More Advanced Electronic Music Projects

R.A. PENFOLD

MORE ADVANCED ELECTRONIC MUSIC PROJECTS

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MORE ADVANCED ELECTRONIC MUSIC PROJECTS

by

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Preface

In "Electronic Music Projects" (BP74) a number of very simple electronic music circuits suitable for beginners were described. This book is intended to complement "Electronic Music Projects" by carrying on where it left off and providing a range of slightly more advanced projects. These include popular effects units such as flanger, phaser, mini-chorus, and ring modulator units. Some useful percussion synthesisers are also described, and together these provide a comprehensive range of effects including drum, cymbal, and gong type sounds.

The format is much the same as for "Electronic Music Projects" (BP74), with an introduction to each project plus circuit diagrams, circuit descriptions, and any necessary construction and setting up notes being provided. Printed circuit layouts and other detailed constructional information have not been included, and the projects are not really in the "dead simple" category as a certain amount of experience at electronics construction is really needed in order to tackle them confidently. However, anyone who has built up a few simple electronic projects should have little difficulty in building most of the designs featured in this book.

R. A. Penfold

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Chapter 1

EFFECTS UNITS

Electronic musical effects units have been popular with electronic musicians for many years, and have also ranked among the most popular of all electronic construction projects for many years. Their popularity shows no sign of waining, and no apologies are made for devoting a large part of this book to a number of effects units. Although effects units are primarily associated with electric guitars, their usefulness with keyboards and other electronic instruments should not be overlooked.

Phaser

Phasing is certainly one of the most useful of effects, and a simple phaser was described in "Electronic Music Projects" (BP74). The effect is generated using a "notch" filter, which is merely a type which allows signals at most frequencies to pass unhindered, but provides a high level of attenuation over a narrow band of frequencies. To generate the phasing effect the notch must be swept up and down over a large part of the audio frequency range. It is a simple single-notch phaser of this type that is featured in the "Electronic Music Projects" (BP74) book. However, high quality phasers produce a much better effect by having several notches in the frequency response which are spread over the audio frequency range and are swept in unison. Unfortunately, a multi-notch phaser is inevitably relatively complex, although not necessarily particularly expensive to build, but is certainly an effects unit that practically anyone involved in electronic music should be able to use to good effect.

A phaser can be based on a delay line, but the more usual arrangement is the one shown in the block diagram of Figure 1 which utilizes a series of phase shifters rather than a true delay line. Each phase shifter provides a degree of phase shift which varies with changes in input frequency. The phase shift starts at 180 degrees at low frequencies, and decreases to zero at high frequencies. The phase shifters are used in pairs connected in series, giving a total phase shift through each pair which varies from 360 degrees at low frequencies to zero at high frequencies. At some frequency between the two extremes the phase shift will be 180 degrees, and it is this phase shift that is of importance.

If we consider the operation of the unit, but ignore phase

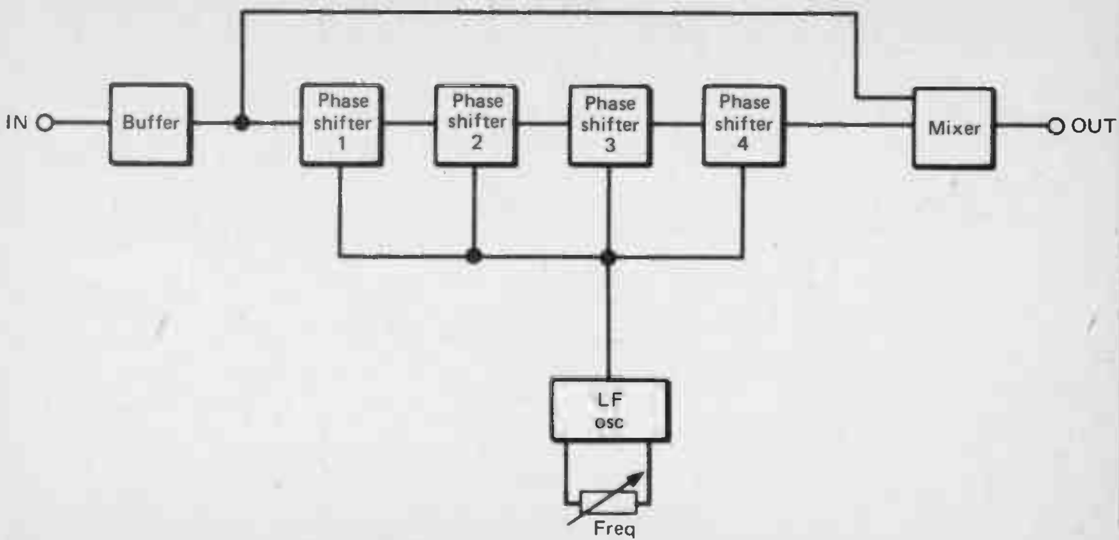


Fig. 1 Block diagram of the Phaser (twin-notch configuration)

shifters 3 and 4 for the moment, the buffered input signal is fed direct to one input of a mixer, and is also fed to the other input of the mixer but via a pair of phase shifters. At low and high frequencies there is no phase shift through the phase shifters, bearing in mind that a 360 degree phase shift brings the two signals back into phase again, and is effectively the same as zero phase shift. At these frequencies the two signals fed to the mixer are in-phase, and add together to give a strong output. Between the two extremes the signals are not perfectly in-phase, and at the frequency where the phase shifters give a 180 degree shift they are actually precisely out of phase. In other words, the two signals rise an fall exactly in unison, but they are always of the opposite polarity. They therefore precisely cancel out one another at the mixer, giving no output at all. At frequencies close to the 180 degree phase shift frequency there is partial cancelling, and the circuit provides strong attenuation.

This gives the required notch in the frequency response, but in order to give automatic sweeping of the notch the phase shifter circuits must be voltage controlled. The sweeping can then be achieved using a low frequency oscillator.

In the set-up of Figure 1 there are four phase shifters connected in series and not just two. This gives a total phase shift which varies from 720 degrees at low frequencies to zero at high frequencies. This gives a 180 degree phase shift at a certain frequency, and a 540 degree shift (360 degrees plus a further 180 degrees) at a somewhat lower frequency. In both cases the two signals reaching the mixer are out of phase and cancel each other out, giving notches in the frequency response. In fact any number of phase shifters can be added in series, and each additional pair will give an extra notch in the frequency response of the unit and a richer phasing effect. Two notches gives quite a good effect, but three are even better. Using more than three notches gives a richer effect, but it then starts to require more and more additional components in order to give an ever decreasing amount of improvement in the effect. Using a large number of phase shifters can also result in a significant degradation of the noise and distortion performance. Three notches probably represents the best compromise between cost and richness of effect, but it is up to the constructor to decide on the number of phase shifters to incorporate in the unit.

Circuit Operation

Figure 2 shows the circuit diagram of the buffer, mixer, and low frequency oscillator circuits. The circuit diagram for a pair of

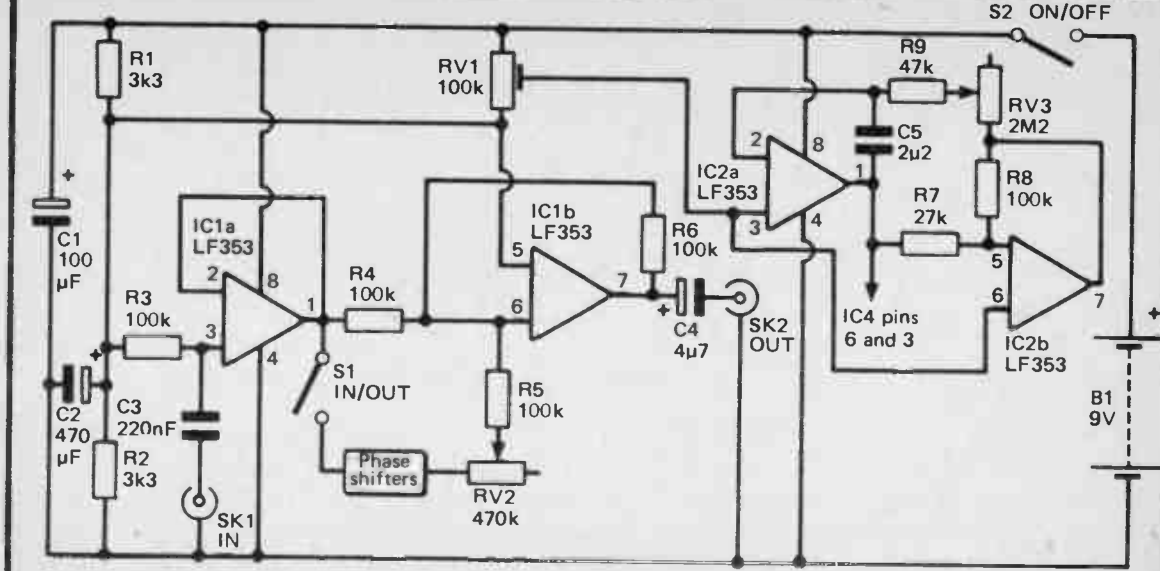


Fig. 2 The main circuit diagram of the Phaser unit

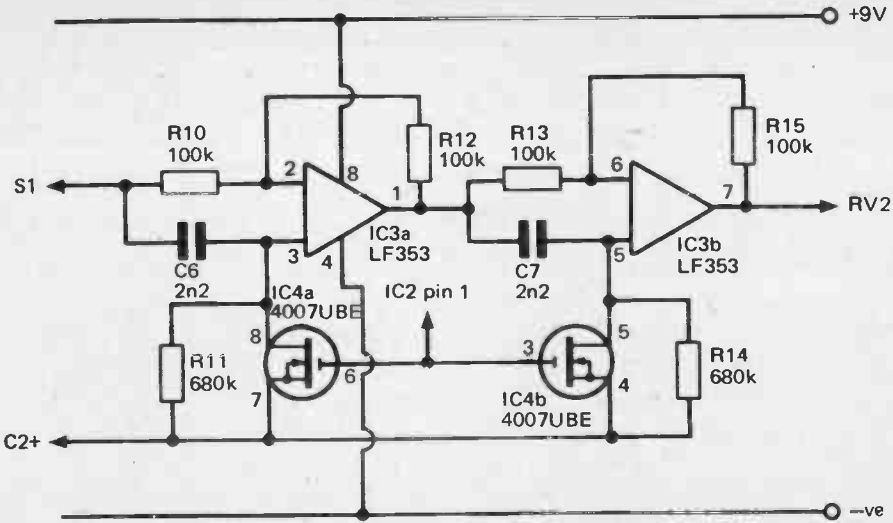


Fig. 3 The circuit diagram of the Phase Shifter

phase shifters is provided in Figure 3.

Starting with Figure 2, IC1a acts as the input buffer stage and this is a perfectly ordinary operational amplifier connected as a unity voltage gain non-inverting amplifier. In common with many of the other circuits in this book, rather than using dual balanced power supplies a single 9 volt battery supply plus a potential divider network to give a central 4.5 volt tapping for biasing purposes is utilized.

IC1b operates as the mixer stage, and this is a conventional operational amplifier summing mode circuit. Apart from providing a mixing action this circuit also acts as an output buffer stage which gives the unit a low output impedance.

With S1 in the open position the signal path through the phase shifters is broken and the phasing effect is switched out. With S1 closed, the phase shifted signal is coupled through to the mixer stage and the effect is generated. With RV2 set at minimum resistance the two signals reaching the mixer will be of equal amplitude, and at the out of phase frequencies precise cancelling will occur, giving notches of very deep attenuation. If RV2 is gradually adjusted for increased resistance the phase shifted signal is mixed with the straight-through signal at a decreasing level, producing only partial cancelling and notches of decreasingly deep attenuation. RV2 accordingly functions as a phasing "depth" control.

The low frequency oscillator is built around two more operational amplifiers, IC2a and IC2b, which are used in a well known triangular/squarewave oscillator configuration. IC2a operates as a Miller Integrator and IC2b functions as a Schmitt Trigger. The basic action of the circuit is for C5 to charge via R9 and RV3 from the output of IC2b, but the negative feedback over IC2a results in the end of C5 which connects to the inverting input of IC2a being maintained at a constant voltage. This voltage is equal to the bias voltage fed to the non-inverting input. Thus, as C5 charges, the output of IC2a goes increasingly negative in order to maintain the voltage balance at its inputs. The charge current into C5 is constant as there is no change in the voltage across R9 and RV3 during the charging process. The output voltage of IC2a therefore changes at a linear rate.

The charging process only continues until the output voltage of IC2a goes below the lower switching threshold of the Schmitt Trigger. The output of IC2b then triggers to the low state. The circuit then operates in much the same way as before, but C5 discharges through R9, RV3 and into the output stage of IC2b. In order to maintain the voltage balance at the inputs of IC2a the output of IC2a goes steadily positive, and this continues until the output voltage of IC2a goes above the upper switching threshold potential of the Schmitt Trigger. The output of IC2b then triggers to the high state, C5 charges via R9 and RV3, and the circuit is back in its original state. It oscillates indefinitely in this manner generating a linear triangular waveform at the output of IC2a, and a squarewave output signal from IC2b.

The modulation signals for most musical effects are either sinewaves or triangular waveforms, and both give good results. I prefer triangular modulation as this always gives a smooth and continuous modulation. With a sinewave, especially when using very low modulation frequencies, very little change is obtained during the peaks in the modulation signal where little voltage variation occurs. A squarewave signal is not suitable as it would simply switch the effect between two fixed states, which is not what is required (although you might like to experiment with this as it could have good creative possibilities). It is therefore the triangular waveform from IC2a which is used to control the phase shifters. RV1 is adjusted to give an output voltage range from the oscillator that is suitable for the phase shifters.

The output frequency of the low frequency oscillator can be varied by means of RV3 from about 10Hz at minimum resistance to 0.2Hz at maximum resistance (i.e. ten cycles per second to one cycle every five seconds). If very low modulation frequencies are required, and these can be quite effective with a multi-notch phaser, a parallel 2M2 resistor and SPST switch can be added in series with R9. When the switch is closed the oscillator will function normally. When the switch is open the frequency range will be modified to cover approximately one cycle every five seconds to one cycle every ten seconds.

Components for Phaser (Figs. 2 & 3)

Resistors (All ¼ watt 5% carbon)

R1,2,	3k3
R3,4,5,6,8,10*,12*,13*,15*	100k
R7	27k
R9	47k
R11*,14*	680k

Potentiometers

RV1	100k sub-min preset
RV2	470k linear
RV3	2M2 linear

Capacitors	
C1	100μF 10V elect
C2	470μF 10V elect
C3	220nF carbonate
C4	4μ7 63V elect
C5	2μ2 carbonate or polyester
C6*,7*	2n2 carbonate
Semiconductors	
IC1,2,3*	LF353 or TL082
IC4*	4007UBE
Miscellaneous	
S1	SPST heavy duty push button
S2	SPST toggle
SK1	Standard jack
SK2	Standard jack
B1	9 volt (PP7 size)

Metal case
Circuit board
14 pin DIL IC holder*
Two control knobs
Battery connector
Wire, solder, etc.

*indicates that a component is required in each phase shifter pair that is incorporated in the unit.

Phase Shifters

Each phase shifter consists of an operational amplifier plus three resistors and one capacitor. If we consider the one based on IC3a, C6 is the capacitor, R10 and R12 are two of the resistors, while the parallel resistance of R11 and the drain to source resistance of IC4a form the third resistor. IC4a is an enhancement mode N channel MOSFET from a CMOS 4007UBE dual complementary pair plus inverter. IC4a is the N channel device from one complementary pair, and IC4b is the N channel device from the other pair (which is used in the second phase shifter circuit). The other parts of the device are just ignored. Note that IC4 should be the unbuffered 4007UBE and not a 4007BE (buffered) version.

The purpose of using the drain to source resistance of a

MOSFET as one of the resistances is that it enables the resistance to be voltage controlled, and therefore permits the 180 degree phase shift frequency of the two inverters to be voltage controlled. The source of IC4a is taken to + 4.5 volts, and a gate voltage of around 4.5 volts or less results in it being cut off. In other words it has a very high drain to source resistance; so high in fact that R11 is needed to maintain a bias to the non-inverting input of IC3a. If the gate voltage is increased above 4.5 volts, IC4a starts to switch on and its drain to source resistance gradually falls, reaching a figure of just a few hundred ohms with the gate in the region of 2 volts higher than the source terminal (pin 7).

If we now consider the operation of the phase shifter, at very low frequencies C6 has a very high impedance and can be ignored. The circuit then functions as a straightforward unity voltage gain inverting amplifier. Taking things to the other extreme, at very low frequencies C6 couples the input signal strongly to the non-inverting input of IC3a, and the circuit operates as a non-inverting unity voltage gain amplifier. At intermediate frequencies a combination of the two operating modes is obtained, with somewhere between 180 and 0 degrees of phase shift. At a certain frequency the required 90 degree phase shift is obtained, and by varying the control voltage to IC4a this frequency can be moved over a large part of the audio frequency range.

Only one pair of phase shifters is shown in Figure 3, but by adding several of these circuits in series (i.e. connect IC3a of one circuit to R10 and C6 of the next) a multi-notch phaser can be produced. Note that one notch per pair of phase shifters is produced, and not one notch for each phase shifter.

Adjustment

There should be little difficulty in constructing this unit. One point to bear in mind is that IC4 is a CMOS device, and although not a particularly expensive type it would still be advisable to take basic antistatic handling precautions. Use a 14 pin DIL IC holder for this device, but do not fit it into the holder until all the wiring has been completed. Leave the component in the antistatic packaging (which usually takes the form of a plastic tube or conductive foam) until it is fitted into circuit. Handle the pins of the device as little as possible while it is being fitted into place.

For a project of this type it is standard practice to use a tough case of all metal construction (to provide screening against electrical interference such as mains "hum"). Diecast aluminium boxes are ideal. Toughness is important as S1 should be a heavy duty (successive operation) push button type mounted on the top of the box so that it can be operated by foot. A box of flimsy construction would simply collapse after a few operations of S1. S1 must be a heavy duty type or it too would be unlikely to survive for more than a few operations. It is common practice for the on/off switch of effects units to be a pair of make contacts on the input socket. The unit is then automatically switched on and off when a plug is inserted into and removed from SK1. If you like this method of doing things it is merely necessary to use a jack socket in the SK1 position which has suitable contacts and to use these for S2, although a socket having a pair of make contacts might prove to be elusive. However, a type having DPDT contacts should be readily available, and can be used with the unwanted contacts simply being ignored.

RV2 is a standard phasing depth control and RV3 is the phasing rate control. RV1 must be set up to give a good phasing effect with the notches swept smoothly over a large part of the audio frequency range. Setting RV1 for a bias potential that is too high will result in the notches never reaching middle and low frequencies, and and for much of the modulation cycle the notch frequencies will remain static. Adjusting RV1 for an inadequate bias potential has a similar effect, but with the notches never reaching high audio frequencies. Phasing works best on signals that have a wide spectrum of component frequencies, such as a low frequency pulse signal, noise, or the output from a polyphonic instrument. It is least effective with an instrument which provides a single sinewave tone, where the accepted phasing sound will not be obtained at all (it would be more like a temolo effect). When adjusting RV1 use an input signal which is rich in component frequencies, and then simply adjust RV1 for the best effect.

The unit has an input impedance of 100k and a low output impedance, and it should fit into most systems with no problems. Input levels of up to about 1 volt RMS or so can be accommodated, although in the interest of good distortion performance a somewhat lower level is preferable.

Parametric Equaliser

The sound of an instrument can often be changed quite dramatically by processing the output signal using a static filter of some kind. Bass and treble boosters are perhaps the best known types of simple filter effects units, but other types of filtering can be used to successfully modify the sound of an instrument and give interesting effects. The ideal tool for this type of thing is a graphic equaliser having around a dozen or more frequency bands, but units of this type are beyond the scope of most home constructors, and whether obtained ready made or built oneself, tend to be quite expensive.

There are two more simple but nevertheless quite effective alternatives to a high quality graphic equaliser. One is to use a relatively simple graphic equaliser with (say) five frequency bands, and the other is to use a parametric equaliser. Both approaches can give very useful results, and a parametric equaliser rather than a simple graphic equaliser is featured here merely because of my personal preference for the former.

For those who are unfamiliar with parametric equalisers it should perhaps be explained that this is a versatile filter which can be used to boost or attenuate a band of frequencies. Whereas a graphic equaliser has a number of controls each covering a small part of the audio range, a parametric equaliser can only provide boost or cut over a single band, but it can be tuned over a wide frequency range. This may seem to be less versatile than even a simple graphic equaliser, and in some respects it certainly is, but in many cases control of a single frequency band is all that is required. If control of two bands is needed it is perfectly feasable to connect two parametric equalisers in series, and they can then be tuned to precisely the required centre frequencies. A final point that should not be overlooked is the "Q" control of a parametric equaliser. This enables either a fairly wide frequency band to be covered (as in each channel of a simple graphic equaliser), or with the unit adjusted for a high Q value only a very narrow band of frequencies is affected. Some of the most interesting effects can be obtained using high Q settings.

Circuit Operation

Figure 4 shows the full circuit diagram of the Parametric Equaliser.

This is a fairly complex form of filter, but looking at things in broad terms, IC1a merely functions as a buffer stage which gives the unit a reasonably high (100k) input impedance and ensures that the filter proper is driven from a sufficiently low source impedance. IC1b, IC2a, and IC3 form a state variable filter circuit which acts only as a variable Q bandpass filter in this application. IC2b is a summing mode mixer circuit, and this combines the input signal with the filtered signal to either give in-phase mixing and bandpass filtering, or out of phase mixing and notch filtering.

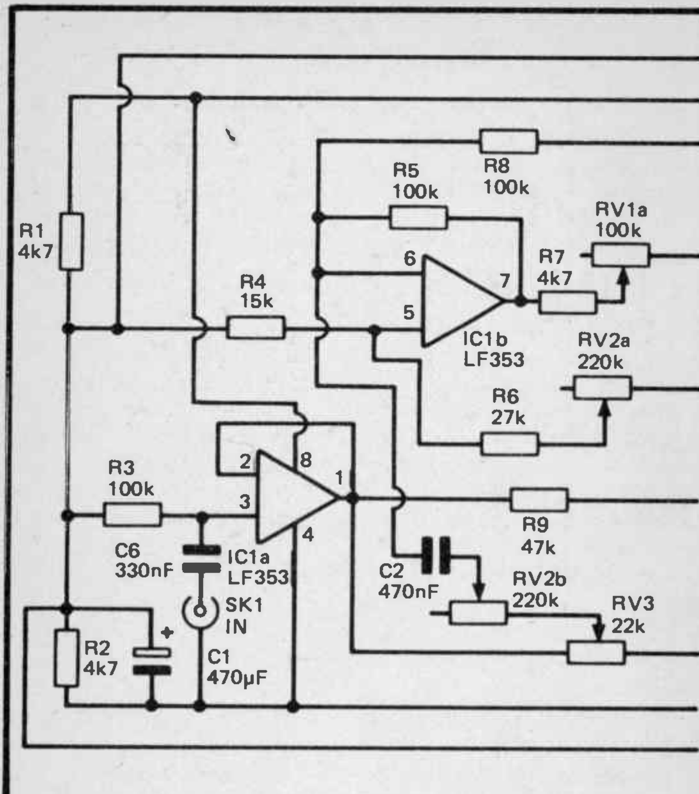
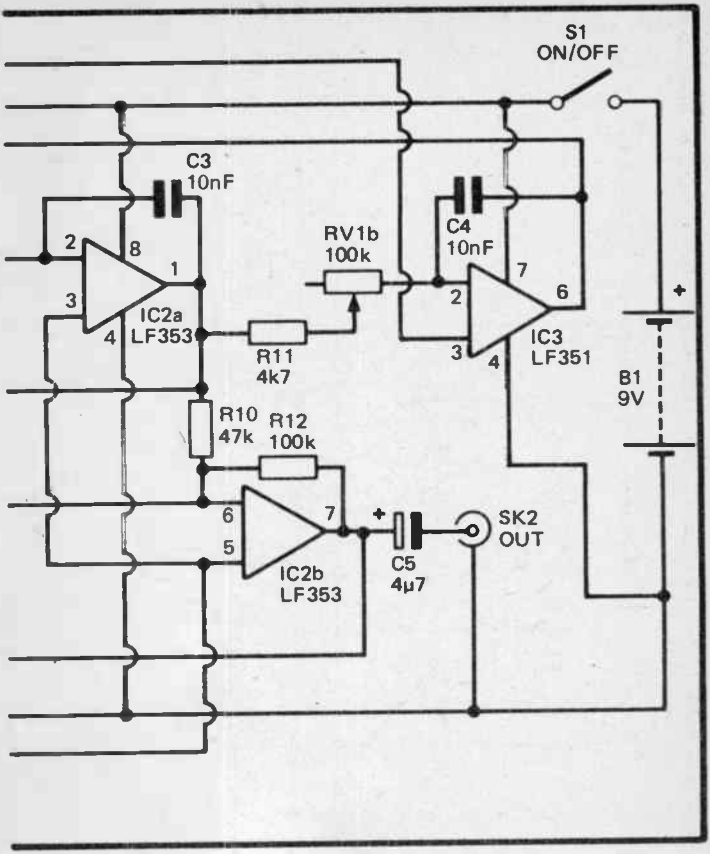


Fig. 4 The circuit diagram of the Parametric Equaliser

There are three controls, and RV1 is the tuning control. This gives a centre frequency range which varies from approximately 150Hz at maximum resistance to 3kHz at minimum resistance. RV3 is the boost/cut control. The level of boost and cut that can be provided depends to some extent on the Q of the circuit, but is generally around 15dB. Boost is obtained with the wiper of RV3 set towards the output of IC1a; cut is produced with its wiper towards the output of IC2b. The response is flat, regardless of the



Q and frequency control settings, with the slider of RV3 at the centre of its track. RV2 is the Q control, and it provides maximum Q value when set at maximum value. Figure 5 shows the basic effect on the frequency response that the Q control provides.

The unit should not be difficult to construct and requires no setting up once completed. As with virtually any effects unit, a little experimentation with the control settings will soon give an idea of the effects and sounds that can be achieved.

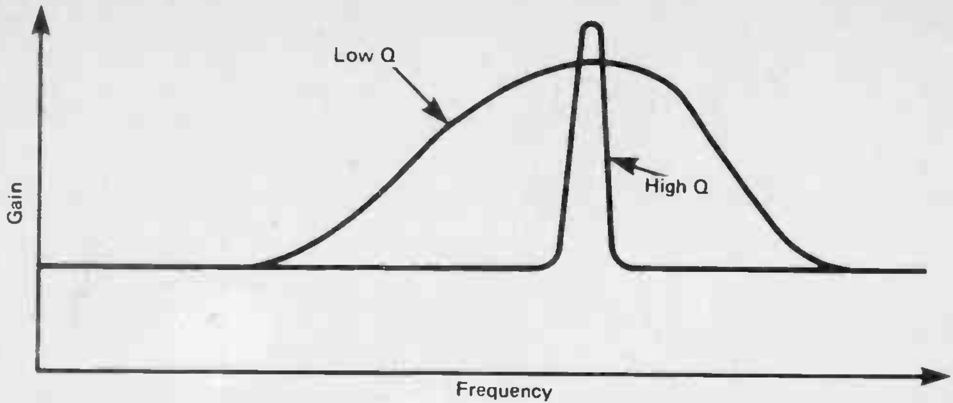


Fig. 5 A high Q setting gives a very narrow bandwidth to the filter

Components for Parametric Equaliser (Fig. 4)

Resistors (All ¼ watt 5%)	
R1,2,7,11	4k7
R3,5,8,12	100k
R4	15k
R6	27k
R9,10	47k
Potentiometers	
RV1	100k linear dual gang
RV2	220k linear dual gang
RV3	22k linear
Capacitors	
C1	470μF 10V elect
C2	470nF carbonate
C3,4	10nF carbonate
C5	4μ7 63V elect
C6	330nF carbonate
Semiconductors	
IC1,2	LF353
IC3	LF351
Miscellaneous	
SK1,2	Standard jack sockets
S1	SPST sub-min toggle
B1	9 volt (PP7 size)

Metal case
Circuit board, control knobs, battery connectors, wire, etc.

Shaped Fuzz

The fuzz effect must be one of the best known of the various musical effects, and could well be the best known of all. It is also one of the easiest to generate, as it is produced using the distortion that we normally try to avoid in audio circuits. There are two main ways of generating the fuzz effect, and one of these is to use the input signal to trigger a monostable multivibrator. The latter gives a series of brief output pulses at a frequency which is equal to the input frequency. This is a severe form of the fuzz effect which generates massive amounts of distortion.

Distortion levels using the second method are relatively low, but the distortion is still quite severe in absolute terms. The second method is the more common one, and it simply consists of using a clipping amplifier or other clipping circuit to flatten the peaks of the input waveform. Quite severe clipping is normally used, so that the output signal becomes virtually a squarewave or pulse signal whatever the input waveform happens to be. "Soft" clipping is sometimes used, and gives a good effect. With normal ("hard") clipping the amplitude of the output signal remains constant once the input signal has taken the circuit beyond the clipping threshold. With soft clipping the output signal does rise and fall in amplitude to a certain extent once the clipping threshold has been exceeded, but not by very much. This gives less distortion than hard clipping, and the higher harmonics are very much less strong with soft clipping. In absolute terms though, the level of distortion is still quite high, which is necessarily so with any effective fuzz unit.

One problem that is common to all simple forms of fuzz effect is the signal compression that is produced, with the envelope of the input signal being greatly modified. The output from a guitar rises almost instantly to full volume, and then decays at a fairly steady rate over a period of a few seconds (the exact decay characteristic varying somewhat from one guitar and pick-up to another). With a fuzz effect unit switched in, the instant attack is retained, but there is then little or no variation in volume until the input signal decays to practically zero. This gives a sound rather like a simple electronic organ, which you may or may not like.

If you do not like the conventional fuzz effect, or would simply like to have a variation on this effect, it is quite possible to have a fuzz unit which retains the original envelope shape of the input signal, or something which closely approximates to it anyway. Figure 6 shows the block diagram for the Shaped Fuzz Unit.

A buffer stage at the input provides the unit with a reasonably high input impedance and gives a low enough output impedance to drive the subsequent circuits properly. One of these circuits is a soft clipping amplifier which generates the basic fuzz effect. From here the signal is coupled to a VCA (voltage controlled amplifier), which is a circuit that provides a level of voltage gain that is determined by a control voltage. In this case the VCA would be more aptly called a voltage controlled attenuator rather than a voltage controlled amplifier, since it never provides more than about unity voltage gain, and provides heavy losses with small

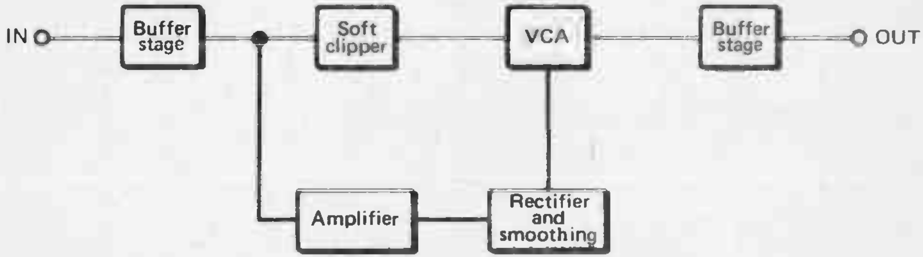


Fig. 6 Block diagram for the Shaped Fuzz unit

control voltages (up to at least 90dB). However, it is a form of amplifier and not a passive circuit, and devices of this type are generally refered to as voltage controlled amplifiers regardless of whether or not they provide any voltage gain.

Under quiescent conditions the VCA will have zero control voltage, and will provide a high level of attenuation. When an input signal is present a control voltage is generated by an amplifier plus a smoothing and rectifier circuit. This generates a control voltage which is roughly proportional to the amplitude of the input signal. The gain of the VCA therefore varies in sympathy with changes in the input signal's amplitude. Normally this would give a volume expander action, with variations in the input signal's dynamic levels being exagerated. This does not occur here due to the compression of the clipping circuit, which gives an output of almost constant amplitude while the input signal is above the clipping threshold. The gain variations from the VCA consequently do no more than restore the dynamic levels of the signal to their original levels. In practice there are slight imperfections in the circuit which result in some distortion of the dynamic levels, but the general envelope shape and character of the input signal is retained. The buffer stage at the output of the unit is needed in order to give the circuit a low output impedance.

Fuzz units often have a rather high noise level due to the fairly high level of voltage gain used in the clipping circuit. The system adopted here avoids this problem as the VCA gives a sort of noise gate action. In other words, under no input signal or low input signal conditions when the noise level of the clipping circuit would normally be at its most obvious the VCA provides a high level of attenuation which reduces the output noise to an insignificant level.

Circuit Operation

Figure 7 shows the complete circuit diagram of the Shaped Fuzz unit.

IC1a is the input buffer amplifier, and is an operational amplifier used as a unity voltage gain non-inverting amplifier. It gives the circuit an input impedance of just over 100k. IC1b is the clipping amplifier, and this is an operational amplifier used in the inverting mode. R3 and R6 set the voltage gain at 20 times (26dB), but the two germanium diodes in the feedback circuit (D1 and D2) introduce non-linear feedback that reduces the voltage gain as the input voltage rises, giving the required soft clipping effect. D1 and D2 can be replaced with silicon types such as 1N4148s if hard clipping is preferred. S2 can be used to switch out the fuzz effect by connecting R20 into the feedback circuit in place of R6 and the two diodes. IC1b then acts as a simple unity gain linear amplifier.

IC2 is an operational transconductance amplifier, and it functions as the VCA. Transconductance amplifiers have similarities to ordinary operational amplifiers, and have inverting and non-inverting inputs for instance, but there are a number of crucial differences, and they are used in very different configurations to ordinary operational amplifiers. The main factor to bear in mind is that they are current rather than voltage operated devices. Whereas an ordinary operational amplifier produces an output voltage which is determined by the differential input voltage, a transconductance amplifier produces an output current that is controlled by the differential input current.

This current operation is often inconvenient in normal use, but resistors can be used to effectively convert the device to voltage operation. In this circuit R7, R8, and C4 provide a bias voltage of 4.5 volts which is coupled to the inverting (pin 4) and non-inverting (pin 3) inputs of IC2 by way of R11 and R10. The input signal is coupled to the inverting input via series resistor R9, and as the current flow through this device is proportional to the applied voltage, this gives the conversion from current to voltage controlled operation at the input. R12 is connected at the output of the transconductance amplifier, and the voltage produced across this component is proportional to the output current. Again, this gives the conversion from current to voltage operation. Circuits using transconductance amplifiers often look strange to the unitiated since, unlike ordinary operational amplifiers, they often use no negative feedback over the amplifier. This circuit is no exception, and IC2 is used open loop.

The output impedance of a transconductance amplifier is often quite high, but IC2 includes a Darlington Pair which can be connected to operate as an emitter follower output buffer stage. This output stage is utilized here, and R15 is its load resistor. IC2 actually contains a second transconductance amplifier and output stage, but neither of these are required in this application and a number of IC2's pins are consequently left unconnected.

The transconductance amplifiers of IC2 incorporate linearising diodes at the input stage, and improved distortion performance

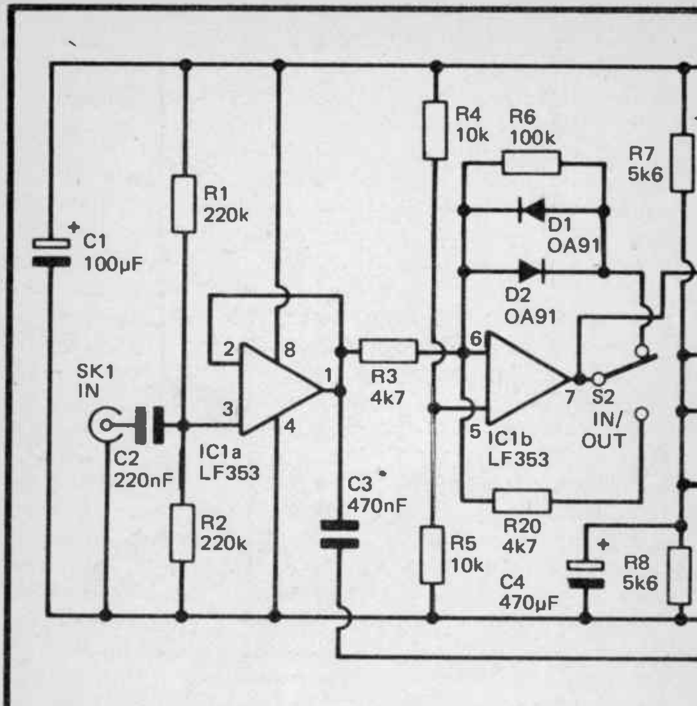
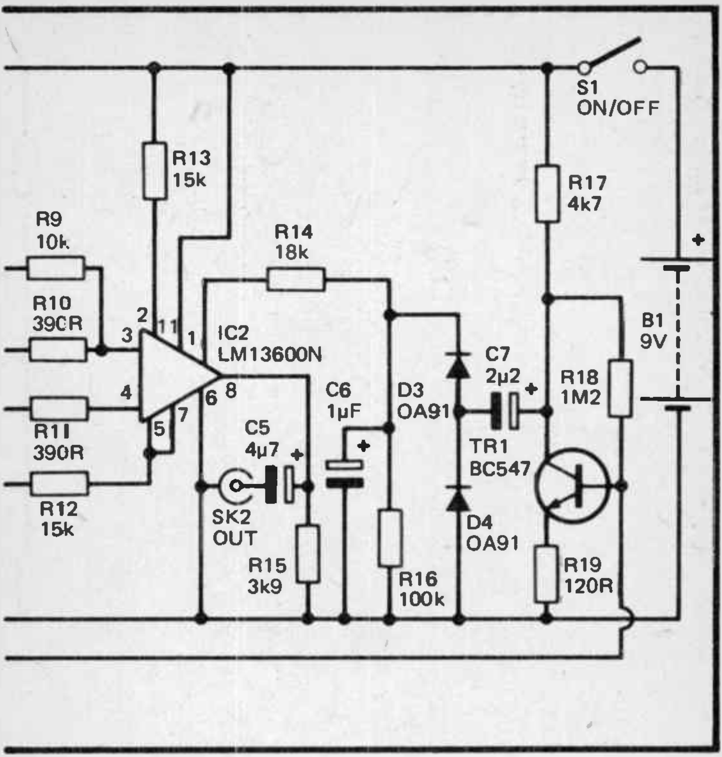


Fig. 7 The circuit diagram of the Shaped Fuzz unit

can be obtained by applying a small bias current to these. In this circuit R13 provides the bias current for the linearising diodes.

It was stated earlier that the output current of a transconductance amplifier is determined by the differential input current, which is true up to a point, but there is in fact a third input, the amplifier bias input, which controls the current gain of the device. The gain of the device is proportional to the input bias current, and by adding R14 in series with this input the circuit is effectively converted from current to voltage control.

Tr1 is the amplifier stage, and this functions as a simple common emitter amplifier. C7 couples the output of Tr1 to a straightforward rectifier and smoothing circuit, and the output of this feeds direct into R14.



Components for Shaped Fuzz (Fig. 7)

Resistors (All ¼ watt 5%)

R1,2	220k
R3,17,20	4k7
R4,5,9	10k
R6,16	100k
R7,8	5k6
R10,11	390R
R12,13	15k
R14	18k
R15	3k9
R18	1M2
R19	120R

Capacitors	
C1	100μF 10V elect
C2	220nF carbonate
C3	470nF carbonate
C4	470μF 10V elect
C5	4μ7 63V elect
C6	1μF 63V elect
C7	2μ2 63V elect
Semiconductors	
IC1	LF353
IC2	LM13600N
Tr1	BC547
D1,2,3,4	OA91
Miscellaneous	
SK1,2	Standard jack sockets
S1	SPST sub-min toggle
S2	SPDT heavy duty push button
B1	9 volt (PP7 size)

Metal case
Circuit board, control knobs, battery connectors, wire, etc.

In Use

In common with the previous project, and many of the others described in this book, the use of a diecast aluminium box as the case is strongly recommended. S2 should be a heavy duty push button type mounted on the lid so that it can be operated by foot. Although an LM13600N is specified for IC2, some component retailers sell the LM13700N which is virtually identical and will function perfectly well in this circuit (and the others in this book where an LM13600N is specified). Similarly, an LF353 is specified for IC1, but similar devices such as the TL082 and TL072 are equally suitable for use in this design, and in the other projects in this book where an LF353 is specified.

An important point to note is that the OA91 diodes used in this project are germanium types. These are much more easily damaged by heat than the more familiar silicon types, and due care should therefore be taken when soldering them into circuit.

The completed unit requires no setting up, and it will work well over a wide range of input levels. However, avoid input levels of more than about 2 volts RMS or the unit will become overloaded. If it is used with a very low output guitar pick-up it will probably be necessary to reduce the value of R19 slightly (say to about 47R) or the circuit may give a very low level of gain. It might also be necessary to raise R6 to about 470k in order to give a good fuzz effect.

A point to note if you are not familiar with fuzz units is that they are only really suitable for monophonic signals (i.e. one note at a time). With more than one note there is strong intermodulation distortion which gives non-harmonically related frequencies and rather non-musical sounds.

Envelope Modifier

The previous project was designed to retain the original envelope shape of the input signal. but interesting effects can be obtained by doing nothing more than modifying the envelope shape of a signal. With many instruments, particularly keyboard types, there is often no difficulty in varying the envelope shape with a fully adjustable ADSR (attack-decay-sustain-release) envelope shaper being provided. On the other hand, with some instruments, including electric guitars, the envelope shape is in no way adjustable, but it can be modified to some extent by an add-on processor.

Probably the most familiar type of envelope modifier is the sustain type, which is a form of compressor. This gives a more or less constant output level from the widely varying input signal level. There are other possibilities though, and the envelope modifier described here is primarily intended for use with an electric guitar to elongate its attack time. Normally an electric guitar has an extremely fast attack time, giving the characteristic "twangy" sound. By slightly slowing up the attack time a much less aggressive sound is produced, and by introducing a fairly slow attack time a quite weird effect is obtained.

Figure 8 shows the block diagram for the Envelope Modifier, and helps to explain the way in which the envelope shaping is achieved.

The main signal path is through an input buffer stage which gives a high input impedance, a VCA which provides the gain variations needed to give the modified envelope shape, and another buffer stage to give a low output impedance. Some of the output from the buffer stage is taken to an amplifier, and the amplified signal is then smoothed and rectified to give a DC bias

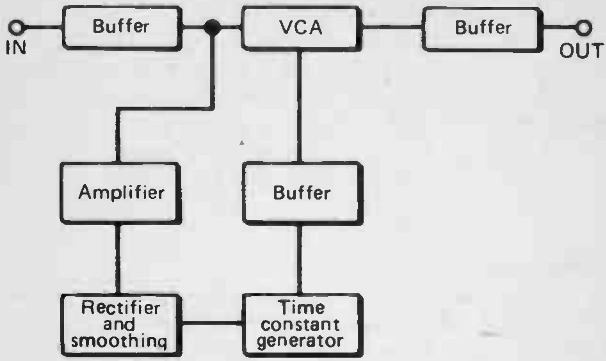


Fig. 8 Block diagram for the Envelope Modifier

that is roughly proportional to the average amplitude of the input signal. The values in the smoothing circuit are chosen to give quite fast attack and decay times so that the DC bias closely follows the envelope shape of the input signal.

The next stage in the side chain is a time constant generator. This has a relatively slow and variable attack time, so that although the output voltage from the smoothing circuit goes to a high level almost immediately when a new note commences, the output voltage from the time constant generator takes a much longer time to build up to its peak level. The output voltage from the time constant generator is fed to the control input of the VCA via a buffer amplifer. The gain of the VCA is initially very low, giving no significant output despite the high input level, but as the voltage from the time constant generator gradually builds up, the output level from the VCA increases. Eventually the gain of the VCA will reach its maximum level, and then the decaying signal level causes the output signal to gradually diminish. The output envelope shape is therefore variable from a moderately slow to a very slow attack time, followed by (more or less) the natural decay of the instrument.

In use each note will not normally be allowed to decay naturally, with each one (apart from the last one in a piece of music) being terminated prematurely to permit the next note to be played. This is not of great importance, but the circuit must be designed to deal with this situation, or the VCA control voltage from the end of one note will hang over to the beginning of the next note, preventing the slow attack from being achieved. This is prevented by having a fast decay time in the time constant circuit. Together with the fast decay time of the rectifier and smoothing circuit this ensures that the control voltage falls back to practically zero during the brief pauses that inevitably occur between notes.

Circuit Operation

This is another circuit which is based on an LM13600N transconductance amplifier, as can be seen by refering to the circuit diagram of Figure 9.

IC2 is the transconductance amplifier, and it is used in a VCA circuit which is essentially the same as that featured in the Shaped Fuzz circuit described previously. IC1 acts as the input buffer stage, and is a simple non-inverting unity gain amplifier which provides the circuit with an input impedance of about 50k.

Tr3 acts as the voltage amplifier stage, and this is a fairly conventional common emitter amplifier. The only slightly unusual aspect is the inclusion of RV2 in the emitter circuit. This provides a variable amount of negative feedback and acts as a simple preset gain control. Tr2 is an emitter follower buffer stage which operates here as a sort of half wave rectifier. R16 provides a small forward bias to Tr2, but this gives only a very low quiescent voltage at the emitter of Tr2, C7 couples the output of Tr3 to the input of Tr2, and negative half cycles switch off Tr2, but positive half cycles bias Tr2 into conduction and charge smoothing capacitor C6. The low source impedance provided by Tr2 gives this circuit a fast attack time, and the fairly low value of R15 gives the smoothing circuit an almost equally fast decay time.

The voltage generated across C6 is coupled to C5 via the fairly high resistance of RV1 and R14. This gives a quite slow attack time which varies from about 100mS with RV1 at minimum value to just over one second when it is at maximum value. An even slower maximum attack time could be obtained by using a 1M or 2M2 component in the RV1 position, but in practice an attack time of much over one second would rarely (if ever) be usable. Tr1 is the buffer stage used to drive the control input of the VCA. S2

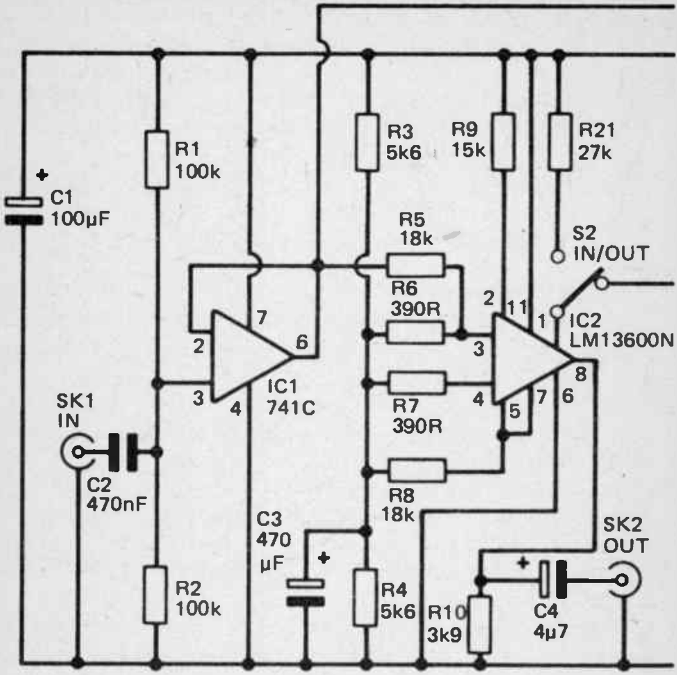
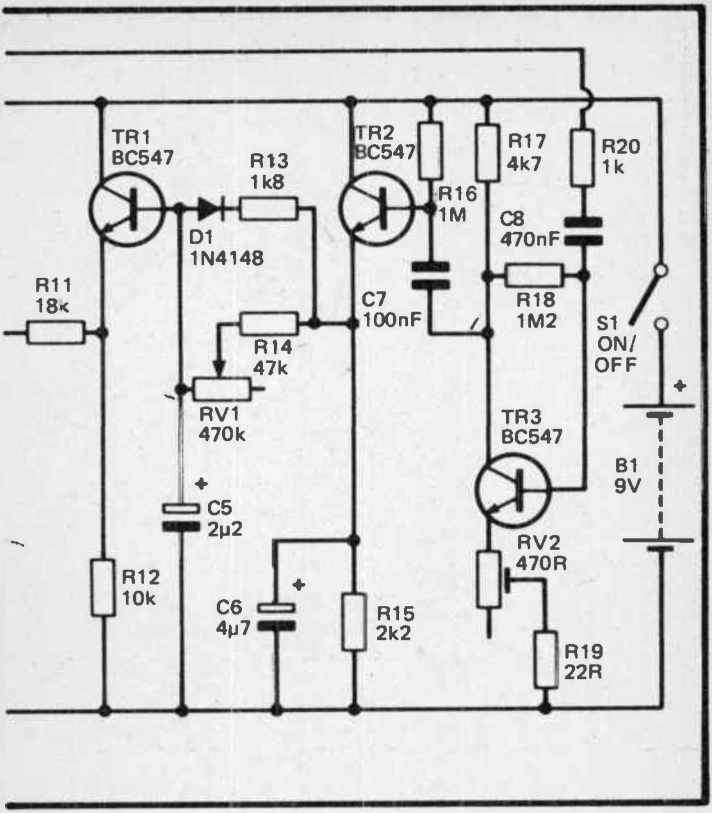


Fig. 9 The circuit diagram of the Envelope Modifier

enables a strong bias to be fed continously to the amplifier bias input of the VCA so that the envelope shaping can be switched out.

Construction of the unit should present no real difficulties. As with the previous project, the case should ideally be a diecast aluminium type and S2 should be a heavy duty SPST push button switch mounted on the lid. The only adjustment required to the finished unit is to give RV2 a suitable setting. With high level input signals a low gain setting (RV2 near maximum resistance) should suffice, but with low level inputs a high gain setting will probably



be needed in order to give satisfactory results. It is really just a matter of trying a few settings in order to find one that gives good results with the particular signal source you are using, and the setting of RV2 should not be at all critical.

When using the unit with a very long decay time bear in mind that it will not be possible to obtain satisfactory results if you play a fast succession of notes. Each note would then have barely become audible before it was terminated, giving very little output from the unit. Like any effects unit, the envelope modifier must be used sensibly and creatively if it is to be of full worth to the user.

Components for Envelope Modifier (Fig. 9)

Resistors (All ¼ watt 5%)

R1,2	100k
R3,4	5k6
R5,8,11	18k
R6,7	390R
R9	15k
R10	3k9
R12	10k
R13	1k8
R14	47k
R15	2k2
R16	1M
R17	4k7
R18	1M2
R19	22R
R20	1k
R21	27k

Potentiometers

RV1	470k linear
RV2	470R sub-min preset

Capacitors

C1	100μF 10V elect
C2,8	470nF carbonate
C3	470μF 10V elect
C4,6	4μ7 63V elect
C5	2μ2 63V elect
C7	100nF carbonate or polyester

Semiconductors

IC1	741C
IC2	LM13600N
Tr1,2,3	BC547
D1	1N4148

Miscellaneous

SK1,2	Standard jack sockets
S1	SPST sub-min toggle
S2	SPDT heavy duty push button
B1	9 volt (PP7 size)

Metal case
Control knob, battery connectors, wire, solder, etc.

Split Phase Tremolo

Tremolo is one of the oldest of electronic music effects, and it is also an extremely simple type. It is produced by varying the amplitude of the input signal rhythmically, and it can be generated manually using a swell pedal. However, it is usually more convenient to use an electronic circuit to produce the effect automatically, and this also allows higher modulation rates to be achieved. Split phase tremolo differs only from ordinary tremolo in that two signals are amplitude modulated using the same modulation signal, but out of phase. In other words, as one signal is brought up in volume the other is reduced in volume, and vice versa.

This effect is normally used with a stereo signal so that first one channel becomes dominant, then the other, and so on. If the same signal is fed to both inputs the effect is to automatically pan the signal signal from side to side across the stereo sound stage, which

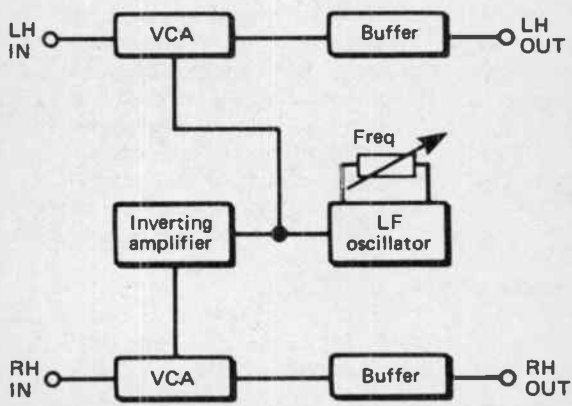


Fig. 10 Block diagram for the Split Phase Tremolo

can give quite stunning results if used thoughfully, or, like most effects units, can be extremely boring if is used to excess.

The block diagram of Figure 10 shows the make-up of the unit. Each input signal is processed by a VCA plus a buffer stage at the output to give a low output impedance. One VCA is driven direct from the output of a low frequency oscillator which has a triangular output waveform, and this gives a smooth variation in the amplitude of the output signal. The modulation frequency is made variable, and optimum results are generally obtained with a modulation frequency of around one or two Hertz. The second VCA is driven from the oscillator via a unity gain inverting amplifier so that the required antiphase operation is obtained.

Figure 11 shows the circuit diagram of the signal processing stages of the unit while the oscillator and inverting amplifier circuit appears in Figure 12.

Starting with the VCAs and buffer stages, these are much the same as the ones used in previous projects, but as two VCAs and buffer amplifiers are required, both sections of the LM13600N transconductance amplifier are utilized. S2 can be used to disconnect the control inputs of the VCAs from the oscillator circuit and provide a fixed bias, cutting out the tremolo effect when it is not required.

The oscillator circuit is also much the same as the ones featured in earlier designs described in this book, being a standard triagular/squarewave type. RV1 gives a frequency range of approximately 0.2Hz to 10Hz. It would be possible to modify the circuit to give lower modulation frequencies, but this is probably not worthwhile as tremolo effects are generally not very impressive at very low modulation rates.

The inverting amplifier is a standard operational amplifier unity gain inverting type. It is biased from the same source (R1 and R2) as the oscillator circuit to ensure that its output is an accurate complement of the oscillator signal.

Once again there should be no constructional difficulties and the unit is perfectly straightforward. No setting up of any kind is required once it has been completed. The input impedance is not very high at about 10k, but this is adequate for the vast majority of likely signal sources. Input signals of up to about 1.5 volts RMS can be tolerated without the processed signals becoming clipped and seriously distorted.

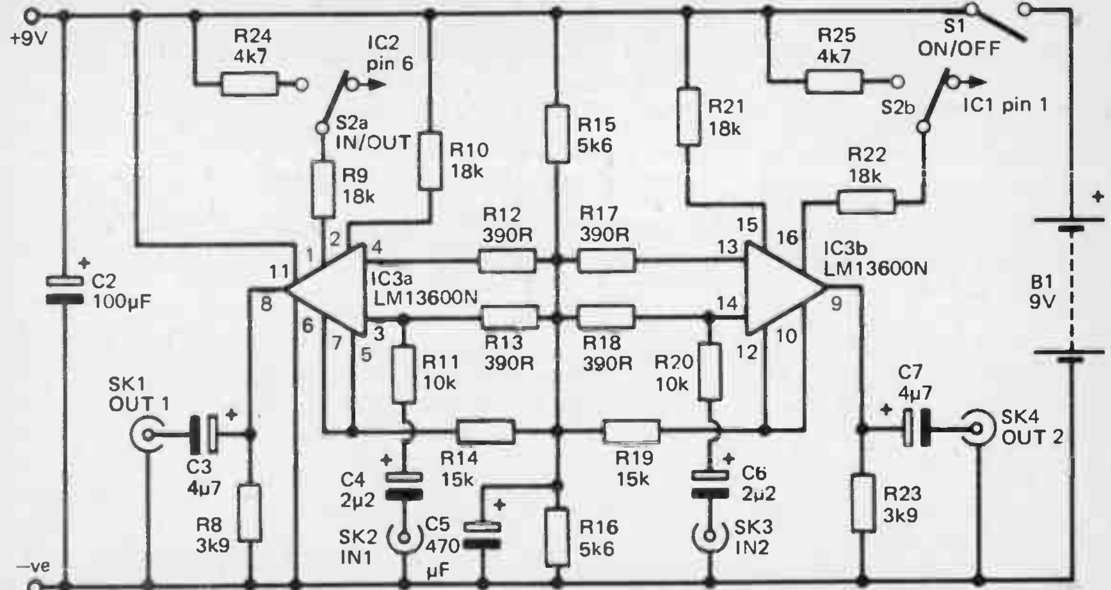


Fig. 11 The VCA stages of the Split Phase Tremolo

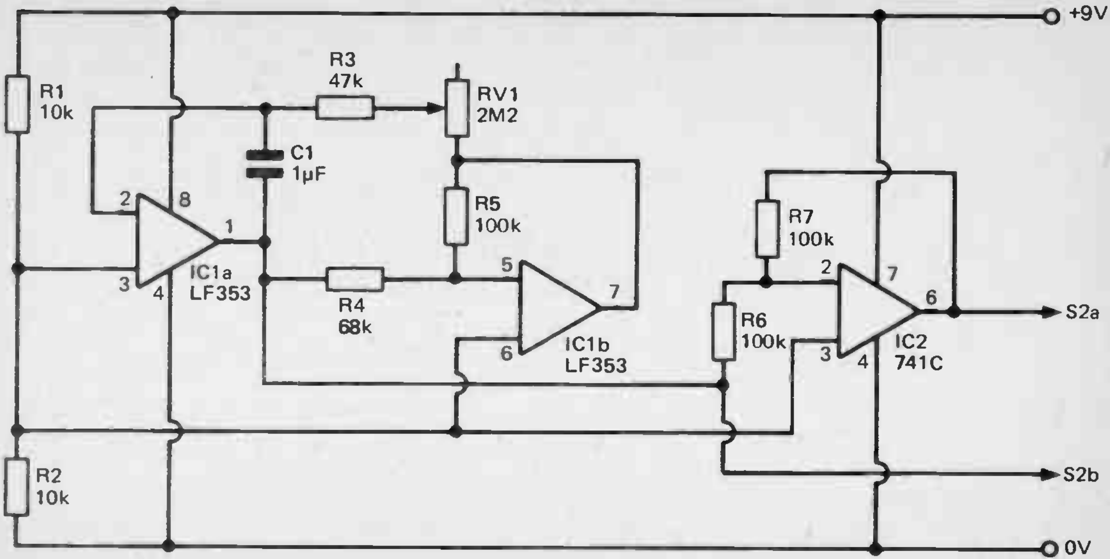


Fig. 12 The oscillator stage of the Split Phase Tremolo

Components for Split Phase Tremolo (Figs. 11 & 12)

Resistors (All ¼ watt 5%)	
R1,2,11,20	10k
R3	47k
R4	68k
R5,6,7	100k
R8,23	3k9
R9,10,21,22	18k
R12,13,17,18	390R
R14,19	15k
R15,16	5k6
R24,25	4k7
Potentiometer	
RV1	2M2 linear
Capacitors	
C1	1μF carbonate
C2	100μF 10V elect
C3,7	4μ7 63V elect
C4,6	2μ2 63V elect
C5	470μF 10V elect
Semiconductors	
IC1	LF353
IC2	741C
IC3	LM13600N
Miscellaneous	
SK1,2,3,4	Standard jack sockets
S1	SPST sub-min toggle
S2	DPDT heavy duty push button
B1	9 volt (PP7 size)

Metal case
Circuit board, battery connectors, control knob, wire, etc.

Ring Modulator

Tremolo is a fairly gentle form of effects unit which does not drastically alter the character of the processed signal, but this is certainly not something that can be said of ring modulation. It is a rather extreme effect which can easily produce decidedly non-musical sounds if it is not used carefully. On the other hand, it can produced some excellent effects when used thoughtfully.

A ring modulator is a form of mixer, but not a mixer in the sense of an ordinary audio type like the summing mode mixer circuit feature in an earlier project. It is a circuit which is more closely related to the mixers used in radio transmitters and receivers, and is a form of balanced mixer. An alternative name for circuits of this type is "four quadrant multiplier". What ever name is used, a circuit of this type generates sum and difference frequencies from two input signals. and things are quite straightforward if the input signals are both sinewave signals which each contain just a single frequency component. As a simple example, assume that signals at frequencies of 100Hz and 500Hz are fed to a ring modulator. The sum frequency is 600Hz (500Hz + 100Hz = 600Hz), and the difference frequency is 400Hz (500Hz − 100Hz = 400Hz). In theory neither of the input signals appear at the output of the circuit, although in practical circuits it is normal for at least one of them to break through at a low but significant level.

In a practical set-up it is unlikely that the input signals would be pure sinewaves, but would probably be complex signals containing a fundamental plus quite strong harmonics over a broad frequency range. The two signals then combine to produce a complex output signal which contains a mass of frequency components. As far as the sound of the output signal is concerned, it is very different to any sound produced by a single oscillator, or a conventional musical instrument which relies on a vibrating string or tube for sound generation. In both cases the spectrum of frequencies on the output signal consists of a fundamental signal plus (usually) some harmonics (signals at multiples of the fundamental frequency). The output from two ring modulated oscillators consists of signals that are to a large extent non-harmonically related, giving what can often be a rather discordant and non-musical sound. If the two oscillators are spaced some musical interval apart (and perhaps just slightly off tune as well in order to give a low frequency beat note), the output can be a very rich musical sound.

In fact ring modulation is primarily used to give so called "metallic" sounds. Metal instruments such as gongs and bells are two or three dimensional, whereas stringed instruments, woodwind instruments, etc., use a vibrator that is essentially one dimensional. This gives stringed and similar instruments a predominance of harmonically related frequency components on their output, while gongs and bells tend to produce mainly non-harmonically related frequency components. Ring modulation can therefore be used to generate quite good bell and gong type sounds.

Circuit Operation

Figure 13 shows the circuit diagram of a ring modulator having a built-in audio oscillator feeding into one of its inputs. The other input is free for use with the output of a guitar, synthesiser, or any other signal source with which you care to experiment.

The audio oscillator has the same basic configuration as the low frequency types featured in previous projects in this book. It differs only in that the timing component values have been modified to give a higher output frequency range of roughly 200Hz to 4kHz. This should be adequate for most purposes, but by making C2 somewhat lower in value and using a 1M component for RV1 it would be possible to cover virtually the whole audio frequency range if desired. In Figure 13 the triangular output signal from pin 1 of IC1a is shown connected through to one input of the ring modulator, but the squarewave signal from pin 7 of IC1b can be used if preferred. Both types of signal usually give good results in this application.

The ring modulator is based on the transconductance amplifier and output buffer stage in IC2a. This device is connected in what is almost the usual VCA configuration, but R11 is taken to the input signal rather than just being biased from R6 and R7. The input signal is taken to the inverting input of IC2a, and the coupling through R11 to the output of transconductance amplifier therefore has a cancelling effect on the signal here. With a certain bias level fed to the control input the two signals will in fact precisely cancel out one another, but varying the control signal either side of this point unbalances the circuit and produces an output signal (the required sum and difference signals). The output of the built-in oscillator is therefore fed to the control input of IC2a, and RV2 is adjusted to give minimum breakthrough of the input signal at the output, although less than optimum balancing can be used if the effect this gives is preferred.

Noise Gate

In theory the circuit of Figure 13 is perfectly satisfactory, but in practice it has a slight flaw that renders it of little practical value.

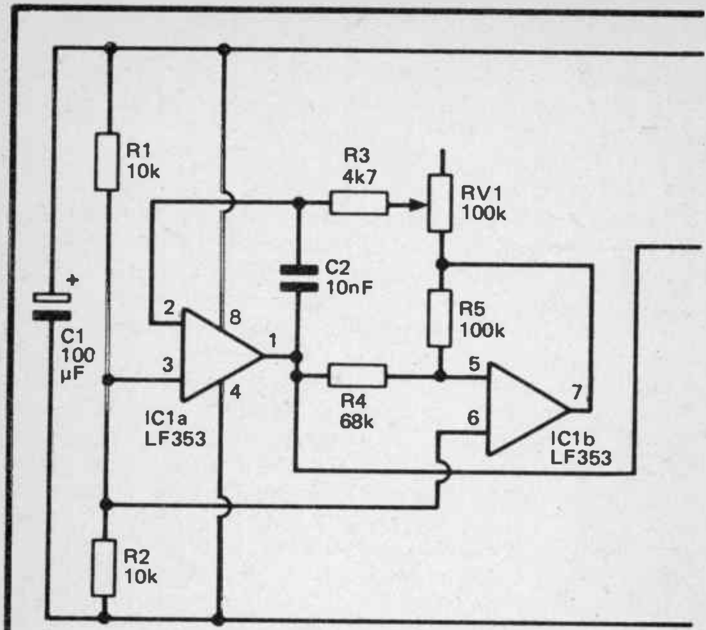
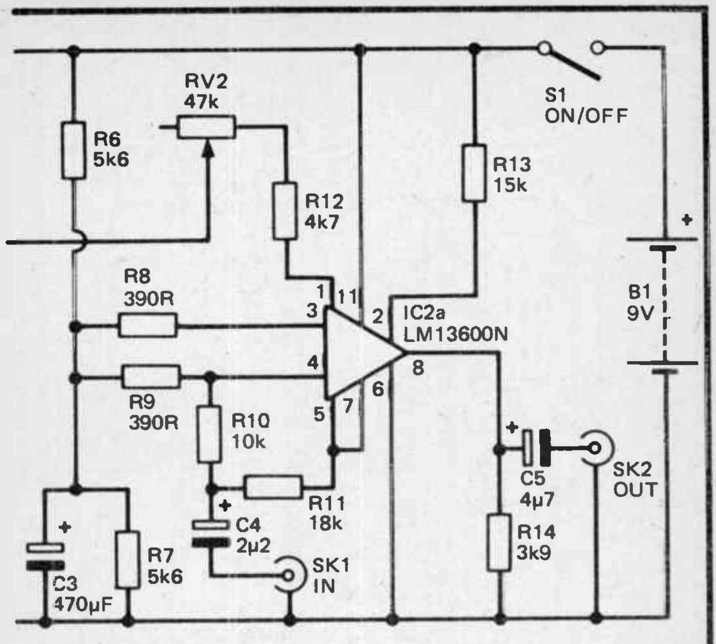


Fig. 13 The circuit diagram of a Ring Modulator

This is a slight but significant breakthrough of the signal from the built-in oscillator at the output of the unit. A simple way of eliminating this breakthrough is to include a noise gate at the output of the unit, and the unused section of IC2 is the obvious basis for this circuit. Figure 14 shows the circuit diagram of a suitable add-on noise gate for the Ring Modulator project.

IC2b functions as a standard VCA which is fed direct from the output from the ring moulator. Some of the ring modulator's output signal is fed via RV3, R20, and C6 to a common emitter amplifier built around Tr1. The output from Tr1 is smoothed and rectified and the resultant DC bias is used to provide the control voltage for the VCA.

RV3 is adjusted so that under quiescent conditions the control voltage produced by the rectifier and smoothing circuit produces strong attenuation through the VCA, and prevents the breakthrough at the output of the ring modulator from giving an



plus Audio Oscillator

audible output from the noise gate. On the other hand, only a slightly higher input level is needed in order to produce a strong bias voltage which enables the output from the ring modulator to pass unhindered to the output of the noise gate. Strictly speaking the circuit is an expander rather than a noise gate since the VCA is not switched between the fully on and fully off states, and over a narrow range of input signal levels it does provide intermediate gain levels. However, this gives better results in the present application, and as the circuit has little effect on the dynamic levels of the processed signal its more like a noise gate than an expander as far as the user is concerned.

When adjusting RV3, start with it at miniumum value. This should result in the breakthrough from the ring modulator being clearly detectable on the output of the noise gate. Carefully adjust RV3 for increased resistance until the breakthrough is reduced to an inaudible level. RV1 and RV2 are simply adjusted to give the

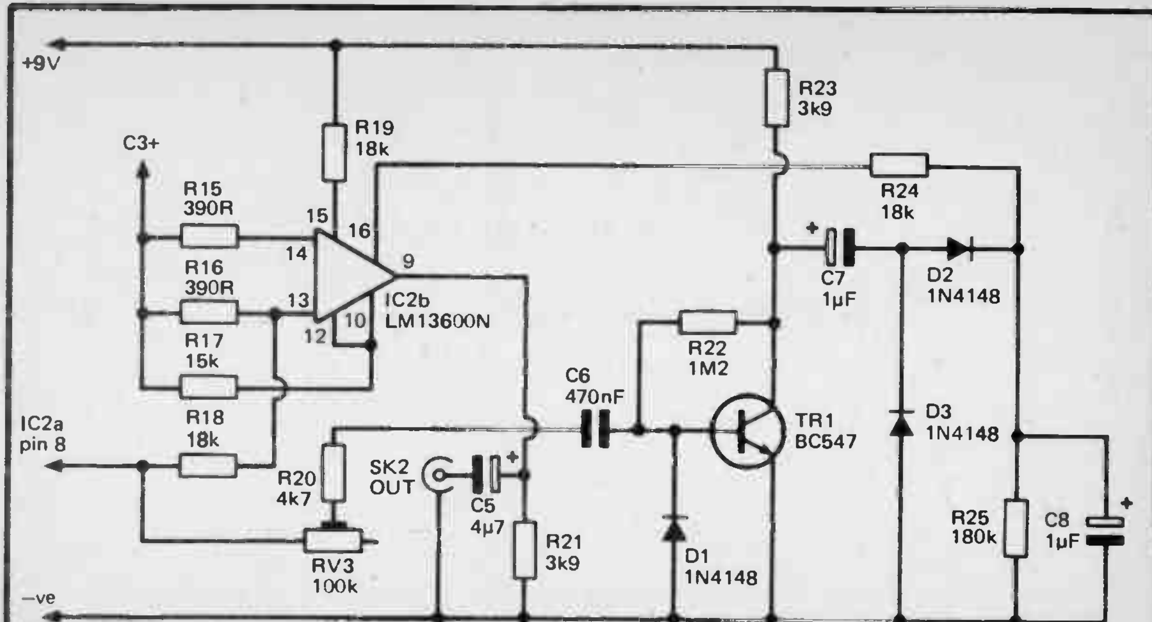


Fig. 14 Adding a Noise Gate circuit to the Ring Modulator. C5 and SK2 are moved to new positions shown here

effect you like best, and some experimentation with these and various input signals will soon demonstrate the range of sounds obtainable.

Components for Ring Modulator (Inc Noise Gate)
(Figs. 13 & 14)

Resistors (All ¼ watt 5%)	
R1,2,10	10k
R3,12,20	4k7
R4	68k
R5	100k
R6,7	5k6
R8,9,15,16	390R
R11,18,19,24	18k
R13,17	15k
R14,21,23	3k9
R22	1M2
R25	180k
Potentiometers	
RV1	100k linear
RV2	47k linear
RV3	100k sub-min preset
Capacitors	
C1	100μF 10V elect
C2	10nF carbonate or polyester
C3	470μF 10V elect
C4	2μ2 63V elect
C5	4μ7 63V elect
C6	470nF carbonate
C7,8	1μF 63V elect
Semiconductors	
IC1	LF353
IC2	LM13600N
Tr1	BC547
D1,2,3	1N4148
Miscellaneous	
SK1,2	Standard jack sockets

S1	SPST sub-min toggle
B1	9 volt (PP7 size)

Metal case
Circuit board, control knobs, battery connector, wire, etc.

Vibrato Unit
Vibrato and tremolo are terms which seem to cause a certain amount of confusion, but it is now generally accepted that tremolo is amplitude modulation and that vibrato is frequency modulation. This is the convention that will be used in this book. Vibrato is something that can be achieved with many electric and electronic instruments without the need for any external assistance, but it is not possible with all instruments. Also, with the aid of a vibrato unit it is possible to frequency modulate a number of instruments (after mixing) if desired, and this can give interesting results. However, it is advisable to only use a moderate amount of vibrato when processing the output of two or more instruments, as decidedly discordant results can be produced if the modulation is excessive.

Vibrato is somewhat more difficult to generate than the effects described so far in this book, and it requires the use of a delay line. The effect is obtained by varying the delay time of the delay line. This is perhaps easier to understand if we consider the basic operation of a delay circuit. There are several types, but in music applications which require a fairly short delay the "bucket brigade" type is the only one in common use. These are more correctly called charge coupled devices (CCDs), although bucket brigade is perhaps a more apt name for them, as we shall see shortly.

Charge coupled delay lines consist essentially of little more than a series of capacitors plus some electronic switches and buffer amplifiers. Figure 15 shows the basic arrangement used in a CCD delay line. Only four stages are shown here in order to illustrate the basic action, but in a practical delay line there would be a few hundred or even a few thousand stages.

Initially the electronic switches (S1 to S4) are in the positions shown in Figure 15, so that C1 charges to the input voltage. The switches are then set to the opposite states so that the charge voltage on C1 is passed via S1, buffer amplifier A1, and S2, to C2. The electronic switches are then set back to their original states so that C1 again charges to the same potential as the input signal.

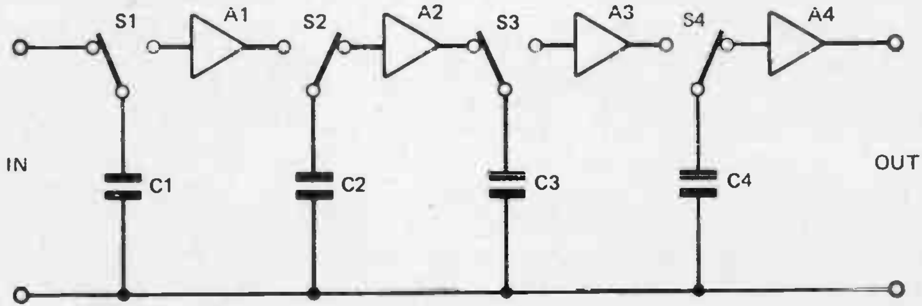


Fig. 15 The basic arrangement used in a CCD delay line

The charge on C2 is then passed through S2, A2, and S3 on to C3. The switches are then once again set to the opposite states, C1 passes on its charge to C2, and C3 passes on its charge to C4 where it provides the output signal via buffer amplifier A4.

The action of the delay line is therefore to repeatedly sample the input signal voltage, and to pass the sampled voltages down a series of capacitors until they emerge, some time later, at the output of the circuit. This is analagous to buckets of water being passed along a human chain, and it is from this that the term "bucket brigade" is derived. The delay provided by the circuit depends on two factors; the number of stages in the line and the frequency at which the switches are alternated from one state to another. In a practical delay line the switching frequency is controlled by a clock oscillator. The delay time is equal to the number of stages divided by double the clock frequency.

In the basic form of Figure 15 there are a couple of flaws in the system. One is simply that the input to A4 is left floating while C4 is receiving a sample from C3. In most delay line integrated circuits this problem is solved by having an additional stage at the end of the delay line which maintains the output while the final delaying stage is receiving a sample. This extra stage merely prevents any breaks in the output signal, and does not contribute to the delay time. The second flaw is that the output signal is not a continuous waveform, but a stepped type which jumps from one voltage to the next as the output stage switches from one sample to the next. This is not important provided the clock frequency is at least double the maximum input frequency, and preferably three or more times the maximum input frequency. The steps in the output waveform can then be eliminated using lowpass filtering, leaving an ordinary audio output signal.

Returning to the vibrato effect, if samples are taken and fed into the delaying stages at a certain rate, and then the clock oscillator is switched to a higher frequency, the samples will obviously be fed to the output at a higher rate than they were taken at the input, giving an increase in output frequency. Similarly, switching the clock oscillator to a lower frequency results in samples being fed from the output at a lower rate than they were taken at the input, giving a reduction in frequency. Of course, the boost or decrease in frequency can only be maintained for a short time (as long as it takes for all the samples in the delaying stages to be clocked through to the output after the clock frequency has been changed). This is of no great importance in this application though, since we require the processed signal to be continuously varied up and down in frequency. However, it does mean that only a very limited amount of frequency shift can be obtained if a very low modulation frequency is used, unless the delay line is very long. A very long delay line is often impractical as it would introduce a noticeable delay between playing an instrument and obtaining any sound, and it could also be very expensive. The delay time therefore has to be something of a compromise.

Circuit Operation

The circuit diagram for the signal processing stages of the Vibrato Unit is shown in Figure 16. Figure 17 shows the clock and modulation oscillator circuits. Starting with the signal processing stages, IC4a acts as a buffer stage which gives the unit an input impedance of 100k and provides the low output impedance needed to drive the next stage of the unit reliably. This next stage is an active lowpass filter based on IC4b, and it is a third order (18dB per octave) type having a cutoff frequency of about 7 to 8kHz. This filtering is needed to prevent signals at frequencies close to the clock frequency from breaking through to the delay line. This would give a form of distortion known as "aliasing" distortion. A 7 to 8kHz bandwidth is obviously somewhat less than the full audio range, but it is wide enough to give an output signal of very reasonable quality.

The delay line chip is a TDA1022 (IC5) and this is a 512 stage device. There are actually 513 stages, but as explained previously, the last one does not provide any delay and is merely used to maintain the output signal during the periods when the penultimate stage is receiving a sample. RV5 acts as a passive mixer which combines the output of stages 512 and 513, and it is adjusted to minimise the clock glitches on the output. R14 and R15 provide a bias voltage to an input of IC5. The output of IC4b is direct coupled to the input of IC5, and RV4 is adjusted to give optimum large signal handling capability from the circuit. R16 acts as a load resistor for the output circuits of IC5.

Pins 1 and 4 of IC5 are both clock inputs, and a two phase clock signal is required. In other words the two clock inputs are fed with antiphase signals. If you refer back to Figure 15 you will see that the odd numbered switches are always in the opposite state to the even numbered switches. It is for this reason that a two phase clock signal is required, with each phase operating alternate

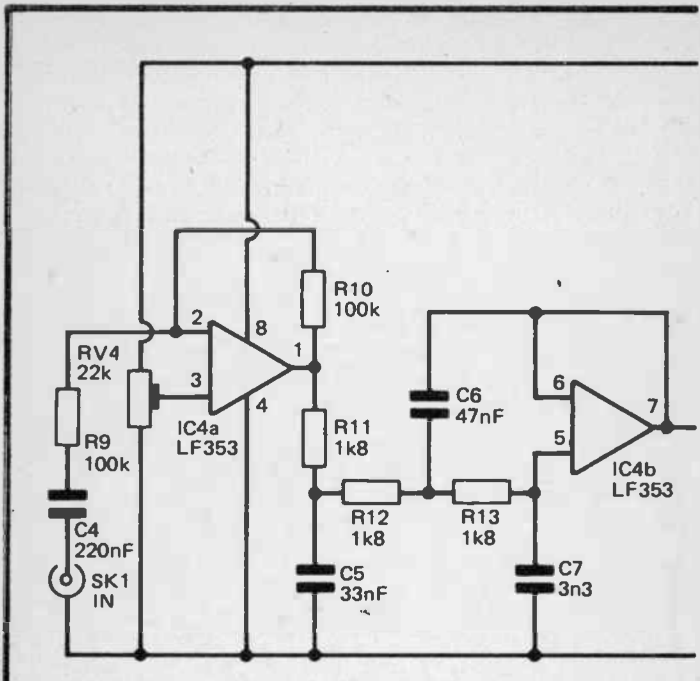
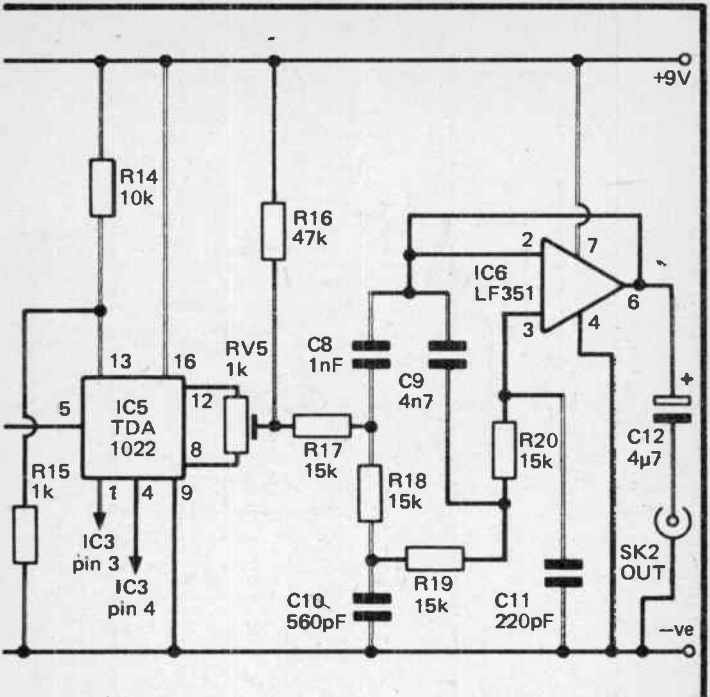


Fig. 16 The circuit diagram of the Vibrato unit signal

switches. It would presumably be feasable to include an on-chip inverter to generate the second clock phase, or even to have an on-chip clock oscillator, but for some reason this does not seem to be a feature of the current CCD delay line chips.

IC6 is used in the output filter which removes the steps on the output signal. This is a fourth order (24dB per octave) filter, again having a cutoff frequency of about 7 to 8kHz. IC6 also acts as a buffer stage which gives the unit a low output impedance.

The clock frequency is in the region of 20 to 30kHz, giving a delay times of around 8 to 13ms. This is not very long, but is sufficient to give a more than reasonable vibrato effect. The clock oscillator is a standard 555 astable circuit based on IC2. RV3 enables the centre frequency of the oscillator to be trimmed to a suitable figure. IC3a and IC3b are CMOS NAND gates, but here



processing stages

they are connected as simple inverters and are used to process the output of IC2 to generate a two phase clock signal that is compatible with the TDA1022.

The modulation oscillator is basically the same as the one used in previous projects, but in this case it has a frequency range of about 0.5Hz to 10Hz. There is little point in using very low modulation frequencies as these would not give a significant vibrato effect. The triangular output of the oscillator provides the modulation signal, and RV2 enables the vibrato depth to be adjusted. S2 can be used to disconnect the modulation oscillator from IC2 and switch out the vibrato effect. Although the clock oscillator is not stricly speaking a voltage controlled type, the output frequency can actually be modulated by applying a suitable signal to pin 5 of IC2, as in this case.

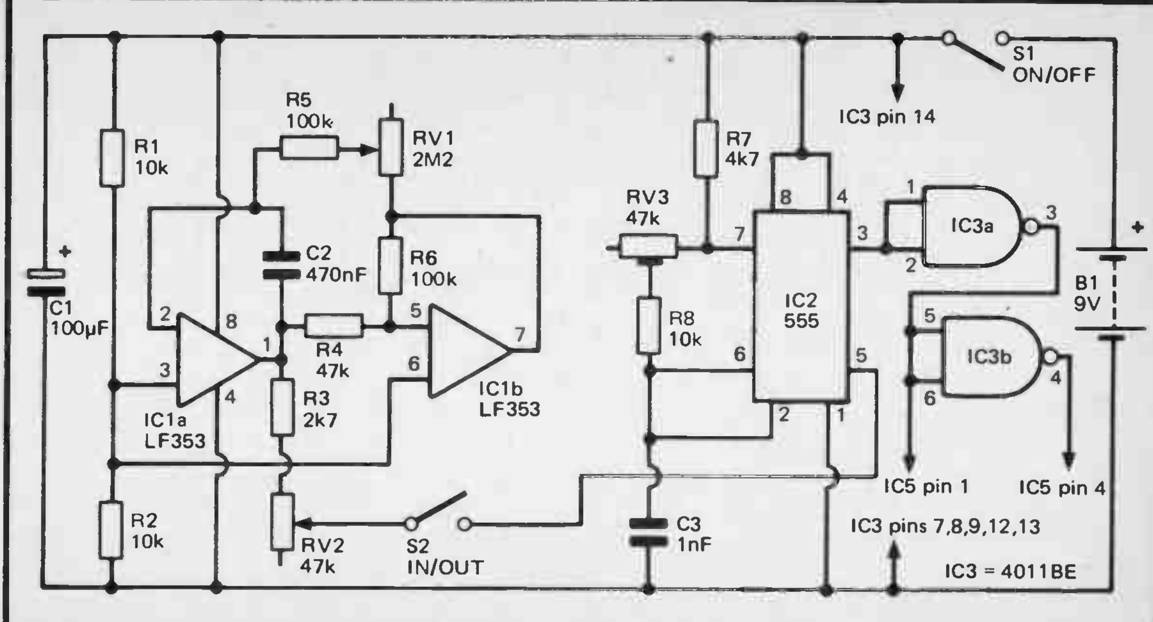


Fig. 17 The circuit diagram of the Vibrato unit clock oscillator

Components for Vibrato Unit (Figs. 16 & 17)

Resistors (All ¼ watt 5%)

R1,2,8,14	10k
R3	2k7
R4,16	47k
R5,6,9,10	100k
R7	4k7
R11,12,13	1k8
R15	1k
R17,18,19,20	15k

Potentiometers

RV1	2M2 linear
RV2	47k linear
RV3	47k sub-min preset
RV4	22k sub-min preset
RV5	1k sub-min preset

Capacitors

C1	100μF 10V elect
C2	470nF carbonate
C3,8	1nF carbonate
C4	220nF carbonate
C5	33nF carbonate
C6	47nF carbonate
C7	3n3 carbonate
C9	4n7 carbonate
C10	560pF polystyrene
C11	220pF polystyrene
C12	4μ7 63V elect

Semiconductors

IC1,4	LF353
IC2	555
IC3	4011BE
IC5	TDA 1022
IC6	LF351

Miscellaneous

SK1, SK2	Standard jack socket
S1	SPST sub-min toggle

S2	SPST heavy duty push button
B1	9 volt (PP7 size)

Metal case
14 pin DIL and 16 pin DIL IC holders
Circuit board, control knobs, battery connectors, wire, etc.

In Use

When constructing the unit remember that IC3 and IC5 are both MOS devices, and that the normal antistatic handling precautions should therefore be taken when dealing with these. IC3 requires a 14 pin DIL integrated circuit holder while IC5 requires a 16 pin type.

There are three preset resistors to be adjusted once the unit has been completed. Start with RV3 at nearly minimum resistance, and both RV4 and RV5 at a roughly mid-point setting. The unit should then work at least to some degree, and if an audio sinewave generator and oscilloscope are available the first task is to adjust RV4 for symetrical clipping. If suitable test gear is not to hand then it is just a matter of feeding an audio input to the unit and then adjusting RV4 to find a setting that gives an output signal which is free from any obvious distortion. It is likely that there will be a small range of suitable settings, and RV4 is then set at the centre of this range.

If an oscilloscope or audio millivoltmeter is available, this should be used to monitor the signal at the wiper of RV5 so that the latter can be set for minimum clock breakthrough. There should be no input signal connected when making this adjustment. If neither a millivoltmeter nor an oscilloscope are to hand, switch S2 to the "OUT" position and set RV3 at maximum resistance. This will bring the clock oscillator to a frequency that is just within the audio range, and the clock breakthrough at the output of the unit will be audible. RV5 is adjusted to minimise the audible clock breakthrough at the output.

In order to permit the unit to achieve the strongest possible vibrato effect RV3 should be adjusted for the longest achievable delay. The maximum delay that can be used in practice is limited by the fact that making the clock oscillator too low results in audible breakthrough at the output. RV3 is therefore adjusted for the highest resistance that does not give any audible breakthrough of the clock signal, even with RV2 set for maximum modulation depth (set at minimum resistance).

Although the unit does not incorporate any form of noise reduction unit, the TDA1022 provides a good signal to noise ratio without any assistance. However, in order to make the most of the device's performance the input signal should be in the range 200 millivolts RMS to about 1 volt RMS or so. The TDA1022 has a minimum recommended supply voltage of 10 volts, but trying several of the devices on a 9 volt battery supply always gave perfectly acceptable results, and it should not be necessary to resort to a 12 volt supply.

Mini Chorus

Delay lines are used in several other types of musical effects unit, and two of the most popular are various types of chorus effect, and flangers. Both will be covered here, and both are based on the vibrato circuit described previously. In fact both are really just minor variations on the vibrato circuit. We will start with a simple chorus unit.

Complex chorus units use several delay lines, with each one having its delay time varied by a low frequency oscillator. The outputs of the delay lines are mixed with the non-delayed signal, and the effect is to make a single signal source sound like an ensemble, or a single voice input sound like a choir (hence the "chorus" name). In fact most chorus units are somewhat less complex than this, and use just a single delay line. These are often called "mini chorus" units, or if the delay time is not modulated the terms "micro chorus" and "ADT" (audio dual tracker) are often adopted. These give a somewhat less rich effect than a full chorus unit, and they turn a solo voice into a duet rather than a choir, but they are nevertheless very useful effects which can greatly enhance many signals. They are probably most used with vocalists, but can be very effective with guitars and synthesisers. They have the advantage over a full chorus unit of only costing a fraction as much to build.

In order to operate properly, the delay time provided by the delay line must be more than 10ms. A shorter time would result in the unit providing a sort of phasing effect, with the delayed signal or signals not being perceived by the listener as seperate sound sources. The delay time should not be more than about 50ms or the effect obtained would be an echo type rather than a chorus effect. The maximum delay of the vibrato unit, at around 13ms, is therefore in the required range.

Flanging is produced using what is really just a slight

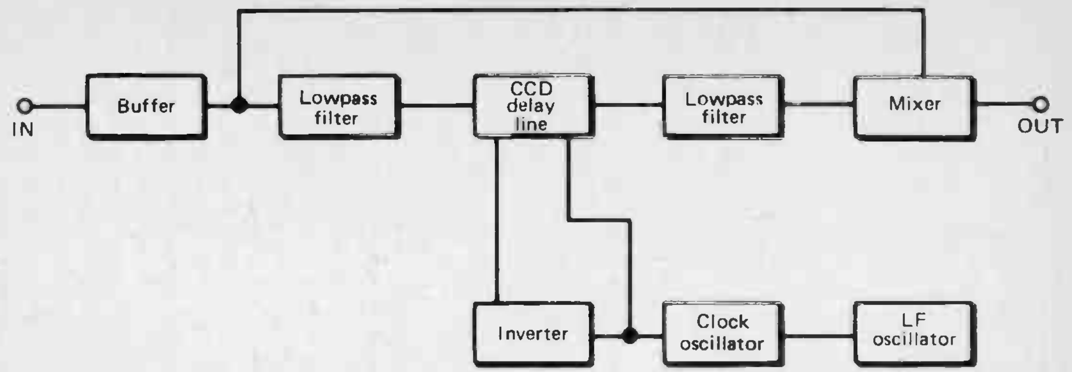


Fig. 18 Block diagram for the Mini Chorus unit

rearrangement of the chorus set-up, but this minor modification to the chorus set-up produces what is a radically different effect. The block diagrams of Figure 18 and Figure 19 show the difference in the way the two effects are generated. The mini chorus effect is generated in the manner described above, with the delayed and non-delayed signals being mixed at the output of the circuit. The flanging arrangement differs from this only in that the mixer is moved to the input of the delay line, but this greatly modifies the action of the circuit, principly due to the feedback that this introduces.

The delay time used for flanging is somewhat less than that required for a chorus effect, with around 4 to 5ms being quite ample. The delay of the vibrato unit can be trimmed down to a suitable level. The feedback from the output to the input of the circuit has two main effects, and one of these is to introduce peaks and troughs in the frequency response. These are swept up and down in frequency by the low frequency oscillator, giving a sort of phasing effect, but with the feedback control set near maximum the irregularities in the frequency response are much more pronounced than with ordinary phasing. The second effect is to circulate signals around the feedback loop, giving a sort of simple and quite short reverberation effect.

With the feedback control set at maximum this is a very strong effect indeed, but it can be tamed considerably by backing off the feedback control slightly. There is a limit to the degree of feedback that can be used, and over-stepping the limit results in the circuit breaking into oscillation.

Chorus Modification

In order to make the vibrato circuit function as a chorus unit it is merely necessary to add a mixer stage at the output, plus two or three other components, as shown in the circuit diagram of Figure 20. Here C12 and SK2 are moved from the original circuit to the new positions, but the other components are all additional to those in the original circuit.

IC7 operates as a straightforward suming mode mixer circuit which combines the non-delayed output from the input buffer stage with the delayed signal from the output of IC4. S3 can be used to cut off the delayed signal and switch out the chorus effect. A potentiometer could be added in series with S3 to enable the strength of delayed signal to be adjusted, but this effect only works well with the two signals at similar levels, and it is probably

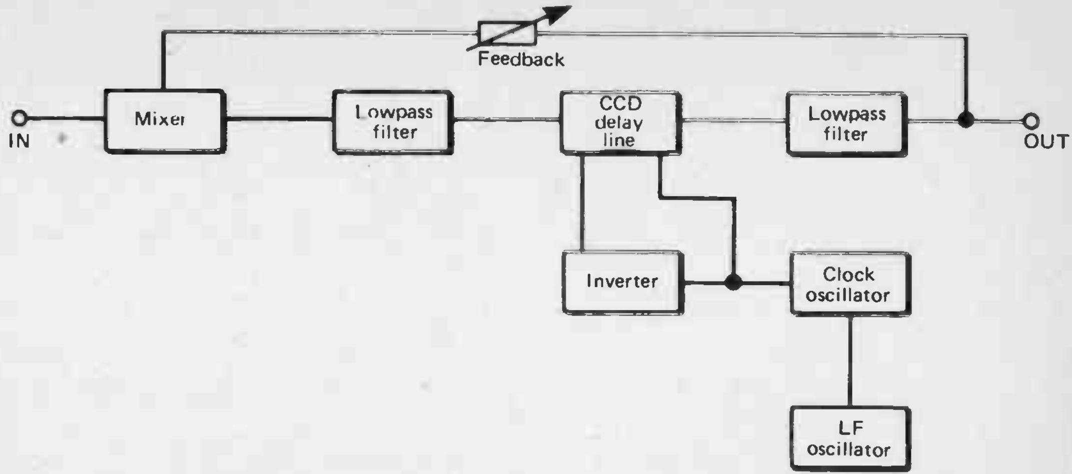


Fig. 19 Block diagram for the Flanger

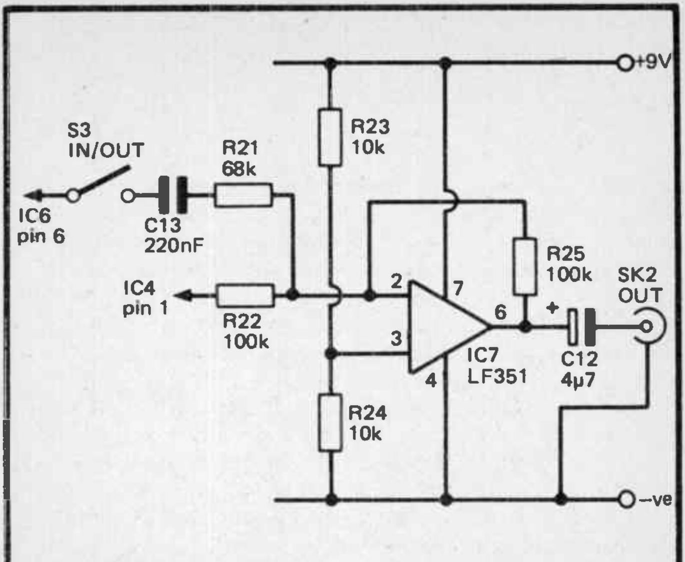


Fig. 20 Modifying the Vibrato unit to act as a Mini-chorus

not worthwhile making this modification. There is a small loss through the delay line circuit, and R21 has been made somewhat lower in value than R22 in order to compensate for this.

The preset resistors are set up in exactly the same manner as for the vibrato unit. S2 is closed in order to give the mini chorus effect, or opened if the micro chorus effect is required.

Components for Mini Chorus Unit (Fig. 20)

All components as for the vibrato unit plus the following additions:-

R21	68k
R22,25	100k
R23,24	10k
C13	220nF carbonate
IC7	LF351
S3	SPST heavy duty push button

Flanger Modification

The flanger modification is even more simple, since the input buffer stage of the vibrato unit can act as the input mixer stage. The necessary modification to the circuit is shown in Figure 21.

There are just four additional components – S2b (which is, of course, an additional pole on the IN/OUT switch, S2), C13, RV6, and RV7. IC4a originally operated as an inverting amplifier, but by adding an extra input resistance it functions as a suming mode mixer circuit. RV6 is set to limit the maximum amount of feedback to a level that does not quite produce oscillation. RV7 is the flanging "depth" control, and C13 is merely a DC blocking capacitor. S2b cuts off the feedback completely when the effect is switched out, and S2 in the original circuit is still required in order to switch off the frequency modulation.

The setting up procedure is much the same as for the vibrato unit, but RV3 in the clock oscillator should be adjusted for a somewhat lower resistance in order to shorten the delay time slightly. To give RV6 the correct setting first set RV7 at minimum resistance. Then adjust RV6 for the lowest resistance setting that does not result in the circuit breaking into oscillation.

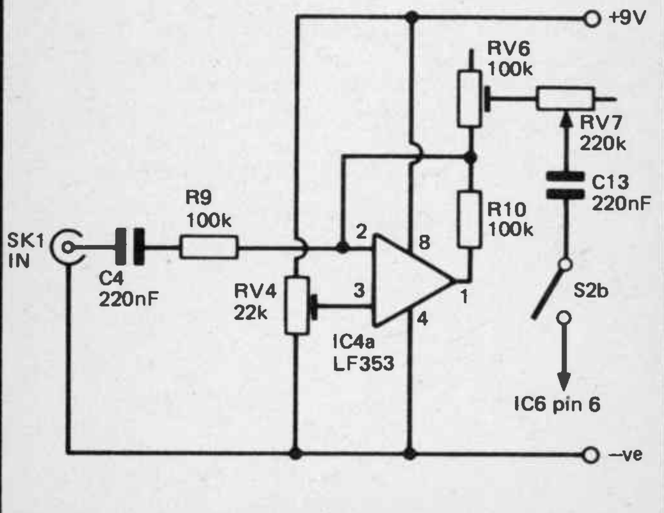


Fig. 21 Modifying the Vibrato unit to provide flanging

Components for Flanger (Fig. 21)

All components as for the vibrato except that S2 becomes a DPST heavy duty push button type, and the following additional components are required:-

RV6	100k sub-min preset
RV7	220k linear
C13	220nF carbonate

Chapter 2

PERCUSSION SYNTHESISERS

Traditional drum kits have been losing ground to electronic equivalents over recent years, and there are definite advantages to electronic percussion synthesisers. Probably the most attractive one for most users is the relatively low cost of electronic drum synthesisers, particularly if they are home constructed. The small size and portability are also important factors for many users.

To synthesise realistic percussion sounds is actually quite difficult, and requires complex digital circuits. Low cost percussion synthesisers based on analogue circuits give what could therefore be considered a rather poor output quality, but this is only if you require conventional drum and cymbal sounds. Even quite simple percussion synthesisers can usually produce a useful range of effects, including sounds that are not possible with conventional instruments. These sounds are just as valid as those produced by conventional percussion instruments, and provide the creative user with almost limitless possibilities. It is therefore advisable to experiment with the units described in this chapter to find the types of sound that can be produced, and then to use these sounds in a creative way, rather than to simply try for the best possible simulations of conventional instruments. Otherwise you are missing out on a lot of the fun and overlooking many of the possibilities of electronic percussion synthesisers.

Envelope Shaper

A percussion synthesiser consists of two main sections; some form of signal source, and an envelope shaper to vary the volume of the sound to give a good percussive effect. An ADSR envelope shaper could be used, but percussive sounds mostly have a fairly simple shape that can be produced satisfactorily using a simple attack/decay envelope shaper. Initially the signal jumps almost instantly to a high volume level, then it falls back at a slower rate (although still quite rapidly), with the rate of decay gradually slowing down as time passes. This is precisely the envelope shape given by an attack/decay envelope shaper set for a fast attack and a medium to slow decay time.

Figure 22 shows the block diagram for a falling pitch disco drum. All the stages apart from the VCO form the envelope shaper.

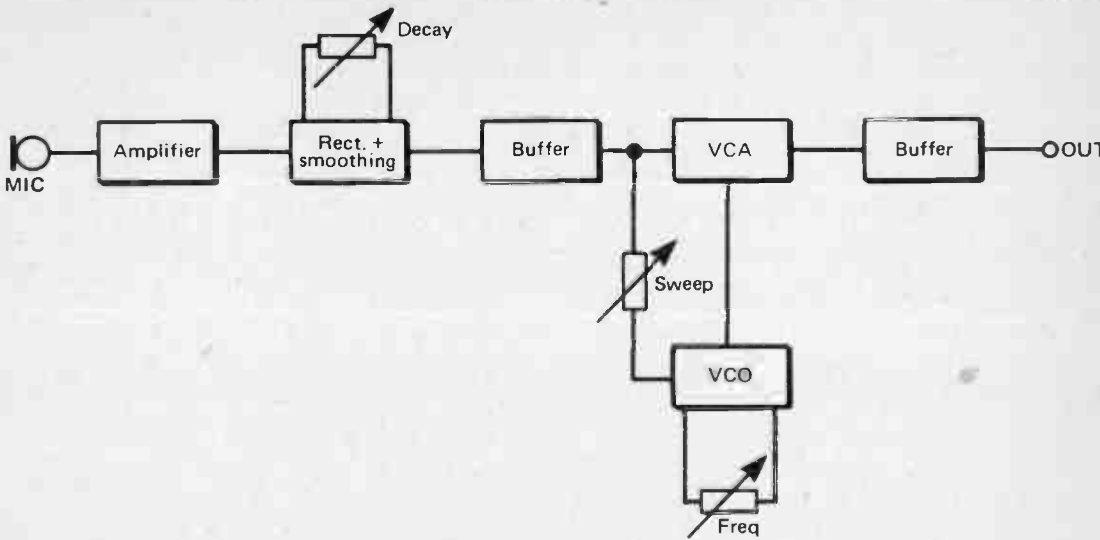


Fig. 22 Block diagram for the Disco Drum

The unit is triggered by tapping the case which vibrates a microphone (or if preferred the microphone can be mounted in a practise pad). The output from the microphone will be much stronger than the few millivolts that would be obtained in normal use, but a certain amount of amplification is still needed in order to give a strong enough signal to operate the subsequent circuits reliably. The amplified signal is fed to a rectifier and smoothing circuit which produces a DC output voltage from the short burst of input pulses. The attack time of this circuit is suitably fast, and the the decay time is much longer and adjustable. The adjustment range is from around 100ms to approximately five seconds, allowing anything from a short "dull" sound to a long and very resonant effect to be produced.

The output from the smoothing circuit is at a high impedance, and a buffer amplifier is therefore used to enable it to drive the relatively low input impedance at the control input of the VCA. The latter is fed from the output of a VCO, and it in turn feeds the output socket by way of a buffer stage which gives the unit a low output impedance. The signal from the VCO is normally blocked from the output, but when the microphone is activated a suitably envelope shaped burst of signal is allowed to pass through to the output. A frequency control enables the pitch of the VCO to be varied, but further than this, the output of the control voltage generator can be coupled through to the control input of the VCO. As the control voltage subsides, the pitch of the VCO is swept downwards, giving the required disco drum falling pitch sound. A "sweep" control enables the sweep width to be controlled.

Although the arrangement of Figure 22 is for a disco drum, by using a different signal source it is possible to obtain other effects. Some of these require the output from the control voltage generator, some do not, but in each case the envelope shaper is the same. We will therefore consider an envelope shaper circuit for percussive sound generation first, and then a variety of signal sources for use with the envelope shaper will be described. These enable an extremely wide range of percussive sounds to be generated.

The circuit diagram of the Envelope Shaper appears in Figure 23.

IC1 is one section of an LM13600N and this is used in a VCA circuit of the type featured in some projects which were discussed earlier in this book. IC2b is the input amplifier, and Mic1 is a

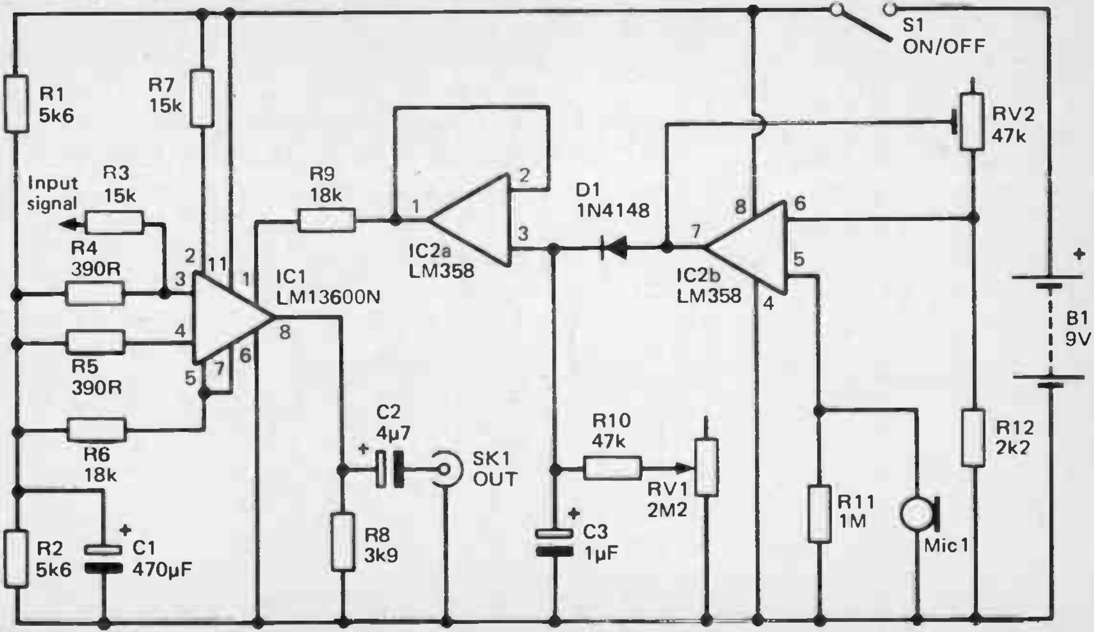


Fig. 23 The circuit diagram of the Envelope Shaper

crystal microphone insert or a ceramic resonator. Both give what is, by microphone standards anyway, a fairly high output level, but from a fairly high source impedance. IC2b is therefore needed as much to provide buffering as it is to give voltage amplification. R11 gives the amplifier a high input impedance of about 1 megohm. RV2 enables its closed loop voltage gain to be varied from unity at minimum resistance to about 22 times at maximum resistance. In practice RV2 is set so that the gain of the input amplifier is only just sufficient to drive the rectifier and smoothing circuit properly. The point of this is to give the unit a degree of touch sensitivity. In other words, if the unit is tapped hard then an output at full volume is obtained, but if it is tapped more gently then an output at something less than full volume is obtained. This gives something approaching the degree of control available when playing a conventional percussion instrument, although precise control at low output levels is admittedly somewhat more tricky with a simple electronic device such as this one.

The output of IC2b is biased to the negative supply rail under quiescent conditions, and the LM358 specified for IC2 is a type having output stages that permit the outputs of both sections to provide output potentials right down to the negative supply voltage. This is an essential feature for correct operation in this circuit, and few other dual operational amplifiers will function properly in the unit (the more expensive CA3240E being one probable exception).

In the presence of a signal from the microphone a burst of positive output pulses are produced from IC2b, and these rapidly charge C3 from the low output impedance of IC2b. D1 provides a low impedance path and does not significantly boost the charge time. This gives the circuit a suitable attack time of just a few milliseconds. The only discharge paths for C3 are into the extremely high input impedance of IC2a and through the series resistance of R10 and RV1. The discharge (decay) time is consequently much longer, and by means of RV1 it can be varied over the range specified previously. C3 does not discharge in a linear fashion, but with the usual exponential characteristic. The voltage drops at a comparatively fast rate initially when the voltage across R10 and RV1 is high, giving a high current flow. As the charge voltage falls, so does the discharge current and the rate of discharge. This gives the required decay characteristic, and is in fact analagous to the sound from a mechanical resonator decaying.

IC2a simply acts as a unity voltage gain buffer stage which matches the high output impedance of the smoothing circuit to the relatively low input impedance at the control input of the VCA.

Power for the circuit is provided by a 9 volt battery, and a PP7 size is specified for all the projects in this book. A smaller type such as a PP3 could be used if small size is important, but battery life would be rather limited and this would be a relatively expensive way of powering the circuits. On the other hand, if space is no problem, a larger type such as a PP9 could be used, and would probably give slightly lower running costs. If the units are likely to receive a great deal of use, in the long term it would almost certainly be worthwhile using rechargeable NiCad batteries as the power source.

Components for Envelope Shaper (Fig. 23)

Resistors (All ¼ watt 5%)	
R1,2	5k6
R3,7	15k
R4,5	390R
R6,9	18k
R8	3k9
R10	47k
R11	1M
R12	2k2
Potentiometers	
RV1	2M2 linear
RV2	47k sub-min preset
Capacitors	
C1	470μF 10V elect
C2	4μ7 63V elect
C3	1μF 63V elect
Semiconductors	
IC1	LM13600N
IC2	LM358
D1	1N4148
Miscellaneous	
SK1	Standard jack socket

S1	SPST sub-min toggle
Mic1	Ceramic resonator (PB2720)
B1	9 volt (PP7 size)

Case
Circuit board, control knob, battery connectors, wire, etc.

Construction

The only unusual aspect of construction is the mounting of the microphone. On the prototype I used a ceramic resonator type PB2720 which, although intended as a sort of miniature loudspeaker in alarm circuits, operates quite well in reverse as a sort of crude microphone. This component has provision for fixing using two small (about 8BA) nuts and bolts, and it should be bolted to the inside of the case at any convenient place.

There is also an uncased version of this component (the PBN2720) which is perfectly suitable, but this has no provision for screw fixing and must be glued in position using a good quality general purpose adhesive. The all-gold coloured side is fixed to the case, and as the component does not come with two leads ready-attached it is necessary for the constructor to solder these in place prior to mounting the component. One lead is connected to the outer (gold coloured) ring and the other is soldered to the inner (silver coloured) disc. In both cases do not leave the iron in place any longer than is absolutely necessary so that the device is not damaged by over-heating. Also, it is advisable not to use a high temperature soldering iron,

A crystal microphone insert is also suitable, and although these seem to provide a slightly lower output level than ceramic resonators, they are perfectly adequate in this respect. Most have no provision for panel mounting and must be fixed to the interior of the case using a suitable adhesive.

It is a good idea to protect the area of the case that will be struck when playing the unit, and some form of thin foam material is ideal for this, but any fairly tough material should suffice. A plastic case is preferable to a metal type in an application such as this, and a type made from a fairly soft but strong plastic is ideal. This ensures a reasonably low level of mechanical noise when the instrument is played. It should not be necessary to have the microphone mounted under the area of the case that is struck, and reliable operation should be obtained with the microphone mounted anywhere inside the case. In order to minimise the risk of damage to the microphone it is probably best to mount it well away from the area of the case that will be struck.

The microphone can be mounted in a practise pad, or somewhere else away from the rest of the unit if this would be more convenient in use for some reason. However, it must be connected to the rest of the circuit via a screened lead, and this should be no more than about 2 metres long.

RV2 is adjusted for the lowest sensitivity (the lowest value) that gives full volume when the pick-up is struck hard.

VCO

The envelope shaper is, of course, no value on its own, and some form of signal source is required in order to make it function as a percussion synthesiser. As the first of several signal sources we will consider the VCO circuit of Figure 24 which enables the envelope shaper to function as a falling pitch disco drum.

There are obvious similarities between this circuit and the twin operational amplifier oscillator circuits featured in previous projects. In fact this circuit is a variation on the standard Miller Integrator/Schmitt Trigger designs described earlier. IC3a acts as the Miller Integrator and IC3b operates as the Schmitt Trigger, although in this case it is an inverting type. It drives common emitter switching device Tr1 which reinverts the signal to give a non-inverting action from the input to the output that drives the charge/discharge current control resistor (R18). However, the charge/discharge rate is not controlled by R18 alone, and R15 plus the control voltage (which is applied to the junction of R15 and R16) also have to be taken into account. This gives voltage controlled operation, with a wide control range and good linearity. A proportion of the control voltage is applied to the non-inverting input of IC3a (rather than a fixed bias) in order to prevent control voltage variations from distorting the output waveforms.

Like an ordinary Miller Integrator/Schmitt Trigger oscillator, this one provides triangle and squarewave output signals. In this application a squarewave signal does not give very good results, and the much purer sound provided by a triangle signal with its low harmonic content is much to be preferred. If you would like to try using the squarewave output though, this is available at pin 7 of IC3.

R13 and R14 form a simple passive mixer at the control input of the oscillator. One input of the mixer is fed with the output

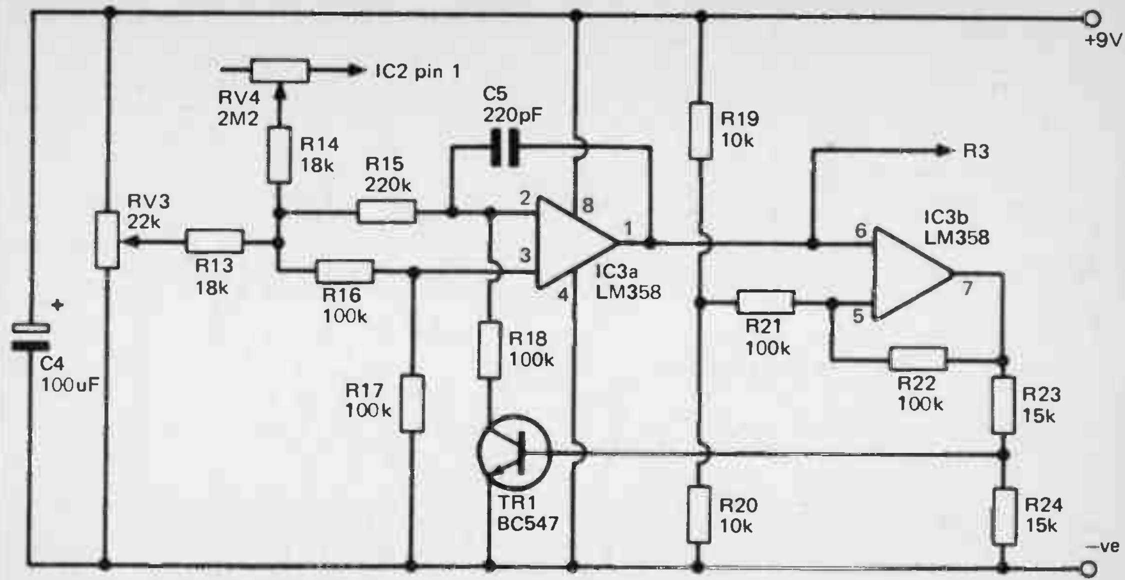


Fig. 24 The circuit diagram of the VCO

voltage of "pitch" control RV3, while the other input is fed from the output of the control voltage generator circuit by way of "sweep" control RV4. There is a certain amount of interaction between the pitch and sweep controls, but by alternating from one control to the the other it does not take long to set up the desired basic pitch and sweep range.

The frequency range of the oscillator is quite wide, and covers from the sub-audio to something approaching the ultrasonic range.

Note that this is another circuit which relies on the output stages of the LM358 to provide large voltage swings, and that few alternative dual operational amplifiers will function properly in this circuit.

Components for Disco Drum (Fig. 24)

Resistors (All ¼ watt 5%)	
R13,14	18k
R15	220k
R16,17,18,21,22	100k
R19,20	10k
R23,24	15k
Potentiometers	
RV3	22k linear
RV4	2M2 linear
Capacitors	
C4	100μF 10V elect
C5	220pF ceramic plate
Semiconductors	
IC3	LM358
Tr1	BC547
Miscellaneous	
Control knobs	

A full set of Envelope Shaper components is also required

Rising Pitch

If you would prefer a rising pitch effect to a falling pitch one, or would like the option of being able to switch from one to the other, the circuit modification shown in Figure 25 can be adopted. This just consists of an operational amplifier connected as a unity voltage gain inverting amplifier, and used to invert the output from the control voltage generator before it is applied to the VCO circuit. S2 enables the inverter to be bypassed and the unit to be switched back to the falling pitch effect when desired.

Note that the CA3140E specified for IC4 has a PMOS input stage and that it consequently requires the antistatic handling precautions described previously. Note also that most other operational amplifiers have an output stage which gives an inadequate output voltage swing for this application, and that the use of other types is not recommended.

RV4 is initially set at a roughly mid-way setting, and it can be trimmed by trial and error to a more suitable setting if this original one does not enable a good sweep range to be obtained. In particular, it might need slight trimming if low sweep frequencies can not be achieved with it at the original setting.

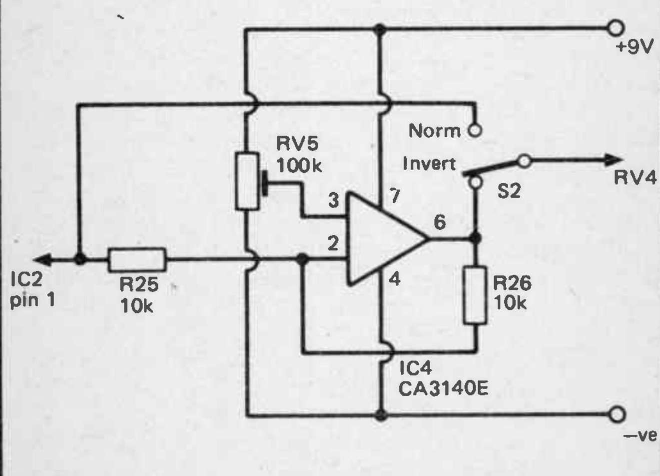


Fig. 25 A circuit to give switchable rising/falling pitch effects

Components for Falling/Rising Pitch Converter (Fig. 25)

Resistors (All ¼ watt 5%)	
R25,26	10k
Potentiometer	
RV5	100k sub-min preset
Semiconductor	
IC4	CA3140E
Miscellaneous	
S2	SPDT sub-min toggle

8 pin DIL IC holder

Fixed Pitch

If a fixed pitch drum effect is required, one way of achieving this would be to used the VCO but to omit R14 and RV4 so that no sweeping is obtained. Things can be simplified still further though, by using a standard Miller Integrator/Schmitt Trigger oscillator circuit, such as the one given in Figure 26.

This is much the same as the oscillators featured in several projects in Chapter 1, but in this case the pitch control (RV3) gives an approximate frequency coverage of 30 to 700Hz. The frequency range has been purposely concentrated at the low frequency end of the audio band as it is these frequencies that give good drum sounds. Higher pitches do not give normal drum sounds at all, but instead give an effect more like a triangle. If you want a unit to generate this type of effect it is merely necessary to reduce the value of C5 in order to boost the output frequencies. For example, a value of 22nF would raise the output frequencies by a factor of ten, giving an approximate coverage of 300Hz to 7kHz.

Components for Fixed Pitch Syndrum (Fig. 26)

Resistors (All ¼ watt 5%)	
R13,14	10k
R15	4k7
R16	33k
R17	100k

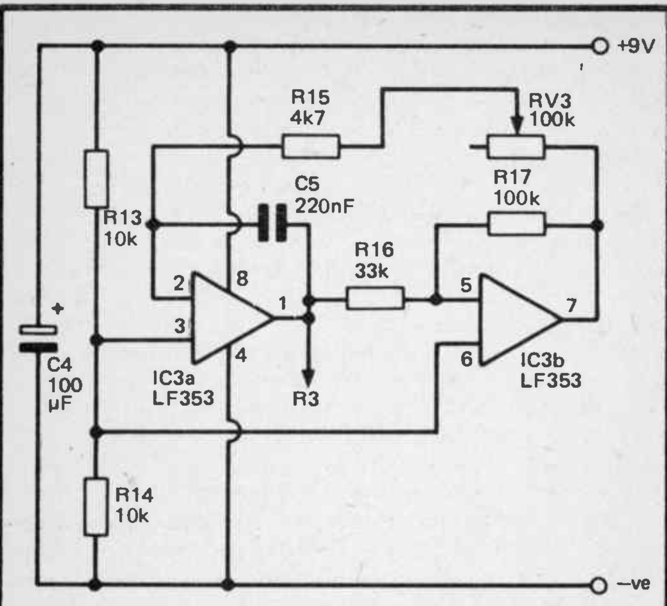


Fig. 26 The circuit diagram of the Fixed Pitch Oscillator

Potentiometer	
RV3	100k linear
Capacitors	
C4	100μF 10V elect
C5	220nF carbonate
Semiconductor	
IC3	LF353
Miscellaneous	
Control knob	

A full set of Envelope Shaper components is also required

Noise Based Sounds

A lot of conventional percussive sounds are noise based, with cymbals, handclaps, and hi-hat being three good examples. Using a noise generator as the signal source for the envelope shaper enables a range of useful simulations of conventional instruments to be obtained, as well as some very interesting effects which have no parallel in conventional percussive sounds.

A simple noise generator circuit is provided in Figure 27. Tr1 acts as the noise source, and here we are only using the base and emitter terminals with the collector being left unconnected. R28 provides a reverse bias to the base-emitter junction, causing it to breakdown and avalanche, much like a zener diode. Also like a zener diode, noise spikes are generated, but by using a transistor rather than a zener diode a narrower bandwidth is obtained. There is a much higher output level though, which suits the current application where only noise at audio frequencies is of importance, and any radio frequency output is irrelevant.

Although the noise output of Tr1 is quite high by noise generator standards, it is still quite low in absolute terms, being typically just a few millivolts RMS. Tr2 is used as a common

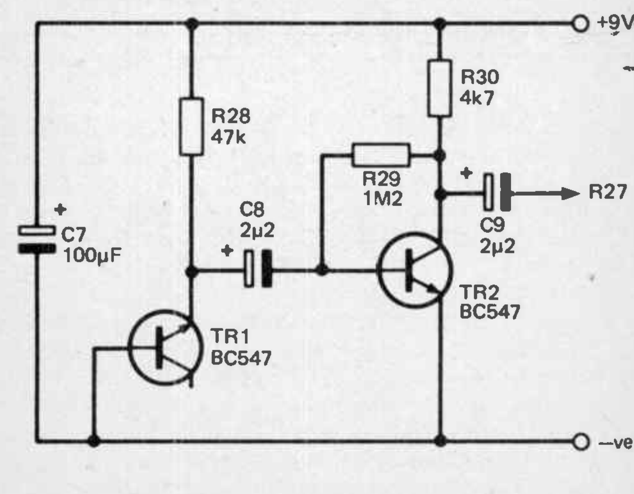


Fig. 27 The circuit diagram of the Noise Generator

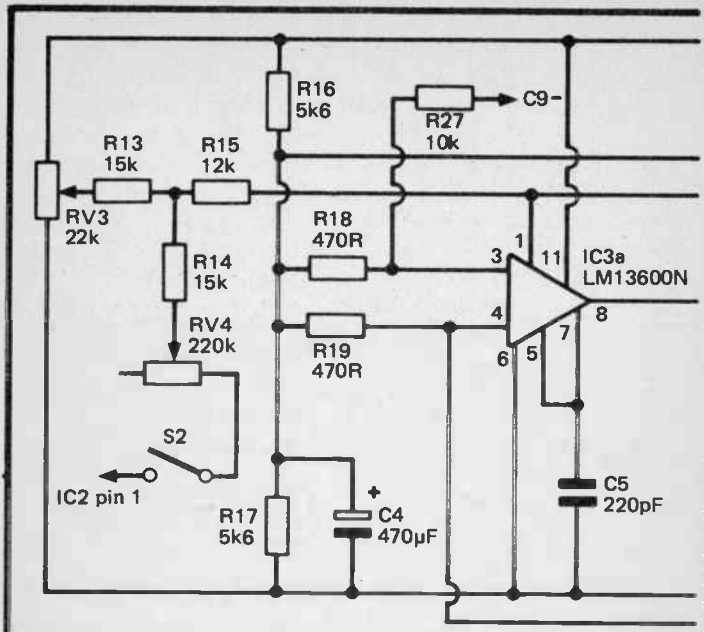
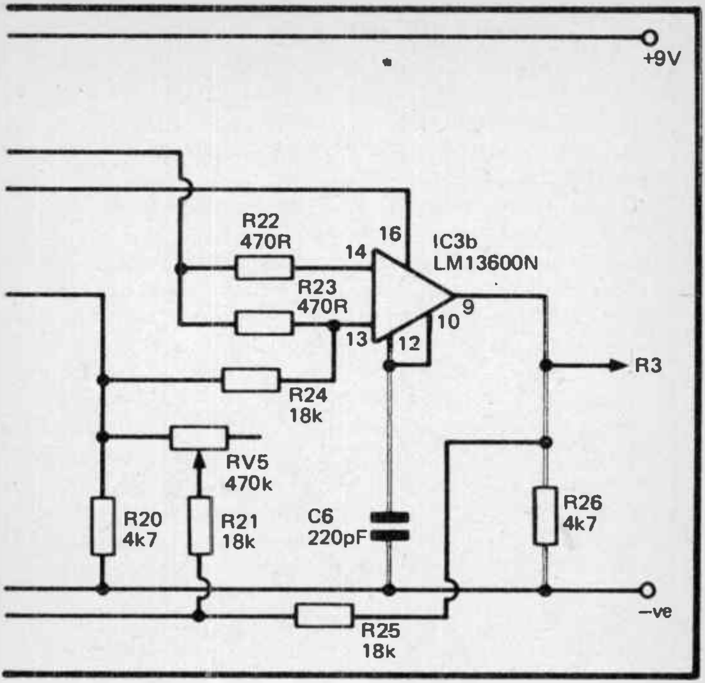


Fig. 28 The circuit diagram of the VCF

emitter amplifier which boosts the signal level by about 40dB or so, giving an output level of around 1 volt RMS, which is sufficient to drive the Envelope Shaper properly.

A BC547 is specified for Tr1, and when a few of these were tried in the prototype they all worked well. However, Tr1 is obviously being used in a non-standard way, and results are therefore slightly unpredictable. In practice it is a good idea to try a few silicon npn transistors from the spares box in the Tr1 position, selecting the first one to give good results. Virtually any silicon npn transistor will operate in the circuit, but some give more noise output than others. Also, some have fairly high breakdown voltages, and although they may work fine with a new battery fitted, the noise output can suddenly subside as the battery voltage falls below the critical level. It is advisable to select the device for Tr1 using a well used battery to power the unit.



VCF

It would be quite possible to connect the output from the noise generator direct to the input of the envelope shaper unit, but this would not give very good results with little control over the output sound being possible. In fact the only control available to provide variations in the output sound would be the duration control. Much more interesting effects can be obtained by processing the output of the noise generator using a voltage controlled filter (VCF), and this gives tremendous control over the output sound. A suitable VCF circuit is shown in Figure 28.

The filter is based on the two transconductance amplifiers in an LM13600N, and it is a 12dB per octave type with adjustable resonance. In fact a bandpass response is also available from pin 8 of IC3, and this can give slightly different effects to the lowpass output. You may prefer to use this output, or a switch to enable either output to be selected could be included in the circuit. The basic operation of the filter relies on the two transconductance amplifiers operating as voltage controlled resistors which, in conjunction with C5 and C6, act as simple (6dB per octave) lowpass filters. Feedback via R21, R25, and RV5 modifies the response at the output of IC3a to a bandpass type, and gives variable resonance by means of RV5 (maximum resistance corresponding to maximum resonance).

The cutoff frequency of the filter can be varied by means of RV3, and with S2 closed the output from the control voltage generator can be mixed with the fixed bias. RV4 then acts as the sweep width control. The sweep facility enables some interesting effects to be obtained, but a vast range of excellent effects can be generated without any sweeping of the VCF. As a few examples, cymbal type sounds can be generated using moderate resonance, a fairly high VCF frequency, and a medium to long decay time. Using higher resonance and a short decay time gives a handclap type sound. Maximum resonance, long decay time, and a fairly low VCF frequency gives a "Sonar" type sound. This effect can be enhanced by a low level of sweep. As with any effects synthesiser, it is important to experiment with various control settings in order to discover all the different types of sound that can be achieved.

Components for Noise Effects Synthesiser (Figs. 27 & 28)

Resistors (All ¼ watt 5%)	
R13,14	15k
R15	12k
R16,17	5k6
R18,19,22,23	470R
R20,26,30	4k7
R21,24,25	18k
R27	10k
R28	47k
R29	1M2
Potentiometers	
RV3	22k linear
RV4	220k linear
RV5	470k linear
Capacitors	
C4	470µF 10V elect

C5,6	220pF ceramic plate
C7	100μF 10V elect
C8,9	2μ2 63V elect
Semiconductors	
IC3	LM13600N
Tr1,2	BC547
Miscellaneous	
S2	SPST sub-min toggle
Control knobs	

A full set of Envelope Shaper components is also required

Noise Squarer

The noise output from the noise generator circuit of Figure 27 is a standard white noise "hissing" sound. There are other types of noise, and computer sound generators often use a form that is basically a pulse signal of random duration. This gives a slightly different sound, which is much more harsh than the gentle white noise "hissing" sound. This type of noise can be better for some types of effect, and it gives good "explosion" sounds for example. After filtering it can also give a somewhat improved cymbal effect.

All that is needed to convert the noise output from the circuit of Figure 27 to a pulse type is a simple comparator circuit. Figure 29 shows the circuit diagram of a suitable add-on circuit.

IC4 acts as a straightforward voltage comparator, and its output goes high if the non-inverting input is at a higher potential than the inverting input, or low if the inverting input is the one at the higher voltage. R31 and R32 provide a fixed bias to the non-inverting input, while RV6 provides a variable bias to the inverting input. If RV6 is adjusted so that the two input voltages are very nearly balanced, the noise signal (which is fed to the non-inverting input) will cause IC4's output to go high on positive half cycles, and low on negative half cycles. This gives an amplified and squared output signal, and it is the squaring process that gives the "roughness" to the sound.

The character of the sound can be modified somewhat by adjusting RV6 away from the balanced setting so that there is a significant difference between the two bias voltages. If we assume that the inverting input is taken to a higher voltage than the fixed bias level at the non-inverting input, the output of IC4 will still

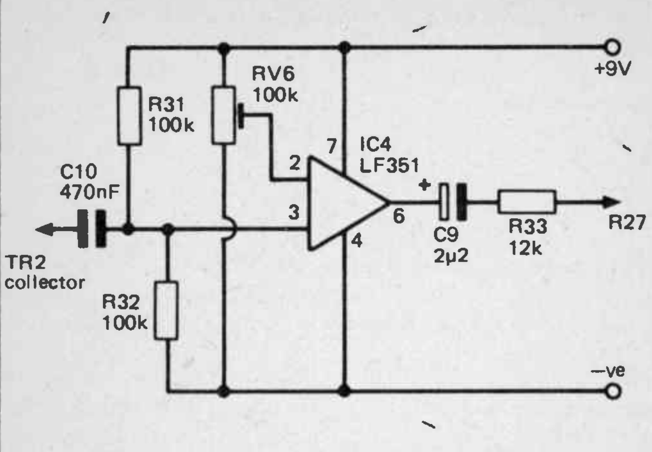


Fig. 29 The circuit diagram of the Noise Squarer

trigger to the high state on positive half cycles, but not on all positive half cycles. Triggering to the high state will only occur on positive excursions that are sufficiently strong to take the non-inverting input above the bias level at the inverting input. The higher the static input voltage difference is made, the fewer the positive half cycles that will operate the circuit, and the shorter the duration of positive output pulses. As far as the sound generated is concerned, the effect of adjusting RV6 away from the central balance point is to give a greater degree of roughness, with the signal degrading to a low frequency random pulse signal if RV6 is set for a large degree of unbalance. In practice RV6 is set to give the degree of roughness you prefer, and a little experimentation with various settings will soon familiarise you with the sounds available.

R33 is needed to attenuate the output of the circuit slightly, as it would otherwise be excessive for the VCF. C9 is part of the original noise generator circuit and is moved to the new position indicated in Figure 29, but all the other parts are needed in addition to those in the noise generator circuit. An LF351 operational amplifier is specified for IC4, but similar Jfet input devices such as the TL071 and TL081 are equally suitable. This is also true where the LF351 is specified for other projects in this book. The use of non-Jfet input devices such as the standard 741C type is not recommended.

Components for Noise Squarer (Fig. 29)

Resistors (All ¼ watt 5%)	
R31,32	100k
R33	12k
Potentiometer	
RV6	100k sub-min preset
Capacitor	
C10	470nF carbonate
Semiconductor	
IC4	LF351

Chime Synthesiser

The use of ring modulation to generate metallic chiming sounds was described in Chapter 1, and the same basic technique can be used with the envelope shaper unit to produce "bell" and similar sounds. The circuit diagram of a suitable signal source for this type of effect is shown in Figure 30.

This consists of two simple operational amplifier astable circuits based on IC3 which provide the signals to be ring modulated. The output signals are squarewaves, and frequency controls RV3 plus RV4 give an approximate frequency range of 50Hz to 5kHz from each oscillator. This is wide enough to enable a useful range of interesting effects to be obtained, ranging from quite low-pitched gong sounds to high-pitched bell type sounds.

The ring modulator described in Chapter 1 was based on a transconductance amplifier, and was capable of dealing with any type of input waveform. A much more simple approach can be adopted here as we are only dealing with squarewave signals, and a simple digital ring modulator can be used. In fact the ring modulator used here is just a two input CMOS logic gate, although it is admittedly an unusual and little used type. It is a 4070BE quad 2 input exlusive OR device, but only one of the gates is used here. The inputs of the other three gates are

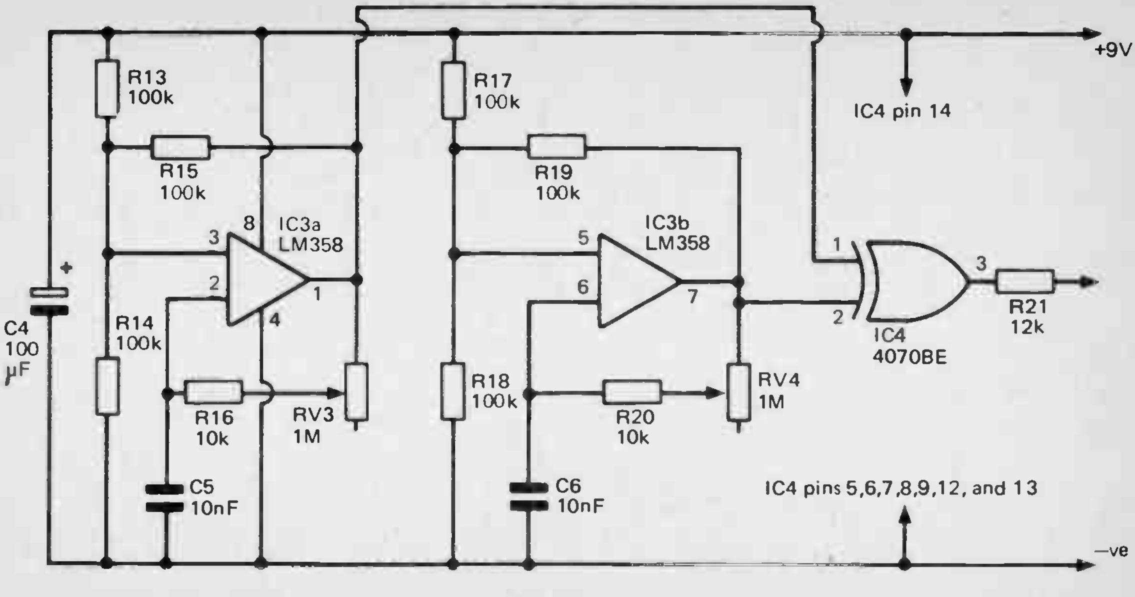


Fig. 30 The circuit diagram of the Oscillators and Digital Ring Modulator

connected to the negative supply rail to prevent possible damage to the device by static charges.

With an ordinary 2 input OR gate the output goes high if either of the inputs is taken high, or if both of them are taken high. An exclusive OR (EOR or XOR) gate is slightly different in that taking one input or the other high still results in the output going high, but taking both inputs high sets the output low. This subtle difference gives a much better mixing effect in an application of this type, and the use of an ordinary 2 input OR, NOR, AND, or NAND gate for IC4 is not recommended.

R21 is used to attenuate the output of the ring modulator as it would otherwise be excessive for the VCA in the Envelope Shaper. Remember that the 4070BE is a CMOS device and accordingly requires the standard antistatic handling precautions to be taken.

These synthesiser circuits permit a very wide range of sounds to be generated, and you might like to experiment to further expand the range of available sounds. For example, the VCF circuit could be used to process the output of the ring modulator prior to feeding the signal through to the Envelope Shaper.

Components for Chime Synthesiser (Fig. 30)

Resistors (All ¼ watt 5%)	
R13,14,15,17,18,19	100k
R16,20	10k
R21	12k
Potentiometers	
RV3,4	1M linear
Capacitors	
C4	100μF 10V elect
C5,6	10nF carbonate
Semiconductors	
IC3	LM358
IC4	4070BE
Miscellaneous	
Control knobs	
14 pin DIL IC holder	

In addition a full set of Envelope Shaper components is required.

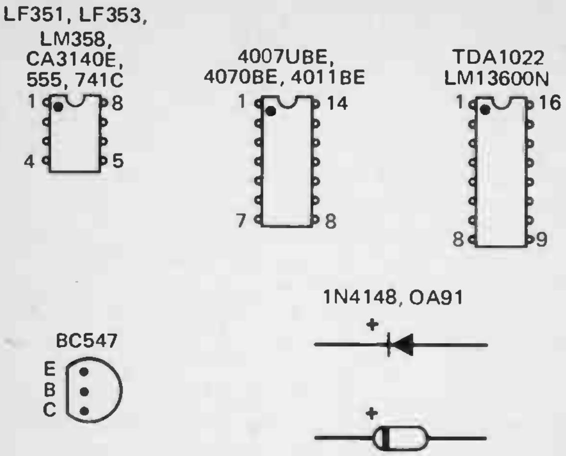


Fig. 31 Semiconductor pinout details (IC top views, transistor base view)

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