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From Sine Waves to Soundscapes: Exploring the Art and Science of Analog Synthesizer Design

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For senior projects:

From Sine Waves to Soundscapes: Exploring the Art and Science of Analog Synthesizer Design

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by
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Abstract

Building an analog synthesizer from scratch can be a challenging yet rewarding endeavor that combines technical know-how with creative expression. This project aims to provide an overview of the process involved in building an analog synthesizer from scratch. The project begins by introducing fundamental concepts related to acoustic music. The science of analog synthesizer design is explored, including circuit theory, signal processing, and sound design. Basic components of analog synthesizers such as oscillators, filters, and amplifiers are discussed, followed by advanced topics such as modulation and envelopes. The project also provides practical guidance on circuit design, software selection, and troubleshooting common issues. Through this project, readers will gain an introductory understanding of the technical and creative aspects involved in analog synthesizer design.

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Dedication

To my father. I became like you, just like you did.

Acknowledgments

I would like to express my sincere gratitude to my advisor, Hal Haggard, for his unwavering support and encouragement during these challenging times. Harold's guidance has helped me to navigate through the difficulties and finally to think beyond the boundaries. His constant support and belief in me have been a source of motivation, allowing me to imagine and achieve the impossible.

I would like to express my sincere gratitude to John Dimarco and Thomas Mark for their invaluable assistance and support throughout this project. Their expertise, guidance, and encouragement have been instrumental in helping me gain a deeper understanding of constructing a synthesizer. Their kindness and patience have made a lasting impression on me.

I also would like to express my appreciation to Matthew Deady for his cookies during exams. I am grateful to him for igniting my curiosity in physics and encouraging me to pursue my goals. His passion and commitment as a professor have been truly inspiring.

I will miss chatting with Shuo Zhang after physics talks on Fridays, and Paul Cadden-Zimansky's wonderful classes as well as his sense of humor. I will miss Antonios Kontos for his generous help relating to PCBs in the project.

1
Introduction

Analog synthesizers are musical instruments that generate sound using analog circuits and signals. They are often used in electronic music production and have been popular since the 1970s. Known for their warm, organic sound, analog synthesizers' high degree of expressiveness and versatility made analog synthesizers a popular choice for electronic musicians, composers, and sound designers. The user can manipulate the sound in real-time using knobs, sliders, and switches. Some of the most famous analog synthesizer models include the Moog Minimoog, the Roland TB-303, and the ARP 2600, etc. Despite the rise of digital synthesizers, analog synthesizers continue to be widely used and appreciated for their unique sonic character and versatility.

In 2016, an unexpected encounter with a friend's beatboxing showcase sparked a new interest within me. I was fascinated by the rhythmic, percussive sounds, and realized that I was, as other teenagers of the same age, eager to express myself artistically. Shortly after this experience, I had the opportunity to meet some DJs who introduced me to the world of electronic music. During the DJ sessions I had with friends, someone mentioned the genre of techno, and I was immediately intrigued. As I delved deeper into this style of music, I discovered the world of modular DJ sets, which use analog equipment to create complex, layered soundscapes. Rather than simply using software to manipulate pre-recorded tracks, a modular DJ can physically

Figure 1.0.1: Minimoog was one of the most popular synthesizers ever built [1].

patch and connect modules in real time. It also allows me to create my own unique signal path, combining and manipulating different modules to create sounds. This was a new frontier for me, and I was eager to learn more. Captivated by the way others were able to manipulate sounds and beats using digital tools and effects, I was inspired to create my own music rather than simply playing others' tracks.

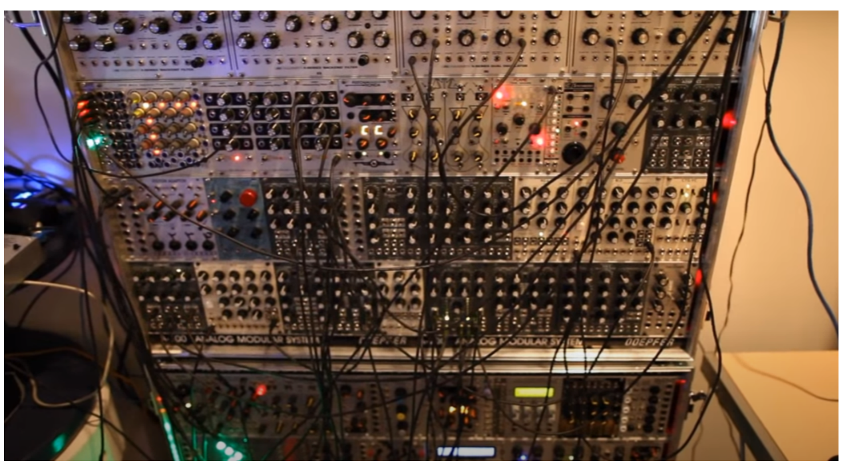


Figure 1.0.2: Deadmau5's impressive rack of synthesizer modules is a collection of various analog and digital synthesis gear used to create electronic music [2].

As I began to explore the possibilities of modular DJing, I found myself drawn to the challenge of creating my own unique sound. The process of patching together various modules, adjusting parameters, and tweaking effects was a labor of love. Each session was a new opportunity to experiment, explore, and create, and beyond that, I realized that this might be the best opportunity for me to combine my passion for physics with music. I eventually selected the topic for my senior project without previous exposure to circuits, and I am glad that I have made it this far.

2
The Science of Sound

The science of sound is the study of the physical properties and behavior of sound waves. Sound is a type of wave that travels through a medium, normally air, and can be perceived by the human ear. It is created by the vibration of an object, which sets the air molecules in motion and produces a pressure wave. In Samuel Pellman's *An Introduction to the Creation of Electroacoustic Music* (Pellman, 1981) [3], he gave a wonderful explanation of the physicality behind the sound in Chapter 1, so I would recommend this book to you if you want to explore more about the physics behind acoustic music.

If we imagine a string being plucked, it vibrates back and forth, creating a disturbance in the air around it. The motion of the string can be described as a series of waves traveling along its length. As the string vibrates, it pushes and pulls on the air molecules. The air molecules in its path vibrate back and forth, thus creating a sound wave, which is then passed to the eardrum and the brain of the listeners. Additionally, I want to point out that vibrating systems tend to produce sinusoidal waves due to their simplicity. I will further explore the magic of sinusoidal waves later as we investigate complex tones as combinations in the part where I explain the Fourier Transformation.

2.1 Pitch

The first property of sound I want to introduce is pitch. The pitch is related to the frequency of the sound. The unit of measurement is in hertz, abbreviated as Hz, named after Heinrich Hertz who discovered radio waves. A cycle of vibration refers to one complete back-and-forth motion, that is, the string vibrates from its initial equilibrium position to its maximum displacement in one direction, through its equilibrium again, to another maximum in the opposite direction, and back to its equilibrium again. The time elapsed during one cycle is called the period, denoted as P. Then, as frequency f refers to the number of cycles that happened in one second, we get this simple equation:

$$P = \frac{1}{f}. \tag{2.1.1}$$

2.2 Loudness

The study of sound encompasses a wide range of topics, including acoustics, psychoacoustics, and signal processing. This leads to the second property of sound, the loudness or the intensity of sound. As the string moves back and forth through the air, the amplitude of the vibration corresponds to how loud our ear perceives the sound. This direct relation is, counter-intuitively, nonlinear with how much energy the vibration contains. From a physics major's perspective, we tend to think that the intensity of sound refers to the amount of energy that is carried by sound waves per unit of area, and it is typically measured in units of watts per square meter ($\mathrm{W/m^2}$). However, nowadays we use the unit dB much more frequently as people have found how the human brain processes and interprets sound, as shown in Figure 2.0.1. From the graph, we see that the human ear is much more sensitive to changes in low-intensity sounds than to changes in high-intensity sounds. By using a logarithmic scale, very large and very small values can be expressed using manageable numbers, and the relative differences between values can be easily compared.

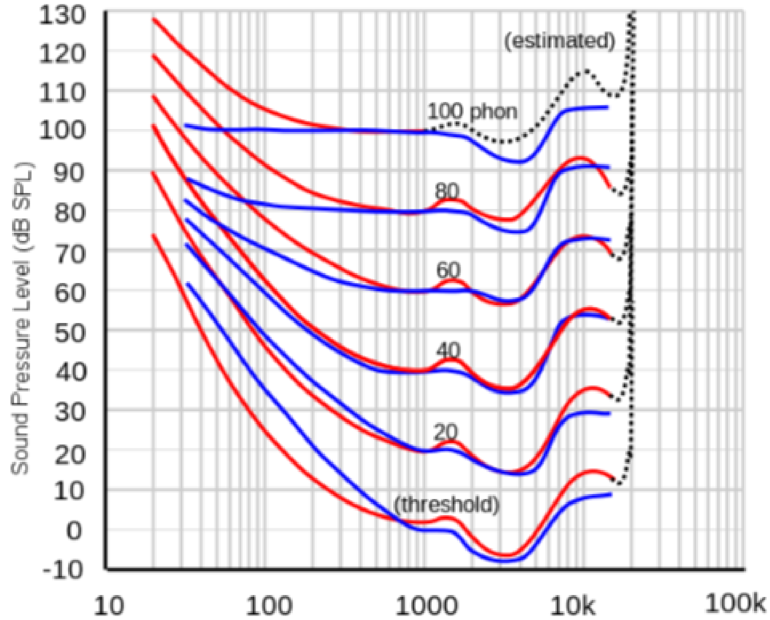


Figure 2.2.1: The equal-loudness curve shows the relationship between sound pressure level and perceived loudness at different frequencies [4].

Moreover, dB is also commonly used to express the relative power of electrical signals or the gain of amplifiers and other electronic components. Each increase of 10 dB represents a tenfold increase in sound intensity, that is, dB$= 10 \cdot \log_{10}\left(\frac{P_{\text{out}}}{P_{\text{in}}}\right)$, where P can either refer to the loudness or the power of the electrical signal.

Timbre is the characteristic quality of a sound that distinguishes it from other sounds of the same pitch and volume. For instance, it is not hard to distinguish a horn from a piano, even though the same note is played. This characteristic is sometimes referred to as the “color” or “tone color” of a sound. Timbre is determined by the harmonic content of a sound, which is the combination of its fundamental frequency (the main pitch or note) and its overtones or partials (additional frequencies that contribute to the sound’s overall character). The relative strengths and frequencies of these overtones give a sound its unique timbre. Hence, I want to introduce one last big concept before we move to the world of circuits, which is the idea of envelopes.

2.3 Envelopes

In analog and digital music, an envelope refers to a set of parameters that control how a sound changes over time. In particular, envelopes are used to shape the amplitude (volume) and/or frequency (pitch) of a sound over a specified period. The most common types of envelopes used in music are amplitude envelopes, which control the volume of a sound over time. Talking about envelopes, it is inevitable to mention John Chowning for his groundbreaking work on frequency modulation (FM) synthesis. In the early 1970s, he developed a new algorithm that allowed for the creation of sounds that were previously impossible to achieve with other synthesis methods for creating complex audio spectra. He explained how FM synthesis involves modulating the frequency of a carrier signal with a modulating signal to create new harmonic partials and how this method allows for the creation of rich, complex sounds that cannot be achieved through other synthesis methods. Therefore, the practical applications of FM synthesis were made possible, including the use in the creation of musical instruments and electronic music. He also described the intensity envelope of different instruments, which helped engineers to produce synthesizer sound which is close to their acoustic counterparts. Intensity or amplitude envelopes typically consist of four stages: attack, decay, sustain, and release (ADSR). During the attack stage, the volume rises from zero to its maximum level. During the decay stage, the volume decreases from its maximum level to the sustain level, which is a steady state volume that persists as long as the musician holds down the note. During the release stage, the volume decreases from the sustain level back down to zero. In my project, I am only building a two-stage AR envelope generator.

The science of sound also has other practical applications, such as in the design of concert halls and auditoriums, the development of hearing aids and cochlear implants, and the creation of noise-canceling headphones. There are also important applications in fields such as medicine recently, where ultrasound is used for diagnostic imaging and in the treatment of certain medical conditions. I would like to give a huge shout-out to the alumni who majored in physics and

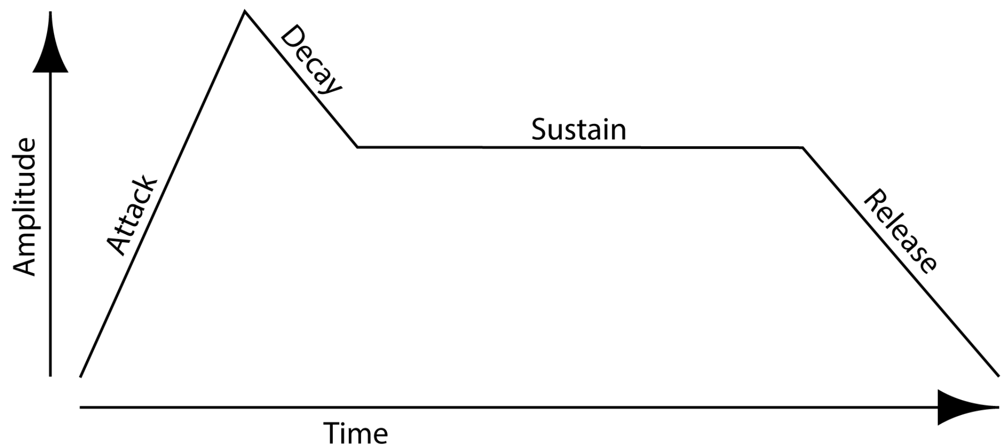
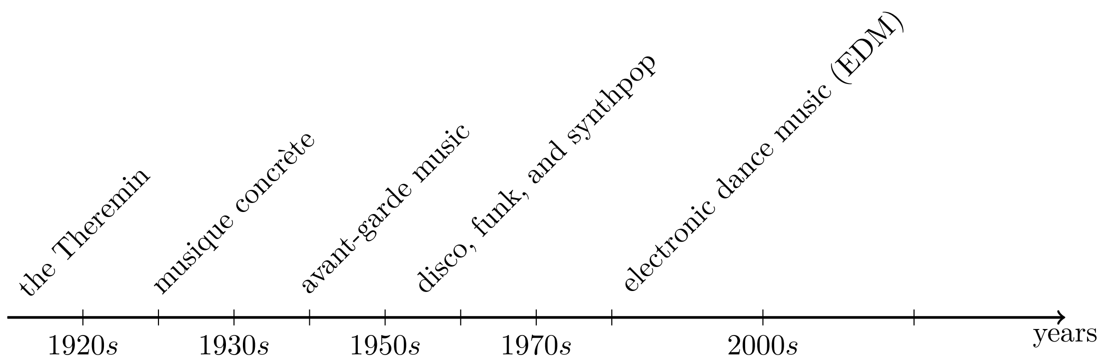


Figure 2.3.1: The four stages in an ADSR envelope are: Attack, Decay, Sustain, and Release [5].

wrote about acoustic and electric music in the past few years[6][7]. I am also grateful to Matthew Deady for guiding us in this direction. These resources are truly wonderful.

3
The History of Electronic Music



The history of electronic musical instruments can be traced back to the early 20th century, with significant developments and innovative ideas emerging since then[8][9]. To provide a manageable account of this history, this chapter focuses on electronic instruments that generate sound purely through analog and/or digital circuitry, magnetic tape, optical disc, software, or other electronic means, excluding electro-mechanical instruments. It is worth noting that many of these instruments have advanced the field of electronic music in significant ways, introducing novel capabilities and innovative concepts. Exploring this history is critical to understanding the wide variety of synthesizers available today. I will highlight some of the key breakthroughs and innovations that have made electronic music what it is nowadays.

There are five important categories for us to consider as we talk about these breakthroughs: control, sound, performance, interface, and composition, based on their intended or best use.

The Control category features instruments that offered new or advanced methods of sound control, such as the Voltage Controlled Oscillator (VCO) invented by Don Buchla in 1963. Sound highlights instruments that advanced the art of electronic sound generation, including the Fairlight CMI (Computer Musical Instrument) which used digital sampling to produce sound. Performance includes instruments optimal for live use, such as the Roland TB-303 Bassline Synthesizer which became popular in the 1980s for its distinctive acid house sound. Interface highlights synthesizers with innovative methods of interaction between the user and the device, such as the Haken Continuum Fingerboard which uses pressure-sensitive keys for expressive control. And Composition showcases synthesizers with extensively improved or extraordinary methods of creating music, such as the Yamaha DX7 which became popular in the 1980s for its digital synthesis capabilities.

The phonautograph was an early mechanical device invented by Édouard-Léon Scott de Martinville in 1857, and it is considered the earliest known device for recording sound. The device consisted of a vibrating diaphragm that was connected to a stylus, which in turn inscribed a visible trace onto a moving surface coated with soot, such as paper or glass. Although the phonautograph was not designed to play back the recorded sound, it allowed for visual analysis of the sound waves and was used primarily for scientific purposes, such as the study of speech patterns and the acoustic properties of musical instruments. The phonautograph served as a precursor to the phonograph, which was invented by Thomas Edison in 1877 and used a similar method of sound recording and playback.

In 1876 just before Thomas Edison invented the Phonograph, Elisha Gray invented the electromagnetic oscillator. An oscillator is an electronic device that generates a periodic signal, typically a sinusoidal wave or a square wave. Gray's oscillator was the first practical device of this kind, and it paved the way for many subsequent inventions, including radio transmitters, electronic musical instruments, and digital clock circuits. In fact, he used the oscillator to create an early prototype of a musical synthesizer, which he called the "musical telegraph". This device used a keyboard to control the frequency of the oscillator, allowing the user to play simple melodies. The invention of the electro-oscillator helped lay the groundwork for many of the electronic devices that we take for granted today.

From 1898 to 1912, Thaddeus Cahill developed the Telharmonium, also known as the Dynamophone, which was an early electronic music instrument. The massive device consisted of spinning electric generators and keyboard consoles and was able to generate electronically produced sounds. It was a precursor to the modern synthesizer which was capable of producing a wide range of sounds, including imitations of traditional instruments like pianos and violins. Although the Telharmonium was not a commercial success due to its size, complexity, and cost, it paved the way for later electronic music instruments and techniques. It also influenced the development of tonewheel organs and ultimately led to the famous Hammond B-3.

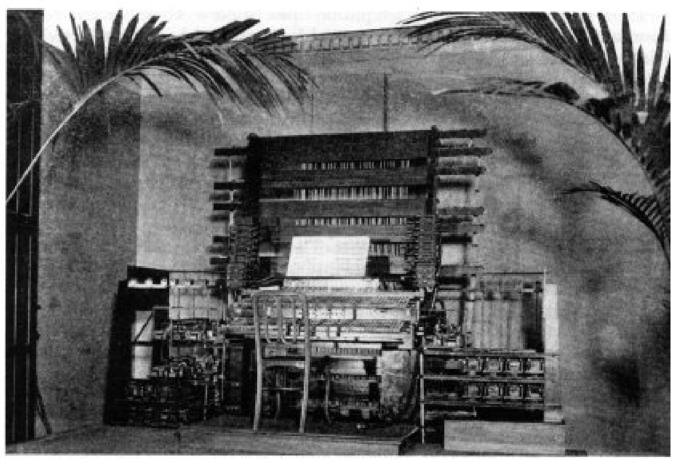


Figure 3.0.1: The Telharmonium was a massive instrument that occupied an entire room, consisting of a complex array of wires, dynamos, and various electronic components [10].

In 1906, Lee De Forest invented the Audion vacuum tube. In the Audition, the cathode is a negatively charged electrode that emits electrons, while the anode is a positively charged electrode that attracts electrons. The grid is a thin metal wire mesh located between the cathode and anode. By adding a third electrode, called the grid, between the cathode and anode, the flow of electrons between the cathode and anode could be controlled by varying the voltage on the grid, thus allowing the Audion to amplify electrical signals. When a voltage is applied between the cathode and anode, electrons flow from the cathode to the anode, creating a current. By adding the grid between the cathode and anode, the flow of electrons can be controlled by varying the voltage on the grid. This is because the grid's voltage affects the electric field between the cathode and anode, which in turn affects the flow of electrons. The device was then used as an amplifier, oscillator, and other sound-related applications.

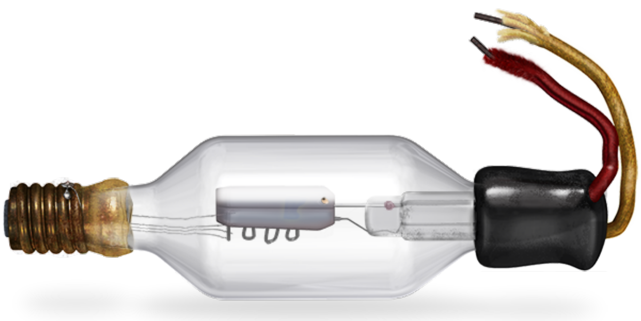


Figure 3.0.2: The triode audion became a key building block in the development of electronic technology [11].

Then in 1920, etherophone was invented by Russian physicist and musician Lev Sergeyevich Termen, better known as Leon Theremin. The Theremin consists of a box with two metal antennas, one vertical and one horizontal. The player moves their hands in the space between the antennas to control the pitch and volume of the sound produced. The Theremin works on the principle of capacitance, with the player's hands acting as variable capacitors that control the frequency of an oscillator circuit. I was especially into this instrument since it proposes an alternative way of interaction between the performer and the instrument. The fact that the instrument is played without physical contact opens up new possibilities for musical expression and paves the way for the development of new electronic instruments that challenge traditional notions of what constitutes an instrument.

At this stage, we had a complete set of tools for electronic music production and eventually transitioned from basic analog techniques to the digital era. After 1940, electronic music advanced rapidly through the use of digital technology and underwent a surge in theoretical innovation, challenging traditional methods. For instance, musique concrète involves the use of recorded sounds and environmental noises as the source material for musical composition. The composers typically manipulate these sounds using various techniques such as speed alteration, looping, filtering, and layering to create unique textures of sounds. This avant-garde approach to music production influenced a wide range of different genres. The experimentation with electronic instruments and techniques paved the way for the use of synthesizers in popular music, which became prominent in the 1970s and 80s. Funk bands such as Parliament-Funkadelic and Sly and the Family Stone incorporated synthesizers into their music, creating a new subgenre known as electro-funk. Synthpop emerged in the late 1970s, featuring synthesizers as the primary instrument, and heavily influenced by the experimental electronic music of the avant-garde. In the 1980s, Disco music also saw the use of synthesizers, creating the popular subgenre known as Italo disco. Today, EDM continues to push the boundaries of electronic music, incorporating various techniques and sounds from the history of electronic music.

4
Understanding Circuits

For this chapter, I will introduce some basic concepts about circuits. Circuits are the backbone of modern electronics and are used in a wide range of applications, from simple light switches to complex computer systems. A circuit is an interconnected network of electronic components that work together to perform a specific function. These components can include resistors, capacitors, diodes, transistors, and integrated circuits.

When designing circuits, engineers need to use the fundamental Ohm's and Kirchhoff's Laws, as well as other theorems such as Thévenin's Theorem and Norton's Theorem. Other techniques such as voltage division and current shunting are also required to determine the behavior of the circuit. Voltage division is a method of dividing the voltage applied to a circuit between two or more resistors in series, while current shunting is a method of dividing current between two or more parallel resistors.

4.1 Voltage Control

The "one volt per octave (V/Oct)" standard is a method of controlling voltage-controlled analog synthesizers that was popularized by Robert Moog in the 1960s. The basic principle behind this standard is that a change of one volt in the control voltage input to an oscillator will cause the oscillator's frequency to increase or decrease by a factor of two. In other words, if the control voltage is increased by one volt, the oscillator's frequency will double, and if the control voltage is decreased by one volt, the oscillator's frequency will halve. This standard allowed for precise and consistent control of analog synthesizers, which previously had been difficult to achieve. It also made it possible to use a keyboard to play melodies and harmonies, by assigning each key on the keyboard to a different voltage level. Today, this standard is still widely used in the design of analog and digital synthesizers, as well as in other electronic musical instruments.

Voltage dividers are used to convert a range of voltages into an appropriate V/Oct range for control over the pitch of the synthesizers. This allows for precise control over the pitch of each instrument in the ensemble, creating a cohesive and harmonious sound. To better understand how voltage dividers are utilized in synthesizers to control pitch, we will analyze a circuit and demonstrate how it functions as a voltage divider:

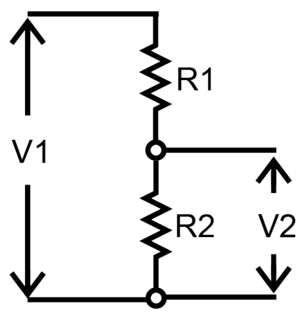


Figure 4.1.1: This is perhaps the simplest voltage divider circuit, as V_1 is the input voltage and V_2 is the output voltage.

Meanwhile, Ohm's Law states that the current passing through a conductor between two points is directly proportional to the voltage across the two points and inversely proportional to the resistance between them. As R_1 and R_2 are in series, we can acquire the simple relation:

$$\frac{V_2}{V_1} = \frac{R_2}{R_1 + R_2}$$

As engineers generally use V_2 for the output voltage and V_1 for the input voltage, we get

$$V_{\text{out}} = V_{\text{in}} \frac{R_2}{R_1 + R_2}$$

as a result.

This is commonly used in the circuit design of my project, especially for adjustable output voltages as I replace R_2 with a potentiometer.

4.2 Kirchhoff's Laws

Kirchhoff's circuit laws are fundamental principles used in the analysis of electrical circuits. These laws were developed by German physicist Gustav Kirchhoff in the mid-19th century, and they form the basis of circuit analysis throughout physics and electrical engineering. The two main laws are:

Kirchhoff's Current Law (KCL): The total current flowing into any node (or junction) in a circuit must equal the total current flowing out of that node. This law is based on the principle of conservation of charge.

Kirchhoff's Voltage Law (KVL): The sum of the voltage drops around any closed loop in a circuit must equal zero. This law is based on the principle of conservation of energy.

Kirchhoff's first law tells us that, in a circuit, a node is arbitrarily selected as a reference point. For each node, all currents entering the node are considered positive, and all currents leaving the node are considered negative. Then, according to Kirchhoff's second law, all voltage drops and voltage sources are listed at the node to obtain an equation. The voltage values of the components in the circuit can be obtained by solving the resulting system of equations. This process allows for the analysis of complex circuits and is a fundamental tool in electrical engineering. By using Kirchhoff's laws, engineers can design, troubleshoot, and optimize circuits for a variety of applications, from consumer electronics to power systems. An important thing to note is that Kirchoff's Laws apply to both linear circuits, where the output of the circuit changes proportionally in response to changes in the input, and to more complex types of circuits.

4.3 Thévenin's Theorem and Norton's Theorem

Thévenin's and Norton's theorems are two important theorems in circuit analysis that can be used to simplify complex circuits and make them easier to analyze. Thévenin's theorem states that any linear circuit containing independent sources and resistances can be replaced by an equivalent voltage source and series resistor. The voltage source has a voltage equal to the open-circuit voltage through the original circuit when the current sources are replaced by open circuits, and the series resistor has a resistance equal to the resistance to the original circuit when the current sources are replaced by open circuits. Similarly, Norton's theorem states that any linear circuit containing independent sources and resistances can be replaced by an equivalent current source and a parallel resistor. The current source has a current equal to the short-circuit current through the original circuit when the voltage sources are replaced by short circuits, and the parallel resistor has a resistance equal to the resistance looking back into the original circuit when the voltage sources are replaced by short circuits. The main difference between Thévenin's theorem and Norton's theorem is that Thévenin's theorem uses a voltage source and series resistor, while Norton's theorem uses a current source and parallel resistor to model the original circuit. Both theorems are used to simplify complex circuits into a single equivalent circuit, which can make analysis much easier. The equivalent circuit can be used to calculate the current, voltage, or power at any point in the circuit.

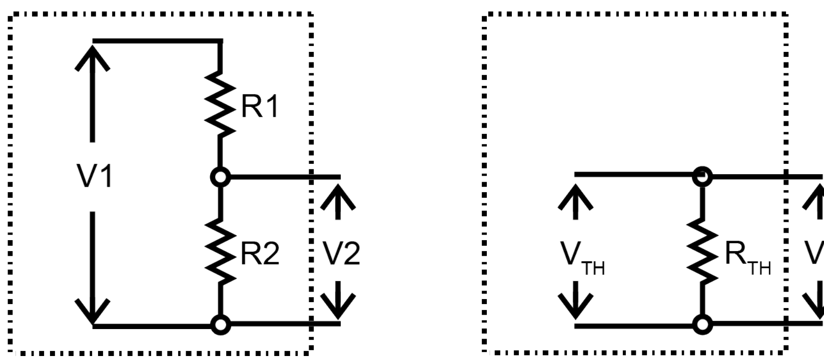


Figure 4.3.1: According to Thévenin's theorem, the two parts within the dotted rectangles are equivalent.

As Figure 4.3.1 shows, we can imagine the circuit within the dotted line to be within a black box, and if we don't know what the actual circuit is like, the two circuits are equivalent according to Thévenin's Theorem. T_{TH} (Thévenin's equivalent voltage) represents the open-circuit voltage between the two terminals of a linear electrical network, while R_{TH} (Thévenin's equivalent resistance) represents the equivalent resistance of the network as "seen" from those two terminals after all voltage and current sources have been replaced. My personal understanding of the theorems is based on the fact that there is no perfectly ideal source in the world, whether it is a voltage source or a current source. Therefore, we can simplify our analysis of circuits by treating them as sources with internal resistance. This simplification can help us to analyze circuits more easily and quickly, even when we don't have all the details of the circuit's internal workings. Similar to the voltage divider example we have above, the result of it would be $V_{\text{TH}} = V_1 \frac{R_2}{R_1+R_2}$ and $R_{\text{TH}} = \frac{R_1 R_2}{R_1+R_2}$. Both theorems only apply to linear circuits.

4.4 Impedance

As we already touched on resistance in the previous sections, now it's time to generalize the concept to impedance. Impedance refers to the total opposition that a circuit presents to the flow of alternating current (AC). It is a measure of the combined effects of resistance, capacitance, and inductance in a circuit, and is measured in ohms (Ω). In a circuit with only resistance, the impedance is equal to the resistance. However, in my project, we are also dealing with conductors, so the impedance becomes more complex and can be calculated using complex numbers. The complex impedance of a circuit is represented by a complex number in the form of $Z = R + jX$, where R is the resistance, X is the reactance ($j \equiv \sqrt{-1}$), and j is the imaginary unit. The magnitude of the impedance, $|Z|$, represents the total opposition to current flow in the circuit, and the phase angle, θ, represents the phase difference between the voltage and current. For a resistor, the impedance is equal to its resistance $Z_R = R$, as the resistor does not have any reactive elements like capacitors or inductors. For a capacitor, the $Z_C = \frac{1}{j\omega C}$, where Z is the impedance in ohms, ω is the frequency of the AC power source in radians per second, and C is the capacitance in farads. This indicates that the impedance of a capacitor does not only depend on its capacitance, but also on the frequency of the circuit. The impedance decreases as the frequency of the AC voltage applied to it increases. When we discuss filters later on, we will delve deeper into the concept of impedance.

5
Circuit Design Fundamentals

Designing a circuit can be a challenging task. It requires careful planning and attention to detail. As a beginner, I always find it intimidating to start building a circuit. However, I learned from this project that practice makes perfect. Experience plays a major role in circuit design, thus I would recommend going to any online forums and websites. Then, try to evaluate others' circuits to begin with.

Before embarking on the design process, it is essential to have some understanding of the circuit's purpose and requirements. This includes identifying the input and output signals, the operating voltage and current, and any specific constraints or limitations. With this information in hand, you can begin selecting the appropriate components, such as resistors, capacitors, transistors, and integrated circuits, based on their electrical characteristics and performance specifications.

After that, I tend to directly draw the circuit schematic with some simulation software. I use Multisim only because this was the first software I encountered. Alternatively, I also recommend LTSpice because it is free. For each software, there is a complete guide on their official website for you to walk through all the steps in order to make your own schematic or PCB. An additional resource that I found helpful is the tutorial page for Altium Designer [13].

Solderless breadboarding is so far the easiest way to accomplish a circuit. It involves using a breadboard, which is a plastic board with a grid of holes that are connected by metal strips. Components can be inserted into the holes to make electrical connections, and wires can be used to connect components together. Solderless breadboarding is a popular method for experimenting with circuit designs, as it allows for quick and easy modifications without the need for specialized equipment or skills. It is also a useful technique for testing and debugging circuits before they are built on a more permanent circuit board using soldering techniques. After that, you may want to try soldering on a prototyping board or a perforated board to practice soldering as well as have some rigid permanent connections. A loose connection can be a real pain for breadboard prototyping, as it can cause unreliable behavior in the circuit. When dealing with linear parts, sometimes making a module soldered board would be helpful for reusing due to superposition theorems, like Thévenin's Theorem and Norton's Theorem in the previous chapter.

When I initially prototyped circuits on a breadboard, it was rare that they worked at first, so troubleshooting skills are essential. A multimeter is a versatile instrument that can be used to test the parameters of components in a circuit, such as resistance, capacitance, and inductance. An oscilloscope is another important tool that can be used to observe the signal waveforms in a circuit and analyze the circuit's operating state. To test the response characteristics of a circuit, signal generators can be used to generate various signals. By using these tools and instruments, circuit designers and engineers can gain valuable insights into the behavior and performance of their circuits, helping them to optimize their designs and ensure reliable operation. However, it is much easier for us to try to make fewer mistakes. I want to give a big shoutout to John DiMarco for always printout the schematic, and then ticking on each node or pin after you carefully check it on your breadboard. This might seem simple, but it greatly reduced my work as my success increased.

After completing everything above and the circuit is ready to go, you can go back to the schematic and convert it to a PCB for a beginner-level circuit without needing to consider advanced concepts such as electromagnetic interference and high-frequency effects.

To summarize the procedures I used in designing a circuit board were:

- Determine the functional and performance requirements of the circuit
- Select the appropriate devices and components
- Draw the circuit schematic
- Perform circuit simulation and analysis
- Start solderless breadboarding and perform wiring
- (You also might want to build and test the soldered circuit board)
- Troubleshoot and optimize the circuit
- Finalize the performance and turn it into a Gerber file for print

6
Op-amp Applications

An operational amplifier (op-amp) is a type of integrated circuit that performs linear operations on an input signal. It is a high-gain direct-coupled amplifier with a differential input and an output that is proportional to the difference between the input signals. Op-amps are versatile electronic components that find application in numerous fields, including amplifiers, filters, voltage regulators, and signal converters, due to their desirable high gain, high input impedance, and low output impedance. They are widely used in analog circuits and can be found in a wide range of electronic devices, including audio and video equipment, medical instruments, and communication systems. In this chapter, I will introduce some basic rules and applications of op-amps.

6.1 Op-amp Golden Rules

There are two simple rules to describe the behavior of op-amps with external feedback, which I found immensely helpful; you can learn more about these rules in *the Art of Electronics* (Horowitz & Hill, 2015)[14].

Rule no.1: The output of an op-amp works to minimize the potential difference between its inputs. It is important to note that op-amps do not alter the voltage of their inputs. Rather, they have a high level of input impedance, which allows us to neglect any currents flowing into or out of the inputs.

Rule no.2: The inputs of an ideal op-amp draw no current. Although in practice, there may be some input bias current, which is generally negligible and can be ignored for most purposes.

6.2 Op-amps as Amplifiers

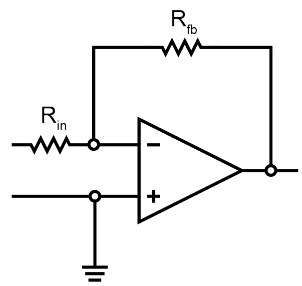


Figure 6.2.1: The output of the op-amp is fed back through the feedback resistor (R_{fb}) to the inverting input. The non-inverting input of the op-amp is grounded

Depending on the circuit configuration, the operational amplifier is classified into two types, namely, the inverting amplifier and the non-inverting amplifier. Figure 6.2.1 shows that the input of the op-amp is connected to the voltage source output and the op-amp output to a voltmeter. The voltage source is applied to one side of the input resistor R_{in}, whose other side is connected to the inverting input of the op-amp. The feedback resistor R_{fb} is connected between the op amp's inverting input and the op amp's output. Both of the resistors will influence the output of the amplifier circuit. By golden rule 2, the op-amp keeps the inverting and non-inverting inputs at the same voltage, that is at ground level 0 volts, thus the voltage across the input resistor is equal to the input voltage V_{in} and the voltage across the feedback resistor is equal to V_{out} and similarly throughout. Using Ohm's law, we have $V_{in} = I_{\text{in}}R_{\text{in}}$ and $V_{\text{out}} = -I_{\text{in}}R_{\text{fb}}$. Therefore, we have that Gain $= -\frac{R_{\text{fb}}}{R_{\text{in}}}$. . The deduction process for deriving the gain formula

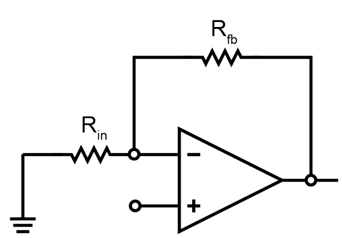


Figure 6.2.2: A non-inverting amplifier has the exact same arrangement, but the inputs were switched.

for a non-inverting op-amp amplifier is similar, except that the input voltage is connected to the non-inverting input terminal of the op-amp, and the feedback resistor is connected between the output and the non-inverting input terminal. The resulting gain formula for the non-inverting op-amp amplifier is Gain $= 1 + \frac{R_{\text{fb}}}{R_{\text{in}}}$.

6.3 Op-amps as Comparators

By removing the feedback resistor, the op-amp amplifier is transformed into a comparator. Comparators evaluate the voltage levels of both the inverting and non-inverting inputs, and subsequently perform an action when one voltage exceeds the other. Without feedback, the output voltage cannot alter the input voltage, leading the op-amp to oscillate wildly between its positive and negative maximum output voltages. When the output voltage reaches its highest or lowest possible value, the op-amp is considered to be in positive or negative saturation respectively. If we flip the inverting with the non-inverting input in Figure 6.3.1, we will get the inverting comparator, and the op-amp will reach saturation inversely. Op-amp comparators are used in the design of the Voltage Controlled Oscillator for square wave generation.

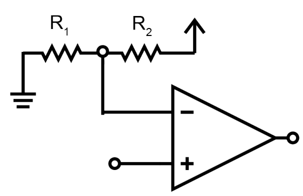


Figure 6.3.1: For a non-inverting comparator, the feedback resistor is replaced with a voltage reference level to determine the threshold for the output.

6.4 Hysteresis Demystified

As we discussed, comparators are commonly utilized to differentiate two distinct voltage levels. Hysteresis, or positive feedback, will reduce the susceptibility to noise. The circuit with hysteresis requires the input signal to drop below a certain threshold to trigger a change in its output state, however, the input signal needs to surpass a significantly higher threshold level to cause the output to switch back to its original state. A study conducted by TI Designs investigated a scenario in which multiple transitions occurred due to noise or signal variation in close proximity to the comparison threshold[15], and they discovered a huge variation in voltage within a short period of time before it stabilizes. To address this issue, hysteresis is implemented by setting an upper and lower threshold, which helps to eliminate multiple transitions caused by noise.

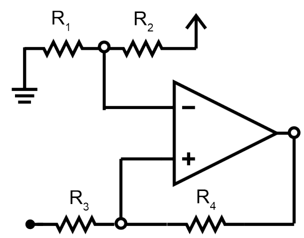


Figure 6.4.1: Comparator with hysteresis resistors R_3 and R_4 added.

As Figure 6.4.1 shows, a positive feedback resistor is added between the op-amp's output and its non-inverting input, as well as a resistor between the applied input voltage and the non-inverting input.

When the output of the op-amp is saturated low, current is being pulled through the positive feedback resistor away from the non-inverting input. This effectively lowers the voltage at the non-inverting input, which helps to alleviate output chatter. When the op-amp goes to high saturation because the voltage level on the non-inverting input went a few millivolts above the voltage level on the inverting input, the current through the positive feedback resistor, R_4, steps the voltage up on the non-inverting input, effectively pushing it out of the region that might result in circuit noise-induced chatter. The hysteresis zone is the region of input voltage over which the output of the comparator remains in a stable saturated state, either at the positive or negative supply voltage. By analyzing the effect of the feedback resistor, the voltage region over which the input can vary without tripping the comparator to one of its stable saturated states.

If the positive feedback resistor's value is too low (that is, if there is too much hysteresis), the op-amp may latch into a saturated state and stay there. Increasing the value of the positive feedback resistor can help to fix this issue. A low-value capacitor can also be placed across the positive feedback resistor (somewhere between 5pF to 20pF) to make the op-amp used as a comparator change states as quickly as possible. This helps to give the non-inverting input a little extra jolt in the right direction when the output changes state.

7
Noise Cornucopia

The noise Cornucopia was the first project I worked on, and the name was taken from Ray Wilson's Music From Outer Space website [16]. The Music From Outer Space is a valuable resource for individuals interested in building their own electronic musical instruments and effects.

On the website, Ray Wilson explained how an NPN transistor operates like a zener diode when the emitter to base breakdown voltage is exceeded. Low-voltage white noise is a byproduct of the zener breakdown mechanism. In order to use the white noise signal produced in this way in a synthesizer, it must first be amplified due to its extremely low amplitude. So I took the upper part of his circuit design and started testing it on Multisim, yet found that Multisim does not allow the collector of the transistor to be cut off and operates in avalanche mode. Thus, I worked on a solderless breadboard instead.

The positive supply voltage is applied to the emitter of Q_1 via R_{33} and R_2, which are 470kΩ resistors in series. Capacitor C_{18} is used to filter the voltage applied to the emitter, reducing the possibility of supply ripple contaminating the noise output. The noise generated at the EB junction of Q_1 is coupled to the non-inverting input of U1-A, which is biased to the ground by a 2MΩ resistor (R_4) that amplifies the input signal by a factor of 48.

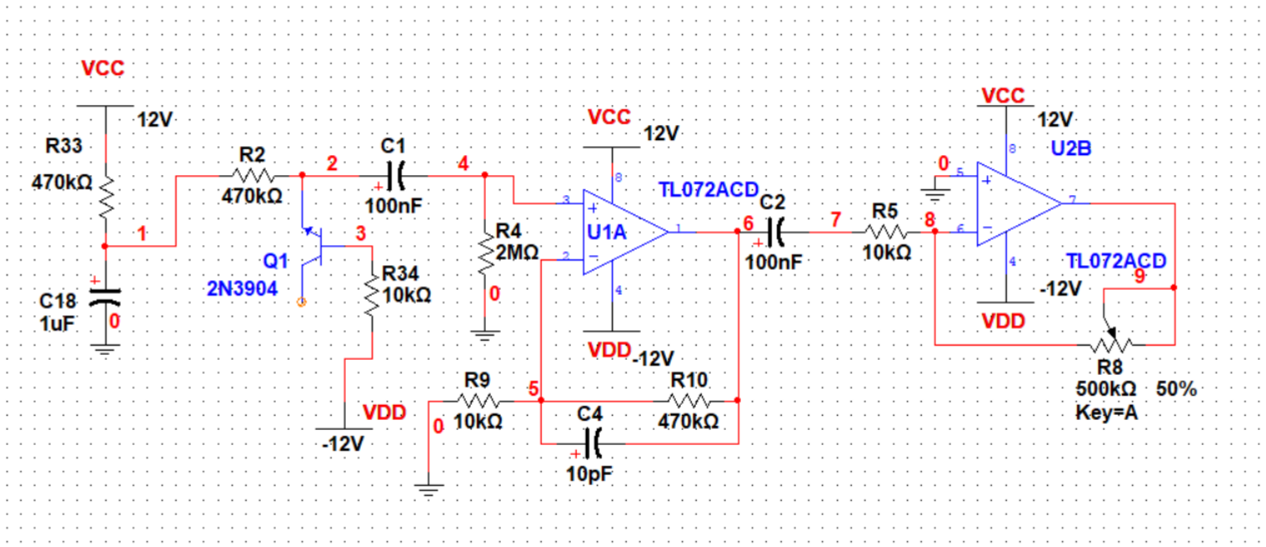


Figure 7.0.1: This is the upper part of the Noise Cornucopia designed by Ray Wilson, using 2N3904 as a low-frequency noise source [16].

The signal is then feed in to U2-B, an active low-pass filter. It consists two part: an active low-pass filter consists of a resistor-capacitor (RC) network in combination with an op-amp. The op-amp is configured as an inverting amplifier, with the RC network placed in the feedback loop. The cutoff frequency of the filter is determined by the values of the resistor and capacitor.

An RC circuit is composed of a resistor R and a capacitor C, and is often used for signal filtering. The circuit works by charging and discharging the capacitor, allowing only low-frequency signals to pass through to the output. When an input signal passes through the RC circuit, the high-frequency components of the signal are short-circuited by the capacitor, while the low-frequency components pass through into the resistor. This filtering process occurs because at high frequencies, the capacitor has a low impedance and easily conducts the current, bypassing the resistor. As a result, the high-frequency signals are removed and only the low-frequency signals are allowed to pass through the circuit, leading to effective signal filtering.

We start with the voltage divider equation, which states that the voltage across a resistor in series with a capacitor is proportional to the ratio of the resistor's resistance and the total circuit resistance:

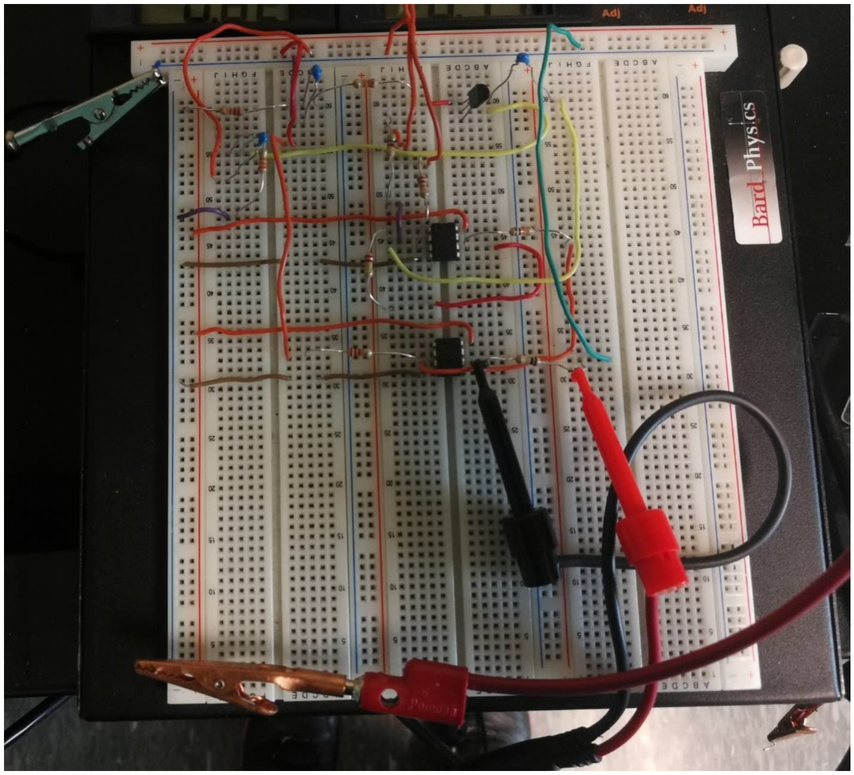


Figure 7.0.2: The breadboard for the upper part of the Noise Cornucopia schematic [16].

$$V_{out} = V_{in} \cdot \frac{R}{R + \frac{1}{j\omega C}} \tag{7.0.1}$$

where V_{in} is the input voltage, V_{out} is the output voltage, R is the resistance of the resistor, C is the capacitance of the capacitor, and j is the imaginary unit.

To simplify this equation, we first multiply the numerator and denominator of the fraction by j C, which results in:

$$V_{out} = V_{in} \cdot \frac{R}{j\omega RC + 1} \tag{7.0.2}$$

We can then rewrite this expression in terms of the filter's cutoff frequency, which is the frequency at which the filter begins to attenuate the signal:

$$V_{out} = V_{in} \cdot \frac{1}{1 + j\frac{\omega}{\omega_c}} \tag{7.0.3}$$

where ${}_c = \frac{1}{RC} = 2\pi f$, so $f = \frac{1}{2\pi RC}$ is the cutoff frequency.

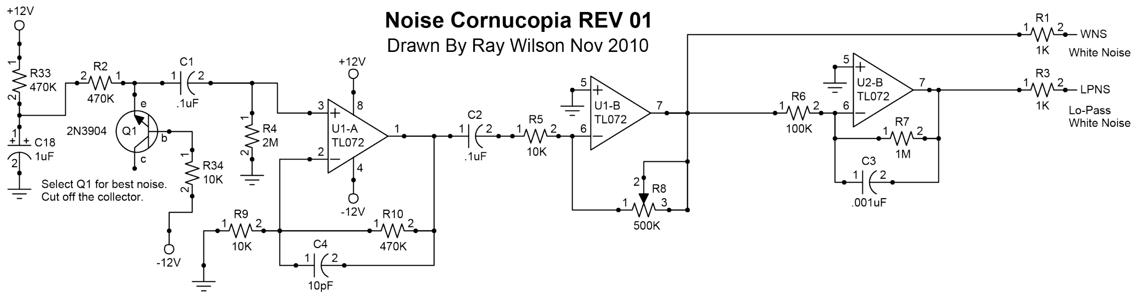


Figure 7.0.3: The original version of Ray Wilson's Noise Cornucopia schematic, and I will discuss the part around U2-B [16].

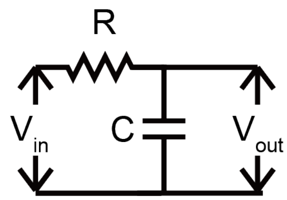


Figure 7.0.4: This is perhaps the simplest voltage divider circuit, as V_1 is the input voltage and V_2 is the output voltage.

This equation describes a low-pass filter, which attenuates signals with frequencies above the cutoff frequency and passes signals with frequencies below the cutoff frequency. The magnitude response is given by:

$$\left|\frac{V_{out}}{V_{in}}\right| = \frac{1}{\sqrt{1+\left(\frac{\omega}{\omega_c}\right)^2}} \tag{7.0.4}$$

Then since the op-amp is powered by an external source, we also get an amplification as we describe in the Op-amp as Amplifier section:

$$G = \frac{V_{out}}{V_{in}}$$

. Therefore, we get a nice wave-like low-pass filter with amplification.

8
Voltage-Controlled Oscillator

In an analog synthesizer, the Voltage-controlled Oscillator (VCO) is a fundamental building block that generates the basic waveform that will be shaped and processed by other modules, such as filters, envelopes, and amplifiers. The frequency of the VCO is determined by the control voltage applied to it and corresponds to the musical pitch. As the range of human hearing is typically from 20Hz to 20kHz, I am trying to build an oscillator with frequencies varying from up to 15kHz, as lower frequencies are also needed for modulation purposes.

A quick note is that this chapter presents a more physics-oriented approach to circuit analysis, with a focus on mathematical deduction and complex verbal explanations of more intricate circuits. If you feel more comfortable with the approach so far, feel free to jump ahead.

8.1 Square Wave Circuit Analysis

A typical sine wave generator circuit is composed of a comparator with inverted input and an RC circuit. The comparator switches the output of the circuit between a high and low level, denoted as $\pm V_o(\text{sat})$ and the RC circuit serves as a delay element to determine the duration of each state, shown in Figure 4.1.1.

In this rectangular wave generation circuit, there is no external input signal. To work properly, the integrated op-amp chip must be connected to an external power supply. When the external power supply is turned on, a certain input signal will be generated nearly instantaneously at the

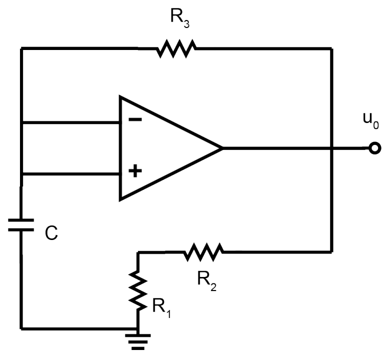


Figure 8.1.1: Square Wave Generator Circuit

input of the integrated op-amp due to the sudden access to the power supply voltage, causing the output voltage to reach the saturation value. Whether the output is a positive or negative saturation is determined by the input signal.

The op-amp inverting input is connected to the capacitor with capacitance C, assuming that the initial value of the capacitor voltage is 0. For resistor R_3, the left side is at a low potential and the right side is at a high potential, hence there is a current I_F. The current flowing into the inverting input of the op-amp is 0, so the current is equal to the current flowing through the capacitor I_C. Therefore, we have $I_F = I_C$.

The current flowing through the capacitor I_C will gradually increase the voltage across the capacitor, thus the capacitor is charging up.

However, u_C cannot get infinitely large. When $u_C > V_{\mathrm{TH}}$, the voltage at the inverting input of the op-amp is greater than the voltage at the non-inverting input, and the voltage at the output of the integrated op-amp will change rapidly from the positive saturation voltage value to the negative saturation voltage value due to the positive feedback, i.e. $u_o = -V_o(\mathrm{sat})$. At this time, the voltage at the non-inverting input of the op-amp $u_+ = -\frac{R1}{R1+R2}V_o(\mathrm{sat})$. Let $u_+ = V_{\mathrm{TL}}$.

When the output voltage is at the negative saturation level, the left side of the resistor R_3 is at low potential and the right side is at high potential, the current and the actual direction are opposite to the reference direction. Therefore, the capacitor is discharging.

When the capacitor is discharged so that the voltage across it u_C is lower than V_{TL}, the voltage at the output of the op-amp will change from negative saturation voltage to positive saturation voltage, and the process keeps repeating, forming a rectangular wave at the output.

Now let's consider node equations in order to find the time period and the frequency of the Square Wave Generator. We already demonstrated that when $u_o = V_o(\mathrm{sat})$, $u_+ = V_{\mathrm{TH}}$ and when $u_o = -V_o(\mathrm{sat})$, $u_+ = V_{\mathrm{TL}}$.

$$\begin{aligned} -C\frac{du_C}{dt} &= \frac{u_C - u_o}{R_3} \\ \frac{du_C}{u_o - u_C} &= \frac{dt}{R_3 C} \end{aligned} \tag{8.1.1}$$

Assuming the initial voltage across the capacitor is 0, we have

$$\int_0^{u_C} (\frac{du_C}{u_o - u_C}) = \int_0^t \frac{dt}{R_3 C} \tag{8.1.2}$$

Thus the solution to such ODE is

$$\begin{aligned} ln(u_o - u_C) = -\frac{t}{R_3 C} + K \\ u_o - u_C = Ke^{-t/R_3 C} \end{aligned} \tag{8.1.3}$$

where K stands for an arbitrary constant.

Take $t = 0$, we can get rid of the constant Kas $K = u_o$. By substitution we get

$$u_C = u_o[1 - e^{-t/R_3 C}] \tag{8.1.4}$$

We can now sum up the contribution of both charging and discharging, let $u_C = 0$, $u_o = V_o(\mathrm{sat})$ for charging up, and $u_o = -V_o(\mathrm{sat})$ for discharging, we get

$$u_C(t) = -V_o(\mathrm{sat}) + \frac{R_1}{R_1 + R_2} V_o(\mathrm{sat}) + V_o(\mathrm{sat}))e^{-t/\tau} \tag{8.1.5}$$

Where the characteristic time $\tau = R_3 C$.

When $t = \frac{T}{2}$, the voltage across the capacitor R_3 is at a saturated level, thus

$$u_C(\frac{T}{2}) = V_{\mathrm{TL}} = -\frac{R_1}{R_1 + R_2} V_o(\mathrm{sat}). \tag{8.1.6}$$

And the period of the square wave generator is

$$T = 2R_3C \cdot ln(1 + \frac{2R_1}{R_2}). \tag{8.1.7}$$

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8.2 Schematic and PCB design

The generated 3D configuration by Altium Designer is shown below:

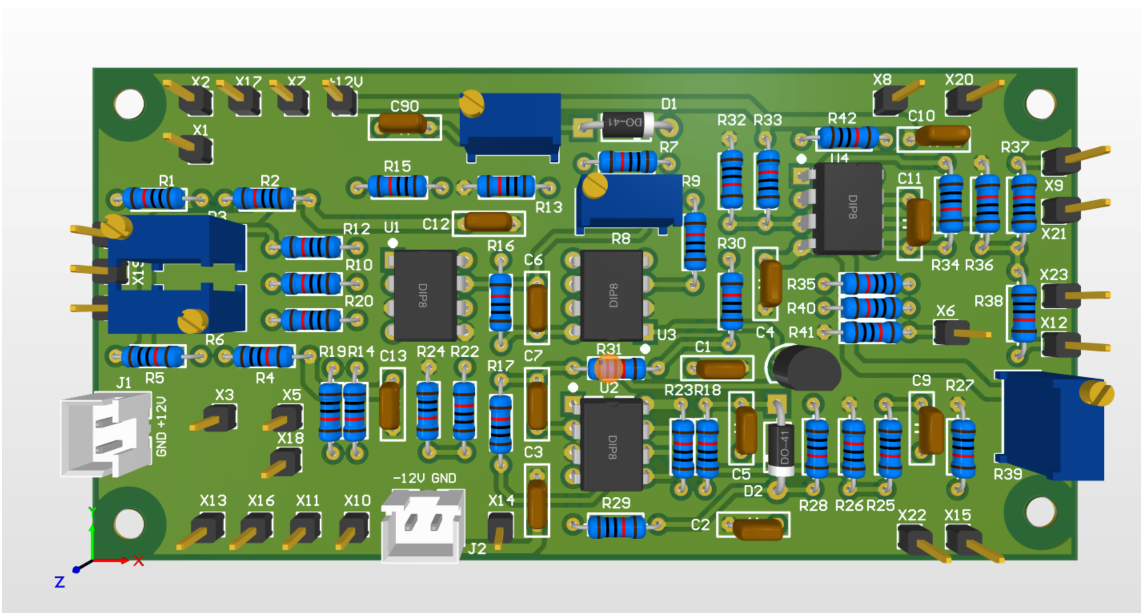
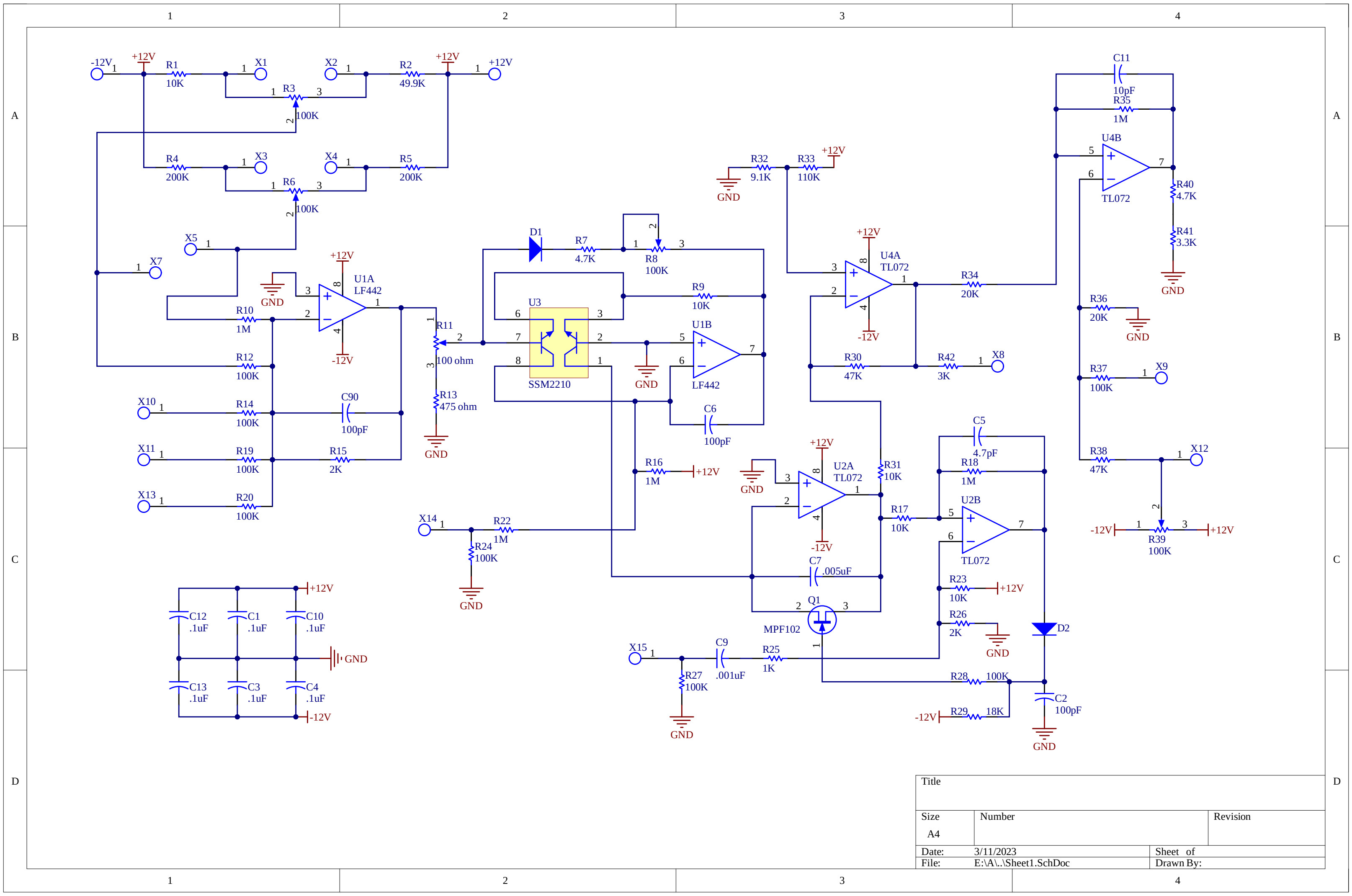


Figure 8.2.1: 3D PCB Demonstration

This is a top-down view of the VCO module, excluding the potential meter and switches that would be installed on the panel. Although I haven't yet reached the point of printing and soldering the entire PCB, I've already learned a valuable lesson: never underestimate the amount of work required for PCB printing and assembly. Even small changes in the schematic can cause significant delays in the project timeline.

I will include the schematic on the next page and explain its design.


1
2
3
4
A
B
C
D
-12V
+12V
R1
10K
X1
X2
R2
49.9K
R3
100K
R4
200K
X3
X4
R5
200K
R6
100K
X5
X7
GND
U1A
LF442
R10
1M
R11
100 ohm
R12
100K
X10
R14
100K
C90
100pF
R13
475 ohm
X11
R19
100K
R15
2K
X13
R20
100K
D1
R7
4.7K
R8
100K
R9
10K
U3
SSM2210
U1B
LF442
C6
100pF
R16
1M
X14
R22
1M
R24
100K
C12
.1uF
C1
.1uF
C10
.1uF
C13
.1uF
C3
.1uF
C4
.1uF
R32
9.1K
R33
110K
U4A
TL072
R34
20K
R30
47K
R42
3K
X8
U2A
TL072
R31
10K
R17
10K
C7
.005uF
Q1
MPF102
X15
C9
.001uF
R25
1K
R27
100K
C11
10pF
R35
1M
U4B
TL072
R40
4.7K
R41
3.3K
R36
20K
R37
100K
X9
R38
47K
X12
R39
100K
C5
4.7pF
R18
1M
U2B
TL072
R23
10K
R26
2K
D2
R28
100K
R29
18K
C2
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Title
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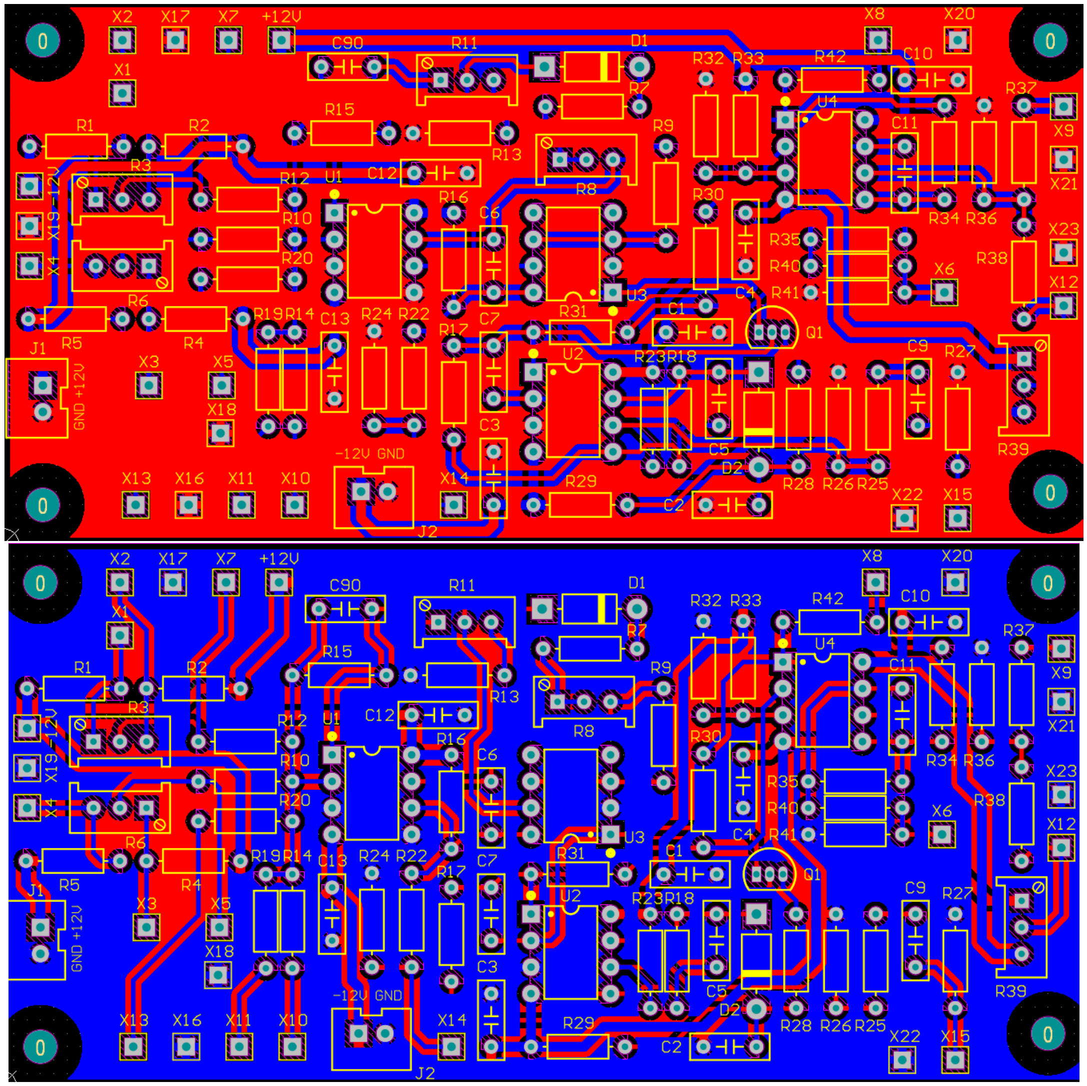

Figure 8.2.2: PCB top and bottom layer

The VCOs will produce sawtooth and variable width rectangular waveform outputs. The circuit is composed of several modules, including a voltage summer, a linear voltage to exponential current converter, and an integrator-comparator. It features six voltage inputs, five exponential ($X5$, $X7$, $X10$, $X11$ and $X13$) and one linear ($X14$), which control the frequency of each oscillator. $X5$ and $X7$ are wired to the Coarse and Fine Frequency Adjust pots (R_3 and R_6 respectively), while $X10$, $X11$ and $X13$ are wired to the banana and 1/4" jacks on the front and auxiliary panels (refer to the wiring diagram section for details). The Coarse pot can adjust from -10V to +4.5V, while the Fine pot provides finer control within a range of -2.4V to +2.4V.

The voltage from the Coarse pot is fed via the wiper of R_3 to voltage summer (U1-A) input resistor R_{12}, allowing for frequency adjustment from a few Hz to about 20 kHz. On the other hand, the voltage from the Fine pot is fed into the voltage summer via R_{10} (1 MΩ).

The voltage summer provides a gain of -.02 for the inputs with 100K input resistors and -.002 for the fine input which has a 1M input resistor. The linear voltage to exponential current converter is made up of U1-B and U3 (SSM2210 matched NPN transistor pair) and converts voltage to current in an exponential relationship. For further reading, I recommend National Semiconductor Application Note *Log Convertors* by Robert C. Dobkin [17] and *Musical Application Of Microprocessors* by Hal Chamberlin Chapter 6 [18]. The voltage applied to the base of the transistor U3 at pin 7 controls the collector current that flows through. The relationship between the voltage and current is exponential, and for every -18mV increment at the base, the current doubles in magnitude. The low end of the current range is about 0.02 A when both coarse and fine pots are turned fully counter-clockwise. When the coarse pot is adjusted to about 40 Hz, the current is at 0.4uA and doubles for every octave thereafter. The resistor R_{16} sets the initial current for the feedback loop of U1-B. By applying a control voltage to R_{22}, the current set by R_{16} is affected directly, and thus, the input $X14$ provides a linear frequency modulation input. Then to stabilize the U1-B, the capacitor C_6 is used. It acts as a low-pass filter, preventing high-frequency signals from causing instability in the amplifier.

A metal film resistor R_{15} is used as the feedback resistor around voltage summer U1-A to cancel out the exponential convertor's temperature coefficient for compensation purpose. This resistance causes the oscillator to go flat at higher frequencies. Adjustment of R_8 allows this error factor to be compensated so that the oscillator works over a wide range of frequencies.

Op amps U2-A and U2-B, along with the JFET Q1 MPF102, are used to generate the ramp wave oscillation that determines the VCO's frequency. U2-A is used as an integrator in association with C7. When the current flows out of U2-A pin 2 into the collector of U3 pin 1, it causes U2-A pin 1 (U2-A's output) to rise linearly in voltage. This is because U2-A's output is trying to hold pin 2 at the same potential as pin 3, which is connected to ground.

Since its output is also connected to U2-B, which is wired as a comparator, when the voltage at U2-A pin 1 rises above the threshold, U2-B's output shoots to +10.5V from its initial state of about -10.5V. This action causes Q1's gate to be brought high via D2 and R_{28}, effectively shorting its drain to source, discharging C7 (.005uF polystyrene or polycarbonate) integrator capacitor, and resetting the integrator's output to ground. Since the voltage on U2-B's non-inverting input is now below the threshold set by R_{23} and R_{26}, its output returns low (-10.5V), and U2-A's output begins to ramp up again, and the cycle continues, resulting in ramp wave oscillation. C_5 conducts the rising output of U2-B to its non-inverting input (positive feedback) to ensure that the output snaps high quickly and completely. The pulse that occurs on the output of U2-B forward biases D2 and charges C_2 (100pF cap). This charge ensures that Q1 stays on long enough to completely discharge C_7. U2-B's output returns low so fast that without this additional charge to keep Q1 on, C7 would not be discharged effectively. Thus, the output of U2-A is a ramp wave whose frequency is directly proportional to the current flowing out of U2-A pin 2.

Op-amp U4-A level shifts and inverts the ramp wave, causing it to become a +/-5V sawtooth wave oscillating about ground. U4-A's output is fed to the front panel connections (X8) via a 3K resistor to protect it from being directly shorted to something it would not like.

The sawtooth waveform is fed into the non-inverting input of U4-B wired as a comparator via R_{34}. When the sawtooth is above the voltage seen at U4-B pin 6 (plus a bit of hysteresis due to R_{35}) the output of U4-B goes high. When the sawtooth is below the voltage seen at U4-B pin 6 (plus a bit of hysteresis due to R_{35}) the output of U4-B goes low. Pulse width adjustment pot R_{39} (100K) sets the voltage level on U4-B pin 6 (via 47K at X12) and thus changes the duty cycle of the rectangular pulse appearing on U4-B's output. External pulse width modulation input is via R_{37} at X9 (20K resistor).

9
Keyboard Controller

I am currently using an unknown keyboard that I acquired from the music department. Despite not knowing the brand or model, I find it to be a decent keyboard with a comfortable feel and responsive keys. It has 61 keys and a 16-pin output, so the main focus of this chapter is trying to reconstruct the schematic from the PCB. The lack of information brings new challenges for reconstructing its schematic. It highlights the importance of being able to analyze and reverse-engineer circuits, as it may be necessary to do so in real-world situations where information about a particular component or device is not readily available.

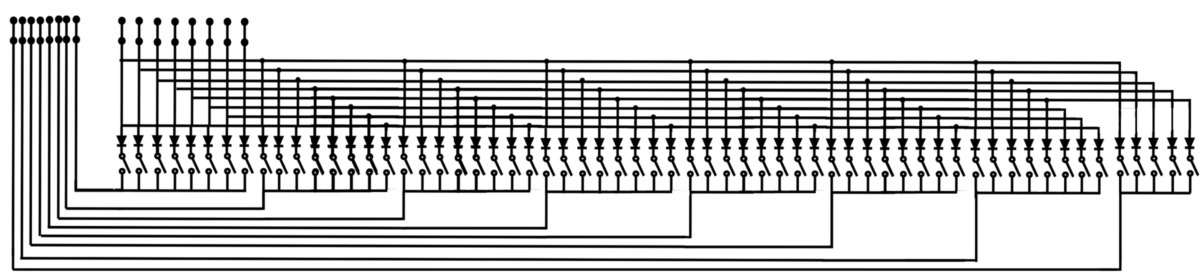

Figure 9.0.1: This is the 61-key keyboard Schematic I reconstructed.

Reconstructing a schematic from a PCB can be a difficult and time-consuming task, especially if the PCB is complex or has many components. Luckily, the keyboard is not too complex. Once again I found some hints from Music From Outer Space [19], and found the design were somehow similar.

Last but not least, I want to show you the picture of the lovely synthesizer. I am looking forward to the day I will finish my synthesizer project. There are many other modules to be built, but I will keep working on this project after submission.

Figure 9.0.2: Big shout out to Angela Yan for drawing the amazing front panel.

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