Electronics Design Project 1

“Triple Engineering”

Analogue Music Synthesiser

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Introduction

Modern digital synthesiser plugins in the VST (Virtual Studio Technology) format have led to an explosion in home-based music recording. A digital audio workstation such as FL Studio can be purchased for £79.00, dramatically lowering the barrier to entry for artists and hobbyists looking to record or produce music. Despite the lowered cost and increased wave shape accuracy of digital synthesisers, many people say that they remove character from the sound and are too “clean” or “thin”. This has caused a resurgence of fully analogue synthesisers in recent years. While during the 1980s, digital synthesisers such as the Yamaha DX7 were seen as the future and certainly shaped the landscape of popular music during that decade, “fully analogue” is once again seen as a point of pride which manufacturers are not afraid to display on their products. These same manufacturers are also not afraid to raise prices accordingly, with synthesisers usually starting in the range of a few hundred to thousands of pounds.

Technical Problem to Solve

The design problem we were faced with was creating an analogue music synthesizer using LTSpice. This meant designing a circuit that generates audible waveforms following the standard musical scale: A at 440 Hz with each octave obtainable by doubling the frequency. We opted for a synthesizer with a larger, 88-key, keyboard, which meant covering the frequencies from 27.5Hz to 4.186kHz, in musical terms, A0 to C8. The synthesiser would include a choice of waveforms, as well as a resonant low-pass filter, which allows the user to adjust the generated sound to their own tastes, an important feature in a musical setting.

Our hope was to design a functional synthesiser with good sound quality and performance, all while keeping the overall budget within reasonable limits.

Design Criteria

Target Product Cost

To determine the price of our design, we created a Bill of Materials, finding the manufactured equivalent of each digital component, as well as its price. This gave us a total component price of £60.52, which of course is only a small part of the total product cost of a commercial-grade synthesiser. The prices of these products are driven up by the assembly process required to produce them, with custom PCBs needing to be printed and cases to manufacture. Additionally, synthesisers are tested and calibrated by professionals with access to extremely accurate testing equipment, which would increase labour and equipment costs in production. Finally, due to the smaller, more niche nature of the analogue music market, product prices tend to be driven up by demand, or lack thereof.

Table 1: Bill of Materials

Component	Quantity	Manufacturer Part Number	Price per unit	Manufacturer
18V Power supply AC 230V	1	T6369ST	9.93	Stontronics
1uF Capacitor	2	A105K20X7RF5UAAP	0.963	Vishay/BC Components
100k Resistor	9	MF50 100K	0.08	Multicomp pro
33k Resistor	2	MF12 33K	0.04	Multicomp pro
10k Resistor	1	MF50 10K	0.08	Multicomp pro
3k Resistor	3	MF50 3K	0.08	Multicomp pro
10nF Capacitor	2	MC0805Y103M101A2.54 MM	0.12	Mulitcomp
100k Stereo Potentiometer	1	PDB182-K220K-104B	1.59	Bourns
100k Mono Potentiometer	1	P160KNP-0EC15B100K	1.64	TT Electronics/ BI Technologies
TL072CP OPAMP, 8-pin	1	TL072CP	0.8	Texas Instruments
50k Resistor	1	CMF7050K000FKEB	2.77	Vishay
22k Resistor	1	MF50 22K	0.08	Multicomp pro
56k Resistor	2	MF50 56K	0.08	Multicomp pro
0.36nF Capacitor	1	CD15FD361JO3F	3.37	Cornell Dubilier
LT1013Dual Precision Op Amp	5	LT1013DN8#PBF	3.87	Analog Devices
3Ohm Resistor	1	MCCFR0W4J030JA50	0.05	Mulitcomp Pro
2N3904 BJT	3	2N3904 PBFREE	0.31	Central Semiconductor Corp
47k Resistor	1	MF50 47K	0.08	Multicomp pro
6.8k Resistor	1	MCCFR0S2J0682A20	0.08	Multicomp pro
370k Resistor	2	RN55C3703FB14	0.49	Vishay
3.3M Ohm Resistor	1	MCCFR0S2J0335A20	0.17	Multicomp Pro
0.01uF capacitor	2	MC0805Y103M101A2.54 MM	0.12	Mulitcomp
20uF capacitor	2	MCAX450V206M13X32	1.7	Multicomp Pro
1k resistor	4	MF50 1K	0.08	Multicomp pro
60k Resistor	1	RS00560K00FE12	3.79	Vishay
3k Trimpot	2	3386P-1-502LF	1.64	Bourns
1Meg Resistor	1	MF12 1M	0.04	Multicomp pro

Logarithmic Slider Potentiometer 10k	2	PTA6043-2015DPB103	2.09	Bourns
		Total Price	60.516	

Testing

In terms of testing, using a digital simulator such as LTSpice came with advantages and disadvantages. Testing our circuit digitally was practical particularly for the simpler tests, such as AC analysis, for which the program could execute its functions almost instantly. This gave us access to more precise measurements, measured using LTSpice's crosshair tracing, and aided greatly in visualizing the spectrum for the filter block, for example. The ability to measure voltages and currents with one click gave us important information when making adjustments or designing our circuits. LTSpice was not without its shortcomings, though. A particular setback we faced with this software was the speed of its audio processing. When testing our circuits using audio ".wav" files, we found that the software would process the file painfully slowly, a problem that also occurred while synthesising waves, a simple "Happy Birthday" took nearly 20 minutes to synthesize. On a design such as this one, which is mainly focused on the quality of the wave synthesis and filtering, these problems were particularly tangible.

Ergonomics

For the user, if we were to build our digital circuit with real components, it would be a generally unpleasant experience. Since our synthesizer is made up of base components without any assembly, the user would be left working with small potentiometers on a caseless breadboard. To remedy this, the circuit would have to be routed to a net which would then be etched onto a PCB and built into a case with larger knobs controlling the potentiometers. These adjustments however require assembly costs or means of production which we, as students, do not easily have access to.

Performance

One of the most crucial components in the music synthesiser is the op-amp. For this reason, we must ensure that the performance of the op-amp can accommodate all the desired functions of the music synthesiser. In the op-amp we are generating a square wave which is later fed into the integrator to achieve a triangle wave. Ideal square waves consist of instantaneous jumps between intervals of horizontal slops, as seen in **Figure 1.b**. However, in practice, this jump is not instantaneous and takes a finite amount of time.

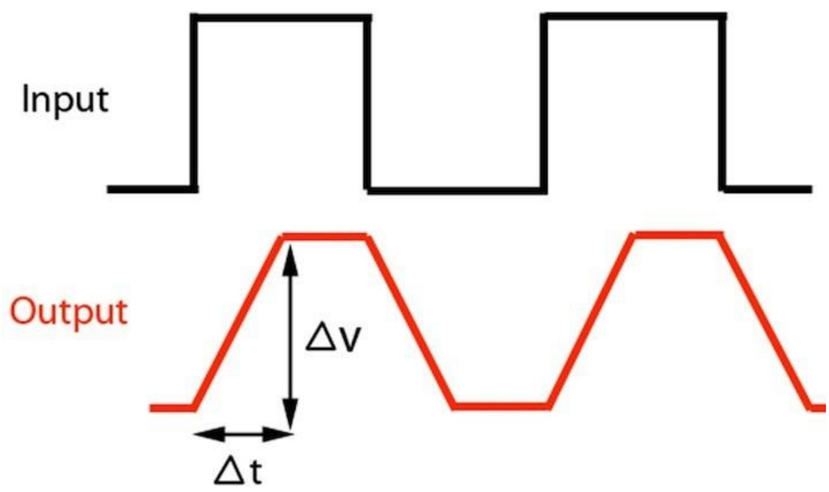


Figure 1.b: Ideal Square Wave (above) vs Non-Ideal Square Wave (below) [1].

In the real world, the fastest time for this jump is given by the slew rate, which is the maximum rate of change of an op amp's output voltage, measured in volts per microsecond [2]. The chosen op-amp, the TL072, has a slew rate of 20V/µs [2].The maximum frequency square wave we are inputting has a frequency of 4200 Hz. This corresponds with a period of approximately 0.238ms. If our square wave is oscillating between +10 and -10 V, the jump time will represent only 0.4% of the total period – a result which enables the music synthesiser to function as desired.

Competition

On the market there are many music synthesisers ranging in price from just under hundred pounds up to several thousand pounds. Since our music synthesiser is not too advanced, we will be comparing it with the MS-1-RD by Behringer. This music synthesiser lies within the price range of around 200 pounds. It has many functionalities that our synthesiser does not have, such as two voltage controlled oscillators, whereas our design only has one [3]. In addition to this, the MS-1-RD can produce saw-waves. Hence, functionality-wise, our music synthesiser would get crushed on the market. However, one must also consider price. The price of building our synthesiser is 60.52 pounds, which is considerably less than 200 pounds. Thus, our simpler synthesiser may still be desirable by individuals who would like a basic synthesiser for a relatively low price.

Linear to Exponential Converter

Justification

As specified in the PDS, the input to the synthesiser will be a voltage that represents a range of frequencies from 27.5Hz (A0) to 4.186kHz (C8). This matches the range of an 88-key piano. A logarithmic input function $f = 2^{V+4.781}$ is used to determine which voltage corresponds to which frequency. This aligns with the commonly used 1V/octave standard, first implemented by Robert Moog [4], which would later become a standard for "control voltages" used in many analogue synthesisers. This standard was also adopted in the Eurorack specification from Doepfer Musikelektronik [5] which has become the primary standard in modular synthesis.

It is important to note that humans do not perceive pitch in a linear fashion. An increase in pitch from 300-400Hz will not sound the same as an increase in pitch from 600-800Hz. In this case, the 300-400Hz increase will sound like a larger frequency change. Human pitch perception is logarithmic, and therefore the perceived pitch change depends on the ratio of the two frequencies. An octave is specified as the doubling of frequency and is one of the only things common to most musical systems. Most music from Western cultures uses 12-tone tuning in equal temperament, where each octave is divided into 12 "semitones" which are all equally spaced. Applying this to 1V/octave, it means that the keyboard output voltage will increase by 1/12V for each semitone. However, this is still a linear equation, so it follows that in order to convert these keyboard voltages to the correct oscillator pitch, linear to exponential conversion is required.

Implementation: Input Adjustment

The linear to exponential converter is based around the 2N3904 bipolar junction transistor. The Ebers-Moll model shows that there is an approximately exponential relationship between V_{be} and I_c. Therefore, V_{be} can be used to control I_c and this current can be used to control the frequency of the VCO core. [6] To double the output current (and therefore increase the oscillator pitch by one octave), V_{be} must be multiplied by the product of V_T and ln(2). More generally, $x \cdot I_c = V_T \cdot \ln(x)$ where x is the desired current multiplier. Assuming a constant temperature of 20˚C, a current increase by a multiple of 2 (and an octave increase in pitch) requires the base voltage to change by 17.53mV, commonly approximated as 18mV. [7]

The first stage of the linear to exponential converter is input adjustment, which takes inputs from several sources (keyboard, LFO and tuning potentiometer), sums them, and scales them down such that a 1V input is equal to 18mV output. The summer function is achieved using an LT1013 op-amp, in an inverting summing amplifier configuration. This configuration scales each of the 3 inputs by $-\frac{1.8k}{R_{in}}$ where R_{in} is the resistor in series with the input (R1, R2, R3 in Fig. 1). The feedback resistor TR1 has been set at 1.8k to provide the 18mV/V output, however it is implemented as a trimmer potentiometer to allow for calibration. The advantages of this setup are that it allows for precise calibration as well as easy routing of low frequency oscillator (LFO) outputs into pitch control. R2 would be implemented as a potentiometer, as most synthesisers have a main tuning knob used to set pitch as well as the more precise trimmers for calibration. Running a DC operating point simulation in LTSpice with a 1V keyboard input confirms that this circuit outputs 18mV as expected (Fig. 2)

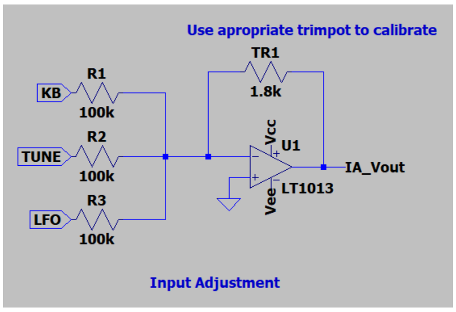


Figure 1: Input Adjustment Circuit

```
V(kb):        1            voltage
V(lfo):       0            voltage
V(tune):      0            voltage
V(ia_vout):   -0.0180216   voltage
```

Figure 2: LTSpice Simulation of Input Adjustment Circuit

Implementation: Linear to Exponential Conversion

To achieve greater temperature stability, a differential pair is used, with an op-amp (U2, Fig. 3) acting as a constant current source to fix I_c. [6] The differential pair requires that the transistors used have hFE values within 10 of each other and are thermally bonded to ensure temperature changes in one transistor are reflected in the other. Due to the through hole mounting of the 2N3904, without thermal bonding air spaces would be left between the transistors. As air is an insulator this could cause temperature mismatches and decrease the accuracy of the conversion.

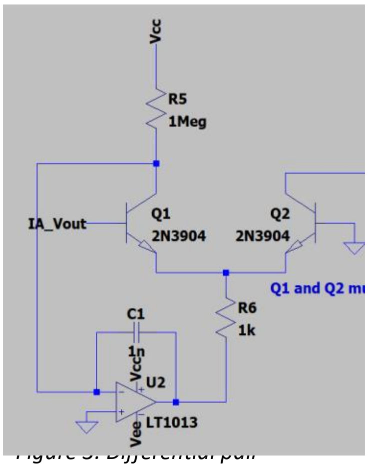


Figure 3: Differential pair

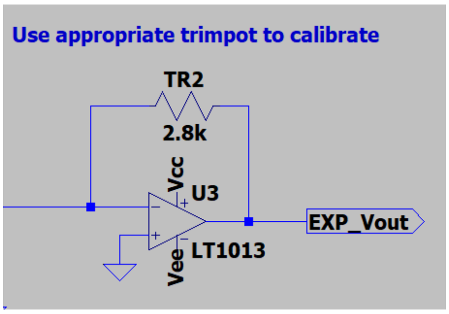


Figure 4: Inverting op amp used as current to voltage converter

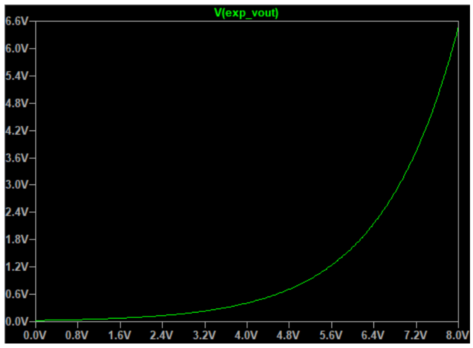


Figure 5: Graph showing exponential output voltage against keyboard input voltage

In Fig. 3 a change in the V_b of Q1 will cause a corresponding change in V_{be} which will exponentially change Q2's collector current. This is then fed into an inverting op-amp configuration (Fig. 4) which will convert the current (flowing out of V_-) into a voltage. The trimmer TR2 is used to calibrate the output voltages. To test the linear to exponential converter, a DC sweep was performed on the keyboard voltage input in LTSpice. This sweep covered all values between 0V and 8V (a range of 8 octaves) and produced exponential outputs on the EXP_Vout node. Specific values were also checked using cursors to ensure a doubling in output voltage for every 1V input voltage. The maximum voltage reached was 6.47V, but as Vcc is 9V in this specification, this will not cause any problems when it is fed into the oscillator.

Voltage Controlled Oscillator (VCO) Core

Justification

The purpose of the VCO core is to take a voltage input from the exponential converter and output a specific waveshape with a frequency corresponding to the input. As the linear to exponential conversion has been done before reaching the VCO core, the output frequency should have a linear relationship to the input voltage. The square wave output is based on a design from Robin Mitchell [8], but a number of adaptations have been made. These include adjustments to component values to enable bipolar output (±9V supply, as opposed to 5V only) and allow the oscillator to work over a range of 8 octaves. A second integrator has also been added to enable a triangle wave to be generated.

Implementation

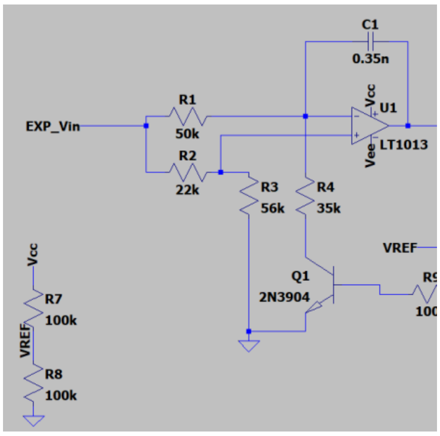


Figure 6: First stage of wave generation

The first stage of wave generation (Fig. 6) is based around an integrator circuit (see integrator section for more detail) and the capacitor charge/discharge is controlled by the BJT Q1 (as well as its series resistors). In this configuration, if Q1 is conducting, the capacitor will discharge, causing the integrator output to rise. However, if Q1 is not conducting, current will instead flow into the capacitor and cause it to charge, which causes the integrator output to fall. The rate of charge/discharge is determined by the time constant, which will be determined by resistors R1, R2, R3, R4 and the capacitance value. It is also determined by EXP_Vin, which allows the oscillator to be voltage controlled. This stage outputs a triangle-like signal (Fig. 7).

This signal is then fed into a Schmitt trigger (Fig. 8) with appropriate threshold values, which converts the triangle wave into a square wave. The Schmitt trigger will go into negative saturation (-9V) if the integrator value crosses above the upper threshold (set by the potential divider) and will go into positive saturation if the integrator value goes below the lower threshold.

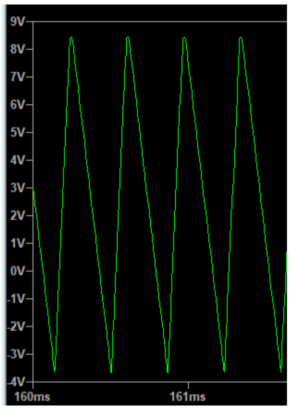


Figure 7: Output of integrator

The system, as is, will not oscillate at this stage, so one final addition is required. The Schmitt trigger output is fed into the base of transistor Q1 so that if the trigger output is high, Q1 is on and if the output is low, Q1 is off. These actions directly counteract each other, so the circuit will oscillate, and the resulting waveform will be a square wave.

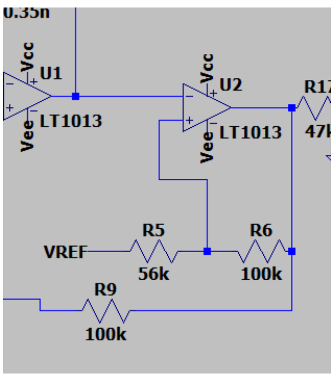
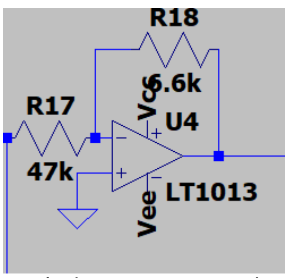


Figure 8: Schmitt Trigger

The last stage before the square wave output can be taken is adding an inverting op-amp with a gain < 1, to bring the signal down to a quieter level so that it doesn't damage any speakers were it to be plugged in.



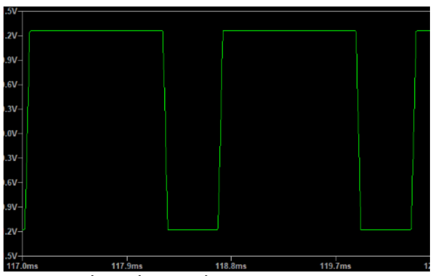


Figure 10: Final square wave output

Figure 9: Inverting op-amp for attenuation

Op-Amp Integrator

The op-amp integrator takes an input waveform and outputs the integral of that waveform.

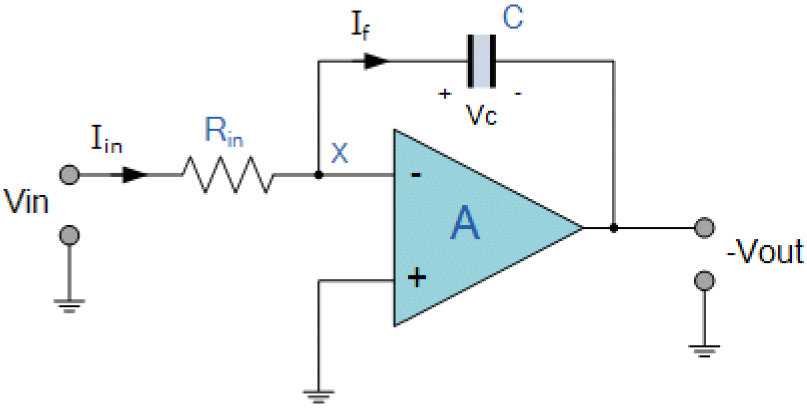


Figure 11: Op-Amp Integrator

The integrator circuit may also be assembled without an op-amp, yet this comes with certain drawbacks. For instance, take a simple RC circuit. An input signal into this circuit can be seen as a voltage or a current. If one looks at it as a current, one may also look at it as charge being injected or dejected into or out of the circuit. A capacitor stores charge, therefore, if one inputs a signal, a capacitor will essentially provide one with the cumulative charge inputted into that circuit over a given period of time. In addition to this, since charge, current and voltage are all related, one can then consider the voltage across the capacitor as the culminated voltage over a given time, i.e. the integral of the inputted signal.

Using an RC circuit does have its drawbacks though. Firstly, a capacitor does not charge linearly. For instance, if there is already a considerable amount of positive charge on one side of the capacitor, this charge will have a repulsive force on any new charge attempting to enter the capacitor. Thus, the output of the RC circuit will incur distortion [9]. Implementing an op-amp solves this issue. The negative terminal is a virtual ground and hence, the current that flows through R_{in} must always remain the same – it behaves as a current sink. Thus, the current being fed onto the capacitor is no longer dependent on the amount of charge that is already on the capacitor. In other words, the current being fed onto the capacitor is no longer affected by the voltage across the capacitor. Hence, the output waveform will be linear, yet inverted due to the inverting behaviour of the circuit [10].

Mathematical Explanation

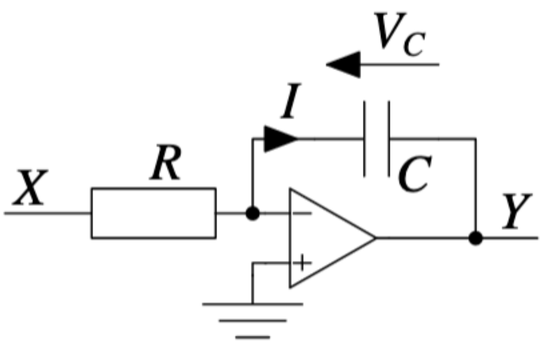


Figure 13: Op-Amp Integrator Circuit

The current flowing onto a capacitor is given by:

$$i = C\frac{dv_c}{dt}$$

Due to the fact that no current flows into the negative terminal of the op-amp, the current that flows through R must equal the current that flows through the capacitor. Hence, using V = IR:

$$i = \frac{x}{R}$$

Also,

$$v_c = -y$$

$$\Rightarrow i = -C\frac{dy}{dt}$$

Hence,

$$i = \frac{x}{R} = -C\frac{dy}{dt}$$

$$\Rightarrow \frac{dy}{dt} = \frac{-x}{RC}$$

Integrating both sides,

$$\int_0^t \frac{dy}{dt}dt = \frac{-1}{RC}\int_0^t xdt \Rightarrow y(t) = \frac{-1}{RC}\int_0^t xdt + y(0)$$

Thus, one may say that the output signal is proportional to the input signal.

Low Frequency Amplification

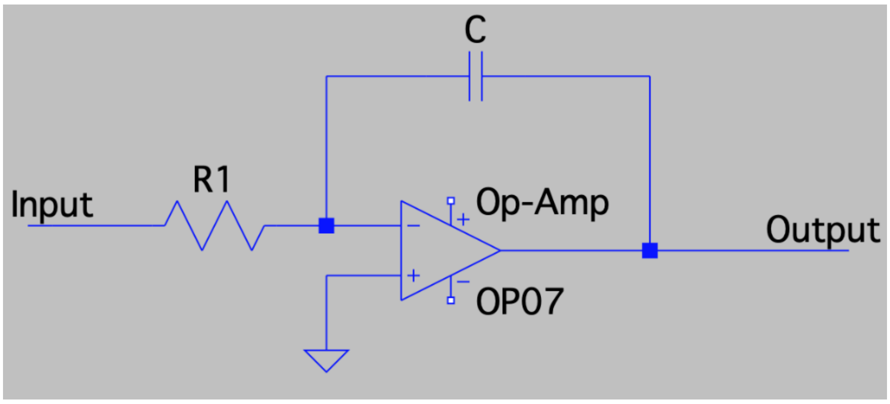


Figure 14: Op-Amp Integrator Circuit reconstructed in LTSpice.

Using nodal analysis at the node of the negative terminal, given that the reactance of the capacitor is 1/jωC, one may find that the gain of the circuit is given by:

$$A_V = -\frac{1}{j\omega RC}$$

Hence, DC currents, which have a frequency of zero, will have infinite gain. In practise, this will result in undesired saturation of any input wave with a DC component. One may argue that if our input waveform does not contain a DC component, one does not have to worry about saturation. However, in the real world there will still be small, unwanted DC offsets present in the input. Although these DC signals may be small, they will be amplified by a very high gain, which may result in distortion or even saturation of the integrated output waveform. To overcome this issue, one must make the gain experienced by DC offsets finite. The physical reason why DC offsets are subject to so much gain is because a capacitor will function as open circuit for low frequencies. Therefore, DC offsets experience the full amplification of the op-amp. This can be solved by introducing an alternate route for offsets to flow from the negative terminal to the output. A resistor, R_2, is introduced:

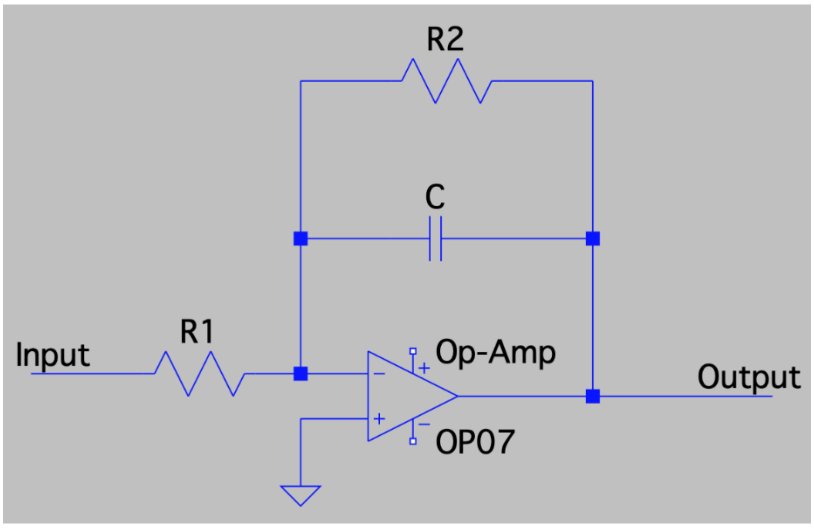


Figure 15: Op-Amp Integrator with an added Resistor, R_2.

Repeating nodal analysis at the negative-terminal node, assuming that the capacitor is open circuit, one obtains the new gain experienced by DC currents:

$$A_{V_{DC}} = -\frac{R_2}{R_1}$$

The gain of this is more finite than the one seen previously. Thus, the small input currents should induce less distortion with the circuit seen in **Figure 15**.

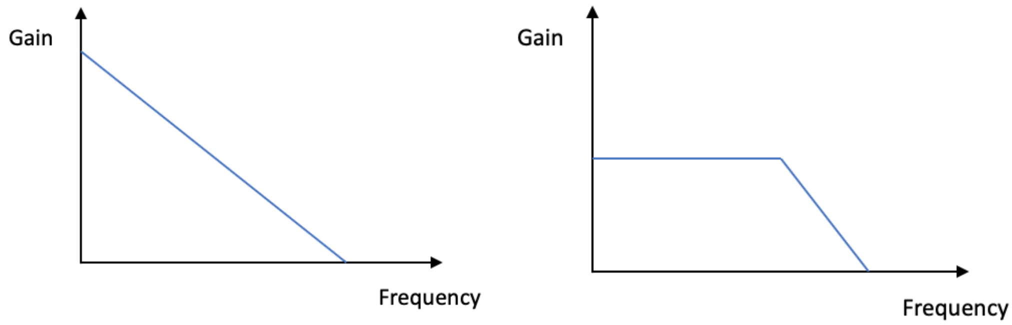


Figure 16: On the left, Gain vs Frequency for Figure 2. On the Right, Gain vs Frequency for Figure 3.

Introducing resistor R_2 into the circuit makes the circuit into a Low-Pass Filter. However, since a non-zero slope is required for the circuit to function as an integrator, one must now operate at frequencies greater than the corner frequency. In reality, one will adjust the component values such that the desired range of input frequencies is within the steep section of the frequency response.

Negating Input Bias Current

Before one can proceed with the design of an integrator, input bias currents must be negated. The ideal op-amp has an infinite resistance at both of its inputs. However, practically, the resistance will be high, yet finite. As a result, there will be undesired currents that leak into the op-amp inverting input of the op-amp.

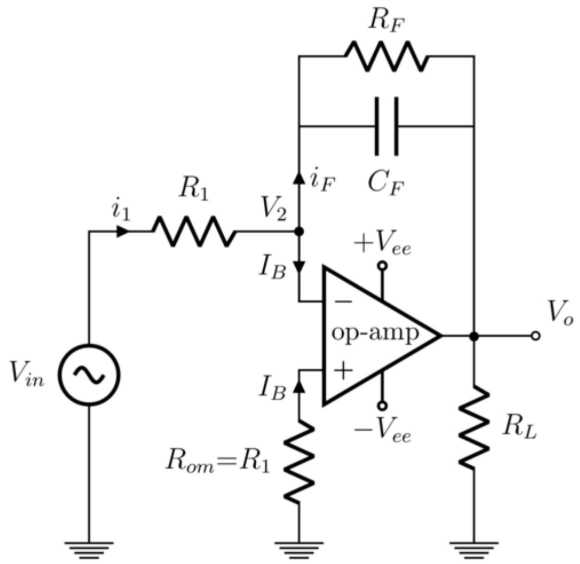


Figure 17: Practical Op-Amp Integrator.

The input bias currents are depicted as I_B in **Figure 17**. Note, the circuit in **Figure 15** would only have a bias current into the inverting terminal. The ideal inverting op-amp is assumed to have $V_+ = V_-$. However, if there are bias currents, V_- will be greater than V_+. To overcome this issue, one must ensure that the impedance at each terminal is approximately equal, such that the non-inverting terminal will have a bias current equal to the one at the inverting terminal. Hence, the assumption $V_+ = V_-$ remains true. Practically, this is achieved by introducing a resistor at the non-inverting terminal – one which is equal to $R_1||R_F||R_L$. However, since R_1 is much greater than the other resistances, one may approximate the resistor value to be R_1. Accounting for this resistor, one obtains an improved circuit design:

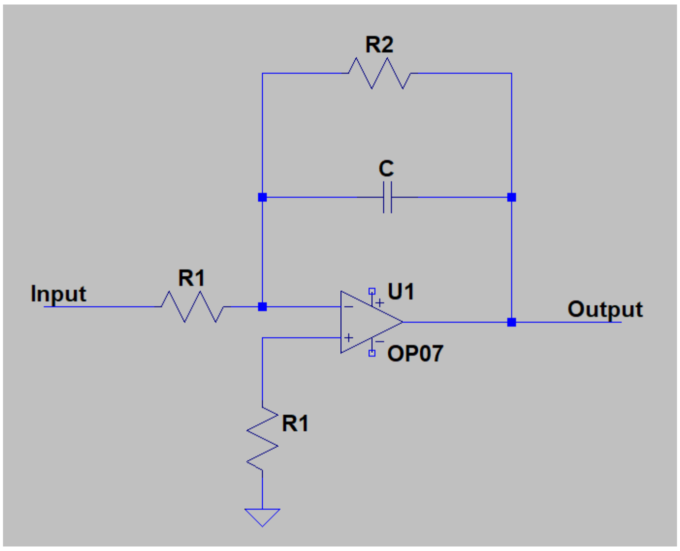


Figure 18: Integrator with Resistor to Negate Input Bias Currents.

Gain Operating Requirements

The gain of an op-amp integrator mirrors the behaviour of a low-pass filter. Looking at this in more detail, one may find the gain for low frequencies will be approximately equal to R_2/R_1 dB. This is due to the capacitor functioning as open circuit for low frequencies, such that the vast majority of current will be flowing through R_2.

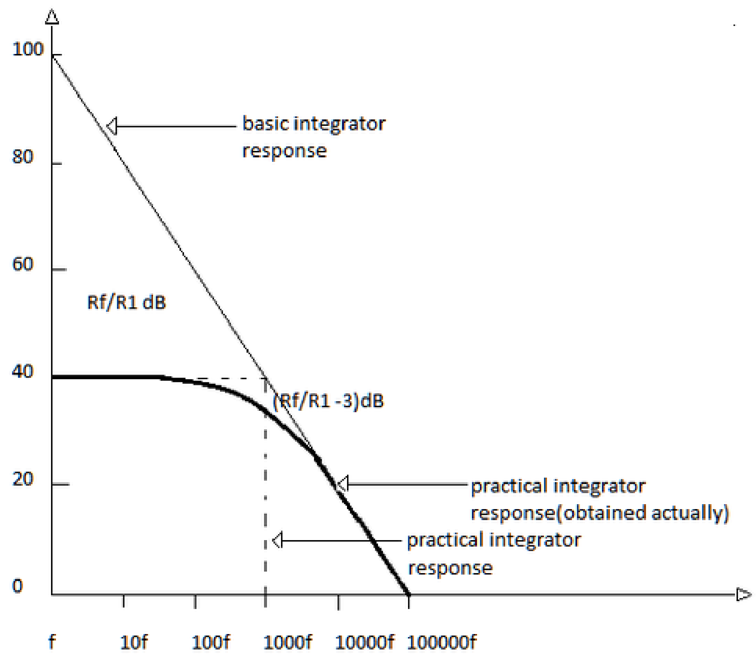


Figure 19: Gain Characteristic for Circuit in Figure 6. Units are dB vs Hz.

The frequency range for which the gain is constant is known as the pass band, i.e. to the left of the corner frequency, while the frequency range for which the gain is not constant is known as the stop band, i.e. to the right of the corner frequency. To utilise, the circuit as an integrator, one must operate within the linear region of the gain characteristic. That is, to the right of the corner frequency. Let the corner frequency, also known as 3 dB cut-off frequency, be called f_a. f_a, in Hz, will be equal to:

$$f_a = \frac{1}{2\pi R_2 C}$$

At this frequency, the gain will be equal to $(R_2/R_1 - 3)$ dB. Furthermore, the frequency at 0 dB, call it f_b, will be equal to:

$$f_b = \frac{1}{2\pi R_1 C}$$

The derivations for f_a and f_b have been omitted due to their complexity. For a full derivation, seek literature.

Although one now has a frequency range to operate the integrator in, i.e. f_a to f_b, the integration will still not be perfect over the entirety of this range. The reason for this is the gain's non-linear behaviour between 1000f and 10000f in **Figure 7**. If one inputs frequencies within this range, the integration will contain distortion. Hence, the actual frequency range one may operate in is between $10*f_a$ and f_b. When designing integrators, the range is often taken to be $f_a < 10*f_b$, where all input signals have a frequency greater than $10*f_b$.

RC Operating Requirements

The RC time constant is the time required to charge a capacitor from 0 V to approximately 63% of the applied DC voltage (Wikipedia). When inputting a signal into the integrator, the RC constant should match the period of the signal. The reason for this is to ensure that enough charge is loaded and discharged from the capacitor within the period of a signal, such that integration can occur.

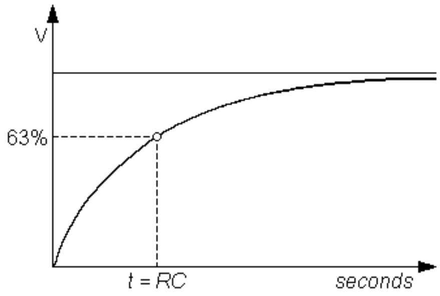


Figure 20: Voltage vs Time display RC point (source).

Op-amp integrators are a linear device. Thus, the charging of the capacitor is required to be linear as well to avoid distortion. However, in **Figure 20** one may observe that the charging of a capacitor is non-linear. To circumvent this issue, the capacitor is operated between 0 seconds and the RC region, i.e. the range over which the behaviour of the capacitor can be approximated to be linear.

The specified range of frequencies to be inputted into the integrator range from 27 to 4200 Hz. For the integrator to function for all these frequencies, an appropriate capacitor value must be chosen that can accommodate the charging and discharging time of all the prescribed frequencies. For instance, if the capacitor is too large, its RC time constant will be larger too. Hence, high frequencies signals which have short periods will not have enough time to load charge onto the capacitor, resulting in a distorted output signal. Similarly, if the capacitor is too small, this will correspond with a small RC time constant. Thus, low frequency signals with periods longer than the RC time constant will load the capacitor with too much charge, overshooting the RC point and exiting the linear region of **Figure 20**. As a result, low frequencies will output distorted signals.

Designing the Op-Amp Integrator

Any input signal, f_s, into the integrator must be at least 10 times greater than the corner frequency $f_c = f_a$. Hence,

$$f_c = \frac{1}{2\pi R_2 C} < 10 \cdot f_s$$

Assuming $f_s = f_b$,

$$f_c < 10 \cdot f_s$$

$$\Rightarrow \frac{1}{2\pi R_2 C} < \frac{1}{2\pi R_1 C}$$

$$\Rightarrow R_1 < 10 \cdot R_2$$

To guarantee this condition,

$$R_2 = 10 \cdot R_1$$

The RC constant must match at the very least match the period of the lowest specified frequency, i.e. 27 Hz. Assuming a capacitor value of .1μF,

$$R_1 C = \frac{1}{27\ Hz} \approx 37ms$$

$$\Rightarrow R_1 = \frac{37 \cdot 10^{-3}}{0.1 \cdot 10^{-6}} = 370k\Omega$$

Given $R_2 = 10*R_1$,

$$R_2 = 3.7M\Omega$$

Design Testing:

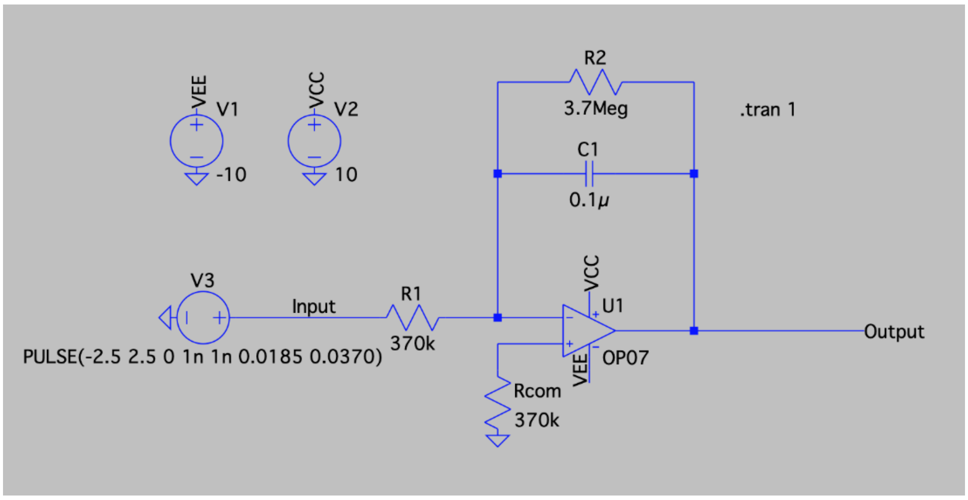


Figure 21: Op-Amp Integrator using Calculated Values.

In **Figure 21** one can see the designed op-amp integrator. The inputted signal is a square wave of amplitude 2.5 V and frequency 27 Hz.

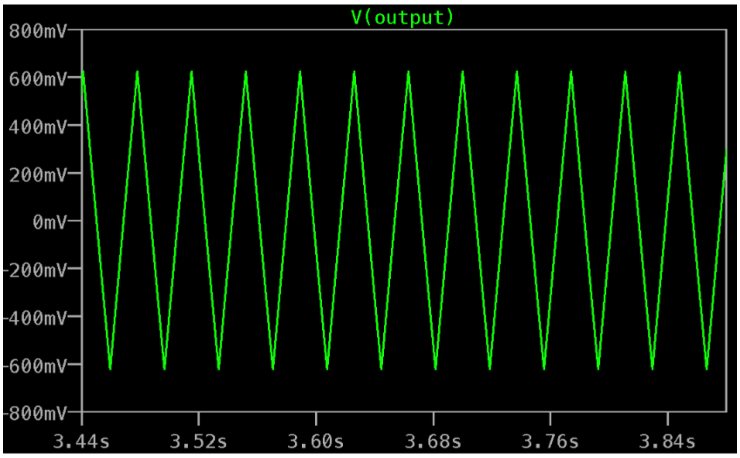


Figure 22: Voltage Output of Integrator.

Based on **Figure 22**, one can assume that the integrator works for a frequency of 27 Hz, as the output is a triangle wave.

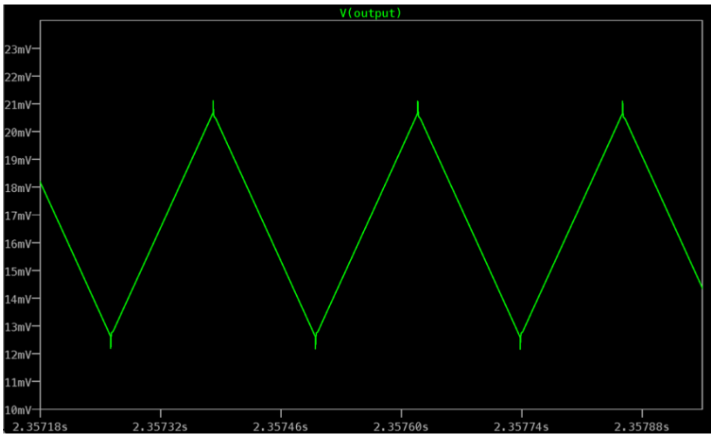


Figure 23: Voltage Output of Integrator for Input Square Wave of 4200 Hz.

Although the output wave for 4200 Hz resembles a triangle wave, there is distortion at the peaks. This may be due to the capacitor being too large. Due to this, higher frequency signals cannot load enough charge onto the capacitor to allow the integrator to function smoothly. Thus, the 0.1μF capacitor is replaced with a 0.01μF one.

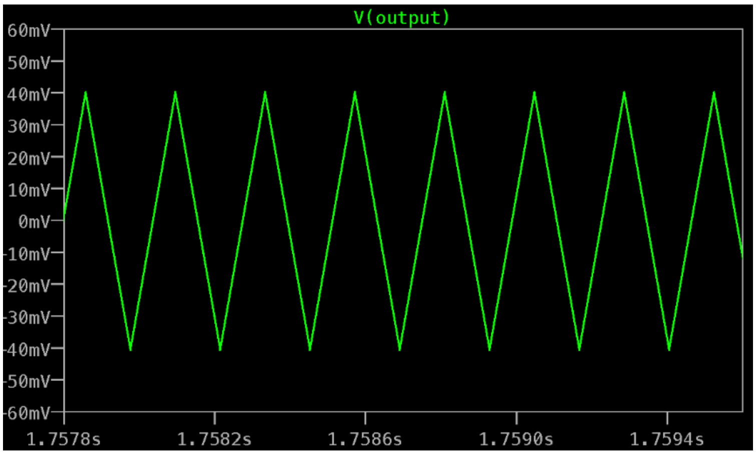


Figure 24: Integrated 4200 Hz Square Wave, using a capacitor value of 0.01μF.

Inspecting **Figure 24** one may conclude that decreasing the capacitor size removed the distortion at the peaks.

Despite the fact that the outputs in **Figure 22** and **Figure 24** are triangles waves, their frequencies must also match the frequency of their inputs, i.e. 27 and 4200 Hz, respectively. To determine whether the frequencies match, one can inspect their frequency spectrums. To achieve this in LTspice, the Fast Fourier Transform (FFT) function is used.

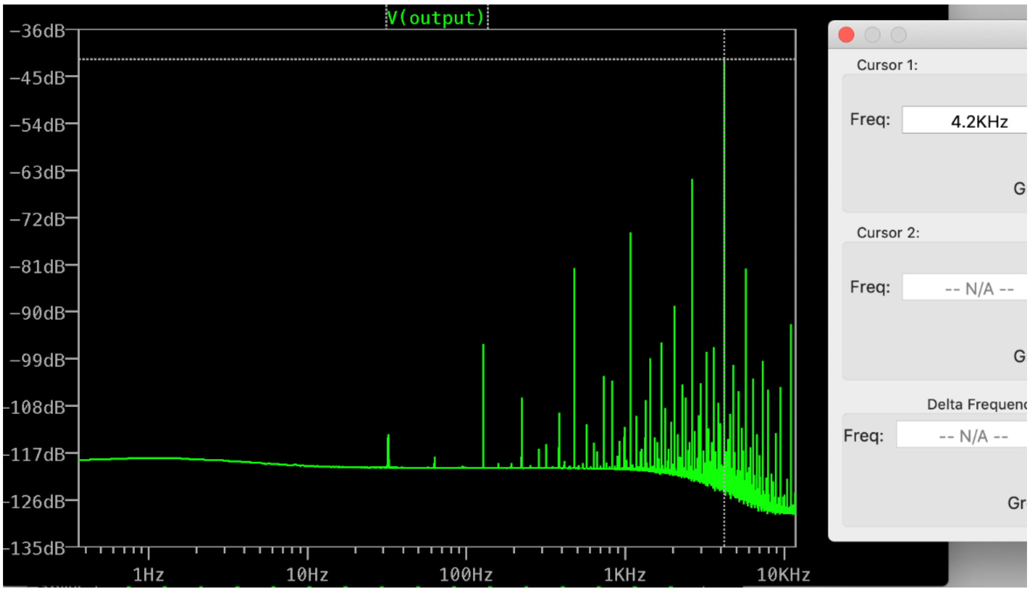


Figure 25: Frequency Spectrum of Integrated 4200 Hz Square Wave.

In **Figure 25** one can see that many frequencies are present in the FFT. However, the frequency at which the amplitude is greatest is 4200 Hz. Hence, the frequency of the output signal matches that of the input signal, as desired.

Thus far, only the minimum and maximum frequencies have been tested. To test whether the op-amp integrator worked over the entire range of 27 to 4200 Hz, a square wave sweep was fed into the input, i.e. the frequency of the square wave changed over time. The results of this can be seen in **Figure 14**.

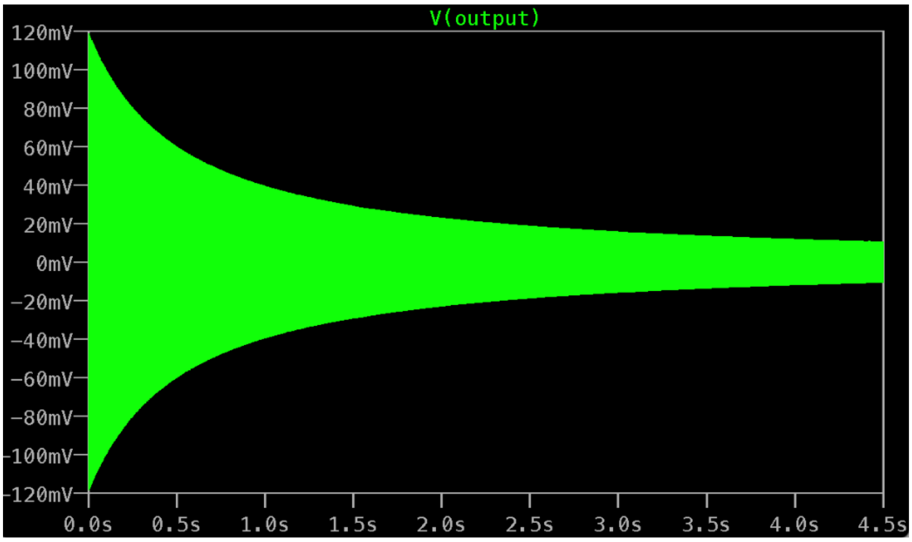


Figure 26: Output for Frequency Sweep of Square Wave.

When **Figure 26** is zoomed into, the wave corresponds to a triangle wave for the specified range of frequencies. However, one may also notice that the amplitude of the output decreases with increased frequency. This is a result of the RC constant – higher frequencies will have shorter periods, meaning they have less time to charge up the capacitor to a high value. This corresponds to a lower amplitude for higher frequencies.

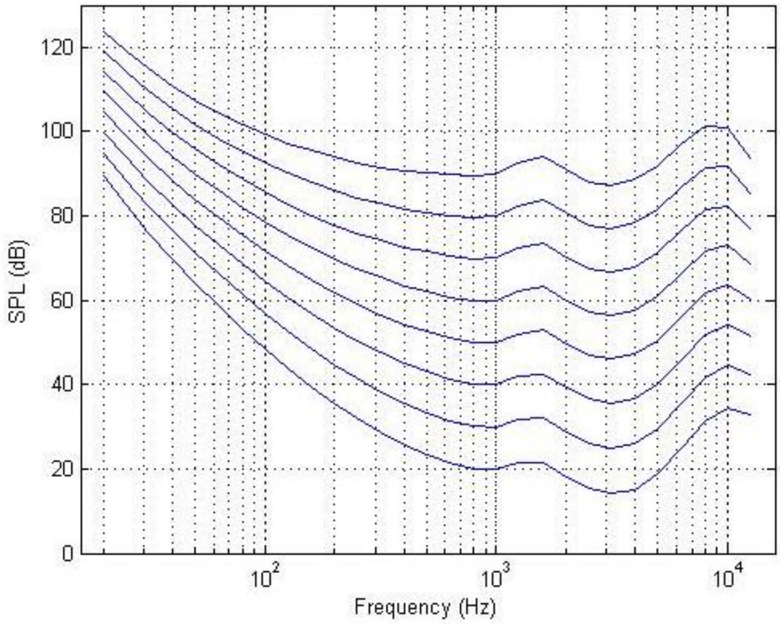


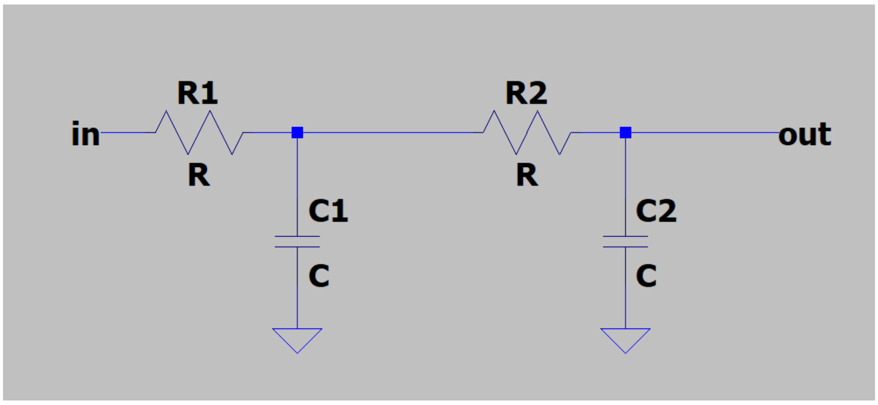
Figure 27: Equal Loudness Contours by Phon. (Source)

The fact that the amplitude of the output wave decreases as frequency increases turns out to be favourable. Looking at **Figure 27**, one can see that high frequencies at lower amplitudes will sound the same as lower frequencies with higher amplitude. This was confirmed by listening to the output wave of **Figure 26**; even at higher frequencies the audio file sounded like it had the same loudness as the low frequencies.

Low Pass Filter

After generating a signal, most synthesizers in the market allow the user to modulate the sound with one or more filters. The most common of these filters is a low-pass with variable cutoff frequency. A low-pass filter allows low frequencies to pass, while cutting off high frequencies. Including a variable cutoff point allows the user to tweak which frequencies get filtered out, thus allowing them to modify the sound to their own tastes.

Simple Low-pass Filter

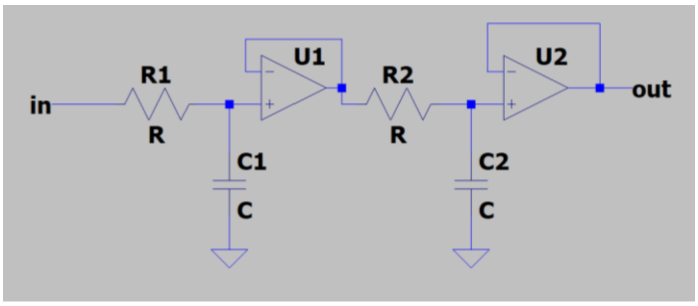


The circuit above represents a simple, two staged low-pass filter, which is made up of two low-pass filters connected in series. On its own, a single stage of a low-pass works as the interaction between a capacitor and a resistor, which causes the frequencies of the input signal to get cut off at a certain corner frequency determined by the values of C and R. Unfortunately, chaining two filters together as shown above will cause unwanted behaviour between the two filters. The current coming out of the first stage capacitor as it discharges will be drawn into the second stage instead of simply flowing back through the first stage resistor.

To remedy this, and to allow us to have multiple stages in our filter block, we must use active filters.

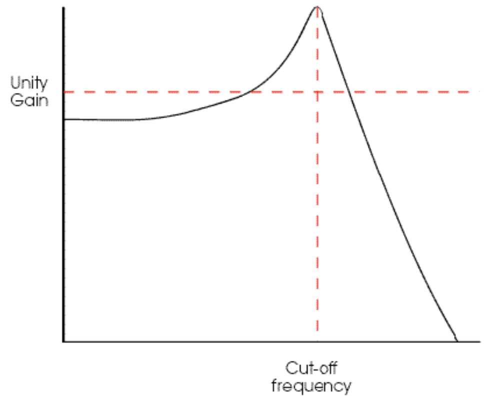
Active Low-pass

The main problem with the previous two stage filter is that the second stage cannot match the first stage's voltages without "robbing" it of its current. To avoid this, we use op-amps, included as buffers, that allow the second stage to match the output voltage of the first stage, without drawing any of its current.



The cutoff frequency of this filter is once again decided by the values of the resistors and capacitors. In this two stage low pass, the two resistors will ideally have the same value. To allow the user to easily change the cutoff frequency, the two resistors need to change value at the same time, all while keeping the same value. This is achievable using a stereo potentiometer, which allows the user to change both resistor values by turning only one knob. Since this is a second order filter, we expect a cutoff slope of approximately -12dB/octave. While this value can be improved and be made steeper by making a larger order filter, it is considered to be the standard value in most synthesizers on the market, such as the Arturia Minibrute 2 (https://www.arturia.com/products/hardware-synths/minibrute-2/overview)

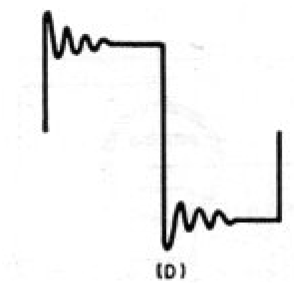
Resonance

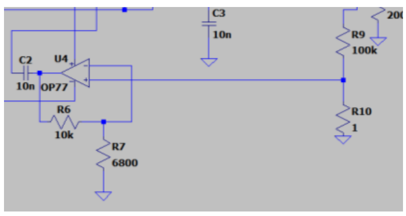


https://www.soundonsound.com/techniques/responses-resonance

When observing the output of these more basic lowpass filters, we notice that for some frequencies, the input is amplified around the cutoff frequency. This is what is called resonance.

To intentionally add resonance, we must add a capacitive connection between the input and the output stages of our filter. As this capacitor charges and discharges, it will cause small, wave-like fluctuations in the output, specifically around the cutoff frequency. Using a square wave as an example, it would cause an “overshoot” at the leading edge of the wave and an “undershoot” at the falling edge, resulting in a swung signal.





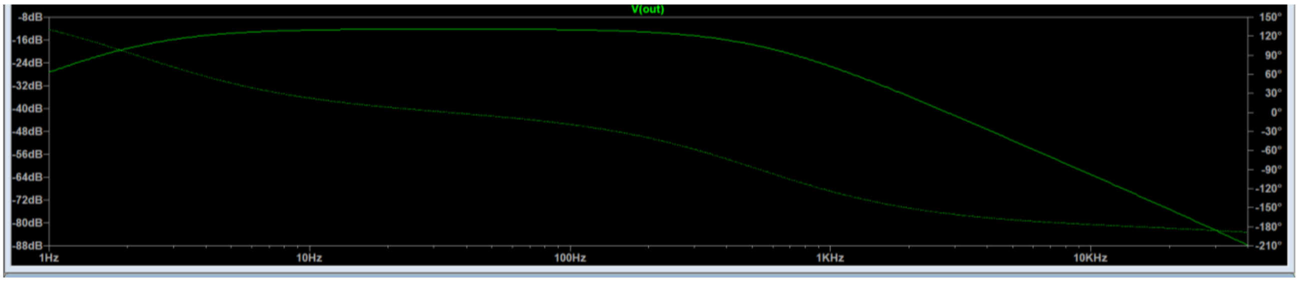
https://www.rfcafe.com/references/radio-news/practical-techniques-square-wave-testing-july-1957-radio-tv-news.htm

When applied to a sound, resonance can give interesting qualities to it that the user may wish to manually adjust. It then becomes interesting to make our low pass filter resonant, with an adjustable resonance frequency.

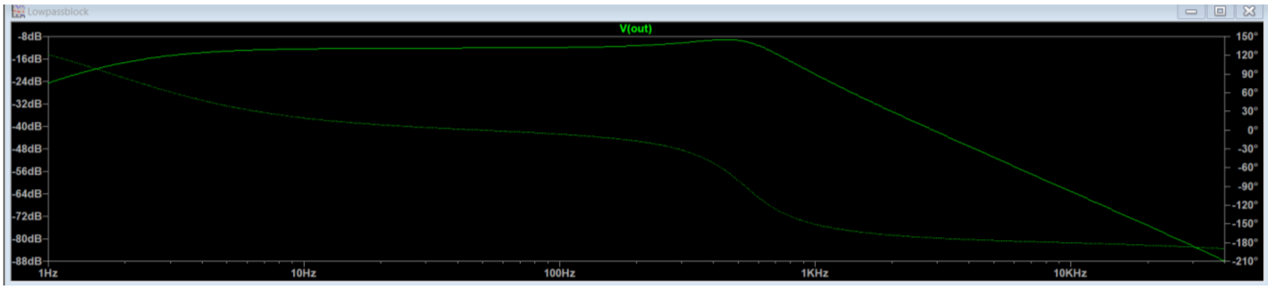
To do so, we route the filter output through a potential divider before sending that voltage through to the resonance capacitor. By adjusting the potential divider’s ratio the user can determine how much voltage contributes to the charging of the resonance capacitor, which determines the amount of resonance. To make this ratio value easily adjustable, we use a potentiometer (modeled above as two resistors) wired as a potential divider between a 100k resistor and ground. Once again, we use an op-amp as a buffer; however, to increase the effect of resonance, we add a gain loop to the previously empty negative feedback loop. The gain on the resonance cannot be modified by the user and must be selected during the synthesizer's design.

After doing this, we achieve the resonance effects needed on the signal, however, the resonance is amplified too much and higher resonance values create distortion in the output signal. An easy fix to this issue is to lower the filter's input voltage, which we achieve by adding a potential divider at the input stage.

With a low resistance on the potentiometer, we get little to no resonance as the filter's output voltage is not fed into the capacitor:



With 100k resistance on the potentiometer, the signal gets significant amount of resonance at the cutoff frequency:



An interesting effect that can be achieved with high values of resonance is self-oscillation. This happens when the feedback through the resonance is strong enough that the resonance's noise becomes its own signal, which has a sine-like waveform. This signal can persist even without an input signal. While it may not seem like a favorable feature, many synthesizers do not use anti-oscillation circuitry as self-oscillation allows circuits to create sinusoidals, which might not otherwise be possible.

Group Work Logistics

Due to the COVID-19 pandemic, group collaboration on a project such as this one was made much more difficult. Restrictions on in-person meetings steered our group work to online platforms; however, our group was able to effectively use these platforms which led to relatively smooth collaboration. We began by creating a roadmap of the different steps we would have to take towards completing our project. Using Notion, an online project planning tool, we brainstormed the tasks we'd face and arranged them in the order we believed was most efficient. Doing so, we separated each group member's workload into equal parts. We created a rough draft of the different stages and blocks of the circuit and decided to work on a block at a time, which allowed us to troubleshoot each part of the circuit individually. Using OneDrive, we were able to collaborate on our documents remotely as well as assist each other with our respective parts. With frequent in-person meetings at the library, and communication through Microsoft Teams, we were able to maintain strong teamwork and bring together each part of the project as planned.

Conclusion

To conclude, the synthesizer our team designed works generally as intended and meets our design specifications. Without the time limitation on this project, it may have been interesting to add more features that appear on commercial synthesizers to our design. The more obvious additions include a variable bandpass and highpass filter, a Low Frequency Modulation (LFO) block, and vibrato/tremolo filtering. In terms of waveforms, it may have been interesting to analyze the sounds of real instruments and create waveforms with the same harmonics to synthesize the sound of those instruments.

Additionally, mostly due to COVID regulations, we were not able to physically build our circuit, which made testing more difficult **(and far less entertaining).** Building the circuit with physical components would have allowed us to better understand the end user's experience, while giving greater appreciation for how our model would compare to commercial models, which are assembled in cases.

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[Due to a problem with OneDrive syncing, the references in different sections of the document lost their numbering. A correctly numbered list of references can be provided if required.]