Analogue Synthesizer Project Technical Report

ELEC40006 – Electronics Design Project 2019-2020

Imperial College London

Team YAK

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Contents

1. Introduction

Popular in the 1980's, synthesizers are electronic musical instruments that use analogue circuits to generate audio signals. Music synthesizers can generate sound through a variety of techniques such as additive, subtractive and phase distortion but the defining characteristic is their modularity. This means that music synthesizers generally have a primary oscillator component which generates a signal that can be fed into other modules which modify the signal to achieve the desired output.

The project brief required the team to design a circuit for an analogue music synthesizer capable of generating tones for the seven notes in the C major scale. After some deliberation, it was decided the final product would function by subtractive synthesis. This process uses oscillators to generate waveforms which are then shaped by passing them through filters and other modular components. The task tested the team's technical skills as well as their ability to work interpersonally. Many challenges were faced such as researching unseen components, becoming familiar with new circuits, and most importantly communicating effectively amid a national lockdown.

2. Design Requirements

The assigned task was to design a circuit for an analogue synthesizer and to carry out simulations of the circuit using LTspice.

The design requirements were split between essential and optional specifications.

The essential requirements were as follows:

I. The synthesizer had to generate audio frequency tones for the seven notes in the C major scale, in an octave of the teams choosing.
II. The synthesizer's input had to be seven voltage sources, each representing one key on the keyboard, with 5V representing a pressed key and 0V a released key.
III. All components used in the circuit simulation had to be 'real' devices with part numbers, with an exception made for voltage sources used as DC power supplies or as keyboard inputs. No 'behavioural' components were permitted in the final design.
IV. The synthesizer had to be capable of driving a loudspeaker with an impedance of 8Ω.

Optional design choices included:

I. A choice between monophonic or polyphonic tone generation.
II. The use of discrete transistors or integrated circuits.

A choice of implementing additional functionality was also left open to the team with possibilities including:

I. Selectable output waveforms (to simulate different instruments).
II. Amplitude and frequency modulation (tremolo and vibrato).
III. Enveloping.
IV. Filtering.
V. Low frequency oscillators (LFO) to drive modulation and filter blocks.
VI. Stereo modulation effects.
VII. Arpeggiators and sequencers.

From the list above, the team chose to implement a choice between Square, Triangular, and Sinusoidal output waves, ADSR (Attack – Decay – Sustain – Release) Enveloping, Voltage Controlled Filtering, Voltage Controlled Amplification, a Low Frequency Oscillator and a final power amplification stage to drive the final output load. These were chosen as they allow the user to freely modify the basic characteristics of the sound produced and would cover the fundamental creative needs of a realistic end user.

3. Product Design Specification

The Product Design Specification (PDS) outlines the requirements of the client. The following elements were chosen as priorities when determining the design criteria:

1. Performance:
   - The aim was to produce the highest quality synthesiser with additional functional modules which enable the user to modify the sound as they please.
   - The notes produced should be accurate to standardised frequencies and tunings, so that the synthesizer could be used to perform with other instruments.
   - The circuit design should be as power efficient as possible.
2. Target Product Cost:
   While no formal budget was imposed, the team thought it was important to make sure the synthesizer design was competitively priced relative to other common analogue synthesizer modules on the market.
3. Materials:
   Although the product was developed and finalised in software, it was important to ensure the design could be built physically, keeping in mind the limitations of the simulation software LTspice. Thus, the component selection was limited to real, obtainable hardware.
4. Standards and Specifications:
   It was decided to conform to two common standards of the analogue synthesizer industry:
   - A 12V dual polarity power supply was used, to ensure compatibility with the majority of commercially available analogue synthesizer modules.
   - The output standard of 1V per octave was conformed to for ease of use and compatibility for the end user.
5. Ergonomics:
   The product had to be easily workable and accessible to any consumer. Hence it must include an intuitive, user friendly interface. In practice, if the synthesizer were to be implemented using hardware, this would require well labelled, tactile knobs and switches.
6. Testing:
   - For each sub-system in each module, comprehensive tests had to be carried out with recordable results in the form of plots and or audio files.
   - The tests covered any output waveforms produced by the circuit and also tested the input and control variables.
   - Reasonable combinations of the completed modules were also tested, and the results were noted.

There is further discussion of these points and more within this report.

4. High Level Design

One of the advantages of analogue synthesizers is that they are completely modular by design. This lets the user rearrange modules to suit their individual needs and to modify the output sound to their liking. To retain this functionality, each of the functions was implemented as discrete modular blocks.

I. Voltage Controlled Oscillator (VCO)
II. ADSR Envelope Generator
III. Voltage Controlled Filter (VCF)
IV. Low Frequency Oscillator (LFO)
V. Voltage Controlled Amplifier (VCA)
VI. Power Amplifier

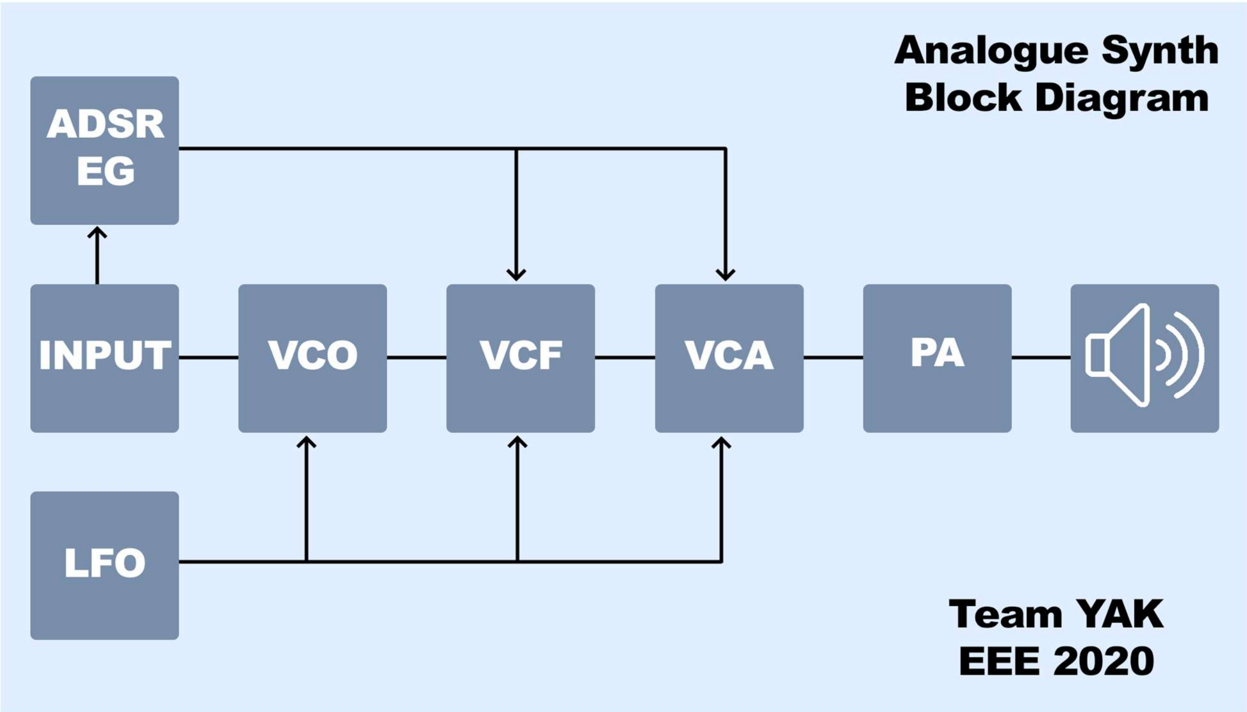

Figure 1: Analogue Synthesizer Block Diagram

5. Module Design Overview

I. Voltage Controlled Oscillator (VCO)

The VCO can be broken down into three main sections: the input block, exponential converter, and the main Voltage Controlled Oscillator.

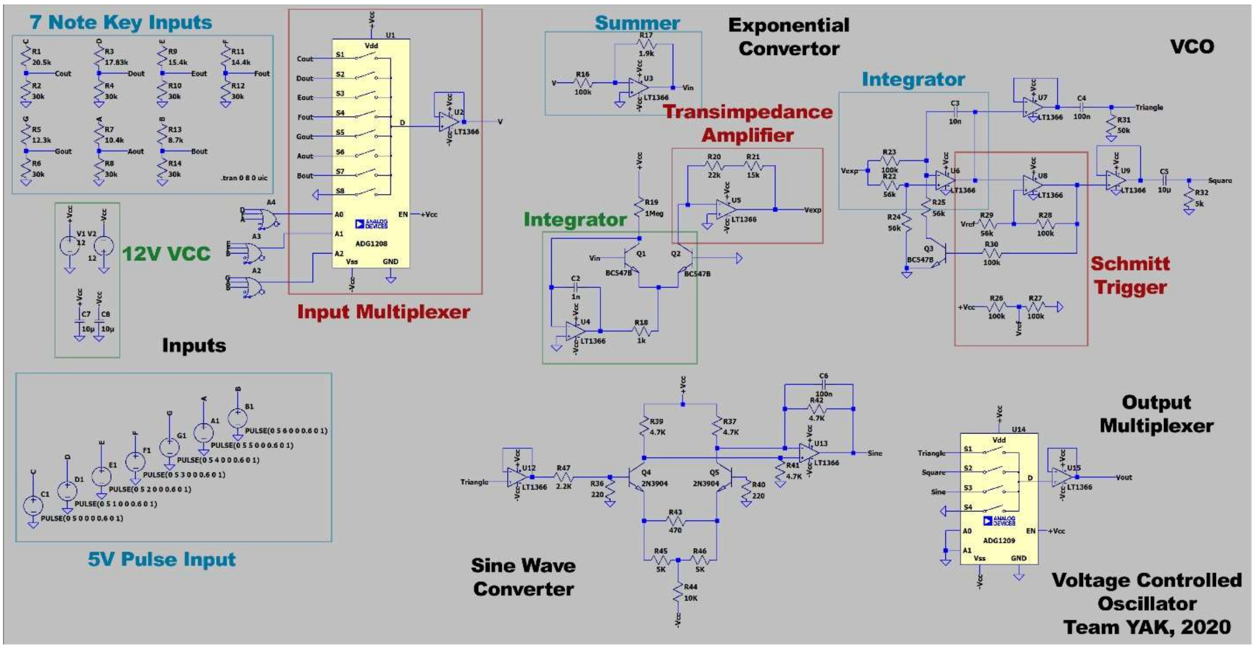

Figure 2: Complete Voltage Controlled Oscillator Circuit Diagram

i. The Input Block

Seven voltage dividers and 5V voltage sources were used, each representing one note of the C-Major scale. This followed the brief which required that 5V represented a pressed key and 0V or open circuit represented a released key. Since the output frequency for each note was tied to the input voltage, potential dividers were used to bias each input to the correct voltage while maintaining the standard set out in the brief.

While practically a potentiometer or variable resistor would be used to perfectly calibrate the voltage to the preferred tuning scheme, in simulation normal resistors were used in a potential divider configuration with values which give as accurate a tuning as possible.

The target frequencies for each note have been summarised in the table below [1]:

Note	Frequency (Hz)
C3	130.81
D3	146.83
E3	164.81
F3	174.61
G3	196.00
A3	220.00
B3	246.94

Table 1: Frequencies for C3 - B3

In the circuit simulation voltage sources were used in place of the push switches that would have been available practically. One resistor in each pair would be replaced with a variable resistor or both with a potentiometer as that would provide more precise control over the voltage and subsequently the input tuning.

The figure below shows the complete circuit diagram for the VCO's input block.

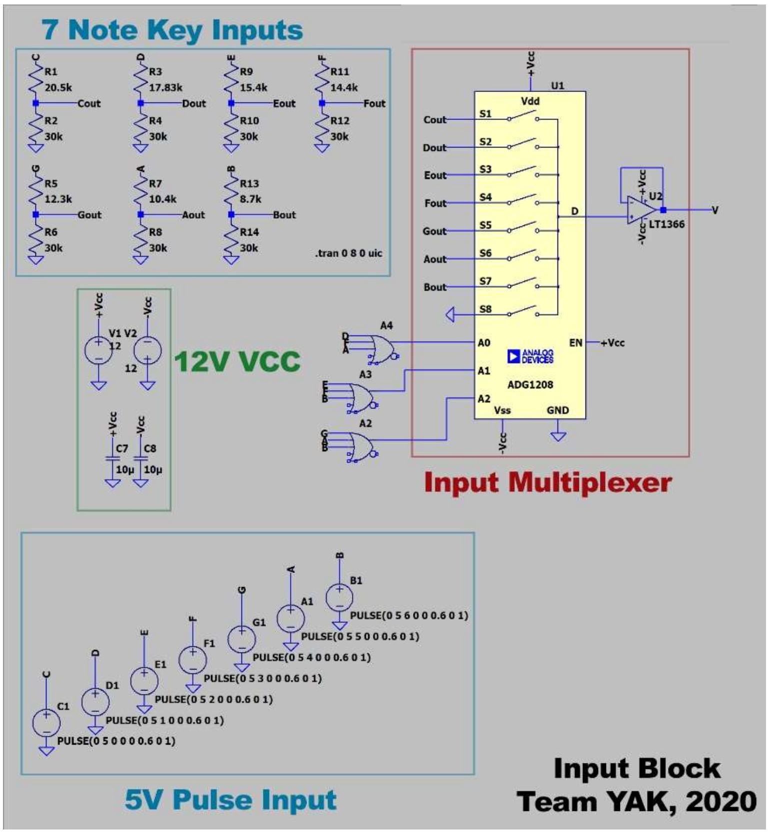

Figure 3: VCO Input Block Circuit Diagram

All seven of the inputs were then fed into an eight-input multiplexer for ease of simulation. Practically, a summing amplifier circuit would be used, which is why the input block is distinct from the exponential convertor block which does include a summer.

ii. The Exponential Convertor

The function of the exponential convertor is to take a linear voltage as its input and convert it into an exponential voltage output. This is necessary as, while doubling frequency makes the tone generated increase by one octave, doubling voltage does not necessarily achieve the same thing. Using an exponential voltage curve allows the voltage to scale at the same rate as frequency and helps achieve the 1V per octave on the output of the VCO.

The converter shown, which is adapted from the one by the creator 'SomethingsGoneWrong' [2], has the following sub blocks:

- Summer:
  Converts multiple inputs into a single output. It has a variable resistor on the inverting input in order to set the output gain of the op-amp.
- Integrator:
  Performs an integration function and thus converts the linear input voltage into an exponential output voltage. The integrator maintains a constant current across NPN transistor Q1. Changes to the input voltage will result in a corresponding change to the VBE of transistor Q2. This results in an exponential change to the collector current of Q2.
- Transimpedance amplifier:
  Since we require the output voltage to be exponential and not the output current, the transimpedance amplifier is used to convert an input current into a proportional output voltage. This active amplifier is preferred to an ideal ohmic resistor as it has a good signal to noise ratio and a flexible output voltage range [3].

The figure below shows the complete circuit diagram for the exponential convertor:

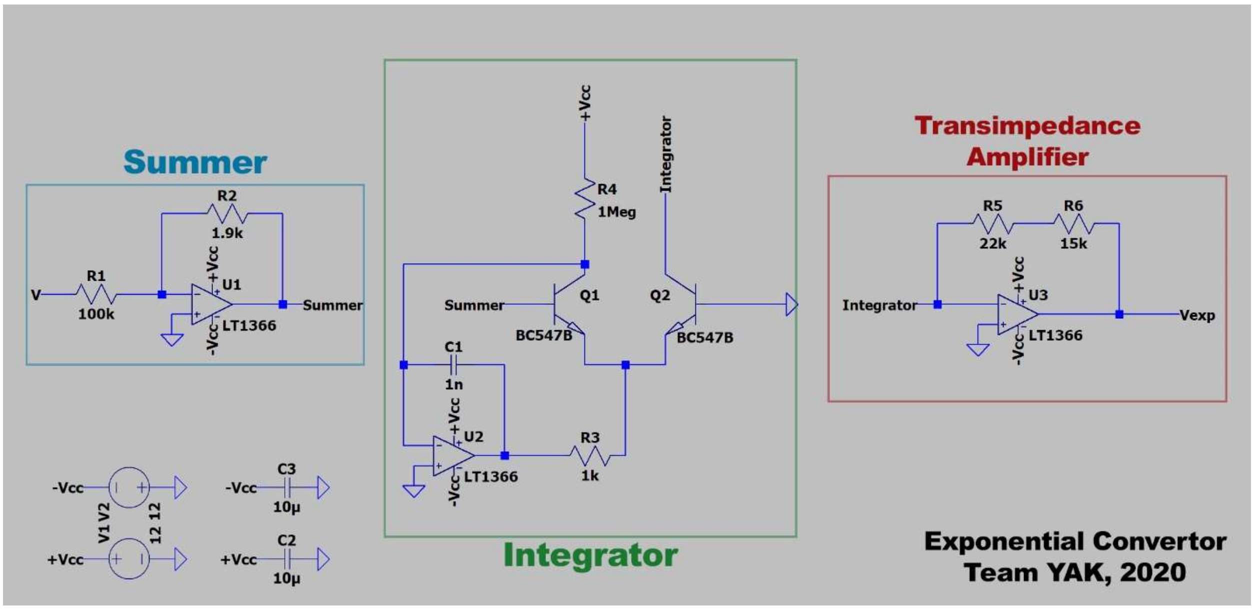

Figure 4: Exponential Convertor Circuit Diagram

The conventional standard for analogue synthesizers is 1V per octave on the keyboard. It was decided to conform to this standard in the team's synthesizer implementation. The graph below shows the input vs output characteristics of the exponential convertor.

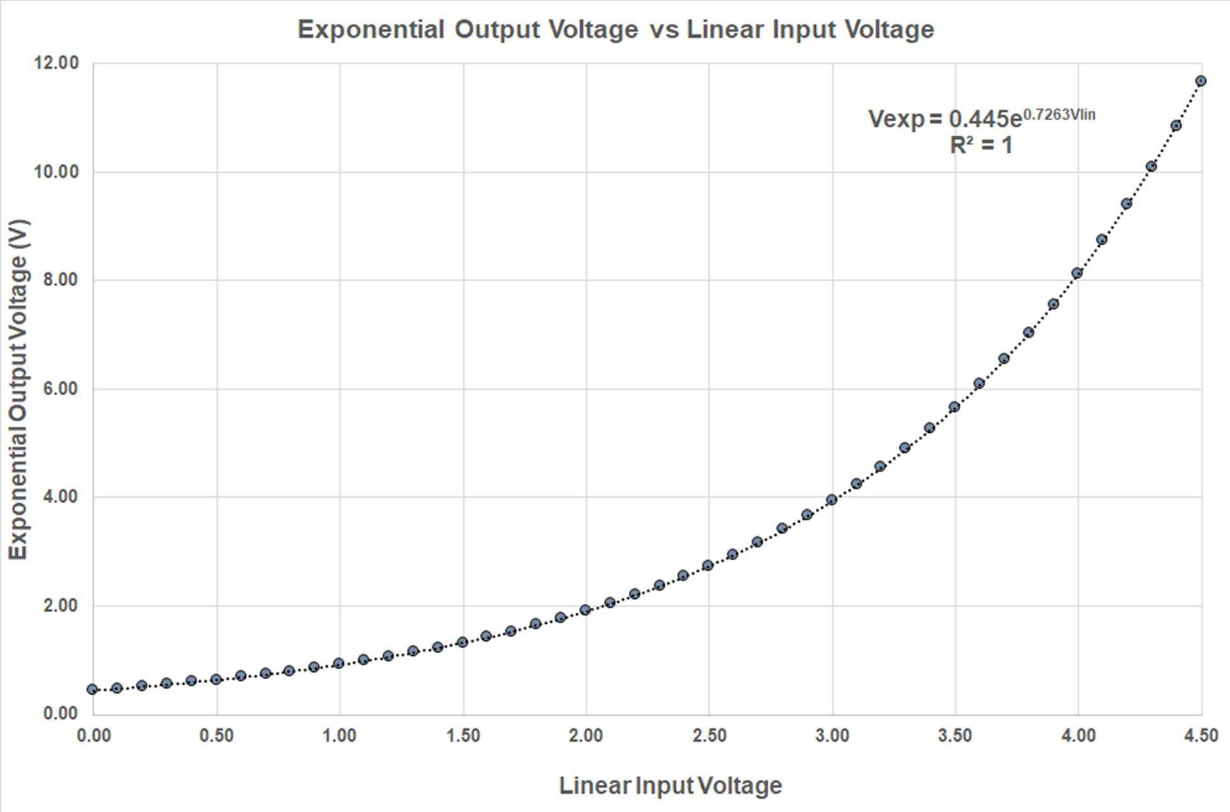

Figure 5: Exponential Convertor Output vs Input Voltage

Clearly, since the components being used are ideal the linear input voltage is perfectly converted into an exponential output voltage which follows the 1V per octave industry standard. In practice there would be some deviations, however these could be corrected for by fine tuning the pitch using the input potentiometers.

iii. The Voltage Controlled Oscillator

This is the signal generator of the synthesizer. It is an oscillator where the output frequency is proportional to the input voltage. In our circuit this will be used to produce output signals in the form of triangle waves or square waves which can be further modified into sine waves. Based on a common design, this VCO comprises two main blocks, an integrator circuit, and a Schmitt trigger [4].

- Integrator:
  Operates similarly to the integrator in the exponential convertor circuit. The output of the integrator depends on the state of Q1. If Q1 is off, then C1 will charge causing the output to fall gradually. If Q1 is on, then C1 will discharge and so the output rises. The output feeds directly into the Schmitt trigger.
- Schmitt trigger:
  The trigger can do two things depending on the input from the integrator; if its input crosses the upper threshold, the trigger output will be 0V. If the input crosses the lower threshold, the trigger output will be high.

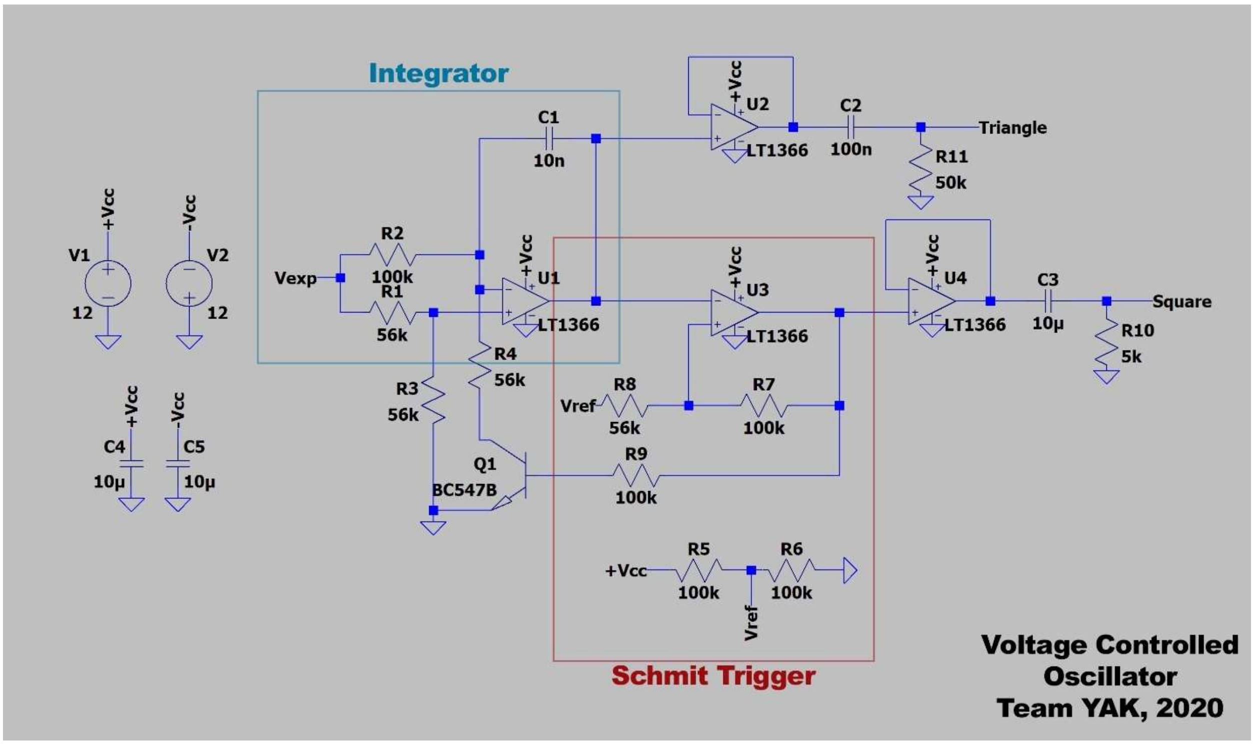

Figure 6: Voltage Controlled Oscillator Circuit Diagram

The output from the Schmitt trigger is then fed into Q1, so the transistor is only on when the trigger's output is high.

There are separate outputs for the triangle and square waves. The triangle wave can be converted into a sine wave using a differential amplifier and an active low pass filter. The differential amplifier consists of two NPN transistors (Q1 and Q2) which is effective in reducing the noise in the signal. The active low pass filter using op-amp U2 then filters the triangle wave input into a sine wave output.

The circuit diagram for the Sine Wave Converter has been shown in the figure above.

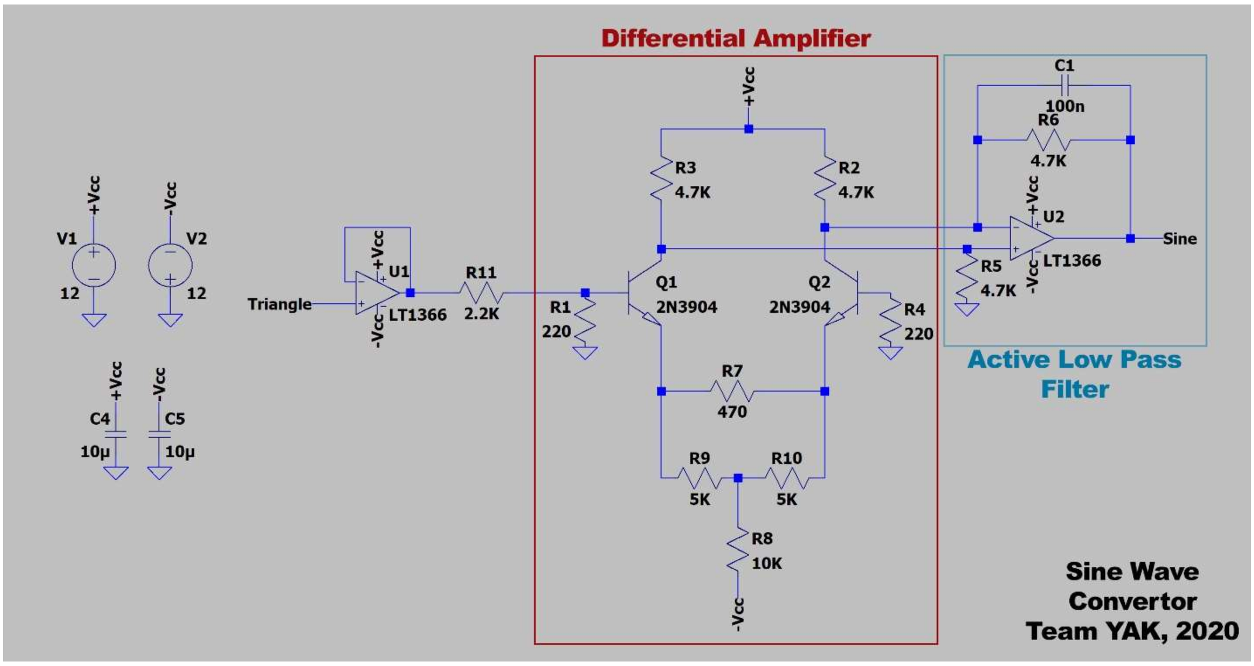

Figure 7: Triangle to Sine Wave Convertor Circuit Diagram

The three output waves from the voltage-controlled oscillator are finally fed into a multiplexer which is used as an input for the user to select which output waveform they would prefer. Alternatively, this could easily be implemented in practice with a three-point switch, however for ease of simulation, a multiplexer was a reasonable alternative when designing a summing amplifier.

The output multiplexer has been shown in the figure below.

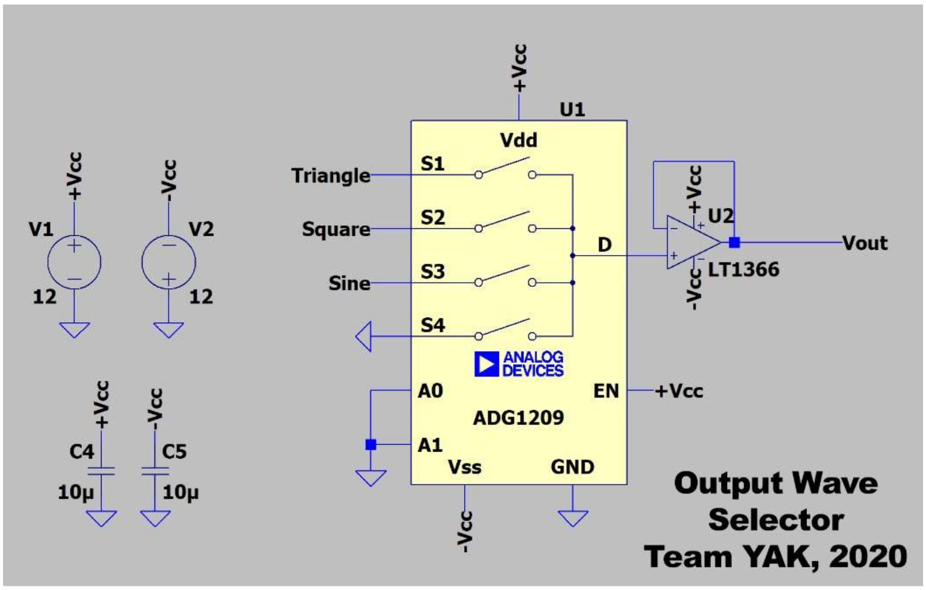

Figure 8: Output Wave Selector Circuit Diagram

II. ADSR Envelope Generator (EG)

An envelope refers to how a parameter changes over time. In analogue synthesis envelops can be applied to any parameter of the synthesizer to modify the parameter over time. ADSR is a fundamental type of envelope generator which controls four basic parameters:

- Attack: This is the amount of time taken for the output voltage to reach its peak value from 0. This time begins when the key is depressed.
- Decay: The time it takes for the output to decrease from its peak value to its designated sustain level.
- Sustain: The level held as long as the key is depressed.
- Release: The amount of time for the level to reach 0 from the sustain level after the key is released.

Adapted from schematics by R.Schmitz [5], the figure below shows the circuit diagram of the ADSR Envelope.

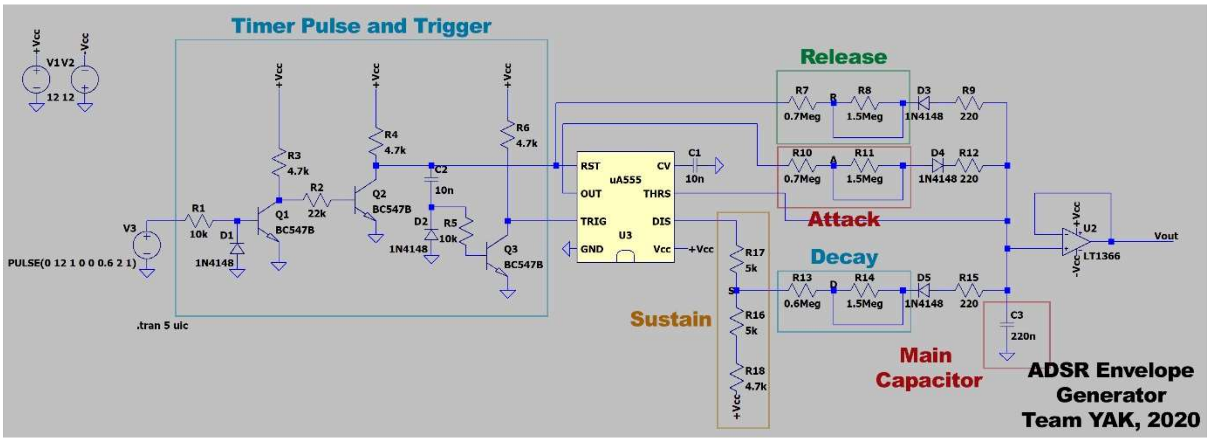

Figure 9: ADSR Envelope Generator Circuit Diagram

The main mechanism used to vary these parameters is the charging and discharging of a capacitor. The main capacitor being charged and discharged at different times is capacitor C3. The transistors (Q1, Q2 and Q3) in the “Timer Pulse and Trigger” section create a short pulse into the trigger input (*TRIG*) of the 555 Timer when an input is detected. They also keep Reset (*RST*) high while the input is high to ensure it the internal timing of the circuit is not reset which also prevents the Release of the envelope from occurring.

After the trigger receives the falling edge generated at the collector of Q3 it sets the *OUT* pin to high and activates the timing interval in the 555 Timer. The current from the output flows through the Attack potentiometer (R10 and R11) and charges capacitor C3 until its potential difference reaches two thirds of *+Vcc* (the threshold voltage) which then sets *OUT* to low.

The capacitor C3 will then lose charge through the Decay potentiometer (R13 and R14) and Discharge pin (*DIS*) until the voltage across the capacitor is equivalent to that dictated by the Sustain potentiometer (R16 and R17) which can be between 0V and 8V.

The voltage across the capacitor remains in this state until the input is no longer high after which it will discharge through the Release potentiometer (R7 and R8).

The timing for the Attack, Decay, and Release phases is dictated by the RC combinations between each potentiometer and the capacitor C3. Therefore, by changing the resistance (via the potentiometers) the shape of the envelope output can be modified.

III. Voltage Controlled Filter (VCF)

The Voltage Controlled Filter is an additional module that allows the user to affect the sound of the incoming signal using a similarly standardised input method.

The filter can take many forms such as:

- Low pass
- High pass
- Band pass
- All pass
- Notch pass

These filters allow certain frequency ranges though while significantly attenuating the frequencies outside that range past a certain 'Cut-Off Frequency'. They also have the effect of producing a significant peak in gain around this cut-off frequency called resonance [6].

It was desired to be able to control these two factors in our filters using our standardised voltage inputs. Auxiliary inputs like that from the LFO that add additional effects to the signal were also chosen to be implemented.

An ideal filter would have an infinitely steep gradient past the cut-off frequency, however, in real terms this would neither be necessary or practical as it would require cascading of multiple orders and provide heavily diminishing returns. Additionally, a steep cut-off might not be beneficial to the output sound of the synthesizer as a first order circuit would likely be able to provide adequate filtering while simultaneously contributing to the 'character' of the sound created.

After researching the function and structure of popular VCFs, the following design was developed as an initial iteration based on designs by R. Wilson [7].

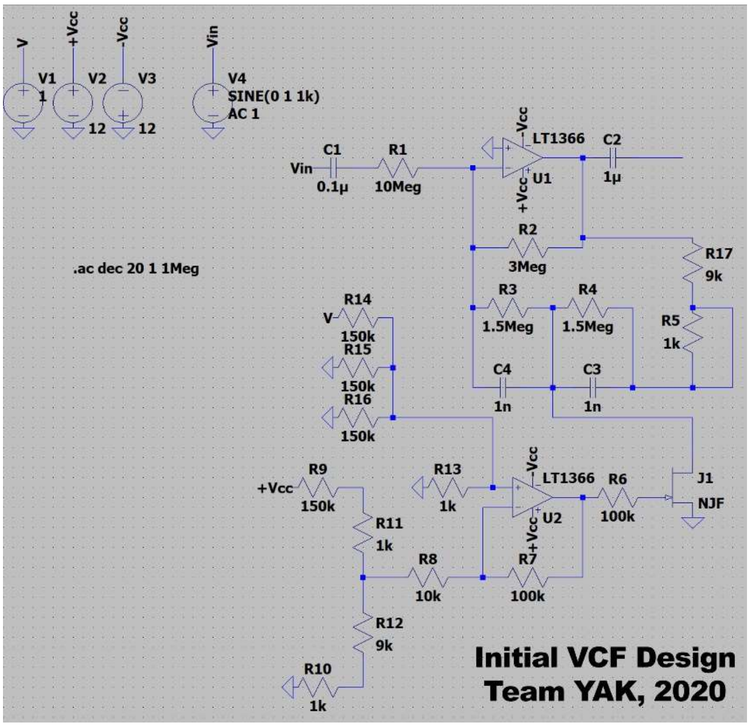

Figure 10: Initial VCF Circuit Design

In this circuit, the input signals were fed into an op-amp configured as an active lowpass filter with a gain of 0.3. Much like the VCA, this circuit also relied on a JFET which behaved as a voltage-controlled resistor. It was connected between two capacitors in the feedback network of the op-amp. The control voltages were used to change the voltage at the gate of the JFET which changed how close the capacitor junction node was to ground value. This consequently altered the cut off frequency of the filter.

Although this filter produced good results when tested as an isolated component, great difficulty was faced when integrating it into the rest of the circuit. The filter was not able to produce the desired output so it was decided that this design would be abandoned. After conducting more research, a second much simpler design was found which worked more effectively with the rest of the circuit [8].

The figure above shows the simple low pass VCF circuit that was implemented.

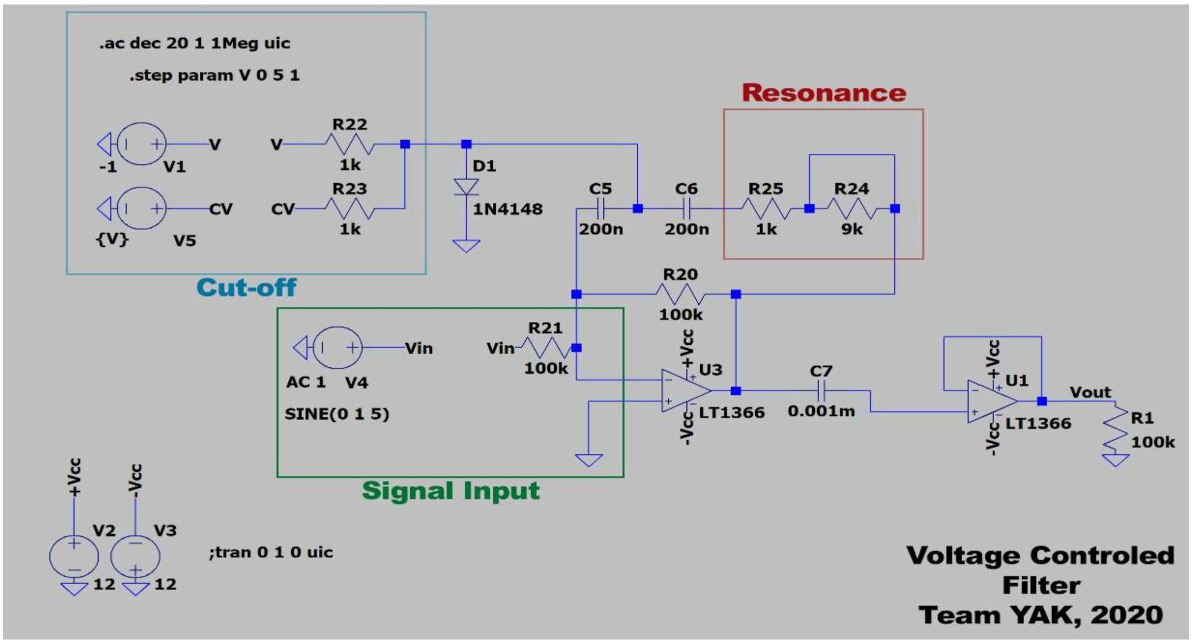

Figure 11: Voltage Controlled Filter Circuit Diagram

The central component in this design was an op-amp configured for use as an active low-pass filter. Unlike their passive counterparts, active filters have a very low output impedance which means the load does not affect the frequency response of the filter. This is what allows the circuit to be a standalone module which is an essential quality for components of a music synthesizer.

A potentiometer (R25 and R24) was placed at the top right of the feedback network. When altered, this changes the resonance of the system, which affects the degree to which frequencies are emphasised at the cut off frequency. The two inputs at the top left of the circuit are very important as they control the cut off frequency of the filter. By changing their value, the two inputs can modify the cut-off frequency which will change the frequencies that are present in the output waveform. The diode D1 will provide a short circuit to ground if the voltage at its terminal is above 0. Additionally, a second op-amp configured as a voltage follower was added at the output node to provide efficient isolation of the output from the input signal.

IV. Low Frequency Oscillator (LFO)

The LFO is another type of oscillator in the circuit which can generate three types of signals, a triangle wave, square wave, and sine wave. The main difference between this and the VCO is that it produces oscillations at a much lower frequency. This is necessary as its output waveforms are used as control signals within the synthesizer circuit rather than as output sound [9].

The figure below shows the Low Frequency Oscillator Circuit adapted from an online source [10]:

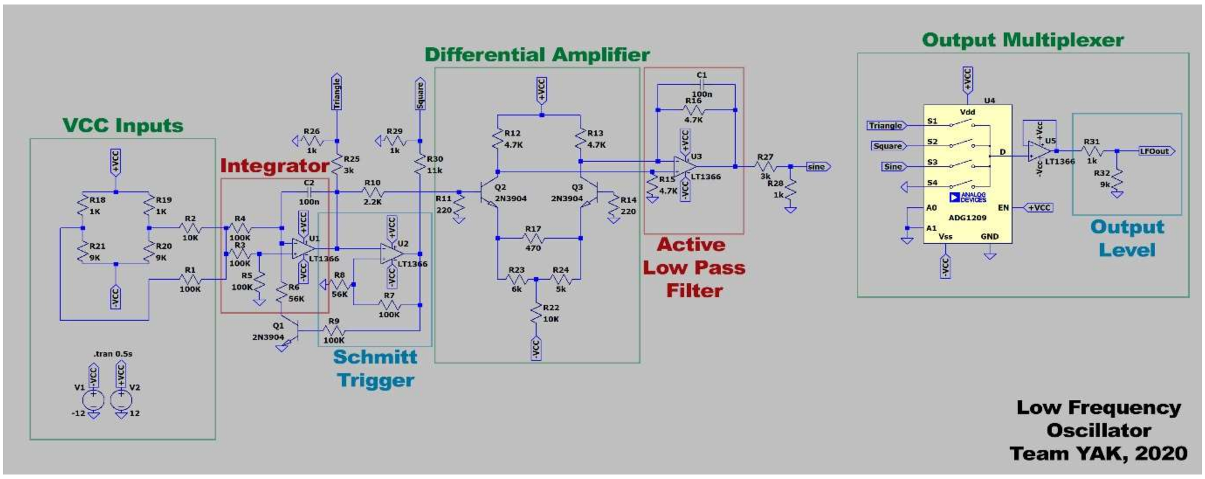

Figure 12: Low Frequency Oscillator Circuit Diagram

By adjusting the potentiometer represented by R18 and R19, the frequency of the LFO can be adjusted. Here a potential divider has been used, but in practice a potentiometer would be preferable.

Additionally, the LFO requires a relatively small peak to peak voltage. This was achieved by adding potential dividers on each of the three waveform outputs, before the output multiplexer, such that each waveform entered the multiplexer with a peak to peak voltage of 1V. After passing through the output multiplexer, one additional voltage divider (R31 and R32) was used to set the output level so as to have a chosen amount of LFO influence on a parameter. Practically, a potentiometer would be used here as well.

V. Voltage Controlled Amplifier (VCA)

The VCA is an amplifier component where the gain is controlled by a control voltage. A higher voltage would allow more of the signal through to the output. The control voltage could be from the output of an envelope generator such as in the image below.

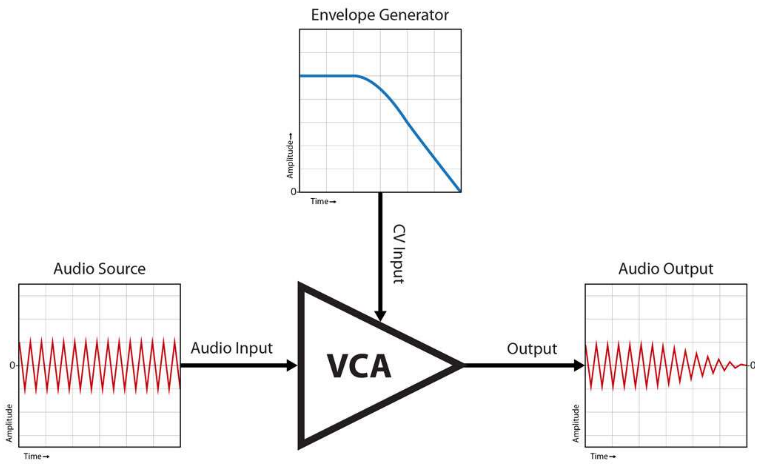

Figure 13: Effect of an Envelope on the Voltage Controlled Amplifier Source adapted from [12]

A setup such as this would be used to achieve more realistic sounds from a music synthesizer. One example of this is how, on a piano, the volume of the sound decreases with time. Another common effect that could be implemented is tremolo, where the control voltage is a sine wave.

After extensive research, the following Voltage Controlled Amplifier circuit was developed [10].

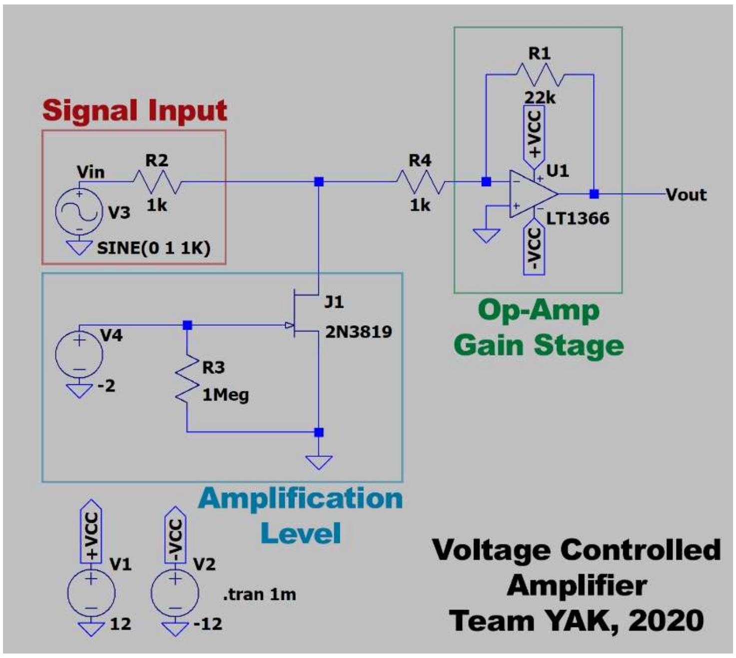

Figure 14: Voltage Controlled Amplifier Circuit Diagram

The main body of the circuit consists of three op-amps. U1 and U2 are configured to work as negative feedback circuits and U3 is simply a voltage follower which essentially isolates the input of U2 from the output of U1. The N-channel JFET which is analogous to a voltage-controlled resistor. When the gate voltage is at 0, the JFET is 'fully open' which is where there is the lowest signal attenuation. A more negative gate voltage will cause the component to be more 'closed' and this would allow more of the input signal through to the gain stage and thus amplifies the signal [9].

VI. Power Amplifier (PA)

The power amplifier was the final stage of the circuit, acting as a current amplifier to drive the 8Ω loudspeaker at the output. The circuit will largely focus on amplifying the current while keeping a unity voltage so that the output power is sufficient. Ideally the output equivalent resistance will be 8Ω (equal to the load) to allow maximum power transfer.

The power amplifier will be the final stage of the circuit and act as to allow the synthesizer to be effective at driving the loudspeaker.

Ideally the stage will have:

- Unity Voltage gain
- High current gain
- 8Ω Thevenin output resistance
- Low power consumption

The requirement on the output resistance is because of the maximum power transfer theorem (citation) which dictates that for the greatest transfer of power from a source to the load the resistance of the output should be equal to that of the load.

The voltage gain is ideally unity as this would dictate the amplitude of the wave which we wish to control via another stage in the circuit so should remain unaffected through this stage. The current gain ensures that the output can drive the loudspeaker and power consumption helps the efficiency of the entire circuit and reduces waste.

With this stage it is important to maintain the fidelity of the signal, avoiding the addition of unwanted noise or distortion, as well as ensuring minimal ambient power consumption.

Initially, a class B power amplifier was considered as shown below.

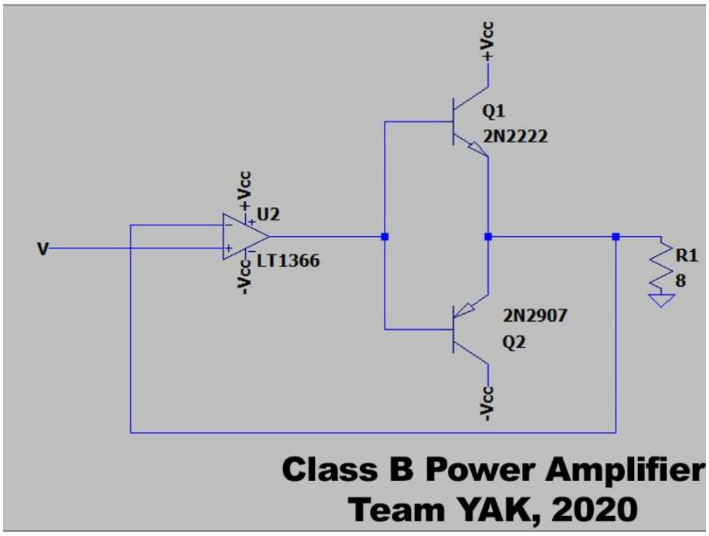

Figure 15: Class B Power Amplifier Circuit Diagram

This is a very common design as it is simple (uses very few components) and seemed quite effective at the start. Significant errors were encountered, however, with the increase of input frequency or amplitude. These mainly appeared as crossover distortion and the loss of a sine shape in the output signal. This is likely to do with limitations in the gain-bandwidth product and slew rate of the op amp.

For these reasons, this design was abandoned.

Due to the previous circuit being incompatible with the team's implementation, it was decided to implement a class AB power output stage. The circuit diagram below was adapted from a piece by Dr P. Mitcheson which describes the structure of a typical class AB amplifier [11].

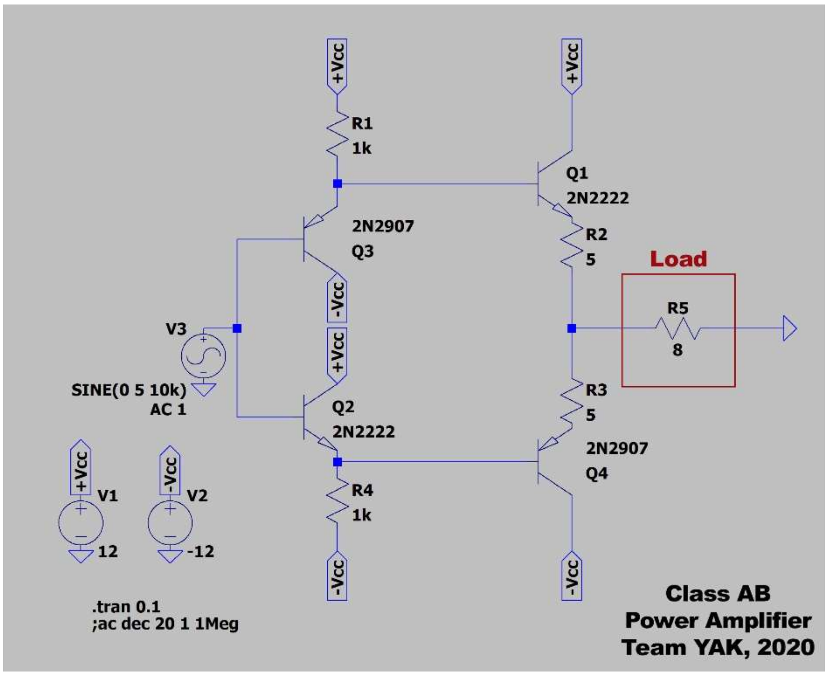

Figure 16: Class AB Power Amplifier Circuit Diagram

The module is the final stage of the design and its main function is to boost current to sufficiently drive the load, which in this case, was an 8Ω load.

This updated circuit overcomes the problem faced by the previous one, cross over distortion. It achieves this by using Q1 and Q2 to ensure that at least one transistor is conducting at all times. Furthermore, the two resistors at R3 and R4 provide some bias stabilization which prevents thermal runaway and protects the transistors.

6. Performance and Results

This section will go over the simulations conducted on the various modules and the operation of the synthesizer with the blocks all operating together. It contains output bode plots and or transfer functions for each of the modules as well as output waveforms for the complete synthesizer.

I. Voltage Controlled Oscillator

The VCO circuit was capable of producing triangular, square, and sinusoidal waves. The plot below shows the three, selectable output waveforms.

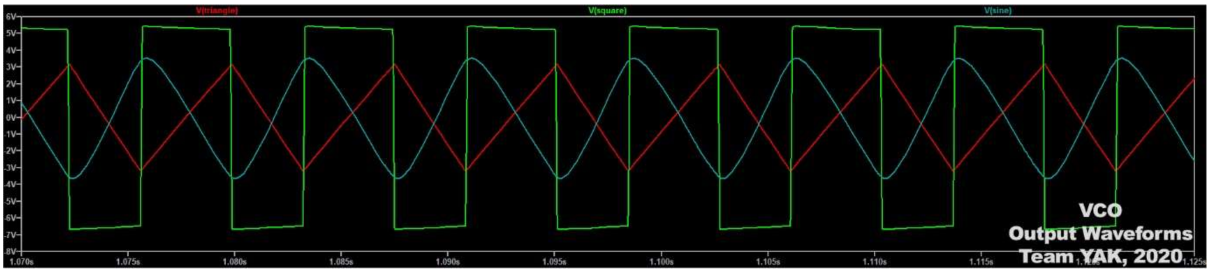

Figure 17: Voltage Controlled Oscillator Output Waveforms

The presence of a sloped voltage at the peak and trough of the square wave is due to the DC blocking capacitor at the end of the oscillator. This was used so that the signals oscillated around 0V allowing more head and legroom without the introduction of distortion. For the same reason, slight curvature can also be seen in the triangular wave at certain frequencies.

The inaccuracy in the square and triangular wave can be reduced if a larger capacitor or resistor is used. However, this also results in a longer time at the start-up of the circuit before the waves oscillate around 0V, as the initial DC bias is around 6V. The chosen values consider the need for a quick decrease of DC bias while also trying to maintain the ideal shape.

The plot below shows the different obtained output frequencies for each of the seven notes in the C-Major Scale. The sine output wave has been used for demonstration purposes; however, the frequency was consistent for all three selectable waveforms:

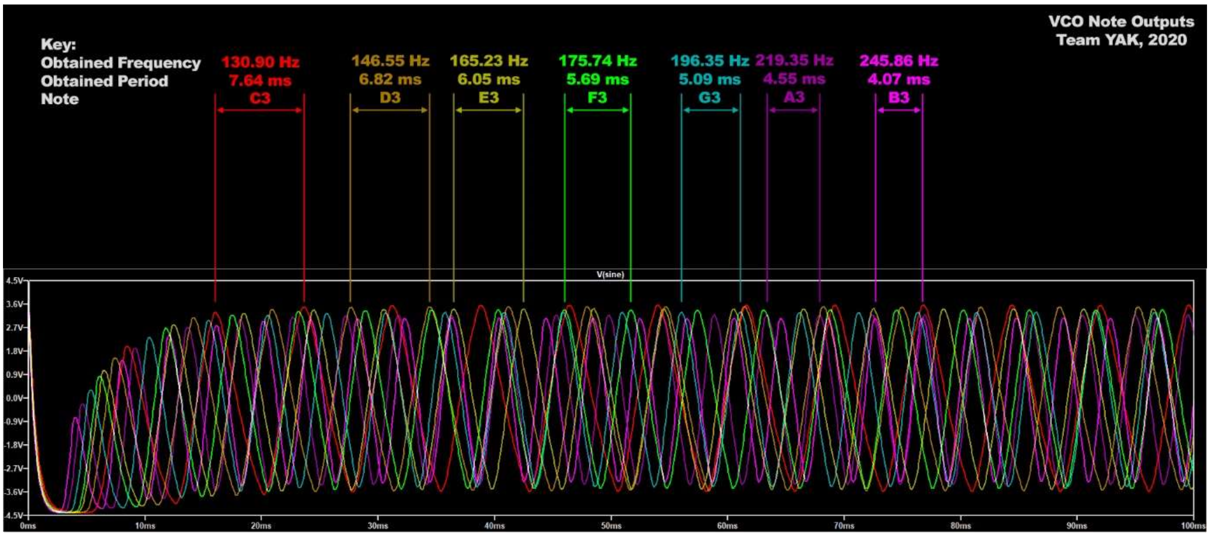

Figure 18: Voltage Controlled Oscillator Note Output Frequencies

II. ADSR Envelope Generator

The goal of having four independently variable parameters (Attack, Decay, Sustain, Release) was achieved. Each parameter was able to be changed using individual voltage divider circuits. The graph below, shows the output curve of the ADSR Envelope Generator:

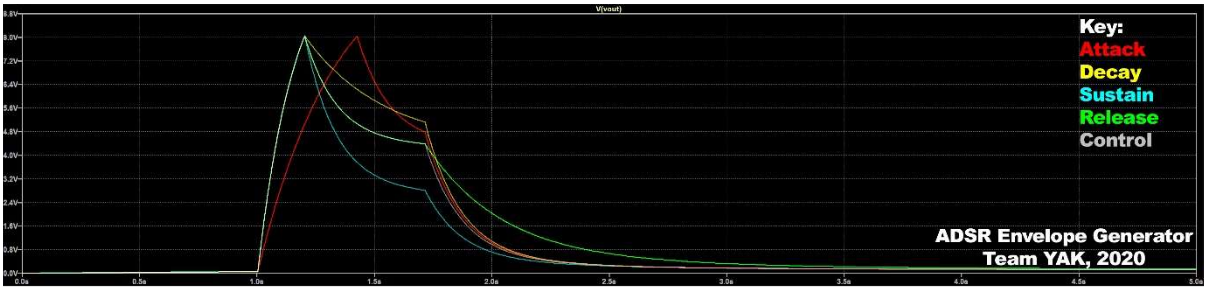

Figure 19: ADSR Envelope Generator Output Curve

In each of the plots above, one parameter was changed, while the other three were kept constant. The grey curve is the control plot.

On the attack curve (red), the attack time was increased from approximately 0.2s to 0.4s. This was done by increasing the effective resistance of the voltage divider, practically equivalent to increasing the variable resistance of the potentiometer towards its maximum value of 2.2MΩ. To decrease the attack time, the user would have to reduce the variable resistance of the potentiometer.

In the decay curve (yellow), decay time was increased once again by increasing the effective resistance of the voltage divider. This meant that for the duration of the pulse the output could not achieve the control sustain level of 4.5V and instead could only reach approximately 5V.

The sustain curve (blue) shows a decrease in the sustain level, from 4.5V to 3V achieved by decreasing the effective resistance of the sustain voltage divider. This leads to a lower output level for the note after the initial rise and subsequent fall.

The final plot is the release curve (green) which demonstrates a longer release time achieved by increasing the effective resistance of the release voltage divider. This results in the output level taking a longer time to reach 0.

The method to vary all of the parameters was simply to increase or decrease the effective resistance of the voltage divider, which in practice would be as simple as twisting the potentiometer trim knob in the desired direction. This would be very user friendly and is traditionally how conventional envelope generators are controlled.

III. Voltage Controlled Filter

The VCF as implemented had two adjustable parameters: Cut-off Frequency, and Resonance level. The figure below shows a bode plot for the VCF circuit. It shows how the frequency response changes as the value of CV (control voltage) is changed.

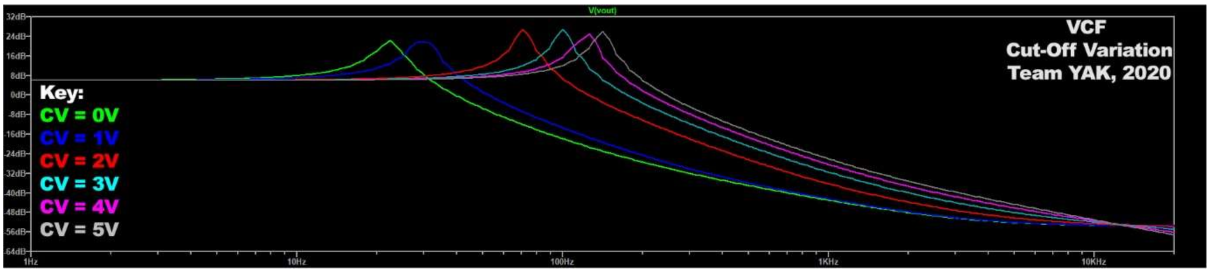

Figure 20: VCF Cut-Off Frequency Variation

The second bode plot shown in the figure below demonstrates the variation in resonance due to change in the value of the potentiometer.

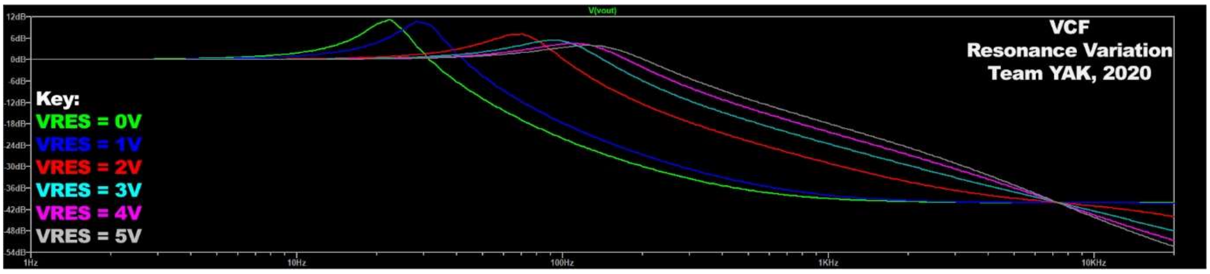

Figure 21: VCF Resonance Variation

The settings show that when resonance is high, a well-defined peak in gain across all ranges of frequency can be observed. This increased the gain from a standard 0dB before the cut-off point up to about 24dB which would cause significant effects to the volume of the output about this frequency.

As the resonance potentiometer value is decreased the peak for later cut-off frequencies is ‘flattened’ but maintain their relative sharpness for the smaller cut-off frequencies however they are still smaller absolutely.

By changing the capacitors or resistors in the circuit the plots can be translated further right along with each change in CV however since the frequencies that are being produced in the VCO fall into the 130-260Hz range a fuller use of the filter can be explored when the cut-off frequency is also around that range.

The change in cut-off frequency and resonance does not appear to be directly proportional to the changes in CV or Potentiometer and there are limits on the varying of these inputs that elicit meaningful differences in the output waveforms. Therefore, additional measures ought to be taken to ensure that user-interaction with these variables stays within the safe limits and is intuitive to use.

IV. Low Frequency Oscillator

The LFO was able to produce distortion free square, triangular and sinusoidal waves. The results of which can be seen in the plot below.

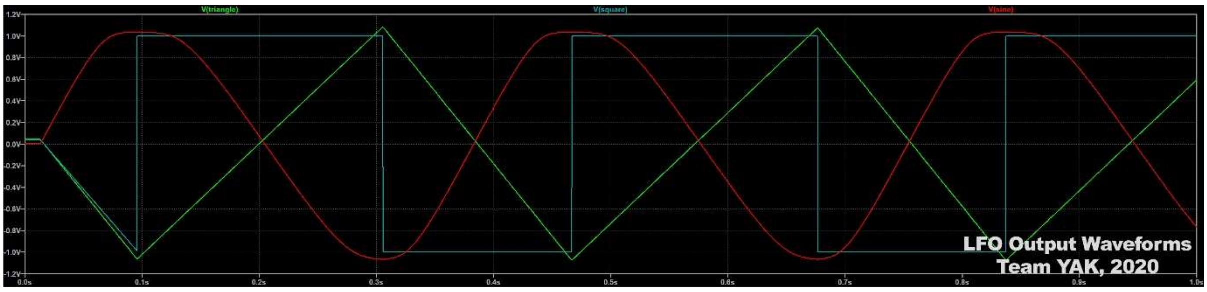

Figure 22: Low Frequency Oscillator Output Waveforms

As mentioned before, by changing the ratio of R18 to R19 or by using a potentiometer trim knob. The greater the ratio of resistance of R19 to R18, the lower the output frequency. The effect of changing this ratio can be seen in the plot below.

Figure 23: LFO Output Frequency Variation

Finally, by adjusting the output potentiometer one can adjust the output level of the LFO. The maximum value chosen was 10kΩ as this is a commonly available potentiometer. The closer to 10kΩ, the lower the output level. This can be seen in the plot below.

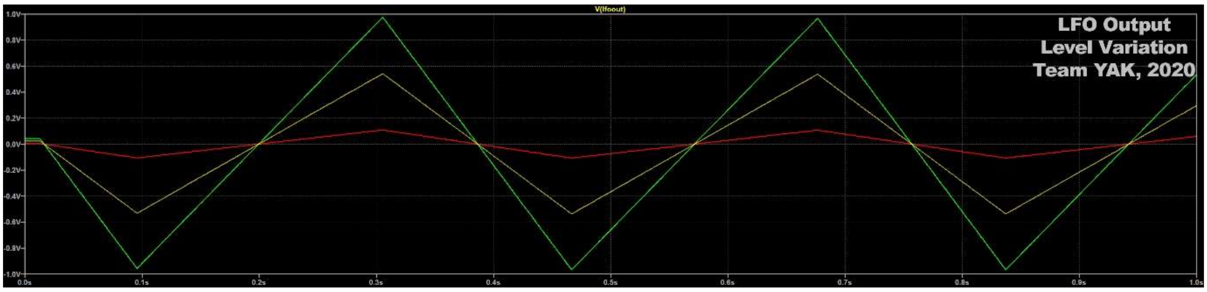

Figure 24: LFO Output Waveform Level Variation

Considering the performance, the LFO circuit was considered to be a success and was capable of working effectively with the other modules, as intended.

V. Voltage Controlled Amplifier

The figure below depicts a DC sweep for the Voltage Controlled Amplifier.

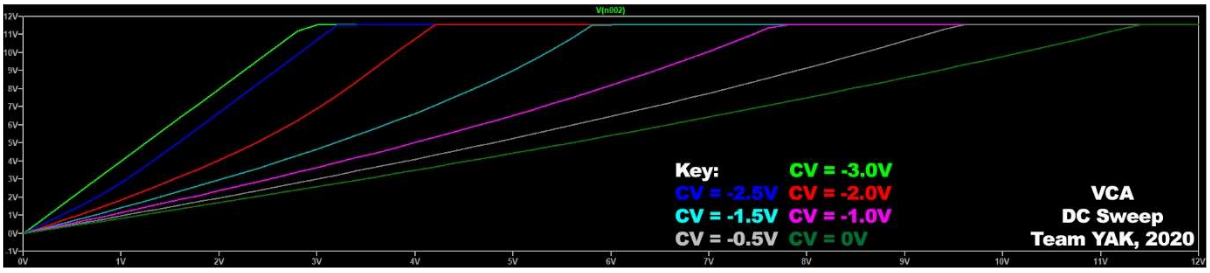

Figure 25: Voltage Controlled Amplifier DC Sweep

The shallowest line represents a control voltage of 0V and, as this becomes more negative, the line becomes steeper. This shows how the gain increases with the magnitude of the control voltage.

Voltage controlled amplifiers must have a neutral output which does not unintentionally ‘colour’ the output sound in any way. Checks were made to ensure that the gain of the amplifier was regular, and the frequency response was flat over the range of frequencies the VCA was expected to handle (20hz – 20kHz). The bode plot in the figure below shows the frequency response for the VCA circuit.

Figure 26:Voltage Controlled Amplifier Frequency Response

From the plot it is evident that there is a slight roll off beyond 10kHz, however this was not seen as too much of a concern as the frequency of the highest note on a conventional keyboard, B8, is 7902.13 Hz and the VCA can effectively handle that.

VI. Power Amplifier

Initially a Class B output stage was considered, as it would be relatively simple to implement and seemed quite effective. However, significant distortion was encountered with the increase of input frequency or amplitude. These mainly appeared as crossover distortion at high frequencies and the loss of sinusoidal shape in the output signal. This was due to the switching between the push and pull transistors.

Once the change was made to a Class AB Power Amplifier, there were significant improvements. The input and output currents of the amplifier have been shown in the figure below.

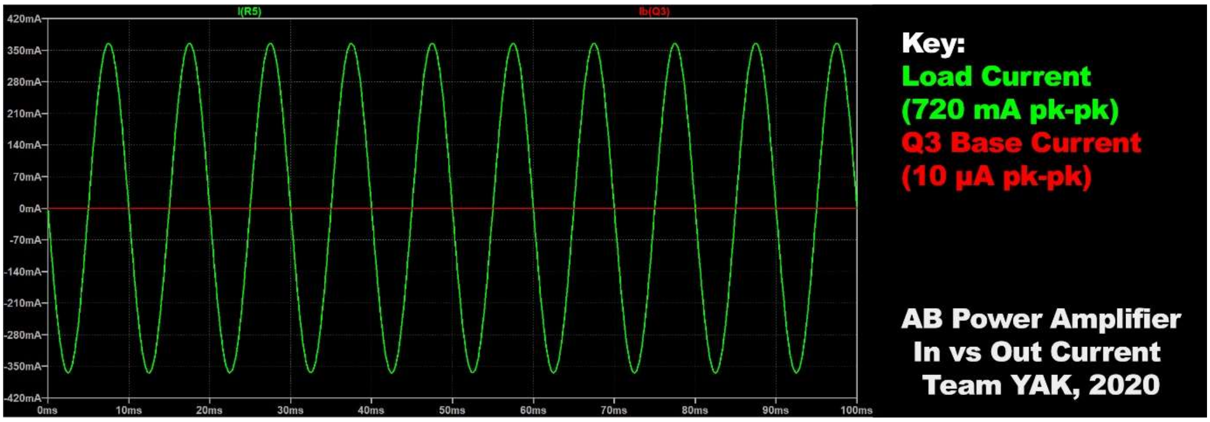

Figure 27: Power Amplifier Current Gain

The figure demonstrates that the Class AB Power Amplifier exhibits a large amount of current gain and completely eliminates the problem of crossover distortion. This is ideal behaviour for an amplifier designed for audio applications.

Since the current gain was so large, it was important to keep in mind that too much current was not supplied as this could cause the loudspeaker and transistors to be overwhelmed and malfunction. Therefore, it would be smart to have current limitations in place.

VII. Module Combinations

The figures below show plots for some common interesting combinations of modules.

i. ADSR Envelope Generator and Voltage Controlled Amplifier

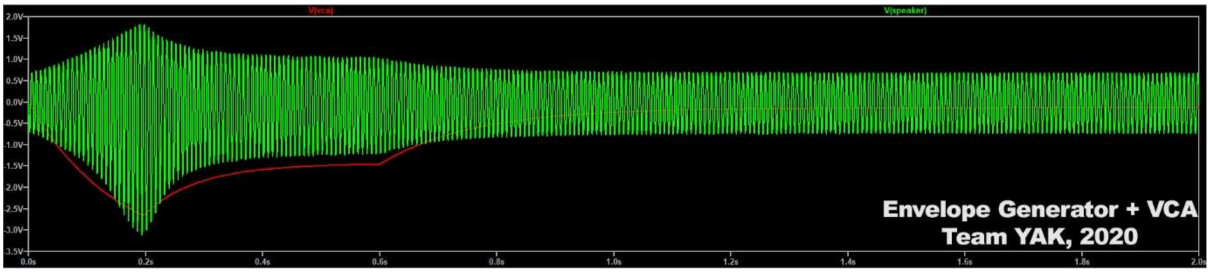

Figure 28: Output Waveform of the VCA when modified by the Envelope Generator

The output of the Envelope Generator is fed into the Control Voltage of the VCA using a summing amplifier. This causes the amplitude of the audio signal to conform with the envelope that is generated producing a distinct increase to peak volume during the Attack, decrease to a constant volume as the key is held (Decay + Sustain), and then final decrease in volume during the Release.

ii. Low Frequency Oscillator and Voltage Controlled Amplifier

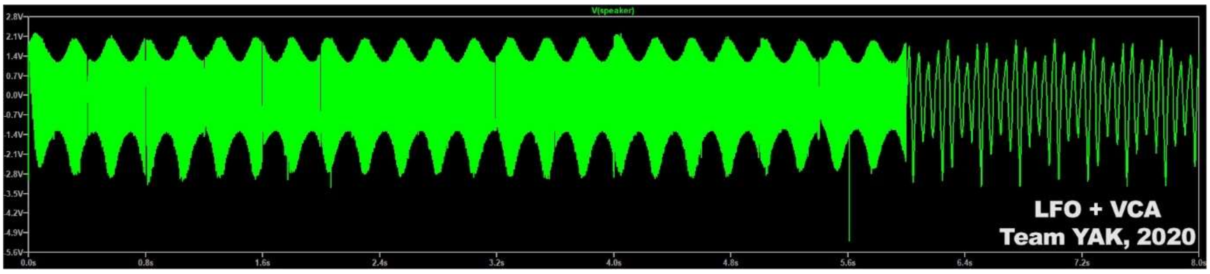

Figure 29: Output Waveform of the VCA when modified by the LFO

The LFO output is summed with the control voltage of the VCA which results in a consistently varying amplitude in the audio signal as the gain of the VCA is influenced by the LFO.

iii. Low Frequency Oscillator and Voltage Controlled Filter

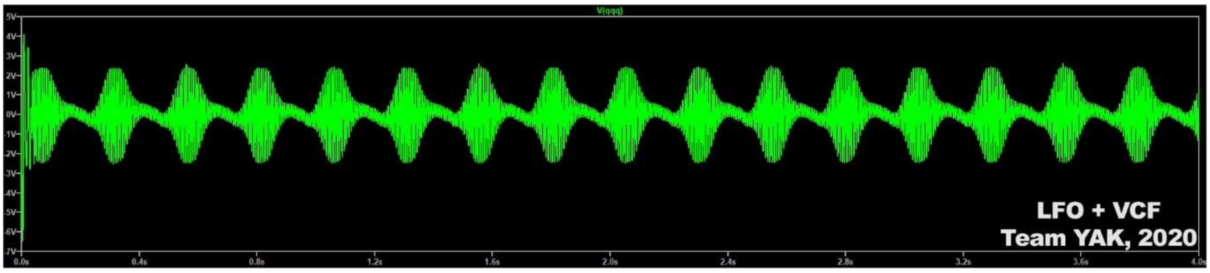

Figure 30: Output of the VCF when modified by the LFO

Again, fed into the control voltage the audio signal produced has the effect of making the sound ‘wobble’. This is caused as the change in control voltage effectively changes the gain on the signal as it moves the resonant peak across the frequency range.

VIII. Power Consumption

To calculate the power consumption of the simulation the inbuilt LTspice waveforms were inspected for both power supplies (± Vcc). Where the power was not a constant the changing power levels were treated as sine waves to provide approximations for RMS.

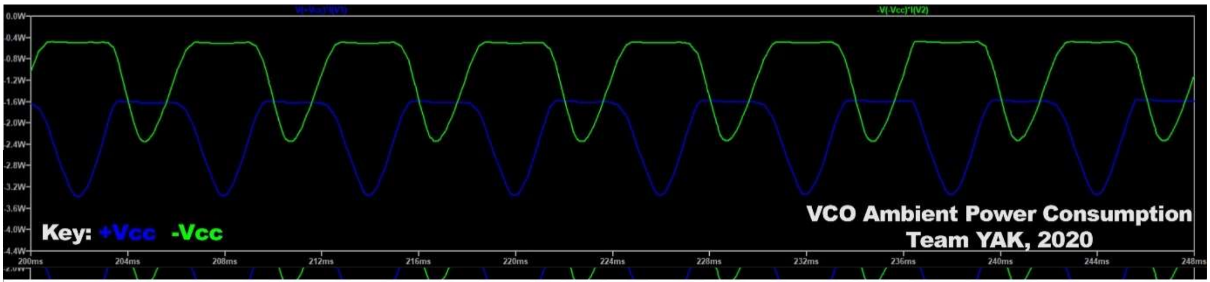

Figure 31: VCO Ambient Power Consumption

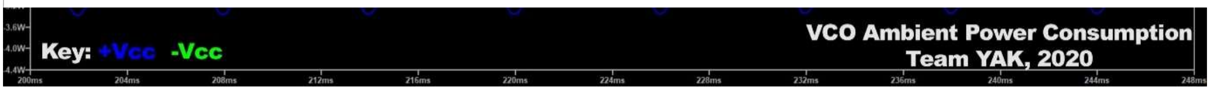

Figure 32: VCO Ambient Power Consumption

The following graphs show that the ambient power consumption of the synthesiser without utilising modules such as the VCF, LFO, EG both when a key is pressed and not pressed. The positive supply provides just above 1.6W without excess current draw while the negative supplies 0.4W in the same fashion.

While a key is pressed the positive supply peaks at about 3.4W while the negative supply peaks around 2.4W. The maximum peak power consumption is around 4W while the RMS consumption is estimated to be 3.1W.

Without the press of a key they peak power draw from the supplies remains similar however the RMS consumption is lower.

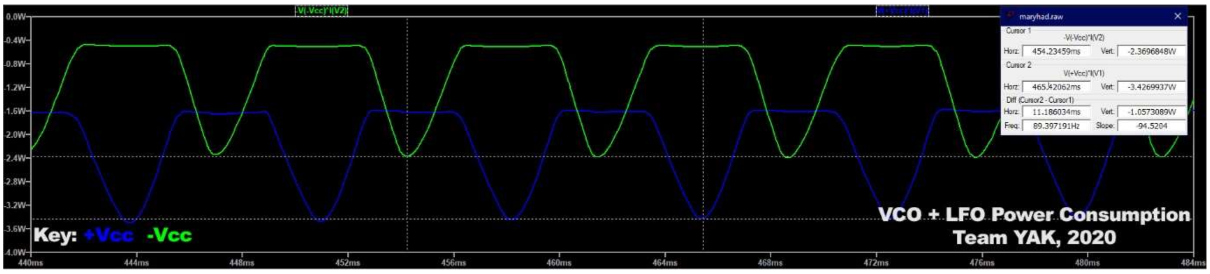

Figure 33: Power Consumption when both VCO and LFO are active

The above graphs show power consumed while the LFO is active and affecting the circuit but does not indicate a significant change in power consumption.

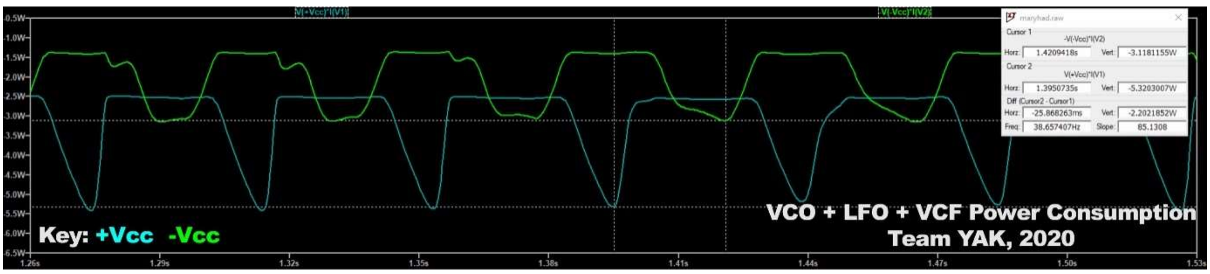

Figure 34: Power Consumption when the VCO, LFO, and VCF are active

While utilising the VCF with the LFO and VCO circuit a significant increase in power consumption is seen at all times. Ambient consumption is 1.4W and 2.5W for the negative and positive supply, respectively.

The peaks are at 3.1W and 5.4W with maximum power consumption at single point being 6.8W and RMS about 4.9W.

While looking at each circuit in their individual files the following results were found.

Module	RMS (W)	Peak (W)
VCO	1.750	2.700
VCF	0.057	0.076
LFO	0.220	0.270
EG	0.005	0.005

Table 2: Power Consumption of Modules

It suggests that the VCO consumes more power than the other sub-systems in part due to the power amplifier contained in the circuit.

While these figures are much lower than some commercial synthesisers the complexity of the circuit is also significantly lower, and the efficiency of the circuit has room for improvement.

IX. Testing Methods and Uncertainties

To measure the frequency of the waveforms the following method was used:

- The output sine waveform was taken and zoomed in so that only three peaks were visible on the graph.
- The inbuilt LTspice cursors were positioned such that they measured the voltage at the top of consecutive peaks.
- It was ensured that the cursors were on the same side of their respective peaks.
- The position of the cursors was adjusted until the voltage readings matched to the second decimal place.
- The frequency was then noted.

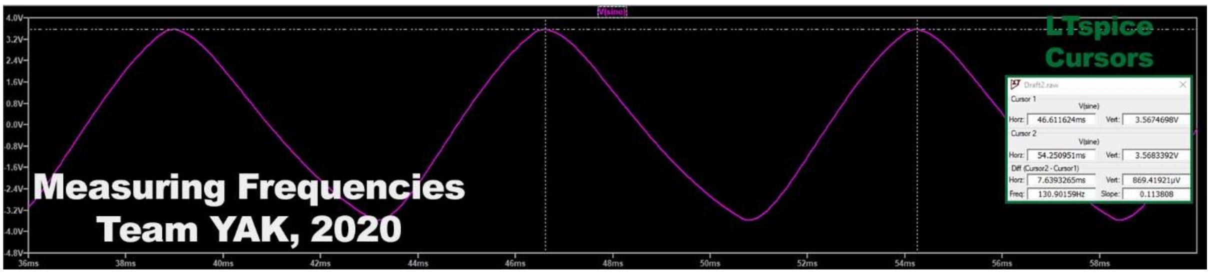

Figure 35: Measuring Frequencies in LTspice

The requirement for three peaks on screen was to try to minimise the difference in error of precision as higher frequency waveforms will have sharper peaks meaning the precision of the cursor will be lower although the time scale will also be lower which increases their precision.

There was found to be approximately a ±0.5 Hz difference in measurement if the cursor was moved one tick to either side of the peak of the signal. If this is taken as an error in precision for each note, it accounts for an error between 0.2 and 0.4%.

The table below summarises the target and obtained frequencies and errors in the VCO:

Note	Target Frequency (Hz)	Obtained Frequency (Hz)	Absolute Error	Percentage Error
C3	130.81	130.90	-0.09	0.07%
D3	146.83	146.55	0.28	0.19%
E3	164.81	165.23	-0.42	0.26%
F3	174.61	175.74	-1.13	0.64%
G3	196.00	196.35	-0.35	0.18%
A3	220.00	219.56	0.44	0.20%
B3	246.94	245.86	1.08	0.44%

Table 3: Summary of Output Frequencies and Errors

Testing during the design phase was carried out with higher margins of error in LTspice as compared to the default values. These include the ‘Gmin’, which is the conductance level added to all non-linear components to improve DC convergence, ‘Abstol’, which is the absolute tolerance for currents, and ‘Reltol’, which is the universal accuracy control. The default and chosen simulation settings have been shown in the figure below:

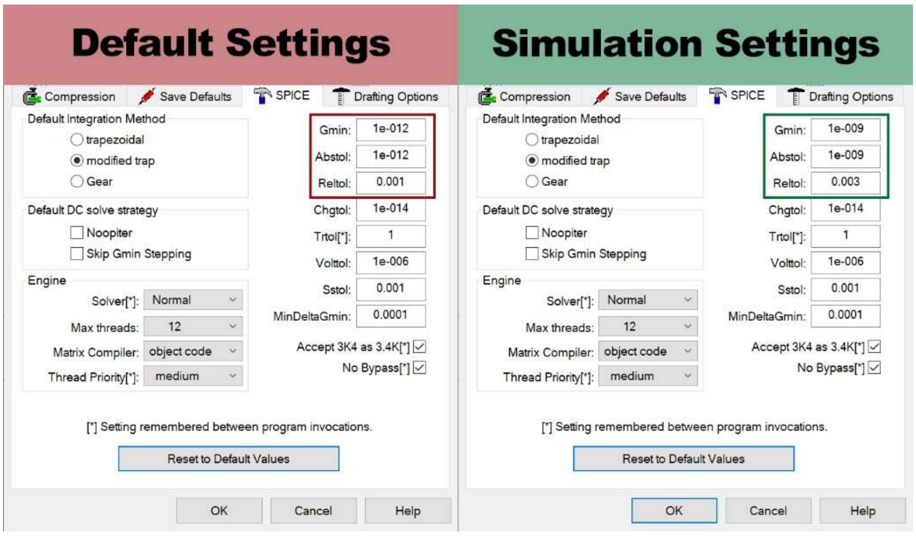

Figure 36: Default vs Optimised LTspice Settings

Increasing the margins of error enabled quicker simulations and thus, a more rapid pace of development during the design phase. This resulted in a noisier signal while testing, however for the final simulations the default tolerances were used in order to obtain as precise of a simulation as possible. This trade off was seen as worthwhile as simulation time during the design phase was brought down to a few seconds, while simulations using the default settings would often take several minutes to complete. This was especially apparent during the testing of the complete synthesizer circuit.

Additionally, the Spice directive “.options cshunt=1e-15” was used, which adds a small capacitance between each node and ground in order to speed up simulation.

7. Project Management

I. Design Process

i. Problem definition

Regular communication between the team was imperative to the design process. During the first team meeting, the team had discussions centred around the project brief. By analysing this, the team were able to develop a detailed design specification that outlines the essential requirements as well as those which are desirable and could "add value".

ii. Conceptual Design process

Once the qualitative and quantitative problems were defined, the subsequent discussions were focused around design. Many different ideas and resources were shared between members in this process. During the initial discussions, a preliminary block design was made which was later developed into the block diagram seen on page 6.

This was useful in giving the group some direction during the early stages as well has helping to give all members an idea of what features a typical synthesizer should include. Following this, a set of online resources were collated which demonstrated circuits that perform the basis for most synthesisers. Some of these modules were selected to form part of the final design.

iii. Preliminary design process

The research process involved assigning each team specific blocks to investigate with the aim of understanding how they function. Members would then produce a working simulation on LTspice of the individual component. The design process was very iterative as there several instances where multiple designs were trialled before one was found to work effectively.

Once individual modules were able to behave as expected they were adjusted so as to work with the other parts of the design enabling better performance.

II. Meeting structure

Due to the working circumstances, communication between the team was an obstacle. As meeting in person was not possible, it was decided that meetings would be held online. The primary medium for communication was an online group conferencing platform which facilitated group work by allowing one's screen to be shared. These meetings usually involved setting a focus for the day's work and then splitting this task into its constituent parts which each team member could work on individually. It was agreed that, when completed, the files worked on by an individual would be uploaded onto a shared cloud file. This allowed members to inspect one another's work which ensured quality control was implemented throughout the design process.

During the first meeting, the team researched many existing music synthesizer designs and collated many resources that could be used to aid in the design process. Sources, ideas, and diagrams were accumulated within a OneNote file in order to make resources easily accessible to other team members. This ensured that all members could develop a good understanding of the components that other teammates were working on; thus, supporting collaboration.

III. Planning and time management

Each of the larger requirements were divided into smaller, manageable tasks. After this, a critical path was created, which contained a breakdown of the time frame to complete the project. This provided a broad, but precise overview of the tasks to be completed and the time frame they must be completed in.

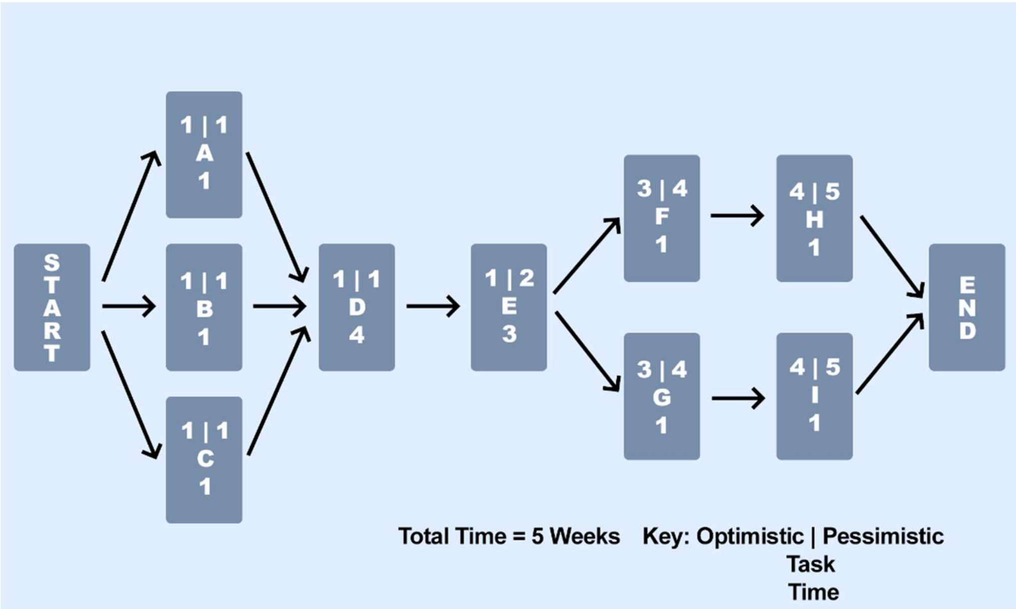

Figure 37: Critical Path

Each box on the chart represents one of the tasks and the lower number represents the number of days needed to complete the task. The upper number shows the optimistic and pessimistic start times.

Task	Duration (Weeks)	Prerequisites
Design Ideas (A)	1	-
PDS (B)	1	-
Design Finalisation (C)	1	-
VCO Circuit (D)	4	A, B, C
Additional Circuits (E)	3	A, B, C
Simulation and Testing (F)	1	D, E
Refinement (G)	1	F
Video (H)	1	G
Technical Report (I)	1	G

Table 4: Task Overview

Additionally, a Gantt Chart was maintained, to keep track of progress and to make sure the plan was being followed. The ability to update the Gantt Chart as the project progressed was found to be useful as it helped keep an objective focus on the task and helped keep the rate of progress flexible without losing sight of the end goal.

Activity	Planned Start	Planned Duration	Actual Start	Actual Duration	Completion	Week 1	2	3	4	5
Design Ideas	1	1	1	1	100%					
PDS	1	1	1	1	100%					
Design Finalisation	1	1	1	1	100%					
VCO Circuit	1	4	1	4	100%					
Additional Circuits	2	3	2	4	100%					
Simulation and Testing	3	1	4	2	100%					
Refinement	3	1	4	2	100%					
Video	4	1	5	1	100%					
Technical Report	4	1	4	2	100%					

Key: On Time, Late, Delayed

Figure 38: Gantt Chart

8. Budget

The initial project brief did not specify a budget, nor did it emphasise cost of material however these are always aspects of a design that engineers should have in mind when designing a product.

Music synthesizers can be purchased at a range of prices depending on what features are included. It was found through research that the cost of a typical commercially available synthesizer offering similar features to the ones developed began at around £100 depending on the brand. The reasonable assumption was made that the selling price was marked up by at least £50 over the cost of materials by manufacturers in order to make a profit and cover the costs of research and development, advertising, etc.

Value added features such as high-quality casings and knobs for parameter adjustment were thought to account for at least 10% of this. Therefore, it was decided that the cost of components for the synthesizer implementation must not exceed £45.

The total cost for the synthesizer was approximately £40. The majority of this was due to the high cost of the LT1366 op-amp which is sold in the form of chips containing 2 op-amps each. If the music synth were to be mass produced, a more affordable op-amp might be opted for, such as the LM324 which has a price of £0.30 per double chip, bringing the total cost down to just £10. Alternatively, through bulk purchasing and economies of scale, agreements could be reached to reduce the cost of components. This would allow for large profit margin and a sizeable return.

The itemised bill of materials has been shown in the table below.

Component	Quantity	Individual Cost (£)	Total Cost (£)
Capacitor	21	0.10	2.10
Bipolar transistor	15	0.02	0.30
Resistor	99	0.01	0.99
8 to 1 mux	1	0.40	0.40
4 to 1 mux	2	0.40	0.80
4075B Triple 3 Input Or Gate	1	0.27	0.27
LT1366 op-amp	19	3.25	32.50
Diode	6	0.08	0.48
JFET	1	0.30	0.30
Potentiometer	1	1.00	1.00
uA555	1	0.07	0.07
		Total Cost	**39.21**

Table 5: Bill of Materials

9. Moving Forward

Although the current circuit designs have been successful, there are means through which the final product could be expanded and improved. The most straightforward would be to incorporate additional expansion modules to allow users to achieve more sophisticated effects. One example is the addition of a ring modulator; a device built using diodes to give a metallic, robotic effect to the sound. Another useful module would be a sequencer which allows the user to program a combination of notes that can be looped together.

In addition to these components, there are also other filter types that could be incorporated. The existing VCF design is configured as a low pass filter however this module could be adjusted to enable high pass, band pass and notch filters.

Furthermore, there are also techniques that can be implemented to improve the signal to noise ratio throughout the synthesizer. Although this may not be as prominent is simulated circuits due to the use of 'ideal' characteristic components, if the designs were used to create a real synthesizer, it is very likely the audio signals would have noise. When noise is present in the output audio signals, the sound can become muffled and rough. The signal to noise ratio describes how prominent the noise in a signal is compared to the desired signal itself. One way to reduce this effect is to have extra amplification stages before signals are processed followed by circuitry to reduce the amplitude when they are output.

Additionally, it would be beneficial to investigate methods to reduce the ambient power consumption so that the circuit can operate more efficiently and is more affordable for the user to operate. This is beneficial as it can improve the sustainability of the product as well as being better for the environment. If the present implementation’s power consumption were lowered it could also allow room for additional features to be added in an equal power budget.

To summarise, the circuit has been very successful in meeting the design specifications that were formulated using the project brief. The final product can successfully generate the seven notes in the C major scale; each controlled by seven input voltages. The product also correctly uses only 'real' components and, due to the power amplification stage, can drive an 8Ω loudspeaker. The synthesizer also meets the optional requirements that were decided during the early stages of development.

10. References

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