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Exploring the Relationship Between Designing and Composing With an Analog Synthesizer

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EXPLORING THE RELATIONSHIP BETWEEN DESIGNING AND COMPOSING WITH AN ANALOG SYNTHESIZER

by

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ABSTRACT

For my electrical engineering capstone, I designed and built a handheld analog synthesizer. This analog synthesizer is a musical instrument which uses electrical analog components to generate sounds with properties based on the user's input. My goal when creating this device was to maximize its utility while remaining within the scope of the capstone project. After the synthesizer was built, I wrote and recorded several pieces of music which each utilized the synthesizer prominently. The aim of this process was to be able to explore the relationship between an instrument's design and the quality of the music produced utilizing it. The resulting music was messy, but it was able to demonstrate the potential that an analog synthesizer like the one I built can have for creating quality music.

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INTRODUCTION

The synthesizer has become an increasingly prominent part of music production over the past century, and by extension, American culture. To put it simply, the synthesizer can be described as an electronic instrument which generates a sound wave with qualities based on the user's inputs. The ability to artificially create sound can effectively allow for limitless possibilities in terms of design and use. A sophisticated enough synthesizer can theoretically replicate any instrument without needing to manipulate a sample. However, given limitations on budget and time, the synthesizer needs to be designed in a mindful way which maximizes flexibility in terms of sound while minimizing components. This process was what I wanted to explore when I decided to design and build an analog synthesizer.

All electrical and computer engineering students are required to put together a capstone project in order to graduate. These are usually done in teams of two, but I was the odd one out, so I had to do mine alone. As an electrical engineering major, I was only required to create my project using analog components. This project idea came about due to my fascination with analog synthesizers, of which I own a small collection. I also enjoy writing and recording music utilizing synthesizers as a hobby. This combination of factors made designing an analog synthesizer the most intuitive choice.

During that same period of time, I had to decide upon a subject for my honors thesis, preferably one which I could tie in with my capstone project. I extrapolated upon my synthesizer idea by choosing to use it to create music. My aim was to write 20 minutes of music while heavily including the synthesizer in the process. By doing so, I could explore the relationship between a piece of music and tools used to make it.

LITERATURE REVIEW

A significant amount of my fascination with analog synthesizers came from how inventive music which featured the instrument heavily often was. The album *An Electric Storm* by White Noise, released in 1969, prominently utilizes rudimentary analog synthesizers. It borrows elements from the genres of psychedelia and Avant Garde which were blossoming during the late 60's and infuses it with a variety of synthesizer sounds and tape manipulation. The result is a pioneer of the electronic genre which sounds nothing like anything that came before or after it. The centerpiece of the album is "The Visitation", an 11-minute song that demonstrates the capability for synths to set a mood. The song is expressive in the way it manages to tell a story. In addition to the synths, sound effects, spoken word, and reverb are utilized to create an experience that sounds immersive and powerful. This album is one that is simultaneously timeless in its construction yet feels like a culmination of the culture and technology of the 1960s. It takes full advantage of the qualities of rudimentary synthesizers that appeal to me, and because of that, I often look to it for inspiration [1].

While *An Electric Storm* explores the more bombastic side of early synthesizers, Brian Eno's *Another Green World* from 1975 finds a more subtle and subdued use for them. When I imagine the former album, I think of giant machines creating immense sounds. With the latter, I think of a modest but substantially more complex machine that's lacking in power, but more than makes up for in texture and endurance. The album lives up to its name with how alien, unconventional, and vibrant the sounds are, even with 50 years of hindsight [2].

In the 1980s, digital synths rose to prominence. In comparison to analog synths, they were more powerful and small enough to fit on chips. This was before digital devices had enough memory to be able to store any substantial amount of high-quality audio. Instead, computers, arcade cabinets, and game consoles began using these chips as a way of producing music and sounds in real time. The music made for these systems was confined to the limitations of the chips being worked with, including the variety of sounds they could make and the number of sound channels they had available. Simple production techniques such as panning and basic effects such as reverb were often rudimentary or unavailable. In addition, every note had to be programmed in, meaning that for better or for worse, the result would be lacking in the flaws and the subtleties which characterized a performance.

While some composers settled with their sound chip work being a downgrade from what they could achieve with better equipment, others played to the chips' strengths. Because the music was programmed, the composers had a level of control which they didn't have with traditional synths. This resulted in arrangements which could have an inhuman level of precision and imitations of the subtleties of a performance. Fast arpeggios became common due to how they could replicate a chord while only using one channel, creating a distinct sound. Chip music has its own characteristic appeal separate from most music, and that's what inspired me to start writing chip music of my own.

I'd also like to mention a YouTube user who goes by maromaro1337. He's a musician who gained popularity on the site by creating covers of songs using nothing but Stylophones, which is a brand of pocket synthesizers from the late 60's. What I admire about his work is how despite how limited the Stylophone is in functionality, he's able to create music which I find enjoyable. I see it as an example of how the capabilities of an instrument only go so far to inform the quality of a piece of music. Performance and production also play an integral role in the result. Essentially, a good song can be made with any device capable of producing sound if enough care is put in to craft the result.

SYNTHESIZER DESIGN

My design came off the basis of wanting to incorporate one of each of five of the components which characterize a synthesizer: a low frequency oscillator (LFO), a voltage-controlled oscillator (VCO), an envelope generator (EG), a voltage-controlled filter (VCF), and a voltage-controlled amplifier (VCA). The LFO creates a wave with an adjustable frequency and amplitude. Its minimum frequency is below that of the minimum frequency humans can hear. Its main purpose is to modulate the VCO, VCF, and/or VCA, creating effects such as vibrato and tremolo. The VCO creates a wave with an adjustable frequency that is within the range that humans can hear. The frequency of this wave determines the pitch of the output. The EG creates a signal which varies over time in a way that depends on whether a key is held or not. The VCF is a low pass filter. Any wave can be described as the sum of individual frequencies. What a low pass filter does in the frequency spectrum is reduce the amplitude of all frequencies above a cutoff frequency. This cutoff frequency is the aspect of the VCF that's adjustable. The cutoff frequency of a filter will influence the shape of the output, resulting in a different sound from the output wave compared to the input wave. The VCA is used to adjust the volume of the output, which could be done with any amplifier circuit. However, it is also required so that the output could drive something such as headphones and speakers, which have a low impedance. Most amplifier circuits can't drive a low impedance device, so I had to find one that could.

I decided to arrange these components such that the VCO led into the VCF, and then the VCA. The LFO would be used to modify the pitch of the VCO, allowing the synthesizer to have vibrato. The EG would be used to control the VCF, allowing for the timbre to change over the duration a note is held. Some of the specifications I decided on for my synthesizer included using a sine wave for the LFO, using a sawtooth wave for the VCO, and using a 9-volt battery to power it. Sine waves and triangle waves sound almost identical when used as vibrato. I chose a sine wave because, before I investigated any LFO designs, I assumed that the design required would be simpler. In addition, I thought the smoothness of a sine wave would result in the vibrato sounding slightly more natural.

When it came to designing my synthesizer, my process involved browsing the internet for designs for each component that were simple and used parts that were easy to access. Most of these designs relied on a power source with a significantly greater voltage across it and access to a ground voltage which was separate from the negative supply voltage. Because of this, I had to modify these circuits to work with the battery I chose. I also had to be careful when determining how these circuits would connect to one another in order to avoid an unexpectedly large current flowing between circuits, which would result in at least one of those circuits not functioning as intended.

In order to be able to perform with the synthesizer, I needed to create a controller. The controller would consist of a keyboard and circuitry which would output a voltage with an amplitude determined by which key is pressed. There is a universal standard for keyboards called Musical Instrument Digital Interface (MIDI). If my synthesizer was able to interpret MIDI, it would be compatible with a large variety of keyboards. However, as the name suggests, MIDI is a digital format, which would place it beyond the scope of this project.

Ultimately, I decided that it would be easier to build my own analog controller using buttons. I chose to implement 25 keys, which is equivalent to two octaves with an extra key so that it starts and ends on the same note. Through my experience with synths and keyboards, I've found that 25 is roughly the minimum number of keys flexible enough to play what I need to. This makes sense considering how the human voice, ignoring falsetto, has a range of about 2 octaves. Therefore, it can be expected that most melodies are within that range.

My controller design consists of a chain of resistors. Holding a key creates a current from it and the ground. As a result, each lower key on the keyboard adds a resistor to the chain, resulting in a lower output voltage. In addition, holding two keys creates a short circuit between them, resulting in only the higher key influencing the output. I added a potentiometer in the chain of resistors, allowing for the master pitch of the synthesizer to be manipulated. This allowed it to have a higher range than the 2 octaves allowed by the keyboard, improving its flexibility.

When no keys are pressed, the synthesizer doesn't stop making sounds. With the controller design I mentioned, releasing a key would result in the pitch becoming low. I added a sample-and-hold circuit to the output to fix this issue. This circuit uses a gate voltage as one of its inputs, which is just the voltage at the highest key. When the gate voltage is large, the output of the sample and hold circuit is just its input. Simultaneously, there's a capacitor which charges and discharges with changes in the input. When the gate voltage is small, the capacitor is effectively unable to discharge, and the output corresponds with the charge of the capacitor.

The layout of the keyboard was based on a 1" by 8" perfboard I ordered from Digikey. I found appropriately sized square buttons which could be arranged in a keyboard pattern on the perfboard and ordered those. The perfboard is just short enough that if I hold the ends in palms, I can reach all the keys with my thumbs. Since holding it this way made all the keys accessible, this seemed like it could be the ideal way to play it. I soldered the tuning potentiometers to the back of the perfboard, spacing them out enough that they could feasibly be used without interfering with one another. I used two lengthy, intertwining wires to connect the controller to the rest of the synthesizer.

When designing the LFO, I learned that I was wrong about my assumption about sine waves being easier to generate compared to triangle waves. What I found was that if the frequency was expected to change, then the easiest way to generate a sine wave would be to create a triangle wave, then approximate a sine wave by limiting the output of it. As a result, the shape of the LFO was far from what I wanted, but it sounded identical, so it wasn't worth trying to improve.

When designing the VCO, I had to decide upon the relationship between the input voltage from the controller and the frequency of the output. For the design I was using as a base, the relationship was 1 volt/octave. However, since I had to adjust the circuit to account for the voltage of the battery, that consistent relationship was lost. I wasn't willing to figure out how I could achieve that relationship again. This led to the decision to add a potentiometer between each key of the keyboard. This decision would reduce the synthesizer's usability because every key would have to be manually tuned, but it would also make it more flexible.

The purpose of the envelope generator is to shape how an aspect of a note progresses over time. That length of time is separated into three stages: the attack stage, the decay stage, and the release stage. When a key is held, the note enters its attack stage, during which the output rises from its initial level to its peak level. Once the peak level is reached, it enters its decay stage, in which it falls to its sustain level. If all the keys are released at any time during these stages, it enters its release stage, during which the output falls back down to the initial level. Each stage has its own independent rate of change.

This envelope generator design is unique from the envelope generators on the other analog synths I own for two reasons. One was that it could control the VCF rather than the VCO or the VCA. At the filter's minimum cutoff frequency, the output is silent, meaning that this would allow the EG to be used to shape notes the same way as the it controlled the VCA. However, the transition between maximum and minimum amplitudes would result in the timbre of the output changing, which would make the sound more complex. The second thing that made it unique is that the initial and peak levels are variables. All the other analog synthesizers I own use a predetermined initial and peak level. For my design, I wanted the user to be able to define these two levels to offer more flexibility.

The VCF was designed with a range of cutoff frequencies large enough that the minimum EG output would result in the VCF output being silent and the maximum EG output would result in it being an unmodified sawtooth wave. The VCA mainly consists of a simple circuit with a potentiometer, allowing for the attenuation of the signal, followed by a pre-designed audio amplifier module. The output of this module leads into a 8 millimeter audio jack, allowing the output to be sent through headphones or a speaker.

In order to construct this design, two custom printed circuit boards were designed and manufactured. One of which contained the LFO, VCO, and VCF. The other only contained the EG. The keyboard, the potentiometers, and both PCBs were wired together on a large piece of perfboard. From left to right, the potentiometers are the volume, master pitch, LFO frequency, LFO amplitude, EG attack rate, EG decay rate, EG release rate, EG initial level, EG peak level, and EG sustain level.

WRITING MUSIC

I ended up writing six songs, totaling almost 19 minutes. I wrote and recorded them using a type of software called a Digital Audio Workstation (DAW). The one I purchased and am most familiar with is FL Studio. This software allows music to be produced either by sequencing the note data and then playing it using a software synthesizer plug-in or arranging audio clips. These clips can be recorded within the software by connecting an input device such as an audio interface. A DAW also handles applying effect plug-ins to the audio as well as mixing the result together. DAWs such as FL Studio often come preloaded with various synthesizers and effect plug-ins, but more can be installed.

I initially hoped my synthesizer was usable enough by the time I was ready to start writing music. However, for several months after it was built, my device was suffering from a variety of issues. The most significant of these included too much current being drawn from the controller from other section, making the pitch inconsistent and the controller difficult to tune. In addition, the EG didn't work properly. As a result, I couldn't follow through with my plan. Instead, I started writing music without it and hoped that I could get the synthesizer more functional by the time I was ready to record it.

I started creating music by arranging notes in FL Studio. In order to emulate the synthesizer I had built, I used a preinstalled plug-in called "3x Osc". As the name suggests, this plug-in produces sound by mixing 3 oscillators, each with a choice between several simple waveforms. This plug-in also includes an EG and LFO for modifying the pitch, volume, panning, filter cutoff frequency, and filter resonance. For this project, I disabled all but the first oscillator and selected the sawtooth waveform. When modifying these sounds, I only changed the pitch LFO and the cutoff frequency EG.

I occasionally varied from this limitation for parts I didn't plan on recording with the synthesizer. This included using the 3x Osc for plucky square waves. For drums, I borrowed from a series of sample packs from 1991 known as *Zero G Datafile*. In "Persistent Presence" I added a part which utilizes flute samples from a 1960s electro-mechanical keyboard known as a Mellotron.

I tend to write music by developing an idea in my head, sequencing the idea, and then using what I sequenced as a springboard to fill out the rest of the song. I usually start a song with a melody in mind because I often find it difficult to try to create a melody out of a chord progression. The first 1:20 of "Surrender" is an example of a song with a chord progression written before the melody. The key I work with is never chosen deliberately, but if a vocal part I end up writing results in notes beyond the range I'm comfortable with, I'll often change the key.

When it comes to time signatures, most of my ideas use four beats per measure, so that's what my music ends up being most of the time. With this project, I wanted to add variety, so I wrote "Artifacts" to be in three beats per measure. One of my favorite moments occurs in the climax of "Steam," which shifts from four beats per measure to eleven beats per measure. The song treats this as alternating between a measure of five beats and a measure of six beats. It sounds as if the former is cut off early, so there is a feeling of resolution in the latter part. During this section, there's a polyrhythm in which there's an arpeggio which continues at four beats per measure. This results in a sound which is more chaotic than the rest of the song, but not so much so that it can be interpreted as noise. It is satisfying when the song returns to four beats per measure because it resolves a tension created by the uncommon time signature.

Once most of a song is written, I start on the lyrics. I have a list of topics which I feel like I could extrapolate on enough to make them into lyrics. I'm not confident in my ability to express simple ideas in vivid ways, so I instead like to try expressing complex topics in direct ways. My main inspiration for this style of lyric writing is The Flaming Lips, whose songs are often emotional and existential, but their lyrics convey those themes with simple statements. "Do You Realize??," one of their most popular songs, is about the importance of living life in the moment. The pairing of this topic and the blunt terms used to discuss it make it powerful. I don't believe I'm nearly at the same level of ease in terms of self-expression that The Flaming Lips are, but I believe that I will continue to improve with practice [3].

I don't take the song's tone into much consideration when choosing which topic to pair it with. Due to the melody of the song shaping the rhythm of the lyrics, I believe that as I workshop the lyrics, the tone of the song will subconsciously guide their tone. My main goal when writing lyrics is to convey facets of the topic I chose while staying within the rhythm of the melody and maintaining a consistent rhyming scheme. Rhyming isn't necessary, but generally, I find rhyming lyrics more memorable. In addition, breaking away from a consistent rhyming scheme can be meaningful. On the other hand, some lyrics, such as the entirety of "Coarse Correction," were written at the last second. In these instances, I decided to favor the aesthetics of the words over their meaning.

RECORDING MUSIC

A couple months after I wrote most of the music, I was able to get my synthesizer functioning well enough to record it. One of its remaining issues is that it can’t be recorded directly. Even if I daisy chain other synthesizers, guitar pedals, and amplifiers together, as long as the output touches either my laptop or my audio interface device, it ends up drawing too much current from its power supply. The way I settled on solving this issue was by plugging it into a guitar amplifier, then recording it with a microphone. This resulted in the ways that sound travels throughout my room influencing the sound being recorded, subtly changing its timbre.

In the process of recording the synthesizer, I would choose a short section of music, tune the synthesizer so that it’s in the right range to play that music, adjust the potentiometers until the sound of the synthesizer reflected my preference for that specific part, and finally record multiple takes until I got one that I considered usable. There are a couple instances where I improvised the section I was recording. I was hoping to do this more often so that the music felt like it was more influenced by the synthesizer, but time restraints made me prioritize pre-written parts.

In order to vary the sound of each part in the recording, I employed a variety of effects: Delay regularly repeats a part after a time interval, except quieter and sometimes with a filter applied. This can help that part stand out among the others. Chorus causes a difference in phase between the left and right channel, making a part sound wider. Phasor adds a filter which oscillates over time. This can give movement to a part that might otherwise be unchanging.

Pitch correction is an effect which adjusts the frequency of a wave so that it's in tune. This is commonly used with vocals as a means of fixing pitch mistakes with a take without having to record a new one. This use of it is meant to be subtle enough that its presence is difficult to identify. On the other hand, it's also used as a vocal effect which can be identified with immediate jumps in pitch and an unnatural lack of variability.

I found this effect useful as a way of ensuring synths are in tune. Unlike vocals, synths don't sound noticeably different when they're autotuned, which means I can apply it to any recording without having to fine tune it. The most significant impact it can have is removing the LFO from a recording that uses it. Considering how synths such as the one I created can be finely tuned, this effect isn't necessary. However, if the LFO's amplitude is set to be high, pitch correction can be used to create a rudimentary arpeggio. There are other synths I have, including a theremin, which aren't designed to be tuned to a quantized pitch, making this effect significantly more useful. The plugin I used to achieve this effect is Graillon by Auburn Sounds. This plugin has an interface which shows both the difference between the pitch of the input and pitch being tuned to, making it useful for tuning any pitched instrument.

Distortion can be described as any warping of the shape of a sound wave, usually in a way that emphasizes dissonant frequencies. An example of this kind of distortion is audio peaking, which occurs when the input exceeds the boundary of the output. When this happens, the peaks of the waveform are cut off. As a result, the details are obscured, and the sound is harsher. I use distortion when I want an instrument to sound more powerful. It's also useful as a method of replicating the sound of an overdriven electric guitar or bass, which I did in "Coarse Correction."

Bitcrusher involves digitally reducing the resolution of a piece of audio either by reducing its sample rate or reducing its bit depth. Reducing the sample rate quantizes the wave in the time domain, causing gradual variations in the waveform to become sudden. This gives the sound a square wave-like quality. Reducing bit depth quantizes the wave in the amplitude domain, making it sound noisier. In recent times this has become one of my favorite effects because of how it changes the timbre of a piece of audio, such as the increased emphasis on high frequencies. Most notably, I applied bitcrusher to the drum samples, giving them a sound that's punchier and unconventional.

Vocoder is an effect which uses the frequency content of one piece of audio to filter the frequency content of another. In music, its most common use is having vocals filter a synthesizer. The result can still be interpreted as speech but contains the same pitch and timbre of the synthesizer. This is often used to give vocals an artificial quality, such as depicting the voices of robots.

I'm not confident in my singing ability. Because of this, using an effect such as vocoder allows me to offload the work of staying in tune to the synthesizer. Pitch correction can also be used for this purpose but can easily result in something that sounds off-putting. In addition, the vocoder has the benefit of allowing me to sing beyond my vocal range if I ever need to. Another reason why I like the vocoder is that it's flexible. There is no limit to what audio can be modulated, which means there is more variety in what the result can sound like compared to unedited vocals. Even a simple sawtooth wave can create pleasant-sounding vocals with a vocoder. Having more flexibility with how my vocals can sound means that it's easier to give separate characters their own distinct voices if I need to.

SYNTHESIZER EVALUATION

My synthesizer design was sufficient for the purpose of this project, but it was host to a multitude of issues which kept it from meeting the same level of usability as the other synthesizers I own. The idea of holding the keyboard rather than laying it down proved to be less ideal than I had hoped. The flatness of the perfboard made it uncomfortable to hold this way. I thought it would be better to play the keys with thumbs rather than laying every finger on the keyboard, but this resulted in time having to be wasted when switching between notes. This made it infeasible to play a run of notes. On monophonic synthesizers, it’s still beneficial to hold multiple keys at time because it results in an immediate shift to the next pitch as soon as one finger is released.

In addition, the flexibility of the perfboard made the keyboard slightly more difficult to play without a surface behind it. Fortunately, the potentiometers on the back provided a sufficient platform so that the synthesizer could be performed with the controller resting on a surface. I was afraid that friction from slight movements of the controller might affect the tuning, but it wasn’t an issue. As a keyboard, it worked sufficiently, which was more than I could ask for. The keys are spaced better and provide better touch feedback than any handheld synthesizers that I own, which are comparatively more compact. The rest of the potentiometers are also well spaced. The only potentiometer which I felt was frustrating to use was the one controlling the master pitch. This is because tuning requires more precision than I needed with any other potentiometer. As a result, the pitch was sensitive to the potentiometer’s movements.

Further impacting the synthesizer’s quality is the sound. The VCA design I used caused distortion, making it sound softer than it should be. The result is distinctive, which can be both a good and a bad thing. It means that the synthesizer fulfills a role other synths don't, but it also means that applying a filter is less effective at changing the shape of the sound. My father compared its sound when a significant attack rate is used and the LFO is minimal to a chord organ. This is an apt comparison considering how pressing a key on a chord organ results in a space being uncovered. When between a fully covered and uncovered state, the sound of the chord organ gets muffled, which is effectively the same as applying a low pass filter.

The issue I have with the synthesizer which most significantly affects its performance is its flawed sample-and-hold circuit. When a key is released, the pitch drops by an inconsistent amount. Any significant unwanted change in pitch like this is going to result in the part being played being interrupted by wrong notes. The solution to this would be to have the release rate constantly set to be instantaneous and the initial level constantly set to low. This wouldn't be such a problem if it didn't tie into another issue I had with it.

The EG is flawed in that each variable is somewhat dependent on another variable which should be unrelated. For example, in order to achieve a long-lasting attack or a fully decreasing decay, the release rate potentiometer can't be instantaneous. The only way to avoid triggering an audible wrong note is to use a low sustain and let the decay stage reduce to its minimum before releasing the key. In some of the recorded parts, some of these wrong notes were inconsequential enough to leave in. However, for the most part, I used an instantaneous release rate to avoid the issue altogether.

CONCLUSION

The music that resulted from this process is unfinished, but I believe it's on track to becoming something great. The composition was given plenty of effort and ideas, making it full of moments that I believe people will find memorable. My largest issue with the compositions is the structure. Some songs have somewhat awkward transitions or lack conclusions. I think what brought down the product the most, however, was the lack of sufficient time spent ensuring that everything that should've been recorded was and that the recordings themselves were of a sufficient enough level of quality. Vocals sang rushed lyrics and were recorded without sufficient practice beforehand. Little time was spent on ensuring that each part lined up with the tempo. This was compounded by a lack of time spent mixing. As a result, I think that despite the quality of the composition, the messiness of how it sounds is going to keeping it from appealing to most people's tastes.

I believe a good metric for determining the utility of the synthesizer would be to compare it to the sound of the software instrument I used to emulate it. Despite both working with the same type of waveform, the software instrument ended up being more flexible due to a lack of inherent distortion. It was also more convenient to use and synced better with the tempo because it doesn't have to be performed, but it still allows for that option. On the other hand, this synthesizer has a sound which I can't easily be replicated in software, and I think that's enough for a synthesizer to be considered useful.

If I were to have the chance to redo this project, my biggest priority would be to reduce the complexity of the synthesizer. As was demonstrated with the sample-and-hold circuit and the envelope generator, some otherwise useful aspects of the synthesizer had to be disregarded when recording due to them not working properly. In fact, there were several other issues with it well into March that had to be fixed before anything could be properly performed using it. By reducing the complexity, the project would be more manageable, giving me more time to fix issues and go on to put together and record the music. This would result in the synthesizer being involved in the process of creating music.

In addition, I would've managed my time better. There were periods such as during September 2022 and winter break where I knew the importance of working on my synthesizer in order to prepare it for this project, but I instead prioritized less important things and had to rush afterwards to catch up. In addition, if I had started the synthesizer design during summer break last year, I could've gotten more comfortable with the process by the time the classes started, leading to that semester going more smoothly.

I think the music I make in the future is going to benefit from this experience. While writing music for this project, I've become more familiar with FL Studio and the features it offers. It's due to these features that I believe I was able to achieve a broad range of sounds despite the limited instrumentation. In addition, the practice this experience offered in lyric writing has been helpful to improving my ability.

Designing and building the synthesizer was a difficult albeit rewarding process. Before my capstone project, the circuits I had worked with were simpler. In addition, I didn't know how to design circuit boards and wasn't as skilled when soldering. By familiarizing myself with aspects of the designing and building process, I believe that I have better prepared myself for a career in engineering.

WORKS CITED

[1] White Noise, *An Electric Storm*. Island, 1969.

[2] B. Eno, *Another Green World.* Island, 1975.

[3] The Flaming Lips, *Do You Realize??.* Warner Bros., 2002.

APPENDICES

APPENDIX A: LYRICS

<u>Persistent Presence</u>
I will never make it back
To the places I once knew
To the person I once was
The first time I met you
But I can feel you still in here
You've made your presence known
I know you want to help me
But I have to go alone

<u>Monumental Dread</u>
I've waited all my life for this moment
And when it comes I'll be relieved
For all the suffering will be worth it
At least that's what I'm told to believe

Keeping me captive
Cause I know there's one other way
For all these years I've been waiting
For today

There are thoughts that will come into my mind that I not allowed to share
There aren't words which could ever describe my overwhelming despair
I can't appeal
I've been naive
Now the lies are revealed
But I can't leave

I've been dying to break free
They need me inside again
Don't think they can make me
I just need to find my
They make offers to me
They know me more than myself

I'd been graced with sickness
Something messing with my head
They gave me forgiveness
Made me lucky I'm not
Don't you know it's wrong to
Question the authority

Keeping me captive
While I know there's no other way
For all these years I'd been waiting

<u>Coarse Correction</u>
For all that they said was true
What do they believe they know
There's zero comfort in your partner's eyes

All hope is gone
At the break of dawn
The spirit leaves and the air is mystified

It's a point of pride
Dry-eyed
Taken for a ride
When there's nothing left to hide

Overture of flight
Twilight
Fade into the night
While there's nothing left to fight

Taken part of every decision
And course correcting to break up division
The storm breaks and I find the eye
So much further in and I begin inspecting it to check for supply
There's little effort in these leaves
After the petal was received

Mark all the trails within the seals in the gateway
I'll let the envelopes in jars wash away
Powerful forces rip the jars and shred the maps
But leave the envelopes enclosed and in tact
Know that the comedy will come to those alive
All of eternity dies

It's a point of pride
Dry-eyed
Taken for a ride
When there's nothing left to hide

Overture of flight
Twilight
Fade into the night
While there's nothing left to fight

The planets overturns
Sideways
Only there to burn
And there’s nothing there to learn

Finally we’re free
In no time
Never to be seen
And there’s no one to believe

<u>Artifacts</u>
Ever since the death of time
Every mind
Had resigned
Isolation left us all powerless refugees

Back then we once had it all
Ruins left from the fall
The structure may be in tact
But there are no guarantees

I’ll stay
Searching for hours
Through man-made
Constructed decay
Stairways
I’m left surrounded
In graceful unbound disarray

Never seen and never heard
Artifacts of printed word
Knowledge of a universe that was yet to be seen

What I’d give to have it back
Get a glimpse through the cracks
I have nothing left to lose
So I’ll wait

I’ll stay
Searching for hours
Through man-made
Constructed decay
Pathways
I’ve fell in love with them
Letting myself go astray

Diversion
From where we are
It feels so close to where we started
More than it appears

We've traveled far
We're braver than when we departed
Wiser to our fears

It's all immeasurable
How life's so difficult
Things never seem to go as planned

Relationships disband
End up where we began
But while we can
We still get up and start again

I can't let go
When everything still weighs upon me
More than what it's worth
You have to know
That everything still preys upon it
Primed for a rebirth

The seasons change again
I'm still the way I am
Take cover overall
It's worth it just to feel it
And we could take it back again

It's all immeasurable
How life's so difficult
Things never seem to go as planned

Relationships disband
End up where we began
But while we can
We still get up and start again

Surrender
I'm asking you to leave
Think it over
How you could ever see

If I leave you long enough
You’ll forget me
What meaning did I have

I think what I can gain is
Worth less that what I’ll lose
But still I push away

I don’t know what I want
Insecure with what I have
There’s nothing set in stone

And as these flaws persist
I continue to resist
And keep it all unknown

I can’t get through to you
I’d compile my thoughts
But they’d get lost
And I'd surrender

I'll only trust myself
I’d rather fail
I’ll fail alone
And start it over

I'll just convince myself
They can't understand
They won't understand
Until it's over

I rely on faith
That the path I've chosen is what’s best for me
No other options

(You’ll make it better)
(I’ll help you through the darkness)
(Let go)

ECE 403 Senior Project Report: Analog Synthesizer University of Maine

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April 5, 2023

Abstract

A monophonic analog synthesizer was designed, built, and tested. The device has a keyboard consisting of 25 buttons which each set the frequency of the output signal when pressed. Several additional inputs control how the signal is modified. Components which these inputs affect include the controller, Low Frequency Oscillator (LFO), Voltage-Controlled Oscillator (VCO), Envelope Generator (EG), Voltage-Controlled Filter (VCF), and Voltage-Controlled Amplifier (VCA). The output port of the device is a 1/8th inch audio jack, allowing the synthesizer to be used as the input for an audio output device such as a speaker. The device is battery powered so that it can be used portably, allowing for an ease of use which is offered by commercial synthesizers of a similar size.

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1 Introduction

This report describes the design and operation of a portable analog synthesizer. This device is capable of generating and manipulating an audio waveform. It differs from a digital synthesizer in that the waveform is produced using analog circuitry. It is typically used for the purpose of sound design due to how it gives the user more precise control over a resulting sound compared with physical instruments. This analog synthesizer in particular employs subtractive synthesis as a method of controlling the timbre of the resulting sound. This type of synthesis utilizes a filter to modify the harmonics of a waveform such as a sawtooth wave.

1.1 Purpose

This synthesizer was designed and built in order to fulfill the requirements of The University of Maine's electrical and computer engineering capstone project. A synthesizer was chosen in particular due to its utility and flexibility in creating sounds and performing music. It was decide that the synthesizer would be analog because it is simpler to work entirely with analog circuits rather than with a mix of analog and digital components, which would require programming a processing unit. The choice to have the device rely on battery power improves its accessibility. In addition, it means that the power source is DC, which is simpler to work with compared to using an AC power source such as a power outlet.

1.2 Specifications

A contract was written which laid out the a set of specifications which were to be met with the design. This contract can be found in appendix A. The design of this synthesizer is broken up into multiple blocks. These blocks consist of a controller, a Low Frequency Oscillator (LFO), a Voltage-Controlled Oscillator (VCO), an Envelope Generator (EG), a Voltage-Controlled Filter (VCF), and a Voltage-Controlled Amplifier (VCA). The controller interprets inputs from a series of buttons and creates a signal with an amplitude which corresponds with those inputs. This series of buttons is referred to as a keyboard and each button is referred to as a key. The LFO creates a waveform with an adjustable frequency which can go below the range of human hearing. The VCO generates a waveform with a frequency which is determined by the voltage input and is within the range of human hearing. The EG creates a multi-stage contour. The VCF filters an input waveform with a cutoff frequency determined by a control voltage. The VCA amplifies or attenuates a waveform so that it can be sent to a device such as a speaker. The contract describes the synthesizer as having one of each of these blocks. Because there is only one VCO, the synthesizer is considered monophonic. The layout of these blocks is described such that the waveform is passed from the VCO to the VCF and then the VCA. The LFO and EG are used to control the VCO and the VCF respectively.

The contract specifies that the inputs must include energy from a battery, button controls, and potentiometer controls. The output must be a synthesized signal which can be accessed through a 1/8th inch mono audio jack. 25 buttons must correspond with an output which has a range of 2 octaves. This assumes that 12 tone equal temperament is being used, which has 12 notes per octave. 25 buttons were chosen instead of 24 so that the keyboard can start and end on the same note of the octave. The LFO must generate a sinusoidal waveform. The shape of the output of the VCO must be a sawtooth waveform. The output of the VCO must have a frequency range with a minimum lower than E2 (82.41 Hz) and a maximum larger than E6 (1318.51 Hz). These pitch-to-frequency conversions were performed such that A4 is equal to 440 Hz. A conversion table created by Suits was used as a reference for these conversions [1]. The EG must be controlled by six parameters. The attack, decay, and release stages of the EG must have a minimum time of $\sim$0 seconds and a maximum time of at least 2 seconds. The synthesizer must be built to include a custom PCB design and not include a development board.

Specifications were verified using a Digilent Analog Discovery oscilloscope and a computer with Waveforms software. Every contract specification can be determined by analyzing the voltage of a node on the device. For example, the frequency and shape of the VCO can be determined by recording the voltage at the VCO's output. In addition, every specification has enough room for error that precise equipment is not required.

1.3 Comparison to Similar Devices

One synthesizer which has similar capabilities to this design is the Dubreq Stylophone Gen X-1. This model of Stylophone features the same components as the one described in this report, but functions differently. Instead of buttons, the controller includes a metal keyboard and a stylus. When the stylus touches a key, it completes a circuit, setting a control voltage which corresponds with the frequency of the output. Alternatively, a ribbon can set the pitch depending on the position where pressure is applied to it. Unlike the keyboard, the ribbon is not quantized, allowing for a smooth transition between notes at the cost of a lack of precision. A potentiometer controls the master pitch of the Stylophone. Only one potentiometer is necessary because every note scales with the pitch, keeping each key in tune relative to one another.

The envelope generator in the Stylophone is simpler than the one in this synthesizer. It only allows the attack and decay to be adjusted. Instead of controlling the VCF, the EG controls both the VCO and VCA. The EG incorporates a potentiometer which adjusts the influence the EG has over the pitch. The LFO features a switch which swaps between a square and triangle waveform. The VCF features a resonance potentiometer, which can amplify the cutoff frequency of the wave. In addition to these features, this Stylophone adds an adjustable delay block between the VCF and VCA. This adds feedback to the sound, creating an echo effect. It also adds three extra VCOs which can each be toggled. The first of these mimics the main VCO, except it has a phase offset, creating a chorus effect. The second and third are one and two octaves lower respectively, adding lower frequencies to the output.

This report will document the process which was used to design, build, and test a portable monophonic analog synthesizer. Section 2 will discuss its functionality. Section 3 will discuss the design process. Section 4 will discuss the method for testing the synthesizer and present the results of those tests.

2 Breakdown

This section discusses the blocks that make up the synthesizer and functionality of each of them. The circuit simulation software Micro-Cap was used in the process of designing this synthesizer. Figure 1 shows the block diagram for it.

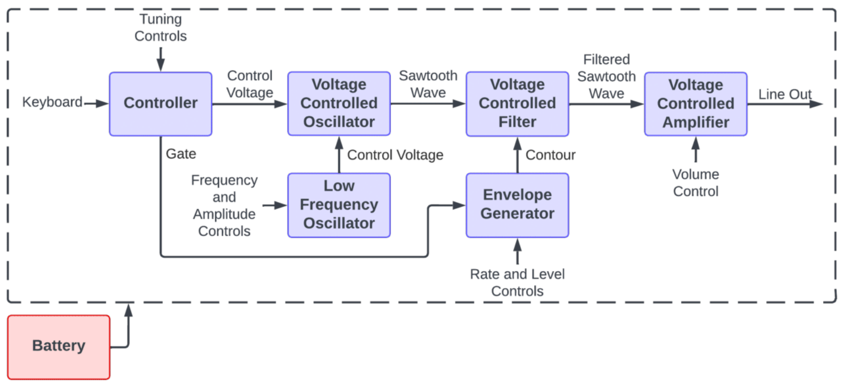

Fig. 1. The functional block diagram for the analog synthesizer.

The synthesizer works by interpreting button presses and potentiometer settings and creates a wave with a pitch and shape which can change over time. This wave's pitch is determined by key presses and tuning potentiometers. In addition, this pitch oscillates over time, with the rate of oscillation being the LFO frequency potentiometer and LFO amplitude potentiometer. When a key is pressed or released, the shape of the wave changes over time. How that shape changes is dependent on a set of rate and level potentiometers. The amplitude of the wave is affected by the shape of the wave, but it can be further modified using the volume control potentiometer, which is used for attenuation. This wave is capable of being interpreted by audio output devices such as headphones and speakers so that it can be converted into sound. The entire system is powered by a battery.

2.1 Controller

The controller interprets key inputs from the user and creates a control voltage output. Each key has a corresponding output voltage which remains for as long as that key is held. This voltage represents the target pitch of the synthesizer's output. If multiple keys are held, the control voltage corresponds with the rightmost key being held. Each of the 25 buttons has a potentiometer which changes the control voltage corresponding with the key as well as every key to the left of it. The potentiometer corresponding with the rightmost key is used as the master pitch control and functions differently from the rest. In addition, the controller creates a gate voltage output which goes high for as long as at least one key is being held. It goes low otherwise.

The control voltage is used as the input of the sample-and-hold circuit, which will either output the control voltage unmodified, or hold on a previously sampled control voltage. Whether the circuit is sampling or holding depends on a separate input which is connected to the gate voltage output of the controller. As a result, when no keys are being held, the output will hold on the voltage corresponding to the last key that was pressed.

2.2 Low Frequency Oscillator

The purpose of the low frequency oscillator in this synthesizer is to modulate the VCO. This causes the output frequency to oscillate around the pitch chosen by the controller. When the LFO is at a frequency of around 5 Hz, this creates an effect known as vibrato. The result of the effect is a sound which is perceived to be more natural. The shape for this wave is a sine wave, which is shown in figure 2. In order to create a sine wave, the LFO generates a triangle wave, then a triangle-to-sine approximator circuit converts the triangle wave into an approximation of a sine wave. This approximation should be audibly indistinguishable from a perfect sine wave when used for vibrato.

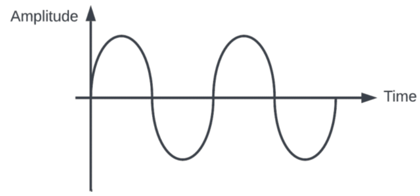

Fig. 2. Sine wave.

2.3 Voltage-Controlled Oscillator

The voltage-controlled oscillator generates a waveform with a frequency determined by the sum of the outputs of the controller and the LFO. The frequency of this waveform determines the pitch of the synthesizer's output. Because of this, the frequency range should be within the range of human hearing, which is between 20 Hz and 20 kHz. The shape of this waveform is a sawtooth wave, which is shown in figure 3. The sawtooth wave was chosen because it is rich in a harmonics, containing both even and odd harmonics. This results in the shape of the wave changing dramatically depending on the cutoff frequency of the filter. This results in the output having a wide range of possible timbres.

Fig. 3. Sawtooth wave.

2.4 Envelope Generator

The envelope generator creates a contour which varies with time. The design used for this synthesizer is a variation on a common four-variable design known as ADSR (attack, decay, sustain, release). With the ADSR design, the initial level and the peak level are the minimum and the maximum possible output voltages respectively. This ADSR design is commonly used with EGs which control a VCA. In the case of this synthesizer, which uses the EG to control the VCF, making the initial and peak levels variables introduces more flexibility in terms of variation in timbre. Figure 4 shows the expected output for the EG design used in this synthesizer.

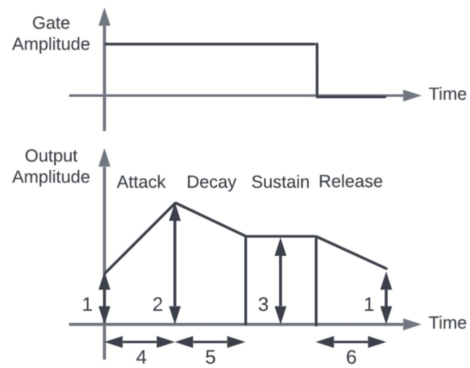

Fig. 4. Envelope generator functionality.

The EG design for this synthesizer involves six variables which are each controlled by a potentiometer. These variables are the initial level (1), the peak level (2), the sustain level (3), the attack rate (4), the decay rate (5), and the release rate (6). When the gate voltage is low, the amplitude remains at the initial level. While the gate voltage is high, the amplitude will raise at a speed determined by the attack rate until it reaches the peak level. This period of time is referred to as the attack stage. Then, the amplitude will decrease at a speed determined by the decay rate until it reaches the sustain level. This period of time is referred to as the decay stage. The amplitude remains at the sustain level until the gate voltage goes low. If the gate voltage is low, it decays at a speed determined by the release rate until the initial level is reached again. This period of time is referred to as the release stage.

The rates are independent of the levels, meaning that the time spent in a stage can vary depending on the difference in amplitude of the starting and ending levels. The peak level should always be greater or equal to the sustain level, which should always be greater or equal to the initial level. If the initial level is set to be higher than the peak level, then the output will remain at the initial level. If the sustain level is too high or too low, then it will effectively be equal to the peak level or the initial level respectively.

2.5 Voltage-Controlled Filter

The voltage-controlled filter takes a waveform as an input and modifies it by reducing some of its frequency components. This filter is a low pass filter, meaning that all frequencies above a specified cutoff frequency are reduced. A control voltage input is used to determine the cutoff frequency of the filter. Changing the cutoff frequency changes the shape of the output waveform, which influences the timbre of the resulting sound. Therefore, having a large range of cutoff frequencies would maximize the range of timbres that the synthesizer can create.

The range of the cutoff frequency should be large enough that when the control voltage is at its minimum, the output is silent. This means that the minimum cutoff frequency should be much less than 82 Hz, which is the minimum frequency of the input waveform. When the control voltage is at its maximum, the output should be unmodified. This means that the maximum cutoff frequency should be much greater than 1318 Hz, which is the maximum frequency of the input waveform. A large roll-off would be preferable because it can result in significant changes in the shape of the output waveform. In addition, at the minimum cutoff frequency, it would be less likely that waveform would still be audible.

2.6 Voltage-Controlled Amplifier

The voltage-controlled amplifier modifies the amplitude of an input waveform according to the setting of a potentiometer. This corresponds with the volume of the output. The output of the VCA is a waveform which is capable of being used as the input for devices such as speakers and headphones. These devices have low impedances. In order for the output to remain consistent regardless of the impedance of the device, the output resistance of the VCA should be exceptionally small.

3 Details

This section discusses how the synthesizer was designed. A 9 V nominal battery was chosen as the system's power source. The reason for this is because 9 V batteries are common among household devices such as smoke detectors, meaning they are commonly available and relatively inexpensive. They have a high voltage compared to most other commonly used batteries, allowing for more flexibility when designing circuits without requiring multiple batteries. Due to variance in the voltage of these batteries in practice, the circuit was designed to work as intended with a power supply voltage of 8.4 V. Neither boost converters nor voltage regulators were incorporated to change the supply voltage. A +4.5 V nominal node was created by using a voltage divider consisting of two 10 kΩ resistors in series which connects the supply voltage and the ground voltage.

For the purpose of accessibility when building, the system was designed so that whenever possible, models which could be found in UMaine's part supply were chosen for frequently used components. Every op-amp used in this synthesizer is a TL074 JFET input op-amp. This model was chosen due to a +8.4 V supply being within its operating range. In addition, according to its data sheet, given a supply voltage of +8.4 V, the TL074 has a voltage range of $\sim$6 V [2]. In addition, this model has a low input bias. This is important for the LFO in particular because the input bias of one of the op-amps determines the ratio between the rise time and the fall time of the resulting wave.

3.1 Controller

The controller is made up of a keyboard and a sample-and-hold circuit. The keyboard portion of this circuit consists of 24 identical circuit segments in series, each corresponding with one key on the keyboard. One of these segments is shown in figure 5. Each of these segments includes a button and a 10 kΩ potentiometer. Figure 6 shows the end of the sequence, which includes the rightmost key and the master pitch potentiometer RV1.

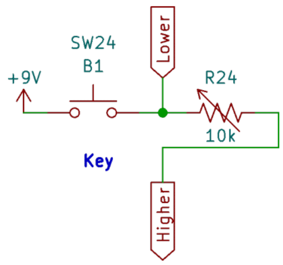

Fig. 5. Circuit segment for each key on the keyboard.

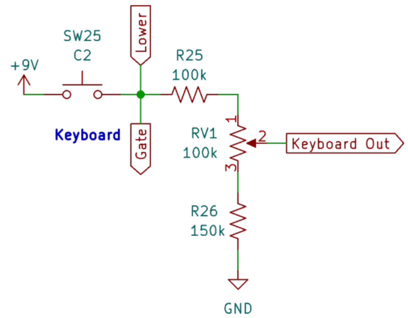

Fig. 6. The end of the sequence of circuit segments.

This circuit functions as an adjustable voltage divider. The more to the left on the keyboard a key is, the larger the resistance of the path the current travels through. The output voltage of a voltage divider is the ratio of the resistance between the output and ground and the total resistance of the path the current travels through. This means that as the total resistance of the path increases, the output voltage decreases. When two keys are held, the path of resistors in series between the two keys is shorted. As a result, the total resistance of the current path only corresponds with the path from the rightmost key to the ground. Adjusting potentiometer RV1 changes the amount of resistance between the output and the ground without changing the resistance of the total current path, resulting in the keyboard output voltage changing with it. Resistors were added before and after RV1 to limit the range of the output voltage. Their values were determined by experimentally adjusting them, then measuring the frequency of the output waveform when RV1 is at its maximum and minimum values. This process was repeated until the output had a frequency range that was slightly beyond the frequency range specified by the contract.

3.1.1 Sample-and-Hold

Figure 7 shows the diagram of the sample-and-hold circuit. This design is based on one shown in Ravi Teja's tutorial on sample-and-hold circuits [3].

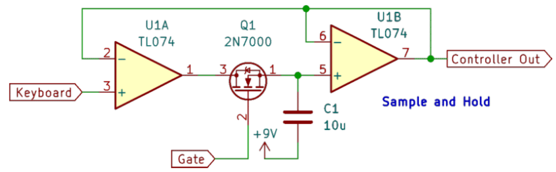

Fig. 7. Diagram of the sample-and-hold circuit.

The sample-and-hold circuit consists of two inverting amplifiers with an NMOS transistor in between. A MOSFET was chosen instead of a BJT because the MOSFET has little to no current to or from the gate, meaning that it will not influence the current path of the keyboard. The 2N7000 model was chosen because it is commonly available and is often used as a switch.

A capacitor connects the supply voltage with the source of the MOSFET. The gate of the MOSFET is the node at the highest key. When at least one key is held, the gate goes high, allowing current to pass from the drain to the source. This results in the output of the sample-and-hold circuit having the same voltage as the input. It also causes the capacitor to charge or discharge to correspond with the voltage. When no keys are held, the MOSFET is off. The capacitor remains charged because neither the MOSFET nor the op-amp allow for a significant current flow. The capacitor will eventually discharge, but over a long period of time. This means that output voltage of the circuit will be effectively constant. The op-amps serve as buffers, preventing a significant amount of current flowing out of the output of the keyboard when the MOSFET is on and preventing current from escaping the capacitor to the output when the MOSFET is off.

The capacitor chosen was the maximum value commonly available. This is because, when holding, smaller capacitor values cause the output voltage to be lower than the input it was sampling. As a result of the capacitor being pulled up, the circuit will only hold if the input is lower than or equal to the sampled voltage. This is acceptable because no situations exist where it would be possible for the input voltage to increase while the circuit is holding.

3.2 Low Frequency Oscillator

The LFO design was based on Staffeld's Simple LFO [4]. Staffeld's LFO generates a triangle wave with a frequency dependent on the setting of a potentiometer. It then uses a triangle-to-sine approximation circuit to translate that triangle wave into a sine wave. In addition, it has a separate square wave output and an LED indicator. The entire circuit works using a ±15 V power supply.

When modifying Staffeld's LFO to work for this synthesizer, the first change that was made was to remove the square wave output buffer and the LED indicator. This is because neither were necessary for the purpose of this synthesizer. The next modification that was made was to get the LFO to function with a power supply of +8.4 V. In order for the wave to be centered around half of the power supply, the output of a voltage divider was used in place of ground. No op-amp models were specified, so the TL074 was used. The value of capacitor C1 was reduced from 1 μF to 220 nF because otherwise, the frequency range would be too low. The final diagram for the section of the LFO that generates a triangle wave is shown in figure 8.

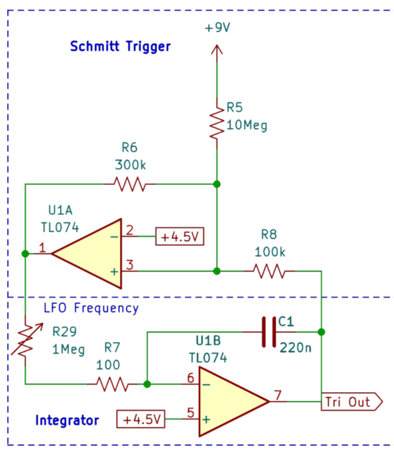

Fig. 8. Low frequency oscillator circuit diagram.

This LFO works by continuously switching between charging and discharging capacitor C1. When the output of the Schmitt trigger is high, the output of the integrator is low, so C1 charges. Once the integrator output reaches its maximum value, it causes the output of the Schmitt trigger to go low, and the capacitor drains at the same rate that it charged at. A potentiometer in between the output of the Schmitt trigger and the input of the integrator determines the rate at which C1 charges and discharges. As the resistance across the potentiometer increases, the rate of charge and discharge decreases, resulting in a lower frequency. This is because the time constant of the RC path between the input and the output of the integrator is proportional to the resistance of potentiometer R29. This means that a larger resistance results in a larger amount of time required to charge C1, reducing the frequency of the wave.

3.2.1 Triangle-to-Sine Approximation

For the triangle-to-sine approximation section of the circuit, the arrangement of the resistors and diodes was modified when compared with Staffeld's LFO. The diodes were placed in series with resistor R13, which did not have any noticeable benefits or drawbacks on the shape of the output. The ground node was replaced with a voltage divider. This voltage divider is unique because the diodes cause a variation in its output voltage. The 1N4148 model diodes were replaced with 1N914 models diodes, which are electrically equivalent. The non-inverting amplifier section was replaced with an inverting amplifier with a variable gain, allowing for amplitude of the LFO to be controlled by the user. The circuit diagram for the approximator and the amplifier is shown in figure 9.

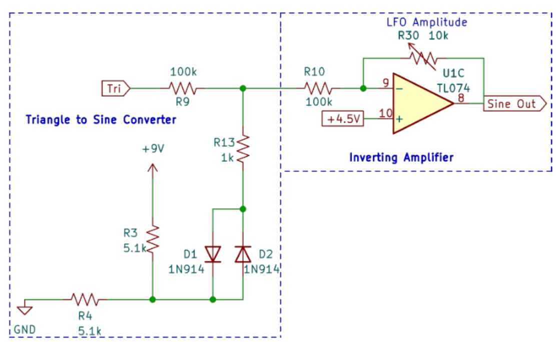

Fig. 9. Triangle to sine approximation circuit diagram.

This approximator works by using diodes to limit the peaks of the triangle wave. If the triangle wave amplitude is significantly larger or smaller than the voltage divider output, then current either flows to or from the voltage divider, limiting the amplitude of the triangle wave. The output of the approximator has a more rounded shape than a triangle wave, but retains a distinctive point at its peaks. The amplitude of this wave is significantly smaller than the triangle wave.

The inverting amplifier was designed such that its output amplitude is between 0% and 10% of its input, depending on setting of potentiometer R30. The maximum gain of this circuit was determined experimentally by listening to the output of the VCO with the LFO at its maximum amplitude. The ideal maximum gain would result in an oscillation in the VCO with a peak of around one semitone.

3.3 Voltage-Controlled Oscillator

The voltage-controlled oscillator takes in both the controller and the LFO outputs as inputs. The two signals are combined into one using a summing amplifier circuit, which consists of a non-inverting amplifier with both inputs connected at the positive node using resistors. Figure 10 shows a diagram of this circuit.

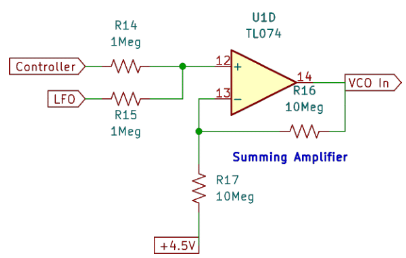

Fig. 10. Diagram of the summing amplifier used in the VCO circuit.

This voltage at the positive node of the op-amp is equal to the average of the two signals. 1 MΩ resistors were used so that the differences in the output resistances of the controller and LFO, would not result in one input taking precedence over the other. In order to get the sum of the two signals instead, the amplifier had to be designed with a gain of +2 V/V. The LFO output is a wave relative to the output of a voltage divider, so the summing amplifier was also designed to be relative to the same voltage divider. 10 MΩ resistors were used to determine the gain in order to maximize the current flow into the input of the VCO.

The main VCO design was based on Charles Staffeld's Yu-synth VCO [5]. Staffeld's synthesizer uses a ±15 V supply, so the resistors and capacitor values were modified in order for the circuit to work with a +8.4 V supply. The resistances of the Schmitt trigger were adjusted so that the output had a V_{PP} of ∼1.5 V and was centered at ∼4.2 V. The BJTs are the same models used in Staffeld's VCO. The diagram for the VCO used in this synthesizer is shown in figure 11.

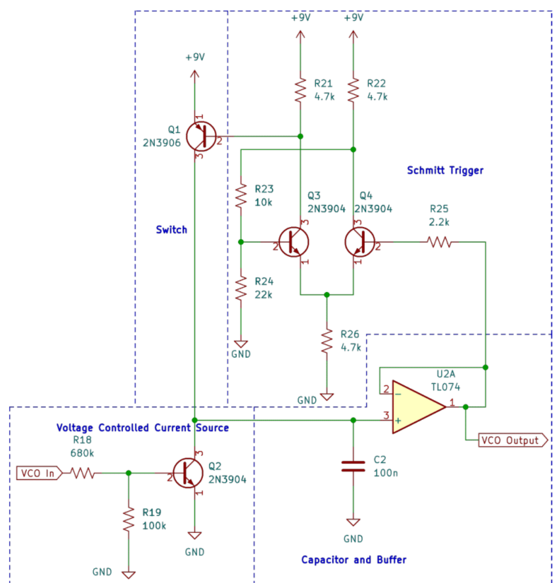

Fig. 11. Voltage-controlled oscillator circuit diagram.

This circuit consists of a voltage-controlled current source, a capacitor and buffer, a Schmitt trigger, and a switch. When the BJT Q1 is off, current drains from the capacitor through the voltage controlled current source at a rate determined by the control voltage. This results in the output voltage decreasing at a constant rate. The output of the circuit is also the input of the Schmitt trigger. Once the output voltage has lowered enough, Q4 turns off, increasing the voltage at the base of Q3.

This reduces the voltage at the base of Q1 turning the switch on. When the switch is on, current flows from the supply voltage to the capacitor, charging it quickly. This increases the output of the circuit, causing Q4 to turn on. This turns off Q3 and turns the switch back off. This cycle repeats indefinitely, creating a sawtooth wave with a frequency dependent on the input voltage.

The range of rates at which the capacitor discharges is determined by value of capacitor C2 and the voltage divider at the input of the circuit. C2 was changed from 10 nF to 100 nF in order to increase the time it would take to discharge, decreasing the frequency. The voltage divider was experimentally modified so that, given the voltage range of the controller output, the frequency range would have a minimum less than 82Hz and a maximum greater than 1318Hz.

3.4 Envelope Generator

This envelope generator design is based on an ADSR circuit created by Nicolas3141 [6]. This circuit is designed to work with a ± 9 V power supply, but switching the op-amps allows it to work with a +8.4 V supply. The BC559 PNP BJT was replaced with a 2N3906 BJT, which is a similar model which can be found in UMaine's part store. The model of diodes is not specified. The purpose of the diodes in this circuit is to direct the path of the current, so it seems like the majority of diode models would work. The model decided upon was 1N914. The diagram for the envelope generator used in this synthesizer is shown in figure 12.

The gate input was connected directly to the positive terminal of op-amp U1A. Nicolas3141's design uses a voltage divider and feedback, but they were causing detuning due to the gate node being connected to the controller, so they were removed. The EG appears to function exactly the same without them.

Op-amps have a property in which if the two inputs are significantly different voltages, they will output either high or low depending on which input is greater. The envelope generator uses this property to create a multistage contour. U1A is used to compare the gate voltage against a voltage of about 33% of the supply voltage. Resistors R4 and R5 were chosen such that it was somewhere between the minimum and maximum value of the gate voltage. This results in a high signal from

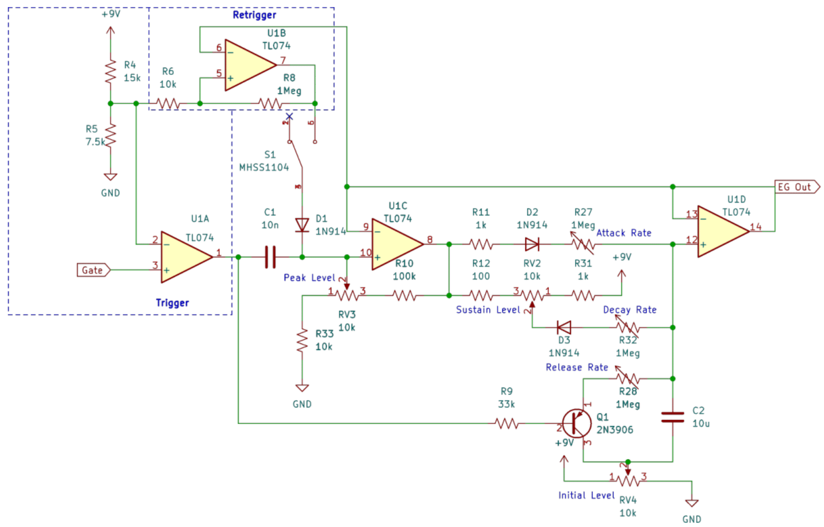

Fig. 12. Envelope generator diagram.

U1A if a key is pressed and a low signal if it is not. When the output of U1A is high, the positive terminal of U1C in turn becomes high, resulting in the output of U1C going high. This produces a current which flows through the attack potentiometer and charges capacitor C2. When this occurs, the circuit is in its attack stage. U1D is a unity gain amplifier, so the voltage at its positive terminal corresponds with the output voltage of the circuit.

Eventually, the output voltage will be greater than the voltage at the positive terminal of U1C, resulting in the output of the op-amp going low and causing a current to flow through the decay potentiometer. During this stage, the capacitor will discharge at a rate dependent on the resistance of the decay potentiometer. The sustain potentiometer creates a voltage divider. Current will continue to flow until the difference between the output voltage and the middle of the voltage divider is equal to the forward voltage of the diode.

While the output of U1A is high, the PNP BJT Q1 will be off. When the output of U1A is low, Q1 turns on, allowing current to flow through it for as long as the capacitor has a charge. This discharges the capacitor at a rate corresponding with the resistance of the potentiometer.

Two potentiometers were added to Nicolas3141's envelope generator design. The first one is RV3, which is used to determined the peak of the envelope generator's output. RV3 creates a voltage divider which determines the voltage that U1C compares the output voltage against during the attack stage. The second is potentiometer RV4, which determines the initial voltage of the circuit. It does this by functioning as a voltage divider which connects the supply voltage and the ground voltage. If the capacitor is discharging, it will eventually stabilize at the initial voltage.

Op-amp U1B compares the output voltage against a constant voltage set by a voltage divider. When the voltage at the negative terminal is less than at the positive terminal, then the output will go high. When the switch is on, current flows through a diode to the positive terminal of U1C, causing the envelope generator to retrigger while a button is held. This causes the envelope to repeat until the button is released.

The rate of C2's charge and discharge corresponds with the resistance of the potentiometer in the path of current flow. The time constant is equal to the resistance of the path multiplied by the capacitance. Large values were chosen for C2 and the attack, decay, and release potentiometers in order to maximize the largest possible amount of time it takes for C2 to charge or discharge. Each stage has a maximum time constant of $\sim$10 seconds, but the time each stage lasts for depends on the initial, sustain, and peak levels.

3.5 Voltage-Controlled Filter

The source of this VCF design was Elektrouwe's simple LED VC-LPF design [7]. This design is based on an active low pass filter, except it adds a variable current path which intersects with the capacitor path. This allows the cutoff frequency to be adjusted based on the control voltage. Elektrouwe's circuit was designed with a $\pm$12 V supply, but managed to work equally as well with a +8.4 V supply. The LT1056 op-amps were replaced with TL074 op-amps. The NSCW100 LEDs were replaced with 1N914 diodes, which do not emit light, but function the same way in the circuit. The ground nodes were replaced with the output of the voltage divider used to create a reference voltage between supply and ground. The diagram for the VCF used in this synthesizer is shown in figure 13.

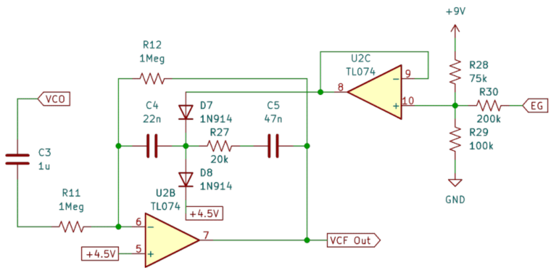

Fig. 13. Voltage-controlled filter circuit diagram.

Resistors R11 and R12 were changed from 1.5 MΩ to 1 MΩ due to a lack of surface mount resistors which have the former value. Resistor R27 influences the resonance of the output. This means that the lower its resistance is, the greater the harmonics of the input near the cutoff frequency will be amplified.

Because of the changes made to this circuit from Elektrouwe's design, the cutoff frequency range ended up being too small and had to be recalculated. Capacitors C4 and C5 had their values multiplied by 10 in order to achieve a larger frequency range. After that, resistors R28, R29, and R30 were modified with the assumption that the EG input would have a ± 3 V swing centered around +4.2 V. The goal when adjusting the cutoff frequency range was that the maximum was significantly higher than 1318 Hz and the minimum was significantly lower than 82 Hz. These changes were made within Micro-Cap, and the resulting simulated frequency response is shown in figure 14.

Based on the simulation, it was determined that the minimum cutoff frequency is 10.9 Hz and the maximum cutoff frequency is 3.13 kHz. In addition, this graph shows that at lower cutoff frequencies, the gain of higher frequencies plateaus at around -34 dB rather than decreasing indefinitely. This means that frequencies which should be inaudible may still be audible.

Resistors R28, R29, and R30 act as a voltage divider which limits the voltage at the positive input terminal of op-amp U2C. U2C acts as a voltage buffer, preventing current from flowing between the node at the positive input terminal and the node at the output. Increasing the voltage at the output of U2C results in the current through diodes D7 and D8 increasing exponentially. This relationship is shown in equation 1. It should be noted that the currents through the two diodes are equal.

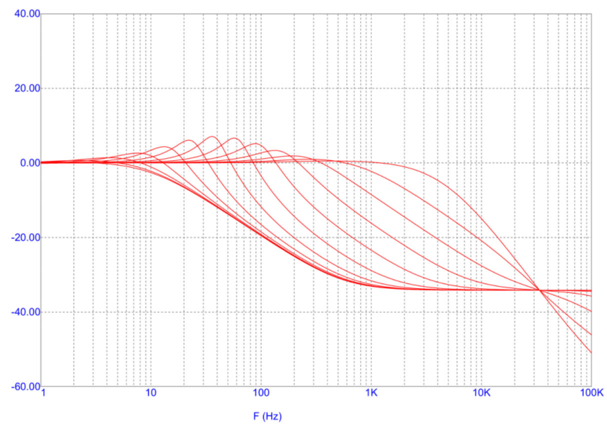

Fig. 14. Simulated frequency response of the VCF when the control voltage between from 1.2 V to 7.2 V.

$$I_D = I_S(e^{\frac{V_D}{nV_T}} - 1) \tag{1}$$

In this equation, I_D is the current across a diode, I_S is its saturation current, V_D is the voltage across it, n is the emission coefficient, and V_T is the thermal voltage. The purpose of this circuit is to modify a signal in the frequency domain, meaning that its small signal model should be used to better understand how it works. This model is shown in figure 15.

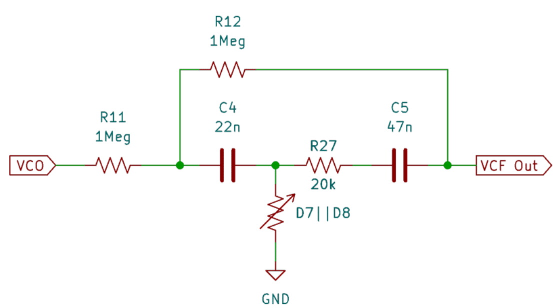

Fig. 15. Voltage-controlled filter small signal circuit diagram.

In the small signal model, D7 and D8 can be modelled as resistors in parallel. They are depicted as a variable resistor because their resistance (r_D) depends on the current flowing through them. This relationship is shown in equation 2.

$$r_D = \frac{V_T}{I_D} \tag{2}$$

As the voltage from the EG increases, the current through the diodes also increases, resulting in their resistance decreasing. The cutoff frequency of a circuit is dependent on the values of the capacitors in the circuit and resistances relative to those capacitors. Equation 3 shows the relationship between the cutoff frequency of a circuit and its components.

$$f_c = \frac{1}{2\pi \sum(R_i C_i)} \tag{3}$$

Because resistance varies inversely with the cutoff frequency, it can be concluded that increasing the control voltage increases the cutoff frequency, which is why the VCF is able to function.

3.6 Voltage-Controlled Amplifier

The VCA circuit consists of a voltage divider and an LC Technology LM386 20x Gain Amplifier Module. The diagram of the VCA is shown in figure 16.

The voltage divider attenuates the signal with a maximum gain of 0.5%. This value was chosen because the VCF output has a maximum amplitude which is significantly greater than the maximum amplitude supported by the devices the output will be sent to. Module J1 has a gain potentiometer, but its potentiometer range is wide enough that it is difficult to obtain a gain that is close enough to silent without being silent. Capacitors C2 and C3 are meant to serve the purpose of reducing noise, though their effectiveness is yet to be determined.

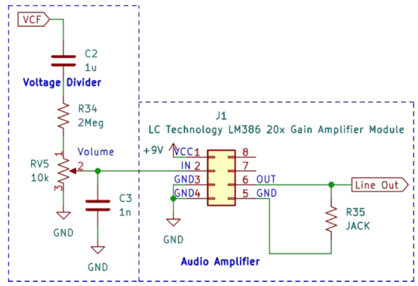

Fig. 16. Voltage-controlled amplifier circuit diagram.

This model of amplifier was chosen because it utilizes an LM386 integrated circuit. The LM386 is a commonly-used audio amplifier which is designed with a low output resistance, allowing it to drive a low resistance load.

4 Results

This section describes the measurements that were performed after the synthesizer was built. Two PCBs were designed in order to streamline the process of building, reduce the possibility of interference from nearby circuit elements, and fulfill the contract requirements. The first PCB contains the LFO, VCO, and VCF. Its schematic is shown in figure 32 in appendix C. The second PCB contains the EG and is shown in figure 33 in appendix C. The controller, the potentiometer, and the VCA were soldered onto perfboard. The buttons, potentiometers, and audio jack are large, meaning that it would be costly to manufacture PCBs which they could fit on.

The overall circuit draws a current of ∼48 mA. The capacity of a 9 V battery varies, but it can be estimated as 300 mAh. This means that this synthesizer has a predicted battery life of ∼6 hours.

In order to keep from draining batteries, a Digilent Analog Discovery was used as a power source in measurements. Its supply outputs were set such that the positive output had a voltage of +4.2V and the negative output had a voltage of -4.2V. All the oscilloscope measurements were taken with the negative probes at the circuit's ground.

4.1 Controller

In order to visualize how the sample-and-hold circuit responds to changes in the input voltage, one waveform generator was used to create a 10 Hz sine wave at the main input and another was used to create a 50 Hz square wave at the gate input. In addition, one set of probes was used with the oscilloscope to measure the sine input and another was used to measure the output of the circuit. Figure 17 shows the result of this setup.

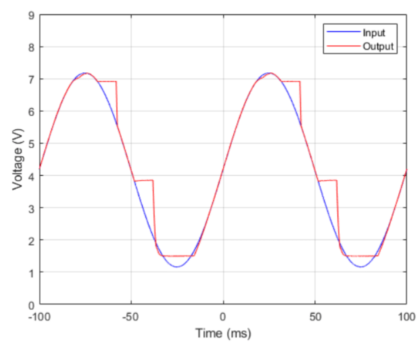

Fig. 17. Measured output of the sample-and-hold circuit with a 10Hz sine wave input and a 50Hz square wave gate input.

When the gate input is low, the output follows the main input. The exception to this was when the main input exceeds the voltage swing of the op-amps. When the gate input is high, however, the voltage remains mostly constant. However, this only occurs when input is less than the held voltage. It is also worth noting while the larger held voltage remains constant, the smaller one shows a gradual increase over time. When the sample-and-hold circuit is used with the synthesizer, the held voltage remains constant for the most part, but that held voltage deviates from the previous voltage significantly enough to be a different pitch.

4.2 Low Frequency Oscillator

The triangle output of the LFO was measured by connecting the positive probe of the oscilloscope to connector J4 on the first PCB. Figures 18 and 19 show the triangle output of the LFO when the LFO frequency potentiometer is at its maximum and minimum settings respectively.

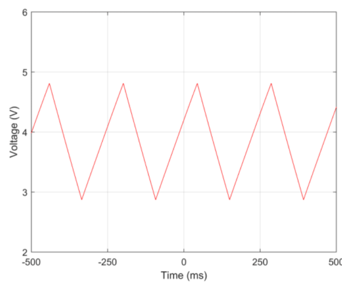

Fig. 18. The triangle output of the LFO when the frequency potentiometer is at its maximum.

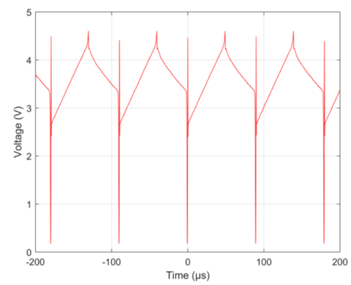

Fig. 19. The triangle output of the LFO when the frequency potentiometer is at its minimum.

The LFO has a frequency range of between 4.13Hz and 22.3kHz. It was designed such that the rise and fall times of the wave would be symmetrical. However, the measured wave has a different rise time compared to its fall time. When at its lowest frequency, the rise time is 52.3 ms and the fall time is 40.5 ms. This means that the fall time has a 22.6% error when compared with the rise time. This does not have a noticeable impact on the LFO's ability to function as vibrato.

At higher frequencies, the LFO fails to produce a triangle wave. Instead, there are small spikes which form at the peak of the wave and large spikes at the troughs of the wave. At points when the wave becomes this distorted, the frequency is either at the edge of or beyond the range of human hearing. This frequency is unlikely to have a purpose, so this anomaly does not influence the synthesizer's usability.

4.2.1 Triangle to Sine Approximation

The sinusoidal output of the LFO was measured by connecting the positive probe of the oscilloscope to connector J7 on the first PCB. The LFO amplitude potentiometer was set to its maximum value. Figures 20 and 21 show the measured output when the LFO frequency potentiometer is at its maximum and minimum settings respectively.

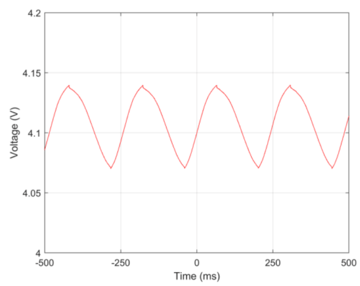

Fig. 20. The output of the triangle-to-sine approximator when the LFO is functioning at the minimum frequency.

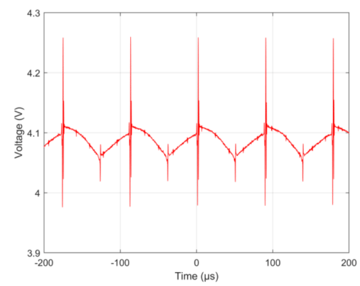

Fig. 21. The output of the triangle-to-sine approximator when the LFO is functioning at the maximum frequency.

At the minimum frequency, the output of the LFO looks more rounded compared to a triangle wave. However, this shape is flawed. At the point where the triangle wave would shift from rising to falling, the approximated sine wave has a sudden decrease in voltage. This shift is not audible. In addition, the valleys are less rounded when compared with the peaks. At this frequency, the peak amplitude is 34.8 mV. At the maximum frequency, the sine approximation retains the same distortion as the triangle wave.

4.3 Voltage-Controlled Oscillator

The output of the voltage-controlled oscillator was recorded by measuring the voltage at connector J12 on the first PCB. While this output was recorded, the rightmost button on the keyboard was held. Figure 22 shows the frequency of the VCO when the master pitch potentiometer is at its minimum setting. Figure 23 shows the frequency of the VCO when the master pitch potentiometer is at its maximum setting.

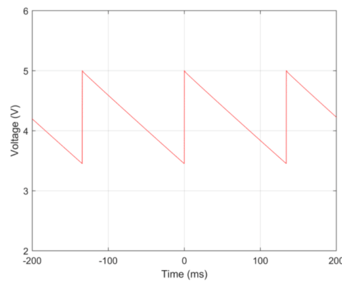

Fig. 22. The output of the VCO when the minimum control voltage is used as the input.

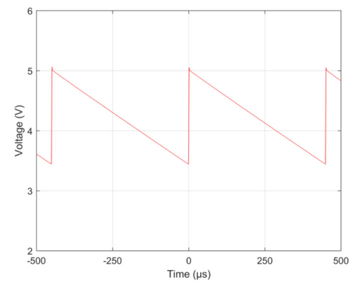

Fig. 23. The output of the VCO when the maximum control voltage is used as the input.

When the rightmost button on the keyboard is held, the VCO has a frequency range of between 7.46 Hz and 2.22 kHz. Holding other buttons on the keyboard can reduce both values of the frequency range, meaning that the total range of the VCO has a smaller minimum. Unlike the LFO, the shape of the output remains consistent regardless of the frequency.

4.4 Envelope Generator

In order to verify if the envelope generator met the contract specifications, its output was measured both when the attack, decay, and release stages were at their minimum and maximum durations. These measurements were taken when the potentiometers were set such that the initial voltage was at its minimum and the peak voltage was at its maximum. This was done in order to ensure that the voltage varies the maximum amount, giving results for what may be considered the default level settings. The switch which toggled retriggering was turned off. The output of the circuit was measured by connecting the positive probe of the oscilloscope to connector J8 on the second PCB. For the minimum duration measurement, the rate potentiometers were set so that the resistance across each potentiometer was at its minimum. For the first measurement, the sustain voltage was set to its minimum and a key on the keyboard was held. This resulted in the EG's attack stage, in which it transitioned from the initial level to the peak level, followed by its decay stage, in which it transitioned from the peak level back to the sustain level. These results are shown in figure 24.

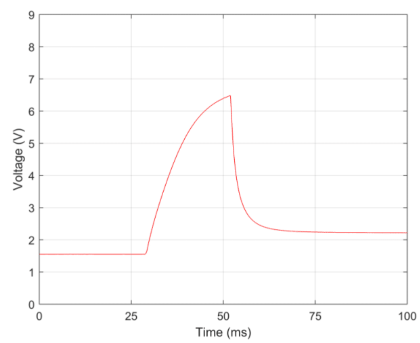

Fig. 24. Attack and decay stages of the envelope generator at the minimum durations.

The duration of attack was measured by finding the time between the two sudden changes in velocity which characterize the start and end of the attack stage. From this measurement, it was found that the attack has a minimum duration of 22.9 ms. The end of the decay stage could not be measured the same way because the stage does not end until all the keys are released. Instead, the decay duration was determined to be the length of time between the end of the attack stage and when the voltage reached its final value plus 10% of the difference between the initial and final value. Through this method, it was determined that the decay stage has a minimum duration of 5.35 ms.

Next, the minimum release duration was measured. This was done by setting the sustain to its maximum value, then holding a key until the voltage plateaued at the sustain value. Then, the key was released, triggering the release stage. The results of this are shown in figure 25.

Much like the decay stage, the release stage also gradually reaches a plateau. The duration of the release stage was measured by measuring the amount of time between when the key was released and when then the voltage reached its final value plus 10% of the difference between its starting and ending values. It was determined that release stage has a minimum duration of 520 ms.

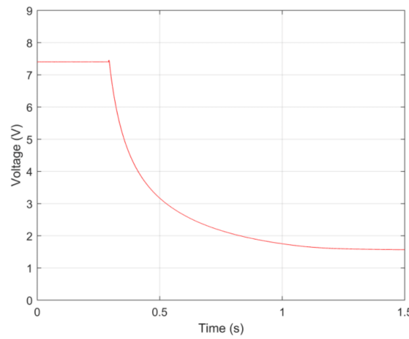

Fig. 25. Release stage of the envelope generator at the minimum duration.

The maximum durations for each stage were found by setting the rate potentiometers such that there was the maximum resistance across each of them. The sustain level was set to be in the middle between its minimum and maximum value. As the output was recorded, one of the keys was held until then voltage began to plateau during the decay stage. Then, the key was released, triggering the release stage. This output is shown in figure 26.

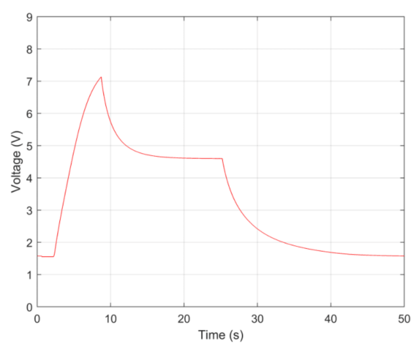

Fig. 26. Attack, decay, and release stages of the envelope generator at the maximum durations.

The maximum attack, decay, and release durations were measured the same way as the minimum equivalents. The attack stage was found to have a maximum duration of 6.48 seconds. The decay stage was found to have a maximum duration of 4.48 seconds. The release stage was found to have a maximum duration of 10.2 seconds.

4.5 Voltage-Controlled Filter

The output of the VCF was measured by connecting the positive probe of the oscilloscope to connector J3 on the first PCB. The VCO was set such that the frequency was ∼440 Hz. Figures 27 and 28 show the output of the VCF when the EG's initial level is at its minimum and maximum value respectively. They correspond with the minimum and maximum cutoff frequency of the VCF.

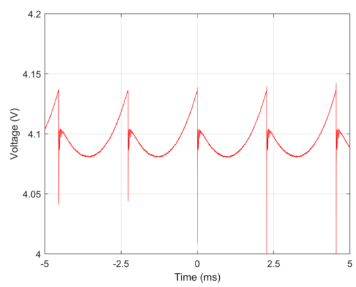

Fig. 27. The output of the VCF at the minimum cutoff frequency.

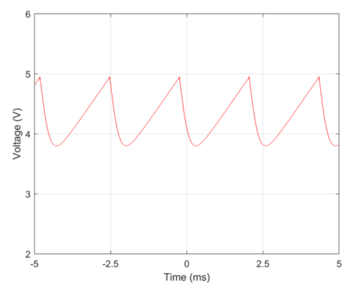

Fig. 28. The output of the VCF at the maximum cutoff frequency.

At the minimum cutoff frequency, the VCF has a peak amplitude 74.4 mV. At the maximum cutoff frequency, it has a peak amplitude of 581 mV. This means that the minimum amplitude is 12.8% of the maximum amplitude, or -17.9 dB.

4.6 Voltage-Controlled Amplifier

In order to compare the shape of the output of the VCF to that of the VCA, one positive oscilloscope probe was connected to connector J3 on the first PCB. A second set of oscilloscope probes was connected at the OUT and GND outputs of the gain amplifier module. The VCO was set so that its frequency was ∼440 Hz and the EG's initial level amplifier was set to its maximum setting. Figure 29 shows the VCF and VCA outputs.

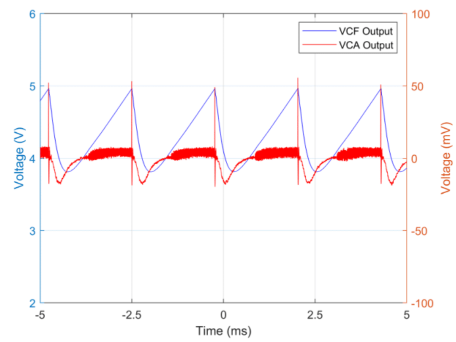

Fig. 29. Comparison of measured VCF and VCA outputs when the VCF is at its maximum cutoff frequency.

The output of the VCA is distorted compared to the output of the VCF. This distortion limits the maximum amplitude of the wave, but not the minimum. This influences of the timbre of the resulting sound. The overall sound is softer and comparable to a free-reed organ. Due to the lack of documentation regarding the audio amplifier integrated circuit used in the design of the VCA, it is unknown what the cause of this distortion is.

4.7 Potential Improvements

Due to the variety of flaws which this design has, a number of improvements could be made.

The controller output is dependent on the supply voltage. As a battery is in use, the voltage across it is constantly decreasing. This means that the pitch of the output is going to slowly decreasing while the synthesizer is in use. A potential solution to this would be to use a voltage regulator.

This would producing a voltage which would stay constant independently of changes in the supply voltage, though this would reduce the voltage range that the synthesizer can work with.

Some blocks of the synthesizer, including the EG, use op-amps in situations where a comparator would be ideal. Op-amps are flawed when used as comparators because they are designed to be linear devices. This can result in them consuming more power than normal or latching up. This can be detrimental to the performance of rail-to-rail op-amps. Bruce Carter discusses the misuse of op-amps as comparators further in his report on the subject [8].

Ideally, the VCO's output frequency would vary logarithmically with its control input. A common relationship is 1V/octave. In this synthesizer, this is not in the case. In addition, the output of the controller is dependent on the ratio between the resistance between the output and ground, and the resistance between the button and ground. This relationship is not linear, meaning that the ratio of frequencies between any two keys will not remain constant when the value of the master pitch potentiometer changes. This makes the synthesizer more difficult to use. A solution to this would be to use a current source in the controller so that each additional resistor adds to output voltage by the same amount.

The sample-and-hold circuit has an issue in which switching to the held state results in a noticeable change in frequency. One way this could be improved would be to incorporate a greater amount of feedback into the circuit. This could result in the gain of the circuit being smaller or greater than 1, so the summing amplifier of the VCO would have to be modified to account for that. Another solution would be to use a capacitor with a higher value.

The output of the synthesizer has a considerable amount of audible noise which reduces the quality of the output. This noise could be reduced through the addition of capacitors in areas which may be the source of the noise. One way of doing this would be to add one in parallel with the feedback path of an amplifier, causing it to function as a noise filter.

The triangle-to-sine approximator used in this circuit could be expanded upon in order to better refine the shape of the output of the LFO so that it better resembles a sine wave. Paul Falstad's triangle-to-sine converter [9] incorporates several of these approximators in parallel in order to create a rounded output which replicates a sine wave much more accurately.

The output of the VCA is distorted when compared with its input. A solution to this issue would be test if the issue is due to the voltage divider or the amplifier module. In addition, it might help for the voltage divider to have its negative end be at a +4.2 V node instead of ground.

The attack and decay durations of the EG are somewhat dependent on the state of the release potentiometer. If the release potentiometer is at its maximum value, it increases the duration of the attack and decay stages. A solution to this issue would be to measure the current flow through the release potentiometer and find ways to minimize it while the gate is positive.

5 Conclusion

Table I shows the minimum and maximum values measured at the output of multiple blocks. These values were used to determine whether or not the goals laid out in the contract were met.

TABLE I
Block minimum and maximum output parameter measurements.

Parameter	Minimum Value	Maximum Value
LFO Frequency	4.13 Hz	22.3 kHz
VCO Frequency	7.46 Hz	2.22 kHz
EG Attack Rate	22.9 ms	6.48 s
EG Decay Rate	5.35 ms	4.48 s
EG Release Rate	520 ms	10.2 s
VCF Amplitude	74.4 mV	581 mV

This project was successful in creating a portable analog synthesizer which could meet all of the specifications described in the project contract. The synthesizer was constructed using custom PCBs and without a development board. It is battery operated. It includes a keyboard and potentiometer controls which allow the user to influence various aspects of the output waveform. The voltage-controlled oscillator range exceeds the range specified by the contract. It has a low frequency oscillator which generates a sinusoidal wave. Each stage of the envelope generator has a maximum duration which exceeds 2 seconds. However, the synthesizer also contains several design flaws which keep it from achieving the same level of reliability and functionality as similar commercially available synths. There is potential for this design to be expanded upon in order to create an instrument as versatile and usable as ones currently worth hundreds to thousands of dollars.

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A Project Contract

Analog Synthesizer

Felix Morrissey (EE)

April 29, 2022

Description

A monophonic analog synthesizer will be designed, built, and tested. The device will use a Voltage-Controlled Oscillator (VCO) to create a signal which will be modified by a Voltage-Controlled Filter (VCF) and a Voltage-Controlled Amplifier (VCA) respectively. The device will have a set of buttons which will each set the frequency of the VCO signal. Pressing the buttons will also trigger an Envelope Generator (EG). The device will have a set of potentiometers which will control the parameters of the EG and Low Frequency Oscillator (LFO), which will in turn modify the VCF and VCO respectively. The gain of the VCA and the offset of the frequency of the signal produced by the VCO will each be controlled by a potentiometer. A port will be used to output the resulting signal. The device will be battery powered so that it can be used portably.

Inputs

- Energy from a battery
- Button controls
- Potentiometer controls

Outputs

- Synthesized signal through a 1/8th inch mono audio jack

Specifications

- VCO generates a sawtooth waveform
    - Frequency range from E2 to E6 or better
- LFO generates a sine waveform
- 25 buttons, corresponding to 2 octaves
- EG controlled by 6 parameters
    - Attack, decay, and release time range each between 0 and at least 2 seconds
- Operates from a battery
- Final project does not use a development board
- Design custom PCB

Fig. 30. Contract for the analog synthesizer capstone project.

35

B Parts List

Qty	P/N	Item	Price	Extended
1	Prototype	2 layer PCB	\$2.00	\$2.00
1	Prototype	2 layer PCB	\$4.00	\$4.00
1	3132-ST-PERF-1-8-ND	1” x 8” Perfboard	\$4.99	\$4.99
1	41-HG8X-N1JS	HANDSKIT Soldering Iron Kit	\$15.99	\$15.99
25	2449-NCA2-ND	KEY STYLE PB, SPST, MOM, BLACK A	\$0.80	\$19.88
4	PTV09A-4020FB103-ND	POT 10K OHM 1/20W CARBON LINEAR	\$0.98	\$3.92
1	PTV09A-4020FB104-ND	POT 100k OHM 1/20W CARBON LINEAR	\$0.98	\$0.98
8	PTV09A-4020FB105-ND	POT 1M OHM 1/20W CARBON LINEAR	\$0.98	\$7.84
4	2092-STX-3000-ND	CONN JACK STEREO 3.5MM TH R/A	\$0.80	\$3.20
2	2057-ASJ-99HR-HT-T/RCTND	AUDIO JACK	\$0.75	\$1.60
4	36-232-ND	IBATT CONN SNAP 9V 1 CEL 4” LEADS	\$0.55	\$2.20
5	296-1777-ND	IC OPAMP JFET 4 CIRCUIT 14DIP	\$0.84	\$4.20
3	MHSS1104	SWITCH SLIDE SPDT 300MA 6V	\$0.65	\$1.95
1	B096Z5DQWQ	AEDIKO 0805 Chip Resistor Assortment Kit	\$7.99	\$7.99
1	B06XD3K15S	MclglcM 0805 Capacitor Assortment Kit	\$7.99	\$7.99
Total				\$92.25

C Project Schematic

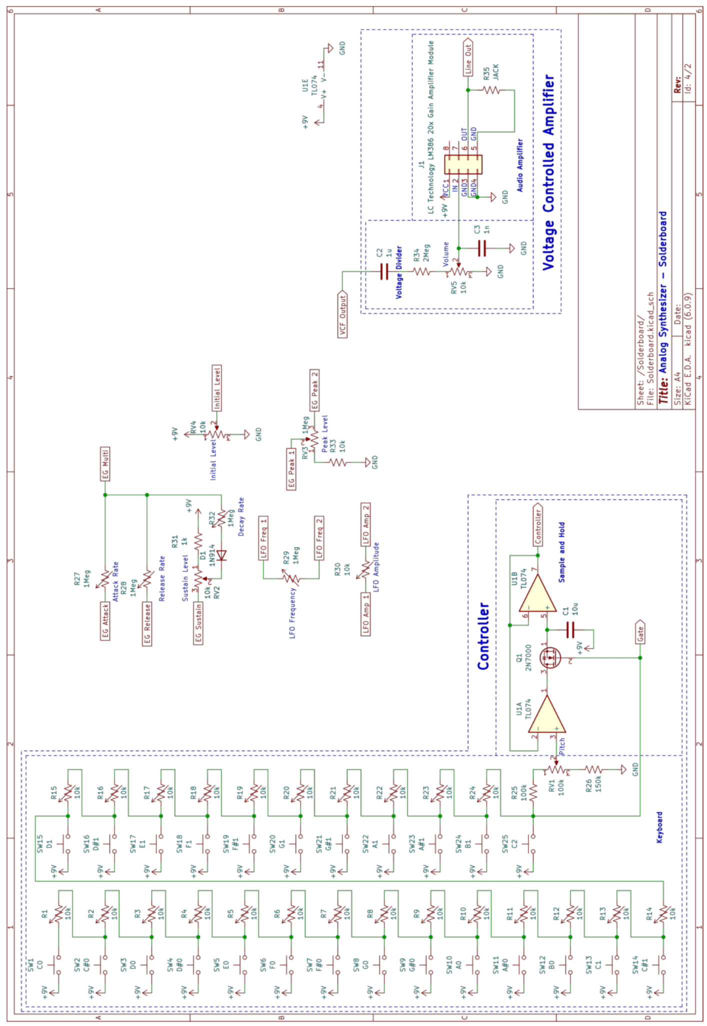

Fig. 31. Schematic for the portion of the synthesizer on perfboard.

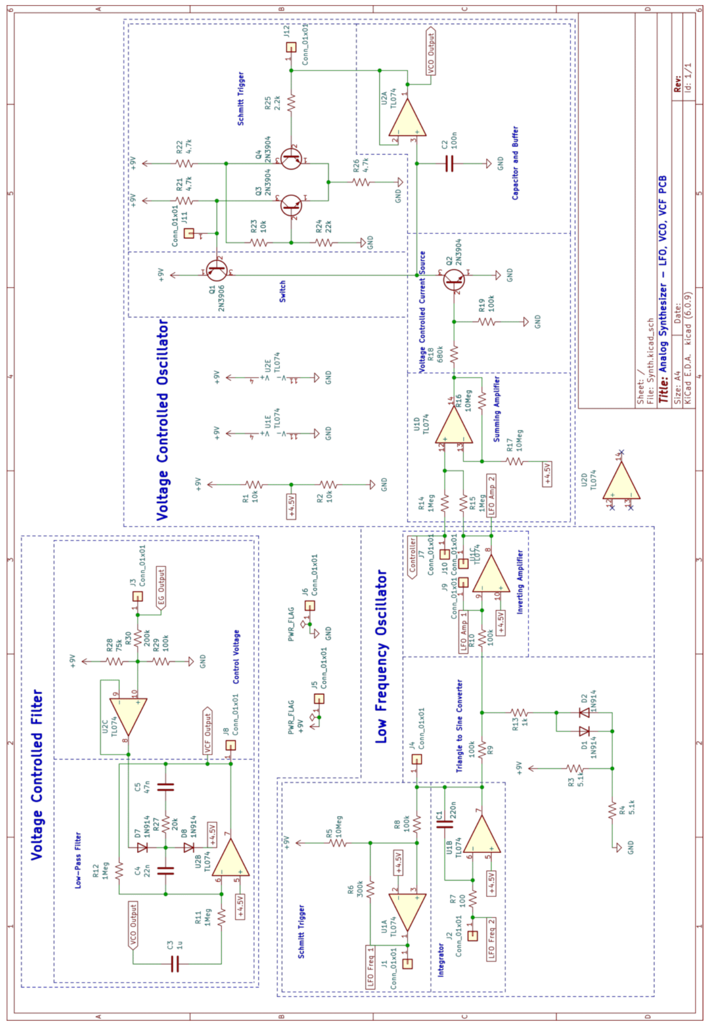

Fig. 32. Schematic for the portion of the synthesizer on the first PCB.

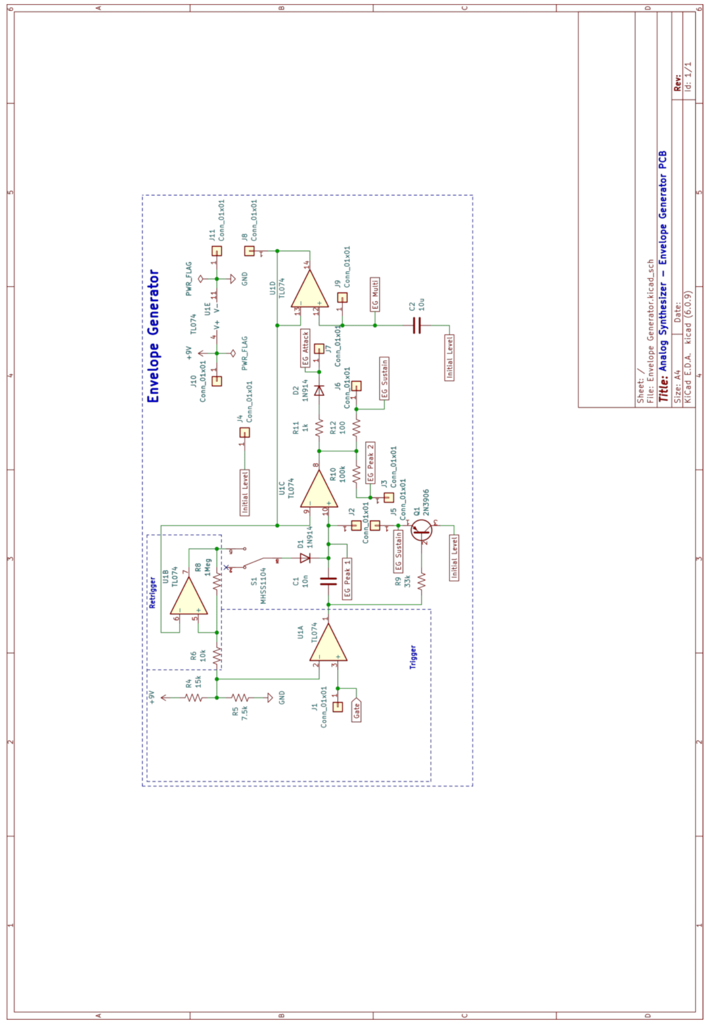

Fig. 33. Schematic for the portion of the synthesizer on the second PCB.

AUTHOR'S BIOGRAPHY

Felix Morrissey was born in Portland, Maine on November 14, 2000. Due to his parents' love of music and his father's hobby as a musician, he was raised with a deep appreciation for music and the process of artistic creation. Between fifth and twelfth grade he played trombone in his schools' bands, including jazz band and jazz combo. He also played violin in Portland's high school orchestra for four years. In addition to musical performance, Felix is a hobbyist in music composition and production, as well as programming, visual art creation, and game design.

Felix graduated from Portland High School in 2019, receiving multiple scholarships, including their Class of 1914 scholarship and the KeyBank scholarship. He went on to attend the University of Maine, where he pursued a Bachelor of Science in Electrical Engineering. He also obtained a minor in studio art. Once he completes his degree, Felix plans on moving out of Maine and finding a job as an engineer.