

# **The Building of an Analogue Voltage-Controlled Modular Musical Synthesiser and the Exploration of its Modes of Sound Production.**

**Brian Leach**

17<sup>th</sup> August 2012

MSc in Music and Technology

CIT - Cork School of Music

**Research advisors:** Hugh McCarthy, lecturer, Cork School of Music.  
Paddy Collins, lecturer, Electronics department, CIT.

## **Abstract**

In order to gain a greater understanding of electronic sound production, a fully analogue musical synthesiser with nine separate patchable modules was built, along with a keyboard to control it. The synthesiser contains two voltage-controlled oscillators, two envelope generators, two voltage-controlled amplifiers, a voltage-controlled filter, a low frequency oscillator, and a two-channel mixer. This synthesiser can produce frequencies up to 11kHz and can be tuned to play different equal tempered musical scales. The instrument is a valuable composition and performance tool, capable of producing a great variety of musical sounds.

## **Introduction**

Musicians have a special relationship with their instruments. When a musician understands their instrument fully, its strengths, weaknesses and limitations, they can perform more confidently and create expressive music. The best way to understand an instrument entirely is to build it from scratch. The practical portion of this thesis involved building seven separate analogue synthesiser modules which can be patched together in many different ways using short cables. The machine is capable of synthesising a large range of sounds and can be controlled with analogue control voltages from a modified keyboard. The goal of this research thesis is to create a unique instrument that is fully understood. The main outcome is a greater understanding of how sounds can be generated electronically.

It is the author's opinion that when sounds are synthesised digitally, as they usually are in popular synthesisers that are currently available such as the Korg MicroKorg, that the sounds they produce can be 'cold'. This may not be for any physical reason as synthesisers generating sounds at a 44,100Hz sample rate does a very good job at recreating waves that sound analogue to our ears (Howard and Angus, 1996). This coldness can be caused by the use of presets and internal menus that change parameters in an unintuitive or even unknown way. By patching modules together with real cables an exact mode of sound production is either decided on by the user,

or easily understood visually and physically by seeing the way the modules are patched together. This understanding leads to a more genuine musical performance.

It was noted by Sinclair (1998) that it is the specific type of distortion which is inherent in Moog's analogue synthesisers that make them so desirable. Digital synthesis can produce precise frequencies but I would argue that this is where the perfect digital mode of synthesis falls down. An element of unpredictability is desirable when creating music. For this reason it was decided to peruse a fully analogue mode of sound production.

The purpose of this thesis is to build an analogue synthesiser which will be unique and fully understood by me. Through the knowledge gained, I will be able to use it to its full extent as a composition and performance instrument.

Through building this synthesiser a greater understanding of how different sounds are generated will be gained. It is not the intent to create altogether new methods of sound production, but to build modules which can be connected together to synthesise sounds using the methods of Additive Synthesis, Frequency Modulation and Amplitude Modulation. These modules must be built simply and cheaply from scratch using only resistors, capacitors, transistors, diodes, op-amps, 555 timers, potentiometers and lengths of wire, which are all readily available and low cost components.

An analogue synthesiser creates voltages and utilises them to create and shape a sound. There is a direct comparison between voltage waves and sound waves. As waves, they behave and interact in the same way (Mansfield and O'Sullivan, 1998) and so an understanding of how electrical waves work leads to an understanding of how sound waves work. A speaker is a transducer that changes voltages into sound waves. While digital information can also be converted into sound waves, this process is not as intuitive or tangible, and is seldom understood by musicians. This research has made the connection between analogue electronics and sound very clear. With this greater understanding of what is going on inside a synthesiser I can understand how to sculpt any sound I desire by manipulating analogue voltages.

When a musician plays any instrument there is an interaction between the instrument and the player. When using a synthesiser the real interaction is not through the keyboard, this is only a means of triggering the sounds, the main interaction is the one that leads to the musician producing a specific type of sound – the mode of synthesis. This is determined by how the musician sets up the synthesiser, selecting a preset does not put any of the musician's self expression into the sound, building the sound up from simple wave shapes by patching cables is how someone can get real artistic expression from a synthesiser. With every parameter visible and changeable in real time the musician can give a convincing performance, which cannot be reproduced using a computer or a digital synthesiser with presets. Scrolling through menus will never be as personal an interaction between the instrument and the musician as patching cables.

The goal of this research thesis is build an advanced electronic instrument from scratch and to understand fully how electronic synthesisers work. The outcome is a unique instrument, capable of creating a wide variety of sounds, which is invaluable as a musical composition and performance tool.

The synthesiser has nine modules: 2 x Voltage-Controlled Oscillators, 2 x Envelope Generators, 2 x Voltage-Controlled Amplifiers, 1 x Voltage-Controlled Filter, 1 x Low Frequency Oscillator, and 1 x Two-Channel Mixer. It has thirty-two knobs, six switches, and thirty-nine inputs/outputs, all of which are accessible from the front panel.

## **Literature review**

### **History of Electronic Instruments**

Electronic instruments were first created at the start of the twentieth century. Instruments such as the Trautonium, the Theremin and the two hundred ton Telharmonium used electronic oscillators to generate sounds via various modes of interaction including keyboards and touch sensitive strips of metal. Many of these instruments were like organs in that they had different electronic oscillators for each note. This made most electronic instruments much more cumbersome than the small keyboard synthesisers we are accustomed to today. So it was not until these machines could be made more manageable that electronic music took off (Webster 1999, p 72).

### **Musical Electronic Voltage Controlled Oscillators**

By using just one oscillator whose pitch could be altered electronically using a control voltage, synthesisers were made less complicated and more affordable. The key to making a musical voltage controlled oscillator was discovered and outlined by Robert Moog in his 1964 paper 'Voltage-Controlled Electronic Music Modules'. He noted:

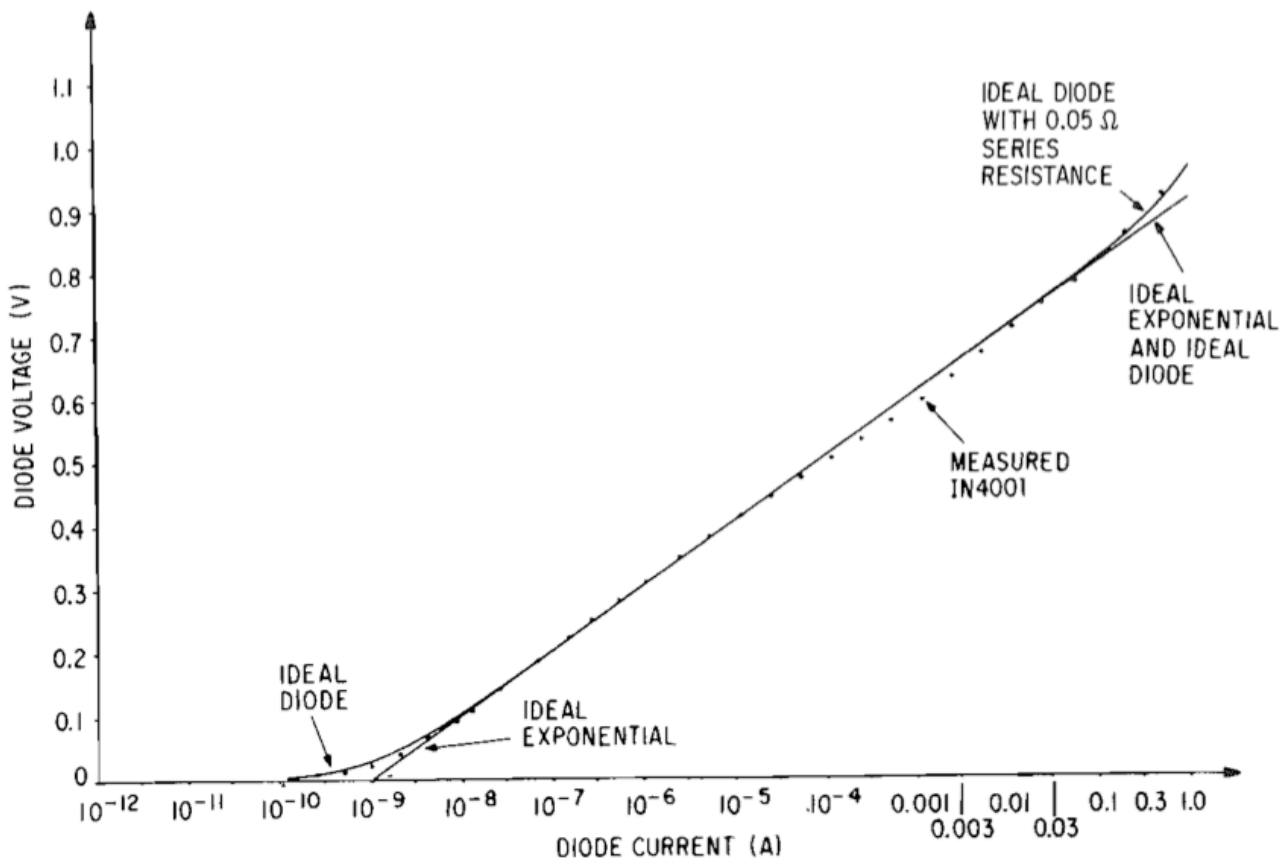
“In Technical measurement and control operations, a linear control voltage – frequency relationship is frequently useful. In the production of music, however, constant frequency differences are of little value. The fundamental subjective quality of frequency change is the interval, which is a ratio of two frequencies. In order to be musically valuable, a VCO should generate a fixed frequency ratio for a given control voltage change. In mathematical language, the frequency should be an *exponential* function of the control voltage.” (Moog, 1964)

Moog went on to have great success manufacturing musical synthesisers that became the standard model for musical synthesisers. Moog was the first to successfully utilise voltage-controlled oscillators for music production (Webster, 1999, p72). The portability, low cost, and ease of use of his instruments lead to electronic sound production integrating into popular music. The form of my synthesiser is the same as

Moog's original synthesizers: separate modules which can be connected in different ways using patch cables to select the desired mode of sound synthesis.

### Exponential Converter

With modern semiconductor technology creating an exponential converter to control an oscillator musically is relatively easy to implement. The diode is a semiconductor device which allows current to pass in one direction only. It has an exponential voltage to current relationships. Chamberlin (1987) proposes the use of a transistor as an exponential converter in his book *Musical Applications of Microprocessors* because it behaves similarly to a diode but has an extra terminal to allow the control voltage to be separated from the output.



**Figure 1. The voltage to current relationship of a real diode compared to that of an ideal diode and a perfect exponential relationship, displayed on logarithmic axes (Chamberlin, 1987, p184).**

The section of *Musical Applications of Microprocessors* on analogue synthesis also outlines circuits to implement all of the other types of modules which were built. His designs for the Exponential Converter, Filter, and Amplifier were used.

## Overview of Operation

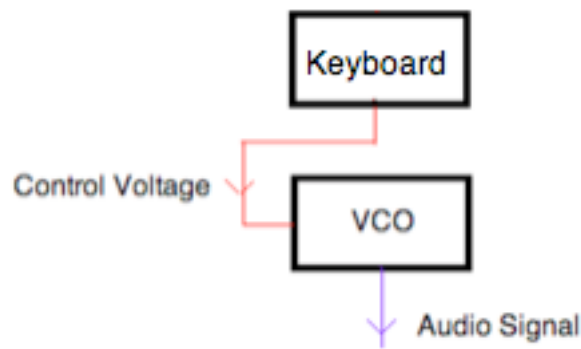
The different modules create two classes of voltages:

- 1) Audio signals, which are the sounds to be processed and eventually heard through a speaker. These are alternating voltages with frequencies within the range of human hearing, 20Hz – 20kHz;
- 2) Control voltages, which are used to determine the frequency of the audio signals and to modulate their amplitude, frequency, and harmonic content.

The modular nature of this project means that each module is separate from the others and also that they can be interconnected in any conceivable way to create many different types of sounds when played via a keyboard or other controller. Some typical modes of sound production and their implementation are outlined below.

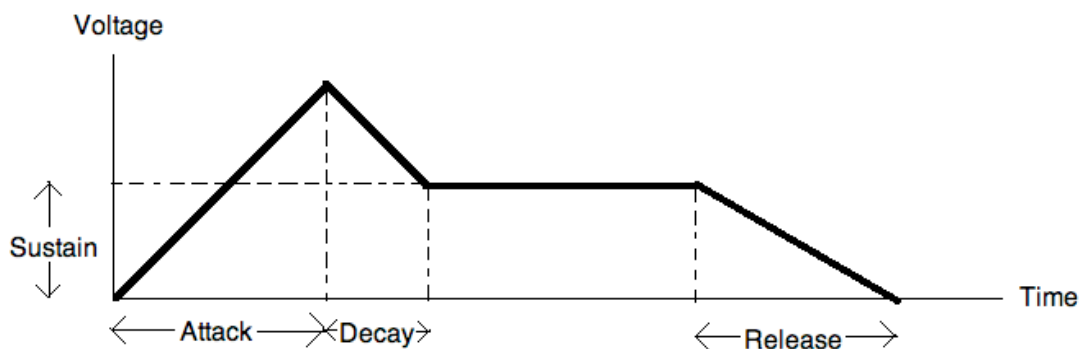
The frequency of oscillation and thus the pitch of the Voltage-Controlled Oscillator (VCO) must be determined by which key on the keyboard is pressed. Thus we need a way to transform keyboard activity into Control Voltages (CVs). A specific circuit for this control voltage will be discussed in the keyboard control section.

This CV output of the keyboard can be connected into the CV input of a VCO. The VCO will then output a signal with a defined frequency. As can be seen in **Fig. 2** the VCO has a CV input and an audio signal output. For clarity I represent control paths with a red line, and audio paths with a blue line.



**Figure 2. A control voltage being transformed into an audio signal with a defined frequency.**

This signal can be patched into the input of the Voltage-Controlled Amplifier (VCA) which multiplies the signal by a number  $\geq 0$ . This number is defined by the control voltage input of the Voltage-Controlled Amplifier. Consider that this CV could be changing over time to give the signal an amplitude envelope. This envelope is created by the Envelope Generator (EG). A typical type of voltage envelope is the ADSR envelope. It has four parameters, Attack, Decay, Sustain, and Release which are set by controls on the front of the synthesiser. The envelope generator can draw out a voltage envelope over times ranging from hundreds of milliseconds to tens of seconds.

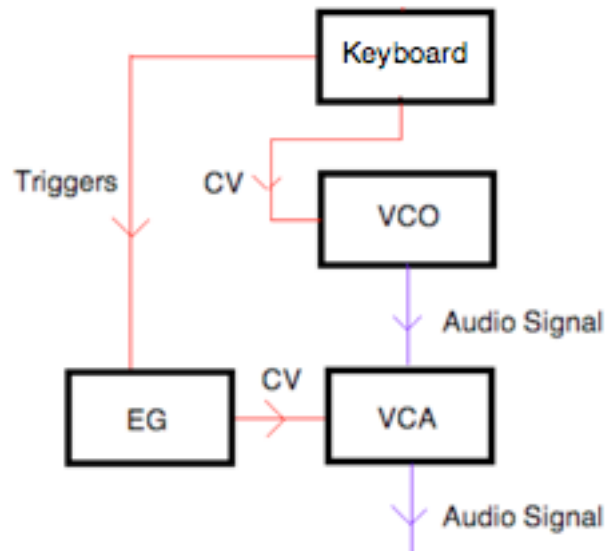


**Figure 3. A typical voltage envelope generated by the Envelope Generator.**

The Envelope Generator must receive signals from the keyboard to tell it when to generate the attack and release portion of the envelope. These signals are

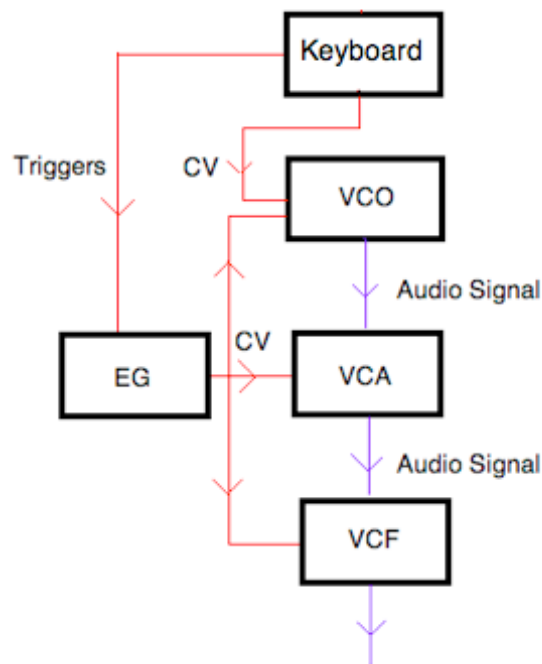


called the Gate and Trigger signals and are discussed in the section on the Envelope Generator circuit.



**Figure 4. An Envelope Generator affecting the amplitude of the audio signal via a Voltage-Controlled Amplifier.**

If a waveform contains harmonics then we can pass it through the Voltage Controlled Filter (VCF) to shape the harmonic content of the sound. The VCF will cut frequencies above or below some defined frequency. This frequency is defined by the CV input of the VCF. This frequency could change over time if we pass the Envelope Generator into the CV input of the Voltage Controlled Filter. The input of the Voltage Controlled Oscillator could also be affected in the same way.

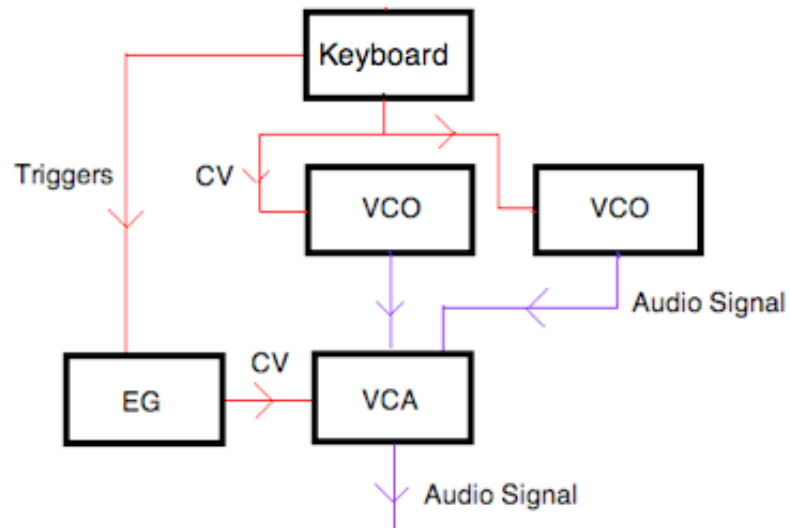


**Figure 5. The Envelope Generator can affect any of the modules by being patched into their Control Voltage inputs. Thus pitch, volume and harmonic content can all be varied over time.**

Note that the Voltage Controlled Oscillator is now receiving two control voltages. These voltages need to be added together. Each of the voltage controllable modules (oscillators, amplifiers and filter) is given two control voltage inputs which are added together to give the total modulation. There are two potentiometers on each module for controlling the amount of modulation being accepted from each control voltage input, with the exception of the oscillators which have one control voltage input which is variable and another which is fixed and calibrated to accept the control voltage from the keyboard. This setup allows one modulator to affect multiple modules by different amounts. This also eliminates the need for a volume control on the output of any of the modules while allowing for total control of all modulation.

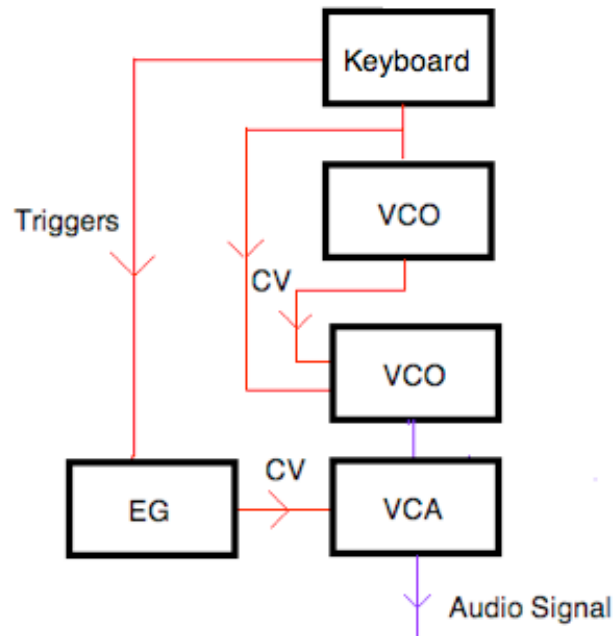
Note that these built in mixers are simply voltage adders and thus they can add audio signal voltages and control voltages indiscriminately.

The two oscillators could be controlled simultaneously and tuned to produce harmonies.



**Figure 6. Two Voltage-Controlled Oscillators being controlled simultaneously to produce additive synthesis.**

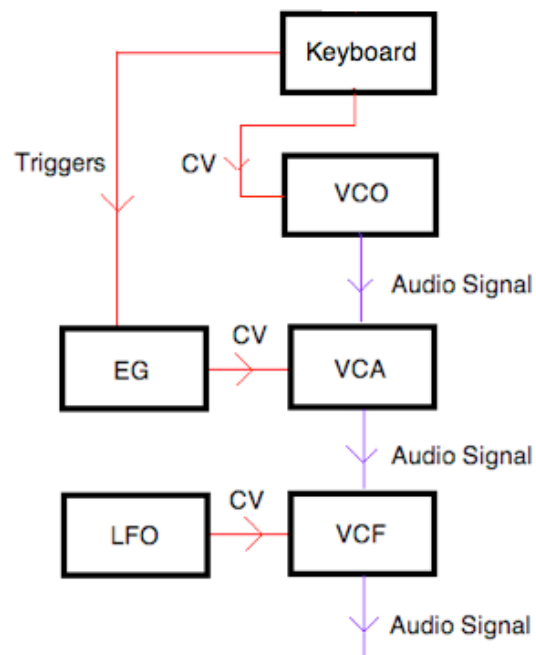
Or the second oscillator's output could be used as a control voltage to modulate another module:



**Figure 7. A Voltage-Controlled Oscillator being used to modulate the Frequency being produced by a second Voltage-Controlled Oscillator.**

Note that the output of the second VCO is acting as a control voltage and not as an audio signal. Used in this way an oscillator can modulate the control voltage of another oscillator to create Frequency Modulation (FM), or if it is affecting the VCA, Amplitude Modulation (AM). This keeps the modulating signal at a fixed ratio of the audio signal which keeps the timbre of the output signal the same no matter what pitch is produced.

If we want a fixed frequency of modulation, rather than one determined by which key we press, we can use a Low Frequency Oscillator (LFO), so called because it generally operates at frequencies below the audible range although this is not necessarily the case. This oscillator needs no control input as its frequency is set by its own onboard control.



**Figure 8. A Low Frequency Oscillator modulating the cutoff frequency of a Voltage-Controlled Filter at a fixed frequency.**

This outlines just some of the many ways the modules can be connected to generate sound using additive, frequency modulation, and amplitude modulation modes of synthesis. The next section looks at each module in detail and analyses their circuit diagrams.

## **Methods**

### **Operational Amplifiers**

Operational amplifiers, or op-amps for short, are integrated circuits which are extremely useful for processing voltages and currents in audio electronics applications.

Three configurations of op-amp circuits are used extensively in this project: voltage followers; inverting amplifiers; and non-inverting amplifiers. One or more of these op-amp circuits show up in all of the modules in this project. They are useful for connecting two sections of circuitry or two modules together. Often when a problem arose while building a circuit it was discovered that a voltage follower between two sections acted as a buffer and allowed the two sections to perform their tasks without interfering with each other. Impedance mismatching between sections can cause these sorts of problems. Op-amps can remedy these problems and hence their usefulness in audio electronics cannot be over emphasised.

Op-amps have a very high input impedance which means that when a voltage is applied to the input of the op-amp we can assume that no current flows into the input. The outcome of this is that a voltage can be processed without affecting the circuit that is creating the voltage. Op-amps also have a very low output impedance which means that they can supply as much current to the next section of the circuit as is required. This current is drawn from the power supply rather than from the previous section of the circuit, hence the previous section is not affected.

### **Modules**

The following circuits were prototyped on a breadboard and tested using a multimeter and oscilloscope. The circuits were taken from various sources and some were modified slightly in order to ensure the compatibility of the modules. Where circuits have been modified the original schematics are given in Appendix B.

## Low Frequency Oscillator

The Low Frequency Oscillator is the only module which is not voltage controlled. It produces a sine wave which can be used to modulate the parameters of the other modules. The frequency is changed via the rate control on the front panel. The switch on the front panel allows the user to select a low frequency range or a high. It should be noted that the amplitude of the output wave is approximately doubled when in low frequency mode. This is due to the capacitors having more time to build up charge when oscillations are slower. Since each module has a control for the amount of modulation accepted this is not a problem as the amplitude of the modulating wave can be reduced using this control.

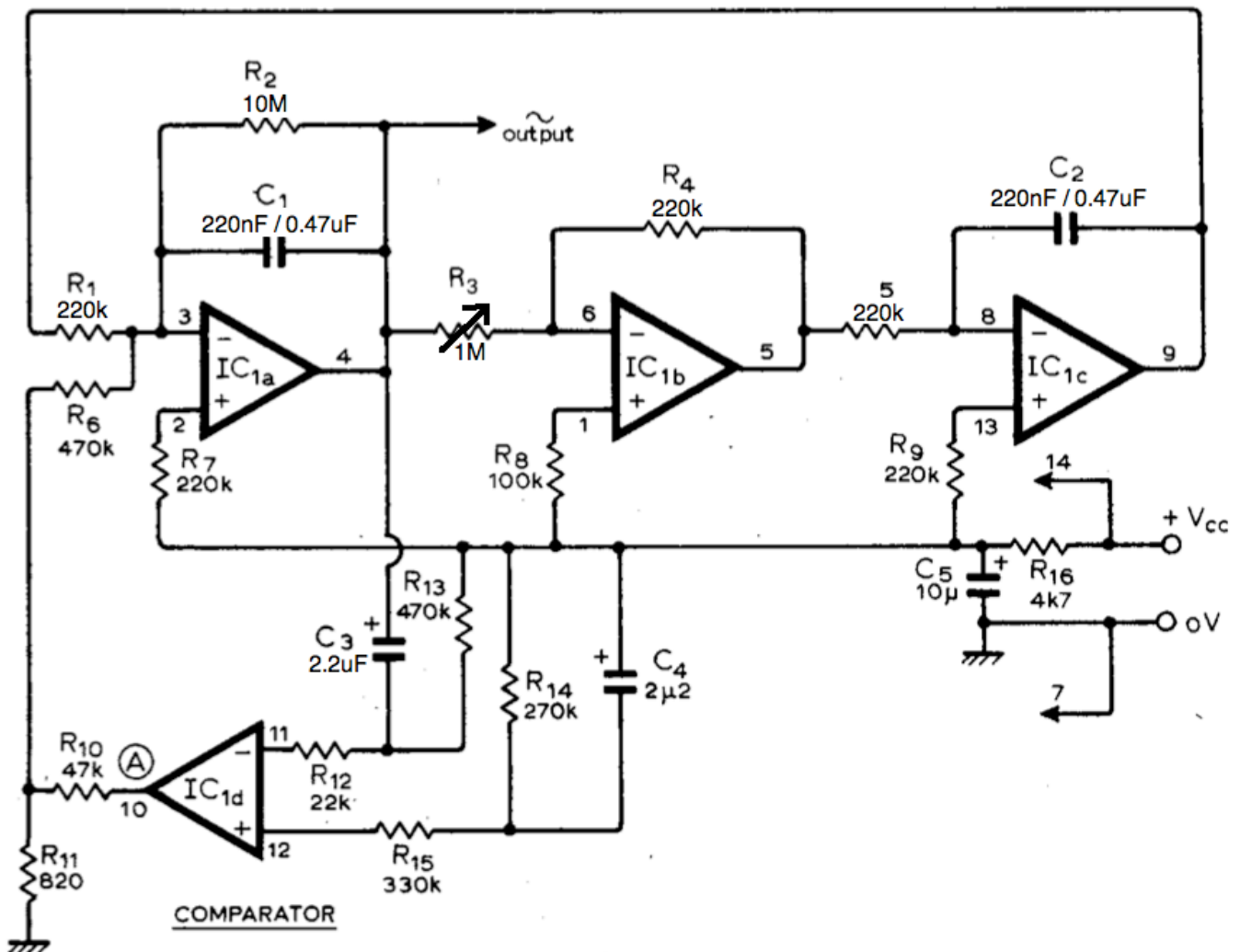


Figure 9. Low Frequency Oscillator Schematic. (Rossiter, n.d.)

When the circuit is powered up,  $C_4$  charges which causes the output of  $IC_4$  to switch from  $-V_{cc}$  to  $+V_{cc}$ . This shocks the band pass filter formed by the two integrators  $IC_1$  and  $IC_3$ , and the inverting amplifier  $IC_2$ . The comparator,  $IC_4$ , is in the feedback loop of  $IC_1$  and so sustains the oscillation of the whole circuit. The constant magnitude of  $IC_4$ 's output helps to keep the amplitude of the output wave constant. The frequency of oscillation is changed using the variable resistor  $R_3$ .

By using relatively large capacitors for  $C_1$  and  $C_2$  very low frequencies of oscillation are produced. A DPDT switch was used to switch the capacitors at  $C_1$  and  $C_2$  simultaneously in order to give even lower frequencies.

### Envelope Generator

Two envelope generators are included to allow separate envelopes for volume and timbre. When the keyboard is being used to control the synthesiser one of the envelope generators should be used to control the voltage-controlled amplifier module.

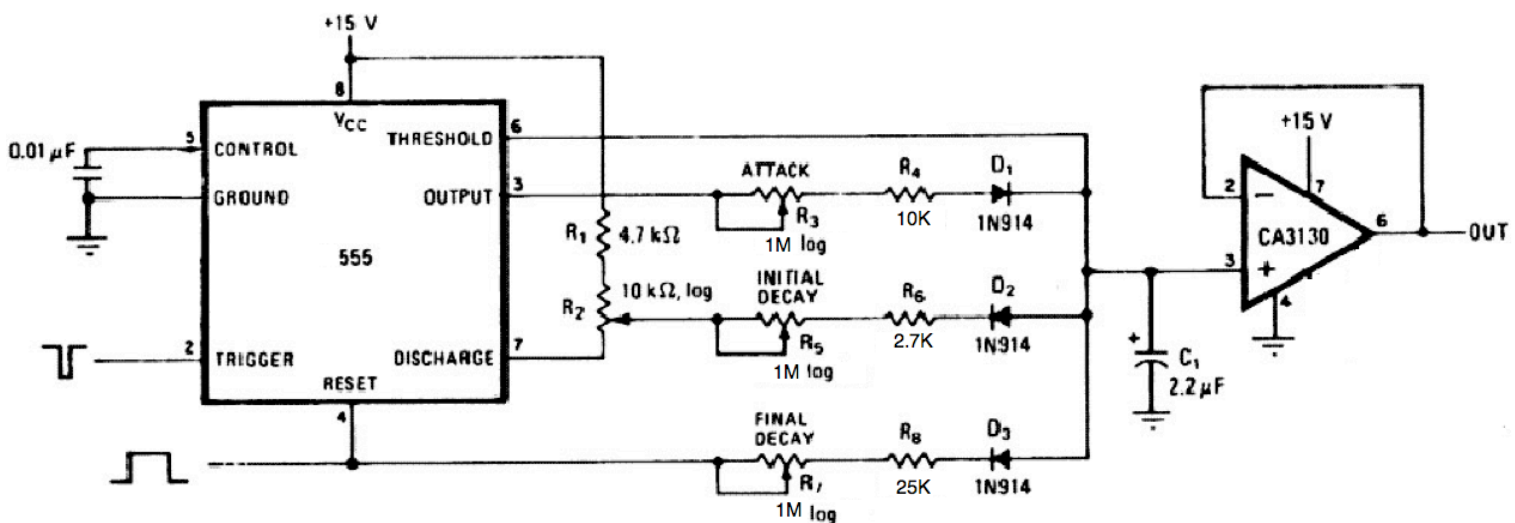


Figure 10. Envelope Generator Schematic. (Jacky 1980)



In order to understand this circuit we must understand the basics of how the 555 timer IC works. Here is its pinout and a short explanation of what each pin does:

- 1) **Ground.** Or in this case -Vcc
- 2) **Trigger.** A short low (less than  $1/3 V_{cc}$ ) pulse on the trigger starts the timer.
- 3) **Output.** During a timing interval, the output stays at +Vcc. Can source up to 200ma.
- 4) **Reset.** Forces pin 3 low if pulled to ground.
- 5) **Control.** Can be used to adjust threshold trigger voltage. Not used in this application so it is connected to ground with a .01uF cap to eliminate supply noise from Vcc.
- 6) **Threshold.** When threshold crosses above  $2/3 V_{cc}$  timing interval ends.
- 7) **Discharge.** connects to ground when output goes low.
- 8) **Vcc.** Power supply. Typical range 4.5v to 16v. (Allen, n.d.)

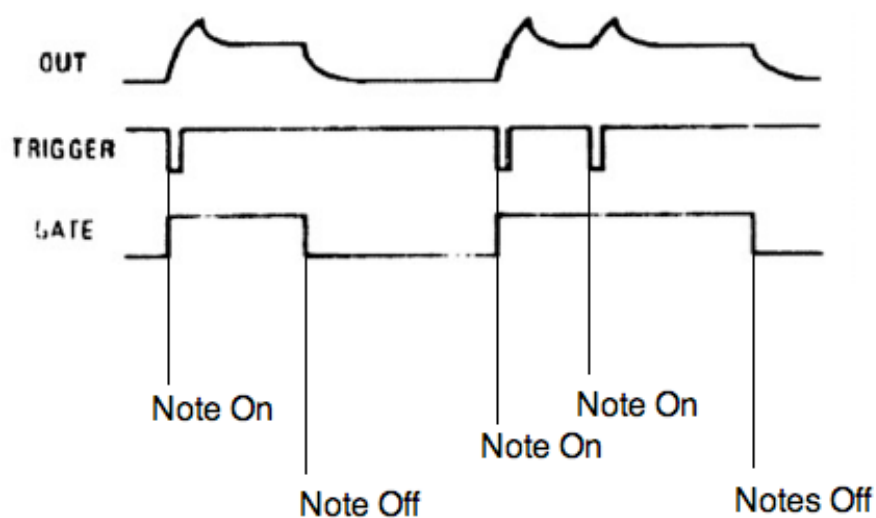


Figure 11. The necessary Gate and Trigger signals to produce the output envelope. (Jacky, 1980)

The capacitor  $C_1$  is charged and discharged through  $D_1$ ,  $D_2$ , and  $D_3$  according to the voltages which are present on pins 3, 7 and 4 of the 555 timer IC. The voltage stored on  $C_1$  is passed to the output through  $IC_2$ , an ordinary op-amp connected as a voltage follower. This allows the voltage to be read without discharging the capacitor.

When a key is held down on the keyboard the gate signal goes high and a low pulse is detected on pin 2 of the 555 timer IC the timing interval begins and pin 3 goes high. This charges  $C_1$  through  $D_1$ . The time taken to charge the capacitor is given by the time constant  $t = RC$ . So the length of the attack phase can be controlled with  $R_4$ .

Once the voltage on  $C_1$  reaches  $2/3 V_{cc}$ , pin 6 senses this and this causes pins 3 and 7 to go to ground. Now  $C_1$  discharges through  $D_2$  but keeps the voltage that is at the wiper of  $R_2$ . Hence  $R_2$  can set the sustain level. During this time another trigger pulse would cause another attack and decay process (although with the keyboard circuit this is not possible)

When the key is released the gate signal returns to ground and the remaining voltage on  $C_1$  escapes to ground through  $D_3$ .

The second Envelope Generator was given a larger decay potentiometer of  $10M\Omega$  to allow it to produce extremely long decays.

### **Voltage Controlled Oscillator**

The following circuit was used to build two oscillators. The circuit consists of an Integrator ( $IC_1$ , an op-amp), followed by a Schmitt trigger ( $IC_2$ , a 555 timer chip). The reset of the 555 is triggered by the voltage which is output by the integrator. The integrator capacitor,  $C_2$ , will charge at a rate which is proportional to the input voltage. Once  $C_2$  reaches 7.5V the timer resets which discharges the capacitor and resets the integrator also. The process repeats indefinitely and the frequency produced is proportional to the input voltage. Hence the voltage-controlled oscillator has a linear voltage to frequency relationship.

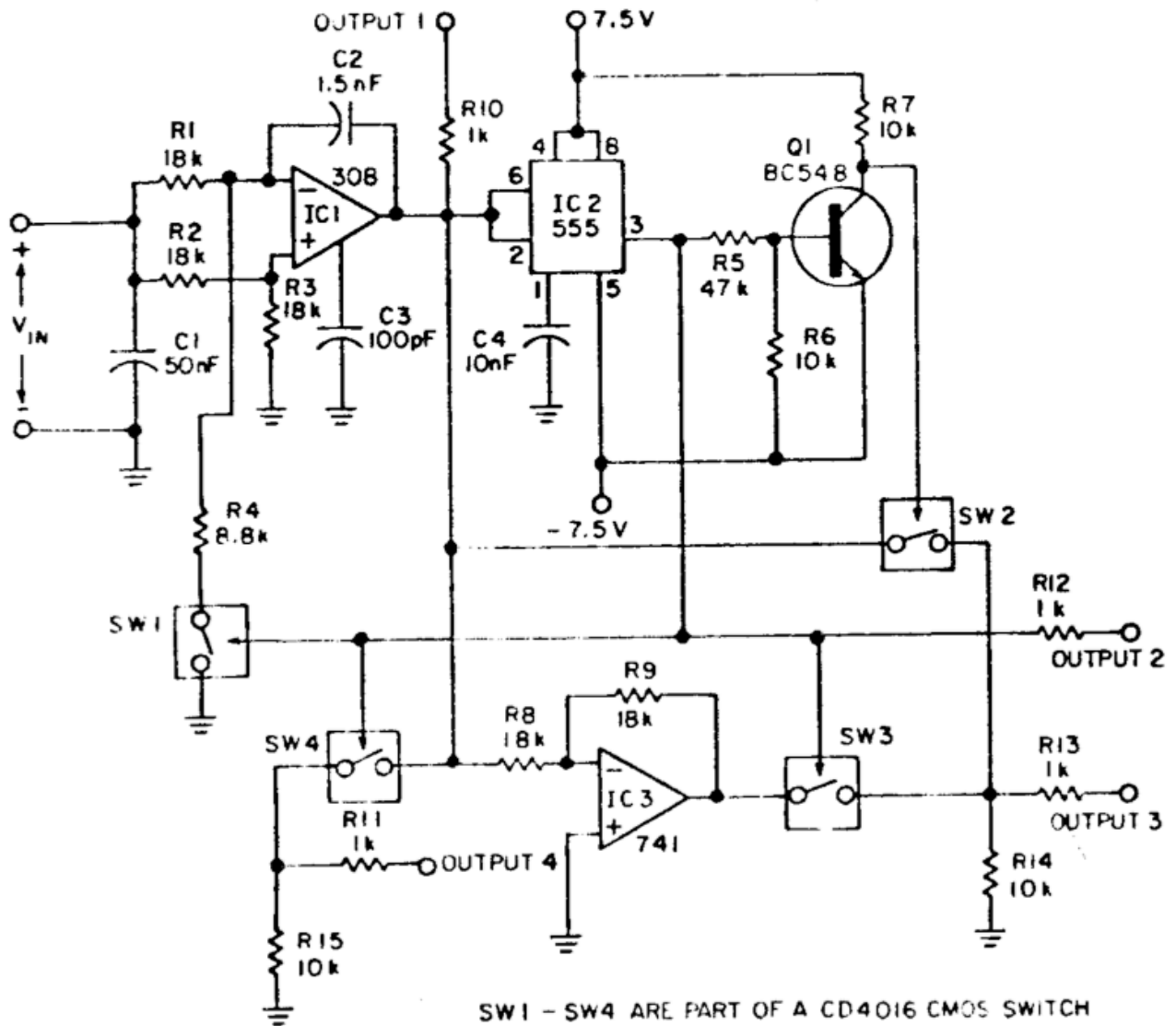


Figure 12. Linear Voltage Controlled Oscillator Schematic. (Malvar, 1980)

A quick test of the frequency response of the oscillator was carried out to aid in the design of the exponential converter which will ultimately drive the oscillator.

A 100kOhm linear potentiometer was used to vary  $V_{IN}$  which was measured using a digital multimeter. Free oscilloscope software *Fftscope* was used to determine the frequency of the output wave.

The following values of  $V_{IN}$  and the corresponding frequency,  $F$ , of the square wave at OUTPUT 1 were measured.

$V_{IN}$ (V)	F (Hz)
(+/- 0.01V)	(+/- 98Hz)
0.17	440
1.18	3185
2.16	6125
3.16	9016
4.16	11956
5.13	14602
6.33	17885

**Table 1. Frequency Response of the Voltage Controlled Oscillator.**

Note that although frequencies as low as 20Hz are desired from the oscillator, it was only tested down to 440Hz at this stage because it was difficult to get more data points below  $V_{IN} = 0.17V$  using a linear potentiometer. This highlights the importance of the exponential converter – that very small changes in control voltage affect the pitch produced by the oscillator greatly in the lower range.

CurveExpert was used to plot the response of the oscillator and determine its constant of proportionality.

The equation of the curve is

$$y = 2870x - 66 \tag{1}$$

So the response of the oscillator is 2870Hz per Volt. This is useful because a 7V input will give a frequency of 20,090Hz so using a +/- 7.5V power supply will allow the entire audible range of frequencies to be produced.

There are four octaves of interest below the range we can accurately reach using a potentiometer. Extrapolating from **Table 1** the voltages which are needed to control the oscillator down to the lower limit of human hearing are determined to be:

<b>F (Hz)</b>	<b>V<sub>in</sub> (mV) (projected)</b>
440	170
220	85
110	42.5
55	21.25
27.5	10.62
13.75	5.31

**Table 2. Projected input voltage required to produce frequencies down to the limit of human hearing for the voltage controlled oscillator.**

To reach this lower range we require an exponential converter that can convert the range 0 - 1V down to the range of 5 – 10mV millivolts.

### **Exponential converter**

Arguably this is the most important circuit in the whole project because it makes the oscillators respond in a musical way. By converting linear voltage changes into a multiplication of a basis voltage, constant changes in pitch are created. This turns the synthesiser from a sound-generating machine into a musical instrument.

The equation of a general exponential curve is

$$y = ae^{bx} \tag{2}$$

where  $a$  and  $b$  are constants of the equation and  $e$  is the exponential constant  $\approx 2.72$ .

In this case  $y$  is the output current and  $x$  is the input voltage.

With this in mind, the following circuit was built with the goal being that a 1V change of the input voltage should lead to a multiplication by 2 of the output voltage. This along with the linear response of the oscillator will lead to a voltage controllable oscillator with a 1 Volt per octave response.

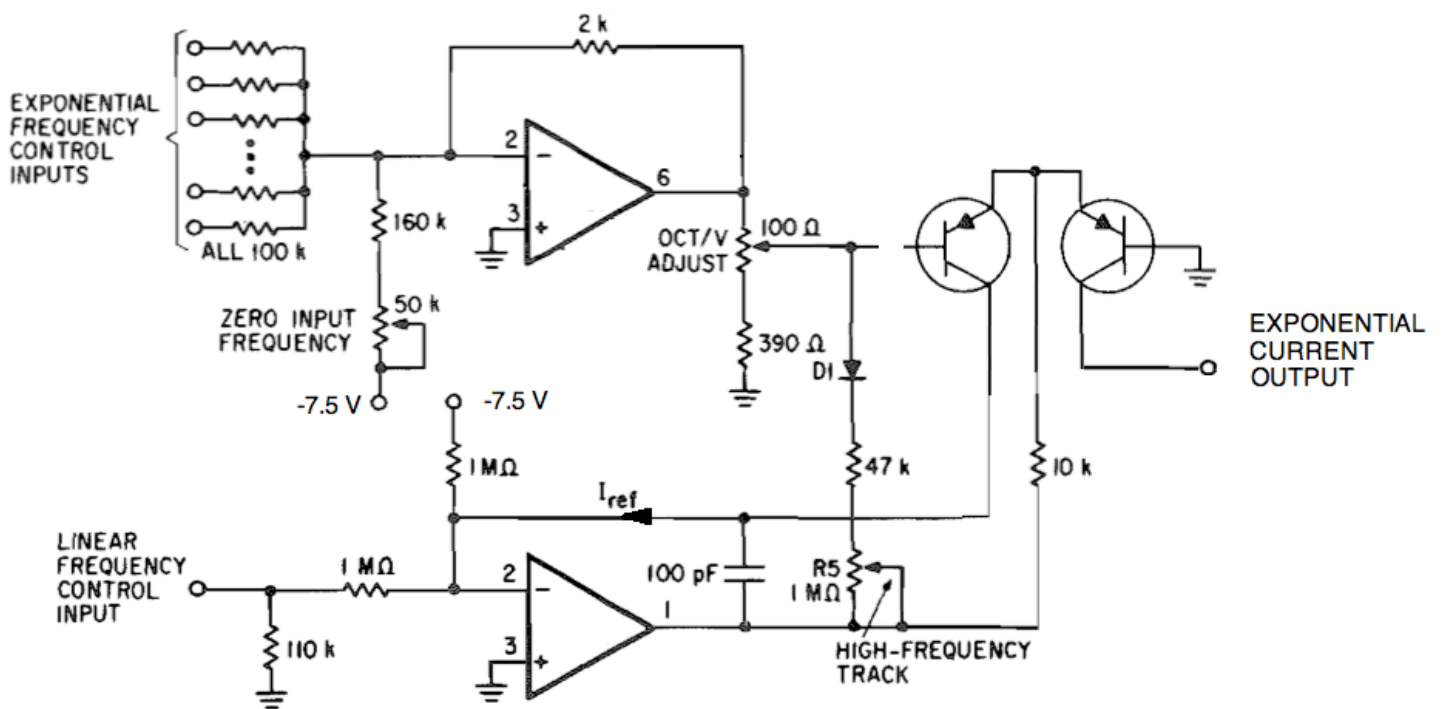


Figure 13. Exponential converter circuit. Adapted from Chamberlin(1987).

### Calibration of the Exponential Converter.

This circuit outputs a current but the oscillator requires a voltage to appear at its input. So the current output of the exponential converter must be converted to a voltage in the right range to produce a lowest frequency of around 20Hz. The 7V control voltage range will then give an upper limit of 2,560Hz. The exponential converter thus reduces the available frequency range of the oscillator to less than the audible range. This is acceptable as we are still left with seven octaves which is just less than the range of a full sized piano keyboard which contains the normal range of usable musical pitches (see appendix A).

The current output connected to a 100kΩ resistor was connected to ground and the input of the oscillator. The oscillator was connected to a small speaker so the pitch being produced could be heard. The **ZERO INPUT FREQUENCY** and **OCT/VOLT** controls were set by ear until a 1 Volt change in input lead to an increase of an octave in output pitch.

The values of input voltage and output current were measured and graphed using CurveExpert so that the constants  $a$  and  $b$  in Eqn 2 could be determined.

The constants of the equation were found to be  $a = 0.272$  and  $b = 0.687$ . This shows that the lowest possible output is  $0.272\mu\text{A}$ . This is the output when there is a  $0\text{V}$  input. This corresponds to the lowest key on the keyboard being pressed.

$$e^b = 2.72^{0.687} = 1.99 \approx 2 \quad (3)$$

So eqn. 2 becomes

$$y = 0.272 \times 2^x \quad (4)$$

This shows clearly that a  $1\text{V}$  increase in  $X$  leads to a doubling of  $Y$ , the output current. This is exactly the response that is needed:  $1$  volt per octave. The next step is to convert the output current of the exponential converter into a voltage that can be passed into the voltage-controlled oscillator to become a frequency. This is simple to implement using just four op-amps and a few resistors.

From **table 2**, we require the lowest possible control voltage to be approximately  $10\text{mV}$ . A current to voltage converter (figure.) with a resistor value of  $330\text{k}\Omega$  will convert the current output from less than a micro Amp to the required voltage range of tens of millivolts according to the equation

$$V = -I \times 330,000 \quad (5)$$

The current-voltage converter outputs a negative voltage for a positive input current so the voltage is passed on to a unity gain inverting-amplifier via a voltage follower to

correct its polarity. A final voltage follower buffers the circuit and a 10K resistor gives the output some impedance to match the input impedance of the voltage-controlled oscillator.

Now that the exponential current has been converted to a voltage in the correct range and polarity, we have a musical way to control the voltage-controlled oscillator.

### **Voltage Controller Filter**

The voltage-controlled filter shapes the harmonic content of a waveform. This filter has four modes of operation: **Low Pass, High Pass, Band Pass, and Band Reject**. The filter has two controls: **Cutoff Frequency (F)**, and **Quality Factor (Q)**.

In Low Pass and High Pass modes the cutoff frequency of the filter is the frequency at which the attenuation begins. For Band Pass and Band Reject (or Notch) modes the cutoff frequency is the frequency at the center of the band of frequencies which is passed or attenuated. A higher Quality Factor leads to a more harsh filter with greater amplification of frequencies just before the cutoff frequency in the case of Low Pass and High Pass modes, and a thinner band of frequencies in band pass and band reject modes.