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TELECOM TECHNOLOGIES AND NETWORKING TECHNIQUES

M-250

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- 4G Features, Mobile communication Technologies, IP based mobile communication Systems,

UNIT I: ELECTRONIC EXCHANGES, TELEPHONE TRAFFIC ENGINEERING AND DIGITAL SWITCHING

NOTES

★ STRUCTURE ★

LEARNING OBJECTIVES

After going through this unit, you will be able to:

- define evaluation of telecommunication
- describe advantages of electronic exchange over electromechanical exchanges
- know about basic concept of telephone traffic engineering
- explain briefly measurement of telephone traffic
- define grade of service
- explain scanning method
- describe basic principles of electronic exchanges
- know about digital switching

PART I: INTRODUCTION TO ELECTRONIC EXCHANGES

NOTES

1.1 INTRODUCTION

To overcome the limitations of manual switching; automatic exchanges, having Electro mechanical components, were developed. Strowger exchange, the first automatic exchange having direct control feature, appeared in 1892 in La Porte (Indiana). Though it improved upon the performance of a manual exchange it still had a number of disadvantages, viz., a large number of mechanical parts, limited availability, inflexibility, bulky in size etc. As a result of further research and development, Crossbar exchanges, having an indirect control system, appeared in 1926 in Sundsvall, Sweden. The Crossbar exchange improved upon many short- comings of the Strowger system. However, much more improvement was expected and the revolutionary change in field of electronics provided it. A large number of moving parts in Register, marker, Translator, etc., were replaced en-block by a single computer. This made the exchange smaller in size, volume and weight, faster and reliable, highly flexible, noise-free, easily manageable with no preventive maintenance etc.

1.2 EVALUATION OF TELECOMMUNICATION

The first electronic exchange employing Space-Division switching (Analog switching) was commissioned in 1965 at Succasunna, New Jersey. This exchange used one physical path for one call and, hence, full availability could still not be achieved. Further research resulted in development of Time-Division switching (Digital Switching) which enabled sharing a single path by several calls, thus providing full availability. The first digital exchange was commissioned in 1970 in Brittany, France.This handout reviews the evolution of the electronic exchanges, lists the chronological developments in this field and briefly describes the facilities provided to subscribers, administration and maintenance personnel.

Analog Exchange

TABLE 1.1 *Chronological development of electronic exchanges*

1965	No.1 ESS	Local	Bell Labs, USA
1972	D 10	Local and Transit	NEC. Japan.
1973	Metaconta	Local	LMT. France
1974	No. 1 ESS Centrex	Local and Transit	Bell Labs. USA
1975	Proteo	Local & Transit	Proteo, Italy
1976	AXE	Local	PTT & LM Ericsson, Sweden
1976	No.4 ESS	Transit	Bell Labs, USA
1978	AXE	Local	LM Ericsson, Sweden.

TABLE 1.2 Development of electronic exchanges

MODEL Analog	Capacity (in thousands)		Traffic	
	Lines	Trunks	Erlangs	Call Attempts Per Second
No. 1 ESS	10–65	-	6,000	30
No. 1 ESS	20–128	32	10,000	65

NO. 4 A XB ETS	-	22.4	6,200	35
No. 4 ESS	-	107	47,500	150
D 10	98	14.3	4,400	30
XE 1	-	13	2,500	3.6
EWSD	30	-	2,000	11–16
EWSP	-	13	5,000	-
TXE-4	40	-	5,000	50
Proteo	30	15	-	-
AXE	64	-	6,000	35
PRX-205	10	-	1,000	10–15
Digital Exchange				
E-10B	30	4	2,400	25
Mentaconta	10–60	-	10,000	28–60
MT 20	-	64	20,000	83–110
E 12	-	65	15,000	86
System X	100	60	25,000	800000
AXE -10	64	-	26,000	800000
FETEX-150L	290	60	24,000	1800000
OCB-283	200	60	25,000	800000
EWSD	250	60	25,200	1000000
No.5ESS	-	-		-
NEAX-61E	100	60	27,000	1000000

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1.3 ADVANTAGES OF ELECTRONIC EXCHANGE OVER ELECTROMECHANICAL EXCHANGES

Electromechanical Exchanges	Electronic Exchanges
Category, Analysis, Routing, translation, etc;, done by relays.	Translation, speech path Sub's Facilities, etc., managed by MAP and other DATA.
Any changes in facilities require addition of hardware and/or large amount of wiring change. Flexibility limited.	Changes can be carried out by simple commands. A few changes can be made by Subs himself. Hence, highly flexible.
Testing is done manually externally and is time consuming. No logic analysis carried out.	Testing carried out periodically automatically and analysis printed out.
Partial full-availability, hence blocking. Limited facilities to the subscribers.	Full availability; hence no blocking. A large number of different types of services possible very easily.
Slow in speed. Dialing speed is max. 11 Ips and switching speed is in l milliseconds.	Very fast. Dialing speed up to 11 digits/sec possible. Switching is achieved in a few microseconds.
Switch room occupies large volume.	Much lesser volume required floor space of switch room reduced to about one-sixth.
Lot of switching noise.	Almost noiseless.
Long installation and testing time. Large maintenance effort and preventive maintenance necessary.	Short installation and testing period. Remedial maintenance is very easy due to plug-in type circuit boards. Preventive maintenance not required.

PART II: BASIC CONCEPT OF TELEPHONE TRAFFIC ENGINEERING

1.4 BASIC CONCEPT OF TELEPHONE TRAFFIC ENGINEERING

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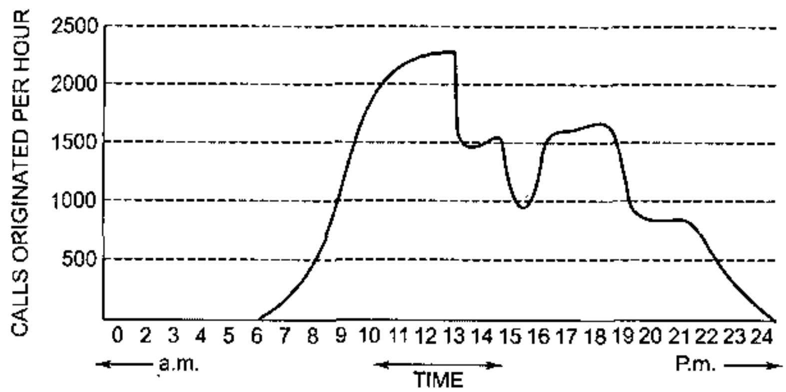
Telephone traffic is originated by the individual needs of different subscribers and so it is beyond the control of telephone administration. Any and every subscriber can originate a call at any and every moment without giving any previous information and the duration of calls is also not previously known. Although the individual telephone traffic originates at random, the average telephone traffic for a particular exchange follows the general pattern of activity in the exchange area. Normally there is a peak in morning, a dip during lunch period followed by a afternoon peak. In some localities the traffic has seasonal characteristic, for example at a holiday resort. A typical 24 hours variations in calling rate is shown below.



Whatever be the nature of variation of traffic, a telephone engineer is interested in maximum traffic that occurs in an exchange. The hour in which maximum traffic usually occurs in an exchange is known as **Busy Hour.** Busy Hour Traffic is the average value of maximum traffic in the busy hour. In computing Busy Hour Traffic the seasonal effects are also taken into account. Sometimes it is convenient to refer to **Busy hour calling rate (BHCR).** Busy hour calling rate is the number of calls originated per subscriber in the busy hour. This provides a simple means for designing the exchange with respect to the number of subscribers. It also provides probable growth of traffic to the estimated growth in number of subscribers. The busy hour calling rate may vary about 0.3 for a small country exchange and 1.5 or more for a busy exchange in business area in a city.

When the volume of traffic is quoted in terms of number of calls originated in a given time, this is insufficient to determine the consequent occupancy of lines and equipment. Therefore, measurement of traffic should not only consider number of calls but also their duration. The duration during which equipments and circuits are held when a call is made is called **HOLDING TIME.** Normally, it is average holding time per call for the particular item of equipment that is taken into account, so far as the caller is concerned the useful time is during the conversation only. However, the total time during which equipments and circuits are held when a call is made also includes, the period during which call is being established and time taken to release the equipment after the call has concluded.

1.5 MEASUREMENT OF TELEPHONE TRAFFIC

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The total cost of providing telephone service can be roughly divided into those charge which are constant and independent of volume of traffic and those, which are determined by the amount of traffic. The cost of subscriber's line and instrument and certain individual equipment in the exchange is totally independent of the volume of traffic. The quantity of common switching equipment required is almost entirely dependent by volume of traffic. The quantity of such equipment is dependent not only on number of calls but also on duration of calls. Therefore to determine the quantity of switching equipment in automatic exchange or staffing in manual exchange telephone traffic may be measured in terms of both the number of calls and the duration of calls.

For certain purpose it is sufficient to specify a **Traffic Volume** which is product of number of calls occurred during the time concerned by their average duration. however for the purposes of automatic exchange a more precise unit of traffic flow is required. this is called **Traffic Intensity**. Traffic intensity is the average number of calls simultaneously in progress. The unit of traffic intensity is **Erlang.**

A traffic intensity of one erlang is obtained in any specified period when the average number of calls simultaneously in progress during that period in unity. The specified period is always one hour and is taken as being the busy hour unless some other period is indicated.

There is a more precise way to define traffic intensity. The average Traffic Intensity during a specified period T, carried by a group of circuits or equipments, is given by the sum of the holding times divided by T. The holding times and period T all being expressed in the same unit.

Sometime it is stated that the average traffic intensity is equal to the average number of calls, which originate during the average holding time.All the above three definitions give the same numerical result.

The foregoing relationships may be expressed symbolically as follows.

Let S be sum of holding times during a given period **T,** both expressed in hours. Then by definition.

$$A = S/T$$

Where A is the average traffic intensity. Let **C** be the total number of calls during the period T then the average holding time '*t*' hours per call, is given by

$$t = S/C$$

Then $\quad A = S/T$ Can also be written as

$$A = Ct/T$$

It also follows that when the average call duration is known, the average call intensity can be obtained by determining the number of calls occurring during the period T. Also because A is equal to average number of calls simultaneously in progress, an approximate value of A can be obtained by counting the number of occupied circuits or equipments at uniform interval during the time T and finding the average value.

1.6 GRADE OF SERVICE

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Owing to the fact that calls originated in a pure chance manner, it is likely that during the busy hour some calls may fail to mature due to insufficiency of switching equipment. To ensure that the number of calls so lost is reasonably small, it is the standard practice switching equipment such that on the average not more than one call out of every 500 in the busy hour is lost at each switching stage, with the provision that loss does not fall below 1 in 100 with a 10 percent increase of traffic.

This allowable loss is termed the grade of service and is usually represented by the symbol 'B' with one lost call in 500 the grade of service is written as

$$B = 1/500 \text{ or } B = 0.002$$

The Grade of service is a factor employed for dimensions of the exchange equipment. A few typical problems are Worked out below to illustrate how the terms and definitions of telephone traffic are *actually applied in practice.*

Example 1. If the calling rate per line per day in an exchange of 5000 lines is 6.0 and proportion of the traffic that occurs in the busy hours is 12 percent, what is the busy hours traffic in Erlangs, assuming an average holding time of 2.5 minutes per call?

Calling rate per line per day	= 6.0
Capacity of the exchange	= 50000 lines
Total number of calls made in a day	= 5000 × 6
	= 30,000
Number of calls originated in the hours	= 30,000 × 12/100
Holding time of a call	= 2.5 minutes
Busy hour traffic	= C × *t*/60
	= *3600* × 2.5/*60*
	= 150 Erlangs or T.u.s.

Example 2. A group of selectors observed for ten busy hours carried an average of twenty Erlangs and the total number of calls lost was twelve. The calls had an average duration of two minutes. What grade of service was given?

Traffic carried by the selectors in one busy hour	= 20Erlangs
Average holding time	= 2 minutes
Total number of calls carried in one busy hour	= 20 × 60/2
	= 600
Number of calls lost in ten busy hours	= 12
Average number of calls lost in one busy hour	= 12/10 = 1.2
Total number of calls offered in busy hour	= 600 + 1.2
	= *601.2*

$$\text{Grade of service} = \frac{\text{Number of calls lost}}{\text{number of calls offered}}$$

$$= 1.2/601.2$$

$$= 0.001996$$

$$\text{Say, } = 0.002$$

1.7 SCANNING METHOD

This is the practical method for measuring traffic in SPC switches.

Here the observation of traffic is not continuous. The group of equipments are scanned at regular intervals and the traffic flow is calculated.

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$$A = 1/S \sum_{v=1}^{s} Fv$$

or $$A = 1/S\, [f_1 + f_2 + f_3 + \ldots + f_s]$$

where A = Tele traffic intensity in Erlangs

S = Number of scans made on the group.

Fv = The number of occupied devices found in the v^{th} scan

Example 3. A group of equipments were scanned for ascertaining the traffic flow. The scanning was done once in 5 seconds for one minute. The number of occupied devices in each scan is as follows

1^{st} scan = 4,	2^{nd} scan = 3,	3^{rd} scan = 2
4^{th} scan = 3,	5^{th} scan = 1,	6^{th} scan = 3
7^{th} scan = 2,	8^{th} scan = 4,	9^{th} scan = 3
10^{th} scan = 5,	11^{th} scan = 4,	12^{th} scan = 2

Calculate the intensity of traffic.

Duration of observation = 60*s*

Frequency of scanning = 5*s*

Number of scans = 12

$$A = \frac{1}{S}\, [f_1 + f_2 + f_3 + \ldots + f_{12}]$$

$$= \frac{1}{12} * 36 = 3 \text{ Erlangs}$$

1.8 QUANTITATIVE INDICATORS FOR QUALITY OF SERVICE

The quality of service of a telecommunications network is characterized by the level of satisfaction of the customers connected to it. There are a number of technical and customer services indicators that determine the quality of service. Technical performance indicators encompass reliability (fault rate and time to clear faults), connectivity (dial tone delay and call completion rates) and operator response time for booking calls (manual operations). Specific technical performance indicators are:

(*a*) fault rate, that is number of faults per main line per year;

(*b*) average number of lines faulty any day as percent (%) of total main lines;

(*c*) percent (%) of faults cleared by next working day;

(*d*) dial tone delay, that is time (in seconds) before dial tone received after call is originated;

(*e*) call completion rates, that is percent (%) of originated calls successfully completed; and

(*f*) time to answer for operator service.

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Fault Rate

The number of faults per main line per year defines the frequency of breakdown of the telephone lines. For a well constructed and well maintained network, the average number of faults per main line per year should be 0.2 or less; that is the telephone line should not be out of order more than once in five years. Because the figure is normally small in industrialized countries, this indicator is often expressed in faults per 100 main lines. The actual situation in developing countries is much worse, with the average number of faults in some countries exceeding three faults per main line per year.

The number of lines faulty on any day as percent (%) of total lines in service is an important performance indicator for the company because it actually represents the percentage of the network that is not generating revenues at any particular time. This indicator is closely related to the fault rate and the time to clear.

Fault Clearance

The time to clear faults is normally expressed in terms of the percentage of reported faults cleared within a given time. The significant time frame normally applied is "by next working day".

Call Completion Rate

The Call Completion Rate (CCR) measures the percentage of originated calls successfully completed. The CCR, which is normally measured during the peak traffic hour, is an indication of the probability of establishing a connection at the end of dialing. In practice, dialing can commence only after the dial tone is received; hence, connectivity also depends on the availability of a dial tone, the ability of the network to establish a transmission path between the calling and the called party and to switch the call to the called party. The network components involved for a local call are:

(*a*) the customer premises equipment (terminal equipment such as a telephone and indoor wiring);

(*b*) the local cable network; and

(*c*) the local switching equipment.

For domestic long distance calls, in addition to the above equipment, long-distance switching equipment and transmission media and equipment are required while for international calls, international switching equipment and transmission media and equipment are required. Hence, the CCR for the international calls depends on the quality of the *total* network - local, domestic long-distance and international.

A successful call could be defined in two ways. First, the call could be considered as successfully completed only if the called party answers and communication (voice, data, fax, etc.) is established. Another interpretation of a successful call could be establishing a connection successfully to the called number although the called party may not answer. In respect of telephone calls, the called party may not answer because of a number of reasons including:

(*a*) called party is not available near the phone and hence the phone keeps on ringing without an answer. In the age of answering machines, the probability of not receiving an answer is low; and (*b*) called line is busy and therefore the telephone at the called number does not actually ring. The probability of this happening is also being reduced through use of "Call Waiting" facility by many users.

The CCR reflects ***directly the degree of congestion*** in the network and *indirectly* the *fault rate*. The CCR depends on the equipment available to switch and transmit the signaling messages. The equipment may not be available either because of under dimensioning in which case the available equipment is not adequate to handle the traffic, or faulty equipment which would cause the same effect. In many developing countries, the poor CCR is mainly due to faulty switching equipment; however, because of poor maintenance, the outside plant network could also contribute to the poor CCR.

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In the international network, the CCR has been further categorized into:

(*a*) Answer Bid Ratio (ABR);

(*b*) Answer Seizure Ratio (ASR); and

(*c*) Congestion (CONG).

The *ABR* is the ratio of successful calls to total originating international calls. The ratio is the measure of effective international calls, reflects the performance of the total international network between the calling and called country and hence is the CCR for the entire international network or the probability of a call being successful. The *ASR* is the ratio of successful calls to total incoming international calls. It is a measure of the performance of the called country's telephone network and hence reflects its CCR. The *CONG* is the percentage of calls lost due to congestion in the international network. It is a measure of the inadequacy in the number of international circuits between the two countries.

PART III: BASIC PRINCIPLES OF ELECTRONIC EXCHANGES

1.9 BASIC PRINCIPLES OF ELECTRONIC EXCHANGES

The prime purpose of an exchange is to provide a temporary path for simultaneous. bidirectional transmission of speech between

(*i*) Subscriber lines connected to same exchange (local switching)

(*ii*) Subscriber lines and trunks to other exchange (outgoing trunk call)

(*iii*) Subscriber lines and trunks from other exchanges (incoming trunk calls) and

(*iv*) Pairs of trunks towards different exchanges (transit switching)

These are also called the switching functions of an exchange and are implemented through the equipment called the switching network. An exchange, which can setup just the first three types of connections., is called a Subscriber or Local Exchange. If an exchange can setup only the fourth type of connections, it is called a Transit or Tandem Exchange. The other distinguished functions of an exchange are

(*i*) Exchange of information with the external environment (Subscriber lines or other exchanges) *i.e.*, signaling.

(*ii*) Processing the signaling information and controlling the operation of signaling network, *i.e.*, control, and (*iii*) Charging and billing

All these functions can be provided more efficiently using computer controlled electronic exchange, than by the conventional electromechanical exchanges.

This handout describes the basic principals of SPC exchanges and explains how the exchange functions are achieved.

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1.10 STORED PROGRAMME CONTROLLED EXCHANGE

In electromechanical switching, the various functions of the exchange are achieved by the operation and release of relays and switch (rotary or crossbar) contacts, under the direction of a control sub-system. These contacts are hard—wired in a predetermined way. The exchange dependent data, such as, subscriber's class of service, translation and routing, combination signaling characteristics, are achieved by hard-ware and logic, by a set of relay, grouping of same type of lines, strapping on main or intermediate distribution frame or translation fields, etc. When the data is to be modified, for introduction of a new service, or change in services already available to a subscriber, the hardware change ranging from inconvenient to near impossible, are involved.

In an SPC exchange, a processor similar to a general purpose computer, is used to control the functions of the exchange. All the control functions, represented by a series of various instructions, are stored in the memory. Therefore the processor memories hold all exchange-dependent data. such as subscriber date, translation tables, routing and charging information and call records. For each call processing step, *e.g.*, for taking a decision according to class of service, the stored data is referred to, Hence, this concept of switching. The memories are modifiable and the control program can always be rewritten if the behavior or the use of system is to be modified. This imparts and enormous flexibility in overall working of the exchange.

Digital computers have the capability of handling many tens of thousands of instructions every second, Hence, in addition to controlling the switching functions the same processor can handle other functions also. The immediate effect of holding both the control programme and the exchange data, in easily alterable memories, is that the administration can become much more responsive to subscriber requirements. both in terms of introducing new services and modifying general services, or in responding to the demands of individual subscriber. For example, to restore service on payment of an overdue bill or to permit change from a dial instrument to a multi frequency sender, simply the appropriate entries in the subscriber data-file are to be amended. This can be done by typing- in simple instructions from a teletypewriter or visual display unit. The ability of the administration to respond rapidly and effectively to subscriber requirements is likely to become increasingly important in the future.

The modifications and changes in services which were previously impossible be achieved very simply in SPC exchange, by modifying the stored data suitably. In some cased, subscribers can also be given the facility to modify their own data entries for supplementary services, such as on-demand call transfer, short code, (abbreviated) dialing, etc.

The use of a central processor, also makes possible the connection of local and remote terminals to carry out man-machine dialogue with each exchange. Thus,

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the maintenance and administrative operations of all the SPC exchanges in a network can be performed from a single centralised place. The processor sends the information on the performance of the network, such as, traffic flow, billing information, faults, to the centre, which carries out remedial measures with the help of commands. Similarly, other modifications in services can also be carried out from the remote centre. This allows a better control on the overall performance of the network.

As the processor is capable of performing operations at a very high speed, it has got sufficient time to run routine test programmes to detect faults, automatically. Hence, there is no need to carry out time consuming manual routine tests.

In an SPC exchange, all control equipment can be replaced by a single processor. The processor must, therefore, be quite powerful, typically, it must process hundreds of calls per second, in addition to performing other administrative and maintenance tasks. However, totally centralised control has drawbacks. The software for such a central processor will be voluminous, complex, and difficult to develop reliably. Moreover, it is not a good arrangement from the point of view of system security, as the entire system will collapse with the failure of the processor. These difficulties can be overcome by decentralising the control. Some routine functions, such as scanning, signal distributing, marking, which are independent of call processing, can be delegated to auxiliary or peripheral processors. These peripheral units, each with specialised function, are often themselves controlled by a small stored programmes processors, thus reducing the size and complexity at central control level. Since, they have to handle only one function, their programmes are less voluminous and far less subjected to change than those at central. Therefore, the associated programme memory need not be modifiable (generally, semiconductors ROM's are used).

1.11 BLOCK SCHEMATIC OF SPC EXCHANGE

Despite the many difference between the electronic switching systems, and all over the world there is a general similarity between most of the systems in terms of their functional subdivisions. In it's simplest form an SPC exchange consists of five main sub-systems, as shown in fig.

(*i*) Terminal equipment, provides on individual basis for each subscriber line and for interexchange trunk.

(*ii*) Switching network, may be space-division or time-division, unidirectional or bidirectional.

(*iii*) Switching processor, consisting mainly of processors and memories.

(*iv*) Switching peripherals (Scanner, Distributor and Marker,) are interface circuits between control system terminal equipment and switching network.

(*v*) Signaling interfaces depending on type of signaling used, and

(*vi*) Data processing peripherals (Teletypewriters, Printers, etc.) for man-machine dialogue for operation and maintenance of the exchange.

1. Terminal Equipment

In this equipment, line, trunk, and service circuits are terminated, for detection, signaling, speech transmission, and supervision of calls. The Line Circuits carry out the traditional functions of supervising and providing battery feed to each subscriber line. The Trunk Circuits are used on outgoing, incoming and transit calls for battery feed and supervision. Service Circuits perform specific functions,

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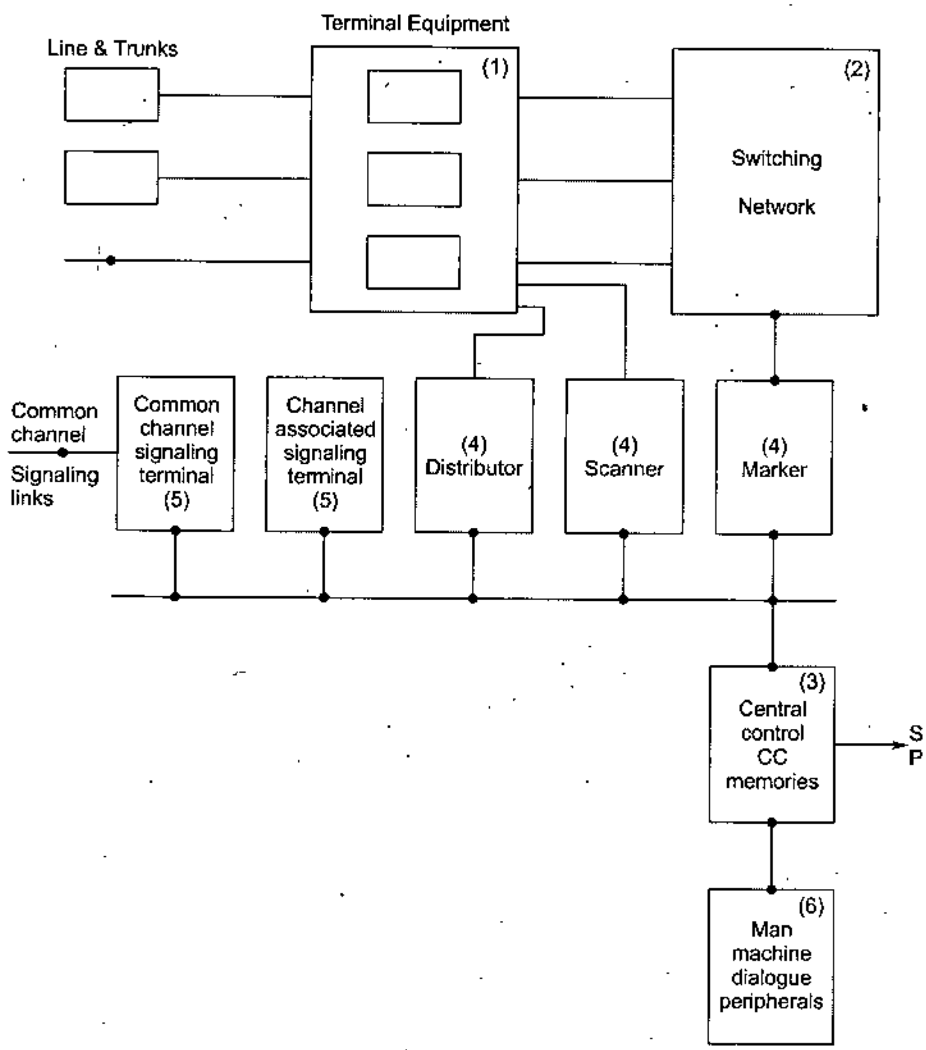

FIG. 1.1 Functional subdivisions of an SPC exchange

like, transmission and reception of decadic dial pulses or MF signals, which may be economically handled by a specialised common pool of circuits. In contrast to electromechanical circuits, the Trunk and Service circuits in SPC exchanges, are considerably simpler because functions, like counting, pulsing, timing charging, etc., are delegated to stored programme.

2. Switching Network

In an electronic exchange, the switching network is one of the largest sub-system in terms of size of the equipment. Its main functions are,

(*i*) Switching, *i.e.*, setting up temporary connection between two or more exchange terminations, and

(*ii*) Transmission of speech and signals between these terminations, with reliable accuracy.

There are two types of electronic switching system. viz. Space division and Time division.

A. Space Division Switching System

In a space Division Switching System, a continuous physical path is set up between input and output terminations. This path is separate for each connection and is held for the entire duration of the call. Path for different connections is independent of each other. Once a continuous path has been established. Signals are interchanged between the two terminations. Such a switching network can employ either metallic or electronic cross-points. Previously, usage of metallic cross-points, viz., reed relay, mini-cross bar derivative switches, etc., were favored. They have the advantage of compatibility with the existing line and trunk signaling conditions in the network.

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B. Time Division Switching System

In Time Division Switching, a number of calls share the same path on time division sharing basis. The path is not separate for each connection, rather, is shared sequentially for a fraction of a time by different calls. This process is repeated periodically at a suitable high rate. The repetition rate is 8 kHz, *i.e.*, once every 125 microseconds for transmitting speech on telephone network, without any appreciable distortion. These samples are time multiplexed with staggered samples of other speech channels, to enable sharing of one path by many calls. The Time Division Switching was initially accomplished by Pulse Amplitude Modulation (PAM) Switching. However, it still could not overcome the performance limitations of signal distortion noise, cross-talk etc. With the advent of Pulse Code Modulation (PCM), the PAM signals were converted into a digital format overcoming the limitations of analog and PAM signals. PCM signals are suitable for both transmission and switching. The PCM switching is popularly called Digital Switching.

C. Compatibility with Existing Network

In this area, the application of electronic techniques has encountered the greatest difficulty. To appreciate the reasons, let us consider the basic requirements of a conventional switching network.

- High OFF resistance and low ON resistance.
- Sufficient power handling capacity for transmitting ringing current, battery feed etc..., on subscriber lines.
- Good frequency response (300–3400 kHz)
- Bidirectional path (preferable)
- D.C. signaling path to work with existing junction equipment (preferable)
- Economy
- Easy to control.
- Low power consumption, and
- Immunity to extraneous noise, voltage surges.

The present day electronic devices cannot meet all these requirements adequately. It is seen that requirement (*iii*), (*v*), (*vi*) and (*vii*) only, can easily be met by electronic devices. These considerations show that substitutions of the analog mode of electromechanical switching network by fully electronic equipment is not, straight way practical. The main virtue of the existing electromechanical devices is their immunity to extraneous noise voltage surge, etc., which are frequently experienced in a telephone network. Moreover, metal contact switches offer little restriction on the voltages and currents to be carried. In the existing network and subscriber handsets, typically, 80 volt peak to peak ringing current is required to be transmitted on the line. This is difficult, if not impractical, for electronic switches to handle. Therefore, to avail of the advantages of the electronic exchanges, either of the two following alternatives may be adopted.

(i) Deploy a new range of peripherals/ equipments, suited to the characteristics of the electronic switching devices, on one hand, and requirements of telephone network on the other hand. *i.e.*, employ Time Division Switching systems, or

(ii) Continue to use metal contact switches, while other sub-systems may be changed to electronic. *i.e.*, semi-electronic type of exchanges rather than fully electronic exchanges, to employ Space Division Switching Systems.

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3. Switching Processor

The switching processor is a special purpose real time computer, designed and optimised for dedicated applications of processing telephone calls. It has to perform certain real time functions (which have to be performed at the time of occurrence and cannot be deferred), such as, reception of dialed digits, and sending of digits in case of transit exchange. The block schematic of a switching processor, consisting of central control programme store is shown in Figure 1.2.

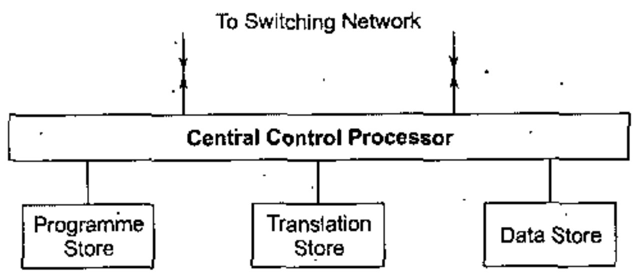

Fig. 1.2 Switching processor

Central Control (CC) is a high speed data processing unit, which controls the operation of the switching network. In Programme store, sets of instructions. called programmes, are stored. The programmes are interpreted and executed by the central control. Data Store provides for the temporary storage of transient data, required in processing telephone calls, such as digits dialed by the subscriber, busy/idle states of lines and trunks etc. Translation Store contains information regarding lines. *e.g.*, category of calling and called line. routing code, charging information, etc. Data Stores is temporary memory, whereas Translation and Programme Stores are of semi-permanent type. The information in the Semi-permanent memories does not change during the processing of the call, but the information in Data Store changes continuously with origination and termination of each call.

4. Switching Peripheral Equipment

The time intervals, in which the processor operates, is in the order of microseconds, while the components in the telephone switching section operate in milliseconds (if the switching network is of the analog type). The equipment, known as the switching peripheral, is the interface between these two equipments working at different speeds. The interface equipment acts as speed buffer, as well as, enables conversion of digital logic signals from the processor to the appropriate electrical signals to operate relays and cross-points, etc. Scanners, Signal distributors and Marker fall under this category of devices.

A. Scanner

Its purpose is to detect and inform CC of all significant events/signals on subscriber lines and trunks. connected to the exchange. These signals may either be continuous or discrete. The equipments at which the events/signals must be detected are equally diverse.

(*i*) Terminal equipment for subscriber lines and inter-exchange trunks and

(*ii*) Common equipment such as DTMF (Dual-Tone Multi Frequency) or MFC digit receivers and inter-exchange signaling senders/receivers connected to the lines and trunks.

In view of this wide diversity in the types of lines. trunks and signaling, the scanning rate, *i.e.*, the frequency at which scan points are read, depends upon the maximum rate at which events/signals may occur. For example, on a subscriber line, with decadic pules signaling with 1:2 make-break ratio, the necessary precision, required for pulse detection, is of the order of ten milliseconds, while other continuous signals (clear, off hook, etc.) on the same line are usually several hundred milliseconds long and the same high precision is not required. To detect new calls, while complying with the dial tone connection specifications, each line must be scanned about every 300 milliseconds. It means that in a 40,000 lines exchange (normal size electronic exchange) 5000 orders are to be issued every 300 milliseconds, assuming that eight lines are scanned simultaneously.

B. Marker

Marker performs physical setup and release of paths through the switching network, under the control of CC. A path is physically operated only when it has been reserved in the central control memory. Similarly, paths are physically released before being cleared in memory, to keep the memory information updated vis-a-vis switching network, Depending upon whether is switching is Time division or Space division, marker either writes information in the control memory of time and space stages. (Time Division Switching), or physical operates the cross-points (Space Division Switching).

C. Distributor

It is a buffer between high-speed-low-power CC and relatively slow-speed-high-power signaling terminal circuits. A signal distributor operates or releases electrically latching relays in trunks and service circuits, under the direction of central control.

D. Bus System

Various switching peripherals are connected to the central processor by means of a common system. A bus is a group of wires on which data and commands pulses are transmitted between the various sub-units of a switching processor or between switching processor and switching peripherals. The device to be activated is addressed by sending its address on the address bus. The common bus system avoids the costly mesh type of interconnection among various devices.

E. Line Interface Circuits

To enable an electronic exchange to function with the existing outdoor telephone network, certain interfaces are required between the network and the electronic exchange.

- **Analogue Subscriber Line Interface:** The functions of a Subscriber Line Interface, for each two wire line, are often known by the acronym : BORSHT

B	:	Battery feed
O	:	Overload protection
R	:	Ringing
S	:	Supervision of loop status
H	:	Hybrid
T	:	Connection to test equipment

NOTES

All these functions cannot be performed directly by the electronic circuits and, therefore, suitable interfaces are required.

- **Transmission Interface:** Transmission interface between analogue trunks and digital trunks (individual or multiplexed) such as, A/D and D/A converters, are known as CODEC, These may be provided on a per-line and per-trunk basis or on the basis of one per 30 speech channels.
- **Signaling Interfaces:** A typical telephone network may have various exchange systems (Manual, Strowger, Cross bar, Electronic) each having different signaling schemes. In such an environment, an exchange must accommodate several different signaling codes.
- **Signaling:** Initially, all signaling between automatic exchanges was decadic *i.e.,* telephone numbers were transmitted as trains of 1 to 10 pulses, each train representing one digit. To increase the speed at which the calls could be set up, and to improve the reliability of signaling, compelled sequence multi frequency signaling system was then introduced. In this system, each signal is transmitted as a combination of 2 out of a group of say 5 or 6 frequencies. In both decadic and multi frequency methods, the signals for each call are sent over a channel directly associated with the inter-exchange speech transmission circuit used for that call. This is termed as channel associated signaling. Recently, a different technique has been developed, known as common channel signaling. In this technique, all the signaling information for a number of calls is sent over a signaling link independent of the interexchange speech circuits. Higher transmission rate can be utilised to enable exchange of much larger amount of information. This results in faster call setup, introduction of new services, *e.g.,* abbreviated dialing, and more retrials ultimately accomplishing higher call completion rate, moreover, it can provided an efficient means of collecting information and transmitting orders for network management and traffic engineering.
- **Data Processing Peripherals**: Following basic categories of Data Processing Peripherals are used in operation and maintenance of exchange.
    (*i*) Man - machine dialogue terminals, like Teletypewriter (TTY) and Visual Display Units (VDU), are used to enter operator commands and to give out low-volume date concerning the operation of the switching system. These terminals may be local *i.e.,* within a few tense of meters of the exchange, or remotely located. These peripherals have been adopted in the switching Systems for their ease and flexibility of operation.
    (*ii*) Special purpose peripheral equipment is, sometimes employed for carrying out repeated functions, such as, subscriber line testing, where speed is more important than flexibility.
    (*iii*) High speed large capacity data storage peripherals (Magnetic Tape Drives, magnetic Disc Unit) are used for loading software in the processor memory.

(*iv*) Maintenance peripherals, such as, Alarm Annunciators and Special Consoles, are used primarily to indicate that automatic maintenance procedure have failed and manual attention is necessary.

1.12 CONCLUSION

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The electronic exchanges work on the principle of Stored Programme Control. All the call processing functions are performed on the basis of pre-designed programme which is stored in the memory of the Central Processor. Though the initially designed Electronic Exchanges had single centralised processor. the control is being decentralised, providing dedicated micro-processor controlled sub-systems for improved efficiency and security of the system. This modular architecture also aids future expansions.

PART IV: DIGITAL SWITCHING

1.13 DIGITAL SWITCHING

A Digital switching system, in general, is one in which signals are switched in digital form. These signals may represent speech or data. The digital signals of several speech samples are time multiplexed on a common media before being switched through the system.

To connect any two subscribers, it is necessary to interconnect the time-slots of the two speech samples which may be on same or different PCM highways. The digitalised speech samples are switched in two modes, viz., Time Switching and Space Switching. This Time Division Multiplex Digital Switching System is popularly known as Digital Switching System.

In this handout, general principles of time and space switching are discussed. A practical digital switch, comprising of both time and space stages, is also explained.

1.14 TIME AND SPACE SWITCHING

Generally, a digital switching system several time division multiplexed (PCM) samples. These PCM samples are conveyed on PCM highways (the common path over which many channels can pass with separation achieved by time division). Switching of calls in this environment, requires placing digital samples from one time-slot of a PCM multiplex in the same or different time-slot of another PAM multiplex.

For example, PCM samples appearing in TS6 of I/C PCM HWY1 are transferred to TS18 of O/G PCM HWY2, via the digital switch, as shown in Figure 1.3.

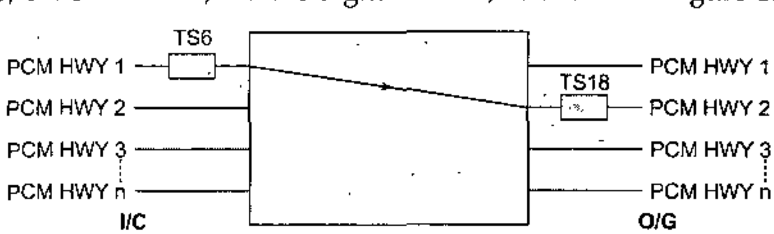

Fig. 1.3 Digital switch

The interconnection of time-slots, *i.e.*, switching of digital signals can be achieved using two different modes of operation. These modes are:

(*i*) Space Switching

(*ii*) Time switching

Usually, a combination of both the modes is used.

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In the space-switching mode, corresponding time-slots of I/C and O/G PCM highways are interconnected. A sample, in a given time-slot, TSi of an I/C HWY, say *HWY1*, is switched to same time-slot, *TSi* of an *O/G HWY, SAY HWY2. Obviously* there is no delay in switching of the sample from one highway to another highway since the sample transfer takes place in the same time-slot of the PCM frame.

Time Switching, on the other hand, involves the interconnection of different time-slots on the incoming and outgoing highways by re-assigning the channel sequence. For example, a time-slot TSx of an I/C Highway can be connected to a different time-slot., TSy, of the outgoing highway. In other words, a time switch is, basically, a time-slot changer.

1.15 DIGITAL SPACE SWITCHING

Principle

The Digital Space Switch consists of several input highways, X_1, X_2, ... X_n *and* several output highways, Y_1, Y_2, ... Y_m, inter connected by a crosspoint matrix of n rows and m columns. The individual crosspoint consists of electronic AND gates. The operation of an appropriate crosspoint connects any channel, a, of I/C PCM highway to the same channel, a', of O/G PCM highway, during each appropriate time-slot which occurs once per frame as shown in Figure 1.4. During other time-slots, the same crosspoint may be used to connect other channels. This crosspoint matrix works as a normal space divided matrix *with full availability* between incoming and outgoing highways during each time-slot.

Each crosspoint column, associated with one O/G highway, is assigned a column of control memory. The control memory has as many words as there are time-slot per frame in the PCM signal. In practice, this number could range from 32 to 1024. Each crosspoint in the column is assigned a binary address, so that only one crosspoint per column is closed during each time-slot. The binary addresses are stored in the control memory, in the order of time-slots. The word size of the control memory is x bits, so that $2x = n$, where n is the number of cross points in each column.

A new word is read from the control memory during each time-slot, in a cyclic order. Each word is read during its corresponding time-slot, *i.e.*,Word 0 (corresponding to TS0), followed by word 1 (corresponding to TS1) and so on. *The word contents are contained* on the vertical address lines for the duration of the time-slot. Thus, the cross point corresponding to the address, is operated during a particular time-slot. This cross point operates every time the particular time-slot appears at the inlet in successive frames. normally, a call may last for around a million frames.

As the next time-slot follows, the control memory is also advanced by one step, so that during each new time-slot new corresponding words are read from the various control memory columns. This results in operation of a completely different set of cross points being activated in different columns. Depending upon the number of time-slots in one frame, this time division action increases the utilisation of cross point 32 to 1024 times compared with that of conventional space-divided switch matrix.

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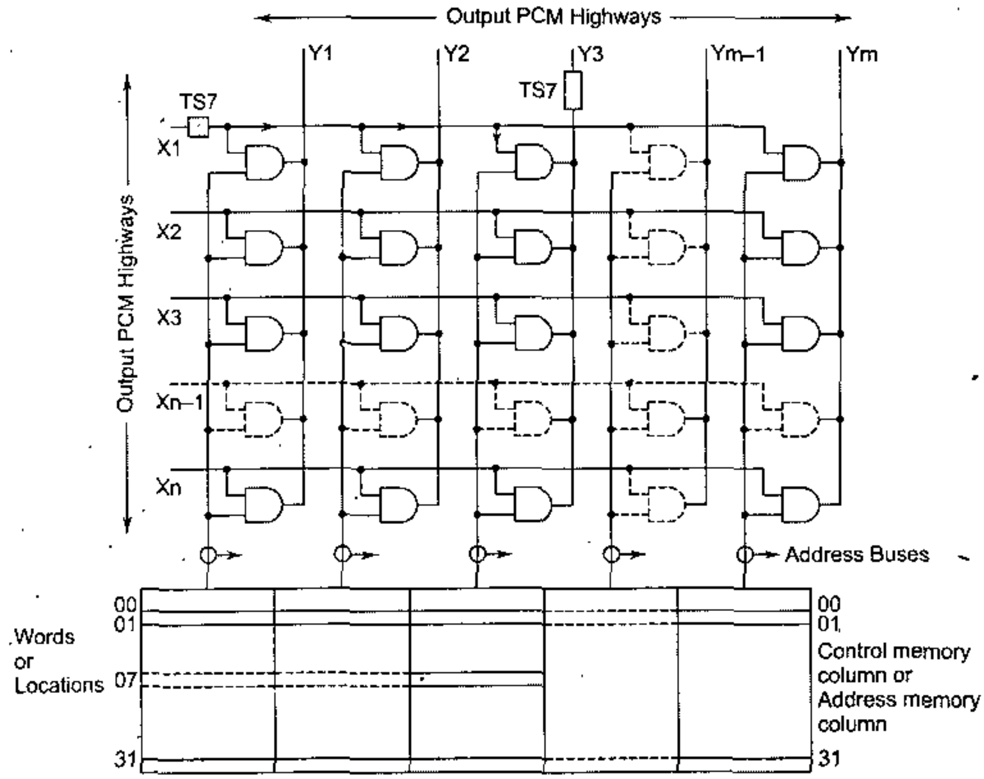

Fig. 1.4 Space switch

Illustration

Consider the transfer of a sample arriving in TS7 of I/C HWY X1 to O/G HWY Y3. Since this is a space switch, there will be no reordering of time *i.e.*, the sample will be transferred without any time delay, via the appropriate cross point. In other words, the objective is to connect TS7 of HWY X1 and TS7 of HWY Y3.

The central control (CC) selects the control memory column corresponding output highway Y3. In this column, the memory location corresponding to the TS7 is chosen. The address of the cross point is written in this location, *i.e.*, 1, in binary, is written in location 7, as shown in Figure 1.4.This cross point remains operated for the duration of the time-slot TS7, in each successive frame till the call lasts.

For disconnection of call, the CC erases the contents of the control memory locations, corresponding to the concerned time-slots. The AND gates, therefore, are disabled and transfer of samples is halted.

Practical Space Switch

In a practical switch, the digital bits are transmitted in parallel rather than serially, through the switching matrix.

In a serial 32 time-slots PCM multiplex, 2048 Kb/s are carried on a single wire sequentially, *i.e.*, all the bits of the various time-slots follow one another. This

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single wire stream of bits, when fed to Serial to Parallel Converter is converted into 8-wire parallel output. For example, all 8 bits corresponding to TS3 serial input are available simultaneously on eight output wires (one bit on each output wire), during just one bit period, as shown in Figure 1.5. This parallel output on the eight wires is fed to the switching matrix. It can be seen that during one full time-slot period, only one bit is carried on the each output line, whereas 8 bits are carried on the input line during this period. Therefore, bit rate on individual output wires, is reduced to 1/8th of input bit rate = 2048/8 = 256Kb/s.

Due to reduced bit rate in parallel mode, the cross point is required to be operated only for 1/8th of the time required for serial working. It can, thus, be shared by eight times more channels, *i.e.*, 32 × 8 = 256 channels, in the same frame.

However, since the eight bits of one TS are carried on eight wires, each cross point have eight switches to interconnect eight input wires to eight output wires. Each cross point (all the eight switches) will remain operated now for the duration of one bit only, *i.e.*, only for 488 ns (1/8th of the TS period of 3.9 μs).

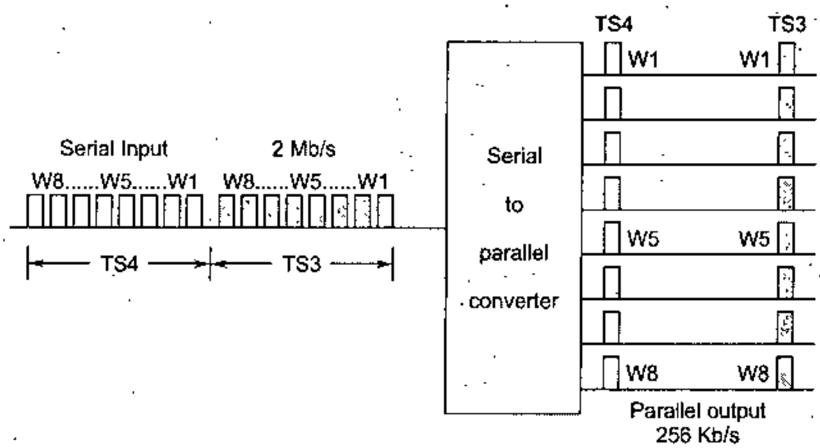

Fig. 1.5 Serial parallel converter

For example, to connect 40 PCM I/C highways, a matrix of 40 × 40 = 1600 cross points each having a single switch, is required in serial mode working. Whereas in parallel mode working, a matrix of (40/8 × 40/8) = 25 cross point is sufficient. As eight switches are required at each cross point 25 × 8 = 200 switches only are required. Thus, there is a reduction of the matrix by 1/8th in parallel mode working, hence reduction in size and cost of the switching matrix.

1.16 DIGITAL TIME SWITCH

Principle

A Digital Time Switch consists of two memories, viz., a speech or buffer memory to store the samples till destination time-slots arrive, and a control or connection or address memory to control the writing and reading of the samples in the buffer memory and directing them on to the appropriate time-slots.

Speech memory has as many storage locations as the number of time-slots in input PCM, *e.g.*, 32 locations for 32 channel PCM system.

The writing/reading operations in the speech memory are controlled by the Control Memory. It has same number of memory locations as for speech memory, *i.e.*, 32 locations for 32 channel PCM system. Each location contains the address of one of the speech memory locations where the channel sample is either written or read during a time-slot. These addresses are written in the control memory of the CC of the exchange, depending upon the connection objective.

A Time-Slot Counter which usually is a synchronous binary counter, is used to count the time-slots from 0 to 31, as they occur. At the end of each frame, It gets reset and the counting starts again. It is used to control the timing for writing/reading of the samples in the speech memory.

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Illustration

Consider the objective that TS4 of incoming PCM is to be connected to TS6 of outgoing PCM. In other words, the sample arriving in TS4 on the I/C PCM has to be delayed by 6 – 4 = 2 time-slots, till the destination time-slot, viz., TS6 appears in the O/G PCM. The required delay is given to the samples by storing it in the speech memory. The I/C PCM samples are written cyclically *i.e.*, sequentially time-slot wise, in the speech memory locations. Thus, the sample in TS4 will be written in location 4, as shown in Figure 1.6.

The reading of the sample is controlled by the Control Memory. The Control Memory location corresponding to output time-slot TS6, is 6. In this location, the CC writes the input time-slot number, viz.,4, in binary. These contents give the read address for the speech memory, *i.e.*, it indicates the speech memory locations from which the sample is to be read out, during read cycle.

When the time-slot TS6 arrives, the control memory location 6 is read. Its content addresses the location 4 of the speech memory in the read mode and sample is read on to the O/G PCM.

In every frame, whenever time-slot 4 comes a new sample will be written in location 4. This will be read when TS6 occurs. This process is repeated till the call lasts.

For disconnection of the call, the CC erases the contents of the control memory location to halt further transfer of samples.

Time switch can operate in two modes, viz.,

(*i*) Output associated control

(*ii*) Input associated control

Output Associated Control

In this mode of working, 2 samples of I/C PCM are written cyclically in the speech memory locations in the order of time-slots of I/C PCM, *i.e.*, TS1 is written in location 1, TS2 is written in location 2, and so on.

The contents of speech memory are read on output PCM in the order specified by control memory. Each location of control memory is rigidly associated with the corresponding time-slot of the O/G PCM and contains the address of the TS of incoming PCM to be connected to. The control memory is always read cyclically, in synchronism with the occurrence of the time-slot. The entire process of writing and reading is repeated in every frame, till the call is disconnected.

It may be noticed that the writing in the speech memory is sequential and independent of the control memory, while reading is controlled by the control memory, *i.e.*, there is a sequential writing but controlled reading.

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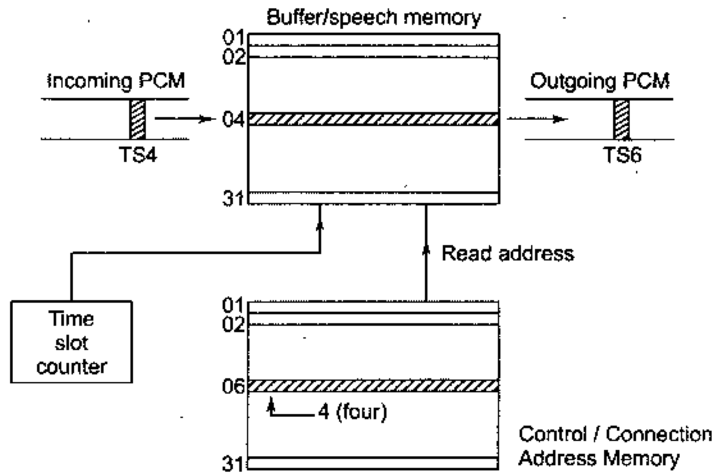

Fig 1.6 Output associated control switch

Input Associated Control

Here, the samples of I/C PCM are written in a controlled way, *i.e.*, in the order specified by control memory, and read sequentially.

Each location of control memory is rigidly associated with the corresponding TS of I/C PCM and contains the address of TS of O/G PCM to be connected to.

The previous example with the same connection objective of connecting TS4 of I/C PCM to TS6 of O/G PCM may be considered for its restoration. The location 4 of the control memory is associated with incoming PCM TS4. Hence, it should contain the address of the location where the contents of TS4 of I/C PCM are to be written in speech memory. A CC writes the number of the destination TS, viz., 6 in this case, in location 4 of the control memory. The contents of TS4 are therefore, written in location of speech memory, as shown in Figure 1.7.

The contents of speech memory are read in the O/G PCM in a sequential way, *i.e.*, location 1 is read during TS1, location 2 is read during TS2, and so on. In this case, the contents of location 6 will appear in the output PCM at TS6. Thus the input PCM TS4 is switched to output PCM TS6. In this switch, there is sequential reading but controlled writing.

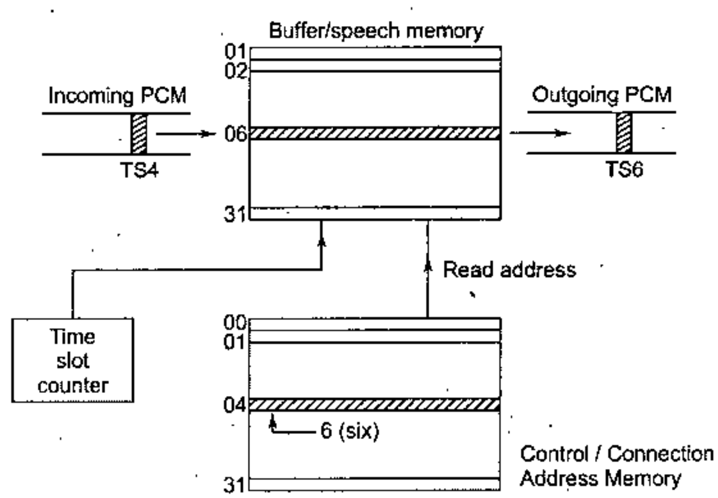

Fig. 1.7 Input associated controlled time switch

Time Delay Switching

The writing and reading, of all time-slots in a frame, has to be completed within one frame time period (before the start of the next frame). A TS of incoming PCM may, therefore, get delayed by a time period ranging from 1 TS to 31 TS periods, before being transmitted on outgoing PCM. For example, consider a case when TS6 of incoming PCM is to be switched to TS5 in outgoing PCM. In this case switching can be completed in two consecutive frames only, *i.e.*, 121 microseconds for a 32 channel PCM system. However, this delay is imperceptable to human beings.

NOTES

Non-Blocking Feature of a Time Switch

In a Time Switch, there are as many memory locations in the control and speech memories as there are time-slots in the incoming and outgoing PCM highways, *i.e.*, corresponding to each time-slot in incoming highway, there is a definite memory location available in the speech and control memories. Similarly, corresponding to each time-slot in the outgoing highway there is a definite memory location available in the control and speech memories. This way, corresponding to free incoming and outgoing time-slots, there is always a free path available to interconnect them. In other words, there is no blocking in a time switch.

Two Dimensional Switching

Though the electronic cross points are not so expensive, the cost of accessing and selecting them from external pins in a Space Switch, becomes prohibitive as the switch size increases. Similarly, the memory location requirements rapidly go up as a Time Switch is expanded, making it uneconomical. Hence, it becomes necessary to employ a number of stages, using small switches as building blocks to build a large network. This would result in necessity of changing both the time-slot and highway in such a network. Hence, the network, usually, employs both types of switches *viz.*, *space switch and time switch*, and. *therefore*, is known as two dimensional network. These networks can have various combinations of the two types of switches and are denoted as TS, STS, TSST,etc.

Though to ensure full availability, it may be desirable to use only T stages. However, the networks having the architecture of TT, TTT, TTTT, etc., are uneconomical, considering the acceptability of tolerable limits of blocking, in a practical network. Similarly, a two-stage two-dimensional network, TS or ST, is basically suitable for very low capacity networks only. The most commonly used architecture has *three stages, viz.*, STS or TST. However, in certain cases, their derivatives, viz., TSST, TSSST, etc., may also be used.

An STS network has relatively simpler control requirements and hence, is still being favoured for low capacity networks, viz., PBX exchanges. As the blocking depends mainly on the outer stages, which are space stages, it becomes unsuitable for high capacity systems.

A TST network has lesser blocking constraints as the outer stages are time stages which are essentially non-blocking and the space stage is relatively smaller. It is, therefore, most *cost-effective for networks handling high traffic*, However, for still higher traffic handling capacity networks, *e.g.*, tandom exchanges, it may be desirable to use TSST or TSSST architecture.

The choice of a particular architecture is dependent on other factors also, viz., implementation complexity, modularity, testability, expandability, etc. As a large number of factors favour TST structure, it is most widely used.

TST Network

NOTES

As the name suggests, in a TST network, there are two time stages separated by a space stage. The former carry out the function of time-slot changing, whereas the later performs highway jumping. Let us consider a network having n input and n output PCM highways. Each of the input and output time stages will have n time switches and the space stage will consist of an $n \times n$ cross point matrix. The speech memory as well as the control memory of each time switch and each column of a control memory of the space switch will have m locations, corresponding to m time-slots in each PCM. Thus, it is possible to connect any TS in I/C PCM to any TS in O/G PCM.

In the case of a local exchange, the network will be of folded type, *i.e.*, the O/G PCM highways, via a suitable hybrid. Whereas, for a transit exchange, the network will be non-folded, having complete isolation of I/C and O/G PCM highways. However, a practical local exchange will have a combination of both types of networks.

For the sake of explanation, let us assume that there are only four I/C and O/G PCM highways in the network. Hence, there will be only four time switches in each of the T-stages and the space switch will consist of 4×4 matrix. let us consider an objective of connecting two subscribers through this switching network of local exchange, assuming that the CC assigns TS4 on HWY0 to the calling party and TS6 on HWY3 to the called party.

The speech samples of the calling party have to be carried from TS4 of I/C HWY 0 and to TS6 of O/G HWY3 and those of the called party from TS6 of I/C HWY 3 to TS4 of O/G HWY 0, with the help of the network. The CC establishes the path, through the network in three steps. To introduce greater flexibility, it uses an intermediate time-slot, TSx, which is also known as internal time-slot. The three switching steps for transfer of speech sample of the calling party to the called party are as under:

Step 1. Input Time Stage (IT) TS4 HWY0 to TSx HWY0

Step 2. Space Stage (S)Tsx HWY0 to Tsx HWY3

Step 3. Output Time Stage (OT)Tsx HWY3 to TS6 HWY3

As the message can be conveyed only in one direction through this path, another independent path, to carry the massage in the other direction is also established by the CC, to complete the connection. Assuming the internal time-slots to be TS10 and TS11, the connection may be established as shown in Figure 1.8.

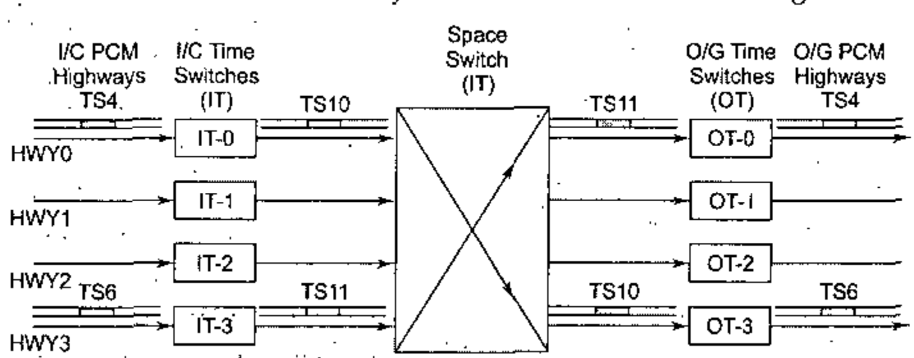

Fig. 1.8 T S T switch

Let us now consider the detailed switching procedure making some more assumptions for the sake of simplicity. Though practical time switches can handle 256 time-slots in parallel mode, let us assume serial working and that there are only 32 time-slots in each PCM. Accordingly, the speech and control memories in time switches and control memory columns in space switch, will contain 32 locations each.

NOTES

To establish the connection, the CC searches for free internal time-slots. Let us assume that the first available time-slots are TS10 and TS11, as before. To reduce the complexity of control, the first time stage is designed as output-controlled switch, whereas the second time stage is input-controlled.

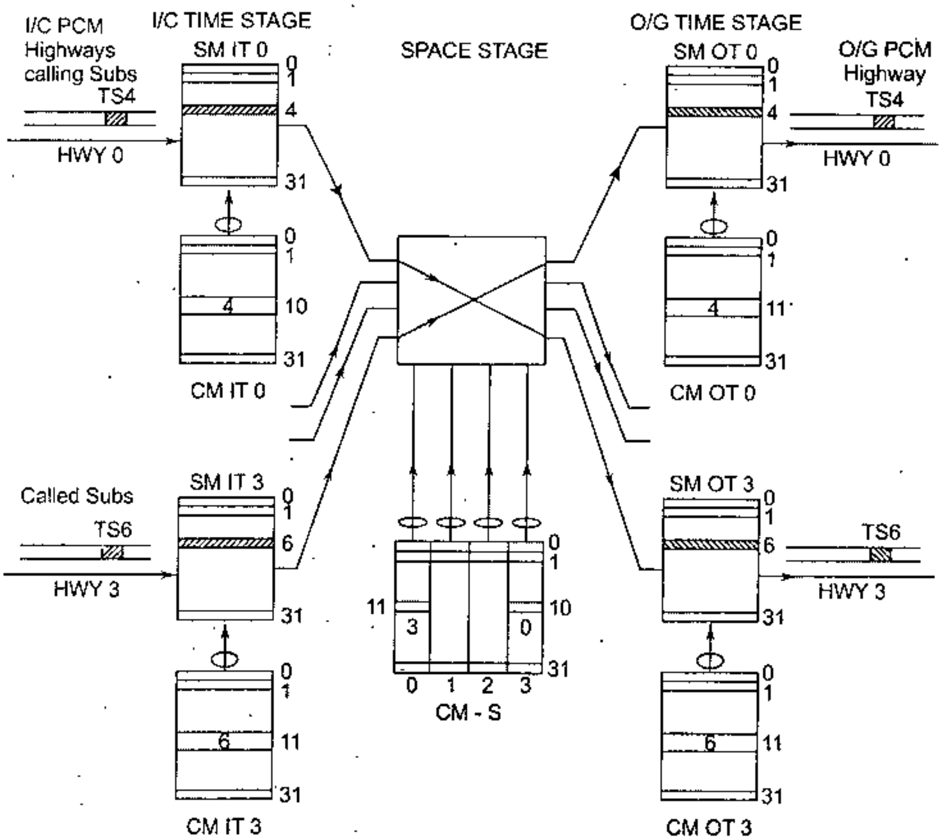

Fig 1.9 T S T switch structure

For transfer of speech samples from the calling party to the called party of previous example, CC orders writing of various addresses in location 10 of control memories of IT-10, OT-3 and column 3 of CM-S of corresponding to O/G highway, HWY3. Thus, 4 corresponding to I/C TS4 is written in CM-IT-0, 6 corresponding to O/G TS6 is written in CM-OT-3 and 0 corresponding to I/C HWY 0 is written in column 3 of CM-S, as shown in Figure 1. 9.

As the first time switch is output-controlled, the writing is done sequentially. Hence, a sample, arriving in TS4 of I/C HWY 0, is stored in location 4 of SM-IT-0. It is readout on internal HWY 0 during TS10 as per the control address sent by CM-IT-0. In the space switch, during this internal TS10, the cross point 0 in column 3 is enabled, as per the control address sent by column 3 of CM-S, thus, transferring the sample to HWY3. The second time stage is input controlled and hence, the sample, arriving in TS10, is stored in location 6 of SM-OT-3, as per the address sent by the CM-OT-3. This sample is finally, readout during TS6 of the next frame, thus, achieving the connection objective.

Similarly, the speech samples in the other direction, *i.e.*, from the called party to the calling party, are transferred using internal TS11. As soon as the call is over, the CC erases the contents in memory locations 10 and 11 of all the concerned switches, to stop further transfer of message. These locations and time-slots are, then, avialable to handle next call.

NOTES

Switching Network Configuration of some Modern Switches

1. E10B - T-S-T
2. EWSD - T-S-S-S-T
3. AXE10 - T-S-T
4. CDOT(MBM) - T-S-T
5. 5ESS - T-S-T
6. OCB 283 - T

SUMMARY

- The first electronic exchange employing space division switching (Analog switching) was commissioned in 1965 at Succasunna, New Jersey. This exchange used one physical path for one call and, hence, full availability could still not be achieved.
- Any and every subscriber can originate a call at any and every moment without giving any previous information and the duration of calls is also not previously known. Although the individual telephone traffic originates at random, the average telephone traffic for a particular exchange follows the general pattern of activity in the exchange area.
- The purposes of automatic exchange a more precise unit of traffic flow is required. this is called **Traffic Intensity**.
- The grade of service is a factor employed for dimensions of the exchange equipment. A few typical problems are worked out below to illustrate how the terms and definitions of telephone traffic are actually applied in practice.
- Digital computers have the capability of handling many tens of thousands of instructions every second, hence, in addition to controlling the switching functions the same processor can handle other functions also.
- A digital switching system, in general, is one in which signals are switched in digital form. These signals may represent speech or data.

REVIEW QUESTIONS

1. Discuss the evaluation of telecommunication.
2. Discuss advantages of electronic exchange over electromechanical exchanges.
3. Discuss the procedure of measurement of telephone traffic.
4. Discuss the grade of service.
5. Discuss the scanning method.
6. Discuss the quantitative indicators for quality of service.
7. Write the short notes
   (*a*) Answer Bid Ratio (ABR);
   (*b*) Answer Seizure Ratio (ASR); and
   (*c*) Congestion (CONG).

8. Write the short notes on stored programme controlled exchange.
9. Draw and discuss the block schematic of spc exchange.
10. Discuss and differentiate time and space switching
11. Discuss the principle of digital space switching
12. Discuss the principle of digital time switch.
13. Discuss
    (*a*) Output associated control
    (*b*) Input associated control

NOTES

FURTHER READINGS

1. *Telecommunication and Information Technology,* Prashant Kaushik, Anmol, 2006.
2. *Optical Networking in Telecommunication,* S. Mukherjee, Jaico.
3. *Wireless Technology and Access of Information,* Ajay K. Srivastav, Shree Pub., 2006.
4. *Elements of Networking Engineering,* Kumar Prasun Ramakrishnan, Shree Pub., 2010.
5. *Trends in Networking and Communication,* Edited by Girish Kumar Srivastav and Charul Bhatnagar, Atlantic Pub., 2009.

NOTES

UNIT II: ISDN, SATELLITE SYSTEM AND OPTICAL FIBRE COMMUNICATION

★ STRUCTURE ★

LEARNING OBJECTIVES

After going through this unit, you will be able to:

- know about what is ISDN?
- define introduction to broadcast channel
- describe overview of satellite system and communication
- know about principles and features of satellite communications
- explain fibre optics

PART I: ISDN INTRODUCTION

2.1 WHAT IS ISDN?

NOTES

The ISDN is an abbreviation of Integrated Services Digital Network. The current communications networks vary with the type of service, such as telephone network, telex network, and digital data transmission network. On the other hand, the ISDN is an integrated network for various types of communications services handling digitized voice (telephone) and non voice (data) information.

Figure 2.1 shows the current network configuration with individual networks, such as telephone network and a data network existing independently, and telephone sets, data terminals, etc., connected individually to each network

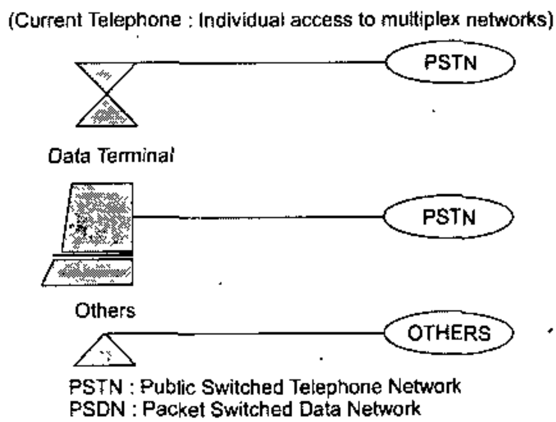

Fig. 2.1 The network configuration without ISDN

Figure 2.2 shows individual networks that will be fully integrated in the future.

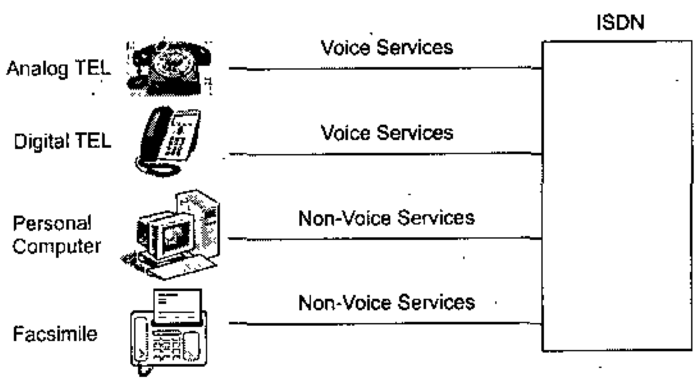

Fig. 2.2 The network configuration with ISDN

2.2 ISDN DEFINITION

The CCITT defines the ISDN as follows :

1. A complete, terminal-to-terminal digital network. Figure 2.3 shows the end-to-end digital connectivity.

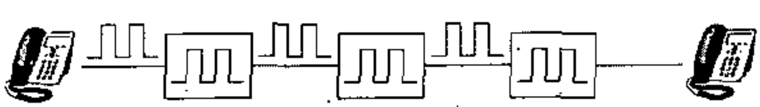

Fig. 2.3 End-to-end digital connectivity

NOTES

2. A network that provides both telephone and non-telephone services in the same network.
3. A network based on a digital telephone network.
4. A network that utilizes Signaling System No. 7 (SS7) for signaling between switching systems. Figure 2.4 shows the signaling connection between Switching Systems.

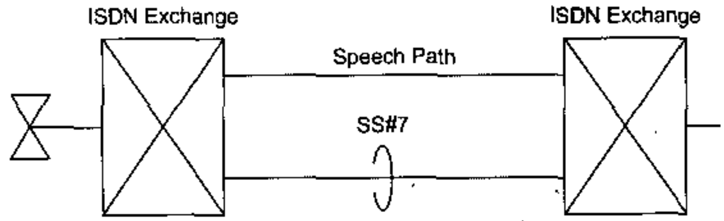

FIG. 2.4 The signaling connection between switching systems

5. A network offers standard user network interface. Figure 2.5 shows the standard user network interface.

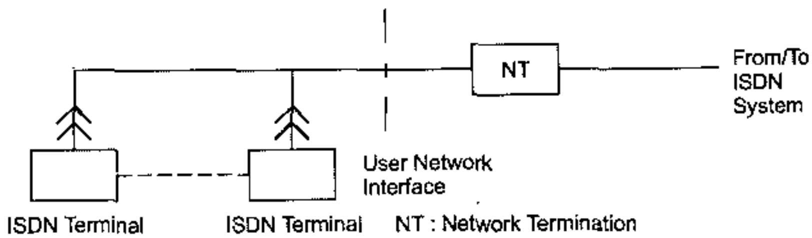

FIG. 2.5 Standard user network interface

2.3 ISDN SERVICES

1. A wide range of services
   (a) The ISDN provides the following functions, as shown in Figure 2.6.
   - Packet switching service
   - Circuit switching service
   - Leased circuit service

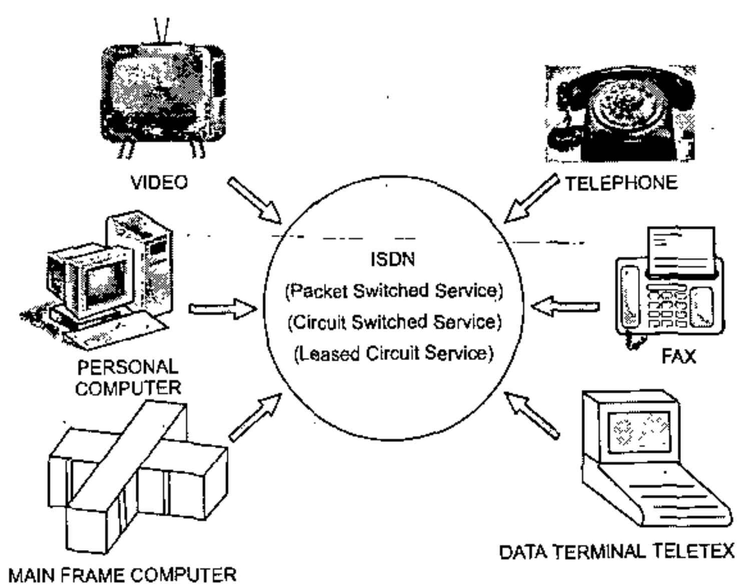

FIG. 2.6 A wide range of services

Circuit switching service includes both telephone and data circuit switching.

(*b*) As shown in the figure, ISDN can interface with various terminals, *such as a telephone set, FAX, Video terminal* or personal computer to provide a wide range of services.

(*c*) The ISDN concept can be summarized by two statements :

- ISDN offers a variety of services, such as telephone, data and image transmission through one network.
- ISDN handles all information digitally.

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2. Standard user-network interface. Figure 2.7 shows the user-terminal/network interface.

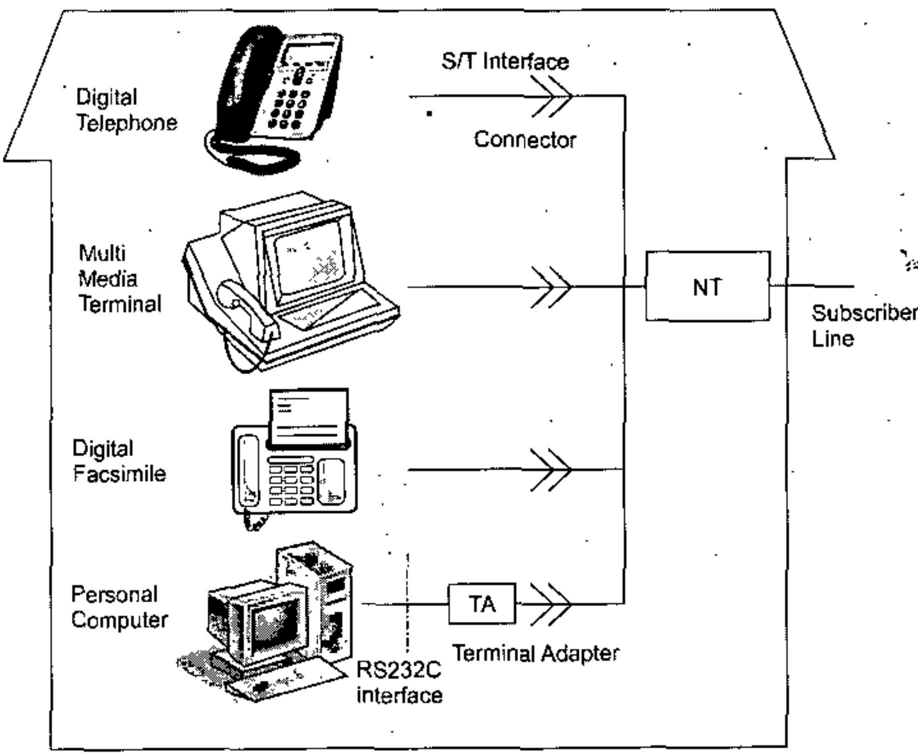

FIG. 2.7 User-terminal/network interface

(*a*) The subscriber line is connected with an NT (Network Termination) installed at the customer premises.

(*b*) Various terminals are connected to the NT. These terminals can include digital telephones, multi media terminal, digital facsimile machines, personal computers, etc., as shown in the figure.

(*c*) The NT and terminals are connected by S or T interface (S/T interface), as recommended by the CCITT. Up to 8 terminals are connected to one S/T interface. The NT and terminals are connected using an 8-pin connector, which is also recommended by the CCITT.

(*d*) As shown in this figure, the personal computer uses the RS232C interface that is different from the ISDN S/T interfaces, so a TA (Terminal Adapter) is provided to adapt the RS232C interface for use with the ISDN interfaces.

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Figure 2.8 shows operation of various terminals in the home.

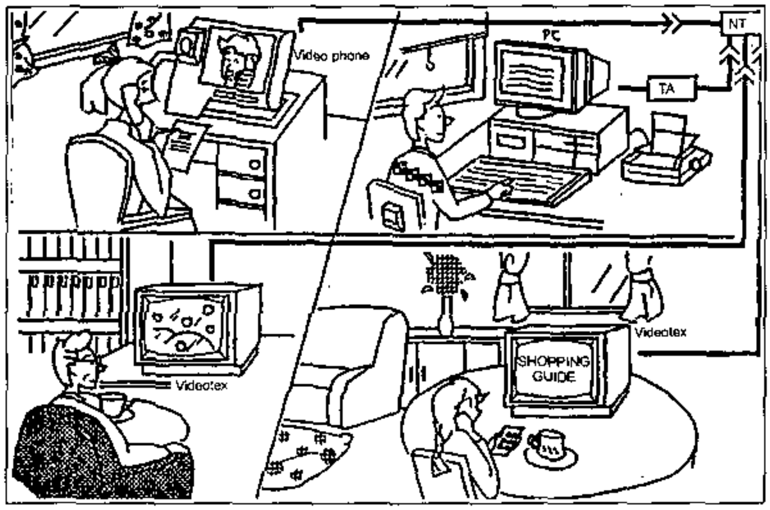


FIG. 2.8 Operation of various terminals in the home

(*a*) Each terminal is connected to the NT through S/T interface which, in turn, is connected to the switching system through the subscriber line.

(*b*) At the upper left of the figure a person is using a television telephone called a Video Phone, at the lower left, a person is watching a picture on a Videotex terminal.

(*c*) At the upper right of the figure, a person is operating a personal computer, which requires the use of a TA to convert the computer's RS232C interface to the S/T interfaces used by ISDN. At the lower right, a person is doing catalog shopping using a Videotex terminal.

3. Home Shopping and Home Banking
   - Figure 2.9 shows home shopping and home banking services.
   - Figure 2.9 shows a typical service made possible by ISDN. It shows something is being ordered to a department store, and then delivered.
   - The goods are ordered using the Videotex terminal, and an instruction is output to the bank to transfer the amount of the bill from your account.

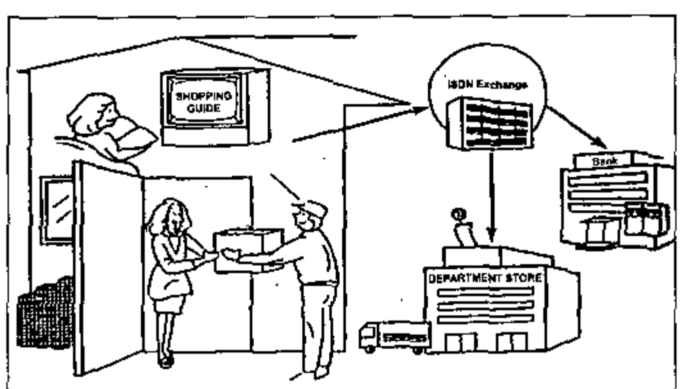


FIG. 2.9 Home shopping and home banking service

   - The department store delivers the ordered goods.
4. Home Medical System
   - Figure 2.10 shows home medical system.

- Figure 2.10 shows another service provided by ISDN : the receiving of medical care at home.

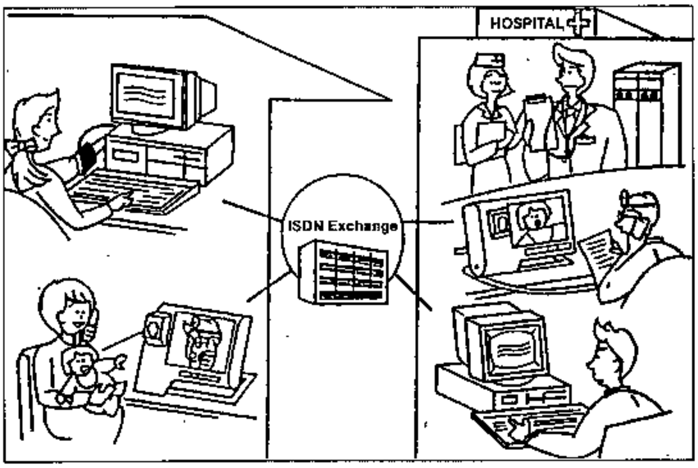


Fig. 2.10 Home medical system

- The upper left shows the measuring of blood pressure, with the result shown on the videotex screen both at home and at a medical facility (show at the bottom right of the figure).
- The lower left shows a consultation for medication using a TV telephone.

NOTES

2.4 USER NETWORK INTERFACE

ISDN User Network Interface Configuration

1. Figure 2.11 shows the interface between the user and the network. Telephone service makes use of two wires for the subscriber line between the switching system and customer's premises. These same two wires can be ued by ISDN to receive ISDN services.

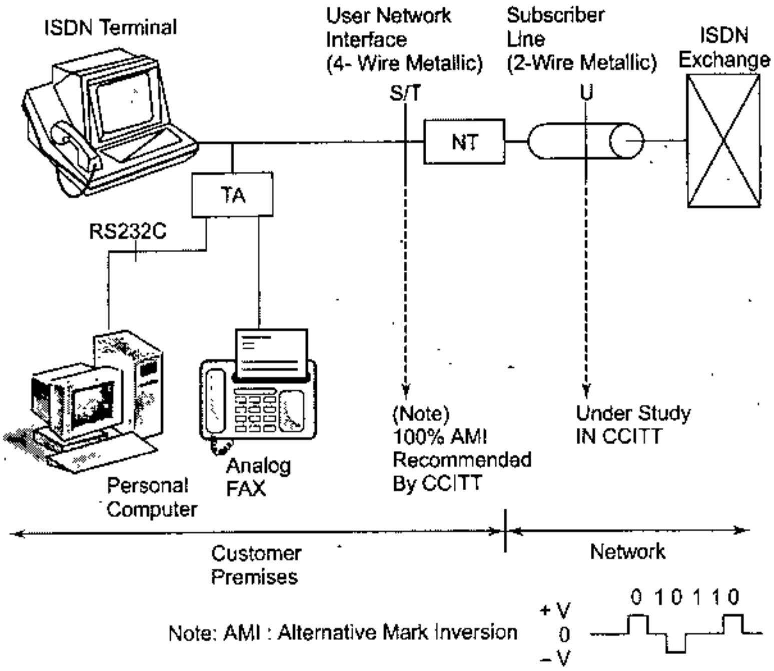


Fig. 2.11 The interface between the user

NOTES

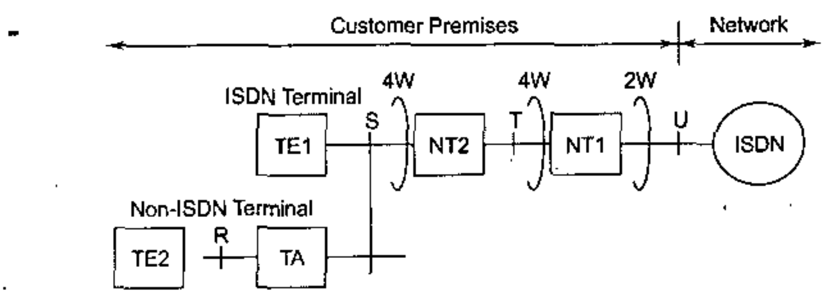
2. An NT (Network Termination) is installed at the subscriber's home and connected to the subscriber line.
3. The Interface between the NT and the ISDN exchange (switching system) is called U interface. This interface has not been defined in the CCITT Recommendations because circumstances are different in each country. The point between the NT and the on-premises terminals is called the S or T reference point. The ISDN user/network interface refers to these S/T points, and is defined in the CCITT Recommendations.
4. The S/T interface uses four wires, two for sending and two for receiving. Since U interface uses two wires, the NT provides a two-wire/four-wire conversion function.
5. CCITT recommends the use of AMI (Alternative Mark Inversion) code at the S/T point. AMI code is a bipolar waveform.
6. As shown in the figure, the ISDN Terminal provides S/T interface that follows the CCITT Recommendations, and can be connected directly to the NT. Since the personal computer and the analog FAX utilize a different interface from S/T interface, they require protocol conversion by a TA (Terminal Adapter).

Service Access Points (Reference Points)

1. *In the existing telephone network, a point at which a service is provided for* a user, that is, a service access point is located at a rossete between the user's telephone set and the subscriber line.

   Since the ISDN provides various types of service other than telephone service through a plural number of terminals, various service access points are provided. Thus, service access points would have to be defined corresponding to the ISDN Services.
2. Figure 2.12 shows the user-network interface reference points which is based on the CCITT reference model and identifies the important reference points of the model.

Point of the Model



NTI : Network termination 1
NT2 : Network Termination 2 (Ex. PBX)
TE1 : ISDN terminal, directly connected to point "S"
TE2 : Non-ISDN terminal containing "R" interface and terminal adapter (TA) for direct connection to point "S".
U : Digital transmission interface reference point
S, T : ISDN interface reference points
TA : Terminal adapter
R : Non-ISDN reference point

Fig. 2.12 User-network interface reference points

3. The following describes the user-access points and the function of each for basic user-network interface.
   (*a*) Network Termination (NT) :
   - The NT can be split into NT1 and NT2. NT1 and NT2 are terminating equipment for the network.
   - In this case, NT1 provides the Layer 1 functions, such as circuit termination, timing and supply of electricity, while NT2 provides the layer 2 functions, such as protocol, control and concentration functions.

   (*b*) Terminal Equipment (TE) :
   - The TE can be split into TE1 and TE2. TE1 is an ISDN terminal which is connected to ISDN via the S/T interface. TE2 is a non-ISDN terminal which is connected to ISDN via a Terminal Adapter (TA) such as personal computer or analog FAX as described in Figure 2.11.

   (*c*) Terminal Adapter (TA) :
   - A TA is a physical device which is connected to a non-ISDN terminal (TE2) to permit access to ISDN.

   (*d*) *S-Interface :*
   - A 4-wire physical interface used for a single customer termination between a TA and NT2 or between TE1 and NT2.

   (*e*) T-Interface :
   - A 4-wire physical interface between NT1 and NT2.

   (*f*) R-Interface :
   - A physical interface used for single customer terminator between TE2 and TA.

   (*g*) U-Interface :
   - The subscriber line is called U-Interface and utilizes 2-wires.

NOTES

ISDN User Network Interface Points

1. Requirements of User-Network Interface: For us to utilize "integrated *services" including voice and* non-voice communications and the use of some new media, such as facsimile in offices and home, the following features must be provided for user-network interfaces :
   (*a*) Different services for each call
   - A switching mode (packet switched/circuit switched function) can be selected.
   - Data transmission speed can be selected.

   (*b*) Plural number of terminals can be concurrently connected.
   (*c*) The portability of terminals can be ensured.
2. Basic Structure of User-Network Interface: *The basic conditions for structuring* the user-network interface that satisfy the preceding requirements can be summarized into the following three points :
   (*a*) Multi services
   - Common use of various services telephone/non telephone and existing/new services. As shown in Figure 2.12, ISDN termianls, personal computers, FAX machines, etc., are connected to S/T points to offer various services.

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(*b*) Multi points

- Up to eight (8) terminals can be connected to one (1) NT as well as point to point connection.
- Figure 2.13 shows the multi points connection.

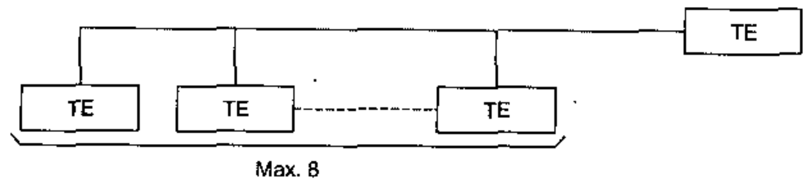

FIG. 2.13 Multi points connection

(*c*) Portability

- Terminals can be carried from place to place and connected to different sockets for use, just as home electrical appliances can be carried around and plugged into AC outlets.

3. Channel Classification: Various channels can be used to transmit information between a terminal and the switching system. These include B, D and H channels. Each channel has a different bit rate and information carrying attributes.

(*a*) B-channel

- The B-channel carries user information such as voice and packet data at a rate of 64 kbps. However, the B-channel does not carry signaling information.

(*b*) D-channel

- The D-channel interface carries mainly signaling information such as originating or terminating subscriber number, call origination and disconnect signals for circuit switching and packet switched user data at 16 kbps or 64 kbps.
- The D-channel also permits multiple logical channels to be established for use in packet communications.

(*c*) H-channel

- The H-channel carries high-speed user information such as high-speed facsimile, video, high-speed data, etc. H channels do not carry signaling information for circuit switching by the ISDN.

(*d*) Table 2.1 outlines these three channel types and characteristics.

TABLE 2.1 Channel types and characteristics

Channel Type	Bit Rate	Function
B	64 kbps	• To carry user information • Circuit switching mode and packet switching mode
D	16 kbps 64 kbps	• To carry signaling information for circuit switching
H	H0 : 384 kbps H11 : 1536 kbps H12 : 1920 kbps	• To carry high-speed packet data such as facsimile and video • An H channel does not carry signaling information for circuit switching by the ISDN
Note :	• H0 : $64^K \times 6 = 384$ kbps • H11 : $64^K \times 24 = 1536$ kbps • H12 : $64^K \times 30 = 1920$ kbps	

3. Typical Interface Structures

(a) Basic Interface

- This interface is primarily for home use.
- The basic interface is set at a transmission speed of 144 kbps. This provides two (2) 64 kbps B-channels for user information exchange and a 16 kbps D-channel for signaling and control.

  The interface is thus referred to as 2B+D.

  Figure 2.14 shows the basic interface structure.

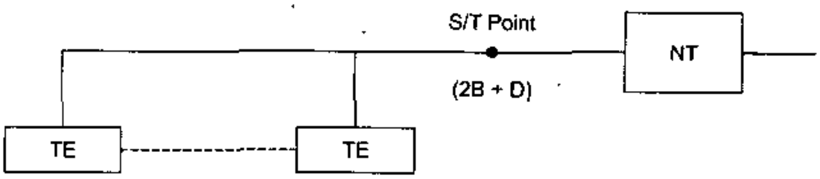

FIG. 2.14 Basic interface structure

(b) Primary Group Interface

- These interface are primarily for business use. The primary group interface for ATT system consists of a 1.544 Mbps line. This line can thus provide up to 23 B-channels at 64 kbps and a single D-channel at 64 kbps.
- In Europe and other countries using CEPT system standards, the primary group is 2.048 Mbps and the interface is 30B-channels and single 64 kbps D-channel.

  This line is used for PABX etc.
- Figure 2.15 shows the primary group interface structure.

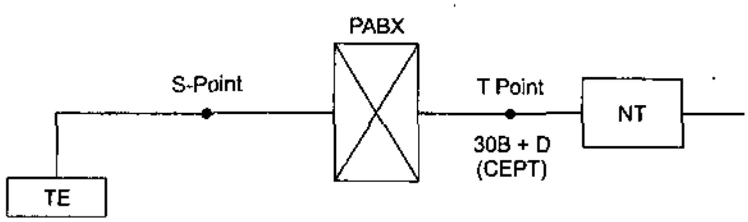

FIG. 2.15 Primary group interface structure

(c) Table 2.2 shows the typical user network interface structure.

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PART II: BRIEF INTRODUCTION TO BROADCAST CHANNEL

2.5 BRIEF INTRODUCTION TO BROADCAST CHANNEL

With the advent of mass scale industrialization in our country, the demand for more communication facilities came up. Several new telephone exchanges have been installed throughout the country for local communication more and more carrier channels have been provided for carrying the trunk traffic. With the planned introduction of Subscriber Trunk Dialing throughout the country, the number of carrier channels required to interconnect different cities became too high to be accomplished by overhead lines. Thus, U/G Cables Carrier Systems

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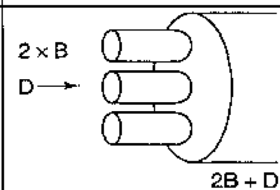
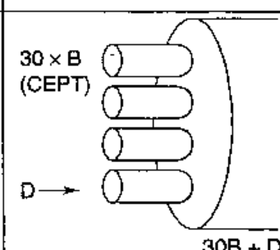
TABLE 2.2

Classification of User Network Interface	Interface Speed	Interface Name	Structure Logical Structure	Channel Speed	Use	Remarks
Basic Interface	192 kbps	Basic Interface (Basic Access)		• B-ch: 64 kbps • D-ch: 16 kbps	• B-ch is used as user information for switched call or packet call. • D-ch is used as signaling control information for switched call, packet call or user information for packet call.	• 192 kbps: 1 Frame = 48 bit in 250 microseconds
Primary Group Interface	2.048 kbps (CEPT)	Primary Group B-ch Interface (Multi Access)		• B/D-ch: 64 kbps	• Ditto	
		Primary Group H-ch Interface (High Speed Access)	(CEPT) • 5H0 + D • H12 + D	• D-ch: 64 kbps	• H-ch is used for user information such as high speed facsimile/data. • D-ch is used as signaling control information for switched call.	• User Study

were introduced, the first of them being the symmetrical pair Cable Carrier System between Calcutta and Asansol with an ultimate capacity of 480 channels. Then came the Co-axial Cable Carrier System linking all major cities in the country.With the development of Microwave technique, which can provide large block of circuits at comparative cost, the problem of long distance communication circuits appear virtually solved. A brief description of the Microwave technique is attempted in the following paragraphs. Electromagnetic waves can be broadly classified in terms of frequencies as follows :

Table 2.3

Range	Name	Wavelength	Uses
0–30 KHz	V.L.F.	Upto 10 km.	Used for long communication. Has limited information. Bandwidth require very high power.
30–300 KHz	L.F.	10 km to 1 km	
0.3–3 MHz	M.F.	1 km to 100 m	Radio Broadcast, Marine Power in KW, ground wave propagation, *i.e.*, follows the curvature of the Earth.
3–30 MHz	H.F.	100 m to 10 m	Long haul point to point communication. Propagation is by one or more reflections from inosphere layers and so subject to variations.
30–300 MHz	V.H.F.	10 m to 1 m	Line of sight, Troposcatter communication.
0.3–3 GHz	U.H.F.	1 m to 10 cm.	——— do ———
3–30 GHz	S.H.F.	10 cm to 1 cm.	Line of sight, terrestrial M/W and Satellite communication.
30–300 GHz	E.H.F.	1 cm to 1 mm.	Experimental.

The term SHF corresponds to "MICROWAVE" Centrimetric waves. As a convention frequencies, above 1 GHz and upto 40 GHz are termed as Microwave. However, most of the m/w systems available are in the range of 1 to 18 GHz.

2.6 APPLICATIONS

M/W frequency bands are used for the following services :

(*i*) Fixed Radio Communication Services.

(*ii*) Fixed Satellite Services.

(*iii*) Mobile Services.

(*iv*) Broadcasting Services.

(*v*) Radio Navigation Services.

(*vi*) Meteorological Services.

(*vii*) Radio Astronomy Services.

To meet the requirements of all above mentioned services, co-ordination among the users of M/W spectrum is necessary. In this regard (in the national context) the wireless planning and co-ordination wing (WPC) of the ministry of communication has allotted m/w frequencies spectrum, on the basis of various wireless users classified as general users and major users. Wireless users who are permitted to plan their services and take action for the development of the required equipments are major users. BSNL has been nominated as a major wireless user by the WPC in 1981 in the following sub baseband of the m/w spectrum for fixed radio communication. Microwave Spectrum Available for BSNL

NOTES

TABLE 2.4

Band	Bandwidth Available	Spectrum Space
2 GHz	300 MHz	2000–2300 MHz
4 GHz	900 MHz	3300–4200 MHz
6 GHz	1185 MHz	5925–7110 MHz
7 GHz	300 MHz	7425–7725 MHz
11 GHz	1000 MHz	10,700–11,700 HHz
13 GHz	500 MHz	12,750–13,250 MHz

In India the first M/w System was completed in December, 1965 between Kolkata and Asansol with a system capacity of 1200 channels. At present many kilometers of M/W systems are scattered throughout the country and further expansion is taking place at a very large rate.

Frequency Characteristics

Microwaves are very short frequency radio waves that have many of the characteristics of light wave in that they travel in line-of-sight paths and can be reflected, boomed and focussed. By focussing these ultra high radio waves into a narrow beam, their energies are concentrated and relatively low transmitting power is required for reliable transmission over long distance.

System Capacity

Microwave communication systems are used to carry telephony, television and data signals. Majority of the systems, however, carry multi-channel telephone signals. The spectrum of the multichannel telephone signal is shown in Figure 2.16. This signal is also called base band (Figure also The system capacity of line of sight systems ranges from 60 telephone channels to 2700 channels over a Radio bearer with a few systems of lower capacities varying from 60 to 60 channels. On the same m/w route one can use more than one radio channels, thus getting still larger capacity. As an example one can accommodate 8 go and 8 return RF channels each with a capacity of 1800 telephone channels in a 500 MHz bandwidth. Of course, in such cases usually one or two RF channels are kept as a standby which are switched over automatically on fading or equipment failure. Usually the system with capacities upto 300 channels is called narrow band system and the systems providing more than 300 channels are called wide band system. M/W systems used to provide communication on major trunk routes with high traffic density and serving long distances are classified as long haul m/w systems. 2, 4, and 6 GHz systems are long haul systems. Systems used to provide communication over short distances for trunk routes with light traffic density are classified as short haul system. 7 and 11 GHz systems are short haul systems.

Figure 2.17 shows the TV spectrum). Individual telephone channels, 4 KHz wide (300 to 3400 Hz for speech and the remaining for signalling and guard band) are multiplexed together in a multiplex equipment to get the base band. The base band frequency given in Table 2.5.

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Table 2.5

Channel capacity	Base band frequency in KHz
60 channels	12–252
60 channels	60–300
120 channels	60–555
300 channels	60–1300
600 channels	60–2540
960 channels	60–4028
1800 channels	312–8120/316–8204
2700 channels	312–12336/316–12388

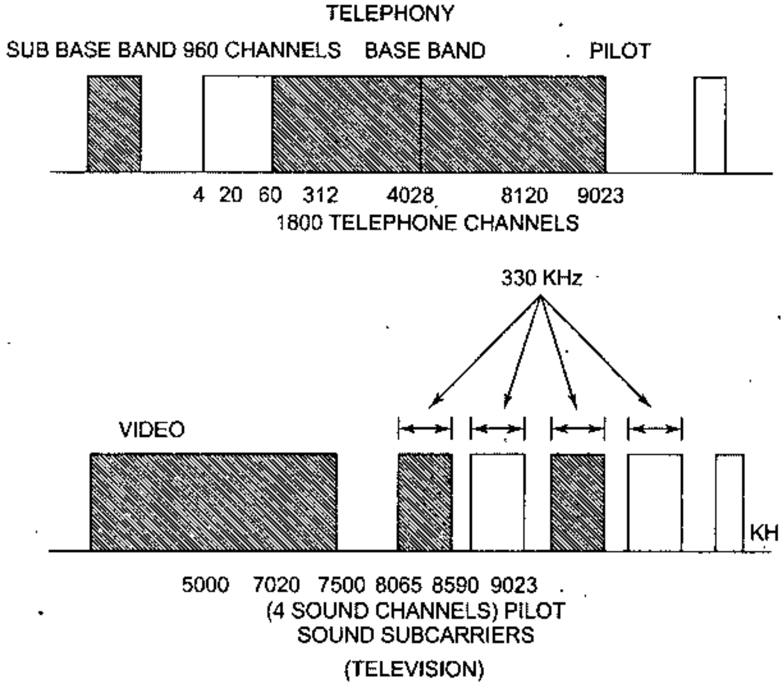

Fig. 2.16

2.7 BASE BAND SPECTRUM

The salient features of various long distance communication systems are summarised below to make a comparative study.

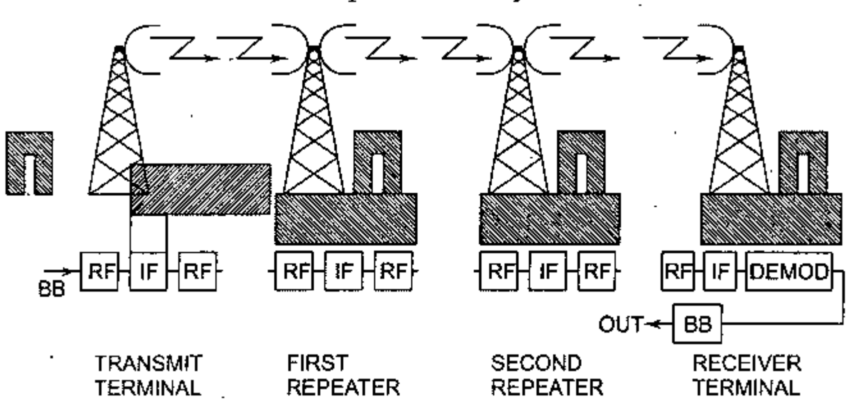

Fig. 2.17 A typical microwave radio system

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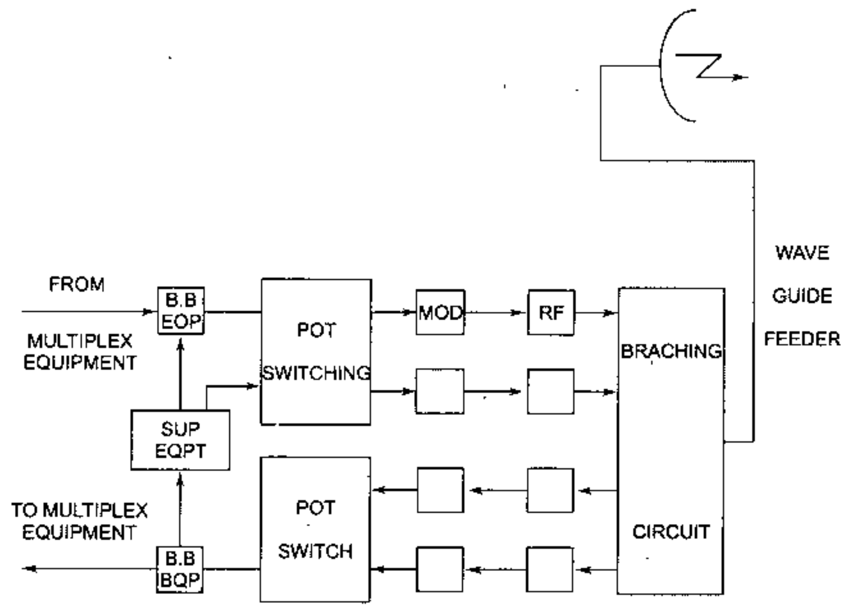

Fig. 2.18 Schematic of a microwave terminal

PART III: OVERVIEW OF SATELLITE SYSTEM & COMMUNICATION

2.8 OVERVIEW OF SATELLITE SYSTEM & COMMUNICATION

Long distance communication using conventional techniques like coaxial cable or microwave radio relay links involves a large number of repeaters. For radio relay links of repeater spacing is limited by line of sight and is of the order of tens of kms. As the number of repeaters increase system performance and reliability are degraded. Tropo scatter propagation can cover several hundred kms. but the channel capacity is limited and costs are high due to necessity of large antennas and high transmit power. HF communication is subject to fading due to ionospheric disturbances and channel capacity is severely restricted due to limited bandwidth available. Large areas could be covered if the height of microwave repeater could be increased by putting it on board an artificial earth satellite (Figure 2.19). Science Fiction writer Arthur C. Clarke in an article in Wireless World in 1945 proposed that worldwide coverage could be obtained by using three microwave repeaters placed in a geostationary orbit at the height of about 36000 kms. with a period of 24 hours (Figure 2.20).

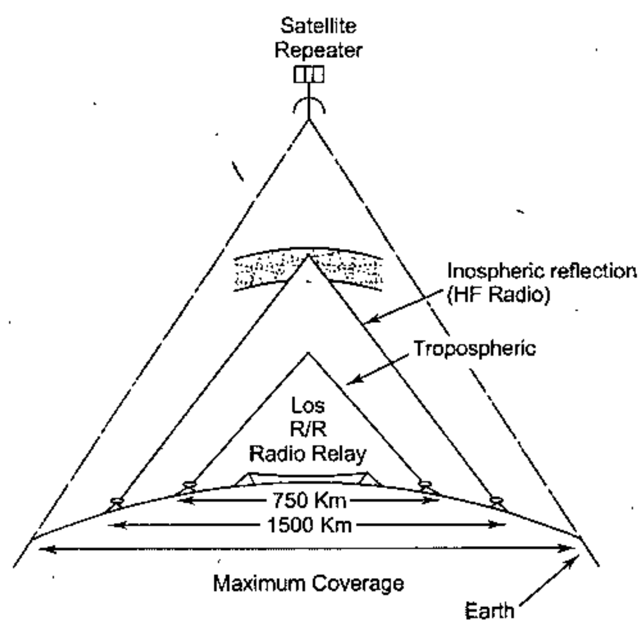

Fig. 2.19

2.9 MODES OF COMMUNICATION

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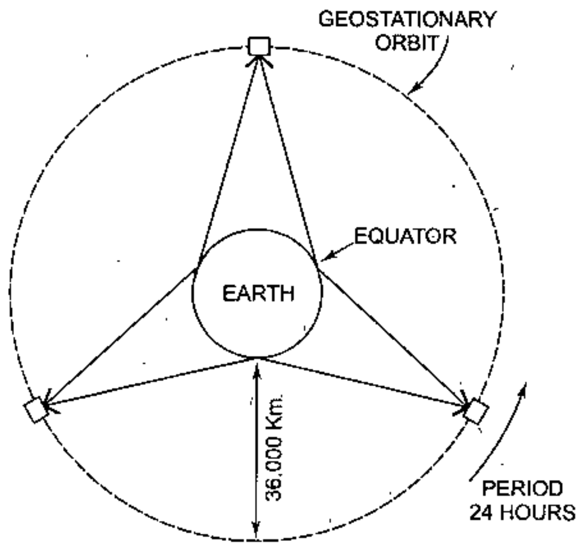


FIG. 2.20 Global coverage with geostationary satellite

Satellite communication provide a practical and economical means of long haul communication traffic in a country with a large geographical area.

It also enables communication service to those areas which are virtually INACCESSIBLE by other conventional forms of communication system due to natural physical barriers.

2.10 PRINCIPLES AND FEATURES OF SATELLITE COMMUNICATIONS

Principles

Figure 2.20 shows the principles of satellite communications. Here, a geostationary satellite with microwave radio repeater equipment receives and amplifies radio waves sent from earth stations and returns them to the earth.

A geostationary satellite is launched above the equator 36,000 km high above the earth. Its period round the earth coincides with that of the earth rotation. Therefore, the satellite looks as if it is stationary from the earth. If three (3) communication satellites are launched equidistantly above the equator (See Figure 2.20), it can serve almost all communication network round the world. Therefore, to facilitate public international telecommunications, INTELSATS IV and V have been launched above the Atlantic, Pacific, and Indian Oceans. These networks cover almost all countries around the world.

Features

For international communication, a submarine cable along the Atlantic Ocean was installed in 1857. Also, short-wave radio communication (invented by Marconi in 1886) has been in use. However, short wave radio communication has disadvantages of :

1. Small transmission capacity; only small telephone channels can be used to transmit.
2. Fading in wave propagation; interferes with stability of transmission. Although over-the-horizon propagation is used for short distance international communications, it is impossible to apply it to transoceanic long distance communications.

NOTES

Unlike other system, geostationary satellite communication systems summarize as follows :

1. Stable and large capacity communication.
2. Costs of establishment and maintenance do not depend on communication distance. The costs of submarine and over-the-horizon systems are proportional to the length, but those of the satellite system do not affect the communication distance. Therefore, the satellite system is ideal for long distance communications.
3. Multiple access is possible. Signals sent from an earth station can be received at several earth stations simultaneously. Therefore, it can transmit signals to many stations simultaneously, such as TV. Actually, increasing of submarine cable's capacity and distance between repeaters, can make submarine cables competitive to satellite communication specially when very large capacity is required but for small traffic size countries, satellite communication is unavailable for the independent communication services.

Advantages of Satellite Communications

(*i*) **Large coverage:** Almost one-third of the earth with exception of polar regions is visible from geostationary orbit. It is, thus, possible to cover about 10,000 kms. distance irrespective of intervening terrain with a single satellite.

(*ii*) **High quality:** Satellite links can be designed for high quality performance. The link performance is highly stable since it is free from ionospheric disturbances, multipath effects or fading.

(*iii*) **High reliability:** Reliability is high since there is only one repeater in the link.

(*iv*) **High capacity:** With microwave frequencies, wide bandwidths are available and large communication capacity can be obtained.

(*v*) **Flexibility:** In a terrestrial system, communication is tied down to the links installed. On the other hand, satellite communication is well suited for changing traffic requirements, locations and channel capacities.

(*vi*) **Speed of installation:** Installation of earth terminals can be achieved in a short time as compared to laying of cables or radio relay links.

(*vii*) **Mobile, short-term or emergency communications:** With ariliftable or road transportable terminals, short-term or emergency communications can be quickly provided. Reliable long distance land mobile, maritime mobile and aeronautical mobile services are feasible only by means of satellite.

(*viii*) Satellite communication is ideally suited for point to multipoint transmission on broadcasting over large areas. Application of satellites for TV broadcasting, audio and video distribution and teleconferencing, facsimile, data and news dissemination is, therefore, increasing rapidly.

(*ix*) All types of common services are possible.

2.11 SATELLITE COMMUNICATION NETWORK

NOTES

Satellite Communication Network could be defined as an ensemble of earth stations of pre-determined size spread over a pre-defined coverage area, interconnected through a suitably designed satellite, placed at a pre-determined location in properly chosen orbit around the earth. Thus, two important elements of a satellite communication network are :

(*i*) Space Segment
(*ii*) Ground Segment

Up Link and Down Link

Up link is the radio path from Ground segment, *i.e.*, earth station to the Space segment, *i.e.*, satellite, whereas Down link is the radio path from space segment, *i.e.*, satellite to the ground segment, *i.e.*, earth station.

Frequency Bands

Choice of Frequency band for space communication depends upon

- Band-width required.
- Noise consideration
- Propagation factors
- Technological developments with regard to component and device.

As the signal levels from the satellite are expected to be very low, any natural phenomenon to aid the reception of the incoming signals must be exploited. Note in Table 2.6 that between the frequencies of 2 GHz to 10 GHz, the level of the sky-noise reduces and this band of frequencies is known as the 'microwave window'.

The most of the communication satellites as on today are using a frequency of 6 GHz for "Up link" and 4 GHz for "Down link" transmission.

These frequencies are preferred because of

- Less atmospheric absorption than higher frequency.
- Less noise both galactic and manmade.
- Less space loss compared to higher frequency.
- A well developed technology available at these frequencies.
- 6 GHz/4 GHz bands are shared with terrestrial services, creating interference problem.
- As equatorial orbit is filling with geostationary satellites, RF interference is increasing from one satellite system to another is increasing.
- 14/11 and 30/20 GHz systems for telecommunication and broadcasting satellite services are slowly coming being.

Frequency Bands in use for Satellite Communication are:

Table 2.6

"L" BAND	1830–2700 MHz	
"S" BAND	2500–2700 MHz	INSAT IS USING
"C" BAND	5925–6425 MHz UP 3700–4200 MHz DOWN	INSAT IS USING

NOTES

"X" BAND	7900–8400 UP 7250–7750 DOWN	
"KU" BAND	14.000–14.500 Hz. UP 10950–11200 GHz/DN. 11450–11700 GHz/DN.	
"K" BAND	27.5–30 GHz UP 17.7–21.2 GHz DOWN	
EXTENDED C BAND	6725–7025 UP 4500–4800 DOWN	INSAT IS USING
V BAND	40–51 GHz UP 40–41 GHz DOWN	
V Band Inter-satellite	59–64 GHz 54–58 GHz	

PART IV: OPTICAL FIBRE COMMUNICATION

2.12 FIBRE OPTICS

Optical Fibre is new medium, in which information (Voice, Data or Video) is transmitted through a glass or plastic fibre, in the form of light, following the transmission sequence given below :

1. Information is encoded into electrical signals.
2. Electrical signals are converted into light signals.
3. Light travels down the fibre.
4. A detector changes the light signals into electrical signals.
5. Electrical signals are decoded into information.

2.13 ADVANTAGES OF FIBRE OPTICS

Fibre Optics has the following advantages :

(*i*) Optical Fibres are non conductive (Dielectrics)
 - Grounding and surge suppression not required.
 - Cables can be all dielectric.

(*ii*) Electromagnetic Immunity :
 - Immune to electromagnetic interference (EMI)
 - No radiated energy.
 - Unauthorised tapping difficult.

(*iii*) Large Bandwidth (> 5.0 GHz for 1 km length)
 - Future upgradability.
 - Maximum utilization of cable right of way.
 - One time cable installation costs.

(*iv*) Low Loss (5 dB/km to < 0.25 dB/km typical)
 - Loss is low and same at all operating speeds within the fibre's specified bandwidth long, unrepeated links (> 70 km is operation).

(*v*) Small, Light weight cables.
   - Easy installation and Handling.
   - Efficient use of space.

(*vi*) Available in Long lengths (> 12 kms)
   - Less splice points.

(*vii*) Security
   - Extremely difficult to tap a fibre as it does not radiate energy that can be received by a nearby antenna.
   - Highly secure transmission medium.

(*viii*) Security - Being a dielectric
   - It cannot cause fire.
   - Does not carry electricity.
   - Can be run through hazardous areas.

(*ix*) Universal medium
   - Serve all communication needs.
   - Non-obsolescence.

NOTES

2.14 APPLICATION OF FIBRE OPTICS IN COMMUNICATIONS

- Common carrier nationwide networks.
- Telephone Inter-office Trunk lines.
- Customer premise communication networks.
- Undersea cables.
- High EMI areas (Power lines, Rails, Roads).
- Factory communication/ Automation.
- Control systems.
- Expensive environments.
   - High lightening areas.
- Military applications.
- Classified (secure) communications.

2.15 TRANSMISSION SEQUENCE

1. Information is Encoded into Electrical Signals.
2. Electrical Signals are Coverted into light Signals.
3. Light Travels Down the Fiber.
4. A Detector Changes the Light Signals into Electrical Signals.
5. Electrical Signals are Decoded into Information.
   - Inexpensive light sources available.
   - Repeater spacing increases along with operating speeds because low loss fibres are used at high data rates.

NOTES

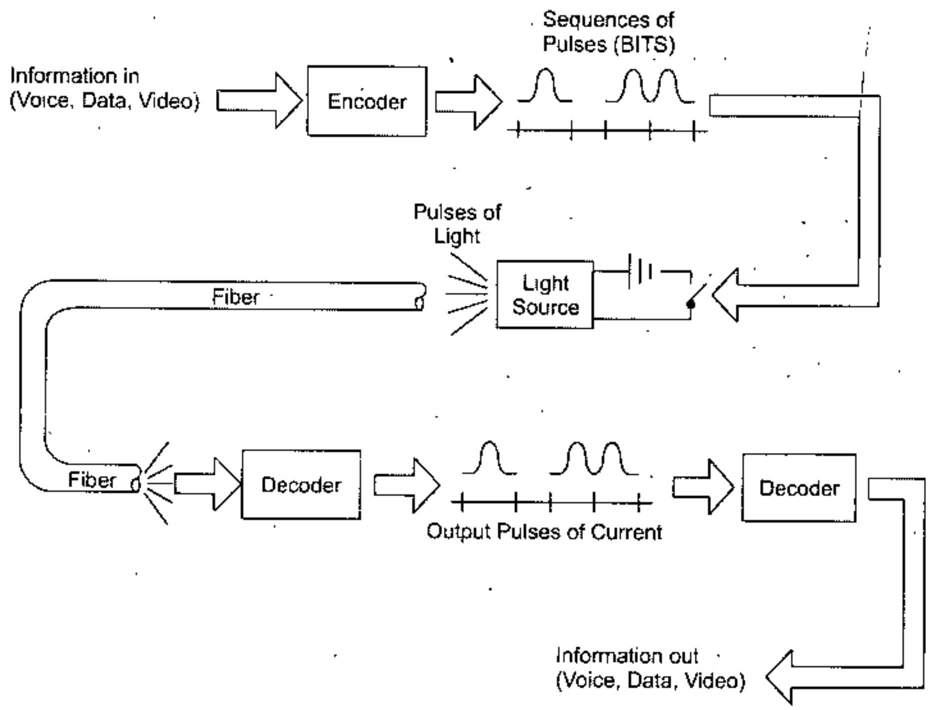


FIG. 2.21 Optical communication system

2.16 PRINCIPLE OF OPERATION - THEORY

- Total Internal Reflection—The reflection that occurs when a light ray travelling in one material hits a different material and reflects back into the original material without any loss of light.

2.17 THEORY AND PRINCIPLE OF FIBRE OPTICS

Speed of light is actually the velocity of electromagnetic energy in vacuum such as space. Light travels at slower velocities in other materials such as glass. Light travelling from one material to another changes speed, which results in light changing its direction of travel. This deflection of light is called Refraction.

The amount that a ray of light passing from a lower refractive index to a higher one is bent towards the normal. But light going from a higher index to a lower one refracting away from the normal, as shown in the figures.

As the angle of incidence increases, the angle of refraction approaches 90° to the normal. The angle of incidence that yields an angle of refraction of 90° is the critical angle. If the angle of incidence increases more than the critical angle, the light is totally reflected back into the first material so that it does not enter the second material. The angle of incidence and reflection are equal and it is called Total Internal Reflection.

By Snell's law, $n_1 \sin \phi_1 = n_2$ sing ϕ_2

The critical angle of incidence ϕ_c where $\phi_2 = 90°$

Is ϕ_c = arc sing (n_2/n_1)

At angle greater than ϕ_c the light is reflected, Because reflected light means that n_1 and n_2 are equal (since they are in the same material), ϕ_1 and ϕ_2 are also equal. The angle of incidence and reflection are equal. These simple principles of refraction and reflection form the basis of light propagation through an optical fibre.

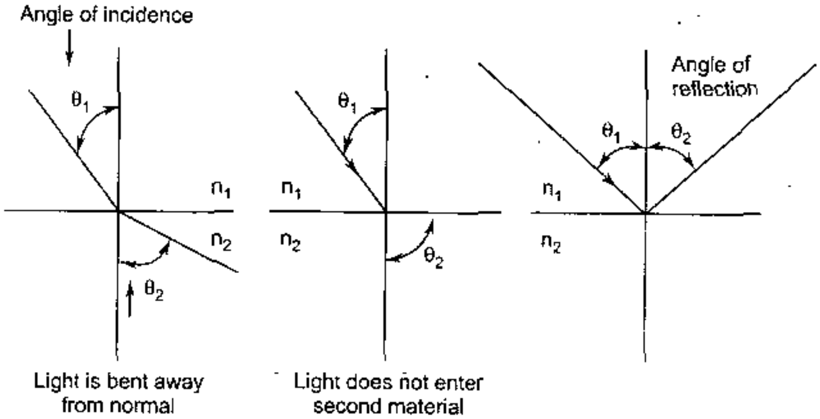


FIG. 2.22 Snell's law

NOTES

2.18 PROPAGATION OF LIGHT THROUGH FIBRE

The optical fibre has two concentric layers called the core and the cladding. The inner core is the light carrying part. The surrounding cladding provides the difference refractive index that allows total internal reflection of light through the core. The index of the cladding is less than 1%, lower than that of the core. Typical values for example are a core refractive index of 1.47 and a cladding index of 1.46. Fibre manufacturers control this difference to obtain desired optical fibre characteristics.

Most fibres have an additional coating around the cladding. This buffer coating is a shock absorber and has no optical properties affecting the propagation of light within the fibre. Figure shows the idea of light travelling through a fibre. Light injected into the fibre and striking core to cladding interface at greater than the critical angle, reflects back into core, since the angle of incidence and reflection are equal, the reflected light will again be reflected. The light will continue zigzagging down the length of the fibre.

Light striking the interface at less than the critical angle passes into the cladding, where it is lost over distance. The cladding is usually inefficient as a light carrier, and light in the cladding becomes attenuated fairly. Propagation of light through fibre is governed by the indices of the core and cladding by Snell's law.

Such total internal reflection forms the basis of light propagation through a optical fibre. This analysis consider only meridional rays—those that pass through the fibre axis each time, they are reflected. Other rays called Skew rays travel down the fibre without passing through the axis. The path of a skew ray is typically helical wrapping around and around the central axis. Fortunately skew rays are ignored in most fibre optics analysis.

The specific characteristics of light propagation through a fibre depends on many factors, including

- The size of the fibre.
- The composition of the fibre.

NOTES

– The light injected into the fibre.

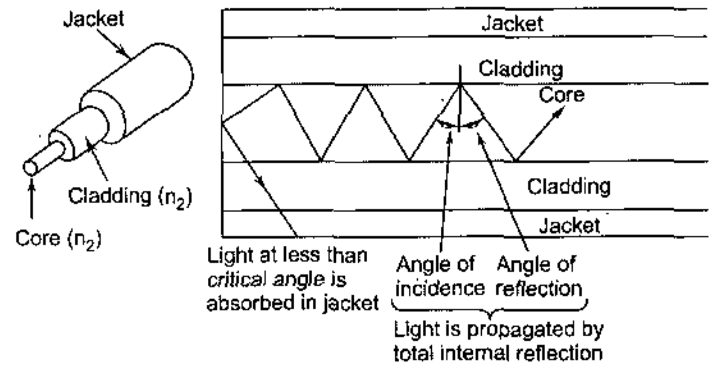


FIG. 2.23 Total internal reflection

2.19 FIBRE GEOMETRY

An Optical fibre consists of a core of optically transparent material usually silica or borosilicate glass surrounded by a cladding of the same material but a slightly lower refractive index. Fibre themselves have exceedingly small diameters. Figure shows cross section of the core and cladding diameters of commonly used fibres. The diameters of the core and cladding are as follows.

Core (μm)	Cladding (μm)
8	125
50	125
62.5	125
100	140

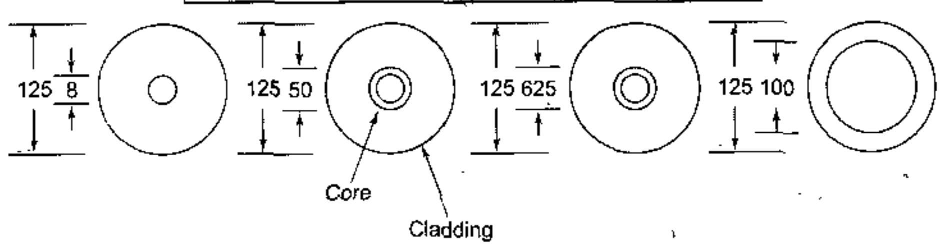


FIG. 2.24 Typical core and cladding

Fibre sizes are usually expressed by first giving the core size followed by the cladding size. Thus 50/125 means a core diameter of 50 mm and a cladding diameter of 125 mm.

SUMMARY

- ISDN is an abbreviation of Integrated Services Digital Network. The current communications networks vary with the type of service, such as telephone network, telex network, and digital data transmission network.
- Several new telephone exchanges have been installed throughout the country for local communication and more and more carrier channels have been provided for carrying the trunk traffic. With the planned introduction of Subscriber Trunk Dialing throughout the country, the number of carrier channels required to interconnect different cities became too high to be accomplished by overhead lines.

- It also enables communication service to those areas which are virtually INACCESSIBLE by other conventional forms of communication system due to natural physical barriers.
- Satellite Communication Network could be defined as an ensemble of earth stations of pre-determined size spread over a pre-defined coverage area, interconnected through a suitably designed satellite, placed at a pre-determined location in properly chosen orbit around the earth.
- Optical Fibre is new medium, in which information (voice, Data or Video) is transmitted through a glass or plastic fibre, in the form of light, following the transmission sequence give below :
- Speed of light is actually the velocity of electromagnetic energy in vacuum such as space. Light travels at slower velocities in other materials such as glass. Light travelling from one material to another changes speed.
- Most fibres have an additional coating around the cladding. This buffer coating is a shock absorber and has no optical properties affecting the propagation of light within the fibre.
- An Optical fibre consists of a core of optically transparent material usually silica or borosilicate glass surrounded by a cladding of the same material but a slightly lower refractive index.

NOTES

REVIEW QUESTIONS

1. What is ISDN?
2. Discuss different ISDN services.
3. Discuss different channels along with their applications.
4. Discuss the advantages of satellite communications.
5. What are different modes of communication in satellite communication?
6. Discuss the principles and features of satellite communications.
7. Discuss fibre optics and its advantages.
8. What are different applications of optical fibre in communications?
9. What is the Transmission sequence?
10. Discuss theory and principle of fibre optics.

FURTHER READINGS

1. *Telecommunication and Information Technology,* Prashant Kaushik, Anmol, 2006.
2. *Optical Networking in Telecommunication,* S. Mukherjee, Jaico.
3. *Wireless Technology and Access of Information,* Ajay K. Srivastav, Shree Pub., 2006.
4. *Elements of Networking Engineering,* Kumar Prasun Ramakrishnan, Shree Pub., 2010.
5. *Trends in Networking and Communication,* Edited by Girish Kumar Srivastav and Charul Bhatnagar, Atlantic Pub., 2009.

NOTES

UNIT III: DATA COMMUNICATION, OSI REFERENCE MODEL, DIGITAL DATA INTERFACE

★ STRUCTURE ★

NOTES

LEARNING OBJECTIVES

After going through this unit, you will be able to:

- define data communication
- know about components
- describe transmission codes
- explain briefly about OSI reference model
- define data encapsulation
- know about protocols
- explain about digital data interfaces
- discuss the lan topology

PART I: INTRODUCTION TO DATA COMMUNICATION

3.1 INTRODUCTION

Communication plays a very important part in our lives because we are almost always involved in some form of communication, *e.g.,*

- Face-to-face conversation
- Sending or receiving a letter
- Watching a film or T. V.
- Attending a lecture
- Reading a book
- Telephonic conversation
- Looking at paintings in an art gallery

There are many other examples of communications and Data Communications is one specific area of whole field of communication. Aim of communication is to transfer some information from one point to another. In data communication, this information is generally called as Data or a message.

3.2 COMPONENTS

In order to send data/message from one point to another, following three components are must:

1. Source
2. Medium
3. Receiver

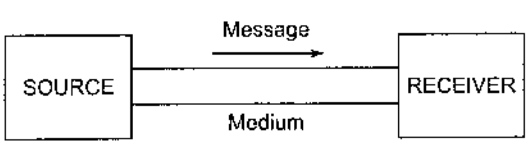


FIG. 3.1 Components

In the above arrangement, it is possible to have both way communication simultaneously. Thus, we need four wire for full-duplex transmission or both way simultaneous communication.

3.4 TRANSMISSION CODES

NOTES

All data communication codes are based on the binary system (1s and 0s). A message can be encoded into a meaningful string of 1s and 0s that can be transmitted along a data line and decoded by a receiver. The string of 1s and 0s is meaningful because it is defined by a code that is known to both the source and the receiver. Code is limited by the number of bits (binary digits) it contains, *e.g.*, one-bit code means that we can have 2 characters so that we can encode the letter A by '0' and B by '1'. Similarly, a 2 bit code will enable us to handle 4 characters. Thus, a n-bit code enables us to handle 2^n characters.

Some commonly used codes are :

1. Baudot code
2. ASCII code
3. BCDIC code
4. EBCDIC code

Baudot Code

This is also called International Alphabet no.2. This is used on International telex network and is often called telex code.

It being a 5 bits code, we can represent 32 characters. This is not enough to handle a full alphanumeric character set so two of the characters are designated as code extension characters. These are letter shift (LS) and figure shift (FS).

LS = 11111 represented by 1. (Downward arrow)

FS = 11011 represented by 1. (Upward arrow)

Pressing LS or FS indicates the receiver that next character will be letter character of figure character as the case may be. So, we can double the number of characters that can be handled by this code.

Limitations

All the 5 bits are used for information and there is no inherent means of error detection.

TABLE 3.1 Baudot codes

Binary	Letters Characters	Figures Characters
00000	Blank	Blank
00001	E	3
00010	?	? Line feed
00011	A	-
00100	SP	SP Space
00101	S	'
00110	I	8
00111	U	7

01000	<	< Carriage return
01001	D	+ Who are you ?
01010	R	4
01011	J	? Bell
01100	N	.
01101	F	
01110	C	?
01111	K	.
10000	T	5
10001	Z	+
10010	L	)
10011	"	2
10100	H	L
10101	Y	6
10110	P	0
10111	Q	1
11000	O	9
11001	B	?
11010	G	$
11011	l	l Figure shift (FS)
11100	M	.
11101	X	/
11110	V	=
11111	l	l Letter shift (LS)

NOTES

ASCII Code (American Standard Code for Information Interchange)

It is an eight-bit code which consists of seven information bits and one bit for parity checking. This is most widely used data code. Seven information bits gives us 128 combinations, which allows us to encode a full keyboard of the computer.

- 52 alphabets (capital and small).
- 0–9 (10 numbers).
- Punctuation marks
- Additional graphic and control characters.

TABLE 3.2 ASCII code set

Bit NO.	765	000	001	010	011	100	101	110	111
4321		0	1	2	3	4	5	6	7
0000	0	NUL	DLE	SPACE	0	@	P		P
0001	1	SOH	DC1	!	1	A	Q	a	q
0010	2	STX	DC2	"	2	B	R	b	r
0011	3	ETX	DC3	#	3	C	S	c	s
0100	4	EOT	DC4	$	4	D	T	d	t
0101	5	ENQ	NAK	%	5	E	U	e	u
0110	6	ACK	SYN	&	6	F	V	f	v
0111	7	BEL	ETB	'	7	G	W	g	w

NOTES

1000	8	BS	CAN	(	8	H	X	h	x
1001	9	HT	EM	)	9	I	X	i	x
1010	A	LF	SUB	*	:	J	Z	j	z
1011	B	VT	ESC	+	;	K	[	k	{
1100	C	FF	FS	,	<	L	\	l	l
1101	D	CR	GS	-	=	M	]	m	}
1110	E	SO	RS	.	>	N	^	n	~
1111	F	SI	US	/	?	U	–	o	DEL

There are 8 columns (0 to 7) and 16 rows (0 to F). *e.g.*, character 'H' is represented as

100	1000	
Column No.	Row No.	Thus ASCII code for character 'H' is P1001000 where P is the Parity Bit.

Parity Bit is placed at (most significant bit) position. Purpose of parity bit is to detect the errors. While transmitting on line, Least Significant Bit (LSB) is transmitted first and MSB at the last, *e.g.*, H is transmitted on line as bit no. 1234567P *i.e.*, 00010011assuming P = 1

We can have odd or even parity. Figure 3.6(*a*) and Figure 3.6(*b*) shows that only odd number of bits going reverse error can be detected. If changes occur in an even number of bits, the parity check will be passed and receiver will assume that it has received a valid character. So parity checking method cannot detect the multiple errors.

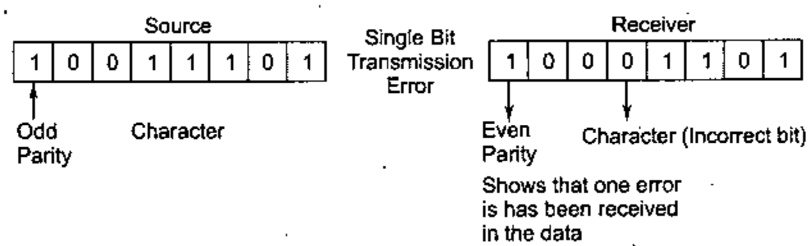


Fig. 3.6 (a) Single bit reversed

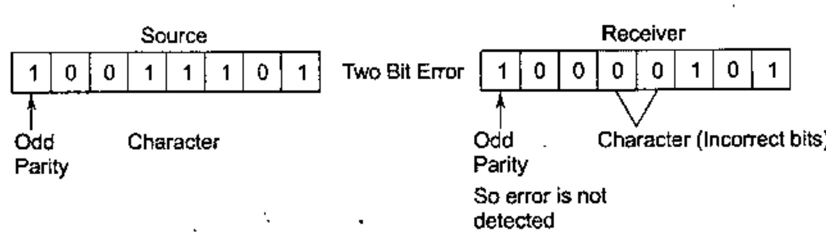


Fig. 3.6 (b) Two bits reversed

BCDIC (Binary Code Decimal Interchange Code)

It is a six-bit code that is used as an internal code by some computers. With 6 information bits, we can have $2^6 = 64$ possible code combinations. For data transmission, code is implemented as 7-bit code containing 6 information bits and one parity bit.

EBCDIC (Extended Binary Coded Decimal Interchange Code)

It is a 8-bit code in which all the 8-bits are used for information (unlike ASCII), giving 256 possible code combinations. EBCDIC is used as an internal machine code in some of the computers.

Data Transmission

(a) Parallel Transmission.
(b) Serial Transmission.

Parallel Transmission

In this method, all bits of encoded character are transmitted simultaneously which means that each bit of the code is having a dedicated channel (Figure 3.7).

NOTES

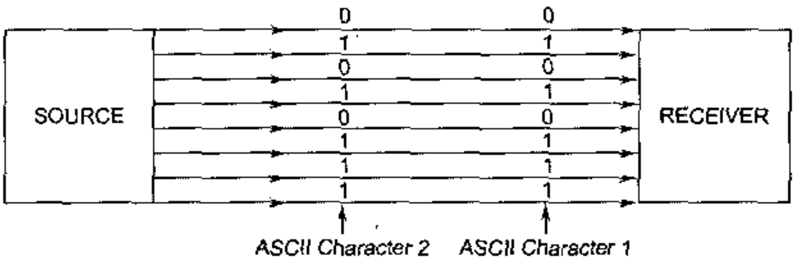


FIG. 3.7 Parallel transmission

It is parallel by bit, serial by character. Here, we need as many numbers of channels as the number of bits in a character.

Serial Transmission

It is the most commonly used method of communication. In this method, bits of the encoded character are transmitted one after the other along one channel serial bit by bit as well as character by character as shown in the Figure 3.8.

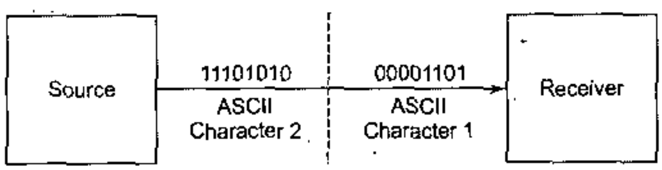


FIG. 3.8 Serial transmission

Receiver then assembles the incoming bit stream into characters. Serial transmission presents problem of synchronization:

(a) Bit synchronization.
(b) Character Synchronization.

Bit Synchronization

Clock is used for synchronization. The source clock tells the source how often to put the bits on to the line and receive clock tells the receiver how often to look at the line, *e.g.*, in Figure 3.9. If we wish to transmit at 100 bits/sec. we set the source clock to run at 100 bits/sec. which tells the source to put the bits on the line 100 times per second. At the receiving end, we would see a bit appearing at the input of the receiver every 1/100th of a second.

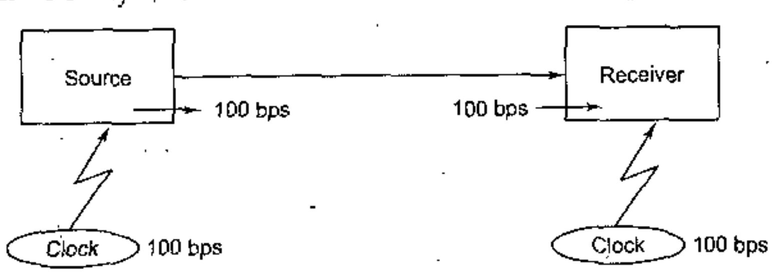


FIG. 3.9 Bit synchronisation

NOTES

We set the receive clock to run at 100 b/s. In most of the systems, timing signals are propagated through the network so that the receiver can derive a clock that is precisely in step with the transmit clock. Extracted clock is applied for sampling the data bits (Figure 3.10).

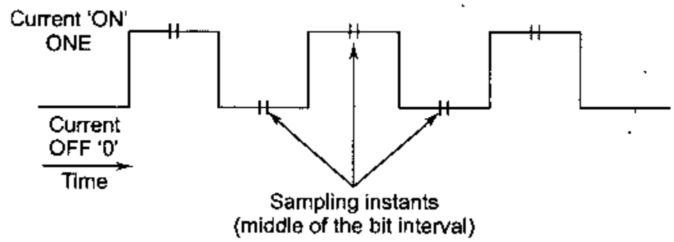


FIG. 3.10

Character Synchronisation

Receiver can identify the character if it knows.

1. How many bits are there in the character ?
2. The speed at which the bits are coming down the line.

Then it can count off the required number of bits and assemble the character once it has identified the first bit of a character. There are two ways to identify the first bit of a character.

1. Synchronous Transmission.
2. Asynchronous Transmission.

Synchronous Transmission

It is used to transmit whole blocks of data at once. Each block of data is preceded with a unique synchronising pattern. This makes use of SYN transmission control character. The SYN character has a bit pattern of 00010110 with odd parity.

Receiver is designed to continuously look towards the 'SYN' character. When it receives the SYN character, it knows the first bit of the information character. But sometimes there is false synchronisation (Figure 3.11) where eight bits of two continuous characters could look like a SYN character.

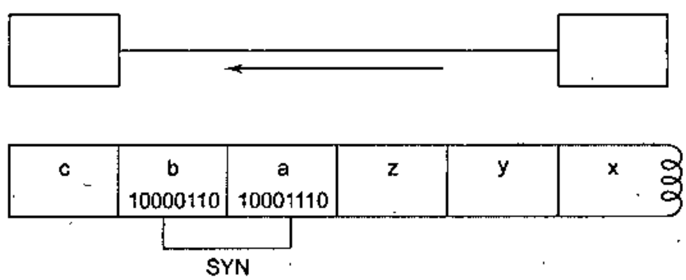


FIG. 3.11 Synchronous transmission

Asynchronous Transmission

It is called start/stop system. In this system Data is transmitted by character. There is no fixed time relationship between one character and the next.

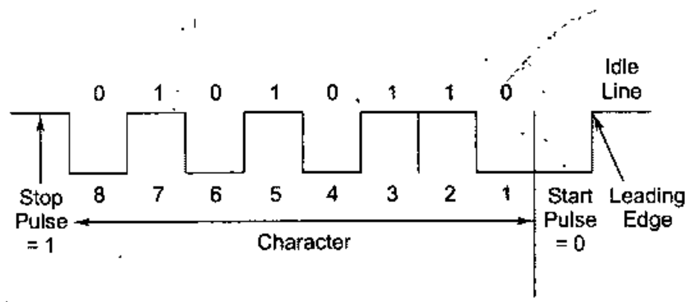


FIG. 3.12 Asynchronous transmission

Receiver re-establishes synchronisation with every character. Each character is preceded by a start pulse '0', which tells the receiver to start receiving character (Leading edge of start pulse synchronises the receiver).

At the end of a character, a stop pulse '1' is applied to allow the receiver to stabilize itself before another character is transmitted. Stop pulse duration varies from 1 bit to 2 bit length. Another reason to add/stop bit is to make the line condition as '1' (Mark) if last bit of last character happens to be '0' (Space). So that the next character will be identified only when polarity changes from '1' to '0', *i.e.*, start pulse is recognised.

PART II: OSI REFERENCE MODEL

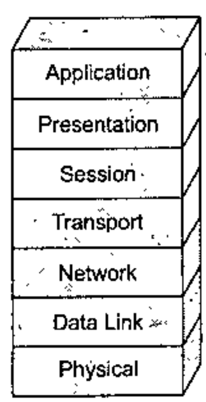
3.5 OSI REFERENCE MODEL

The International Organization introduced the OSI layer for Standardization (ISO) in 1984 in order to provide a reference model to make sure products of different vendors would interoperate in networks. OSI is short for Open System Interconnection.

The OSI layer shows WHAT needs to be done to send data from an application on one computer, trough a network, to an application on another computer, not HOW it should be done. A layer in the OSI model communicates with three other layers: the layer above it, the layer below it, and the same layer at its communication partner. Data transmitted between software programs passes all 7 OSI layers. The Application, Presentation and Session layers are also known as the Upper Layers.

The Data Link and Physical Layers are often implemented together to define LAN and WAN specifications.

- Data Encapsulation
- Application Layer
- Presentation Layer
- Session Layer
- Transport Layer
- Network Layer
- Data Link Layer
- Physical Layer



3.6 DATA ENCAPSULATION

Data Encapsulation is the process of adding a header to wrap the data that flows down the OSI model. Each OSI layer may add it's own header to the data received

NOTES

from above. (from the layer above or from the software program 'above' the Application layer.)

There are five steps of Data Encapsulation :

1. The Application, Presentation and Session layers create DATA from users' input.
2. The Transport layer converts the DATA to SEGMENTS
3. The Network layer converts the SEGMENTS to PACKETS (or datagrams)
4. The Data Link layer converts the PACKETS to FRAMES
5. The Physical layer converts the FRAMES to BITS.

At the sending computer the information goes from top to bottom while each layers divides the information received from upper layers in to smaller pieces and adds a header. At the receiving computer the information flows up the model discarding the corresponding header at each layer and putting the pieces back together.

The Figure 3.13 shows layered model of two directly interconnected end systems. The transmission media is not included in the seven layers and, therefore, it can be regarded as layer number zero. Functions and services of various layers are described

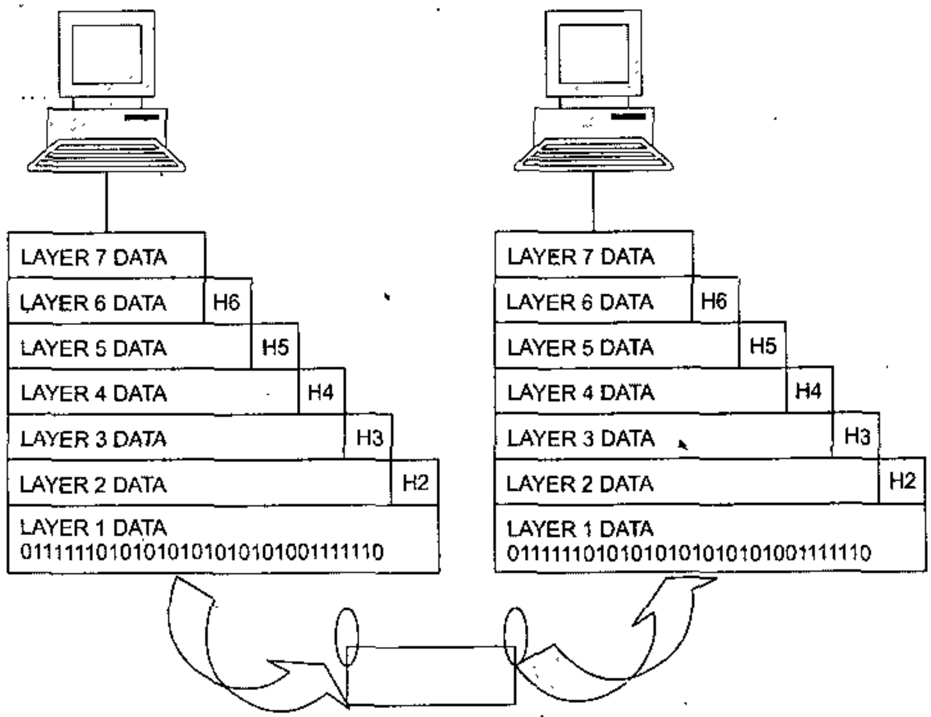


FIG. 3.13

3.7 CHARACTERISTICS OF THE OSI LAYERS

The seven layers of the OSI reference model can be divided into two categories: upper layers and lower layers.

The upper layers of the OSI model deal with application issues and generally are implemented only in software. The highest layer, the application layer, is closest to the end user. Both users and application layer processes interact with software applications that contain a communications component. The term upper layer is sometimes used to refer to any layer above another layer in the OSI model.

The lower layers of the OSI model handle data transport issues. The physical layer and the data link layer are implemented in hardware and software. The lowest layer, the physical layer, is closest to the physical network medium (the network cabling, for example) and is responsible for actually placing information on the medium.

3.8 PROTOCOLS

NOTES

The OSI model provides a conceptual framework for communication between computers, but the model itself is not a method of communication. Actual communication is made possible by using communication protocols. In the context of data networking, a *protocol* is a formal set of rules and conventions that governs how computers exchange information over a network medium. A protocol implements the functions of one or more of the OSI layers. A wide variety of communication protocols exist. Some of these include:

LAN protocols operate at the physical and data link layers of the OSI model and define communication over the various LAN media.

WAN protocols operate at the lowest three layers of the OSI model and define communication over the various wide-area media.

Routing protocols are network layer protocols that are responsible for exchanging information between routers so that the routers can select the proper path for network traffic.

Network protocols are the various upper-layer protocols that exist in a given protocol suite. Many protocols rely on others for operation.

For example, many routing protocols use network protocols to exchange information between routers. This concept of building upon the layers already in existence is the foundation of the OSI model.

3.9 OSI MODEL & COMMUNICATION BETWEEN SYSTEMS

Information being transferred from a software application in one computer system to a software application in another must pass through the OSI layers. For example, if a software application in System A has information to transmit to a software application in System B. The application program in System A will pass its information to the application layer (Layer 7) of System A.

The application layer then passes the information to the presentation layer (Layer 6), which relays the data to the session layer (Layer 5), and so on down to the physical layer (Layer 1). At the physical layer, the information is placed on the physical network medium and is sent across the medium to System B. The physical layer of System B removes the information from the physical medium, and then its physical layer passes the information up to the data link layer (Layer 2), which passes it to the network layer (Layer 3), and so on, until it reaches the application layer (Layer 7) of System B. Finally, the application layer of System B passes the information to the recipient application program to complete the communication process.

3.10 INTERACTION BETWEEN OSI MODEL LAYERS

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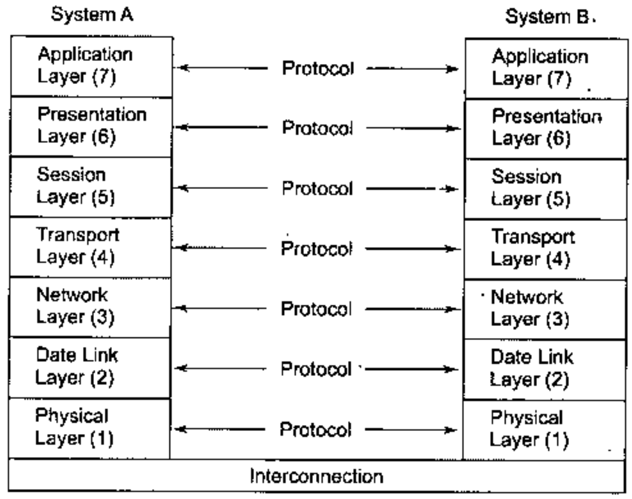
A given layer in the OSI model generally communicates with three other OSI layers: the layer directly above it, the layer directly below it, and its peer layer in other networked computer systems. The data link layer in System A, for example, communicates with the network layer of System A, the physical layer of System A, and the data link layer in System B. Figure below illustrates this example.



One OSI layer communicates with another layer to make use of the services provided by the second layer. The services provided by adjacent layers help a given OSI layer communicate with its peer layer in other computer systems. Three basic elements are involved in layer services: the service user, the service provider, and the service access point (SAP).

In this context, the *service user* is the OSI layer that requests services from an adjacent OSI layer. The *service provider* is the OSI layer that provides services to service users. OSI layers can provide services to multiple service users. The SAP is a conceptual location at which one OSI layer can request the services of another OSI layer.

3.11 APPLICATION LAYER (LAYER 7)

Application Layer provides network services directly to applications. Type of software programs vary a lot: from groupware and web browser to Tactical Ops (video game). Software programs itself are not part of the OSI model. It determines the identity and availability of communication partners, and determines if sufficient resources are available to start program-to-program communication. This layer is closest to the user. Gateways operate at this layer. Following are the examples of Application layer protocols:

(*i*) Telnet
(*ii*) SMTP
(*iii*) FTP
(*iv*) SNMP
(*v*) NCP
(*vi*) SMB

3.12 PRESENTATION LAYER (LAYER 6)

Presentation Layer defines coding and conversion functions. It ensures that information sent from the application layer of one system is readable by the application layer of another system. It includes common data representation formats, conversion of character representation formats, common data compression schemes, and common data encryption schemes, common examples of these formats and schemes are:

(*i*) MPEG, QuickTime
(*ii*) ASCII, EBCDIC
(*iii*) GIF, TIFF, JPEG

Gateways operate at this layer. It transmits data to lower layers.

NOTES

3.13 SESSION LAYER (LAYER 5)

The session layer establishes, manages, maintains and terminates communication channels between software programs on network nodes. It provides error reporting for the Application and Presentation layer. Examples of Session layer protocols are:

(*i*) NFS
(*ii*) SQL
(*iii*) RPC
(*iv*) Zone Information Protocol (ZIP)

Gateways operate at this layer. It transmits data to lower layers.

3.14 TRANSPORT LAYER (LAYER 4)

The main purpose of this layers is making sure that the data is delivered error-free and in the correct sequence. It establishes, maintains and terminates virtual circuits. It provides error detection and recovery. It is concerned with reliable and unreliable transport. When using a connection-oriented, reliable transport protocol, such as TCP, acknowledgments is send back to the sender to confirm that the data has been received. It provides Flow Control and Windowing. It provides multiplexing; the support of different flows of data to different applications on the same host. Examples of Transport layer protocols are:

(*i*) TCP (connection-oriented, reliable, provides guaranteed delivery.)
(*ii*) UDP (connectionless, unreliable, less overhead, reliability can be provided by the Application layer)
(*iii*) SPX

Gateways operate at this layer. It transmits data to lower layers.

3.15 NETWORK LAYER (LAYER 3)

This layer defines logical addressing for nodes and networks/segments. It enables internetworking, passing data from one network to another. It defines the logical network layout so routers can determine how to forward packets trough an internet-work. Routing occurs at this layer, hence Routed and Routing protocols reside on this layer. Routed protocols are used to encapsulate data into packets. The header added by the Network layer contains a network address so it can be routed trough an internet-work. Examples of Network layer Routed protocols are:

- Flexible to accommodate
  - Changes in physical location of the stations
  - Increase in number of stations.
  - Increase in LAN coverage.
- Consistent with the media access method
- Minimum cost of physical media.

NOTES

Bus Topology

In bus topology, a single transmission medium interconnects all the stations (Figure 3.14) All stations share this medium for transmission to any other station. Every stations listens to all the transmissions on the bus. Every transmission has source and destination address so that stations can pick the messages meant for them and identify their senders.

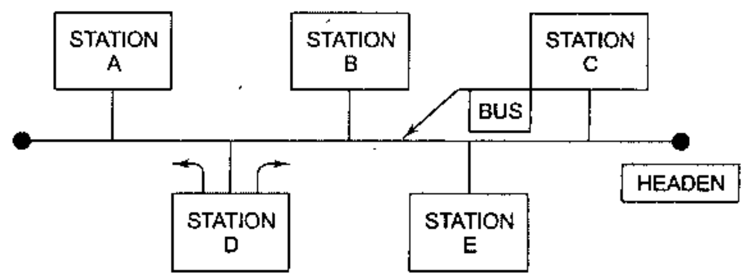


Fig. 3.14

Figure 3.14 shows a two-way bus. Each station injects its signals on the bus, which flow in both the directions. To avoid signal reflection at the ends of the bus, the bus is terminated by appropriate impedance (characteristic impedance) called head end. Note that signal flow is bidirectional, therefore amplifiers can not be used to compensate for bus attenuation. Repeaters which interconnect two buses are used for extending the physical coverage of the network (Figure 3.15). A repeater is transparent to rest of the system in the sense that it does not have buffer and interconnects the two sections to make them virtually one section.

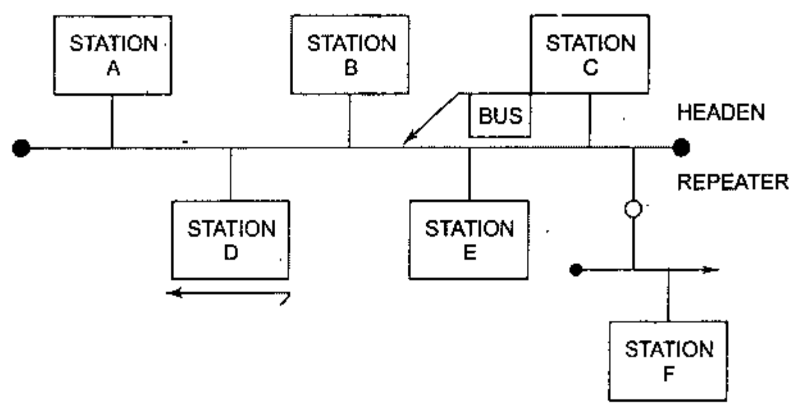


Fig. 3.15

If signals are amplified along the bus, the bus becomes unidirectional. Therefore two separate channels are required. Every station injects signal on one bus and listens to the other bus. At one end of the LAN, the buses are looped. These two channels can be provided on a signal bus also using frequency division multiplexing. In this case, the head-end contains a frequency translator. Stations transmit on one frequency and listen to other frequency.

Advantages of Bus Topology

- Stations are connected to the bus using a passive tap.
- Least amount of media is used.
- Coverage can be increased by extending the bus using repeaters.
- New stations are easily added by tapping working bus.

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Disadvantages of Bus Topology

- Fault diagnostics is difficult
- Fault isolation is difficult
- Nodes must be intelligent

Ring Topology

A ring network consists of a number of transmission links joined together in form of a ring through repeaters called Ring interface Units (RIU). The transmission is usually unidirectional on the ring. Thus each repeater receives the signals at its input and after regeneration, sends it to the repeater of the next station. If the frame belongs to the station, a copy of the incoming frame is retained. Each frame contains source and destination addresses. Figure. 3.16 shows a ring network.

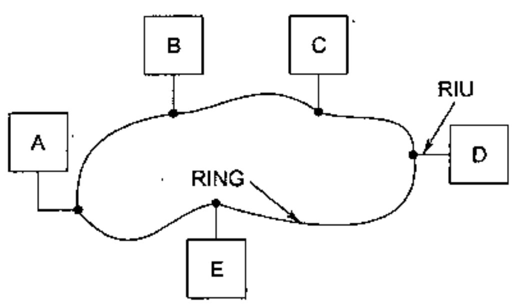

Fig. 3.16

Unlike a bus, signals on the ring never reach an end. They will keep circulating in the ring unless removed. This responsibility is usually given to the source because the destination may not be available. Possibility of source going down after transmitting a frame cannot be ruled out. Therefore a monitoring station is required to remove continuously circulating frames. Rings are not as flexible as bus because to add a station means breaking the ring and adding an RIU. Another possible problem can be an RIU may fail resulting in total network failure. A "Dead Man Relay" is usually provided to bypass a failed RIU. Wire centers are provided to improve flexibility of removing or adding a station and to isolate a faulty section to add a new station, cables are laid to the wire center. The bypass relays are also moved to the wire center. Wire centers can be connected together to increase geographic coverage of the network. A ring network does not economize on cables.

Advantages of Ring Topology

- Short cable length
- Suitable for optical fiber

Disadvantages of Ring

- Node failure cause network failure

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- Difficult to diagnose fault
- Network reconfiguration is difficult

Star Topology

A star network consists of dedicated links from the stations to the central controller (Figure 3.17). Each interconnection supports two-way communication. The central controller acts as a switch to route the frames from source to the destination unlike ring or bus topologies where communication is in broadcast mode.

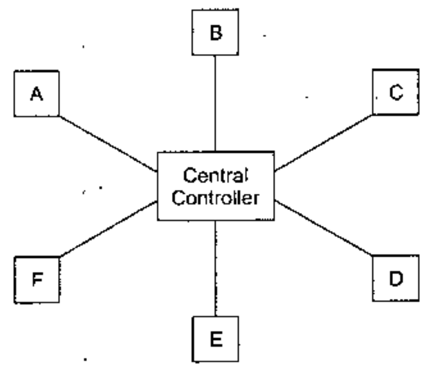

FIG. 3.17

Advantages of Star Topology

- Control/fault diagnostics is centralized
- Simple access protocols are employed
- Ease of service
- One device per connection

Disadvantages of Star Topology

While star topology is well understood and is based on prove technology (telephone network), its disadvantages are:

- Single point of failure
- No sharing of transmission
- Long cable lengths involved
- Difficult to expand
- Central node dependency

3.21 MEDIA ACCESS CONTROL

The physical topology of local area networks can take shape of a bus or ring or a star but the network attributes in terms of delay, throughput, expandability etc., are determined by the mechanisms utilized for sharing the use of physical interconnecting media. There are many methods sharing the media and they are, in general, called Media Access Control methods. These methods can be categorized as :

- Access Centrally Controlled
- Distributed Access Control

In the former access to the media is controlled by a central controller. There are several ways in which the media is shared, e.g., polling, demand assigned time division or frequency division multiple access, etc. But in local area networks, *distributed access control* methods are more common and are described in the following sections.

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Distributed Access Control

As name implies, there is no single controller for the shared media. A disciplines *built up among the various stations* (station refers to a LAN terminal/host/any other device) of the local area network so that a fair opportunity is given to each station to transmit its data which is in the form of frames. The basic advantage of distributed control over centrally controlled methods is that there is no single point of network failure. Distributed access control methods are available both for bus and ring topologies.

Media Access Control—Bus Topology

An interconnecting bus can be thought of as *single data transmission channel* to which all the stations of the network are connected. A bus can be a twisted pair cable, a coaxial cable or even an optical fiber cable.

The bus operates in broadcast mode, *i.e.*, all the stations are always listening to all the transmissions on the bus. Access control mechanisms are so designed that transmissions from *different stations do not intermingle and all the stations get* fair chance to transmit. There are two techniques which dominate the present day market : Token passing and CSMA/CD. These two techniques are described below. CSMA/CD is a contention access technique and covered under this general heading.

Token Passing

In token passing method, the stations connected on a bus are arranged in a logical ring *i.e.*, the addresses of the stations are assigned a logical sequence with the last number of the sequence followed by the first (Figure 3.18) each station knows identity of the stations proceeding and following it.

Access to the interconnecting bus is regulated by a control frame known as token. At a time only one station which holds the token has right to transmit its frame on the bus. The operation of token passing bus is as follows:

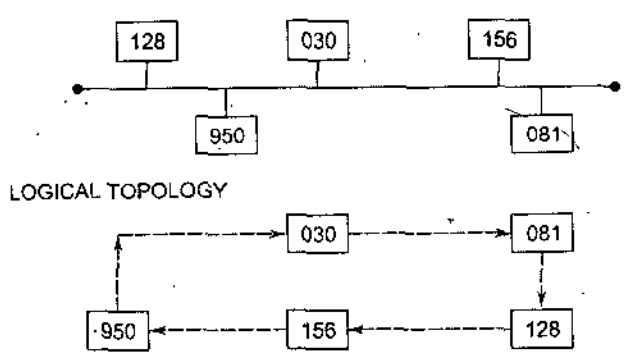

FIG 3.18

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On the bus all the stations operate in broadcast mode so that every station can hear every transmission. When a station detects a token on the bus with its address, it transmits its data frame(s) each containing the source and destination addresses (Figure 3.19). In the end, it transmits the token with address of the next station in the logical ring. Thus, in one cycle each station gets an opportunity to transmit. It is possible for a station to have more than one turn in a cycle (e.g. by giving it more than one address).

To maintain continuity of communication, it is necessary that when turn comes each station transmits the token frame even if there is no data to send. The transmission sequence can get disrupted if a station is down. To account for such eventuality, a timer is provided. Station holding the token must release the token before time out else next station takes over and deletes the station from the logical ring for future transmissions.

PREAMBLE	SD	FC	DA	SA	DATA	FCS	ED

PREAMBLE	-	BIT SYNCHRONISATION	SA	-	SOURCE ADDRESS
SD	-	FRAME START DELIMITER	DATA	-	DATA FIELD
FC	-	FRAME CONTROL (TYPE)	FCS	-	FRAME CHECK SEQUENCE
DA	-	DESTINATION ADDRESS	ED	-	END DELIMITER

FIG. 3.19

The frame format shown above is as per IEEE 802.4 standard. The "FC" field indicates whether it is a data frame or token. Delay in completing data transfer depends on number of stations, propagation and transmission times and traffic. Token passing LANs operate at data rates 1 Mbps to 10 Mbps.

Contention Access

CSMA/CD is a contention access method in which there is no scheduled time or sequence for stations to transmit on the medium. They compete for the use of the medium. It is, therefore, quite likely that more than one station will transmit simultaneously and the data frames will "collide". There are rules and mechanisms to reduce the likelihood of these collisions. These methods in general are called contention access methods for the way in which access to the medium is gained. Origination of contention access techniques is Aloha Radio network which used a random access mechanism now called Pure Aloha. We shall, therefore, first look at the Aloha access mechanism.

Pure Aloha

This access mechanism was originally used in Aloha Radio Network which provides a single radio channel for access by number of stations to a central computer. The scheme is as under :

- A station can transmit whenever it wants. There is no pre-assigned time or sequence. If a station starts to transmit when another transmission is already in progress, collisions will occur. but there will be some instances when transmissions will reach the destination without any collision.
- A mechanism to detect collision is established (*e.g.*, acknowledgement). Collision is assumed to have occurred and the message is retransmitted if the acknowledgement is not received within specified time (twice the propagation time plus processing time).

The link utilization of Pure Aloha access method is somewhat modest, mere 18%.

The reason for poor efficiency is large wasted time when a collision occur (Figure 3.20). Note that even though there may have been only a few bits of overlap, the whole of the two packets time is wasted due to the collision.

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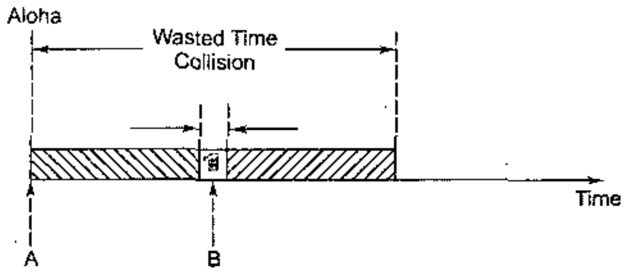

FIG. 3.20

Provided that there is relatively little traffic on the network and the packet size is fairly small, then this technique is reasonably good.

Slotted Aloha

Wasted time due to collisions can be reduced if all the transmissions are synchronized. The channel time is divided into time slots and the stations are allowed to transmit at specific instants of time so that all transmissions arrive aligned with the time slot boundaries (Figure 3.21).

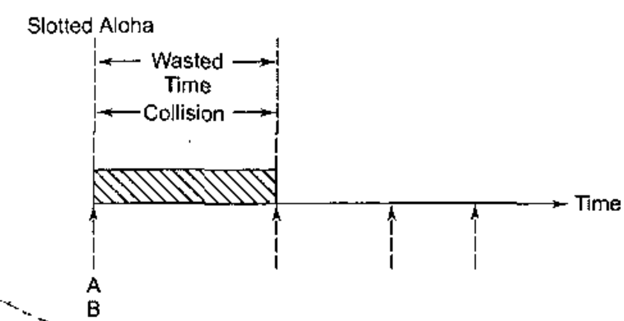

FIG. 3.21

3.22 CARRIER SENSE MULTIPLE ACCESS (CSMA)

In Aloha channel discussed above, possibility of collision can be reduced if some discipline is built into totally random access mechanism. If a station senses the carrier (we are using the term carrier, despite the fact that most of the baseband local area networks do not use carrier to transmit data) before starting its own transmission, a collision can be avoided. CSMA as the name suggests, is based on this principle. Consider a situation where frame transmission time is much more than propagation time, *i.e.*, once a transmission starts, it is soon heard by all. If a station has a frame to send, it listens to the channel, and if it is quite, it starts to transmit. It will be soon heard by other stations, which defer their transmission on sensing the carrier. Contention for the channel can take place only during the first few bits of the frame when the first bit is still in transit. Once, the first bit is heard by every terminal, there cannot be a collision. To account for possibility of collision during first few bits, acknowledgement is transmitted by the recipient. In CSMA, an algorithm is needed to specify when a station can transmit once the

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channel is found busy. There are several ways in which the waiting frames can be transmitted (Figure 3.22).

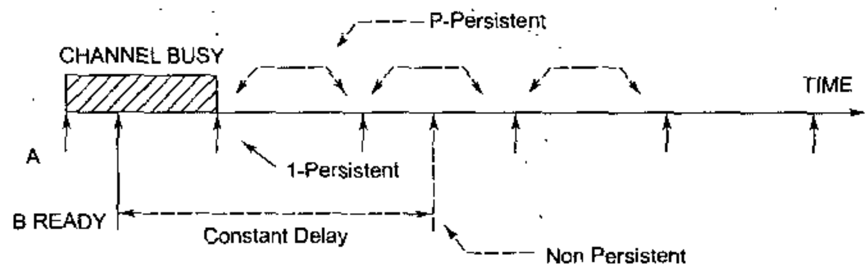

FIG. 3.22

Non-Persistent CSMA

In this scheme, when a station having a frame to send finds that channel is busy, it backs off and waits for a fixed interval of time. Then it again senses the channel for transmission of its frame. If the channel is free, it transmits. The back off delay is determined by transition time, propagation time and other system parameters There is likelihood of some wasted idle time when channel is not in use by any station.

1-Persistent CSMA

In this scheme, stations wishing to transmit, monitor the channel continuously until the channel is idle and then transmit immediately. The problems with this strategy is that if two stations are waiting to transmit, then they will collide always, and require retransmission.

P-Persistent CSMA

To reduce the probability of collision in 1-Persistent CSMA, not all the waiting stations are allowed to transmit immediately after the channel is idle. A waiting station transmits with probability p if the channel is idle. For example, if 10 stations are waiting with $p = 0.1$, on average only one station will transmit and the rest nine will wait. Optimized P-persistent CSMA is about 82% efficient from link utilization point of view while 1-persistent CSMA achieves about 53% efficiency.

CSMA/CD

One of the most commonly used multiple access technique in the current local area networks is CSMA/CD where CD stands for Collision Detection. One limitation of CSMA techniques discussed above is that even after a collision has occurred, the stations continue transmissions till all the bits of the frames are over. This result in unnecessary wastage of channel time. If the stations listen to the channel while they are transmitting, a collision can be detected as soon as it occurs and further transmission can be abandoned thus saving the channel time. This scheme is known as CSMA/CD and as illustrated in Figure 3.23. Naturally for the technique to work properly, each station should not attempt to transmit again immediately after a collision has been detected. Otherwise the same frames will collide again. Usually the stations are given a random back off delay for retry. If collision repeats; back off delay is increased. In Ethernet back off delay is doubled on each repetition of collision. In this way, the network adapts itself to the loading conditions. By careful design, it is possible to achieve efficiencies more than 90% using CSMA/CD.

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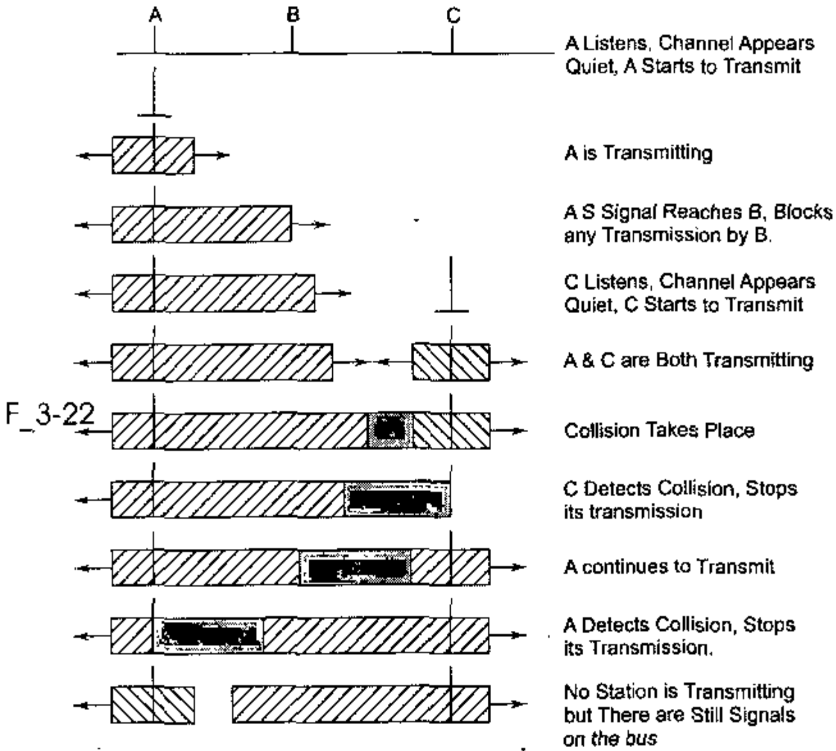

FIG. 3.23

Frame format as per IEEE 802.3 standard is shown in Figure 3.24. The format is very close to that of Ethernet. The individual fields are as given below :

- Preamble (PR) – used for bit synchronization
- Start Frame Delimiter – identifies start of the frame (SFD)
- Destination Address – specifies the destination (DA)
- Source Address (SA) – identifies the source
- Length (L) – specifies number of data bytes
- Data – contains data bytes
- Pad – contains additional bytes to make up the frame of required size.
- Frame Check Sequence - (FCS)

PR	SFD	DA	SA	L	DATA	PAD	FCS

PR	:	PREAMBLE	SA	:	SOURCE ADDRESS
SFD	:	START FRAME DELIMITER	L	:	LENGTH
DA	:	DESTINATION ADDRESS	FCS	:	FRAME CHECK SEQUENCE

FIG. 3.24

PART IV: DATA LINK CONTROL

3.23 DATA LINK CONTROL

The services provided by the physical layer are limited to conversion of bits into electrical signals and transmission of these signals over the interconnecting media. But it is not sufficient for reliable transmission of the bits. Disturbed line conditions of the media may introduce errors which must be taken care of. The basic function-error control and other associated functions are carried out by the second layer of the OSI model, data link layer. In the following sections, we shall discuss in general the mechanisms used for implementation of these functions in various data link protocols.

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3.24 NEED FOR DATA LINK CONTROL

Let us consider a situation, as shown in Figure 3.25 where two digital devices A and B need to exchange information. These devices could be computers, concentrators or other data terminal equipment.

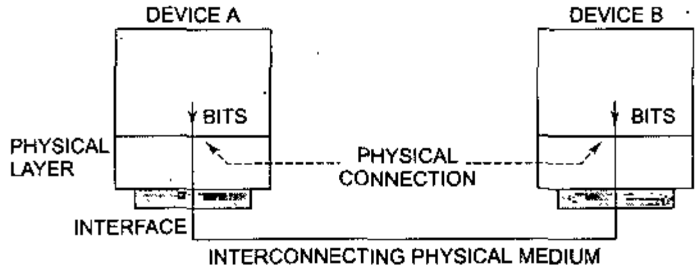

FIG. 3.25

To transfer digital information from device A to B and in the opposite direction, we require :

- An interconnecting transmission medium to carry the electrical signals (*e.g.*, copper wires).
- A standard interface between the device and the interconnecting transmission medium (*e.g.*, RS-232-C).
- Services of the physical layer to convert bits (1s and 0s) into electrical signal.

These three elements provide only the capability for transparent exchange of bits over the physical connection. The electrical signal, however, may get corrupted by the noise encountered during transmission over the physical medium and this may result in introduction of errors in the data bits. Therefore, a mechanism to control the transmission errors is required.

The bits could also be lost if the receiving device is not ready for incoming bit stream. Therefore, a data flow control mechanism also needs to be implemented. Flow control also enables error control because an error is detected, the receiver should be capable halting further incoming flow of bits till the errors have been corrected.

Error control and flow control functions ensure reliable transfer of bits from one device to the other. These functions are not carried out by the physical layer but are implemented in the data link layer.

3.25 DATA LINK LAYER

The data link layer constitutes the second layer of the hierarchical OSI model. It incorporates certain data link control processes which carry out error control, flow control and the associated link management functions. The data link layers together with physical layers and interconnecting medium provide a data link

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connection for reliable transfer of data bits over imperfect physical connection (Figure 3.26).

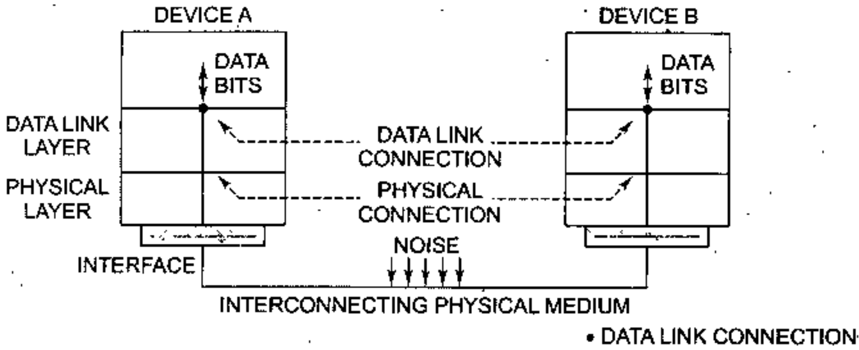

FIG. 3.26

To carry out error and flow control functions, the data link layer receives the data to be sent to the other device from the next higher layer, adds some control bits to a block of data bits as a separate field and hands over the data block containing control bits (called a frame) to the physical layer. The physical layer converts the bits to an electrical signal for transmission over the interconnecting transmission media (Figure 3.27).

At the receiving end, the incoming electrical signal is converted back to bits by the physical layer and the frame is handed over to the data link layer. The data link layer removes the control bits and hands over the received data bits to the next layer. These control bits are used to realise the data link layer functions.

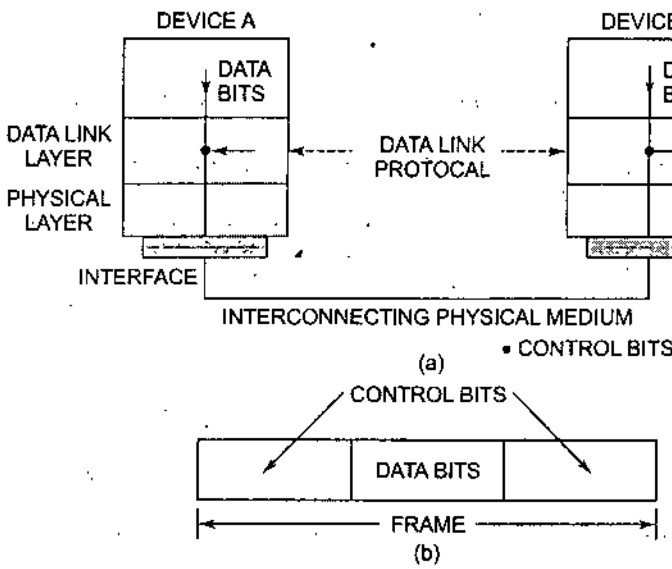

FIG. 3.27

It is essential that the structure of the frame should be known to both the devices so that control bits could be identified. The devices should also agree on the set of procedures to be adopted for exchange of the control information. The specified set of rules and procedures for carrying out data link control functions is called data link protocol.

3.26 DATA LINK PROTOCOLS

Protocols are rules and procedures governing the interaction between two peer (equal) entities. In the present context these entities are the data link control processes constituting the data link layers. There are many data link protocols developed by various manufacturers and organisations. While all the protocols broadly satisfy the basic requirements of the data link layer, the services offered are different. Examples of data link protocols are :

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- Binary Synchronous Data Link Control (BISYNC, BSC).
- Synchronous Data Link Control (SDLC).
- High Level Data Link Control (HDLC).
- Advanced Data Communication Control Procedure (ADCCP).

BISYNC and HDLC are the most commonly used protocols today, but HDLC is going to be the most important in future because it has been accepted as a standard. A data link protocol must specify :

- the format of the frames, *i.e.*, the position and size of the various fields.
- the contents of the fields containing control bits.
- the sequence of the messages to be exchanged to carry out error control, flow control and link management functions.

Data link functions are implemented differently in different protocols. But they consist of similar set of processes which are described below.

3.27 FRAMING

The first and foremost task of the data link layer is to format the user data as series of frames each having a predefined structure. The frame format, in general, consists of three components as shown in Figure 3.28(*a*).

- Header
- Data
- Trailer

The data field contains the data bits received from the next higher layer and which are to be transmitted across the data link. The header and the trailer consist of one or more fields containing data link protocol control information.

Composition of an HDLC information frame is shown in Figure 3.28(*b*). Note that header and trailer consist of several fields. The frame starts with a flag to identify the start of a frame. The flag is followed by an address field. The control field contains sequence number of the frame, acknowledgement of having received a good frame or other link control information.

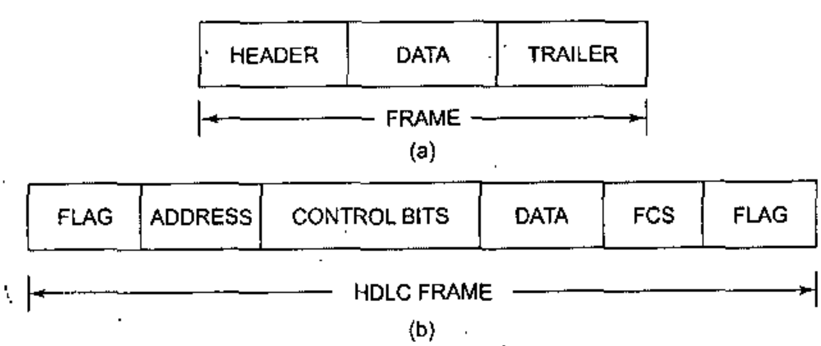

Fig. 3.28

The trailer consists of Frame Check Sequence (FCS) containing error detection bits and a flag indicating end of the frame.

It is not necessary that all frames contain all the fields. Some of the frames contain only the control information. These frames are used for link management, error and flow control functions.

3.28 FRAME FORMAT

NOTES

A frame consists of number of fields and, therefore, its format is so designed that the receiver is able to

- locate the beginning of each frame.
- locate various fields in a frame.
- separate the data field.

For this purpose, some field identifiers and frame identifiers, are incorporated. For example, in the HDLC frame as shown in Figure 3.28(*b*), the flags identify the start and end of the frames of the other fields, address, control and FCS fields have fixed number of bits, so they can be easily separated once flags have been identified. The remaining bits of the frame constitute the data field.

3.29 TRANSPARENCY

Problems may arise if the data field in a frame contains bit patterns similar to the bits in the header or trailer. For example, if the data contains a bit pattern same as the flag in a HDLC frame, the receiver may mistake it for end of the frame. Therefore, the frame format should be so designed that such situations are taken care of an no restrictions are placed on the user as to what bit patterns can/cannot be sent.

3.30 FLOW CONTROL

Flow control mechanisms are incorporated to ensure that the data link layer at the sending end does not send more data frames than what the data link layer at receiving end is capable of handling. Therefore, the receiver is to be provided with a control to regulate the flow of the incoming frames. This control is in form of an acknowledgement (ACK) which is sent by the receiver, the acknowledgement serves two purposes :

- It clears the sending end to transmit the next data frame.
- It acknowledges receipt of previous frame(s).

The two commonly used flow control mechanisms are STOP-AND-WAIT and SLIDING WINDOW.

3.31 STOP-AND-WAIT FLOW CONTROL

In stop-and-wait flow control mechanism, the sending end sends one frame at a time and waits for an acknowledgement from the receiver. The receiver can temporarily stop flow of frames by witholding the acknowledgement. Alternatively, it can request for temporary suspension of transmission of the frames by sending an acknowledgement and wait signal (WACK). On receipt of WACK, the sending end has to wait for ACK to commence transmission of next frame. Figure 3.29 illustrates the mechanism.

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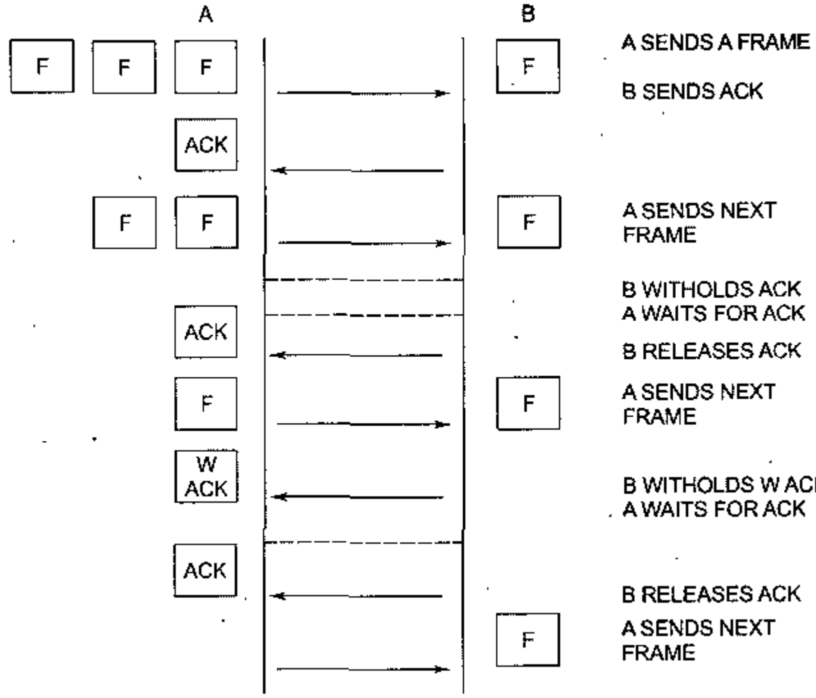

FIG. 3.29

Link Utilisation

In stop-and-wait flow control mechanism, only one frame is sent at a time and each frame is individually acknowledged. The frame and the acknowledgement take certain amount of propagation time to travel across the transmission media.

Propagation time can be as large as 270 milliseconds for a satellite communication link or a few milliseconds for a terrestrial link. Large propagation time makes stop-and-wait mechanism very inefficient from the point of view of link utilisation.

3.32 SLIDING WINDOW FLOW CONTROL

Sliding window flow control mechanism allows transmission of multiple frame without acknowledgements for individual frames. Before going into the actual operation of the mechanism, let us first see its basic features :

- Each frame carries a sequence number for its identification.
- The sending end maintains a window containing a fixed number of frames ready for transmission.
- Frames in a window can be sent without waiting for any frame acknowledgement. But a copy of each transmitted frame is retained in the window till it is acknowledged.
- The number of frames in a window is called size of the window. Its typical value is seven.
- The receiver acknowledges receipt of one or more frames by sending back a numbered acknowledgement (Receive Ready, RR-N) signal. N is the sequence number the next frame it expects to receive. Note the change in terminologies. Instead of ACK, RR is used.

- When an acknowledgement is received all previous frames are assumed acknowledged. For example, by sending RR-5, the receiver is acknowledging receipt of frames bearing numbers 4, 3, 2, etc.
- When an acknowledgement is received by the sending end, it slides the window deleting the copies of acknowledged frames and inserting same number of new frames from the queue of frames waiting for transmission.
- To stop the transmission temporarily, the receiving end can send a Receive Not Ready (RNR-N) signal. RNR-N is acknowledgement up to frame N-1 and a request to stop further transmission temporarily. The transmission can be resumed when RR-N is released by the receiving end.

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Figure 3.30 illustrates the operation of the mechanism. Device A is sending frames to B. Let us assume that the window size is seven and the window is initially located on the frames F1 to F7.

A initiates the transmission with its first frame F1 followed by frames F2, F3, etc. A can send upto F7 without waiting for an acknowledgement from B.

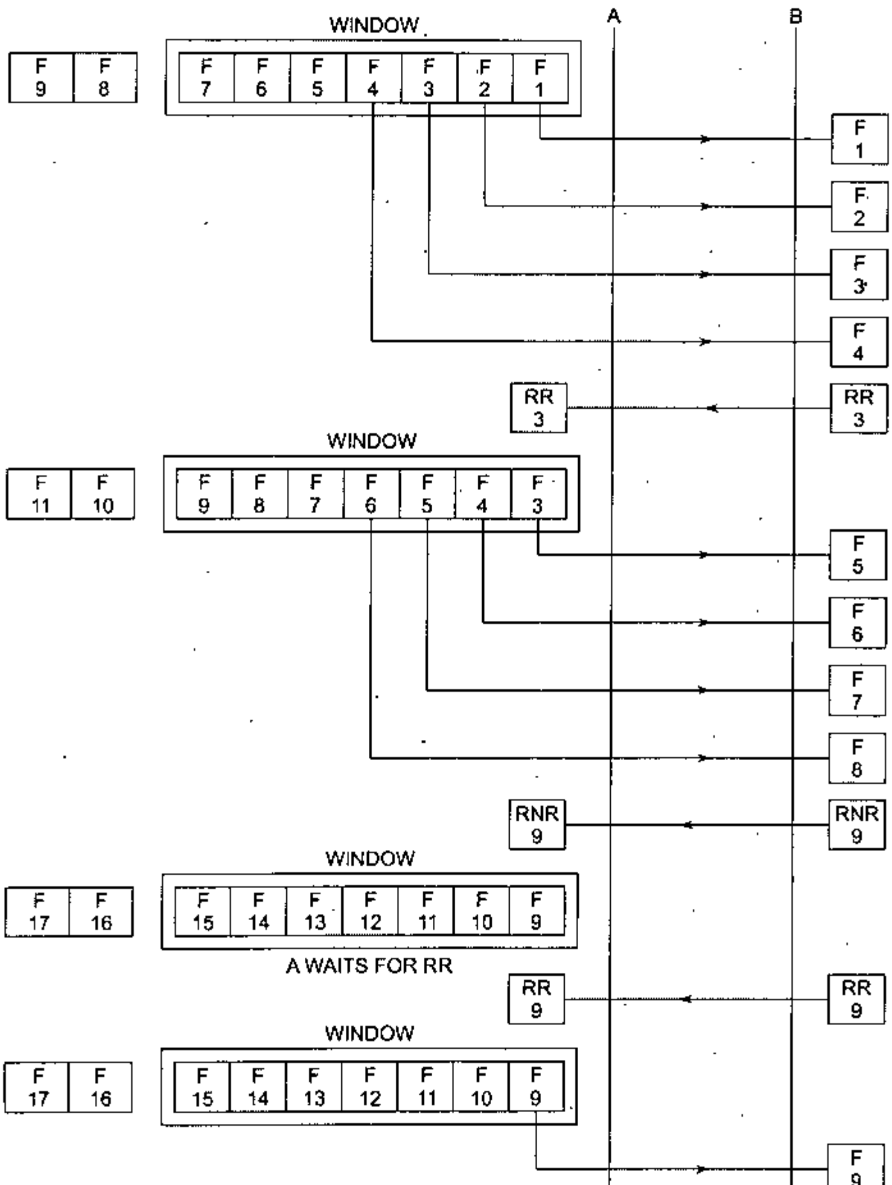

FIG. 3.30

While A is in the process of sending F4, it receives an acknowledgement RR-3 from B indicating to A that F1 and F2 have been received by B. A slides the window by two frames deleting F1 and F2 from the window. F3 to F9 now occupy the window and frames up to F9 can be sent without waiting for further acknowledgement. To temporarily stop the flow of frames, B sends RNR-9. Transmission is resumed when B sends back RR-9.

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3.33 SEQUENCE NUMBERING

In sliding window, all frames are given a sequence number which is a binary number of fixed number of bits. The number of bits are determined by the window size. If the sequence number consists of "*n*" number of bits, the maximum size of the window can be $2n$-1.

Link Utilisation

Unlike stop-and-wait mechanism, in sliding window flow control, each frame is not individually acknowledged and, therefore, the sending end can send number of frames one after the other without waiting for acknowledgement.

3.34 DATA LINK ERROR CONTROL

Two types of errors can occur during transmission of frames from one device to the other.

- Content errors.
- Flow integrity errors.

Errors contained in a received frame are termed as content errors. Flow integrity errors refer to the lost/duplicate frames and acknowledgements. Error control involves three phases :

- Error Detection.
- Error Correction.
- Recovery.

Content errors are detected using parity check or cyclic redundancy codes. The parity bits or CRC check bits are added as the trailer in a frame at the sending end.

The most common method of error correction is retransmission of the frame. The receiver informs the sending end of the error and the sending end retransmits. Note that it is essential for the sending end to retain a copy of the transmitted frame until it is acknowledged by the receiver.

Recovery refers to how the system gets back into normal operating mode after a correction has been made. Generally, the sending end treats each retransmission same as the first transmission and continues with the next frame. It keeps a historical record to detect if the link has become very noisy resulting in very frequent retransmissions. In such an eventuality, it may initiate recovery procedures which may involve re-establishment of the link.

For the flow integrity errors, the data link protocols specify the procedures to be adopted to detect and recover the missing frames and acknowledgements. It must be remembered that no error method is 100% effective. There will always be some undetected content and flow integrity errors. Residual Error Rate (RER) refers to the errors that still exist in the data stream after all error control procedures have been completed.

3.35 ERROR CONTROL IN STOP-AND-WAIT MECHANISM

In the stop-and-wait mechanism, the receiver *sends a positive acknowledgement* (ACK) if there is no content error in the received data frame else it responds with a negative acknowledgement (NAK). The sending end continues with the next frame if it receives an ACK or it repeats the previous frame if it receives a NAK. Figure 3.31 illustrates the mechanism. To deal with flow integrity errors the sending end is equipped with a timer. After sending a frame, the sending end waits for a acknowledgement (ACK or NAK) for a specified time period. When the time expires, it challenges the other end by *sending an Enquiry (ENQ). The receiver responds* with previous ACK or NAK.

The algorithm described above does not cover all contingencies. If a frame is lost *during the transmission*, the receiver will have no knowledge of the frame and, therefore, when it is challenged by the sending end, it shall respond with the acknowledgement of the previous frame. The sending end will obviously misinterpret it as acknowledgement for the lost frame. So a frame will be completely lost without the sender or the receiver being aware of the loss (Figure 3.31).

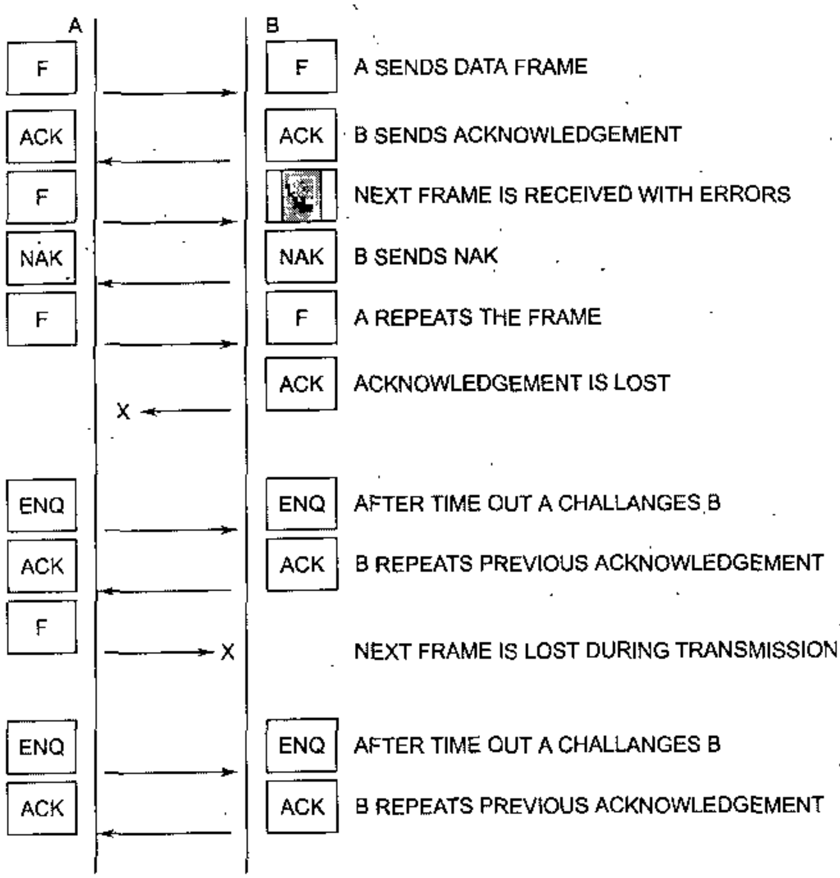

FIG. 3.31

A variant of the above scheme overcomes this problem by distinguishing between the acknowledgements of *the consecutive frames. Each acknowledgement carries*

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a number indicating whether it is acknowledging an even or odd frame. The frames do not carry any number. It is for the sending end to remember the designation (EVEN,ODD) of each frame. The mechanism is illustrated in Figure 3.32.

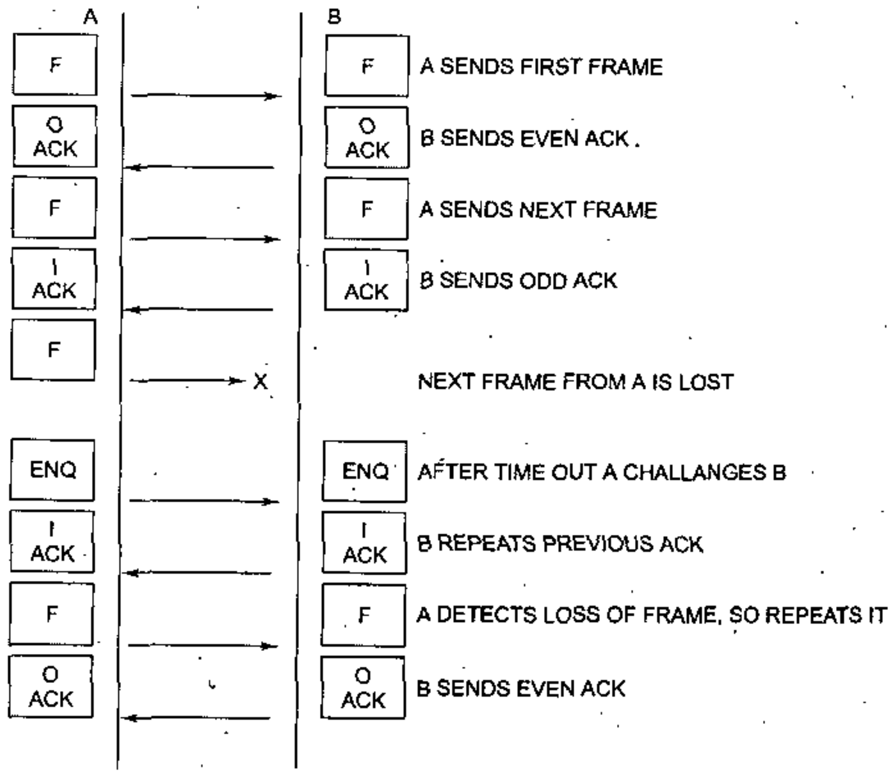

Fig. 3.32

3.36 ERROR CONTROL IN SLIDING WINDOW MECHANISM

In sliding window mechanism each frame is assigned a sequence number and, therefore, a more elaborate error control scheme is feasible. The receiver keeps track of the sequence numbers of the incoming data frames. If any out of sequence frame is received, immediately a request for retransmission of the missing frame is sent. There are two alternatives :

- If the receiver can sequence the frames on its own. It request retransmission only of the missing frame by sending a Selective Reject (SREJ-N) frame. N is the number of the missing frame. On receipt of SREJ-N, the sending end retransmits frame number N only. If the succeeding frames are received meanwhile, the receiver accepts them and arranges all the frames in proper sequence when frame N is also received.
- Alternatively, the receiver requests for retransmission of the missing frame and all the following frames by sending a Reject (REJ-N) signal. REJ-N indicates request for retransmission of the frames starting with the frame bearing sequence number N.

Both SREJ-N and REJ-N also acknowledge receipt of frames up to N-1. Although SREJ is more efficient, the receiver needs to be more complex having capability to put the frames in proper sequence. Figure 3.33 illustrates the error control mechanism.

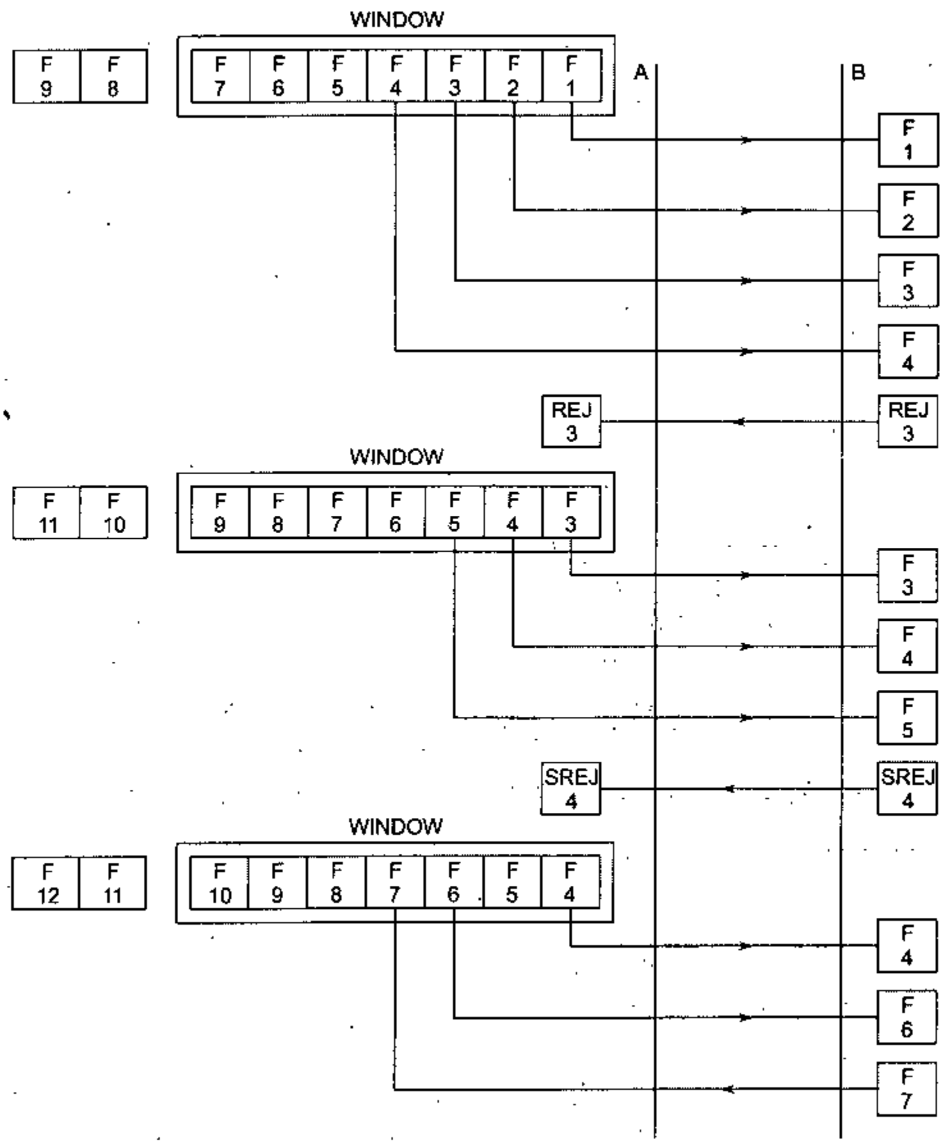

Fig. 3.33

3.37 DATA LINK MANAGEMENT

The data transfer process between two devices can be viewed as consisting of the following phases (Figure 3.34).

- Link establishment phase.
- Information transfer phase.
- Termination phase.

Link establishment includes processes required to initialise the data link, call/poll the other end, set mode of data transfer (synchronous/asynchronous) etc.

Information transfer phase involves exchange of data and acknowledgements.

Disconnection phase consists of those processes associated with relinquishing control of the link. Control is normally returned to the master/primary station.

Data link protocols define a set of control symbols and procedures to execute the above mentioned functions. These symbols and procedures are specific to a data link protocol and, therefore, cannot be generalised.

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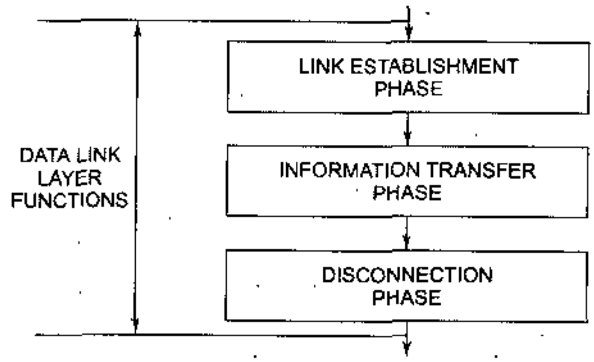

Fig. 3.34

SUMMARY

- Communications and Data Communications is one specific area of whole field of communication.
- All data communication codes are based on the binary system (1s and 0s). A message can be encoded into a meaningful string of 1s and 0s that can be transmitted along a data line and decoded by a receiver.
- All data communication codes are based on the binary system (1s and 0s). A message can be encoded into a meaningful string of 1s and 0s that can be transmitted along a data line and decoded by a receiver.
- The International Organization introduced the OSI layer for Standardization (ISO) in 1984 in order to provide a reference model to make sure products of different vendors would interoperate in networks.
- Data Encapsulation is the process of adding a header to wrap the data that flows down the OSI model. Each OSI layer may add it's own header to the data received from above.
- Layers of the OSI model deal with application issues and generally are implemented only in software. The highest layer, the application layer, is closest to the end user.
- Protocol is a formal set of rules and conventions that governs how computers exchange information over a network medium.
- Information being transferred from a software application in one computer system to a software application in another must pass through the OSI layers.
- OSI layer communicates with another layer to make use of the services provided by the second layer. The services provided by adjacent layers help a given OSI layer communicate with its peer layer in other computer systems.
- Application Layer provides network services directly to applications.
- Presentation Layer defines coding and conversion functions. It ensures that information sent from the application layer of one system is readable by the application layer of another system.
- The session layer establishes, manages, maintains and terminates communication channels between software programs on network nodes.
- It establishes, maintains and terminates virtual circuits. It provides error detection and recovery. It is concerned with reliable and unreliable transport.
- The physical topology of local area networks can take shape of a bus or ring or a star but the network attributes in terms of delay, throughput, expandability etc. are determined by the mechanisms utilized for sharing the use of physical interconnecting media. There are many methods sharing the media and they are, in general, called Media Access Control methods.

- The basic function-error control and other associated functions are carried out by the second layer of the OSI model, data link layer.
- *The data link layer constitutes the second layer of the hierarchical OSI model. It* incorporates certain data link control processes which carry out error control, flow control and the associated link management functions.
- Protocols are rules and procedures governing the interaction between two peer *(equal) entities.*
- Flow control mechanisms are incorporated to ensure that the data link layer at the sending end does not send more data frames than what the data link layer at receiving end is capable of handling.
- Sliding window flow control mechanism allows transmission of multiple frame without acknowledgements for individual frames.
- In sliding window mechanism each frame is assigned a sequence number and, therefore, a more elaborate error control scheme is feasible.

REVIEW QUESTIONS

1. What do you understand by data communications?
2. What are the different components in data communications?
3. Differentiate between Simplex Transmission, Half Duplex Transmission and Full-Duplex Transmission.
4. *What are the different transmission codes?*
5. Differentiate between Parallel Transmission and Serial Transmission.
6. What do you understand by OSI reference model?
7. Discuss the operations and protocol examples of
   (*a*) Application Layer
   (*b*) Presentation Layer
   (*c*) Session Layer
   (*d*) Transport Layer
   (*e*) Network Layer
   (*f*) Data Link Layer
   (*g*) Physical Layer.
8. What do you understand by LAN?
9. What are different LAN topologies?
10. Discuss media access control.
11. Discuss carrier sense multiple access (CSMA).
12. Write short notes on
    (*a*) Non-Persistent CSMA
    (*b*) 1-Persistent CSMA
    (*c*) P-Persistent CSMA
    (*d*) CSMA/CD
13. What do you understand by data link control and its need?
14. Discuss data link layer and its applications.
15. What are different data link protocols?
16. What do you understand by framing?
17. *Discuss sliding window flow control.*
18. Discuss error control in stop-and-wait mechanism.
19. Discuss error control in sliding window mechanism.

FURTHER READINGS

1. *Telecommunication and Information Technology,* Prashant Kaushik, Anmol, 2006.
2. *Optical Networking in Telecommunication,* S. Mukherjee, Jaico.
3. *Wireless Technology and Access of Information,* Ajay K. Srivastav, Shree Pub., 2006.
4. *Elements of Networking Engineering,* Kumar Prasun Ramakrishnan, Shree Pub., 2010.
5. *Trends in Networking and Communication,* Edited by Girish Kumar Srivastav and Charul Bhatnagar, Atlantic Pub., 2009.

NOTES

UNIT IV: TCP/IP, CONNECTIVITY AND INTERNETWORKING, ATM, SDH, AND ACCESS TECHNIQUES

★ STRUCTURE ★

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LEARNING OBJECTIVES

After going through this unit, you will be able to:

- define TCP/IP addressing
- know about concept of IP address
- describe LANs defined
- explain briefly about ATM
- describe virtual path/virtual connection or channel or circuit/transmission path
- know about VPI/VCI
- define historical overview
- know about importance of access network

PART I: TCP/IP, ADDRESSING

4.1 INTRODUCTION

In the mid-1990s, the Internet is a dramatically different network than when it was first established in the early 1980s. There is a direct relationship between the value of the Internet and the number of sites connected to the Internet. Over the past few years, the Internet has experienced two major scaling issues as it has struggled to provide continuous and uninterrupted growth. The eventual exhaustion of the IPv4 address space The ability to route traffic between the ever increasing number of networks that comprise the Internet. The first problem is concerned with the eventual depletion of the IP address space.

4.2 TCP

TCP is a connection-oriented transport protocol that sends data as an unstructured stream of bytes. By using sequence numbers and acknowledgment messages, TCP can provide a sending node with delivery information about packets transmitted to a destination node. Where data has been lost in transit from source to destination, TCP can retransmit the data until either a timeout condition is reached or until successful delivery has been achieved. TCP can also recognize duplicate

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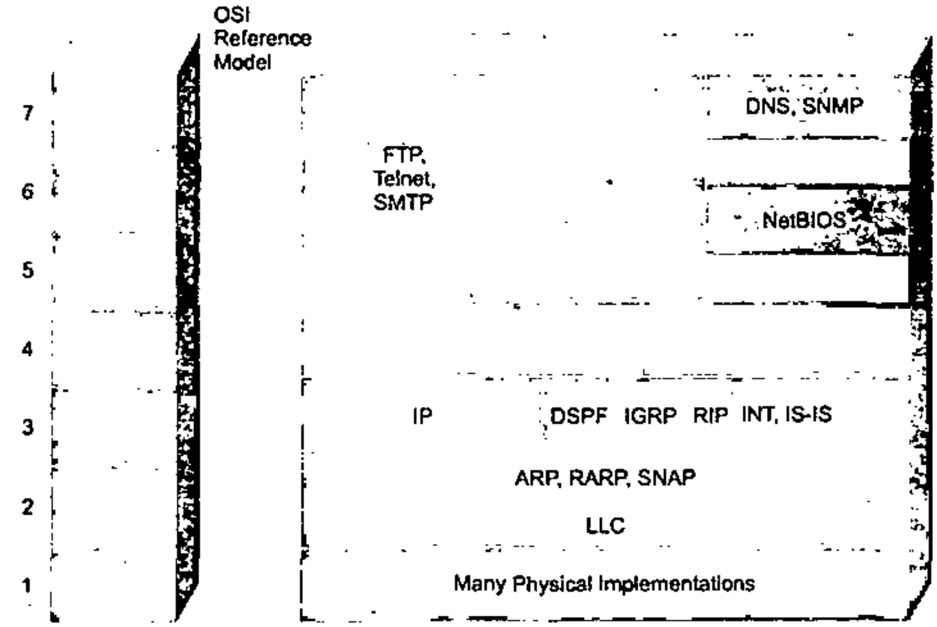
messages and will discard them appropriately. If the sending computer is transmitting too fast for the receiving computer, TCP can employ flow control mechanisms to slow data transfer. TCP can also communicate delivery information to the upper-layer protocols and applications it supports. Figure below shows the relationship of the Internet Protocol Suite to the OSI Reference Model



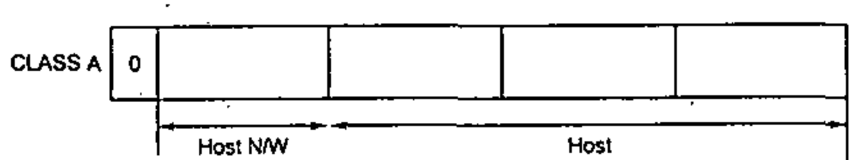
4.3 IP

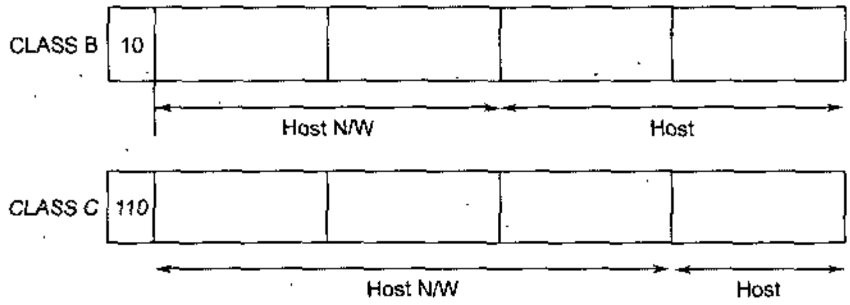
IP is the primary layer 3 protocol in the Internet suite. In addition to internetwork routing, IP provides error reporting and fragmentation and reassembly of information units called datagrams for transmission over networks with different maximum data unit sizes. IP represents the heart of the Internet protocol suite.

IP addresses are globally unique, 32-bit numbers assigned by the Network Information Center. Globally unique addresses permit IP networks anywhere in the world to communicate with each other.

An IP address is divided into three parts. The first part designates the network address, the second part designates the subnet address, and the third part designates the host address.

IP addressing supports three different network classes. Class A networks are intended mainly for use with a few very large networks, because they provide only 8 bits for the network address field. Class B networks allocate 16 bits, and Class C networks allocate 24 bits for the network address field. Class C networks only provide 8 bits for the host field, however, so the number of hosts per network may be a limiting factor. In all three cases, the leftmost bit(s) indicate the network class. IP addresses are written in dotted decimal format; for example, 34.0.0.1. Figure below shows the address formats for Class A, B, and C IP networks.





4.4 CONCEPT OF IP ADDRESS

The current version of IP, IP version 4 (IPv4), defines a 32-bit address which means that there are only 232 (4,294,967,296) IPv4 addresses available. This might seem like a large number of addresses, but as new markets open and a significant portion of the world's population becomes candidates for IP addresses, the finite number of IP addresses will eventually be exhausted. The address shortage problem is aggravated by the fact that portions of the IP address space have not been efficiently allocated. Also, the traditional model of classful addressing does not allow the address space to be used to its maximum potential.

In order to provide the flexibility required to support different size networks, the designers decided that the IP address space should be divided into three different address classes - Class A, Class B, and Class C. This is often referred to as "classful" addressing because the address space is split into three predefined classes, groupings, or categories. Each class fixes the boundary between the network-prefix and the host-number at a different point within the 32-bit address.

One of the fundamental features of classful IP addressing is that each address contains a self-encoding key that identifies the dividing point between the network-prefix and the host-number.

4.5 CLASS A NETWORKS (/8 PREFIXES)

Each Class A network address has an 8-bit network-prefix with the highest order bit set to 0 and a seven-bit network number, followed by a 24-bit host-number. Today, it is no longer considered 'modern' to refer to a Class A network. Class A networks are now referred to as "/8s" (pronounced "slash eight" or just "eights") since they have an 8-bit network-prefix. A maximum of 126 (27-2) /8 networks can be defined. The calculation requires that the 2 is subtracted because the /8 network 0.0.0.0 is reserved for use as the default route and the /8 network 127.0.0.0 (also written 127/8 or 127.0.0.0/8) has been reserved for the "loopback" function. Each /8 supports a maximum of 16,777,214 (224-2) hosts per network. The host calculation requires that 2 is subtracted because the all-0s ("this network") and all-1s ("broadcast") host-numbers may not be assigned to individual hosts.

4.6 CLASS B NETWORKS (/16 PREFIXES)

Each Class B network address has a 16-bit network-prefix with the two highest order bits set to 1-0 and a 14-bit network number, followed by a 16-bit host-number.

Class B networks are now referred to as"/16s". since they have a 16-bit network-prefix. A maximum of 16,384 (214) /16 networks can be defined with up to 65,534 (216 -2) hosts per network.

4.7 CLASS C NETWORKS (/24 PREFIXES)

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Each Class C network address has a 24-bit network-prefix with the three highest order bits set to 1-1-0 and a 21-bit network number, followed by an 8-bit host-number. Class C networks are now referred to as "/24s" since they have a 24-bit network-prefix. A maximum of 2,097,152 (221)/24 networks can be defined with up to 254 (28 -2) hosts per network.

4.8 DOTTED-DECIMAL NOTATION

To make Internet addresses easier for human users to read and write, IP addresses are often expressed as four decimal numbers, each separated by a dot. This format is called "dotted-decimal notation."Dotted-decimal notation divides the 32-bit Internet address into four 8-bit (byte) fields and specifies the value of each field independently as a decimal number with the fields separated by dots.

The classful A, B, and C octet boundaries were easy to understand and implement, but they did not foster the efficient allocation of a finite address space. A /24, which supports 254 hosts, is too small while a /16, which supports 65,534 hosts, is too large. In the past, the Internet has assigned sites with several hundred hosts a single /16 address instead of a couple of /24s addresses.

PART II: CONNECTIVITY AND INTERNETWORKING

4.9 LANs DEFINED

A *local area network* is a form of local (limited - distance) shared packet network for computer communications. LANs interconnect computers and peripherals over a common medium so users might share access to host computers, databases, files, applications, and peripherals. LANs conform to the *client/server* architecture, which essentially is a distributed computing architecture that takes advantage of the fact that both the client workstations and the servers are intelligent, programmable devices and exploits the capabilities of each. In such a network, *client* applications on microcomputers run against one or more centralized *servers*, which are high - performance multiport computers with substantial processing power and large amounts of memory. Some servers are positioned as devices that control the operational, administrative, and executive functions for the network, including authenticating legitimate users, granting access privileges to a database or perhaps a shared printer, and recording usage data. Some servers are positioned as database engines, that is, application or data repositories, capable of processing client requests for information and managing the resident data. *Note:* LANs also support peer-to-peer communications between clients and between servers.

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Generally, LAN specifications are the province of the Institute of Electrical and Electronics Engineers (IEEE), although the American National Standards Institute (ANSI) and other standards bodies are involved, and the regulators are very much involved in spectrum allocation in the Wireless LAN (WLAN) domain. LANs operate at Layer 1, the Physical Layer, and Layer 2, the Data Link Layer, of the Open Systems Interconnection (OSI) Reference Model. Raw bandwidth ranges up to 10 Gbps, although actual throughput often is much less. LANs are limited to a maximum distance of only a few miles or kilometers, although they often operate within a much more confined area measured in feet or meters. LANs support the transmission of data in frame format, with the frames varying in size within specified minimums and maximums.

LANs are used almost exclusively for data communications over relatively short distances such as within an office, office building, or campus environment. LANs enable multiple workstations to share access to multiple host computers, other workstations, applications and databases, printers and other peripherals, and WAN connections. LANs traditionally are used in computer data applications, although they increasingly support video and voice communications as well.

4.10 LAN DIMENSIONS

LANs can be characterized along a number of common dimensions, for ease of understanding. Those dimensions include transmission medium, physical and logical topology, baseband versus broadband, and medium access control method.

Transmission Media

Although coaxial cable was the original medium, fiber-optic cable has superseded coax in the LAN backbone. Unshielded Twisted Pair (UTP) replaced coax to the desktop beginning in the early 1990s. Radio Frequency (RF) wireless technologies more recently have become extremely popular, particularly in providing the final link to portable and mobile computers. While Wireless LANs (WLANs) generally are limited to special radio technologies, InfraRed (IR) technology is used in certain applications, and microwave and IR systems connect LANs and LAN segments in a campus environment. Satellite rarely is used in any way because propagation delay renders it unsatisfactory for interactive communications. Satellite links also defy the notion of a *local* area network, although they sometimes are used to link LANs and LAN segments in remote areas.

1. *Coaxial Cable:* Coaxial cable was the transmission medium first employed in LANs. Although coax is expensive to acquire and to configure and reconfigure, its performance characteristics are excellent. Additionally, Data Processing/Management Information Systems (DP/MIS) managers traditionally were comfortable with coax, which routinely was specified in the mainframe and midrange computer world. In fact, the technology did not exist until fairly recently to make effective use of other options such as twisted pair, fiber optics, and radio systems. In retrospect, perhaps the use of coaxial cable lessened the resistance of DP/MIS managers to the concept of LANs. Those who lived in the mainframe world (most did) regarded PCs with disdain and sneered at twisted pair, which they referred to as *telephone wire*. The advantages of coaxial cable include high bandwidth and exceptional error performance over relatively long distances as the thick inner core

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conductor results in fairly modest signal attenuation. Further, the outer shield rejects ElectroMagnetic Interference (EMI) and Radio Frequency Interference (RFI) as well as providing excellent security. Coax is also highly durable, but the costs of acquisition, deployment, and reconfiguration are high. While the disadvantages of coaxial cable have been mitigated to a large extent through the development of new coax designs, those designs also affect system performance. By way of example, consider three variations on the coax theme: ThickNet, ThinNet, and Twinax.

- **ThickNet:** *Thick* Ether *net*, also known as *10Base5*, was approved by the IEEE in 1983. 10Base5 uses traditional thick coax, often referred to as *goldenrod*, referring to its high cost, high value, and the yellow cable sheath used by some manufacturers. Other manufacturers used orange cable sheaths for thick coax, giving rise to the term *orange hose*. *10Base5* translates to 10 Mbps, *Base* band (one transmission at a time over a single, shared channel), and 500 m maximum segment length. While individual devices can be separated by much greater distances across the network, issues of signal attenuation limit each segment, or link, in the network to approximately 500 m.
- **ThinNet:** *Thin* Ether *net*, also known as *10Base2*, was approved by the IEEE in 1986. 10Base2 uses coax of thinner gauge. The thinner cable is less costly to acquire and deploy, although its performance is less in terms of transmission distance. *10Base2* translates to *10* Mbps, *Base* band, and 200 m maximum segment length (actually 185 m, rounded up).
- **Twinax:** *Twinaxial* cable, resembles ThinNet coax, but with *twin coax*ial conductors, rather than one. Twinax is used in older IBM midrange systems such as Systems 34, 36, and 38 as well as the younger IBM AS/400 and RS/6000. More recently, the IEEE has developed the *10GBase-CX4* standard in support of *10-Gigabit Ethernet* (10GbE). Based on the *Infiniband* high-speed cable assemblies, the specification calls for twinax assemblies operating over distances up to 50 ft. The standard calls for four transmitters and four receivers operating differentially in simplex mode over a bundle of eight twinax cables, with each simplex transmission occurring at 2.5 Gbps at a frequency of 3.125 GHz per channel with *8B/10B* line coding. The cost of this patch cord technology is expected to be approximately $1/10^{th}$ that of comparable 10GBase-optical solutions.

2. *Twisted Pair:* Since the early 1990s, *unshielded twisted pair* has become very popular as a LAN medium. Although its performance characteristics are less appealing than coax, its low cost and high availability certainly are very attractive. UTP of various categories performs very nicely at signaling speeds from 10 Mbps up to 1 Gbps over relatively short distances. The advantages of UTP include its low costs of acquisition, deployment, and reconfiguration. The disadvantages of UTP include its relatively low bandwidth and poor error performance over long distances. Because the carrier frequency must be high to support a data rate of 10/100 Mbps or more and as high-frequency signals attenuate relatively quickly, error performance suffers considerably over a distance. Therefore, distances are severely restricted. Additionally, the radiated electromagnetic field is considerable at the high frequencies required to support high speeds, which poses security concerns. Security at this level, however, generally is not considered to be a significant issue, as the cabling

system is restricted to the premises. More importantly, the radiated electromagnetic field can create noise that affects signals traveling on adjacent pairs in the same cable and in nearby cables. At 10 Mbps, however, Category 5 (Cat 5) cable commonly is used in a *structured wiring plan* to support both voice and data, with two pairs typically pulled to each duplex jack—one pair for voice and one for data. The disadvantages of UTP have been mitigated to some extent, and the LAN applications have increased through the development and use of Cat 3, 4, and 5 UTP. Since Cat 5 is by far the most capable of these standard options, it currently is the inside wire default for both voice and data. Category 6 is now enjoying application in high-speed LANs and the Cat 7 specification is under development. The following discussions of 1Base5, 10Base – T, 100Base – T, 1000Base – T, and 10GBase – T serve to illustrate the evolution of twisted - pair applications in the LAN domain:

NOTES

- *1Base5* (IEEE, mid - 1980s) translates to 1 Mbps, *Base* band, and 500 m maximum segment length and was the predecessor to 10Base - T. AT & T spearheaded the 1Base5 initiative in support of its StarLAN product. 1Base5 runs over Cat 3, 4, or 5 UTP. 1Base5 is considered obsolete.
- *10Base - T* translates to 10 Mbps, *Base* band over *T* wisted pair (IEEE, 1990) and refers to Ethernet running over Cat 3, 4, or 5 UTP. The maximum segment length between the 10Base - T hub and the attached device (*e.g.*, workstation or printer) is specified at 100 m or less, although good Cat 5 cable will perform well over somewhat longer distances. The 10Base - T hub is a wire hub that serves as a multiport repeater as well as a central point of interconnection.
- *100Base - T* (IEEE, 1995) is similar to 10Base - T Ethernet hub technology, running at 100 Mbps and requiring Cat 5 UTP or better. Distances originally were limited to 100 m over Cat 5 cable but now extend to 350 m over Cat 5e.
- *1000Base - T* (IEEE 802.3ab, 1999) is similar in concept to the predecessor 10/100Base - T. The original specifications called for Cat 5 cable to support Gigabit Ethernet (GbE) over four pairs and distances up to 100 m. Category 6 cabling specifications include UTP, *Shielded Twisted Pair* (STP), and *Screened Twisted Pair* (ScTP) rated at 250 MHz over distances up to 220 m.
- *10GBase - T* (IEEE 802.3an, June 2006) is a specification for 10GbE over Cat 6 cable for distances up to at least 55 m, although distances generally can be extended to 100 m. The expectation is that Cat 7 cable will extend those distances even further. Cat 7 is STP with a combination foil and braided screen construction. Cat 7 supports signaling rates up to 600 MHz, although the usable spectrum can be up to 750 MHz.

Category 3 (Cat 3) UTP also is used for 4 - Mbps Token Ring LANs. Category 4 (Cat 4) UTP, developed for 16 - Mbps Token Ring LANs, has a bandwidth of 20 MHz. In addition to its application in Cat 6 and 7 cables, STP sometimes is used in high - noise environments in which UTP data transmission is especially susceptible to EMI or RFI. Examples include manufacturing environments where there are large numbers of powerful machines, power plants, and old buildings (*e.g.*, hospitals, or government or military facilities) where it might be impossible when installing a LAN to avoid placing wires close to electric motors or older fluorescent light fixtures.

NOTES

3. ***Fiber-Optic Cable***: Because of its outstanding performance characteristics, optical fiber also is used extensively in contemporary LAN applications. Its cost and fragility, however, generally relegate it to use as a backbone technology in *Fiber Distributed Data Interface* (FDDI) networks, for example. The advantages of fiber certainly include the combination of high bandwidth and excellent error performance. Additionally, fiber performs well over long distances and offers excellent security. The disadvantages of fiber transmission systems include their high cost of acquisition, as compared to UTP systems. While the fiber, itself, is not significantly more expensive than Cat 5e UTP, the light sources and detectors are considerably more expensive than the metallic interfaces used with UTP. As fiber is very fragile, it must be protected carefully, and redundancy is always a good idea.
4. ***Wireless***: Wireless LANs (WLANs) offer the obvious advantage of avoiding much of the time and cost associated with deploying wires and cables. This is especially important in a dynamic environment where portability is desirable, such as an office where cubicles are frequently reconfigured. WLANs have found acceptance in providing LAN capabilities in temporary quarters, where costly cabling soon would have to be abandoned, and in older buildings, where wires are difficult or impossible to run. WLAN technologies include both RF and IR. The most common approach is that of RF, which involves fitting each device with a low - power transmit/receive radio antenna, which traditionally is in the form of a PC card. Newer laptop, tablet, and hand - held computers boast built - in antennas. Frequency assignments for commercial applications generally are in the 2.4 - and 5 - GHz bands. The physical configuration involves a hub antenna located at a central point (see Figure 14.1), such as the center or the corner of the ceiling, where Line - Of - Sight (LOS) or near - LOS connectivity can be established with the various terminal antennas. While LOS is not strictly required at these frequencies, it is always desirable and is particularly important at higher frequencies, which suffer greater attenuation from physical obstructions. The hub antenna then connects to the servers, peripherals, and other hosts via cabled connections, which also connect together multiple hub antennas for transmission between rooms, floors, buildings, and so on. In order to serve multiple workstations, *spread - spectrum* radio technology often is employed to maximize the effective use of limited bandwidth.

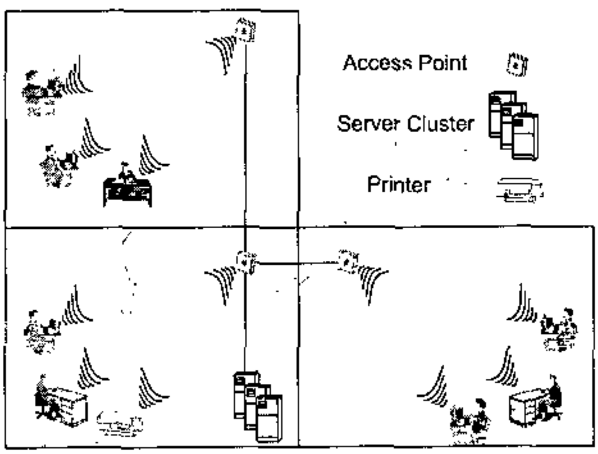

Fig. 4.1

PART III: ATM

4.11 ATM

NOTES

ATM stands for Asynchronous Transfer Mode.Here

Mode means specific method or way.

Transfer means transmission and switching aspects.

Asynchronous means information packets will be transferred based an irregular or random occurrence pattern as they are filled according to the demand. Hence "ATM is a method of transmission & switching of information in the form of packets which may occur an irregular occurrence pattern as they are filled according to the demand of the user".

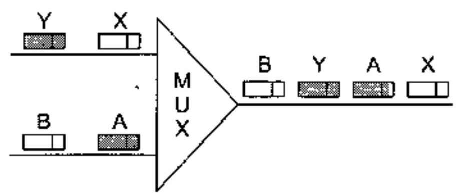

FIG. 4.2 Synchronous transmission mode

In the above Fig. 4.2, even though the Cell X and B are empty, they will also be Multiplexed and sent on the output side. By this, the bandwidth is not used effectively.

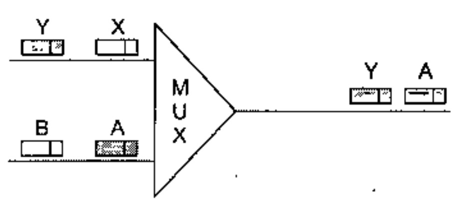

FIG. 4.3 Synchronous transmission mode

In the above Figure 4.3, the empty Cells X and B are not at all transferred towards output side. By this, the output bandwidth is effectively used.

Advantages

Hence ATM is a standardized technology that enables the convergence of a variety of services such as:

1. Low bandwidth and Very high bandwidth.
2. Synchronous and Asynchronous.
3. Voice, Video and Data.
4. Constant Bit Rate (CBR) and Variable Bit Rate (VBR).
5. Real-Time (RT) and Non-Real-Time (NRT).
6. Slotted and Pocketsize.
7. Switched and Non Switched.
8. In addition, ATM is an independent of Transmission medium, which means the medium can be Wire (Twisted Pair/Copper Pair/Co-axial/ Fiber) or Wireless.

ATM technology allows a variety of bit rates to be transported, with which sophisticated bandwidth management enables the network to be more efficient and at the same time, maintain a QoS (Quality of Service) that is custom suited to each other.

NOTES

4.12 ATM PROTOCOL

ATM is the protocol designed by ATM Forum and adopted by the ITU-T. ATM can be thought of as the "Highway" of the information super highway. So ATM can do every thing that N-ISDN can do but with better quality.

In ATM System, the packet size is fixed to 53 octets known as a CELL. Any type of traffic viz Voice, Data, Video, Synchronous or Asynchronous, Short or Long packets can be converted into ATM Cells by a process known as emulation. So ATM can also be called as Cell relaying technology or Cell switching technology. Primary rate of transmission in ATM is 155.52 Mbps.

Cell Switching

Switching means creating a temporary connection between two or more devices linked to the switch, Hardware and/or Software devices. Traditionally, 3 methods of switching have been important called Circuit Switching, Packet Switching and Message Switching.

Circuit switching

Circuit switching create a direct physical connection between two devices such as phones or computers. As in Figure 4.4, devices A & G are connected by the switches 1,2 and 4 *via path I and III. Circuit switching is mostly used at the physical layer* of OSI Model

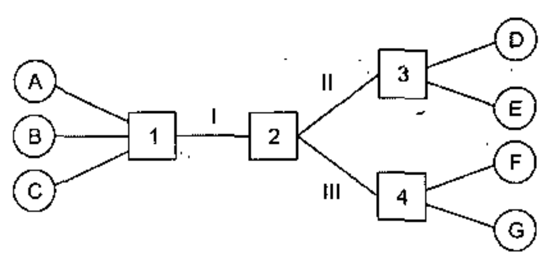

FIG. 4.4

Packet Switching

For Data communication Packet switching technology was designed. User data are packetized and sent packet by packet using the path in shared manner. Two different approaches are available under packet switching. One is called Datagram approach and second is called Virtual circuit approach. The later is used in ATM. The identifier that is actually used for data transfer in Virtual circuit approach is called the Virtual circuit identifier. A VCI is a smaller number that only has switch scope. It is used by a frame. When a frame arrives at a switch, it has one VCI. When it leaves, it has another VCI. Fig. 4.5 shows how the VCI in a data frame changes from one switch to another

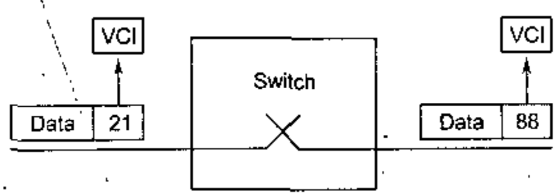

Fig. 4.5

4.13 ATM INTERFACES

ATM has 2 interfaces namely

1. User to Network Interface (UNI)
    1. Private UNI
    2. Public UNI
2. Network to Network Interface (NNI)

UNI is used between user and network where as NNI is used between networks.

4.14 ATM CONNECTIONS

ATM or B-ISDN offers 2 types of connections called PVC & SVC and ATM services are connection oriented.

Permanent Virtual Connection (PVC)

A source and a destination may choose to have a dedicated virtual circuit. In this case, the corresponding table entry is recorded for all switches by the system administrator. An outgoing VCI is given to the source and an incoming VCI is given to the destination. The source always uses this VCI to send frames to that particular destination. The destination knows that the frame is coming from that particular source if the frame carries the corresponding incoming VCI. In a simple word, PVC is like a Hotline/P Wire/ Point to Point/ Leased line and the nature is static. Fig. 4.6 shows the PVC setup.

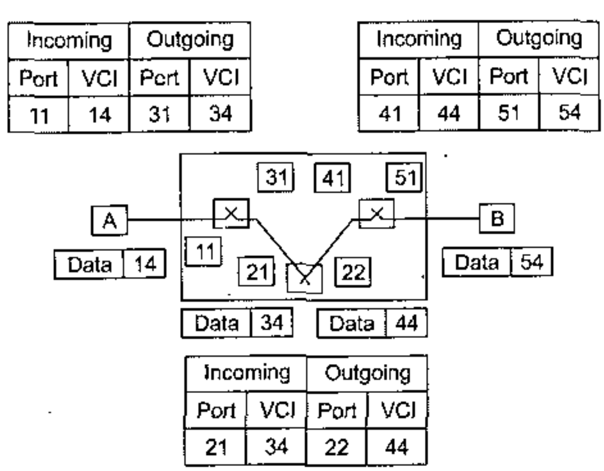

Fig. 4.6

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Switched Virtual Circuit (SVC)

If a source needs connection with several destinations or any other destination, it needs a PVC for each destination which is costly. An alternative approach is the SVC. So SVC creates a temporary, short duration connection which exists only whenever data are being transferred by the end users. In other words, this is dynamic in nature. This approach requires a series of action called connection setup, setup acknowledgement, data transfer and tear down phases. ATM supports both types of connections.

4.15 ATM NETWORK ARCHITECTURE

ATM network consists of access devices called the end points, available at user end, are connected through a interface called UNI to the ATM switch. Another ATM switch of the network is connected through an interface called NNI. The architecture is shown in the Fig. 5.7.

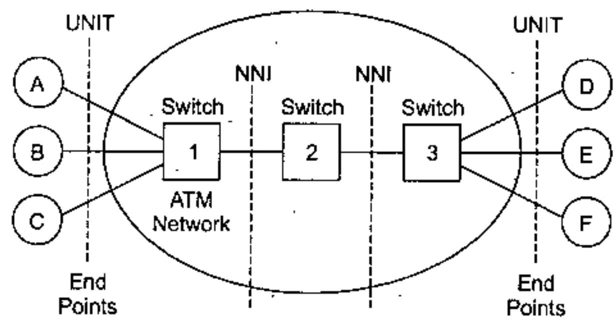

Fig. 4.7

4.16 VIRTUAL PATH/VIRTUAL CONNECTION OR CHANNEL OR CIRCUIT/TRANSMISSION PATH

Connection between two end points is accomplished through transmission path (TP), virtual path (VP) and virtual circuit (VC). A transmission path (TP) is the physical connection (wire/wireless) between an end point and a switch or between two switches. A TP is divided into several virtual paths (VPs). A virtual path provides a connection or set of connections between two switches. Within a VP, many circuits called virtual circuits (VCs) will be available which is used for connection. Cell networks are based on virtual circuits. All cells belonging to single message follow the same VC and remain in their original order until they reach their destination. TP, VP and VC are shown in Figure-4.8.

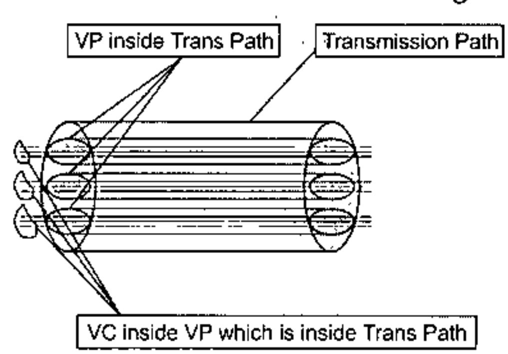

Fig. 4.8

4.17 VPI/VCI

In a virtual circuit network, to route data from one end point to another, the virtual connection need to be identified. For this purpose, the designer of ATM, created a hierarchical identifier with 2 levels called virtual path identifier (VPI) and virtual circuit or channel identifier (VCI). The VPI defines the specific VP and the VCI defines a particular VC inside the VP. Both the connection identifier are shown in Figure 4.9.

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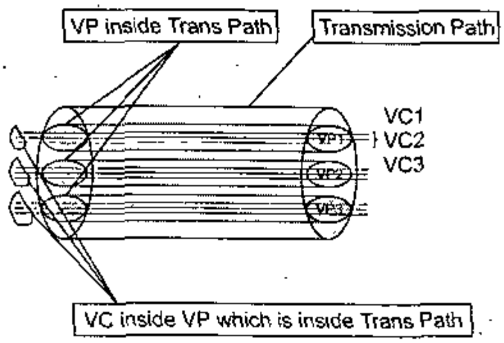

Fig. 4.9

VP Switch/VC Switch

Most of the switches (Core switch) within typical ATM network are routed using VPI (VP switch). (*i.e.*,) The switching can be taken place by changing the VPI but keeping VCI within VPI intact. Such switches are called VP switch. If switching can be taken place by changing both the VPI and VCI, then such switches are called VC switch. The switches at end points (Edge switch) of the ATM network use both VPIs and VCIs (VC switch). Both switches are shown in Figure 4.10

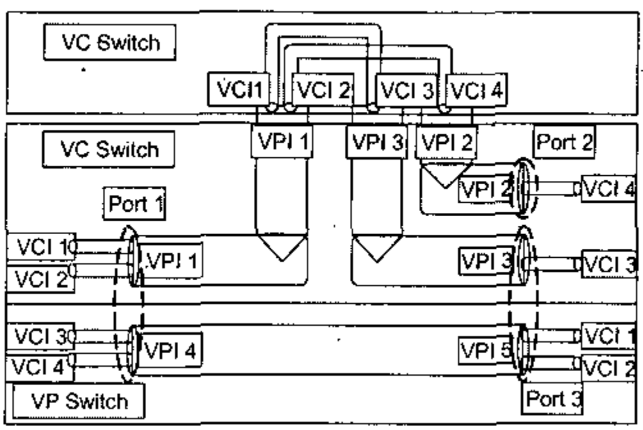

Fig. 4.10

ATM Transmission Rates

At present, rate of transmission is 155 Mbps called primary rate. Higher order is also possible in multiple of 4 times.

4.18 ATM CELL FORMAT

ATM Cell consists of 2 fields called Header Field and Information Field as in Figure 4.11

HEADER FIELD OCTETS	INFORMATION FIELD 48 OCTETS

FIG. 4.11

Header Field

Header field is different for UNI and NNI in the ATM network

NOTES

	BIT 1		BIT 8
OCTET 1	GFC	VPI	
5 OCTETS	VPI	VCI	
UNI	VCI		
FORMAT	VCI	PTI	CLP
OCTET 5	HEC		

GFC => GENERIC FLOW CONTROL (4 BITS)
VPI => VIRTUAL PATH IDENTIFIER (8 BITS)
VCI => VIRTUAL CHANNEL IDENTIFIER (16 BITS)
PTI => PAYLOAD TYPE IDENTIFIER (3 BITS)
CLP => CELL LOSS PRIORITY (1 BIT)
HEC => HEADER ERROR CONTROL (8 BITS)

GFC (Generic Flow Control - 4 bits)

It is used to assist the customer network in the cell flow control, but not carried through the network.

VPI/VCI (Virtual Path Identifier-8 bits/Virtual Channel Identifier-16 bits)

This label identifies a particular virtual path and virtual channel or circuit on a transmission link. The switching nodes use this information and along with the routing information established at connecting setup, routes the cells to the appropriate output ports. The switching nodes changes the input value of VPI/VCI fields to new output values. Since VPI field is 8 bits (at UNI) and VCI has 16 bits field, a host can have theoretically 256 bundles, each containing up to 65,536 circuits.

8 VPI bits provide $2^8 = 256$ bundles

16 VCI bits provide $2^{16} = 65{,}536$ circuits

CLP (Cell Loss Priority-1 bit)

Having one of the two values '0' or '1', the CLP indicates priority of a cell when the network element has to make the decision to drop the cell when its throughput bandwidth exceeds its transfer rate. In congestion situations, cells with CLP =1 may be dropped and not transferred at all.

PTI (Payload Type Identifier-3 bits)

It identifies the payload type *i.e.*, whether the cell payload contains user data or network information and also provides congestion identification.

HEC (Header Error Control-8 bits)

HEC code detects and corrects a single bit error or detects multi bit errors in the header field. It is based on CRC-8 with the devisor polynomial as X^8+X^2+X+1.

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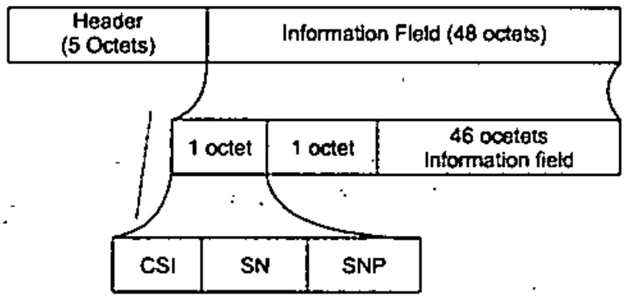
Information Field (48 Octets)



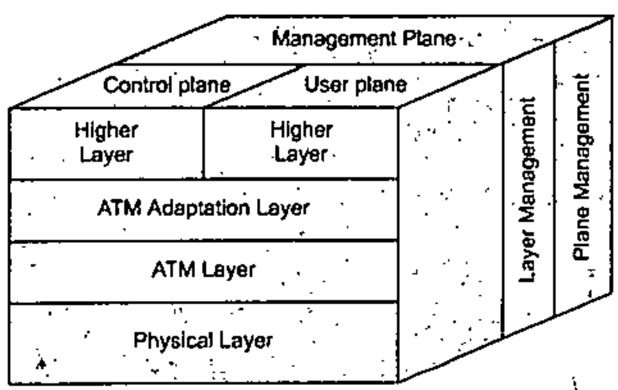
CSI => Convergence Sub Layer Indicator (1 bit)
SN => Sequence Number (3 bits)
SNP => Sequence Number Protection (3 bits)

The Information Field does not contain all the 48 octets of user data. One or two octets are dedicated for administration and call sequence purpose. The first octet (after the overhead bits or Header octets) consists of three sub fields. The first bit is known as the convergence sub layer indicator (CSI). It is used to indicate whether the pointer is used or not. The next three bits are sequential number (SN) from 000 to 111 used to detect the type of cells. The next three bits are the Sequence Number Protection (SNP). It performs error detection on the CSI and SN sub fields. One bit is not used at present. The second octet is optional and is used as a pointer to mark the start of long encapsulated messages. 48 octets information field is only scrambled.

4.19 ATM REFERENCE MODEL

ATM functionality is organized in a stack of layers; each layer assigned a specific function. It consists of three planes called

1. User Plane
2. Control Plane
3. Management Plane



Management Plane

All the management functions that relate to whole system are located in the management plane, which is responsible for providing coordination between all planes.

Two types of functions (i) Layer Management (ii) Plane Management.

Layer Management

1. Management functions relating to resources and parameters residing in its protocol entities.
2. Handles specific OAM information flow for each layer.

NOTES

Plane Management

Management of all the planes for its proper functions.

Control Plane

1. Responsible for the call control and connection control functions.
2. These are all signaling functions for setup, supervise and release a call or connection.

User Plane

Deals with transport of user information, flow control and recovery from errors.

PART IV: SDH CONCEPTS AND PRINCIPLE

4.20 SDH CONCEPTS AND PRINCIPLE

It is an international standard networking principle and a multiplexing method. The name of hierarchy has been taken from the multiplexing method which is synchronous by nature. The evolution of this system will assist in improving the economy of operability and reliability of a digital network.

4.21 HISTORICAL OVERVIEW

In February 1988, an agreement was reached at CCITT (now ITU-TS) study group XVIII in Seoul, on set of recommendations, for a synchronous digital hierarchy representing a single world wide standard for transporting the digital signal. These recommendations G-707, G-708, G-709 cover the functional characteristic of the network node interface, *i.e.*, the bit rates and format of the signal passing over the Network Node Interface (NNI). For smooth transformation from existing PDH, it has to accommodate the three different country standards of PDH developed over a time period. The first attempt to formulate standards for Optical Transmission started in U.S.A. as SONET (Synchronous Optical Network). The aim of these standards was to simplify interconnection between network operators by allowing inter-connection of equipment from different vendors to the extent that compatibility could be achieved. It was achieved by SDH in 1990, when the CCITT accepted the recommendations for physical layer network interface. The SONET hierarchy from 52 Mbit per second rate onwards was accepted for SDH hierarchy.

4.22 MERITS OF SDH

(*i*) Simplified multiplexing/demultiplexing techniques.

(*ii*) Direct access to lower speed tributaries, without need to multiplex/demultiplex the entire high speed signal.

(iii) Enhanced operations, Administration, Maintenance and provisioning capabilities.

(iv) Easy growth to higher bit rates in step with evolution of transmission technology.

(v) Capable of transporting existing PDH signals.

(vi) Capable of transporting future broadband (ATM) channel bit rates.

(vii) Capable of operating in a multi-vendor and multi-operator environment.

NOTES

Advantages

(i) *Multi-vendor environment (mid span meet):* Prior to 1988 international agreement on SDH all vendors used proprietary non-standard techniques for transporting information on fibre. The only way to interconnect was to convert to the copper transmission standards (G702/703/704). The cost and complexity levels were very high.

(ii) *Synchronous networking*: SDH supports multi-point or hub configurations whereas, asynchronous networking only supports point-to-point configurations.

(iii) *Enhanced OAM&P*: The telecoms need the ability to administer, surveil, provision, and control the network from a central location.

(iv) *Positioning the network for transport on new services*: LAN to LAN, HDTV, interactive multimedia, video conferencing.

(v) *HUB*: A hub is an intermediate site from which traffic is distributed to 3 or more spur. It allows the nodes to communicate as an angle network, thus reducing the back-to-back multiplexing and demultiplexing.

4.23 S.D.H. EVOLUTION

S.D.H. evolution is possible because of the following factors :

(i) **Fibre Optic Bandwidth:** The bandwidth in Optical Fibre can be increased and there is no limit for it. This gives a great advantage for using SDH.

(ii) **Technical Sophistication:** Although, SDH circuitary is highly complicated, it is possible to have such circuitary because of VLSI technique which is also very cost effective.

(iii) **Intelligence:** The availability of cheaper memory opens new possibilities.

(iv) **Customer Service Needs:** The requirement of the customer with respect to different bandwidth requirements could be easily met without much additional equipment. The different services it supports are :

1. Low/High speed data.
2. Voice
3. Interconnection of LAN
4. Computer links
5. Feature services like H.D.T.V.
6. Broadband ISDN transport (ATM transport)

4.24 S.D.H. STANDARDS

The S.D.H. standards exploit one common characteristic of all PDH networks namely 125 micro seconds duration, *i.e.*, sampling rate of audio signals (time for 1

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byte in 64 k bit per second). This is the time for one frame of SDH. The frame structure of the SDH is represented using matrix of rows in byte units. As the speed increases, the number of bits increases and the single line is insufficient to show the information on Frame structure. Therefore, this representation method is adopted. How the bits are transmitted on the line is indicated on the top. The frame structure contains 9 rows and number of columns depending upon synchronous transfer mode level (STM). In STM-1, there are 9 rows and 270 columns. The reason for 9 rows arranged in every 125 micro seconds is as follows :

For 1.544 Mbit PDH signal (North America and Japan Standard), there are 25 bytes in 125 micro second and for 2.048 Mbit per second signal, there are 32 bytes in 125 micro second. Taking some additional bytes for supervisory purposes, 27 bytes can be allotted for holding 1.544 Mbit per second signal, *i.e.*, 9 rows x 3 columns. Similarly, for 2.048 Mbit per second signal, 36 bytes are allotted in 125 micro seconds, *i.e.*, 9 rows x 4 columns. Therefore, it could be said 9 rows are matched to both hierarchies.

A typical STM-1 frame is shown in Figure. Earlier this was the basic rate but at present STM-0 which is just $1/3^{rd}$ of STM-1, *i.e.*, 51.840 Mbit per second has been accepted by CCITT. In STM-1 as in the first 9 rows and 9 columns accommodate Section Overhead (SOH) and 9 rows x 261 columns accommodates the main information called pay load. The interface speed of the STM-1 can be calculated as follows :

$$(270 \text{ columns} \times 9 \text{ rows} \times 8 \text{ bits} \times 1/125 \text{ ms}) = 155.52 \text{ Mbps}.$$

The STM-0 contains just $1/3^{rd}$ of the STM-1, *i.e.*, 9 rows × 90 columns out of that 9 rows x 3 columns consist of section overhead and 9 rows × 87 columns consist of pay load. The STM-0 structure was accepted so that the radio and satellite can use this bit rate, *i.e.*, 51.840 Mbit/s across their section.

The different SDH level as per G-707 recommendations is as given in Figure 4.

4.25 PRINCIPLES OF SDH

- SDH defines a number of "Containers", each corresponding to an existing plesiochronous rate.
- Each container has a "Path Overhead" added to it
  - POH provides network management capability.
- Container plus POH form a "Virtual Container".
- All equipment is synchronised to a national clock.
- Delays associated with a transmission link may vary slightly with time-causing location of VC within the STM-1 frame to move.
- Variations accommodated by use of a Pointer
  - points to beginning of VC.
  - pointer may be incremented or decremented.
- G.709 defines different combinations of VCs which can be accommodated in the "payload" of an STM-1 frame.
- When STM-1 payload is full, more network management capability is added to form the "Section Overhead".
- SOH remains with payload for the fibre section between synchronous multiplexers.
- SOH bytes provide communication channels to cater for :

– OA&M facilities.
– user channels.
– protection switching.
– section performance
– *frame alignment*
– other functions.

NOTES

4.26 BASIC DEFINITIONS

(*i*) **Synchronous Transport Module.** This is the information structure used to support information pay load and over head information field organised in a block frame structure which repeats every 125 micro seconds.

(*ii*) **Container.** The first entry point of the PDH signal is the container in which the signal is prepared so that it can enter into the next stage, *i.e.*, virtual container. In container (container-I) the signal speed is increased from 32 bytes to 34 bytes in the case of 2.048 Mbit/s signal. The additional bytes added are fixed stuff bytes (R), Justification Control Bytes (CC and C′), Justification opportunity bytes (s).

In container-3, 34.368 Mbit/s signal (*i.e.*, 534 bytes in 125 m seconds) is increased to 756 bytes in 125 m seconds adding fixed stuff bits(R). Justification control bits (C-1, C-2) and Justification opportunity bits (S-1, S-2).

Detail follows : 756 bytes are in 9 × 84 bytes/125 m seconds frame. They are further subdivided into 3 sub frames 3 × 84 (252 bytes or 2016 bits). Out of this

- 1431 information bits (I),
- 10 bits (two sets) (C-1, C-2)
- 2 Justification opportunity bits (S-1, S-2)
- 573 (fixed bits)

In container-4, 139.264 Mbit/s signal (2176 bytes in 125 m seconds) is increased to 9 × 260 bytes. Details as follows : 9 × 260 bytes are partitioned into 20 blocks consisting of 13 bytes each. In each row one justification opportunity bit(s) and five justification control bit(s) are provided. The first byte of each block consists of either eight information bit (I) or eight fixed stuff bits (R) or one justification control bit (C) plus five fixed stuff bits (R) plus two overhead bits (o) or six information bits (I) plus one justification opportunity bit (s) plus one fixed stuff bit (R). The last 12 bytes of one block consists of information bits (I).

(*iii*) **Virtual Container.** In Virtual container the path over head (POH) fields are organised in a block frame structure either 125 m seconds or 500 m seconds. The POH information consists of only 1 byte in VC-1 for 125 m seconds frame. *In VC-3, POH is 1 column of 9 bytes.* In VC-4 also POH 1 column of 9 bytes. The types of virtual container identified are lower orders VCs VC-1 and VC-2 and higher order VC-3 and VC-4.

(*iv*) **Tributary Unit.** A tributary unit is a information structure which provides adaptation between the lower order path layer and the higher order path layer. It consists of a information pay load (lower order virtual container) and a tributary unit pointer which indicates the offset of the pay load frame start relating to the higher order VC frame start. Tributary unit 1 for VC-1

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and Tributary unit 2 is for VC-2 and Tributary unit 3 is for VC-3, when it is mapped for VC-4 through tributary group-3. TU-3 pointer consists of 3 bytes out of 9 bytes. Three bytes are H1, H2, H3 and remaining bytes are fixed bytes. TU-1 pointers are one byte interleaved in the TUG-2.

(*v*) **Tributary Unit Group.** One or more tributaries are contained in tributary unit group. A TUG-2 consist of homogeneous assembly of identical TU-1s or TU-2. TUG-3 consists of a homogeneous assembly of TUG-2s or TU-3. TUG-2 consists of 3 TU-12s (For 2.048 Mbit/sec). TUG-3 consists of either 7 TUG-2 or one TU-3.

(*vi*) **Network Node Interface (NNI).** The interface at a network node which is used to interconnect with another network node.

(*vii*) **Pointer.** An indicator whose value defines frame offset of a VC with respect to the frame reference of transport entity, on which it is supported.

(*viii*) **Administrative Unit.** It is the information structure which provides adaptation between the higher order path layer and the multiplex section layer. It consists of information pay load and a A.U. pointer which indicates the offset of the pay load frame start relating to the multiplex section frame start. Two AUs are defined (i) AU-4 consisting VC-4 plus an A.U. pointer indicating phase alignment of VC-4 with respect to STM-N frame, (ii) AU-3 consisting of VC-3 plus A.U. pointer indicating phase alignment of VC-3 with respect to STM-N frame. A.U. location is fixed with respect to STM-N frame.

(*ix*) **Administrative Group.** AUG consists of a homogeneous assembly of AU-3s or an AU-4.

(*x*) **Concatenation.** The procedure with which the multiple virtual container are associated with one another, with the result their combined capacity could be used as a single container across which bit sequence integrity is maintained.

PART V: DIFFERENT ACCESS TECHNIQUES

4.27 DIFFERENT ACCESS TECHNIQUES

Across Network, the network between local exchange and subscribers in the Telecom Network accounts for A major portion of resources both in terms of capital and manpower. So far, the subscriber loop has remained in the domain of the copper cable providing cost effective solution in the past. Need for quick deployment of subscriber loop, coverage of inaccessible and remote locations and requirement of more bandwidth for new services coupled with advances in technology have led to the emergence of new Access Technologies. Modern access network technologies are discussed here.

4.28 IMPORTANCE OF ACCESS NETWORK

At present the access network represents approximately 45% to 50% of the total capital investment in the telecom network. It is therefore a very substantial portion of total network and must be given due attention. If this access network is properly maintained most of the problems in the telecom network could be avoided.

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However the copper pair cables still dominate the subscriber loop (local network) due to certain reasons primarily based on techno-economic considerations. This copper based local network is considered to be responsible for most of the faults in telecom network. The obvious reasons are congestion of underground facilities, complex network planning and limitation of copper cables to handle digital signals leading to a network inappropriate for extending broadband integrated services digital network (ISDN).

Introduction of Digital Technology coupled with radio transmission and optical fibre cable has revolutionized Telecom Network worldwide. The overall reliability of network has improved vastly. In India too these concepts have been field tried in the access network. The implementation of the above technologies in the access network can be as follows:

(a) Using radio in the access network (WILL Technology)
(b) Using fibre in the access network (*FITL* Technology)
(c) Exploiting the existing copper network for higher bandwidths. (HDSL,ADSL,VDSL Technologies)

Let us discuss above modern technologies which are largely set to replace copper in subscriber loop.

4.29 WIRELESS IN LOCAL LOOP (WILL)

Radio communication has been employed as a replacement for copper based cables in the long distance media for several years. More recent developments of digital radio and advances in micro-electronic circuits have given rise to *wireless in local loop (WILL)*. It involves using radio to replace the wired link between PSTN switch and the subscriber. WILL is generally used as "the last mile solution" to deliver basic phone services expeditiously where none has existed before. It shall facilitate cordless telephony for residential as well as commercial complexes where people are highly mobile. It is also used in remote areas where it is uneconomical to lay cables and for rapid development of telephone services. The main advantages of this technology are:

1. Fast deployment and hence early access to revenue.
2. Reduced service interruptions.
3. Low maintenance and operational costs.

The radio technology is able to offer the same level of service quality as that provided by wire line technology. The subscribers have no knowledge of their radio connection and may access all the offered PSTN services in exactly the same way as if they were directly connected by wire line. Application of wireless local loop has just started worldwide. The technology employed shall depend upon various radio access techniques like FDMA, TDMA, CDMA.

There is no international standard for this so far. However, a number of national and regional air interface standards for Digital Cellular Mobile Telephone system and cordless telephony are available. These are being adopted for fixed wireless in local loop application. The various technologies available in International market for WILL application are as follows.

4.30 TECHNOLOGY OPTIONS FOR WILL

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1. Basically two types of technology options are available for wireless in local loop. The first one based on cellular mobile Telephone system can be adopted for fixed wireless in local loop application. These systems have *Macro* cell *architecture with cell radius of tens of kilometers (typically 10–20 Kms), the* second type based on *Micro* cell architecture are extension of cordless telephone systems. These systems have cell radius of few hundred meters (typically 50–200 mts).
2. **Point to Multi Point (PMP)** systems also called Digital **MARR** systems are becoming available. These systems can cover long range depending upon line of sight conditions (LOS) and repeaters. These systems can be found attractive in hilly areas, isolated islands or largely dispersed habitation where multiple of subscribers are to be served.
3. **Satellite** media can also be used to extend local loop to subscribers who are otherwise located at inaccessible places where laying of cables or line of sight radio media is not economically justified. For example certain villages have been extended *gram panchayat* telephones using satellite media for providing rural communication. Generally social factors dominate cost considerations for providing local loops in such cases. Now a days very small aperture terminals (VSATs) being used for interactive data communication have proved successful in business/corporate applications. Multichannel per carrier VSATs are also used to extend trunk junctions to remote and inaccessible/ hilly areas.

Through its wide area broadcast capability, a geostationary earth orbit (GEO) satellite is able to deliver essentially the same throughput signal throughout the country or region at an attractive cost per user. Taking advantage of this factor direct to home (DTH) satellite broadcasting with a smaller antenna at the subscriber roof top is also an extension of local loop over the satellite.

Many personal communication services (PCN) services have been planned using low earth orbit (LEO) satellites which permit users with portable/handheld terminals to connect themselves directly to the LEO satellites. In the process the local loop is extended for fixed/mobile application.

4.31 FREQUENCY BAND

The WILL technologies available in different frequency and their important parameter are indicated below:

SYSTEM TECHNOLOGY	MACRO CELLULAR			MICRO CELLULAR		
FREQUENCY	GSM 890–915 935–960 MHz	DAMPS 824–849 869–894 MHz	CDMA 824–849 869–894 MHz	*CT2 864–868 MHz*	DECT 1810–1900 MHz	PHS 1895–1918 MHz
CELL SIZE	LARGE	LARGE	LARGE	*SMALL*	SMALL	SMALL
MULTIPLE ACCESS	TDMA	TDMA	CDMA	*FDMA*	TDMA	TDMA
RF CHL SPACING	200 KHz	30 KHz	1250 KHz	*100 KHz*	1728 KHz	300 KHz

VOICE CHL / CARRIER	8	3	25-45	1	12	4
MODULATION	GMSK	PIE/4-QPSK	CDMA	FSK	GFSK	PIC/4 QPSK

GSM Global System for Mobile Communication
DAMPS Digital Advance Mobile Phone Service
CDMA Code Division Multiple Access
CT-2 Cordless Telephony-2
DECT *Digital Enhanced Cordless Technology*
PHS Personal Handiphone Service

4.32 FIBRE IN LOCAL LOOP (FITL)

In order to overcome the limitations of copper cable and to be able to support value added broadband service like data, cable Video, HDTV and increased use of computer which require bandwidth on demand, optical fibre is introduced in the local loop. In light of its infinite bandwidth and high reliability, optical fibre cable is the automatic choice for the local loop.

4.33 ADVANTAGES OF OPTICAL FIBRE IN THE LOOP

(*i*) Impact of environmental factors is almost negligible on optical fibre cable.

(*ii*) Optical fibre cables are not susceptible to electromagnetic interference and hence there is no possibility of intercepting information at any point.

(*iii*) The limitation on loop resistance is eliminated by introduction of active elements providing appropriate amplification.

(*iv*) Because of fibre's unlimited bandwidth capacity upgradation is very simple as it can be affected by simply changing the end terminal equipment. Repeated digging and cable laying is not required.

(*v*) Small size of fibre cable avoids congestion in ducts and crowding at MDF.

(*vi*) Due to their inherent wide bandwidth capability optical fibre cable can support narrow band and broadband ISDN services. They can also support video transmission, thus bringing the telephone services and cable TV operations together.

The advantages offered by FITL and limitations of copper access network can be tabulated as:

Copper Access Network Challenges	FITL Advantages
• Bandwidth Limited	• Virtually unlimited bandwidth
• Planning/Engineering	• Reduced impact of forecasting errors • Simplified engineering with universal access
• Maintenance	• "No maintenance with fibre"
• Security	• Requires physical security only
• Reliability	• Dielectric media
• Obsolescence	• Permanent outside plant

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4.34 FITL NETWORKING

The long term objective of FITL is to take the fibre right upto the subscriber premises or else to extend the fibre as close to subscriber as possible. The various approaches towards the end goal depending upon its penetration in the access network can be listed below

(*i*) Fibre to the Building (FTTB)
(*ii*) Fibre to the Curb (FTTC)
(*iii*) Fibre to the Home (FTTH)

Implementation

While the use of fibre optics in the access network is extremely advantageous and promising but it is still in the initial stages. Currently fibre costs are higher as compared to copper but there is a trend towards decreasing costs of optoelectronics and optical fibre cables. Some of the implementation issues and limitations of FITL specially in Indian contest are as follows:

(*i*) The reliable power supply at the remote end other than the exchange end is a must to exploit the reliability and other advantages promised by FITL concept.
(*ii*) There is no major identified demand for broad band ISDN service. Thus unless the cost economics of FITL is justified its application may be some what slow.

4.35 HIGH BIT RATE DIGITAL SUBSCRIBER LINE (HDSL)

While there is no denying the fact that the fibre will eventually take over the last mile (access portion) of the network,it is felt and generally accepted that it would not be economically justified immediately and, would take some more time before it penetrates in the access network. The Telecom Administrations world over have already invested a lot in, terms of their copper based network and will continue to do so until the fibre becomes more techno-economically feasible. Till such time there is no alternative but to exploit the already buried (but not dead) copper to carry more and more bandwidth.

HDSL is one such[1] technology employing a transmission technique which derives substantial capacity advantage in transporting digital signals in local network over existing copper pairs by providing clear 64 kb/s channels supporting *voice, FAX* and data services with a improved transmission performance. In other words HDSL is able to convert the copper pairs into high speed digital line carriers what were essentially individual voice telephone line carriers. HDSL technology promises following advantages

(a) The subscriber connectivity over a distance of 4.5 Kms on 0.5 mm copper pairs carrying 2.048 Mb/s data.
(b) The adaptive digital signal processing used in HDSL allows near fibre-quality transmission.
(c) Remote power fading over copper wires from exchange to subscribers.
(d) There is no need for any cable conditioning or selection of pairs.
(e) It is possible to extend the range of operation to 9 Kms over single repeater.

HDSL Technology can prove very useful in improving inter exchange junction working and subscriber access network utilizing the already buried copper pair cables.

Recently some new technologies Asymmetric Digital Subscriber Line (ADSI) and very high Speed Digital Subscriber Line (VDSL) have come up which promise to implement digital TV broadcast, video on demand interactive distance learning and home shopping on the same simple existing copper lines. ADSL can carry 6 Mb/s signal over 3.5 Kms (approx) and VDSL can carry 26 Mb/s to 52 Mb/s over a short distance of 600–1000 meters approximately. Later on it is expected to carry even 620 Mb/s over 100 meters. VDSL may find its use in business applications within a building. Many Telecom Administrations are beginning to evaluate and install ADSL services.

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4.36 CONCLUSION

Subscriber loops form a very important part of telecom network. The increasing appreciation of draw backs of present copper based network and introduction of high band width services have necessitated many alternate modern access technologies. Flexibility and expediency are becoming the key driving factors behind the deployment of WILL. This is a step towards mobile communications leading towards personal communication services (PCS).

The deployment of optical fibre in the access network promises many advantages as compared to traditional copper cable network. Fibre extension closer to subscriber premises will provide an economical, flexible and easily upgradable transport media for carrying existing and emerging range of services including telephony, distributive video services, high bit rate data and broadband ISDN services. HDSL, ADSL and VDSL technologies allow a techno-economically feasible migration from copper based network to a broad band fibre based network of future.

These modern Access Technologies shall avoid further large scale deployment of copper cable and shall pave the way for setting up a strong Access Network infrastructure required to step in the future Telecom Network of 21st Century.

SUMMARY

- TCP is a connection-oriented transport protocol that sends data as an unstructured stream of bytes.
- The address shortage problem is aggravated by the fact that portions of the IP address space have not been efficiently allocated.
- LANs interconnect computers and peripherals over a common medium so users might share access to host computers, databases, files, applications, and peripherals.
- Across Network, the network between local exchange and subscribers in the Telecom Network accounts for A major portion of resources both in terms of capital and manpower.
- At present the access network represents approximately 45% to 50% of the total capital investment in the telecom network. It is therefore a very substantial portion of total network and must be given due attention.
- More recent developments of digital radio and advances in micro-electronic circuits have given rise to *wireless in local loop (WILL)*.
- In other words HDSL is able to convert the copper pairs into high speed digital line carriers what were essentially individual voice telephone line carriers.

NOTES

REVIEW QUESTIONS

1. What is TCP?
2. What is IP?
3. Discuss different classes of IP addresses.
4. What do you understand by LAN?
5. What are different Transmission Media in LAN?
6. What are different standards for Twisted Pair?
7. Discuss Fibre-Optic Cable as a transmission media.
8. Discuss Wireless LANs (WLANs).
9. What is ATM?
10. Discuss ATM protocol.
11. What are different ATM protocols?
12. What are different ATM interfaces?
13. What are different ATM Connection?
14. Discuss SDH principle.
15. What are different merits of SDH?
16. Discuss different SDH standards.
17. Discuss Different Access techniques and their comparison.

FURTHER READINGS

1. *Telecommunication and Information Technology,* Prashant Kaushik, Anmol, 2006.
2. *Optical Networking in Telecommunication,* S. Mukherjee, Jaico.
3. *Wireless Technology and Access of Information,* Ajay K. Srivastav, Shree Pub., 2006.
4. *Elements of Networking Engineering,* Kumar Prasun Ramakrishnan, Shree Pub., 2010.
5. *Trends in Networking and Communication,* Edited by Girish Kumar Srivastav and Charul Bhatnagar, Atlantic Pub., 2009.

UNIT V: TELECOM NETWORK

★ STRUCTURE ★

NOTES

LEARNING OBJECTIVES

After going through this unit, you will be able to:

- know about telecom network
- define vertical network
- explain briefly about history
- describe the features of NGN
- define characteristics of next generation networks

- know about broadband
- describe evolution of mobile communication technologies

PART I: TELECOM NETWORK

NOTES

5.1 WHAT IS A NETWORK?: THE FUNCTIONAL CONTENT OF A SERVICE PRODUCTION UNIT

To make it simple, let us start with the traditional telecom networks and split them into two logical systems: the *traffic system* and the *management system*. The traffic system serves the end-users, and the management system supports the traffic system with the activities that can be found in Figure 5.1. This is a general system model, which is valid for all types of network architectures.

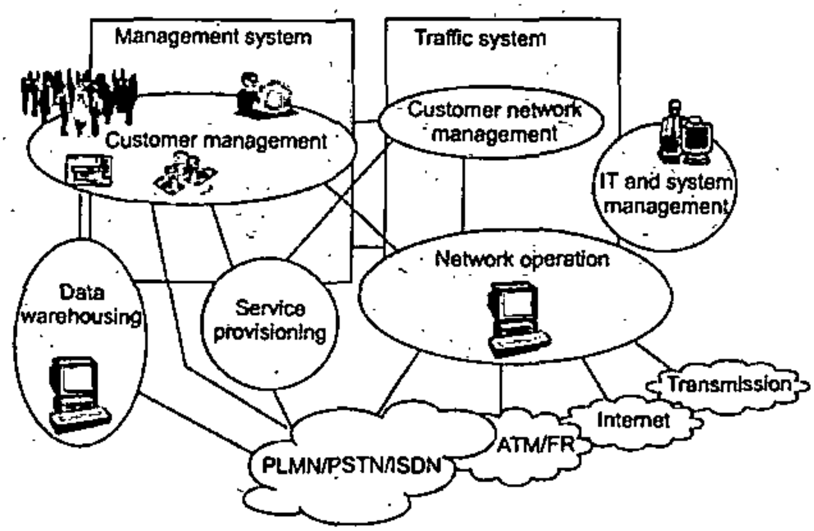

Fig. 5.1 The management and the traffic systems

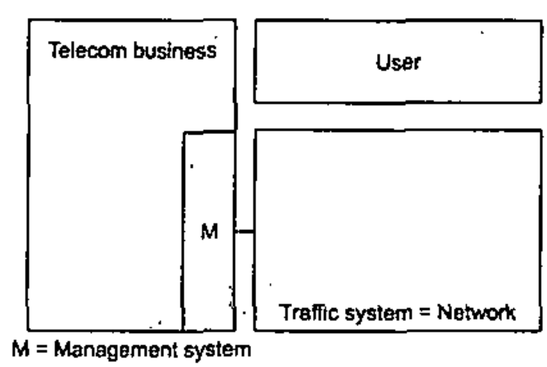

Fig. 5.2

The management system is at least partly a function of the traffic system, that is first we have to understand the traffic system before defining how to manage it. Adding the users we get the triangle which is found throughout this book (see Figure 5.2).

Any traffic system is characterized by two main functions: resources and resource control. This is a programmed control, not to be confused with management. In

ISDN, the concepts of *user plane* with the user traffic, and *control plane* were introduced and defined. In PSTN the planes are not described separately, and consequently there are no standardized interfaces between them. In the evolving horizontal network, the distinction between these planes is pronounced. An example is the split into a control layer and a connectivity layer.

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Packet-switched networks (or packet-mode networks) are sometimes regarded as having no control plane. Especially in connectionless networks, the control layer is difficult to identify, since for example routing is done for every single packet. (Connectionless = 'Any packet can theoretically take its own path to the receiving end'.) However, there is always a kind of control plane. In packet-switched networks it just partly deals with other issues than the tasks of the control plane of circuit-switched networks. For IP networks this is a heritage from the original US defence-based requirements on robustness, with quite a lot of intelligence located at the terminal devices, and a less intelligent but more autonomous network part in comparison with the circuit-mode networks. Among IP control functions are traffic supervision (ICMP, Internet control message protocol), routing (BGP, border gateway protocol, for routing to/from external domains; OSPF, open shortest path first, for internal routing; and IGMP, Internet group management protocol for multicast routing), QoS (different alternatives such as RSVP, resource reservation protocol), DiffServ and MPLS (multi-protocol label switching) and security (IP security).

IP network nodes are able to learn the network topology by themselves and perform routing autonomously. This minimized network administration is in line with network robustness and reliability policies so that the network might automatically recover from any intentional damage.

The importance of control is related to business. Most markets are deregulated today. In the resulting competitive environment business models and charging possibilities become central questions. What we can charge for depends on the services and network control we have. Control means awareness. And awareness means charging possibilities. The awareness might concern duration, calling and called IP address, bandwidth, security and QoS requirements.

The telecentric operators are used to a firm control of the traffic through their networks. The networks are fairly intelligent, often with sophisticated charging possibilities. David Isenberg in the USA wrote a famous article in 1997, called 'The rise of the stupid network', see http://www.rageboy.com/stupidnet.html, contrasting the intelligent telecentric networks with a more advantageous future network where most intelligence was located at the terminals. The contemporary Internet was considered a rudimentary form of this future network.

Service and *QoS assistance* from the network were however regarded as necessary. Another possible area for network assistance is *security*. So, also in a 'stupid' network there is a need for assistance to the end user from the network operator. This assistance means at the same time that the operator gains control of the subscriber calls.

A key component for awareness in the new horizontal networks is called the call session control server (CSCF). The CSCF will gather charging information and send it to a billing server.

Control signalling is transported in very different ways in circuit and packet-mode networks. In circuit-switched networks control messages are normally handled and transported separately from user traffic by the well-standardized separate sub-network Signalling System 7 (SS7). In packet-switched networks data of the control plane and the user plane are transported in the same messages. This fact may have added to the impression that packet-switched networks lack control planes. In the horizontal networks the separation is not just logical (as in ISDN) but there is also a physical implementation defined called the gateway control protocol, GCP/H.248.

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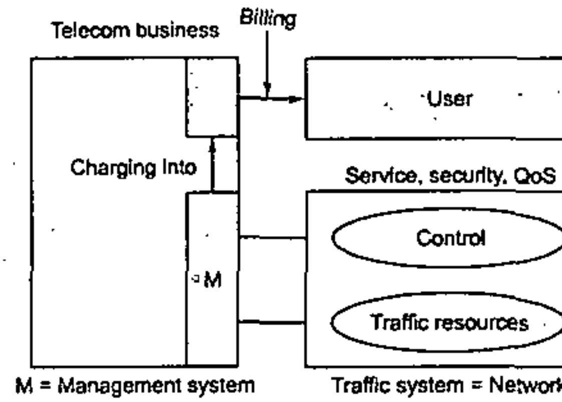

Fig. 5.3 Control, traffic resources and charging based on control

5.2 WHAT IS A VERTICAL NETWORK?

Today, almost all networks are 'vertically integrated' single-service networks where the operator offers everything from subscriber access to service creation and service delivery across a wholly owned network infrastructure, optimized for a particular service category. Each vertically integrated network has its own protocols, nodes, end- user equipment/ terminals and builds on different principles and practices to ensure reliability of a single service.

These vertical 'monolithic' networks have basically their own fundamental plans with some exceptions: PSTN and ISDN are strongly related, and the transmission facilities are often shared between different networks. To a more limited extent an IN (intelligent network) network service layer is also common for a number of networks. An obvious difference between them is the user service-oriented top layers, which are much more developed in the IP networks. See Figure 5.4.

The new network has to interwork with the networks already present. Therefore a good knowledge of current 'vertical' networks, in particular their network properties, benefits and drawbacks, is an excellent starting point (Figure 5.5).

The various network types were originally dedicated to various media type services according to Table 5.1. GPRS is considered a step between GSM and UMTS but is no doubt supporting mobile access to the information society media sources. The telephony and data service domains are still more or less kept separate.

Historically, most functionality was located on telephone exchanges with vertical interfaces offering a number of signalling interfaces for adaptation to the existing network environment. Taking Sweden as an example, hundreds of signalling systems and variants have been used. Apart from signalling the interfaces were fairly homogeneous. The carried voice traffic mainly had only two shapes: either analogue with a bandwidth of 300–3400 Hz or digital with 64 kbit/s bandwidth. It should be borne in mind that both forms represent the circuit transfer mode. Today, the situation is different, voice can be carried by many transfer modes (VoIP, VoATM, VoFR) with many different bandwidths.

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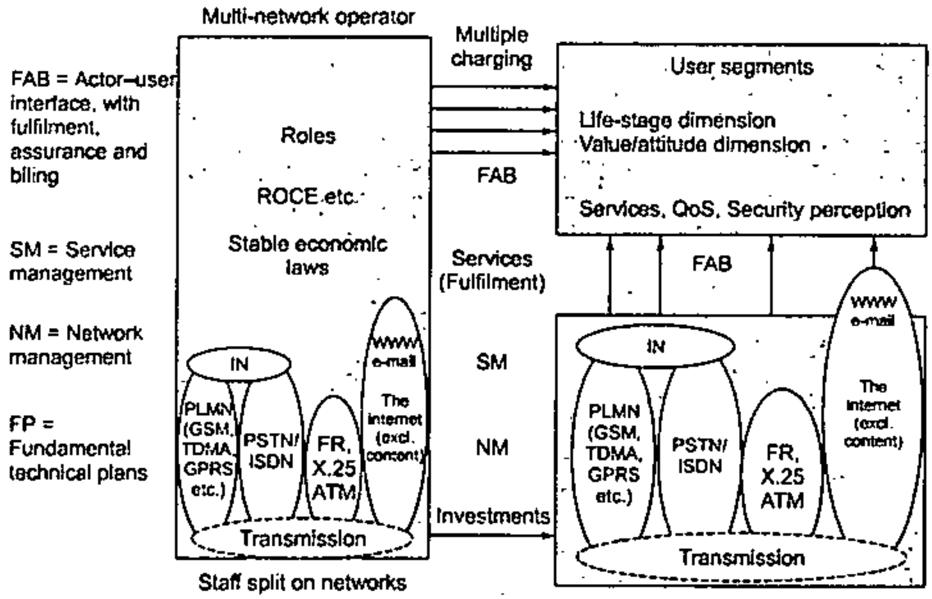

FIG. 5. 4 The vertical 'monolithic' networks demand a staff split on networks, multiple subscriptions and multiple charging

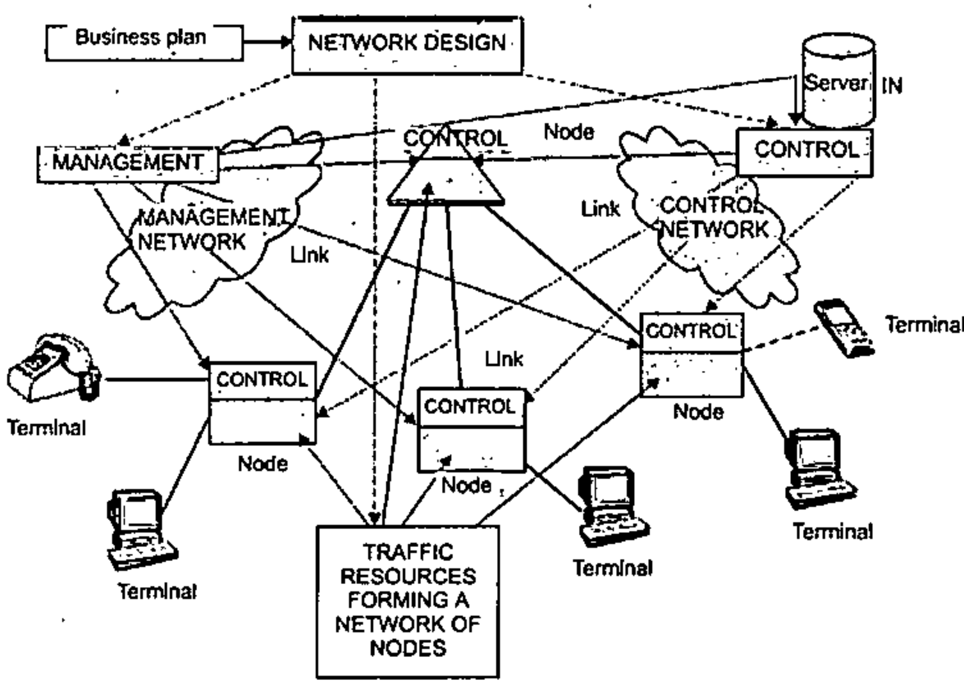

FIG. 5.5 A simple distributed and node-oriented tele-centric view of a traditional network

In order to introduce new functions into the 'historical' PSTN it was necessary to upgrade the exchanges or build a more service-rich overlay network, such as an ISDN network. It is easy to realize that individual upgrades of exchanges is a very timeconsuming way of improving network functionality. However, it was in the monopoly era, and the risk of losing subscribers was very low (and the waiting lists were often very long!).

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TABLE 5.1 The vertical networks and the targeted services

Service	Network
Voice	PSTN
Data, wide area	X.25
Data, local area	Ethernet, Token Ring
Data, 'Faster X.25'	Frame relay
Data + voice 'dream'	ISDN, ATM
Data internetworking	Internet
Voice with mobility	GSM, TDMA, CDMA, etc.
Voice + data + mobility	Next generation network, 3G
+ information society	(e.g. UMTS, WLAN)

The IN architecture brought the first 'server' into the fixed network at the end of the 1980s. This was before the convergence between voice and data networks, and the term 'server' was not used by telecom staff. The IN concept with a service control point (SCP) node on top of a group of exchanges also became integrated into the mobile systems. The SCPs offer a centralized way to introduce new services all over the network, proving the advantage of a server-supported 'layered' network. The time to customer for the operator could now be shortened. However, the interfaces between telephone exchanges were not simplified. On the contrary, since the IN concept adds new signalling in the network, the overall interface complexity grew. Managing an IN network has also offered more problems than expected.

It should be borne in mind that the IN services to a large extent are 'network oriented', often meaning advanced routing, advanced charging or advanced numbering. Such services have little or no content to offer, as opposed to for example WWW services. A more system-oriented approach was necessary when the mobile telephony systems were standardized. Here the functionality was more distributed, with many 'servers' in the network (such as HLR, VLR, EIR, AUC in GSM) starting with the second generation of systems from the beginning of the 1990s. IN was also introduced in this generation of mobile systems. A mixed fixed-mobile system came with 'fixed cellular systems'.

The data-oriented networks, on the other hand, were layered from start, which enables a smooth transition to the new horizontal network. However, except for ATM, they are optimized for data only. The introduction of TCP/IP networks meant something new, especially a possibility for global communication, since IP could 'surf' on other networks. The service offerings on top of TCP/IP, headed by WWW services and e-mail, were another strong feature. So it is not curious that the future network might lend many features from TCP/IP networks. See Figure 5.6.

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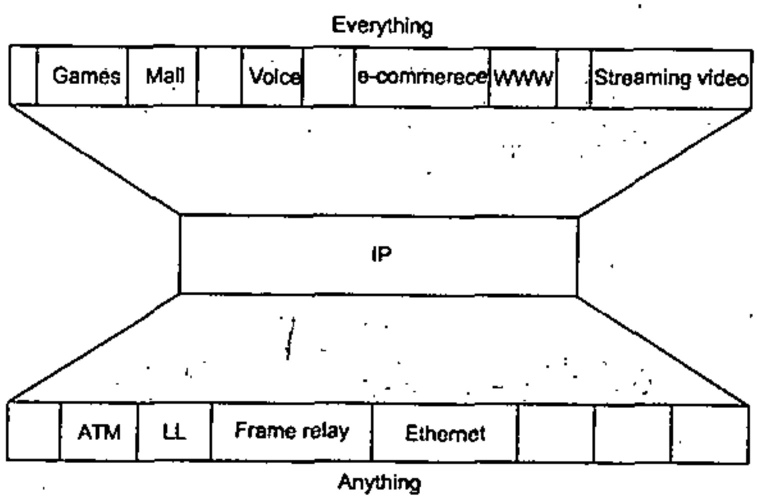

FIG. 5.6 The 'secret' behind Internet success and its worldwide connectivity

ATM was designed to become a multimedia-capable type of network. ATM has mainly found its niche in the network backbones as an important layer 2 technology. Closer to the subscribers Ethernet is a very strong candidate at layer 2.

PART II: NEXT GENERATION NETWORKS

5.3 NEXT GENERATION NETWORKS

Next Generation Networks (NGN) are the next step in world communications. NGNs are the culmination of 100 years of telecommunications evolution, combining the scalability and reliability of the public telephone network with the reach and flexibility of the Internet. The next-generation network seamlessly blends the public switched telephone network (PSTN) and the public switched data network (PSDN), creating a single multi service network.

Traditionally, now there are three separate networks: the PSTN voice network, the wireless network and the data network (the Internet). NGN converts all of these three networks into a common packet infrastructure. This intelligent, highly efficient infrastructure delivers universal access and a host of new technologies, applications, and service opportunities. The fundamental difference between NGN and today's network is the switch from current 'circuit-switched' networks to 'packet-based' systems such as those using Internet Protocol (IP). The need for global standards is critical as most operators expect to move to an IP infrastructure. One area to be addressed is the concept of 'nomadicity', which will give fixed line and mobile users completely seamless communication. It means that the underlying technology will be invisible to the user regardless of a multi-service, multi-protocol, multi-vendor environment.

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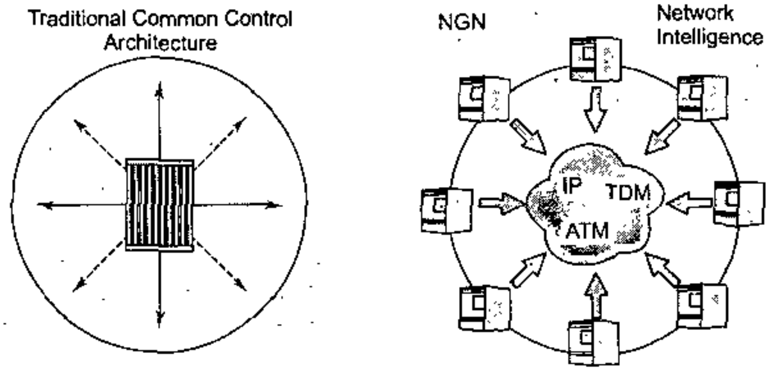

FIG. 5.7

5.4 HISTORY

The global telecommunications infrastructure has evolved over the past 100 years. The last two decades, however, have heralded seminal change that has accelerated this evolution manifold. The emergence of the converged network—driven largely by growth in video, voice and data traffic across the globe has been a major primer for change and all industry watchers agree that this is only the beginning.

Traditional circuit-switched telecommunications infrastructure is the foundation for the public switched telephone network (PSTN) that delivers telephony connections to homes and businesses today. This network is extremely demanding in its requirements for reliability and high availability. People expect, and generally receive, a dial tone when they pick up the phone.

How is such a reliable network assured? Under the existing paradigm, the phone system creates a dedicated circuit between the caller and the destination to complete a call. This line cannot be used by the system for other purposes during the duration of the call. Time division multiplexing (TDM) technology, on which circuit-switched telephony is based, allows the system to place multiple calls on its major trunk lines, but the dedicated circuit still consumes more network bandwidth than necessary.

High reliability and voice quality—as well as the lack of any viable alternative—meant that TDM based communication technologies grew and flourished. Till the Internet emerged! The Internet is a network of network, connecting millions of computers across the world. Widespread adoption of PC devices, evolution of killer applications such as the WWW and e-mail, as well as its efficiency in transfer of data traffic across the world saw a surge in Internet users through the 1990s.

The telecommunications world is at the crossroads today. As the amount of data traffic crossing the globe increases every second, the conventional infrastructure is seen to be increasingly incapable of handling it. On the other hand, the flexible and efficient data network — the Internet — can carry all forms of service traffic over it, but has been found to be unsuited for telephony.

As is usually the case — the market found a way out. The clash of the old-world and the new led to a wave of innovation and evolution for telecommunications. Today, copper and fibre optic lines that used to carry voice traffic now also transmit data, fax, and video. Traditional circuit switching is giving way to more efficient and flexible packet switching technologies as a result of the explosive growth of IP (Internet Protocol) networks.

New companies are entering the telecommunications space as service providers and old companies are adopting new business models built on new technology. In this competitive marketplace, telecommunication firms are looking to enhance the services they provide to their customers and reduce the costs of delivering them.

NOTES

One critical area of communications infrastructure that has been rapidly evolving in recent times has been switching technologies, as traditional switching functions give way to next generation of telecommunication switches. Switching is the core of all telecommunication networks, allowing efficient point-to-point communications without direct connections between every node.

To operate in the demanding and highly intensive PSTN domain, telecom switches are needed to be compatible with existing legacy systems and standard communications protocols. They are expected to deliver the high reliability that is expected today from a TDM network.

They are also expected to support value-added features and services that service providers allow carriers to differentiate themselves based on service and scale on demand. Such increasingly open architecture demands switching technology to upgrade to accommodate the emerging requirements from a communications network.

Rapid Progress in the Late 1990s

During the late 1990s, very rapid progress was made in overcoming these limitations. Gateways that can pass traffic between IP networks and the PSTN have been available since early 1998, and various groups have been working on the development of software that can be used to control gateways, in order to enable managed delivery of voice over IP. The era of circuit-switched telecommunication networks is drawing to a close. We are seeing the beginnings of a transition that will gather pace over the coming decade, from distinct and separate sets of infrastructure for telephony and data, towards the 'next-generation network', a single IP-based infrastructure for carrying all the voice, data and multimedia traffic associated with an increasingly wide range of network services.

One of the key reasons for the rapid acceptance for this technology has been its open-standards based architecture, which provides great flexibility for carriers to develop custom solutions based on best-of-breed hardware and software components.

5.5 DIFFERENT TYPES OF NETWORKS

Circuit Switching

In this method, a connection called a circuit is set up between two devices, which is used for the whole communication. Information about the nature of the circuit is maintained by the network. The circuit may either be a fixed one that is always present, or it may be a circuit that is created on an as-needed basis. Even if many potential paths through intermediate devices may exist between the two devices communicating, only one will be used for any given dialog. This is illustrated as follow:

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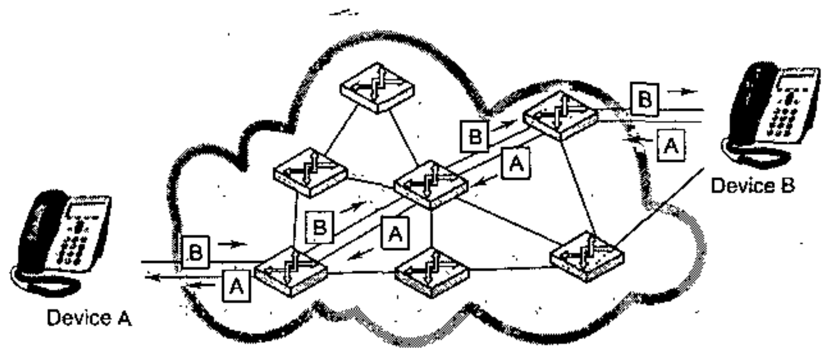

FIG. 5.8 Circuit switching

In a circuit-switched network, before communication can occur between two devices, a circuit is established between them. Communication link from A to B, and B to A are shown in figure. Once set up, all communication between these devices take place over this circuit. The classic example of a circuit-switched network is the existing telephone system. When A calls B and he answers, a circuit connection is established. That circuit function the same way regardless of how many intermediate devices are used to carry the voice. You use it for as long as you need it, and then terminate the circuit. The next time you call, you get a new circuit, which may (probably will) use different hardware than the first circuit did, depending on what's available at that time in the network.

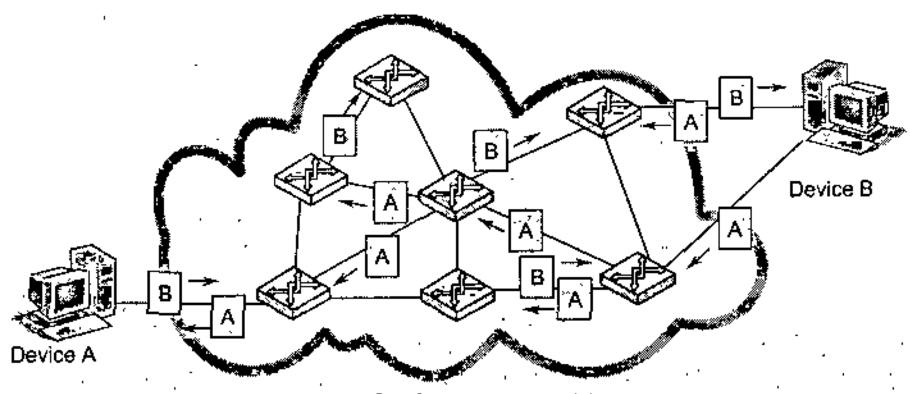

FIG. 5.9 Packet switching

In this network type, no specific path is used for data transfer. Instead, the data is chopped up into small pieces called packets and sent over the network. The packets can be routed, combined or fragmented, as required to get them to their eventual destination. On the receiving end, the process is reversed—the data is read from the packets and re-assembled into the form of the original data.

Packet Switching

In a packet-switched network, no circuit is set up prior to sending data between devices. Blocks of data may take any number of paths as it journeys from one device to another. In circuit switching, a circuit is first established and then used to carry all data between devices. In packet switching no fixed path is created between devices that communicate; it is broken into packets, each of which may take a separate path from sender to recipient.

The traditional Public Switched Telephone Network (PSTN)

- Built to provide VOICE service
- Intelligence at the core (central switch)

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- Dedicated circuit set up for each call
- Dumb terminals (cheap CPE)
- ATM, SDH, copper local loop technology
- Very reliable
- Licensed and highly regulated
- Usually monopoly
- Universal service obligation
- Emergency call service

The Mobile Telecom Network

- Built to provide VOICE/data service
- Intelligence at the core (central switch)
- Dumb mobile devices
- BSS, MSS, HLR/VLR, SIM cards
- Dedicated circuit set up for each call
- Less reliable than PSTN
- Licensed and highly regulated
- Two or more competing providers
- Emergency call service
- Interconnect to other mobile networks and PSTN by agreements

The Internet

- Built over PSTN to provide data service
- Information is routed, not switched
- Best efforts rather than guaranteed QoS
- Intelligence at the edge, large variety of devices and services connected to the internet
- Unregulated
- Many competing providers
- No universal service obligation or emergency call service
- Interconnect between clouds by peering or transit agreements

Voice over Data

As data traffic began to equal and surpass voice traffic on telecommunications networks it became economic for operators to consider transporting their voice traffic over packet switched networks. This convergence would help reduce the costs associated with operating and maintaining separate networks. However there are many problems associated with obtaining circuit switched levels of service for real-time traffic (*e.g.*, voice) on packet switched networks which may not always have the sufficient capacity (packets are discarded under congested conditions in packet switched networks resulting in delayed or lost data which is unacceptable during telephone conversations).

5.6 DEFINITION OF NEXT GENERATION NETWORK BY ITU

A Next Generation Network (NGN) is a packet-based network able to provide services including Telecommunication Services and able to make use of multiple broadband, QoS-enabled transport technologies and in which service-related functions are independent from underlying transport-related technologies. It offers unrestricted access by users to different service providers. It supports generalized mobility which will allow consistent and ubiquitous provision of services to users.

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5.7 FEATURES OF NGN

- Packet-based transfer
- Separation of control functions among bearer capabilities, call/session, and application/ service
- Decoupling of service provision from network, and provision of open interfaces
- Support for a wide range of services, applications and mechanisms based on service building blocks (including real time/streaming/non-real time services and multi-media)
- Broadband capabilities with end-to-end QoS and transparency
- Interworking with legacy networks via open interfaces
- Generalized mobility
- Unrestricted access by users to different service providers
- A variety of identification schemes which can be resolved to IP addresses for the purposes of routing in IP networks
- Unified service characteristics for the same service as perceived by the user
- Converged services between Fixed/Mobile
- Independence of service-related functions from underlying transport technologies
- Compliant with all regulatory requirements, for example concerning emergency communications and security/privacy, etc.

5.8 APPLICATIONS OF NGN

Telepresence

Telepresence is the ability to interact in real-time with another person who is at a different location using telecommunications. Telephony is a Telepresence application in its most simple form. Advanced Telepresence systems operating on next generation networks will enhance users' experiences of realism while communicating. Applications such as high quality video-conferencing systems would require capacities of between 2 and 8 Mbit/s1 per user. (Current video conferencing systems can operate at capacities of between 128 and 384 kbit/s but provide a low quality service.)

Video conferencing technology is currently most common in the business world, and applications are also being developed in the fields of education and medicine. When NGNs make ample capacity available it is conceivable that video conferencing could be adopted on a mass basis as a replacement or augmentation of basic telephony.

3D Imaging

Adding three-dimensional aspects to the imaging systems of Telepresence will further enhance the experience of Telepresence. Initially, this sort of enhancement could have applications for business users, enabling delegates to sit down to a virtual meeting and hold real time discussions while viewing other delegates on three dimensional monitors. Other applications are in the medical and educational fields. At a more advanced stage Telepresence will become interchangeable with virtual reality, and applications in entertainment are envisioned.

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Virtual Reality

When we think of virtual reality we often think of applications involving complete Teleimmersion. However it is likely that applications will develop that blend reality and virtual reality forming hybrid realities to enhance our experiences. An example of this could be a type of visual display that could project images onto a user's normal field of view using devices mounted on eyeglasses, allowing them to receive augmented information relating to their environment such as directions to the nearest hospital or police station.

To further enhance users' sense of realism the sense of touch could be incorporated into virtual reality systems through interfaces. Such systems allow users to touch and manipulate virtual objects. This aspect is essential for telesurgery applications,It is conceivable that in the future the senses of taste and smell could also be incorporated in virtual reality systems.

Data Augmentation

Further value can be added to Telepresence applications by augmenting services with additional information. In many ways this could allow Telepresence to surpass real face to face communication. For example, future face to face communications may often have files attached to them such as work that had been jointly undertaken during a Telepresence meeting.

Tele-Learning/Tele-Education

Tele learning or Tele-education is the application of telecommunications technology in education and training. Next generation Tele-education applications will use advanced graphical visualisation tools to help users understand difficult or abstract topics and also provide users with an opportunity to learn in a safe and non-critical environment (*e.g.*, flight simulation training, surgical procedure training). Some of these applications will require the use of three dimensional and virtual reality simulators.

Interactivity is also an important feature of Tele-education, allowing users in remote locations to focus on areas where they are experiencing particular difficulties for example, and will enable a higher level of one to one interactivity with tutors (real or virtual). Applications of this type could involve a mixture of real-time and stored data. Interactive Tele-education is also applicable in class-room environments. Already on-line learning is a growing Internet application.

Tele-education provides users with the convenience of being able to learn at more convenient times and places (*e.g.*, from home in the evenings instead of at a college during the day, or in work at the desktop). Also, Tele-education gives users the opportunity to select more specific course material that is directly applicable or tailored to their individual interests.

The capacity requirements of these systems will vary according to the level of quality sought from the video images, and it can therefore be expected that capacities of 2 Mbit/s or more would be required for video-conferencing applications.

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Tele-Medicine

Tele-medicine or medical informatics is the use of telecommunications technology in medical applications. These applications would be greatly facilitated by highly reliable next generation networks. Tele-medicine will allow the transfer of records or actual medical conditions between patients and medical personal in geographically diverse locations. Furthermore, Telepresence applications will enable medical staff to conduct face to face meetings with other staff and patients without the need to travel.

An important future Tele-medicine application is Tele-surgery, in which a surgeon views the patient through a three dimensional display and conducts a surgical operation via robotic instruments from a remote location using a high capacity telecommunications link.

Other medical imaging techniques are well suited to Tele-medicine allowing for the diagnosis process to occur at a different location from the patient and collection of information (*e.g.*, digital imaging, tissue sample analysis). This form of Tele-medicine is now common on hospitals' local area networks with the transmission of x-ray images. Next generation networks will enable widespread use of such applications.

Tele-education also has applications in the medical area. Similar imaging techniques to those used by the remote surgeons mentioned above can be used in the training of medical staff.

Home Care

Home care involves monitoring and caring for patients at home using telecommunications technology. Time and costs can be saved by allowing nurses to conduct daily virtual visits to patients in geographically dispersed areas. Furthermore, the concept of person to machine communications could be utilised here as home care patients could be constantly monitored, reducing the recovery times needed in hospitals. Home care using telecommunications links can allow the elderly to extend the time that they can live independent lives in their own homes.. Although many of these applications do not require high data rates their mass adoption could produce significant traffic loads on next generation networks.

Data Integrity and Privacy

Important data integrity and privacy issues arise from the application of Tele-medicine. Tele-medicine applications that involve real time data concerning the well being of patients are critical in terms of data integrity. Any erroneous transmissions could result in mistreatment with potentially serious consequences. Also, as medical information is of a highly private nature security is a priority and will become a key consideration in the design of next generation networks.

Social Interactivity and Entertainment

High capacity applications will emerge in the areas of gaming, movies and social interactivity. Interactive gaming with multiple participants is already an established internet application. However, with increasingly intense gaming applications (*e.g.*, high resolution video graphics) more and more capacity is needed from telecommunications networks to support multi-player real-time use.

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Streaming video and audio entertainment will be important applications of next generation networks as traditional broadcasting services and delivery methods converge with telecommunications (*e.g.*, interactive TV). Applications such as video on demand (VOD) providing users with personalised viewing services and applications with added interactivity will require high capacity networks to serve them. Peer to peer networking of video, audio and even 3D virtual reality archives could also bear heavily on next generation networks as users swap massive amounts of data.

Machine to Machine Communication

In the paragraphs above mostly human to human and human to machine communications have been considered. As the number of devices or machines that are able to communicate continues to increase, telecommunications traffic between these machines will continue to increase exponentially. A number of commentators have suggested that machine to machine communication will exceed person to person communication on next generation networks in around five years. Although, for the most part, the early applications envisioned here would be narrow band applications (*e.g.*, environmental sensors to detect temperature, moisture levels, light intensity, movement etc.), the vast numbers of routine communications will make their aggregate capacity significant. Machine to machine communication could also allow for the development of smart environments which are environments or workspaces that are aware of the context in which they are being used. For example, if a child approached a TV terminal, children's' programs could be shown instead of stock market information. In a business environment a user could automatically receive relevant information based on a particular caller, or the attendees at a meeting.

Other applications of machine to machine communication could include improved safety on our roads by allowing road traffic to be automated. This would enable guidance systems in vehicles to communicate with one another to ensure that collisions did not occur.

Business Applications

Increasing levels of e-commerce will place increasing demands on next generation networks. Highly secure and reliable next generation networks will in turn encourage the growth of business applications as users become accustomed to and develop trust in e-commerce applications.

Increased telecommunications traffic from applications such as online banking and shopping will create large amounts of e-commerce traffic. Furthermore, video conferencing and virtual reality show rooms may change the way in which we choose products and services.

5.9 CHARACTERISTICS OF NEXT GENERATION NETWORKS

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Next generation networks will for the most part be high speed packet based networks capable of delivering a multitude of broadband services. Among other things these networks need to be both flexible and reliable. Although next generation networks will develop in many different ways they will all have a common set of broad characteristics. These characteristics are

- Protocol Independence
- Reliability
- Controllability and Quality of Service
- Programmability
- *Scalability*

Protocol Independence

In order to facilitate multiple forms of communications, next generation networks will need to be capable of operating a multitude of different communications protocols. Traditionally networks have been designed and implemented to transmit certain specific types of data such as voice, video or data. This required separate networks, using different sets of equipment (although usually using the same cables or transmission media) to support multi-media communications.

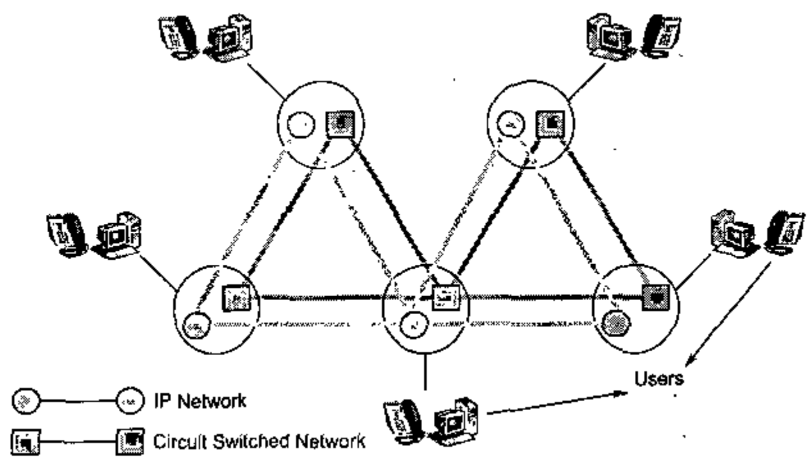

FIG. 5.10 A simplified diagram of overlaid IP and circuit switched networks showing the duplication of network resources.

Essentially, protocol independence is the ability of a network to operate any protocol that may be required. The ability of equipment to be multi-functional is increasingly required by telecommunications operators. It enables them to save on operational costs as equipment is managed from a single platform. Also, the physical space and hence costs that are saved with multi-functional equipment is a critical factor. Another significant factor is a reduction in the amount of power consumed by using less equipment.

Reliability

Increased dependency on advanced new applications in the future will place even greater reliability requirements on next generation networks. Individuals' *expectations of availability and quality of service*, grounded in a perception of high quality in traditional telephony and television services, will impose high standards of performance.

E-commerce applications will lead to highly resilient telecommunications networks as businesses become increasingly reliant on telecommunications to function. For other highly sensitive applications, such as tele-medicine, network reliability and resilience is imperative, since a patient's health could depend on the quality of the information transmitted.

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In order to achieve the necessary levels of resilience and reliability next generation networks will need more diverse topologies and redundant elements than is normal in today's networks.

Controllability

It is essential for network managers to be able to design, adapt and optimise their networks to accommodate simultaneously different types of media with varying network requirements. The main issue here is of quality of service, (*i.e.*, the ability of a network to provide a particular level of service or to guarantee a certain amount of bandwidth and response time over a specified period). For example a voice or video conferencing application could not normally afford to have information packets (*i.e.*, pieces of the conversation) lost or even delayed. Therefore these types of services need a guaranteed high level of quality of service to function adequately. On the other hand, non critical applications such as internet browsing can afford to lose occasional packets of information as these can be resent without degrading the service.

Control of these aspects of a network is an important characteristic since it allows network managers and network management software to optimise utilisation of network resources by dynamically setting the balance between the amount of capacity that is dedicated to real time applications and mission critical applications. Network managers also need to control the amount of flexibility that is applied to non-real time services such as file transfers (*e.g.*, downloading of design files from a design centre to the manufacturing plant). This is known as traffic engineering. Traffic engineering features of next generation networks will help overcome both the problems of guaranteed quality of service in current packet switched networks (*e.g.*, IP) and the problem of wasted capacity in dedicated circuit switched networks. See annex 1.

A common shortcoming of current packet switched networks is that it can be difficult for telecommunications network operators to specify or guarantee an end to end quality of service, particularly where part of the communications link is carried over a third party's network. For example a call originating on a network with a sufficiently high quality of service may terminate on a network, perhaps in a different country, where the quality of service is noticeably lower, thus resulting in a poor quality call. Using traffic engineering, operators can define specific levels of service and then enter into service level agreements with other operators who have similar traffic engineering capabilities. This process facilitates further interconnection between operators and networks.

Programmability

The more programmable and re-configurable next generation networks are the more flexible they will be, and the more they will be able to cope with new services

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and user requirements. Programmability will allow for traffic engineering and the dynamic allocation of network resources enabling next generation networks to adapt quickly to new services or requirements.

Programmability yields more simple scalability since the less manual configuration that has to be performed during a network upgrade the more quickly services can be expanded. The time it takes to provision new capacity in networks can be reduced from several weeks (in manually configurable networks) to a few hours or less through programmability. Fully programmable networks could be upgraded remotely from a single location eliminating the need for expensive site visits.

To aid interoperable and programmable networks open standards need to be supported by all equipment vendors. This will mean the provisioning of open Application Programming Interfaces (APIs) enabling developers to create software for equipment from various vendors to operate in interconnected networks.

Scalability

Scalability is an important attribute that can help protect next generation networks from becoming obsolete. In order to cope with growing traffic loads network operators will have to over-provision transmission capacity (*i.e.*, lay more fibre optics than currently needed). Next generation network equipment will need to be scalable to allow for the addition of capacity as required without the need to replace equipment once it reaches its design capacity.

The more general purpose that telecommunications equipment is the greater the chance that it can be programmed, adapted and scaled to cope with future needs.

Furthermore, next generation networks will need to be scalable in terms of address space (*i.e.*, the number of devices that can be connected and individually identified on a network).

5.10 TYPICAL NEXT GENERATION NETWORK ELEMENTS

Some typical next generation network elements are described below:

Soft switches

Soft switches are the key component that enables next generation networks to be built. They can be programmed to act as gateways allowing communication between packet based networks (*e.g.*, IP) and traditional circuit switched networks. The soft switch can mediate between IP-centric, or VoIP services and circuit switched telephony services converting all of the necessary added services accordingly. Soft switches execute the same functions as traditional switches and are completely transparent to end-users. Telecommunications companies are embracing soft switches because they are functionally equivalent to conventional phone switches; only better, faster, and cheaper. Soft switches tend to be modular, smaller, and less expensive than their conventional switching counterparts. This modularity makes scaling easy, critical when telephony markets and technologies can change overnight. All this is accomplished without any compromises on the high availability and reliability delivered by conventional switches.

DSLAM

Digital Subscriber Line Access Module, used to connect multiple DSL users to the rest of a network. A multi-service DSLAM interconnects to voice networks as well as other data networks.

Next Generation Edge Switch

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A multi-protocol switch that can connect users various access methods (*e.g.*, ISDN, Dial-up modem, Analogue telephony) to next generation core networks.

Broadband Access Switch

Connects broadband access networks (*e.g.*, Broadband leased circuits) directly to core networks. These devices connect network segments that are suitable for direct connection to core next generation networks.

PART III: BROADBAND ACCESS (WIRED AND WIRELESS)

5.11 BROADBAND ACCESS (WIRED AND WIRELESS)

Advances in telecommunications and data technology are creating new opportunities for countries, businesses and individuals—just as the industrial revolution changed fortunes around the globe. The new economy is defining how people do business, communicate , shop, have fun, learn, and live on a global basis—**connecting everyone to everything.** The evolution of Internet has come into existence and Internet service is expanding rapidly. The demands it has placed upon the public network, especially the access network, are great. However, technological advances promise big increases in access speeds, enabling public networks to play a major role in delivering new and improved telecommunications services and applications to consumers. The Internet and the network congestion that followed, has led people to focus both on the first and last mile as well as on creating a different network infrastructure to avoid the network congestion and access problems. The solution to this is Broadband.

5.12 WHAT IS BROADBAND?

A definition to broadband is a must as different service providers defines in their own terms and context. **TRAI (Telecommunication Regulatory Authority of India)** defines broadband as follows:

An '**always-on**' data connection that is able to support interactive services including Internet access and has the capability of the **minimum download speed of 256 kilo bits per second (kbps)** to an individual subscriber **from the Point Of Presence (POP)** of the service provider intending to provide Broadband service where multiple such individual broadband connections are aggregated and the subscriber is able to access these interactive services including the Internet through this POP. The interactive services will exclude any services for which a separate licence is specifically required, for example, real-time voice transmission, except to the extent that it is presently permitted under ISP licence with Internet Telephony.

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5.13 BROADBAND ACCESS

Broadband access technology is broadly classified into two categories. They are Wired Line & Wireless and further classified as detailed in the following diagram.

Broadband Access Technologies

- Wired line
  - DSL (Digital Sub's Line)
  - Cable Modem
  - PLC (Power Line Communication)
  - Optical Fibre Technologies
- Wireless
  - 3G Mobile
  - Wi-Fi (Wireless Fidelity)
  - WiMAX
  - FSO (Free Space Optics)
  - LMDS & MMDS
  - Satellite

5.14 WIRED LINE ACCESS

DSL (Digital Subscriber Line) :

DSL uses the exisiting twisted-pair telephone lines as the access media. Over a period of time, a number of technologies (**xDSL**) have been introduced to provide faster data speeds over this medium. The various xDSL technologies are given below.

1. ADSL (Asymmetric Digital Subscriber Line)
2. VDSL (Very High-Speed Digital Subscriber Line)
3. RADSL (Rate Adaptive Digital Subscriber Line)
4. HDSL (High Data-Rate Digital Subscriber Line)
5. SDSL (Symmetric Digital Subscriber Line)

5.15 ADSL (ASYMMETRIC DIGITAL SUBSCRIBER LINE)

Asymmetric Digital Subscriber Line (ADSL) is a form of DSL, a data communications technology that enables faster data transmission over copper telephone lines than a conventional modem can provide.ADSL has the distinguishing characteristic that the data can flow faster in one direction (used for download streaming) than the other(used for upload streaming) *i.e.*, asymmetrically.

Why ADSL?

ADSL is in place due to both technical and marketing reasons. On the technical side, there is likely to be more crosstalk from other circuits at the DSLAM (Digital Subscriber Line Access Multiplex) end (where the wires from many local loops are close together) than at the customer premises. Thus the upload signal is weakest, while the download signal is strongest at the noisiest part of the local loop. It therefore makes DSLAM transmit at a higher bit rate than does the modem on the customer end. Since the typical home user in fact does prefer a higher download speed, thus telecom companies chose to make a virtue out of necessity, hence ADSL come to place.

How ADSL Works?

To obtain the asymmetrical data transfer to suit requirement of Internet and LAN access, ADSL works by firstly splitting the available bandwidth on the twisted copper wire (telephone wires) into three different channel:

1. A high speed downstream channel (ranges from 1.5 to 8 Mbps)
2. A medium speed upstream channel (ranges from 16 kbps to 1 Mbps
3. POTS (Plain Old Telephone Service) channel

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ADSL uses two separate frequency bands. With standard ADSL, the band from 25.875 kHz to 138 kHz is used for upstream communication, while 138 kHz – 1104 kHz is used for downstream communication.

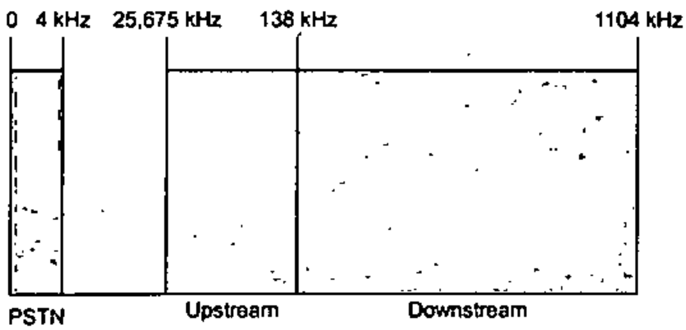

Fig. 5.11 Frequency plan for ADSL

First the POTS channel is splits off from the digital modem by filter, thus guaranteeing uninterrupted POTS. After the POTS channel are splitted from the digital data transfer bandwidth, the 26 kHz to 1.1 MHz data bandwidth could be further separated by using one of two ways as describe below: **(1)Frequency Division Multiplexing (FDM)** : FDM assigns one band for upstream data and one band for downstream data. Time division multiplexing divides the downstream path into one or more high speed channels and one or more low speed channels. But the upstream path is only multiplexed into corresponding low speed. **(2)Echo cancellation:** Echo cancellation assigns the upstream band to over-lap the downstream. To separate them is by local echo cancellation. This technique is common in V.32 and V.34 modems (Conventional Modems). By using either one of the above techniques, ADSL splits off a 4 kHz region for POTS at the DC end of the band.

5.16 ADSL MODULATION

ADSL uses two types of Modulation *i.e.*, CAP(Carrierless Amplitude Phase Modulation) & DMT(Discrete Multi Tone) & DMT is the most widely used one.

CAP(Carrierless Amplitude Phase Modulation): It is a variation of QAM (Quadrature Amplitude Modulation).QAM generates a DSSC (Double Sideband Suppressed Carrier) signal constructed from two multi-level PAM (Pulse Amplitude Modulated) signals applied in phase quadrature to one another. CAP modulation produces the same form of signal as QAM without requiring in-phase and quadrature components of the carrier to the first be generated. The following diagrams illustrates the CAP modulation.

5.17 CAP TRANSMITTER AND RECEIVER

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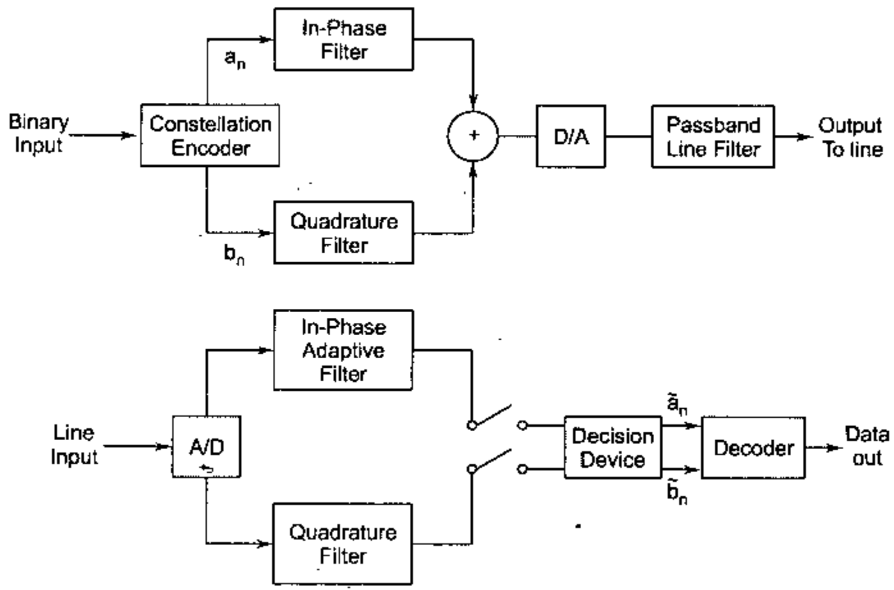

FIG. 5.12

5.18 DISCRETE MULTITONE MODULATION (DMT)

DMT is basically a multicarrier modulation technique. DMT spread the original spectrum of the input signal over numerous sub-channels each of which carries a fraction of the total information. All these sub-channels transmit data in parallel to each other and are independently modulated with a carrier frequency. By using DSP techniques, multiple sub-channels could be established using Fast Fourier Transform (FFT), where the sub-carriers had to have orthogonality with each other.

As mentioned before, DMT utilizes the spectrum between 26 kHz and 1.1 MHz. After using FDM or echo cancellation technique, this spectrum of bandwidth is split up into upstream band (26 kHz to 138 kHz) and downstream band (138 kHz to 1.1 MHz), which is then further divided into 256 discrete sub-channels each of which had a bandwidth of 4 kHz.

One of DMT most significant feature is that it is able to dynamically adapt to the line condition to obtain the maximum throughput for each unique telephone line. DMT does this by framing the data bits into chunks and spreads them over the sub-channels. The allocation of data into each sub-channel is dependent on the characteristics of the line and on the SNR (Signal to Noise Ratio) of the line. There could be no data at all in a really noisy channel and there could be as high as 15 bits/Hz in a channel where SNR is optimum. The major stages in transmitting and receiving could be seen in the following block diagram.

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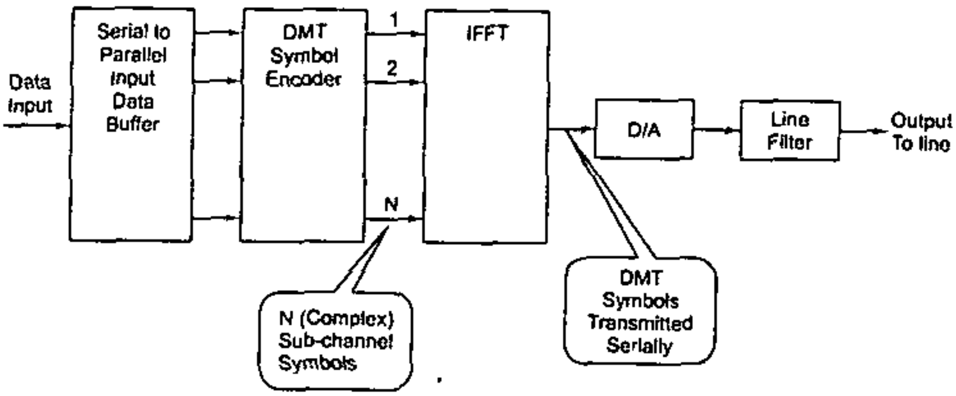

FIG. 5.13

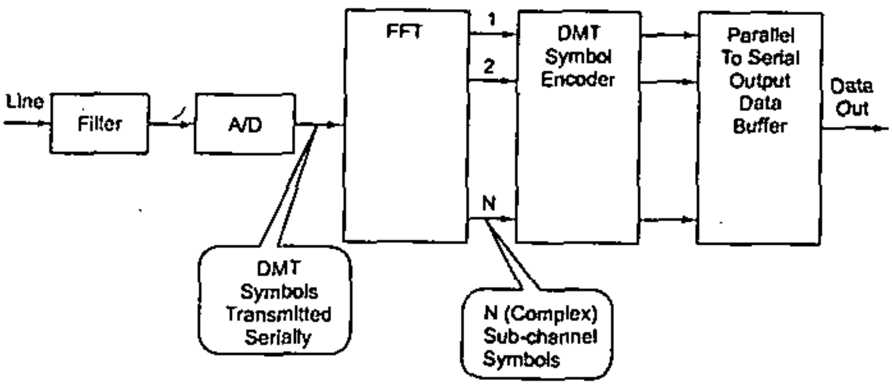

FIG. 5.14

The chunk of bits that are being assigned to each sub-channel as described above are encoded as a set of quadrature amplitude modulated subsymbols. These subsymbols are then pass into an Inverse Fourier Transform(IFFT) which combines the subsymbols into a set of real-valued time domain samples, the output of the IFFT is then send a Parallel-to-Serial block with cyclic prefix which is added to remove Inter Symbol Interference (ISI) between the sub-channels. The output is then pass into an digital to analog converter which is then send through the twisted copper telephone wire. The receiver would receive the signal from the twisted copper telephone wire and does the reverse process to obtained the required data.

To reduce error in transmission and to counter those problem of using telephone lines as a data transfer medium, DMT had uses Reed Solomon forward error correction method. The size of this Reed Solomon codeword depends on the number of bits assigned to each sub-channel.

5.19 COMMON ELEMENTS IN ADSL

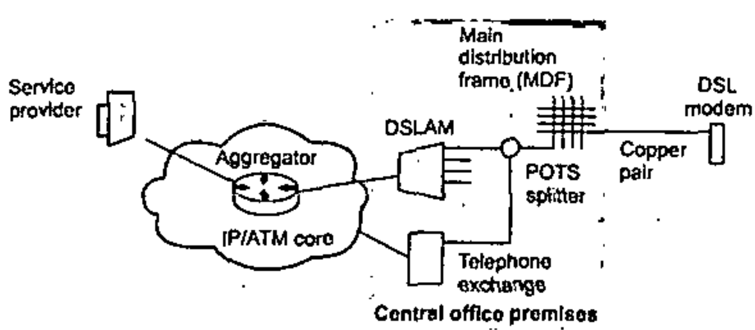

FIG. 5.15

The common elements of ADSL are

(a) CPE(Customer Premises Equipment) containing a Splitter, ADSL Modem & a PC.

(b) Central Office Premises Equipment containing DSLAM (Digital Subscriber Line Access Miltiplex), MDFs & PSTN.

(c) Aggregator and ATM core consists of Tier II, Tier I switches, BRAS (Broad band Remote Access Service), Servers and Core routers.

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Factors Determining ADSL Connectivity

More the distance from the DSLAM(Digital Subscriber Line Access Multiplex) to the customer end the data rate reduces. Signal attenuation and Signal to Noise Ratio are defining characteristics, and can vary completely independently of distance (*e.g.*, non-copper cabling, cable diameter). The performance is also dependent to the line impedance, which can change dynamically either dependent on weather conditions (very common for old overhead lines) or on the number and quality of joints or junctions in a particular cable length.

Data Rate - Wire Size-Distance

Data Rate	Wire Size	Distance	
1.5–2.0 Mbps	0.5 mm	18000 Feet	5.5 Kms
1.5–2.0 Mbps	0.4 mm	15000 Feet	4.6 Kms
6.1 Mbps	0.5 mm	12000 Feet	3.7 Kms
6.1 Mbps	0.4 mm	9000 Feet	2.7 Kms

5.20 ADSL STANDARDS

Standard name	Standard type	Downstream rate	Upstream rate
ANSI T1.413–1998 Issue 2	ADSL	8 Mbit/s	1.0 Mbit/s
ITU G.992.1	ADSL (G.DMT)	8 Mbit/s	1.0 Mbit/s
ITU G.992.2	ADSL Lite (G.Lite)	1.5 Mbit/s	0.5 Mbit/s
ITU G.992.3/4	ADSL2	12 Mbit/s	1.0 Mbit/s
ITU G.992.3/4 Annex J	ADSL2	12 Mbit/s	3.5 Mbit/s
ITU G.992.3/4 Annex L[1]	ADSL2	12 Mbit/s	1.0 Mbit/s
ITU G.992.5	ADSL2+	24 Mbit/s	1.0 Mbit/s
ITU G.992.5 Annex L[1]	ADSL2+	24 Mbit/s	1.0 Mbit/s
ITU G.992.5 Annex M	ADSL2+	24 Mbit/s	3.5 Mbit/s

Additionally, the non-Annex ADSL2 and ADSL2+ support an extra 256 kbit/s of upstream if the bandwidth normally used for POTS voice calls is allocated for ADSL usage.While the ADSL access utilizes the 1.1 MHz band, ADSL2+ utilizes the 2.2 MHz band.

5.21 VDSL (VERY-HIGH-SPEED DSL)

Very-high-speed DSL (VDSL) promises even higher speeds than ADSL, although over much shorter distances. Originally named VADSL (A -Asymmetric) but was later extended to support both symmetric and asymmetric. Requires one phone line and supports voice and data. It works between 0.3–1.37 kms depending on speed. It supports upstream data rate of 1.6–2.3 mbps and downstream data rate of 13–52 mbps. The following figure illustrates shows the data rate, wire size and distance.

Downstream	Upstream	Distance Feet Kms	
12.96 Mbps	1.6–2.3 mbps	4500 Feet	1.37 Kms
25.82 Mbps	1.6–2.3 mbps	3000 Feet	0.91 Kms
51.84 Mbps	1.6–2.3 mbps	1000 Feet	0.30 Kms

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5.22 RADSL (RATE-ADAPTIVE DSL)

As the name implies, rate-adaptive DSL (RADSL) modems adjust the data rate to match the quality of the twisted-pair connection. Emerging software should make this an automated process with little human intervention.

5.23 HDSL (HIGH-DATA-RATE DSL)

HDSL modem is viewed as equivalent of PCM stream(2 Mbps) and offers the same bandwidth both upstream and downstream. It can work up to a distance of 3.66 to 4.57 kms depending upon the speed required. It can deliver 2048 kbps

(a) On 2 pairs of wires, each line carrying 1168 kbps
(b) On 3 pairs of wires, each line carrying 784 kbps.

5.24 SDSL(SYMMETRIC DSL)

Symmetrical digital subscriber line (SDSL) is similar to HDSL but requires only one pair of wires. Transmission speed ranges from n x 64 kbps to 2.0 Mbps in both directions. In this the upload and download streams are of equivalent bandwidth.

PART IV: THE 4G FEATURES

5.25 THE 4G FEATURES

The rapid growth of mobile communication systems has been overwhelming. Since the mobile communication was first introduced, it has evolved into several generations namely the first (1G), second (2G) and third (3G). Even before the 3G is being fully deployed, works on the next generation has already begun. Figure 5.16 illustrates the evolution of mobile communication systems, including both the technologies and services aspects.

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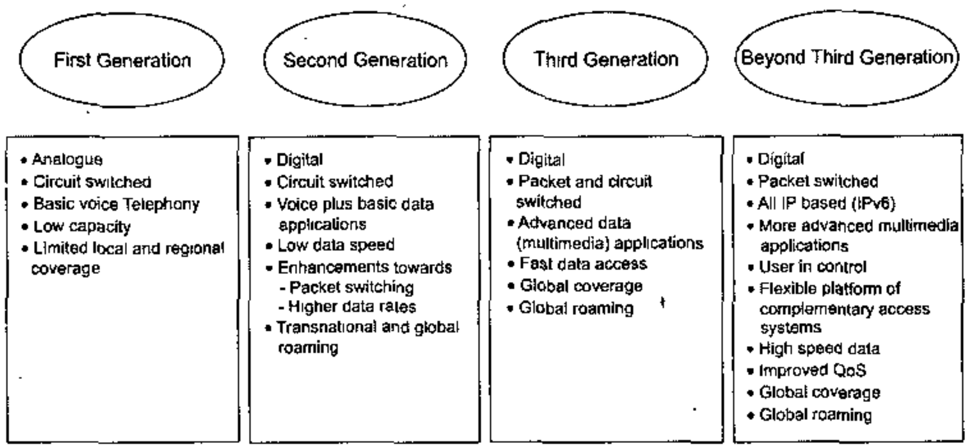

FIG. 5.16

The launch of the 3G mobile communication systems with the capability of mobile multimedia services at any time and anywhere shows how advanced has the area progressed starting from just a mere voice transaction in the first generation.

5.26 EVOLUTION OF MOBILE COMMUNICATION TECHNOLOGIES

Before the characteristics of the future wireless communication systems are discussed, it is thought appropriate to discuss briefly about the evolution of the technology.

The First Generation (1G)

The first generation of mobile systems used analogue transmission. It was introduced about three decades ago and offered basic voice telephony to and from mobile subscribers. Operated in circuit-switched mode, these systems offered voice band data transmission at a relatively low bit rate. The frequency band that were used were 450 and 800 MHz. The bands used variants of Frequency Division Multiple Access (FDMA) schemes. The 1G radio systems lack the ability to support roaming between different network operators. Consequently, it has a limited local and regional coverage.

The Second Generation (2G)

The second generation of mobile systems began to emerge in the early 1990s. Digital signal processing and transmission were introduced to replace the analogue transmission. Global System for Mobile Communication (GSM) is the most popular 2G cellular system standard. In its original form, GSM in the 900, 1800 and 1900 MHz frequency bands uses a Time Division Multiple Access (TDMA) scheme and offers a digitised speech and digital data at up to 9.6 Kbit/s. The introduction of Subscriber Identity Module (SIM) cards and the GSM Mobile Application Part (MAP) protocol enabled internetworking between different networks, allowing subscribers to roam worldwide.

Since its commercial introduction in the early 1990s, GSM has been constantly upgraded. This is manifested by the introduction of High Speed Circuit-Switched

Data (HSCSD), General Packet Radio Service (GPRS), Enhanced Data Rates for Global GSM Evolution (EDGE), Enhanced Circuit-Switched Data (ECSD) and Enhanced GPRS (EGPRS).

Essentially, the transition to third generation services has been prepared for by the introduction of high speed circuit switched data transmission using HSCSD at up to 57.6 Kbit/s, and ECSD at up to three times the HSCSD rate. Moreover, GPRS and EGPRS allow the efficient operation of "always-on" data/Internet services through packet mode transmission, realizing best case performances of more than 100 kbit/s and 384 kbit/s, respectively.

Third Generation (3G)

The standardization efforts for the third generation mobile systems began in the late 1990s. The introduction of Universal Mobile Telecommunication System (UMTS), based on Wideband Code Division Multiple Access (WCDMA) technology, is a further step towards satisfying the ever increasing demand for data/Internet services. Circuit-mode speech and data as well as packet-mode data transmission are possible with UMTS. It offers services like wireless voice, video e-mail, web browsing, videoconferencing, multimedia and e-commerce, at any time and from anywhere. The data rates is at 384 Kbit/s, with a maximum of 2 Mbit/s per customer, are made available by 3G cellular systems using WCDMA.

5.27 THE NEXT GENERATION TECHNOLOGY (4G)

The cellular networks have evolved considerably over the period. The early cellular networks suffered from many limitations and carried only speech. Today's the cellular systems are able to carry packet switched data and support various type of application like Internet browsing, email access etc. In these days, the emphasis is on integration of all kinds of services like web access, file transfer and facsimile with the traditional voice services. The cellular networks are now heading towards all IP-based network and it is likely that in future all services will be made available over IP. The 3G cellular system is a step towards standardization of the next generation mobile cellular networks widely known as 4G.

4G in a broad sense could means, connectivity across a wide range of access technologies and access networks such as Wireless Local Area Network (WLAN), Bluetooth, Wireless Personal Area Network (WPAN), CDMA, WCDMA, xDSL, PSTN etc. The term which is being used by the International Telecommunication Union (ITU) to reflect this feature is inter-working. Now the 4G technology is in the process of standardization mainly by ITU.

There have been many people and organizations who have envisioned the features of the 4G communication systems, and it can be said that the most outstanding features which have been proposed are: Personalization; Seamless Access; Quality of Service; and the last but not least IP-based systems (refer Figure 5.17).

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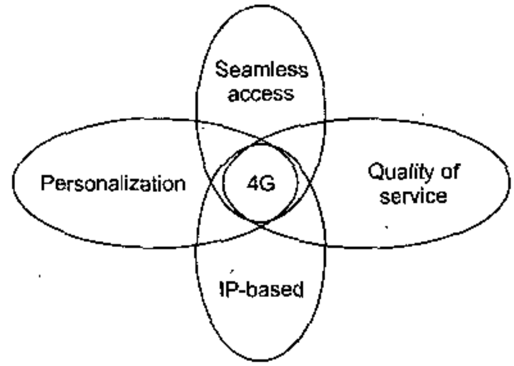

Fig 5.17 The next generation mobile communication systems features

Personalization

Much has been said about customer personalization in the mobile communication industries. Operators are struggling to provide customers with the appropriate signaling mechanisms and service discovery technologies in order to make sure that the right information is consumed at the right place and time. This is made difficult with the heterogeneity of the networks and devices in placed. It is safe to say that different customer has its own requirement and they expect to be treated differently by the operators according to their needs. It is expected that personalization will play an important role for the 4G communication systems.

One of the personalization function that the customer is expected to need is the ability to choose their own service provider from the so called "open market" scheme. It is envisioned that the service and network functions of communications systems will be decoupled which each of them is being run by different organizations. The trend of Mobile Virtual Network Operator (MVNO) nowadays is paving the way for the realization of this scheme. Decoupling of service provision from network and provision of open interfaces is also envisioned by ITU-R.

The usage of network agent as the representatives of customer and service providers when communicating with each other and also between service providers and network providers have also been much discussed. This communicating process is better known as negotiations. In his paper, Le Bodic discussed about the using of agent-based middleware in a "Digital Marketplace" environment where all the negotiations of the service provider, network provider and also the customer take place. In a digital marketplace environment, the customers will be given the chance to choose his or her service provider according to his needs. This will result in a more competitive market in the mobile communication industry.

According to Kellerer, service personalization will be a key success factor for forthcoming data and communication services. The services should also includes; exclusive support of individual customers and personal access to mobile information portals or seamless service adaptation according to the varying context of a customer. Indeed, the 4G does give strong emphasized on the customers' needs.

Seamless Access

According to Kellerer, conceptually, seamless access is somewhat similar to roaming as users can experience it today in mobile telephony networks of the second generation, in particular GSM. However, seamless access in 4G will go much beyond the roaming as it is known today and will be a much more sophisticated affair.

In the up-coming 4G, it is anticipated that the users will be provided with connectivity to heterogeneous wireless networks inter-working with one another. This is more advanced than the current context for roaming.

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At present, the user may be able to roam between similar public wireless access networks. In other words, roaming is allowed within homogeneous environment and in particular in mobile networks. Advanced mobility concepts proposed in the 4G will allow the customers to reach their personal services and preferences anywhere, anytime, and overall access networks. In order to achieve that, it was proposed in for suitable servers in the core network to handle the above mentioned issues. Inter-working wireless access networks can be categorized as horizontal (intra-system) and vertical (inter-system) handover. For this reason, service negotiation, and global roaming are the most important issues in providing seamless access to the users. According to Aretz , 4G systems will be characterized by a horizontal communication model between different access technologies to the customer terminals such as cellular, cordless, WLAN type systems, short range connectivity, broadcast systems, and wired systems. In the literature also, it was found that in order to integrate these different access technologies, a common, flexible, and expandable platform that can complement each of the technologies is needed.

That is why in the literature, it was found that a seamless IP-based core network was proposed to take up this responsibility.

Quality of Service

In a basic definition, Quality of Service or more widely known as QoS can be defined as the ability of a network to provide some consistent level of assurance for data delivery over the network with the levels being different for different classes of traffic. Traffic in a higher priority class is given more resources than traffic in a low priority class in a fair manner.

To have the QoS assurance is also part of the features for the up-coming 4G systems. According to Kellerer, it is expected that the 4G service quality will be the collective effect of the performance of all system elements in combination with the customer expectations, which determines the degree of satisfaction of the 4G customer. This is due to the varying QoS requirements and the heterogeneous nature of multimedia traffic *e.g.*, time-varying transmission rates and also limited radio resources. Furthermore, the ability of a user to roam between different wireless access networks making the task of providing QoS during the traffic lifetime much tougher.

Developers would need to do much more work to address end-to-end QoS. They may need to modify many existing QoS schemes, including admission control, dynamic resource reservation, and QoS renegotiation to support 4G user's diverse QoS requirements. The other challenge is to find a QoS mechanism that can address fairness and fair queuing schemes.

IP-Based

It has been observed that the most talked about feature for the 4G systems is the switching to an all-IP or IP-based systems. By using IP-based system, data will be transferred over the access networks by using IP packets. In the telecommunication world, this is known as packet-switched. The definition of packet-switched network itself means that the networks move the data in separate,

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small blocks called packet, based on the destination address which contains in the packet. Each packet will contain certain information such as destination address, source address etc., which are needed in order for the packet to reach its destination. When received, the packets will be reassembled in the proper sequence to make up the original message. In a circuit-switched system, a dedicated point-to-point connection is needed for each call. This means that, the line has to be reserved for a particular call in order for the call to take place. The original message was not being separated into small pieces, as they were in the packet-switched system.

5.28 IP-BASED MOBILE COMMUNICATION SYSTEMS

The term "All-IP" is often used in conjunction with mobile networks and has come to mean a number of different things to different people. The phrase could mean that all the traffic is encapsulated, within IP packets in the network core; as in UMTS Release 5, where IP packets from mobile are only reconstructed at the SGSN and then tunnelled to the Internet. It can be defined as an access network that transports IP packets and provides QoS and mobility support.

Whatever it is, all of the definition has the same objective, *i.e.*, to provide users with data, voice and multimedia services etc., over an IP bearer. Such network architectures are referred as "All-IP" networks. As advances are being made towards this goal in the different standardization areas, it is becoming obvious that the core network of the next generation mobile network will be pure IP-based, most likely leading to a network convergence between the various telecommunication systems as depicted in Figure 5.3.

Apart from that, it is also envisaged by many researchers that the next generation access network will also work hand in hand with the legacy networks as well, which will form a truly IP-based network, *i.e.*, both the core and the access networks are running on IP. This concept will allow the deployment of a unified backbone, federating different access technologies *e.g.*, narrow/broadband, fixed/ wireless, and public/private access networks.

Although there are some disadvantages of using IP packet in the delivery of data, the advantages of it have overwhelmed the user. It is anticipated that in the future, the disadvantages will be solved and IP is going to play an important role in the next generation of mobile communication systems. In the following sub-sections, the advantages of IP will be discussed.

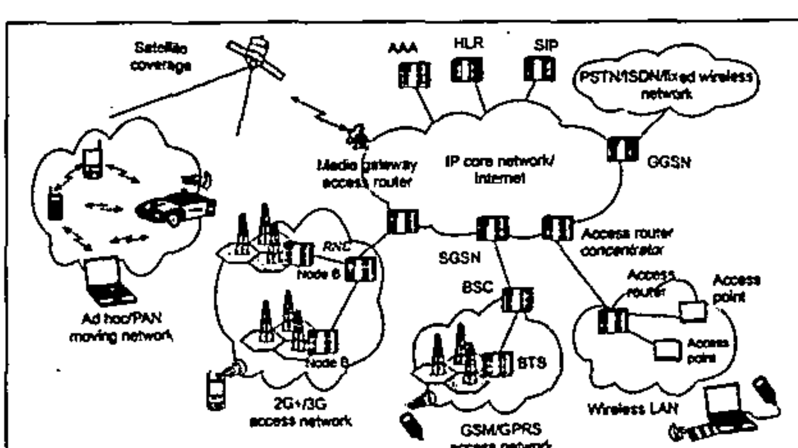

FIG. 5.18 Open All-IP network architecture

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Seamless Access Enabler

Seamless Access is one of the features for the 4G systems. IP technology can facilitate this feature with its capability to support different access technologies such as 802.11, WCDMA, Bluetooth, and HyperLAN. The flexibility comes with the concept of having an IP core network supporting various IP access networks. The beauty of IP technology is that its core and access networks can evolve independently. By having this, the developer for access networks can come up with new technology without having to worry with how to connect to other access networks.

Efficiency

With the IP packet delivery system, the scarce radio spectrum issue can be tackled. The usage of packet-switched network for the non voice services in the GPRS systems has proved that packet delivery system is more efficient than a circuit-switched network.

With the number of users increasing drastically each year, network and service providers need to cater for this problem, and IP packet delivery systems can give a solution to this problem. In the IP networks, the networks are fully utilized, all applications can share the same connection and QoS levels can still be guaranteed. This is in contrast with the conventional mobile communication systems which are using the circuit-switched networks.

Cost

IP technology has been around for quite sometime now. The foundation of the network is in place and a lot of researches are taking place for the betterment of the system. By using IP, the users do not have to think about setting up a new core network as such. Everything is in place. That is why it is cheaper by using the IP in the mobile communication systems. It is expected that the 4G equipment costs are four to ten times cheaper than equivalent circuit-switched equipment for 2G and 3G wireless infrastructure. An open system of wireless IP environment would probably further reduce costs for service providers by ushering in an era of real equipment interoperability.

5.29 CONCLUSION

This unit presents the trends for future mobile communication systems. From the discussion, it can be concluded that the next generation of mobile communication system is migrating towards an IP-based mobile communication systems.

SUMMARY

- These vertical 'monolithic' networks have basically their own fundamental plans with some exceptions: PSTN and ISDN are strongly related, and the transmission facilities are often shared between different networks.
- Next Generation Networks (NGN) are the next step in world communications. NGNs are the culmination of 100 years of telecommunications evolution, combining the scalability and reliability of the public telephone network with the reach and flexibility of the Internet.
- The global telecommunications infrastructure has evolved over the past 100 years. The last two decades, however, have heralded seminal change that has accelerated this evolution manifold.

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- A Next Generation Network (NGN) is a packet-based network able to provide services including Telecommunication Services and able to make use of multiple broadband, QoS-enabled transport technologies and in which service-related functions are independent from underlying transport-related technologies. It offers unrestricted access by users to different service providers.
- Next generation networks will for the most part be high speed packet based networks capable of delivering a multitude of broadband services.
- The demands it has placed upon the public network, especially the access network, are great. However, technological advances promise big increases in access speeds, enabling public networks to play a major role in delivering new and improved telecommunications services and applications to consumers.
- A definition to broadband is a must as different service providers defines in their own terms and context.
- DMT is basically a multicarrier modulation technique. DMT spread the original spectrum of the input signal over numerous sub-channels each of which carries a fraction of the total information.
- Very-high-speed DSL (VDSL) promises even higher speeds than ADSL, although over much shorter distances.
- The rapid growth of mobile communication systems has been overwhelming. Since the mobile communication was first introduced, it has evolved into several generations.
- The cellular networks are now heading towards all IP-based network and it is likely that in future all services will be made available over IP. The 3G cellular system is a step towards standardization of the next generation mobile cellular networks widely known as 4G.
- The term "All-IP" is often used in conjunction with mobile networks and has come to mean a number of different things to different people.

REVIEW QUESTIONS

1. What do you understand by Network?
2. What do you understand by telecom network?
3. What is a vertical network?
4. What do you understand by Next Generation Networks?
5. What are different types of networks?
6. Write short notes on
   (a) The traditional Public Switched Telephone Network (PSTN)
   (b) The Mobile Telecom Network.
7. Discuss definition of next generation network by ITU.
8. What are different features of NGN?
9. What are different applications of NGN?
10. What are different characteristics of next generation networks?
11. What do you broadband understand?
12. What are different broadband access techniques?
13. What is ADSL?
14. Discuss 4G of telecom and its features.

FURTHER READINGS

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