Analog/Digital Polyphonic Guitar Synthesizer

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***Abstract* — This paper demonstrates the design approach of creating a polyphonic (multi-channel) guitar synthesizer that utilizes mixed-mode analog and digital circuitry. The design augments the standard array of musical tools used by many guitarists, and allows for the creation of sounds not generally available to the musician in the form of wave shaping and various other sound effects such as variable envelope shape, tremolo, vibrato, and an arpeggiator. This paper also presents the challenges and reasoning behind choosing analog or digital methods for certain functions of the system.**

***Index Terms* — Active filters, analog circuits, digital circuits, microcontrollers, signal synthesis, voltage-controlled oscillators.**

I. Introduction

The purpose of this project is to design an entirely new instrument that is controllable with a standard electric guitar that possesses musicality, accuracy, playability, and portability. The tones produced should be on a frequency range which can be processed by the human ear and are harmonious when played in progressions among chords. The synthesizer generates tones that correspond to an electric guitar and obey the fundamental theory of which a guitar player is familiar. The features of the control interface are be organized in a way which they can be easily identified and accessed in a live setting. The final synthesizer enclosure will be small enough in size and weight that it can be transported without difficulty and set up at any venue with ease.

The design is a polyphonic system that is structured primarily with analog circuitry. Multiple strings can be played and manipulated at the same time with each having an individual response.

II. System Overview

In order to make a system that is polyphonic, separate pickups were made for each string, so that six different channels may be operated independently. These signals pass through a filtering and amplification stages before their frequency information is converted to control voltages. These control voltages are then passed to voltage-controlled oscillators followed by wave shaping circuitry. To get a more dynamic sound, amplitude modulation is performed using voltage-controlled amplifiers coupled with a user-controllable envelope generator. This envelope generator is microcontroller based, and receives signals called the gate and trigger which supply information concerning the beginning of a new note and its duration. The six channels are then summed together and passed through a filtering stage that allows the user to control both cutoff and resonance, allowing more elaborate control over the dynamics of the overall sound. Other effects such as tremolo, the time-varying rise and fall of volume, and an arpeggiator, a time-varying effect that changes the pitch, both utilize a low-frequency oscillator which also allows user control of speed and depth (amplitude). Finally, these systems are powered by a switching bipolar power supply with ±12V and 3.3V outputs.

III. Signal Acquisition

The first step in synthesizing an analog guitar signal is acquire the signal. The reason a normal guitar output is not sufficient is because a guitar outputs all six strings on one signal, thus not allowing the synthesized signal to be polyphonic. In order to overcome this issue a six channel guitar pickup was designed using AlNiCo magnets with magnetic wire windings, but with six individual inductors versus one large one. This way every string can be processed individually and allow for a complex polyphonic output. In order to send this output to an off-board processor, a 15 pin VGA cable was implemented, which has more than enough channels for task. A six channel pickup requires six different signals and then one common ground. This leaves 8 open pins to put some on board options on the guitar.

IV. Input Stages

Each of the six signals from the pickup will need to pass through a series of input stages for filtering, frequency acquisition, and note beginning and length information. The output of these sections pass to various parts of the system that utilize the information gathered every time a note is played.

A. Low Pass Filters

Because of the nature of the guitar, it's steel, round-wound strings cause harmonics that inhibit the accuracy of frequency to voltage conversion. Since there is a strong second harmonic on the lower range any given string, each channel is filtered individually using a Sallen-Key low-pass filter. This topology was chosen for its small size and ease of design for multiple channels. The choice of op-amp for this task was given to the OPA2132, for it's low input offset (<500µV), wide range of supply voltages (±2.5 to ±18V), and low distortion (0.00008%).

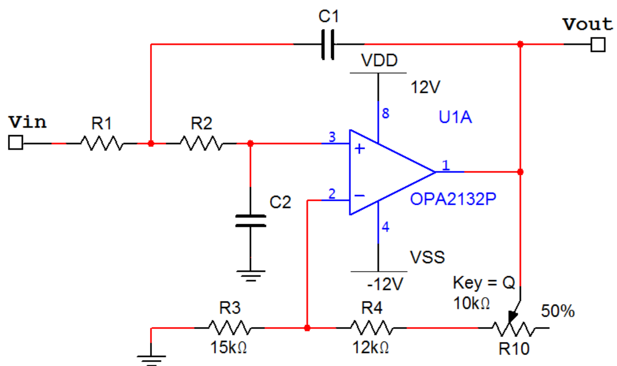

Fig. 1. Sallen-Key filter schematic for single channel. R_1, R_2, C_1, and C_2 are selected for each channel according to the string's open note frequency.

The choice for cutoff frequency, quality factor and gain were made by testing the harmonic content of the pickups without filters, and measuring the second harmonic vs the fundamental at various points along the string's length. It was found that as the note frequency was raised (the string's length being shortened), the second harmonic reduced considerably. Also, the second harmonic of the lowest notes appear as playable notes on the neck at the 12th fret and higher. With these considerations, a cutoff of about 20% past the open note frequency was chosen. A variable Q from about 0.83 to 1.89 was used to allow for tuning the gain K with trimmer potentiometers. The component values in Fig. 1 for each string were chosen by using common capacitor values, and solving for the resistor values. This approach is typical due to the higher availability and lower cost of more accurate 1% thick film SMD resistors.

B. Frequency to Voltage Conversion

When a note is played, its fundamental frequency must be converted to a control voltage to be used by the voltage-controlled oscillators. There were several methods to achieve this, including using DSP, but a tachometer IC, the LM2907N was chosen since it is designed for use with variable reluctance magnetic pickups. It has great linearity (0.3%) and can be used with the +12V supply used throughout the system. It features a very simple design equation for achieved a desired V/Hz output slope, shown in (1), where R and C are component values within the circuit, and Fin is the frequency input.

$$Vout = V_{CC} * R * C * Fin \tag{1}$$

The slope was designed to have a maximum output of 5V at the frequency of the 20th fret of each string. The component values were selected similarly to the filters, with R in series with a 100kΩ trimmer potentiometer to tune the slope for accurate performance. Fig 2 shows the output of the LM2907N at various notes along the range of the lowest E string along with the desired slope with a maximum error of 1.65%.

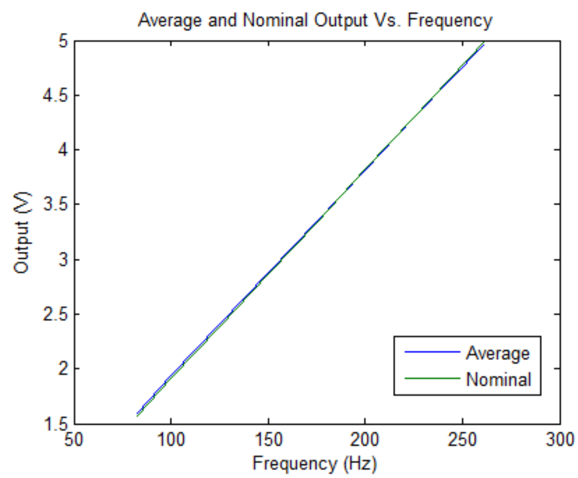

Fig. 2. Average and nominal output of the LM2907N for the low E string.

One initial problem with the LM2907N was output ripple. Minimum ripple could be obtained by trading off response time. However, response time is critical for real-time use, so it was designed to have a minimum settling time of 16ms. Ripple was reduced by utilizing a 2-pole Butterworth filter from [1]. The pole frequency is:

$$f_{pole} = \frac{0.707}{2 * pi * R * C} \quad . \tag{2}$$

C. Triggers and Gates

The trigger and gate block takes the input signal and outputs binary signals which signify when a new note begins and how long it stays active. The trigger is a binary pulse that is activated whenever a new note is hit on the guitar string, and the gate is active whenever the input signal is above a certain threshold. The information from this block is passed to the envelope generator microcontroller, which determines how to set the envelope according to these signals. The trigger and gate circuits are realized by passing the signal through an envelope follower and into separate comparator circuits, each with a different reference voltage and hysteresis. For the envelope follower, the smallest, but still effective design approach was a small signal diode and parallel RC path to ground. The LM339 quad comparator was chosen for a more compact design. It is powered by 3.3V, so the output can be coupled directly to the microcontroller's input pins without reducing the voltage.

The trigger's reference voltage is set at about 1V, and gate's to about 160mV. These values were selected based on average experimental output amplitude of plucking the string as a normal guitarist would. The main problem with this method is the inconsistency of every player. For our prototype, the reference voltages will be variable with trimmer potentiometers, to allow for someone who tends to pick lighter or softer. A more commercial version would have these controls on board, so that the user may change the parameters according to his/her playing style.

V. Wave Shaping, Mixing and Control

Three waveforms are generated from voltage received from the FVC: square, triangle, and sine. The user is able to select either the square wave or the triangle wave to mix with the sine wave, and also control the proportion between the two mixed signals.

In order to generate a square and triangle wave, the LM13700 transconductance amplifier is used since it is a current controlled device; the voltage from the FVC can be applied across a resistor to generate a current, and with the circuit arranged as an oscillator (as shown in the data sheet), the resulting waveforms will oscillate with a frequency that is proportional to the input current. The oscillating frequency is determined by the following equation:

$$f_{OSC} = \frac{I_C}{4CI_AR_A} \tag{3}$$

Ia is this equation is constant because it is generated from the positive power supply rail; therefore, the oscillating frequency is a function of the control current Ic (thus, the corresponding resistor Rc) and the capacitor. Each guitar string has different values of Rc and C to generate waveforms within different frequency ranges in Table 1.

String	E(low)	A	D	G	B	E
Δf (Hz)	82.4-261.6	110-349.2	146.8-466.2	195.9-622.3	246.9-783.9	329.6-1046
C	100nF	100nF	47nF	47nF	33nF	22nF
Rc	22k	16k	22k	16k	22k	22k

Table 1. Component values chosen for tuning of each channel.

The pcb schematic of the LM13700 is shown in the Fig. 4, where V+ is the input control voltage, and the triangle output and square wave output are indicated by TRI_O and SQ_O, respectively.

The input current into the LM13700 drains to the negative power supply rail after passing through two transistors, resulting in a -12+1.4V drop. In order to compensate for the negative 12V and the transistor voltage drop, the voltage coming from the FVC is added to -10.6V through a simple mixing circuit, that way the voltage entering the LM13700 is initialized to zero. This schematic is shown in Fig. 3, where V+ is the control voltage coming from the FVC and R7 is a trim potentiometer that will aid in precisely achieving -10.6V.

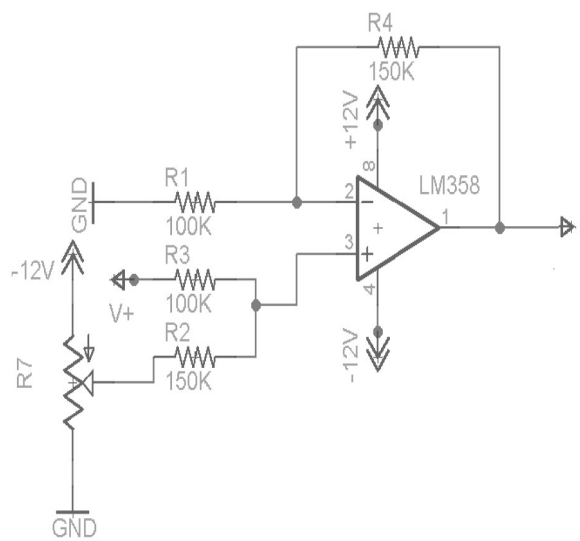

Fig. 3. Negative voltage summer and amplifier for VCO control voltage.

To create a sine wave, the triangle wave from the LM13700 is sent through a simple integrating circuit. The RC values of the cutoff frequency were chosen based on the average frequency of each guitar string. Since the

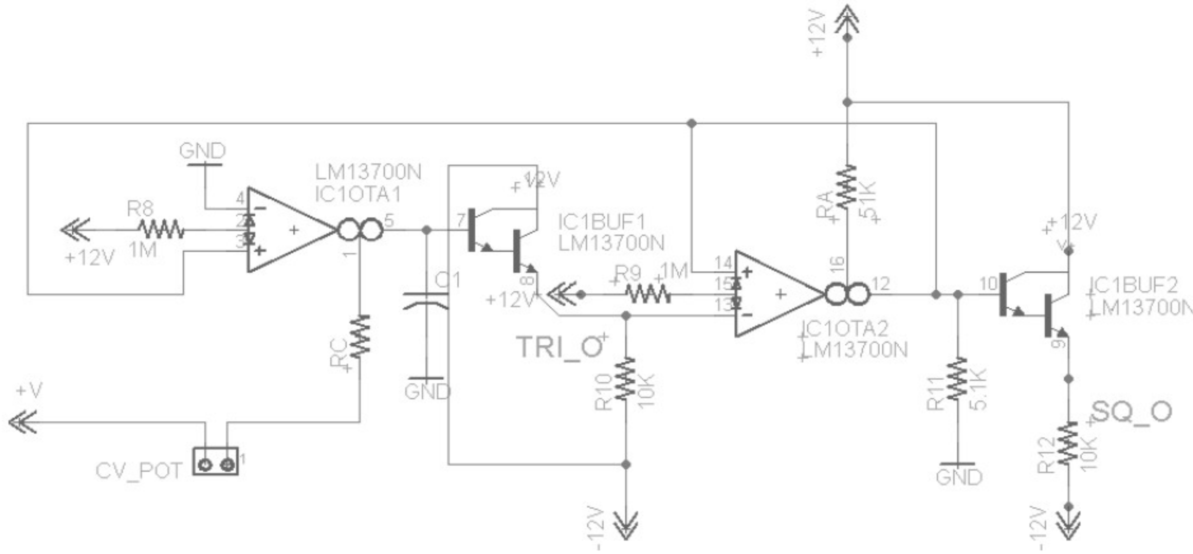

Fig. 4. Schematic of voltage controlled oscillator.

frequency range of a guitar string is exponential, the average frequency is not the center frequency, but instead, approximately one fret higher. Equal amplitude of all waveforms is required, so the configuration of R1 and R2 in the integrator circuit contribute a gain of 1.41 to the output signal since the sine wave will naturally only reach magnitude .707 of the input triangle wave.

With three waveforms available for modulation and synthesis, the user can toggle between the square wave or triangle wave to mix with the sine. LED's on the top panel of the synthesizer indicate which waveform has been selected (triangle or square). Finally, the user can vary the proportion between the sine wave and the selected wave with the use of sliding potentiometers (one for each string). The output of each potentiometer is sent into a simple mixing circuit to combine the resolved signals of each guitar string separately, prior to entering the Voltage Controlled Amplifier.

VI. ADSR Envelope Generation

The envelope generation block focuses on creating a more dynamic sound by adding configurable contour to the volume profile, which would otherwise be constant. To achieve this more dynamic sound, and also allow for user-controlled flexibility, a voltage-controlled amplifier is used in conjunction with a microcontroller-based envelope generator. The only downfall to this method is size, because each channel requires its own envelope in order to maintain polyphony.

ADSR stands for Attack, Decay, Sustain, Release, which describes the signals characteristic shape. Fig. 5 shows the output of the ADSR (top) with the envelope of a guitar signal as the input (bottom). The attack is initial rise (set to a longer time for illustration); the decay falls to the sustain level, which is constant as long as the gate signal is active, at which point the release cycle begins. The shape of each rise and fall is modeled digitally after an RC charging circuit. This method is used to imitate earlier synthesizer ADSR circuits, which have a more “natural” sounding rise and fall in volume due to the ear's logarithmic nature.

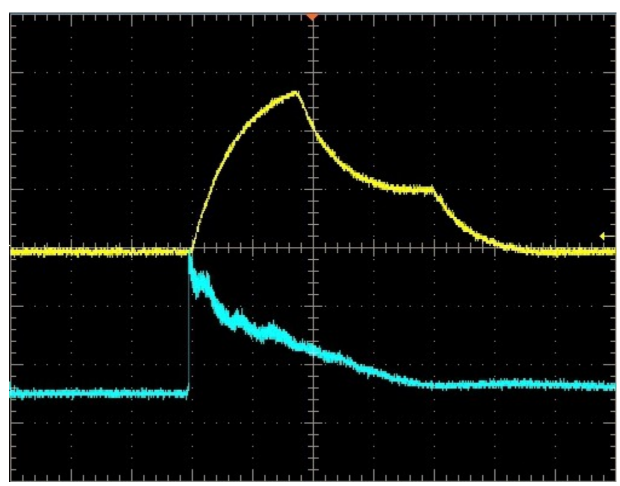

Fig. 5. Top: output of ADSR envelope generator. Bottom: envelope of the guitar input signal.

A. Voltage Controlled Amplifier (VCA)

In order to modulate the amplitude of the oscillator's output, the VCA circuit uses the LM13700 transconductance op amp which has an input bias pin. The VCA has two inputs: the signal to be amplified and the control voltage. The control voltage is generated by a microcontroller, then amplified and summed with -8.1V DC to work with the amplifier's bipolar supply voltages.

B. Microcontroller

The actual envelope is generated using a microcontroller with a low-pass filtered PWM output. An MSP430F4152 was selected for the following requirements:

- MSP430F4132 microcontroller
- 16KB Flash, 512B SRAM, 8MHz
- 56 GPIO, 64 pin LQFP package
- Has multiple port pin interrupts
- 8 10-bit ADC channels
- Outputs up to 6 PWM signals from capture/compare peripheral

The msp430 accepts the trigger outputs as port pin interrupts, as well as the gate signals into general purpose input/output pins.

C. Software

The MSP430 software is programmed in C using Texas Instruments' Code Composer Studio and sent to the chip using the Launchpad development platform, which uses the Spy-Bi-Wire interface common on MSP430 chips for communication.

To output exponential curves, it was decided to use a wave table rather than calculating the result with time-costly floating point math. Similar to direct digital synthesis, a variable phase pointer allows samples to be skipped and the time shortened. The wave table holds values that map directly to the PWM timer values, so that they may be copied directly to the capture/control registers.

The attack, decay, and release cycles have potentiometers to control their time, while the sustain has a potentiometer for level. The ADC ports are polled every 32ms within the MSP430's watchdog timer routine.

The software waits for trigger signals to activate port pin interrupts, then, within the interrupt routine, determines which trigger occurred. This then begins the attack cycle. When the final value is reached, it begins the decay cycle, and then the sustain, which is held at a specified DC level until the gate signal goes high (inverting comparator output), then the release cycle begins. If the gate goes low at any point, then the release cycle begins. If a new trigger occurs, then the attack cycle restarts.

VII. Signal Modification

Once the signal from the guitar is converted into a synthesized electric signal, the user then has the option of several different modifications they can run their signal through. This modification can occur when the signal is in a DC control voltage or when the waveform has already been synthesized.

A. Low Frequency Oscillator

The heart of all the signal modulation is the low frequency oscillator. This provides a square and sine wave oscillation from the range of about 3Hz to 33Hz. This was done by using a Wein-Bridge oscillator with the oscillation frequency being controlled by a stereo potentiometer.

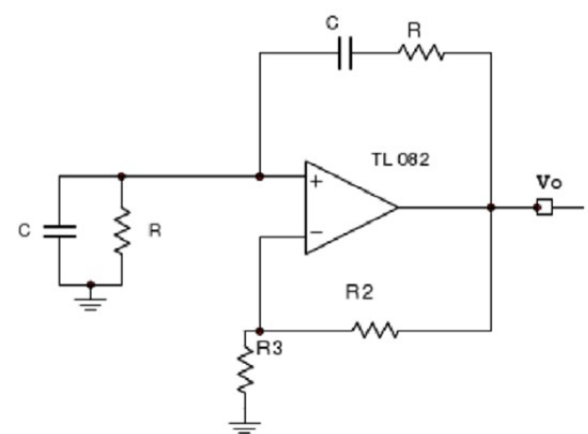

Fig. 6. Wein-Bridge Oscillator

The low frequency oscillator is used to drive different circuits and modify the signal in various ways. The voltage is also stepped up or stepped down in order to provide proper voltages to other circuits. Each separate effect has its own depth control but all of these share the same rate of oscillation which is controlled by the stereo potentiometer.

B. Tremolo

Tremolo is the rabid increase and decrease in the amplitude of a signal. This effect is realized by the LM13700's voltage controlled amplifier configuration. The voltage is being driven by the LFO with a depth option added. The complex signal enters into this design and gradually swells and decays In volume. A potentiometer attenuates the LFO signal in order to provide the user with variability of how much they want the signal to decrease and increase in volume. The speed of the oscillator will also determine the speed of volume swells.

C. Vibrato

Vibrato is known as the rapid increase and decrease in frequency of a signal. In music this means that the pitch of the note is being increased and decreased. This is done by passing the DC controlled voltage from the frequency to voltage converter through a summing op amp configuration where the controlled voltage is being added with a stepped down version of the oscillators signal. Thus the Controlled voltage with gradually increase and decrease according to the rate on the LFO. A depth potentiometer is added so the user can control how much they want the pitch to increase and decrease. This effect also takes place in the same schematic as the arpeggiator.

D. Arpeggiator

Another modulation that takes place after the frequency to voltage converter is the arpeggiator. An arpeggiator automatically steps through a sequence of notes based on the input note creating an arpeggio of the root note, major third and perfect fifth. This is done by taking advantage of the linearity between the frequency to voltage convert. Meaning if an output frequency is to be doubled to create an octave, all that needs to be done is doubling the input controlled voltage. So the arpeggiator is created by sending the DC controlled voltage through various stages of gain. In order to keep the root note a gain of 1:1 should be added. This is simply done by keeping the signal the same value. A major third harmony is created by giving the signal a gain of 1.25 or a 5:4 ratio and a perfect fifth is done by giving the signal a gain of 1.5 or a 3:2 ratio. In order to switch between gain stages a non inverting configuration was used. A different resistor value gives different appropriate gain values to the input signal. This is done by using a 4:1 demultiplexer with resistors at the input and the output at the output of the op-amp.

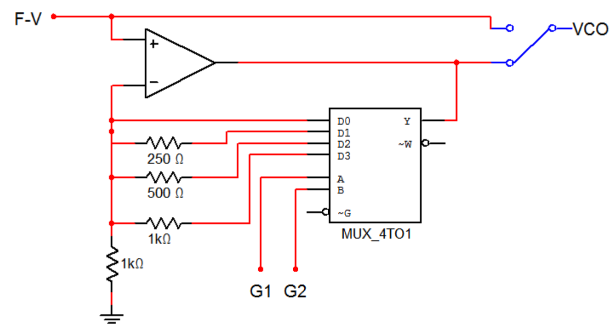

Fig. 7 Arpeggiator layout.

The multiplexer is driven by two controlled voltages with control the switching speed and pattern of the arpeggiator. This is done by passing the LFO through two different gates. Each gate has an adjustable threshold voltage which is controlled by a potentiometer. By varying these two separate gates a different switching time can be achieved. So this makes the multiplexer switching speed dependent on the speed of the LFO and the switching pattern according to the settings of the two gates. If both the threshold voltages are set to high then the output of the MUX will always be on the 00 logic setting. Here the circuit acts as a buffer allowing the control signal to pass unaltered. This is so the effect is always on but the user can choose to bypass it without having to flip six switches.

VIII. Final Stage Filtering.

When mixing many complex signals sometimes the output signal results in many rough overtones and harmonics that are overall undesired. At the final stages the signal passes through a final filter to take out any unwanted harmonics. This filter takes advantage of the voltage controlled resistor configuration of the LM13700. The filter works as both a bandpass filter and low pass filter depending on where the signal is acquired from. The two options on this filter are cutoff frequency and resonance. The cutoff filter varies where the filter begins to attenuate frequencies. If at the lowest setting the LP cutoff filter will cut off all frequencies passing through. At the highest setting the cutoff filter allows most frequencies to pass. When these types of instruments were first developed many users enjoyed the way the resonance sounded on some of these filters. So the second option on the filter is for the user to set the resonance value. This determines how much resonance is output and feeds back into the circuit. These final two stages should allow the user to create a custom effect out of the already synthesized signal.

IX. Power Supply

The power supply will output +/-12VDC for all IC's involved in the design, as well as +3.3VDC for the microprocessor and envelope detector, while supplying no more than 400mA of current. The National Semiconductor's LM2576 Simple-Switcher voltage regulator was used for each of the supply rails since it requires only a few external components and will offer regulating efficiency of at least 75%.

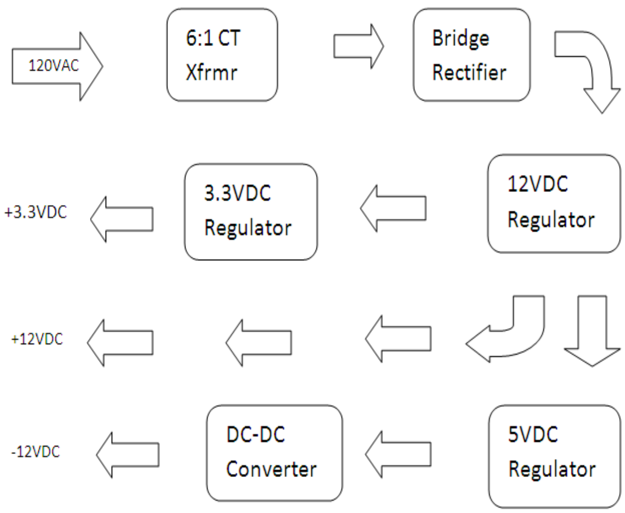

Fig. 8. Flow diagram of the power distribution.

A linear Technology LT3579 DC-DC converter is used to step-up and negate a regulated 5V input voltage to -12VDC output. Fig. 8 shows the flow diagram of the power distribution for the synthesizer.

A dual primary/dual secondary, center tap transformer is used to step down mains voltage from 120VAC to 10VAC rms. Following the transformer is a Varo Semiconductor, Epoxy Bridge Rectifier rated up to 100V. The unregulated DC output is filtered for ripple and fed into the 12V LM2576 regulator. The regulated 12V will act as the input voltage for both the 3.3V regulator and a 5V regulator (which feeds the DC-DC converter).

The components for each of the regulators were determined using the layout guidelines offered from the manufacturer's data sheet. Toroidal inductors were used for the 12V and 5V regulators to reduce EMI, and Schottky diodes were used because of their low forward voltage drop and fast switching capabilities. 8.9degC/W heat sinks are mounted to each of the regulators and the bridge rectifier. All components are surface mounted onto pcb with a combined weight of less than 2lbs.

The power supply was tested by varying the input voltage and the load resistance, and monitored for voltage deviations; the output voltages remained constant under these conditions, proving the proper functionality of the power supply.

Fig. 9 shows the pcb schematics for each regulating stage of the power supply, along with component values.

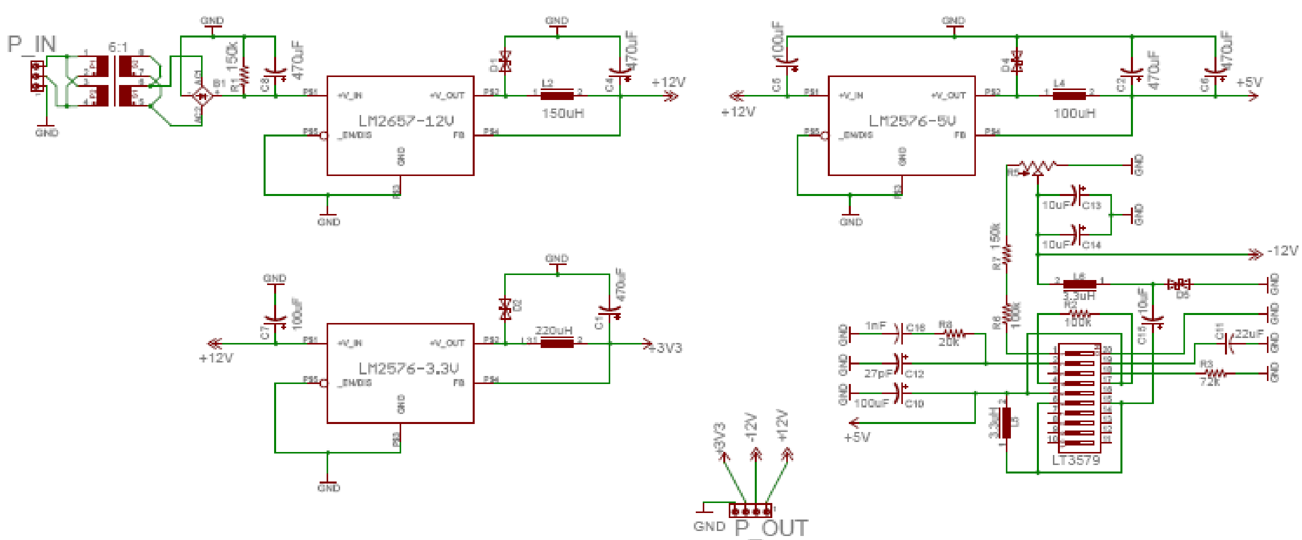

Fig. 9. Schematic of switching power supply.

X. Conclusion

Designing a functional polyphonic guitar synthesizer that was both accurate and included features that would attract performing musicians was challenging for the group to complete. Overall, the end result came out to be successful in both areas. With some functional tuning, the accuracy could be achieved, and the added effects from the wave shaping circuitry, ADSR, tremolo, vibrato, and arpeggiator give the user a wide array of sounds to create with.

Acknowledgement

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Biography

Erin Browning is an Electrical Engineering Student graduating from UCF in May 2011. She currently works in the power generation industry for Turbine Technology Services. She has a passion for music and jumped on board with the guitar synthesizer project because she knew it would involve engineering practices that are different from her past experiences.

Matthew Mohn is currently a senior at UCF studying Electrical Engineering and plans on graduating in May 2011. His interests are in analog and digital signal processing as well as microcontrollers with a focus on media applications such as musical equipment. He also designs and sells guitar effects to musicians across the Orlando area.

Michael Senejoa is a musician and currently a senior at the University of Central Florida. He plans to graduate with a Bachelor's in Electrical Engineering in May of 2010 and pursue a career in audio signal processing.

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