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- Again the circuits contained in this book are primarily intended for those who have some experience, even if only limited, of electronic construction techniques and are capable of building projects working from just a circuit diagram and without the aid of detailed constructional information or board layouts, etc.

- The circuits cover a wide range of subjects including audio, test gear, radio, house and car, miscellaneous, etc., so there should be a number of circuits of interest to most readers. The circuits are all based on modern components that should be readily available and generally inexpensive.

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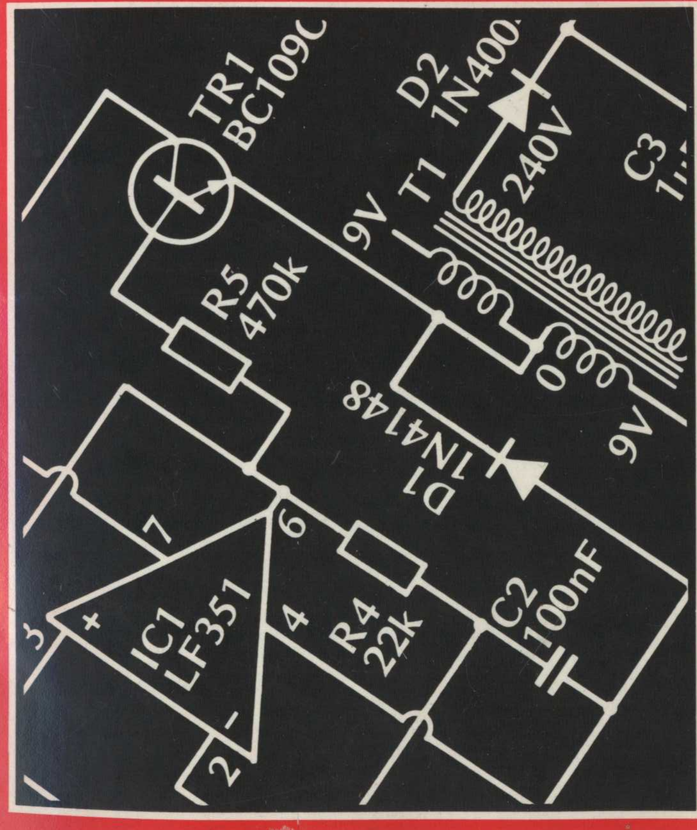
Popular Electronic Circuits Book 2

R. A. PENFOLD



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POPULAR ELECTRONIC CIRCUITS BOOK 2

by

R.A. PENFOLD

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PREFACE

Like "Popular Electronic Circuits Book 1", this book is designed to provide a wide range of circuits for electronics enthusiasts who have some experience of electronic project construction, and can produce projects working from just a circuit diagram and without the aid of any constructional information. Also as was the case with "Popular Electronic Circuits Book 1", most of the projects are fairly simple and straight forward, and should not be beyond the capabilities of constructors who have only a limited amount of experience.

The circuits cover a wide range of subjects, including audio, radio, test gear, household gadgets, etc. There should therefore be a number or projects of use and interest to most electronics enthusiasts. The circuits are all based on modern readily available components, and in general are quite inexpensive to construct. Most of the circuits are for devices that, when completed, form projects in their own right. However, a few are intended as electronic building block for use in larger projects or systems, and a few fall into both of these categories. Where relevant, any special setting up procedures are described.

None of the circuits in this publication duplicate those in "Popular Electronic Circuits Book 1".

R. A. Penfold

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CHAPTER 1

AUDIO CIRCUITS

Guitar Preamplifier

If you have ever tried to use an electric guitar with a hi-fi amplifier you will amost certainly have run into problems. High level inputs intended for use with tuners or tape decks have inadequate sensitivity, while the magnetic cartridge input has an equalised frequency response which results in excessive bass and very limited treble response.

A simple way around the problem is to use a preamplifier to boost the output of the guitar to a level which is sufficient to drive a tuner, tape, or auxillary input. The low noise design shown in the circuit diagram of Fig.1 is based on a low noise operational amplifier (IC1) which uses a JFET input stage to give the low noise and distortion levels. The unit can also be used to give the popular treble boost effect.

The circuit is basically just an opertional amplifier used in the non-inverting mode. R1 and R2 are used to bias the non-inverting input, while R4 and R5 bias the inverting input from the output of IC1. The voltage gain of the amplifier is equal to the sum of R3, R4 and R5, divided by R3, and this gives a voltage gain which is adjustable from about 5.5 to 27 times by means of R5. This range of adjustment should enable any guitar to be matched to any hi-fi amplifier.

When S2 is closed, R6 and C6 are shunted across R3, and this gives a boost in voltage gain. However, the gain is only boosted at high frequencies where C6 has a low impedance, and this gives the treble boost effect.

The current consumption of the circuit is only about 2mA., and a small 9 volt battery is a suitable power source. The circuit will also work on higher supply voltages up to a

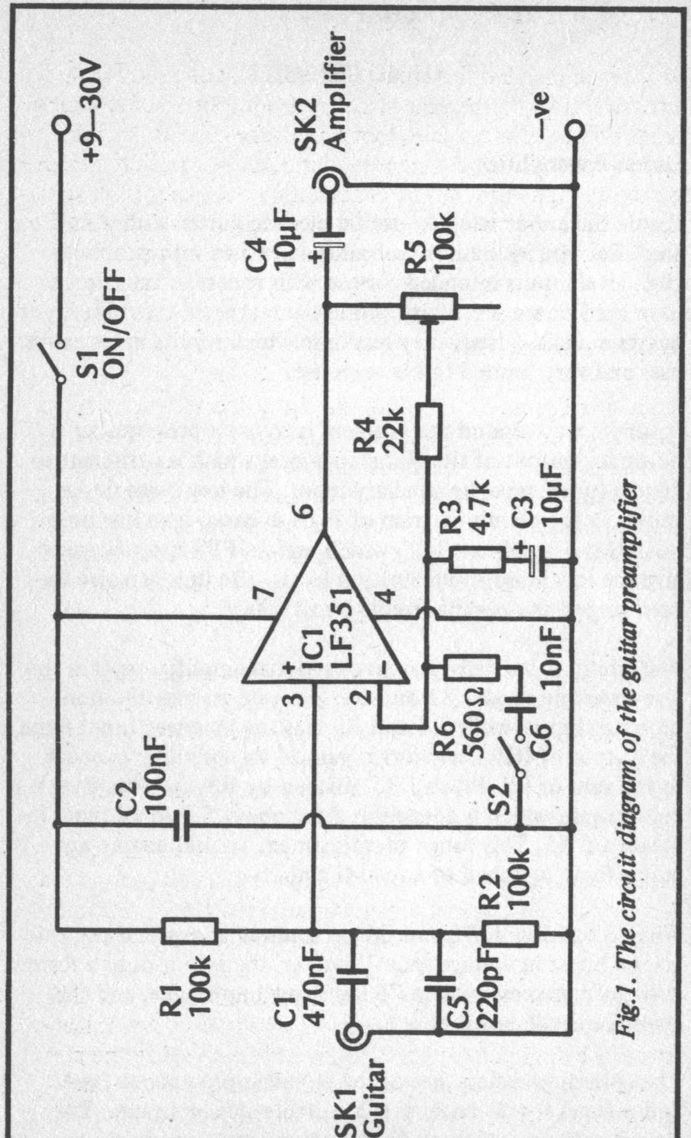

Fig.1. The circuit diagram of the guitar preamplifier

maximum of about 30 volts.

It is recommended that a metal case should be used for the unit as this will screen the circuitry from sources of electrical interference such as mains hum (earth the case to the negative supply rail). The component layout must be carefully designed since the input and output of the unit are in-phase, and stray feedback could therefore result in instability, although C5 helps to minimise the risk of problems of this type.

R.I.A.A. Preamplifier

During the recording process the signal transferred onto an ordinary record is given a substantial amount of treble boost and bass cut. The treble boost enables an improved signal to noise ratio to be obtained when complementary treble cut is used at the record player. The bass cut prevents excessive low frequency groove modulations, and must be counteracted by complementary bass boost at the record player in order to obtain a flat frequency response.

This preamplifier circuit has the necessary equalisation to give a flat frequency response overall when used with a magnetic pick-up, and it also gives the necessary amplification to boost the output from such a pick-up from around 5mV. R.M.S. to the 500mV. R.M.S. or so needed by most power amplifiers for maximum output. The circuit of Fig.2 is for one channel only, but the LM387 I.C. used in the unit is a dual amplifier and can be used in a stereo version of the circuit (the pin connections for the second channel are shown in brackets – the supply connections are common to both sections of the device).

The LM387 is virtually an operational amplifier, but it differs from a straight forward operational amplifier in that it has an internal bias circuit which supplies a bias voltage to the non-inverting input. Normally in circuits of this type the non-inverting input is biased to half the supply voltage and 100% negative feedback is provided at D.C. so that the output is

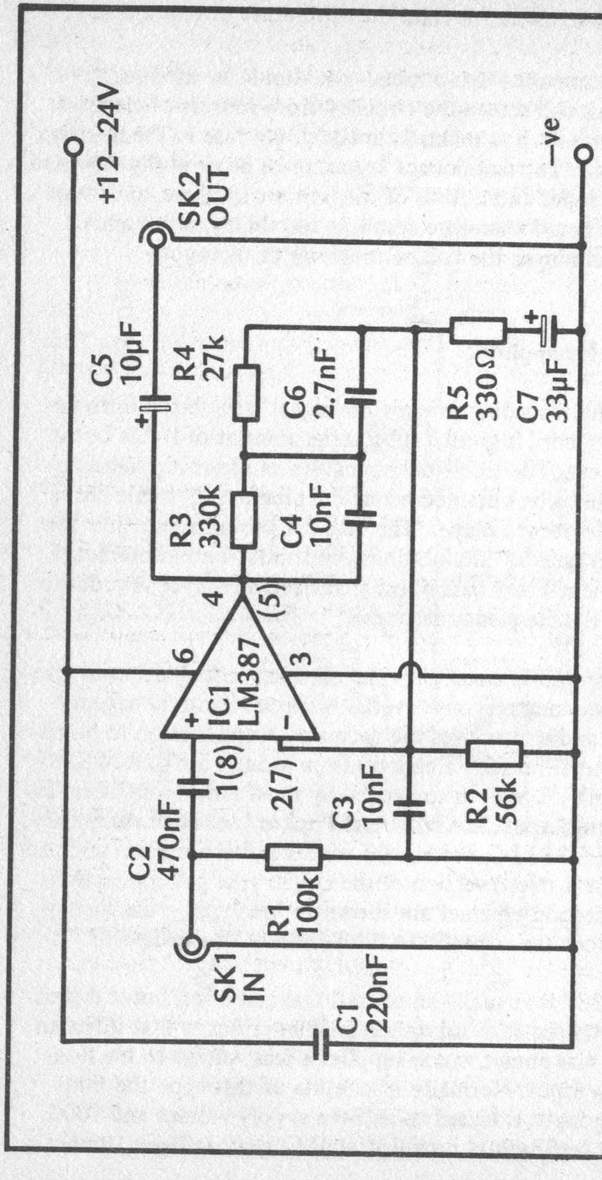

Fig. 2. The circuit diagram of the R.I.A.A. preamplifier

also biased to this level and the optimum overload margin is obtained. The LM387 has a much lower input bias voltage, and a certain amount of voltage gain at D.C. is therefore needed in order to give the optimum quiescent output voltage. This is the reason for including R2 in the circuit, and not just the usual A.C. feedback decoupling components which are R5 and C7 in this case. The values of R3, R4, C4, and C6 are chosen to give the necessary voltage gain and equalisation to the circuit. R1 shunts the input impedance of the circuit to a level that gives a good match for a magnetic pick-up.

The LM387 gives low levels of noise and distortion, and the circuit gives true hi-fi performance. The circuit will operate on any supply voltage from about 12 volts to 24 volts (30 volts absolute maximum).

As with the previous circuit, a metal case to provide screening is recommended, and due to the very high gain of the circuit at low frequencies this is even more important in this case. Also in common with the previous design, the input and output of this circuit are in-phase, and a carefully designed layout is needed, although this problem is eased somewhat by the fairly low gain of the circuit at high frequencies.

Cassette Preamplifier

This simple preamplifier is intended for use with a cassette mechanism to produce a cassette player for use in conjunction with a hi-fi amplifier (the output of the unit cannot directly drive loudspeakers). The circuit is based on the LM387 dual low noise audio amplifier (as featured in the previous project), and is suitable for stereo or mono operation, see Fig.3.

The output from a tape head increases with output frequency at a rate of 6dB. per octave, and treble boost for noise reduction in used during the recording process. The circuit must therefore provide equalisation in order to give a flat frequency response to the overall system, but the degree of

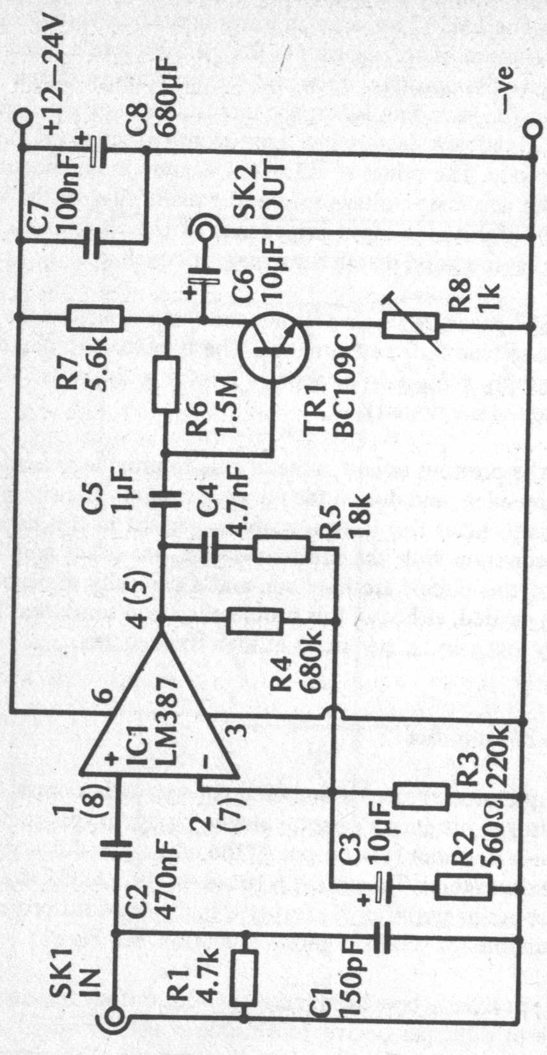

Fig. 3 The circuit diagram of the cassette preamplifier

treble cut required is far less than might at first appear necessary. This is simply because tape heads and recording tapes are not perfect, and tend to be rather inefficient at the higher audio frequencies. Thus the actual playback response needs to be rolled off with increased frequency only into the middle audio frequency range, and then the response must flatten out.

This circuit is very similar to the previous one, but the equalisation components (R2, R3, R4, R5, and C4) have values that give the appropriate response for this application. C1 is needed to prevent instability.

Although the voltage gain of IC1 is quite high, it does not give sufficient output to drive most amplifiers due to the very low output level of a cassette tape head which is typically only about 250µV. for a stereo head, or about double this figure for a mono type. The output of IC1 is therefore boosted by Tr1 which is used as a straight forward common emitter amplifier with a controlled amount of negative feedback provided by R8. The latter is adjusted to give the required output level with minimum value corresponding to maximum output.

It is likely that the supply will contain a certain amount of noise from the cassette deck motor, and C8 is used to filter out this noise.

As was the case with the previous circuits, the component layout should be carefully designed to avoid stray feedback and consequent instability, and the circuitry should be kept a reasonable distance from sources of electrical interference or should be screened.

Stereo Microphone Preamplifier

High impedance (50k) dynamic microphones have an output level of only about 2 to 5mV., and this is far too low to drive most amplifiers and many other items of equipment. This preamplifier has low levels of noise and distortion, and will boost the output voltage of a high impedance dynamic microphone by 40dB. (100 times). It is built around the LM387 dual amplifier, and by using both sections of this device a stereo preamplifier can be produced.

Fig.4 shows the circuit diagram for one channel of the unit (the numbers in brackets are the pin connections to IC1 for the other stereo channel, and the two channels are the same apart from the differing pin connections).

R1 is used to shunt the input of the unit in order to reduce the input impedance to the required level, and C2 is an R.F. filter capacitor that helps to prevent the breakthrough of radio signals. C2 also helps to avoid instability, especially when no microphone is plugged into the unit.

The inverting input of IC1 is biased by R3 and R2, while the voltage gain of the amplifier is governed by R3 and R4 (the shunting effect of R2 on R4 is too minor to have any significant effect on the gain of the unit). The voltage gain is approximately equal to R3 plus R4 divided by R4, which works out at fractionally over 100 times with the specified values. The voltage gain can be increased if necessary by raising the value of R3, or reduced by decreasing the value of this component, but it is not advisable to boost the gain to more than about 300 times as greater voltage gains could result in noticeably reduced audio quality and (or) instability.

Like the previous circuits, this one is sensitive to stray pick-up of mains hum and other types of electrical interference, and it is recommended that it should be housed in a metal case which is earthed to the negative supply rail. Also as before, the input and output of the circuit are in-phase and the component layout should be arranged with reasonable care in order to avoid excessive stray feedback and possible instability.

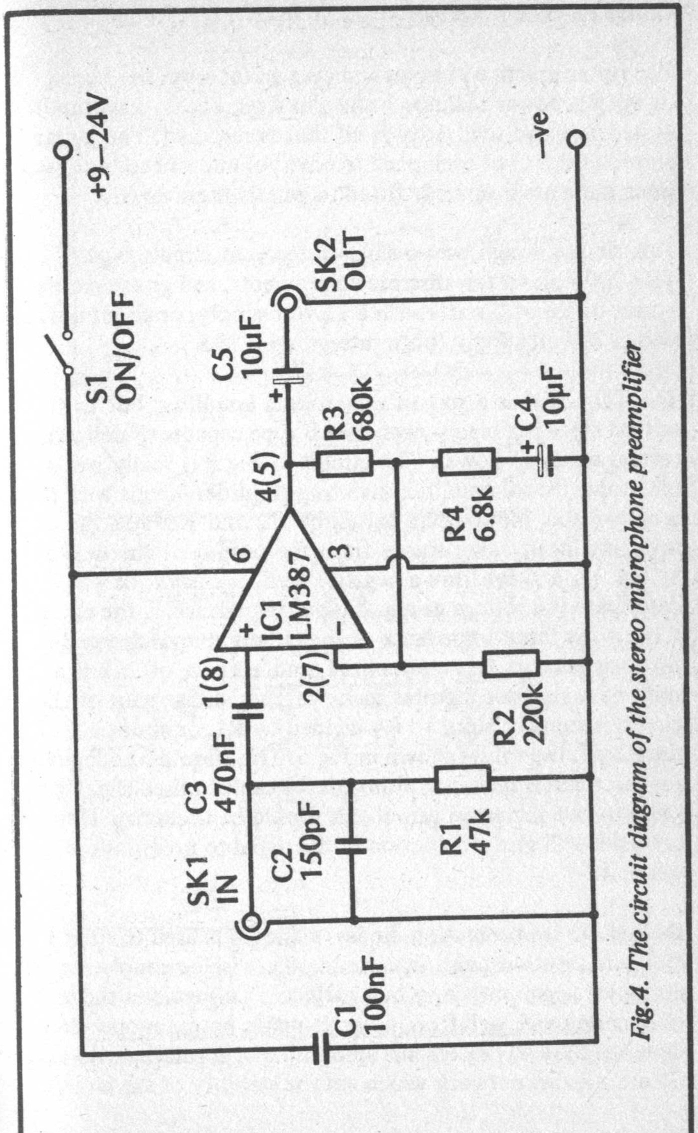

Fig.4. The circuit diagram of the stereo microphone preamplifier

Guitar Practice Amplifier

For guitar practice at home a simple guitar amplifier having an output power of a few watts plus a reasonably high input impedance and sensitivity is all that is required. The guitar amplifier does not even need to have volume and tone controls since these are invariably fitted to guitars these days.

This simple design uses a single integrated circuit type TDA2006 plus a few discrete components, and gives an output power of about 2 watts with a 12 volt supply, or about 8 watts with a 24 volt supply (both ratings are R.M.S.).

The TDA2006 is a sort of operational amplifier, but the output stage is a high power class B type capable of delivering several watts of power. The circuit of Fig.5 is really just a basic operational amplifier inverting amplifier circuit with the non-inverting input being biased by R2 and R3, and the inverting input being biased from the output of the device by R4. R1 and R4 form a negative feedback network which determine the voltage gain and input impedance of the circuit. R1 sets the input impedance of the circuit at a value equal to the resistance of this component, and a figure of 22k is a suitable match for a guitar pick-up. The voltage gain of the circuit is roughly equal to R4 divided by R1, or about 123 times with the values shown in Fig.5. This should be adequate for most guitar pick-ups, although R4 can be raised slightly in value to give increased gain if this should be necessary. However, this will give a reduction in the signal to noise ratio of the unit.

D1 and D2 are protection diodes, while C4 is used to filter out any hum or other noise that could otherwise be coupled to the non-inverting input via its bias network. This enables the unit to operate quite well from a simple mains power supply that does not have any electronic smoothing or regulation. R5 and C6 are a Zobel network which aid the stability of the circuit.

The unit requires a supply current of about 250mA. for an

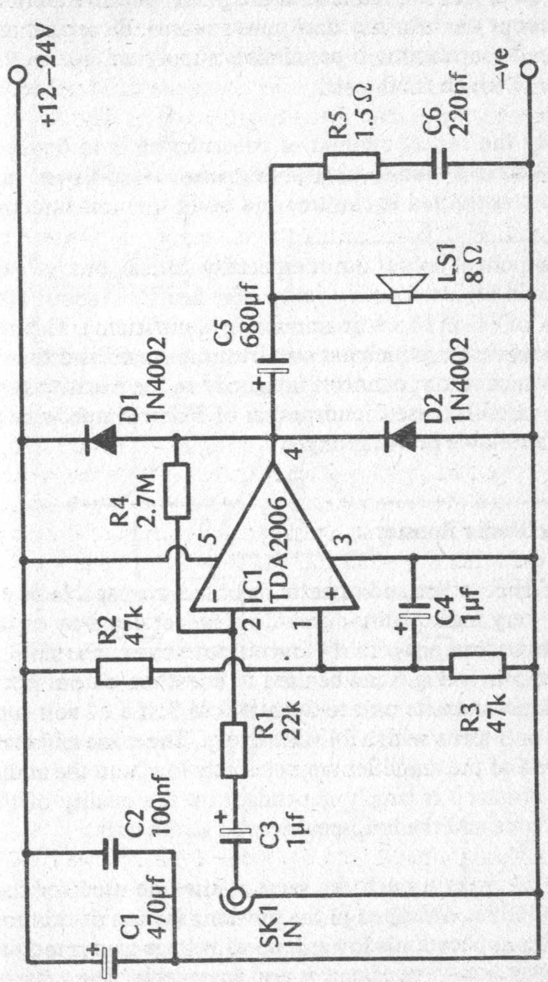

Fig. 5. The circuit diagram of the guitar practice amplifier

output of 2 watts R.M.S., or a supply current of around 500mA. for an output of 8 watts R.M.S. This makes battery operation rather impractical, and a mains power supply is really about the only practical power source. Be careful not to exceed the maximum permissible supply voltage of the TDA2006 which is 30 volts.

Probably the easiest method of construction is to buy a medium sized 8 ohm speaker in an enclosure (or fit one into a home-constructed enclosure) and build the unit into this.

The component layout is not especially critical, but note that IC1 should be fitted onto a fairly large heatsink (about 100 sq. cms. of 16 or 18 s.w.g. aluminium is sufficient). The heat-tab of this device, which has output short circuit and thermal shutdown circuitry, connects internally to the negative supply pin. Be careful to use a loudspeaker of 8 ohms impedance and having a suitable power rating.

Cassette/Radio Booster

Many V.H.F. radios and cassette recorders are capable of providing a very high quality output, but are let down by small internal speakers and a fairly low output power. The simple circuit shown in Fig.6 can be used to boost the output power of a radio or cassette unit to 2 watts R.M.S. if a 12 volt supply is used, or 8 watts with a 24 volt supply. The noise and distortion levels of the amplifier are extremely low, and the audio quality obtained is largely dependant on the quality of the signal source and the loudspeaker used in the unit.

The circuit is very much the same as the one used for the guitar amplifier described in the previous section of this book, but in this application a lower level of voltage gain is required, and a lower input impedance is also acceptable. The values of R2 and R5 have therefore been chosen to give a voltage gain of 20dB. (10 times) and an input impedance of 4.7k. The input impedance is shunted somewhat by preset input

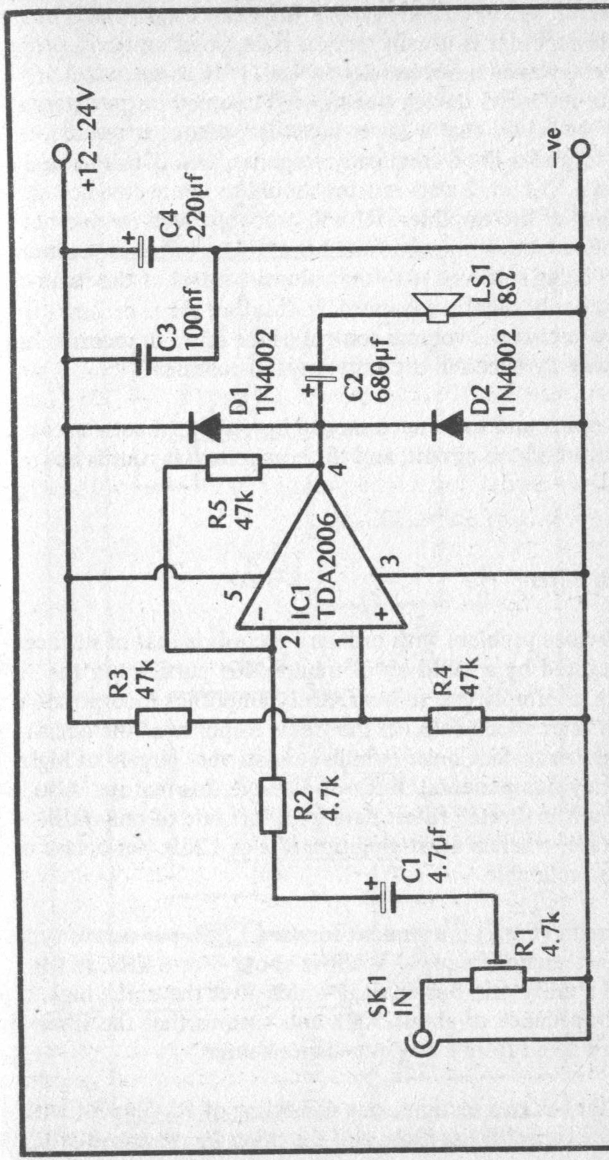

Fig. 6. The circuit diagram of the cassette/radio booster

attenuator R1, but as the output impedance of a radio or cassette recorder is usually only a fraction of an ohm, the input impedance is not usually critical. With some radios or cassette recorders having transformer coupled output stages it may be found that a lower input impedance is needed in order to give a good frequency response, and if this should happen a 15 ohm 2 watt resistor should be connected across the input of the amplifier. R1 will probably need to be set at full volume, but it may be found that this results in maximum output being achieved with the volume control of the radio or recorder only slightly advanced. In this case R1 is backed off slightly so that the volume control of the radio or recorder has to be well advanced in order to give full volume.

It was not found to be necessary to have a Zobel network at the output of this circuit, and the component layout is not critical.

Scratch Filter

One obvious problem with ordinary records is that of surface noise caused by a build up of minute dirt particles in the grooves, or simply due to wear. Many amplifiers incorporate a scratch filter which rolls off the treble response of the circuit and reduces surface noise (which consists very largely of high frequency components), but not all have this feature. Also, some built in scratch filters have a roll-off rate of only 6dB. per octave, whereas an attenuation rate of 12dB. per octave or more is preferable.

This circuit (Fig.7) is a straight forward 12dB. per octave type which attenuates frequencies above about 5 or 6 kHz. IC1 is used as a unity gain buffer stage which gives the unit a high input impedance of about 500k and ensures that the filter circuitry is fed from a low impedance source.

The filter has two sections, one consisting of R3 plus R4 with C4 as the capacitive section, and the other is comprised of R3

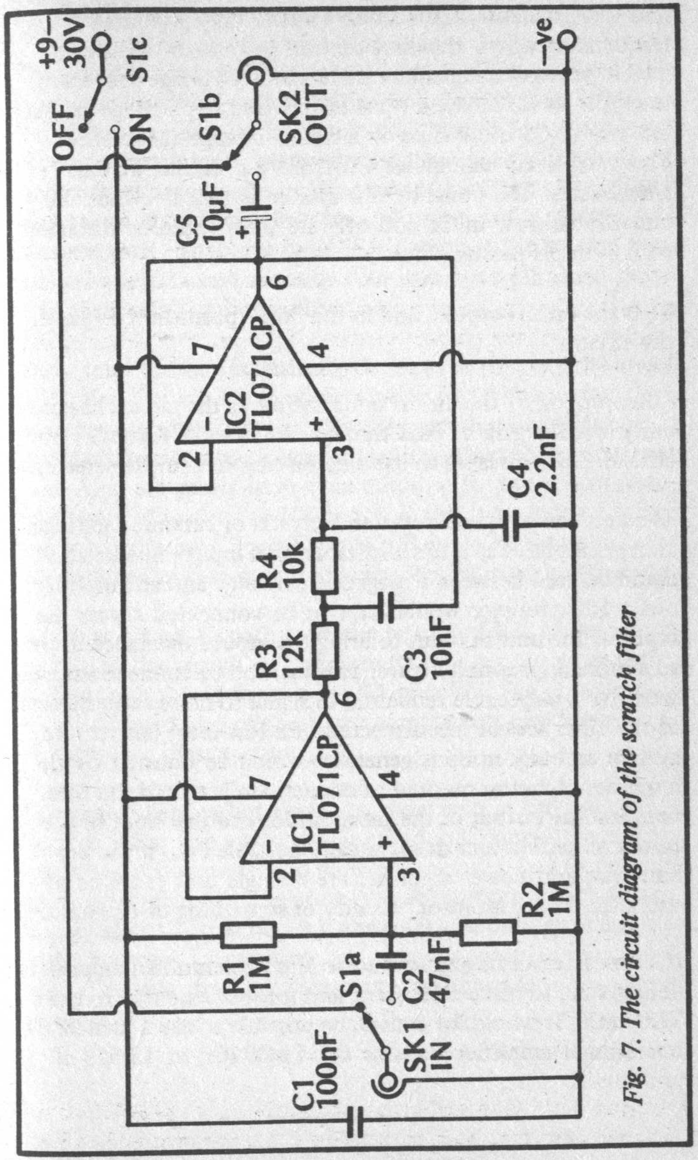

Fig. 7. The circuit diagram of the scratch filter

and C3. However, the second section only operates at frequencies where the first section introduces significant losses. At other frequencies there is unity voltage gain from the input at R3 to the output of IC1, and any voltage gain at one end of C3 is matched by a similar change at the other. This effectively eliminates C3 from the circuit at pass frequencies. The point of this system is that it avoids the undesirable slow initial roll off rate that would be obtained with a simple passive filter.

S1 is the on/off switch, and in the "off" position it bypasses the circuit.

Construction of the unit is quite simple as the circuit has only unity voltage gain at pass frequencies and the layout is not critical. It is advisable to use a metal case to provide screening.

The unit can be fitted between a crystal or ceramic cartridge and an amplifier as it has a suitably high input impedance. It could be used between a magnetic cartridge and an amplifier, but a 100k resistor would have to be connected across the input of the unit in order to bring the input impedance down to a more appropriate figure. This method of connection could well give a noticeable reduction in signal to noise ratio though, as the filter would be connected in a low level part of the system and any noise is generates would be boosted by the amplifier. A better method of connection is to add the filter between the output of the preamplifier and the input of the power amplifier, and this is usually possible even if the pre-amplifier and power amplifier are a single unit (such as by using the "Tape Monitor" facility or something of this nature).

IC1 and IC2 can be any low noise JFET operational amplifiers (TLO81CP, LF351, or similar), and it is not essential to use a TLO71CP. It would, of course, be possible to use a dual JFET operational amplifier such as the TLO82CP or LF353 if preferred.

Rumble Filter

The quality of most modern record decks is such that rumble from the turntable and motor is not usually a problem. Records are a different matter, and low frequency noises often occur due to record warps or bad pressings. These low frequency noises can be greatly reduced using a rumble filter which rolls off the low frequency response of the system. As is the case with any simple noise filter, some of the wanted signal is also lost, but using a rumble filter can often give a subjective improvement in audio quality.

The circuit of Fig.8 is basically the same as that of the Scratch Filter described in the previous section of this book, but the resistors and capacitors of the filter sections have been swopped over to give a high pass action rather than a low pass one. Also the values have been adjusted to give a turn over frequency of just under 100 Hertz. If this is considered to be too high, altering C3 and C4 to 47nF and 470nF respectively will reduce the turn-over frequency to a little under 50 Hertz.

As there is no D.C. path from the output of IC1 to the input of IC2, in this circuit IC2 requires its own biasing potential divider, and this is formed by R4 plus R5. The combined (parallel) resistance of these two components also forms one element of a filter section.

The notes given earlier about the construction and use of the Scratch Filter also apply to this circuit. Of course, for stereo operation it is necessary to build two circuits, one for use in each stereo channel. The current consumption of each filter is only about 3 to 4mA., and a small 9 volt battery makes a suitable power source, although a supply potential of up to 30 volts (36 volts absolute maximum) can be used.

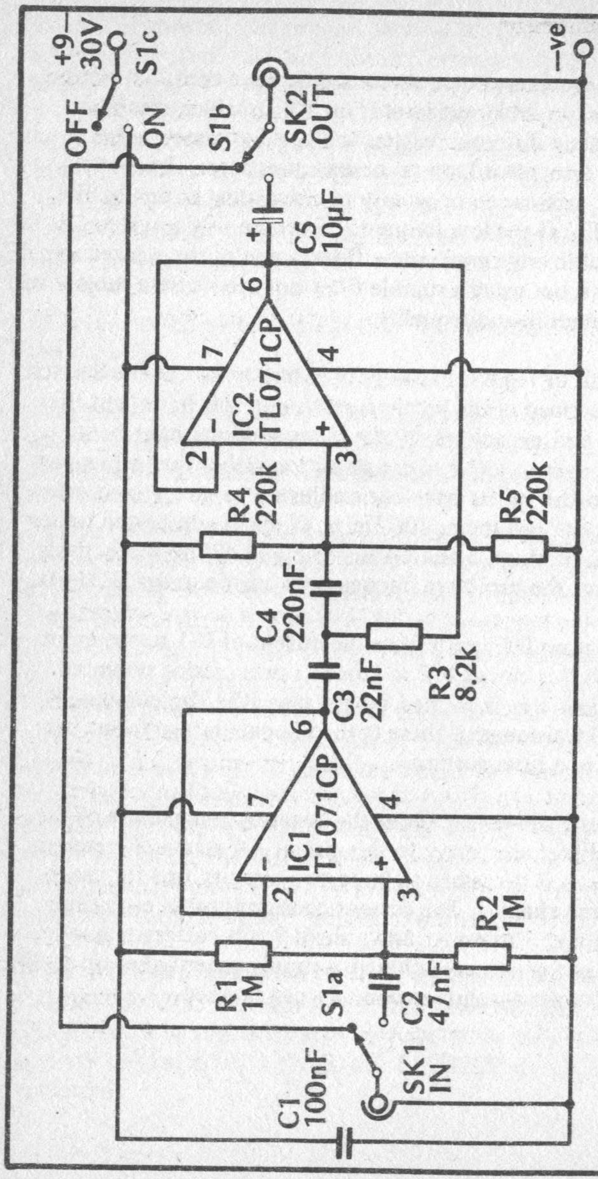

Fig. 8. The circuit diagram of the rumble filter

Audio Compressor

An audio compressor is a device which has a constant voltage gain at low input signal levels, but a gain which steadily diminishes as the input signal is increased above some threshold level. This limits the maximum output level of the unit, but without causing serious distortion as would be generated if a simple clipping type limiter was to be used. Units of this type are used in tape recording to prevent an excessive recording level, in radio communications to ensure a high modulation level, and in similar applications.

The circuit of Fig.9 is based on a MC3340P electronic attenuator device, and R2 biases the control input of this to give unit voltage gain from the input to the output of the circuit. Some of IC1's output signal is coupled by C4 to an operational amplifier which is used in the inverting mode. This amplifies the signal by an amount which is adjustable by means of R7 from a minimum of less than unity to a maximum of about 45 times. The amplified output signal is rectified and smoothed by D1, D2, and C5 to give a negative D.C. bias that is fed to Tr1's base via R3.

At low input signal levels the bias voltage is not large enough to bias Tr1 into conduction, but above a certain level Tr1 starts to conduct and increases the bias voltage fed to the control terminal of IC1 and reduces the gain of this device. This gives the required limiting effect, and on the prototype (with R7 at about 250k) a voltage gain of unity was obtained with inputs of up to 40mV. R.M.S., and increasing the input to 500mV. R.M.S. gave an increase in the output of just 10mV. R.M.S.; 500mV. R.M.S. is the maximum input level the circuit can handle without causing severe distortion on the output signal. The threshold level can be raised by reducing the value of R7, or lowered by increasing the value of R7.

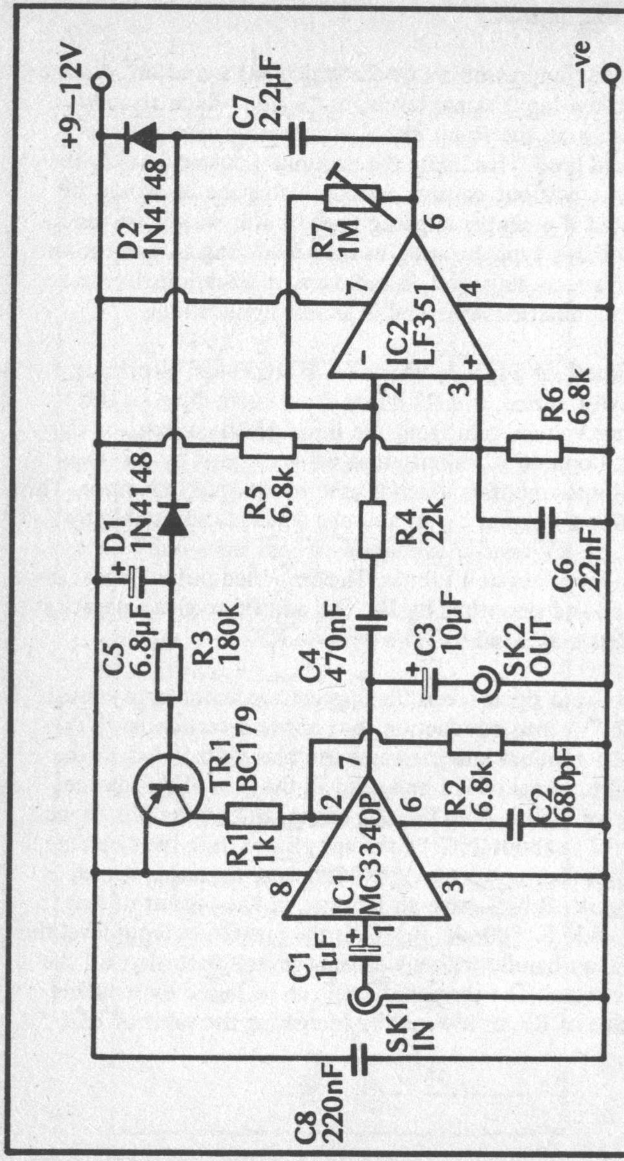

Fig. 9. The circuit diagram of the audio compressor

Volume Expander

A volume expander has an action that is the opposite of that provided by a compressor circuit, with the voltage gain of the unit rising somewhat as the input signal amplitude is raised. Units of this type are used in hi-fi systems to increase the dynamic range of a signal that is lacking in this respect. It is only possible to increase the apparent dynamic range by about 10dB. or so as a larger amount would probably result in the action of the unit being apparent to the listener.

Like the previous circuit, this one (see Fig.10) is based on the MC3340P electronic attenuator device, and the control input is biased by R1 to give IC1 a voltage gain of about unity. Also as in the previous circuit, an operational amplifier is used in the inverting mode, and here it is fed with the input signal. C8 couples some of the output of the amplifier to a rectifier and smoothing circuit that gives a positive D.C. bias, and it is referenced to the negative supply rail.

If the input signal is sufficiently large, the bias signal will bias Tr1 into conduction due to the base current that flows through R2. This reduces the control voltage fed to IC1 and boosts the gain of this device. Increasing the input signal level causes Tr1 to conduct more heavily and further boost the gain of IC1 until the saturation point of IC1 is reached and it is at maximum gain. This occurs when the voltage gain has been boosted by the required amount of about 10dB.

R6 determines the level at which full expansion is achieved, and this should be set so that full expansion is only just produced at the highest dynamic levels, otherwise the action of the unit may well become obvious, giving unsatisfactory results.

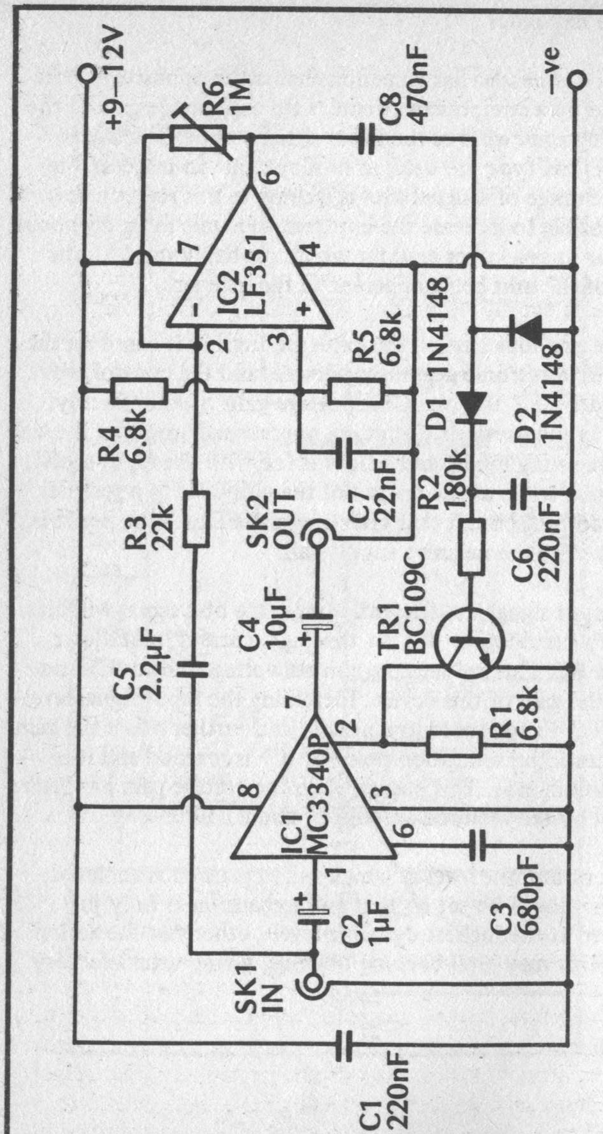

Fig. 10. The circuit diagram of the volume expander

Noise Gate

A noise gate is a device which cuts off or severely attenuates the signal being processed unless it is above a certain dynamic level. This type of circuit is used in communications systems, public address systems, etc., to give a silent background level during pauses in the wanted signal, and is primarily of benefit where there is a substantial amount of background noise.

The volume expander of Fig.10 can be modified to act as a noise gate simply by raising the value of R1, or simply removing it altogether, depending upon whether the signal is to be attenuated or cut off during pauses. R6 then sets the signal level at which the attenuation of the unit is removed.

With either version of the circuit the input signal should be kept to 500mV. R.M.S. or less in order to avoid severe distortion. The current consumption of the unit is approximately 10mA.

Car Radio Booster

The output power of most car radios and cassette players is fairly modest, usually being just a few watts. This is probably due to the difficulty in obtaining higher output powers using modern transformerless output stages, a supply potential of just 12 volts, and an ordinary loudspeaker having an impedance in the range of 4 – 8 ohms.

This booster amplifier (Fig.11) can be used to increase the output power of a car radio or cassette unit to 9 watts R.M.S. into an 8 ohm loudspeaker, or 18 watts R.M.S. into a 4 ohm unit. This comparatively high output level is obtained by using a bridge amplifier circuit, and this really consists of two power amplifier stages having anti-phase outputs. The loudspeaker is driven from between the outputs of the amplifier stages, and neither speaker lead connects to the earth rail. This method gives a high output power because one output is fully

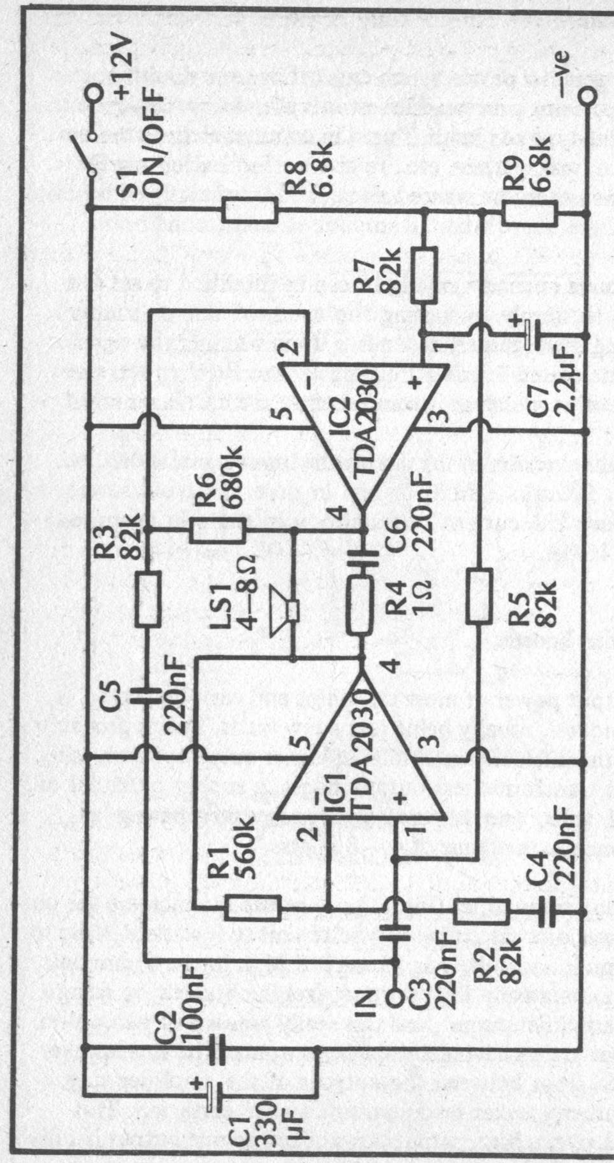

Fig. 11. The circuit diagram of the car radio booster

positive when the other is fully negative, and vice versa. In theory the peak signal voltage is equal to the supply potential, and the peak to peak output signal voltage is double the supply potential. In practice there will always be losses through the output stages, but this method is nevertheless capable of giving a high output power for a given supply voltage and speaker impedance.

In this circuit IC1 is used as a non-inverting amplifier, and IC2 is used as an inverting amplifier having a similar voltage gain (so that the two outputs are driven to the clipping threshold more or less in unison, and the optimum output power is attained). The two inputs are simply connected together and the speaker is fed direct from the two outputs. There is only a very low quiescent voltage across the two outputs and a coupling capacitor here would be superfluous.

As with any amplifier of fairly high output power, the component layout needs to be designed with reasonable care to avoid stray feedback and possible instability. The two integrated circuits should be bolted to a substantial heatsink, and note that their heat-tabs connect internally to their negative supply terminals.

L.E.D. Vu Meter

Probably most readers will be familiar with L.E.D. Vu meters which are faster acting and more accurate than most types which incorporate a moving coil meter. Circuits of this type are very easy to produce these days due to the availability of L.E.D. bargraph driver I.C.s. These drive a series of L.E.D.s, and as the input voltage to the device is raised first one L.E.D. switches on, then a second as well, then a third, and so on, giving a sort of L.E.D. voltmeter. Some of these devices, including the LM3915N device used in this circuit (Fig.12) have a logarithmic scale so that they provide ideal scaling for a Vu meter. The LM3915N can drive up to ten L.E.D.s, although only seven are used in this circuit. There is a 3dB. gap

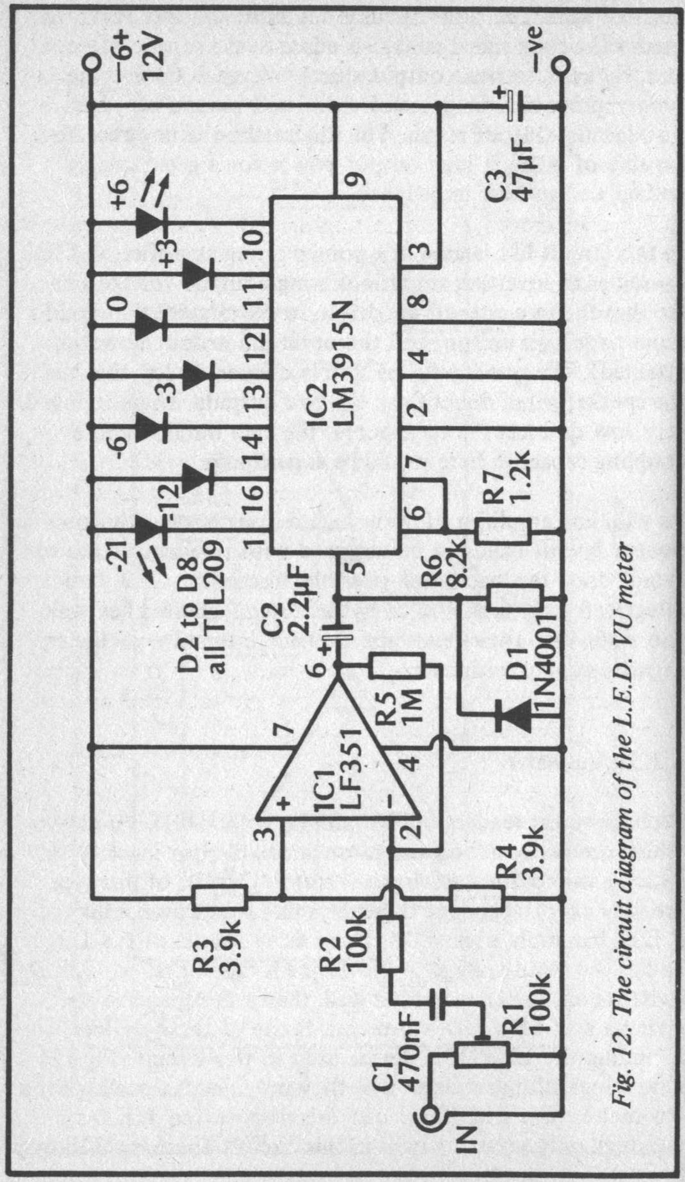

Fig. 12. The circuit diagram of the L.E.D. VU meter

between the L.E.D. threshold levels, although there are larger gaps between some L.E.D.s in this circuit as some of the L.E.D.s at the lower end of the scale were felt to be unnecessary and were therefore omitted. The three missing L.E.D.s can be added if desired, and would be driven from pins 18, 17, and 15 of IC2.

IC1 is used as a simple audio amplifier having a voltage gain of 20dB. (10 times), and this is needed to boost the sensitivity and input impedance of the circuit so that it can be driven from any likely source. R1 enables the sensitivity of the unit to be adjusted to the correct level. C2 couples the output signal of IC1 to the input of IC2, and R6 is used to bias the latter to the negative supply rail. IC2 responds to positive half cycles from IC1, and switches on the appropriate number of L.E.D.s (D2 to D8) for the input voltage present at that instant. The circuit is thus a peak reading rather than an average reading type, and it gives good accuracy even on waveforms having a high peak but low average amplitude.

Construction of the unit is quite straight forward and the component layout is not critical. The quiescent current consumption of the unit is only about 9mA., but it rises to over three times this figure with high input levels as the average L.E.D. current is about 5mA. per L.E.D.

In order to calibrate the unit it is first necessary to apply a signal to the main equipment that gives a 0dB. signal level. R1 is then adjusted for the lowest sensitivity that causes the 0dB. L.E.D. of the unit to glow reasonably brightly.

CHAPTER 2

TEST GEAR CIRCUITS

Signal Injector

A signal injector is a device which generates an audio frequency signal which is rich in harmonics that extend well into the radio frequency spectrum. This enables such a unit to be used for most A.F., I.F., and R.F. testing. The basic technique for testing equipment using a signal injector is to first apply the signal to the loudspeaker or earphone, then to the input of the output stage, then to the input of the driver stage, and so on through to the input of the equipment. If the tone is produced from the loudspeaker or earphone after a test, and at the expected volume level, then the stages through which the test signal is passing are obviously functioning. When eventually a test is made and the output tone is absent, the fault lies in the circuitry between this test point and the previous one (or at least, in this general part of the equipment under test). Component tests and voltage measurements are then used to locate the precise fault.

The circuit of the signal injector appears in Fig.13, and is a simple oscillator based on an operational amplifier. R1 and R2 bias the non-inverting input of the device while R3 biases the inverting input from the output of the device. This gives 100% D.C. negative feedback, but the decoupling effect of C2 gives greatly reduced feedback and high voltage gain at audio frequencies. C3 provides positive feedback over the circuit, and due to the high voltage gain of the amplifier this produces strong oscillation and a roughly squarewave output which contains the required high frequency harmonics. The fundamental operating frequency is approximately 800 Hertz.

VR1 is the output level control, and this can be ganged with on/off switch S1 if desired. When making R.F. tests it will usually only be necessary to loosely couple the output of the

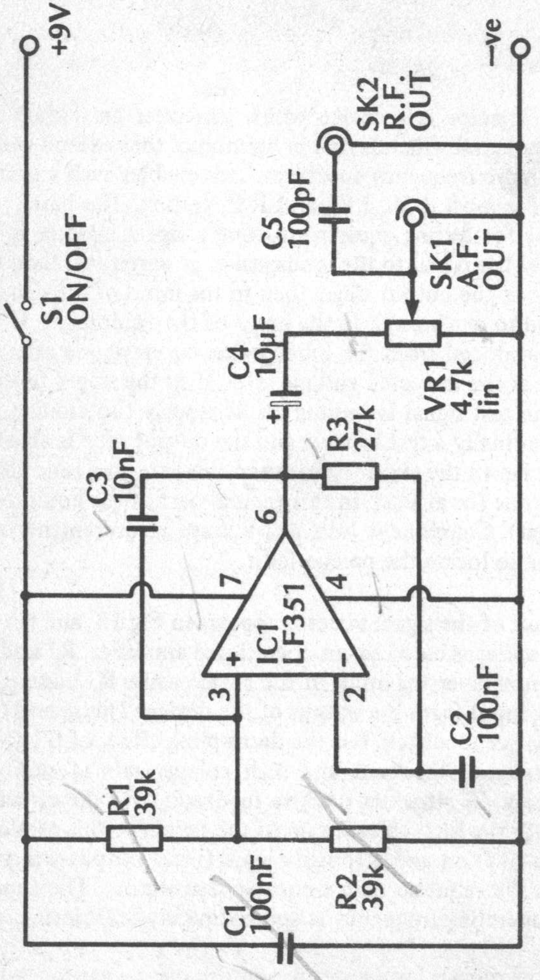

Fig.13. The circuit diagram of the signal injector

unit into the equipment under test, and this can be achieved by taking the output from SK2 so that the output signal is filtered by C5.

The unit should give no constructional difficulties as it is very simple and the component layout is not at all critical.

Signal Tracer

Like the signal injector described in the previous section of this book, a signal tracer is used to locate the general area in which a fault lies in a piece of radio or audio equipment (plus a few other types of equipment). The method of fault finding is slightly different though, in that the first test is made at the input of the unit under test, and then subsequent tests are made at points which progress towards the output. As before, the tests are made until the signal ceases, and the fault lies in the circuitry around or immediately preceding the point at which the last test was made. Of course, this method can only work if a suitable input signal is applied to the equipment being checked, and this signal can either be from a signal generator of some kind or an ordinary programme source for the equipment.

The circuit of the signal tracer is shown in Fig.14, and the unit breaks down into two amplifying sections. The first one uses Tr1 as a common emitter amplifier having aubstantial amount of negative feedback introduced by R3. This lowers the gain of the input stage to only about 20dB. (10 times), but gives a high input impedance of a few hundred kilohms so that the unit only lightly loads the circuit under investigation. C2 is an R.F. filter capacitor, and Tr1 acts as a sort of crude A.M. demodulator if an R.F. signal is fed to the input. This enables the unit to trace both A.F., and A.M. R.F. signals without the need for a separate R.F. probe or any A.F./R.F. switching.

IC1 is an oprational amplifier used as a fairly straight forward inverting amplifier. VR1 controls the amount of negative

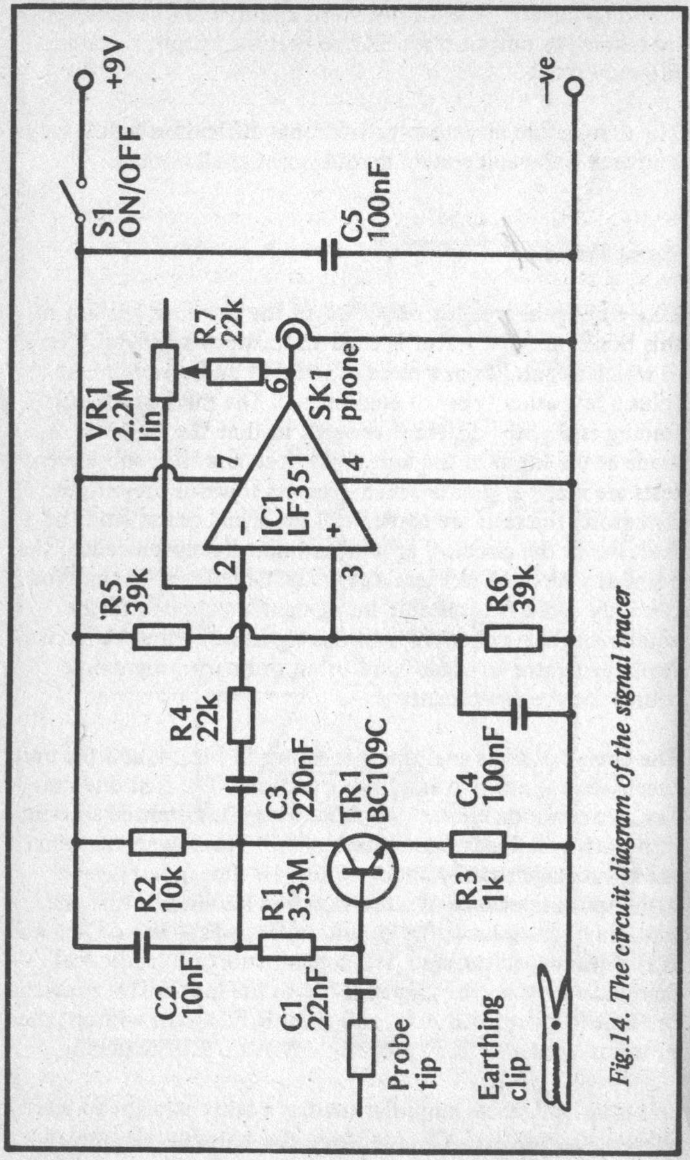

Fig.14. The circuit diagram of the signal tracer

feedback applied over IC1 and therefore controls the voltage gain of the amplifier. This varies from unity with VR1 at minimum resistance to just over 40dB. (100 times) with VR1 at maximum resistance. It is useful to be able to reduce the gain of the unit as this can be used to avoid overloading the unit when tracing higher level signals.

The output is coupled to a crystal earphone (no other type is suitable) and the high maximum gain of the unit coupled with the high sensitivity of a crystal earphone enables signals of very low amplitude to be detected. The unit can therefore be used for A.F., I.F., and often for R.F. signal tracing. R.F. signals in excess of 50 MHz can be detected by the unit.

Units of this type are usually constructed in a small plastic case with a probe at the front, and the latter can simply be a long bolt secured to the case. This avoids using a long screened cable at the input of the unit, which would greatly reduce the input impedance at radio frequencies, probably giving rise to unwanted loading effects. In other respects construction is quite normal, but the unit has a high level of voltage gain and the component layout needs to be designed with appropriate care.

Zener Diode Tester

Most series of zener diodes are available in values up to about 33 volts, and it obviously necessary to have a supply voltage slightly in excess of this figure in order to test any diode from such a series. There are several ways of obtaining this voltage, such as a mains power supply, high voltage batteries, or from low voltage batteries via a voltage step-up circuit. The last method is the one employed in this unit, the circuit of which is given in Fig.15.

IC1 is used as an oscillator producing a squarewave output at a frequency of about 100 Hertz or so. The output of IC1 is coupled by current limiting resistor R5 to an emitter follower

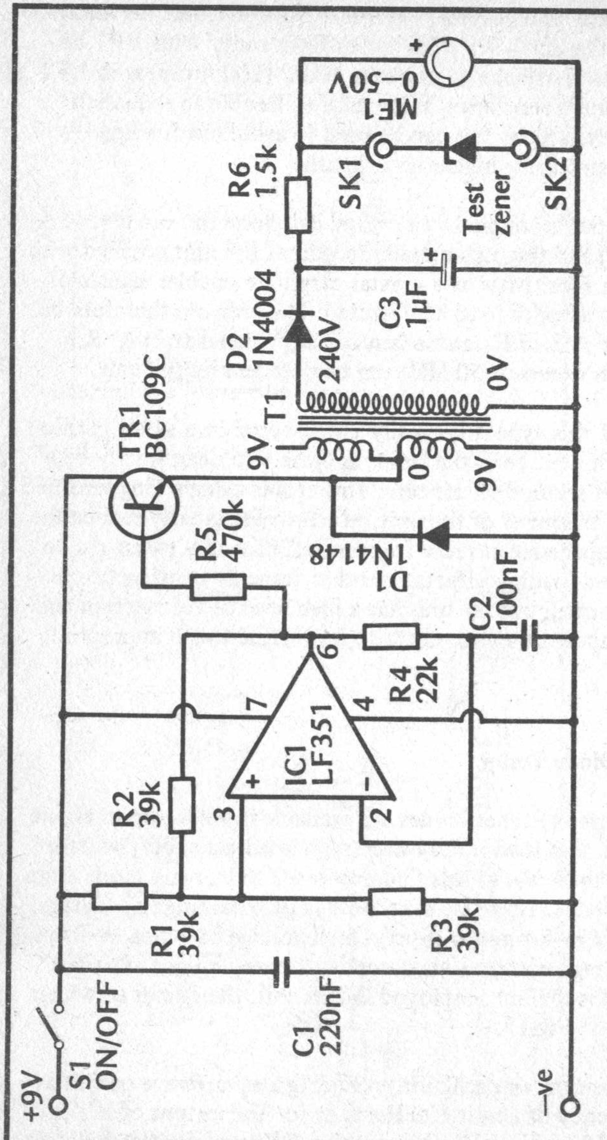

Fig. 15. The circuit diagram of the zener diode tester

buffer stage (Tr1). The latter drives the secondary winding of a 9V – 0V – 9V 75mA. mains transformer, but only half the secondary winding is used here. Also, the transformer is being used in reverse so that it provides the required voltage step-up, and the secondary winding is really the primary winding in this circuit.

The output from the 240 volt winding of T1 is rectified by D2 and smoothed by C3 to produce an unloaded D.C. voltage of about 42 volts. This is coupled to the zener under test by R6, and this component ensures that a high discharge current cannot flow from C3 into the test device, which would probably damage the latter. ME1 measures the zener voltage of the test device, and a multimeter switched to a suitable D.C. voltage range can be used here. Alternatively a 50 μA. meter in series with a 1M 1% resistor is suitable. As most zener voltages are quoted at a zener current of 5mA. and the output current of the unit is somewhat less than this figure, readings will be marginally on the low side (but only very marginally).

Construction of the unit should present no difficulties. SK1 and SK2 can be 1mm. panel sockets mounted about 20mm. apart on the front panel of the unit. Zener diodes should readily plug into these, but be sure to connect them with the correct polarity or a low reading of only about 0.7 volts will be obtained.

Op. Amp. Power Supply

Many circuits that are built around operational amplifiers require dual balanced power supplies, usually of 12 volts each. The power supply circuit of Fig.16 can be used to power equipment of this type provided it does not consume more than 100mA. per supply rail. The unit is primarily intended for use as a bench power supply for use when developing and testing operational amplifier based circuits, but it can be used as the power supply for a final piece of equipment.

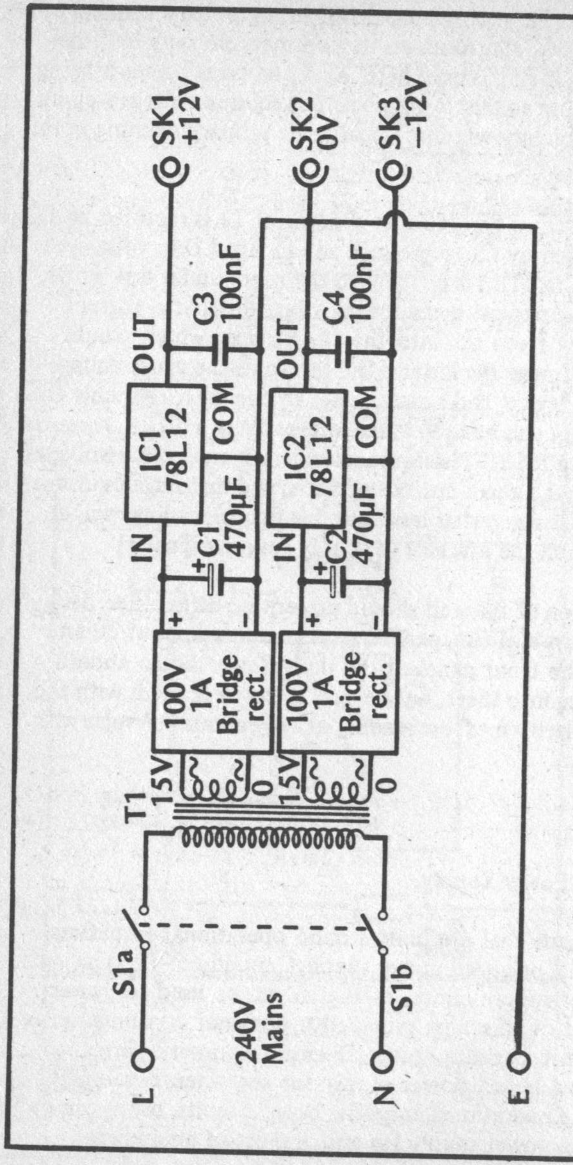

Fig. 16. The circuit diagram of the op. amp. power supply

The mains is applied to the primary winding of step-down transformer T1 by way of on/off switch S1. T1 has dual 15 volt secondary windings having a current rating of at least 100mA. each. The two secondary windings have their outputs fullwave rectified by separate bridge rectifiers, and then smoothed by separate smoothing capacitors (C1 and C2). Each of the supplies is then regulated by a small 12 volt monolithic voltage regulator device (IC1 and IC2), and the two regulated outputs are connected in series to give the required dual balanced supplies. C3 and C4 are decoupling capacitors that prevent IC1 and IC2 from becoming unstable.

Construction of the unit is quite simple, but bear in mind that the unit is mains powered and the normal safety precautions should be observed. Any exposed metalwork must be earthed to the mains earth lead, as should the 0V. output of the unit. The unit should be housed in a case which has a screw-on lid, and not one that can simply be unclipped to reveal dangerous mains wiring.

IC1 and IC2 have output current limiting that protects the unit if the output is inadvertently short circuited.

Supply Splitter

An alternative method of obtainig dual balanced supplies is to use a supply splitter that gives a low impedance centre-tap on a single supply, rather than generating two equal supplies and connecting them in series as was done in the previous project. A supply splitter such as the one shown in the circuit diagram of Fig.17 is intended for use with an existing bench power supply, and is not an independent unit. This particular circuit will work well with supply voltages of between 15 and 30 volts, and each of the output rails has a potential equal to half the input voltage (thus 30 volts is needed at the input to give dual 15 volt outputs for example). The unit will readily handle currents of up to 100mA. and it should not be necessary to fit Tr3 and Tr4 with heatsinks.

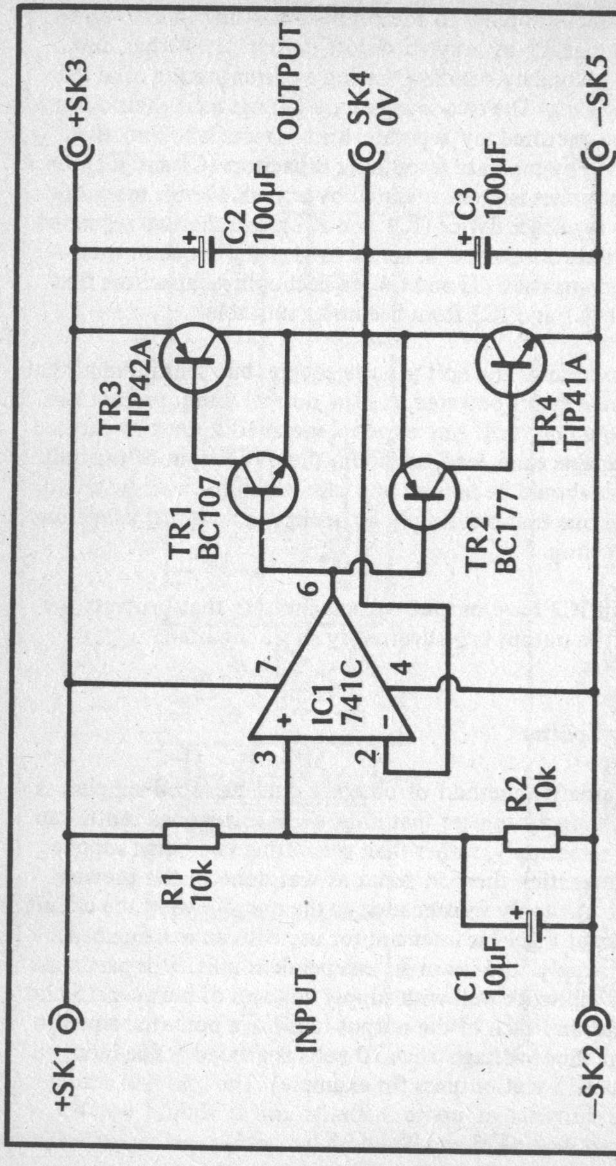

Fig.17. The circuit diagram of the supply splitter

The circuit is really just a unity gain amplifier having a high input impedance and a low impedance class B output stage. A class B output stage is used as this gives the unit a low quiescent current consumption (only about 2mA.). Tr1 and Tr3 are common emitter amplifiers, but as there is 100% negative feedback from Tr3's collector to Tr1's emitter these give only unit gain. Tr2 and Tr4 form a complementary output pair for Tr1 and Tr3. There is 100% negative feedback from the output of the unit (Tr3 – Tr4 collectors) to the inverting input of IC1, and this gives unity voltage gain from the non-inverting input of IC1 to the output. R1 and R2 bias the non-inverting input of IC1 to half the supply potential, giving the same voltage at the output, but at a very low impedance. The three capacitors are needed to maintain stability.

Capacitance Meter

This capacitance meter has four measuring ranges of 1nf, 10nf, 100nf, and 1μf f.s.d., and the scaling is linear on all four ranges. The circuit diagram of the unit is shown in Fig.18, and this uses a conventional arrangement with an astable multivibrator driving a monostable type. The astable is based on IC1 which is a standard 555 type oscillator. The monostable uses IC2 in the standard 555 monostable configuration. R8 and D1 form a simple zener shunt stabiliser circuit, and these give stabilised 5.6 volt pulses from IC2. R9 and ME1 form a simple voltmeter circuit, and this responds to the average output voltage of IC2.

The frequency of the output pulses from IC2 depends upon the frequency at which IC1 oscillates, since the output of IC1 consists of a series of short negative pulses which are used to trigger IC2. The length of the output pulses depends on the value of the capacitor under test (which is used as the timing capacitor for IC2), and the timing resistance value selected using S1. The length of the output pulses is proportional to the value of both timing components.

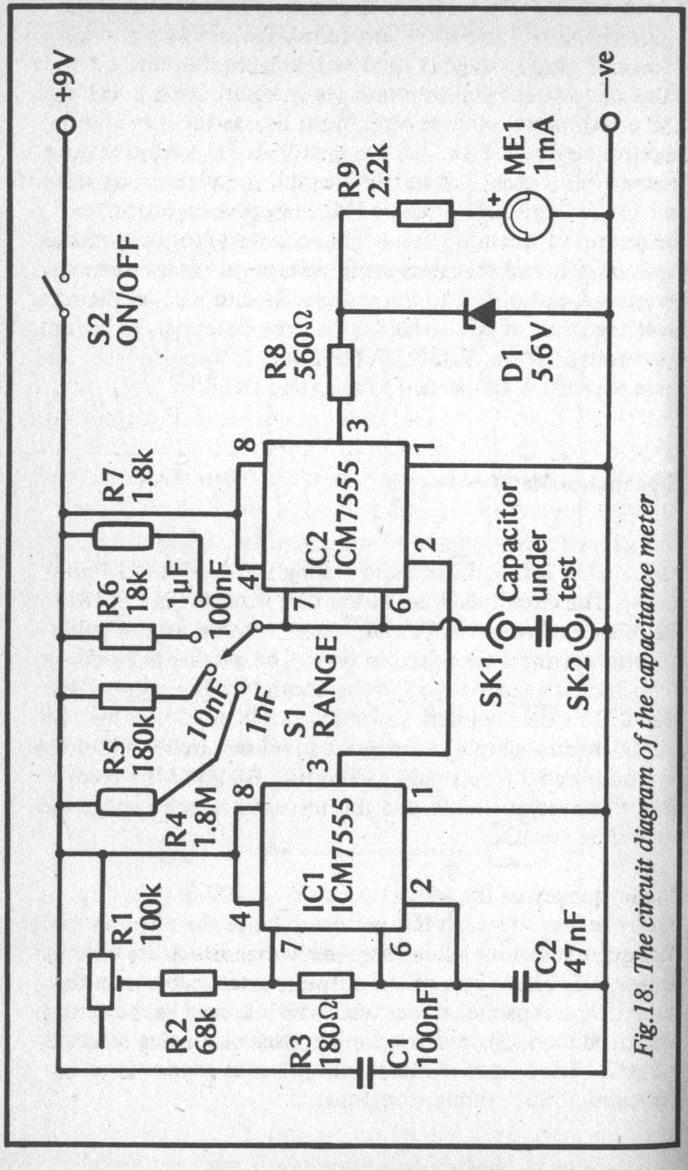

Fig.18. The circuit diagram of the capacitance meter

In practice R1 is adjusted so that with a capacitor of 1nf in value connected across the test terminals, and S1 set to the 1nf position, the circuit gives full scale deflection of ME1. A lower value of test capacitance gives a shorter output pulse length, and a proportionate reduction in the average output voltage (and reading on ME1). The required linear scale capacitance meter action is thus obtained. Switching S1 to the 10nf, 100nf, and 1µF ranges causes the timing resistance to be reduced by a factor of ten at each new setting, so that the test capacitance needs to be ten times greater in order to produce a given meter reading. This gives four measuring ranges, but R4 to R7 must be close tolerance (2% or better) types in order to give consistent accurancy on all ranges.

CMOS 555 devices are used in the circuit as these have a much lower supply current drain, and a lower self capacitance so that slightly better accuracy is obtained when measuring low value components.

The layout of the components is not critical, but you should obviously arrange the circuit to give a low level of stray capacitance across the test sockets. The circuit can be calibrated on any range using a capacitor having a close tolerance and a value equal to the f.s.d. value for that range. With the test capacitor connected into circuit R1 is adjusted for precisely full scale deflection (but make sure that the meter is properly zeroed mechanically before calibrating the unit). The unit is then ready for use.

Audio Frequency Meter

Although for really precise frequency measurements a simple analogue frequency meter is inadequate, for audio frequency measurements a unit of this type is often more than adequate. The circuit shown here (Fig.19) has four measuring ranges of 0.1, 1, 10, and 100 kHz f.s.d. and the scaling is linear on all ranges.

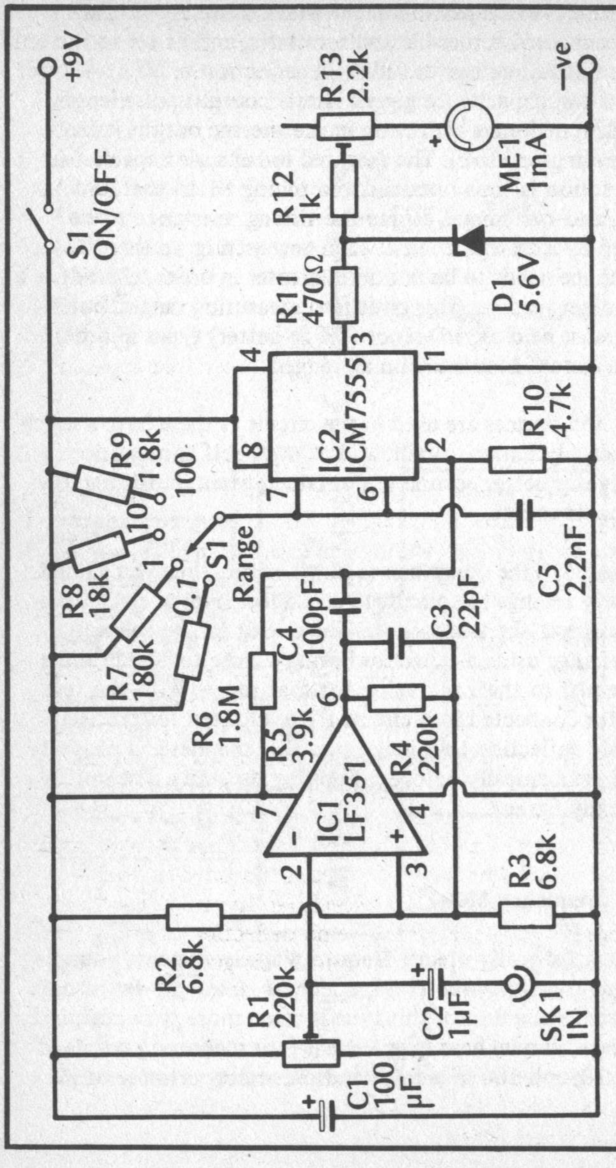

Fig. 19 The circuit diagram of the audio frequency meter

This is really an adaptation of the previous circuit. IC2 is a monostable which provides pulses of fixed duration, and its average output voltage therefore depends on the frequency of the signal applied to its trigger input, provided the input frequency is not so high as to cause the output pulses to effectively overlap one another. S1 and R6 to R9 give four switched output pulse lengths, and this gives the unit its four ranges since the shorter the pulse length, the greater the number of pulses (and hence input frequency) needed to produce full scale deflection of ME1. R6 to R9 should all be close tolerance components. R13 enables the unit to be calibrated against a signal of known frequency, and suitable signal sources include calibrated signal generators and some calibration oscillators.

R5 and R10 bias the trigger input of IC2 just above the 1/3V threshold level which this terminal must be taken below in order to trigger the monostable. IC1 is used in a simple Schmitt trigger circuit and this gives a rectangular output waveform regardless of the input waveform. C4 couples the output of IC1 to the trigger input of IC2, and this component has purposely been given a low value so that only very brief input pulses are applied to IC2. This is necessary because the monostable is a type where the output pulse is lengthened by the input pulse, and a significant lengthening of the output pulses would upset the proper operation of the circuit, especially on the higher ranges. R1 is adjusted by trial and error to obtain good sensitivity and reliable operation. When correctly set the unit has an input impedance of over 100k and only about 200mV. peak to peak is needed in order to operate the unit. Lower input levels fail to produce any output from IC1, and therefore give no deflection of ME1. This avoids the possibility of low input levels giving misleading readings.

Construction of the unit is quite straightforward, and the component layout is not especially critical, although some care should be taken to avoid stray feedback and possible consequent instability.

Fuse Checker

This circuit, which is shown in Fig.20, is extremely simple and inexpensive to build, but is no less useful because of this. If a serviceable fuse is placed in one of the fuseholders it completes the circuit so that a current flows through D1 and it lights up to indicate that the fuse is usable. If the fuse is a dud and has been blown it will fail to complete the circuit, and D1 will fail to light up, indicating that the fuse is unusable. Three fuseholders of different sizes are provided so that the unit is easy to use with any normal type of fuse.

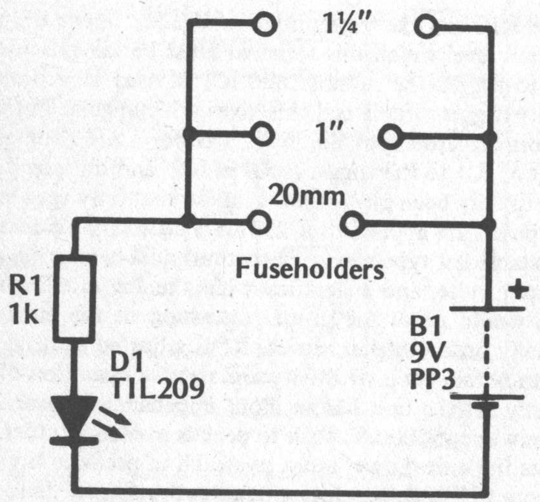

Fig. 20 The circuit diagram of the fuse checker

Resistance Meter

Most if not all multimeters have resistance measuring ranges, but on many instruments there are only two ranges, and the simple resistance meter circuits used in multimeters have reverse reading non-linear scales. This can make them difficult to use and also gives relatively low accuracy on occasions.

The resistance meter circuit shown in Fig.21 has five measuring ranges with full scale values of 1k, 10k, 100k, 1M, and 10M. The scaling is linear on all ranges and the scale is forward reading. This makes the unit easy to use and gives excellent accuracy whether measuring a resistance of just a few hundred ohms or several megohms.

The circuit uses the well known principle of feeding a constant current to the test resistor, and then measuring the voltage developed across the meter. With a constant current, the voltage developed across the test component is proportional to its resistance, and the meter can therefore be calibrated in ohms rather than volts.

Tr1 and its associated components form the constant current generator circuit, and there are five switched emitter resistors to give five currents, and the five ranges of the unit. The unit is calibrated separately on each of the ranges against a close tolerance resistor having a value equal to the full scale value of the range concerned. With the calibration resistor connected to the unit and S1 switched to the correct range, the appropriate preset is adjusted for precisely full scale deflection of ME1. Note that a push button switch of the non-locking type is used for the on/off switch, and this should not be operated until a test resistor is connected into circuit. This is simply because ME1 will be driven beyond f.s.d. with no test resistor in circuit, and while this is unlikely to damage it, this is something that should obviously be avoided as far as possible.

It is essential that the meter circuit does not take a significant

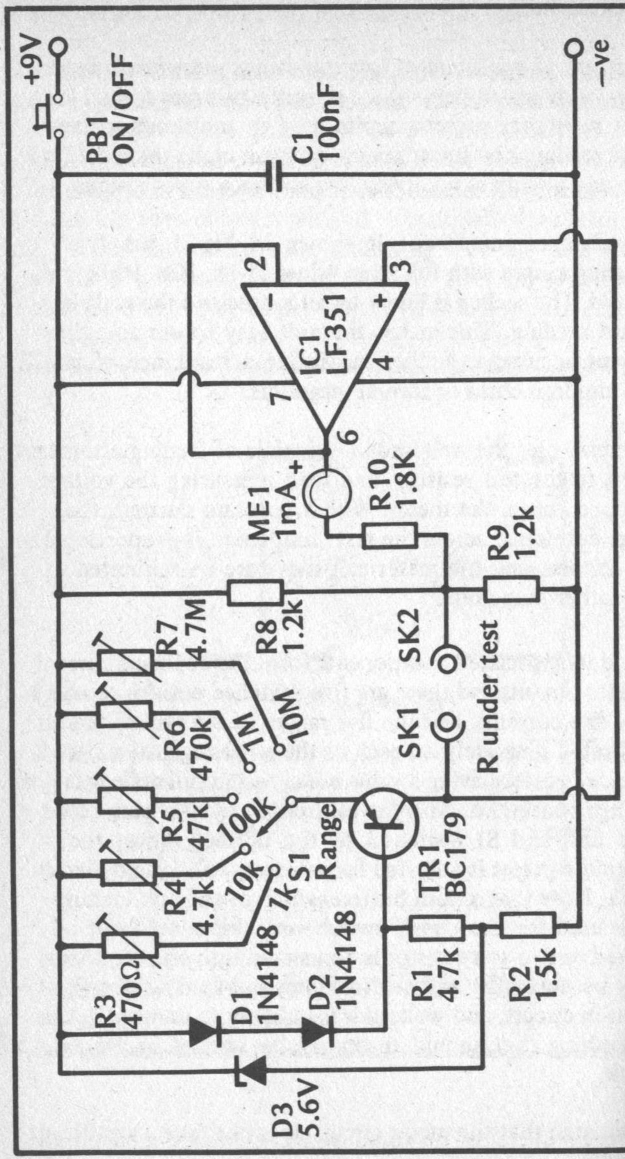

Fig. 21. The circuit diagram of the resistance meter

current as it would tap off some of the current (and a varying amount of current) from the test resistor, thus producing inaccurate results. IC1 has therefore been used as a high impedance buffer stage to prevent this from happening.

As the circuit only handles D.C. signals the layout of the unit is not critical, and construction is perfectly straight forward.

Transistor Tester

This transistor tester gives a rough indication of the current gain of the test device, and it can be built at very low cost. The circuit diagram of the unit is shown in Fig.22, and if we first consider the circuit in the n.p.n. mode, if the test device conducts hard enough to pass a collector current of about 4mA. there will be insufficient voltage across D1 to bias this component into conduction. D2 will not pass a current either as it will be reverse biased. Thus neither L.E.D. indicator lights up. If the test transistor passes a current of less than about 4mA. D1 will pass a current and will light up.

In practice VR1 is adjusted just low enough in value to extinguish D1. VR1 varies the base current fed to the test device, of course, and the base current needed to switch off D1 depends upon the current gain of the test transistor. With VR1 at maximum resistance a current gain of about 1,000 is needed to switch off D1, and this reduces to only about 10 with VR1 at minimum resistance. It is therefore possible to roughly gauge the gain of the test transistor by the setting of VR1 when it is adjusted to the switch off point.

In the p.n.p. mode S1 switches the supply polarity so that it is correct for p.n.p. devices, and D2 becomes the indicator light while it is D1 that becomes reverse biased and plays no active role in the unit. In other respects the unit operates and is used as before.

The three sockets can be 1mm. wander types grouped closely

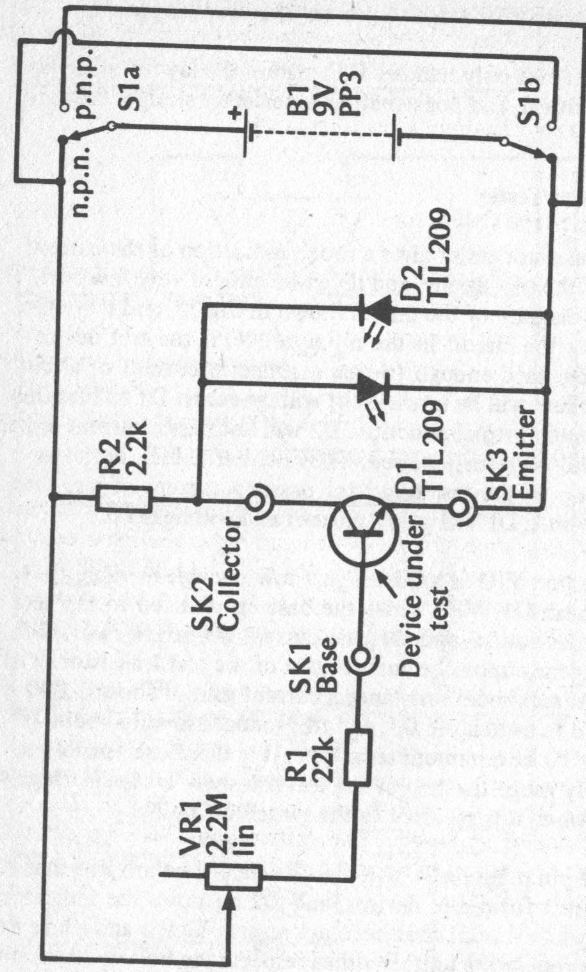

Fig. 22. The circuit diagram of the transistor tester

together on the front panel so that most test devices can be readily plugged into them. A set of test leads having 1mm. wander plugs at one end and crocodile clips at the other can be made up so that power transistors and other types that will not connect direct to the sockets can be connected to the unit. It is a good idea to use different coloured test leads so that they can easily be identified and the risk of connecting them to the wrong leadout wires is reduced.

Simple Scope Calibrator

A simple oscilloscope calibrator of the type described here simply provides a squarewave signal of known peak to peak value. The circuit diagram of the calibrator is shown in Fig.23, and this has IC1 as a standard 555 astable providing a roughly squarewave output at a frequency of about 500 Hertz. The output of IC1 is used to switch VMOS transistor Tr1 hard on and hard off at the astable operating frequency.

D1, R5, R6, and Tr2 form a simple series regulator circuit having an output voltage which can be varied from zero to about 5 volts by means of R5. R4 is a load resistor for Tr2, and R3 performs the same function for Tr1. It also couples the output voltage of the regulator to the drain of Tr1.

When Tr1 is switched off it provides a very high resistance and virtually the full output voltage of the regulator circuit is fed to the output socket (provided the output is fed to a reasonably high impedance load such as the input of an oscilloscope). When Tr1 is switched on it has a drain to source resistance of only about one or two ohms. The potential divider action across R3 and Tr1 thus gives an output voltage of practically zero.

Thus it is possible to obtain a peak to peak output amplitude of the required value simply by adjusting R5 to give a D.C. potential of this value across R4. The output is in fact marginally lower, but the error here is too low to be of

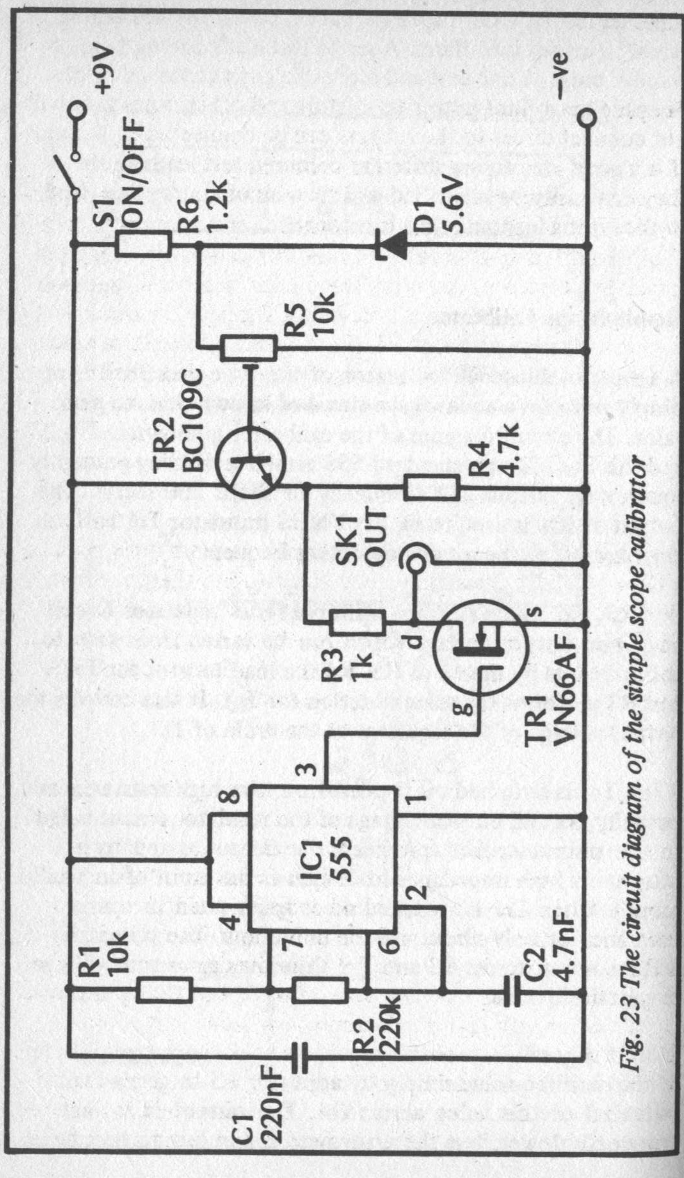

Fig. 23 The circuit diagram of the simple scope calibrator

significance.

Simple Signal Generator

If a simple signal generator for such things as frequency response measurements is all that is required, and not a sophisticated instrument suitable for distortion measurement on hi-fi equipment, a circuit based on a function generator integrated circuit is perfectly adequate. The circuit described here and shown in Fig.24 is based on the 8038CC device, and it covers a frequency range of about 30 Hertz to 30 kilohertz in three ranges (30Hz to 300Hz, 300Hz to 3 kHz, and 3kHz to 30kHz). It provides the usual sine and squarewave outputs.

The 8038CC produces a squarewave signal at a frequency which is determined by the control voltage fed to pin 8 of the device, and a capacitor connected from pin 10 to the negative rail. The control voltage can be varied by means of VR1 which is the fine frequency control, and there are three switched timing capacitors (C2 to C4) which are selected by S1. These give the unit its three frequency ranges. The mark space ratio of the signal is controlled by balance control R3, and with the output viewed on an oscilloscope this is used to trim the ratio to the normal one-to-one.

The sinewave signal is produced from the squarewave signal by a shaping circuit, and R6 is used to trim the sinewave signal to the correct waveform. R7 is the load for the squarewave output, but the sinewave output (pin 2) has an integral load. S2 selects the desired output and couples it to IC2 which is used as a buffer stage to give the unit a low output impedance. VR2 is the output level control. The maximum sinewave amplitude is about 1.5 volts peak to peak, but the squarewave signal has a much higher maximum amplitude.

The circuit requires an 18 volt supply which can be provided by two 9 volt batteries wired in series. It also seems to operate from a single 9 volt battery, albeit with reduced performance.

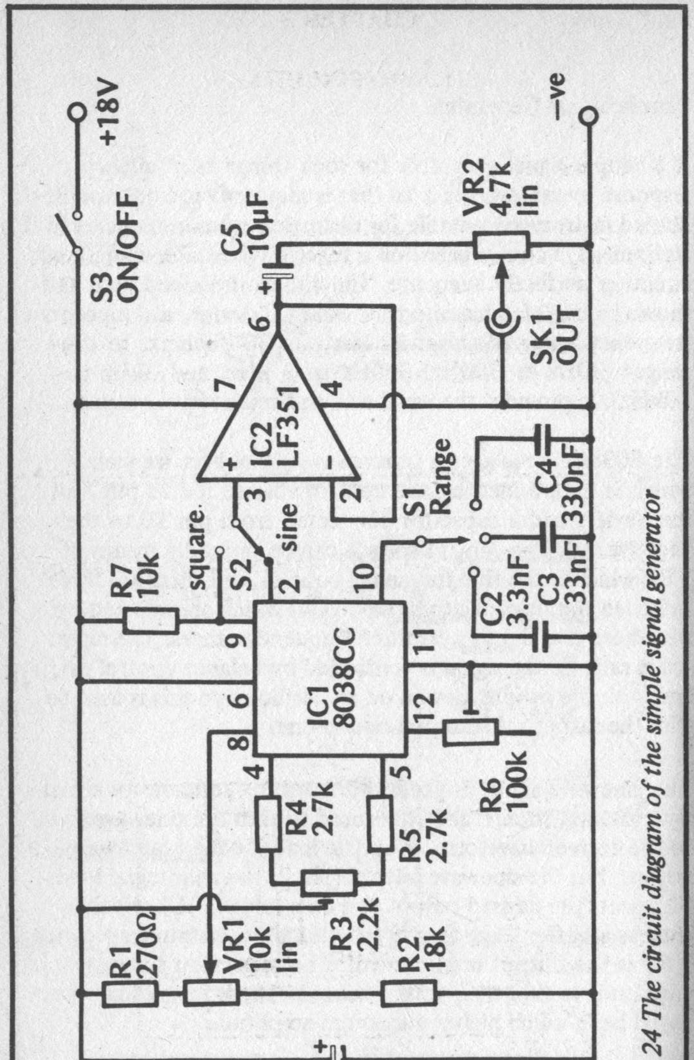

Fig. 24 The circuit diagram of the simple signal generator

CHAPTER 3

RADIO CIRCUITS

S.W. Crystal Set

With a short wave crystal receiver it is possible to pick up a surprisingly large number of stations over a period of time, many of which emanate from quite distant transmitters. The crystal set circuit of Fig.25 covers a frequency range of roughly 5 to 15MHz, and therefore covers the 25, 31, 39, and 49 metre broadcast bands. The output is for one or two crystal earphones.

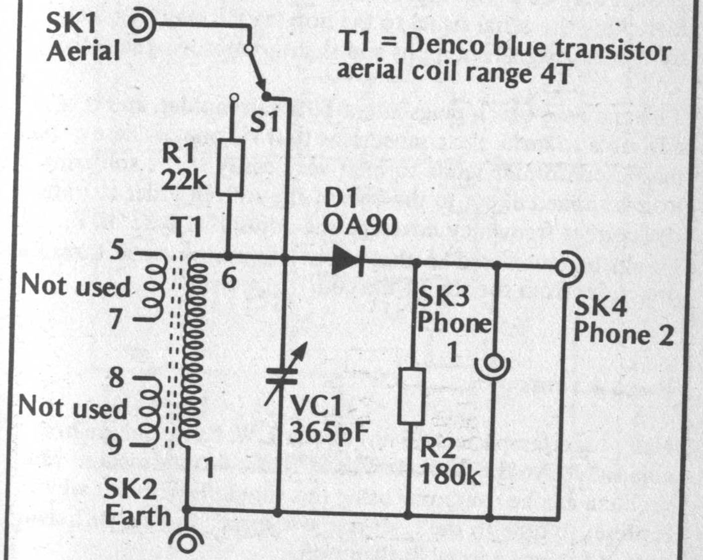

Fig. 25. The circuit diagram of the S.W. crystal set

The circuit uses the main winding of T1 and VC1 as the tuned circuit, and VC1 is, of course, the tuning capacitor. D1 is the demodulator diode, and R.F. filtering is provided by R2 and the capacitance of the earphone(s). The aerial is a longwire type, which merely consists of a piece of insulated wire which is as long as possible, and positioned as high as possible, with one end connected to SK1. An earth connection can considerably aid signal strengths on the lower end of the frequency range covered by the set, and an earth can be a metal pipe buried in the ground and connected to SK2 via a lead which should be as short as possible.

The number of signals and their strengths will vary according to the time of day, time of year, and other factors. However, it will normally be possible to receive at least one or two stations at reasonable strength. The selectivity of a receiver such as this is inevitably quite poor, and when conditions are good it may be found that several stations are received at once. Switching the aerial signal to the unit via R1 may then help by reducing signal strengths and slightly improving selectivity.

T1 has a base which plugs into a B9A valveholder, and it is advisable to make the connections to it via one of these as the plastic coilformer tends to melt very easily if the soldering iron is applied direct to the pins of the coil. In order to obtain the correct frequency coverage the adjustable core of T1 should be unscrewed so that about 10mm. of metal thread protrudes from the top of the coil.

Radio 4 Tuner

Many tuner/amplifiers do not have a L.W. band and are not able to receive the 200kHz B.B.C. Radio 4 transmission. This problem can be overcome using this simple L.W. tuner which is preset to tune to the L.W. Radio 4 signal, and has sufficient output for any normal hi-fi amplifier.

Fig.26 shows the circuit diagram of the unit, L1 is the ferrite

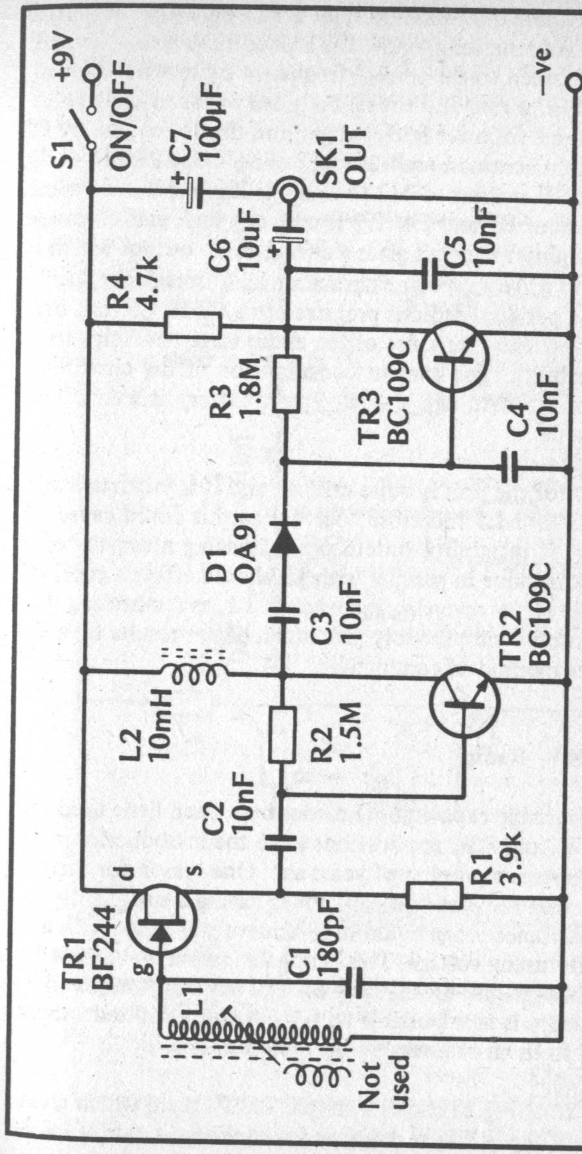

Fig. 26. The circuit diagram of the Radio 4 tuner

aerial (an Ambit International type LWC1 on a 9.5mm. ferrite rod about 140mm. long) and C1 is a fixed tuning capacitance. The unit is tuned to the correct frequency by moving the coil along the ferrite rod, and the coil is glued or taped in place. Tr1 is a source follower buffer stage, and this is coupled by C2 to a high gain common emitter stage using Tr2. C3 couples the amplified R.F. output of Tr2 to a simple diode detector which is comprised of D1 and C4. Tr3 is used as a high gain common emitter amplifier and this gives a strong audio output which is adequate to drive a crystal earpiece or high impedance headphones if a personal radio is preferred to a tuner. C5 rolls off the high frequency response of the audio stage and helps to avoid instability. The current consumption of the circuit is only about 3.5mA., and a small 9 volt battery is a suitable power source.

The layout of the unit is quite critical, and it is important not to have L1 and L2 too close together as this could cause instability. If instability should occur, placing a resistor of about 10k in value in parallel with L2 should effect a cure. It is advisable to try reversing the leads to L1, as connecting it one way round will probably give much better results than the alternative method of connection.

Varicap M.W. Radio

Varicap (variable capacitance) diodes have been little used in M.W., L.W., and S.W. applications since the introduction of suitable devices a number of years ago. One reason for this is probably that the early varicap diodes having a sufficiently wide capacitance range could only achieve this range with a fairly high tuning voltage. This is not the case with current devices though, and tuning over the full medium waveband using varicaps is now possible with a 6.8 volt stabilised supply obtained from an ordinary 9 volt radio battery.

The circuit of Fig.27 shows a simple T.R.F. radio which gives coverage of the full M.W. band and is tuned by a pair of back-

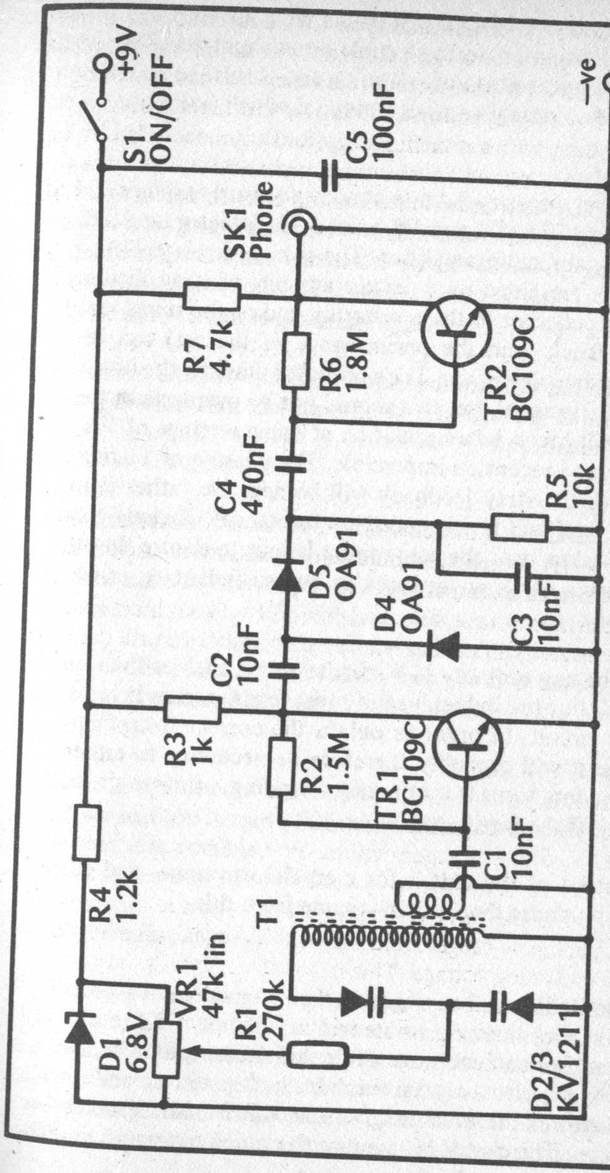

Fig. 27. The circuit diagram of the M.W. varicap radio

to-back varicap diodes (D2/3) which are contained in a single package. R4 and D1 form a simple zener shunt stabiliser which gives an output voltage of 6.8 volts, and this is fed to VR1 which gives a variable output potential which is used to tune the receiver.

The receiver circuitry is quite straight forward and uses Tr1 as an R.F. amplifier, D4 and D5 as an A.M. detector, and Tr2 as a high gain audio amplifier. The gain and selectivity of the circuit are aided by a certain amount of stray feedback from Tr1 collector to the non-earthy end of the tuned circuit. This feedback (and the performance of the set) can be improved by a lead from Tr1's collector close to the hot lead of the ferrite aerial, but this should not be overdone or the circuit will break into oscillation at some settings of VR1, making good reception impossible. The phasing of T1 must be correct or the stray feedback will be negative rather than positive, and will give reduced performance! Obviously care must be taken over the component layout to ensure that the innate level of stray feedback is not sufficient to cause instability.

T1 can be any ordinary M.W. ferrite aerial, such as the Denco MW5FR, but the inductance of most ferrite aerials is too high for this circuit. In order to obtain the correct frequency coverage it will probably therefore be necessary to either remove a few turns from the main winding, or move the coil slightly off the ferrite rod.

The output of the unit is for a crystal earphone, and there should be more than ample volume from this.

Bedside Radio

This simple medium wave radio has an output which is intended to drive a crystal earpiece, but an add-on audio output stage can be used to give low volume loudspeaker reception. The circuit is based on the well known and popular

ZN414 integrated circuit, and this is used in the standard configuration. L1 is the ferrite aerial, and any commercial ferrite aerial can be used here (Denco MW5FR etc.). Most commercial medium wave ferrite aerials have a small low impedance coupling winding, and this is not required here. It can therefore be either removed or just ignored. VC1 is the tuning control, and any value between about 200 and 300pf should give satisfactory results here. The aerial coil is given a position on the ferrite rod that gives full coverage of the medium waveband, and then it is glued or taped in that position. The circuit is given in Fig.28.

The ZN414 circuitry needs a supply potential of only about 1.2 to 1.3 volts, and this supply should peferably be stabilised. A suitable supply voltage is provided by amplified diode Tr1 and its load resistor R4.

The output from the ZN414 is only about 30mV. R.M.S. and this needs to be boosted somewhat in order to give good volume from a crystal earphone. The necessary extra amplification is produced by IC2 which is used as a straightforward inverting amplifier having a voltage gain of approximately 45 times. S1 is the on/off switch and C4 is a supply decoupling capacitor. The current consumption of the receiver is about 6mA.

The component layout is not too critical, but L1 should not be positioned too close to IC1, and C2 should be mounted as close to IC1 as possible.

Add-On Audio Stage

The audio add-on stage for the bedside radio of Fig.28 is shown in Fig.29, and this is just an emitter follower buffer stage based on Tr1 which lowers the output of the unit so that it is able to drive a high impedance loudspeaker. R1 is the load resistor for Tr1 and C1 gives D.C. blocking at the output.

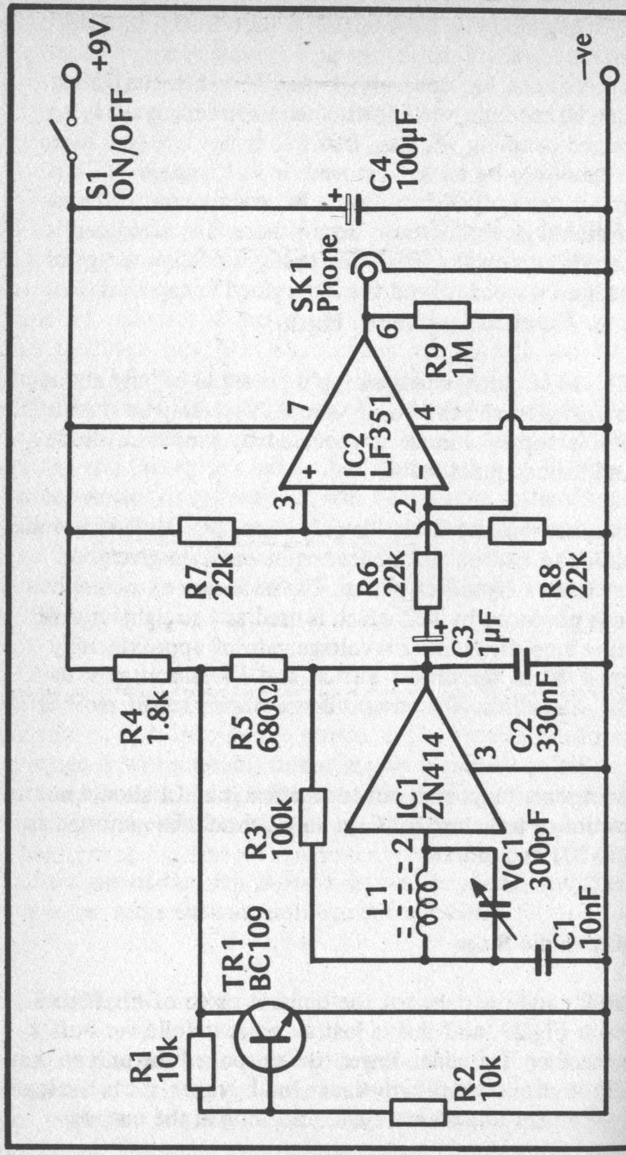

Fig. 28. The circuit diagram of the bedside radio

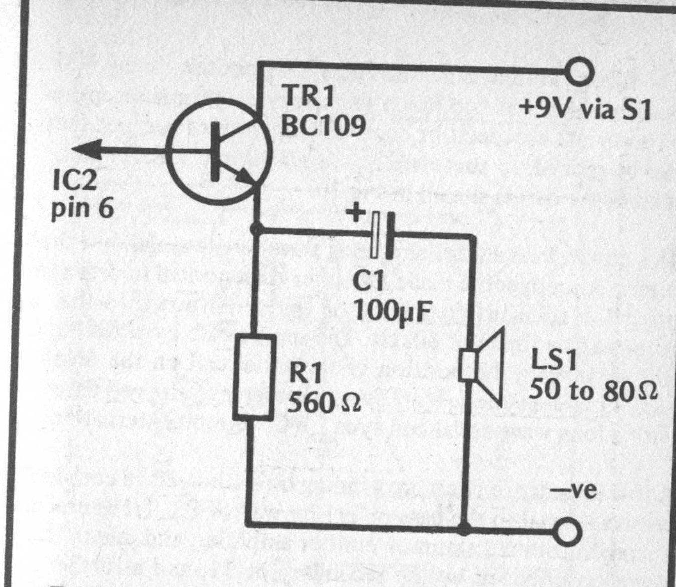

Fig.29. An add–on audio stage for the circuit of Fig.28

Only a modest output power of milliwatts rather than watts is provided by this output stage, but this is adequate in this application where high volume would not normally be desirable. It may be found that higher volume can be attained by slightly increasing the value of R9 in the main receiver circuit, but this should not be overdone or severe distortion will result.

The addition of the output stage increases the current consumption of the unit by about 8mA. Note that LS1 should not be positioned too close to L1 of the main receiver or instability or poor sensitivity may result. Note also that LS1 must be a high impedance type, and should not be a low impedance speaker.

60kHz Time Receiver

The 60kHz atomic time transmissions from the Rugby MSF transmitter are at too low a frequency to permit reception on a normal broadcast or communications receiver, but they can be picked up successfully using a simple T.R.F. radio such as the design shown in Fig.30.

This circuit uses an ordinary long wave ferrite aerial, but the tuning capacitance is made far larger than normal in order to bring the resonant frequency of the aerial down to the required low figure of 60kHz. The set is tuned by adjusting C2 and by varying the position of the aerial coil on the ferrite rod. T1 is a 140mm. by 9.5mm. diameter ferrite rod fitted with a long wave aerial coil type LWC1 (Ambit International).

A low impedance coupling winding on T1 is used to couple the received signal to the base of Tr1 by way of C3. Tr1 is used as a straight forward common emitter amplifier, and due to the inversion provided by the secondary of T1, and a further inversion provided by Tr1, positive feedback is provided by C4. In practice this is adjusted for a level of feedback that is just sufficient to cause oscillation to occur.

C5 couples the output of Tr1 to a simple diode detector, and the output of this is amplified by a high gain common emitter audio stage based on Tr2. The output signal can be monitored using high impedance (200 ohms or more) headphones or a crystal earphone. The output can also be fed to an audio amplifier.

The set is tuned slightly off the 60kHz transmission frequency so that an audio beat note is produced when the 60kHz carrier wave is present. The 1 second marker signals (and other signals) are transmitted as breaks in the carrier wave, and will be heard as short gaps in the audio output signal.

Due to its modest gain and fairly low operating frequency this circuit is unlikely to give any problems in construction.

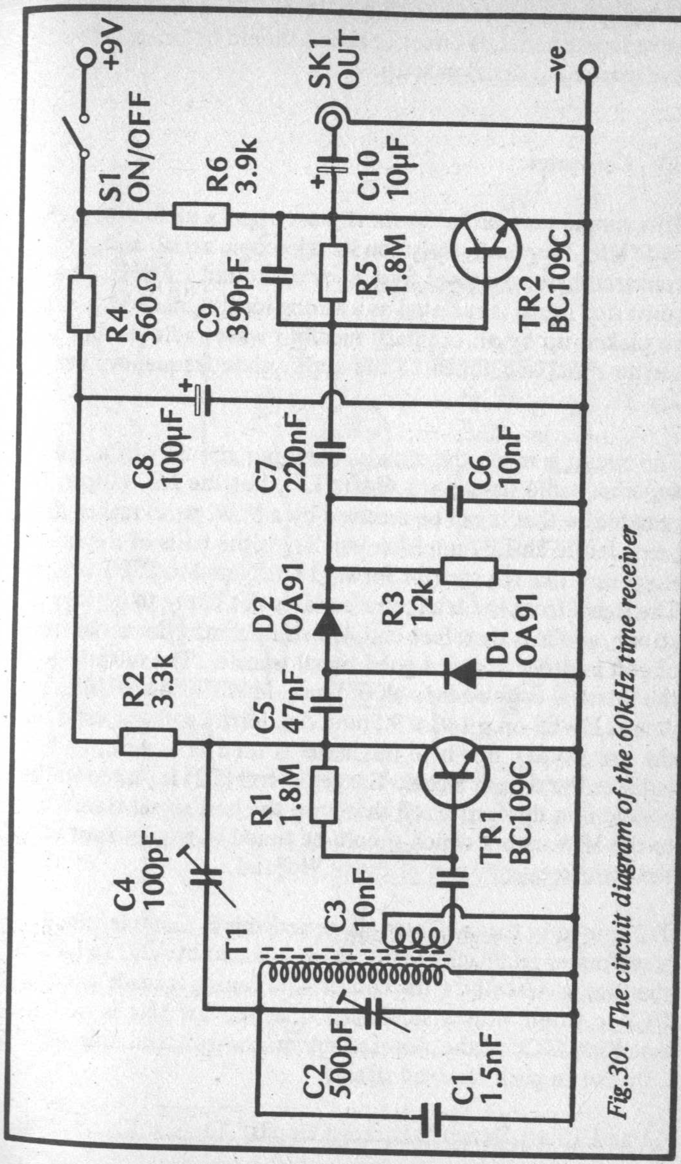

Fig.30. The circuit diagram of the 60kHz time receiver

Note that the ferrite aerial, like an ordinary medium or long wave ferrite aerial, is directional and should be orientated to give maximum signal pick-up.

S.W. Converter

This simple unit picks up short wave signals in the range 5 to 15MHz (approximately) on its telescopic aerial, and converts these to a fixed frequency at around 1.6MHz. The converted signal is radiated as a strong local signal which can be picked up by an ordinary medium wave radio having a ferrite aerial and tuned to the appropriate frequency. See Fig.31.

The circuit is much the same as the input circuitry of a S.W. superhet radio having a 1.6MHz I.F., but the I.F. output is radiated so that it can be received by a M.W. radio rather than being fed to an I.F. amplifier etc. Tr1 is the basis of the mixer stage, and this is a straight forward dual gate MOSFET design. The signal from the telescopic aerial is not likely to be very strong, and it is therefore coupled straight into the aerial tuned circuit in order to give a good signal transfer. The output of the mixer is coupled into an ordinary M.W. ferrite aerial (an Ambit MWC2 on a 140 x 9.5mm. dia. ferrite rod was used on the prototype), but here the aerial is used in reverse, and radiates the output signal. The aerial coil (T2) is placed in the position on the ferrite rod that gives the best signal transfer to the M.W. radio, which should be tuned to a quiet spot at the high frequency end of the M.W. band.

Tr2 is used in the oscillator stage, and this is a simple tuned, transformer feedback circuit which is tuned by VC2 (which is the tuning control for the converter). Tuning is made easier if VC2 is fitted with a slow-motion drive, but this is not essential. VC1 is the aerial trimmer control, and this is adjusted to peak received signals.

T1 and T3 are Denco D.P. range 4 coils: T1 is a Blue aerial

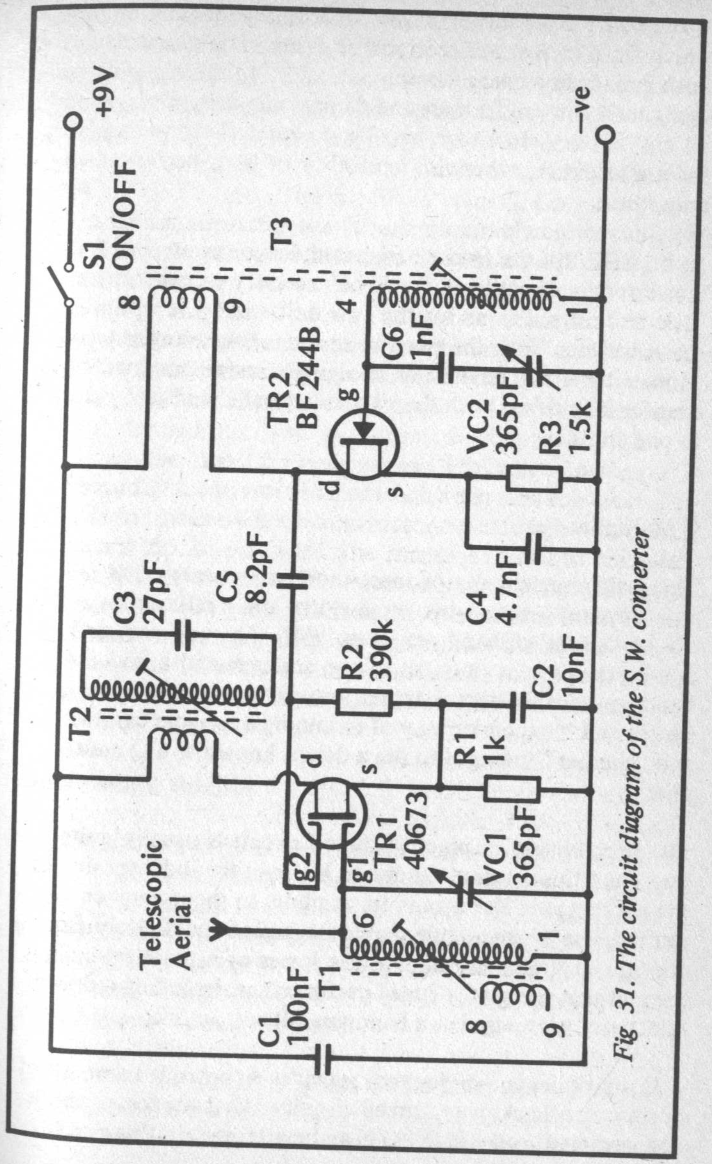

Fig 31. The circuit diagram of the S.W. converter

type and T3 is a White oscillator coil. These have a 9 pin base which fits a B9A valveholder, and it is a good idea to mount both coils in holders as direct connections to the coil pins can easily melt the coil formers and damage the coils. Do not mount T1 and T3 very close together. Try to keep all the wiring fairly short and direct, otherwise instability or poor performance may result.

As both T2 and the ferrite aerial of the receiver used with the converter are directional, it will be necessary to experiment a little to find positions for the two units that give optimum signal transfer. With the two ferrite rods at right angles to one another there will be little or no signal transfer, and maximum transfer is achieved with the two rods parallel and fairly close to one another.

Q Multiplier

The main shortcoming of most short wave receivers is that they have only one or two bandwidths and a rather slow roll-off of the out-of-band response. With the very crowded conditions present on most of the amateur and broadcast bands poor selectivity is rather a major drawback. A very simple and inexpensive way of obtaining a variable bandwidth and "sharper" tuning is to use a device known as a Q multiplier.

The selectivity of a practical tuned circuit is usually quite poor, and this is primarily due to losses in the inductor due to the resistance of the wire in its winding. In theory any energy put into the tuned circuit should be passed indefinitely from the coil to the capacitor, but the losses of a practical coil prevent this and give reduced performance, including reduced selectivity when used as a bandpass filter.

A Q multiplier is a device that provides a controlled amount of positive feedback over a tuned circuit so that the losses can be counteracted, and a high "Q" can be attained. In this circuit

(Fig.32) the tuned circuit is the main winding of a Denco 465kHz I.F. transformer type IFT13, and this can be tuned to any of the popular I.F.s around 455 to 470kHz. Tr1 is a common emitter amplifier which has its output coupled into the tuned winding of T1, while its input is fed from the secondary winding of T1 via VR1. The latter controls the effective Q of T1, and is the selectivity control. Care must be taken not to advance this too far or an unusably narrow and peaky response will be obtained, or the circuit will break into oscillation.

The output of the unit is taken to the primary winding of the first I.F. transformer of the receiver via a coaxial cable. Transistorised receivers normally have a tapped primary winding on each I.F. transformer, and the output is coupled to this tapping on the first I.F. transformer. The unit is not recommended for use with valved receivers, and it is not advisable for inexperienced constructors to attempt to add a unit such as this to a receiver. The core of T1 must be adjusted to align the unit with the I.F. circuits of the receiver or there will effectively be a tuning shift as the Q multiplier is adjusted for increased selectivity. Due to the capacitance in the lead connecting the Q multiplier to the receiver it will probably be necessary to realign the first I.F. transformer of the receiver.

Radio Control Monitor

When adjusting a radio control transmitter it is often useful to have a field strength monitor that gives a relative indication of the signal strength from the transmitter. This circuit covers the 27MHz radio control band if the core of the tuning inductor is adjusted correctly, and it can be tuned up to the 35MHz band by removing C3 and adjusting the core of the tuning inductor to suit this band.

Fig.33 shows the circuit diagram of the unit, and this is centred around a simple dual gate MOSFET common source amplifier which utilizes Tr1 in a standard configuration. T1 is

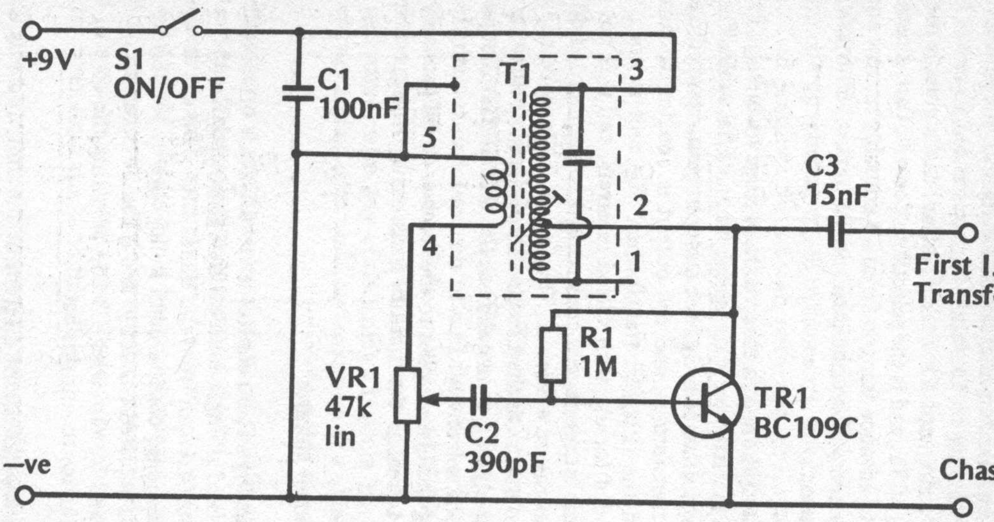

Fig. 32. The circuit diagram of the Q multiplier

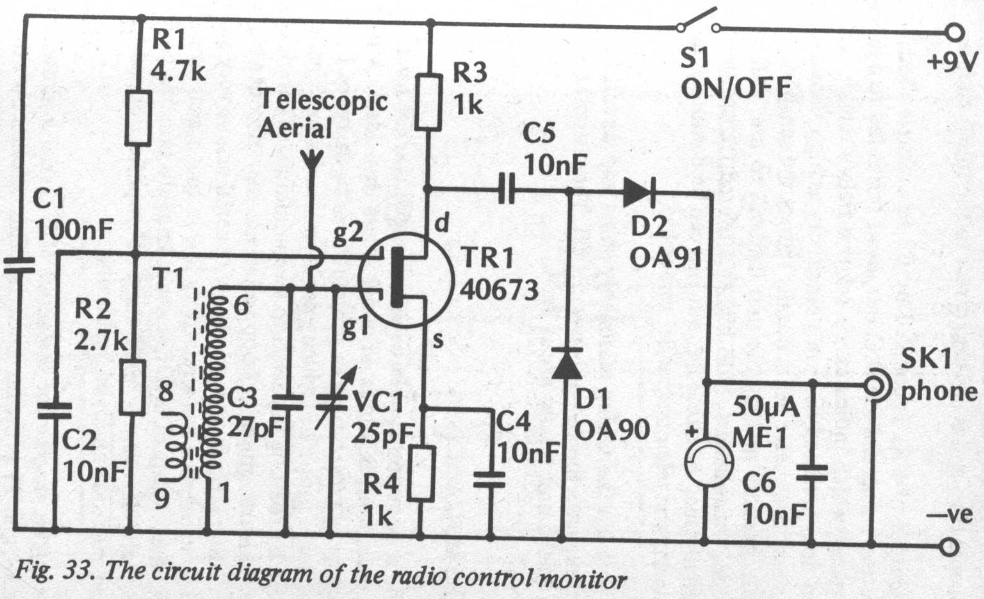

Fig. 33. The circuit diagram of the radio control monitor

a Denco D.P. range 5 blue aerial coil, and the main winding of this forms the input tuned circuit in conjunction with VC1, or C3 and VC1 for 27MHz band operation, as explained earlier. VC1 is, of course, the tuning control. The aerial is a short telescopic type and is coupled direct to the tuned circuit.

The output of TR1 is coupled by C5 to a simple diode detector circuit, and the D.C. output of this is fed to a moving coil meter which indicates the relative field strength. The meter does not need to have a sensitivity as high as 50μA, but a less sensitive meter will naturally reduce the sensitivity of the circuit, possibly making the unit difficult to use with low powered transmitters. With higher powered transmitters the sensitivity may be excessive, but this can be counteracted by retracting the telescopic aerial somewhat.

Try to keep the wiring reasonably short and direct, but in other respects the layout is not critical. The current consumption of the unit is only about 2mA.

Active M.W. Aerial

While it is quite possible to use a longwire aerial for M.W. DX reception, this type of aerial is less than ideal. A loop aerial or a ferrite type is preferable as both types have directional properties which enable an interfering signal to be nulled by rotating the aerial to give minimum signal pick-up from this transmission. Loop aerials are very effective but are also rather large in general. A ferrite aerial can be very small and compact, but has a relatively low output signal level. However, the output of a ferrite aerial is easily boosted to an acceptable level by a single stage of amplification, and this is the method employed in this unit.

The circuit diagram of the active M.W. aerial is shown in Fig.34, and T1 is the ferrite aerial. This is tuned over the M.W. band by means of VC1, and in practice the latter is adjusted to peak received signals.

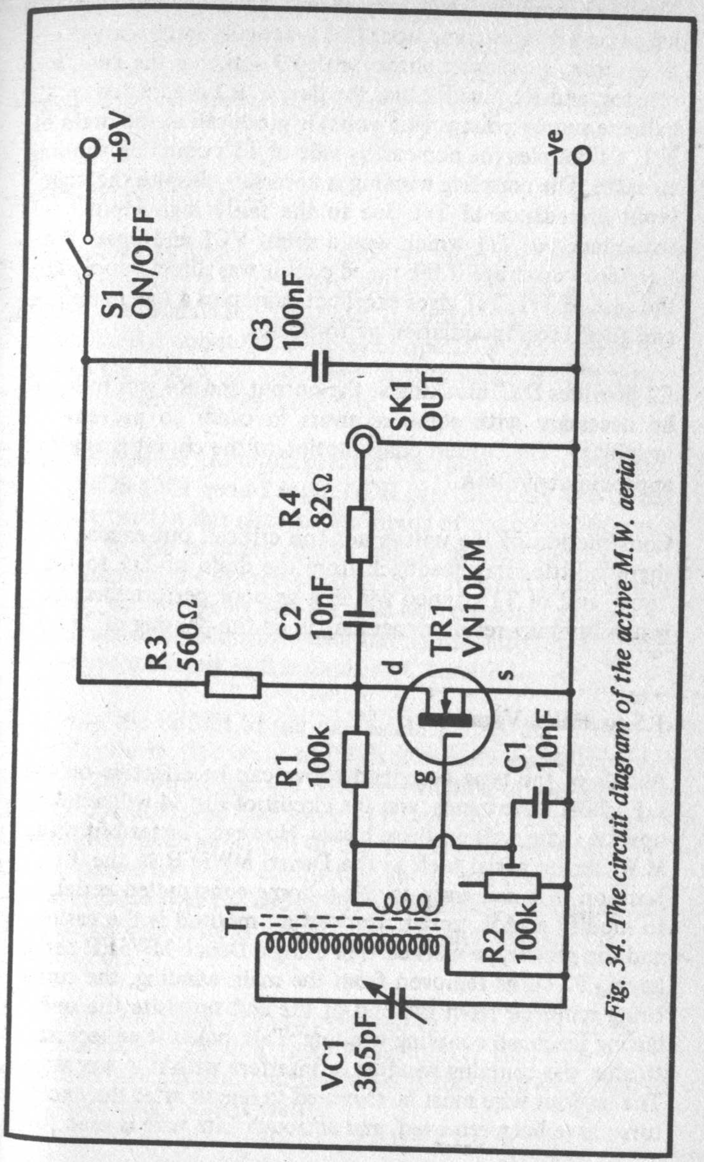

Fig. 34. The circuit diagram of the active M.W. aerial

The R.F. amplifier stage may seem a little unusual as it is based on a VMOS transistor. This is actually quite similar to an ordinary bipolar amplifier, with R3 acting as the drain load resistor, and R1 plus R2 bias the device. R2 is adjusted so that half the supply voltage (4.5 volts) is produced at the drain of Tr1. C1 couples the non-earthy side of T1's coupling winding to earth. The coupling winding is necessary despite the high input impedance of Tr1 due to the fairly high input capacitance of Tr1 which would shunt VC1 and upset the frequency coverage if the tuned circuit was direct coupled to the gate of Tr1. Tr1 gives excellent gain, plus a low noise level and good cross modulation performance.

C2 provides D.C. blocking at the output and R4 was found to be necessary with some receivers in order to prevent instability. The current consumption of the circuit is approxi- approximately 8mA.

Construction of the unit is not too critical, but ensure that there is little stray feedback from the drain of Tr1 to the "hot" end of T1's tuned winding or poor performance or instability may result, depending upon the phasing of T1.

1.5 to 4MHz Version

Aerials of the type described above can be effective on the L.F. short wave bands, and the circuit of Fig.34 will actually operate quite well on these bands. However, instead of using a M.W. ferrite aerial such as the Denco MW5FR in the T1 position it is necessary to use a home-constructed aerial, or to modify a M.W. aerial. The second method is the easier, and the prototype worked well using a Denco MW5FR aerial having 52 turns removed from the main winding, the turns being removed from the end of the coil opposite the one having the small coupling winding. This makes it unnecessary to alter the coupling winding or interfere with it in any way. The leadout wire must be trimmed to length after the excess turns have been removed, and although Litz wire is used for

the coil, this can be easily tinned with solder, but the iron must be applied for a little longer than normal in order to remove the insulation. This leadout wire must be glued or taped to the coil former to prevent the coil from unwinding.

Auto Switch-Off Timer

This extremely simple timer circuit is intended to be used with an ordinary 9 volt battery operated transistor radio which is used as a bedside set. The timer automatically switches off the radio after about ten minutes of operation, and this is the same as the "sleep" or "slumber" facility which is built into a few commercially produced sets.

Fig.35 shows the circuit diagram of the timer, and it is based on a CMOS 4001 quad 2 input NOR gate, but only two of the gates are used in this circuit. The inputs of the other two gates are tied to the negative supply rail to prevent spurious operation, and the unused outputs (pins 10 and 11) are simply ignored. The two gates that are used are connected to form a simple monostable multivibrator having the output pulse length set at approximately 600 seconds by C2 and R2.

Normally the output of the monostable is low and VMOS switching device Tr1 is cut off. This gives no negative supply path to the output and the radio is switched off. If S1 is briefly operated, the 600 second (10 minute) output pulse commences, and the output of the monostable goes high. Tr1 is then biased hard into conduction and power is connected to the radio set. At the end of the output pulse the output of the monostable returns to the low state and the radio is switched off.

S2 enables the timer to be bypassed so that the radio can be used normally. The unit requires no on/off switch as it consumes no significant current when in the stand-by mode, or when operating for that matter.

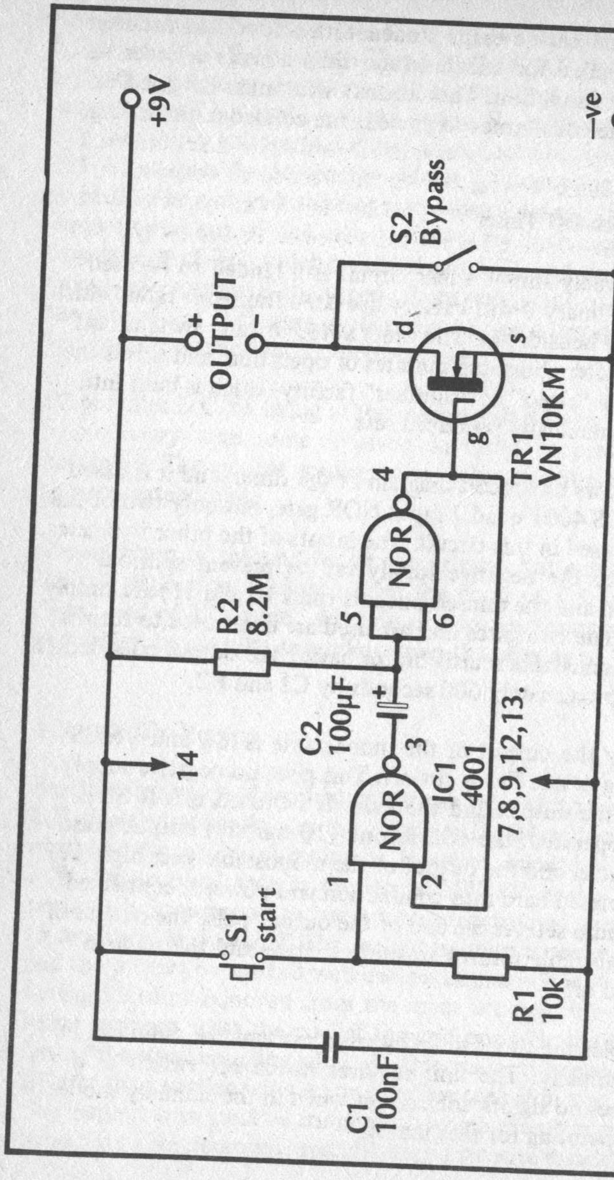

Fig.35. The circuit diagram of the auto switch–off timer

C2 must be a type of capacitor that has a low leakage current, and due to the fairly high value required this really means using a tantalum bead component. The output of the timer can be taken to a battery clip of the type used in the radio, and the battery is, of course, fitted inside the timer unit and not in the radio. Remember that as the output connector of the timer is sourcing power, rather than collecting it as would normally be the case for a battery connector, the two press studs and leads have the opposite roles to normal. Therefore the red lead connects to the negative output of the timer, and the black lead carries the positive output.

CHAPTER 4

HOUSE AND CAR CIRCUITS

Mains Failure Alarm

While a brief, or even fairly prolonged failure of the mains supply is often of little or no consequence, such a failure can sometimes be a nuisance and can even cause costly damage. Some items of equipment will not operate properly if there is an interruption of the supply, and will give misleading results in this eventuality. A longer break in the supply could result in food inside a freezer unit being ruined if the break in supply passes unnoticed for some time and remedial action is not taken.

The circuit of Fig.36 is for a simple alarm unit that monitors the mains supply and sounds an alarm if there is even a brief break in the supply. The supply is monitored by a mains relay, and this is connected to the mains via relay contact RLA2 and push button switch S1. The latter is operated in order to get the relay to operate initially, and then RLA2 connects the power to the relay coil. RLA1 is a normally closed relay contact, and therefore opens when the relay is energised. This cuts off power from the simple audio alarm circuit which is based on IC1 which is used as a straight forward 555 astable.

If the mains supply fails, RLA2 opens and cuts off power to the relay. Even if the mains then returns to normal, power will no longer reach the relay coil and the unit latches with the relay de-energised. In this state RLA1 closes, and power is fed from BY1 to the oscillator circuit, causing an audio alarm tone to be generated. Once the mains supply has been restored, the unit can be reset by briefly operating S1 again.

If the latching action is not required, simply omit RLA2 and S1, and connect the relay coil direct across the mains.

The unit is very simple with regard to construction, but as it

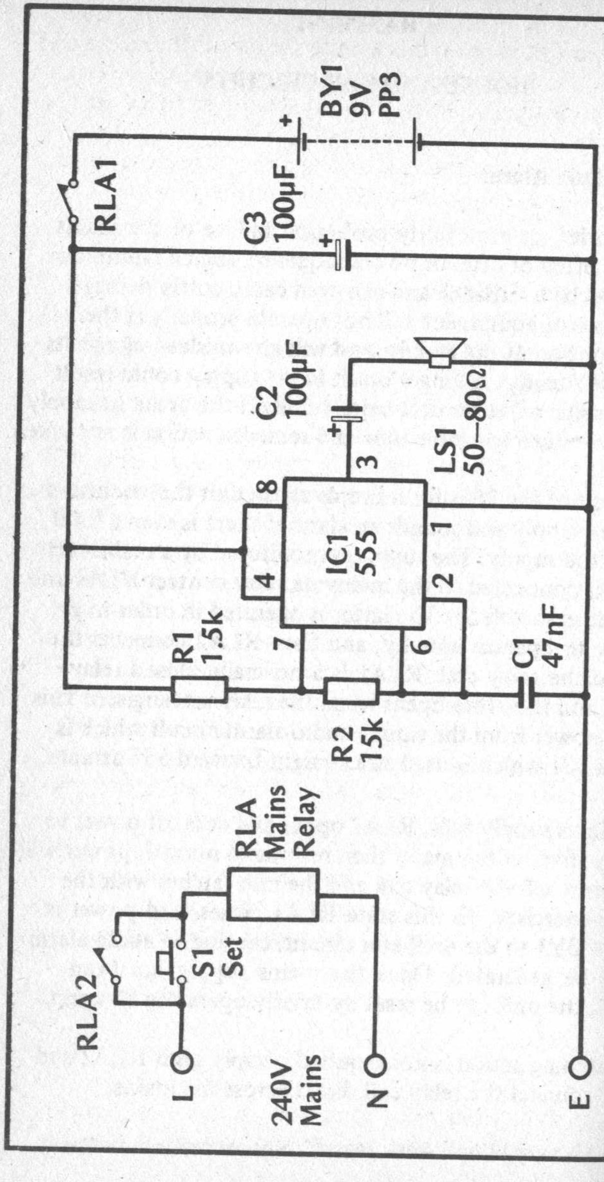

Fig.36. The circuit diagram of the mains failure alarm

connects to the mains it is necessary to observe the usual safety precautions. Also make quite sure than there is no risk of an electric shock from the mains being sustained when the battery is changed. Ideally the unit should be fitted with a battery compartment that enables the battery to be changed without opening up the case and exposing dangerous mains wiring. Be sure relay and switch S1 are mains types.

Soldering Iron Reminder

It is very easy to leave a soldering iron switched on, giving an effective reduction in bit and element life, as well as producing a potential fire hazard. The simple circuit of Fig.37 can be connected in parallel with the power lead of a soldering iron, and will produce an audible reminder signal that will help to prevent the iron being inadvertently left switched on.

The left hand section of the supply is just a simple fullwave power supply giving an output potential of about 9 volts. The right hand section is a 555 astable circuit having an operating frequency that is just a fraction of one Hertz. Thus the loudspeaker is not fed with an audio tone, but is fed with a series of pulses of alternate polarities. This gives a clicking sound from the loudspeaker each time the output of IC1 changes state, and these "clicks" are produced at roughly three second intervals. This gives an effective reminder signal, but is not over-obtrusive.

As with the previous project, construction of the unit is very straightforward, but it should be borne in mind that the circuit is mains powered and the appropriate safety precautions should be observed.

Bit Saver

Some soldering irons seem to run at a rather high temperature, giving relatively poor bit life and perhaps making it

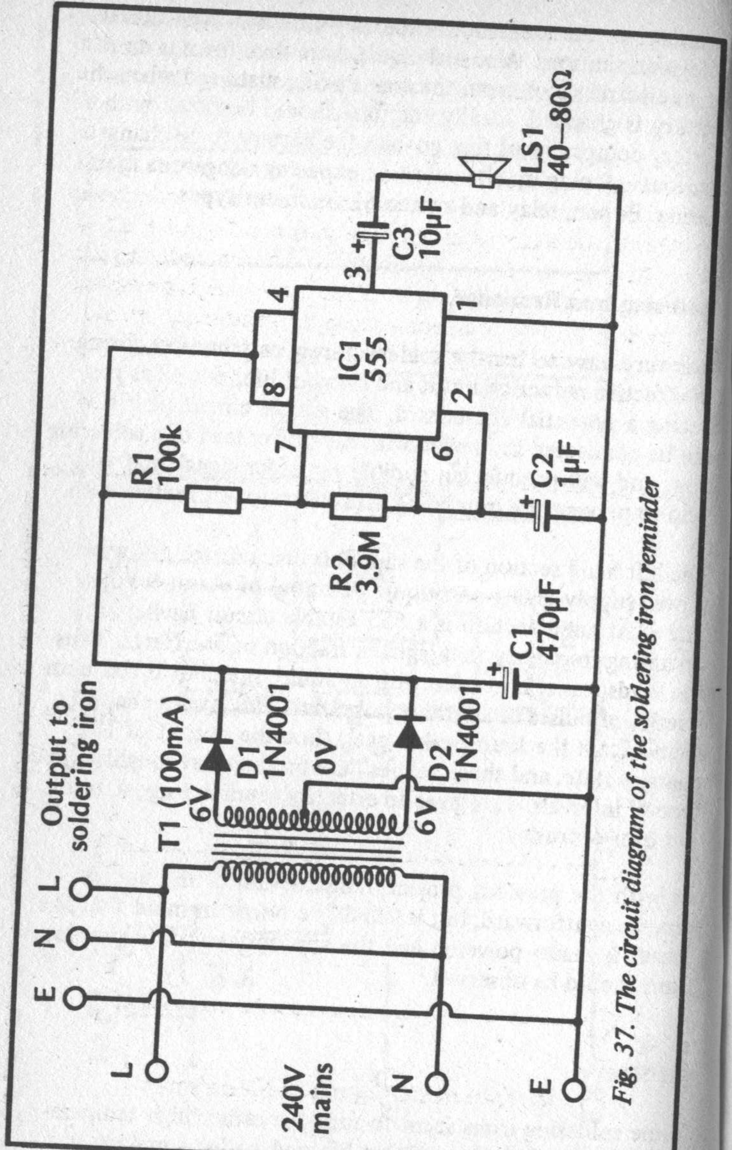

Fig. 37. The circuit diagram of the soldering iron reminder

difficult to produce good soldered joints. An excessive bit temperature also increases the risk of overheating and damaging semiconductor devices when soldering them into circuit.

One simple way of reducing the power fed to the iron (and therefore reducing the bit temperature to a more acceptable level) is to use a circuit of the type shown in Fig.38. Here the diode half wave rectifies the supply to the iron, reducing the power dissipated in the iron by 50%. Little heat is generated in the diode since it is not a dropper resistance, but is effectively switching the iron on during one set of half cycles and off during the other set.

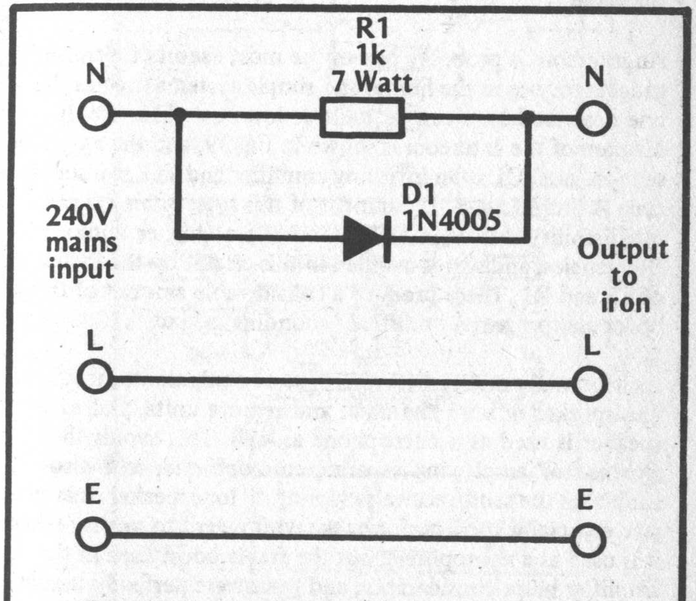

Fig. 38. The circuit diagram of the bit saver

Unfortunately, this gives an excessive reduction in power, and D1 is therefore bypassed to some extent by R1 on half cycles where D1 is reverse biased. This results in the iron being run at roughly 80% of full power, which in practice seems to be sufficient to increase bit life, but not so much as to make soldering impossible or difficult. The value of R1 shown in Fig.38 is for a 15 watt iron, and it should be reduced to 680 ohms for a 25 watt iron.

Once again, construction of this project is very straight forward, but the appropriate safety precautions should be taken to prevent the possibility of an electric shock from the mains supply being received by anyone handling the unit.

Intercom

An intercom is probably one of the most useful of electronic gadgets for use in the home, and simple systems such as the one described here can be built at low cost. The circuit diagram of the Intercom is shown in Fig.39, and the amplifier section uses IC1 as an inverting amplifier and Tr1 as a simple class A output stage. Equipment of this type often has poor intelligibility due to a lack of output at higher audio frequencies, and this is avoided in this circuit by the inclusion of C3 and R1. These produce a considerable amount of treble boost and prevent a "muffled" sounding output.

As is normal practice with this type of equipment, there is a loudspeaker in both the main and remote units, and each speaker is used as a microphone as well. This avoids the expense of employing separate microphones, and also simplifies the send/receive switching. A loudspeaker does not give especially good performance with regard to quality when it is used as a microphone, but the treble boost used in the amplifier helps considerably, and results are perfectly usable. S2 is the send receive switch, and connects the appropriate speaker to the input so that it acts as the microphone, and the other speaker to the output so that it operates as a speaker.

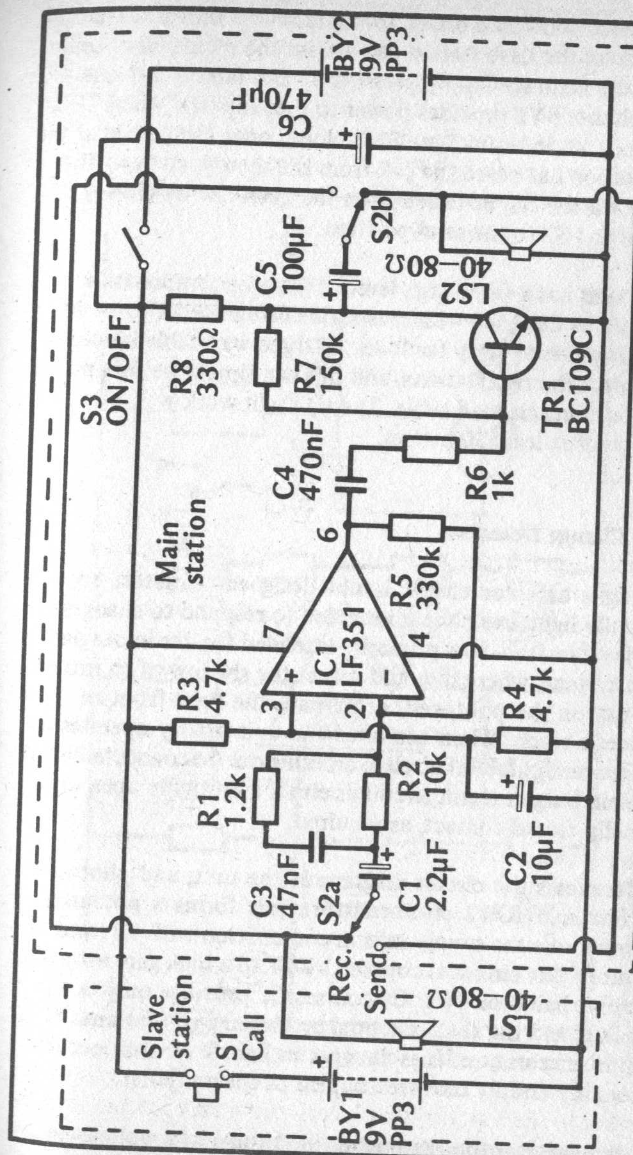

Fig. 39. The circuit diagram of the intercom

This switch should be biased to the receive position so that the operator of the slave station can attract the attention of someone at the main station by pressing S1 and talking into the microphone. BY1 provides power to the amplifier when S1 is operated, but in normal use S3 is closed once the person at the main station has heard the call from the remote station. The remote station can be called from the main one by closing S3 and moving S2 to the send position.

As the unit has a fairly high level of gain the component layout must be designed with reasonable care in order to avoid instability due to stray feedback. A three way cable is needed to connect the two stations, and this can simply be thin mains cable or twin screened cable. The unit will work well over a distance of at least 20 metres.

Light Change Detector

This light detector circuit is not designed to detect any particular light level, but is intended to respond to changes in light level instead. It is primarily intended for use in burglar alarm systems where it would detect the shadow of an intruder as it fell on the photocell, or perhaps the light from an intruder's torch. When activated the unit briefly operates a changeover contact of a relay, and this can be connected into the main burglar alarm circuit as either a normally open or a normally closed contact, as required.

Fig.40 shows the circuit diagram of the unit, and photocell PCC1 is an ORP12 photoresistor. This forms a potential divider across the supply rails in conjunction with R1, and the output of this circuit is coupled by C1 to a high gain inverting amplifier based on IC1. Of course, C1 provides only A.C. coupling, and the D.C. potential produced by PCC1 and R1 is of little importance. It is changes in light level that occur reasonably rapidly that are coupled to the amplifier.

IC1 is biased so that its output is at too low a voltage to

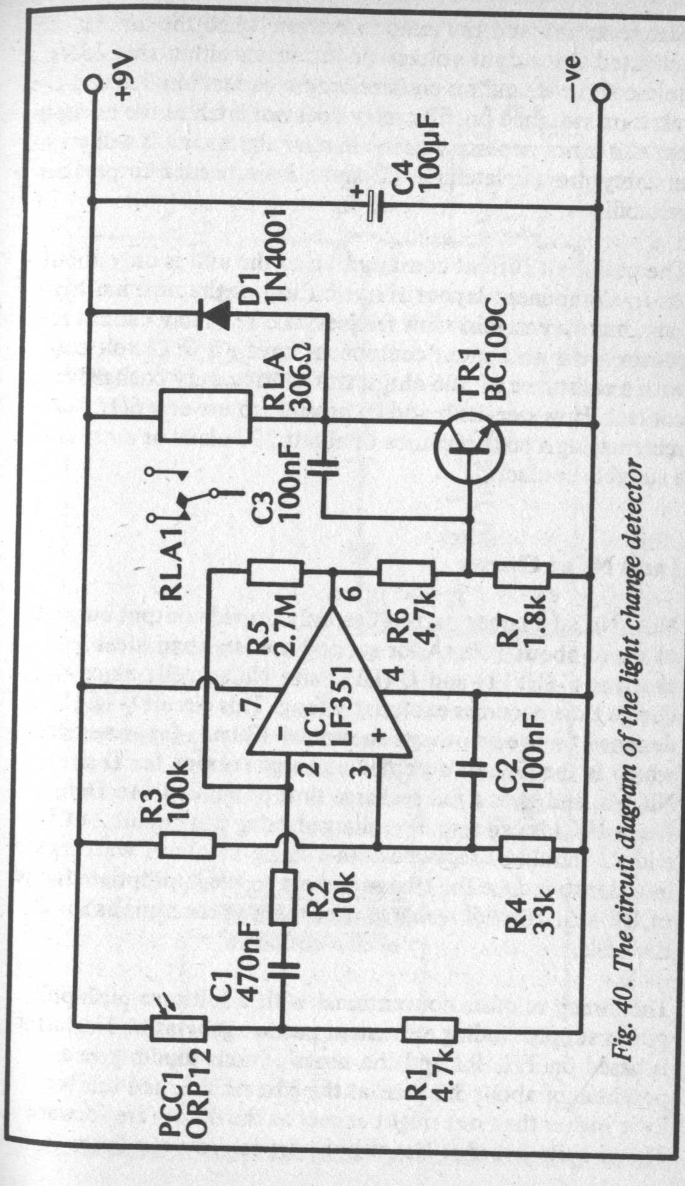

Fig. 40. The circuit diagram of the light change detector

switch on Tr1 and the relay. However, when the unit is activated the output voltage of IC1 varies either side of its quiescent level, and on positive output excursions Tr1 and the relay are switched on. The relay does not latch in the on state, but this is not necessary as the burglar alarm circuit will presumably provide latching. C2 and C3 are needed to prevent instability.

The quiescent current consumption of the unit is only about 2mA. Component layout is not critical as the unit has high gain, but only at fairly low frequencies. The relay used in the prototype is an Omron component having a 6/12 volt coil with a resistance of 306 ohms, and a heavy duty changeover contact. However, it should be possible to use any 6/12 volt relay having a coil resistance of about 200 ohms or more and a suitable contact.

Large NiCad Charger

Most NiCad charger circuits can only provide output currents of up to about 100mA. or so, and are less than ideal for charging C (HP11) and D (HP2) size NiCad cells, since the charge time becomes excessively long. This circuit (Fig.41) is designed to give an output current of 400mA. (approximately) which is the highest acceptable charge current for D size NiCads, and gives a full recharge time of about 14 to 16 hours. C size NiCads can also be recharged using this circuit, but R2 and R3 should be replaced with a single 15 ohm 1 watt resistor in order to reduce the charge current to the appropriate figure of 180mA. The full recharge time remains the same as for D size cells.

The circuit is quite conventional with a fullwave push-pull power supply feeding a constant current generator. The latter is based on Tr1: R1 and the series of four diodes give a potential of about 3.2 volts at the base of Tr1, and this is a little higher than one might expect as the diodes are forward biased quite heavily. The voltage drop across the base –

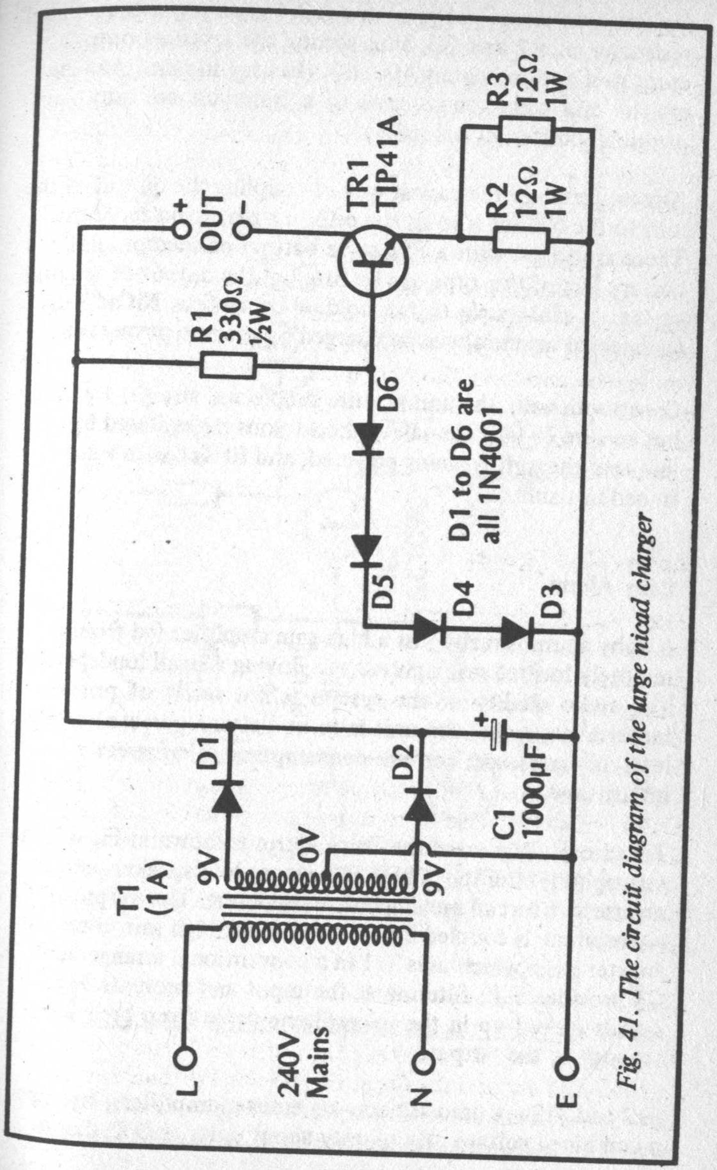

Fig. 41. The circuit diagram of the large nicad charger

emitter junction of Tr1 gives about 2.4 volts across the parallel resistance of R2 and R3, thus setting the required output current of approximately 400 mA. (bearing in mind that the emitter and collector currents of a transistor are only marginally different normally).

Almost certainly the easiest way of coupling the output of the unit to the NiCads is to fit the cells in a plastic battery holder. These are fitted with a PP3 style battery connector, and a battery clip of this type can be fitted at the output of the unit so that it can couple to the holder. Up to four NiCad cells (connected in series) can be charged by the unit at one time.

Constructionally the unit is quite simple and straight forward, but be sure to take the safety precatuions necessitated by the fact that the unit is mains powered, and fit Tr1 with a small finned heatsink.

Baby Alarm

A baby alarm is really just a high gain amplifier fed from a remotely located microphone and driving a small loudspeaker. The audio quality of the system is not really of prime importance, and if the unit is to be battery operated a low level of quiescent current consumption is of greater importance.

The circuit diagram of the Baby Alarm is shown in Fig.42. The microphone is simply a high impedance loudspeaker used in reverse as a sort of moving coil microphone. The output of the microphone is coupled by C2 to a simple, high gain common emitter stage which uses Tr1 in a conventional arrangement. C3 provides R.F. filtering at the input and prevents R.F. signals picked up in the microphone cable from breaking through to the output.

Tr2 and Tr3 are used as common emitter amplifiers, but have a combined voltage gain of only about unity at D.C. due to

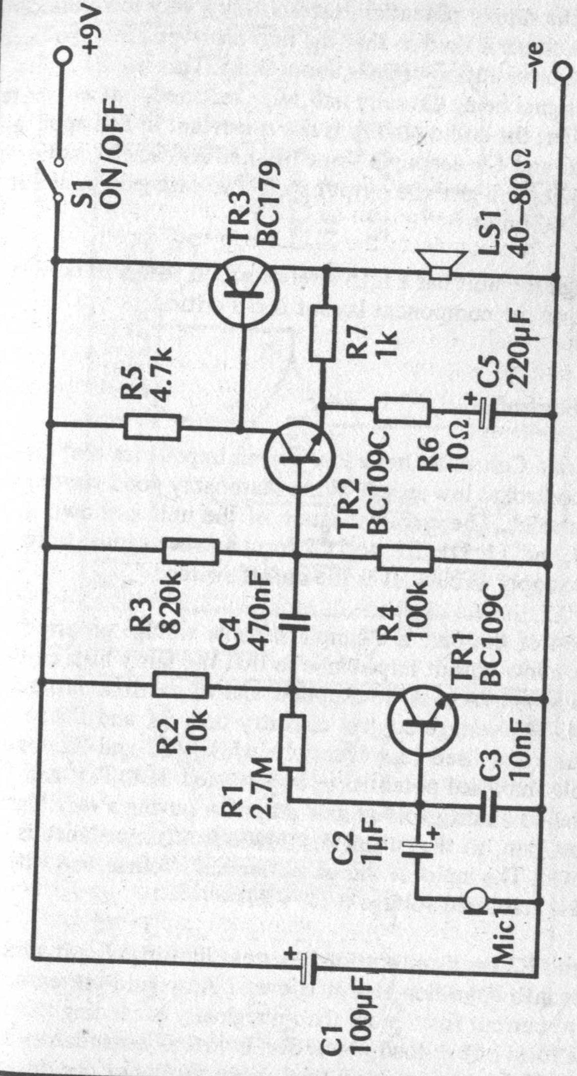

Fig.42. The circuit diagram of the baby alarm

the 100% negative feedback through R7. R3 and R4 bias the amplifier, but do not give the usual quiescent output voltage of half the supply potential. Instead only a very low quiescent output voltage is used so that the unit achieves a low quiescent current consumption of only about 3mA. This results in the output signal being virtually half wave rectified, but as pointed out earlier, the audio quality is not important in this application. R6 and C5 decouple some of the feedback at audio frequencies and give the output stage a voltage gain of about 40dB. 100 times.

Although the unit has a high level of gain it seems to be very stable and the component layout is not critical.

Train Controller

This Train Controller has a low output impedance that gives good control at low speeds plus a reasonably good starting performance. The circuit diagram of the unit is shown in Fig.43, and T1, D1, D2, and C2 form a conventional fullwave power supply circuit. S1 is the on/off switch.

The rest of the unit is a simple variable voltage generator having a low output impedance so that the fairly high current (up to about 1A.) can be supplied, and an overload protection circuit. The voltage control circuitry uses R1 and D3 to provide a stabilised 15 volt supply, while VR1 and R2 enable a variable stabilised potential to be obtained. IC1, Tr1, and Tr2 are used as a unity voltage gain amplifier having a very high current gain, so that a suitably low output impedance is obtained. The input of this amplifier is, of course, fed with the variable stabilised voltage from VR1.

Tr3 and R3 are a conventional current limiting circuit which comes into operation at a little over 1 Amp. and prevents the output current from more than marginally exceeding this level. Some form of overload protection is virtually mandatory in a circuit of this type as occasional short circuits on the output

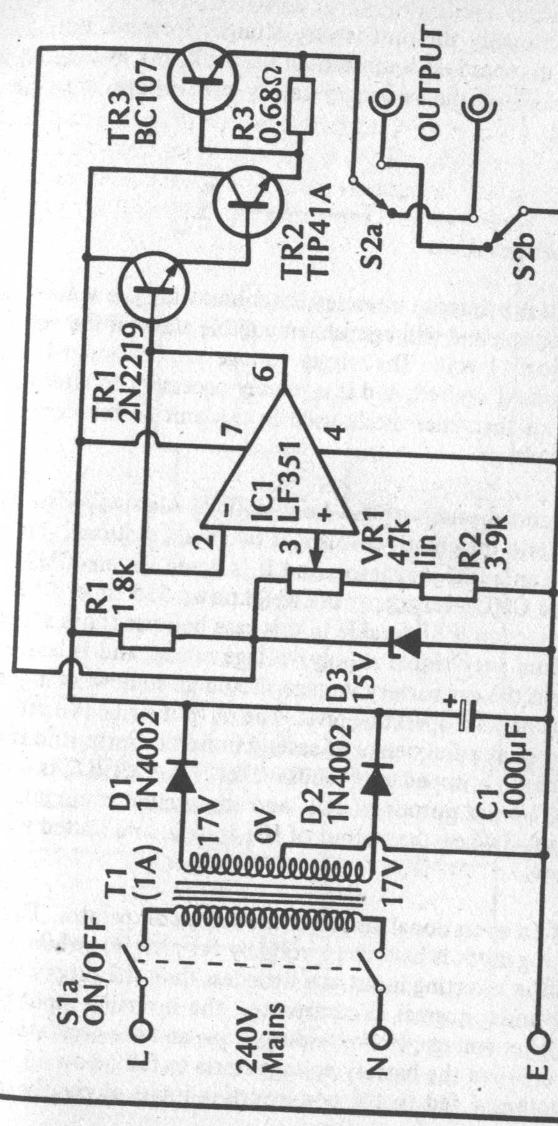

Fig.43. The circuit diagram of the train controller

are inevitable. S2 is the forward/reverse switch.

Constructionally the unit is very straight forward, but Tr2 must be mounted on a substantial heatsink, and as the unit is mains powered the necessary safety precautions must be observed.

Low Voltage Alarm

This unit is primarily intended for monitoring the voltage of a car battery, and will operate an audible alarm if the voltage falls below 11 volts. The trigger voltage can be changed to 10 or 12 volts if desired, and it is merely necessary to alter the voltage of the zener diode used in the unit to the desired trigger voltage.

The circuit diagram of the Low Voltage Alarm appears in Fig.44, and the circuit consists of two main sections. One of these is an audio oscillator, and it is based on an ICM7555 I.C. (the CMOS version of the well known 555 timer I.C.). The CMOS version is preferable in this case because it has a higher maximum permissible supply voltage rating, and is less in danger if the car battery voltage should go somewhat higher than its normal operating level. The output signal is a pulse waveform at a frequency of several hundred Hertz, and it is fed to a high impedance loudspeaker. Pin 4 of IC2 is controlled by the output of IC1, and the oscillator circuit will be enabled when the output of IC1 is high, and muted when it is low.

IC1 is an operational amplifier used as a comparator. The non-inverting input is held at 11 volts by R1 plus D1, while R4 holds the inverting input at a little less than the supply voltage. Thus, under normal circumstances, the inverting input is at the higher voltage, IC1's output is low, and the oscillator is muted. When the battery voltage starts to fall below 11 volts the potential fed to the non-inverting input obviously starts

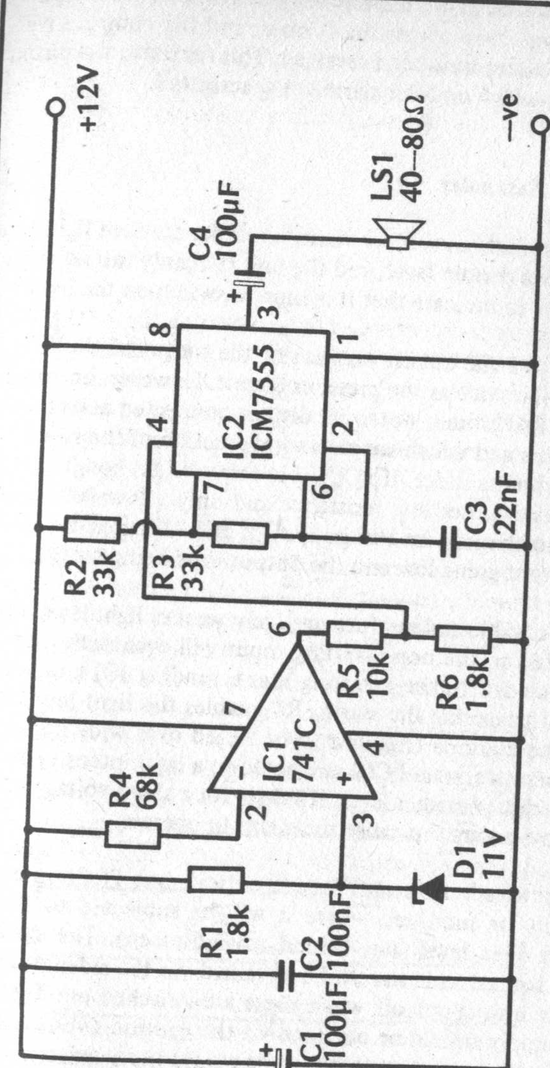

Fig.44. The circuit diagram of the low voltage alarm

to fall. However, as R1 has a lower value than R4 there is a lower voltage drop across the former, and the comparative input states have now been reversed. This results in the output of IC1 going high and the alarm being activated.

Lights-On Reminder

This unit sounds an audible alarm when the ambient light level falls below a certain level, and the unit is mainly intended for use in a car to indicate that it is time to switch-on the lights.

Fig.45 shows the circuit diagram of the unit, and this is basically the same as the previous circuit. However, one input of IC1 is fed from a potential divider connected across the supply lines and which supplies some fraction of the supply voltage from its slider. If PCC1 is in fairly bright conditions it exhibits a rather low resistance and only a low voltage is applied to the non-inverting input of IC1. This results in IC1's output going low and the output tone being muted.

If PCC1 is subjected to a progressively weaker light level, the voltage fed to the non-inverting input will eventually rise above that fed to the inverting input, sending IC1's output high and triggering the alarm. R1 enables the light level at which the alarm is triggered to be varied over wide limits, and in practice, with PCC1 subjected to a light intensity equal to the required trigger level, R1 is set for a slider voltage which is just low enough to cause the alarm to operate.

Constructionally the unit is not critical, but PCC1 must obviously be mounted where it will be subjected to the ambient light level (e.g. behind a windscreen). The earth supply for the unit should be obtained via the sidelights so that the unit is cut off when these are switched on. The other supply should be obtained via the ignition switch so that the unit does not operate when the car is not in use.

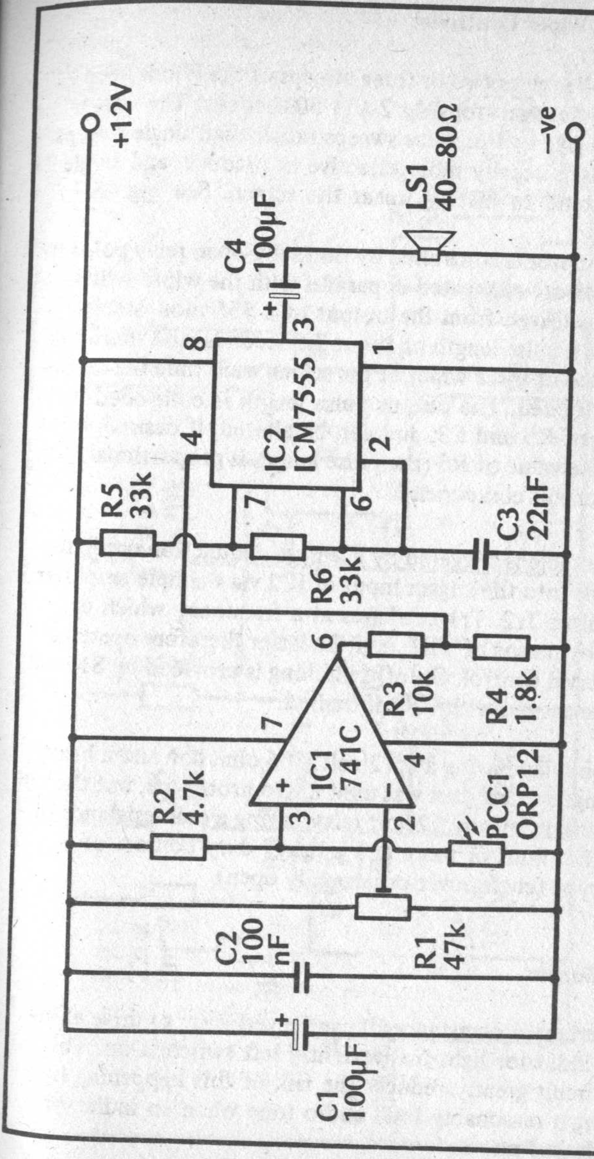

Fig. 45. The circuit diagram of the lights—on reminder

Windscreen Wiper Controller

This controller gives two or three sweeps of the Windscreen at intervals of between roughly 2 and 50 seconds. The unit is designed to give two or three sweeps rather than single sweeps because this is usually more effective in practice, and single wipes can tend to merely smear the screen. See Fig.46.

The wiper motor is controlled by normally open relay contacts RLA1 which are connected in parallel with the wiper switch. The relay is driven from the output of a 555 monostable which gives a pulse length of about 2.5 seconds, and therefore gives the two or three wipes of the screen each time the monostable is triggered. The output pulse length is controlled by the values of R5 and C3, and can be altered, if desired, by changing the value of R5 (the pulse length is proportional to the value of this component).

The monostable is triggered by a simple unijunction oscillator which feeds into the trigger input of IC2 via a simple amplifier which utilizes Tr2. Tr1 oscillates at a frequency which can be varied by means of VR1, and the latter therefore operates as the interval control. On/off switching is provided by S1, and this can be ganged with VR1 if desired.

An Omron relay having a 6/12 volt, 306 ohm coil and a heavy duty changeover contact was used in the prototype, but the unit should work using any 12 volt relay having a coil resistance of around 200 ohms or more and a heavy duty contact of a suitable type (changeover or normally open).

Flasher Buzzer

Under certain circumstances it can be very easy to drive along with an indicator light inadvertently left switched on. This simple circuit greatly reduces the risk of this happening by producing a reasonably loud audio tone when an indicator light is flashed on.

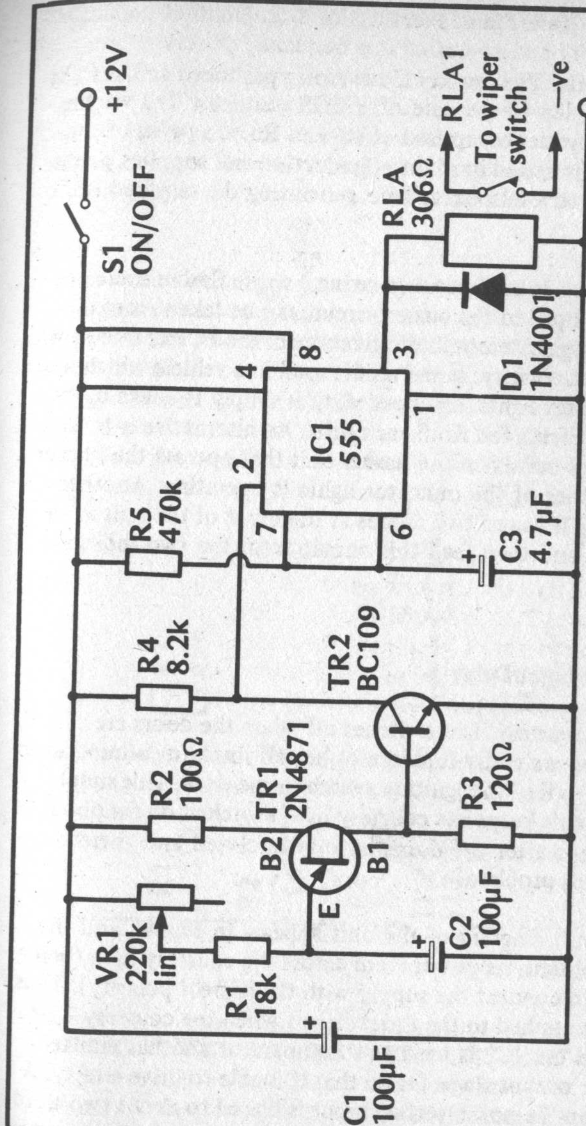

Fig.46. The circuit diagram of the windscreen wiper controller

The circuit (see Fig.47) consists of a unijunction oscillator based on Tr1 and operating at a frequency of very approximately 1kHz. The sawtooth waveform produced across C2 is direct coupled to the gate of VMOS transistor Tr2 via the potential divider comprised of R4 and R5. On peaks of this signal Tr2 is biased hard into conduction and supplies pulses of current to loudspeaker LS1, generating the required audio tone.

If the circuit is used in a car having a single flasher indicator light, the input to the buzzer circuit can be taken from the indicator light, being careful to connect the buzzer circuit with the correct polarity. If the unit is used in a vehicle which has two indicator lights, one possibility is simply to make up two units, one being fed from each light. An alternative is to find connection points on the flasher unit that operate the buzzer unit if either of the indicator lights is operating. Another possibility is to use two diodes at the input of the unit as an OR gate, and then feed this circuit from the two indicator lights.

Courtesy Light Delay

As a car courtesy light switches off when the doors are closed, the driver can easily find him or herself fumbling around in an attempt to find the ignition switch in the dark. This simple delay circuit keeps the courtesy light switched on for about 20 seconds after the doors have been closed and therefore avoids this problem.

The circuit diagram of the unit appears in Fig.48, and the circuit obtains its power from across the courtesy light (being careful to connect the supply with the correct polarity). Thus power is applied to the timer circuit when the courtesy light is switched on. IC1 is used as a comparator and has emitter follower output stage Tr1 so that it is able to drive a relay. A switch on the non-inverting input is biased to about two thirds of the supply voltage by R1 and R2, but the inverting input is

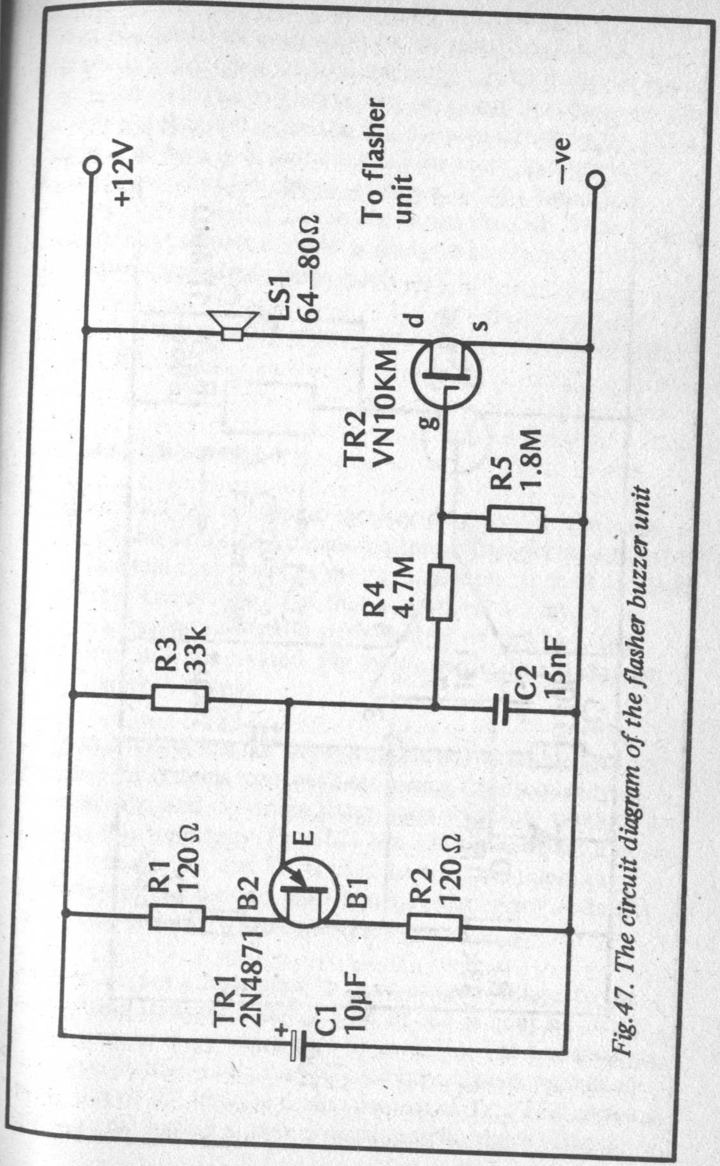

Fig.47. The circuit diagram of the flasher buzzer unit

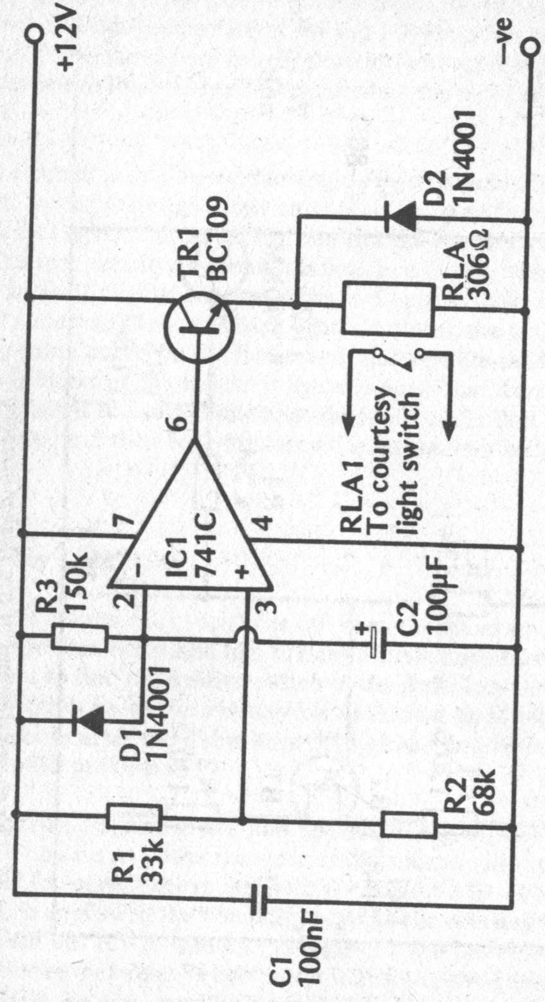

Fig.48. The circuit diagram of the courtesy light delay

at zero volts at this stage since C2 will be uncharged. The output of IC1 therefore goes high and the relay is activated. A pair of normally open relay contacts then latch the timer and courtesy light in the "on" state, but only until the charge on C2 exceeds the bias voltage fed to IC1's non-inverting input. IC1's output then goes low, switching off the relay, and provided the door switches are closed, switches off the timer and courtesy light as well. C2 then discharges through D1 and the courtesy light so that the unit is ready to start another timing run and no significant charge is left on C2.

The delay time is easily altered, and is proportional to the value of R3.

Ultrasonic Transmitter

Together with the ultrasonic receiver described in the next section of this book, this ultrasonic transmitter forms a remote control system that operates over a maximum range of about 12 metres. The system is of the type where a relay at the receiver is activated when a push button on the hand-held transmitter unit is operated. Fig.49 shows the circuit diagram of the transmitter unit.

Ultrasonic transducers of the type generally available for use in ultrasonic systems have peak efficiency at a frequency of about 40kHz, and the transmitter must therefore produce a signal at this frequency. Here IC1 is a 555 timer device used as a simple oscillator, and R2 is used to trim the operating frequency of the unit to the frequency that gives optimum range. This is done by trial and error.

In order to give a high peak to peak output voltage to the transmitting transducer the output of IC1 is coupled to a simple inverter stage which uses Tr1 and R3. The transmitting transducer is then fed with the antiphase signals present at the output of IC1 and the drain terminal of Tr1. This gives an output signal having a peak to peak amplitude of almost

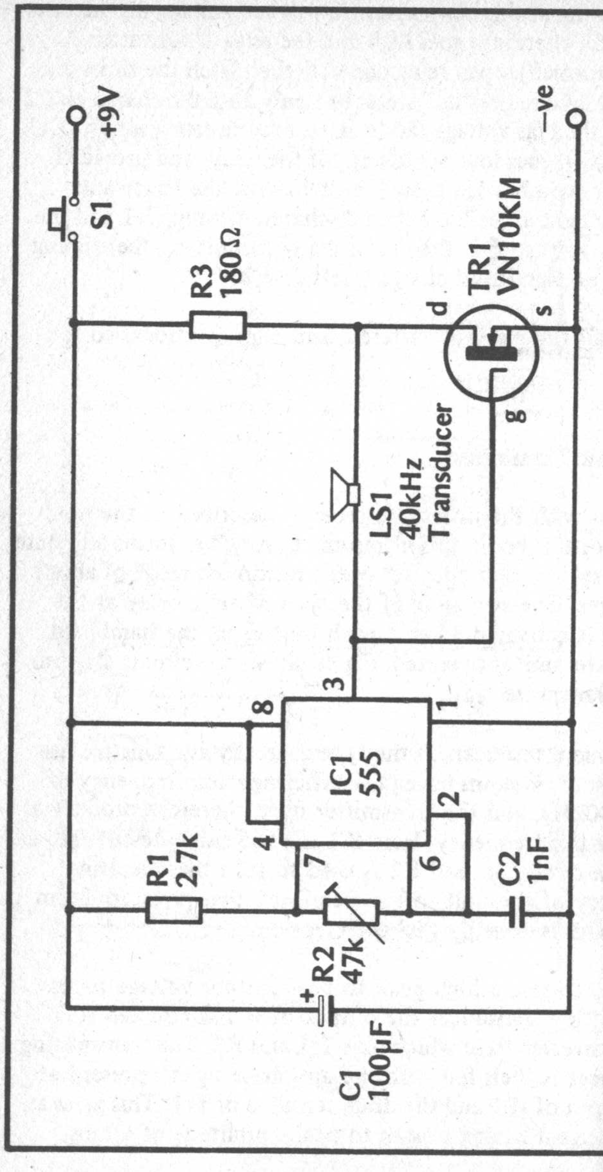

Fig.49. The circuit diagram of the ultrasonic transmitter

double the supply voltage.

The circuit should work well with any normal 40kHz ultrasonic transducer, but note that some types have one transducer for use in the receiver and a slightly different transducer for use in the transmitter. The transducers must be used in the correct circuits in these cases if optimum results are to be obtained. The retailers or manufacturers data will show whether or not the transducers are of this type, and if so, which transducer to use in which circuit.

Ultrasonic Receiver

In order to give a useful range the receiver circuit must provide a high level of gain as the output from the receiving transducer will normally be less than a millivolt. Two stages of A.C. amplification are used in this circuit (Fig.50), and the first of these uses operational amplifier IC1 in the non-inverting mode, and with a voltage gain of approximately 40dB. (100 times). The second stage uses Tr1 as a common emitter amplifier having a similar voltage gain to the input stage.

The output from Tr1 is coupled by C3 to a simple detector circuit which is based on Tr2, and produces a strong D.C. bias across C4 and R7 in the presence of a suitable input signal. This signal is used to drive the relay, and Tr3 is used as an emitter follower buffer stage which operates the relay from the relatively high impedance output signal from Tr2. D3 is a L.E.D. indicator that is switched on in unison with the relay. The controlled equipment is powered by a set of normally open relay contacts (RLA1).

As in the previous circuit which employed a relay, this can be any type having a resistance of about 200 ohms or more, an operating voltage of 6/12 volts, and suitable contacts for the intended load. Due to the high gain and fairly high bandwidth of the circuit, the component layout is inevitably quite critical, and must be carefully designed. As with any unit of

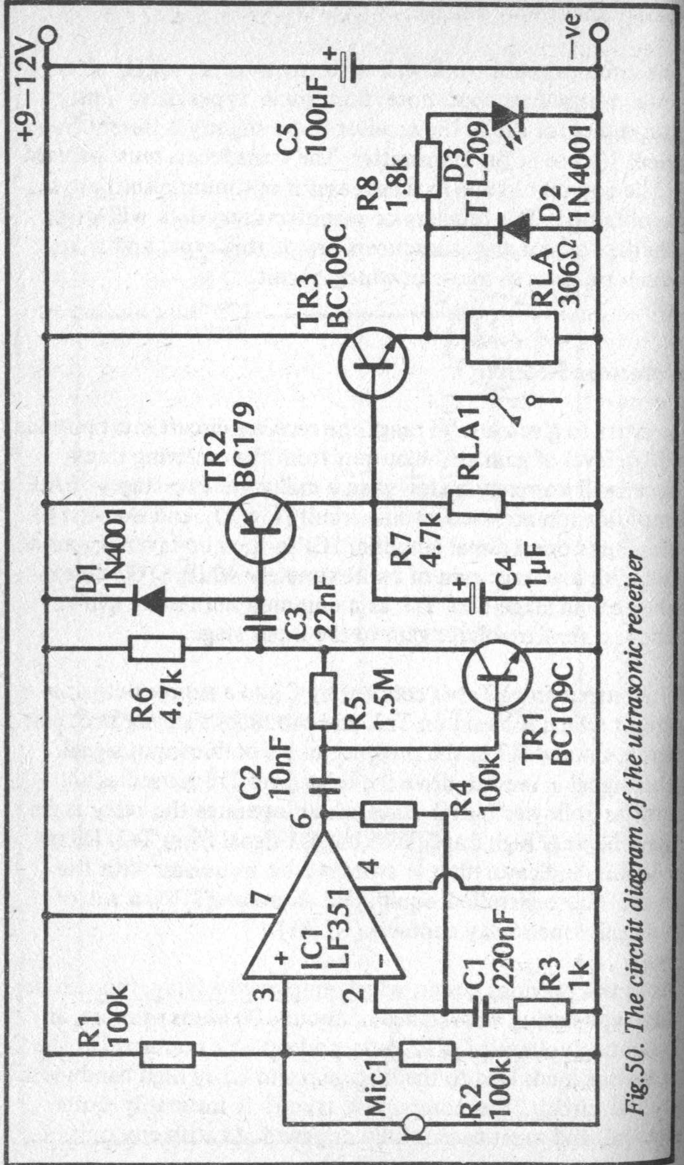

Fig.50. The circuit diagram of the ultrasonic receiver

this type, the transmitter is highly directional and must be aimed at the receiver for best results.

Infra-Red Transmitter

In conjunction with the infra-red receiver unit described in the next section of this book, the infra-red transmitter described here acts as a burglar alarm of the type where breaking a light beam triggers the alarm. Since the beam is infra-red it is not visible, and is not readily detectable. The alarm briefly makes or breaks a set of relay contacts when activated, and the relay contact is wired into the detector switch circuit of an existing alarm system. The unit does not have built-in latching, and is not suitable for use as a burglar alarm when used on its own.

The circuit diagram of the transmitter is shown in Fig.51, and is very straight forward. Systems of this type use a modulated infra-red beam since an unmodulated beam and D.C. detector circuit would be susceptible to false alarms due to changes in the ambient infra-red level. Modulated beam systems, for all practical purposes, are immune to this problem.

This circuit simply consists of a 555 timer I.C. used in the astable mode, and driving an infra-red L.E.D. via a VMOS switching transistor. The L.E.D. is pulsed by a current of about 90mA., and at a frequency of about 2kHz. The specified infra-red L.E.D. does not give a significant output in the visible light spectrum, and it is therefore unnecessary to fit it with an infra-red filter.

Infra-Red Receiver

The circuit of the infra-red receiver is the same as that used in the ultrasonic receiver (Fig.50) which was described earlier in this book. However, the ultrasonic transducer must be replaced by an infra-red detector of some kind, and Fig.52 shows the circuit diagram of a suitable detector. D1 is an infra-

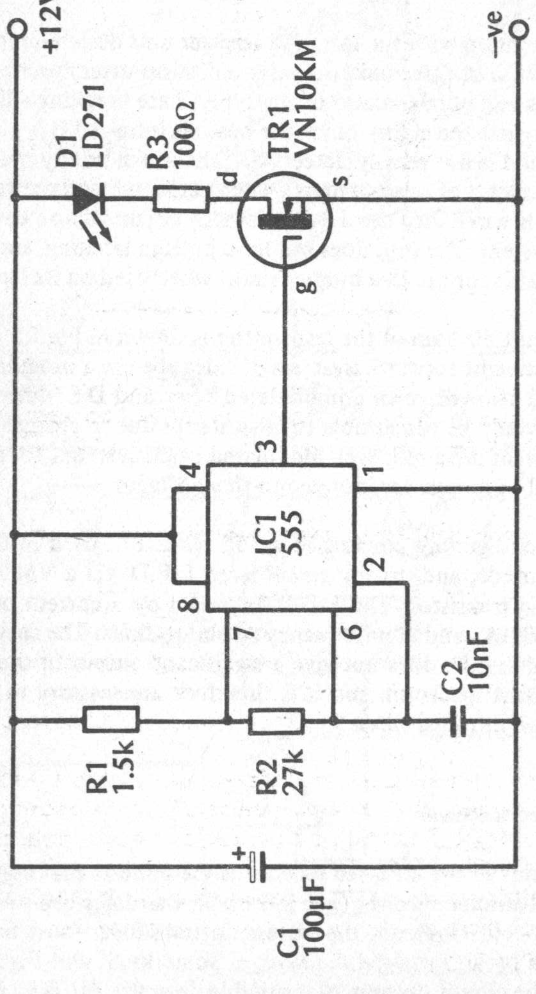

Fig.51. The circuit diagram of the infra–red transmitter

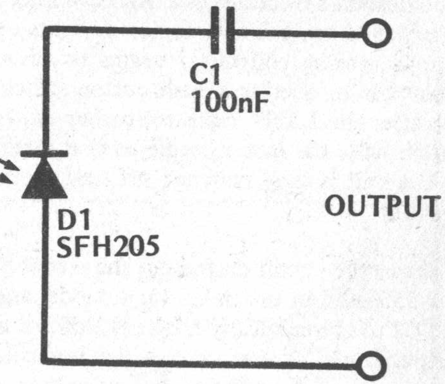

Fig.52. The circuit of Fig.50. can be modified to an I.R. receiver by using this detector in place of the ultrasonic transducer

red photo-diode that has a built-in infra-red filter, and does not require a separate filter. When subjected to the pulsed infra-red beam D1 produces a small voltage pulse in response to each pulse of infra-red light, and these are coupled to the input of the receiver circuit by D.C. blocking capacitor C1.

The circuit gives a maximum range of about 20 feet, and this should be adequate for most purposes. The output of the transmitting L.E.D. should be directed towards the sensitive (curved) surface of the detector diode, and the receiver and transmitting units should obviously be constructed in such a way that there is a clear path from the emitting diode to the receiving one.

Reaction Tester

This simple reaction testing game gives a relative indication of the contestant's reaction time. About ten seconds after the unit is switched on, a L.E.D. indicator lights up, and the needle of a moving coil meter begins to advance. The contestant has to operate a push button switch as soon as possible after the L.E.D. indicator comes on, and operating this switch halts the meter needle so that a reading can be taken. The unit is then switched off, and is ready to start another run.

Fig.53 shows the circuit diagram of the reaction tester unit. IC1 is a 555 used in the monostable mode, and having R2, R3, and C2 to automatically trigger the circuit at switch on. The output at pin 3 therefore goes high immediately, but only for about ten seconds as the output pulse then ends. L.E.D. indicator D1 then switches on, and C3 (which rapidly becomes fully charged at switch-on) starts to discharge through R4, S1, and IC1. This gives a rising voltage across ME1 and R6 which are fed from C3 via the emitter follower buffer stage formed by Tr1. The needle of ME1 therefore starts to advance, and continues to do so until S1 is operated and the discharge circuit of C3 is broken. ME1's needle then stays at a constant setting for a fairly long period of time so that a reading can be easily taken. Note that C3 does not discharge at a linear rate, and the unit does not have linear scaling. It is merely intended to give a relative indication of reaction time.

Construction of the unit is very straight forward since it is a D.C. circuit, and the component layout is in no way critical. The current consumption is about 8mA.

Two Tone Horn

This circuit (Fig.54) is intended for use with model railways, and gives a two tone horn sound, as used on some electric and diesel locomotives. Horns of this type usually have an initial

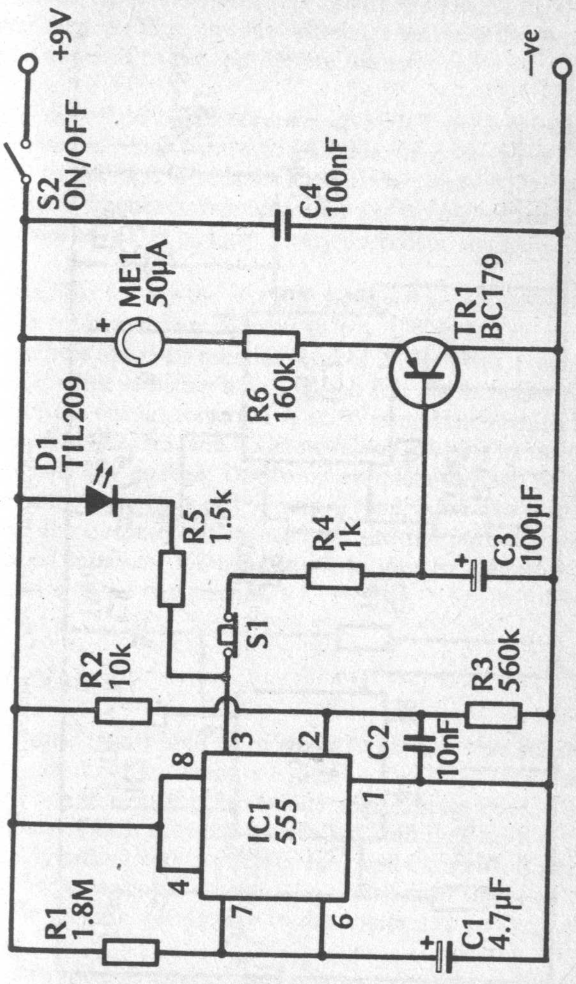

Fig. 53. The circuit diagram of the reaction tester

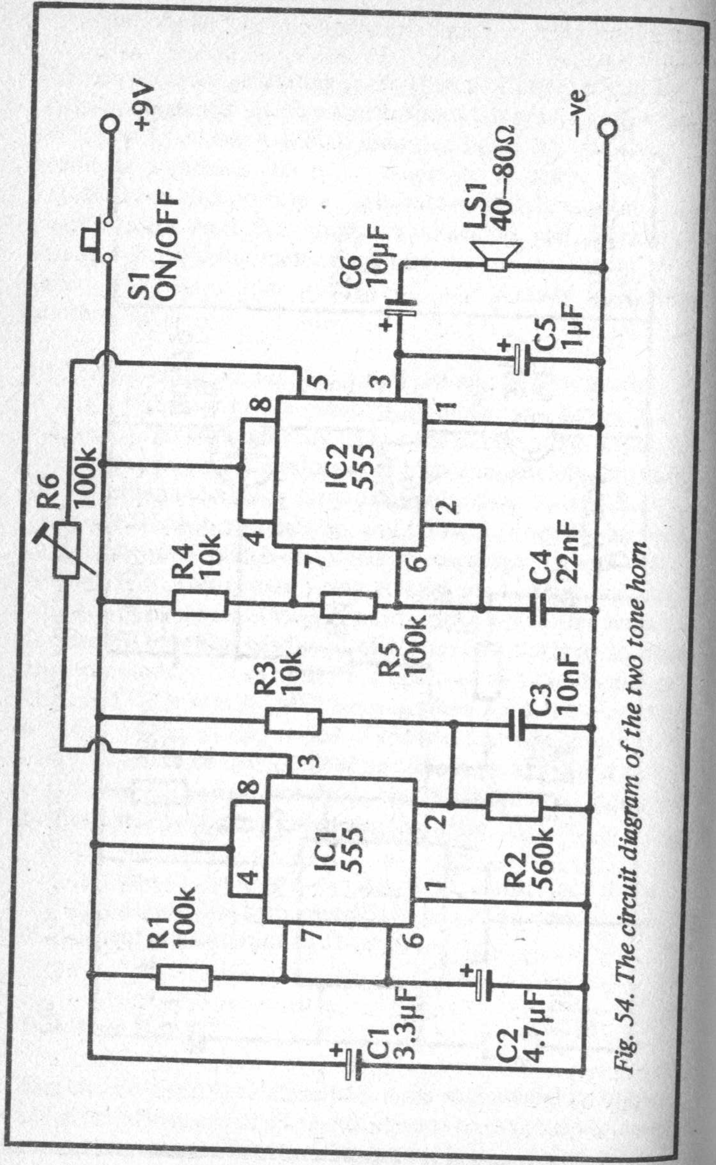

Fig. 54. The circuit diagram of the two tone horn

tone of about half a second or so in duration, followed by a higher or lower pitch sound, the difference in pitch being about half an octave. With this circuit the second note is higher than the first, and the difference between the two notes can be adjusted to the appropriate amount.

The audio output signal is generated by IC2 and its associated components, which form a 555 astable. C6 couples the output of IC2 to the high impedance loudspeaker, and C5 attenuates the higher frequency harmonics on the output which are otherwise excessive and give a very unrealistic sound.

It is possible to vary the pitch of the signal generated by IC2 using a control voltage applied to pin 5 of IC2. Taking the voltage here above its nominal level of 2/3V+ gives a reduction in pitch, while reducing the voltage at this pin increases the pitch of the output tone. IC1 is a 555 monostable which is triggered by R2, R3, and C3 at switch-on, causing its output to immediately go high. Due to the coupling through R6 this reduces the frequency of the output tone. After about half a second the output of IC1 goes low, and the output tone is increased in pitch. R6 is adjusted to give the required difference in the two pitches.

Points Controller

The electric points used on model railways have two solenoids; one to set the point from one state to the other, and the second one to reset the point to its original state again. The resistance of each solenoid is quite low, and they must only be briefly pulsed with current or they will burn out. A points controller that simply consists of a switch must therefore be used very carefully if damage to the points is to be avoided.

A better type of points controller is one where the solenoids can only be briefly pulsed on, and cannot be burnt out. Such a points controller circuit is shown in Fig.55. This circuit is powered from a 15 volt A.C. output of the train controller,

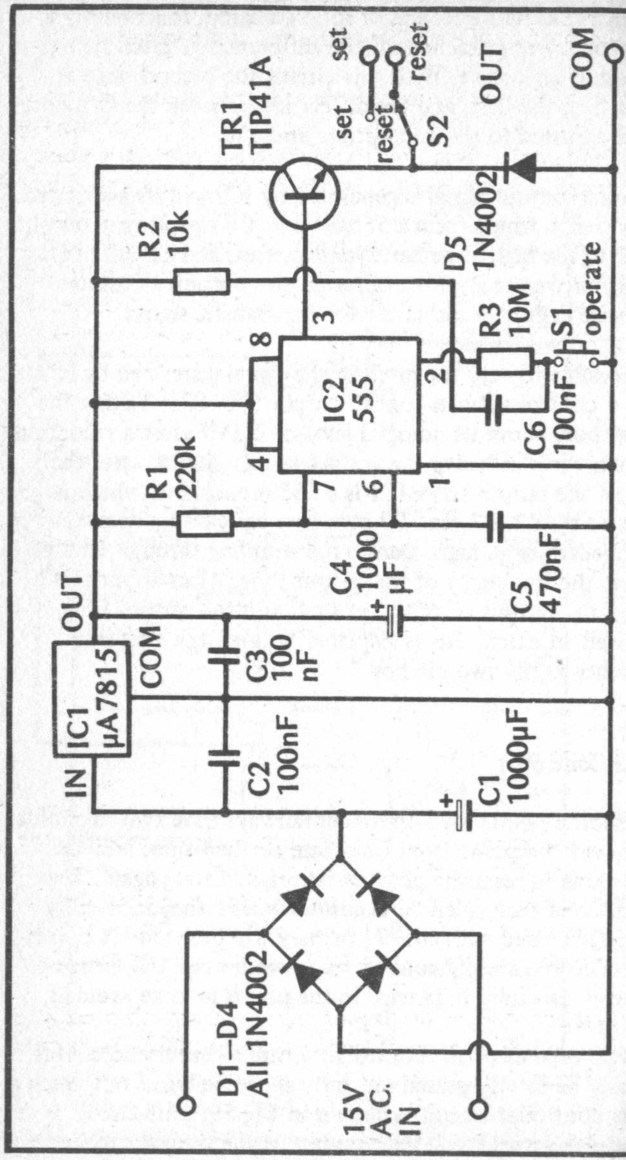

Fig. 55. The circuit diagram of the points controller

and this supply is rectified by D1 to D4 and then smoothed and regulated by IC1 and its associated components.

IC2 is a 555 timer device used in the monostable mode and having an output pulse length of about 100ms (which is quite sufficient a pulse length to reliably operate an electric point). Tr1 is used as an emitter follower stage that enables the high current needed by the point to be supplied by the circuit, and D5 is simply a protective diode. S2 selects the solenoid of the point that is connected to the output of the controller. S1 is operated in order to trigger IC2 and activate the point. R3 discharges C6 so that the unit is soon ready to operate again when S1 is released, but the discharge time of C6 is such that there is a delay of a second or so before the point can be pulsed again, and this ensures that there is no possibility of the point being burnt out.

Although the circuit shown here is only suitable for use with a single point, the unit is easily modified to enable several points to be controlled. S2 would need to be duplicated for each point to be controlled, and then the pole of these switches would be connected to the output of the controller via a selector switch.

Call Charge Reminder

When making telephone calls it is remarkably easy to lose track of the passage of time, with surprisingly high telephone bills in consequence. This simple circuit can be used to give an audible signal (a "clicking" sound) that indicates the commencement of each new unit, and this prevents one from inadvertently making excessively long and expensive calls.

Fig.56 shows the circuit diagram of the unit which is really little more than a 555 astable circuit. The ICM7555 version of the 555 timer is used here in order to minimise the current consumption of the unit, which is approximately 4mA. D1 is a steering diode that gives C1 a rather brief charge time, and

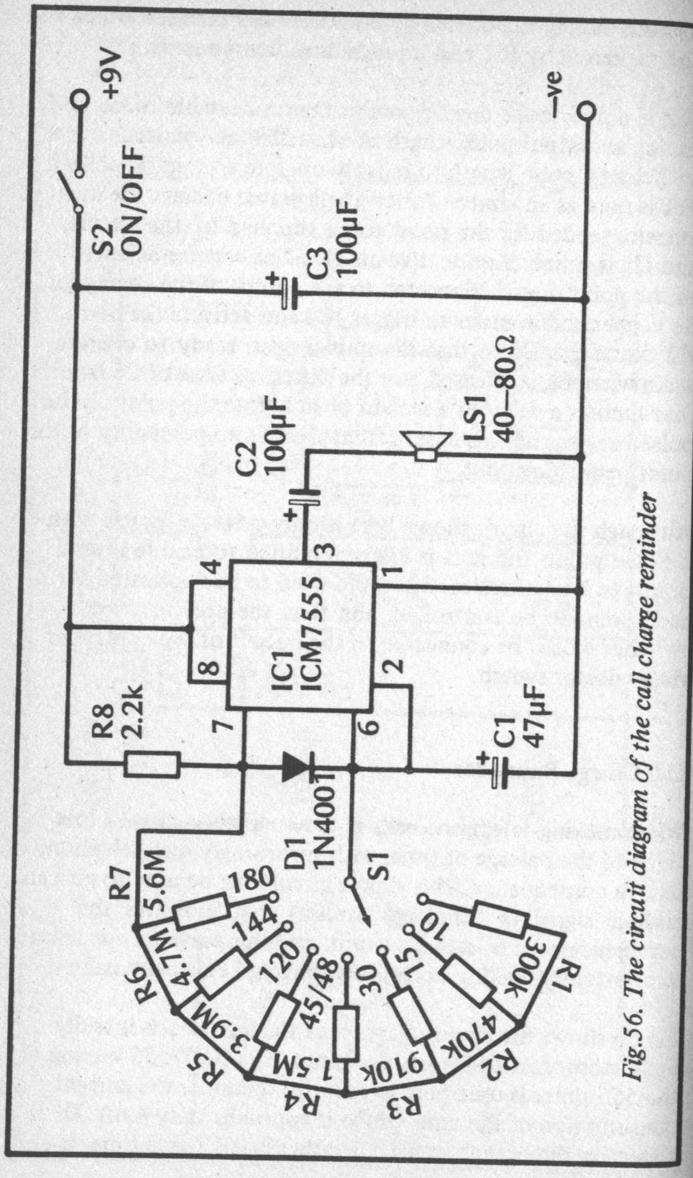

Fig.56. The circuit diagram of the call charge reminder

this triggers IC1 to the discharge mode almost immediately at switch-on, and a "click" sound is produced by the loudspeaker due to the pulse it receives. In practice the unit is switched on when the person being called answers the phone, and the "click" sound indicates that the first unit has been started.

C1 now starts to discharge through whichever of the seven timing resistors is selected using S1, and the appropriate switch position is selected after first consulting the telephone call charge booklet to find the correct unit time for the call being made. The times between output pulses are only approximations of the specified times, but in this application a high degree of accuracy is not really necessary, since the unit is not intended to be a call charge calculator.

C1 must be a high quality component if the unit is to function properly, and a tantalum bead capacitor is ideal.

CHAPTER 5

MISCELLANEOUS CIRCUITS

Gas And Smoke Detector

This gas and smoke detector is mainly intended for use in a boat or caravan having a 12 volt electrical system, and using bottled gas for cooking and (or) heating. The unit will respond to butane, methane, most smoke, practically any inflammable gas or vapour in fact.

Fig.57 shows the circuit diagram of the unit, and the circuit is built around a TGS812 gas sensor. This has a heating element which requires a 5 volt supply, and monolithic voltage regulator IC2 is used to provide a stabilised 5 volt supply from the 12 volt input. The entire circuit is powered from the 12 volt supply, and not just the heating element. The element heats a piece of semiconductor material, and has an oxidising effect which produces a high resistance through this material. The semiconductor section of the sensor is connected to form a potential divider in conjunction with R1, and the latter is adjusted to give an output voltage from the divider of a little under 0.5 volts under quiescent conditions.

IC1 is a 555 timer device used as an audio oscillator and feeding a loudspeaker. However, under quiescent conditions the voltage applied to pin 4 of IC1 is insufficient to permit the device to operate, and no tone is emitted from the loudspeaker. When the sensor detects a suitable concentration of inflammable gas or vapour (the unit responds to concentrations well below the lower explosive limit incidentally), the semiconductor material is partially de-oxidised, and its resistance greatly reduces as a result. The voltage fed to pin 4 of IC1 therefore increases considerably, and the oscillator begins to operate with a warning tone being produced by the speaker in consequence.

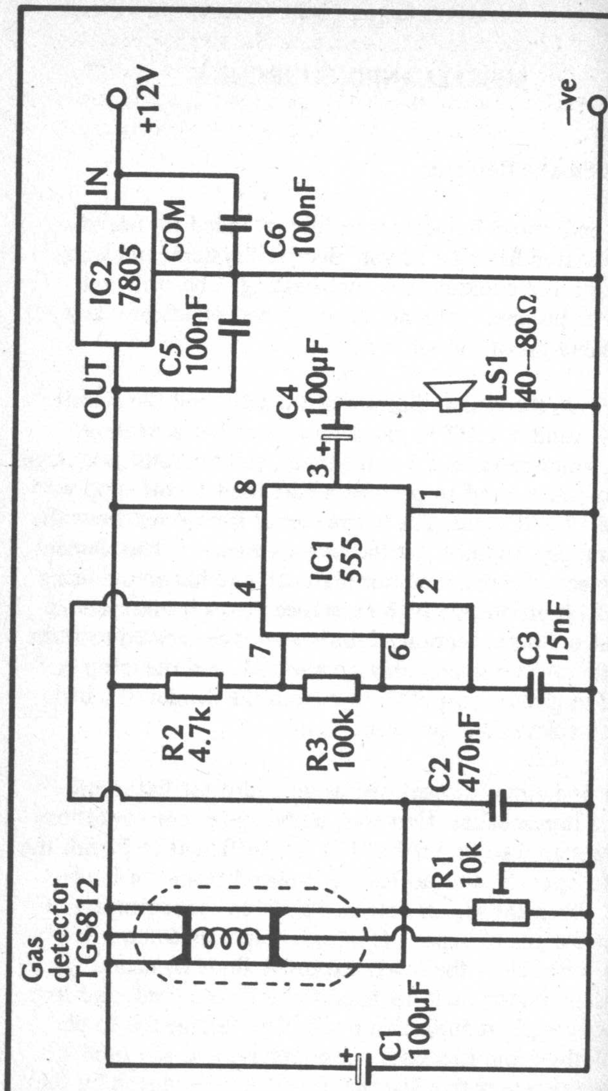

Fig.57. The circuit diagram of the smoke and gas detector

Construction of the unit is very straight forward, and is made even easier if the gas sensor is fitted into the special holder that is available (both the sensor and the holder are available from Watford Electronics). The gas sensor is symetrical, and can be connected either way round. The current consumption of the unit is about 120mA.

R1 is adjusted for the highest resistance that does not cause the alarm to operate, or false alarms to be obtained.

One Second Flasher

When making a long exposure using a camera with the shutter set to "B" or "T", some sort of timer is needed in order to obtain accurate results. Unless very long exposures are required, a simple one second flasher such as the one described here is probably the ideal solution. The unit simply pulses on a L.E.D. indicator at one second intervals. The shutter is opened as the L.E.D. flashes, and then closed after the appropriate number of flashes have been counted off. There are almost certainly a great many other uses for a simple timer of this type, and it could perhaps be used as a simple enlarger timer for example.

The circuit diagram appears in Fig.58, and is based on an operational amplifier that is biased by R1, R2, R3 to act as a form of Schmitt trigger. The output goes to the low state if the inverting input is taken about 2/3V+, and high if it is taken below 1/3V+. The output therefore goes high initially, but C2 soon charges to 2/3V+ via R4, and then the output goes low. C2 then discharges to 1/3V+ via R4, sending the output high again, and producing continuous oscillation.

R4 is adjusted to give an operating frequency of one Hertz (the unit being calibrated against a watch or clock with a seconds hand by empirical means). The output of IC1 is coupled to the L.E.D. indicator by way of D.C. blocking capacitor C3 and current limiting resistor R5, and the L.E.D.

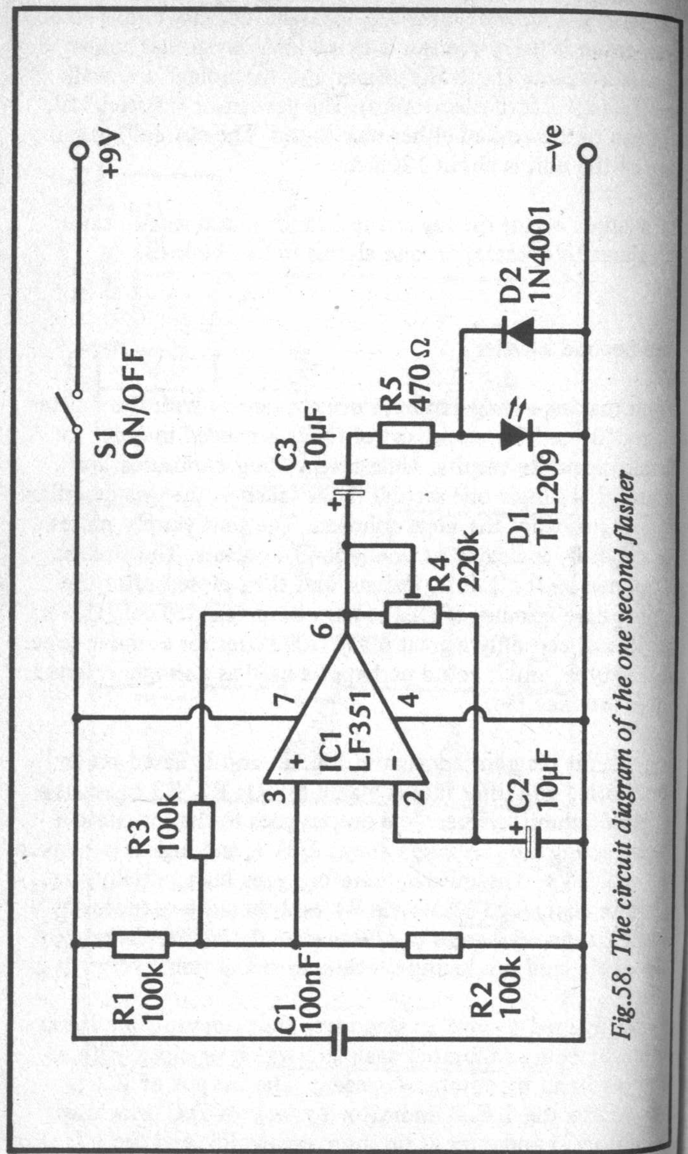

Fig.58. The circuit diagram of the one second flasher

is briefly pulsed on as the output swings positive. D2 ensures that there is both a charge and discharge path for C3 so that the output signal is properly coupled to D1.

The current consumption of the unit is only about 2mA.

Sound Activated Switch

The sound activated switch circuit of Fig.59 is suitable for use in a sound activated tape recorder and similar applications. Provided a reasonably sensitive crystal microphone or insert is used, the unit will operate from speech of normal volume at a range of several feet.

The circuit uses IC1 as a low noise preamplifier having a voltage gain of 270 times, and this is followed by a second stage of amplification which gives a similar voltage gain. The second stage uses Tr1 in the common emitter mode and this is a conventional arrangement except for the inclusion of C4. This gives a considerable amount of high frequency attenuation, and this is necessary in order to prevent instability.

C5 couples the greatly amplified output of Tr1 to a rectifier and smoothing circuit which gives a positive D.C. output signal which is roughly proportional to the input signal level. If the input signal is sufficiently strong, the bias voltage at the gate of VMOS device Tr2 will be adequate to bias this transistor into conduction, and the relay which forms its drain load is then activated. A pair of normally open relay contacts are used to control whatever item of equipment is operated by the unit. Of course, the voltage at the gate of Tr2 soon decays as C6 discharges through R8 if the input signal ceases, and the relay is then switched off. The decay time is roughly one second, which is about the optimum time for most applications. The attack time of the circuit is only a fraction of a second, and the unit responds almost immediately when a sound is initially picked up by the microphone.

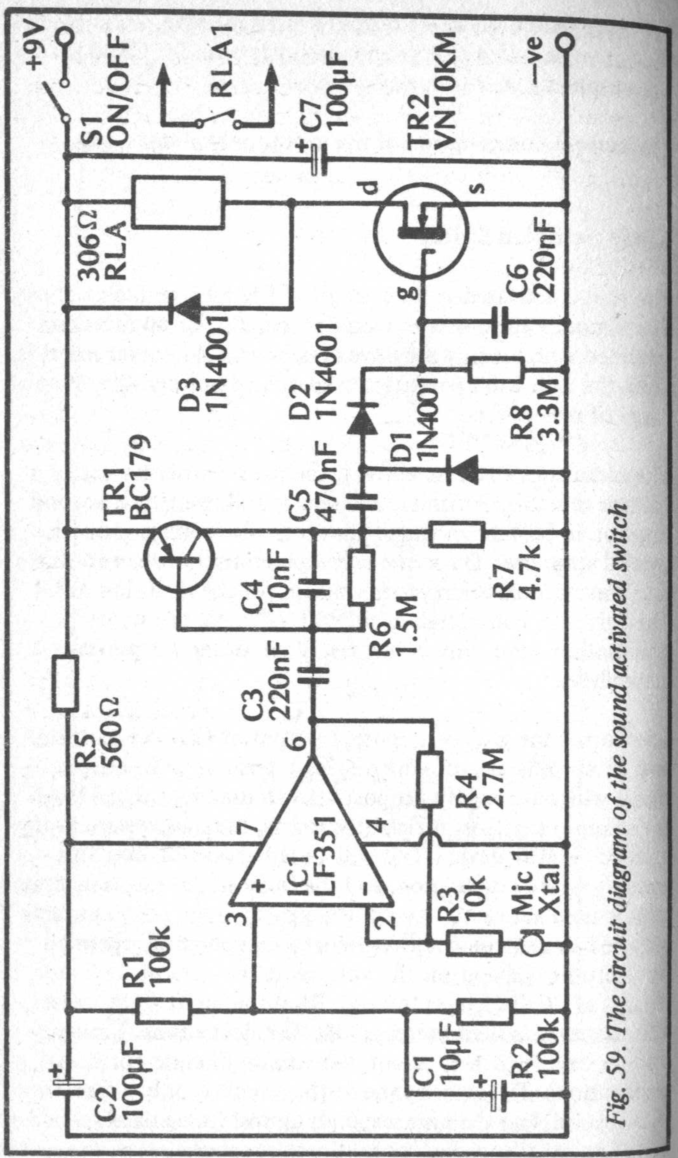

Fig. 59. The circuit diagram of the sound activated switch

As the circuit has a fairly high level of gain the component layout must be designed with reasonable care, although the very limited bandwidth of the circuit greatly reduces the risk of instability. The current consumption of the unit is only about 3mA. under quiescent conditions, but this increases considerably when the relay is activated.

Flash Slave Unit

Photographs taken using a single flashgun for illumination often tend to look rather unnatural due to the rather harsh shadows, and the use of two or more flashguns usually gives much better results. A flash slave unit such as the one described here is very useful for multiple-flash shots as it enables the additional flashguns to be triggered without the need for any connecting leads. Instead, each additional flashgun is connected to a slave unit which is a form of light activated switch, and triggers the additional flash unit(s) when it senses the pulse of light from the primary flashgun.

Fig.60 shows the circuit diagram of the unit, and the photosensitive device is a photo-Darlington transistor (Tr1). This has good sensitivity plus the fairly high operating speed that is essential in this application. The photo-Darlington device is actually a little over-sensitive, and is likely to saturate under bright ambient conditions, and R1 is therefore used to reduce the sensitivity of the device slightly. Tr1 is connected with R2 to form a potential divider across the supply lines, and this gives a negative output spike when the primary flashgun fires. This is coupled to Tr2 by C2, causing Tr2 to conduct heavily and trigger the triac. The latter fires the flashgun. R3 is a current limiting resistor.

Ideally the unit should be housed in a clear case so that light can readily reach Tr1. Alternatively a small hole can be drilled in the case, and the sensitive surface of Tr1 (the curved surface) is then positioned behind this. Flash sockets can be obtained from a few sources, but a simple alternative is to use

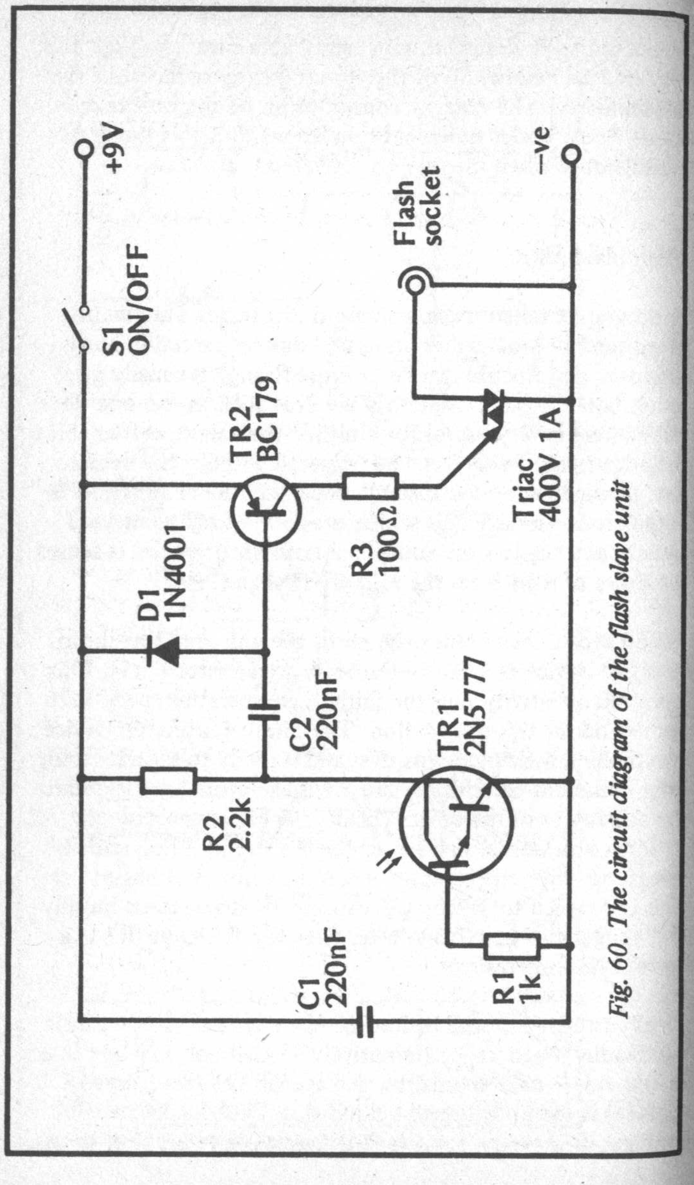

Fig. 60. The circuit diagram of the flash slave unit

a socket cut from one of the flash extensions leads that are readily available. Under some circumstances, such as when used out-of-doors, the unit is highly directional and the photo-Darlington must be aimed at the primary flashgun in order to obtain reliable results. The unit should work reliably over a range of more than 10 metres it it is set up properly.

The quiescent current consumption of the unit is typically well under 1mA.

'Heads' Or 'Tails'

This extremely simple circuit (Fig.61) electronically simulates the tossing of a coin. It has two L.E.D. indicators, only one of which remains lit when a push button switch is released. One L.E.D. is designated "heads" and the other is designated "tails".

Basically the circuit is just a 555 astable which drives two L.E.D. indicators from its output (D1 and D2). D2 is switched on when the output of IC1 is high, and D1 is switched on when the output is low. When IC1 is oscillating D1 and D2 switch on alternately as the output of IC1 switches from one state to the other, and as the value of R1 has been made high in comparison to that of R2, the output waveform is a square-wave having a mark space ratio of almost exactly one to one. This action cannot be perceived by an observer since IC1 is oscillating at a frequency which is slightly too high to permit this.

Of course, IC1 can only oscillate if S1 is closed, as both the charge and discharge paths of C2 are otherwise interrupted. When S1 is released, oscillation ceases, and IC1 latches with whatever output state it happened to have at the instant S1 broke the circuit. There is an evens chance as to which output state this happens to be, and to whether the "heads" or "tails" L.E.D. is the one which is left switched on. The required simulation of a coin being tossed is thus obtained.

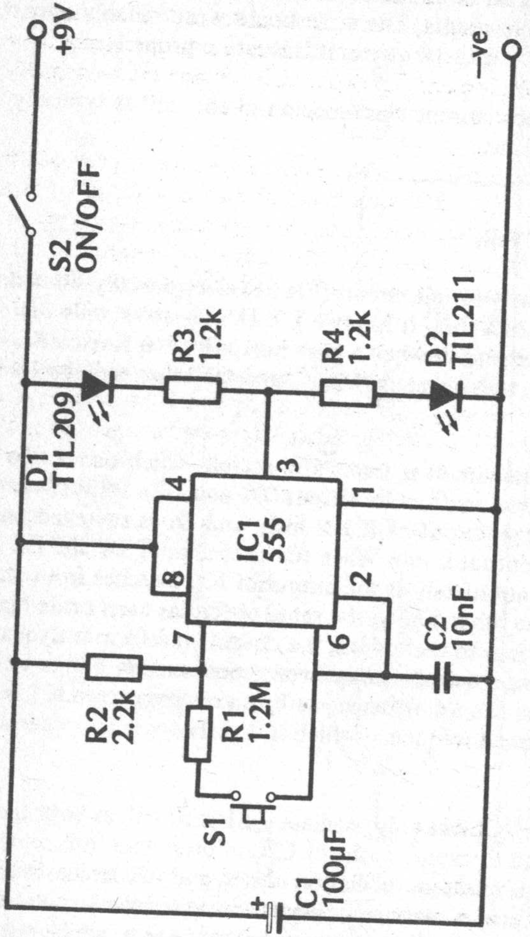

Fig. 61. The circuit diagram of the "heads" or "tails" unit

Tone Generator

This tone generator covers a frequency range of about 200 Hertz to 2kHz in one range. It drives a built-in loudspeaker, but there is an output socket that can be used to take the 8 volt peak to peak, low impedance output signal to other items of equipment. The internal loudspeaker is automatically shut off when the output socket is used.

The circuit diagram of the Tone Generator is provided in Fig.62. This is a 555 astable circuit having a virtually square-wave output waveform, and the operating frequency is adjusted by means of VR1. The harmonic content on the output of the unit would be rather high unless some filtering was to be used, and C3 is therefore used to attenuate the higher frequency harmonics and thus give improved tone. The output signal is coupled to the high impedance internal loudspeaker via D.C. blocking capacitor C4 and a break contact on output jack JK1. The latter automatically switches out the speaker when a jack plug is inserted into JK1. The internal speaker should not be a low impedance type, and it is not advisable to take the output signal from JK1 to a low impedance load.

C1 provides decoupling of the supply rail and S1 is the on/off switch. The circuit has a current consumption of about 8mA. when JK1 is in use and the output is fed to a fairly high impedance load, but the current consumption rises to around 40mA. when the internal loudspeaker or an external load of a similar impedance is used.

Drum Synthesiser

Although extremely simple, the circuit shown in Fig.63 produces a quite realistic simulation of a drum sound. MIC1 is a crystal microphone insert which is fitted on the case of the unit, and will produce a wide range of output frequencies if the case is struck. Only low output frequencies are required here, and the higher frequencies are therefore filtered out

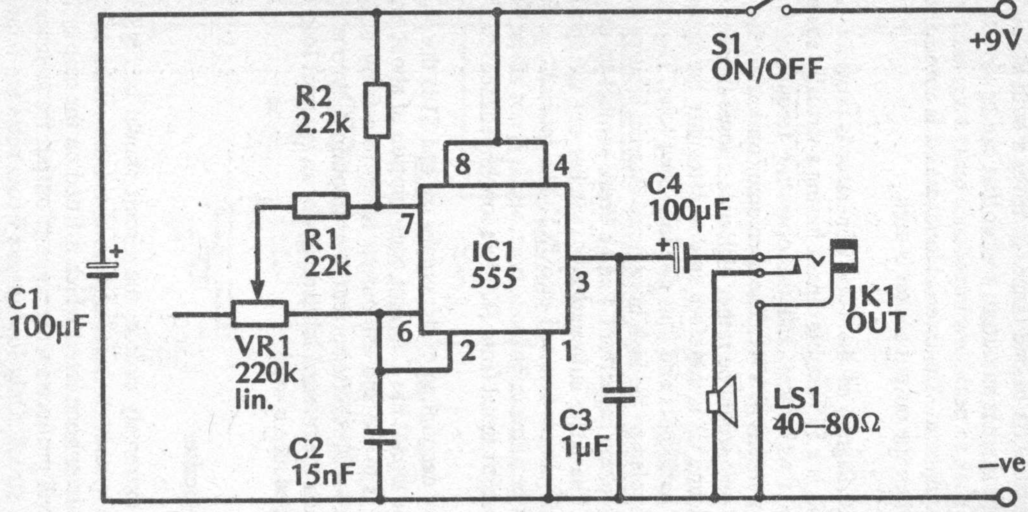

Fig. 62 The circuit diagram of the tone generator

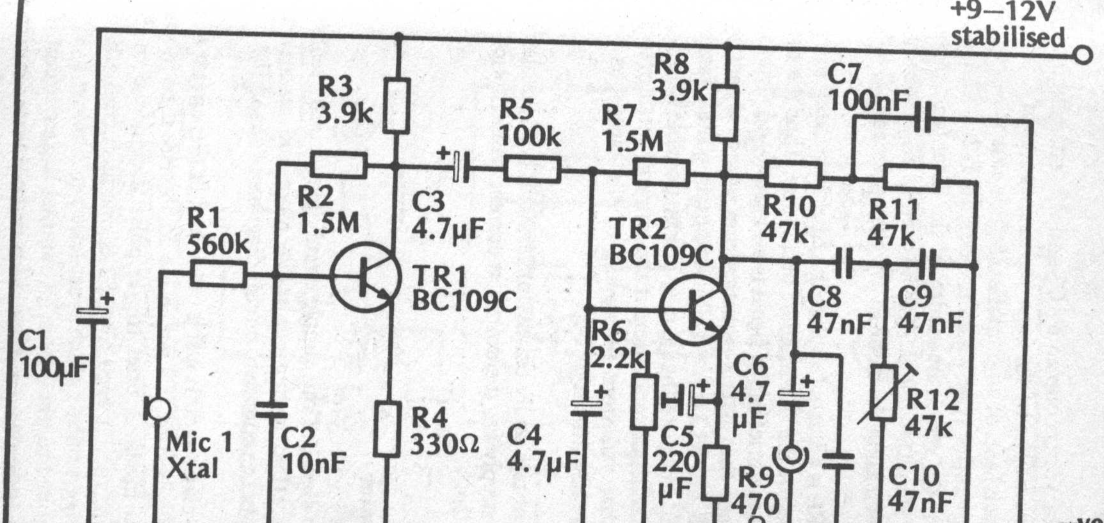

Fig. 63. The circuit diagram of the drum synthesiser

using R1 and C2. Tr1 is a common emitter amplifier which is used to boost the output of MIC1.

The output of Tr1 is coupled by C3 and R5 to the input of a simple bandpass filter of the type which uses a twin T filter in the feedback loop of an amplifier. In this case the amplifier is a common emitter type which is based on Tr2. R12 can be used to adjust the centre frequency of the filter (and the pitch of the drum sound), while R6 can be used to adjust the Q of the filter. Decreasing the value of R6 gives increased Q value.

In practice R6 is adjusted so that the filter has a very high Q and is just below the threshold of oscillation. Thus the filter "rings", and the brief input signal is elongated and given an envelope shape that gives a realistic drum sound. In fact by adjusting R6 and R12 it is possible to obtain quite a range of drum sounds, but note that it may not be possible to bring the filter to the verge of oscillation at the extreme settings of R12. Note also that variations in loading of the output may necessitate readjustment of R6.

In order to obtain really consistent results it is advisable to use a stabilised supply for the unit. The current consumption is only about 2mA.

Clap Generator

The output signal of this simple circuit gives a quite realistic clap sound effect having adjustable pitch and resonance. Fig.64 shows the complete circuit diagram of the unit.

Basically a clap sound is just a pulse, and in this circuit the pulse is generated by IC1 which is a 555 device used in the monostable mode. A brief trigger pulse is delivered to the trigger input of IC1 each time S2 is operated, and a brief negative trigger pulse (which must take pin 2 below 1/3V+) can be supplied to pin 2 from an external trigger circuit if this method of triggering is preferred for some reason. The length

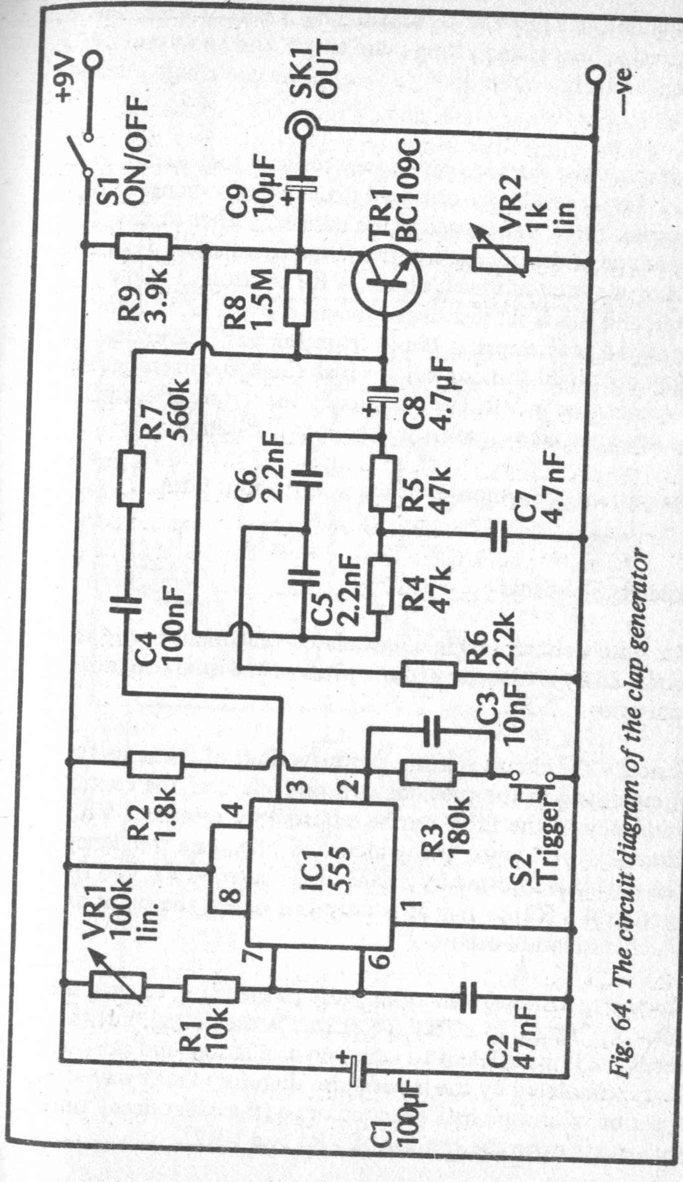

Fig. 64. The circuit diagram of the clap generator

of the output pulse can be adjusted by means of VR1, with a high value here giving a long pulse length and an output of low pitch, and a low value giving a short pulse and a high pitched output.

A simple pulse does not give a very realistic clap sound, and much better results are obtained if the pulse is shaped using a bandpass filter. Tr1 is used as the basis of a filter that is of the same type as the one used in the Drum Synthesiser described in a previous circuit in this book. VR2 controls Q of the filter, and this is set just high enough to give a fairly resonant (cupped hand) clapping sound from the unit. Care must be taken not to set this control too high (high Q corresponding to low resistance in VR2) or completely the wrong effect will be obtained, and the circuit could go into oscillation.

The current consumption of the unit is about 9mA.

Triangle Simulator

The simple circuit of Fig.65 simulates electronically the sound produced by a triangle, and the pitch of the note produced is adjustable.

Basically the circuit is just a bandpass filter of the same type as employed in the previous two projects, and the centre frequency of the filter can be adjusted by means of VR3 (decreased resistance giving increased operating frequency). The circuit is adjusted by means of Q control VR1, and fine Q control VR2, so that it is only just below the point at which oscillation occurs.

Operating S1 applies an input pulse to the filter, causing the filter to "ring". In effect the circuit is oscillating, but the feedback is insufficient to maintain oscillation, and the oscillations stimulated by the input pulse therefore fade away after a period of around half a second or so (the exact decay time depending upon the settings of VR1 and VR2).

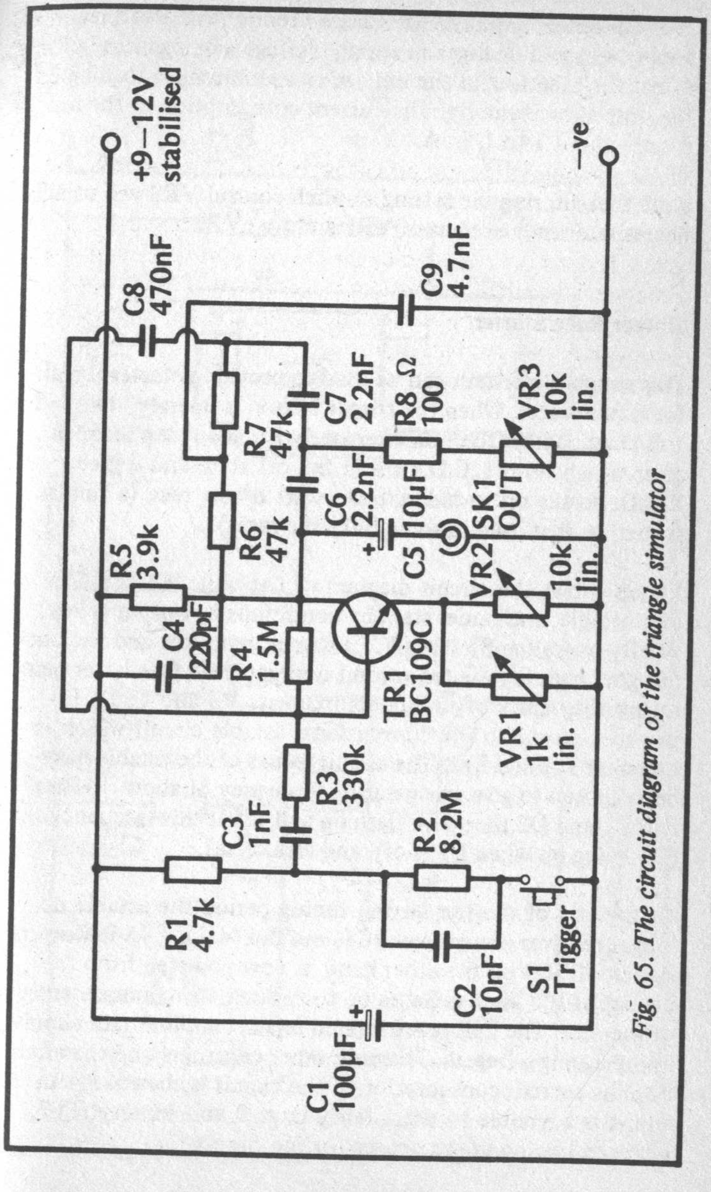

Fig. 65. The circuit diagram of the triangle simulator

It is advisable to power the circuit from a well stabilised supply as small changes in supply voltage can significantly affect the Q setting of the unit, as can variations in loading of the output incidentally. The current consumption of the unit is only about 1 to 1.5mA.

Note that altering the setting of pitch control VR3 will usually necessitate readjustment of VR1 and (or) VR2.

Slotcar Race Starter

This simple timer unit can be used to provide a starter signal for slotcar races. When the trigger button is operated two red L.E.D. indicators flash on alternately for about ten seconds, after which both L.E.D.s are in the off state and a green L.E.D. comes on to indicate the start of the race (a similar system to that used in some full scale races).

Fig.66 shows the circuit diagram of the unit. IC1 is a 555 monostable, and under standby conditions its output is low. Briefly operating S1 takes IC1's trigger input low and the output goes high for the ten second timing period (the latter being set by the values of timing components R6 and C3). This provides power to the conventional astable circuit which is based on Tr1 and Tr2. The circuit values of the astable have been chosen to give an operating frequency of about 1 Hertz, and D1 and D2 therefore flash on and off at this frequency (D1 being on when D2 is off, and vice versa).

At the end of the ten second timing period the astable no longer receives power from IC1, and the two L.E.D. indicators switch off. D3 on the other hand, is now powered from the output of IC1, and switches on to indicate the commencement of the race. The unit is retriggered to start another race simply by operating S1 again. The only other control is on/off switch S2. The current consumption of the circuit is about 15mA., and it is advisable to use a fairly large 9 volt battery (PP7, PP9, etc.) as the power source for the circuit.

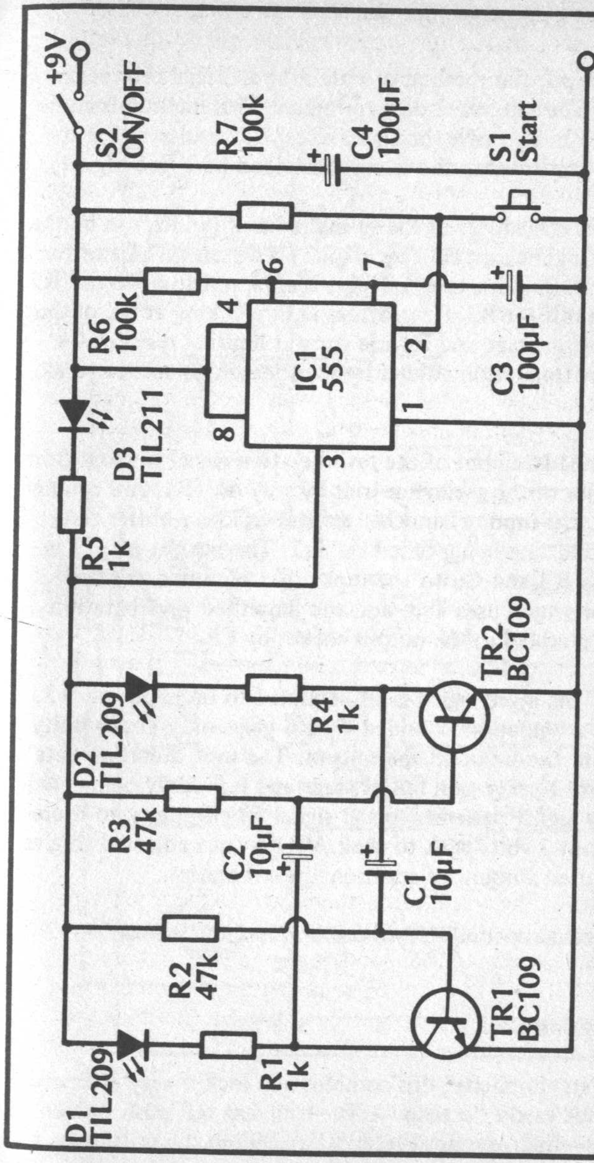

Fig. 66. The circuit diagram of the slotcar starter

Springline Reverberation Unit

The most popular method of obtaining artificial reverberation is to use a circuit based on a springline. This method is comparatively inexpensive, but gives excellent results, and is the method employed in the circuit described here (see Fig.67).

The input transducer of the springline unit (which can be the short springline unit sold by Maplin) is driven via an emitter follower stage. This uses Tr1 plus biasing resistors R1 and R2, and load resistor R3. C3 provides D.C. blocking at the output of the buffer stage and R4 is a current limiting resistor. C4 provides attenuation at high frequencies and is needed to aid stability.

A controlled amount of the reverberation signal is taken from the output of the springline unit by way of VR1, and coupled by C5 to the input of another emitter follower buffer stage; this second one being based on Tr2. The output of Tr2 is taken via R8 and C6 to the input of a common emitter amplifier which uses Tr3, and the amplified reverberation signal is coupled to the output socket by C8.

Some of the input signal is also coupled to the input of Tr3, but the attenuation provided by R6 gives only about unity gain from the input to the output. The unit therefore acts rather like a unity gain buffer stage and it is easily connected into any signal path where the signal amplitude is no more than about 2 volts peak to peak. VR1 is then adjusted to give the required amount of additional reverberation.

The current consumption of the unit is about 12mA.

Combination Lock

Despite its simplicity, this combination lock is very effective and is not easily "cracked". The unit has ten push button switches which are numbered "0" to "9" in the usual way, but

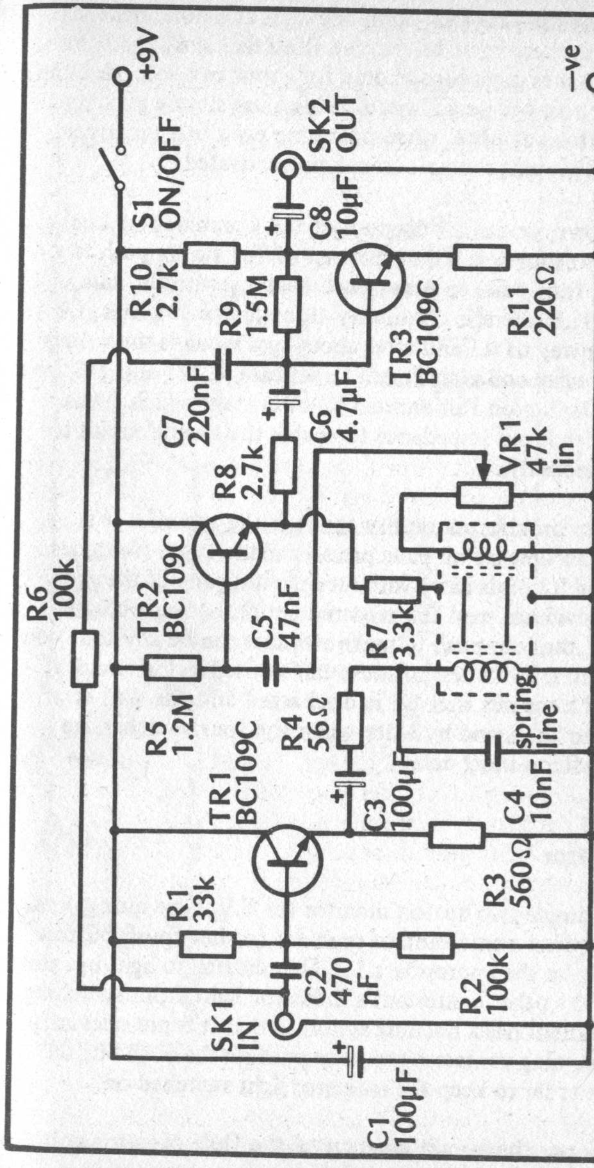

Fig. 67. The circuit diagram of the springline reverberation unit

the combination is not keyed in one digit at a time. Instead, the four numbers must be entered simultaneously, and the correct switches must be operated for about two seconds or so before the unit will be activated. It is no use simply pushing all the buttons at once, since operating even one incorrect switch will prevent the unit from being activated.

Fig.68 shows the circuit diagram of the Combination Lock. The positive supply is fed to the base of Tr1 via six push to break and four push to make push button switches, and resistor R1. If there is continuity through the switches, C1 charges by way of R1 and after about two seconds the voltage across the relay coil is sufficient to activate it. Tr1 and Tr2 are simply a Darlington Pair emitter follower stage which give a suitably high input impedance to enable the timing circuit to function properly.

In order to provide continuity through the switches it is necessary to operate all four press to make types (switches 1, 3, 6, and 9 in this case) without operating any of the press to break switches, and the required action is thus obtained. Of course, the four push to make switches can be any four you choose, and more or less of these can be used in the circuit if desired. R2 ensures that C1 is discharged and the unit is returned to the stand-by state when the four switches are released, after a short delay.

Quiz Monitor

This is a simple two button monitor for T.V. type quiz games, where the first contestant to push his (or her) push button switch causes the appropriate L.E.D. indicator to light up, and prevents the other contestant's indicator light from switching on. The circuit has a latching action so that it is not necessary for the winning contestant to keep pushing the push button switch in order to keep the indicator light switched on.

Fig.69 shows the circuit diagram of the Quiz Monitor unit.

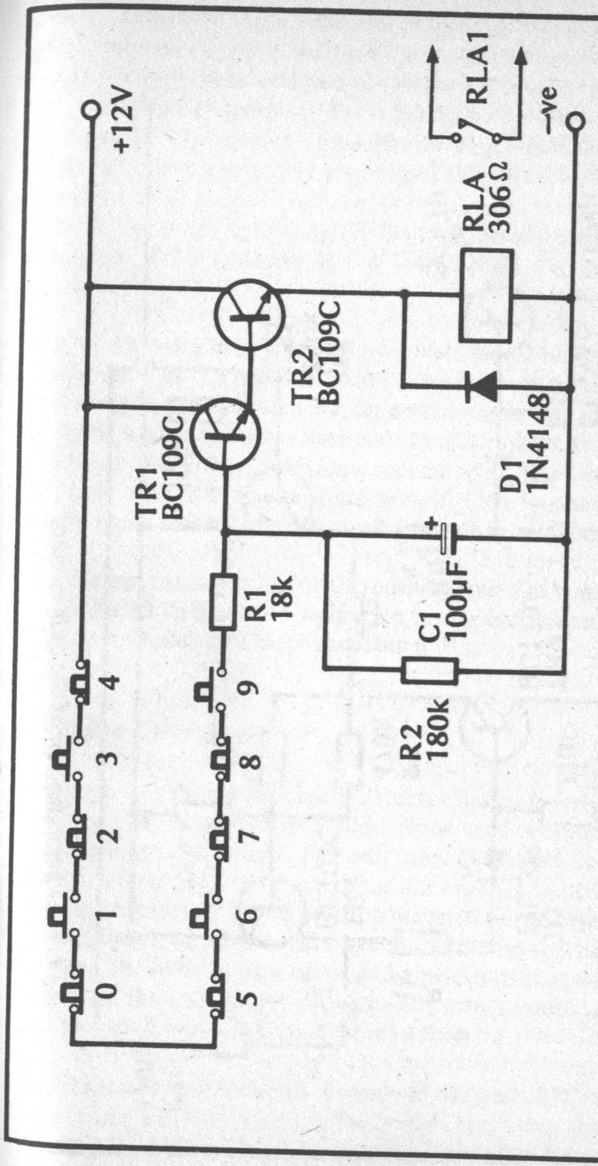

Fig. 68. The circuit diagram of the combination lock

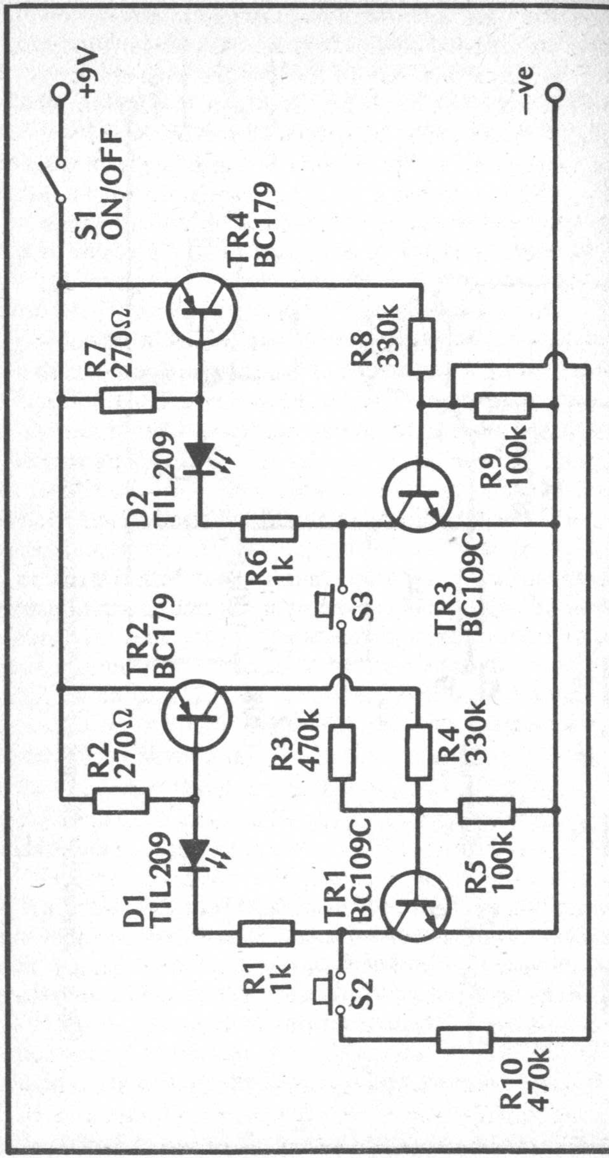

Fig. 69. The circuit diagram of the quiz monitor

This is basically just a conventional bistable multivibrator with the cross coupling provided via the push button switches, and a L.E.D. indicator as part of the load for each transistor. Thus, when S3 is operated, Tr1 is biased into conduction and D1 lights up. Operating S2 then has no effect as the voltage at Tr1's collector is too low to bias Tr3 into conduction. Similarly, if S2 is operated first, Tr3 is biased into conduction and D2 lights up. Operating S3 then has no effect as the voltage at Tr3's collector is too low to bias Tr1 into conduction.

The latching action is provided by an additional transistor in each section of the bistable. If Tr1 is biased into conduction for example, the potential divider action across R1, D1, and R2 gives a high enough voltage across R2 to bias Tr2 into conduction. Tr2 then provides a bias current to Tr1 by way of R4, holding Tr1 in the on state even if S3 is released. The circuit can be reset by simply switching off briefly using S1.

The current consumption of the unit is about 7mA. when one of the L.E.D. indicators is operating, and is insignificant (less than 1μA) under quiescent conditions.

Proximity Detector

Although this simple proximity detector design is put forward here as an amusing game, it could well be used in a number of more serious applications. The unit must be housed in a plastic case (or one made from some other non-metallic substance), and a sensor plate is fitted on the underside of the case's lid. A coin, sweet, or other small object is placed on top of the case just above the sensor plate, and the idea of the game is to remove the coin, sweet, or whatever, without activating the unit and causing a L.E.D. indicator to flash on.

Fig.70 shows the full circuit diagram of the unit. IC1 is used as an inverting amplifier having a fairly high input impedance of about 100k, and a voltage gain which can be varied from zero

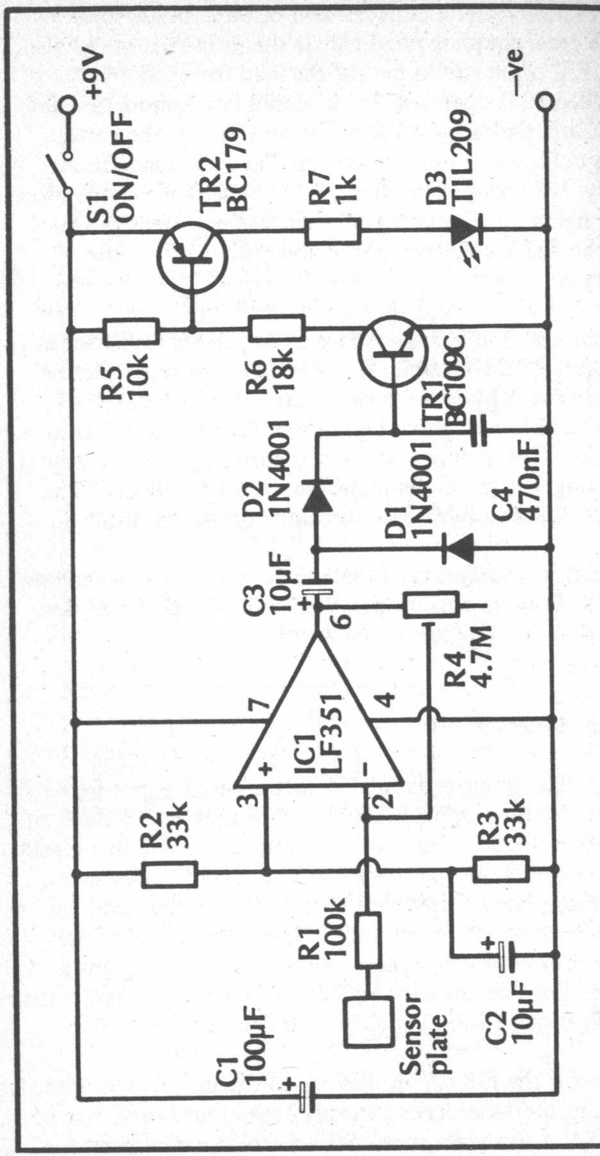

Fig. 70. The circuit diagram of the proximity detector

to 47 times by means of R4 (maximum value here producing maximum voltage gain). The input of the amplifier is fed from the sensor plate, and the latter is simply a small piece of aluminium or any other convenient metal, say a piece having dimensions of about 60mm. square.

Hum and other electrical noise are picked up by the plate and amplified by IC1. D1, D2, and C4 rectify and smooth the output of IC1 to give a positive bias that is amplified by Tr1 and Tr2, and used to drive L.E.D. indicator D3. In practice R4 is adjusted for a level of gain that is not quite sufficient to cause D3 to switch on. However, if a hand is placed near the sensor plate, stray capacitance results in signals picked up in the contestant's body being coupled to the sensor plate, giving an increased input signal level, and causing D3 to light up.

Slide/Tape Synchroniser

The purpose of a slide/tape synchroniser is to put short bursts of audio tone onto one channel of a stereo tape at the point where slide changes are required; the other channel being fed with music and (or) a commentary. When the tape is played back, the synchroniser is fed with the signal from the channel of the tape which contains the tone bursts, and when it receives each burst of signal it closes a pair of relay contacts which operate the slide change mechanism of the projector. The slide changes are thus provided automatically at the appropriate places.

Fig.71 shows the circuit diagram of the Slide/Tape Synchroniser, and this consists of two main sections: the tone generator and the tone detector. The tone generator uses a 555 device (IC1) as a simple astable which has a virtually square-wave output. R3 and C3 filter the output signal to reduce the harmonic content, and VR1 is the output level control. The maximum output level is several volts peak to peak, and should be sufficient to drive any tape deck or recorder. C4 and R4 are used to feed some of the tone to the tone detector

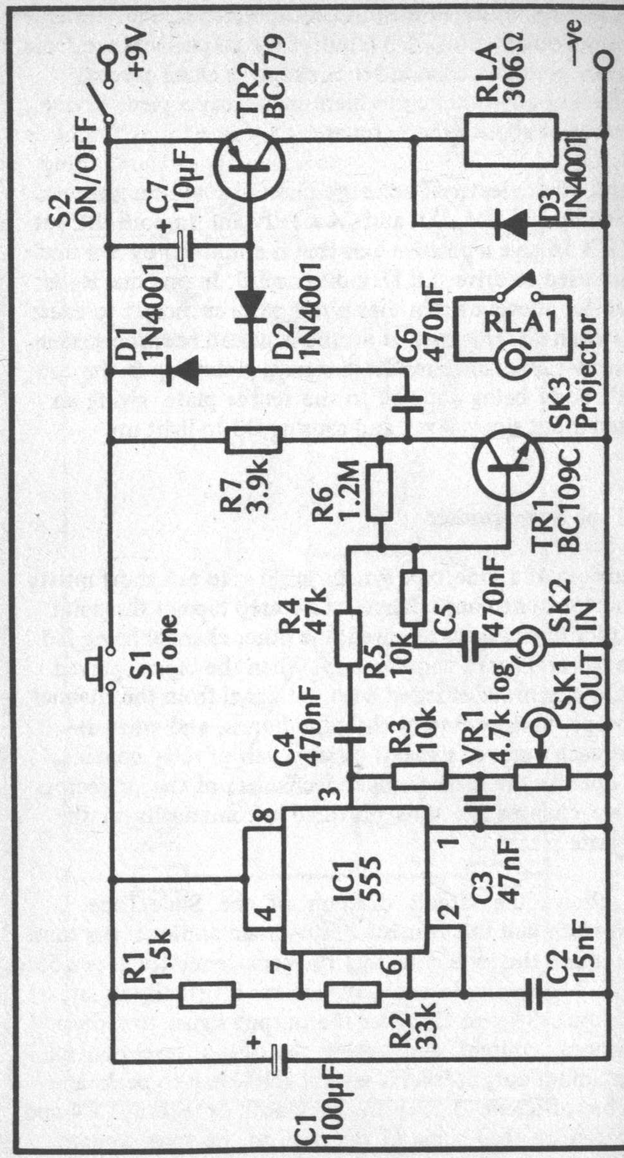

Fig. 71. The circuit diagram of the slide/tape synchroniser

circuit so that the projector can be operated automatically as each tone burst is recorded (this is very helpful when making a recording). S1 is operated to produce each tone burst.

The tone detector uses Tr1 as a high gain common emitter amplifier, and its output is fed to a rectifier and smoothing circuit which consists of D1, D2, and C7. The negative output of this circuit is fed to the base of Tr2, and is strong enough to switch on Tr2, activate the relay, and operate the projector's slide change mechanism, when the input tone is present. An input level of only about 100mV. R.M.S. or more is needed to operate the unit, and any normal tape deck or recorder should be capable of delivering this.

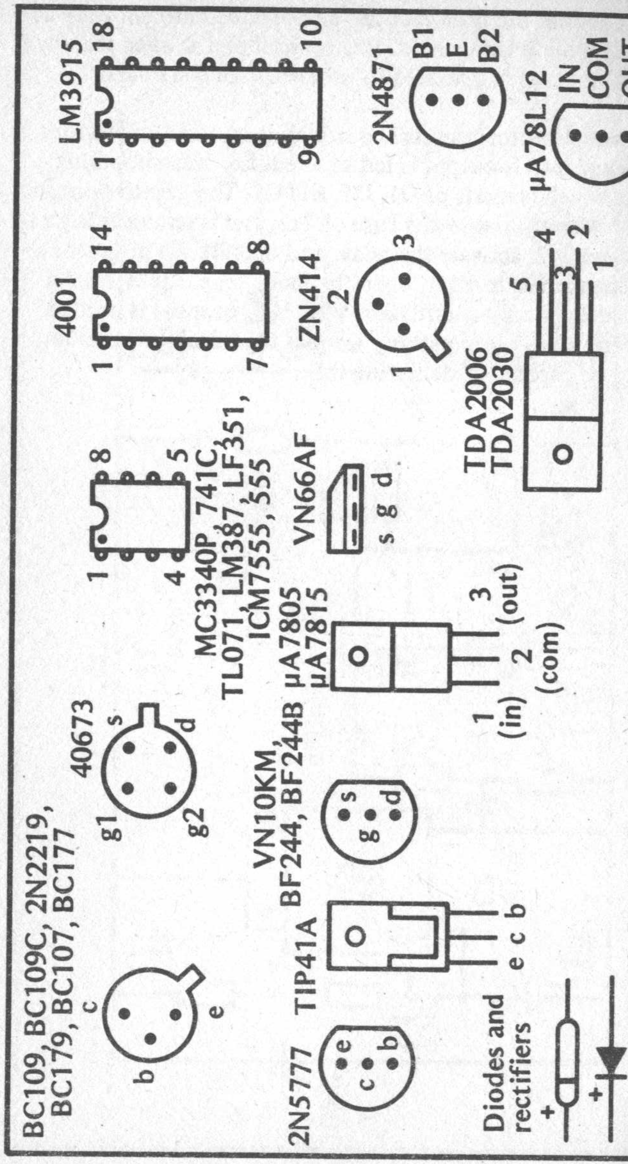

Fig. 72. Semiconductor leadout and pinout details

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