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Block Diagram

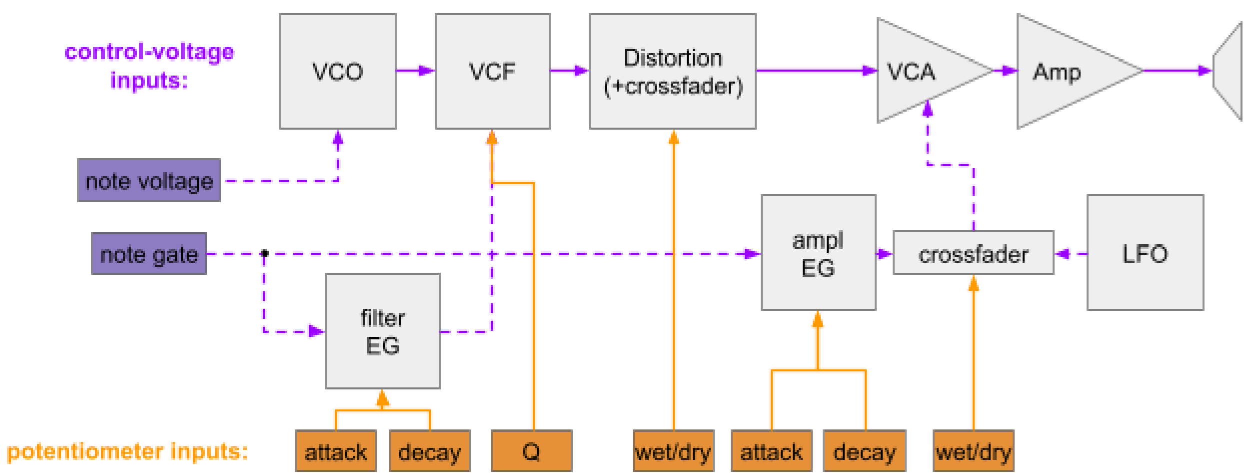

Figure 1. Potentiometers are shown in orange, and time-varying signals are shown in purple.

Input/Output specifications

Module	Input	Output
VCO	0.2 V - 3.3 V note voltage	4Vpp, 60-2000 Hz sawtooth
Filter EG	0/3.3 V note gate	0-8 Vpp tunable AR signal
Ampl. EG	0/3.3 V note gate	0-4 Vpp tunable ADSR signal
LFO	-	0-4 Vpp 500mHz sine wave
Crossfader	0-4 Vpp LFO & 0-4 Vpp Ampl. EG	0-8 Vpp tunable weighted sum of LFO & Ampl. EG
VCF	4 Vpp VCO and 0-8 Vpp filter EG	~4 Vpp filtered sawtooth tunable
Distortion	~4 Vpp VCF	~4 Vpp distorted signal
VCA	~4 Vpp Distortion and 0-8 Vpp crossfader	0-4 Vpp filtered output
Amp	0-4 Vpp audio signal	0-4 Vpp audio signal

User Interface

The musician's primary method of interaction with the instrument is through a microcontroller that generates a digital note-on/note-off signal (0/3.3 V) for the envelope generator and control voltage for the voltage controlled oscillator (0 to 3.3 V using 12 bit DAC on STM32H7). The musician can also control the synthesizer through the potentiometer inputs into the envelope generators, voltage controlled filter, and distortion modules.

Voltage Controlled Oscillator (VCO)

The voltage controlled oscillator is the heart of the subtractive synthesizer. It generates a sawtooth waveform with rich harmonic content which is later passed through additional stages of modulation to produce the desired output sound. The desired output frequency should be a linear function of the input note voltage.

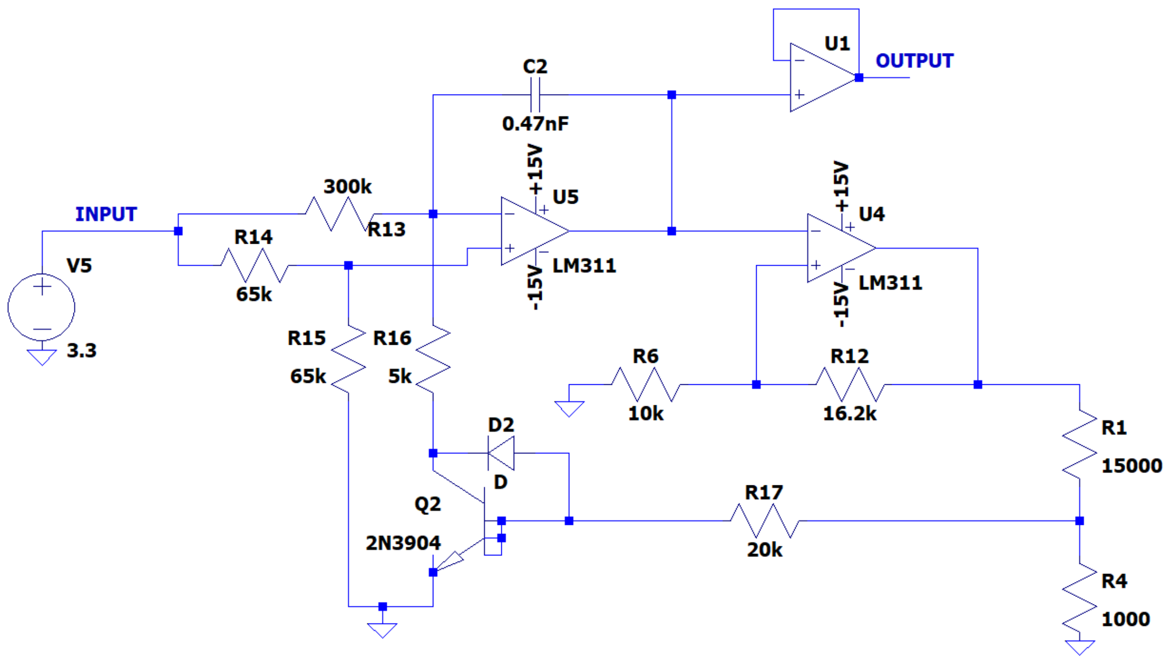

Figure 2. Voltage Controlled Oscillator Topology. Input voltage defines the charging current of the capacitor and Q_2 provides a path for fast discharge. This results in a sawtooth on the output
.

Operation of VCO

Initially, Q_2 is off and the non-inverting input to the Schmitt trigger (U_4) is at -5 V. To create the sawtooth, the input note voltage charges the capacitor (C_2) with constant current through R_{13}. After the output voltage gets below -5 V, the Schmitt trigger switches and opens Q_2. This creates the low-resistance discharging path for the capacitor through R_{16}, which quickly brings the output to +5 V. At that point, the Schmitt trigger switches again, which closes Q_2, and the cycle starts again.

Human ears are incredibly sensitive to small deviations in frequency, so it is critical for the VCO output frequency to be a linear function of the input. In order to have the capacitor switch very quickly, and for the frequency to be linear in input voltage, Q_2 must be able to switch on and off very fast Otherwise, there will be an overshoot on the output as the capacitor will keep discharging for some time after the output crosses the threshold. To make sure that that does not happen, we use LM311 comparators together with a baker clamp on the switching BJT.

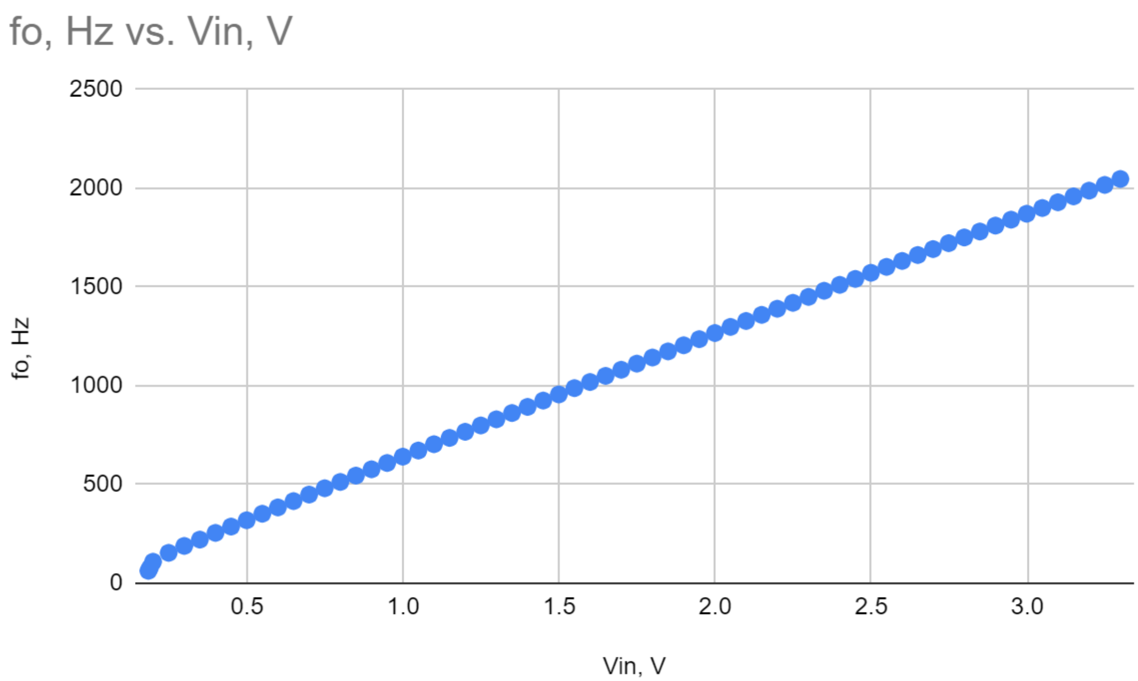

Figure 3. Measured output frequency vs Vin.

As shown in Fig. 3, the measured frequency is mostly linear with the input voltage. However, to improve the sound quality, this curve is approximated by several linear pieces and a 5th order polynomial closer to the origin. Each linear piece has a correlation coefficient of 1, and the polynomial section close to the origin has a coefficient of 0.9998.

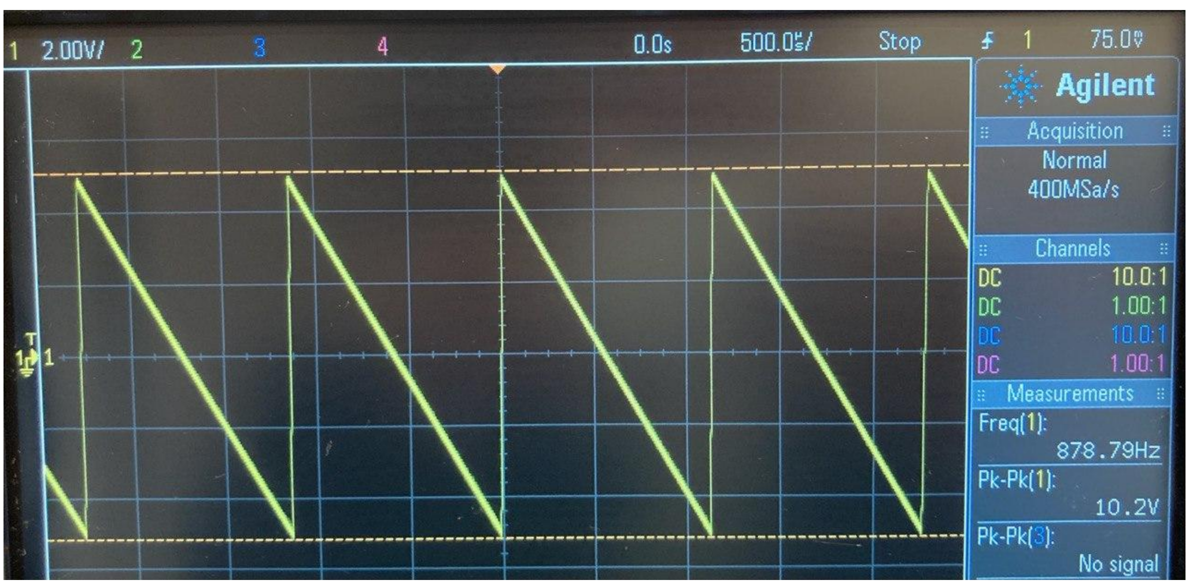

Figure 4. Oscilloscope trace of VCO output.

Voltage Controlled Filter (VCF)

The voltage controlled filter gives the synthesizer the capability to produce different timbres from a single sawtooth wave. By applying a filter to the oscillator output, the VCF changes the presence of harmonic content in the audio signal. This module is a 4-pole lowpass filter with tunable feedback to create corner peaking (resonance near the pole frequency).

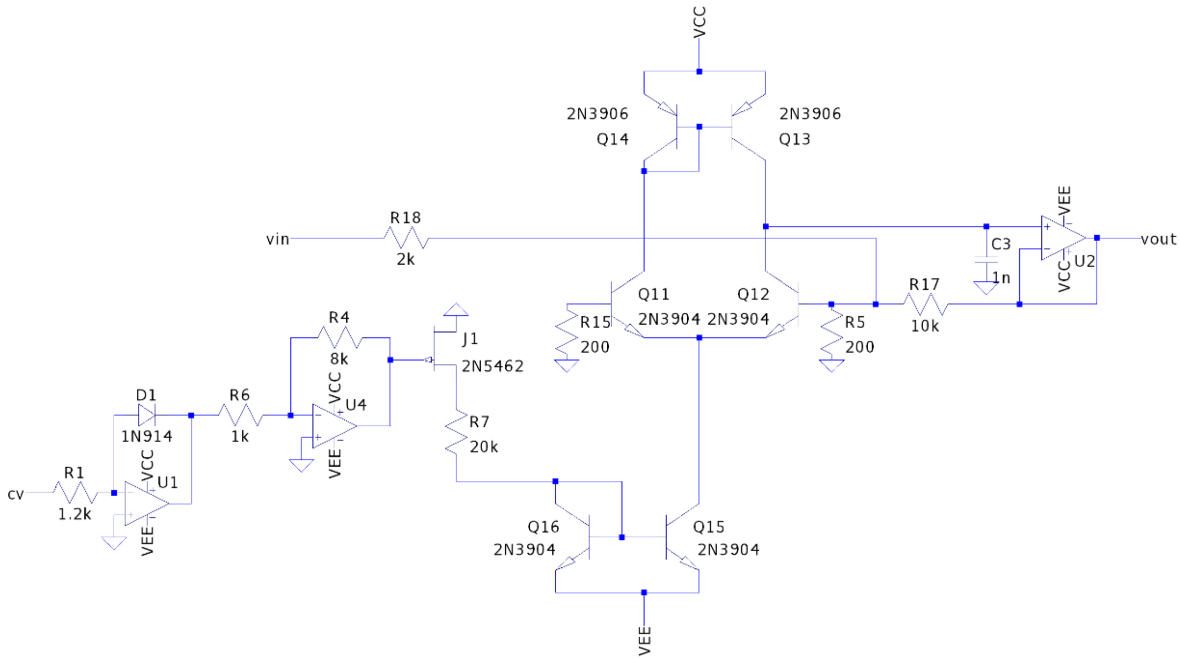

Figure 5. Single-pole lowpass filter. Bias current through Q_{15} controls the frequency of the dominant pole

Operation of single-pole cell

A single cell of the filter is configured as a differential op-amp in negative feedback, with tunable bias current and large capacitive load after the first stage. If the input signal is small, then we can linearize the circuit using small-signal parameters. The transconductance g_m of Q_{11} and Q_{12} will then be a function of the bias current set by Q_{16}. This therefore allows the differential output current between the collector of Q_{13} and Q_{12} to be tunable based on the bias current of the circuit. This will set the charging time of C_3 for a differential input voltage. If g_m is large, then for the same differential input voltage, the output current will be larger, so the charging time for the capacitor will be faster, making the frequency of the dominant pole higher. If g_m is small, then the charging time of the capacitor will be slower, making the dominant pole frequency lower.

The bias current is set through a current mirror. The JFET J_1 is used to create nonlinear changes in bias current as a function of input voltage. The logarithmic amplifier and inverting amplifier stage is used to create a spread of bias currents which produces the desired changes in cutoff frequency as a function of input control voltage.

Originally, the design was to use the highly nonlinear relation between I_c and V_{BE} of a BJT, but this proved too difficult to control without dramatically sacrificing the range of currents available. If V_{BE} were set too high, then the current mirror would draw too much current through the circuit, causing the BJTs to exceed their maximum ratings and explode. The use of J_1 and R_7

keep the current through Q_{16} and therefore Q_{15} and the rest of the differential input stage well below 1 mA. A higher current isn't really necessary, as this setup allows for a -3dB cutoff frequency above 71 kHz (measured), which is well above the audible range, so it will have no perceptible impact on the signal.

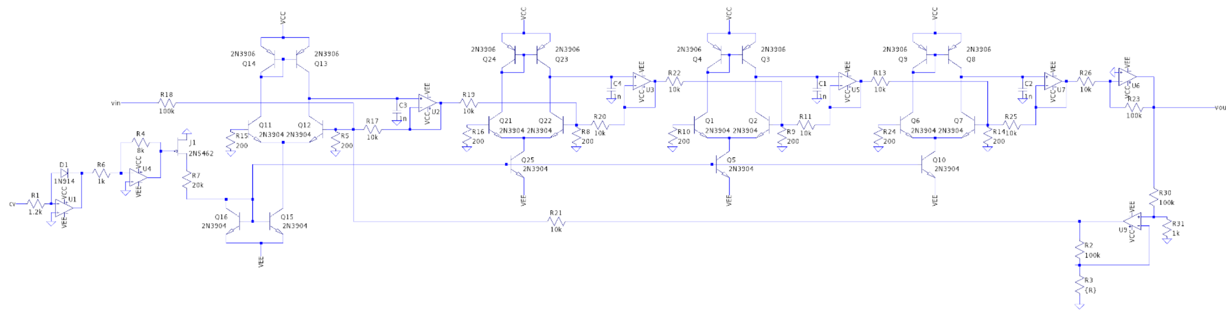

Figure 6. Four-pole lowpass filter with corner peaking. A potentiometer controls the amount of output fed back into the input to adjust the amount of corner peaking.

Operation of four-pole filter

The four-pole lowpass filter works by cascading multiple single-pole stages together to achieve a steeper roll-off. In practice, it would've been more effective to implement a more sophisticated filter structure to achieve steeper roll-off (such as a chebyshev or elliptic filter), but more analysis would need to be done on the impact of resonant feedback on the performance of the circuit, as the ability to control the corner peaking of the transfer function was very important to us in terms of the ability of the circuit to make interesting sounds.

The corner peaking (resonant feedback) is achieved by feeding some of the output back and summing it with the input into the inverting input of the first stage. The feedback amount is tunable by a potentiometer attached between the output of an op-amp and ground, with the wiper tied back to the inverting input of the op-amp. This corner peaking manifests as a small rise (resonance) in the transfer function of the filter around the pole frequency. Simultaneously, the pass-band gain is reduced. This can be compensated by changing the operation of the feedback circuit to have two voltage-controlled amplifiers, but we chose to keep it this way so that the overall amplitude of the waveform didn't change too substantially.

Corner peaking low-pass filter as a VCO

For large feedback fractions, this circuit self-oscillates (i.e. no input signal is required to start oscillation) with a fairly pure sinusoid tone (harmonic distortion between fundamental and first overtone measured as 38 dB, so it's pure enough that it sounds sinusoidal), the frequency of which is tunable by the bias current.

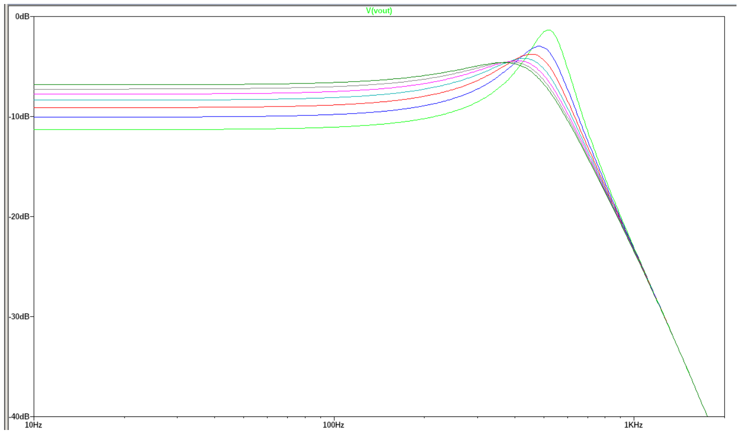

Figure 7a. Simulation of effect of corner peaking on filter transfer function

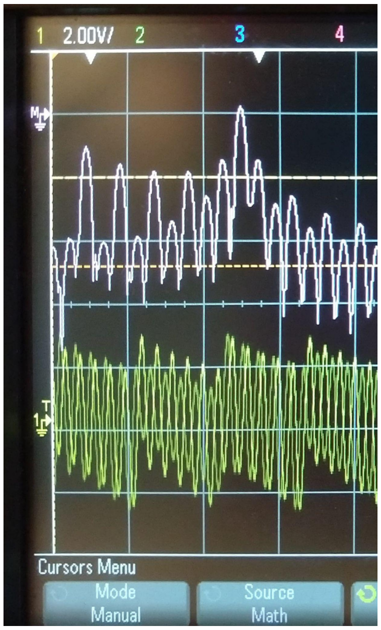

Figure 7b. Oscilloscope trace of corner peaking (time-domain in yellow, FFT in pink)

Wavefolder

The wavefolder circuit distorts the signal by introducing vertical jump discontinuities. With a simple adder circuit and potentiometer, the wavefolder input and output can be crossfaded to change the amount of distorted signal present in the output.

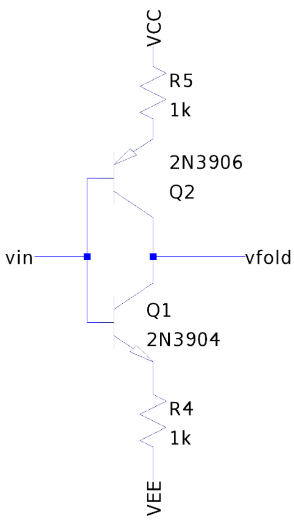

Figure 8. Single Lockhart wavefolder (load not shown). Only one transistor is on at a time.

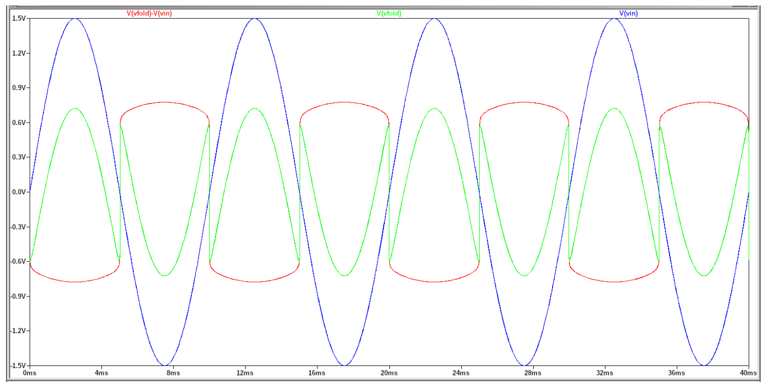

Figure 9a. V_{in} (blue), V_{fold} (green) and V_{fold} - V_{in} (red)

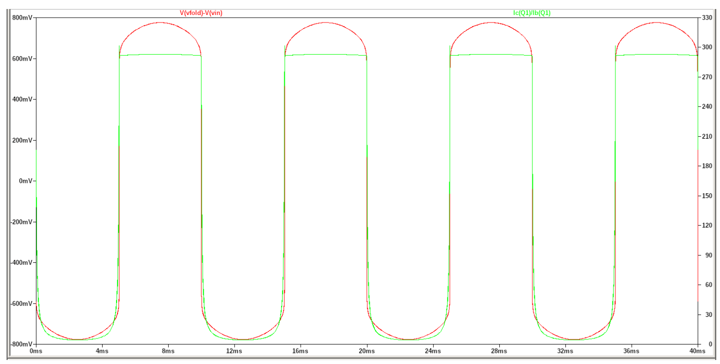

Figure 9b. V_{fold} - V_{in} (red) and $I_{C,1}/I_{B,1}$ (green)

Wavefolder operation

In this circuit, the BJTs don't always act as current amplifiers, and the input signals change by a large amount, so a small signal analysis is inappropriate. In order to model the "off" state of the BJT, a large signal model which models both base-emitter and base-collector junctions as PN diodes is appropriate. The base-emitter diodes in Q_1 and Q_2 are always on, but the base-collector diodes switch from operating in a sort of "reverse current" mode (the normal "on" mode of a BJT where charge from the emitter flows directly to the collector) to forward current (where charge flows from both the emitter and collector into the base). This transition is what causes the 1.2 V swing in the output voltage V_{fold} and creates the distortion.

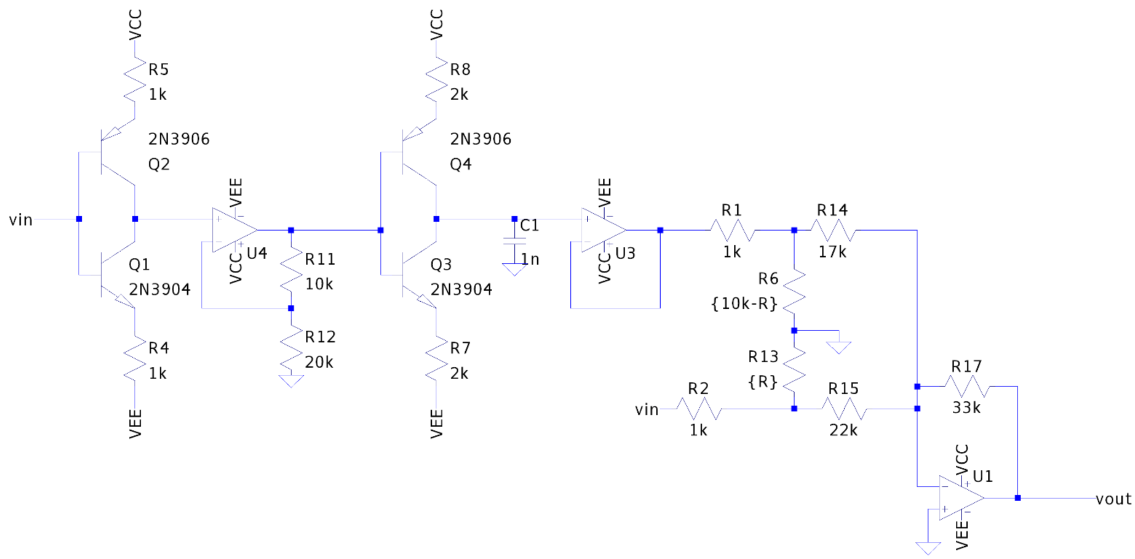

Figure 10. Two-stage wave folder circuit with crossfader

Multistage Operation

By cascading these stages, every zero crossing in the output of each stage creates more zero crossings in the output of the next stage, increasing (very strongly) the presence of high frequency harmonics in the output sound. We can see this effect in an oscilloscope trace of the VCF output with wavefolding applied to it.

A crossfader is also added to allow control over the “presence” or strength of the high frequency harmonics in order to change the timbral qualities of the sound.

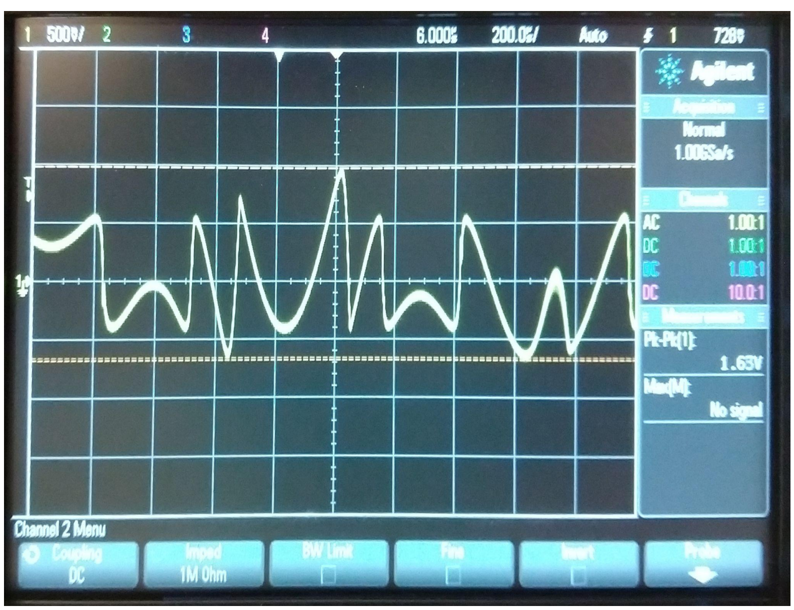

Figure 11. Oscilloscope trace of wavefolder output (input was resonantly filtered sawtooth)

Voltage Controlled Amplifier (VCA)

The voltage controlled amplifier allows the synthesizer to change the amplitude of the audio signal, which combined with the envelope generator, allows the synthesizer to vary the transient amplitude of notes.

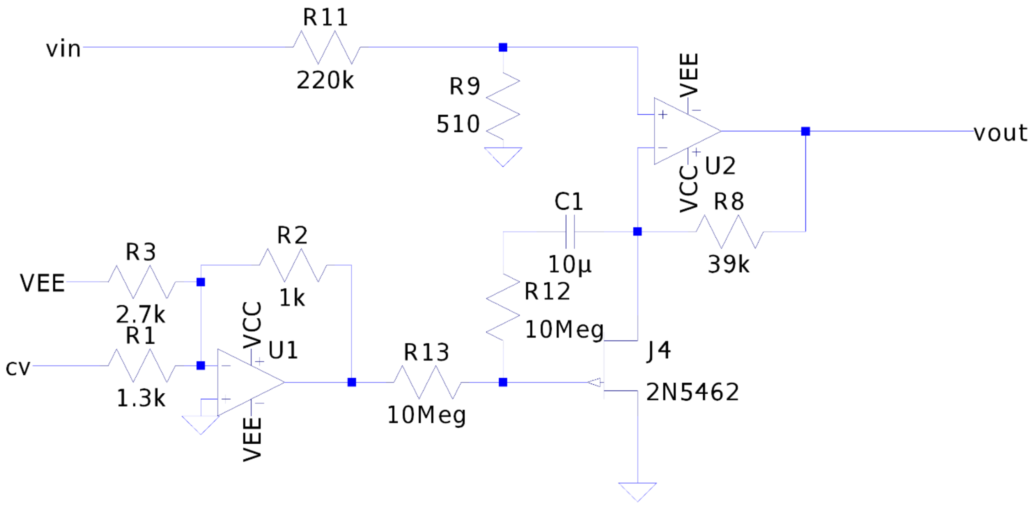

Figure 12. Voltage Controlled Amplifier Topology

The voltage controlled amplifier is configured as a non-inverting amplifier with tunable feedback fraction. By changing the effective impedance of J_4 by changing its gate-source voltage, the ratio between R_8 and R_{J4} changes.

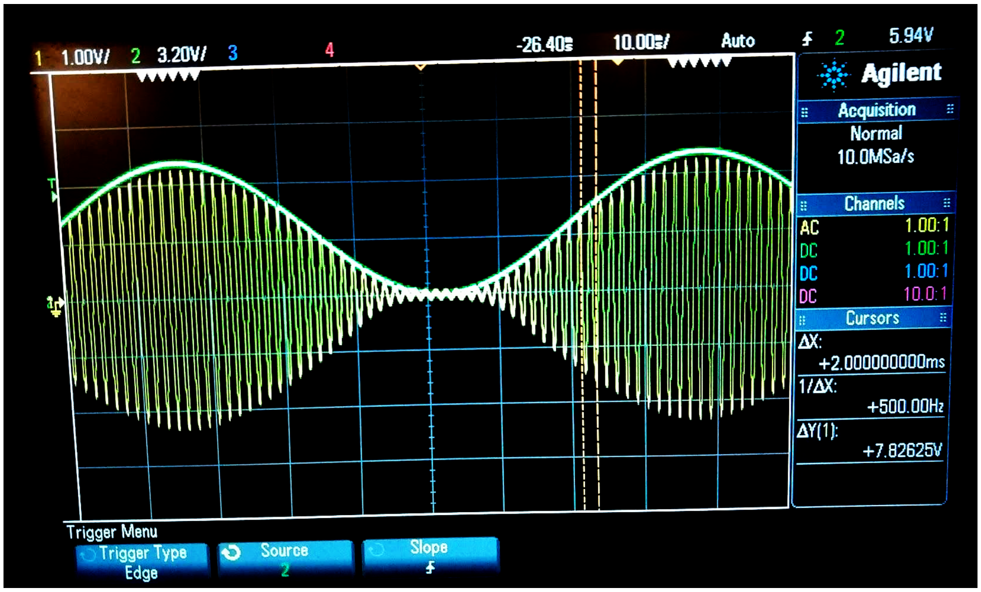

Figure 13. Waveform of VCA (control voltage in green, output waveform in yellow)

Envelope Generator (EG)

The envelope generator gives the synthesizer the capability to change the transient behavior of a note's timbre and amplitude. This allows the musician to create sounds ranging from bright and plucky, to more soothing, warm and flowing sounds.

The synthesizer uses two separate envelope generator versions — one to set the VCF control-voltage and the other to set the VCA control voltage. The envelope generator controlling the VCF is a simple attack-release EG. The envelope generator controlling the VCA is an attack-decay-sustain-release (ADSR) EG.

Attack-Release Envelope Generator (AR)

Attack-Release EG Operation

A positive gate voltage opens Q_1, which opens Q_2, which provides the path to charge the capacitor C_2 through D_2 and R_7. This is the attack stage.

Zero gate voltage closes Q_1, which closes Q_2, and the capacitor C_2 starts discharging through D_1 and R_8. This is the release stage.

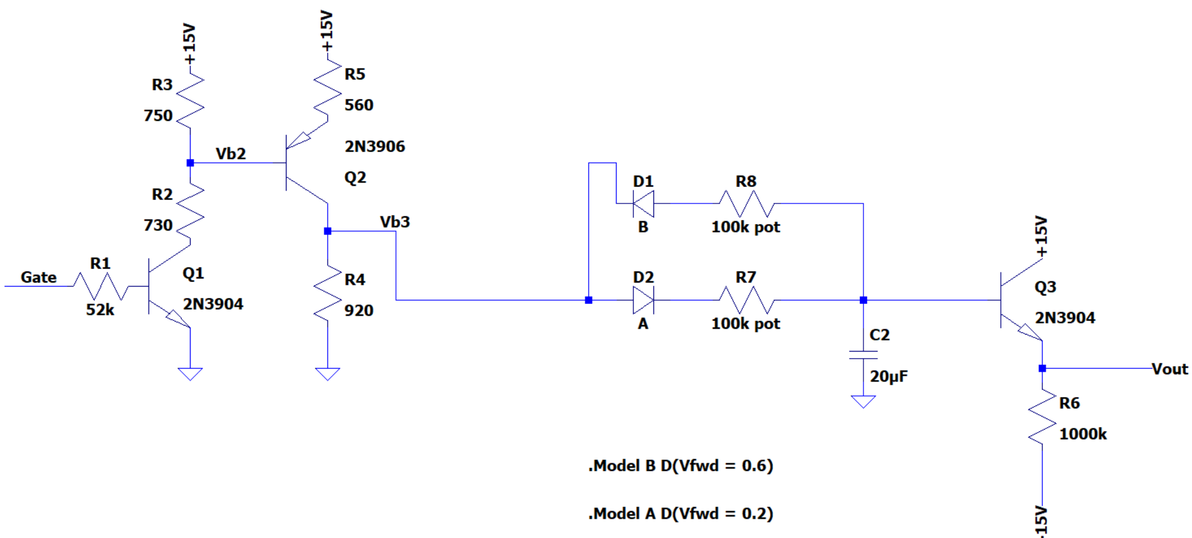

Figure 14. AR schematic

A musician can control the attack and release time with potentiometers R7 and R8.

Figure 15. AR EG output.

Attack Decay Sustain Release Envelope Generator (ADSR)

To allow the musician to create more interesting amplitude transients which more closely model the amplitude transients of acoustic instruments with percussive elements (such as drums, or a piano), we want the input to the VCA similar to that shown Fig. 17, given the gate input:

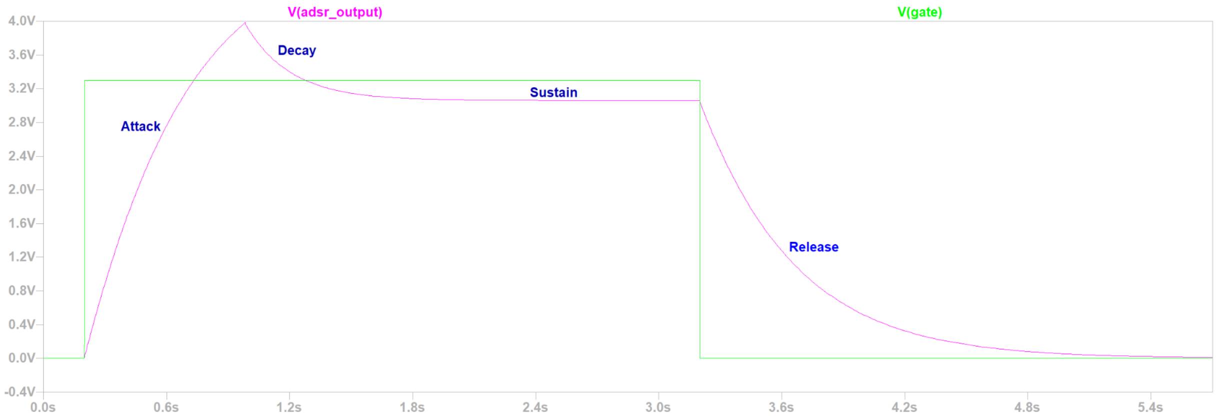

Figure 16. Simulation of the ADSR circuit output, V_gate (green), V_adsr(pink)

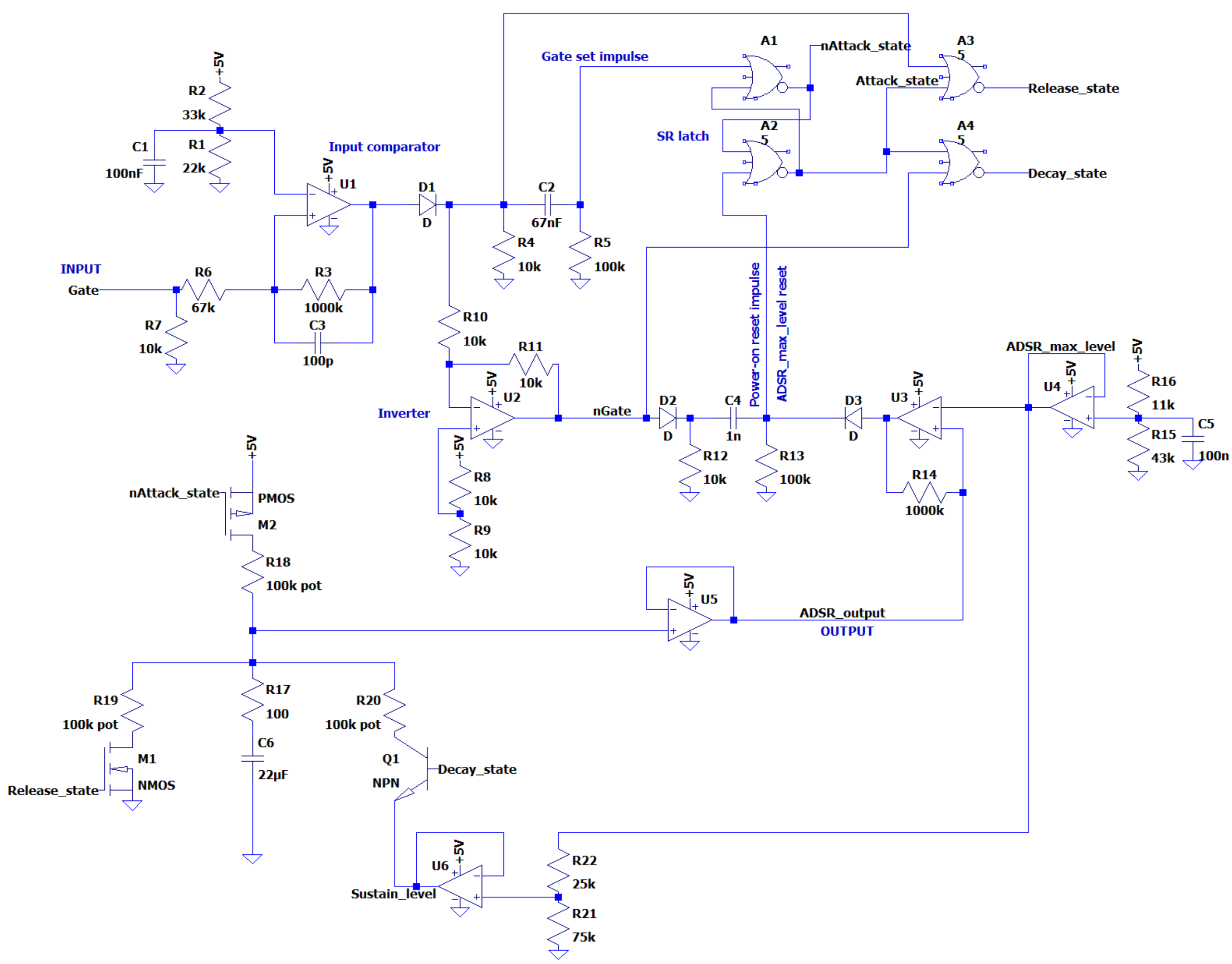

Figure 17. ADSR schematic.

ADSR operation

The output of the circuit is the buffered capacitor voltage (C_6 on the left bottom of Fig. 17), the charging cycle of which is controlled by three state logic levels: Release_state, Attack_state and Decay_state. Note that the sustain part of the ADSR curve is also managed by the Decay_state as it will discharge the capacitor down to the Sustain_level.

The ADSR cycle of the circuit can be described by the following steps:

- When gate voltage goes high, the gate comparator (U_1) output goes to the positive rail, which with the help of the high-pass RC filter (C_2 & R_5), sends a set impulse to the SR latch (A_1 & A_2), which puts the system into the Attack_state. The capacitor starts charging and the output increases.
- When the output of the circuit reaches the maximum level set by U_4, the U_3 comparator resets the SR latch, which puts the circuit into the Decay_state, in which the circuit will stay until the gate voltage goes low. The capacitor will discharge to Sustain_level and maintain it.
- When the gate voltage goes low, the inverted gate signal sends a reset impulse to the SR latch, using the C_4 & R_{13} high pass RC filter. This puts the circuit into its last state - Release_state. The capacitor voltage decays to zero and stays at 0 voltage until the gate voltage goes high again.

A musician can change atack, decay and release times by changing the values of the variable resistors (R_{18}, R_{19}, R_{20} in Fig. 17). Also, the sustain level can be adjusted by changing the potentiometer connected to U_6.

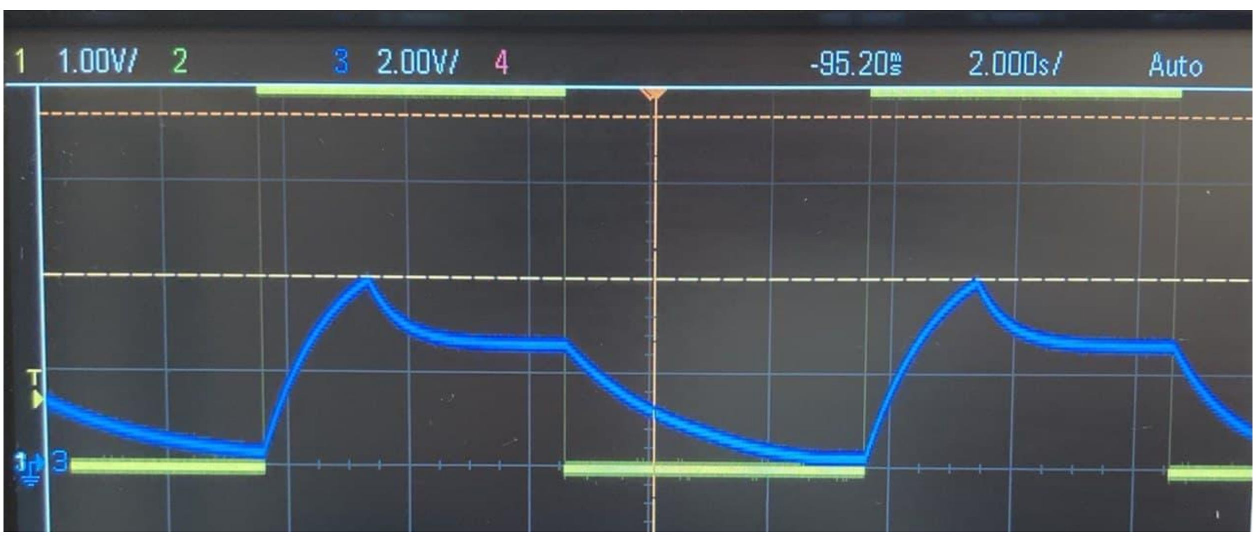

Figure 18. Output of ADSR (ADSR voltage in blue, gate voltage in yellow)

Low Frequency Oscillator (LFO)

The low frequency oscillator (1-10 Hz) can be used to modulate the amplitude of the audio signal to create a tremolo effect, or create an interesting rising and falling flow to a melody.

Phase-shift LFO operation

Self-sustainable oscillations can be achieved if the output of a system in feedback can exist without the input. The closed loop transfer function of this circuit is:

$$\frac{V_{osc}}{i_{input}} = \frac{A}{1 + A \cdot f}$$

Where A = -R_1 is the open loop gain, and f is V_osc/i_f. When A*f = -1, we get an oscillating output without any input, which could be triggered by noise in the circuit. f = -1/A = 1/R_f is achieved by cascading a three pole RC network.

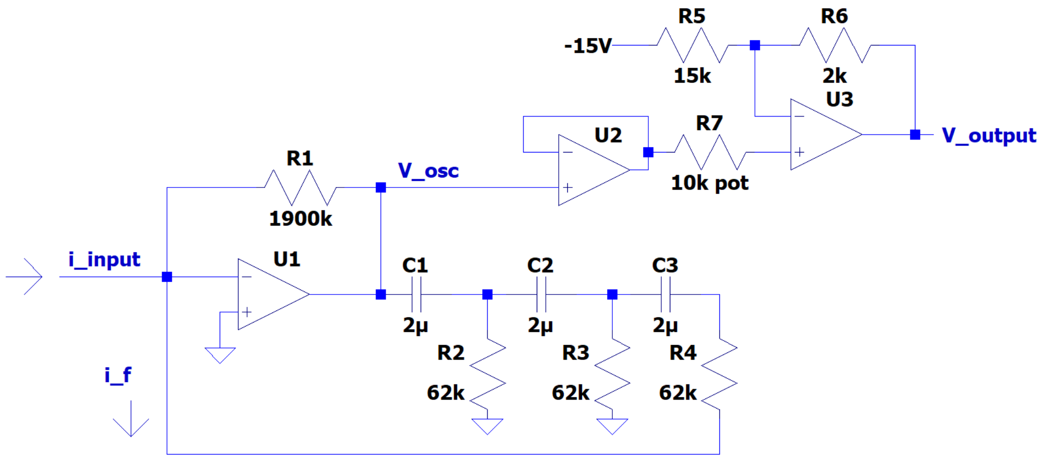

Figure 19. LFO circuit. The closed loop gain from i_input to V_osc is made infinite to achieve the output without explicit input. V_osc oscillates at the frequency at which A*f = -1.

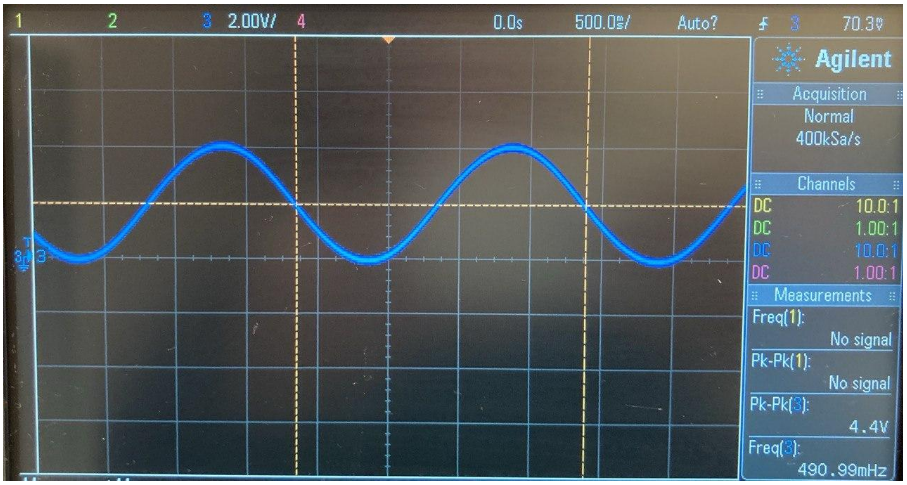

Figure 20. LFO output V_output after shifting and scaling

Note about the R_7 (Fig 19.) variable resistor

Because the oscillator amplitude was observed to be sensitive to changing power outlets, we used R7 to adjust the gain of the V_osc for different setups. We assume that the reason for variation in the amplitude was caused by different frequency content of different power outlets.

VCA input crossfader

For the VCA to function properly, its input must cover the range from 0 to 8 V. At the same time, the musician should have control over the effect of the LFO and ADSR on the input to VCA. This was achieved by constructing a crossfader, using a potentiometer with its wiper tied to ground in the inverting adder circuit, as shown in Fig. 21. Fig. 22 shows the effect of adding LFO to ADSR. Again, using this signal as an input to VCA, the musician can achieve something similar to a tremolo effect.

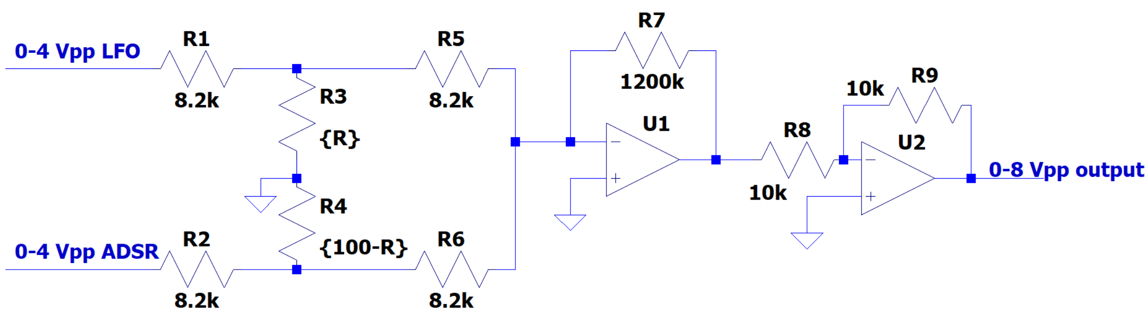

Figure 21. Crossfader topology. By adjusting the potentiometer division, we get a desired ratio of LFO vs ADSR impact on the input to VCA. The resulting amplitude stays almost constant.

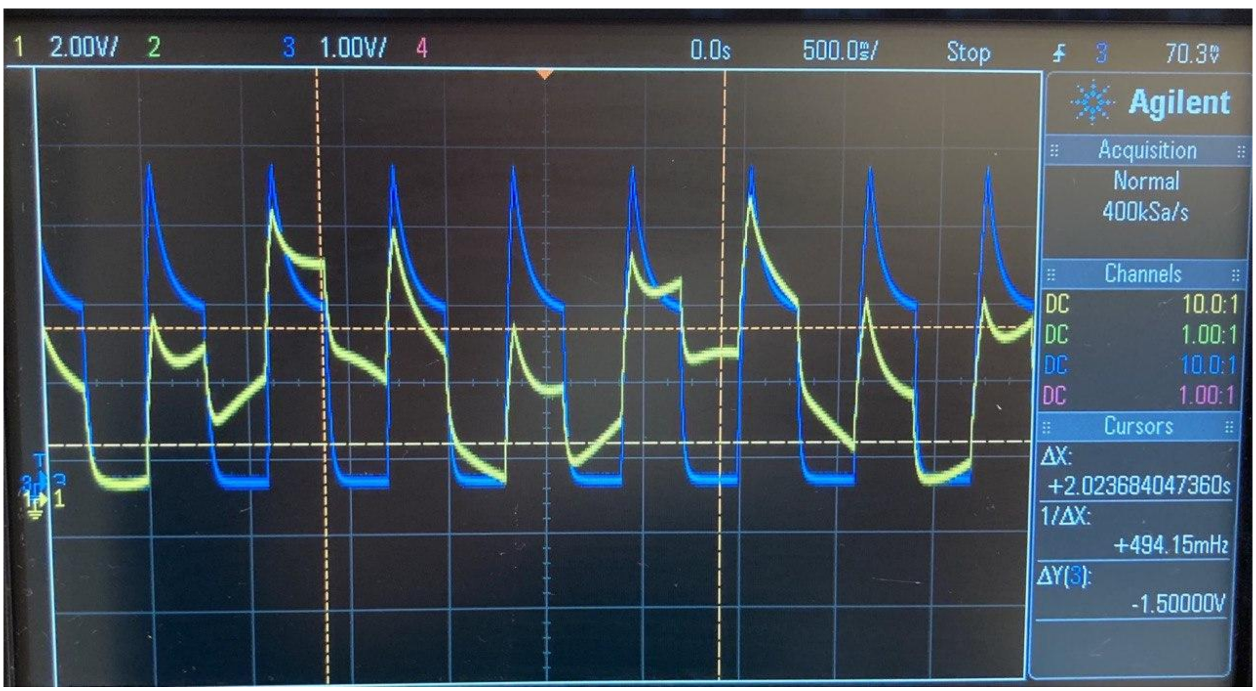

Figure 22. Crossfader output. (ADSR output in blue, weighted sum of LFO & ADSR in yellow)

Push-Pull Output Stage

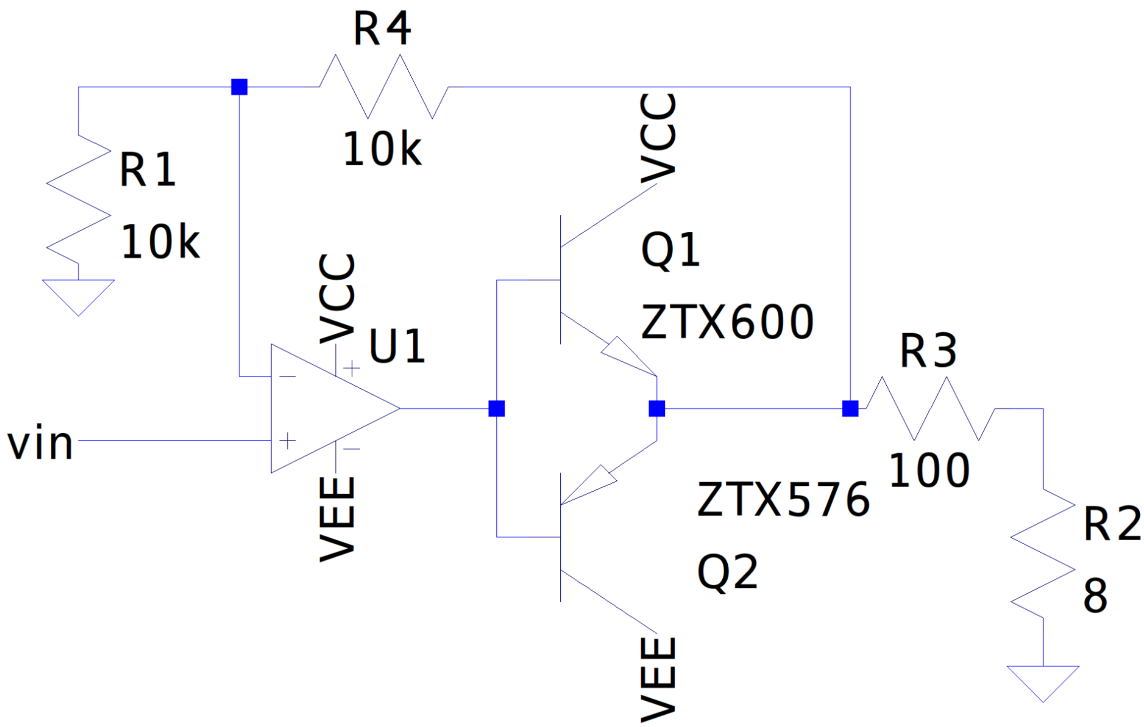

Figure 23. Two high-current BJTs drive the low impedance load. The op-amp is used to remove crossover distortion from forward-bias V_{BE} drop. A 100 ohm resistor is in series with the load R_2 to prevent large current draw, and keep the output volume at a reasonable level for testing in the lab.

Possible improvements

VCO

At low frequencies (< 80 Hz), the VCO output frequency wasn't linear with the input voltage. We suspect that the reason for that is that the input offset current at low frequencies has the same order of magnitude as the charging current. To decrease the effect of the offset current, we would need to increase the minimum charging current. The solution could be decreasing R13 and increasing capacitance C2 to keep the period constant. However, this would result in higher discharge time through R16 and less ideal saw-tooth.

ADSR

Because of the 4 digital logic NOR gates, the design could be improved by changing comparators to Schmitt triggers to increase the noise rejection.

LFO

The current design output amplitude turned out to be sensitive to changing the power outlet and had to be manually adjusted with a variable resistor. Another topology or op amp might be considered to remove this effect. A tunable frequency LFO would be very useful for creating more interesting sounds.

VCA Input Crossfader

To have constant output amplitude independent of how we weight the sum of LFO and ADSR, in the current topology, the potentiometer resistance R_3+R_4 (Fig. 21) has to be much smaller than R_5 and R_7. This results in a very high gain of the adder and some undesired offset on the output of the crossfader. It wasn't noticeable, but it certainly could be improved by removing the offset or using a different topology that doesn't require high gain.

VCF

With four poles, it is possible to create very sharp cutoff filters using an elliptic or Chebyshev filter. Even though these types of filters do not have a flat passband, it could have been useful to experiment with them to determine their usefulness for this type of application.

With corner peaking, the low-frequency passband gain of the filter is attenuated substantially. Adding more sophisticated control to perform resonant feedback while also adjusting output gain would eliminate this reduction in DC gain for high values of corner peaking, keeping the voltage swing of the VCF output about the same.

Push-pull Output Stage

Using high-power MOSFETs would have been a good idea for the output stage to drive the low impedance speaker with large power. As is, the circuit we constructed cannot drive an 8 ohm load without damaging the push-pull BJTs. In order to fix this problem, the maximum current rating for the push-pull transistors would need to be over 2 A.

Conclusion

Through the tunability of the modules that we created, we successfully achieved a great variety in sounds that our synthesizer can create. By adding potentiometers that control the VCF cut-off frequency & corner peaking, ADSR and AR times & sustain level, wavefolding and crossfader outputs, the musician has significant control over the sound characteristics. The synthesizer design met the initial specifications and possible improvements have been proposed to improve the robustness of the synthesizer.