1
The telephone system

Each subscriber's line consists of a single pair of wires, connecting the subscriber's premises to the local exchange. Such a line is called a 'direct exchange', line or 'DEL'. (In American terminology, the exchange is called the central office, abbreviated to 'CO', and a direct exchange line is called a 'CO line'.)

This simple connection, however, is used for a remarkable number of functions. Firstly, the line carries the voice frequency (VF) communication, in both directions. Such communication may be in the form of speech, of modem or facsimile (fax) tones, or of other tones. Secondly, the line also carries signalling information, both from the subscriber to the exchange and vice versa. Some signalling is in direct current (DC) form, and some is in alternating current (AC) form.

Let us consider a telephone call from initiation, and observe how the progress of the call is marked by various forms of signalling. Figure 1.1 shows the essential parts of a crude telephone and exchange line connection, which may aid understanding.

Stage 1. The subscriber lifts the handset, which releases a switch mechanism in the telephone called a hookswitch. (The name recalls the days when the earpiece ('receiver') was suspended from a hook when not in use, this hook being arranged to operate a switch. The term 'hang up' has the same origin.) This has the effect of connecting some part of the telephone's circuitry across the ends of the two wires of the exchange line, providing a DC path. At the exchange, a 48 volt (V) power supply is indirectly connected to each subscriber's line. When the DC path is completed, current will flow, and the 'local loop' is formed. Providing the current flow is greater than about 20 milliamperes (mA), the line relay in the exchange, whose

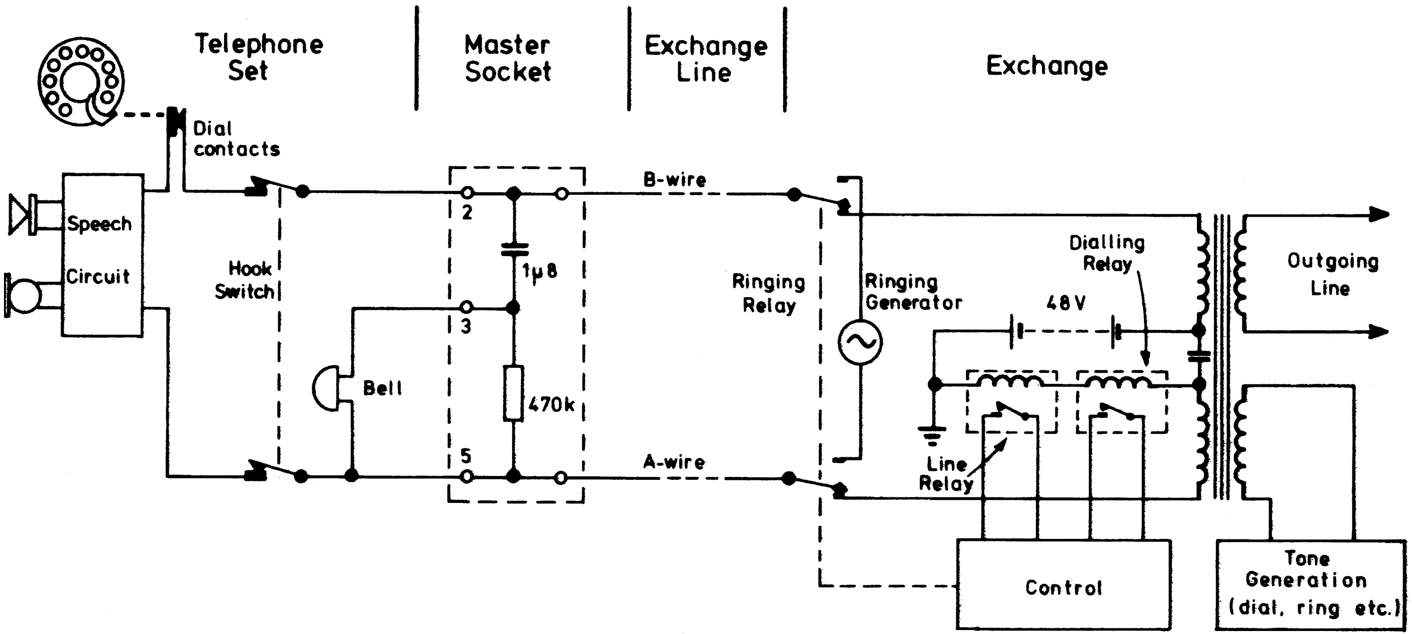


Figure 1.1 *Crude telephone with exchange line connection.*

coil is connected in series with the local loop, will operate, thus signalling to the exchange that service is required.

Note that the description of the operation here and in ensuing paragraphs refers to the use of relays. In specific exchanges, these may well be replaced by solid-state sensors and switches, but this does not affect the theory of operation.

Stage 2. The exchange now connects the subscriber's line to the system and indicates that it has done so by applying the familiar 'dialling tone' to the line.

Stage 3. The subscriber now indicates to the exchange the number of the called party. This may be achieved in one of two ways:

- By means of loop disconnect (LD) or 'pulse' dialling – turning the current on and off.
- By means of tone (dual tone multi-frequency – DTMF) – dialling – sending 'musical' tones for the number dialled, without interrupting the connection.

Loop disconnect dialling has been in general use since the early days of telephony. It operates by temporarily disconnecting the local loop – once for number one, twice for number two, etc., and ten times for number zero. The temporary disconnection is too short for the line relay in the exchange to drop out, since these relays are designed to be of sluggish action. However, another relay in the local loop has a faster reaction time, and it operates in sympathy with the makes and breaks of the dialling contact. While dialling is in progress, the impulses are prevented from reaching the earpiece by closure of a muting switch. Obviously, for the dialling to be successful, the timing of the pulses in respect of both the pulse width and the time between successive pulses, as well as the time between successive digits, has to be controlled. Figure 1.2 shows the timing of these pulses as applicable to the UK telephone network.

(Loop disconnect dialling is a development of the first form of automatic dialling to be used. It was patented in 1891 by Almon Strowger, a funeral director of Kansas City. He suspected that he was losing business to a rival because calls for 'a funeral director' were being manually routed by the telephone operator to his rival. Legend has it that the telephone operator was the rival's sister!)

In the case of tone dialling, which can only be accepted by the more modern telephone exchanges (such as System X), each digit is represented by two simultaneous tones. These tones are generated within the telephone, and selected according to which button of the keypad is pressed. Figure 1.3 shows the keypad layout and the corresponding combination of tones that will be transmitted along the telephone line to the exchange.

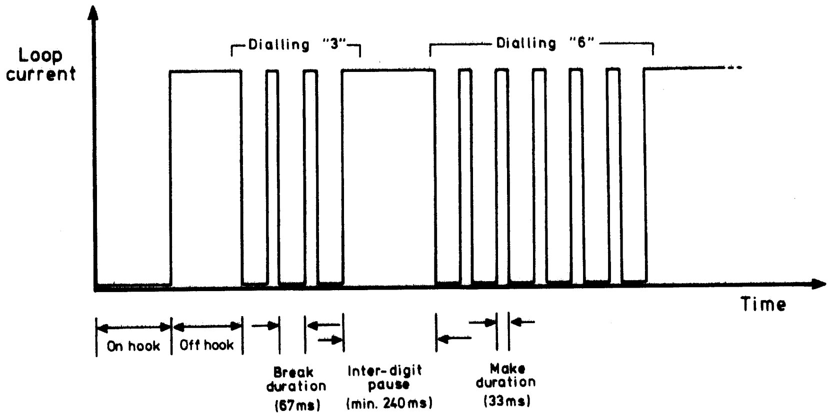


Figure 1.2 *Loop disconnect dialling – pulse timing.*

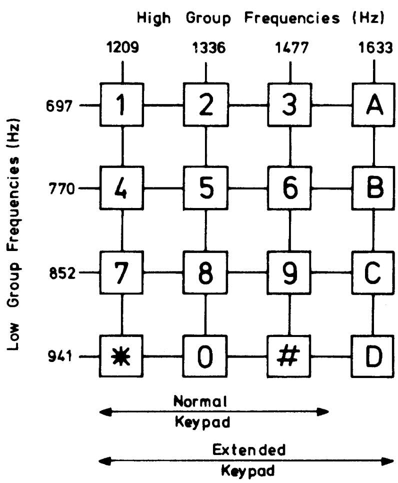


Figure 1.3 *MF dialling – keypad layout and tone frequencies.*

The tones are decoded at the exchange and the call routing controlled accordingly. Tone, or DTMF, dialling is both faster and more reliable than the older loop-disconnect method.

- It is faster because each digit takes just a few tens of milliseconds (ms) to transmit – the speed is dictated largely by the dexterity of the operator in pressing the buttons. However, pulse-dialled digits will take between 1 and 2 seconds (s) each.
- It is more reliable because the decoding and switching operations are

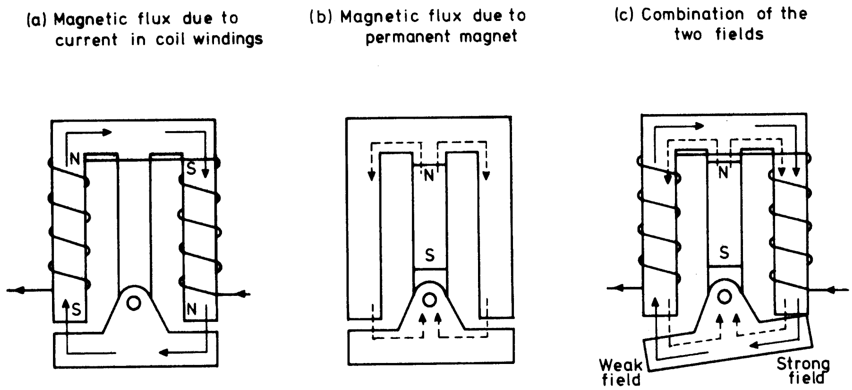


Figure 1.4 *Telephone bell – principle of operation.*

accomplished electronically, so sensitive electromechanical devices can be eliminated.

- There is another benefit to the tone-dialling system. The tones may be used for signalling purposes after the connection has been made. This permits, for instance, certain answering machines to give up their stored messages in response to a predetermined code. It has also allowed BT to introduce their 'Star Services', whereby subscribers may store frequently used telephone numbers at the exchange, set up conference calls, and control call diversion, call barring, and other facilities.

Stage 4. The subscriber now hears one of a number of possible tone signals indicating number ringing, number engaged, or number unobtainable.

The distant exchange now applies a ringing signal to the line of the called party. It is perhaps worth remembering that at this stage of the call, a considerable amount of equipment is in use, but there is no revenue resulting, since the voice frequency path has not been established. It is therefore in the interest of the service provider that the attention of the called party is attracted as soon as possible. The bell, or other audible signal, needs to be as loud as possible, within acceptable limits.

The ringing signal (or 'call arrival indication') is an AC signal of around 90 V, at a frequency of about 20 Hertz (Hz). The telephone bell, which is energized by this signal, is shown diagrammatically in Figs 1.4 and 1.5. The bell consists of an armature that is pivoted between two electromagnets consisting of two coils wound onto permanent magnet cores. The two coils are wound so that the magnetization due to the ringing current is of opposite polarity on each side of the structure. A permanent magnet bias ensures that the armature will be attracted first to one side, then the other, in sympathy with the alternating magnetic field.

Note that the bell is powered through a capacitor, normally located in the master socket, so that the bell coil windings do not provide a DC path to

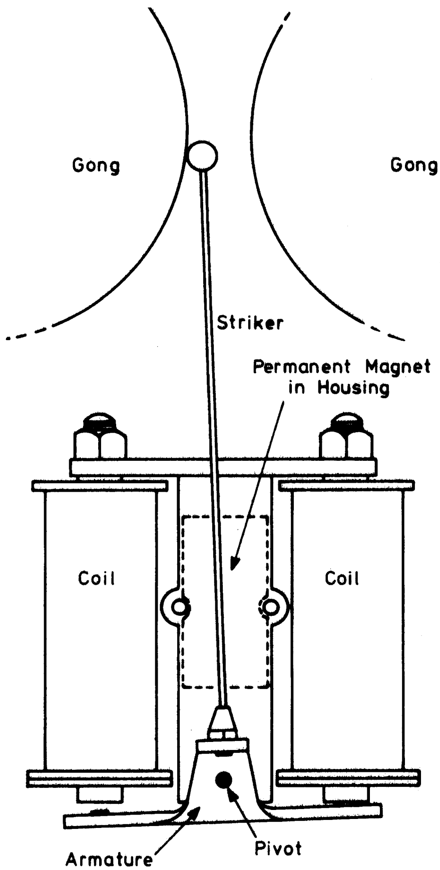


Figure 1.5 *Telephone bell – physical arrangement.*

complete the local loop. At the time when the ringing signal is applied to the line, only the bell circuitry is connected to the line, the rest of the telephone being disconnected by means of the hook switch.

When one of a number of parallel-connected telephones is being used for pulse-dialling, the continual interruptions of the line current may cause the bells of the other telephones to 'tinkle' in sympathy. In order to overcome this annoying phenomenon, a second pair of contacts in the hook switch is used to connect a low resistance across the bell circuit.

Electronic ringing circuits are becoming increasingly common in telephones today. There are a number of benefits associated with the use of such circuits.

- They are generally smaller and lighter than the conventional bell.
- Variation of both pitch and of volume is quite easy to arrange. Use of different pitches allows the identification of one of several telephones in an

office, and electronic volume control may allow the volume to increase the longer the phone rings.

In the conventional bell system, the physical movement of the gong is directly controlled by the amplitude and frequency of the ringing signal. In the case of the electronic ringing circuit, the signal is merely rectified and smoothed, and used as a power source for the ringer. One advantage of this approach is that it provides another method of overcoming the problem of bell tinkle.

Ringing circuits are commonly available as integrated circuits, and so elaborate tone generation is quite feasible – for instance, the low-frequency alternation between two different tones to produce a warbling effect.

The electrical signals still have to be converted to audible tones, though, and electromagnetic or piezoelectric transducers may be used for this purpose. When the telephone is answered by the called party, the hook switch is closed, thus completing the local loop. The exchange recognizes the off-hook condition, and responds by replacing the ringing signal with the voice frequency path from the calling end. This state is maintained until one party hangs up, and their local loop is broken.

It is a feature of the telephone system that the call path will be maintained until the calling party hangs up. This means that the called party can replace the receiver or unplug the telephone and move to a new location to take the call in comfort and/or privacy. It also means that charges for the call continue to accrue until the calling party hangs up. There have been occasions when a call has been made, and terminated properly by the recipient, but not by the call originator who inadvertently left the receiver off the hook. In one instance, a transatlantic call was made immediately prior to a two-week absence, and a very large charge indeed was incurred. Happily, in this case, BT waived the charge when advised of the circumstances.

The speech circuit

The speech circuit performs three basic functions:

- To balance the send and receive signal levels so that the correct talk and listen sensitivities are achieved.
- To provide a degree of 'sidetone' – i.e. the application of some of the microphone signal to the earpiece. If this were not done, the user would complain of a 'dead' earpiece.
- To provide regulation to compensate for the varying line lengths. The resistance of the local loop may vary between virtually zero and 1000 ohms (Ω) – for which the telephone must be able to compensate. This is classically achieved (in the 700-series telephone, for instance) by introducing a circuit whose resistance decreases with increasing DC line

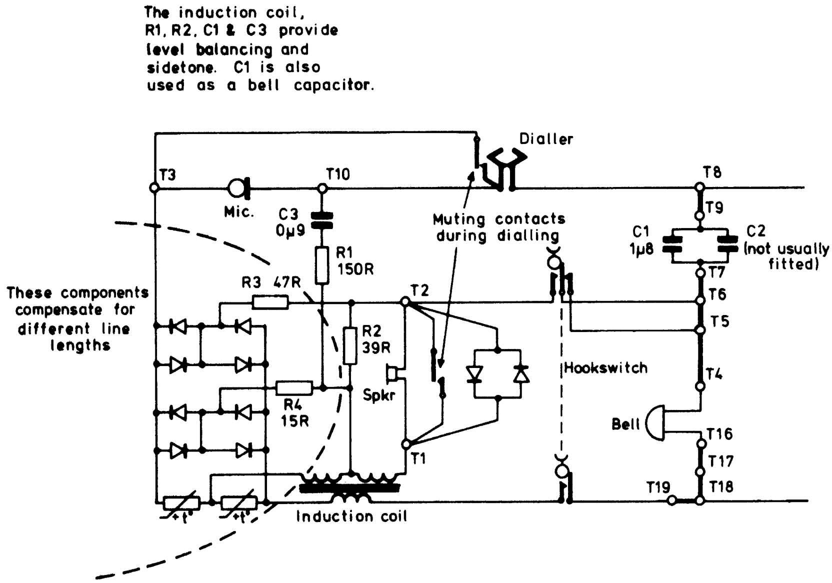


Figure 1.6 *700-Series telephone circuit diagram.*

current, and using this to shunt the microphone. A similar arrangement, though with rather less loss, is used for the receive signal.

The BT 700-series telephone was the standard issue for many years and will be familiar to all. It was the first to be available in a variety of colours, and was equipped with an internal bell and rotary dial, although some push-button versions were produced. The circuit diagram of the basic 700-series telephone is reproduced in Fig. 1.6.

Elimination of induced noise

Some readers may be surprised that the telephone network could ever work, given the very long line lengths employed and the amount of electrical pollution, mostly mains hum, which must be picked up. The use of a balanced line and differential amplifiers solves this problem.

The principle is that the voice frequency signals are applied as a difference signal between the two wires of the pair, and at the distant end, only the difference signal is amplified. This of course assumes that the electrical characteristics of each leg are identical. Any imbalance due to a fault condition will cause hum and other noise to become apparent. Such faults may be caused by leakage paths to earth from one wire and not to the other (perhaps in turn due to dampness in a cable), or by complete disconnection

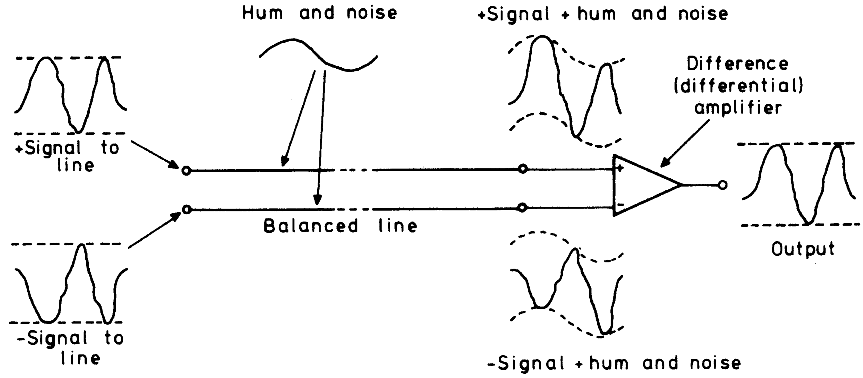


Figure 1.7 *Principle of balanced line.*

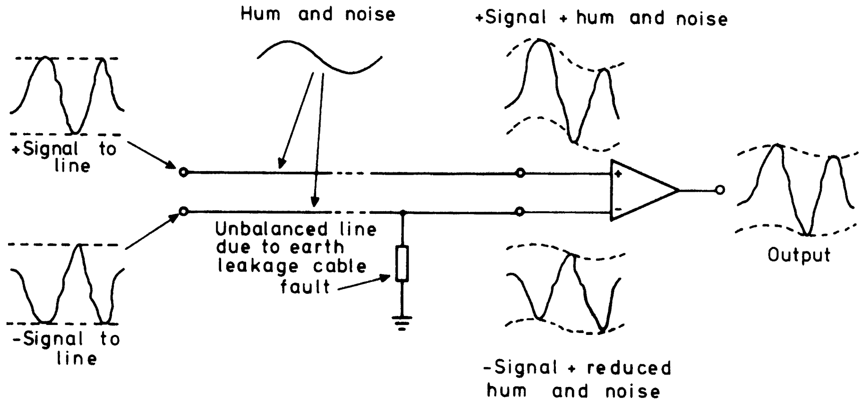


Figure 1.8 *Fault condition on balanced line.*

of one wire further down the line. Figs 1.7 and 1.8 illustrate the principle and possible problems that may be encountered.

Voice channel bandwidth

As any telephone user is aware, the quality of the transmission of the human voice over a telephone link is far from perfect. This is not a problem however. The human voice contains many more sounds than are actually required for intelligible speech, and savings may be made in the costs of transmission equipment if only the essential range of frequencies is transmitted. The voice bandwidth, as it is called, is limited to the range 300–3000 Hz (3 kHz), which is transmitted within a voice channel that is 4000 Hz (4 kHz) wide (see Fig. 1.9). The apparent wastage (the difference between the voice channel and the voice bandwidth) provides a guard band

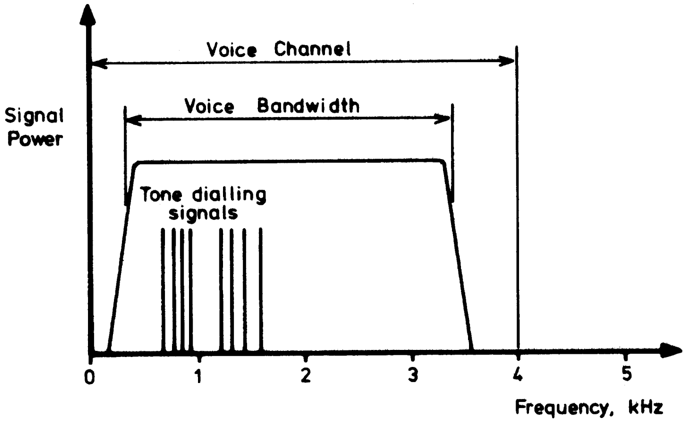


Figure 1.9 *Voice channel bandwidth.*

between adjacent channels when frequency division multiplexing (shortly to be described) is in use and also allows various system control signals to be inserted in the transmission path.

Beyond the local loop

The connection between each subscriber and the local exchange is normally made by means of a pair of wires, which will almost certainly be incorporated at some stage within a multipair cable carrying many subscribers' lines. Continuation of this method of distribution beyond the local exchange would attract a very high cost indeed due to the amount of copper in use, and the associated amplifiers, etc.

For this reason, methods have been devised of transmitting many channels at once through one physical, or radio, connection. In one such system, channels are formed and stacked one above the other in frequency to form a 'baseband'. The baseband is then transmitted by coaxial cable, by a radio link (usually in the microwave range) or by fibre-optic cable. This method is known as frequency division multiplex, or FDM.

There is another method of multiplexing, by means of time division. In its simplest form, each channel is sampled briefly in sequence, in the manner of a motor-driven rotary switch. At the distant end, a similar rotary switch is driven in synchronism with the first, and thus each sampled channel may be separated out again (see Fig. 1.10). Providing there are sufficient samples per unit time, the original signals may be reconstructed. Of course, the sampling and switching are performed electronically rather than mechanically, and modern systems convert the analogue signals to digital form first, which then provides the basis of the pulse code modulation (PCM) method of transmission.

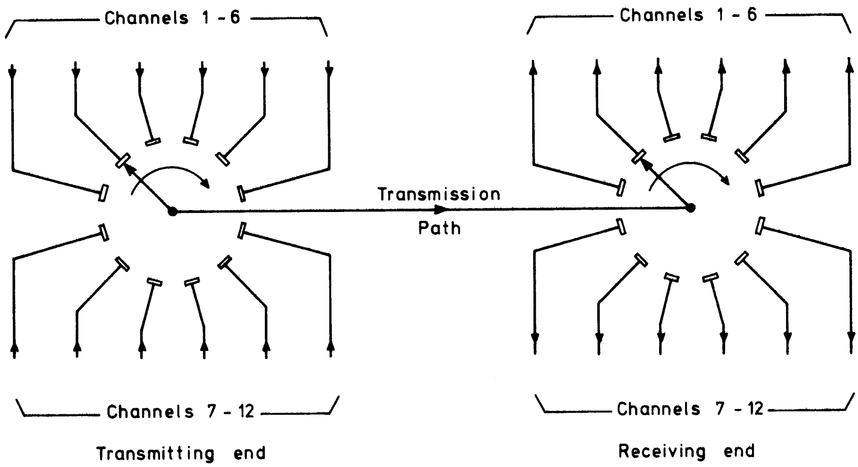


Figure 1.10 *Principle of time-division multiplex.*

By using fibre-optic links, where the transmission medium is light, very wide bandwidths with associated high traffic densities may be employed. Fibre-optic links have a high immunity to both interference and eavesdropping, and the cables employed are less bulky and require less protection from chemical contamination and moisture than their copper counterparts.

Using service providers other than BT

Since the liberalization of the telephone service, licences have been granted to other service providers such as Cable and Wireless. Immediately, the problem of fair competition arises, since BT have inherited an enormous infrastructure of telephone equipment and cabling, and other carriers would need to install duplicate systems before starting to earn any revenue. The difficulties have been overcome to some extent by making it possible for customers to make use of the local loop for access to the trunk networks owned and operated by other carriers, where the traffic may be carried at a lower charge.

In the simplest case, a customer wishing to make a long-distance call will decide to use Cable and Wireless. From a telephone connected to the BT network, first of all, he will dial a Cable and Wireless access code, followed by his own unique code (PIN) to enable the call charge to be billed to the right customer, and then the destination telephone number. This is a generous sequence of digits to be dialled, to say the least, and so telephones have appeared with a programmable button that does nothing more than access a memory location in which may be stored the access code and the customer's PIN. The dialling sequence for a long-distance call over a Cable and Wireless line is now the same as for a BT call, with the addition of just one button press. However, the customer still needs to decide if the call is going to be cheaper via BT or Cable and Wireless, and this will depend upon his geographical location and the destination of the call. A BT local call is usually cheaper, but calls around the outskirts of the local call area may require a reference document to determine the least-cost routing (LCR) for any particular call.

Special adaptors, when inserted between the telephone and the BT socket and suitably programmed, will not only insert the access code and customer PIN, but also decide by which carrier the call will be cheaper, and route the call accordingly. This approach is the most obvious if a number of people will be making calls. Many private exchanges (PABXs) now offer automatic LCR built in to the exchange, thus obviating the need for an additional adaptor unit.

More recently, service providers have made use of the calling line identification (CLI) facility that transmits the caller's telephone number when a call is set up. This number uniquely identifies the user to the service provider, and the use of a PIN becomes unnecessary.

Integrated services digital network (ISDN)

While the 4 kHz voice channel is perfectly adequate for ordinary telephone conversations, it is quite limiting when the connection is used instead for data communication purposes. An increase in bandwidth will allow higher frequencies, and therefore higher data rates, to be accommodated. In order to meet the demands of the current age, where large amounts of data are being transferred between business users, and many people are requiring high speed access to the Internet, another network, running in parallel to the analogue PSTN, has been introduced. By using digital techniques, customers may benefit from the use of wider bandwidths, and therefore higher data-transfer rates. A different interface from the simple local loop just described is required, but customers are still able to initiate and receive calls via the PSTN. The advantages become apparent when an ISDN customer wishes to communicate with another ISDN customer, where the wide bandwidth available allows, for example, high-speed data transmission, high-quality speech, and video conferencing.

An ISDN connection is divided into channels of two different types. The first, the 'B' channel, carries digital data at a rate of 64 000 bits per second (64 kbit/s), where a bit is a Binary digIT. The second, known as the 'D' channel, is restricted to a data rate of 16 kbit/s, and is normally used for signalling purposes. Because the data channels are separate from the signalling channels, the data stream is uninterrupted by control signals, therefore allowing the highest possible rate of data flow.

There are two types of ISDN access, known as basic rate access, abbreviated to BRA, and primary rate access, or PRA.

Basic-rate ISDN provides a customer with two 'B' channels and one 'D' channel. With appropriate terminal equipment, this may use the existing ordinary telephone line to a customer's premises. The network operator may also find it to be more cost-effective to provide BRA to a customer who requires an additional telephone line than to install a new line.

Primary-rate ISDN provides for 30 'B' channels and two 'D' channels. This requires an upgraded line between the customer and the local exchange, since any existing telephone lines will be unable to operate at the required 2 million bits per second (2 Mbit/s) data rate.

Advantages of using ISDN

The wider bandwidth and higher speed of ISDN mean that applications that were once unthinkable have now become a reality. For example, many business meetings do not need the physical presence of the members, but could be effectively accomplished if all the parties were connected together by voice and by video. Two or more ISDN customers can do this by a system known as 'video conferencing'. The benefits are obvious – thousands of pounds may be saved in travel and accommodation costs and travel time.

Recent case studies have identified other areas where substantial cost and time savings can be made through the use of ISDN.

- A customer with branch offices in three other countries used to exchange computer files by sending floppy discs by mail or by courier. Now, by transmitting the files by ISDN, the data are up to date and readily accessible to all.
- A graphic-design company relied on mail, couriers, or modems for communicating their designs to customers. By switching to ISDN, they have been able to save substantially on courier fees, and have expanded their business by targeting potential customers who also have ISDN capability.
- The head office of a chain store has installed ISDN in all of its branches. This enables daily sales figures to be routinely retrieved from each branch by the head office and stock to be automatically replenished. It has also speeded up credit-card transactions due to the higher connection speed of ISDN.
- Many people whose work is information-based are now able to work at home because they are able, thanks to ISDN, to download computer files from their place of employment as quickly as if they were using a computer in the office. Apart from the time and money being saved due to reduced travelling (with environmental benefits as a spin-off), the employer can reduce the amount of accommodation required for his desk-bound staff.

Basic rate access

The original version of BRA, dating back to the mid-1980s, was known as ISDN 2. Subsequent harmonization with European standards has brought about a modified, and now current, version – ISDN 2e. The service is aimed at the medium-sized business market and arrives at the customer's premises via an ordinary telephone cable pair. At what is known as the 'U' reference point, this pair is terminated at a network terminating unit, or 'NT', which converts the two-wire network connections to a four-wire interface, where the transmit and receive paths have been separated. The customer may now connect any ISDN-compatible equipment to this interface, known as the 'S-bus'. If non-ISDN-compatible equipment, such as an ordinary telephone or modem, is to be connected, then a terminal adaptor ('TA') is connected to this point. Fig. 1.11 shows the general scheme. Under normal circumstances, power for the NT and for ISDN-compatible terminal equipment ('TE1') will be supplied by a power adaptor connected to the local mains supply.

Up to eight items of TE1 may be connected, in parallel, to the S-bus, and each one may be separately addressed by the network connection. This means that up to eight telephones, say, may be connected. The network

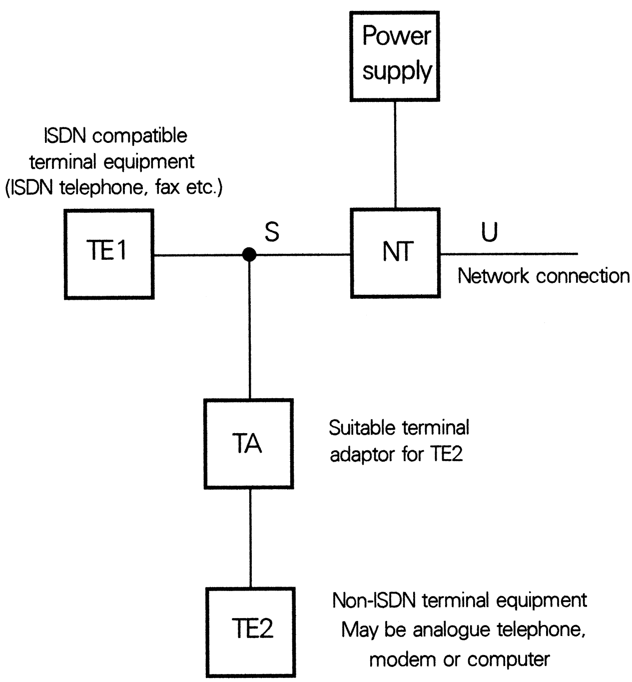


Figure 1.11 *ISDN 2e implementation at customer's premises.*

operator will allocate a block of consecutive telephone numbers to the connection, the last digit(s) of which will not be used for routing the call through the public network, but will be broadcast to all of the terminals connected to the S-bus. Each terminal will be programmed to respond to a specific number. The scheme is known as multiple subscriber numbering, or MSN.

In the case of a power failure, only one item of TE will be operative (normally a telephone for emergency use), the power being derived from the network.

While this level of complexity is acceptable to the medium-sized business community, it is rather a daunting prospect for the small business and for the domestic customer. This is a considerable market segment that nevertheless would embrace the idea of high-speed Internet access with open arms. Mindful of this market, BT has introduced more user-friendly forms of ISDN 2e, known as Business Highway and Home Highway.

In both cases, the network terminator and two terminal adaptors have been combined into one unit. This scheme is shown in Fig. 1.12. In this implementation, the customer has two familiar analogue telephone sockets to which may be connected ordinary telephones, fax machines and the like. There are also two digital connection points, primarily intended for one or two computers to be connected. Suitable terminal adaptors will be required for the digital connections, which could be either internal computer expansion cards, or separate external units. Each analogue socket has a separate telephone number, and a third number is provided for the direct digital access.

Not all of the sockets may be used at once, however. The permissible combinations are shown in Fig. 1.13. Each connection point, or 'port', uses 64 kbit/s of the available 128 kbit/s bandwidth. However, to achieve the highest possible data rate, the entire bandwidth may be devoted to one computer link, in which case, no other connection may be used simultaneously. It should be noted that in order to convert to a high-speed Internet connection, the user must contact his ISP (Internet service provider) who will provide a different telephone number for ISDN access.

Unlike ISDN 2e, BT Highway does not provide a power source for ISDN-compatible equipment. Any devices connected to the digital ports will therefore need their own power supplies.

BT Home Highway differs from both ISDN 2e and Business Highway in that multiple subscriber numbering – MSN – is not supported. However, the requirement for this feature is unlikely in domestic circumstances.

Primary rate access – ISDN 30

ISDN 30 is aimed at the large business customer, with more than ten analogue lines. A single ISDN 30 connection, usually made by fibre-optic link

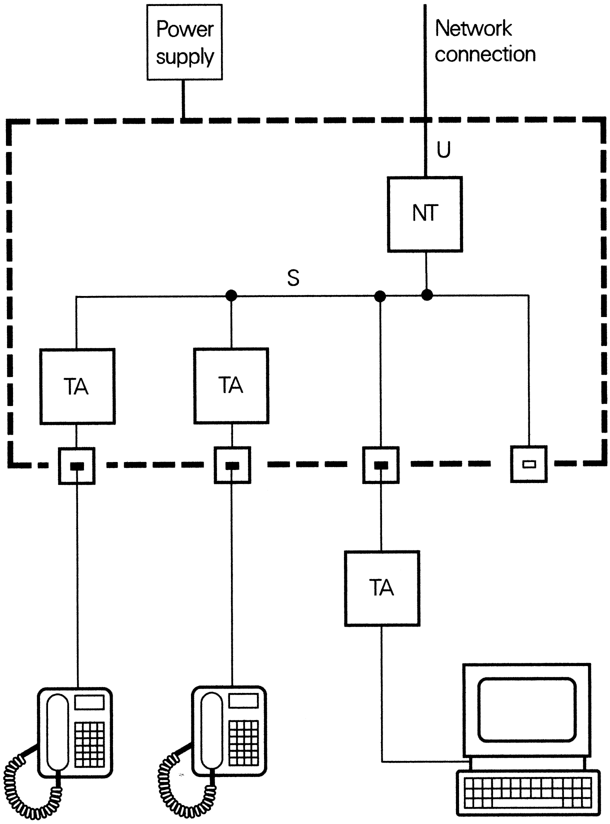
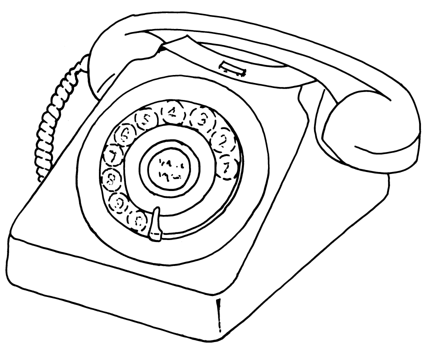


Figure 1.12 *BT Highway implementation at customer's premises.*

between the customer's premises and the local exchange, will support up to 30 ISDN channels of 64 kbit/s each, although the minimum configuration is eight such channels. At the customer's premises, these channels would normally be terminated in a suitable telephone exchange that would be able to handle the complex procedures involved. Because the precise method of installation will be largely manufacturer-dependent, as will the facilities available and the methods of programming, installers would need to equip themselves with the specific knowledge required by attending a manufacturer's course. For those readers wishing to learn more about ISDN, Appendix 2 lists some of the titles available.

Analogue telephone 1	Analogue telephone 2	Digital 1	Digital 2
✓	✓	✗	✗
✓	✗	64 k	✗
✗	✗	64 k	64 k
✗	✗	128 k	✗

Figure 1.13 *BT Highway permissible access combinations.*



'700 Series' U.K. 1959