ANALOG VIOLIN
AUDIO SYNTHESIZER

by

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Acknowledgements

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Second, my loving parents, Gary and Linda Davis, for their extreme generosity towards me and encouraging words. Thank you for believing in me and being an amazing example. Because of you I am determined to give my best effort in everything I do.

And finally, Jesus Christ, for giving me a purpose that is far greater than electronic design.

Abstract
In the past decade, music electronics have almost completely shifted from analog to digital technology. Digital keyboards and effects provide more sound capabilities than their analog predecessors, while also reducing size and cost. However, many musicians still prefer analog instruments due to the perception that they produce superior sound quality. Many musicians spend extra money and accommodate the extra space required for analog technologies instead of digital.

Furthermore, audio synthesizers are commonly controlled with the standard piano keyboard interface. Many musicians can perform sufficiently on a keyboard, but requiring a specific skill set limits the size of the market for a product. Also, when reproducing instruments such as a violin, a keyboard will not suffice in simulating a controllable vibrato from a fretless fingerboard. There is a need for an interface that allows the user to successfully reproduce the sound of the desired instrument. The violin is just one example of instruments that cannot be completely reproduced on a keyboard. For example, cellos, trombones and slide guitars all have features that a keyboard cannot simulate in real time.

The Analog Violin Synthesizer uses oscillators and analog technology to reproduce the sound of a violin. The user controls the synthesizer with a continuous touch sensor, representing the fretless violin fingerboard. The continuous interface allows for a violin sound played as a standard note, or a warmer sound with adjustable vibrato, based on how the user moves his or her hand. This product provides an innovation and next step to the use of analog technology in sound synthesis. However, as digital technology continues to improve, this product could potentially cross over into digital, with the continued use of the touch interface. Currently, there are products that utilize touch input, however they are often used for sound effects, and atmospheric sounds. Rarely are they used to allow for the digital playability of a synthesized acoustic instrument.

I. Introduction

The Analog Violin Synthesizer simulates an acoustic violin waveform complete with adjustable vibrato. The customer needs a synthesizer with high quality audio output. The customer also needs to play the instrument with ease. An acoustic violin has a steep learning curve and the synthesizer provides an easy alternative. The touch input makes sound generation simple, because the user does not need to have prior knowledge of a violin fingerboard or a piano keyboard. The design allows for a wide range of customers. Musicians with sufficient musical talents can add the synthesizer to their collection and an aspiring musician can also use the synthesizer without facing the frustration of learning a new instrument.

2. Background

The Analog Violin Audio Synthesizer makes use of subtractive synthesis to produce the violin waveform. The complex waveform of a violin includes many harmonics of different amplitudes. The spectra changes not only over time as the bow runs along the string, but changes also occur as the instrument plays in different registers. Figures 1 – 3 display the complexity of the violin waveforms.

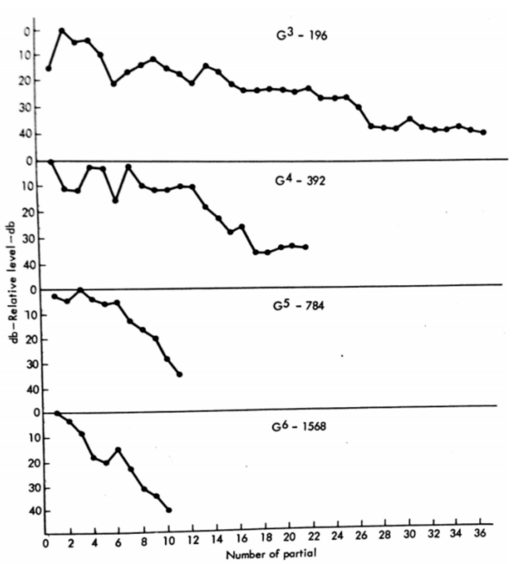

FIGURE 1
SPECTRA OF VIOLIN TONES [14]

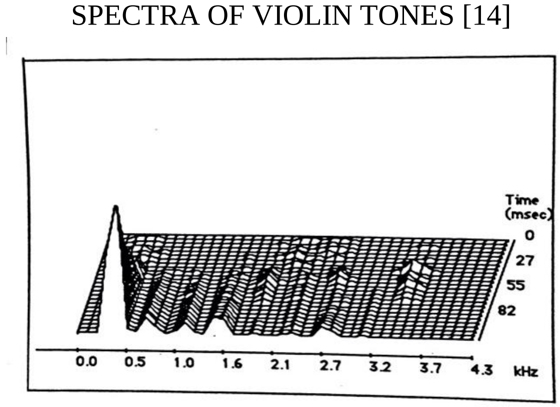

FIGURE 2
START OF VIOLIN SOUND (FFT) [14]

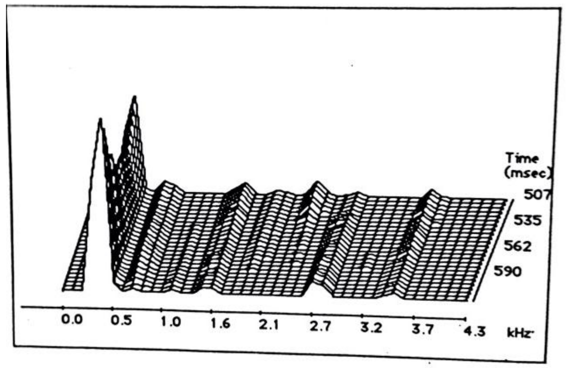

FIGURE 3
SUSTAINING PORTION OF VIOLIN SOUND (FFT) [14]

Because of this complex sound, simulating each harmonic individually would require over 30 oscillators each with their own specified parameters that change over time. Subtractive synthesis involves the use of signals that already contain harmonic content. Then the unused harmonics can be filtered or "subtracted" out. This requires less technology and therefore less money and time. Of course, subtractive synthesis does not account for every sound characteristic. Howard C. Massey writes in *A Synthesist's Guide to Acoustic Instruments*:

> "Subtractive synthesizers—particularly analog systems—are renowned for their ability to emulate the warmth and richness of string sounds with uncanny accuracy. This is largely due to their ability to modulate the width of pulse waves periodically with an LFO. In addition, the many random distortions that are sometimes generated by analog circuitry can seem to reflect anomalies of the acoustic string sound which include wood resonances, air resonances, and the complex sound radiation patterns that occur because of the irregular shape of the typical stringed instrument's body.[14]"

Later sections of this report describe how I used subtractive synthesis in the simulation of this complex sound.

3. Requirements and Specifications

Based upon the customer needs, the following marketing requirements, given in Table 1, define the project outcomes that must be achieved. Requirements 1 and 2 meet the customer need pertaining to the output audio signal. The customer needs a high quality violin sound with vibrato. Requirements 3 and 4 describe the installation and playability of the instrument. The customer wants to play the synthesizer without extensive technical knowledge and musical training. Adding a touch interface will make the synthesizer easier to play because the user does not need to know the fingerboard and strings of a violin. Furthermore, they do not need piano training to play either. Requirement 6 explains the frequency range that the user will be able to play, this information is important for a consumer to know before purchasing.

Also listed in Table 1, the engineering specifications support each of these important marketing requirements. To further simulate the violin sound, and make the instrument truly analog, the touch input is a continuous interface. This means that the synthesizer can play all notes including frequencies

between the notes. This is necessary to simulate the effect of a fretless violin and the vibrato effect, simulated in real time. A standard piano keyboard divides the frequencies into notes, excluding the intermediate between frequencies. With a continuous interface, the user can add vibrato and control the vibrato depth and speed. Other engineering specifications explain the dimensions and ranges for each aspect of the project.

To meet all requirements and specifications by the end of the academic year, the project follows the deliverables listed in Table 2.

TABLE 1
ANALOG VIOLIN SYNTHESIZER REQUIREMENTS AND SPECIFICATIONS

Marketing Requirements	Engineering Specifications	Justification
1	1. Up to sixteen oscillators produce a violin's harmonic frequencies for each note played monophonically.	An instrument produces a note containing 16 different sine wave frequencies. These frequencies inhabit the natural harmonic series [3], [4].
1, 6	2. Produces notes (fundamental frequencies) between 220Hz and 880Hz.	These frequencies represent the fundamental frequencies of the two most commonly played octaves on an acoustic violin.
2, 3, 6	3. Touch sensors receive the x-position touch input from the continuous interface to specify frequencies.	A violin player produces vibrato by moving his or her finger up and down the fretless fingerboard. The continuous interface allows for note frequencies as well as the "in-between" frequencies for a real-time vibrato.
4	4. The system should connect to a ¼" output jack with maximum voltage output of $2V_{peak\text{-}to\text{-}peak}$.	Almost all speaker inputs connect to instruments with a ¼" instrument cable. Therefore the users will have no difficulty producing sound with the synthesizer.
4	5. The system runs off 120V AC (wall). The system uses an AC/DC converter.	Plugs into a standard wall outlet for easy and versatile installation and play. An AC/DC converter can be purchased online or at an electronics store.
5	5. The dimensions should not exceed 24" x 12" x 6"	Fits on a typical keyboard stand. Also fits on a pedal board or equipment rack.
4,5	6. A removable casing encloses the circuitry and components.	Resembles modern keyboards. Improves the visual simplicity.

Marketing Requirements
1. The system should have excellent sound quality.
2. The system should replicate the vibrato effect.
3. The system should be easy to play.
4. The system should easily connect to power and speakers.
5. The system should be visually pleasing to musicians.
6. The system should play two octaves of notes.

The requirements and specifications table format derives from [12], Chapter 3.

TABLE 2
ANALOG VIOLIN SYNTHESIZER DELIVERABLES

Delivery Date	Deliverable Description
March 10, 2014	EE 461 Report
March 12, 2014	EE 461 Demo Device
May 28, 2014	EE 462 Report
May 30, 2014	EE 462 Demo
June 4, 2014	ABET Senior Project Analysis
June 5, 2014	Senior Project Expo Poster

4. Design Approach Alternatives

To synthesize a violin, many synthesis techniques can be employed. The violin can be synthesized using additive, FM, or digital phase distortion synthesis.

Additive synthesis uses basic sine waves to define every partial present in the sound. This technique produces an extremely complex sound that can be manipulated at virtually any frequency. Additive synthesis benefits in that you can define every parameter, without dealing with unwanted sounds. On the downside, defining every single parameter, especially with analog circuitry would require numerous oscillators, all with different parameters. If even one of these oscillators are out of tolerance, the sound quality will be forfeited. Digital synthesis can make use of additive techniques because the output frequencies are nearly exact, and specifying each parameter is done in digital design rather than with hardware.

FM synthesis uses a carrier and modulator signal to create sidebands representing the harmonic series. The ratio of carrier frequency to modulator frequency determine the frequencies of the sidebands. However, FM designs lack qualitative timbral shifting effects such as pulse-width modulation [14]. FM synthesis is best used for the synthesis of percussive instruments (vibraphones, xylophones, etc.) due to the physical construction of these instruments. The oscillation of a violin string, combined with the body resonances, create a waveform in a different method than FM techniques suggest.

Phase distortion synthesis also uses a carrier and modulator signal, similar to FM synthesis. This method of synthesis adjusts the phase angle of a cosine signal to obtain new waveforms. This method can only be accomplished using digital techniques. FM synthesis can be done with analog circuitry, though many digital systems (ex. DX-7) have been developed with extensive capabilities.

Subtractive synthesis best accomplishes the violin sound emulation and analog circuitry can easily be implemented to complete the task. The other design approaches require digital design, or extensive circuitry, outside of the specifications for this project.

5. Project Design

5.1 Level 0 Functional Decomposition

Figure 4 and Table 3 describes the Level 0 functionality of the synthesizer. Besides power, the user controls all inputs with his or her hand. The inputs affect the amplitude and frequency of the audio output signal. The LED outputs were added at the end of the design as an extra feature. There are 3 LED's that light up based on the user's finger position on the touch sensor.

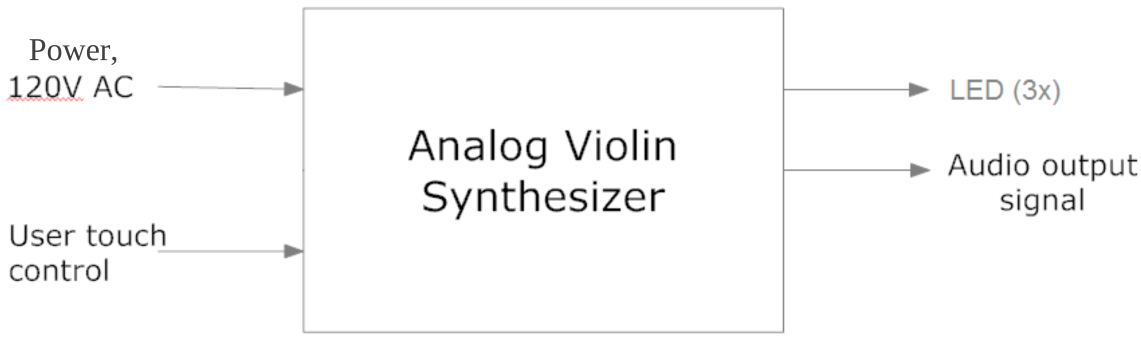

FIGURE 4
ANALOG VIOLIN SYNTHESIZER LEVEL 0 BLOCK DIAGRAM

TABLE 3
LEVEL 0 FUNCTIONAL REQUIREMENTS

Module	Analog Violin Synthesizer
Inputs	- Power: 120 V AC rms, 60 Hz. - User touch control: human finger on touch interface.
Outputs	- Audio output signal: ranges two octaves of notes. Variable peak value. - 3 Red LED's
Functionality	The synthesizer plays the output frequency according to the user touch input. The output waveform matches the sound wave produced by an acoustic violin. The 3 LED's are placed along the touch screen and light up according to the finger position.

All Functional Requirements Tables Derive from [12]

5.2 Level 1 Functional Decomposition

Figure 5 displays the Level 1 functional decomposition of the synthesizer. The system contains seven main components: power supply, touch sensor, oscillators, filter, mixer, amplifier, and LED control. The system contains a mixing, filtering, and amplifying stage due to the unique timbre of the violin. To add further complexity to the sound, the envelope generators control the parameters to change over time.

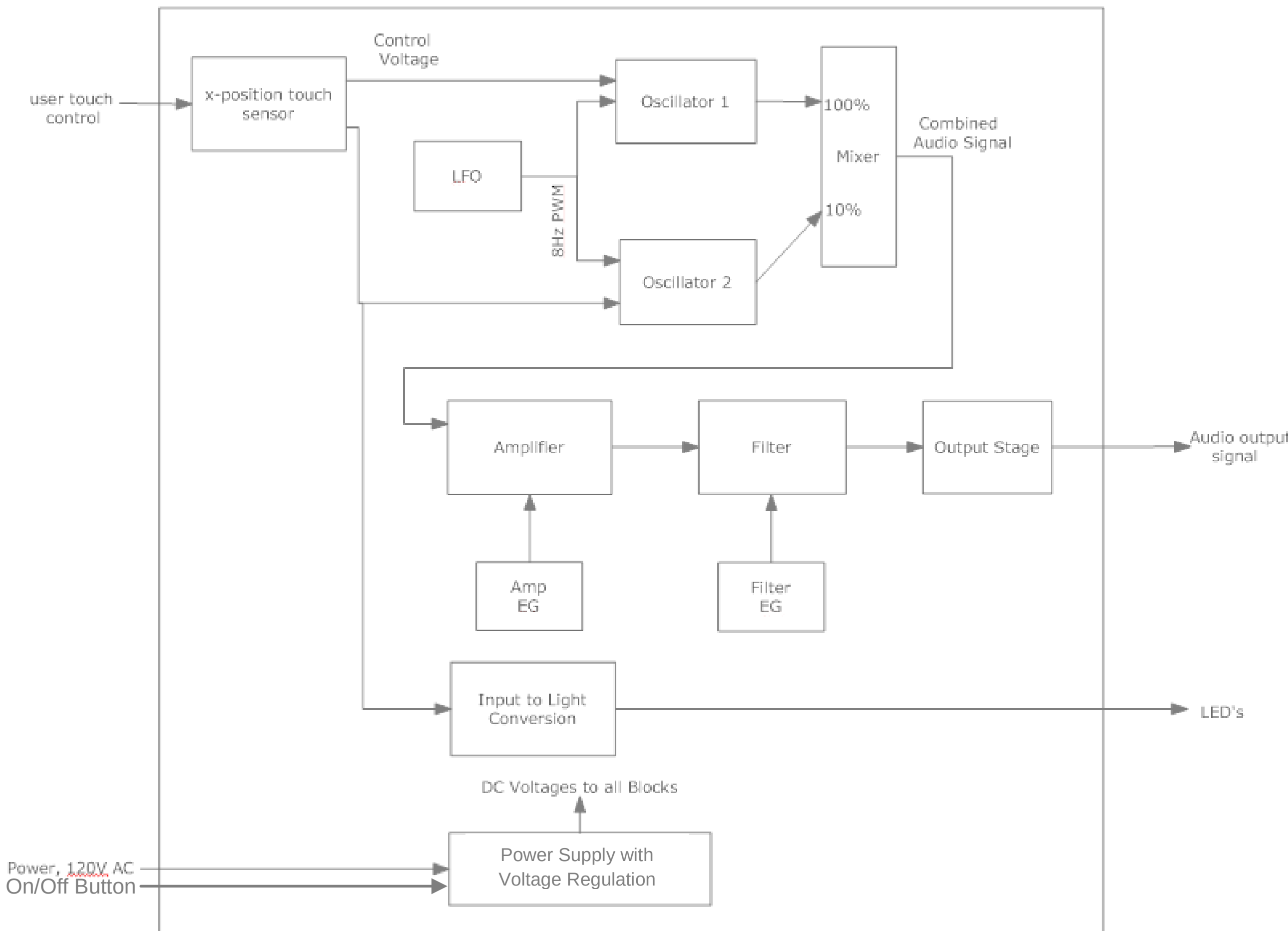

FIGURE 5
ANALOG VIOLIN AUDIO SYNTHESIZER LEVEL 1 BLOCK DIAGRAM

5.3 Level 2 Functional Decomposition

NOTE: The following subsections contain design schematics and theoretical output values. Actual results and oscilloscope captures can be found in Section 7; "Integrated System Tests and Results."

5.3.1 Touch Sensor

Table 4 displays the functional requirements for the touch sensor.

TABLE 4
TOUCH SENSOR FUNCTIONAL REQUIREMENTS

Module	X-Position Touch Sensor
Inputs	- User touch input, variable upon x-axis input - Power: 5V DC
Outputs	- Reference voltages to set frequency of oscillation: 0 to 5V DC
Functionality	Read the touch input and specify the fundamental frequency of oscillation with an output reference voltage

The touch input implements a 4-wire resistive touch sensor. These sensors have two transparent layers each coated with a conductive material. With a voltage applied across one layer, the second layer creates a voltage divider when contact is made. As shown in Figure 6, these sensors read in both X and Y directions, with a digital controller rapidly switching between the layers.

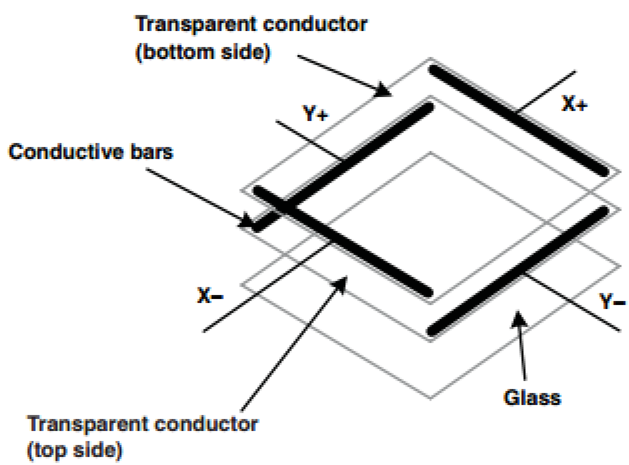

FIGURE 6
4-WIRE RESISTIVE TOUCH SENSOR DIAGRAM [16]

The synthesizer requires the use of only the x-position touch input. With this specification, the sensor can be read with analog circuitry, and doesn't need a controller to switch between the two layers. Figure 7 describes how to read the voltage divider output for the x-position.

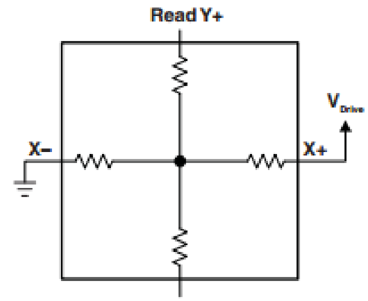

FIGURE 7
READING X-POSITION TOUCH ON THE 4-WIRE TOUCH SENSOR [16]

Only 3 terminal are required to read the x-position input. With a voltage applied across the two x-terminals, the y+ terminal outputs the voltage based on the resistive divider produced by the users hand. The position can be calculated with the following equation.

$$x = \frac{V_{Y+}}{V_{Drive}} \times width_{screen}$$

[16]

This means that the output at y+ will swing from 0V ground to the rail voltage of 5V used in this design.

I used the VS101TP-A1touch sensor made by VS Display Technology in China. This product is 9 ¼” X 5 ½”[15]. The product is marketed by it’s diagonal dimension as a “10-inch touch 4-wire touch sensor.” The product datasheet specifies that the y-terminals measure the long edge and the x-terminals measure the short edge. Therefore, to make use of as much space as possible, the sensor must be positioned reading the long edge as x. This means that the rail voltage is placed across y+ and y-, and the voltage division is read from x+. See Figure 8 for the pin configuration used in this design.

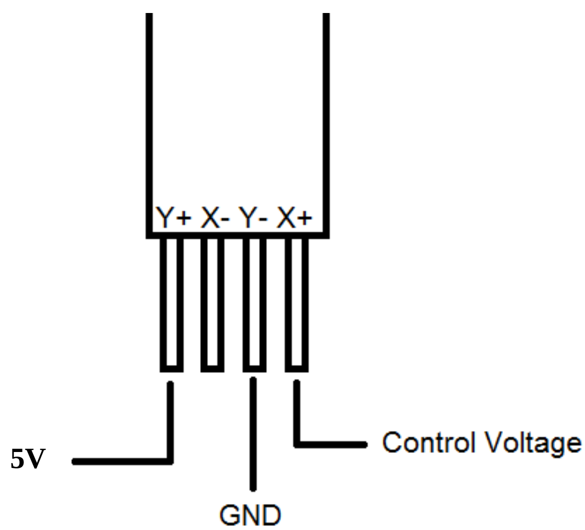

FIGURE 8
4-WIRE TOUCH SENSOR PIN CONFIGURATION

5.3.2 Oscillator 1 and Oscillator 2

Because the touch controller provides a voltage output, the system requires voltage controlled oscillators. At this point in the design, a decision had to be made about what waveform would best replicate the violin spectrum. In *A Synthesist's Guide to Acoustic Instruments*, Howard C. Massey's suggests using two oscillators with 20% duty cycle square waves. Each of these also contains a slight pulse-width modulation (PWM) provided by an 8Hz sine wave [14].

Note that many of the parameters chosen in each of the blocks in this synthesis design follow Massey's notes on violin synthesis.

With two oscillators present, the output sound has an added complexity with a slight "beating" between the sounds. Also with pulse-width modulation, the overtones are periodically altered. The presence of overtones is dependent upon the pulse width of the sound. For example, a 1/5 duty cycle square wave (20%) contains all harmonics, except the 5^{th} harmonic and its multiples. With a slight pulse width modulation, the harmonic content changes. Table 5 defines the functional requirements of the oscillators. The following circuitry was design to match these specifications.

TABLE 5
OSCILLATOR STAGE FUNCTIONAL REQUIREMENTS

Module	Oscillators
Inputs	- Power: 5V DC - Input Voltage: 0 to 5V DC (From Touch Screen)
Outputs	- 20% duty cycle square wave with PWM
Functionality	Generate a 20% duty cycle square wave with frequency dependent upon the touch control input.

Initial Design Idea: LTC6992 PWM TimerBlox

Initially the oscillators were designed using the LTC6992 TimerBlox Voltage-Controlled Pulse-Width Modulator. This was built and tested, providing a stable output wave with low noise. Figure 9 shows the setup configuration for an oscillator with pulse width modulation. However, this design makes use of a frequency setting resistor. This means that the fundamental frequency of oscillation is determined by a resistor value, not a voltage. With the touch sensor providing a voltage output, converting this data to a relevant resistance value would only be possible with the use of excessive circuitry (Digital Potentiometer, DAC, ADC, Clock generator). Therefore this circuit is not practical to meet the requirements for oscillators 1 and 2.

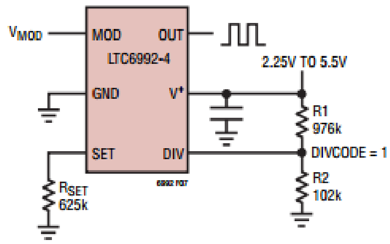

FIGURE 9
SQUARE WAVE OSCILLATOR WITH PWM USING LTC 6992 [17]

Design Used: Sawtooth Wave Oscillator + Comparator

The final oscillator design achieves a PWM square wave using a sawtooth wave oscillator connected to a single-threshold non-inverting comparator. When the sawtooth wave crosses above the reference voltage, the comparator outputs a high signal, and when the wave is below the reference voltage, the output drops low. Therefore the reference voltage of the comparator determines the pulse width of the output signal (See Figure 10). PWM is achieved when the reference (or "level") voltage varies periodically, thus modulating the width of the output pulse.

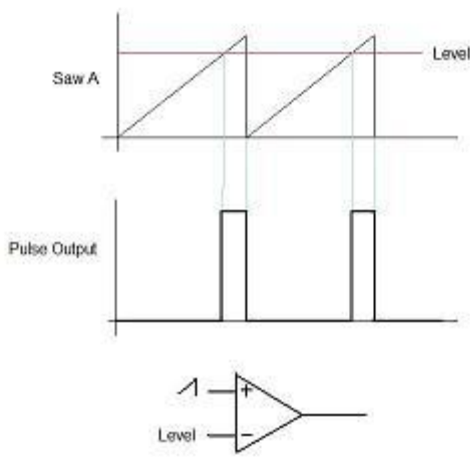

FIGURE 10
USING A COMPARATOR AND SAWTOOTH WAVE TO SET PULSE WIDTH [18]

To design this system, I used a 555 timer current-controlled oscillator, with a current mirror to source current into the timer. The schematic in Figure 11 shows the design used.

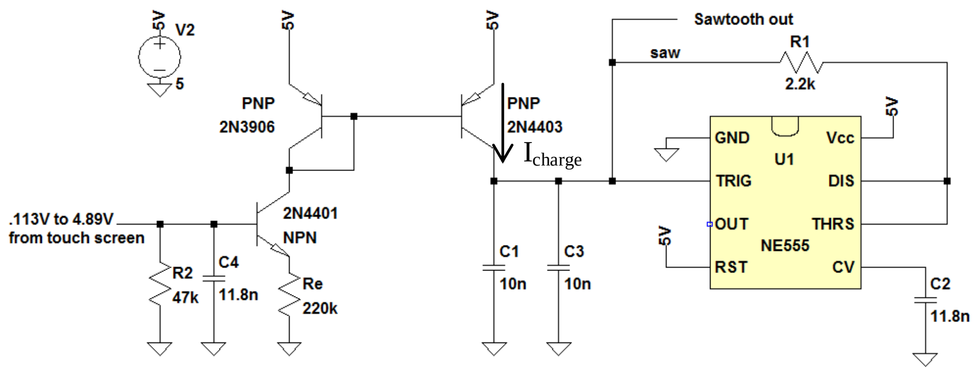

FIGURE 11
555 TIMER-BASED VCO

The 555 timer with resistor and capacitors behaves as a current-controlled oscillator. The current leaving through the collector of the 2N4403 PNP determines the frequency of oscillation by the following equation.

$$f_{osc} \approx \frac{3}{(C1 + C3)V_{cc}} I_{charge}$$

The NPN and two PNP's are present to convert the voltage input from the touch screen into $I_{charge.}$ As soon as the input voltage rises high enough to turn on the NPN, current flows through the NPN as well as each of the PNP's, pumping current into the oscillator.

Oscillators 1 and 2 are designed to match in fundamental frequency as close as possible. To provide this match, the following frequency setting components were chosen accordingly in Table 6, with names referring to the schematic in Figure 11. These component values provided a frequency difference within 10Hz when calibrated to 440Hz.

TABLE 6
MEASURED COMPONENT VALUES

Component	Oscillator 1	Oscillator 2
R_e	221.09kΩ	220.89kΩ
C_1	9.308nF	9.26nF
C_3	10.415nF	10.563
C_1+C_3	19.723F	19.823nF

To produce an output pulse wave with 20% duty cycle, the LM339 Quad OP Comparator IC was used, configured for two non-inverting single-threshold comparators. Figure 12 shows the circuit diagram.

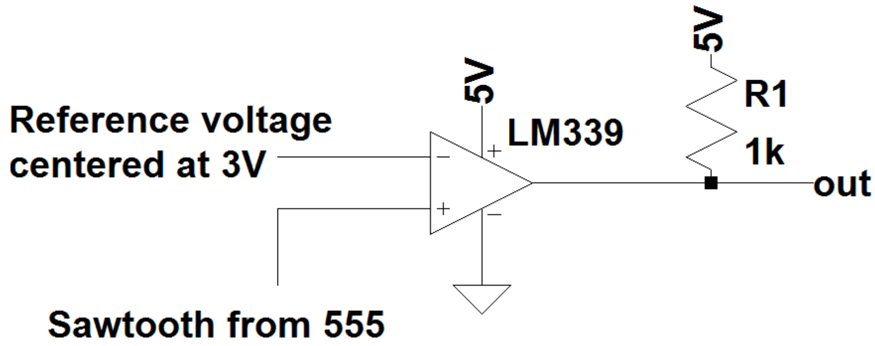

FIGURE 12
COMPARATOR STAGE SCHEMATIC

Equations to determine reference voltage:

$1.667V \leq V_{saw} \leq 3.333V$

$V_{peak\text{-}peak} = 3.333\text{-}1.667 = 1.667V$

20% of 1.667V = 0.333V

$V_{ref} = 3V$

Therefore, the PWM input will contain a DC offset at 3V.

5.3.3 Low Frequency Oscillator (LFO)

Many sine wave oscillators can provide a low frequency sine wave. However most oscillators provide a sine wave that swung from rail to rail. Because the PWM requirement is for a small amplitude signal, buffering such a large signal down to a small value without filtering out the low frequency fundamental was difficult. The Wein-Bridge and Phase-Shift Oscillator were investigated, yet neither of these designs provided a very reliable signal for the system. Table 7 displays the functional requirements of the low frequency oscillator.

TABLE 7
LFO FUNCTIONAL REQUIREMENTS

Module	Low Frequency Oscillator
Inputs	Power: 5V DC
Outputs	Sine, triangle, or similar waveform - Frequency: 8Hz - DC Offset: 3V - Small amplitude: less than $30mV_{pp}$
Functionality	Apply PWM to the square wave signal.

Scott R. Gravenhorst uploaded an LFO design that uses the 555 timer and a low-pass filter [21]. Rather than using an op-amp circuit that produces a sine wave, Gravenhorst makes use of the 555 timer outputting a low frequency square wave, then applies a low-pass filter to achieve a sine-like signal. My design adapted Gravenhorst's idea and adds a buffer to set the peak-to peak voltage and DC offset to the correct values. See Figure 13 for the circuit diagram.

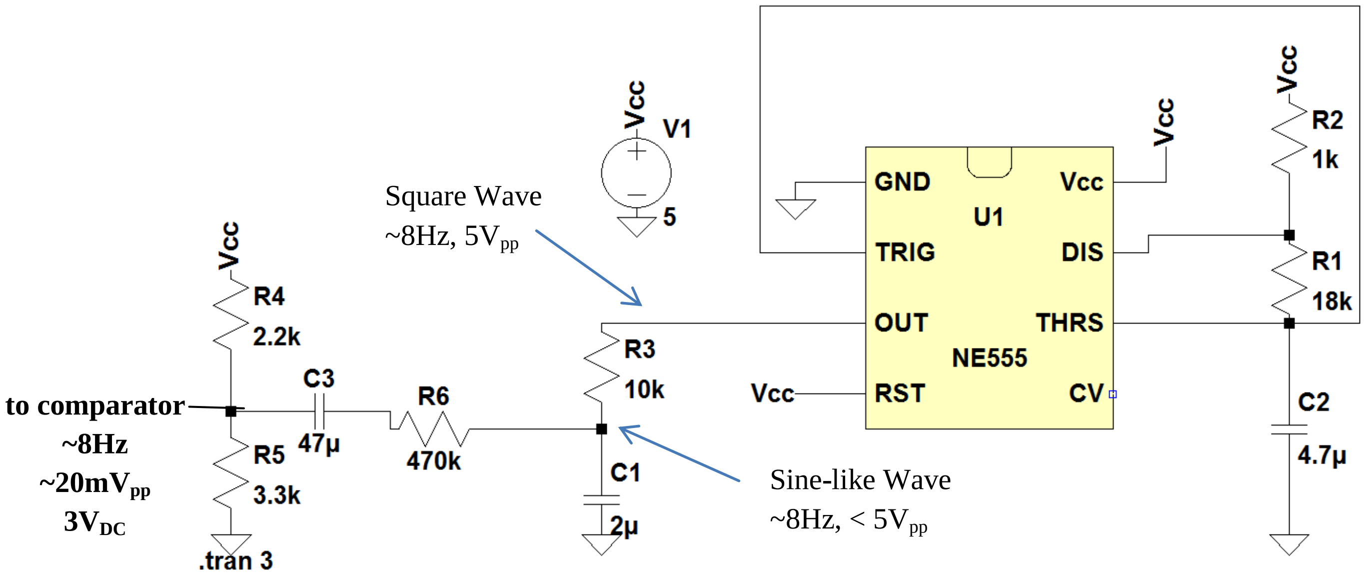

FIGURE 13
LFO + BUFFER SCHEMATIC

In the circuit of Figure 13, components R2, R1, and C2 set the fundamental frequency of oscillation, producing a square wave at the OUT terminal of the 555-timer. The frequency is determined by the following equation.

$$T_{LFO} = 0.693(R2 + 2 \times R1)\ C2 = 0.120s$$
$$f_{LFO} = 1/T_{LFO} = 8.3Hz$$

Components R3 and C1 create a low pass filter with cutoff frequency f_c = 7.957Hz, just below the desired fundamental frequency of oscillation. The filter will cut out most of the harmonics, leaving a single sine-like waveform at the fundamental frequency.

The remaining components R4, R5, R6, and C3 buffer the signal to a DC voltage of 3V and amplitude of less than $20mV_{PP}$.

5.3.4 Mixer

The mixer was designed as a simple summing amplifier. In Massey's book, he recommends sending both oscillators through the mixer, yet Oscillator 2 should only be about 10 to 20 percent of the amplitude of oscillator 1. This way the beating effect will only be slightly noticeable. If they are both at full volume, you will begin to hear more of an ensemble timbre. Since this system focuses on the solo violin sound, oscillator 2 is brought in at a lower volume.

Figure 14 shows the circuit diagram for the mixer.

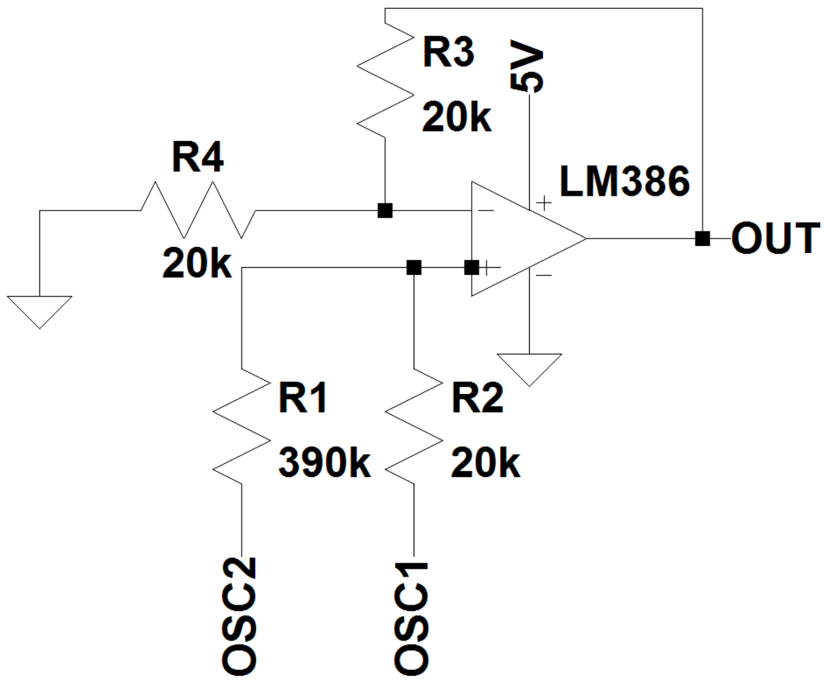

FIGURE 14
MIXER SCHEMATIC

$$V_{out} = \left(1 + \frac{R_3}{R_4}\right) \times (V_{OSC1} \times \frac{R_1}{R_1 + R_2} + V_{OSC2} \times \frac{R_2}{R_1 + R_2})$$

Resistors R1 and R2 determine the individual gain of each input as shown in the above equation. This circuit meets all functional requirements listen in Table 8.

TABLE 8
MIXER FUNCTIONAL REQUIREMENTS

Module	Mixer
Inputs	- Power: 5V DC - Two pulse-width modulated Square Waves (0-5V_{PP}), up to 1.4kHz
Outputs	- One combine output wave o Oscillator 1 at full volume o Oscillator 2 at 10 to 20% of Oscillator 1 amplitude o Frequency varying from below 100Hz to at least880Hz
Functionality	Adds both signals from Oscillators 1 and 2 into a single output

5.3.5 Envelope Generator Circuits

An envelope generator is used in synthesis to add complexity to the sound. An envelope is responsible for changing the sound over time. In a violin for example, the sound does not stay the same over time, but has a rich complex attack sound as the bow strikes the violin, then a more smooth sustaining sound. An envelope can be applied to many different synthesizer modules (Oscillators, Filters, Amplifiers) to adjust the parameters of the blocks. The most common envelope in synthesis is the ADSR (Attack, Decay, Sustain, Release) Envelope. The four stages of this envelope can each be adjusted separately to provide a sound that is most natural for common acoustic instruments. The diagram in Figure 15 shows each stage of the ADSR Filter.

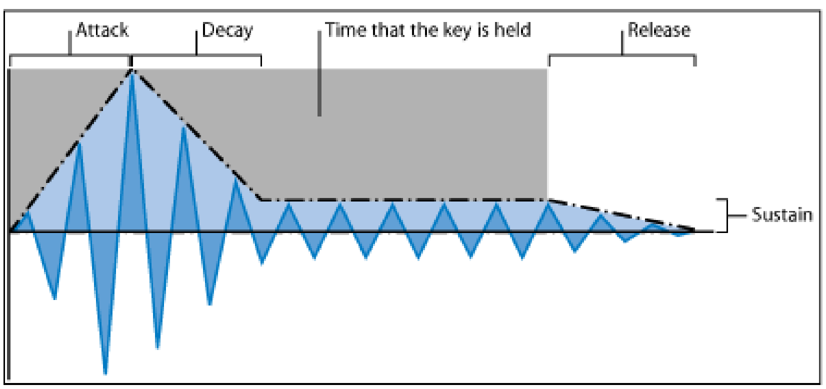

FIGURE 15
THE ADSR ENVELOPE [22]

The makers of Logic Express (Apple Inc.) characterize each stage of the ADSR envelope with the following definitions:

> *Attack:* Controls the time it takes for the initial slide from an amplitude of zero to 100% (full amplitude).
> *Decay:* Determines the time taken for the subsequent fall from 100% amplitude to the designated sustain level.
> *Sustain:* Sets the steady amplitude level produced when a key is held down.
> *Release:* Sets the time it takes for the sound to decay from the sustain level to an amplitude of zero when the key is released [22].

Envelope generators can easily be designed in digital synthesis, using delays and interrupts. However, creating a timed parameter change in analog design requires careful component selection that will change

a voltage signal in ADSR stages. The model used for the analog ADSR envelope generator was adapted from a design given by Rene Schmitz in 1999. This design again uses the 555-timer due to its simple time-adjusting capabilities. The original circuit design by Schmitz is shown in Figure 16.

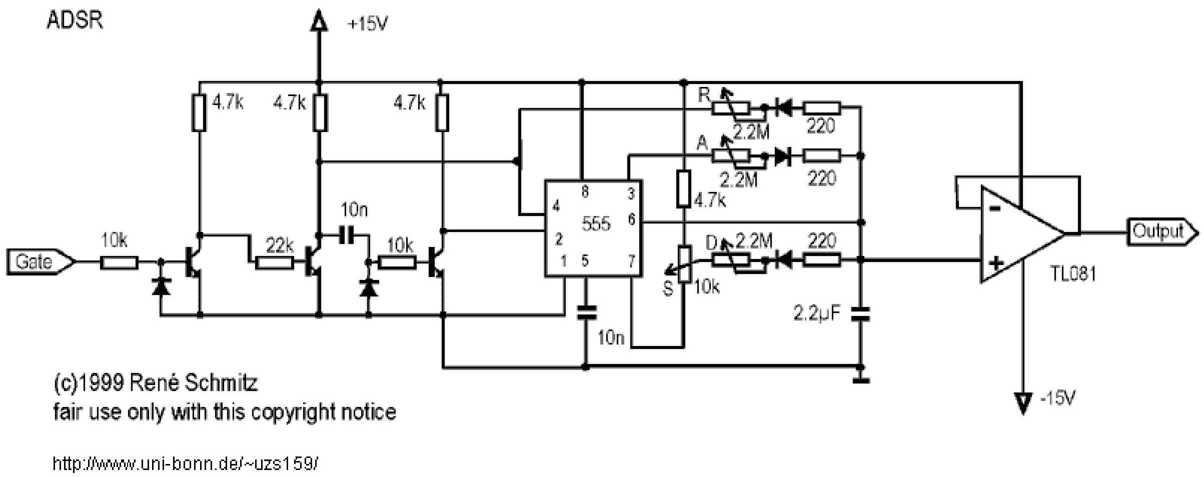

FIGURE 16
ORIGINAL ENVELOPE GENERATOR BY RENE SCHMITZ [23]

In this design, the three NPN transistors, configured as common emitter amplifiers, turn the input into a gate that swings from rail-to-rail as soon as the input rises above a diode drop of 0.7V. At the first CE amplifier, the output goes from high to low, but also inverts, so the second amplifier inverts the gate a second time. The third CE amplifier with capacitor turns the gate into a simple trigger that drops low every time the gate turns on. This signal is sent to the "TRIGGER" input of the 555 timer. The 555 timer set-up resembles the configuration for a monostable multivibrator (or one-shot). However the ADSR diodes and resistors direct the signal to control each side of the pulse. For example, the "A" branch is connected to the "OUTPUT" pin (pin 3). The attack time is determined by adjusting the potentiometer, thus changing the pulse width of the one-shot. Similarly, the release time is controlled by adjusting the decay time from the amplified gate input falling edge. The diodes ensure that only the rising or falling edge is adjusted. The decay and sustain parameters are controlled together. The sustain potentiometer acts as a voltage divider from the Vcc voltage, determining what level the decay leg will fall to.

In this design, an envelope will control both the cutoff frequency of the low-pass filter stage and the gain of the amplifier stage. Table 9 lists the functional requirements for both generators.

TABLE 9
ENVELOPE GENERATOR FUNCTIONAL REQUIREMENTS

Module	Envelope Generators
Inputs	- Power: 12V DC - Voltage gate in, from touch controller
Outputs	- Voltage with adjustable ADSR stages
Functionality	As soon as the touch screen is pressed and provides a voltage about ~0.7mV, the envelope generator outputs a timed voltage that changes in 4 stages to adjust either the low-pass filter or the amplifier.

5.3.5.1 Amplifier Envelope Generator

The envelope applied to the amplifier contains all four ADSR stages. A solo violin player can adjust each of these stages according to his or her bow control. Therefore certain design choices were made to simulate a violinist playing at full volume with no crescendos or decrescendos. Table 10 organizes the choices made for each section.

TABLE 10
ADSR DESIGN CHOICES FOR AMPLIFIER

Stage	Acoustic Solo Violin	Design choice
Attack	Fast or slow crescendo, dependent upon violinist	Less than 0.5 seconds, simulating an aggressive playing style
Decay	Fast, but noticeable, about half a second	0.5 seconds
Sustain	Dependent upon dynamics of piece	High, just below attack maximum. Allows for full sound
Release	Depends on violinist. Can be immediate or have a decrescendo	Fast, for immediate bow release, but allows the sound to resonate through the body of the violin. Release time of less than 0.5 seconds.

Figures 17 and 18 show the schematic for the design as well as the simulated envelope. Notice the stages match the design choices from Table 10.

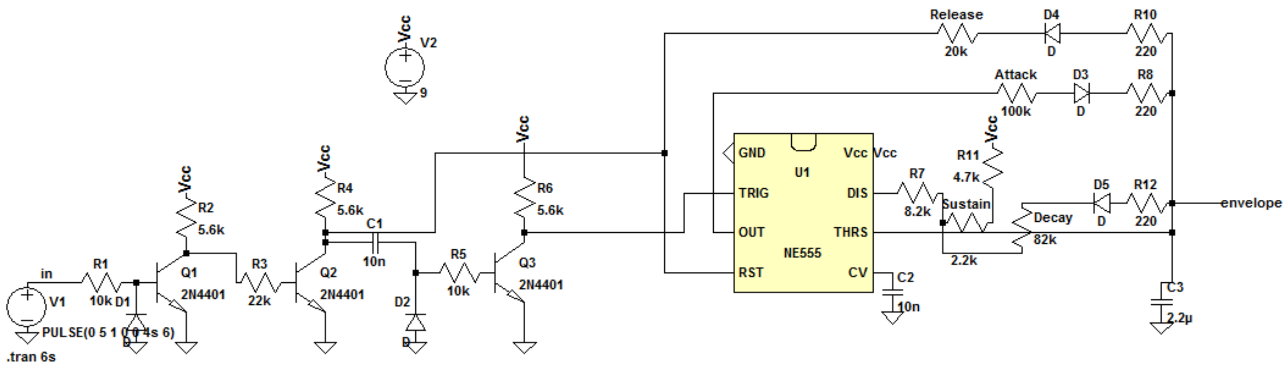

FIGURE 17
AMPLIFIER ADSR ENVELOPE SCHEMATIC

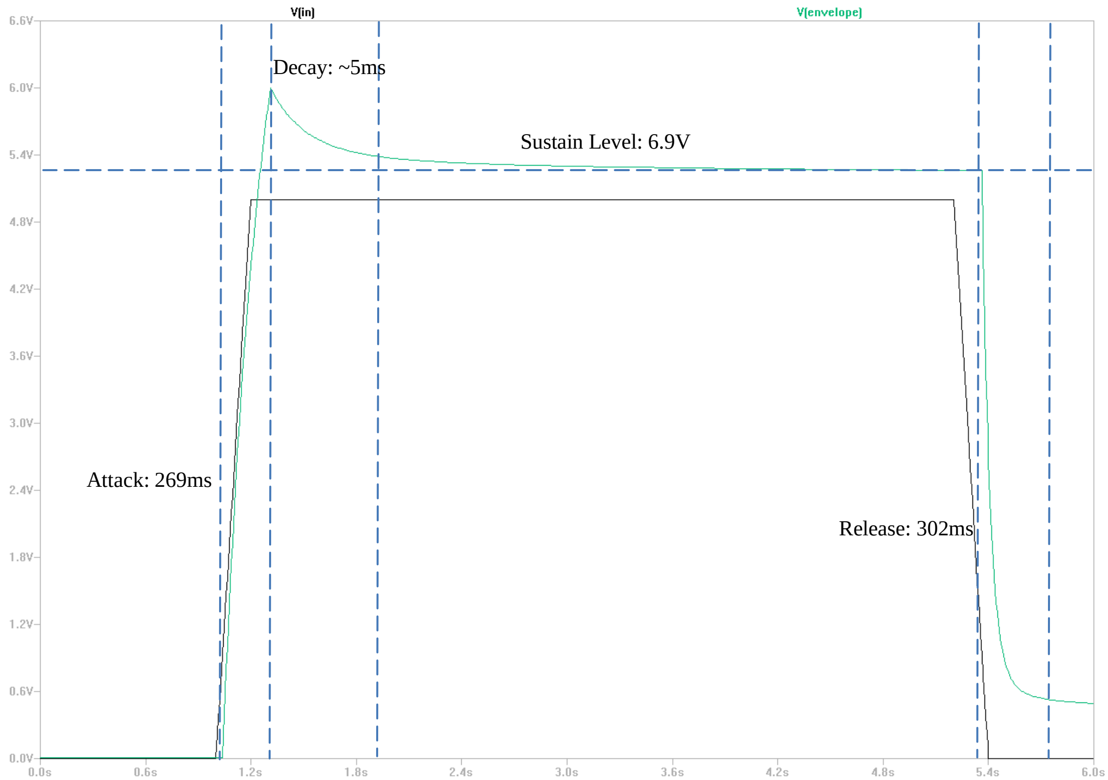

FIGURE 18
AMPLIFIER ADSR ENVELOPE SIMULATION

Also, note that the circuit design allows for both legato and stacatto playing. The attack and decay stages only play when the note is initially pressed and the gate goes from low to high. If a note is changed with no release, the envelope remains in the sustain stage. This effect simulates a note change without changing the bow direction.

5.3.5.2 Low-Pass Filter Envelope Generator

The most dramatic change of harmonic contact in a violin sound exists between the initial bowing section when the bow strikes the string and the sustaining portion when the bow is smoothly drawing sound from the instrument. The harsh attack of the bow striking the string contains higher partials to grasp the sound of an attack and add definition to the start of the note. The harmonic content undergoes very little noticeable change during the sustain or release sections of the sound. Therefore only the attack and decay stages are needed to produce this quality of sound. Once the decay stage is over, the envelope output parameter of zero will correspond to a nominal cutoff frequency of the lowpass filter, rather than 0Hz. This will be explain in section 5.3.7.

The attack and decay times chosen simply match the attack and decay stages of the amplifier. Figures 19 and 20 show the circuit diagram for the AD filter as well as the simulation. The idea of only using two stages was sugested by Massey in his design for synthesizing an acoustic violin [14].

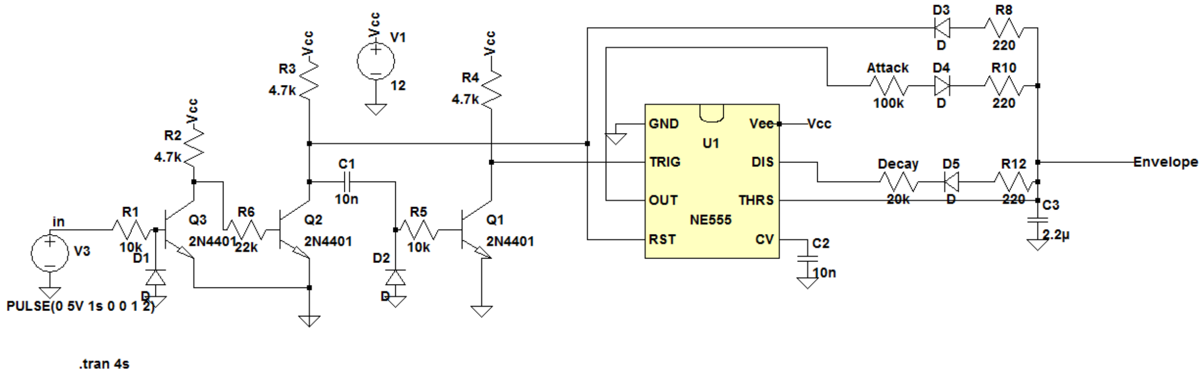

FIGURE 19

LOW-PASS FILTER AD ENVELOPE GENERATOR SCHEMATIC

To create only two stages of the envelope, the sustain leg was completely removed, allowing the decay stage to drop all the way low. The resistor controlling the release time was removed, though it's value is negligable because with no sustain, there is no level to release from.

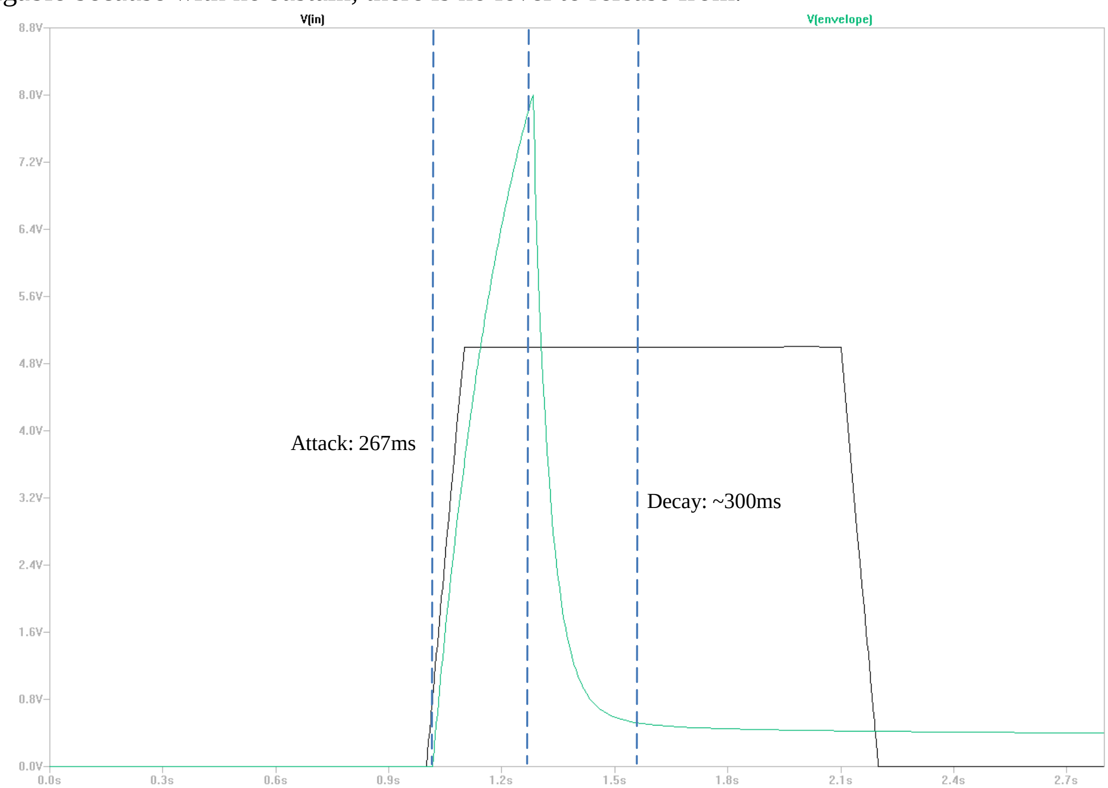

FIGURE 20

LOW-PASS FILTER AD ENVELOPE GENERATOR SIMULATION

In the simulation, the decay time falls faster in the filter than on the amlifier. This parameter will work for this design, because the 300ms corresponds to the steepest portion of the amplifier decay, before the curve begins to smooth out. There is little noticeable difference during the smoothing out portion. If the decay time of the filter were extended, then the sharp change in harmonic content will become inaudible.

5.3.5.3 Envelope Buffer Stages

Before sending the two envelopes to the amplifier and low-pass filter, an emiter-follower buffers the envelope outputs for a low output resistance. The schematic in Figure 21 displays the circuit diagram of the unity gain buffer. Rene Schmitz uses a unity gain op-amp as an output buffer, however the design change depended upon availability of NPN transistors verses op-amps.

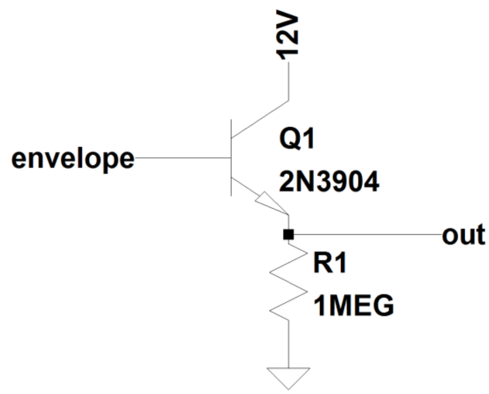

FIGURE 21
ENVELOPE OUTPUT BUFFER SCHEMATIC

5.3.6 Voltage-Controlled Amplifier (VCA)

Because an ADSR voltage envelope controls the sound amplitude, the amplifying stage must be designed as a voltage-controlled amplifier. A voltage controlled amplifier contains two inputs, an AC audio signal and a DC conrol voltage. The control voltage directly controls the AC gain of the amplifier. Table 11 lists the functional requirements of the voltage-controlled amplifier.

TABLE 11
VOLTAGE-CONROLLED AMPLIFIER FUNCTIONAL REQUIREMENTS

Module	Voltage-Controlled Amplifier

Inputs	- Power: +/- 12V DC
	- AC audio input
	    ○ ~ $4V_{pp}$
	    ○ Up to 880Hz
	    ○ ~$2.5V_{DC}$ offset
	- DC control voltage
	    ○ 0 to $9V_{DC}$

Outputs	- Audio output signal: Amplified as high as $8V_{PP}$ For more than enough amplitude in audio, $8V_{PP}$ is a generous maximum. ○ Goes at least as high as 880Hz
Functionality	Amplifies audio signal for ¼” output. Changes gain based on envelope control input.

To achieve a voltage controlled gain, synthesists recommend the use of the LM13700 “Transconctuance Amplifiers with Linearizing Diodes and Buffers” IC, made by Texas Instruments Inc. [24][25]. The chip

contains two current-controlled transconductance amplifiers, with differential inputs and darlington buffer outputs[25]. Both amplifiers on the chip were used for both the VCA and Voltage-Controlled Filter. Figure 22 shows the schematic and pinout of the IC.

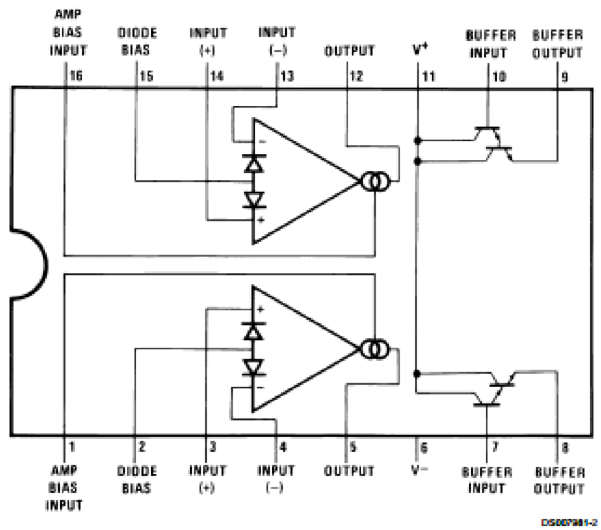

FIGURE 22
LM13700 PINOUT [25]

The transconductance amplifier takes in a differential voltage and converts it into a current output. It also includes a curent bias input to control the amplifiers transconductance. The output of the amplifier is a result of the difference in input voltage between the inverting and noninverting inputs, multiplied by the transconductance, which is controlled by the bias input curent [26]. The darlington pair buffers the output for a low output resistance, and converts the current output to a voltage when tied to a load resistor.

The LM13700 datasheet provides a VCA circuit design, which was used for this design. Figure 23 shows the circuit. The VCA of the synthesizer uses the same component values.

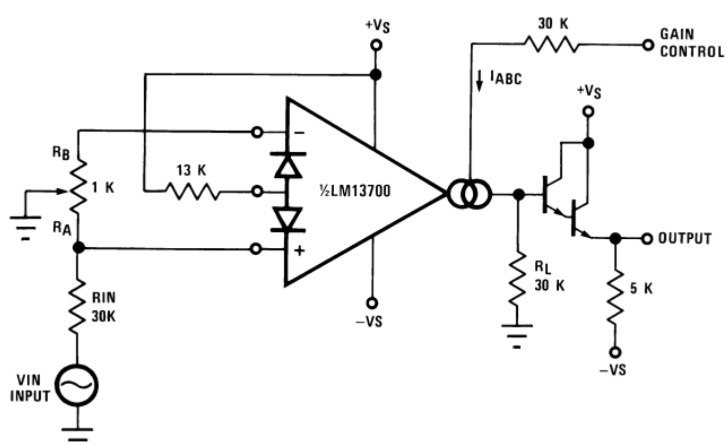

FIGURE 23
VOLTAGE CONTROLLED AMPLIFIER SCHEMATIC [25]

A rail of +/- 12V supplied +/-Vs to the circuit. The 30KΩ resistor at the gain control input converts the voltage into a bias current I_{ABC}. The 1kΩ potentiometer is set to limit the effect of the control signal at the output, because the changing control voltage can affect the DC value of the output [25]. Holding the potentiometer at halfway best limits this effect.

5.3.7 Voltage-Controlled (Low-Pass) Filter (VCF)

Fortunately, Texas Instruments also provides a useful circuit diagram using the LM13700 as a VCF as well as a VCA. However before designing the filter, the correct cutoff frequency was determined. To determine the cutoff frequency, I first determined the frequencies of each harmonic present in the note A-440 as shown below.

f_1	f_2	f_3	f_4	f_5
440Hz	880Hz	1320Hz	1760Hz	2200Hz

As shown in Figure 1 (Section 2), more partials are present in the lower register, and as the fundamental increases in frequency, the higher partials drop off and are no longer heard. Therefore, for 440Hz, a nominal cutoff frequency of about 3Hz will still include many partials and as the frequency increases to 880Hz and above, the harmonic content will simplify. Upon attack, the envelope will increase the filter to a value determined by the circuit design theory. If a higher frequency were chosen, then the effect of the filter will become unnoticeable. Table 12 lists the functional requirements of the VCF as determined by these design decisions.

TABLE 12
FILTER STAGE FUNCTIONAL REQUIREMENTS

Module	Filter Stage

Inputs	- Power: +/-12V DC
	- AC audio input
	    ○ ~ $4V_{pp}$
	    ○ Up to 880Hz or higher
	    ○ ~$2.5V_{DC}$ offset
	- DC control voltage
	    ○ 0 to $9V_{DC}$

Outputs	- Audio output signal: Filtered with cut-off frequency varying from 3kHz to 5kHz.
	Filters outs harmonic content above 3 to 5kHz, depending on envelope control input. Cutoff frequency changes directly with control voltage.

The VCF also uses a transconductance amplifier provided by the LM13700, so the second amplifier on the IC is used. The LM13700 datasheet provides a useful VCF design schematic. For this system, the design was adapted from the datasheet recommendation. Figure 24 shows the circuit diagram for the VCF.

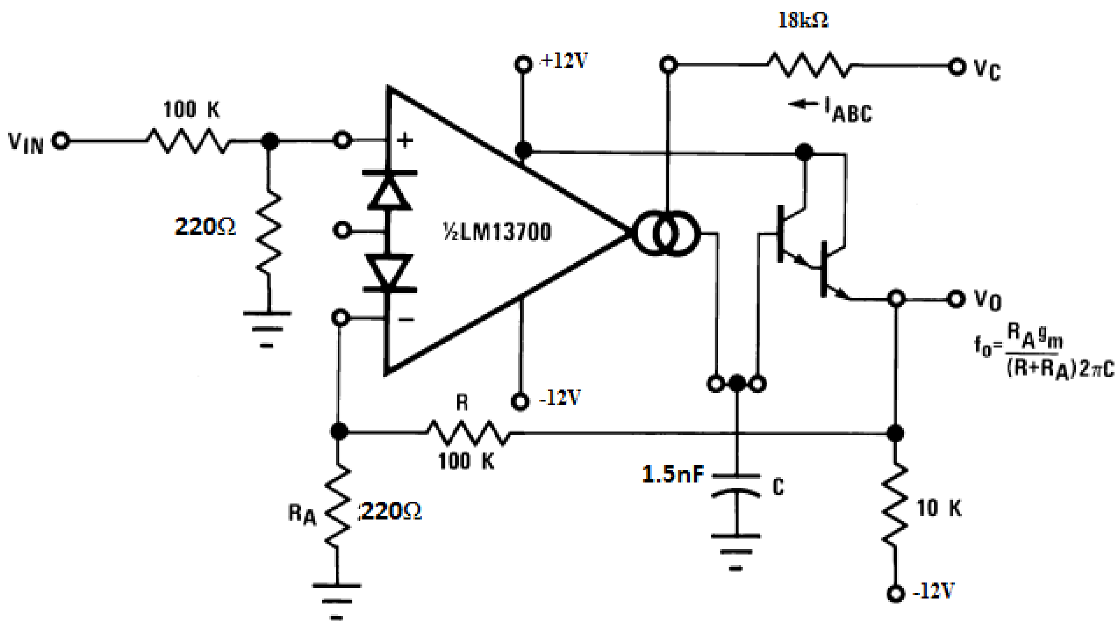

FIGURE 24
VCF SCHEMATIC [25]

The circuit performs the unity-gain buffer amplification at frequencies below cut-off. The cutoff frequency is determined by the point where X_c/g_m equals the closed loop gain (R/R_A) [25]. The resistor and capacitor values were chosen with the following equations.

$$g_m = 19.2 \times I_{ABC}$$

$$g_{m\,min} = 19.2 \times \frac{0V - (-13V)}{18k\Omega} = 0.013866\,S$$

$$g_{m\,max} = 19.2 \times \frac{9V - (-13V)}{18k\Omega} = 0.023477\,S$$

$$f_c = \frac{R_A g_m}{(R_A + R)2\pi C}$$

$$f_{c\,min} = \frac{220\Omega \times 0.013866\,S}{(220\Omega + 100k\Omega)2\pi 1.5nF} = 3.229kHz$$

$$f_{c\,max} = \frac{220\Omega \times 0.023477\,S}{(220\Omega + 100k\Omega)2\pi 1.5nF} = 5.468kHz$$

Therefore, the chosen resistor and capacitor values successfully provide cutoff frequencies from 3kHz to 5kHz.

The VCF was designed as the last block before the audio signal goes to the speaker. As the last module, the filter will remove the high-frequency noise that accumulates as a result of the many different system components.

5.3.8 Output Stage

The system is design for a ¼" audio output cable that can connect to various audio speakers with different characteristics. A simple emitter follower will successfully buffer the output to comply with speakers of various load resistances. Also, the VCF operates on a split rail supply. However speakers may run on only a single supply voltage, so the audio output cannot swing below ground. Therefore the output stage contains a voltage divider to raise the DC value of the circuit and ensure that the signal does not drop to negative voltage. Figure 25 shows the circuit diagram for the output stage.

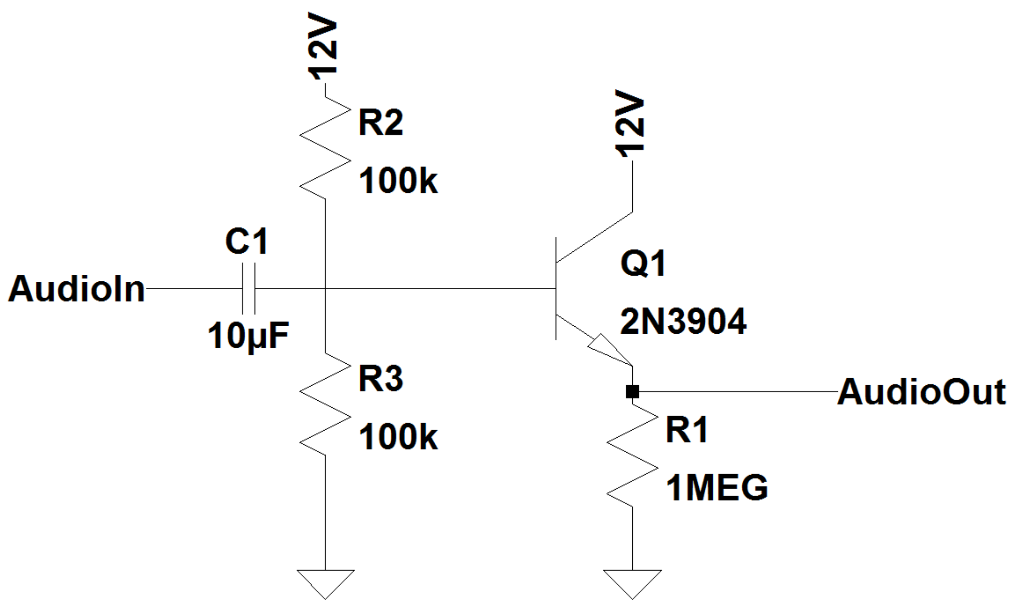

FIGURE 25
OUTPUT STAGE SCHEMATIC

The capactor C1 acts as an AC coupling capacitor to set the new DC voltage of the signal at $6V_{DC}$, set by resistors R2 and R3. With the signal centered at $6V_{DC}$, there is plenty of room for the signal to oscillate without clipping at V_{CC} or ground.

5.3.9 Power Supply

As specified in the marketing requirements, the system must run from the wall power supply. However all of the circuits use DC power. Also, there are three different DC supplies needed: $5V_{DC}$ single supply, $+12V_{DC}$ single supply, and +/- $12V_{DC}$ split supply. Therefore power conversion circuitry is required to provide power to all blocks. Table 13 lists the functional requirements of the power conversion circuit.

TABLE 13
POWER SUPPLY FUNCTIONAL REQUIREMENTS

Module	Power Supply
Inputs	- Power: 120 V AC rms, 60 Hz
Outputs	- DC voltages o +5V DC o +12V DC o +/-12V DC split supply
Functionality	AC to DC converter to provide DC supply to all components (single and split supply)

The power conversion circuit makes use of a step-down transformer and bridge rectifier to lower the voltage level and convert from AC to DC voltage. However, to hold the voltage at the correct DC levels, 5V and 12V voltage regulators were implemented into the circuit. The following part numbers, listed in Table 15, were used in the circuit.

TABLE 14
PARTS LIST FOR POWER SUPPLY

Part Number	Description
CES-67-1241	24 Volt 1A Center-Tapped Transformer
W04G	Single-Phase Bridge Rectifier
LM7805	$+5V_{DC}$ Voltage Regulator
LM7812	+12V Voltage Regulator

Using the listed components, the power-conversion circuit was designed as shown in Figure 26.

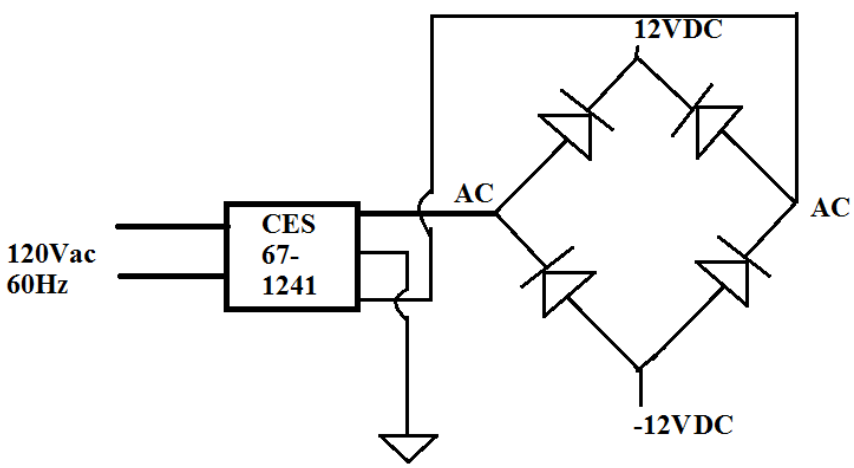

FIGURE 26
AC (WALL) TO +/- $12V_{DC}$ CONVERSION CIRCUIT

The two voltage regulators were applied to the positive rail to hold the voltage at +12V and +5V. No voltage regulator was used on negative rail, because even a small variance in negative rail did not affect

the operation of the VCA and VCF (the only two blocks using negative rail). Bypass capacitors were used with the two voltage regulators as shown in Figures 27. These capacitors remove any AC from the signal to ensure that the voltage stays at a constant DC value. A 220uF bypass capacitor was also used on the negative rail as well, without a voltage regulator present.

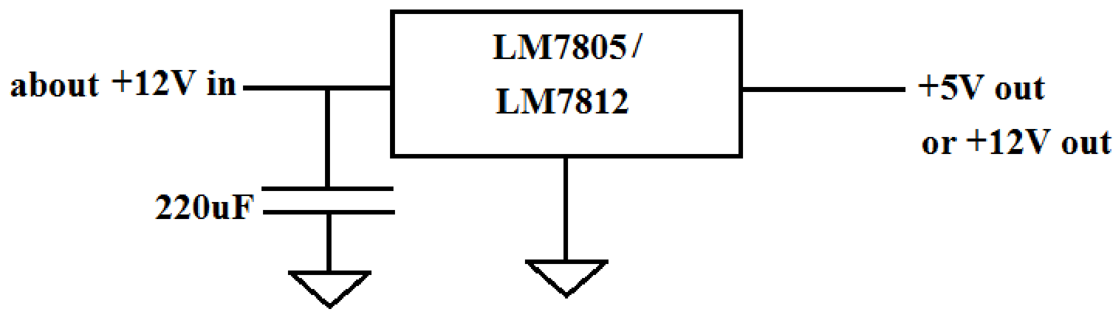

FIGURE 27
VOLTAGE REGULATOR CONFIGURATION FOR POSITIVE RAILS (+5V AND +12V)

6. Physical Construction and Integration

6.1 Breadboard and Proto-board Design

Each block underwent breadboard construction first to ensure correct operation. After determining that each module operated correctly, all modules were connected to each other with each module still on the breadboard. Figure 28 shows the final breadboard configuration of the circuit before converting over to the proto-board.

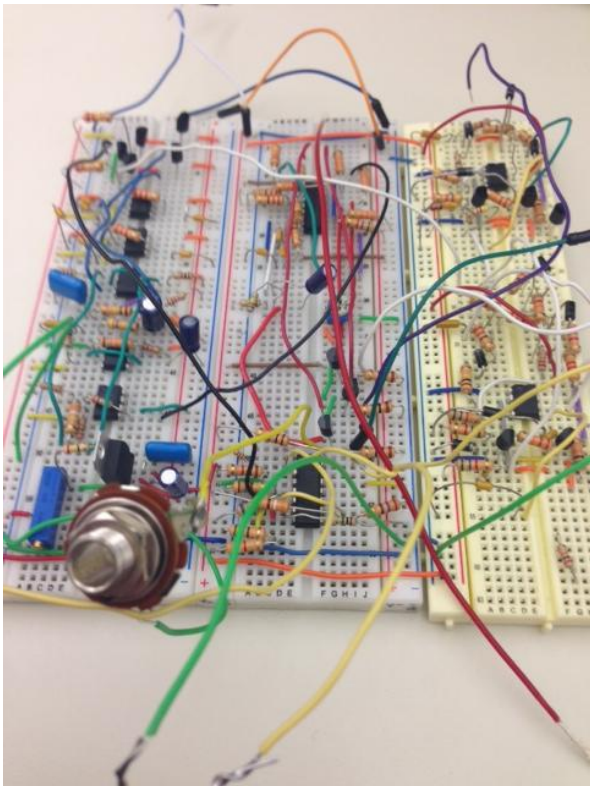

FIGURE 28
BREADBOARD CONFIGURATION

In order to best place the circuitry into an enclosure, each module was placed onto a separate proto-board. This design choice was made to ensure that no errors occurred in the conversion of the circuit from one board to another. Figure 29 shows examples of the proto-boards used for each block. The components were secured onto the proto-board with solder on the pads under the board. Components with adjacent connections were joined with a large solder bead. Wires connected components with nodes that did not lie next to each other. Figure 30 shows the bottom side of a proto-board, with connected pads.

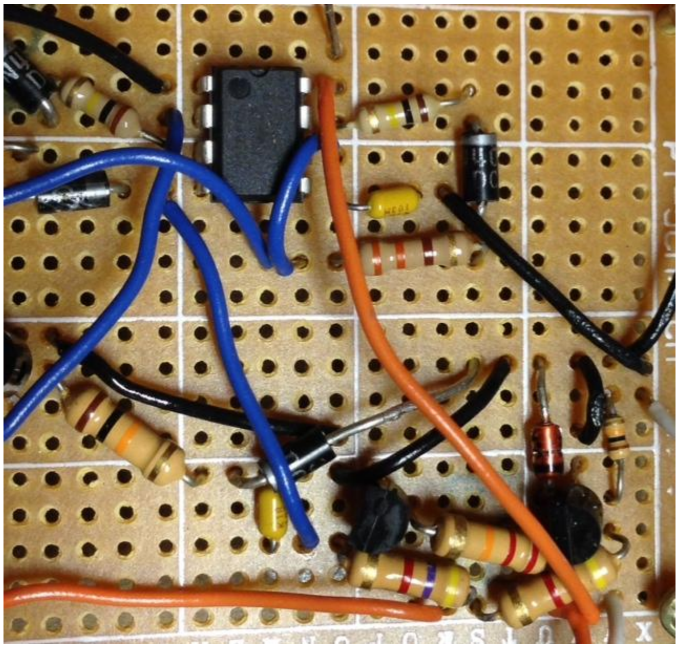

FIGURE 29
PROTOBOARD TOP SIDE

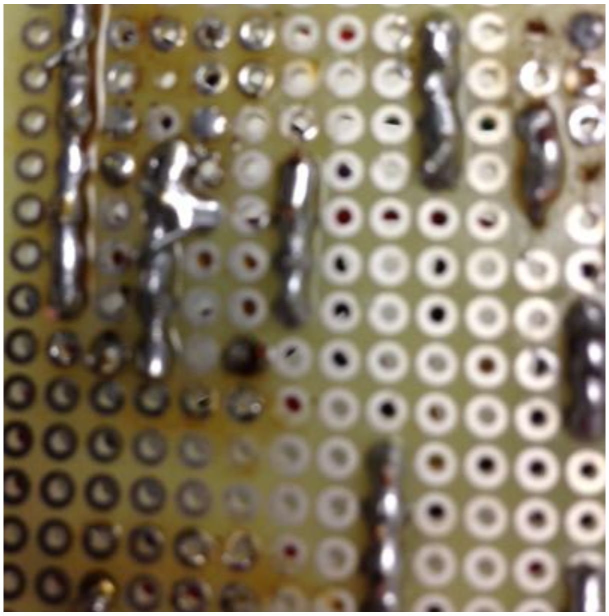

FIGURE 30
PROTOBOARD BOTTOM SIDE WITH SOLDERS

The following components each took up a single proto-board each: ADSR Envelope Generator, AD Envelope Generator, Bridge Rectifier, +12V DC voltage regulator, LED comparators, output stage. Proto-board sizes depended on number of components.
The VCF and VCA both took up a single board, because the LM13700 contains two transconductance amplifiers on a single IC, therefore both the filter and amplifier used one chip, thus requiring only one board.

Because of the many connections between the Oscillators, Comparators, LFO, and Mixer, all 5 of these stages were placed on a single board, along with the 5V regulator. However, proto-board configuration continually introduced new problems to the circuits in this group. After two attempts to solder these components down, the circuits were placed back onto a single breadboard. Breadboard configuration proved to be the most reliable setup for this portion of the system.

Also, breadboard configuration allows for the other blocks to be easily connected by placing leads into pins on the breadboard. If a block ever needed to be removed, the proto-boarded block can simply be pulled out from the breadboard without any desoldering.

6.2 Component Layout

With many different small proto-boards, and a single breadboard, each block was laid out so that each circuit was closest to the block that it connected to. Ground and rail voltages were daisy-chained from block to block, rather than having a central ground, though the ground rail on the breadboard provided many available pins. The photo in Figure 31 shows each section laid out for the shortest wires for signal connections. The photo corresponds to the layout diagram in Figure 32. The layout diagram shows a simplified layout of the various wires connecting the blocks to each other. These connections justify the layout choice. Each block separate from the breadboard represents a circuit soldered on to a proto-board (except for the transformer).

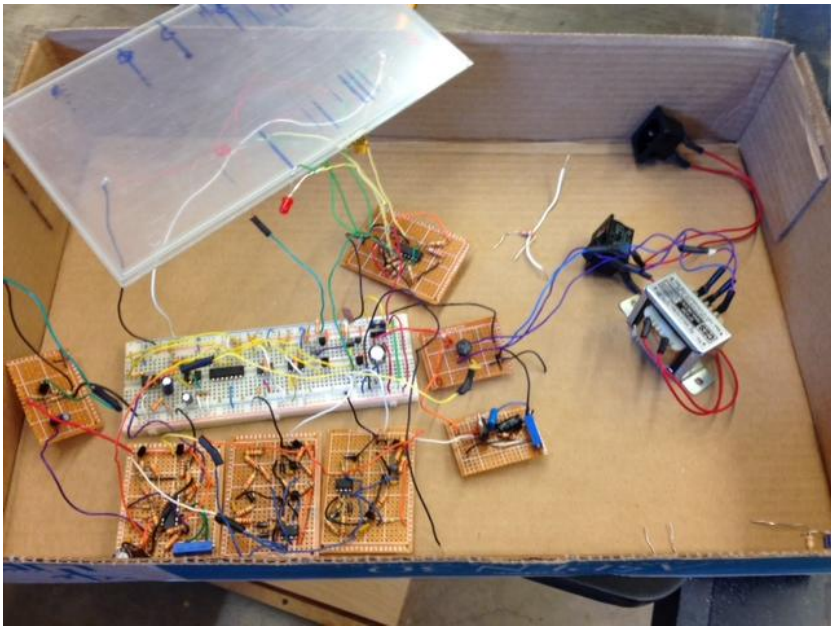

FIGURE 31
COMPONENTS LAYOUT PHOTO

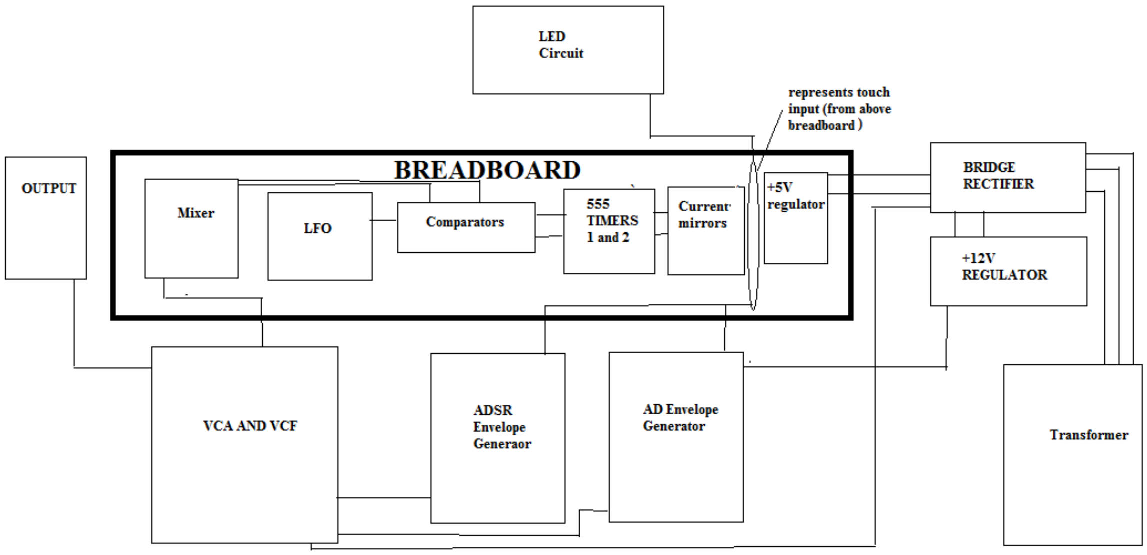

FIGURE 32
COMPONENT LAYOUT DIAGRAM

6.3 Enclosure Design

The enclosure was designed as a simple rectangular box, using recycled wood pieces and screws and nails, found in the EE woodshop. The box dimensions were determined based on the dimensions of the touch screen: 5 ½" x 9 ¼". To provide enough room for a power switch on the top next to the screen, the box was designed with dimensions as shown in Figure 33. The bottom board is a thin piece of wood, with soft texture, useful for screwing down spacers by hand. The side boards are beams with ¾" thickness. The beams were connected to each other and the bottom piece using wood glue.

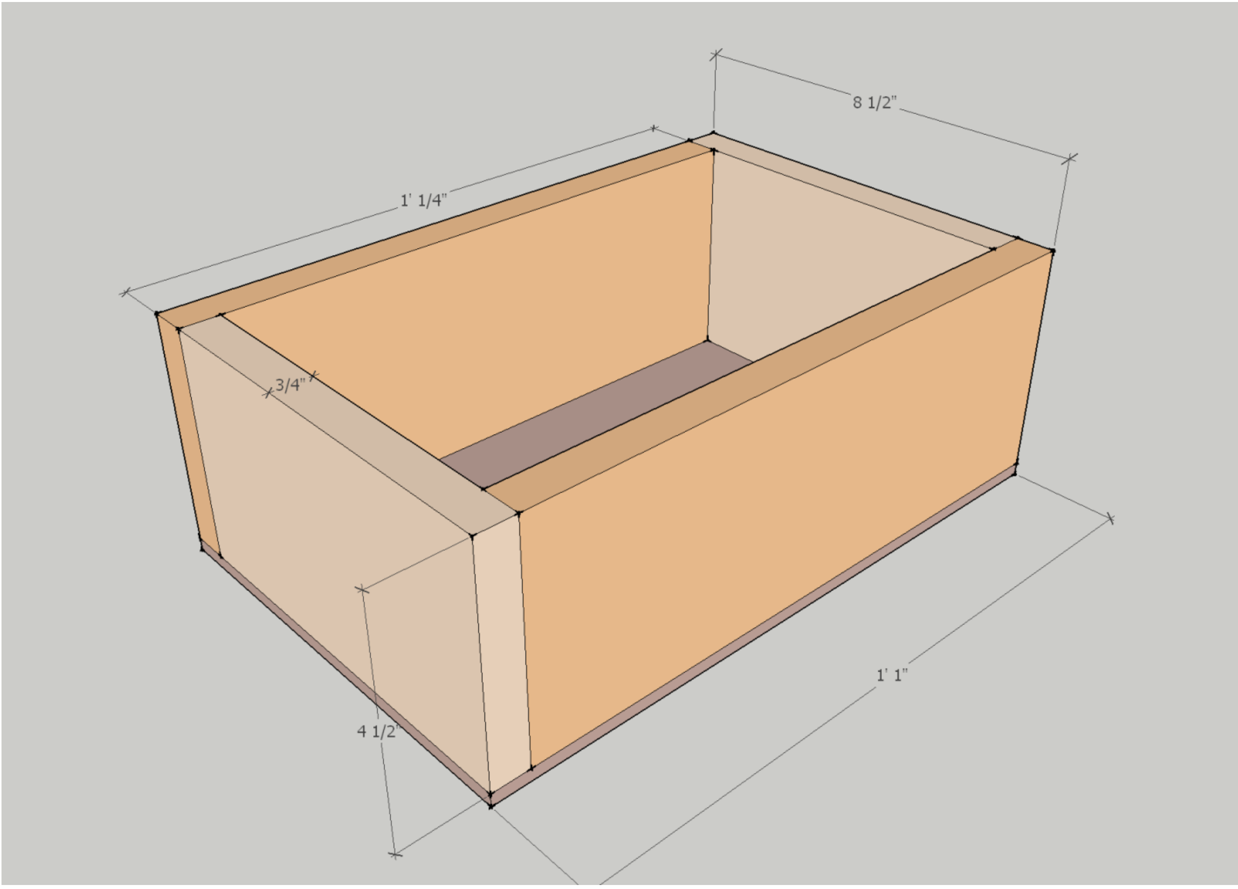

FIGURE 33
BOX DESIGN

Holes were drilled in the side beams for the wall plug and ¼" audio jack. Figures 34 and 35 show photos of these two ports.

FIGURE 34
QUARTER-INCH AUDIO JACK

FIGURE 35
POWER SUPPLY INPUT PORT

The breadboard was placed inside the box and fastened to the bottom using adhesive. The proto-boards were fastened to the bottom and sides with small screws and spacers. Figures 36 and 37 show photos of the circuitry mounted into the box.

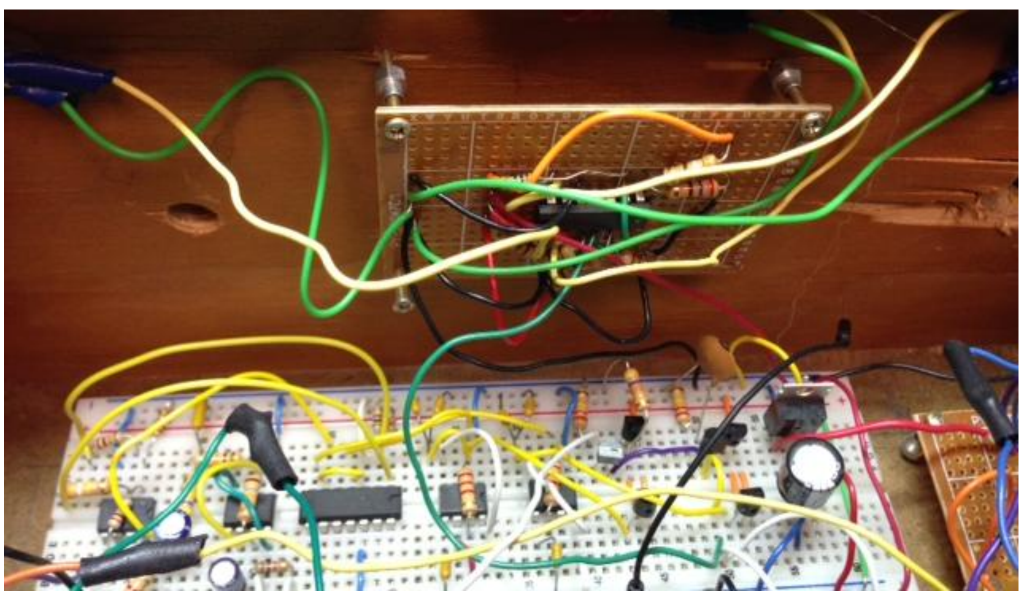

FIGURE 36
BREADBOARD AND PROTOBOARD MOUNTING

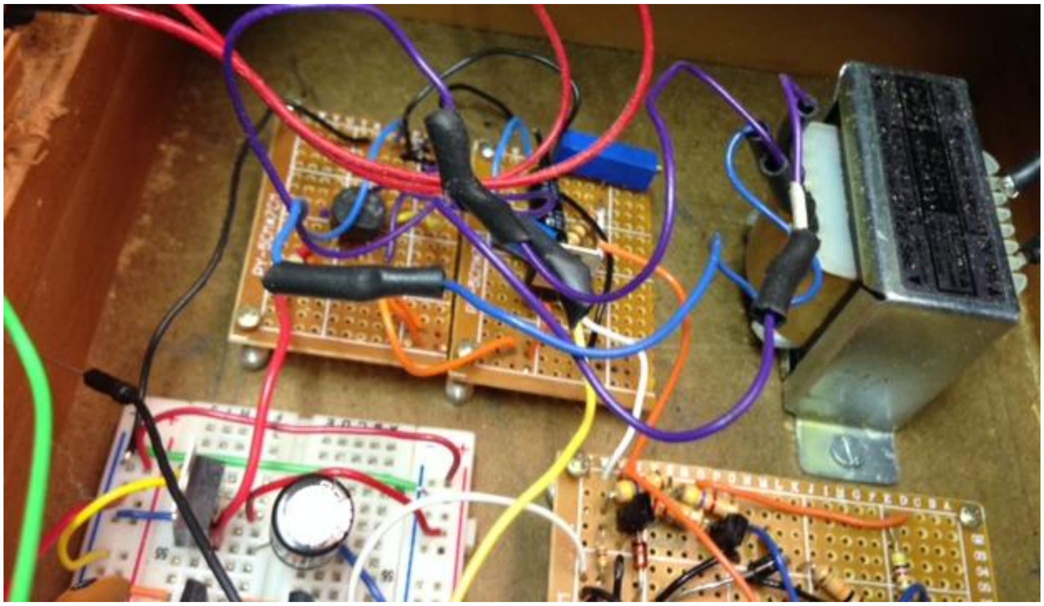

FIGURE 37
PROTO-BOARD AND TRANSFORMER MOUNTING

The top panel was constructed with the same wood as the bottom panel. Holes were cut out for the power switch and the touch screen. The hole for the touch screen is slightly smaller than the screen, so that the screen can sit on the panel, without falling into the box. The photo in Figure 38 shows the box with the top panel and touch screen mounted to the top of the box.

FIGURE 38
COMPLETED BOX WITH PARTS MOUNTED

7. Integrated System Tests and Results

To confirm that each block individually met the functional requirements, the circuit outputs were measured with the Agilent 34401A Digital Multimeter (DMM) and the Agilent MSO-X-3014A Oscilloscope (Scope). The measurement tools and cursors on the oscilloscope were used to measure frequencies, amplitudes, and other signal characteristics.

7.1 Resistive Touch Sensor Test Results

The 4-wire resistive touch sensor was tested by measuring voltage from the X+ terminal using the DMM. Figure 39 displays the test setup for this device.

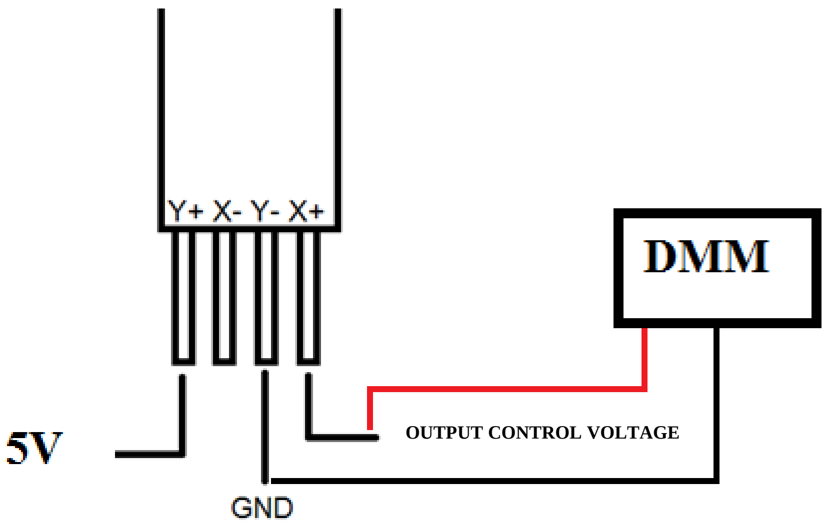

FIGURE 39
TOUCH SENSOR TEST SETUP

Upon testing the touch sensor with a 5V rail, the output swung from **0.113V to 4.89V** with the user input from furthest left to furthest right position.

$$0.113V \leq V_{OUT} \leq 4.89V$$

7.2 Oscillator Test Results

The 555 timer sawtooth wave oscillator was tested to ensure that the oscillation frequency matched the theoretical equation:

$$f_{osc} \approx \frac{3}{(C1 + C3)V_{cc}} I_{charge}$$

Upon testing the circuit, the following results were obtained, shown in Table 15.

TABLE 15
OSCILLATOR MEASUREMENTS (OSCILLATOR 1)

	Input Voltage (V)	I_{220k} (mA)	F_{osc} (Hz)
At Minimum Oscillation Frequency	0.5V	0.18uA	≈ 22Hz
At Maximum Oscillation Frequency	4.89V	19.4uA	≈ 1.4kHz

Note that if $I_{220k} = I_{charge}$, then these measurements do not follow the equation given for the frequency of oscillation. The charge current measured should provide a lower frequency of oscillation. However the two PNPs are not matched transistors. As given in the datasheets for both the 2N3906 and 2N4403, the two transistors have different current characteristics. The 2N4403 has higher current characteristics for the base-emitter voltage. Figure 40 compares the two characteristics. This means that I_{charge} is greater than I_{220k} in the µA range, causing a higher frequency of oscillation than expected. Even with a difference as small as single microamp, frequency will change dramatically. For example, the difference between 22Hz and 1.4kHz is only dependent on about a 19µA change in charge current.

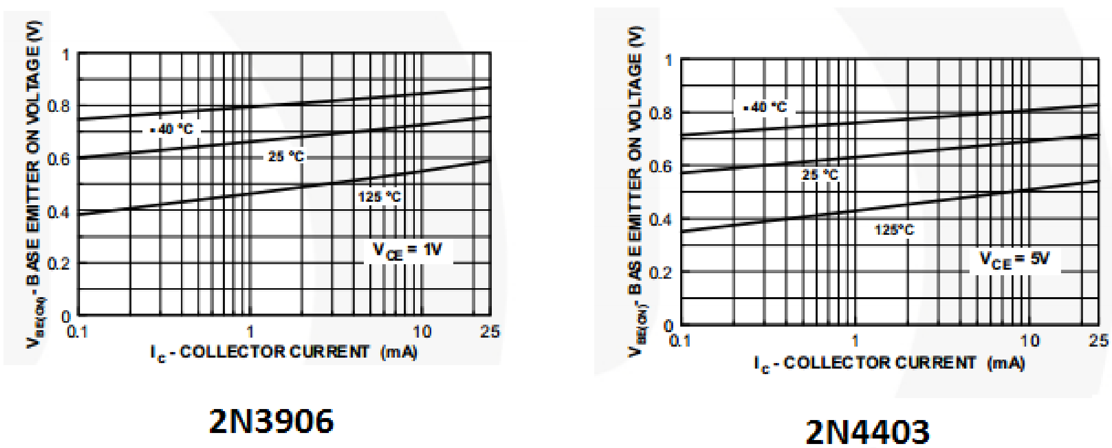

FIGURE 40
FORWARD ACTIVE CHARACTERISTICS FOR EACH PNP [19] [20]

According to the design specifications, the oscillator must at least reach 880Hz. This design goes above to 1.4kHz, almost a whole octave above requirements. This is an acceptable range, but if the oscillator went any higher, it would become too hard to control the touch screen to the desired frequency. One small change in position would cause a large jump in frequency.

Next, the oscillators were tested for frequency matching characteristics. Ideally, these oscillators should only differ by a 5-10Hz in the 440Hz register. As mentioned in Section 5, the circuit was calibrated with components that provided this small frequency difference. However, over time the component values change due to operating temperatures and other added resistance and capacitance. The task of more closely matching these two frequencies still needs to be mastered.

Each oscillator signal was viewed on the oscilloscope and measured individually. At times the frequency difference spreads as far as 150Hz, even in the lower register. The problem includes more than just different capacitor and resistor values. The transistors and 555 timer IC's all have minor differences that add up to an unpredictable outcome.

Figures 41 and 42 display the oscilloscope readings for each of the oscillators with a 3V touch input. Here, the oscillators are out of calibration after about a week since initial component selection. The frequencies differ by over 70Hz. The sawtooth oscillation rises and falls between $2/3V_{cc}$ and $1/3V_{cc}$ (1.6V and 3.3V). Due to scope measurement errors, the max and min are recorded as about 300mV higher and lower than the actual values. These values are due to the 555 timer architecture. The IC contains two comparators with reference voltages separated by three resistors of equal values, dividing the voltage into thirds. This is why the wave oscillates from $1/3V_{cc}$ to $2/3V_{cc.}$

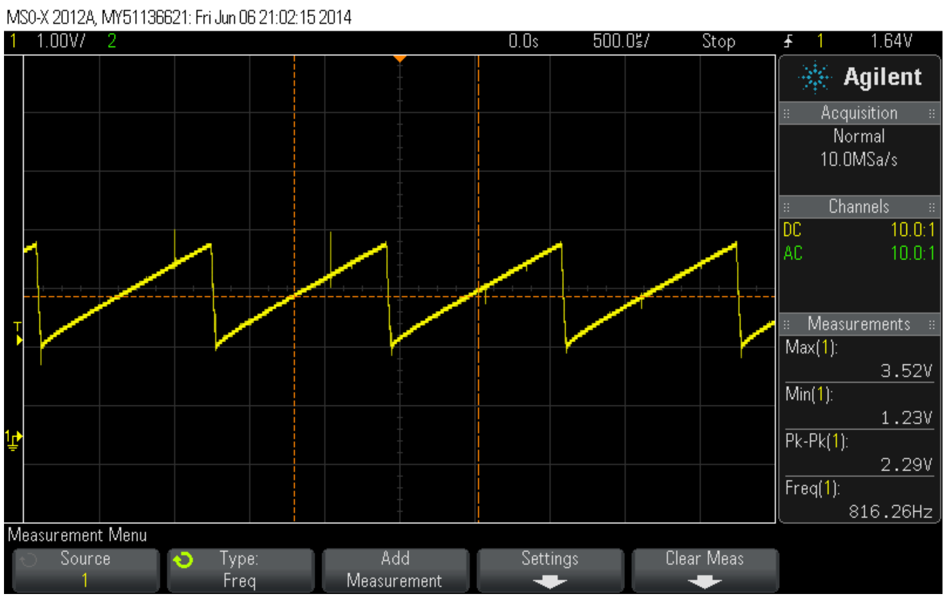

FIGURE 41
OSCILLATOR 1 555 TIMER

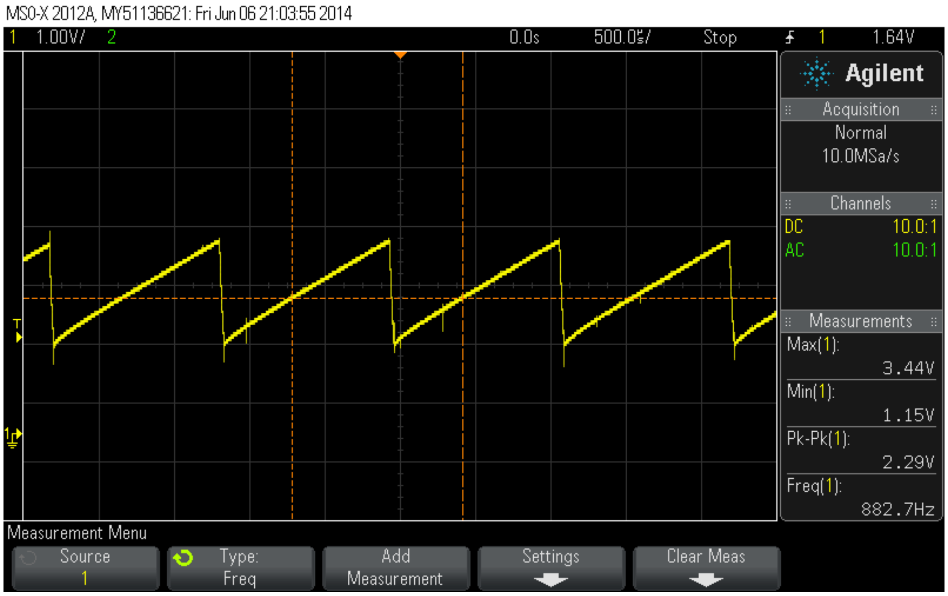

FIGURE 42
OSCILLATOR 2 555 TIMER

Again, this problem still needs to be addressed. A simple recalibration does not fully address the issue because over time, the frequency will again fall out of calibration as one of the many components undergoes normal wear-and-tear. However, in this design, since only 20% of Oscillator 2 is sent to the mixer, the frequency difference goes through unnoticed by the user and can be neglected. In the synthesis of other acoustic instruments, designers cannot overlook such a difference in frequencies.

The output of each comparator was also measured with the scope, as shown in Figure 43. The cursors measure the pulse width of the signal to ensure a 20% Duty Cycle wave to match the functional requirements of the block.

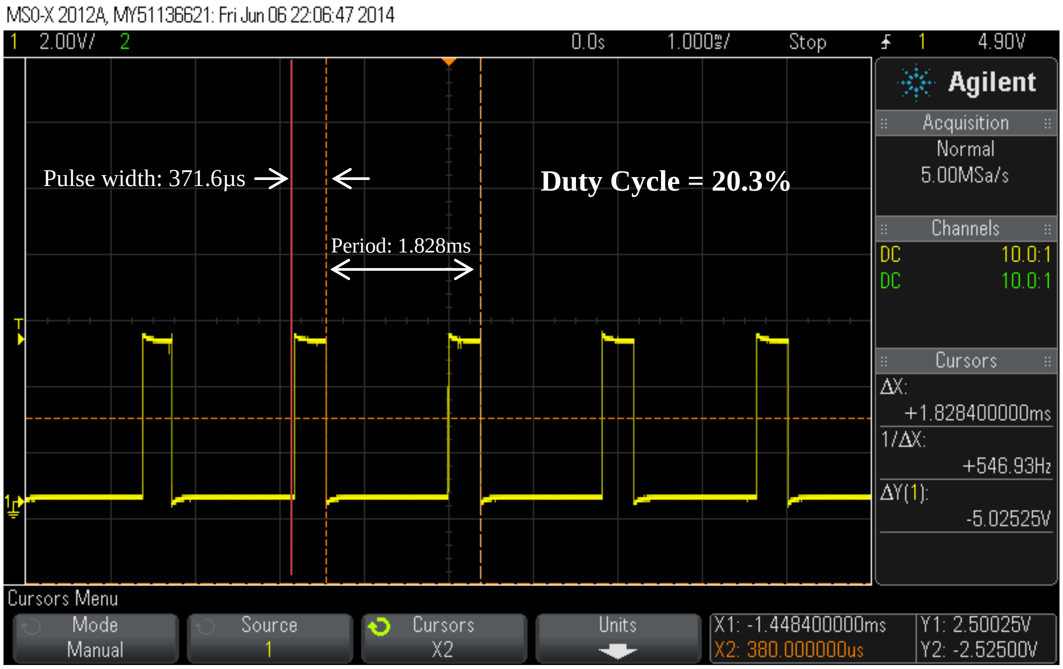

FIGURE 43
COMPARATOR OUTPUT TEST

7.3 Low Frequency Oscillator Test Results

The output of the LFO was viewed on the oscilloscope as shown in Figure 44. Cursors measured the parameters of the output sine wave to ensure that it meets the functional requirements. The signal does contain a noticeable amount of noise when viewed on such a small scale, but when added to the system, the noise has little to no effect.

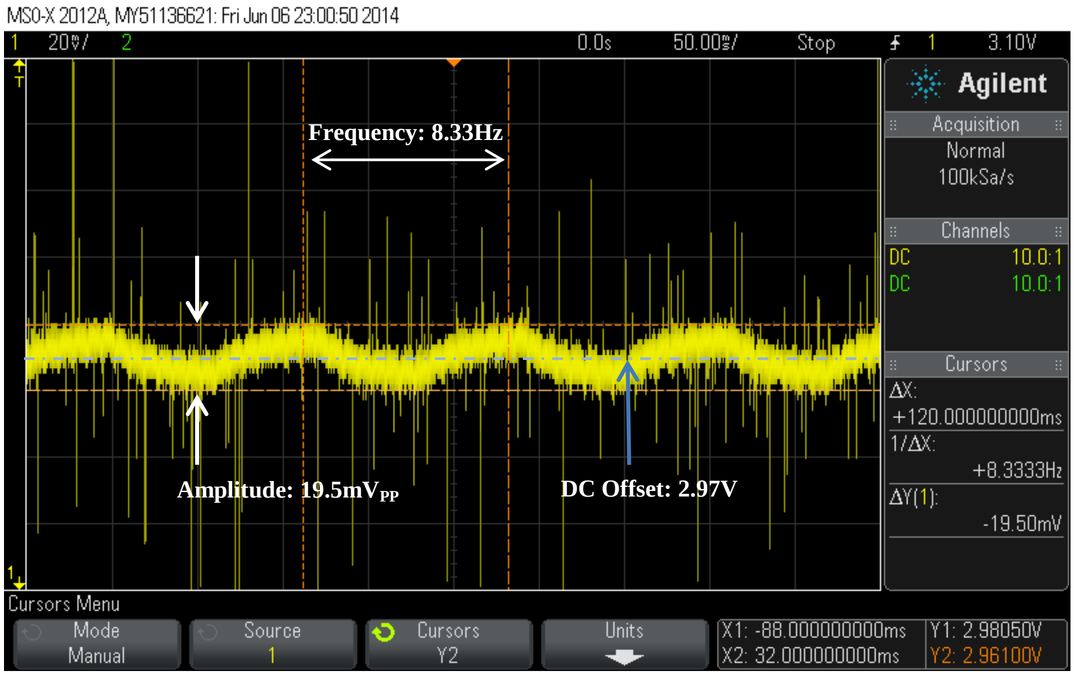

FIGURE 44
LFO OUTPUT

After this test, the oscilloscope also viewed the new output wave from each comparator. Here, the 20% duty cycle wave still oscillated, yet now the pulse with slightly modulated side to side at about 8 cycles per second. As anticipated, none of the noise from the LFO signal affected the PWM and the amplitude of 19.5mV$_{PP}$ provided just enough oscillation without drastically changing the duty cycle.

7.4 Mixer Test Results

The circuit was tested using the oscilloscope to ensure the correct output. Figure 45 displays the output waveform. Notice the presence of both oscillator frequencies, slightly detuned from each other. Also, the amplitude of Oscillator 2 is approximately 10.3% the amplitude of Oscillator 1, allowing for a slight beating effect to be present in the sound.

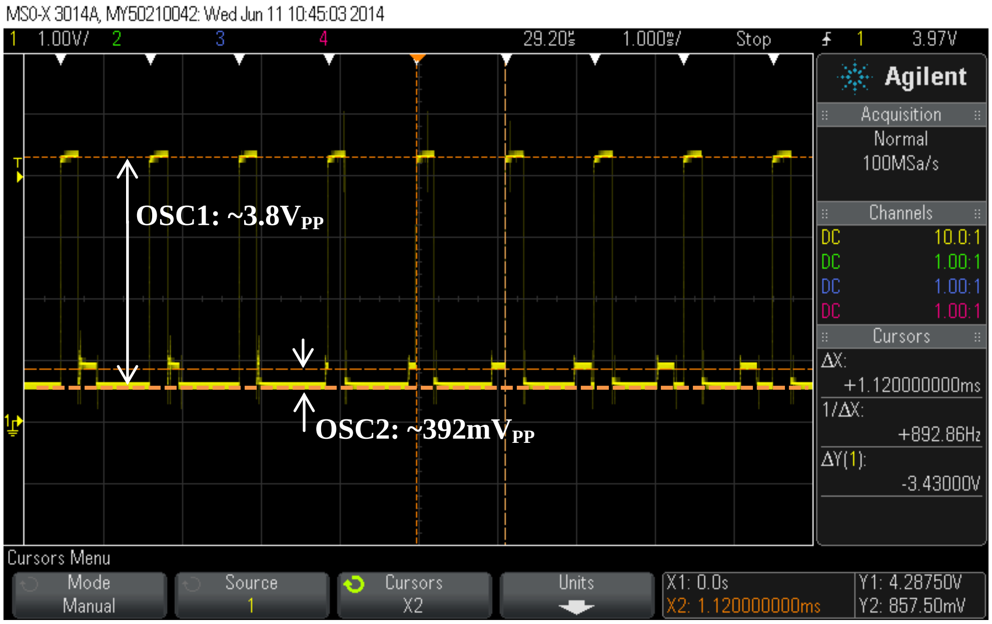

FIGURE 45
MIXER OUTPUT SCOPE CAPTURE

This design is helpful for the frequency matching problem because with such a low volume, the large frequency difference will only be slightly noticeable, and may even add to the sound. Violins contain many different resonances from the body and neck. Any error in frequency matching may just sound like a natural resonance of the violin.

7.5 Envelope Generators Test Results

The oscilloscope capture in Figure 46 shows the envelope generated for the VCA.

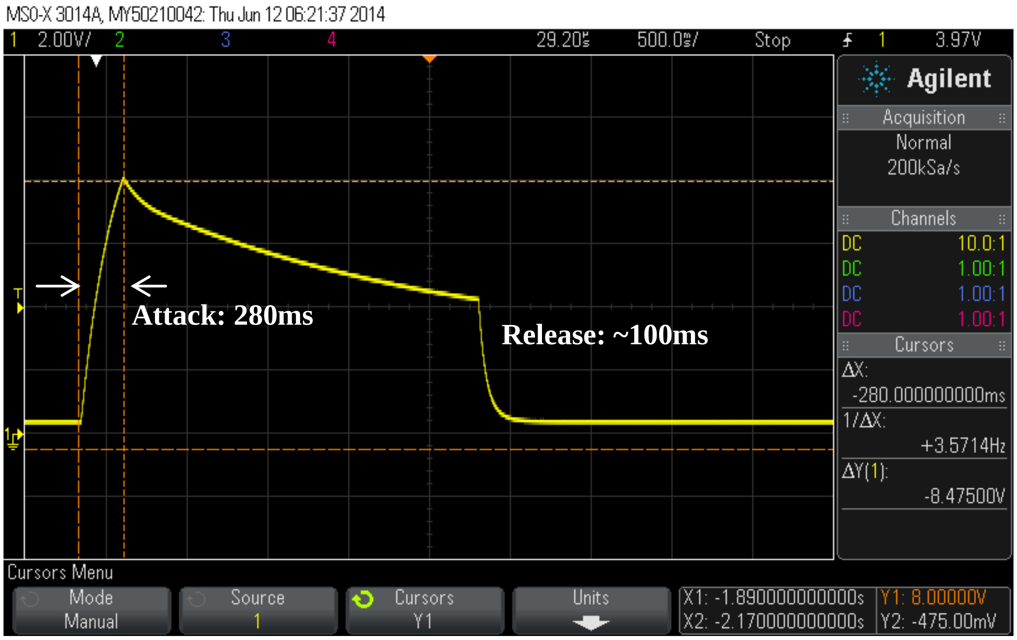

FIGURE 46
ADSR ENVELOPE SCOPE CAPTURE

The attack time rises with a time similar to the simulation of Section 5. The release time falls faster than simulated. The decay and sustain time do not exactly follow the simulation, with sustain decreasing at a speed much faster than the simulation. There still is a noticeable point when the decay slope elbows out, beginning the sustain portion. The falling sustain, however, does not greatly affect the circuit, because the VCA does not begin at 0 for a control voltage of 0V. If the key is held long enough, the sustain will continue to fall, though at a slower rate, but the sound will not disappear.

This characteristic of the VCA also affects the importance of a release and attack time. Both attack and release begin and end at 0V (or close). This means that the amplifier gain will start at a nominal value instead of starting and finishing at 0 gain. For the attack time, this characteristic is acceptable and we can assume that the user is very aggressive and the attack simply increases the gain as the bow initially pulls the string. For a release time, this doesn't matter at all because the oscillator will not even sound once the note is released. Therefore, the release time depends entirely on the oscillators and the envelope has no effect.

These results reveal that the presence of sustain and release stages are somewhat unnecessary, or provide few changes to the gain. Therefore, the same AD envelope from the Filter EG would also suffice, and the two parameters would match. This would also remove a whole block from the system, decreasing production time and cost.

The AD envelope generator scope capture is shown in Figure 47.

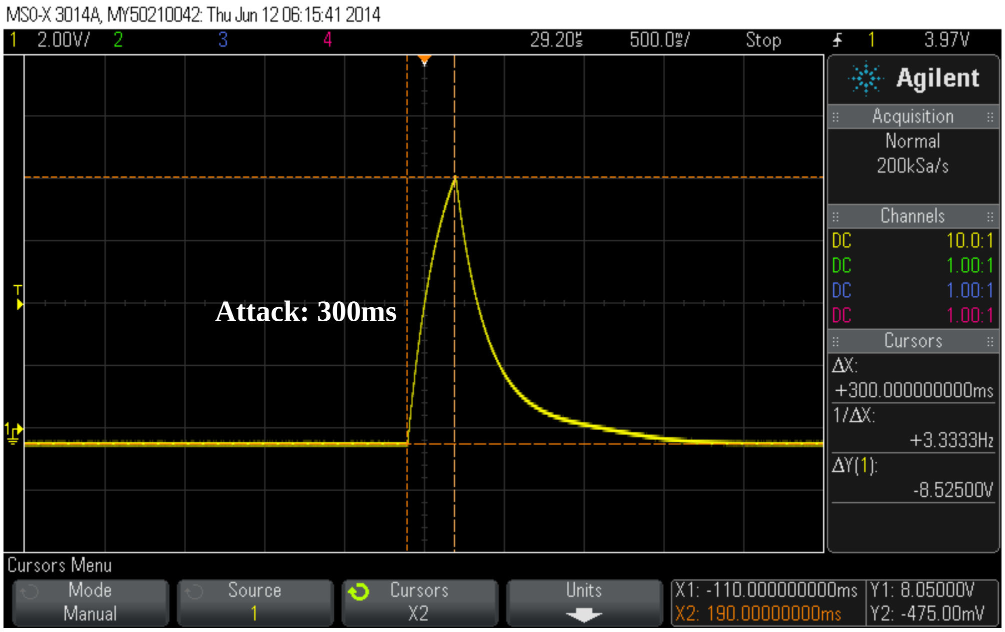

FIGURE 47
AD ENVELOPE SCOPE CAPTURE

Most importantly, the attack time matches the attack time of the ADSR envelope (within 20ms). The decay time is also comparable with the decay of the other envelope. To ensure that both of these envelopes begin together, both envelopes were viewed on the oscilloscope simultaneously (see Figure 48). The cursors indicate that both envelopes begin simultaneously with the same attack duration. This stage is the most noticeable, and it is most important that these match between the two envelopes.

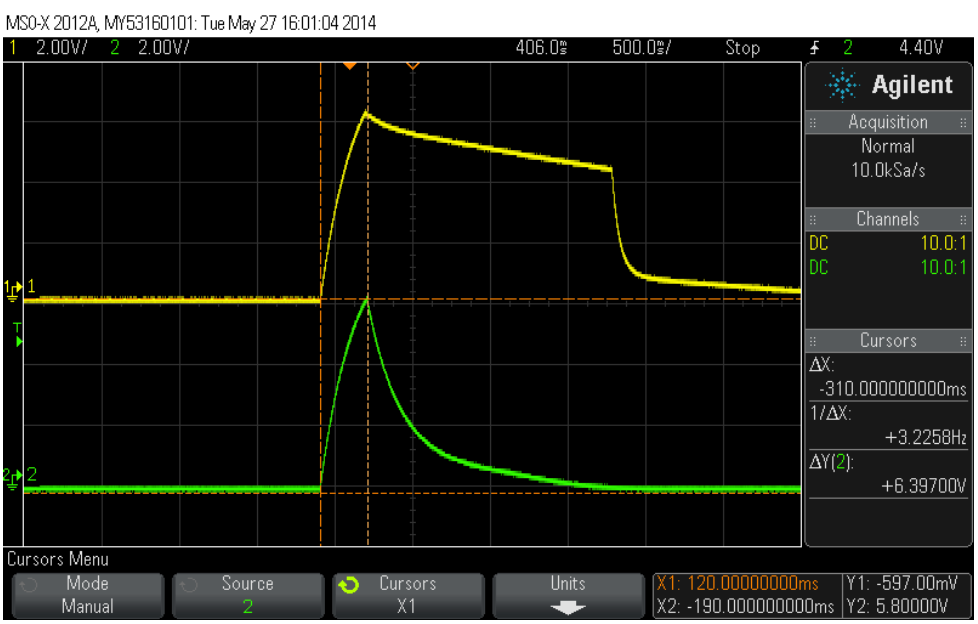

FIGURE 48
BOTH ENVELOPES TESTED

7.6 Voltage-Controlled Amplifier Test Results

The VCA was tested with the oscilloscope was tested on the oscilloscope by connecting the ADSR envelope generator to the $V_{control}$ input and the mixer output to the audio input. Observation showed that a note pressed produces the square wave that is sent through the amplifier. The amplitude of the output signal immediately rises to a peak and then slightly decays. When connected to a speaker, the gain changes is just noticeable enough without sacrificing the quality of sound. Figure 49 shows the signal at the output of the VCA after the key has been pressed for several seconds. This amplitude represents the nominal gain of the circuit.

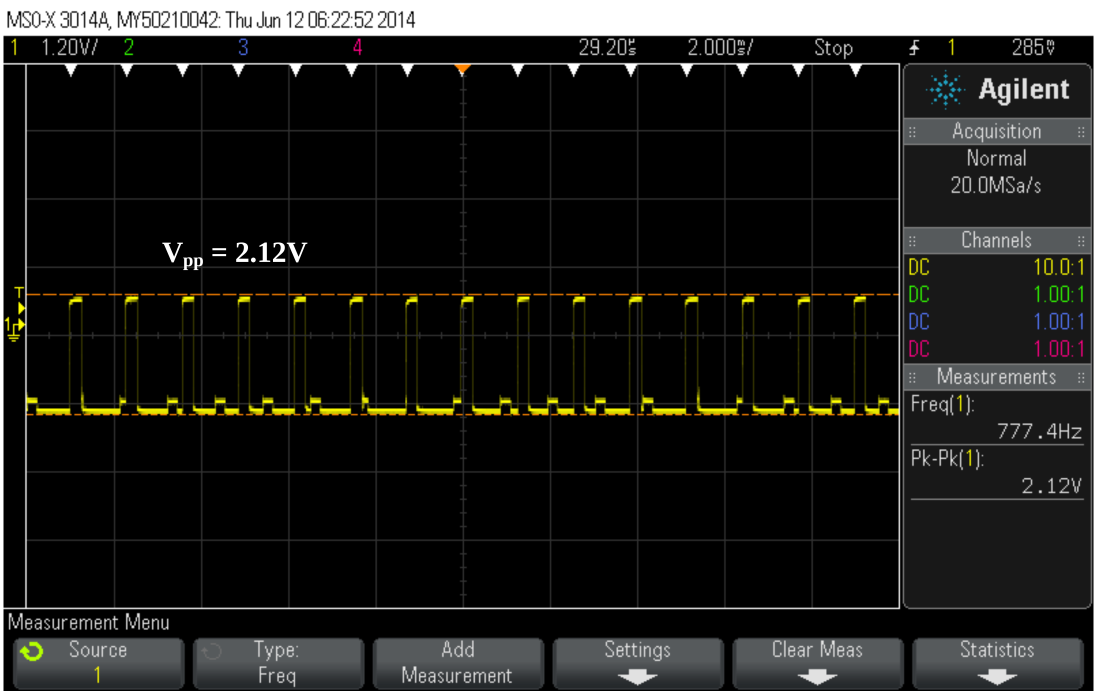

FIGURE 49
VCA OUTPUT SCOPE CAPTURE

7.7 Voltage-Controlled Filter Test Results

With the control and audio signals connected, the output of the filter was viewed on the oscilloscope, as shown in Figure 50. The scope capture shows both oscillators undergoing filtering. The wave is smoothed out, yet a narrow pulse can still be observed. Also notice that the peak-to-peak voltage was not greatly affected by the filter, so no further amplification is necessary.

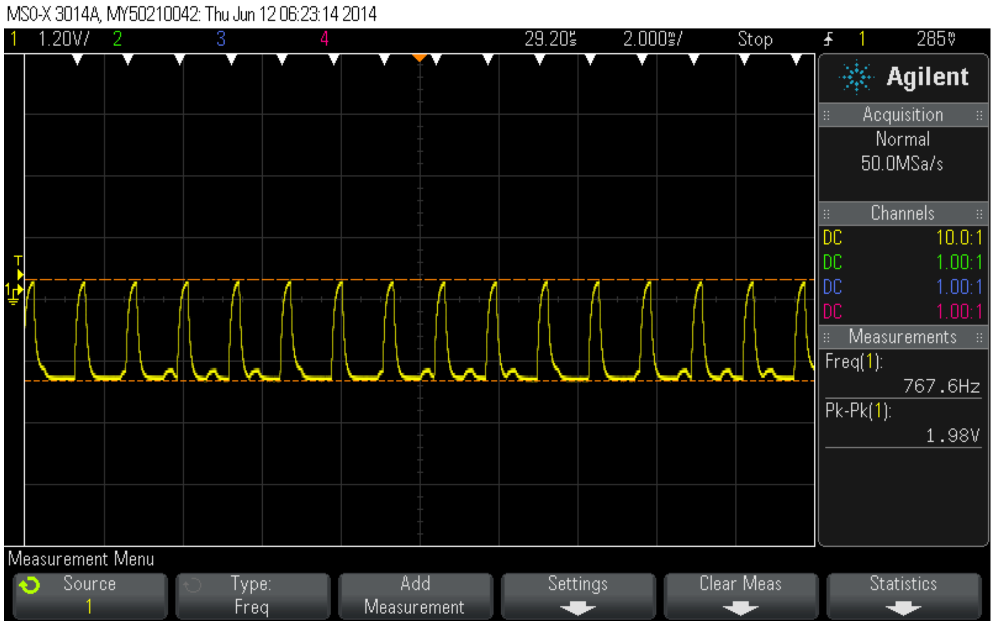

FIGURE 50

VCF FILTER OUTPUT SCOPE CAPTURE

7.8 Output Stage Test Results

Figure 51 displays the final audio signal sent to the speaker. The most important test is to determine that the signal was not altered by the presence of the speaker. By comparing Figure 51 to FIGURE 50, the signal was not affected.

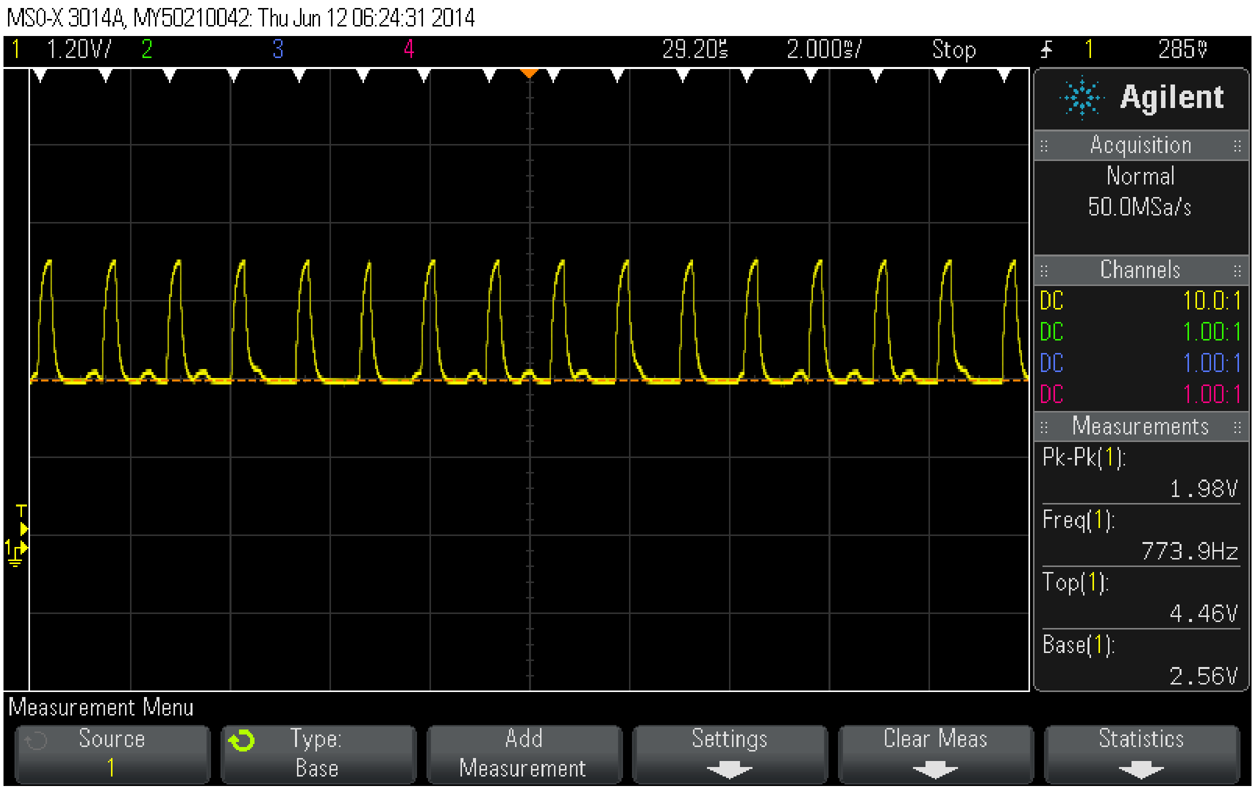

FIGURE 51

FINAL AUDIO SIGNAL SCOPE CAPTURE

Also, notice that the signal sits at about $3V_{DC}$. This will ensure that the signal does not fall below 0V or clip at 12V rail. According to theory, the equal voltage divider should shift the signal up to $6V_{DC}$. The results display a lower offset due to loading and leakage current, caused by the emitter follower and speaker. These components create a parallel resistance with the bottom resistor, decreasing the lower resistance of the divider.

Finally, using the speaker, the sound was tested while playing the touch screen to test for successful violin synthesis. My observations conclude that the sound does indeed resemble a violin. Higher frequencies best replicate the sound, due to the presence of less partials. Sometimes the lower frequencies can sound too electronic. When quickly moving the finger back and forth, the vibrato effect best reminds the listener of a solo violin timbre.

Also, when listening to the sound, the speaker clicked periodically when the sensor was note pressed. This problem was traced back to the 555-timer oscillators. It seems that even with no input, a small amount of current leaks into the CCO oscillator, causing a small oscillation in the mHz range. This frequency is far below audible, yet the changes still produce a “pop” that the speaker picks up. This problem cannot be solved without completely redesigning the oscillators, because even a picoamp leakage current will cause a small oscillation.

7.9 Summarized Test Results based on Design Specifications

The six engineering specifications were met by this design, still with room for improvement. Table 16 lists the test results corresponding to each specification.

TABLE 16
TEST RESULTS BASED ON SPECIFICATIONS

SPECIFICATION	TEST RESULT
1. Up to sixteen oscillators produce a violin's harmonic frequencies for each note played monophonically.	1. Using subtractive synthesis, only two oscillators produced signals to account for all of the required harmonic frequencies.
2. Produces notes (fundamental frequencies) between 220Hz and 880Hz.	2. The system oscillates from 22Hz to 1.4kHz, a greater range than specified.
3. Touch sensors receive the x-position touch input from the continuous interface to specify frequencies.	3. The touch sensor reads x-position and converts the input into a control voltage from 0.113V to 4.89V
4. The system should connect to a ¼" output jack with maximum voltage output of $2V_{peak\text{-}to\text{-}peak}$.	4. The final output has amplitude of $1.98V_{PP}$.
5. The system runs off 120V AC (wall). The system uses an AC/DC converter.	5. The power supply converts 120VAC into +6V, +12, and +/-12V DC supplies.
5. The dimensions should not exceed 24" x 12" x 6"	6. The box designed confined the system to the following dimensions: 13"x 8 ½" x 5 ½"
6. A removable casing encloses the circuitry and components.	7. A wood box uses nails and wood glue to enclose and protect the circuitry. The touch screen can be removed from the top, opening a hole to reach in and adjust the components.

8. Conclusions

In conclusion, the Analog Violin Audio Synthesizer design met all of the design specifications. Some specifications were exceeded, such as dimensions and operating range. The marketing requirements to drive these specifications were not entirely met. For example, marketing requirement #1 states that the system should have excellent sound quality. Upon hearing the final output sound, a violin sound can be recognized, yet by industry standards, the sound would not pass as an excellent emulation of a solo violin. Also, marketing requirement #3 states that the system is easy to play. This is true in a sense that the touch sensor simply needs to be pressed lightly and sound can easily be changed. However, to select a standard note, such as A-440, the finger must be placed in the *exact* spot corresponding to 440Hz. The slightest change in position will make a noticeable difference in output frequency.

To improve the design, a more stable oscillator would be used, possibly a more complex VCO as recommended by other designers. For example, Andre Lundkvist presents a VCO, VCA, and VCF analog design using multiple LM13700 amplifiers and other audio-rated op-amps [24]. These circuits are much more reliable than 555-timers, capacitors and resistors.

Also, a second touch screen can be implemented, this way the large touch screen can be quantized for discrete sounds, and a second smaller touch screen can control fine-tuned frequencies, for the vibrato effect.

A third improvement can be made by better interfacing the envelope generators with the VCA and VCF. More research will discover either a new circuit that goes low at 0V control rather than -12V, or an envelope generator that operates at negative voltages. Alternatively, a buffer design can level shift the envelope down to the desired negative start voltage.

Overall, the design proved successful and the use of a touch screen combined with analog circuitry provided a quality output that could easily be controlled. Minor changes can increase the quality and playability of the system to provide a more user-friendly design. The touch screen provides an easy vibrato effect that even a first-time user can simulate. Learning to play solo violin with vibrato and expression takes years, this synthesizer with touch interface allows aspiring musicians to simulate the real thing without piano or synthesizer knowledge.

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APPENDIX A – ANALYSIS OF SENIOR PROJECT DESIGN

Analog Violin Synthesizer

Student: Brandon Davis
Advisor: Wayne Pilkington

1. Summary of Functional Requirements

The Analog Violin Synthesizer produces an audio signal simulating a violin sound. The user controls the synthesizer via touch input. The continuous touch input allows the user to directly emulate a violin's vibrato effect.

2. Primary Constraints

The design and implementation process faces many challenges. Sixteen harmonic frequencies combine, creating the violin waveform, each a multiple of the fundamental frequency. To simulate this sound with subtractive synthesis, a complex waveform containing all harmonics is required. . A continuous input also means that frequencies between standard notes will be played as well. This creates a challenge for the designer to create a system that can change the frequency continuously, or on a very small scale. Also, as frequency changes, the violin spectra also changes, adding another challenge to the development of the synthesis technique [11]. Not only do I need to adjust frequency, but the amplitude of partials change depending on the frequency range.

3. Economic

- **What economic impacts result?**
  Human Capital is impacted by the work that I, as the designer, builder, and tester put in. I must practice time management, and planning to successfully complete the project while maintaining a job, relationships, and acceptable grades in other courses. Also, the time of my advisor is affected as Dr. Pilkington assesses and assists with the project. The companies that manufacture the parts ordered are affected in human, financial, and real capital. The parts are purchased, ordered, manufactured, and shipped. This process affects my money, Cal Poly's money (reimbursement), and the company's money and workflow. Natural capital is impacted for the metals and substrates used in the parts and in the tools used for production of the parts. For the consumer, this product costs more than other alternatives because analog technology is generally more expensive than digital. If a consumer wanted a continuous touch violin sound with analog quality, then they will spend more money to acquire the product.
- **When and where do the costs and benefits accrue throughout the project's lifecycle?**
  In this project, material costs accrue mostly in parts ordering. Costs may also be added to labor for specialty parts, or to purchase text and necessary literature. However benefits accrue from funding and donations offered from the school or other companies. For example, Cal Poly compensates up to $200 of part expenses. Also, Texas Instruments and other companies send free samples or other parts to fund student projects.
- **What inputs does the experiment require? How much does the project cost? Who pays?**
  The project requires student time and purchased parts. The estimated cost given by the cost estimate is $106.25. The money and parts are provided by Cal Poly, Texas Instruments, and myself. The Cal

Poly facilities provide all necessary test equipment (oscilloscopes, multi-meters, etc.) needed to complete this project. There is an estimated cost of $0 for equipment.

- **How much does the project earn? Who profits?**

  A standard analog synthesizer sells for a wide range of prices ($100-$3000) based upon functionality. This project is estimated to sell for $150, due to the single sound output. The actual labor and parts cost for design is much greater than $150, but after completing the project, duplicates will be made for an estimated $40-$50 dollars. The designers and manufacturers will profit from purchase.
- **Timing**

  Synthesizers last until a better model or similar product emerges and replaces the old one. Continuous improvements to old models are necessary for a surviving synthesizer. Research and development costs exist, and design, build and test costs exist for new models.

  The Gantt chart and cost estimates chart (TABLES 17 and 18) display the projected economic and timing impacts of the project.

TABLE 17

PROJECT GANTT CHART [12]

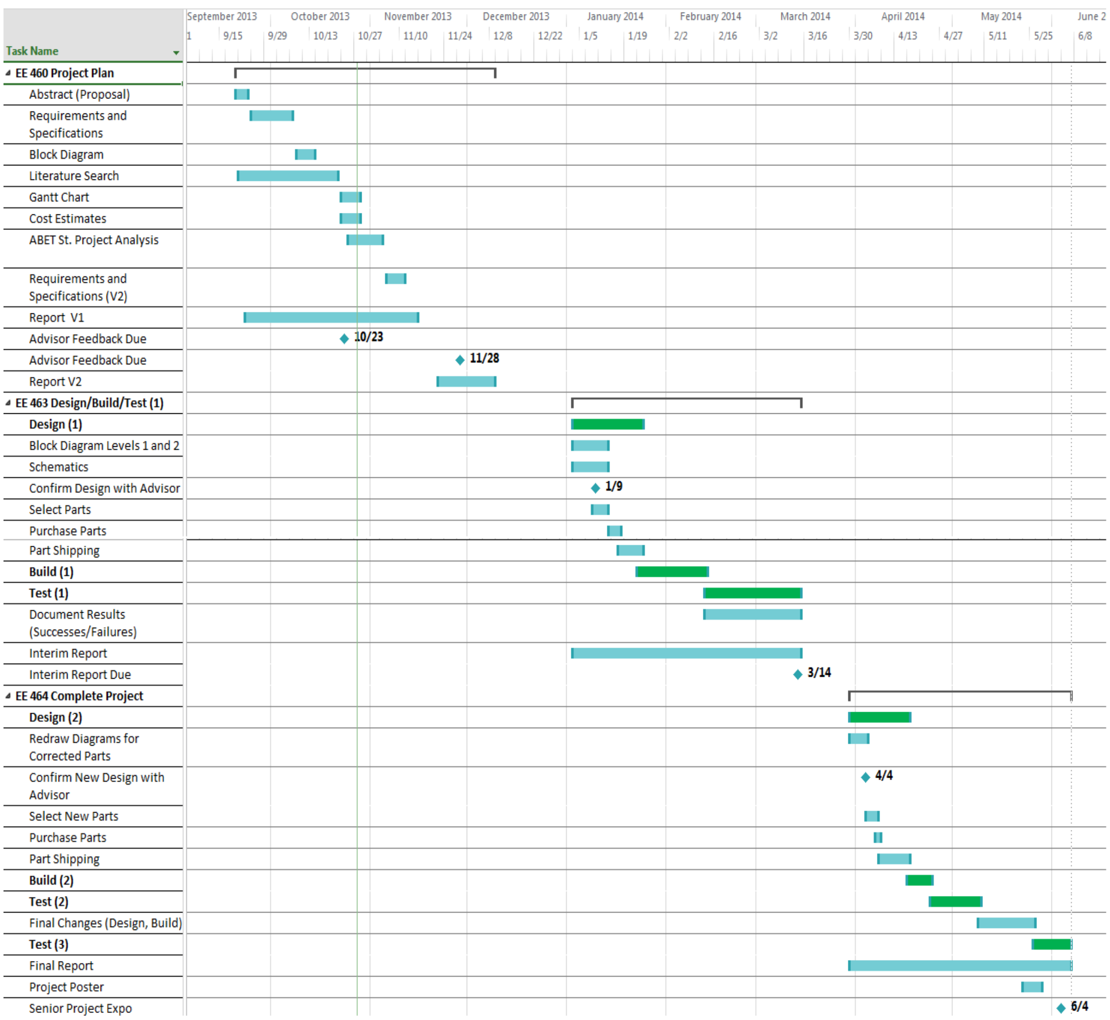

TABLE 18
PROJECT COST ESTIMATES [12]

	Worst-Case Cost ($cost_b$)	Expected Cost ($cost_m$)	Best-Case Cost ($cost_a$)	Cost Estimate from Equation 6
EE 460 Project Planning				
Labor Costs				
Report Writing and Documentation	$70.00	$50.00	$30.00	$50.00
EE 463 Design, Build Test (1)				
Parts Costs				
FSR-408 Touch Sensor	$25.00	$17.00	$15.00	$18.00
Various Resistors (max. 50)	$10.00	$5.00	$2.50	$5.42
Crystals (SER3301-ND) (16x)	$30.00	$16.00	$12.00	$17.67
Op-Amp (TLV2471) (10x)	$30.00	$20.00	$15.00	$20.83
Various Capacitors (20x)	$7.00	$4.00	$3.00	$4.33
Wire/Leads	$5.00	$2.00	$1.00	$2.33
Labor Costs				
Design	$50.00	$40.00	$20.00	$38.33
Build	$50.00	$40.00	$20.00	$38.33
Test	$90.00	$70.00	$30.00	$66.67
EE 464 D,B,T (2) and Completion				
Parts Costs				
Extra Resistors	$5.00	$2.00	$1.00	$2.33
Extra Crystals or Oscillators	$20.00	$10.00	$5.00	$10.83
Extra Op-Amps	$5.00	$3.00	$2.00	$3.17
Extra Capacitors	$5.00	$2.00	$1.00	$2.33
Poster Board	$15.00	$9.00	$3.00	$9.00
Casing	$15.00	$10.00	$5.00	$10.00
Labor Costs				
Design	$30.00	$20.00	$10.00	$20.00
Build	$30.00	$20.00	$10.00	$20.00
Test	$40.00	$30.00	$20.00	$30.00
Report Writing and Documentation	$30.00	$20.00	$15.00	$20.83
TOTAL PARTS COST	$172.00	$100.00	$65.50	$106.25
TOTAL LABOR COST	$390.00	$290.00	$155.00	$284.17
TOTAL COST	$562.00	$390.00	$220.50	$390.42

Equation 6 from Ford and Coulston Chapter 10 [12]

$$Cost = (cost_a + 4cost_m + cost_b)/6$$

4. If manufactured on a commercial basis

If manufactured on a commercial basis, the product is estimated to sell up to 5,000 per year [11]. The product is estimated to cost $40-$50 for manufacturing and be sold for $200. Therefore the estimated profit is $775,000 per year. With this profit, I would have to account for employee payment, advertising, factory maintenance, and other business expenses. When the user purchases the product, they only need to spend money on electricity and other audio equipment (speakers and cables). Most musicians have speakers and cables, so the cost to use the product compares to the price of electricity for a standard appliance. Upon manufacturing, companies would need to design and build tooling and test fixtures for fabricating and assembling a printed circuit board, the enclosure, and the final product. This would require non-recurring engineering and tooling charges that would have to be liquidated across the number of units sold as an additional cost; reducing the profitability.

5. Environmental

- **Describe any environmental impacts associated with manufacturing or use, explain where they occur and quantify.**
  The manufactured synthesizer includes printed circuit-boards and other electronic components. These part will eventually add to the world's e-waste. Printed circuit boards have a complex disassembly process, which eventually leads to dumping in third-world countries. During manufacturing, the main impact arises from the use of chemicals [13]. Of course, power consumption is increased from the product's manufacturing and usage.
- **Which natural resources and ecosystem services does the project use directly and indirectly?**
  The synthesizer directly uses metals including copper, nickel, and aluminum for the various electrical components and wiring. Polystyrene and polyethylene terephthalate are widely used in capacitors. The project also directly uses silicon, a very common semi-conductor material [13]. Indirectly, the product impacts the resources and ecosystems from which we derive power from, as the synthesizer consumes power from a wall outlet.
- **Which natural resources and ecosystem services does the project improve or harm?**
  The project harms the resources and ecosystems associated with the metals and materials listed above. Also, as an audio device, the use of this product will impact the use of recording equipment, CD's, microphones, computers, mixing boards, etc. All of these audio devices further impact these ecosystem services.
- **How does the project impact other species?**
  Since the product is designed to make sound, the use of the product outdoors with heavy amplification greatly impacts the outdoor noise level. The amplified sound impacts animals that rely on long distances communication. The extraction of metals used in electronic components affects species such as bugs and earth rodents, who live in areas of extraction.

6. Manufacturability

The main challenge associated with manufacturing the product is developing capital to support mass production in a factory. Often, companies have a difficult time starting up, because they cannot purchase the parts and build the product fast enough for the orders being placed. There are often long delays in distribution and slow payment periods, thus limiting capital. Also, many companies face the ethical issue of sending the product overseas for manufacturing, limiting jobs in America. However, some design practices can be implemented to improve the manufacturability of the product. For instance, I can implement materials that are common among analog synthesizers so that existing tools and fixtures can work to build the product rather than building/designing new ones.

7. Sustainability

- **Describe any issues or challenges associated with maintaining the completed device or system.**
  The completed synthesizer is designed to run without any further maintenance. If a knob or switch breaks, the user can order a new part. This is also true of the touch sensor. However, if a component in the main circuitry fails, then most likely the manufacturer will provide a completely new board.
- **Describe how the project impacts the sustainable use of resources.**
  The product consumes power from a standard 120VAC wall plug. Power can be provided by solar, wind, or other sustainable practices of producing power. as an analog device, it uses more electronics, circuitry, parts, and space, for less functionality. Digital synthesizers practice a more efficient use of silicon and other materials. Also, the use of recycled wood for the enclosure promotes sustainable use of resources.
- **Describe any upgrades that would improve the design of the project.**
  The project can be upgraded by implementing more efficient circuit design to decrease the number of components used to the bare minimum. Also, I can implement low power circuits and devices to decrease power consumption. One might argue that going digital would upgrade the product to add simplicity and functionality, though that would go against the purpose of the project. Also, to remove unneeded components, the same AD envelope generator could be used for both the VCA and VCF.
- **Describe any issues or challenges associated with upgrading the design.**
  Upgrades require more money for both labor and parts. Once the product is complete, money refocuses to manufacturing and marketing. When including upgrades, CEO's spend more money on design, build, and test phases.

8. Ethical

- **Describe ethical implications relating to the design, manufacturing, use, or misuse of the project.**
  The IEEE code of ethics statements 1, 3, and 9 discuss the importance of honesty and integrity with regards to safety and product performance. The synthesizer must come complete with warnings and directions for the installation. To "avoid injuring the user's property, I can recommend surge protectors and safety when plugging in the synthesizer. Recommending a safe environment for playing the synthesizer also represents professional conduct, consistent with IEEE code of ethics statement 3. The code further explains the importance of professionalism and responsibility in design. My design must contain the full specifications of components used and design practices in order to fully disclose the information, practicing honesty. Furthermore, I must accept responsibility for the product if the design causes harm to the manufacturer or user. The code extends not only to health and safety matters, but also includes ensuring functionality and fulfillment of specifications. For example, if the finished product does not fulfill every specification that the report defines, then I am practicing dishonest design and documentation and must either change the specifications or spend the time and money to improve the design.

9. Health and Safety

- **Describe any health and safety concerns associated with design, manufacture or use of the project.**
  The design stage begins with research, simulations, and other desk work. This may lead to back, neck, and eye problems, as an engineer spends hours at a desk. This lifestyle also impacts the morale and psyche of the engineer who is kept indoors for multiple days working in his or her office. When manufacturing the product, workers face the risk of exposure to toxic chemicals used in electronics [13]. Upon usage, the user faces the danger of minor electrocution when plugging the product into an outlet. Audio equipment always comes with the risk of ear damage. Many users are uninformed about the risks of permanent damage to the eardrum without proper volume control.

10. Social and Political

- **Describe social and political issues associated with design, manufacture and use.**
  The project directly affects the electronic music industry and the issue concerning simulated sound. Because the project simulates a natural violin sound, it can imply that this is how music is "supposed" to sound. Some audio engineers argue that synthesis should be used to create sound that cannot be produced acoustically rather than "putting a box" around what sounds can be produced.
- **Who does the project impact? How does the project benefit or harm the direct and indirect stakeholders? Consider equities and inequities. Consider locations, environments, political and social power of stakeholders.**
  The direct stakeholders include engineers, manufacturers, and consumers. The consumer category includes amateur musicians, households, performing musicians, and professional sound engineers. The product benefits the engineers and manufacturers by providing jobs. The households of these families indirectly benefit via the income obtained. Consumers benefit from the acquisition of new technology with touch control. Many inexperienced consumers can purchase the product and easily learn to play the instrument adequately. Indirectly, professional musicians, especially violin players, are directly affected. This product allows for musicians to simulate a quality violin sound without knowing how to play the violin. Studio violin players may lose jobs because the analog quality sound of the synthesizer becomes a cheaper and faster replacement for studio and live play. The addition of another electronic device into the music industry has the potential to take the place of a classical instrument for studio and live play. in the household, music students may not learn how to play the violin if they have access to a synthesizer that will simulate the sound of a violin with a smaller learning curve.

11. Development

- **Describe any new tools or techniques, used for either development or analysis that you learned independently during the course of your project.**
  During the project planning phase of design, I used Smartsheet to develop block diagrams of the analog synthesizer. I independently learned Microsoft Project 2013 to develop a Gantt Chart and Cost Estimate. The Literature Search below also includes references that I researched independently. Upper level design courses have taught me design of the various components used in this project. Along with EE courses, the Cal Poly Music Department also offers Sound Design courses taught by Dr. Antonio Barata. I had to expand my studies into electronic music studies to learn about additive and subtractive synthesis techniques. The following sources listed below proved useful in the development of the project.

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APPENDIX B – SPECIFICATIONS

Marketing Requirements	**Engineering Specifications**	**Justification**
1	1. Up to sixteen oscillators produce a violin's harmonic frequencies for each note played monophonically.	An instrument produces a note containing 16 different sine wave frequencies. These frequencies inhabit the natural harmonic series [3], [4].
1, 6	2. Produces notes (fundamental frequencies) between 220Hz and 880Hz.	These frequencies represent the fundamental frequencies of the two most commonly played octaves on an acoustic violin.
2, 3, 6	3. Touch sensors receive the x-position touch input from the continuous interface to specify frequencies.	A violin player produces vibrato by moving his or her finger up and down the fretless fingerboard. The continuous interface allows for note frequencies as well as the "in-between" frequencies for a real-time vibrato.
4	4. The system should connect to a ¼" output jack with maximum voltage output of $2V_{peak\text{-}to\text{-}peak}$.	Almost all speaker inputs connect to instruments with a ¼" instrument cable. Therefore the users will have no difficulty producing sound with the synthesizer.
4	5. The system runs off 120V AC (wall). The system uses an AC/DC converter.	Plugs into a standard wall outlet for easy and versatile installation and play. An AC/DC converter can be purchased online or at an electronics store.
5	5. The dimensions should not exceed 24" x 12" x 6"	Fits on a typical keyboard stand. Also fits on a pedal board or equipment rack.
4,5	6. A removable casing encloses the circuitry and components.	Resembles modern keyboards. Improves the visual simplicity.

Marketing Requirements

1. The system should have excellent sound quality.
2. The system should replicate the vibrato effect.
3. The system should be easy to play.
4. The system should easily connect to power and speakers.
5. The system should be visually pleasing to musicians.
6. The system should play two octaves of notes.

APPENDIX C – PARTS LIST AND COST

TABLE 19

PARTS LIST AND COST

Part	Manufacturer	Cost ($)
Assorted NPN transistor package 2N3904, 2N3904, 2N4401	Fairchild Semiconductors	$2.49
Assorted PNP transistor package 2N2907, 2N3906, 2N4403	Fairchild Semiconductors	$2.49
LM7805	Fairchild Semiconductors	$1.99
LM7812	Fairchild Semiconductors	$1.99
Assorted Resistor Package RS150-ND	Yageo	$16.95
ICM 7555 IPAZ (x7)	Intersil	$6.16
10.1 inch 4-wire Touch Panel	Chinatobby	$39.88
Bridge Rectifier	Fairchild Semiconductors	$1.50
1kΩ potentiometer (2x)	Fairchild Semiconductors	$6.28
LM339 Quad-Op Comparators (2x)	Fairchild Semiconductors	$5
Capacitors from Radioshack: 1uF, .1uF, .1uF	Fairchild Semiconductors	$5.69
LM-386 (2x)	Texas Instruments Inc.	$1.86
LM-13700 (8x)	Texas Instruments Inc.	$15.92
TL-0821 (10x)	Texas Instruments Inc.	$5.60
CES-67-1241 Center-tapped Transformer	CES	$16.90

Total Parts Cost: $130.70

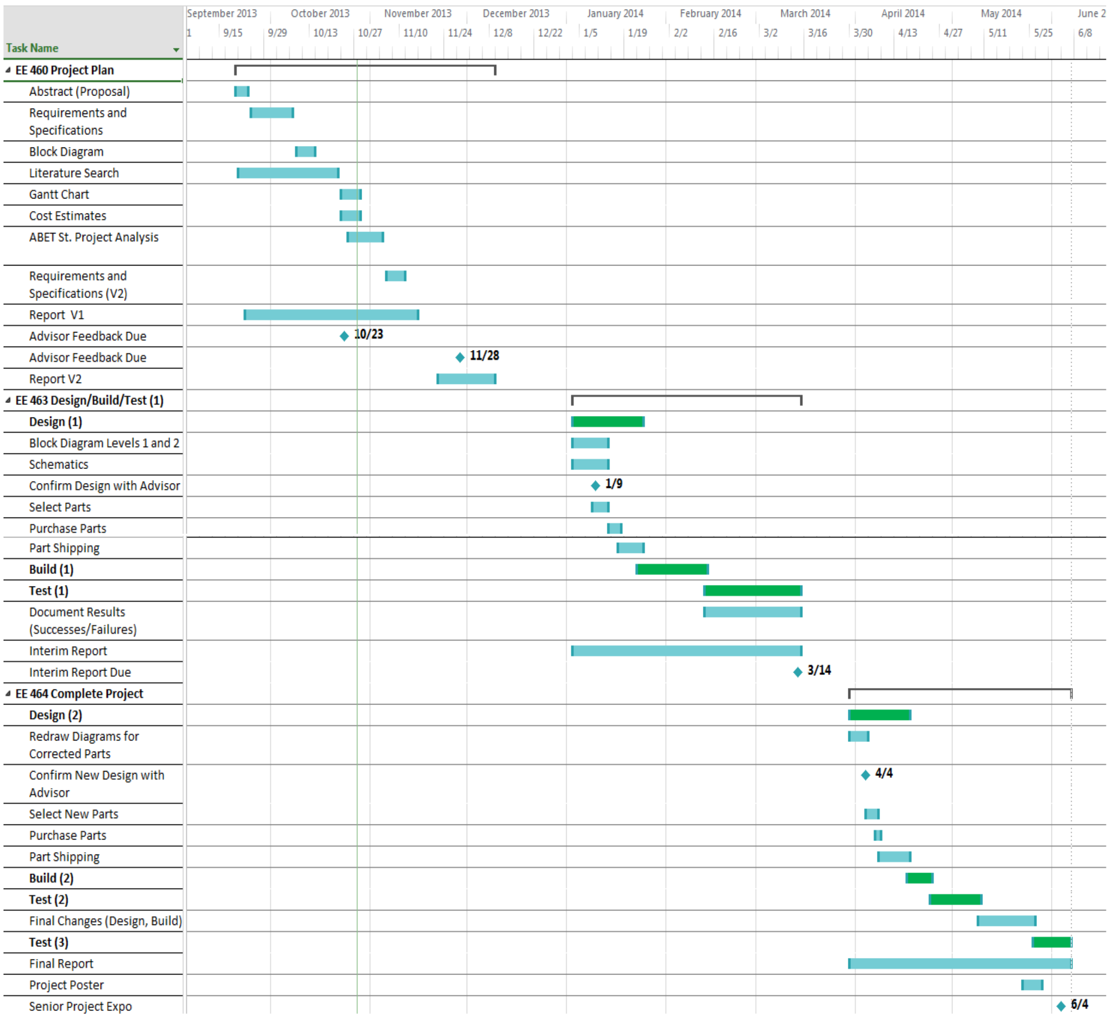
TABLE 20

FREE PARTS LIST

FREE PARTS	FROM
Various resistors and capacitors	EE senior project lab (SPL)
Recycled Wood	Woodshop
Power switch	SPL
Audio Jack	SPL
Power supply input port	SPL
SOLDER AND SOLDER WICK	SPL

APPENDIX D – SCHEDULE – TIME ESTIMATES AND ACTUALS

TABLE 21 GANTT CHART ESTIMATES



TABL 22 ACTUAL SCHEDULE GANTT CHART