Analogue Synthesizer

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Contents

1 - Introduction

With the digitalization of modern society, music producers, singers and musicians ought to find the best way possible to improve their music quality by diversifying their sound outcomes. Analogue synthesizers, which are also called "Electronic Sound Synthesizers" provide just that, a way to produce new sounds that would have been very difficult and nearly impossible to mimic with real, old-fashioned instruments. Synthesizers can also alter a sound's Timbre[1], duration, frequency, and intensity. Sounds can be produced in a controlled environment in an easier way. When synthesizers were first created, they came in the form of big and bulky machines but as the decades pass, they have evolved into compact easy-to-use keyboards. The first analogue synthesizer was invented by Harry Olson and Herbert Belar, in the early 1950s and was later developed by the Radio Corporation of America (RCA). In 1974, Minimoog was the first company to mass produce analogue synthesizers for commercial use. With their wide range of use, synthesizers help unravel the full spectra of frequency, intensity, and timbre of sounds. They were created brilliantly bringing together the electronic and musical knowledge of that era to create a new instrument that can produce a great variety of sounds (more than any other instrument). Synthesizers are an excellent example, amongst others, of how electricity can positively affect all fields of life, include music.

In this project, we aim to create our version of a fully functional Analogue Synthesizer and simulate it on LTSpice. This is done by combining the knowledge given to us in the first year of the EEE course as well as individual research.

2 - Design criteria

Synthesizers are commonly built with a keyboard, each of who's keys represent a different not. In this project we decided to use 88 keys, each of which represent a different voltage, which in turn represents a certain frequency, following the formula: $f = 2^{\mathrm{V}+4.781}$. Our synthesizer is aiming at producing audible signals and will thus try operating in the frequency range of 27.5Hz to 4.186kHz. In addition to these frequency requirements, we require the ability to filter, and modulate the signal, all of which has to be done through the use of real components. The synthesizer must also produce a signal that can be driven through an 8Ω loudspeaker.

In addition, these main technical criteria, we must also consider other key specifications such as performance, testing, materials, and customer. Regarding performance, we want our synthesizer to be able to operate in the 27.5Hz to 4.186kHz frequency range. Additionally, we want the synthesizer to be able to high pass and low pass filter in this range while making the cut-off frequency and extent of resonance variable. Furthermore, we would like to be able to change the evolution of one's audio signal over time and would therefore like to implement an ADSR and amplitude modulation module. Lastly, with respect to power consumption we would also like the full system to consume below a full watt of power.

When having built the circuit, we also have certain testing requirements. First is that we must simulate the frequency and voltage time (if necessary) responses of the individual module circuits (VCO, VCF etc.). Before integrating these circuits, the individual tests must show that each circuit is working according to the module specifications. Our second testing criteria is with regards to the system. It must be tested for each voltage input corresponding to a certain frequency. Checking whether the VCO is producing the correct wave

[1] **Timbre**, also called **timber**, quality of auditory sensations produced by the tone of a sound wave.

with correct frequency and thus whether our VCF, VCA and ADSR are changing the wave as required. A final audio test through MATLAB is also required.

Regarding materials, we must aim at using the same operational amplifier and diodes throughout the design. Some op-amps differ in how much power they require, so just using one across the design would simplify the power infrastructure in the circuit.

Our design must also focus on the end user. Analogue synthesizers are unfortunately much less common in today's world of music production. Digital ones are more versatile; however, analogue ones are more unique, and thus remain an attractive instrument to buy for synthesizer purists in the industry. In order to compete with digital synthesizers, we must thus try to implement as much choice as possible, which plays into the performance; variable filtering through variable cut-off frequency and resonance, ADSR, amplitude modulation and the choice of different oscillator waves.

3 - Project planning and management

3.1 - Strengths and Weaknesses

After reading about the synthesizer, and the components that make it, we had to think of the optimal way to design our own. It is only logical that for the task to be finished in the least amount of time, everyone in the team should do what they are best at. Thus, to prepare accordingly for the design, we needed to assess everybody's strengths and weaknesses. Based on that, we would later assign the appropriate roles to utilize everyone's strengths as much as we can.

3.1.1 - Andy

Andy showed great leadership skills throughout the project. He was very good at assigning roles for all the members (including himself) according to their strengths and weaknesses (as mentioned above). He was always up to date with what everyone was working on and how focused they were in what they were doing. Furthermore, he is great at planning, so with the help of Athanasios, they were able to create a timetable for the circuit and made sure everything[2] was on time. Lastly, Andy's area of expertise is the use of LTSpice. Out of all the members, he showed the optimal solutions to design issues[3] and he demonstrated great skill while preforming the testing of the complete synthesizer. Moreover, when Andy had a "tough" time dealing with technical challenges, he showcased having difficulty dealing with time management. As a result, some of his work took longer than expected to be finished thus the team had to adjust the timetable.

3.1.2 - Pawan

By using his experience with projects, Pawan was able to thrive in an environment filled with individual research. He showed restlessness in finding more details about how the synthesizer works and the components it is made of which was a particularly great help in the initial research of the project. When needed, Pawan was also able to provide information about the components that other members of the team were working on apart from his own. Furthermore, Pawan illustrated his great writing skills in the final report of this project and the script of the final video presentation. On the contrast, Pawan showed difficulty

[2] Everything includes all the circuit components, testing, the report etc.

[3] E.g.: Use of wrong type of transistor or diode, but also shortcuts to make circuits simpler

communicating with the team as he is good at individual work. This, sometimes, created barriers between teammates which later caused some timing issues[4].

3.1.3 - Athanasios

Together with Andy, Athanasios was able to showcase his great organizing abilities when creating the timetable for the team, so that the project would be finished in the least/optimal amount of time. Athanasios preformed excellent under pressure, by always being on time with his work. When someone was behind on the time schedule, Athanasios was quick to adjust it to fit their needs in a way that would not impact anyone else. When building the ADSR circuit, he faced tremendous difficulties trying to make it work but showed great adapting skills by quickly deciding to let it go and by building the ASR instead. That way, he was able to save a lot of time that was used in different areas of the project that were most needed. Through this technical difficulty, Athanasios realized that using LTSpice was not one of his strengths but was able to learn and adapt quick. Furthermore, since he had little inexperience with projects and since English is not his first language, Athanasios had some difficulties with writing the report thus needed the team's help on certain areas.

To summarize:

	Strengths	**Weaknesses**
Andy	• Leadership • Organization / Planning • Use of LTSpice • Helping others	• Time management • Responding under pressure
Pawan	• Research skills (Individual and for the team) • Writing skills	• Lack of Communication
Athanasios	• Organization / Planning • Time management • Adapting/ Responding under pressure	• Use of LTSpice • Writing skills

From the information above, it is clear that the team consists of diverse individuals. What is nice, is that the members of the team "complete" each other, as someone's weakness is another person's strength.

[4] Components took longer to be completed, due to the lack of communication. As a result, members of the team took longer to move on to the next task.

3.2 - Assignment of roles

After doing our research to understand the tasks that needed to be completed and after closely examining each member's strengths and weaknesses, the team decided to divide the labor. Andy took the initiative to do most of the LTSpice work since that is what he is feeling most confident at. To be more precise, he completed the VCO, VCA, LFO, Integration and testing of the circuit. Furthermore, he and Athanasios assigned the roles to the team by using both of their organization and planning skills. Meanwhile, we utilized Pawan's great research abilities to help the team with research about the circuits that needed to be made so that the synthesizer was complete. Pawan was also a great help while everybody was writing their own parts of the report when the circuits were complete by using his writing skills. Moreover, Pawan oversaw making the VCF circuit. Lastly, Athanasios used his organizing abilities and took all the individual writing bits from all the members to compose the complete and final report. Furthermore, he was in charge of making the ASR circuit as well as helped assign the roles (as mentioned before).

Member	**Roles**
Andy	• VCO • VCA • LFO • Integration • Testing • Assignment of roles
Pawan	• Extensive research • VCF • Provided help with writing
Athanasios	• Report • ASR • Assignment of roles

3.3 - Meeting schedule and time management

When planning how we would approach the project, we decided to meet in the library every Monday to discuss what we would do each week. Furthermore, we added a weekly team checkup every Wednesday to confirm weather everyone is on time with their work. The initial research took the team a full week to complete so that we made sure we knew as much information as we could about the synthesizer. By the beginning of the third week, the circuits for the Exponential curve generator, the Low-Pass Filter and the Low-Frequency Oscillator (LFO) were finished. Furthermore, we dedicated this week to completing the final programming assignment. By the beginning of the fourth week, we had completed the circuits for Voltage Controlled Amplifier (VCA), Voltage controlled Oscillator (VCO), Envelope Generator (ASR) (NOTE: the ADSR envelope generator was meant to be completed by the third week, but we encountered timing issues, as discussed earlier, that caused us to extend the deadline for the circuit) and Voltage Controlled Filter (VCF). The integration of all circuits was finished by the Thursday and the testing of it was finished by the Sunday of the fourth week. On the fifth week we encountered some timing drawbacks, as a member of the team was diagnosed positive with Covid-19 and an extension of three days was given to us.

On the fifth Monday, we planned the Final Report and Video, which we finished by the sixth Tuesday. Overall, we tried to follow the initial plan as much as possible and that lead us to having an enjoyable time without worrying too much. Below we can see the original plan, in blue are team meetings in green are deadlines set by the group, and in red are deadlines set by the department.

Monday	Tuesday	Wednesday	Thursday	Friday	Saturday	Sunday
Team meeting		**Team checkup**				
Team meeting **Research complete**		**Team checkup**				
Team meeting **Exponential complete** **Low pass filter complete** **LFO complete** **Beginning of programming**	**Break to complete programming assignment**					
Team meeting **ASR complete** **VCA complete** **VCO complete** **VCF complete**		**Team checkup**	**Integration complete**			**Testing complete**
Team meeting **Report plans complete** **Video plan complete**		**Team checkup**				**Deadline**
Team meeting	**Report complete** **Video complete**	**Extended deadline**	-	-	-	-

4 Design process

4.1 - Initial research

To start off the project we were first concerned with what an analogue synthesizer does, and the way in which it enables users to change audio signals. After consulting multiple sources, we broke down our synth into five essential stages: oscillators (VCO's), filters (VCF's), low frequency oscillators (LFO's)'s, amplifiers (VCA's) and the ADSR envelope / amplitude envelope.

The oscillator is the first fundamental building block of our synthesizer and is what produces the audio signal used in our system. Oscillators in synthesizers are voltage controlled, as it is convenient to change the fundamental frequency of the input signal by altering voltage. Typically, these voltage-controlled oscillators (VCO's) produce sine waves, square waves, triangle waves and sawtooth waves. Sine waves consist of one fundamental frequency, whereas square waves consist of many overtones and harmonics. The diversity in frequency of each typical signal means it is useful to have access to multiple VCO's when producing music. Since each VCO creates a different characteristic signal with different overtones and harmonic frequencies, the use of multiple VCO's can allow creative layering of input signals to produce more interesting and complex audio.

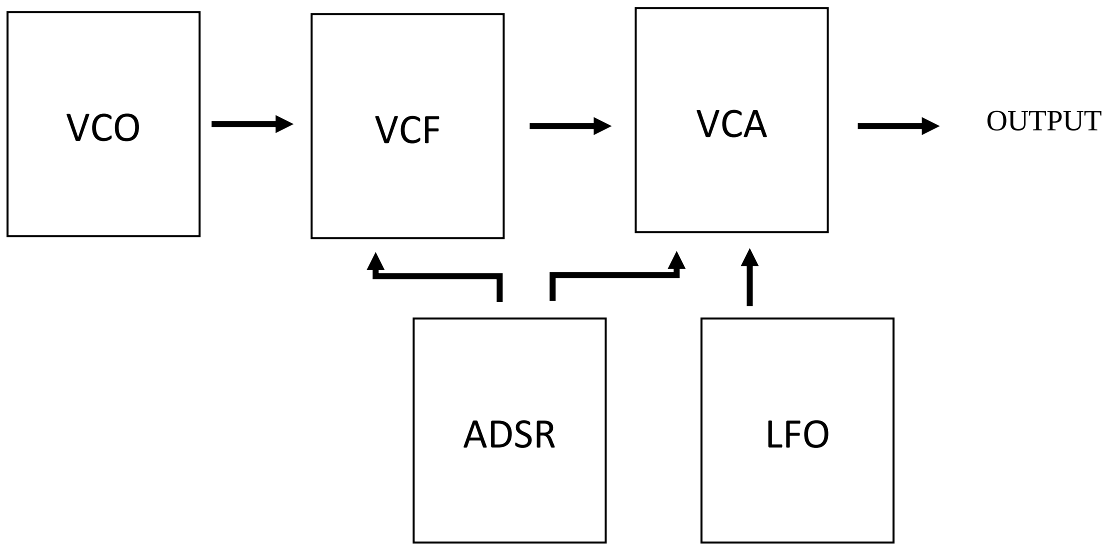
In addition to the oscillator, the filter is an important part of sound design when using a synthesizer. Filters in synthesizers are voltage controlled (VCF) and can filter out frequencies from the VCO output. Low pass filters will filter out high frequencies while high pass filters filter out low frequencies. Additionally, VCF's have voltage adjustable cut-off frequencies and voltage adjustable levels of resonance.

Although the VCO's produce the primary audio signals used in the system, LFO's (Low frequency oscillators) are used to modulate signals. Modulation of a signal means you can change the sound of the audio signal according to the LFO signal. For example, modulating amplitude using the LFO signal introduces tremolo, a trembling effect, while modulating pitch introduces vibrato, which is a pulsating change of pitch. If the LFO signal rises and falls like a sine wave, the audio signal's pitch being modulated will thus also rise and fall.

Lastly, the VCA and the ADSR are key in synthesizers as well. VCA's are voltage-controlled amplifiers and are used to vary the volume of the sound over time. The ADSR is concerned with the timing of the synthesizer. It dictates how long it takes for the synthesizer to reach full volume and the way in which it fades away. The ADSR consists of the Attack, Delay, the Sustain and the Release. The attack stage is concerned with the amount of time it takes to reach full volume, while the decay stage dictates time taken to reach the secondary volume. The exact secondary volume is determined by the sustain and the release dictates how long it takes for the sound to fade out.

4.2 - Creating a black box outline

To obtain a better understanding of the system we are trying to build can create a block diagram:



4.3 - VCO

The VCO is an essential piece to any analogue synthesiser as it is the first thing that creates the sound to be used later. Its fundamental purpose is to take a single DC voltage input and create an oscillating wave at a specific frequency. The desired frequency output is directly based on how humans hear and interpret sounds. The human ear does not hear sound additively but multiplicatively. This means if you added 100Hz A4 which is 440Hz, resulting in 540Hz, hearing this sound would not sound like an A. However, doubling the frequency to 880Hz would sound like a higher pitch A, similarly halving the frequency to 220Hz, would sound like a lower pitch A. Nevertheless, both As, are 12 notes (including minors) away from A4, but the frequency difference is 200Hz, or 440Hz. This implies we need a relationship which takes a linear additive input, voltage, and changes it to a multiplicatively scaling output frequency. The previous points above imply an exponential relationship of base 2, hence the frequency relationship below:

$$f = 2^{V+4.781}$$

This fact then demands immediately a circuit to exponentiate the input voltage. The 2 components which are known to have exponential relationships are both dependent on semi-conductors, diodes and BJTs. We decided to go with a well-known exponentiator circuit which relies on a diode and resistor connected in a negative feedback loops with an Op-Amp. The diode only exhibits exponential behaviour up until around 0.7 volts where it then becomes extremely conductive. However, the input to the VCO needs a maximum input of around 7.25V. This will not be as high for the exponential circuit as it will be scared so we have four diodes in series effectively extending the range to around 2.8 volts. To test the exponential circuit, we swept the dc input and viewed the output. First issue we noticed with the raw circuit was that it was inverting the input. This was an easy fix as we just added another inverting amplifier with a gain of 0. Now the circuit clearly outputs an exponential relationship for an input voltage below 2.8 volts as shown in Figure 1.

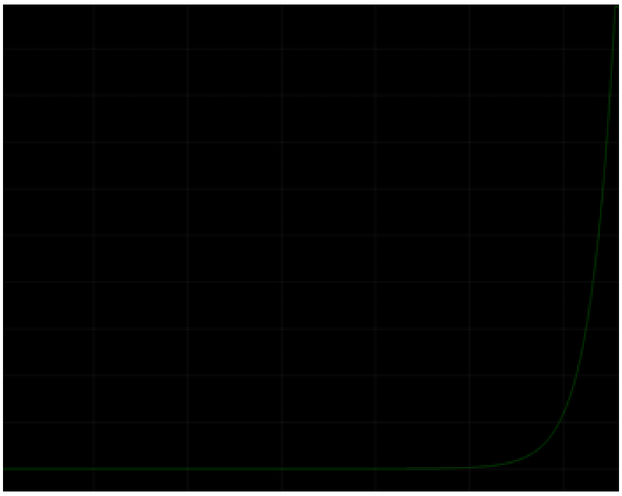


Figure 1, the exponential curve (light green)

Now to work out the exponential relationship between the input and output we took values and ran them on a solving algorithm we made on excel. The good thing about exponential functions is that in the form below:

$$V_{out} = Ae^{bV_{in}+C}$$

A and C are 2 parts of one scaling constant, so we can set one and determine the other allowing me to select an easy to manipulate value of A and work out its corresponding C. The key with the exponential circuit is that it fundamentally must exhibit the exponential relationship, the actual scaling of the frequency is controlled later via gain stages and the oscillator circuit. How this is done is explained below.

An import property of the exponential function is that all constants raised to the power of x, are just equal to *e* raised to constant times x:

$$C^x = e^{kx}$$

$$k = \ln(C)$$

This fact allows me to manipulate the above function by scaling V_{in} by a factor of ln(2) to “convert” it to a function of 2^x. The second important thing to note is that by adding another constant, r, such that:

$$r + C = \ln(2) - 4.781$$

Will allow us to factor to out the ln2 leaving the function as approximately:

$$V_{out} = A2^{V_{in}+4.781}$$

The next step is to take this scaled exponentiated V_{out} and convert it to a frequency. The design the oscillating circuit is based on is the one on “allaboutcircuits” (Mitchell, Robin).

The circuit essential to achieve this is the VCO, the actual circuit which takes an input voltage and output an oscillating waveform. This circuit can be seen below in Figure 2. For this we used a classic triangle and square wave generator. There are to main reasons for this, from an auditory perspective both these signals are rich in harmonics and allow for more interesting sound manipulation. Second, both these waves are created co-dependently so just having one circuit always nets both waves. The circuit has 3 main operators that allow it to work, an integrator, a Schmitt trigger and a MOSFET. The output of the integrator is always the input voltage plus or minus a scaled and integrated version of the input voltage. In the case of a constant dc input X the output Y looks like one of the functions below:

$$Y = mX \pm nX\Delta t$$

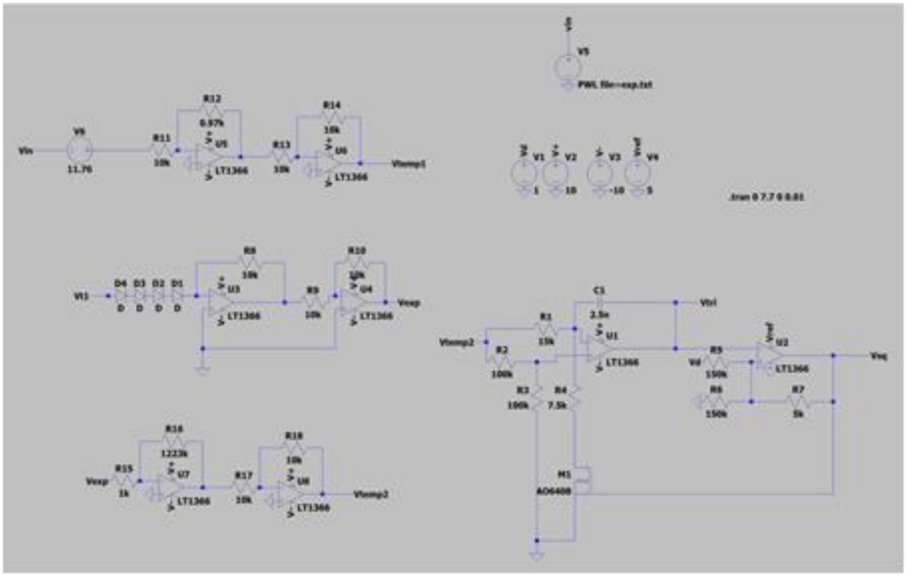


Figure 2 VCO circuit

It is obvious that the function changes linearly with time. The output at the Schmitt trigger is determined by whether the input voltage is above a certain threshold voltage or below another, Vt1 and Vt2. When the input voltage crosses Vt1 the output becomes the value at the positive rail and when the input voltage goes below Vt2 the voltage becomes the value at the negative rail. Finally, when the MOSFET is on it connects the resistor and ground to the negative rail of the integrator and cause the output Y to be a decreasing linear function and, when it is off the resistor and ground are disconnected and the output Y is an increasing linear function of time. These 3 facts together cause the circuit to generate 2 oscillating waves for a given input voltage, this is shown in the series of steps below.

1. MOSFET is off
2. Y is increasing
3. Y > Vt1
4. Schmitt output is High
5. MOSFET is on

6. Y is decreasing
7. Y < Vt2
8. Schmitt output is Low
9. MOSFET is of

Once the fundamental circuit was in place and working, we had to work out what values to use for resistors capacitors and voltages. After trying to work out a relationship between voltage and frequency via the circuits the below equation was derived:

$$f = \frac{X}{2(V_{ref} - X)R_1C}$$

Evidently, the above equation is not a linear relationship between frequency and voltage which makes it much harder to manipulate it into the desired exponential equation. Despite this, after plotting a graph and testing different values we noticed there is a range where the function can be approximated to a straight line. The first thing we realised was that there is a max theoretical input voltage of V_{ref} and this limited both the frequency and the input voltage. Moreover, V_d controls how quickly it approaches said maximum frequency. After days of tuning, we found 5 volts on the positive rail and 1 Volt V_{ref} gave it the maximum upper range of 4.186KHz. The next thing to tune was the R and C which along with the 1 volt input it allowed me to reach the minimum frequency of around 23.8Hz. Once determining values that work over the full range of output frequencies, we needed to approximate a linear relationship to allow me to better manipulate the form.

I selected various voltage inputs and viewed what frequency waves they were outputting and recorded them on excel. we then approximated it using a linear relationship to get an understanding. The closest relationship we found was with an added constant however in implementation we could not use this as we had to cancel it out by subtracting a constant resulting in negative input voltages. This forced me to set the y-intercept to 0 and simply get a function in the form f = kv, even though it was not as accurate with an R^2 mean value of 0.977. Finally, below we have shown how to put all the equations together to achieve the desired original function.

$$X = AV + B$$

$$W = 10^{-6}e^{3.222M-9.21}$$

$$Y = KW$$

$$f = 817.3Y$$

$$f = 817.3K10^{-6}e^{3.222AV+3.222B-9.21}$$

$$f = 817.3K10^{-6}2^{\frac{3.222AV+3.222B-9.21}{\ln(2)}}$$

To get this into the desired form:

- $817.3K10^{-6} = 1$
- 3.222A = ln (2)
- 3.222B - 9.21 = 4.781*ln (2)

I then worked these theoretical values for A B and K and implemented them in smaller circuits. The addition of B was done via a voltage source, and the scaling of A and K was done via Op-Amps.

4.4 - VCF

The VCF (Voltage Controlled Filter) is another module we have decided to implement in our synthesizer. It, like any other filter, controls the output signal power associated with certain frequencies. Having the choice to variably filter out higher or lower frequencies and to add resonance to different extents is an important musical choice many synthesizer users like to make. The following therefore discusses the methodology behind our VCF, the ways in which it varies cut-off frequency and resonance and the ways in which it aids the user of the synthesizer make musical choices.

4.4.1 - Initial Ideation and Passive Filters:

We started off our VCF design process by looking at a simple passive low pass filter, and the ways in which it falls short of our specifications. The simple resistor capacitor combination was initially used and has been assigned arbitrary values that would produce a frequency response in the 1Hz -100kHz region:

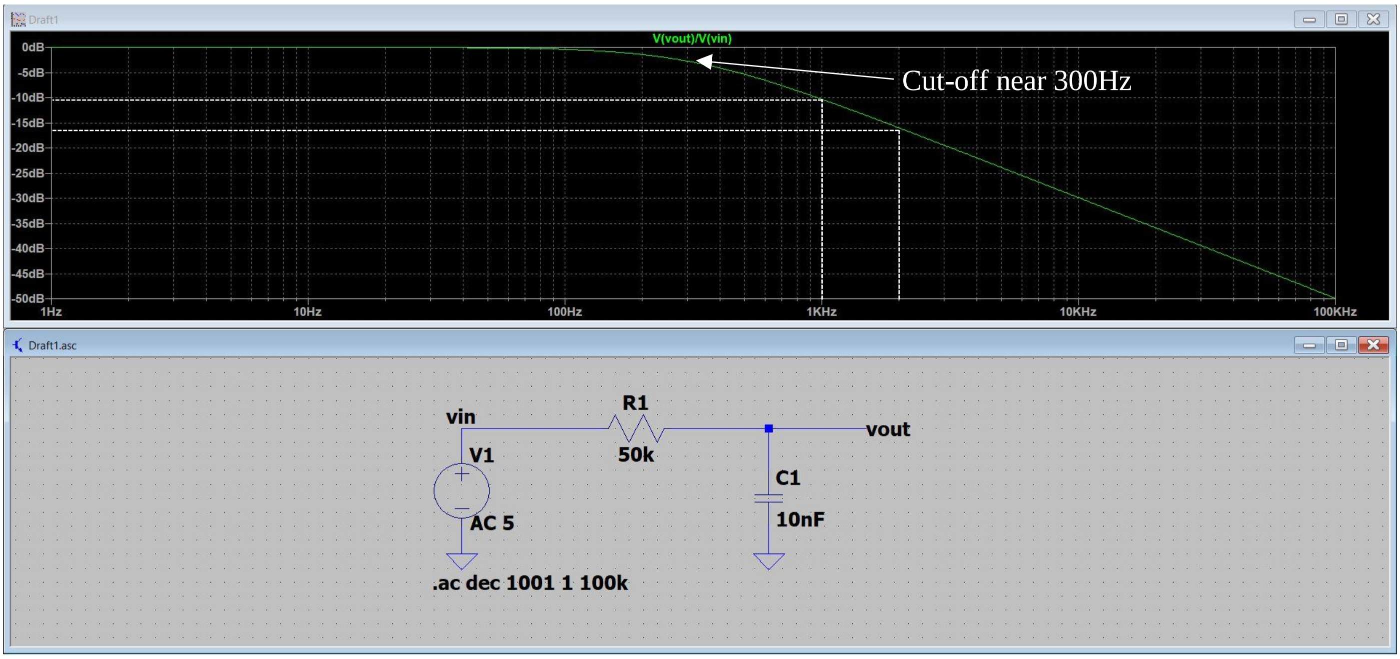


Figure 3, first order Low pass filter

From the simulation it is evident that the circuit is indeed a low pass filter. At low frequencies like 10Hz and 100Hz, the circuit provides no gain and allows frequencies to pass, whereas at high frequencies like 1kHz and 10kHz, we see a significant loss in power, and thus volume. Deriving the transfer function of the circuit also provides insight into the low pass behaviour:

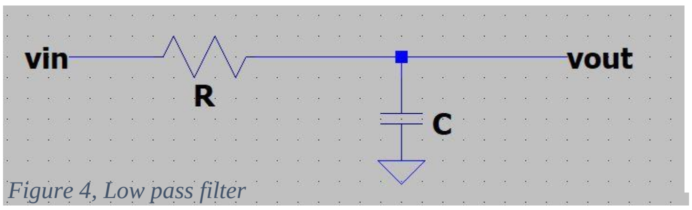


Figure 4, Low pass filter

$$\rightarrow \frac{v_{out}}{V_{in}} = \frac{\frac{1}{j\omega C}}{R + \frac{1}{j\omega C}} \rightarrow H(\omega) = \frac{1}{1 + j\omega RC}$$

Looking at the limits of angular frequency ω $(2\pi f)$, we see that as $\omega \rightarrow 0, H(0) = 1$ and as $\omega \rightarrow \infty, H(\infty) = 0$, indicative of a low pass filter. The transfer function also, however, gives us an insight into cut-off frequency and the way in which it can be altered. The cut-off frequency is the frequency at which the power output of the circuit has fallen by half in proportion to the power of the passband. This sits at -3dB:

$$|H(\omega_c)| = \frac{1}{\sqrt{2}} \rightarrow \frac{1}{|1 + j\omega_c RC|} = \frac{1}{\sqrt{2}} \rightarrow \frac{1}{\sqrt{1 + \omega_c{}^2 R^2 C^2}} = \frac{1}{\sqrt{2}} \rightarrow \omega_c{}^2 R^2 C^2 = 1 \rightarrow \omega_c = \frac{1}{RC}$$

This formula can be verified by plugging in the values of figure 1: $f_c = \frac{1}{2\pi \cdot RC} = \frac{1}{2\pi \cdot 50 \cdot 10^3 \cdot 10 \cdot 10^{-9}}$ = 318Hz.

Having verified our formula works, we can see that to vary our cut-off frequency, we can vary the value of resistor R. This will be the main mechanism through which we change cut-off frequency throughout our design.

Despite the ease of implementation, our circuit has several flaws. The first being that frequencies beyond the cut-off point are not completely excluded from the output, as they are still relatively well associated certain amounts of power. Ideally, we would like the roll-off of the transfer function to be much steeper and thus tune out higher frequencies more effectively. Secondly, we would like to introduce resonance, as it is part of our specification.

We decided to tackle the first flaw by simply cascading our first order filter with itself to produce a second order filter. Applying additional filtering to an already filtered signal seemed logical and would ideally result in a faster roll off:

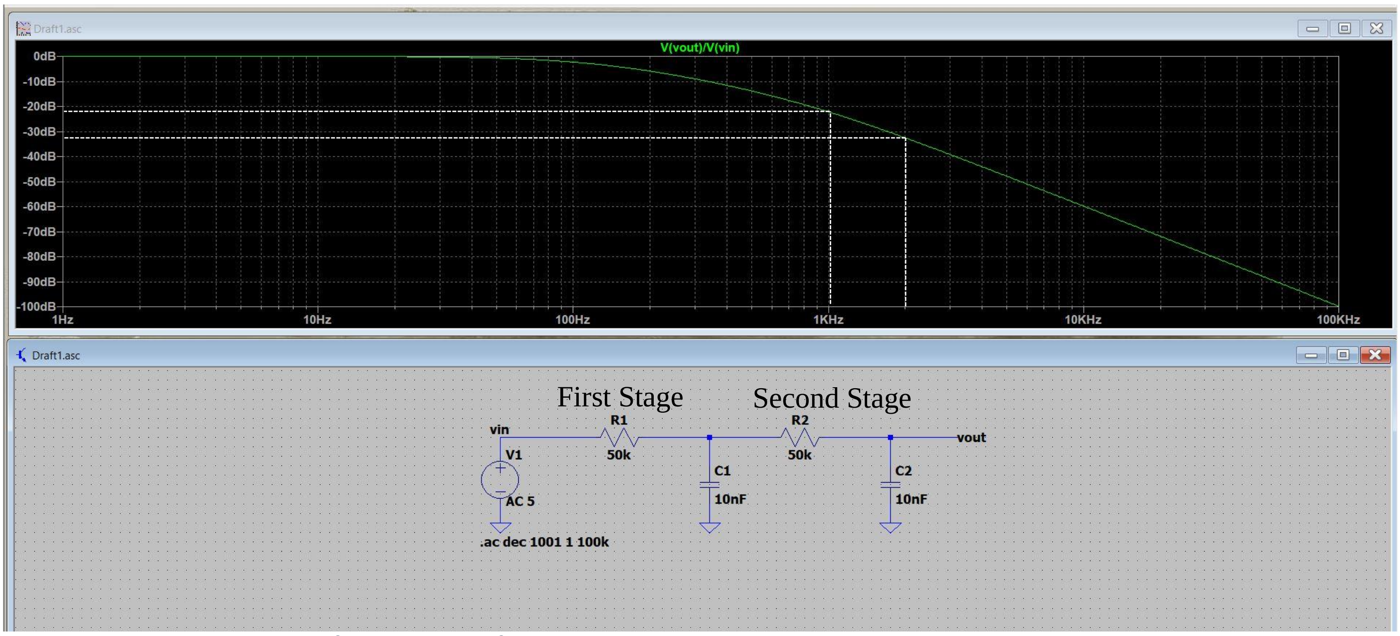


Figure 5, second order Low pass filter circuit and frequency response

From the simulation above it was clear that the roll off, and thus extent of the exclusion of higher frequencies was much steeper. From figure 3, 1kHz is associated with -22dB and 2kHz is associated with -32dB, implying a roll-off of 10dB/octave ≈ 12dB/octave, indicative of a second order circuit. For comparison, our first order filter associated 1kHz with -10dB and 2kHz with -16dB, implying a -6dB/octave roll-off, indicative of a first order filter.

Having created an effective low pass filter from the frequency perspective, we still see major shortcomings. The first is intrinsic to passive filters, which is that as they remove parts of the input signal, they also reduce the overall volume level of the signal. This implies that we require amplification built into our VCF. Additionally, our passive filter setup cannot produce resonance, since resonance is inherently the amplification of select frequencies, which again requires power.

4.4.2 - Active Filters:

To visualise our first short coming, we can load the above filter with a 1kΩ resistor and check its frequency response. We must load the circuit to see if there is a power loss:

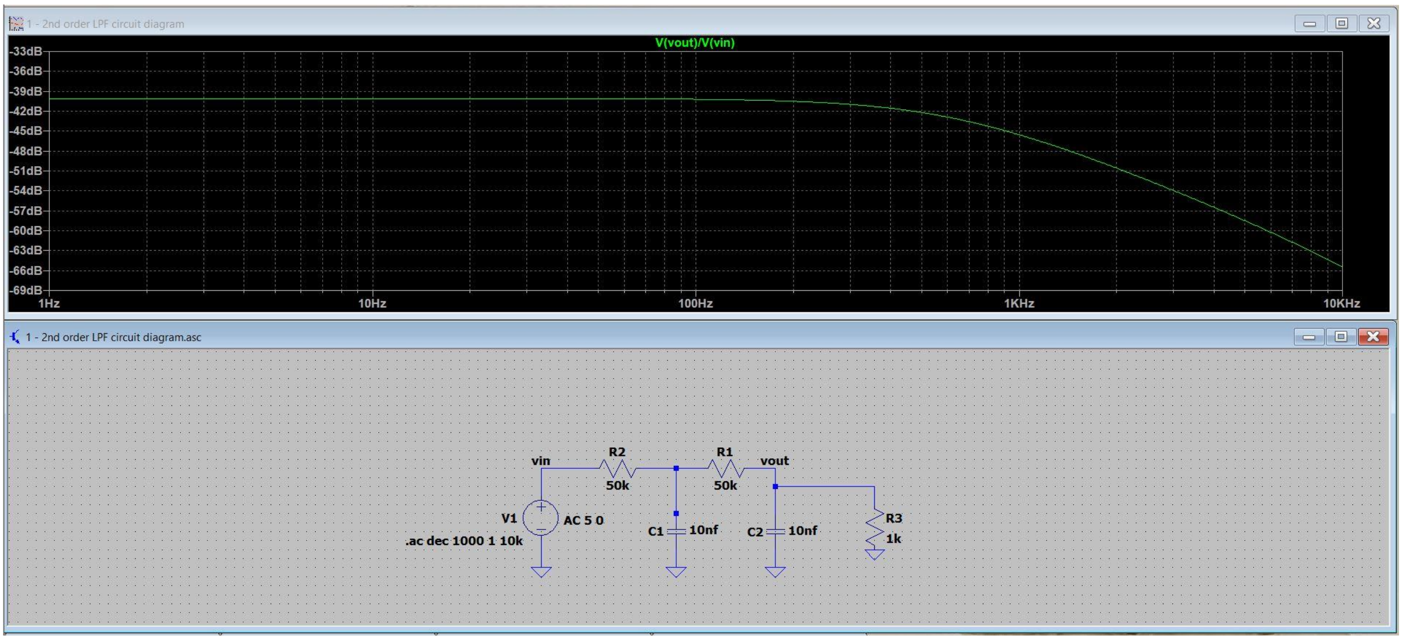


Figure 6, Loaded second order Low pass filter circuit and frequency response

We see that at frequencies we want to keep, there is a significant power loss of almost -40dB. To solve this problem, we took inspiration from Mortiz Klein's videos [Klein, Moritz] and decided to use simple op-amp based buffers throughout our design. Buffers are useful here as they provide copies of their input voltage at their output, but at the same time have a low output impedance and can thus provide substantial current. This means that one stage's input impedance is prevented from loading the previous stage's output impedance, which in our case limits the interferences between the two passive filter stages. We create a buffer by applying our signal at the positive terminal of the op-amp and creating a negative feedback path between the negative terminal and output. As per the specifications we use -10V and +10V power rails. We can see the effect of our buffers below:

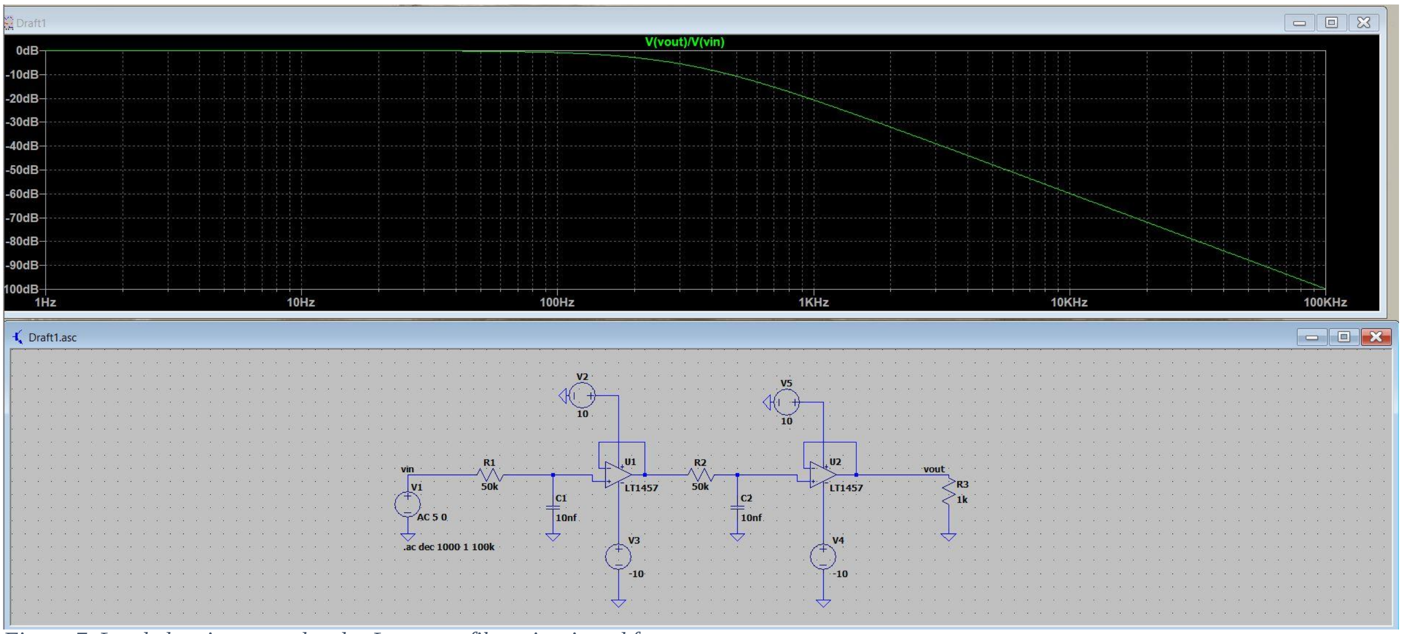


Figure 7, Loaded active second order Low pass filter circuit and frequency response

Above we can see that we retain the overall low pass frequency response, but do not systematically lose signal power across all frequencies throughout the circuit.

4.4.3 - Resonance:

Having created a cut-off adjustable (through adjusting resistors R1 and R2 (simulated later)) filter with unit gain at low frequencies we can move on to adjusting resonance. In active filters, resonance is created by amplifying select frequencies surrounding the cut-off frequency. We implemented this by connecting our output to the capacitor of our first stage:

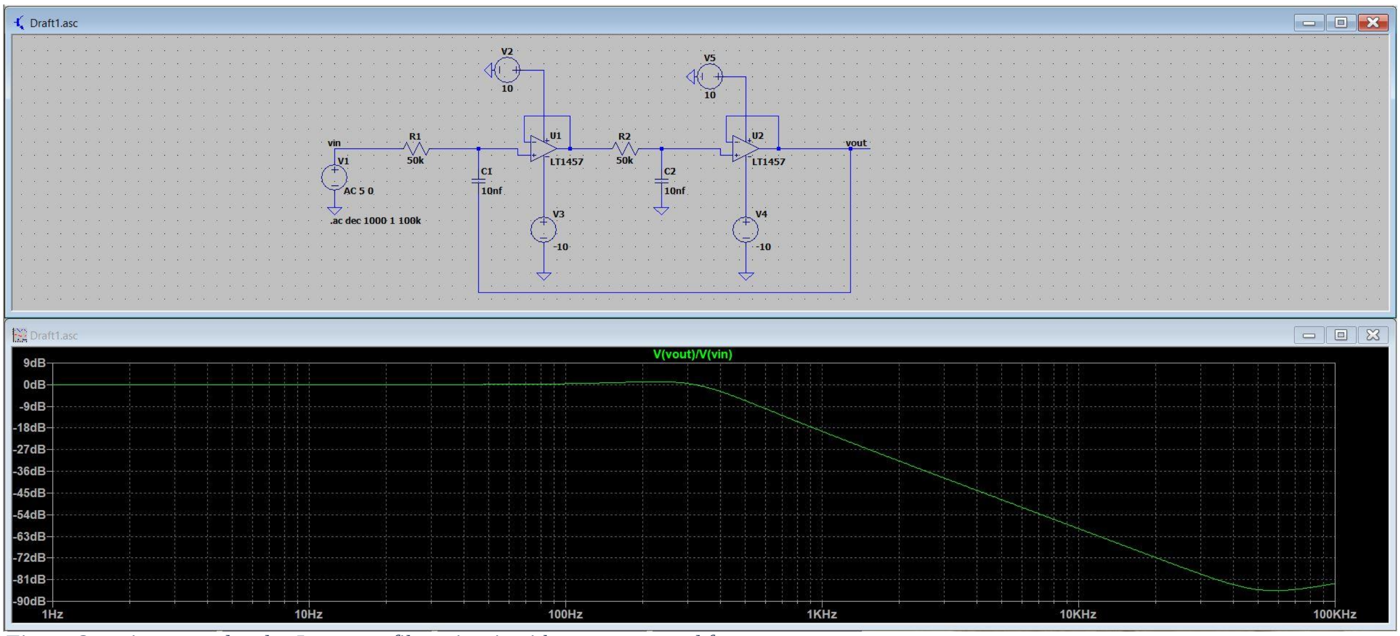


Figure 8, active second order Low pass filter circuit with resonance and frequency response

↓ Zoom in on 120Hz – 560Hz region

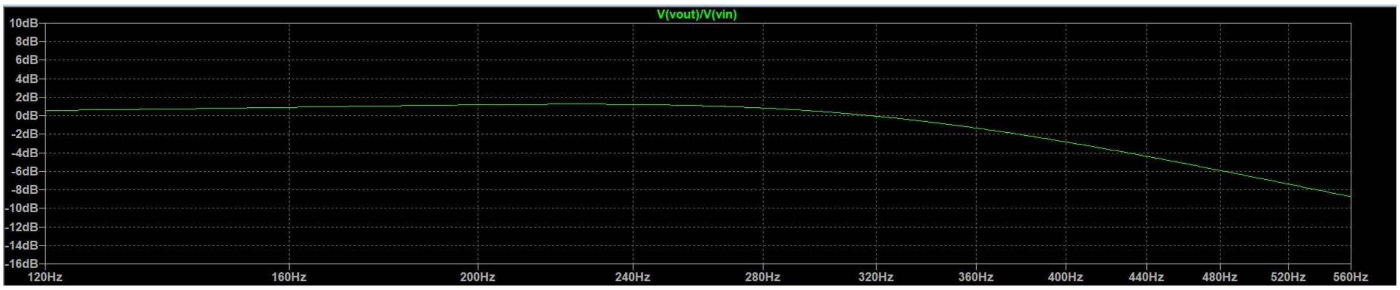


Figure 9, zoomed in frequency response

From the two figures above is evident that frequencies around 200Hz to 300Hz are amplified, indicative of resonance. However, as with the cut-off frequency, we would like to make the extent to which the system resonates voltage controlled using variable resistors. It is clear from our simulation that the extent to which resonance occurs is dependent on the feedback from the output. Therefore, we decided that to increase or decrease the level of resonance, we had to increase or decrease the amount of feedback from the output.

To lower the level of feedback we decided to use a voltage divider while simultaneously implementing a buffer afterwards. This is because a voltage divider is limited in the amount of current it can sink. To increase the level of feedback we use our buffer and turn it into an amplifier. This is done using an external circuit shown below:

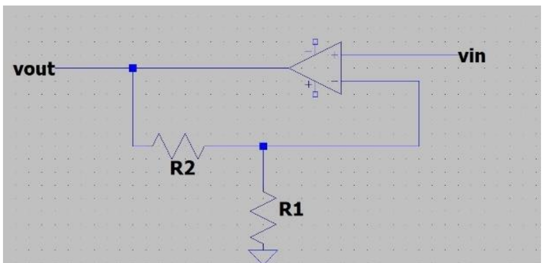


Figure 10, resonance feedback circuit

$$\rightarrow \frac{V_{in}-V_{out}}{R_2} + \frac{V_{in}}{R_1} = 0 \rightarrow \frac{V_{out}}{V_{in}} = \frac{R_2+R_1}{R_1}$$

Although arbitrary, we decided to make the amplifier have a gain of 2.5. Setting R2 as 10kΩ, R1 would thus be 6.8kΩ. Having this amplifier with set gain will thus allow us to increase or decrease resonance using our voltage divider. This is implemented below:

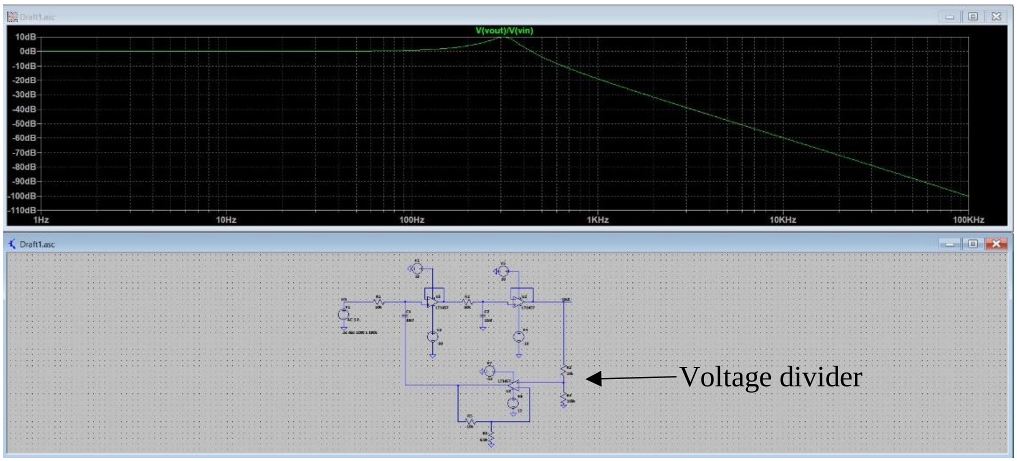


Figure 11, Active low pass filter with resonance and frequency response

Here we can see the full frequency response of our filter. Around the 100Hz – 400Hz range there is a visible level of resonance occurring. We can make the extent of this resonance adjustable through altering our voltage divider, simulated in section 4.4.5.

4.4.4 - LPF tests:

To test the Low pass part of the VCF, we make each of the relevant resistor's variables that can be adjusted. R1 and R2 must be the same value to alter cut-off frequency. R2 and R3 are changed in order to change the level of resonance.

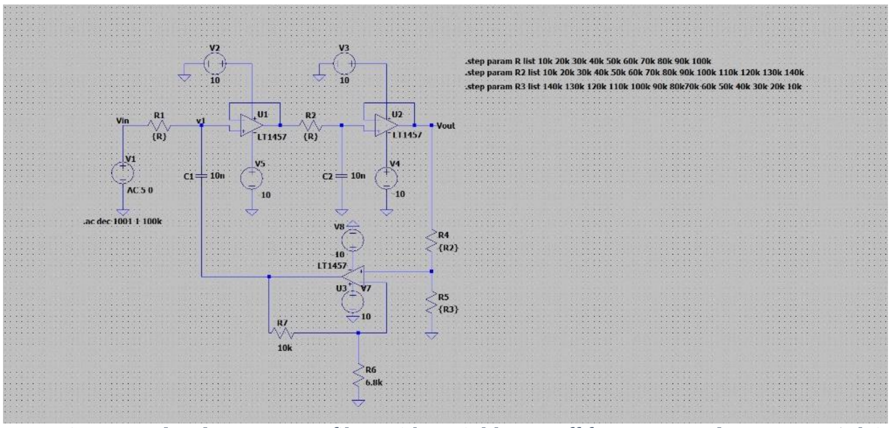


Figure 12, active second order Low pass filter with variable cut-off frequency and resonance (Klein, Moritz)

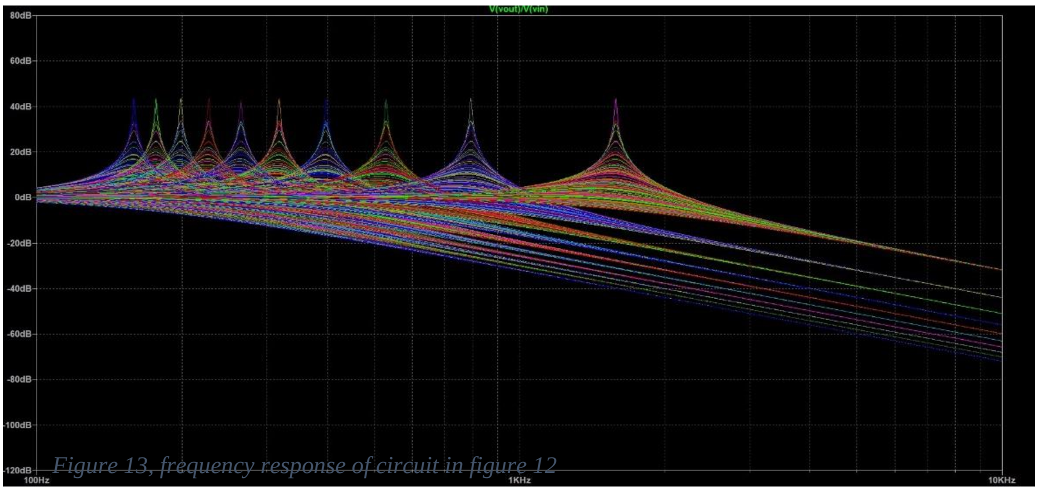


Figure 13, frequency response of circuit in figure 12

The plot shows that our filter is a good fit for our specification. The different peaks evidence that we can shift the cut-off frequency across the frequency range we use. Secondly, at those specific cut-off frequencies we see that we can significantly vary the level of resonance occurring, from 2-3dB associated with the resonant frequencies to 30-40dB.

4.4.5 - HPF tests:

The high pass part of our filter is similarly constructed to our low pass. We use a resistor capacitor combination but take our output voltage across the resistor instead of the capacitor. We saw, however, that it was implausible to switch the R1 and R2 resistors with C1 and C2 since C1 is connected to our resonance feedback path. As in Klein's circuit, we decided to connect our input signal through C2 and ground the connection to R1. In this way we produce a first order high pass filter.

In addition to this compromise, we have included a high frequency high pass filter comprised of a 1 microfarad capacitor and 200kΩ resistor at both the input and output. These provide AC coupling. In the case that our input signal has a DC offset we want to make sure that we do not run into headroom issues. Centring our input and output signals around 0 is thus done through these two filters. We also buffer our input into C2 since we do not want the AC coupling high pass filter to interfere with our second filter.

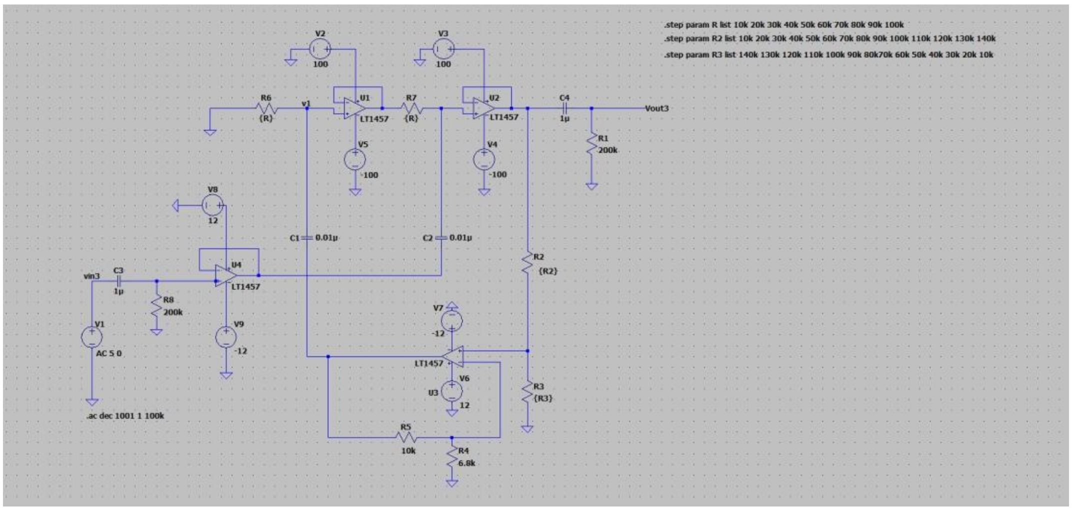


Figure 14, active second order High pass filter with variable cut-off frequency and resonance [Klein, Moritz]

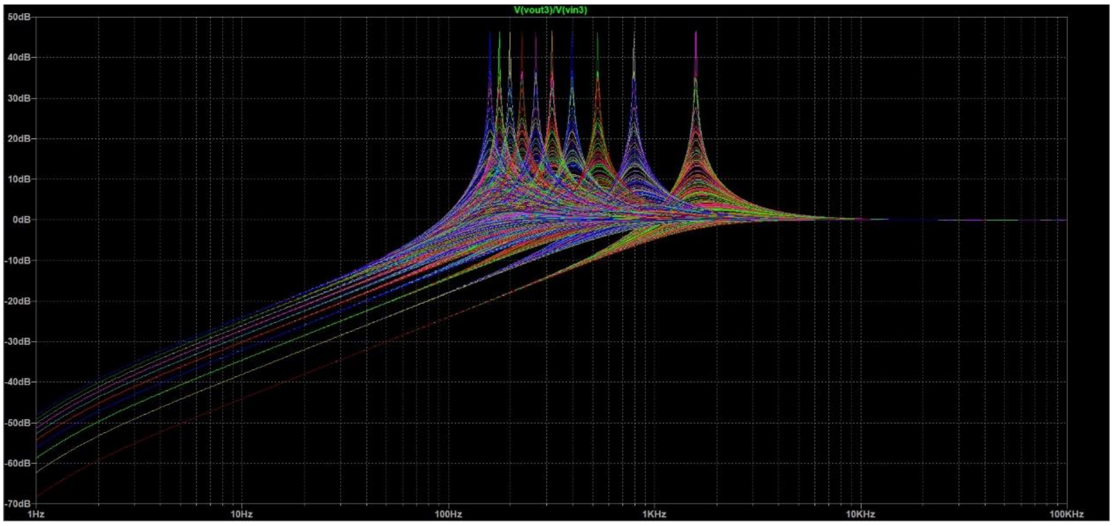


Figure 15, frequency response of circuit in figure 14

The frequency response shows how we can change both cut-off frequency and resonance like in our low pass. However, it is evident that the roll-off is less steep with respect to lower frequencies when compared to the low pass filter.

4.5 - VCA

The main role of the Voltage Controlled Amplifier (VCA) is to amplify the incoming voltage signal according to a second control voltage. This control voltage can be an envelope signal such as the ASR, or a low frequency oscillator, and the higher its voltage the more original signal needs to be amplified. This helps us incorporate the idea of amplitude modulation or *tremolo* into our synthesiser. The main method of doing this is via a differential amplifier. The current source for this differential amplifier is a resistor connected to the collector of an NPN transistor powered by a scaled and manipulated control signal. The VCA circuit used for the design is the one found on "Saggitronics" (Staffeld, Charles). The basic idea is that as the voltage on the base of the NPN changes the voltage of the collector will change, altering the emitter current through the current source, changing the gain according to the control signal. The gain of the differential amplifier is given by:

$$A_d = -g_m R_c/2$$

$$g_m \cong \frac{I_0}{2V_t}$$

Where I_o is the current created by the source, and about half this current is the I_c flowing through the differential amplifier. This current is obviously controlled by the negative feedback loop involving V+, V- and V_c. The full circuit can be seen below in figure 16. The result and effect of the amplification is shown in figure 17.

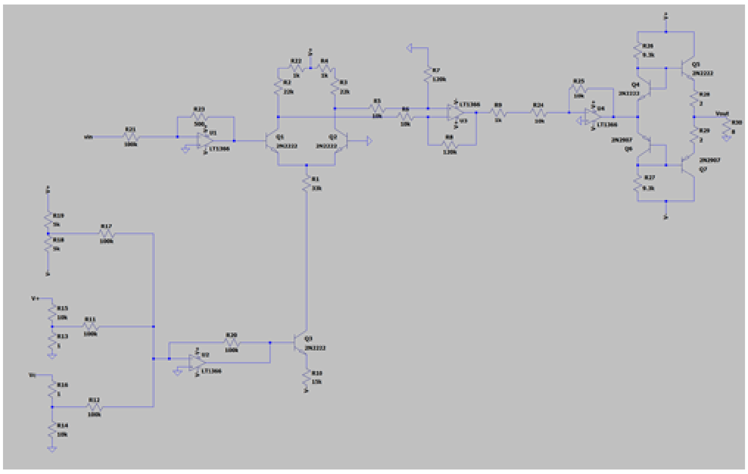


Figure 16 VCA circuit

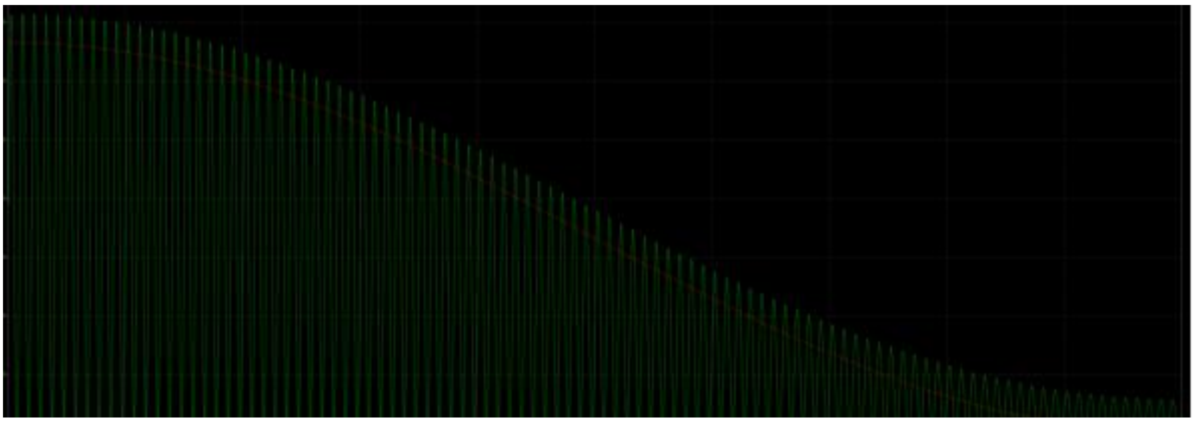


Figure 17 Amplitude modulated sine wave

The second stage of the VCA contains a power amplifier. The point of this is to amplify the current coming out of the previous circuit to load the speaker correctly. The typical current coming into the power amp is between 0.5mA and 1mA, if these were used to load the speaker there would be an output voltage peak of 8mv, which cannot be heard.

Thus, the power amplifier is used to amplify these to peaks of around 120mA which then result in in voltage peaks of around 1 volt, which are heard clearly. This could be louder however we decided not to for 2 reasons. Primarily, the BJTs being used have a current max of 600mA, and thus we decided to limit the current flowing in the circuit to a max 300mA, reducing the typical running current. Moreover, 1 volt is enough to hear a clear crisp sound. The main issues that come with power amps, are cross-over distortion and its power consumption. The best circuit which reduced these issues is the class-AB power circuit, which is the one we decided to implement. This can again be seen in figure 3.

4.6 - ASR

Since the synthesizer is made to simulate ordinary, non-analogue instruments, it is only fitting that the sounds that are produced by the synthesizer sound natural and just as expected. They do that by applying an envelope to the sound signal which simulates the “smoothness” of a sound played by a piano, the “sharpness” of a violin or the “speed” of a drum. Thus, the sound is divided into four parts: *Attack, Decay, Sustain, Release*. **Attack** is the initial rise of the amplitude of the signal when the person presses the key. **Decay** is a small exponential decrease to a given amplitude, where it remains constant for a short time (**Sustain**). Lastly, the **Release** is the exponential decrease of the Sustain amplitude to zero. By changing the duration of each step, sounds such as the “kick”, “snare”, “pad”, and many others are made. These are all sounds that are used in modern music.

Figure 18, Different versions of the ADSR envelope to simulate different sounds.(Swisher, Drew)

In our version of the synthesizer, we decided to use an ASR envelope generator instead of the full ADSR because of some timing problems that we came across, which will be discussed thoroughly later[5]. Both circuits were originally designed by “Kassutronics”. The difference between our ASR and the traditional ADSR is that our circuit combines the Decay and Sustain phases into one single special Sustain. Our Sustain consists of an exponential decrease to a given value (sustain value) followed by the release phase.

[5] The initial ADSR circuit will be discussed in section 4.1.

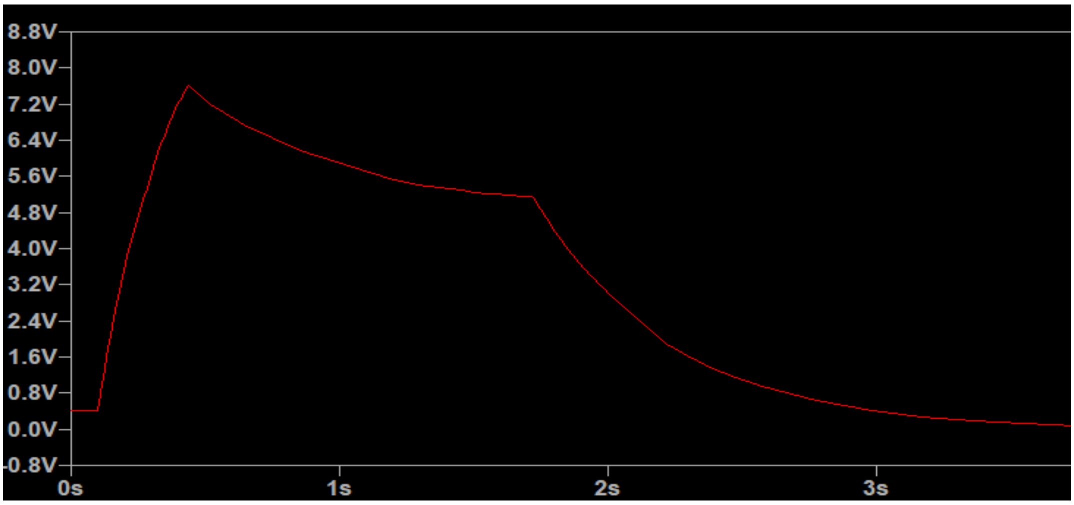


Figure 19, Our version of the ASR.

Figure 20, The input, output, and control values of our ASR circuit (ignore loop)

From the outside, the ASR circuit consists of **one GATE input**, **one output** (of the envelope) and **three potentiometers** (which we represent through three control resistors in LTSpice). The GATE input consists of a pulse of desired duration that dictates the width of the output/envelope. The whole circuit is powered by +/-12V rail supplies. The input block is a Schmitt Trigger that goes high (+12V), when the input pulse is high and low (0V), when the input pulse is 0. This allows for the creation of a well-defined pulse going from -11V to +11V. Furthermore, a short pulse is created from the rising edge of the well-defined pulse, which is directed towards the second Schmitt trigger, which we will refer to as a flip-flop due to its close resemblance to an SR flip-flop. Before the pulse arrives, the flip-flop is in a RESET state of -11V. As soon as the trigger pulse arrives, the flip-flop state changes to SET and goes high to +11V thus, the timing[6] capacitor begins to charge (*ATTACK PHASE)* until it reaches its maximum value, at approx. +8V. The charging rate is dictated by the **Attack potentiometer**. The flip-flop goes low when the timing capacitor charges to +8V, where (because of negative feedback of the flip-flop with the output) the flip-flop state goes to RESET, and the timing capacitor begins to discharge. During that time that the input pulse is still high, and the capacitor has stopped charging, we are in the sustain phase of the output, which means that the discharge of the timing capacitor is limited to the sustain value. The positive input of that op-amp (OP07) is controlled by the **Sustain potentiometer**, that controls the final voltage value of the discharge capacitor (*initial discharge, SUSTAIN PHASE*) which is the initial value of the final phase. When the input pulse goes low, the final *DISCHARGE PHASE* begins. The timing capacitor begins to discharge through the grounded 1MΩ resistor (R12) because of the loop created from the output. The rate of discharge is controlled by the **Release potentiometer**. The output of the timing capacitor is also buffered by an op-amp with negative feedback and is stabilized by a resistor to form the output envelope.

[6] The capacitor is referred to as "timing capacitor" due to its use.

4.7 - LFO

The next circuit that had to be designed was the LFO, as shown in figure 5. This is like the VCO however it does not need to be voltage controlled, just output a set range of low frequency oscillator. The circuit works almost identically to the voltage controlled one, consisting of a Schmitt trigger and integrator. The triangle waves outputs are fed into a Schmitt trigger which causes it to output a square wave. Each of the values the square wave takes are then integrated to give the increasing and decreasing slopes of a triangle wave. The formula for frequency is much easier to derive and interpret.

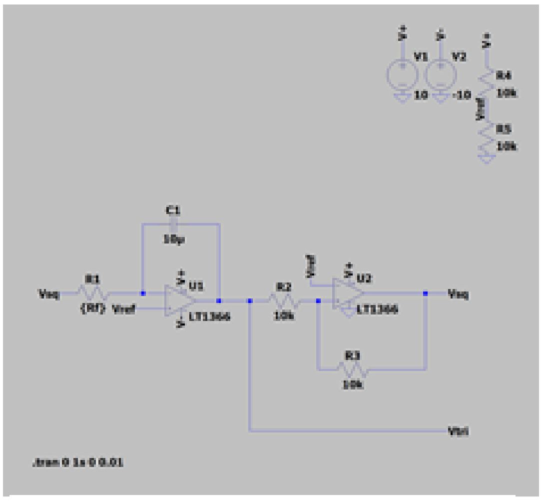


Figure 21 LFO circuit

It is given by:

$$f = \frac{1}{R_1 C} \times \frac{R_3}{R_2}$$

The peak of the output curves is controlled by R3/R2 and in order to maintain them to be the same as the square wave their ratio was set to one, leaving frequency as a function of just R_1 and C. In the real world the resistance can easily be change by a potentiometer, so we set the capacitance and worked out resistance to get desired frequencies of 5, 10 and 20Hz. These waveforms can be viewed in figure 6.

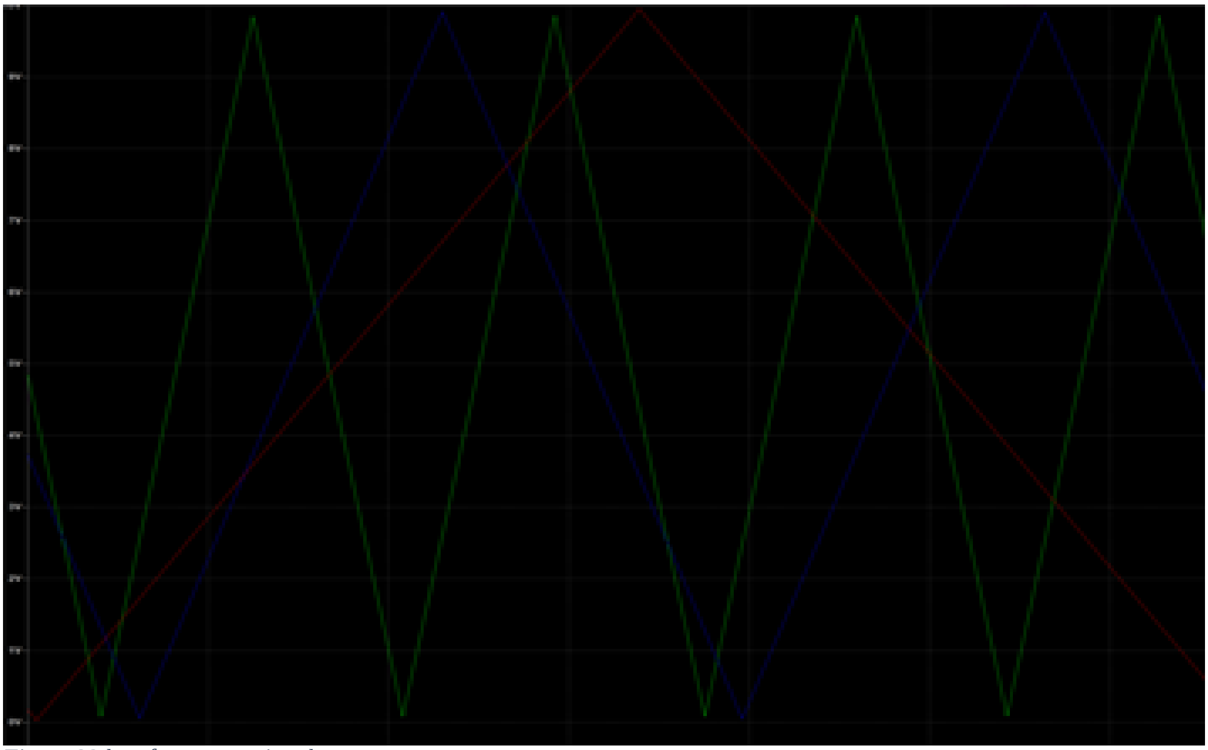


Figure 22 low-frequency triangle wave

4.8 - Integration

Once all the circuits were completed, the logical next step was to put them all together to complete the full analogue synthesiser. The order of the circuits follows that mentioned in the design process. The voltage input goes to the VCO first. Next, the waveform is put through either the low or high pass filter in the VCF. The signal output of the VCF is fed into the VCA control envelope signal is used from either the LFO generator or the ASR signal generator. This signal is used as the control signal in the VCA. The output of the VCA is the final voltage signal. Contrary to our hopes, but in line with our expectation the circuits did not work perfectly together.

Most of these issues arise with the VCA, probably because the myriad of restrictions on its inputs. The first issue is that control voltage signal cannot be too large. This forced me to scale down the LFO and ASR outputs before connecting them to the VCA. Moreover, the input to the VCA that is being amplified must also be scaled down as the gain of the differential stage is already very high. The second condition was on the output of the VCA being inverted which forced me to add an inverting amplifier, to return the original scaled signal.

5 Solutions to technical problems

5.1 - Envelope timing issues

The initial plan was to create the full ADSR envelope instead of the ASR. However, we stumbled upon some timing issues that did not allow us to do so. In contrast to the ASR circuit, the ADSR makes extensive use of transistors as well as contained a quad op-amp and a timer. The specific model used for the quad op-amp was not in the LTSpice library and we did not know what to replace it with. Furthermore, the timer is a circuit which we have never discussed in the lectures and the exact model used in the schematic was an Intersil ICM7555 that was also nowhere to be found in the LTSpice library. As a result, the circuit we built kept outputting an AR (Attack-Release) envelope instead of the full ADSR. According to that, we concluded that there was a problem with the timing capacitor of the circuit; that it was not charging and discharging at the correct times. Since we, as a team, have set deadlines for every circuit component (so we do not lose track of time), we made the calculated decision to stop working on the ADSR and instead work on creating the ASR. Overall, by sacrificing the decay (D) phase of the ADSR, we saved a great amount of time that we believed could be better utilized elsewhere. The ASR circuit is significantly simpler due to the fact that it does not use transistors or a timer. As a result, it works seamlessly with no errors.

5.2 - VCO Tuning

After going through and working out all the expected values we logically tested it with some input voltages and comparing them, to their expected outputs. The circuit did a great job emulating the low frequency waves with errors of around 5-50Hz, however this is probably because as the voltages are really close to 0 the change in output is very small regardless. Despite the accuracy on lower voltages for higher voltages the recorded frequency was sometimes around 500Hz lower than expected. This was obviously an issue that had to be solved before moving on to testing. The reason for the massive error was probably our constant approximation and error propagation throughout all our calculations and putting the circuits together. Nevertheless, this had to be solved. To solve the issue our approach was to solve the scaling on the exponential calculation, and to make the circuit work correctly for the highest potential input voltage. We used Desmos graphing software to help us visualise how they frequency would change by changing different parameters to approximate the effect of each of our changes.

Once we were eventually getting 4.186KHz for 7.25V input as expected we tried a random arbitrary value, to see how far skewed the rest of the curve was. The most essential factor in tuning now was to shift the curve while keeping the 7.25V->4.186KHz relationship intact. To do this we worked what the input values had to sum to with a 7.25 input to result in 4.186Khz wave and made sure all inputs with 7.25 upheld this constant. After establishing this input constant, we could freely change the input scaling and offset until we got a curve that suited the yielded the expected frequency for a second voltage input. Once we had both these 2 points fixed, we now had to test the whole curve, as we could no longer change the inputs without changing one of these. To finally test that it was working as expected, we tried a final arbitrary voltage. It was to our surprise and joy that our circuit outputted a wave with a frequency with only +_30Hz error, which corresponded to around an 8% error. This is error is far less significant towards the extremes of the curve closer to 0.5-3.8% at the max and min frequencies.

The reason we did not continue testing and fine-tuning it was that each Simulation took 10-20 minutes to run so it was extremely tedious and frustrating to make small changes and see the effects. For example, imagine a person trying to tune a guitar but after tightening each string the user had to wait 20 minutes to hear the effects of that change. As a result, we decided this time could be used in other areas that were needed.

6 Testing

After the all the circuits were completed and integrated it was imperative to test how waveforms were synthesised and then how they sounded, hence the testing phase. The testing was conducted primarily by inputting a series of varying input voltages and times via PWL source, to emulate a series of keyboards inputs and analysing the output. The output was checked to see if mainly the frequency but also the waveform and pitch of said wave matched the expectations after being manipulated by the VCF and VCA respectively. The first apparent issue was that the simulations took too long to run. Originally simulations had to be left running overnight to see a result over the full run time. This obviously made testing the synthesiser like this impossible as any changes would take so long to see the effects of and improve upon. To shorten the simulation time, we opted to run each circuit and convert the output to a file of a list of voltages and times to be used as an input to the following circuit. For this to be done reliably circuits have to be coupled correctly to make sure they do not load each other.

After finally being able to reliable view circuit a second issue presented itself, the input to the ASR. The ASR relies on a pulse input to turn it on and off to create the pulse signal, however on LT-Spice we do not have access to button to do this and were originally forced to use a behavioural model originally to test it. However, this is not allowed the final model and thus an alternative had to be found. To bypass this a circuit was added to scale up the original PWL input by 40000 ensuring that the output for any input is the positive power rail of 10v for as long as the key is being simulated. This will always create an ASR signal for the exact same length as the original key input is pressed. After ironing out these first 2 issues below in figure “” a series of waveforms modulated by the ASR can be seen.

The second and final way to test the circuit was to listen to the output waveforms and compare them to certain expectations. Doing this was simple as we only had to select a single output waveform and convert it to a text file to be played by MATLAB. On MATLAB 3 different signals were tested to get a good understanding about the functionality of each of the circuits as there was always many different options for each waveform, such as a low or high pass filter and an ASR or LFO envelope. Moreover, to effectively compare the sounds to real world expectation we converted the tune of Beethoven’s Fur Elise to frequencies and then voltages and played a snippet of the melody on the synthesiser to evaluate how accurate in sounded. The 3 signals played were: a shorter set of a wide range of pulses with an LPF and ASR envelope, a snippet of the melody of Fur Elise with an ASR and no filter (the signal shown in figure 7), and finally the same snippet of the melody but with a 10Hz LFO and HPF. How accurate these sounded will be discussed further in the evaluation.

7 Evaluation

The evaluation of the final circuit consists of 3 main parts, the sound and quality of the analogue synth, the BOM and size of the physical circuit if it were to be built and finally its power consumption. The quality of sound was fully judged based on the sound that was produced and outside opinions were used to attempt to receive biasing, as this is very qualitative and subjective evaluation.

The quality of the raw waveforms coming out of the VCO was alright. It did not sound to sharp or electronic and when playing the melody from Fur Elise it was identifiable despite the timing issues. However, the quality of sound suffers, especially if un-modified from being too low. It was the consensus that the melody sounded off and often people needed to be told beforehand that it was Fur Elise to recognise it. The notes being too low can singly be attributed to the VCO producing waveforms with an error close to 20-30Hz, and this sound difference was especially noticeable in the ranges the notes were played, of 300Hz. Despite this, the signal which is modified via an HPF and LFO sounded better, and the low pitch of the notes was not as intrusive.

In terms of the bill of materials, the actual circuit does not cost too much. The costs here are averages for each component. 91 resistors at $0.79, 11 capacitors at $0.24, $1.89 for a pack of 50 diodes, 30 LT1366 Op-Amps at $0.72 and 7 NPN and PNP BJTs at $0.60. Adding these all together sums to around $102.22 as a cost for all its parts. Despite the low BOM, the glaring issues is the sheer number of Op-Amps. This is a result of a lack of use of non-inverting amplifiers due to an unfamiliarity with he those arrangements and an inefficient integration as certain circuits could have probably been re-used to accomplish more than one function.

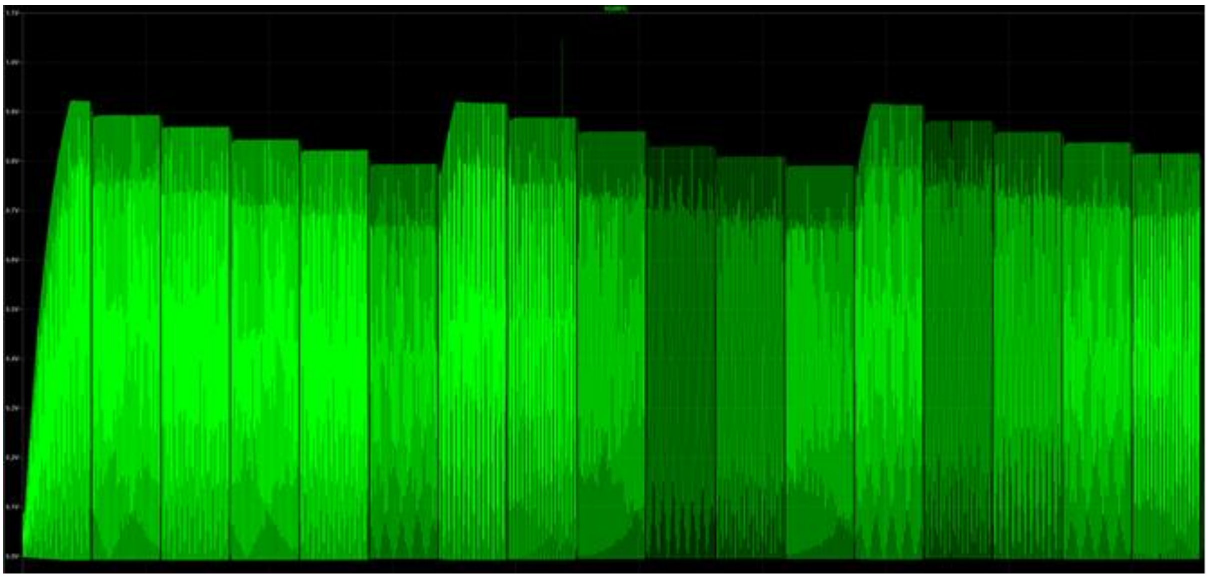


Figure 23 ASR modulated triangle waves.

The final area of evaluation is the power consumed by each sub circuit. To calculate this power consumption of each circuit we looked at the product of the current flowing though the positive voltage rail which is 10V for each circuit. Because of Kirchhoff's current law, all the current that power the circuits must come from here hence giving me the power needed to make each circuit work. In the table below we can see the power consumption of each circuit.

Sub-Circuit	Power (mA)
VCO	92
VCF	125
VCA	50
LFO	56
ASR	180
Total	503

Figure 24 table of power consumption

What can be observed from the table above is that the ASR uses more than triple the power of the LFO, the 2 ways to amplitude modulate our signal. Despite the favourable power consumption of the LFO the ASR is a much more common envelope to be used as it sounds like the classical instruments raising and lowering their voltage. Typically, all these circuits except either the LFO or ASR will be operating so we can expect a total power consumption of either 323mW (LFO on) or 447mW. It should also be noted changing the potentiometers in the LFO, VCF and ASR can reduce the power consumed, however the values above are the maximums.

8 Conclusions / recommendations for future work

After integrating and testing the circuits it is evident the synthesiser works effectively but not perfectly. The primary improvements can be made by considering our 3 flaws, the quality of sound, the functions of the synthesiser, and the build and materials of the synth. The first apparent issue of the quality of the sound being on the low end could have been solved by more careful tuning in the VCO. This was impractical with our time frame due to the sheer sensitivity of the VCO and the time it takes to simulate each waveform. Moreover, there were a lot of factors that affected the frequency relationship so all these would have to be tuned accurately which would have required many more hours than available to us to complete on time.

The second flaw that could have easily been improved upon was the total functions of the synthesiser. Despite the fact that we implemented many different, tough functions of the amplifier such as the full range, the envelope and amplitude modulation we missed out on others. Namely, we did not implement a sawtooth or sine wave, frequency modulation (vibrato) or an arpeggiator. These probably could have all been implemented with an extra week or 2 to design said circuits or had we designed and tested the circuits more efficiently. Nevertheless, we believe we implemented the more important and defining functions of an analogue synthesiser.

The final thing which could have been improved dramatically was the build of the circuit. Frequently, our lack of experience with building circuits is obvious, which resulted in an overuse of certain parts. For example, an over-reliance of inverting amplifier op-amp arrangements for gain functions, resulting in twice the Op-Amps that could have been used using non-inverting amplifiers. Moreover, in certain circuits such as the VCO there is unconventional and probably impractical design choices, such as a DC voltage source used to scale the input voltage. This is a blatant example of a lack of experience with circuit design as there are probably better methods of doing this.

Analysing and evaluating the above improvements it is conclusive that with more time we could have made a more efficient, accurate and impressive synthesiser. However, that is disregarding all the extra challenges that would have arisen in attempt to implement them, implying that we made a good decision on the amount of work our group decided to undertake. We also would have greatly benefitted from more experience with LTSpice, and circuit design. It is inevitable that we must start somewhere, and the experience gained from designing this analogue synthesiser will be invaluable in our future in engineering.

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10 Appendix A - Circuits:

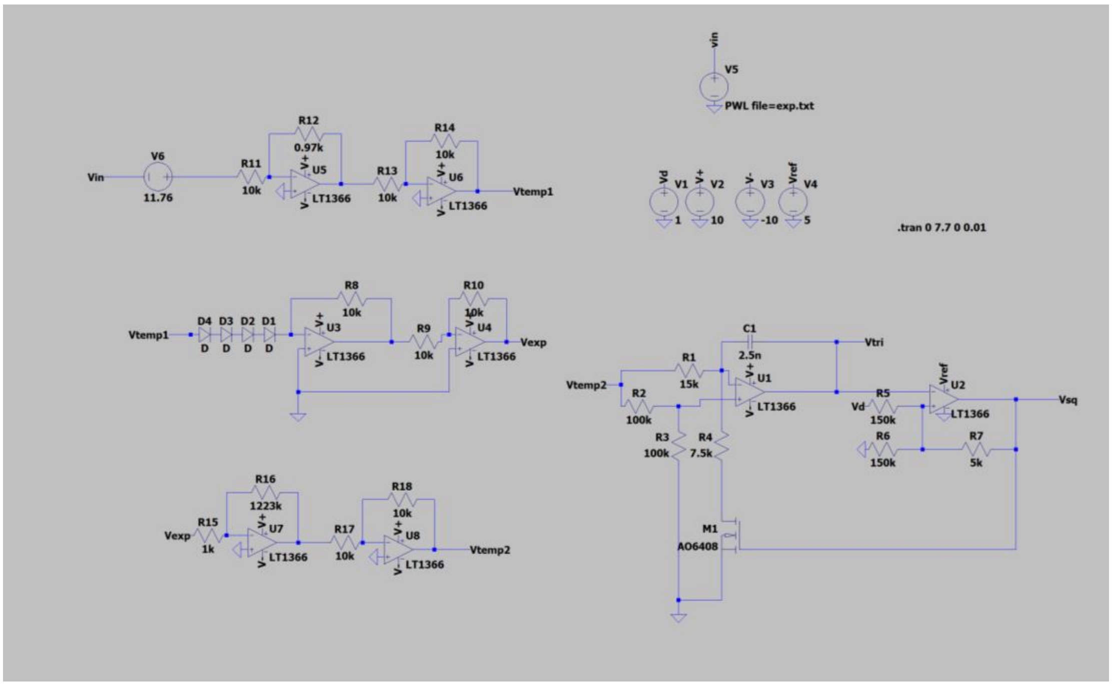

Figure 25 Final VCO circuit

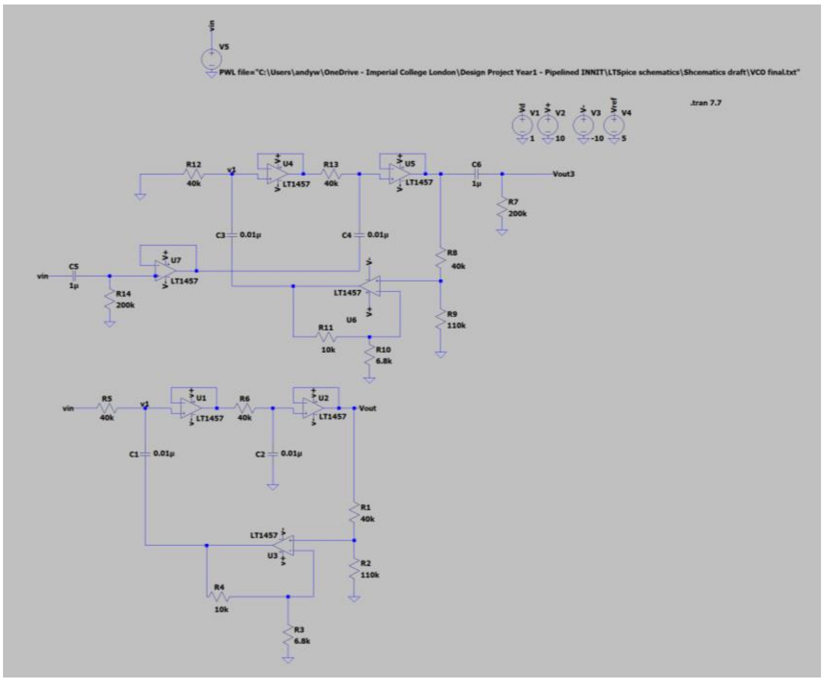

Figure 26 Final VCF circuit

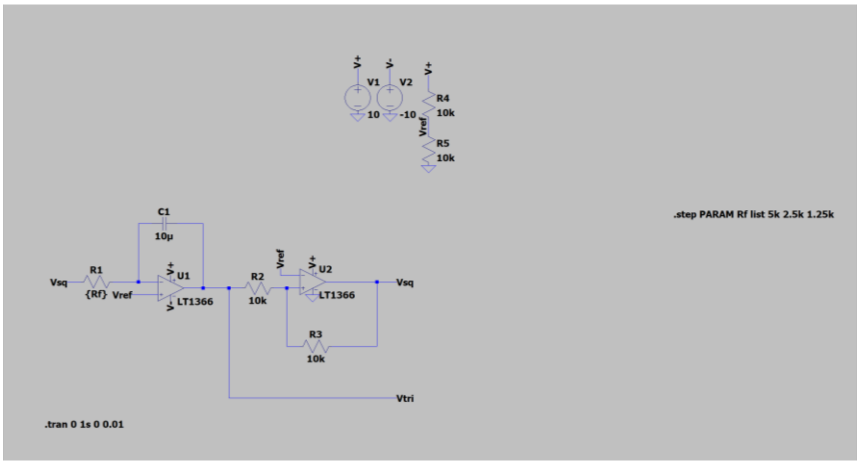

Figure 27 Final LFO circuit

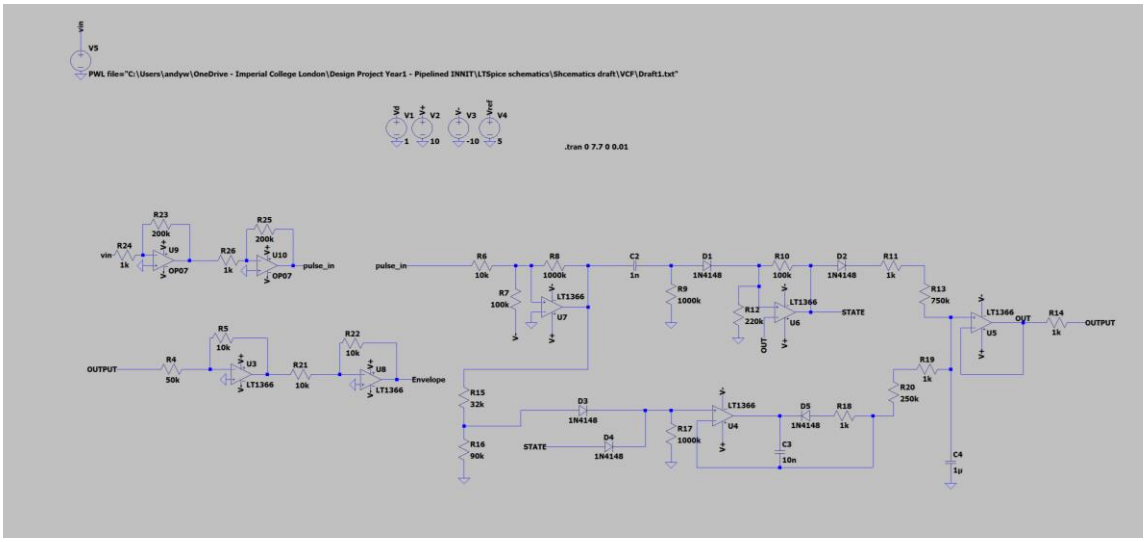

Figure 28 Final ASR circuit

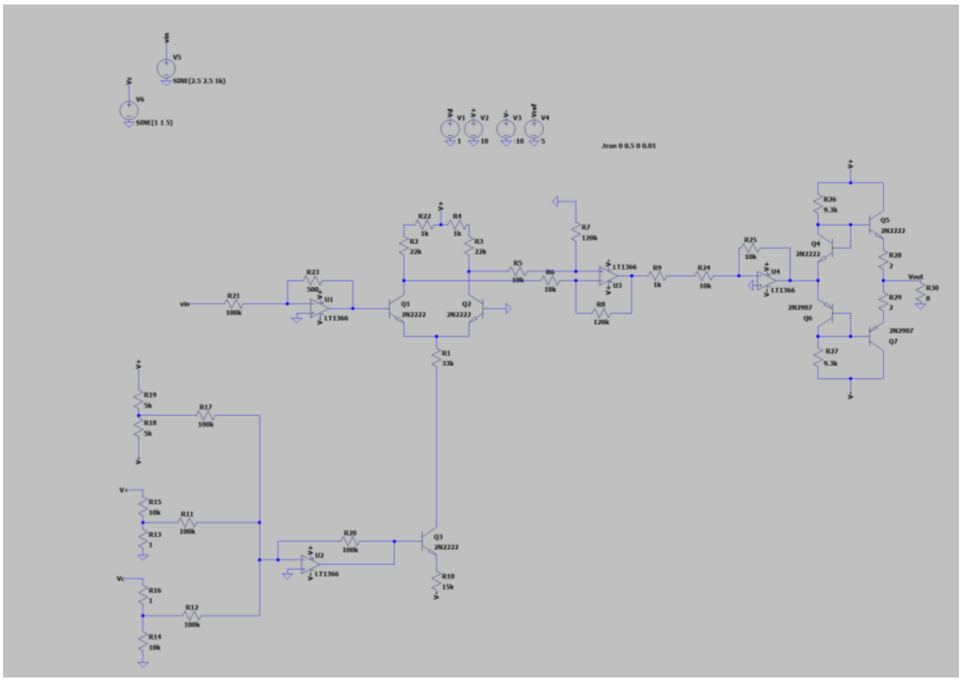

Figure 29 Final VCA circuit

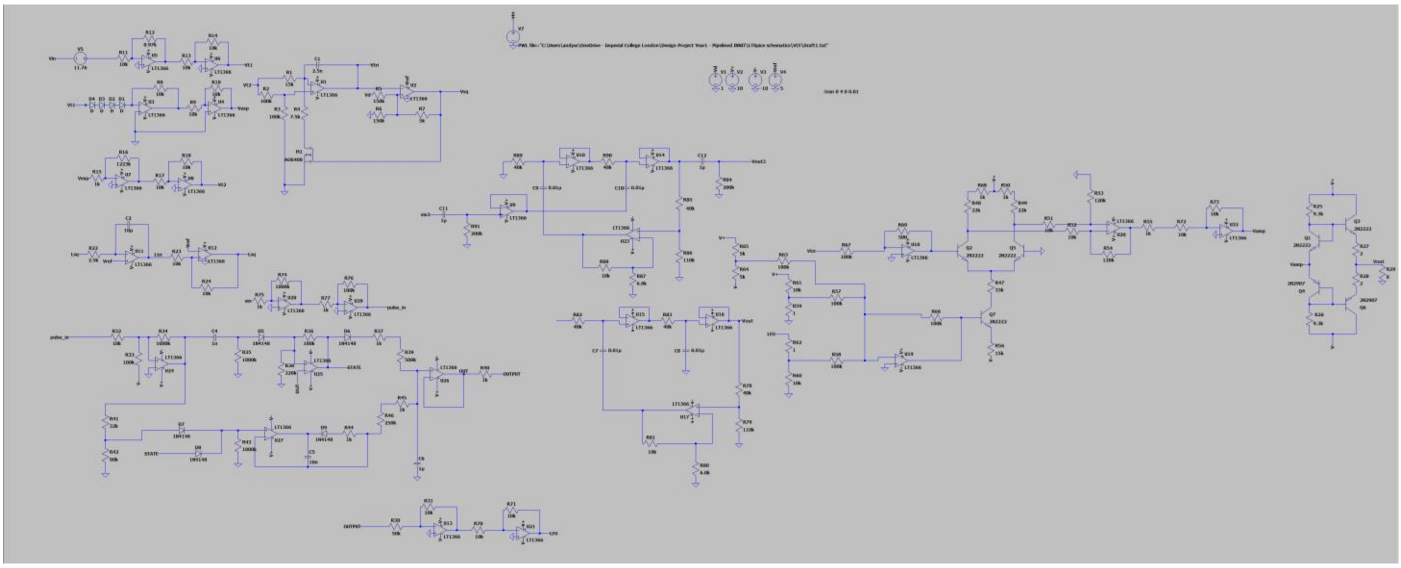

Figure 30 Final VCA circuit