Creative Music Production

Professional Project

Kieran Murphy

Is the synthesiser community's preference for Analog equipment justifiable, Can the average user hear the difference between Analog and Digital Subtractive Synthesis.

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Supervisor

Mr. Conor Brennan

Abstract:

A dual development process was completed in this project, an analog synthesiser was constructed and a digital synthesiser was programmed. The effectiveness of the devices was confirmed in the listening test, Digital emulations are possible and the analog model produced the desired characteristics according to the survey.

A survey was designed to determine if a synthesiser user could identify the generation source, be it analog or digital. The results found that most participants had a favourite between the two, but could not accurately determine its source in an A/B listening test. Research into words most commonly used to describe audio was carried out, it found several academic sources with "warmth" being the most common descriptor. A verbal association test was designed and confirmed that warmth is the most common descriptor. However, the sample materials did not contain the warmth characteristic that has been previously defined by other researchers in the field. An interview with an industry expert supported by comments from the community helped to further understand why people have a preference, listing tactile nature and mood as key factors. A complete analog synthesiser design can now be considered for further work.

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Introduction:

An entire growing industry is based on developing modular analog synthesisers, once taught to be obsolete technology, it is now making a revival. The revival of analog synthesis has been well documented by fans but it possesses the question as to why. The subculture movement seems to have a bias towards analog synthesisers, however digital emulations of the classics have greatly improved in recent years. Can the average synthesiser user hear the difference between analog and digital subtractive synthesis.

This project highlights previous research on the synthesiser subculture, and the history of analog synthesisers. It also documents the recent developments in virtual analog emulations. A technical exploration takes place discussing a dual development of an analog and a digital subtractive synthesiser. This dual development highlights the pros and cons of each system and the difficulties in developing them.

The builds served a purpose as the method of sound generation in a A/B listening test. The test was designed to answer the question on whether the user can determine the difference, as well as gather insights into why people prefer one form of synthesis over another. The results give a good indication on the users ability to identify the sound source.

Unexpected results were gained via an interview with industry experts and comments from the users themselves, as they explained their own views on the research question.

Literature Review

"I Dream of Wires" documents the coinciding dual development of the modular synthesiser by Robert Moog and Donald Buchla and its inevitable introduction to the mainstream audience. Working independently, Moog and Buchla arrived at fundamentally different instruments that the individuals in the documentary fawn over. Moog developed a traditional interface using a 1 volt per octave keyboard controller, and Buchla developed an experimental capacitive touch controller (Fantinatto 2014). The documentary features high profile interviews from pioneers such as Morton Subotnick, Trent Reznor and Gary Numan as they detail the turbulent rise and fall of the modular synthesiser.

Notable moments in synthesiser history are met with criticism, Subotnick in particular was left unimpressed with the commercial success of "*Switched on Bach*", the first synthesiser album to go platinum. Subotnick made comments about how this new form of musical expression is not intended for the classics. This attitude draws some similarities to the modern-day subculture that is responsible for collecting and maintaining analog modular systems.

The documentary would have been more relevant if the researchers had included a comparative section detailing the differences between analog and digital synthesisers. Instead, it focuses on the avant-garde musicians and niche hobbyists, whose only references to working "inside the box" are negative. Selection bias is a potential concern for this documentary. The work lacks technical exploration and presents the analog synthesisers of the past as the holy grail of synthesis, presumably to avoid alienating the modular fan base. This leads to the question of whether sound quality over nostalgia is even the main concern for the modular enthusiasts.

"Synthesised" explores the analog synthesis world. The documentary focuses on the social identity often exhibited by analog enthusiasts, while also taking a look at the analog revival and the culture behind the movement (Kresse 2016).

The end of the 1990s saw the rise of computer emulations of analog synths, often indistinguishable from the real thing. The first analog revival started around 2001 when a sense of nostalgia awakened an interest in the analog domain, with the rise in Vinyl sales and an aesthetic appreciation of the items themselves and their output. Kresse suggests that the analog community does not wholeheartedly resist the digital domain, as it caters to its own palette of sounds unachievable in the analog domain. However, the documentary states that the draw towards analog gear is often down to its tactile nature.

Recent synthesizer meetups have exhibited similarities to other subculture movements, showing distinct symbolism between clothing and the music. A key method of communication is through the association of owning and performing with a piece of hardware which influences the artist's sense of identity. Kresse also suggests that the eurorack movement, while gaining popularity, is still an underground scene. It is also suggested that the analog revival has grown ritualistic behaviour such as patching cables, tuning oscillators, and regulating CV (Control Voltage) even before a single melody is played.

The bonds formed between musicians and their instruments will inherently cause bias. Since the tactile nature of hardware is a massive selling point, it leads to the question if the users can even distinguish between a decent virtual analog emulation and real analog circuitry.

Stanley Lunetta was an early adopter of digital electronics in music and a musical sculpture enthusiast. The early workings of Stanley's synthesizers were inspired by the logic-based CMOS (Complementary Metal-Oxide Semiconductor) chipset, specifically the 4000 series. "*Introduction to the Lunetta style CMOS Synths*" by Sergio Gonzalez catalogues various methods and resources on building a DIY Lunetta system. Gonzales states that the Stanley Lunetta spirit is to not worry about a machine being "pure". This links in with the Do It Yourself attitude of the modular subculture movement presented in "*I Dream of Wires"* and "*Synthesised*".

Today the CMOS chipset is rarely used apart from for educational purposes and is considered obsolete technology. However, the 4000 series allows quick and easy implementation, with low power consumption, of various modular oscillators and utility features. The 40106 is a common chip among the Lunetta community, the 40106 is a Hex Schmitt-Trigger inverter. The term inverter refers to the chip's operational voltage thresholds. Gonzalez fails to explain the circuits in the fan zine in technical detail, often presenting schematics straight from the manufacturer's data sheet. A more detailed reference is the "*CMOS Cookbook*" by Don Lancaster.

Additional research is required to construct a feasible analog circuit. While the zine provides a jumping off point it fails to teach the user how to implement a stable and musical oscillator conforming to the western scale. With the addition of an operational amplifier and other common components, a circuit can be designed that produces classic analog waveshapes, which can be used in the subsequent listening tests.

Audio morphing is the technique of encoding the spectral shape and pitch of one sound on independent axes and blending it with the interpolated amplitude of a second sound source (Williams 2). By creating a new sound from a pair of existing sounds, both audio morphing and additive synthesis share some similarities. Williams and Brooks coded a device capable of blending the spectral "warmth region" of one sound with another and validated the effectiveness of the device with a series of listening tests. The results were published to Audio Engineering Society convention, a reputable peer-reviewed publisher.

Credible academic research on analog warmth is scarce, however, Williams and Brookes have clearly defined "warmth" as the spectral energy 3.5 times above the fundamental frequency. They further explained how audio enthusiasts denote the nonlinearities of analog equipment to possess the warmth characteristics. Interestingly the research conducted a verbal elicitation experiment, where the participants described the perceived spectral differences using additional adjectives. It found that audio samples with the enhanced warmth characteristic were often described as bassy, wide, fat, thick and rounded (Williams 8).

A similar approach will be taken as participants will be asked to select common adjectives associated with audio from a glossary. The resulting digital and analog oscillators and filters will undergo spectral analysis for the presence of a warmth region.

A Brief History Of Virtual Analog Synthesis quantifies the number of research papers related to the relatively newly coined term "Virtual Analog". VA synthesizer algorithms have been developing since the release of the Clavia NordLead synthesizer, released in 1995. Since then, research on the methods for virtual subtractive synthesis has increased in academia and the music technology industry. Subtractive synthesis is the generation of a spectrally rich source later passed through a filter, typically a low pass filter.

The 1980s onward saw the focus shift from this method of synthesis towards the digital domain with the rise in popularity in FM (Frequency modulation) and wavetable algorithms. However, the market for the classic “warm” sounds of analog circuitry was beginning to emerge. (Pekonen and Valimaki 2).

Current trends and future research is discussed as half of all publications on VA were released between 2005 and 2010 from the date of publication. Most of the research is focused on source signal generation linked to classic setups of modular structures, with issues arising from trivially generating waveshapes. The concern with harsh aliasing in digital systems is highlighted as analog signals are infinite.

The paper highlights methods used to generate signals to improve the realism of VA waveshapes, methods such as bandlimited algorithms to produce harmonic components based around a non-modulating fundamental frequency, quasi-bandlimited algorithms to allow aliasing to happen outside of the human hearing range, and anti-aliasing methods which utilises higher sampling rates to further suppress aliasing.

At the core of every synthesizer is the oscillator, and so far analog oscillators have been discussed. Dr Jussi Pekonen put forward several methods on how to produce raw waveforms in his dissertation “*Filter-Based Oscillator Algorithms for Virtual Analog Synthesis*.” Several algorithms are presented and methods for producing anti-aliasing oscillators. Alaising is an unwanted artifact often heard as distortion, and can occur during the sampling process.

An important theorem in the field of digital signal processing is the Nyquist rate, which is generally understood as the process of doubling the sample rate of the highest frequency present in the signal. In relation to continuous time signals (or analog signals) this would mean a minimum of 44,100 samples per second to reconstruct the signal without introducing aliasing errors.

Ideally, bandlimited oscillator algorithms try to avoid aliasing completely by generating waveforms with a finite number of harmonics. One such approach would be additive synthesis, however, the computational load can be extreme. This can be eased by storing the waveshapes in a look-up table but this is not the ideal solution for memory sensitive applications.

The band-limited impulse train method takes advantage of human hearing by creating a softened discontinuity. The BLIT technique generates a stream of low pass filtered impulses, the impulse response of an "Ideal filter" otherwise known as the "sinc" function is bandlimited by nature. With the help of oversampling aliasing still happens, but much higher in the frequency spectrum where human hearing is not as sensitive (Pekonen 15) .

While these two algorithms focus on removing aliasing, the aliasing suppression algorithm allows aliasing to happen across the entire audio band. Trivally generated waveforms like ones produced in the analog domain (Triangle, Square) create noticeable amounts of aliasing when done in a digital system. In the digital domain, earliest forms of aliasing suppression were based on oversampling. Doubling the sampling rate can reduce aliased components by 6db in square waves and up to 12db in triangle waves. The higher the sampling rate the better the results, however the more computational strain put on the system. Each of these methods will be explored in Max and analyzed spectrally to determine which method produces the least distorted signal.

Electronic Music and Sound Design discusses the low-level workings of digital synthesis. In the chapter "Equations for digital filters" Cipriani and Maurizio highlight how filters modify the audio spectra by altering its amplitude and phase, leaving the fundamental oscillation unchanged. The book discusses the methodologies of filtering a signal, two common methods are the finite impulse response (FIR) and the infinite impulse response algorithms (IIF) which can be further classified by their filter order, described as the level of attenuation in decibels.

The FIR algorithm is a non-recursive filter, so-called as it only requires an input sample as the basis of its calculation. The calculation can be described as the output being equal to the summation of its input multiplied by a coefficient. To increase the filter order a single sample from the previously sampled input is stored in a delay line and multiplied by the coefficient. Then the rescaled samples are summed together resulting in the new sample output (Cipriani and Maurizio 355).

The IIF algorithm is a recursive filter, it requires both an input sample and an output sample. For this filter, the samples derived from the output are subtracted from the sum of the input samples. Calculating filter coefficients is outside the scope of this study, however audio originated programs such as Max and Csound provide simple functions to carry out the calculations.

The book highlights that the FIR algorithm is much easier to implement in Max, and allows variable control of the filter cut off. Using the FIR algorithm in the first order will offer good attenuation similar to analog filters.

The writers discuss “Harmonic and Non-Harmonic spectra” in a subsequent chapter in the book detailing how harmonic and non-harmonic spectrums can be generated using additive synthesis. At its core, additive synthesis is the summation of many sine waves at multiples of a fundamental frequency. The fundamental of a harmonic sound is the component with the lowest frequency value and the highest amplitude, with all other components being integer multiples of the fundamental, i.e 100Hz, 200Hz, 300Hz and so on. This is referred to as a periodic shape where each component is equidistant. The spectral envelope is calculated from the amplitude of a given harmonic, which gives the wave its shape (Cipriani and Maurizio 217).

Humans can still distinguish the pitch of a sound if the fundamental is missing, the period of the waveform corresponds to the inverse of the frequency. The more lower components are removed, the more harmonicity is lost. A non-periodic sound, by contrast, has no discernible repetition of frequencies.

Additive synthesis can be calculated for a buffer where predetermined values are stored in an array. An array is an ordered series of values consisting of an index value and a phase value. In the time domain, index value would be a position on the X-axis and phase would be the position of the cycle at a given time. Additive synthesis will be explored in the methodology chapter as a method to generate the classic waveform shapes.

Methodology:

Research into virtual analog emulations has been extensive in recent years, the field of study has made great improvement in terms of timbre and spectral content when compared to analog circuitry. This has made it more difficult to distinguish the method of sound generation. With a strong connection established between the analog synthesizer enthusiast and their hardware, can the musician accurately distinguish an analog synthesizer from its digital counterpart?

The literature review discussed methods of creating oscillators and filters in both the analog and digital domain. In this methodology a dual development process is presented, a circuit design for an analog oscillator and filter inspired by Stanley Lunetta's work, and a digital synthesiser programmed in Max MSP presenting common topologies and algorithms often associated with digital oscillators and filters. The resulting frequency output of the builds is then analysed for the existence of a "warmth characteristic" often associated with analog synthesis.

An A/B style listening test was carried out to help determine if the participants could detect this characteristic, as well as being presented with a glossary of adjectives sometimes used to describe an audio source's timbral qualities. Each participant was asked a series of questions to determine their prior synthesiser experience and split them into several demographics, those who prefer analog synthesis and those who prefer digital.

During the course of developing the methodology several members involved in the synthesiser community were contacted for a comment, this was done to gain the most recent insight into the community from the experts who document it. Robert Fantinatto co-wrote and directed "I Dream of Wires" and agreed to answer questions on the film. Questions and comments can be referenced in full in Appendix A.

Prototyping and Design - Analog Domain

An extensive prototyping stage took place early on in the project, with the purpose of successfully breadboarding a stable oscillator and filter for use as a sound source in the listening test. An important factors to note with the voltage controlled oscillator is the fact that they rely on ambient room temperature to regulate their tuning stability.[1] Since there is no definitive VCO based on the 40106 oscillator, a number of improvements can be made from the data sheet which shows a basic astable operation.[2] Suggested improvements and features found in off the shelf eurorack modules involve temperature compensation, Op-amp buffers, Op-amp amplifiers, weighted voltage divider inputs and outputs and lastly, control voltage for sequencing.

The voltage controlled oscillator:

The initial design for the oscillator was based around the 40106 Hex schmitt-trigger inverter. A hex schmitt-trigger is a comparator circuit that provides hysteresis. Hysteresis means that the input signal is compared with two voltage threshold values, a low and a high threshold,[3] To which the voltage threshold is set by the supply voltage of the 40106 IC, this is essentially a logic gate outputting a square wave. To turn this logic gate into an monostable oscillator only two components are needed, a resistor and a capacitor.

Figure 1 shows the basic astable oscillation provided by the manufacturer. It will be assumed that the capacitor (C) is fully discharged, meaning a voltage of 0v is at the input of the trigger. This meets the minimum threshold of the inverting input causing it to turn high, the output now registers 5 volts. The resistor (R) in this schematic prevents feedback as the voltage drop across the resistor charges the capacitor.[4] Once the capacitor is fully charged and the maximum pressure is met, it discharges to ground and therefore the input again meets the minimum threshold, producing a single cycle oscillation.

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[1]Schmitz, René. Voltage controlled oscillators, accessed November 2020
https://www.schmitzbits.de/vco2.html
[2] *CD40106B CMOS Hex Schmitt-Trigger Inverters Datasheet,* Texas Instrument, pp 15, November 1998
[3] Cadence PCB solutions. “Hysteresis in Analog Circuits: Comparator and Operational Amplifier Circuits.” *Cadence*, 2019.
[4] Hutchison Tyler , *Analog Circuit Design*, Matched resistor networks for precision amplifier applications, Chapter 415, pp 893-894, 2015

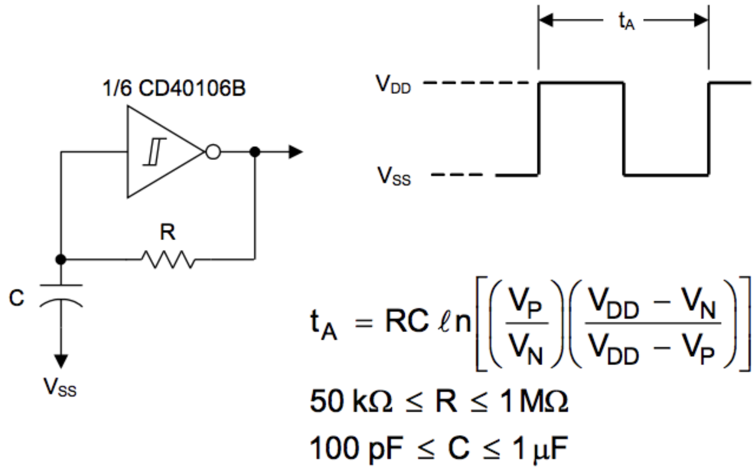


Figure 1. Astable schematic from the texas instruments data sheet.[5]

A slight modification can be made on the datasheet to produce a saw tooth waveshape. The discharge by the capacitor in figure 1 is instantaneous as it meets no resistance to ground. Modifying the same principle as above a diode (D1) is added in place of the resistor, this prevents negative feedback voltage.[6] Yet again the inverting input meets its low threshold and switches, the capacitor (C1) charges, but this time it will now flow to ground via a resistor (R1) when the maximum pressure is met at the inverting input. The capacitor now meets resistance and depending on the resistor value the discharge can be much slower, causing an instantaneous ramp up and gradual decline in our single cycle oscillation, Which can be seen in figure 2.

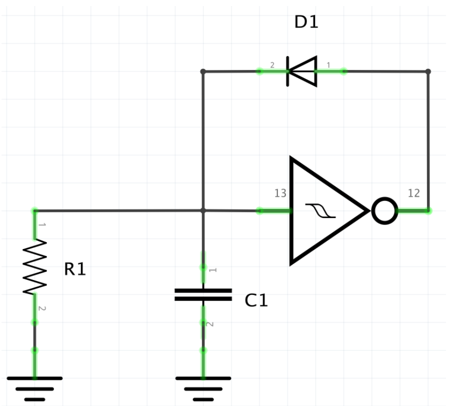


Figure 2. Astable oscillator modification.

[5] *CD40106B CMOS Hex Schmitt-Trigger Inverters Datasheet,* Texas Instrument, pp 15, November 1998

[6] Blom, Jim. "Diodes - Learn.Sparkfun.Com." *Sparkfun*, 2018, learn.sparkfun.com/tutorials/diodes/all.

The next step was to listen to the oscillation. Problems occur if a speaker is directly connected to the circuit. The most viable place to connect would be R1 in figure 2, however this provided another path to ground and the output would not be audible. To prevent loss of signal an Op-Amp buffer was used. In this case a voltage follower was used, which is an amplifier with a gain of 1. This does not offer an extra amplification of the signal, its task is to balance the impedance of the input signal to the output source's impedance.[7] Audio oscillators produce AC (Alternating Current) as they oscillate across a zero crossing. To properly patch the signal into a listening source an AC coupling circuit needs to be added to remove unwanted DC offset. DC components happen in the positive range, with the help of a capacitor the signal can be offset to be more audio friendly.[8] Just like a speaker cone it needs to push forwards and backwards to vibrate, without AC coupling the speaker would just push forward and cause unwanted clicks.[9]

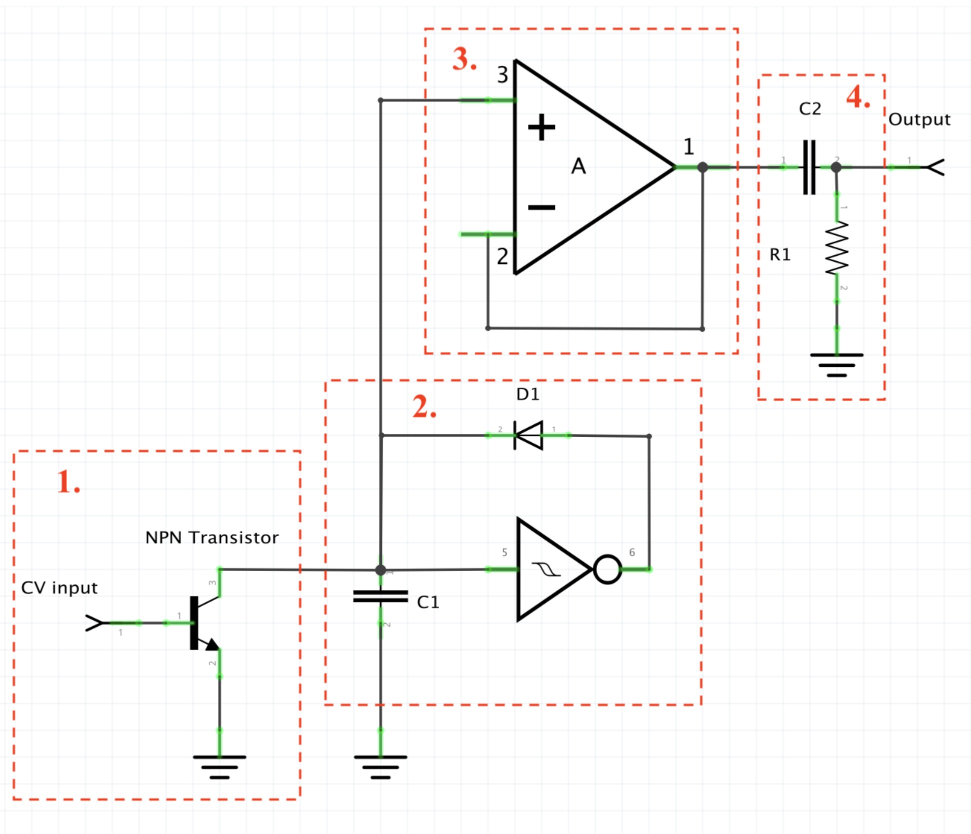


Figure 3.

1. Voltage controlled input, 2. Astable oscillator, 3. Non-Inverting amplifier, 4. AC Coupling.

[7] Robbins, Michael. “Op-Amp Voltage Buffer Chapter 7.2.” *Ultimate Electronics Book*, 30 July 2020,
[8] Jacquot, Blake. “AC Coupling.” *YouTube*, uploaded by Blake Jacquot, 2 July 2012,
[9] Marty. “How Do Speakers Work? A Speaker Guide For Everyone – With Diagrams.” *Sound Certified*, SoundCertified.com, 6 May 2021.

Adding voltage control to the circuit is relatively simple, with the addition of a transistor in place of R1 as seen in figure 3. The pressure regulated by the resistor as the capacitor drained to ground determined the pitch. By applying pressure to the base of the NPN transistor the speed at which the capacitor drains through the emitter can be changed, varying the pitch.

The limitations of the current iteration of the design is tuning stability. Up to this point the circuit has used foundational design methods and is quite easy to implement. The 40106 is oscillating and has variable voltage control, but is at the mercy of ambient room temperature. More stable functionality was added to the circuit when inspiration was taken from René Schmitz. His design takes the classic VCO approach and sums together the input control voltages to create an exponential current source used in volt per octave tuning standard, Accompanied by NTC thermistors to provide additional temperature stability.[10]

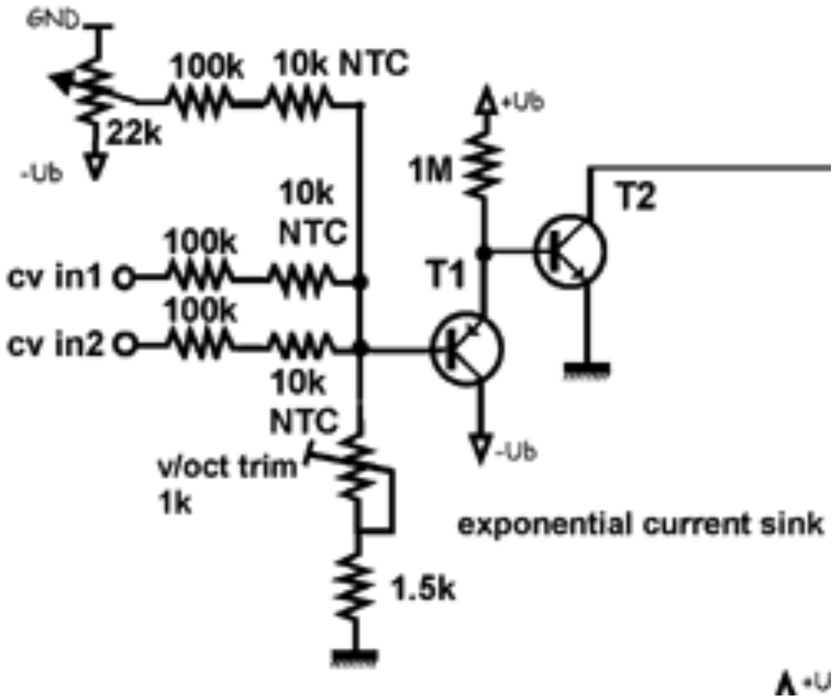
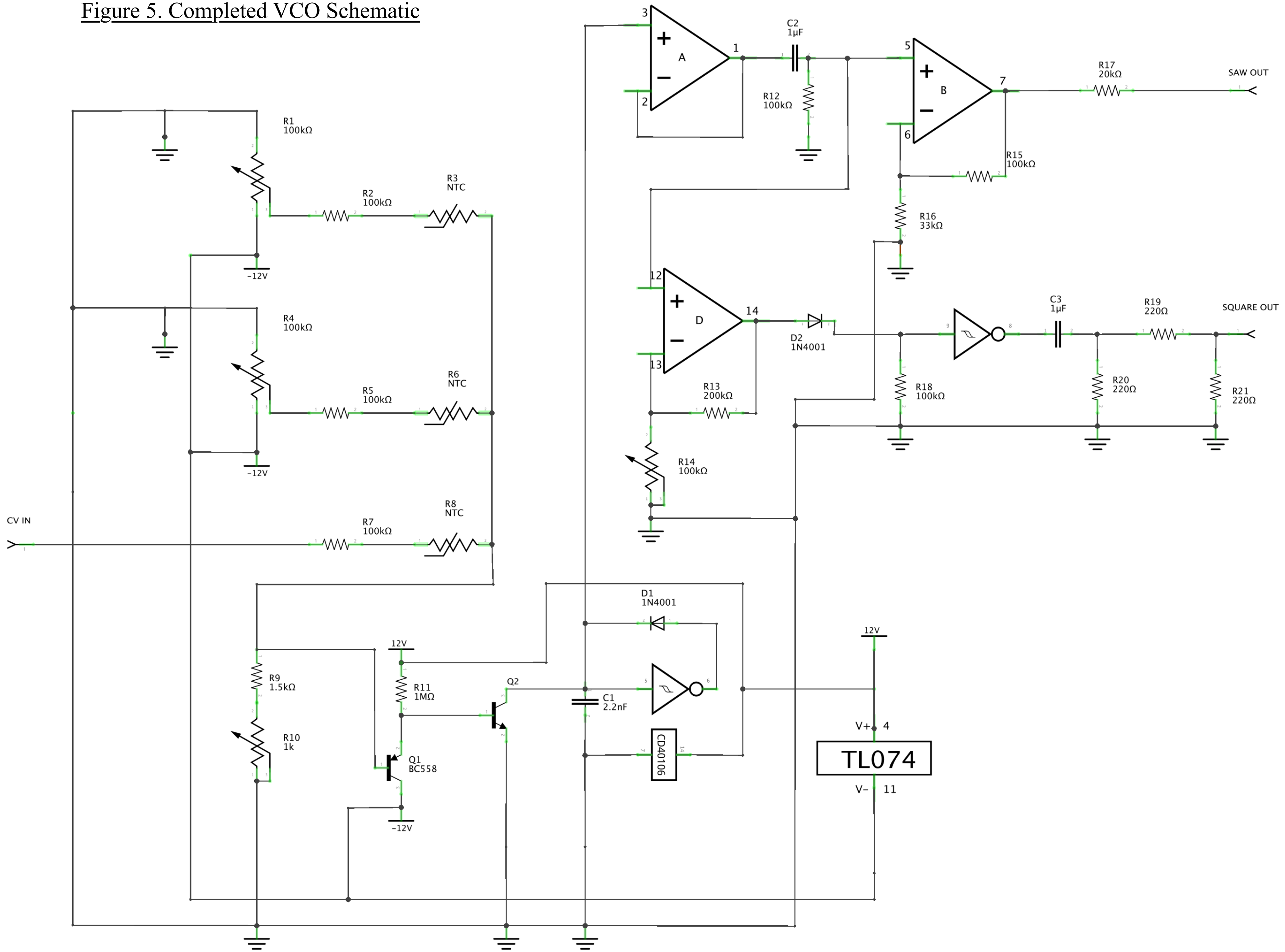


Figure 4. René Schmitz voltage control "current sink" This schematic can be used to control all desired functions, such as tuning stability, course and fine pitch, and even pulse width.

[10] Schmitz, René. "Synth Schematics VCO 4069." *Synth'R'us*, 2001.

Figure 5. Completed VCO Schematic



Low Pass filter :

The next step in the prototype and design phase was to construct a filter for use in the listening and comparison tests. Analog filters found in off the shelf synthesisers are incredibly complex and difficult to construct. The main goal for this design was to have a variable cutoff and produce musical resonance. The fundamentals of a low pass filter are however quite simple, as passive filters are used to filter frequency in almost every electronic circuit. A simple first order filter consists of one filter stage, this stage is made up of a resistor and a capacitor. The resistor restricts the flow of electrons and allows the capacitor to "fill" slowly, this in turn affects the waveshape as it passes through the circuit.[11]

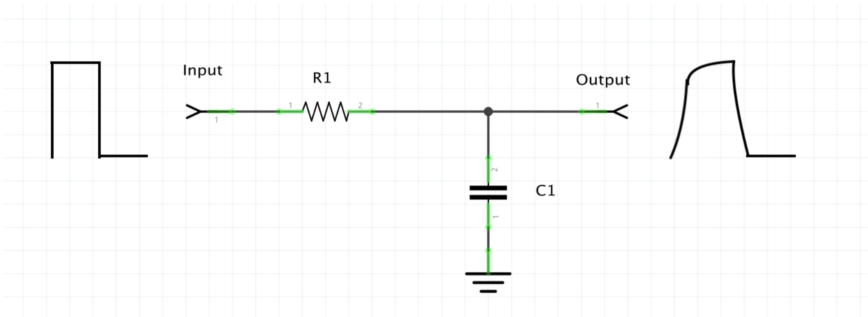


Figure 6. Passive low pass filter circuit, affecting the waveshape.
This will filter the harmonic content generated by a trivial square wave.

An ideal filter would have a brick wall cutoff frequency, with this first order design the decibel per octave range is poor and a large amount of harmonic content still passes. It is generally accepted that the above RC combination has a cutoff attenuation of -6dB per octave.

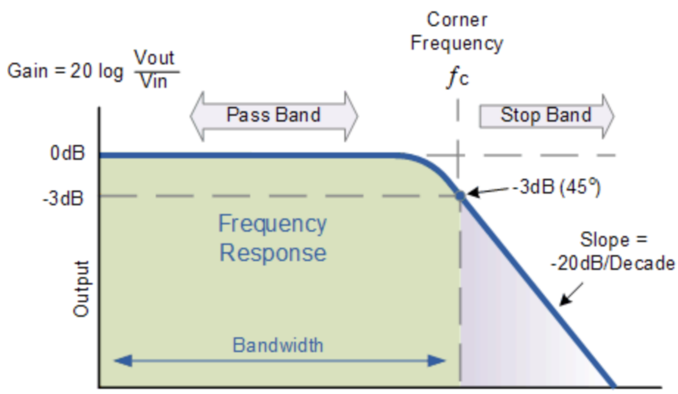


Figure 7. Cutoff slope[12]

[11] "Passive Low Pass Filter." *Electronics Tutorials*, AspenCore, www.electronics-tutorials.ws/filter/filter_2.html.
[12] "Frequency Response 1st Order Filter" *Electronics Tutorials*, AspenCore, www.electronics-tutorials.ws/filter/filter_2.html

To increase the attenuation a series of filters can be chained together to achieve the desired effect; increasing the slope. The limitation of increasing the filter order is volume, the resulting output filtered signal will be drastically quieter than then input. Knowledge gained from the VCO section can be put to use here. The addition of a buffer to the circuit can transform this design to an active filter. An op-amp works by differentiating the voltage levels at its two inputs and multiplying the difference by the op-amps gain value, the maximum voltage output is determined by the supply voltage. In this case the op-amp is utilizing negative feedback voltage as explained below;

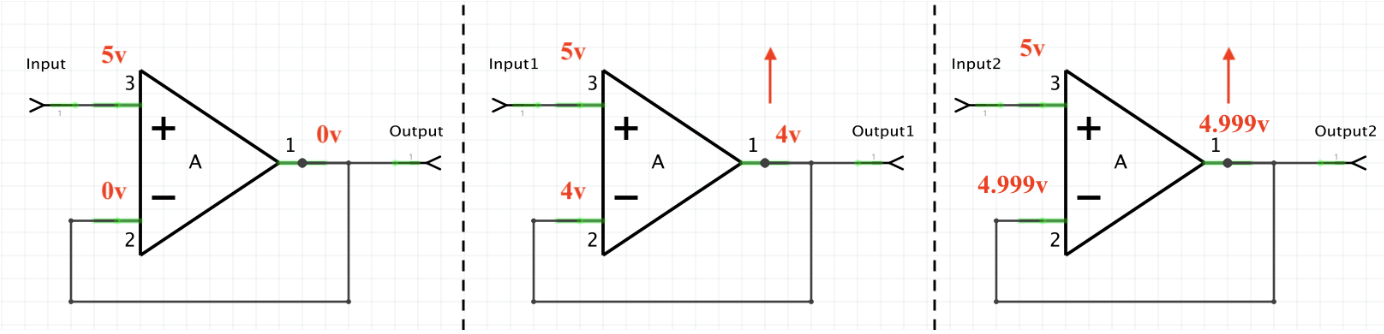


Figure 8. Showing the operation of a negative feedback buffer, and how over time the voltage level at its output matches its input.

The buffer's non-inverting input (Pin 3) probes a voltage for example 5 volts, while the inverting input (Pin 2) connects directly to the output (Pin 1) creating the negative feedback loop. Before the op-amp calculates the gain it will switch its output to 0v, meaning the inverting input and output start at the same voltage level. The difference between the inverting input and non-inverting input is then multiplied by the gain factor, increasing the output level. Since the inverting input is connected to the output this in turn increases. The differences between the two inputs begin to decrease, eventually meaning the output stabilizes slightly below the non-inverting input,[13] Creating a unity gain close to one. This process loops forever, keeping the voltage stable through negative feedback. After adding the buffer the filter schematic now performs as expected, with no volume issues.

[13] "Active Low Pass Filter" *Electronics Tutorials*, AspenCore, https://www.electronics-tutorials.ws/filter/filter_5.html

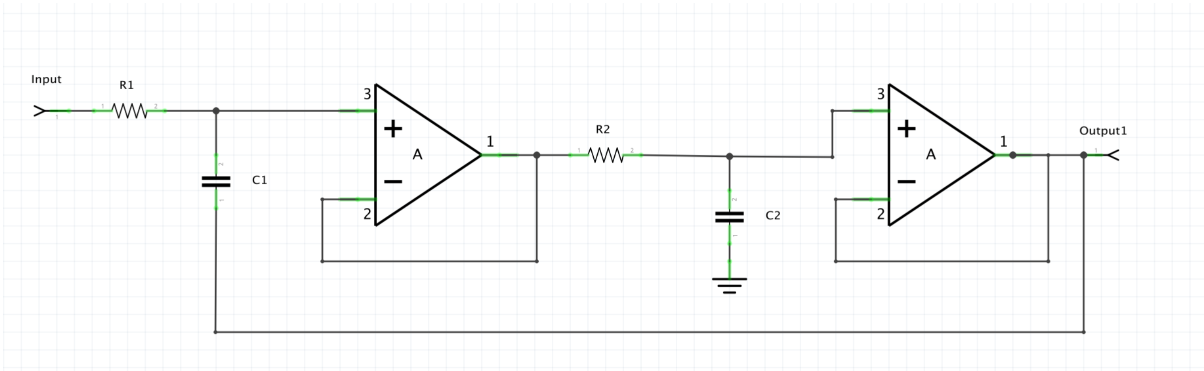


Figure 9. 2nd order low pass filter.

The last step in the filter design was to implement resonance control. In figure 9 above C1 looks to be connected incorrectly, as we learned in the passive design that the capacitor should drain to ground. However it is connected to the output of the second filter stage. This feedback voltage creates a cascading effect that drives the input into the first non-inverting input past 5 volts, increasing the peak of the signal, also known as resonance. This feedback design is a characteristic seen in digital FIR filters. To add control to this resonance peak, inspiration was taken from René Schmitz once again. Schmitz changed the clipping diode feedback loop to work with LEDs instead of schottky diodes used by Korg in its filters.[14]

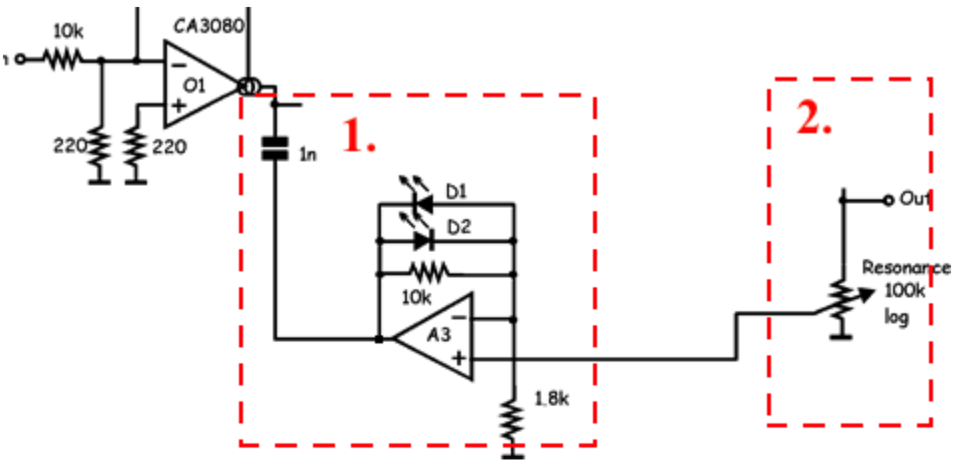
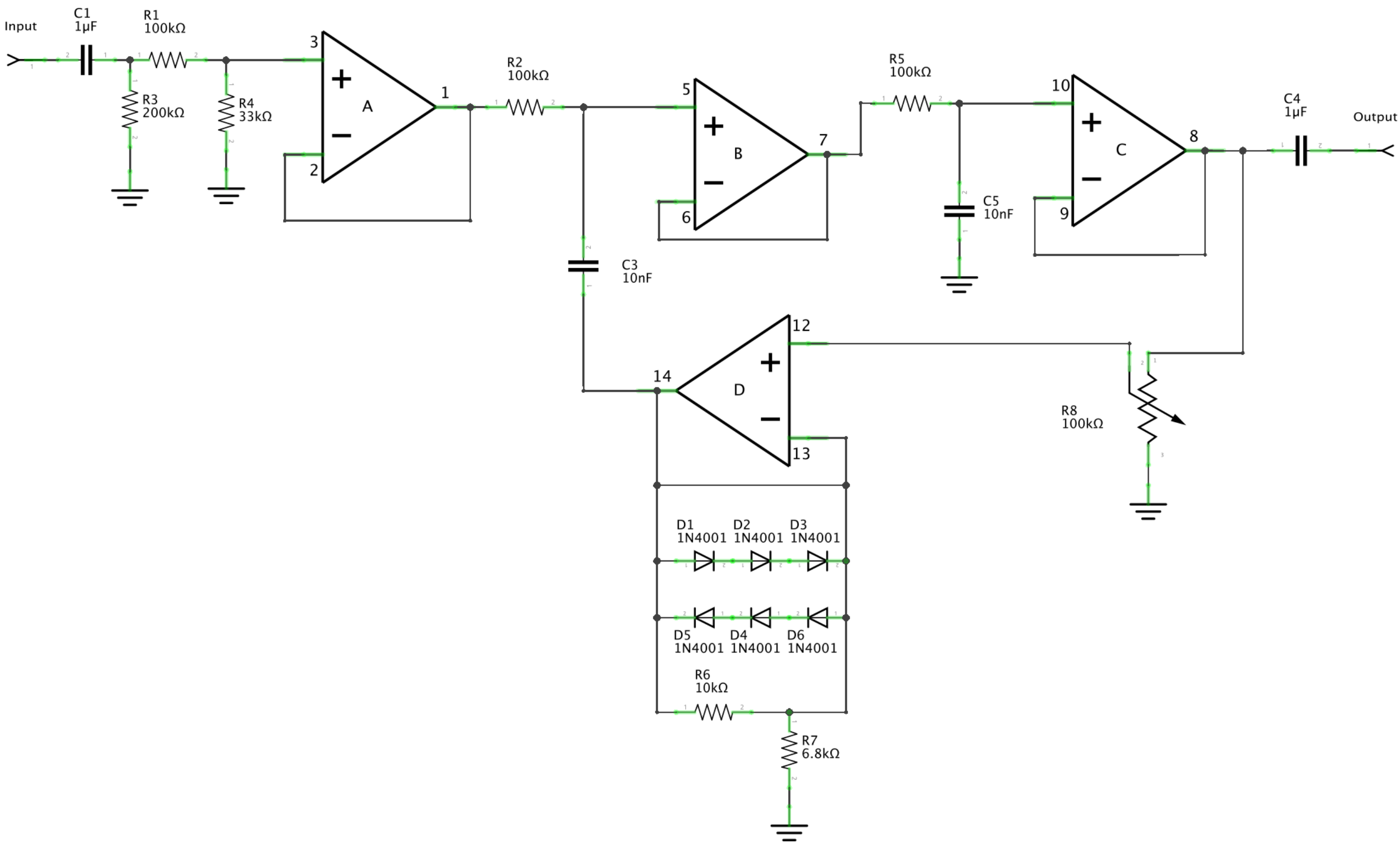


Figure 10. René Schmitz MS20 inspired design. 1, Diode feedback clipping for controlled resonance. 2, Resonance peak control.

[14] Chip. “Polysix Drive - Diode Distortion.” *Synth Hacker*, 16 May 2021, synthhacker.blogspot.com/2015/11/polysix-drive-diode-distortion.html.

Figure 11. Completed Filter Schematic



Limitations of Design:

The two designs highlighted above would not be considered perfect circuits. The VCO in its final iteration is still subject to ambient room temperature which makes tuning an issue. Advanced methods for creating a truly stable circuit are beyond the scope of this project, however, the VCO is musical enough for the purpose of the listening test. The filter is a simple implementation of a common active low pass filter. This design met the criteria of having variable cutoff and resonance control, however the resonance proved to be unwieldy in nature, and quite dominant in context of the mix. Even with the addition of the MS20 style clipping diodes, the resonance control only has a small working range before it starts to self oscillate and distort. The resulting builds are not ready for mass production but are suitable designs for the A/B listening test.

Unfortunately due to mistakes made along the way the circuits failed several attempts at being transferred to a printed circuit board. Much like the nature of prototyping, several of the order PCBs and the attempts at etching and milling boards at home failed. A case and knobs for the synthesiser were designed and 3D Printed. Photos of the attempts can be viewed in and 3D models can be viewed in Appendix B.

Prototyping and Design - Digital Domain:

The next step in the method was to patch a device in Max MSP utilising objects that come as standard. The requirement for the digital synthesiser was to replicate the waveshapes created by the analog synthesizer. The device was designed to have “hot swappable” oscillators and filters allowing for a variety of sound generation and filter methods to be explored. The desired methods had to produce a sawtooth and square wave as well as a second order filter. This assumption was made so that the style of subtractive synthesis was consistent between the analog and digital design.

The following is a list of objects that met the criteria for the study. Since this is not an exercise in creating an emulation from scratch, Max objects that already serve the purpose of being based on virtual analog algorithms and methods were chosen. The objects and patches used fall under the following categories: Virtual analog emulation, Anti-aliasing, Trivelly generated and second order filters.

List of Max objects used in synthesiser;

1. Rect~ is an anti-aliasing square wave oscillator, when combined with the slide~ object its characteristics can be altered to roughly emulate the capacitor discharge in the 40106 oscillator. This object also has pulse width modulation, having the same feature set as the 40106.
2. Saw~ is an anti-aliasing sawtooth oscillator.[15]
3. Gen Square~ is a customer built object that trivally draws a square wave. This is not a band limited oscillator. This was built to serve as the most recognisable digital sound, as historically digital oscillators contained aliasing components.[16]
4. Gen Saw~ is a customer built object that trivally draws a saw wave.
5. Cycle~ This object was used to create both a saw and square wave oscillator using additive synthesis. This method was not used in the listening test as it contains less harmonic content then the other methods and does not contain the "warmth" characteristic.
6. Biquad~ Implements a 2 pole filter
7. FIR filter in Gen~ this was used to create a second order low pass filter similar in sound to the analog filter design.
8. IIR filter in Gen~

The Max synthesiser acted as the base of all samples used in the listening test. Envelope, amplifier and MIDI control was also added to the instrument. Full screen shots of the patches can be viewed in Appendix C.

Analysing spectrum:

With the circuits and Max topologies completed, a method for analysing their spectrums and timbral qualities was designed. This will help compare the resulting waveforms and their effectiveness of creating "warmth" in the analysis chapter. The method for analysing the spectrum will involve looking at the resulting waveforms and viewing the frequency content with a FFT spectrogram. Each synthesiser was recorded with considerations for having consistent amplitude and frequency for both the analog and digital sources, this will allow for

[15] Max v8. 1.3 Documentation, Saw~, Accessed November 2020
[16] "Gen Overview - Max 8 Documentation." *Cycling74*, 2020,

accurate comparisons. The output was monitored to reach a level of -12db at a frequency of 1kHz at time of recording, a short audio sample or 1 to 2 seconds was required, this provided a healthy signal for analysis.

The resulting recording was then aurally examined for distortion and aliasing artefacts, given the correct sampling, aliasing should, in theory, be non-existing due to the non-linearities of the analog signal. However, varying rates of aliasing were expected in the Max implementations depending on the generation method.

Survey:

To determine whether or not the synthesizer enthusiast can hear the difference between analog and digital subtractive synthesis, a survey was devised. This survey was split into three key sections. First, word association for both analog and digital systems. Second, to determine the participants previous experience with synthesisers and their preferences. Third, an A/B style listening test asking the participants a series of questions on timbral attributes and identifying the sample's audio source.

Descriptive words associated with audio are often arbitrary, and several adjectives often share a similar dictionary definition.[17] When Robert Fantinatto was asked about his word association for analog synthesis he listed "warm","harmonic" and "unstable". For digital he listed "precise", "aggressive" and "cold". Previous research has referenced warm and cold as audio descriptors but sources on the other words provided by Robert proved difficult to reference, further reinforcing the arbitrary nature of audio word association. A decision was made to make a short glossary of words prefacing the survey, The adjectives most commonly found in research were removed of all suffixes. Examples "*Warmness*", "*Warmer*", "*Warmest*" would be stemmed to "Warm". Words relating to the volume or pitch of a sound were removed from consideration as to not persuade the listener, and words used to describe a musical attribute were also not considered examples include "*dynamic*" or "*expressive*". Words used to describe an audio effect such as "*Reverb"* and "*Delay"* were also disqualified.

[17] Pearce, Andy, et al. "Timbral Attributes for Sound Effect Library Searching." *Audio Engineering Society Conference on Semantic Audio*, 22 June 2017, p. 2.

The following is a list of descriptive words associated with audio and their sources in literature. The words were split into an analog category and digital category. The last question of the survey will present all the words to the participant accompanied by an audio sample. This list of words from now on will be referred to as the "*Glossary of words*";

Word:	**Category:**	**Source:**
Warm	Analog	Robjohns, Hugh[18] Fantinatto, Robert Williams, Ducan, et al[19] Pearce, Andy, et al[20]
Clean	Digital	Robjohns, Hugh
Natural	Analog	McGowan, Paul[21]
Harsh	Digital	Robjohns, Hugh Williams, Ducan, et al
Dry	Digital	Williams, Ducan, et al
Thick	Analog	Robjohns, Hugh Williams, Ducan, et al
Sterile	Digital	Robjohns, Hugh
Distorted	Analog	Robjohns, Hugh
Round	Analog	Williams, Ducan, et al
Crisp	Digital	Robjohns, Hugh
Thin	Digital	Williams, Ducan, et al

The next section of the survey was to split the participants into two demographics, people whose preference is analog synthesis and peoples whose preference is digital synthesis. To do this, the survey was distributed to musicians via synthesiser associated subreddits and forums. The moderators of r/synthdiy, r/synthesisers, r/synthesizercirclejerk and r/softsynth were contacted and asked to facilitate the survey request, ensuring that the sample pool consisted mostly of synthesiser enthusiasts from all backgrounds. The questions asked were designed to align the participants with analog or digital synthesis, sample question "Do you own a hardware synthesiser?" If yes, a logic system would redirect the participant to divulge information into what type of hardware they own, if they have a preference if so then why? If no, participants would be moved onto the following question; "Do you use any "softsynths"

[18] Robjohns, Hugh. "Analogue Warmth." *Sound On Sound*, SOS Publications Group, Feb. 2010.
[19] Brookes, Tim. "Perceptually-Motivated Audio Morphing: Warmth." *AES E-Library*, 1 May 2010.
[20] Pearce, Andy, et al. "Timbral Attributes for Sound Effect Library Searching." *AES E-Library*, 13 June 2017.
[21] McGowan, Paul. "Natural Sounding – PS Audio." *Psaudio*, 2 Dec. 2018.

or plugins on your computer?” This was done to ensure everyone involved has some prior knowledge, participants who could not answer were excluded from the results.

The following flowchart in figure 12. shows the questionnaire logic in action. The red lines progress the participants towards the listening test when they align with digital synthesis, While those who prefer analog progress through the questionnaire.

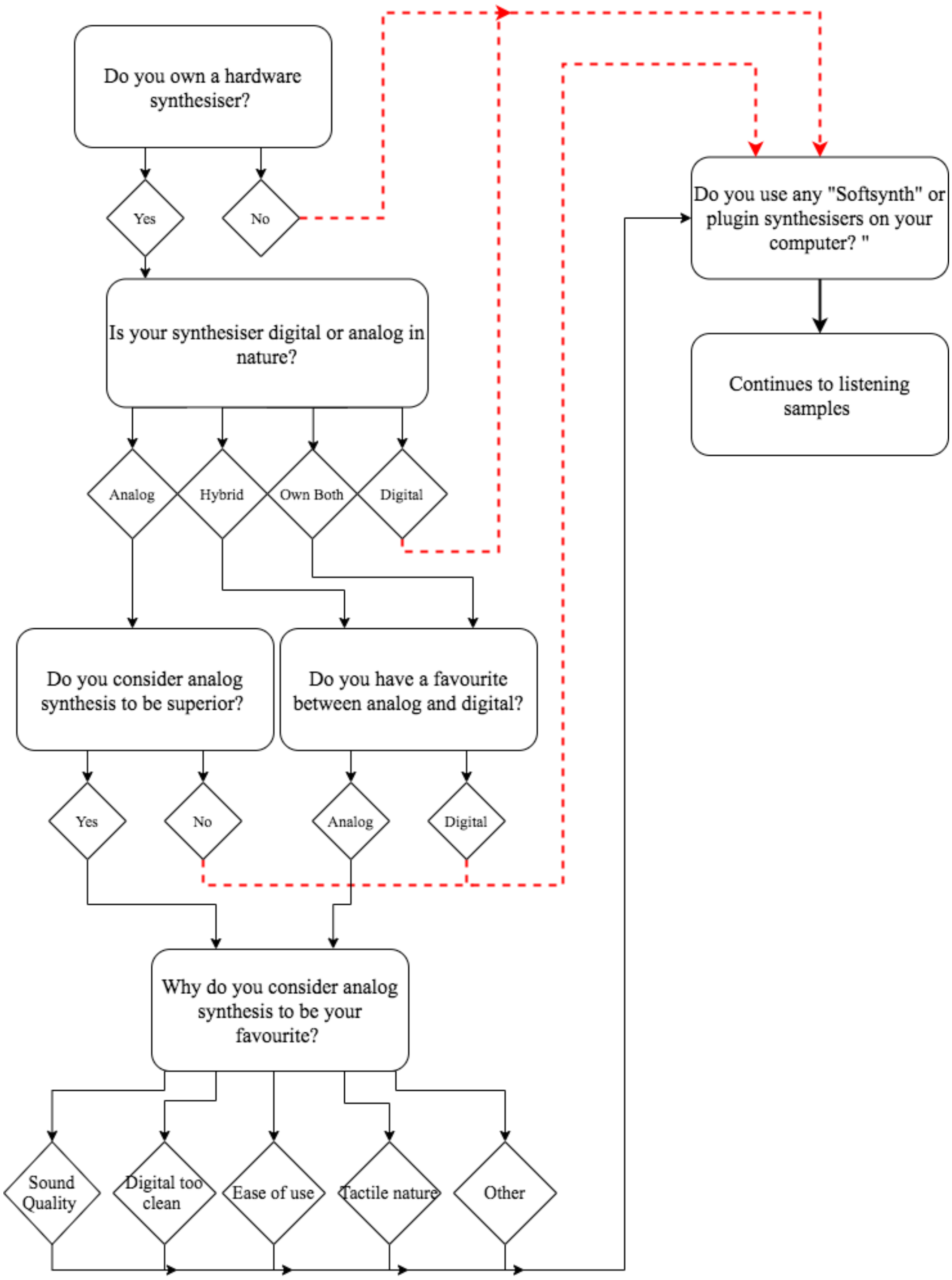


Figure 12. Questionnaire logic separates the participants into those who prefer digital versus those who prefer analog.

The logic was inspired by responses from Robert Fantinatto, when asked why he thinks the musicians in his documentary prefer analog synthesis he said:

> "I think it is more an appreciation of the tactile nature of the interface and less so about the purity of the analog signal"

Lastly, the participants answered the listening test. This section of the survey was designed to bluntly ask what the method of sound generation of the provided sample is. The same method for recording samples for spectrum analysis was followed. The participants were also asked to link the most commonly found words from the glossary to a sample. The literature review highlighted academic sources for what warmth actually means in the context of audio samples. The following questions were asked to determine if an analog or a digital signal inherently possesses the warmth characteristic and if so, is it audible. The audio sources for the listening test were provided by the Max implementations, the 40106 synthesiser and Arturia microbrute synthesiser. The Arturia is an off the shelf analog subtractive synthesiser.

List of questions provided to participants an accompanying sample:

Question:	**Sample:**	**Source:**	**Pulse Width**
Which sample sounds more "warm" to you?	Sample 1: Analog	40106 Square	25%
	Sample 2: Digital	Max rect~ with slide~ parameter	25%
Which sample sounds more "Distorted"?	Sample 1: Digital	Max Saw~	0%
	Sample 2: Analog	40106 Saw	0%
Which sample do you prefer?	Sample 1: Analog	Microbrute Square	50%
	Sample 2: Digital	Max rect~	50%
Which sample do you think is from a Digital synthesiser?	Sample 1: Digital	Max Gen~ Trivial Square	50%
	Sample 2: Analog	40106 Square	50%
Which sample do you think is from an Analog synthesiser?	Sample 1: Digital	Max Saw~ with filter sweep	0%
	Sample 2: Analog	Microbrute Saw w/ filter	0%
Which filter sweep	Sample 1: Digital	Max Saw~ w/ filter	0%

sounds “Fatter”?	Sample 2: Analog	Microbrute Saw with filter sweep	0%

Care was taken to ensure that the volume matched between samples, pulse width of the cycles matched and that the filter sweeps had the same dB per octave attenuation, resonance, the same starting point in time, and the same frequency.

It is hypothesized that the more information provided to the participant, the higher the chances are of the recordings being interpreted negatively, reinforcing an individual's existing beliefs[22]. When trying to ascertain if participants can notice the difference between an analog signal versus a digital signal, all prior thoughts and theories should be removed by introducing blind testing, allowing for qualitative results over personal opinions. Each sample is named ambiguously, and can be downloaded. No metadata can be derived from the records to influence a person's decision.

The designed survey is limited to discovering the participants’ reasons for preferring analog synthesis, if they in fact do so. If they choose digital synthesis they get rushed onto the listening test, creating a gap in the results with no quantifiable reason as to why they prefer digital synthesis.

[22] Giroux, Megan E., et al. “Confirmation Bias, and Degraded Audio Recordings.” *OSF,* 22 Jan. 2020. Web.

Analysis:

Throughout the course of research and carrying out the methodology, one word kept cropping up when analog or virtual analog emulations were discussed and this was “warmth”. The definition for warmth was discovered to be the spectral content 3.5 times the fundamental frequency. With the two builds complete and samples recorded following the procedure laid out in the methodology, it's time to see if this warmth region can be identified.

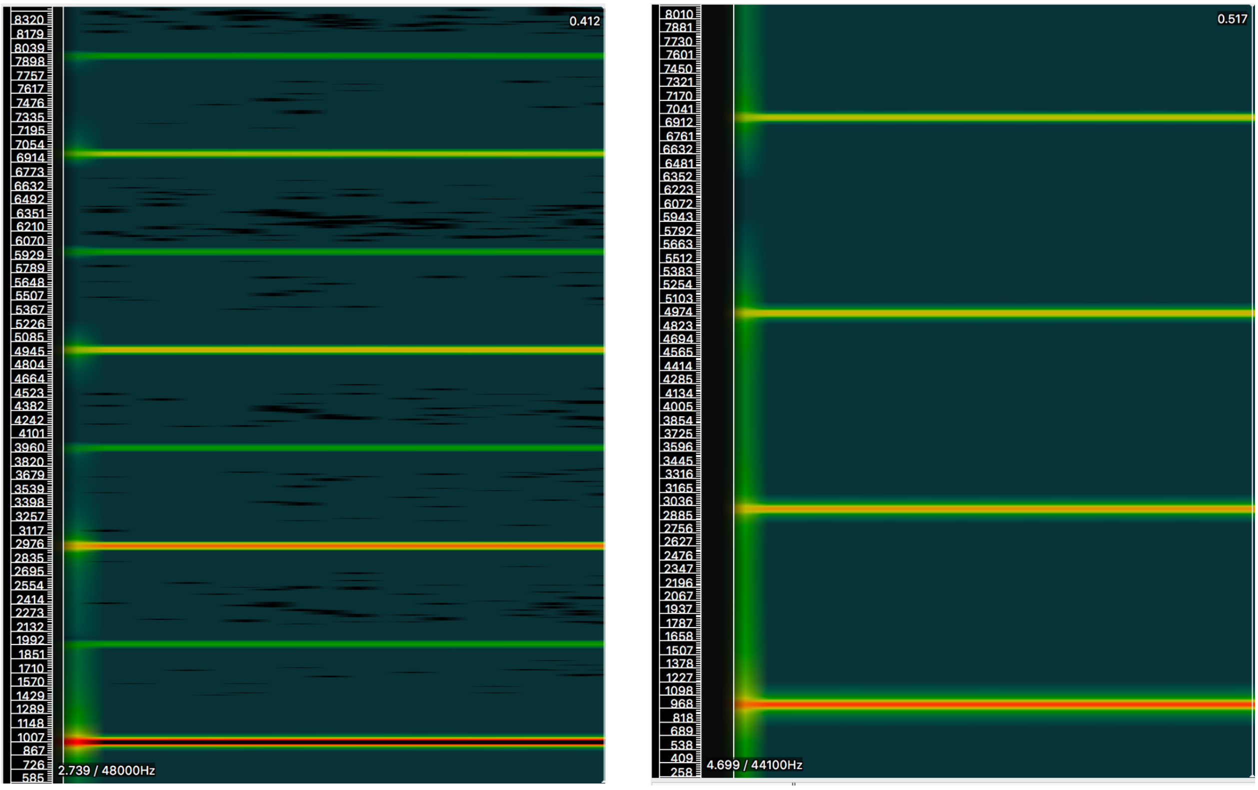


Figure 13. 40106 square oscillator on the left, Max Rect~ oscillator on the right

The above figure shows two screenshots from sonic visualiser.[23] using the spectrogram function the spectral content of the oscillators can be seen. The fundamental frequency at 1kHz is seen as the thick red line in both images. If these oscillators possess the warmth characteristic, extra harmonic content would be viewed between 3kHz and 4kHz. However there is no extra harmonic content in either implementations. Interestingly both implementations have varying amounts of harmonics. Waveforms can be approximated by a

[23] “Sonic Visualiser: A Brief Reference 6.3 Spectrogram.” *Sonicvisualiser*, www.sonicvisualiser.org/doc/reference/4.3/en/#spectrogram. Accessed 12 May 2021.

series of convenient formulas: A square wave consists of only odd-numbered sine wave components with a phase of 0 degree:

Fundamental 1/1
2nd harmonic 0
3rd harmonic 1/3
4th harmonic 0
5th harmonic 1/5
6th harmonic 0
7th harmonic 1/7

The rect~ object follows this rule and exhibits no other form of colouration, this may be down to the method of generation happening under the hood of Max. It is possible Max uses either BLIT or BLEP algorithms, which create a series of bandlimited impulse responses, integrated together over time. This creates an ideal low pass filter, filtering out any unwanted harmonics.[24] However, the 40106 square wave operating in the time domain has extra harmonic content, possibly due to the non-linearity of analog circuits.

Perhaps the warmth characteristic comes from the filter stage? The 40106 was filtered through the analog filter and examined. It showed a slight addition of harmonic content in the range expected, between 3kHz and 4kHz when the cut off frequency passes by. Figure 13 shows a state zero of the analog synthesiser with a resonant peak, while figure 14 shows the filter sweep and added harmonic components.

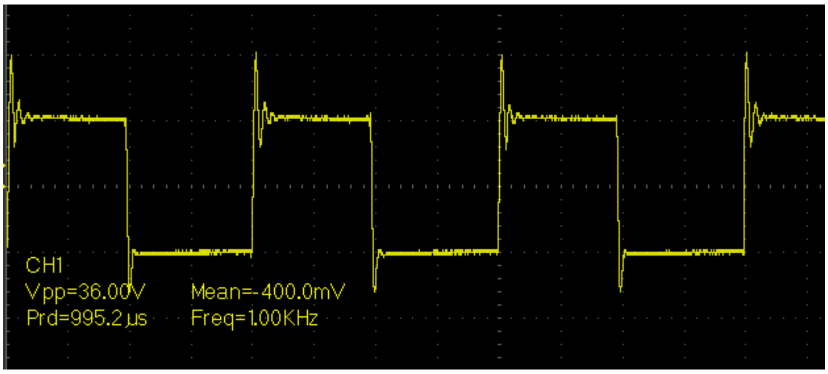


Figure 13. State zero analog synthesiser.

[24] Phelankane. "All About Digital Oscillators Part 2 – BLITS & BLEPS." *Meta Function*, 4 Dec. 2020, www.metafunction.co.uk/post/all-about-digital-oscillators-part-2-blits-bleps.

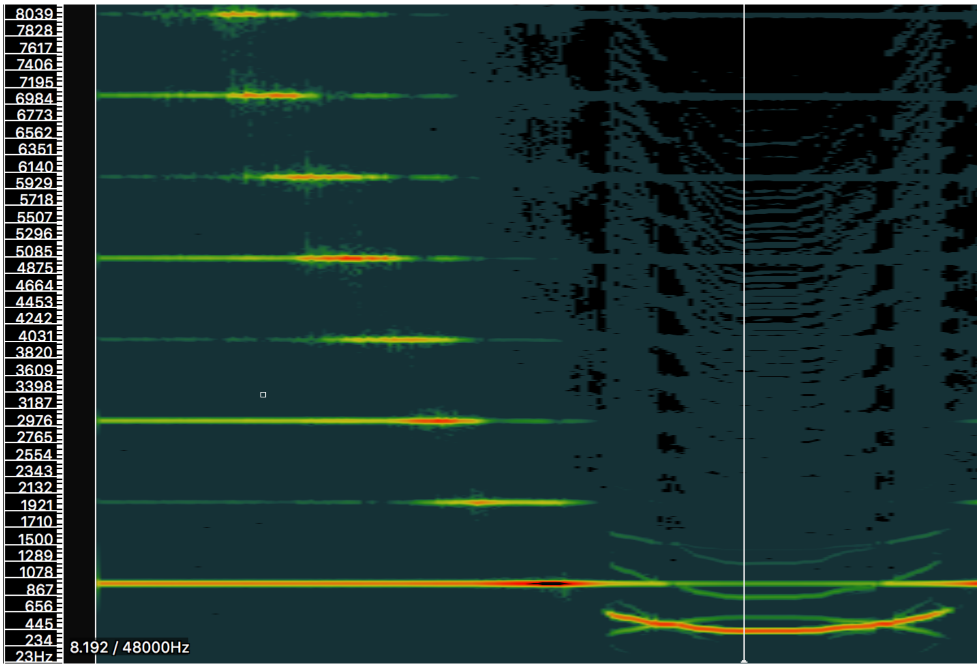


Figure 14. Filter sweep, added harmonic content around filter cutoff.

Neither the analog or digital emulations created large amounts of the "warmth" characteristic, perhaps this is added to the signal at different stages of the chain. The implementations used were designed to create classic waveshapes, and consideration of the methods used to create harmonic distortion were not considered. The other objects in Max behaved exactly as they were designed to do by the developers and produced no extra aliasing artefacts. The trivally drawn waveshapes did, as expected, contain aliasing artefacts.

Survey analysis:

The following survey results highlight the findings from the online survey distributed to musicians on the 16th of April and concluded on the 26th of April. Musicians were contacted through synthesiser associated subreddits and students studying music related disciplines in further education institutes. The total number of respondents was 94. When the data was examined each respondent needed to meet the following criteria: 1. Agree to ethics approval, 2. Have experience with hardware or software synthesisers, 3. Complete the survey to the end. Respondents who did not meet all the required criteria were removed from the final results, leaving the survey with a total of 57 total respondents, a completion rate of 60.6%. For access to the raw and filtered data please see Appendix D.

Question 1 asked the participants to select words from the glossary that they would associate with analog synthesis with a total of 125 selections. Research predicted that "warm" would be the prominent choice, and it was, with 31% of all respondents selecting the adjective. There was an even percentage spread seen for each of the other adjectives. However, this group featured the largest amount of positively associated words used to describe analog synthesis such as Natural (16.80%) and Thick (16.80%) boasting positive nostalgia. The word "warm" is the most associated word with analog synthesis.

Glossary:	Percentage:	Number of Sections:
Warm	31.20%	39
Natural	16.80%	21
Thick	16.80%	21
Distorted	12.80%	16
Round	16.80%	21
None of the above	5.60%	7
Other	0.00%	0

Question 2 presented the glossary of words most commonly associated with Digital synthesis with 116 selections in total. The adjective "clean" received 32% of all selections by a larger margin when compared to question 1. The intended attribute "Harsh" which was most commonly found in literature received one of the lowest scores with 6.9%, Suggesting opinions have changed over recent years. The second most common attribute was "Sterile" with 18.9%. "Clean" and "sterile" being directly correlated by dictionary definition means that over 50% of all selections associate digital with a clean or in other words pristine signal. Break down of the selections can be seen in the table below.

Glossary:	Percentage:	Number of Sections:
Clean	32.76%	38
Harsh	6.90%	8
Dry	12.07%	14
Sterile	18.97%	22
Crisp	19.83%	23
None of the above	6.90%	8
Other	2.59%	3

The next set of questions was designed to separate participants into two key demographics, those who align with Analog synthesis and those who align with Digital synthesis. Referencing the logic flow chart from earlier, the results have now been overlaid, this demonstrates how participants were filtered into the two demographics.

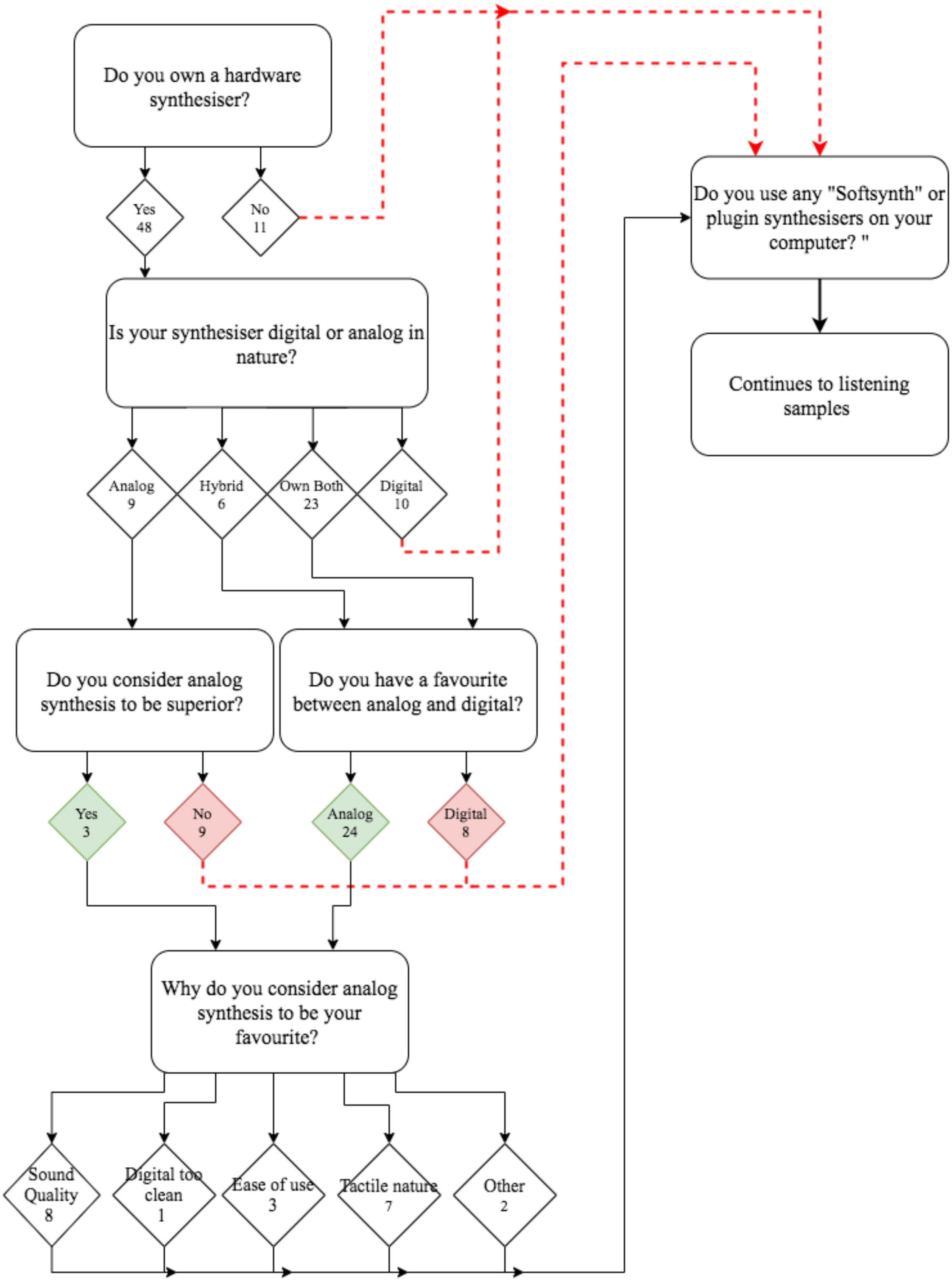


Figure 15. Flow chart with selection results numbered.

Out of all the participants, 81% owned some form of hardware equipment. This was further broken down into different types of hardware. Participants were split between Analog (18.7%) Digital (20.8%) Hybrid (12.5%) and those who owned both analog and digital (47.8%), meaning the majority of those who answered own analog equipment. Unsurprisingly 81% of all participants have also used software synthesisers on their computer.

Next, the logic moved the participants onto the questions, “Do you consider analog synthesis to be superior?” and “Do you have a favourite between Analog and Digital?”. It can be derived that of the participants who have a preferred form of synthesis that 62% preferred analog synthesis prior to taking the listening test. The reasons cited as to why were broken down into the following:

Selection:	Percentage:	Number of selections:
Sounds quality	38.10%	8
Digital sounds too clean	4.76%	1
Ease of use	14.29%	3
Tactile nature	33.33%	7
Other	9.52%	2

Other answers: More technically interesting to me, More fun to build.

The results here specifically line up with insights provided by Robert Fantinatto when he was asked “Why do you think that the musicians interviewed in the documentary prefer analog synthesis?”

“I think it is more an appreciation of the tactile nature of the interface and less so about the purity of the analog signal, there are many modules that are digital but the controls are directly physical along with the ability to interconnect as the user pleases.”

With the key demographics determined, all the participants were moved onto the listening section. A key take away to remember so far is that 62% of participants prefer analog synthesis. The listening test consisted of 6 A/B style questions asking the participants to distinguish one sample from another, and 1 question of glossary association to an audio sample.

Each question in the listening section presented two samples playing a melody, please refer to methodology for information on which method of generation matches which particular sample. The people who listened to question 1 could not determine what “warm” was, warmth does not have an inherent sound or is difficult for the participants to determine, with a close to even 50/50 split.

On question 2 participants determined that the digital sample sounded more distorted, Digital 59.6% to Analogs 40.4%. However, the Max Square was shown to have less harmonic content than the 40106 square due to its method of anti-aliasing generation. On question 3 when asked which sample they prefer, two thirds of the participants chose the analog sample over the digital.

Question 4 asked which sample is from a digital synthesiser. 61.4% chose the digital sample in the blind listening test. This sample was generated with the trivial oscillator, meaning it produces high amounts of aliasing artefacts. Most synthesis enthusiasts are familiar with the sound of aliasing artifacts, possibly due to the fact that most participants have used software synthesisers that come in varying degrees of quality.

Question 5 “Which sample is from an analog synthesiser?” The results from this question proved inconclusive as the selection was, evening split, Even with the majority of participants preferring analog synthesizer.

The final question asked the users to select words from the glossary they best felt matched its timbre. The recorded sample came from a performance on the 40106 synthesiser. It found that “warm”, “distorted” and “thick” made up the majority of selections, the most common words associated with analog synthesis being;

Selection:	%	Number of selections:		Selection:	%:	Number of selections:
Warm	18.90%	31		Sterile	0.61%	1
Clean	3.66%	6		Distorted	21.34%	35
Natural	3.05%	5		Round	10.37%	17
Harsh	10.98%	18		Crisp	6.71%	11
Dry	6.71%	11		Thin	3.05%	5
Thick	12.80%	21		Other	1.83%	3

Discussion:

What the results show us is that most synthesiser users given the option prefer Analog synthesis to Digital. However in a blind listening test the participants were unsuccessful in picking out which sample was analog. The reasons cited as to why the users preferred analog relate back to the work done by Robert Fantinatto, who had this comment to make;

"I believe most listeners of music can't tell the difference whether the music was created using analog or digital synthesis, but it does affect the musician and their ability to form a personal connection with their instrument, that I believe will never change."

The personal connection is a key point here, synthesiser enthusiasts identify with a medium based on its functionality and how they feel in a particular moment, as seen by participants mentioning "vibe" and current "state of mind" for their rationale behind synthesiser choice. It is a fluid decision and not based on signal purity or warmth. Robert Fantinatto also went onto comment;

"I think it is more an appreciation of the tactile nature of the interface and less so about the purity of the analog signal, there are many modules that are digital but the controls are directly physical along with the ability to interconnect as the user pleases."

Participants were also given the choice to weigh in on the survey and make a comment on their preference, some key takeaways relating to the appreciation of synthesis are displayed below;

1. Most of the time I can't really tell the difference, also my sound preferences regarding analogue/digital depend on the instrument. Quality analogue drum machines definitely have a unique quality that I feel I can pick out.
2. I'm a big fan of analog synthesizers. Not only because of the warm nostalgic sound but because of the aesthetic look. It reminds me of the 80s-90s. Adore it. Moreover, analog synthesizers provide a really unique sound and feel. That's why I believe my favorite music producer Mick Gordon doesn't use digital one.
3. Both can sound thin or thick depending on usage. I slightly prefer analog since I'm not a fan of stepping filters or oscillators, but digital is far more versatile. Very very few people would hear the difference in a mix.

4. Digital synthesis offers more features per dollar spent, and is free from the limitations of analog circuitry and physical space. It trades the character-defining noise and harmonic distortion of analog synthesis for aliasing, which can be mitigated to inaudible levels with the right algorithms
5. No preference as far as sound goes.
   Digital : complete precision and not having to deal with noise is nice. Analog: playing is satisfyingly direct - eg. When you move a filter cut-off control you are physically coupled to the sound generating circuit, there's no layer of abstraction sampling your movements into digitized control parameters.

When the survey was first submitted to Reddit one user spotted a mistake in the way the survey asked the question about which words the user associated with analog or digital synthesis. The question was not worded the same for both, the user suggested the question be unified to prevent bias. On top of this reddit proved to provide a harsh critique, with users arguing over the finite details for the survey. Suggestions were taken on board and the survey was edited and relaunched.

The logic in the listening test served the purpose of identifying those who preferred analog synthesis and why, however it failed to identify why some may prefer digital. If this project was launched again research would help detail why users prefer digital by interviewing relevant people in the field. There is a gap in the data due to this and further research could be carried out to rectify this.

Some of the words associated with either form of synthesis have negative meanings in the music industry and are not desirable traits, words such as “harsh” which was used in the glossary for digital synthesis was selected the least amount of time. Perhaps the use of negative words alienated the digital synthesiser fanbase. In the end all samples in the survey were digitised, this process may have had an effect on the resulting samples, and this contingency was not planned for. The ideal scenario would be to perform the survey in person in a treated room. This would mean the analog signals do not undergo analog to digital conversion.

The project highlighted that analog synthesisers do not contain the “warmth” characteristic just by association, and that it is possibly down to other factors not looked at in this project.

While “warm” is proven to be the most commonly associated word with analog synthesis it dont not have an inherent sound.

The project has successfully answered the research question, the average user can not hear the difference between analog and digital subtractive synthesis.

<u>Conclusion:</u>

In conclusion the methodology set out to design methods of sound generation in the analog and digital domain, successfully creating classic subtractive synthesis waveforms. Further research is needed in order to create a more commercially viable analog synthesizer that can reproduce the characteristics associated with it. Similarly, research into the field of virtual analog synthesisers in the digital domain is currently ongoing and future developments will no doubt further blur the line between the analog and digital domain. Soon the audible difference between analog and digital synthesis will be finite, and audio quality will no longer be the main selling point. Instead users may prioritise their relationships to the instruments and their tactile nature. Digital emulations are possible, and the analog model produced the desired characteristics, according to the survey.

The Analog vs Digital is a purist issue in some circles. It transcends the fact that the basic function of instruments is to make music. If making music is all that counts, analog and digital synthesis are just tools to be used.

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Appendix A. Interview

1. What inspired you to make "I dream of wires"?

I used to play electronic music in the 80s as an amateur musician. I still had my old analogue synthesizers but was not active in music for many years. I had only seen photos of modular synthesizers on the covers of albums such as Switched-On Bach, but never had any direct experience with them. My oldest son shared an image of a modern-day modular synthesizer with me and I was immediately intrigued. I had no idea that what seemed to me to be an archaic and obsolete technology was still being used to make synthesizers. After looking into it a bit more I discovered there was a whole subculture of people who loved this stuff and there were many small companies making modules. I've always been interested in things that were supposedly obsolete coming back into the zeitgeist and why people are sometimes drawn to old technologies. I began interviewing the odd person about it and it slowly turned into a huge project that would go on for three years.

2. Why do you think that the musicians interviewed in the documentary prefer analog synthesis?

I think it is more an appreciation of the tactile nature of the interface and less so about the purity of the analog signal, there are many modules that are digital but the controls are directly physical along with the ability to interconnect as the user pleases.

3. What 3 words would you use to describe the sound of analog synthesis?

Warm, harmonic and unstable.

4. What 3 words would you use to describe digital synthesis?

Precise, aggressive and cold.

5. In "I Dream of Wires" musicians spoke about how they prefer to work without a laptop or digital instruments. Do you think there is a bias towards analog gear?

The general feeling attitude was the more time you can spend away from laptops the better as we spend so much of our time in front of screens. The analog interface is designed to do one thing only and there are no distractions, you can't check Facebook while you're in the middle of creating a patch.

6. Have you ever seen any studies on analog versus digital synthesis?

I'm not aware of any, my personal observation is that as long as the interface style is "analog" most people are fine mixing analog and digital sound synthesis techniques.

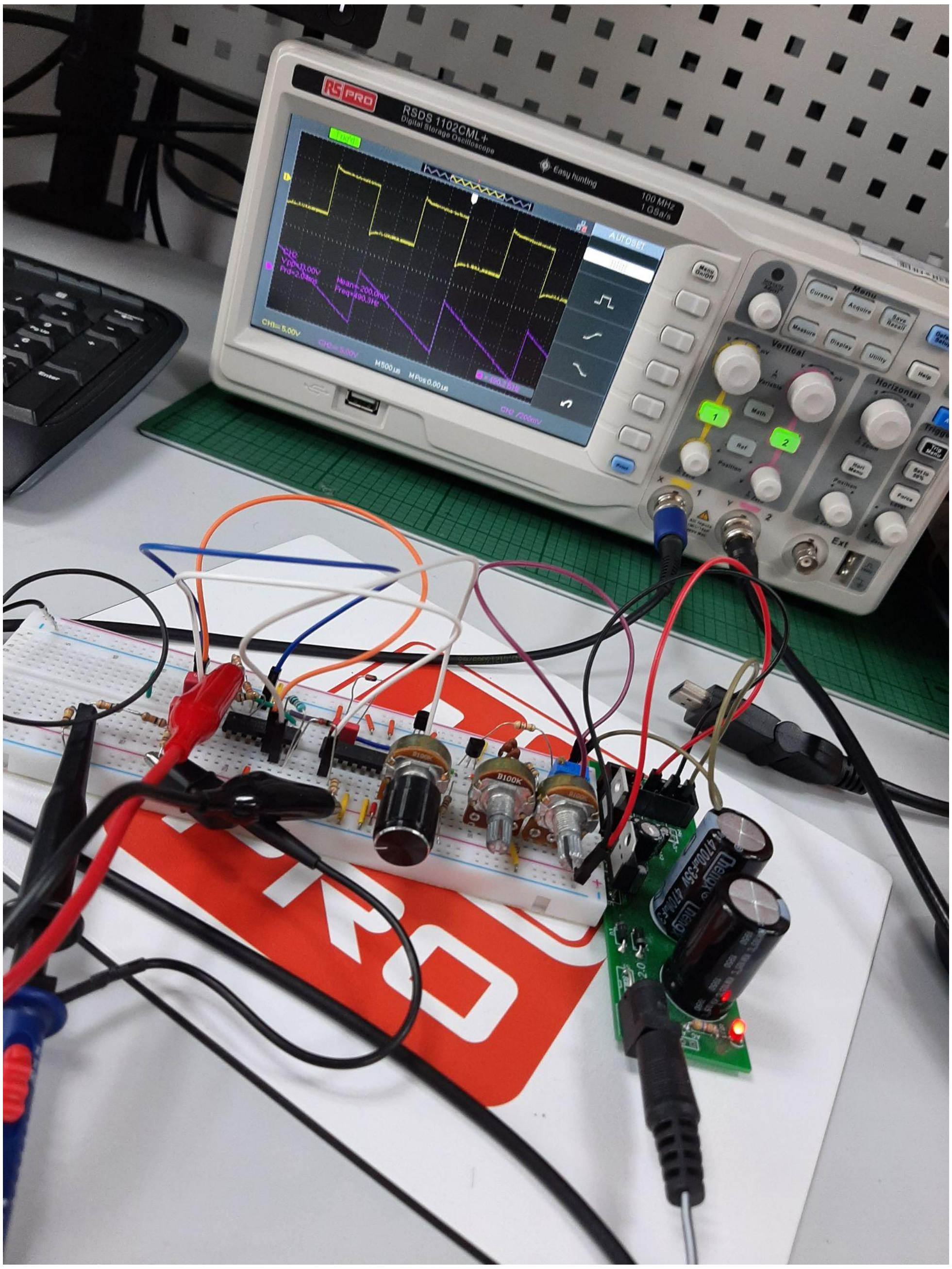
7. Five years on from the documentary virtual analog emulations have improved, soon enough the audible differences between analog and digital synthesis will be finite, if the sound is no longer the main selling point, what is?

I believe most listeners of music can't tell the difference whether the music was created using analog or digital synthesis, but it does affect the musician and their ability to form a personal connection with their instrument, that I believe will never change.

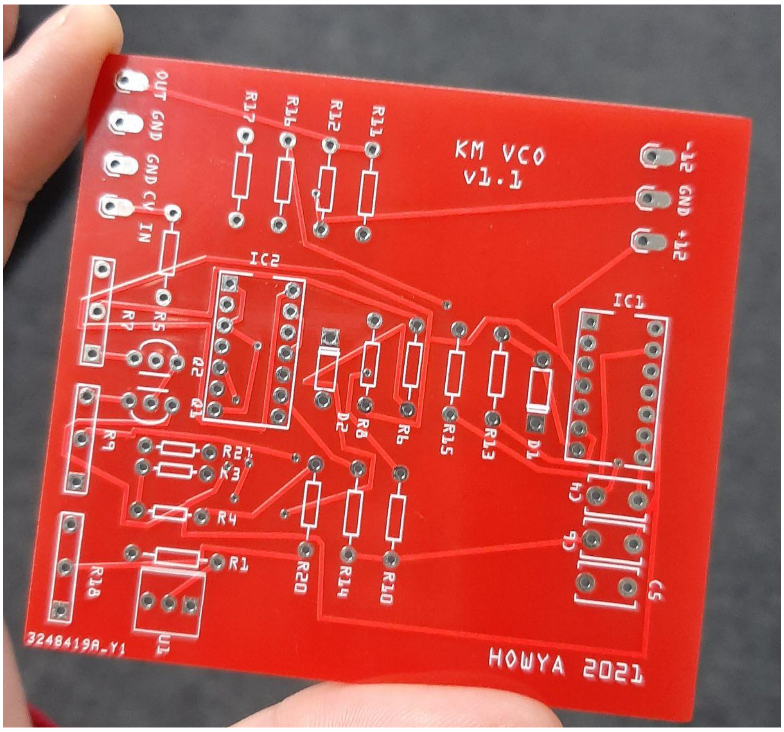
8. Do you think modular analog synthesisers will continue to see a rise in popularity over the coming years? Even with the competitive price and constant improvements in digital synthesisers.

Digital synthesizers will always be less expensive to develop, but I see more and more merging of digital and analog technologies, this is already happening. I think the ability to customize your own instrument using modules from many different manufacturers has been the most important factor in the return of this style of instrument. I think the Eurorack format will just continue to grow. At this point, large manufacturers like Roland are starting to offer modules which show how seriously they take what was once a very niche market

Appendix B Analog Build:

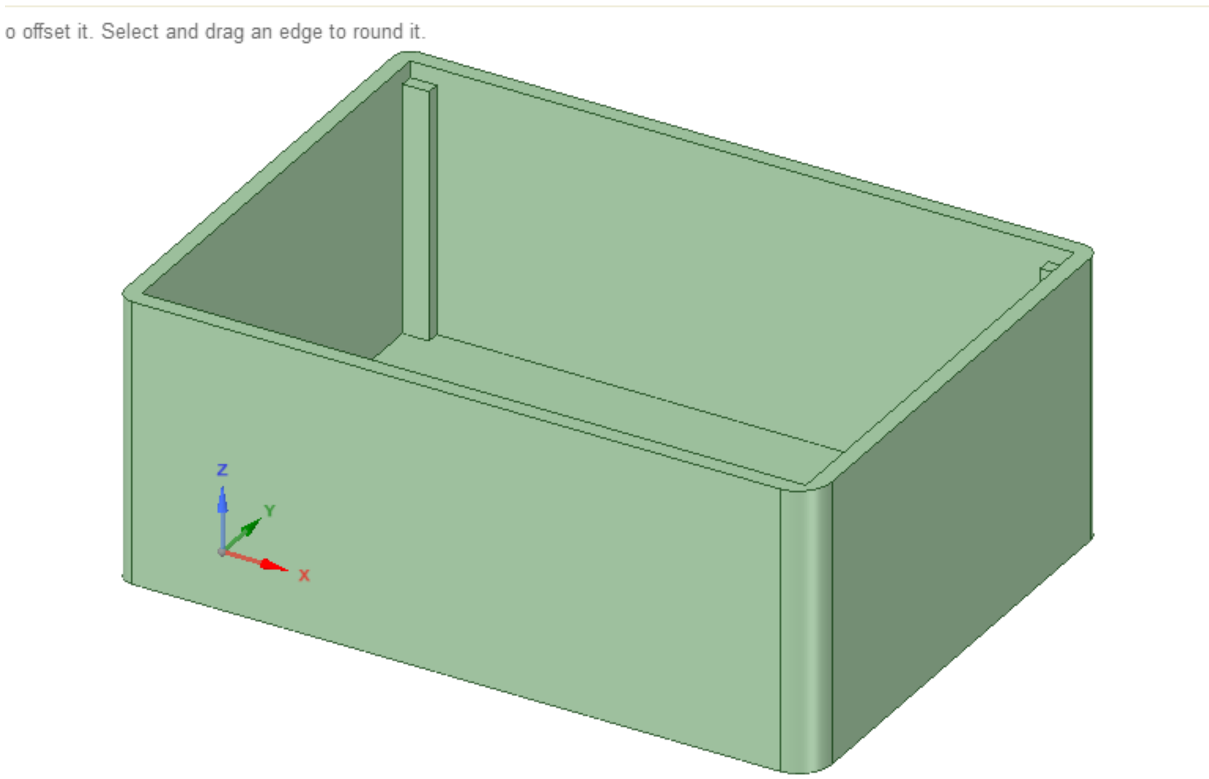


Prototyping stage:



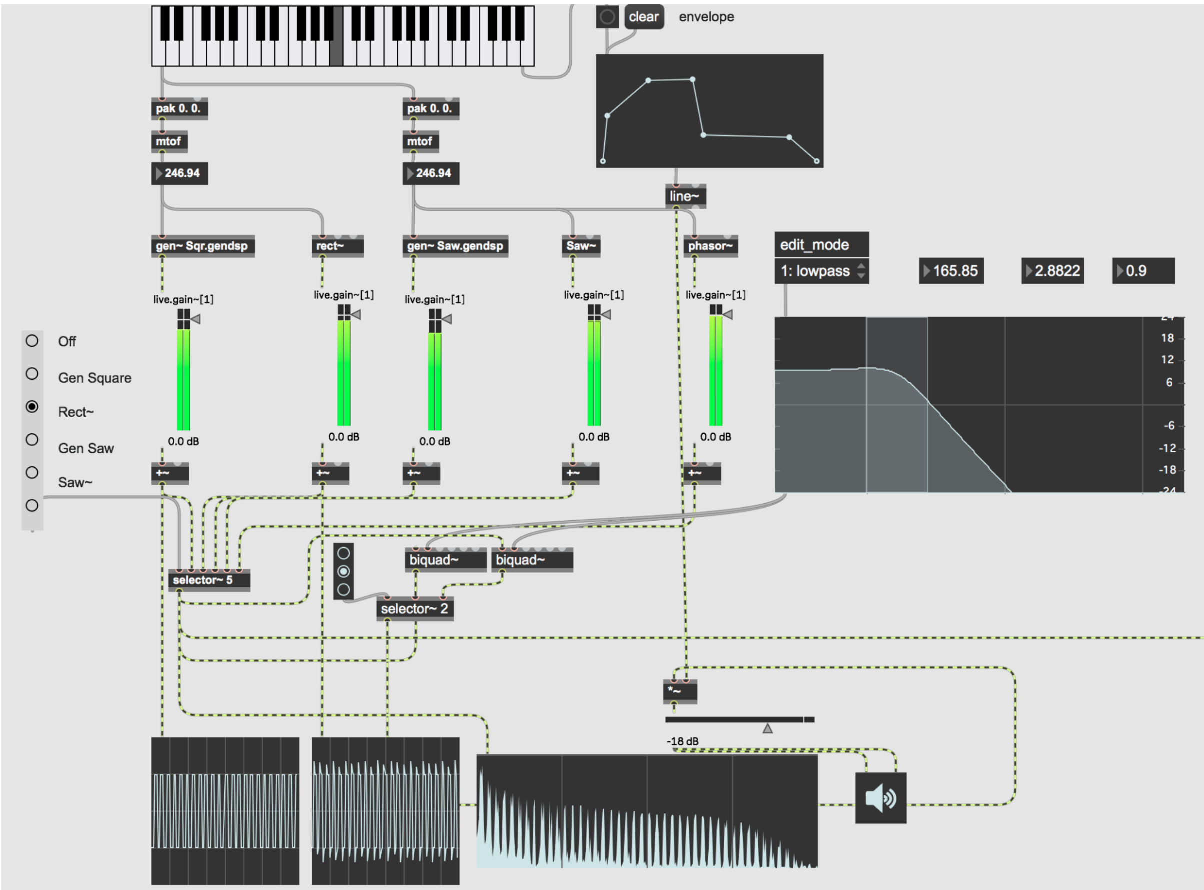
Failed attempts at PCB boards. Red board was printed from China. Green board laser engraved and etched in acid with home CNC machine.

3D Printed case model:

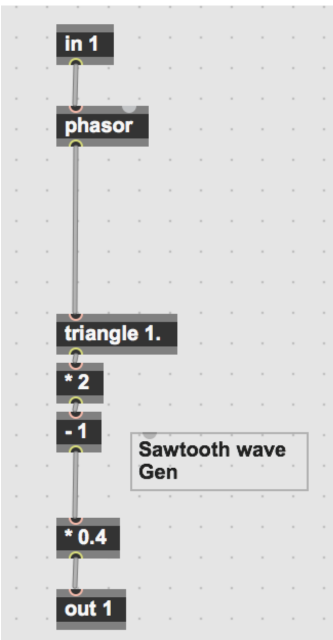


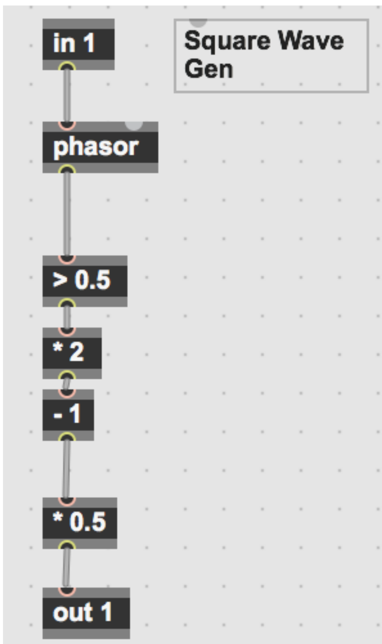
<u>Appendix C Max Patches;</u>

Proof of concept for the block synthesiser. Allows for swapping multiple oscillators and filters;

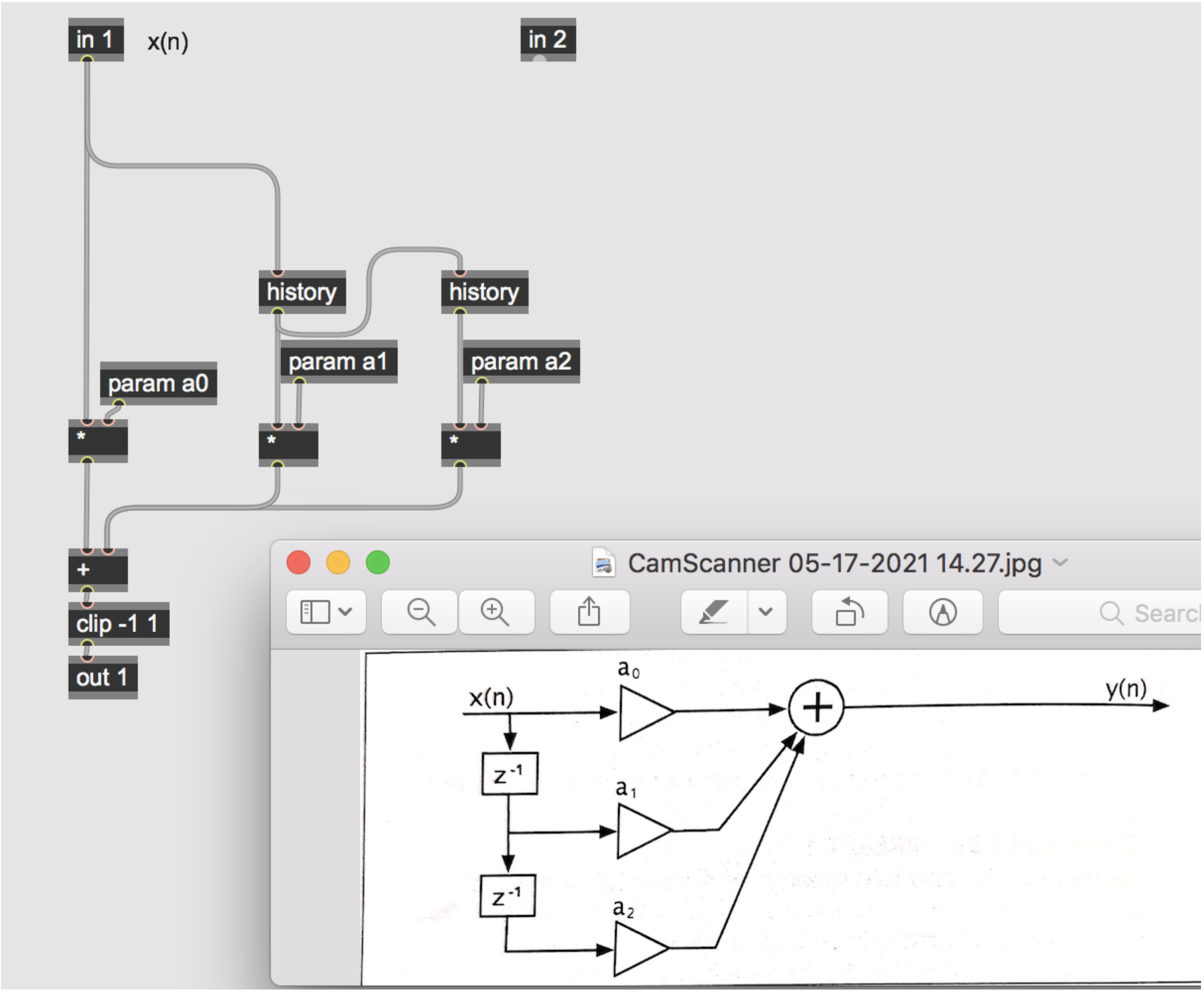


Trivially generated Gen~ Sawtooth and Square waves.





Second order FIR filter in Gen~ with block diagram;



Appendix D: Survey Results

Question:	Option:	Number of Selection	Total Answer For Question:	Precentage:
I confirm that I have read and understand the information sheet for the above study and have had the opportunity to ask questions.	Yes	60	60	100%
	No	0		0.00%
The following are words often used to describe Analog audio. What words would you associate with Analog synthesis?	Warm	39	125	31.20%
	Natural	21		16.80%
	Thick	21		16.80%
	Distorted	16		12.80%
	Round	21		16.80%
	None of the above	7		5.60%
	Other	0		0.00%
The following are words often used to describe Digital audio.What words would you associate with Digital synthesis?	Clean	38	116	32.76%
	Harsh	8		6.90%
	Dry	14		12.07%
	Sterile	22		18.97%
	Crisp	23		19.83%
	None of the above	8		6.90%
	Other	3		2.59%
Do you own a hardware synthesiser?	Yes	48	59	81.36%
	No	11		18.64%
Is your synthesiser digital or analog in nature?	Analog	9	48	18.75%
	Digital	10		20.83%
	Hybrid	6		12.50%
	I own both Analog and Digital	23		47.92%
	Not sure	0		0.00%
Do you consider the sound of analog synthesis to be superior to digital?	Yes	3	9	33.33%
	No	6		66.67%
Do you think you have a favourite between Analog or Digital synthesisers?	Analog is my favourite	24	32	75.00%
	Digital is my favourite	8		25.00%
Why do you consider analog synthesis to be your favourite?	Sounds quality	8	21	38.10%
	Digital sounds too clean	1		4.76%
	Ease of use	3		14.29%
	Tactile nature	7		33.33%
	Other	2		9.52%
Do you use any Softsynth" or plugin synthesisers on your computer? "	Yes	48	59	81.36%
	No	11		18.64%
Please lower volume first then readjust to a comfortable listening level. Headphones advised.Which sample sounds more "warm" to you?	Sample 1	28	57	49.12%
	Sample 2	29		50.88%
Lower volume:Which sample sounds more "Distorted"	Sample 1	34	57	59.65%
	Sample 2	23		40.35%
Which Sample do you prefer?	Sample 1	37	57	64.91%
	Sample 2	20		35.09%
Which sample do you think is from a Digital synthesiser?	Sample 1	35	57	61.40%
	Sample 2	22		38.60%
Which sample do you think is from an Analog synthesiser?	Sample 1	28	57	49.12%
	Sample 2	29		50.88%
Listen to the previous sample again Which filter sweep sounds "Fatter"?	Sample 1	21	55	38.18%
	Sample 2	34		61.82%
Listen to the following sample What words would you use to describe it?	Warm	31	164	18.90%
	Clean	6		3.66%
	Natural	5		3.05%
	Harsh	18		10.98%
	Dry	11		6.71%
	Thick	21		12.80%
	Sterile	1		0.61%
	Distorted	35		21.34%
	Round	17		10.37%
	Crisp	11		6.71%
	Thin	5		3.05%
	Other	3		1.83%