Building an Analog Synthesizer

George Wieser

August 25, 2025

Acknowledgements

I give special thanks to my advisor, Charles Collett, for his constant support throughout this project. His willingness to take on a thesis that was separate from his own research was commendable and ultimately allowed me to explore a topic that I am truly interested in and passionate about. I also thank Brian Collett for providing me with resources that informed major aspects of this project. Finally, I thank the Hamilton College Physics Department for their support and encouragement throughout not only this project, but the past four years at Hamilton.

1 Abstract

The making of an analog synthesizer is an ideal interdisciplinary study of both music and physics, as it expands upon work done in Phys 245 while also introducing an element of personal interest when it comes to sound creation in music. As a music producer with a physics background, I hope to use my circuitry knowledge to help me create synthesized sounds that can ultimately be used in my own music. The primary focus of this project is the construction of a Dual Integrator Oscillator. The circuit was first constructed on a breadboard, and then soldered to a CPU board.

2 Introduction

The process of production started with researching synthesizers, specifically identifying that they are machines capable of producing oscillations using electricity. These signals can then be altered by the user with different effect modules. I decided on a fully analog approach, meaning circuits of my synthesizer operate solely on continuous signals as opposed to discrete, binary signals. The first and most critical component of the synthesizer I started with is the primary oscillator. The function of the primary oscillator is to generate waveforms, which is implied given that a synthesizer is a tonal instrument.

For the purposes of this project, the oscillator chosen was a Dual Integrator, a circuit that produces two sinusoidal waveforms and whose output frequency depends on the input voltage. The reason for this choice was informed by the variety in outputs, as one has a more distorted timbre compared to the other. Additionally, the relationship between output frequency and input voltage allows for a setup that can model a traditional keyboard instrument, as 12 different voltage triggers can be used to create the standard 12 notes found on a keyboard. The first iteration was constructed on a breadboard with the intention of troubleshooting and confirming the circuit functions as expected. From here, I transferred the components to a more compact board with permanent solder connections, also known as a CPU board. This will result in a compact, easily movable final product that can be easily paired with other effect modules. Although I was unable to build any additional modules, if given more time, I would have built a system capable of triggering twelve distinct frequencies with different voltages. Furthermore, having an additional voltage-controlled filter to further alter the waveform would have been a desired addition.

In this paper, I introduce background information on synthesizers and how the development of more efficient electrical components fueled their development. Then, the relevant theory needed to understand the physics behind how these circuits function will be outlined, first with the key components, followed by the complete circuit. This will be followed by the construction process of the Dual Integrator, using both the breadboard as well as the CPU board. I will close with the results of my builds and a discussion around what I would have done differently and where this project could go in the future.

3 Background

Understanding the definition and history of the synthesizer is key in order to discern which components are necessary to include in the end product. There are two main criteria that the instrument needs to follow in order to be considered a synthesizer: all sounds coming from the instrument need to be created solely by electrical signals which can be either digital or analog. From here, these sounds need to be able to be manipulated by the user through the use of different effect modules [10]. Further expanding upon the first definition, given that a synthesizer is a tonal instrument, i.e. the sound produced from it comes in the form of an oscillation with a distinct frequency, its primary function is to create tonal sound waves using electricity. This waveform is generated by an oscillator, the most fundamental component within a synthesizer for this project. It is quite common for a synthesizer to have an oscillator with up to four different waveforms such as sine waves, square waves with pulse width modulation[1], triangle waves, and sawtooth waves. The user of a synthesizer also needs to be able to change the timbre of the waves produced. This can be achieved with the use of modules, with volume envelopes, low frequency oscillators, and frequency filters being the most common. Unique combinations of both fundamental waveforms and different effects are what makes synthesizers stand out among others within the market.

The beginnings of the synthesizer date back to the early 20th century, with the Audion Piano being the first notable instance where sound was electronically generated with musical intentions.[10] Sound generation was achieved using vacuum tubes, invented in 1906 by Lee De Forest[6]. This was the first active electronic component, giving it the ability to amplify signals, ultimately leading to the development of radio and telephones. Amplification was achieved through thermionic emission, where electron flow could be varied using two electrodes, a cathode, and an anode.

[1]The ability to change the width of each peak of a square wave.

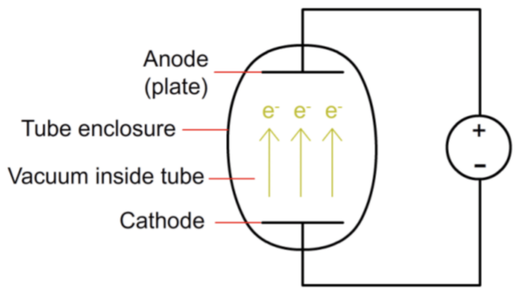

Figure 1: A basic schematic of a vacuum tube, where the electron flow between anode and cathode can be observed.[3]

By heating the cathode and applying a negative voltage to the cathode, electrons are displaced and move toward the cathode due to the positive voltage. By varying these two factors, changes in voltage could be achieved during certain time intervals, allowing both amplification and oscillation to take place[3]. These components were an industry standard for electronics as well as the first attempts at creating electronic instruments during the first half of the 20th century[3].

Figure 2: A sketch of the Audion Piano in action. Due to the components available at the time, the instrument needed to take up a large amount of space.[6]

Although the Audion Piano was an instrument which produced tones created from electricity, if the formal definition of a synthesizer is considered in this context, these early electronic instruments could not formally be considered synthesizers, as the introduction of effects such as filters, envelopes, reverb, etc. were not yet incorporated. The first instance of this occured in 1937, with the Warbo Formant Orgel built by Harald Bode and Christian Warnke[10], as their instrument allowed the user to modulate filter parameters. However, this single effect parameter was not enough for this to be considered the first synthesizer.

Things began to change with the introduction of the transistor in 1947. Similar to vacuum tubes, transistors have the ability to increase the energy in a signal by controlling currents between their three terminals, as seen in Figure 3.

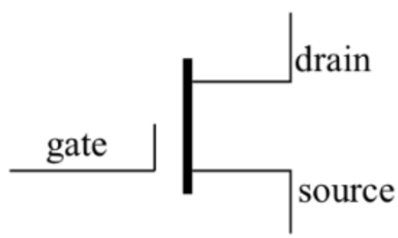

Figure 3: The general layout of a transistor, which shows its three terminals. Voltages between gate and source terminals will change the current flowing between the drain and source. [4]

This device allows voltages to be actively varied. This is achieved by applying a voltage between the gate and source terminals, which alters the internal resistance of the device, ultimately changing the current flow between the drain and source terminals[4]. This opening can be fully closed, opened, or anywhere in between depending on the circuit. This system can be better understood by thinking of it as a plumbing system, as seen in Figure 4.

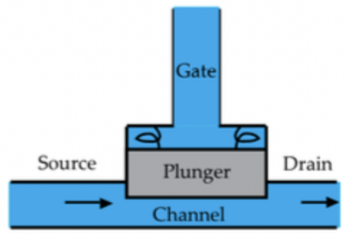

Figure 4: The water model of a transistor. [4]

By applying more water pressure on the plunger, or the gate, this constricts water flow between the source and drain[4]. Applying less pressure on the gate will result in more water flow between the source and drain. By replacing water pressure with voltage and water flow with current, this system can be used to accurately model the behaviour of a transistor.

Transistors were smaller, cheaper, and more reliable than vacuum tubes, making them a superior component compared to vacuum tubes[2]. When considering transistors in the context of electronic instruments, their introduction into modern electronics was crucial for their development both sonically and commercially. Robert Moog, a renowned engineer and pioneer within the electronic music world, was a key figure in harnessing the power of transistors in a musical context. His first prototype was created in October of 1964, where he collaborated with Herbert Deutsch to create one of the first modular synthesizers. This was a large scale instrument operated using patch cords. by plugging these cords into different modules, it connects them to already active components, allowing for their effect to be audible in the output[10].

Figure 5: One of Moog's modular synthesizers. The inputs and outputs used for patch cords can be seen all throughout the surface of the instrument.[2]

Although sonically impressive, modular synthesizers were inconvenient in a live performance setting given their size and unconventional user interface, which prompted Moog to create the Minimoog, a more compact keyboard instrument which incorporated all the formal elements outlined in the primary definition. By 1970, the Minimoog was commercially available, revolutionizing the sonic language being used by the world's most popular artists. Well known companies such as Yamaha and Roland entered the market and began creating synthesizers of their own, expanding and popularizing the instrument in the music industry[2].

Synthesizers such as the Minimoog used analog components, meaning that all circuitry within the breadboards were run using continuous voltages. In 1978, Sequential Circuits produced a groundbreaking instrument known as the Prophet 5[2], the first synthesizer to harness digital components, making it the first hybrid synthesizer. Unlike analog components, these new chips utilized discrete signals, or more simply, 0s and 1s to operate. The first application of this new technology was used for saving presets in a memory bank, meaning that certain combinations of settings within the effects and waveforms could now be accessed with a touch of a button, allowing the user to save and access sounds. The application of digital components within synthesizers would then grow into CPU units within the sound making process, as having digital components as opposed to pure analog allowed for simpler and more compact circuits within the instruments. An example of this is the master oscillator within the Roland Juno 106[5]. A master oscillator, different from a primary oscillator, is a single oscillator which produces waveforms with frequencies in the MHZ range. This wave is then divided into audible frequencies using a digital frequency divider, with no change to the actual waveform itself besides the change in frequency. By triggering a key on a keyboard, the master oscillator is activated followed by the divider, where the output is ultimately the frequency of the key pressed. This is the setup used in the Juno 106, which was released in 1984, which gives an idea of when these components were first introduced.

The influence of digital components continued to change within the 21st century with the introduction of digital instruments and plugins. Music creation has become accessible to a wider range of people due to digital audio workstations such as Logic, FL Studio, Ableton, and Pro Tools being available for the general public. Electronic instruments are code based and incorporate all the sonic elements that exist within the analog version, allowing for users to create sounds without any analog gear. This distinction becomes relevant when considering the synthesizer being built, as the same instrument could be created using fully analog, fully digital, or a combination of the two resources. With this in mind, a completely analog approach was decided. Although this was a personal aesthetic choice, it also allowed for a more in-depth understanding of foundational topics established in Electronics and Computers 245.

4 Theory

The basis for the entire synthesizer is the primary oscillator system, that being a Dual Integrator with two sinusoidal outputs. Discussion around this circuit will be provided in great detail later in the paper; this section will outline details surrounding the most important components. This circuit operates on several basic components such as resistors, capacitors, a single transistor, and diodes. The main functionality is reflected in the five op-amps within the system, as their fundamental actions, such as feedback and amplification, are present within the Dual Integrator.

4.1 LF 356 / 741 op amp

Created in the early 1970s, this chip was originally designed to perform mathematical operations such as addition, subtraction, multiplication, and division among others, but was quickly recognized for its ability to amplify signals and was ultimately integrated into many low-frequency electronic circuits[4]. Operational amplifiers, as the name suggests, are able to take any input voltage and amplify the output. They function as a very high gain DC coupled amplifier with two reverse sign inputs, with DC coupling meaning that both AC and DC currents can pass through. This allows the output to have a reverse sign compared to the input, with the difference depending on the strength of the input signal. These chips are powered with +/- 15 volts, with specified pins being available in Figure 6. It is recommended to ground the power pins with 0.1 microfarad capacitors to give the +/- 15 volts a reference point.

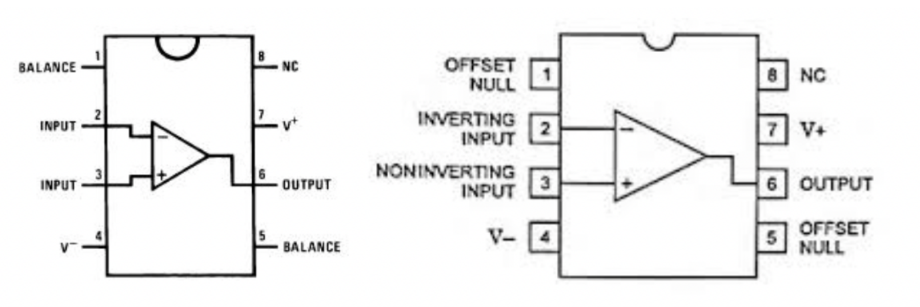

Figure 6: The pins of both a standard LF356 (left) and 741 (right). Terminals are practically identical, with 4 and 7 acting as terminals for powering the device using +/- 15 volts respectively. 2 acts as the inverting input with 3 as the non-inverting input, and 6 being the output. All other terminals are not used within the scope of this project.[1][11]

A popular method used to improve op amps in a variety of ways is negative feedback, a technique where a portion of the output is subtracted from the input. Systems which incorporate negative feedback have a slight reduction in overall gain, but benefit from more stability[4]. Circuits with negative feedback obey a set of laws called “the golden rules”, which are as follows:

1. The input draws no current
2. The inputs are at the same voltage

The Dual Integrator is a system that utilizes negative feedback, which means these rules are a helpful tool for analysis. Knowing that op amps try to get their inputs to be the same value can aid in understanding how signals behave within the circuit. In the context of synthesizers, where input voltages are often an AC signal, op amps are not reliable to reproduce a flawless, amplified output. Incorporating negative feedback is a common and useful workaround and can be observed in the Dual Integrator.

4.2 CA3080

The basis of amplification and frequency variation within the Dual Integrator is the CA3080. Although its functionality is quite similar to that of the LF356, a key difference is that it is an operational transconductance amplifier, whereas the LF356 is just a standard op amp. Op amps which use transconductance produce an output current that is a function of the input voltage[4]. This property can be harnessed using the amplifier bias input seen in Figure 7:

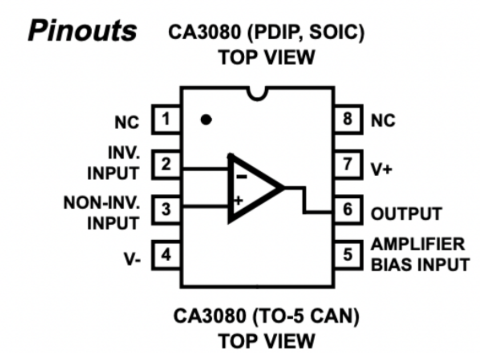

Figure 7: The pin diagram of the CA3080. The pins are nearly identical to the 356 and 741, however a main difference can be seen in the 5 pin, as it is being used for the amplifier bias input. [14]

The CA3080 allows for the main functionality of the Dual Integrator, that being the ability to vary frequency with different input voltages, using its amplifier bias input. This bias input generates high output impedance, or more specifically, allows the output to function more as a current as opposed to a voltage. This property allows the input voltage to vary the output frequency. The higher the input voltage, the higher the transconductance current coming from the CA3080, the faster the integrator capacitor, which will be explained in more detail in the next section, charges. All factors at play will affect the output frequency of the system, the CA3080 being a central component of this process.

5 The Oscillator: Theory Overview

There are many varieties of oscillators that can produce waveforms using analog components, with some approaches more useful than others. There are multiple valid methods in creating a usable oscillator, with systems varying in difficulty and practicality; a good example of a simple oscillator is the CMOS oscillator, such as the one shown in Figure 8. This circuit is a straightforward build, but difficulties arise in its application in a musical instrument context, with impracticalities appearing in the ways in which it controls tone frequency. A central element of the synthesizer is its ability to control the output frequency, as this is the basis for tonal harmony. The CMOS is an RCO, or resistance controlled oscillator, meaning frequency is resistance dependent as opposed to voltage dependent. This would require a separate oscillator to be constructed for each note and corresponding key.

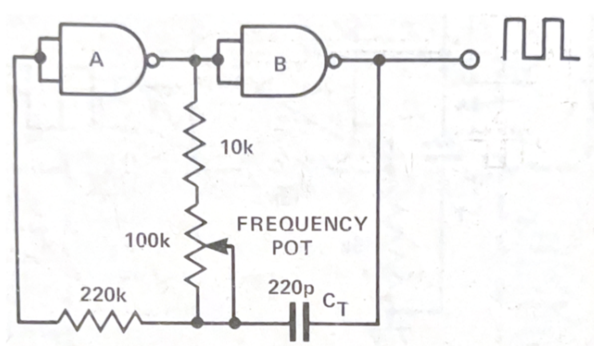

Figure 8: A schematic of the CMOS oscillator.[9]

By analyzing the functions of the CMOS gates within this circuit, a better understanding of its functionality can be achieved. If the output of B is high, then the input of A is also high since it is connected to capacitor Ct. If the output of A is low, Input B is then low and output of B is high. The switching of states occurs due to Ct being discharged through the 100k variable resistor and 10k resistor, resulting in a logic 0. Once the voltage for input A reaches its critical point, the output for B goes low and the capacitor is charged to the binary value of 1, and the process repeats itself creating a square-wave oscillation. Frequency is controlled through the 100k variable resistor, as it changes the rate at which the capacitor charges, ultimately dictating the rate of state change[9]. Although the circuit is reliable and creates a tonal output, it is using discrete binary states as opposed to continuous analog voltages. Additionally, this oscillator is only capable of producing a square wave, which limits the sonic potential of the instrument.

A more practical but involved oscillator is a VCO, or voltage controlled oscillator. The VCO chosen for this project is the Dual Integrator, with a detailed circuit diagram shown in Figure 9. This fully analog circuit is an ideal candidate for this project, as this system produces two outputs, one distorted and the other smoothed, ultimately providing a variety of timbres which is desirable in a musical context. Furthermore, frequencies are voltage dependent, allowing for just a single oscillator to be able to create multiple frequencies, as opposed to the CMOS, which would require a separate oscillator for each key. In the following section, each chip will appear in a detailed circuit diagram of the Dual Integrator, given in Figure 9. During construction, some of these components were replaced with usable alternatives due to convenience. These substitutions will be specified when relevant.

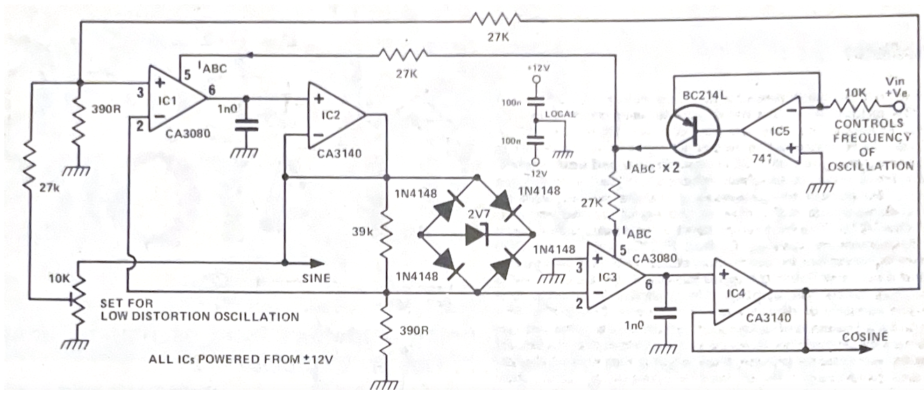

Figure 9: The Dual Integrator Quadrature VCO, a system capable of producing two sinusoidal outputs that are voltage controlled.[9]

The behavior of the system is a solution to the second order differential equation used to model a simple harmonic oscillator, as seen in Equation 1.

$$d^2x/dt^2 + w^2x = 0 \tag{1}$$

There is a damping factor of 0 within this equation, which corresponds to a beta value of 0. As mentioned in Sec. 4.1, these chips were originally used to solve mathematical equations. This behavior can be observed here, as there are two systems within the larger circuit performing integration. These smaller systems are located in two respective areas within the larger circuit, those being the connection between IC1 and IC2, as well as IC3 and IC4. These integrators are what allows for the opposite sine and cosine outputs, as each integrator produces a phase shift of 90 degrees.

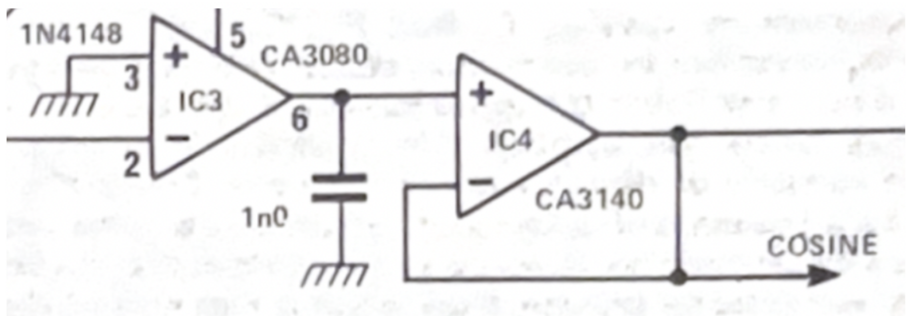

Figure 10: An integrator within the Dual Integrator circuit, whose behavior is identical to the corresponding integrator between IC1 and IC2.[9]

The input signal is a DC voltage which enters the inverting input on IC5. This area of the circuit is designed to supply the CA3080s with the transconductance current, labeled I_{abc} in Figure 9. IC5 is used as a unity gain buffer, allowing for the input voltage to remain stable and unaffected upon entering the BC214L. V_{in} is connected to the gate of the BC214L, as well as the drain through the output of the op amp. The output current, $2I_{abc}$, is supplied by the source, and split into I_{abc} using a 27k resistor divider. These output currents are then entered into the amplifier bias pins on the CA3080s.

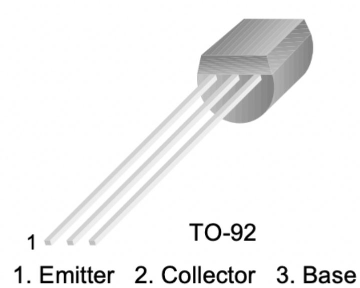

Figure 11: Pins on the BC214L. In the context of the previously stated transistor pins, Emitter corresponds to Source, Collector corresponds to Drain, and Base corresponds to Gate.[12]

This current, I_{abc}, is the value equal to the multiplication factor, omega squared, found in the simple harmonic oscillator equation. By varying V_{in}, I_{abc} will change and alter the resonant frequency as a result, and in this case, the oscillation frequency. This is how the input voltage can vary the output frequency, which is the entire basis for the practicality of this circuit in a musical sense. Next, one terminal of the 1nF capacitor is connected between the output of IC3 and the inverting input of IC4, while the other is grounded. The presence of the 1nF capacitor creates the integration, with the charge being equal to the integral of the current flowing into it, as seen in Equation 2.

$$Q = \int_0^t I(t)dt \tag{2}$$

The voltage across the capacitor is simply the output voltage of IC3, given that one end of the capacitor is connected to ground. With that being established, a relationship between V_{in} and V_{out} can be established, as seen in Equation 3:

$$V_{out} = (-1/R * C) * \int_0^t V_{in}dt \tag{3}$$

This equation models the behavior we would expect from the circuit, as the integrator performs a 90 degree phase shift on the sine wave input, thus producing the cosine wave output we observe in IC4. IC4, similarly to IC5, acts as a unity gain buffer with low output impedance, creating a stable waveform as an output. The first integrator, IC1 and IC2, has the same theory as that of IC3 and IC4.

Positive feedback is also present within this system, as the output of both IC4 and IC2 are connected to the non-inverting inputs of IC1 and IC3, allowing oscillations to build up and create a stronger output signal. This feedback is also controlled using the 10k variable resistor, with resistance being proportional to signal strength.

The final section of the Dual Integrator that requires analysis is the diode bridge which limits the amplitude of both outputs, as seen in Figure 12. A bridge of four 1N4148s and a single zener allows for frequency dependent feedback to occur. The placement of the zener within the bridge, both positive and negative signals will be limited and symmetrical. This allows for both outputs to have relatively similar amplitudes, although slight variations in amplitude are possible. Smoothing is only applied to the IC3 and IC4 stage, meaning that the cosine output will appear less distorted than the sine output.

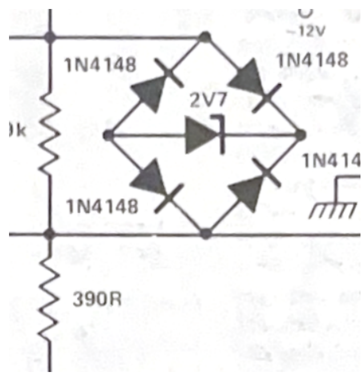

Figure 12: The zener diode amplitude limiter, with the 2V7 zener being responsible for limiting both signals.[9]

A drawback to such a system is that only one key can be played at a time, however monophonic synthesizers are quite common, especially for old fashioned analog synths. Separate signal paths need to be created for every additional voice being activated which is very difficult to achieve using only analog components[8], which is beyond the scope of this project.

6 The Oscillator: Construction Phase 1

The purpose of this section is to walk through a practical experience of the construction of the Dual Integrator. The construction process began by building the circuit in Figure 9 on a breadboard. The reason for this was to confirm that the outputs functioned as expected, and to streamline troubleshooting if needed due to the impermanence of a breadboard setup. The circuit was powered using a K and H ETS 7000 Digital-Analog Training System. This system provided all the power and ground supplies previously mentioned for the Dual Integrator. A Tektronix TBS 1052B digital oscilloscope was used to troubleshoot and measure voltages and oscillations. In order to hear the output of the system, a two terminal speaker was used on each output. The ultimate goal is to wire this circuit on a small scale CPU board, allowing me to easily transport the circuit, as the designer is not a practical way to power a musical instrument.

The initial construction consisted of laying out every op amp side by side, horizontally. This is the simplest way to power and ground each of the five op amps. A +15 volt rail was powered using the designer on the upper end, with a -15 volt rail being powered on the lower end. Ground rails were connected one unit above and below each respective power rail, as seen in Figure 13. This allowed for the shortest possible wiring route between op amps and their respective power sources.

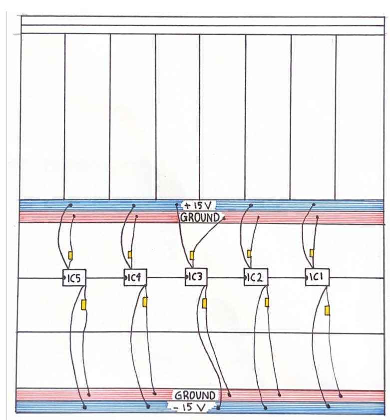

Figure 13: The layout chosen for the ICs. Each is connected to both a + and - 15 volt power source, with each power pin being grounded with a capacitor. Capacitors are represented with yellow boxes.

As mentioned before, it is important to use short wires and resistor series to ensure maximum signal stability. The format used for Figures 13 through 16 are not standard circuit diagrams, but rather sketches of what the circuit realistically looks like. This choice was made to demonstrate how I laid out my components with these things in mind, with the one caveat that I used resistor series which I do not recommend.

Once each op amp was powered, the oscilloscope was used to confirm that the pins on each op amp were receiving the correct voltages. A common malfunction within circuits are power and grounding issues, so checking pins at an early stage was an easy way to eliminate potential problems. In Figure 9, IC2 and IC4 are specified as CA3140s. Given that these chips were not present in the laboratory upon construction, and that their behavior is quite similar to LF356s which were available for use, IC2 and IC4 were replaced with LF356s. IC5 originally listed as a 741, which is practically indistinguishable from LF356s, so IC5 was also replaced by an LF356.

Now that each op amp was powered, the logical progression of setup was to build outward from Vin. This involves following the direction of current originating from IC5, then moving to each integrator, and finally ending at the diode bridge. The build of IC5 was straight forward, with the only notable difficulty being the orientation of the BC214L. Since each gate of the transistor has a very important function, it is critical to make sure that each terminal is aligned and in the correct position, as it is easy to orient the transistor in the opposite direction.

The next phase in building was to implement the 27k resistor divider. 27k resistors are not the most common, so an alternative approach that was initially attempted was to create a resistor series using one 20k, three 2k, and a single 1k resistor. Although mathematically equal to a 27k resistor, this setup can lead to signal instability, so reducing the number of resistors in series is crucial to creating a stable oscillator. Uncommon resistor values such as 27k and 39k appear multiple times within the circuit, and it is important to apply the same principle of minimal resistor series in every instance in which they appear. The use of short wires when possible is also important for the same reason.

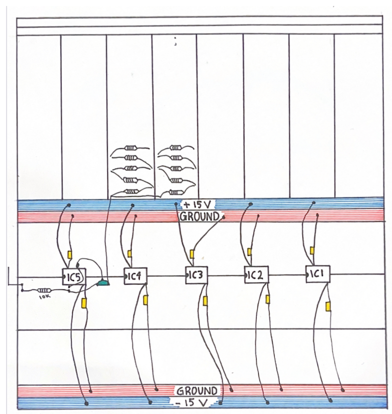

Figure 14: Vin, the IC5, and the 27k resistor series have been added to the breadboard. Although it is advised to use a single resistor as opposed to a series, the original build was the one chosen to be represented in the schematic. The B214L is represented in green to the right of IC5.

The circuit around the second integrator, IC3 and IC4 was built next. This decision was made as it follows the direction of the current, I_{abc}. The steps of setting up this section included creating the negative feedback loop between the inverting input and output of IC4, grounding the non-inverting input of IC3, and adding the 1nF capacitor which facilitates the integration process. The inverting input of IC1 and IC3 also need to be connected, as well as adding another 27k resistor between the output of IC3 and the non-inverting input of IC1.

The construction of the first integrator is largely the same as the second, with the only differences being the inclusion of a 390 ohm resistor, 27k resistor, and 10k variable resistor connected to the non-inverting input on IC1. It should be noted that a 370 ohm resistor was used in place of the 390 seen in Figure 9, as a resistor series of that value was impossible to construct given the resistors available in the lab.

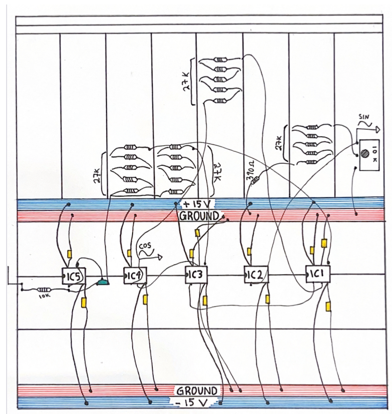

Figure 15: Each of the two integrators have now been wired with their components. Each of the sinusoidal outputs have also been labeled.

The final addition of the circuit is the amplitude-limiting diode bridge. Four 1N4148s are connected in the orientation seen in Figure 16. An easy mistake is to configure the 1N4148s in the incorrect direction, which would result in complete circuit failure. Any standard zener diode can be used to connect the bridge, and in this case a 1N4733 with a voltage rating of 5.1 was used[13]. A 40k and 370 ohm voltage divider was connected between the inverting input of IC1 and the output of IC2, with the 40k being constructed using a series of two 20k resistors.

At this point, the entire circuit is constructed. By activating power on the Digital-Analog Training System, oscillations can be observed using either the speaker or the oscilloscope, with the speaker being the ideal method since this is fundamentally an instrument, and it is important to hear the actual sounds made from the device. Grounding one speaker wire and connecting the other to either output will allow the waveforms to be audible. If the result is unsatisfactory, the 10k variable resistor may have to be adjusted to increase the amplitude of the output. I also had to decrease the +/- 15 volt power supplies to +/- 12 volts to reduce saturation within the output. The audio should be able to be heard once these parameters are tweaked to optimal levels.

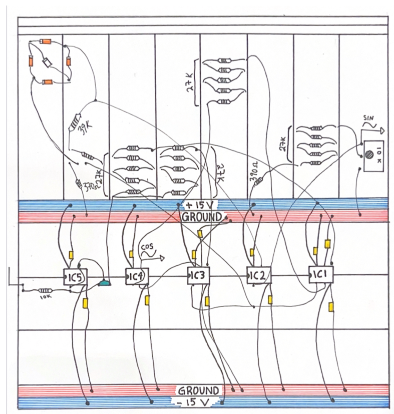

Figure 16: The inclusion of the diode amplitude limiter. Diodes are represented in orange in the top left of the schematic.

7 The Oscillator: Construction Phase 2

Although it is satisfying to be able to hear results from the previous building setup, it is not practical to have a single oscillator take up an entire breadboard of space, as commercial synthesizers have dozens of circuits that are equally complex as the Dual Integrator within it. In addition, the main purpose of the breadboard was to troubleshoot the original circuit, given the changes that were made to certain components. With this in mind, the Dual Integrator can be built on a CPU board that is a fraction of the size of the breadboard used in the first construction phase. The goal of this section is to walk through the necessary steps of soldering a circuit, as this technique is required to build circuits on CPU boards. It is assumed that the builder is familiar with the circuit being built since they constructed it in the previous section. That being said, the soldering process will be the main focus of this section.

Before the building process even begins, it is important to introduce the tools and basic principles used in soldering. Soldering is the process of connecting two pieces of conducting metal with a filler metal, with the intention of creating a permanent electrical and mechanical connection. The filler metal comes in both wire form as well as solder paste, as seen in the top half of Figure 17. Soldering wire can come with or without lead; I used the lead-free version on my CPU board. The soldering paste used was Chipquik SMD291SNL, which worked well in my process. The solder paste and wire serve identical purposes; which one to use at whatever point can be up to the discretion of the builder.

Possibly the most critical tool in soldering is the soldering iron, as seen in the bottom left of Figure 17. The tip can reach temperatures upwards of 700 degrees fahrenheit, and is responsible for melting the soldering material, ultimately bridging two separated areas of the CPU. It is easy for the tip of the soldering iron to corrode and oxidize, which can not only allow the formation of faulty electrical connections, but also prevent the metal itself from reaching the necessary temperature to melt the soldering material[7]. With this in mind, it is critical to regularly clean the tip of the soldering iron. The soldering iron I used is a Weller TC201T, which gave me no issues throughout the construction process.

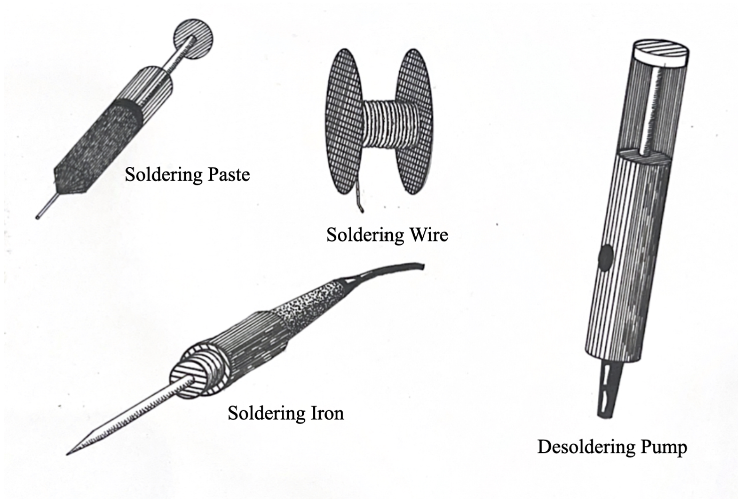

Figure 17: Sketches of Soldering Paste, Soldering Wire, Soldering Iron, and Desoldering Pump are layed out.

There are several methods that can be used to clean the soldering iron. The most straightforward solution is to scrape a sharp metal razor against the tip regularly during soldering, removing any oxidized material. Additionally, wiping the device against a wet sponge can also be an effective method for cleaning. The final technique involves melting a soldering material (ideally soldering paste) onto the tip (a process known as tinning)[7], and dipping the tip into a brass sponge. It is important not to be overly aggressive when removing soldering material from the tip, as hot metal could result in physical injury. For this project, a combination of the first two methods were used to maintain the cleanliness on the iron.

In the event that excess soldering paste forms an unwanted connection on the CPU board, a desoldering pump, seen on the right of Figure 17, can be used to remove the metal. By heating the unwanted solder using the iron, and positioning the desoldering pump directly above the liquid, the press of a button will suck away the excess material.

The wires I used are not the same as the ones used on the breadboard. The new wire was thinner and wrapped in a coil, allowing me to cut off specific lengths needed for different parts of the circuit, which was performed using wire cutters. The wire also had a plastic coating around it, which was removed at the tips using a wire stripper. Expose just enough of the inner conductor as necessary to make the solder connection, as excess exposed wire can cause shorting within the circuit.

The final material needed in this process are wire cutters. Once a component (such as a resistor) is attached to the CPU board, it is unnecessary to keep the entire length of the wires on either side, as this will clutter the underside of the board. By snipping the extra length after they are soldered to the board, it will allow for easier connections between other parts of the circuit.

The process of soldering a Dual Integrator onto a CPU board is extremely meticulous, as there are dozens of connections which need to be formed in a very small area. The first step is to create power rails for the +/- 12 volt power supplies and ground. This is done by melting connections between every port on both the top and bottom channels of the CPU board for the power supplies. The same is done for ground, but on the rightmost vertical channel, as seen in Figure 18. In retrospect, having one small rail for ground on one side of the CPU board isn't optimal, as there are many connections which end up spanning one side of the CPU board to the other. Additionally, it is imperative to make sure that the rails don't connect to any other channels, as this could apply power or ground to areas of the circuit which should not have them.

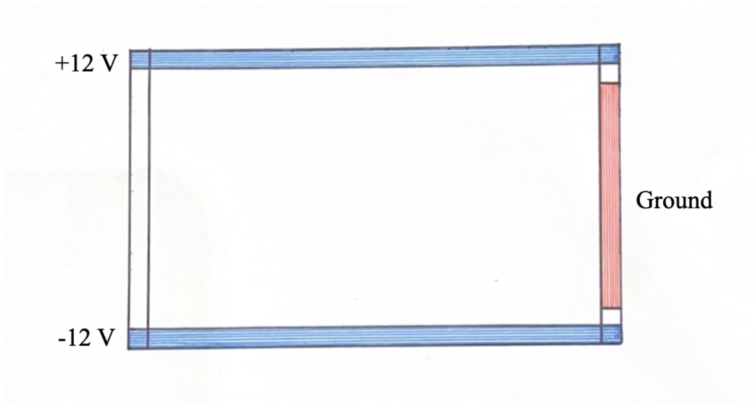

Figure 18: The placement of both power and ground on the CPU board, used primarily for the ICUs.

The Dual Integrator on the CPU board was assembled using the same procedure outlined in Section 6, and I will therefore not go into detail about the process of building the circuit itself. In Figure 18, a detailed layout of the completed CPU board can be seen. Connections which require wires should use the shortest connections possible, as long, looping wires can cause instability within the circuit.

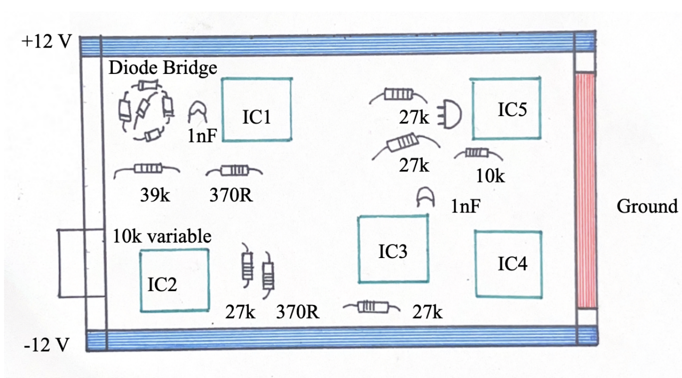

Figure 19: A schematic of the final CPU board layout. The general layout and locations of all major components are labeled. Wires as well as power and ground for the ICUs are not included.

8 Discussion and Conclusion

In this section, I will summarize my experience and acknowledge what I would do differently, followed by next steps if I were to continue this project. Constructing the Dual Integrator turned out to be an extremely time consuming and frustrating process, as it took three attempts in order to yield a successful output on the breadboard setup. The sine output was far more reliable than the cosine, as I only witnessed an audible signal a single time on the cosine output. Using shorter wires and less resistor series would have resulted in a much more useful end product.

Additionally, I had attempted, on two occasions, to build a voltage controlled amplifier, or VCA. The circuit was an even more difficult build than the Dual Integrator, due mostly to the eight variable resistors being used, and in retrospect, it was not the best use of time when considering a reasonable final product. A detailed circuit diagram of this amplifier can be seen in Figure 20. There are several easier solutions to signal amplification that would not have been nearly as time consuming, such as connecting the output to a speaker jack, or using an already functional CPU amplifier. A solid option for this is a pam8302a, as it is compact and has a high gain. Time spent on this circuit would have been better used constructing an effect circuit, such as a voltage controlled filter.

The next thing I would do is to create a mechanism that would allow for notes to be played from the Dual Integrator. This would consist of twelve separate buttons which activate voltages specific to tonal frequencies found on standard keyboards. Upon the activation of an individual button, a voltage would be sent to the input of IC5, producing the desired frequency within the Dual Integrator. This was the entire basis for choosing to construct this oscillator, and it would have been a practical way to demonstrate the relationship between input voltage and output frequency.

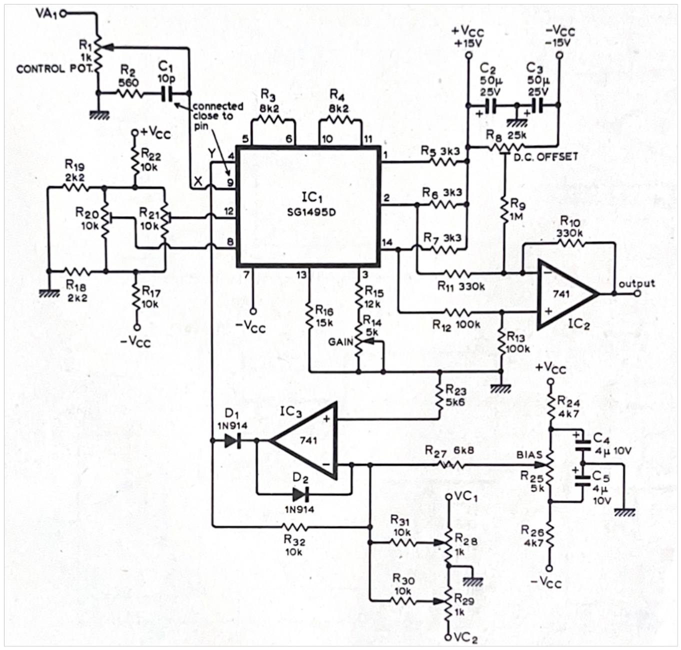

Figure 20: Circuit diagram of the Voltage Controlled Amplifier I attempted to build.[15]

The best way to have completed the synthesizer would have been the addition of an effect module, with the most desirable being a voltage controlled filter. This would have restricted certain wavelengths from passing through the output, allowing for shallower sounds. With the use of automation, or tweaking the frequency range over time, this would result in ways to change the timbre of the sounds in live time, which is a tool many musicians actively use.

The hope for this thesis is for it to act as a framework for future musicians with physicist backgrounds who are interested in the crossover between circuitry and sound creation in a music context. I hope that the resources and processes that I've discovered throughout the past four months will allow for one to quickly understand which areas of this project are important. Although I did not get to build every module I would have liked to, I feel as if I have a much deeper and meaningful understanding of the instrument that I use on a daily basis. This will ultimately allow for more informed decisions around my own creative process.

References

[1] Components 101. Lf356 jfet input op-amp ic. https://components101.com/ics/lf356-jfet-input-op-amp-ic-pinout-features-datasheet, 6/28/2021.

[2] Stewart Adam. May 17 1965, moog introduces the first analog synthesizer. https://creativeaudioworks.com/equipment/may-17-1965-moog-introduces-the-first-analog-synthesizer/, May 17 2012.

[3] Michael Alba. Vacuum tubes: The world before transistors. https://www.engineering.com/vacuum-tubes-the-world-before-transistors/, January 19, 2018.

[4] Brian Collett. Introduction to electronics and computers, vol, 1, 2024.

[5] Roland Corporation. Roland juno 106 service manual. https://www.synthxl.com/wp-content/uploads/2018/02/Roland-Juno-106-Service-Manual.pdf, July 31 1984.

[6] Simon Crab. 120 years of electronic music. https://120years.net/the-audion-pianolee-de-forestusa1915/, November 19, 2023.

[7] Will Donaldson. A comprehensive guide to soldering: Techniques, tools, and tips. https://www.instructables.com/A-Comprehensive-Guide-to-Soldering-Techniques-Tool/, 3/27/2024.

[8] Dave Erickson. Polysynth polyphonic analog synthesizer project. https://www.djerickson.com/polysynth/, 2/18/2023.

[9] Electronics Today International. Single board synthesizer, July 1978.

[10] Jerry Kovarsky. History of the synthesizer, part 1. https://hub.yamaha.com/keyboards/synthesizers/history-of-the-synthesizer-part-1/, April 4, 2022.

[11] Netsonic. Lm741 operational amplifier op-amp datasheet and pinout. https://netsonic.fi/en/lm741-op-amp-datasheet-and-pinout/, 7/14/2021.

[12] Fairchild Semiconductor. Bc214l pnp general purpose amplifier. https://www.silicon-ark.co.uk/datasheets/BC214L.pdf, October 2003.

[13] General Semiconductor. 1n4728 thru 1n4764 - zener diodes. https://www.futurlec.com/Diodes/1N4733.shtml, 12/3/1998.

[14] Harris Semiconductor. Ca3080, operational transconductance amplifier (ota). https://www.elde.cz/datasht/ca3080.pdf, May 1994.

[15] Wireless World. Electronic sound synthesizer: Part 2, September 1973.