
UNIT 7 SWITCHING AND TELEPHONY

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7.1 INTRODUCTION

A telephone is a commonly used instrument that you access now and then. It allows you to communicate a voice signal with others. The advances in telecommunication technology have now made it possible to communicate even the text messages, pictures, video signals etc. from one place to another using the telephone lines. This is closely related to advances in electronics and device technologies as well as increased efficiency in the data transmission due to the use of fast data transmission media and modulation techniques. The use of computers in handling of telephone traffic has expanded the capacity of telephone networks to cater to large number of users. Whenever you use a telephone, you must be wondering, what system works behind the scene that allows you to communicate with your friends either in the same town or anywhere in the country, or for that matter, any where in the world. In this unit, you will learn about the telecommunication system and how it evolved, what are its components, and also briefly review the modern technological trends in the field.

Basically the journey of telecommunication began in the mid nineteenth century with Samuel Morse inventing the Telegraph. He was the first person to wonder, *whether a message can be passed through a metal wire*. In 1844, with the help of a predecided code, he achieved the communication between Washington DC & Baltimore for the first time using electrical signal.

Direct communication of voice was pioneered by Alexander Graham Bell in 1876 when he transmitted voice using the telephone instruments he had developed. Later on, with development in technology, telephone exchanges to handle multiple users were established. The scientific and technological advancement in the field of

electronics, computers and communication has given a big boost to the modernisation of telecommunication media. One of the most striking developments has been in the form of cellular mobile telephony. This has revolutionised the access in remote areas since no physical cable path is required for its functioning. With many public and private sector service providers coming in the market, the customers are getting better and better telephone service options these days.

In Sec. 7.2 you will learn about the construction of a telephone instrument and its working. The telephone sets in our homes are connected by a pair of wires with the telephone network world over via a telephone exchange. Here, both transmitted and received signals are carried by the same pair of wires. This is called a two-wire communication. When the signal is to be transmitted over a long distance, it needs amplification. For this purpose, the transmitted and the received signals are to be separated and carried on two separate pairs of wires. This is called four-wire communication. You will learn about the two- and four-wire communication in Sec. 7.3. The dialling of a number from a phone can be done in two major ways, viz. pulse dial and tone dial. In Sec. 7.4 we will discuss the phone dialling. Each telephone is identified by its unique number. There is a universal convention of allotting the numbers to the telephones. This is called numbering plan. In Sec. 7.5 we will briefly discuss the numbering plan. The telephone in our house has to be connected to the main network of telephones, in order that we can achieve connectivity to any telephone in the most remote part of the world. This telephone networking is discussed in Sec. 7.6.

The connection between any two telephones is done by the process of switching. Switching means providing a path for transfer of signal between two telephones. This is done with the help of telephone exchanges. In the earlier days, the exchanges were manually operated. Later on electromechanical switches replaced the manual operator. This historical evolution of telephone switching is detailed in Sec. 7.7. Later on, with the emergence of solid-state devices like transistors and now microprocessors, the switching is done in automatic way where either space division or time division switching is done. We discuss these switching modes also in the same section. In Sec. 7.8 we describe the working of electronic exchange along with the important services provided by it. Modern telephones allow many additional facilities like teleconference, relay of video signals etc. This is done via an Integrated Services Digital Network (ISDN). A brief account of the facilities provided by ISDN is taken in Sec. 7.9. In recent years, mobile telephones are replacing the conventional telephones very fast. In Sec. 7.10 we shall briefly discuss the basics involved in the mobile telephony. A list of some important terms related with telecommunication is given in Appendix A. In Appendix B we shall discuss the working of facsimile machines in brief.

Objectives

After studying this unit, you should be able to:

- describe the construction of a telephones set;
- explain the methods of phone dialling;
- describe the numbering plans in telephony;
- enumerate the stages in the evolution of telecommunication;
- list the components of a typical telecommunication system;
- explain the telephone network;
- describe the working of an electromechanical switching system;
- explain the components and working of automatic exchange;
- list the services provided by ISDN; and
- describe the working of mobile telephones.

7.2 WORKING OF TELEPHONE

In our house, we have a telephone set, through which we dial a number and get connected to the other phone. Then through the mouth piece (microphone) of the hand set, we can speak and through the speaker we can listen to the voice coming from the other end.

Essentially, any telephone has these two basic components viz. a speaker and a microphone. A schematic diagram of basic telephone is shown in Fig. 7.1

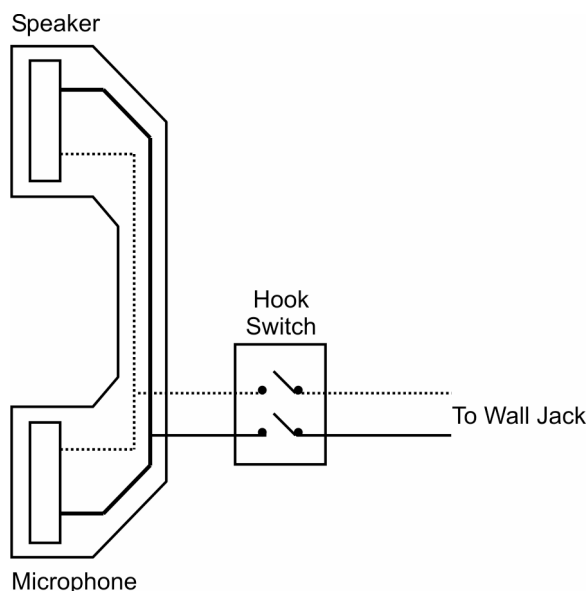


Fig. 7.1: Basic telephone set

The microphone in this set is an actuator, which converts mechanical vibrations into electrical signals. In old sets, the microphones were made of carbon granules pressed between two thin metallic plates or diaphragms. When we speak in this microphone, the air pressure is exerted on this metal plate, which presses the carbon granules, thereby altering the resistance of the microphone. Hence the current flowing through the microphone is changed, which can be used as an electrical signal for transporting the voice signal over the connecting wires. Recently, more efficient devices like piezoelectric actuators are used as the microphones.

The speaker in the telephone is a diaphragm, which vibrates in accordance with the electrical signal coming from the sender side. In earlier telephone sets, an electromagnetic armature attached to a diaphragm was used as the speaker. The incoming signal in the form of varying current was passed through the coil of the armature. The armature plunger would move according to the current in the coil. This would move the diaphragm attached to the plunger and generate the corresponding sound signal. In modern telephone sets, the armature-coil assembly is replaced by solid-state devices like piezoelectric actuators.

A telephone is connected to the outer world with the help of a pair of wires. In order to obtain the current variations in the speaker and microphone, they should be provided with an electric supply. For this purpose, a dc voltage is supplied to the telephone line from the exchange which is a centre catering for a group of telephones in the given locality. The exchange mainly serves the purpose of handling of the telephone calls, i.e. it provides connection between the two telephones as per requirements. The schematic of connection between two telephones is shown in Fig. 7.2.

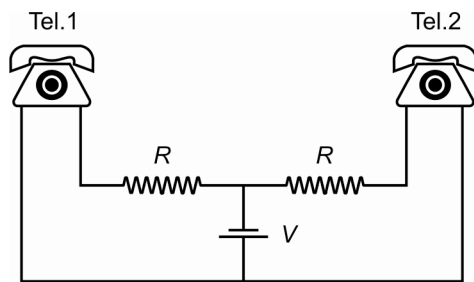


Fig. 7.2: Connectivity between two telephones

Now, you must be wondering, how is it possible to carry out a two-way connection between the two telephones, using just two wires. This is achieved by a simple circuit called **hybrid**, which facilitates the current flow from one telephone to another and backwards without two signals interfering with each other. You will be learning the details of hybrid circuit in the next section.

Now, along with speaker and microphone, we need some more features, in order to make the telephone a useful instrument. In the example shown in Fig. 7.2 we considered the connection only between the two fixed sets. However, in practice we should be able to choose to contact any telephone by dialling a relevant number. For this purpose, a numbering dial or numbering pad has to be provided.

Secondly, we also need an indication, when some call is coming in. This is done with the help of a ringer, which rings a bell, when somebody dials our number.

Hence any basic telephone should contain (i) a speaker; (ii) a microphone; (iii) a dialler and (iv) a ringer. The connection of these 4 components is shown in Fig. 7.3.

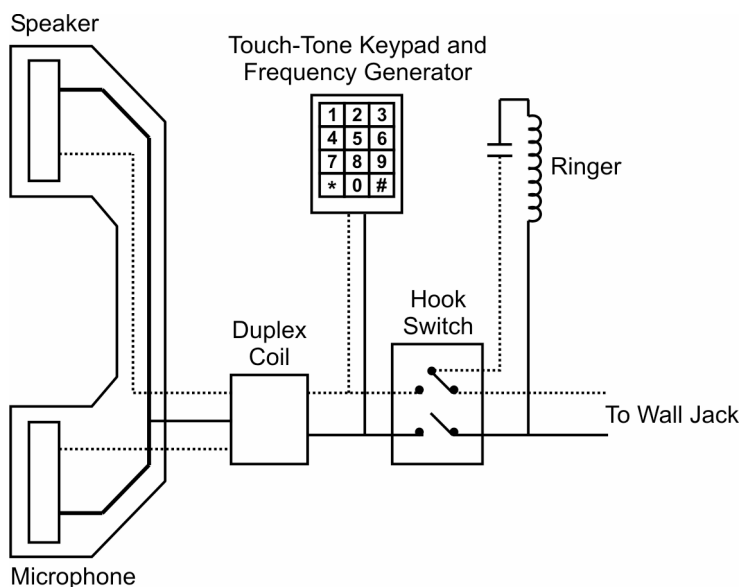


Fig. 7.3: Telephone with dialler and ringer

In this diagram there is another additional block called **hook switch**. This is a switch, which makes the contact of the telephone with the supply line. It is activated by picking up the telephone from the hook.

SAQ 1

You must have noticed that in Fig. 7.3, the ringer is attached to the telephone line before the hook switch. Can you give the reason for it?

*Spend
3 Min.*

With the improvements in technology, the telephone sets have been continuously modified. The miniaturisation in electronics has reduced the size of telephones quite considerably. The operation of entire telephone set can be controlled by one small integrated circuit (IC).

7.3 TWO- AND FOUR-WIRE COMMUNICATIONS

In the earlier stages of development of telephone communication, there were three basic modes of operation: (i) simplex; (ii) half duplex (semi-duplex) and (iii) full duplex.

In **simplex communication** there is only one-way communication as shown in Fig. 7.4a. In this case station *A* is transmitting the signal and *B* is receiving it. There is no back communication from *B* to *A*. Hence, if any error occurs in communication, *B* cannot send a message to *A* informing about the error or request to resend the signal. This type of communication is similar to radio or TV broadcast, where the station sends the signal and we listen to the broadcast. We cannot talk back to the transmitting station.

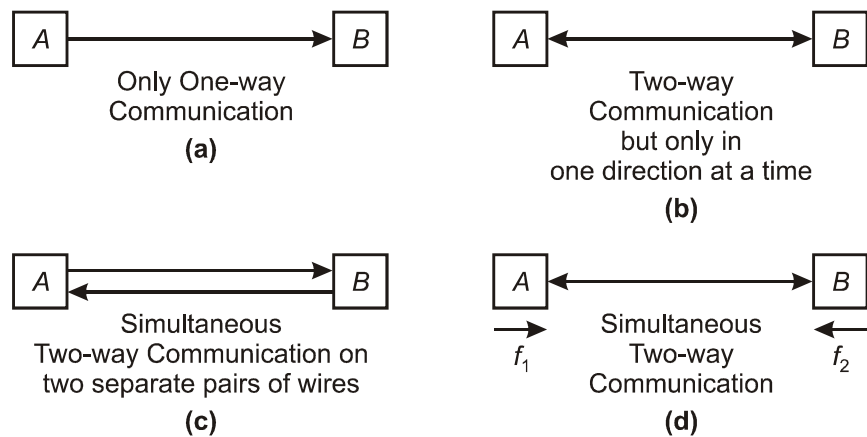


Fig. 7.4: a) Simplex communication; b) Half duplex communication; c) Four-wire Full duplex Communication; and d) two-wire Full duplex communication

In **half duplex** mode of communication shown in Fig. 7.4b both the stations, *A* and *B* can transmit and receive, but at a given point of time, only one of them talks and the other listens. This you might have seen being used in wireless hand-held sets. The third and most common mode of transmission is **full duplex**. Here, both stations can talk and listen simultaneously. This can be achieved by having two sets of cable pairs, one dedicated to transmission and other for reception. So effectively this is a four-wire communication as shown in Fig. 7.4c. It is possible to achieve full duplex communication by using only a single pair of wire (two-wire communication), but then the transmitted signals from both the stations should be riding on different frequencies as shown in Fig. 7.4d.

The telephone instruments in our homes are connected to the local exchange by a two-wire circuit. Hence, the oppositely directed portions of a telephonic conversation occur over the same electrical path. However, in two-wire communication system, a long distance call suffers a large attenuation due to line resistance. Hence, the long distance communication needs amplification of signal. This amplification can be efficiently carried out when the two opposite direction signals are separately carried on a four-wire communication circuit. The conversion of two- to four- and four- to two-wire circuits is done using a **hybrid circuit**. A hybrid circuit is schematically shown in Fig. 7.5a. It typically consists of a balance network connected to a hybrid transformer as shown in Fig. 7.5b. This circuit is essentially a power splitter with four

sets of wire-pair connections. Two pairs are connected to the transmit and the receive lines of a four-wire circuit; one pair is used to connect the two-wire circuit i.e. the telephone instrument; and the remaining pair is used to connect the balancing network that provides matched impedance to the power splitted signal. The balancing network consists of a resistance-capacitor combination, which provides an electrical balance over the frequency range of operation.

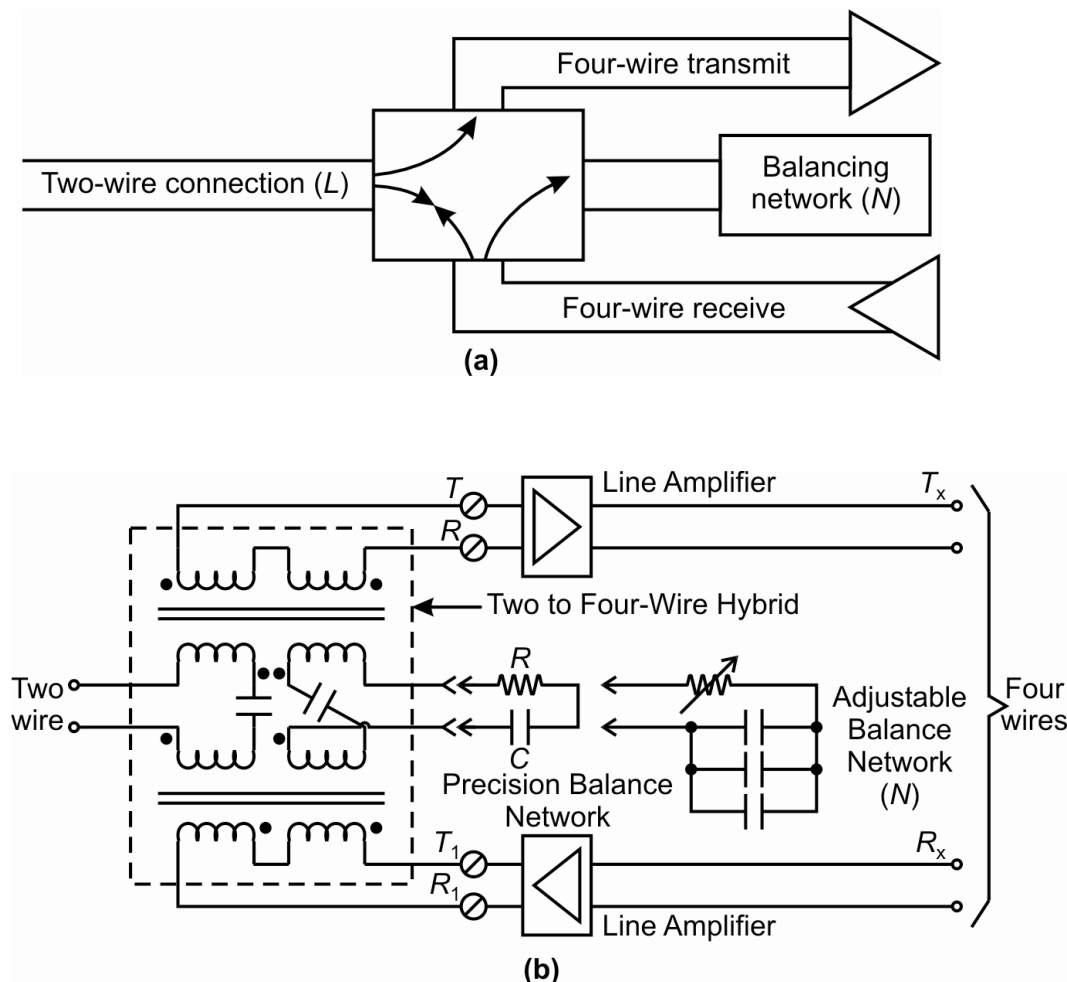


Fig. 7.5: a) Schematic; and b) circuit implementation of hybrid

Signal entering from the two-wire side gets equally splitted between the transmit and receive branches of the four-wire network. The signal on the transmit side travels towards the long distance receiving station, while the remaining half of the signal is dissipated in the impedance of the receiver pair of wires. In such an ideal condition, there is no dissipation in the balancing network.

The signal entering from the four-wire side (from the receiver pair of wires) also gets equally split. One half the signal travels to the two-wire circuit while the other half dissipates in the balancing network.

You must have noticed that half the signal in either direction (i.e. from four-wire to two-wire circuit or from two-wire to four-wire circuit) is available for useful communication while half is lost in dissipation. Also, while inserting a passive network like hybrid, there is some *insertion loss* caused to the signal. Hence, in the hybrid operation, we loose more than half of the signal. However, the amplifier in the four-wire circuit can compensate for these losses and provide reasonable signal transmission over a long distance.

After learning about two-and four-wire communication and the interconverting hybrid circuit, let us now discuss about different phone dialling techniques.

7.4 PHONE DIAL

Two types of phone dials are used around the world. The most common one is called **pulse, loop disconnect**, or **rotary**. The other dialling method is more modern and is called **Touch-tone** or **Dual Tone Multi-Frequency (DTMF)**.

7.4.1 Pulse Dialling



Fig. 7.6: A typical telephone with rotary dial

Pulse phone dialling is traditionally accomplished with a rotary phone dial shown in Fig. 7.6. It is essentially a speed governed wheel with a cam that opens and closes a switch in series with your phone and the phone line. It works by actually *disconnecting* or *hanging up* the phone at specific time intervals. The common standard is one disconnect per digit, so if you dial a "1," your telephone is *disconnected* once. Dial a '3' and you will be *disconnected* three times; dial a zero, and you will be *disconnected* ten times. To allow for the time required to achieve these many disconnects (pulse generations), the rotary dial is rotated from the intended number till the mechanical stopper stops the finger movement any further. This time for rotation and the time required to bring back the dial to its original position is utilised for generating the appropriate *pulses* or *disconnects* corresponding to that number. The accepted standard of pulse rate is 8 to 10 pulses per second (PPS). Some modern digital phone exchanges can accept a PPS rate up to 20.

Most pulse dialling phones produced today use a CMOS IC and a keyboard. Instead of pushing your finger round in circles, then removing your finger and waiting for the phone dial to return before dialling the next digit, you punch the phone buttons as fast as you want. The IC stores the phone number and pulses it out at the correct rate. Because the IC has already stored the dialled phone number in order to pulse it out at the correct rate, it is simple to keep it in the memory and allow the phone to store, recall, and redial the Last Number Dialed (LND). This feature enables you to redial by picking up the handset and pushing just one phone button.

7.4.2 Dual Tone Multi-frequency (DTMF) Dialling



Fig. 7.7: Touch Tone Phone

Touch Tone or Dual Tone Multi-frequency (DTMF) (Refer Fig. 7.7) is the most modern form of phone dialling which is fast and less prone to error than pulse phone dialling. DTMF is an example of Multifrequency shift keying (MFSK) system. In this case audio frequency signals are used to identify the pressed number. These audio band signals can travel down the phone lines farther than pulse, which can travel only as far as your local phone exchange. Touch-tone phone dialling can therefore send signals around the world via the phone lines.

Bell Labs developed DTMF in order to have a phone dialling system that could travel across microwave links and work rapidly with computer controlled phone exchanges. Each transmitted digit consists of two separate audio tones that are mixed together. The four vertical columns on the phone keypad are known as the high group and the four horizontal rows as the low group. For example the digit 3 is composed of 1477 Hz and 697 Hz. The amplitudes of these two frequencies are within 3dB of each other. A complete touch-tone phone pad has 16 digits, as opposed to ten on a pulse phone dial as shown in Table 7.1. Besides the numerals 0 to 9, a DTMF pad has *, #, A, B, C, and D. Although the letters are not normally found on consumer phones, the IC in the phone is capable of generating them.

Table 7.1: DTMF Keypad Frequencies

1	2	3	A	697 Hz
4	5	6	B	770 Hz
7	8	9	C	852 Hz
*	0	#	D	941 Hz
1209 Hz	1336 Hz	1477 Hz	1633 Hz	

The tone frequencies defined above are selected such that harmonics and inter-modulation products will not cause an unreliable signal. No frequency is a multiple of another, the difference between any two frequencies or the sum of any two frequencies does not equal any of the frequencies. The frequencies may not vary more than $\pm 1.5\%$ from their nominal frequency; otherwise the exchange will ignore the signal.

All the telephones are identified by unique numbers as per international conventions. Before discussing about the networks facilitating the connectivity of these telephones, let us discuss the numbering plan used in telephony.

7.5 NUMBERING PLAN

The most important feature of any telephone system is the numbering system adopted for identifying each telephone subscriber line. Every subscriber is provided with a unique number by the exchange. This is typically 3 to 5 digits number depending on the capacity of the exchange i.e. an exchange with 1000 lines can have 3 digits (000-999) subscriber number, while one with 1 lakh lines will have 5 digits (00000-99999) number. These numbers could be used for dialling to the subscriber within the exchange. In the past, when the subscribers were less in number, a single exchange could cater for the whole town. But as the number of subscribers increased, more than one exchanges were needed to cater for the town or city. Hence, to call within the same town, the subscribers belonging to one exchange have to dial a number belonging to another exchange. To facilitate this, the exchanges are identified with a 2 to 4 digit number and the telephone number of any line is the combination of the exchange and subscriber line number. These are the numbers printed in the telephone directory of the town. For example in IGNOU, a typical telephone number is 29532167. In this case, the last 4 digits (2167) are subscriber line number, while first 4 digits (2953) indicate the exchange code. When there are more than one telephone service providers, the first digit of the telephone number can be used to identify them..

Thus, the telephone numbers facilitate communication within the city, i.e. the local calls. When calling a subscriber in another city, it is a subscriber trunk dialling (STD). In this case, it is necessary to dial the code assigned to that city before the telephone number of the called subscriber. To alert the local exchange, when the number dialled is the code of another city (or the call is STD), the dialling starts with '0'. The following numbers are then treated as the city code followed by the subscriber number.

In India, the city codes are 2 to 4 digit long e.g. to dial a number in Delhi from outside, the subscriber has to dial 011 followed by the number in Delhi. Here, 11 is the STD code of Delhi and 0 is the prefix code for indicating that it is an STD call.

The STD codes of some important cities in India are tabulated in Table 7.2.

Table 7.2: STD code of Important Cities in India

City	STD Code	City	STD Code
Agartala	381	Jammu	191
Ahmedabad	79	Kohima	370
Aizwal	389	Kolkata	33
Bangalore	80	Lucknow	522
Bhopal	755	Mumbai	22
Bhubaneshwar	674	New Delhi	11
Chandigarh	172	Panji	832
Chennai	44	Patna	612
Dehradun	135	Port Blair	3192
Gangtok	3592	Raipur	771
Guwahati	361	Ranchi	651
Hyderabad	40	Shillong	364
Imphal	385	Simla	177
Itanagar	360	Thiruvananthapuram	471
Jaipur	141		

Source: <http://www.bsnl.co.in>

A call made out of country is the International Subscriber Dialling (ISD). Here it is necessary to dial the country code of the called subscriber followed by the city code and then the telephone number. To initiate an ISD call from India, it is necessary to dial '00' before the country code.

If you want to call India from another country, the country code for India is 91. So if you are dialling from any other country to IGNOU office then you should dial,

00	91	11	2953	2167
ISD Call Initialisation	Country Code for India	City Code of Delhi	Exchange Code	Subscriber Number

By international convention, the maximum number of digits to be dialled for any international call is limited to 12. The Country Codes of some nations are listed in Table 7.3.

Table 7.3: Country codes of some nations

Country	Code	Country	Code
Australia	61	Japan	81
Bangladesh	880	Mexico	52
Bhutan	975	Nepal	977
Canada	1	New Zealand	64
France	33	Pakistan	92
Germany	49	South Africa	27
India	91	Sri Lanka	94
Iran	98	United Kingdom	44
Italy	39	United States of America	1

Now you may like to attempt an SAQ.

*Spend
2 Min.*

SAQ 2

What is the maximum number of subscribers that can be catered to by a 7 digit telephone number?

After learning about the numbering system in telephones used for local, STD and ISD calls, let us now discuss the evolution of telephone networks.

7.6 TELEPHONE NETWORKS

The telephone demonstrated by Bell was working between two points only. However, practical telecommunication system needs communication from one place to various other places. Now let us learn about the components of a typical telecommunication system. In telephone system, each customer or *subscriber* is identified with a unique telephone number. Hence, when we wish to call somebody, we have to dial the telephone number of that person from our telephone set. How does this dialling allow us to get connected to the intended person? This happens because of a network which switches the connection between the two telephone sets in such a way that there is a direct electric path established between them.

The simplest way to connect two telephones is by using a single pair of wires as in the case of *point-to-point communication*. This was in practice in the initial stages of telecommunication era.

7.6.1 Point-to-Point Communication

Graham Bell invented a point-to-point telephone connection. Fig.7.8 shows telephones connected from one point to another point.

In this case, telephone T_1 could communicate with telephone T_2 , similarly telephone T_3 could communicate with telephone T_4 and so on. But T_1 could not communicate with T_3 or T_4 etc. In this case one telephone was connected to only one other telephone. It means that the communication was allowed only between the two points, which was known as **point-to-point communication**. Even though several telephones were working, a person from one telephone could communicate with only another

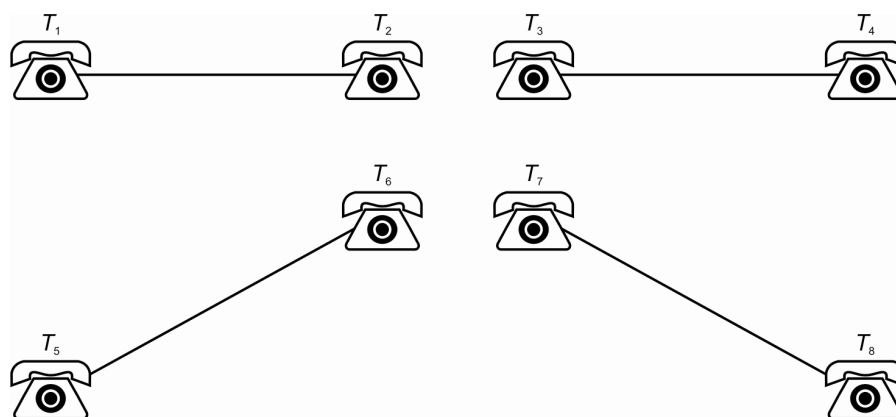


Fig. 7.8: Point-to-point communication

particular phone. To overcome this problem, consider a scheme shown in Fig. 7.9, which is modification of the above scheme.

In this type of networks, all the telephones were connected to the other telephones. A calling subscriber chose the appropriate link to establish connection with another subscriber. Now, in order to draw the attention of the called subscriber some form of signalling in the form of ringing was required with each link. If the called subscriber was engaged, a suitable indication was to be given to the calling subscriber by means of signalling. In Fig. 7.9 there are five telephones and ten point-to-point links. Networks with point-to-point links between all the telephones are known as *fully connected networks*.

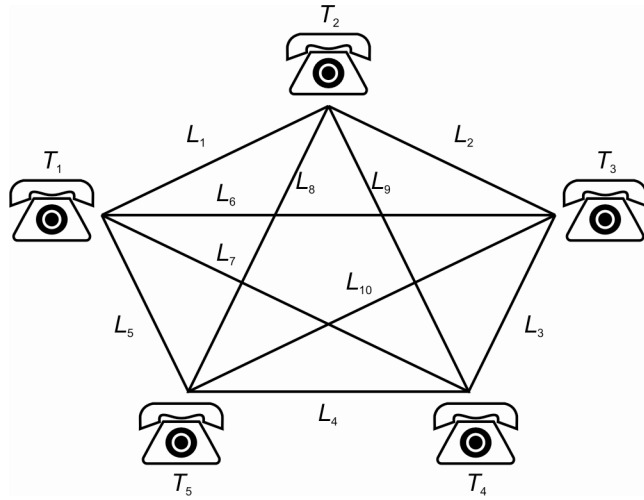


Fig. 7.9: Modified point-to-point network

Spend
3 Min.

SAQ 3

What will be the number of links required to fully connect n telephones?

After solving the SAQ you will appreciate that the number of links required becomes very large even with moderate value of n . For example we require 1225 links for fully interconnecting 50 subscribers. To overcome this problem of large number of links, a *multi-point network* approach is used.

7.6.2 Multi-point Communication

The solution to avoid direct physical links between all the individual telephones but still achieving almost full connectivity lies in the concept of **Exchange**. In this scheme, all the telephone lines are connected to a central place as shown the Fig. 7.10.

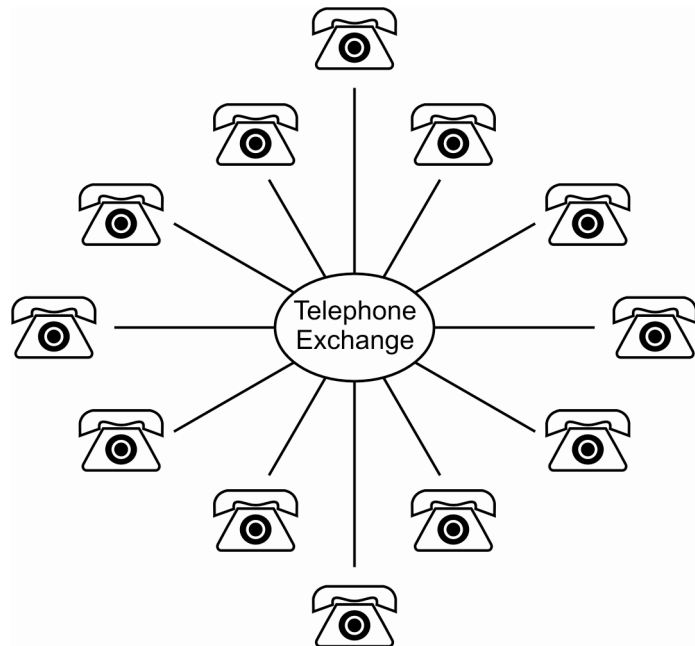


Fig. 7.10: Multi-point communication

This central place, where all the telephone lines are terminated is called a **telephone exchange**. The equipment on which the telephone lines are terminated is called a **switch board**. So, now the numbers of links required are just equal to the number of telephones connected to an exchange.

7.7 EVOLUTION OF SWITCHING SYSTEM

7.7.1 Manual Switching

Previously, an attendant known as *Telephone Operator* manually operated the switch board. The job of the Telephone Operator was to connect one telephone line to any other telephone as seen in Fig. 7.11. In this way any subscriber was able to communicate with any other subscriber of the same telephone exchange. So this scheme worked well for the telephones placed in a locality or neighbourhood. To achieve connectivity between the telephones belonging to different exchanges, it is necessary to establish a link between the telephone exchange of one town with that of another town. These interconnecting links are known as **trunk lines**. The calls made within the same exchange area are known as **local calls**. The inter-city calls are known as **trunk calls**. Thus, huge telephone networks could be established throughout the world which was controlled through the Telephone Operators. The exchanges controlled by Operators are termed as **Manual Telephone Exchanges**.

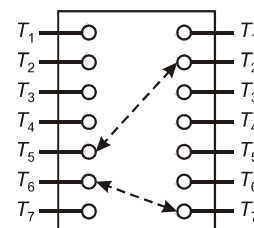


Fig. 7.11: Terminations on switch board

This system was solely dependent on Telephone Operators. To eliminate this dependency, manually operated switchboards were converted into automatic switching equipment. Thus Manual Telephone Exchanges are now replaced by *Automatic Telephone Exchanges*. These exchanges were initially of electromechanical type and now they are replaced by electronic and computer controlled exchanges. In the following sub-section we describe the working of electromechanical switching system.

7.7.2 Electromechanical Switching

The first automatic telephone exchange was developed by A.B. Strowger in 1889. This is also known as **Step-by-Step exchange**. In this exchange, electro-mechanical devices called **selectors** are used to perform the switching operations.

A typical selector starts in the 'home' position and with each impulse generated by pulse dialling the wiper contacts would progress round the output bank to the next position. Each output would be connected to a different subscriber, thus the caller could connect to any other subscriber who was connected to that bank, without any manual assistance from an operator.

In Fig.7.12, the selector has 10 outputs, so a caller can choose to connect to any of the 10 different subscribers by dialling any digit from 1 to 0 (0=10). This sort of automatic selector is known as a **Uniselector**, as it moves in just one plane (rotary).

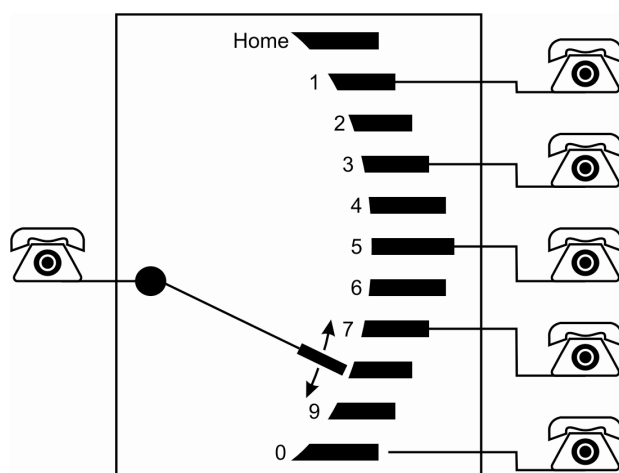


Fig.7.12: Diagram of a Uniselector

By mounting several arcs of contacts discussed in uniselector on the top of each other, the number of outlets can be increased significantly. But the wipers are then required to move both horizontally to select a bank and then vertically to move around that bank to the required outlet. Such a selector is known as a **Two-Motion Selector** (Fig.7.13). Two-motion selectors typically have 10 rows of 10 outlets, thus 100 possible outlets altogether. A two-motion selector can therefore accept two dialed digits from a subscriber and route the call to any of the 100 numbers. The selector 'wipers' always start in their resting 'home' position. The first digit moves the selector vertically up to the corresponding level and then the second digit moves the wipers around the contacts of that level. The figure indicates the connections made when the caller dials 89.

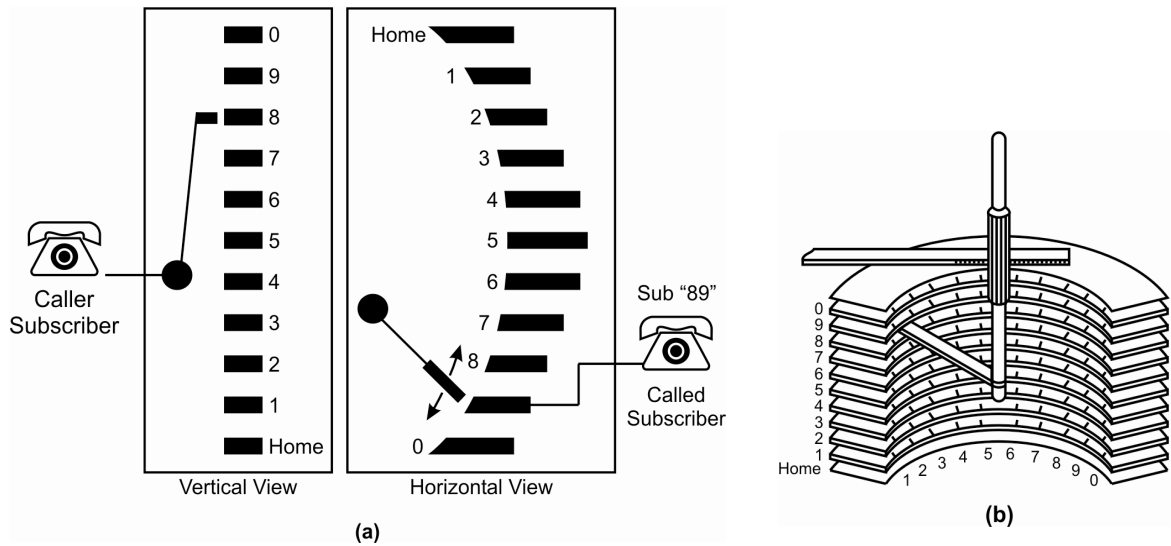


Fig.7.13: a) Vertical and horizontal view; and b) construction of Two-Motion Selector

Call routing through Strowger (step-by-step) switching system

The schematic of Strowger Switching network is shown in Fig. 7.14. It consists of three main stages. The first stage is **line finder** or **selector hunters**, which is a

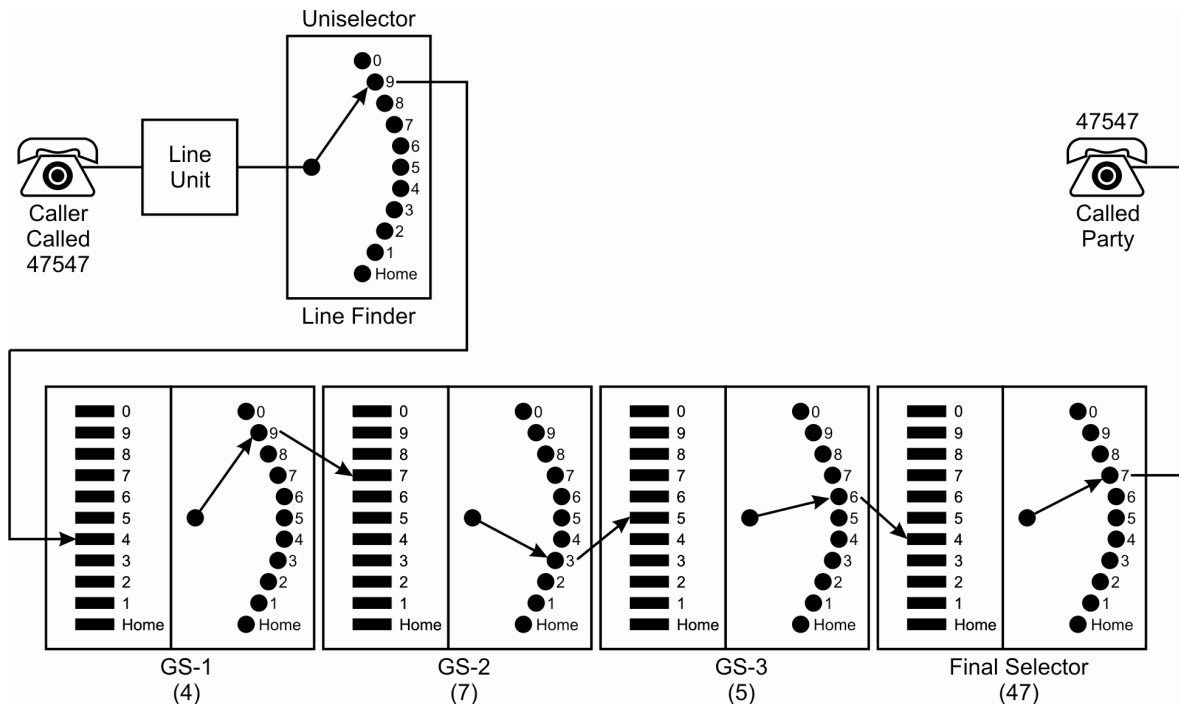


Fig. 7.14: Call routing in Strowger Automatic Exchange

uniselector switch. When the calling subscriber lifts the hand set from the cradle, the line finder searches an empty (free) line through which it connects the line of the calling subscriber to the first selector. This section is known as '**preselector**'. When this line is found, the dial tone is provided to the calling subscriber. There is one line finder for each subscriber, so the number of line finders is decided by the number of subscribers. The second stage, called **switching network**, consist of one or more sets of two-motion selectors known as **group selectors (GS)**.

After the subscriber starts dialling, first selector cuts the dial tone and receives the pulse train corresponding to the first digit. According to this number the wiper assembly moves first vertically to attain the dialled number. Once the vertical wiper selects the appropriate number, the horizontal wiper can access any of the contacts on that particular horizontal plate. The contacts on this plate are attached to the second group selectors. The wiper sweeps in horizontal plane to detect a free second group selector and establishes the contact. If it fails to find any free GS, an engaged (busy) tone is sent back to the caller subscriber.

Once the contact with the second GS is established, the pulse train corresponding to second digit dialled by the subscriber is received by the second group selector via the first selector and this process continues till the last two digits are dialled. In case of the last two digits, the Final selector (FS) takes over and this two motion selector establishes contact with the called subscriber. If the called subscriber is already on the line then a busy tone is sent to the calling party. If the called phone is free, then a ringing current is sent to that phone, so that it alerts the called subscriber while a ringing tone is sent to the calling subscriber. When the called party picks up the hand set, the ring current and ringing tones are cut and audio communication is established.

In this way, the automatic exchange establishes a physical path between the two subscribers via the pre-selector, group selector and final selector.

You must have guessed it correctly that as the number of digits in telephone number increases, the number of GS increases.

You may attempt one SAQ now.

SAQ 4

*Spend
2 Min.*

What is the number of group selectors required for a Strowger exchange with 8-digit telephone number?

All the manual telephone exchanges were replaced by the Strowger automatic exchanges and this technology was working successfully for many decades. However, the Strowger technology involves several mechanical switches each having moving parts and wiping contacts. Due to this, the maintenance becomes costly and difficult. This led to a new electromechanical technology using comparatively less number of movable parts. This was called a **Cross-Bar exchange**.

The cross-bar exchange equipment consisted of two blocks viz. a **common control unit** and a **connection unit**. The common control unit handled the call processing and switching functions while the connection unit actually established the connection.

But by the time the cross-bar exchanges were popularised, the communication technology transferred rapidly into electronic era. So the cross-bar technology was short-lived and many of the Strowger exchanges were directly replaced by *electronic exchanges*.

In electronic exchanges the transistorised switches took over. Nowadays, the special telecommunication ICs provide digital switch arrays, which can be activated by control signals. In this case, switching is done in two ways:

- Space Division Switching
- Time Division Switching

We shall now discuss in details these two switching methodologies.

7.7.3 Space Division Switching

This switching is also called *analog switching*. This concept has emerged from the primitive electromagnetic switches, which provide a metallic path for the signal flow between the two telephones. Hence, each call has an individual physical path, which remains in existence throughout the duration of the call. It shares the exchange in space domain and hence the name **space division switch**. The schematic of space division switch is shown in Fig. 7.15. It is in the form of a matrix, which has calling lines in horizontal direction (rows) and called lines in vertical direction (columns). The connection between calling line and the intended called line is established by

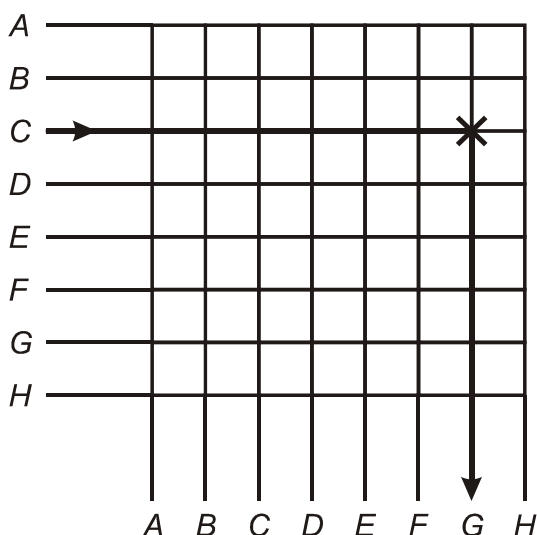


Fig. 7.15: Space division switch

activating the cross point connection between these two lines (In Fig. 7.15 we indicate a call establishment from caller C to subscriber G). These connections can be done by either electromagnetic or electronic switches situated at the cross points. The electromagnetic contacts can be in the form of reed relays which can be either electrically or magnetically latched. The electronic switches can be bipolar junction transistors or field effect transistors. These switches provide a physical path between the calling and called subscribers. However, with advances in technology, it is possible to use computer for data transfer in time division mode. Let us now discuss about the *time division switching*.

7.7.4 Time Division Switching

With digital technology, it is possible to handle the speech traffic with computers. For this purpose, the analog speech signal is first converted into discrete signal by using sampling process. This discrete signal is then modulated using standard PCM technique, which typically converts the signal voltage into an 8 bit code. The frequency range of a speech signal is typically between 300 and 3400 Hz. Hence to satisfy the sampling theorem (Nyquist theorem) normally the signals are sampled at

8 kHz frequency, i.e. the sample arrives at every 125 μ s interval. Now, if each sample is taken by opening the gate for 5.2 μ s, then typically 24 different samples can be handled in the interval of 125 μ s. By reducing the time per sample to 3.9 μ s, we can accommodate 32 samples in 125 μ s. In this way, the same communication path can be utilised for many channels on time division basis. Fig. 7.16 shows the schematic of time division data transmission by digital switching.

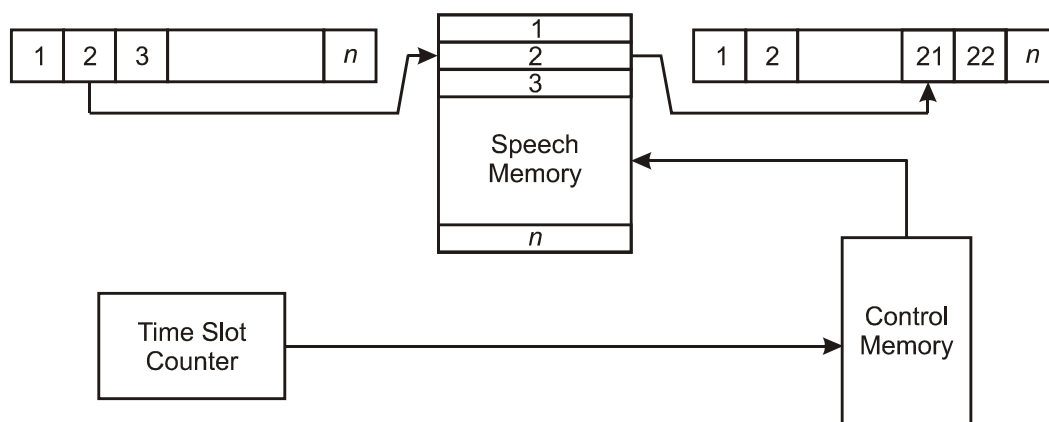


Fig. 7.16: Time division switching

The time division switching scheme shown in the figure is also called *time-slot interchanger*. Time slot interchanging involves moving the data contained in each time slot from the incoming bit stream to the outgoing bit stream, but with different time slots arrangement in accordance with the destination of each time slot e.g. In this example the data from the second incoming time slot is transferred to the 21st time slot of the outgoing bit stream.

Here, within a 125 μ s time period, the incoming calls are serially received and stored in speech memory sequentially. The size of speech memory is equal to number of slots (24 or 32) handled in this time period.

In the outgoing bit stream, the slots are allotted for different (24 or 32) destinations, the addresses of which are controlled by the control memory.

7.8 ELECTRONIC EXCHANGES

Initially the electronic exchanges used the transistor a switching element. But with further technological advancements, integrated circuits took over. The controls in this system are computer controlled. These advanced systems usually comprise four major blocks as shown in Fig. 7.17.

- Connection unit (Interface Unit)
- Switching networks
- Control unit
- Maintenance unit

Let us discuss the functions of these units in brief:

- Connection Unit:** Interface block shown in Fig.7.17 represents the connection unit. It performs the job of providing interface between subscribers and the switching networks. On the one end, the subscriber lines are connected to the connection unit while the other end is connected to the switching network. The connection unit is also connected to the control unit, which monitors all the actions of the exchange. The major functions of the connection unit are:

- providing battery supply to subscriber lines;
- Converting analog speech signals received from callers in to digital form;
- Feeding various tones to subscriber (dial, busy, ringing tone etc.); and
- Play recorded announcements.

b. Switching Network: It is the largest sub-system in size comprising many electronic switches performing routing of call from one interface unit to another, i.e. it carries out the function of establishing connection between calling and called subscriber. This switching facilitates transmission of speech signal between the two points. It is controlled by a control unit.

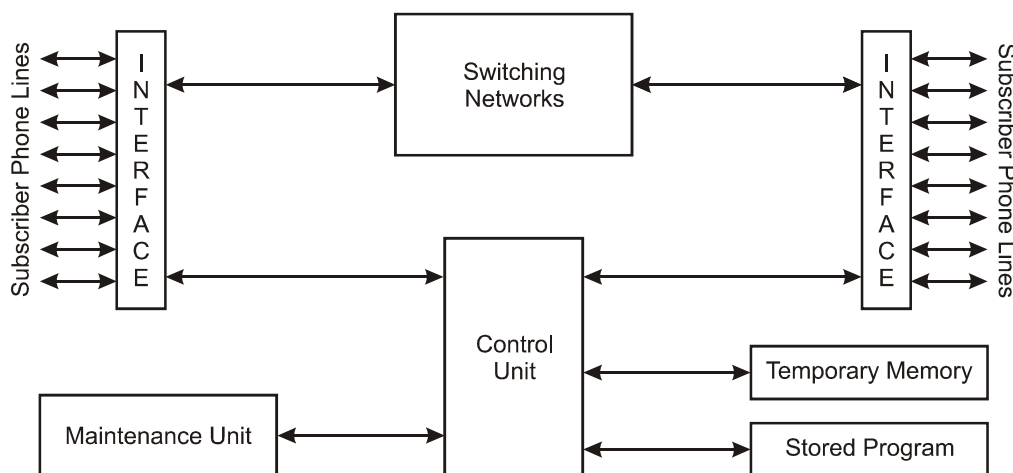


Fig. 7.17: Block Schematic diagram of electronic exchanges

c. Control Unit: The entire system is controlled by this centralised unit. It monitors and controls all the activities of the exchange. It is basically an electronic controller in the form of microprocessor, microcontroller, microcomputer or advance computer. With the advances in technology, the capacity of controllers has improved and along with monitoring the switching operations, it can handle many more other functions that the subscribers demand. Mostly, the control unit operates on the principle of **Stored Programme Control**, commonly termed as SPC. In this scheme, all the switching procedures to set up a call are already stored in the memory files known as programme modules. The processor handling the call accesses this programme and executes it.

The main functions of the control unit are:

- **Call set up:** This includes monitoring of subscriber lines through connecting unit to detect new calls, releasing subscriber lines after call completion, and receiving and transmitting the dialled digits;
- **Making and Breaking the call:** Issuing the instruction to switching networks to connect the calling subscriber to the called subscriber and disconnecting it after call completion;
- **Supervision:** This covers identification of type of subscriber; detection of error condition; confirming the fault and determining its location; and controlling alarm circuits; and
- **Charging:** This covers handling the subscribers accounts; counting the number of metered charges; registering the duration of each call and charging it as per the type of the call i.e. Local, STD , ISD.

Modern electronic exchanges working with computer control can handle many additional features like

- **Push button dialling:** DTMF dialling which increases the speed of number dialling;
 - **Conference call:** A subscriber can set up connection with more than one subscriber simultaneously to conduct a conference on telephone;
 - **Call waiting:** By this facility subscriber can receive another call even if already one call is in progress;
 - **Call transfer:** Subscriber can receive the telephone call on any other pre-decided telephone number automatically;
 - **Announcements:** Any type of pre-recorded announcements can be played on subscriber lines in electronic exchanges. e.g. *this route is busy* or *this number is changed* etc;
 - **Caller ID:** The telephone number of calling subscriber is displayed on the called subscribers telephone;
 - **Detailed billing:** The record of time, date and duration of the calls made by the subscriber can be made available; and
 - **Dynamic Lock:** Telephone can be barred for outgoing calls such as STD, ISD or even local calls by dialling a code.
- d. **Maintenance Unit:** This is also known as Operation and Maintenance Centre (OMC). It performs the important functions of monitoring the working of all the units; watching subscribers meters and store metered information for generating the bills; and management of traffic data.

The Advantages of Electronic Exchanges are

- The electronic exchange allows handling of analog as well as digital signals. This improves speed and quality of data transmission as well as allows facilities like Internet Connectivity.
- It requires less maintenance since there are no mechanical or electromechanical switches.
- Since there are no physical wire links involved, the installation and maintenance of these exchanges is cost and time effective.
- Capacity of these exchanges can be easily expanded since only electronic switching boards need to be added which do not occupy large space.
- Possibility of using time division switching reduces the physical space and hardware involved.

Activity

Visit the nearest telephone exchange in your neighbourhood and find out the type of exchange and its subscriber handling capacity.

With the development of digital communication, many more features are being offered by the telephone service providers. These are called Integrated Services Digital Network (ISDN). In the following section, we shall take a brief account of the important ISDN services.

7.9 ISDN

In initial stages of telephony, the analog signals were transmitted and hence the traffic was restricted mostly to voice signals along with limited text and picture transfer. As you know, the signals generated by the equipment handling picture signals are mostly in digital form. They had to be converted into analog form and transmitted to the called subscriber. Again at the receiving end, these analog signals had to be converted

into digital form, in order to display them. In analog telephony, the bandwidth of the signal channel is only 4 kHz. Hence the large information generated by picture documents takes long time to transfer.

With the emergence of digital technology, the signal in digital form can be transmitted as such. Further, the bandwidths of digital communication channels are very large like typically 64 kilobytes per second (kbps). This enhances data handling capacity of the communication channel and large quantities of data can be transferred at faster rate. Due to the digital communication, it is possible to transmit the computer based data along with voice and picture signals. Fast transmission rate allows even the transmission of images in continuous video mode, which can be used for teleconferencing purposes.

In India, Bharat Sanchar Nigam Limited (BSNL) provides many Integrated Services on Digital Network (ISDN) facilities like:

- Normal telephone and fax (Group-III);
- Digital telephone with facility to identify calling subscriber number;
- Group-IV fax;
- Data transmission at 64 kbps;
- Video conferencing at 128 kbps;
- Video conferencing at 384 kbps (with 3 ISDN lines); and
- ATM (Asynchronous Transfer Mode).

ISDN allows data transmission across the world using end-to-end digital connectivity. In this system, voice and data are carried by **bearer channels (B channels)** occupying bandwidth of 64 kbps. Obviously, all the components involved in data transmission like transmission media (metal wire or optical fibre), switches etc. need to be capable of handling this data transmission rate.

A **data channel (D channel)** is used for handling the signalling. It works at 16 kbps or 64 kbps and carries the control signals necessary for transmission process and in general total functioning of the network.

Typical ISDN connection comprises two *B* channels of 64 kbps each and one 16 kbps *D* channel for a total of 144 kbps. For users with greater capacity requirement, the channel structure is 23 *B* channels plus one 64 kbps *D* channel for a total of 1536 kbps. ISDN subscriber can establish two simultaneous independent calls (except when the equipment at subscriber end does not occupy both *B* channels for single call itself) using a single pair of telephone wires. In analog mode, it is possible to have a single call at a time. The equipment for ISDN are connected to the telephone network via an interface called *S/T interface*. It allows multiple access of terminating equipment, so that many types of equipment can be connected to the same subscriber line.

The terminal equipments include:

- **Normal telephone set** for voice transmission;
- **G-III facsimile machine** that allows transmission of printed pages, including graphics, drawings, pictures, handwritten texts etc. The Group-III fax machines can also be used with conventional analog telephone lines. Working of fax machine is described in Appendix B;
- **G-IV facsimile machine** works only in ISDN mode. This uses digital technology and has faster transmission rate than G-III machines;
- **Digital Telephone set** consists of a keypad for number dialling. The dialling can be either in pulse mode or tone mode. These phones have many additional features like number display panel, speaker mode, call charging display, etc;

- **Data transmission card** allows telephone network based computer data transmission and effectively establishes computer-to-computer communication; and
- **Video conferencing equipment** consisting of digital camera and mike for video and audio signal transmission and video monitor and speaker for signal reception. Video conferencing requires both B channels for establishing the connection, since the data size involved in this case is very large. Hence it is not possible to provide the multiple equipment access when videoconferencing is in progress.

You must have noticed that the journey of telecommunication era, started by Graham Bell, has reached to the turning point by the development of very modern electronic exchanges. But the main developments till date have been in switching equipment only. The external network (exchange to subscriber's premises) has remained mostly unchanged. It means that this connection is through cables and open wires only. Recently the latest technology like WLL (Wireless in Local Loop), GSM (Global Service for Mobile communication) has made it possible to eliminate the cables and open wires from the outdoor plant. By using these advanced technologies, it is possible to establish telephone connections between moving subscribers in cars, trains or even Airplanes. This also avoids the bandwidth limitation posed by the wire connection at the subscriber end (last mile problem) and facilitates fast data transfer.

The complicated cable connections in the external networks which limit the bandwidth of data transmission is known as *last mile problem* in the telephone network terminology.

In the following section we shall take a brief review of how mobile telephone works.

7.10 CELLULAR MOBILE TELEPHONY

In the last few years, cellular mobile phones have changed our life completely. These phones allow us to access any other telephone in the world without having any physical wire connection between the phone and the exchange. Hence we can use them while we are moving. These phones use wireless techniques where radio frequency is used to establish the connection. Naturally in this case the switching mechanism is completely different from the one used for fixed line (or land line) telephones discussed so far.

In mobile telephony, a region of telephone service is divided geographically into small cells as shown in Fig. 7.18.

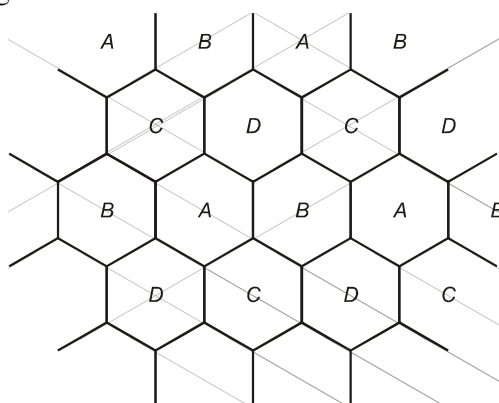


Fig. 7.18: Cell structure for mobile communication working with four sets of frequencies

Though we have shown the cells in hexagonal shape, in reality they may be of irregular shapes (and also sizes depending on number of mobile phone users in a given region). Each cell has a base station, which acts as a local exchange for the mobile phones. Every base station has a low power transmitter and a computer based control system which carries on the functions like call setting/breaking, handing over the call to neighbouring cell when the mobile phone user crosses from one cell region to other cell region. The transmitter power is kept very low so that the signal attenuates fast as it reaches beyond the boundary of the cell and does not disturb the

functioning of the neighbouring cell. Naturally, each cell is operated at different frequency than its neighbouring cells in order to avoid any interference between the phones in two cells. However since the signal diminishes fast, it is possible to reuse the same frequency beyond the first neighbouring cell as shown in Fig. 7.18, where four frequency sets *A*, *B*, *C* and *D* are used to cover the entire area of mobile service divided in small cells.

The frequency bands to operate the mobile service are allotted to different service providers by the Regulatory Authority of the Government. This is called spectrum allocation.

Let us assume that a 25 MHz band between 824 MHz and 849 MHz is allocated for cellular mobile communication in a certain country. Then this band is shared by all the mobile service providers in that country. Let us assume that there are two service providers. Then each one of them would get a 12.5 MHz band.

The cellular mobile telephony commonly uses a narrow band frequency modulation, which can be of about 25 kHz bandwidth. Hence each channel in mobile communication dedicated for each conversation would be 25 kHz wide. To separate the two consecutive channels, normally 5 kHz are added in the channel bandwidth and each channel is allotted a 30 kHz band.

Now with 30 kHz spacing we can fit 416 channels in 12.5 MHz band. However if about 21 channels are reserved for control signalling then 395 channels will be available for mobile users. If we have to operate four sets of frequencies (*A*, *B*, *C* and *D*) in the cells, then each set would contain about 100 channels, i.e. each cell can handle about 100 conversations at a time. Of course it is not necessary to always divide the total number of channels equally. If certain cells have more number of mobile users (may be due to clustering to many offices in one part of the town) then these channels can be allocated unequally as per the traffic in those cells.

Each mobile base station contains a radio transmitter-receiver (transceiver), antenna, and a controller. The base station keeps track of mobile handsets active in its cell area by receiving the signals sent by these phones at periodic intervals (7 to 10 sec). The levels of these signals are monitored and the information is communicated to the Mobile Switching Centre (MSC) controlling the total system comprising all base stations. The MSC keeps track of positions of the mobile telephone subscribers and co-ordinates the function of handing the calls over from one base station to another as the subscriber moves across the cells. Fig. 7.19 shows a mobile handset

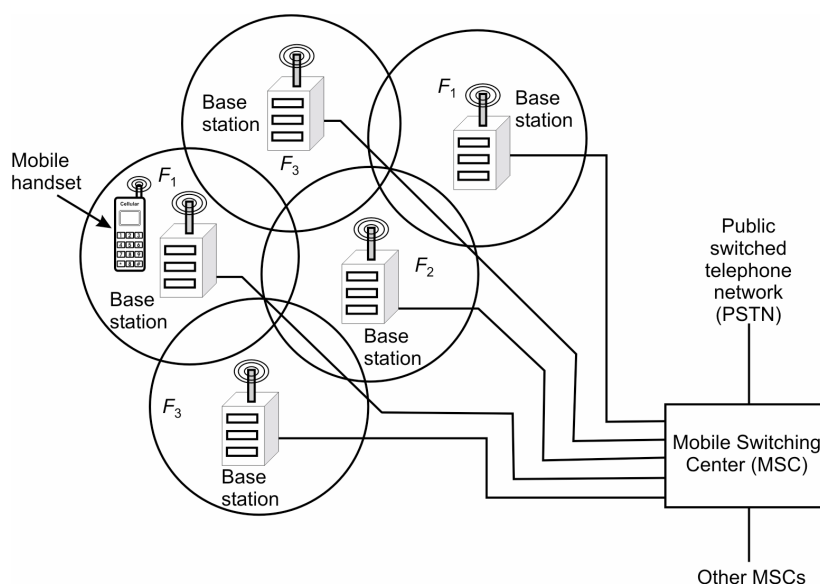


Fig. 7.19: Call handling in cellular mobile system

communicating with a base station which then relays the signal to MSC. The MSC completes the call to another MSC or to the public switched telephone network (PSTN).

The system controlling all the call handling is powered by very sophisticated software systems about which you will be learning later.

The mobile phone handset consists of a keypad to choose a number to dial; an LCD display to inform the user about the operation of the mobile phone and a radio frequency transmitter receiver with antenna to communicate with the outside world. Each mobile phone is powered by a battery which can be recharged from time to time. Most significant part of the mobile phone handset is its controller. This controller remembers the telephone number assigned to that handset, sends signals for setting up a call, receives the information about the frequency channel allotted for conversation from the base station and keeps other information like number directory, track of missed call etc. Mobile phones employ the full duplex system with different frequencies used for transmitting and receiving.

A schematic block diagram of a typical mobile phone is shown in Fig. 7.20.

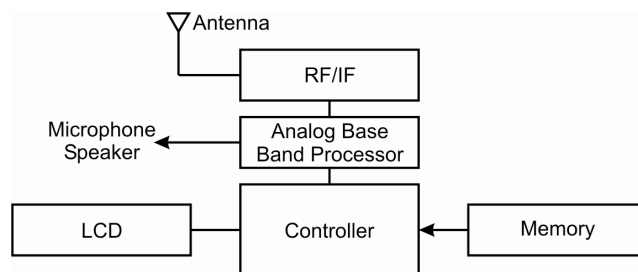


Fig. 7.20: Schematic block diagram of mobile phone

When a call is to be made, the control circuit in the mobile phone sends a signal to the nearest base station. The base station looks up for a free frequency channel and sends the information back to the mobile phone and blocks this channel for the communication. Further, the base station contacts MSC which, in turn, contacts the concerned exchange where the called number is subscribed and the call is set up. Now the ringing tone is given on the mobile phone. Once the call is established, the voice signal is modulated on the allotted frequency channel and the conversation continues. The base station always keeps track of the signal level coming from the mobile phone. If the caller is moving and the base station finds that the signal level is going down, other neighbouring cell base stations pick-up the signal strength. The cell nearest to the caller then takes up the call under the control of MSC by allotting a free frequency channel from that particular cell. This information is sent to the control unit of the mobile handset, so that it starts trans-receiving on this new frequency channel. This handing over of calls takes about few milliseconds and is completely transparent to the mobile phone user. However, if there is any high speed digital data transfer (like file transfer) being done then the handing over affects the quality of transfer. Various multiplexing techniques used in mobile communication will be discussed in the next unit.

Let us now summarize the points you learnt in this unit.

7.11 SUMMARY

- By using telephone instrument and network, two way voice communications is possible.
- Earlier there was point-to-point communication, later it developed to multi-point.

- Automatic exchanges reduced manual dependency on operators and a new era of automatic switching started.
- Invention of electronic switching reduced a large amount of maintenance cost and increased the efficiency to maximum extent.
- Electronic exchanges provided additional facilities on telephone line, which led to real revolution in telecommunication field.
- Telecommunication, which was limited to voice communication has now become a medium for text and image transmission, video streaming, Internet access etc. via ISDN.
- The telecommunication field is going through another revolution of technology of wireless leading mobile communication.

7.12 TERMINAL QUESTIONS

Spend 15 Minutes

1. How has the technological development helped in realising a single common network for handling all telecommunication services?
2. Explain the working of a two-motion-selector as a final selector in a Strowger switching system.
3. Is it possible to generate the effect of rotary dialling using a touch-tone pad phone? How can it be implemented?

7.13 SOLUTIONS AND ANSWERS

Self Assessment Questions

1. The ringer should always be connected to a battery supply, whenever the telephone is not in use. The hook switch disconnects the supply from the telephone, when the receiver or handset is kept on the hook. Hence the ringer has to be placed before this switch, constantly connected to the supply line.
2. 10 Million. (This is a theoretical limit for numbers from 0000000 to 9999999, however, in practice many of the numbers are reserved for special purposes and are not allotted to the subscribers.)
3. Number of links required will be $n(n-1)/2$.
4. 6 Group Selectors.

Terminal Questions

1. Refer to the text.
2. Refer to Sec. 7.7.2.
3. Yes. The number dialled on a touch-tone pad can be stored in the temporary memory of the phone and corresponding pulses are sent out serially with appropriate break to distinguish between the consecutive numbers. The pulses can be generated by using digital/analog pulse generator circuits or ICs.

Reference Material:

1. *Telecommunication Systems Engineering* by Freeman, Roger; (III Edition) (Wiley-Interscience Publication)
2. *Telecommunication Switching Systems and Networks* by Viswanathan, Thiagrajan; (I Edition) (Prentic-Hall of India)

APPENDIX A: TERMINOLOGY IN TELECOMMUNICATION

In telecommunication world, some specific terms are used for the components and activities of the system. Here we are listing these items.

1. **Local Call:** A call made with in the city.
2. **STD:** Subscriber Trunk Dialling, a call made from one city to other within a country.
3. **ISD:** International Subscriber Dialling, a call made to another country.
4. **Calling subscriber:** One who initiates a telephone call.
5. **Called subscriber:** One who receives the telephone call.
6. **Dial:** A rotary disk on the telephone instrument to dial the telephone number.
7. **Dial tone:** This is the tone you hear when you lift the hand set. It is supplied by the exchange, indicating to the calling subscriber that the exchange is ready to accept dialling.
8. **Ring tone:** This is sound of bell in telephone instrument to indicate that there is incoming call on the phone.
9. **Ring back tone:** This tone is returned to the calling subscriber from the exchange when the bell of the calling subscriber's telephone is ringing.
10. **Busy tone:** This tone is returned to the calling subscriber from the exchange when the called subscriber is already busy.

The tones described above are collectively called **signalling tones** in telephony.

APPENDIX B: FACSIMILE MACHINE

It is essentially another form of text communication service. It is faster than any other communication in delivery of documents. It can transport any type of document such as printed material, hand written copy, picture or diagram. Operations are simple. It does not require skilled operator. It is cheaper than Telex or other modes. Nowadays there is possibility of scanning the documents, storing them in computer memory and sending them as an attachment to the e-mail. However, in case of fax, there is no need of the separate scanner and the computer with Internet access. Therefore, facsimile is most popular data transfer technique in business community.

The object in Facsimile telegraphy (FAX) is to reproduce (remotely) a faithful copy of the original document.

Fax machine consists of a Scanner and a Printer. In a Fax machine the document is electrically scanned line by line in sub-millimetre dimensions to read the alternating black / white contents on it, so as to reproduce them through a printer. In the Fax, these two units are not co-located but linked by a telecom medium. A document to be sent via fax from machine *A* to machine *B* is scanned in machine *A* and printed on machine *B*. Similarly, a fax sent by machine *B* is scanned there and sent over to *A* to be printed. The efficiency of delivery depends upon the functioning of each unit viz. scanner, the telecom link and the printer.

Scanner: There are several methods of scanning, all of which work on the principle of focusing a powerful light source on the document and then evaluating progressively the reflection of each small element (0.26 mm × 0.125 mm) on a line left to right and repeating it line by line till the end.

The scan dimensions are eight scans per millimetre horizontally and 3.85 scans per millimetre vertically. In term of inches, each inch is scanned horizontally 203 times and vertically 98 times. Thus every horizontal line of an A-4 size sheet is scanned 1728 times and each spot is interpreted as black or white element. Earlier Fax machines were very slow. Now the latest machines are faster which can simultaneously scan all 1728 elements comprising the entire line in a single stroke by an array of 1728 sensors called CCD (Charged Couple Device).

Printer: Thermal printing is most common, using thermal sensitive paper rolls. Costlier machines incorporate laser printers, using normal papers.

Classification of Fax: As per CCITT (Consulting Committee of International Telephone & Telegraph) recommendations, Faxes are introduced in G-1 to G-4 Series, which are based on speed, capability & technology.

G - 1 : Uses double sideband modulation without any measures for bandwidth compression. In G - 1, a white element is indicated by a tone of 1300 Hz and black element by 2100 Hz. It is slow; hence it is no longer in use.

G - 2 : Uses bandwidth compression technique (amplitude - phase modulation) to reduce transmission time to approximately 3 minutes. In G - 2 the frequency for white and black elements remains at 2100 Hz.

G - 3 : Makes use of special encoding schemes which allow compression techniques exploiting redundancy to condense information control. Document transmission is possible approximately in one minute.

G - 4 : This machine works only in ISDN mode. This uses digital technology and has faster transmission rate than G-3 machines. The time for transmission of A4 sheet is 15 seconds.