# VERS-1

Erin Browning

Matthew Mohn

Michael Senejoa

Motivation: Increase the versatility of the electric guitar by allowing polyphonic, real-time processing capabilities for modulation in a live setting.

## Isolated stage Single coil Guitar pickups

When designing a synthesizer guitar, the first big issue of the project is how to read, process and synthesize a note correctly. The best solution for this problem was found in the past by the audio processing company Roland. Their solution was to first isolate every string's pickup versus the normal guitar pickup configuration that mixes all six strings into a complex signal. By doing this a computer can interpret a note being played much easier without having to worry about certain harmonics or overtones. A normal guitar pickup configuration would only allow for once not to be played at a time (monophonic) but this new design allows the synth guitar to now be polyphonic. The single coil magnetic pickup is hand built beginning with a small cylindrical soft magnet such as Neodymium, Alnico or ceramic. Then a small magnetic wire with the gauge of #42 or #43 must be wound around this magnet for at least 300 turns. This makes the equivalent circuit of the magnetic wire to be an inductor with a resistor in parallel with a capacitor. The magnetic coil is then placed facing the string and when the string vibrates, this induces a small AC signal into the coil and the load is to be read as if it was over a capacitor. Once the strings are wound they are to be submerged in wax (Besswax 20%, Paraffin 80%) to coat the turns and hold them in place over time. Normal guitar pickups need to be wound around 8,000 turns in order for it to provide a good strong clean signal. This is because the signal that is being picked up is ultimately going to be the amplified signal that is being heard. In this case the pickup is used only to generate the frequency that is going to be read by the frequency to voltage converter and the envelop is going to be read by the envelop detector to be used by the voltage controlled amplifier. This means that the actual tone of the guitar is not as important other than the envelope and frequency. Each coil will sit diagonal to the next in order to provide the necessary space between them because of the isolated windings. Because of specially designed guitar pickups such as these, a special type of cable is required because the standard 1/4' coaxial guitar cables only allow for the transfer of a single signal. This will be done by using a cable with a minimum of 12 pins in order to transfer all six signals at once.

- 42 gauge magnetic wire
- 150 ft. of wire
- Output an amplitude of 200mV

Cost

- .5 lbs of 42 gauge magnetic wire - $30
- 6 alnico magnets – $9.60
  Total = $40

## Switching Mode Power Supply

A switching mode power supply (SMPS) will be designed and constructed to convert 120VAC to 20Vdc, providing a 16W supply to the synthesizer box. An SMPS will provide a more efficient means of delivering power to the load than a linear power supply because instead of regulating output voltage with Ohmic losses in the form of heat, a SMPS utilizes the ideal storage characteristics of inductors and capacitors in configuration with transistors (which have no resistance when "closed"); therefore, nearly all the input power is delivered to the load, with few, minor losses due to the non-ideal circumstances of reality.

The first stage of a SMPS is the rectification of the AC signal source to DC. This will be done with a diode bridge or a voltage doubler circuit. In the next stage, the dc signal is inverted to pass through a high frequency transformer to be stepped down to the desired voltage level. The high frequency hum will be inaudible to humans as long as it is above 20 kHz (but dogs may become irritated). It is then rectified again to produce a dc output, followed by additional capacitors for smoothing.

Adequate voltage regulation will be implemented to monitor the output voltage of the power supply and generate feedback that automatically increases or decreases the supply voltage as necessary, to compensate for any tendency of the output to change. (Output variations could occur because of load changes and/or changes in temperature).

Specs:

- 120VAC Voltage In
- 20VDC out
- 20Watts Out
- Transistors, Capacitors, Inductors
- Autotransformer

Cost:

- Storage Elements: $10
- Autotransformer: $15
- Plastic Casing: $5

Total: $30

## Frequency to Voltage Converter and inter-stage conditioning

The output waveforms from each of the pickups will be sent their own frequency-to-voltage converter. This F/V converter must drive the next block, the voltage-controlled oscillator (VCO), with a voltage that is scaled properly, so that the resulting frequency coming from the VCO matches that of the string's fundamental frequency of vibration. Depending on how the VCO is designed, the scaling of the F/V output may be either linear or logarithmic. Either way, a simple single operation amplifier circuit should suffice. The resistor and capacitor values will have to be carefully chosen in order to achieve an accurate representation of the string's frequency. The F/V converter can be realized using a frequency to voltage converter IC which outputs a linear voltage with respect to the input frequency. The envelope of each of the pickup's output signals (from hitting a string) are not taken into consideration in this part of the circuit. An envelope follower will run in parallel with the F/V converter which will detect the “attack” of the string being plucked.

Specs:

- Low power quad op amps
- 5-9V range of power supply
- 4mA supply current (F/V converter)
- 60Hz – 2kHz (full range of guitar)

Cost:

- 6 * $2 for LM2907 f/v converters
- 3 * $0.50 for LM324 quad op amps
- 2$ for resistors/capacitors

Total: $18 with shipping

Voltage controlled Low frequency Oscillator.

The purpose behind the VCLFO is to create an oscillation based on the voltage that is being read from the output of the frequency to voltage converter. This consists of several Operational Amplifiers in a configuration that would provide a 360 degree phase shift feedback loop creating specific oscillations based on a “control voltage”. The output frequency is going to have to be adjusted and the values of the resistor and capacitors will have to be chosen in order to create a proper frequency of oscillation. In standard American music theory the value of 440Hz must correspond to an A. If the frequency is doubled, meaning 880Hz then this should complete a whole octave. Every octave there is twelve half steps and this would be the smallest increment of voltage that the VCLFO would have to be able to convert to. The standards of Controlled Voltage Oscillators for synthesizers is 1 volt per octave, but in this project it will be different based on the output voltage of the frequency to voltage converter. When the user hits an A on the guitar this should then be read into the frequency to voltage converter, converted into a dc voltage and fed into the VCLFO which then creates an artificial tone mimicking that of the guitar string. Because there will be special magnetic guitar pickups that isolate each string it will be possible for the user to play polyphonic notes (more than one note at a time). This means there will have to be six separate oscillators, one for each string and from those six oscillations a total of 18 will be created with integrators. The original oscillations will be generated in the form of square waves. From this the signal will be integrated to form a triangle wave oscillation and once more for a sinusoidal wave oscillation. The sin wave will be used as the original tone and then the user is given the option between adding a square wave or a triangle wave to the sin wave. There will be an option where the user chooses how much of each oscillation they would like by turning a knob, which adjust the volumes proportionality. Although this frequency mimics the same frequency played on the guitar it provides only that information of the original waveform. This means that the envelope of the original wave will need to be controlled by the voltage controlled amplifier.

Specs

- Lm13700 with two transductance with linearizing diodes and buffers
- Low power and high accuracy
- +/- 18 volt

Cost

- LM13700 = $1.67 x 24 = $40
- Resistors and capacitors = $10

Total = $50

Envelope Detector and VCA

As well as the frequency information, the envelope of each string pluck must be captured and accurately synthesized. To do this, an envelope follower circuit will accept input from each pickup. If needed, a buffer will drive each envelope follower. The buffers should counteract any loading issues concerning both the envelope follower and the F/V converter inputs. The resulting signal from each envelope follower will be sent into the control voltage input of the voltage controlled amplifier (VCA) for each of the six oscillations. The oscillation signals from each of the VCOs will be sent into the signal input side of the VCAs. The resulting output of each VCA should be a frequency that matches the string plucked as well as the amplitude envelope of the "pluck" of the string, as well as its decay.

Specs:

- Low power quad op amps
- 9-12V single supply voltage
- ~700µA supply current

Cost:

- $2 for resistors/capacitors/diodes (envelope detector)

- $3-6 for VCAs
  Total:  $10 with shipping

Voltage Controlled Filter

Unlike all the other components in this build that are controlled by a voltage coming in from another source, this voltage-controlled device is driven by the users input on the controls and not by the guitar. When you have several tones and several frequencies that are all mixing together, harmonics and overtones start appearing on the spectrum. Although many of these are desired effects some harmonics can be harsh to listen to and quite undesirable. The purpose of the voltage-controlled filter is to provide the user with two options for filtering their overall sound, cutoff frequency and resonant frequency. Most of the undesired harmonics and overtones that appear are in the higher range of the audible frequency spectrum. In order to solve this problem there will be a low pass filter implemented where by turning a potentiometer the user chooses how high or low they would like the cutoff frequency to be. If the knob is turned all the way down, then only very low frequencies would pass through the filter and vice versa. The other option would be a potentiometer to adjust the Q value of the resonant frequency. This is a spike in the amplitude of frequencies that appear right before the cutoff 3db point. The higher the knob is turned the higher the spike in amplitude of the frequencies before the 3db cutoff frequency. This sort of effect appeared in early 70s synthesizers and was usually and undesired result of having a low pass filter where the user chooses the cutoff frequency, but many people enjoyed the tone that a spiked resonant frequency would produce and engineers started implementing this into their design. Since a voltage controls this filter, a low frequency oscillator can be implemented to give the cutoff frequency an oscillating range versus a fixed range. The same low frequency oscillator that corresponds to the VCA can also be used to give both of the parameters a synchronous effect.

Specs

- Wide bandwidth dual JFET input operational amplifier (TL082)
- +- 18 volts

Cost

- 2 x TL082 = $4
- Resistors and capacitors = $3

Mixer

Since each guitar string will have individual oscillators, they will be generating unique voltage signals independent of each other.  All of the signals will have to be mixed (or added) in order for the final output to be single audible signal which will undergo further modulation prior to audio amplification.

The goal is to have all six signals mixed together with proportional control between the treble and bass notes of the guitar. This will take place in two stages: stage one will be the direct coupling of all six voltage signals to an inverting op-amp, and the voltage output will be the sum of all six. Stage two will be the additional proportional control coupling to the inverting op-amp in parallel with the stage one coupling. The two stages will be added together and sent through a final inverting op-amp with unity gain to produce a positive output signal.

During stage 2, each voltage signal will be coupled to one of two inverting op-amps. The three treble notes of the guitar will be summed together, and the three bass strings will be summed together. The outputs of each of these stage 2 adders will be fed into an analog multiplier which will multiply the difference between the signals with an adjustable voltage controlled gain value set by the user via turning knob. This output will unite in parallel with the stage 1 adder in combination with the sum of all six individual signals. The result is a proportionally controlled adder.

An analog multiplier will be a linear quadrant multiplier so the output voltage can swing between polarities. This means the user could swing the control voltage from a negative minimum to a positive maximum. Polarity will also define the difference between the treble (higher octave) output and the bass (lower octave output). If the control voltage is set to its maximum value, the output voltage would be only one signal. If the control voltage is set to the minimum value, the output would be the other signal. If the control voltage is exactly between the maximum and minimum values, the output will be the average of the two input signals. Anywhere in between the maximum and minimum voltage supply values would produce a proportional output between the two input signals. In this case, the two input signals are the sum of treble strings and the sum of the bass strings. Combined with the sum of all six strings at the stage one op-amp, the result is a full additive mixer of all six signals (strings) with a proportional control between the treble and bass notes which will be further modulated and finally audio amplified.

One disadvantage of using analog multipliers is if any errors exist on the input signals, they will be multiplied on the output. This may require fine tune filtering prior to the signals entering the multiplier.

<u>Specs:</u>

- 4 general purpose op-amps rated at 680mW/ea: (2.72W)
- 15 resistors for coupling at approximate power rating of 15mW/each: (.5W)

- 1 analog multiplier rated at 170/mW
- 1 variable resistor for DC voltage control: .005W
- 1 turning knob for user control of voltage

Cost:

- 1$ for Analog Multiplier
- 5$ for Resistors
- 4*$0.50 for OPA

ADSR

After all six voltage signals from the guitar have been mixed together, an ADSR envelope will function as a variable affects module for the overall voice of the guitar, or timbre, allowing a wide range of instrumental options via guitar interface.

ADSR stands for Attack Decay Sustain Release. These are characteristics that differentiate one musical instrument from another. They are unique to all instruments playing the same note in the same key. By varying the ADSR, a user can generate sound effects that are similar to common instruments or create new sounds that have not even been imagined. In this project, the sounds created will be entirely experimental.

The ADSR is an envelope detector. Based on voltage settings of each parameter (Attack Decay Sustain Release), a signal will be generated that follows the shape of the ADSR with the same frequency characteristics of the output of the mixer. The ADSR generator will be voltage controlled by a user via turning knobs, and requires a gate voltage and a trigger voltage. The gate signal, or turn on voltage, is a switched socket connected to the system dc bus. The trigger signal is fed into an IC monostable timer such as a 555. When the clock is triggered, the ADSR cycle begins through an RC network with a characteristic pulse width that is determined by the components of the circuit. On the leading edge of the trigger, the ADSR cycle begins with attack and decay profiles as determined by the user. On the falling edge of the trigger, the cycle switches to the sustain and release of the sound. Each time the clock is triggered, the ADSR cycle repeats itself. The output voltage can be used as a control voltage for any further voltage controlled devices such as VCF's and VCO's.

Specs:

- Timer IC
- Low powered transistors
- Low powered operational amplifier
- 15V Power Supply
- 10mA Supply Current

Cost:

- 1$ for L7555 time
- 3$ for transistors
- $ for Resistors and Capacitors
- $0.50 for OPA

Total: $8.00

DSP Post Effects

Before the mixed signal is sent to the user's amplifier, it will be sent into a digital signal processor which to add some effects to the basic sound of the synthesizer. The general list of effects to be performed by the DSP are delay, reverb, chorus, phaser, and others if time allows. The delay effect will simply hold the incoming signal in memory for a selectable amount of time and release it after that time is passed. The user may select the amount of mix between the “wet” and “dry” signals, “wet” being the delayed signal and “dry” being the unaltered signal. The “wet” signal may be fed back into the delay module at a selectable ratio called “Repeats” which determines how many times the echo is played back until its volume has decreased to below a threshold, where it is ended. Reverb is a simulation of the natural reverberation, or echo, of a sound within a space, such as a hallway. It may be digitally synthesized by using multiple short delays with feedback that allows it to decrease in amplitude after a short amount of time. Chorus is the simulation of multiple instruments playing in unison. Realistically, these instruments will be slightly off in pitch and time, which seems to “thicken” and “enrich” the sound. To simulate this effect, the input signal will be sent through a slight delay and pitch modulation, which is then mixed with an unaltered copy of the original signal. A low-frequency oscillator (LFO) may also be implemented to modulate the chorus effect with sine wave, which will vary the intensity of the chorus with respect to time. This adds cadence to the effect which further simulates the changing tonality of a group of instruments. Finally, the phaser effect is a “swooshing” sound modulated onto the input signal by passing a copy of the input through a series of all-pass filters, which keep the magnitude but change the phase. Like the chorus, an LFO will modulate the speed and intensity of this phase shifting. The result is a rich, dynamic sound due to the adding and canceling of the “wet” and “dry” signals with respect to time. Each effect is selectable and controllable, using switches and knobs on the face of the main enclosure, or may be bypassed entirely for a “clean” signal to the amplifier.

Specs:

- 12-bit A/D conversion
- 16-bit D/A conversion
- 44.1kHz sampling rate
- LFO rate of 0.5Hz to 20Hz
- Low power 3-3.6V operation voltage
- 40 MIPS
- 16kB RAM

Cost:

- $10-12 for dsp chip and shipping
- MPLAB C Compiler for Academic Use is Free