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Electronic Sound Synthesizer

First of three articles describing the operation and construction of a modular system with
manual or electronic voltage control of synthesized waveforms

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The electronic sound synthesizer is an instrument that can generate a variety of complex outputs the parameters of which are variable and are controlled by the device itself. In its most common form, the synthesizer is used as an electronic musical instrument usually being a monophonic keyboard device. It is also to be found in more fixed purpose applications, such as animal "alarm call" generators. Basically, the synthesizer is capable of generating and processing signals, and by employing such techniques as frequency and amplitude modulation, filtering and mixing, it is usually possible to produce a desirable output. The feature that makes the synthesizer unique from other instruments, such as organs or electric pianos, is its voltage control capability. This enables parameters such as frequency, amplitude modulation, attack and reverberation, to be not only manually controlled, but also electronically controlled. Couple this voltage control capability to a flexible programming unit and the result is an instrument with an enormous range of possible tone colours. The versablity of the synthesizer can be further extended by the inclusion of more and more functional units, but this approach is over-sophisticated. It is better to try to analyse just what is required and how best to achieve it. For instance, what particular types of sounds should the synthesizer generate; is it for instance, going to be used as a piece of educational equipment or for quantitatively synthesizing known waveforms, for example bird calls, engine noises, spoken words etc? This is the "deep end" of synthesizer technology where a great deal of effort has been expended for few retums. Where reasonable returns have been achieved it has been, generally, with computer backup.

Sound synthesis

As a musical instrument the synthesizer is well cast. Tbe world of qualitative descriptions is an ideal environment for a machine that continually defies a quantitative approach. The synthesizer is often used to generate special effects and can also be used to produce pseudo-instrumental sounds via keyboard control, or by modifying real instrument sounds. To synthesize implies the process of generating a result by the summation of many parts, and a musical synthesizer should produce a musical output by the summing of a group of semi-musical elements. Musical instruments produce sounds that have a discernible harmonic structure, the perceived sounds being the result of exciting a resonant structure by percussion, bowing, plucking or blowing. The envelope of the signal is modified by various sorts of damping and excitation, and the pitch of the fundamental is either pre-selectable or in some cases continuously variable. To make an electronic synthesis of a "pseudo-instrument", a selection of resonators (oscillators) is required. These resonators should have a variable multi-pitch control (voltage controllable) with a large dynamic range and possibly a selection of different harmonic structures (sinewave, square, ramp, etc which have different harmonics; pure tones only have a limited use). Three or four of these resonators can be considered as a basic minimum for any sort of modest synthesizer arrangement. The signal amplitude from the resonators must be controllable and so a means of control (a voltage controlled amplifier, the gain varying with respect to a control voltage) and a source of control (voltage control sources such as other oscillators, joystick, keyboards, potentiometers, waveform generators etc) must be provided. Also, a means is necessary of bringing these units together so that they interact (the patch panel and the voltage summing networks). When a rapid series of randomly distributed percussions is initiated (for

instance, brush drums) the pitch information is low. This group of "pitchless" sounds is characterized by the lack of a significant harmonic structure and can be synthesized by modifying the amplitude and spectrum of a noise source. When a musical instrument is played an amount of reverberation is always introduced, thus a means of adding a controlled amount of reverberation is provided.

The synthesizer is operated to its best advantage using a set of keyboards. However, no dynamic function - i.e. a means of generating a louder note the harder the key is pressed - has been provided as in some other synthesizers. To simulate a percussion envelope, waveform generator having a variable exponential attack and decay has been included. Other circuit functions are included (described later) and these combine
with those units already mentioned to produce a system that is capable of generating a very large range of special effects.

The total collection of units was chosen after monitoring the format of commercially available synthesizers. Such items as oscillators, voltage controlled amplifiers, noise sources, mixer reverberation, patch panel, keyboard, voltage controlled filter, and waveform generator are common to most devices but unusual items included are a joystick, summer/inverter, exponential transfer function, and a very low frequency noise source. These units extend the range of special effects that can be generated.

Items that appear in other synthesizers, but which had to be left out due to time, space and money limitations are: the internal amplifer, loudspeaker, an input preamplifer for microphone and pickups (these provide some excellent electronic effects), envelope followers that try to mimic instruments and voices), electronic two-way switches and a programmable memory.

Faced with all the possible combinations of units, the newcomer to sound synthesis will probably be somewhat at a loss to make any decisions as to what units are needed to meet his requirements. Firstly, the system is going to need a power supply. If the synthesizer is likely to be built in modules, which are added when time and money permit it is advisable to allow a more than suffcient power supply capability to enable an unhindered growth.

A current-limited supply would be an improvement over the one given later in this series. The amplifier loudspeaker combination and the patch panel are also essential. The heart of the synthesizer is its oscillators; they generate nearly all of the sound that is produced.

The next most important are the voltage-controlled amplifiers. These are reasonable quality devices, but a cheap f.e.t. modulator could be used if money is tight. Such parameters as linearity and harmonic distorrion will suffer from this particuIar economy. It now becomes more difficult to decide which particular units are most important, so they have been grouped together; the audio mixer, noise sources (coloured), voltage controlled filter, reverberation, waveform generator and keyboards. Lastly, probably the low priority units are the joystick, sample and hold, exponential transfer function, summer/inverter, white and very low frequency noise sources. Even though these last units have the lowest priority, they add considerably to the synthesizer's versatility. As a guide to cost, the synthesizer described in this article was produced for approximately £100. The performance of the machine, as with other synthesizers, is not sufficient for it to be a main instrument for live performances, due mainly to speed consideratons in setting up patches and pots. The only way to obtain a versatile performance entirely from the synthesizer is to use multi-track recording techniques.

The system

The synthesizer may be considered as a series of separate units, each with their own respective sub-groupings (see Fig. 1).

Voltage controlled units

This is probably the most important set of units, for it is these devices that have their parameters controlled by external electrical signals. Voltage controlled oscillators. Each oscillator's fundamental frequency is controlled by the sum of the input control voltages and a bias voltage, there being a fixed relationship between the voltage and frequency. From three oscillators, several waveforms are simultaneously available, these being sinusoidal, square, triangular, sawtooth, variable mark/space ratio, pulse and a sequential signal. The operating ranges extend down to frequencies of a fraction of 1Hz and to requencies above the audo range. These oscillators perform

all the frequency modulation functions of the synthesizer.
Voltage controlled amplifiers. The gain of the unit is linearly controlled by the sum of the input control voltages and a bias voltage. There are two v.c.a:s and these provide all of the amplitude modulation capacity. Voltage controlled filter. This unit is a bandpass filter, the value of the resonant frequency being linearly proportional to the sum of the input control voltages and a bias voltage. The Q factor is manually adjustable and increases linearly with frequency.

Signal processors

The voltage controlled units require input control signals and produce either control or audio signals at their outputs. Note that the distinction between control and audio signals is not absolute, but as a lineralization, control signals exist from d.c. up to the low frequency end of the audio spectrum. There is no physical reason against control signals extending to high frequencies, except that the effect is rarely a pleasant one! By processing audio and control signals, the range of outputs is considerably enlarged.
Audio mixer and reverberation unit. These two processors are only compatible with audio signals as they are both a.c. coupled. The mixer has three channels, each channel having its own attenuator, and there is also a master gain control. The reverberation unit also has a gain control and provides a source of reverberation up to approximately 4kHz.
Summer/inverter and exponential transfer function. These devices were designed essentially for control signals, but audio signals may also be used. Two of each are used in the synthesizer.
The summer/inverter has three inputs, two with a gain of - 1, one with a gain of - 10.
Sample and hold. This is the only form of analogue memory provided. Sampling is initiated by a positive input pulse that causes the unit to sample the analogue signal for a preset time. This signal is then held for an unspecified period.

Noise sources

Three different outputs are simultaneously available. The noise may be used as a control signal or as an audio signal.
White noise. The noise source provides on average a continuous flat spectrum (within certain limits and tolerances).
Coloured noise source. The output noise spectrum is arbitrarily variable and is controlled by a conventional tone control network.
Very low frequency noise source. One of two v.l.f outputs may be selected, the signal's function being a random control voltage.

Control voltage sources

The units of this group generate control voltages, and provide the main active link between the operator and the synthesizer.
Joy stick control. Two bias voltages are produced, one associated with each degree of freedom of the device. By physically moving the joystick, the bias voltages change, the modified signals being linearly proportional to the stick's position.

Waveform generator. A "rectangular" waveform with an exponential attack and decay is generated, the process being initiated by a manual or electronic signal. The attack and decay time constant, and the duration are all arbitrarily variable.
Key boards. A standard four-octave key-board is used to generate a d.c. control voltage, which is linearly proportional to the key position. As the synthesizer is essentially a monophonic instrument, then only one key may be pressed at a time. If two or more are pressed simultaneously, the highest note is automatically selected. Also a pulse is produced at the start of each new note.
Three other units must be introduced to complete the total system. The first is the patch panel which enables the rapid interconnection of units into any desired configuration.
Secondly, an external amplifier and loudspeaker is required. The third requirement is an external feedback system with pattern recognition facilities and a versatile complement of servo systems - an operator. The selection of units may be varied to suit one's particular requirements.

Design in general

There are certain rules that have to be enforced if the synthesizer is to work satisfactorily.
Firstly, it is essential to generate and measure all signals relative to OV, and this requires a reliable grounding system. A stack of star terminals was employed for this, to which were connected the ground wires from the control pots and all the OV supply lines from the edge connectors.
A signal level of 3V was selected, this giving ample room for larger signal excursions. Also as there is a considerable amount of wiring between the pots circuits and patch panel, the input and output impedance of the units was kept low so that unscreened wiring could be used without any serious interference or crosstalk problems occurring. The input impedances are typically lkOhm and the output impedances must be correspondingly lower to avoid loading. Some control signals are low frequency or even direct voltages and so a.c. coupling between units is not a practical proposition (with the exception of the audio mixer and the reverberation unit). The most significant problem with direct coupling is the fact that control signals are never what they ought to be, but always have an offset voltage added to them. Most of these offset voltages are
only a few hundred millivolts (positive), but this is enough to cause disturbing effects. However, the variable bias on the voltage controlled units should be capable of overcoming most offsets.
The general layout of the synthesizer can be seen in the photograph. Most of the circuitry was constructed on plugin boards and although the connectors increase the cost, they do provide the advantage of making the boards removeable for servicing. Also a spacious layout has been used, enabling clear access to the control pots. Even with a stablized supply and a reasonable ground system it may prove necessary to decouple the power supply on each board. Minor transients of the supply levels can be disturbing as they can build up into a noticeable background noise, and may even cause the v.c.os to lock on to each other's harmonics. The synthesizer bears a strong resemblance to an analogue computer, with an array of control pots to vary parameters, a patching system and a selection of functional electronic units. However, whereas the analogue computer makes an attempt at being quantitative and accurate, this synthesizer does not, relying strongly on the qualitative perception of the operator.

First voltage controlled oscillator

This oscillator has a linear frequency/voltage characteristic and produces four outputs as shown in Fig. 2. These are square, triangular, sinusoidal and a variable mark/space ratio rectangular waveform. The oscillator has three frequency ranges, the top range covering the audio spectrum, the bottom two extending to subsonic frequencies. The quiescent operating point may be shifted by altering the bias level, and the input control voltages (VC1, VC2) may be attenuated by control pots. The final operating frequency is linearly proportional to the sum of the bias voltage and the attenuated control voltages, and should have a dynamic range of at least three decades.
The heart of the oscillator is a triangle-squarewave generator (Fig. 3) where a Schmitttrigger provides positive feedback around an integrator; the integrator's output thus ramps up and down inside the hysteresis window of the Schmitt trigger. The oscillator is both self starting and stable, having a large dynamic operating range and a defined amplitude. Two outputs are produced, a triangle at the integrator's output and a square wave from the Schmitt trigger. The ramp rate, and hence the operating frequencv, may be varied by altering either the ntegrator's gain and/or the drive voltage.
The two voltages V and V (Fig. 3) are alternately switched into the integrator by the electronic switch (a diode ring switch D7, 8, 9,10, Fig 4), which is controlled by the Schmitt trigger. The voltage V is produced at the output of IC3, where the output is depressed by the forward drop across diode D6. Ideally D6-10 should all be matched and so should resistors R21, 24, 36 and R22, 23, thus preserving as far as possible the linear voltage/frequency characteristic and signal symmetry. However, as matched diodes are relatively expensive, it was decided to use unmatched unselected diodes. This had the effect of causing some nonlinearities which were only noticeable at low frequencies where the diodes were conducting very low currents. To obtain the required gain from IC3, resistor R36 had to be much larger than R21,24, and this resulted in a loss of voltage/frequency linearity at low freqnencies. This effect is

not very noticeable, but imbalance in the ring switch may cause a disturbing loss of symmetry (Fig. 7). This can be nulled by preset R2 (Fig. 4) which is set to cancel the offset caused by the ring switch's imbalance at its minimum operating point. To preserve as much symmetry as possible, R21-24 are all 2% tolerance resistors.
Diode D3 (Fig. 4) is included to protect Tr1, TrZ, against emitter-base breakdown; if for any reason the feedback loop is broken, the output of IC1, may ramp down unhindered, with irreversible results. The Schmitt trigger used is the SN7413N, a ttl integrated circuit.
The whole of the circuit operation relies upon the stability of the hysteresis levels; if they alter, then the amplitude and frequency of the output will change. Thus it is particularly essential to have a stabilized and decoupled 5V supply for IC2 as well as for Vcc. If this is not achieved then spikes on the power supplies will cause oscillators VCO1 and VCO2 to have a tendency to lock onto one another's harmonics. To reduce the generation of spikes, the output of the Schmitt trigger is capacitively loaded; this however, has little effect on the square wave production at audio frequencies.
It should be pointed out that using the SN7413N for the Schmitt trigger has its drawbacks. The separation between its hysteresis levels is small, making it vulnerable to interference by other v.c.os. Its fast rise and fall times can generate significant interference and also it does not like driving long lengths of cable. These difficulties have been largely overcome, but a Schmitt trigger of discrete components would still be an improve-
ment. Also, delays in the loop cause some unwanted amplitude modulation. This becomes apparent at frequencies above 10kHz, but the change in amplitude and harmonic content (in the case of the piecewise generated sinewave) is not obvious to the observer. The sinewave output is generated by feeding the triangular wave at the output of IC1, (Fig. 4) into a diode function generator (Fig. 5). Thus, by adjusting the bias, R2, and the gain, R3 a sinewave can be produced as shown in Fig. 8.
The mark/space signal is produced by driving the circuit shown in Fig. 6 with the "triangle" waveform. Transistor Tr1 forms a level sensitive switch, and R4 effectively shifts the d.c. level of the input sigal. The resultant mark/space output is buffered by Tr2. Preset R3, is adjusted so that Tr1, comes on just at the peaks of the input drive with the wiper of R4, set at - Vcc. This should provide a mark/space range from about 15 to 85%.
To set up VCO1 select the highest frequency range, disconnect any inputs, set the bias to mid position and set R2 and R32 (both as in Fig. 4) to mid position. Monitor the triangle output and switch on. Tum the bias level down to zero and if the oscillations stop increase R32 until they start again. If the oscillations become badly asymmetric just before stopping, compensate by adjusting the offset control R2. Thus by adjusting R2 and R32, optimize the balance between minimum operating frequency and symmetry. Having done this, increase the bias pot setting to give an output frequency of about 1kHz. The triangular wave should now be symmetrical and the diode function generator and mark/space generator presets can now be aligned.

Second voltage controlled oscillator

This oscillator is similar to VCO1. It produces sine, square and triangular waveforms as before and also pulse and ramp waveforms (Fig. 9). The heart of the oscillator is basically the same as shown in Fig. 4, except that four frequency ranges are employed (see Fig. 10), thus giving an extended low frequency range. The sinewave generator is the same as before (Fig. 5), but two new generators, a pulse and a ramp generator are provided (Fig. 11).
The pulse generator is a monostable; it is triggered on the positive edge of thesquare-wave output and produces a pulse of approximately 20us duration (Fig. 12).
The ramp generator is a differential amplifer with a switched gain (Fig. 13). The square-wave is used to control switching transistor Tr1 , so that the differential amplifier has an alternately positive and then negative gain. As the triangle and square-wave are always phase locked, output of the differential amplifier is a ramp. As the triangular wave will have a d.c. offset voltage associated with it, a step will be produced in the middle of the ramp, but this can be zeroed by cancelling out the offset. For this purpose, preset R11 in Fig.13 has been provided. There will, however, be some distortion generated at the crossover point which cannot be removed, but this is relatively small.
In the article by R. A. Moog', the v.c.o. described takes a different approach to the waveform synthesis. It first generates a ramp using a current-driven unijunction

relaxation oscillator, and then converts this ramp into a triangle. This type of
v.c.o. has a smaller dynamic range than VC01,2, but has a much higher immunity to
locking onto harmonics of other oscillators.
The series will be continued with details of a sweep frequency oscillator, VCO3,
voltage controlled amplifiers and filters, mixer and summer/inverter, sample and hold
and noise sources. The final part describes the joystick control, waveform
generator, keyboards, patch panel and power supply.

References
1. Moog R. A., "Voltage Controlled Electronic Music Modules",
Journal of the Audio Engineering Society , July 1965.
2. Kindlmann and Fuge. "Sound Synthesis",
IEEE Transactions on Audio and Electroacoustics, Dec. 1968.

Electronic Sound Synthesizer: Part 2   (Wireless World September 1973)

Continuing the construction with descriptions of voltage control circuitry,
reverberation and the exponental converter

The first part of this series of constructional articles (August issue) described the
philosophy behind the design of the synthesizer and its capabilities as a musical or
educational instrument. The series continues with constructional detail of the
circuitry. Each basic modular unit is described in full, but the number of units
employed can be varied to suit the constructor´s needs.

Sweep frequency oscillator

By driving VCO1 with a ramp, generated by VCO2 (both described last month), it is
possible to produce a sweep frequency oscillator capable of covering the entire audio
spectrum in one sweep (Fig. 14). If the swept sinewave output of VC01 is then fed into
a network, the amplitude-frequency response of that network can be rapidly determined.
A three decade sweep is available and the peak to peak amplitude is virtually constant.
However, the sinewave generated by VC01 is by no means pure, having a harmonic content
of between 3 and 4%. This limits the resolution to a rather modest value, but even so,
a reasonable representation of the network's frequency response can be obtained. (It is
particularly useful for directly observing the effect of tone controls in audio
amplifiers.) To display the amplitube-frequency response, the ramp drives both the
oscillator VCO1, and an oscilloscope (in the x-direction),whilst the network response
is displayed in the y-direction. The drive need not be a ramp; in fact any continuous
function could be used.

Voltage controlled oscillator, VCO3

This oscillator produces a sequence of steps, the amplitude of the steps being
individually controllable. The number of steps in the sequence can be varied up to a
maximum of six (Fig. 15) and a series of pulses is also available (1 to 1 mark/space
ratio) each being coincident with the leading edge of each step. The oscillator is
voltage controlled and has a pair of summing inputs.
The frequency-voltage relationship is exponential and extends from subsonic frequencies
to above 20kHz, all in one range. The oscillator, which consists of a voltage
controlled astable driving a binary counter, is shown in block diagram form in Fig. 16.
The b.c.d. output is decoded into decimal form, attenuated by pots 1 to 6 and then
summed. The length of the resulting sequential output can be modified by selectively
resetting the binary counter. The circuit diagram of VCO3 is given in Fig.17.
Transistors Tr2 and Tr3 are voltage driven and provide a current drive to the astable.
The result is an exponential current-voltage relationship and an exponential
frequency-voltage response.
The useful range of the control voltage applied to Tr2,3 is a few hundred millivolts,
and must be generated relative to +Vcc. Preset R7 is adjusted so that the clock
frequency produced is approximately 2OkHz with the bias set at maximum. Also, when the
bias is set to a minimum, preset R10 is adjusted so that the clock frequency is
approximately 0.2Hz. Unfortunately the effects of R7 and R10 are interdependent and
hence they must be set up iteratively until convergence is achieved on the desired
settings.
The logic section is self explanatory ; however, the logic power supply must be

decoupled with a 0.1uF capacitor. Note that this oscillator does not lend itself to construction on plug-in boards, as the edge connections required are exceptionally large.

Voltage controlled amplifier, VCA1,2

The heart of this unit which performs the function of amplitude modulation is a linear four quadrant multiplier, the device being an integrated circuit SG1495D. This device operates from a + 15V supply, and when used in the circuit shown in Fig.18 can accept inputs of +/-5V. The frequency response is greater than that required. The output is taken between two load resistors and a differential amplifier (IC2) is required to remove the common mode signal. Design of the multiplier and differential amplifier is very nearly the same as that given in the applications sheet for the SG1495D but some component values have been modified and lower tolerances are used. A scale factor of 0. 1 is employed.

The multiplier accepts two inputs, X and Y, and generates an output that is linearly proportional to the product XY. The X input is the audio signal VA1, and the Y input is the output of the control cinuit. This circuit is a voltage summer with inputs of VC1, VC2 and a bias voltage. This arrangement is that of a "perfect" halfwave rectifier, thus, when the sum of the control and bias signals goes negative, the output of IC3 remains at OV. Only when the sum is positive will an output (a control voltage9 be produced. In this way, the control section has a threshold characteristic, this threshold occurring at OV and being used as the reference level for zero output from the v.c.a. When long leads an used, parasitic oscillation may occur, but this can be suppressed by an RC network (C1R2).

Aligning the v.c.as

Four presets (Fig. 18) have to be aligned ; these are R20, R21, R14, and R8. The first two are Y and X "offset adjust", the third is "gain" and the last is "output offset". Set X and Y to 0V making sure that the bias (R25) is set at its most negative setting. Monitor the output of IC2 and adjust the output offset (R8) until it is at zero potential.

Set X to + 5V, but keep Y at 0V, and adjust R20, the Y "offset adjust" until the output is again at zero potential.

Set Y to +5V, set X to 0V, and adjust R21, the X "offset adjust", until the output is once more at zero potential.

Now repeat the first step. The last preset to be adjusted is R14, the gain control, which alters the scale factor of the generated output. The unit is now a functional v.c.a., and some amplitude modulation can be demonstrated. Also the audible and visible effect of varying the X and Y "offset adjust" can be observed.

Let VA1 be a 1kHz sine wave and VC1 be a l00Hz triangular wave. By varying the bias control the product can be made to rise or fall above the reference level horizon (Fig. 19a). If the sinewave and triangle are produced by VCO1 and VCO2, then the output will also be onesided as both of these signals are one-sided (Fig. 19b). If however, the signal VA1 is alternating, then a double sided output will be produced.

The effects of misalignment of the X and Y "offset adjust" can now be observed. For the Y "offset addust" the result is that the output is non-zero when the control voltage reaches the threshold level. The output may never reach zero or may even pass through zero and become inverted (Fig. 19c). The dynamic range of the v.c.a. is thus severely limited by errors in the setting of the Y "offset adjust" . Misalignment of the X "offset adjust", results in the unwanted components of VC1, VC2 and the bias appearing at the output, this being particularly disturbing when VA1, is zero.

Voltage controlled filter

The v.c.f. is a bandpass filter with presettable Q factor and a variable (and voltage controllable) centre frequency. The circuit diagram of the v.c.f. is given in Fig. 20. The use of a multiplier makes the circuit appear rather complex; an alterative approach is to use f.e.t. modulators. The result would be a reduction in circuit complexity and cost, paid for at the expense of increased distortion and a loss of linearity of the centre frequency with respect to the control voltage.

The circuit operation is as follows. The sum of the bias and the input control voltage is squared and used to drive the multiplier. The transfer function of the square law generator is given in Fig.21, transistor Tr2 being designed to saturate when its

collector reaches a potential of - 5V.
The multiplier is similar to VCA1 and VCA2, but in this case forms part of the main
loop with the two integrators. Integrator gains of 3.3 x 10E4 are used and the first
integrator is limited by zener diodes D10 and D11. If the output swing of this
integrator were allowed its full range of movement the maximum input of the multiplier
would be exceeded, causing the loop to become unstable and then "hung-up". The Q factor
can be modified by adjusting R22 and theoretically when the wiper of this pot is at 0V
there is no damping term, and the loop becomes unstable. This situation may or may not
occur it being dependent on component imperfections and the way in which the circuit is
constructed.
It was decided that the filter should not become oscillatory at high Q factor settingd.
Thus to eliminate this possible state, R23 has been included, the value of which is
chosen to make the loop non-oscillatory throughout its ranges. The v.c.f can be
modified externally so that it forms a low distonion oscillator, but this will be
decscribed later. Other synthesizers tend to prefer the use of low pass v.c.fs, and
this is easily achieved by taking the output from IC4 the low pass output. In fact
there is no reason why the bandpass and the low-pass
outputs should not be simultaneously available.

Filter alignment

Careful adjustments of presets R42, R44, and R35 are required, because the multiplier
is part of a loop that can easily become unstable, particularly at low values of gain.
 Break link A, Fig. 20, and align the multiplier using the method described for VCA1
and VCA2. Note that there is no "gain preset" to adjust. Replace link A andmonitor the
output of IC5. 'This potential may need to be zeroed by adjusting R35, this offset
being particularly large when the bias control is set at minimum. If the multiplier has
been carefully aligned to give a maximum dynamic range, then a range in centre
frequency of about one octave can be expected. Note that the multiplier's gain should
not quite reach zero or (even worse) pass through it and hence change sign, as both of
these states are unstable.
 The v.c.f. is now functional and can be used to perform a variety of operations. The
more common uses are to filter inputs such as white noise, producing various effects
similar to wind, rain, jet engines etc.
 The v.c.f can be used as a low distortion oscillator (Fig. 22a). The Q factor and the
bias are set at maximum and the control pot R18 (Fig. 20) is increased so that the
filter just oscillates. The v.c.f. can now be swept throughout its range, producing a
sinewave of virtually constant amplitude and of variable frequency (a 9.5 to 1 range
was obtained). There is a 90-degree phase difference between thee low-pass and
band-pass output, sine and cosine waveforms being simultaneously available.
The v.c.f. can also be used as a notch filter (Fig. 22b). To set up the mixer, use a
sine wave input and set the v.c.f. to the same frequency, with the Q factor at maximum.
By varying the pots VA1 och VA2, the input and the bandpass output can be made to
cancel out. The "notch" can then be swept throughout the v.c.f's range. This technique
can be used to examine the harmonic content of any signal that lies within the
frequency range of the v.c.f.
 The v.c.f is also useable as a spectrum analyser, measuring energy per hertz versus
frequency (Fig. 22c). However, this is no more than a demonstrarion piece, due to the
narrow range. Also, the sweep time must be rather long if a "high" resolution (i.e.
high Q factor) is required, and the constant bandwidth makes interpretation of results
difficult.

Audio mixer and summer/inverter

The audio mixer is a three channel virtual-earth mixer, each channel having its own
attenuator and being a.c. coupled. A master volume control determines the overall
signal level at the output (Fig. 23).
Also, two direct coupled virtual-earth mixers are provided (Fig. 24). These both have
three inputs, two having a fixed gain of - 1 and one of - 10, and are used for signal
processing, such as inversion, summing or amplification.

Reverberation unit

The reverberation unit consists basically of three sections; the driver, the springline
reverberation unit and the equalised pickup amplifier (Fig. 25). The springline

reverberation unit used was the "Hi242" obtainable from Henry's Radio. This unit is moderately inexpensive, but suffers from a loss of high frequency reverberation, dropping considerably at about 4kHz. However, a useful response can be obtained - enough in fact for this unit to be used in one of the commercially available synthesizers.

By operating switch "S" a choice of the input signal plus reverberation, or just reverberation is available. Thus the reverberation can be separately controlled, by using a v.c.a. and/or a v.c.f., as well as being manually controllable (R3). To reduce any microphonic effects, the HR42 unit should be mounted on a pair of rubber pillars.

Exponential converter

This unit has an exponential transfer function of the form, $V_{out} = \exp(V_{in} + constant)$
The base-emitter junction of transistor Tr3, Fig. 26, is voltage driven, the collector current being monitored. The relationship between Vbe and Ic is very nearly exponential, modified by the fact that the voltage drive is imperfect and the value of Vce (Tr3) is changing. The suitable working range for the base emitter voltage of Tr3 is from 0.5V to about 0.7V, a width of only 200mV. This requires that Tr3 is biased to a Vbe of about 0.5V and that the control voltage drive is suitably attenuated, the bias being preset by adjusting R9. As two exponential converters are included in the synthesizer, both should be adjusted so that their respones are matched.
The need for an exponential transfer function is twofold. One, the subjective response to volume can be loosely described as "logarithmic". And two, the subjective response to a change in frequency is governed by the ratio of the two frequencies. Thus, frequency generation should be an exponential function of the control signal, if the control, from say a set of keyboards, is to be considered musically useful.
The construction of the synthesizer will be completed with a description of the sample and hold function, noise sources, waveform generator and power supply. All the syntbesizer functions will then be linked via the patch panel, keyboards and joystick control. Details of i.c. pin connections will be given and also examples of the synthesizer's use.

Correction

In Fig. 13, a resistor R10 of value lkOhm should be inserted between the + 15V supply and zener D3.

Reterence
1. D. T. Smith, þ'\þiultivibrators with Sevendecade Range in Period", Wrrefess World,
February l972, pp. 85-86.

Electronic Sound Synthesizer: Part 3   (Wireless World Oktober 1973)

Final circuit details, interconnection of functions by patch-panel , inteconnection of functions by patch-panel, keyboard and joystick control The final part of this series describing the construcrion and operation of a sound synthesizer completes the circuit functions provided with sample and hold, noise sources and the waveform generator ciruitry.

Sample and hold

It is very useful to have an analogue memory function, for use in such cases as a long fadeout where a constant control signal may be required throughout. One method of implementing this requirement is to use a sample and hold device with the following characteristics. The output should have a very small offset voltage coupled with a low output imþbance; also a long storage time, so that the output voltage will only drift by a few per cent per minute ; and a high accuracy over the specified input range. The sampling period is relatively short, being initiated by a positive-going pulse. Also, then is no input buffer because the output impedance of all the units of the synthesizer is low. The input voltagr range is approximately -0,5V to + 6,5V, being deliberately limited by D1 (Fig. 28). The signal is stored on C3, a low Ieakage capacitor, which is connected to the input voltage by an f.e.t. (Trl). This transistor is used as an analogue gate and is controlled by a monostable (Tr5,6,7). During the

monostable period, the gate is opened and the signal is sampled: The voltage stored on C3 is monitored by Tr3 , a current-driven source follower which can be preset to give a zero input/output offset voltage. Using a 500 Ohm source resistor, the spread in Vgs may range from about -0,5V to -5,0V, for drain current drives from about 0,5mA to 10mA, respectively. The constant current source may be preset to lie anywhere in this range. Thus by keeping Tr3 operating in its saturation region, and maintaining Id virtually constant, variations in Vgs can be kept very low for considerable changes in Vds.

  Setting up procedure: set R4 to about 500 Ohm (this is the "fine adjust" and it is preferable that R4 is a trimmer) and, with the input short-circuited, initiate the sampling with a positive pulse (this clears any charge on C,). Adjust R8 until the output voltage is as near to zero as possible and then use R4 to finely "zero" the output.

  Storage time with input short-circuited is 30 minutes for 5% droop and sampling time l4ms.

Noise sources

The noise sources fill two functions, firstly, a source of noise that can be filtered and modulated, and secondly, a low frequency source that can be used as a randomly fluctuating control voltage. This was achieved by constructing a white noise source and injecting the output into a spectrum shaping network and a low pass filter.

White noise source

The major difficulty in producing a simple reliable white noise source is the very nature of noise itself; it is non-deterministic. Several methods were available but the simplest and cheapest seemed to be the use of the leakage current Icbo of a faulty (high leakage) germanium transistor. However, this approach requires that the leaky transistor is specially selected, or even manufactured by gentle frying! A suitable device (Trl) Fig. 29, should produce an average noise level of approximately 40mV pk-pk, when used in the configuration shown. The white noise generator consists of three parts; the noise source Trl, an equalized high gain amplifier, and an output buffer. A high gain amplifier is used because the signal level from Tr1 is relatively low, thus particular care must be taken to isolate Tr1 and the input of the amplifier from any power supply fluctuations. Preset R3 is adjusted to give a suitable output level of between 2 to 3V pk-pk average.

Colored noise source

Coloured noise is produced by driving a spectrum shaping network with white noise, this network being a Baxandall tone control. Preset R17 is adjusted so that with both tone control pots at maximum the output
shows no signs of clipping.

V.L.F. noise

Very low frequency noise is extracted from the white noise source by two low pass filters, only one of which is available at one time, the selection being made by operating switch S1, Fig. 29. One of the drawbacks of
this method of producing v.l.f. noise, is that very little signal remains after filtering, the amplitude rapidly diminishing with decreasing cut-off frequencv.
  Preset R42 is adjusted so that the two v.l.f. outputs have the same amptitude, of approximately 3V pk-pk average.

Waveform generator

The waveform generator produces a control voltage that may be used to either frequency or amplitude modulate other units. The start of the waveform is initiated by a pulse input, the output rises "exponentially" and, after a predetermined period, falls "exponentially" (Fig. 30). Three controls are provided, attack, duration and decay and tbe pulse may be introduced electronically or from a manual pulse source.
  The circuit operation is as follows (see Fig. 31). The first section is a current driven monostable, the monostable period or duration being controlled by the current drive which is proportional to the wiper

setting of R6. The monostable is triggered by either a positive going input pulse or from a manual pulse upon release. The square wave produced is then fed into the attack/decay section where a capacitor is charged
via the attack conþrol R14 and diode D3. When the monostable period is over, the capacitor discharges via R12, the decay control, and D2. The potential across the capacitor is monitored, and an attenuated and buffered output signal is produced. A choice of duration times is available (C3 or C3 + C4 with S1 closed) and also a choice of time constants (C6 or C6 + C7 with S2 closed).

Joystick control

The joystick is a mechanically controlled voltage source having two degrees of freedom, and thus generating two independent control voltages, which are proportional to the stick's position. The device is essentially
a position transducer (Fig. 32) with two sense pots (R4 and R8, Fig. 33) mounted orthogonally. The range of the joystick is limited by the rectangular opening in the front panel giving approximately 90degrees of freedom in both the x and y directions. An extra pot can also be seen (Fig. 32) but this is used only as a spindle. The connecting cable should be thin and flexible so as to present as little restriction as possible to the stick's movement. Also, this cable should be firmly held by two P-clips, one on the joystick assembly and one on the front panel so as to stop continual wear on the soldered connections.
 The circuit function is illustrated in Fig.34. A constant potential is maintained across the control pots R4,8, in Fig. 33 and by the zener diodes D2,3. Also, the potential of these pots relative to 0V may be shifted by presets R2,6. Wiper crackle is attenuated by capacitors C3,5 and the wiper is buffered to the output by Tr2,3 and Tr5,6. With the joystick in the bottom left hand corner of its range, the two outputs x and y are zeroed by adjusting R2, 6 ; movement of the joystick in the x and y directions will then produce corresponding positive increases in the potential of the respective outputs.

Keyboard

The keyboard generates a control voltage that is linearly proportional to the status of the key that is pressed. This voltage is produced for the duration of the key's depression, retuming to 0V when the key is released. If two or more keys are pressed, the highest frequency key is selected automatically. Also, when a key is pressed, a pulse is generated, this being intended to trigger the waveform generator or the sample and hold unit. However, if the production of this trigger pulse is required, then care must be taken when playing the keyboard to ensure that each key is released before the next key is pressed. If this procedure is not observed, then, even though the control voltage does change correctly, no pulse will be generated. The result is the production of a signal somewhat different to that intended.
The keyboard control circuit is shown in Fig. 35. A constant potential is maintained across resistors R1 to R48 and as all these resistors are the same, they forrn a potential divider composed of equally spaced steps. The switches S1 to 49 are operated by the keyboard and form, with diodes D1 to D49 and resistor R51 a "Minof" analogue gate. Thus, whatever combination of switches are pressed the most negative voltage is selected, this voltage appearing at the emitter of Tr3. Note that when no switches are pressed, the emitter of Tr3 rises to nearly + Vcc. This voltage must be modified so that it is in a suitable form to act as a control signal. It is attenuated (R55), inverted and its d.c. level is shifted (R59) so that the range of outputs is from 0V to + 3V. Also the feedback around IC1 is such that when no keys are pressed, and the emitter of Tr3 rises to nearly + Vcc, the output (Vc) is prevented from going negative, and stays at 0V.
It is required that a pulse is generated a the moment when a key is pressed, but not when it is released. This would be a simple response to achieve (by detecting the transition direction of the "Minof" voltage) if it were not for the phenomenon of contact bounce. The spikes produced by the bounce can be largely suppressed (C3) but there is still a possibility of generating a pulse by mistake. One method of overcoming this dilemma is to use a Schmitt trigger with a sizeable hysteresis loop, so that, as the "Minof" signal plus spikes rises or falls, it causes the Schrnitt to change state only once. The direction of this change is determined by whether the input is rising or falling (i.e. whether the key is being released or pressed) and can thus be made to produce a pulse only on the falling transient.

Some applications of the keyboard are given in Fig. 36. Fig. 36(a) shows a patch diagram of simulated piano sound. A sinusoidal signal is given a fast attack and a slow decay. Note that the control output (Vc) from the keyboard is modified by the exponential converter, so that an equally tempered scale is produced. However, if the key is prematurely released, the output promptly changes frequency. Fig. 36(b) overcomes this difficulty, by using the sample and hold circuit to store the control signal. Also, reverberation with a slow sinusoidal modulation has been added producing a pleasant effect similar to a xylophone. Fig. 36(c) shows a network for producing bell-like "clanging" noises.

Patch panel

To provide a flexible means of programming the synthesizer, a patch panel similar to the type used in analogue computers has been included. As the input and output impedance of all the units is low, it was possible to use an unscreened system. In fact, ordinary 4mm banana plugs and sockets were eventually chosen, this deci-sion being greatly influenced by cost factors. This choice, however, presents a danger of damage due to misuse. If two outputs are connected together, then it is possible that some damage will eventually occur, although how long it takes is difficult to predict. Certainly, from previous experience of a similar synthesizer, no lasting damage was seen to occur when an error of this sort was made. To minimize this danger the sockets are coloured, all the inputs being yellow, the outputs being any other colour.
The synthesizers on the market appear to have overcome this difficulty, but at some cost. One method is to employ a series of horizontal and parallel conductors, one set being the inputs, the other set the outputs. Pins are then plugged in to make a connection between an input and an output, thus the danger of an "output to output" never arises. Other methods are to use switches or jack plug instead of pins. These systems an all pre-wired and so another problem, that of the "birds' nest" of patchcords (an all too familiar sight to those who have ever used an analogue computer) has also been eliminated. However, this advantage has been gained at some expense. The layout of the patch panel was determined on a logical basis ; that is, all the oscillators on one section, the v.c.as and v.c.f. in another, the noise sources in one block etc. Also, to make connections with an external amplifier, a coax. socket was included as well as two sockets which were connected to "ground"   potential, these being used as a 0V reference point for external equipment such as voltmeters or oscilloscopes.

Power supply

Many units of the synthesizer are sensitive to power supply fluctuations and so a stabilized supply is desirable. The circuit diagram of the supply used is given in Fig. 37. Without this suppression it is possible to trigger a response by switching on and off unconnected (except via the mains) equipment. Care should be taken in constructing the power supply to avoid introducing any high rurrent paths that might adversely affect the circuit operation.

Appendix

Voltage controlled filter

Consider a bandpass filter consisting of a series LCR network. The behaviour of this system is characterized by a linear second order ditferential equation with constant coeffcients. Using anatogue techniques, it is possible to model this system, but more important it is possible to make the coeffcients variable, in fact, voltage controlled.
The general equation of a linear second order system is
$F(t) = xi + 2kwnx + wnE2x$
Where wn is the undamped natural frequency, k is the damping factor (note, the quality factor Q = 1/2k); and F(t) is a generalized forcing function. The solution of this equation consists of two parts ; the particular integral that depends on F(t), and the complementary function that depends on the solution of the right hand side only. Using the network shown in Fig. 38(a) it is possible to implement the complete solution. Different forms of F(t)
can be inserted, and by varying pots 4 and 5, the values of wnE and 2kw can be

modified. By monitoring the voltage at the output of integrator l (-x), the response of a bandpass filter, with the same coefficients, under
the influence of the same forcing function F(t), is observed. (The coefficients for a series LCR cinuit would be wn = 1/LC and k = R/2 C/L). By monitoring x, a low pass response would be seen, and xi a high pass response. If pot 4 were an electronic multiplier, then wn (and hence k) could be voltage controlled. Now 1/2k = Q, so it is thus possible to control both the resonant frequency and thus the quality factor. Two points are immediately noticeable; one, the
Q factor increases with frequency. This is because if pot 5 remains constant, we have
    2kwn = constant

but   k =   1 / 2Q .:. wn/Q = constant.

Fractional changes in pot 4 i.e. wnE2, result in the square root of that change in wn. One mefhod of curing both of these effects is to use two multipliers Fig. 38(b). It is easily shown that there is a linear relationship between the control voltage Vc and wn". Also the Q factor is invariant with resonant frequency changes (assuming multipliers 7 and 8 are matched), and the dynamic range of the filter is equal to that of one of the multipliers. It would also be possible to control tbe Q factor with yet another multiplier, but the use of multipliers is both expensive and introduces complications. It was for these reasons that the configuration shown in Fig. 38(c) was finally choosen. Hence, the relationship between Vc and wn is "linear", the dynamic range is nearly 10 to 1 and the Q factor increases with frequency. The variation of the Q factor is not as disturbing an effect as it may appear to be, especially when it is considered qualatively.

Acknowledgement

Capacitor ratings

Voltage ratings of electrolytic capacitors shown in Figs. 28-38 an as follows :
Fig. 28-C2/35V.
Fig. 29-C1/25V, C3/l0V, C4/10V,
        C7/l0V, C8/25V, C9/25V,
        C10/16V, C11/40V, C14/40V,
        C17/16V, C19/16V, C20/16V,
        C27/l0V.
Fig. 31-C3/40V, C4/16V, C5/40V,
        C6/40V, C7/25V, C8/25V.
Fig. 33-C1/25V, C2/25V, C3/l0V,
        C4/25V, C5/l0V.
Fig. 35-C1/25V, C2/25V, C6/16V.
Fig. 37-C2/40V, C3/16V, C4/25V,
        C6/25V, C7/ 16V, C8/40V.

① voltage controlled units

VCO₁ voltage controlled oscillator 1

VCO₂ voltage controlled oscillator 2

VCO₃ voltage controlled oscillator 3

VCA₁ voltage controlled amplifier 1

VCA₂ voltage controlled amplifier 2

VCF voltage controlled filter (bandpass)

audio mixer

reverberation

summer/inverter 1 & 2

programming unit patch panel

joy stick control

waveform generator

keyboards

exponential transfer function 1 & 2

sample and hold

external amplifier/loudspeaker

white noise source

coloured noise source

very low frequency noise source

② signal processors

③ noise sources
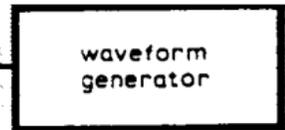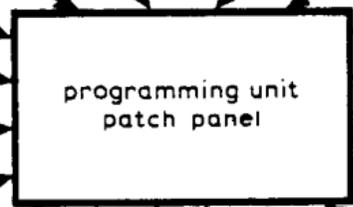
④ control voltage sources
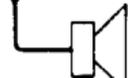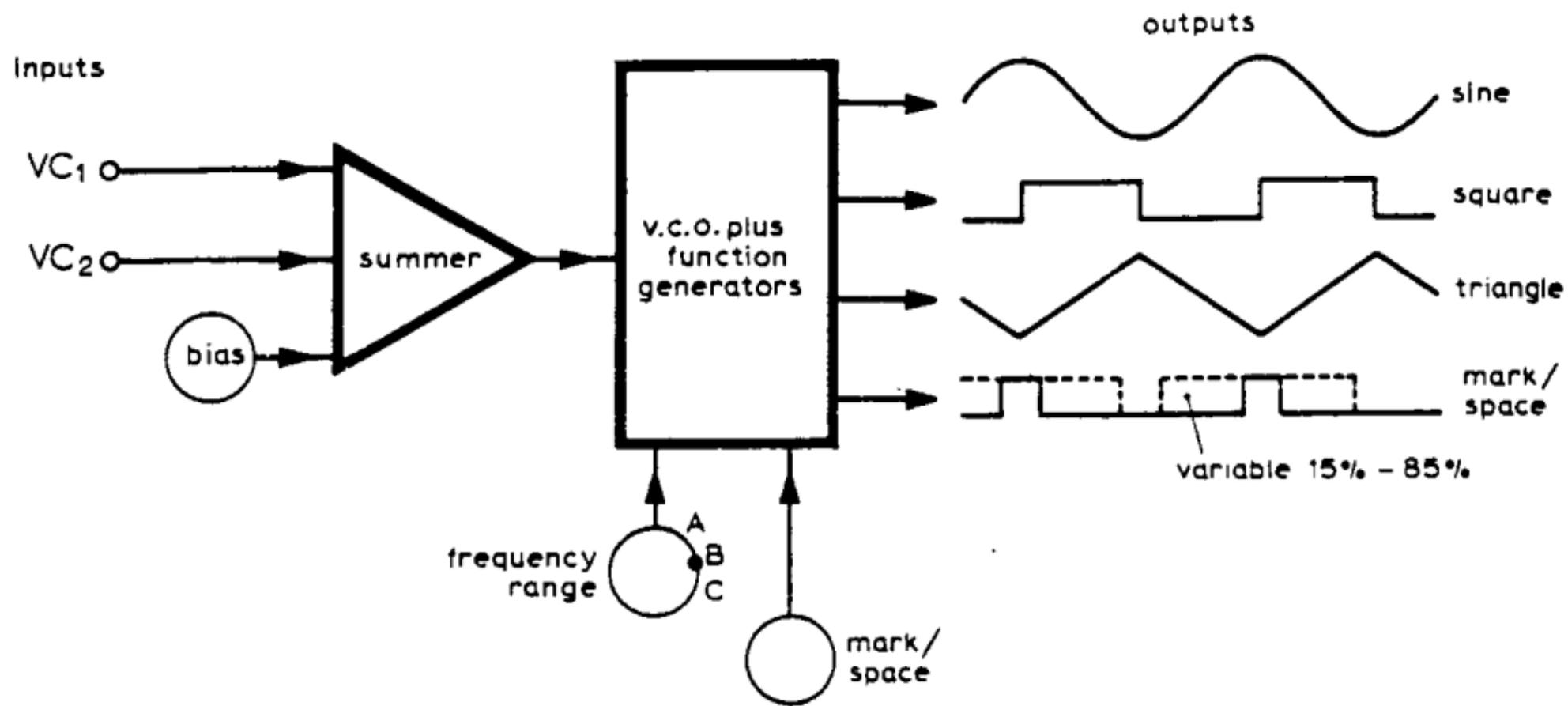
Fig. 2. Functions of voltage controlled oscillator, VCO_1.
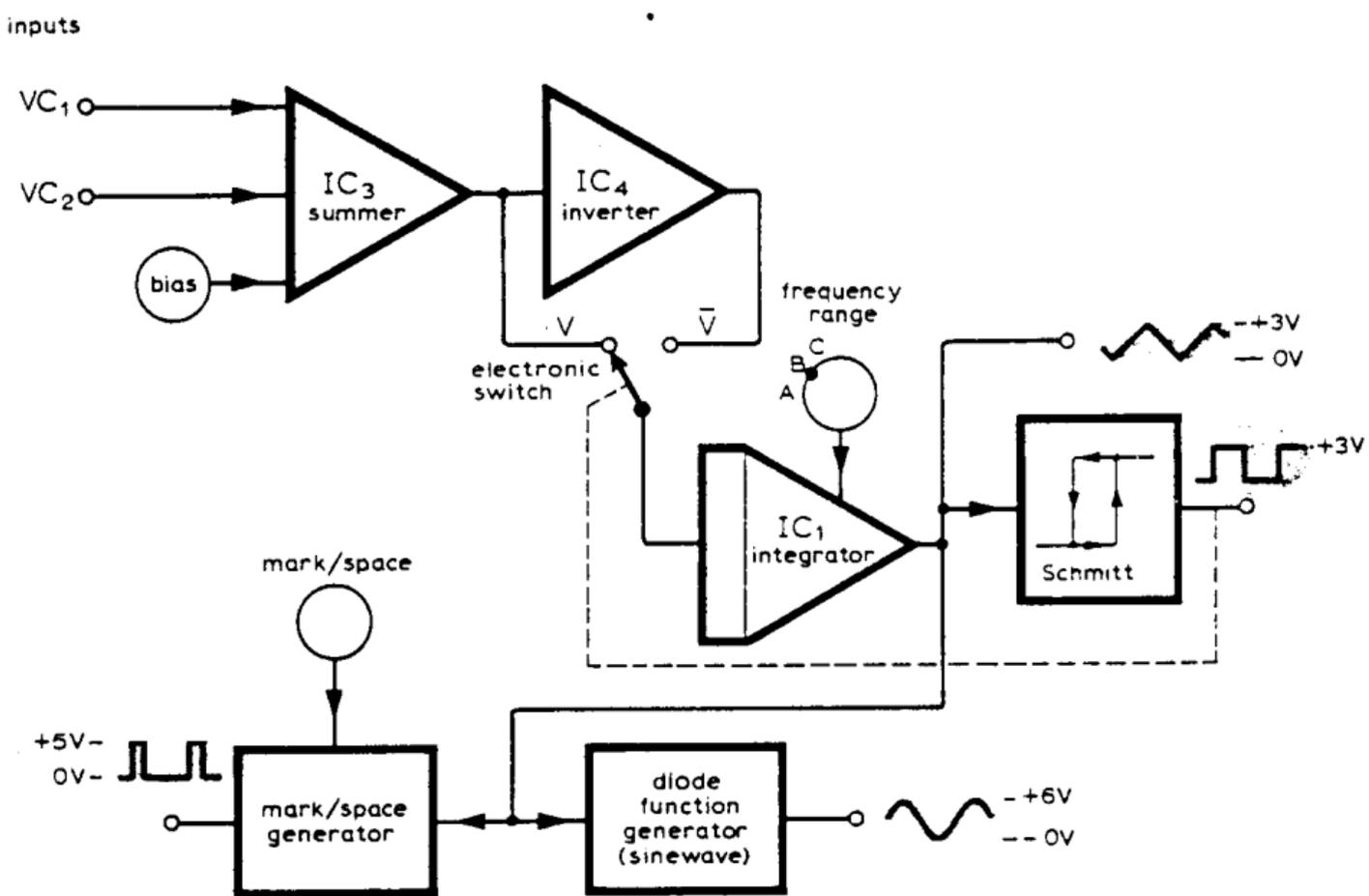
Fig. 3. Oscillator VCO_1 *in block diagram form.*
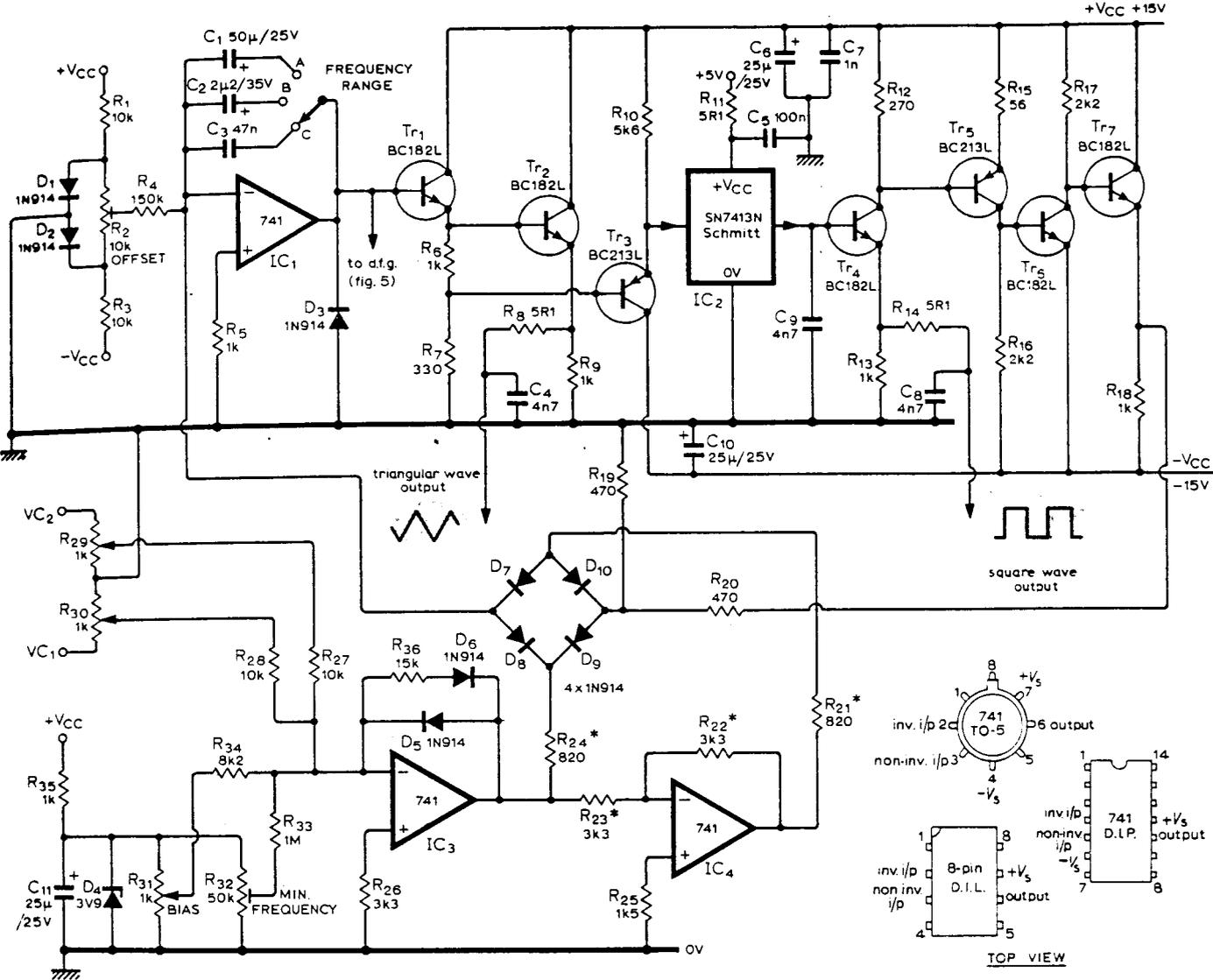
Fig. 4. Circuit of VCO₁. All resistors are 5%, ¼W unless asterisked — these are 2%.
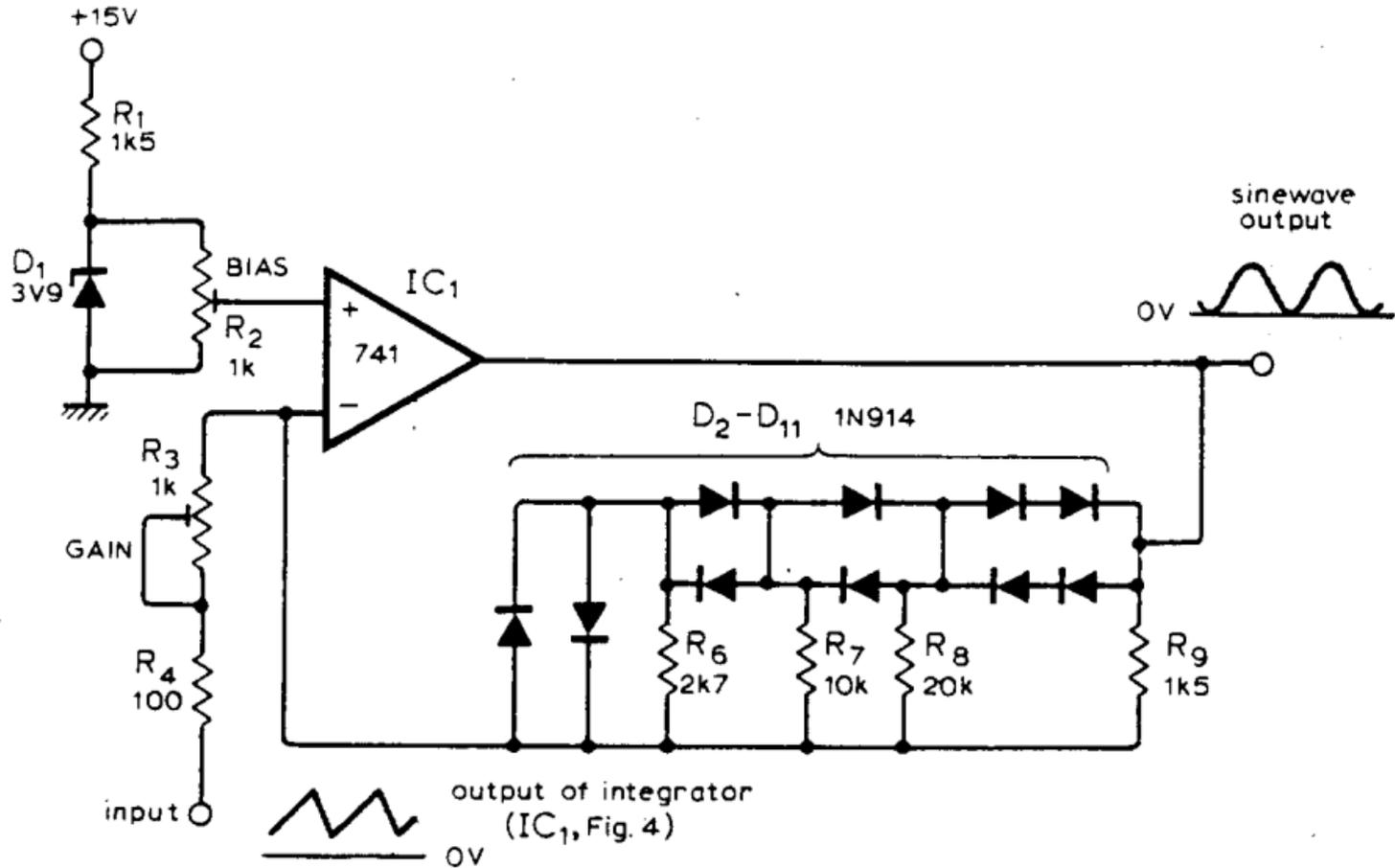
Fig. 5. Diode function generator which produces a sinewave output when fed with the triangular wave output from the integrator IC_1 in Fig. 4. All resistors are $\frac{1}{4}W$, 5%.
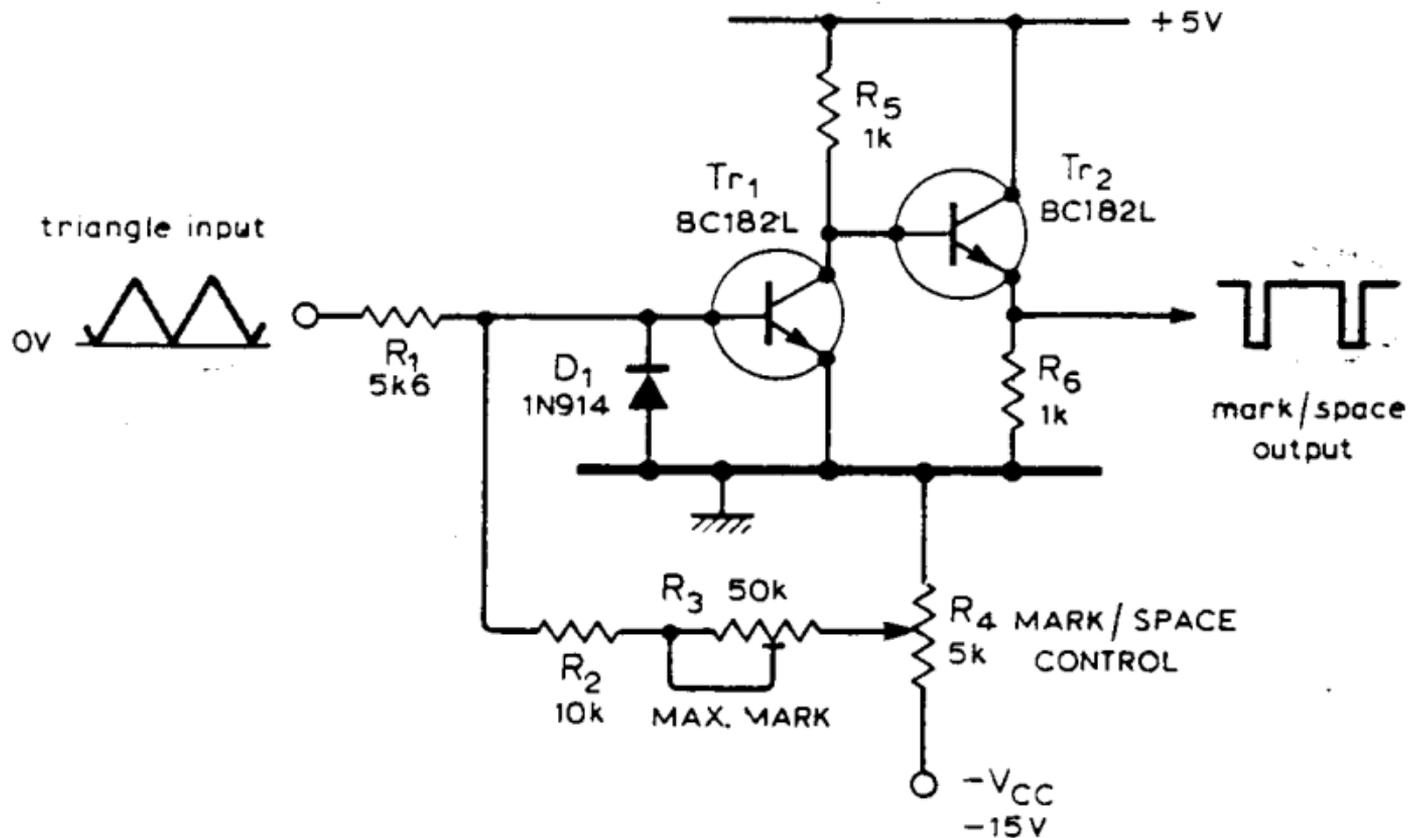
Fig. 6. Mark/space generator whose output mark/space ratio is variable from 15-85%.

triangle output

generated 'sinewave' output
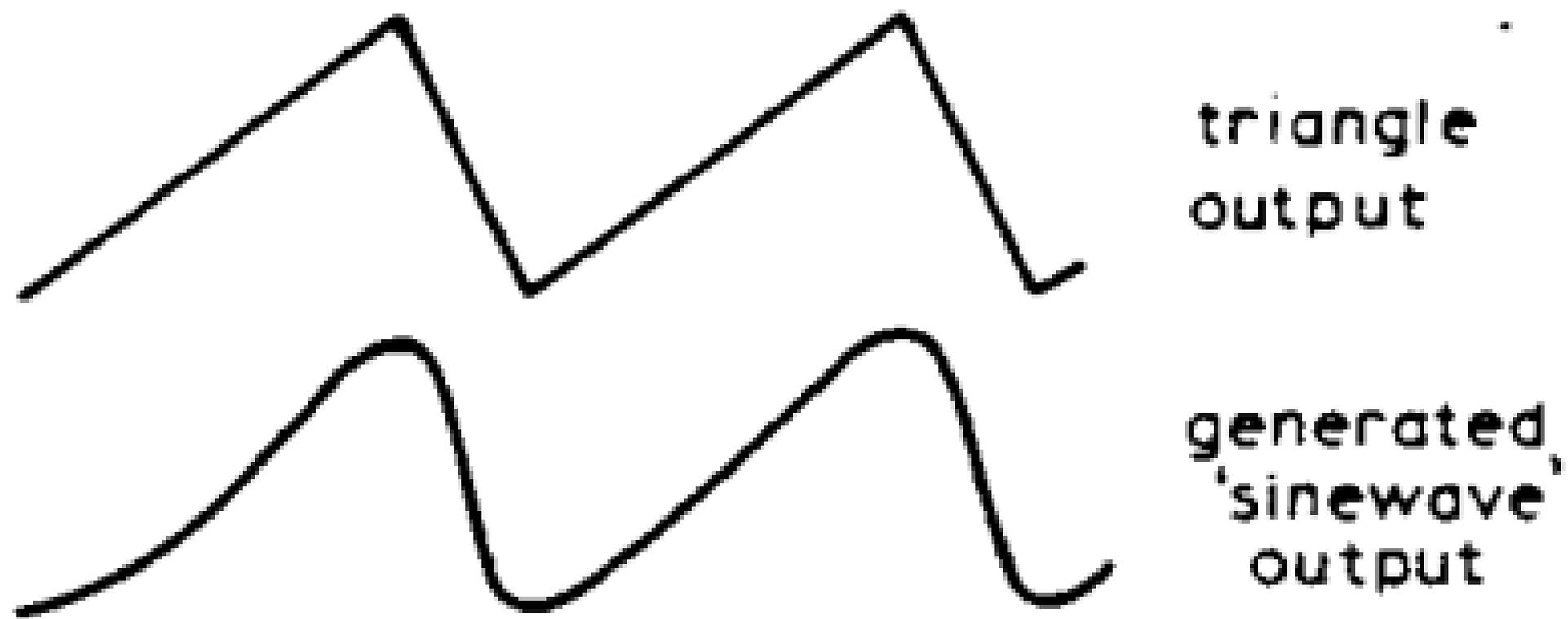
Fig. 7. Asymmetry caused by an imbalance in the diode ring switch.

bias wrong

not enough gain

to much gain

optimum output a 'sinewave'

Fig. 8. Output of the diode function generator with cause and effect of incorrect bias and gain adjustment.
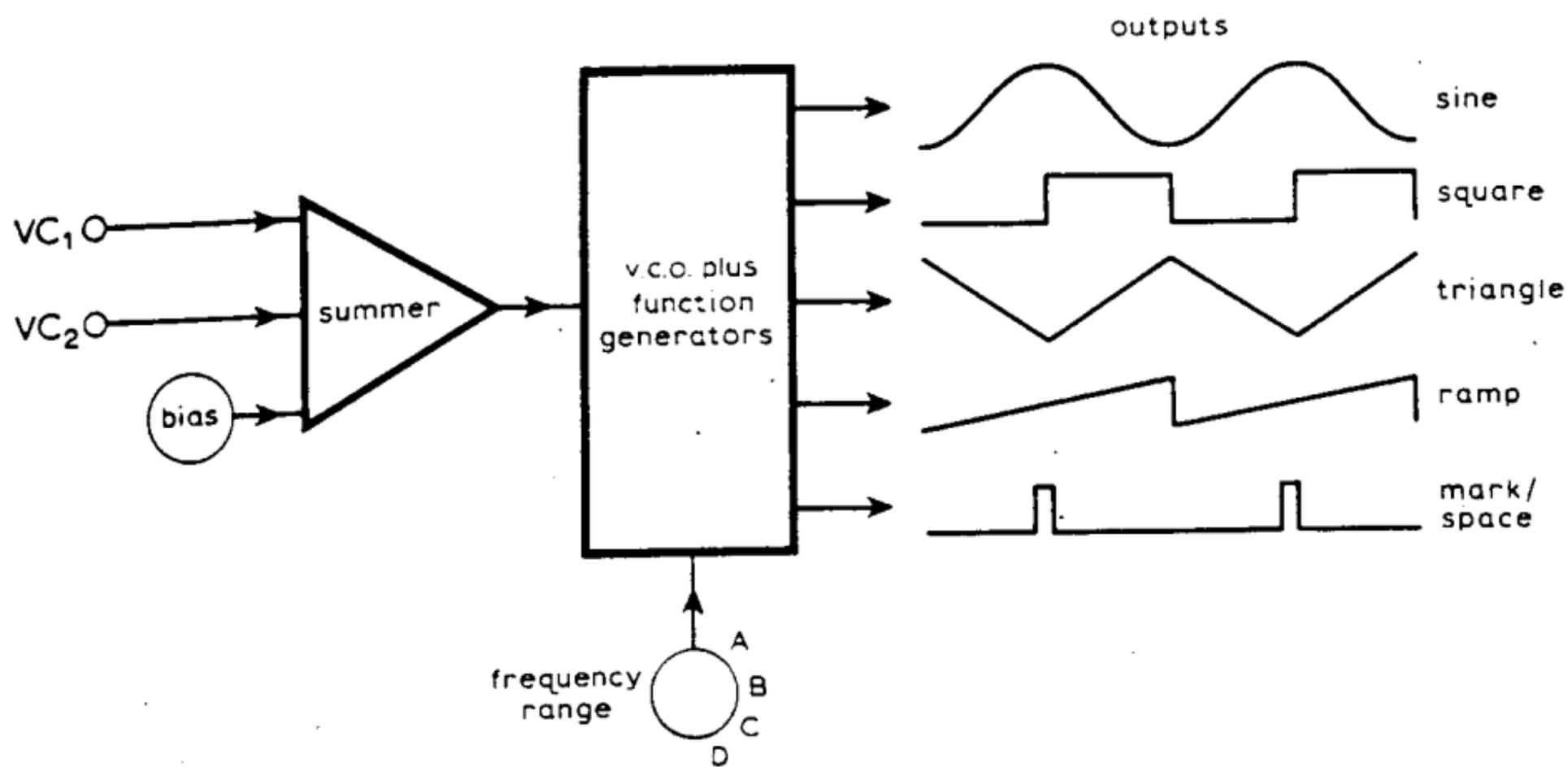
Fig. 9. Functions of voltage controlled oscillator VCO₂
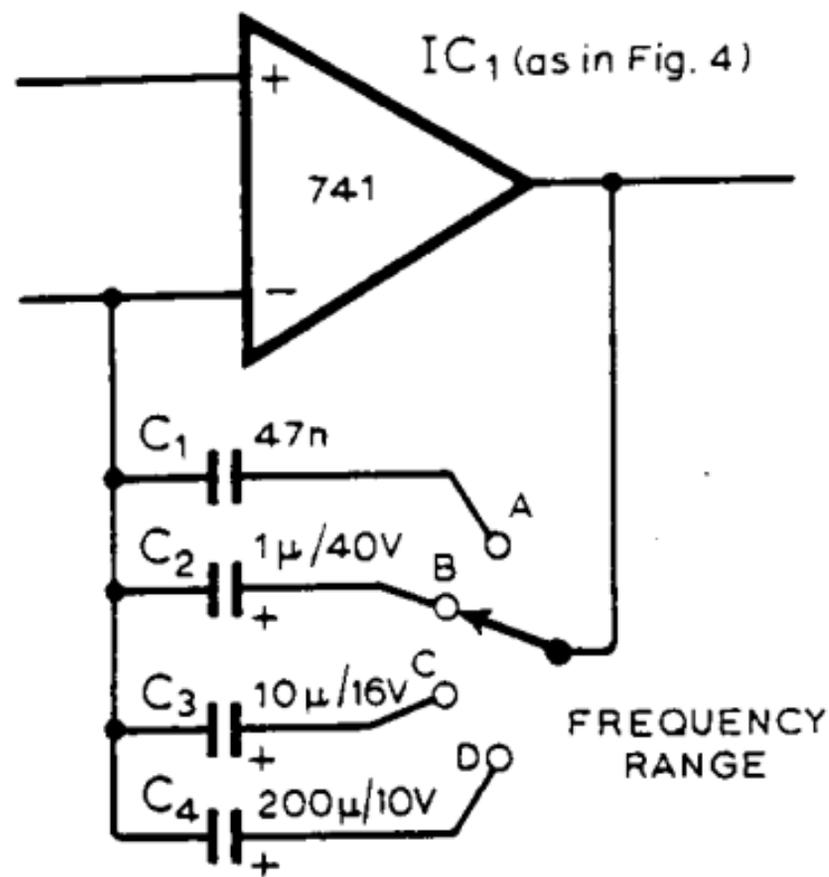
Fig. 10. Different frequency ranges of VCO₂. The rest of the circuit is the same as in Fig. 4.

square-wave input

pulse

--0V

--0V

monostable +V_e edge triggered

triangle to sawtooth converter

--0V

--0V

triangle wave input
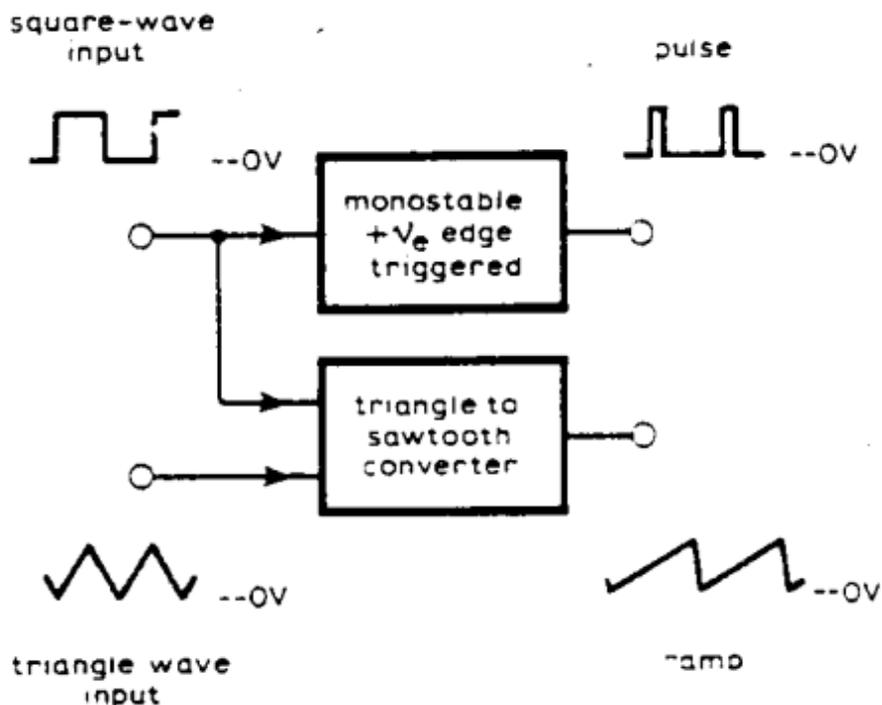
ramp

Fig 11. Function generators of VCO_2 providing a pulse or ramp output.
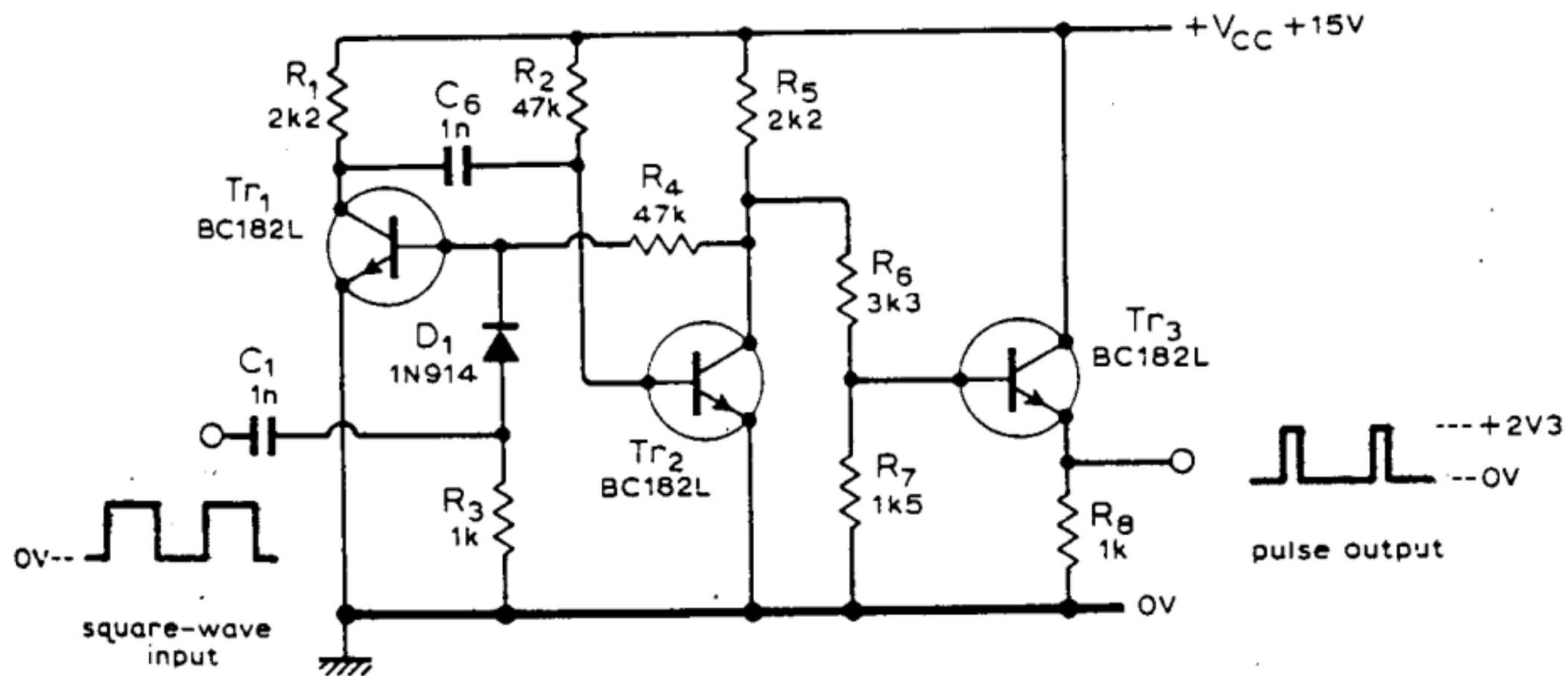
Fig. 12. Circuit of the pulse function generator.

Fig. 13. Circuit of the ramp function generator.

Fig. 14. Using VCO_1 and VCO_2 as a sweep frequency oscillator.
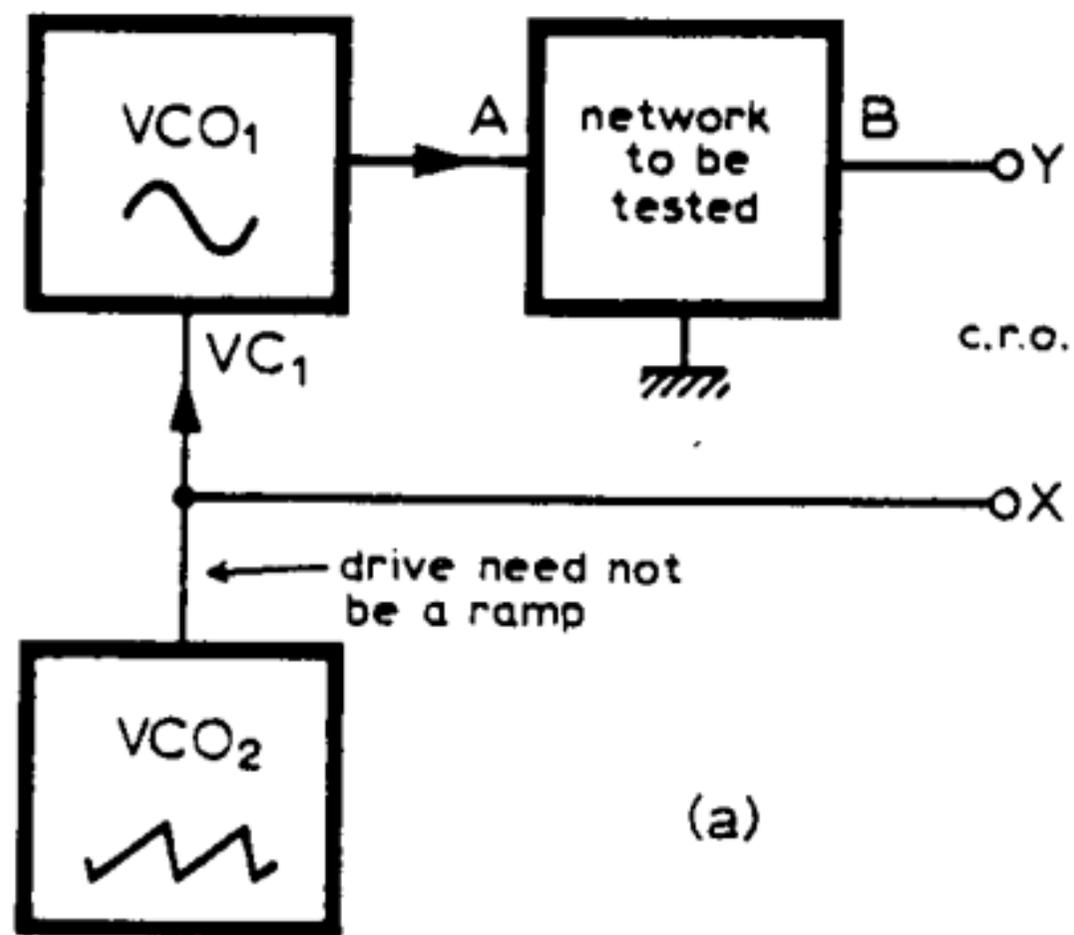
Fig. 15. Functions provided by voltage controlled oscillator, VCO_3.
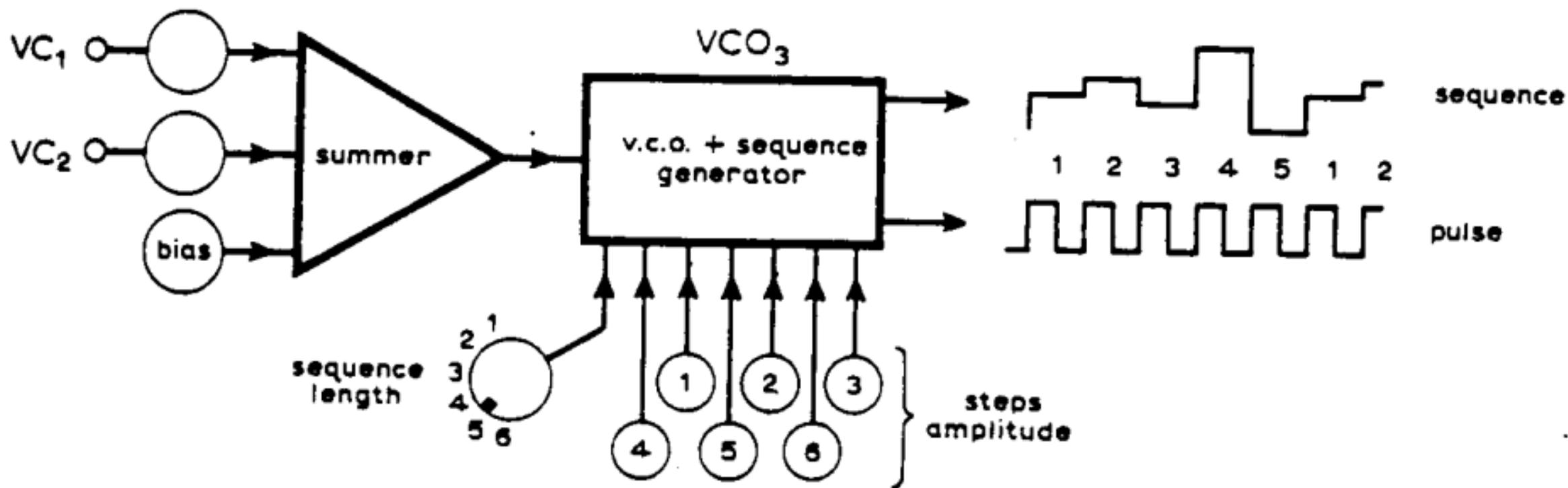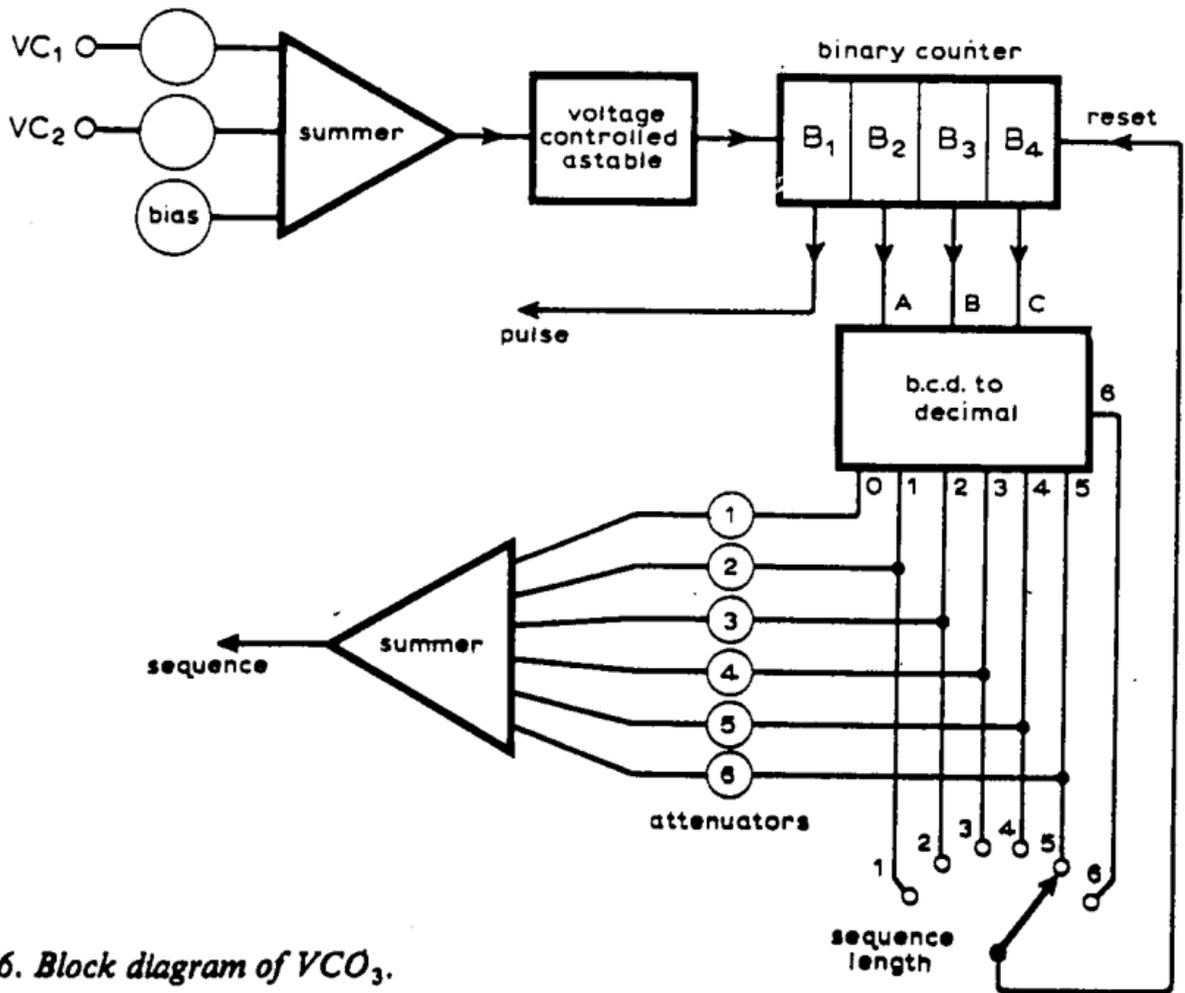
Fig. 16. Block diagram of VCO₃.
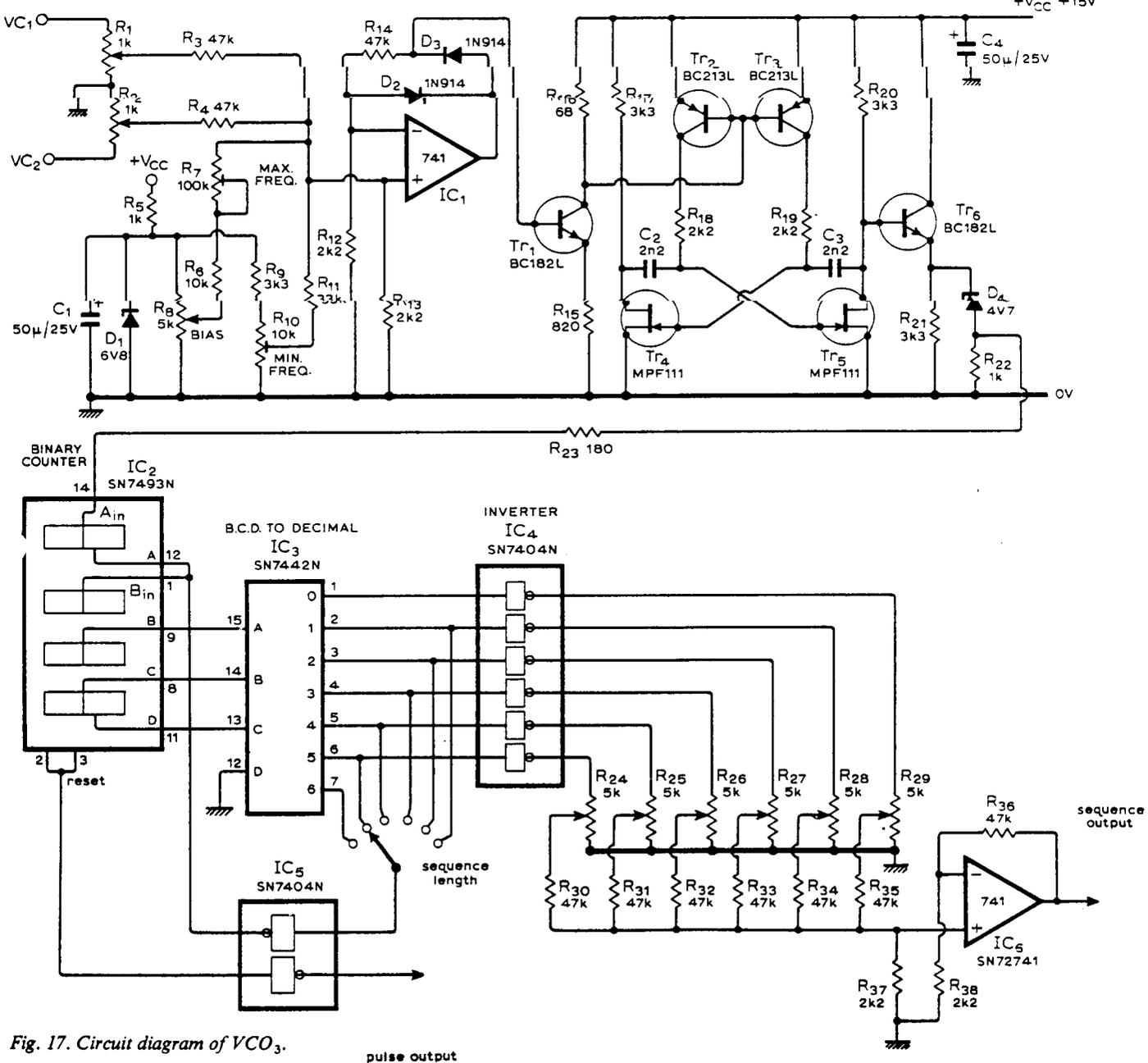
Fig. 17. Circuit diagram of VCO₃.
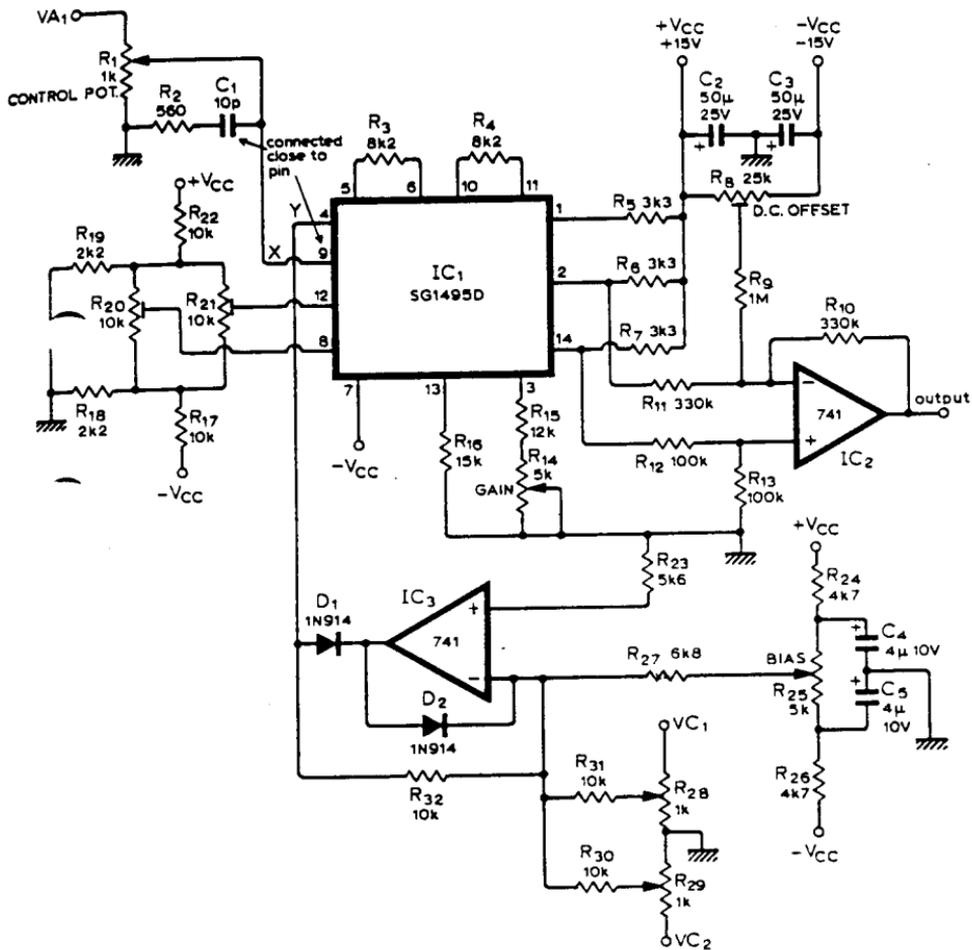
Fig. 18. Circuit diagram of voltage controlled amplifier $VCA_{1,2}$.
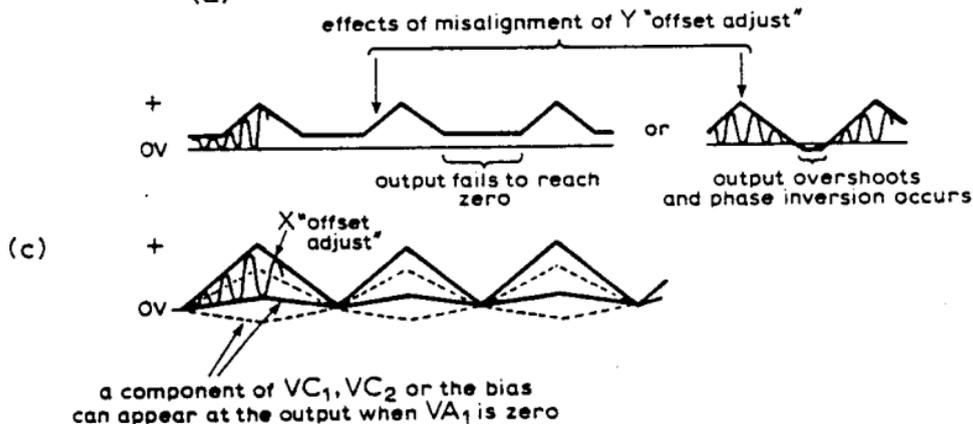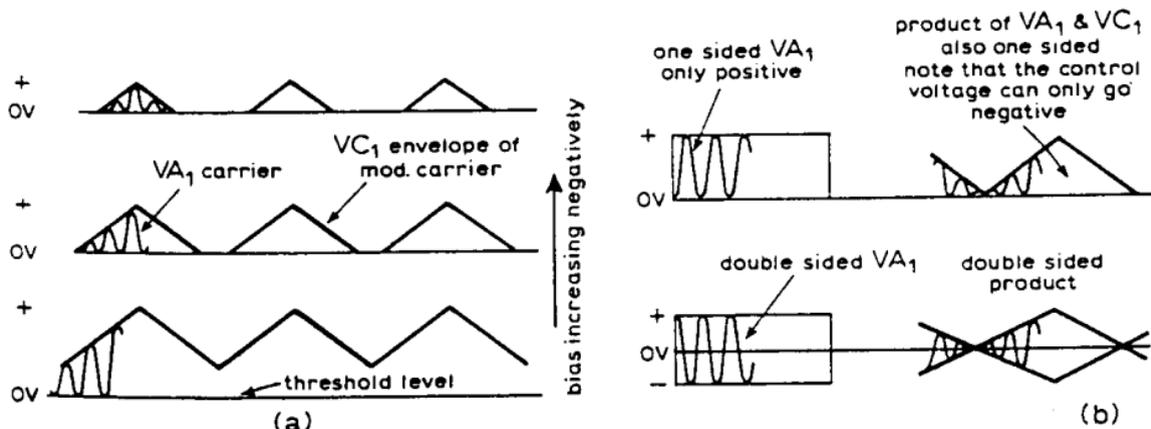
Fig. 19. (a) Variable threshold effect in aligning the v.c.as. (b) Waveform envelopes from VA_1 and VC_1 products. (c) Effects of misaligning the Y "offset adjust". See text for full explanation.
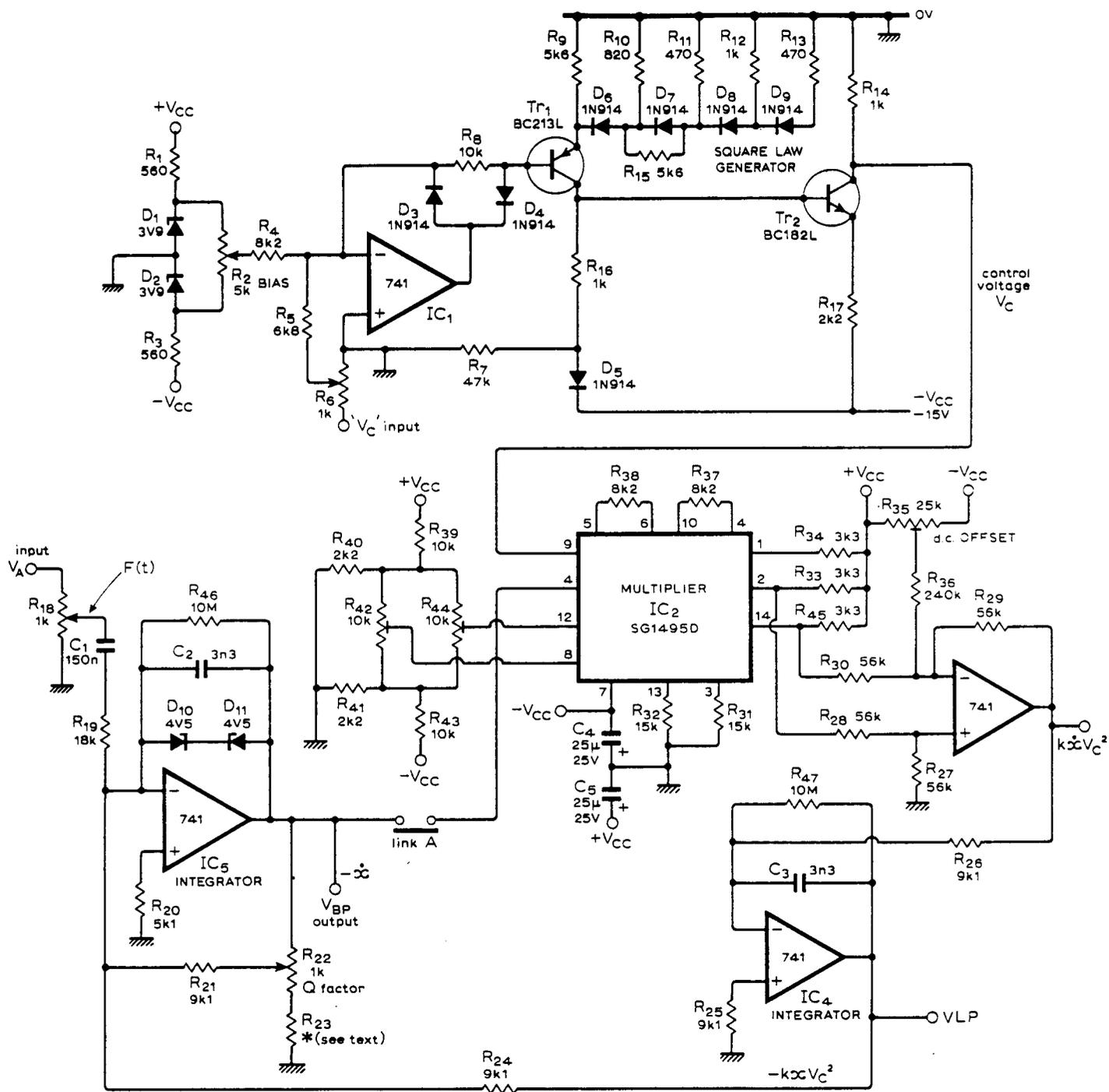
Fig. 20. Circuit of the voltage controlled filter.
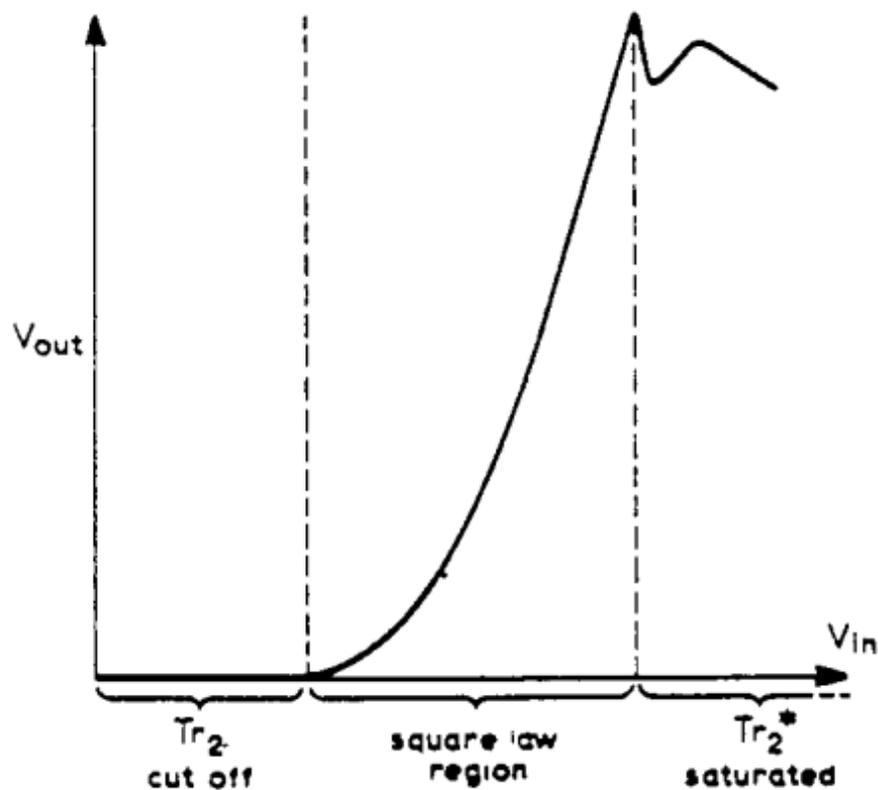
*as in fig. 22

Fig. 21. Transfer function of the square
law generator in Fig. 20.
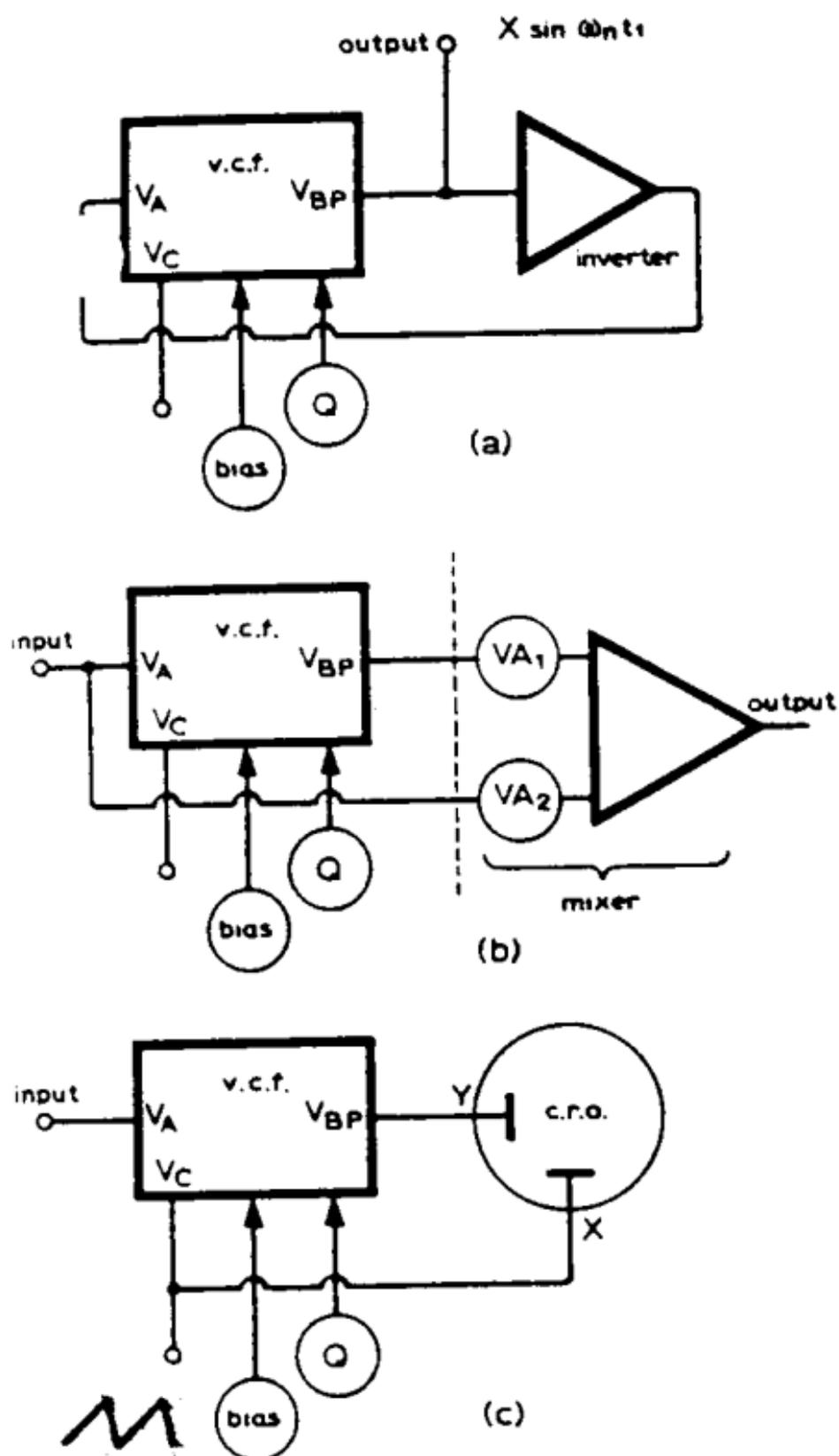
i   22. Using the voltage controlled filter
as (a) a low distortion oscillator (b) a
notch filter and (c) a spectrum analyzer.
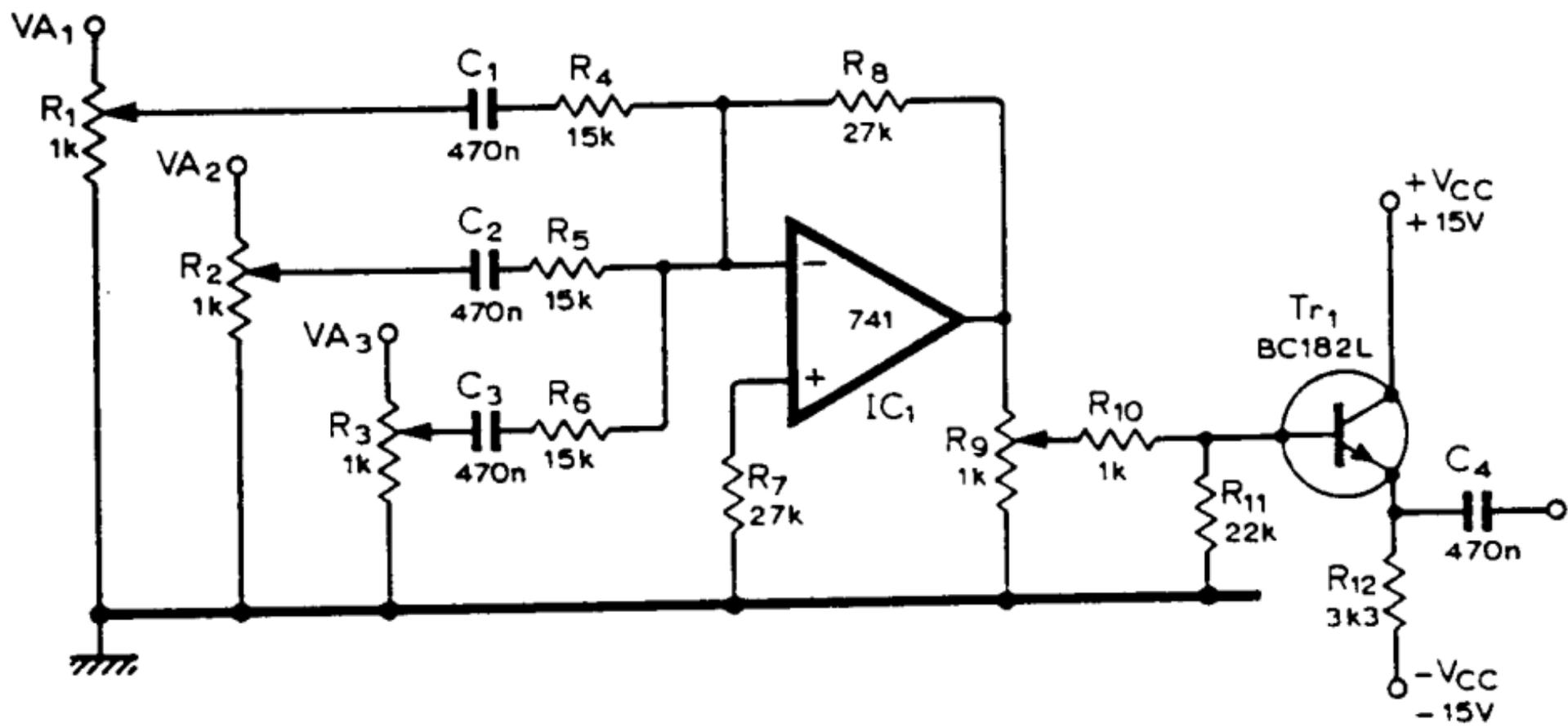
Fig. 23. Three-channel virtual-earth a.c. coupled mixer circuit.

Fig. 24. Direct coupled virtual-earth mixer.
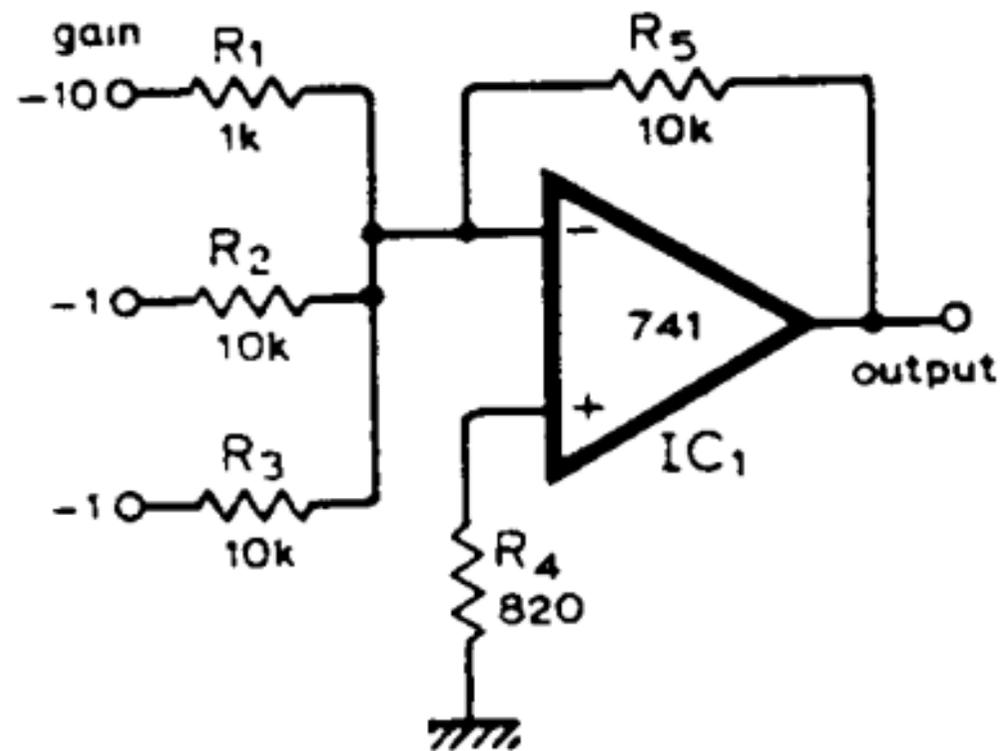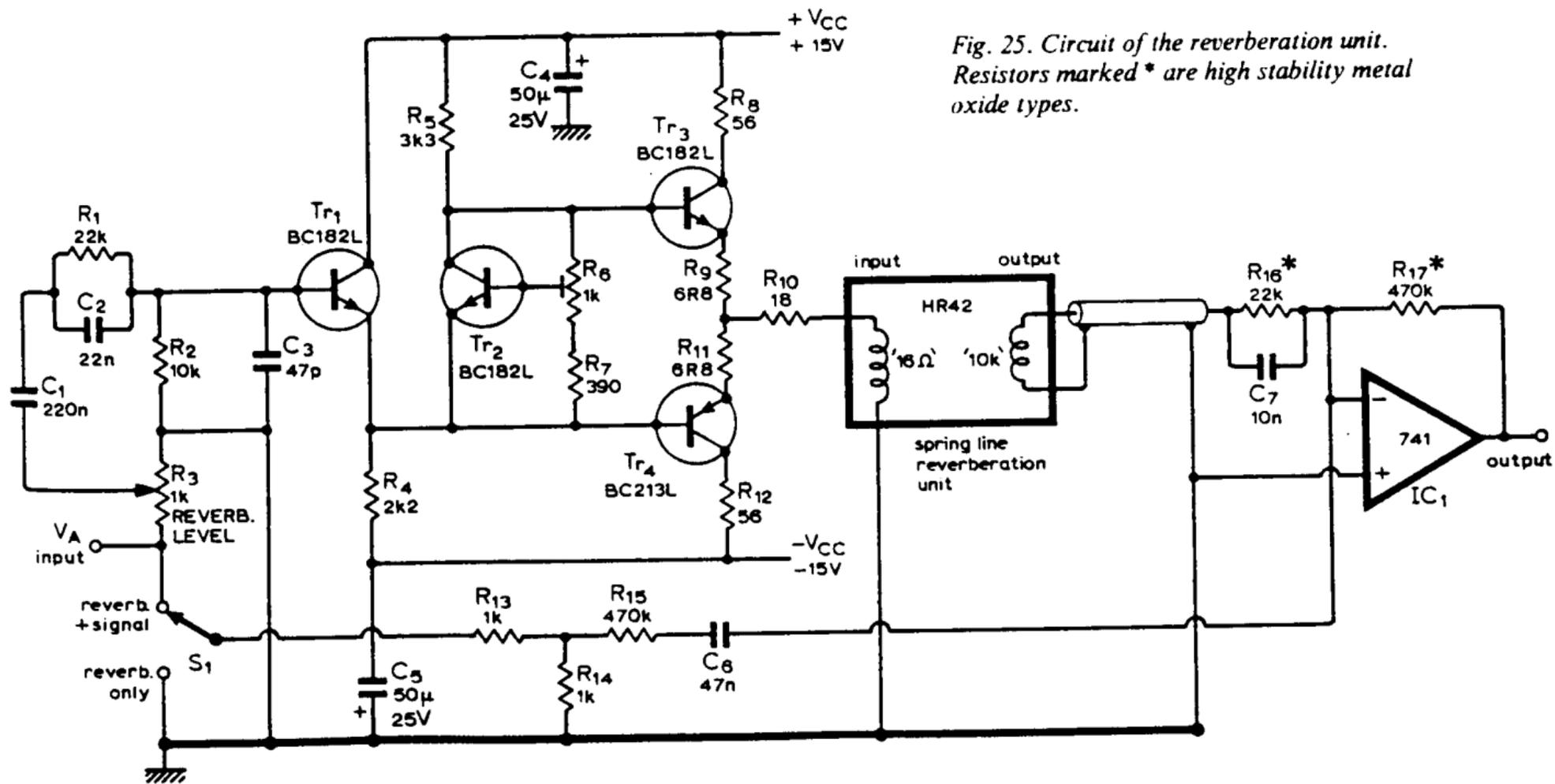
Fig. 25. Circuit of the reverberation unit. Resistors marked * are high stability metal oxide types.
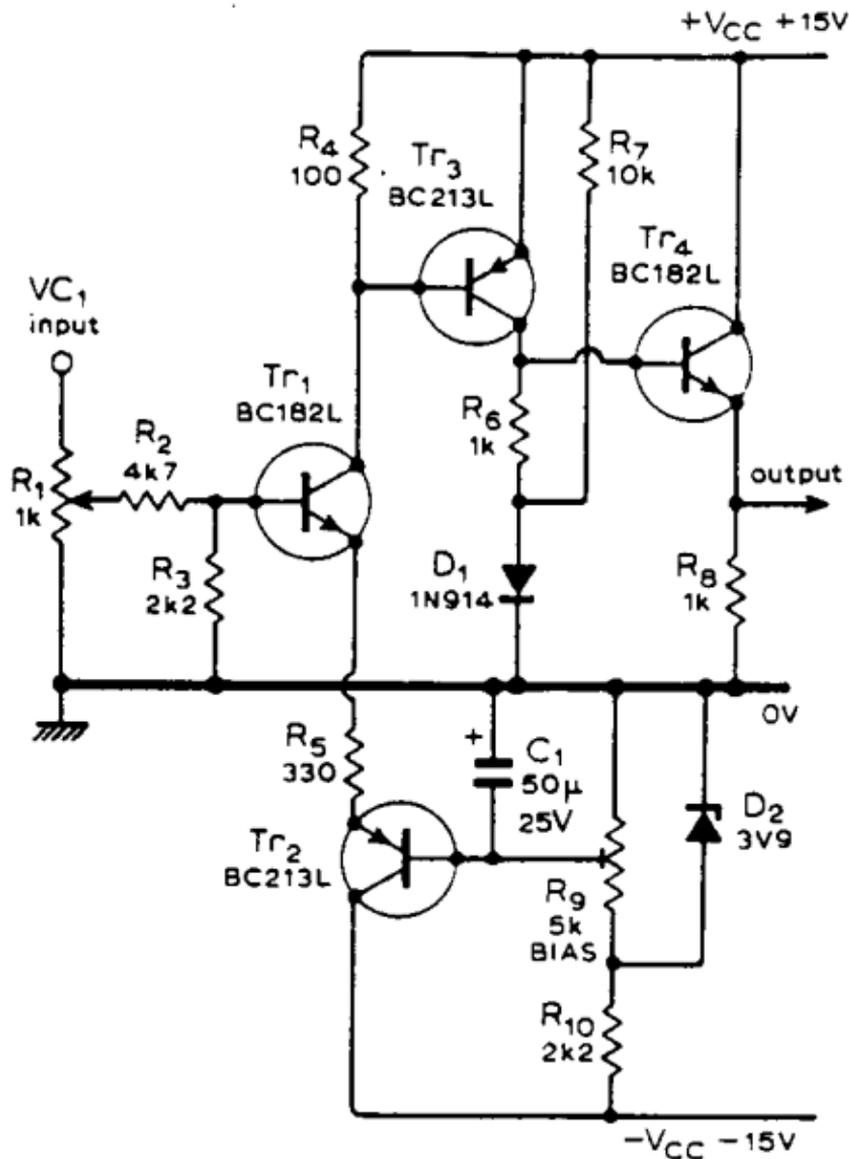
Fig. 26. Circuit of the exponential converter. Two converters are included in the synthesizer.

Fig. 28. Circuit providing the sample and hold functions. Resistors are 5%, $\frac{1}{4}W$, capacitor C_3 should be a polyester low-leakage type.
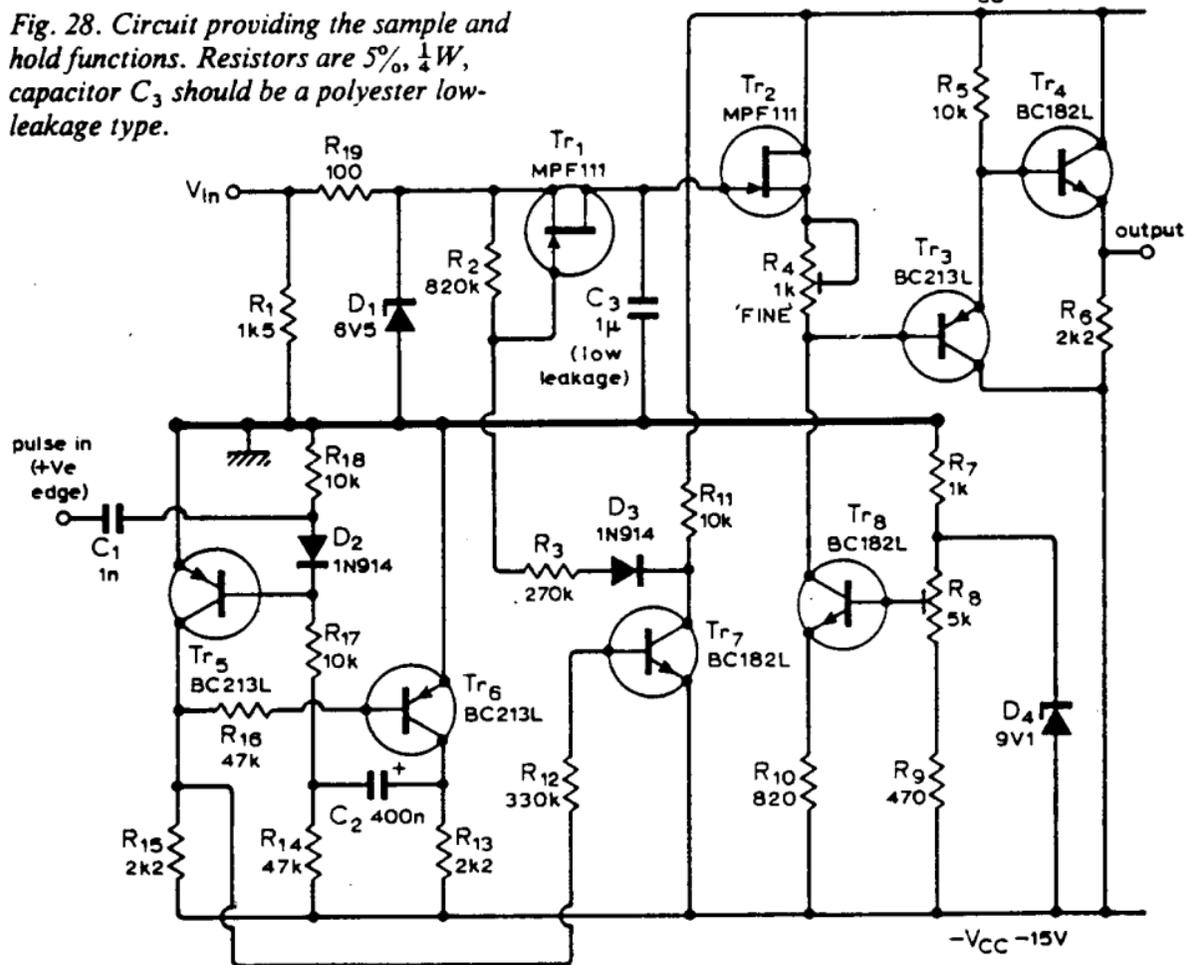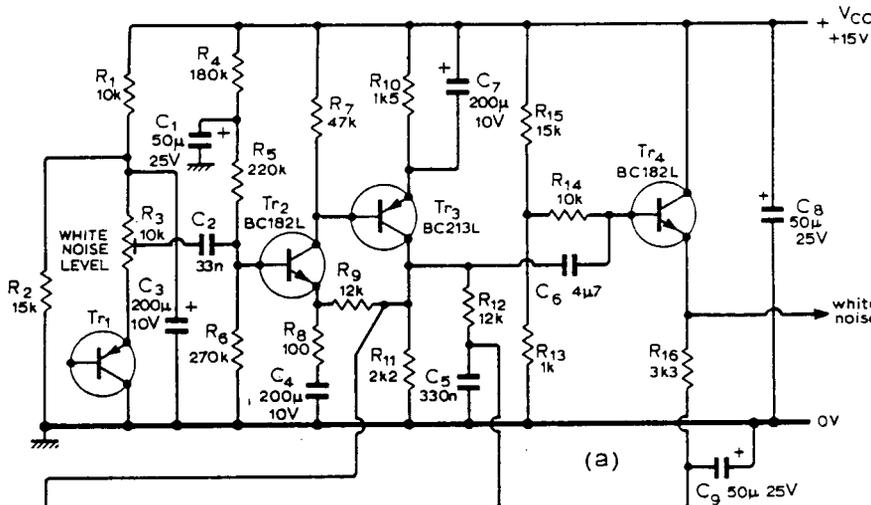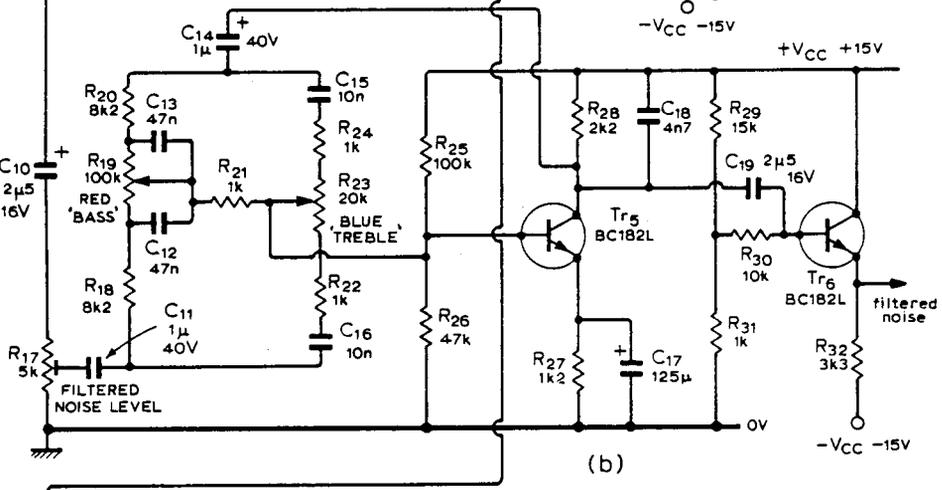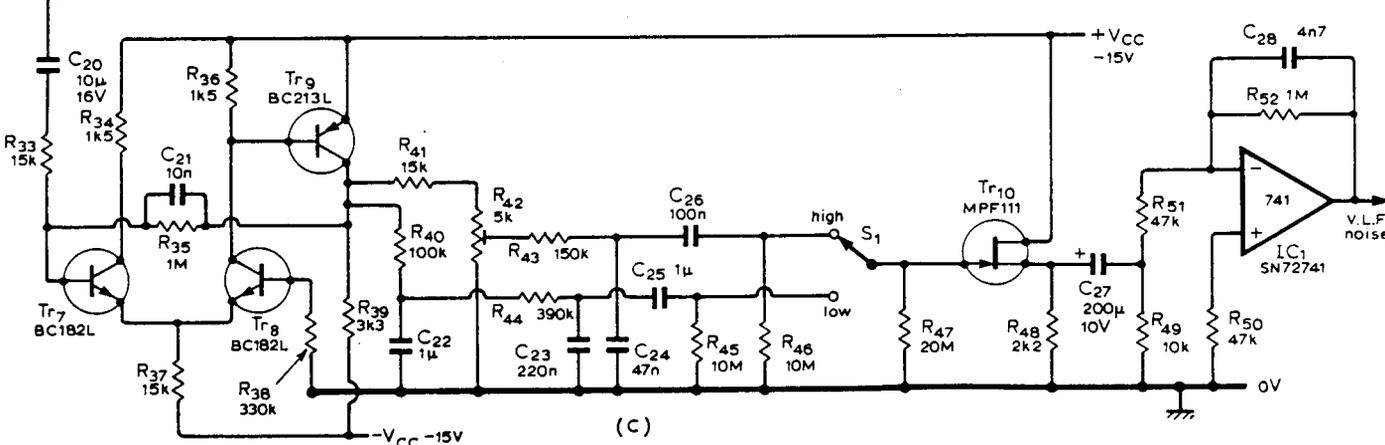
Fig. 29. Noise source circuitry which provides white, filtered or v.l.f. noise. Resistors are 5% ¼ W, capacitors C_{22}–C_{26} are polyester low-leakage types.
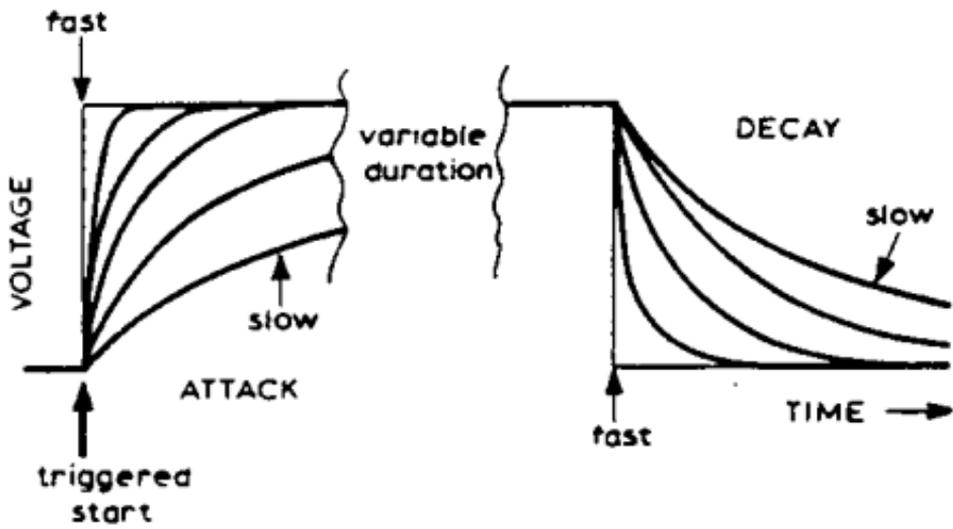
	ATTACK	DECAY	DURATION
	fast	slow	short
	slow	slow	long
	fast	v. fast	short
	slow	fast	long

Fig. 30. Outputs available from the waveform generator.
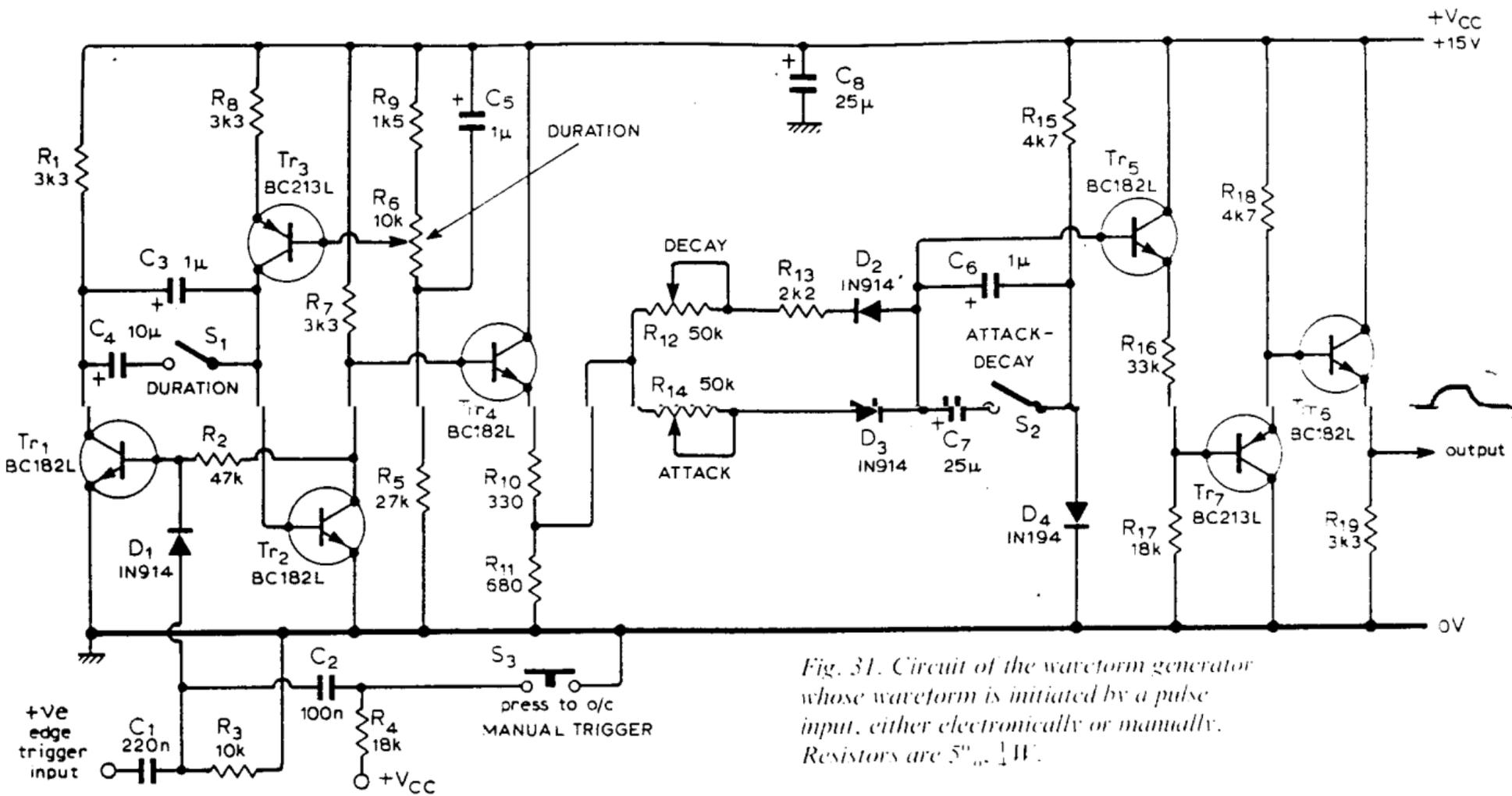
Fig. 31. Circuit of the waveform generator whose waveform is initiated by a pulse input, either electronically or manually. Resistors are 5%, ¼W.

spare pot used as a spindle

alloy

90° of movement in X and Y directions

connecting wires should be firmly clamped by two 'P' clips (the connecting wire should be thin and flexible)

joystick assembly mounted on two supporting brackets fixed to front panel
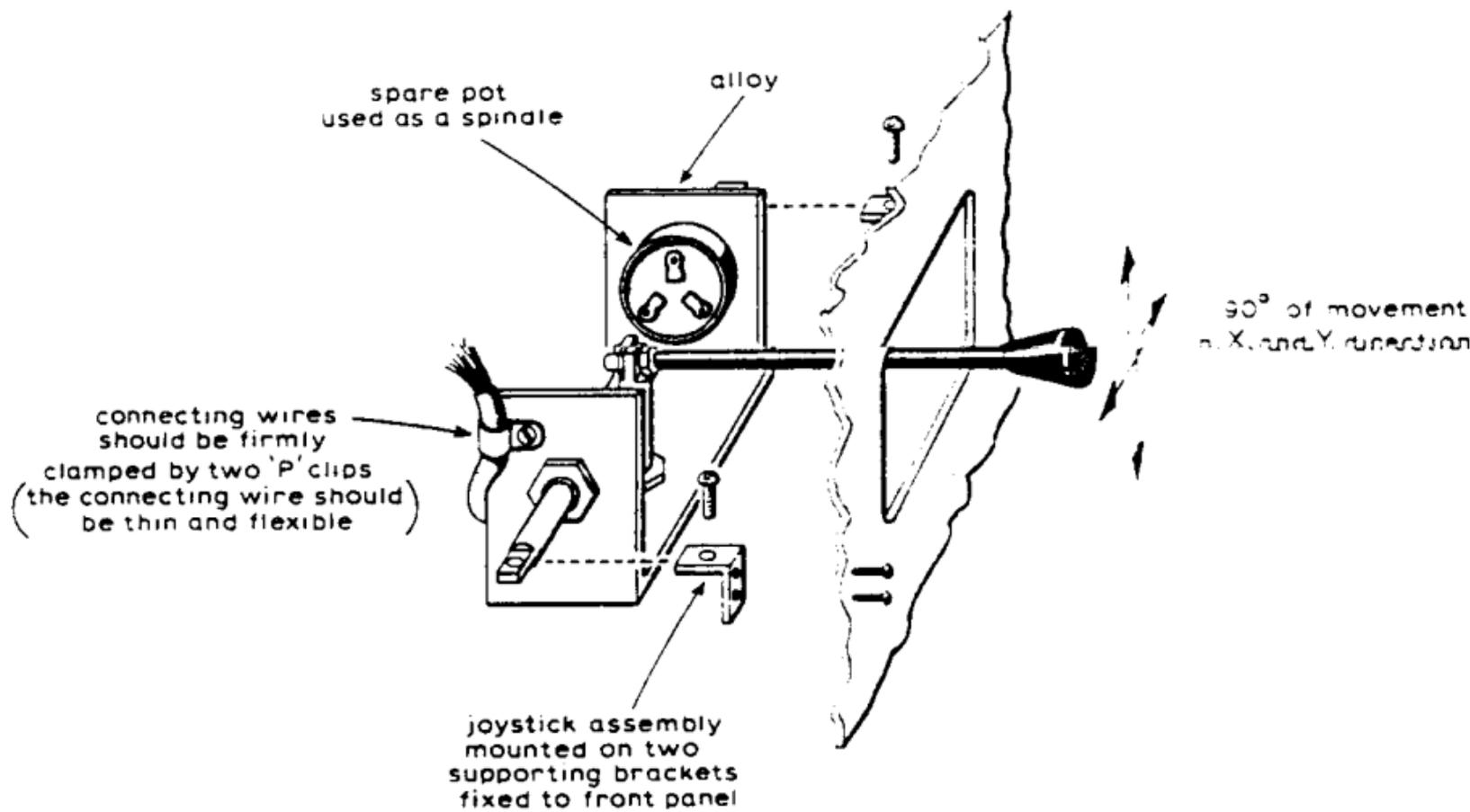
Fig. 32. Mechanical assembly of the joystick control.
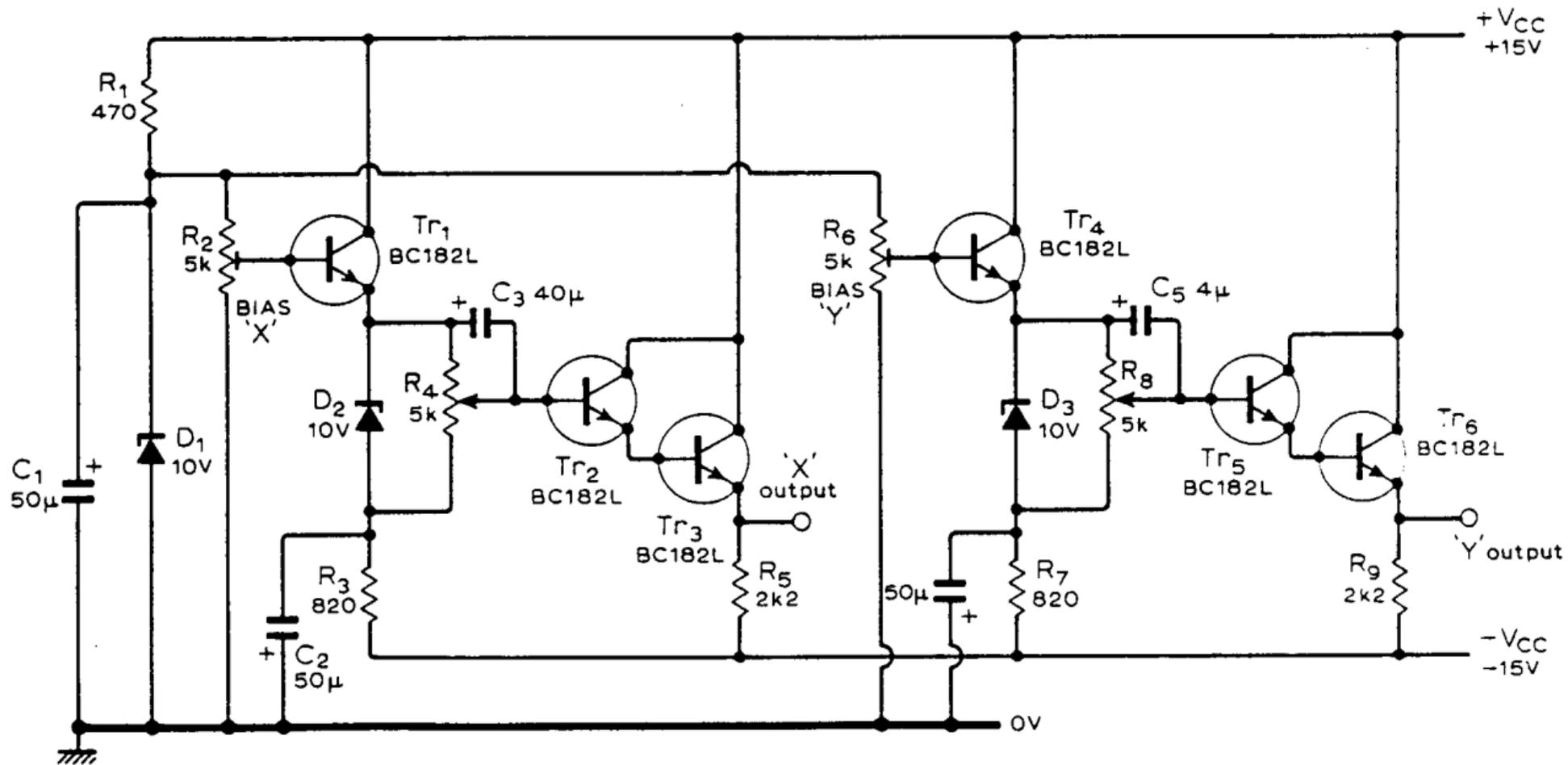
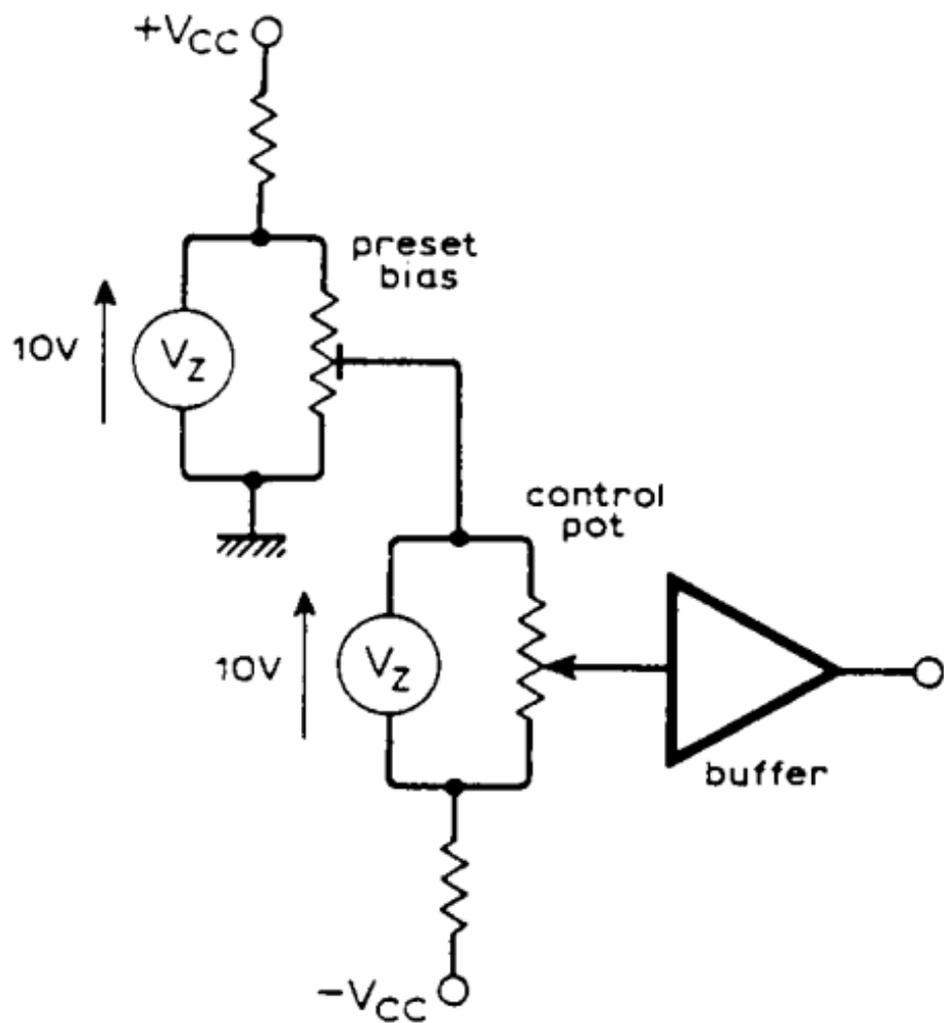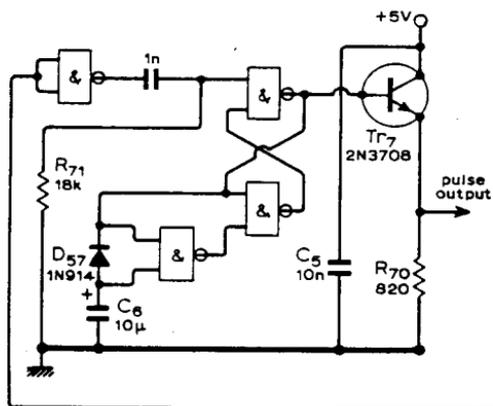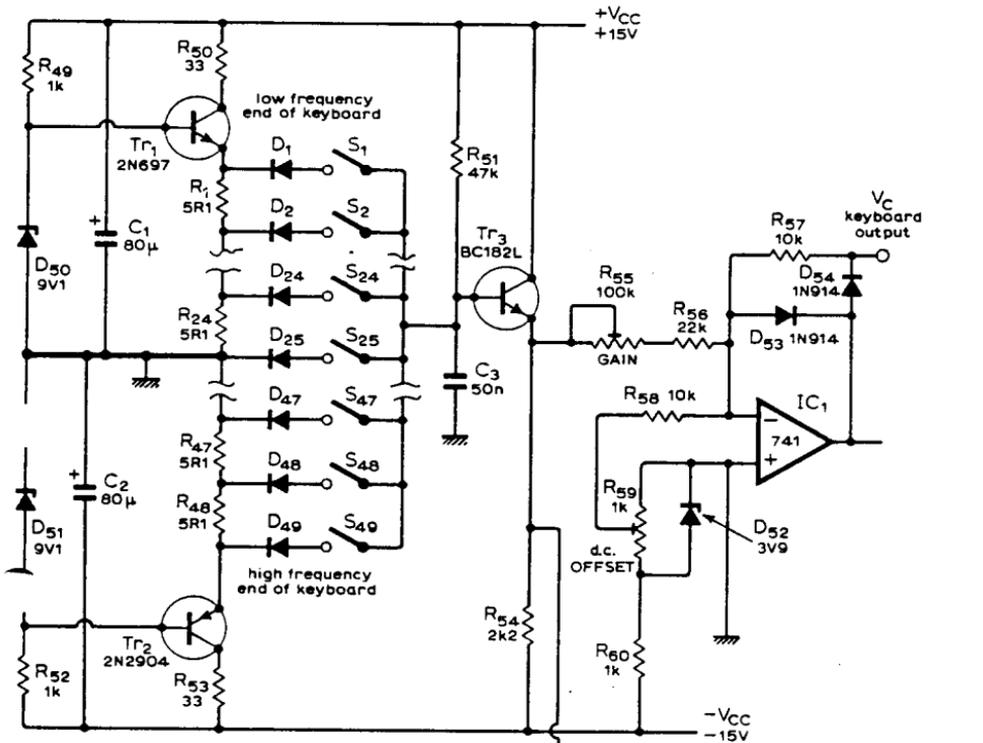*Fig. 33. Circuitry associated with the joystick control. Resistors are 5%, ¼W.*
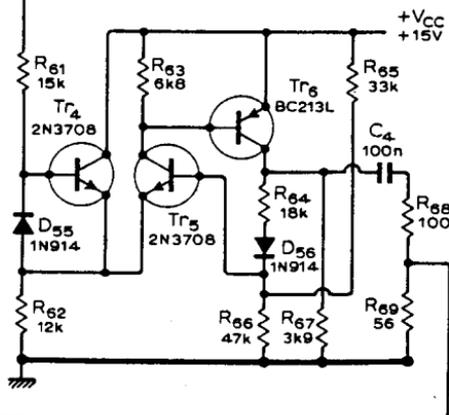
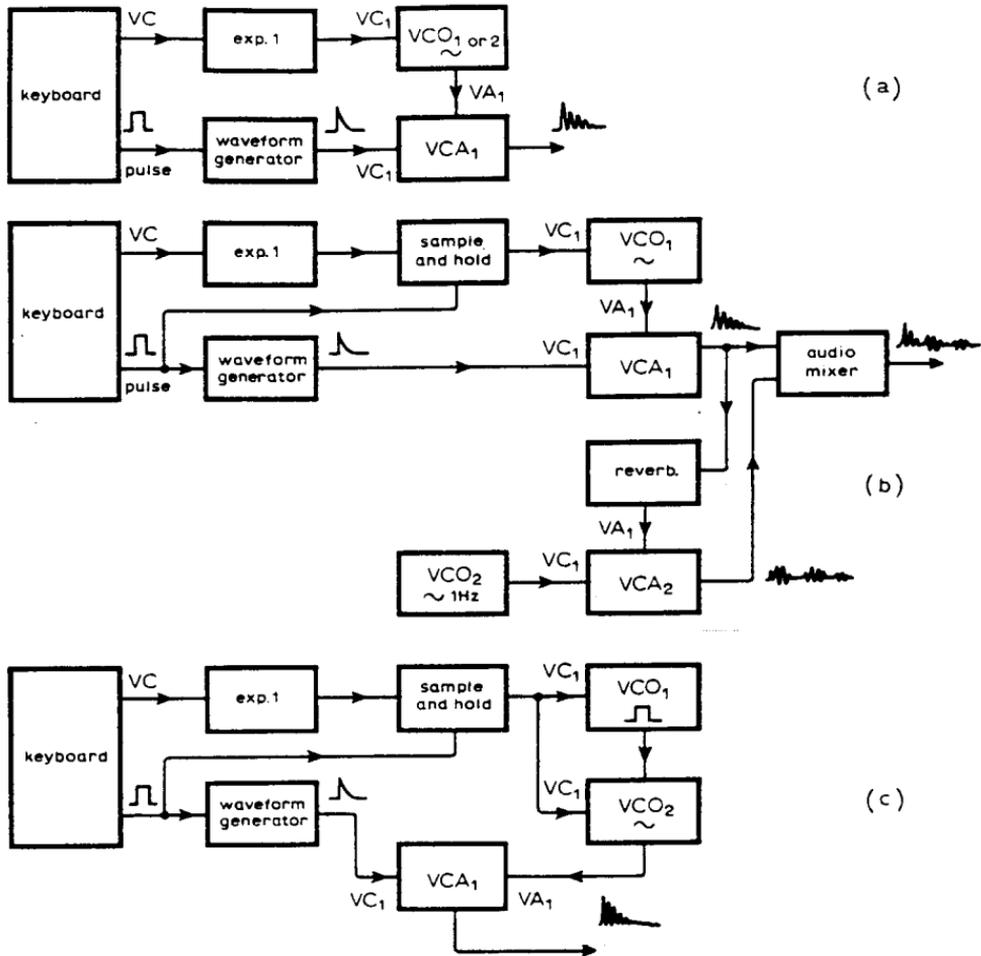*Fig. 34. Illustration of the joystick control circuit function.*

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Fig. 36. Three examples of how the patch-board can be programmed for a particular sound synthesis. See text for explanation.
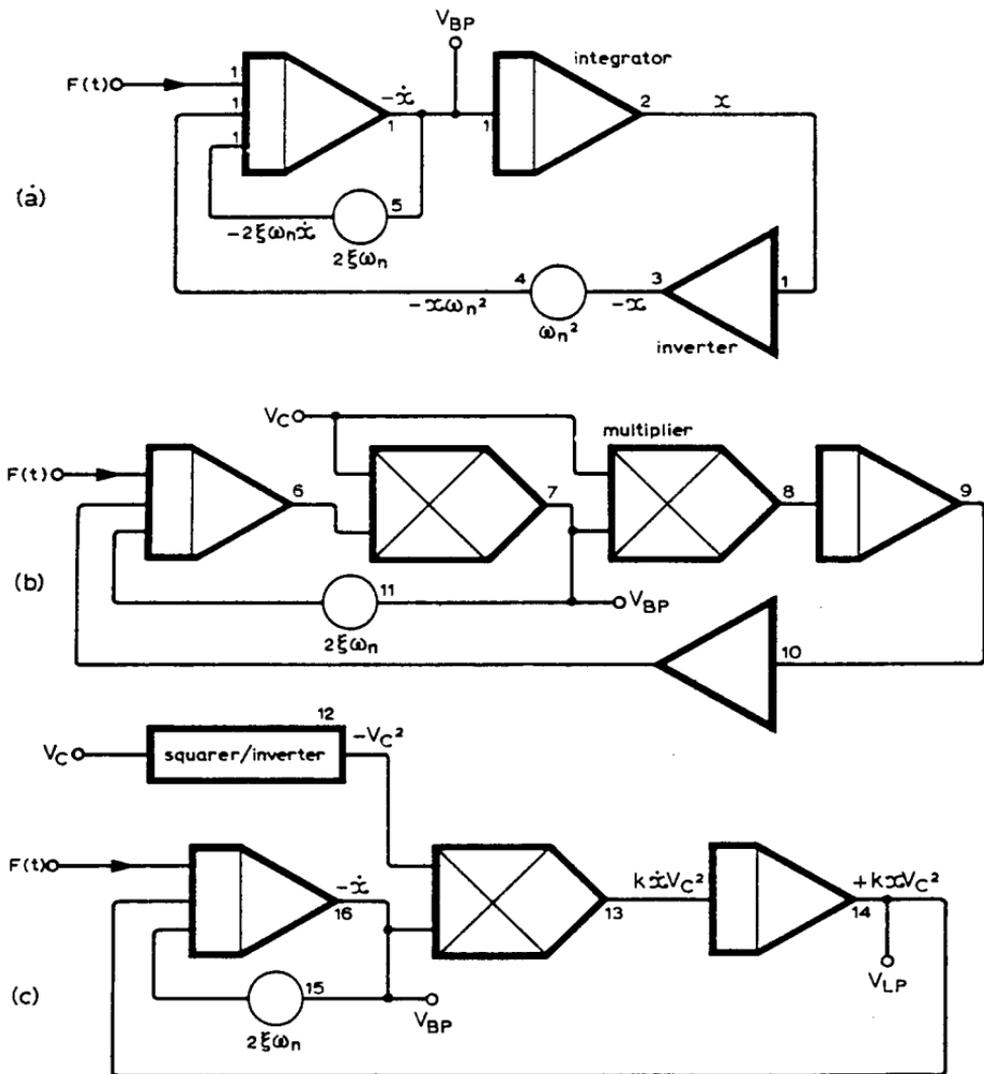
Fig. 37. Power supply circuit. Note the use of the suppressor capacitor C_1 to reduce any impulsive interference present on the mains. Resistors are 5%, $\frac{1}{4}W$ except R_{10} and R_{12} which are 1W. Heat sinks are required by Tr_6 (0.6W dissipation) and Tr_5 and Tr_7 (both 14W dissipation). D_{1-4} is a bridge rectifier 100 p.i.v. at 2A.

Fig. 38. Programmes for solving the second order differential equations of the form
$F(t) = \ddot{x} + 2\varsigma\omega_n\dot{x} + \omega_n^2 x$ where $Q = 1/2\varsigma$ (see appendix).