LECTURE NOTES
ON

TELECOMMUNICATION SWITCHING THEORY AND APPLICATIONS

B.Tech V semester
(IARE-R16)

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ELECTRONICS AND COMMUNICATION ENGINEERING

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Syllabus

UNIT-I	INTRODUCTION	Classes: 10
Introduction: Evolution of telecommunications, simple telephone communication, manual switching system, major telecommunication networks, strowger switching system, crossbar switching; Electronic Space Division Switching: Stored program control, centralized SPC, distributed SPC, enhanced services, two stage networks, three stage network n-stage networks.		
UNIT-II	**TIME DIVISION SWITCHING**	**Classes: 09**
Time Division Switching: Time multiplexed space switching, time multiplexed time switching, combination Switching, three stage combination switching, n-stage combination switching; Traffic Engineering: Network traffic load and parameters, grade of servicVe and blocking probability, modeling switching systems, incoming traffic and service time characterization, blocking models and loss estimates, delay systems.		
UNIT-III	**DATA NETWORKS**	**Classes: 08**
Data networks: Block diagram, features, working of EPABX systems, data transmission in PSTNs, data rates in PSTNs, modems, switching techniques for data transmission, circuit switching, store and forward switching data communication architecture. ISO-OSI reference model, link to link layers, physical layer, data link layer, network layer, end to end layers, transport layer, session layer, presentation layer, Satellite based data networks, LAN, metropolitan area network, fiber optic networks, and data network standards.		
UNIT-IV	**TELEPHONE NETWORKS**	**Classes: 08**
Telephone Networks: Subscriber loop systems, switching hierarchy and routing, transmission plan, transmission systems, numbering plan, charging plan, signaling techniques, in channel signaling, common channel signaling, cellular mobile telephony.		
UNIT-V	**INTEGRATED SERVICES DIGITAL NETWORKS**	**Classes: 10**
Integrated Services Digital Networks: Motivation for ISDN, new services, network and protocol architecture, transmission channels, user network interface, signaling, numbering and addressing, service characterization, interworking, ISDN standards, broadband ISDN ,voice data Integration.		

UNIT-I
TELECOMMUNICATION SWITCHING SYSTEM AND NETWORK

Introduction

The world has undergone many changes since the evolution of man. For instance, the exchange of information was initially in the form of signs and sounds. This transitioned to the language and script form with advanced inventions. The communication from one place to another which called for distance between individuals was carried through letters; sent by pigeons and between two groups through drum beats or semaphores. Men used to travel long distances to pass on messages.

Today‗s world is more an age of communication. The advancement of communication techniques has increased the speed with which the transfer of information takes place. This development has not been an easy process. At the onset of the invention of communication systems, the invention and usage of telephony was the most important one. The way the telephone systems evolved from a basic system into an essential multi- purpose friendly gadget today, leaves one and all astonished knowing the innovations made out of the meagre resources available in those days.

Telecommunications

The exchange of information between two or many individuals is called **Communication**. The word **tele** is a Greek word which means distance. Hence, **Telecommunication** means the exchange of information between two distant places.

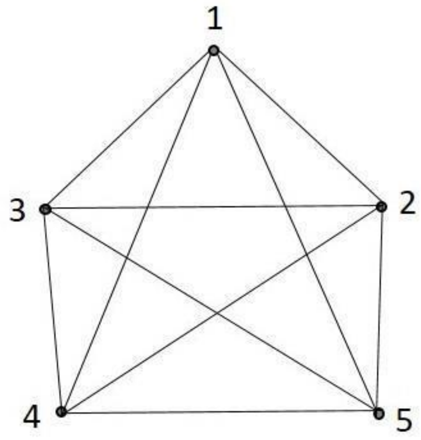
Telecommunications represent the transfer of information, from an entity at one place to an entity at another place, whereas the information can be in the form of data, voice or symbol. The entities can be human beings, computers, facsimile machines, telegraphy machines, phones or so on. In telephone conversation, the one who initiates the call is referred to as the **Calling Subscriber** and the one for whom the call is destined is the **Called Subscriber**. In other cases of information transfer, the communicating entities are known as **Source** and **Destination**, respectively.

In March 1876, Alexander Graham Bell invented and demonstrated his telephone set and the possibility of long distance voice communication. He demonstrated the **point-to-point** communication, in which a calling subscriber chooses the appropriate link to establish connection with the called subscriber. This system also requires some mode of **Signalling** to alert the called subscriber about the incoming call and a signal to indicate the calling subscriber, when the called subscriber is busy on another call.

Need for Switching Exchanges

The point-to-point connection for establishing communication requires the telephone sets to be linked using wires. If the number of telephone sets or the subscribers present is low in number, the type of connection will be a little complex. However, if this number is high or moderate, then the connections will lead to a mess. To understand the complication, let us consider a network of 5 subscribers.

The following illustration shows a point-to-point connection for five subscribers (telephone sets):



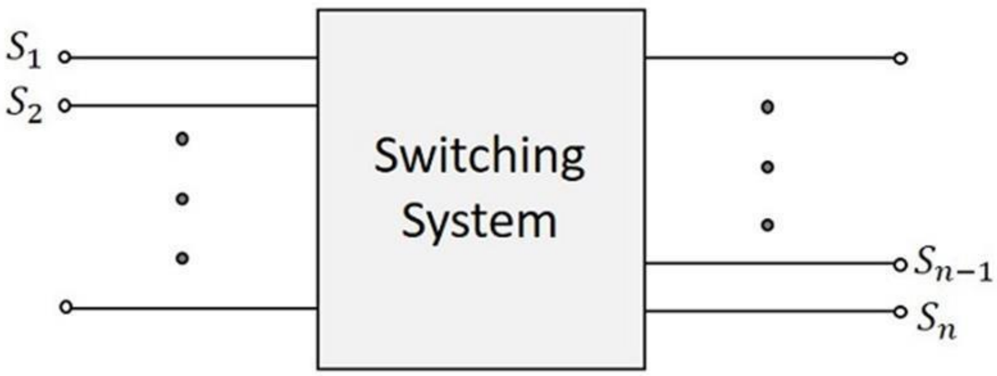
In the point-to-point connection, for **n** entities, we need **n(n-1)/2** links. All these links form a network. Networks with point-to-point links among all the entities are known as **Fully Connected Networks**. The number of links required in a fully connected network becomes very large even with moderate values of **n**.

Hence, a system of switching the networks is needed in-between these subscribers. Alexander Graham Bell recommended the Switching between the subscribers using a switching office that maintains the telephone connections.

Switching Systems

This network connection cannot be simply made with telephone sets and bunch of wires, but a good system is required to make or break a connection. This system is known as the **Switching System** or the **Switching Office** or the **Exchange**. With the introduction of the switching system, the subscribers instead of getting connected directly to one another, are connected to a switching office and then to the required subscriber.

The following figure will help you understand the switching system



With the introduction of switching systems, the need for traditional connections between the subscribers reduced. All the subscribers need to **have a connection with the switching system**, which makes or breaks any connection, requested by the calling subscriber. The switching system, which is also called the **Telephone Exchange**, takes care of establishing the calls. Hence, the total number of such links is equal to the number of subscribers connected to the system.

The early systems required manual operations to establish telephone calls. An operator used to receive a call from the calling subscriber and then connect the call to the called subscriber. Later on, the system was automated.

Telephone Model

The following figure will help you understand the model of telephones in the early stage of its invention.

When you see the telephone in the above figure, the dialer part and the microphone are connected to a stationary wooden plank; and the speaker to listen, was connected by a wire at the side. The top portion of the telephone has two bells connected - these bells ring when there is an incoming call. This is one of the earlier models of the telephone.

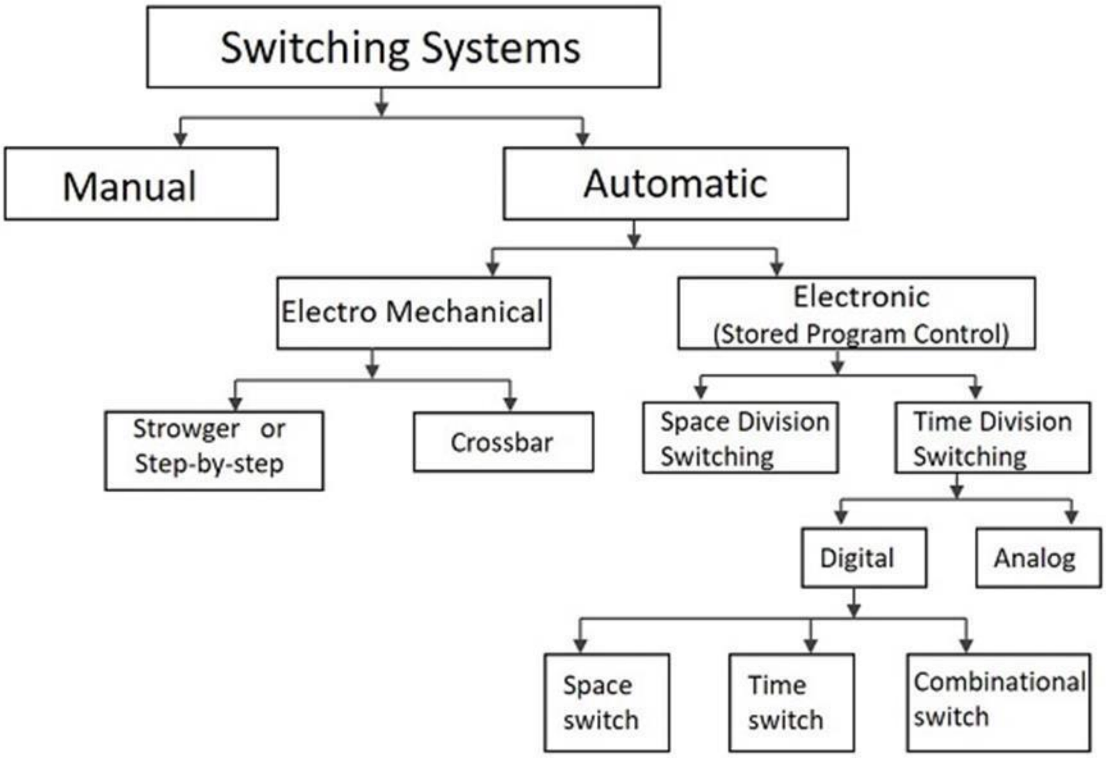
The telephone sets of the calling subscriber and the called subscriber are connected through a switching system or a telephone exchange in order to establish the calls requested.

In the following sections, we will learn about the switching system in detail.

In this chapter, we will understand how the switching systems work. A Switching system can be understood as a collection of switching elements arranged and controlled in such a way as to set up a common path between any two distant points. The introduction of switching systems reduced the complexity of wiring and made the telephony hassle-free.

Classification of Switching Systems

In the early stages of telecommunication systems, the process and stages of switching, played an important to make or break connections. At the initial stages, the switching systems were operated manually. These systems were later automated. The following flowchart shows how the switching systems were classified.



The switching systems in the early stages were operated **manually**. The connections were made by the operators at the telephone exchanges in order to establish a connection. To minimize the disadvantages of manual operation, automatic switching systems were introduced.

The **Automatic** switching systems are classified as the following:

- **Electromechanical Switching Systems** - Here, mechanical switches are electrically operated.
- **Electronic Switching Systems** – Here, the usage of electronic components such as diodes, transistors and ICs are used for the switching purposes.

Electromechanical Switching Systems

The Electromechanical switching systems are a combination of mechanical and electrical switching types. The electrical circuits and the mechanical relays are deployed in them. The Electromechanical switching systems are further classified into the following.

Step-by-step

The **Step-by-step** switching system is also called the **Strowger** switching system after its inventor A B Strowger. The control functions in a Strowger system are performed by circuits associated with the switching elements in the system.

Crossbar

The **Crossbar** switching systems have hard-wired control subsystems which use relays and latches. These subsystems have limited capability and it is virtually impossible to modify them to provide additional functionalities.

ElectronicSwitching Systems

The Electronic Switching systems are operated with the help of a processor or a computer which control the switching timings. The instructions are programmed and stored on a processor or computer that control the operations. This method of storing the programs on a processor or computer is called the **Stored Program Control** (SPC) technology. New facilities can be added to a **SPC** system by changing the control program.

The switching scheme used by the electronic switching systems may be either **Space Division Switching** or **Time Division Switching**. In space division switching, a dedicated path is established between the calling and the called subscribers for the entire duration of the call. In time division switching, sampled values of speech signals are transferred at fixed intervals.

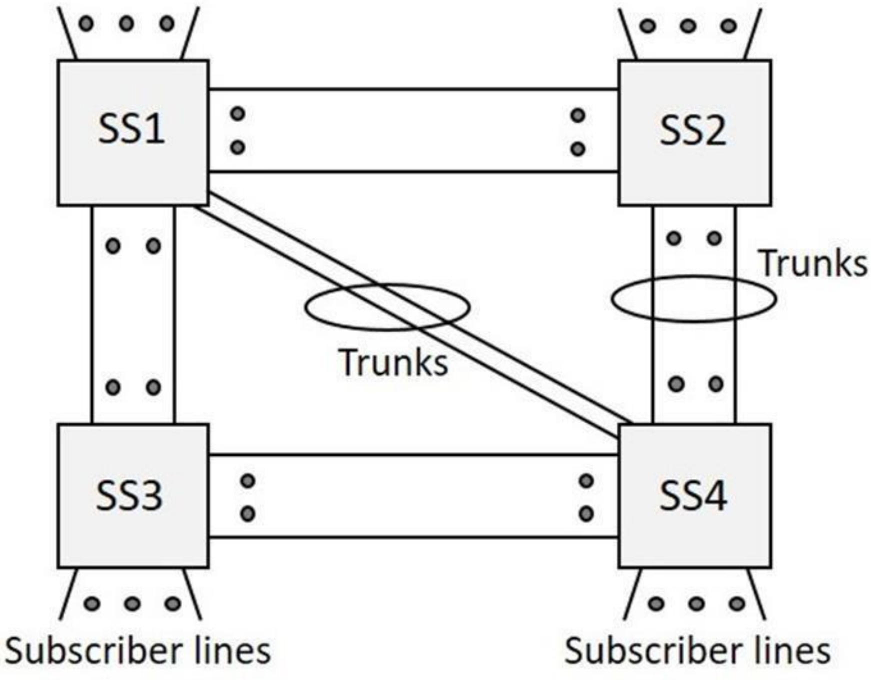
The time division switching may be analog or digital. In analog switching, the sampled voltage levels are transmitted as they are. However, in binary switching, they are binary coded and transmitted. If the coded values are transferred during the same time interval from input to output, the technique is called **Space Switching**. If the values are stored and transferred to the output at a time interval, the technique is called **Time Switching**. A time division digital switch may also be designed by using a combination of space and time switching techniques.

Telecommunication Network

A Telecommunication network is a group of systems that establishes a distant call. The switching systems are part of a telecommunication network.

The switching stations provide connection between different subscribers. Such switching systems can be grouped to form a telecommunication network. The switching systems are connected using lines called the **Trunks**. The lines that run to the Subscriber premises are called the **Subscriber Lines**.

The following figure shows a telecommunication network.

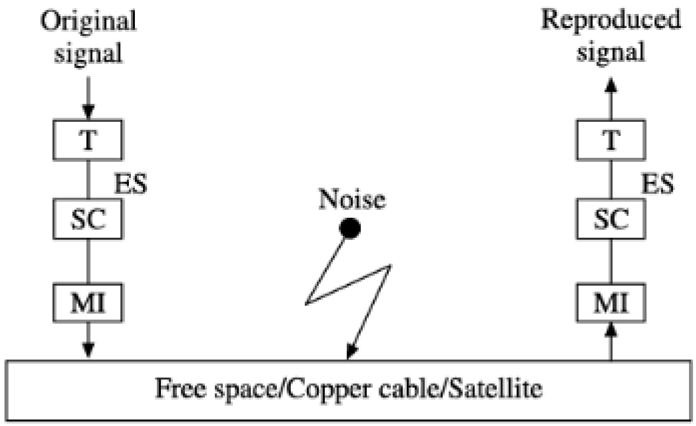


From the early to the later stages of the 20th Century (1900-80), when a person needed to make a distant call, the call was first routed to the operator at the nearest switching center and then the number and location of the called subscriber was noted down. Here, the job of the operator was to establish a call to the remote switching center and then recall the calling subscriber to establish the connection. This system of making calls was called the **Trunk call** system.

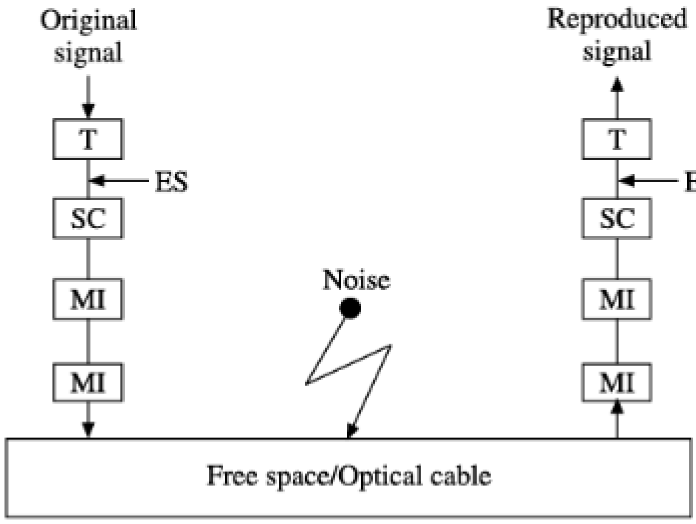
For example, a person at Hyderabad can book a trunk call to Mumbai and wait for the operator to call back when the operator establishes connection through the trunk lines and the switching systems.

Communication Links

A telephone switching network is made tip of switching systems trunks. subscriber lines and telephone instruments. Trunks and subscriber lilies are essentially communication links which carry information signals from one point to another. There are basically only two forms of communication links electrical and optical. In the former information is conveyed by means of electrical energy and in the latter. by means of light energy.



(a) An electrical communication link



(b) An optical communication link

EOC = electrical to optical converter; ES = electrical signal; MI = medium interface
OEC = optical to electrical converter; SC = signal conditioner; T = transducer

The information to be conveyed is not always in the form of electrical or optical signals. For example. human speech signals are essentially sound waves. As a consequence conversion from one form of energy to another may be required before information signals can be carried by communication links. Transducers perform this energy conversion they are available for converting sound. light or heat energy to electrical energy and vice versa. But at the present state of technological development, there are no known transducers that can directly convert sound energy into light energy. As a result, one is required to go through a two-step process of converting sound into electrical energy first and then into optical energy to be able to use the optical communication links. In other words, today's optical sources accept only electrical signals as input, and the optical detectors produce only electrical signals as output. Hence, at the transmitting end. The original signals are first converted into electrical signals by using known transducers. Then electrical to optical converters (EOC). i.e. optical sources. are used to obtain optical signals. At the receiving end. optical to electrical converters (OEC). i.e. optical detectors. are used first and then the transducer to reproduce the original signal.

Service Specific Networks

With the concept of switched connections for telephony taking its firm roots. the idea of offering oilier non voice services using switches and switched networks caught the attention of tele communication specialists in the first half of 20th century. Different services nican different types of end equipments at the customer premises. e.g telex, teleprinter and facsimile machines. The signal characteristics of such end equipments vary widely. For example. the electrical characteristics. i.e. voltage. current and power levels and bandwidth of die signal. of a teleprinter are completely different from those of a telephone. In addition. Signalling requirements for different types of end equipments also differ significantly. Such wide variations in electrical characteristics and signalling requirements have led to the development of different service.
specific telecommunication networks that operate independently.

Examples are:

1. Telegraph networks

2. Telex networks

3. Telephone networks

4. Data networks

Of the service specific networks mentioned above, telephone networks and data networks form the contents of this text. The most stupendous telecommunications network in existence today is the public switched telephone network (PSTN) or sometimes known as plain old telephone system (POTS). In standards documents. PSTN is often referred to as general switched telephone network (GSTN). In this text. the popular nomenclature. PSTN is used. PSTN has evolved over a period of 120 years since the beginning of telephony in 1879. There are over a billion telephones in the world connected via land lines (copper cables) to the network. PSTN is highly demanding in its requirements for reliability and availability.

SIMPLE TELEPHONE COMMUNICATION

In the simplest form of a telephone circuit. there is a one-way communication involving two entities. one receiving (listening) and the other transmitting (talking). This form of one-way communication shown in shown as simplex communication. The microphone and the earphone are the transducer

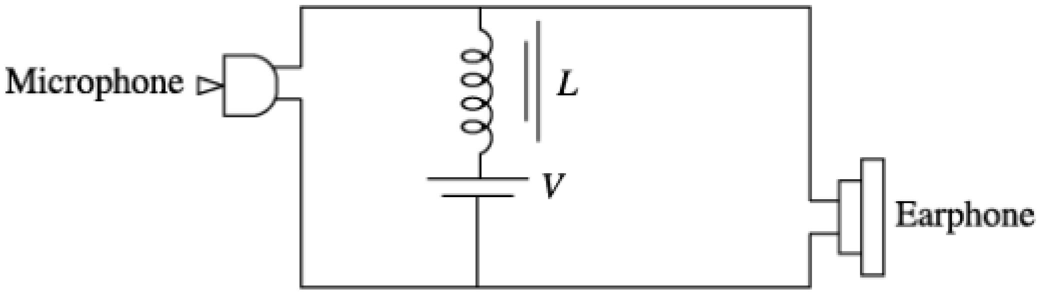
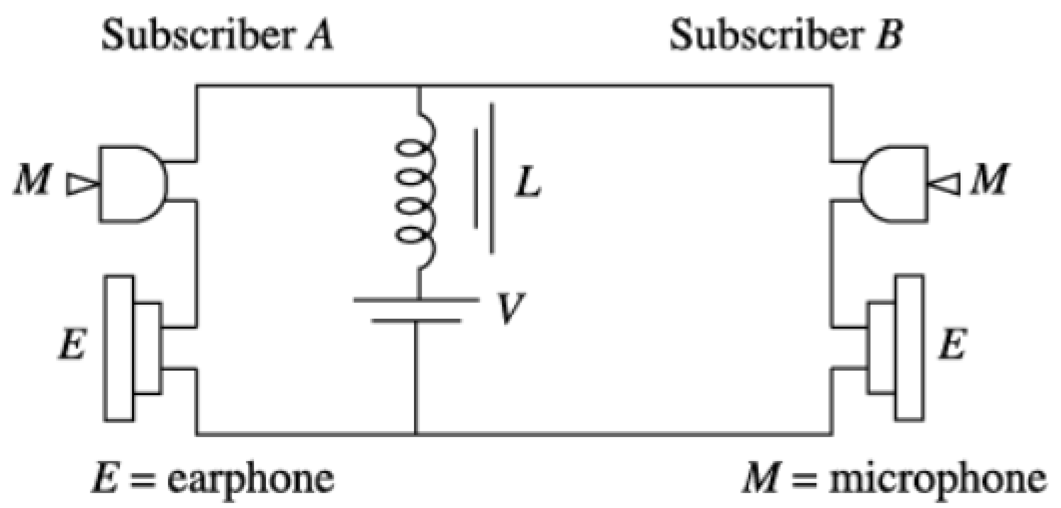


Figure 1.6 A Simplex telephone circuit.

elements of the telephone communication system. Microphone converts speech signals into electrical signals and the earphone converts electrical signals into audio signals. Most commonly used microphone is a carbon microphone. Carbon microphones do not produce high fidelity signals but give out strong electrical signals at acceptable quality levels for telephone conversation.

In a normal telephone communication system. Information is transferred both ways. An entity is capable of both receiving and sending although these do not take place simultaneously. An entity is either receiving or sending at any instant of time. When one entity is transmitting the other is receiving and vice versa. Such a form of communication where the information transfer takes place both ways but not simultaneously is known as half-duplex communication. If the information transfer takes place in both directions simultaneously. Then it is called full-duplex communication. may be modified to achieve half-duplex communication by the introduction of a transmitter and receiver at both ends of the circuit as shown in Figure. In this circuit. the speech of A is heard by B as well as in A's own earphone. This audio signal. heard at the generating end. is called side tone. A certain amount of side tone is useful. or even essential. Human speech and hearing system is a feedback system in which the volume of speech is automatically adjusted. based on the side tone heard by the ear. If no side tone is present. a person tends to shout. and if too much of side tone is present. there is a tendency to reduce the speech to a very low level. In the circuit of Figure .The entire speech intensity is heard as side tone. which is not desirable.



Basics of a Switching System

In this section, we will learn about the different components and terms used in switching systems.

Inlets and Outlets

The set of input circuits of an exchange are called **Inlets** and the set of output circuits are called the **Outlets**. The primary function of a switching system is to establish an electrical path between a given inlet-outlet pair.

Usually, **N** indicates the inlets and the outlets are indicated by **M**. So, a switching network has **N** inlets and **M** outlets.

Switching Matrix

The hardware used to establish connection between inlets and outlets is called the **Switching Matrix** or the **Switching Network**. This switching network is the group of connections formed in the process of connecting inlets and outlets. Hence, it is different from the telecommunication network mentioned above.

Types of Connections

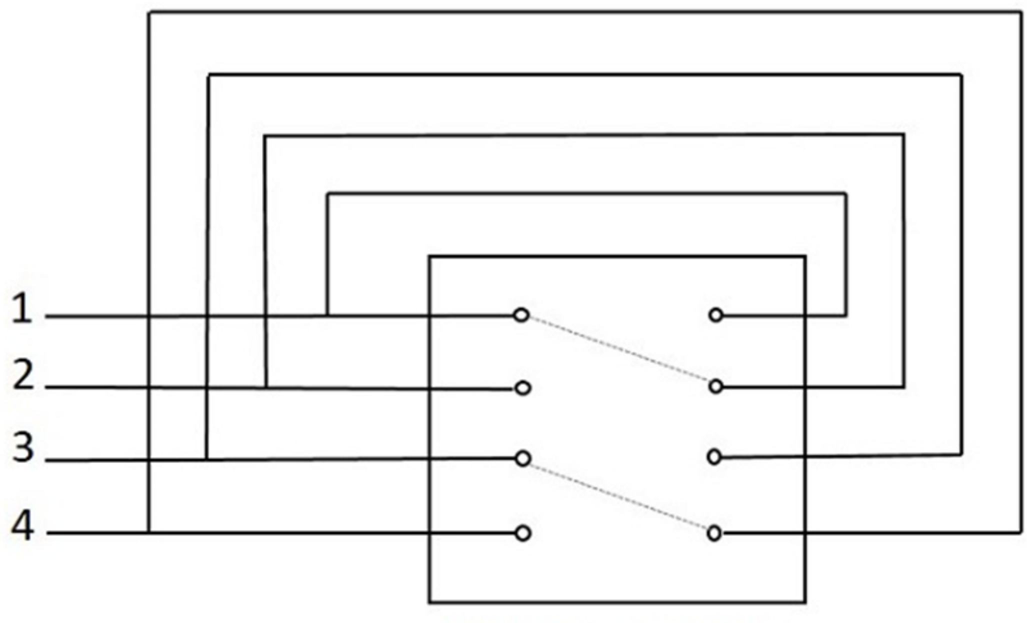
There are four types of connections that can be established in a telecommunication network. The connections are as follows:

- Local call connection between two subscribers in the system.
- Outgoing call connection between a subscriber and an outgoing trunk.
- Incoming call connection between an incoming trunk and a local subscriber.
- Transit call connection between an incoming trunk and an outgoing trunk.

Folded Network

When the number of inlets is equal to the number of outlets for a switching network, such a network is called the **Symmetric Network**, which means N=M. A network where the outlets are connected to the inlets, is called the **Folded Network**.

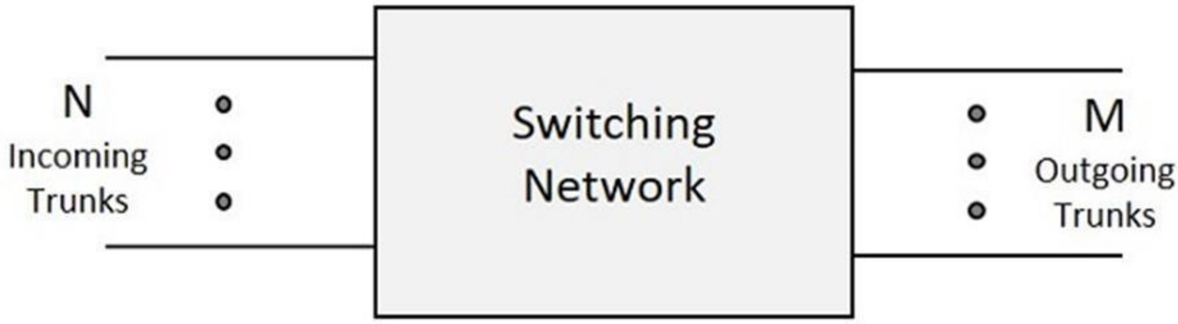
In a Folded Network, the N number of inlets which come as outlets are again folded back to the inlets. Nevertheless, the switching network provides connections to the inlets and outlets as per the requirement. The following figure will help you understand how the Switching Network works.



As one connection can be given to one line per time, only N/2 connections are established for N inlets of a folded network. Such a network can be called as **Non-blocking network**. In a non-blocking network, as long as the called subscriber is free, a calling subscriber will be able to establish a connection to the called subscriber.

In the above figure, only 4 subscribers were considered - where line 1 is busy with line 2 and line 3 is busy with line 4. While the call is in progress, there used to be no chance for making another call and hence, only a single connection was made. Hence for N inlets, only N/2 lines are connected.

At times, it might happen that the inlet and outlet connections are continuously used to make Transit calls through trunk lines only, but not among the local subscribers. The inlet and outlet connections if used in an **Inter-exchange transmission** such that the exchange does not support connection between local subscribers, then it is called the **Transit Exchange**. A switching network of such kind is called the **Non-folded network**. This is shown in the following figure:



Blocking Network

If there are no switching paths free in the network, the call requested will be denied, where the subscriber is said to be **blocked** and the network is called the **Blocking Network**. In a blocking network, the number of simultaneous switching paths is less than the maximum number of simultaneous conversations that can take place. The probability that a user may get blocked is called the **Blocking Probability**. A good design should ensure low blocking probability.

Traffc

The product of the calling rate and the average holding time is defined as the Traffic Intensity. The continuous sixty-minute period during which the traffic intensity is high is the Busy Hour. When the traffic exceeds the limit to which the switching system is designed, a subscriber experiences blocking.

Erlag

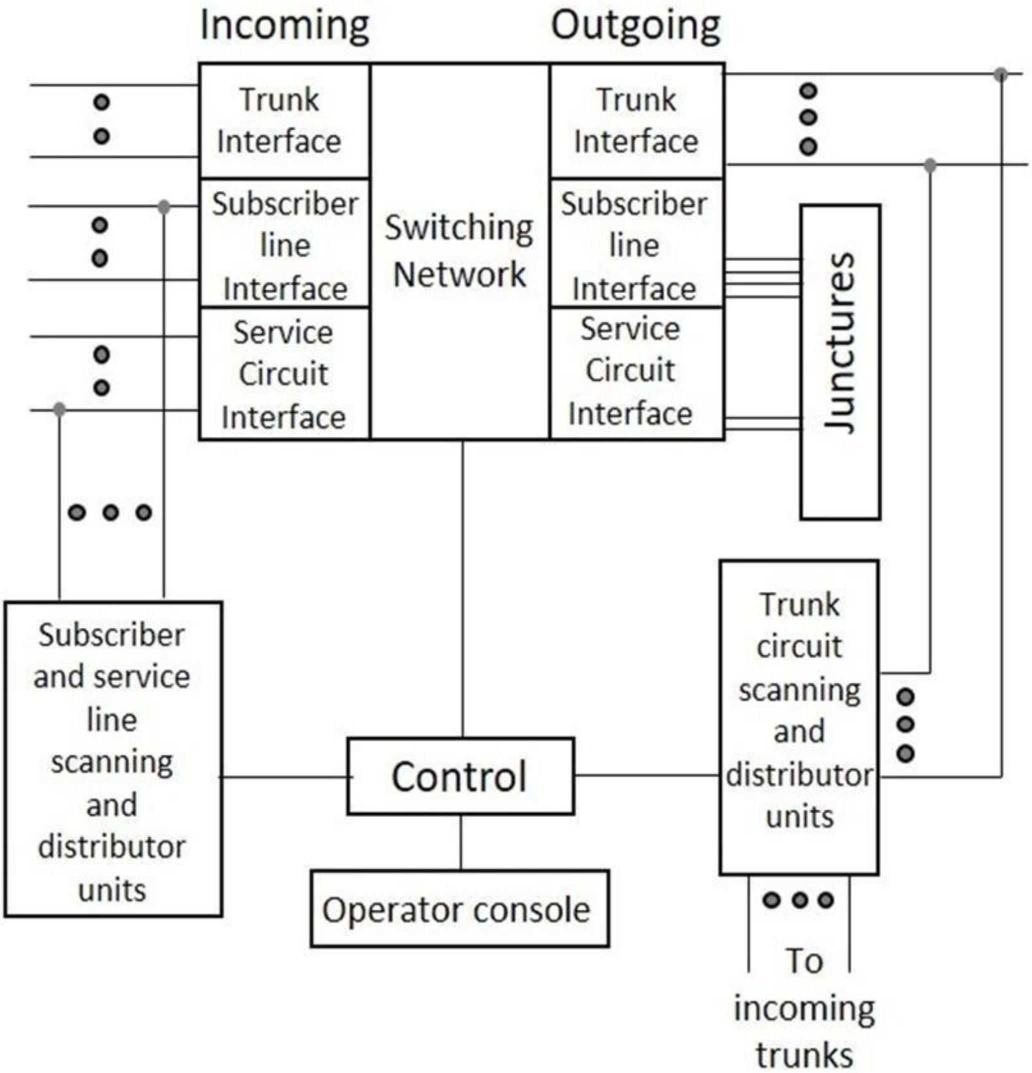
The traffic in a telecommunication network is measured by an internationally accepted unit of traffic intensity known as **Erlang** (E). A switching resource is said to carry one Erlang of traffic if it is continuously occupied through a given period of observation.

Elements of a Switching System

In this chapter, we will discuss the elements of a switching system. Though there are different kinds of switching systems from manual to automatic, a few basic elements play an essential role for the functioning of a switching system. Along with the switching network, there are different sub systems such as control sub system, signaling system, trunk and subscriber line interfaces, distributor units, operator console, juncture circuits, essential for the operation of the whole switching system.

Switching System

In this section, we will understand the structure of the switching system. We will also understand how the different elements work in it. The block diagram of the switching system given below show the essential elements of a switching system.



The diagram shown above contains different blocks of the switching system. The blocks are discussed below.

Switching Network

It provides the switching paths between the called subscribers and the calling subscribers.

Control Subsystem

This is the critical part of the switching system, which actively establishes the switching paths, by identifying the inlet and outlet lines and interpreting the signaling information received on these lines.

This control subsystem, controls the making and breaking of the connection by sensing the signal transfer on the lines. The control sub system sends out signaling information to the subscriber and other exchanges connected to the outgoing trunks.

Signaling

The signaling formats and requirements for the subscriber, the trunks and the sub systems differ significantly. Accordingly, a switching system provides for three different forms of signaling:

- Subscriber loop signaling
- Interexchange signaling
- Intraexchange or register signaling

A switching system is composed of elements that perform switching, control and signaling functions.

Trunk Interface

The Trunk lines used for connections between the switching systems, are terminated at this port. The Trunk interface is the point where the trunk lines are connected to the system.

Subscriber Line Interface

The Subscriber lines used for connections between the subscribers and the switching systems are terminated at this port. The subscriber line interface is the point where the lines from the subscribers are connected to the system.

Line Scanning Unit

The line scanning unit senses and obtains the signaling information from the respective lines. The information obtained from these lines are given to the control sub system to identify the inlets and outlets.

Distributor Units

The distributor units are used for distributing or sending out the signaling information on the respective lines. The distribution of information through the trunk lines, is done through the distribution units

Operator Console

The operator console permits interaction with the switching system for maintenance and administrative purposes.

Service Circuit Interface

The service circuit interface provides interaction between circuits for maintenance and testing purposes.

Junctures

The Junctures is a junction that provides a folded connection for the local subscribers and the service circuits. If the called subscriber and the calling subscriber both are local, then the folded connection helps in making the connection to a local call, whereas the trunk lines will not be in use.

Direct and Indirect

The switching systems are of the following two types:

- the direct control switching system
- the indirect control switching system

Direct Control Switching System

The Switching systems where the control sub systems form an integral part of the network are called the Direct Control Switching systems. For example, the Strowger switching system.

Indirect Control Switching System

The Switching system in which the control sub system is present outside the switching network is called the **Indirect Control** Switching system or the **Common Control** Switching system or the **Register Control** switching system. The examples of this system include Crossbar switching system, Electronic switching system or Stored Program Control method of switching systems.

Strowger Switching System

In this chapter, we will discuss how the Strowger Switching system works. The first ever automatic telephone switching was developed by Almon B Strowger. As the operator at the Manual telephone exchange was the wife of his competitor and was diverting all the business, Strowger thought of developing a switching system, which does not require an operator. This led to the invention of the automatic switching system developed by Strowger.

The **Strowger Switching system** is also called the **step-by-step** switching system as the connections are established in a step-by-step manner.

Automatic Switching System

The Manual Switching system requires an operator who after receiving a request, places a call. Here, the operator is the sole in-charge for establishing or releasing the connections. The privacy of the calls and the details of the called and the calling subscribers are at stake.

Overcoming the disadvantages of Manual Switching systems, the Automatic Switching systems come with the following advantages:

- Language barriers will not affect the request for connection.
- Higher degree of privacy is maintained.
- Faster establishment and release of calls is done.
- Number of calls made in a given period can be increased.
- Calls can be made irrespective of the load on the system or the time of the day.

Let us now throw some light on how a call is made and how dialing is done without the help of an operator.

Dialing

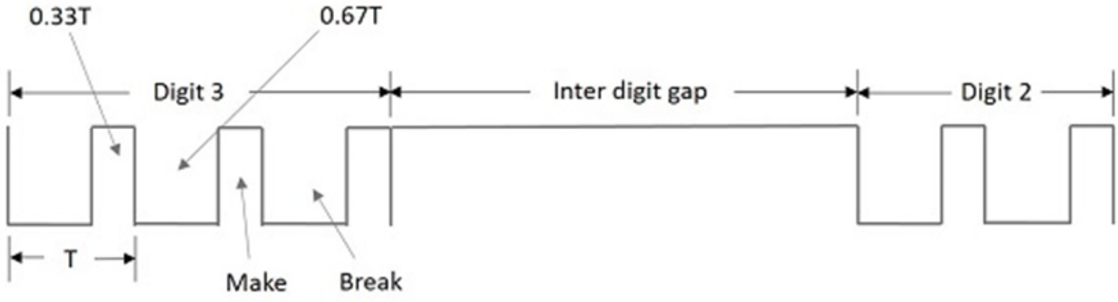
Unlike in Manual Switching system, an automatic switching system requires a formal numbering plan or addressing scheme to identify the subscribers. Numbering plan is where a number identifies a subscriber, is more widely used than the addressing scheme in which a subscriber is identified by the alpha numerical strings. So, there needs to be a mechanism to transmit the identity of the called subscriber to the exchange.

This mechanism should be present in the telephone set, in order to connect the call automatically to the required subscriber. The methods prevalent for this purpose are **Pulse Dialing** and **Multi Frequency Dialing**. Of them, the Pulse dialing is the most commonly used form of dialing till date.

Pulse Dialing

As the name implies, the digits that are used to identify the subscribers are represented by a train of pulses. The number of pulses in a train is equal to the digit value it represents except in the case of zero, which is represented by 10 pulses. Successive digits in a number are represented by a series of pulse trains. These pulses have equal number of time intervals and the number of pulses produced will be according to the number dialed.

Two successive trains are distinguished from one another by a pause in between them, known as the **Inter-digit gap**. The pulses are generated by alternately breaking and making the loop circuit between the subscriber and the exchange. An example pulse train is shown in the following figure.



The above figure shows the pulsating pattern. The pulse rate is usually 10 pulses per second with a 10 percent of tolerance. The gap between the digits, which is called the Inter-digit gap is at least 200ms.

The pulse dialing pattern in recent times employs the duty ratio (ratio between the pulse width and the time period of the waveform) of the pulse as 33 percent nominally and there exists an upper limit for the inter-digit gap.

Rotary Dial Telephone

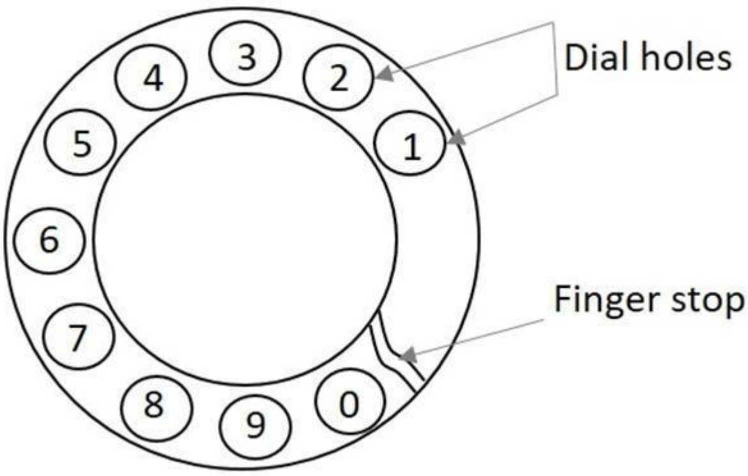
In this section, we will learn about what the Rotary Dial Telephone is and how it works. To start with, we will discuss the drawbacks that were prevalent before the invention of the Rotary Dial Telephone.

The pulse dialing technique is where there is making and breaking of the subscriber loops. This might disturb and affect the performance of speaker, microphone and bell contained in the telephone. In addition, the dialing timings should not affect the timing of the pulse train as this will lead to the dialing of a wrong number.

The Rotary Dial Telephone came into existence to solve the problems prevailing then. The microphone and the loudspeaker are combined and placed in the receiver set. The set has a finger plate the arrangement of which makes the dialing time appropriate. The below figure shows how a rotary dial looks like. The pulse dialing technique is where there is making and breaking of the subscriber loops. This might disturb and affect the performance of speaker, microphone and bell contained in the telephone. In addition, the dialing timings should not affect the timing of the pulse train as this will lead to the dialing of a wrong number. As the name implies, the digits that are used to identify the subscribers are represented by a train of pulses. The number of pulses in a train is equal to the digit value it represents except in the case of zero, which is represented by 10 pulses. Successive digits in a number are represented by a series of pulse trains. These pulses have equal number of time intervals and the number of pulses produced will be according to the number dialed.

The dial is operated by placing the finger in the hole appropriate to the digit to be dialed. Now, drawing the fingerplate round in the clockwise direction to the finger stop position and letting the dial free by withdrawing the finger, makes a number dialed. The fingerplate and the associated mechanism now return to the rest position under the influence of a spring. The dial is ready for the next number.

The dial pulses are produced during the return travel of the fingerplate, thus eliminating the human element in pulse timings. The following figure shows the dial holes and finger stop.



A rotary dial phone uses the following for implementing pulse dialing:

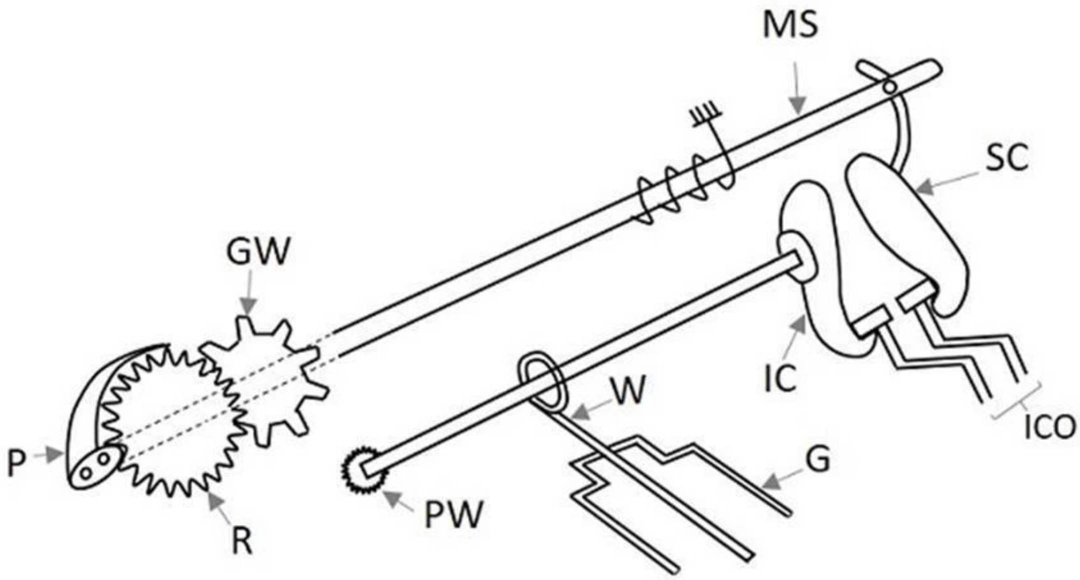
- Finger plate and spring
- Shaft, gear and Pinion wheel

- Pawl and ratchet mechanism
- Impulsing cam and suppressor cam or a trigger mechanism

- Impulsing contact
- Centrifugal governor and worm gear
- Transmitter, Receiver and bell by-pass circuits

Internal Mechanism

The cam mechanism or trigger mechanism helps in dialing. This mechanism is used in operating the Impulsing contact. Let us consider the operation of the rotary dial telephone using the cam mechanism. The following figure will help you understand the internal mechanism.



G = Governo
GW = Gear Wheel
IC = Impulsing Cam
ICO = Impulsing Contacts
MS = Main Shaft

P = Pawl
PW = Pinion Wheel
R = Rachet
SC = Suppressor Cam
W =Worm gear

The suppressor cam helps in keeping the Impulsing cam away from the Impulsing contacts. When the rotary dial is in rest position, then the Impulsing contacts are away from the Impulsing cam. When a number is dialed, by placing the finger in the dial hole, which means the dial is displaced from its position, then the Impulsing contacts come near the Impulsing cam. This rotation of the finger plate, causes the rotation of the Main shaft.

As the dial is rotated in clockwise direction, the pawl slips over the ratchet durin g this clockwise rotation. The ratchet, gear wheel, pinion wheel and the governor are all stationary during the clockwise movement of the dial. When the dial returns, the pawl engages and rotates the ratchet.

All the gear wheel, pinion wheel, the governor rotate, and the uniformity in the speed of the rotation are maintained by the governor. The Impulsing cam, which is attached to a pinion shaft, now breaks and makes the Impulsing contacts that in turn causes the pulses in the circuit. The shape of the Impulsing cam is such that the break and make periods are in the ratio of 2:1. When the dial is about to reach the rest position, the suppressor cam again, moves the Impulsing contacts away from the Impulsing cam. This action of getting back to the rest position and waiting for the other number to be dialed creates a gap called the Inter-digit gap, the timing of which is independent of the pause that may occur between two successive digits, due to human dialing habit. This gap is also provided prior to the dialing of the first digit through a small change in the suppressor cam design.

The Pulse generated through this mechanism is then transmitted to the switching systems where the connection to the dialed number is established. The procedure of switching systems is discussed in a subsequent chapter. Meanwhile, let us have an idea on the signaling tones that are used to indicate the condition of the subscribers.

SignalingTones

In this section, we will understand what are signaling tones and how these work. As the manual exchanges were replaced, the operator who used to communicate the calling subscribers regarding the situation of the called subscribers, needed to be replaced with different tones indicating different situations.

Consider the following five subscriber related signaling functions that are to be performed by the operator:

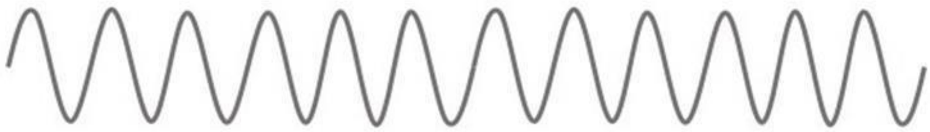
- Respond to the calling subscriber that system is ready to receive the identification of the called party.
- Inform the calling subscriber that the call is being established.
- Ring the bell of the called party.
- Inform the calling subscriber, if the called party is busy.
- Inform the calling subscriber, if the called party line is unobtainable for some reason.

The function 2 is not signaled in the Strowger switching system. The signaling function 1 is fulfilled by sending a dial tone to the calling subscriber.

Dial Tone

The dial tone is the signaling tone, which indicates that the exchange is ready to accept the dialed digits from the subscriber. The number should be dialed only when this signal is heard. Otherwise, the digits dialed before this signal will not be considered. This will lead to the dialing of a wrong number.

The dial tone is generally a 33 Hz or 50 Hz or 400 Hz continuous tone as shown below.

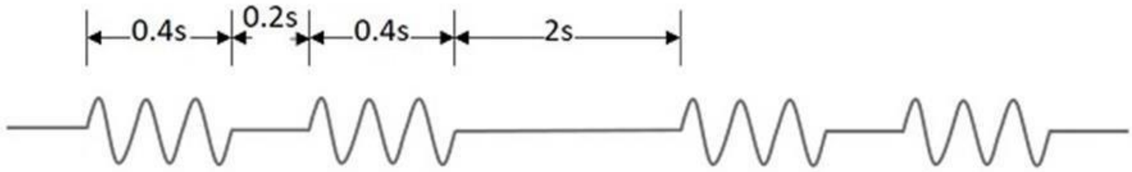


33Hz or 50Hz or 400Hz Continuous signal

Ring Tone

After dialing the number of the called party, when the line of the called party is obtained, the exchange control equipment sends out the ringing current to the telephone set of the called party, which is a familiar double-ring pattern.

Simultaneously, the control equipment sends out a ringing tone to the calling subscriber, which has a pattern similar to that of the ringing current. The two rings double-ring pattern are separated by a time gap of 0.2s and two double-ring patterns by a gap of 2s, as shown in the below figure.



400Hz or 133Hz tone

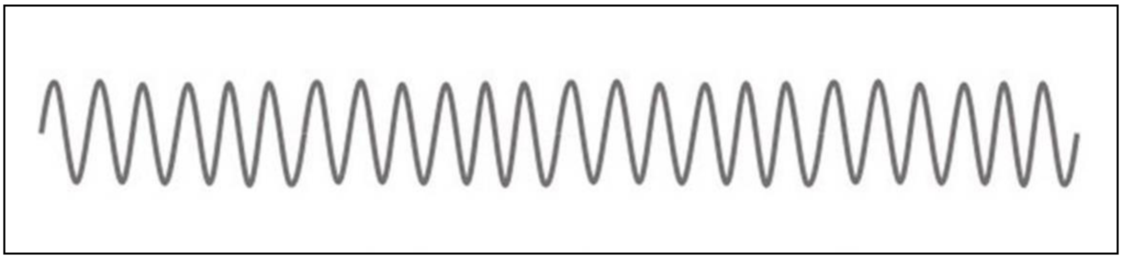
Busy Tone

After dialing the required number, if the called subscriber or the lines at the exchange are not free to place a call, the calling subscriber is sent a busy tone indicating that the lines or the subscriber is busy; this is called a busy tone.

A busty tone of 400Hz signal with silence period in between. The burst and silence durations have the same value of 0.75s or 0.75s.

Number Unobtainable Tone

If the called party is out of order or disconnected or if an error in dialing leads to the selection of a spare line, such a situation is indicated using a continuous 400Hz signal, called as Number Unobtainable tone. The following illustration shows a continuous 400Hz signal.



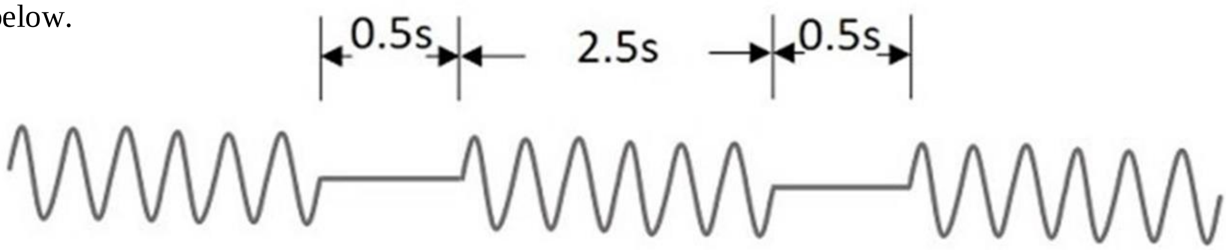
Routing Tone or Call-in-Progress Tone

When a subscriber call is routed through a number of different types of exchanges, one hears different call-in-progress tones as the call progresses through different exchanges. Such a signal is a 400Hz or 800Hz intermittent pattern. This signal has different patterns in different systems.

- In electromechanical systems, it is usually 800Hz with 50 percent duty ratio and 0.5s ON/OFF period.
- In analog electronic exchanges, it is a 400Hz pattern with 0.5s ON period and 2.5s OFF period.
- In digital exchanges, it is 400Hz signal with 0.1s ON/OFF

periods. The signal for routing tone or call-in-progress tone is as shown

below.



In order to overcome the problem of recognizing the difference in these tones for those who are not familiar with telephone signalling and for those who rarely make calls, voice recorded messages were introduced, later on.

Switching Mechanisms

In this chapter, we will discuss the switching mechanisms in Telecommunication Switching Systems and Networks.

In our previous chapters, we discussed the mechanism in the telephone set. Let us now see what happens when this telephone set sends a signal to the switching system. The switching system at the exchange should be able to connect the line automatically to the called subscriber. In the Strowger switching system, there are two types of selectors; these selectors form the building blocks for the switching systems.

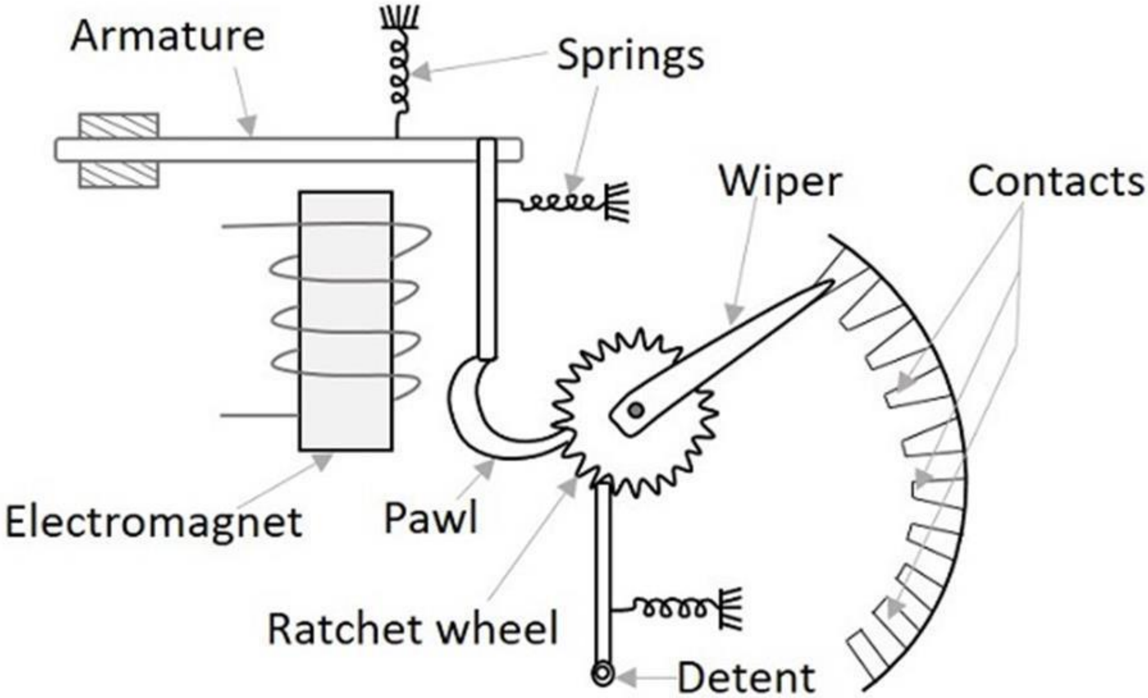
- Uni-selector
- Two-motion selector

Both of these selectors are constructed using electro-mechanical rotary switches. The Uni-selector has a single selector pole and multiple throws to reach the bank of contacts for each number dialed. The two-motion selector has two rotary switches for vertical and horizontal stepping movement, to reach the bank of contacts.

Uni-selector Switching

The Uni-selector switching mechanism consists of an Electromagnet, an Armature with springs, a Pawl, a ratchet wheel with wiper attached and a detent. The wiper is made to move on the bank contacts in clock-wise direction. As the wiper moves in one-direction, the process is called Uni-selector switching. The contacts onto which the wiper moves are called Bank contacts as a number of contacts are placed in this shape of an arc.

The following figure shows the drive mechanism of the Uni-selector Strowger switching system. Both of these selectors are constructed using electro-mechanical rotary switches. The Uni- selector has a single selector pole and multiple throws to reach the bank of contacts for each number dialed. The two-motion selector has two rotary switches for vertical and horizontal stepping movement, to reach the bank of contacts. When a subscriber call is routed through a number of different types of exchanges, one hears different call-in-progress tones as the call progresses through different exchanges. Such a signal is a 400Hz or 800Hz intermittent pattern. This signal has different patterns in different systems. Once the electromagnet gets de-energized, the armature is released and this action moves the pawl upwards, which further moves the ratchet wheel to one position above. Hence, the wiper moves one position below or in clockwise direction, to make a contact. If the electromagnet is energized and de-energized five times, by applying five pulses, the wiper moves by five contacts. Usually three sets (or more) of wipers are placed associated with the banks of Uni-selector, one for each bank. The sets are rigidly mounted to a wiper assembly, which moves whenever the ratchet wheel rotates. The interrupter spring releases the magnet and enables it to make another step.



When the input voltage energizes the Electromagnet, the armature is pulled down towards the magnet. Now as the armature gets attracted towards the electromagnet, the pawl falls one position down the previous one in the ratchet wheel. The detent prevents the movement of the ratchet wheel.

Once the electromagnet gets de-energized, the armature is released and this action moves the pawl upwards, which further moves the ratchet wheel to one position above. Hence, the wiper moves one position below or in clockwise direction, to make a contact. If the electromagnet is energized and de-energized five times, by applying five pulses, the wiper moves by five contacts. Usually three sets (or more) of wipers are placed associated with the banks of Uni-selector, one for each bank. The sets are rigidly mounted to a wiper assembly, which moves whenever the ratchet wheel rotates. The interrupter spring releases the magnet and enables it to make another step.

The following figure shows a practical Uni-selector Strowger switching system.
This signal has different patterns in different systems. Once the electromagnet gets de-energized, the armature is released and this action moves the pawl upwards, which further moves the ratchet wheel to one position above. Hence, the wiper moves one position below or in clockwise direction, to make a contact. If the electromagnet is energized and de-energized five times, by applying five pulses, the wiper moves by five contacts. Usually three sets (or more) of wipers are placed associated with the banks of Uni-selector, one for each bank. The sets are rigidly mounted to a wiper assembly, which moves whenever the ratchet wheel rotates. The interrupter spring releases the magnet and enables it to make another step.

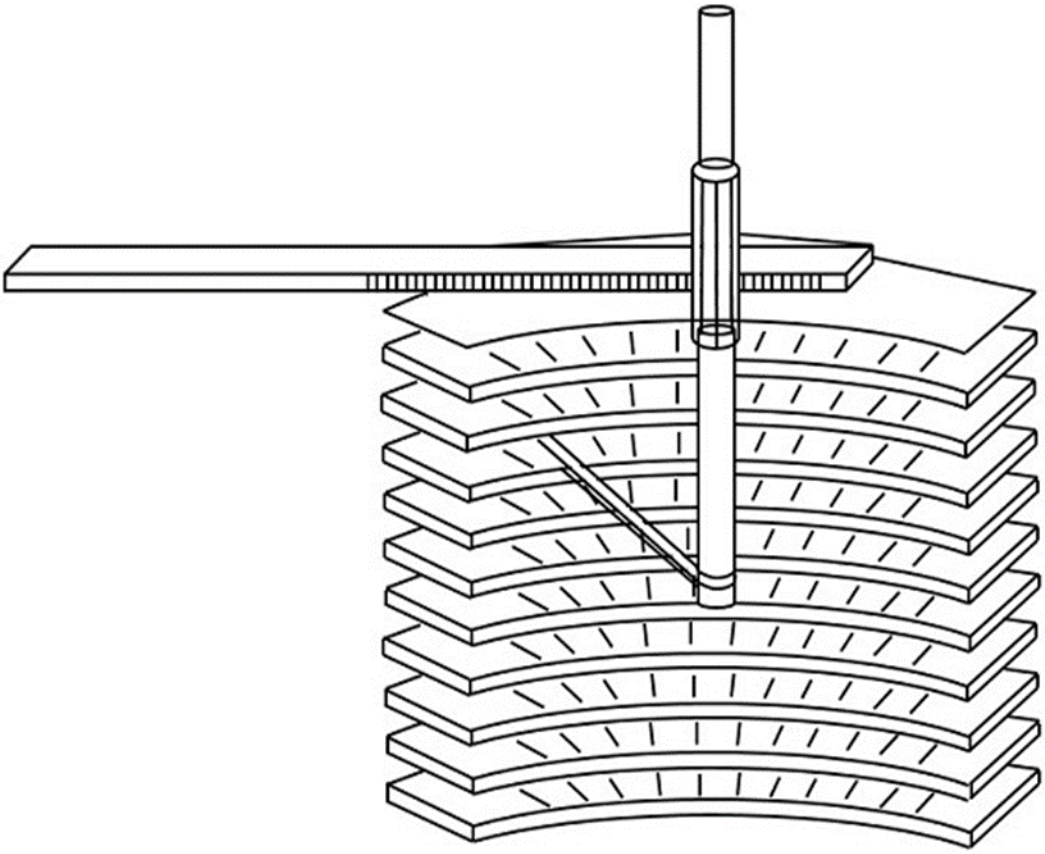
The type of switching mechanism discussed here is known as the **reverse drive type** because, here the ratchet wheel moves when the armature return to its rest position. If it is arranged such that the wheel moves during the forward motion of the armature it is known as the **forward drive type**. The Reverse drive type mechanism is prevalent in uni- selectors and the forward drive type mechanism in the two-motion selectors.

There is an **interrupter contact** associated with the Uni-selector, which is normally closed. When the armature is energized, the interrupter contact opens and allows the movement of armature, which helps the armature return to its rest position after breaking up the armature energizing circuit.

Two-motion Selectors

Unlike in Uni-selector, the motion in these selectors is two-way, vertical and horizontal. An upward movement is made in vertical and horizontal directions; there are no contacts made in the vertical movement. However, bank contacts are made in the horizontal movement. If the two-motion selector has 10 levels, each having 10 contacts, then 100 contacts are accessible, by the vertical and horizontal movement of the two-motion selector switching system.

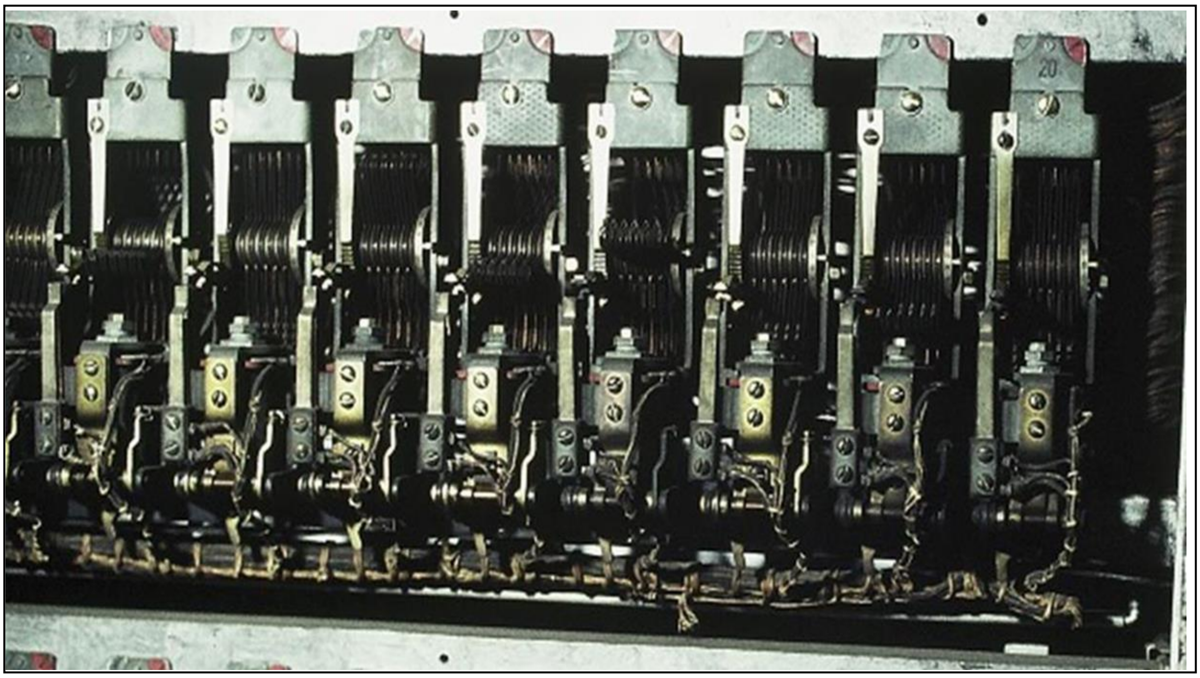
The following figure shows the internal structure of two-motion switching selectors.



- When the first digit is dialed, the pulses energize and de-energize the vertical magnet according to the number dialed, with the help of ratchet and pawl mechanism. This is called as **Vertical Stepping**.

- When the second digit is dialed, the dialing pulses are diverted to horizontal magnet, with the help of a relay where the pulses energize and de-energize the horizontal magnet according to the number dialed, with the help of ratchet and pawl mechanism. This is called **Horizontal Stepping**.

Normally, there are 11 vertical positions and 11 horizontal contacts in each vertical position. The lowest vertical position and the first horizontal contact in each vertical level are **home positions**, and the remaining ones are actual switching positions. Thus, the wiper in a two-motion selector has access to 100 switching contacts. The following figure shows a practical two-motion switching selector.

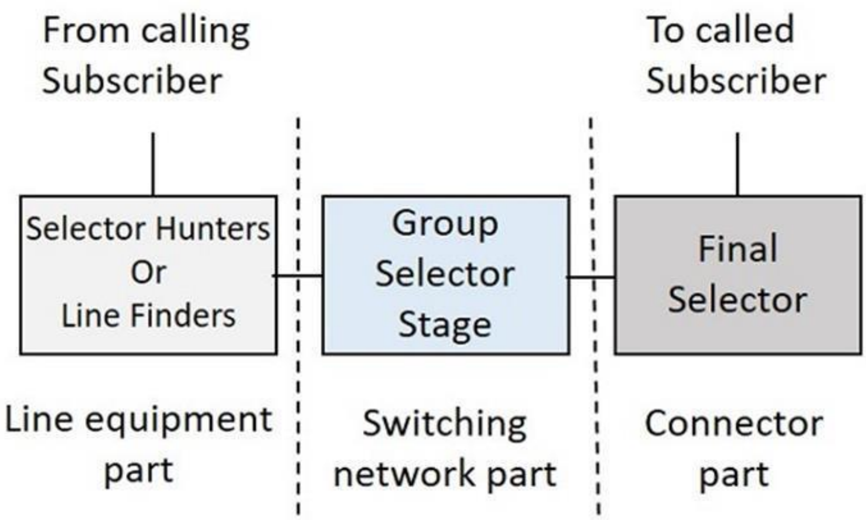


Thus, the wiper assembly establishes the call; after completion of which it comes back to the home position. For this purpose, the rotary magnet is operated by the current and thus the wiper assembly moves through the remaining contacts of the level. A restoring spring forces the wiper assembly to drop vertically and then return horizontally to the home position.

Step-by-Step Switching

The Step-by-step switching system is a very popular and widely-used switching system, which may be constructed using Uni-selectors or two-motion selectors or the combination of both. The wiper present in this switching, steps forward by one contact and then moves forward according to the number of dialed pulses or according to the signaling conditions and hence the name, **step-by-step** switching is given.

A step-by-step switching is also called the **Direct control** system as the relevant signaling tones are sent out to the subscriber by the switching elements or selectors at the appropriate stages of switching. This system has three main stages of configuration. The following figure shows the different stages.For this purpose, the rotary magnet is operated by the current and thus the wiper assembly moves through the remaining contacts of the level. A restoring spring forces the wiper assembly to drop vertically and then return horizontally to the home position.



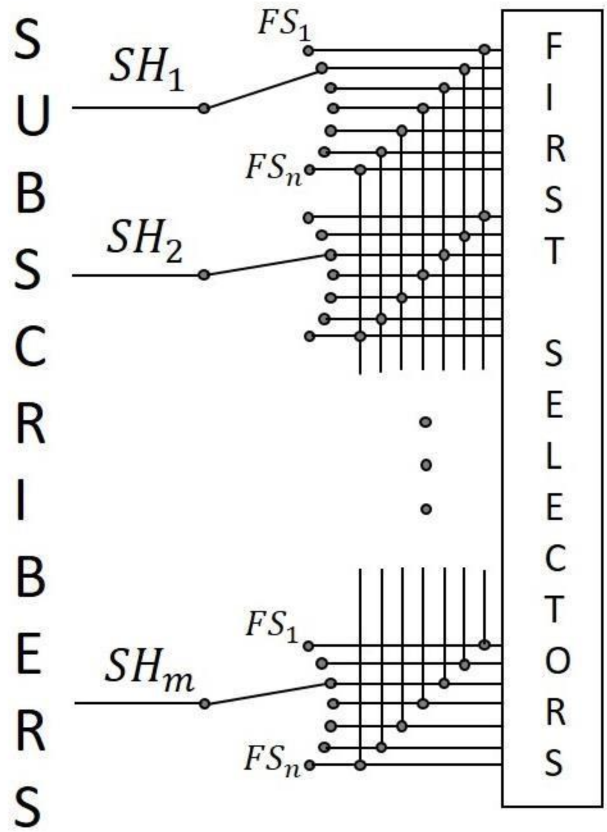
Let us now see how these blocks function.

Selector Hunters

As soon as the calling subscriber gets ready to dial the number, by lifting the handset from the telephone, a dial tone is heard. We have already learnt that a number is not accepted unless the dial tone is heard. But to get that dial tone, the line has to be established when the handset is lifted up. The **Selector Hunter** circuit, establishes the line to make a call as immediately as the calling subscriber lifts up the handset to make a call.

The Selector Hunters hunt for selecting a switching matrix part. Usually, 24-outlet Uni-selectors are used as selector hunters. and so this can be called as **Subscriber Uni-selector** scheme as there is a dedicated Uni-selector for each subscriber in the system. These can also be build using two-motion selectors.

The selector hunter mechanism can also be replaced with the line finder mechanism, where there is small difference between the two in construction. Here, we shall discuss the selector hunter mechanism. The figure below gives an idea about its construction.



When a calling subscriber lifts the handset to make a call, the selector hunter activates the interrupter mechanism, which steps up the wiper until a free first group selector is found at the outlet. One of the bank contacts of the selector hunter, at this point, senses whether the first group selector is free or busy. Once a free first selector is sensed, the interrupter is disabled and the connection is established, where the first selector sends out a dialer tone to the calling subscriber.

The line finder approach is used where the traffic is low and the exchange is small, whereas the selector hunter mechanism described above is used for large exchanges with heavy traffic and this approach is cost-effective.

Group Selector Stage

The Group Selector stage has the main switching network. The calling subscriber dials the number after hearing the dial tone. The first number when dialed activates the first selector. To be more precise, the group selector consists of certain selector stages. We used to have 5 numbers as an identification number, for the land connection. Hence, there were three selector stages present.

To dial the first number, the number plate is rotated by placing the finger in the finger gap given according to the subscriber number. After taking out the finger, the number plate is rotated back to its previous position, which sends the dialing pulses to the first selector. The first selector then moves accordingly, to place a contact.

When the subscriber starts dialing, the dial tone produced till then, cuts off and the pulse train is received according to the number dialed. The wiper assembly of the first selector then moves vertically upward, according to the number dialed. The wipers then move in the horizontal plane across the contacts until they come across a contact to which a free second group selector is connected. This horizontal stepping is completed within the inter- digit gap of about 240ms. From there, the first group selector connects the electrical path to the available second group selector.

Likewise, every group selector connects path according to the number dialed and then extends the connection to the next selector until the final selector. The action of the final selector is a bit different. As discussed above, three selectors are present and the fourth and the fifth numbers are connected to the matrix by the final selector.

Final Selector

The last two digits are processed by the final selector. This selector moves vertically according to the fourth digit dialed and then it moves horizontally according to the last digit, as there are no further digits to connect it to some other connector. The last digit dialed, establishes electrical connection to the called subscriber.

Since the final selector responds to both the digits in vertical and horizontal directions unlike the group selectors, this final selector is also called a **Numerical Selector**. If the called subscriber is free, as sensed from a signal at the corresponding bank contact, the final selector sends out a ringing current to the called subscriber and a ringing tone to the calling subscriber.

When the called subscriber lifts his handset, the ringing current and the ringing tone provided till then, are cut off and the call metering circuits are enabled by the control circuits associated with the final selectors. Otherwise, if the called subscriber is found to be busy on some other line, then the final selector sends out a busy tone to the calling subscriber. At any stage of switching, if there is no free selector available at the next stage, a busy tone is returned to the calling subscriber.

The magnets and mechanical linkages used in rotating the shafts vertically and horizontally while connecting a call, will release the magnet (generally called the release magnet) and armature release the shaft when the call is completed.

Common Control Subsystem

In this chapter, we will discuss how the Common Control Subsystem works in Telecommunication Switching Systems and Networks.In order to establish calls between different exchanges, which may further lead to a long distance trunk call, the Crossbar switching system was developed and the first patent was given in 1915. However, AT&T developed the first Crossbar switching system in 1938. The Crossbar switching system introduced the **Common Control Subsystem** in its switching system.

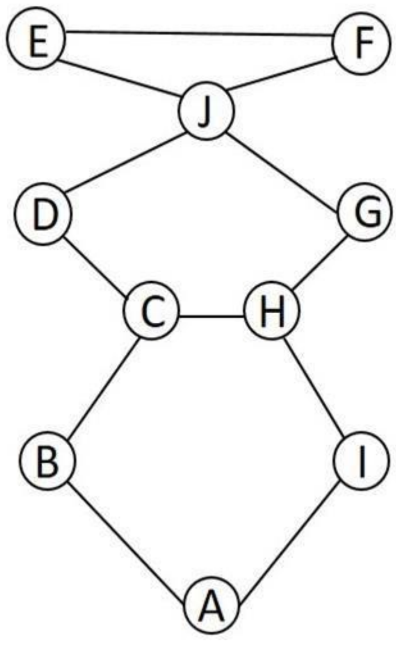
To understand this, let us have an idea on the problem created by Multi-exchange network of the Strowger system.

Multi-Exchange Network

When a subscriber belonging to a particular network has to be contacted, a number of ways can help you contact the particular exchange; also, there is not one but any exchanges present in the route.

In a Multi exchange network, the routes used to establish connection with a particular subscriber differs from time to time. In the Strowger exchange following the Multi- exchange network, the subscriber has to be more concerned with the routing. A subscriber should have the details of all the numbers of exchanges present in the route. There may arise situations where a subscriber may be required to establish a connection on other routes; this becomes cumbersome at times.

The following figure is an example of the topology of a Multi-exchange network.



The level is reserved in each Strowger exchange, where the outgoing calls are connected to neighboring exchanges. These exchanges are contacted as per the exchange numbers dialed, when the calls are made.

Hence, the disadvantages of implementing Multi-Exchange network in switching are:

- The subscriber identity number is changed depending on the calling route.
- The user must have knowledge on the topology of the network and the numbers of the exchanges present in it.
- The number and size of the called subscriber varies depending upon the exchange from where the call originates.

In order to overcome these problems, the common control subsystem was introduced.

Common Control Subsystem

In order to avoid the complication and to make it easier for a subscriber to place a call, two main ideas were implemented by the Common Control Sub system. The ideas are listed below:

- The routing of the call should be done by the exchange, but not by the numbers dialed.
- A Unique Identification Number should be allotted to the subscriber. The UIN contains the number of the exchange of the subscriber and the number indicating the line of the subscriber.

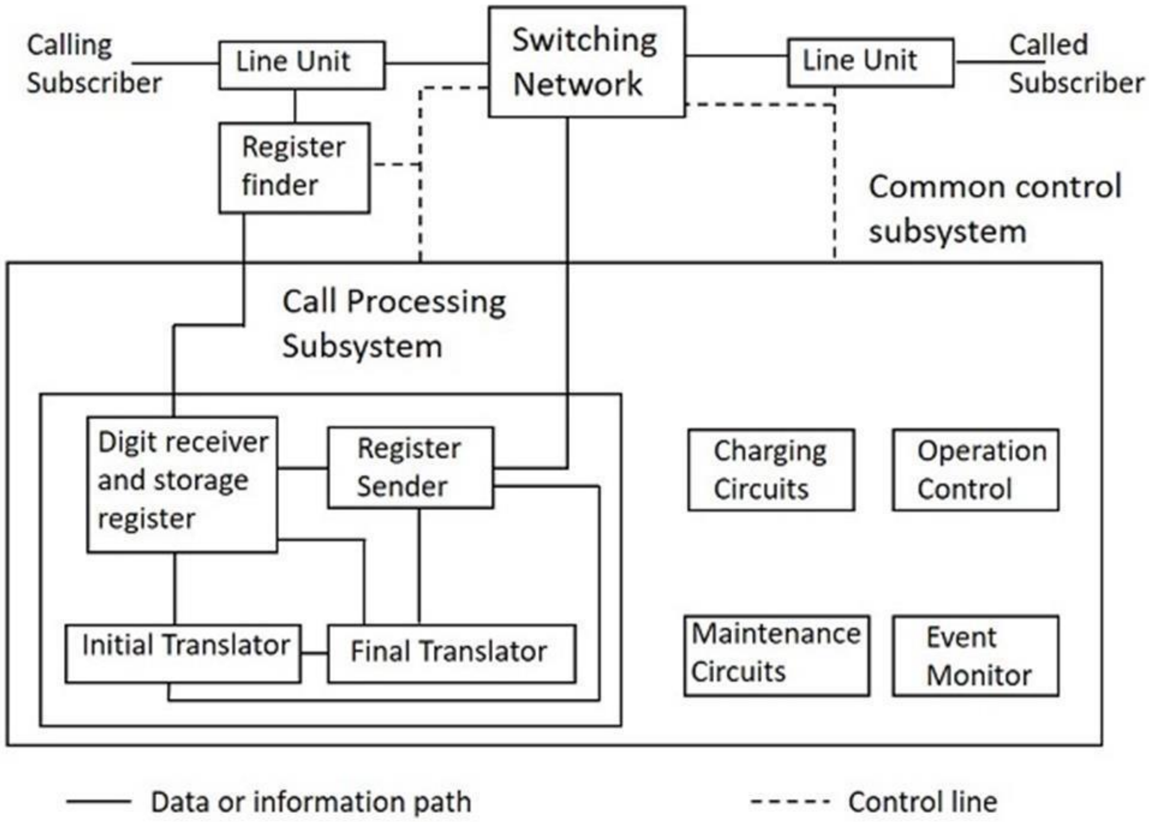
The above ideas helped solve the problem associated with the placement of calls. These two solve the problem and make the work simple. Wherever the calling subscriber calls from, the call is directed to the particular called subscriber, in a particular network. The routing of this call is taken care by the exchange itself. Hence, the uniform numbering scheme identifies the subscriber based on the aspects described below.

Exchange Identifier + Subscriber Line Identifier

This is a combination of STD (Subscriber Trunk Dialing) code and the subscriber‘s number; consider this as the physical line address. Every user is assigned a logical number irrespective of the physical line number. An Address translation mechanism translates the logical address to actual physical address for connection establishment. The call processing takes place independent of the switching network.

A Director system is employed in the common control sub system. As soon as the translated digits are transmitted, the Director is free to process another call and is not involved in maintaining the circuit for the conversation.

The following figure shows the diagram of the Common Control Subsystem, which contains Call Processing Sub system, Charging Circuits, Operation Control, Maintenance Control and Event Monitor.



The above block diagram is a simple indication of the common control switching system. The control functions in a switching system can be categorized as the following.

Event Monitoring

Event Monitoring Section of the Control Subsystem monitors the events occurring outside the exchange at the line units, trunk junctures and inter exchange signaling and sender/receiver units. The events at the **line units** are - call request and call release. The control of relays to establish connection to the required line is an event at the **junctures**. There is control of relays between the exchanges for connection and also for signaling the required tones both to the sender and receiver circuits at the **inter exchange**. This event monitoring may be distributed.

Call Processing

The Call Processing units contain digit receiver and storage register, which receive and store the dialing number from the calling party. The units also contain the initial and final translators. The **Initial translator** is the **Office Code translator** that determines the route for the call through the network or charging method or rate. The **Final translator** is the **Subscriber Code translator** which determines the line unit to which a call must be connected and category of the called line. The Register Sender transfers the route digit and dialed digit using proper signaling, depending on the requirements of the destination exchange.

Charging

This is related to the charges levied on the calls made. It depends upon the type of subscriber and the service of the subscriber. For example, some services like emergency lines or fault repairs are free of charge; a few commercial services also may offer charge- free services.

Operation and Maintenance

The control and operation of the switching network with two main techniques known as Map-in-memory and Map-in-network.

Map-in-Memory

The path in this technique is determined by marking the switching elements at different stages in accordance with a set of binary data defining the path, whereas the control unit supplies the data. At this stage, the command for the actual connection of the path is given. This Map-in-memory technique is present in Stored Program Control.

Map-in-Network

In this technique, the Path finding may be carried out at the level of common control unit, where it marks the inlet and outlet to be connected and the actual path is determined by the switching network. This Map-in-Network technique is common in Crossbar exchanges using markers for control.

The administration and maintenance of a switching system, involves activities such as laying the new subscriber lines and trunks into service, modifying subscriber service entitlements and changing routing plans based on the network status, which are performed with the coordination of control systems. Maintenance personnel do the maintenance activities such as supervision for proper functioning, performing tests and making measurements for different line parameters.

Touch-tone Dial Telephone

In this chapter, we will learn about the Touch-tone Dial Telephone technology. When we talk about the technological development of the telephone set, the rotary dial was used in the initial stages. Slower dialing was one major disadvantage associated with the Rotary dial. It took 12 seconds to dial a 7-digit number on a Rotary dial. The step-by-step switching elements of the Strowger switching system, cannot respond to rates higher than 10-12 pulses per second.

It uses the DTMF technology, prior to which the pulse dialing technique was used. In the **Pulse dialing** technique which is also called a **Loop disconnect** technique, repeated connecting and disconnecting of the lines is done, like clicks of a switch; this is interpreted by the exchange as the number dialed, according to the number of clicks.

Need for Touch-tone

With the introduction of the Common Control subsystems into switching exchanges, there came the feasibility for higher rates of dialing. Hence, a new system called the **Touch- tone dialing** was developed in Telephony to replace the Rotary dial; this was considered to benefit the customer with higher speed. This has also removed the disadvantages of limited usage and limited signaling capacity along with lower speed.

The Pulse dialing is limited to signaling between the exchange and the subscriber, but not between two subscribers, which is called End-to-End signalling. **End-to-End signaling** is a desirable feature and is possible only if the signaling is in voice frequency band so that the signaling information can be transmitted to any point in the telephone network to which voice can be transmitted.

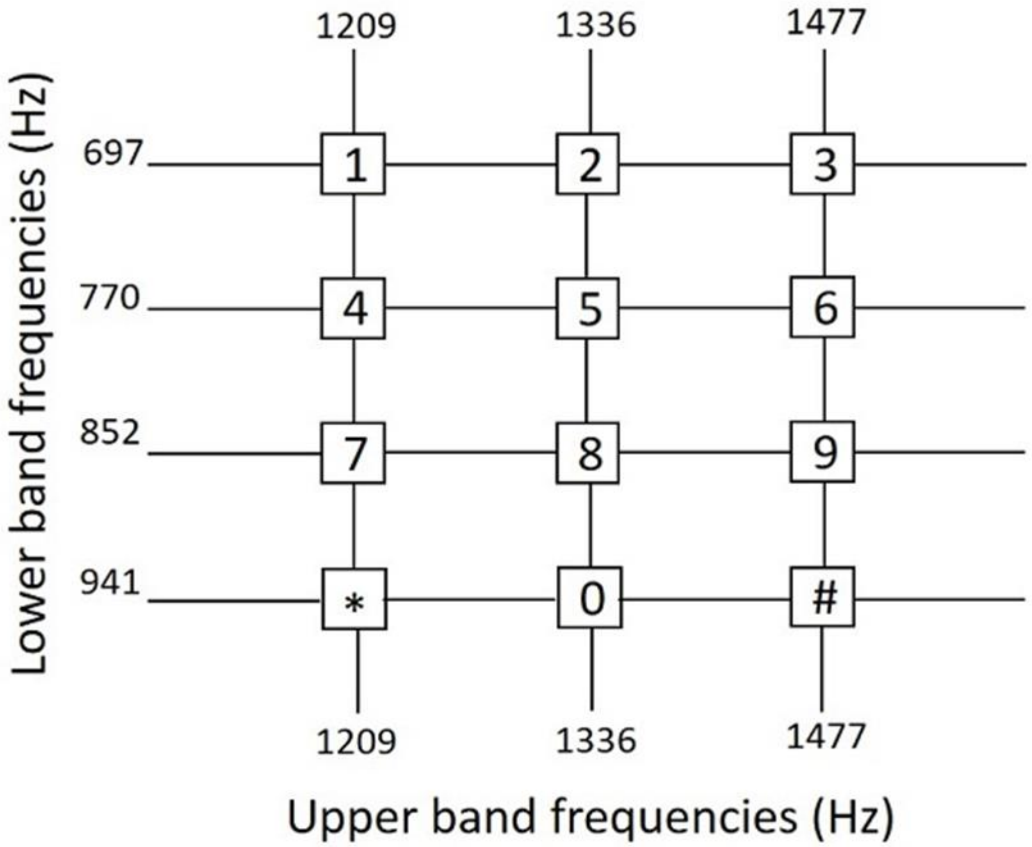
Hence replacing the inconvenience of using the rotary dial, the touch-tone dial telephone was introduced. The development of the touch-tone dial telephone came around 1950. However, the usage of it started somewhere around 1964. The following figure shows a practical touch-tone dial telephone.

The above figure will help you understand that the rotary dial is replaced with a push button keyboard, where the buttons, if touched to ―press‖ the button will generate frequencies related to the number dialed. The hassle-free rotation was replaced and a feature to redial the number was added to this push button keyboard, where the dialed number is stored until another number is dialed. This eased the process of redialing a 7- digit number all over again.

How does the Touch-tone Dial Telephone Operate?

The press of a button on the touch-tone dial telephone indicates the number dialed using certain frequencies. –**Touching**‖ or light pressing of a number generates a –**tone**‖ which is a combination of two frequencies, one from lower band and the other from upper band.

For example, by pressing the button 9, two frequencies such as 852 Hz the lower frequency and 1477Hz the upper frequency are produced. The design of touch-tone dialing producing two frequencies is as shown below.



The DTMF (Dual-tone Multi-frequency) dialing can be done through the touch-tone dialing technique as shown above. As two frequencies, one being higher and the other being lower are transmitted at the same time in the touch-tone dialing technique, it is called the **Dual- tone Multi Frequency** (**DTMF**) dialing. The two signals produced are for a duration of 100ms, which are selected by the key pressed from the matrix as shown above. Each key is uniquely referenced by selecting one of the four lower band frequencies associated with the matrix rows, coupled with selecting one of the three higher band frequencies associated with the matrix column.

Design Considerations

The design considerations are

- Choice of Code
- Band Separation
- Choice of Frequencies
- Choice of Power Levels
- Signaling Duration

The **choice of code** for touch-tone signaling should be such that the imitation of code signals by music and speech must be difficult.

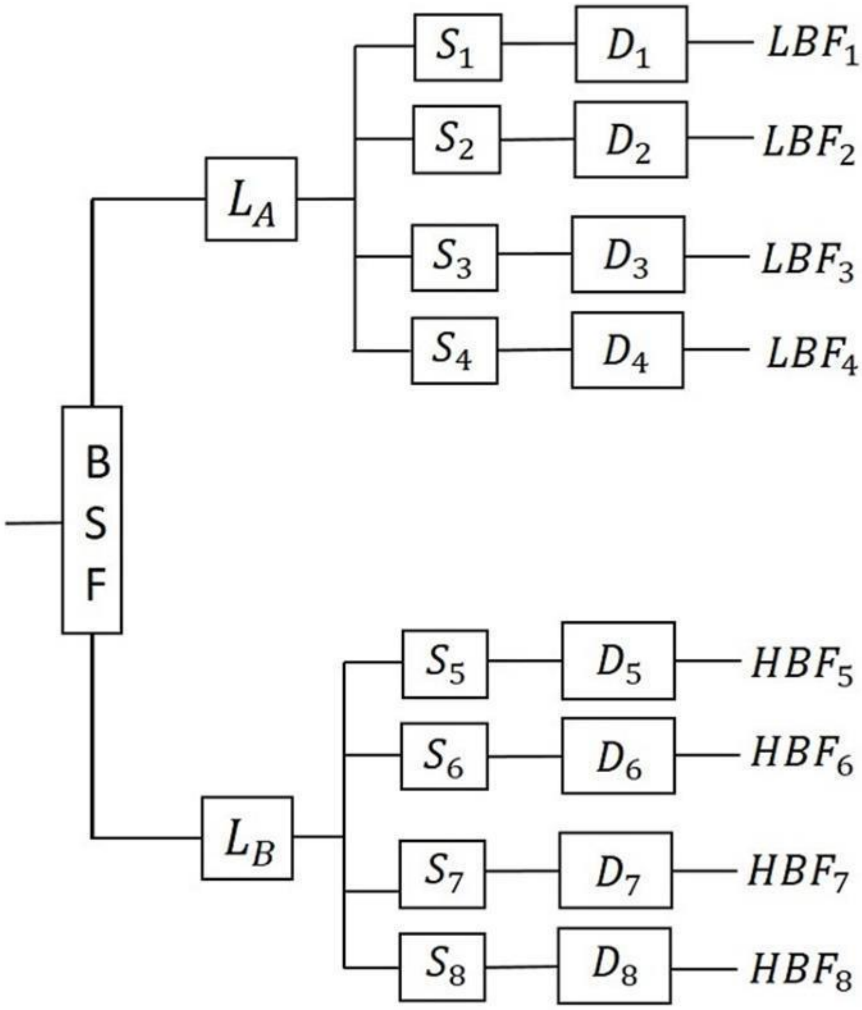
Consider the following reasons for separating the **band** of two frequencies:

- At the receiver, band filtering is used to separate the frequency groups; this helps to determine the specific frequencies in a simple way.
- Easy amplitude regulation of each frequency component separately.
- Limiters can be used to guard the action of each frequency separately.
- The probability of false response is reduced.

The attenuation and delay distortion characteristics of the telephone network circuits determine the **choice of frequencies**. A flat amplitude response with a very low attenuation and a uniform delay response with a low relative delay value are desirable. Though the design is high enough for reliability, the **choice of power levels** should be planned according to attenuation characteristics of the channel. The **signal duration** although inefficient is longer and helpful to combat talk-off.

Internal Mechanism

The internal mechanism of the touch-tone receiver can be explained by a simple block diagram which contains Band Separation Filter (BSF), Limiters (L), Selector Circuits (S) and Detectors (D) which give out Low Band Frequency (LBF) signals and High Band Frequency (HBF) signals, as indicated below.



The Band separation filter present at the receiver is used to separate the frequency groups. This helps to determine the specific frequencies, separately. In addition, the filter also regulates the amplitudes of each component. Then the signal reaches the limiter, which has two of the frequencies at its input. It allows the dominant signal through it bypassing the weak signal. If both of the signals have the same strength, the limiter output is much below the full output and neither of the signal dominates.

The selectors present in the circuitry, are designed to recognize the signal when it falls within the specified narrow passband and has an amplitude within the range of 2.5dB of full output of the limiter. Both of the limiter and selector circuits are efficient in recognizing the difference between the **touch-tone** and the **voice signal**, to avoid talk-off. For further improvement, Band Elimination filters are sometimes used in place of Band Separation filters as they permit a wide spectrum of speech to pass through the filters. The high and low band frequency signals reach the output separately through the detector outputs.

Crossbar Switching

In this chapter, we will discuss the concept of Crossbar Switching. The Crossbar exchanges were developed during 1940s. They achieve full access and non-blocking capabilities with the Crossbar switches and common control equipment, used in the Crossbar exchanges. The active elements called **Crosspoints** are placed between the input and the output lines. In the common control switching systems, the separation between switching and control operations allows the usage of switching networks by a group of common control switches to establish many calls at the same time on a shared basis.

The Features of Crossbar Switches

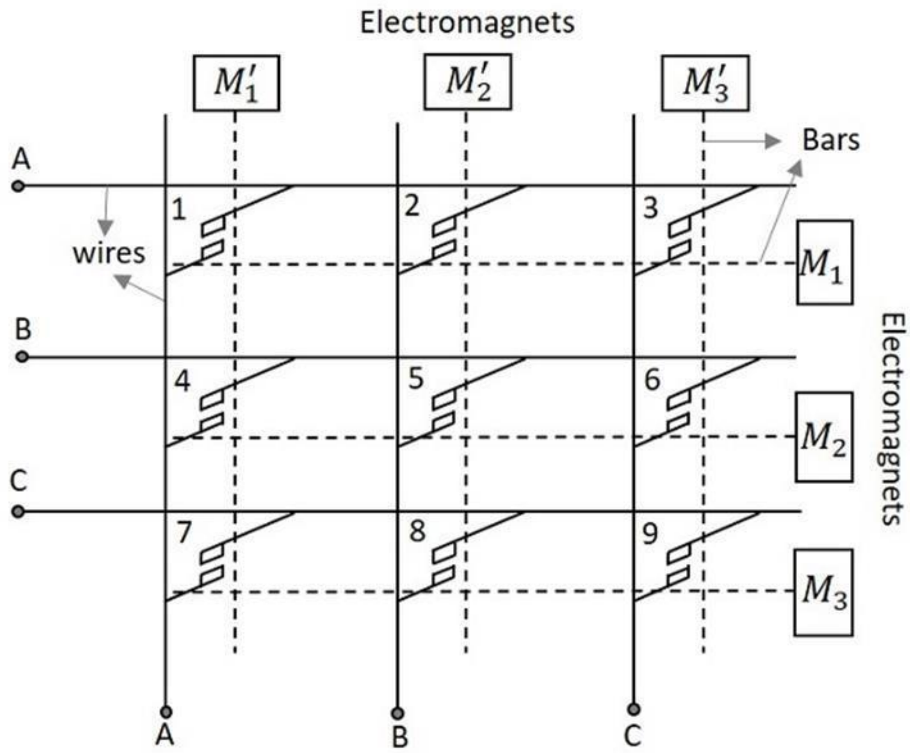
In this section, we will discuss the different features of the Crossbar Switches. The features are described in brief below:

- While processing a call, the common control system helps in the sharing of resources.
- The specific route functions of call processing are hardwired because of the Wire logic computers.
- The flexible system design helps in the appropriate ratio selection is allowed for a specific switch.
- Fewer moving parts ease the maintenance of Crossbar switching systems.

The Crossbar switching system uses the common control networks which enable the switching network to perform event monitoring, call processing, charging, operation and maintenance as discussed previously. The common control also provides uniform numbering of subscribers in a multi-exchange area like big cities and routing of calls from one exchange to another using the same intermediate exchanges. This method helps to avoid the disadvantages associated with the step-by-step switching method through its unique process of receiving and storing the complete number to establish a call connection.

Crossbar Switching Matrix

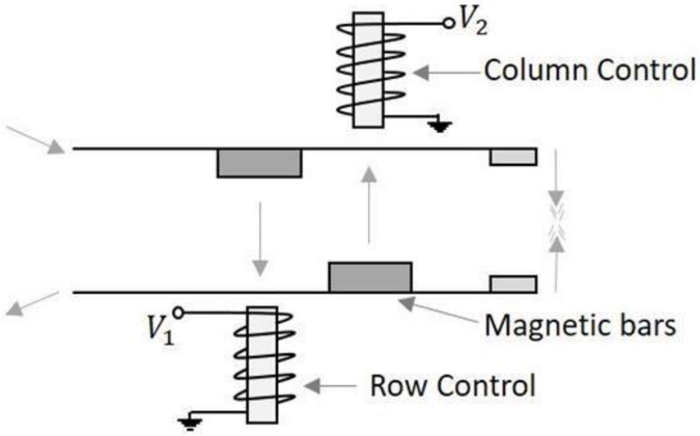
The Crossbar arrangement is a matrix which is formed by the M X N sets of contacts arranged as vertical and horizontal bars with contact points where they meet. They need nearly M + N number of activators to select one of the contacts. The Crossbar matrix arrangement is shown in the following figure.



3 X 3 Crossbar Switching

The Crossbar matrix contains an array of horizontal and vertical wires shown by solid lines in the following figure, which are both connected to initially separated contact points of switches. The horizontal and vertical bars shown in dotted lines in the above figure are mechanically connected to these contact points and attached to the electromagnets.

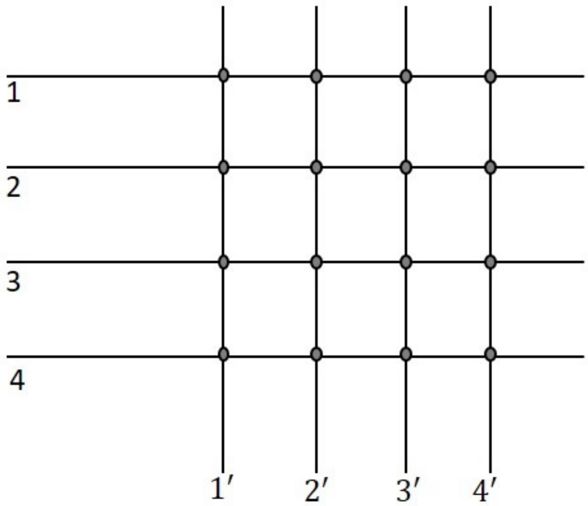
The Crosspoints placed between the input and the output lines have electromagnets which when energized, close the contact of intersection of the two bars. This makes the two bars to come closer and hold on. The following figure will help you understand the contact made at the Crosspoints.



Once energized, the electromagnets pull the small magnetic slabs present on the bars. The column control electromagnet pulls the magnet on the lower bar, while the row control

electromagnet pulls the magnet on the upper bar. In order to avoid the catching of different Crosspoints in the same circuit, a procedure is followed, to establish a connection. According to this procedure, either horizontal or vertical bar can be energized first to make a contact. However, to break a contact, the horizontal bar is de-energized first; the vertical bar being de-energized follows this.

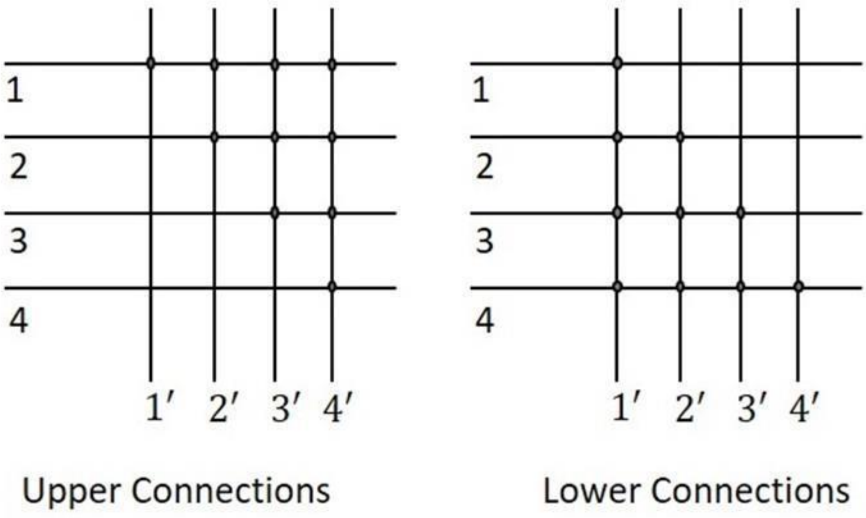
As all the stations are allowed to be connected with all possible connections as long as the called party is free, this Crossbar Switching is called the **Non-Blocking Crossbar configuration**, which requires N^2 switching elements for N subscribers. So, the Crosspoints will be highly greater than the subscribers. For example, 100 subscribers will require a 10,000 Crosspoints. This means that this technique can be applied to a group having a small number of subscribers.



There is an external switch called the **Marker**; this can control many switches and serve many registers. The switch decides the operation of magnets such as the select magnet and the bridge magnet that should be energized and de-energized for connecting and releasing the subscriber respectively.

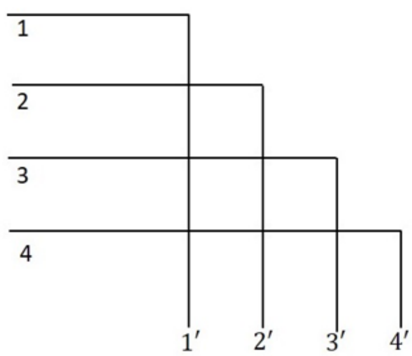
Diagonal Crosspoint Matrix

In the matrix, as 1,2,3,4 indicate input lines and 1‘,2‘,3‘,4‘ indicate output lines of the same subscribers, if a connection has to be established between the 1st and the 2nd subscriber, then 1 and 2‘ can be connected or 2 and 1‘ can be connected using the Crosspoints. In the same way, when a connection has to be established between 3 and 4, then 3-4‘ Crosspoint or 4-3‘ Crosspoint can do the work. The following figure will help you understand how this works.



Now, the diagonal portions are the Crosspoints connecting to the same subscriber again. A line that is already connected to the terminal has no need of connecting it again to the same terminal. Hence, the diagonal points are also not necessary.

So, it is understood that for N number of subscribers, if the diagonal points are also considered, the total number of Crosspoints will be,For N number of subscribers, if the diagonal points are **not** considered, then the total number of Crosspoints will be,As the number of nodes N increases, the Crosspoints proportionally increase up to N^2.
The Crosspoints will always be linear. Therefore, as either the lower part or the upper part of the diagonal points in the matrix, can be considered, the whole matrix considering the lower portion, will now be as shown in the following figure.



This is called the **Diagonal Crosspoint Matrix**. The matrix is in a triangular format and can be called the **Triangular Matrix** or the **Two-way Matrix**. The diagonal Crosspoint matrix is fully connected. When the third subscriber initiates a call, to the fourth subscriber, then the third subscriber's horizontal bar is initiated first and then the fourth subscriber's vertical bar is energized. The diagonal Crosspoint matrix is a non-blocking configuration. The main disadvantage of this system is that, the failure of a single switch will make some subscribers inaccessible.

The Crosspoint switch is the abstract of any switch such as the time or space switch. If N connections can be made simultaneously in an NXN switch matrix, it is called the **Non-blocking Switch**. If the number of connections made are less than N in some or all cases, then it is called the **Blocking** switch. These blocking switches are worked upon using Multiple Switches and such networks are called **Line frames**.

Crossbar Switch Configurations

In this chapter, we will discuss how the Crossbar switch configuration works. The Crossbar switch configurations are Non-blocking configurations, which have N^2 switching elements for N subscribers and can make N/2 simultaneous conversations. The usage of Crosspoint depends upon the calling subscriber.

This is a modified Non-blocking scheme with Diagonal Crosspoint matrix as discussed above having N(N-1)/2 elements. The number of elements is same as that of a fully connected network. The connection in this method is established by first energizing the horizontal bar and then the vertical bar. However, this Non-blocking scheme has few disadvantages such as:

- Large number of switching elements are required.
- This is difficult to implement in practice.
- This is neither a cost-effective process.

In order to overcome these disadvantages, the blocking Crossbar switching was introduced.

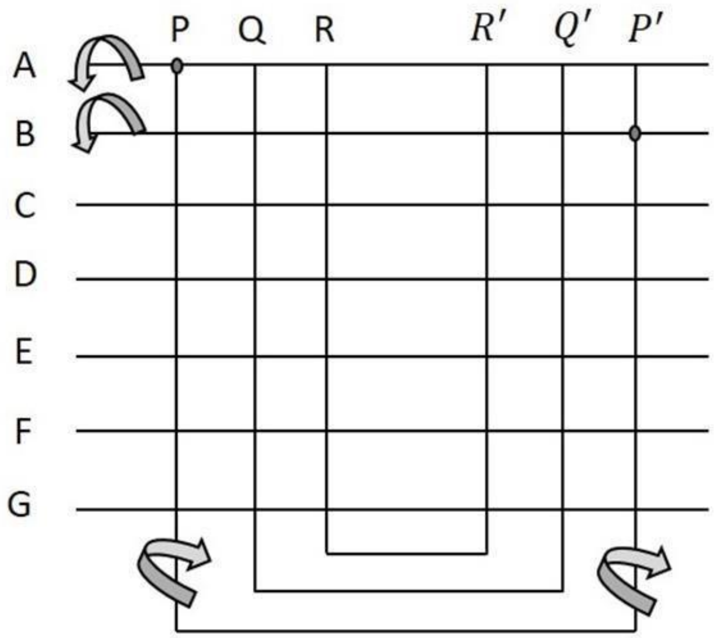
Blocking Crossbar Switches

The main aim of blocking Crossbar switches is to reduce the number of Crosspoint switches. There are single stage and multi-stage switches. The number of Crosspoint switches can be reduced with the help of two different methodologies. In the first method, two subscribers share one vertical bar. With this, the number of bars will be reduced but the number of Crosspoint switches remain the same. The second method is where all the subscribers share a number of vertical bars. With this, the number of bars and Crosspoint switches are reduced.

Method 1

This method contains **2NK** switches, where **N** is the number of subscribers and **K** is the number of simultaneous connections. Four bars operate to establish a connection. If a connection has to be established between A and B, then the horizontal bar A is energized first and then one of the free vertical bars say P is energized. Now, the Crosspoint AP is latched. If the horizontal bar B is energized now, BP will not be latched, as the P vertical is energized before B was energized. To connect A and B, we need another vertical Crossbar which should electrically correspond to the vertical bar P, which is P‘ as shown in the following figure. When this P‘ is energized after B, the Crosspoint BP‘ is latched and a connection between A and B is established.

The connections are as shown in the following figure.



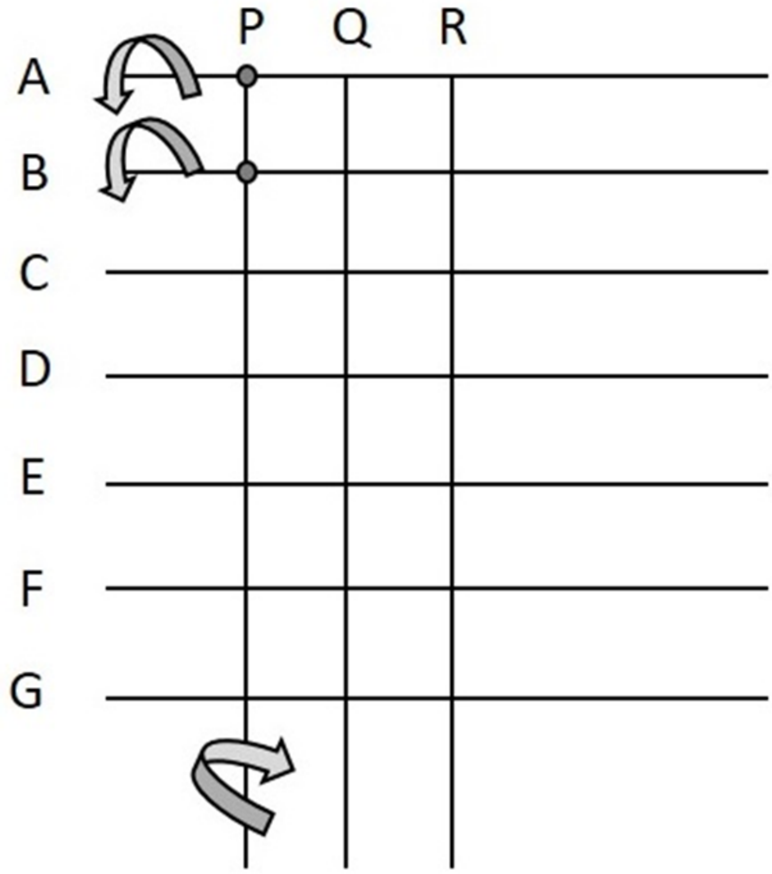
Hence, the steps associated with the establishment of connection follows a sequence:

- Energize horizontal bar A
- Energize free vertical bar P
- De-energize horizontal bar A
- Energize horizontal bar B
- Energize free vertical bar P‘ (associated with P)
- De-energize horizontal bar B

Method 2

This method contains **NK** switches, where **N** is the number of subscribers and **K** is the number of simultaneous connections. Here, three bars operate to establish a connection. If a connection has to be established between A and B, then the horizontal bars A and B are energized first and then one of the free vertical bars say P is energized. Now, the connection is established using one vertical bar P only instead of two bars. The horizontal bars A and B are de-energized now.

The connections are as shown in the following figure.



Hence, the establishment of connection follows a sequence:

- Energize horizontal bars A and B
- Energize free vertical bar P
- De-energize horizontal bars A and B

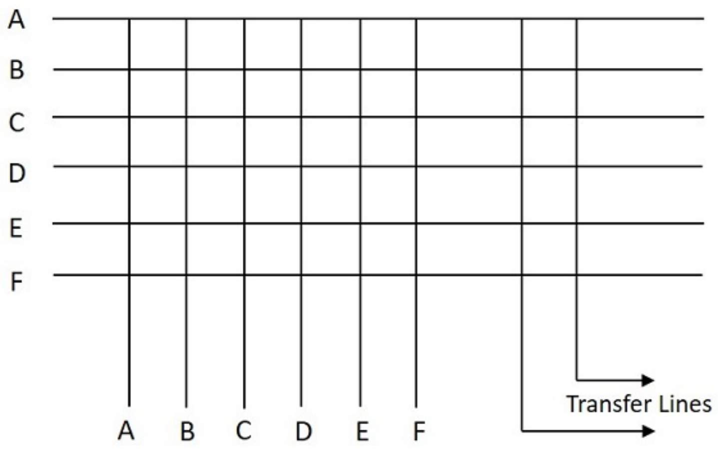
Transfer Line Support

In this section, we will discuss how the Transfer Line Support works. Both of the above discussed blocking and non-blocking type Crossbar switches can support transfer lines. This is done by introducing additional vertical Crossbars and Crosspoint switches.

There are two methods to introduce additional vertical Crossbars and Crosspoint switches:

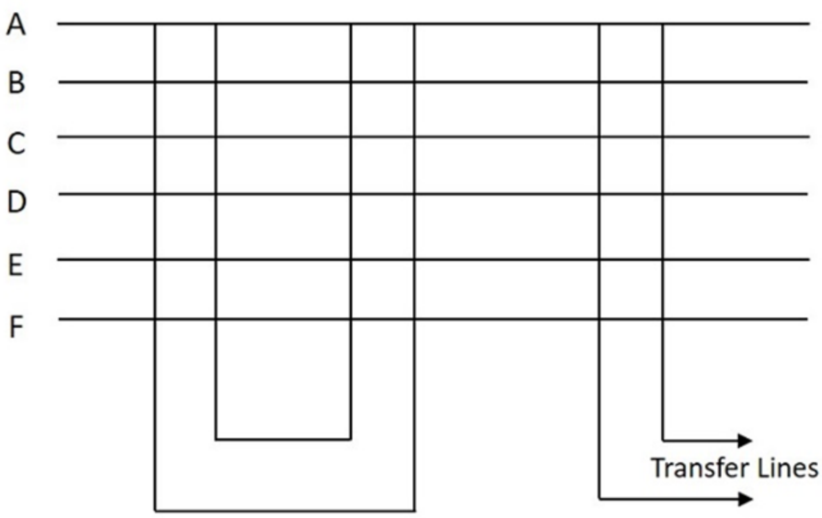
- Internal non-blocking and external blocking
- Blocking both local and external

The internal non-blocking and external blocking method is as shown in the figure below.



The switch shown in internal non-blocking has two transfer lines. The number of Crosspoint switches in this case is **N(N+L)**, where **N** is the number of subscribers, **L** is the number of transfer lines.

The method of blocking both local and external is as shown in the figure below.



The switch shown in the above figure is blocking both internally and externally with two simultaneous internal and two simultaneous external calls. The number of Crosspoint switches in this case is **N(2K+L)**, where **N** is the number of subscribers, **L** is the number of transfer lines and **K** is the number of simultaneous calls that can be supported locally.

Crosspoint Technology

In this chapter, we will discuss the Crosspoint Technology in Telecommunication Switching Systems and Networks.

The Crossbar system mainly consists of the Crosspoint switches, which increases the cost of the system. The cost of the Crossbar system increases in direct proportion to the number of Crosspoint.

Challenges for the Crosspoint Technology

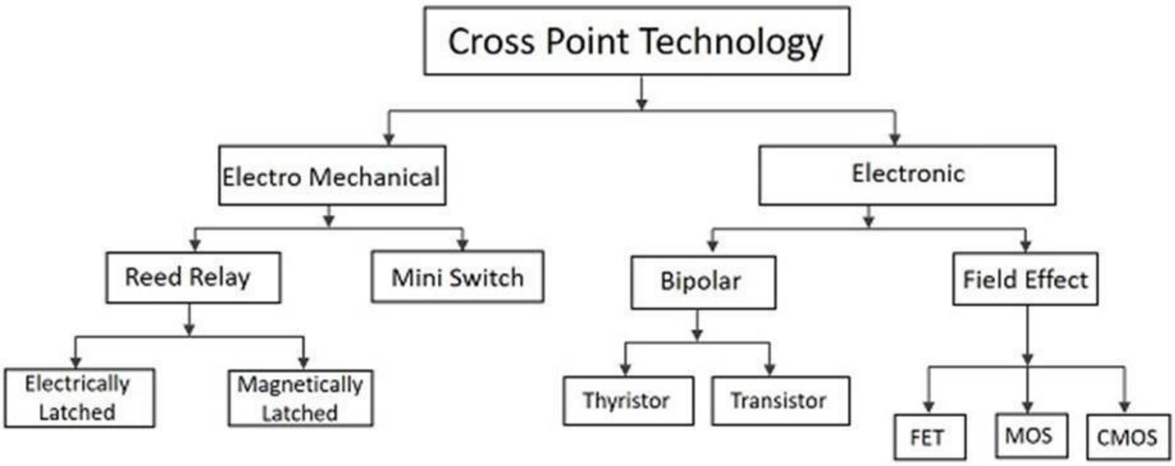
In this section, we will discuss the challenges associated with the Crosspoint technology. The challenges are described below:

- Reduction in the size of a Crosspoint
- Reduction in the cost of a Crosspoint
- Improvisation of the switching time

In the process of finding solutions to the existing challenges, the Crosspoint technology evolved. Crosspoint technology is an amalgamation of two related technologies. The technologies are:

- Electromechanical
- Electronic

The flowchart given below chows the different categories of the Crosspoint technology:



In our subsequent sections, we will discuss more about the related technologies:

Electromechanical Crosspoint Technology

The Electromechanical Crosspoint switches which are capable of making and breaking contacts in 1-10ms of time duration for several million times without any wear and tear

are being extensively used even today. The two types of switches widely used are **Mini switches** and **Reed relay**.

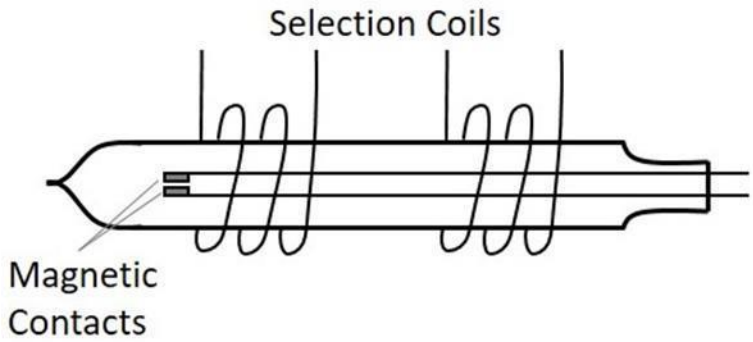
Mini Switches

These switches are made up of a precious metal like Palladium, which makes the contacts work quieter, with their bifurcated design and high resistance to corrosion for long lasting design. These mechanically latched switches use –V‖ notches for this purpose and are highly reliable in Crossbar switching systems.

These switches mounted on Crossbars move horizontally and vertically to establish and release contacts with a switching time of 8-10ms.

Reed Relay Switches

In order to reduce the usage of mechanical switches and increase the operating life of the switches further, the Reed relay switches were introduced. These switches are made up of magnetic material contacts sealed in a glass tube; this protects the contacts from getting contaminated. The following figure illustrates the design of a reed relay switch.



A reed relay switch may be electrically or mechanically latched; it contains the contacts very close to each other having a displacement of 0.2mm resulting in a fast switching speed of 1ms. The construction of this relay is such that the glass tube is surrounded by a pair of coils and when current is passed through both the coils simultaneously, a field is created. This further leads to the reed contacts moving together. As long as it is switched on, the electrical connection is latched and current passes through the coil.

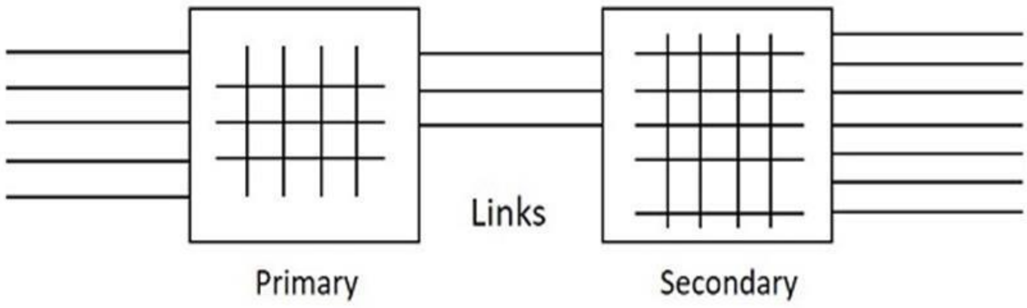
In magnetic latching, the hysteresis of the magnetic material decides the performance. The magnetic pole pieces required may be placed outside the glass or the contacts may act as poles by choosing an appropriate ferromagnetic material. The reed relay is called the **remreed** due to the remnance property of the contact strips. The residual magnetism lets the contacts stay intact even after the currents are withdrawn and hence a demagnetizing current needs to be applied to open the contacts.

These reed relays are placed at each Crosspoint to construct a Crosspoint matrix. Crosspoint selection is achieved by connecting one of the coil windings of each relay in series with its vertical neighbor and the other winding in series with its horizontal neighbor. The reed relay is excited when the required Crosspoint is selected by pulsating the corresponding vertical and horizontal bars simultaneously.

Crossbar Exchange Organization

The organization of a Crossbar exchange consists of three basic building blocks such as link frames, control markers and registers. Link frames contain primary and secondary stages having Crossbars, connected with links between them. This two-stage arrangement with links has the effect of increasing the number of outlets for a given number of inlets. If the number of outlets is high, the selectivity is higher too.

The Markers control the connections between the inlets and outlets via the primary section, links and secondary sections. Registers are there to store the number dialed, in order to establish the connection. The following figure shows the arrangement of a link frame and its control by a marker.



The two main sections of the Crossbar Exchange organization are:

Line Unit

The line link frames along with associated markers and registers can be termed as **Line Unit**. The line units are two-way units that help in the origination and termination of calls. Because of its two-way capability, the secondary section in the line link frame is called the terminal section. The subscriber lines are terminated on the outlets of the terminal section frames.

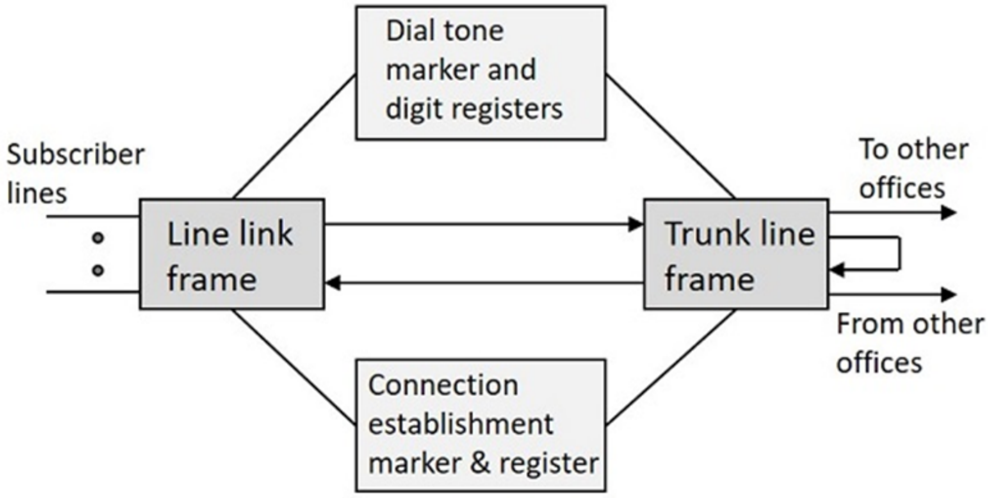
Group Unit

The trunk link frame along with its associated circuitry can be termed as the **Group Unit**. The trunk link frame may be sub-divided into two or three link frames like local office link frame and incoming link frame, etc. Group unit is a uni-directional device that receives the calls from the line unit or from distant exchanges. It is capable of handling local, outgoing, incoming, terminating and transit calls.

Call Processing

A Simplified organization of a Crossbar exchange is shown in the following figure.

The line link frames along with associated markers and registers can be termed as **Line Unit**. The line units are two-way units that help in the origination and termination of calls. Because of its two-way capability, the secondary section in the line link frame is called the terminal section. The subscriber lines are terminated on the outlets of the terminal section frames



The call processing in a Crossbar exchange is done in three stages, named as Pre- Selection, Group Selection and Line Selection.

Pre-Selection

The originating marker does the pre-selection. When the calling subscriber lifts the handset, the dial tone is heard. The register send this tone. This stage that stars from lifting the handset to sending the dialed tone is called **Pre-Selection**.

Group Selection

Once the dial tone is heard, the number can be dialed. The call is switched through the desired direction as decided, in accordance with the code given by the translator. This stage of selecting the desired group for making a call is called **Group Selection**.

Line Selection

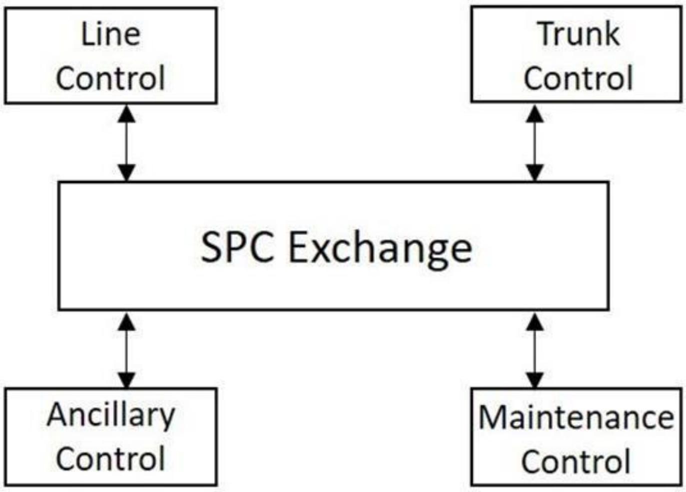
Once a number is dialed, the calling subscriber is connected to the called subscriber by the terminating marker. The line of the called party is controlled by the terminating marker which also sets up ringing on the line. This stage of selecting the line of the desired subscriber can be called as the **Line Selection**.

With these three sections, a call can be connected and processed in a Crossbar exchange.

Stored Program Control

In this chapter, we will discuss the Stored Program Control works in Telecommunication Switching Systems and Networks. In order to increase the efficiency and speed of control and signaling in switching, the use of electronics was introduced. The **Stored Program Control**, in short **SPC** is the concept of electronics that ringed in a change in telecommunication. It permits the features like abbreviated dialing, call forwarding, call waiting, etc. The Stored Program Control concept is where a program or a set of instructions to the computer is stored in its memory and the instructions are executed automatically one by one by the processor.

As the exchange control functions are carried out through programs stored in the memory of a computer, it is called the **Stored Program Control** (**SPC**). The following figure shows the basic control structure of an SPC telephony exchange.



The processors used by SPC are designed based on the requirements of the exchange. The processors are duplicated; and, using more than one processor makes the process reliable. A separate processor is used for the maintenance of the switching system.

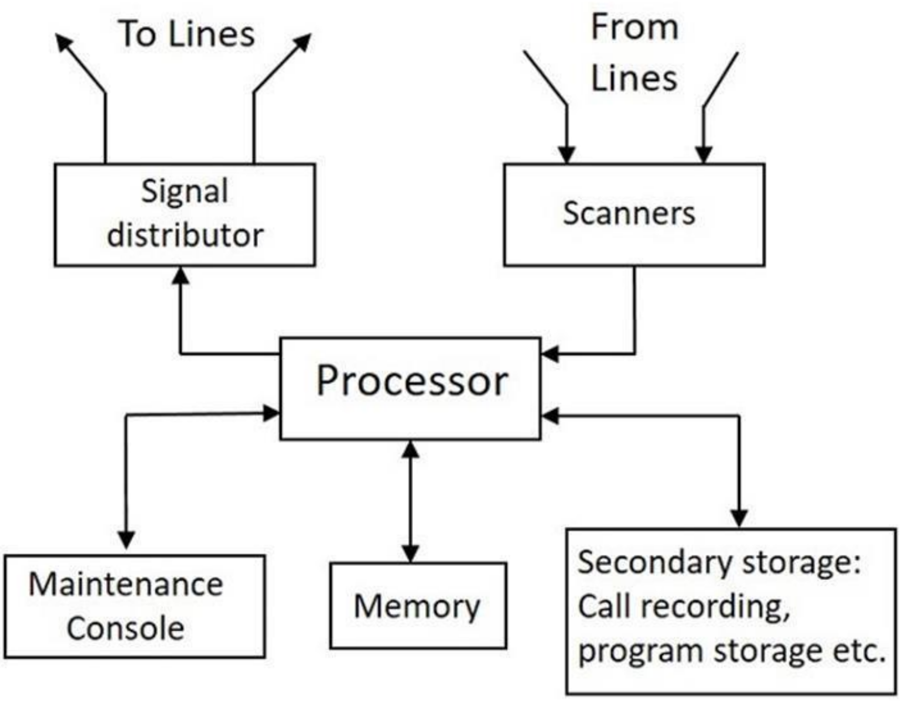
There are two types of SPCs:

- Centralized SPC
- Distributed SPC

Centralized SPC

The previous version of Centralized SPC used a single main processor to perform the exchange functions. The dual processor replaced the single main processor at a later stage of advancement. This made the process more reliable. The following figure shows the organization of a typical Centralized SPC.

As the name implies, in the two processors present, one processor is active and the other is in the standby mode. The processor in the standby mode is used as a backup, in case the active one fails. This mode of exchange uses a secondary storage common to both the processors. The active processor copies the status of the system periodically and stores in the axis secondary storage, but the processors are not directly connected. The programs and instructions related to the control functions, routine programs and other required information are stored in the Secondary storage.



A dual processor architecture may be configured to operate in three modes like:

- Standby Mode
- Synchronous Duplex Mode
- Load Sharing Mode

Standby Mode

As the name implies, in the two processors present, one processor is active and the other is in the standby mode. The processor in the standby mode is used as a backup, in case the active one fails. This mode of exchange uses a secondary storage common to both the processors. The active processor copies the status of the system periodically and stores in the axis secondary storage, but the processors are not directly connected. The programs and instructions related to the control functions, routine programs and other required information are stored in the Secondary storage.

Synchronous Duplex Mode

In the Synchronous Duplex mode, two processors are connected and operated in synchronism. Two processors P1 and P2 are connected and separate memories like M1 and M2 are used. These processors are coupled to exchange the stored data. A Comparator is used in between these two processors. The Comparator helps in comparing the results.

During the normal operation, both of the processors function individually receiving all the information from the exchange and also related data from their memories. However, only one processor controls the exchange; the other one remains in synchronism with the previous one. The comparator, which compares the results of both the processors, identifies if any fault occurs and then the faulty processor among them is identified by operating them individually. The faulty processor is brought into service only after the rectification of fault and the other processor serves meanwhile.

Load Sharing Mode

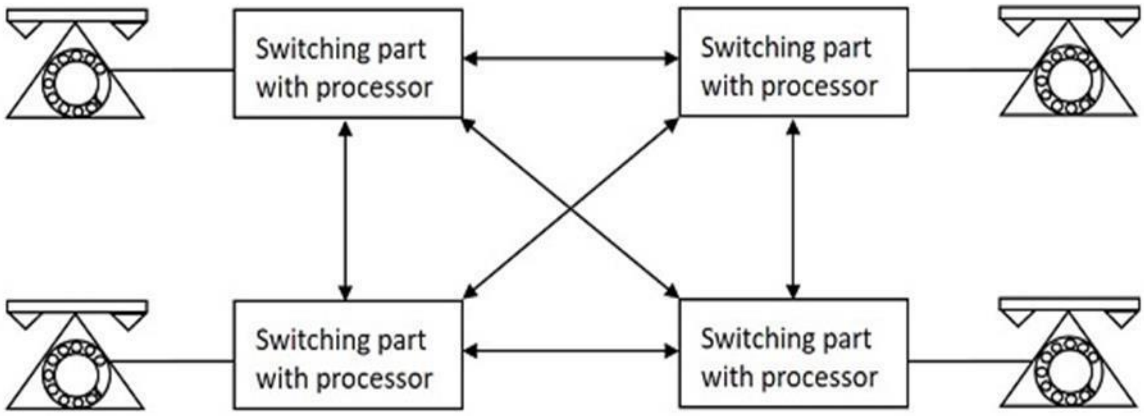
Load sharing mode is where a task is shared between two processors. The Exclusion Device (ED) is used instead of the comparator in this mode. The processors call for ED to share the resources, so that both the processors do not seek the same resource at the same time.

In this mode, both the processors are simultaneously active. These processors share the resources of the exchange and load. In case one of the processor fails, the other one takes over the entire load of the exchange with the help of ED. Under normal operation, each processor handles one-half of the calls on a statistical basis. The exchange operator can however vary the processor load for maintenance purpose.

Distributed SPC

Unlike Electromechanical switches and Centralized SPC, the introduction of Distributed SPC has enabled to provide a wide range of services. This SPC has separate small processors called the **Regional Processors** that deal with different works, rather than just one or two processors working on the whole thing like in the centralized system. However, when these regional processors are required to perform complex tasks, the centralized SPC helps by directing them.

The Distributed SPC has more availability and reliability than Centralized SPC, because entire exchange control functions may be decomposed either horizontally or vertically for distributed processing. Such distributed control where switching equipment is divided into parts, each of which have its own processor, is indicated in the figure below.



The exchange environment in vertical decomposition is divided into several blocks and each block is assigned to a processor that performs all the control functions that are related to specific block of equipment, whereas each processor in horizontal decomposition performs one or some of the exchange control functions.

Software Architecture

In this chapter, we will learn about the Software Architecture of Telecommunication Switching Systems and Networks.

The software of the SPC systems can be categorized into two for better understanding – **System Software** and **Application Software**. The Software architecture deals with the system software environment of SPC including the language processors. Many features along with call processing are part of the operating system under which operations and Management functions are carried out.

Call Processing is the main processing function, which is event oriented. The event that occurs at the subscriber's line or trunk triggers the call processing. Call setup is not done in one continuous processing sequence in the exchange. This entire process is consistent of many elementary processes that last for few tens or hundreds of milliseconds and many calls are processed as such simultaneously and each call is handled by a separate **Process**. A **Process** is an active entity which is a **program in execution**, sometimes even termed as a **task**.

Process in a Multiprogramming Environment

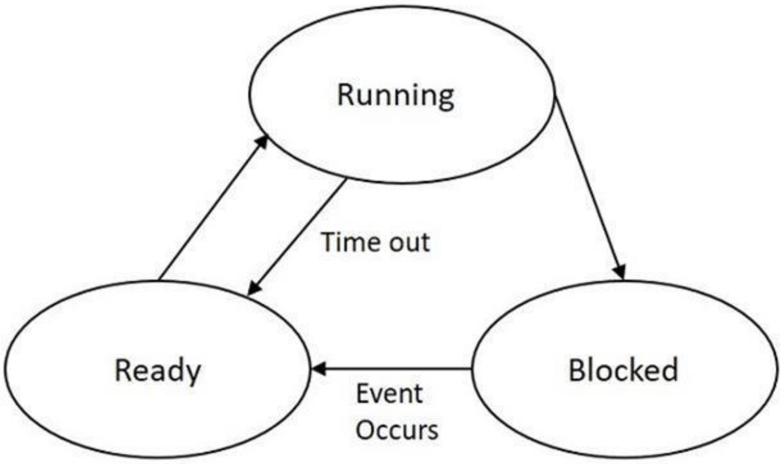
In this section, we will see what a process in a multiprogramming environment is. A Process in a multiprogramming environment may be one of the following:

- Running
- Ready
- Blocked

The state of a process is defined by its current activity and the process it executes and the transitions that its state undergoes.

- A Process is said to be **running**, if an instruction is currently being executed by the processor.
- A Process is said to be **ready** if the next instruction of running a process is waiting or has an instruction that is timed out.
- A Process is said to be **blocked**, if it is waiting for some event to occur before it can proceed.

The following figure indicates the process that shows the transition between running, ready and blocked.



While some processes are in the running state, some will be in the ready state while others are blocked up. The processes in the ready list will be according to the priorities. The blocked processes are unordered and they unblock in the order in which the events are waiting to occur. If a process is not executed and waits for some other instruction or resource, the processor time is saved by pushing such process to the ready list and will be unblocked when its priority is high.

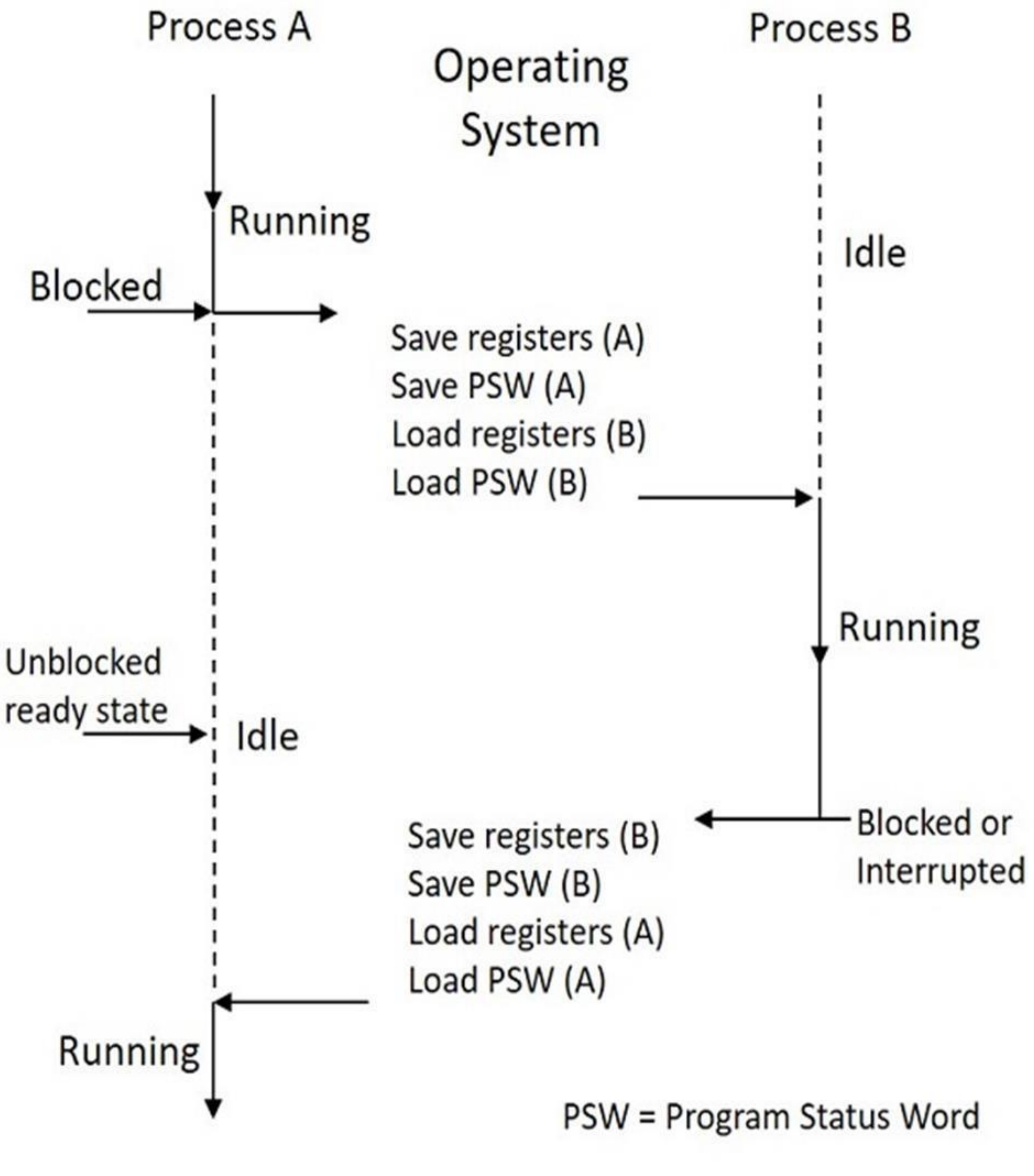
Process Control Block

The Process Control Block represents each process in the operating system. PCB is a data structure containing the following information about the process.

- Current running state of the process
- Process priority which are in the ready state
- CPU scheduling parameters
- Saves the content of CPU, when a process gets interrupted
- Memory allocation to the process
- The details of process like its number, CPU usage etc. are present
- Status of events and I/O resources that are associated with the process

PCB has all the information about the processes to be executed next when it gets the CPU. The CPU registers include a **Program Status Word** (PSW) that contains the address of the next instruction to be executed, the types of interrupts enabled or disabled currently, etc.

While the CPU executes some process, that process needs to be switched when the currently running process becomes blocked or an event or interrupt that triggers a high priority process occurs. Such situation is called **Process Switching**, which is also known as **Context Switching**. Such interrupt priority mechanism is described in the following figure.



If a process **A** scans a particular subscriber line and finds it free, then the process establishes a call with that subscriber. However, if another process B claims the priority and establishes a call with the same subscriber at the same time, then both the processes need to make a call to the same subscriber at the same time, which is not suggestable. A similar problem might occur with other shared tables and files also.

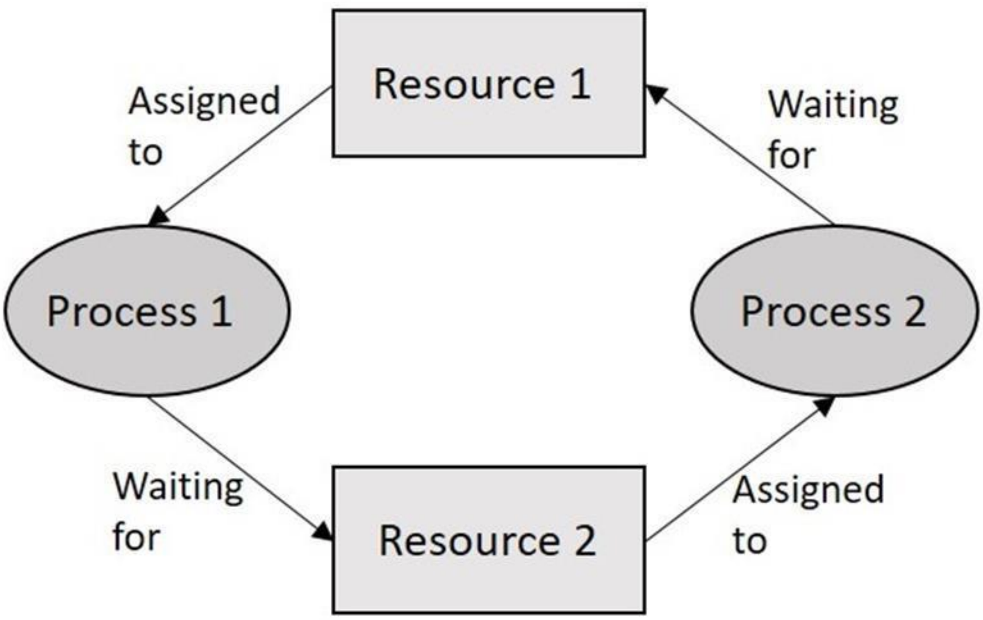
Information about the resources of the exchange (trunks, registers etc.) and their current utilization is kept in the form of tables. These tables when needed are shared by different processes. The problem occurs when two or more processes opt for the same table at the same time. This problem can be solved by giving access to each process to a shared table.

Sharing Resources

Whenever a process uses a shared table or any shared resource, all the other processes that needs the same are to be kept waiting. When the running process finishes using the resource, it will be allotted to the first prioritized ready process which is kept waiting. This process of using the shared resources is called **Mutual Exclusion**. The process, which is accessing the shared resource, is said to be in its **Critical Section** or **Critical Region**. Mutual Exclusion implies that only one process can be in the critical region at any instance for a given shared resource. The coding for the process to be in the critical section is done very carefully that there are no infinite loops. This helps in the process not being blocked. The work done is more accurate and efficient. This helps the other processes that are waiting.

If two processes in a semaphore have to share a common resource, it is shared by them for certain time intervals. While one uses the resource, the other waits. Now, while waiting, in order to be in synchronism with the other one, it reads the task that was written until then. This means, the state of that process should be non-zero and should keep on incrementing, which otherwise would be sent out to the blocked list. The processes that are in the blocked list are stacked one over the other and are allowed to use the resource according to the priority.

The following figure shows how the process works:



If two or more processes in a semaphore wait indefinitely for a resource and does not get zero to return to the block state, while other processes wait in the blocked state for the use of the same resource while none could use the resource but wait, such a state is called the **Deadlock State**.

The techniques have been developed for deadlock prevention, avoidance, detection and recovery. Therefore, these cover the salient features of operating system for switching processors.

Software Production

The SPC software production is important because of its complexity and size of the software along with its long working life and reliability, availability and portability.

Software production is that branch of software engineering that deals with the problems encountered in the production and maintenance of large scale software for complex systems. The practice of software engineering is categorized into four stages. These stages make up for the production of software systems.

- Functional specifications
- Formal description and detailed specifications
- Coding and verification
- Testing and debugging

The Application software of a switching system may be divided into call processing software, administrative software and maintenance software; the application software packages of a switching system use a modular organization.

With the introduction of the Stored Program Control, a host of new or improved services can be made available to the subscribers. Many kinds of enhanced services such as abbreviated dialing, recorded number calls or no dialing calls, call back when free, call forwarding, operator answer, calling number record, call waiting, consultation hold, conference calls, automatic alarm, STD barring, malicious call tracing, etc. are all introduced with these changes in the telephony.

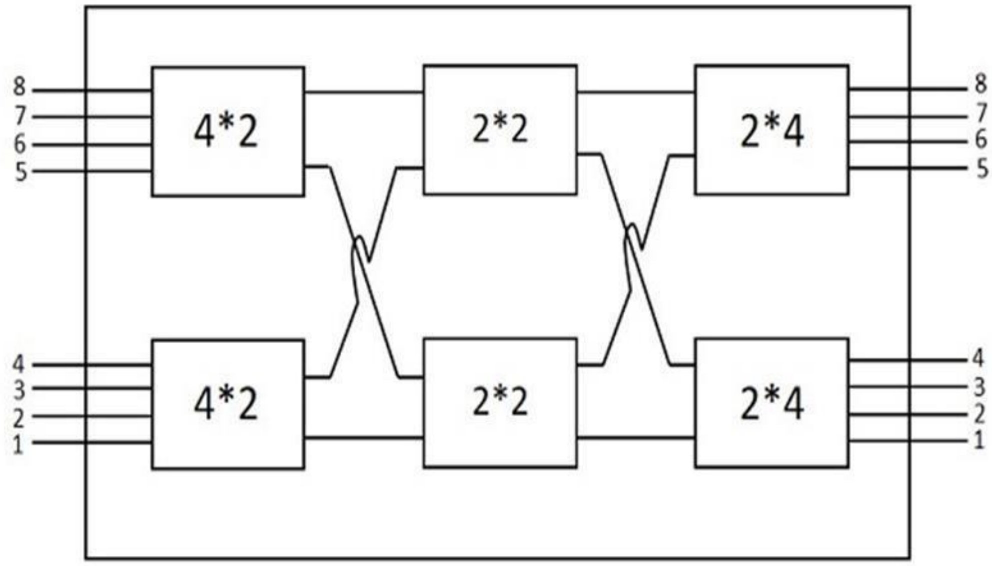
Multi-Stage Networks

The multi-stage networks are the networks built to provide connections between more subscribers more efficiently than the Crossbar switching systems.

The Crossbar switching networks discussed previously have some limitations as described below:

- The number of Crosspoint will be the square of the number of attached stations and hence this is costly for a large switch.
- The failure of Crosspoint prevents connection with those two subscribers between which the Crosspoint is connected.
- Even if all the attached devices are active, only few of the Crosspoints are utilized. In

order to find a solution to subsidize these disadvantages, the multistage space division switches were built. By splitting the Crossbar switch into smaller units and interconnecting them, it is possible to build multistage switches with fewer Crosspoints. The following figure shows an example of a multistage switch.



The multistage switch like the above one needs less number of Crosspoints than the ones needed in Crossbar switching. According to the example shown above, for the 8 (input) and 8 (output) various subscribers (both called and calling subscribers), the Crosspoints needed in a normal Crossbar network will be square of them, which is 64. However, in the multistage Crossbar network, just 40 Crosspoints are enough. This is as shown in the diagram above. In a large multistage Crossbar switch, the reduction is more significant.

Advantages of a Multistage Network

The advantages of a multistage network are as follows:

- The number of Crossbars are reduced.
- The number of paths of connection can be more.

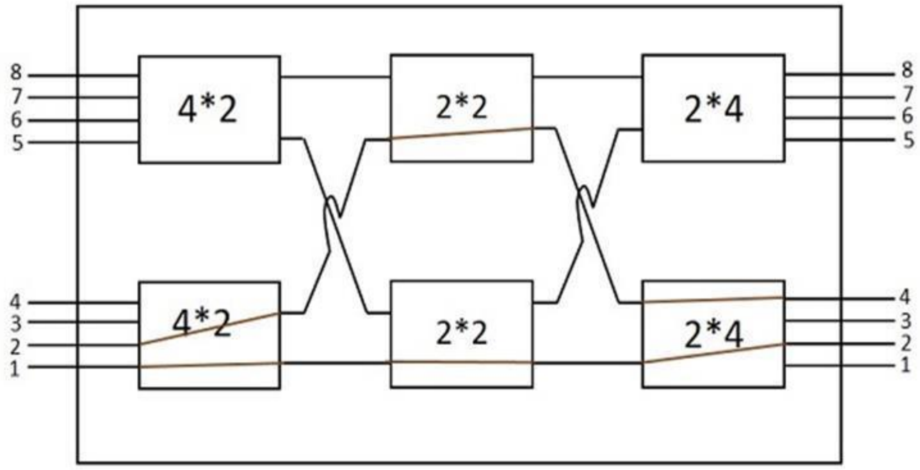
Disadvantages of a Multistage Network

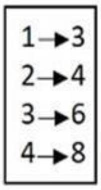
The disadvantage of a multistage network are as follows:

- Multistage switches may cause **Blocking**.
- The number or size of the intermediate switches if increased can solve this problem, but the cost increases with this.

Blocking

Blocking reduces the number of Crosspoints. The following diagram will help you understand Blocking in a better way.





In the above figure, where there are 4 inputs and 2 outputs, the Subscriber 1 was connected to Line 3 and the Subscriber 2 was connected to Line 4. The red-colored lines indicate the connections. However, there will be more requests coming; a calling request from subscriber 3 and subscriber 4 if made cannot be processed, as the call cannot be established.

The subscribers of the above block also (as shown in the above diagram) face the same problem. Only two blocks can be connected at a time; connecting more than two or all of the inputs cannot be done (as it depends on the number of outputs present). Hence, a number of connections cannot be established simultaneously, which is understood as the calls being blocked up.

Switching Techniques

In this chapter, we will discuss the switching techniques in Telecommunication Switching Systems and Networks.

In large networks, there may be more than one path for transmitting data from the sender to the receiver. Selecting a path that data must take out of the available options can be understood as **Switching**. The information may be switched while it travels between various communication channels.

There are three typical switching techniques available for digital traffic. They are:

- Circuit Switching
- Message Switching
- Packet Switching

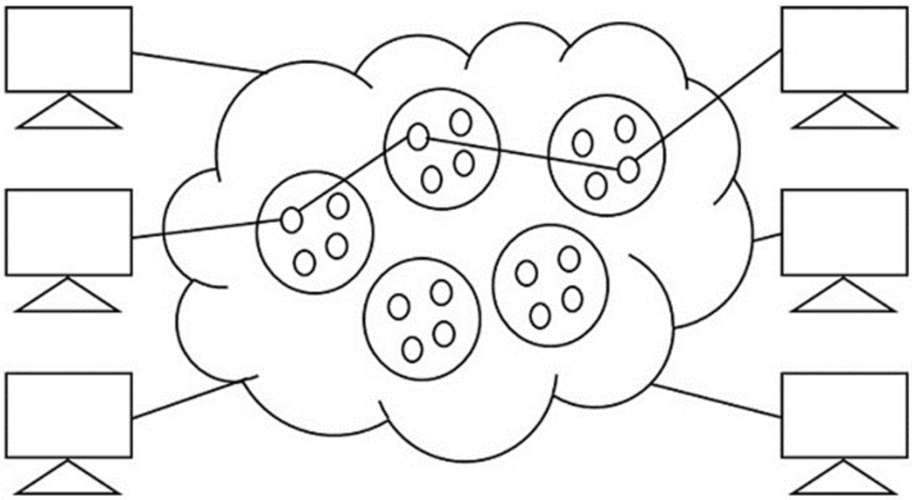
Let us now see how these techniques work.

Circuit Switching

In Circuit switching, two nodes communicate with each other over a dedicated communication path. In this, a circuit is established to transfer the data. These circuits may be permanent or temporary. Applications that use circuit switching may have to go through three phases. The different phases are:

- Establishing a circuit
- Transferring the data
- Disconnecting the circuit

The following figure below shows the pattern of Circuit switching.



Circuit switching was designed for voice applications. Telephone is the best suitable example of circuit switching. Before a user can make a call, a virtual path between the called subscriber and the calling subscriber is established over the network.

The drawbacks of circuit switching are:

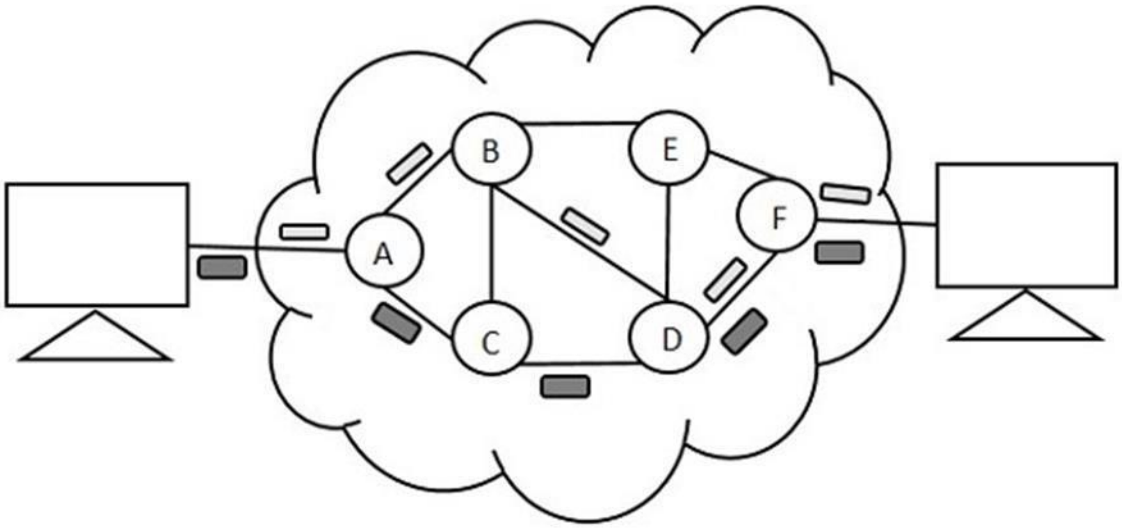
- The waiting time lasts long, and there is no data transfer.
- Each connection has a dedicated path, and this gets expensive.
- When connected systems do not use the channel, it is kept idle.

The circuit pattern is made once the connection is established, using the dedicated path which is intended for data transfer, in the circuit switching. The telephone system is a common example of Circuit Switching technique.

Message Switching

In message switching, the whole message is treated as a data unit. The data is transferred in its entire circuitry. A switch working on message switching, first receives the whole message and buffers it until there are resources available to transfer it to the next hop. If the next hop is not having enough resource to accommodate large size message, the message is stored and the switch waits.

The following figure shows the pattern of Message switching.



In this technique, the data is stored and forwarded. The technique is also called the **Store-and- Forward** technique. This technique was considered a substitute to circuit switching. But the transmission delay that followed the end to end delay of message transmission added to the propagation delay and slowed down the entire process.

Message switching has the following drawbacks:

- Every switch in the transit path needs enough storage to accommodate the entire message.
- Because of the waiting included until resources are available, message switching is very slow.
- Message switching was not a solution for streaming media and real-time applications.

The data packets are accepted even when the network is busy; this slows down the delivery. Hence, this is not recommended for real time applications like voice and video.

Packet Switching

The packet switching technique is derived from message switching where the message is broken down into smaller chunks called **Packets**. The header of each packet contains the switching information which is then transmitted independently. The header contains details such as source, destination and intermediate node address information. The intermediate networking devices can store small size packets and don't take many resources either on the carrier path or in the internal memory of switches.

Individual routing of packets is done where a total set of packets need not be sent in the same route. As the data is split, bandwidth is reduced. This switching is used for performing data rate conversion.

The figure below shows the pattern of Packet switching.

The line efficiency of packet switching can be enhanced by multiplexing the packets from multiple applications over the carrier. The internet which uses this packet switching enables the user to differentiate data streams based on priorities. Depending upon the priority list, these packets are forwarded after storing to provide quality of service.

The packet switching technique was proved to be an efficient technique and is being widely

used in both voice and data transfer. The transmission resources are allocated using different techniques such as Statistical Multiplexing or Dynamic Bandwidth allocation.

Statistical Multiplexing

Statistical multiplexing is a communication link sharing technique, which is used in packet switching. The shared linking is variable in statistical multiplexing, whereas it is fixed in TDM or FDM. This is a strategic application for maximizing the utilization of bandwidth. This can increase the efficiency of network, as well.

By allocating the bandwidth for channels with valid data packets, statistical multiplexing technique combines the input traffic to maximize channel efficiency. Each stream is divided into packets, and delivered on a first-come, first-served basis. The increase in priority levels allow to allocate more bandwidth. The time slots are taken care not to be wasted in the statistical multiplexing whereas they are wasted in time division multiplexing.

Network Traffic

As the name implies, network traffic is simply the data that moves along the network in a given time. The data transmission is done in the form of packets, where the number of packets transmitted per unit time is considered as load. The controlling of this network traffic includes managing, prioritizing, controlling or reducing the network traffic. The amount and type of traffic on a network can also be measured with the help of a few techniques. The network traffic needs to be monitored as this helps in network security; high data rate might cause damage to the network.

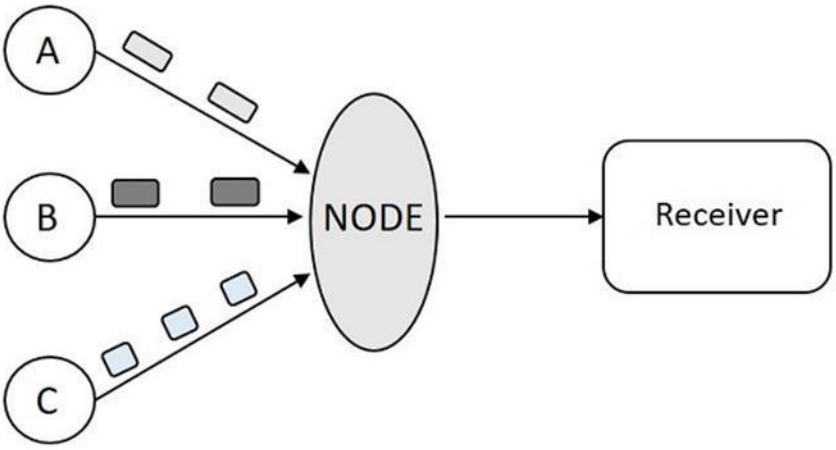
A measure of the total work done by a resource or facility, over a period (usually 24 hours) is understood as **Traffic Volume** and is measured in Erlang-hours. The traffic volume is defined as the product of the average traffic intensity and the period of the study.

$$Tai\,V = Tai\,Iiy \times Ti\,i$$

Congestion

Congestion in a network is said to have occurred when load on the network is greater than the capacity of the network. When the buffer size of the node exceeds the data received, then the traffic will be high. This further leads to congestion. The amount of data moved from a node to the other can be called as **Throughput**.

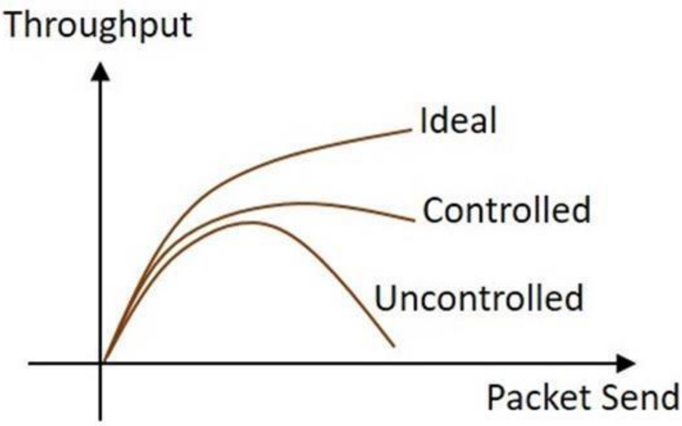
The following figure shows congestion.



In the above figure, when the data packets arrive at Node from the senders A, B and C then the node cannot transmit the data to the receiver at a faster rate. There occurs a delay in transmission or may be data loss due to heavy congestion.

When too many packets arrive at the port in a packet switched network, then the performance degrades and such a situation is called **Congestion**. The data waits in the

queue line for transmission. When the queue line is utilized more than 80%, then the queue line is said to be congested. The Congestion control techniques help in controlling the congestion. The following graph, drawn between throughput and packet send shows the difference between congestion controlled transmission and uncontrolled transmission.



The techniques used for congestion control are of two types – open loop and closed loop. The loops differ by the protocols they issue.

Open Loop

The open loop congestion control mechanism produces protocols to **avoid congestion.** These protocols are sent to the **source** and the **destination**.

Closed Loop

The closed loop congestion control mechanism produces protocols that allow the system to enter the congested state and then **detect** and **remove** the congestion. The **explicit** and **implicit** feedback methods help in the running of the mechanism.

UNIT-II

TIME DIVISION SWITCHING

In this chapter, we will discuss how the Time Division Switching works in Telecommunication Switching Systems and Networks.

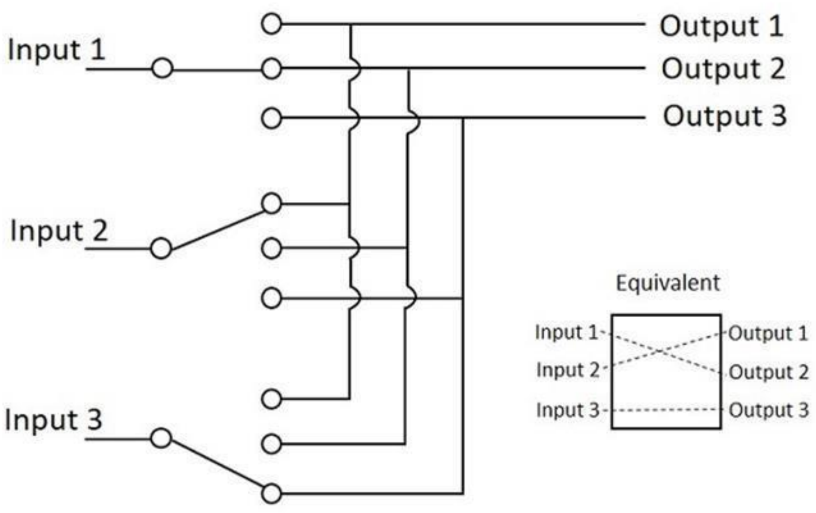
The switching scheme used by the electronic switching systems may be either **Space Division Switching** or **Time Division Switching**. In space division switching, a dedicated path is established between the calling and the called subscribers for the entire duration of the call. In time division switching, sampled values of speech signals are transferred at fixed intervals.

The time division switching may be analog or digital. In analog switching, the sampled voltage levels are transmitted as they are whereas in binary switching, they are binary coded and transmitted. If the coded values are transferred during the same time interval from input to output, the technique is called **Space Switching**. If the values are stored and transferred to the output at a late time interval, the technique is called as **Time Switching**. A time division digital switch may also be designed by using a combination of space and time switching techniques.

Space Division Switching

The paths in a circuit are separated from each other, spatially in space division switching. Though initially designed for analog networks, it is being used for both analog and digital switching. A Crosspoint switch is mostly referred to as a space division switch because it moves a bit stream from one circuit or bus to another.

The switching system where any channel of one of its incoming PCM highway is connected to any channel of an outgoing PCM highway, where both of them are spatially separated is called the **Space Division Switching**. The Crosspoint matrix connects the incoming and outgoing PCM highways, where different channels of an incoming PCM frame may need to be switched by different Crosspoints in order to reach different destinations.



Though the space division switching was developed for the analog environment, it has been carried over to digital communication as well. This requires separate physical path for each signal connection, and uses metallic or semiconductor gates.

Advantages of Space Division Switching

Following is the advantage of Space Division Switching:

- It is instantaneous.

Disadvantages of Space Division Switching

Following is the disadvantage of Space Division Switching:

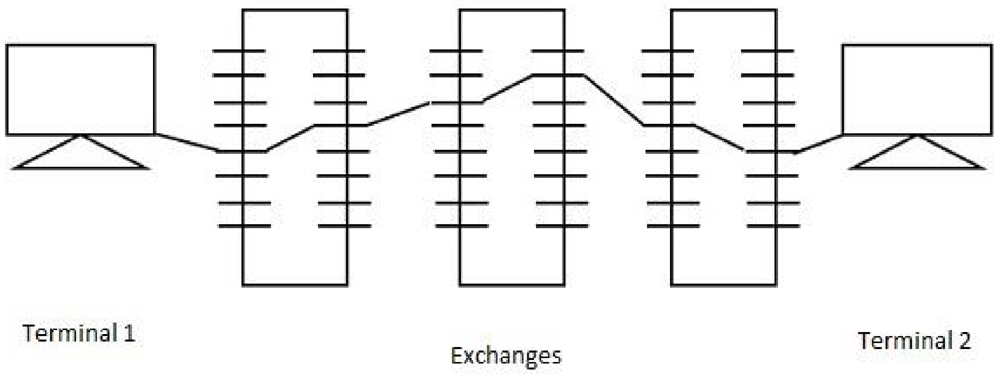
- Number of Crosspoints required to make space-division switching are acceptable in terms of blocking.

Time Division Switching

Time division switching comes under digital switching techniques, where the Pulse Code Modulated signals are mostly present at the input and the output ports. A digital Switching system is one, where the inputs of any PCM highway can be connected to the outputs of any PCM highway, to establish a call.

The incoming and outgoing signals when received and re-transmitted in a different time slot, is called **Time Division Switching**. The digitized speech information is sliced into a sequence of time intervals or slots. Additional voice circuit slots, corresponding to other users are inserted into this bit stream of data. Hence, the data is sent in time frames.

The main difference between space division multiplexing and time division multiplexing is sharing of Crosspoints. Crosspoints are not shared in space division switching, whereas they can be shared in time division multiplexing, for shorter periods. This helps in reassigning the Crosspoints and its associated circuitry for other connections as well.



Time division switches use time division multiplexing, in switching. The two popular methods of TDM are TSI (Time and Slot Interchange) and TDM bus. The data sent at the transmitter reaches the receiver in the same order, in an ordinary time division multiplexing whereas, in TSI mechanism, the data sent is changed according to the ordering of slots based on the desired connections. It consists of RAM with several memory locations such as input, output locations and control unit.

Both of the techniques are used in digital transmission. The TDM bus utilizes multiplexing to place all the signals on a common transmission path. The bus must have higher data rate than individual I/O lines. The main advantage of time division multiplexing is that, there is no need of Crosspoints. However, processing each connection creates delay as each time slot must be stored by RAM, then retrieved and then passed on.

Time Division Multiplexing

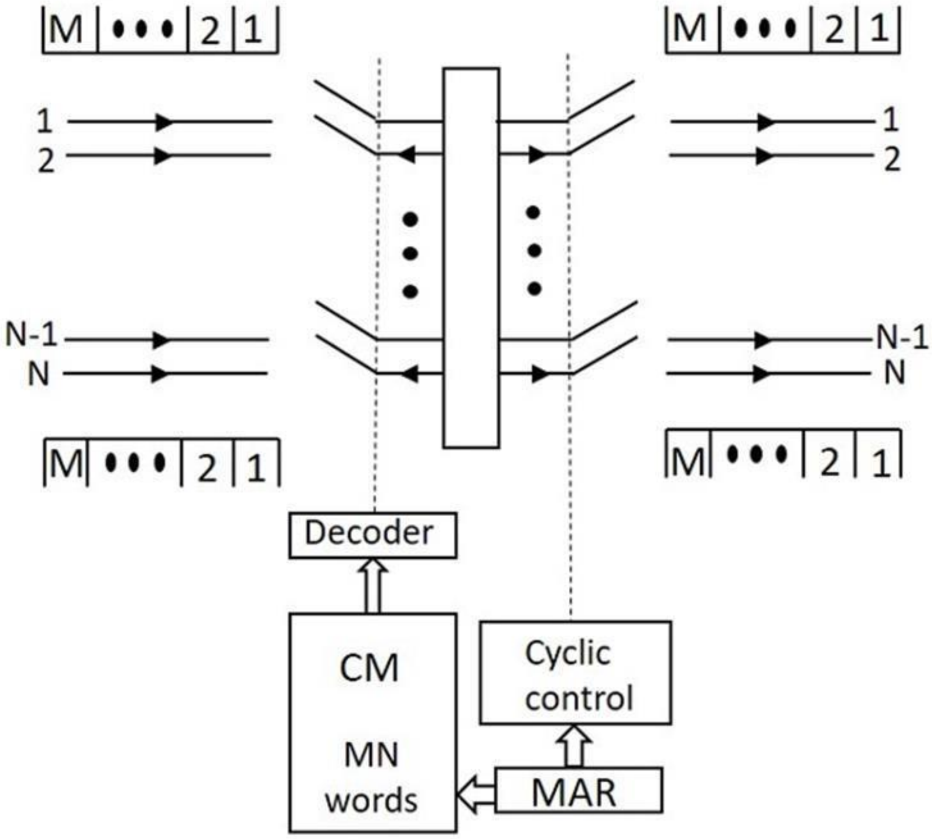
When the transmission of data or signals is done in digital means, using the limited number of resources available, then the Time Division Multiplexing is used for the transmission of such data. **Multiplexing** is the process in communication, which merges two or more signals at its input into a single output, which when de-multiplexed, offers all those signals separately as they were.

The Multiplexers are broadly classified as Analog and Digital, where the Time Division Multiplexing comes under Digital Multiplexing. There are two types of TDM called as Synchronous and Asynchronous TDM.

Time Division Space Switching

Time division switches may also employ space division switching techniques, whereas an appropriate mixture of both time and space division switching is advantageous in various circumstances.

A Time division space switch takes outputs of several time-division switches (say, TSI switches) which are then given as inputs to space division switches. This means that one of the two similar outputs produced by a TDM switch, can be selected by space switch to deliver to another output path which reduces the number of Crosspoints. The model of time division space switch is as shown in the following figure.



The interchange of time slots is not possible in time division switching, as the incoming time slot transfers the data to its dedicated output time slot only. Hence, time multiplexed switches do not provide full availability.

A time multiplexed Time Division Space Switch can be configured around a space array, which has M input horizontals and N output verticals. If both inputs and outputs are equal, M=N the switch leads to non-blocking. If inputs are greater than outputs; for concentrating switch we have M>N and if the outputs are higher, the switch expands gathering one more connection. In every time slot, one logic gate per vertical if M>N, or one logic per horizontal if M<N is enabled for one-to-one connections.

In every time slot, up to N or M samples are switched simultaneously. Because of the parallel transfer of N or M data samples in each time slot, a large number of channels can be multiplexed per input line. If along with multiplexing for N control memory modules, full availability has to be achieved, one should opt for time division time multiplexing technique.

Time Division Time Switching

The main advantage of time division time multiplexing technique is that, unlike time division space switching, it allows **time slot interchange** (**TSI**) of sample values. In TSI, a speech sample input during a time slot may be sent to the output during a different time slot, which implies a delay between reception and transmission of a sample.

The rate at which the time slot clock runs is 125-μ sec. The time slot counter increments by one, at the end of each clock pulse, the contents of which provide location addresses for data memory and control memory. The input sample is read at the beginning of the slot and it is clocked at the end of the clock pulse. Because of the storage action, the sample is delayed at least by a single time slot in passing from the input to the output, even if there is no time slot interchange.

A TSI which can be expanding or concentrating, has different number of time slots per frame at input and output also. For an expanding switch, the output bit rate is higher, whereas for a concentrating switch, the input bit rate is higher. The handling of input and output subscribers in this technique can be done in four ways, such as serial-in/serial-out, parallel-in/parallel-out, serial-in/parallel-out, parallel-in/serial-out.

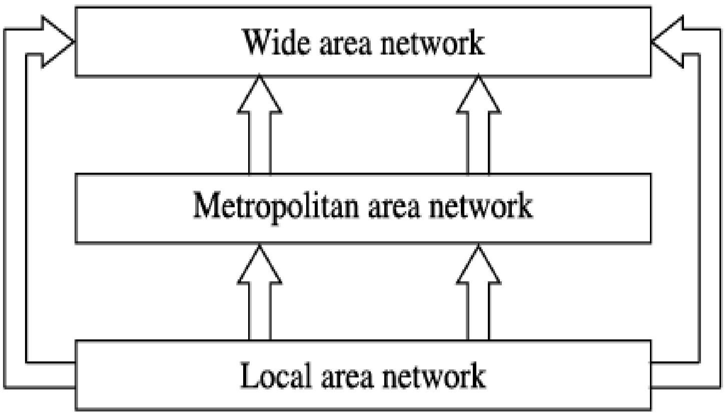
UNIT-III
DATA NETWORKS

Data networks are classified according to their geographical coverage:

- Wide area networks (WANs)
- Metropolitan area networks (LANs)
- Local area networks (LANs)

Intercity inter country and intercontinental networks are known as ‗WANs. Based on the communication infrastructure used, they may be classified as terrestrial data networks (TDNs) or satellite based data networks (SBDNs). In TDNs, data communication is organised using cables, fibre optic lines or radio links. A geo stationary or a geosynchronous satellite is used for Communication in SBDNs.

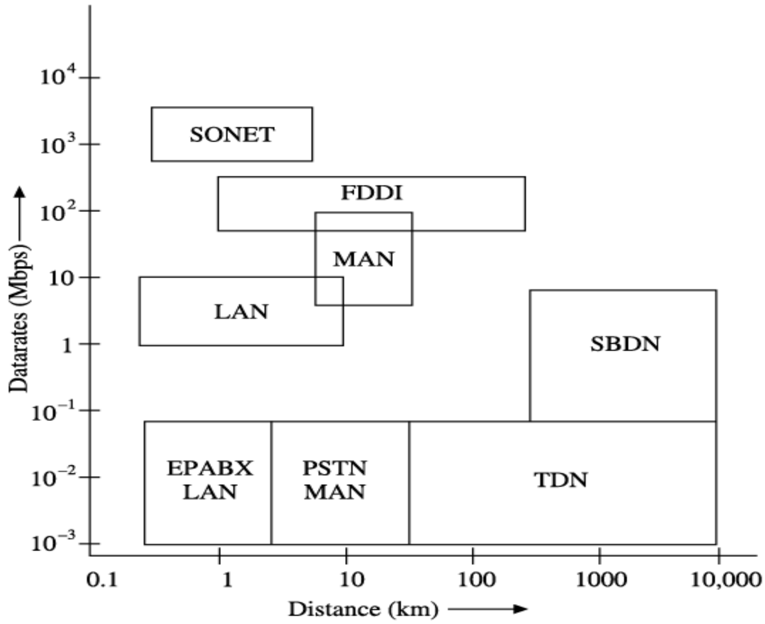
A metropolitan area network interconnects computers within a metropolitan city. Community antenna television (CAB‘) cables, twisted pair wires or shielded lines, optical fibres, radio links or line-of-sight (LOS) optical communication links provide the communication medium for MAN. The broadband capability of CATV cables permits carrying voice, data and video simultaneously. Thus MANs are usually multimedia networks. Local area networks are confined to a single building or a group of buildings generally belonging to the same organisation. Optical fibres, twisted pair and coaxial cables are used as the communication media for LANs. Fibre optic networks (FONs) are suitable for both LANs and MANs. Synchronous optical networks (SONET) are designed to operate at high speeds. LANs, MANs and WANs are generally interconnected in a hierarchical manner to form a global network as shown in Figure. LANs are often connected directly to WANs, particularly in places where MANs are not installed or have not developed well. Apart from the different geographical coverage, the range of data rates supported on these networks also different widely. Figure seminaries the typical data rates and geographical coverage for these networks.



Data network hierarchy

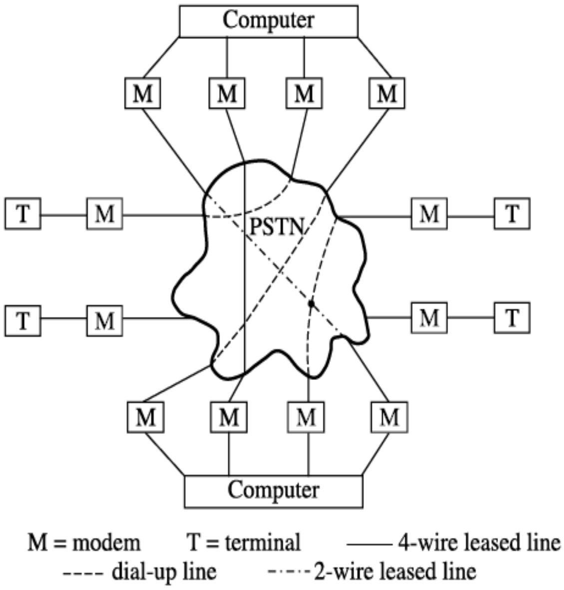
DATA TRANSMISSION IN PSTNS

Public switched telephone networks and electronic PABXs are designed to carry analog voice signals. They can however, be used for data transmission by employing suitable interfaces. As seen from Figure 10.2. LAYs can be designed around PABXS. and MANs around PSTNs. In these cases, the data rates are usually limited to a maximum of 64 kbps. Terrestrial data networks and the integrated services digital networks however, support data rates of 1.544 or 2.048 Mbp& modulator translates the data pulses into voice band analog signals at the transmitting end. At the receiving end, the analog signals are demodulated to recover the digital information. A
Combined modulator demodulator unit is called a modem.



Geographical coverage and speeds of data networks

Initially, modems were used to connect terminals, located in remote places, to a central computer. Later, computer-to-computer communication was established using modems and PSTNs. A data communication scheme using modems and a PSTN is shown in Figure 10.3. The digital interface of a modem is connected to the computer and the analog interface to the telephone network. In addition to data exchange, the digital interface permits control signals to be exchanged between the modem and the computer.



Data communication using PSTN.

In a PSTN, a connection may be established using a dial-up facility or a dedicate non witched leased line. A 2-wire or 4-wire leased line may be used. The modems usually have automatic dialling and answer facility which is useful when working with dial-up line. Modems are driven by a communication software package that runs on the terminal or the personnel computer of the user. The transmission may be half-duplex or

Data Rates in PSTNs

A voice channel in a PSTN is band limited with a nominal bandwidth of 3.1 kHz. What is the maximum data rate that a voice channel can support with its limited bandwidth? A first-cut estimate of this can be obtained from Nvquist‘s theorem which applies to noiseless channels and states:

$$R = 2H \log_2 V \text{ bps}$$

where

R = maximum data rate

H = bandwidth of the channel

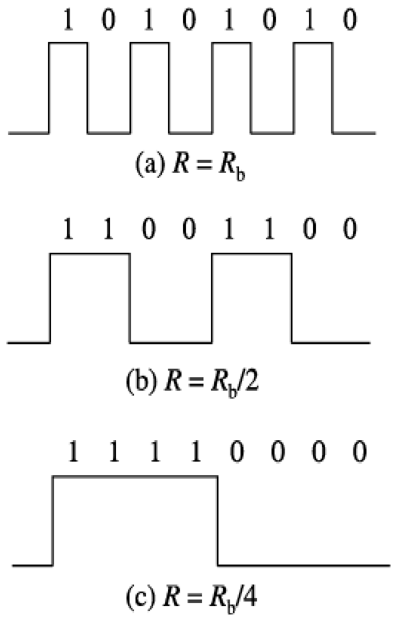
V = number of discrete levels in the signal

For a 3-kHz channel, and a binary signal, the maximum data rate works out to be 6000 bps, if the channel is ideal In a practical channel, the maximum rate would come down. By increasing the number of levels used to represent the signal, the bit rate may be increased arbitrarily in a noiseless channel. It is important to recognise that the actual number of signal transitions is still limited to the binary level limit; the effective bit rate goes up with more than two signal levels as each signal level can now represent a group of two or more bits. The maximum rate of signal transitions that can be supported by a channel is known as baud rate or symbol rate. In a channel where noise is present, there is an absolute maximum limit for the bit rate This limit arises because the difference between two adjacent signal levels becomes comparable to the noise level when the number of signal levels is increased. Claude Shannon extended Nyquist‘s work to the case of noisy channels affected by random or thermal noise. Shannon‘s result states:

$$R_b = H \log_2\left(1 + \frac{S}{N}\right)$$

where

R_b = the maximum bit rate obtainable
H = bandwidth of the channel
S/N = signal-to-noise ratio



Baud rates and bit rates.

Modems

Amplitude, frequency and phase modulation are all used in the design of modems. In amplitude modulation, zeros and ones are represented by two different voltage levels. A signal waveform s(t), called baseband signal, is generated from the digital data. This is then multiplied by a sinusoidal carrier, say cos (2tfot), to generate a modulated signal,

$$m(t) = s(t)\cos(2\pi f_0 t) \qquad (10.3)$$

At the receiver end, the modulated signal is again multiplied by $\cos(2\pi f_0 t)$, yielding a received signal

$$r(t) = s(t)\cos^2(2\pi f_0 t)$$

$$= \frac{s(t)}{2} + \frac{s(t)}{2}\cos(2\pi(2f_0)t) \qquad (10.4)$$

The high frequency component at 2tfo is then filtered out, leaving the demodulated signal s(t) from which the original digital signal may be attained. As mentioned in Section 101.1. it is not necessary that only two voltage levels are used when generating s(t). A new digital message may be chosen at every symbol interval, which is mapped to one of a set of discrete voltage levels. There is a one-to-one correspondence between the discrete message set and the voltage set. The voltage levels are then used to modulate the carrier. When more than two voltage levels are used, the modulation is known as M-ar. In M ary scheme, a message sample contains k bits. such that = M. For example, in a ló-arv scheme. each sample contains four bits of information.

While the message sample rate is the baud rate, the bit rate is four times the baud rate in this example. This technique of varying the amplitude of sinusoidal carrier using the voltage levels of the baseband signal is known as amplitude shift keying (ASK) If the carrier is sinusoidal and the baseband voltage level is used to vary the frequency or phase of the carrier, the modulation is known as frequency shift keying (FSK) or phase shift keying (PSK) respectively. In FSK, two or more different tones are used to represent zeros and ones, or a group of message bits. Binar: FSK modems operating at 1200 bps on dial-up lines are widely used for personal computer communications. Separate half-duplex channels are used for each direction. FSK has the advantage of constant amplitude signal, which makes it robust against the nonlinearities of power limiting devices in the channel.

SWITCHING TECHNIQUES FOR DATA TRANSMISSION

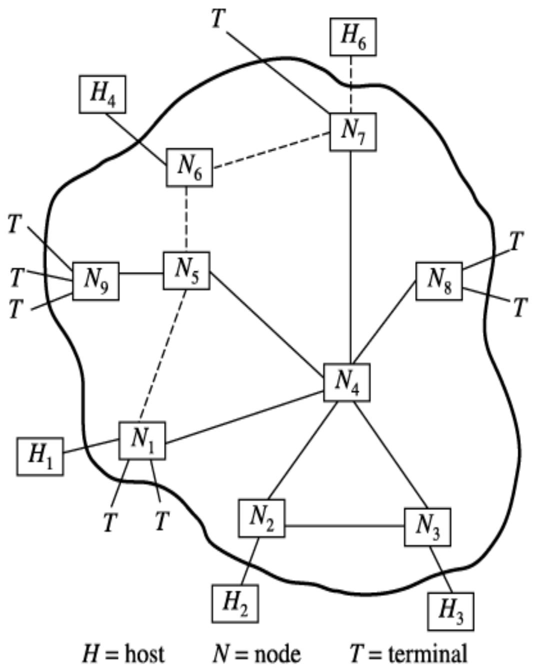
We saw how telephone networks are used to carry data. They are basically designed to carry voice traffic and there are some significant differences in the nature of voice and data traffic. Voice traffic is generally continuous (except for the silence periods in normal speech), whereas data traffic is bursty in nature. When a user sits at a terminal and works with a computer, interactively, he she spends tune thinking, keying in the query or command to the computer, and waiting for a response from the computer before proceeding further. In essence, the user is in one of three states thinking keying or waiting during thinking and aItm2 states no data transmission takes place. During the waiting state the computer is busy processing the user command. In order to have a low waiting time. the user input must be transmitted to the system expeditiously soon after the keying-in is over. Similarly, once the response is computed by the machine, the same must be transmitted to the user at a fast ratt Thus, the data traffic is generally bursty in nature and calls for high bandwidth for short duration This is true of the data traffic between two computers also.

In contrast, voice traffic needs low bandwidth (3.4 kHz) for long durations. Typically, the transmission line is idle for 85—95°ó of the holding time in the case of data transmission and is busy for a similar period in the case of a telephone conversation. (Pauses in normal speech are considered as active transmission periods.) While voice traffic is half-duplex, data traffic mr be half or full-duplex. Another important difference lies in acceptable error and loss rates- No errors or losses are acceptable in data transmission whereas a small amount of speech loss is often not noticeable. While the speech traffic always takes place in real time, the data traffic may or may not occur in real time. Interactive use of computers requires real time or near real time response, whereas file transfer between two computers need not be in real time. Table summarises the differences between voice and data traffic.

Voice traffic	Data traffic
Continuous	Bursty
Low bandwidth for long duration	High bandwidth for short duration
Typical line utilisation 85–95%	Typical line utilisation 5–15%
Half duplex	Half or full duplex
Real time	Nonreal time or near real time
Loss acceptable	Loss unacceptable
Error tolerable	Error unacceptable

Circuit Switching

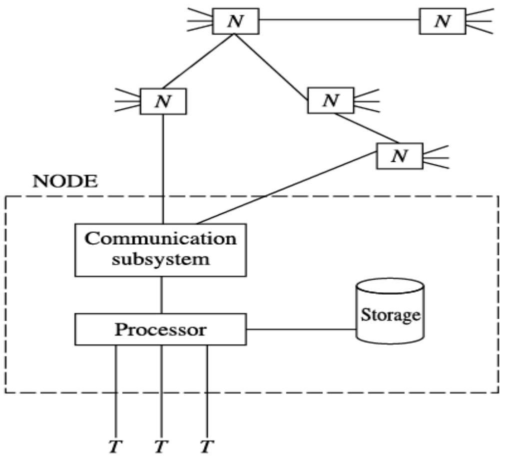
In circuit switching, an electrical path is established between the source and the destination before any data transfer takes place. The electrical path may be realised by physical wires or coaxial cables or radio or satellite links. It remains dedicated to the communicating pair for the entire duration of the transmission irrespective of either data is actually transferred or not. No other potential user can use the path even if it is idle. The connection is released only when specifically signalled so by either of the communicating entities. Data transmission using a PSTN connection is a typical example of a circuit switched data transfer. Figure illustrates the principle of circuit switching. When the host computer Hl wants to transfer data to the host computer H5, a connection request is made to the switching node Nl which. in turn, selects a suitable neighbouring node through which the desired connection may be established. say in this case node N5. Node N5 in turn, selects a suitable onward path and so on until an electrical path is established between Hl and H. The path selection is generally based on a routing algorithm that may take into account the network traffic, path length etc. Once a path is established, data transfer begins. There are three explicit phases involved in circuit switched data transfer.



Circuit switching Network

Store and Forward Switching

A store and forward (S&F) network configuration is shown in Figure 1(19. In S&F switching, the switching nodes have the ability to store user messages and forward the same towards the destination as and when the links become available. For this purpose, each node is equipped with a processor and some buffer storage. No end-to-end link is set up prior to data transmission. The user deposits his:her message to the nearest switching node and then on, thenetwork takes the responsibility for delivering the message to the destination user or host.
(Observe the analov to the postal system.)



The network moves the user information from node to node. One such movement is called a hop. Since the communication links are used one at a time between any two nodes, line speeds can be utilised efficiently. S&F switching may be classified as:

- Message switching
- Packet switching

In message switching, once the transmission is initiated, a message is transmitted in its entirely without a break from one node to another. The node processor performs the following functions:

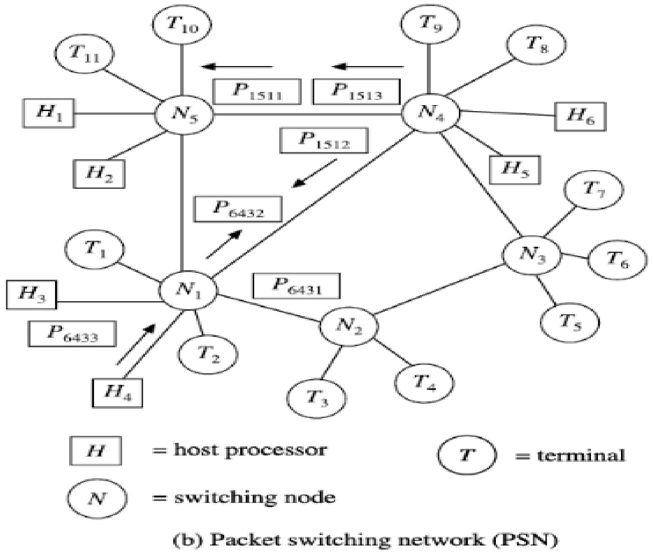
1. Receive the full user message and store the same.
2.Check the message for data transmission errors and perform error recover if required.
3. Determine the destination address from the user message.
4. Choose an appropriate link towards destination based on certain routing criterion.
5. Forward the message to the next node on the chosen link.

Message switching has certain drawbacks. For long messages, it becomes important to ensure that there is adequate storage space on the receiving node before the transmission is initiated. Otherwise the buffer storage may become full, and part of the message may not be stored, thereby requiring retransmission of the entire message. Similarly, if an error occurs during transmission, the entire message may have to be retransmitted. Retransmission of long messages results in large communication overheads in the network. If a high priority short message arrives while a long message is in transmission, it will have to wait until the transmission of the long message ends. Such drawbacks are overcome in packet switching. In packet switching, messages are split into a number of packets, often fixed in size, and the packets are transmitted in a S&F fashion. Messages are split at the source host and reassembled at the destination host. Each packet transmission is independent of the others. The packets of a single message may travel via different routes and arrive at the destination with different delays.

This may lead to the situation where the packets of the same message arrive out of sequence at the destination node. Every packet needs to carry the complete address information, viz. destination identifier (id), source id, message id, and packet id, in addition to the actual user data A typical packet format is shown in Figure .A packet switching schematic is shown in Figure .In this schematic, a packet is numbered using a four subscript quantity where the first subscript is the destination host id, the second the source host id, the third the message id, and the fourth the packet id of the message .For example, P6432 indicates that this is the second packet of the third message originating from the host 4 and destined to host 6. It may be observed that the source host delivers the packets of a message in sequence to the network node and it is natural to expect that the packets are delivered to the destination host in proper sequence. However as the packets may arrive out of sequence at the destination node, it becomes the responsibility of the network to sequence the packets before deliver to the destination host. This calls for considerable overhead in terms of buffer storage and processing power at the network nodes and hence turns out to be somewhat an expensive service. In order to be cost effective, packet networks offer two different forms of services**:**

Header					User data
Destination id	Source id	Message id	Packet id	Control	

(a) A typical packet format



(b) Packet switching network (PSN)

Data communication among computers involves a number of functions such as physical transmission of bits, error control, and routing and session establishment. In order to efficiently implement these functions, vendors of computer systems evolved their own architectures.

Examples of vendor specific architectures are System Network Architecture (SNA) of IBM and Digital Network Architecture (DNA) of Digital Equipment Corporation (DEC). Such architectures permit interconnection of computers from the same vendor but not from different vendors. Systems or networks, which are not open to other vendor systems for networking, are known as _closed‘ systems or networks. In order that heterogeneous computer systems from different vendors be interconnected as a network. an architecture which is used as a standard by all the vendors is required. The heterogeneity covers the following aspects:

1. Systems of different vendors

2. Systems under different managements

3. Systems of different complexities

4. Systems of different technologies.

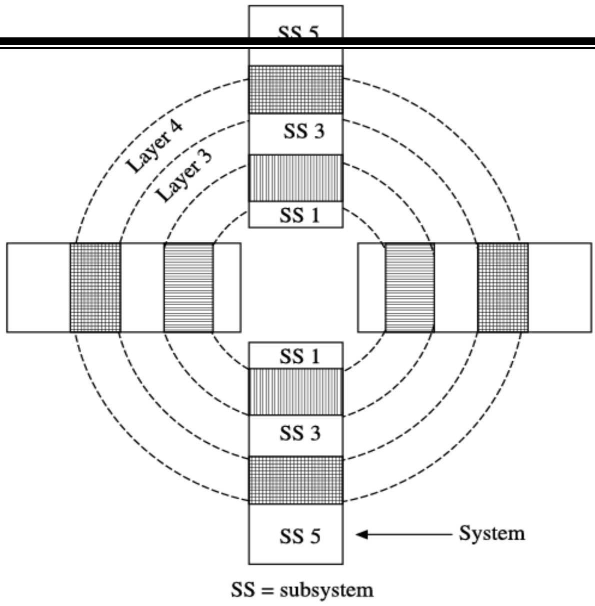
ARPANET. the network project supported by Advanced Research Projects Agency of the Department of Defence, United States, was one of the pioneering efforts in interconnection heterogeneous systems. The efforts put in and the experience gained in the project significantly contributed the emergence of a world standard architecture for computer communication, largely pursued and set out by International Standardisation Organisation (ISO). These standards now wel known as ISO-Open System Interconnection (ISO-OSI) standards, are widely accepted. The standards are based on a reference architecture which is described in the ISO standard IS 7498. CCITT has also adopted this standard under its own number X.200. The architecture is considered _open‘ as any vendor‘s system conforming to this reference model is capable of organising information transfer with any other vendor‘s system which also conforms to the same architecture.

ISO-OSI Reference Model

Before we discuss this model. a few definitions are in order

System: A system is one or more autonomous computers and their associated software, peripherals and users, which are capable of information processing and or transfer

Subsystem: A logically independent smaller unit of a system A succession of subsystem make up a system as shown in Figure.



Systems, subsystems and layers in ISO-OSI model.

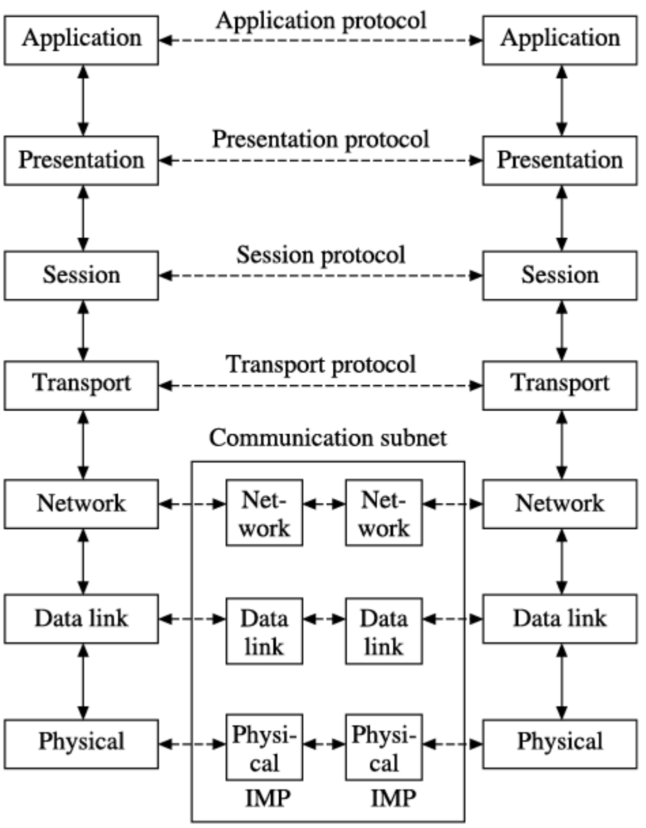
Layer: A layer is composed of subsystems of the same rank of all the interconnected systems. The concept of a laver is illustrated in Figure. Which shows a five-layer network The subsystems and the layers are numbered starting with one at the bottom level.

Entity: The functions in a laver are performed by hardware subsystems and or software packages. These are known as entities. ISO-OSI architecture is a layered one. Layering is a natural choice for communication architectures. This is well illustrated by an example. Consider the activities that are involved when executives A and B of two companies in different cities want to converse over a trunk telephone connection. For the sake of illustration, let us assume that there is no subscriber trunk dialling (STD) facility between the two cities. Let executive A be the calling party. As a first step, he requests his secretary to connect him to executive B. His secretary in turn calls up the trunk operator and communicates the calling number. called number, nature of the call the name of the particular person called etc. Then, the local trunk operator calls up the other trunk operator in the other city and communicates the details. Remote trunk operator now calls up the secretary of executive B. who in turn confirms with executive B that he would like to receive the call and requests the operator that the call be put through. This process is depicted in Figure.

.

1.A three-layer structure is used in this communication process

2. The conversation between an upper and lower laver is strictly business like.
3. There is generally a little private conversation between the trunk operators and the two secretaries on account of their familiarity. In other words entities in the same level or laver exchange information using their own private protocols.
4. A layer obtains services from its immediate lower layer and provides services to its immediate upper layer. In this sense, a layer acts both as a user as well as a service provider.
5. There are fairly well defined functions to be performed by each layer.
6. It is immaterial as to how the functions of each laver are implemented. For example, the secretary may ask his assistant to book the call and as far as the executive is concerned, it is immaterial who books the call as long as the call is booked.



ISO OSI reference model.

Entities in these layers always communicate with peer entities in the adjacent system. In other words, in the first three layers, the communication proceeds on a link- by-link basis. In contrast, entities in layers 4 —7 communicate with peer entities in the end systems. There is no communication with entities in the intermediate systems. In this sense, layers 4 —7 are often called end-to-end layers. The 7-layer architecture has been arrived at after a careful application of a broad set of layering principles. The important principles are:
1. Create layers to handle Functions which are manifestly different in the process performed or technology involved.
2. Collect similar functions into the same layer and create a boundary at a point where the number of interactions across the boundary are minimised.

3. Create a layer of easily localised functions so that the layer could be totally redesigned and its

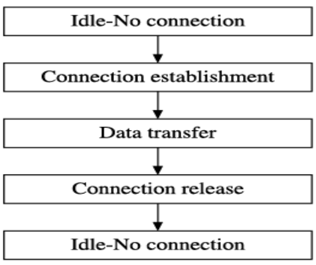
protocols changed in a major way to take advantage of new advances in architectures. Hardware and software technology without changing the services offered or the interfaces with the adjacent layers.

LINK-TO-LINK LAYERS

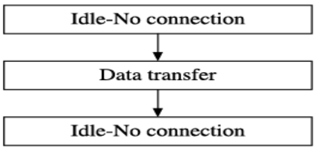
The first three layers, viz. physical, data link and network layers, form the link-to-link layers of OSI reference model Entities in an 051 layer perform certain functions to fulfil the stated purpose of the layer. They obtain services from the immediate lower layer and provide services to the immediate upper laver. OSI services may be placed under two broad categories:

- Connection oriented services
- Connectionless services.

In connection oriented services, a connection is first established between the sender and the receiver before data transfer can commence. The connection may be virtual (logical) or physical, depending upon the network capabilities and facilities. The essence of a connection oriented service is that a connection acts like a tube or a pipe delivering the data to the receiver strictly in the same order in which the data was put into the connection by the sender. Connection oriented service is modelled after the telephone system. In contrast connectionless service is modelled after the postal system. Each submission by the sender is treat independently of others and is sell-contained with the full address of the destination and the source indication which may be the full address too. In connectionless service: when two messages are sent to the same destination one after another, it is possible that the fast one is delayed and the second one arrives first. Datagram service and the virtual circuit service discussed in Section are examples of connectionless and connection oriented services respectively. The operation of the two categories of services is shown in Figures (a) and (b). A connection oriented service has provision for acknowledgements, flow control and error recovery; whereas a connectionless service does not generally have such provisions. Peer entities of OSI layers communicate using peer protocols. Protocols are strict procedures and sequence of actions to be followed in order to achieve orderly exchange of information among peer entities. Corresponding to the two categories of services, there are two sets of protocols; one set for the connection oriented services and the other for connectionless services. It is important to recognise that layer protocols relate to the implementation of services of the laver and therefore are not visible to the users or other layers. This separation of services and the protocols provides complete freedom to change protocols at will without affecting the services.



(a) Connection oriented service



(b) Connectionless service

Physical layer

It is essential that the OSI architecture permits the usage of a realistic variety of physical media and control procedures. Keeping this in mind, the lowest layer of the architecture has been identified as the physical laver This laver performs functions associated with the activation and deactivation of physical connections. It deals with encoding decoding of signals and the bit level transmission of electronic signals throu2h the available transmission medium The transmission may be synchronous or asynchronous. Mode of data transmission may be simplex, half duplex or full duplex. Transmission and data encoding schemes have been discussed in detail in Chapters 5 and 7 for cables and optical fibres respectively. The physical layer provide mechanical. electrical, functional and procedural characteristics to activate, maintain and deactivate physical connections for transmission of bits.

Data Link Layer

Some physical communication media like telephone lines have error rates that are not acceptable for the great majority of data network applications. Special techniques are required to ensure error free transmission of data. The data link laver deals with error detection and automatic recover procedures required when a message is lost or corrupted. For this purpose, a user of this layer, ic. the network layer, is required to break up the data to be transmitted into frames which are then numbered and transmitted sequentially. The layer provides functional and procedural means to establish, maintain and release data link connections for the entities in the network layer. A data link connection may be built upon one or several physical connections. Another important function performed by the data link layer is the link level flow control of frames. Flow control is essentially a traffic regulation mechanism that will have to be enforced when the receiver is unable to accept frames as fast as the transmitter is able to send. A data link may be of point-to-point type as in the case of terrestrial networks or broadcast type as in the case of SBDNs, LAYs or MANs. In the case of broadcast type channels, the data link laver will have to perform an additional function of acquiring or accessing the channel before data transmission can be gut Media access techniques for satellite channels are discussed in Section for LAN configurations in Section and for MANs In this section, we confine ourselves to point-to-point links. In this case, the media access is an insignificant operation as there are no contenders for the channel Hence, our discussions mainly concentrate on efficient channel utilisation, error recovery and flow control mechanisms.
These discussions are also applicable for data transmission in broadcast type channels, once the channel is acquired. The main sources of error in a system are:

1. Thermal noise which is internal to the system
2. Impulse or spike noise which originates from man-made sources like automobiles or signalling in telephone systems, and from natural sources like lightning
3. Crosstalk which occurs through electromagnetic radiation from parallel and adjacent wires which behave like an antenna. Errors are of two types: isolated and bursty. Errors due to thermal noise are generally isolated. whereas errors due to spike noise or crosstalk are burstv in nature. Error control mechanisms are chosen depending upon the type of error that is predominant in a given system. There are three error control mechanisms that are commonly used:

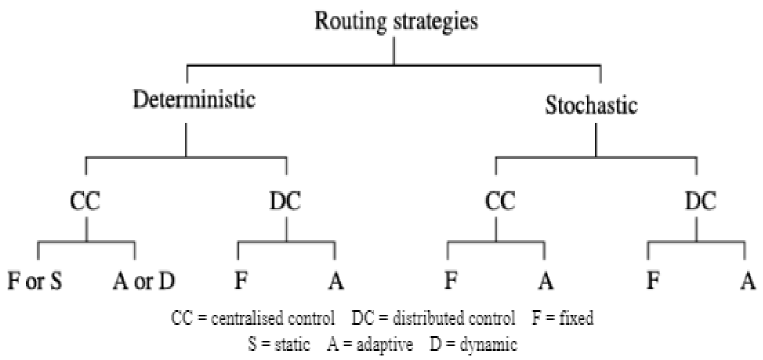
1. Echo checking

2.Forward error correction (FEC)

3. Automatic repeat request (ARQ).

Network Layer

The highest link-to-link layer in the OSI model is the network layer. Although this layer functions on a link-to-link basis, it is concerned with transmission of packets from the source node to the destination node. It deals with routing and switching considerations that are required in establishing a network connection which may involve the use of several transmission resources in tandem. including a number of intermediate switching nodes of different sub networks. The network laver makes invisible to the transport laver, the details of the underlying communication media and the different characteristics of the transmission and network technologies. It only assures a certain quality of service to the upper layers. Since an end-to-end connection may involve routing through a number of different networks, internetworking is an important function of the network layer. Addressing schemes, network capabilities, protocol differences, and accounting and billing are all issues to be handled in internetworking. Network congestion, which may occur due to too many messages on a particular route, is also tackled by the network layer.



Classification of routing algorithms.

A number of measures may be used in assessing the performance of a routing algorithm:

1. Minimum delay
2. Minimum number of intermediate nodes or hops
3. Processing complexity
4 Signalling capacity required on the network
5. The rate of adaption in the case of adaptive algorithms
6. Fairness to all types of traffic
7. A reasonable response time over a range of traffic intensities
8. Robustness: the ability to reach the destination even when parts of the network fail
9. Stability: the ability to reach the destination quickly without wandering.

END TO END LAYERS

Many data networks were operational before the OSI model was designed. These networks were well thought out up to the network laver, but little had been done about the transport layer and above. As a consequence, the design of the bottom three layers of OSI vas highly the flounced by pre-OSI developments.

Transport Layer

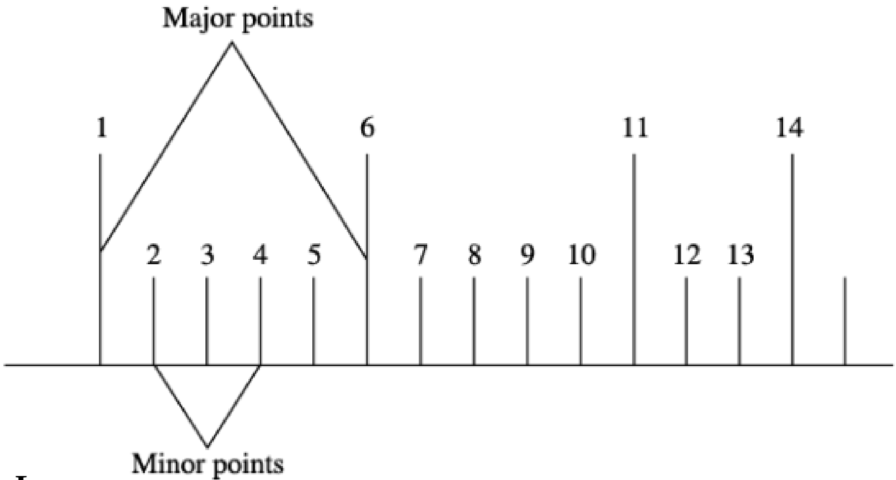
Transport layer is the first end-to-end layer in the OSI architecture. It is responsible for matching user message characteristics and service requirements with that of the network capabilities. In a packet switched network, the transport entity breaks up a long user message into packets to match the network capabilities. The packets are reassembled at the destination transport subsystem to reconstruct the user message. The transport entities may invoke sophisticated error control protocols to provide a reliable session service on an unreliable network. Similarly a number of low rate user services may be multiplexed to use efficiently a single network connection, or a high data rate requirement from the user may be split into a number of network connections. Multiplexing and splitting, performed by the transport layer are transparent to the sessions layer. Like every other layer, transport laver is also concerned with the establishment control and release of the transport connections between peer entities in the source and destination systems. End-to-end flow control and error recovery are also functions of the transport layer. The need for flow control arises when the speed or the buffer space of the destination machine does not match with that of the source machine. Similarly, end-to-end error recovery becomes necessary when the source or destination system fails or when the network becomes disconnected due to link failures.

As far as a user is concerned, it is the transport layer that offers transport services regard less of the underlying subnetvork. The user makes his sen-ice requests to this layer by specifying certain ‗quality of service' (QOS) parameter values. Some QOS parameters that are of direct interest to the users are:

1. Transit delay
2. Residual error rate
3. Protection
4. Transfer failure probability
5. Priority
6. Throughput.

Session Layer

The main function of the session layer is to organise different sessions between cooperating entities and perform all related functions like synchronisation, failure management, control, etc. for the successful execution of a session. Online search of databases, remote job entry, remote login to a time sharing system and file transfer between two systems are all examples of sessions. Different sessions have different requirements. For example, a dialogue session may be two-way simultaneous or one-way alternate. A large file transfer session may call for rollback points being established in order to recover from stem crashes. An online transaction processing session calls for semaphore management, and file, record and sometimes even item level lock mechanisms. A quarantine service is one which enables a specified number of presentation layer SDUs to be transported to the destination system but not actually deliver them unless explicitly so requested by the sender. The session layer may also offer a director service.



Synchronisation points in session layer.

Presentation Layer

The purpose of the presentation laver is to represent information to the communicating application entities in a way that preserves the meaning while resolving syntax differences. Syntax differences are resolved by encoding application data into a standard abstract notation that is valid throughout the network. Thus, file format differences (e.g. IBM or DEC format), data representation differences (e.g. ASCII or EBCDIC) or data structure differences are all resolved by using a standard notation. Data transformation and formatting may include data compression, encryption etc. There are two aspects associated with network wide handling of a variety of data in a standard form. First, the representations of the data in a standard form, and second, the transmission of the data as a bit stream across the network.

Identifier	Length	Data	Flag

Figure 10.26 ASN.1 transfer syntax.

1. Universal
2. Application
3. Private
4. Context specific.

Application Layer

As the highest layer in the OSI reference model, the application laver provides services to the users of OSI environment. The laver provides all services that are directly comprehensible by the users. Which include

1. Electronic mail or message handling sen-ices

2. Directory services

3. Cost allocation

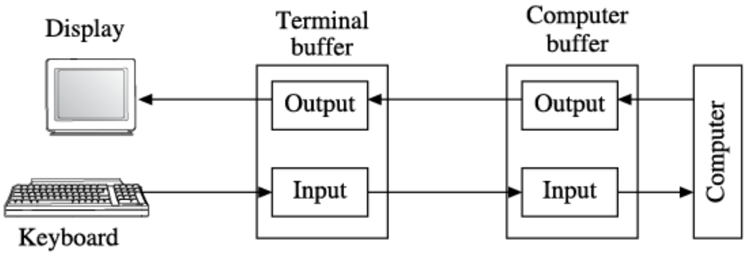
4. Determination of quality of sen-ice

5. File transfer and management

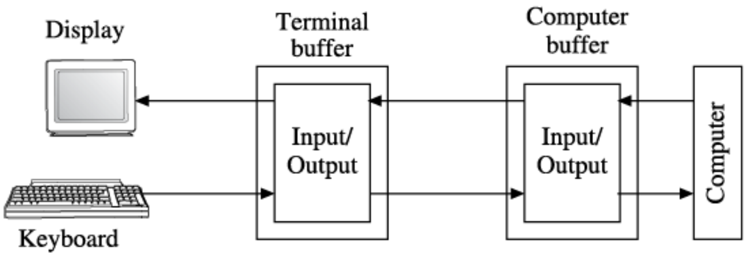
6. Editors and terminal support services

7. Telematic services like videotex.

In general, every application requires its own software which in turn uses a number of generic supporting packages. These generic packages and the application specific packages are part and parcel of the application laver. What is listed above is a set of generic applications and support packages. For example, file transfer or remote file access may be used by airlines reservation system. banking applications etc. Similarly, electronic mail may be used in order



(a) Independent buffers for input and output: asynchronous mode



(b) Common buffer for both input and output: synchronous mode

Virtual terminal data structure models.

SATELLITE BASED DATA NETWORKS

In the context of data networks and the 051 reference model, there are some important aspects of satellite communication 5v stems which require consideration:

1. Satellite network topology and confutations modulation schemes and bandwidth utilisation; these are aspects related to the physical layer functions of the reference model.
2. Being a common communication resource accessible by all or a group of earth stations simultaneously. media access becomes a nontrivial function in the data link layer.
3. Satellite communication being broadcast in nature, routing becomes a trivia function; however. organising point-to-point or point-to-multipoint connections in a broadcast.
4. Since a geostationary communication satellite is placed at an altitude of about 36,000 km above the equator, the signal will have to travel a distance of 72,000 km or more between the source and the destination, resulting in a significant propagation delay of 250 — 300 ms. Session and transport layer have to be concerned about this factor.

LAN

A local area network (LAN) typifies a distributed environment and finds applications in a number of areas. Some examples are:

1. Office automation

2. Factory automation

3. Distributed computing

4. Fire and security systems

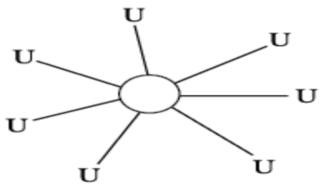
5. Process control

6. Document distribution.

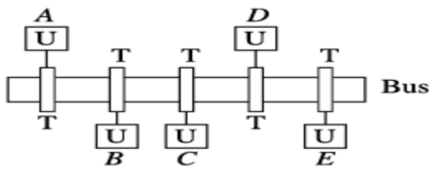
Of these, the first two applications have evolved to the extent that world standards for these are emerging. Office automation standard. known as Technical and Office Protocol (TOP), and the factory automation standard, known as Manufacturing Automation Protocol (MAP), are discussed in Section 10.11. Being distributed in nature, LANs offer a number of benefits and for the same reason suffer from certain disadvantages too.

The advantages offered by the LANs are:

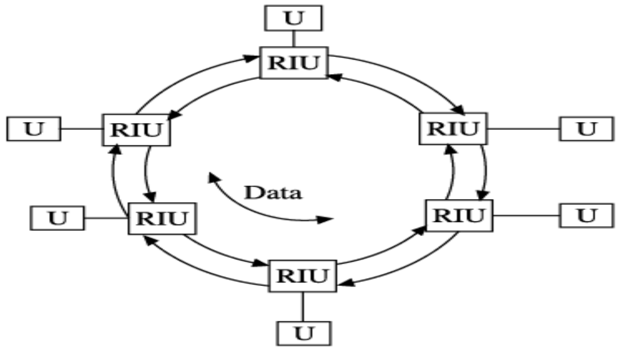
1. Unlike a large centralised system, a LAN may evolve with time. It may be put into operation with a small investment, and more systems may be added as the need arises.
2. Since LAN is a set of multiple interconnected systems, it offers a good back up capability in the event of one or two systems failing in the network. This, in turn, enhances the reliability and availability of the systems to users.
3. LAN provides a resource-sharing environment. Expensive peripherals, hosts and databases may be shared by all the LAN users.
4. A LAN adhering to a certain standard permits multivendor systems to be connected to it. Thus, a user is not committed to a single vendor.
5. In LAN, the systems are generally so chosen as to meet most of the user



(a) Star topology



(b) Bus topology



RIU = ring interface unit
(c) Ring topology

Three access methods are predominantly used in LANs: switched access, contention or multiple access, and token passing access. Switched access is used in LANs that are designed around CBXs. Electronic switching systems have been extensively discussed in Chapters 4 and 6. In this chapter, we are concerned only with contention and token passing access techniques. There are also other access techniques like reservation access, load adaptive access, and tree structure based access. Discussion of such schemes is beyond the scope of this text.

Different types of LAYs are obtained by choosing different combinations of medium. topology, and access method. If we consider three media, viz, twisted pair, coaxial and optical fibre, three topologies, viz, star, bus, and ring and two access methods, viz, contention and token passing, we would have a total of 18 different types of LAN. But not all of them are technically feasible or practically possible. For example, there are still unsolved engineering problems with regard to the use of optical fibres as bus media. Token passing access is a natural choice on the ring topology although contention access is technically feasible. As a result, only three combinations of access techniques and topologies are popularly used:

1. Multiple access bus

2. Token passing ring or token ring

3. Token passing bus or token bus.

METROPOLITAN AREA NETWORKS

A metropolitan area network (MAN) usually covers a geographical area spanning a distance of 5—50 km. In functionality MANs support services that require guaranteed bandwidth and bounded delay performance, in addition to data services that do not pose such restrictions. Additional services may include voice and video: depending on the available bandwidth. MANs may operate at speeds of 1 Mbps and above although a more common range is 50—150 kbps. Network structures to MAN are similar to the ones used in LANs: star, bus and ring. In this sense. the)LANs may be considered as an extension of LANs. However, the need to support services like voice and video renders many of the LAN access methods unusable in MAN. The large distances involved in LAN affect the performance adversely if LAN access methods are used.

FIBRE OPTIC NETWORKS

Optical fibre networks are characterised by

1. High speed operation (typically loo Mbps or more)

2. Ability to span large distances (100—200 km)

3. Ability to support a moderate number of stations; typically 10 to a few hundred stations are supported with a maximum limit around 1000.

These characteristics make the fibre optic networks suitable for high speed LASs and MANs with a limited number of stations. Fibre networks may be configured around a star. ring or bus structure. The number of stations that can be supported in a star or a bus structure is relatively low compared to that in a ring configuration. Optical fibres are inherently unidirectional and this influences the way in which the network structures are realised, and the consideration for medium access- Some of medium access considerations discussed in the context of MANs in Section are also applicable to fibre optic networks (EONs).

The central hub in a star FON may be passive or active as shown in Figure.. The passive hub is usually a silica cylinder. The incoming fibres are fused to one end of the cylinder and the outgoing fibres to the other end. The light emitted by the transmitters (LEDs) of the stations diffuses inside the passive hub illuminating all the receivers (,photodiodes). The incoming energy is divided among all the outgoing lines. The output power of the transmitter and the sensitivity of the receiver would determine the number of nodes that can be connected to the passive hub. The limited number of stations in passive-star FONs permits a simple medium access protocol to be used. Following the end of a transmission, a fixed number of time slots usually equal to or twice the number of stations, is defined as illustrated in Figure. Each station is assigned one or two slots, as the case may be, in which it may start transmission provided no other station prior to it has started transmission. In Figure. The station assigned to slot 5 (station 5) starts transmission and all stations after slot 5 will have to wait until the end of transmission by station 5, when the cycle repeats. Obviously;; stations allotted to earlier slots attain priority over other stations, resulting in an unfair access scheme. To bring about fairness in access. iV slots are used where a station is allotted,

DATA NETWORK STANDARDS

Three major international bodies have been sigrnficantlv contributing to the data network The bodies, their major area of concentration, and the standard series identifiers are

Body	Main thrust areas	Series
ISO	OSI reference model; end-to-end layers (4–7)	7xxx 8xxx 9xxx 10xxx
CCITT	Link-to-link layers (1–3) of wide area networks: PSTN, PDN and ISDN; electronic messaging and directory services	V series X series I series
IEEE	Link-to-link layers (1–3) of local and metropolitan area networks	802.x

Standards laid down by ISO and CCITT have international legal standing, whereas IEEE standards have to be adopted by ISO to attain this legal standing. Standards work at IEEE is sponsored by ANSI which is affiliated to ISO. Standards evolved by CCITT are adopted by ISO under its own series number and vice versa. For example, ISO basic OSI reference model IS 7498 is adopted by CCITT as X.200. Similarly, X.400 message handling system of CCITT is adopted by ISO under the name message oriented text interchange system (MOTIS)‖ as IS 1002L The X4OO message handling system is described in Section .While adopting standards these bodies may introduce minor changes.

UNIT-IV
TELEPHONE NETWORKS

Public switched telephone network (PSTN) or the plain old telephone system (POTS) is perhaps the most stupendous telecommunication network in existence today. The length of telephone Wire-pairs buried underground exceeds a billion kilometres. A unique feature of the telephone network is that every piece of equipment, technique or procedure which has evolved in the last lOO years from a number of different giant corporations, is capable of working with each other. The enormous complexity of the telephone network is managed by using a hierarchical structure. worldwide standardisation. and decentralisation of administration. operation and maintenance.

Any telecommunication network may be viewed as consisting of the following major

systems:

1. Subscriber end instruments or equipments

2. Subscriber loop systems

3. Switching systems

4. Transmission s:cstems

5. Signalling systems.

Telephone instruments and switching systems have been dealt with in Chapters Fibre optic transmission system has been discussed in Chapter 7. In this chapter, we discuss other transmission systems, subscriber loop systems, signalling systems and network management related aspects.

In this chapter, we will learn about the Public Switched Telephone Network (PSTN). This extraordinary telecommunication network is counted as one of the achievements in the field of technology advancement. However, there come a few problems when we come to these networks. We will discuss these problems in our subsequent sections.

PSTN

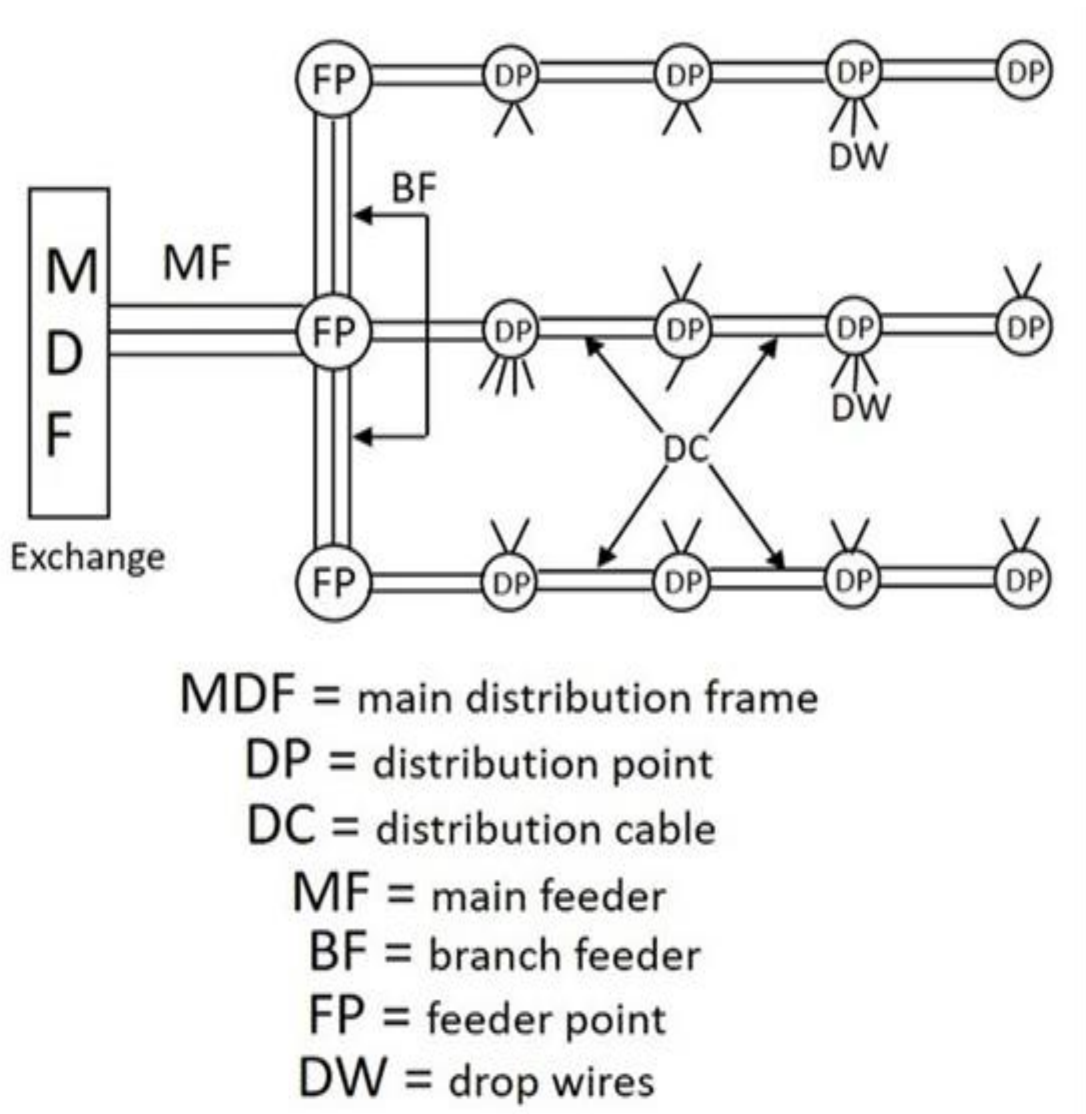
The Public Switched Telephone Network is understood as an aggregate of world‘s circuit switched telephone networks, used for providing public telecommunication. The PSTN networks are called POTS (Plain Old Telephone Systems). These networks are operated regionally, locally, nationally and inter-nationally using telephone lines, fiber optic cables, microwave transmission links or cellular communications.

PSTN consists of switches at centralized points on the network, which act as nodes for communication between any point and any other point on the network. All the types of Switching techniques discussed previously, such as circuit switching, packet switching and message switching are different modes of using PSTN.

Subscriber Loop Systems

In a general telephone network, every subscriber has two dedicated lines connecting to the nearest switching exchange, which are called the **Loop lines** of that subscriber. The laying of lines to the subscriber premises from the exchange office is called **Cabling**. As it is difficult to run cables from each subscriber‘s premises to the exchange, large cables are used through which the drop wires (subscriber lines) are taken to a distribution point.

The drop wires are connected to wire pairs



distribution point, in the cables. Such distribution cables from nearby geographical area are connected at a same feeder point where they connected to branch feeder cables which in turn, are connected to the main feeder cable. This whole process can be understood with the help of the following figure.

The subscriber cable pairs from the exchange will also terminate at MDF through main feeder cables that carry large number of wire pairs. These subscriber pairs and exchange pairs are interconnected at the MDF using jumpers, which makes MDF to provide flexible mechanism for reallocating cable pairs and subscriber numbers. This means a subscriber who shifts to a different location though in the same exchange area, can be allowed to use the same number using appropriate jumper, while his old drop wires can be used by another subscriber with a new number.

Switching Hierarchy and Routing

The next important system in this is the switching hierarchy and routing of the telephone lines. The interconnectivity of calls between different areas having different exchanges is done with the help of **trunk lines** between the exchanges. The group of trunk lines that are used to interconnect different exchanges are called the **Trunk Groups**.

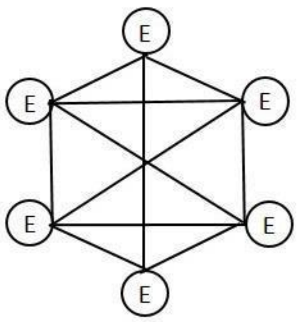
In the process of interconnecting exchanges, there are three basic topologies, such as

- Mesh Topology
- Star Topology
- Hierarchical

Mesh Topology

Mesh topology, as the name implies, is a fully connected network. The number of trunk groups in a mesh network is proportional to the square of the exchanges being interconnected. Hence, these mesh topologies are widely used in metropolitan areas where there is heavy traffic.

The following figure shows how a mesh topology looks like.

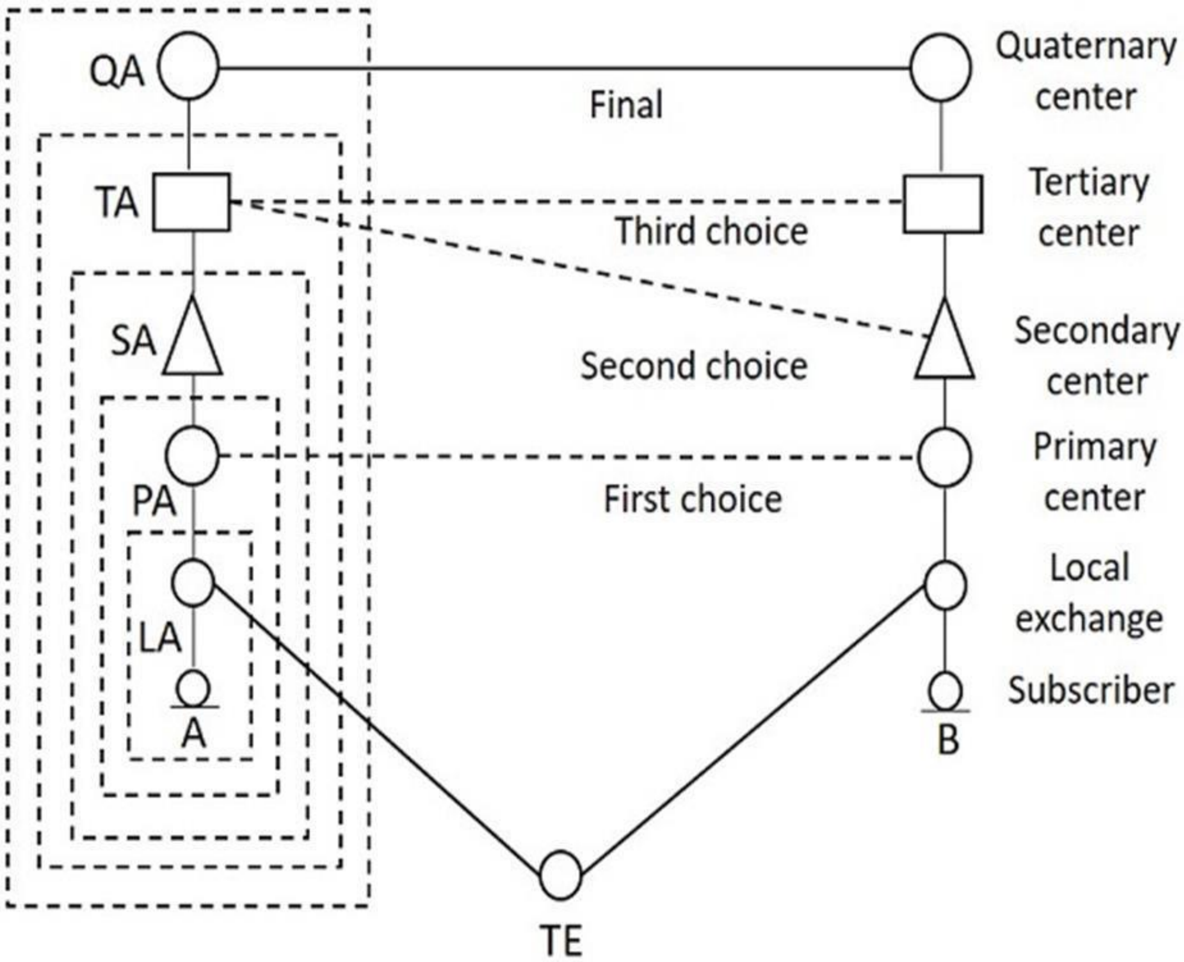


Star Topology

Star topology is connected in the shape of a star, which utilizes an intermediate exchange called a **tandem exchange** through which all other exchanges communicate. The figure given below shows the model of a star network. The star network is used when traffic levels are comparatively low. Many star networks can be used by interconnecting through additional tandem exchange, leading to a two-level star network as shown in the following figure.

Hierarchical

The hierarchical topology is used to handle heavy traffic with minimal number of trunk groups. The traffic flows through the **Final route** which is the highest level of hierarchy. If the traffic intensity between any pair of exchanges is high, direct trunk routes may be established between them as indicated by dashed lines in the figure given below. These direct trunk routes are **High Usage routes**. Wherever these high usage routes exist, the traffic flows through them. Here, the overflown traffic is routed along the hierarchical path. No overflow traffic is permitted from the final route.



To decide the routing on a particular connection, the following three methods are used:

- Right-through routing
- Own-exchange routing
- Computer-controlled routing

Transmission Plan

For reasons of transmission quality and efficiency of operation of signalling, it is desirable to limit the number of circuits connected in tandem. In a tandem chain, the apportionment of links between national and international circuits is necessary to ensure ‗quality' telecommunications. CCITT lays down certain guidelines in this regard in its recommendations Q40:

1. The maximum number of circuits to be used in an international call is 12.

2. No more than four international circuits be used in tandem between the originating and the terminating international switching centres.

3. In exceptional cases and for a low number of calls, the total number of circuits may b 14, but even in this case, the international circuits are limited to a maximum of four.

Taking the guidelines 1 and 2 above, we have eight links available for national circuits, which implies a limit of four for each national circuit National network designs should take into account this limit. The transmission loss is defined in terms of reference equivalents TRE, RRE and ORE as discussed in Section CITT recommends that for 97% of the connections the maximum TRE be limited to 20.8 dB and RRE to 12.2 dB between the subscriber and the international interface in the national network. This gives an overall reference equivalent ORE of 33 dB. Telephone administrations and companies attempt to design networks in such a way as to reduce as much as possible the ORE to improve subscriber satisfaction. From country to country OREs range from 6 dB to 26 dB. Transmission loss budget should provide for two factors other than the line and switch losses:

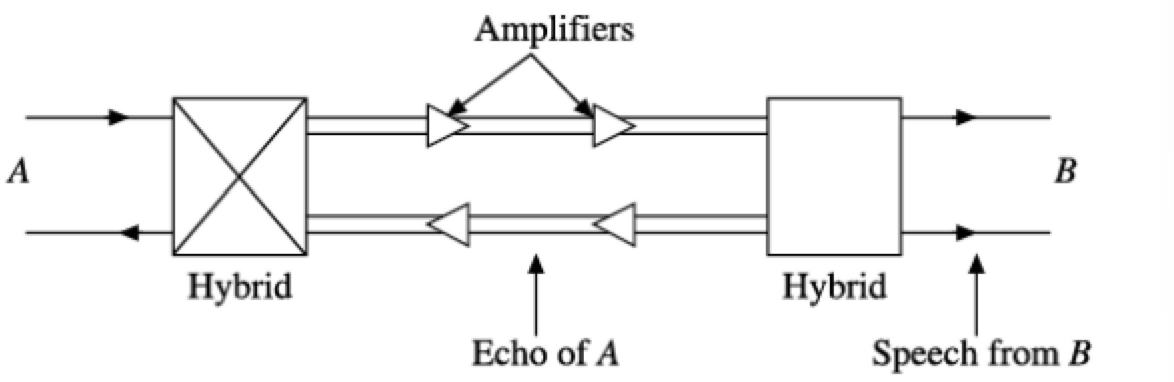
.Keeping echo levels within limits
. Control singing.

The Transmission of signals through cables should be high in quality in order to ensure better communication. The transmission links between national and international circuits should be better to connect in tandem for establishing calls.

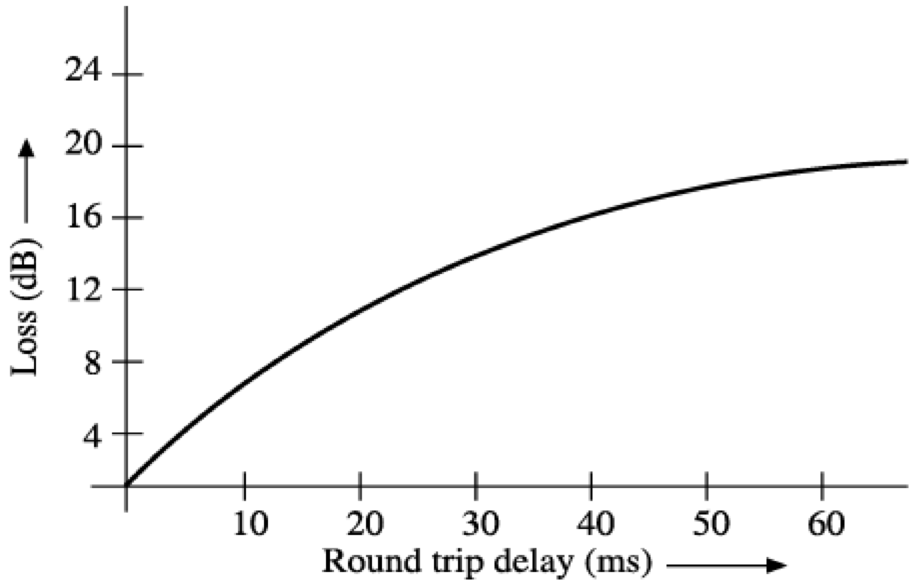
To have high quality standards, the following guidelines were put forward by the CCITT:

- The maximum number of circuits to be used in an international call is 12.
- No more than four international circuits be used in tandem between the originating and the terminating international switching centers.
- In exceptional cases and for a low number of calls, the total number of circuits may be 14, but even in this case, the international circuits are limited to a maximum of four.

Along with limiting the number of circuits required, the losses such as line loss or wire loss and switch loss or contact loss should also be minimized. These aspects come under the transmission loss budget, which provides for factors such as keeping echo levels within limits and control singing.



Echo as reflected signal.



Attenuation vs. echo delay.

Because of the long distances, the circuits need amplifiers and repeaters at appropriate intervals to boost the signals. At the subscriber-line interfaces, mismatch occurs; this results in reflecting a part of the incoming signal onto the outgoing circuit, which returns to the speaker as **Echo**. The echo suppressor or cancellation circuits are used to minimize the effect of the echo. The signal attenuation and echo are the main losses in the transmission lines along with contact and wire losses.

TRANSMISSION SYSTEMS

Modem long distance transmission systems can be placed under three broad categories:
1. Radio systems

2. Coaxial cable systems

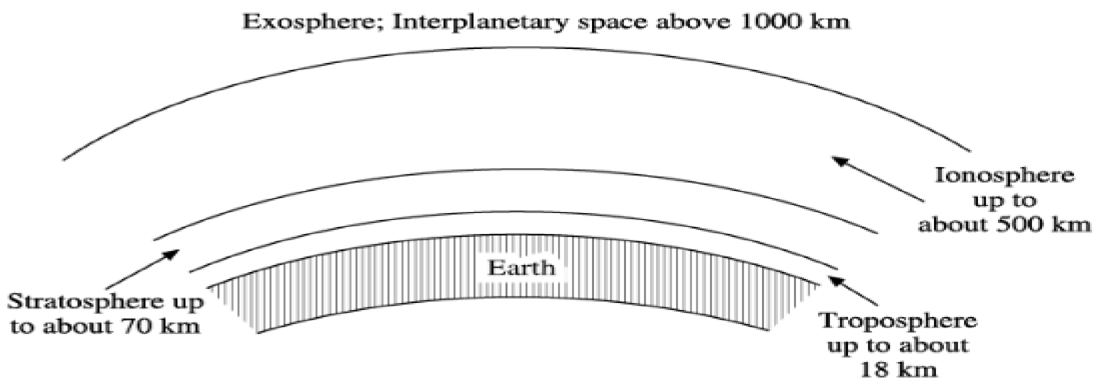
3. Optical fibre systems.

Fibre optic transmission systems have been discussed in detail in Chapter 7. In this section, we concentrate on radio and coaxial cable systems. Radio communication deals with electronic radiation of electromagnetic energy from one point to another through the atmosphere or free space. It is possible only in a certain portion of the electromagnetic frequency spectrum. This portion includes frequencies from 9 kHz to 400 0Hz. While there are international allocations for the radio spectrum up to 275 0Hz, most of the commercial uses take place between 100 kHz and 20 0Hz. Different layers of the atmosphere play a role in propagating radio waves. The atmosphere consists of four layers as shown in Figure 9.11. Of the four layers, the ionosphere and troposphere are useful for radio communication in certain frequency ranges. Certain other radio frequencies pass straight through the atmosphere and can be beamed towards satellites placed in the inter planet an space. Depending on the mechanism of propagation, long distance radio communication can be placed under four categories:
1. Sky wave or ionosphere communication

2. Line-of-sight (LOS) microwave communication limited by horizon

3. Tropospheric scatter communication

4. Satellite communication.



Layers of atmosphere.

Numbering Plan

During the early stages of development, the numbering scheme was confined to a small single exchange, which used to connect to the other exchanges by identifying them with the names of the towns in which they were located. But with the increase in the number of subscribers, many exchanges were introduced

A large central exchange which serves the main business center of a town, can be called the **Main Exchange** and the smaller exchanges serving different localities are called the **Satellite Exchanges**. The area containing the complete network of the main exchange and the satellites is known as the **Multi-exchange area**. A common numbering scheme was required to identify the location of the exchange of called subscriber, especially when the call is from a location outside the Multi-exchange area.

The common numbering scheme is called the **Linked Numbering Scheme**, where all the exchanges in a town were collectively identified by the name of the town. With the introduction of **Subscriber Trunk Dialing** (STD) or **Direct Distance Dialing** (DDD) for inter-city and inter-town long distance communications, the Multi-exchange areas were also allotted unique identification number. In order to make very long distance communications possible, the international dialing called the **International Subscriber Dialing** (ISD) was introduced, where the international numbering plan and national numbering plan came into existence.

Types of Numbering Plans

In this section, we will discuss the Numbering Plans for telephone networks. The plans are described in brief below:

Open Numbering Plan

This is also called the **Non-Uniform Numbering Plan** and it permits wide variation in the number of digits to be used to identify a subscriber within a multi-exchange area or within a country.

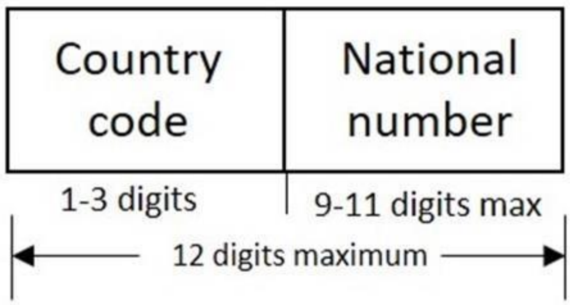
Semi-Open Numbering Plan

This plan permits number lengths to differ by almost one or two digits. The semi-open numbering plan is commonly used in countries such as India, Sweden, Switzerland and UK.

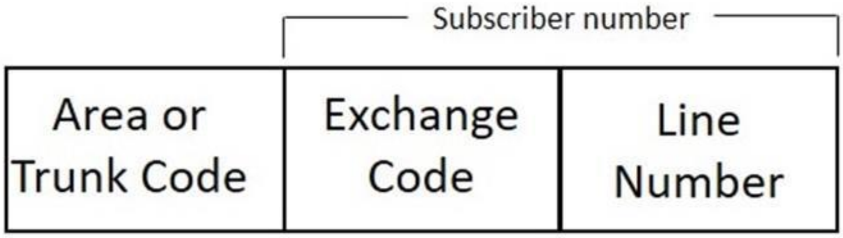
Closed Numbering Plan

This is also called the **Uniform Numbering Plan** where the number of digits in a subscriber number are fixed. This is used in a few countries such as France, Belgium, Canada, Hawaii and in a few parts of USA.

An International Numbering Plan or World Numbering Plan has been defined by the CCITT. For numbering purposes, the world is divided into zones. The following figure indicates the telephone number structure.



International telephone number



National Telephone Number

A national number consists of three parts. The parts are described below:

The Area Code or the Trunk Code

This code identifies a particular numbering area or the multi-exchange area of the called subscriber. It is with this code, the routing for a trunk call is determined and charged for it.

Exchange Code

This code identifies a particular exchange within a numbering area. It determines the routing for incoming trunk call from another numbering area or for a call originating from one exchange and destined to another in the same numbering area.

Subscriber Line Number

It is used to select the called subscriber line at the terminating exchange. The combination of the exchange code and the subscriber line number is called the Subscriber Line number in CCITT terminology.

Charging Plan

The calls are charged as accounted by the metering instrument connected to each subscriber line or as per a metering register that is assigned to each subscriber in case of electronic exchanges. A **meter** counts the number of charging units, and that count is incremented by sending a **pulse** to the meter. For the number of units, the meter reads, a bill is raised by assigning a rate to the charging unit.

The individual calls can be charged based on the following categories.

- Duration independent charging
- Duration dependent charging

Local calls within a numbering area are usually charged on a duration independent basis. For duration dependent charging, the meter starts incrementing, once the called subscriber answers the call. Depending upon the number of exchanges involved in setting up a call, more than one pulse is sent to the charging meter, which is called **Multi- Metering**. The metering pulse rate keeps on increasing per min with the distance between the called and the calling subscribers.

Signaling Techniques

Signaling techniques enable the circuit to function as a whole by inter connecting all varieties of switching systems. There are three forms of signaling involved in a telecommunication network.

- Subscriber loop signaling
- Intraexchange or register signaling
- Interexchange or inter-register signaling

The **subscriber loop** signaling depends upon the type of telephone instrument used. The **intra exchange** signaling refers to the internal portion of a switching system that is heavily dependent upon the type and design of a switching system, which varies depending upon the model. The **inter-exchange** signaling takes place between exchanges. This helps in the exchange of address digits, which pass from exchange to exchange on a link- by-link basis. The network-wide signaling that involves end-to-end signaling between the originating exchange and the terminating exchange is called the **Line signaling**.

The two main types of signaling techniques are:

In-Channel Signaling

In-Channel Signaling is also known as **Per Trunk Signaling**. This uses the same channel, which carries user voice or data to pass control signals related to that call or connection. No additional transmission facilities are needed, for In-channel signaling.

Common Channel Signaling

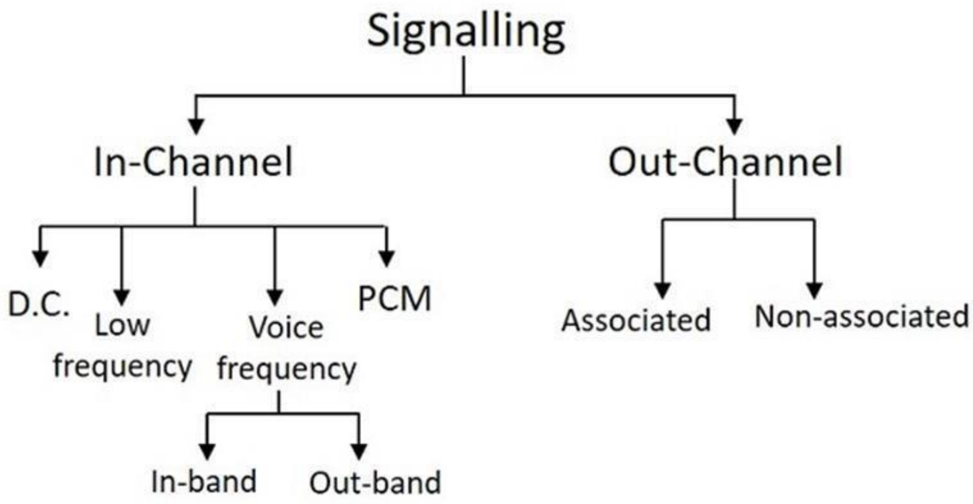
Common Channel Signaling uses a separate common channel for passing control signals for a group of trunks or information paths. This signaling does not use the speech or the data path for signaling.

We will discuss the signaling techniques in depth in our subsequent sections.

Types of Signaling Techniques

As discussed above, the signaling techniques are categorized into two, the In-channel signaling and the Common channel signaling. However, these are further divided into few types depending upon the frequencies and frequency techniques used.

The division is as shown in the following figure:



In-channel Signaling

This type of signaling is used to carry voice or data and pass control signals related to a call or connection. There are different types of In-channel Signaling, as seen in the above figure. The D.C. signaling is simple, cheap and reliable even for unamplified audio circuits. However, for amplified audio circuits, low frequency A.C. signaling may be adopted.

The Voice Frequency signaling is used when FDM (Frequency Division Multiplexing) transmission systems are used, because low frequency signaling and D.C. signaling cannot be provided. This Voice Frequency signaling may be **In-band** or **Out-band**.

In-band Signaling

In-band voice frequency uses the same frequency band as the voice, which is 300-3400 Hz, which has to be protected against false operation by speech. One such instant took place when a lady‘s voice which has generated a tone at around 2600Hz lasting for a duration of 100ms was detected as the line disconnect signal due to which her calls were frequently being disconnected in the middle of her conversation. Such problems precluded the in-band signaling during speech phase.

The advantages of In-band signaling are:

- The control signals can be sent to every part where a speech signal can reach.
- The control signals will be independent of the transmission systems as they are carried along with the speech signals.
- The Analog to digital and Digital to analog conversion processes will not affect them.

Out-band Signaling

The out-band signaling uses frequencies which are above the voice band but below the upper limit of 4000 Hz of the nominal voice channel spacing. The signaling is done throughout the speech period and thus continuous supervision of the call is allowed. Extra circuits are needed to handle the extremely narrow band width of this signaling, due to which it is seldom used.

Both of these in-band and out-band voice frequency signaling techniques have limited information transmission capacity. In order to provide enhanced facilities, common channel signaling is used.

Common Channel Signaling

Common Channel Signaling uses a separate common channel for passing control signals for a group of trunks or information paths as it does not use the speech or the data path for signaling. The common channel signaling consists of two types of nodes such as **Signaling Transfer Points** (STP) and **Signaling Points** (SP).

A Signaling point is capable of handling control messages directly addressed to it but is incapable of routing messages. Signaling transfer point is capable of routing messages and can perform the functions of SP.

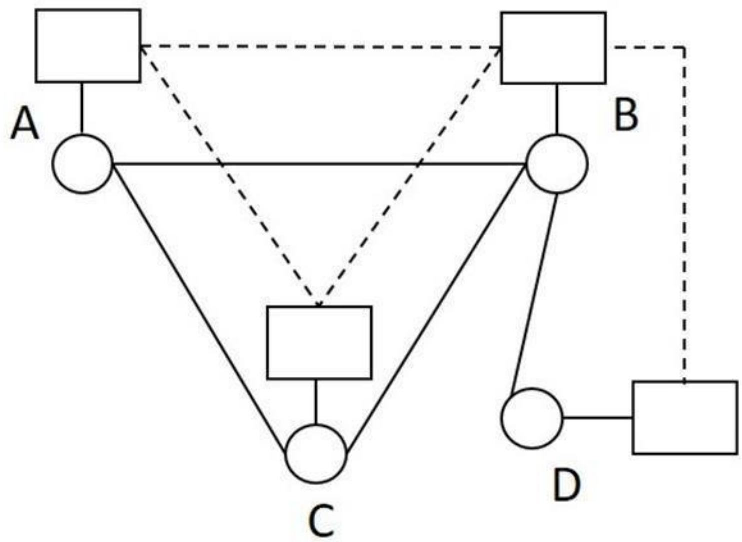
This common channel signaling is implemented in two modes:

- Channel associated mode
- Channel non-associated mode

Channel-associated Mode

In the channel-associated mode, the channel closely tracks the trunk groups along the entire length of the connection. Here, the signaling is done on a separate channel; the signaling path passes through the same set of switches, as does the speech path.

The following figure shows the associated mode of operation in common channel signalling.

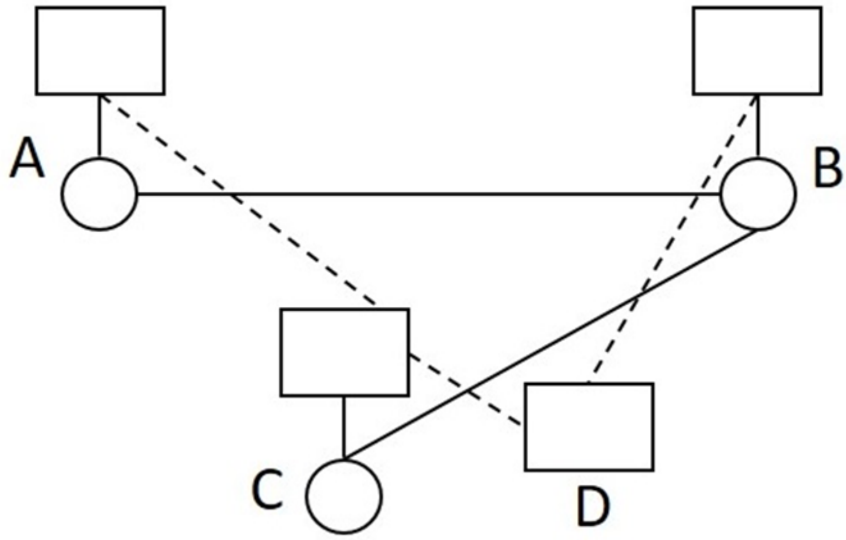


The signaling paths for the speech paths A-B, A-C-B and B-D are A-B, A-C-B and B-D respectively. The advantages of this signaling are:

- The implementation is economic
- The assignment of trunk groups is simple

Channel Non-associated Mode

In the channel non-associated mode, there is no close or simple assignment of the control channels to trunk groups. It follows a different path from that of the speech signal as shown in the following figure.



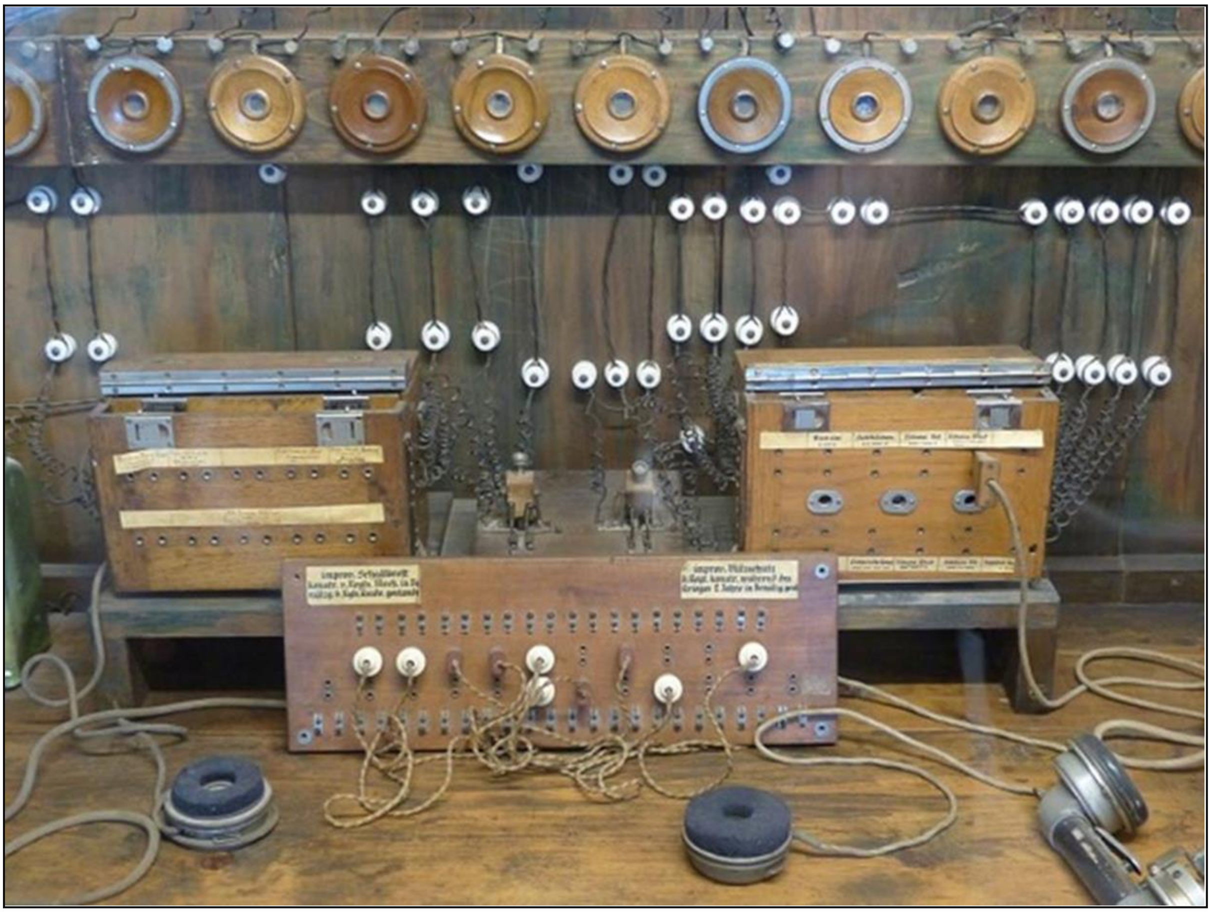
The signaling paths for the speech paths A-B and B-C are A-C-D-B and B-D-C respectively. The network topologies are different for signaling and speech networks. Though this scheme offers flexibility as there is no switching center, it is a bit complex, as the signal messages may be transferred between the two end switching systems via any available path in the common channel signaling network according to its own routing principles.

Private Branch Exchange (PBX)

Private Branch Exchange or PBX can be understood as a local exchange within an office or a building, in order to communicate within themselves. As the name implies, it is a private exchange, which is a branch to the main exchange similar to a local loop connected to the main loop as a branch.

Private Branch Exchange is a telephone system within a local area that switches calls between those users on local lines while allowing all users to share a certain number of external phone lines. The main purpose of PBX is to save the cost of requirement for a line to each user to the central exchange office.

The following figure shows the model of a PBX.



The above figure shows an early model of the PBX system. The PBX is usually operated and owned by the local office where the users are connected through it within that limited area.

The parts of a PBX include:

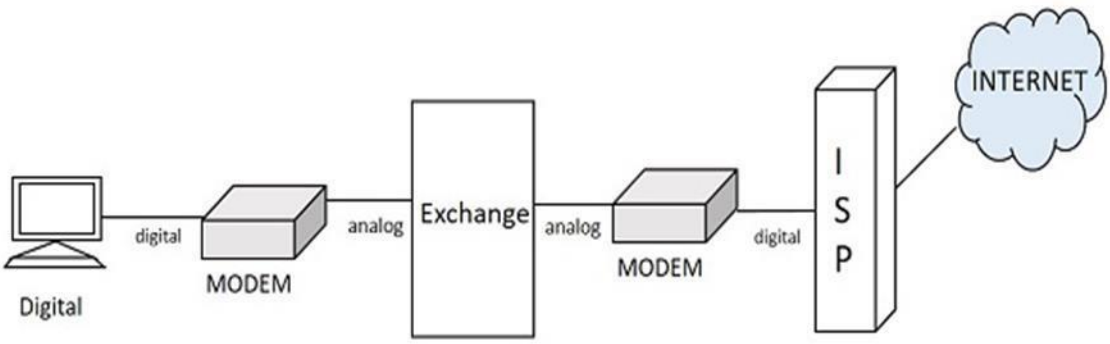
- A telephone trunk that contains many phone lines, which are terminated at PBX.
- A computer that handles the incoming and outgoing calls of PBX along with switching between different calls within the local loop.
- The network of lines within the PBX.
- A human operator console, which is optional.

Having all these along with the PBX equipment, the local branch exchange is built. The PBX exchanges previously operated using the analog technology. However, these exchanges operate on digital technology. The digital signals are converted to analog for outside calls on the local loop using Plain Old Telephone Services (POTS).

UNIT-V
INTEGRATED SERVICES DIGITAL NETWORK

In this chapter, we will learn about the Integrated Services Digital Network. Earlier, the transmission of data and voice both were possible through normal POTS, Plain Old Telephone Systems. With the introduction of Internet came the advancement in telecommunication too. Yet, the sending and receiving of data along with voice was not an easy task. One could use either the Internet or the Telephone. The invention of ISDN helped mitigate this problem.

The process of connecting a home computer to the Internet Service Provider used to take a lot of effort. The usage of the modulator-demodulator unit, simply called the **MODEM** was the essential thing to establish a connection. The following figure shows how the model worked in the past.



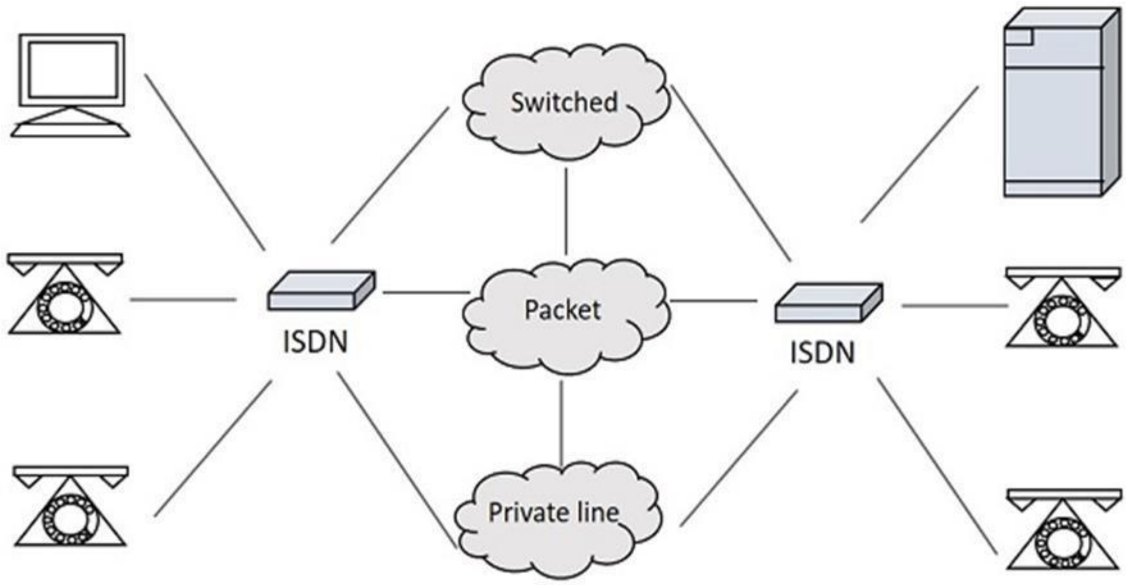
The above figure shows that the digital signals have to be converted into analog and analog signals to digital using modem during the whole path. What if the digital information at one end reaches to the other end in the same mode, without all these connections? It is this basic idea that lead to the development of **ISDN**.

As the system has to use the telephone cable through the telephone exchange for using the Internet, the usage of telephone for voice calls was not permitted. The introduction of ISDN has resolved this problem allowing the transmission of both voice and data simultaneously. This has many advanced features over the traditional PSTN, Public Switched Telephone Network.

ISDN

ISDN was first defined in the CCITT red book in 1988.The **Integrated Services of Digital Networking**, in short ISDN is a telephone network based infrastructure that allows the transmission of voice and data simultaneously at a high speed with greater efficiency. This is a circuit switched telephone network system, which also provides access to Packet switched networks.

The model of a practical ISDN is as shown below.



ISDN supports a variety of services. A few of them are listed below:

- Voice calls
- Facsimile
- Videotext
- Teletext
- Electronic Mail
- Database access
- Data transmission and voice
- Connection to internet
- Electronic Fund transfer
- Image and graphics exchange
- Document storage and transfer
- Audio and Video Conferencing
- Automatic alarm services to fire stations, police, medical etc.

Types of ISDN

Among the types of several interfaces present, some of them contains channels such as the **B-Channels** or Bearer Channels that are used to transmit voice and data simultaneously; the **D- Channels** or Delta Channels that are used for signaling purpose to set up communication.

The ISDN has several kinds of access interfaces such as:

- Basic Rate Interface (BRI)
- Primary Rate Interface (PRI)
- Narrowband ISDN
- Broadband ISDN

Basic Rate Interface (BRI)

The Basic Rate Interface or Basic Rate Access, simply called the **ISDN BRI Connection** uses the existing telephone infrastructure. The BRI configuration provides **two data** or bearer channels at **64 Kbits/sec** speed and **one control** or delta channel at **16 Kbits/sec**. This is a standard rate.

The ISDN BRI interface is commonly used by smaller organizations or home users or within a local group, limiting a smaller area.

Primary Rate Interface (PRI)

The Primary Rate Interface or Primary Rate Access, simply called the ISDN PRI connection is used by enterprises and offices. The PRI configuration is based on T-carrier or T1 in the US, Canada and Japan countries consisting of **23 data** or bearer channels and **one control** or delta channel, with 64kbps speed for a bandwidth of 1.544 M bits/sec. The PRI configuration is based on E-carrier or E1 in Europe, Australia and few Asian countries consisting of **30 data** or bearer channels and **two-control** or delta channel with 64kbps speed for a bandwidth of 2.048 M bits/sec.

The ISDN BRI interface is used by larger organizations or enterprises and for Internet Service Providers.

Narrowband ISDN

The Narrowband Integrated Services Digital Network is called the **N-ISDN**. This can be understood as a telecommunication that carries voice information in a narrow band of frequencies. This is actually an attempt to digitize the analog voice information. This uses 64kbps circuit switching.

The narrowband ISDN is implemented to carry voice data, which uses lesser bandwidth, on a limited number of frequencies.

Broadband ISDN

The Broadband Integrated Services Digital Network is called the **B-ISDN.** This integrates the digital networking services and provides digital transmission over ordinary telephone wires, as well as over other media. The CCITT defined it as, ‒Qualifying a service or system requiring transmission channels capable of supporting rates greater than primary rates.‖

The broadband ISDN speed is around 2 MBPS to 1 GBPS and the transmission is related to ATM, i.e., Asynchronous Transfer Mode. The broadband ISDN communication is usually made using the fiber optic cables.

As the speed is greater than 1.544 Mbps, the communications based on this are called **Broadband Communications**. The broadband services provide a continuous flow of information, which is distributed from a central source to an unlimited number of authorized receivers connected to the network. Though a user can access this flow of information, he cannot control it.

Advantages of ISDN

ISDN is a telephone network based infrastructure, which enables the transmission of both voice and data simultaneously. There are many advantages of ISDN such as:

- As the services are digital, there is less chance for errors.
- The connection is faster.
- The bandwidth is higher.
- Voice, data and video – all of these can be sent over a single ISDN line.

Disadvantages of ISDN

The disadvantage of ISDN is that it requires specialized digital services and is costlier.

However, the advent of ISDN has brought great advancement in communications. Multiple transmissions with greater speed are being achieved with higher levels of accuracy.

Motivation for ISD

Three factors are responsible for the developments towards ISDN:

1 Sociological or societal needs

I Economic necessities

3. Technological developments.

The rapid developments in various facets of society call for increasing and complex communication facilities- A biotechnologist today would like to examine a blood sample remotely simultaneously compare the analytical results of other samples stored in a centralised database. Consult his assistant who is presently in a laboratory some distance away, and report the findings as the investigation progresses, to his superior who is in another building. To meet such a demand, we need to electronically transmit the microscopic image of the blood sample and reproduce the same graphically on the computer screen of the biotechnologist, at a rate fast enough to faithfully reproduce the movements of living cells, etc. Simultaneously, the same communication medium is used to access a remotely located database and hold an audio conference with his assistant and superior. As another example, a senior executive of a company, who often takes important decisions at home late in the evening or while on a holiday, would like to give instant effect to his decisions- This may call for access to different computer systems connected in the form of network. Electronic banking facilities, facsimile transmission and desktop image processing facilities, all in the place there he is at present. In other words, automobiles, homes, boats, hotel rooms and railway compartments will need to house electronic workstations. Automatic calling party identification and guaranteed transaction privacy and security are important signalling requirements of systems that can support such application tin effect, the society needs a telecommunication system that can support universal access to a host of service& In such a system, it should be possible for a user to attach to the network anywhere in the world the equipment of his choice to obtain a particular service.

The user will be allotted a permanent identification number or code, like the income tax permanent account number or the social security number, which would be valid for his lifetime. No matter where a user lives, or how often he shifts residence, dialling the number assigned to the user would always ring his telephone, computer system or any other equipment. Similarly irrespective of wherefrom a user obtains his services, the related charges are debited to his account traditionally, network providers have put up separate and independent networks to support different services. Telex network. Data network. Telephone network and CATV networks are examples of such a development. Independent networks call for separate administration, maintenance staff and build mg for housing switching systems. The independent and duplicate infrastructural facilities lead to high capital cost low maintenance efficiency and high management cost.

In addition, the network facilities are never full utilised as the services are independently supported on different networks. The net result is that the overheads mm out to be excessive, leading to become economically unviable network services. Obviousl3; the independent network approach is not viable for the current and fracture services. One should consider the possibility of sharing resources to keep the costs down. Thus, sheer economic necssity is forcing network providers to look for solutions –-here many different services maybe integrated and supported on common network resources. Searching for new solutions is of no avail unless technology developments make possible such solutions. In fact, it is the technology factor that brought about the independent network solutions earlier.

The end equipment‘s for different services were analog in nature and had different electrical, electronic, signal and communication characteristics. It was necessary to design different networks to suit each of these devices. For example, the vokage, current and bandwidth requirements of a tele printer, a computer terminal and a telephone instrument are vastly different Accordingly, the switching systems, signalling systems and transmission media have to be different If the end equipment, switching, transmission and signalling systems can all function in the digital domain, the interface and communication requirements can be uniform for all the services.

Today, the digital technology has matured to a level where all the above mentioned functions of a telecommunications network can be realised in the digital domain It is interesting to trace the history of the process of digitalisation in the field of telecommunications. Traditionally, digital technology y always appeared first in the field of computers. In the late l930s and the early 1940s. while the theory of digital communication (pulse code modulation, sampling theorem, etc.) was coining of age, digital computers were already being built. The first digitalisation step in telecommunications appeared when slow channel was introduced for long distance transmission in the early 1960s. Transistorised or integrated circuit based minicomputers opened up the possibility of digital switching in the late 1960s. These computers were used in the exchanges, and switching wa.s organised in space domain or time domain. Developments in the field of packet switching networks led to the possibility of digital signalling in the mid 1970s. Since late 1970, digital end equipment‘s such as digital telephone and digital facsimile are being developed.

Each Plenary Assembly of CCITT, which takes place once every four years, assigns specific topics related to ISDN to the concerned study groups for an in-depth study during the ensuring four-year period- The result of such an organised study in the last many decades is that the ISDN architecture has evolved and a plethora of services have been supported in ISDN. As must be obvious by now. ISDN is the apt nomenclature for the telecommunications network in which all network functions are handled in digital domain and a single network supports a set of integrated services.

ISDN SERVICES

ISDN supports a variety of services. A short list of some of the important services is:

1 Videotex

2. Electronic mail

3. Digital facsimile

4. Teletex

5. Database access.

These services are described in the following sections.

1. Videotex

Videotex is a generic term for systems that provide easy to use. Low cost computer based Services via communication facilities. Three forms of videotex exist:

1. View

2. Teletext

3. Open channel teletext.

View data is fully interactive videoteL This means that requests for information or service from a user are acniallv sent to, received by, and acted on by a centralised computer. Teletext is broadcast or pseudo-interactive videotex service. Teletext users may select the information to be seen, the pace at which the information is to be displayed and, often, the sequence of display. The information is cast in the form of frames and a set of frames which is called a magazine is recycled continuously. Teletext is a one-way communication system and there is no real interaction between the user and the computer.

Open channel teletext is totally noninteractive one-way videotex. With this form of

videotex, the user receives preselected information in a predetermined order. There is no interaction, either real or apparent The user has no control over the pace or sequence of display. He just ins to watch what appears on the screen and pick up the information that is of use to him, as and when it appears on the screen. Open channel teletext may be classified into three categories according to the way the preselected information is displayed and the way the display
channel is used:

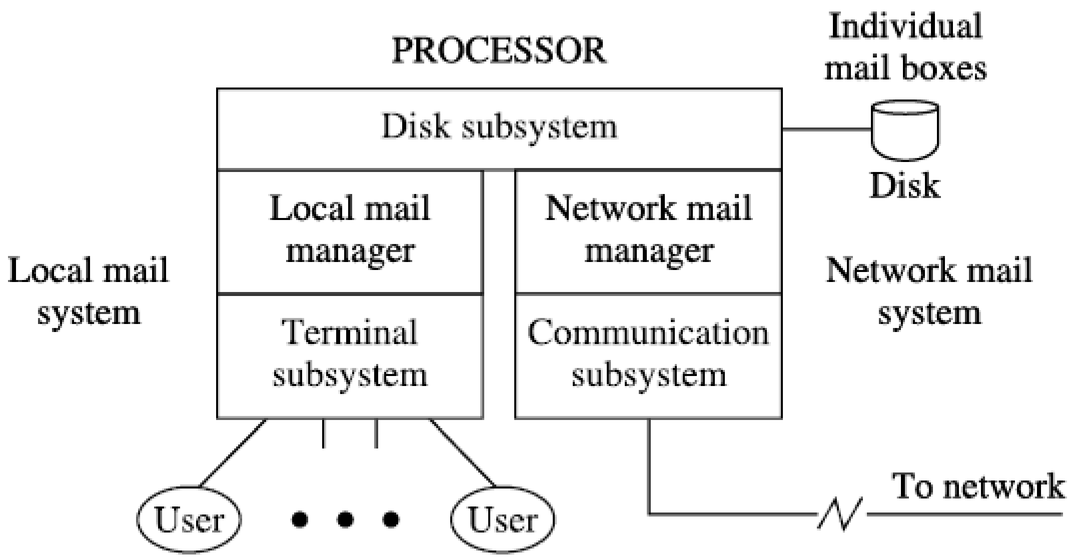
1. Dedicated open channel
2. Open captioning
3. Closed captioning.

In dedicated open channel teletext, a separate transmission channel is dedicated for the display of preselected information. Open captioning shares a normal display channel and the teletext display appears at fixed intervals along with other programmes on the channel.

2 Electronic Mail

Electronic mail. popularly known as email, may be defined as the communication of textual messages via electronic means. Even the telex communication is electronic in nature. Then, what is the difference? There are many. While telex communication is terminal-to-terminal. electronic mail communication is user-to-user. In telex, messages destined to a number of users are sent to the same terminal from where it is distributed by an operator or a messenger. On the other hand. Electronic mail is delivered to the mail boxes of individuals. Telex works on a circuit switched mode. Whereas electronic mail is a store and onward (S&F) service. Electronic mail is a computer based messaging system, whereas the telex is generally not. Being a person-to-person communication system, electronic mail turns out to be a cheaper alternative to telephone conversation and eliminates the time spent in establishing phone calls. Early electronic mail systems were organised around a single time sharing or multiuser computer system, wherein electronic mail was exchanged among the users of the system. Since the late 1970s, electronic mail facility is being extended over networks. A typical configuration of an electronic mail system is depicted in Figure.. There are two maor components of the system: one to handle mail within the system and another to handle mail over the network. Both share a common disk storage where mail boxes are maintained. The terminals connected to the system may be distributed throughout the organisation and kept on the tables of individuals.

Every registered user is provided with a mail box in the system. Any user may log on to the system from any of the terminals and send mail to another user on the same system or on the network. If the recipient is currently logged on, a message may be flashed on his terminal about the arrival of mail Otherwise, he is informed about the mail waiting in bis mail box as soon as he logs on to the machine. If the mail arrives when a recipient is logged on to the machine, he may suspend the present work and get into a conversational mode with the sender- Thus, real time exchange of messages is possible in an electronic mail system, if the two concerned parties are logged on at the same time. Electronic mail, being a S&F service on a network, real time exchange may not be possible.



A Typical configuration of the an electronic mail system

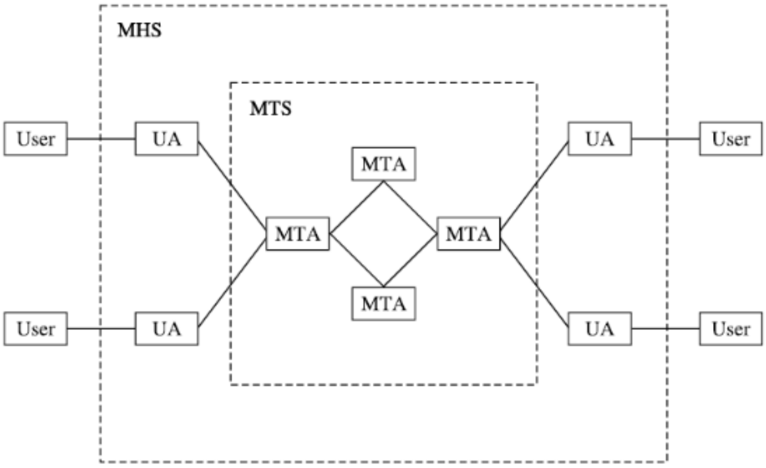
Users of homogeneous computer systems were the first to introduce electronic mail over networks there are a number of computer networks in the world which support electronic mail service. Some of the well-known networks that used to offer electronic mail service in the begging were UUNET, BITNET, CSNET and JÆNET As a next step in the development. Internetwork electronic mails have received the attention of the world network communities. An important problem in internetwork messaging is the highly varying mail address structures used by different networks. Users of a particular network are familiar with their own addressing schemes which need to be modified and recast into another format when sent over to another netvorlc This is conveniently carried out by using gateways without burdening the user with the knowledge of different addressing schemes and the associated conversion process. Most of the networks in the world have established a gateway to one or more networks for messaging purposes. Thus it is possible now to send electronic mail to almost anyone on any network in the world. There is also an attempt to evolve a uniform addressing scheme for all networks. In the context of Open System Interconnection (051) networks, electronic mail is considered as an application process running on the seventh layer. Standard electronic mail service components have been defined and approved by 1984 CCITT Plenary Assembly.

These are known as X.400 family of standards for message handling systems (MHS) Presently defined standards do not deal with the user interface or the services available directly to the user. They do, however. specify a set of base services that can be used for building suitable userlevel interfaces.

Table 11.1 X.400 Family of Standards	
Number	**Subject dealt with**
X.400	System model—service elements
X.401	Basic service elements and optional user facilities
X.408	Encoded information-type conversion rules
X.409	Presentation transfer syntax and notation
X.410	Remote operations and reliable transfer server
X.411	Message transfer layer
X.420	Interpersonal messaging—user agent layer
X.430	Access protocols for teletex terminals
X.435	Electronic data interchange messaging system

The MHS model as defined in X.400 has two types of entities:

- . User agent entity (UAE)
- . Message transfer agent entity (MTAE).



MHS = message handling system
MTS = message transfer system
MTA = message transfer agent
UA = user agent

It also assists the user in other message functions such as filing, replying, retrieving and forwarding. Message transfer agent (MTA) is concerned with transfer of messages across the network and functions in an environment designated as message transfer system (MTS) It obtains messages from the source UA and delivers the same to the destination UA On accepting a message, the message transfer agent (ffA) performs either a deliver function or a routing function. If the destination UA is in the same system as the MTA or is attached to the MTA direct1; then the MTA performs a delivery function otherwise it performs a routing function.

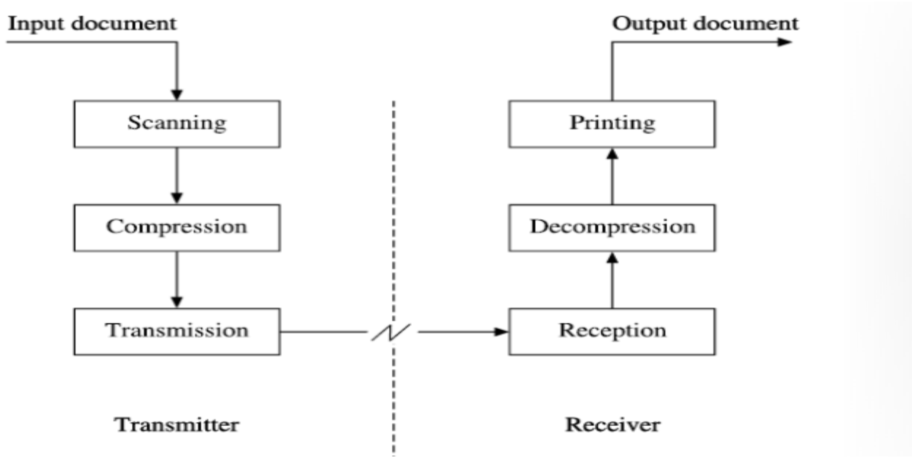
Depending on the physical location of the MTAs and UAs, a number of different physical realisations of email systems are possible. Some typical realisations are shown in Figure .In general, in a multiuser system, a single MTA and manic UAs are present. In a personal computer no it is usually implemented and there is only one UA. In a workstation, one MTA and one UA. are present In a computer dedicated to mail transfer functions, there is only one MTA which interacts with a number of UAs outside the machine. Figure shows an electronic mail configuration around a minicomputer system, Figure a number of personal computers attached to the dedicated mail transfer system, and Figure a workstation attached to a minicomputer.

3 Digital Facsimile

Document exchange through facsimile systems has emerged as a major application of telecommunication systems. Unlike electronic mail, facsimile is capable of transmitting and receiving printed matter which may include graphics, drawings, and pictures. Two types of facsimile systems exist:

- Photographic facsimile
- Document facsimile.

In photographic facsimile, the gray level information is transmitted and printed in addition to black and white. Typically, there are 8 or 16 gray levels that can be recognised by the system. Document facsimile system handles only black and white levels, i.e. only two gray levels. Document facsimile system is more popular than the photographic system. The receiver transmitter functions applicable to both the types of facsimile systems are shown in Figure. A scanner converts the print material into a set of points represented by bit-mapped electrical signals. In document facsimi e, every point is represented as a zero or one, whereas in photographic facsimile, each point is represented by 3 or 4 bits giving the gray level values. Scanning resolution is of the order of 4—16 dots per mm (100—400 dots per inch) horizontally across the page and 4—8 lines per mm (100—200 lines per inch) vertically down the page. An A4 size page (210 x 297 mm) generates about 3 million picture elements (pels).



Functions of facsimile system

At a transmission rate of 9600 bps, the time required to transmit 3 million bits is of the order of 300 seconds or about 5 minutes. Facsimile information is transmitted using regular

telephone lines at the same tariff as the telephone charges. There is a need to compress the scanned data before it is transmitted. This is the second step in facsimile transmission.
There are two types of compression techniques:

- Information preserving techniques
- Approximate techniques.

CCITT has standardised on two compression techniques, both belonging to the first category. These techniques reproduce an exact replica of the scanned image, whereas techniques belonging to the second category approximate the original in the output. The two techniques standardised by CCITT are:

- Modified Huffman (I‖JH) technique
- Modified READ (MR) technique.

Before applying the Huffman or relative element address designate (READ) technique, the scanned information is coded using a basic technique known as run-length coding. In a document facsimile system. the pels are represented by zeros and ones. A quick observation of a printed document, taking into account the horizontal scan resolution of about 200 dots per inch, reveals that the scanned information would contain strings of zeros and ones. Longer strings of zeros and ones are more probable than short strings- Run length coding conveys a count of continuous ones and zeros instead of the pattern itself Huffman coding is based on the principle that a more efficient coding scheme can be evolved by using short code words for frequently occurring symbols and long code words for sparingly occurring symbols, instead of using a uniform size for all symbols. In the case of document facsimile, longer strings of zeros or ones are coded with short code words as they are more probable and vice- versa with the shorter strings. However, the string sizes vary over a very large range, theoretically one to a few millions per page. Consequently, the average size of the code word is large. To overcome this problem, the Huffinan code is modified to view the run-len2ths in two parts and code them independently taking into account their probability of occurrence. The two parts of the run length are known as made-up code part and the terminating code pan. The first part is used for coding run lengths that are less than 64 and the latter part is used for run lengths greater than or equal to 64. A detailed treatment of the coding scheme is beyond the scope of this text.Interested readers may refer to relevant CCITT standards or Further Reading [3).

Relative element address designate (READ) code is based on the principle that further coding efficiency may be gained by coding the relative position of changing elements. There is a strong correlation between the black—white patterns of two adjacent scan lines in a document. This fact is exploited in modified READ (MR) coding. A changing element is coded in terms of its distance from a preceding changing element on the same or on the previous line. Which one of the two elements is chosen as reference depends on the relative position of the element to be coded. In general, the reference element is so chosen as to minimise the size of the code word required. A detailed treatment of this coding scheme is beyond the scope of this text. Many studies have been conducted to assess the effectiveness of MH and MR coding schemes over a representative set of document pages. Both coding schemes achieve considerable compression. It has been observed that the MR coding is more effective than the MH coding. Compression ratios of 1:8 and 1:16 on an average have been reported for MH and MR coding schemes, respectively Since their introduction in the early 1 970s. four models of facsimile machines, known as Group I, II, IN and IV machines, have evolved. Group I and II machines are analog in nature, whereas Group III and IV use digital technology. Group I, II and III machines have been designed for operation with PSTN, whereas Group IV machines with ISDN. A comparison of the features of these machines is given in Table . Group IV machines have further models known as class 1. 2 and 3 machines. Class 2 and 3 machines have a horizontal and a vertical resolution of 11.8 pels mm each Class 2 machine is capable of receiving teletex messages, whereas class 3 machine can receive and transmit teletex messages.

4 Teletex

Teletex is an upgrade to the conventional telex service. The terminal-to-terminal communication service of telex has been turned into office-to-office document transmission system by teletex.Teletex enables direct communication between electronic typewriters, word processors and personal computers. These units have storage for transmitting and receiving messages. The use of such equipments considerably enhances the character set available for document preparation. In addition to the standard character set, a rich set of graphic symbols and a comprehensive set of control characters are supported in teletex. The set of control characters helps in the preparation and reproduction of documents. In particular, the‘ permit the positioning of the printing element, specification of page orientation, left and right margins, vertical spacing and the use of underlining. The page control feature allows standard A4 size papers to be used for receiving messages instead of the continuous stationery used in conventional telex systems.

A background foreground operation is provided in teletex. Transmission reception of messages should proceed in the background without affecting the work which the user may be carrying out in the foreground with the equipment. In other words, a user may be preparing a new document, while another document is being transmitted or is received. The teletex would also maintain compatibility with the present telex system and inter-operate with them. Teletex procedures call for the exchange of header information before the actual document transfer takes place. The header information consists of four parts:

Part1: Destination id

Part2: Originator id

Part 3: Date and time stamp

Part 4: Document reference.

Twenty four characters are used for source destination ids, 14 characters for date and time stamp, and 7 characters for document reference which also specifies the number of pages in the document.

Destination Source id consists of four fields:

Field 1: Country Network code

Field 2: National subscriber number

Field 3: Reserved for future use

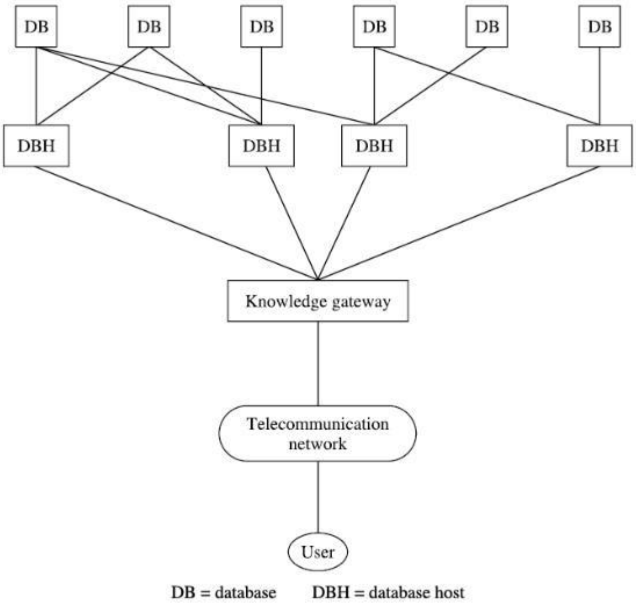
Field 4: Terminal Owner code.

The number of characters allotted to each of the above fields is variable subject to a maximum for each field, the total being 24 characters.

5 Database Access

We are in an age of information explosion. There is so much of new information being generated published periodically that it has become almost impossible to keep track of the new information in any field. It is in this context that electronic databases are becoming a boon Here, a user can, by a suitable search query, obtain all the information generated in a particular topic. There are large number of databases in different parts of the world, covering a vide variety of subjects which include humanities and social sciences. science and technology. engineering and industry. These databases may be accessed online using the telephone network, modem and a personal computer. Accessing a database calls for an agreement to be signed with the database vendors and each vendor may have different terms and conditions- For an individual or an organisation to deal with a number of database vendors is cumbersome and difficult. In order to facilitate access to various databases through a single contact point: agents have sprung up and set up what are known as database _hosts‘. These host machines are connected to a number of databases.

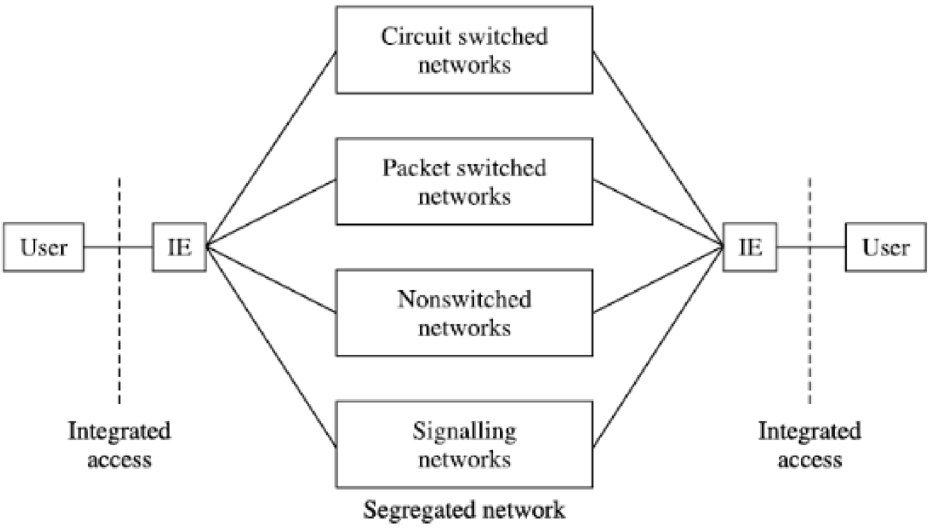
By entering into an agreement with one database host agent: the user can gain access to a variety of databases. bile a database host simplifies the access procedure to databases, the user will still have to consult directories and other publications to decide on the right type of database to be accessed, involving considerable knowledge, expertise and effort on his part. Casual users and even frequent users would like a simpler way of accessing information of interest to them- In order to assist the users select the right type of database and to provide them transparent access to information, a new type of service is introduced by setting up what are known as knowledge gateways. The scheme of access is shown in Figure 113. The knowledge gateways use concepts from artificial intelligence and the techniques of expert systems. A user interacts with the knowledge gateways which provide him expert help to choose the right database pertaining to the subject of interest.



Electronic access information

NETWORK AND PROTOCOL ARCH ITECTURE

Network architecture of [SDN followed an evolutionary path. It is natural that an evolutionary approach was taken with regard to ISDN. The wide range of telecommunication equipments and networks that were existing could not replaced overnight by ISDN. As a first step, the then existing analog telephone networks were converted to digital networks. These networks were then be operated along with other existing data and signalling networks. Such networks are termed Integrated Digital Networks (ON). Thus, initially. ISDN functioned as a collection of a few segregated networks to which an integrated access is provided as illustrated in Figure.It is seen from the figure that four different types of networks form part facilities.



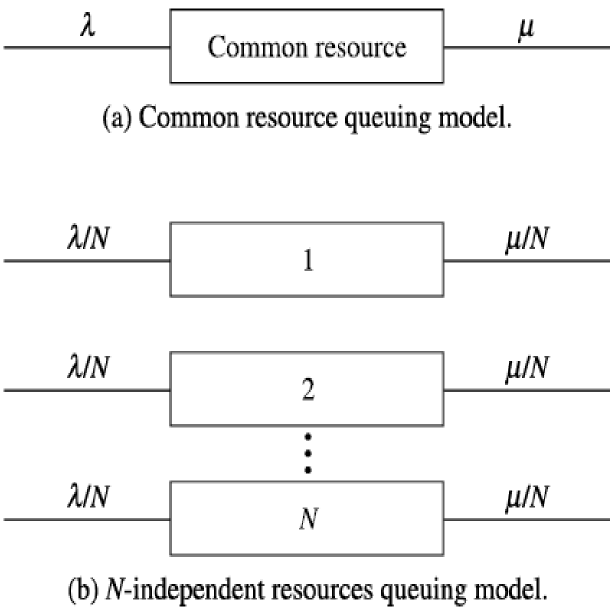
ISDN Architecture

The motivation for moving into an integrated infrastructure is based on factors like operational issues, management complexities and economies of realisation as discussed in Section 11.1. It is also based on the general belief that the pooling of resources would lead to an efficient utilisation of the resources and better performance of the system. This is true in most cases. We illustrate this point by modelling the segregated and the integrated networks as MMc and MM1 queuing systems respectively, and by analysing their performance. The models used are shown in Figure 11.9. The dynamics of the queuing systems are governed by the steady state birth-death Eqs. (8.10) and (8.11). For the systems under consideration

A = total mean arrival rate

ji = total mean service rate

N= number of servers for the case of segregated system



(a) Common resource queuing model.

(b) *N*-independent resources queuing model.

Different queuing configurations

The total arrival rate and the total service capacity remain the same in both cases. For the sake of simplicity, we assume that the service capacity and the arrivals are uniformly distributed over the N servers in the segregated model. Arrival and service rates are assumed to be constant and independent of the state of the system. Under these conditions. Eqs. (8.10) and (8.11) reduce to

$$M = \sum_{k=0}^{\infty} k p_k = (1-\rho) \sum_{k=0}^{\infty} k \rho^k$$

$$\lambda p_{k-1} + \mu p_{k+1} - (\lambda + \mu) p_k = 0 \qquad \text{for} \quad k \geq 1 \qquad (11.1)$$

$$p_1 \mu = p_0 \lambda \qquad \text{for} \quad k = 0 \qquad (11.2)$$

For different values of k, we get

$$\begin{aligned} p_1\mu &= p_0\lambda \quad \text{or} \quad p_1 = \rho p_0 \\ p_2\lambda &= p_1\lambda \quad \text{or} \quad p_2 = \rho^2 p_0 \\ p_3\lambda &= p_2\lambda \quad \text{or} \quad p_3 = \rho^3 p_0 \\ &\vdots \qquad\qquad\quad \vdots \\ p_k\mu &= p_{k-1} \quad \text{or} \quad p_k = \rho^k p_0 \end{aligned} \qquad (11.3)$$

where $\rho = \lambda/\mu$ is known as traffic intensity. To eliminate p_0 from Eq. (11.3), we use the fact that the probability sums to 1:

$$\sum_{k=0}^{\infty} \rho^k \ p_0 = 1 \qquad (11.4)$$

$\sum_{k=0}^{\infty} \rho^k$ is a geometric series and the sum is given by $1/(1 - \rho)$. Therefore,

$$p_0 = 1 - \rho, \qquad P_k = (1 - \rho)\rho^k \qquad (11.5)$$

Let us now use the mean waiting time of a customer 2 as a measure of performance of the system To determine T, we first determine the mean number of customers in the system .4 by using the state probabilities

Now differentiating both sides of the equation

$$\sum_{k=0}^{\infty} \rho^k = \frac{1}{1-\rho}$$

with respect to ρ, we get

$$\sum_{k=0}^{\infty} k\rho^{k-1} = \frac{1}{(1-\rho)^2}$$

Multiplying both sides by ρ, we obtain

$$\sum_{k=0}^{\infty} k\rho^{k} = \frac{\rho}{(1-\rho)^2}$$

Therefore,

$$M = \frac{\rho}{1-\rho} \qquad (11.6)$$

We now use the well-known Little's result which states that the mean number of customers in any queuing system is equal to the product of the mean waiting time for a customer and the mean arrival rate. The result is valid for all queuing system is irrespective of the arrival time or service time distributions Hence,

$$M = \lambda T \quad \text{or} \quad T = M/\lambda$$

For the integrated model, we have

$$T_1 = \frac{\rho}{1-\rho} \times \frac{1}{\lambda}$$

Substituting $\rho = \lambda/\mu$, we get

$$T_1 = \frac{\lambda}{\mu(1-\lambda/\mu)} \times \frac{1}{\lambda} = \frac{1}{\mu - \lambda} \tag{11.7}$$

For the segregated model, we have

$$T_s = \frac{1}{(\mu/N)-(\lambda/N)} = \frac{N}{\mu-\lambda} = NT_1 \tag{11.8}$$

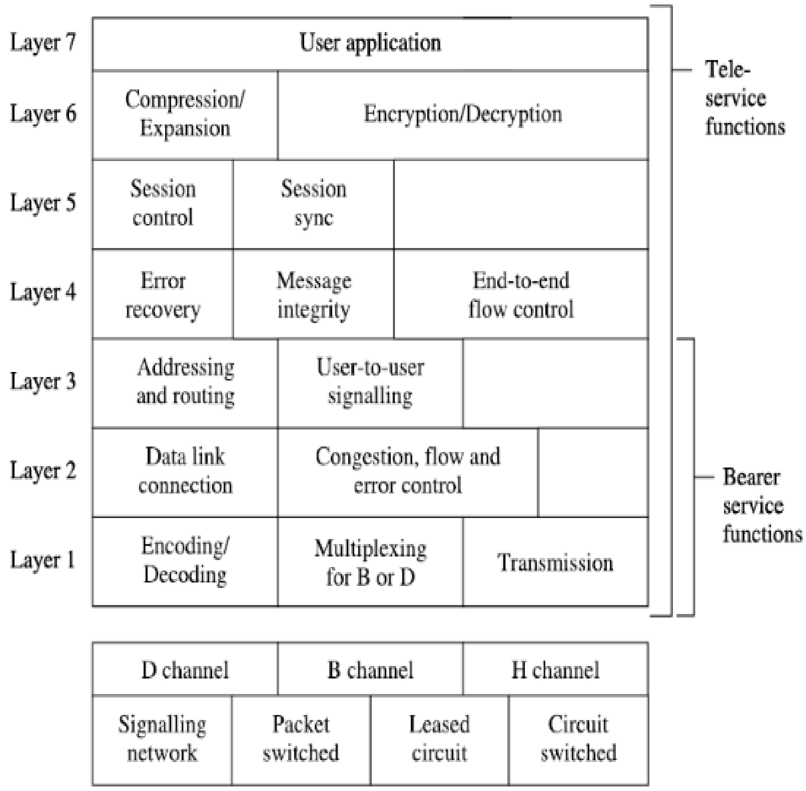
Equation (11.8) shows that the delay performance of the segregated system deteriorates by Ntimes if the integrated system is broken mm N segments .It should be noted that this analysis is valid only under the following assumptions:

1. The total capacities of the integrated and segregated networks are identical.

2. Input services have identical requirements in both the cases.

3. Service rate distributions are identical.

These assumptions may not be valid in ISĐN environment. Incoming traffic characteristics vary for different services like voice, data facsimile. etc. Similarly, the service time characteristics may also be different. Under the circumstances, it is possible that the integrated network performs worse than the segregated one. An analysis in this regard is beyond the scope of this text and the interested readers are referred to Further Reading [5]. The protocol architecture of ISDN is layered according to the OSI reference model this is as shown in Figure. However, the specific functionalities of the various layers are different from those of the OSI layer functions.



ISDN Protocol architecture

The differences stem from the different fundamental characteristics of ISDN and data networks. While ISDN is largely a circuit switched network with provision for store and forward (S&F) communication, data networks are largely S&F networks with provision for virtual circuit switching. As a network, ISDN is largely concerned with the lower layers l—3. Layers 4—7 are the concern of the service providers. Accordingly, the lower three layer functions are known as bearer sen-ice functions. Services offered by service providers are known as tele senices. Functionalities of layers 1—7 are required to offer tele services which are built over bearer.service functions. Layers 4—7 flinctionalities are not vet fully defined. Some of the important functions performed in
the lower three layers are:

Layer 1 (Physical Layer)

1. Encoding and decoding of signals

2. Transmission of B, D, and H channel data

3. Multiplexing to form basic or primary rates

4. Activation and deactivation of physical circuits.

Layer 2 (Data Link Layer)

1. Establishing and clearing data links

2. Error, flow and congestion control

3. Synchronisation

Layer 3 (Network Layer)

1. Addressing and routing

2. Establishing and clearing network level connections

3. User-to-user signalling

4. Network level multiplexing

5. Internet working multiplexing.

Transmission channels B. D and H are discussed in Section 11.4. Channel level multiplexing is also dealt with in that section. Basic and primary rates are discussed in Section 11.5 which also deals with some layer 3 functionalities. User-to-user level signalling is covered in Section 11.6. Addressing structure is discussed in Section 11.7, and Section 11.8 deals with interworking of networks. Further details of bearer and teleservices are presented in Section 11.8. Laver 1 protocol definitions apply uniformly to all transmission channels and services. But, from layer 2 upwards, the protocol structure differs for different channels and services. For
example, in layer 2, a new protocol, LAP—D. has been defined for handling D Channel information. Level 2 of X.25 (see Section 10.4) is used for a packet switched connection over B channel. LAP—D protocol is modelled after LAP—B protocol of X.25, which has been discussed in Section 10.4. Other functions of layer 2 are similar to datalink layer functions of the OSI reference model, discussed in Section 10.4. Hence. ISDN laver 2 functionalities are not covered in this chapter. Interested readers may refer to Further Reading [1, 2, 4, 5].

TRANSMISSION CHANNELS

- There are three types of fundamental channels in ISDN around which the entire information transmission is organised. These are:

- Basic information channel B channel, 64 kbps
- Signalling channel D channel. 16 or 64 kbps
- High speed channel FI channels
- HO channel, 384 kbps
- HI 1 channel, 1536 khps
- H12 channel, 1920 kbps

B channel and D channel are basically adopted from telephone digital networks with common channel signalling. Digital telephone networks have evolved around A-law and g-law encoding of speech signal at 64 kbps. ISDN has the same rate for the basic information

channel although the technology has advanced to a stage where quality voice can be transmitted at 32 kbps or even less. Having adopted a 64 kbps rate. ISDN. However, permits lower rate signals to be transmitted by using rate adaptation.

Alternatively, a number of low rate information channels may be multiplexed and sent on the B channel. Information streams at rates of 5. 16 and 32 kbps are rate adapted by placing the information bits in the first 1, 2 and 4 bits, respectively of an octet (S-bit quantity) in the B channel and filling the remaining bits by binary ones- Streams at rates other than 8, 16 or 32kbps and lower than 32 kbps are rate adapted in two stages. First, they are adapted to one of S, 16 or 32 kbps rates and then to 64 kbps by using the technique described above. Rate adaptation in the first stage is done by creating a frame similar to HDLC frame with the entire capacity filed in by the flag bits. The selection of frame size and frame rate is governed by the user information stream rate and the rate to which it is being adapted.

USER-NETWORK INTERFACES

Comprehensive user-network interface definitions are key to ensuring worldwide ISDN compatibility. An excellent example of an interface standard that serves us so well and goes almost unnoticed is the electrical power user interface. \Ve can purchase an electrical appliance almost anywhere in the world and plug it in our house socket and expect it to work. In fact, we are totally unmindful of the internal wiring, power distribution or generation scheme or the physical dimensions of the socket etc. as the final user interface conforms to a universal standard. By contrast. data communications suffer due to the presence of a variety of interface connectors. pin assignments and software interfaces. In ISDN, user-network interfaces have been given careful consideration to avoid potential inconsistencies that may arise. ISDN caters to a variety of services such as voice, data, telemetry and image In a situation like this, one encounters conflicting requirement & On the one hand, a number of custom designed interfaces may ideally suit each service but would lead to a proliferation of interfaces. On the other hand, one single multipurpose interface may turn out to be an ‗overkill‘ for most of th services. One needs to adopt a via media. Keeping such factors in mind. two information rate access interface have been standardised for ISDN:

. Basic rate access

. Primary rate access.

In some sense, these rate accesses are analogous to 5 A and 15 A power sockets at home The basic rate access caters to low bit rate services and the primary rate access caters to services that demand high bit rates in the same way as the two types of power sockets cater to low-power and high-power electrical appliances. The selection of access interfaces is somewhat influenced by the ISDN approach plan to achieve rapid and cost effective realisation.
. Use of much of the existing telephony networks; expensive replacement is not to be restored to except for some special ft1nction or locations.

. Use of low cost l SI technology to add new functionalities and to provide upgrades capabilities.

SIGNALLING

ISDN uses a common channel signalling scheme. The signalling is done over the D channel which acts as the common signalling channel for the B and H channels which carry the user information D channel may also be used for carrying some user information, if there is spare capacity. In such cases also, the required signalling is done on the D channel. The concepts of common channel signalling and the CCITT‘s Signalling System 7 (SS7) have been discussed Section 9.9. ISDN adopts SS7 for its interoffice and other network related signalling. It is ISDN User Part (1St?) of SS7 architecture (see Figure 9.36) that deals with signalling facilities in ISDN. Unlike in other networks, signalling in ISDN falls into two distinct categories:

. User level signalling

. Network level signalling.

All user generated signalling and the signalling features that are open to the user are treated as user level sign1Iing and are defined as part of the layer 3 user-network interface standards. The signalling facilities employed by the network to support user level signalling and to implement network control fùnctions, not directly related to the user, are treated as network level signalling and are defined as part of the ISUP of SS7. In this section, we describe the user level and network level signalling features.

1 User Level Signalling

User level signalling in ISDN permits a user to

1. establislt control and terminate circuit switched connections in B channeL

2. carry out user-to-user signalling, and

3. establish, control and terminate packet switched connections in B or D channels.

User-to-user signalling is achieved by employing a symmetrical protocol for outgoing and incoming calls. User level signalling is of two types:

. Message based signalling

. Stimulus signalling.

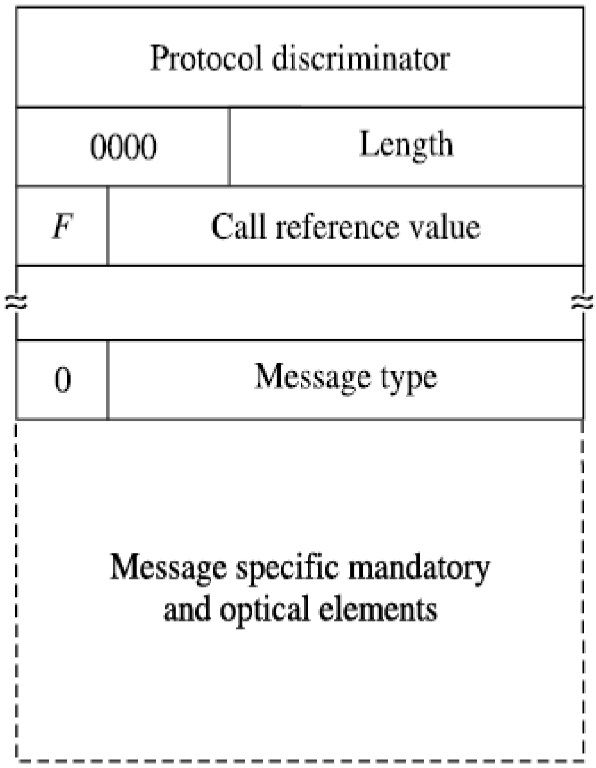
Message based signalling is employed when the user end equipment is an intelligent terminal. In ISDN parlance, an intelligent terminal is known as functional terminal It provides a user friendly interface for signalling and performs the functions of forming, sending, receiving and replying messages. The process of establishing, controlling and terminating a call is achieved by exchanging messages between the network and the terminal. The messages may be placed under four groups:

1. Call estabhsbment messages

2. Call control messages

3. Call disconnect messages

4. Miscellaneous messages.



User Level Signalling message structure

.2 Network Level Signalling

Network level signalling in ISDX is concerned with interoffice signalling, signalling features accessible by the user to obtain enhanced services from the network and other network related signalling. Circuit suspension and call supervision messages are examples of network related sign1Iing. The procedures for network level signalling are defined as the ISDN user part(ISUP) of the signalling system 7. One of the main aims in the context of ISUP has been to evolve a flexible design for the signalling system to accommodate new services and connection types that may come about in the future to be supported on ISDN. Flexibility is achieved by making the signalling entirely message basei For the transport of the signalling messages, ISUP relies on the services of the message transfer part or the network services part of SS7 .40 network level messages have been standardised so far and these messages may be placed under nine broad categories:

1. Forward address

2. General setup

3. Backward setup

4. Call supervision

5. Circuit supervision

6. Circuit group supervision

7. In-call modification

8. End-to-end

9. User-to-user.

NUMBERING AND ADDRESSING

In telephone and data networks, the end equipment are more often single units than multiple devices units like MEX or LAN. Historically a telephone, a computer, or a terminal has been the predominant end equipment. The numbering systems for these networks have also evolved to identify single equipment end points. In ISĐN, multiple devices at the end points are more of a norm than single units, in view of the multiple services environment. It then becomes necessary to identify a specific end equipment, e.g. facsimile or computer, to render the service. Identifying the specific equipment is a two-level process; first the end point is identified as in the case of telephone or data networks and then the equipment at the end point. ISDN addressing structure provides for this requirement. The component of the ISDN address which is used to identify the end point is known as the ISDY number: and the component for identifying the specific equipment at the end point is called the ISDN sub address. The numbering plan for ISDN is evolved using the following guidelines:

1. It is based on, and is an enhancement of: the telephone numbering plait In particular, the country codes evolved for the telephone numbering as defined in CCITT standard E. 163 are adopted in toto for ISDN.
2. It is independent of the nature of the science (e.g. voice, facsimile or data) or the performance characteristics of the connection (e.g. 32 kbps voice or 64 kbps voice).
3. It is independent of routing, i.e. the numbering or addressing does not specify the intermediate exchanges through which the service is to be put through. In contrast, some addressing schemes in data networks demand that the complete route from source to destination be specified as part of the address. UNIX based networks are typical examples of this.
4. It is a sequence of decimal digits. No alphabet or other characters are permitted as part of the address.
5. Its design is such that interworking between ISĐNs requires only the use of ISĐN number and no other additional digits or addressing signals.

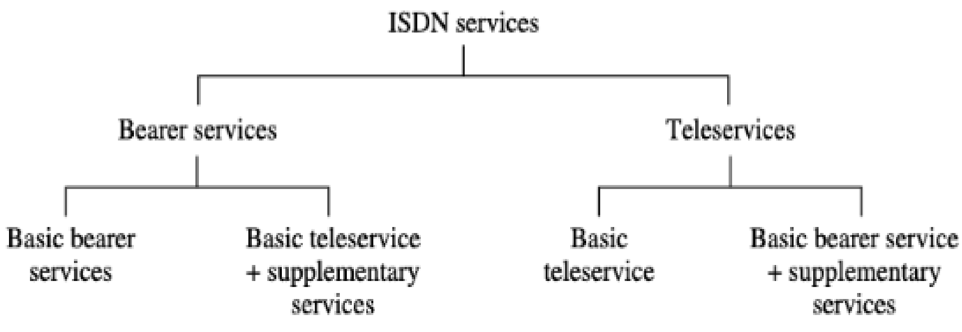
SERVICE CHARACTERISATION

In Section we described some of the new services that may be supported on ISDNs. ISDN services are placed under two broad categories:

. Bearer Services

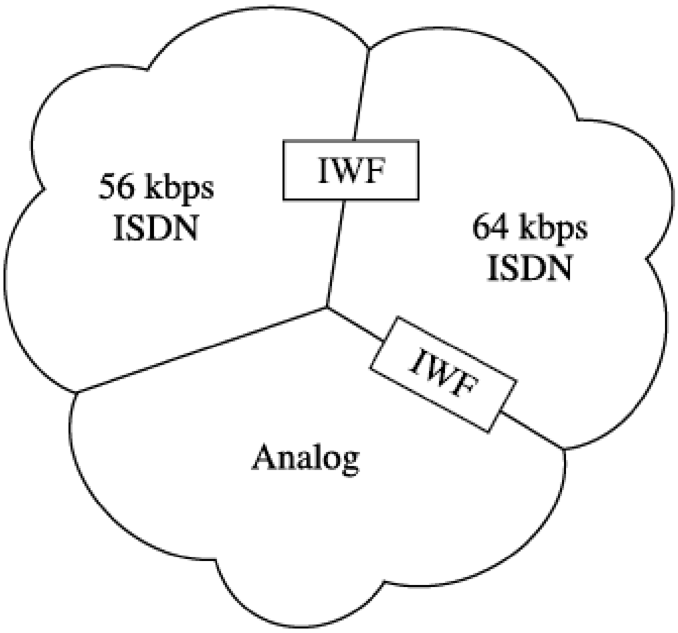
. Tele services.

In Section it was stated that the ISDN protocol architecture is layered according to the OSI reference model. Bearer services correspond to the functionalities offered by layers 1—3 (see Figure 11.10). Provision of these three lower layer functionalities is generally the responsibility of the network provider (subject to the regulations in force in a country as mentioned in Section 11.5). It is for this reason that the functions are termed bearer service functions, the bearer being the network provider. Bearer services are accessible at T and S reference points for ISDN compatible equipment (see Figure 11.11). For non ISDN terminals the bearer services are accessible at R reference point of the user-network interface. The upper laver functions, viz, layers 4—7, support tele services which employ the bearer services to transport information as per the requirements of the specific tele services offered. A packet data transmission is a typical example of a bearer service, whereas telephony, teletex. videotex and facsimile fall in the class of teleservices. Both the bearer service and the teleservice functionalities may be enhanced by adding to the basic service, the fbnctionalities of what are known as supplementary services. Supplementary services cannot stand alone and are always offered in conjunction with either a bearer service or a teleservïce. Figure 11.17 summarises the ISDN service. Supplementary services call for additional functionalities both in the lower layers and in the upper layers, depending on whether they supplement a basic bearer service or a basic teleservice.



INTERWORKING

ISDN has to coexist with other networks and this is essential for achieving market penetration and business success. Figure depicts the situation.



IWF = interworking functions

Typical interworking functions to ensure interoperability between ISDN and other networks

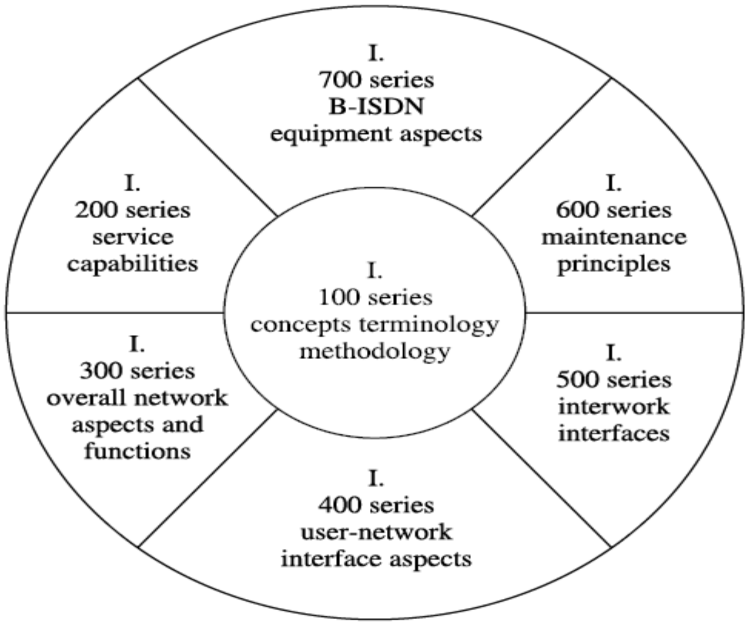
1. Determine if the network resources of the non ISDN network are adequate to meet the ISDN service demand.
2. In case the network resources are inadequate, take suitable action to cancel the call. reroute the call or renegotiate with ISDN to be consistent with the available resources.
3. Map signalling messages between the two (ISDN and non ISDN) signalling systems.
4. Ensure sen-ice and connection compatibility such as compatibility of call process tones, call failure mdication and signal processing modules like echo canceller.
5. Provide transmission structure conversion, including modulation technique and frame structure conversions.
6. Provide error and flow control.
7. Collect data for billing and charging.
8. Coordinate operation and maintenance procedures to be able to isolate faults.
9. Enable interworking of the numbering schemes.

ISDN STANDARDS

Standardisation is an essential process in the introduction of any major and complete international service. The capability of providing true international connectivity and interoperability between networks is critically dependent on the availability of standards and the strict adherence to them. The importance of standards has been well recognised in the context of ISDN from the very early stages. CCITT has been playing a leading role and acting as a coordinating body by issuing ISDN related recommendations and thereby guiding the introduction of ISDN internationally. The first definition of ISDN appeared in CCITT‘s recommendation, G.702 issued in 1972. Subsequent studies led to the emergence of the first ISDN standard: G.705 in 1980.

The contents of the G.702 and G.705 have been presented in the beginning of this chapter. The conceptual principles on which ISDN should be based are laid down in G.705.

ISDN was declared as the major concern of CCITT during the study period 1981—1984, and considerable efforts were put in to evolve standards for ISDN. These efforts culminated in a comprehensive set of recommendations in 1984 (although many were incomplete), known as I-Series recommendations. Figure 11.20 shows the general structure of I-Series recommendations.



Structure of ISDN recommendations

There are four others, teletex. facsimile. videotex. and message handling that have been standardised À CCITT service is said to be completely standardised only when

1. end-to-end compatibility is guaranteed;
2. terminals to provide the service are standardised;
3. procedures for obtaining the service are specified;
4. service subscribers are listed in an international directory:
5. testing and maintenance procedures are standardised; and
6. charging and accounting rules are spelt out .

The main aim of 1.200 series is to provide an unambiguous description of service features but at the same time permit complete flexibility and freedom in the implementation of these services. In the past, definitions used in the context of different networks to mean broadly the same service bave had subtle differences that often turned out to be important. 1.200 series is an attempt to ensure that such problems do not arise in the future.

The 1.300 series which deals with network capabilities discusses the protocol reference model, numbering and addressing principles, etc. The 1.300 series maps the various services defined in 1.200 series into elementary network connections and functions. The protocol reference model is an extension of the open system interconnection model described in X.200 series of recommendations. There are three major extensions:

1. Separation of signalling and management operations from the flow of application information within a piece of equipment
2. Definition of communication contexts that operate independent of each other
3. Application of extensions 1 and 2 to internal network components. Other extensions include multipoint and asymmetric connections, changeover change back of signalling links, and communications for maintenance and network management.

5.12 BROADBAND ISDN

Broadband ISDN (BISDN) is defined as a network capable of supporting data rates greater than the primary rate (1S44 or 2.048 Mbps) supported by ISDN. In the context of BISDN, the original ISDN concept is often termed narrowband ISDN (NISDN). The main aim of BISDN is to support video and image services. BISDN services are broadly classified as

. Interactive services

. Distribution services.

Interactive services may be classified as

1. Conversational sen-ices

2. Messaging services

3. Retrieval services.

Distribution services are classified as

. Broadcast services

. Cyclic services.

Conversational services support end-to-end information transfer on real time, bidirectional basis. There is a wide range of applications that may be supported using conversational services, the most important one being th video telephony or videophone. In this science the telephone instrument bas the capability to transmit, receive and display video signals. A dial-up connection brings about both video and audio transmission. Other applications include video conferencing and video surveillance. A number of data oriented conversational applications may also be supported. These include distributed databases, program downloading, inter process communication and large volume, high speed data exchange as encountered in CAD CAM, or graphics based applications. Messaging services offer store and forward communication. Analogous to X.400 messaging services on ISDN. voice mail, video mail and document mail containing text, graphics etc. may become the important messaging services on BISDN.

5.14 VOICE DATA INTEGRATION

Before evaluation of BISDN many research investigations have been undertaken to study the issues related to the voice data integration. Lots of study has only considered the integration two types of traffic: digitised voice and data. In this section. we discuss some of the schemes that have been proposed for integration of voice and data on a single channel. Selection of voice and data traffic for integration studies is not accidental. The traffic characteristics and requirements of voice and data are so different that finding an efficient solution here well turn out to be the key for solving the integration problem. Certain differences in the characteristics of voice and data traffic are given in Table Other important parameters of voice and data traffic are summarised in Table.

Digitised voice	Data
Periodic bursty in nature	Aperiodic bursty in nature
Fixed length bursts	Variable length bursts
Small packet size	Large packet size
Packetisation time critical	Packetisation time is not critical
Hard bound on delay	Soft bound on delay
Hard bound on the variance of delay	Soft bound on the variance of delay
Loss of parts of speech acceptable	Loss of parts of data unacceptable
Low overhead as there is no error recovery	High overhead due to error detection and recovery