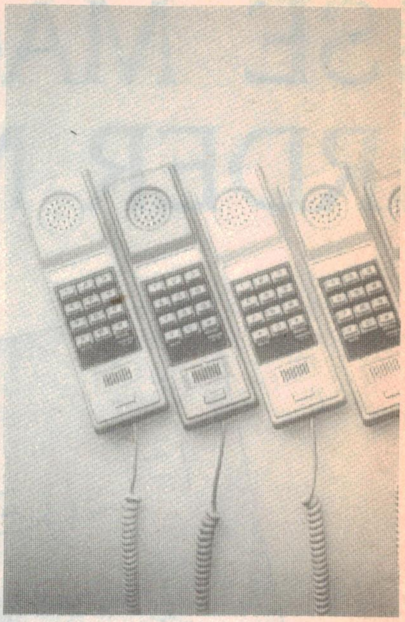
How they work, how to fix them:

Electronic phones



So the beaut little second telephone that you bought from the local supermarket at a bargain price just a few months ago has given up the ghost. At the cost of repair charges nowadays you'll probably have to throw the thing out. Or will you? With a bit of an idea about how these things tick, and some spare time, you could probably fix it yourself. Read on, your troubles may be over.

by CHRIS KING

To begin with, let's look at the basic functional sections a telephone needs, in order to do what it does. These are the speech circuitry, signalling circuitry and the call indicator or ringer.

The speech circuitry translates our voice into electrical signals which are applied to the phone line, and produces sounds in the earpiece derived from signals received from the 'other end' via the same line. In its simplest form the speech circuit requires a microphone and a receiver or earpiece.

Until a few years ago, common telephones employed a carbon microphone. This consists of a chamber of carbon granules, which varies its resistance in response to sound waves. The varying resistance modulated a direct current passing through the granules.

Carbon microphones are relatively large and do not provide consistent performance. The carbon granules also tend to settle and become 'packed' after a while, causing a loss in sensitivity. This is why tapping the telephone handset can sometimes improve its signal quality. The frequency response of these microphones is peaky and is also subject to variations.

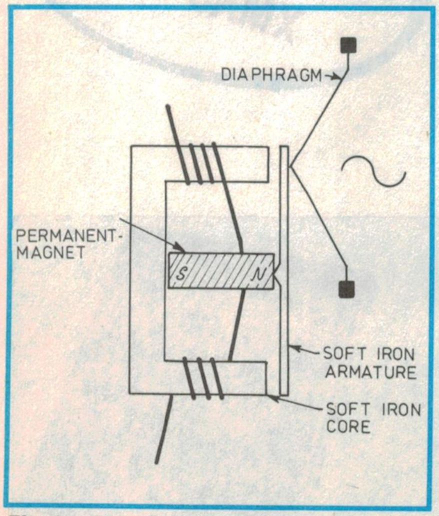


***Fig.1:** The basic principle of the rocking-armature receiver, as used in non-electronic phones.*

Given their imperfections, it is not difficult to understand why they are now rarely used. The main advantage of the carbon microphone is its ability to develop a signal that is large enough for application to the phone line, without the need for further amplification; that is, it's quite *sensitive*.

Early telephones employed a simple receiver made up of a thin, compliant, iron diaphragm mounted in the field of a permanent magnet which biased it. Speech signal currents applied through a coil wound on the magnet modulated the magnet's field, causing the diaphragm to vibrate. Because a permanent magnet has a high reluctance, i.e., it tends to impede changes in its magnetism, this type of receiver was barely sensitive enough for the job.

A similar but improved receiver is the *rocking armature* type, shown in Fig.1. Here speech currents modulate the strength of two magnetic paths through the soft-iron armature and core, in a complimentary manner, causing it to rock. This action is coupled to the diaphragm. As the armature is able to apply bi-directional force on the diaphragm, no biasing field is necessary. Although the rocking armature receiver is a bit more complex, it is more sensitive and can be designed to have better acoustic properties.

The telephone line, which simultaneously conducts the signals from both parties in a telephone conversation, is just a single pair of wires. If the speech circuitry in each telephone were nothing more than a microphone and a receiver, each person would hear their own voice in their receiver louder than that of the other, as it would not have suffered the losses of the telephone line. The existence of our voice in our own earpiece is called *sidetone*.

Fortunately, it is possible to significantly reduce sidetone with a clever balancing circuit called the anti-sidetone circuit, or sometimes the *hybrid*.

Reducing sidetone

Fig.2 shows a simplified diagram of a telephone's anti-sidetone speech circuit when transmitting speech signals. Speech currents from the microphone split into two paths, I_B and I_L, flowing in opposite directions through the transformer primaries. By choosing the correct relationship between the ratio of the telephone line impedance (Z_L) and the balance impedance (Z_B) and the

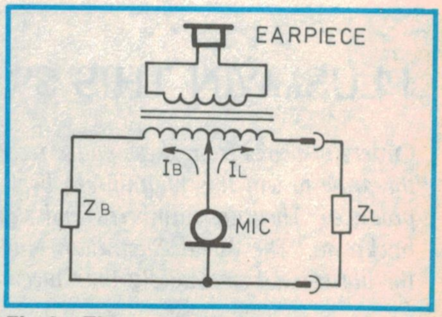


***Fig.2:** The anti-sidetone circuit of a conventional phone, in simplified form.*

ratio of the turns in the two primaries, the two opposing speech currents will produce no resultant signal in the secondary winding.

In a practical telephone the anti-sidetone transformer is wound as an autotransformer and the turns ratio and balancing impedance are chosen such that an adequate proportion of the speech current is passed to the phone line.

Because the impedance of the phone line is somewhat unpredictable, total sidetone cancellation is not practical. Fortunately it also unnecessary, because the phone sounds more natural when we hear our voice at about the same level as we do in normal conversation without the handset.

Dialling circuitry

To get the telephone exchange equipment to connect us to another phone, it has to know that we want to initiate a call and also the address – i.e., the phone number – of the telephone we're calling. This information is generated by the phone's *signalling* circuitry.

When the telephone handset is 'hung-up' or 'on-hook', the telephone draws little if any DC current and the full open-circuit exchange voltage of about 50 volts is present across the phone line terminals. Lifting the handset allows a switch, known as a gravity switch, to connect the phone's speech circuits to the line. Equipment at the the exchange senses the current drawn by the phone, and allocates the necessary circuits to decode address signals from the phone and then make the connection through to the phone being called.

The majority of telephones in use in Australia employ what is known vari-

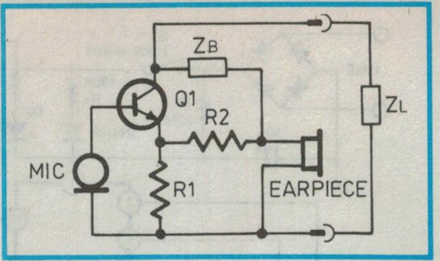


Fig.3 (above): The basic sidetone nulling circuit used in many electronic phones.

ously as pulse, loop-disconnect or *decadic* signalling to transfer address information to the exchange. With this signalling method, the DC loop established when the phone is taken 'off-hook' is sequentially opened for about 67ms (milliseconds) and then closed again for about 33ms before the next 67ms open period.

The number of open or 'break' periods represents the digit being dialled. For example, if the digit '5' is dialled, the dialling circuitry in the phone open-circuits the phone line for five 67ms periods, each one separated by a 33ms closed period.

To separate one digit from the next, the pulse train for each digit is followed by a closed circuit or 'make' period of at least 800ms; this is referred to as the *inter-digital pause* or IDP.

Decadic signalling is easily achieved by the electro-mechanical rotary dial used in most older telephones.

To activate the incoming call indicator or 'ringer' in our phone, the exchange applies an AC signal of about 75 volts to the line. This signal is on for about 200ms then off for about 400ms, then on for 200ms and off for about 2s, before repeating again. This pattern, or *cadence*, results in the familiar 'bring-bring' ringing sound.

When the exchange equipment senses that the phone has been answered, ring current is removed from the line and the connection to the calling phone is made.

The ringer in the phone must monitor the line when the phone is 'on-hook'

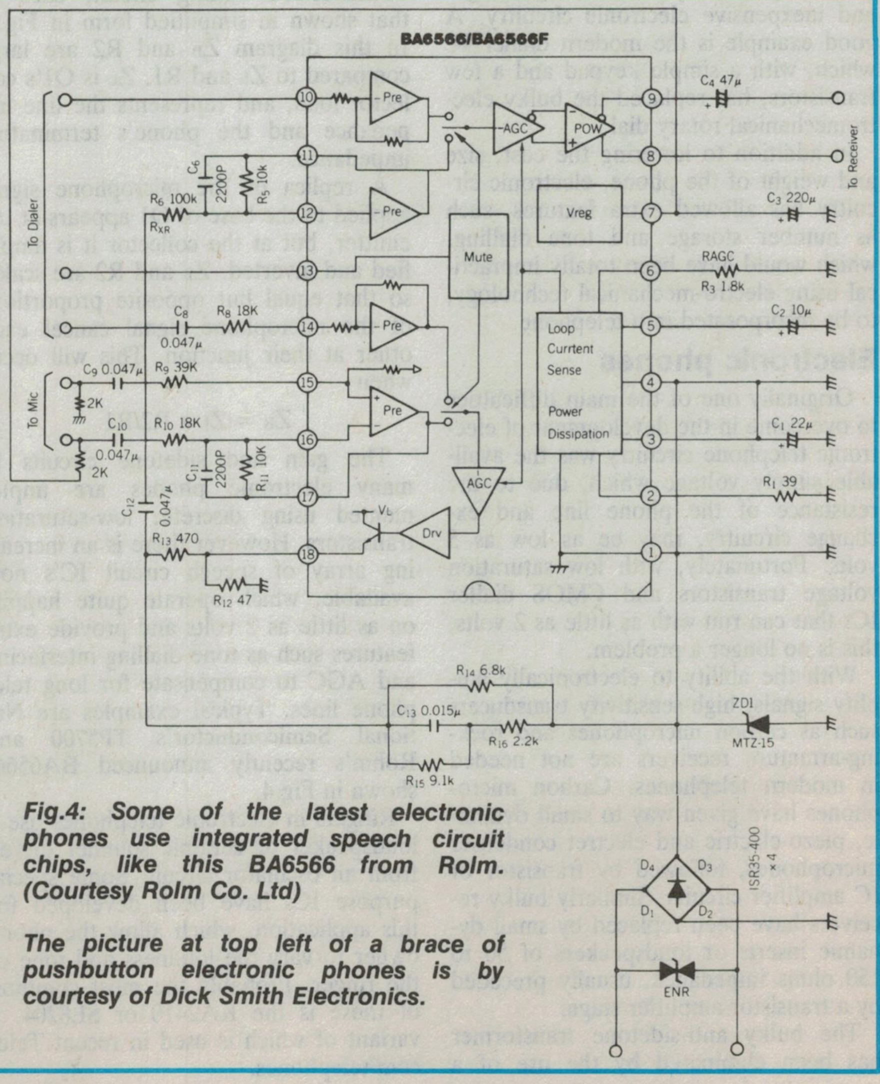


Fig.4: Some of the latest electronic phones use integrated speech circuit chips, like this BA6566 from Rolm. (Courtesy Rolm Co. Ltd)

The picture at top left of a brace of pushbutton electronic phones is by courtesy of Dick Smith Electronics.

and must draw no DC current. In phones for use in Australia, the ringer is always coupled to the line via a high-voltage capacitor of 1-1.5uF. Besides blocking DC, the capacitor provides a standardised 'on-hook' characteristic for line testing purposes.

In older telephones, the ringer itself is an electro-mechanical bell operating on the same principle as the rocking armature receiver already described.

We have now looked at the main functional blocks of a basic telephone. All of the functions can be implemented using rugged, tried and proven passive devices. Telephones constructed with not much more than the above, while having only basic features, have provided us with a reliable and easy to use means of communication for over half a century.

However developments in semiconductor technology have allowed many of the components in the basic telephone to be replaced by smaller, lightweight and inexpensive electronic circuitry. A good example is the modern dialler IC which, with a simple keypad and a few transistors, has replaced the bulky electromechanical rotary dial.

In addition to lowering the cost, size and weight of the phone, electronic circuitry has allowed extra features, such as number storage and tone dialling, which would have been totally impractical using electro-mechanical technology, to be incorporated into telephone.

Electronic phones

Originally one of the main difficulties to overcome in the development of electronic telephone circuitry was the available supply voltage which, due to the resistance of the phone line and exchange circuitry, may be as low as 5 volts. Fortunately, with low saturation voltage transistors and CMOS dialler ICs that can run with as little as 2 volts, this is no longer a problem.

With the ability to electronically amplify signals, high-sensitivity transducers such as carbon microphones and rocking-armature receivers are not needed in modern telephones. Carbon microphones have given way to small dynamic, piezo-electric and electret condenser microphones, followed by transistor or IC amplifier circuits. Similarly bulky receivers have been replaced by small dynamic inserts or loudspeakers of 50 to 150 ohms impedance, usually preceded by a transistor amplifier stage.

The bulky anti-sidetone transformer has been eliminated by the use of a transistorised nulling circuit, such as that shown in simplified form in Fig.3. In this diagram Z_B and R2 are large compared to Z_L and R1. Z_L is Q1's collector load, and represents the line impedance and the phone's terminating impedance.

A replica of the microphone signal applied to the base of Q1 appears at the emitter, but at the collector it is amplified and inverted. Z_B and R2 are scaled so that equal but opposite proportions of the microphone signal cancel each other at their junction. This will occur when

$$Z_B = Z_L \times R2/R1$$

The gain and sidetone circuits in many electronic phones are implemented using discrete, low-saturation transistors. However there is an increasing array of speech circuit IC's now available, which operate quite happily on as little as 2 volts and provide extra features such as tone dialling interfacing and AGC to compensate for long telephone lines. Typical examples are National Semiconductor's TP5700 and Rohm's recently announced BA6566, shown in Fig.4.

Ringers in electronic telephones use a loudspeaker or ceramic sounder driven from an oscillator circuit. Some special purpose ICs have been developed for this application, which allow the phone owner to vary the loudness and tone of the ringer. Probably the most common of these is the KA2410 or SL8204, a variant of which is used in recent Telecom telephones.

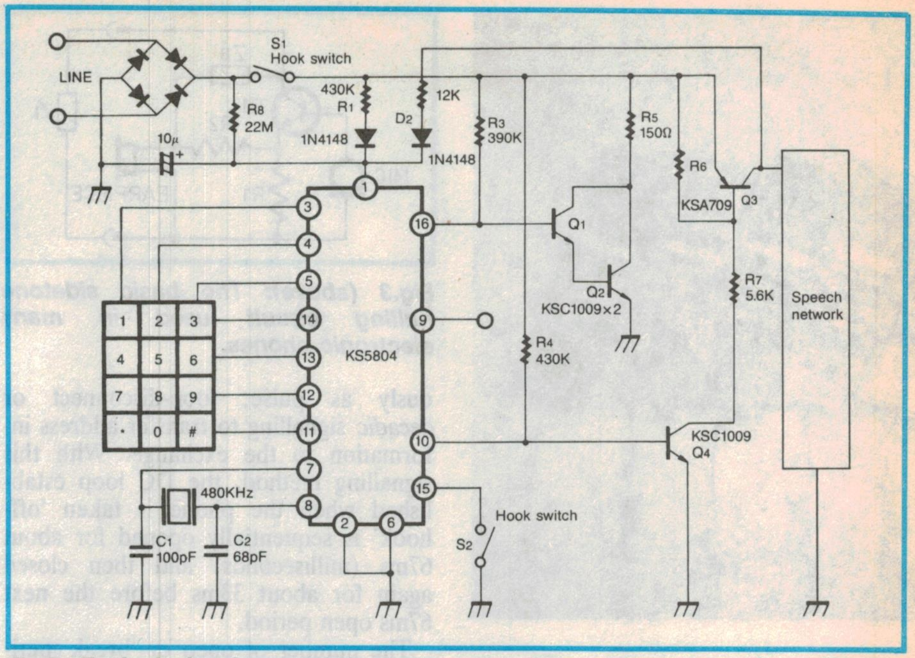


Fig.5: Typical circuit application of a pulse dialler IC, the Samsung KS5804. Pin 16 switches Q1 and Q2 on and off, producing the dialling pulses via R5. (Courtesy Samsung Semiconductor)

In this chip a slow oscillator operating at about 10Hz sweeps a tone oscillator around 600Hz, producing a pleasant ringing sound. This IC is produced, with varying part number prefixes, by a number of manufacturers including Plessey, SGS, Samsung and Mitel. Like most ringer circuits, the chip is normally powered by rectifying the AC ring signal from the exchange.

Dialler chips

The first section of the telephone to be implemented in IC form was the loop-disconnect (decadic) dialler. An enormous range of these ICs has become available, all providing similar basic functions. Some provide extra features such as ten (or more) number storage and the ability to select different dial rates and break:make ratios to suit varying national standards. Table 1 lists some of the more common types.

Fig.5 shows an application of a typical pulse dialler IC, the Samsung KS5804. Numbers are entered via a 7-key keypad, which is scanned as a 3 by 4 matrix by the IC. As a timebase, the chip uses a 480kHz ceramic resonator. In general, pulse diallers use either this timebase or a low frequency (2-20kHz) RC oscillator.

To conserve power, the oscillator only runs whens necessary – i.e., immediately following the detection of a key press, and during transmission of dialling pulses.

Pin 10 is an output used to mute the speech circuitry during dialling, so as to prevent loud dialling clicks from being heard in the earpiece. Pin 16 provides the actual dial pulses and is used, in this case, to connect and disconnect R5 – which is the load on the line during 'make' periods. In many phones, the speech circuitry itself is used as the dialling load.

Pin 9 of this IC selects the break:-make ratio, at either 60:40 or 67:33.

Many pulse dialler ICs have provision to select different dialling pulse rates and inter-digital-pause periods. For Australian conditions, the dialler should be configured as follows:

Dial rate	10 pulses per second
Break:make	67:33
IDP	800ms or more

Tone dialling

A more modern method of sending dialling information, which has only become practical with the advent of IC technology, is dual-tone multi-frequency (DTMF) encoding. Here instead of opening and closing the DC loop, standardised pairs of carefully selected audio tones are transmitted along the phone line in the same manner as speech signals. Decoding equipment at the exchange identifies these special tones codes and initiates the appropriate call routing action.

The DTMF tones are shown in Table 2. There are 16 different combinations, each made up of one of the low group of tones with one of the high group. Ten of these combinations are used to represent the digits 0 to 9, leaving the other six available for special uses.

Once a connection has been made, the same tones can then be used for remote control of equipment, such as telephone answering systems, or for entry of data into a remote banking system for example.

DTMF dialling is faster than pulse dialling. For example, to dial the 6-digit number '565656', a decadic dialler would take over 8 seconds, whereas the DTMF encoded number could be transferred in just 1 second. However, few DTMF dialler ICs are equipped with last-number redial or multi-number memory features. Also, because most subscriber lines in Australia are configured for pulse dialling, DTMF dialling is not common here in simpler telephones.

The nitty gritty

Armed with this understanding of the basic sections of a telephone, let's now take a look at how they all fit together in Fig.6, the actual circuit of a typical modern electronic phone.

The 2-wire phone line enters at the top left of the circuit and with the hookswitch, SW1, in the on-hook position, as shown, the line is connected to the ringer circuit in the bottom-left corner.

The ringer comprises a 2-transistor multivibrator configuration, driving a ceramic piezo element, PZ1. Capacitor C1 isolates the line's DC from the ringer so that it only operates from the high-voltage, AC ring signal.

In this particular circuit, the ringer oscillator only operates during positive half-cycles of the ring signal. Commonly, the ringer oscillator is powered via a full-wave rectifier.

Zener diode D14 sets a threshold voltage for the ringer to prevent it from 'tinkling' in response to dialling transients produced by parallel connected phones. The ringer switch shorts out the oscillator in the 'off' position, preserving the standardised 1uF on-hook line termination.

Resistor R1, which bypasses the hook switch in the on-hook position, provides the minute current needed to keep the dialler IC's last-number-redial memory alive.

When the phone is picked up, the hookswitch contacts changeover, connecting the dialling and speech circuits and in this case disconnecting the ringer

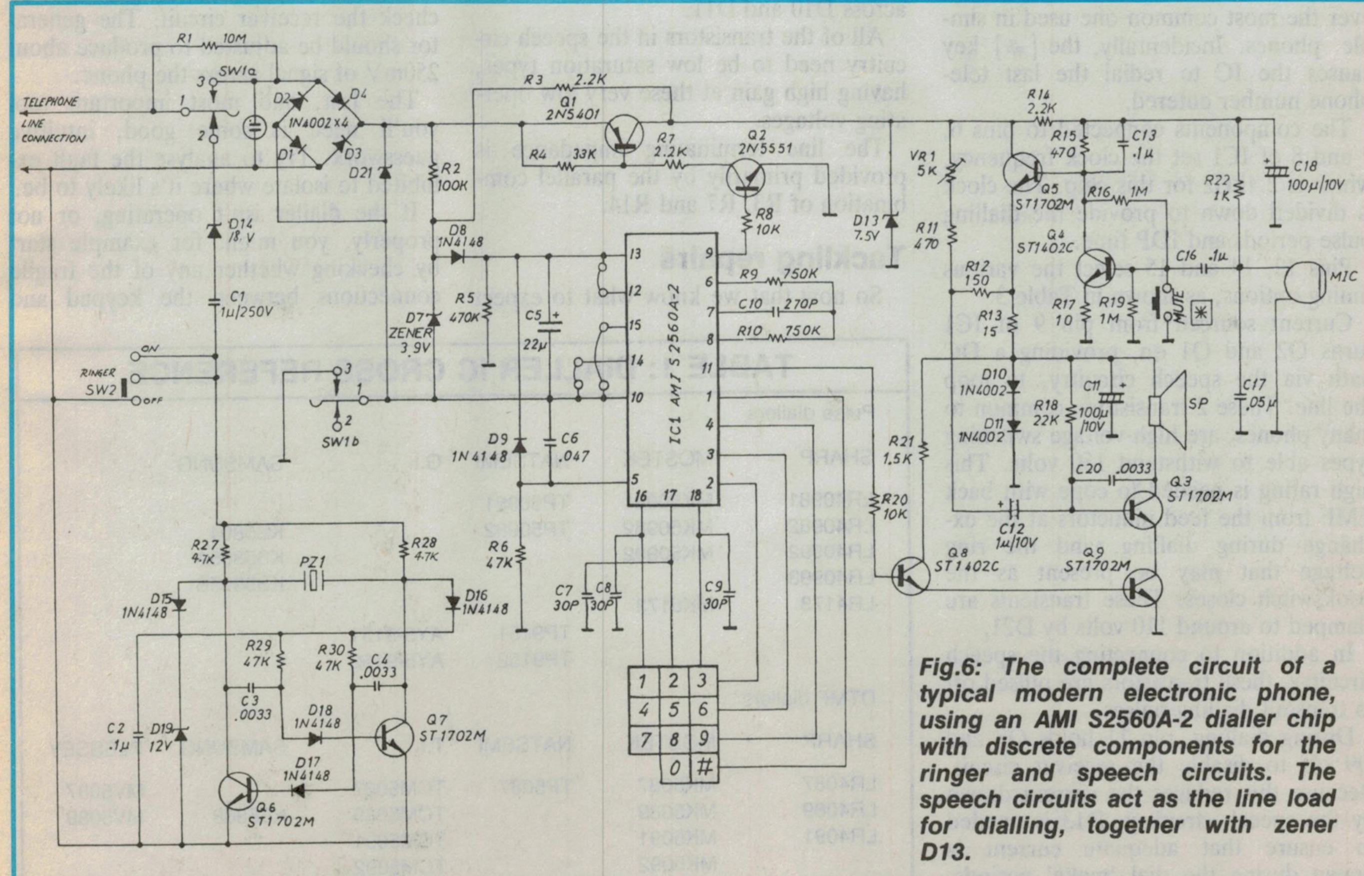


Fig.6: The complete circuit of a typical modern electronic phone, using an AMI S2560A-2 dialler chip with discrete components for the ringer and speech circuits. The speech circuits act as the line load for dialling, together with zener D13.

Electronic phones

circuit. In many phones the ringer stays across the circuit in the off-hook state.

As the line polarity is unknown, a full wave bridge, D1-D4, is needed to correctly route the line current to the polarity-sensitive speech and dialling circuitry.

In the off-hook state, the dialler IC is powered via R2 and, except during dialling break periods, R3. D7 and D8 limit the supply voltage to the IC to just over 3 volts.

When the contacts of SW1b close, the 'ground' side of the 'phone circuitry is connected to the negative output of the supply, pulling pin 5 of dialler IC1 low via R6. This indicates to the IC that the phone is on-line. The IC assumes an operational condition, sourcing current from pins 9 and 11 to activate the speech circuitry at the right of the diagram.

Pins 16, 17 and 18, and pins 1, 2, 3 and 4 are the keyboard column and row inputs respectively. The IC detects the connection of a row with a column and generates the appropriate dialling pulse train. These pins are referred to as 'inputs' on this particular IC because there are 2 other methods of interfacing to the chip where the pins are true inputs. The connection method shown, is however the most common one used in simple phones. Incidentally, the [#] key causes the IC to redial the last telephone number entered.

The components connected to pins 6, 7 and 8 of IC1 set the clock frequency, which is 2.4kHz for this chip. The clock is divided down to provide the dialling pulse periods and IDP time.

Pins 12, 14 and 15 select the various timing options, as shown in Table 3.

Current sourced from pin 9 of IC1 turns Q2 and Q1 on, providing a DC path via the speech circuitry, to loop the line. These 2 transistors, common to many phones, are high-voltage switching types able to withstand 150 volts. This high rating is needed to cope with back EMF from the feed inductors at the exchange during dialling, and the ring voltage that may be present as the hookswitch closes. These transients are clamped to around 110 volts by D21.

In addition to connecting the speech circuitry, these transistors are pulsed off to transmit dialling pulses.

During dialling, pin 11 holds Q8 and Q9 off to disable the receiver circuit. Because this reduces the current drawn by the speech circuitry, D13 is needed to ensure that adequate current is drawn during the dial 'make' periods.

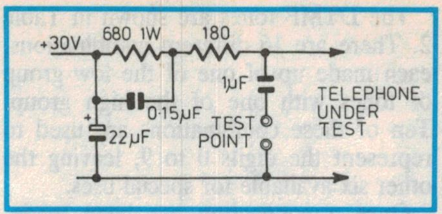


Fig.7: A simple test circuit which can be used for troubleshooting.

With the receiver circuitry enabled, D13 is effectively out of the circuit as the voltage across it will have dropped below its knee point.

In common with most electronic phones, a common 2-terminal electret microphone is used. These devices appear electrically as an open-drain FET. R22 is the microphone's 'drain' load.

The microphone signal is amplified by Q4 and applied to the line via the collector of Q5. This transistor is configured in the anti-sidetone circuit discussed before. The sidetone balance is adjusted by VR1. Note that the lower end of R13, being heavily bypassed by C11, is an AC ground.

Received signals, and attenuated transmit signals, are coupled from the junction of R11 and R12 to Q3, the receiver amplifier. Q3 drives the earpiece, a small 150 ohm speaker, in a class A configuration. The receive circuit is powered from just 1.5 volts developed across D10 and D11.

All of the transistors in the speech circuitry need to be low saturation types, having high gain at these very low operating voltages.

The line terminating impedance is provided primarily by the parallel combination of R3, R7 and R14.

Tackling repairs

So now that we know what to expect inside the little beast, what else might be handy before we attack it?

To make things easier, try to get as much information about your particular phone as possible. The telephone supplier should let you have a copy of the circuit, if they have one. Data about the dialler IC could be invaluable. With a bit of luck, your dialler IC, or an equivalent, might be detailed in one of the more readily available data manuals, such as National Semiconductor's linear series. If all else fails, a bit of intuitive circuit tracing might see you through.

You won't want to work on the phone while it is connected to the telephone line. Besides the possibility of being bitten by line transients, you can expect a hefty financial 'bite' from Telecom if you cause any interference or damage to its exchange equipment.

The test circuit shown in Fig.7 can be cobbled together in a plastic case and powered from an existing DC supply. Note that neither side of the supply should be earthed, as this would prevent you from moving signal generator and CRO earth leads around the phone circuits.

The test point can be connected to a CRO to monitor dialling pulses and transmitted speech signals. Alternatively it can be fed from an AF generator to check the receiver circuit. The generator should be adjusted to produce about 250mV of signal across the phone.

The last, and most important tool you'll need is some good, intuitive guesswork. Try to analyse the fault exhibited to isolate where it's likely to be.

If the dialler isn't operating, or not properly, you might for example start by checking whether any of the fragile connections between the keypad and

TABLE 1: DIALLER IC CROSS REFERENCE

Pulse diallers

SHARP	MOSTEK	NATSEMI	G.I.	SAMSUNG
LR40981	MK50981	TP50981		
LR40982	MK50982	TP50982		KS5804
LR40992	MK50992			KS5805A
LR40993				KS5805B
LR4173	MK5173			
		TP9151	AY5-9151	
		TP9152	AY5-9152	

DTMF diallers

SHARP	MOSTEK	NATSEMI	T.I.	SAMSUNG	PLESSEY
LR4087	MK5087	TP5087	TCM5087		MV5087
LR4089	MK5089		TCM5089	KS5808	MV5089
LR4091	MK5091		TCM5091		
	MK5092		TCM5092		

TABLE 3: AMI S2560A-2 DIALLER PIN PROGRAMMING			
Pin #	Function	Low-Vss	High-Vdd
12	Break/Make Ratio	67:33	60:40
14	Dialling Pulse Rate	10 pps	20 pps
15	IDP length	800ms	400ms

Above: The mode control pin programming for the AMI dialler chip used in Fig.6. Right: The tone pair combinations used for DTMF dialling. The table at the bottom of the facing page shows a cross reference between many of the dialler chips currently available.

TABLE 2: DTMF TONE COMBINATIONS					
		High group tones (Hz)			
		1209	1336	1477	1633
Low Group	697	(1)	(2)	(3)	(A)
	770	(4)	(5)	(6)	(B)
	852	(7)	(8)	(9)	(C)
	941	(★)	(0)	(#)	(D)

This table shows the relationship between keypad digits and DTMF tone pairs. Pressing the (5) key, for example, produces a 770Hz tone along with a 1336Hz tone.

the IC has parted company. Check to see if the chip shows any signs of activity, such as the oscillator running, as keys are pressed. If all seems in order, you could trace the path of the dial pulses through the switching transistors.

Always check to see whether the various circuits seem to have the right supply voltages. Inspect any flexible connections between switches, microphones etc. Generally speaking, no circuit will give the right output if its not provided with the right input.

If you do isolate a faulty component, your next problem is to locate a suitable replacement. Most parts suppliers have suitable replacements for the high voltage switching transistors, and common resistors, capacitors and diodes should present no problem. As replacements for speech circuit transistors, BC337s will often suffice, especially if higher gain types can be selected.

Probably the most difficult component to replace will be the dialler IC. The telephone supplier may be able to sell you a spare or, with a bit of luck, one of the semiconductor distributors or specialist parts importers may carry the part you need.

Although there is always the chance of disappointment when attempting to repair economy 'throw-away' equipment such as this, just as often your efforts could result in many more years of trouble free performance. At worst, there is still the garbage bin option, but at least you will have gained a bit more knowledge and experience in the process.

Knowing what a phone's internal circuitry has to do, and at least the broad way in which it's done, is usually more than half the battle. So hopefully the above information will get you off to a good start. EA