VRS-1

Guitar Synthesizer

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1. Introduction

1.1. Executive summary

In the universe, there is no limit to sound; there exists frequencies which human beings could not fathom. Music has been around since the spawn of humanity and will continue to awe and inspire generations to come. Music is known to have very strong effects on the way a persons brain processes, stores and retrieves information. Music can make or break a day. The evolution of music is reaching a peak of expansion and possibilities because modern day technological revolution. By experimenting with unusual tones of musical instruments and generating unique envelopes of sonic character, a musician has the power to broaden the sensation which satisfies his passion.

Motivation

The world of synthesizers is primarily filled by keyboard-based instruments that utilize analog and/or digital circuitry to create a sound scape over which the musician has full control. This control is part of what is so attractive about synthesizers. It allows for the ability shape sound by ways that are only limited by the imagination of the circuit designer and the musician. Keyboard synthesizers have evolved from purely analog monophonic tone generators to being fully polyphonic, digital-orchestra-at-your-fingertips devices. Somewhere in between, the guitar synthesizer appeared, acting as the little cousin of the keyboard synth, trying to keep up technologically and musically. By creating this product, players who are already well trained on a guitar but lack knowledge of the keyboard can create the same sounds heard on modern day analog synthesizers, including all the popular options such as selecting waveforms, filtering, attack, decay, cutoff and resonance, in a package that the musician is already familiar with.

The guitar is a vastly different beast from a keyboard. Converting a keyboard's inputs into sounds are a matter of reading which key is being pressed and generating an appropriate tone. But the guitar brings new challenges which require far different approaches. Strings are hit by the player's pick or even bare fingers and forced to resonate at some desired note. This noisy, mess of frequencies has to somehow translate into a fully-controllable tone. Strings may bring challenges to the design, but they also allow for musical freedom that only a stringed instrument such as a guitar can offer. The ability to bend notes, for example, doesn't exist natively on a keyboard. Creating a polyphonic synth is yet another challenge that pays off with the ability to play chords, or let certain notes ring while playing new notes. New sounds can be created by converting information about each note into control voltages that can in turn be modulated, shaped, and shifted to add substance to the music. This is what will add to the functionality of a guitar, and give the musician one more weapon in their arsenal of sound. The ability to shape and manipulate several parameters of an analog signal can result in many new and unique sounds that are unique to this specific design.

Goals

VRS-1 should possess 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. VRS-1 should generate 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 should be organized in a way which they can be easily identified and accessed in a live setting. The synthesizer should be small enough in size and weight that it can be transported without difficulty and set up at any venue with ease.

VIRS-1 should be 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. Over time the analog circuitry will experience some drift, so the ability for the tracking to be tunable is key for the products lifetime.

1.2 Design Summary

Each of the different systems within the VRS-1 are shown in figure 1.2-1 and described below as functional blocks which appear in the block diagram of the synthesizer. This section is intended as a brief overview of each stage, describing its purpose, uses, and general design. Every design will either wise be controlled or control another block in the design.

Multichannel Pickups

For the synthesizer to be polyphonic, meaning possessing the ability to create multiple voices, each string of the guitar must have its own pickup. This is because processing a complex signal becomes a lot more difficult than a mostly pure tone with harmonics. At any given time the every string on the guitar can only produce one note at a time. The pickup will be similar to standard guitar pickups, except for having separate windings located under each string. Small gauge insulated wire will be wrapped around cylindrical magnets which will slightly magnetize the portion of the nickel-coated guitar string suspended just above it. When the string is plucked, the corresponding change in the magnetic flux within the coil will create the electrical current directly proportional to the vibration of the string. This method is important because each string produces (in theory) one fundamental frequency when plucked and is scaled accordingly to match the intervals of a 12 note musical system. Each frequency must be sensed independently and used to produce the desired sound.

First-Stage Filters

The signals straight out of a guitar pickup are noisy, and filled with harmonics that will add a level of uncertainty to the next stage, which needs to see a single frequency input. Simple Sallen-Key low-pass filters will make up this stage, which act to boost the frequencies in the lower end of each string, while cutting off higher ones that are usually present with any guitar pickup.

Frequency-to-Voltage Converter

Using the signal straight from the guitar to produce the oscillations is a basic way of creating a pseudo-synthesizer, but doesn't allow for the control over the sound that is achieved when coding the frequency information as a voltage. This voltage can then be sent to later stages and modulated, shifted up or down, resulting in a shift in frequency, and other sonic effects. This stage will utilize the LM2907 tachometer chip which outputs a voltage proportional to an input frequency. The voltage is scalable, and the device suffers only from an oscillating output voltage if the spectrum of the input signal does not mostly consist of one spectral line, or frequency. This however, is taken care of by the first-stage filters.

Voltage-Controlled Oscillator

The Voltage-Controlled Oscillator, or VCO, is the heart of the system. It receives a voltage signal that specifies a certain frequency to produce, where the voltage signal is receives is directly related to the frequency of a guitar string. This stage can also create different waveform shapes such as sinusoidal, triangle and square waveform and mix them as the user specifies. By adding and subtracting these types of signals different harmonic and overtones are produced giving the sound a lot more depth than a pure sine wave oscillation.

Attack, Decay, Sustain, Release Envelope Generator

The ADSR is responsible for generating a desired envelope signal with which the static-amplitude VCO outputs will be modulated to simulate the envelope of a guitar or other instrument, depending on the settings chosen. This stage takes in two signals from each string channel: the trigger and gate. The trigger tells the ADSR when a new note is hit, and the gate tells it how long it is to be held. The envelope signals are generated from an MSP430 microcontroller which reads in the trigger, gate, and various potentiometer inputs that are used for varying the shape of the envelope. The envelope signal will be in the form of a control voltage that will be sent to the voltage-controlled amplifier.

Voltage-Controlled Amplifier

An important part of shaping the sound is varying the amplitude in such a way as to simulate the characteristics of real instruments. The VCA takes on the task of modulating the signals from the VCO with another control voltage. Once again, the LM13700 will be used for the job. Its ability to use changes in bias current linearly is important to having full control over the amplitude of the wave.

Voltage-Controlled Filter

One of the most important concepts in music is timbre, which can be described as the frequency content of a certain sound wave. Changing this spectrum can be done with a filter that allows for a cutoff that is variable with a control voltage. A Current-Generalized Impedance Converter Biquad with a voltage-controlled resistance circuit designed around an LM13700 will make up the circuitry of this stage.

Low-Frequency Oscillator

Since rhythm is essential to music, a low-frequency oscillator is used to modulate certain stages to a selectable "beat". This can vary from a few Hertz up to about 30Hz. The oscillation is used as a control voltage and can be coupled into the VCA for a tremolo effect which varies volume, the VCO for a vibrato effect which changes pitch with respect to time, or an arpeggiator, which uses the LFO signal to play different notes in a scale at a certain rhythm.

Arpeggiator

The arpeggiator uses the LFO and two selectable gates to change the static control voltage from the F/V converter and send the VCO a time-varying signal. This signal switches from the voltage associated with the fundamental frequency to the musical third, fifth, and eighth. Changing each gate threshold alters the time within the wave that the voltage takes to change to the next note in the arpeggio.

Mixer

A dual channel mixer will be used to mix two selected waveforms in order to enhance the strength of the signal received as well as generate additional overtones available for modulation. Proportional control will be added in order to isolate one of the signals from the other if so desired. A six channel mixer will be used prior to audio amplification. Also, for added control, each channel will have its own proportional control in the form of volume control on the panel in the form of six slider potentiometers in the style of "faders" used in mixing consoles for live or studio sound engineering.

Power Supply

To supply power to each stage, a switching power supply unit will be built that can handle up to 3W and be able to supply both positive and negative voltages. The supply rails will extend from -12V to +12V at their maximum to allow for enough headroom for a low distortion signal path. A switching circuit will be used for stability, since any voltage changes may introduce changes in the various control voltages throughout the circuit. For instance, if the power supply voltage drifts, there may be a resulting drift in the frequency of the VCO's output. Stability in this case is key to designing a musically pleasing device.

All of the functions are shown in figure 1.2-1.

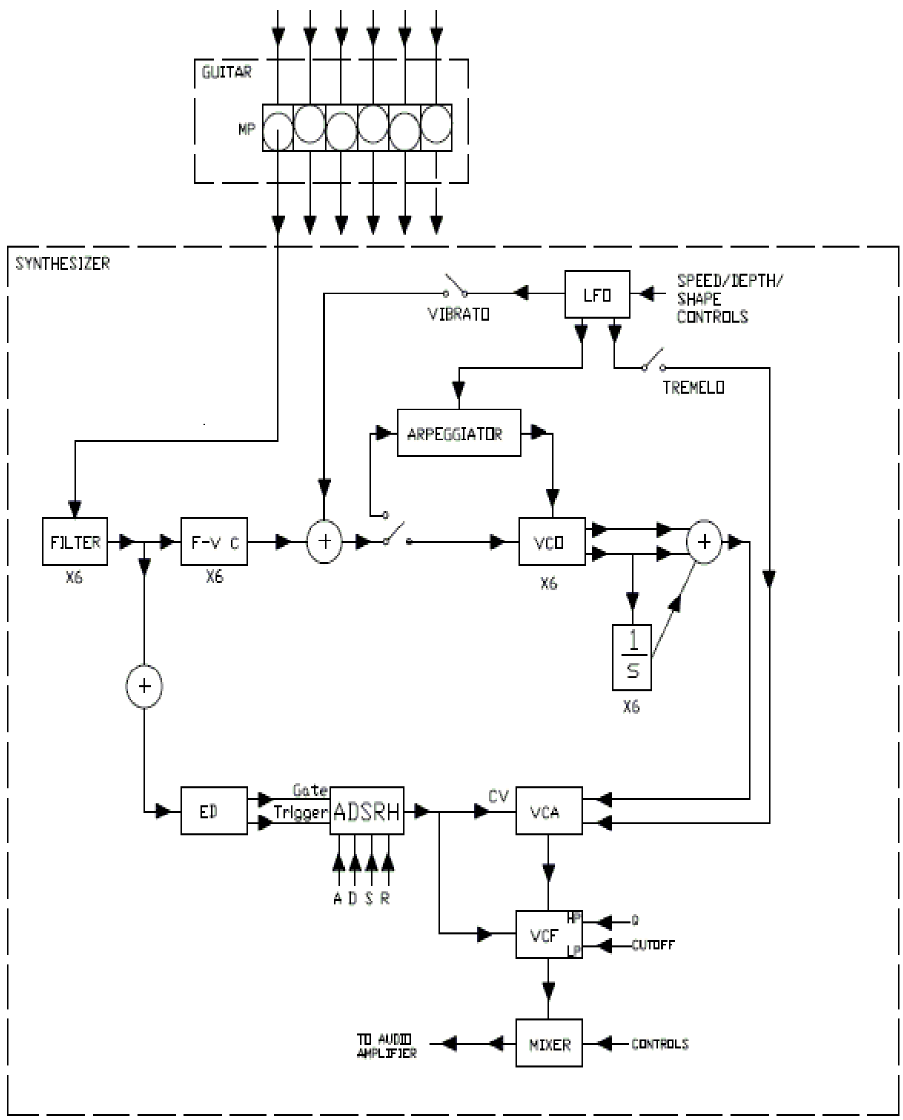


Figure 1.2-1: Synthesizer block diagram

2. Research

This section explores the different technologies available to design the individual components of the synthesizer.

2.1 Guitar Theory

Throughout the sequence of synthesizer design, the relationship between frequencies and electric signals must be preserved in order to maintain a consistent and familiar interface for guitar players; therefore, the underlying theory of a guitar will be explored in order to introduce concepts which will determine the necessary electronic design in order to reproduce the nature of an electric guitar.

2.1.1 Harmonics

Upon being plucked, a guitar string will vibrate with a number of frequencies: the lowest being the fundamental frequency, with the other, higher frequencies called overtones. The overtones exist at integer multiples of the fundamental frequency, at locations on the fret board known as nodes. The nodes are points on the guitar fret where the string being plucked does not vibrate. Figure 2.2-1 shows the locations of the natural harmonics that exist along the length of the guitar neck.

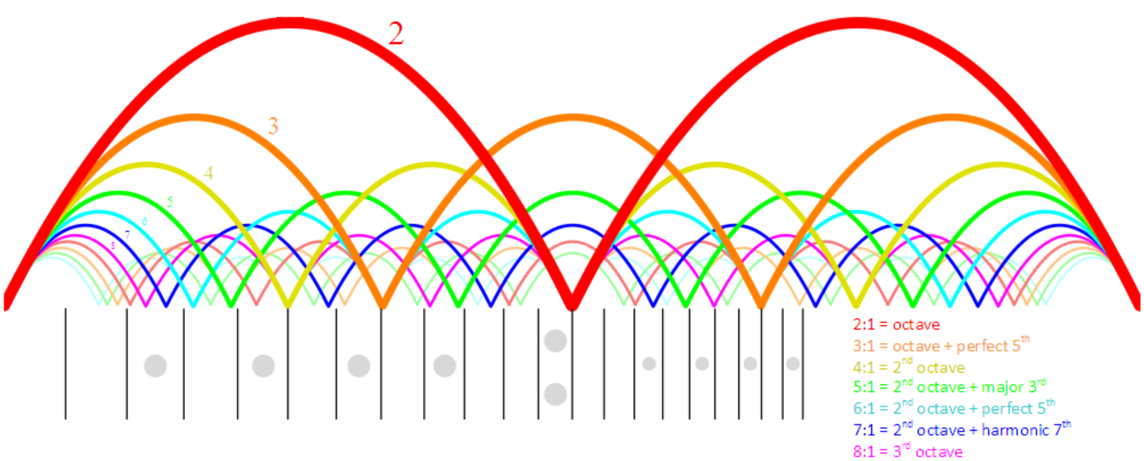


Figure 2.1-1: Natural harmonics on a guitar fret board (Permission Pending)

Once a string is plucked, all of the overtones in action will sound like a single tone, but they enhance the intensity of the overall sound. A guitar player will often use special techniques to play the harmonics which are otherwise unreachable by the fret board alone.

The lifespan of each overtone is not constant for a given note. They will decay at different rates during the duration of the note, where each overtone involved will rise or fall independently of the note being played. The fashion in which the overtones decay or sustain contributes to the timbre, or uniqueness, of the instrument being played. A synthesizer would like to take advantage of these naturally occurring overtones and manipulate them in order to produce unfamiliar, experimental sounds.

As part of the synthesizer design, an ideal sine wave will be generated which will be defined only by the fundamental frequency, and filtered of all higher harmonic overtones for the purpose of remaining within operating range of the electronic components involved. An ideal sine wave has no partials, which is why it is often described as a pure sound, since there are no added overtones to the note being played. Since there are no overtones to the sine wave, there is not an opportunity to filter and therefore change the characteristics of the sound, which is a primary feature of a synthesizer. For this reason, square waves and triangle waves will also be introduced, which consist of odd harmonics that exist by contributing to the shape of the waveforms.

Figure 2.1-2 shows the frequency spectra for each of the three aforementioned waveforms. It can be noted in the figure that the triangle wave and square wave both have odd harmonics, but the roll off rate of the harmonics is greater for the triangle wave.

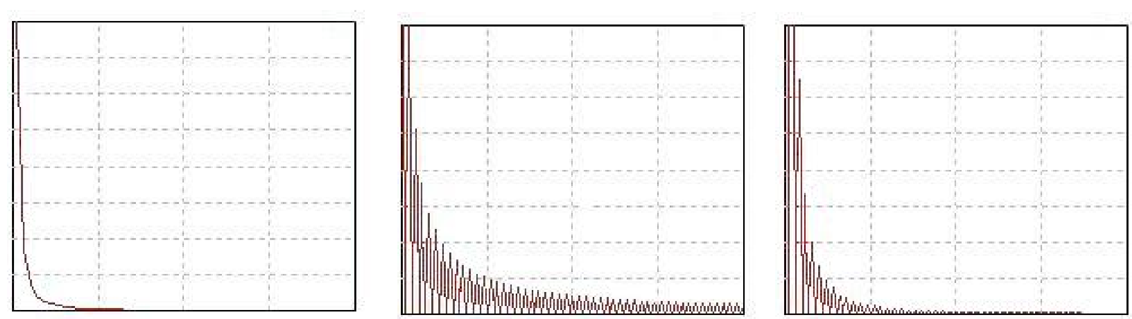


Figure 2.1-2: Frequency Spectra of Sine, Square and Triangle Waves

Following modulation and filtering of the generated waveforms, the overtones extracted during the frequency-to-voltage conversion of the notes being played will be replaced, since the combination of overtones and the fundamental frequency will add much “color” to the sound, and will resemble the traditional tone of an electric guitar. As shown earlier in figure 2.1-1, the major 3rd and perfect 5th exist in the overtones of a guitar. A given note combined with these two harmonics form the basis of any chord progression, and can be sequenced as an arpeggio, which is one of many features of modern synthesizers. When more overtones are available, there are more options during synthesis.

2.1.2 Fret Board Frequencies

A musical octave is the interval between two different pitches of a single note, either half or doubled the frequency of the base note. On a guitar, the fret board is divided in steps of half notes, with a frequency increase or decrease of $2^{1/12}$ per step. The fundamental frequency range is plotted in 2.1-3 which reveals the exponential nature of the frequencies on a fret board as the note being played travels up and down the guitar between 20 frets. A "0" on the x-axis indicates an open note. It can be noted from the figure that most frequencies are repeated among the different strings at different positions on the fret board. For a complete chart of guitar notes and frequencies, reference Appendix A.

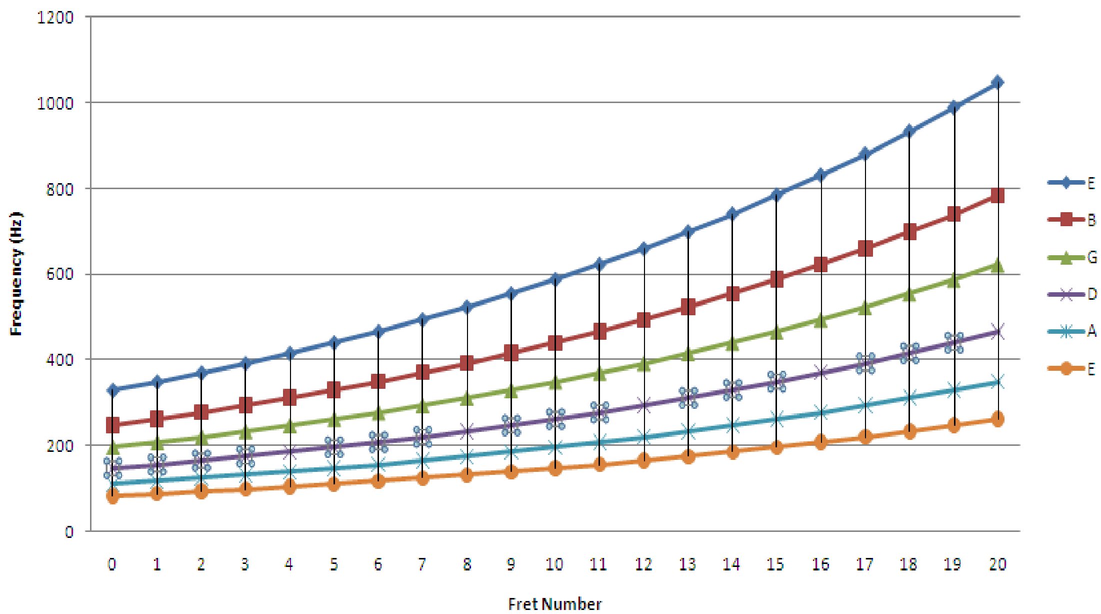


Figure 2.1-3: Range of Guitar Frequencies on a Fret Board

<u>Exponential Conversion</u>

If scaled appropriately, the frequency characteristics of each guitar string will increase linearly as the note being played travels up and down the guitar. The conversion takes place with the following equation, by converting the log base 10 of each frequency to the log base $2^{1/12}$.

$$\frac{\log_{10} f}{\log_{10} 2^{(\frac{1}{12})}} = \log_{2^{(\frac{1}{12})}} f$$

where f is the fundamental frequency of any given note.

Figure 2.1-4 below shows each guitar string as a linear function of frequency over the fret range of a guitar neck if each frequency is scaled logarithmically by the base $2^{(1/12)}$.

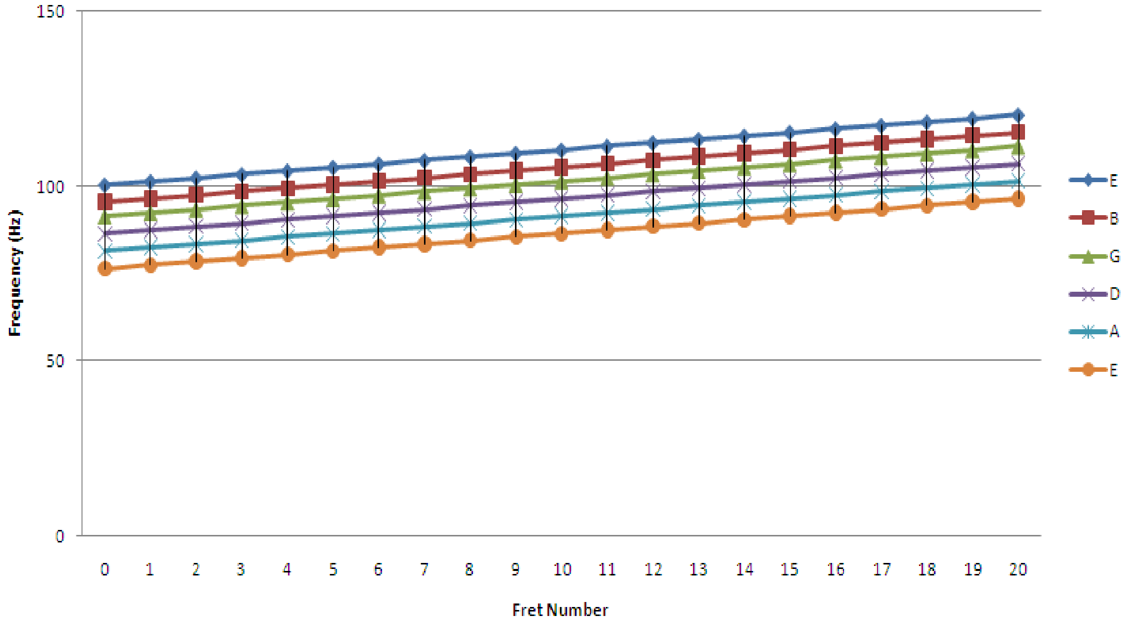


Figure 2.1-4: Linearized frequencies of a guitar fret board

After observing the two figures above, it is noted that the complete frequency range of the guitar, prior to scaling, ranges from 82Hz to 1046Hz. This is a difference of 964Hz between the maximum and minimum frequencies of notes on a fret board, with a scaling factor of $2^{1/12}$ between each note. On the other hand, if a given frequency is converted to log base $2^{1/12}$ prior to driving a frequency to voltage converter, a smaller range of 76Hz to 120Hz can be achieved, with a difference of 1Hz between each note on the guitar fret. This could potentially correspond to a discrete change in voltage on the output of the frequency-to-voltage converter. By implementing the conversion, it is possible that greater precision could be achieved and more redundancy could exist in designing the circuits that the voltages will drive. Also, if proportionality between frequency and voltage becomes a challenge, scaling the frets logarithmically with a microprocessor could be an option to maintain a constant relationship between frequency and voltage. Prior to mixing and audio amplification, the frequencies could be un-scaled to resemble the nature of the tones produced by a guitar.

2.1.3 Magnetic pickups

Magnetic pickups work on the theory of magnetic induction and magnetic flux. A cylindrical shaped magnet is wound with fine copper wire a few thousand turns. The more the windings the higher the output swing voltage, when the magnetic flux of the magnet is changed an induced current passes through the copper wire. The number of turns will determine the magnitude of the output. If a high output (hot) pickup is desired, this requires a few thousand turns. This gives the tone of the sound coming out to be a very warm and fat sound. Pickups with a less amount of windings have a thinner more twangy sound. Personal preference determines what type of sound the player would like when playing guitar. Within the first 100ms of plucking a string the volume decreases by more than half. This is known as the envelope of the wave.

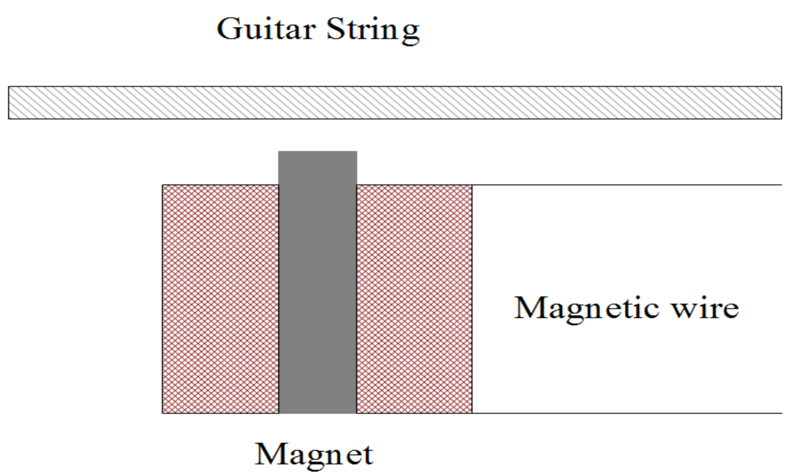


Figure 2.1-5 Magnetic guitar pickup

This same type of figure can be drawn as an equivalent of having an inductor in parallel with a capacitor and terminated with a resistor. The pickup in a way acts as a bandpass filter passing frequencies within an audible range.

The more amount of windings around the magnet the more active the response will be. Too much winding on a single coil can cause some issues. A problem that

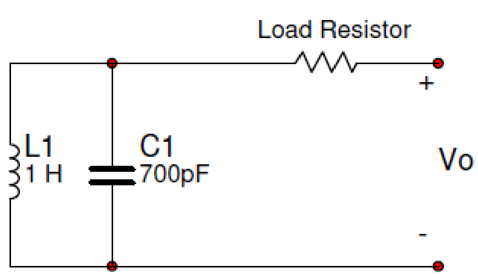


Figure 2.1-6 Pickup equivalent circuit

arose when engineers were designing these pickups is that the coils were receiving a great amount of interference noise from everyday electronic devices, transformers and other devices that would cause the pickups to receive electromagnetic interference. One way engineers tried to solve this issue was to shield the housing the pickups were inclosed in. But the single coil pickup instead acted like an antenna receiving a buzz at around 55 Hz. In order to solve this problem two electrical engineers developed the humbucker. Equivalent circuit to the humbug circuit is seen in Figure 2.1-7. The design consisted of two single coil pickup with opposite magnetic polarity. Since the interference was common mode it would be equally induced on both of the pickups and since they are 180 degrees out of phase and wired in series the common mode noise is eliminated and the induced current is doubled.

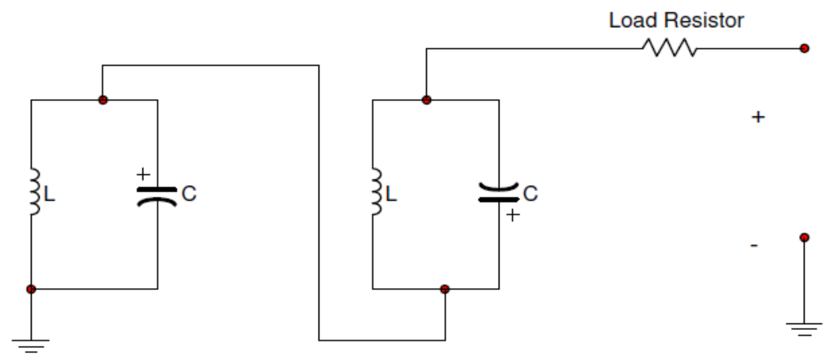


Figure 2.1-7: Equivalent circuit to a humbucker pickup

2.2 Signal Synthesis

The idea of signal synthesis is based on the idea that an analog signal can be used as an audible tone with each waveform having its own distinct voicing. In order to do this the board must be able to take an input signal and mimic two characteristics, the pitch and the volume. Pitch being frequency of the note being played and volume being the amplitude of the waveform. In order to do this accurately and without using a micro controller is to acquire these characteristics in an analog fashion. For example if the guitar string is resonating at 440Hz then the board should be able to read this frequency and create the same type of frequency. The following sections will discuss the theories behind how a tone will be generated from the guitar signal and then duplicated with oscillators.

2.2.1 Multichannel Pickups

For the purpose of this design the tone of the output is not as important as the information it carries. From the original signal there are several characteristics that need to be noted to be used on the schematic. First there is the gate, like the name says this senses weather there is a signal passing through not. Thus the name gate refers to an open or closed switch. Second comes the trigger, this is similar to the gate, but is triggered every time there is a new attack of higher amplitude. If the guitar player is to play two chords consecutively there would be one gate for both the notes and two separate triggers. This is information is to be used by the ADSR later in the schematic. Third the envelope of the wave is going to be used for several different filters; this measures the peaks of all the different amplitudes and plots it out in form of a voltage output. Lastly the pickups need to receive a key piece of information, the frequency of the signal. In order to get high tracking accuracy this signal needs to be passed through a band pass filter attenuating most resonant frequencies that are not found within the range of that specific string. The first design that is going to be used is a single coil design, although early designs showed this to have a lot of noise this issue will be solved with the band pass filter. It would be unnecessary and an arduous task to make humbucker pickups when the final output is not at all dependent on the tone of the guitar.

Because a normal guitar pickup design mixes all 6 strings into one complex signal a single string pickup is desired for this design in order for the board to be able to read an polyphonic (more than one note) signal. Roland has designed a guitar pickup called GK-3 which is used for their model of compatible synthesizer guitar stomp boxes. This design uses a normal instrument cable, this could have been done by signal modulation and having a receiver that picks up a much wider bandwidth and adding all the six signals together then later subtracting them. An easier design is to have six individual pickups that send the signal using a 12 pin computer monitor cable. Before the signal goes into the filters they are going to have a gain added to them which makes the signal an active signal. The target amplitude is to have the signal be transferred within a 1 volt magnitude.

Materials

The type of magnet that is going to be used is a aluminum nickel cobalt because this has been proven to be very responsive and inexpensive. AlNiCo pickups are among the most popular types of magnets used in the magnetic guitar pickup industry. A rod with the diameter of .1875 inches and a length of 1 inch will be wound with copper wire (#42 gauge) a few hundred times. The first trial will have a total of 500 turns and the target magnitude is .5mV. Once the turns are in place the pickup can be coated with a layer of wax to keep the turns from moving around and loosening over time. The two outputs of the magnetic wires will be terminated with a load resistor which will determine the output signal. Different loads need to be tested to see which one would provide the best current flow and voltage difference. The usual capacitance of the pickups depends on the capacitance of the coils and the capacitance of the wire. The usual capacitance ranges from 300pF to 1000pf. The inductance of the windings is generally 1H.

Spacing

The spacing of the magnets will also be variable that needs to be tested. A bad placement of the magnet on the string would result in the magnet picking resonant frequencies or attenuated areas of the string. The following image shows how placement of the pickup effects the response. Since the tone output is not as important as the frequency being carried in the signal, the pickup should be placed near the bridge of the guitar where the harmonics and overtones are kept at a minimum. And in order to reduce cross interference noise all of the magnets should be lined up with the same polarity

2.2.2 Filtering

Before the pure guitar signal is seen by the frequency to voltage converter, there must be some initial filtering to reduce the higher harmonics associated with a plucked string. In a perfect world, the guitar string, when vibrating, would produce a single fundamental frequency. However, the case is different. The "timbre" of a guitar, referring to the sound, is related to its frequency spectrum, as in any instrument. The shape, wood type, string material, winding, and other factors effect its spectrum. Most guitars are equipped with a "Tone" knob, which is simply a passive RC low-pass filter with a variable cutoff. Fig 2.2-1 shows the spectrum of the open note, E, on the low E string (82.4 Hz fundamental) plucked on a Squire Strat guitar. The pickup closest to the center of the string was chosen, as well as the "Tone" knob rolled off as much as possible to reduce as much higher harmonic content as possible

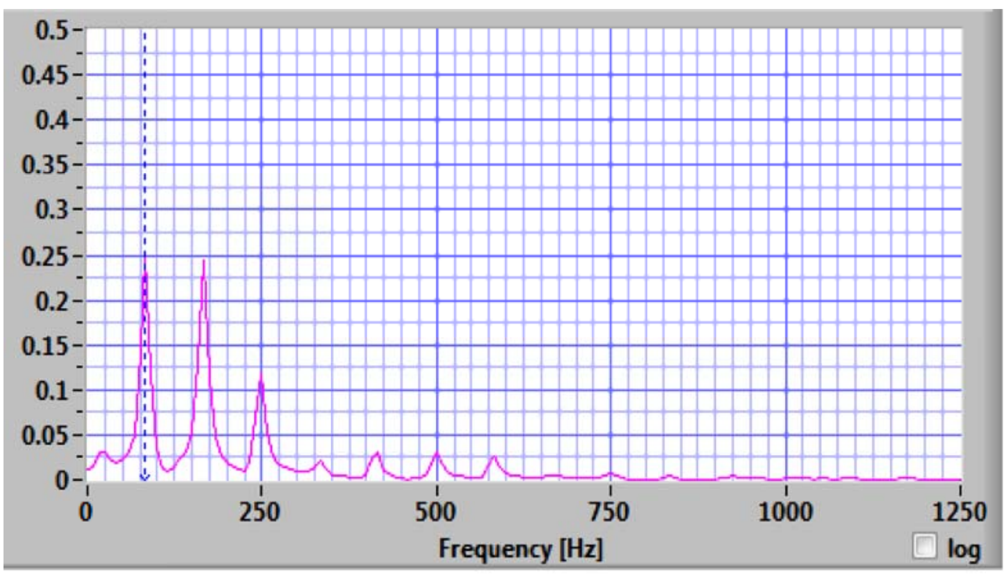


Fig 2.2-1: Spectrum of an open E on the low E string.

Clearly, the second harmonic is similar in amplitude to the fundamental. This could potentially create problems with tracking the correct frequency. The frequency to voltage converter may in incorrectly choose the octave, or even second octave, resulting in undesirable output. The output would in theory be the same "note", E in this case, but may not correspond exactly to the octave being played. It is also possible that the F/V converter's output may ramp up to the level associated with the octave, resulting in a non-stable frequency output of the VCO. Fig 2.2-2 shows the same string being plucked, but an octave up on the fret board. The note played at the 12th fret doesn't produce a strong second harmonic or third harmonic like the open not does. This is beneficial, and may help with the filter design, since the cutoff doesn't need to be substantial after the octave.

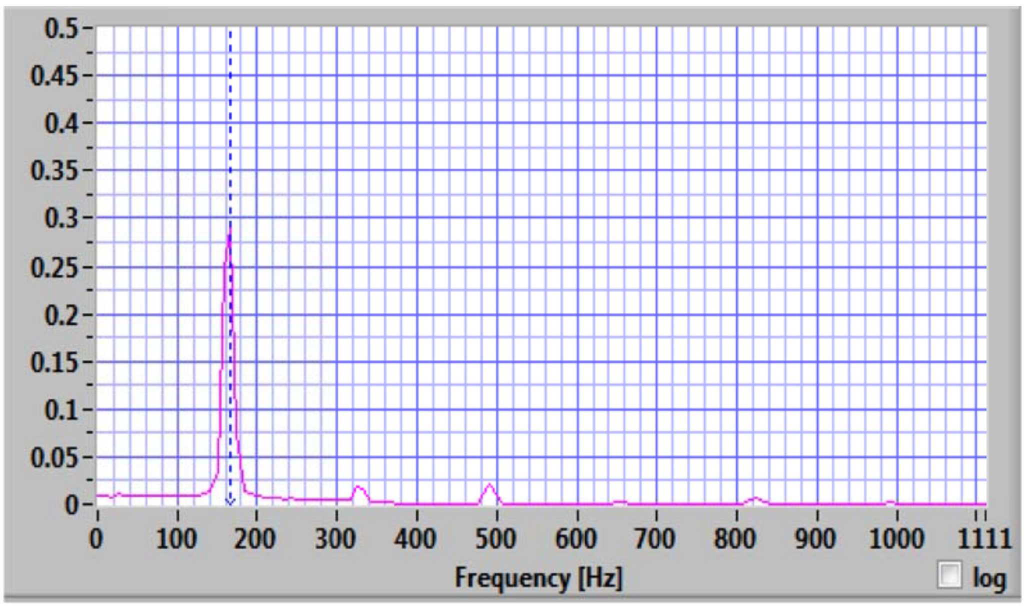


Fig 2.2-2: Spectrum of 12th fret of low E string.

The filters should reduce the second harmonic, but not too much, since guitars allow for playability past one octave on each string (sometimes up to the second octave). The threshold of the F/V converter must be considered when designing the filters so that the octave may still be picked up.

<u>Comparison of Topologies</u>

In order to select appropriate filters, several different circuits will be compared. The parameters that will be of importance to these filters are size, quality factor, and simplicity of tuning. Size is important since each string's signal will have its own filter. The availability of dual and quad op-amps will assist in keeping the board space to a minimum. A good quality factor will ensure the reduction of the second harmonic. Finally, the filter should be tunable by trimming some resistance with a potentiometer. The filters could then be tuned for each string to make up for tolerances. The single op-amp Sallen-Key low pass filter is a possible candidate. It is described by an RC network built around an amplifier. It is actually capable of high pass and band pass by reconfiguring the RC network accordingly. Fig 2.2-3 shows the circuit in low-pass configuration.

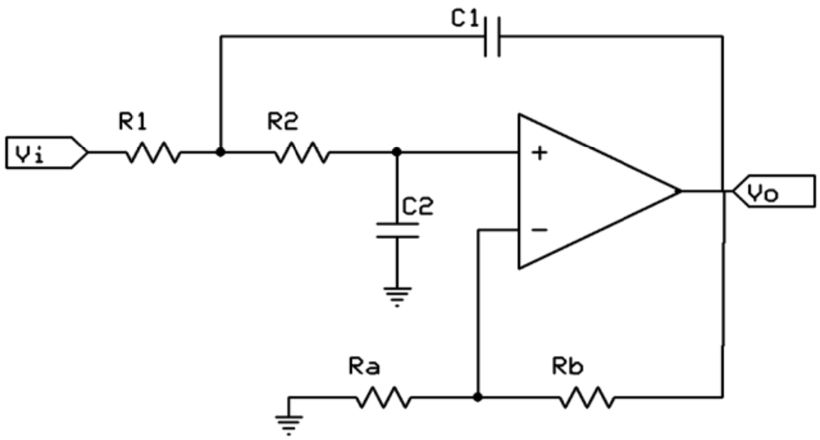


Fig 2.2-3: Sallen-Key Positive Low Pass Filter

The transfer function associated with the filter is

$$T(s)=\frac{\frac{K}{R_1R_2C_1C_2}}{s^2+[\frac{1-K}{R_2C_2}+\frac{R_1+R_2}{R_1R_2C_1}]s+\frac{1}{R_1R_2C_1C_2}}$$

To simplify the transfer function, and hence the quality factor, Q, and cutoff, ω_0, the resistors and capacitors are defined as

$$R_1=R_2=R \quad \text{and} \quad C_1=C_2=C$$

which simplifies Q to

$$Q=\frac{1}{3-K}$$

and ω_0 to

$$\omega_0=\frac{1}{RC}$$

where K is the amplifier gain

$$K=\frac{R_b}{R_a}$$

The resistors would need to be tuned to match each other accurately. Also, the amplifier gain, K, can be tuned with either R_a or R_b, which in turn will affect Q.

Another single-amplifier biquad is the Rauch Lowpass Filter. It boasts features such as low component sensitivity, easily adjustable parameters, great stability, and, being a single-amplifier filter, keeps power and size to a minimum. Fig 2.2-4 shows the circuit in low-pass configuration.

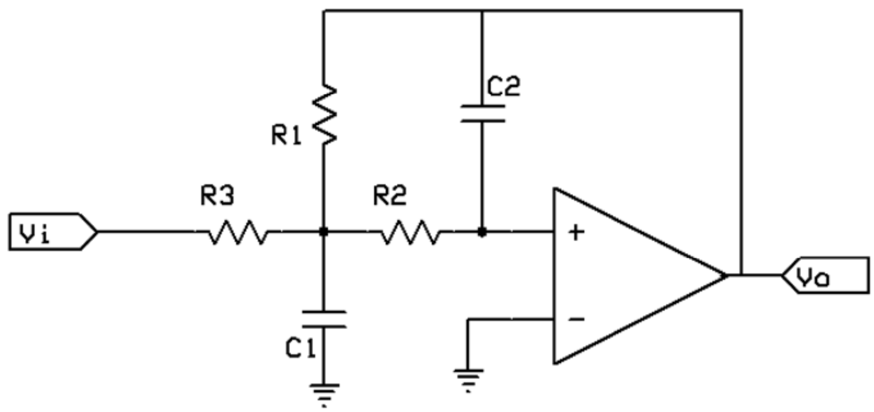


Fig 2.2-4: Rauch Low-Pass Filter

The transfer function associated with the filter is

$$T(s)=\frac{-\frac{G_2 G_3}{C_1 C_2}}{s^2+s\left(\frac{G_1+G_2+G_3}{C_1}\right)+\frac{G_1 G_2}{C_1 C_2}}$$

The dc gain and pole frequency are

$$H_0=\frac{G_3}{G_1} \qquad \omega_0=\sqrt{\frac{G_1 G_2}{C1\,C2}}$$

with a quality factor of

$$Q=\frac{\sqrt{C_1/C_2}}{\sqrt{G_1/G_2}+\sqrt{G_2/G_1}+G_3/\sqrt{G_1/G_2}}$$

<u>String-Specific Filters</u>

Since each string covers a different span of frequencies, each filter must be designed for proper cutoff. Table 2.2-5 shows the strings and their respective open note frequencies, or the frequencies at which the string is plucked under standard tuning without pressing down on any frets.

String (open note)	Frequency (Hz)
E (Low)	82.4
A	110
D	146.8
G	196
B	246.9
e (High)	329.6

Table 2.2-5. Standard guitar open string frequencies.

Each of the filters looked at have similar pole frequency equations, making designing the string-specific filters a matter of finding suitable resistor and capacitor values. Some error due to tolerances and availability of specific values will have to be considered and reduced as much as possible. The capacitors used will be chosen first, and metal film resistors which have a tolerance of 1% should be used in this portion of the circuit.

Some guitarists prefer to use non-standard tunings, the most popular being “drop D”, where the low E string is “dropped” to a D, which corresponds to a frequency of about 73.416 Hz. If time permits, the design may include a switch which changes the pole frequency to account for this lower note, so that the guitar player may choose their own tuning of the low E string.

Op-Amps

Choice of operation amplifier integrated circuits for the initial filtering circuitry relies much more on parameters such as cost, size, and noise than other considerations such as bandwidth. The absolute minimum gain-bandwidth product ω_t depends on the highest string's open frequency of 329.6 Hz and a rough estimate of the required DC gain of 5. This corresponds to a minimum ω_t of about 1.6 kHz. To be safe, this could be raised to 5 kHz. Given today's requirements for much higher bandwidth, this should leave most options open for ICs. Considering there will be six op amps for the initial filtering alone, not counting op amps in later stages, size and cost are two important parameters. Clearly, quad op amp, surface mount packages will be ideal. Also, noise will need to be avoided as much as possible without raising the project's overall size and price considerably. The LM324a from Texas Instruments seems like a viable option. It offers low input bias and offset parameters, which will be important in noise avoidance. Cost from Mouser is $0.49 for the low-profile SOIC package. Table 2.2-6 shows relevant parameters of the LM324a.

Parameter	Min	Typ	Max	Units
Input Offset Voltage		3	7	mV
Input Offset Current		2	50	nA
Input Bias Current		-20	-250	nA
CMRR	65	80		dB
Supply Voltage Rejection Ratio	65	100		dB
Supply Current		0.7	1.2	mA
Supply Voltage	± 1.5		± 16	V

Table 2.2-6: LM324a Relevant parameters

2.2.3 Frequency to Voltage Conversion

All synthesizers that include voltage-controlled oscillators require some sort of initial voltage stage in order to feed an appropriate signal into the VCO to get a matching output frequency. How this section works is dependent on what the input controller looks like, or how the instrument itself is played. Take the example of a keyboard synthesizer: each key may be designed as a simple binary switch which adds in resistance to a voltage divider, or some other similar scheme. This in turn sends a specific voltage for each key to the VCO and in turn creates a frequency relative to that key pressed on an old fashioned string-and-hammer piano. In this case, there is no real input frequency, rather, the keyboard simulates a piano's keys, and the corresponding frequency is realized through circuitry.

The case using of a guitar as a controller is different task entirely. There are six strings, each capable of up to 24 notes (on some guitars), instead of 88 keys, each directly assigned to a single note. It would be possible to emulate a keyboard style using a guitar body with keys on the neck, each sending it's own trigger signal, but that would require manufacturing of a special instrument. The goal of this project is to allow for minimal modifications of the user's own guitar as possible. The installation of a new pickup is a fairly common guitar modification, and should not hinder serious players. That being said, the frequency to voltage converter (FVC) must be able to read an input frequency from the pickups and output a predefined, corresponding voltage to the VCO.

There are some important specifications/considerations before designing an appropriate circuit to do the job:

- Is the output voltage a linear function of frequency?
- Size (there will be one for each of the six strings)
- Low power
- High accuracy

LM2907/2917

Through some research, an integrated circuit was found which does exactly what the FVC requires. The LM2907/2917 requires minimal components and comes in an 8 or 14 pin soic package. It uses a charge pump system to convert frequency to voltage and allows for ground reference or differential input. There is a built in comparator with hysteresis to assist in eliminating jitter, especially from the possible existence of a second harmonic. Also, the output goes to ground when the input is at zero frequency. It has a very easy to use output equation:

$$V_{out} = f_i V_{cc} R_1 C_1$$

where f_i is the input frequency and V_{cc} is the supply voltage. Some importance specs are tabulated in table 2.2-7.

Parameter	Condition	Min	Typ.	Max	Unit
Supply Voltage				28	V
Supply Current			3.8	6	mA
Input Threshold	V_{in}=250mV_{pp} @ 1kHz	±10	±25	±40	mV
Hysteresis	V_{in}=250mV_{pp} @ 1kHz		30		mV
Output Current	$V_2 = V_3 = 6V$	140	180	240	μA
Linearity	f_{in}=1kHz,5kHz, 10kHz	-1	0.3	1	%
Frequency				10	kHz

Table 2.2-7: LM2907 Parameters

Overall, this device looks like an exceptional candidate for the FVC. Linearity and supply current seem to fit within specs. The maximum frequency is well beyond the natural frequencies encountered by guitar strings. Also, the price ranges from $0.81 to $1.86 on Digikey, depending on package and chip version. One thing to always keep in mind is that most of the sections in this project will be multiplied by six, so cost is a very important issue.

Voltage Scaling

Depending on how the VCO circuit is designed, the input to it may need to be linear, such as 10mV/Hz, or logarithmic, such as 1V/octave. The 1V/octave is a common spec for modular synthesizer VCOs, allowing musicians to swap out different equipment without compatibility issues. These older style synthesizers usually required patch cables to be connected to varying sections of the synthesizer, such as the control voltage from a keyboard into a VCO. The output of the LM2907 is linear, which means that the VCO should preferably accept a linear input. Also, since the Guitar Synth will not be modular in the sense that each section has external inputs and outputs, the 1V/octave standard does not need to be met. However, if the VCO requires a logarithmic input, then one must be designed for it, as long as the size of the final circuit is not bigger than it would be in the purely linear case. However, this does not seem likely.

2.2.4. Wave Generation

To achieve a polyphonic system, each frequency to voltage converter (FVC) will drive a voltage controlled oscillator (VCO) to produce a waveform that will form the basis of modulation and signal synthesis. The voltage supplied to the VCO will be scaled proportionally to the frequency of the note played on each individual guitar string. The VCO should generate a repetitive signal with a frequency varying linearly, as a function of the voltage applied to the VCO. Important requirements include:

- Tuning Linearity – measured percentage of a signal to maintain a linear fit to a tuning curve should be no greater than .02% error
- Tuning range – oscillator should operate for frequencies between 80Hz and 1100Hz
- Output Noise – relationship should be small between output noise current and frequency
- Thermal Stability – oscillator should maintain adequate output at operating temperatures up to 70 degrees Celsius
- Maximum Input Voltage – should be no less than +/-10V
- Power Dissipation – should be less than 500mW

Varactor Diode Oscillator

Harmonic oscillators that utilize variable capacitors (varactor diodes) control the resonant frequency of oscillation by adjusting the voltage applied to the varactor diode. As all diodes act as a capacitor to some degree, a varactor is specifically designed to allow the depletion layer to vary according to the applied voltage. Since capacitance is inversely proportional to the depletion layer thickness, it is also inversely proportional to the applied voltage. The amount of capacitance present in a circuit will determine the resonant frequency at which the DC voltage will oscillate. Hartley oscillators include series inductors, tapped to allow a feedback based on voltage division across the impedance, which provides the right amount of voltage to maintain constant amplitude. An amplifier is included to control the amplitude: if the oscillations produced are too high in amplitude, the bias of the amplifier is increased, and the gain of the amplifier decreases. On the contrary, if the amplitude of oscillation is small, the feedback to the amplifier is reduced, along with the bias, resulting in higher gain.

A varactor diode circuit can be assembled with just a few components but there is no regulation of a relationship between a control voltage and an oscillating frequency. A precise, linear relationship between the two is necessary in order for the frequency to be truly related to the voltage supplied by the frequency-to-voltage converter. The supply voltage carriers the frequency characteristic of every note being played on the guitar, and the oscillator needs to be able to interpret and reproduce the frequency of a note.

Monolithic Oscillators

Voltage Controlled Oscillators are available in the form of monolithic circuits, often at a low price with high reliability within frequency and temperature ranges. They are relatively simple to test and troubleshoot, and are available both through-hole and surface-mounted. With such a large number of components expected throughout this project, size is also a concern, and monolithic oscillators perform a lot of technology in a small package.

LM231

An LM231 VCO is designed with a comparator which drives a one shot timer to operate when a selectable threshold frequency is reached. The schematic is shown in figure 2.2-8. The applied voltage Vi is greater than Vx since the current source is disconnected from the Vx loop. As a result, the comparator goes high and the frequency output transistor turns on, as well as the switched current source. With the switched current source in the ON mode, the capacitor CL will begin to charge and the voltage Vx will build up and eventually exceed the input voltage Vi. When this occurs, the output of the comparator goes low, the timer shuts off, and the voltage Vx will begin to decrease as the capacitor discharges. At the point when Vx is lower than Vi the cycle repeats itself, producing a repeatable frequency output dependent on the control input voltage, Vi. The current flowing out of the capacitor is a function of Vx and the load resistor, RL (Iout = Vx/RL). At the point when current is actually flowing out of the capacitor (because it is discharging), Vx has reached the value of Vin, therefore,the current flowing out of the capacitor is also a function of Vin and the load resistor, R_L.

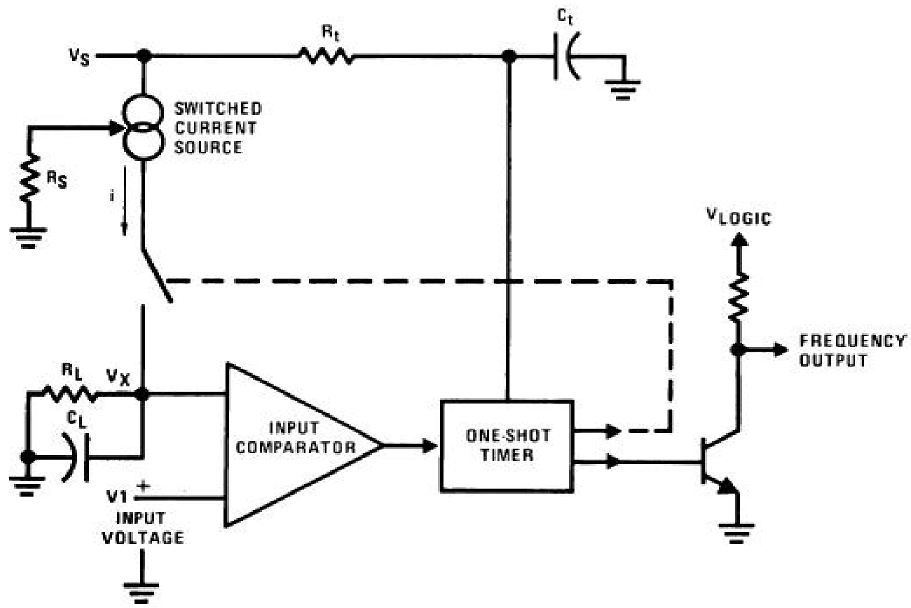


Figure 2.2-8: Schematic, LM231 Voltage Controlled Oscillator (Permission Pending)

The current flowing into the capacitor is a function of the period which the timer is ON, and the frequency of the circuit. Since current going in must equal current going out, if the input voltage doubles, the frequency must also double to maintain the balanced relationship. In this sense, the frequency of the output is directly proportional to the applied, control voltage, Vin. Another way of looking at it: the higher the voltage Vin, the faster the capacitor CL will charge and discharge, resulting in a higher frequency of oscillation.

The LM231 produces an output with a frequency that is precisely proportional to the input control voltage, with a maximum linearity error of .01%. It also consumes low power at 15mW when supplied with 5V power; the supply voltage operating range is between 4V and 40V and it is a low current device, with a maximum supply current of 4mA. The LM231 has a thermal operating range between -25 degrees Celsius to 85 degrees Celsius. The lowest input voltage is -.02V, and can be as great as the positive power supply voltage. Finally, it operates primarily at low frequencies, from 1Hz to 100kHz. Only a single output can be generated by the LM231; therefore, two additional circuits would need to be integrated in the design of the VCO in order to generate the three desired waveforms.

LM13700

The LM13700 operates based on dual, current controlled operational transconductance amplifiers (OTA), where the two op-amps function independently of each other while simultaneously producing two output waveforms. One of the two operational transconductance amplifiers that makes up the internal circuitry of the LM13700 is shown in figure 2.2-9.

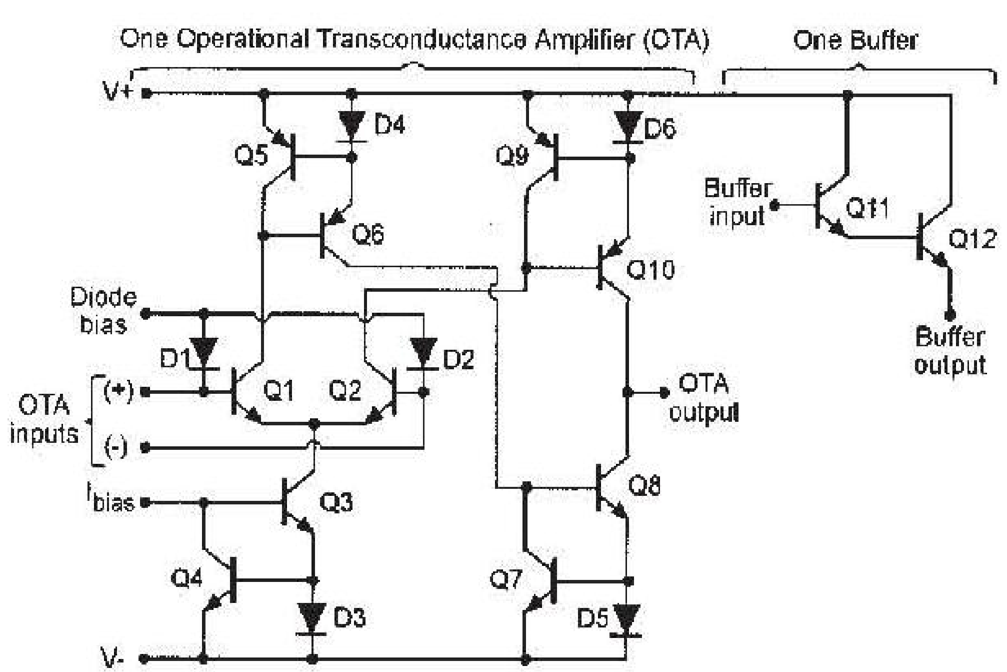


Figure 2.2-9: LM13700 Operational Transconductance Amplifier (Permission Pending)

A complete LM13700 is composed of two differential voltage-to-current amplifiers and eight current mirrors that together produce an output proportional the transconductance of the amplifier and the difference between the two input terminals of the amplifier in the form of current. The transconductance is directly related to the emitter current of the OTA which can be controlled externally. The current mirror consisting of Q3-D3-Q4 allows control of the emitter current by adjusting the bias current externally, while the other three current mirrors make it possible to extract the difference between the input signals of the OTA. Current mirrors Q5-D4-Q6 and Q9-D6-Q10 produce values equal to the collector currents of the transconductance transistors Q1 and Q2, and feed the current mirror Q7-D5-Q8 on both the bias and sink terminals. The current mirrors are designed in the LM13700 so that if a load is placed across the mirror, the difference between the sink/source current and bias current will be sent across the load.

The two diodes between the positive and negative terminals of the op-amp reduce distortion caused by the non-linear transfer characteristics between Ic and Vbe of the transconductance transistors. Impedance buffers appear on the output and input of the op-amps to prevent the high input impedance of the second op-amp from loading the first op-amp improperly and possibly affecting the operation of the first op-amp.

Triangle waves and square waves are possible with the LM13700 because the push pull current mirror circuits are symmetrical and cancel out even harmonics; both triangle and square waves only possess odd harmonics. Figure 2.2-10 shows how the bias current Ic can be utilized as an oscillating circuit.

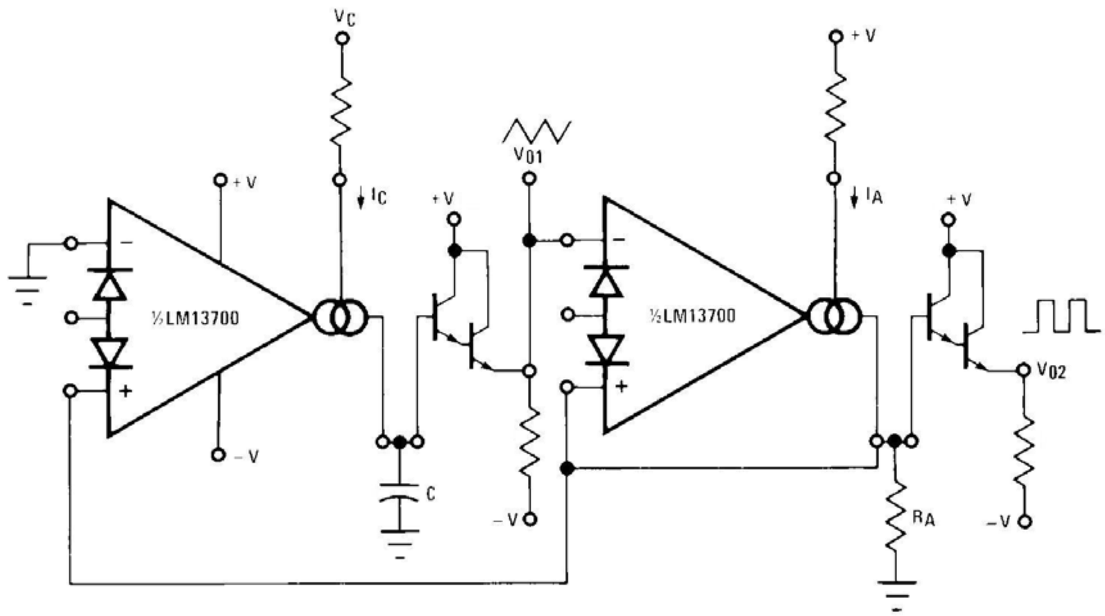


Figure 2.2-10: LM13700 Voltage Controlled Oscillator
(Permission Pending)

If the capacitor is negatively charged and the square wave output goes high, Ra sees positive voltage and sends it to the non-inverting terminals of both op-amps. Since the op-amps are wired as comparators in this application, with no external signal voltage, the first amplifier produces a positive output current that is equal to Ic. This current flows into the capacitor, causing it to charge positively. The ramped voltage from the capacitor is fed into the negative terminal of the second amplifier after passing through an impedance buffer, until the voltage on the negative terminal equals the voltage on the positive terminal. When this happens, the output of the second amplifier switches to low and resistor Ra sees a negative voltage which is fed back into the non-inverting terminal of the first amplifier. This causes the first amplifier to produce a negative output current equal to Ic. At this point, the capacitor discharges and sends a negative voltage into the inverting terminal of the second amplifier. At the point when the voltage at the non-inverting terminal of the second amplifier is greater than that at the inverting terminal, the output goes high and the cycle repeats itself. A triangle wave can be captured following the capacitor between the two amplifier stages since its charging characteristics will appear linear if the correct capacitor and resistor values are chosen.

The LM13700 will oscillate signals between low frequencies up to 2MHz. The power dissipation within the IC is 570mW which is significantly greater than the LM231 previously discussed, but the amplitudes of both outputs are equivalent and determined by the current Ia*Ra. This offers automatic gain control between both output waveforms, reducing the integration of additional gaining components in order to achieve unity gain while integrating a square wave to generate a triangle wave. Since the LM13700 produces two waveforms proportional to an input voltage, only a single integrator circuit would need to be included in the overall design: that in order to generate a sine wave. LM13700 operates between 0 and 70 degrees Celsius, and the output noise varies between 300pA/√(Hz) to 175pA/√(Hz) over the frequency range which it would be operating. The common mode rejection ratio between the differential inputs of both op-amps is a minimum 80dB, ensuring high rejection of common-mode signals between the two terminals and therefore amplifying mostly the difference between the two signals as desired. Finally, the two amplifiers are well matched and possess a linearizing error of only .3db between them do to the diodes at the input terminals of each op-amp.

Direct Digital Synthesis

Direct Digital Synthesis (DDS) is an option for waveform generation that is almost entirely integrated in an IC to produce waveforms of different profiles, while requiring little input voltage and consuming even less power. DDS can form a sinusoidal and/or a triangle wave based on a time varying signal in digital form, and with a digital-to-analog converter, it produces an accurate waveform with high resolution.

A sinusoid exhibits nonlinear characteristics while it is expressed in amplitude; however, the phase of a sinusoid rotates in a linear manner, with a fixed angle during each segment of time. The frequency of a signal dictates the angular rate which it travels during time through the relationship $\omega=(2\pi f)$. In conjunction with a reference signal (providing a clock period), a phase rotation per period can be calculated. A DDS chip uses the relationship between the frequency of a sinusoid and the phase rotation per period to construct a sinusoidal output and then converts it to an analog signal. The internal components include that perform these tasks are as follows:

- Numerical Controlled Oscillator (NCO)
- ROM Look-Up Table
- Digital-to-Analog Converter

The NCO consists of a phase accumulator that scales the range of phase values of a complete cycle of a sine wave into a digital number. Using the reference signal as a clock, the frequency of a sinusoid corresponding to the digital phase-word can be determined. At each new clock cycle, a phase step occurs and the new phase information is added to the contents previously stored in the accumulator. This results in a digital value that is linearly increasing throughout the cycle of a sine wave; however, only ¼ of the period is calculated since the characteristics of a sine wave will repeat itself every pi/2 periods. A sine look up table is generated to compare the phase information of the signal to amplitude information stored in the table, thus giving shape to the sinusoid. A digital-to-analog converter is used to convert the binary, time-divided information of the sinusoid to a useful, continuous analog signal in the form of voltage or current. DDS chips require an external reference clock, precision resistors, and decoupling capacitors. A simplified block diagram of the DDS architecture is shown in figure 2.2-11, as well as the signal outputs following each stage. A filter is used after the D/A converter to rid the output of high frequency sampling components.

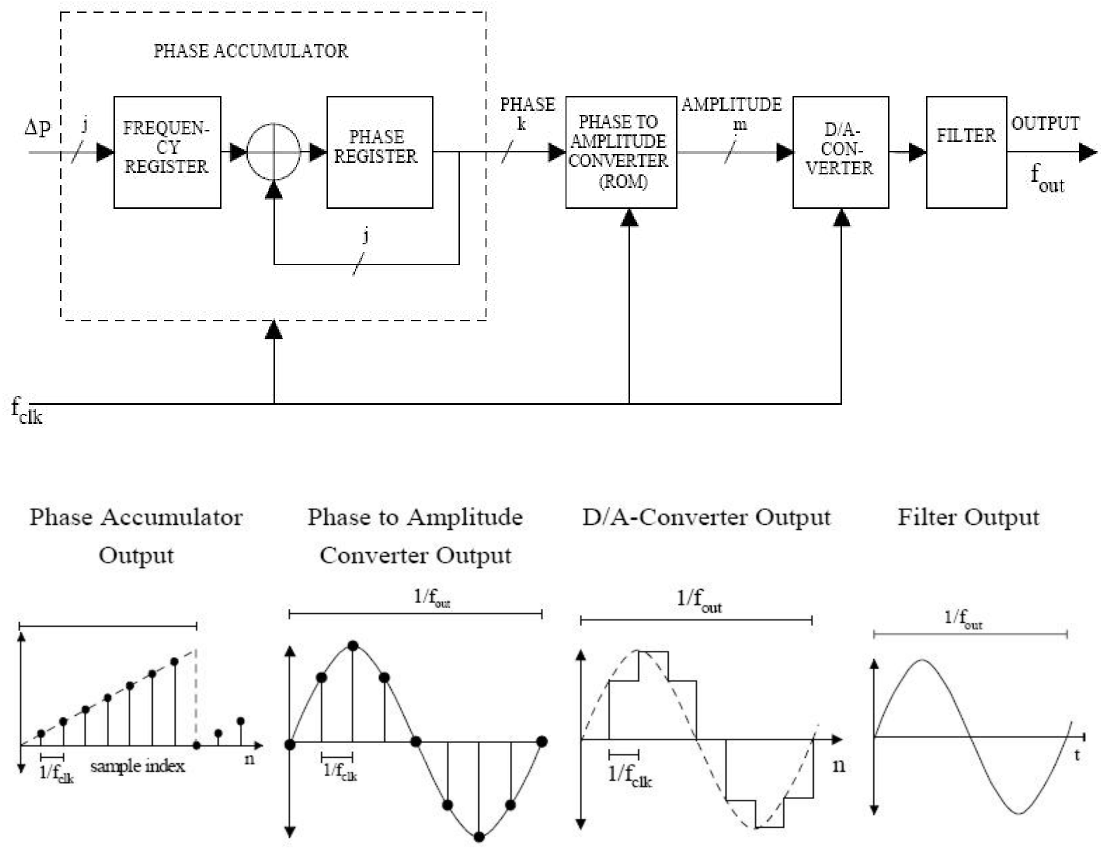


Figure 2.2-11: Direct Digital Synthesis, Architectural and Functional Diagrams(Permission Pending)

A DDS chip produces a very precise sine wave, but implementing one into the design of the synthesizer would require a microprocessor in order to convert the DC output of the Frequency-to-Voltage converter into a digital number that the DDS could read. The microprocessor would require programming in order to relate phase information with a DC voltage level. A DDS chip with sufficient resolution (determined by the reference clock cycle) will most likely not be available for under $9, and purchasing quantity six of these would not be the most cost effective solution to waveform generation.

Comparison

The monolithic circuits above seem to have simpler implementations than the DDS chip. They are also cheaper and require only few external components to operate. The table below summarizes some important specifications in choosing the best IC for the application of generating a waveform with a frequency linearly related to an input voltage/current.

Though the LM13700 consumes more power than the LM231, the voltage input and frequency output are linearly related, so it is promising that all phase information transposed to a voltage from the FVC will be interpreted accurately upon entering the oscillator. Also, the 13700 produces two output waveforms as opposed to one, both of which are desirable for the application of the synthesizer. Finally, the oscillator circuit for the LM13700 can be reconfigured to generate a sawtooth waveform which is especially advantageous in the advent of electronic music since both even and odd harmonics exist in the waveform. The LM13700 can contribute more to the overall design of the synthesizer than any other oscillator described in this section and therefore will be used as the voltage controlled oscillator for the synthesizer.

2.2.5. Wave Shaping

The voltage controlled oscillators will individually produce a waveform with a frequency proportional to the note played on a particular guitar string. It is desirable to have alternative wave forms (as well as mixed waveforms) in increase the number of overtones available for filtering and modulation. The waveforms that the synthesizer should produce include sine waves, square waves, and triangle waves. The following sections explore the different ways to achieve such waves from one produced by a voltage controlled oscillator.

Integrator Circuits

A triangle wave and a sine wave can be generated from the given square wave using a series of integrator circuits composed of operational amplifiers, capacitors, and resistors. The circuit in figure 2.2-12 shows how the integrators would be set up in order to generate and capture both triangle and sine waves from a square wave. Theoretically, an integrator circuit is made up of only 1 resistor, that which is labeled R1 in the first segment and R3 in the second segment. The resistors in the feedback path of both op-amps are to prevent amplification of any DC offset present at the input terminals. Without the resistors, the output signal would be offset by the amplified DC voltage, resulting in larger amplitude than desired.

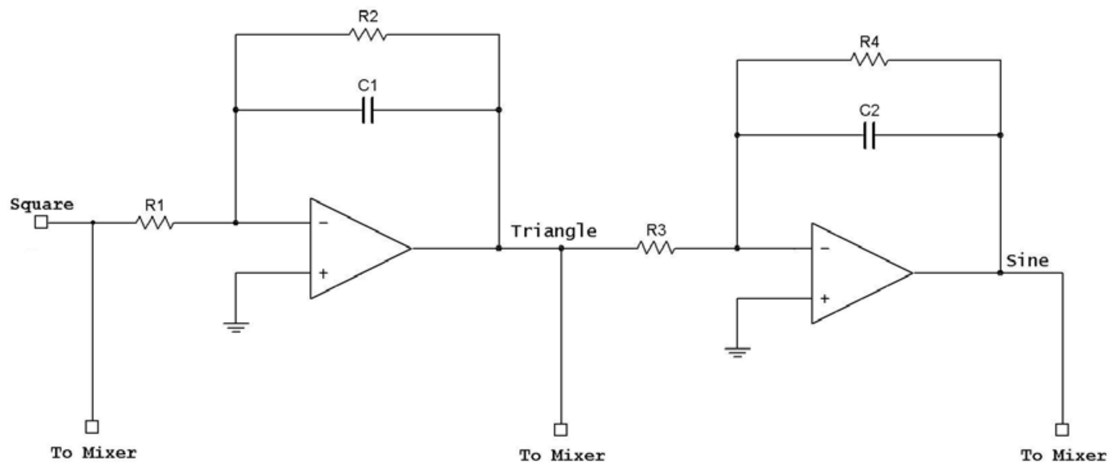


Figure 2.2-12: Tri-state integrator schematic

An integrator produces a voltage proportional to the product of the input voltage and time. A square wave is a series of step changes, composed of a finite impulse and straight line with a zero slope. Since a square wave begins with a rapid step change in voltage, a capacitor will quickly generate a charging current, which is proportional to the voltage across it. As the square wave completes a cycle and returns to the 0dc level, the capacitor will discharge at an equal rate at which it charged, as indicated by the symmetrical, negative slope of the triangle wave shown below in figure 2.2-13.

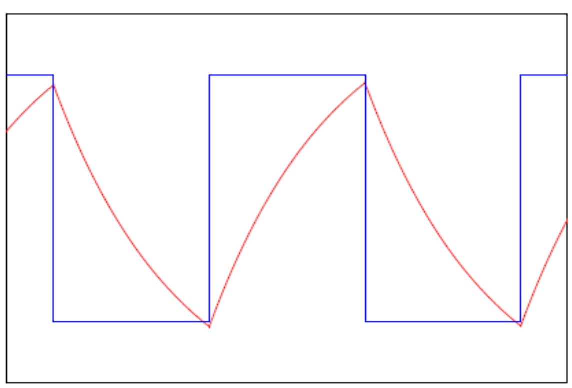


Figure 2.2-13: Output of a square-to-triangle integrator circuit

A sinusoidal waveform can be generated by placing another integrator circuit in series with the triangle wave output. A triangle is a series of alternating positive and negative ramps. Since a ramp is a function that increases linearly with time, integrating one produces a function that is the square of time, which takes the shape of a parabola. The series of alternating positive and negative parabolas resembles a sine wave.

The charging characteristic of the capacitor in the integrator circuit is the indicator of the precision and the amplitude of the output waveform. Greater capacitance means a greater resistance to change. If a capacitor is too small, the voltage will change very quickly, and the desired output amplitude may not be achieved. Since the impedance of a capacitor is a function of the frequency running through it, design considerations must be made to produce an output signal that resembles the integral of both square waves and triangle waves while maintaining unity gain at a given frequency.

Phase Compensation

Integrator circuits do not require many components, and the desired gain can easily be achieved; however, because the capacitor and resistor are set up similar to a filter, a 90 phase shift is introduced on the output of the integrator. With two stages in series, a total of 180 degrees would be introduced at the output, so phase compensation with an all pass filter would have to be integrated into the circuit for at least one stage of integration, while a simple inverting op-amp could compensate another stage. Figure 2.2-14 shows an all pass filter that can be designed in terms of R and C to produce a 90 degree phase shift between the input and the output.

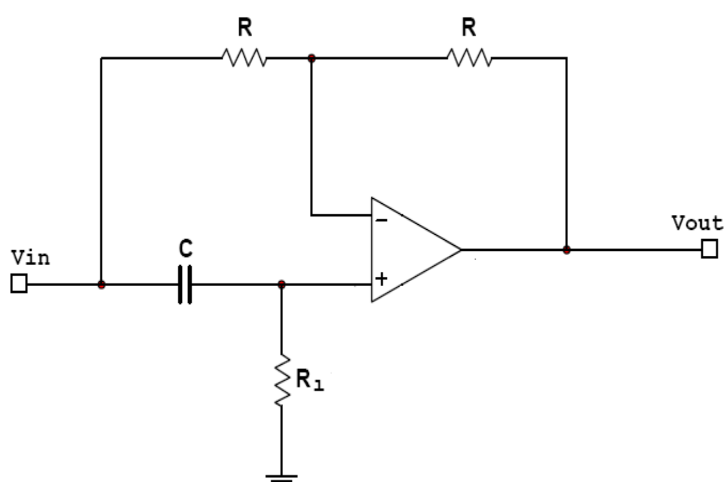


Figure 2.2-14: All Pass Filter used for phase compensation

During high frequencies, the capacitor in the circuit is approximately a short circuit since the frequency term lies in the denominator of the impedance (1/jwC) and the term tends to a value of zero. The filter then behaves as a voltage-follower operational amplifier, contributing a zero phase shift and unity gain. During low frequencies (or at DC voltage), the impedance term becomes very large and can be approximated as an open circuit. In this case, the filter behaves as an inverting op-amp, introducing a 180 degree phase shift. Setting both R's equal to each other would provide unity gain.

Since the all pass filter produces different phase shifts at different frequencies, design considerations must be made in selecting the values of R and C to achieve the desired phase shift at a given frequency.

Triangle-Sine Converter

An alternative method for generating a sine wave from a triangle wave is with non-linear circuit components, such as a diode bridge. This circuit introduces a zero phase shift between input and output. The circuit in figure 2.2-15 shows a triangle-to-sine wave converter, consisting of multiple diode bridges and resistor circuits. The input voltage would be a triangle wave.

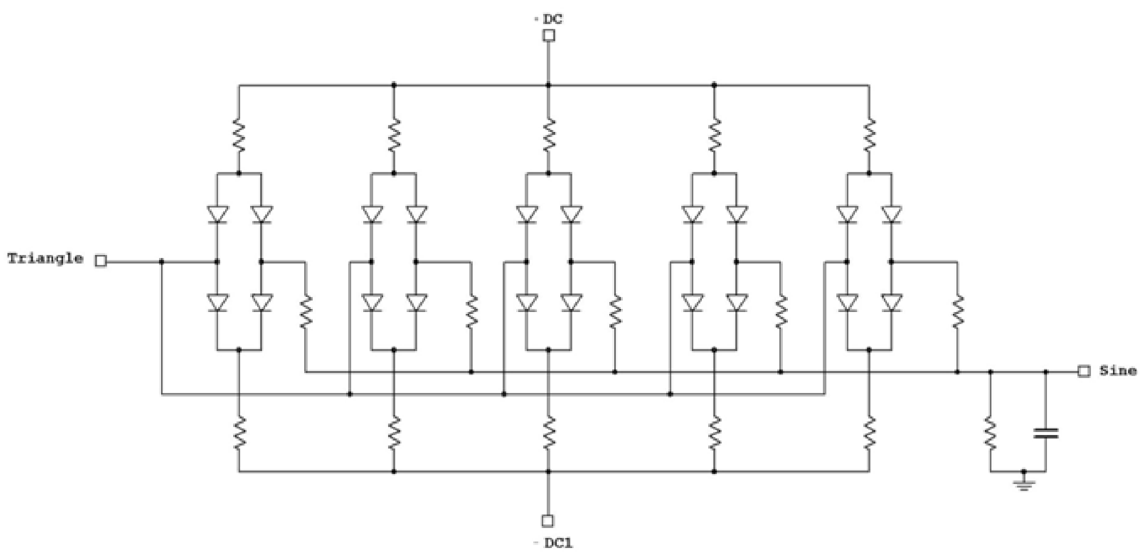


Figure 2.2-15: 5 stage triangle-to-sine wave bridge converter

A sinusoidal output is achieved by producing a piece-wise linear model of a sine wave via transfer characteristics of the diode bridges. The optimal voltage breakpoints are calculated and the appropriate resistor values are selected to minimize the error induced by the linearity of the model. The error is distributed across the range of the function to reduce the linearity of the output, producing a rather accurate sinusoidal output. The precision of the sine wave increases as more bridge stages are integrated into the circuit.

Figure 2.2-16 shows a single stage diode bridge. The resistor Ro in each stage are the slope resistors and determine the voltage breakpoints. V+ and V- are the DC supply voltages, and Vin is a triangle wave. As Vin increases, D1 becomes reverse biased and conducts less. As a result, the current in D2 becomes greater. D4 also conducts less as Vin increases because it becomes reverse biased; therefore, the current in D3 becomes greater. Iout is then equal to the current across I2, which is the current Iu from the DC supply V+. The current across the load, IL, is equal to the current across D3, which is equal to the current from the source, Iin.

The contributing current across Ro becomes independent of Vin. The current across each slope resistor will add linearly until its respective breakpoint voltage has been reached. At this point, the current across Ro remains constant, while each successive diode bridge continues to increase current across its respective Ro, until its breakpoint has been reached. The result is a linear accumulation of Ro currents from the first bridge to the last bridge.

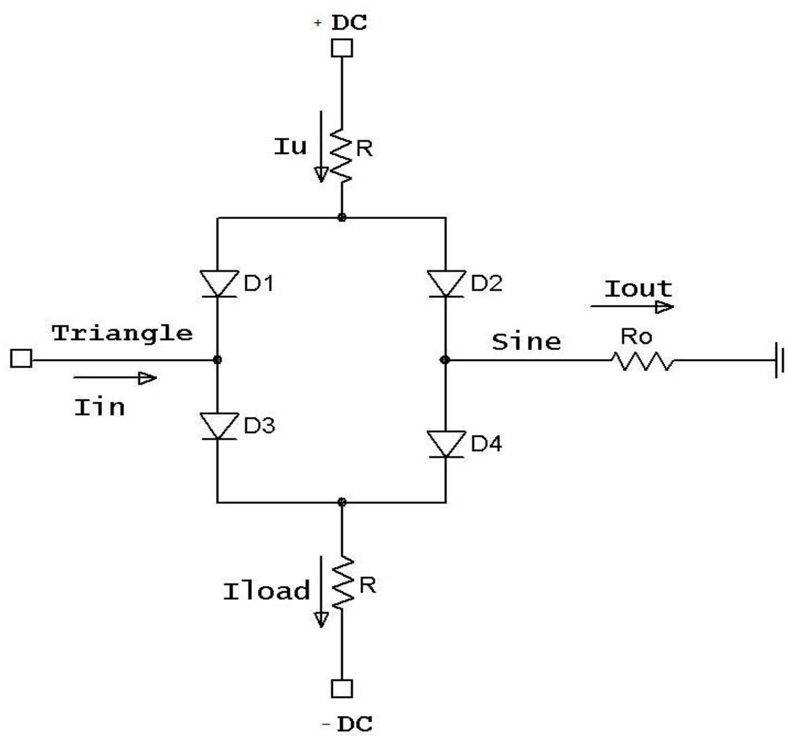


Figure 2.2-16: Triangle-Sine converter, single stage detail

The 5 stage diode bridge circuit can be classified as a symmetrical circuit if resistor pairs are matched. If this is the case, the circuit will behave similarly during negative cycles of the input. The combination of all currents across the slope resistors will smooth the output to the point that it resembles a sinusoid. The output of the 5 stage circuit is shown in figure 2.2-17.

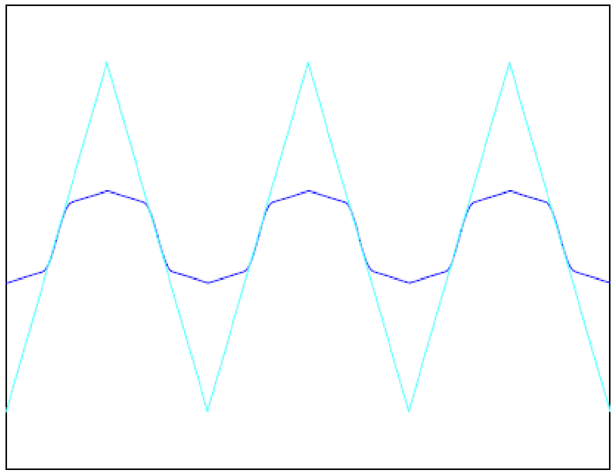


Figure 2.2-17: Triangle-to-Sine bridge converter output

The sinusoid produced by the circuit is much more precise than that produced by integrating a triangle wave; however, many more components are involved, and the circuit is delicate to resistor values since they ultimately determine the breakpoints of the diodes. Since resistors are only available with 5% tolerance at best, the sinusoidal output may be only as precise as that produced by the integrator circuit. Also, troubleshooting the circuit would not be as easy as the integrator since so many components are present, introducing more nodes to check voltages and more loops to check currents. Finally, a gain stage would have to be integrated at the output of the circuit to obtain the desired amplitude of the sine wave, further increasing the number of components in the circuit.

2.2.6. Dual Wave Mixing

With three waveforms available for modulation, it is desirable to mix two together for sound enhancement. When two signals that are in phase are added together, the result is an overall stronger signal than a single wave on its own. Adding waveforms of different frequencies introduces partials, a mixture of harmonics and overtones, that are susceptible to modulation. Many overtones are desirable since they contribute to the uniqueness of an instrument and are ideal in the generation of experimental tones. Mixing two waveforms together will introduce a greater variety of sounds. The synthesizer will allow either a square wave or triangle wave to be mixed with a sine wave. Table 2.2-18 describes the characteristics of individual waveforms, as well as mixed waveforms, to emphasize the desire to mix two waveforms together.

	Sound	Harmonics	Mixed with Sine
Square	Rich, hollow sounds; resembles wind instruments and bass & brass tones; adjusting amplitudes of highs and lows can produce a reed like sounds	Has a rich spectrum, but only odd overtones; it is symmetrical about the pulse width-changing pulse width changes number of overtones	Provides a subtle, psych-acoustic boost; creates fuller sound;
Sine	Pure sounding tone; however, many sinusoids acting together create disharmony	One basic tone; not able to filter any overtones and manipulate sound quality	Extra pure sound, no new added overtones; stronger sounding; not available in the VIRS-1 synth
Triangle	Smooth pitch transitions; ideal for creating flute sounds, pads, and vocal “oohs”; sometimes can offer a muted sound;	Has more overtones than a sine wave, but only odd overtones like a square wave; The overtones vary in amplitude and roll off much quicker than a square wave.	Brighter sound, glitter; emulates presence; if triangle is tuned an octave lower, will generate a male reinforcement

Table 2.2-18: Sound wave comparison among sine, square and triangle waves

The mixer will also allow proportion control between either a square wave or a triangle wave in relation to the sine wave. Both waveforms being mixed will be able to contribute fully to the mixed signal, or not at all, allowing isolation between the two waveforms if so desired by the user. Proportion control will be possible for each of the 6 strings on a guitar. This will allow more options for a guitar player, but will require six times the number of parts as to implement a single dual channel mixer. As a result, a circuit of minimal components will be sought after, as well as cost effective and requiring the least complicated circuitry.

There are a variety of ways to perform dual channel mixing. The mixer should include a gain stage to amplify the output waveform, since the sum of two signals is expected to be smaller than either of the original. This is because the signals will cancel out if they are at all out of phase with each other, and since the two different waveforms will rise and fall at different rates, there will be instances where one waveform is at peak value, while the other is still rising. In this circumstance, the average of the two signals at that point is expected to be generated, lessening the amplitude of the two signals being mixed.

Summing Circuits

The simple summing circuit in figure 2.2-19 allows two signals to be added together. The summed input signals are fed into a FET transistor. Because of the capacitors at the input side of the circuit, the effective impedance of the circuit changes between high and low frequencies. As a result, the output wave shape will distort from the expected mixed signal as the frequencies of the signals increase or decrease, due to the charging characteristics of the capacitors. This effect may not be significant to an untrained ear, but the end result would not match the scope of the project without further circuitry to maintain consistent mixed waveforms. Also, if a gain stage became necessary on the output of this circuit, an additional component such as an operational amplifier would be implemented in the design, adding to the overall component count.

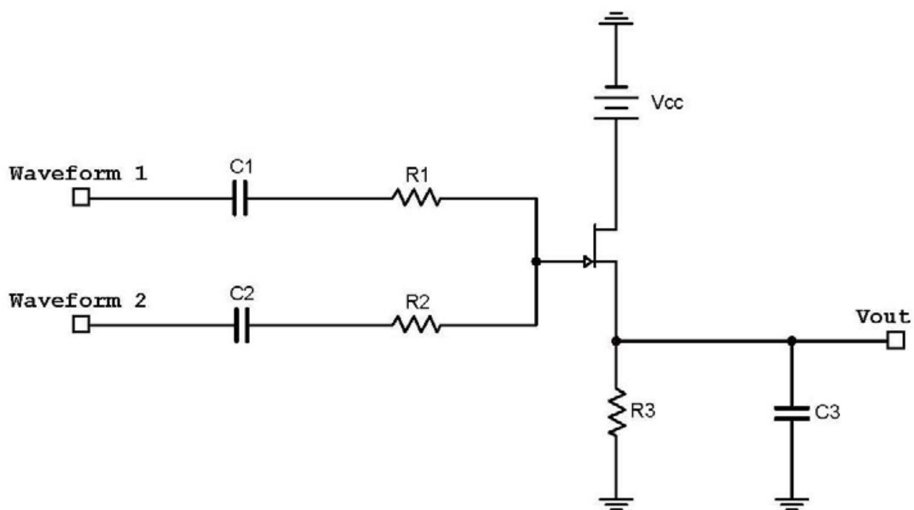


Figure 2.2-19: FET transistor summing circuit

A similar circuit using a summing amplifier could be used. This circuit requires less components since the gain stage can be implemented using a single op-amp, which also takes the place of the transistor seen above. The op-amp in this figure is acting as a voltage follower, so the circuit produces unity gain, but the gain can be increased or decreased by selecting the appropriate values of resistors on the feedback and the on the input of the negative terminal. A capacitor may also be required in the input to filter out any noise on the terminals of the input since the signals that appear at the input will be amplified through the op-amp.

Proportion Control

Potentiometers can replace R1 and R2 in either of the circuits above to introduce proportion control between the two input signals. This capability will offer emphasis of a given channel by allowing the user to adjust the potentiometers: if either of the potentiometers is set to the maximum resistance, nearly none of the respective channel will contribute to the output. This feature allows the user to isolate one of the two waveforms being mixed. Conversely, if both potentiometers are set to 50 percent resistance, each of the waveforms will contribute equally to the output. Figure 2.2-20 shows how potentiometers would be implemented into a summing op-amp.

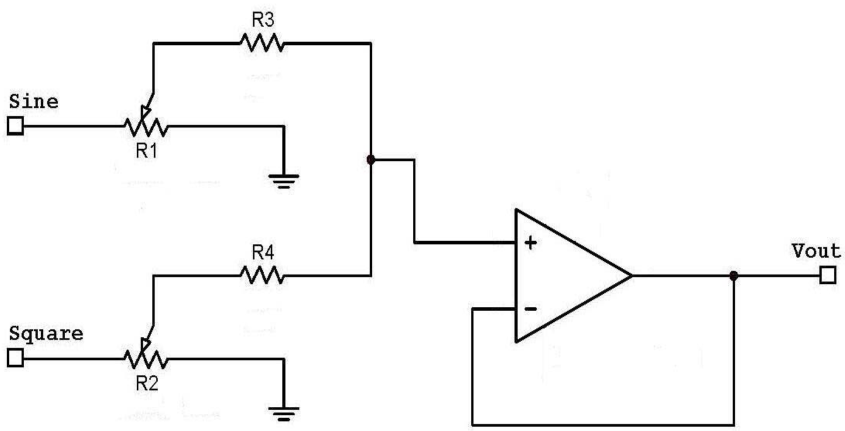


Figure 2.2-20: Summing circuit with potentiometers for proportion control

The outputs in figure 2.2-21 show the resulting waveform when the potentiometers are varied between two input signals. In 2.2-21a, the sine wave contributes more to the output because the resistor R2 is set to a higher value than R1; In 2.2-21b, the square wave contributes more because resistor R1 is set to a higher value than R2.

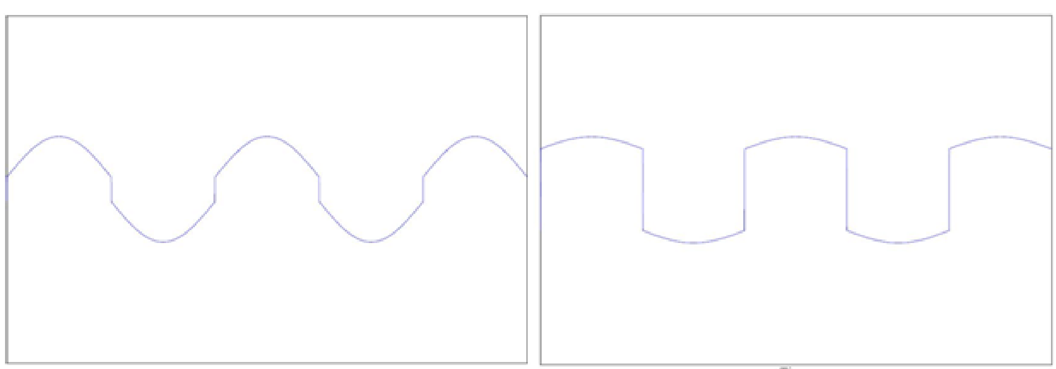


Figure 2.2-21: Proportionally adjusted sine wave and square wave. Figure A (left) is 65% sine mixed with 35% square. Figure B (right) is 35% sine and 65% square

As can be seen in the figures above, two potentiometers function very well for proportion control, and since they are available with tolerance ratings as low as 5%, producing nearly equivalent proportions between two signals would be possible; however, one potentiometer would be needed for each signal that would available to vary proportionally. Since two signals will be mixed for each of the six strings of the guitar, implementing this design would require a total of 12 potentiometers to allow dual-channel proportional control for each string. Not only would this be overwhelming for the user, but it would be inconvenient and spacious on the synthesizer. Also, unity gain cannot be maintained: if both resistors are set to full capacity, the output amplitude would be twice that of the amplitude.

Dual-Gang Potentiometer

An alternative to using two potentiometers would be a dual gang potentiometer, which is two potentiometers sharing a common shaft, controlled by a single knob. Dual gang pots adjust the setting of the potentiometer on two channels simultaneously: as the resistance of one channel is increased, it is decreased equally on the other channel. This allows unity gain to be maintained.

Figure 2.2-22 below verifies the operation of the dual gang potentiometer. Figure 2.2-22a shows how a square wave distorts into a sine wave as its respective side of the potentiometer increases in order to increase the amount of sine wave present on the output signal. At the same time, the potentiometer on the side of the sine wave decreases in value, allowing more of the sine wave to pass through. Figure 2.2-22b shows the opposite, when a sine wave distorts into a square wave as the proportion is controlled to allow more square wave on the output signal in comparison to the sine wave with which it is being mixed. In each of the images, the resulting output would be the average of the two signals.

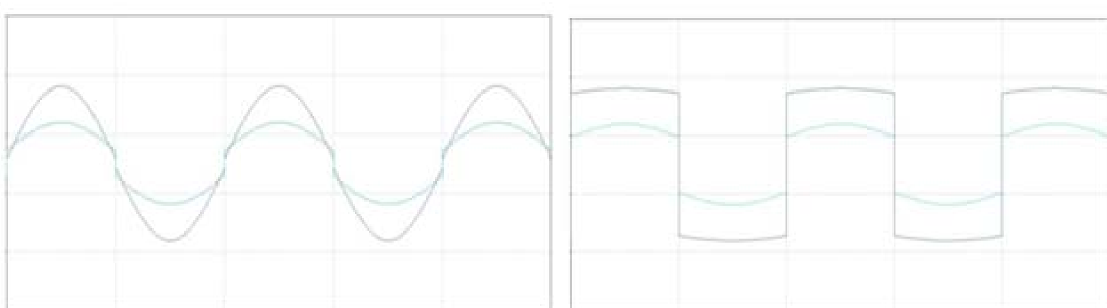


Figure 2.2-22: Effects of dual gang potentiometer on square wave (left) and sine wave (right).

Dual gang pots do not have as high of tolerances as compared to single gang pots; their tolerance is on the range of 10-20%. The range of deviation can be compensated with an attenuator or a trim potentiometer, internal to the synthesizer, therefore reducing the number of controls for the user and saving space on the synthesizer box.

Analog Multiplier

Linear, 4-quadrant analog multipliers are a further approach for proportion control. They are available as an integrated circuit and operate based on variable transconductance of transistors. There are two differential inputs, and when the voltage applied to either input changes, the current at the output changes. Both of the signals being mixed could be fed into each of the positive and negative (differential) terminals of one of the inputs of the multiplier. The second differential input would contain a bipolar DC control voltage. The multiplier would produce an output that is the product of the two inputs, and depending on the polarity of the control voltage, will either add or subtract the relative proportion of either waveform by adjusting the control voltage.

Multipliers require only one control knob to vary the proportion of the mixed signals, but any noise that exists on the input terminals of the device will appear multiplied on the output terminals. They are also more costly than electromechanical devices, but consume low power and can be compensated to achieve excellent linearity between the output voltage and input voltage. For applications where the proportion control is sensitive and significant, a multiplier would perform more reliably than a potentiometer.

2.3. Signal Modulation

Signal modulation consists of a system that takes in a signal and manipulates something about the signal, then giving you a new out. Most figures in our design consist of schematics that are taking in a signal, either wise the waveform or a controlled voltage. The following sections will discuss the theories of the designs and schematics that will be used.

2.3.1. Low Frequency Oscillator

Like the name says the purpose of an LFO is to create a very low frequency of oscillation and to provide the output in terms of a plus minus voltage. Because this is going to be used to modulate filters but never for tone generation tracking is not a vital part of this circuit. Meaning there needs to be no scaling in order for the output to still sound in tune with the input signal. The LFO will however need to be scaled depending how powerful the system that it is driving needs it to be. For example in the tremolo the controlled voltage is very sensitive to change, this means the LFO will have to have a very small plus minus voltage to not vary the pitch too much. The LFO has the options of rate of oscillation and depth of each oscillation. The schematic will be based of a Wein-Bridge oscillator (figure 2.3-1) which uses the op-amp TL 082 . The oscillator has several features that can be tuned. The first is the depth of the oscillation, the depth defines the amplitude of the oscillator, a higher depth would provide a larger plus-minus voltage output. The effect this will have on a system this varies based on what the LFO is driving. For example if this oscillation is going into a voltage controlled resistor, a resistance output will be varied based on the rate of oscillation Because most components in this design are voltage controlled the LFO provides an input voltage that is used to drive the filters and modulate the signal. The LFO can also be used to drive several functions at the same time. If both the tremolo and vibrato are being driven with the same rate of oscillation, every time the note goes sharp the volume will peak and when the note is flat the volume is attenuated. A function can be added to simply invert the wave if the opposite effect is desired.

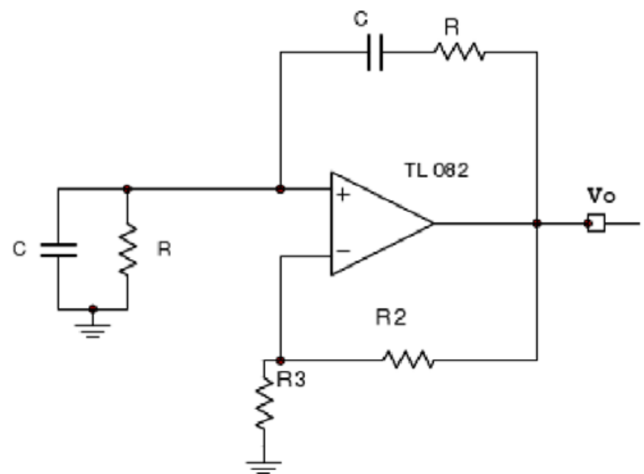


Figure 2.3-1 Wein Bridge Oscillator

<u>Vibrato</u>

The first effect that can be applied using the low frequency oscillator is a vibrato. The vibrato effect is a gradual increase and decrease in the pitch of the signal. The rate of how fast the increase and decrease of pitch occurs is controlled by the rate of the low frequency oscillator and how much the pitch is shifted by is controlled by the depth of the LFO. First the signal must enter phase splitter. Figure 2.3-2 shows a schematic that creates two identical signals but one is 90 degrees out of phase from the other. In this configuration is an all pass filter with a 90 degree phase shift at 159Hz, where Fout is the target frequency for a 90 degree phase shift.

$$F_o = \frac{1}{2\pi RC}$$

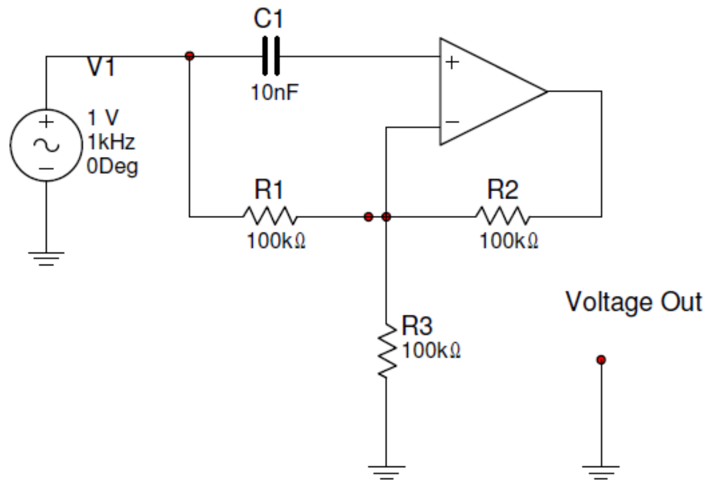


Figure 2.3-2 Unity Gain 90 Phase shift schematic

The LFO will be controlling a two voltage controlled amplifiers that are being used to drive both of the phases. If one of the signals is at full amplitude then the other signal that is 90 degrees out of phase should be completely attenuated. As both switch on and off (rate dependent on the LCO) they are then recombined into one signal. The variances of the two signals out of phase will provide for a slight frequency modulation and the depth based on the amplitude driving the VCA. This flowchart can be seen in Figure 2.3-3.

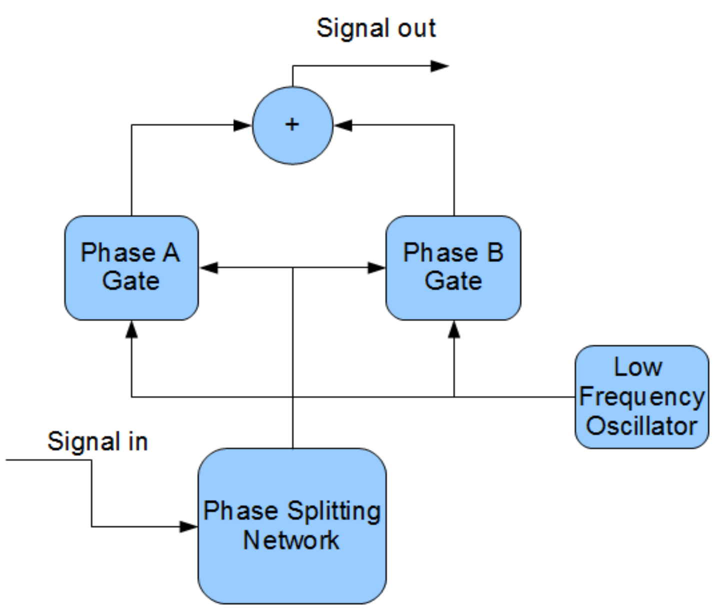


Figure 2.3-3: Vibrato phase splitting flowchart

Another option for creating a tremolo effect is where the pitch is turned into a DC voltage right after the frequency to voltage converter. Here the line is reading a voltage that represents a frequency. If this signal is added to the LFO with a very low amplitude, this in terms would cause the VCO to fluctuate in pitch. When the LFO is high the note will sound sharp and when the LFO is low the note will sound flat.

Tremolo

The next effect that the LFO is driving is a tremolo effect. This circuit is realized a lot easier than the vibrato. Tremolo is a rapid swell and decay in volume of the signal. The amplitude of the LFO will be driving a VCA which will control the magnitude of the signal. The rate is dependent of the rate of the LFO as well. Here the output of the LFO is feeding into the gain control and the original signal is being put into the input. If the LFO is putting out a square wave voltage then the tremolo would be a hard increase and decrease in value. If the LFO is generating a triangle wave the volume goes up and down linearly. Figure 2.3-4 shows a VCA which the LFO will drive.

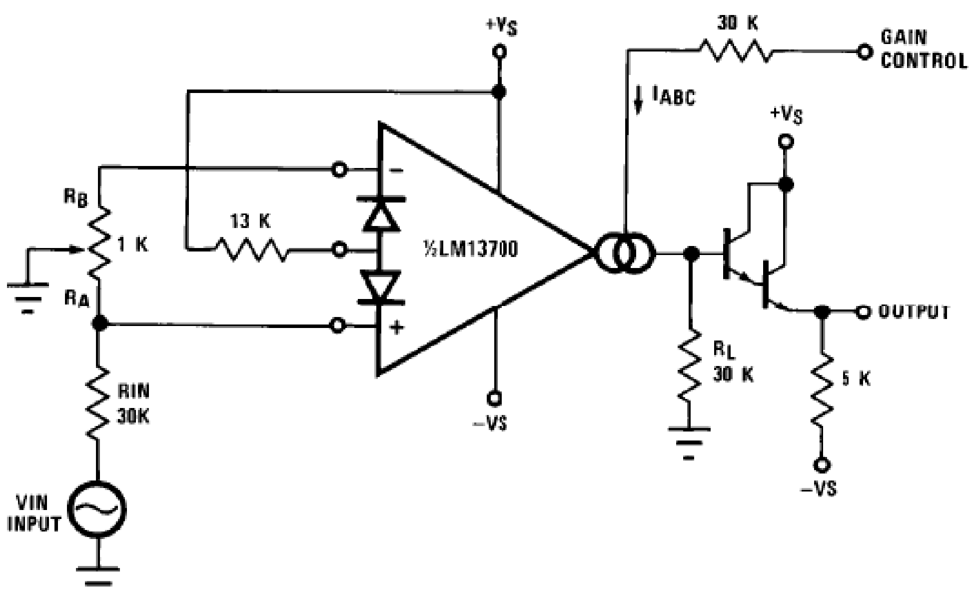


Figure 2.3-4. Voltage controlled amplifier
(Permission Pending)

2.3.2. Voltage Controlled Filter

Voltage controlled filtering consists of real-time filtering of a system. Here the signal will pass through and certain frequencies will be boosted and certain frequencies will be attenuated. A very early option in synthesizers design is a control for cutoff and resonance. At first resonance was seen as a consequence of non ideal filters which did not have a brick wall response. Resonance arose as a spike in energy immediately before the cutoff frequency, as shown in figure 2.3-5. When this was applied to audio filters it was also seen as an unwanted consequence but when sound synthesis arose many people found resonance to sound appealing. Filters were then designed to spike the resonance in a filter according to the user inputs. When mixing waveforms such as square waves and triangle waves, there are many harsh overtones that are produced. These are unwanted effects so in order to solve this problem engineers implemented the design of having a low pass filter where the user can adjust the resonance and the cutoff. This option is controlled by two pots with the cutoff pot varying where the cutoff frequency is on the low pass filter. If the cutoff is turned all the way down this should mean that the cutoff frequency of the low pass filter is so low that almost no frequencies are passing. If the cutoff pot is turned all the way this should be around 1.3kHz. The resonance pot is controlling the quality factor of the filter. If the n Both of these options are going to be controlled by a low pass filter with very specific design factors in order to optimize it for this use.

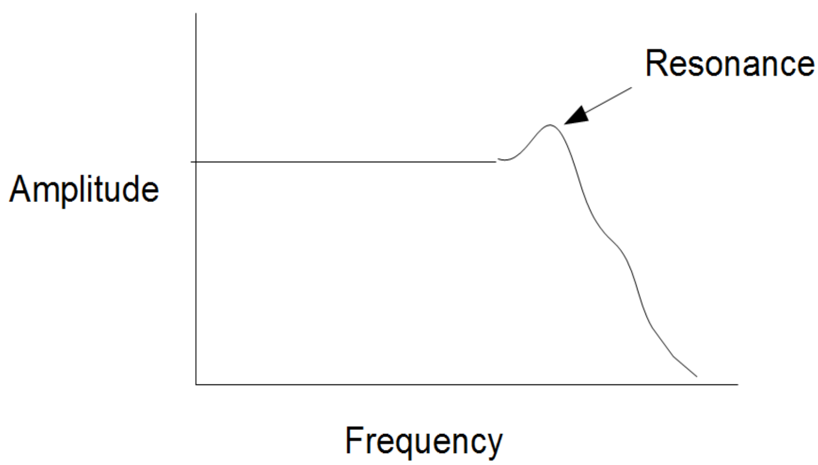


Figure 2.3-5 Diagram where the low pass resonance is labeled

Cutoff Frequency

The cutoff dial represents the cutoff frequency in a low pass filter. If the cutoff is turned all the way down the target cutoff frequency is basically attenuating the whole frequency response as the knob gets turned higher the cutoff frequency then becomes a much higher frequency. The reason this is a low pass filter is because when synthesizing and modulating these tones there may be a lot of harsh and undesired overtones that occur just because of the nature of music. Most of these occur at high frequencies and the highest frequency achievable by the guitar (harmonics ignored) is about 1.3kHz. So a low pass filter with the cutoff frequency max value of around 1.4 or 1.5kHz would work out well. The filter shown in figure 2.3-6 is voltage controlled low pass filter that could be used to filter out frequencies above 1.4kHz.

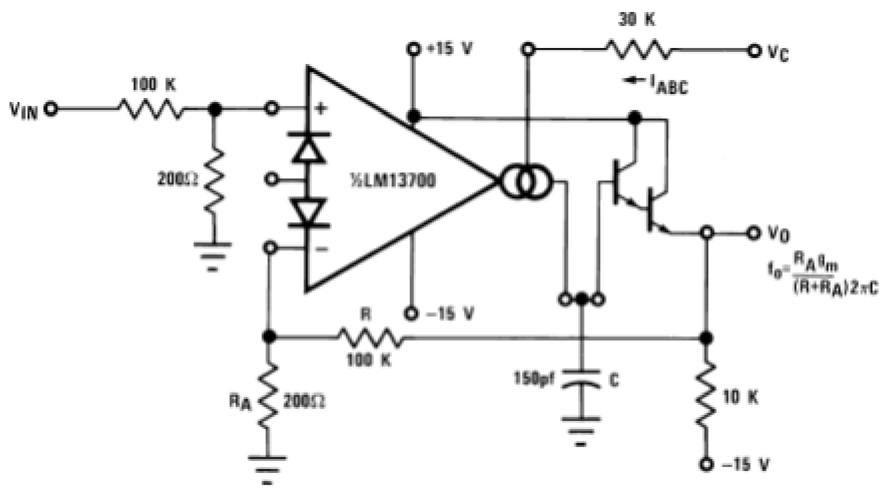


Figure 2.3-6: Voltage Controlled Low pass filter
(Permission Pending)

<u>Resonance</u>

Resonance is defined as a spike in amplitude at the cutoff frequency. Resonance is determined by the quality factor of the low pass filter. A higher quality value results in higher spike at the resonant frequency. As the user turns the resonance knob up the Q value on the low pass filter will gradually grow. The quality factor cant be too large because the gain in Q is multiplied by the gain of the filter. A spike too high in resonant frequency might result in a very nasty peaking in the circuit which could harm components later on in the design. A viable solution would be to have a design that as the Q value gets tuned the overall gain of the system gets attenuated, this way the filter isolates the resonant frequency and works more like a bump filter. These controls are based off of variable resistors. If one was to want some other device to control this filter with a voltage, a good solution is to design a voltage controlled resistor in place of the variable resistor. This way a voltage is turned into a resistance which then changes the parameters on the low pass filter. The schematic for a voltage controlled resistor is shown in figure 2.3-7. This can be used to tune either the cutoff frequency or the resonance of the filter. This can then be driven by an LFO or even the envelope. So when the user would play and the amplitude of the attack increases so would the cutoff and resonance.

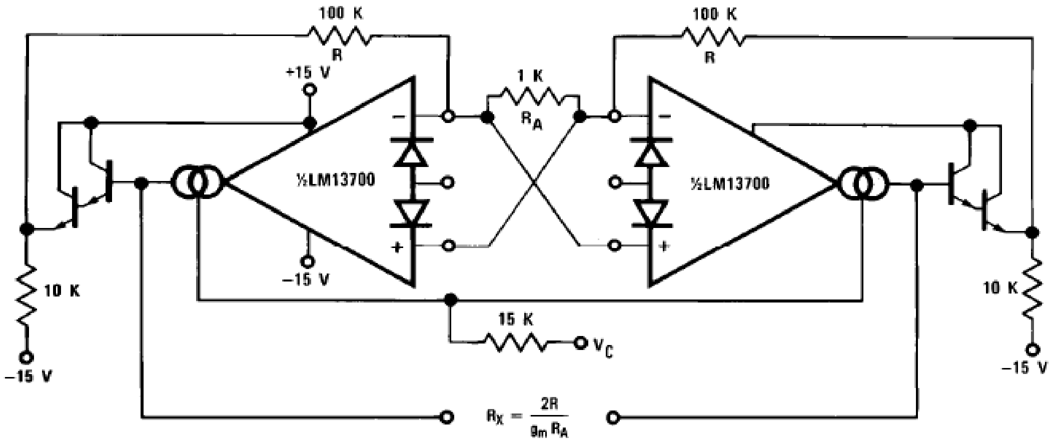


Figure 2.3-7: Voltage controlled resistor

2.3.3 Amplitude Modulation

Multiplying an input signal by another signal is a method used for modulating the amplitude of the the carrier signal with a modulation signal. Synthesizers use a similar process which involves amplifying the created wave by a non-periodic contour signal. Without the contour (or envelope), the resulting sound would not have any realistic volume dynamics, such as “attack” or “decay” which help to add musicality to function. To simulate this contour, a piecewise function was created in Matlab that has the characteristic shape of an ADSR signal. This signal is shown in Fig 2.3-8 and has a maximum amplitude of 1.

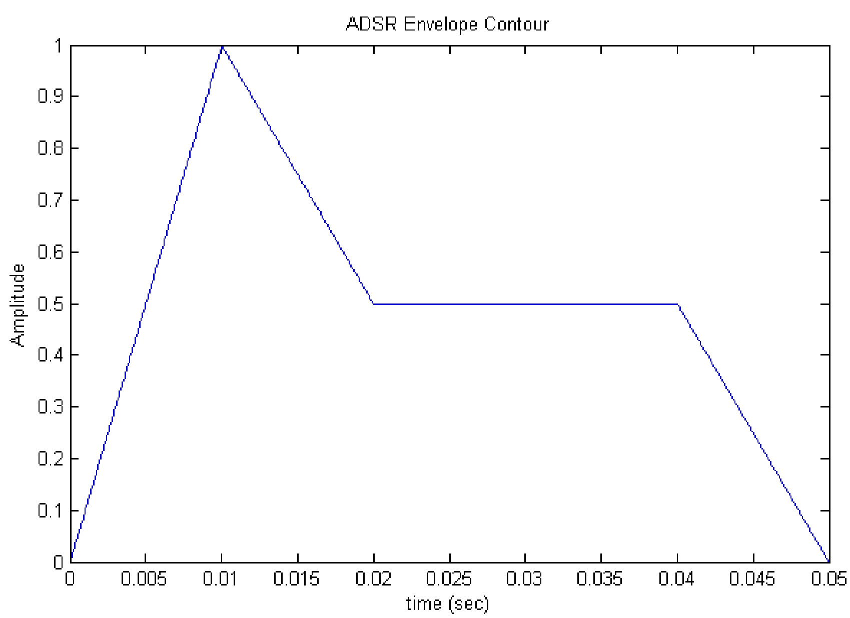


Figure 2.3-8: ADSR Envelope Signal

Using the same technique as in Fig 2.3-8 of multiplying the carrier signal's amplitude by the message signal, a properly modulated signal output was simulated. In this case, there was no DC offset, because the minimum value of the ADSR signal was 0. This is actually ideal, since the ADSR signal represents the note being played at that particular point in time, and should not allow the oscillator's signal to exist anywhere else in time. Fig 2.3-9 shows this modulated carrier signal.

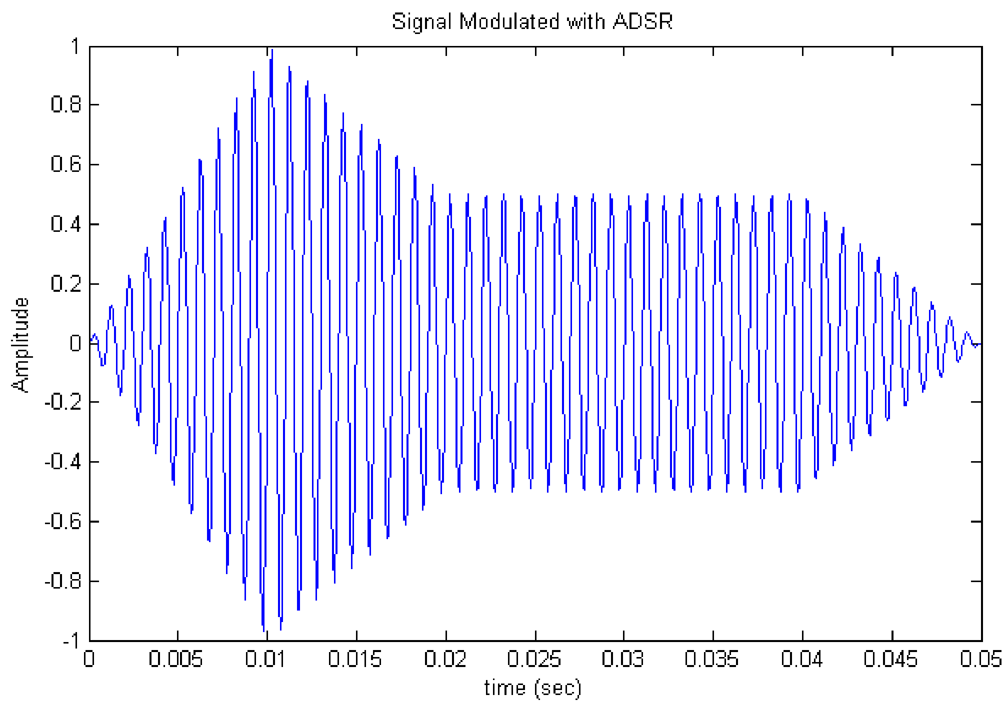


Figure 2.3-9: Signal Modulated with ADSR Envelope

Clearly, the amplitude of the new signal takes the shape of the envelope. This is the what should be achieved by the voltage-controlled amplifier working with the ADSR.

<u>Voltage-Controlled Amplifiers</u>

One of the main signal blocks in a synthesizer is the voltage-controlled amplifier (VCA), which dynamically amplifies the basic, constant-amplitude, synthesized wave by a non-periodic signal called the envelope. The envelope is generated to synthesize not a signal, but the amplitude of the original signal. However, most synthesizers use a system that allows the user to modify the envelope's parameters, such as attack (the initial volume and rate shape), decay (the time for the volume to reduce to its average level), sustain (the level and length of time of the note while it is being held) and the release (the rate at which the volume decreases after the note is released). This part of the system will be discussed more in depth in the Attack, Decay, Sustain, Release (ADSR) section.

For the purpose of the discussion on VCAs, the following assumptions will be made:

- The ADSR envelope is finite in time
- The ADSR parameters are externally controllable

Voltage controlled amplifiers generally accept at least two signals:

1) The source signal
2) The envelope control voltage (also called “contour”)
3) Other control signals as needed, summed with the envelope

There are many different circuit topologies that realize a voltage-controlled amplifier. Some initial considerations in researching possible VCAs are size, voltage supply requirements, linearity, and good signal-to-noise ratio. A component manufactured by National Semiconductor, the LM13700, is a dual transconductance amplifier that seems to fit the needs for the VCA circuit. These devices are actually current-controlled, and fit a variety of applications such as current-controlled amplifiers, filters, and oscillators, to name a few. Being current controlled means that the control voltage from the ADSR circuit must be dropped across a resistor coupled to the “amp bias input” of the LM13700. The integrated circuit includes linearizing diodes which help linearize the transfer function of the amplifier. These are biased through the “diode bias” pins using current sources. Output buffers are also included in the device, which saves circuit board space otherwise used up by another op amp or some other kind of buffer. Having two amplifiers and buffers on each chip is another benefit that will lower the cost of the circuit board, considering that each of the six oscillators will need its own VCA. Some important specifications taken from the LM13700 datasheet are shown in Table 2.3-10.

Parameter	Min	Typ	Max	Units
Supply Voltage	± 5		±18	V
Differential Input Voltage			± 5	V
Input Bias Current		0.4	5	μA
Forward Transconductance	5400			μmho
g_m Tracking		0.3		dB
Input Resistance	10	26		kΩ
Open Loop Bandwidth		2		MHz
Peak Buffer Output Voltage	10			V

Table 2.3-10: LM13700 selected parameters

One possible VCA circuit utilizing the powers of the LM13700 was found at a synthesizer website called "Music From Outer Space" (musicfromouterspace.com). This circuit is intended for use in synthesizers with multiple control voltages, and has the capability of converting the control voltages to a logarithmic response, or keeping their linearity. We only intend on using the linear portion of the VCA, so some modification would be required during design. Once again, cutting out the log portion further reduces circuit board space by removing one op amp, two transistors, and several resistors; a little goes a long way! The first op amp, U1-C, sums and inverts the incoming control voltages and reduces their amplitudes. This is smaller signal is then converted to a logarithmic scaling by transistors Q1 and Q2. In the linear path, the output of U1-C is re-amplified to its original level and inverted again using U1-D. This circuit is intended to be used with an off-board switch to choose between log and linear. It also includes potentiometers for trimming and biasing to achieve optimal performance such as minimum output when the input signal is at 0V through the "Bias Adjust" 100kΩ potentiometer. The "TRIMA" 2kΩ potentiometer is used for properly biasing the input signals to the amplifier. Finally, the "Offset Adjust" 100kΩ potentiometer is simply a voltage divider that adds some DC offset to the control voltage, effectively working as a volume control for the VCA.

Tremolo Effect

Changing the amplitude of the oscillator's output is the main function of the VCA, but by utilizing its ability to accept multiple control voltages, one could very well modulate the signal in other interesting ways. One such modulation is called tremolo, with which the the amplitude is modulated with a low-frequency oscillator (LFO) to "sweep" the volume of the sound from high to low in a periodic fashion. The speed of this oscillation, as well as its amplitude and shape, are controllable by the user, and contribute to creating a wide range of sound effects spanning from a slow volume swell to a jagged, stutter effect. Generally, the tremolo speed varies from little less than 1Hz to about 30Hz. The range is short, but very effective as a tool for matching the tempo of the song being played. The LFO itself is discussed in detail in section 2.3.1 under Signal Modulation. Adding functionality and sonic variety to the guitar synthesizer is highly desirable, in order to give the player more to create with. Tremolo is actually one of the oldest guitar effects, and is seen in many vintage tube guitar amplifiers. Those circuits used vacuum tubes set in oscillation, and used to change the bias of the preamp or power tubes. Even with the ability of much smaller, cheaper, and energy efficient op amps to create oscillations, some amplifier manufacturers still choose the old tube method.

Fig 2.3-11 shows a 1kHz sine wave modulated with a lower frequency oscillation (50Hz here, so that the full period of the wave could be seen). The LFO signal is has some DC offset to allow it to be positive at all times, resulting in a wave which cuts off fully at the times where the LFO is at its minimum. This shows that the amount of DC offset in the LFO signal corresponds to the “depth” of the LFO.

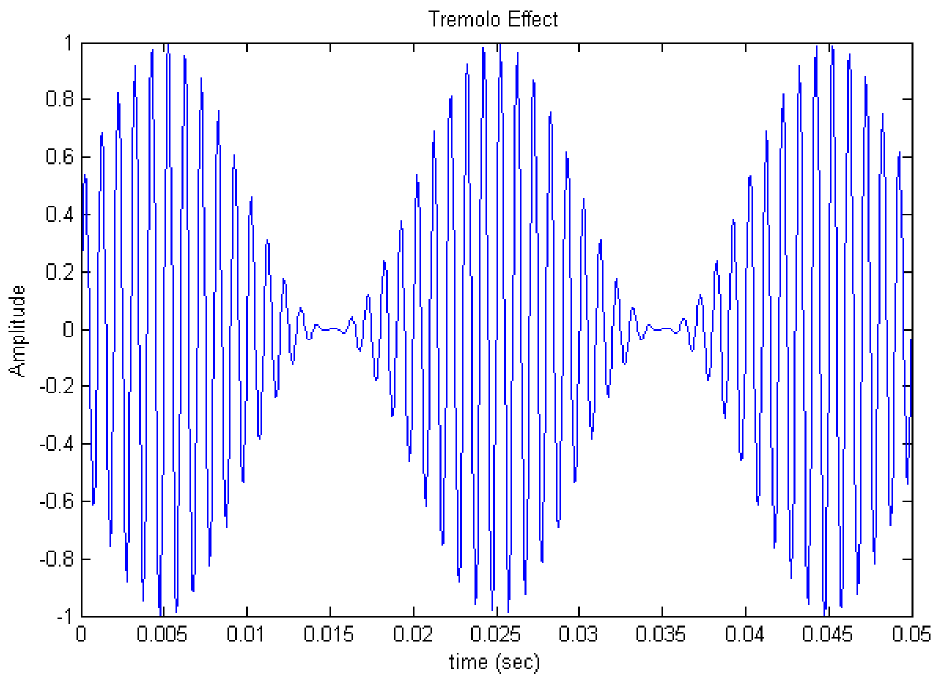


Figure 2.3-11 Tremolo Effect: Signal modulated with an LFO

2.3.4 Attack, Decay, Sustain, Release (ADSR)

The ADSR can be described simply as a controllable envelope. This section allows the user to shape the envelope by changing the duration and level of each part of a note's envelope (shown in Fig 2.3-11). The first section is the “attack” and is a ramp up from no volume to the attack level. After the attack reaches its peak, the “decay” cycle begins, which is a ramp down to the sustain level. The sustain is held at that level for as long as the note is held, and, upon release, is sent into the “release” cycle, which ramps down to no volume.

For an ADSR circuit to know when to begin its cycle, and for how long to remain in the sustain portion of the cycle, a signal call a gate is required from the synthesizer controller. In general, the synthesizer controller may be any device which is played by the user. It has several outputs which pass information about which note is being played, when the note began, and how long the note is played. This usually corresponds to three signals: the pitch control voltage, the trigger, and the gate. The pitch control voltage is discussed in more detail, and specific to the guitar synthesizer, in the section concerning the frequency to voltage converter. The other two signals, the trigger and the gate, are sent to the ADSR for beginning the cycle and staying in the sustain portion for a user-chosen length of time.

Trigger Signal

The trigger is simply a pulse that occurs at the onset of a synthesizer controller event (i.e. a note being played on the keyboard or a guitar string being plucked). This trigger signal is digital, and carries no information about the desired pitch, or the length of time of the note to be created. The trigger always has the same "high" voltage (which could be 5V or up to 15V, depending on the ADSR topology), and the same duration. The length of the trigger must be short enough to allow the player to strike several notes in succession as fast as they possibly can without the trigger trying to fire off when again during its high state. One of the underlying goals of the project is that the system does not get in the way of the player's ability or creativity. On the contrary, it must enhance the musician's “tool set”, giving them more to create with. So, a trigger that disallows fast playing is not desirable. An arbitrary upper limit on speed may be placed at sixteenth notes (four notes per beat) played at 200 bpm (beats per minute), which is 200/60*4 = 13.3 notes per second. To anyone who plays guitar, at least, this seems like a reasonably high enough limit. The time length of each note in this case is 0.075 seconds. Note that the trigger can be split into three parts in time: the rise time, the pulse time, and the fall time. It must therefore be required that these three times summed must not be greater than 0.075 seconds, in order for the trigger to pulse every time a note is played, even up to our specified limit. The trigger's pulse time must be designed according to this limit and the rise and fall time.

Gate Signal

The gate signal is required to keep the ADSR in its sustain period for as long as the gate remains high. The gate, like the trigger, is also a digital “high” or “low”, but is arbitrary in length of time. Gate circuits may be as simple as a momentary switch with some filtering to reduce bounce on a keyboard, or more complicated if the synthesizer controller requires it. For a guitar synthesizer that still uses strings, one does not have the advantage of being able to use push buttons. The gate circuit has to read the guitar’s signal from the pickups and decide whether a note is being played, and for how long. One way of realizing a gate signal is to use a circuit with a comparator. The input signal may be passed through a circuit with a diode that charges a capacitor when the amplitude is above a certain threshold, and then the comparator outputs a digital “high” when it reads the capacitor’s voltage to be above a certain level. The comparator should use hysteresis so that the gate output does not swing randomly when the voltage is near the threshold.

Circuits for ADSR

Several ADSR circuits were researched and one major distinction was made between most of them. The analog ADSR circuits were essentially digital circuits, but used op amps, comparators, and some gates to create what is essentially a logic circuit. The last stage charges or discharges a capacitor, which is buffered to create the envelope output. Upon realizing how much work was involved, it was decided to move to a microcontroller to accept the trigger and gate inputs and deploy an appropriate output. The microcontroller (MCU) will receive each trigger and gate signal as an input to one of its I/O pins, each being set to receive inputs through the software. The trigger inputs will be set up as external interrupts, in order to begin the attack cycle. Once the decay cycle is complete, the MCU will need to read the corresponding gate input and keep the output in the sustain cycle until that gate input goes low. Ideally, there should be analog inputs for attack time, decay time, sustain level, and release time. Using these specifications, a suitable microcontroller could be chosen that will meet all needs.

The microcontroller must be able to receive multiple interrupts from its input pins (up to six, one for each trigger). Interrupts may be configured such that the interrupt service routine (ISR) is triggered on either a rising or falling edge of the input signal. There must be at least 22 I/O ports on the MCU, six for the gates, six for the triggers, six for the outputs, and four for the analog inputs.

<u>Microcontrollers</u>

Microchip PIC

One family of microcontrollers considered was the PIC family from Microchip. Their line of chips range from 8-bit up to 32-bit bus width. Memory ranges from as little as under 1KB to 128KB of flash. They can reach speeds of 40 MIPS and boast peripherals such as PWM, SPI, I^2C, UART, ADC to name a few. They can be programmed using a variety of interfaces, from parallel port to USB powered programmers. There are many circuits that can be built for cheap that will allow programming through either the serial port or parallel port, but the results would not be as fast as using a USB port programmer, which is hard to build. Prices range from about $24 upwards to $200. The other main consideration is programming language. These may be programmed in their native assembly language using the free assembler packaged with Microchip's MPLab. However, it would be much more practical to use a higher level language such as C. C compilers for the PIC micro line start at $495, and there is no free compiler that doesn't have code size limitations. There is a free compiler called sdcc that is unfortunately not fully supported for the PIC architecture. It is geared more towards Z80 and Atmel controllers, so using it would be a risk, despite being free.

- 8-bit to 32-bit
- up to 128KB flash memory
- up to 40 MIPS cpu speed
- PWM, SPI, I^2C, UART, ADC peripherals

Atmel AVR

AVR is a similar family of microcontrollers that could be considered due to their comparison in cost, specs and peripherals. Some of the chips have up to 86 I/O ports, 256KB flash memory, work up to 20MHz, and include PWM and ADC, to mention the important peripherals. The AVR Dragon is a programming board that is supported for devices with up to 32KB of flash memory, and runs about $50 on Mouser. Programming in C is much less of a problem, since there are free compilers such as the one included in the AVR32 GNU Toolchain. This software package is multi-platform and includes a debugger and programming interface. Another option with programming is to use an Arduino board as an in-circuit programming interface, which costs as little as $20 for the Arduino Pro (Sparkfun).

- 8 to 32-bit
- up to 256KB flash memory
- up to 86 I/O ports
- up to 20MHz cpu speed

Texas Instruments MSP430

The MSP430 line of microcontrollers is one possible choice that features very low power design. These 16 bit chips range from 1KB to 256KB of flash memory, up to 25MHz cpu speed, and up to 87 I/O ports. The peripherals include ADC, PWM, comparator, 8 and 16-bit timers, and SPI/I^{2}C interfaces. The major pull for this family of controllers is the TI Launchpad development board, priced at just $4.30. This includes two of the Value-Line chips (2KB, 1 16-bit timer, 1 PWM channel) which alone don't have what the ADSR needs. However, the Launchpad uses a USB connection to which (through on-board flash emulation) programs any chip that uses their Spy-By-Wire interface. This includes chips with more advanced features and more I/O pins, which is what the ADSR needs.

TheMSP430 may be programmed in C with Code Composer Studio 4 with a code size limitation of 2KB, unless a license is purchased at $495. There is, however a free alternative called mspgcc. It is a multi-platform C compiler which boasts competitive code efficiency. It is command line driven, but may be linked to Eclipse for an IDE. Programming and debugging the chip can be done with another tool, mspdebug, which is also command line driven, but free. The lack of graphical interface is hardly an issue when the price difference is so great.

- 16-bit architecture
- up to 256KB flash
- up to 25MHz cpu speed
- up to 87 I/O ports
- ADC, PWM, comparator, 8 and 16-bit timers, SPI/I^{2}C
- Ultra-low power 1.8 – 3.6V

2.3.5. Frequency Multiplier

There are several ways about making harmonizers. The concept of harmonies was explained in the music theory section. In order to make a perfect fifth there needs to be a ratio of 3:2 meaning that within every three waveforms of the note being created there needs to be a ratio of two waves inside of that wavelength to create a 3:2 ratio and a perfect fifth harmony. In a 12 tone system the half steps are divided by 100 cents so if there was 7 half steps to get to a perfect fifth to total shift needs to be 700 cents. The harmonizer will use the controlled voltage coming from the frequency to voltage converter. If you use a resistor value of 100K and a capacitor value of .01 uF and you apply a frequency of 16.35Hz (C) an output voltage of .1635V will be the controlled voltage being used to feed the VCO. A perfect fifth of 16.35Hz is a 24.5Hz waveform which is a G. If 24.5Hz is applied to the frequency to voltage converter an output controlled voltage of .245v is produced. The ratio between both these numbers is .245/.1635 is 1.5. If this same ratio is applied to a 5th of a G the desired note is a D (37.51Hz). Then .245 is applied to the formula below to achieve an original waveform of 37.51Hz. This linear relationship is made possible by the fact that the frequency to voltage converter creates a linear curve for a logarithmic input as seen in figure 2.3-12. A signal can be buffered out of the F-V then ran through a simple gain of 1.5 and fed into a sin wave VCO properly scaled the same way the original VCO. This harmony is only going to have the option of a sine wave in order to keep costs down and because the sin wave is the smoothest tone this would eliminate the possibilities of undesirable overtones to all the controlled voltages being fed into the VCO a perfect fifth harmony can easily created and added back into the original VCO.

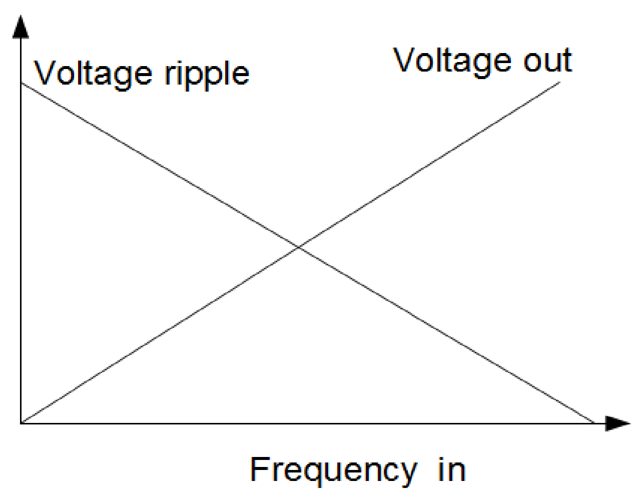


Figure 2.3-12: The linear response of the voltage to frequency to voltage converter

This type of response makes it possible to have harmonies scaled on in a linear fashion even though the response is ultimately exponential.

$$\frac{\Delta Q}{T} = i_c(AVG) = C_1 x \frac{V_{cc}}{2} x (2 f_{¿}) = V_{cc} x f_{¿} x C_1$$

$$V_{ripple} = \frac{V_{cc}}{2} x \frac{C_1}{C_2} x \left(1 - \frac{Vcc\, x\, f_{¿} x C_1}{I_2}\right)$$

$$f_{max} = \frac{I_2}{C_1 x V_{cc}}$$

In order to have harmonies there would need to be another VCO to drive the other voicing. The design would require minimal components in order to not be a very bulky feature. Figure 2.3-13 schematic is for the single op amp sine wave VCO that will be used to make the harmony.

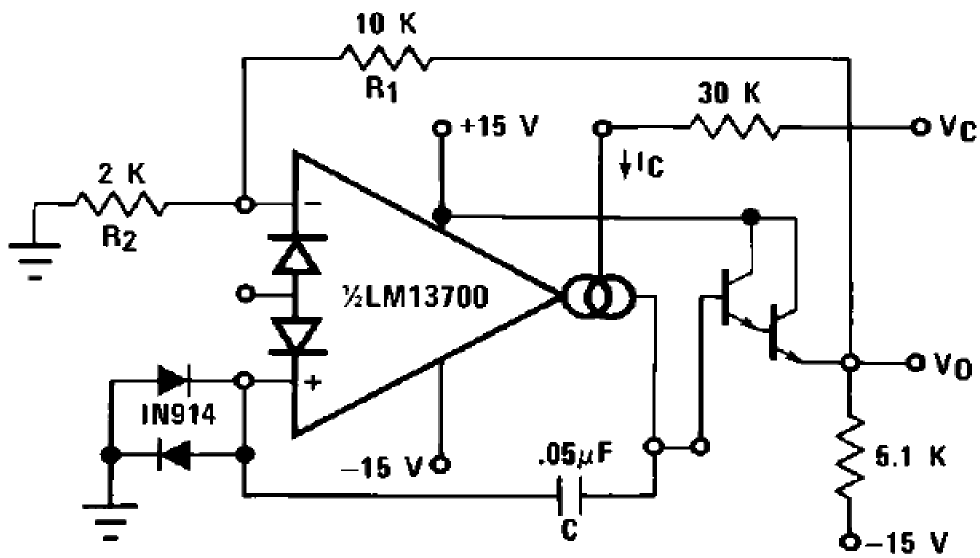


Figure 2.3-13 Single op amp VCO

The following formulas are shown for the VCO which outputs a square wave. If a triangle or sin wave are desired there would just need to be two integrators at the end of the design.

$$V_{pk}=\frac{(V^{+\pm.8}V)R_2}{R_1+R_2} \quad t_H=\frac{2V_{pk}C}{I_f} \quad t_l=\frac{2V_{pk}C}{I_c} \quad f_o=\frac{i_c}{2V_{pk}C}\ for\ i_c \ll i_f$$

There is another option if one doesn't want to build another VCO, it can be an arpeggiator. An arpeggio is three intervals played consecutively. For example a 1st, 3rd and 5th is considered an arpeggio. If the switching rate is fast enough, this can give a ringing harmony to the sound, if the switching speed becomes really low then the individual steps can be heard. This switching can be controlled by a multiplexer and two controlled voltages. The speed and pattern depends on the rate of the control lines. The control lines can be created by passing gates at different thresholds.

2.4. Sound Synthesis

Once the whole design is laid out, the final output should be on a form that an amplifier can match its impedance and read the signal clearly without any distortion.

<u>Single Signal Amplification</u>

Because the signal of this device is going to fed into an amp for final amplification the current and voltage cannot be too high or the amplified signal will experience a lot of distortion due to the threshold limitations of guitar amps. At very high output the waveform experiences a lot of harmonics and a quick decline in amplitude. A high input would be considered around 1.2 volts peak-peak (424 mV RMS), while a weak signal would be around 84 mv pk-pk (30 mv RMS). A good middle ground for circuit can be 0.423 v pk-pk (150 mV RMS). This can be done quite easily by scaling the output of the final output down so it can work in a comfortable range for the amplifier to produce an accurate clean tone. This can be down with a simple inverting op amp configuration and current limiters.

2.5. Power Supply

A DC power supply will be necessary to power the electronics in the synthesizer. The basic components of the DC power supply will provide the following functions:

- Step down mains voltage
- Rectify AC voltage to DC
- Filter out frequency harmonics and smooth rectified waveform
- Regulate voltage output from power supply
- Protect load and power supply from over current conditions

Step-Down Transformer

A transformer will be used to step down mains single phase 120VAC to a lower AC voltage which will further provide DC power to the synthesizer. The transformer used should be mountable to a printed circuit board with a single primary winding. The secondary side of the transformer will provide at least 15VAC in order to achieve a DC power supply of 12V. It is expected that during rectification of the AC voltage, some of the amplitude will be lost due to the required circuit components for rectification, such as diodes.

The secondary windings of a transformer are available as single windings, dual windings, or with a center tap. A center tap may be used in conjunction with a rectifying circuit which will be discussed below. If such a configuration is implemented, the turns ratio between the primary and secondary windings would be less than a single winding on the secondary side, since a center tap is referenced to ground. This implies that if +/-12V was desired on the secondary side, 24V would be required since the winding will be referenced to ground in the center, producing two sections of 12V with opposite polarities.

Rectifiers

Different methods of rectification are possible in order to obtain a DC voltage from an AC voltage. In any case, the use of diodes is necessary since their operation only allows current to pass in one direction relative to their position in a circuit. It is desirable to rectify the AC voltage with as few diodes as possible in order to minimize voltage losses across the diode due to their inherent turn-on voltage requirements, as well as reduce the total number of components which make up the circuit.

A diode bridge consisting of four diodes and one resistor is a common technique for rectifying AC voltages. The single resistor passes voltage in the same direction during both positive and negative half cycles. Voltage drops are accumulated across each diode according to their turn on voltage, and power is dissipated across the center resistor; however, a diode bridge is simple to implement and troubleshoot, and does not consume much space on a printed circuit board.

Since a transformer will be used to step down the mains voltage, an alternative method of rectification which requires less diodes can be considered if the transformer is center tapped on the secondary windings. The configuration requires only two diodes as compared to four, yet still has a single resistor which passes voltage in a common direction during both positive and negative half cycles of the voltage. The primary and secondary windings would need to have grounds independent of each other. Figure 2.5-1 shows the configuration to do so and how the circuit reacts to the positive and negative half cycles.

If the center tap is referenced to ground, then terminal A is positive with respect to the center tap and terminal B is negative with respect to the center tap. During the positive half cycle of the sine wave, diode 1 is on and diode 2 is off, and current flows in the upper loop through the resistor. During the negative half cycle, diode 2 is on and diode 1 is off, and current flows in the opposite direction and through the resistor. Since current flows through R in the same direction in both cases, the circuit produces only positive waves at the output.

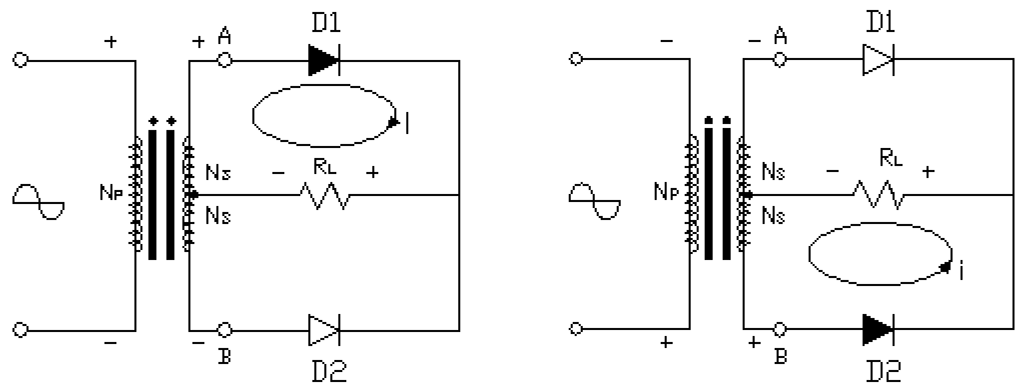


Figure 2.5-1: Transformer rectifying circuit

The reverse bias voltage forced on the diodes of the transformer configuration is twice that of a diode bridge configuration. Noting in figure 2.5-1 above, when diode D1 is forward biased and D2 is reverse biased, the voltage across RL is Va from the A terminal to the center tap referenced to ground. In this case, the cathode to ground voltage of D2 is Va+Vb when considering also the voltage Vb from the B terminal to the center tap (which is referenced to ground). The same is true for the reverse bias voltage across D1 when D2 is forward biased. As a result, the diodes in the circuit must be able to withstand twice the peak of the rectified voltage without exceeding their maximum reverse bias voltage rating.

Capacitor Filters

A full wave rectified signal has twice the frequency of a half rectified waveform because the period has been halved. As a result, the fundamental frequency and the harmonics are twice the value of the original, unrectified waveform. An average, DC value of the rectified waveform also exists on the rectified voltage. A capacitor filter will be placed in parallel with the load resistance to filter out the harmonics and suppress the DC average value. The capacitor will produce a ripple on the rectified voltage since it will charge and discharge as the voltage surpasses the turn on voltage of the diodes and then falls below the turn on voltage. The capacitor filter will be designed in such a way to minimize the percent ripple on the rectified voltage, thus reducing noise on the unregulated DC voltage; if necessary, multiple capacitors will be used in parallel.

2.5.4. Voltage Regulation

The voltage drawn by the synthesizer is expected to vary as the load current increases or decreases. A voltage regulator working with the power supply will maintain a constant output voltage as the load increases or decreases. It can be considered a feedback control loop because it monitors the output voltage and compares it to an internal reference voltage, then generates a response that automatically adjusts the supply voltage as necessary to maintain the desired output. In this section, regulated power supplies will be explored to determine the most suitable design approach for the application of a synthesizer, while considering power efficiency and simplicity of design, as well as ability to maintain a thermal operating range less than 70 degrees Celsius.

Linear Regulators

A power supply with a linear regulator operates based on a voltage dividing network on the output of the regulator that is variably adjusted to achieve a desired output voltage. The actual regulating device is an active component with linear characteristics, such as a BJT or FET.

Two types of linear regulators exist: series regulators and shunt regulators. Figures 2.5-2 and 2.5-3 show functional block diagrams of each. Both possess a control element which is adjusted to achieve a desired output voltage. In the series regulator, the control element is located in series between the unregulated DC voltage and the regulated output voltage. The sampling circuit produces a voltage that is proportional to the output, which is compared with the reference voltage to generate a control signal to the control element. The signal causes the control element to adjust until there is no difference detected by the comparator between the reference voltage and the sampling signal.

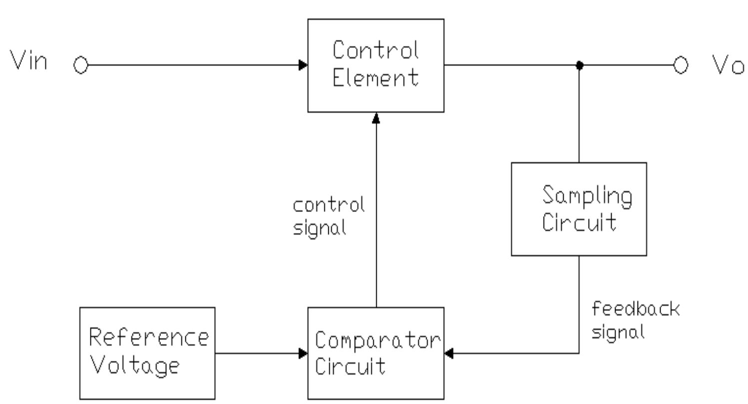


Figure 2.5-2: Series linear regulator, functional block diagram

The shunt regulator control element is in parallel between the unregulated DC voltage and regulated output voltage. It operates by the control element shunting more or less of the load current to ground to achieve the desired output voltage. When the load voltage increases, the resistance across the control element decreases, drawing more current away from the load and reducing the output voltage. When the output voltage decreases, the resistance across the control element increases, sending more current towards the load, thereby increasing the output voltage. The current is shunted to ground, reducing the efficiency of the regulator.

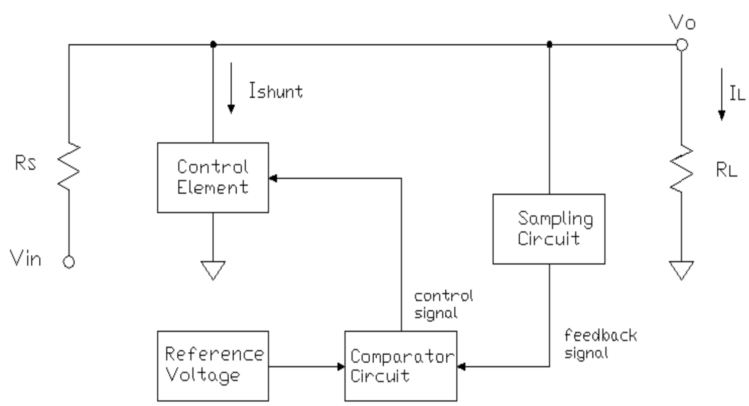


Figure 2.5-3: Shunting linear regulator, functional block diagram

Switching Regulators

The Regulators discussed above operate on a voltage dividing network. The control element is designed to function as a variable resistor, adjusting itself based on the difference between the actual output voltage and the reference voltage. As output voltage increases, the control element will dissipate voltage to compensate the error. It dissipates voltage in the form of heat, lessening the overall efficiency of the regulator, and therefore the power supply. Since the control element operates in the active region, the power dissipation can be large when there is a large difference between voltage in and voltage out.

Switching regulators are an alternative method of regulation with little power loss across the components. The primary operator is a pulse width modulator that adjusts the duty cycle of the input according to the desired output. The pulse width modulator is an IC with an inherent oscillating device that produces a train of rectangular pulses. The pulse widths are proportional to the average value of the input voltage. Higher input voltage results in a wider pulse width. Figure 2.5-4 shows the pulse width characteristics of input voltages with varying average values.

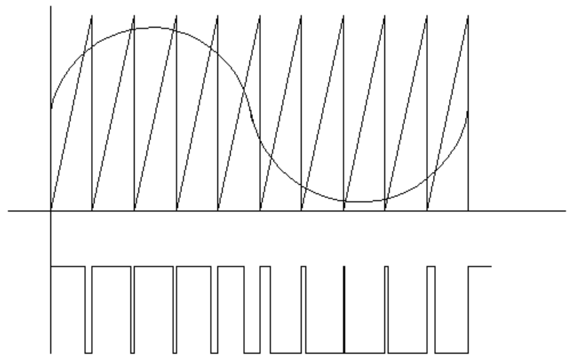


Figure 2.5-4: Pulse width modulator output with an input of varying average value

A typical switching regulator circuit is shown in figure 2.5-5. The output of the pulse width modulator drives the pass transistor Q1. When the pulse is high, the transistor is saturated; when the pulse is low, the transistor is cut off. Thus, the state of Q1 is directly related to the instantaneous properties of the pulse width modulator.

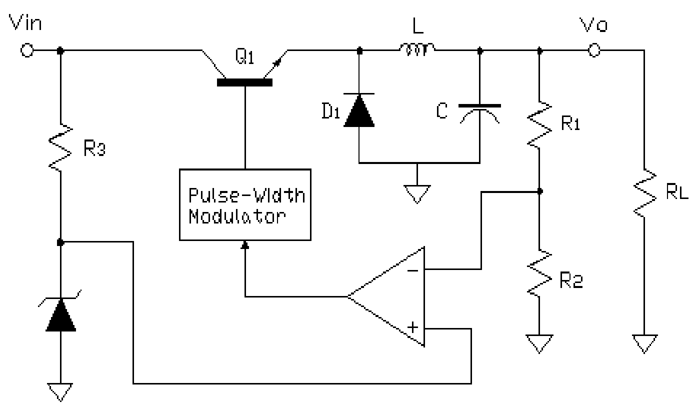


Figure 2.3-23: Typical switching mode regulator

A zener diode provides a reference voltage which is compared to a voltage directly proportional to the output voltage via voltage diving network. When the output voltage increases, the gain of the op-amp decreases since the gain is proportional to the difference between the two terminals. As the gain decreases, the pulse widths decrease (corresponding to a smaller average DC value, therefore, smaller duty cycle) and the transistor conducts less heavily. This produces a smaller load current, decreasing the output voltage. The opposite is true when the output voltage decreases: the gain of the opamp increases as the difference between the terminals increases, driving the transistor into a greater conducting state, thus producing more load current to increase the output voltage of the regulator.

Regulator Comparison

The table below summarizes the tradeoffs between linear and switching mode power supplies.

	Linear Power Supply	Switching Mode Power Supply
Size/Weight	Bulky because heat sinks are required and large transformers because low operating frequency	Smaller transformer used because higher operating frequency; Shielding will add to overall size and weight
Output Voltage	If used with a transformer, any output voltage can be achieved; without a transformer, cannot exceed the input voltage	Voltage limited only by transistor breakdown voltage; input voltage can vary significantly before the output voltage is affected
Efficiency, Heat, Power Dissipation	Excess voltage is dissipated in the form of heat; efficiency on the range of 30-40%; results in high temperature	Small resistive losses between input and output of power supply because switching transistor is either fully on or fully off based on duty cycle control
Complexity	Simpler circuit consisting of IC regulating device or discrete circuit and noise filtering capacitor	Consists of pulse width modulator, at least one transistor, diode, inductor, filter capacitors, and transformer; requires noise considerations in design
Radio Frequency Interference	Interference caused by rectifier diodes during heavy current loading	EMI/RFI produced by high switching frequency of transistor state; requires filters and shielding
Electronic Noise at Output Terminals	Noise a caused by AC ripple superimposed on DC voltage; can be reduced with parallel capacitors	Noisy due to switching frequency; can cause noise in audio circuits
Electronic Noise at Input Terminals	Harmonic distortion at the input; little to no frequency noise	May feedback switching noise into the input
Acoustic Noise	Hum from transformer windings; else, inaudible	Inaudible unless switching frequency is within audible range of humans
Inrush Current	Large inrush current unless a slow start circuit is used	Very large inrush current limited by input resistance of power supply and resistors in series with filter capacitors

Table 2.3-24: Comparison of power supplies, switching mode regulator and linear regulator

3. Design

3.1 Multichannel Pickup

The first step in constructing multichannel pickups is setting up the magnets in the correct alignment. Because each channel is an individual pickup the direction of the pole does not matter but all the magnets will be aligned so the north pole is facing the guitar strings. There are six cylindrical AlNiCo magnets with the diameter of .1875”. Each magnet is individually wound with 42G magnetic wire, since the usual design consists of a few thousand turns the easiest way of doing this is to attach the cylindrical magnet to an electric drill having one end glued to a flat base so the magnetic wire will not come loose during the winding. A few several tests are going to be applied to see how many turns would be necessary to attain a clean signal. The first test will consist of 500 turns and increment each test with 500 more turns. Once desirable output amplitude is found 5 more pickups will be made with the same design specifications. Each individual pickup will be submerged in a pool of wax so the wire does not come loose or out of place over time. The six pickups will be attached in a checkered pattern to ensure there is enough space between them in order to have a proper number of windings as seen in figure 3.1-1. If less windings are needed than expected, the magnets can all be lined up versus checkered in order to follow more traditional designs as seen in figure 3.1-2.

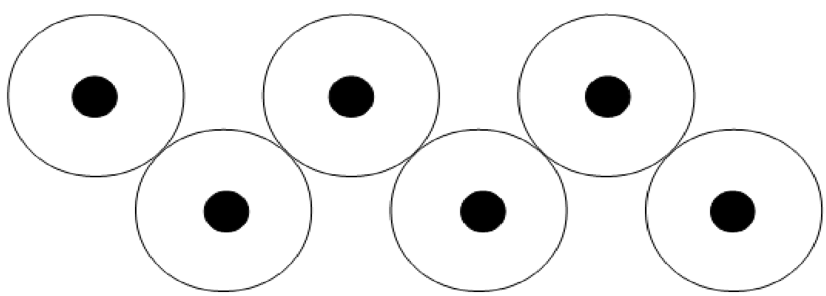


Figure 3.1-1: Checkered pin layout for the pickup layout if large amount of windings are necessary.

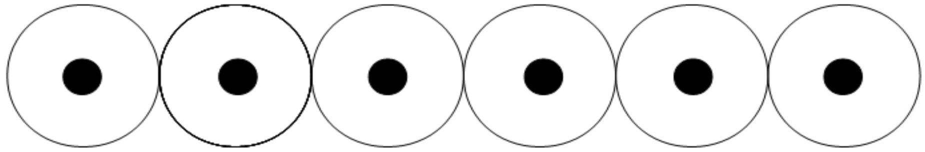


Figure 3.1-2: The second design of the layout of the pins; will be used if minimal windings are needed

Each pickup has two open ends. There are two types of schematic that are commonly used in guitar pickups. One option is to ground one end of the magnetic wire as shown in Figure 3.1-3**.** In this figure the other end of the resistor is terminated with a ground wire and the signal is taken over the resistor.

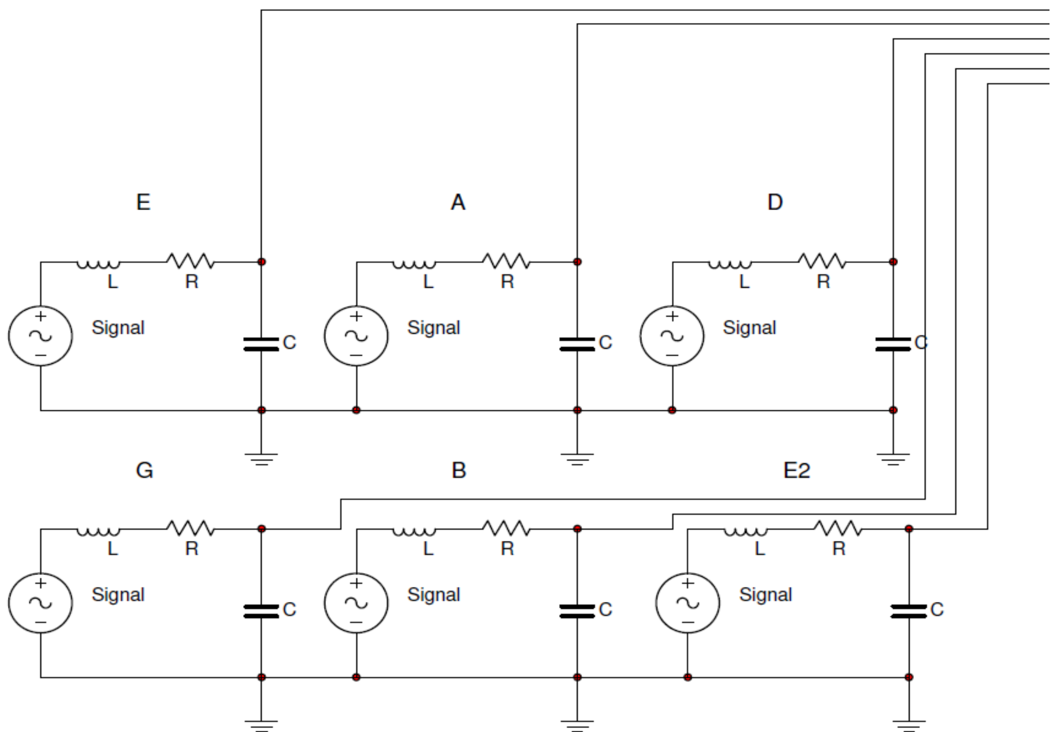


Figure 3.1-3: Grounded string circuit; notice the output only has 6 pins and then a grounded terminal

This configuration would reduce the number of pins needed to supply the board to 6 (one per string) and a ground. This design would be able to use a 7 pin MIDI cable to connect to the board as shown on figure 3.1-4. A midi cable would be a good choice because it is already an industry standard among electronic instruments.

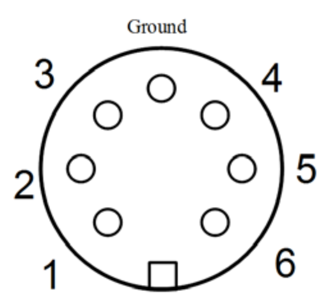


Figure 3.1-4: 7 Pin midi cable

If grounding one of the terminals seems to become an issue through testing, the second option would be a design that does not include the ground wire; this schematic is seen in figure 3.1-5.

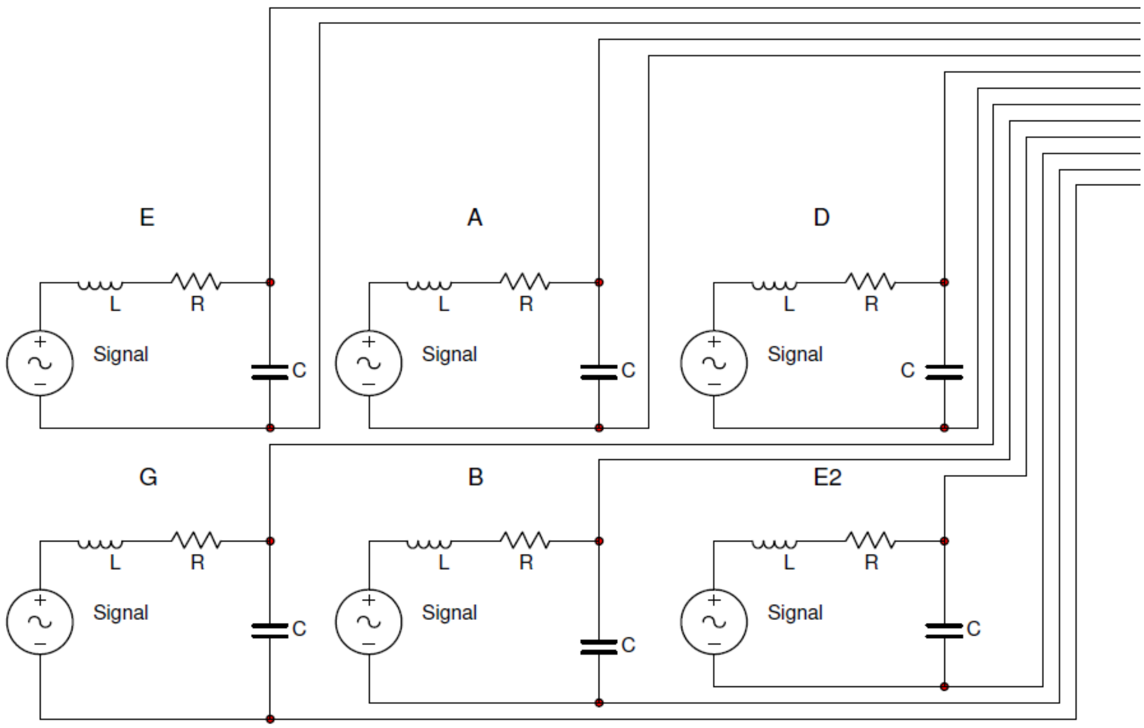


Figure 3.1-5: Magnetic pickup output with no grounded terminal; notice there are 12 outputs versus 7 in the previous design.

This design is more complicated than the previous and would require a cable with at least 12 pins; for this application, a good cable to use would be a VGA monitor cable. This cable has 15 pins available and would leave three available for on board guitar effects. This type of design is seen in figure 3.1-6**.** This type of design means the load resistors terminating the magnetic wire can be placed on the board versus on the guitar pickups themselves. There will be several resistor values that will be tested to see which value gives a better response. The values that will be tested are 1MOhm, 100KOhm and 10Kohm.

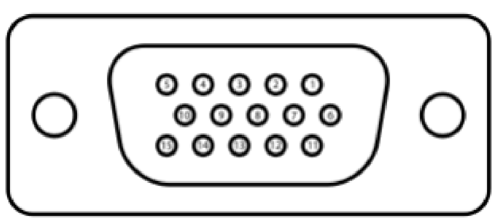


Figure 3.1-6: 15 Pin VGA cable

3.2 Frequency to Voltage Converter

The frequency to voltage converter will be implemented using the LM2907 tachometer chip. Each string will need to have its own channel for conversion since the chip will not be able to read in multiple inputs and have multiple outputs. Also, the scaling will be different for each string depending on the frequency range associated with the standard tuning of a guitar. The output must stay constant for the entire

<u>3.2.1 Initial LM 2907N Testing</u>

Some initial testing was done to verify the functionality of the frequency to voltage converter (F/V). The design includes using an LM2907 (LM2907N/LM2917/LM2917N) to read the output frequency from each pickup and output a linear voltage. The output of the F/V device was very easy to design using the datasheet, which gives an equation for the output voltage:

$$V_{out} = f_{in} * V_{cc} * R_1 * C_1$$

where f_{in} is the input frequency,V_{cc} is the supply voltage, and R_1, C_1 are components in the recommended circuit given in the datasheet, shown in Fig 3.2-1.

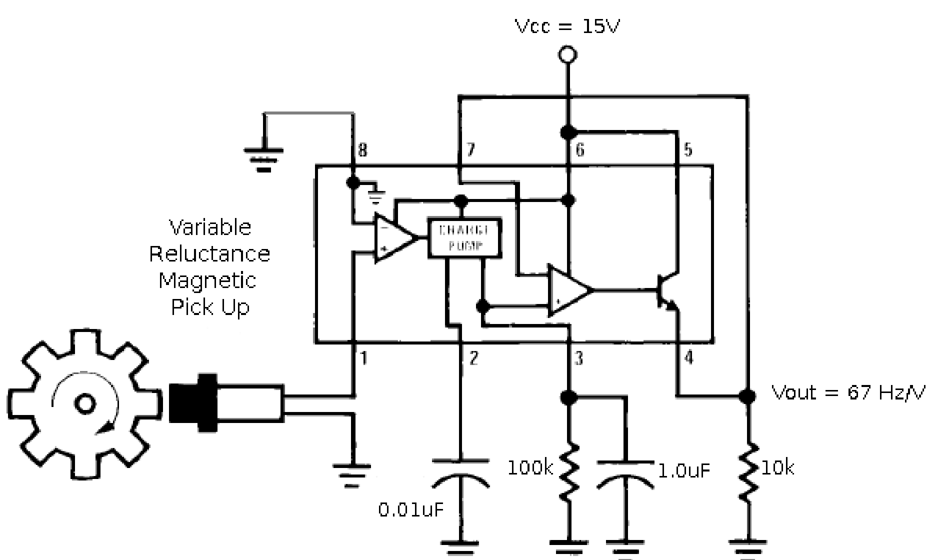


Fig 3.2-1 LM2907 Basic circuit configuration.
(Permission requested from National Semiconductor)

An LM2907N was found and purchased at Skycraft to be tested for functionality. The values chosen were V_{cc} = 10V, R_1 = 100kΩ and C_1 = 0.01μF. This would give an output voltage corresponding to 1V/100Hz, or 10mV/Hz.

Before using a guitar pickup to as an input frequency, a function generator was built that could accurately vary the output frequency and amplitude. It was important to know at what minimum input amplitude does the LM2907N need to see before it starts generating an output. So the device chosen was the XR2206, a monolithic function generator IC that was on hand at the time. The device needs only a few components to generate sine, triangle, and square waves. The circuit was designed using the schematic in the datasheet as shown in Fig. 3.2-2.

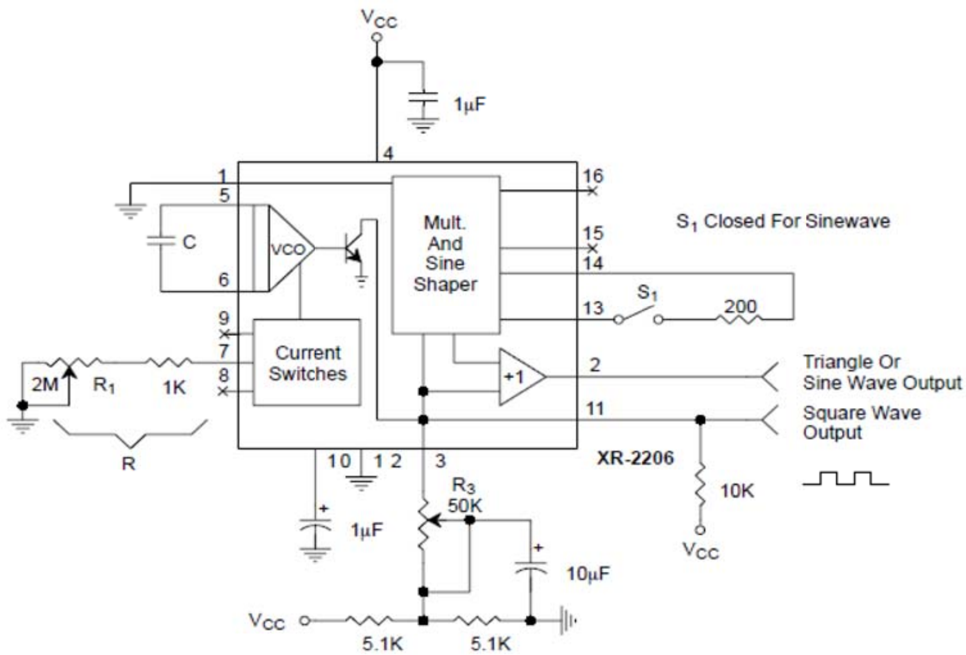


Fig. 3.2-2. XR2206 Circuit for sine/triangle/square wave generation(Used with permission from Exar Coorporation)

The values selected were C = 0.01µF and R being two linear potentiometers (1MΩ and 50kΩ) in series with a 47kΩ resistor. This resistance R determines the output frequency of the function generator. With these components, the lowest and highest frequencies that could be obtained were about 80Hz and 1kHz. This range is ideal for testing within the range of a normally-tuned, standard guitar.

A table of test frequencies corresponding to the frequencies commonly used in standard tuning for each of the open guitar strings (the lowest note an any of the six strings), as well a few other values up to three octaves of the lowest frequency (the low E string) was constructed using the values from a musical frequency table (http://www.vaughns-1-pagers.com/music/musical-note-frequencies.htm). Not having access to a bench top oscilloscope, the output from the function generator was fed into the LM2907N as well as the sound card of the test computer. Using a free program called Soundcard Scope (http://www.zeitnitz.de/Christian/scope_en), these test frequencies could be dialed in with the help of the program's frequency measurement tool. The output voltage of the LM2907N was then measured using a multimeter, and the values tabulated in Table 3.2-3.

Freq (Hz)	LM2907N Vout (V)	Note
82.4	0.8	Low E string
110	1.06	A string
146.8	1.41	D string
164.8	1.58	E on D string
196	1.87	G string
246.9	2.36	B string
329.6	3.14	High E string
440	4.18	A on E string
659.2	6.22	Oct on E string

Table 3.2-3. LM2907 Output from various guitar note inputs.

The output voltage was around 9.4 to 9.7mV/Hz, which is close to the desired output of 10mV/Hz. These values were graphed in Fig. 3.2-4 to determine linearity and the results were very desirable. Linearity in this system is especially important since frequencies are relatively close together at the low end of the guitar, around the 80-100Hz range.

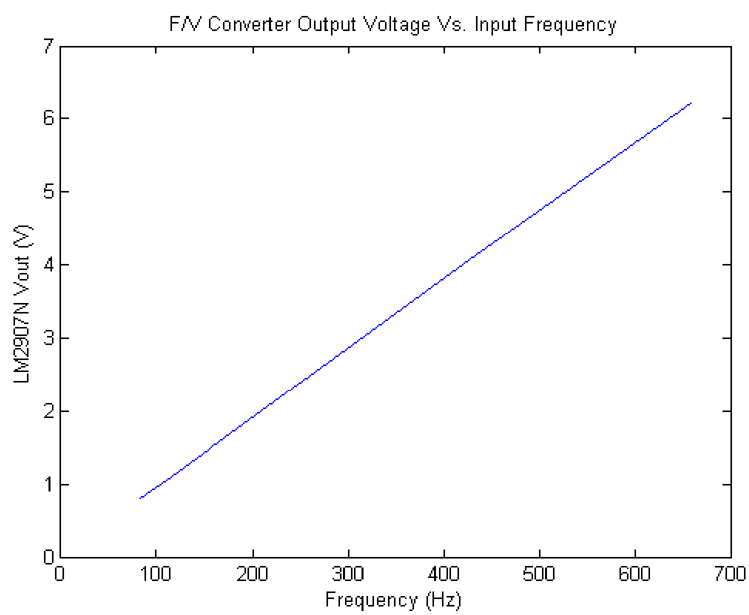


Fig. 3.2-4. F/V Converter Output Voltage Vs. Input Frequency.

The results are very linear, which makes the LM2907N a good candidate for the F/V converter. The output will most likely need to be conditioned for the next stage, the voltage-controlled oscillator (VCO). If the VCO is designed to require a 1V/octave input, (as is common with most synthesizers) then the output of the F/V converter must go through a logarithmic amplifier. However, to simplify the circuit and reduce board space, the 1V/octave specification is likely to be avoided, given the first stage's linearity.

One way to improve these results would be to add a small trim potentiometer of about 10kΩ in series with the 100kΩ resister (R_1 in the LM2907N circuit) to make adjustments for each of the six circuits that will require one of these stages. Then, calibrating would be a simple process: input a frequency from a function generator and match the output frequency from the LFO by adjusting the trim pot.

3.2.2 Guitar Pickup Tracking with the LM2907N

A guitar was used as the input to the LM2907 to test the circuit's ability to track the frequency of a single guitar string being plucked. It was noted before that the in spectrum of an open E string, for example, there exists a strong second harmonic that should probably be reduced before being sent to the F/V. Fig 3.2-5 shows the circuit under test, with some modifications due to a change in power supply. The new power supply voltage is +12V, which means that the resistor had to change to about 83.3kΩ to obtain a linear output of 10mV/Hz. The design has been altered to include an 82kΩ resistor in series with a 10kΩ linear potentiometer, wired as a variable resistor, to counteract the tolerances of both the resistor and the 0.01µF capacitor. Components with lower tolerances will be chosen for the final design for certain areas of the circuit, this being on of them since the volts-per-Hz ratio must be fairly exact in order for the output of the synthesizer to be in tune with the guitar. Note that this represents only one channel and will be repeated for the other five.

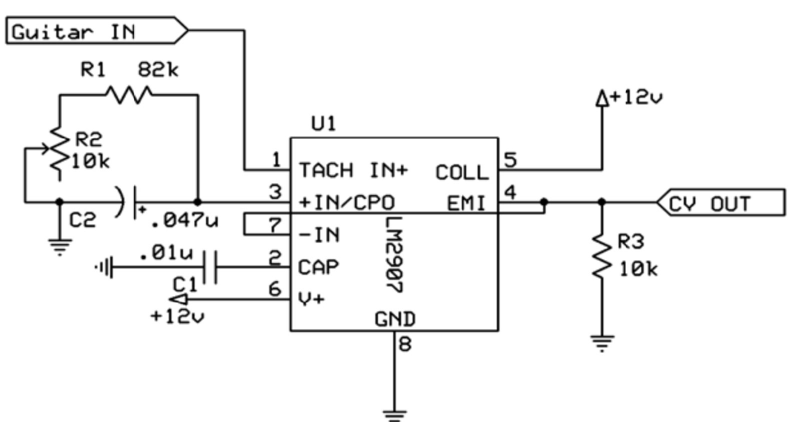


Fig. 3.2-5. Single channel F/V converter design schematic.

The guitar was directly coupled to Guitar IN on pin 1, and a multimeter set to read voltage was connected to the CV OUT, measuring voltage across R_3. Watching the multimeter readout while plucking single notes at various points on each string to gain a perspective about the circuits sensitivity to harmonics gave results that were expected, given what was shown in the guitar string's spectrum. When an open note was played, the output was not constant, but jumped to the voltage associated with the second harmonic of that frequency. The amount of time it stayed on that voltage varied with how hard the string was plucked, and how far up the neck the note was. For instance, playing the open note on the lowest string led to an output that started at 0.81mV, but oscillated between that and 1.62V. However, when the octave of that note was played on the same string (half way up the string), the output was constant at 1.62V. This was also predicted since the amplitude of the second harmonic reduced with notes traveling up the neck, and were maximum when not pressing any frets at all.

Using the results from the initial testing, each channel was assigned a different range according to each string's natural frequencies. The goal for this section was to output a voltage that ranges from 0V to 5V according to the input voltage. For the lowest string, the highest frequency which occurs at the 20th fret has a value of 261.63Hz. This frequency is assigned the output value of 5V while 0V stays at 0 Hz, or simply not input frequency. Each channel's R and C value in the equation for CV_{OUT} are shown in Table 3.2-6. The maximum frequency for each string was forced at an output voltage of 5V, and the slope taken as

$$slope=5/f_{max}$$

where f_{max} is the maximum frequency for each string. This relationship makes designing the VCO section easier, since each maximum frequency is known to occur at 5V. The only problem with this scheme is the loss of what can be called “resolution” or the output voltage interval for each note interval as the notes go up in frequency. This is due to the natural exponential nature of frequencies with respect to rising notes, since notes are divisors of octaves. Because of the imperfect tolerances in the resistors and especially capacitors, trim pots will be used in series with a slightly smaller than specified resistor value, which will be tuned during the testing phase. This makes tuning easy, by inserting an input with a frequency equal to that of the 20th fret of the channel's string under test, and tuning for an output of 5.00V. This scheme also helps reduce rise time for the highest frequencies on the top strings. For example, the 20th fret on the high e string is a 1046.5 Hz note, which is four times the frequency of the 20th fret on the lowest string. If the slope was kept the same for each string, then the highest note on the guitar would result in a voltage four times that of the output from plucking the lowest string's 20th fret note. The LM2907 cannot produce an output with an infinite slope, since a capacitor is used for charging. So, any way to reduce the maximum voltage output is desired for reducing the response time of the F/V converter in whole. Any perceived delay subtracts from the musicality and playability, two major goals of the project.

Open	20^{th} fret	Slope	R desired	R + 10k trimpot	C
82.41	261.63	0.0191	16k	10k	0.1uF
110	349.23	0.0144	12k	8.2k	0.1uF
146.83	466.16	0.0107	16k	10k	0.056uF
196	622.25	0.0080	12k	8.2k	0.056uF
246.94	783.99	0.0064	24k	20k	0.022uF
329.63	1046.5	0.0048	18k	12k	0.022uF

Table 3.2-6: R and C design values for each F/V channel.

One problem that could potentially prohibit the use of the LM2907 is output delay, or the time the output takes to ramp up to the prescribed voltage, given the input frequency. If the frequency changes instantaneously, as in a note being plucked, or another fret being pressed, then the output must change as fast as possible, otherwise the playability begins to decrease. During testing, the output took around 0.5sec to reach its final value. This would severely limit how fast the guitarist could be allowed to play and still get a relevant output. To reduce this time, the capacitor from pin 3 to ground (C_2) was reduced from 2.2μF, as in the datasheet, to 0.047μF. Increasing this capacitor increases output stability but also increasing response time and decreases output ripple. A balance therefore must be found that keeps ripple and response time to a minimum without compromising stability. Ripple is a concern since this would result in unwanted ripple in frequency in the later oscillator stage.

3.3 First Stage Filters

Due to the uncertainty of the output of the F/V converter when a second harmonic is presented, the first stage filters were designed to help boost the fundamental while slightly reducing the second harmonic. The problem with simply cutting the second harmonic as much as possible is that the player may want to (as many guitarists do) play notes above the 12^{th} fret, corresponding to the octave of the lowest note present on each string. This means that the octave must be low enough in proportion to the fundamental, but high enough to be sensed by the LM2907 when it is present alone. The Sallen-Key circuit was chosen do to its ease of design and its small size.

Each string's filter must have a different cutoff corresponding to its open note frequency. Table 3.3-1 shows each filter's component values, chosen to match as close as possible to the required frequency with standard component values. A Butterworth approximation was chosen (Q = 0.707), which led to a gain K of about 1.6. Also, using the recommended simplification,

$$C_1 = C_2 = C \quad \text{and} \quad R_1 = R_2 = R$$

and for gain K,

$$K = \frac{R_b}{R_a}$$

String freq (Hz)	C	R	w0 (Hz)	Ra	Rb
E 82.4	0.1uF	20k	79.6	10k	16k
A 110	0.1uF	15k	106.1	10k	16k
D 146.8	0.05uF	22k	144.7	10k	16k
G 196	0.027uF	30k	196.5	10k	16k
B 246.9	0.027uF	24k	245.6	10k	16k
E 329.6	0.018uF	27k	327.5	10k	16k

Table 3.3-1: Low-Pass Sallen-Key positive filter component values

The simulated frequency response for the low E string filter is shown in Fig 3.3-2 as an example. The others resemble this, except for a shift to the relevant frequency. It is unknown if the string-specific pickup will have different a different spectrum, so these filters may need some additional tuning when the unit is built. The response has an amplitude of about 6 at the low E frequency of 82.4kHz, and drops to an amplitude of 0.79 at 164.8kHz. This was chosen as acceptable until some testing could be done with a real circuit and guitar input.

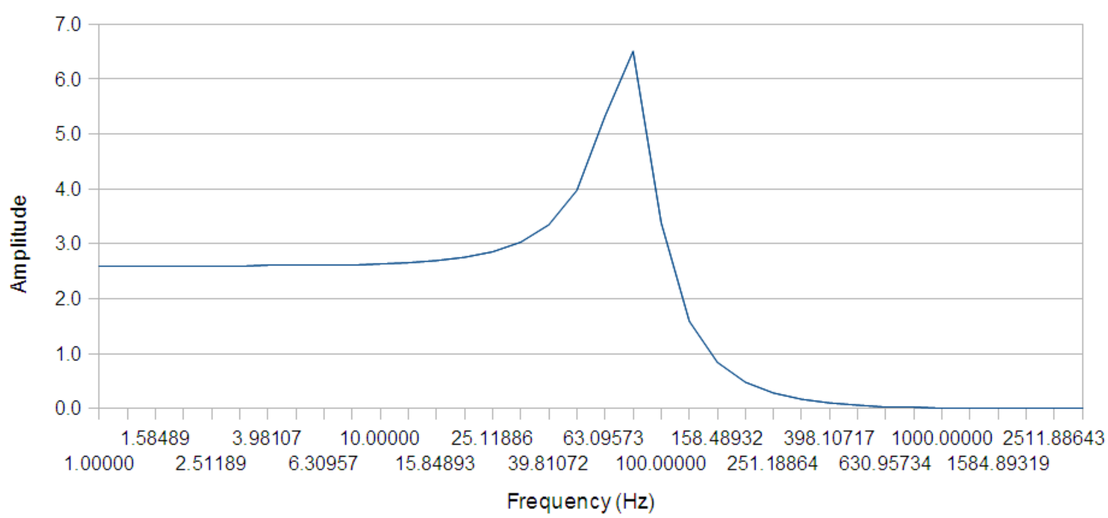


Figure 3.3-2: Sallen-key Filter Response for Low E String

Initial testing of these filters was done to rule out any potential flaw in the design that would require finding a new way of doing the pitch to voltage conversion, such as a DSP front end that would act as a frequency counter, but would also need programming. This is not the final test but more of a preliminary proof of concept test to make sure the design has potential. The circuit was tested with component values corresponding to the filter for the low E string. An LF-351N op amp was used with a split supply voltage of ± 12V. Fig 3.3-3 shows the filter used, with $R_1 = R_2 = 20k\Omega$ and $C_1 = C_2 = 0.1\mu F$. The results were very positive: there was no detectable voltage oscillation when playing the open note on the low E string, as was seen without the filter present. Also, the LM2907 was able to detect the higher notes along the string, meaning that the cutoff was just enough to allow the fundamental be sensed by the chip at any place along the neck of the guitar. This satisfies the requirements for the initial filtering and frequency to voltage conversion. The chips used will be the TL084 quad op amp and TL072 dual op amp, since six op amps will be needed. These can be purchased as SOIC packages for $0.63 and $0.69 respectively.

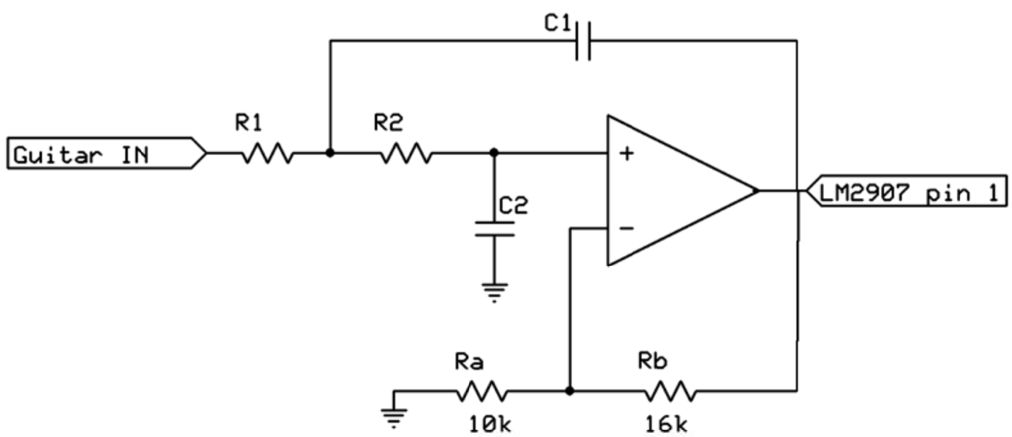


Fig. 3.3-3. First stage filter (one of six).

3.4. Voltage Controlled Oscillator

The LM13700 voltage controlled oscillator (VCO) will receive a voltage between 0 and 10Vdc from the frequency to voltage converter (FCV). It must internally generate a current which is linearly proportional to the control voltage input. The current must drive the oscillator to form two output waveforms that are related to the control voltage input which inherently is scaled and represents a frequency. The oscillator must translate that frequency and replicate it.

Parts Required:

- LM13700 Operational Transconductance Amplifier (x6)
- 5-7 Resistors (x6)
- 1 Capacitor (x6)

The control voltage into the LM13700 will appear across resistor Rc in figure 3.4-1, which will generate a current that, when externally adjusted, will vary the transconductance of amplifier 1. The magnitude of the current will determine the charging and discharging rate of the capacitor, who's voltage will initiate the change of state of the second amplifier's output; therefore, at larger currents, voltage changes faster, and the frequency of the square wave output increases. The process for choosing component values has been described below.

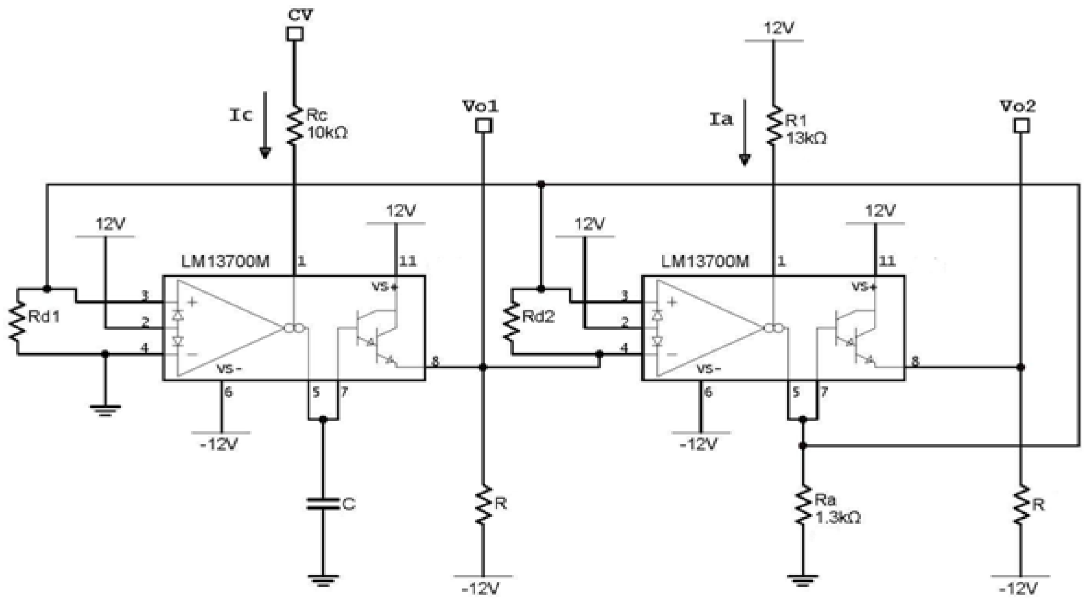


Figure 3.4-1: LM13700 Voltage Controlled Oscillator, Design schematic

The frequency at which the square wave oscillates must be proportional to the current Ic across Rc. The equation relating the frequency to current Ic is as follows:

$$f = \frac{I_c}{4 \times C \times I_a \times R_a}$$

Ia is the current across Ra which remains constant because its supply voltage is constant. The control voltage will vary from 0 to 10V where 0V will correspond to 0Hz. Each guitar string has a different frequency range, so each will require a different value of current in order to change the frequency by 1Hz. The following equation was used to determine the correlation between voltage and frequency for each guitar string, where Cvmax is the control voltage:

$$V/_{Hz} = \frac{CV_{max}}{f_{max}}$$

The bias current Ic sinks to the negative power supply rail after passing though the two transistors associated with the current mirror. The voltage drop across each transistor is 0.7V, totaling 1.4V. Performing a KVL from the control voltage to the power supply rail leads to the equation for Ic shown below.

$$I_c = \frac{CV - V^- - 2V_{BE}}{R_c}$$

Rc in the above equation must be large enough to prevent Ic from exceeding 2mA. The minimum value of Rc was found to be 5.3kQ, but a value of 10kQ was chosen to prevent Ic from dangerously approaching the 2mA limit, and also to prevent any additional linearity error which exists close to the current limit.

Ic was solved for in terms of uA/Hz by making substitutions among the equations above. The results of the above calculations are shown in figure 3.4-2 below.

String	Frequency Range	mV/Hz	Bias Current Range (mA)	uA/Hz
E(low)	82.4 to 261.66	38.22	0.333 to 1.0464.	4.05
A	110 to 349.23	28.63	.334 to 1.062	3.04
D	146.83 to 466.16	21.45	.335 to 1.063	2.28
G	195.99 to 622.25	16.07	.335 to 1.064	1.71
B	246.94 to 783.99	12.75	.333 to 1.058	1.35
E(high)	329.62 to 1046.5	9.55	.333 to 1.057	1.01

Table 3.4-2: Current spectrum of LM13700 in terms of voltage and frequency range of each guitar string

The value of C was chosen for each string by substituting the desired output voltage (Ia*Ra) of 2V into the frequency equation above. Because the frequency ranges vary among the guitar strings, different C values were determined for each VCO circuit that will be built. Table 3.4-3 lists the capacitor values for each string.

E(low)	A	D	G	B	E(high)
510nF	380nF	285nF	212nF	168nF	126nF

Table 3.4-3: Capacitors used for each VCO circuit with the LM13700

The bias current Ia remains constant in the circuit and must be less than 2mA. Based on the desired output voltage of 2V, the product Ia*Ra determined the value of Ra should be 1.3kQ without exceeding an Ia current of 1.5mA. (A lesser value than the maximum Ia was used in the equation to prevent approaching the maximum value 2mA to maintain adequate circuit operation). The equation for Ia is found by writing the KVL from the positive supply rail to the negative supply rail:

$$I_a = \frac{V^+ - V^- - 2V_{BE}}{R_1 + R_a}$$

Rearranging the equation, R1 is found to be 13.7kQ.

Capacitors are usually tolerant to +/-10%, and not as wide a range of values are available on the market. Once the capacitors are purchased, modifications may need to be made to the resistor values on each of the six oscillating circuits that will be built. Other potential modifications include expanding the input bias current range Ic closer to 2mA from 0 in order to expand the resolution of the voltage-frequency relationship. Finally, it is undetermined prior to testing whether the resistors across the input terminals of each amplifier are necessary.

3.5 Wave Shaping

The synthesizer is to have three available waveforms: sine, triangle and square. Since the oscillator driving the waveforms is an LM13700, both triangle and square waveforms will be generated simultaneously due to the characteristics of the IC; therefore, only a sine wave needs to be generated with an integrator circuit. It was discussed previously that an integrator circuit with a triangle wave input will generate a sine wave on the output. The integrator circuit must maintain unity gain at all frequencies which it operates. Since each sine wave generator will operate on a range of frequencies, each will be designed specifically to achieve unity gain at the center frequency (f).

Parts Required:

- 3 LF351 Operational Amplifiers (x6)
- 4 Resistors (x6)
- 1 Capacitor (x6)

Table 3.5-1 below lists the center frequency of each guitar string and the corresponding conversion to omega (2πf) which will be considered during the design.

String	E(low)	A	D	G	B	E(high)
f	172	229.6	306.5	409.1	515.5	688.1
w=2πf	1080.8	1442.7	1925.8	2570.6	3238.8	4323.2

Table 3.5-1: Average frequencies of each guitar string in terms of radians/second

The circuit in figure 3.5-2 shows the integrator circuit with a triangle wave input on the first op-amp, followed by a voltage bugger and an inverting amplifier. Because the integrator is designed using an inverting op-amp, it is necessary to invert the output by 180 degrees; however, a buffer is used in between to prevent the second inverting stage from loading the first stage and possibly causing it to function improperly.

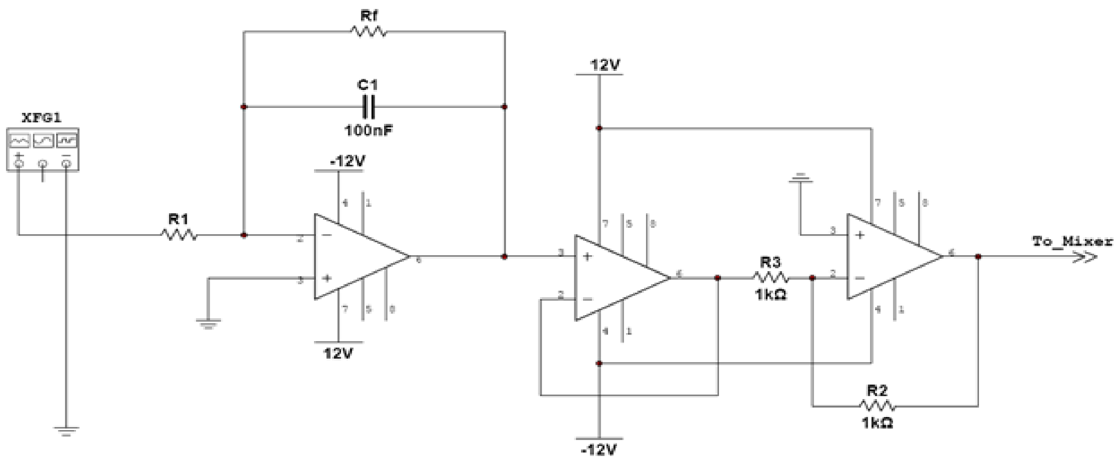


Figure 3.5-2: Triangle-to-sine wave integrator design

The values of R and C in the first op-amp are determined by the frequency at which unity gain is to be achieved. The capacitor value is chosen first because they are typically manufactured with +/-10% tolerance. Once the capacitor is chosen, a resistor value can be selected that fits the design requirements of the circuit. To determine the C and R values of the circuit, the transfer function must be set to 1 in conjunction with the ideal operating frequency. Figure 3.5-3 shows the output of a triangle integrating circuit with unity gain. This is the desired output that way no additional circuit components need to be added prior to signal mixing.

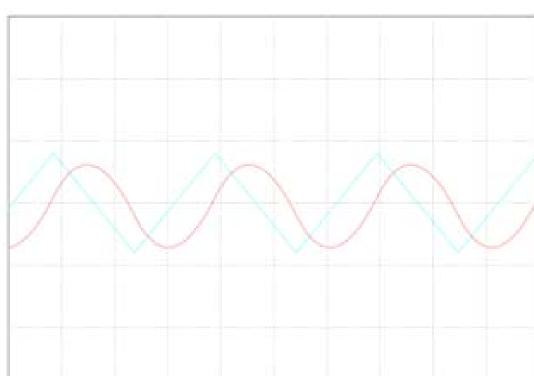


Figure 3.5-3: Triangle-to-sine with unity gain

Figures 3.5-4 shows the output waveform when a triangle wave is integrated at the lowest frequency (82.4Hz) which a Low E string will face during operation.

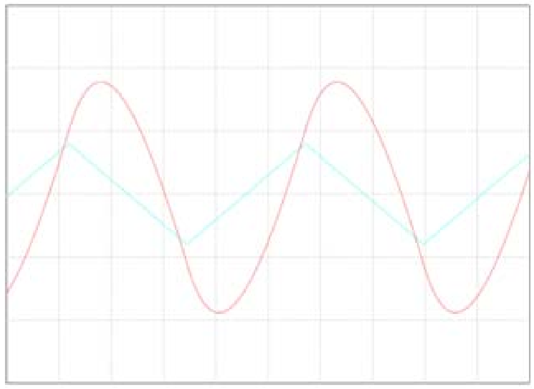


Figure 3.5-4: Triangle to sine wave, low frequency input

The opposite effect is true when the triangle wave is integrated at a frequency higher than which the circuit was designed to achieve unity gain: the output is smaller than unity. Since the circuit was designed for the center frequency of the range of notes on an E string, the deviation of unity gain is inevitable. Since the range of notes on a guitar string increases exponentially by $2^{(1/12)}$, the center resonant frequency which determines the component values in the circuit can be shifted slightly off center in order to balance the deviation.

After testing the design, it will be determined if an all-pass-filter will be necessary in the circuit to offer a 90 degree phase shift.

3.6. Dual Wave Mixing

An operational amplifier configuration was chosen for the dual channel mixing that will take place between two of three selectable waveforms. The decision was made based on the ease of modifying gain to the circuit, which is anticipated to vary from expected during testing of the circuit. A dual gang potentiometer was chosen for proportion control in order to reduce the components on the synthesizer interface, while offering the user equivalent control as a circuit that would use two separate potentiometers. The circuit shown in figure 3.6-1 represents one of six dual channel mixers that will be implemented for each of the six strings on a guitar.

A selector switch is shown on the input between the square and triangle wave and again between two LED circuit branches. A double-pole double-throw (DPDT) switch will be of toggle type located on the surface of the synthesizer, where the user can select between either the square or triangle waveforms to mix with the sine wave. LED's 1 and 2 serve the purpose of indicating to the user which waveform is selected. Each is preceded by a resistor to step down the input voltage and not exceed the forward operating range of the diode. LED1 is associated with the square wave and LED2 corresponds to the triangle wave. A single toggle switch will change the position of both arms simultaneously. The dashed line between the potentiometers Pa and Pb indicate a dual gang potentiometer, where the two share a single shaft. Resistors R3 and Rf determine the overall gain of the operation amplifier. Positive and negative power supply voltages will be supplied to the opamp and will ultimately limit the output voltage of the signal to their respective levels.

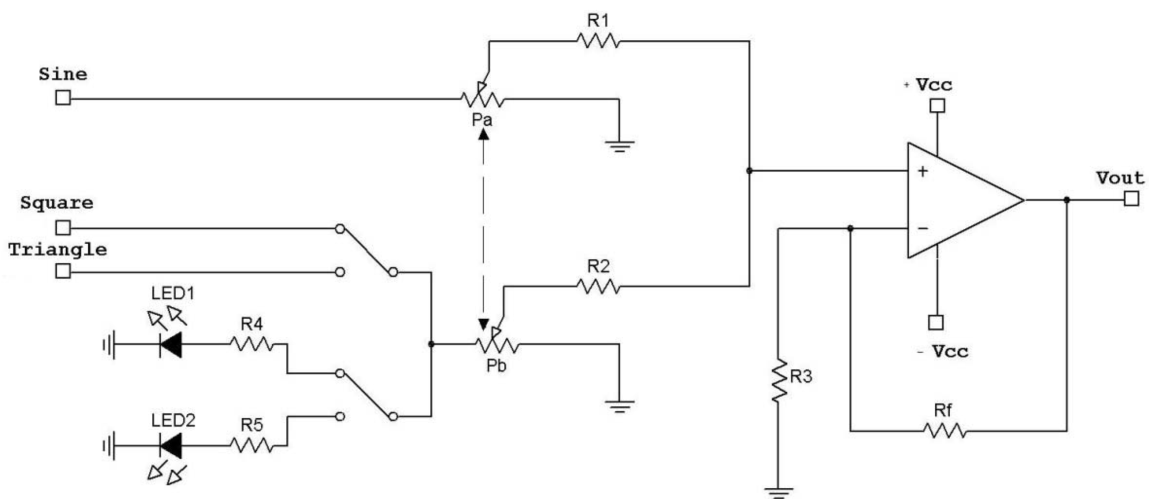


Figure 3.6-1: Dual wave mixer with dual gang potentiometer

Parts required (x6):

- 1 Operation Amplifier
- 1 Toggle Switch
- 2 LED's
- 6 Resistors
- 1 Linear, Dual Gang Potentiometer
- +/- DC Power Supply Voltages

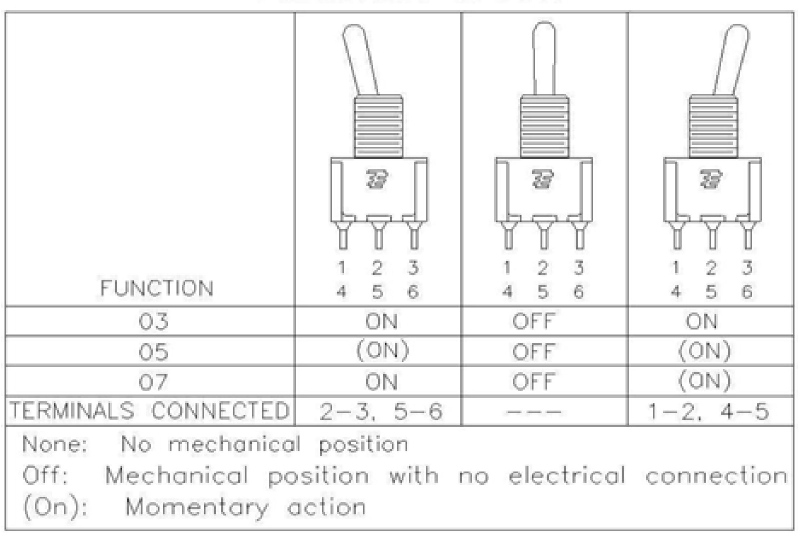
<u>LF351 Operational Amplifier</u>

An LF351 operational amplifier will be used to sum the individual waveforms. It operates at power supply voltages less than +/-18V which is greater than the power supplied to the synthesizer. The opamp will have some noise on the input terminals of which would be susceptible to amplification through the opamp. It is desirable to keep the input noise at a level in the nanovolt range. The opamp will face signals with frequencies between 80Hz and 1400Hz, and the LF351 handles frequencies within that range with input noise voltage less than 20nV which is manageable. Also, harmonic distortion is less than .004% during the working range of frequencies which it will be exposed. The LF351 has low power consumption of 680mW, low input bias current between 20 and 200nA, and low offset current between 100pA and 4nA. It has a relatively high slew rate of 16V/us. They are also low cost and readily available on the market, making them a good choice for the op-amp in the mixing circuit.

<u>Toggle Switch</u>

The toggle switch in the circuit will be double pole, double throw (DPDT), On-Off-On type with a center “off” position. This will allow the user another way to isolate the sine wave from both the triangle and square waves. The toggle will be panel mounted with solder lugs for the printed circuit board which it will be electrically connected. The contacts on the switch must be rated for 10VAC or greater since the input of the mixer will see maximum 5V signals, and typical ratings are determined by doubling operating parameters. Figure 3.6-2 shows a the functional diagram of an On-Off-On toggle switch.

When the square wave is selected, terminals 2 and 3 will be connected to the input waveform while terminals 5 and 6 will be connected to the LED corresponding to the square wave. When the triangle wave is selected, terminals 1 and 2 will be connected to the input waveform and terminals 4 and 5 will be connected to the LED corresponding to the triangle wave. When neither of the waveforms are selected, the toggle will rest in the center position with no electrical connection to either waveform or LED.



FUNCTION	1 2 3 / 4 5 6	1 2 3 / 4 5 6	1 2 3 / 4 5 6
03	ON	OFF	ON
05	(ON)	OFF	(ON)
07	ON	OFF	(ON)
TERMINALS CONNECTED	2–3, 5–6	---	1–2, 4–5

None: No mechanical position
Off: Mechanical position with no electrical connection
(On): Momentary action

Figure 3.6-2: DPDT toggle switch functional diagram
(Permission Pending)

The LED's used in the circuit will be purchased from Skycraft and are expected to be yellow or red since they are typically the cheapest colors available. In the case of either, a voltage drop on the range of 1.6V to 2.2V is expected with no more than 120mW of power dissipated across the LED. Significance will not be placed on luminous intensity or operating temperature during selection. Consideration will be allotted to the maximum reverse voltage, which will not be less than 5V, and the maximum reverse current, which will not be less than .0002% of the maximum reverse voltage.

The resistors labeled R1 and R2 in the circuit will be 1kQ resistors as to not attenuate the input signal too greatly. The resistors labeled Rf and R3 will be determined upon testing the input signals, once generated by the voltage controlled oscillators. These resistors control the gain of the circuit and will be chosen in order to maintain unity gain between the input and the output of the mixing stage shown in figure 3.6-1 above. Resistors with a tolerance of 5% will be chosen and purchased from Skycraft.

Dual Gang Potentiometer

A linear, rectangular, 10kQ dual gang potentiometer will be used for proportion control with a tolerance no greater than +/-20%. Linear was chosen as opposed to logarithmic because it will be offering proportion control to voltage signals which are inaudible. The dual gang pot will be through-hole mounted with soldering lugs for PCB attachment. It will dissipate no more than 100mW of power and have dimensions no greater than (3.5"Lx0.6"Wx0.35"H) for purposes of conserving space on the synthesizer box. The potentiometer will be purchased at Skycraft based on availability and price while maintaining compliance with the listed parameters.

Figure 3.6-3 shows the output waveforms as each of the two inputs are proportionally adjusted. In the figure on the left, the potentiometer was set to 65% sine wave and 35% square wave. In the figure on the right, the potentiometer was set to 35% sine wave and 65% square wave.

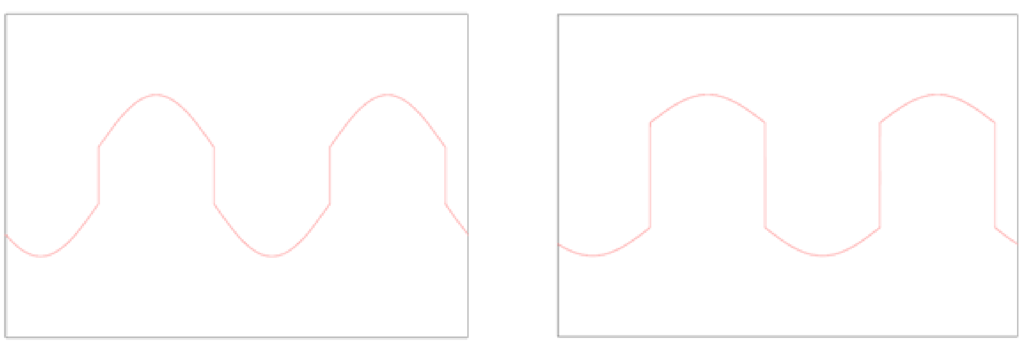


Figure 3.6-3: Proportionally adjusted sine wave and square wave; 3.6-3a (Left) is set to 65% sine and 35% square; 3.6-3b (right) is set to 65% square and 35% sine

As can be seen from the figures above, the proportion can be controlled very well with a dual gang potentiometer; however, they are typically manufactured with a tolerance of +/-20%. This implies that the maximum in-equivalence between the two states of the potentiometer is 40%. Figure 3.6-4 below shows the effect on the output signal if the two waveforms are at the maximum 40% out of synch with each other. In each case, the sine wave was intended to appear 65% on the output signal as shown above in figure 3.6-3, while the square wave was to contribute 35%. Considering a maximum 40% error on either direction, figure 3.6-4 shows the output waveform when such an error exists. In the figure on the left, the sine wave contributes 85% and the square wave 15%. In figure on the right, the sine wave contributes 45% while the square wave contributes 55%. In each photo, the red wave is desired proportion of 65% sine and 35% square, and the blue wave is the deviation caused by tolerance of the potentiometer.

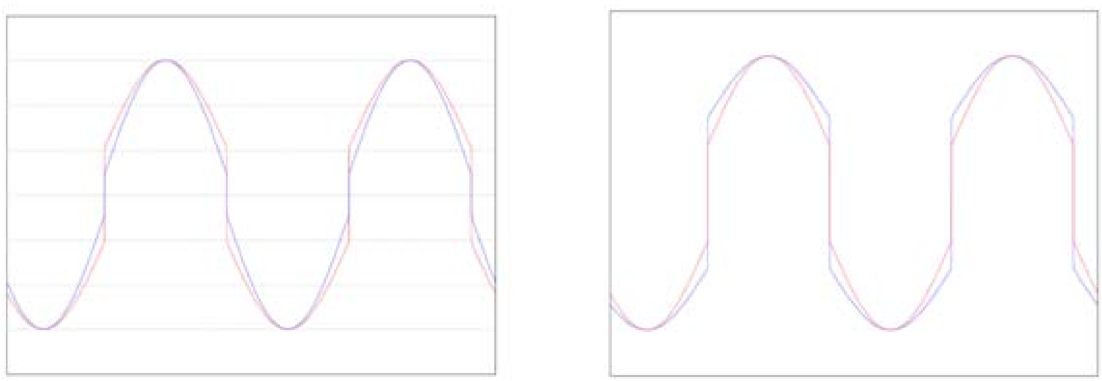


Figure 3.6-4: Proportionally Adjusted Sine Wave and Square Wave with Deviation Due to Tolerance of Dual-Gang Potentiometer

Upon testing the actual tolerance of the potentiometers purchased, trim potentiometers may be used in order to equalize the resistance of both ends of the dual gang potentiometer by attenuating the input if the resistance is too low; however, as seen in the figures below, the deviation is quite small and may be deemed acceptable after testing has taken place.

3.7 Low frequency Oscillator

The low frequency Oscillator is based off a Wein Bridge Oscillator as seen in figure 3.7-1. This type of oscillator is able to produce a sine wave output. The frequency is determined by the R, R1, C and C1 values and the feedback gain is produced by ratio between R2 and R3. In order to maintain a stable oscillation R2 should be roughly over twice the size of R3, which would offer a gain of two to the overall circuit.

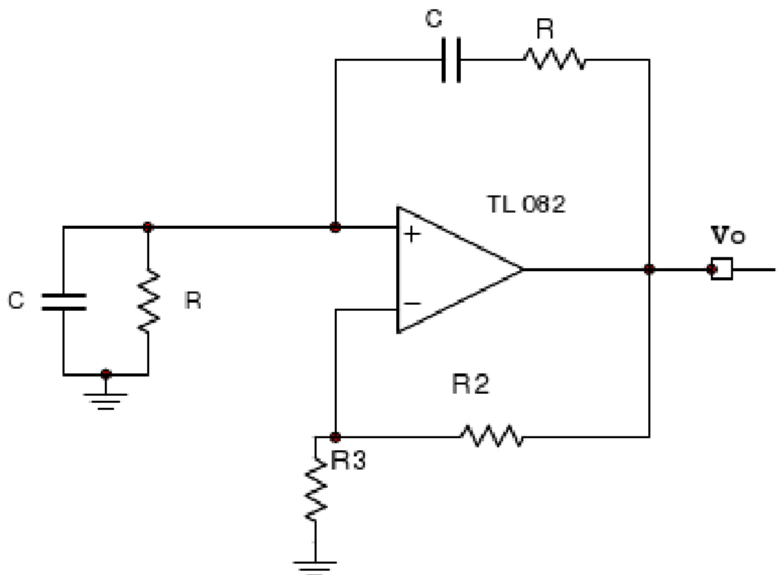


Figure 3.7-1: Simple design of the Wein-bridge Oscillator

The relationship between frequency and the components are shown in the following equations.

$$f = \frac{1}{2\pi RC} \qquad R_2 = 2R_3$$

The design will use TL082 Op-amps. The design above is improved on by adding limiting diodes D1n4448 to each input of the op amp. This makes sure the gain stays manageable and the circuit oscillates at a stable rate. Two resistors that control the frequency are now split into two series resistors each. The first will have a set value of 100 Ohms and the second is a 100kOhm potentiometer. Because there are two potentiometers and both need to have the same value they are condensed into a logarithmic 10kOhm stereo potentiometer, this way both can be varied proportionally to each other. This potentiometer and capacitor values will determine rate of oscillation. The capacitors are bipolar electrolytic capacitors with a value of 47uF. The following equation shows that when the resistor in parallel with the capacitor in figure 3.7-1 is at a minimum value, the circuit oscillates at 33.86Hz.

$$f = \frac{1}{2\pi\, 47\, uf\; 1\Omega} = 33.86\, Hz$$

When the resistor is at a maximum value, the oscillation is at a minimum value. When the resistor is at its minimum, this defines the highest rate of oscillation. The gain is controlled by the ratio of R2/R3. This will be varied later in the design to control amplitude of the oscillator. Since R2 needs to be slightly higher than R3 the value for R3 is chosen as 1K and R2 is chosen to be 2.2K. The gain will be controlled later by separate op amps in order to keep the design at a stable state.

There are two options as to what type of oscillation is produced, both square and sin wave oscillation, depending on where in the circuit the voltage is being read. In order to be able to read these voltages but not disturb the circuit a total of 3 buffers will be added, one for the square waveform, sinusoidal waveform and one for the LEDs that visually show the rate of oscillation. The LEDs are terminated by a 4.7kOhm resistor in order to limit the current so the diodes will not burn. The last two op amps that are terminating buffers have a 100kOhm stereo potentiometer in order be able to adjust the gain of oscillation without becoming unstable. The final design for the LFO is shown in figure 3.7-2.

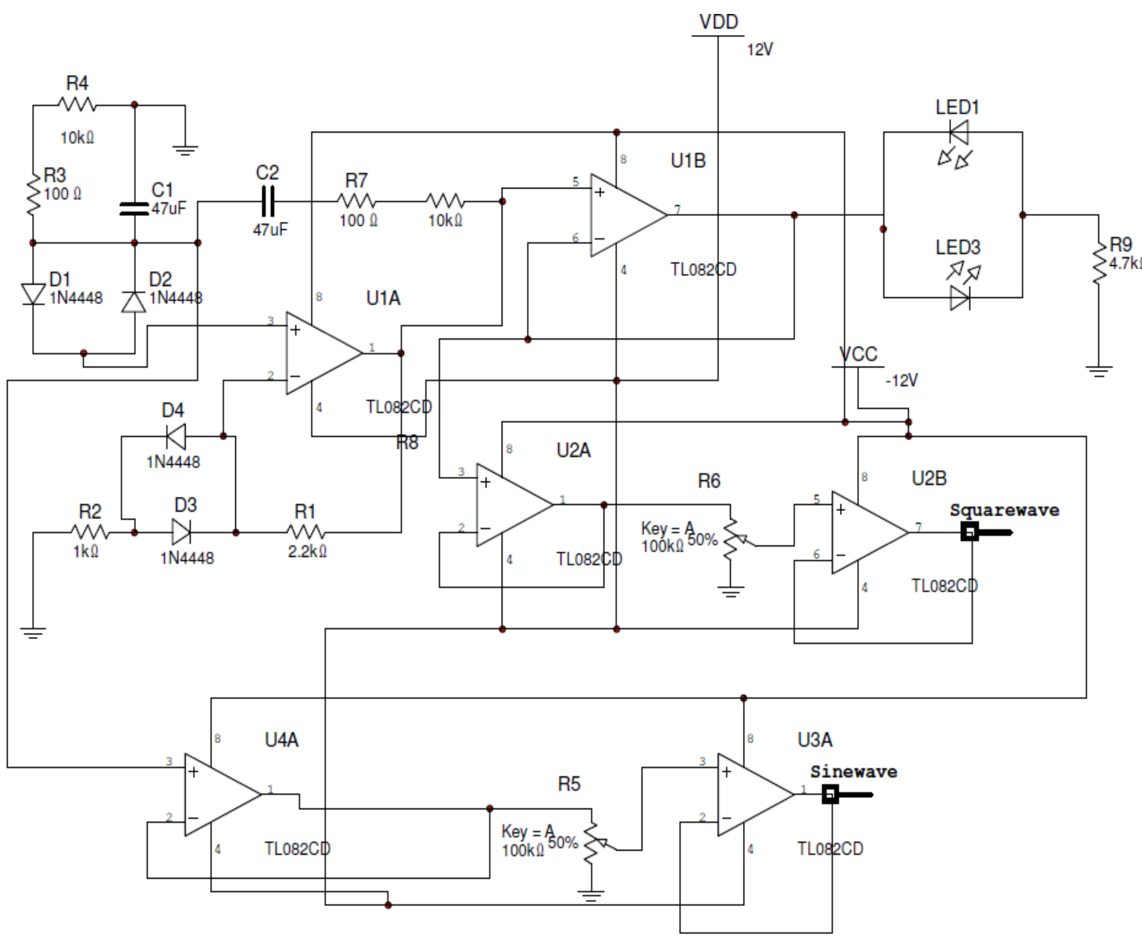


Figure 3.7-2: Low frequency Oscillator

The first op-amp in the circuit is a Wein-Bridge oscillator. This ranges from .33Hz to 33.3Hz. U1B is used as a buffer to light up LEDs when the signal is high and Low. U2A is used to turn the signal into a square-wave and U4A is outputting a sine wave. The two op amps following them are used as a gain control.

Vibrato

The vibrato effect is the rapid increase and decrease in volume and can be done quite easily. In our design it is the easiest to add the vibrato right after all the signals get mixed into one. This is because there is little to no purpose of having vibrato on individual strings. When all the signals are mixed into one the complex signal then gets fed into the input of a voltage controlled amplifier. The voltage feeding the VCA will be obtained from the LFO assuming that the maximum value the LFO can output is +-1v added to a DC voltage of 4v This makes the max value of the CV feeding the VCA to be 5v and the minimum value to be 3v. The rate on the LFO will effect how quickly the swells and decays of volume occur and the depth will determine how how amplified or attenuated the output will be. With a normalized input of 4v the VCA is configured to have unity gain. At the min input of the 3V, the VCA is configured to have a very small output volume.

Tremolo

The tremolo effect is the quick decrease and increase of pitch in a note. The flow chart is shown on figure 3.7-3. This effect can be achieved easiest where then pitch is converted into a DC voltage which is right after the frequency to voltage converter. The LFO signal is used for this effect but the max value is attenuated to be +-.1v (because of the sensitivity of the VCO). Then this signal is added to the signal from the frequency to voltage converter to vary it by +- .1v which will ultimately effect the pitch of the note after it goes into the VCO as seen in the figure below. The rate on the LFO will effect how fast the pitch goes up and down and the depth will effect how high and how low it goes in pitch.

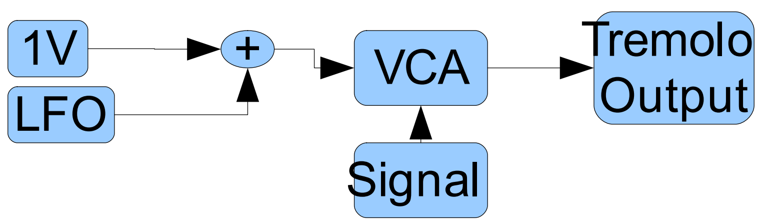


Figure 3.7-3: Tremolo flow chart

3.8 Voltage Controlled Filter

There are several options for the Voltage Controlled filters all based of a current generalize admittance structure (CGIC). These structures provide a second order low pass filter with the ability to tune certain resistance values to affect the gain, cutoff frequency and Q value. The design that is being used is seen below in figure 3.8-1**.** Because of the nature of the design, if one component gets trimmed, it can effect more then just the cutoff frequency or the resonance. The design below allows for several different options and designs by changing which impedance will be trimmed. This analysis is done by looking at the formula shown for the low pass filter.

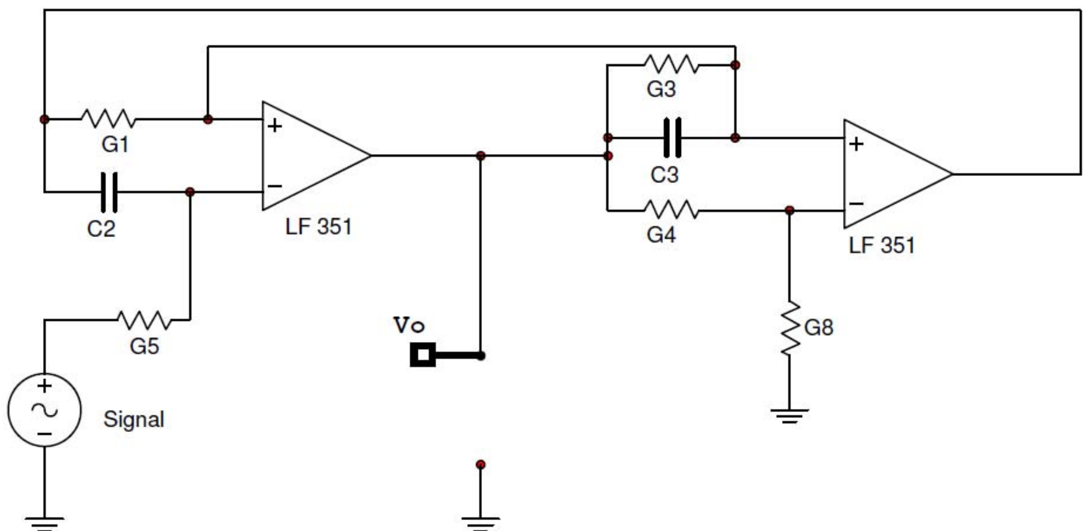


Figure 3.8-1: CGIC Design

In the above image, each individual parameter can be tuned to effect different results in the gain, cutoff frequency and resonance. This design can work as a low pass, band pass or high pass depending on where the signal is inputted and where the signal is outputted. For a low pass the design the signal should be read after the first op-amp The formula is seen below for the transfer function. Through manipulating the transfer function the cutoff frequency and resonance.

$$h(s)=\frac{sC_2(sC_3+G_3)}{G_1G_4} \qquad T(s)=\frac{G_1G_5(G_4+G_8)}{G_1G_5G_4+sC_2G_3G_8+s^2C_2C_3G_8}$$

Transfer function for CGIC is shown above. Rearranging the transfer function shows the Q and w values.

$$T(s)=\frac{1+\frac{G_8}{G_4}}{s^2\frac{C_2C_3G_8}{G_1G_5G_4}+s\frac{C_2G_3G_8}{G_1G_5G_4}+1}$$

If one parameter is changed in one formula it can effect the next formula depending on if its present within that formula. Notice that the gain has G8 and G4 controlling the gain. If one was to want to change the cutoff frequency, either the G8 and G4 would have to be tune simultaneously or another parameter will need to be chosen.

$$w_p^2 = \frac{G_1 G_5 G_4}{C_2 C_3 G_8}$$

$$Q_p^2 = \frac{C_3 G_1 G_4 G_5}{C_2 G_3^2 G_8}$$

Here the gain value is effected by the ratio of G8 to G4. In order to tune the gain of the circuit these are one of the two attendances that can be changed. The formula for the cutoff frequency is shown above. If one wanted to change the cutoff frequency and maintain the same amount of gain, then the values G4 and G8 cannot be trimmed. This leaves two other impedances that can be trimmed in order to change the cutoff frequency of the circuit. The bode plot of this configuration can be seen in figure 3.8-2.

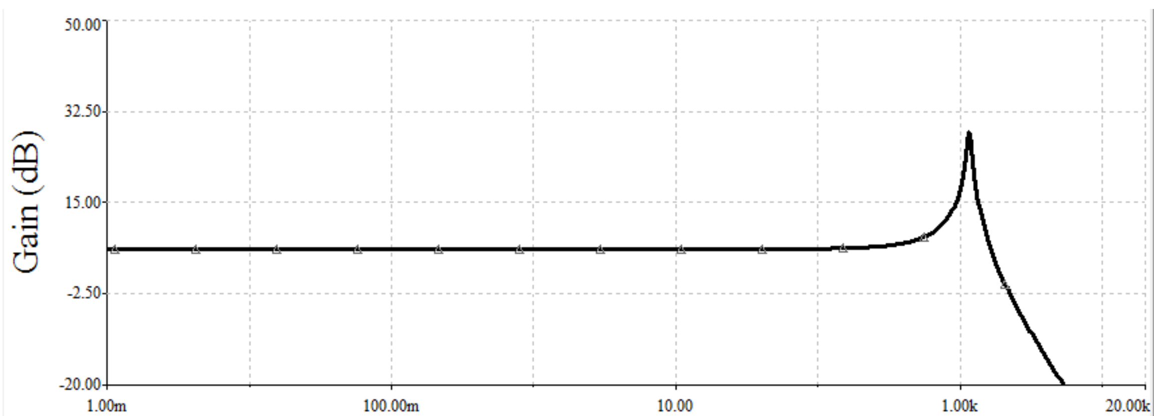


Figure 3.8-2: Cutoff & Resonance with high Q and w values

The plot above corresponds to the filter where the Q and w values are at their maximum and the over all gain is not attenuated. This is likely to peak off the op-amps and create some distortion and undesired effects in the sound of the circuit.

Looking at the formula for the resonance, there is only one impedance that can be changed in order to not effect the rest of the parameters. The resistor G3 can be tuned in order to only effect the resonance of the circuit. Ideally this filter will keep a unity gain as the user tunes the cutoff frequency, and tuning the cutoff frequency should not effect the Q value. The Bode Plot of this circuit is seen figure 3.8-3**.**

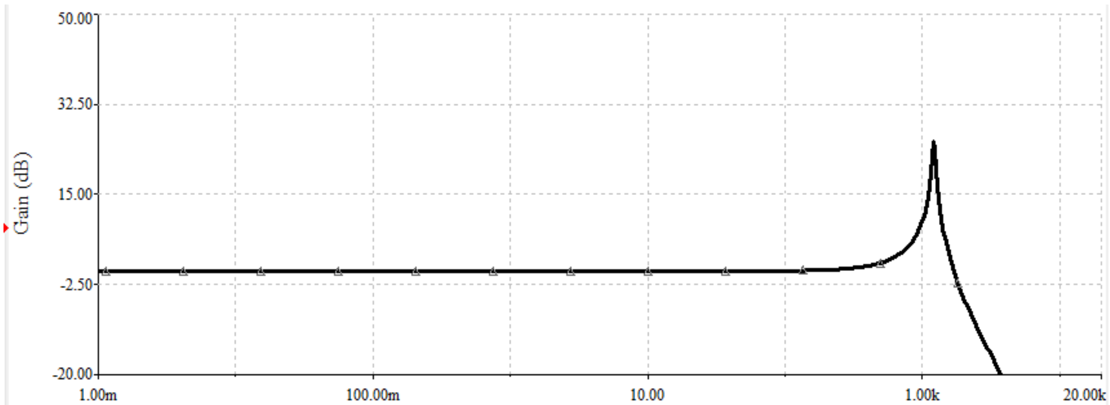


Figure 3.8-3: Bode Plot for design 2.

Here the cutoff frequency is at its highest value and the Q value is also at the highest value. In this design as the resonance gets spiked up, the gain begins to go down gradually. This is more of a desired effect because instead of giving the system too much gain (unity gain is ideal for filter), it isolates the resonance frequencies. This possible design would provide too much gain in the resonant frequencies and would need diodes to limit the amount of gain that the filter is putting out. The second configuration Requires changing the tuning parameters to another resistor. A better option for resonance would be a design that as Q value goes up. The gain of the circuit goes down proportionally, this way effecting Q would not only spike the resonant frequency but also attenuate all the frequencies seen passing the low pass filter. This is a more desired resonant effect because it isolates the resonant frequencies without giving it too much overall gain. This circuit can be seen below in 3.8-4.

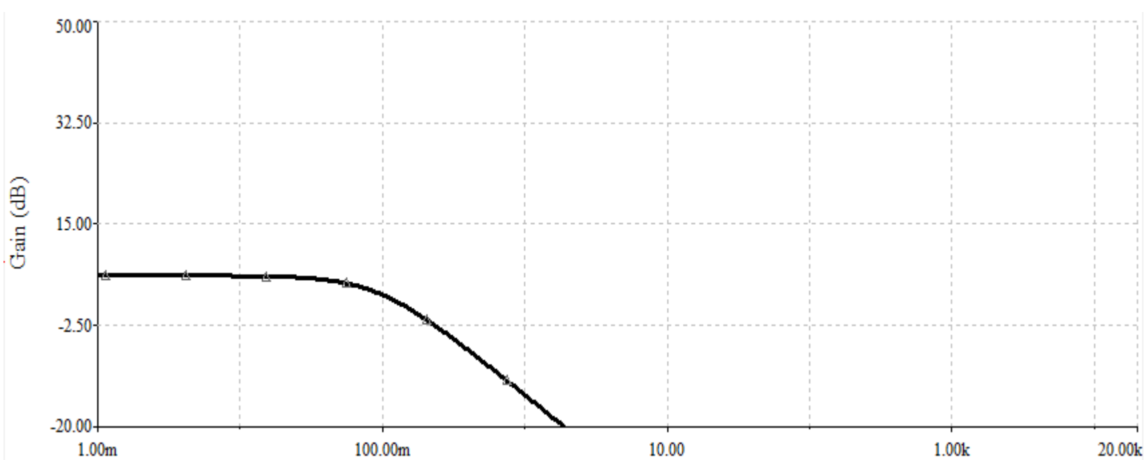


Figure 3.8-4: Bode Plot for Design 1 with low Q and w values.

This design gives a higher lowest value than the design above with a cutoff of approximately 60Hz which is still below the lowest frequency attainable by the guitar. This means at the lowest cutoff and resonance value are still below the natural guitar range. There is also no resonance in this plot.

In this circuit the Q value is tuned by changing the G4 value, by changing this value the gain of the overall circuit will be decreased. This will also sacrifice how high the cutoff frequency is because G4 is also seen in that formula. The effect of this can be seen below in figure 3.8-5.

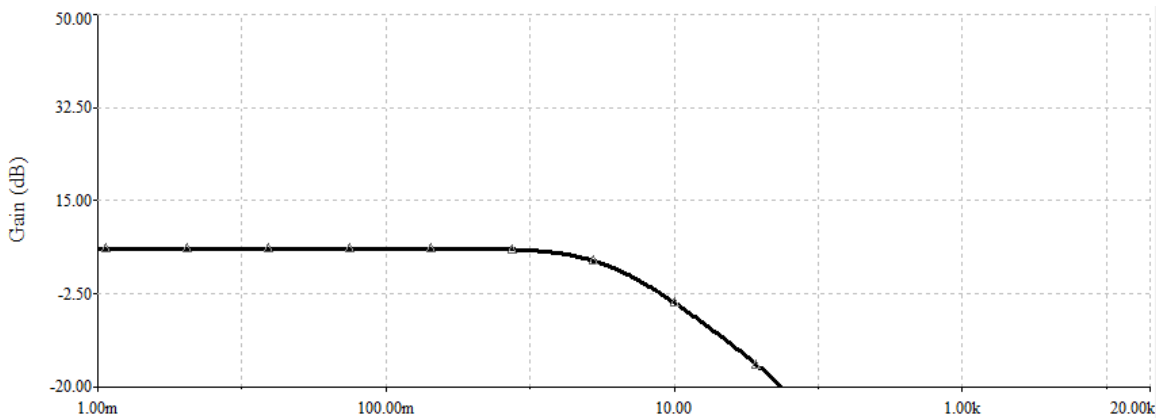


Figure 3.8-5: Bode plot for second configuration with both the cutoff frequency and resonance at a minimum value.

Since these types of parameters are based on resistance values there are other options other than just trimming resistances to vary them. A voltage controlled resistor would be a possible option when being controlled by something such as the low frequency oscillator or possibly the envelope detector. The circuit in figure 3.8-6 takes in a voltage and outputs a resistance to a certain designed value. The voltage can be read from the envelope detector with a proportionate amount of gain so the cutoff and resonance of the circuit will swell and decay proportionate to the envelope of the notes played. From the designs above for the low pass filter, if G4 is increased, this increases the Q value and Cutoff frequency but reduces the overall gain. This option will require a switch that removes the resistor that is being used for the cutoff and resonance and replace it with a voltage controlled resistor in series with 15.1kOhm. The output of the envelope will be scaled proportionately so the highest average value is 1V. This is done with a simple non inverting amplifier. This way the filter is at its max points for the maximum points in the envelope. This resistor is used to in the place of a resistor (G4) in the CGIC in order to make that circuit tunable by an input voltage. The desired value of this output resistance should be 15.1kOhm which will be in series with a 100 Ohm resistor. The controlled voltage will come from the envelope detector so when the user is actively playing loud the resonance and cutoff frequency will follow to give the effect of a “wah” following the envelope.

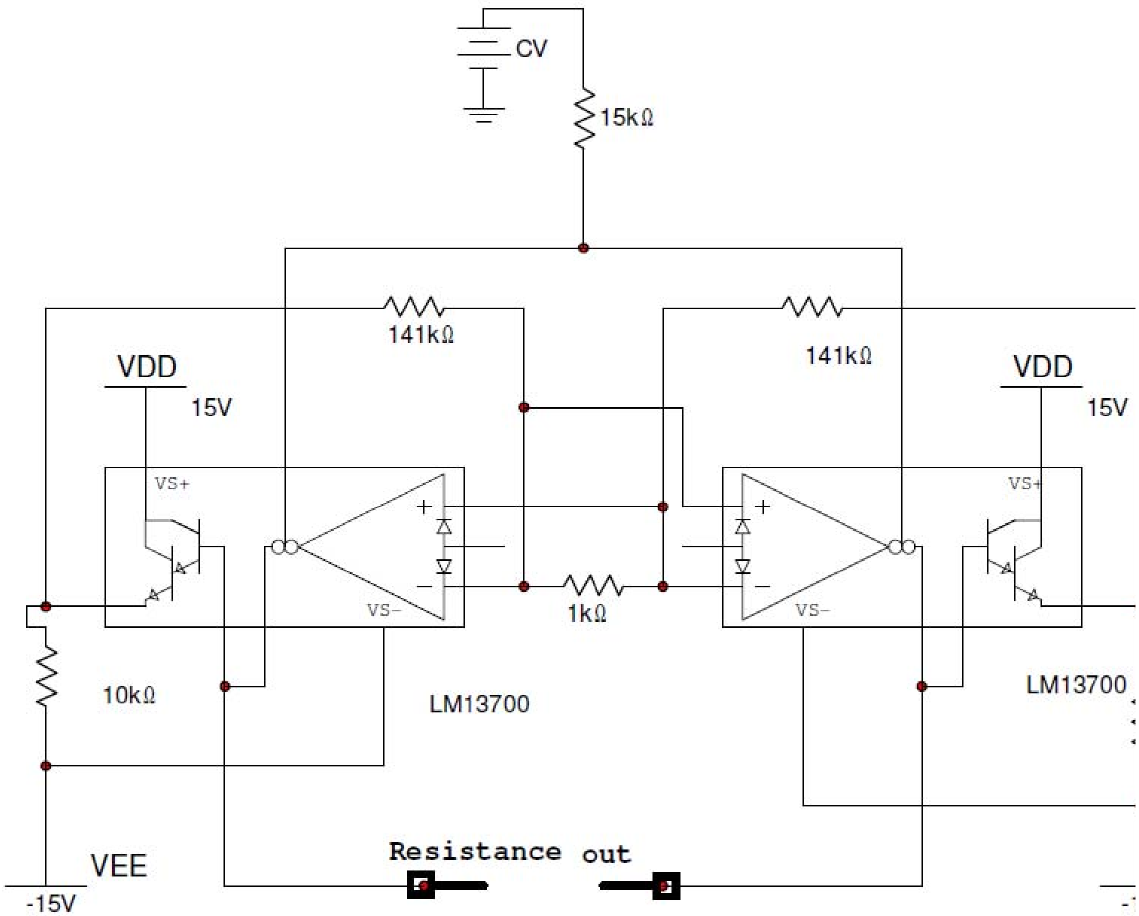


Figure 3.8-6: Voltage Controlled Resistor.

3.9 Voltage Controlled Amplifier

For the voltage-controlled amplifier, the decision was to use an operational transconductance amplifier, which takes in a control voltage, outputs a proportional current which is sent into a buffer circuit. One of the six VCA circuits is shown in Fig. 3.9-1, and uses a LM13700 operation transconductance amplifier with a supply voltage of ±9V. A split supply was chosen to avoid having to use a buffered voltage divider for the middle bias point of the amplifier. Also, the supply range is 18V instead of the usual 24V seen elsewhere in the design. This was chosen to keep the control voltage from being to close to the negative supply rail of its buffer. The control voltage, CV, is applied at the CV In, and a current, I_{ABC} is goes through R_9 and R_7 which is a 10kΩ potentiometer that allows the user to change the sensitivity of g_m, the transconductance, to the control voltage. It will act similarly to a volume setting, but will also affect the LFO, since this signal will also be sent through this resistor. The input signal, applied at the inverting terminal of the amplifier, is then amplified proportionately to the control voltage. The 2kΩ trim potentiometer, R_{10}, is used for biasing the differential input so that rapid changes in the control current are not bled into the output, which would result in an audible “click” or “pop”. The control voltage from the ADSR circuit is effectively a 0 to 5V signal, which is lowered with the same scale to a range from -9V to -4V, since the control voltage must be at -9V for the signal to be fully (or close to fully) attenuated. This is achieved by applying the control voltage to a positive summing amplifier with a -9V input, effectively, shifting the control voltage range negatively by 9V. Finally, the output is AC coupled with C_2.

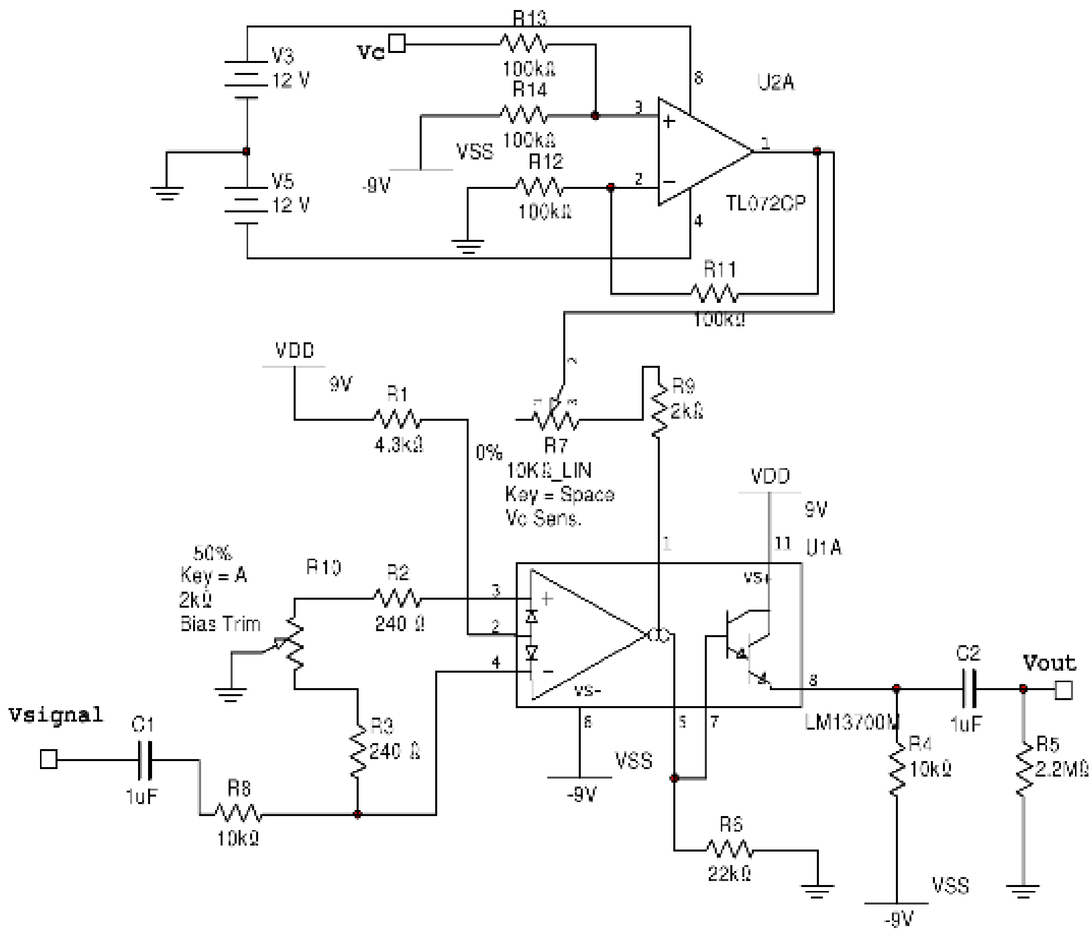


Figure: 3.9-1. Voltage controlled amplifier circuit.

This circuit has a fairly linear Vo/Vin with respect to the control voltage. The cutoff point, defined here as the control voltage needed for full attenuation of the input signal, is about -8V for any value of the series combination of R_7 and R_9. Table 3.9-2 shows a map of the output voltage given a 1kHz sinusoidal input voltage with an amplitude of 1V. The results show that with the resistance at its minimum of 2kΩ, the maximum amplification of the input signal is 2, and conversely, when the potentiometer is at its max resistance, the input signal is attenuated with any value of the control voltage. The potentiometer will be wired as a variable resistor across its second and third pins such that when it is turned clockwise, the resistance is lowered, and conversely when it is turned counter clockwise. This gives the user a volume control for each VCA is desired.

	Vout Amplitude Vin = 1V		
CV	R7+R9=2k (pot 100%)	R7+R9=7k (pot (50%)	R7+R9=12k (pot 0%)
-9	14nV	14nV	13nV
-8	147μV	19μV	18μV
-7	357mV	113mV	70mV
-6	898mV	276mV	166mV
-5	1.43V	440mV	262mV
-4	1.92V	600mV	360mV

Table 3.9-2: Output voltage versus control voltage R7 position

When the results were graphed against each other in Fig. 3.9-3, the amplitude showed a fairly linear response to control voltage after about -7V. Each colored line represents the different resistance that was used for the simulation. The results are positive and show that the envelope of the VCA output will match, with good accuracy, to the envelope that is generated by the ADSR circuit, with mainly a difference in amplitude and possibly a small difference close to the signal's minimum, since the graphs is not perfectly linear in that region.

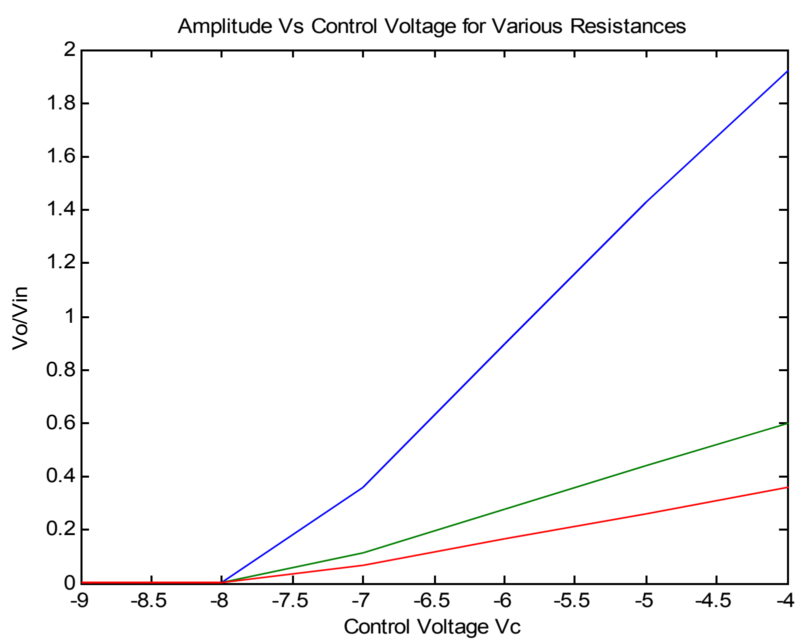


Fig. 3.9-3. Vo versus Vc for min (blue), mid (green), and max (red) R_7 values.

Another way of introducing control to the output amplitude is by changing R_6 at the output of the amplifier to ground, before the buffer. Increasing this resistor increases the overall amplitude. Raising this could help keep the input bias current from reaching its rated max of 2mA. This could be done be raising both R_9 and R_6, resulting in less input bias current, but more current reaching the buffer, since R_9 is connected across the input of the buffer to ground. Testing will need to be done with real components to ensure that the output is stable and that the input bias current is not too high.

3.9.1 Tremolo Input

The tremolo effect will be added as an input to the summing amplifier to each of the control voltage inputs of the VCA section. The effect is not string specific, but is added to the entire sound of the instrument as a variable volume swell. All that is needed is another 100kΩ resistor in parallel with the other two in the control voltage summer, which would add in a low frequency sine,square, or triangle to the control voltage. The amplitude will need be variable in order to let the user select the amount of volume swell. This variable amplitude will be taken care of by the low frequency oscillator section, or the LFO. Fig. 3.9-4 shows a simulated tremolo effect. The triangle wave in green is the LFO signal, which varies from 0V to 5V. The input is a 1kHz sine wave (in red) and the output it the blue wave. Clearly, the amplitude of the output follows the voltage of the triangle wave.

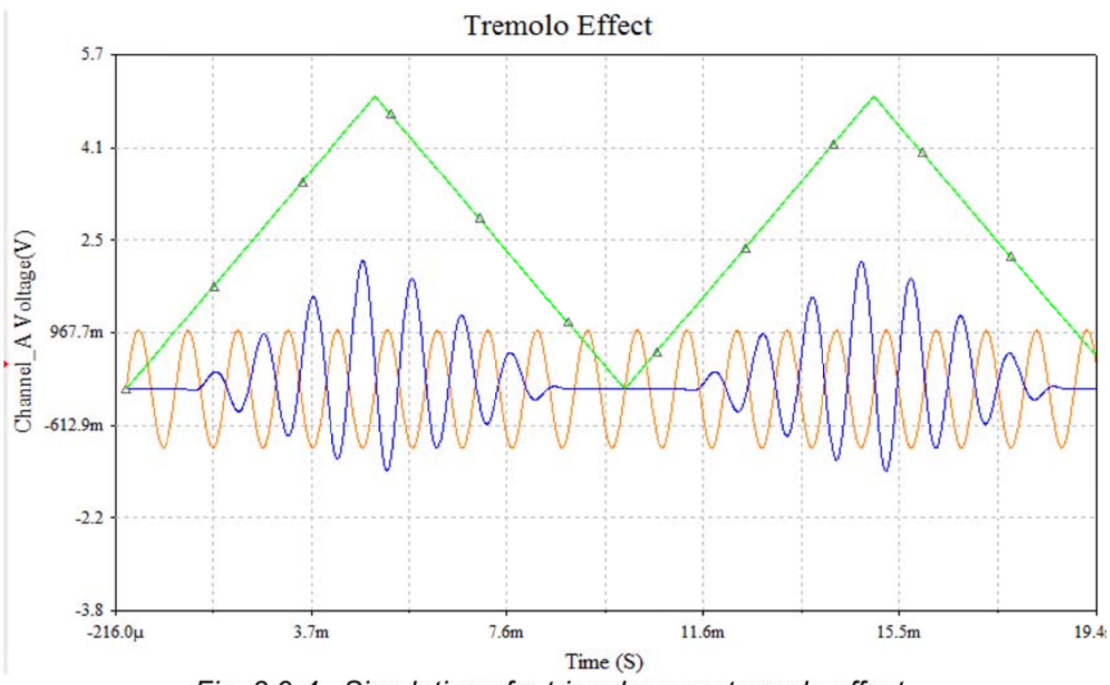


Fig. 3.9-4. Simulation of a triangle wave tremolo effect.

There is another "effect" that can be generated using the tremolo. By increasing the LFO frequency to within the guitar's audio range, then the tremolo acts as what is called a ring modulator. Ring modulation is the sum and difference of the frequencies being multiplied together. The side bands are present at the output, creating a very strange sound that interacts seemingly unpredictably when playing different notes. In the case of the guitar synthesizer, ring modulation should begin to become apparent close audio frequencies, somewhere around 20Hz and upwards. Ring modulation is an effect that is generally found in synthesizers and even as stand alone guitar effects, so bringing in this feature adds not only a richness to the sound created, but another creative avenue for the player to experiment with.

The cost involved in building the VCA sections is dependent mainly on accuracy of components and the fact that there will be six of each component, not to mention the added board space. Because there needs to be six VCAs for a polyphonic sound, this section might be the most costly. However, it is one of the most important parts of the synthesizer, since Clearly, surface mount components will be purchased for the final prototype, and soldered to a double-sided circuit board. The estimated cost of the components is broken down below:

- 1% metal film : $0.06 mouser (13 each VCA)
- 10% multilayer ceramic capacitors : $0.19 mouser (2 each VCA)
- NJM13700M (cross reference for LM13700): $0.61 mouser (0.5 each VCA)
- TL074 dual op-amp : $0.69 mouser (qty. 1)
- TL084 quad op-amp : $0.63 mouser (qty. 1)
- Bourns 2kΩ trim potentiometer : $0.50 mouser (1 each VCA)
- 10kΩ 16mm linear potentiometer : $1.25 smallbearelec.com (1 each VCA)

Total: $20.61

3.10 ADSR Envelope Generator

The ADSR envelope generator will be realized using a microcontroller, and a gate and trigger circuit. The microcontroller will use the principle of direct digital synthesis (DDS) to create the envelope. This envelope signal will then be sent into the control voltage input of the VCA, to vary the amplitude of the VCO output in proportion to the desired shape. Shaping controls will give the user the ability to change how long each of the attack, decay, sustain, and release cycles will last. For the attack, decay, and release, the control will determine the length. For sustain, the control actually determines the sustain level, or the level which the decay will drop down to, then hold while the gate signal is still high.

3.10.1 Generating Gate Signals

The main inputs to the ADSR block are the trigger and the gate, which gives the MCU two pieces of information from which to output an appropriate envelope. The trigger must be a short pulse corresponding to a digital “high” that triggers the event of beginning the attack cycle. This information corresponds to new string pluck, which is where the user begins to articulate a new note. Sometimes guitarists will not pluck a string to change notes, but change the position of their fretting fingers to get a new tone. This doesn't create a new attack in a real guitar's envelope, so it should not in the case of the guitar synthesizer. The gate, however, will remain open for as long as the input signal from the pickup will be above a certain threshold. This will effectively keep the amplitude of the VCO's output above zero until the note dies off, or the string reaches a point where its vibrations no longer have a large enough amplitude to be heard on a guitar. The circuitry for the trigger must therefore include a one-shot pulse that is sensitive to a quick positive edge input signal, so that a player's string pluck is interpreted as a trigger.

To realize this, the input signal must first be sent into a simple envelope detector, which will take the positive portions of the input and filter them to get a rough interpretation of the amplitude of the signal with respect to time. Fig. 3.10-1 shows the envelope detector followed by the comparator. A standard small signal diode, 1N4148, has been chosen for its relatively low cost and size. The RC time constant must be high enough to reduce voltage ripple, which could possibly send false trigger signals, and low enough to reduce negative peak clipping. Fortunately, negative peak clipping isn't as much of a concern as voltage ripple, since it would only result in a slightly longer gate time than desired. But this could be counteracted by turning the gate sensitivity down, which would lower the sustain time by cutting off the gate before the envelope signal reaches a lower voltage. The RC time constant was chosen to be 20ms, which keeps ripple down. The comparator uses hysteresis to control when the gate signal goes high and for how long. The length of time depends on the actual amplitude of the resonating string, and how long it stays above the comparator threshold. The hysteresis was set up to have a high threshold of 0.5V and a low threshold of 0.1V. Until the pickup is made, these values will have to suffice for design. The pickup will most likely have a lower output voltage swing than a standard guitar pickup since the number of turns will be reduced in order to keep size down. However, with the first stage filters, there should be some amplification at least at the lower frequencies of each string. The ratio of R_2/R_1 was determined using the equation,

$$\frac{R_2}{R_1}=\frac{V_P-V_N}{V_{TH}-V_{TL}}=12.5$$

where V_P = 5V, V_N = 0V, V_{TH} = 0.5V, and V_{TL} = 0.1V. This results in a ratio of about 12.5. Using a close ratio of 12, the values chosen were R_1 = 10kΩ and R_2 = 120kΩ. For the reference voltage, the equation used was,

$$V_R = \frac{V_{TH} R_2 + V_{TL} R_1}{R_1 + R_2} = 0.47\text{V}$$

where V_R is the output of a simple resistor voltage divider involving R_3 and R_4, such that R_3 = 100kΩ and R_4 = 10kΩ. Each of the resistor values in this section are designed using common component values. If more exact value resistors can be purchased without too much of a cost increase, then the realization will become closer to the design.

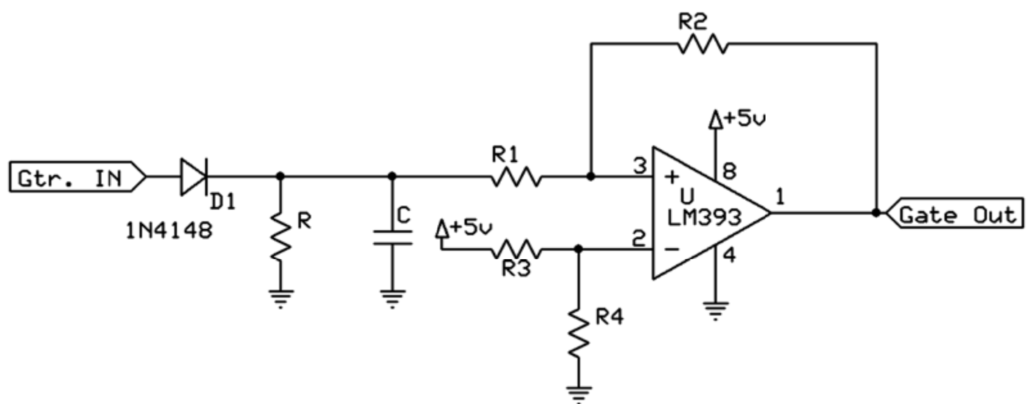


Figure 3.10-1: Gate generator circuit.

This circuit can be realized for six channels using one LM339 quad comparator and one LM393 dual comparator. Each can be purchased for under $0.50 from Mouser Electronics.

3.10.2 Generating Trigger Signals

The trigger signal is merely a one-shot pulse generator that reacts to a spike in amplitude. It must be sensitive to react to slight string picking, but not sensitive enough to trigger every for every positive swing of a wave. The problem with trying to use a guitar's signal to generate a trigger is that there is potentially a lot of noise in the signal, which could produce false triggers. Such false triggers could be as simple as bumping one's hand against a string. Also, a successful trigger must be produced when a string is plucked at any intensity. So simply using a certain threshold for the signal to rise above in order to generate a trigger is problematic if the play decides to strum softly for the next part of the song, for instance. Using the positive edge of the gate as the trigger carries with it a similar problem. If the gate is still high when the string is plucked again, then there will be not positive edge to detect as a trigger for a new attack. The threshold level must therefore be variable and able to handle any picking intensity. The circuitry involved in this process is much to large to be done in an analog sense and certainly too large to include all six channels. Another method had to be produced since this is such a critical component in the synthesizer.

Another method is to take on the task digitally through the software of a microcontroller. If the output of the envelope detector, before the comparator input of the gate generator, was taken and sent into the analog input of the microcontroller, then the software could sample the input and track the signal level in real time and dynamically allocate threshold levels. Logically, a new note could be determined by looking at the successive peaks of the envelope signal. Unfortunately, a simple diode and RC filter envelope detector, like the one used in this design, suffers from peak ripple which must not be mistaken for a new attack. If the input is sampled often enough, then each new peak in the ripple could be recorded and compared with the next peak. A peak in this case is found when the current sample is lower than the previous sample. To be sure of a peak, multiple samples could be used in this comparison to ensure that there was no noise creating a false peak. Once the software finds a new peak, it holds that value and compares it with the previous recorded peak value. A trigger in software goes high when the new value is higher than the previous value. The assumption made here is that the signal's amplitude only decreases after the initial string hit and never again reaches that height if there is no other attack. To be safe, a threshold of something like 10% could be used to determine an attack's presence. The trigger in this case is not an electrical signal, but is a Boolean variable that the software uses to determine when to begin the attack cycle.

3.10.3 Microcontroller for Envelope Generation

The microcontroller that was chosen for the job of taking in each gate, trigger, and analog control input is the MSP430f4132, which offers all of the necessary peripherals at the lowest cost. Several factors led to the decision of the MSP430 series. Texas Instruments offers a small development platform called the MSP430 Launchpad (MSP-EXP430G2), which ships with two of their low cost, value line chips. These chips only have one PWM channel and not enough I/O pins for the requirement of 22 for the ADSR (6 trigger inputs, 6 gate inputs, 4 analog inputs, 6 PWM outputs). The Launchpad uses flash emulation to connect to a computer via USB, making programming and debugging less problematic and more useful on new computers than a serial or parallel port programmer. The two best features of the TI Launchpad is its low cost of $4.30 and its ability to program any of the MSP430 line of microcontrollers that feature the Spy-Bi-

Wire interface. This means that a low cost programmer may be utilized for programming high-end MSP430 chips, such as the MSP430f4132.

Specs for the MSP430f4132

- 8KB Flash, 512B SRAM
- 56 programmable I/O pins
- Timer_A0 with 3 capture/compare/PWM
- Timer_A1 with 5 capture/compare/PWM
- Watchdog Timer, Brown-Out Reset
- 8 channel 10-bit ADC
- low-profile LQFP or VQFN 64-pin packages
- Spy-Bi-Wire programming interface
- Low Power 1.8 to 3.6 V
- Low cost: $4.56 (mouser)

<u>3.10.4 Software Flow</u>

Programming the MSP430f4132 will be achieved by using the mspgcc tool chain which includes the GNU C compiler, an assembler and linker, and a debugger. It is an open source, multi-platform software package that offers unlimited code size and functionality, unlike the version of the CCS Compiler that originally comes with the TI Launchpad, which has a code limitation of 2kB. Fig. 3.10-2 shows the general input and output flow of the ADSR. The triggers will cause appropriate interrupts which will begin each ADSR cycle. The chip will then periodically (on the same order of speed as the trigger) read the input pin which corresponds to the gate for that string. Periodically, but not while an ADSR cycle is engaged, the chip will sample the analog inputs, which are tied to linear potentiometers set up as a simple voltage divider. If any value has changed, the chip will update the value in memory. These values are attack time, decay time, sustain level, and release time.

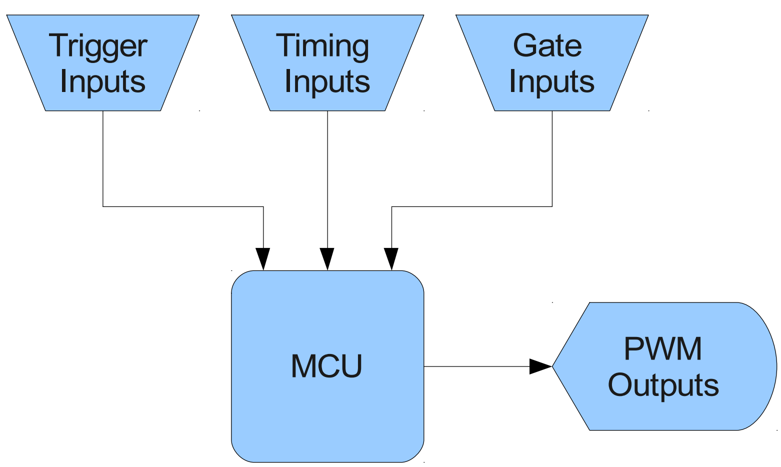


Figure. 3.10-2: Input/Output flow.

The attack, decay, and release cycles will be shaped like an exponential to synthesize an analog RC time constant-based ADSR circuit. Also, being shaped thusly, the ear should interpret the volume change as linear with respect to time, since the human ear senses volume on a logarithmic scale. The exponential curve will be stored in program memory as a lookup table with a width of 256 values that range from 0 to 1023. The inner loop will cycle through these values with a slight delay, in order to keep the timing on a relevant and useful scale. Each value will be stored into the chip's corresponding capture/compare register (CCRx) which will output a PWM with a duty cycle relative to that number. The frequency of each PWM will be fixed so that an appropriate digital to analog converter (DAC) can be designed that correctly outputs the desired signal. The capture/compare peripheral will be configured to count up to the value in CCR0, then restart at zero. Then, using the chip's set/reset mode, the pulse will be reset when the timer reaches the value stored in CCRx and set when it reaches the value stored in CCR0. For example, if the PWM is configured to have a 10-bit resolution and set to have a 75% duty cycle, then the values stored in CCR0 and CCRx will be 1023 and 767, respectively. In this case, CCR0 will remain static while CCRx will become whatever value is grabbed from the exponential curve information stored in program memory. The delay mentioned earlier will allow some time for the PWM to output at the selected duty cycle. Each cycle will end when the final value of the lookup table is reached. This is unlike a standard DDS system which cycles repeatedly, resulting in a periodic wave at some given frequency.

Fig. 3.10-3 shows the ADSR cycle flow chart. Once a trigger causes an interrupt, the attack cycle begins, the chip outputs a signal that is proportional to

$$V_{max}(1-e^{-t})$$

where V_{max} is the supply voltage of the chip, around 3.3V. Periodically, the gate input is read. If the gate is still high, then the attack cycle continues. If the gate drops low during the attack, then the current lookup value is saved and the release cycle begins at this value. However, if the cycle reaches the last lookup value with the gate still high, then the attack cycle ends and the decay begins. The decay behaves similarly, except the wave shape is reversed vertically and scaled according to begin at the last value of the attack and end at the sustain level, read by the analog input and interpreted into a digital number. Once again, the gate is read and the decay either stops and goes into the release cycle or continues until its final value is reached. If the gate is still high at that point, then the sustain cycle begins, which simply holds the value read from the analog input until the gate is dropped low. At that point, the release cycle begins, which looks like the decay cycle except its beginning value is dependent on the last value of the previous cycle (which could be the attack, decay, or sustain), and drops to a value of zero. The full cycle being over, the program then continues to wait for further trigger inputs while periodically sampling the analog inputs in case the user is changing settings.

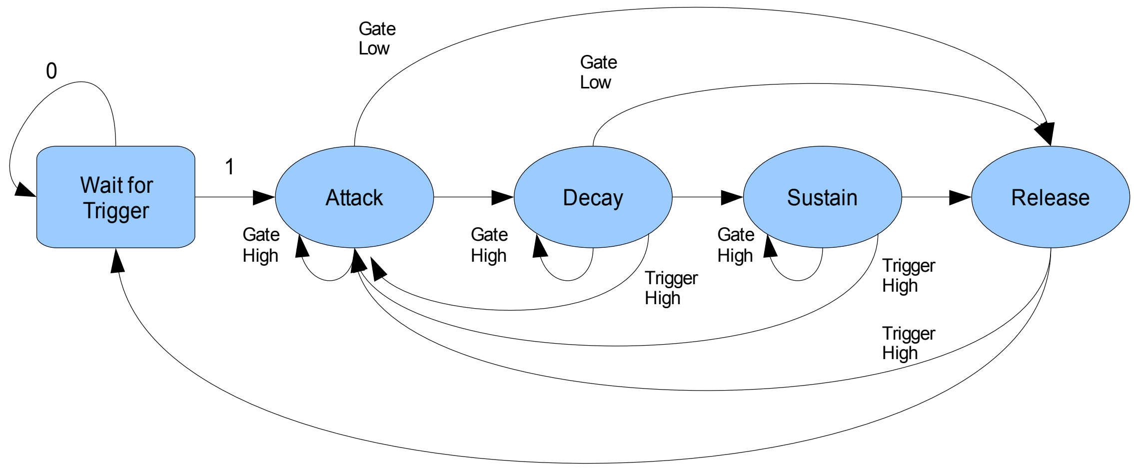


Fig. 3.10-3. ADSR cycle flow graph.

If at any point during any of the cycles the trigger causes an interrupt, then the current cycle must end, save its current value, and return to the attack cycle. This could happen if a string is plucked while resonating without fully muting it. There is a possibility that the gate signals alone could also be used as triggers, instead of generating and reading separate gate and trigger signals. In software, the gate input pins could be set up as interrupts that fire when a positive edge is detected. The interrupt should wait for the gate to fall low before it can read another positive edge. The one problem with this scheme is the potential inability to play very fast. For instance, if the gate remains high while the string is plucked again, then the MCU won't read another positive edge, and the ADSR cycle will not restart and the attack that was desired wouldn't appear. However, when a string is plucked, it is also muted for a very short duration while the plectrum makes contact with the string. If this is enough time to bring the gate low, then there will certainly be a new trigger and a new attack cycle, creating the abrupt “picking” volume swell. It is yet unclear if this will be a considerable negative effect on the overall playability, so it will need to be tested with real equipment before making a judgment to cut the trigger circuitry.

3.11. Harmonizer

One advantage of having linear response for the frequency to voltage converter is the ability to create harmonies just by a proportionate gain. For example a gain of 2 will create a perfect octave harmony, and a gain of 1.5 will give you a perfect fifth harmony. The diagram in figure 3.11-1 shows how the harmonizer will work. The first step is to acquire the signal out of the frequency to voltage converter and buffer it so the system has no loss. This signal is then sent to three separate circuits. Two of the signals will be sent to have different gain, the first one will pass through a non inverting amplifier to have a gain of 2. This means R1 is 1kOhms and R2 is 2kOhms. The signal at the other end is now scaled to be a perfect octave.

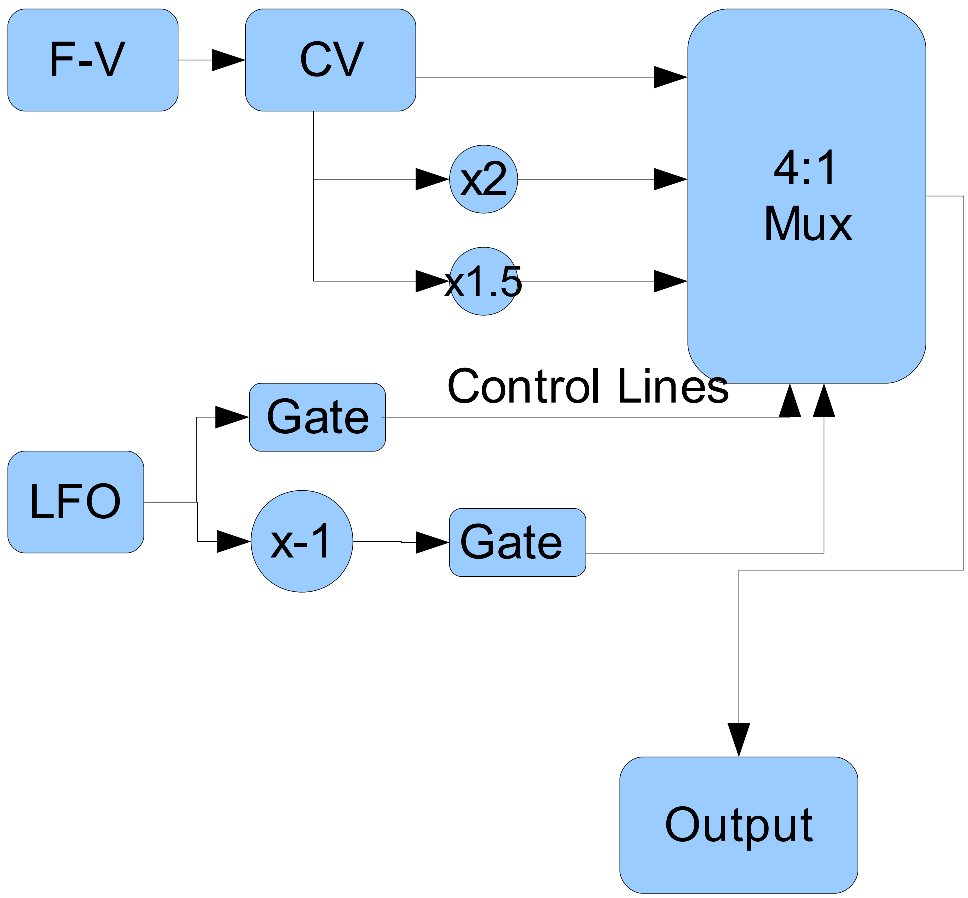


Figure 3.11-1: Arpeggiator flow chart; The controlled voltage is attained from the frequency to voltage converter and multiplied by a factor of 1.5 (Perfect fifth) and 2 (octave) and used to feed in a multiplexer. The multiplexer is driven by an LFO triggering two gates with different threshold voltages. The rate of the control lines changing is based on the rate of the LFO. The gain on the LFO will define the time between the gates triggering.

The next signal is sent to have one of two different gains chosen. The first option is to have a gain of 1.5, this makes R1 1kOhm and R2 4.5kOhm. The second option is set to have a gain of 1.2 which makes that option a major third, this makes R1 1kOhm and R2 1.2kOhm. Now there is three signals total and two possibilities between them, 1st, 3rd, 5th and 1st, 5th, and 8th. The low frequency oscillator will then have two lines coming out, one going directly into a gate (same design used for the ADSR, and the second going into an inverting amplifier with unity gain and then being passed through the LFO, this is seen in figure 3.11-2.

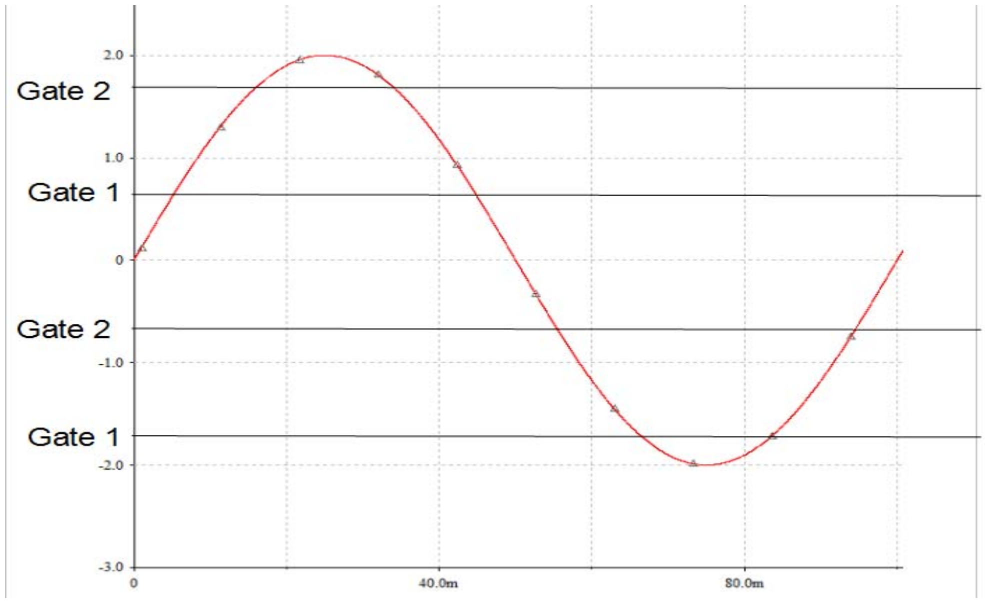


Figure 3.11-2: A low frequency oscillator is being passed through 2 gates both with different threshold voltages.

The threshold will be set at around .7 volts and the low frequency oscillator will have a varied maximum value, so this will vary when the gate is triggered. The second line goes out of the LFO into an inverting amplifier in order to activate that gate when the voltage is negative. This design gives two gates being triggered opposite to each other and then a gap (based on the threshold) between both the outputs. The expected gate output is shown in figure 3.11-3.

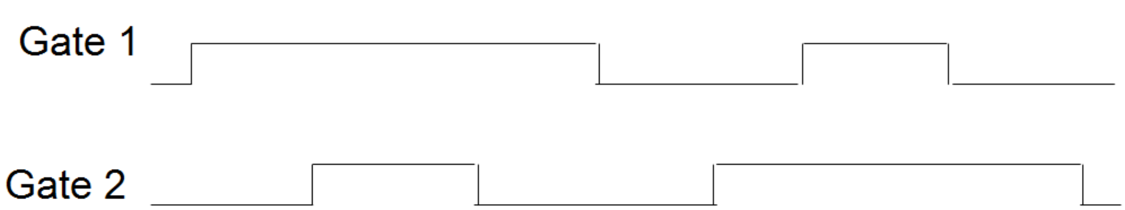


Figure 3.11-3: The triggering sequence of both the gates with the threshold voltages shown above

These two lines will be used to control a 4:1 multiplexer. The three signals will be fed into the multiplexer and one output will be chosen at a time. This signal replaces the original DC voltage that is being used to drive the VCO. The logic control lines allows for four possible outputs to come out. This means the 1st should be repeated as one of the lines feeding the multiplexer. The amount of time each harmony takes is based on where the gate threshold is compared to LFO output. If the threshold is relatively high compared to the max LFO output then the two harmonies will be quick and the main signal will take up most of the time. The opposite occurs if you lower the depth of the LFO, you have much more harmonies being passed than the original signal. If two gates with different thresholds are passed this gives a quicker switching option and you can have a total of 8 interval steps during one oscillation. At a switching rate of 33 Hz passing 8 intervals, this can give the effect of a ring modulator. Combining this effect with the the tremolo using the LFO as a control a syncopation would occur that every time the octave is being produced it is also peaking in the tremolo meaning that the pitch is going to be slightly higher and when the 5th is produced it will always be somewhat flat. This type of design could then be inverted just by passing the LFO through an inverting amplifier before it gets read by the two gates. The multiplexer schematic is shown in figure 3.11-4.

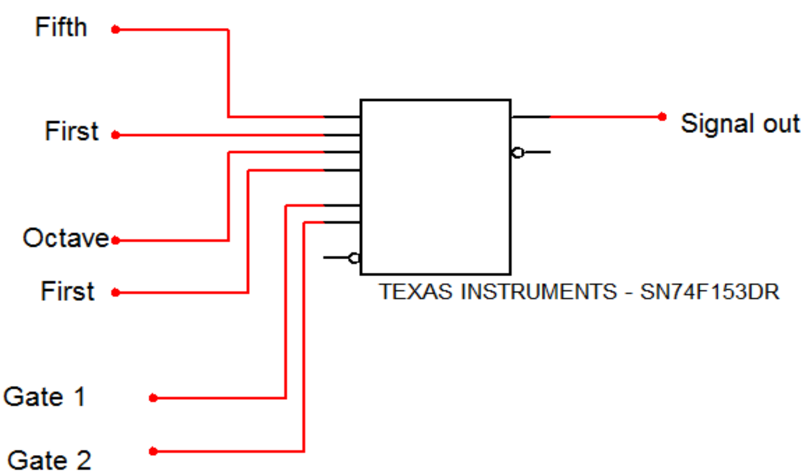


Figure 3.11-4: Multiplexer schematic layout

3.12 Switching Mode Power Supply

A switching mode power supply will regulate 12VDC from a 120VAC voltage source. A transformer will both step down mains voltage and perform rectification. A capacitor filter will smooth ripple from the DC voltage.

Parts Required:

- LM2576 Voltage Regulator
- Inductor: 150uH
- Capacitor: 470uF(x2); 240uF
- Diode (x3)
- Resistors: 1kQ(x2)
- 3:1 Center Tap Transformer

<u>LM2976 Regulator</u>

The regulator must be capable of outputting a fixed voltage of 12VDC to the synthesizer with a current between 2 and 3A while not exceeding an input voltage of 20V. An LM2576 step-down, switching mode regulator will be used because it is readily available and can meet these specifications. Because it is a step-down regulator, the unregulated input voltage must be greater than the output voltage. The regulator requires only a few external components and has internal frequency compensation and current foldback limiting protection. It also has a thermal shutdown mechanism and an overall efficiency of 88%. The data sheet for the device simplifies the design by offering component selection guides for choosing the values of the external components. Figure 3.12-1 shows the circuit diagram for a fixed output voltage design.

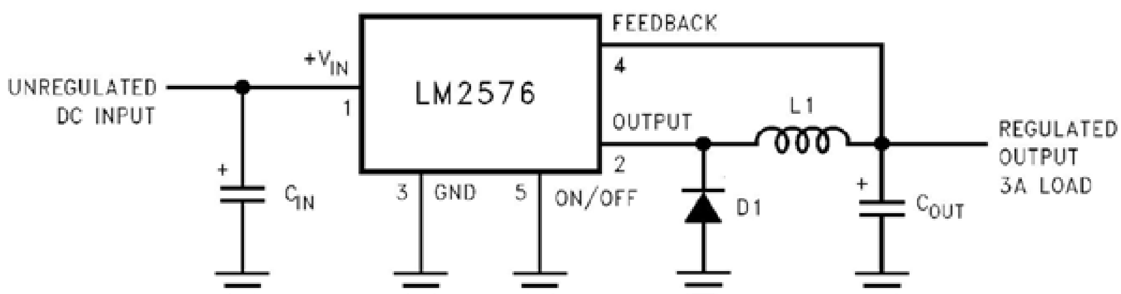


Figure 3.12-1: LM2576 Switching mode regulator, schematic
(Permission Pending)

The series inductor and shunt capacitor on the output of the regulator form a low pass filter to suppress the harmonics and the fundamental frequency of the pulse train from the internal pulse width modulator. The modulator drives a transistor that is frequently and instantaneously switching between saturation and cut off modes. A diode is added to the circuit to relieve the stress on the inductor while it tries to prevent the current from changing instantaneously across it. The result is a ripple on the current and therefore the voltage. Paired with a capacitor, the ripple in the current and voltage can be greatly reduced and should not exceed 1% so long as the correct capacitor value is chosen. The input capacitor is used to stabilize the regulator.

The regulator will operate in a continuous mode, where the current through the inductor will flow continuously instead of reaching zero during the switching cycle. For a maximum input voltage of 20V and maximum load current of 3A, a 150uH inductor was selected from the datasheet inductor selection guide. The inductor purchased must withstand the LM2576 switching frequency of 52kHz and a maximum current rating of 115% of the load current. A capacitor between 100uF and 470uF must be used in order to maintain a ripple voltage on the output of less than 1%. The voltage rating of the capacitor should be 18V since the output voltage will be 12V and the capacitor should be rated for at least 1.5 times greater than the output voltage. The current rating of the diode must be at least 3.6A and the reverse voltage rating must be at least 25V. The input capacitor must be made of aluminum or tantalum electrolytic material. The inductor, capacitors and diode will be selected upon availability from Skycraft with the best possible compliance of these requirements.

Transformer & Rectification

Since a transformer will be integrated into the design of the power supply in order to step down the mains voltage, it was decided to utilize the capabilities and perform full wave rectification. The load resistance seen by the transformer will be determined by the input resistance to the voltage regulator. For the purpose of testing the transformer design, a load resistance of 1kQ was selected. The diodes in the circuit are preceded by 1kQ resistors as well as to not exceed the forward operating voltage of the diodes. The designed circuit uses a transformer with a 3:1 turns ratio, indicating that a 120VAC input on the primary side will step down to 40 VAC on the secondary. There will be a voltage drop across the resistors and diodes, further decreasing the voltage on the secondary side. It is necessary to keep the output voltage above 12V in order for the regulator to function properly. Figure 3.12-2 shows the schematic of the transformer acting as both a device for stepping down the voltage as well as the means for rectifying the AC voltage.

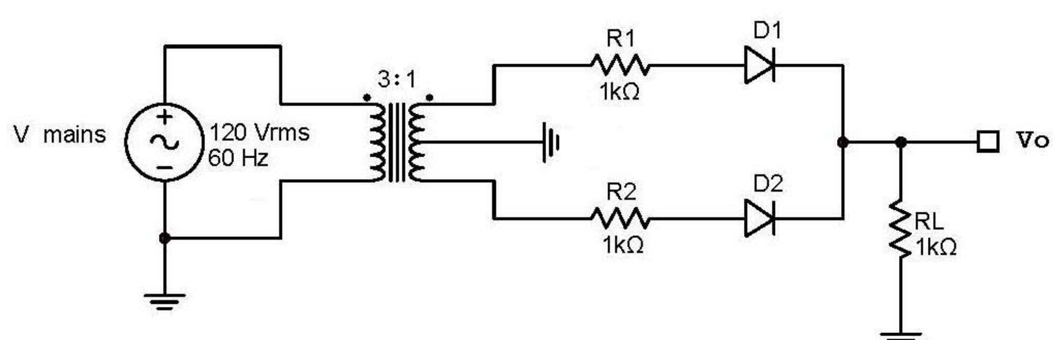


Figure 3.12-2: Transformer rectifying circuit

In this circuit, 120VAC at 60Hz is supplied to the positive and negative terminals of the transformer. The voltage on the secondary side is +/-20V with respect to the center tap. This allows flexibility in the load resistance that the transformer will face while maintaining the 12V needed for the regulator.

The output waveform is shown in figure 3.12-3 along with the input waveform to show the quality of rectification that can be achieved using a transformer circuit. The output waveform has an amplitude of 14V due to the voltage drop across the resistors R1 and R2, as well as the diodes. Upon determining the input resistance of the regulator, appropriate resistor values can be chosen in order to maintain 12V on the output of the circuit. The diodes and resistors will be purchased from Skycraft. The transformer should be PCB mounted to reduce the size of the power supply, but cost and availability are the two largest concerns when it comes to choosing one. If only larger, chassis mounted transformers are available which meet the requirements, they can be integrated into the design of the power supply.

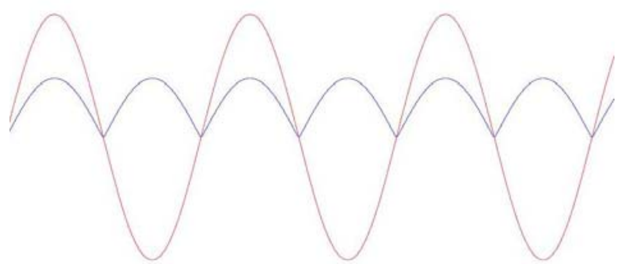


Figure 3.12-2: Output of transformer rectifying circuit

Capacitor Filter

A filter is to be designed that limits the percent ripple in the fully rectified waveform to less than 1%. The capacitor will be in parallel with the load resistor. The smaller the RC time constant, the further the capacitor discharges before another positive pulse arrives and recharges the capacitor. The filter will be designed to keep the capacitor from decaying significantly before the next charging cycle occurs. Percent ripple is calculated with the following equation:

$$r = \frac{1}{2\sqrt{3} \times f \times R \times C} \times 100\%$$

where root(3) represents the ripple voltage in terms of its rms value, f is the frequency of the rectified waveform, R is the load resistance, and C is the capacitor. In order to maintain a ripple of less than 1%, the necessary capacitor is determined to be

$$C = \frac{1}{.01 \times 2\sqrt{3} \times 120 \times 1000} = 240\mu F$$

where .01 represents 1% ripple, 120 is the frequency since the mains frequency is doubled during full wave rectification, and 1000 is the value of the load resistor.

When a 240uF capacitor is placed in parallel with load resistor shown in figure 3.12-2 above, the output waveform more closely resembles a DC voltage 3.12-4 below shows the modified circuit output superimposed on the unfiltered rectified output. Because the capacitor value is as large as it is, the capacitor does not charge and discharge at a high enough rate during positive and negative slopes of the rectified output to generate significant ripple on the output. The output voltage can be further smoothed with additional parallel capacitors across the load resistance. The capacitor value needed to satisfy the requirement of 1% ripple value is subject to change based on the load resistance. If a maximum 1% ripple is not satisfactory for the synthesizer components, additional capacitors can be added in parallel across the load resistor in order to further smooth the rectified voltage.

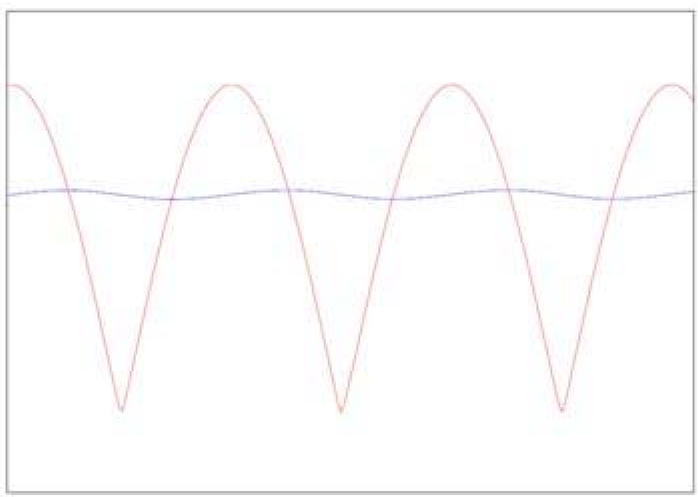


Figure 3.12-4: Capacitor-Filtered Rectifying Circuit, Output

Current Limiting

The LM2576 has internal current limiting protection for both the power supply and the load. In the event of a short circuit on the output, the oscillator frequency reduces to 11kHz, causing the regulated output to drop 40%. As a result, current to the load is reduced to an amount of (12V*.4)/R_L.

4. Testing Procedures

4.2 Frequency to Voltage Converter

Tuning each F/V converter will be a crucial step to giving the VCO an accurate voltage for the corresponding frequency. For accuracy to be ensured, an oscilloscope will be used to input the highest frequency for each string channel (the 20^{th} fret frequency). The 10kΩ trim potentiometer will then be adjusted until the output is exactly 5V. Once this is done, some more checking may be done with lower frequencies to make sure that the slope is as designed. Successful tuning will give the VCO the proper control voltages for each channel. Also the speed of the response time will be tested. Nominal speed should be lower than 10ms for the output voltage to ramp up to its required value. Lowering C_2 from pin 3 to ground will decrease this time, at the expense of becoming more sensitive to input noise. This has already been tested on a smaller scale using a breadboard and a guitar being fed into the LM2907 through a filter set up for the low E string. Testing will have to be done will every string channel and each component value changed accordingly. Trim capacitors are not as small or reliable as trim potentiometers, so the test for response time will have to be done for each string on a breadboard, and suitable capacitor values will have to be chosen for the final PCB design.

4.3 First Stage Filters

The first stage filters will be tested by using a sequence which will first tune for the cutoff frequency ω_0, then tune for the quality factor. The cutoff is only dependent on R and C in the Sallen-Key low pass filter, while the quality factor is dependent on Ra and Rb by the gain K. Table 4.3-1 shows the testing sequence, the parameter under test, and the component value to alter. This test could only be done on a bread board if the surface mount components are guaranteed to be exactly the same as the leaded components. Since this won't be the case, trim pots will have to be included on the circuit board to ensure that proper trimming is done.

Table 4.3-1. Testing Procedure for First Stage Filters.

Parameter	Test Condition	Nominal Value	Adjust Component
ω_0	Phase at ω_0	-90°	R
Q	Gain at ω_0	1.6	Rb

Testing the phase and gain will require using a function generator to input a frequency ω_0, which will be the lowest frequency for each string, into the filter, and measuring the output on an oscilloscope. The sequence here is important.

The correct ω_0 must first be found before the quality factor, which is the amplitude at ω_0, can be adjusted.

4.4. Voltage Controlled Oscillator

Equipment:

- DC Adjustable Power Supply 0-10V
- Oscilloscope
- Breadboard
- Multimeter
- LM13700 Operational Transconductance Amplifier
- Resistors
- Capacitors
- Diodes

Procedure:

1. Assemble the circuit using the component values calculated in the design portion of this document
2. Connect the dc power supply across R_c so it will function as the Control Voltage
3. Vary the control voltage between 1V, 5V, and 10VDC according to the V/Hz value computed in table (3.4-2) and measure Ic. Verify that Ic equals the expected value from table (3.4-2)
4. Using an Ammeter, verify that the current Ia remains constant as the control voltage is varied between 0-10VDC
5. Vary the control voltage between 1V, 5V, and 10VDC, and measure the corresponding frequency change on the oscilloscope. Verify that f on the oscilloscope equals the expected value based on Ic values calculated in part 3 of this procedure.
6. Measure the voltage across the input terminals of each OTA and verify it does not exceed +/-5V

4.5. Integrator Circuit

Equipment:

- Function generator
- LF351 Operational Amplifiers
- DC Power Supply
- Bread Board

- Resistors
- Capacitors

Procedure:

1. Apply +/- 12 VDC to the V+ and V- terminals of all op-amps
2. Apply a triangle wave of 2V amplitude to the inverting terminal of the first op-amp
3. Adjust the frequency between 172 and 688Hz for the circuit corresponding to the Low E string.
4. Observe the amplitude shift and phase shift and determine if sine wave output is acceptable along the range of frequencies
5. Repeat procedure with circuits for the remaining 5 strings and adjust the frequencies corresponding to those strings (reference table 3.5-1).

4.6. Dual Channel Mixer

Equipment:

- Function Generator
- Oscilloscope
- DC Power Supply
- Bread Board
- LF351 Operation Amplifier
- Dual gang potentiometer
- Resistors
- Capacitors
- Diodes
- Toggle switch

Procedure:

1. Apply +/- 12VDC to the V+ and V- of the opamp
2. Apply waveforms to respective terminals according to figure 3.6-1.
3. Measure the resistive tolerance of the potentiometer and match as best as possible with use of trim potentiometers
4. Toggle the switch between square and triangle waves and observe on the oscilloscope that is functions correctly
5. Vary the proportions between the two mixed waveforms and observe the results
6. Vary the input frequencies (while keeping them equal) and measure the output to confirm that the amplitude of the signals are being added as expected.

4.7. Low Frequency Oscillators

The testing for the low frequency oscillator will consist of designing the circuit on a breadboard and and connecting the outputs to an oscilloscope. The purpose of this oscillator is to make frequencies that are in the inaudible range. The designed frequency range should go from 3 Hz to 33Hz.

Magnetic Pickups

The Magnetic pickups will be tested by observing the outputs of both the terminals. The most important function of these will be to produce a decent amplitude (the active filters will boost them for processing) and the frequency being produced. With a simple design like this the frequency should line up exactly to the frequency being played on the string. Once the design is built, the placement of them on the guitar will also prove to be key when trying to produce a good amplitude. There are certain places where the strings vibration is attenuated, and then others where they are at the maximum. The output will need to be terminated by a load resistor. Several values of load resistors need to be tested to see which one will produce a more desirable effect. The load resister will effect the overall response of that pickup, meaning the cutoff and resonance is different for every load resistor. The values of 100Ohm, 1kOhm, 10kOhm, 100kOhm.

4.8. Vibrato

The vibrato will be tested by inputting a 1 volt DC sine wave into the VCA and then having the controlled voltage come from the addition of 1v and a .5v pk pk 10Hz wave. If the vibrato works correctly the output frequency should be the same and the output amplitude should be swelling and decaying following the frequency that is being inputted.

4.9. Tremolo

The tremolo can be tested the the same way the vibrato wast tested. Inputting a DC voltage of 1 volt and adding it with an attenuated signal from the LFO. This will make the the output will vary by going up and down according to the LFO rate and depth. There will be several values tested for how much gain the LFO has. This is because the CV line going into the VCO is very sensitive to any sort of variances meaning the addition and subtraction has to be very minimal in order to not make the tremolo effect too much. The goal is to have the maximum range be plus or minus one interval of what is being played.

4.10. Voltage Controlled Filter

The VCF will be tested on a breadboard and variable resistors. When each parameter is changed a frequency will then go into the input and the output will be observed with an oscilloscope. For example if you are testing resonance, the input frequency should be that of the cutoff and as the quality factor goes up then the amplitude at that frequency should rise while all the other frequencies are attenuated. If both settings are at the minimum then anything above 10 Hz will be completely attenuated and nothing should pass. The gain of both the separate designs will be tested. For the first one, there should be a unity gain for all the frequencies that are passing (except at the cutoff), and for the second design as the quality factor goes up the gain of the passing frequencies should become more attenuated.

4.11. Arpeggiator

The testing for this will be done in several areas. First the gates will be tested by inputting a voltage and monitoring the output. The gate should become active once a voltage is applied that is higher than the threshold voltage. Both the gates are tested individually at different voltages then LFO is passed through both the gates. The second part of the testing is creating the harmonies and making sure they are scaled correctly. This is done by testing the new controlled voltage by passing it through the VCO and the output should be exponential. This will be done by passing in frequencies such as an E, measuring its output frequency, then passing in the same voltage but multiplied by a factor of 1.5. This should then create the perfect fifth out of the VCO and can be tested. The same thing is done for the octave. This creates three different signals that are being fed into a multiplexer. The third step in testing is feeding the two lines coming out of the gates into the multiplexer. The lines should now be switching inputs and outputs so here every line is going to to be held active in order to ensure that the properly scaled CV is being produced in the output. Once all three components have been tested they are then connected to each other, but instead of having a function generator create a low oscillation, the LFO will instead be used to trigger the gates. The output of the multiplexer should be then read into a oscilloscope which should read different DC Voltages as the switching is occurring.

4.12 Voltage Controlled Amplifier

Testing the VCA circuitry will involve isolating the VCA and injecting both an input signal and a control voltage and reading the output versus the input on an oscilloscope. The input will be a sinusoid from a function generator with a frequency of 500Hz, about the middle of the range of the guitar. Then, a DC control voltage will be sent in, starting at 0V and moving up incrementally to 5V. The output amplitude should behave according to the table in the VCA design section, or at least roughly so. The main goal is to get a range of 0V amplitude when the control voltage is 0V, and a maximum of around twice the magnitude of the input when the control voltage is at its max of 5V. Trimming will be done by changing R_1 until the expected output occurs.

4.13 ADSR Envelope Generator

There are several parts of the ADSR system to put under test. Testing this section involves seeing if the gate, trigger, and envelope generation portions all work correctly. The gate is the most simple to test, since it requires putting in a signal from a function generator and checking for an output that is stable above the specified threshold. The trigger could be tested by using an LED as an output from the microcontroller and blinking the LED when a trigger is detected. This will have to be done with an actual guitar input, since the peaks have to be detected when a string is plucked. Once these systems are active and working, the ADSR envelope generation can be tested by inputting a guitar signal and viewing the output of the ADSR on an oscilloscope set up as a one-shot trigger. The attack time, decay time, sustain level, and release time can therefore be varied with the potentiometers and their effects viewed on the screen. The ADSR system as a whole must respond only when a guitar string is plucked, or more specifically, when a new peak is detected that is higher than the last peak.

4.14 Switching Mode Power Supply

Equipment:

- 120VAC power source
- Oscilloscope
- Multimeter
- 3:1 Transformer
- LM2576 Voltage Regulator
- Capacitors
- Inductor
- Resistors
- Diodes

Procedure:

1. Connect the circuit shown in figure3.12-2
2. Measure the output across RL on the oscilloscope and observe the AC ripple on the output
3. Replace RL with the LM2576 voltage regulator and connect the circuit as shown in figure 3.12-1 by using a 1kQ resistor as the load to the regulator
4. Observe the AC noise present across the input terminals of the LM2576
5. Determine if additional filter capacitors are necessary to reduce the amount of noise.
6. Disconnect the Transformer circuit from the LM2576 and replace with DC power supply

7. Vary the DC power supply between 12, 15, and 18V and verify that the output of the LM2576 remains constant at 12V.
8. Replace the load resistor in figure 3.12-1 with a wire (simulating a short circuit to ground) and measure the output voltage of the LM2576 across pin 2 to ground
9. Verify that the output voltage decreases to 7.2V

5. Construction

5.1 Prototype Plan

For the initial prototype only one channel will be built using a breadboard and through-hole versions of the various op-amps and ICs. The op amps and most other chips can be purchased cheaply as through-hole versions, but the microcontroller only comes in surface mount packages. To get around this hurdle in the initial prototyping, a breakout board will have to be made which the surface mount part could be soldered to, making each pin accessible with headers. The materials for a small printed circuit board are already available to the group, and it is mainly a matter of designing the layout, transferring the design to a board, etching away the copper and drilling. Another route is simply buying a breakout board that could also be used for the final design, so two chips don't have to be purchased for the ADSR. The best choice might be to use the MSP430G2231 which is a through-hole microcontroller that doesn't have the amount of features or pins as the 64-pin MSP430F4132, but is already available as part of the TI Launchpad development kit that was already purchased. It has the features necessary for one channel: 1 PWM and up to 8 analog inputs, as well as 10 I/O pins. This could be programmed and placed on a breadboard with the rest of the components, cutting down the clutter that could easily plague an initial prototype. The other ICs are cheaper and it is not an issue to purchase one of each in a DIP package for the prototype. Because this is only for one channel, the result would be a monophonic synth, with the goal of producing a fully polyphonic synth by the end of the building phase.

5.2 Main Enclosure

For the final build, printed circuit boards will be designed and their construction outsourced to an appropriate facility. A mixture of surface mount and through-hole components will be used, in order to balance size and ease of soldering. Another space-saving method will be to build multiple boards (one for each channel, for instance) that could be connected in rows vertically. This is only necessary if the area of the main board is too large to allow for the project to be portable. Size should not deter a musician from being able to set up the device in a stage setting. Fig. 5.2-1 shows the prospective enclosure with its knobs and switches labeled. The input to the device will be a MIDI cable connector which carries the signals from each of the pickup coils. The user will be able to connect the device to an external amplifier using a 1/4" inch audio cable. Power will be supplied by a standard wall plug cable, since the power supply will be housed within the enclosure.

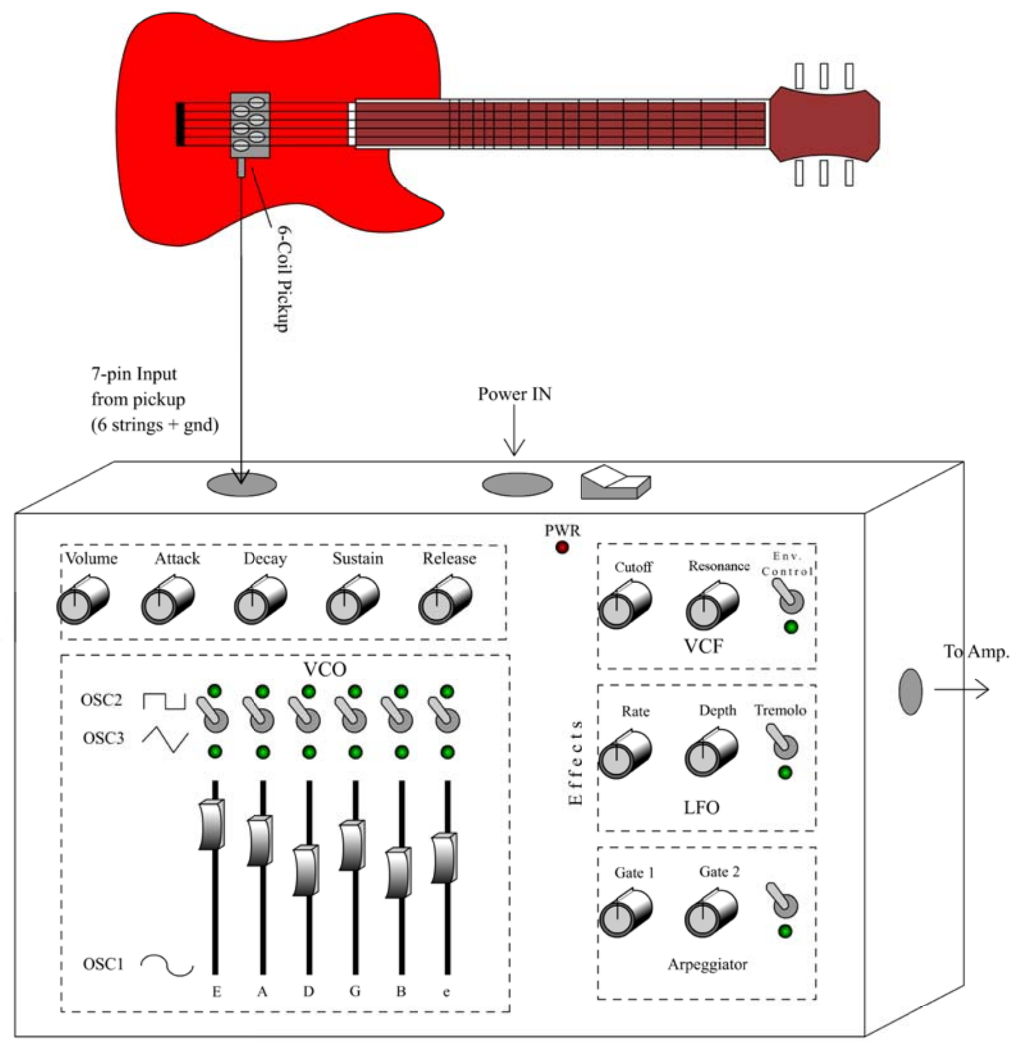


Fig. 5.2-1. Prospective final enclosure design.

6.1. Project Management

6.1. Milestone Chart

The Gant chart below summarizes the critical tasks that took place in semester 1 of this project.

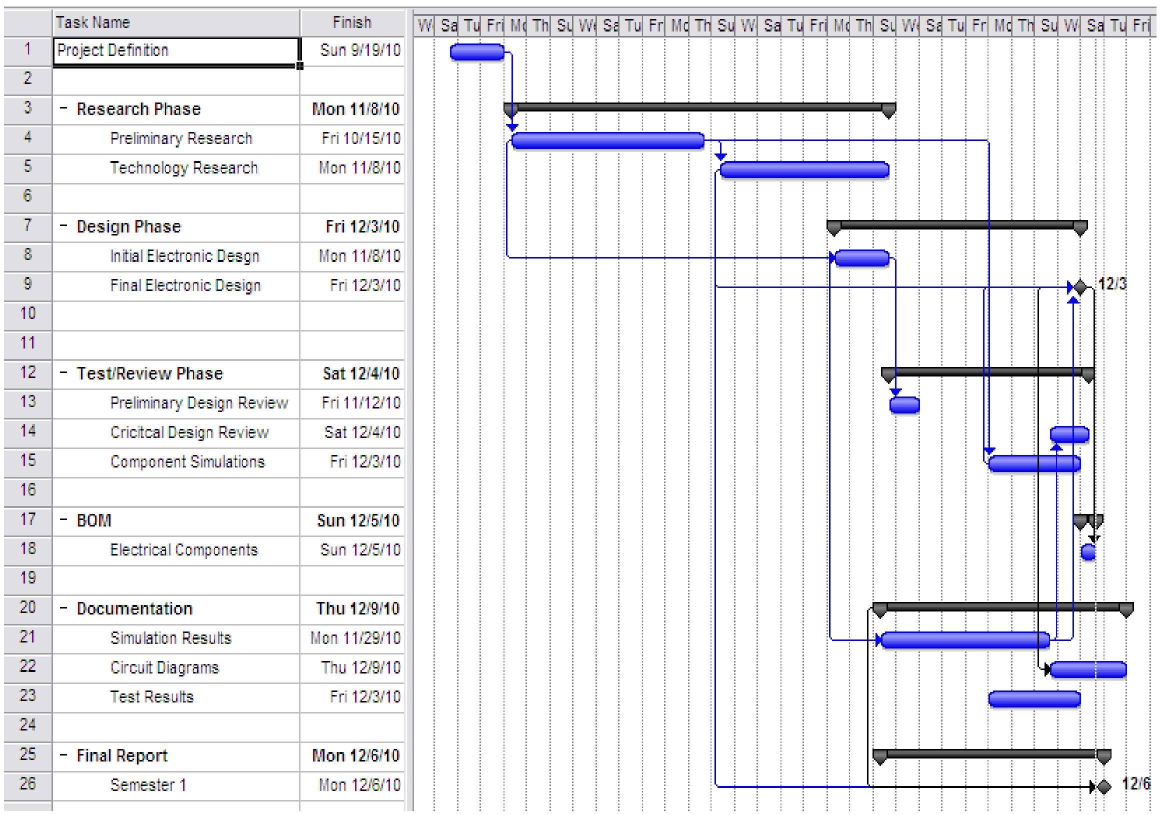


Figure 6.1-1: Milestone Chart

6.2. Bill of Materials

The rough estimate of the parts cost is tabulated in the Bill of Materials below. The initial prospective project budget was under $300. This goal still seems reasonable considering the total cost of the parts, including the circuit board. Shipping is still uncertain and will be calculated at the time of purchase. Also, an enclosure is not included in the pricing list due to the uncertainty of the final size of the build. Once the circuit board(s) is designed, then an appropriately-sized enclosure may be purchased or built given there is enough time. Also, the guitar itself will be sourced from one of the group members, cutting down cost in that area. All of the estimated parts to be purchased are listed in the Bill of Materials in table 6.2-1 below, including quantities and prices.

Part	Description	Qty.	Cost ea.	Total
LF351	Operation Amplifier	28	$0.30	$8.40
Potentiometer	Linear Dual Gang	6	$2.50	$15.00
LED's	Green/Red	12	$0.00	$0.00
LM2576	Voltage Regulator	1	$3.44	$3.44
Toggle Switch	SPST Center Off	10	$1.90	$19.00
Inductor	150µH	1	$1.80	$1.80
Diode	1N4003	3	$0.32	$0.96
Transformer	3:1 PCB Mounted Center Tap	1	$4.00	$4.00
LM13700	Transconductance OP AMP	18	$0.41	$7.38
LM2907	Tachometer	6	$1.86	$11.16
TL072	Dual Op Amp	2	$0.69	$1.38
TL084	Quad Op Amp	2	$0.63	$1.26
TL082	Dual Op Amp	10	$0.62	$6.20
LM393	Dual Comparator	1	$0.28	$0.28
LM339	Quad Comparator	1	$0.49	$0.49
MSP430F4132	Microcontroller	1	$4.56	$4.56
Trim Potentiometer		18	$0.50	$9.00
Panel Potentiometer		9	$1.25	$11.25
AlNiCo	Magnet	6	$2.00	$12.00
Wire	42 Gauge	1	$10.00	$10.00
1N4148 Diodes		18	$0.00	$0.00
Midi Cable		1	$15.00	$15.00
Multiplexer	4:1 TI 5n74…	6	$0.55	$3.30
Resistors		325	$0.06	$19.50
Capacitors		72	$0.19	$13.68
Circuit Board	Double sided 4pcb.com	1	$33.00	$33.00
			Total	$212.04

Table 6.2-1: Bill of Materials

Appendix A……………………………………………………...Guitar Frequency Spectrum

OPEN	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20
E	F	F#	G	G#	A	A#	B	C	C#	D	D#	E	F	F#	G	G#	A	A#	B	C
330	349	370	392	415	440	466	494	523	554	587	622	659	698	740	784	831	880	932	988	1047
B	C	C#	D	D#	E	F	F#	G	G#	A	A#	B	C	C#	D	D#	E	F	F#	G
247	262	277	294	311	330	349	370	392	415	440	466	494	523	554	587	622	659	698	740	784
G	G#	A	A#	B	C	C#	D	D#	E	F	F#	G	G#	A	A#	B	C	C#	D	D#
196	208	220	233	247	262	277	294	311	330	349	370	392	415	440	466	494	523	554	587	622
D	D#	E	F	F#	G	G#	A	A#	B	C	C#	D	D#	E	F	F#	G	G#	A	A#
147	156	165	175	185	196	208	220	233	247	262	277	294	311	330	349	370	392	415	440	466
A	A#	B	C	C#	D	D#	E	F	F#	G	G#	A	A#	B	C	C#	D	D#	E	F
110	117	123	131	139	147	156	165	175	185	196	208	220	233	247	262	277	294	311	330	349
E	F	F#	G	G#	A	A#	B	C	C#	D	D#	E	F	F#	G	G#	A	A#	B	C
82	87	92	98	104	110	117	123	131	139	147	156	165	175	185	196	208	220	233	247	262

Appendix B……………………………………………………………………..…..Datasheets

(LM13700)
http://www.national.com/ds/LM/LM13700.pdf

(LM2907)
http://www.national.com/ds/LM/LM2907.pdf

(LM324)
http://focus.ti.com/lit/ds/symlink/lm324a.pdf

(LM2576)
http://www.national.com/mpf/LM/LM2576.html

(LM231)
http://www.national.com/ds/LM/LM231.pdf

Appendix C……………………………………………………………………..….Bibliography

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http://en.wikipedia.org/wiki/Overtones

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Manual of Guitar Technology: The History and Technology of Plucked String Instruments
Franz Jahnel

(Filters, Note Frequencies)
http://shakahara.com/pianopitch2.php

(F/V Converter)
http://www.vaughns-1-pagers.com/music/musical-note-frequencies.htm

(VCO, LM231 Theory of Operation)
http://www.national.com/ds/LM/LM231.pdf

(VCO, Operational Transconductance Amplifier, Theory)
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http://beavishifi.com/articles/Volume_Control/index.htm

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Appendix D……………………………………………………………..…Copyright Permissions

Request to use datasheet image [Incident: 101114-000015]

From: **Exar Customer Support** (CustomerSupport@exar.com)
Sent: Mon 11/15/10 12:08 PM
To: mattmohn@knights.ucf.edu

Recently you requested personal assistance from our on-line support center. Below is a summary of your request and our response.

If this issue is not resolved to your satisfaction, you may reopen it within the next 7 days.

Thank you for allowing us to be of service to you.

To access your question from our support site, click here.

Subject
Request to use datasheet image

Discussion Thread
Response (Linda Craig) 11/15/2010 09:08 AM
Hi Matthew,

Thank you for your interest in Exar product. Approval for you to use of the circuit schematic image from the XR2206 datasheet is granted.

Best Regards,
Linda

Customer (Matt mohn) 11/14/2010 05:55 PM
My name is Matthew Mohn and I am an electrical engineering student at the University of Central Florida. I am emailing to inquire about the use of a circuit schematic image from the XR-2206 datasheet. The image would be used in my group's final documentation for our senior project.

Thank you!
Matthew Mohn

Question Reference #101114-000015
Category Level 1: General
Category Level 2: Other
Date Created: 11/14/2010 05:55 PM
Last Updated: 11/15/2010 09:08 AM

Request to use datasheet image

From: **mattmohn@knights.ucf.edu**
Sent: Sun 11/14/10 8:49 PM
To: customersupport@exar.com

My name is Matthew Mohn and I am an electrical engineering student at the University of Central Florida. I am emailing to inquire about the use of a circuit schematic image from the XR-2206 datasheet. The image would be used in my group's final documentation for our senior project.

Thank you!
Matthew Mohn

Inquiry to use datasheet images

From: **mattmohn@knights.ucf.edu**
Sent: Mon 10/25/10 1:11 AM
To: copyrightagent@national.com

I am an electrical engineering student at the University of Central Florida. I am asking permission to use some of the information and data sheet images from the National Semiconductor website for my group's senior design project documentation. The information and images are from the data sheets for the LM2907, LM324 and LM13700. Thanks for your help,

Matthew Mohn