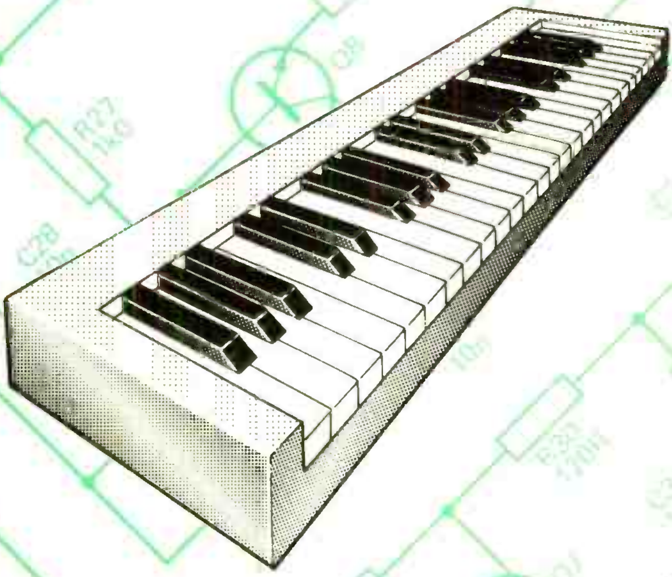
Electronic Synthesiser Construction

R.A. PENFOLD



ELECTRONIC SYNTHESISER CONSTRUCTION

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by

R.A. PENFOLD

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PREFACE

The playing of music at home is a pastime which has undergone a resurgence in recent years, and one reason for this must be the availability of some excellent electronic instruments at quite modest prices. Electronic organs and portable keyboards have led the way, but the immense and ever increasing versatility of synthesisers has resulted in a steady growth in their popularity. They have moved on from being something largely regarded as suitable only for those interested in the more avant-garde side of music to being something that has achieved general acceptance.

The term "synthesiser" probably conjours up images of sophisticated items of ready made equipment, but it is a field which was originally the domain of the do-it-yourself musician, and it is one where this approach is still a valid one. Building your own instruments, apart from being an interesting and rewarding hobby in its own right, can provide very worthwhile savings in cost. It also gives a good insight into the way in which synthesisers operate, which in turn helps the user to fully utilize the equipment and not to overlook any possibilities. The home constructor also has the possibility of experimenting with set-ups, and generating sounds, which are not available using anything but the most expensive of ready-made equipment.

This book describes a number of circuits which can be built up into a quite sophisticated synthesiser, or they can be used to extend the capabilities of an existing instrument. The individual circuits are all reasonably simple, and as printed circuit designs are provided for many of them, they should not be beyond the capabilities of inexperienced constructors.

R. A. Penfold

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Chapter 1

SYNTHESISER OPERATION

Synthesisers were once the domain of a relatively small number of dedicated enthusiasts, and music produced by synthesisers tended to be treated as something not to be taken too seriously by the general public. Things have changed considerably over recent years, and synthesiser music is very much an everyday part of modern life, Apart from the increasingly large band of people who play synthesisers as a hobby, most people regularly hear synthesised music in the form of television and film theme music, and incidental music. It does, of course, play a large part in the world of popular music, and many synthesiser players and groups are household names.

Like so many of today's high technology pastimes, synthesisers can be something of a puzzle to the beginner, with a lot of jargon to overcome. The instruments are no more difficult to play than any other keyboard type, and with the addition of a sequencer it is possible to compose or arrange music without having any playing skills whatever. The problem is more one of obtaining the desired sounds, and the instruments can be a little difficult to understand at first. Those which have rows and rows of controls can be more than a little intimidating for the beginner. Paradoxically, those instruments which have few controls can actually be more difficult to use and understand than those having dozens of controls, as the reduction in number is usually achieved by having each one perform a variety of tasks. Also like many of today's other high technology pastimes, things are surprisingly straightforward once a few fundamentals have been grasped.

This book provides an insight into the way in which a synthesiser functions and various sounds can be produced, and a number of practical projects are provided. These projects can be fitted together to produce a useful monophonic synthesiser, enabling a beginner to build a worthwhile instrument at low cost and learn a great deal about synthesis in the process. The book should also be useful to experienced users who could usefully add some of the designs to an existing system to increase its capabilities for a very modest monetary outlay.

In this first chapter we will cover some basics of synthesiser operation, and in the next chapter a number of modules will be described. These are the basic building blocks for a synthesiser, and they can form the basis of anything from a very simple instrument to a full-featured monphonic type. Later chapters will deal with some effects units and sequencing. Printed circuit designs are provided for the main modules, but not for some of the more simple ones. These can easily be constructed on stripboard though, if and when required, and inexperienced constructors should have little difficulty in building and experimenting with the modules. It is worth emphasizing the experimental aspect, as this is really the only way to fully come to terms with sound synthesis. Books can teach you some of the fundamentals of the subject, but it is only by practical experience that you can become properly familiar with the role of each module, and how it affects the final sound.

A common question is "how does a synthesiser differ from an electronic organ?". The main difference is in the way in which the two types of instrument generate the range of notes. Early electronic organs were highly complex as they used a separate oscillator for the generation of each note, and the keyboard effectively connected each oscilllator through to the output when its respective key was operated. This is analagous to a pipe organ with its separate pipe resonator for each note. Most modern instruments use a different technique where all the notes are derived from a single oscillator operating at a very high frequency. The various notes are obtained by dividing this high frequency signal by the appropriate figures. In both cases the instruments are fully polyphonic, and the number of notes that can be generated simultaneously is limited only by the number of keys (or more realistically, the number that the player can operate at one time).

An analogue synthesiser operates on a totally different principle, and one which the block diagram of Figure 1 helps to explain. The keyboard is more than just switches operated by the keys, and it includes a circuit which provides an output voltage that is dependant on the note selected. The general scheme of things is to have an output which steadily increases as higher notes are played. Most synthesisers use a logarithmic control voltage law of 1 volt per octave, or 83.33mV (0.0833 volts) per semitone.

The keyboard circuit can simply consist of a series of resistors and a voltage source, but in some of the more advanced instruments it is based on digital circuits and a digital to analogue converter. One advantage of the digital approach is that once a

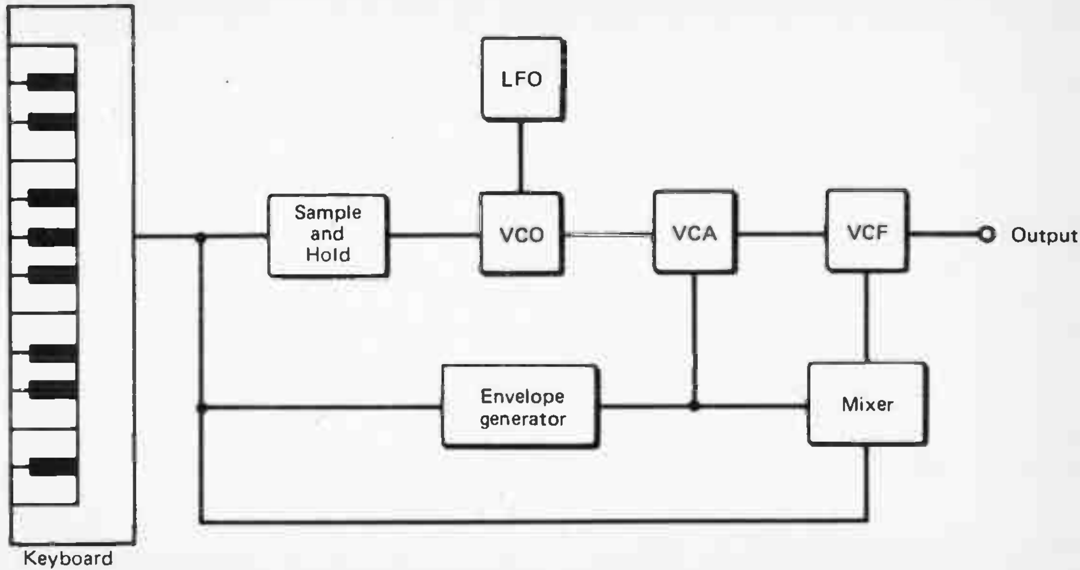


Fig. 1 Block diagram of a typical synthesiser

key has been depressed the appropriate output voltage will be maintained accurately until the next note is played, however long the delay might be. With a simple resistor network circuit the output voltage will only be maintained for as long as a key is depressed, and as soon as it is released the output voltage falls to zero. In order to overcome this the keyboard must be followed by a sample and hold circuit. This merely samples the output voltage when a key is initially depressed, and then maintains an output equal to that voltage until the next key depression is detected. Practical sample and hold circuits are not 100% effective, and can not maintain the output potential indefinitely. However, in practice it is not necessary to have a circuit that will hold the output level for weeks at a time, and the output voltage would normally only need to be held steady for a few seconds at most. Using modern components this level of performance is easily achieved.

The heart of a synthesiser is the VCO (voltage controlled oscillator), and it is this that generates the basic audio output signal. Its output frequency depends on the voltage fed to its control input, and this voltage is, of course, provided by the keyboard circuit. The VCO has a control characteristic which results in it producing an output of the correct pitch each time a key is operated. A point that should be noted here is that there is just a single oscillator, and that the instrument is consequently monophonic (i.e. it will only play one note at a time). Many synthesisers do in fact have two VCOs, but these are fed with the same control voltage and they operate together to provide a richer sounding output, and they can not be played independently from the keyboard. Typical arrangements would be to have one oscillator playing one or two octaves higher than the other, or perhaps a fifth higher than the other, usually with the two oscillators just fractionally out of sychronisation. This gives a low frequency beat note on the output and a much fuller sound.

Of course, polyphonic synthesisers are available, and most commercially produced instruments are now of this type. These consist of what is essentially a number of separate synthesisers fed from the same keyboard, with a digital circuit to channel notes to the synthesisers so that polyphonic operation is possible. Just how notes that are played are assigned to the available channels is something that depends on the particular instrument concerned, and some are much more versatile than others. Although polyphonic operation is possible, there are usually substantially fewer channels than keys, and normally only six or eight notes at a time can be played (sixteen notes with some of the more expensive instruments). Polyphonic operation is not something that will be considered further here as it is an advanced topic which goes beyond the scope of this book.

Waveforms

VCOs normally offer more than one output waveform, and the two that are usually available are square and triangular waveforms. Some additionally offer waveforms such as pulse, sawtooth, and sinewave. The waveform of the signal is of fundamental importance as it is this that probably has more influence on the final sound than any other parameter. A signal which contains just one frequency has the sinewave shape of Figure 2(a). This has a very distinctive "pure" sound, but is not greatly used in electronic music as it is generally considered to be a sound that quickly becomes boring. All other repetitive waveforms consist of two or more frequencies in the form of the fundamental frequency plus harmonics of this frequency. Harmonics are merely signals at multiples of the fundamental frequency. For example, a note at 220Hz (the 'A' below middle C) would have harmonics at 440Hz, 660Hz, 880Hz, etc. It is the particular harmonics present and their relative strengths which determines the waveshape, and of more importance, which determines the sound of the signal.

A triangular waveform (Figure 2(b)) has relatively few harmonics, and those that are present are not particularly strong. This gives a slightly harsher sound than a sinewave signal, but it is far less harsh than the sound of a sawtooth waveform (Figure 2(c)). This is in turn less harsh sounding than the squarewave of Figure 2(d). Pulse waveforms are the most harsh sounding of all, and actually have more harmonic content than fundamental signal (Figure 2(e)).

In practice it is unusual to use a straightforward waveform such as a triangular or squarewave type, and it is normally modified in some way. This is a topic to which we will shortly return.

Most synthesisers include an LFO (low frequency oscillator) for modulation purposes. The most common use for this is to frequency modulate the VCO to give a vibrato effect. However, it can be connected in other ways to give other effects such as tremolo. So far all we have is a tone which changes pitch in sympathy with keys of the keyboard being operated. To be of any

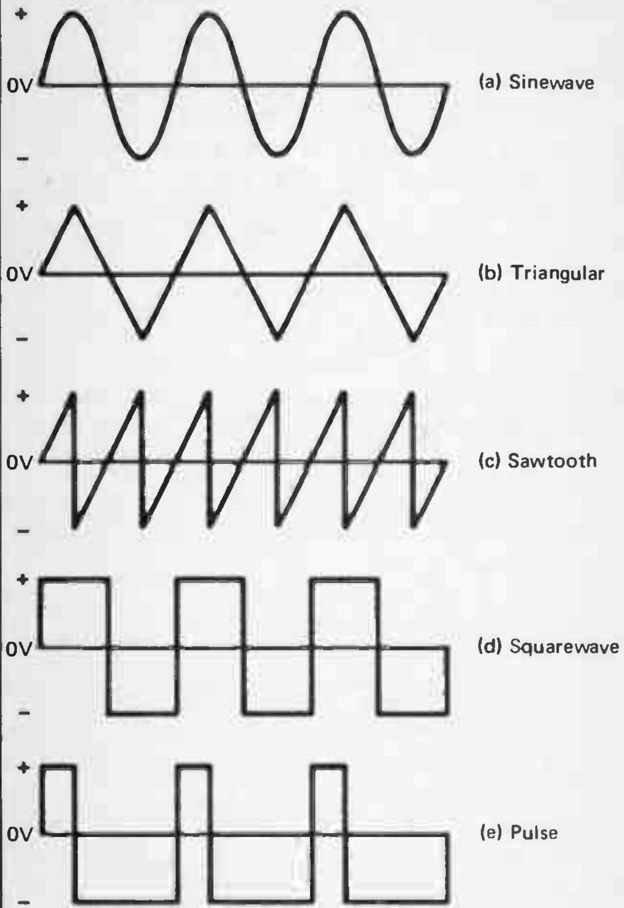


Fig. 2 Waveforms

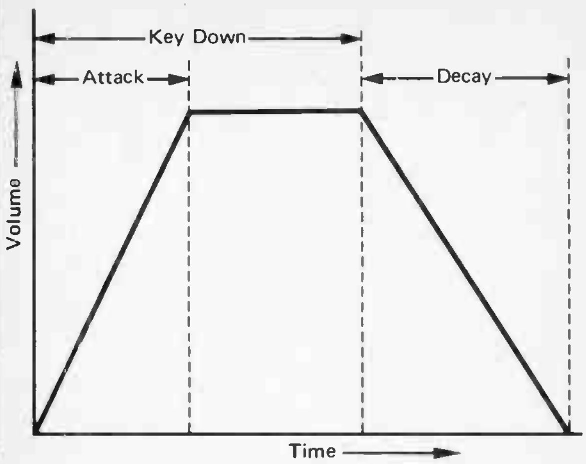


Fig. 3 Attack/Decay envelope shape

practical value some envelope shaping must be provided. Envelope shaping is merely controlling the amplitude (volume) of the output signal to give the desired effect. In its most basic form the envelope shaping simply consists of switching the signal to full volume the instant a key is operated, and cutting it off at once when the key is released. While this is very simple to implement it does not permit a very wide range of effects to be produced, and does not give particularly interesting or musical sounds.

Synthesisers incorporate a circuit known as a VCA (voltage controlled amplifier) which can set the output signal at anything from zero to maximum volume. It is controlled by means of an input voltage, and this control voltage is derived from the keyboard by way of an envelope generator circuit. The VCA and envelope generator together form what is termed an "envelope shaper". The most simple of synthesisers use an envelope generator of the attack/decay type. With this type the volume of the output signal rises steadily when a note is depressed, until the signal either reaches full volume or the key is released. If it reaches full volume it remains there until the key is released. When the key is released the signal falls back to zero amplitude. The rates at which the signal rises and falls in volume are both independently adjustable over wide limits. A typical range would be from 10 milliseconds to around 5 seconds. This gives simple envelope shapes of the kind shown in Figure 3. Although this type of envelope shaper is extremely simple, it enables some interesting effects and musically pleasing results to be obtained.

Most synthesisers use a more complex type of envelope shaper; the ADSR (attack, decay, sustain, release) type. The attack phase is much the same as for an attack/decay envelope shaper, and is adjustable over similar limits. The decay phase is somewhat different in that this section of the envelope shape is entered as soon as the signal reaches its peak value. This occurs regardless of whether or not the key is released. Another difference between this section of an A/D and an ADSR envelope is that the decay phase does not necessarily last until the signal has reached zero amplitude. The "sustain" control enables the decay section of the envelope to be terminated at any level from zero to the peak amplitude. As its name suggests, the sustain control determines the level at which the signal will be maintained for a period of time. This period lasts until the key is released, and the release phase is then entered. During this phase the signal dies back to zero amplitude. Figure 4 shows the classic ADSR envelope shape, and helps to illustrate the way in which this scheme of things operates.

In practice it is true that an ADSR envelope shaper represents a substantial increase in complexity when compared with an attack/decay type, but it is capable of producing a much wider range of sounds, including some types which are excellent for musical purposes. In particular, the classic ADSR shape of Figure 4 is one which is a feature of many acoustic instruments, including pianos. With the sustain level set quite low it is possible to produce a very "spikey" sound, like a harpsichord. Both ADSR and attack/decay envelope shapers are capable of producing envelope shapes for which there is no acoustic instrument equivalent, permitting some weird but musically very useful sounds to be generated. Sound synthesis is not just a matter of trying to mimic natural sounds and conventional instruments, but is more a matter of trying to create new sounds and use them to good effect musically.

The final stage in the synthesiser is the VCF (voltage controlled

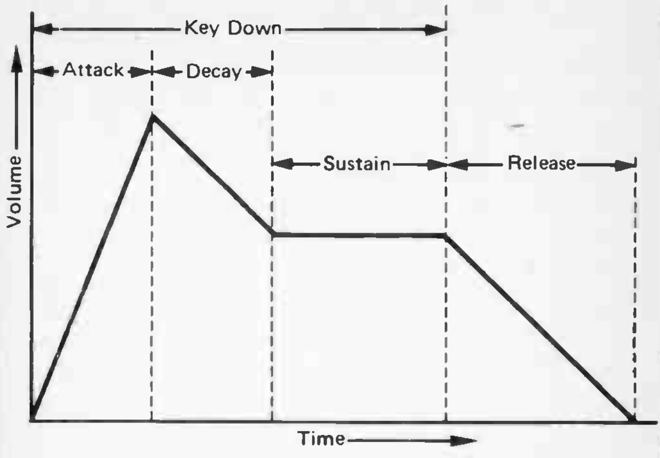
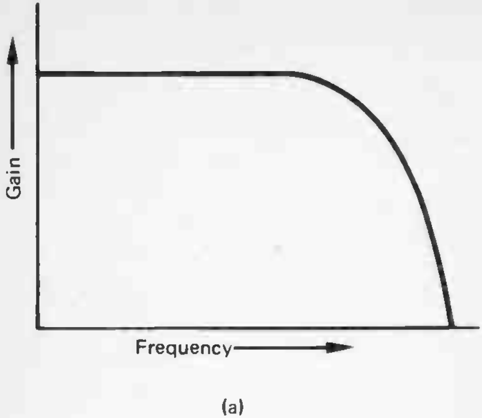


Fig. 4 ADSR envelope shape

filter). This is sometimes connected between the VCO and the VCA rather than following the VCA, but as far as the sounds generated are concerned it does not make a great deal of difference. VCFs vary a great deal in the facilities they offer and the attenuation rate they provide. With many synthesisers only lowpass filtering is available, but there is usually a "resonance" or "Q" control which enables a sort of pseudo bandpass filtering to be obtained. With simple lowpass filtering a response of the type shown in Figure 5(a) is obtained. Here frequencies up to a certain figure are allowed to pass unhindered, but above this cutoff frequency an increasingly high degree of attenuation is provided. The attenuation rate is usually either 12 or 24dB per octave. In other words, above the cutoff frequency the gain of the circuit reduces by a factor of 4 or 16 respectively for each doubling of frequency. The resonance control has the effect of placing a peak in the response just below the cutoff frequency, as in Figure 5(b). The higher the resonance setting, the more pronounced the peak.



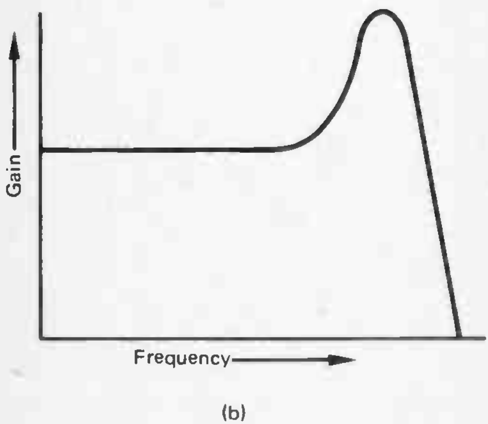


Fig. 5 (a) LPF response (b) with resonance

With some synthesisers the resonance level can be advanced beyond the point at which the VCF breaks into oscillation.

The control voltage for the VCF is derived from three sources. One of these is the front panel "filter frequency" control, and this merely enables the cutoff frequency to be manually set at the required figure. The second source is the keyboard, and this is an important feature. The purpose of the filtering is to modify the harmonic content of the input signal, and what this usually means is removing some of the higher frequency harmonics. However, with the aid of the resonance control it is possible to boost certain harmonics as well. There is a problem here in that setting the filter frequency to give the right sound with a high-pitched note being played will not give the same sound when a lower pitch is being played. This is simply because the frequency of the fundamental and harmonic signals reduces as lower pitches are selected, resulting in unwanted harmonics being taken below the cutoff frequency of the filter. This deadening of the sound as the pitch increases might sometimes be what you need, but in most cases what is required is for the sound to stay the same over the full range of the keyboard, or to change only in a quite subtle manner.

This can be achieved by mixing some of the keyboard voltage into the VCF input. In fact with correct adjustment the filter can be made to precisely track up and down in frequency exactly matching the VCO. This is useful if (say) you wish to filter a triangular waveform to produce a reasonable sinewave output. A good sinewave output can only be achieved with the filter accurately tracking the VCO over the full keyboard range.

The third control voltage source is the envelope shaper. Many natural sounds have a stong harmonic content initially, but as the signal dies away the relative harmonic content diminishes. The same result can be obtained by introducing the envelope voltage to the control input of the VCF, so that the filter's cutoff frequency reduces slightly as the envelope voltage (and amplitude of the output signal) diminishes.

Another common use of this feature is to give very interesting sounds by setting the VCF for a high resonance setting and using a large envelope content on the VCF's control voltage. In conjunction with a VCO waveform that is rich in harmonics, this sweeps the peak in the response down through the harmonics giving a sort of waa-waa effect. It has been assumed here that the envelope voltage is obtained from the same envelope generator that controls the VCA, but this is not always the case, and in many designs the filter has its own envelope generator.

Some VCF's offer alternative types of response such as bandpass, highpass, and notch filtering. A bandpass response gives an effect which is similar to a lowpass type with the resonance control advanced, but with low frequencies and not just the high frequencies attenuated. Highpass filtering gives a rather harsh effect with the fundamental and possibly the lower harmonics being attenuated, and the higher harmonics being passed through to the output. Notch filtering attenuates a narrow band of frequencies but lets other frequencies pass unattenuated. A fixed frequency notch filter tends to be largely unnoticeable, and only has a significant effect if it removes the fundamental or an important harmonic. It is more effective if it is swept by the envelope voltage, and it then gives a sort of simple phasing type effect.

Although this is a rather brief description of a very complex piece of equipment, if you have been able to follow everything so far then looking at the front panel of a conventional analogue synthesiser you should find the rows of controls easy to understand, with things such as VCO waveform selection, and envelope controls for attack, decay, and release times, plus the sustain level. You should certainly have no real difficulty in connecting up the modules described in the next chapter and setting them up to give a range of interesting sounds. However, as pointed out earlier, it is really a matter of experimenting with a synthesiser until you are able to relate control settings to their sounds, and this is not something that can be learnt from a book.

Chapter 2

THE MODULES

The basis of any analogue synthesiser is its VCO (or VCOs), and this is the module that we will consider first. The circuit diagram of the VCO appears in Figure 6.

IC1 is an LM13600N dual transconductance amplifier (or the virtually identical LM13700N), and devices of this type are much used in electronic music. What makes transconductance amplifiers so useful is their ability to operate as a sort of voltage controlled resistance, a feature which makes them suitable for use in VCOs, VCAs, VCFs, and certain types of modulator circuit. In this case the two amplifiers are connected in an arrangement that is similar to a conventional triangular/squarewave oscillator using a Miller Integrator and a Schmitt Trigger. IC1a operates as the integrator while IC1b is the trigger circuit. The triangular waveform is available from IC1a and IC1b provides the squarewave output.

Although this circuit has been described as a VCO, this is not really a very accurate description in that transconductance amplifiers are current rather than voltage operated. It is therefore the control current fed to the amplifier bias input (pin 1) which controls the operating frequency, and not the voltage applied here. In order to obtain voltage operation it is merely necessary to add a resistor in series with the bias input, as the current flow is then roughly proportional to the applied voltage.

This is not a suitable solution in this case as it would result in a VCO having a reasonably linear voltage/frequency characteristic, whereas what is needed is a type having a logarithmic control law. In other words, we require a circuit where the pitch is raised in octave steps by control voltages of 1 volt, 2 volts, 3 volts, 4 volts, etc. With a linear control law the control voltage must be doubled in order to provide each octave increment. Designing a VCO which has a logarithmic control law would be a difficult task, and the normal approach is to use a linear VCO fed from a logarithmic to linear converter. While it might seem easier to just design the keyboard circuit to suit a linear VCO and eliminate the need for what is a difficult form of voltage conversion, this is not really the case. The logarithmic law enables a very simple keyboard circuit based on readily available components to be adopted, whereas

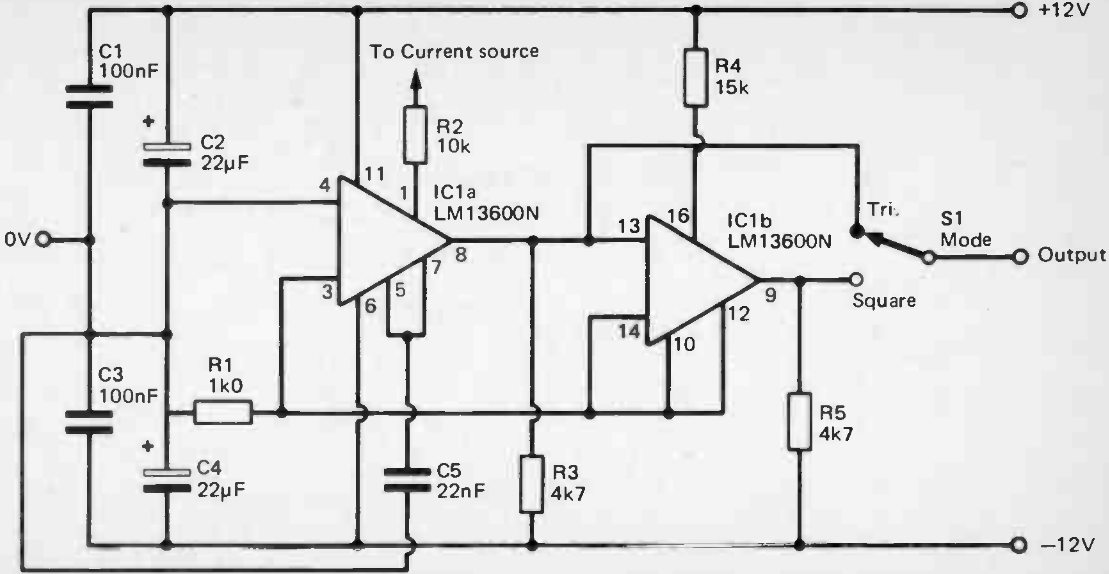


Fig. 6 The circuit diagram of the Linear VCO

with a linear characteristic it would probably be impossible to obtain suitable precision resistors. This would make it necessary to tune each note of the keyboard individually. It would also make digital control of the circuit a more difficult proposition.

The circuit diagram of a suitable logarithmic to linear converter appears in Figure 7. This is also a voltage to current converter, and the VCO remains a current rather than a true voltage controlled type despite the inclusion of series resistor R2 at its input. R2 is only a current limiting resistor to protect IC1a against an excessive input current, and it plays no active role in the VCO.

This converter circuit, in common with most other types of logarithmic amplifier, relies for its operation on the fact that the current through a forward biased silicon diode rises exponentially with linear increments in the input voltage. Although this characteristic is only maintained over certain limits, in this application there is no need to exceed these limits as only a modest range of output currents is involved (no more than a range of about 100 to 1). What does complicate things is that the voltage across a forward biased silicon diode varies significantly with changes in temperature, and diodes are often used as electronic temperature sensors. The circuit must therefore include temperature compensation to avoid the need for very frequent readjustment to correct tuning drift.

Having tried a variety of configurations, this one seems to give reasonably accurate and stable results without the need for any "difficult to obtain" components. IC2 is a CA3046 transistor array, which consists of three individual transistors plus two others connected as a long-tailed pair (i.e. having their emitters connected together). In this circuit only two of the individual transistors are used, and no connections are made to the other devices. IC2a operates as the converter, while IC2b provides a degree of temperature compenstation. The point of using a transistor array to provide these two devices is that this ensures excellent thermal contact between the two, and consequently gives instant and accurate temperature compensation. The alternative of using two ordinary silicon npn transistors with their cases glued together seems to work reasonably well, but the additional expense of using a transistor array is probably justified.

The input voltage range is far too large to directly drive IC2a, and a potential divider to provide a suitable degree of attenuation is therefore included at the input of the circuit. RV1 is adjusted to give the required 1 volt per octave characteristic. RV2 is the

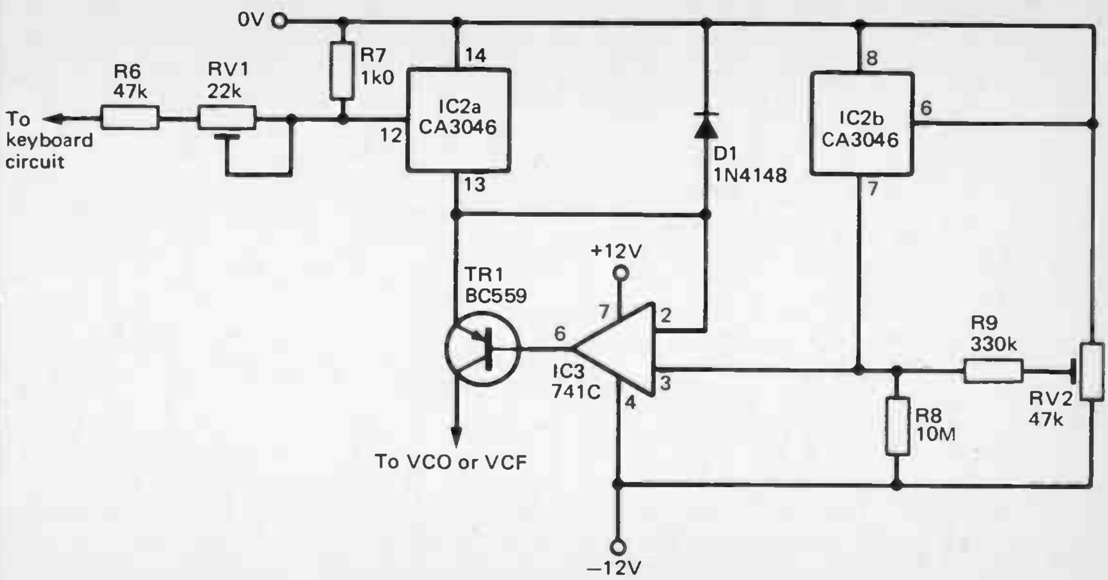


Fig. 7 The circuit diagram of the Logarithmic Converter

frequency control, and this enables the output current of the circuit to be adjusted. In practice this acts as the tuning control, and it is adjusted to give the required pitch range from the VCO. It provides a large control range, and enables the pitch range to be shifted over at least three octaves. A more detailed description of setting up RV1 and RV2 will be provided later.

VCO Components (Fig. 6)

Resistors (all 1/4 watt 5%)

R1	1k
R2	10k
R3,5	4k7
R4	15k

Capacitors

C1,3	100nF ceramic
C2,4	22μF 25V radial electrolytic
C5	22nF miniature polyester

Semiconductors

IC1	LM13600N or LM13700N

Miscellaneous

S1	SPDT miniature toggle switch

Printed circuit board, wire, etc.

Log/Lin Converter Components (Fig. 7)

Resistors (all 1/4 watt 5%)

R6	47k
R7	1k
R8	10M
R9	330k

Potentiometers

RV1	22k miniature horizontal preset
RV2	47k miniature horizontal preset

Semiconductors

IC2	CA3046
IC3	741C
Tr1	BC559
D1	1N4148

Keyboard Circuit

Keyboard circuits can be quite complex and difficult to set up, but good results can be obtained using a very basic type such as the one shown in the circuit diagram of Figure 8.

The first requirement is for a potential divider to generate all the voltages for the required notes. These voltages are provided by RV3 plus R10 to R17, but only eight fixed resistors are shown in Figure 8 for the sake of clarity. In practice this resistor chain must include one less fixed resistor than there are notes on the keyboard. Most keyboards available to the home constructor seem to be four or five octave types, which require 48 and 60 resistors respectively. The use of a logarithmic VCO means that all the resistors must be of the same value, and there is no need to use any unusual and difficult to find values. In fact the exact value used is not critical, but it must not be too low or the resistor chain will draw a high supply current. On the other hand, using a high value could result in the keyboard circuit being unable to drive the subsequent circuitry properly, and could also give problems with excessive stray pick up of mains "hum" or other electrical noise. A value of 47 ohms seems to be a good compromise. Ideally the keyboard resistors should have a tolerance of 0.5% or better, but suitable components might prove to be difficult to obtain, and very expensive if a suitable source can be located. Good results should be obtained using resistors having a tolerance of 1%, and these are readily available at quite low cost.

RV3 is adjusted so that there is 83.33 millivolts across each resistor, or 1 volt per 12 resistors in other words. This adjustment does not need to be particularly accurate if the synthesiser is to be used on its own, and RV3 can then be given any setting which enables the VCO to be set up for the correct pitch range. Greater accuracy is needed if the unit is to be used with other synthesisers (which must be of the 1 volt per octave type). In the absence of suitable test gear it is still possible to give RV3 the correct adjustment. First use the keyboard voltage from the other synthesiser to drive the VCO, and set up the VCO for the correct pitch range.

The sample and hold circuit has IC6 as a very high input impedance buffer stage. This is required in order to provide a sufficiently low output voltage to drive one or more VCOs, while providing a high enough input impedance to ensure that there is no significant loading on charge storage capacitor C12. The input impedance of IC6 is about one million megohms, and in theory

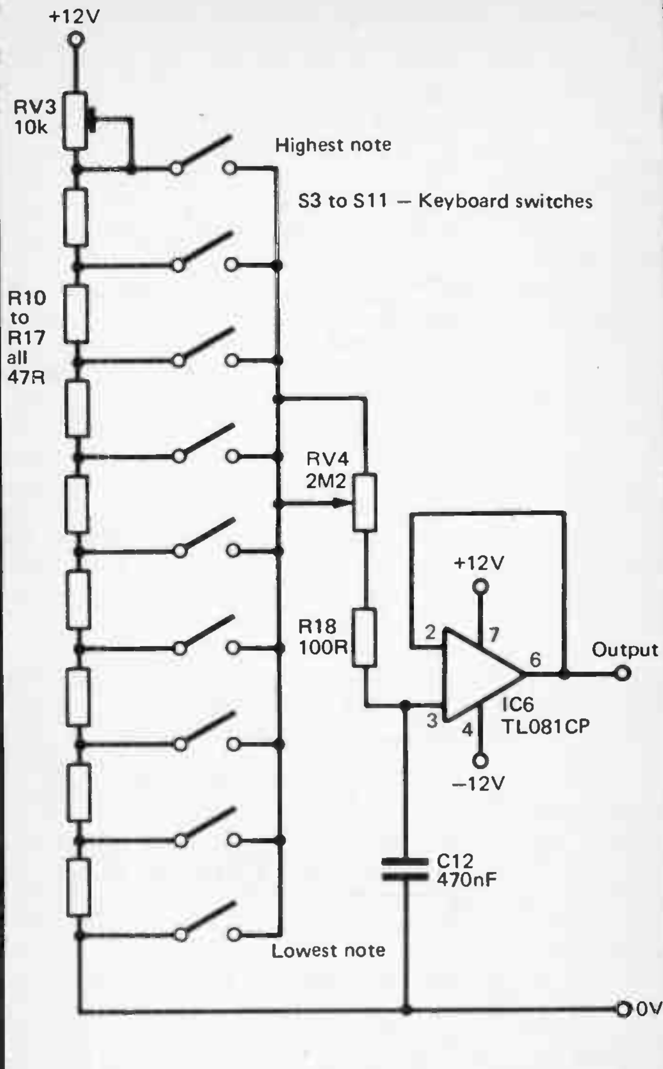


Fig. 8 A simple keyboard circuit with portamento

there should be no significant reduction in the charge on C12 over a long period of time. In reality things are less certain, as the leakage resistance of C12 itself plus any leakage through the printed circuit board also have to be taken into account. However, provided C12 is a good quality type such as a carbonate or polyester component, there should be no significant drop in the charge voltage over a period of at least several seconds.

If we assume that RV4 is at minimum resistance, when any keyboard switch is operated C12 is almost instantly charged to the new keyboard voltage, and it retains that charge after the key is released. Adjusting RV4 for increased resistance gives much the same effect, but it takes C12 longer to adjust to each new voltage. This can be used to give a glide from one note to the next, rather than abrupt switching from one note to another. This effect is usually termed "portamento", but it is also known as "glissando". The higher the value of RV4, the longer it takes the synthesiser to move on to new notes. However, beware of using an excessive glide time as this could result in the desired notes never being reached.

With its relative simplicity this keyboard circuit inevitably has a shortcoming. With more sophisticated keyboard circuits the lowest note is normally the one that is obtained if more than one key at a time is depressed. With this circuit you will get a more or less random note if more than one key at a time is operated. Of course, this is not really a major drawback, and is perfectly satisfactory provided the keyboard is played reasonably crisply. Certainly no real problems were encountered when playing the prototype.

Keyboard Components (Fig. 8)

Resistors

R10 to R17	47R (1% or better, quantity to suit keyboard)
R18	100R (¼ watt 5%)

Potentiometers

RV3	10k miniature horizontal preset
RV4	2M2 linear

Capacitors

C12	470nF miniature polyester

Semiconductors
IC6 TL081CP or LF351

Miscellaneous
Printed circuit board
Keyboard with materials for key contacts
Wire, control knob, etc.

Stylus Keyboard

One advantage of this ultra-simple keyboard arrangement is that it can easily be applied to a stylus type keyboard. This is not a keyboard at all in the true sense, but is a printed circuit board etched with a miniature keyboard type pattern. Touching a stylus of some kind onto one of the "keys" completes an electrical contact and plays the required note. This type of keyboard is more difficult to use than a conventional type, but it does have a very large advantage in that it can be constructed for only a small fraction of the cost of a real keyboard.

Figure 9 shows the stylus keyboard circuit, and this only differs from the original in that the keyboard switches have been replaced by the keyboard pads and the stylus. The latter can be something like a test prod or a 4 millimetre plug. Keep the keyboard pads well cleaned so that a good electrical contact can easily be made to them.

VCA Circuit

The circuit diagram of the VCA appears in Figure 10. This is based on one of the transconductance amplifiers in an LM13600N or LM13700N (IC7), and unlike the previous circuits this one operates from a single 12 volt positive supply rather than dual balanced 12 volt supplies. R19, R20, and C13 are used to provide a centre tap on the supply for biasing purposes. The VCA is an entirely conventional type which makes use of the linearising diodes at the input of the amplifier. A bias current is fed to these by R25, and this gives improved distortion performance from the circuit. The output impedance of the transconductance amplifier is quite high, but IC7 includes an output buffer amplifier in the form of a Darlington Pair emitter follower stage, and this provides the circuit as a whole with a low output impedance. R26 is the discrete load resistor for this stage.

R27 is included in series with the amplifier bias input of IC7 to provide a conversion from current to voltage control. The VCA

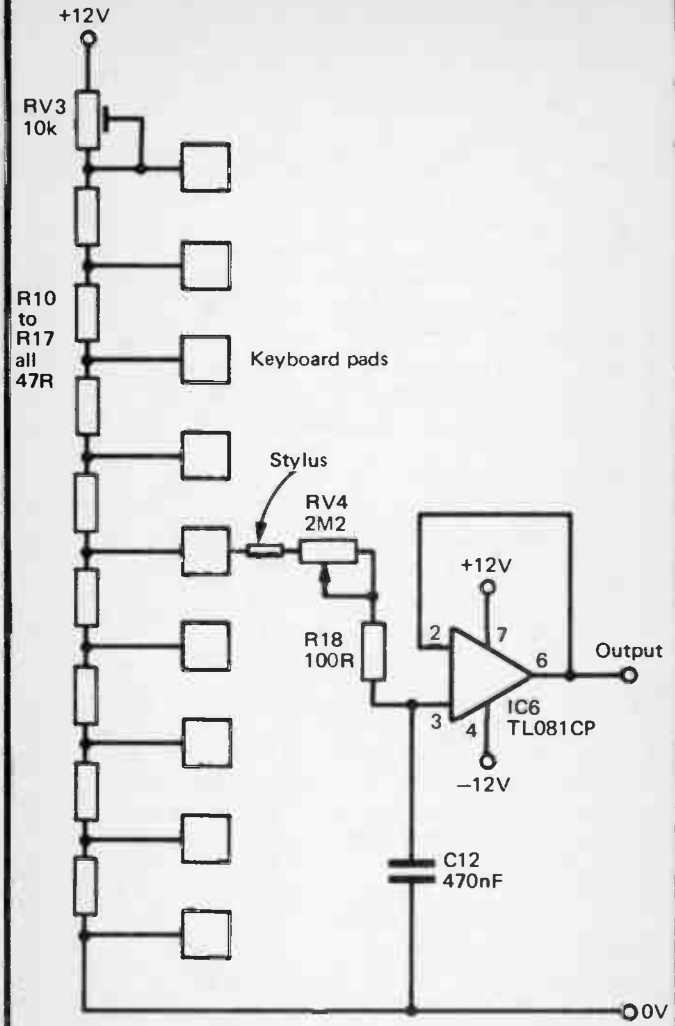


Fig. 9 A simple stylus keyboard circuit

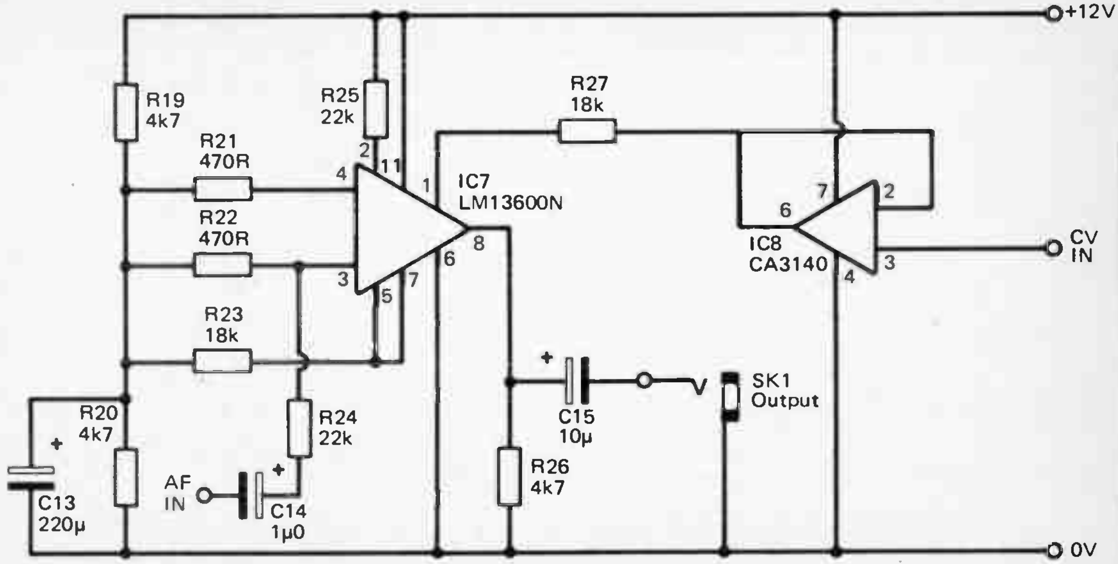


Fig. 10 The circuit diagram of the VCA

has a linear rather than a logarithmic control characteristic, but this is perfectly satisfactory and there is no need to add a logarithmic converter ahead of the control input. What is needed here is a buffer amplifier to provide a very high input impedance at the control input of the unit. This is necessary because of the very high output impedance of the envelope generator circuit.

VCA Components (Fig. 10)

Resistors (all ¼ watt 5%)

R19,20,26	4k7
R21,22	470R
R23,27	18k
R24,25	22k

Capacitors

C13	220μF 10V radial electrolytic
C14	1μF 63V radial electrolytic
C15	10μF 25V radial electrolytic

Semiconductors

IC7	LM13600N or LM13700N
IC8	CA3140

Miscellaneous

SK1	Standard jack socket
Printed circuit board	
Wire, solder, etc.	

Envelope Generators

If reduced to its most fundamental form an attack/decay envelope generator need have no more than five components plus some keyboard switches, as shown in the circuit diagram of Figure 11. This requires two sets of keyboard switches, one set which are normally open and another set which are normally closed. Note that with this circuit, and with the other envelope generators described here, the switches which activate the circuit are in addition to those used to generate the control voltage. In other words, with this envelope generator circuit a total of three switches per key must be available (two normally open and one normally closed type).

Operation of the circuit is very straightforward indeed. Normally capacitor Ca is connected to the negative supply rail via

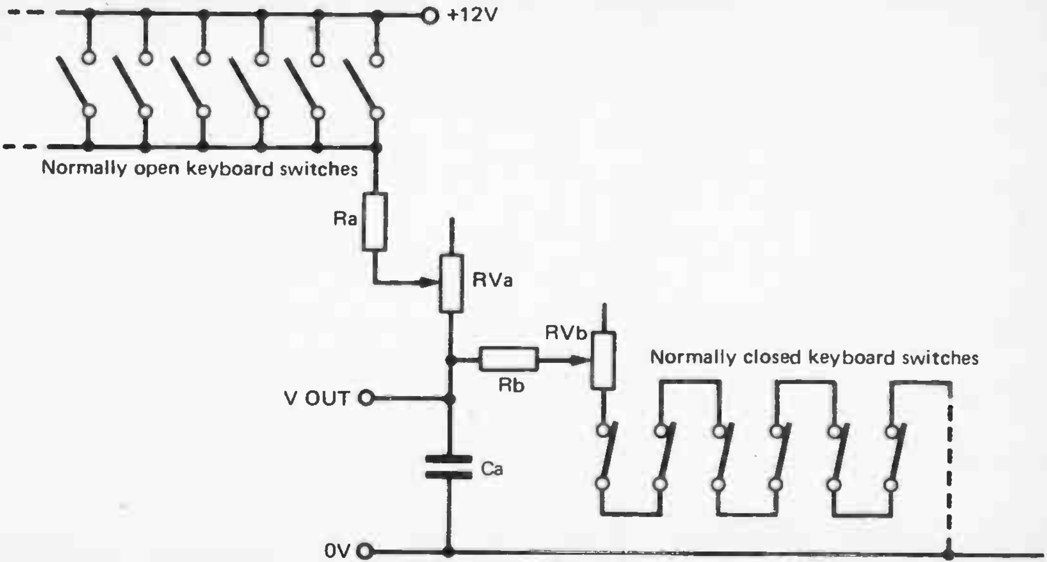


Fig. 11 The most basic type of envelope generator

Rb, RVb, and the series of normally closed switches. There is no path to the positive supply rail through Ra, RVa, and the normally open switches, as all the latter will be open. Ca is therefore held in the uncharged state. If one of the keys is operated this provides a path to the positive supply rail through Ra and RVa, and also breaks the path to earth through the normally closed switches. The voltage across Ca consequently rises as this component charges up, and it continues to do so until the full supply potential is reached or the key is released. When the key is released the circuit returns to its original state, and Ca discharges through Rb, RVb, and the normally closed switches. The charge and discharge rates are controlled using RVa and RVb respectively, and these therefore act as the attack time and decay time controls respectively.

Although having the advantage of extreme simplicity, the envelope generator of Figure 11 is not very practical in that it requires two sets of key contacts, and one set is of the normally closed variety (which can be difficult to arrange with some keyboards). The envelope generator circuit of Figure 12 is more practical as it requires just one set of normally open key contacts.

IC101 is one of the 2 input NOR gates from a CMOS 4001BE device, but here its two inputs are simply wired together so that it operates as a simple inverter. Normally R101 holds the input of IC101 low, which takes the output high. This results in Tr102 being cut off and Tr101 being biased hard into conduction. C101 is therefore held in the discharged state. If one of the keyboard switches is operated, this takes the input of IC101 high and sends the output low. It is then Tr102 that is switched on and Tr101 that is cut off. C101 then charges via Tr102, RV102, and R105. When the key is released the circuit reverts to its original state and C101 discharges by way of R104, R101, and Tr101. RV101 and RV102 therefore act respectively as the decay and attack time controls. Both times are adjustable from a few milliseconds to several seconds in duration.

Attack/Decay Envelope Generator Components (Fig. 12)

Resistors (all ¼ watt 5%)

R101	10k
R102,103	18k
R104,105	6k8

Potentiometers

RV101,102	2M2 lin

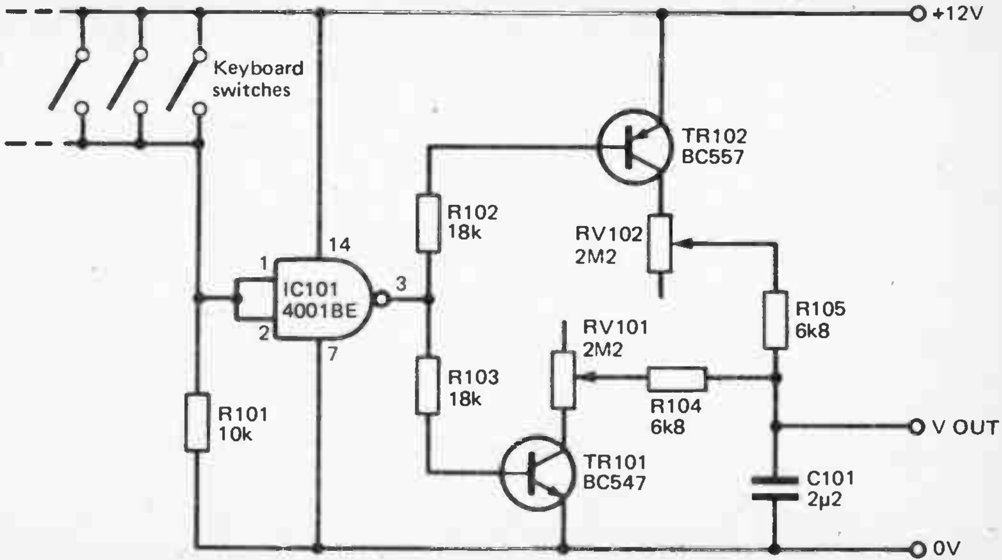


Fig. 12 A simple attack/decay envelope generator

Capacitors

C101	2μ2 miniature polyester

Semiconductors

IC101	4001BE
Tr101	BC547
Tr102	BC557

Miscellaneous
Circuit board
Two control knobs
14 pin DIL IC holder, wire etc.

ADSR Generator

Although an ADSR envelope shaper is rather more complex than an attack decay type, it does not greatly increase the cost of the synthesiser as a whole, and is probably well worth the additional expense. The block diagram of Figure 13 helps to explain the operation of the ADSR envelope generator featured here.

The flip/flop circuit operates two electronic switches, and the circuit is arranged so that when one of these is closed the other is open. This is very much like the system used in the attack/decay envelope generator described previously, and these two switches do in fact control the attack and decay phases of the envelope. Initially the attack switch is held open and the decay switch is closed, but when a key is operated this triggers a monostable multivibrator which generates a brief pulse. This pulse sets the flip/flop to its alternative state, so that the attack switch is closed and the decay switch is opened. The voltage on the storage capacitor then starts to rise, and it continues to do so until the charge potential reaches a certain threshold level. This is then detected by one of the voltage comparators which resets the flip/flop to its original state. The storage capacitor then starts to discharge via the decay potentiometer, and it continues to do so until a certain threshold level is reached. This level is detected by the second voltage comparator, and the sustain control enables the threshold level to be varied from zero to the peak envelope voltage. The voltage comparator cuts off the control signal to the decay switch, which is consequently opened, ending the decay phase.

There is a third electronic switch, the release switch, and this is operated from the keyboard switches via an inverter stage. This

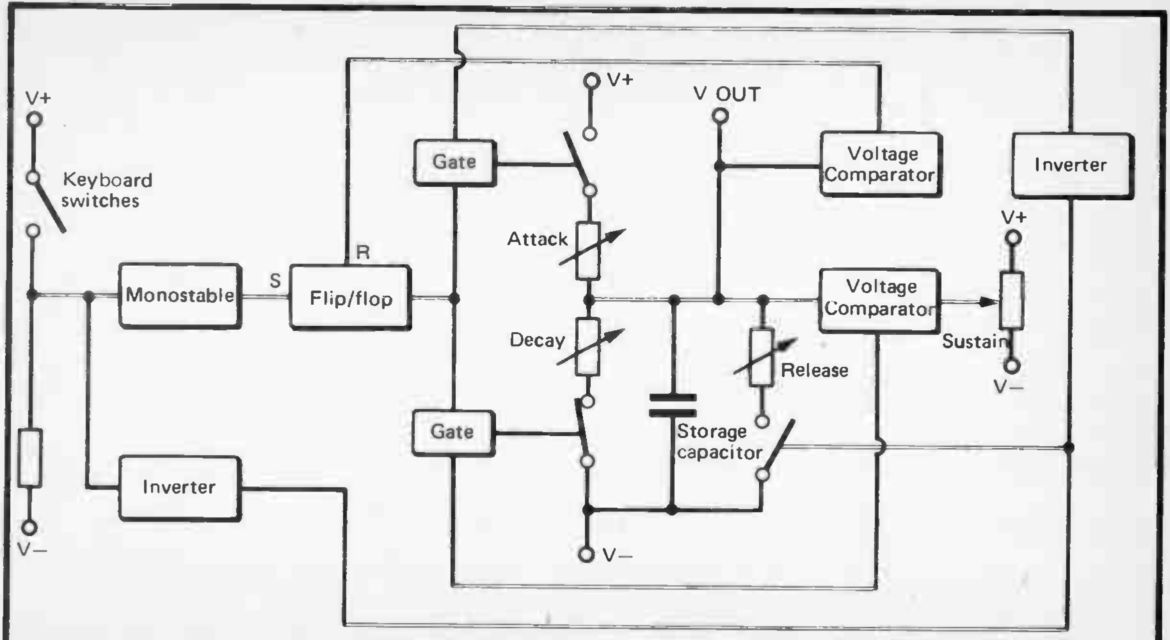


Fig. 13 Block diagram of the ADSR Envelope Shaper

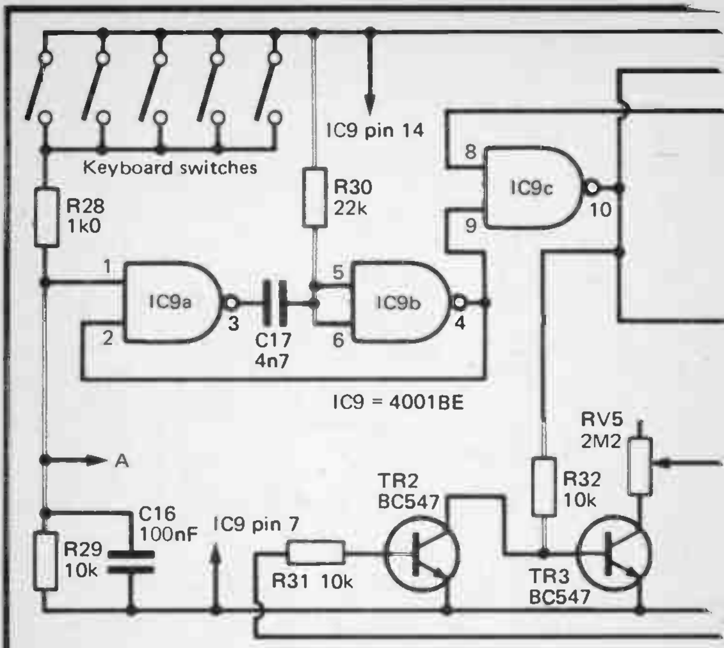
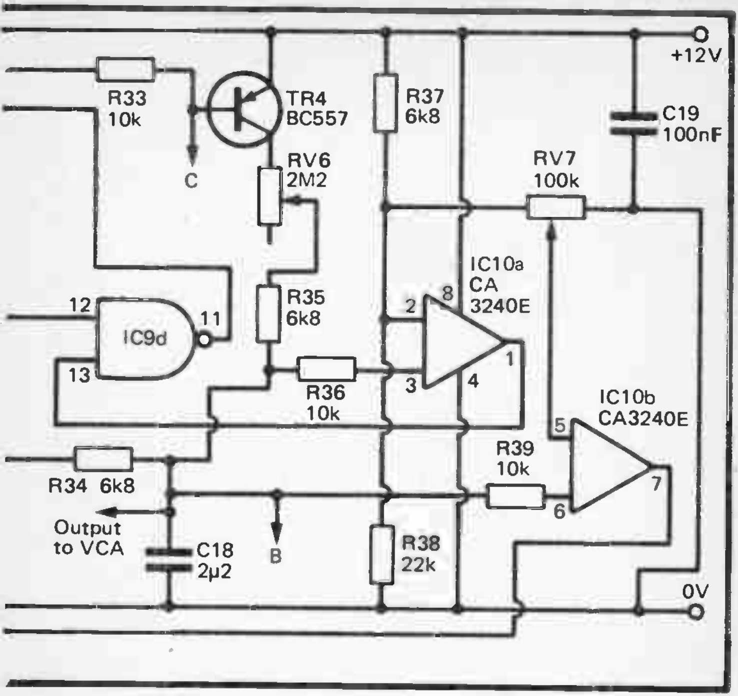


Fig. 14 The main section of the ADSR Envelope Shaper circuit

switch is therefore open when a key is operated, and closed when a key is released. Thus, when a key is released, the release phase of the envelope is commenced. A further gate and inverter stage are used to ensure that when the release switch is closed the attack switch must be open. This is necessary because there is otherwise a risk that both switches would be in the closed state if the key is released early in the envelope. This could result in the circuit hanging-up in an intermediate state, rather than immediately entering the release phase.

The main circuit of the ADSR envelope shaper is shown in Figure 14, and the rest of the circuit appears in Figure 15.

The monostable and bistable (flip/flop) circuits are each formed from two of the CMOS 2 input NOR gates of IC9. IC9a and IC9b are used in the monostable, while IC9c and IC9d form the basis of the flip/flop. Both are conventional circuit configurations. C16 helps to avoid spurious triggering of the monostable due to



contact bounce when a key is released. C18 is the charge storage capacitor, while RV5 and RV6 are respectively the decay and attack controls.

IC10 is a dual operational amplifier, but in this circuit both sections act as voltage comparators. IC10a is the one which resets the flip/flop at the end of the attack period, while IC10b ends the decay period in conjunction with Tr2. RV7 is used to set the required sustain level. Tr5 is the release switch, and it is driven from the keyboard circuit via inverter Tr6. Tr7 are Tr8 are used to inhibit the attack circuit when a key is released.

One problem with both the attack/decay and the ADSR envelope shapers is that they require an additional set of contacts on the keyboard. When using a proper keyboard there is not usually any problem in having two sets of make contacts per key, but with a stylus keyboard there are obviously no switch contacts to trigger the envelope shaper. This problem is not

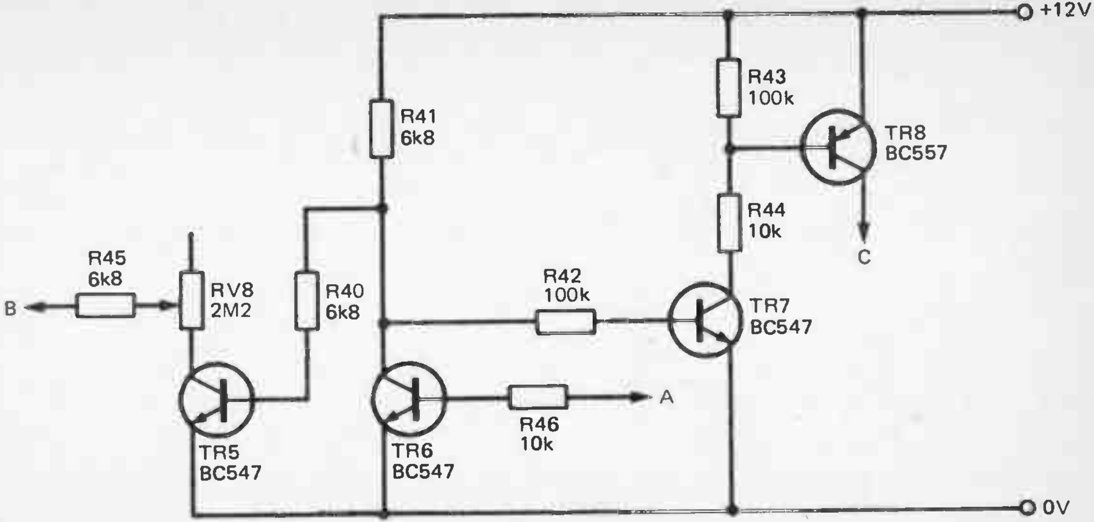


Fig. 15 The remaining part of the ADSR Envelope Generator circuit

insurmountable, and all that is needed is a simple trigger circuit of the type shown in the circuit diagram of Figure 16.

The circuit is a voltage comparator, and it compares the keyboard voltage with a low reference potential of about 120 millivolts which is supplied by R101 and R102. With no note being played, the non-inverting input of IC101 is taken low by R103, and the output of the circuit goes low. When a note is played, the input voltage is more than 120 millivolts, and the output of the circuit goes high, triggering the envelope shaper. In fact the input voltage will only exceed 120 millivolts when the third or higher notes on the keyboard are played. In practice this means that the lowest two keys should be omitted, and the keyboard should start at "D" rather than "C".

R103 must have a high value so that the circuit does not significantly load the potential divider circuit in the keyboard circuit (which would cause the higher notes to sound flat). However, this makes the circuit prone to spurious triggering due to stray pick up of electrical noise in the lead to the stylus. To avoid this the lead should be kept reasonably short, and the stylus should be a type that ensures that the player is well insulated from the metal tip (the human body picks up substantial amounts of electrical noise from the environment).

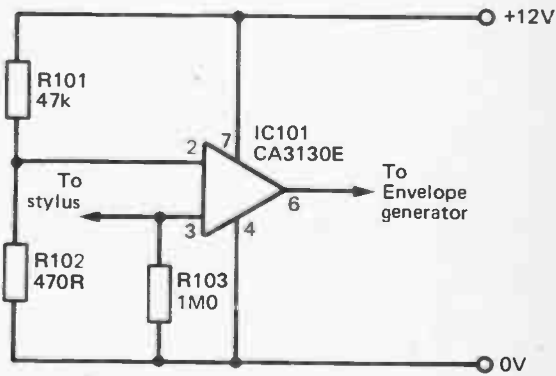


Fig. 16 Driving an envelope shaper from a stylus keyboard

ADSR Envelope Generator Components (Figs. 14 & 15)

Resistors (all ¼ watt 5%)

R28	1k
R29,31,32,33,36,39,44,46	10k
R30,38	22k
R34,35,37,40,41,45	6k8
R42,43	100k

Potentiometers

RV5,6,8	2M2 linear
RV7	100k linear

Capacitors

C16	100nF ceramic
C17	4n7 miniature polyester
C18	2µ2 miniature polyester
C19	100nF ceramic

Semiconductors

IC9	4001BE
IC10	CA3240E
Tr2,3,5,6,7	BC547
Tr4,8	BC557

Miscellaneous

Printed circuit board

Four control knobs, IC holders, wire, etc.

VCF

Figure 17 shows the circuit diagram of the VCF. Like the VCA, this should have its control input driven by way of the logarithmic to linear converter (Figure 7).

Also in common with the VCA, this circuit is based on the two transconductance amplifiers and buffer stages in an LN13600N or LM13700N. The circuit consists of two single stage lowpass filters connected in series. In the first of these IC11a acts as a voltage controlled resistor while C20 is the filter capacitor, and in the second IC11b acts as the voltage controlled resistor while C21 is the filter capacitor. Being a two stage filter the attenuation rate is 12dB per octave.

Feedback via R48 and R53 –RV9 provides a bandpass response at the output of IC11a, and also gives variable

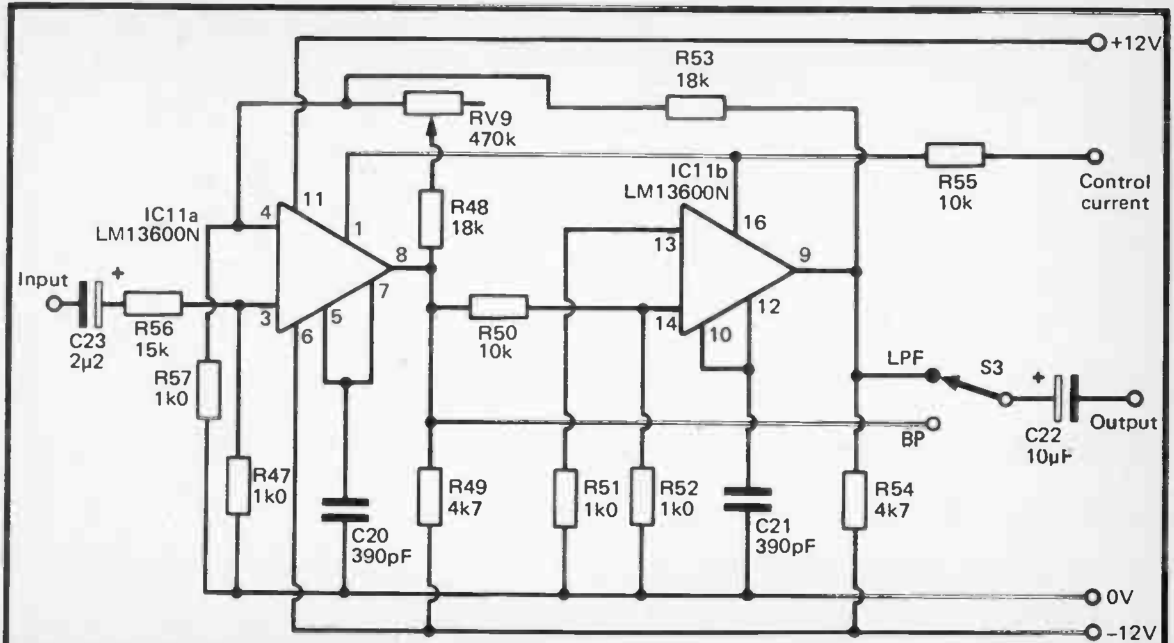


Fig. 17 The circuit diagram of a 12dB per octave VCF circuit

resonance. The resonance control can be used to both give a peak in the response just below the cutoff frequency of the filter in the lowpass mode, or to narrow the response in the bandpass mode. S3 is used to select the required type of filtering. The filter has a wide operating frequency range, and its cutoff frequency can in fact be varied over the full audio range.

VCF Components (Fig. 17)
Resistors (all ¼ watt 5%)

R47,51,52,57	1k
R48,53	18k
R49,54	4k7
R50,55	10k
R56	15k

Potentiometers

RV9	470k linear

Capacitors

C20,21	390F ceramic plate
C22	10μF 25V radial electrolytic
C23	2μ2 63V radial electrolytic

Semiconductors

IC11	LM13600N or LM13700N

Miscellaneous
Printed circuit board
Control knob, wire, solder, etc.

Power Supply Unit
To power the circuits described in this chapter a power supply capable of supplying +12 volts and −12 volts is required, and the supply must be well smoothed and stabilised. The maximum current required depends on which modules are used, and on how many of each one are incorporated into the instrument. For a simple single oscillator type the current consumption is likely to be less than 100 milliamps from both supplies, but with a two VCO type with a separate envelope shaper for the filter and some of the extra modules to be described later, the current consumption could be well over 100 milliamps per supply rail.

Battery operation is not really a very practical method of

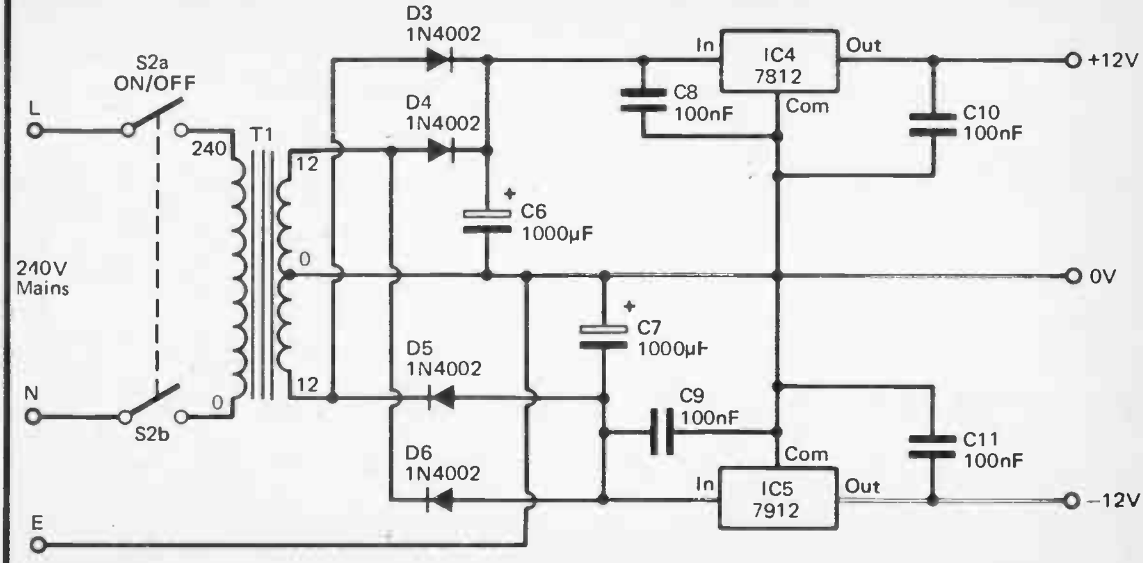


Fig. 18 The circuit diagram of the Mains Power Supply Unit

powering a unit of this type, and a mains power supply is really needed. A suitable circuit is shown in Figure 18.

This is a very straightforward design using push-pull rectification to provide positive and negative supplies. C6 and C7 smooth the positive and negative supplies respectively. IC4 and IC5 then provide electronic smoothing and regulation of the two outputs. A mains transformer (T1) having a secondary rating of 500 milliamps should be adequate to supply even an an instrument which uses all the circuits described in this chapter, including two VCOs and two envelope generators. For a simple instrument a type having a secondary current rating of 250 milliamps should be more than adequate. However, if you are likely to add to the basic synthesiser and gradually build it up into a fully fledged instrument it would be advisable to start with a 500mA component. An alternative to using a 12V – 0V – 12V transformer is to use a twin 12 volt type with the two secondaries wired in series. Mains transformers having twin secondary windings seem to be the more readily available these days.

PSU Components (Fig. 18)

Capacitors

C6,7	1000μF 25V radial electrolytic
C8,9,10,11	100nF ceramic

Semiconductors

IC4	7812 (+12V 1A regulator)
IC5	7912 (–12V 1A regulator)
D3,4,5,6	1N4002

Miscellaneous

T1	Mains primary, 12–0–12 volt 500mA secondary
S2	Rotary mains switch

Printed circuit board
Wire, control knob, solder, etc.

VCF Mixer

The modules described so far are the main building blocks of a synthesiser, but a practical instrument usually has some additional circuits. Just what additional circuits are required depends on the way in which the main modules are to be interconnected and used, and on how many of these modules the instrument is to

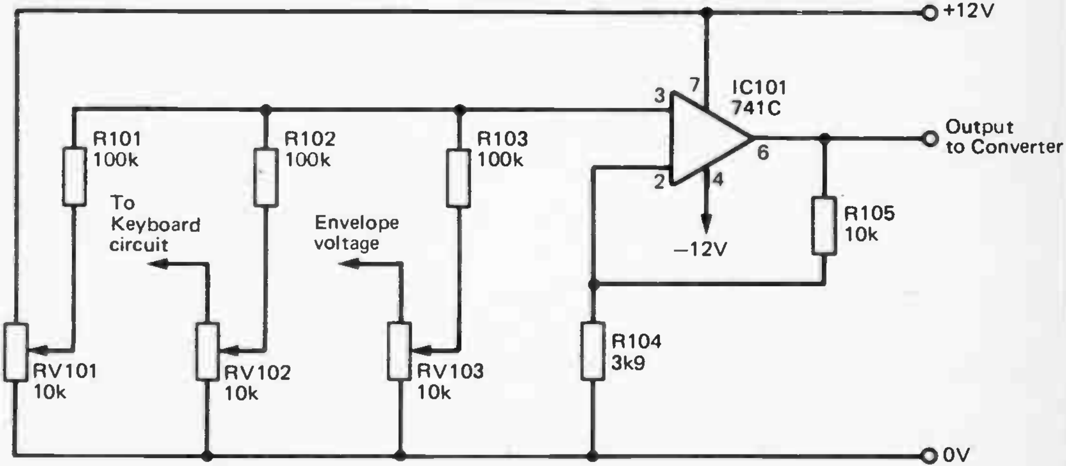


Fig. 19 The circuit diagram of the VCF Mixer

incorporate. Anyway, the circuits described in the following section of this book should prove useful in getting the most out of the instrument and in extending its capabilities.

As mentioned in Chapter 1, in order to get the most from the VCF it should be driven from three voltage sources, the keyboard, an envelope shaper, and a manually controllable voltage. In order to do this a suitable mixer circuit is required, and such a design is shown in Figure 19.

Although the circuit incorporates an operational amplifier it is not the usual summing mode type of mixer circuit. This type of mixer is unsuitable in this case as it provides an unwanted inversion of the signal. This circuit is really a passive mixer using R101, R102, and R103, followed by a non-inverting amplifier which compensates for the losses through the passive mixer. RV101 to RV103 enable the contributions from the three voltage sources to be mixed in the desired proportions. Note that this circuit must drive the VCF via the logarithmic current source, and that it should not be used to drive it directly.

DC Mixer Components (Fig. 19)

Resistors (all ¼ watt 5%)	
R101,102,103	100k
R104	3k9
R105	10k
Potentiometers	
RV101,102,103	10k lin
Semiconductors	
IC101	741C

Miscellaneous

Circuit board
Three control knobs
Wire, solder, etc.

Extendable Mixer

With a simple, single VCO synthesiser there is only one audio output signal, but with a design that incorporates two or more VCOs, plus perhaps a noise source as well, there will be several audio signals which must be mixed together to give a single audio output. All that is needed is a basic summing mode mixer such as the design shown in the circuit diagram of Figure 20.

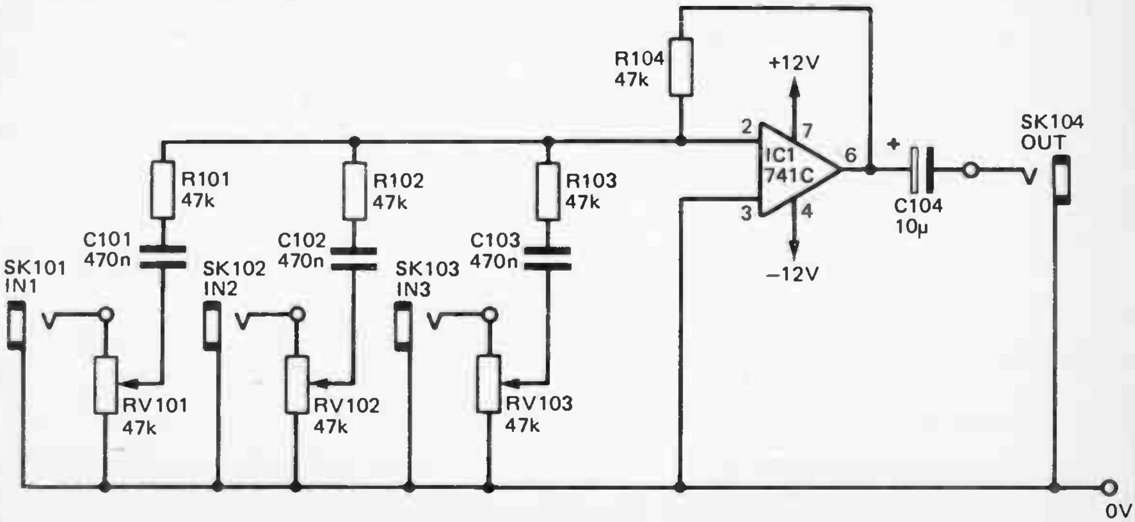


Fig. 20 The circuit diagram of the Extendable Mixer

As shown in Figure 20 the mixer has three inputs, but if only two are required it is merely necessary to omit RV103, C103, and R103. If more inputs are required this can easily be achieved. Just add an extra input resistor, DC blocking capacitor, and level potentiometer for each additional channel. In theory any number of extra channels can be added to the circuit, but in practice problems with noise pick up might be experienced if a large number are used. In the present application no more than about five inputs should be needed though, and this number should certainly not result in any problems.

Extendable Mixer Components (Fig. 20)

Resistors (all ¼ watt 5%)	
R101,102,103,104	47k
Potentiometers	
RV101,102,103	47k log
Capacitors	
C101,102,103	470nF miniature polyester
C104	10μF radial electrolytic
Semiconductors	
IC101	741C

Miscellaneous
Circuit board
Three control knobs
8 pin DIL IC holder, wire etc.

Noise Generator

Not all synthesisers incorporate a noise generator, but it is certainly a circuit which I would never omit from one. In this context the noise we are talking about is of the "white" noise type, which is the "hissing" sound that we normally try to avoid in electronic circuits. When suitably envelope shaped this basic noise sound can produce some interesting effects, but it is much more useful and versatile when used in conjunction with a VCF as well. A lot of percussive sounds can be synthesised, such as cymbal, handclap, and woodblock type sounds. By sweeping the filter from the envelope shaper some unusual but very worthwhile effects can be obtained. Another use of noise is in conjunction with a VCF which is set up to accurately track the pitch dictated by the keyboard. This enables melodies to be played on the noise generator/VCF combination provided the VCF's resonance control has been well advanced. A more conventional but similar use would be to mix the filtered noise with the VCO's output signal, and this can give some good pipe organ type sounds.

Figure 21 shows the circuit diagram of a noise generator based on a Z5J noise diode. This is fed from the two 12 volt supply lines via load resistor R101, and the total supply potential of 24 volts is sufficient to cause D101 to break down and avalanche like a zener diode. Also like a zener diode, a certain amount of noise is generated across the component, and this is coupled by C101 to a high gain amplifier based on IC101. This provides a voltage gain of about 40dB (100 times), and this is needed because the noise voltage generated across D1, although higher than from an average zener diode, is still not very great. The amplified output from IC101 is about 7 volts peak to peak. Using a noise diode gives a good quality output with no significant bias to any frequency band within the audio range.

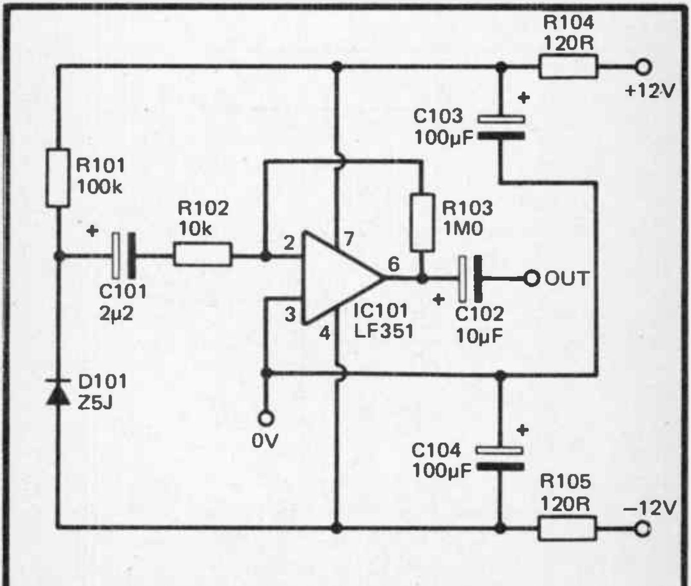


Fig. 21 A simple noise generator circuit

R104, R105, C103, and C104 are used to provide additional smoothing of the two supply rails. This is necessary as the high closed loop gain of IC101 leaves the circuit vulnerable to stray pick up of signals from the VCOs.

Noise diodes have the disadvantage of being relatively expensive and difficult to obtain. An inexpensive alternative is a silicon transistor connected as shown in Figure 22. Only the base and emitter terminals are connected, and these are connected so that the base – emitter junction is reverse biased. This results in the reverse breakdown voltage being exceeded, causing the junction to avalanche. It generates a fairly strong audio noise signal, but the typical output level is slightly less than that of a Z5J noise diode. It is adequate though, and it is not necessary to use any additional amplification. The noise purity is also likely to be not quite as good as when using a proper noise diode, but it is still perfectly adequate for most purposes.

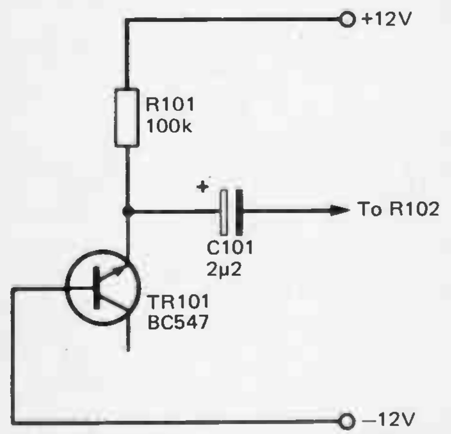


Fig. 22 Alternative noise source

Noise Generator Components (Fig. 21)

Resistors (all ¼ watt 5%)

R101	100k
R102	10k
R103	1M
R104,105	120R

Capacitors

C101	2μ2 63V radial electrolytic
C102	10μF 25V radial electrolytic
C103,104	100μF 16V radial electrolytic

Semiconductors

IC101	LF351
D101	Z5J noise diode

Miscellaneous

Circuit board

8 pin DIL IC holder, wire, etc.

LFO

The use of an LFO (low frequency oscillator) for frequency modulating a VCO to give the vibrato effect was mentioned in Chapter 1. LFOs can actually be used in other ways, such as providing one of the voltage sources for a VCF in order to give dynamic effects. It is well worthwhile having one or two LFOs in the system and experimenting with these. Figure 23 shows the circuit diagram for a simple LFO.

This is a circuit of the type which uses a Miller Integrator (IC101a) driving a Schmitt Trigger (IC101b). RV101 is the frequency control, and it gives an adjustment range of very approximately 0.2Hz to 10Hz. The suggested way of using the LFO is to couple the triangular output from IC101a to the frequency control in the VCO. This is better than trying to couple the signal into the keyboard circuit, where there would be problems with stronger modulation at high notes than when low notes were being played. RV102 is the vibrato depth control. This is only a suggestion for using the unit, and it would probably be worthwhile trying out other ways of using it. The triangular output is the one which is most suitable for the majority of modulation purposes, but do not overlook the fact that a squarewave output signal is available from IC101b.

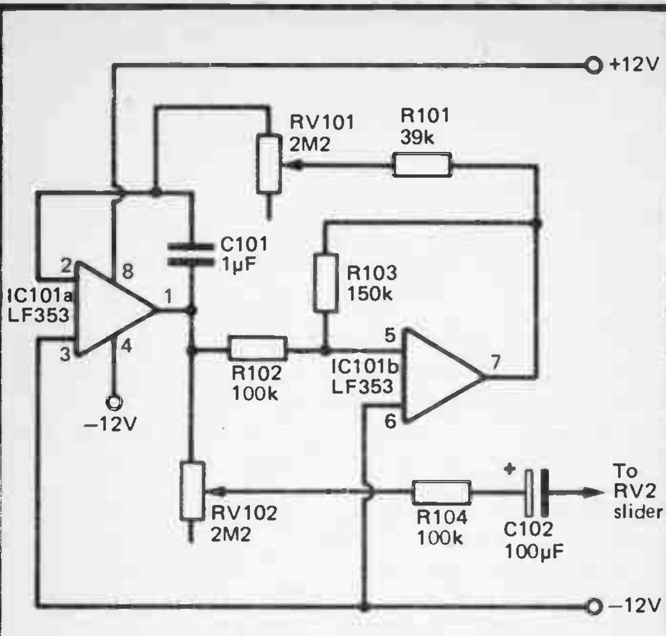


Fig 23. A simple triangular LFO circuit

LFO Components (Fig. 23)

Resistors (all ¼ watt 5%)

R101	39k
R102,104	100k
R103	150k

Potentiometers

RV101,102	2M2 lin

Capacitors

C101	1μF miniature polyester
C102	100μF 25V radial electrolytic

Semiconductors

IC1	LF353

Miscellaneous
Circuit board
Two control knobs
8 pin DIL IC holder, wire, etc.

Construction
Printed circuit designs for the main modules are provided in Figures 24 to 35, and these should considerably ease the task of building what is not a particularly simple piece of equipment when taken as a whole. The minor circuits, if required, can easily be built on stripboard if you do not wish to design your own printed circuit boards for them.

Construction should be generally straightforward from the electrical point of view. Bear in mind that the CA3140E, CA3240E, and 40U1BE devices utilized in some of the modules are MOS input types, and that they consequently require the standard antistatic handling precautions. Use integrated circuit holders for these devices, and do not plug them into circuit until the unit has been completed in all other respects. Until then, leave them in the antistatic packaging (usually this is a plastic tube or conductive foam). When fitting the devices handle them no more than is really necessary, and avoid any obvious sources of strong static charges (nylon carpets, etc.).

Mechanically the standard synthesiser arrangement is to have the keyboard and the electronics in one unit, with the control panel just to the rear of the keyboard, and either vertical or at an angle of about 30 to 45 degrees. For a home constructed instrument I prefer a slightly different arrangement, with a large ready-made case to house the main electronics and the keyboard constructed as a separate unit. An attractive case for the keyboard can easily be made using chipboard and a plastic veneer of some kind, but the exact dimensions and form of the case must be varied to suit the particular type concerned. Note that many keyboards are little more than the keys and a frame on which they are mounted, and the switch contacts must be added by the constructor using parts available separately. You should therefore make quite sure you know what you are buying, and what extras are needed, before ordering the keyboard. The keyboard circuit should be fitted in the same housing as the keyboard, and not with the main electronics. This avoids having a cable with a large number of ways connecting the keyboard to the main unit. In fact just four connections are required (+12V,

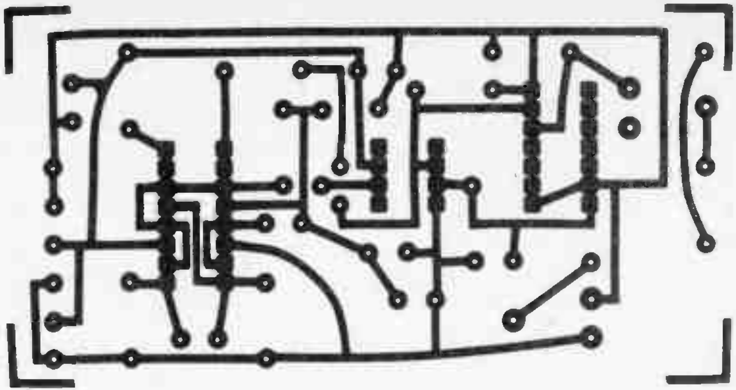


Fig. 24 The track pattern for the VCO

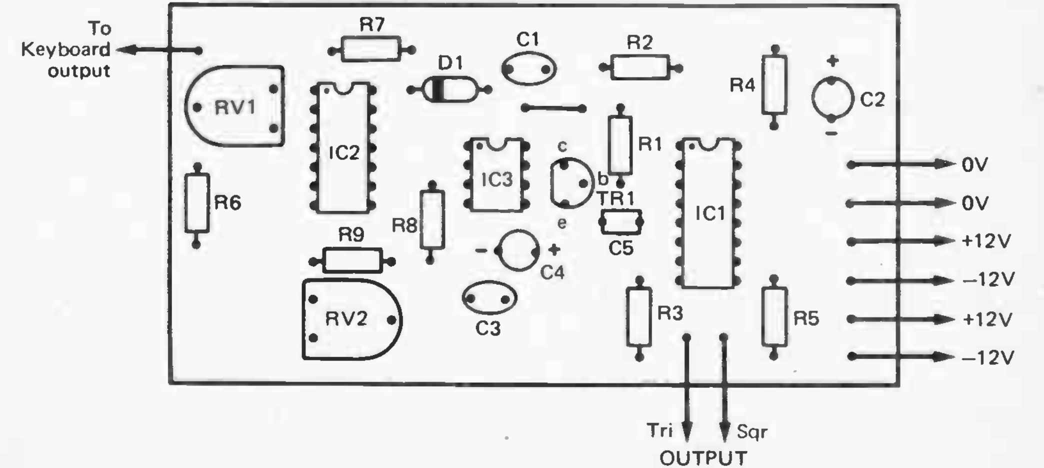


Fig. 25 The component overlay for the VCO

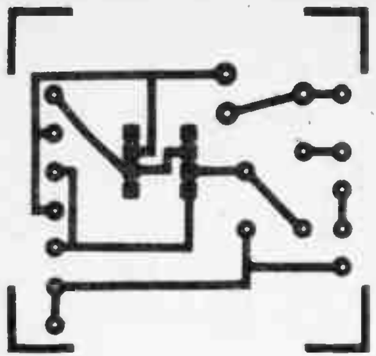


Fig. 26 *The track pattern for the Keyboard*

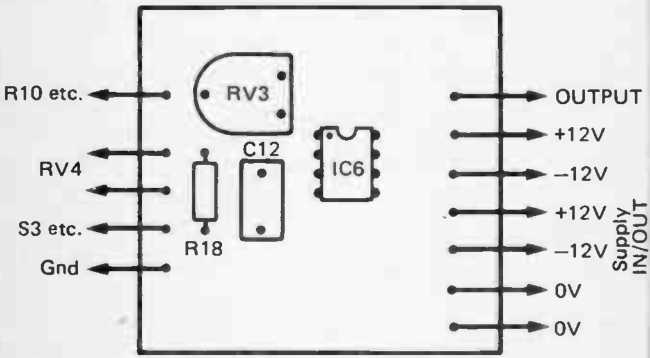


Fig. 27 *The component overlay for the Keyboard*

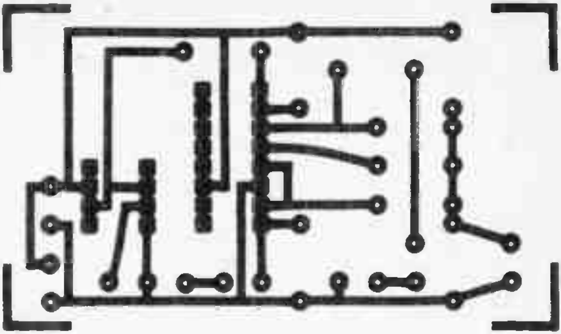


Fig. 28 The track pattern for the VCA

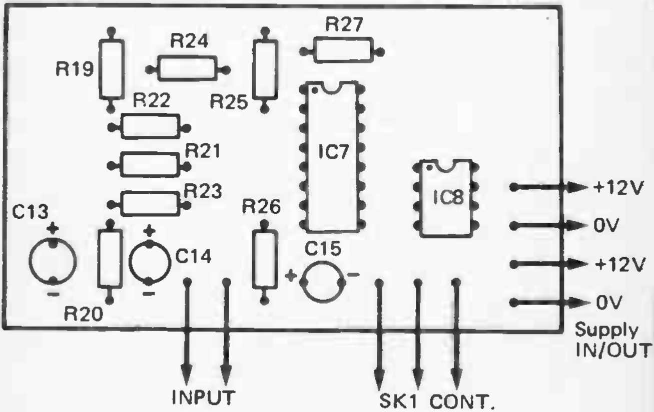


Fig. 29 The component overlay for the VCA

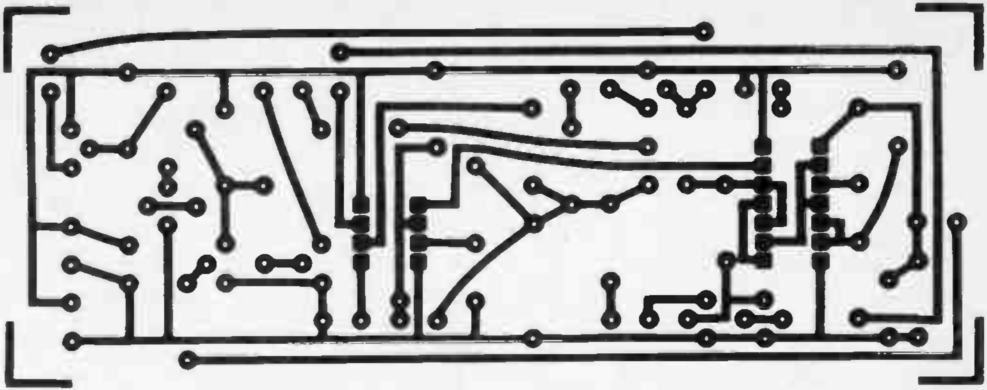


Fig. 30 The track pattern for the ADSR Envelope Shaper

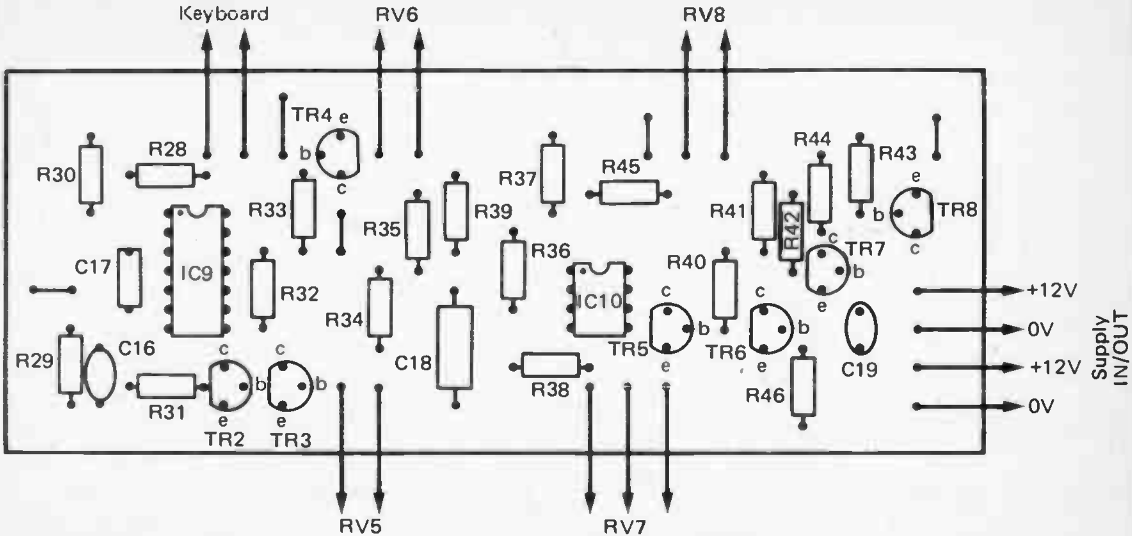


Fig. 31 The component overlay for the ADSR Envelope Shaper

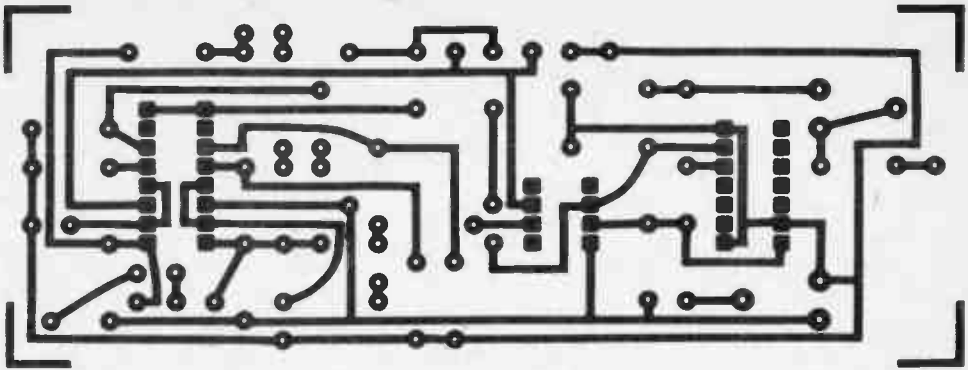


Fig. 32 The track pattern for the VCF

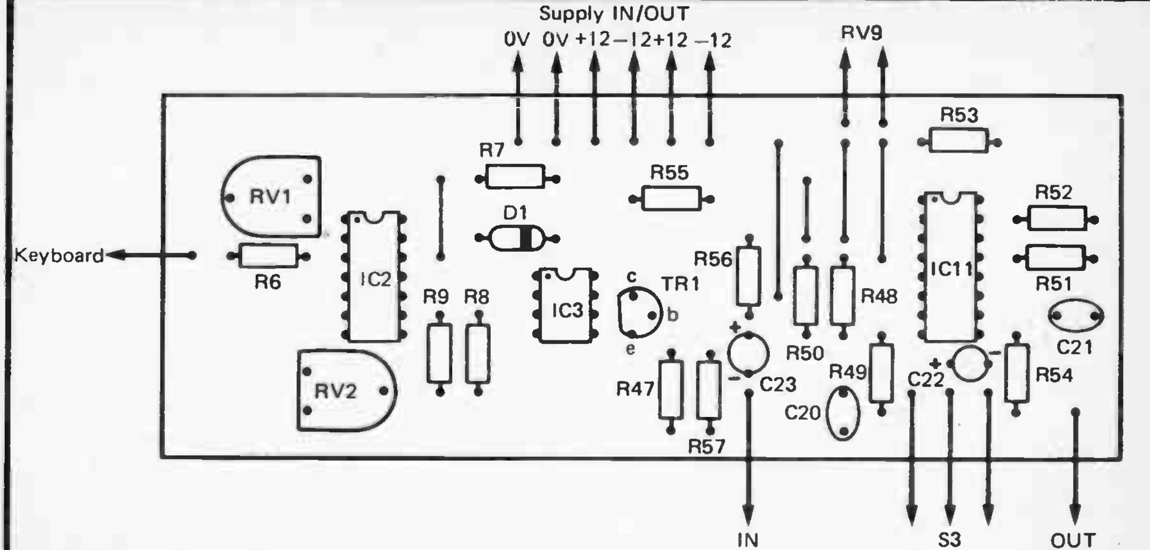


Fig. 33 The component overlay for the VCF

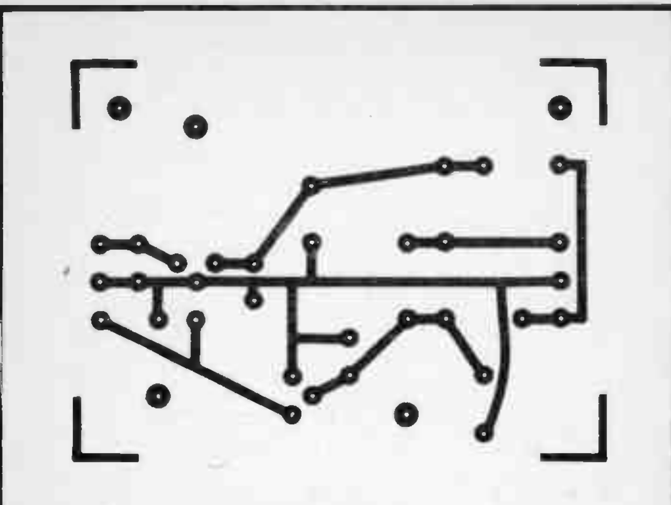


Fig. 34 The track pattern for the PSU

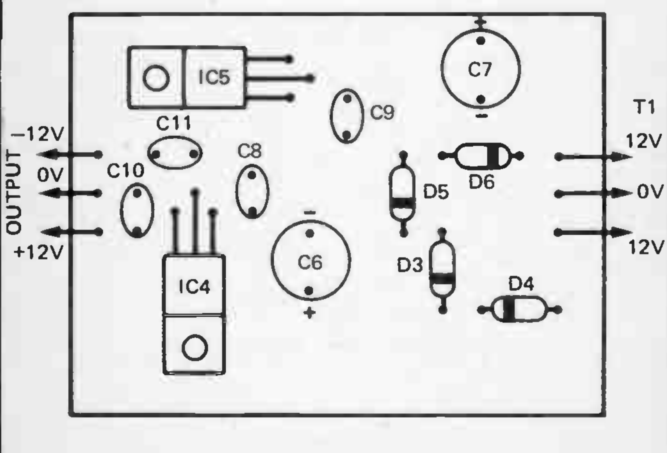


Fig. 35 The component overlay for the PSU

−12V, 0V, and the output). Ideally the coupling should be made via a length of coiled cable, a system used successfully with many business computers, although as yet little used in the field of electronic music.

Note that the keyboard printed circuit does not accommodate the resistor chain, apart from the tuning preset that is. Obviously the number of resistors required will vary depending on which particular keyboard you use, which would make it awkward to design a printed circuit board to suit any keyboard. Anyway, where possible it is far better to mount the components in the resistor chain actually on the keyboard switches, as a lot of hard wiring will otherwise be required.

When dealing with the mains power supply the normal safety precautions should be observed. The case must be a type which has a screw fitting lid or cover so that easy access to the dangerous mains wiring is not possible. Ideally any exposed mains wiring should be insulated. Any exposed metalwork should be earthed to the mains earth lead, as should the 0 volt supply rail. The most practical approach is probably to use a case of all metal construction, and to earth this to the mains earth lead. Anything metal such as a transformer which is mounted on the case will then be earthed, as will any fixing bolts.

Some synthesisers are constructed in such a way that leads can be used to patch the modules together in the desired way, but this is probably only worthwhile in the case of a large instrument which has many modules and a large range of possible methods of interconnection. With more simple instruments a few switches can usually take care of all the likely possibilities. Also, if you use the mixer circuits described earlier, the appropriate controls can be set back to zero to cut out a feature that is not required (to cut out the noise generator for example, or to prevent any keyboard voltage from being fed to the control input of the VCF).

Adjustments

The modules have been designed so that as far as possible no adjustments are required before they are ready for use. In a few cases this is not possible, and there is one preset resistor in the keyboard circuit. However, adjustment of this was covered previously, and this ground will not be covered again here.

The only other presets are the two in each logarithmic to analogue converter circuit. When adjusting a converter that is driving a VCO, RV2 is adjusted to give the correct note with the lowest note of the keyboard being operated. Obviously pitch pipes or another (in-tune) instrument is required as a tuning reference when doing this. RV2 gives a wide adjustment range, and the base note can be set at anything from about one octave above middle "C" to two octaves below middle "C". You may prefer to have this as a front panel control rather than a preset type. This has two advantages, one of which is that it makes it easier to retune the instrument should the tuning drift slightly, and all analogue synthesisers seem to be prone to slight tuning drift. It also enables the tuning to be easily shifted if you wish to (say) offset one VCO from another by a fifth, or to move the compass of the instrument down an octave.

RV1 is adjusted to give the correct pitch with a note near the upper end of the keyboard operated. It is advisable to adjust RV2 and RV1 in turn a few times to make quite sure that the instrument is tracking properly over the full range of the keyboard.

If the converter is used to drive a VCF, the way in which it is set up will to some extent depend on the exact way in which the VCF is being utilized, and in general things will be less critical than when driving a VCO. The basic effect of the two controls is the same as when driving a VCO, with RV2 setting the base frequency and RV1 setting the sensitivity. When driving the circuit from the mixer circuit of Figure 19 the settings of RV1 will not be at all critical, and simply setting it at about half value should suffice. RV2 can be given any setting which enables the three mixer controls to be set up so that the filter tracks the keyboard correctly. This is done by setting the frequency control to give the correct frequency with the lowest note played, and adjusting the "keyboard" control to give the correct cutoff frequercy with a high note played. The envelope control should be set at minimum. Adjustment of the "frequency" and "keyboard" controls should be repeated a few times in order to ensure accurate tracking on occasions when accurate tracking is essential.

Using The Modules

Although primarily intended to be utilized as a full synthesiser in their own right, there are other ways of using the modules. For example, an arrangement such as that shown in Figure 36 can be adopted in order to add to the facilities of an existing analogue synthesiser (which must have a 1 volt per octave CV characteristic and a +5 volt gate output signal). The idea here is to effectively

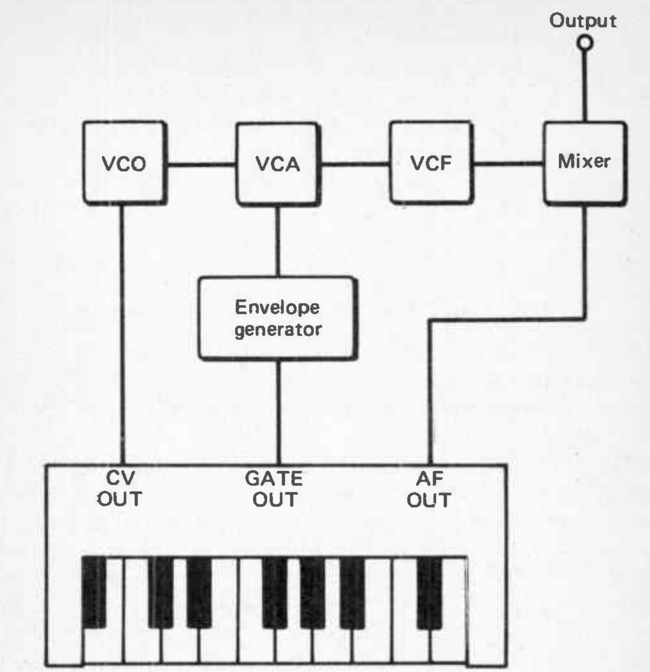


Fig. 36 Using the modules with a synthesiser

give the synthesiser an extra VCO, and this can provide a very worthwhile improvement in performance with a simple single oscillator instrument, but is also a useful expansion with more complex instruments. The control voltage for the VCO is provided by the CV output of the synthesiser. If you require the ability to switch between the home constructed synthesiser's own keyboard and an external CV source, all that is required is a simple switching circuit of the type shown in Figure 37. It would in fact be possible to use switch contacts on the CV input socket to automatically provide the switch-over, but in practice it will probably be more convenient to have a separate switch.

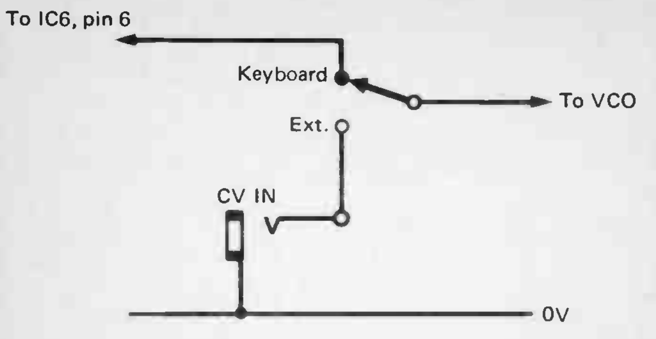
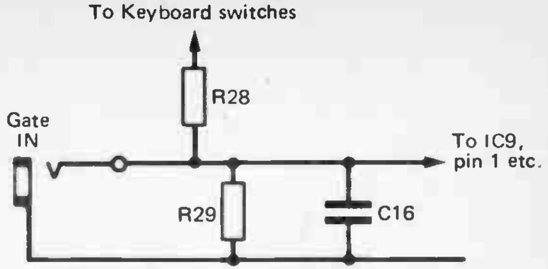


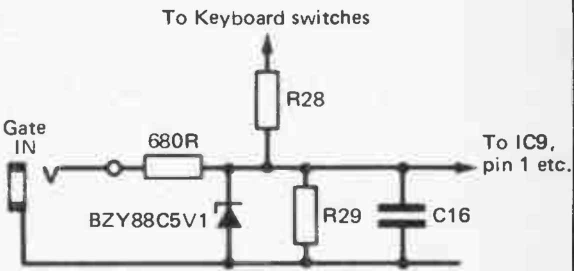
Fig. 37 Adding a CV input to the instrument

The gate output of the synthesiser is used to drive the envelope generator in the manner shown in Figure 38(a). This seems to work perfectly well provided the gate output signal is at standard 5 volt logic levels (which is, unfortunately, not the case with all instruments). With a type which has a +15 volt gate output signal the zener clipping circuit of Figure 38(b) would have to be added to protect the input circuitry of the envelope generator. With the type that provides a "short-to-ground" gate signal the arrangement of Figure 38(c) should give the desired result, although I have not had the opportunity to fully test this circuit and it is only put forward as a recommended line of experiment for those with the appropriate type of synthesiser.

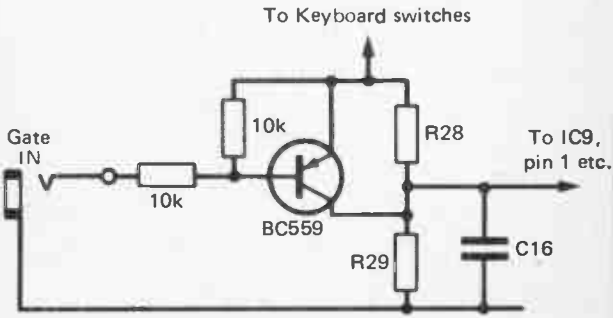
A mixer is used to combine the signals from the two instruments into a single output. With a simple arrangement of the type outlined here it is possible to generate some interesting sounds. The main point to bear in mind, and to exploit to the full, is that the two instruments have separate envelope shapers, and the signal from one synthesiser can have a totally different envelope shape to the other. This enables some interesting results to be obtained, with the system effectively providing envelope shapes that are not available with a single ADSR envelope shaper. The two synthesisers have separate filters as well, which can also be used to give some excellent effects. Bear in mind that the two instruments do not need to be tuned to the same octave, or even the same note.



(a) Adding a gate input socket



(b) Driving the unit from a +15V gate pulse source



(c) Driving the unit from a short-to-ground gate signal

Fig. 38 (a), (b) & (c)

In Figure 36 the home constructed synthesiser is shown as a very basic type, but by adding extra modules such as a separate envelope generator for the VCF it is possible to further expand the range of available sounds.

A very useful application for a home constructed synthesiser in conjunction with a ready-made type (or another home built instrument for that matter) is as a simple percussion synthesiser. In this role the gate signal is provided by an external source, which would normally be in the form of some kind sequencer. It could be provided by the main synthesiser, but this would result in the percussion sound being generated each time a note was played, which is not normally what is required. Sequencers are covered in the next chapter, and this subject will not be persued further here. Many percussion sounds are noise based, and are therefore only available if the instrument is equipped with a noise generator. However, by using an envelope shape which has a fast attack and slow decay, a fairly low VCO frequency, and the VCF set to give only a low harmonic content on the output signal, some quite good drum sounds can be obtained. By sweeping the VCO with the output of the envelope generator some good falling pitch and "disco" drum sounds can be produced.

Of course, many synthesisers these days do not have the CV/gate type of interface, but instead have a MIDI (musical instruments digital interface) type. This is a fully digital type of interface which is completely incompatible with CV/gate interfaces. However, analogue instruments can be driven successfully from MIDI equipment with the aid of a suitable converter. MIDI goes well beyond the scope of this book, but details of some MIDI equipment (including a MIDI to CV/gate converter circuit) can be found in Book No. BP182 "MIDI Projects", from the same publisher and author as this publication.

Chapter 3

SEQUENCING

Automatic sequencing of a synthesiser is a common practise, and one which has two main applications. Probably the more popular of these is to have a short sequence of notes which is repeated indefinitely, and this acts as a backing while a "live" melody line is played. The other application is where long sequences of notes are programmed, and systems of this type often have numerous channels. This type of sequencing enables composers and arrangers to experiment with ideas, and to try out pieces of music which are beyond their playing ability. In fact sequencers can play any music note – perfectly, no matter how demanding it may be, and some of the experimental music currently being written can only be performed by electronic means.

Whether short repetitive sequences are required, or long and difficult pieces are to be played automatically, there are two basic types of sequencer which can be used. These are the "real-time" and "step-time" varieties. With a real-time sequencer the notes are entered into the system by playing them on the keyboard, and the sequencer records the notes that are played, the times between notes, and with some systems even how hard the notes are played. This information is extracted via the gate and CV sockets, or via the MIDI interface, as appropriate. The sequencer can then play back exactly what was played during the recording process, but refinements often allow such things as note duration correction (making notes precisely the correct duration even if the playing was a bit untidy), and the ability to speed up or slow down the tempo. Real-time sequencing is preferred by many musicians as it provides a fast means of entering the required sequence of notes, but it obviously demands a fair degree of playing competence on the part of the user, even if multipart music is entered one or two lines at a time.

Step-time sequencing involves no playing whatever, and the exact method of entering the notes depends on the particular system in use. With a simple system it might involve setting a few switches or connecting patch-cords, with a computer based system and a light-pen being used in top of the range equipment. Entering music into a step-time sequencer is a relatively slow business, but it is an attractive method for many users as it requires no playing skill at all, and gives note-perfect results every time. At least, it gives perfect results provided no programming errors are made, but even if a mistake should be made it is usually quite easy to edit it out.

It is quite feasable to build complex step-time or real-time sequencers, but it is generally easier to base such systems on a home computer rather than build them from scratch. This is a subject which we will not consider here as it really goes well beyond the scope of this book. However, it is a topic which is covered in the books No. BP173: "Computer Music Projects" and No. BP182: "MIDI Projects", both of which are from the same publisher and author as this publication.

Percussion Sequencer

Simple step-time sequencers are something that can be easily built by the home constructor, and although their usefulness is low in comparison to complex sequencers, taking into acount the very low costs involved they are still a worthwhile proposition. If a percussion synthesiser (or an ordinary synthesiser used as a percussion synthesiser) is to be sequenced, then it will often only be necessary to generate a series of gate pulses having the required rhythm, and no control voltage output will be required. Figure 39 shows the main circuit for a simple percussion sequencer.

The clock signal is generated by IC1 which is a 555 timer device connected in the standard astable (oscillator) mode. RV1 enables the operating frequency to be adjusted from about ten pulses per second to around one pulse every four seconds. RV1 is effectively the tempo control.

IC2 is a 4017BE decade counter and one of ten decoder. In this application it is only the one of ten decoder action that is required. In this role the device has ten outputs, one of which gocs high while all the others take up the low state. Initially it is output '0' that is high, but on subsequent clock pulses output '1' goes high, then output '2', and so on. After output '9' has gone high the circuit cycles back to output '0' going high on the next clock pulse, and this sequence repeats indefinitely.

The 4017BE is popular for simple sequencing applications as it is not difficult to use the outputs to provide simple rhythm patterns. In this case each output is taken to a 2 millimetre socket, and the required outputs are patched through to a sort of mixer circuit based on D1 to D8. For instance, if we assume that outputs

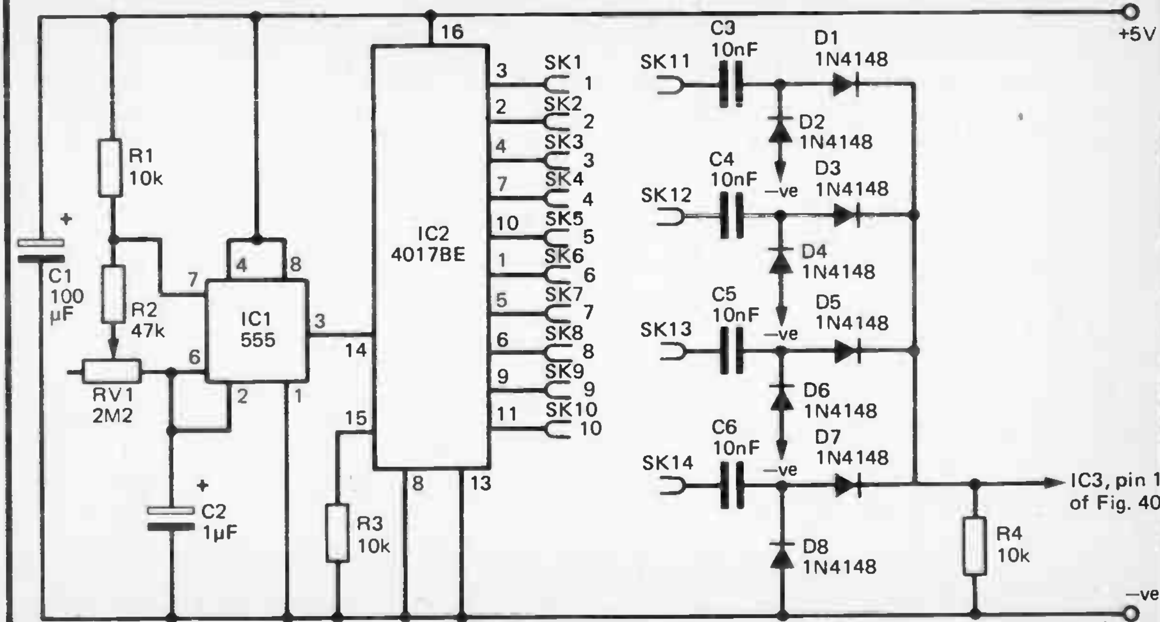


Fig. 39 The circuit diagram of the Main Percussion Synthesiser

1, 4, 5, and 7 are to be used, these would be connected through to SK11, SK12, SK13, and SK14 (one output connecting to each input of the mixer). The purpose of the mixer is to generate short positive pulses on the leading edge of each output pulse from IC2. Although at first sight it might seem that a conventional OR gate action is all that is required, things are not quite as simple as this. If (say) output '1' and '2' were to be logic ORed, as output '1' went low output '2' would go high, giving no change in the output from the gate. In other words, rather than two distinct output pulses, one double length pulse would be obtained. By giving a brief pulse on the leading edge of each output pulse the mixer circuit does provide two distinct output pulses. By adding extra stages to the mixer circuit it could provide more than four output pulses per IC2 cycle, but in practice four stages should be sufficient.

The output pulses from this circuit are incompatible with the gate and trigger inputs of most synthesisers, but the simple circuit

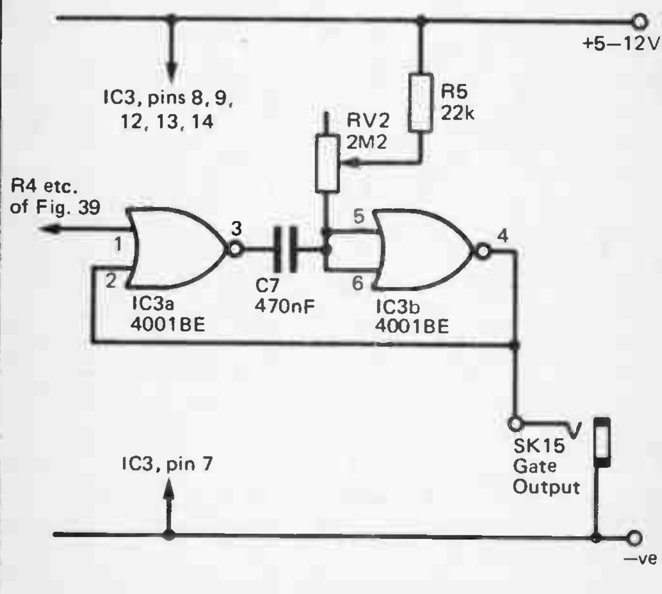


Fig. 40 The circuit diagram of the Gate Pulse Generator

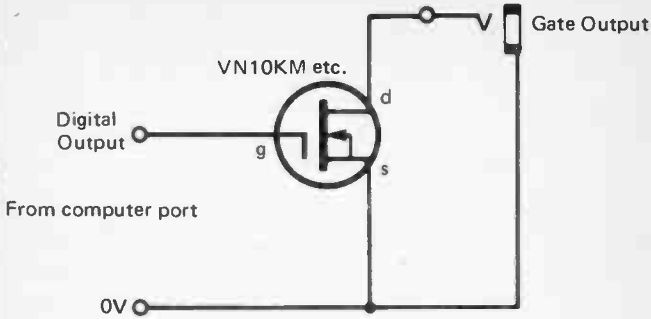


Fig. 41 Converting 5 volt logic levels to a short to ground action

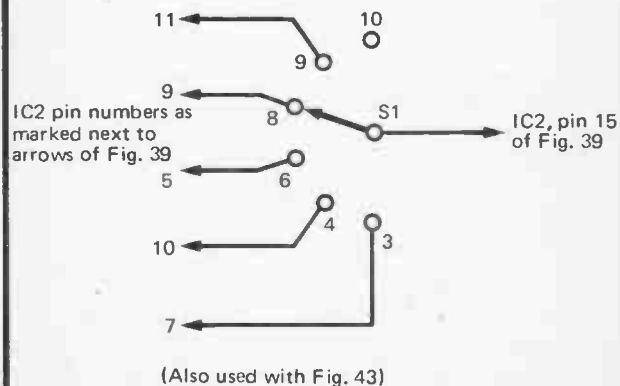


Fig. 42 A rotary switch is all that is needed in order to reduce the sequence length

of Figure 40 is all that is needed in order to give an output at standard 5 volt logic levels. Most synthesisers which require nominal +15 volt trigger pulses are also designed to respond to +5 volt signals, and should work perfectly well with this circuit. For synthesisers that require a short-to-ground gate signal a VMOS transistor should be added at the output of the circuit, as shown in Figure 41. The circuit of Figure 40 is a basic CMOS monostable circuit, and RV2 enables the gate output pulse duration to be varied from around 5 milliseconds to approximately 500 milliseconds. However, bear in mind that when using a fast tempo the gate pulse must be quite short, or a trigger pulse might be received before the previous output pulse has ceased, resulting in missing beats.

A major drawback of the unit is that it can only produce rhythms based on ten units of time. It can be made much more versatile by adding a six way switch, as shown in Figure 42. This switch enables a selected output to be connected to the reset input of IC2. Thus, when the selected output goes high, IC2 is immediately reset to '0', and each cycle is cut short. This enables the unit to produce rhythms based on 3, 4, 6, 8, 9, or 10 units of time.

Percussion Sequencer Components (Figs. 39 & 40)

Resistors (all ¼ watt 5%)

R1,3,4	10k
R2	47k
R5	22k

Potentiometers

RV1,2	2M2 linear

Capacitors

C1	100μF 10V radial electrolytic
C2	1μF 63V radial electrolytic
C3,4,5,6	10nF miniature polyester
C7	470nF miniature polyester

Semiconductors

IC1	555
IC2	4017BE
IC3	4001BE

Miscellaneous

SK1 to SK14	2mm sockets
SK15	Standard Jack socket
S1	6 way 2 pole rotary switch

Printed circuit board
Four leads fitted with 2mm plugs
Two control knobs, IC holders, wire, etc.

10 Note Sequencer Components (Figs. 40, 42 & 43)

Resistors (all ¼ watt 5%)

R1,3	10k
R2	47k
R5	22k

Potentiometers

RV1,2	2M2 linear
RV3 to RV12	10k miniature horizontal preset

Capacitors

C1	100μF 10V radial electrolytic
C2	1μF 63V radial electrolytic
C7	470nF miniature polyester

Semiconductors

IC1	555
IC2	4017BE
IC3	4001BE
D1 to D10	1N4148

Miscellaneous

S1	6 way 2 pole rotary
SK1,15	Standard jack socket

Printed circuit board
Two control knobs, IC holders, wire, solder, etc.

16 Note Sequencer Components (Figs. 40, 44 & 45)

Resistors (all ¼ watt 5%)

R1	10k
R2	47k
R3,4,6,7	1k
R5	22k

Potentiometers

RV1,2	2M2 linear
RV3 to RV18	10k miniature horizontal preset

Capacitors

C1	100μF 10V radial electrolytic
C2	1μF 63V radial electrolytic
C7	470nF miniature polyester

Semiconductors

IC1	4067BE
IC2	555
IC3	4001BE
IC4	4024BE
IC5	74HC85

Miscellaneous

S1	SPST miniature toggle type
SK2,3,4,5	Four SPST miniature toggle switches or one hex switch
SK1,15	Standard jack socket

Circuit board, IC holders, wire, solder, etc.

Note Sequencing

Sequencing notes rather than just rhythms is, on the face of it, a much more difficult prospect, as a range of voltages must be provided. In fact sequencing notes is not particularly difficult, and requires little more than the addition of a few potentiometers. Figure 43 shows the circuit diagram of a simple note sequencer circuit.

This has obvious similarities with the Percussion Sequencer which was described previously, and the main difference is the addition of a preset potentiometer at each output of the 4017BE. A simple gate is formed by the ten diodes, and its purpose is to enable the voltage from whichever output is activated to reach the output socket unhindered. Without the diodes the preset resistors in the other channels would load the active output, reducing the maximum available voltage. It would also make the unit practically impossible to tune since adjusting one preset would affect the tuning of all the others.

The circuit can provide output voltages of up to about 4.3 volts, The maximum output voltage is slightly less than the supply

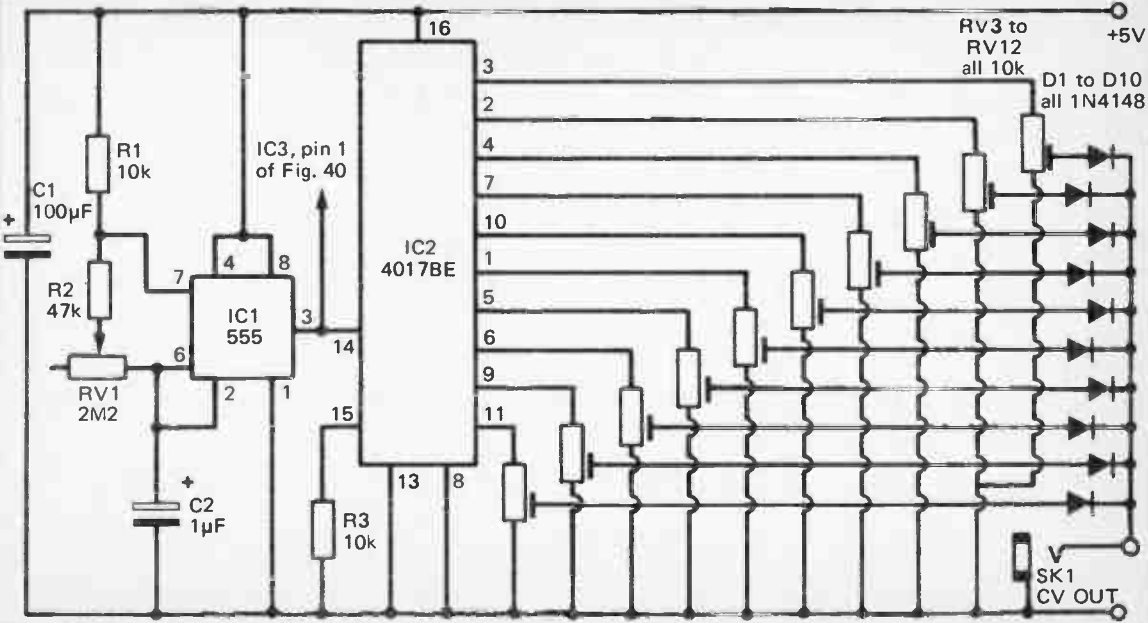


Fig. 43 A basic sequencer circuit

voltage due to the voltage drop through the output diodes, and because IC2 provides a "high" output voltage which is slightly less than the full supply voltage. This enables a range of just over four octaves to be covered with a standard 1 volt per octave synthesiser, and this should be adequate for most purposes. A larger range could be provided by increasing the supply voltage, which can be up to 15 volts. However, this would also increase the gate output potential, and this is unacceptable with synthesisers that are designed to accept a 5 volt signal. It is, of course, quite acceptable with synthesisers that are designed for use with a +15 volt gate pulse, and it is also satisfactory if a VMOS transistor is used to provide a suitable gate signal for a "short to ground" gate input.

The inclusion of a diode in series with the output of each tuning potentiometer does result in slight variations in the tuning voltage with changes in the ambient temperature, but in practice this factor does not seem to introduce significant tuning drift. Any variations in the power supply voltage will introduce tuning drift though, and for this reason it is important to power the unit from a well smoothed and regulated power supply.

Of course, a gate pulse generator is required, and the most simple way is to derive the pulse from pin 1 of IC3 as shown in the circuit of Figure 40. Alternatively, the diode mixer circuit from the percussion synthesiser can be used to drive the gate impulse generator circuit. The first method has the advantage of simplicity, but it is restrictive in that each note has to be of the same duration, and there is no control over the rhythm apart from the tempo. The second method permits simple rhythm patterns to be generated, but at the cost of reduced note capacity and increased complexity. Whichever method is adopted, it is advisable to include S1 (Figure 42) so that the number of notes in the sequence can be reduced if required.

16 Note Sequencer

It is possible to produce longer sequences than 10 notes, but for really long sequences a complex circuit based on analogue-digital-analogue conversion techniques is required. However, if only a few extra notes are required it is possible to achieve this using relatively simple circuits. The circuit diagram of Figure 44 shows how a CMOS analogue switch can be used as the basis of a sequencer.

IC1 is a CMOS 4067BE sixteen way single pole type. The sixteen inputs are fed from individual tuning potentiometers (but for clarity only two of these are shown in Figure 44). Which one of the tuning voltages is fed through the device and out to the output socket depends on the four bit binary code fed to the address inputs of the device.

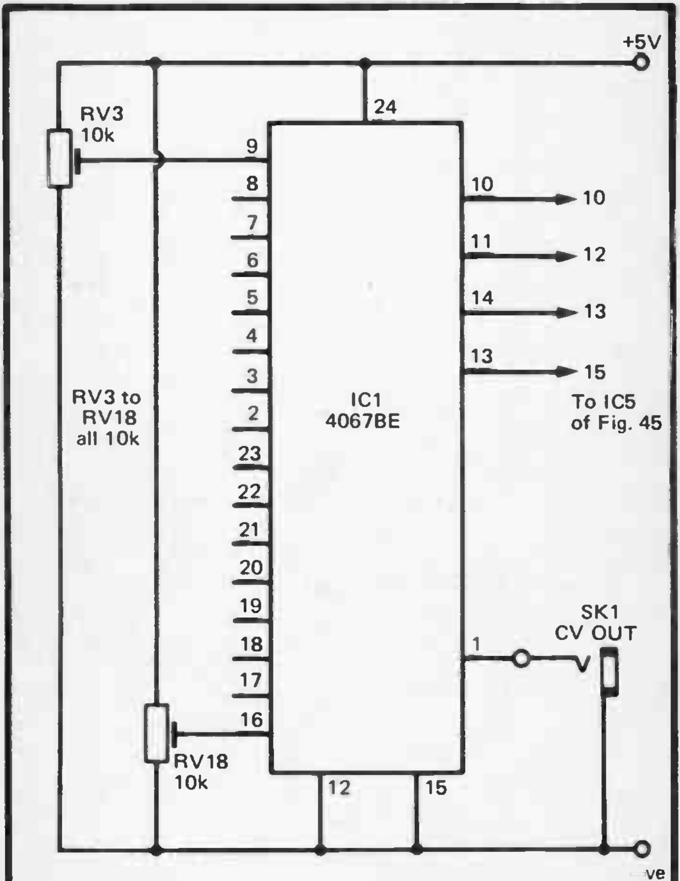


Fig. 44 Deriving the CV from a multiplexer

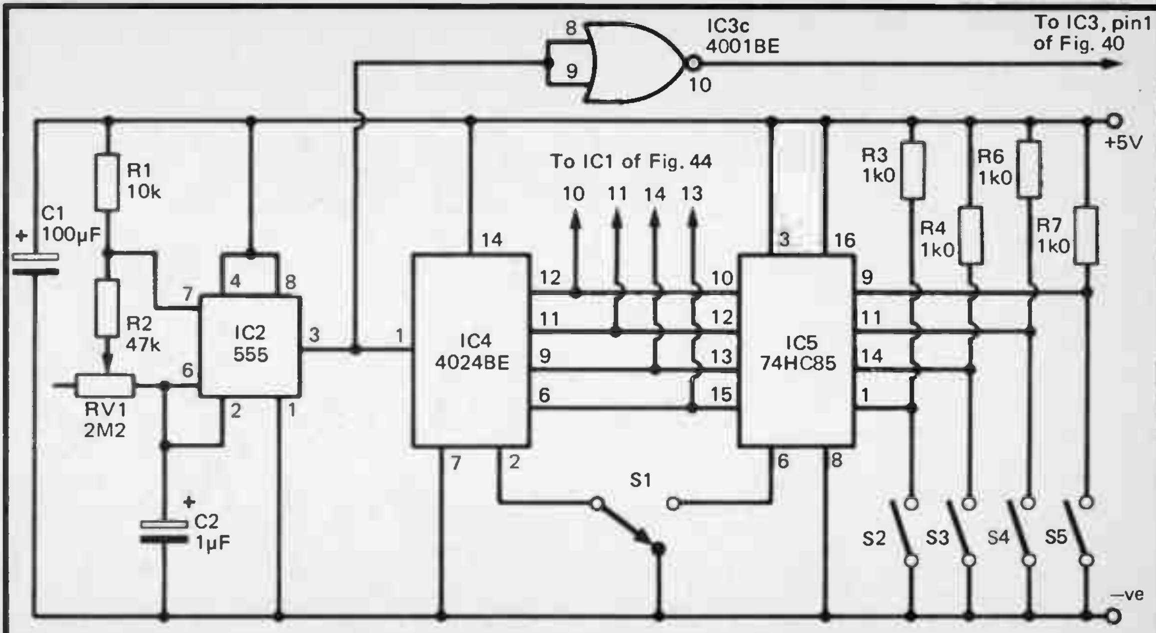


Fig. 45 The circuit diagram of the Clock Generator for the 16 Note Sequencer

Figure 45 shows the circuit diagram for the clock generator and counter which generates the sequence of four bit addresses. This has a 555 clock oscillator (as in the previous sequencer circuits) driving a seven stage binary counter (IC4). Only the first four stages of IC4 are utilized, and these provide the four bit addresses for IC1. This results in RV3 to RV18 being switched through to the output, in sequence, and the circuit cycles in this way indefinitely.

Obviously it is advantagous if there is some means of reducing the sequence length available, and this is made possible by the inclusion of IC5. This is a 74HC85 four bit comparator, and it compares the output code of IC4 with the binary code provided by S2 to S5 and the four load resistors. When the two codes are the same the output of IC5 goes high, and provided S1 is set to connect IC5 through to the reset input of IC4, IC4 is reset to zero and the count is shortened. S1 connects the reset input of IC4 to the negative supply rail when IC5 is switched out of circuit.

S2 to S5 can be miniature toggle switches, or a type of switch known as a "hex" (hexadecimal) switch can be used. The latter is a sixteen way rotary switch which is calibrated "0" to "15" and provides the appropriate binary output code for each number. This option is by far the more convenient in use since it avoids having to convert decimal numbers to their four bit binary alternative before the switches can be set correctly. If you choose to use a hex switch, be careful to obtain the correct type. The switch must be of the type where an output which is to be at binary 0 is connected to the common terminal, and not a type where the switch is open when the output it drives is at binary 0. Of course, in the current context switch positions "0" and "1" are of no practical value, since sequences 0 or 1 note long are obviously nonsensical, and switch position "2" is the lowest that should ever be selected. Accidentally setting the hexadecimal switch to "0" or "1" will not damage the circiuit though (it will simply result in the first note being held continuously), and there is no need to implement some means of eliminating these positions. Incidentally, hexadecimal switches are usually printed circuit rather than panel mounting types, and this is something which must be taken into acount when designing the component layout, etc. Alternatively, with a little ingenuity it should be possible to devise a method of panel mounting one of these components.

A limitation of this circuit is that it only permits each note to be of the same duration, and there is no easy way around this problem. The gate circuit used in the Percussion Synthesiser is not applicable in this case as there are no separate outputs from the 4067BE at logic levels to drive this circuit. The gate pulse generator circuit of Figure 40 is used to provide the gate output signal, but note that this circuit is driven via one of the previously unused gates of IC3 (IC3c) in order to synchronise the gate pulses with changes in the tuning voltage.

Construction

Printed circuit designs for the Percussion Sequencer and the 10 Note Sequencer are provided in Figures 46 to 49. No printed circuit design is given for the 16 note type, but there should not be too much difficulty in building this on 0.1 inch stripboard, The 74HC85 device specified for the 16 Note sequencer might be a little difficult to obtain, and this is the high speed CMOS version of the 7485 TTL device. The standard CMOS 4063BE type is a pin for pin equivalent to the 74HC85, and in this circuit where ultra-fast operation is not required it is a suitable alternative, Note that the 74HC85, 4063BE, 4001BE, 4017BE, 4024BE, and 4067BE are all CMOS devices, and that they consequently require the standard MOS antistatic handling precautions.

Construction of these projects should not be difficult, but with the percussion synthesiser try to use a well layed out front panel so that there is no difficulty in wiring the sockets together in the required manner. It is also advisable to clearly label the sockets so that it is easy to wire things up in the required way, with mistakes being avoided. In fact with any music project it is advisable to clearly label all controls and sockets, especially with projects that have numerous controls and (or) sockets. Rub-on letter transfers are available from some of the larger electronic component retailers, and they are also available from many stationers and graphic art supply shops. With the aid of these it is possible to produce some really professional looking front panels, but it is advsable to protect the finished panels with a spray-on lacquer, as rub-on transfers rub-off almost as easily from many plastic and metal surfaces. If you do not wish to go to the trouble of using rub-on transfers a simple alternative is to use self-adhesive paper labels with the legends just written on using a pen. This may not give a particularly neat looking finished product, but it will do the job perfectly well.

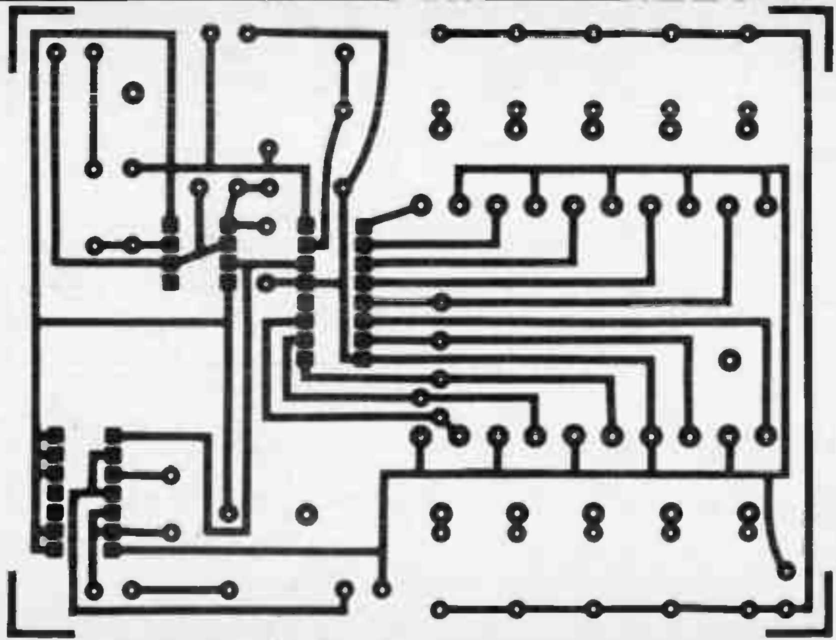


Fig. 46 The track pattern for the Percussion Sequencer

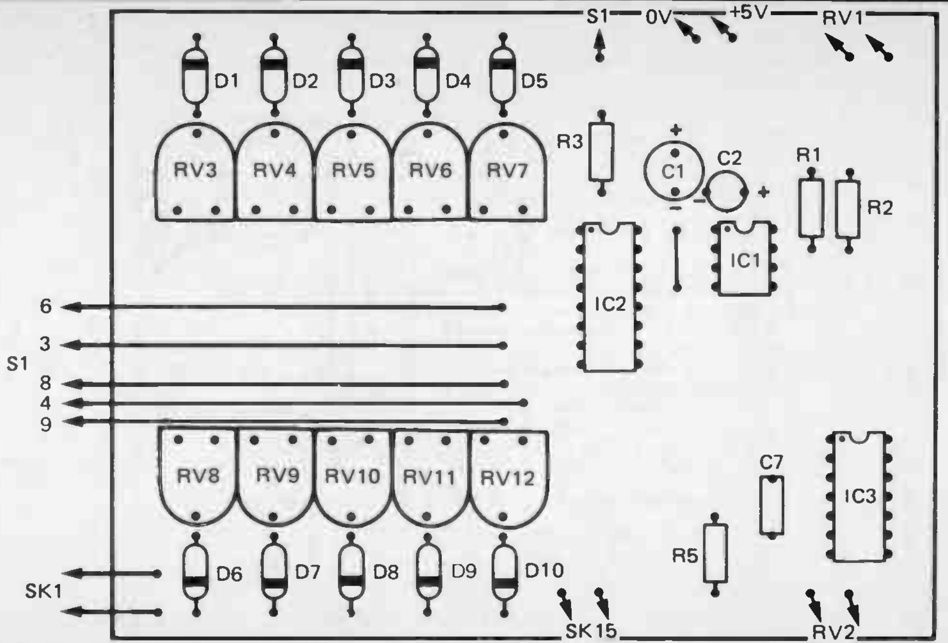


Fig. 47 The component overlay for the Percussion Sequencer (combines Figs. 40, 42 & 43)

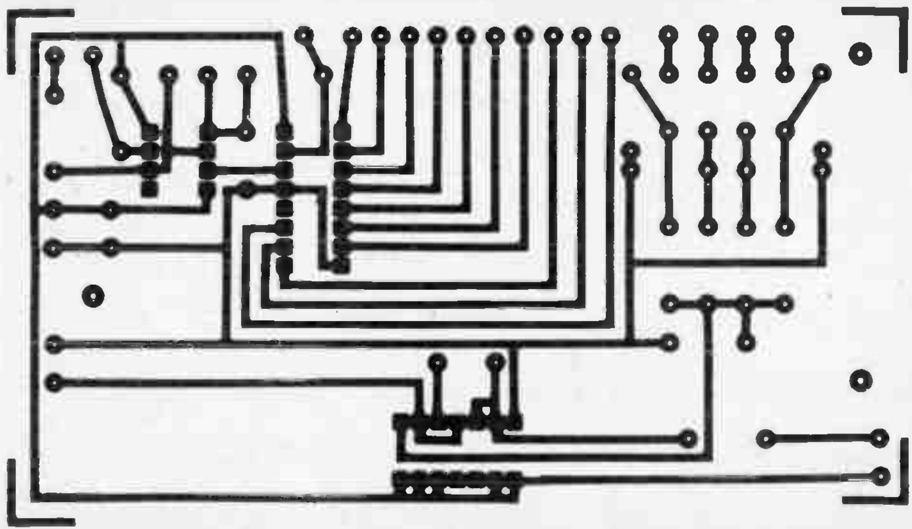


Fig. 48 The track pattern for the 10 Note Sequencer

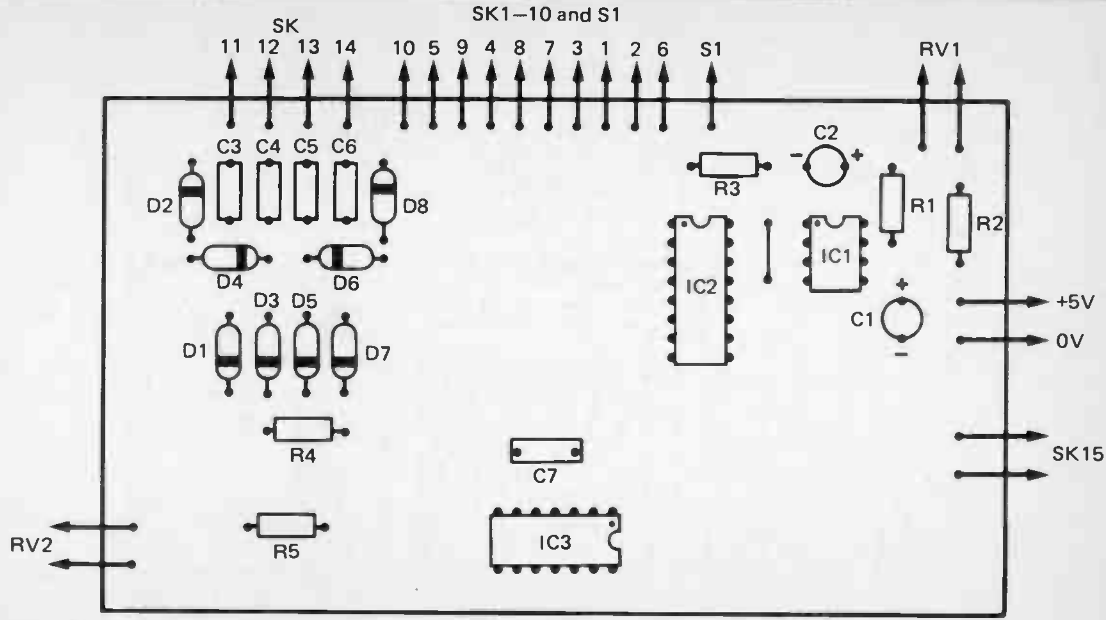


Fig. 49 The component overlay for the 10 Note Sequencer (combines Figs. 39, 40 & 42)

Tuning

With the tempo control set at minimum you may well find that you can tune the preset resistors without too much difficulty, although it will probably take more than one run through the sequence of notes before the tuning of every preset is spot on. If you are likely to want to make frequent readjustments to the presets it will be much more convenient to have a "hold" facility, so that the sequencing can be halted at any required note, thus enabling the appropriate preset to be accurately tuned at your leisure. There is more than one way of implementing a "hold" facility with these designs, but probably the most simple one is to connect a switch into the clock oscillator circuit, as shown in Figure 50. When the switch is open the charge/discharge path to the timing capacitor is cut, and oscillation ceases. When the switch is closed the circuit is able to function normally.

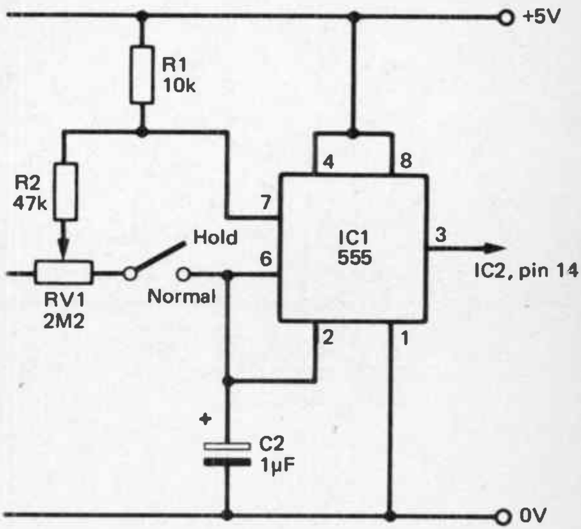


Fig. 50 Adding a "hold" switch to the clock oscillator

In use the switch is set to the closed position until the sequence reaches the preset that controls the first note in the sequence. The switch is then opened, and left open until the preset has been accurately tuned. The switch is then closed to advance the sequence to the next note, after which it is opened and the second preset is adjusted. This process is continued until all the presets in use have been adjusted correctly. Of course, it is advisable to use a fairly slow tempo setting so that operating the switch does not become something of a reaction testing game. The "hold" switch also makes it quite easy to edit just one or two notes in the sequence if desired.

Power Supply

Figure 51 shows the circuit of a simple 5 volt mains power supply unit which is suitable as the power source for these sequencer projects. This is a conventional design using a small 5 volt monolithic voltage regulator to provide a well smoothed and regulated output. For battery operation the mains transformer, the two rectifiers, and the smoothing capacitor could be replaced by a fairly high capacity 9 volt battery such as a PP7 or PP9 size.

5 Volt PSU Components (Fig. 51)

Capacitors

C1	1000μF 16V electrolytic
C2,3	100nF ceramic

Semiconductors

IC1	78L05 (+5 volt 100mA regulator)
D1,2	1N4002

Miscellaneous

FS1	20mm 160mA Antisurge
T1	Mains primary, 9–0–9 volt 100mA secondary
S1	Rotary mains switch

20mm fuseholder
Circuit board
Case, mains lead, wire, solder, etc.

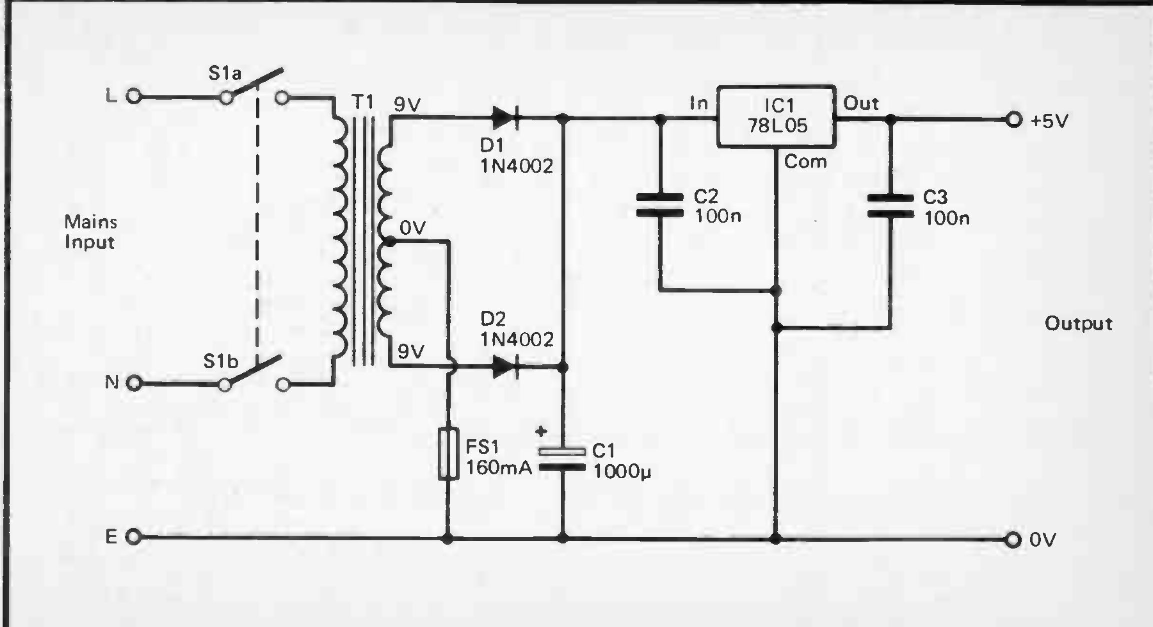


Fig. 51 The circuit diagram of the 5 Volt PSU

Chapter 4

ADDING STEREO EFFECTS

Effects units are probably more associated with electric guitars than with synthesisers by most electronic musicians, and it is true that there is little point in adding some types of effects units to a synthesiser. This is simply because the built-in facilities of most synthesisers make some effects units superfluous. As a couple of examples of this, a distortion box is not of great value since using a squarewave or pulse signal gives much the same sound as taking some other waveform and then clipping it in a distortion unit. A waa-waa type effect can be obtained by using the VCF with a high resonance setting, and then sweeping the filter from (say) an envelope generator or an LFO.

Although some types of effects unit are quite pointless when applied to synthesisers, there are some that can be used to good effect. Probably the chorus effect is the most popular amongst synthesiser players, and a popular application of this effect is in the generation of the so-called "string ensemble" sound. Effects such as flanging and phasing can also usefully boost the range of sounds available from a synthesiser. A range of popular effects units will not be described here as space does not permit this, but a number of simple effects circuits can be found in the Book No. BP74: "Electronic Music Projects". Some more complex designs can be found in the Book No. BP174: "More Advanced Electronic Music Projects". Both of these are from the same publisher and author as this publication. What we will concentrate on here are some less common types of circuit for processing the output of a synthesiser, and in particular designs for generating true and pseudo stereo signals will be discussed.

Pseudo Stereo

With most electronic music being recorded in stereo, or reproduced through stereo loudspeakers when played "live", the production of a good stereo image is a topic which is of great importance to many musicians. With a monophonic synthesiser, or a multichannel instrument which has a common audio output socket for all the channels, it is not possible to produce a genuine stereo output. Of course, it is possible to combine the outputs of several instruments to generate a real stereo output, or a pseudo stereo signal can be generated from a monophonic source. The latter is the subject that we will persue first.

The most simple form of stereo simulation is to simply reverse the phase of one channel. The stereo effect relies on the two loudspeakers being driven in-phase (i.e. the diaphragms of the two loudspeakers move backwards and forwards in unison, rather than with one going forwards as the other goes backwards). With the same signal applied to the loudspeakers and the two units driven in-phase, the sound seems to emanate from a point mid-way between the two loudspeakers. Making the volume from one loudspeaker higher than that from the other shifts the stereo image towards the loudspeaker which provides the higher volume level. In fact making one loudspeaker twice as loud as the other has the effect of moving the stereo image right over to the louder of the two units. Reversing the phase of one of the loudspeakers destroys the phase relationship needed to give a good stereo image, and tends to spread out the sound between the two loudspeakers, giving a very basic (and admittedly rather crude) form of pseudo stereo effect.

The most simple way of reversing the phase of one channel is to reverse the leads to one of the loudspeakers, and it does not matter whether this is the left or right speaker. This is often an inconvenient way of doing things in practice since the connections must be returned to the normal (in-phase) method of connection for normal stereo listening. The alternative is to add an inverter stage into the signal path to one input of the amplifier, as shown in the circuit diagram of Figure 52. This is just a basic operational amplifier inverting mode circuit, with negative feedback network R1 – R4 having values which give unity voltage gain.

This ultra-simple approach is not without its drawbacks, and the main one is that the sound tends to move away from the middle of the sound stage and seems to come predominantly from the two loudspeakers. This gives the so-called "hole in the middle" effect, and a what is often a rather unconvincing stereo effect.

There are other approaches to generating pseudo stereo signals, but these really boil down to just two basic methods. One method is based on phasing, and consists of more sophisticated versions of the technique described previously. The other relies on frequency selective channeling of signals to the two loudspeakers. In its most basic form the frequency selective

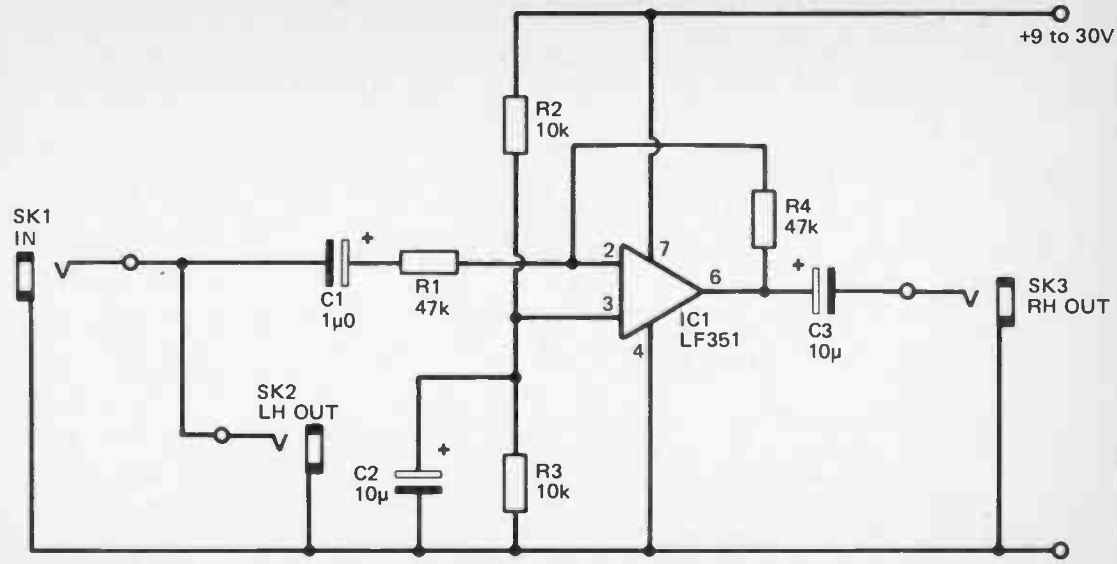


Fig. 52 The circuit diagram of the Inverter for stereo synthesis

approach has a highpass filter to channel signals at high-middle and treble frequencies to one loudspeaker, and a lowpass filter to channel low-middle and bass frequencies to the other. The two filters have complementary responses so that there is no overall effect on the frequency response of the system. In theory a fairly high pitched instrument appears in one channel while a low pitched type appears on the opposite side of the sound stage. Medium pitched instruments appear at or near the middle of the sound stage.

In practice this system often fails to give really convincing results. One problem is that only a very narrow range of frequencies give a central stereo image, and the "hole in the middle" effect is often very evident. Another is that any noise on the input signal of the white noise "hissing" variety is fed predominantly to one channel, giving a relatively poor signal to noise ratio from one channel and a very high signal to noise ratio from the other. This can be a little disconcerting when listening to a pseudo stereo system of this type. Also, any low frequency hum may be channelled to one loudspeaker, making its presence more obvious than would otherwise be the case.

Improved results can be obtained using two complementary comb filters. These are filters which have numerous peaks and troughs in their frequency responses, and the idea is to have the peaks of one filter matching the troughs of the other filter. This gives a reasonably flat overall frequency response, but also gives the required channeling of some frequencies to one channel and other frequencies to the second channel. As some high frequency signals go to one channel, and others go to the second channel, this arrangement does not suffer from the problem of having all the background "hiss" going to one channel. The only real drawback of this system is that it is relatively expensive as the two comb filter responses can not be obtained using very simple circuitry.

The best low cost approach that I have tried is a variation on the out-of-phase system. However, rather than simply having the two channels out-of-phase, improved results can be obtained by using a frequency selective phase shift circuit in one channel. The basic idea is to have the two signals in-phase at low frequencies. As the input frequency is increased the phase relationship is gradually reversed, taking the two signals out-of-phase. At still higher frequencies the two signals gradually slip into phase once again. In fact the system can use a multiple phase shifter circuit so that the signals repeatedly slip in and out of phase as the input frequency is increased, but quite good results can be obtained using just a couple of phase shift circuits.

The main point of this system is that it gives a combination of in-phase signals to give a strong central stereo image, and out-of-phase signals to spread the sound stage from one loudspeaker to the other. It consequently gives better results than the simple out-of-phase system, with the "hole in the middle" problem being absent. It is advisable to use a system that has the signals in-phase at low frequencies, as this gives a good bass response. With the signals out-of-phase at low frequencies there tends to be cancelling of bass signals, with the system having ar apparent lack of bass output.

Inverter Components (Fig. 52)

Resistors (all ¼ watt 5%)

R1,4	47k
R2,3	22k

Capacitors

C1	1µF 63V radial electrolytic
C2,C3	10µF 25V radial electrolytic

Semiconductors

IC1	LF351

Miscellaneous

SK1,2,3	Standard jack sockets

Circuit board, 8 pin DIL IC holder, wire, etc.

Phase Shifter Circuit

A simple phase shifter circuit for use as a stereo simulator is shown in the circuit diagram of Figure 53, IC1 merely acts as an input buffer stage, and it is IC2a and IC2b that act as the phase shifters. These use the standard configuration which is much used in phaser effects units, but in this case there is no need to sweep the operating frequency of the shifters, and so no voltage controlled resistances are required.

The two phase shifters are identical, and we will therefore only consider the operation of the first of these. At low frequencies C3 has an impedance which is extremely high in relation to R5, and C3 consequently has no significant effect on the circuit. IC2a then

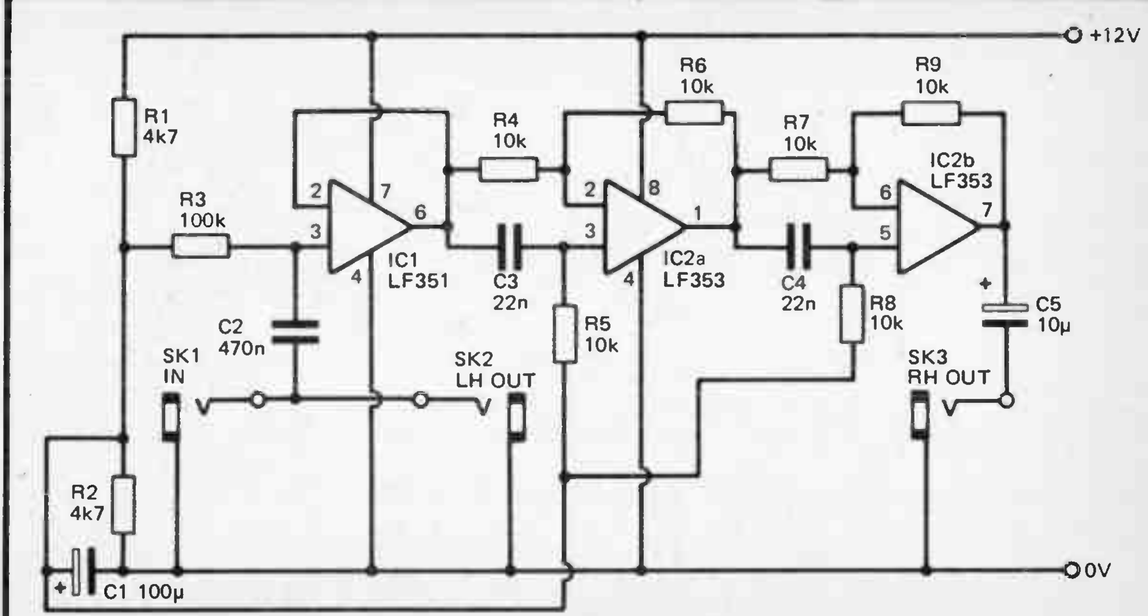


Fig. 53 The circuit diagram of the Phase-shift Stereo Simulator

operates as a straightforward inverting amplifier having unity voltage gain. At high frequencies C3 has a very low impedance, and effectively couples the input signal direct to the non- inverting input of IC2a. The circuit then operates as a non-inverting amplifier having unity voltage gain. What makes this type of circuit so useful is that at intermediate frequencies it works in what is a combination of the inverting and non-inverting modes, giving unity voltage gain and somewhere between zero and 180 degrees of phase shift.

Taking the overall effect of the two phase shifters, the double inversion at low frequencies gives no phase change through the circuit. At middle audio frequencies there is about 90 degrees of phase shift through each shifter, giving a total phase shift of 180 degrees, and taking the two output frequencies out of phase. The frequency at which precisely 180 degrees of phase shift is provided is approximately 1kHz, which is roughly in the middle of the audio range. At high frequencies there is no significant phase shift through either of the shifters, bringing the two pseudo stereo channels back in-phase again.

Slightly improved results can be obtained by adding more phase shifters into the circuit, but in order to maintain zero phase shift at low frequencies it is advisable to use pairs of phase shifters, and avoid odd numbers of shifters. Although the circuit is shown as being added in the right hand channel, it makes no difference to the effect which channel it is added into. Note though, that only one circuit added into one channel is required. Adding a phase shift circuit into both channels would simply result in the effect of one being cancelled out by the other, giving no pseudo stereo effect whatever.

Stereo Simulator Components (Fig. 53)

Resistors (all ¼ watt 5%)

R1,2	4k7
R3	100k
R4,5,6,7,8,9	10k

Capacitors

C1	100μF 16V radial electrolytic
C2	470nF miniature polyester
C3,4	22nF miniature polyester
C5	10μF 25V electrolytic

Semiconductors

IC1	LF351
IC2	LF353

Miscellaneous

SK1,2,3	Standard jack sockets

Printed circuit board
Two 8 pin DIL IC holders, wire, solder, etc.

Panning Mixer

If you have a number of channels available, then it is possible to mix these to give a genuine stereo signal. In its most fundamental form a stereo signal can be produced simply by feeding the output from one instrument into the right hand channel and the output from a second instrument into the left hand channel. This is preferable to mixing the two signals to produce a monophonic signal, but it gives a rather crude stereo effect with nothing at the centre of the sound stage.

An improved arrangement is to have three or more signal sources, with some signals being fed to one or other of the channels, and other signals being fed to both channels in order to give a central stereo image. Feeding a signal to both channels gives a very good central stereo image, and many professional recordings are made using a technique which consists basically of producing a three channel (left, right, and centre) tape, and then mixing it down into a conventional two channel stereo type. Conventionally the lead instrument or vocalist are positioned somewhere near the middle of the sound stage, with backing instruments or vocalists placed to the sides, simulating a typical stage set up during a "live" performance. However, the instruments can obviously be positioned wherever you like within the stereo sound stage, and there is no need to always opt for the conventional approach. It is not even necessary to have the instruments static within the sound stage, and some dramatic effects can be obtained by moving instruments within the sound stage. Like any effect though, it should be used sensibly and not to excess.

For this type of mixing many stereo mixers are far from ideal, especially the more simple types. The main problem with many is that they do not provide an easy means of panning a monophonic input signal across the sound stage, either for effect, or simply when initially setting everything up and deciding on the positions

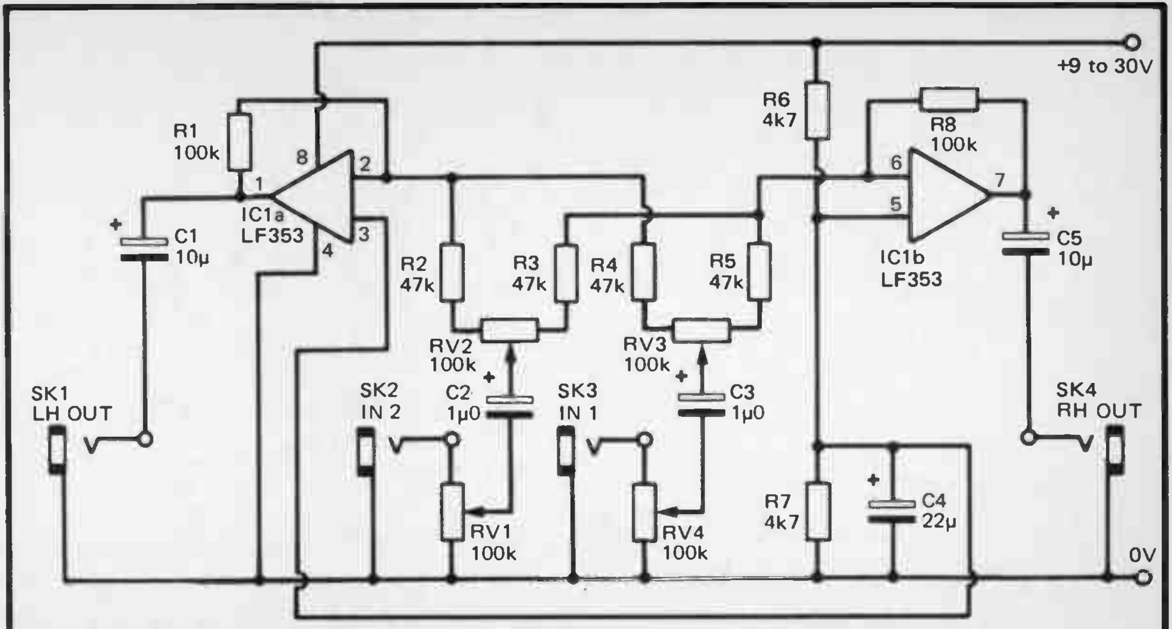


Fig. 54 The circuit diagram of the Panning Mixer

of the various instruments. The mixer circuit of Figure 54 has been designed specifically for electronic music applications, and as well as a level control for each input it also provides a panning control.

The circuit is basically two conventional summing mode mixers of the type described in Chapter 2, with one mixer being used in each stereo channel. There is a slight difference in this case in that the circuit is designed to operate from a single supply rather than dual balanced supply rails. R6, R7, and C4 are therefore used to provide a centre tap on the supply lines which is used for biasing purposes. Each input is taken to a volume control type "fader" potentiometer in the normal way, and these two controls are RV1 and RV4. The output of each fader control is taken to both mixer circuits, and both input signals are present at the output of each channel. With "pan" controls RV2 and RV3 at a middle setting the two signals are both balanced at the two stereo outputs, and the input signals appear at the centre of the sound stage. By moving the controls off the central setting the signals can be panned across the sound stage. At the extreme settings the pan controls provide about 6dB of boost in one channel and around 3db of cut in the other. If desired, a higher degree of separation can be achieved by making R2 to R5 somewhat lower in value, but the specified values give sufficient adjustment range to enable the input signals to be positioned anywhere within the sound stage. Also, when stereo recordings are made of "live" performances there is usually a fair amount of cross-talk due to sounds on one side of the stage being picked up by the microphone on the opposite side, and having a massive amount of stereo separation is not particularly authentic.

Although only two inputs are shown in Figure 54, any required number (within reason) can be added by including an input socket, fader potentiometer, panning potentiometer, and two 47k input resistors for each additional input.

Panning Mixer Components (Fig. 54)

Resistors (all ¼ watt 5%)

R1,8	100k
R2,3,4,5	47k
R6,7	4k7

Potentiometers

RV1,4	100k log
RV2,3	100k lin

Capacitors	
C1,5	10μF 25V radial electrolytic
C2,3	1μF 63V radial electrolytic
C4	22μF 16V radial electrolytic
Semiconductors	
IC1	LF353
Miscellaneous	
SK1,2,3,4	Standard jack sockets
Circuit board	
8 pin DIL IC holder	
Four control knobs	
Wire, solder, etc.	

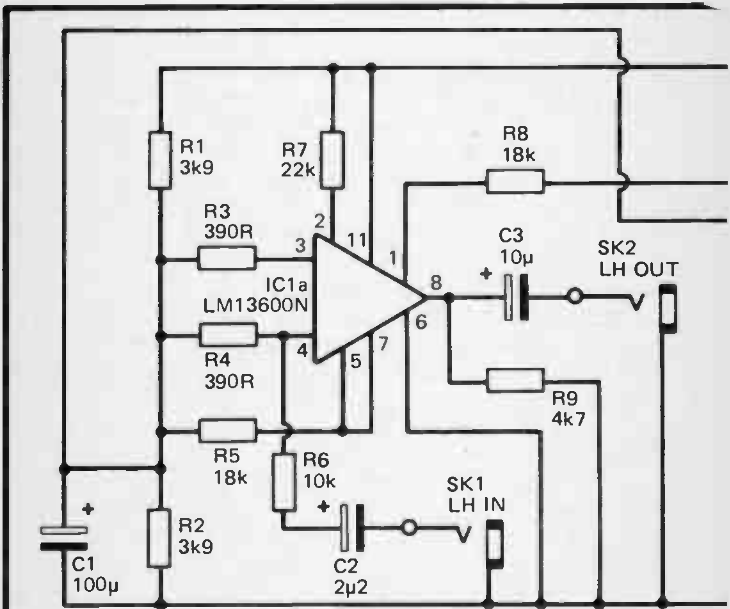
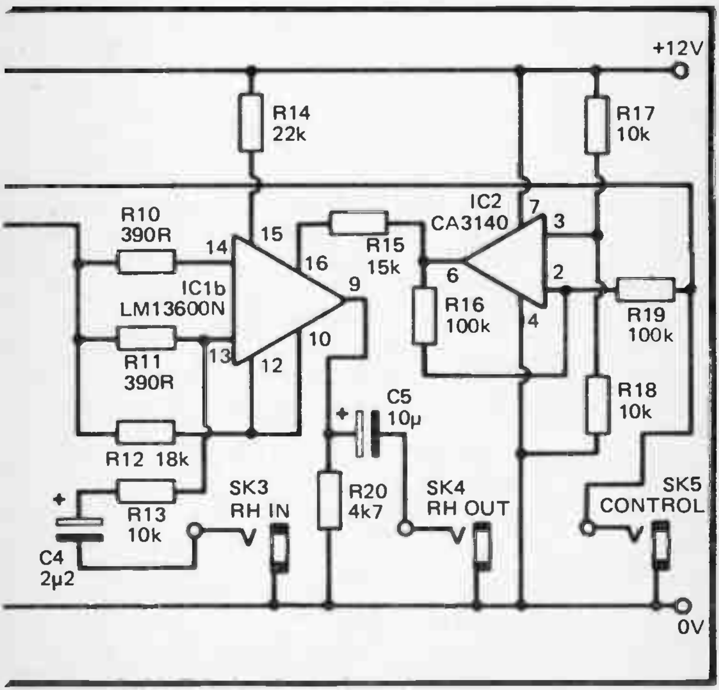


Fig. 55 The circuit diagram of the Voltage Controlled Panning unit

Voltage Control

A voltage controlled panning circuit can be useful on occasions, and circuits of this type can be reasonably simple. Figure 55 shows the circuit diagram for a voltage controlled panning circuit based on the two transconductance amplifiers in an LM13600N or LM13700N package.

The two transconductance amplifiers are used as straightforward voltage controlled amplifiers of the type featured in Chapter 2, with one VCA connected into each channel. The only slight complication is that the two VCAs require antiphase control voltages. In other words, as the control voltage to one VCA increases the control voltage fed to the other must decrease. This gives the required increase in gain on one channel and complementary decrease in gain on the other stereo channel, so that the signal moves from one side of the sound stage to the other.



One VCA is driven direct from the control voltage input, but the other is driven via IC2 which is wired as an inverting amplifier with a voltage gain of unity. The non-inverting input is biased to half the supply voltage, and the output voltage is therefore an exact complement of the input voltage (e.g. if the input voltage is 2 volts positive of the negative supply, the output voltage will be 2 volts negative of the positive supply rail). This gives the desired effect with the two VCAs being controlled in almost perfect antiphase fashion.

The circuit can operate as a manual panning control if the track of a 10k linear potentiometer is wired across the supply rails, and the wiper is used to drive the control input of the circuit. This does not really utilize the circuit's capabilities to the full though, and it can be used more effectively with the control voltage provided by an LFO or the output of an envelope generator. The effect obtained by driving the unit from an envelope generator can be extremely good, with the signal moving from one side of the sound stage to the other as the volume rises, and then back again as it decays. This type of effect is often used with noise based sounds, but it can also work quite well with other types of sound.

The circuit does not have to be used with the same signal fed to both signal inputs, and they can be fed from separate signal sources. When used in this way the unit is probably most effective with the two outputs mixed together in some way. The action of the unit is then to feed one signal through to the output when the control voltage is low, but as it is increased the second signal is gradually introduced, and a further increase causes the first signal to be faded out. This can give some interesting results when used in conjunction with suitably complementary signal sources.

Voltage Controlled Panning Components (Fig. 55)

Resistors (all ¼ watt 5%)

R1,2	3k9
R3,4,10,11	390R
R5,8,12	18k
R6,13,17,18	10k
R7,14	22k
R15	15k
R9,20	4k7
R16,19	100k

Capacitors

C1	100μF 16V radial electrolytic
C2,4	2μ2 63V radial electrolytic
C3,5	10μF 25V radial electrolytic

Semiconductors

IC1	LM13600N or LM13700N
IC2	CA3140E

Miscellaneous

SK1,2,3,4	Standard jack sockets

Printed circuit board
16 pin CIL IC holder
8 pin DIL IC holder
Wire, solder, etc.

Construction

The inverter, stereo simulator, and mixer circuits can easily be constructed on stripboard, including whatever number of channels you require in the case of the mixer circuit, and the required number of phase shifters in the case of the stereo

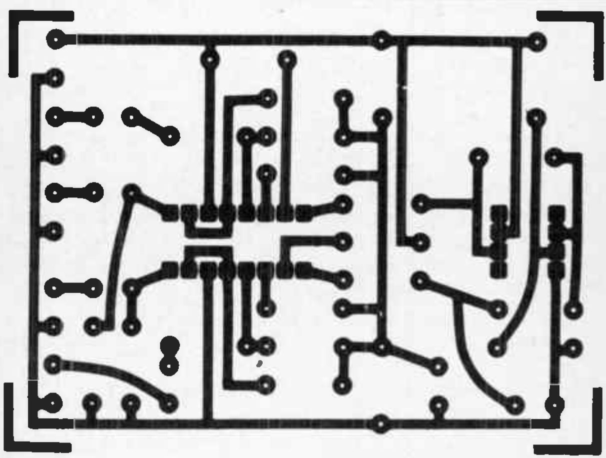


Fig. 56 The track pattern for the Voltage Controlled Panning unit

simulator. A suitable printed circuit design for the voltage controlled panning circuit is provided in Figures 56 and 57.

None of the circuits present any real constructional difficulties, but bear in mind that the CA3140E used in the Voltage Controlled Panning project is a MOS device which requires the normal antistatic handling precautions. The most popular constructional approach with mixers is to use a large sloping front case with slider potentiometers as the faders (and the panning controls in the case of the present mixer design). This is a slightly awkward approach from the constructional point of view since the slits for the slider potentiometers can be rather difficult to cut neatly. However, they can be made with the aid of miniature files, and neat results can be obtained if due care is exercised. Alternatively, ordinary rotary potentiometers can be used, and this is the type I prefer anyway, as precise adjustments seem to be slightly easier to make with this type.

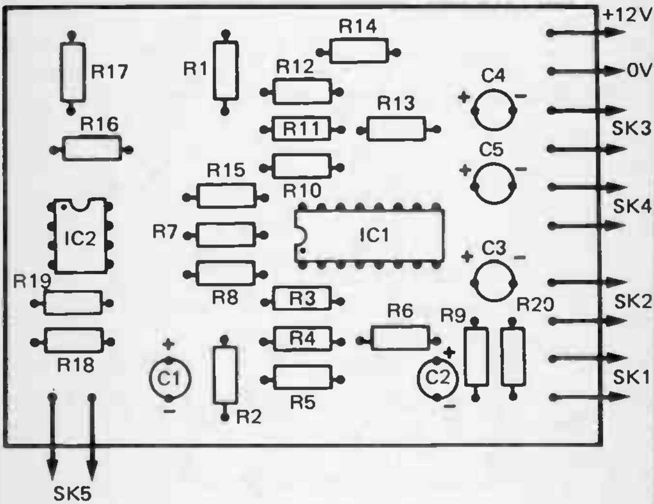


Fig. 57 The component overlay for the Voltage Controlled Panning unit

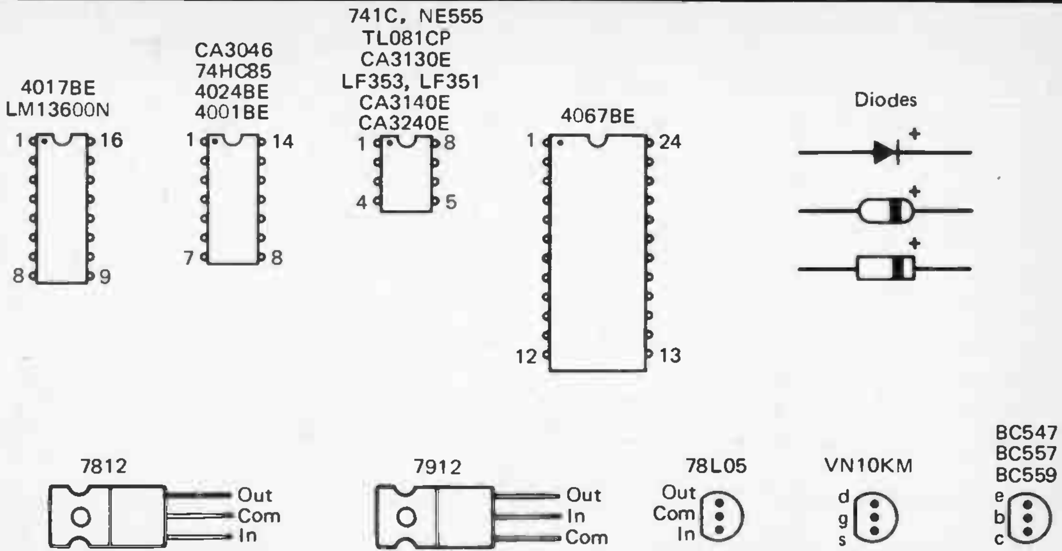


Fig. 58 Semiconductor leadout and pinout details

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R.A. Penfold

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