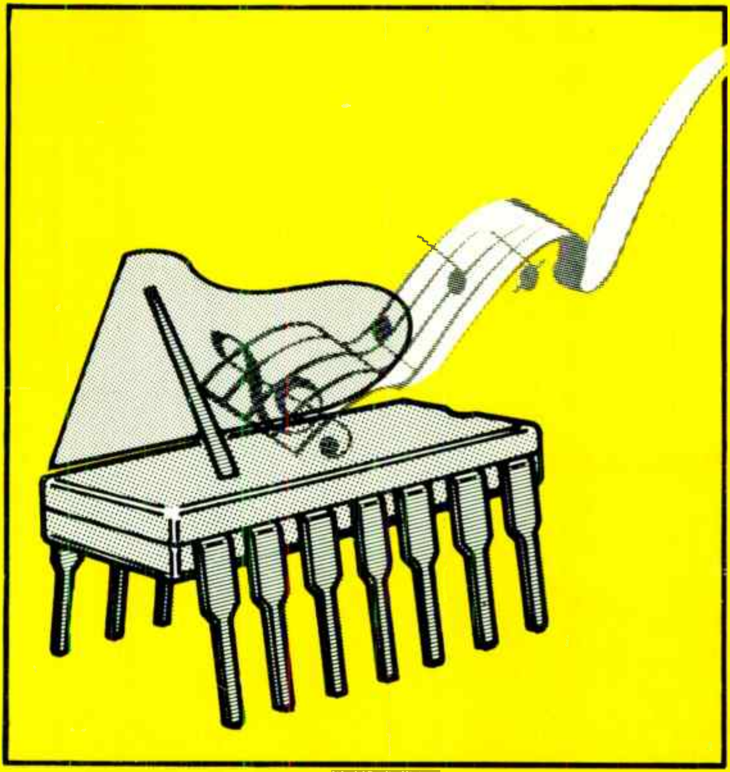
Practical Electronic Music Projects

R.A. PENFOLD



PRACTICAL ELECTRONIC MUSIC PROJECTS

by

R. A. Penfold

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Preface

I have been building and designing electronic gadgets for what must be about 30 years now, and electronic music projects have been a popular facet of the hobby for all that time. This popularity is not really surprising. You can have a lot of fun building music projects, and the finished units are genuinely useful. Unlike some other areas of electronics, the home constructed units are usually substantially cheaper than the ready-made alternatives. Although at one time music projects generally meant simple effects units and amplifiers, it is now possible to build a wide variety of music related devices. These range from basic effects units to complex MIDI accessories.

This book provides a number of circuits for electronic music projects of various types. These include guitar effects units and other guitar related projects, and general projects such as metronomes, audio mixers, and preamplifiers. The final chapter describes several MIDI related projects, including MIDI testers, a MIDI pedal, noise gates, and a THRU box. The circuits cover a range of complexities, but none are beyond the capabilities of the average electronics hobbyist. Some are simple enough for beginners to tackle (the soft and hard distortion units, and the MIDI THRU box for example). Where appropriate, notes on any awkward aspects of construction are provided, as are notes on setting up and using the devices. No test equipment is needed in order to get any of the circuits adjusted correctly. All the circuits are based on reasonably inexpensive components that are readily available.

R. A. Penfold

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Chapter 1

GUITAR PROJECTS

Home constructed add-ons for guitars have been a popular branch of the electronics hobby for many years now. In fact the earliest designs of this type were probably published before many readers of this book were born. When add-ons for guitars are mentioned, all manner of effects units tend to spring to mind. However, there are other useful electric guitar accessories that can be home constructed at less than the cost of ready-made equivalents. The units featured in this chapter include amplifiers, tuners, and effects units.

Tuning Fork

This very simple circuit has six switched output frequencies that are reproduced via a ceramic resonator. These notes correspond to the six open strings of a guitar (E, A, D, G, B, and E). The unit can therefore be used in much the same way as conventional tuning forks or pitch-pipes when tuning a guitar "by ear". The full circuit diagram for the electronic tuning fork appears in Figure 1.1.

The circuit is based on a standard 555 oscillator. A low power version of the 555 is used for IC1 in order to keep the current consumption to a minimum, and greatly extend the life of the battery. The timing components are C2, R1, R2, and whichever of the six preset resistors (VR1 to VR6) is switched into circuit via S1. The six switched preset resistors provide the unit with its six output frequencies. The bottom E is produced with VR1 in circuit, the A is produced with VR2 in circuit, through to the top E with VR6 switched into circuit.

The output of IC1 is fed to a seven stage binary divider (IC2) which gives a final output frequency that is 1/128th of IC1's output frequency. One reason for using a divider is that it gives a good quality squarewave output signal, which for the present purposes is better than the short pulse output waveform from IC1. These pulses would give rather a "thin" sound, with little fundamental content, like a synthesiser set

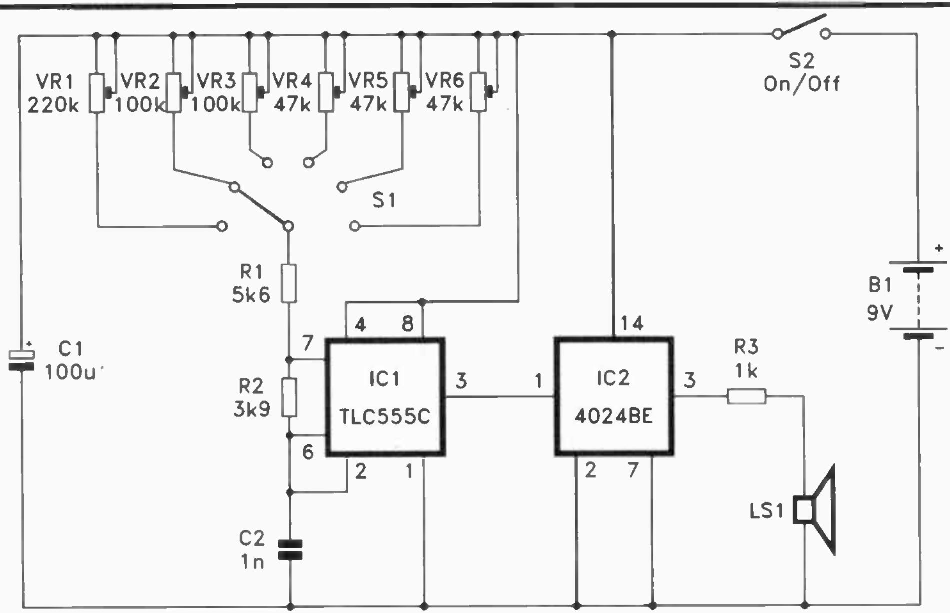

Fig.1.1 The electronic tuning fork circuit diagram

to produce a brief pulse signal. The main reason though, is that good oscillator stability is essential for this application. Good frequency stability is aided by the use of a 555 based circuit. Changes in the supply voltage due to battery ageing produce no significant variations in the output frequency.

The main cause of drift is likely to be temperature changes causing variations in the value of C2. As the output frequency of IC1 is relatively high it is possible to use a reasonably low value for C2. This means that it is possible to use a very high quality capacitor for C2 without arranging a mortgage first! A close tolerance (1%) polystyrene capacitor should provide excellent stability, but a silvered mica capacitor would probably offer the best performance. It is advisable to use multi-turn "trimpots" for VR1 to VR6. These are much more expensive than ordinary presets, but they have better stability and resolution. Using ordinary preset resistors it could be difficult to set the output frequencies with adequate accuracy. I generally use the horizontal mounting 18-turn "trimpots" in critical applications, but any multi-turn trimmers should give good results in this circuit.

IC2's final output drives the loudspeaker via R3. Note that loudspeaker is a cased ceramic resonator and not a normal moving coil loudspeaker. The circuit can not drive any form of moving coil loudspeaker properly, including high impedance types. The volume from the circuit was found to be slightly excessive on the higher output frequencies, and R3 was therefore included in order to reduce the volume to an acceptable level. With some ceramic resonators it might be better to use a lower value for R3, or even to drive the resonator direct from pin 3 of IC2. This depends on the exact characteristics of the particular resonator used.

The current consumption of the circuit is quite low, and may well be under one milliamp. A PP3 size battery is therefore adequate as the power source, and should have a very long operating life.

Construction and Adjustment

Construction of the project should be perfectly straightforward. However, bear in mind that IC2 is a CMOS device, and that it therefore requires the usual anti-static handling precautions. LS1 should be a cased ceramic resonator, not an uncased type. An uncased resonator would probably not provide enough volume even if driven direct from IC2, and it can also be difficult to make the connections to this type of component. Cased ceramic resonators normally have "flying" leads, so there should be no difficulty in making the connections to one of these. Although the leads of LS1 may be different colours, this is not a polarised component, and it can be connected either way round.

VR1 to VR6 must be set to give the correct notes, and this is just a matter of tuning them "by ear", one-by-one, against a set of pitch-pipes, an in-tune guitar, or any instrument which is in-tune and can provide the right notes. It is worth making the effort to get all six presets adjusted really accurately. Unless you like playing out of tune there is little point in having a tuning reference which lacks good accuracy!

Components for Electronic Tuning Fork (Fig.1.1)

Resistors (all 0.25 watt 5% carbon film)

R1	5k6
R2	3k9
R3	1k

Potentiometers (all multi-turn trimpots)

VR1	220k
VR2	100k
VR3	100k
VR4	47k
VR5	47k
VR6	47k

Capacitors

C1	100μ 10V elect
C2	1n 1% polystyrene or silvered mica

Semiconductors

IC1	TLC555C or similar low power 555 timer
IC2	4024BE

Miscellaneous

S1	6 way 2 pole rotary (only one pole used)
S2	SPST min toggle
LS1	Cased ceramic resonator
B1	9 volt (PP3 size)
	Case
	Circuit board
	Battery connector
	Control knob
	8 pin DIL IC holder
	14 pin DIL IC holder
	Wire, solder, etc.

Guitar Tuner

This project is based on the previous one (the electronic tuning fork), but it takes things a stage further. Like the tuning fork project, this device produces the same six notes as the open strings of a guitar. If desired, it can be used as a simple electronic tuning fork. However, it includes additional circuitry and a twin LED display which make accurate tuning very easy. In fact it is possible to tune an electric guitar really accurately using this device, even if you are literally tone deaf!

Provided the tuning of the guitar is reasonably close to the correct pitch, the two LEDs flash on and off at a rate which is equal to the difference between the guitar's output frequency and the reference frequency from the guitar tuner. In other words, the LEDs flash at what is generally termed the "beat rate". In order to accurately tune the guitar it is merely necessary to adjust the guitar for a very low flash rate from the LED display. There is no difficulty in tuning all six strings to within about 0.25Hz of the correct pitch.

System Operation

The block diagram of Figure 1.2 shows the general arrangement used in this guitar tuner. The reference oscillator and divider stages are basically the same as the equivalent stages in the tuning fork project described previously. The output from the divider chain is therefore six switched reference frequencies, which are the same as the six open string notes

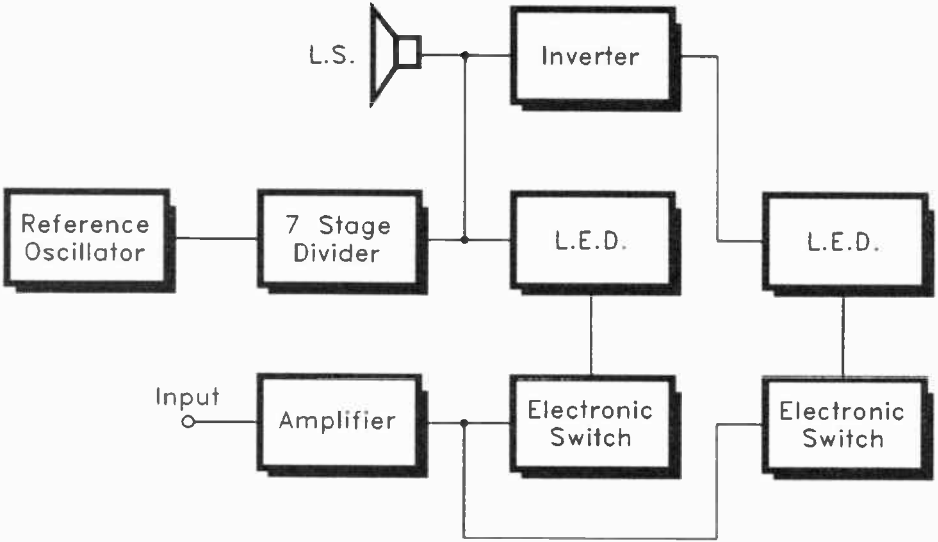

Fig. 1.2 The guitar tuner block diagram

from a guitar. These notes are available via a loudspeaker if the user wishes to use the unit to tune a guitar "by ear". The LEDs will only flash at a perceivable rate if the guitar's tuning is within about 25Hz of the correct frequency. The tones from the loudspeaker are useful for coarse adjustment of the guitar when fitting a new string, or retuning one that has slipped badly out of tune. An inverter stage generates an anti-phase oscillator signal.

An amplifier stage boosts the output from the guitar to a level that will operate two electronic switches. A LED is controlled by each electronic switch, but the positive supply for the LEDs is obtained via the inverted and non-inverted reference oscillator signals. A LED will only be activated if the electronic switch driving it is switched on, and the relevant oscillator signal is high.

The waveform diagram of Figure 1.3 shows example waveforms at various points in the circuit, and helps to explain the way in which the unit operates. The signal at the output of IC4 is the output from the amplifier stage. The electronic switches are on while this signal is high, and off when it is low. Therefore, the LEDs can only be activated while this signal is high. The signals at IC3 pin 3 and IC2 pin 3 are respectively the inverted and non-inverted oscillator signals. Again, each LED can only be switched on while its respective oscillator signal is high.

In the top set of waveforms the non-inverted oscillator signal is at the same frequency as the input signal, and is in-phase with the output from IC4. LED D2 is therefore switched on each time the output of IC4 goes high, but D1 is always switched off. Although D2 is pulsed on and off, the switching frequency is too high for the flashing to be perceived by a human observer, and D2 appears to light up continuously. The situation is similar in the middle set of waveforms, but it is now the output of IC3 that is in-phase with the input signal. Therefore, it is D1 that appears to light up continuously, and D2 that is continuously switched off. In the lower set of waveforms the input signal is 90 degrees out-of-phase with the two oscillator signals. Both LEDs are switched on, but only for about 25 percent of the time. Consequently, both LEDs appear to light up

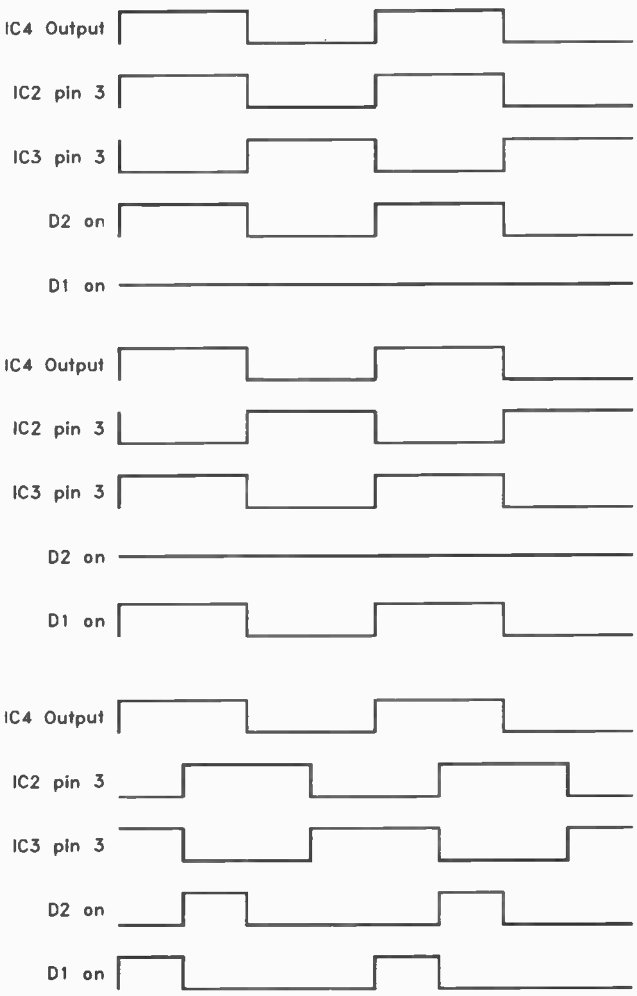

Fig. 1.3 Example guitar tuner waveforms

continuously, but at only about half brightness.

The LED circuit operates as a sort of basic phase indicator. If the pitch of the guitar exactly matches the selected reference frequency, the display will be "frozen". The display will indicate the relative phasing of the input signal and the reference oscillator, but this is of no consequence in the present application. It is the fixed state of the two LEDs that is of importance as it indicates that the guitar's tuning is accurate. If the guitar and reference oscillator produce slightly different frequencies, the phasing of the two signals will constantly vary. This results in the LEDs flashing on and off at a rate that is equal to the difference between the two frequencies. The guitar is therefore adjusted to produce the lowest achievable flash rate.

Strictly speaking, only one or other of the LEDs are needed, as they are providing what is essentially the same information. In practice a single LED display tends to be a bit confusing when the beat rate is very low. Adding the second LED gives a display which is significantly easier to use. However, if preferred the inverter stage, second LED, and the electronic switch which drives it can be omitted.

The Circuit

Refer to Figures 1.4 and 1.5 for the guitar tuner circuit diagram. Figure 1.4 shows the circuit for the oscillator, divider, and inverter stages. The oscillator and divider stages are the same as those used in the tuning fork project. The inverter is one NOR gate from a CMOS 4001BE, which is wired to give a simple inverter action. The other three gates in IC3 are unused, but have their inputs connected to the 0 volt supply rail in order to avoid spurious operations.

Turning to Figure 1.5, IC4 is used as the basis of the amplifier stage. This is a non-inverting amplifier which has its closed loop voltage gain controlled via VR7. Gains of up to about 200 or so (46dB) can be provided with VR7 set at maximum resistance. With low output guitar pick-ups it is necessary to use VR7 at or close to maximum resistance. With high output pick-ups results will probably be better with VR7 set well back towards minimum resistance. C5 provides a certain amount of high frequency roll-off which

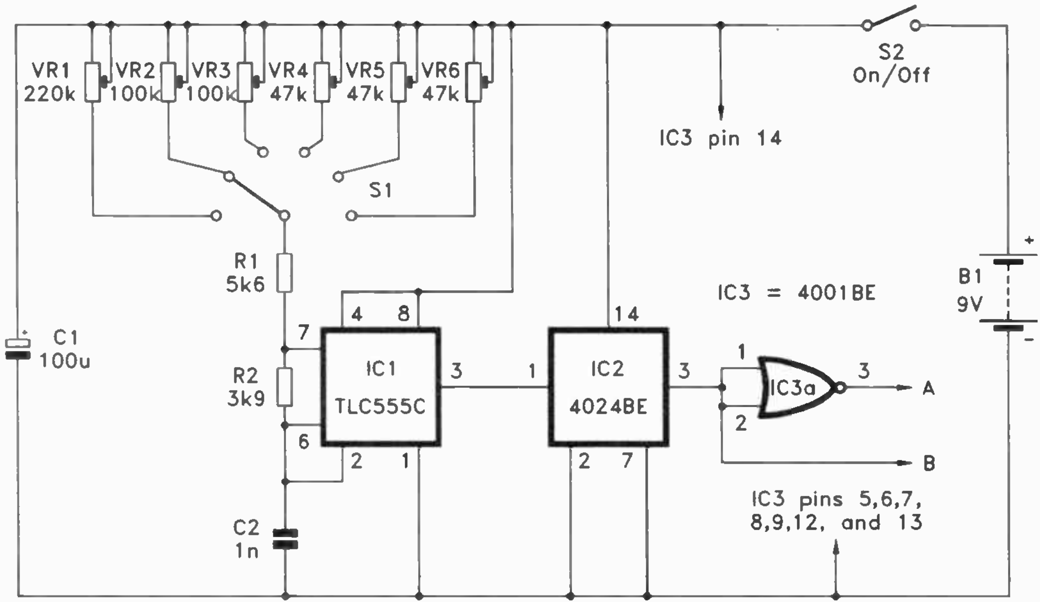

Fig.1.4 The oscillator, divider, and inverter stages

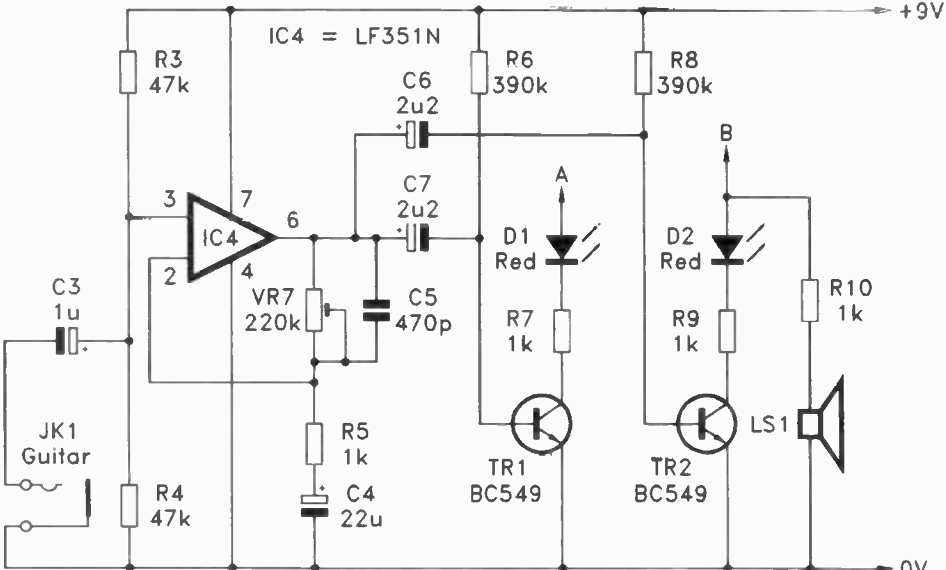

Fig.1.5 The amplifier and electronic switch stages

attenuates the harmonic content of the input signal. This gives a slightly better output waveform from IC4, and a clearer indication from the display. Even with this filtering, the output from IC4 will be something less than a perfect squarewave, but the LED display will still flash reasonably "cleanly" at the beat rate.

The two electronic switches are identical, and are simple common emitter types based on TR1 and TR2. R6 and R8 provide bias currents to TR1 and TR2 so that they are at an in-between state under quiescent conditions. On positive half cycles from IC4 they are switched on – on negative half cycles they are switched off. LS1 provides the audible signal to permit tuning "by ear". If this facility is not required, simply omit R10 and LS1. Alternatively, adding an SPST switch in series with these two components would enable the audible tone to be switched on and off, as desired.

The current consumption of the circuit is approximately 5 milliamps. A PP3 size battery is more than adequate as the power source, bearing in mind that the unit will only be used briefly and intermittently.

Adjustment and Use

Construction of this project presents few difficulties, but remember that IC2 and IC3 are CMOS devices. The normal anti-static handling precautions should be taken when dealing with these two devices. The average "on" current to D1 and D2 is not very high, so it is advisable to use high efficiency LEDs. VR1 to VR6 should be multi-turn "trimpots", and C2 must be a high quality component if the oscillator is to achieve good stability.

VR1 to VR6 can be adjusted "by ear" against tuning forks, pitch-pipes, or any instrument that is in-tune and provides the right notes. Alternatively, connect an in-tune electric guitar to JK1, and adjust each preset for the right note using the LED display as the tuning indicator. Initially it is probably best to start with VR7 at around one-third of maximum resistance. If the LED display tends to produce a lot of random flashing under standby conditions, results might be better if VR7 is adjusted for a slightly lower resistance. If the tuner only works for a second or two after a note has been played, setting VR7 for a higher resistance should improve matters.

If a preset should accidentally be set to produce the right note but in the wrong octave, the unit should still operate properly. The method used in this tuner has a big advantage over the main alternatives in that it is very tolerant of unusual input waveforms, and will even work if the input signal and reference oscillator are not on the same octave. However, the LED display will give a clearer indication of the beat frequency if the reference oscillator is on the right octave. Therefore, it is worth making the effort to make sure that the reference oscillator is set to the correct octave for each note.

Components for Guitar Tuner (Figs 1.4 and 1.5)

Resistors (all 0.25 watt 5% carbon film)

R1	5k6
R2	3k9
R3	47k
R4	47k
R5	1k
R6	390k
R7	1k
R8	390k
R9	1k
R10	1k

Potentiometers (VR1 – VR6 all multi-turn trimpots)

VR1	220k
VR2	100k
VR3	100k
VR4	47k
VR5	47k
VR6	47k
VR7	220k min preset

Capacitors

C1	100μ 10V elect
C2	1n 1% polystyrene or silvered mica

C3	1μ 50V elect
C4	22μ 25V elect
C5	470p ceramic plate
C6	2μ2 50V elect
C7	2μ2 50V elect

Semiconductors

IC1	TLC555C or similar low power 555 timer
IC2	4024BE
IC3	4001BE
IC4	LF351N
D1	Red LED
D2	Red LED

Miscellaneous

S1	6 way 2 pole rotary (only one pole used)
S2	SPST min toggle
LS1	Cased ceramic resonator
B1	9 volt (PP3 size)
	Case
	Circuit board
	Battery connector
	Control knob
	8 pin DIL IC holder (2 off)
	14 pin DIL IC holder (2 off)
	Wire, solder, etc.

Hard Distortion Unit

The distortion effect, which is also known as the "fuzz" effect, is one of the most simple to generate. In fact it is almost certainly the most simple of all the musical effects. There are several practical approaches to producing this effect, but the vast majority of distortion units are based on a clipping circuit of some sort. The effect of a clipping amplifier is much the same as over-driving an audio amplifier. In fact distortion units of this type are sometimes called "overdrive" units.

There are two basic types of clipping circuit, which are the "hard" and "soft" varieties. The waveforms of Figure 1.6

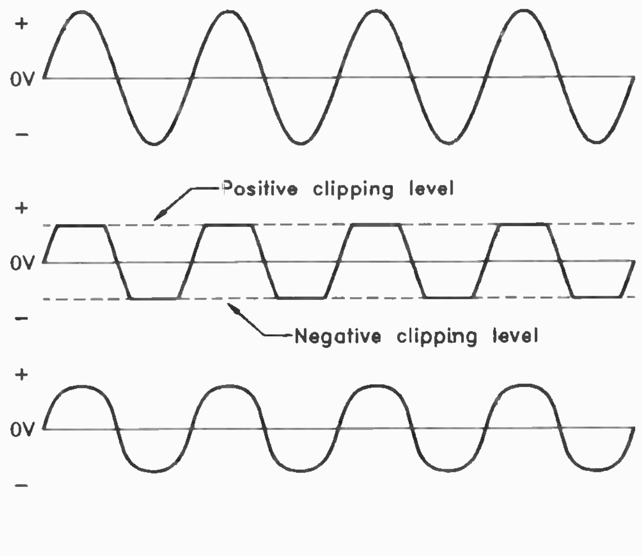

Fig.1.6 Hard and soft clipping waveforms

show the difference between the two types. The top waveform is the sinewave input signal, and the middle waveform is this signal after it has been subjected to hard clipping. With this type of clipping the signal rises from zero volts in the usual way, but once the positive threshold voltage is reached the signal voltage increases no further. No matter how much the input signal increases, the output will not significantly exceed the positive clipping voltage. Once the input voltage has fallen back below the threshold voltage, the output voltage falls back to zero in the normal way. It then starts to increase in the negative direction, but once the negative clipping level is reached the output voltage is unable to rise any further.

Hard clipping obviously produces a severely distorted output signal, even if the input signal only slightly exceeds the clipping levels. This is less than ideal in a musical context, and it gives rise to a number of practical problems. One of these is simply that a hard clipping distortion unit does not give a very controllable effect. Whether the input signal is set well above the clipping threshold, or only marginally above it, the output signal contains strong distortion products.

If you try to adjust the degree of clipping so that only the signal peaks are clipped, the result is not a moderate amount of distortion on the output signal. With hard clipping it tends to be a case of all or nothing. A slightly clipped signal and a heavily clipped signal do not sound very different. Initially a large amount of distortion is produced because the signal is strong and the peaks are clipped. As the signal decays it drops beneath the clipping level, and the distortion is abruptly removed. To my ears at any rate, this does not give a particularly musical effect.

Strong high frequency harmonics are generated by hard clipping. This gives a very "bright" sound which is favoured by some, but is not to everyone's liking. Strong intermodulation products are also generated, and this gives rise to the most serious practical drawback of hard clipping distortion units. The intermodulation distortion is so strong that it produces some very discordant sounds if more than one note at a time is played. It is therefore necessary to play with great precision when using an effects unit of this type.

Soft clipping is a much milder form of distortion, and the bottom waveform of Figure 1.6 shows the result of soft clipping a sinewave signal. Like hard clipping, at low voltages the waveform is not significantly altered. At the peaks of the signal there is a rounding down of the waveform, rather than the complete flattening of hard clipping. The clipping level is indistinct, and I suppose that strictly speaking there is no clipping level with soft distortion. As the signal amplitude increases, the gain is steadily reduced. Increases in the signal amplitude always produce some increase in the output level though, albeit very small increases at high input voltages where the clipping is at its hardest.

It is easy to underestimate the differences in the sounds produced by hard and soft clipping. Looking at the clipped waveforms there may not seem to be too much difference between the two. The important difference is the sharp angles in the hard clipped waveform, and the lack of them in the soft clipped one. Soft clipping generates strong lower harmonics, but only weak high frequency harmonics. It gives a much less "bright" but more "full bodied" sound. To my ears, the soft distortion effect is much more musical than the hard distortion variety.

Although the hard distortion effect was popular some years ago, recently there has been increased interest in the more traditional ("Hendrix") distortion effect, which is a soft distortion type. This is also known as the "valve" or "tube" distortion sound, because it is soft clipping that occurs when a valve amplifier is over-driven. Valves are called "tubes" in America incidentally. There is a strong point in favour of soft distortion in that it generates relatively little intermodulation distortion. This makes it possible to play two or more notes at once without the discordance sounds that occur with hard clipping.

The Circuit

Both types of distortion effect are normally generated using an operational amplifier clipping circuit. Figure 1.7 shows the circuit diagram for a hard distortion unit. IC1 is used in what is almost a standard non-inverting amplifier. R1 and R2 bias the non-inverting input to about half the supply potential, and set the input impedance of the circuit at about 50k. This should give good results with any normal guitar pick-up. R3, R4, and VR1 act as the negative feedback network. The voltage gain of the circuit can be varied from 11 times with VR1 at minimum value to 231 times with VR1 set for maximum resistance.

D1 and D2 are connected in parallel with VR1 and R4, and also form part of the negative feedback circuit. These are silicon diodes, and as such they only begin to conduct significantly once they are forward biased by about 0.6 volts. This means that they have no significant effect on the circuit with peak-to-peak output voltages of less than about 1.2 volts (i.e. plus and minus 0.6 volts). The diodes are brought into conduction when the output level exceeds 1.2 volts peak-to-peak, but only on the signal peaks where the plus and minus 0.6 volt levels are exceeded.

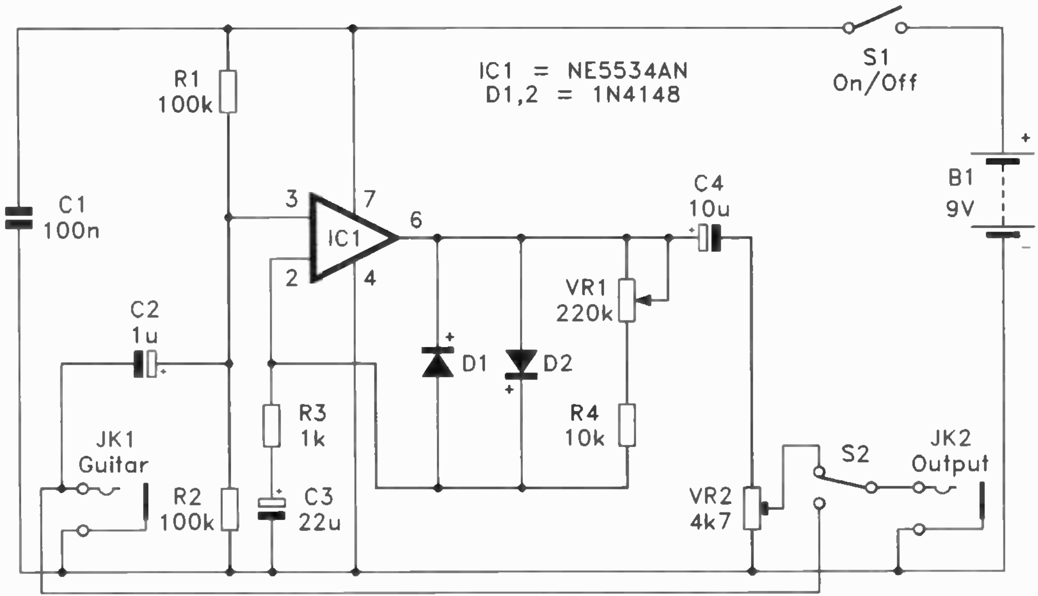

Fig.1.7 The hard distortion unit circuit diagram

Although a silicon diode has a very high resistance below its forward threshold voltage (typically many megohms), its resistance falls very rapidly above the conduction threshold voltage. With a forward bias of only 0.65 volts or so the resistance might be no more than a few ohms. Thus, as the output voltage tries to go beyond 1.2 volts peak-to-peak, the extra feedback through the diodes reduces the closed loop voltage gain of IC1, and prevents a significant increase in the output voltage. This gives a clipped output signal, and due to the abrupt switch-on characteristic of the silicon diodes it is hard clipping that is produced.

As explained previously, with hard clipping it is not really possible to control the strength of the effect. VR1 does give some control over the amount of distortion added to the input signal, but not enough to be of any great practical value. It is needed more to permit the basic gain of the circuit to be adjusted to suit a wide range of guitar pick-ups. The output level from electric guitars varies enormously from one to another. In fact the highest output pick-ups produce at least ten times the output voltage available from low output types. In general, the higher the cost of a pick-up, the greater its output level.

Adjustment of VR1 should permit good results to be obtained using any normal guitar pick-up(s). For the best results VR1 should be set for the lowest gain (lowest resistance) that gives a properly distorted signal, even once notes have decayed somewhat. Setting a higher level of gain will probably still give good results, but the background noise level will be higher than is really necessary. Also, the more gain that is added into the signal chain, the greater the risk of problems with feedback and "hum". Bear in mind that a fair amount of extra gain is involved even when the unit is used with VR1 at minimum resistance. With any conventional distortion unit added into the system it is always necessary to take extra care to avoid problems with feedback, "hum", and general noise.

The NE5534AN specified for IC1 is a very low noise and distortion bipolar operational amplifier. The low distortion performance of this device is obviously of no practical importance in the current application, but its low noise level does help to keep the background "hiss" level to a minimum. Although devices such as the NE5534AN were once many times more expensive than "ordinary" operational amplifiers such as the Bifet LF351N, the difference in cost is quite small these days. Although the circuit will work quite well using an LF351N, TL071CP, etc., for IC1, the slightly higher cost of the NE5534AN is probably well justified, especially if the unit is used with a low output guitar pick-up. The background noise level is about ten times higher using an LF351N for IC1.

VR2 is a variable output attenuator. This is simply set to give approximately the same volume whether the effect is switched in, or bypassed using S2. It is not necessary to use any test equipment to make precise measurements when doing this. It is just a matter of setting VR2 by trial and error, checking for differences in volume "by ear". With guitars that have very high output levels it might be necessary to back-off the guitar's volume control and set VR2 for maximum output. In other words, the guitar might have a higher output level than the distortion unit, making it necessary to adjust the guitar to match the distortion unit, rather than vice versa.

S2 provides a very basic form of bypass switching, but it is a method I have always found to be perfectly adequate in practice. It connects the output socket to VR2's wiper in order to switch in the effect, or connects the output socket direct to the input socket in order to bypass the effect. This leaves the guitar connected to the input of the distortion unit even when the effect is bypassed. This is not likely to cause any problems in use, but the full bypass switching via an DPDT switch can be used if preferred.

Most users will wish to use the "look no hands" method of switching the effect in and out. This means that S2 must be operated by foot, which in turn means that it should ideally be a heavy duty push-button switch. The usual choice is a heavy duty push-button switch of the successive operation variety. With a switch of this type the effect is switched in the first time the switch is operated, switched out on the next operation, switched in again the third time it is operated, and so on. One slight drawback of this system is that switches of this type tend to be a bit slow in operation, and they are also quite noisy (in terms of the sound they make, not noisy electrically). You may prefer to use a large push-button switch of the non-locking variety. A switch of this type will switch in the effect when it is pressed, and switch it out again when it is released. The only problem with switches of this type is that they are unlikely to be as hard wearing as a heavy-duty type, and might need to be replaced periodically.

Of course, if S2 is a foot operated switch it must be mounted on the top panel of the case where it is accessible, and the case must be a suitably tough type. Diecast aluminium boxes are generally regarded as the best choice as they are extremely strong, and they also provide excellent screening against mains "hum" and other electrical noise. Unfortunately, they are also relatively expensive. Simple folded aluminium boxes offer a good inexpensive alternative. These are nothing like as tough as diecast aluminium boxes, and they have slightly inferior screening properties. However, I have always found them to be perfectly adequate when used as cases for effects units. I would not recommend using plastic cases, some of which are made from a rather brittle type of plastic. Also, plastic cases do not provide electrical screening.

The current consumption of the circuit is only a few milliamps, and a small (PP3 size) 9 volt battery is adequate to supply this. The circuit can be powered from a 9 volt mains power supply unit, but it must be a type which has a low ripple content on its output. In practice this means a good quality type having a stabilised output. Non-stabilised types generally have quite high "hum" levels on their output, and the distortion circuit does not include filtering to deal with this. The circuit for a low noise mains power supply unit is provided in the final section of this book.

Components for Hard Distortion Unit (Fig.1.7)

Resistors (all 0.25 watt 5% carbon film)

R1	100k
R2	100k
R3	1k
R4	10k

Potentiometers	
VR1	220k lin carbon
VR2	4k7 min preset
Capacitors	
C1	100n ceramic
C2	1μ 50V elect
C3	22μ 16V elect
C4	10μ 25V elect
Semiconductors	
IC1	NE5534AN
D1	1N4148
D2	1N4148
Miscellaneous	
S1	SPST min toggle
S2	SPDT heavy duty push-button (see text)
JK1	Standard 6.35mm jack socket
JK2	Standard 6.35mm jack socket
B1	9 volt (PP3 size)
	Battery connector
	8 pin DIL IC holder
	Control knob
	Metal case
	Circuit board, wire, solder, etc.

Soft Distortion Unit

Figure 1.8 shows a modified version of the distortion unit which provides a form of soft clipping. All that has been done here is to add a germanium diode in series with each of the silicon diodes. Germanium and silicon diodes have very different forward transfer characteristics. The forward resistance of a germanium diode is quite high at very low forward voltages, but nothing like as high as the equivalent figure for a silicon diode. Furthermore, the resistance starts to fall at a much lower voltage, and the transition from the "off" state to the "on" state is far more gradual.

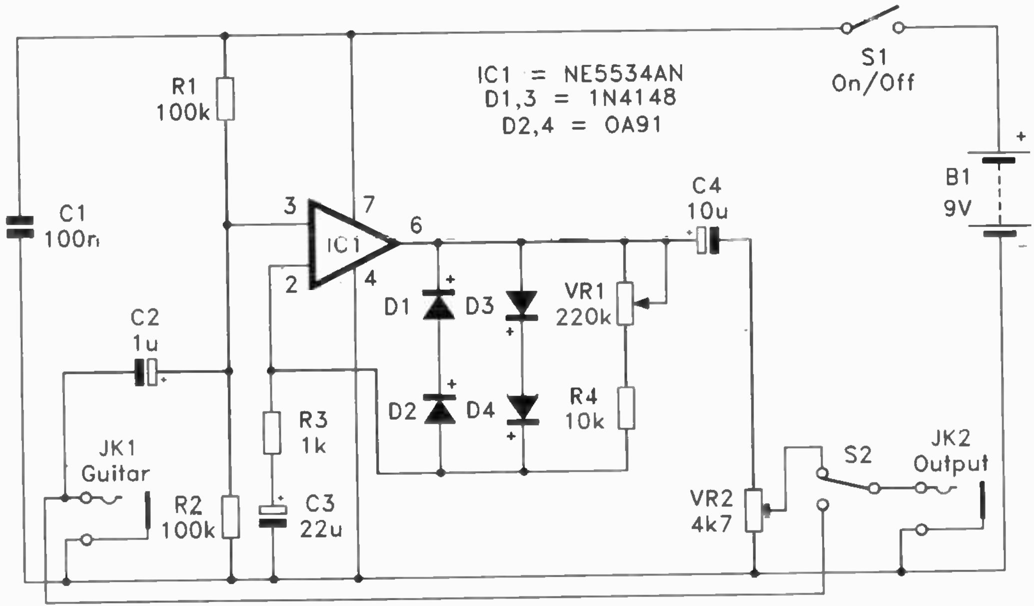

Fig.1.8 The circuit diagram for the soft distortion unit

A typical germanium diode begins to conduct more readily at a forward potential of well under 0.1 volts, but does not reach saturation point until the forward voltage reaches almost 0.2 volts. When applied to a clipping amplifier such as the one used in the distortion circuit, a germanium diode therefore gives a more gradual reduction in gain as the input voltage rises. This gives the required soft clipping, and a very good distortion effect.

This circuit uses a combination of silicon and germanium diodes, which results in harder clipping than is obtained using germanium diodes alone. On the other hand, the clipping is much less hard than using only silicon diodes, and the circuit provides what is really a form of soft clipping. In terms of the effect obtained, results are very good with a strong distortion effect being produced. The higher harmonics are tamed though, as is the intermodulation distortion. This produces a slightly less "bright" effect, and makes it safe to play more than one note at a time.

The notes about constructing and setting up the hard distortion unit apply equally to the soft clipping version, and will not be repeated here. One additional point which should be noted is that germanium diodes are not as hardy as silicon types. In particular, germanium diodes are much more vulnerable to heat damage. D2 and D4 must therefore be soldered into place reasonably quickly, with the iron being applied to each soldered joint for no longer than is really necessary. I have never found it necessary to use a heat-shunt when soldering germanium devices into circuit.

In this version of the unit VR1 provides rather more control over the strength of the distortion effect. However, to a large extent it is still needed to compensate for variations in the output levels from guitar pick-ups. It is worth experimenting with different settings for VR1, and listening carefully to the changes that are produced in the effect. With this soft distortion effect you should find that the effect gradually dies away reasonably smoothly on notes that are allowed to fully decay. This contrasts with the hard clipping effect where the distortion tends to end very abruptly as the signal drops below the rigidly defined clipping level.

The input and output of these distortion circuits are in-phase, and they have quite high voltage gains with VR1 set at maximum resistance. It is therefore necessary to take due care to avoid stray feedback when designing the component layouts.

Components for Soft Distortion Unit (Fig.1.8)

Resistors (all 0.25 watt 5% carbon film)

R1	100k
R2	100k
R3	1k
R4	10k

Potentiometer

VR1	220k lin carbon
VR2	4k7 min preset

Capacitors

C1	100n ceramic
C2	1µ 50V elect
C3	22µ 16V elect
C4	10µ 25V elect

Semiconductors

IC1	NE5534AN
D1, D3	1N4148 (2 off)
D2, D4	OA91 (2 off)

Miscellaneous

S1	SPST min toggle
S2	SPDT heavy duty push-button (see text)
JK1	Standard 6.35mm jack socket
JK2	Standard 6.35mm jack socket
B1	9 volt (PP3 size)
	Battery connector
	8 pin DIL IC holder
	Control knob
	Metal case
	Circuit board, wire, solder, etc.

Pseudo Echo Unit

This circuit is really a variation on a tremolo generator, but when used with a guitar it can produce an effect that is reminiscent of an echo unit. An ordinary tremolo unit varies the volume at a rate which is variable between (typically) about 0.2Hz and 10Hz. The variations in volume are quite gradual, and the effect is basically the same as the one produced by continuously operating a swell pedal. This unit is rather less subtle, and it operates by simply switching the output signal on and off. The output signal is therefore a series of signal bursts, and due to the envelope shape of a guitar's output signal, the strength of these bursts steadily diminishes. This gives a sort of echo-style effect.

The waveforms of Figure 1.9 show this process. The top waveform is the output signal from the guitar, and the bottom waveform is the signal from the l.f.o. that controls the switching of the output signal. The middle waveform is the output from the unit, which is a series of signal bursts of diminishing amplitude.

System Operation

Figure 1.10 shows the block diagram for the pseudo echo unit. The main signal path is through an input buffer amplifier, an electronic switch, and an output buffer stage. The output signal is gated on and off by applying a low frequency pulse signal to the control input of the electronic switch. This signal is provided by a gated low frequency oscillator (l.f.o.).

On the face of it, the stages mentioned so far are all that is needed in order to give the desired effect. However, there is a flaw in this basic arrangement. The l.f.o. is not synchronised with the playing of notes on the guitar. This means that a new note could be commenced just as the l.f.o. switches off the output signal. This would give rather poor results, particularly when the unit is used with the l.f.o. set for a very low switching frequency. Apart from other considerations, it would seriously disrupt the timing of the output signal, and to the listener this would appear to be due to inept playing.

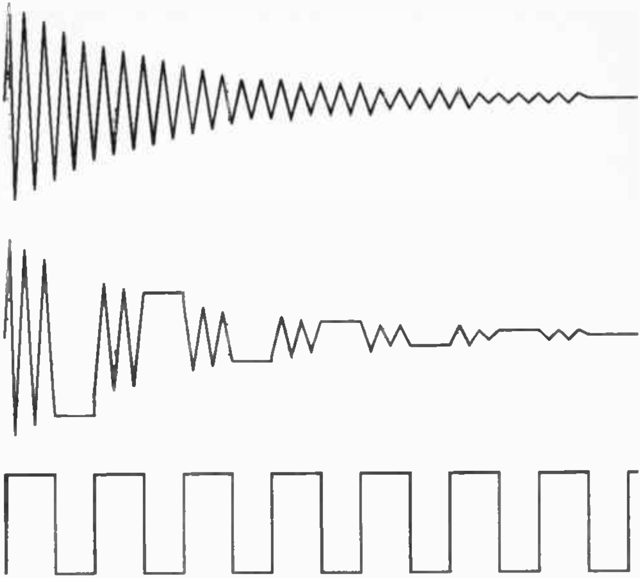

Fig. 1.9   Example waveforms for the pseudo-echo unit

This problem is overcome by using a gated oscillator. The gate signal is produced by first amplifying some of the input signal. The amplified signal is then fed to a rectifier and smoothing circuit which produces a positive d.c. output signal. The amplitude of this signal is proportional to the strength of the input signal. The smoothing circuit has quite short attack and decay times so that its output signal accurately follows changes in the input level. The output from the smoothing circuit is fed to the input of a voltage detector circuit. Provided there is at least a small output voltage from the smoothing circuit, the output of the voltage detector goes high and activates the l.f.o.

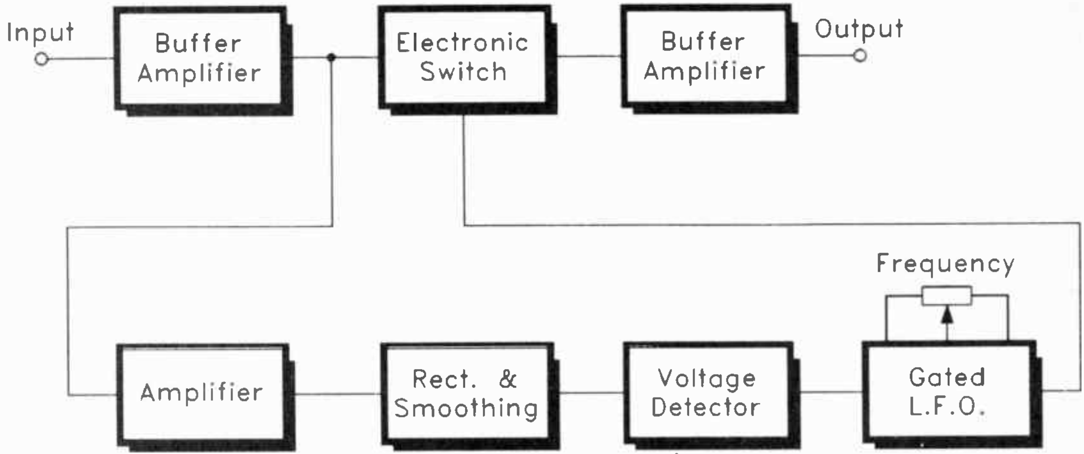

Fig. 1.10 The pseudo-echo unit block diagram

If there is a gap in the input signal, even a very brief one, the output voltage from the smoothing circuit will rapidly decay to an insignificant level, and the voltage detector will then switch off the l.f.o. When the input signal commences again, the voltage detector is activated, and the l.f.o. is switched on again. When gated on, the l.f.o. always starts from scratch with a new cycle, rather than carrying on where it left off. This means that its output goes high initially, letting the input signal pass through to the output. This gives the required synchronisation, with the beginning of each new note being passed through the unit.

The Circuit

The circuit diagram for the pseudo echo unit appears in Figures 1.11 and 1.12. Taking Figure 1.11 first, this shows the circuit for the input and output buffer stages, and the electronic switch. The input buffer stage is a simple non-inverting type based on IC1. Its output feeds direct into the electronic switch, which is one of the four switches in a 4016BE CMOS quad analogue switch (IC2). No connections are made to the other three switches in IC2. Note that the 4066BE is not recommended for use in an audio application such as this. The 4066BE is pin for pin compatible with the 4016BE, and tends to be regarded as a superior version of the 4016BE having a lower "on" resistance. However, the 4066BE is not totally compatible with the 4016BE, and it tends to produce switching "clicks" when used in audio applications. Therefore, only the 4016BE should be used in the IC2 position of this circuit.

The output buffer amplifier is based on IC3, and it is another non-inverting type. Together with IC2 and C4 this constitutes a basic sample and hold circuit. C4 has no significant affect on the circuit when the electronic switch is closed as it is driven from a fairly low source impedance. When the switch opens, C4 holds whatever signal voltage happened to be present at that instant. The ultra-high input impedance of IC3 ensures that C4 does not discharge significantly while the switch remains open. The charge on C4 immediately jumps to the new signal voltage when the electronic switch closes again. This helps to minimise switching "clicks",

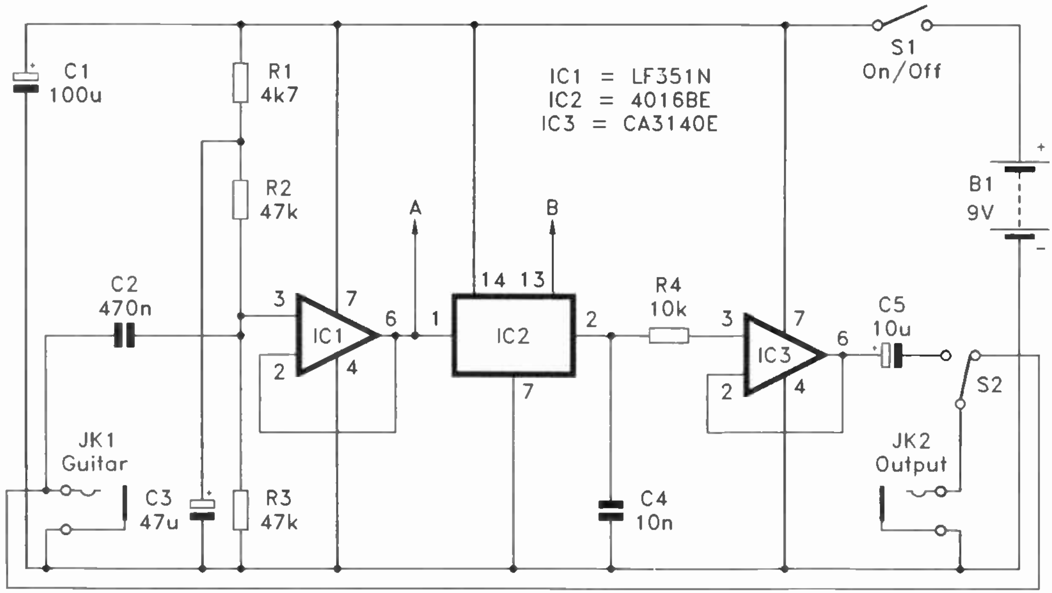

Fig.1.11 The circuit for the buffer and switch stages

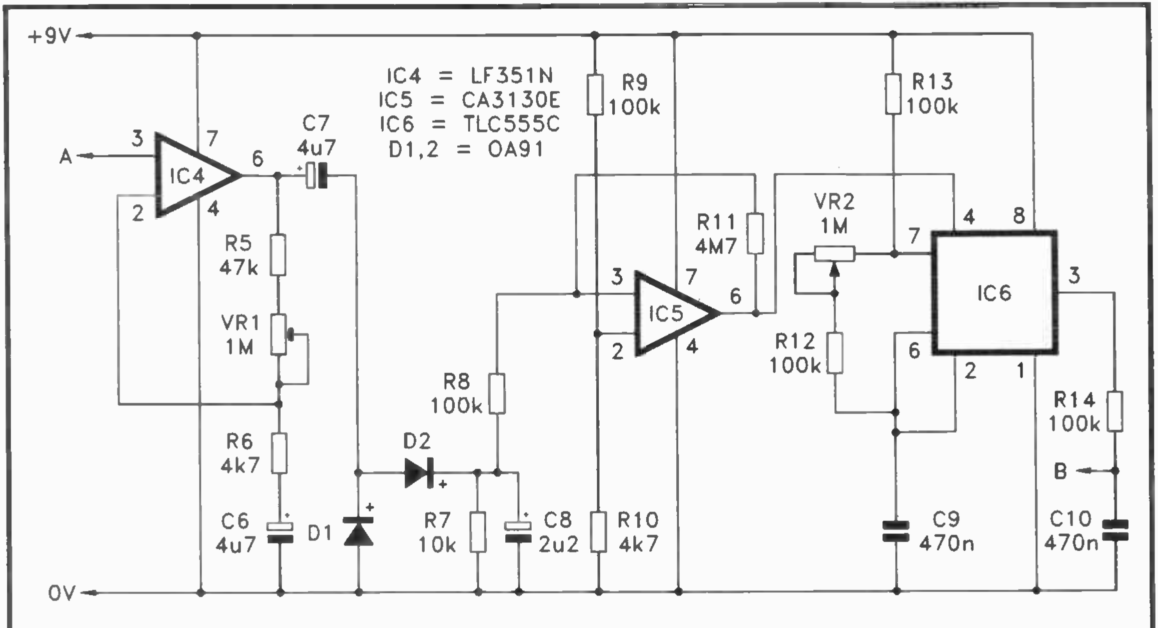

Fig.1.12 The circuit for the remaining stages of the pseudo-echo unit

whereas simply leaving the input of IC3 floating during the "off" periods of IC2 would almost certainly exacerbate the problem.

In the circuit of Figure 1.12 IC4 acts as the amplifier stage which drives the rectifier and smoothing circuit. This is a non-inverting circuit which has a closed loop gain that can be varied by means of VR1. The closed loop voltage gain of IC4 is about 11 times with VR1 at minimum resistance, rising to approximately 222 times with VR1 set at maximum resistance. This enables the unit to be set up to operate efficiently with high, medium, and low output guitar pick-ups. The smoothing and rectifier circuit is a conventional half-wave type based on D1 and D2. Germanium diodes are used for D1 and D2 as these have lower forward voltage drops which provide better results in this application.

The voltage detector is based on IC5. This is a conventional non-inverting type which has a small amount of hysteresis provided by R11. This helps to provide clean switching of the output. R9 and R10 provide a small reference voltage to the inverting input of IC5. It is when this voltage is significantly exceeded that the output of IC5 triggers to the high state.

The gated oscillator is based on IC6, which is a low power 555. This is used in a conventional gated astable circuit which has the control signal applied to the reset input (pin 4). The output frequency is variable from about 1Hz with VR2 at maximum resistance, to about 12Hz with it at minimum resistance. It is advisable to use a low power version of the 555 for IC6, since the standard 555 tends to "crowbar" the supply on output transitions. This could easily couple "clicks" into the signal path. Low power versions of the 555 are free from this problem. Any low power version of the 555 (TLC555C, L555, etc.) should work well in this circuit. R14 and C10 form a lowpass filter which slightly slows down the switching action, and helps to minimise switching noises on the output signal.

The current consumption of the circuit is about 9 to 10 milliamps. This would give a rather short battery life using a PP3 size battery, and it is advisable to use a higher capacity type such as six HP7 size cells in a plastic holder.

Setting Up

Construction of the unit is fairly straightforward, but bear in mind that IC2 and IC3 are MOS integrated circuits, and that they require the normal anti-static handling precautions. It should also be borne in mind that D1 and D2 are germanium diodes, and that they are more vulnerable to heat damage than the more familiar silicon diodes. Complete the soldered joints reasonably quickly when connecting D1 and D2 to the circuit board.

This pseudo echo unit does not provide the most subtle effect ever devised, and you should certainly have no difficulty in hearing the action of the unit on the output signal! VR2 should be set for a fairly low frequency in order to produce an echo type effect. To my ears at any rate, the effect with VR2 set for higher switching frequencies is actually a rather more interesting one. As with practically any effects unit, it is a good idea to try various control settings in order to determine which effects you like the most. It will also be necessary to experiment a little with various settings for VR1. The best setting is probably the lowest resistance that gives reliable operation, without the modulation being cut off before notes have had time to decay to a fairly low level. If the resistance provided by VR1 is too low, notes will be severely truncated. If the setting of VR1 is too high, the oscillator will not be retriggered each time a new note is played.

Components for Pseudo Echo Unit (Figs 1.11 & 1.12)

Resistors (all 0.25 watt 5% carbon film)

R1	4k7
R2	47k
R3	47k
R4	10k
R5	47k
R6	4k7
R7	10k
R8	100k
R9	100k
R10	4k7

R11	4M7
R12	100k
R13	100k
R14	100k

Potentiometers

VR1	1M min preset
VR2	1M lin carbon

Capacitors

C1	100µ 10V elect
C2	470n polyester
C3	47µ 16V elect
C4	10n polyester
C5	10µ 25V elect
C6	4µ7 50V elect
C7	4µ7 50V elect
C8	2µ2 50V elect
C9	470n polyester
C10	470n polyester

Semiconductors

IC1	LF351N
IC2	4016BE
IC3	CA3140E
IC4	LF351N
IC5	CA3130E
IC6	TLC555C or similar
D1	OA91
D2	OA91

Miscellaneous

S1	SPST min toggle
S2	SPDT heavy duty push-button
JK1	Standard 6.35mm jack socket
JK2	Standard 6.35mm jack socket
B1	9 volt (6 × HP7 size cells in holder)
	Battery connector (PP3 type)
	8 pin DIL IC holder (5 off)
	14 pin DIL IC holder

Control knob
Metal case
Circuit board, wire, solder, etc.

Guitar Headphone Amplifier

This project is useful for practice at home where a high power amplifier would be too large and too loud. It is also useful if your practising is simply not appreciated by the neighbours or others in your household! This amplifier drives a pair of headphones at good volume so that you can hear your playing loud and clear, but others only hear the acoustic output of the guitar. With all the electric guitars I have ever heard, the acoustic output is not loud enough to seriously annoy someone even just a few feet away.

The full circuit diagram for the headphone amplifier appears in Figure 1.13. The circuit is based on an NE5532N dual low noise operational amplifier. Although the low noise level provided by this device is useful, and gives virtually "hiss" free results, its relatively high output drive current is probably of greater importance in this application. It permits a pair of medium impedance headphones (the type sold as replacements for personal stereos) to be driven at quite high volume levels.

IC1a acts as an input buffer stage which provides an input impedance of about 100k. The input signal is fed to IC1a via volume control VR1, which shunts the input impedance down to about 30 to 40k, depending on the setting of VR1. This is still high enough to give good results with any normal guitar pick-up. IC1b is an inverting amplifier which provides the voltage amplification. R4 and R5 set the voltage gain at 10 times (40dB). This is not particularly high, but it should be borne in mind that the unit is designed for use with headphones that require only a low drive voltage in order to produce good volume levels. The gain of the circuit is therefore adequate to produce good results with a low output guitar pick-up, and with a high output pick-up the volume control will probably need to be kept well backed off. R6 slightly reduces the loading on the output of IC1b, which gives improved audio quality. The stereo headphones should be

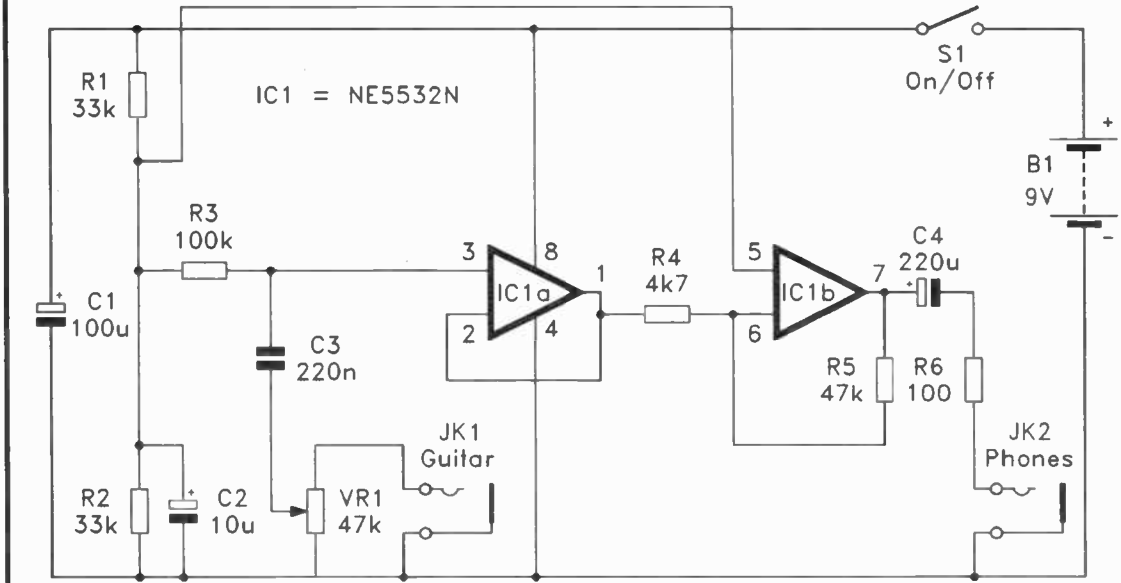

Fig. 1.13 The guitar headphone amplifier circuit

driven in series incidentally, and not in parallel (which might produce excessive losses through R6).

The current consumption of the circuit is approximately 6 milliamps, and a PP3 size battery should therefore be adequate as the power source. As the gain of the circuit is not particularly high, and the input and output of the circuit are out-of-phase, the layout is not critical.

It is advisable to use an all-metal case, and to earth it to the 0 volt supply rail. The case will then screen the circuit board and wiring from mains "hum" and other electrical interference, giving a very low background noise level. JK2 should be wired in such a way that the headphones are driven in series. This means ignoring the common earth tag, and connecting the output of the amplifier to the other two tags (either way round). An insulated socket should be used for JK2, as there will otherwise be an unwanted connection from the earth tag to the 0 volt supply via the case.

The gain of the circuit should be adequate for use with low output pick-ups, but if necessary the gain can be boosted somewhat by increasing the value of R5 to 100k or so. It would be beneficial to reduce R5 to about 10k if the unit will only be used with high output pick-ups. A better control characteristic will then be obtained from the volume control. Of course, it is perfectly acceptable to connect an effects unit between the guitar and the headphone amplifier, and the effect should be clearly audible via the headphones.

Components for Guitar Headphone Amplifier (Fig.1.13)

Resistors (all 0.25 watt 5% carbon film)

R1	33k
R2	33k
R3	100k
R4	4k7
R5	47k
R6	100R

Potentiometer

VR1	47k log carbon

Capacitors

C1	100μ 10V elect
C2	10μ 25V elect
C3	220n polyester
C4	220μ 10V elect

Semiconductor

IC1	NE5532N

Miscellaneous

S1	SPST min toggle
JK1	Standard 6.35mm jack socket
JK2	3.5mm stereo jack socket
B1	9 volt (PP3 size)
	Metal case
	Circuit board
	Battery connector
	Control knob
	8 pin DIL IC holder
	Medium impedance stereo headphones
	Wire, solder, etc.

Guitar Practice Amplifier

This amplifier, like the headphone amplifier described previously, is intended for use when practising at home. Instead of driving headphones it operates with an 8 ohm loudspeaker which it can drive at a few watts r.m.s. This is not likely to be enough power for "live" stage performances, but it is about right for practising at home without disturbing the neighbours. The circuit incorporates bass and treble tone controls. The amplifier can be battery or mains powered.

The guitar practice amplifier circuit is shown in Figures 1.14 and 1.15. The volume and tone control stages are shown in Figure 1.14. The guitar feeds straight into the volume control, and from here the signal is taken to the input of a conventional active tone control circuit. This is based on IC1, which is used as an inverting mode amplifier. The tone control network provides frequency selective negative feedback that produces the required tailoring of the frequency response.

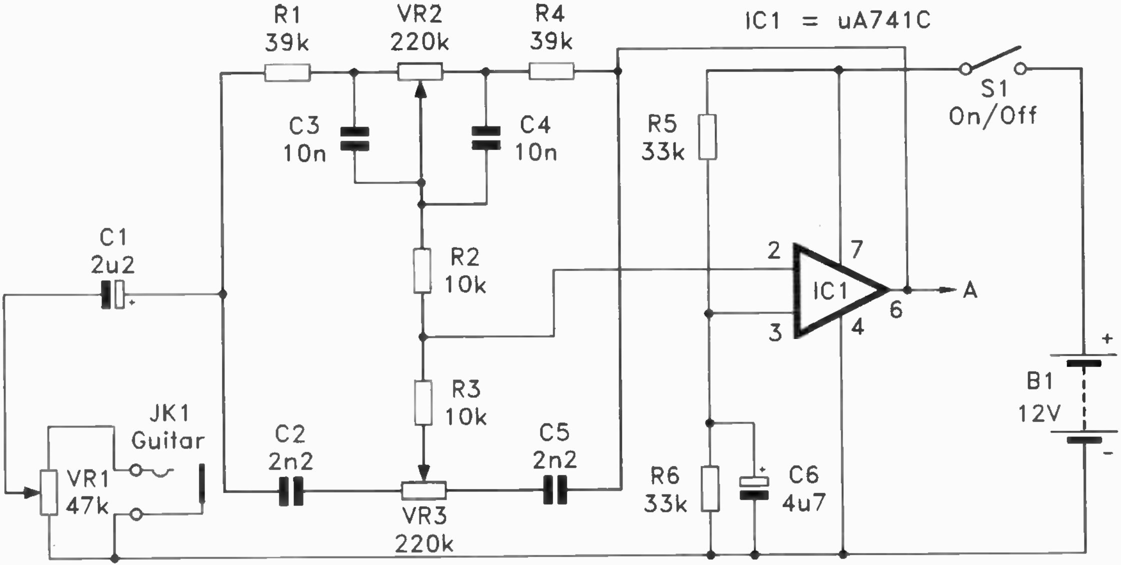

Fig.1.14 The tone control stage of the guitar practice amplifier

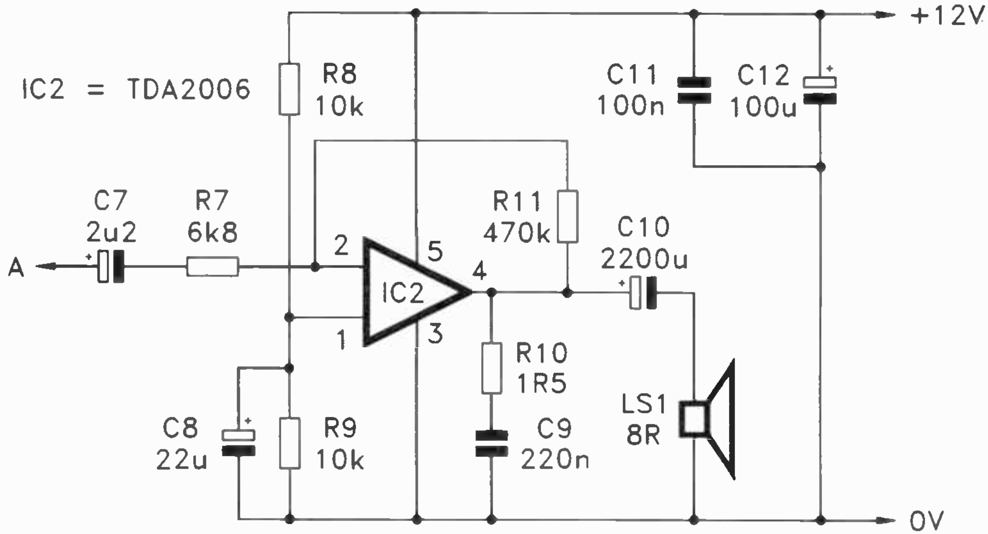

Fig. 1.15 The power amplifier section of the guitar practice amplifier

VR2 is the bass control, and VR3 is the treble control. Maximum boost is produced with the wipers set full to the left – maximum cut is produced with the wipers set fully to the right.

Figure 1.15 shows the circuit diagram for the power amplifier stage. The circuit is based on the TDA2006, which is effectively a high power operational amplifier, complete with inverting and non-inverting inputs. Unlike some other power amplifier chips that are basically operational amplifiers, the TDA2006 does not have any built-in bias or feedback components. It can therefore be used in standard operational amplifier configurations.

In this case IC2 is used in a standard inverting mode configuration. A fair amount of voltage gain is needed in order to obtain good volume using low output guitar pick-ups. Accordingly, the values of feedback resistors R7 and R11 have been chosen to give a closed loop voltage gain of about 70 times. The value of R11 can be reduced to about 100k if the amplifier will only be used with high output guitar pick-ups. This will give a better control characteristic from VR1, and will also give improved fidelity from IC2. Even with R11 at 470k, the quality of reproduction is more than adequate for the present application. The TDA2006 incorporates both output short circuit and thermal overload protection incidentally.

R10 and C9 may be needed in order to aid high frequency stability, although in my experience the TDA2006 is one of the more stable audio power amplifier chips, and this output circuit is not always necessary. It is probably best to include R10 and C9, as they should ensure good stability if the unit should happen to be used with an "awkward" loudspeaker. The quiescent current consumption of the circuit is approximately 40 milliamps, but it can be many times higher than this when the amplifier is used at high volume levels. Powering the circuit from batteries is only a practical proposition if a high capacity type is used (e.g. eight HP2 (D) size cells in a plastic holder). Ten rechargeable D size cells would be a more economic choice in the medium to long term. If portability is not important, a mains power supply (such as the one described in the following section of this book) is the most practical choice.

Construction of any unit of this type presents a few difficulties. The main point to note is that IC2 must be mounted on a reasonably large heatsink. IC2 is not being run at anything like full power in this circuit, but it still requires something more substantial than a small bolt-on heatsink. Probably the best method is to use a metal case for the project, and then mount IC2 on this via a small bracket fabricated from 18 s.w.g. aluminium. There is no need to use an insulation set to electrically isolate IC2 from the bracket and case. IC2's heat-tab connects internally to its 0 volt supply lead, and the case will presumably be connected to the 0 volt supply as well. For those who are not familiar with devices such as the TDA2006, it should perhaps be explained that it has an encapsulation that is very much like a plastic power transistor, but it has five "legs" rather than three.

This circuit provides a substantial amount of power gain, which makes the layout rather more critical than normal. In particular, take care over the arrangement of the earthing points. The order of the earthing points should follow along basically the same lines as the earthing points in the circuit diagrams (JK1, IC1, IC2, LS1, C11 – C12, and then the 0 volt supply input point). The unit can be built as a combination amplifier/loudspeaker, but my preference would be to build it as an amplifier to drive a separate loudspeaker. As the output power is only a few watts, the loudspeaker should obviously have a reasonably high efficiency so that it gives plenty of volume from the relatively low output power. The loudspeaker must also be a type that is capable of handling about 4 watts r.m.s. Miniature 8 ohm loudspeakers can only handle a few hundred milliwatts r.m.s., and are not suitable.

Components for Guitar Practice Amplifier (Figs 1.14 & 1.15)

Resistors (all 0.25 watt 5% carbon film)

R1	39k
R2	10k
R3	10k
R4	39k

R5	33k
R6	33k
R7	6k8
R8	10k
R9	10k
R10	1R5
R11	470k

Potentiometers

VR1	47k log carbon
VR2	220k lin carbon
VR3	220k lin carbon

Capacitors

C1	2µ2 50V elect
C2	2n2 polyester
C3	10n polyester
C4	10n polyester
C5	2n2 polyester
C6	4µ7 50V elect
C7	2µ2 50V elect
C8	22µ 25V elect
C9	220n polyester
C10	2200µ 16V elect
C11	100n ceramic
C12	100µ 25V elect

Semiconductors

IC1	µA741C
IC2	TDA2006 or TDA2006V

Miscellaneous

S1	SPST toggle switch
LS1	8R impedance loudspeaker (see text)
JK1	6.35mm jack socket
B1	12 volt (e.g. 8 × HP2 size cells in holder)
	Metal case
	Circuit board
	8 pin DIL IC holder
	Heatsink (see text)

Control knob (3 off)
Wire, solder, etc.

Mains P.S.U.

If you wish to power the unit from the mains supply, the power supply circuit of Figure 1.16 is suitable. This has a low ripple and general noise content on its 15 volt regulated output. A 15 volt supply gives a slightly higher maximum output power than that provided using a 12 volt battery supply. However, IC2 is still used well within its maximum voltage and power ratings, and there is no risk of it being "zapped" by a 15 volt supply. It is actually quite in order to use a 15 volt battery supply to power the amplifier, but this requires two extra batteries, and reduces battery life. This would significantly increase running costs. The circuit is a straightforward type having push-pull full-wave rectification. Both electronic smoothing and voltage regulation are provided by IC1.

This power supply circuit is included for those constructors who are suitably experienced at electronic project building, and who are competent to deal with a project that connects to the dangerous mains supply. Bear in mind that mistakes with a circuit of this type could prove lethal. This circuit must be constructed to comply with the normal safety regulations. In particular, it must be housed in a case of all-metal construction, and both the case and the 0 volt output of the supply must be reliably earthed to the mains earth lead. The case must be a type which has a lid secured by means of screws, not a clip-on type that would give easy access to the dangerous mains wiring.

Components for Mains Power Supply (Fig.1.16)

Capacitors

C1	2200µ 40V elect
C2	220n ceramic
C3	100n ceramic

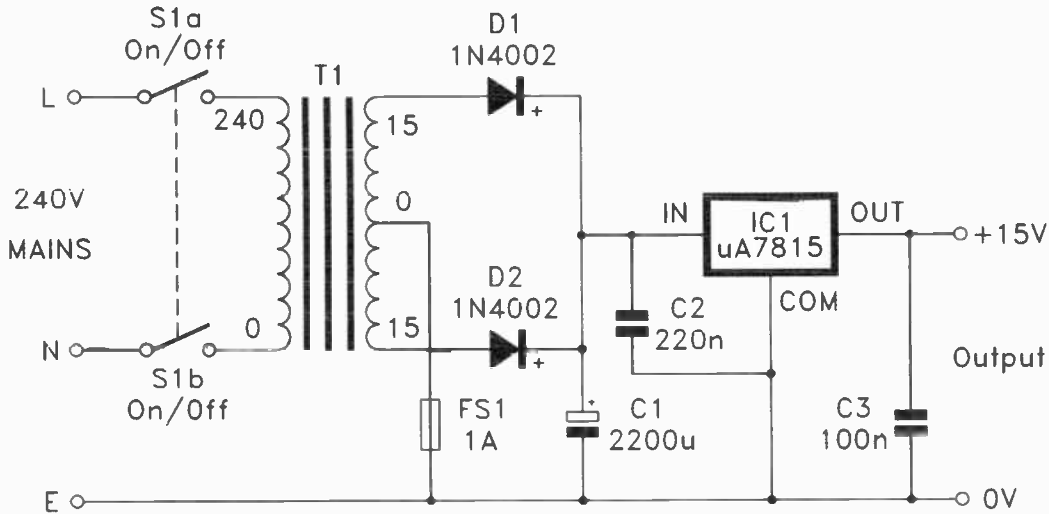

Fig. 1.16 The circuit for the mains power supply unit

Semiconductors

D1	1N4002 (100V 1A)
D2	1N4002 (100V 1A)
IC1	μA7815 (15 volt 1A positive regulator)

Miscellaneous

T1	Standard mains primary, 15–0–15 volt 1A secondary
FS1	1A 20mm anti-surge
S1	Rotary mains switch
	All-metal case with screwed lid
	Circuit board
	20mm fuse holder
	Control knob
	3 core mains lead and fused plug (2A)
	Wire, solder, etc.

Chapter 2

MISCELLANEOUS MUSIC PROJECTS

The projects featured in this chapter are of use to keyboard players and guitarists. In fact the two metronomes are not even specific to users of electric or electronic instruments, and are also usable by those who play acoustic instruments.

Simple Metronome

A metronome simply produces a "clicking" sound at regular intervals, and it enables players to perform pieces at the correct tempo. Sheet music is often marked with the tempo in beats per minute. The "click" rate control of a metronome has a calibrated scale which enables the desired beat rate to be set quite accurately. The original metronome (Maelzel's metronome) was a purely mechanical device which was driven by a form of clockwork motor, and had a swinging arm that pivoted at the bottom. It therefore provided both audible and visual indicators of the beat rate.

Mechanical metronomes are still very much in everyday use, but they are now starting to become musical curios. Modern metronomes are electronic devices, but they perform the same basic function as Maelzel's metronome. In most cases they produce very much the same sort of "click" sound as mechanical metronomes. Some also produce a visual indication of the beat rate, usually via some form of LED display.

This electronic metronome is very basic, but it provides good results. It provides reasonably loud "clicks" from its built-in loudspeaker, and it provides a visual indication of the beat rate by way of a LED that flashes each time a "click" is generated.

The Circuit

The circuit diagram for the metronome appears in Figure 2.1. This is really little more than a standard 555 astable (oscillator). A low power version of the 555 is used in order to keep the current consumption of the circuit to a minimum, but a

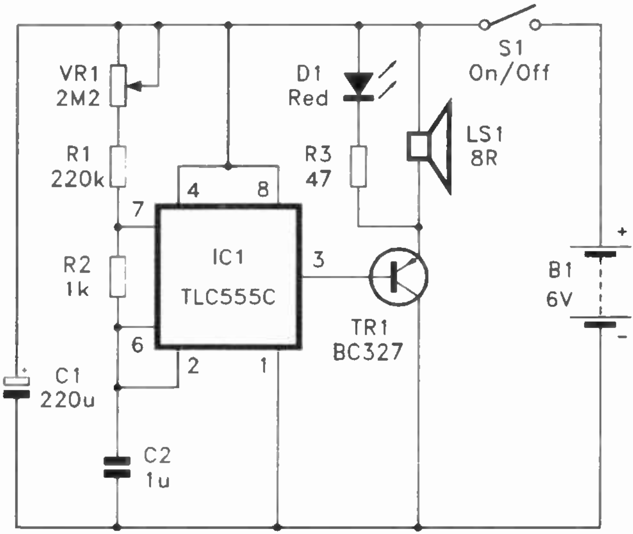

Fig.2.1 The metronome circuit diagram

standard 555 can be used if you are prepared to accept a somewhat reduced battery life. The timing components are VR1, R1, R2, and C2. VR1 enables the operating frequency to be varied over an approximate range of 0.5Hz to 5Hz. In terms of beats per minute this represents about 30 with VR1 at maximum resistance, rising to about 300 with it set at minimum resistance.

The output of IC1 (pin 3) is high while C2 charges via the three timing resistances, and low while C2 discharges via R2 and an internal switching transistor of IC1. Due to the relatively low value of R1, the output waveform is a series of brief negative pulses. It is brief pulses that are needed in order to produce the required "clicking" sounds, and the actual pulse duration should be no more than about one millisecond. The specified value for R2 gives a pulse duration of roughly 0.7 milliseconds. This is short enough to give a good "click" sound, but long enough to produce reasonable volume. The output pulses are coupled to the loudspeaker via an emitter follower buffer stage (TR1). This also drives the LED indicator (D1) via current limiting resistor R3.

The current consumption while TR1 is driving LS1 and D1 is quite high. Supply decoupling capacitor C1 has been given a fairly high value so that it can aid the supply of these large current pulses. Even with the beat rate control at maximum, the average current consumption of the circuit is only a few milliamps. The circuit is powered from either four AAA or four AA (HP7) size batteries in a plastic battery holder. Either way, even with frequent use the batteries should last at least a few months.

Construction

Construction of this very simple project should not present any real difficulties. The volume of the "clicks" produced by the unit is likely to be more satisfactory if LS1 is a reasonably large loudspeaker having a diameter of about 76 millimetres. Sub-miniature types having diameters of about 40 to 50 millimetres are usable, but are likely to give substantially lower volume than a slightly larger type. High impedance loudspeakers can be used, but again, they are likely to provide a relatively low volume level. Although D1 is driven with fairly hefty pulses of current, the duration of the pulses is quite short. In order to obtain reasonably bright flashes it is necessary to use a high efficiency LED. I would also recommend the use of a fairly large LED (about eight millimetres in diameter).

An electrolytic capacitor can be used for C2, but I would strongly urge the use of a polyester type. Some electrolytic capacitors have quite high leakage currents, and will not work well in a low current timing circuit such as the one used in this project. The tolerances of electrolytic capacitors are quite high, and can actually be as large as plus 100 and minus 50 percent! At best the tolerance is likely to be plus and minus 20%, which is also the normal tolerance for a potentiometer (including VR1 in this circuit). Using a polyester capacitor for

C2, with its closer tolerance rating, should ensure that the range of beat rates provided by VR1 is reasonably close to the specified range.

Mounting miniature loudspeakers on front panels tends to be a bit awkward, since these components almost invariably lack any form of mounting bracket. I usually drill a matrix of 5 millimetre diameter holes in the front panel to act as a loudspeaker grille, and then glue the loudspeaker in place behind this. Any general purpose adhesive should do the job quite well, but try to avoid smearing the adhesive over the loudspeaker's diaphragm. Use a minimal amount of adhesive applied only to the front rim of the loudspeaker.

A rotary potentiometer can be used for VR1, but there is some advantage in using a large slider type. The long travel of the slider enables a large scale to be used, which makes it much easier to obtain good calibration accuracy. Also, it is then possible to use a logarithmic potentiometer, but to use it upside-down as it were, to give an anti-logarithmic law. This gives something approximating to a linear scale, which again aids good calibration accuracy. Essentially the same ploy can be used with a rotary potentiometer if it is connected so that clockwise rotation produces increased resistance. Unfortunately, this gives a reverse reading scale (i.e. clockwise rotation of the control knob gives a reduced beat rate). This problem could be avoided by using an anti-log potentiometer, but it is unlikely that a suitable component will be available. If you use a rotary potentiometer it should be fitted with a large control knob so that a reasonable scale length is obtained.

The calibration points have to be located by trial and error, and this is likely to be a time consuming business. To determine the beat rate simply count the number of "clicks" in a period of one minute. Good accuracy should be obtained at the higher beat rates if you count the number of beats in a period of twenty seconds, and then multiply by three to get the number of beats per minute. The beat rate scale is easily marked on the front panel using ordinary rub-on transfers.

Components for Metronome (Fig.2.1)

Resistors (all 0.25 watt 5% carbon film)	
R1	220k
R2	1k
R3	47R
Potentiometer	
VR1	2M2 lin carbon (see text)
Capacitors	
C1	220µ 10V elect
C2	1µ polyester
Semiconductors	
IC1	TLC555C or similar
TR1	BC327
D1	High efficiency red LED
Miscellaneous	
S1	SPST min toggle
LS1	8 ohm moving coil loudspeaker, about 76mm diameter
B1	6 volt (four AA or AAA size cells in a plastic holder)
	Small case
	Circuit board
	Control knob
	8 pin DIL IC holder
	Battery connector (PP3 type)
	Wire, solder, etc.

Accented Metronome

Some mechanical metronomes have a bell which can be used to give an accented beat facility. If there are (say) three beats to the bar, the bell mechanism is set so that the bell sounds on every third beat. This electronic metronome is the modern equivalent to a mechanical accented beat metronome. Rather than a bell, or bell-type sound, it uses a lower

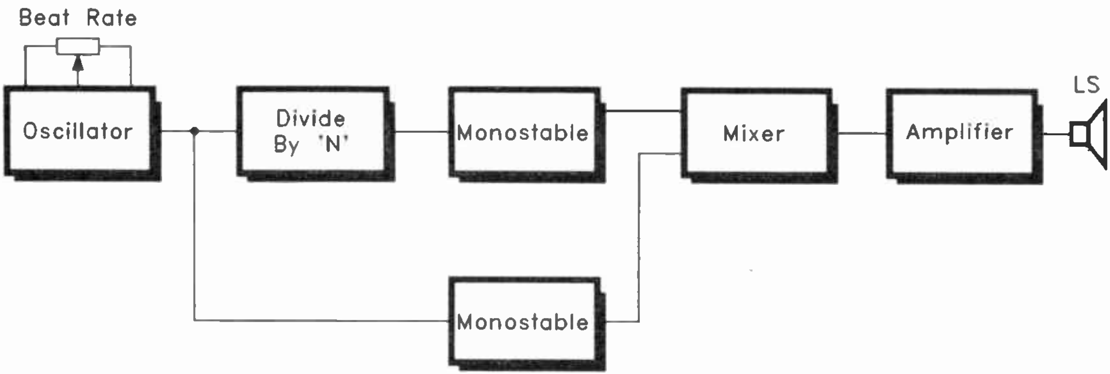

Fig.2.2 The accented metronome block diagram

pitched "click" to accent beats. In other words, on an accented beat it provides what is more of a "thud" sound than a "click". This method is very simple, but it is one that I have always found to be quite good in practice. It is easy to distinguish between normal and accented beats, and the "thud" sound is less distracting than the "ting" of a bell.

Like the metronome described previously, this one can be set for any beat rate from about 30 to 300 beats per minute. It can be set to accent every second, third, fourth, sixth, or eighth beat. It can also be set to provide normal operation with no beats accented. It has optional LED indicators.

System Operation

The block diagram of Figure 2.2 shows the general make-up of the accented beat metronome. An oscillator controls the beat rate, and covers the range mentioned previously. The oscillator drives a monostable which provides an output pulse duration of well under one millisecond. This monostable generates the normal (non-accented) "clicks". The short pulse duration gives fairly high-pitched "click" sounds which are easily distinguished from the lower pitched pulses on accented beats. These pulses are fed to a mixer stage, and then on to an amplifier which drives the loudspeaker.

The output from the oscillator also drives a divide by 'N' counter. This can be set to divide by two, three, four, six, or eight. The output of this counter drives another monostable, and this one has a somewhat longer output pulse duration than the other monostable. It produces pulses that are still little more than a millisecond in duration, and these provide the lower pitched "thuds" for accented beats. Of course, this monostable is not triggered on every clock cycle. It is triggered on every second, third, fourth, sixth, or eighth clock cycle, depending on the division rate of the divide by 'N' counter. In other words, to accent every fourth beat, the divide by 'N' counter is set to divide by four. The output from this monostable, like the other monostable, drives the loudspeaker via the mixer stage and the output amplifier.

The Circuit

Figures 2.3 to 2.5 show the circuit diagram for the accented

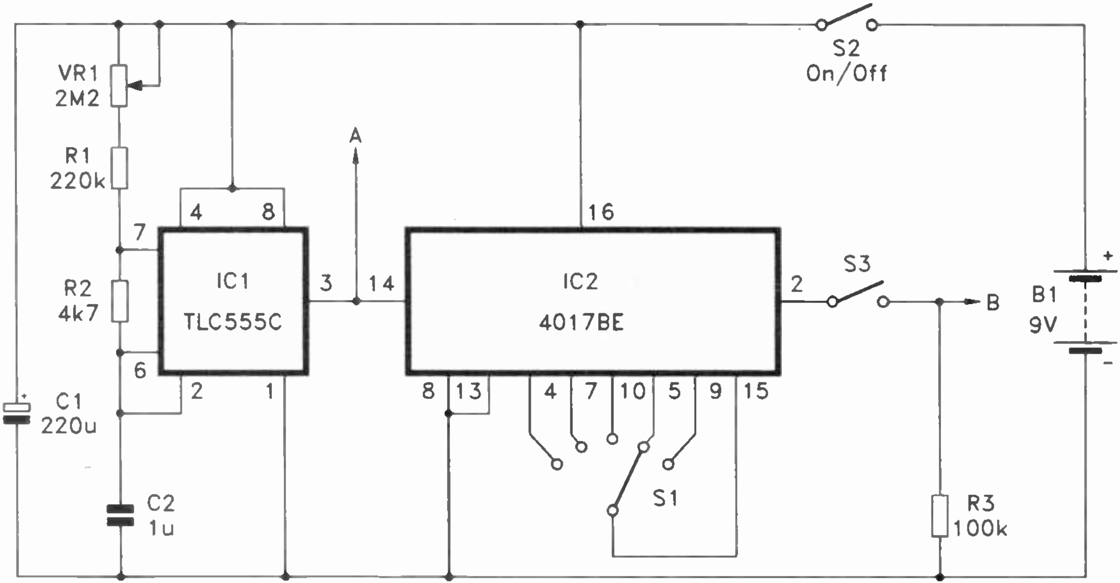

Fig.2.3 The clock oscillator and divide by 'N' stages

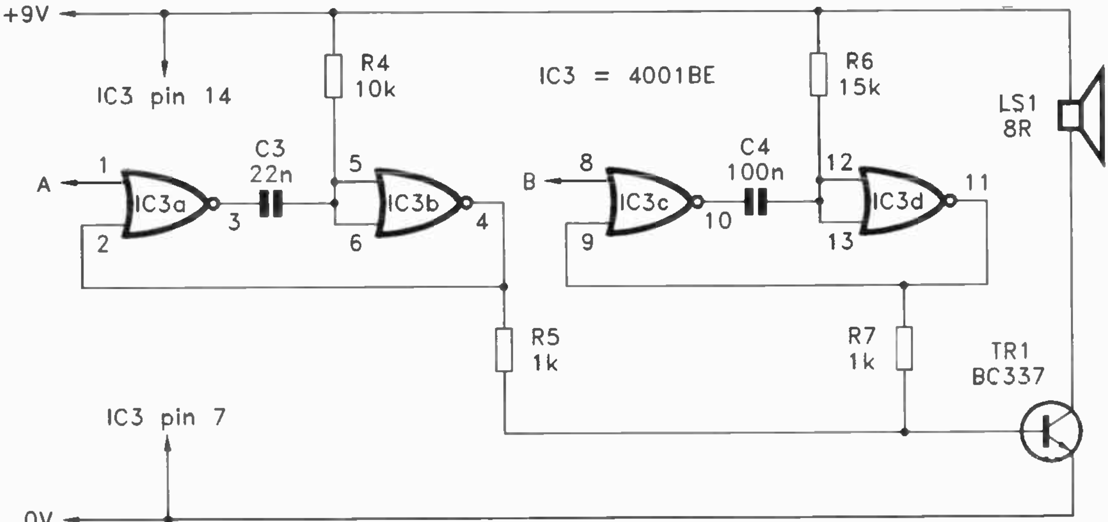

Fig.2.4 The monostable and output amplifier stages

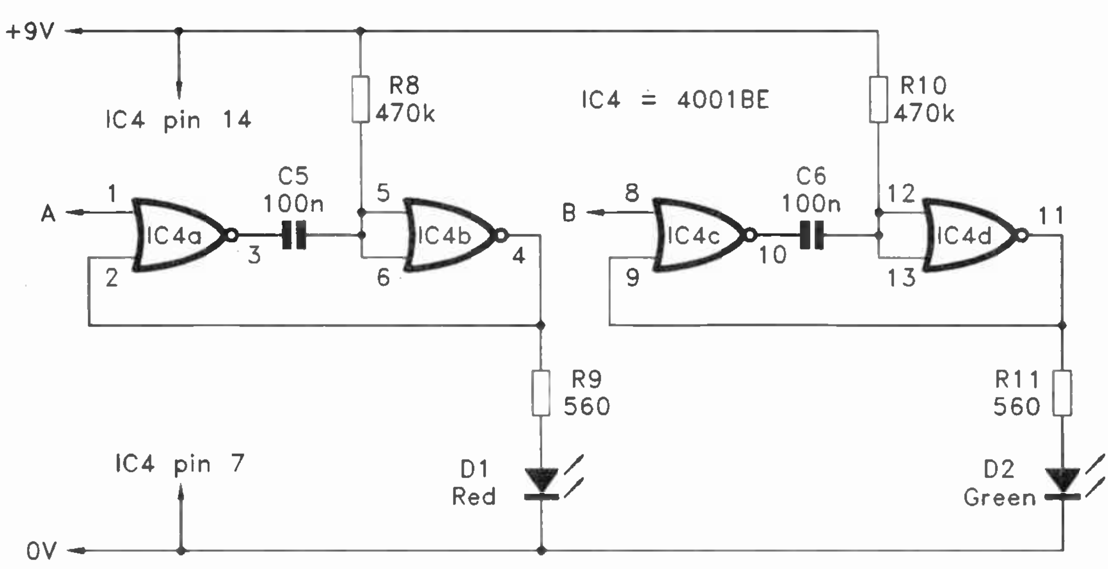

Fig.2.5 The optional LED indicator stages

metronome. Taking Figure 2.3 first, this shows the circuit diagram for the oscillator and divide by 'N' stages. The oscillator stage is based on a low power 555, and is much the same as the one used in the metronome project described previously. The only difference is that R2 has a slightly higher value which gives a slightly longer output pulse duration, but this is not really of any practical significance. The monostables determine the length of the pulses fed to the loudspeaker.

The divide by 'N' stage is based on a CMOS 4017BE, which is a one of ten decoder. This has the standard "carry out" output, plus ten other outputs. It is some of these extra outputs that are utilized in this circuit. The ten extra outputs are numbered from "0" to "9", and initially output "0" is high, and the other nine are low. The first clock cycle takes output "0" low, but sends output "1" high. On the next clock cycle output "1" goes low and output "2" goes high. This process continues on subsequent clock cycles, until output "9" goes high on the ninth clock cycle. On the tenth clock cycle output "9" goes low, output "0" goes high, and the circuit is back in its original state. This counting process continues indefinitely.

S1 connects the reset input of IC2 to output "2", "3", "4", "6", or "8". Suppose that output "3" is selected. The counter operates normally on the first and second clock cycles, but on the third clock pulse output "3" goes high and immediately resets the counter back to zero. The count then continues normally again for two more clock pulses, but on the third pulse output "3" goes high again, and resets the counter back to zero again. The circuit therefore provides a divide by three action. The monostable which provides the longer pulses is driven from output "1", and will therefore be triggered on every third clock pulse. This obviously accents every third beat. By using the other outputs it is possible to accent every second, fourth, sixth, or eighth beat. S3 enables the accented beat facility to be switched off.

Figure 2.4 shows the circuit diagram for the monostable, mixer, and output amplifier stages. The monostables are based on the same configuration, and are each formed from a pair of CMOS NOR gates. The output pulse duration is approximately equal to 0.7CR seconds, or a little over one millisecond for the monostable based on IC3c and IC3d. IC3a and IC3b produce the shorter pulses (about 0.15 milliseconds). The mixer is a basic passive type (R5 and R7), and the output amplifier is a common emitter stage (TR1).

The optional LED indicator circuit is provided in Figure 2.5. This utilizes two monostables of the type used to produce the pulses that drive the loudspeaker. However, in this case they both provide the same pulse duration, and the pulse duration is much longer at approximately 33 milliseconds. This is long enough to give a bright flash from the two LEDs. D1 flashes on every beat, but D2 only flashes on accented beats. The unit therefore provides both visual and audible indications of accented beats. Due to the use of CMOS devices the average current consumption of the circuit is very low, and a PP3 battery is more than adequate as the power source.

Most of the notes on constructing the previous project apply equally to this one, and they will not be repeated here. In addition, note that the 4017BE and 4001BE are CMOS devices, and that the standard anti-static handling precautions should be observed when dealing with these devices. S1 is a 6 way 2 pole rotary switch having an adjustable end-stop. It is set for 5 way operation, and one pole is left unused. It should be a break before make type, since a make before break switch would place a short circuit across two outputs of IC2 each time it was adjusted.

Components for Accented Metronome (Figs 2.3, 2.4 & 2.5)

Resistors (all 0.25 watt 5% carbon film)

R1	220k
R2	4k7
R3	100k
R4	10k
R5	1k
R6	15k
R7	1k
R8	470k
R9	560R

R10	470k
R11	560R

Potentiometer

VR1	2M2 lin carbon (see text)

Capacitors

C1	220μ 10V elect
C2	1μ polyester
C3	22n polyester
C4	100n polyester
C5	100n polyester
C6	100n polyester

Semiconductors

IC1	TLC555C or similar
IC2	4017BE
IC3	4001BE
IC4	4001BE
TR1	BC337
D1	High efficiency red LED
D2	High efficiency green LED

Miscellaneous

S1	6 way 2 pole rotary with end-stop, break before make
S2	SPST min toggle
S3	SPST min toggle
LS1	8 ohm moving coil loudspeaker, about 76mm diameter
B1	9 volt (PP3 size)
	Small case
	Circuit board
	Control knob (2 off)
	8 pin DIL IC holder
	14 pin DIL IC holder (2 off)
	16 pin DIL IC holder
	Battery connector
	Wire, solder, etc.

Pseudo Stereo Unit

There are several common methods of generating a pseudo stereo output from a monophonic input signal. These vary greatly in complexity and effectiveness. The most simple method is to feed the unprocessed signal to one channel, and an inverted version of this signal to the other channel. Feeding a monophonic signal to both channels of a stereo system normally places all the sounds at the centre of the sound stage. Inverting one channel prevents a proper stereo effect from being obtained, since it is necessary to have in-phase signals in order to generate a vivid central stereo image. The problem with this method is that it provides identical signals at the sides of the sound stage, with no central image. This gives a rather unconvincing stereo effect, with a "hole" in the middle of the sound stage. In fact it is really just a different form of monophonic reproduction, rather than something most people would accept as a form of stereo signal.

I have always found results to be much better using a slightly more sophisticated version of the phase inversion technique. Rather than simply having the two channels out-of-phase, greatly improved results can be obtained using a frequency selective phase shift circuit in one channel. The basic idea is to have the two channels in-phase at low frequencies. As the input frequency is increased, the phase relationship gradually changes, eventually taking the two channels out-of-phase. At still higher frequencies the two signals are gradually brought back in-phase once again. In fact the system can use a multiple phase shifter circuit so that the two channels repeatedly slip in and out of phase as the input frequency is increased. However, good results can be obtained using a circuit based on just a couple of phase shifters.

The main point of this system is that it gives a combination of in-phase signals to provide a strong central stereo image, and out-of-phase signals to spread the signal right across the sound stage. It therefore gives better results than a simple phase inversion system, with the "hole in the middle" problem being avoided. It is advisable to use a system that has the signals in-phase at low frequencies as this gives a better bass response. With the signals out-of-phase at low frequencies there tends to be cancelling of the bass signals, with the system

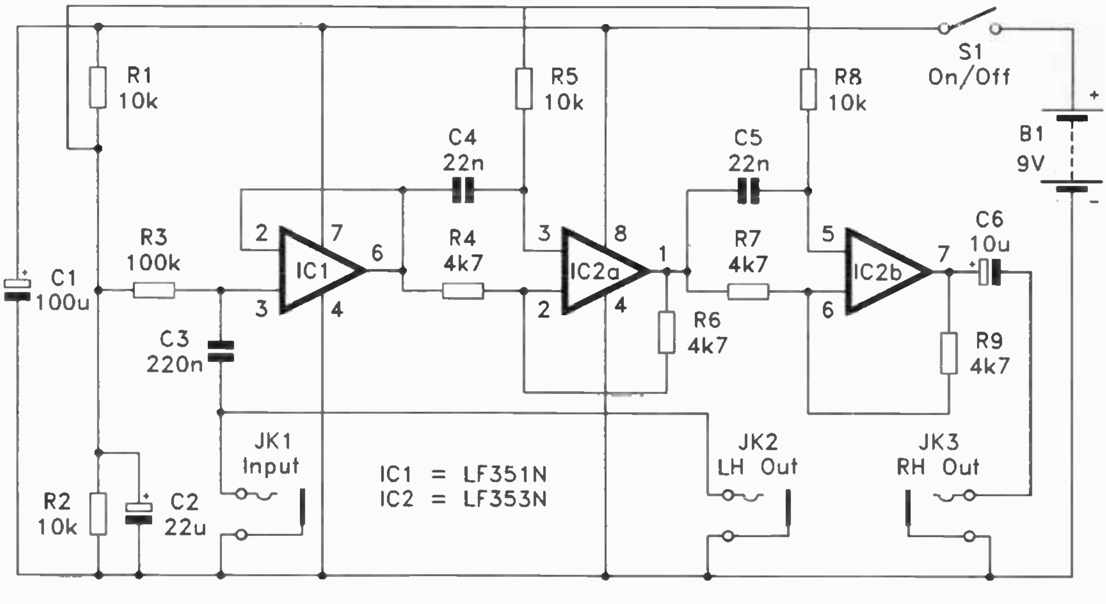

Fig.2.6 The pseudo-stereo unit circuit diagram

having an apparent lack of bass response.

The Circuit

Figure 2.6 shows the circuit diagram for the pseudo stereo unit. IC1 merely acts as an input buffer stage, and it is IC2 that acts as the basis of the two phase shifters. These use the standard configuration that is much used in phaser effects units. In this case there is no need to sweep the operating frequency of the phase shifters, so no voltage controlled resistances are required.

The two phase shifters are identical, so we will only consider the operation of the one based on IC2a. At low frequencies C4 has an impedance which is extremely high in relation to the resistance of R5, and C4 consequently has no significant effect on the circuit. IC2a therefore operates as a straightforward inverting amplifier having unity voltage gain. At high frequencies C4 has a very low impedance, and effectively couples the input signal direct to the non-inverting input of IC2a. The circuit then operates as a non-inverting amplifier having unity voltage gain. At intermediate frequencies the circuit operates somewhere between these two extremes. The factor which makes this type of circuit so useful is that it operates in a combination of the inverting and non-inverting modes at these intermediate frequencies, and it provides a phase shift that varies from 180 degrees at low frequencies, to 0 degrees at high frequencies.

Taking the overall effect of the two phase shifters, at low frequencies the phase inversion through the first phase shifter is cancelled out by the phase inversion through the second phase shifter. This gives no overall phase shift through the circuit, and the two stereo channels are therefore in-phase. At middle frequencies there is a phase shift of about 90 degrees through each phase shifter, giving an overall shift of about 180 degrees. This produces anti-phase output signals, and takes middle frequency signals to the sides of the sound stage. An overall phase shift of exactly 180 degrees is produced at approximately 1kHz incidentally. At high frequencies there is no significant phase shift through either of the shifters, bringing the two pseudo stereo channels back in-phase again.

Slightly improved results can be obtained by adding more phase shifters into the circuit. In order to maintain zero phase shift at low frequencies it is advisable to use pairs of phase shifters, and to avoid using odd numbers of them. Although the circuit is shown as being added in the right-hand channel, the effect is the same whichever channel it is used to process. The important thing is that one channel should be processed and the other should be left unaltered, so that relative phase shifts are produced. Phase shifting both channels would result in them being in-phase at all frequencies, which would be the same as not processing either channel.

A basic circuit having one pair of phase shifters has a current consumption of about 4 milliamps. A PP3 size battery is adequate to supply this.

Components for Pseudo Stereo Unit (Fig.2.6)

Resistors (all 0.25 watt 5% carbon film)

R1	10k
R2	10k
R3	100k
R4	4k7
R5	10k
R6	4k7
R7	4k7
R8	10k
R9	4k7

Capacitors

C1	100µ 10V elect
C2	22µ 16V elect
C3	220n polyester
C4	22n polyester
C5	22n polyester
C6	10µ 25V elect

Semiconductors

IC1	LF351N
IC2	LF353N

Miscellaneous

S1	SPST min toggle
B1	9 volt (PP3 size)
JK1, 2, 3	6.35mm jack socket (3 off)
	Case
	Circuit board
	Battery connector
	Wire, solder, etc.

Audio Mixers

Feeding several instruments into one amplifier, tape recorder, etc., with its one audio input socket, is a common problem in the world of electronic music. The solution to the problem is an audio mixer. The term "mixer" tends to conjure up images of large mixing desks of the type used in professional studios. However, for many purposes, particularly in an electronic music context, a very basic mixer is often all that is really needed. The mixer described here is a basic three input monophonic type, but it is easily modified to provide more inputs, and (or) for stereo operation. It can also have pre-amplifiers for low output guitar pick-ups and microphones added, so as to further enhance the versatility of the basic mixer.

The Circuit

The circuit diagram for the simple three input mixer appears in Figure 2.7. It is basically just a conventional summing mode mixer circuit. IC1 operates in the inverting amplifier mode, but it has three input resistors, one for each input signal. The output of IC1 therefore has to balance the sum of the input voltages, and this gives the required mixing. A virtual earth is formed at the inverting input of IC1, which gives almost complete isolation between the three inputs. This is important, because there would otherwise be at least a certain amount of interaction between the three "fader" potentiometers (VR1 to VR3). The virtual earth gives such good isolation between the inputs that adjustment of one "fader" potentiometer has no significant effect on the other two.

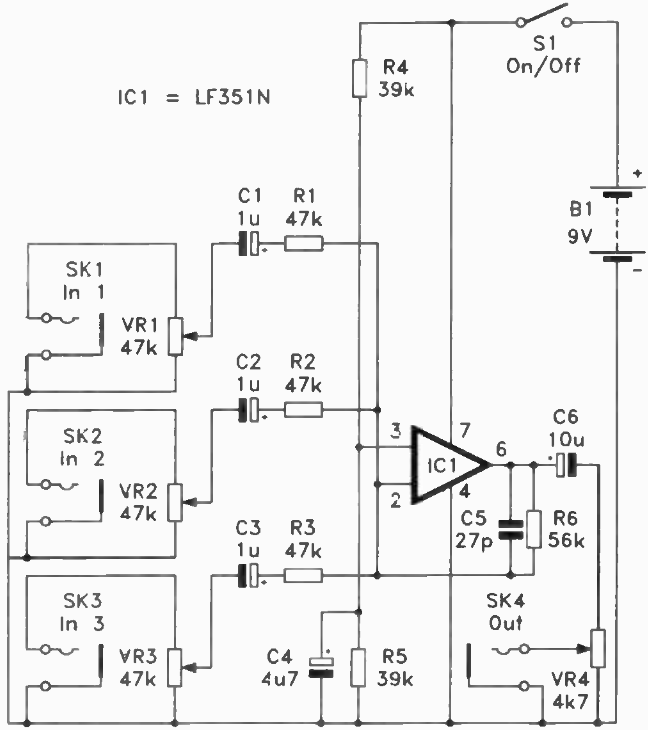

Fig.2.7 The circuit diagram for the basic three input mixer

C5 provides some high frequency roll-off to the circuit, and this helps to avoid problems with r.f. breakthrough. C5 also aids good stability. VR4 is the output "fader", and it simply enables the output level to be varied. The importance of this control (or lack of it) depends on the type of system in which the mixer will be used, and the precise manner in which it will be used. However, it is as well to include the output level control as it often proves to be useful.

With all the "fader" controls set at maximum gain the circuit has slightly more than unity voltage gain from each input to the output. This should be satisfactory for most purposes, but if necessary the gain can be boosted slightly by increasing the value of R6 to 100k or so. The current consumption of the circuit is only about 2 milliamps.

If more inputs are required it is just a matter of adding an extra input socket, 47k "fader" potentiometer, 1μ d.c. blocking capacitor, and 47k input resistor per additional input. This raises the inevitable question, what is the maximum number of inputs that can be used? This is very much a "how long is a piece of string?" type question. There are two main problems when using this type of circuit with a large number of inputs. One is simply that this inevitably results in a large amount of wiring at the input. This wiring tends to pick up mains "hum" and other electrical noise. Using a metal case to provide screening from the outside world helps to minimise this problem, but the larger the amount of input wiring, the more difficult it becomes to obtain an insignificant level of noise pick up.

The second problem is also one of noise, and it is a more serious problem. The noise in this case is the "hiss'" type noise generated by IC1. The more inputs that are added, the higher the effective voltage gain of IC1, and the higher the noise level at its output. In some cases this might not matter, since the noise levels on the input signals could be too high for the noise generated by IC1 to be of any real consequence. On the other hand, if the circuit has ten or more inputs, and the input signals are all very low noise types, the noise generated by IC1 could account for the majority of the noise on the output signal. With up to about eight to ten inputs there should be no major problem with the noise level of the circuit. With more than about ten outputs it would be advisable to use a higher quality operational amplifier for IC1, such as an NE5534AN.

Many electronic keyboard instruments and MIDI modules now have stereo outputs. One way of adapting the audio mixer featured here for stereo operation is to build two mixer boards, and use one in each stereo channel. For a stereo unit the four fader potentiometers should be dual gang types, with one gang of each potentiometer being used in one stereo channel, and the other gang being used in the opposite channel.

If slider potentiometers are used there is an alternative method available. This is to use single gang potentiometers, but with the pairs of potentiometers for each input mounted side-by-side with a minimal gap between them. It is then easy to operate the two potentiometers in unison if straightforward fading is all that is needed. A skilful operator can also operate the two controls independently, if changes in the channel balance are required. Probably most users will not wish to modify the channel balances, and using dual gang potentiometers is then the better way of handling things.

Construction of the mixer does not present any major difficulties, but try to keep the wiring at the input of the unit as short and direct as reasonably possible. If a few long wires can not be avoided, it is advisable to use screened cable even if the circuit has overall screening provided by a metal case. Although standard 6.35 millimetre jack sockets are specified for SK1 to SK4 you may prefer to use a different type of socket for some or all of these, if this would fit in better with your music system. For example, it might be better to use a phono socket for SK4, if this matches the sockets on the amplifier or tape recorder used with the mixer.

Components for Mixer (Monophonic) (Fig.2.7)

Resistors (all 0.25 watt 5% carbon film)

R1	47k
R2	47k
R3	47k
R4	39k
R5	39k
R6	56k

Potentiometers

VR1	47k log carbon
VR2	47k log carbon
VR3	47k log carbon
VR4	4k7 log carbon

Capacitors

C1	1µ 50V elect
C2	1µ 50V elect
C3	1µ 50V elect
C4	4µ7 50V elect
C5	27p ceramic plate
C6	10µ 25V elect

Semiconductor

IC1	LF351N

Miscellaneous

SK1 to SK4	6.35mm jack socket (4 off, see text)
S1	SPST min toggle
B1	9 volt (PP3 size)
	Case
	Circuit board
	Battery connector
	Control knob (4 off)
	8 pin DIL IC holder
	Wire, solder, etc.

Microphone Mixer

A very basic mixer is perfectly adequate when all the signal sources provide a high level signal. For example, when mixing the outputs from synthesisers, samplers, and high output guitar pick-ups. Problems arise when trying to mix high and low level signals, such as the output from a synthesiser with the signal from a microphone. To some extent the fader controls can be used to compensate for differences in the input levels, but there is a limit to their effectiveness in this role. Basically all you can do is reduce the level of the stronger signals to match that of the weakest input signal. If the lowest input level is less than a few hundred millivolts peak-to-peak, the output level from the mixer will probably not be strong enough to drive the amplifier, tape recorder, etc.

The output level from a low impedance dynamic microphone is very much lower than the signal level from a synthesiser, sampler, or other high level signal source. It is even lower

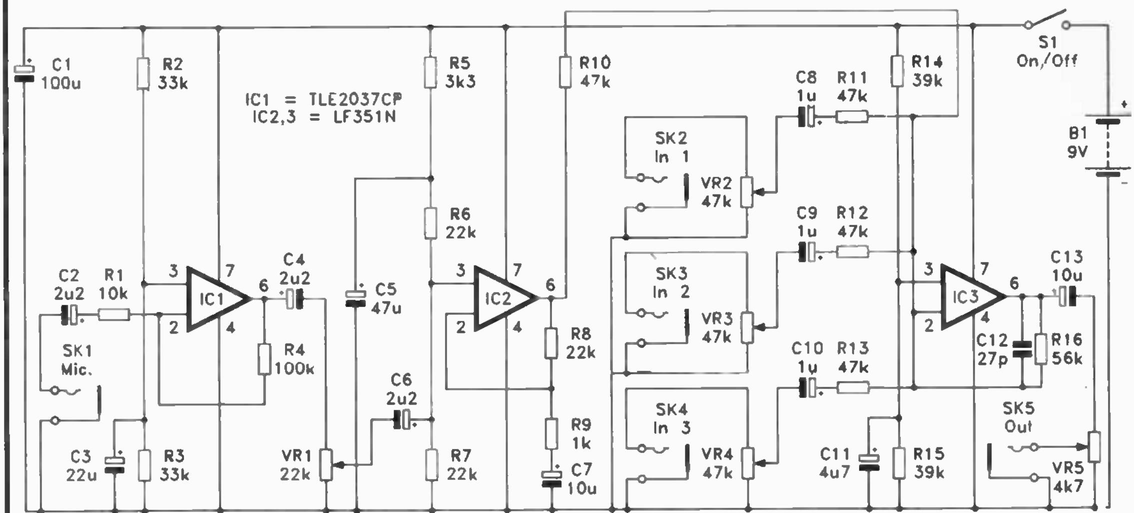

Fig.2.8 The microphone mixer circuit

than the output level from a low output guitar pick-up. In fact the output from a low impedance microphone is often under one millivolt r.m.s., and the output level from a high impedance type is usually no more than a few millivolts r.m.s. The signal from a high level source such as a synthesiser is typically about one hundred to one thousand times higher in amplitude. A high gain preamplifier is therefore needed in order to boost the signal from the microphone to a level that is comparable to the other input signals.

Figure 2.8 shows the circuit diagram for a four input mixer which has three high level inputs and one microphone input. A two stage microphone preamplifier is used, and this has IC1 as a very low noise input stage. The TLE2037CP specified for IC1 is an ultra low noise and distortion bipolar operational amplifier which is specifically designed for use in high quality preamplifier circuits. The circuit will work using a Bifet operational amplifier for IC1, such as a LF351N or a TL081CP, or even a standard 741C bipolar type, but the background noise level will be about ten times higher. A very high quality bipolar operational amplifier such as the TLE2037CP or NE5534A is a much better choice for a critical application such as this.

IC1 is used as a standard inverting mode amplifier. R1 and R4 are the negative feedback network for IC1, and these set the closed loop voltage gain and input impedance at 20dB (10 times) and 10k respectively. This input impedance should give good results with any high impedance dynamic microphone, or any microphone which has similar output characteristics (an electret type having a built-in step-up transformer for example). If the unit is to be used with a low impedance dynamic microphone, or a transformerless electret type, reduce the value of R1 to 1k and increase the value of C2 to 22μ. This will give the necessary boost in gain and reduction in input impedance.

C4 couples the output of IC1 to the microphone fader potentiometer, VR1. From here the signal is coupled to the second stage of the preamplifier, which is an operational amplifier non-inverting mode circuit based on IC2. Noise performance is less important for this stage as it is handling a much higher signal level than the input stage. A Bifet operational amplifier such as the LF351N is therefore perfectly adequate for IC2. R8 and R9 are the negative feedback resistors for this stage, and they provide a closed loop voltage gain of just over 26dB (20 times). This gives the preamplifier a maximum overall gain in excess of 46dB (200 times), which should be sufficient to give good results with any normal high impedance microphone. The total voltage gain is over 66dB (2000 times) if the circuit is modified for operation with a low impedance microphone. This should give an adequate output level with any normal low impedance microphone. The output of IC2 is direct coupled to mixer input resistor R10.

Components for Microphone Mixer (Fig.2.8)

Resistors (all 0.25 watt 5% carbon film)

R1	10k
R2	33k
R3	33k
R4	100k
R5	3k3
R6	22k
R7	22k
R8	22k
R9	1k
R10	47k
R11	47k
R12	47k
R13	47k
R14	39k
R15	39k
R16	56k

Potentiometers

VR1	22k log
VR2	47k log
VR3	47k log
VR4	47k log
VR5	4k7 log

Capacitors

C1	100μ 25V radial elect
C2	2μ2 50V radial elect
C3	22μ 16V radial elect
C4	2μ2 50V radial elect
C5	47μ 25V radial elect
C6	2μ2 50V radial elect
C7	10μ 25V radial elect
C8	1μ 50V radial elect
C9	1μ 50V radial elect
C10	1μ 50V radial elect
C11	4μ7 50V radial elect
C12	27p ceramic plate
C13	10μ 25V radial elect

Semiconductors

IC1	TLE2037CP or NE5534A
IC2	LF351N
IC3	LF351N

Miscellaneous

SK1 to SK5	Standard 6.35mm jack socket (5 off)
S1	SPST min toggle
B1	9 volt (PP3 size)
	Case
	Circuit board
	Control knob (5 off)
	8 pin DIL IC holder (3 off)
	Wire, solder, etc.

Guitar Mixer

Many guitar pick-ups provide an output level that is comparable to the output from a synthesiser or other high level source. However, some pick-ups, particularly older and the less expensive types, have much lower output levels. In some cases the output signal is only about 100 millivolts peak-to-peak on the initial transient, which is less than a tenth of the output level from most synthesisers. Figure 2.9 shows the circuit diagram for a four input mixer having three high level

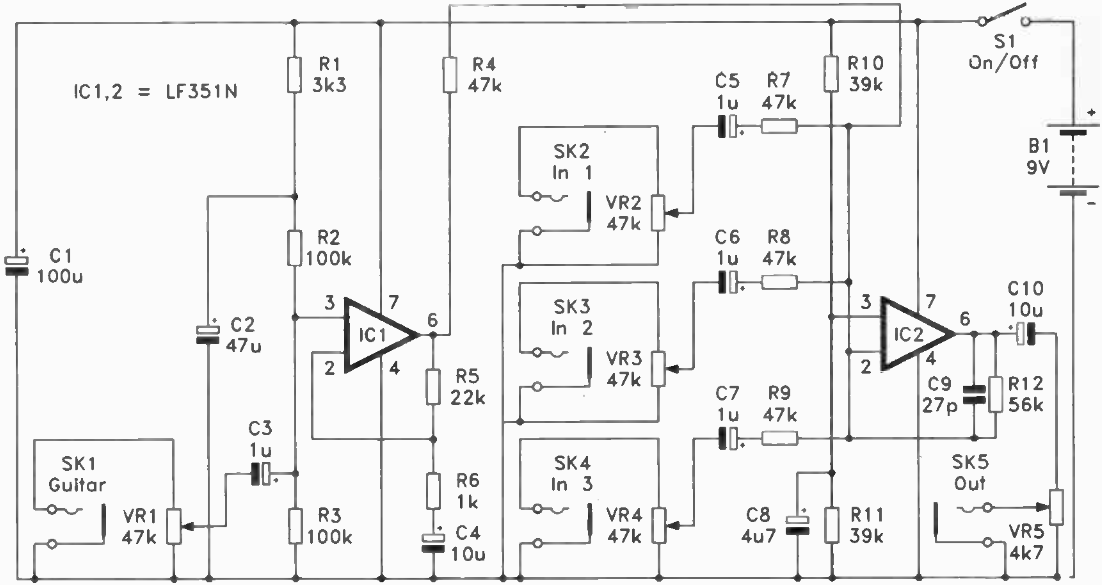

Fig.2.9 A four input mixer with built-in guitar preamplifier

inputs and one input for a low output guitar pick-up. The guitar's output signal is boosted by a simple preamplifier based on IC1. This has a voltage gain of about 23 times, which should be sufficient to give an adequate output signal with virtually any guitar pick-up. However, if necessary the closed loop voltage gain of the circuit can be boosted somewhat by increasing the value of R5 to about 47k.

Components for Guitar Mixer (Fig.2.9)

Resistors (all 0.25 watt 5% carbon film)

R1	3k3
R2	100k
R3	100k
R4	47k
R5	22k
R6	1k
R7	47k
R8	47k
R9	47k
R10	39k
R11	39k
R12	56k

Potentiometers

VR1 to VR4	47k log carbon (4 off)
VR5	4k7 log carbon

Capacitors

C1	100μ 10V elect
C2	47μ 16V elect
C3	1μ 50V elect
C4	10μ 25V elect
C5	1μ 50V elect
C6	1μ 50V elect
C7	1μ 50V elect
C8	4μ7 50V elect
C9	27p ceramic plate
C10	10μ 25V elect

Semiconductors

IC1	LF351N
IC2	LF351N

Miscellaneous

SK1 to SK5	Standard 6.35mm jack socket (5 off)
S1	SPST min toggle
B1	9 volt (PP3 size)
	Case
	Circuit board
	Control knob (5 off)
	8 pin DIL IC holder (2 off)
	Wire, solder, etc.

Finally

The basic mixer circuit is not restricted to use with one preamplifier. If necessary, several preamplifiers (microphone, guitar, or both types of preamplifier) can be added ahead of the mixer. Provided you have a reasonable understanding of what you are doing, it is therefore possible to put together something like a six input mixer having a microphone input, two guitar inputs, and three high level inputs. However, bear in mind that more preamplifiers also means more noise on the output signal.

Chapter 3

MIDI PROJECTS

MIDI (musical instruments digital interface) provides a means for one electronic musical instrument to communicate with another. In fact it is not limited to use with two instruments, and it is probable that most systems are based on a sequencer of some kind plus several instruments. It can also be used with items of equipment such as digital effects units and audio mixers. Unlike many earlier systems of interfacing, including the once popular gate/CV type, it provides a sophisticated link between the master device and the slave units. Multi-channel polyphonic sequencing can be accommodated, as can sophisticated control of the dynamics, pitch bending, etc. Although MIDI was not exactly an overnight success when it was launched, some twenty years or so later it has become central to electronic music making.

This chapter features several projects which, in a variety of ways, enable MIDI to be more fully exploited. You do not need to be a MIDI expert in order to build and use these projects, since any essential technical information is provided. However, anyone using MIDI equipment needs to be familiar with the basic working of the system. There is insufficient space available here for a detailed discussion of the way in which MIDI operates and is used. Anyone needing general information on using a MIDI system should consult a book on the subject, such as "A Beginners Guide To MIDI" (BP331) from the same publisher and author as this publication.

Simple MIDI Tester

Although connecting up and using a MIDI system is very simple in theory, in practice there are inevitably a few problems from time to time. In some cases this is due to problems with cables. These tend to be the "weak link" in any electronic system that consists of several units wired together. Leads tend to get trodden on, tripped over, and generally abused. This does not do a great deal for their reliability, even if they start out as top quality leads! Another problem is that modern instruments are very sophisticated, and have MIDI implementations which reflect this sophistication. Getting everything set up and working correctly can be difficult, and it is not made any easier by the method of control used on most modern instruments. It can take an awful lot of button pushing to set even a few basic MIDI functions such as the reception and transmission channels.

This very basic tester simply lets you know whether or not the MIDI signal is getting to the point in the system where it is fitted. The unit provides audible and visual indications if an input signal is received. The main use of the device is for checking cables. If you know that the master unit is working properly and sending a signal into the cable, but no signal is received at the other end of the cable, then there is clearly a broken lead or connection somewhere in the cable. If the signal is passing through the cable satisfactorily, then the slave unit that is not responding properly might be faulty.

In most cases the problem is not due to a fault in the slave unit, but is simply due to it being incorrectly set up. A common cause of slave units failing to respond properly is that they have been inadvertently set to the wrong MIDI channel. I was once asked to check over a MIDI unit that was failing to respond to any messages received. The problem was simply that someone has accidentally leant on one of the control buttons and had switched off the MIDI sockets! Many MIDI units have built-in MIDI filtering which enables certain types of message to be ignored. A MIDI controlled feature might fail to operate because it has not been activated, or it has been accidentally switched off. Unless a MIDI unit shows obvious signs of being faulty it is advisable to thoroughly check all the settings are correct before sending it off for repair.

Opto-Isolation

It is a requirement of the MIDI standard that all inputs should be equipped with opto-isolators, so that there are no direct electrical connections between MIDI units via the MIDI cables. This helps avoid problems with earth loops, and digital noise being coupled into the audio signal paths. MIDI is a form of serial interface, and it operates at the relatively high baud rate of 31,250 baud. Ordinary opto-isolators such as the TIL111 can not operate at baud rates as high as this, and tend to struggle at about one-tenth of that rate. In order to successfully opto-couple a signal at such a high baud rate it is necessary to either use some external circuitry to effectively speed up a "bog standard" opto-isolator, or to opt for a more up-market device.

The circuits in this chapter which require opto-isolation all use a 6N139, which is a high quality type that can easily handle a 31,250 baud signal. In fact it seems to be able to operate properly at baud rates at least ten times higher than this. Figure 3.1 shows the internal circuit and pinout arrangement for the 6N139. On the input side it has the usual infrared LED. On the output side there is a photodiode driving a

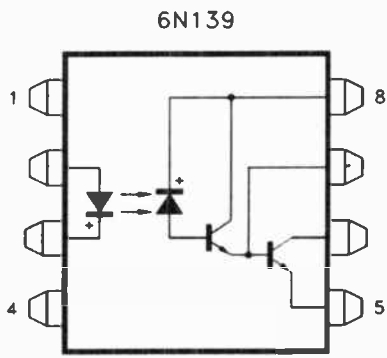

Fig.3.1 The internal circuit of the 6N139 opto-isolator

common emitter switch via an emitter follower buffer stage. This arrangement provides far higher efficiency than simply using a photo-transistor at the output of the device. Of more importance in the current application, it also provides a much quicker switching rate.

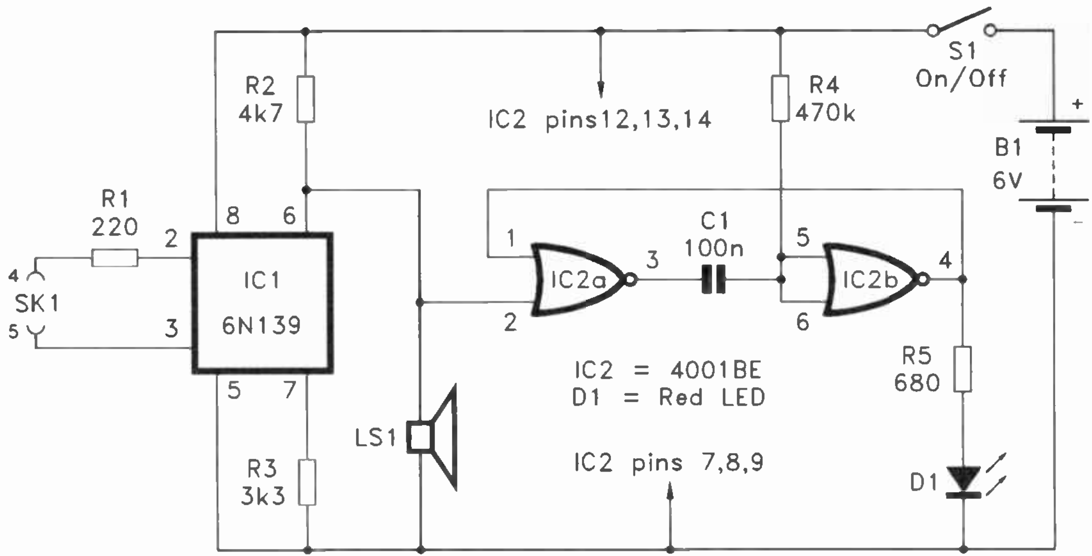

Fig.3.2 The circuit diagram for the simple MIDI tester

The Circuit

Figure 3.2 shows the circuit diagram for the simple MIDI tester. IC1 is the 6N139 opto-isolator, and this has R1 to provide part of the current limiting for its LED. The rest of the current limiting is provided by resistors in the MIDI output circuit that drives IC1. Incidentally, MIDI operates at a nominal current of 5 milliamps. On the output side of IC1, R2 and R3 are respectively the load resistor for the common emitter output transistor and the emitter follower stage.

LS1 is a ceramic resonator, and it provides the audible indication when a MIDI signal is received. It works well from the very limited drive current available from IC1. It still only provides a rather low volume output, but this is all that is needed in this application. Note that the circuit will not work if a moving coil loudspeaker (high or low impedance) is used for LS1. The visual indication is provided by a simple CMOS monostable which drives LED indicator D1. The LED could be driven direct from IC1, but the pulses from IC1 are likely to be so short and intermittent that the LED brightness would usually be very low. Each time IC2 is triggered by an input pulse it switches on D1 for about 33 milliseconds. Although this is not very long, it is sufficient to give a very obvious flash from D1.

The standby current consumption of the circuit is very low, and might be less than one microamp. A supply current of a few milliamps flows at times of high activity on the MIDI input. The 6 volt battery should therefore have an extremely long operating life.

Construction of this project offers little out of the ordinary, but bear in mind that IC2 is a CMOS device which requires the usual anti-static handling precautions. Figure 3.3 shows the correct method of connection for SK1. This shows SK1 viewed from the rear (i.e. looking at the pins to which the connections will be made). Incidentally, if you look carefully at the rear of SK1 you should find that it is marked with the pin numbers.

The tester connects to a standard MIDI (DIN type) cable in the usual way. D1 will flash if the unit detects any MIDI activity. You will also hear various "clicks" and "buzzes"

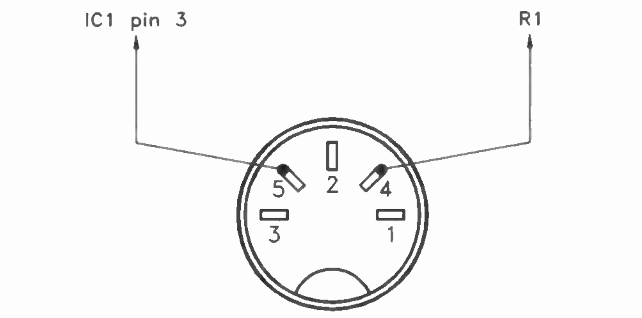

Fig.3.3 Connection details for SK1

from LS1. Although a single MIDI message usually consists of several very brief pulses, these pulses will only produce one perceivable "click" from LS1. If there is a lot of MIDI activity (such as when using pitch wheel modulation) there will be so many pulses that they will merge into a "buzzing" sound. The greater the MIDI activity, the higher the general pitch of the "buzzing" sound.

Components for Simple MIDI Tester (Fig.3.2)

Resistors (all 0.25 watt 5% carbon film)

R1	220R
R2	4k7
R3	3k3
R4	470k
R5	680R

Capacitor

C1	100n polyester

Semiconductors

IC1	6N139 opto-isolator
IC2	4001BE
D1	Red panel LED

Miscellaneous

S1	SPST min toggle
B1	6 volt (4 x AA or AAA cells in plastic holder)
LS1	Cased ceramic resonator
SK1	5 way 180 degree DIN docket
	Small case
	Circuit board
	Battery connector
	8 pin DIL IC holder
	14 pin DIL IC holder
	Wire, solder, etc.

MIDI Comparator

This project is another form of MIDI tester, but it is rather more sophisticated than the unit described previously. Instead of just indicating that a MIDI signal of some sort is present and correct, it flashes the LED indicator when a certain binary pattern is detected. This binary pattern is user selectable, and could be any part of a MIDI message, including a data byte. However, the normal way of using the unit would be to set it up to detect a particular MIDI message type on a certain channel. Devices being accidentally set to the wrong channel is a common cause of problems with MIDI systems. Therefore, it will often be the channel number rather than the message type that is primarily of interest. Either way, if the right type of message is being sent on the right channel, this unit will confirm the fact.

System Operation

The block diagram for the MIDI comparator appears in Figure 3.4. A device called a UART (universal asynchronous receiver/transmitter) is at the heart of the unit. A UART can provide a conversion from parallel to serial data, and (or) a

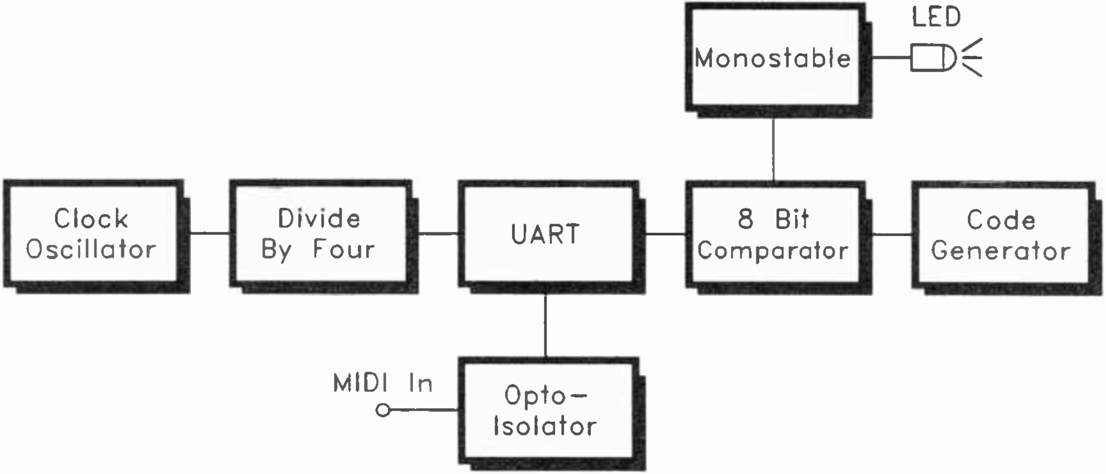

Fig.3.4 The MIDI comparator block diagram

conversion in the opposite direction. In this case it is a conversion from the serial MIDI signal to parallel data that is required. The MIDI signal is coupled to the input of the UART's receiver section via an opto-isolator. The baud rate of the UART is controlled by a clock signal, and this signal must be at sixteen times the required baud rate. In this case the baud rate is 31,250 baud, which means that a clock frequency of 500kHz is required (31,250 × 16 = 500,000Hz (500kHz)). The clock oscillator is a crystal type operating at 2MHz, but a 500kHz signal is derived from this via a divide by four circuit.

Each time a complete byte is received, the eight bits of new data are placed onto the parallel output bus of the UART. These eight bit codes are compared to the eight bit code set up by the user. The output of the eight bit comparator goes low when a match is found, and only when a match is found. The output of the comparator drives the input of a monostable, and the latter is triggered when the output of the comparator goes low. A LED indicator then flashes to indicate that the correct binary pattern has been detected.

The Circuit

The circuit diagram for the MIDI comparator is shown in Figures 3.5 and 3.6. Starting with Figure 3.5, this shows the circuit for the clock, input, and UART stages. The clock oscillator is a conventional crystal type based on TR1. There is no need to bother with trimmer capacitors to permit fine tuning of the output frequency. The raw accuracy of the oscillator is more than adequate for the current application. The divide by four action is provide by two stages of IC1, which is a CMOS seven stage binary counter.

IC2 is the opto-isolator, and it is used in the same circuit that was used in the simple MIDI tester unit described previously. In this case there is some additional circuitry (R6, R7, and SK2) which provides a MIDI THRU output. If required, the unit can therefore be used to monitor the output of the master unit connected to SK1, while passing on the signal to a slave unit connected to SK2.

IC3 is the UART, and this is an industry standard 6402, or any true equivalent. A UART can handle any normal word

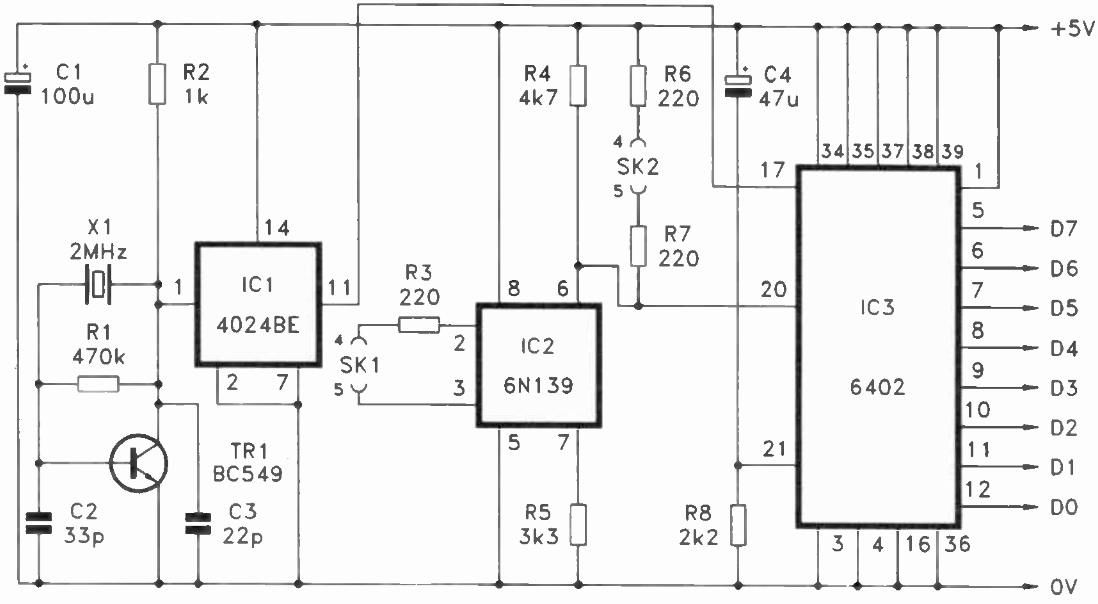

Fig.3.5 The MIDI comparator clock, input, and UART stages

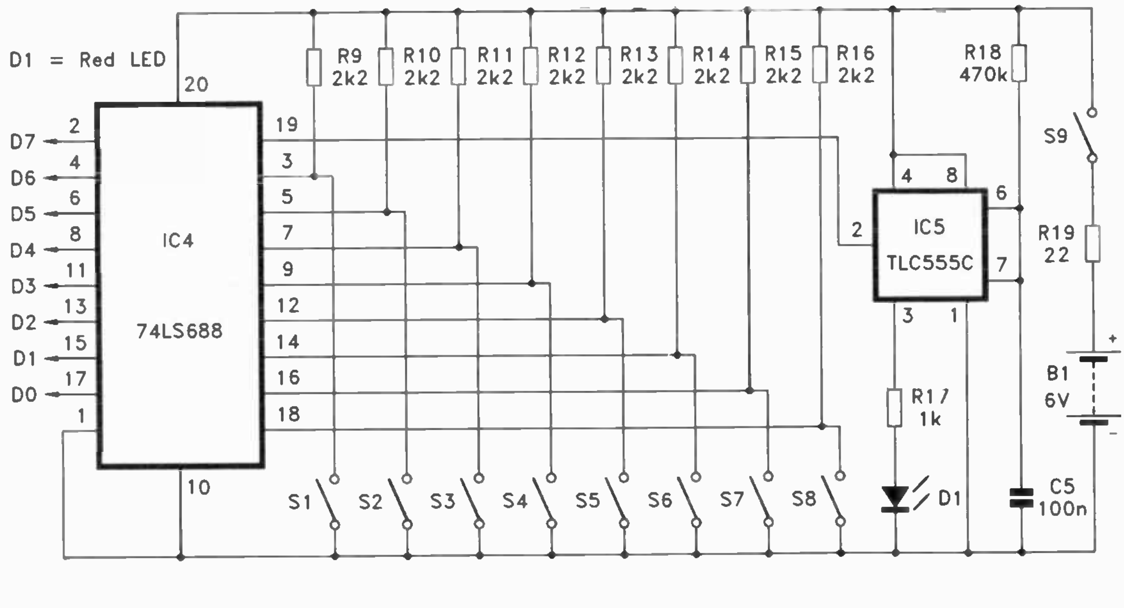

Fig.3.6 The MIDI comparator code detector and monostable stages

format, including the more unusual ones such as one start bit, five data bits, and one and a half stop bits! The MIDI word format is the much more common format of one start bit, eight data bits, one stop bit, and no parity checking. The required format is obtained by connecting pins 34 to 39 to the appropriate logic levels. C4 and R8 provide a reset pulse to IC3 at switch-on.

Turning now to Figure 3.6, IC4 is the 8 bit magnitude comparator. It has one set of inputs fed from the UART, and the other set fed from a basic 8 bit binary code generator. R9 to R16 pull the inputs of IC4 high, but closing the relevant switch will take an input to the low state. Any required eight bit code can therefore be set up using S1 to S8. I used eight individual switches, but a couple of "hex" switches could be used if preferred. In an application of this type it is probably as easy (or difficult) whether you work in binary or hex. It is just a matter of using whichever method suits you best. S1 to S8 could be a bank of eight DIP switches, but as you may wish to make frequent changes to their settings, it is probably best to use ordinary switches (miniature toggle, slider, etc.).

The output of IC4 drives a simple monostable based on IC5. This is a conventional 555 monostable which pulses LED indicator D1 for about 33 milliseconds each time the monostable is triggered. The circuit requires a supply potential of about 5 volts, and this is provided by a 6 volt battery and a dropper resistor (R18). The current consumption of the circuit is about 45 milliamps or so, which is largely the current consumed by IC4. Although this is quite high, each 6 volt battery pack can power the unit for many hours.

Construction of a circuit such as this tends to be a bit awkward, with a tangle of interconnections to contend with. The easy way is to use wire-wrapping techniques and to not worry too much about making everything too neat. Stripboard can also be used, and can give a reasonably neat finished assembly if due care is taken over the layout. A custom printed circuit board is ideal, but unless you are particularly good at designing and producing these I would recommend sticking to wire-wrapping or stripboard construction. Although the circuit is handling some high frequency signals, the component layout does not seem to be particularly critical.

Even if the finished assembly does not look very neat, provided everything connects together correctly it should work properly. Note that IC1 and IC3 are CMOS devices, and that they require the standard anti-static handling precautions.

In Use

In order to use this unit it is clearly necessary to understand a little about the way in which MIDI messages are coded. All MIDI messages start with a header byte that contains details of the message type, and where appropriate its MIDI channel number. Most MIDI messages have one or more data bytes after the header byte. The data bytes contain information such as which note to turn on, how much pitch bend to apply, etc. The most significant bit is set to 1 for header bytes, and to 0 for data bytes. When checking the MIDI output of an instrument it is usually header bytes that are being tested, so S1 should be switched "off".

The four least significant bits of a header byte carry the channel number. An important point to note here is that the binary value used in the message is in the range 0 to 15, but the convention is for MIDI channels to be numbered from 1 to 16. The MIDI channel is therefore one higher than the binary value used in the header byte (e.g. a value of 8 is used for a message on channel 9). Table 1 shows the switch positions needed to select each of the sixteen MIDI channels.

Bits 4, 5, and 6 indicate the message type, and there are eight different types of message. Table 2 shows the switch settings needed to select each of these.

Note that the system messages are directed to the entire system, and they do not use channelling. The four least significant bits of a system message byte indicate the type of message. Many of the sixteen available codes are not yet allocated to message types, or have been allocated to messages that are little used in practice. Table 3 shows the switch settings needed in order to select some of the more common system messages.

As an example, in order to detect note-on messages on channel 2, this set of switch settings would be used.

S1	*S2*	*S3*	*S4*	*S5*	*S6*	*S7*	*S8*
Off	On	On	Off	Off	Off	Off	On

Table 1

Channel	*S5*	*S6*	*S7*	*S8*
1	On	On	On	On
2	On	On	On	Off
3	On	On	Off	On
4	On	On	Off	Off
5	On	Off	On	On
6	On	Off	On	Off
7	On	Off	Off	On
8	On	Off	Off	Off
9	Off	On	On	On
10	Off	On	On	Off
11	Off	On	Off	On
12	Off	On	Off	Off
13	Off	Off	On	On
14	Off	Off	On	Off
15	Off	Off	Off	On
16	Off	Off	Off	Off

Table 2

Message Type	*S2*	*S3*	*S4*
Note Off	On	On	On
Note On	On	On	Off
Poly Key Pressure	On	Off	On
Control Change	On	Off	Off
Program Change	Off	On	On
Overall Key Pressure	Off	On	Off
Pitch Wheel Change	Off	Off	On
System Message	Off	Off	Off

Table 3

Message Type	*S5*	*S6*	*S7*	*S8*
Start Sys. Excl.	On	On	On	On
Song Position	On	On	Off	On
End. Sys. Excl.	On	Off	Off	Off
Clock	Off	On	On	On
Start	Off	On	Off	On
Continue	Off	On	Off	Off
Stop	Off	Off	On	On
Active Sensing	Off	Off	Off	On

Components for MIDI Comparator (Figs 3.5 & 3.6)

Resistors (all 0.25 watt 5% carbon film)

R1	470k
R2	1k
R3	220R
R4	4k7
R5	3k3
R6	220R
R7	220R
R8 to R16	2k2 (9 off)
R17	1k
R18	470k
R19	22R

Capacitors

C1	100μ 10V elect
C2	33p ceramic plate
C3	22p ceramic plate
C4	47μ 16V elect
C5	100n polyester

Semiconductors

IC1	4024BE
IC2	6N139
IC3	6402
IC4	74LS688

IC5	TLC555C or similar
D1	Red panel LED
TR1	BC549

Miscellaneous

S1 to S9	SPST min toggle (9 off)
X1	2MHz wire-ended crystal
B1	6 volt (4 x AA size cells in holder)
SK1	5 way 180 degree DIN socket
SK2	5 way 180 degree DIN socket
	Case
	Circuit board
	Battery connector (PP3 type)
	8 pin DIL IC holder (2 off)
	14 pin DIL IC holder
	20 pin DIL IC holder
	40 pin DIL IC holder
	Wire, solder, etc.

THRU Box

The normal method of connecting a MIDI system together is the "chain" system. With this method the THRU output on the first slave unit connects to the IN socket of the second slave unit. The THRU output of the second slave unit connects to the IN socket of the third slave device, and so on. Figure 3.7 shows an example chain system having a keyboard controller and four slave synthesisers. This method of connection is not favoured by all MIDI users due to potential problems with so-called MIDI delays. On each journey from an IN socket to a THRU type the signal passes through an opto-isolator. There is inevitably a certain amount of waveform distortion through each opto-isolator, and if the signal passes through several isolators, this distortion can build up to significant proportions.

In practice it is unlikely that anyone would connect enough devices into a chain system to produce corruption of any messages. Some MIDI users prefer not to take the chance though, and instead use the "star" method of connection. There is sometimes no choice but to opt for the star system,

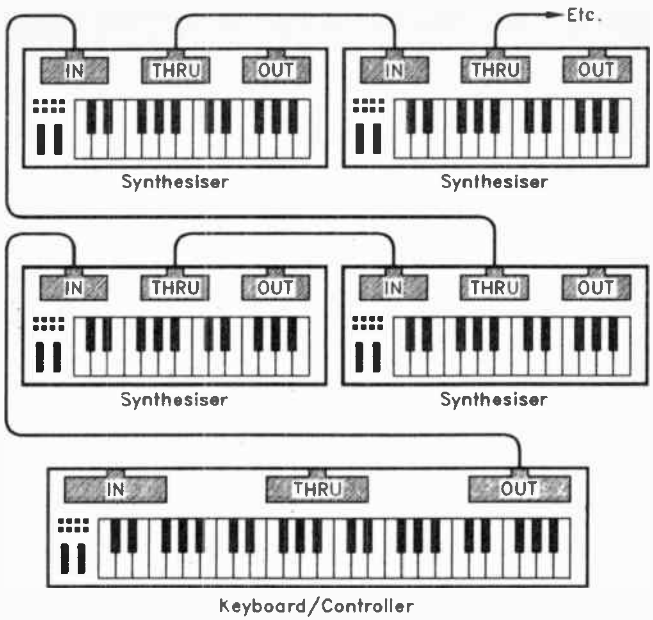

Fig.3.7 The "chain" method of connection

because not all MIDI units have a THRU socket. It is not a requirement of the MIDI standard that this facility should be included on all equipment that has an IN socket. Fortunately, a THRU port seems to be standard equipment for all modern MIDI instruments, but there are plenty of instruments from some years ago which lack this facility.

Unless the MIDI master unit has several outputs, which is unlikely, the star system can only be implemented with the aid of a THRU box. Figure 3.8 shows the star system equivalent of the chain system of Figure 3.7. As will be apparent from Figure 3.8, a THRU box has one input socket and several THRU output sockets. The output signal from the master

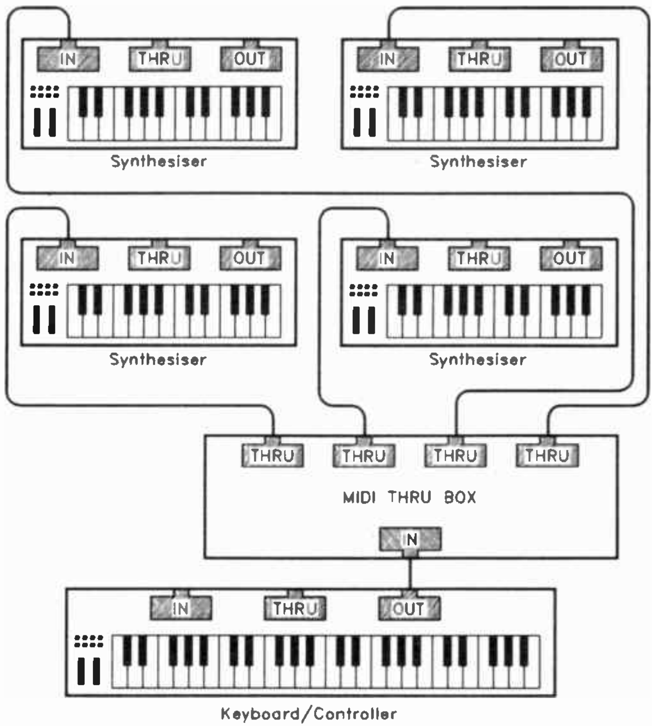

Fig.3.8 The "star" method of connection

unit feeds the input of the THRU box, and the THRU outputs connect to the various slave units. The THRU box must therefore have one output per slave unit.

The Circuit

Figure 3.9 shows the circuit diagram for the THRU box. This is basically just a 6N139 opto-isolator driving four THRU sockets (SK2 to SK5). The 6N139 has a maximum output

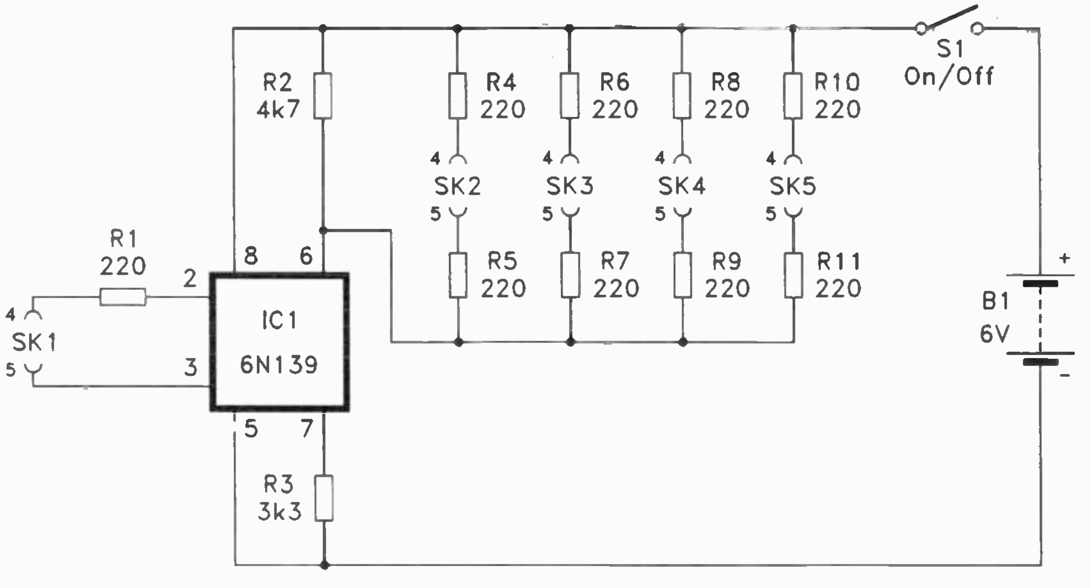

Fig.3.9 The THRU box circuit diagram

current rating of 60 milliamps, so it is well able to drive four THRU outputs. In fact it can drive up to about a dozen THRU outputs. If you need more than four outputs it is just a matter of adding further sockets, complete with pairs of 200R current limiting resistors.

The current consumption of the circuit is negligible under standby conditions. It obviously increases substantially when the unit is passing MIDI data. The greater the MIDI activity, and the larger the number of outputs in use, the higher the current consumption. The average current consumption is never likely to exceed about 25 milliamps though, even if you use about ten or twelve outputs. The four HP7 (AA) size batteries therefore have a very long operating life.

Construction of this project is very easy indeed, and even those of limited experience should have no difficulty in building it.

Components for MIDI THRU box (Fig.3.9)

Resistors (all 0.25 watt 5% carbon film)

R1, R4 to R11	220R (9 off)
R2	4k7
R3	3k3

Semiconductor

IC1	6N139

Miscellaneous

S1	SPST min toggle
B1	6 volt (4 x HP7 size cells in plastic holder)
SK1 to SK5	5 way 180 degree DIN socket (5 off)
	Case
	Circuit board
	Battery connector (PP3 type)
	8 pin DIL IC holder
	Wire, solder, etc.

MIDI Noise Gates

A noise gate is a form of electronic switch. It is used in an audio signal path, and it cuts the signal path if the input signal falls below a certain level. A noise gate provides a very basic but effective form of single-ended noise reduction. When the input signal drops to a very low level it is predominantly "hum" and general noise. It then fails to serve any useful purpose, and it is beneficial to switch the signal off. This gives silent gaps in the signal, where there would otherwise be some very noticeable background noise.

A conventional noise gate is activated by the input signal, and cuts off the output when the input signal drops below a certain threshold level. The noise gate circuits featured here operate in a very different manner. They are operated by the MIDI input signal. The signal is switched through to the output when there is some MIDI activity, and switched off if there is no activity for a time that is user adjustable.

Users of these noise gates should be aware that there is a potential flaw in this system. It is possible for a MIDI system to produce a flow of MIDI data at times when there is no audio output signal. The most likely cause of problems are MIDI timing signals, although in practice even these are often only sent when notes are being played. Anyway, in practice it should not be too difficult to arrange things so that the MIDI controller does not keep activating the gate at the wrong times.

A true noise gate provides a switching action, with the output signal either fully switched on or totally cut off. Many practical noise gates fade the signal out rather than simply switching it on and off, in an attempt to make the gating action less obvious. Two versions of this noise gate are provided. One provides a true gating action, and the other provides a fade-out. In general, the circuit which provides the fade-out is better if there is a fairly high background noise level to contend with. The switching circuit is likely to be better if there is only a moderate amount of "hum" and noise to counter.

System Operation

The block diagram of Figure 3.10 shows the general arrangement used in the fader version of the noise gate. The audio

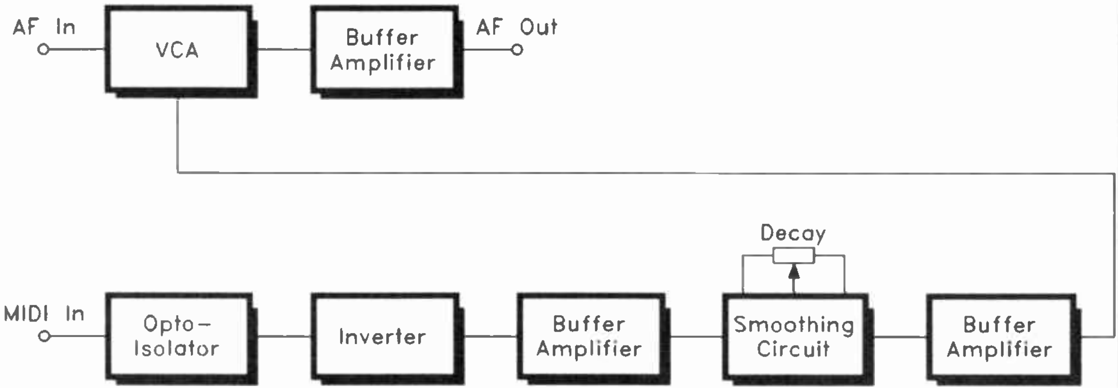

Fig.3.10 The MIDI noise gate block diagram

signal path is via a VCA (voltage controlled attenuator) and a buffer amplifier. The VCA provides unity voltage gain at maximum control voltage. Reducing the control voltage produces losses through the VCA, with what for practical purposes can be regarded as infinite attenuation at zero control voltage.

The control voltage is generated by first feeding the MIDI input signal through the usual opto-isolator input stage. This provides negative output pulses which are inverted to produce positive pulses. The output from the inverter is fed to a buffer amplifier, which in turn feeds a smoothing circuit. A noise gate must have the shortest possible attack time so that it does not significantly "chop" the beginning of the audio input signal each time it is activated. The buffer amplifier ensures that the smoothing circuit is driven from a very low source impedance, and that it therefore has a very short attack time.

In this respect there is a definite advantage in using the MIDI signal to activate the unit, rather than using the audio signal. There has to be a slight lag between the beginning of the MIDI signal and the start of the audio signal from a slave unit, because the audio signal can not commence until a three byte note-on message has been received and processed. This gives the noise gate a chance to switch on before the audio signal commences. There is no guaranteed lag of this type between the start of the MIDI signal from the controller, and the commencement of its audio output signal. However, in practice the unit does not seem to produce any detectable chopping of the MIDI controller's audio output signal.

The decay time of the smoothing circuit is relatively long, and is adjustable from about 0.5 to 5 seconds. The output from the smoothing circuit drives the control input of the VCA via a buffer amplifier. The output voltage from the smoothing circuit is sufficient to produce unity voltage gain from the VCA while there is a reasonable amount of MIDI activity. If the flow of MIDI data halts, the voltage from the smoothing circuit starts to fall, and eventually fades out the audio output signal.

The switch-type noise gate operates in a similar manner, but the VCA and buffer stage are replaced with an electronic switch. Also, the smoothing circuit feeds into a trigger circuit rather than a simple buffer amplifier. The output voltage from the smoothing circuit is sufficient to operate the trigger circuit and switch on the gate provided there is a certain amount of activity on the MIDI input. If the MIDI input signal ceases, the output voltage from the smoothing circuit declines, and the unit switches off the audio output signal.

The Circuits

The circuit diagram for the fader type noise gate appears in Figures 3.11 and 3.12. Figure 3.11 shows the circuit for the stages which generate the control voltage. IC1 is used as an opto-isolator input stage. This is the same as the input stage featured in some of the projects described previously. A THRU output facility is included. IC2 is a CMOS inverter stage which gives positive output pulses from the negative pulses received from IC1. The other five inverters in IC2 are left unused, but have their inputs connected to the 0 volt supply rail in order to prevent spurious operation.

TR1 is the buffer stage, and this is a simple emitter follower type. Its output feeds direct into the smoothing circuit, which is comprised of VR1, R6, and C2. No diode is needed in the emitter circuit of TR1, since TR1 can only source current. When switched off it does not provide a significant discharge path for C2, and the discharge period is therefore controlled by the resistance through VR1 and R6. IC3 acts as the buffer stage, and this is a simple non-inverting type. Note that IC3 must be a device that can operate with its output at very low voltages. Devices such as the μA741C and LF351N will not give a low enough minimum output voltage, and will not operate properly in this circuit.

Figure 3.12 shows the circuit for the VCA and output buffer stage. IC4 is a transconductance amplifier which is used in the standard VCA configuration. The output buffer amplifier is an internal emitter follower stage, but it requires discrete load resistor R16. The circuit will work using an LM13600N or an LM13700N (which are pin for pin compatible, and seem to be all but identical). These are dual transconductance amplifiers, so a stereo noise gate can be produced by using the other section of IC4 in an identical

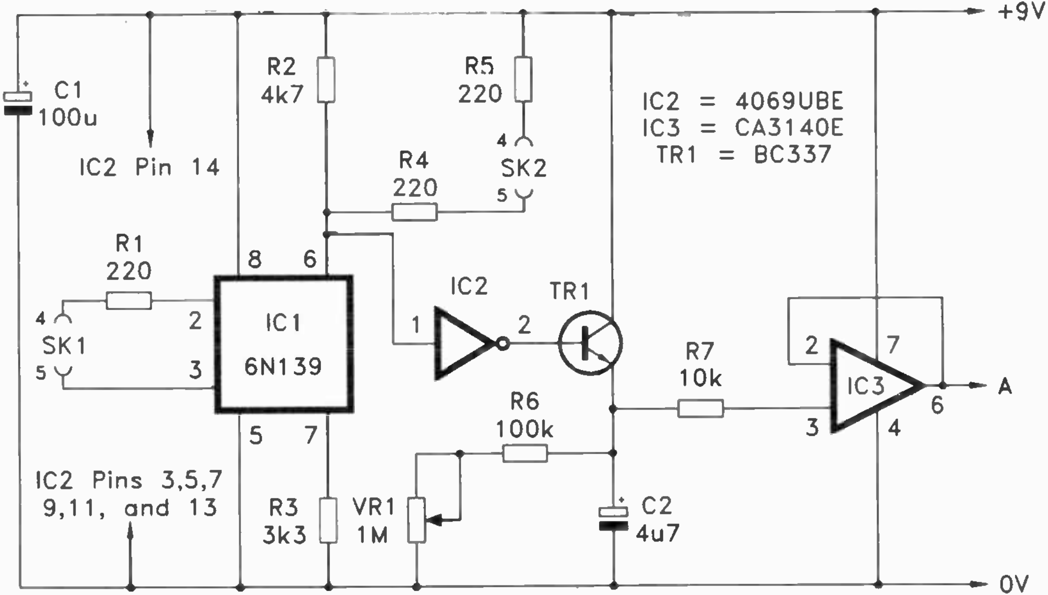

Fig.3.11 The control section of fader type noise gate

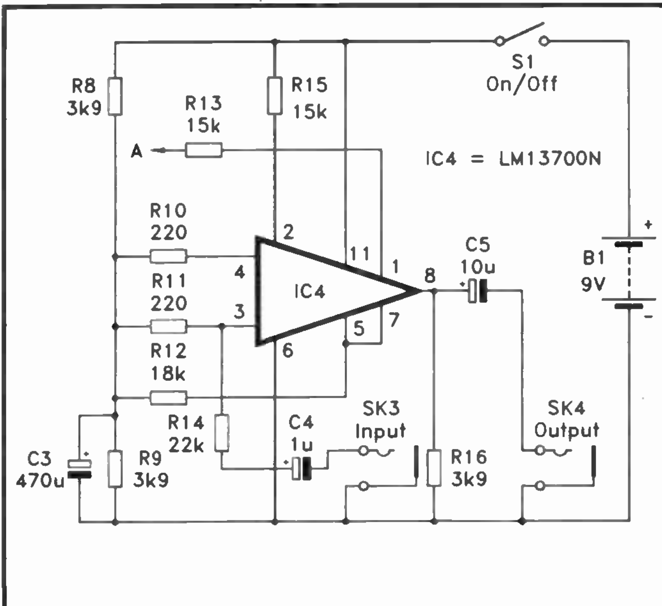

Fig.3.12 The audio section of the fader type noise gate

VCA circuit.

The circuit diagram for the true noise gate appears in Figures 3.13 and 3.14. The control part of the circuit (Figure 3.13) is much the same as the equivalent section of the fader type noise gate. The only difference is that IC3 has been reconfigured so that it operates as a voltage detector circuit rather than a simple buffer stage. Its output goes high when the output voltage from the smoothing circuit is greater than the reference voltage produced by R8 and R9 (about 1.4 volts). The electronic switch circuit (Figure 3.14) is based on a 4016BE quad CMOS analogue switch. In this case only one switch of IC4 is utilized, and no connections are made to the other three switches. The various resistors and capacitors bias the input and output of the circuit, and provide d.c. blocking at the input and output.

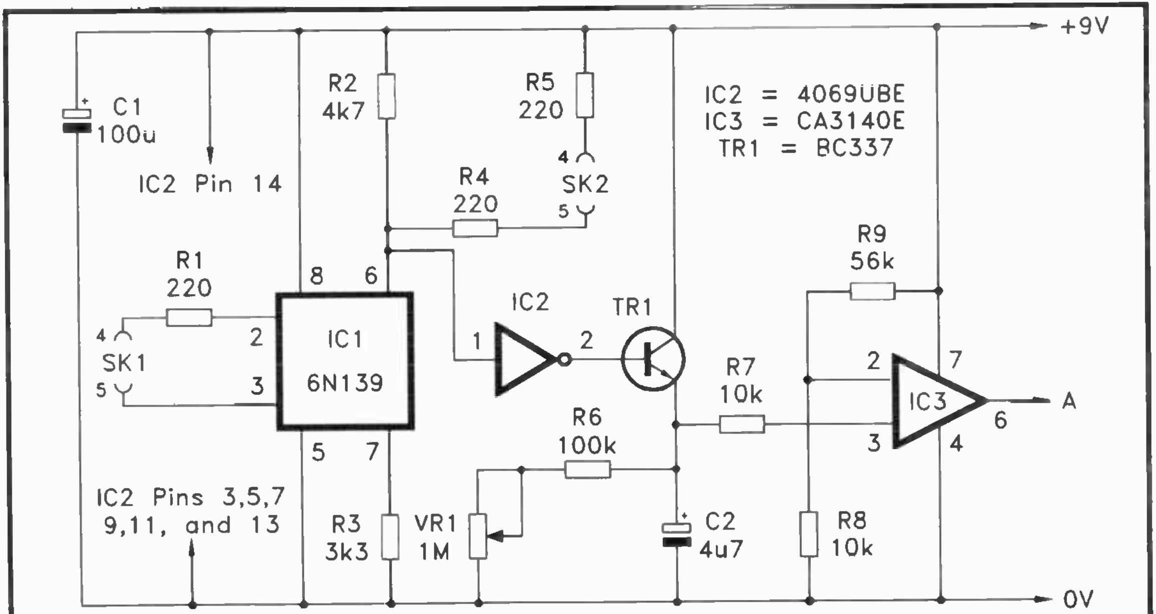

Fig.3.13 The control section of the true noise gate

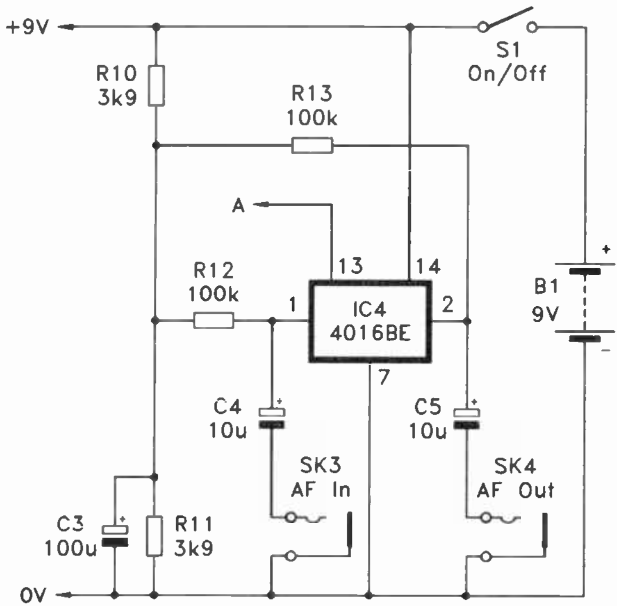

Fig.3.14 The switching section of the true noise gate

The current consumption of the true noise gate is quite low at about 3 milliamps. The fader type noise gate consumes about twice as much current, but a PP3 size battery is adequate as the power source for both circuits. Construction of both units should not present any major difficulties, but bear in mind that the 4069UBE and 4016BE are static sensitive devices. I would not recommend the use of a 4066BE for IC4 in the true noise gate circuit, even though the 4016BE and 4066BE are pin-for-pin compatible. The 4066BE is not designed for use in audio circuits, and can produce strong switching "clicks" on the output signal.

In general, it is best to set VR1 for the shortest switch-off delay (lowest resistance) that does not result in notes being truncated. The optimum setting therefore depends on the types of sound in use. If one or more of the sounds you are using has a long decay period, VR1 must be set for a corresponding long switch-off delay.

Components for Noise Gate (Fading) (Figs 3.11 & 3.12)

Resistors (all 0.25 watt 5% carbon film)

R1	220R
R2	4k7
R3	3k3
R4	220R
R5	220R
R6	100k
R7	10k
R8	3k9
R9	3k9
R10	220R
R11	220R
R12	18k
R13	15k
R14	22k
R15	15k
R16	3k9

Potentiometer

VR1	1M lin carbon

Capacitors

C1	100µ 10V elect
C2	4µ7 50V elect
C3	470µ 10V elect
C4	1µ 50V elect
C5	10µ 25V elect

Semiconductors

IC1	6N139
IC2	4069UBE

IC3	CA3140E
IC4	LM13700N or LM13600N
TR1	BC337

Miscellaneous

S1	SPST min toggle
B1	9 volt (PP3 size)
SK1	5 way 180 degree DIN socket
SK2	5 way 180 degree DIN socket
SK3	6.35mm jack socket
SK4	6.35mm jack socket
	Case
	Circuit board
	Control knob
	8 pin DIL IC holder (2 off)
	14 pin DIL IC holder
	16 pin DIL IC holder
	Battery connector
	Wire, solder, etc.

Components for Noise Gate (Switch) (Figs 3.13 & 3.14)

Resistors (all 0.25 watt 5% carbon film)

R1	220R
R2	4k7
R3	3k3
R4	220R
R5	220R
R6	100k
R7	10k
R8	10k
R9	56k
R10	3k9
R11	3k9
R12	100k
R13	100k

Potentiometer

VR1	1M lin carbon

Capacitors

C1	100µ 10V elect
C2	4µ7 50V elect
C3	100µ 10V elect
C4	10µ 25V elect
C5	10µ 25V elect

Semiconductors

IC1	6N139
IC2	4069UBE
IC3	CA3140E
IC4	4016BE
TR1	BC337

Miscellaneous

S1	SPST min toggle
B1	9 volt (PP3 size)
SK1	5 way 180 degree DIN socket
SK2	5 way 180 degree DIN socket
SK3	6.35mm jack socket
SK4	6.35mm jack socket
	Case
	Circuit board
	Control knob
	8 pin DIL IC holder (2 off)
	14 pin DIL IC holder (2 off)
	Battery connector
	Wire, solder, etc.

MIDI Control Pedal

Most MIDI equipped instruments permit a number of parameters to be controlled via MIDI control change messages. Some parameters are controlled via continuous controls, and these permit such things as modulation depth and panning to be altered. Others are controlled via simple switch type controllers, which only provide on/off switching of what is usually some form of built-in effect. The most common switch type controller is the sustain on/off control, and virtually all MIDI instruments seem to respond to this one.

In some cases this is the only switch type controller that is implemented, but many instruments implement one or two others, such as portamento switching.

This pedal generates MIDI controller on messages when the pedal is operated, and controller off messages when the pedal is released. As described here the unit operates on MIDI channel 1, and it operates MIDI control number 64 (the number allocated to the sustain pedal). However, it can easily be modified to operate on a different channel and (or) to operate a different control number. It can therefore provide pedal control of any MIDI switch type control.

System Operation

The block diagram in Figure 3.15 shows the arrangement used in the MIDI controller pedal. This looks a bit involved, but it is basically quite simple. The UART is at the heart of the unit, and in this case it is only the transmitter section that is utilized. The output of the UART drives the MIDI OUT socket via a simple inverter having an open collector output stage. A 2MHz clock oscillator and divide by four circuit provide the UART with a 500kHz clock signal, which sets the required baud rate of 31,250.

A MIDI controller message is a three byte type. The first byte contains the MIDI control change code and the channel number. The second byte contains the number of the control which must be changed, which in this case will normally be 64. The final byte is the new value for the control. The original MIDI specification stated that for switch type controls only values of 0 (off) and 127 (on) should be recognised. Some years later this was changed by an amendment to the MIDI specification, and new equipment should recognise values from 0 to 63 as off, and 64 to 127 as on. In order to be certain that the unit will have the desired effect with all instruments it is therefore necessary to use values of 0 and 127 in the third byte.

The basic action of the unit is to feed the first eight bit code to the UART, and then send a trigger signal to the UART so that this code is transmitted. After a suitable delay the second code and another trigger signal are supplied to the UART, followed by the third code and another trigger signal

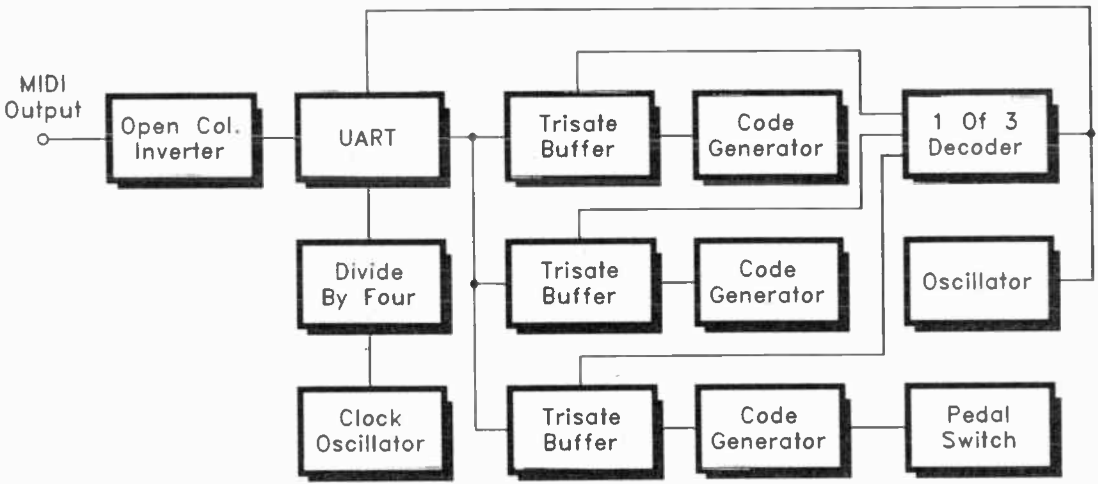

Fig.3.15 The MIDI controller pedal block diagram

after a further delay. The unit could be designed so that it would only send a three byte message each time the pedal was closed or released. In practice there seems to be no problem if the unit sends a continuous stream of controller messages, with the third byte reflecting the state of the pedal switch (127 for closed – 0 for open). The only potential problem is that a continuous stream of end-to-end MIDI messages could overload the instrument used with the pedal. Data therefore has to be sent at a rate which is high enough to give a reasonably quick response when the pedal is operated, but low enough to avoid MIDI "choke".

The parallel inputs of the UART are fed from three octal tristate buffers. These are in turn fed from code generators which produce the appropriate values for the three byte message. In the case of the code generator which produces the final byte, it is controlled by the pedal switch, and the code it produces reflects the position of the pedal switch. The tristate buffers are controlled by a one-of-three decoder. This is in turn fed from an oscillator. On the first oscillator cycle the first tristate buffer is activated, and the oscillator sends a pulse to the UART that results in the first byte being transmitted. On the second oscillator cycle the second tristate buffer is activated, and the oscillator again triggers the UART. On the third oscillator cycle this process is repeated, and the third byte is transmitted. The one-of-three decoder then cycles back to its original state, and this whole process is repeated indefinitely.

The Circuit

The circuit diagram for the MIDI controller pedal appears in Figures 3.16 to 3.18. The clock oscillator, divider circuit, output stage, and UART circuits are shown in Figure 3.16. The clock oscillator is the same as the one featured in the MIDI comparator project which was described previously. The only difference is that the 500kHz output from IC1 feeds into pin 40 of the UART, not pin 17. This is simply because the 6402 has spearate transmitter and receiver clock inputs. In this case it is obviously only the transmitter clock input that is used. As in the previous project, the UART is wired for a word format of one start bit, eight data bits, one stop bit, and

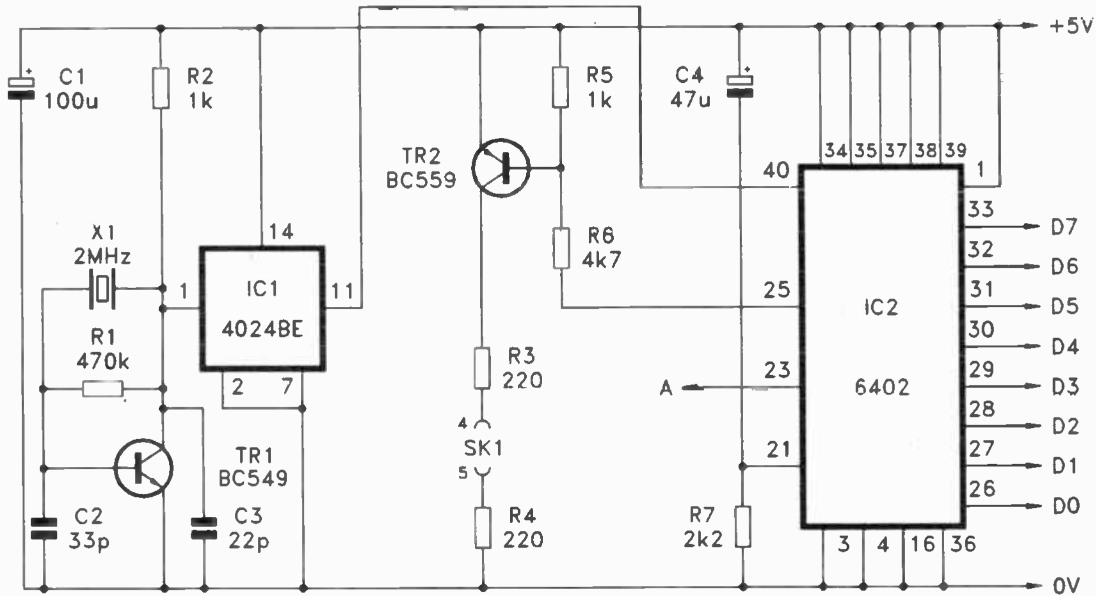

Fig.3.16 The clock, output stage, and UART circuitry

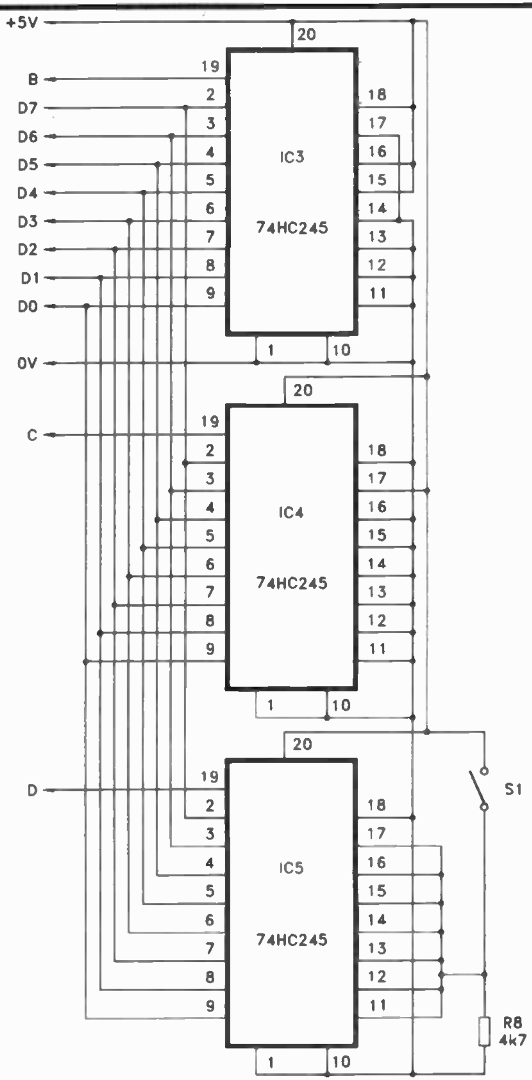

Fig.3.17 The tristate buffers and code generator circuitry

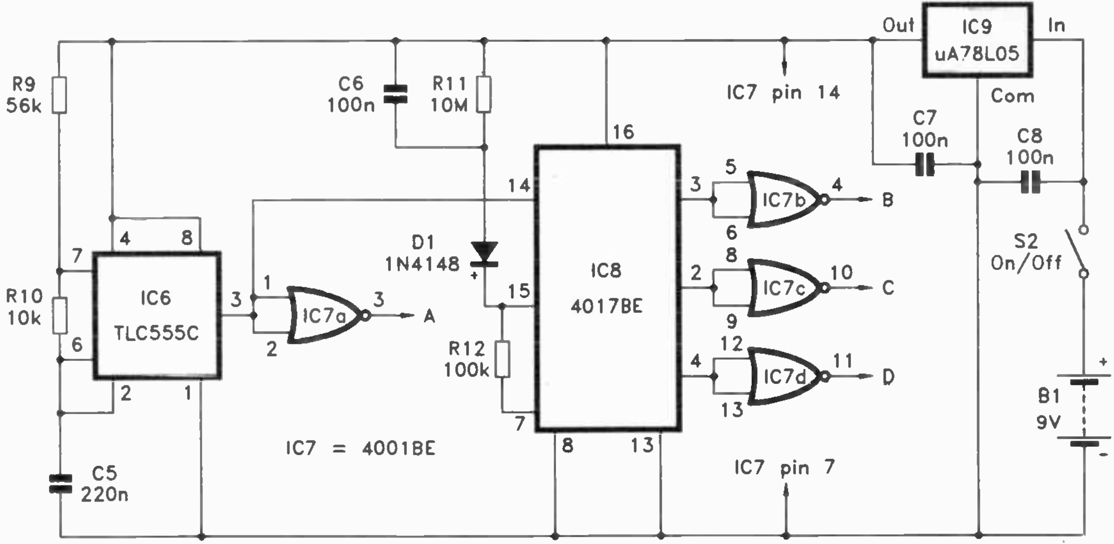

Fig.3.18 The MIDI pedal control stages

no parity. The serial output of the UART is fed to a simple common emitter switch (TR2). Pin 23 of IC4 is fed from the oscillator in the control circuit, and it is the signal here that causes data to be loaded into IC4 and transmitted.

Figure 3.17 shows the tristate buffer and code generator circuitry. The tristate buffers are actually 74HC245 octal transceivers, but in this case the send/receive input of each device is tied to earth, and they act as straightforward tristate buffers. The inputs of IC3 are hard wired to produce the control change code (1011) on channel 1 (0000). It would not be difficult to make the MIDI channel adjustable, but there is probably no point in doing this. There should be no difficulty in using the unit with an instrument set to channel 1. In most cases there will probably be no point in using channels at all, and the slave instrument could be used in mode 1 (omni on/poly).

IC4 has its inputs wired with the code 01000000 (64 in decimal), which is the control number for a sustain pedal. If necessary, these inputs can be wired with a different binary code. For example, use 01000001 (65) if the unit is to be used as a portamento pedal. IC5 normally has all eight inputs taken low, and it therefore supplies a value of zero in the third byte of each message. This switches off the sustain effect (or whatever). Closing S1 pulls data lines 0 to 6 high, and the unit then transmits a value of 127 in the third byte of each message. This switches on the sustain or other effect.

The control circuitry appears in Figure 3.18. The oscillator is a standard 555 astable based on IC6. It operates at about 100Hz, which is high enough to give a reasonably short response time, but long enough to avoid problems with MIDI "choke". A shorter response time can be obtained by reducing the value of C5, but doing so increases the risk of overloading the processor in the slave instrument. Values as low as 10n should provide correct operation of the pedal circuit. The output of IC6 directly drives the clock input of the one-of-three decoder (IC8), but it drives the UART via an inverter formed from IC7a.

The one-of-three decoder is actually a CMOS 4017BE one-of-ten decoder, but the coupling through R12 results in IC8 resetting itself to zero each time the count reaches three.

This effectively eliminates outputs "3" to "9", leaving outputs "0", "1", and "2" to provide a one-of-three action. These respectively control IC3, IC4, and IC5. The outputs of IC8 are active high, but a low control level is needed in order to activate one of the tristate buffers. IC8 therefore controls the tristate buffers via inverters formed from IC7b to IC7d. C6, R11, and D1 provide a reset pulse to IC8 at switch-on, which ensures that the unit does not initially generate invalid MIDI signals.

The circuit requires a reasonably stable 5 volt supply. This is derived from a 9 volt battery via a small monolithic voltage regulator (IC9). The supply current is surprisingly low, and is typically about 5 milliamps. This is due to the fact that the circuit is based on low power CMOS integrated circuits. A PP3 size battery is adequate to power the unit. When constructing the unit, bear in mind that apart from IC6 all the integrated circuits are static sensitive types. The unit must be built into a fairly tough case, and S1 should be a heavy duty non-locking push-button switch mounted on the top of the case, so that it can be operated by foot. Alternatively, S1 can be a separate footswitch connected to the main unit via a twin cable.

Components for MIDI Controller Pedal (Figs 3.16, 3.17 & 3.18)

Resistors (all 0.25 watt 5% carbon film)

R1	470k
R2	1k
R3	220R
R4	220R
R5	1k
R6	4k7
R7	2k2
R8	4k7
R9	56k
R10	10k
R11	10M
R12	100k

Capacitors

C1	100μ 10V elect
C2	33p ceramic plate
C3	22p ceramic plate
C4	47μ 25V elect
C5	220n polyester
C6	100n polyester
C7	100n ceramic
C8	100n ceramic

Semiconductors

IC1	4024BE
IC2	6402
IC3	74HC245
IC4	74HC245
IC5	74HC245
IC6	TLC555C
IC7	4001BE
IC8	4017BE
IC9	μA78L05 (5V 100mA positive regulator)
TR1	BC549
TR2	BC559
D1	1N4148

Miscellaneous

S1	SPST heavy-duty non-locking push-button switch
S2	SPST min toggle
B1	9 volt (PP3 size)
X1	2MHz wire-ended crystal
SK1	5 way 180 degree DIN socket
	Case
	Circuit board
	Battery connector
	8 pin DIL IC holder
	14 pin DIL IC holder (2 off)
	16 pin DIL IC holder
	20 pin DIL IC holder (3 off)
	40 pin DIL IC holder
	Wire, solder, etc.

MIDI Lead Tester

The lack of reliability from MIDI leads is something that was mentioned previously in this chapter. There is a further problem in that many supposed MIDI leads do not provide the correct interconnections. This simple device provides a quick means of determining the interconnections between the two 5 way 180 degree DIN plugs of an alleged MIDI lead. It achieves this via a display of five LEDs. The circuit diagram for the MIDI lead tester appears in Figure 3.19.

The Circuit

IC1 is a 555 timer which acts as a low frequency oscillator operating at about 5 to 6Hz. It feeds into a CMOS seven stage binary counter (IC2), but in this circuit only two stages of IC2 are utilized. It therefore provides a divide by four action, and gives an output frequency of a little over 1Hz. This is used as the clock signal for IC3, which is a CMOS one-of-ten decoder. However, in this case IC3 resets itself when the count reaches "6". Accordingly, IC3 actually provides a one-of-six action on outputs "0" to "5", and outputs "6" to "9" are effectively eliminated.

The five display LEDs (D1 to D5) are driven from outputs "0" to "4" of IC3 via SK1, SK2, and the MIDI lead under test. The common connection point goes to the output of IC1 rather than to the 0 volt supply rail. If the MIDI lead provides a "straight" connection between all five pins, D1 to D5 switch on, one at a time, and in sequence, for a little under one second each. When activated a LED does not light up continuously though, but is instead flashed on and off four times by IC1. The display then goes blank for a little under one second while output "6" of IC3 goes high. D1 to D5 then switch on in sequence again, and the circuit repeats this process indefinitely.

In theory, a MIDI lead does not provide interconnections between all five pins, and this action should not be obtained. In reality, many MIDI leads do seem to have all five pins linked, and will give this action with the five LEDs being turned on in sequence. The extra connections should not be of any real significance, and a lead of this type should work perfectly well if used with standard MIDI IN, OUT, and THRU sockets.

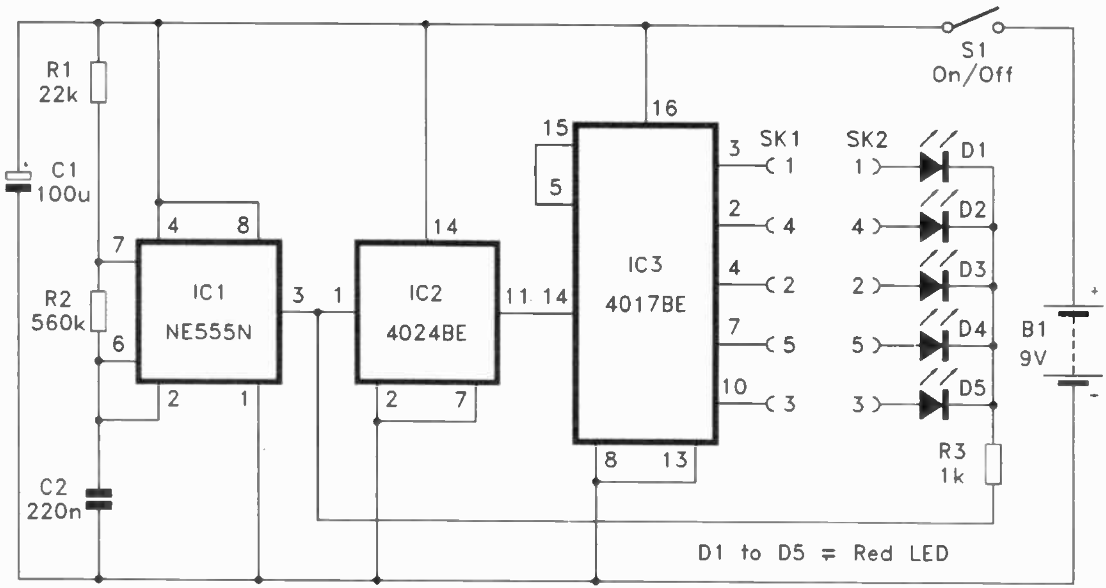

Fig.3.19 The MIDI lead tester circuit diagram

A MIDI lead should only provide a straight connection between pins 2, 4, and 5. With a lead of this type D2, D3, and D4 will be activated, in that order, followed by a period of about two seconds with none of the LEDs activated.

If a lead is failing to provide an interconnection between a pair of pins, the appropriate LED will fail to light up. For example, if D4 fails to switch on, there is no interconnection between pin 5 of SK1 and pin 5 of SK2. Of course, D1 and D2 failing to switch on is correct, as these pins are not used by a MIDI link. The lead should still work even if D3 does not switch on, but the screen of the lead might be ineffective, and the lead might radiate significant amounts of radio frequency interference.

If the LEDs switch on in the wrong sequence, the lead is providing a cross-coupling. This means that it is actually an audio type and not a MIDI lead at all. A lead of this type is unsuitable for use as a MIDI lead unless one of the plugs is rewired. If two LEDs light up simultaneously, this means that one of the plugs has two pins shorted together. If one of the unused pins is shorted to one of the pins that is used (e.g. pin 1 to pin 4) this should not matter, but it could give problems with a non-standard MIDI port such as the combined OUT and THRU type of the Atari ST computers.

The average current consumption of the circuit is about 8 milliamps when a LED is activated, and two or three milliamps less than this with all the LEDs switched off. The unit can therefore be powered from a PP3 size 9 volt battery.

Components for MIDI Lead Tester (Fig.3.19)

Resistors (all 0.25 watt 5% carbon film)

R1	22k
R2	560k
R3	1k

Capacitors

C1	100μ 10V elect
C2	220n polyester

Semiconductors

IC1	NE555N
IC2	4024BE
IC3	4017BE
D1 to D5	Red panel LEDs

Miscellaneous

S1	SPST min toggle
B1	9 volt (PP3 size)
SK1	5 way 180 degree DIN socket
SK2	5 way 180 degree DIN socket
	Case
	Circuit board
	8 pin DIL IC holder
	14 pin DIL IC holder
	16 pin DIL IC holder
	Battery connector,
	Wire, solder, etc.

Mains Power Supply

Apart from the guitar practice amplifier, all the projects featured in this book are designed for battery power. If you wish to avoid the expense of replacement batteries, NiCad rechargeable types are probably the best option. These give very low running costs in the medium to long term, and do not compromise portability. The circuits can be powered from a ready-made battery eliminator provided it has an appropriate output voltage, and it also has a properly smoothed output. Many inexpensive battery eliminators have unregulated outputs which have a high ripple content. These are unlikely to be suitable for use with the circuits featured here.

If you wish to build your own mains power supply unit, Figure 3.20 shows the circuit diagram for a 9 volt mains power supply which has a low ripple and general noise content on its regulated output. This circuit is only included for those who are suitably experienced at project construction, and who are competent to deal with a project that connects to the dangerous mains supply. Bear in mind that mistakes with a

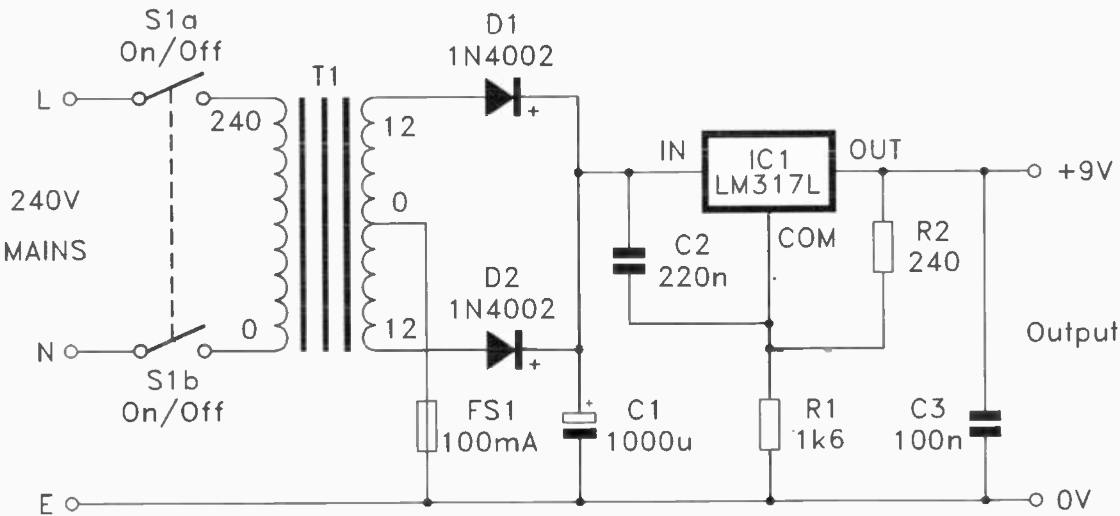

Fig.3.20 The 9 volt mains P.S.U. circuit

circuit of this type could prove lethal. This circuit must be constructed to comply with the normal safety regulations. In particular, it must be housed in a case of all-metal construction, and both the case and the 0 volt output of the supply must be reliably earthed to the mains earth lead. The case must be a type which has a lid secured by means of screws, not a clip-on type that would give easy access to the dangerous mains wiring.

Components for Mains Power Supply (Fig.3.20)

Resistors	
R1	1k6 1% 0.5 watt metal film
R2	240R 1% 0.5 watt metal film
Capacitors	
C1	1000µ 25V elect
C2	220n ceramic
C3	100n ceramic
Semiconductors	
D1	1N4002 (100V 1A)
D2	1N4002 (100V 1A)
IC1	LM317L
Miscellaneous	
T1	Standard mains primary, 12–0–12 volt, 100mA secondary
FS1	100mA 20mm anti-surge
S1	Rotary mains switch
	All-metal case with screwed lid
	Circuit board
	20mm fuse holder
	Control knob
	3 core mains lead and fused plug (2A)
	Wire, solder, etc.

For an output voltage of 6 volts T1 should be a 9–0–9 volt 100mA type, and R1 should have a value of 1k. For a 5 volt output potential T1 should be a 9–0–9 volt 100mA component and R1 should have a value of 750R.

BERNARD BABANI BP363

Practical Electronic Music Projects

- While many facets of electronic project construction have waxed and waned over the years, music related projects of various types have remained as popular as ever. This is perhaps not surprising, since many electronic music projects can be home constructed for much less than the cost of equivalent ready-made products. Also, you can 'broaden your horizons' with home constructed music projects that have no true commercial equivalents.

- This book provides practical circuits for a number of electronic music projects. All can be built at relatively low cost, and use standard, readily available components. The projects covered can be broadly divided into three categories: Guitar Projects, Miscellaneous Music Projects and MIDI Projects.

- The projects cover a range of complexities, but most are well within the capabilities of the average electronics hobbyist. None of the projects require the use of test equipment in order to get them set up correctly, and several of the projects are suitable for near beginners.

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