Modular Analog Synthesizer

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Abstract: This project presents an analog audio synthesizer capable of generating rich, customizable sounds across three octaves of the A major scale. The system integrates a JFET-controlled voltage-controlled oscillator (VCO), waveform shaping modules for square, sine, triangle, sawtooth, and noise generation, a resonant filter core for tonal shaping, and an AHDSR envelope generator for dynamic sound control. A voltage-controlled amplifier (VCA) further refines the amplitude response, while a mixer and soft-clipping stage produce a warm, musical output. Designed with simplicity and flexibility in mind, this synthesizer demonstrates how analog circuitry can create expressive and versatile audio suitable for music production and sound design.

Project Overview:

The entire model can be broken down into 4 sub circuits namely:

1. The Oscillator Core
2. Resonant Filter Core
3. The ADHSR filter Core
4. The VCA Modules to amplify the result till ADHSR filter to human audio levels
5. The Interaction Module

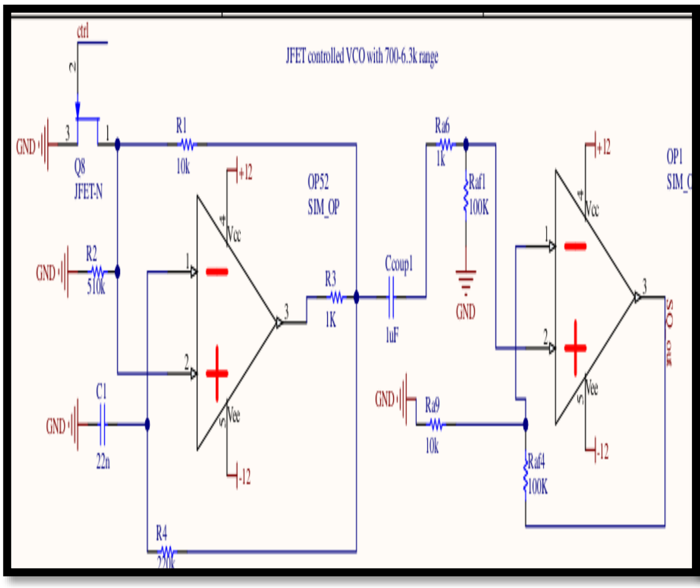

Fig 1: JFET controlled VCO with 700-6.3k range

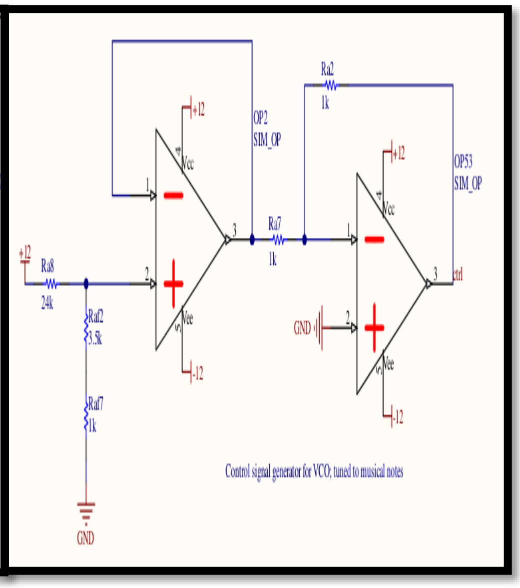

Fig 2: Control Signal generator for VCO tuned to musical notes

Oscillator Core:

Here the task of the JFET is very simple, it basically acts as a variable resistor, and modifies the gain depending on the control element. Leaving that as it is analyzing the remaining oscillator core circuit. The 510K and 10K and the JFET are connected to the non-inverting terminal of the first Op-amp as positive feedback to start the oscillations and gain control. The main timing is controlled by the C1 Capacitor along with the feedback path R4 resistor connected to the negative terminal. Like a 555 timer IC, the capacitor here works in a similar fashion, that is when the output is positive, the capacitor charges and develops a voltage across it. This modulates the non-inv input of the op-amp, the oscillations increase until the op-amp switches state. When the output is negative, (i.e after reaching a threshold, op-amp's output flips), now the capacitor 22n, starts discharging. When the capacitor's voltage crosses the switching thresholds, it flips the output back, thus the cycle repeats. This acts as a Schmitt trigger configuration to buck noise. This is the fundamental circuit for generating the square wave, works kind of like a Schmitt trigger with the timing controlled by the charging and discharging of the 22n capacitor. The output of the first op-amp passes through another capacitor 1uF to just remove any DC biassing voltage present and center the output to the origin. Upon getting the fundamental origin aligned square wave we pass the output via the 2nd op-amp just as a distortion eliminator and buffer through a voltage divider to keep the signal within the audible range. It is responsible for a clean output.

Simplifying this and addressing the key points:

1. In a JFET-controlled Voltage Controlled Oscillator (VCO), the oscillation frequency is determined by a capacitor's charging and discharging cycle, with a dynamically varying hysteresis window controlled by a JFET. This document discusses the role of the JFET in adjusting the hysteresis window, how it affects the oscillator frequency, and why this necessarily impacts the duty cycle of the output square wave.
2. In the given oscillator circuit, the op-amp operates with positive feedback, forming a Schmitt trigger configuration. The hysteresis window, defined by the upper and lower switching thresholds, is crucial in determining when the op-amp toggles between high and low outputs. The JFET is introduced into the feedback network, specifically influencing the voltage divider that sets these thresholds. The JFET acts as a variable resistor whose drain-source resistance changes with the applied ctrl voltage.
    a. As the ctrl voltage increases, the JFET's resistance decreases, strengthening the positive feedback.
    b. As the ctrl voltage decreases, the JFET's resistance increases, weakening the positive feedback.
3. The frequency of oscillation is determined by the RC time constant, particularly how fast the capacitor charges and discharges between the switching thresholds. The basic formula for frequency is:

$$f = \frac{1}{2\pi RC}$$

R = effective resistance, C= Capacitance (22n here)

As the JFET changes the hysteresis thresholds:

- A larger hysteresis window (lower JFET resistance) results in slower toggling and higher frequency.
- A smaller hysteresis window (higher JFET resistance) results in faster toggling and lower frequency.

Thus, the JFET effectively controls the oscillation frequency by dynamically modifying the reference voltages that govern capacitor charging and discharging.

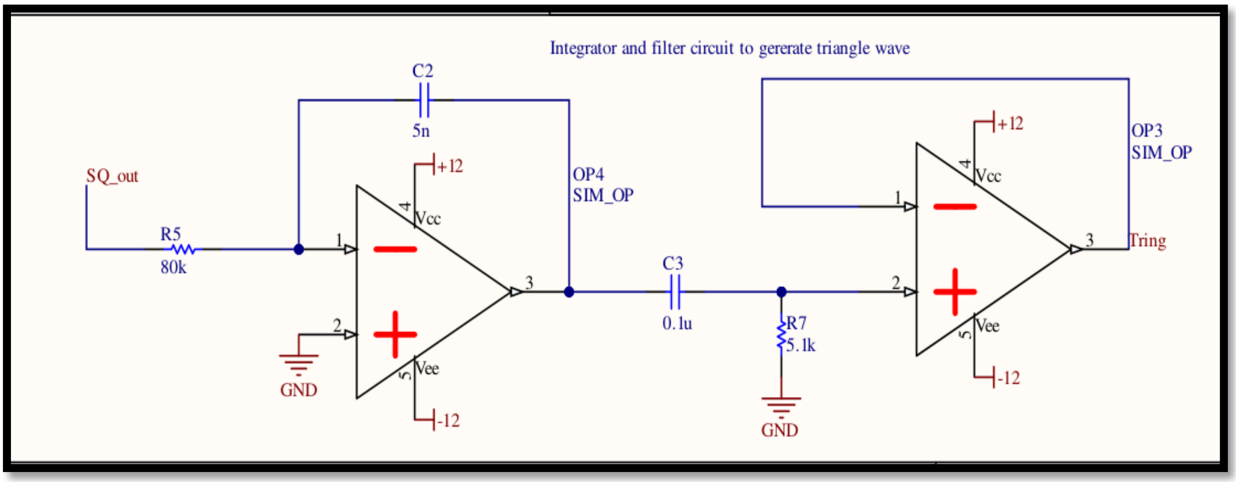

Fig 4: Integrator and filter Circuit to generate triangle wave

4. Impact on Duty Cycle
Because the charging and discharging thresholds are shifted asymmetrically by the JFET, the time it takes for the capacitor voltage to reach the upper threshold (charge) is not equal to the time it takes to reach the lower threshold (discharge).

- When the hysteresis window becomes asymmetric, the capacitor spends unequal amounts of time charging and discharging.

- Consequently, the output waveform's high and low times differ, resulting in a duty cycle that deviates from 50%.

Criteria	Hartley JFET VCO	Our JFET-Integrated Op-Amp VCO
Oscillator Topology	Resonant LC tank (Hartley); frequency set by inductors and JFET capacitance.	RC charging/discharging with op-amps; frequency controlled via time constants.
Component Requirements	Requires high-quality inductors; limited tuning range due to parasitic elements.	Uses resistors and capacitors; easy to assemble and tune.
Frequency Stability & Linearity	Sensitive to supply drift and inductor tolerances; poor linear control.	Stable, predictable frequency; supports linear control.
Audio Quality & Harmonic Content	Sawtooth-like waveform with rich harmonics; less control over shape.	Multiple shaped outputs (triangle, square, saw); high fidelity.
Complexity vs Versatility	Simple and compact; limited waveform outputs.	More complex but flexible and suited for audio synthesis.
Suitability for 1 V/Octave Tracking	Hard to stabilize for 1 V/octave tracking without trimming circuits.	Supports accurate 1 V/octave tracking through voltage scaling.
Operating frequency	Works in Radio Frequency range only, Thus, can't be used in audio frequency range.	Works in low to moderately high frequency range thus is optimal for using in audio frequency range.
Tuning the frequency range	Using Varactor diodes to change the capacitance of the LC tank circuit to change the	Simply changing the Resistances values, that is by changing the voltage division we can change the operating frequency

5. The Trade-off:

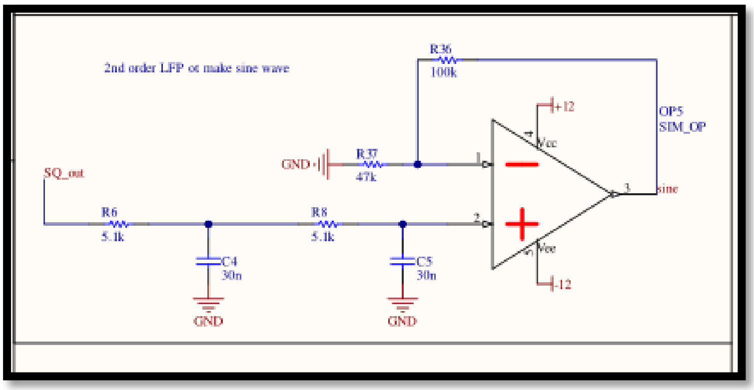

Fig 3: 2nd order LPF to make sine wave

In the JFET-controlled VCO, the JFET serves as a dynamic element within the feedback voltage divider, adjusting the hysteresis window and hence the oscillation frequency. However, because the reference voltages for toggling are altered asymmetrically, this method intrinsically affects the duty cycle of the generated square wave. Therefore, frequency control and duty cycle stability cannot be independently maintained without additional circuit complexity. And thus, unable to produce very low thumping sounds like plucked bass. But since we have not designed the circuit for transparent single processing but for tasteful audio generation, this variation of duty cycle is not much of an issue.

Control Signal Generator

The first Op-amp with R=24K connected to +12VCC, passes through a voltage divider (3.5K+1K) to scale down the +12V DC to a lower DC voltage, and gets buffered. (non inv config). The second Op-amp provides a unity negative gain thus inverts the input waveform for the VCO while keeping the magnitude same. This is done as Musical Notes in VCOs often need 1V per octave tuning, so precision voltage scaling and buffer helps us to keep it within our desired octave of A major scale. Needs adjustment and clarity to show that this voltage output is used to control the JFET's gate that is discussed below. Put this section while discussing how the JFET works in the circuit. Alternative approach to a JFET based VCO is a Hartley Type Oscillator instead of our Schmitt trigger type Op-amp based, however here's a

Our Synth will be producing 3 main music notes namely, Square Wave, Sine Wave, Sawtooth Wave (derived from the triangular wave).

Using the Square wave, it is passed through a second order active Butterworth low pass filter to generate the sine wave. Any higher order would make the cut off better but it comes with 2 trade-offs, firstly PCB size adjustment and secondly in audio electronics we don't need the finest of the waveforms to get a musically pleasing wave. The 2nd order low pass filter produces a slightly distorted sine wave instead of a clearer sine that a 3rd or 4th order would produce; but this is to our advantage as sine waves are sonically very un-interesting and thus a slight distortion benefits us by adding sonically interesting elements. First Order filter isn't used as it would have resulted in a poorer cutoff and a proper sine wave would not have been possible with commercially available values of R and C, i.e., the distortion in the wave will be too much.

Triangle Wave generation: a prelude to saw-tooth wave generation

Following the generation of a square wave, as shown in the final stage of the system, an integrator circuit is used to convert the square wave into a triangle wave. This is a key intermediate step in generating a sawtooth waveform. The integrator circuit comprises an op-amp with a capacitor (C2 = 5nF) in the feedback path and a resistor (R5 = 80k) at the input. The square wave drives the integrator, and the op-amp integrates the voltage over time:

- When the input square wave is high, the integrator output ramps up linearly.
- When the input is low, the output ramps down linearly.

This continuous integration results in a triangle wave due to the constant positive and negative levels of the square wave input. Following this, a passive RC high-pass filter (formed by C3 = 0.1μF and R7 = 5.1k) removes low frequency, near DC components. The filtered output is buffered by another op-amp (configured as a voltage follower) to produce the final triangle wave output labelled "Tring". This triangle waveform is an essential precursor to forming a sawtooth wave by controlling the rate of change in only one direction.

Saw tooth wave generation

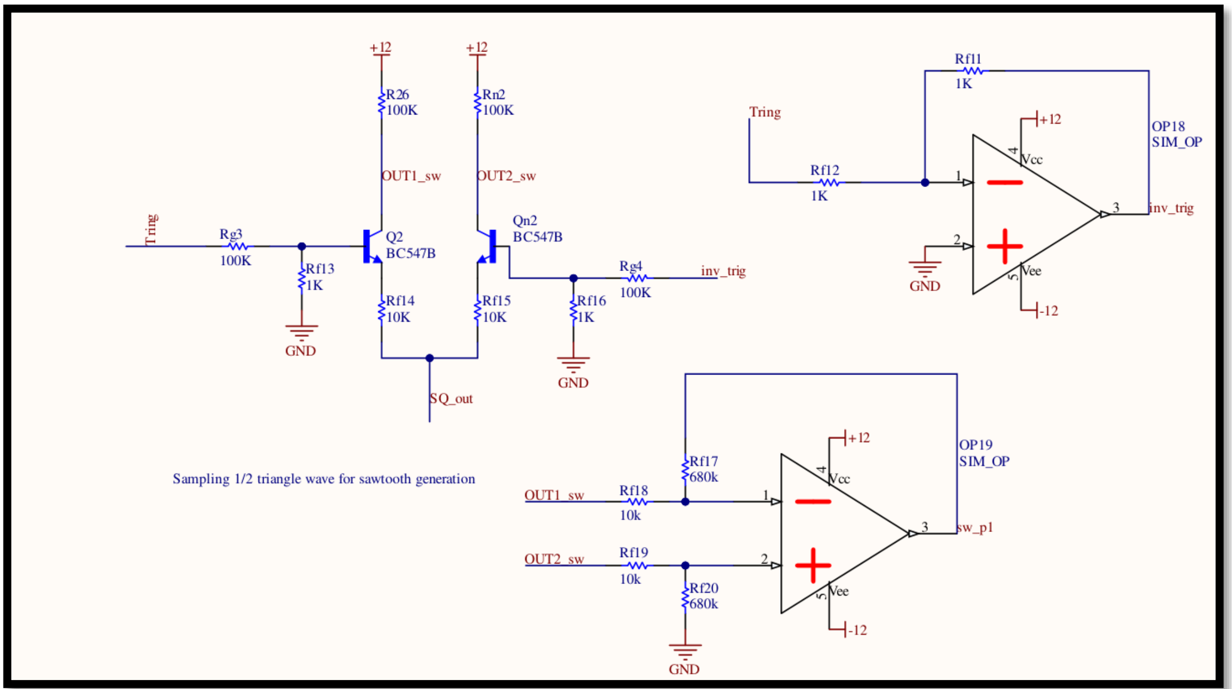

Fig 5: Sampling 1/2 triangle wave for sawtooth generation

Output Stage: OUT1 and OUT2 are differentially amplified by an op-amp (as shown as OP17 setup later)

1. Purpose:
   This stage samples the rising half of the triangle wave to generate a sawtooth waveform by slicing it accordingly.
2. VCA Configuration for Sampling:
   - This circuit mimics a transistor-based Voltage-Controlled Amplifier.
   - Q2 samples the positive-going half of the triangle (Tring), and Qn2 samples the negative-going half (inv_trig).
3. Switching Controlled by Square Wave:
   - The emitter node for both Q2 and Qn2 is biased with the raw square wave (SQ_out).
   - This square wave toggles the operating point of the transistors, selectively pushing one into conduction and the other into cutoff.
   - This alternating conduction allows only one half of the triangle signal to pass through each transistor path at a time.
4. Breakdown Behavior:
   - With SQ_out as the emitter reference, when its potential is significantly different from the base input, the transistors may operate near reverse-active or near-breakdown regions, effectively sharpening the switch and allowing cleaner half-cycle sampling.
   - This "bias modulation" allows precise slicing of the waveform and is critical to creating a sharply defined sawtooth.
5. Reconstruction of Sawtooth:
   - OUT1_sw and OUT2_sw carry the sampled halves.
   - These are then fed to a differential op-amp stage which adds the halves together, reconstructing the asymmetric sawtooth wave.

The full sawtooth output is thus synthesized from:

- The rising half sampled using SQ_out.
- The falling half sampled using SQ_out_inv.

As shown in Fig. 6, this technique ensures accurate and consistent waveform shaping using discrete analogue logic and signal gating. Now ultimately the 2 sampled saw tooth waves are fed to an inverting amplifier then passed through a 2nd order Butterworth Low-pass filter to minimize the very high frequency components and give a musical sounding Sawtooth wave.

Alt Simpler approach:

An asymmetrical integrator (Fig. 7) (with unequal charge/discharge paths) can generate a sawtooth wave by default. However, this results in a very clean saw tooth but we can't use that in audio purposes.

Key Components and Their Roles

- C1: Input coupling capacitor (blocks DC).
- R1: Controls the charge rate of capacitor C2.

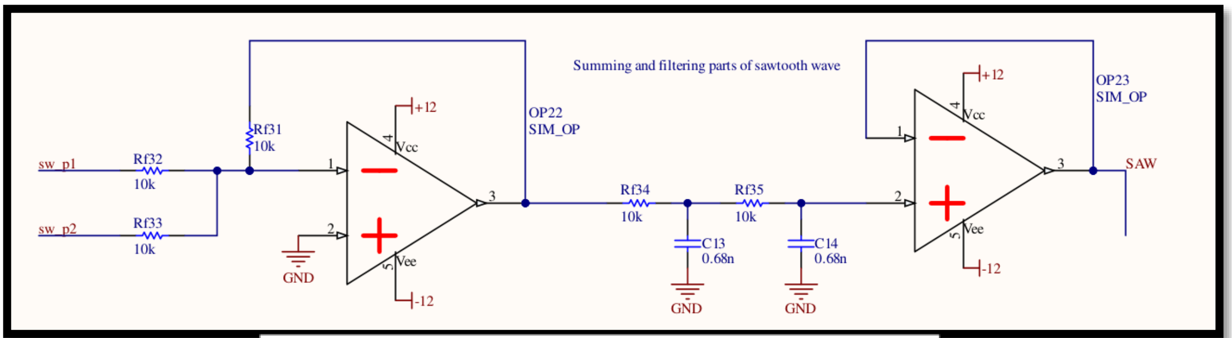

Fig 6: Summing and filtering parts of sawtooth wave

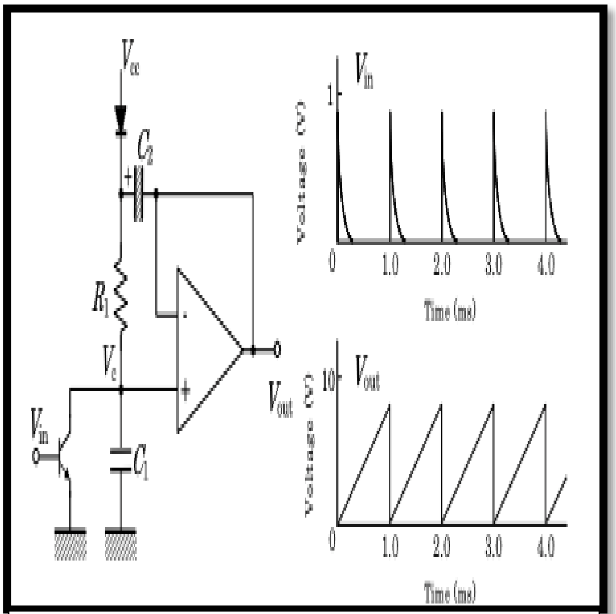

Fig 7: Alternative approach to saw-tooth

- C2: Integration capacitor — the heart of the sawtooth shape.
- Diode (D): Provides fast discharge path — this causes asymmetry.
- Op-Amp: Acts as a comparator or buffer to support waveform generation.
- V_in: Sharp pulses (typically from a square or narrow pulse wave).
- V_out: Resulting sawtooth waveform.

Working Principle

1. Input Pulse (V_in):
Sharp, narrow pulses are applied periodically. Each pulse briefly forward-biases the diode, rapidly discharging C2 through the diode path.

2. Capacitor Charging (Slow Ramp-Up):
After the pulse ends, the diode is reverse-biased, blocking the discharge. C2 slowly charges through resistor R1, creating a linear ramp voltage at the op-amp input.

3. Capacitor Discharge (Fast Drop):
When the next pulse arrives, the diode conducts again. C2 discharges rapidly, causing a sudden voltage drop — the falling edge of the sawtooth.

Waveform Behavior

- V_in: Periodic, narrow pulse train.
- V_out: Classic sawtooth waveform:
- Slow rise (linear ramp due to R1 charging C2)
- Sharp fall (due to diode discharge path)

Problems with the asymmetrical integrator:

1. Low-Pass Filtering Effect:
   - To stabilize the waveform, component values (especially R and C) are often selected to reduce high-frequency ripple.
   - This unintentionally filters out high-frequency harmonics, dulling the character of the waveform.

2. Reduced Harmonic Content:
   - A rich, bright sawtooth wave requires strong odd and even harmonics.
   - Asymmetrical integrators inherently smooth out these transients, leading to a less aggressive, muted tone.

3. Loss of Synth Edge:
   - In subtractive synthesis, sawtooth waves are preferred for their harmonic richness, essential for cutting through mixes.
   - A low-pass-affected saw compromises this edge — making the synth sound less lively.

White Noise Generation:

The circuit (Fig. 8) utilizes the inherent noise generated by a reverse-biased base-emitter junction of an NPN transistor to produce white noise. In this configuration:

- The base of the transistor is connected to ground, while the collector is left unconnected (NC).
- The emitter is connected through a resistor R to the supply voltage Vcc, creating a reverse bias across the base-emitter junction.
- This reverse bias causes avalanche breakdown or shot noise, resulting in a broad-spectrum noise signal.

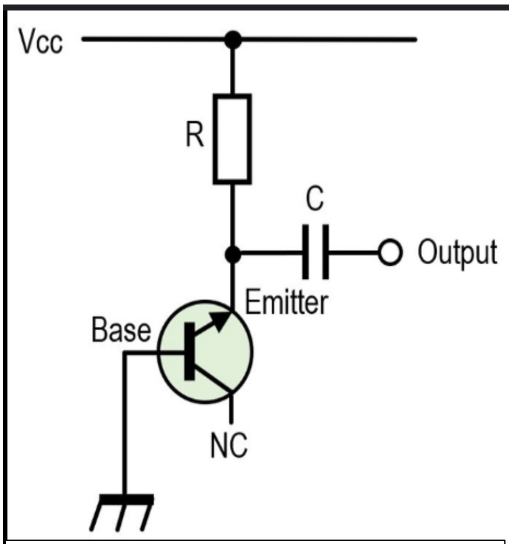

Fig 8: white noise generator

- A capacitor C is used to couple the AC noise signal from the emitter to the output, blocking any DC component.

This minimalistic design serves as a compact, low-cost white noise source useful in audio testing, random number generation, and other signal processing applications.

Resonant Filter Core

We have used the same Resonant Low/ High Pass Filter design (Fig. 9) to shape the Square, Sine and Saw Tooth waves. This circuit takes an input (SQ_out) and processes it through a resonant high-pass filter followed by a low-pass stage, resulting in a band-pass shaped output (sq_f). This configuration is commonly used in synthesizer circuits to introduce resonance and shape harmonic content.

1. High-Pass Filter Input Stage (R11, C6, OP7)
   - R11 (2.2kΩ) and C6 (10nF) form a first-order high-pass filter.
   - The cutoff frequency is given by: fc= 1/ (2pi R11 * C6)
   - This filtered signal is fed into the non-inverting input of OP7 (configured as a non-inverting amplifier with feedback).
   - OP7 amplifies the high-passed signal, and through its feedback, participates in resonance shaping.
2. Resonance Feedback Path (OP6)
   - OP6 receives the output of OP7 and feeds it back through R9 (10k) and R10 (6.8k) into the non-inverting input of OP7.
   - This introduces positive feedback, increasing the Q factor or resonance of the high-pass filter.
   - The feedback strength (resonance peak sharpness) is determined by the ratio of R9 and R10.
   - You can tune the resonance frequency by adjusting this ratio.
3. Low-Pass Filter Stage (R12, C7, OP8)
   - The signal from OP7 goes through R12 (2.2kΩ) and C7 (10nF) to ground.
   - This acts as a low-pass filter, forming a second pole: fc= 1/(2pi R12 * C7)
   - The combination of the HPF (R11-C6) and LPF (R12-C7) forms a band-pass filter centred around the cutoff/resonant frequency.
   - OP8 buffers the filtered signal, giving the output

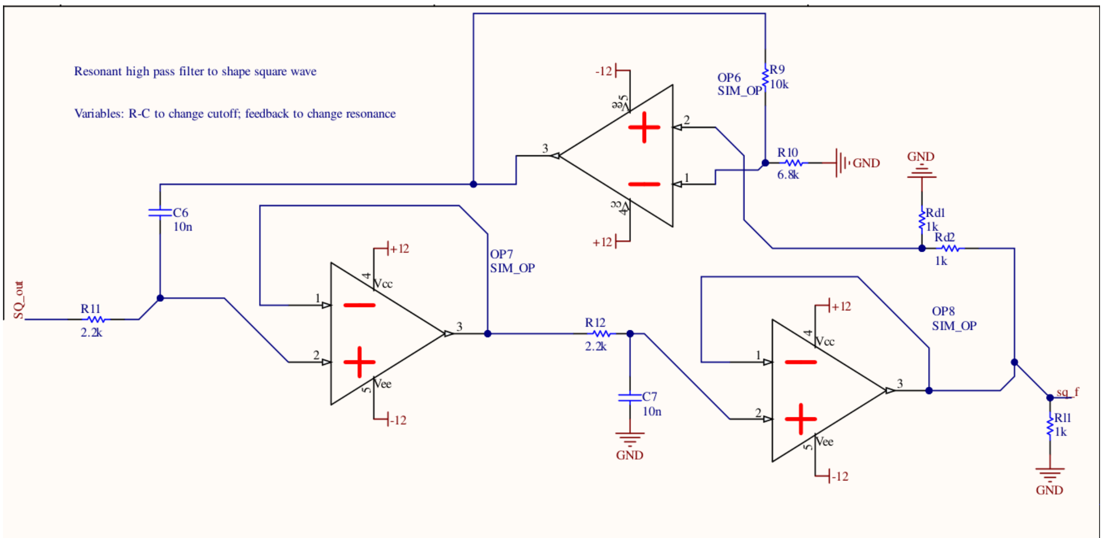

Fig 9: Resonant high pass filter to shape square wave

4. Output and Attenuation Network

- The final output sq_f is taken across Rd1 parallel Rd2, a voltage divider (1kΩ each), slightly attenuating the output signal for downstream compatibility.
- A final resistor R11 (1kΩ) provides impedance matching and protection.

ADHSR Envelope generator

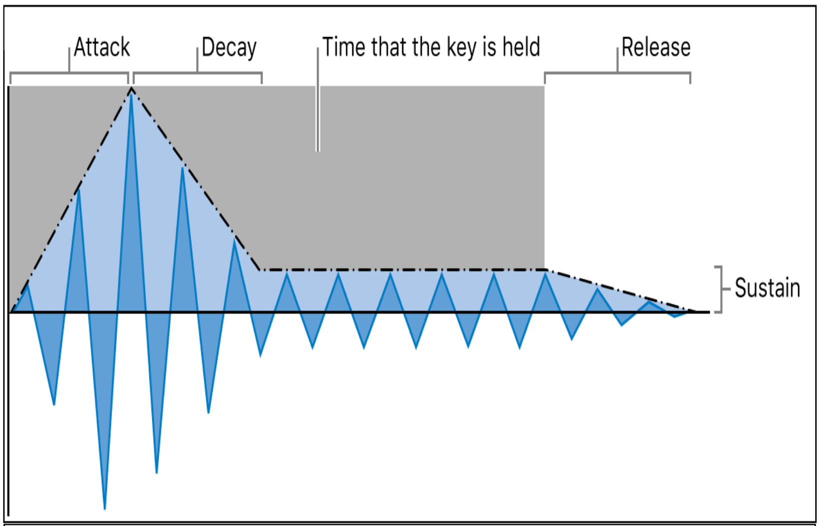

Fig 10: AHDSR envelope

Envelope controls

- Attack: Sets the time it takes for the signal to rise from an amplitude of 0 to 100% (full amplitude).
- Decay: Sets the time it takes for the signal to fall from 100% amplitude to the designated sustain level.
- Sustain: Sets the steady amplitude level produced when a key is held down.
- Release: Sets the time it takes for the sound to decay from the sustain level to an amplitude of 0 when the key is released.

1. Purpose:
   This circuit generates an ADSR envelope (Fig. 10) to shape a square wave input into a dynamic control signal, typically used in synthesizers.

2. Input Control - Simulated Button (BTN):

   - The BTN simulates a key press. (a square function)
   - When the button is pressed, a positive voltage is applied to begin the attack phase.

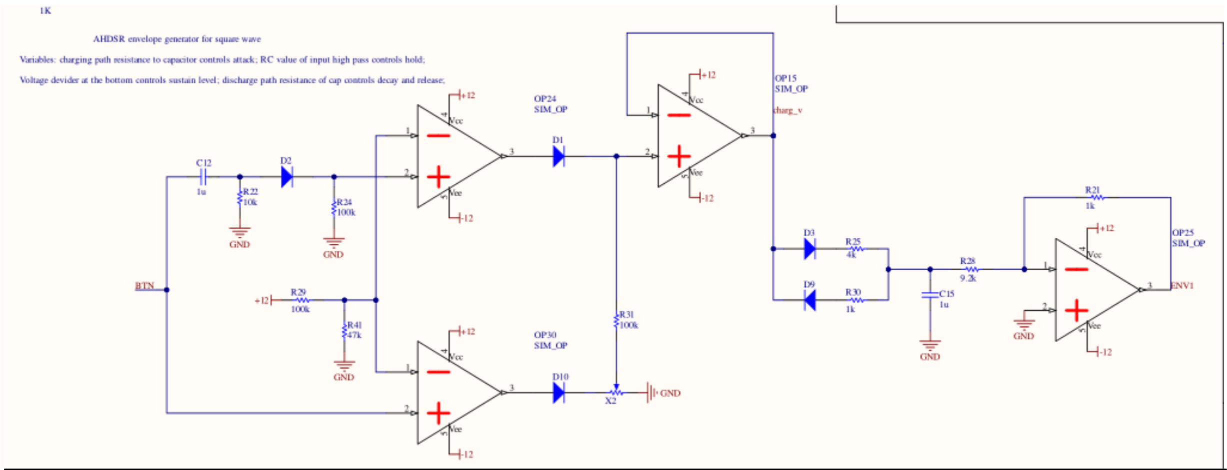

Fig 11: AHDSR envelope generator

3. Attack and Hold Phase (Left Side):

   - C12 (1 μF) begins charging through resistors R22 and R34 (each 100kΩ).
   - This defines the attack time: the rate at which the envelope voltage rises.

   Sustain Level Control:

- The voltage divider formed by R29 (100kΩ) and R34 defines the sustain voltage level (i.e., the voltage the envelope holds when the button remains pressed).
- The op-amp OP30 compares this sustain reference with the envelope signal, likely helping maintain or clamp the voltage at this level.

4. Decay and Release Path:
    - Once the button is released, the capacitor discharges through R31 (100kΩ) and X2 (adjustable pot or control).
    - The diode D2 allows current to flow only during discharge (decay/release).
    - This path defines:
        - Decay: Time to reach sustain level after attack.
        - Release: Time to return to zero once the button is released.
5. Output Shaping Stage:
    - OP15 buffers the final envelope signal after decay and release phases.
    - Diode D3 and R35 form a peak-detection or edge-smoothing block.
    - C15 (1 µF) and R36 further smooth the envelope.
    - Final output is buffered by OP5, producing ENV1, the final ADSR envelope signal.
6. Tuning Parameters:
    - Attack: Set by R22 + R34 and capacitor C12.
    - Decay & Release: Set by R31, X2, and diode D2.
    - Sustain: Controlled via the voltage divider (R29, R34).

Why D3 and D9 Are Used – Backup and Protection Purpose:

1. Directional Control of Signal Flow:
    - D3 and D9 ensure that current only flows in the intended direction — this is crucial for correct envelope generation.
    - D3 ensures the capacitor (C15) charges through R35 only when a valid signal is present at charge_v.
    - D9 ensures the voltage is correctly passed to the buffer stage (OP5), blocking any reverse signal that might corrupt the envelope.

2. Prevents Reverse Feedback or Latch-Up:
    - Without D3 or D9, there's a risk that unexpected feedback from OP5 or other parts of the circuit might:
        - Discharge the capacitor prematurely.
        - Interfere with proper decay or release timing.
    - These diodes isolate the stages to avoid such latch-up conditions.
3. Stability in High-Impedance Conditions:
    - During simulation or real-world cases, high-impedance states or floating nodes can result in erratic behaviour. D3 and D9 force logical transitions, ensuring clean rise/fall edges of the envelope.

4. Backup Against Noise or Glitches:

   - If glitches or high-frequency transients exist in the system, D3 and D9 act like gates that ignore them unless they're in the correct polarity and magnitude — adding reliability.

     Voltage-Controlled Amplifier (VCA) configuration using two complementary NPN transistors (Q1 and Qn1) with a shared emitter node that acts as the control point, which in this case is modulated by an envelope signal (ENV1).

1. Differential Amplifier Structure:

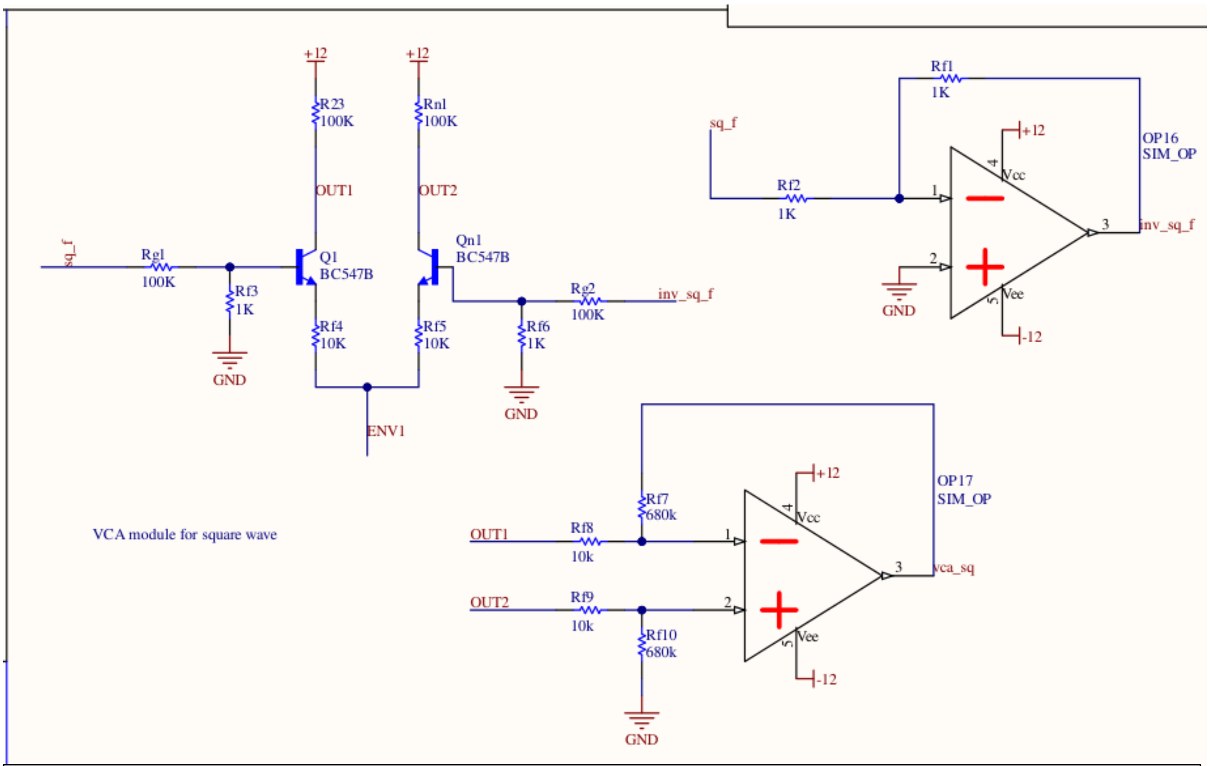

Fig 12: VCA module

- Q1 and Qn1 form a differential amplifier with their emitters tied together at ENV1.
- The bases receive complementary square wave signals:
  - sq_f goes to Q1's base via Rg1.
  - inv_sq_f goes to Qn1's base via Rg2.

2. Signal Path:

- The differential pair compares the voltages on their bases (sq_f and inv_sq_f) and passes the signal to the collector outputs:
  - Q1 collector → OUT1
  - Qn1 collector → OUT2

3. Biasing and Load:

- Each collector is connected to +12V via 100k resistors (R23 and Rn1).
- These form the load resistors that allow the amplified signal to be extracted from the collector terminals.

4. Gain Control via Envelope (ENV1):

- The common emitter node (ENV1) receives the ADSR envelope.

- This node controls the emitter current of both transistors simultaneously.
- Since the gain of a differential amplifier is proportional to emitter current, the envelope voltage directly modulates the amplitude of the output.

5. Purpose of Resistors Rf4 and Rf5 (10k):

- These helps stabilize the emitter voltages and control bias currents for proper transistor operation.
- They ensure symmetrical behaviour in the pair and influence linearity.

End Benefits:

- The circuit modulates the amplitude of the square wave signal using the envelope signal.
- As the envelope voltage (ENV1) increases, more current flows, boosting the gain.
- This results in an envelope-shaped square wave output at OUT1 and/or OUT2.
- In short, this is a dynamically amplitude-controlled square wave—very useful in synthesizers for volume shaping.

This part of the circuit collects the differential outputs (OUT1, OUT2) from the transistor pair and combines them using an op-amp (OP17).

1. Inputs: Differential Combination

- OUT1 and OUT2 from the collectors of Q1 and Qn1 go into two voltage divider paths:
    - OUT1 → R8 (10k) → inverting input of OP17
    - OUT2 → R9 (10k) → non-inverting input of OP17
- This arrangement allows for the differential signal across the transistor collectors to be combined and amplified.

2. Resistive Feedback Network

- Rf7 and Rf10 (both 680kΩ) form the feedback resistors for OP17.
- This sets the gain and ensures symmetrical contribution from both OUT1 and OUT2.

3. Output: vca_sq

- The final audio-ready square wave output is taken from the output of OP17 and labeled as vca_sq.
- This output is the envelope-controlled, amplitude-shaped square wave—perfectly suitable as an audio signal.

Utility:

- The op-amp essentially acts as a differential summing amplifier with high input impedance and unity gain configuration.
- Because OUT1 and OUT2 are complementary (from the diff pair), this stage cancels common-mode noise and gives you a clean, amplified square wave.
- The envelope (ENV1) still controls the dynamics of the signal, but now it's buffered and conditioned for further use (e.g., output to speaker amp and mixer).

Mixer

Finally, we use a 3-channel mixer and an Op-amp based Adder to mix the 3 input channels to get a musical note. Further we add a buffer stage and perform a soft clipping at around 4V to make the Output less harsh to the ears. Finally with an inverting Amp config we end one proper music note.

1. Soft Clipping Stage

Purpose: To gently limit peaks and add warm harmonic content.

- Key Component: First op-amp (leftmost) with diodes in feedback path (D23, D24).
- Mechanism:
    - Diodes are placed in the feedback loop of the op-amp (OP48).

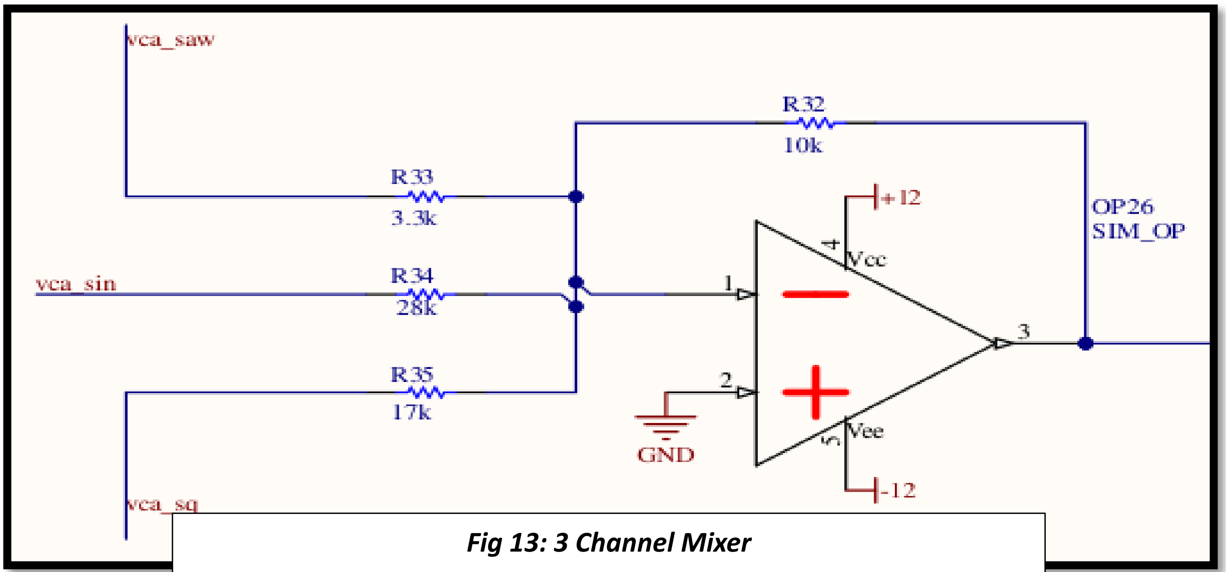

Fig 13: 3 Channel Mixer

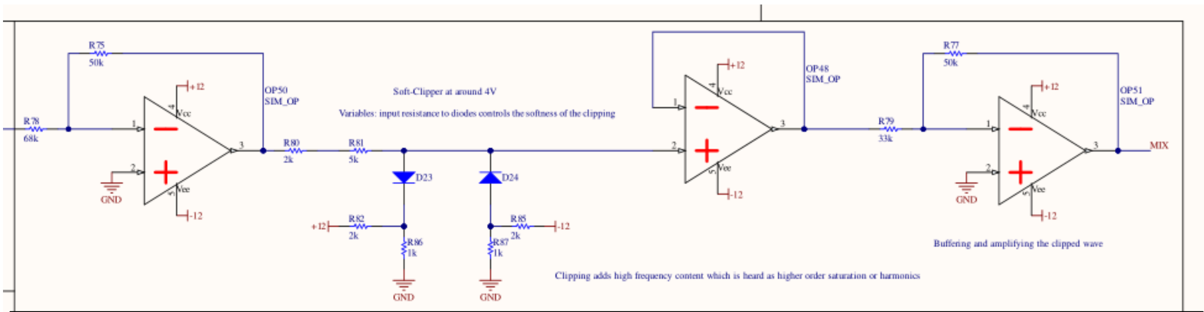

Fig 14: Soft -Clipper at 4v

    - Soft clipping occurs when input signal exceeds the diode's forward voltage threshold (~0.6–0.7V for silicon diodes).
    - Instead of abrupt cutoff, the output voltage gradually compresses, producing even-order harmonics.
- Tuning Softness:
    - R86 and R85 control diode current and affect clipping softness.
    - Lower resistance → more current through diodes → earlier and softer clipping.
    - Higher resistance → more signal required before clipping starts.
- Sonic Effect: Produces tube-like warmth, mild distortion, and a musical compression of transients.
- Clip Voltage: Circuit is set to clip around ±4V.

2. (Optional/Implied) Hard Clipping Stage

Purpose: Add aggressive saturation if signal exceeds op-amp headroom.

- While not explicitly marked, hard clipping naturally happens:
    - When the output of the soft clipper exceeds the op-amp's supply rails (±12V), the op-amp itself hard clips the signal.
    - Creates flat-topped waveforms if driven too hard.
- Sonic Effect:
    - Adds odd harmonics and can sound harsh or buzzy.
    - Useful for aggressive tones like distorted synth basses or leads.

3. Final Gain & Buffering Stage

Purpose: Amplify, restore level, and provide low output impedance.

- Final Op-Amp Stage (OP51):
    - Acts as a buffer and output gain amplifier.
    - Input from soft clipper passes through resistor R79 (33kΩ) to non-inverting op-amp configuration.
- Gain Control:
    - Determined by the resistor values (R77 = 56kΩ).
    - Provides fixed gain to boost the clipped signal to desired line level.
- Sonic Effect:
    - Maintains waveform shape while increasing loudness.

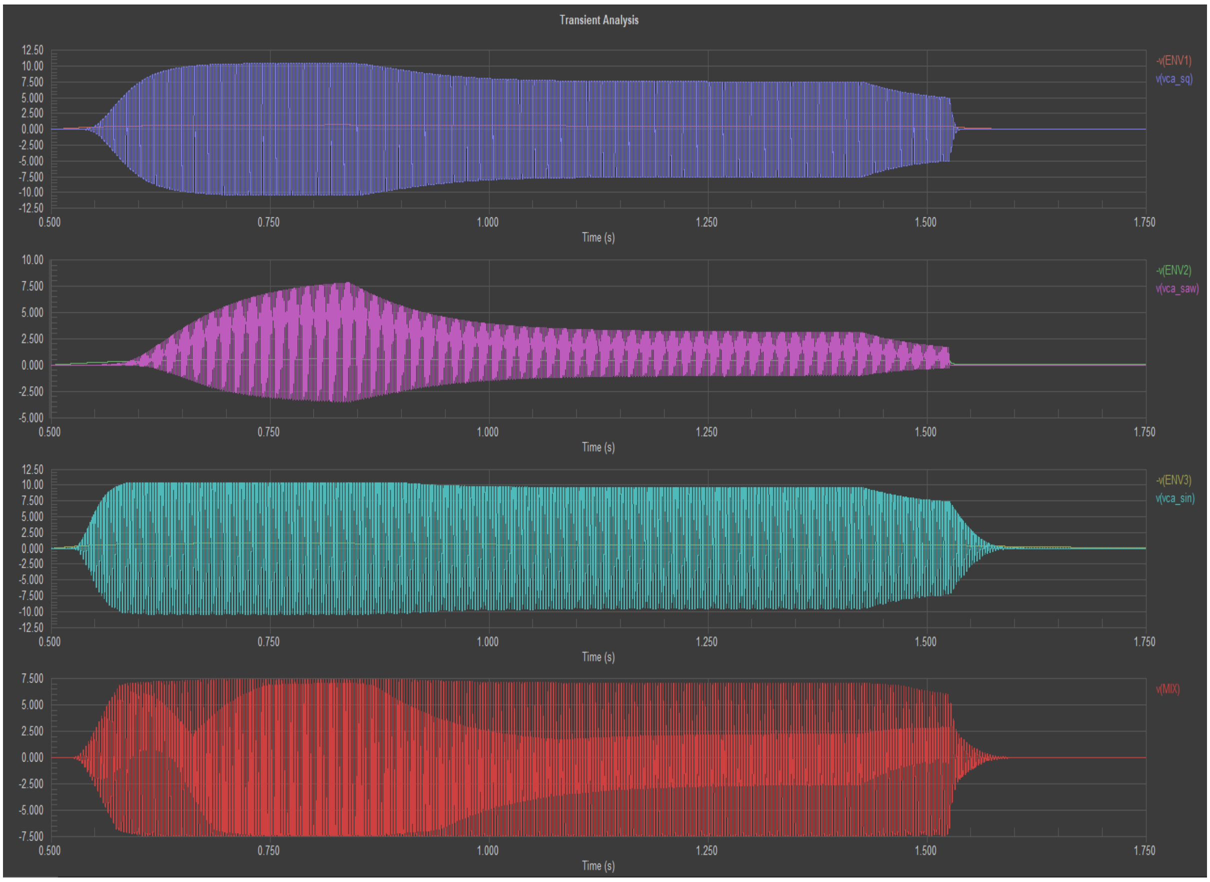

Fig 15: Time response of the circuit

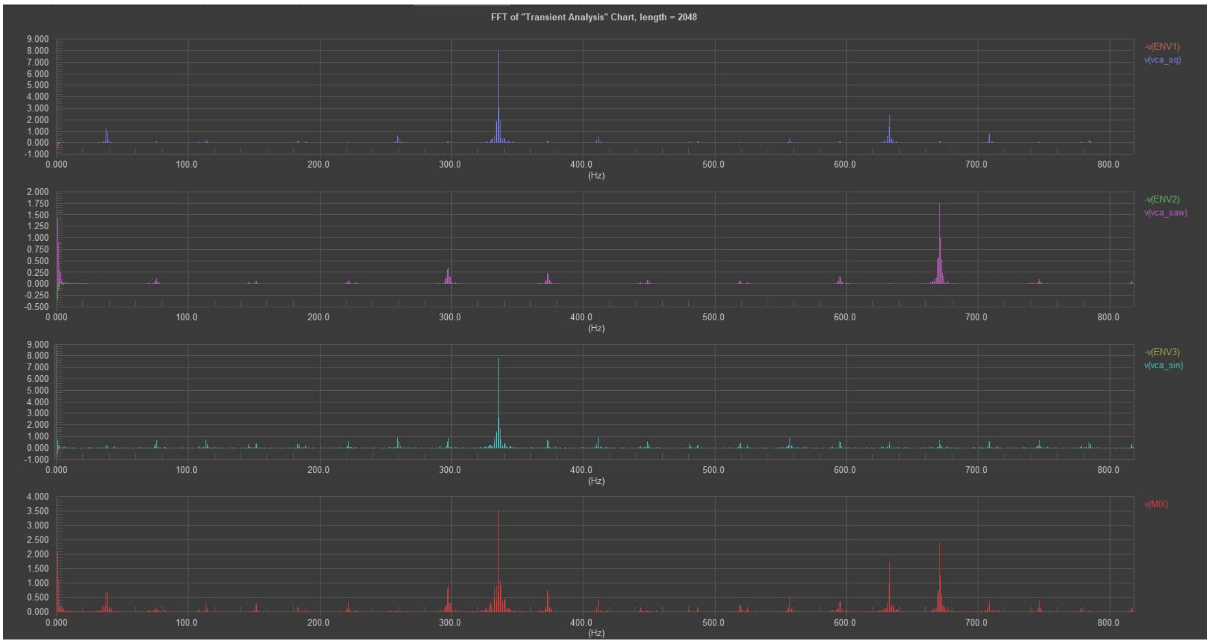

Fig 16: Frequency response of the circuit

Conclusion

The transient and frequency response analyses confirm that the designed analog synthesizer successfully generates and shapes multiple waveforms—sine, square, and sawtooth—with distinct harmonic content. The mixed output demonstrates smooth integration of individual signals, maintaining musical richness while avoiding excessive distortion. The AHDSR envelope and resonant filtering provide dynamic control, resulting in a versatile and expressive sound output. Overall, the circuit achieves stable operation across the intended audio range, validating its effectiveness for analog music synthesis.

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