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6.101 Final Project Proposal
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Analog Synthesizer

Motivation

From the birth of popular music, with the invention of the phonograph, to the increased availability of music, due to various music stores and libraries such as iTunes, technology shaped the creation of music. This effect has touched every aspect of the industry, from the business model to the music itself. One particular interest is how technology has changed the style and sound of music. For example, the electric guitar forever changed music by ushering in rock, while computers caused an equally large shift by enabling modern electronic music, such as techno. The analog synthesizer was another influential device that changed rock music as well as helped to create electronic music.

Analog synthesizers were the first purely electrical instruments. Although electric guitars preceded analog synthesizers, analog synthesizers produce purely electronic signals while electronic guitars convert mechanical signals (the vibration of wires) into voltages that are then processed to produce sound. Due to this mechanical input, mechanical instruments produce distorted waveforms, which are impossible to replicate in purely analog systems. These uniquely distorted waveforms give the instruments their distinctive sound. Similarly, the unique sound of an analog synthesizer is due to the perfect waveforms generated by these instruments. This perfect signal has a sound that is nothing like a conventional instrument and has helped to create as well as heavily influenced electronic music. Influential bands such as Kraftwerk and Brian Eno used analog synthesizers prominently in their seminal work.

In the eighties, analog synthesizers were largely replaced by digital synthesizers, which in the nineties were replaced by computers. Both digital synthesizers and computers enable the generation of arbitrary waveforms. Despite the advantages of these alternatives, analog synthesizers continue to remain popular among some musicians, because the analog signals provide a precise sound that digital systems cannot match. In addition, the modularity of the synthesizer encourages musicians to create new and unique combinations of sounds in a way that a preprogramed instrument cannot.

In addition to their musical uses, analog synthesizers are of great interest to electrical engineers because it uses concepts and building blocks from a diverse set of electrical systems. A single module may incorporate elements from AM and FM radios, timers, and filters in addition to digital circuitry. Common building blocks include multipliers, exponential current sources, and voltage controlled resistors. In addition, in order to drive the generated music through speakers loud enough for a live audience, a lot of work must be done to create a high power amplifier and power supply. Therefore engineers can gain a lot of useful knowledge and experience from building an analog synthesizer.

Project Overview

Elliot, Lauren and Elaine will be dividing our analog synthesizer project into three components: the power supply, signal processing, and power amplifier stage, with Elliot focusing on the signal processing, Lauren designing and building the power supply, and Elaine working on the power amplifier. These three systems are can be designed separately to later be integrated together, a process during which we may need to tweak the power supply to meet the other systems' needs.

The Synthesizer Stage

Elliott will be building the signal processing block. An analog synthesizer consists of several modules that can be wired together in different ways to create different sounds. A particular wiring setup is called a "patch". Modules connect to each other using two different types of signals. One type is called a waveform signal, which is a waveform being generated, modified, or played through a speaker. The other type of signal is a control signal, which determines how a module works. A control signal may be a 5 volt pulse to trigger an event in another module or a varying waveform to control the amplitude of an output signal at a given time. Waveform signals have a voltage range of plus and minus 5 volts while control signals range from zero to 5 volts. Because any module can be plugged into any other module, signals are carried between modules via quarter inch mono cables through standard output and input buffers. The output resistance of the output buffer is set to 1k to prevent damage should the output be grounded. The input resistance of the input buffer is set to 100k to reduce attenuation.

Because the synthesizer is modular, a fixed block diagram of the system does not accurately describe how the system works. However, a fairly standard patch is shown in Figure 1 to provide the reader with a reference to help understand the function of each module and how they may be used together.

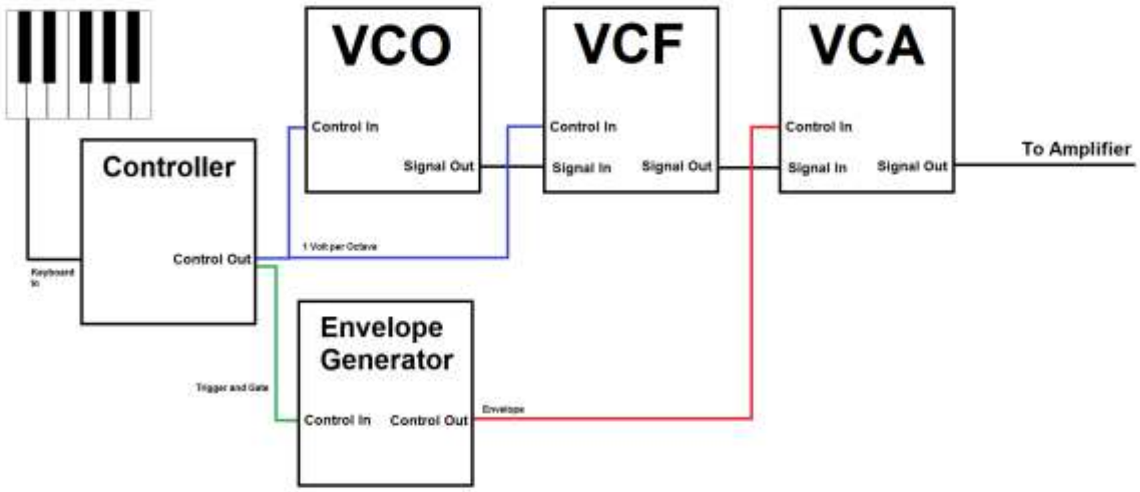

Figure 1. Block Diagram of a Typical Patch on an Analog Synthesizer: The blocks represent modules while the lines represent signals. The black lines are waveform signals while the colored lines represent different control signals. Note that analog synthesizers are inherently modular and thus this block diagram only represents one such configuration. In this setup, the controller reads the keyboard presses and produces the corresponding control signals. The one volt per octave signal is fed into both the VCO and the VCF so that filter will track the tone generated. The output of the VCF is then modified by the output of the envelope generator to produce a output signal that is fed to the amplifier.

The standard analog synthesizer is made up of five basic modules, the controller, the voltage controlled oscillator, the voltage controlled filter, the voltage controlled amplifier, and the envelope generator. Additional modules such as a sample and hold or a noise source are possible stretch goals.

The controller is the user interface of the synthesizer. It converts key presses on a keyboard into three control voltages that define the signal to be generated. One signal is a voltage proportional to the key that was pressed. This voltage is set so that an increase in one octave will increase the control voltage by one volt, which will result in a doubling in frequency at the output of a VCO. This one volt per octave signal is an exception to the typical voltage limits of a control signal and may range from -5 volts to 5 volts to provide an output bandwidth from around 30 Hz to 20 kHz. The other two output signals are a 5 volt trigger and a 5 volt gate. The trigger is simply a pulse that indicates exactly when a note has been pressed. A gate is a signal that is high whenever a note is held down and low when all notes are released. These two signals can be used by an envelope generator to control the amplitude of the output waveform over time based on the inputs on the keyboard.

A voltage controlled oscillator produces several waveforms dependent on the input signals and its knob settings. Output signals include sine, triangle, sawtooth, square, and PWM waveforms. The frequency of oscillation can either be controlled by a potentiometer or by a 1 volt per octave signal from a controller. VCO's sometimes have an optional input that can perform frequency modulation of the input signal and the generated signal.

A voltage controlled filter is a bank of filters controlled by input voltages and potentiometer settings. While there are many types of VCFs with different output types, the proposed system will have low pass, band pass, band stop, high pass, and notch filter outputs. The basic VCF filters an input signal at a frequency set by control knobs and input voltages. Typically the one volt per octave input voltage output of the controller is inputted to both the VCO and the VCF. In this setup, the filter will track the frequency of the outputted waveform and will thus produce the same effect at different frequencies. This is typically used to remove some selection of harmonics to alter the timbre of the output waveform. Advanced VFC designs provide options to adjust properties of the filter such as its Q.

The voltage controlled amplifier is a two quadrant multiplier that allows a control input to modify the amplitude of an input signal. Clearly the VCA takes in one control input and one signal input. VCA's are typically used in combination with envelope generators to give notes a realistic time response. Instead of turning notes on and off immediately, which introduces high frequency components that create a clicking sound, notes are turned on and off more gradually like a mechanical instrument. VCA can also be used to add other effects such as AM modulation. A two quadrant multiplier is used so that a slightly negative control voltage will ensure the signal is turned off. This makes it easier to control the amplitude of the output signal. Four quadrant multipliers, referred to as ring modulators, are also used in analog synthesizers albeit less frequently. Ring modulators are used to create audio effects such as tremolo. A ring module is one of the many options for a stretch goal.

Finally, the envelope generator is a module that takes in a gate and trigger signal and generates a time dependent output signal to be used by a VCA to give the output signal a more realistic sound. A typical envelope generator creates an ADSR signal. ADSR stands for the four sections of the envelope, attack, delay, sustain, and release. The attack phase is an exponential rise to some peak voltage when the generator receives a trigger. Once the peak voltage is reached, the exponentially drops to a final value that is held until the gat signal ends. The exponential drop is the delay stage while the held final value is the sustain stage. Once the use lifts their finger, the sustain voltage exponentially drops down to zero volts. This last stage is called the release stage. Each of the voltage levels and timing values of the ADSR envelope can be modified. This allows an envelope generator to simulate a large number of possible envelopes.

Other modules are typically found in an analog synthesizer though not as prominently as the five described above. These modules are all possible stretch goals with each additional module greatly enhancing the range of output sounds that could be produced. Additional modules include the aforementioned ring modulators, mixers, sample-and-holds, noise generators, timber modifiers, delay circuits, and more. There are enough additional modules that one can spend a lifetime working with analog synthesizers (as many hobbyists do). Therefore, the goal is to produce the five main modules at a minimum and then to add as many additional modules as time will allow.

The Power Amplifier Stage

Elaine will be designing a class G amplifier, with +/- 25V and +/- 15V supply rails at the output stage. The input signal into this amplifier from the signal processing unit will be +/- 5V, so the gain of this amplifier will be designed to be under 7. As the human ear can hear from about 20Hz to 20kHz, the bandwidth of the amplifier will be designed to extend from 15Hz to 100kHz. A class G amplifier was chosen for the power amplifier stage because it is more efficient than a class B amplifier but still maintains linearity (unlike a class D amplifier). Linear systems generally give less distortion at the output than non- linear ones, so with the class G amplifier we can strive for both efficiency and sound quality. The design of the Power Amplifier can be divided into five smaller stages: The input stage, the voltage amplifying stage, the class G output block, the output network, and the negative feedback loop that extends from the output back to the input stage.

The input stage of the amplifier will be a differential amplifier with current set using a pnp bjt above the differential amplifier and npn current mirror below. Naturally, the differential amplifier has two inputs, one being for the raw input signal from the signal processing unit, the other being for the negative feedback of the scaled down output signal. The output current can be determined by the following relationship: $I_{out} = I_e * \tanh(-V_{in}/2V_t)$, I_e being the total current of the circuit and V_t being the thermal voltage of about 26 mV. Maximizing I_e thereby increases transconductance (we double I_e by using a current mirror instead resistors at the collectors of the differential amplifier); it also gives us the opportunity to increase the slew rate, which is determined by $I_{e,}$/Cdom (Cdom is the dominant capacitor). The slew rate must be taken into account for producing clean high frequency signals; the required slew rate for this amplifier is 2*pi*frequency*peak voltage. Optimizing the input stage is also critical for noise performance; therefore a technique involving bootstrapping between the signal input and feedback point will be researched.
The Voltage-Amplifier Stage (VAS) will be critical in providing the current necessary to produce the desired voltage gain, as well as establish the dominant pole of the circuit. The Voltage-Amplifying Stage will likely look like a common emitter with an active load, and a capacitor connected between the input signal and the base of the amplifying bjt. This capacitor will determine the dominant pole of the circuit so that the total loop gain falls below unity before enough phase-shift accumulates to cause high frequency oscillation (Audio Amplifier Design Handbook), i.e. a phase shift of -180 degrees at unity gain. It is suggested in the literature that two pole compensation in the VAS can be used to further decrease Class-G stage distortion, this technique will also be further researched.

The Class G stage is characterized by its two supply rails, which as previously stated will be +/- 25V and +/- 15V. The Class G stage is composed of two Class-C outer power devices, and two inner Class-B power devices. Most of the signal will be powered by the inner 15V rail, the 25V rail will only be turned on when the signal exceeds 15V. Care will need to be taken to eliminate distortion that can occur from switching supplies. Schottky diodes leading into the circuit from the 15V power supply should be able to give a fast enough switching speed to minimize distortion. The negative feedback will extend from the output of this class G amplifier stage back to the differential amplifier in the input stage; this is where the actual gain of the overall amplifier will be determined.

The output network may need the most tweaking due to the presence of inductors. It will be comprised of a shunt Zobel network and series output inductor/damping resistor. The Zobel network provides stability into inductive loads, while the latter circuit gives stability into capacitive loads. It is important to note that the output inductor should be air-cored to eliminate the possibility of distortion due to magnetic materials. The coil wire should be thick, as thin wire will add more parasitic resistance and, reducing the voltage drop across the 8 ohm speaker load (Audio Power Amplifier Design Handbook). A very small resistor can be placed in parallel to the output inductor to reduce overshoot and ringing. Further considerations such as this will need to be taken into account in choosing the output inductor.

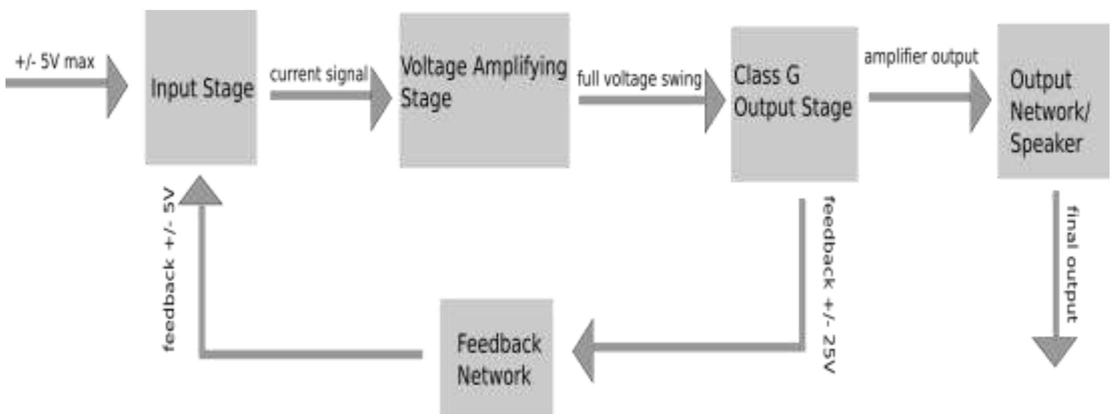

Figure 2: The Power Amplifier Design. This design consists of five modules: the input stage, the voltage amplifying stage, the class G output stage, the output network, and the negative feedback stage.

The Power Supply

Lauren is designing and building the power supply, which will be a \ fixed-voltage, linear power supply. This design will provide +/-25V and +/-15V to the power amplifier and +/-15V to the signal processing of the analog synthesizer. The power supply must be able to supply a peak of 3.2A at 25V to the power amplifier (25V/8 Ohm = 3.2 A) and ~2A at 15V, as well as 2A at 15V to the signal processing. Overall, the power supply must be able to supply 4A at 15V as well as 4A at 25V. Therefore, the power supply will require six transformers, two rated at 24VAC at 4A and four rated at 18VAC@2A. Working at such high currents will require high current rated components as well as heat sinking.

The power supply design is comprised of five major modules: the transformer, the rectifier, the filter, and the regulator. These modules will be modified and designed to output both +/-25V and +/-15V as shown below in Figure 3 and Figure 4.

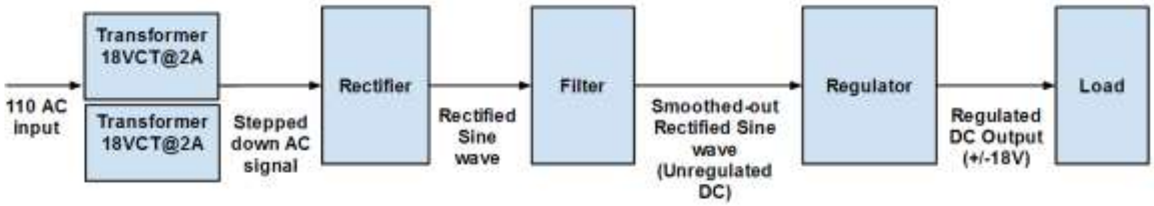

Figure 3: Fixed-Voltage, Linear Power Supply for +/-15V. Above is the proposed design of the +/-15V rails of the power supply. The power supply will take an AC signal, step down the signal from 110Vac to 18VAC across the outer taps of the transformer. Two of these transformers will be placed in parallel so that they can output 4A total. The signal is then filtered and smoothed, and the remaining ripple is eliminated to produce a regulated DC output of +/-15V.

As shown in Figure 3, the first module includes two Transformers, which are responsible for stepping down the AC line from 110VAC to 18VAC . With 18VAC, there is approximately 25V to work with to generate an output of 15V. By putting two transformers in parallel, 4A of current is able to be obtained. The AC signal is then rectified by a diode circuit, which is then filtered into an unregulated DC signal by a large capacitor. The signal is then regulated to reduce output ripple, and produce a constant +/-15V output.

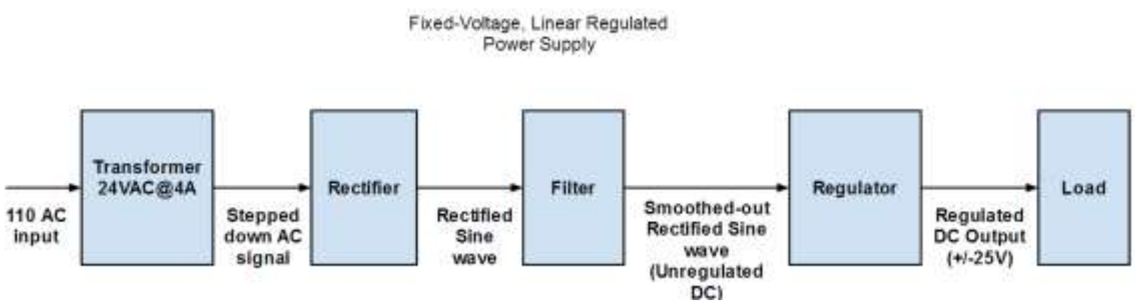

Figure 4: Power Supply for +/-25V. Above is the proposed design of the +/-25V rails of the power supply. The power supply will take an AC signal, step down the signal from 110VAC to 24VAC across the outer taps of the transformer. The signal is then filtered and smoothed, and the remaining ripple is eliminated to produce a regulated DC output of +/-25V.

As shown in Figure 4, the +/-25V is similar to the +/-15V design, however the +25V and the -25V require 24VAC@4A transformers. Because we were able to purchase 4A transformers, putting two transformers in parallel is not necessary. The entire voltage will be rectified, filtered, and regulated to output the final output voltage. Overall, the transformer converts down the AC line voltage to a smaller peak voltage, which is approximately 34V, which gives 9V to work with. The rectifier uses diodes to produce a rectified sine wave, which has a large DC component. The filter then smooths out the rectified signal wave, created an unregulated DC voltage with ripple. The final regulating circuitry eliminates ripple, producing a DC output. In addition to meeting the power requirements of the signal processing, the power supply will need heat sinking, lower gauge wires, higher rated components to accommodate for the large currents and voltages it must supply.

Conclusion

This project consists of three modules: the analog synthesizer, power amplifier, and power supply. Each component will present its unique challenges in design; it is also critical that each module meets the other components' requirements. Our goal is to design and build these subcomponents and link them together to produce an easy to use, fun analog synthesizer that can be further built upon in the future.