DIGITAL SWITCHING AND TELECOM NETWORKS

PEEC5404

7th semester, ETC

Objective of the Module- I:

- **To learn about different basic components that are used in telephone exchanges**
- **Difference between an automatic exchange and manual exchange**
- **Learn about different types of electronic exchanges and about the software architecture used in an electronic exchange.**
- **To differentiate between a single stage and a multistage network.**
- **To understand the advantages of multistage network over a single stage network.**
- **Design of multistage network to reduce blocking of calls.**
- **Design of multistage network to reduce the number of switching matrices**
- **To teach different types switching techniques that are used in exchanges such as time division time switching, time division space switching and combination of both types of switching**

MODULE-1:

1. INTRODUCTION:

Telecommunication networks carry information signals among entities, which aregeographically far apart. An entity may be a computer or human being, a facsimile machine, atele-printer, a data terminal and so on. The entities are involved in the process of informationtransfer, which may be in the form of a telephone conversation (telephony) or a file transferbetween two computers or message transfer between two terminals etc.

A switch transfers signals from one input port to an appropriate output. A basic problem is then how to transfer traffic to the correct output port .In the early telephone network, operator's closed circuitsmanually. In modern circuit switches this is done electronically in digital switches. If no circuit is available when a call is made, it will be blocked (rejected). When a call is finished a connection teardown is required to make the circuit available for another user.

Fundamentals of switching systems:

Types of communication transmission mode:

Simplex : one way communication ex: Radio

Half Duplex : Two way communication shared by single channel ex: walkie Talkie

Full Duplex : Two way communication simultaneously ex: Telephone

Therefore, telephone comes under the Full Duplex type of communication.

Point – Point Links/Fully Connected Network/Bell Proposed Network:

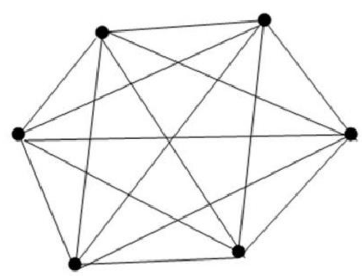

Fig 1.1. Point to Point link [1]

To connect 'N' Points the number of Links required is as below:

Problem: Calculate the Number of links required to fully connect 5000 links and the numberof additional links required to fully connect 5001 links

Solution: to connect 5000 points, numbers of links required are: 12497500

To connect 5001 points, number of links required: 12502500

Therefore, the additional links required to connect extra 1 point on a 5000 points network of fully connected are: **5000**

From the above problem it is understood that it is highly impossible to connect large number points (telephones) as fully connected network/point-point network. To resolve this problem "Telephone Exchange" came into existence

2. TELECOMMUNICATION NETWORKS:

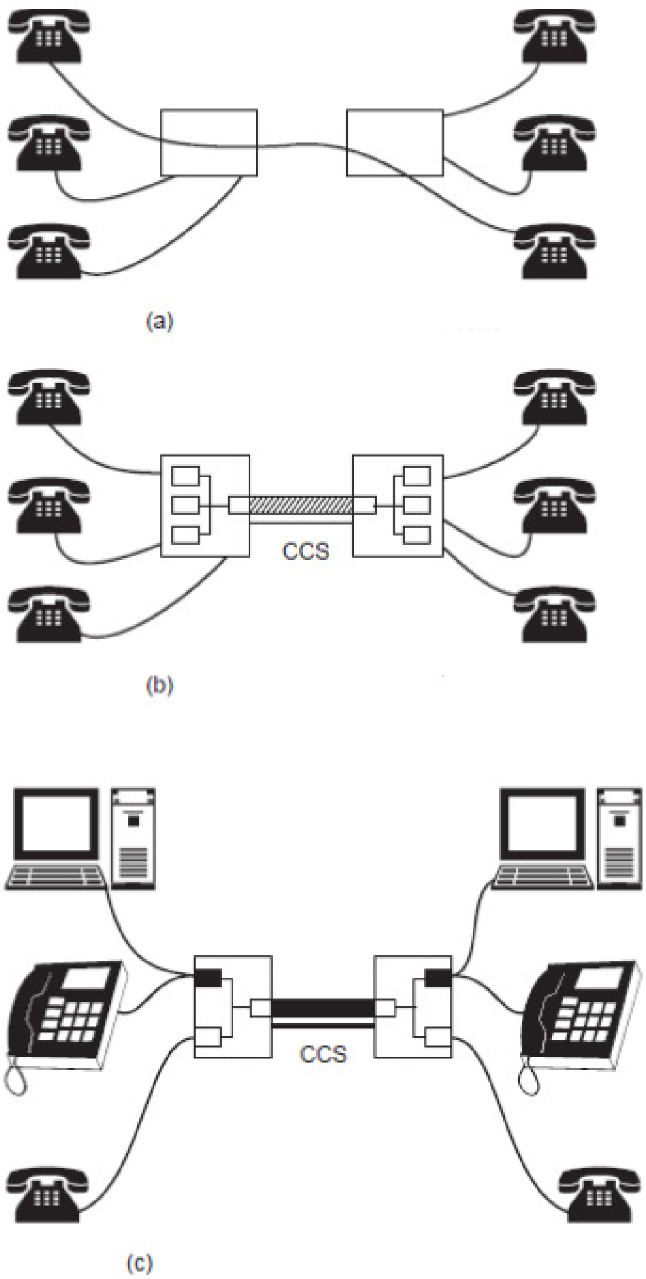

Fig 1.2 Various telephone networks[6]

(a) telephone network around 1890 , (b) telephone network around 1988

(a) telephone networks after 1990 with ISDN

The electromechanical switching systems have been replaced by computer controlled switching systems referred to as stored program control (SPC). In SPC, switching is controlled by software program. The first computer controlled switch was introduced in 1960. Till 1965, computer controlled switching used transistors and printed circuit technology. Since 1965 switchingis based on microprocessors.

3. SIGNAL CHARACTERISTICS:

Telecommunication is mainly concerned with the transmission of messages between two distant points. The signal that contains the messages is usually converted into electrical waves before transmission. Our voice is an analog signal which has amplitude and frequency characteristic.

Voice frequencies: The range of frequencies used by a communication device determines the communication channel, communicating devices, and bandwidth or information carrying capacity.

Table 1.1. Bandwidth requirements of various applications

Type	*Bandwidth*	*Bit Rate*
Telephone (speech)	300—3400 Hz	—
Music	50 Hz—16 kHz	—
Facsimile	40 kHz	—
Broadcast televison	0—55 MHz	—
Personal communication	—	300 to 9600 bits/sec
E—Mail transmission	—	2400 to 9600 bits/sec
Digitized voice phone call	—	6400 bits/sec
Digital audio	—	1 to 2 M bits/sec
Compressed video	—	2 to 10 M bits/sec
Document imaging	—	10 to 100 M bits/sec
Full motion video	—	1 to 2 G bits/sec

Speech spectrum: The telephone channel over which we wish to send data are designed to transmit electrical oscillations (microphone converts sound into equivalent number of electrical oscillation) of voice.

Decibels: The decibel is a valuable unit for telecommunication because losses or gains in signal strength may be added or subtracted if they are referred to in decibels. The power ratio is expressed as-

$$G = 10 \log_{10} \frac{P_2}{P_1}$$

Voltage and current level can be quoted in decibel as follows

$$G = 10 \log_{10} \frac{P_2}{P_1} = \frac{V_2 I_2}{V_1 I_1} = \frac{I_2^2 R}{I_1^2 R}$$

$$G = 10 \log_{10} (I_2/I_1)^2$$

$$G = 20 \log (I_2/I_1)$$

Similarly in voltage ratio,

$$G = 20 \log_{10} V_2/V_1$$

Example 1.1. If input power is 16 μW and output power is 30 mW, find the power ratio and express it in decibel and nepers.

$$\text{Power} = \frac{P_2}{P_1} = \frac{30 \times 10^{-3}}{16 \times 10^{-6}} = 1875 = 1.875 \times 10^3$$

Power in decibel, $G = 10 \log_{10} 1.875 \times 10^3 = 32.73$ dB

Power in nepers, $G = 3.76$ N.

4. ELEMENTS OF COMMUNICATION SWITCHING SYSTEM

The purpose of a telecommunication switching system is to provide the means to pass information from any terminal device to any other terminal device selected by the originator.Telecommunication system can be divided into four main parts. They are

1. End system or Instruments

2. Transmission system

3. Switching system

4. Signaling.

End Systems or Instruments: The end system or instruments are transmitters or receivers that are responsible for sending information or decoding or inverting received information or message into an intelligible message. End systems in the telephone network have evolved from analog telephones to digital handsets and cellular phones. However, endless arrays of other devices are being attached to telephone lines, including computer terminals used for data transmission. Fig. 1.3 shows some of the end instruments.

Transmission System: Signals generated by the end system or the instruments should be transported to the destination by some means. The transmission on links conveys the information and control signals between the terminals and switching centers.

In general a communication path between two distinct points can be setup connecting a number of transmission lines in tandem. The transmission links include two-wire lines, coaxial cables microwave radio, optical fibers and satellites. Functionally, the communication channels between switching system are referred to as trunks.

Switching System: The switching centers receives the control signals, messages or conversations and forwards to the required destination, after necessary modification (link amplifications) if necessary. A switching system is a collection of switching elements arranged and controlled in such a way as to setup a communication path between any two distant points.

Signaling Systems:A signaling system in a data communication networks exchangessignaling information effectively between subscribers. The signaling systems are essentialbuilding blocks in providing the ultimate objective of a worldwide automatic telephone servicesstandardized. Signaling provides the interface between different national systems. Theintroduction of signaling system was the big step in improving the PSTN.

The consultative committee on international telegraphy and telephony (CCITT) based in Geneva, recommended seven formats related to signaling.

5. CRITERIA FOR THE DESIGN OF TELECOMMUNICATION SYSTEM

The design for telephone switching center or equipment requirement in a telecommunication system is determined on the basis of the traffic intensity of the busy hour. The traffic intersity is defined as the product of the calling rate and the average holding time. The busy hour is defined as that continuous sixty-minute period during which the traffic intensity is highest.

Otherwise the average holding time is the average duration of occupancy of traffic path by a call.

Grade of Service: In telephone field, the so called busy hour traffic is used for planning purposes. Once the statistical properties of the traffic are known, the objective for the performance of a switching system should be stated.

GOS is a measure of congestion expressed as the probability that a call will be blocked or delayed. Thus when dealing with GOS in traffic engineering, the clear understanding .of blocking criteria, delay criteria and congestion are essential.

Blocking criteria: If the design of a system is based on the fraction of calls blocked (the blocking probability), then the system is said to be engineered on a blocking basis or call loss basis. Blocking can occur if all devices are occupied when a demand of service is initiated.

Delay criteria: If the design of a system is based on the fraction of calls delayed longer than a specified length of time (the delay probability), the system is said to be a waiting system or engineered on a delay basis. Delay criteria are used in telephone systems for the dimensioning of registers. In waiting system, a GOS objective could be either the percentage of calls which are delayed or the percentage, which is delayed more than a certain length of time.

Congestion: It is the condition in a switching center when a subscriber cannot obtain a connection to the wanted subscriber immediately. In a circuit switching system, there will be a period of congestion during which no new calls can be accepted. There are two ways of specifying congestion.

1. **Time congestion:** It is the probability that all servers are busy. It is also called the probability of blocking.

2. **Call congestion:**It is the proportion of calls arising that do not find a free server. Call congestion is a loss system and also known as the probability of loss while in a delay system it is referred to as the probability of waiting.

Measure of GOS:GOS is expressed as a probability. The GOS of 2% (0.02) mean that 98% of the calls will reach a called instrument if it is free. Generally, GOS is quoted as P.02 or simply P02 to represent a network busy probability of 0.02. GOS is applied to a terminal-to-terminal connection. For the system connection many switching centers, the system is generally broken into following components.

(i) An internal call (calling subscriber to switching office)

(ii) An outgoing call to the trunk network (switching office to trunk)

(iii) The trunk network (trunk to trunk)

(iv) A terminating call (switching office to called subscriber)

The GOS of each component is called component GOS. The GOS for internal calls is 3 to 5%, for trunk calls 1-3%, for outgoing calls 2% and for terminating calls 2%. The overall GOS of a system is approximately the sum of the component grade of service. In practice, in order to ensure that the GOS does not deteriorate disastrously if the actual busy hour traffic exceeds the mean; GOS are specified 10% or 20% more of the mean.

6. FUNDAMENTALS FOR THE DESIGN OF TELECOMMUNICATION NETWORK:

A telephone network is composed of a variety of all processing equipment, interstate switching links and inters office trunks. Because of the random nature of the call request, the design ofequipment switching links and trunks are quite difficult. Thus, the traffic analysis is thefundamental request for the design of cost effective, efficient and effective configuration ofnetworks. The effectiveness of a network can be evaluated interms of how much traffic itcarries under normal or average loads and how often the traffic volume exceeds the capacity ofthe network.Fundamental problem in the design of telecommunication networks concerns thedimensioning of a route. To dimension the route, volume of traffic required grade of serviceand capacity (in bits per sec) must be known.

Traffic: In telecommunication system, traffic is defined as the occupancy of the serverin the network. There are two types of traffic viz. voice traffic and data traffic. For voice traffic,the calling rate is defined as the number of calls per traffic path during the busy hour. In a day,the 60 minutes interval in which the traffic is highest is called busy hour (BH).

Average occupancy: If the average number of calls to and from a terminal during aperiod T second is 'n' and the average holding time is 'h' seconds, the average occupancy of theterminal is given by

$$A = \frac{nh}{T}$$

The average occupancy is also referred as traffic flow of traffic intensity. The internationalunit of telephone traffic is the Erlang.

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7.DISTRIBUTED&CENTRALIZED SWITCHING SYSTEM

Distributed Model

A simplest way of structuring the telecommunication switching is the terminal-toterminalconnection. This kind of switching is called distributed switching and applied only to small telephone system. Some examples of distributed switching are shown here. Fig. 1.3 shows the full interconnection of five terminals.

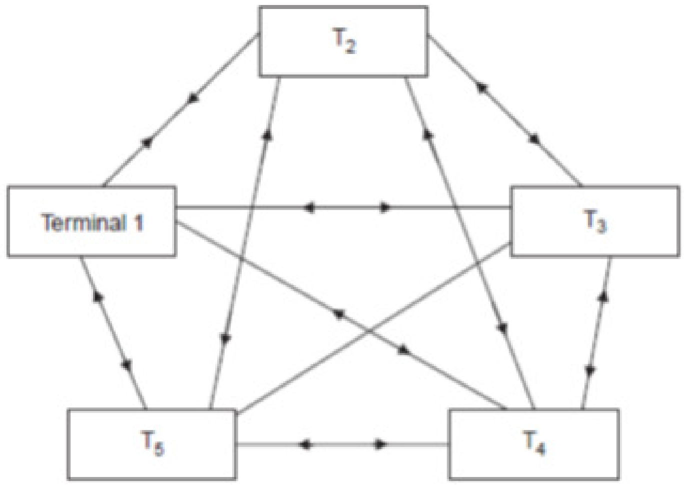


Fig 1.3 Distributed model[4]

Each terminal has two kinds of switches, one to make required link and other to connect a link to receive a call. By this method, for N terminals, the numbers of links required are 1 2 N (N – 1) Fig. 1.4 shows the interconnection of four terminals but only with 4(N) links. Here each terminal is connected permanently to one channel and all other terminals may be accessed by operating a switch. Also it removes the need to connect a terminal to a link for an incoming cal.

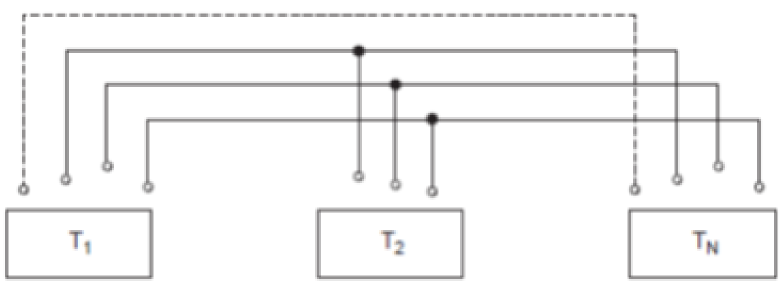


Fig 1.4 Centralized model with (N-1) links [5]

In this arrangement, a calling terminal sends a calling signal to indicate the called terminal to which the terminal should be switched in order to receive the call. The recognition of an incoming call and switching operation may be performed automatically in system using coded signals.

Centralized Model:

The distributed system cannot be extended to large terminal cases and the increased geographical separation of terminals. A simple centralized system, which reduces the average length of transmission link, and hence the transmission cost is shown in Fig. 1.8. But this system increases the total switching costs. Introducing more local centers instead of one national center switching machine can further reduce the transmission cost. Two local centers are connected by links called trunk. A trunk in telephone system is a communication path that contains shared circuits that are used to interconnect central offices. Fig. 1.5 shows the telecommunication system for short distances, with two exchanges (switching offices).

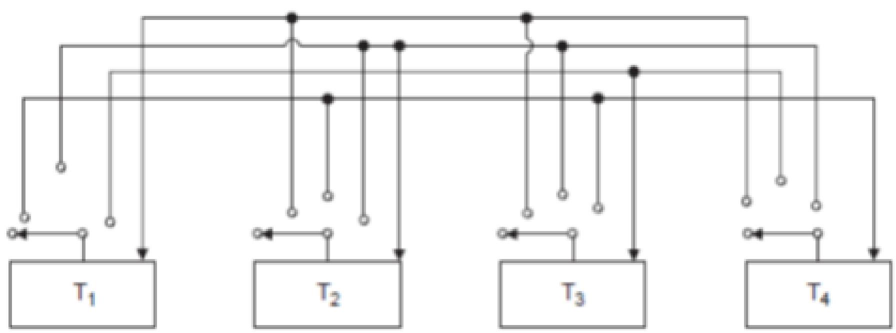


Fig 1.5 Type of centralized model [5]

7.1 Classification of Switching System:

In early days, the human exchange provided switching facilities. In manual exchanges, a humanoperator and the elements like switches, plugs and sacks were used to connect two subscribers.Around 1890's many electromechanical switching devices were introduced. Till 1940, differentelectromechanical switching system were invented, of which strowger switching system andcross bar switching system were still popular. The later invention of electronic switching system(ESS) which uses stored program control (SPC) and computer controlled switching systemsare presently dominating the worldwide exchanges. Fig. 2 below shows the classification of switchingsystem.

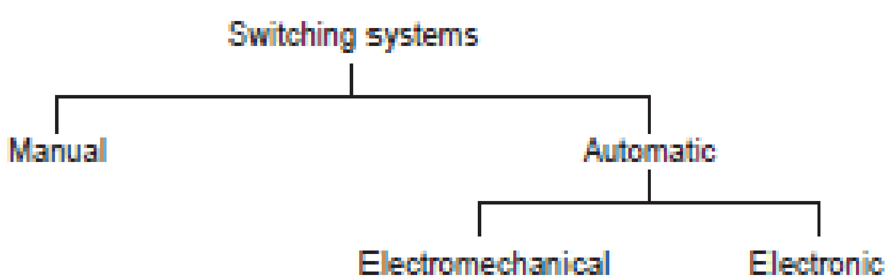


Figure 2. Classification of the switching system [1]

8. BASICS OF SWITCHING SYSTEM

i) Functions of Switching System

The switching office performs the following basic functions irrespective of the system whetherit is a manual or electromechanical or electronic switching system. Fig. 4.4. shows the simplesignal exchange diagram

ii)**Identity.**The local switching center must react to a calling signal from callingsubscriber and must be able to receive information to identify the required destination terminalseize.

iii)**Addressing.**The switching system must be able to identify the called subscriberfrom the input information (train of pulses or multiple frequency depends on the dialing facility).The address may be in same local center or some other exchange. If the terminal or trunkgroup is busy, a suitable signal must be returned to the calling subscriber. If more than onefree circuit, particular one will be selected.

iv)**Finding and Pathsetup.**Once the calling subscriber destination is identified andthe called subscriber is available, an accept signal is passed to the switching system and callingsubscriber. Based on the availability, suitable path will be selected.

v)**Busy testing.**If number dialed by the calling subscriber is wrong or the calledsubscriber is busy (not attending the phone) or the terminal may be free (lifting the phone) butno response (not willing to talk or children handling), a switching system has to pass acorresponding voice message or busy tone after waiting for some time (status).

vi)**Supervision.**Once the path is setup between calling and called subscriber, it shouldbe supervised in order to detect answer and clear down conditions and recording billinginformation.

vii)**Clear down.**When the established call is completed, the path setup should bedisconnected. If the calling subscriber keeps the phone down first, the signal called clear forwardis passed to the switching system. If the called subscriber keeps the phone down first, a signalcalled clear backward signal is passed to the switching system. By clear signal, the switchingsystem must disconnect the path setup between calling and called subscriber.

viii) **Billing.**A switching system should have a mechanism to meter to count the numberof units made during the conversation. The cumulative number of units made for a particularduration by the calling subscriber is calculated. This information and if any should be sent tothe called subscriber.

8.1 Requirements of Switching System

All practical switching system should satisfy the following requirements for the economic useof the equipment’s of the system and to provide efficient service to the subscribers. Depends onthe place (Rural or town, big town, city or big cities). The local exchange located, the serviceprovided to the subscriber may vary. Some important requirements are discussed briefly.

High availability. The telephone system must be very reliable. System reliability canbe expressed mathematically as the ratio of uptime to sum of the uptime and down time. The uptime is the total time that the system is operating satisfactorily and the down time is thetotal time that is not. In telephone switching networks, the availability or full accessibility ispossible if all of the lines are equally accessible to all incoming calls.The full accessibility is also defined as the capacity or number of outlets of a switch toaccess a given route. If each incoming trunk has access to a sufficient number of trunks oneach route to give the required grade of service is known as limited availability. The availabilityis defined as

$$A = \frac{\text{Uptime}}{\text{Uptime} + \text{down time}}$$

Also
$$A = \frac{\text{MTBF}}{\text{MTBF} + \text{MTTR}}$$

where, MTBF = Mean time between failure

MTTR = Mean time to repair.

The unavailability of the system is given by

$$U = 1 - A < \frac{\text{MTTR}}{\text{MTBF} + \text{MTTR}}$$

9. STORED PROGRAM CONTROL (SPC) EXCHANGE

In the strowger's step by step switching system and crossbar switching system,electromechanical components were used for both switching andcontrol elements. In 1965, Bell system installed the first computer controlled switching systemwhich uses a stored program digital computer for its control functions. The SPC conceptspermits the features like abbreviated dialing, call forwarding, call waiting etc. The SPC providessignificant advantages to end users. The SPC enables easier number changes, automated calltracing message unit accounting (for billing) etc.

Basic of SPC

Two types of SPC switching system

(1)Electro-mechanicalSwitching

SPC+Electro-mechanical switching network

(2)ElectronicSwitching

SPC+Electronic switching network

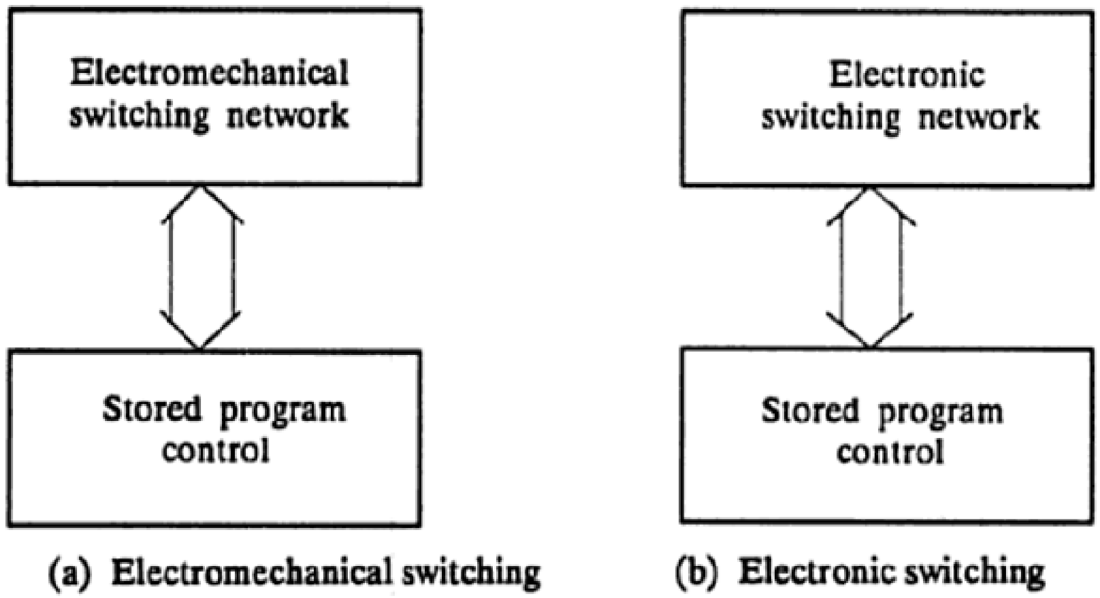


Fig.3 Electronic space division switching system [1]

9.1 Organization of SPC:

CENTRALIZED SPC

It finds broadly application in early SPC switching systems.

DISTRIBUTED SPC

It is Gaining popularity in modern switching systems.

Early electronic switching systems are centralized SPC exchanges and used a single processor to perform the exchange functions. Presently centralized exchanges uses dual processor for high reliability.

Concept

„All the control equipment is replaced by a single powerful processor.

†Configuration of centralized SPC

„Typical organization

„Redundant configuration

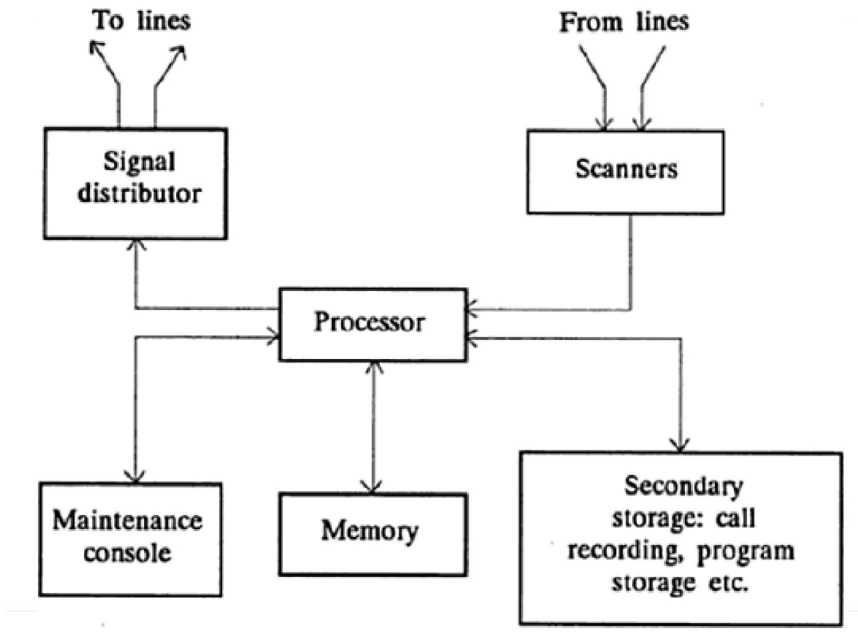


Fig. 3 (a) Centralized SPC organization [1]

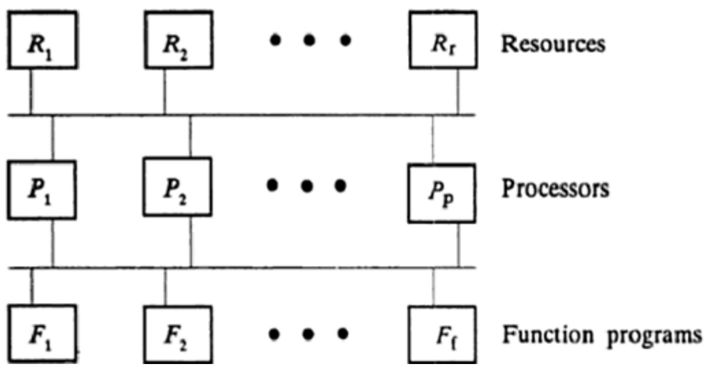


Fig. 3 (b): A redundant Centralized SPC control structure [1]

Operation modes in redundant configuration (e.g. dual processor)

- **Standby mode**
- **,,Synchronous duplex mode**
- **,,Load sharing mode**
- **Standby mode**

How does it work?

In this mode, any one of the processors will be active and the rest is standby. The standby processor is brought online only when the active processor fail. This mode of exchange uses a secondary storage common to both processors. The active processor copies the status of the system periodically and stores in axis secondary storage. In this mode the processors are not connected directly. In secondary storage, programs and instructions related to the control functions, routine programs and other required information are stored.

„All processors have the same capability to control the switching procedure.

„One processor is active and the other is on standby, both hardware and software wise.

„The standby processor is brought online only when the active processor fails.

How does the standby processor take over the control properly?

„State of the exchange system should be clear to the standby processor as its starting point.

Which of the subscribers are busy or free?

Which of the trunks are busy or free?

Which of the paths are connected through the switching network?

Reconstitution of the state

Scanning:

- ✓ **The standby processor scans all status signals as soon as it is brought into operation.**
- ✓ **Only the calls which are being established at the time of failure are disturbed.**
- ✓ **Only feasible for small exchanges.**
- ✓ **Shared secondary storage: popular.**

Shared secondary storage:

- ✓ **The active processor copies system status into a secondary storage periodically, say every 5 seconds.**
- ✓ **As soon as a switchover occurs, the online processor loads the most recent update of the system status from the secondary storage and continues the operations.**
- ✓ **Only the calls which changed status between the last update and the failure are disturbed.**
- ✓ **Feasible for large exchanges.**

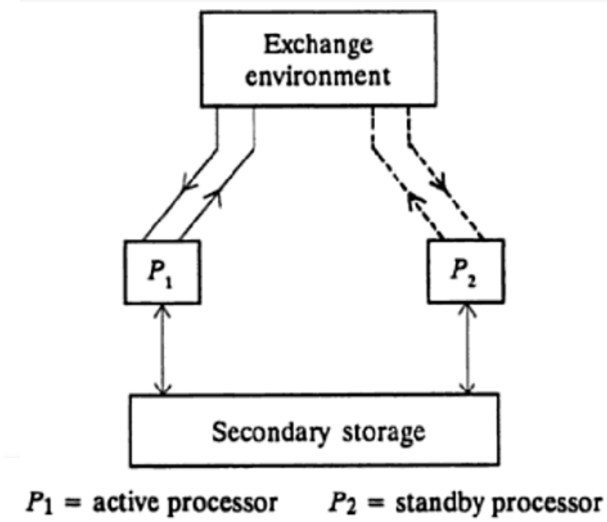


Fig. 3 (c): Standby dual processor configuration [1]

9.2 Synchronous duplex mode

How does it work?

In this mode, the processors $p1$ and $p2$ are connected together to exchange instructions and controls. Instead of a secondary storage common to P1 and P2, separate memory M1 and M2 are used. These processors are coupled to exchange stored data. This mode of operation also uses a comparator in between $p1$ and $p2$. The comparator compares the result of the processors.

During normal operation, both processor receives all the information from the exchange and receives related data from their memories. Although only one processor actually controls the exchange and remaining is in synchronism with first one. If a mismatch occurs, the fault is identified by the comparator, and the faulty processor is identified by operating both individually. After the rectification of fault, the processor is brought into service.

- ✓ Both two processors execute the same set of instructions.
- ✓ One of the processor actually controls the exchange.
- ✓ The results from two processors are compared continuously by a comparator.
- ✓ If the results match, the system works normally. Otherwise, a fault occurs, a check-out program is run independently in both two processors to determine which one is faulty.
- ✓ The faulty processor is taken out of service, and the other one works independently.

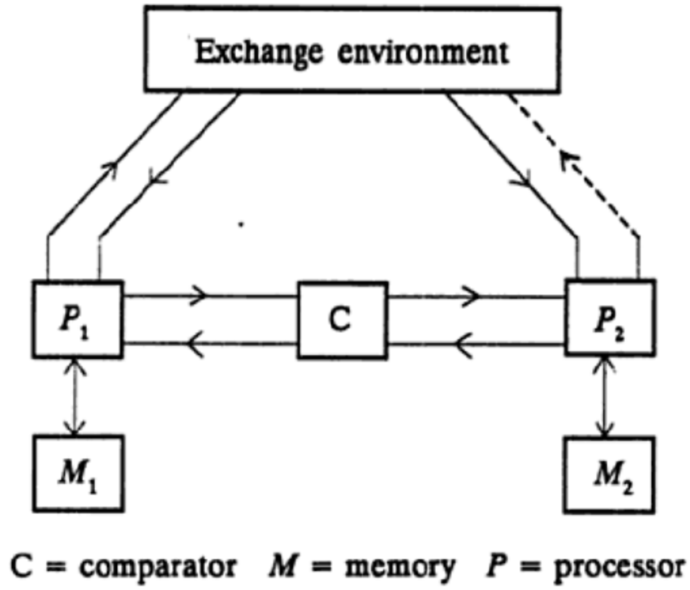


Fig. 3 (d): Synchronous duplex operation [1]

9.3 Synchronous duplex mode

- ✓ In case of transient failure of comparator, there are three possibilities exist:
- ✓ Continue with both processors.
- ✓ Take out the active processor and continue with other processor.
- ✓ „Continue with the active processor but remove the other processor from service.

9.4 Load sharing mode

How does it work?

- ✓ **„** Both two processors have access to entire exchange environment. Each of them has independent memories for redundancy purpose.
- ✓ „ Both two processors are active simultaneously and share the load and the resources dynamically.
- ✓ „ An incoming call is assigned randomly or in a predefined order to one of the processors which then handles the call right through completion.
- ✓ „ Inter-processor links are configured for processors to exchange information needed for mutual coordination and verifying the 'state of health' of the other.

If a processor fails, the other processor takes over the entire load including the calls already set up by the failing processor.

Exclusion mechanism in resource sharing

- ✓ „The processors should not seek the same resource at the same time.
- ✓ „Implementation: hardware & software.

In this mode, the comparator is removed and alternatively an exclusion device (ED) is used. The processors call for ED to share the resources, so that both the processors do not seek the same resource at the same time. In this mode, both the processor are active simultaneously and share the resources of exchange and the load dynamically. If one processor fails, with the help of ED, the other processor takes over the entire load of the exchange. Under normal operation, each processor handles one half of the calls on a statistical basis. However the exchange operator can vary the processor load for maintenance purpose.

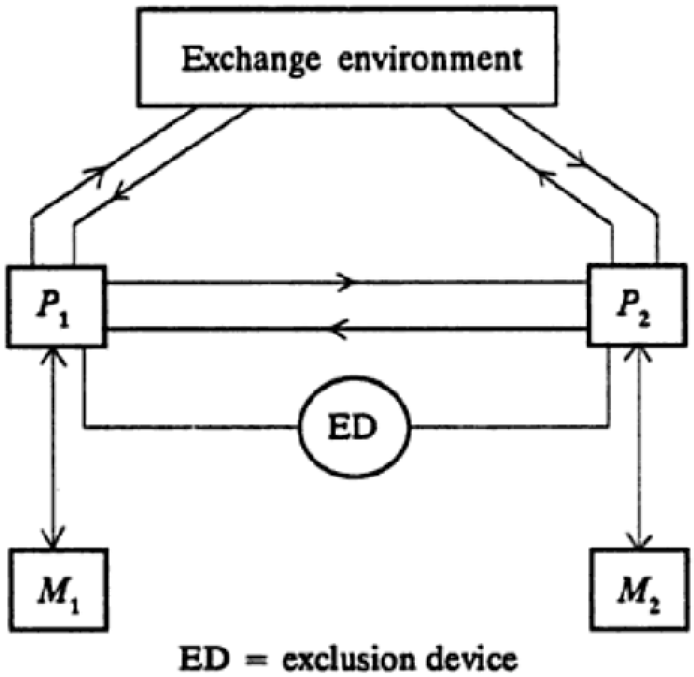


Fig. 3 (e): Load sharing configuration.[1]

Traffic distribution between processors

Load sharing increases the effective traffic capacity by 30 percent compared with synchronous duplex.

Single processor. Availability $A = \frac{MTBF}{MTBF + MTTR}$ ...(4.4)

where MTBF = Mean time between failures

MTTR = Mean time to repair

Unavailability = 1 – A

$$U = 1 - \frac{MTBF}{MTBF + MTTR}\,;\; U = \frac{MTTR}{MTBF + MTTR} \qquad ...(4.5)$$

If MTBF >> MTTR, $U = \frac{MTTR}{MTBF}$ ...(4.6)

Dual Processor. A dual processor system is said to have failed only when both processor fails and the total system is unavailable. The MTBF of dual processor is given by

$$(MTBF)_D = \frac{(MTBF)^2}{2MTTR} \qquad ...(4.7)$$

where $(\mathrm{MTBF})_D$ = MTBF of dual processor

MTBF = MTBF single processor

Availability $$A_D = \frac{(\mathrm{MTBF})_D}{\mathrm{MTTR} + (\mathrm{MTBF})_D}$$

Substituting $(\mathrm{MTBF})_D$ in the above equation, we have

$$A_D = \frac{(\mathrm{MTBF})^2 / 2\mathrm{MTTR}}{\mathrm{MTTR} + \dfrac{(\mathrm{MTBF})^2}{2\mathrm{MTTR}}}$$

$$A_D = \frac{(\mathrm{MTBF})^2}{(\mathrm{MTBF})^2 + 2(\mathrm{MTTR})^2}$$

Unavailability $$U = 1 - A_D = 1 - \frac{(\mathrm{MTBF})^2}{(\mathrm{MTBF})^2 + 2(\mathrm{MTTR})^2}$$

$$= \frac{2(\mathrm{MTTR})^2}{(\mathrm{MTBF})^2 + 2(\mathrm{MTTR})^2}$$

If MTBF >> MTTR, $$U_D = \frac{2(\mathrm{MTTR})^2}{(\mathrm{MTBF})^2}$$

Distributed SPC

The introduction of distributed SPC enabled customers to be provided with a wider range of services than those available with centralized and electromechanical switching system. Instead of all processing being performed by a one or two processor in centralized switching, functions are delegated to separate small processors (referred as regional processors). But central processors are still required to direct the regional processors and to perform more complex tasks. The distributed SPC offers better availability and reliability than the centralized SPC. Entire exchange control functions may be decomposed either horizontally or vertically for distributed processing. In vertical decomposition, the exchange environment is divided into several blocks and each block is assigned to a processor that performs all control functions related to that block of equipment. In horizontal decomposition, each processor performs one or some of the exchange control functions. Figure shows the distributed control where switching equipment is divided into parts, each of which has its own processor.

CONCEPT OF DISTRIBUTED SPC

The control functions are shared by many processors within the exchanges.

- ✓ ***Background***
- ✓ **Low cost processors**

Advantages

- Better Availability
- Better Reliability

Decomposition of Control Functions

- **Vertical decomposition**

The exchange environment is divided into several blocks.

Each block is assigned to a processor.

A processor performs all control functions related to the corresponding block.

The processor in each block may be duplicated for redundancy purposes.

Obviously, the control system consists of a number of control units.

The modular structure is flexible for system expanding.

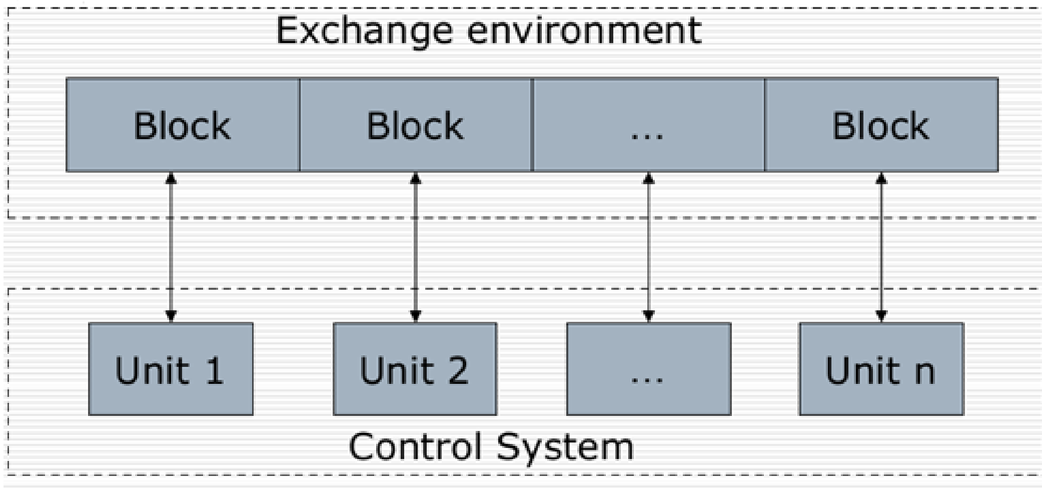


Fig. 3 (f): Vertical decomposition[1]

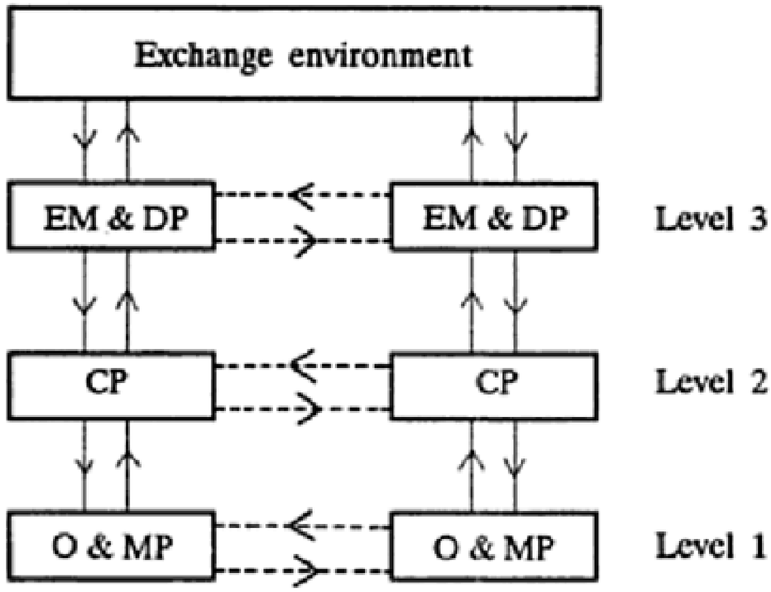
- **Horizontal decomposition**

✓ The control functions are divided into groups, e.g. event monitoring, call processing, and O&M functions.

✓ Each processor performs only one or some of the exchange control functions.

✓ A chain of processors are used to perform the entire control of the exchange.

✓ The entire chain may be duplicated to provide redundancy.



CP = call processor EM&DP = event monitoring and distribution processor O&MP = operation and maintenance processor

Fig. 3 (g): Dual chain distributed control [1]

- **Interrupt Processing:**before studying the Distributed SPC there is need to studyabout interrupt processing
- In Exchange, all the functions are classified into following categories:

  Event Monitoring and Distribution

  Call Processing

  Operation, Maintenance & Call Charging

- In exchange, the processor will process all the functions according to "Interrupt Processing"
- "Interrupt Processing" is done using the priority levels of the programs are being executed by the processor
- ✓ The following will interpret the concept of levels of processing or priority of the processing

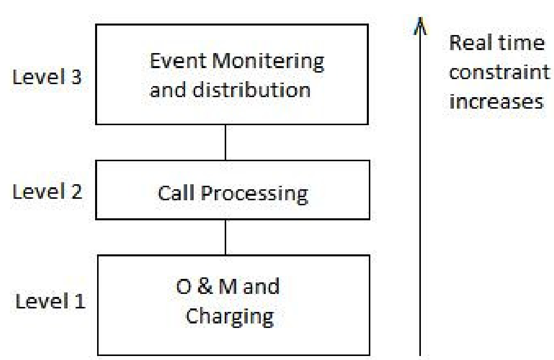


Fig. 4 (a): Interrupt processing [1]

From the above diagram it is understood that

- o Level 3 Process : Event Monitoring and Distribution
- o Level2 Process : Call Processing
- o Level 1 Process : Operation, Maintenance & Call Charging
- o and the priority of Level 3 > Level 2 > Level 1

Let a processor is currently executing a Level 1 Process like "Bill generation" of a consumer, meanwhile if exchange is assigned a new level 2 Process like "Call Connecting" then Processor will currently pause the Level 1 Process and will continue level 2 Process. After completing level 2 Process it will resume level 1 Process this depicts the "Interrupt Processing" as shown below:

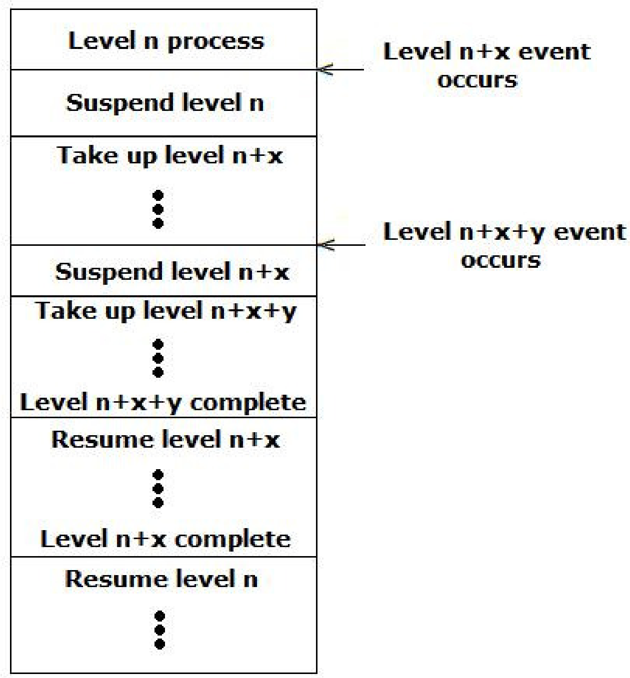


Fig. 4 (b): Interrupt processing priority [1]

Interrupts are basically in following types:

- o**Maskable Interrupt** : process instructions with different Priorities
- o **Non Maskable Interrupt**: The Highest Priority Interrupts like Power Failure,Reset Instruction

In interrupt processing when an interrupt occurs program execution is shifted to an appropriate service routine address in the memory through branch operation, this accomplishes with two methods:

Vectored Interrupt: In this method, the set of branch addresses are suppliedto the processor with different interrupting sources

Non Vectored Interrupt: In this method, the set of branch addresses aresupplied to the processor from fixed source

In case of Centralized SPC, only one processor is used to process all the exchange functions where as in case of “Distributed SPC” three different processors are used for different levels of operations

Level 3 Processing

Level 3 Processing will include the functions like:

- Scanning
- Distribution
- Marking
- Controlling all incoming & Outgoing **local calls, STD Calls, ISD Calls, Fax &Data** services
- Control of all functions are carried by specially designed Processors with
- “Micro-programmed Control”

All levels of processing are depicted in the following figure:

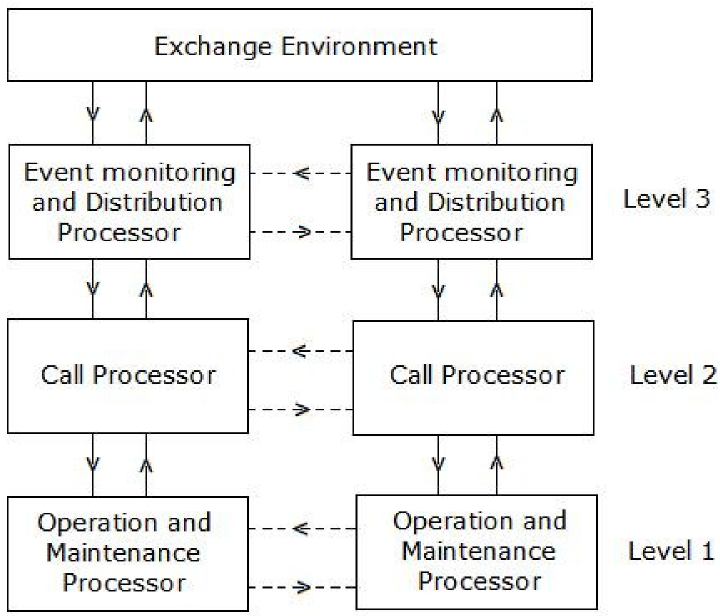


Fig. 4 (c): Level processing [1]

Here it is understood that, at each level of processing there are two set of processors are available one set is for active processing and another set is for standby

Coming back to Level 3 processing, here there is a keen difference between the classical “hard-wired control” and “Micro-programmed control”

Micro-programmed control	Hard-wired control
Flexible, Slower, More Expensive for small exchanges, easier to implement, complex programming, Introduction to new services & Easier to maintain	Non flexible, Faster, Less Expensive for small exchanges, Fixed Processing Speed, Difficult to Implement, Complex Functions, No scope of introducing new services, Difficult to maintain because of Electromechanical and lot of wires and connectors

Control Word: In case of Micro-Programmed Control, all the control functions arecontrolled by a Control Word which contains the status and control actions in the binary codes by which processor able to understands what operations to be performed by the exchange

For example an exchange contain number inlets and outlets in which some line are active and some lines are idle, in the following diagram indicates the status of lines and its binary codes in control word:

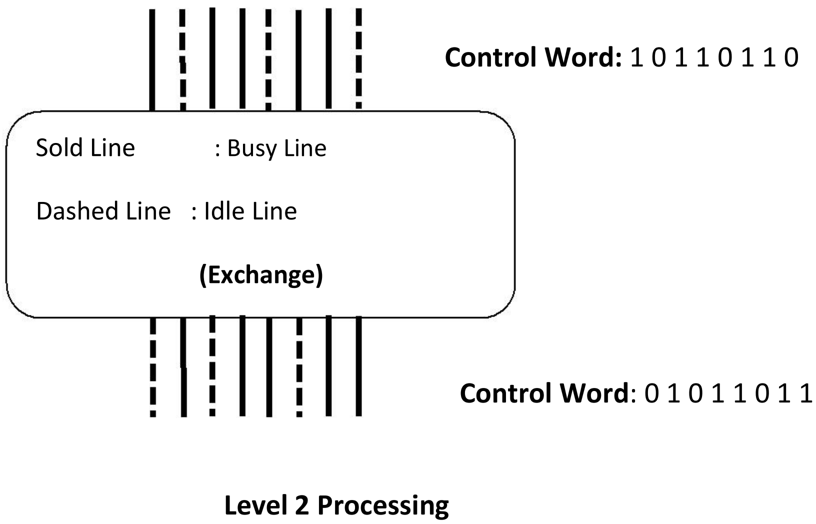


Fig. 4 (d): Level 2 processing [1]

Level 2 Processing

- Level 2 Processing or Processor is also called Switching Processing or Switching Processor
- we the concept of "system availability" it is understood that, the availability of a telephone exchange for a user is completely depending upon the availability of Switching devices at the exchange to connect any two phones are computers. So switching unit play a major role in telephone exchange
- The architecture of switching processors is designed to for 99.9% availability and fault tolerance and security operations
- **Switching Occupancy:** The traffic handling capacity of the control equipment isusually limited by the capacity of the switching processor. The load on the switching processor is measured by its occupancy "t" estimated by the simple formula:

$$t = a + bN$$

o**t = Switching Processor capacity**o**a = fixed overhead**

o**b = average time to process a call**
o **N = Number of calls per unit time**

Level 1 Processing

The Level 1 Processing includes a common general purpose computer to handle the following operations:

- Bill Charging
- Bill distributing
  Monitor Traffic
- Fault tolerance
- Customer Support
- Making a new Service
- Disconnecting a requested serviceProcuring new Equipment
- Paying power bills of exchange.

This kind of operations is not required in a huge demand like Level 3 & Level 2 Operations ·Because of this reason, a central telephone exchange will provide service of Level 1 Processing. Meaning, all the nearby exchanges of a central exchange contain their own level 3 & Level 2 Processing units but Level 1Procesing unit is available at a central telephone exchange. In this way expenses of small exchanges are reduced

The below diagram will depict the concept of a central Operation and Maintenance and Call Charging Unit of some nearby exchange.

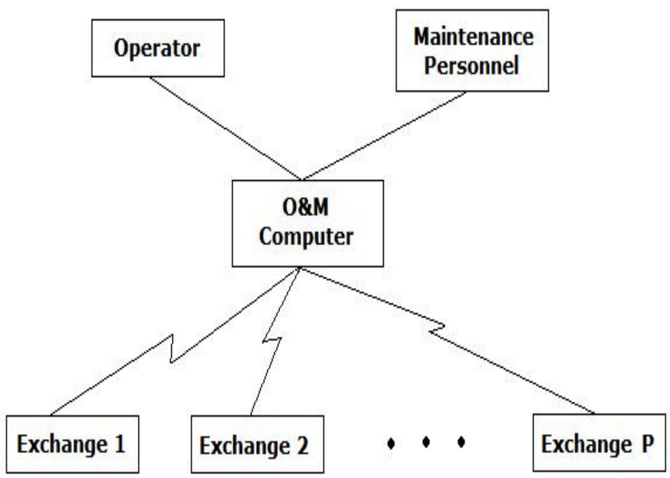


Fig. 4 (e): concept of a central Operation and Maintenance and Call Charging Unit [1]

9.5 Software Architecture

Software is basically two types:

i. System Software (Operating System)

ii. Application Software (Software based on Operating System)

Therefore a Special Design and Development is to be done for Switching Operating System
Process: an instruction executed by the processor is commonly called as a "Process"

Running Process: an instruction is currently executing by the processor

Ready Process: next instruction of running process and an instruction timed out isnormally called as a Ready Process

Blocked Process: a Process or instruction is said to be blocked when it is conditionallike "if", "while" because the execution of this instructions are depending on the results of the conditional statements. The below diagram will depict the state transitions between the "Running Process", "Ready Process" & "Blocked Process"

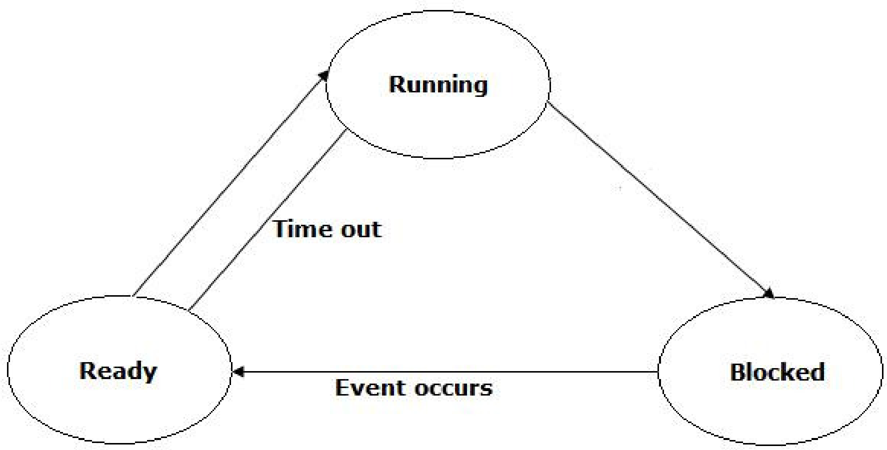


Fig. 4(f): Process States and Transitions [1]

Process Control Block: Each Control Process is represented by the operating systemby a "Process Control Block (PCB)" which is a data structure containing the following information about the process:

- Current State of the Process
- Process Priority and CPU Scheduling parameters
- Memory allocated to process
- Status of events and I/O resources associated with the process

✓

Program Status Word: which contains the address of the next instruction to be executed, the types of interrupts enabled or disabled currently

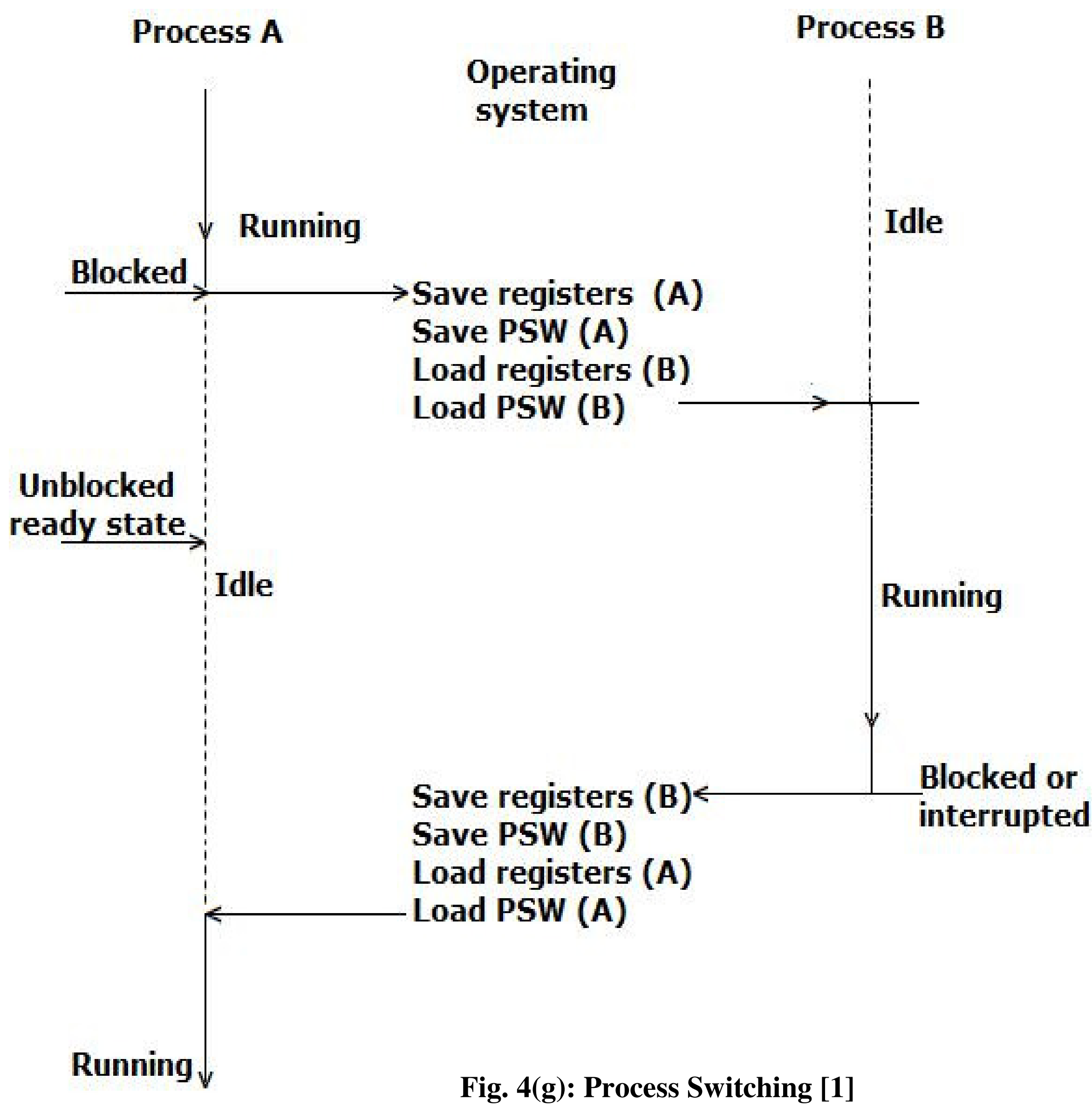


Fig. 4(g): Process Switching [1]

The above diagram depicts the process switching control of an operation system depending upon priority

Critical Region:when numbers of parallel processes are running by the operatingsystem, any time, any process may access common resources like memory space.

"When a process is accessing a common resource in any time of its execution, then the process is said to be in "Critical Region"

Semaphore:in order to avoid the problem of accessing any two or more processesare in critical state and to avoid "Deadlock" a variable "Semaphore" used

Semaphore contain a number (which is equal to the number processes of accessingthe common resources or be in critical state) by accessing this number operating system can manage between different processes in “Critical State” and by which, “Deadlock” is avoided.

9.5.1 SOFTWARE PRODUCTION

Basic factors associated with switching software

- Complexity and size of the software
- Long working life required
- Real time operation
- Stringent reliability and availability
- Software portability

Four stages in software production

- ✓ Functional specification
- ✓ Formal description and detailed specification
- ✓ Coding and verification
- ✓ †Testing and debugging

9.6 ENHANCED SERVICES

Categories of enhanced services

1. Services associated with the calling subscriber and designed to reduce the time spent on dialing and the number of dialing errors.

2. Services associated with the called subscriber and designed to increase the call completion rate.

3. Services involving more than two parties.

4. Miscellaneous services.

Category 1

Abbreviated dialing

Recorded number calls or no dialing calls.

Call back when free

Category 2

Call forwarding

Operator answer

Category 3

†Calling number record

†Call waiting

†Consultation hold

†Conference calls

Category 4

†Automatic alarm

†STD barring

†Malicious call tracing

STD : subscriber trunk dialing

9.7 TWO-STAGE NETWORKS

For any single stage network, there exists an equivalent multistage network.

Simple Two-stage NxN network

ANxN single stage network with a switching capacity of K connections can be realized by a two-stage network of NxK and KxN.

Fig. 5 (a): A two-stage representation of an NXN Network [1]

•First Stage: Any of the N inlets can be connected to any of the K outputs. NK switching elements.

•Second Stage: Any of the K input scan be connected to any of the N outlets. NK switching elements.

•There are K alternative paths for any inlet/outlet pair connection.

Simple Two-stage NxN network

Full connectivity/full availability

Any of the N inlets can be connected to any of the N outlets.

Example

Assume 10% of the subscribers to be active on average. Set K to be N/16. The number of switching elements is S=N2/8.

For N=1024, we have K=64, S=131072.

Note: Feasibility & Flexibility

Two-Stage Networks

Single stage vs. multistage networks

Inlet to outlet connection

Quality of link

Utility of cross-points

Establishment of a specific connection

Cross-point & path Redundancy

Number of cross-points

Capacitive loading problem

Blocking feature

Call establishing time

General two-stage networks

Terminology

Expanding network: M<N

Concentrating network: M>N

Square network: M=N

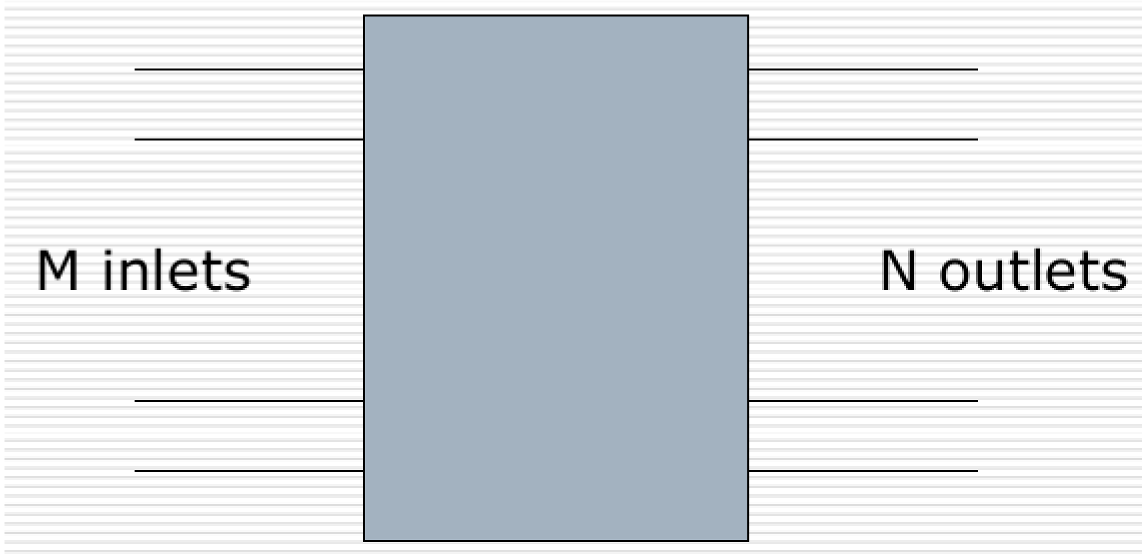


Fig. 5(b): General two-stage networks [1]

Architecture of General two-stage networks

Multiple small size matrices are used in each stage.

Easy to be realized in practice.

Flexible in system design.

NxN two-stage network design

Decomposition: M = p x r, N = q x s

Switching matrices: p x s and r x q

Full availability: There must be at least one out let from each block in the first stage terminatingas inlet on every block of the second stage.

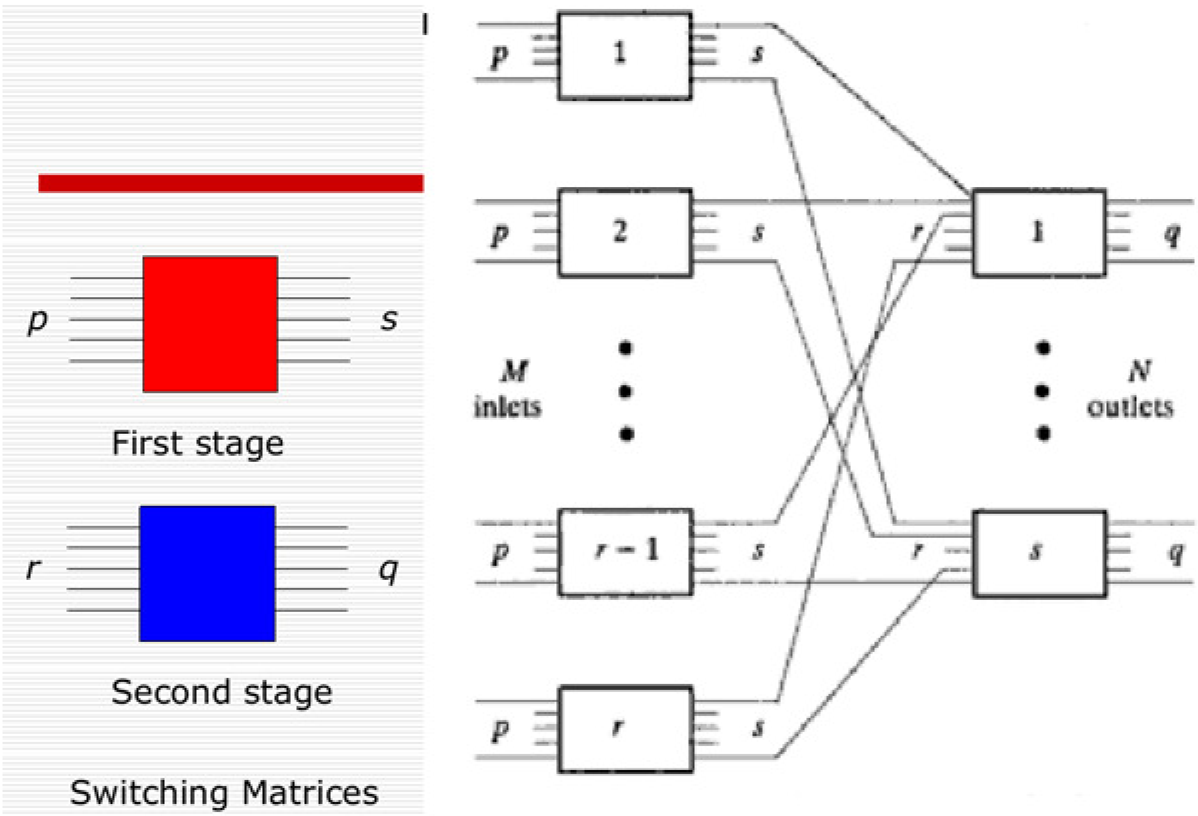


Fig. 5(c): General two-stage networks with multiple switching matrices. [1]

General two-stage networks

Parameters

Number of switching elements

S=psr+qrs=Ms+Nr

Switching Capacity i.e., the number of links between the first and the second stages.

SC=sr

General two-stage networks

Parameters

Blocking probability

Blocking condition 1

There are rxs calls in progress, and the (rs+1)-th call arrives;

The blocking probability PB is dependent on the traffic statistics.

Blocking condition 2

There is a call in progress from I-th block in the first stage to the J-th block in the second stage,and another call originating in the I-th block destined to the J-th block.

Blocking probability

$$P_B = \frac{M \propto (s-1)\left(\left(\frac{M}{r}\right)-1\right) \propto}{rs(s-1)}$$

General two-stage networks

†How to choose values of r and s?

„Both S and SC are proportional to r & s.

„Blocking probability PB is reversely proportional to r & s.

„Strategy: Tradeoffs should be made between cost and quality of service.

The values of r & s should be as small as possible but give sufficient links to provide a reasonable grade of service to subscribers.

Square two-stage networks

Baseline networks

Square switching matrices are used as building blocks.

p=r=s=q=N1/2

There are N1/2blocks, each block is a switching matrix of N1/2xN1/2 inlets and outlets.

Switching elements: S=2NxN1/2

Switching capacity: SC=N Support

Non-blocking networks

Why does blocking occur?

Only one link exists between a pair of first stage and second stage blocks.

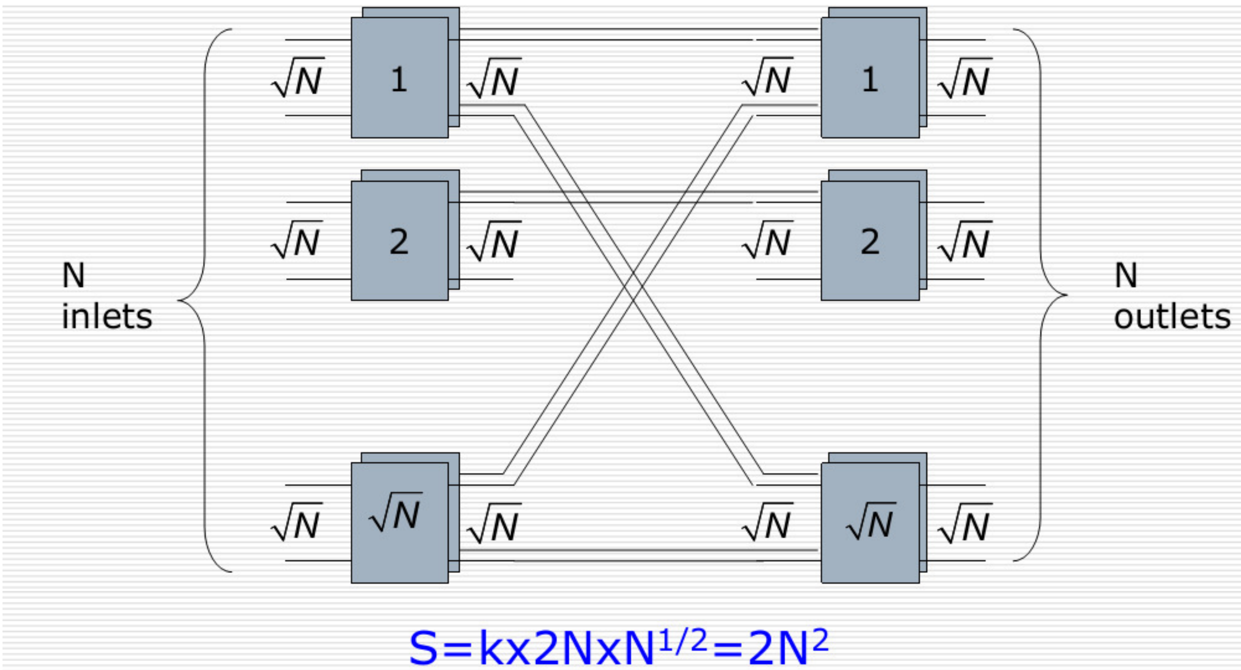
How to reduce the probability of blocking?

Provide more links between the first stage and second stage blocks.

„ How many links should be provided?

A group of k=N1/2links should be provided for each pair of first stage and second stage blocks. S=2N2.

In comparison with single stage network, the number of switching elements is doubled. N simultaneous calls only if the traffic is uniformly distributed.



Three-Stage networks

General structure of an NxN three-stage blocking network

Stage 1: p x s switching matrices

Stage 2: r x r switching matrices

Stage 3: s x p switching matrices

N=pxr, s is changeable

Compared with a two-stage network, there are s alternative pathsbetween apair of inlet and outlet.

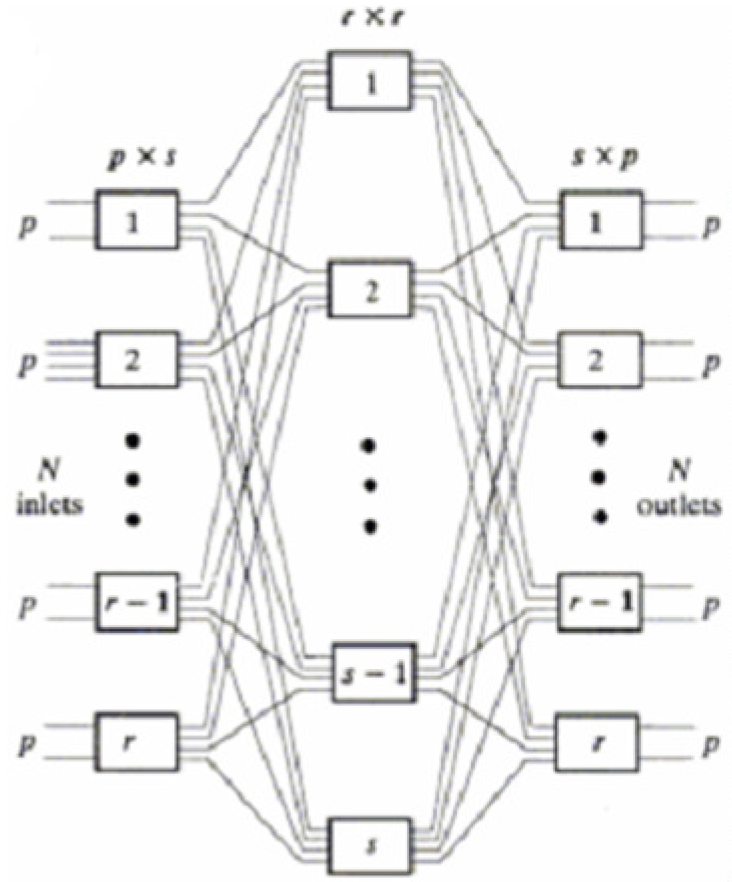


Fig. 5(d): General three-stage networks [1]

NxN three-stage blocking network

Number of switching elements

$$S=rps+sr2+spr=2Ns+sr2=s(2N+r2)$$

If square matrices are used in both the first and third stages, then p=s=N/r and S=2N2/r+Nr.

For a given value of N, there exists an optimal value of r which minimizes the value of S.

The optimal value of r is r= (2N) 1/2 and the corresponding minimum of S isSmin=2N (2N)1/2.

p=N/r=(N/2)1/2

NxN three-stage blocking network

Blocking probability analysis

Probability graph

Circle: stage

Line: link

A graph can be broken down into serial and parallel paths.

Notation

β: probability that a link is busy.

β’=1-β: probability that a link is free.

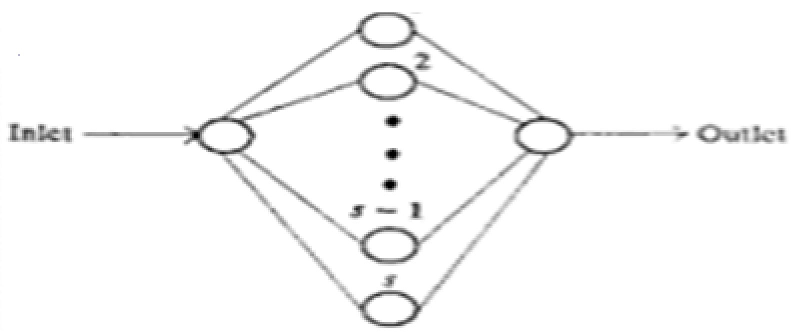
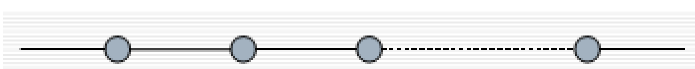


Fig. 5(e): Lee’s graph for a three-stage network. [1]

Blocking probability analysis:

When s serial links complete a connection, the blocking probability is

PB=1-$(1-\beta)^s$



If there are s parallel links, the blocking probability is

PB=βs

Non-blocking three-stage networks

C. Clos: pioneer researcher of multistage non-blocking networks.

†Clos networks

- Multistage non-blocking and fully available networks.
- Much less switching elements are used than that in single stage networks.

Design strategy

Providing adequate number of blocks in middle stages. For three-stage networks, the value of s should be large enough.

Three-stage non-blocking configuration

Worst situation for blocking

p-1 inlets in a block I in the first stage are busy;

p-1 outlets in a block O in the third stage are busy;

The p-1 second-stage blocks, on which the p-1 outlets from block I are terminated on, are different from the p-1 second-stage blocks from which the links are established to the block O.

The free inlet of block I need to be terminated on the free outlet of block O.

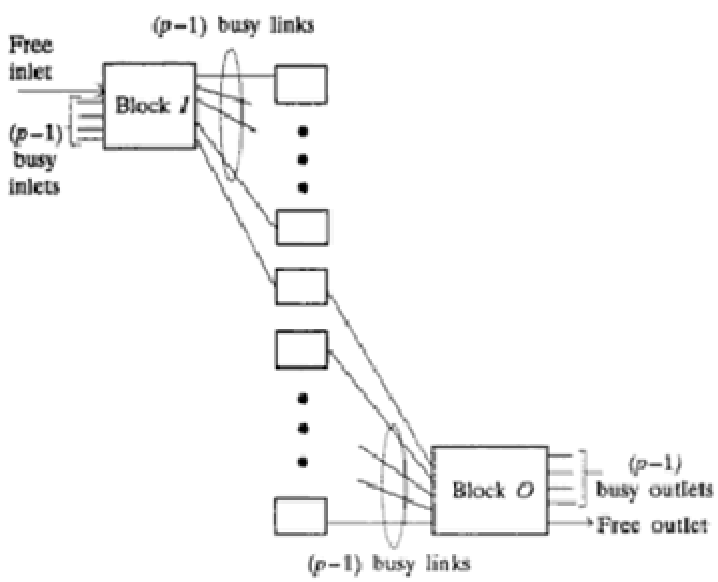


Fig.5 (f): Three-stage non-blocking configuration [1]

The number of blocks required in the second stage for non-blocking operation is

s=2(p-1)+1=2p-1.

The number of switching elements in the non-blocking configuration is given by

$$
\begin{aligned}
S &= p(2p-1)r+(2p-1)r^2+p(2p-1)r \\
&= 2N(2N/r-1)+r^2(2N/r-1) \\
&= (4N^2-2Nr)/r+2Nr-r^2
\end{aligned}
$$

†

There exists an optimal value of r for minimizing the value of S.

Let dS/dr=0, we have $r^2(N-r)=2N^2$

For large values of N, we have $N-r \approx N$.

Hence, $r=(2N)^{1/2},\ p=N/r=(N/2)^{1/2}$

The minimum S is $S_{min}=4N(2N)^{1/2}$

Switching elements advantage ratio

$$\lambda = \frac{\text{S in nonblocking single-stage network, } N^2}{\text{S in nonblocking three-stage network, } 4N(2N)^{1/2}}$$

N-Stage networks

Further reduction in the number of switching elements is possible by using even higher number of stages than three.

Construction of multi-stage networks

By replacing the middle blocks with three-stage network blocks continually, any number of stages can be obtained.

Construction of five-stage ~

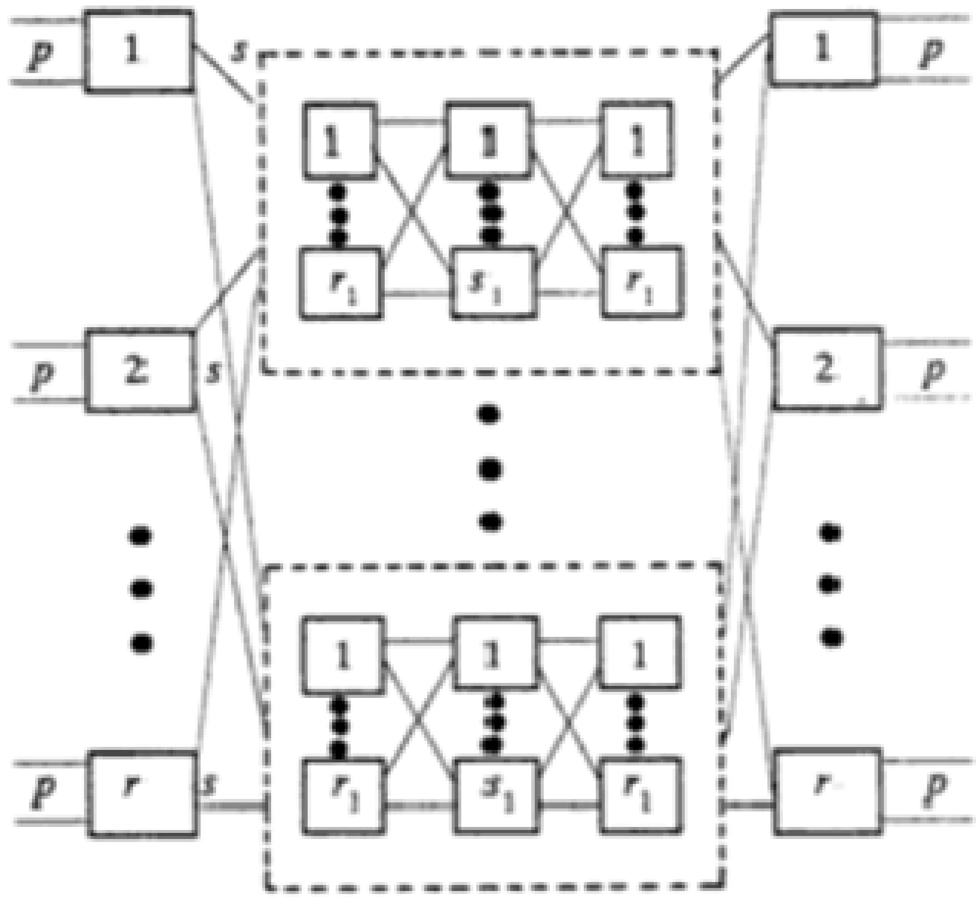


Fig.5 (g):Five-stage switching network [1]

10. Telecommunication Switching:

Various facilities of digital switching and transmission are the reason why the analog switching is slowly getting replaced by digital switching. The incorporation of digital switching and transmission technique into telecommunications altered the whole telecommunication industries setup. The reliability of digital switching system is becoming increasingly important for users of telephone services. Voice and/or data can be represented using digital signals efficiently than analog signals.

A switching system is called digital when the input to and output from the switching system can directly support digital signal. Many basic elements of the digital switching system and its operation are very similar to the stored program control (SPC) switching system. The cost of an analogue switch is roughly proportional to the number of cross points, but the cost relationship in digital switching is different.

The functions of the digital switching network are to connect pairs of channels. So that information arriving at the switching centre in a particular channel on one PCM multiplex system can be passed to some other channel on an outgoing PCM multiplex systems. To achieve this switching, two processes referred to as time switching and space switching are used. The principles of these two switching process are described in this chapter.

In digital data communication (analog or digital signal), a fundamental requirement is that the receiver should know the starting time and duration of each bit that it receives. To meet this requirement a synchronous and synchronous transmission are used. These two transmission techniques are described in this chapter.

Evolution of Digital Swtching System:

10.1 Evolution of Digital Switching System:

The early version of electronic switching system is the stored program control (SPC). The SPC systems have temporary memory for storing transient call information and to carry programming information. The SPC performs line control, trunk control, ancillary control, maintenance control etc. The instructions required for performing these operations are resided in a single processor. For reliability or high availability, the processor may be duplicated. Thus SPC uses centralized software and hardware architectures.

A modern digital switching system employs a number of processors not uses distributed software and hardware architectures. The digital switching system also referred as Electronic Switching System–III generation is purely electronic in operation, the switching process is by time division/digital transmission, the type of control is stored program common control and the network uses pulse code modulation.

10.2 TIME DIVISION SWITCHING

Time division switching involves the sharing of cross points for shorter periods of time. This paves way for the reassign of cross points and its associated circuits for other needed connections. Therefore, in time division switching, greater savings in cross points can be achieved. Hence, by using dynamic control mechanisms, a switching element can be assigned to many inlet-outlet pairs for few microseconds. This is the principle of time division switching. Time division switching uses time division multiplexing to achieve switching. Two popular methods that are used in time division multiplexing are (*a*) the time slot interchange (TSI) and (*b*) the TDM bus. In ordinary time division multiplexing, the data reaches the output in the same order as they sent. But TSI changes the ordering of slots based on the desired connections. The de-multiplexer separates the slots and passes them to the proper outputs. The TDM uses a control unit. The control unit opens and closes the gates according to the switching need. The principle of time division switching can be equally applied to analog and digital signals. For interfacing sampled analog signals but not digitized, the analog time division switches are attractive. But for larger switches, there are some limitations due to noise, distortion and crosstalk which normally occurs in PAM signals. Thus analog switching is now used only in smaller switching systems. In this section, the analog time division switching and digital time division switching are described briefly.

10.3 Analog Time Division Switching

Fig. 6 shows a simple analog time division switching structure. The speech is carried as PAM analog samples or PCM digital samples, occurring at 125 μs intervals. When PAM samples are switched in a time division manner, the switching is known as analog time division switching. If PCM binary samples are switched, then the switching is known as digital time division switching. A single switching bus supports a multiple number of connections by interleaving PAM samples from receive line interfaces to transmit line interfaces. There are two cyclic control stores. The first control store controls gating of inputs onto the bus one sample at a time. The second control store operates in synchronism with the first and selects the appropriate output line for each input sample.

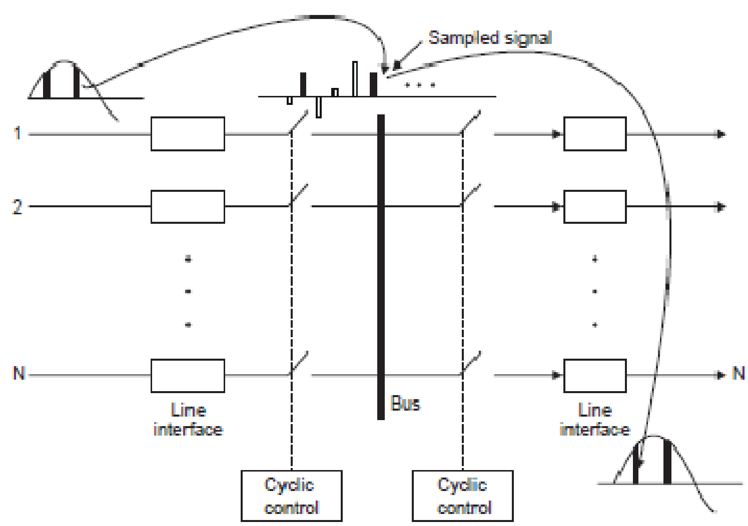


Figure 6: Analog time division switching [1]

The selection of inlet/outlet is controlled by various ways. The (a) cyclic control and (b) memory based control are the important controls and described in the following paragraphs.

Cyclic control: The cyclic control is organized by using Modulo-N counter and k to 2k decoder as shown in Fig. 7.

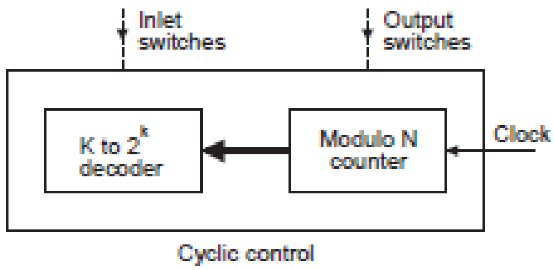


Figure 7: The cyclic control [1]

The k and N are related by $\lceil \log_2 N \rceil = k$

where N = number of inlets/outlets

k = decoder size.

$\lceil \; \rceil$ = gives the lowest integer. It means k may be assumed lowest integer or more than that.

This kind of switching is non-blocking but lack of full availability as it is not possible to connect inlet to any outlet. The switching capacity or number of channel supported by cyclic controlled system is

$$C = \frac{125\,\mu\,\text{sec}}{t_s}$$

The numerator 125 µsindicates the time taken to scan inlet and outlet and the denominator t_s is the time in µs to setup connection. Memory based control. Full availability can be achieved if any one control is made memory based. If the input side is cyclically switched and the outlets are connected based on the addresses of the outlets stored in contiguous location is referred as input controlled or input driven. If the outlets are cyclically switched, the switch is referred as output controlled or output driven. As the physical connection is established between the inlet and the outlet through the common bus for the duration of one sample transfer, the switching technique is known as time division space multiplexing. For this system,

$$C = \frac{125\,\mu\text{sec}}{t_i + t_m + t_d + t_t}$$

Where, t_m = time to read the control memory

t_d= time to decode address and select the inlet and outlet.

t_i= time to increment the modulo N counter.

t_t= time to transfer the sample.

The capacity equations are valid only for an 8 kHz sampling and non-folded network (can be used for folded network with certain changes in network). The switching Capacity in the memory controlled is equal to N. The use of cyclic control in input or output controlled switches restricts the number of subscribers on the system rather than the switching capacity since all the lines are scanned whether it is active or not. No restrictions on subscriber number and full availability of the switching system can be achieved by designing a switching configuration with control memory for controlling both inlets and outlets. This configuration referred to as memory controlled time division space switch is shown in Fig. 8. As each word of the control memory has inlet address and an outlet address, the control memory width is 2 $\log_2$ N. The control memory words are readout one after another. Modulo counter is updated at the clock rate. For the path setup of kth inlet and jth outlet, the addresses are entered in control memory and path is made. Then the location is marked busy. When conversation is terminated, the addresses are replaced by null values and location is marked free. Hence

$$C = \frac{125}{t_s}\ \mu\ \text{sec, where } t_s = t_i + t_m + t_d + t_t$$

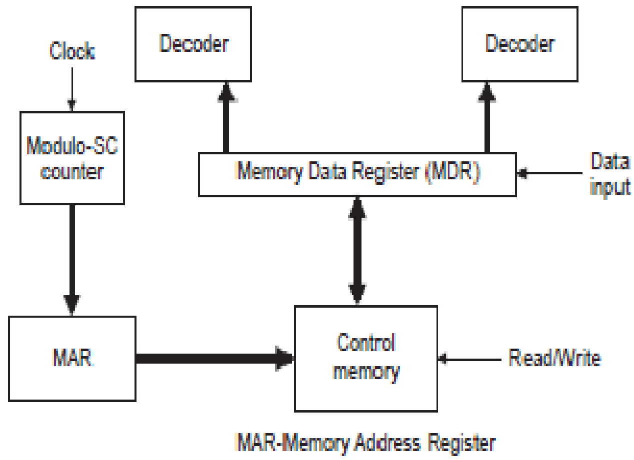


Figure 8: Memory controlled time division space switch [1]

The switching matrix described above is referred to as time multiplexed switching as the switch in this configuration is replicated once for each time slot.

10.4 Digital Time Division Switching

The analog time division switching is useful for both analog and digital signals. The digital time division multiplexed signals usually requires switching between time slots as well as between physical lines. The switching between time slots is usually referred as time switching. Similar to analog time division switching the switching structure can be organized expect the use of memory block in place of the bus. This adds the serial to parallel and parallel to serial bit conversion circuitry's as the input to the memory block should be in parallel form. The time division switch can be controlled in any of the following three ways.

Basic operation: The basic requirement of time division switching is that the transfer of information arriving at in a time slot of one input link to other time slot of any one of output link. A complete set of pulses, arriving at each active input line is referred to as a frame. The frame rate is equal to the sample rate of each line. A time switch operates by writing data into and reading data out of a single memory. In the process the information in selected time slots is interchanged as shown in Fig. 9.

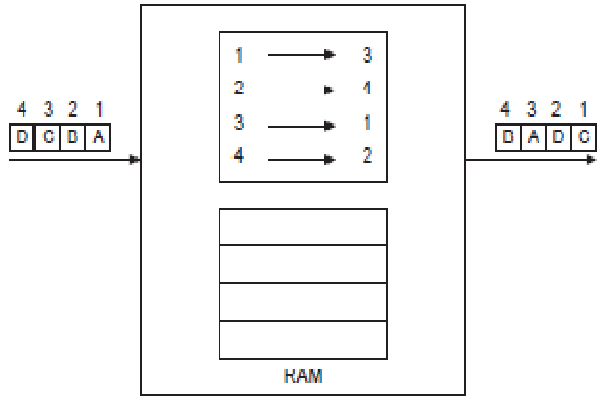


Figure 9: Time division switch[1]

In TSI operation, inputs are sequentially controlled and outputs are selectively controlled. The RAM has several memory locations, each size is the same as of single time slot. Fig. 10 shows the general arrangement of the time division time switching.

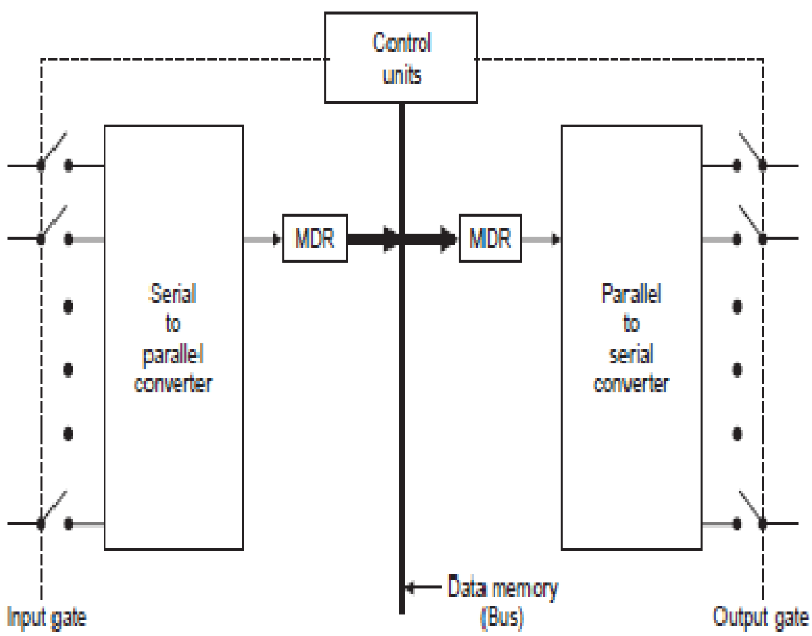


Figure 10: General arrangement of time division switching [1]

The serial to parallel and parallel to serial converter are used to write the data into the memory and read the data out of memory. For convenience, two MDR are shown, but MDR is a single register. Gating mechanism is used to connect the inlet/outlet to MDR. The input and output lines are connected to a high speed bus through input and output gates. Each input gate is closed during one of the four time slots. During the same time slot, only one output gate closed. This pair of gates allows a burst of data to be transferred from one input line to a specific output line through the bus. The control unit opens and closes the gates according to switching need. The time division time switch may be controlled by sequential write/random read or random write/ sequential read. Fig. 10 depicts both modes of operation and indicateshow the memories are accessed to translate information from time slot 2 to time slot 16. Both methods use a cyclic control.

Fig. 10 (a) implies that specific memory locations are dedicated to respective channels of the incoming TDM link. Data are stored in sequential locations in memory by incrementing modulo N counter with every time slot. Thus incoming data during time slot 2 is stored in the second location within the memory. On output, information retrieved from the control store specifies which address is to be accessed for that particular time slot. Thus sixteenth word of control store contains the number 2, implying that the contents of data store address 2 is transferred to the output link during outgoing slot 16. Random write/sequential read mode of operation is opposite to that of sequential write/ random read. Incoming data are written into the memory locations as specified by the control store, but outgoing data are retrieved sequentially under control of an outgoing time slot counter. The data received during time slot 2 is written directly into data store address 16 and it is retrieved during outgoing TDM channel number 16.

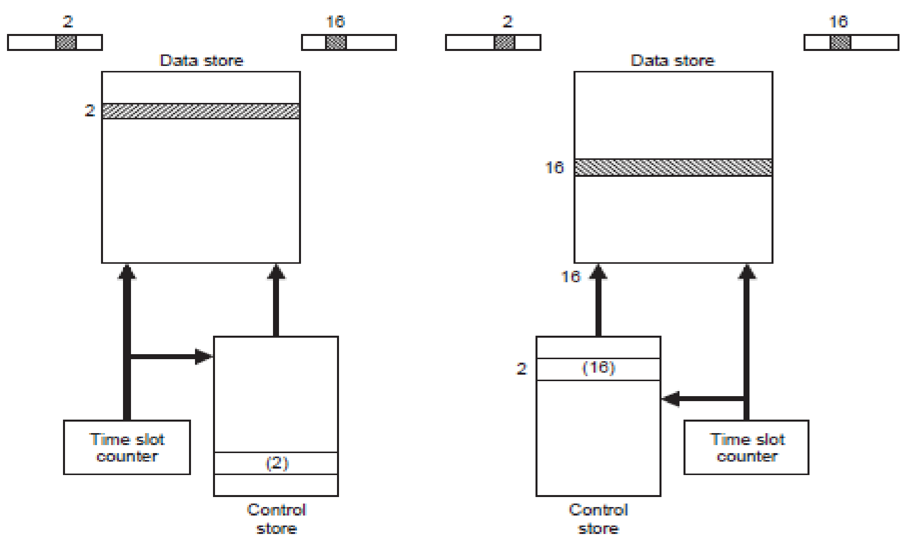


Figure 10 (a) : Memory locations allocated to TDM [1]

10.5 TWO DIMENSIONAL DIGITAL SWITCHING

Combination of the time and space switches leads to a configuration that achieved both time slot interchange and sample switching across trunks. These structures also permit a large number of simultaneous connections to be supported for a given technology. Large digital switches require switching operations in both a space dimension and a time dimension. There are a large variety of network configurations that can be used to accomplish these requirements. The incoming and outgoing PCM highways are spatially separate. So the connection of one line of local exchange obviously requires space switching to connect to the channel of outgoing highways. Thus the switching network must be able to receive PCM samples from one time slot and retransmit them in a different time-slot. This is known as time slot interchange, or simply as time switching. Thus the switching network must perform both space and time switching.

The space switching and time switching may be accomplished in many ways. A two stage combination switch may be organized with time switch as first stage and the space switch as the second stage or vice versa. The resulting configurations are referred as time space (TS) or space time (ST) switches respectively. Three stage time and space combinations of TST and STS configurations are more popular and flexible. Very large division switches includes many combinations of time and space switches. Typical configurations are TSST, TSSSST, and TSTSTSTS. These switches support 40000 lines or more economically. The general block diagram involving time and space switching is shown in Fig. 11.

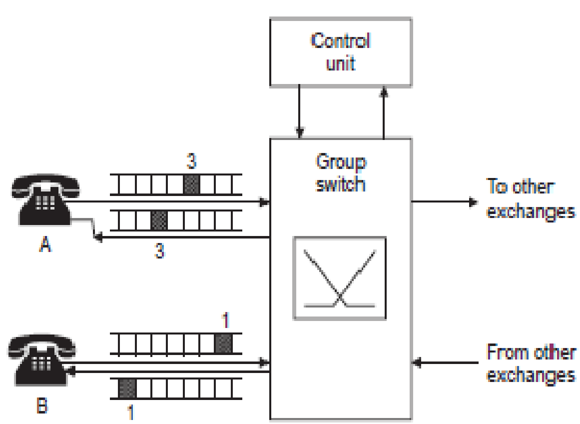


Figure 11: Time and space switching [1]

The main task of the switching part is to interconnect an incoming time slot and an outgoing time slot. The unit responsible for this function is group switch. There are two types of building block in the digital group switch. They are time switch and space switch. In Fig. 7, the subscriber makes a local call to B. The control unit has assigned time slot 3 to the call on its way into the group switch, and time slot 1 on its way out of the group switch (to B). This is maintained during the entire call. Similarly B to A also carried out. The fundamental design and structure of the two switches viz. time switch and space switch are described in the following sections.

11. Space and Time Switches

11.1 Space switch: Fig. 12 shows a typical space switch. It uses a space array to provide switching generally the space switch consists of a matrix of $M \times N$ switching points where M is number of inlets and N is number of outlets. A connection between an inlet and an outlet is made by the simple logic gates (AND gates). As logic gates are unidirectional, two paths through switching matrix must be established to accommodate a two way conversation. The logic gate array can serve for concentration, expansion or distribution depending on M is larger, smaller or equal to N. Fig. 8 shows only one voice direction. However, the corresponding components are available for the opposite direction too. A number of M, of X slot multiplexers, provide the inputs and the outlets are connected to N, X slot de-multiplexers. The gate select memory has X locations. The word containing information about which cross point is to be enabled is decoded by the translator. During each internal time slot, one cross point is activated. In the shift to the next interval time slot, the control memory is incremented by one step, and a new cross point pattern is formed in the matrix.

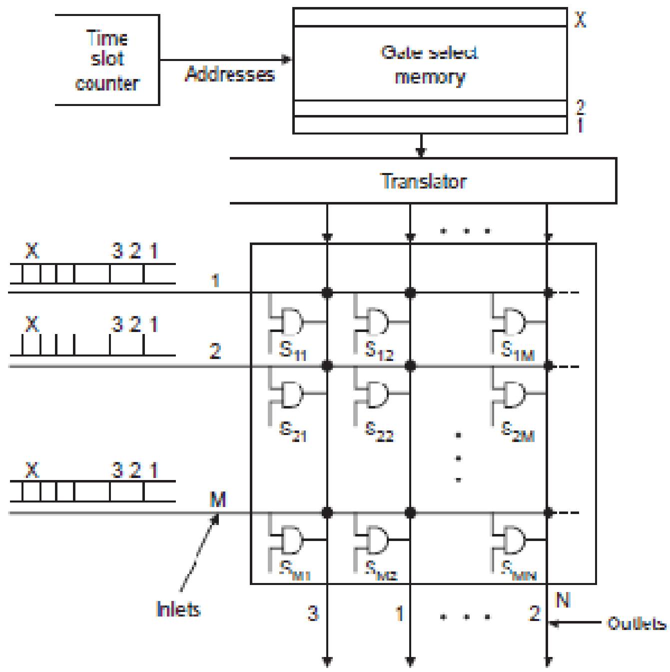


Figure 12: Space switch [1]

11.2 Time switch: The time-slot interchange (TSI) system is referred to as time switching (T-switching). Fig. 13 shows the block diagram of time switch.

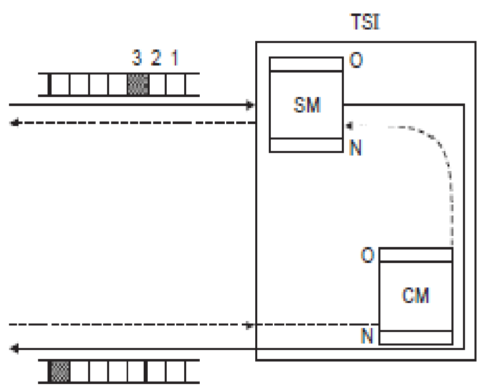


Figure 13: Time switch [1]

Each incoming time slot is stored in sequence in a speech memory (SM). The control memory (CM) determines in which order the time slots are to be read from SM. This means that a voice sample may be moved from say incoming time slot 3 to outgoing time slot 1.

11.3 Time-space (TS) Switching: This switch consists of only two stages. This structure contains a time stage T followed by a space stage S as shown in Fig. 14. Thus this structure is referred to as time-space (TS) switch. The space arrays have N inlets and N outlets. For each inlet line, a time slot interchangerwith T slots is introduced. Each TSI is provided with a time slot memories (not shown). Similarly a gate select memory needs to be provided for the space array (not shown).

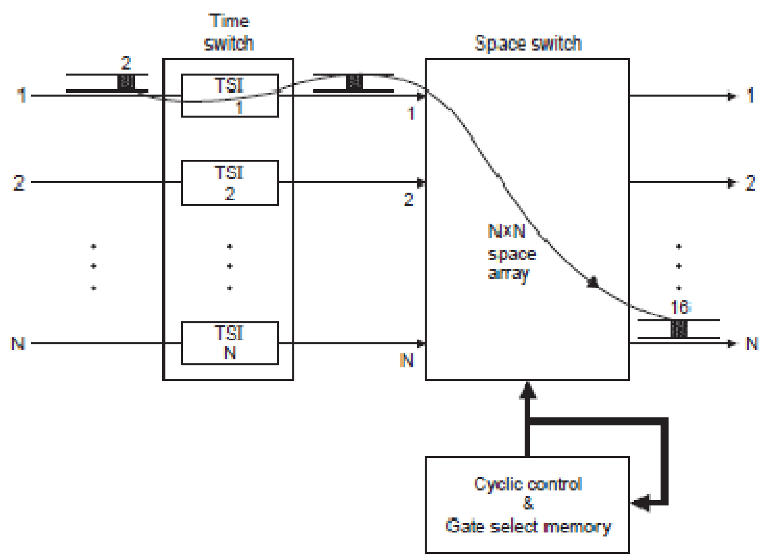


Figure 14: Time space switching [1]

The transmission of signals carried out from sender to receiver through multiplexer input and de-multiplexer output. The reverse communication is also similar. Thus a hybrid arrangement is needed to isolate the transmitted signal from the received signal. The basic function of the time switch is to delay information in arriving time slots until the desired output time slot occurs.

Let the communication is to take place between subscriber A and B. Let A is assigned time slot 2 and line 7 and subscriber B is assigned time slot 16 and line 11. Then the signal moved from time slot 2 to time slot 16 by the time-slot exchanger and is transferred from line 7 to line 11 in the space array. Similarly, the signal originated by B is moved from slot 16 to slot 2 through line 11 to 8. The cyclic control and gate select memory contains the information needed to specify the space stage configuration for each individual time slot of a frame. The time stage has to provide decays ranging from one time slot to a full frame. During each outgoing time slot, control information is accessed that specifies inter stage link number to output link. During other time slots, the space switch is completely reconfigured to support other connections.

Let each time slot interchanger have T slots. If the space array is a N × N, then the simultaneous connections possible is NT. If T = 128 and N = 16, 2048 connections can be supported. This structure is not free of blocking. The control store is a parallel end around shift resister. If space array is at the inlet side and time switch is at the output side, the structure is referred as space time (ST) switching. Both TS and ST arrangements are equally effective. TS system is used in DMS 100 digital switching system developed in Canada (1979). It handles 61000 trunks and accommodates 39000 trunks.

Blocking probability :

The blocking probability of TS switching is calculated as follows.

The probability that a subscriber A is active $= \dfrac{\rho}{T}$ ...(5.28)

where ρ = fraction of time that a particular link is busy measured in Erlangs

T = number of time slots in a frame.

The probability that any other subscriber is active on the same link

$$= \frac{(T-1)\rho}{T} \qquad \text{...(5.29)}$$

The probability that a particular called subscriber is chosen by A

$$= \frac{1}{NT} \times \frac{1}{T} \qquad \text{...(5.30)}$$

where N = Number of inlets (or outlets) for N × N space array.

NT = Simultaneous connection ...(5.31)

The probability that the same time slot on a different outlet is chosen by the other subscribers on the same inlet

$$= \frac{(T-1)(N-1)\rho}{T(NT-1)} \qquad \text{...(5.32)}$$

From Blocking probability $= B = \left(\dfrac{\rho}{T \times NT}\right)\left(\dfrac{(T-1)(N-1)}{T(NT-1)}\right)$

As T >> 1 and N >> 1, & NT >> 1

$$B = \frac{P}{NT^3} \qquad \text{...(5.33)}$$

The TS switch can be made non-blocking by using an expanding time switch (T to T^2 slots) and a concentrating space switch (which is complex).

The number of cross points in space stage can be easily calculated which is based on the array size. The time stage uses significant amount of memory which adds the cost of the whole system. To take this into account the cost of memory bit is assumed one hundredth of the cost of cross point. Thus,

$$\text{Implementation complexity} = N_X + \frac{N_B}{100}$$

where N_X = Number of space stage cross points

N_B = Number of bits of memory.

The N_B not only includes the time stage memory arrays, but also the control memory (store) of the time stage and space stage. Thus,

$$N_B = N_{BX} + N_{BT} \qquad \text{...(5.35)}$$

where N_{BX} = Number of memory bits for the space stage control store

= N × (Number of control words) (number of bits per control word)

N_{BT} = Number of memory bits in the time stage equal to sum of time slot interchange and the control store bits.

= N × number of channels × number of bits per channel + N × number of control words × number of bits per control world.

Example 5.3. If N = 80, N_{BX} = 13, 440 and N_{BT} = 24,960 for a typical TS switch, calculate the implementation complexity.

$$IC = N_X + \frac{N_{BX} + N_{BT}}{100} = 80 \times 80 + \frac{13440 + 24960}{100}$$

IC = 6784 equivalent cross points.

As the number of cross points in space array is equal to 6400, the total cost is dominated by the space stage.

11.4 STS and TST Switching:

The TS structure is of blocking nature. Let A and B are the subscribers using different time slot on the same line want to connect to two subscribers C and D using same time slot on different lines. A and B can be moved to the same time slot but during that time slot, the inlet line can be connected to C's line or D's line but not both. This is the significant limitation of the structure. Moreover, time stage switching is generally less expensive than space stage switching as digital memory is much cheaper than digital cross points (AND gates). The multiple stages overcome the limitations of the individual switches and cost savings can also be achieved. TST, STS, TSST, TSSSST and TSTSTSTSTSTSTS are the switching system configurations used in digital switching system. However, the TST structure is the most common.

11.4.1 STS Switching:

In STS switching, the time stage is sandwiched between two space arrays. The digital switching system ITS 4/5 of USA (1976) uses the STS switching configuration. It handles 3000 trunks and accommodates 1500 Erlangs of traffic. Fig. 15 shows the space-time- space (S-T-S) switching network for M incoming and outgoing PCM highways. Establishing a path through an STS switch requires finding a time switch array with an available unit's access during the incoming time slot and an available read access during the desired outgoing time slot. The input side space stage as well as the output side space stage is free to utilize any free time switch modules. In the diagram shown in Fig. 11, the time slot 2 is connected to the TSM 2 where the time slot allotted is 16 and passed to the $(M - 1)^{th}$ line of output space array. Thus the path is provided. This structure is of non-blocking nature.

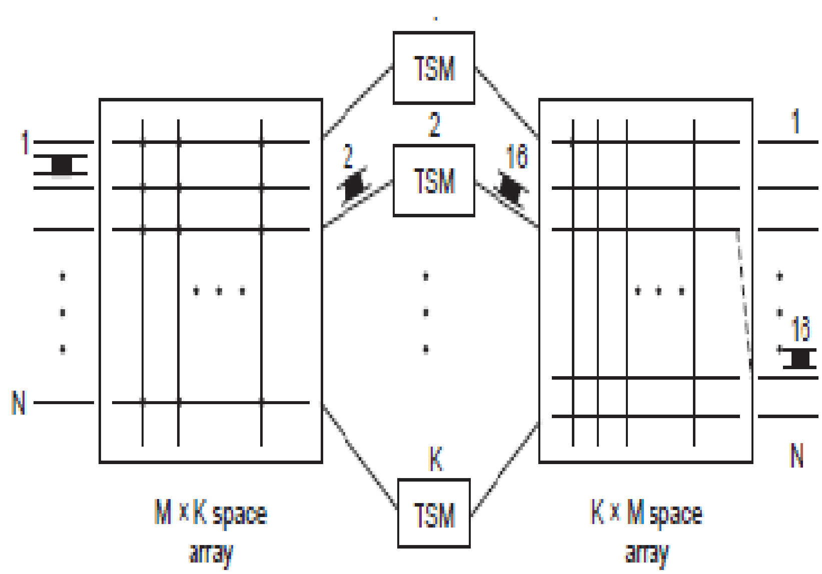


Figure 15: STS Switching[1]

Blocking probability: The STS switch is identical to the probability graph of three stage space switches. Similar to that, the blocking probability of an STS switch is

$$B = \left[1 - \left(1 - \frac{p}{\beta}\right)^2\right]^K$$

Where p = probability that a link is busy

N = is the factor by which the percentage of links that are busy is reduced. ($\beta < 1$)

K = number of center stage TSM.

Implementation capacity (IC):

While calculating IC, the total number of two space stage cross points, total number of two space stage control bits, number of time stage memory bits and number of time stage control bits are to be considered. Thus,

$$IC = 2KN + \frac{2KC \log_2 N + KC(8) + KC \log_2 C}{100}$$

Where K = Minimum number of center stage TSM to provide desired grade of service

C = number of channel.

11.4.2 TST Switching:

In TST switching the space stage is sandwiched between two time stage switches. Of all the multistage switching, TST is a popular one. Popular digital switching systems using TST are tabulated in table 2.

Some important features of TST switches are:

(i) Low blocking probability. An incoming channel time slot may be connected to an outgoing channel time slot using any possible space array time slot. Thus there are many alternative paths between two subscribers. This concept reduces the blocking probability of a three stage combination switch.

(ii) Stage independency. The space stage operates in a time-divided fashion, independently of the external TDM links. The number of space stage time slots L does not coincide with the number of external TDM time slots T.

Type	Characteristics	
	Max. no. of trunks	Traffic (Erlangs)
E 10 B (France, 1970)	3600	1600
AXE 10 (Sweden 1978)	65000	30000
EWSD (Germany 1980)	60000	30000
GTD SEAX (USA, 1982)	49000	36000
C-DOT MAX-XL (India)	40000	47000

Table 2: Digital switching systems using TST and its characteristics [1]

(iii) Implementation advantage. The factors to be considered for switching design and implementation are traffic loads, modularity, testability, expandability and simple control requirements. For large switches with heavy traffic loads, the TST have good implementation advantage.

(iv) More cost effective. If the input channel loading is high, the time expansion of TST and space expansion of STS are required. Time expansion of TST can be achieved at less cost than space expansion of STS. In comparison with STS, the TST have certain limitations. For small switches, the STS architectures are less complex to implement than TST. The control requirements of STS are simpler than TST.The principle of operation of TST switching is shown in Fig. 16. In figure, two flows of time slots, one for each direction are connected together.

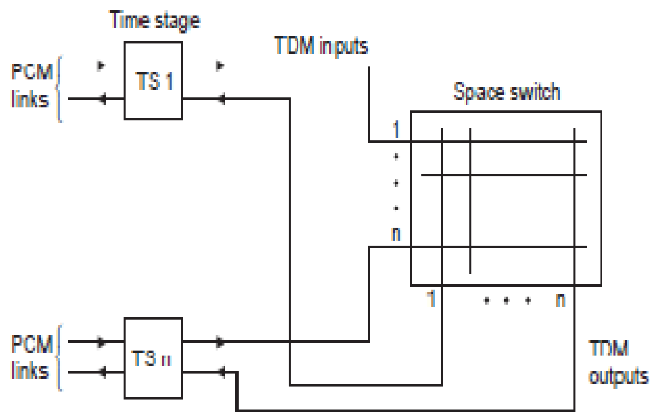


Figure 16: Principle of TST switching [1]

The functional block diagram which explains the transfer of signals from inlet to outlet is shown in Fig. 17. The information arriving at the incoming link of TDM channel is delayed in the· inlet times stage until an appropriate path through the space stage is available. Then the information is transferred through the space stage to the appropriate outlet time stage. Here the information is held until the desired outgoing time slot occurs. Any space stage time slot can be used to establish a connection. The space stage operates in a time divided fashion, which isIndependent of the external TDM links. There are many alternative paths between a prescribed input and output unlike a two stage network which has only one fixed path.

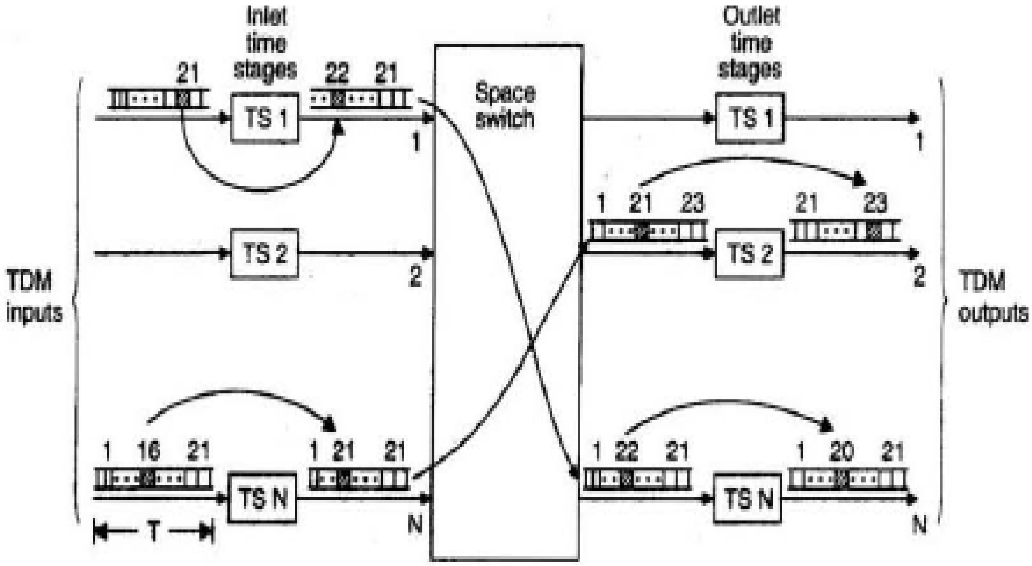


Figure 17: TST switching structure [1]

Blocking probability: The blocking probability is minimized if the number of space stage time slots L is made to be large. By direct analogy of three stage space switches, the TST switch is strictly non-blocking if

L = 2T - 1

Where T = number of time slot of time switch.

L = number of space slot of space switch.

The probability graph of TST switch with non-blocking stage is shown in Fig. 18.

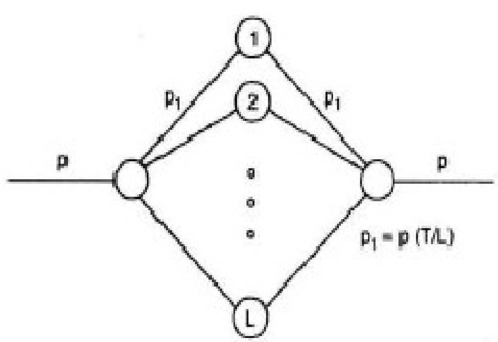


Figure 18: Probability graph[1]

The general expression of blocking probability for a TST switch with non-blocking individual stage is

$$B = [1 - (1 - pT/L)^{n}]^{L}$$

For 3 stage $$B = [1 - (1 - p(T/2))^{2}]^{2}$$

Implementation complexity:

The implementation complexity (IC) of a TST switch can be derived as

$$IC = N^2 + \frac{NL \log_2 N + 2NT(8) + 2NL \log_2 T}{100}$$

Where N=No. of TDM links

T= No. of channel

L= No. of Time slot of space switch

Objective of the Module- II:

- What is traffic network load?
- What is the meaning of blocking probability and its physical significance in the telecommunication network?
- What is grade of service?
- Difference between the grade of service and the blocking probability.
- Markov Model of a switching system in a telecommunication network.
- Traffic arrival and service time probability density function
- Design of blocking model and delay model
- Estimation of call loss in a blocking model and in a delay model.
- Design of a subscriber loop system
- Different types of signaling techniques that are used in a telecommunication network.
- Routing protocol used in a telecommunication network.
- Switching hierarchy in a telecommunication network.

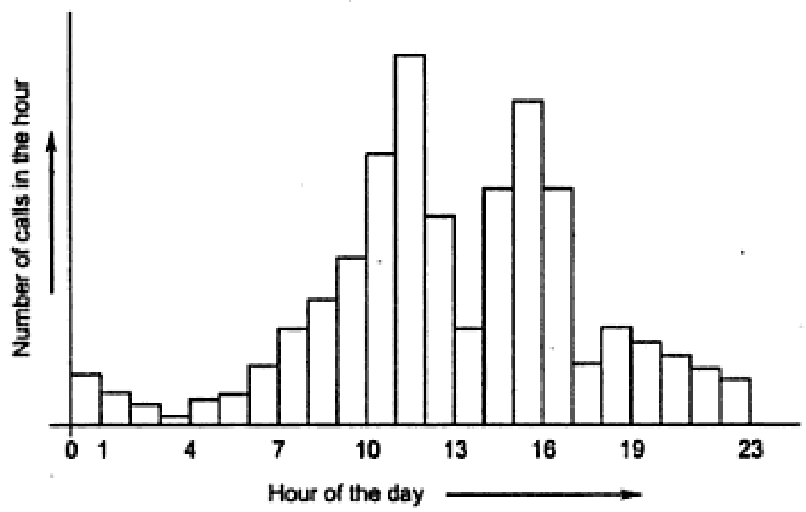
MODULE-II

12. TRAFFIC ENGINEERING:

Due to the non-availability of switching paths blocking of a subscriber call will occur. For this we have calculated blocking probabilities as discussed in the previous chapters [1]. These problems can be avoided by the help of traffic engineering. Traffic engineering analysis enables one to determine the ability of a telecommunication network to carry a given traffic at a particular loss probability [1].

12.1 Network traffic load and parameters:

In a telephone network the traffic load on a typical working day during 24 hours is as shown in figure;



Telephone traffic pattern of a working day

Fig19: Telephone traffic pattern statistics of a working day. [1]

From this statistics:

i) There is a little use of network during 0 and 6 hours when most of the population is asleep.
ii) There is a large peak around mid-forenoon and mid-afternoon, which shows busy office activities.
iii) The afternoon peak is however slightly smaller.
iv) The load is low during the lunch hour period i.e. 12:00-14:00 hours.
v) During the period 17:00-18:00 hours shows low traffic because the people are move from office to their residences.

vi) The peak of domestic calls occurs after 18:00 hours when person reach home and reduces tariff applies.
vii) During holidays and festival days the traffic pattern is different.

Generally there is a peak of calls occurs around 10:00 hours just before people leave their homes on outings and another peak occur in the evening when people returns to their home.

There are 3 types of busy hours are defined by CCITT:

1) **BUSY HOUR:** In a day the 60 minute interval in which the traffic is the highest is called the busy hour.
2) **PEAK BUSY HOUR:** The busy hour on each day is called peak busy hour; It varies from day to day or over a number of days.
3) **TIME CONSISTENT BUSY HOUR:** The one hour period starting at the same time each day for which the number of call attempts is greatest over the days.

Again all the call attempts are not materialize into actual conversations for variety of reasons: Those are due to called line busy, no answers from called lines, and blocking in the trunk groups or the switching centers [1].

A call attempt is said to be successful or completed if the party answers, successful call attempts is again categorized into three types: [1]

1. **CCR (call completion rate):** It is defined as the number of successful calls to the number of call attempts.
2. **BHCA (busy hour call attempts):** The number of call attempts in the busy hour is called busy hour call attempts.

13. GRADE OF SERVICE AND BLOCKING PROBABILITY:

1. The amount of traffic rejected by the network is an quality of service offered by the network, This is known as **grade of service**.[2]
2. Grade of service is defined as the ratio of lost traffic to offered traffic.

$$GOS = \frac{\text{Blocked Busy Hour calls}}{\text{Offered Busy Hour calls}}$$

$$GOS = \frac{A - A_0}{A}$$

where A_0 = carried traffic

A = offered traffic

$A - A_0$ = lost traffic.

3. The smaller is the value of grade of service, the better is the service.

4. The blocking probability P_B is defined as the probability that all the servers in a system are busy.
5. When all the servers are busy no further traffic can be carried by the system and the arriving subscriber traffic is blocked.

13.1 DIFFERENCE BETWEEN GOS AND P_B

1. Grade of service is also known as call congestion or loss probability where as blocking probability is otherwise known as time congestion.[2]
2. GOS is a measure from the subscriber point of view i.e. the GOS is zero as there is always a server available to a subscriber where as blocking probability is a measure from the network or switching system point of view i.e. the blocking probability is non zero as there is a definite probability that all the servers are busy at a given instant.
3. GOS is arrived by observing the number of rejected subscriber calls where as P_B is arrived by observing the busy servers in the switching system.

13.2 DELAY PROBABILITY:

If the offered load or the input rate of traffic far exceeds the network capacity, then the queue lengths become very large and the calls have undesirably long delay. The probability that the call experiences a delay termed as delay probability [2]. In this case the delay systems are said to be unstable as they would never be able to clear the load.

The technique of queued up traffic cleared to an acceptable limit to maintain a stable operation is called **flow control.[1]**

13.3 MODELLING SWITCHING SYSTEMS:

In a telecommunication network the call generation by the subscribers and the behavior of the network or the switching system are random process [1] . A random process or a stochastic process is one in which one or more quantities vary with time such that the instantaneous variables predictable with certain probability.

We have four different types of stochastic process namely

i. Continuous time continuous state
ii. Continuous time discrete state
iii. Discrete time continuous state
iv. Discrete time discrete state

A discrete state stochastic process is called **chain.**

Random processes whose statistical parameters do not change with time are known as **stationary process** [1].

The random processes which have identical time and ensemble averages are known as **ergodic process** [1].

In random process if the mean and variance alone are stationary and other higher order moments may vary with time are known as **wide sense stationary process [1].**

13.4 MARKOV PROCESS:

Markov process is an important class of random processes that have some special properties. The properties were defined by A.A Markov in1907.This property is used for modeling of our switching system [2].

A discrete time markov chain i.e. discrete time discrete state Markov process is defined as

$$P\,[\{X(t_{n+1}) = x_{n+1}\}/\{X(t_n) = x_n, X(t_{n-1}) = x_{n-1}, \ldots\ldots X(t_1) = x_1\}]$$
$$= P\,[\{X(t_{n+1}) = x_{n+1}\}/\{X(t_n) = x_n\}]$$

where $t_1 < t_2 \ldots\ldots < t_n < t_{n+1}$ and x_i is the ith discrete state space value.

The above equation states that the entire past history (n-1,n-2,n-3) is summarized in its current status (n),hence the next state (n+1) is determined only by the current state [2].

The interstate transition time in a discrete time markov process is **geometrically distributed** and in a continuous time markov process it is **exponentially distributed [1].**

13.4.1 BIRTH-DEATH PROCESS:

The birth and death process is a special case of the discrete state continuous time Markov process, which is often called a continuous-time Markov chain. The number of calls in progress is always between 0 and N. It thus has N + 1 state. If the Markov chain can occur only to adjacent states (i.e. probability change from each state to the one above and one below it) the process is known as birth-death (B–D) process [2].

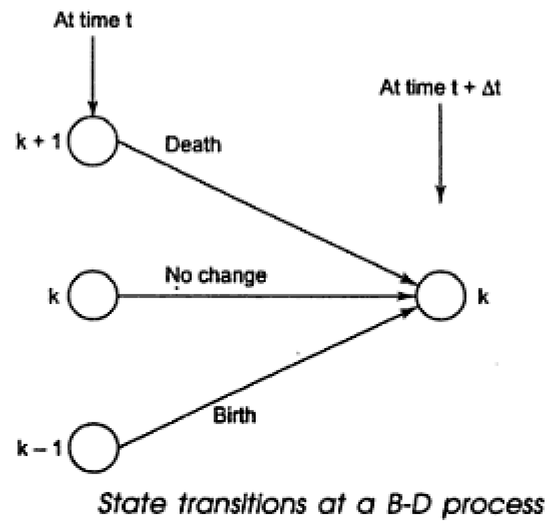


Fig20: State probability transition diagram in a birth death process. [1]

The basic feature of the method of Markov chains is the Kolmogorov differential-difference equation, for the limiting case, can provide a solution to the state probability distribution for the Erlang systems and Engset systems.

Let N (t) be a random variable specifying the size of the population at time t. For a complete description of a birth and death process, we assume that N (t) is in state k at time t and has the following properties:

1. P (k) is the probability of state k and P (k + 1) is the probability of state k + 1.

2. The probability of transition from state k to state k + 1 in short duration Δt is $\lambda_k \Delta t$,

Whereλ_k is called the birth rate in state k.

3. The probability of transition from state k to state k – 1 in the time interval Δt is $\mu_k \Delta t$,

Whereμ_k is called the death rate in state k.

4. The probability of no change of state in the time interval Δt is equal to $1 - (\lambda_k + \mu_k) \Delta t$.

5. The probability in Δt, from state k to a state other than k + 1 or k – 1 is zero.

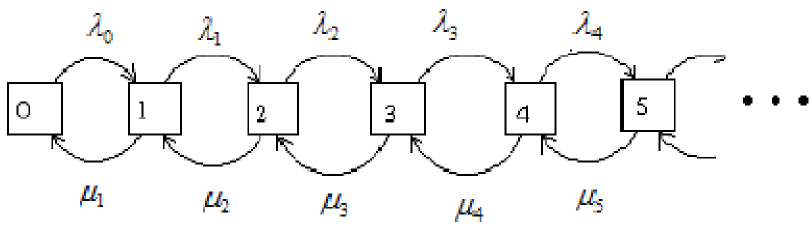


Fig21: State transition diagram in a birth death process with mean arrival and service rate.

Rate transition matrix for Birth-Death process:

$$Q = \begin{bmatrix} -\lambda_0 & \lambda_0 & 0 & 0 & 0\cdots \\ \mu_1 & -(\mu_1+\lambda_1) & \lambda_1 & 0 & 0\cdots \\ 0 & \mu_2 & -(\mu_2+\lambda_2) & \lambda_2 & 0\cdots \\ 0 & 0 & \mu_3 & -(\mu_3+\lambda_3) & \lambda_3\cdots \\ \ldots & \ldots & \ldots & \ldots & \ldots \end{bmatrix}$$

The stationary **probability state** (**steady state) vector** $\vec{p} = (p_0, p_1, p_2, p_3, \ldots)$ we get from the equation $\vec{p}Q = \vec{0}$.

This equation gives the system

$$-\lambda_0 p_0 + \mu_1 p_1 = 0, \qquad \text{(eq1)}$$
$$\lambda_0 p_0 - (\mu_1+\lambda_1)p_1 + \mu_2 p_2 = 0, \qquad \text{(eq2)}$$
$$\lambda_1 p_1 - (\mu_2+\lambda_2)p_2 + \mu_3 p_3 = 0 \qquad \text{(eq3)}$$
$$\lambda_2 p_2 - (\mu_3+\lambda_3)p_3 + \mu_4 p_4 = 0 \qquad \text{(eq4)}$$

Equations eq1, eq2,… are called the **balance equations** because they show the balance between the average rates of entering and leaving each state, so called **rate-in =rate out principle**.

The system is equivalent with the following system:

$$-\lambda_0 p_0 + \mu_1 p_1 - 0, \qquad \text{eqA } (-\text{eq1})$$
$$-\lambda_1 p_1 + \mu_2 p_2 = 0, \qquad \text{eqB } (= \text{eqA+eq2})$$
$$-\lambda_2 p_2 + \mu_3 p_3 = 0 \qquad \text{eqC } (= \text{eqB+eq3})$$
$$-\lambda_3 p_3 + \mu_4 p_4 = 0 \qquad \text{eqD } (= \text{eqC+eq3})$$

We use this system to express the steady state probabilities p_k, k=1, 2, 3… as the functions of p_0.
From equation A, B, C, D… we have

$$p_1 = \frac{\lambda_0}{\mu_1} p_0,$$

$$p_2 = \frac{\lambda_1}{\mu_2} p_1 = \frac{\lambda_0 \lambda_1}{\mu_1 \mu_2} p_0,$$

$$p_3 = \frac{\lambda_2}{\mu_3} p_2 = \frac{\lambda_0 \lambda_1 \lambda_2}{\mu_1 \mu_2 \mu_3} p_0$$

….

$$p_n = \frac{\lambda_{n-1}}{\mu_n} p_{n-1} = \frac{\lambda_0 \lambda_1 \lambda_2 \cdots \lambda_{(n-1)}}{\mu_1 \mu_2 \mu_3 \cdots \mu_n} p_0$$

Substituting p_n, n=0, 1, 2, 3…in the equation

$$p_0 + p_1 + p_2 + p_3 + \cdots = 1,$$

we find p_0.

14. INCOMING TRAFFIC AND SERVICE TIME CHARACTERISATION:

As we know whenever a subscriber originates a call, he adds one to the number of calls arriving at the network and no way by which he can reduce the number of calls that have already arrived. This process can be treated as a special case of B-D process in which the death rate is equal to zero, i.e. no death occurring in the process. Such a process is known as renewal process.

It is a pure birth process in the sense that it can only add to the population as the time goes by and cannot delete the population by itself. The equation arrived in renewal process by setting $\mu_k = 0$ in B-D process.

i.e. $\frac{d\,P_k\,(t)}{dt} = P_{k-1}\,(t)\lambda_{k-1} - \lambda_k P_k$ (t) for k ≥ 1 ………….. 3.17

$\frac{dP_0\,(t)}{dt}$ = $-\lambda_0 P_0$(t) for k=0 ……………………….3.20

$$Let\ the\ time\ t = 0$$

In equation (3.17) and (3.20), the birth rate is dependent upon the state of the system. Let us assume a constant birth rate λ which is independent of the state of the system and then we get a Poisson process. The governing equations of a Poisson process are —

$$\frac{d\,P_k(t)}{dt} = \lambda\,P_{k-1}(t) - \lambda\,P_k(t) \text{ for } k \geq 1 \qquad \text{... (3.21)}$$

$$\frac{d\,P_o(t)}{dt} = -\,\lambda\,P_o(t) \text{ for } k = 0 \qquad \text{... (3.22)}$$

Now, we assume certain boundary conditions *i.e.*, at time $t = 0$, the system is in state zero or no births have taken place. So, we have,

$$P_k(0) = \begin{cases} 1 & \text{for} \quad k = 0 \\ 0 & \text{for} \quad k \neq 0 \end{cases}$$

Thus, equation (3.22) can be written as,

$$P_o(t) = e^{-\lambda t} \qquad \text{... (3.23)}$$

From equations (3.21) and (3.23) we get, for $k = 1$,

$$\frac{d\,P_1(t)}{dt} = -\lambda\,P_1(t) + \lambda e^{-\lambda t}$$

$\Rightarrow$ $\qquad P_1(t) = \lambda\,te^{-\lambda t}$

for $\qquad k = 2,$

$$P_2(t) = \frac{(\lambda t)^2\,e^{-\lambda t}}{2\,!}$$

Thus, the general solution or equation is,

$$\boxed{P_k(t) = \frac{(\lambda t)^k\,e^{-\lambda t}}{k\,!}} \qquad \text{... (3.24)}$$

The equation represents the probability of k arrivals in the time interval t and equation (3.23) express the probability of zero arrival in a given time interval or probability distribution of inter arrival times *i.e.*, the time that elapses between two arrivals. So, in a Poisson Process, the inter arrival time is exponentially distributed as interstate transition times in case of Markov process.

15. BLOCKING MODELS AND LOSS ESTIMATES:

The behavior of loss system is studied by using blocking models. In loss system the over flow of traffic is rejected i.e. the overflow of traffic experiences a blocking from the network[2] . There are three waysin which overflow traffic may be handled:

1. The traffic rejected by one set of resources may be cleared by another set of resources in the network.

2. The traffic may return to the same resource after some time.
3. The traffic may be held by the resource as if being serviced but actually serviced only after the resources become available.

Corresponding to the above three cases, we consider three models of loss systems:

a. Lost calls cleared (LCC)
b. Lost calls returned (LCR)
c. Lost calls held (LCH)

a. Lost calls cleared (LCC):

a.1. Lost calls cleared system with infinite sources :

Let's assume the LCC system using infinite number of subscribers. This model is well suited for study of the behavior of the trunk transmission systems. Usually there are many trunk groups emanating from a switching office and terminating on adjacent switching offices. Whenever a direct trunk group between two switching offices is busy, it is possible to divert the traffic via other switching offices using different trunk groups. In this way the blocked calls in one trunk group are cleared via other trunk group

The LCC model assumes that the subscriber on hearing the engaged tone, hangs up and waits for some length of time before reattempting. He does not reattempt immediately or within a short time. Such calls are considered to have been cleared from the system and the reattempts are treated as new calls.

The LCC model was first studied by A. K. Erlang in 1917.The main purpose of the analysis is to estimate the blocking probability and the grade of service.Consider the Erlang loss system with N fully accessible lines and exponential holding times. The Erlang loss system can be modeled by birth and death process with birth and death rate as follows.

$$\lambda_k = \begin{cases} \lambda, k = 0, 1, \ldots\ldots N-1 \\ 0 \quad k \geq N \end{cases}$$ equation 1

$$\mu_k = \begin{cases} k\mu, k = 0, 1, \ldots\ldots N \\ 0, \quad k > N \end{cases}$$ equation 2

From $$P(k) = \frac{\lambda_0 \, \lambda_1 \ldots\ldots \lambda_{k-1}}{\mu_1 \, \mu_2 \ldots\ldots \mu_k} P(0),\ k = 1, 2, 3$$

Substituting equation 1 and 2 in the above equation, we get

$$P(k) = \frac{1}{k!}\left(\frac{\lambda}{\mu}\right)^k P(0),\ k = 1, 2, 3, \ldots\ldots N$$

From equation (8.4), the offered traffic is

$$A = \frac{\lambda}{\mu}$$

$$P(k) = \frac{1}{k!}(A)^k P(0),\ k = 1, 2, 3, \ldots\ldots N$$

The probability P(0) is determined by the normalization condition

$$\sum_{k=0}^{N} P(k) = P(0) \sum_{k=0}^{N} \frac{A^k}{k!} = 1$$ equation 3

$$P(0) = \frac{1}{\sum_{k=0}^{N} \frac{A^k}{k!}}$$ equation 4

Substituting equation 3 in 4 we get

$$P(k) = \frac{A^k / k!}{\sum_{k=0}^{N} \frac{A^k}{k!}}$$

The probability distribution is called the truncated **Poisson distribution** or **Erlang's loss distribution.** In particular when k = N, the probability of loss is given by

$$P(N) = B(N, A) = \frac{A^N}{N! \sum_{k=0}^{N} \left(\frac{A^k}{k!}\right)}$$ where $A = \lambda/\mu$.

This result is variously referred to as **Erlang's formula of the first kind, the Erlangs-B formula or Erlangs loss formula.**

A.2 Lost Calls Cleared system with finite subscribers:

In case of LCC model with finite subscribers, the call arrival rate is dependent on the number of subscribers, who are not occupied as the busy subscribers do not generate new calls. The traffic in this case is known as Engest traffic or pure chance traffic type 2.

Let, λ_R = arrival rate per subscriber

K = number of busy subscribers.

S = number of server

N = total number of subscriber.

The offered traffic (arrival rate) when the system is in state k is given by,

$$R_k = (N-K)\lambda_R \quad \text{for} \quad k \le 0 \le S$$

The mean offered traffic rate is given by,

$$R = \sum_{k=0}^{S}(N-k)\lambda_R P_k = N\lambda r \sum_{k=0}^{S} P_k - \lambda_R \sum_{k=0}^{S} kP_k$$

$$= \lambda_R \left(N - \sum_{K=0}^{S} kP_k \right)$$

$\left[\sum kP_k \text{ represents the average number of busy servers} \right]$

$$= \lambda_R (N - A_o)$$

The offered traffic is,

$$A = Rt_n = \lambda_R t_h (N - A_o)$$

when the system is in state S, the offered traffic rate is $(N - S)\, \lambda_R$, but all the arrivals are rejected. so, the lost traffic.

$$A - A_o = (N - S)\, \lambda_R P_R t_h$$

Therefore, the grade of service,

$$GOS = \frac{N - S}{N - A_o} P_R$$

So, we see that Engest traffic, the blocking probability and the GOS are not same or we can say that the time congestion and the call congestion values are different.

By analyzing the steady state characteristics of B-D process, we can calculate the blocking probability and grade of service. The expressions are as follows—

$$P_B = \frac{P^R \binom{N}{S}}{\sum_{k=0}^{R} P^k \binom{N}{K}}$$

and

$$GOS = \frac{P^R \binom{N-1}{S}}{\sum_{k=0}^{R} P^k \binom{N-1}{K}}$$

B. Lost Calls Returned System:

LCR model based on the assumption that the blocked calls return to the system as a form of retries.So that the offered traffic comprisestwo components:

Offered traffic = new traffic + retry traffic

The following assumptions are made to analyse the LCR model with regard to the nature of returning calls [1] .

I. Retries calls get the service even if multiple retries are required.

II. No new call is generated when a blocked call is being retried.
III. Time between call blocking and regeneration is random stastistically independent of each other.
IV. Typical working time before a retry is longer than the average holding time .
V. If retries are immidiate ,congestion may occur or the network operation becomes delay system.

Suppose a system with first attempt call arrival ratio of λ=100 (say).If a percentage B of the call is blocked,B times λ retries.

Thus, the infinite series, total arrival rate λ′ is given as,

$$\lambda' = \lambda + B\lambda + B^2\lambda + B^3\lambda + \ldots\ldots$$

$$\boxed{\lambda' = \frac{\lambda}{1-B}}$$

where B is the blocking probability.

Similarly we can say

$$\boxed{\lambda' = \frac{\lambda}{1-GOS}}$$

The effect of returning traffic is insignificant when operating at low blocking probabilities or grade of service.

C. Lost Calls Held System:

I. In this case, the blocked calls are held in a queue and serviced when necessary facilities become available.
II. In this system, the total time is not dependent on the waiting time. The total time is determined by the average service time required.
III. The sources need service continuously for a period of time whether or not the services are available. As the service becomes free from other calls it handles the queued calls.
IV. If a number of calls blocked, a portion of it is lost until a server become free to service a call as shown in figure

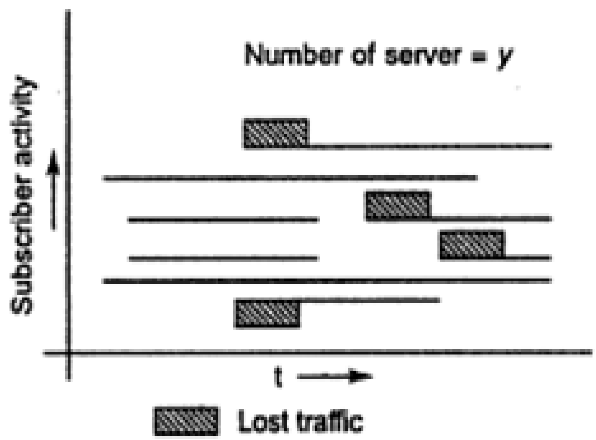


Fig22: Lost traffic in LCH Model. [1]

LCH model is used to find the probability of the total number of calls in the system at a particular time. The number of active sources is identical to the number of call arrivals in that particular time.
i.e. given as Poisson distribution equation

$$P(x) = \frac{\mu^x}{x!} e^{-\mu}$$

16. DELAY SYSTEMS:

1. In telecommunication network, such as data network places the call or messages in a queue in the absence of resources and services them when the resources become available. Such systems are known as delay systems, which are also called lost call delayed systems [1]. For example:
   - Message switching
   - Packet switching
   - Digital receiver access
   - Automatic call distribution
   - Call processing

2. Delay systems are analyzed using queuing theory or waiting line theory. The elements of a queuing system are shown in figure

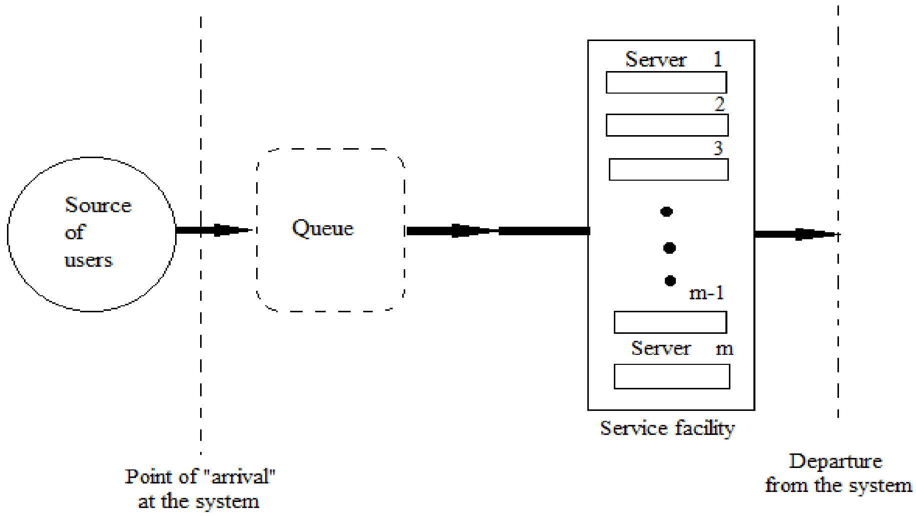


Fig23: Queuing diagram of users waiting to be served. [1]

There is a large number of populations of sources that generates traffic or service requests to the network. There is a service facility that contains a number of identical servers, each of which is capable of providing the desired service to a request. When all the servers are busy, a request arriving at the network is placed in a queue until a server becomes available. Hence

$K = k_q + R$, i.e. the mean time a call or a request spends in the system is the sum of the mean wait time t_q and mean service or holding time t_h.

3. If a delay system has infinite queue capacity during operation then a necessary condition for the system is stable is:

   $\frac{mean\ arrival\ rate}{mean\ service\ rate} < 1$ or $\frac{offered\ traffic}{number\ of services} < 1$

4. A queuing system is characterized by a set of six parameters such as A/B/c/K/m/Z and the parameter specifications are

   A= Arrival process specification (values are GI, G, ER, M, D, Hk)

   B= Service time distribution (values are GI, G, E_R, M, D, H_k)

   c = number of servers (nonzero positive finite number)

   K= queue capacity (may or be finite an infinite number)

   m = number of sources (input population) (may or be finite an infinite number)

   Z=service discipline (first-come-first-served (FCFS))

17. TELEPHONE NETWORK

A telephone network is a telecommunications network used for telephone calls between two or more parties.

There are a number of different types of telephone network:

- A landline network where the telephones must be directly wired into a single telephone exchange. This is known as the public switched telephone network or PSTN.
- A wireless network where the telephones are mobile and can move around anywhere within the coverage area.
- A private network where a closed group of telephones are connected primarily to each other and use a gateway to reach the outside world. This is usually used inside companies and call centers and is called a private branch exchange (PBX).

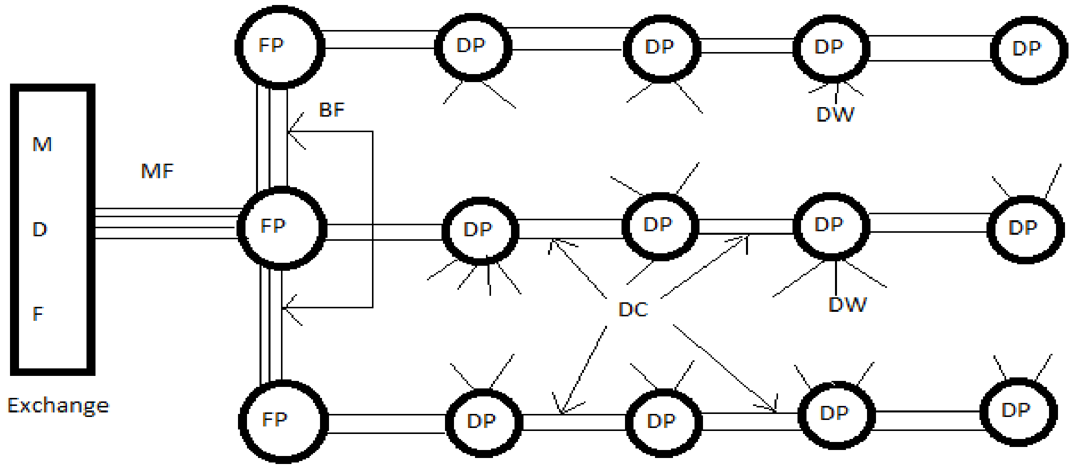
Public telephone operators (PTOs) own and build networks of the first two types and provide services to the public under license from the national government. Virtual Network Operators (VNOs) lease capacity wholesale from the PTOs and sell on telephony service to the public directly.

A telecommunication network may be categorized into major systems:

1. Subscriber end instruments
2. Subscriber loop systems
3. Switching systems
4. Transmission systems
5. Signaling systems

Subscriber loop system

In a telephone network every subscriber is connected generally to the nearest switching office by means of dedicated pair of wires. Its very difficult to connect each individual pairs from every subscriber premises to the exchange. So, generally four levels of cabling are used as shown below



MDF=main distribution frame MF=main feeder FP=feeder point BF=branch feeder DP=distribution point DC=distribution cable DW=drop wires

Fig. 24: Cable hierarchy for subscriber loops [1]

In a communications network, a drop is the portion of a device directly connected to the internal station facilities, such as toward a telephone switchboard, toward a switching center, or toward a telephone exchange. A drop can also be a wire or cable from a pole or cable terminus to a building, in which case it may be referred to as a down lead. These cables may be reinforced to withstand the tension (due to gravity and weather) of an aerial drop (i.e., hanging in air), as in "messenger" type RG-6 coaxial cable, which is reinforced with a steel messenger wire along its length. These wires are individual pairs that run into subscriber premises and they are connected to wire pairs in the distribution cables at the distribution point. This cable can also be used as an external grade telephone cable and be clipped to the side of a building. A drop wire is over head telephone wire or cable used from a telegraph pole to a house or a building, drop wire no 10 has 2 pairs (4 wires) with 3 straining wires to withstand the tension (due to the long length of the cable and weather) along its entire length.

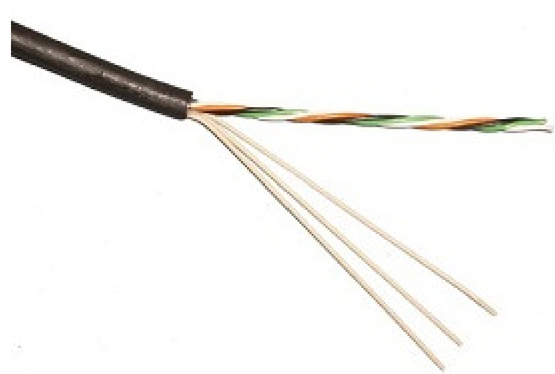


Fig.25: Drop wire no 10 [3]

Thus, many such distribution cables from nearby geographical regions are terminated on a feeder point where they are connected to branch feeder cables which are thus connected to main feeder cable. These DC carry only 10-500 pairs while MF cables large numbers i.e. 100-2000 wire pairs. So, such MF cables are terminated on a MDF at the exchange and moreover the subscriber cable pairs emanating from exchange also gets terminated on MDF.

MDF:The MDF is a termination point within the local telephone exchange where exchange equipment and terminations of local loops are connected by jumper wires at the MDF. All cable copper pairs supplying services through user telephone lines are terminated at the MDF and distributed through the MDF to equipment within the local exchange e.g. repeaters and DSLAM. Cables to intermediate distribution frames (IDF) terminate at the MDF. Trunk cables may terminate on the same MDF or on a separate trunk main distribution frame (TMDF). Like other distribution frames the MDF provides flexibility in assigning facilities, at lower cost and higher capacity than a patch panel. The most common kind of large MDF is a long steel rack accessible from both sides

Each jumper is a twisted pair. The MDF usually holds telephone exchange protective devices including heat coils, and functions as a test point between a line and the exchange equipment. Due to its flexibility it is very useful in reallocating cable pairs and subscriber numbers. For example: when any subscriber moves his house to a nearby served by same exchange but different DP then he can be allowed to have the same number or if he releases his wires then he can be given a new number and that number is given another subscriber. Moreover feeder and distribution points have flexible cross point connection capability in newer installations which helps in easy reconnection of subscriber drop wire to any pair of wire in the distribution cables and there after any pair from DC to any other pair of feeder cable at FP. These helps in utilization of the cable pairs efficiently as well as management during faults for e.g.: if particular cable is faulty then important subscribers assigned to this cable can be reassigned to free pairs in other cables.

From economy point of view it is desirable that the subscriber loop lengths are as large as possible so that single exchange can serve large area. But there are two factors that limit their length.

(1) Signaling

(2) Attenuation

Signaling: as discussed before dc signaling is used for subscriber lines like off-hook signals and dial pulses. A certain minimum current is required to perform these signaling functions properlyand thus exchanges must be designed so that they can deliver such a current. A bound on loop resistance also limits the loop length for a given gauge wire. So, the dc loop resistance Rdc for copper conductors is $Rdc = 21.96/d^2$ ohms/km

Where d = diameter of conductor in mm

As resistance is a function of temperature equation holds good for resistance values at 20°C. So, smaller gauge wires use thicker conductors and often less dc resistance per unit length

Attenuation:its limit arises from ac response of the loop and refers to loop loss in decibels. The criteria here are to ensure that the quality of reception at the subscriber end is satisfactory.

- A need arises to connect to existing subscribers who are located beyond the maximum prescribed distance. For example: it is uneconomical to install a new exchange for alone or a few remote subscribers in such cases special technique is used to meet the resistance and attenuation constraints.
  So, the dc resistance constraint is met by:
  (1) Use of higher diameter(lower gauge)
  (2) Use of equalized telephone sets wire
  (3) Unigauge design or use of higher supply voltage.
  And attenuation constraints are overcome usually by the use of loading coils
  Moreover as significant portion (30-40%) the cost of telecommunication network comes from cost of copper in the subscriber lines so a larger value of loop resistance is acceptable.

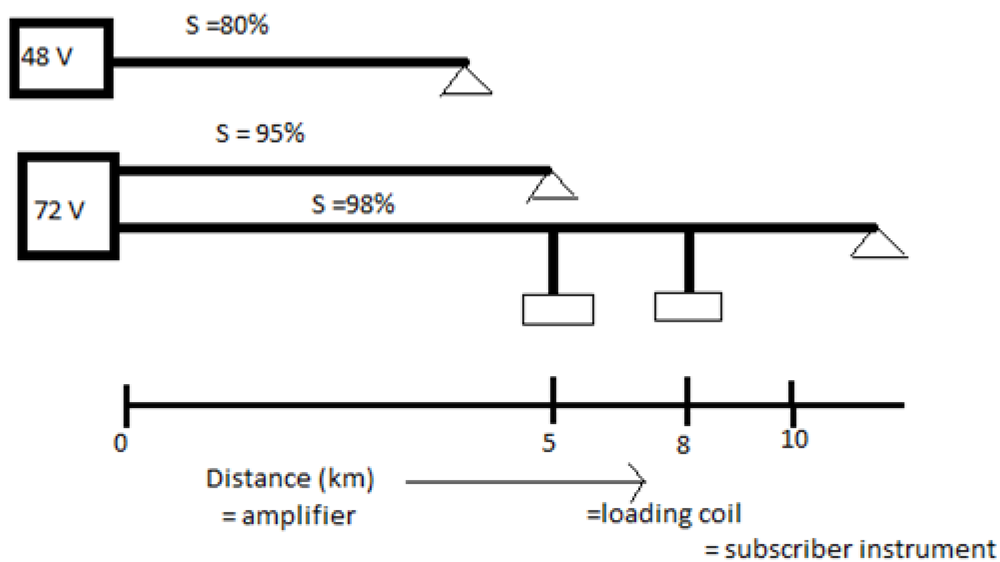


Fig.26: Unigauge design of subscriber loops [1]

This design attempts to use wire with as small a diameter as possible while retaining the resistance and attenuation limits. For both long and short distances, the same gauge wire is used and hence the name unigauge design.

- Loading coils are identified by using a standard convention as 19-D-44, 24-A-88 etc. Here the 1st number indicates with which coil the wire gauge is used,the letter specifies the spacing between the coils and the last number specifies the inductance value in mH. This table 3 below shows the standard letters and their associated spacing and the most commonly used spacing is B,D and H.

Table 3- Loading coil spacing [1]

Letter code	A	B	C	D	E	F	H	X	Y
Spacing (km)	0.21	0.92	0.28	1.37	1.7	0.85	1.83	0.2	0.65

- In rural places subscribers are generally dispersed and so it is both unnecessary and expensive to provide a dedicated pair for every subscriber. Hence, 3 techniques are used to gain on number pairs:
    (1) Party lines
    (2) Concentrators
    (3) Carrier systems

  Party lines: here two or more subscribers are connected to one line termed as party line. This technique is generally not used due to many drawbacks. Only one subscriber can be used at a time. Selective ringing is difficult and privacy is not maintained and dialling between two subscribers on the same line is not possible.

  Concentrator: here a concentrator expander is used near the cluster of users and another one at exchange end as shown

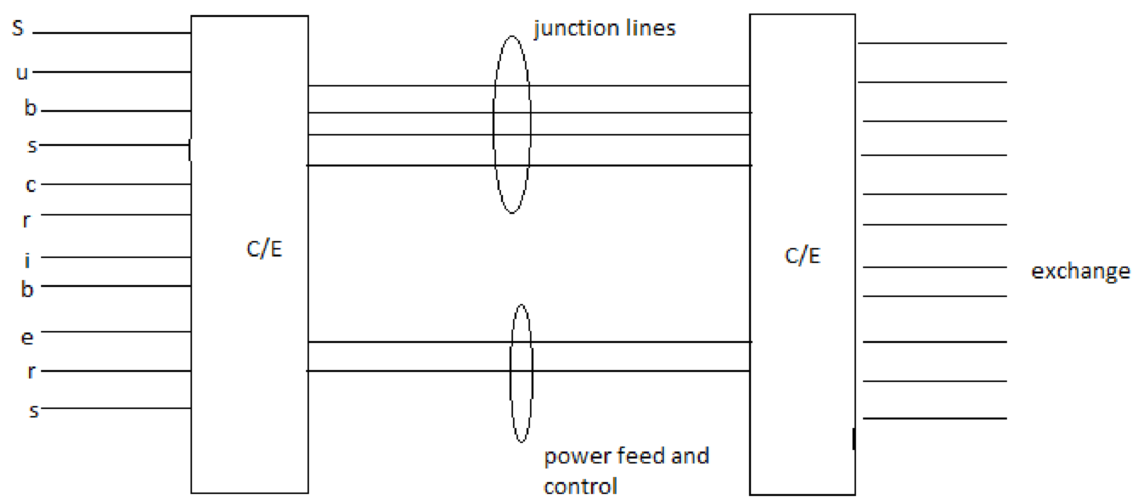


Fig. 27: Concentrator-expander connection for dispersed subscribers [1]

Only few junction lines are run between the CEs which have switching capability. A ratio of 1:10 between the junction lines is used and the CE at the exchange end remotely powers and controls the CE at the subscriber end.

Carrier systems: they employ multiplexing techniques and enable all the users to access the exchange over a single line, analog FDM and digital TDM systems are used.

Signalling and voice transmission of the subscriber lines requires that the exchange performs a set of functions. These functions are performed by an interface at the exchange end called as subscriber loop interface.

The complete set of functions is known by an acronym BORSCHT which stands for:

B = battery feed
O = overvoltage protection
R = ringing
S = supervision
C = coding
H = hybrid
T = test

Functions B and R are well known. Overvoltage protection deals with equipment and personnel protection from lightining strikes and power line surges. Detection of off-hook condition is a supervisiory function. Functions C and H are exclusive to digital switch interfaces. As we know, digital switching demands that analog to digital and digital to analog conversions and some form of coding/decoding be done. Subscribers are connected to the exchange via 2-wire circuits.these circuits use balanced connections as shown below:

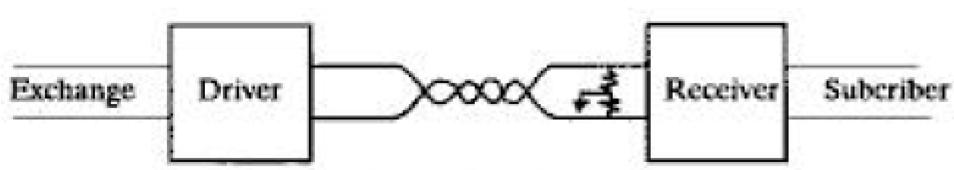


Fig.28: balanced circuit connection [1]

Balanced connections overcome many drawbacks of unbalanced circuits. The transmission lines have equal impedances to ground and hence don't act as an antenna to pick up signals. Since the ground is not a part of the signal path and hence is eliminated. Differential inputs improve noise immunity as any interface affects both lines equally and does not introduce differential currents.

Digital exchanges require receive and transmit signals on separate 2-wire circuits. This call for 2-wire to 4-wire conversion and the circuit that performs it is called hybrid. Such a conversion is normally required for trunk transmissions in analog exchanges. The circuit is shown below. The main function of a hybrid is to ensure that there is no coupling of signal from the input to the output in 4-wire circuit.

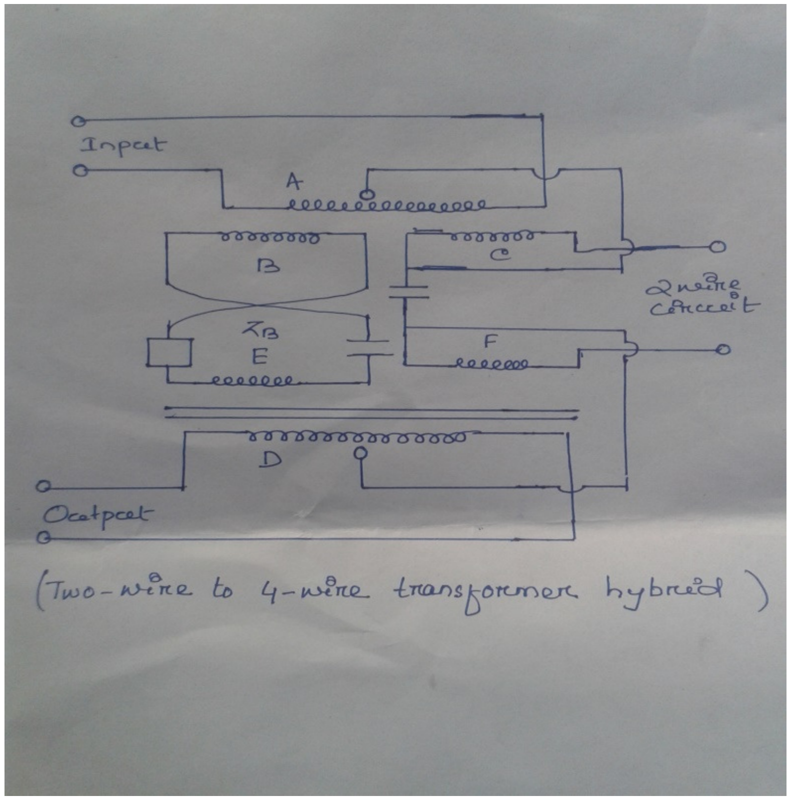


Fig.29: two- wire to 4-wire transformer hybrid [1]

The operation of the circuit is as follows: The input signal is coupled to the B and F windings equally. Through the C winding, the input is coupled to the 2-wire circuit. The same signal when it flows the balanced 2-wire couples the signal to winding D through winding C. The signal induced in B flows through E and induces a current in D that opposes the current induced by F. If the impedance Z_B exactly matches that of the 2-wire circuit, the effect of input signal from the subscriber end is completely nullified from coupling into the winding A.

Integrated circuit manufacturer have successfully developed a single chip called subscriber loop interface circuit (SLIC) and an all-electronic telephone automatic redialing is feature that enabled in the telephone instrument if we use a microprocessor.

18. Switching Hierarchy and Routing

- Telephone networks require some form of interconnection of switching exchanges to route traffic effectively and economically. Exchanges are interconnected by groups of trunks called as trunk groups that carry traffic in one direction.
- Between any Two exchanges 2 trunk groups are required and 3 basic topologies are adopted for interconnecting exchanges: mesh, star and hierarchy. Where mesh is a fully

connected network which is used mostly in heavy traffic among exchanges i.e. in metropolitan area as the number of trunk groups are proportional to the square of exchanges being interconnected.

- A star connection utilizes an intermediate exchange called a tandem exchange through which all other exchanges communicate.
- An orderly construction of multilevel star networks leads to hierarchical networks.

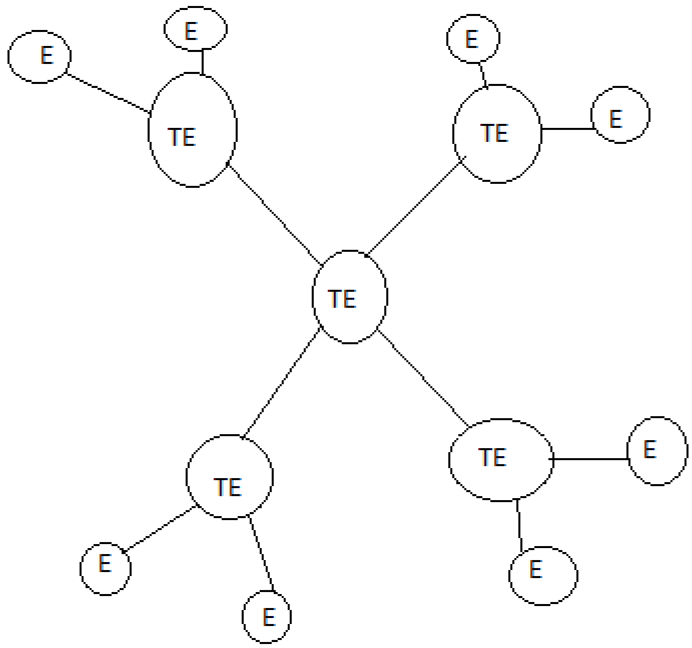


Fig. 30: Two level star [1]

Hierarchical networks are capable of handling heavy traffic where required, and at the same time use minimal number of trunk groups. A 5-level switching hierarchy is recommended by CCITT as shown fig. 27 below. In a strictly hierarchical network, traffic from subscriber A to subscriber B and vice versa flows through the highest level of hierarchy, viz quaternary centers in fig.27 below. A traffic route via the highest level of the hierarchy is known as final route. However, if there is high traffic intensity between any pair of exchanges, direct trunk groups may be established between them as shown in figures below. These direct routes are known as high usage routes.

Three methods are commonly used for deciding on the route for a particular connection:

1. Right-through routing
2. Own-exchange routing
3. Computer-controlled routing

In right-through routing the originating exchange determines the complete route from source to destination. No routing decisions are taken at the intermediate routes.Own-exchange routing or distributed routing allows alternative routes to be chosen at the intermediate nodes. Thus the strategy is capable of responding to changes in traffic loads and network configurations.Computers are used in networks with common channel signaling (CCS) features. With computer in position, a number of sophisticated route selection methods can be implemented.

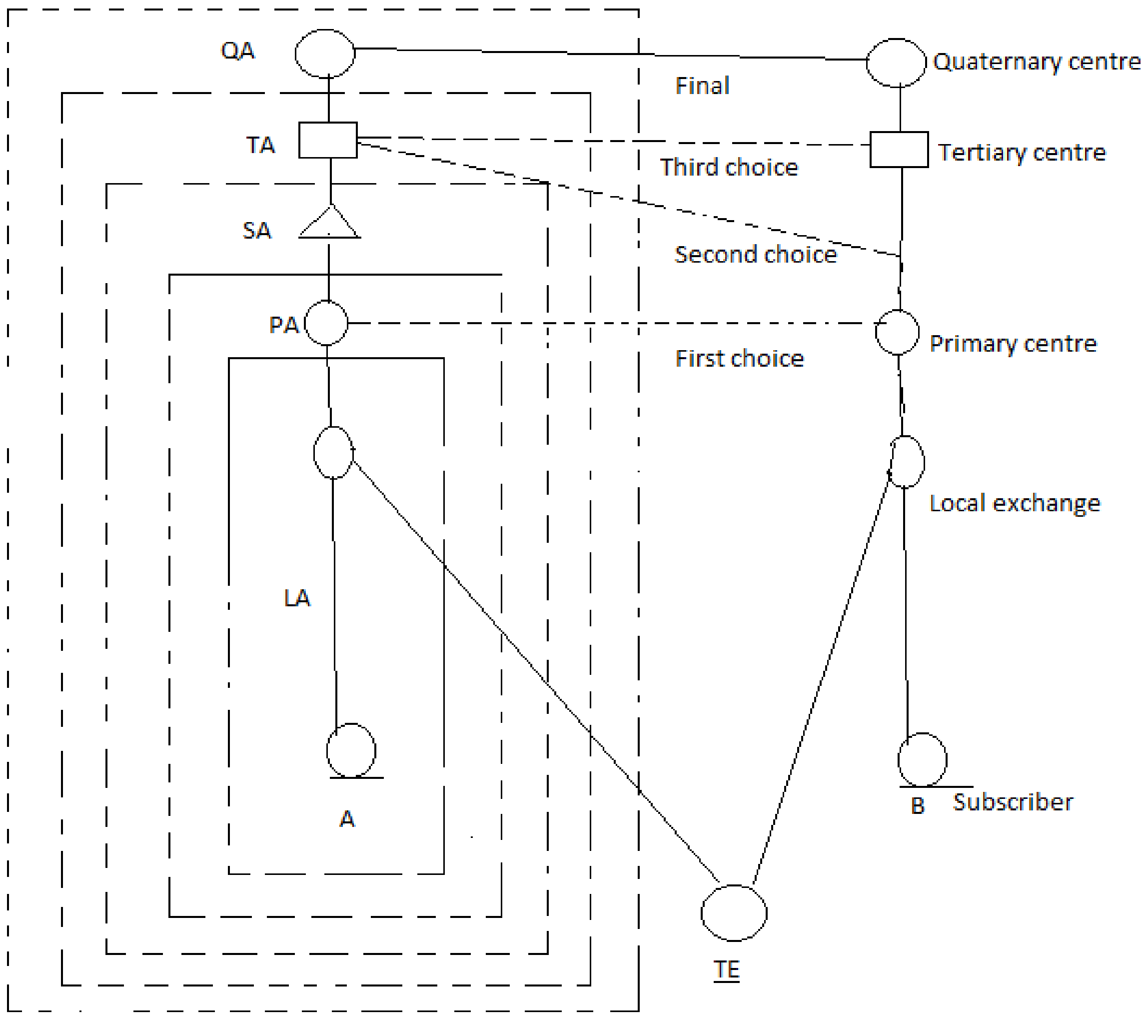


Fig.31: A five level CCITT hierarchical structure [1]

18.1 Transmission Plan

For reasons of transmission quality and efficiency of operation of signalling, it is desirable to limit the number of circuits connected in tandem. In a tandem chain, the apportionment of links between national and international circuits is necessary to ensure "quality" telecommunications. CCITT guidelines in this regard are:

1. The maximum number of circuits to be used in an international call is 12
2. No more than 4 international circuits to be used in tandem between the originating and the terminating international switching centers.

3. In exceptional cases and for low number of calls the total number of circuits may be 14, but even in this case, the international circuits are limited to a maximum of 4.

Taking guidelines 1 & 2 we have 8 links available for national circuits, which implies a limit of four for each national circuits.

The transmission loss is defined in terms of reference equivalents of TRE, ORE and RRE. CCITT recommends that for 97% of the connections the maximum TRE is limited to 20.8dB and RRE to 12.2dB between the subscriber and the international interface in the national network. This gives an overall reference equivalent ORE of 33 db. From one country to another the OREs range is from 6 to 26 db.

Transmission losses budget should provide for 2 factors other than line and switch losses:

1. Keeping echo levels within limits
2. Control singing.

In analog exchanges, local calls are established on 2 wire circuits. But long distance calls require 2 wire to 4 wire conversion at the subscriber line trunk interface. Due to long distance involved, the bearer circuits need amplifiers or repeaters at appropriate intervals to boost the signals. The amplifiers are almost invariably one way devices. Since for long distance circuits need separate for each direction, leading to conversion into a 4 wire circuits.

The important function of the hybrid is to ensure that the received signal is not coupled. The coupling is zero only when the 2 wire circuit and the 4 wire circuit impedances are perfectly matched. However impedance mismatch occurs in most of the connections at the subscriber line trunk interface. The effect of such mismatch results in echo. If the distance are short , the round trip delay experienced by the echo is small and becomes unnoticeable.

The echo suppressors are voice attenuators. Normally the echo suppressors remain in a deactivated state. Speech in one channel activates the echo suppressor in the return path. One drawback of echo suppressors is that they may clip the beginning portion of speech segments. New designsof echo suppressors attempt to minimize the time required to reverse directions. Typical reversal times are in the range of 2-5 ms.

The operation of a system with echo suppressors is clearly half duplex. When telephone lines are used for data transmission full duplex operation is required. Echo suppressors are usually turned off for data transmission as it is very difficult to organize data transmission. This is done by providing a disabler feature in the echo suppressor and triggering the same with a special signal. Usually a 2025 Hz or 2100 Hz tone transmitted for at least duration of 300 ms with a signal not less than -5 dB is used to trigger the disabler.Once the time increases the echo becomes noticeable and annoying to the speaker.

Short delay echo are controlled byusing attenuators and the long delay ones by echo suppressors or echo cancellers. CCITT recommends use of the echo cancellers if round rip delay exceeds 50 ms. For delay up to 50 ms simple attenuators in the transmission path limit the loudness of echo to a tolerable level. The attenuation increases as the delay increases.

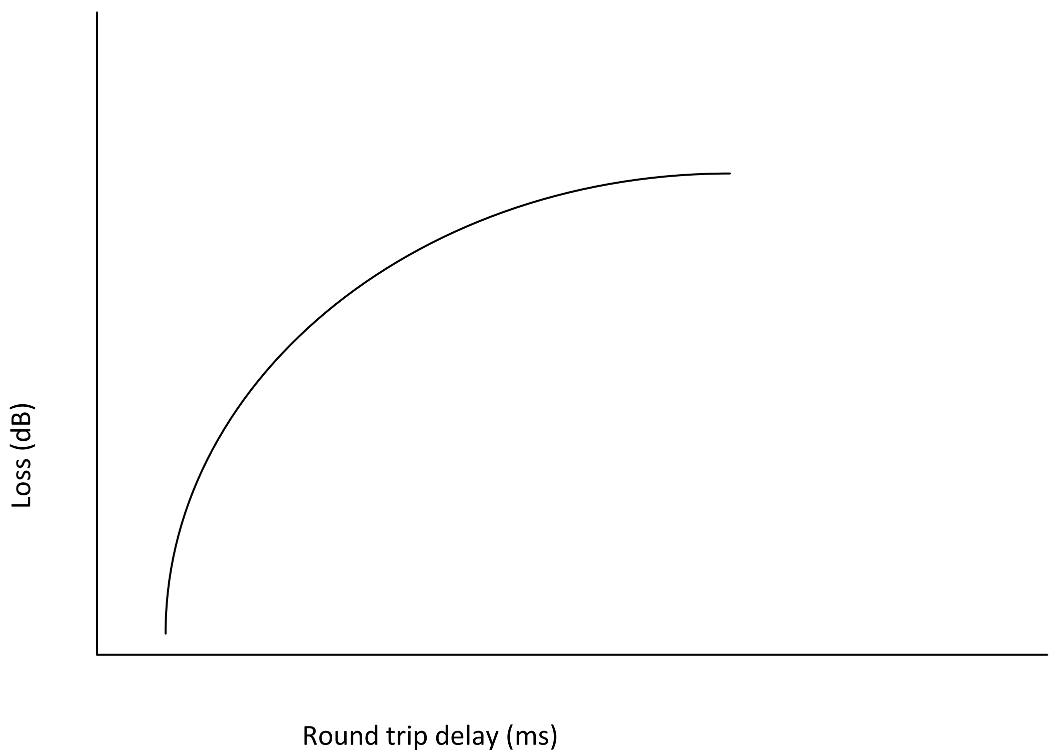


Fig.32: Transmission systems [1]

Long distance transmission system can be placed into 3 categories:

(1) Radio systems
(2) Coaxial cable system
(3) Optical fiber system

- A radio communication system sends signals by radio. Types of radio communication systems deployed depend on technology, standards, regulations, radio spectrum allocation, user requirements, service positioning, and investment.
- The radio equipment involved in communication systems includes a transmitter and a receiver, each having an antenna and appropriate terminal equipment such as a

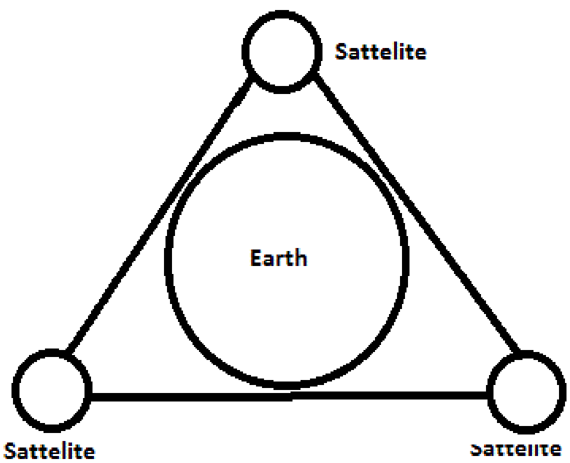
microphone at the transmitter and a loudspeaker at the receiver in the case of a voice-communication system.

- The power consumed in a transmitting station varies depending on the distance of communication and the transmission conditions. The power received at the receiving station is usually only a tiny fraction of the transmitter's output, since communication depends on receiving the information, not the energy that was transmitted.

19. Satellite Communication

Satellite is powerful long distance and point-to multi point communication system. A communication satellite is an R.F (Radio Frequency) repeater. To overcome disadvantage of Line of sight communication which is only 45 - 55 km, the transmitting antenna is placed on the satellite and the satellite is placed in the orbit high above the earth. The function of satellite is to communicate between different earth stations around the earth, thus with the help of satellite, it is easy to communicate over thousands of km, a com-satellite is a combination of ROCKET to put the satellite in the orbit, micro wave electronic devices for the communication, solar cells are used to convert the solar energy into a power supply (ELECTRICAL ENERGY) for the electronic equipment.

The satellite placed in GEO- STATIONARY and placed at an altitude of 22300 miles or 35900 km above the ground level. The satellite travels at the same speed at which the earth rotates around the sun. The rotation of satellite is synchronized with earth rotation as a result satellite appears to be stationary in the sky w.r.t the earth station is constant. There are 3 satellites are placed at angle 120° in GEO-STATIONARY orbit, they provide 100% coverage from one earth station to anywhere on the earth, this concept is shown below



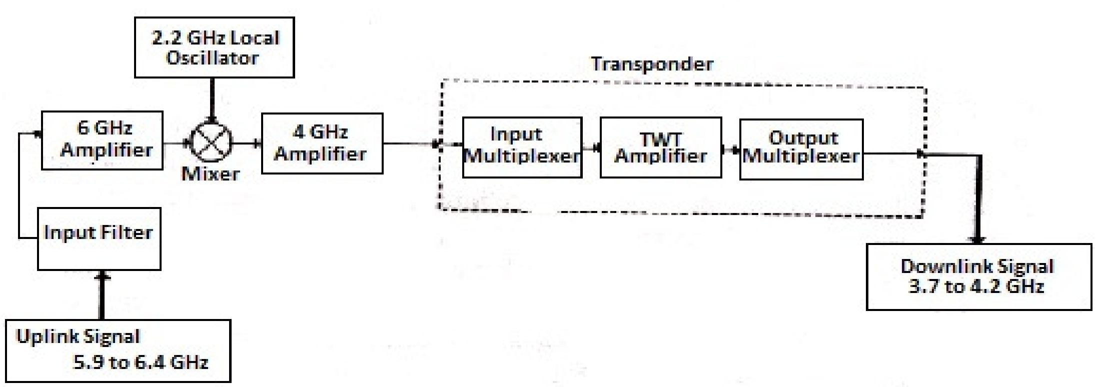


Fig.33: Block Diagram of Satellite Communication System [5]

The uplink frequencies (5.9---6.4 GHZ) are used for T/N from the earth station to the satellite and down link frequencies (3.7—4.2GHZ).

The above frequencies are used for T/N from the satellite to the earth station , the uplink frequencies are converted to lower frequencies by the mixer and local Osc, the com satellite acts as a repeater station it receives the signal, amplifiers it and then transmitted over a next frequencies to avoid interference between the uplink signal and down link , the two way communication is established with the help of transponder , a com satellite has multi transponder per satellite has increased over the year ,a satellite with 2 transponder can support a signal T.V channel or 240 telephone lines , a satellite with 48 transponder can accommodate 4000T.P CKTS and 2 T.V channels now-a-days in satellite using a digital tech , due to which 0ne satellite can handle 120,000 T.P4 channels and more than 500 T.V channels.

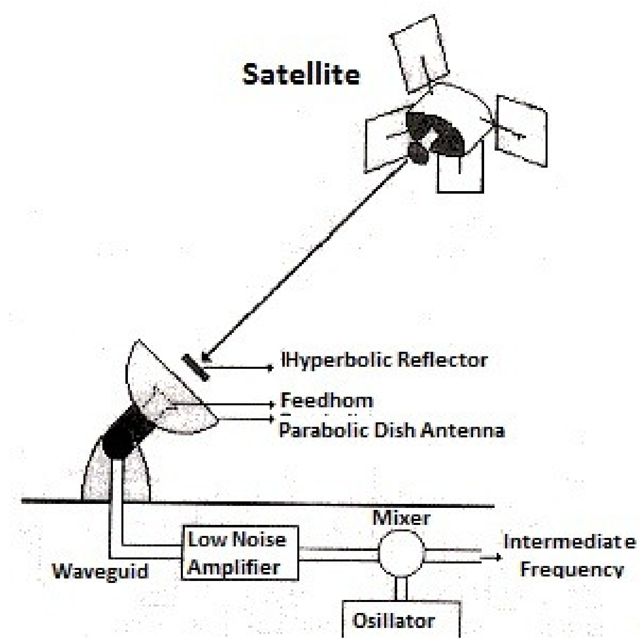


Fig.34: Satellite Communication Earth Station [2]

The equipment used in satellite earth station are shown in fig , the earth station consist of a dish antenna transmitter which can transmit a high frequencies (5.9—6.4GHZ) micro wave signals, some earth stations also called ground station , which can transmit and receive the signals while others can only receive signals.

A high directive and a high gain antenna is necessary at the earth station , because the losses over the long T/N path is very high , the signals power reaching back to the earth station from satellite is very small . Therefore at receiving end a parabolic dish antenna with 61m diameter provides a high gain and thus amplify the signal power, it is important to have a low noise amplifier before the mixer stage in the receiver C, K, T at the satellite earth terminal.

19.1 Geostationary Satellite

The satellites were placed in low earth orbit. as a result the satellite at a such high speed that it visible to the ground only for a short time at each day , the satellite appeared below the horizon and dies appear below the opposite horizon , the ground station was cut-off for long time in day , to maintain the communication link another station had to be activated , this problem was solved by placing the satellite in circular orbit of approximately 22300 miles or 35900 km radius, as the satellite height increases from the earth surface , the speed of satellite decreases by the same manner , at that height the angular velocity of satellite will be proportional to the angular velocity of earth , the satellite rotates with the same speed as that of the earth due to which the satellite will always be at the same place where it has been fixed , this type of satellite is called geo stationary satellite.

19.2 Telephone Link via Satellite

The satellite communication can be used for Telephone telecom. Around the world, the block diagram of such a system is shown in fig.31. The block diagram of earth station working with three satellites here, the national long distance Telephone network of a 4 countries (A,B,C,D,) through international switching centre are connected, consider country “A” the 0/p of the Telephone exchange is applied to the MUX, the multiplexed signal is send to the micro-wave station and from there to the satellite earth station , at the earth station the signal is multiplexed and directly applied to the modulator stage of earth station where it demodulated with a high frequency signal and transmitted towards the satellite as uplink , in other case the earth station “A” receive three down link signal , the 3 carriers are demodulated and then transmitted toward the micro-wave station and from there international switching centre.

Many earth stations are designed to transmit several carriers from direct communication with other station through one satellite; the other wire (OW) facilities are transmitted for message carriers from the band of 300HZ----12KHZ,

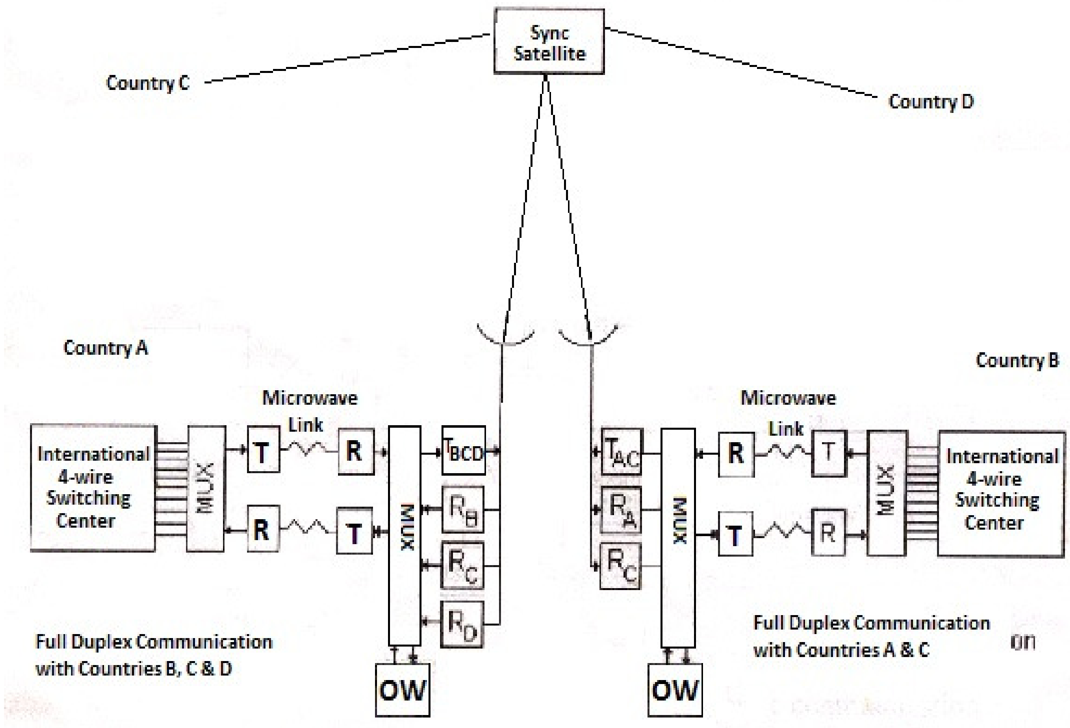


Fig.35: Satellite communication used for telecom [2]

Merits

1. No tracking is required by Geostationary Satellites.
2. Multiple access points are available in Satellite communication.
3. 24 hour communication can be achieved with the help of satellite.
4. The signal quality of Satellite communication is higher.
5. To put more information on the carrier a broad band can be used.
6. Satellite Communication is used for long distance communication or across oceans.
7. Low transmitting Power and low receiver sensitivity is required by the Satellite in close elliptical orbits.

Demerits

1. The transmitter and receiver used in satellite communication require high power, most sensitive transmitters and large diameter antennas.
2. Satellite communication is disturbed by solar activities and cyclones in the space.
3. Due to ageing effect the efficiency of Satellite components decreases.
4. The longer propagation times (APPOX, 300ms) is one of a disadvantage of satellite communication.
5. The cost for Initial design and launching of the satellite in the orbit results in extremely high.

20. Numbering plans

The objective of numbering plan is to uniquely identify every subscriber connected to a telecommunication network. In early stages of development a numbering scheme was confined to a single local exchange, and exchanges were identified by the names of the towns in which they were located. This scheme works well as long as there is only one exchange per town. But as the subscriber volume grew, it became necessary to introduce more than one exchange in a town. Generally, a largecentrally located exchange called the **main exchange** serving the main business center of the town, and a number of smaller exchanges known as **satellite exchanges** serving different residential localities were used to cope with the growing traffic in a large area. The area containing the complete network of the main exchange, and the satellite is known as **multiexchange area.** A common numbering scheme was then required for the area so that the digits dialed to identify a given terminating exchange do not vary with the exchange originating the call. For call originating from a location outside the multi-exchange area, there is a need to identify the area by a common code. The common numbering scheme is sometimes called as the **linked numbering scheme.** In this scheme, all exchanges in a townwere collectively identified by the name of the town.The introduction of subscriber **trunkdialing(STD)** or **directdistance dialing(DDD)** for intercity and intertown long distance connection called for a national numbering plan, where multiexchange areas are identified uniquely by numbers. Subsequent development of **internationalsubscriber dialing (ISD)** makes it necessary to have an international numbering plans and to have the national numbering plan conform to the international one. A numbering plan may be open, semi open, or closed. An **open numbering plan,** also known as **non-uniform numbering scheme**, permit wide variation in the number of digits to be used to identify a subscriber within a multiexchange area or within a country. This plan is used in countries equipped extensively with non-Director Strowger switching systems. In such cases, the numbering scheme is usually an exact image of the network structure and requires to be changed if the networkstructure changes.

A **semiopen plan** permits number lengths to differ by at most one or two digits. Today the scheme is the most common and is used in many countries including India, Sweden, Switzerland, and the United Kingdom. In **closed numbering plan** the **uniform numbering scheme,** the number of digits in a subscriber number is fixed. This scheme is used by a few countries which include from France, Belgium and the countries in the North America(USA,Canada). An **international numbering plan** or **world numbering plan** has been refined by **CCITT** in its recommendation. For the number of purposes, the world is divided into zones. Each zone is given a single digit code. For the European zone, two codes have been allotted because of the large number of countries within this zone. Every international telephone number consists of two parts.

The country code contains one, two or three digits, the first digit being the zone code in which the country lies. For example France has the country code '33'. In cases where an integrated numbering plan already covers an entire zone, the countries in that zone are identified by the single digit zone code itself. All the countries in the North America zone have the code as '1' and all the countries in the USSR have the code as '7'. It is not clear if this code would change on account of the recent changes in the political setup of the USSR and the formation of the new commonwealth of Independent States (CIS).

The existence of a world numbering plan places restrictions on the national numbering plan of each country. The number of digits in an international subscriber number is limited to an absolute maximum of 12. In practice, with a few exceptions, world numbers are limited to 11 digits. As a result, the number of digits available for a national numbering plans is11-N where N is the number ofdigits in the country code. In general, a national number consists of three parts. The **area** or **trunk code** identifies a particular numbering area or the multiexchange area of the called subscriber, and thus determines the routing for a trunk call and the charge for it. A numbering area is defined as that area in which any two subscribers use identical dialing procedure to reach any other subscriber in the network. An **exchangecode** identifies a particular exchange within a numbering area. It determines the routing for incoming trunk call from another numbering area or for a call originating from one exchange and destined to another in the same numbering area. **Subscriber line number** is used to select the called subscriber line at the terminating exchange. In CCITT terminology, the combination of the exchange code and the subscriber line number is known as the **subscriber number** which is the number listed in the telephone directory. The term ' local call' here implies a call within a numbering area and the term 'national call' a trunk call between two different numbering areas within the same country. Basically there are four possible approaches to dialing procedures:

1. Use a single uniform procedure for all calls, viz. local, national and international calls.
2. Use two different procedures, one for international calls and the other for local and nationalCalls.
3. Use three different procedures, one for international calls, second for national trunk calls,and the third for local calls.
4. Use four different procedures, three procedures same as given in 3 above and a fourth procedure for calls in the adjacent numbering areas.

Approach 1 demands that 11 or 12 digits be dialed uniformly for any number to be obtained. This places an unnecessary burden on the subscriber and the digit processing subsystems of the exchanges and hence the approach is not retorted to. In approach 2 , a common procedure of dialing the full national number for both local and national trunk calls is adopted. This is justified only if the national trunk traffic represents a high proportion of the total traffic which generally is not the case. Hence the approach is also not favored.Approach 3 is commonly adopted by most of the nations. Here a subscriber directory number is dialed for calls within a numbering area, a national number for national trunk calls, and an international number for foreign calls. A need to distinguish one class of numbers from the other now arises. A standard technique that has been adopted for this purpose by most of the countries is the use of a set of one or more **prefixdigits.** Since the first digit of the country codes can be any of the digits 1-9, the prefix digit conveniently starts with a zero. Usually, a single digit '0' prefix is used to distinguish between a local call and a national call. A two digit '00' or a three digit, e.g. '010' prefix is used to differentiate between national and international calls. The first prefix digit '0' routes a call to atrunk exchange and the following prefix digit(s), if any , causes the call to be switched to the international gateway exchange. The dialing procedure calls for the required prefix to be dialed followed by the appropriate (national or international number).Approach 4 suggests special short dialing codes for adjacent numbering areas. This approach tends to reduce the number space available to users, hence is not extensively used.Size and delineation of numbering areas require careful consideration. Large numbering areas necessitate long and expensive junctions from satellite exchanges to central trunk switching centers. A numbering plan must make a generous allowance for growth in the number of subscribers for up to 50 years ahead. The numbering area should not be so large that the number of subscribers it will ultimately serve is beyond its

numbering capacity. A large number of subscribers within a numbering area imply long subscriber numbers and cumbersome dialing. A maximum size of seven digits for local numbers appears to be at the acceptable limit. If the numbering area is small, the number of such areas is large and so is the number of trunk switching centers. In such a case, the number of area codes available in the national numbering plan may run out too soon. If the numbering areas are small, nearby exchanges having considerable traffic between them may belong to different numbering areas, thus complicating the dialing procedure and entailing higher charges.
Any special provision in numbering scheme has the effect of reducing the number space available for identifying unique subscribers.Another important aspect of numbering plan is direct inward dialing(DID) with the advancement of electronics switching must of the PABXS provide direct dialing access to the public networks i.e. the direct outward dialing(DOD) from extension. However the incoming calls are routed through the operator.Increasingly the customers now desire that it should be possible to dial a PABX extension directly from a public network without having to go through the operator. The DID facility is desirable particularly in the context of any distance dialing when a caller has to pay for the time that lapses before the operator establishes a connection to the desire extension.

There are basically two approaches to provide DID facility:-

(i) Use of set suffix digits to the national number to identify PABX extension.
(ii) Allot national number to a PABX extension.

Under the restriction of maximum 12 digits in the international no. approaches & has the effect of increasing the length of the subscriber no. and thereby reducing the digits available for exchange and area codes and approaches to reduce the no. of subscribers who can be accommodated within the numbering scheme of the local exchange. Both approach reduce the effective no. space available, approach 1 more severely than approach 2.A better approach would be to add suffix digits over and above the international, national or local no. This can be exchange or modified in the exchange system.An elegant solution seems to be in the use of DIMF signaling once the connection is established to PABX system.A data invoice answer(DIVA) feature is required in the PABX system for this purpose. Here as soon as PABX is dialed the PABX enables a digit receiver and recorded voice using dialer to dial additional extension digits. These digits are analyzed by the PABX digits receiver and an appropriate connection is established.

Another facility demanded by the customer is likely to have a major impact on numbering scheme.

20.1 Charging Plan

- Providing a telecommunication service calls for investment in capital items as well as meeting operational expenses. The capital cost includes that of line plant, switching systems, buildings and land. Operating costs include staff salaries, maintenance cost, water and electricity charges and miscellaneous expenses.

- A telecommunication system administration receives its income from its subscribers. A charging plan provides for recovering both the capital costs and the operating costs from subscribers.
- The cost of shared resources like the switching equipment is amortized among a large number of subscribers over a period of time.
- The cost of dedicated resources like telephone instrument and the subscriber line must be recovered from individual customers. The operating cost must be worked out depending on the quantum of resources used I providing a service and the duration for which these resources are used.
- Taking all these factors in account a charging plan for a telecommunication service levies three different charges on subscriber:
  (1) An initial charge for providing a network connection
  (2) A rental or leasing charge
  (3) Charges for individual calls made
- A subscriber's share of the capital costs of the common resources is generally covered in the initial connection charge and the rental component.
- The rental may be levied on monthly, bimonthly, quarterly, half yearly or annual basis. Certain operating costs are incurred even if the network carries no traffic. These are covered by the rental.
- The charges for the individual calls include the operating costs in establishing and maintaining the calls and a component for capital resources used. There are also other factors like market policy and government regulation. For example: often government regulations demand that the revenue from a trunk network be used to subsidies the cost of local networks. So the local networks still continue to be expensive.
- By feeding revenue from one service to another the subscribers are given reasonable tariff structures for both local and long distance services.

Charging for individual calls is accounted for by using either a metering instrument connected to each subscriber line or metering register assigned to each subscriber in the case of electronic exchanges. The term 'meter' is used to denote the instrument or the register. The account in the meter represents the number of charging units. A bill is raised by assigning a rate to the charging unit. The count is incremented by sending a pulse to the meter. Charging methods for individual calls falls under two categories:

(1) Duration independent charging
(2) Duration dependent charging

Local calls within a numbering area are usually charged on a duration independent basis .the charging meter is incremented once for every successful call, i.e. whenever the called party answers.

In the past, some systems made a distinction between the calls within an exchange and the calls across the exchanges within a numbering area. Depending upon the number of exchanges involved in setting of a call, more than one pulse for one call is sent to the charging meter. The scheme of sending more than one pulse for a call is known as multimetering. Today it is more usual to apply one unit charge to all the calls within a numbering area irrespective of the number of exchanges involved.

- To avoid the capital cost of providing meters and operating costs of reading them at regular intervals and preparing the bills some administrations have adopted a flat rate tariff system where some fixed charges for an estimated average number of local calls are included in the rental and this scheme is advantageous to subscribers who make a large number of calls but unfair to sparing users.
- To reduce the disparity, business subscribers are charged a higher flat rate compared to domestic subscribers. When flat rate charging is used, the subscribers naturally tend to make more calls. This necessitates local exchanges to be designed for a higher traffic level. Some administrations combine both flat rate and call rate charging.
- The rental covers a certain number of free calls per rental period and only calls above this number are charged for. India uses this scheme .this method is usually adopted from the marketing angle but this scheme doesn't provide any particular advantage in terms of reducing the capital or operating costs.
- With the introduction of STD and ISD automatic charging demands that the subscriber meters be installed. Moreover with the advancement of data transmission, local calls tend to be longer in duration and due to these reasons flat charges are going to be discarded soon.
- In case of duration dependent charging a periodic train of pulses from a common pulse generator operates the calling subscribers meter at appropriate intervals. This method is called periodic pulse metering. In this case the charge for a call is proportion to its duration.

21. SIGNALLING TECHNIQUES

Signalling systems link the variety of switching systems , transmission systems and subscriber equipment's in a telecommunication network to enable the network to function as a whole. Three forms of signaling are involved in telecommunication network:

1. Subscriber loop signaling
2. Intraexchange or register signaling
3. Intraexchange or interregister signaling

Subscriber loop signaling depends upon the type of telephone instrument used. Multifrequency signaling has brought about new services like data in-voice answer, which fall in the class of user to user signaling facilities.

Intraexchange signaling is internal to the switching system and is heavily dependent upon the type and design of a switching system. It varies from one model to another even with the same manufacturer.When inter exchange signaling takes place between exchanges with common control subsystems and is called interregister signaling. In this main purpose is the exchange of address digits which pass from exchange to exchange on a link by link basis. Network wide signaling also involves end to end signaling between originating exchange and terminating exchange and that form is called line signaling.

Signalling techniques falls under two broad classes:

(1) Inchannel signaling
(2) Common channel signaling

21.1 INCHANNEL SIGNALING

The rang of CCITT specified in-channel signaling systems reflects the evolution of international signaling requirements to meet the continually changing condition of the international network. The CCITT in-channel signaling systems and the signaling techniques used in each of them along with the envisaged applications are presented. The early systems, SS1, SS2 and SS3 are of historical interest only. At present, interest in the international inchannel signaling is confined to SS4, SS5 and SS5 bits and in the regional systems to R1 and R2. The other signaling system of the interest is the PCM signaling. The international signaling systems SS4, SS5 and SS5 bits adopt in-band signaling using a combination of two voice band frequencies or a single voice frequency. In additional system SS5 and SS5 bits use multi frequency (MF) signaling for interregister signaling. In SS4 there is no separate interregister signaling.

In SS5 the line signalling comprises either a compound of the two VF frequency or a continuous single frequency. Interregister signalling used 2 out of 6 MF code. Initially the system was jointly developed by UK office and the Bell laboratories for dialing over time assigned speech international (TASI) equipped transatlantic cables. This was the first application of intercontinental dialing and of TASI equipment. The system was subsequently specified by the CCITT as a standard in 1964 and has since found increasing application on other parts of the world.

In TASI each channel is equipped with a speech detector which on detecting speech arranges for a circuit to be assigned to that channel. Since this process of speech detection and establishment of trunk channel association takes definite time the speech burst is clipped for that duration. Typical clip duration is about 15ms when a channel is available. It increases under busy traffic conditions when a free channel may not be available immediately. In order to reduce the extent of interpolation a circuit is not disassociated from the channel for short gapes of speech. For this purpose the speech detector are provided with a short hangover time and a circuit is disconnected only when the speech gap is longer than the detector hangover time. The digital counterpart of TASI is known as digital speech interpolation (DSI)

As with speech burst in-channel signalling information also experiences clipping in a TASI environment. This calls for special consideration in designed signalling system for TASI environment. Unless signals are of sufficient duration to permit trunk channel association and reliable recognition at the receiving end there is the like hood of the signal being lost partially or fully. With pulse signalling it has been determined that a 500ms duration is required to account for the extreme trunk channel association condition. Allowing for reliable recognition a pulse of 850± 200ms duration is considered suitable. But pulse signal of such length would slow down the signalling process considerably. The pulse gapes would result in the channel being disassociation thus leading to unnecessary TASI activity. Moreover fixed length pulses cannot take advantage of lightly loaded condition when the channel assignment time is slow

Interregister signalling carrying address information would be far too slow if continuous compelled signalling is adopted. Pulse signalling is preferred here and two different techniques are used to maintain trunk channel association during the signalling period:

1. The address information transmitted en bloc after gathering all the address digits and the gaps between the pulses are ensured to be less than the speech detector hangover time.
2. Address digits are transmitted as and when they arrive and a lock tone is transmitted during the gaps.

The SS5 adopts en bloc transmission scheme whereas SS5 bits uses the lock tone method. The en bloc method facilitates checking of the validity of the address by digital count and avoids expensive intercontinental circuits being ineffectively taken during incomplete dialing. However this method increases the post dialing delay because the digital are accumulated before the signalling begins. The lock tone method permits overlapped operation of digits received and transmitted. It reduces the post dialing delay. The trunk however is not used as efficiently as in en bloc transmission.

The CCITT R2 signalling system combines an out-band line signalling system and an interregister MF signalling system. Conceptually out-band signalling includes D.C low frequency A.C in slot PCM and signalling above the effective speech band. But in practice the term out-band is applied to system based on A.C signalling using frequency above the upper frequency limit of the effective speech band i.e. above 3400 Hz and within the 4 KHz speech channel spacing. Out-band signalling is generally performed only in the FDM transmission environment.

In FDM system incorporating out-band signalling the channel bandwidth is divided into a speech sub-channel and a signalling sub-channel using suitable filters. Usually a single frequency signalling is used. A 4 wire circuit is used for forward and backward signalling paths and for full duplex operation. The frequency is chosen to lie approximately midway between two adjacent channels and the CCITT recommends 3825Hz.

As the signalling frequency must not be extended to the switching equipment out-band signalling is done on link by link basis and end to end signalling is precluded. Since the signalling is independent of speech there is virtually complete freedom in the choice of signalling mode. Signalling may be two states (on/off) continuous tone or pulse signalling. The former is simple to implement but its potential to support a large signal repertoire is limited/ moreover the signal level must be relatively low to avoid overloading of transmission system. A higher level is permissible with pulse signalling and a large signal repertoire can be supported.

21.2 Common Channel Signalling:

In CCS, signalling is completely separate from switching and speech and in this case signalling is done over a channel that is different from the one which carries the voice or data. Here separate analog voice channel is used for signaling.

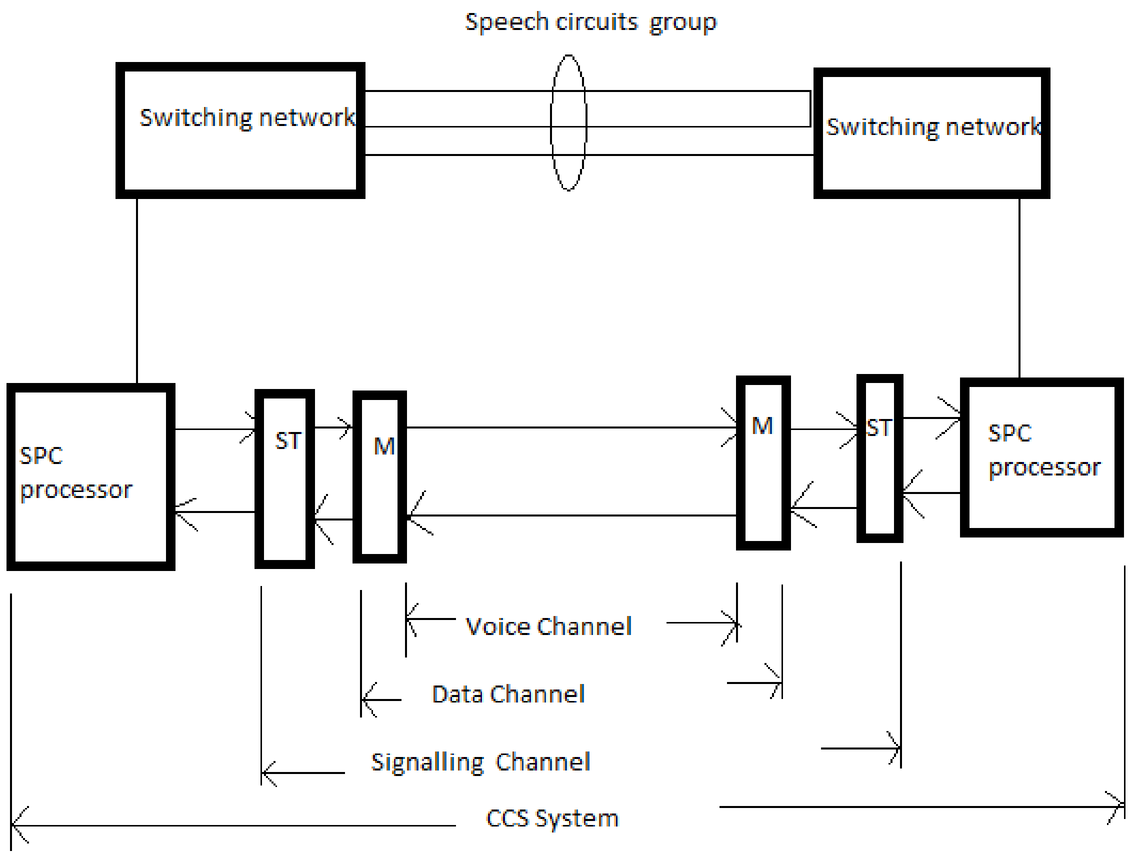


Fig. 36: Basic scheme of CCS [1]

In the above fig.32, two signalling channels, one for each direction, are used in a dedicated manner to carry signalling information. Since the channels are dedicated for signalling they are capable of carrying signalling information for a group of circuits. A phase-equalized voice channel is capable of supporting a bit rate of 2.4 k bps with acceptable error rates for signalling. At this bit rate, one CCS link can carry signals for 1500-2000 speech circuits.

The CCS network is basically a **store and forward (S&F) network** where signalling information travels on a link-by-link basic along the route. When the signalling information is received at a node, it is stored, processed and forwarded to the next node in the route.

In CCS, signalling information is transferred as message of varying length usually defined as one or more fixed length is called single units(SUs).A message of one signal unit length is called **single unit message(SUM)** and one with multiple signal units as **multiunit message(MUM)**.

Header	Signallling information	Circuit label	Error check

(a) Signal Unit Message

Header	Signallling information	Circuit label	Error check
Sub Header	Length	Other signalling information	Error check
Sub Header	Length	Address digits	Error check

(b) Three unit message

Fig.37: Typical CCS signalling message formats [1]

There are three types of signalling units (SUs) defined in SS7:

1. Message signal unit(MSU)
2. Link status signal unit(LSSU)
3. Fill-in signal unit(FISU)

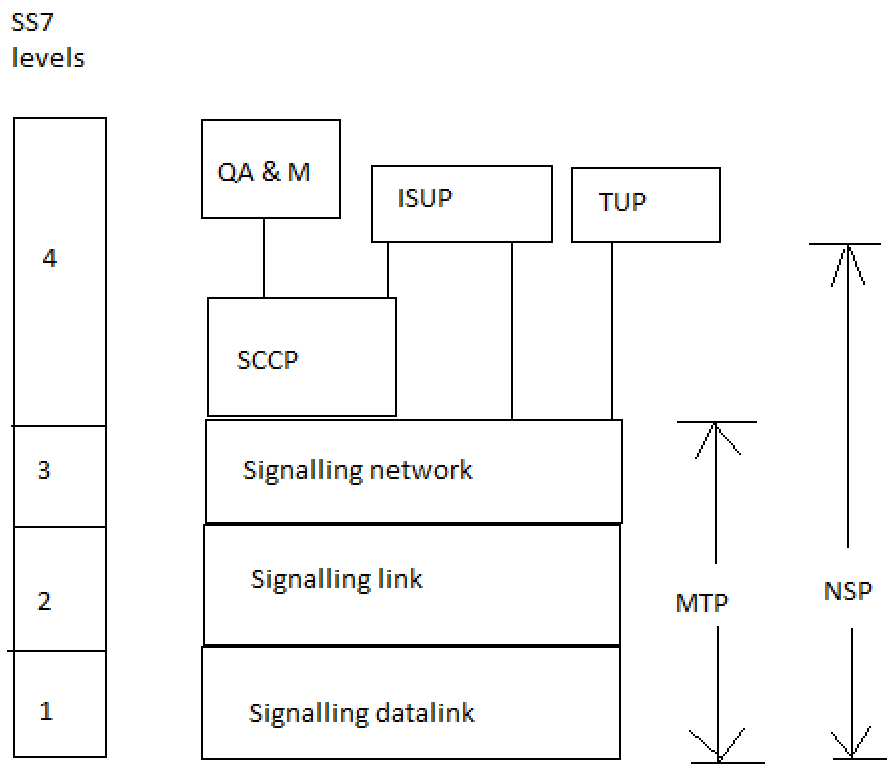
All the SUs begin and end with flag field which has the unique bit pattern 01111110. The flags act as delimiters for the SUs. A common flag may be used as the closing flag for one SU and the opening flag for the next if the SUs are transmitted in continuum.

A bit stuffing technique is used inside SU to avoid destroying synchronization problem. In this technique, the transmitter inserts an extra 0 whenever it comes across five consecutive 1's in the data part of SU. The receiver on detecting five 1's deletes the zero following it. Thus only the flag pattern can contain six 1's.

All the SUs in SS7 contain a 16bit error checking field. Cyclic redundancy code (CRC) is used for error checking.

The control field consists of five subfields (i) the backward sequence number(BSN) and the backward indicator (BI) bit together permit piggybacked acknowledgment of the SUs received from the other side. The negative acknowledgment is indicated by inverting the BI bit which remains unchanged for all subsequent positive acknowledgments. The forward sequence number (FSN) identifies the SU uniquely using modulo 128 counts. A retransmission is indicated by inverting the forward indicator (FI) bit which remains unchanged until another retransmission occurs. A value of 0 indicates a FISU, a value of 1 implies a LSSU and a value of 3 to 63 specifies a MSU.

The level 3 signalling network functions relate to message handling and network management. Message handling involves **discrimination, routing and distribution of messages**. The discrimination function analyses the destination code in the address label to decide whether a message is to be routed to another node or distributed to one of the user parts in the local destination node is decided based on an analysis of the type of message information in the SER field. The discrimination function is needed only in STPs.



MTP = message transfer part   NSP = network service part

TUP = telephone user part   ISUP = ISDN user part

SCCP = signalling connection control part

QA & M = operation, administration and maintanence

Fig.38: Architecture of SS7 [1]

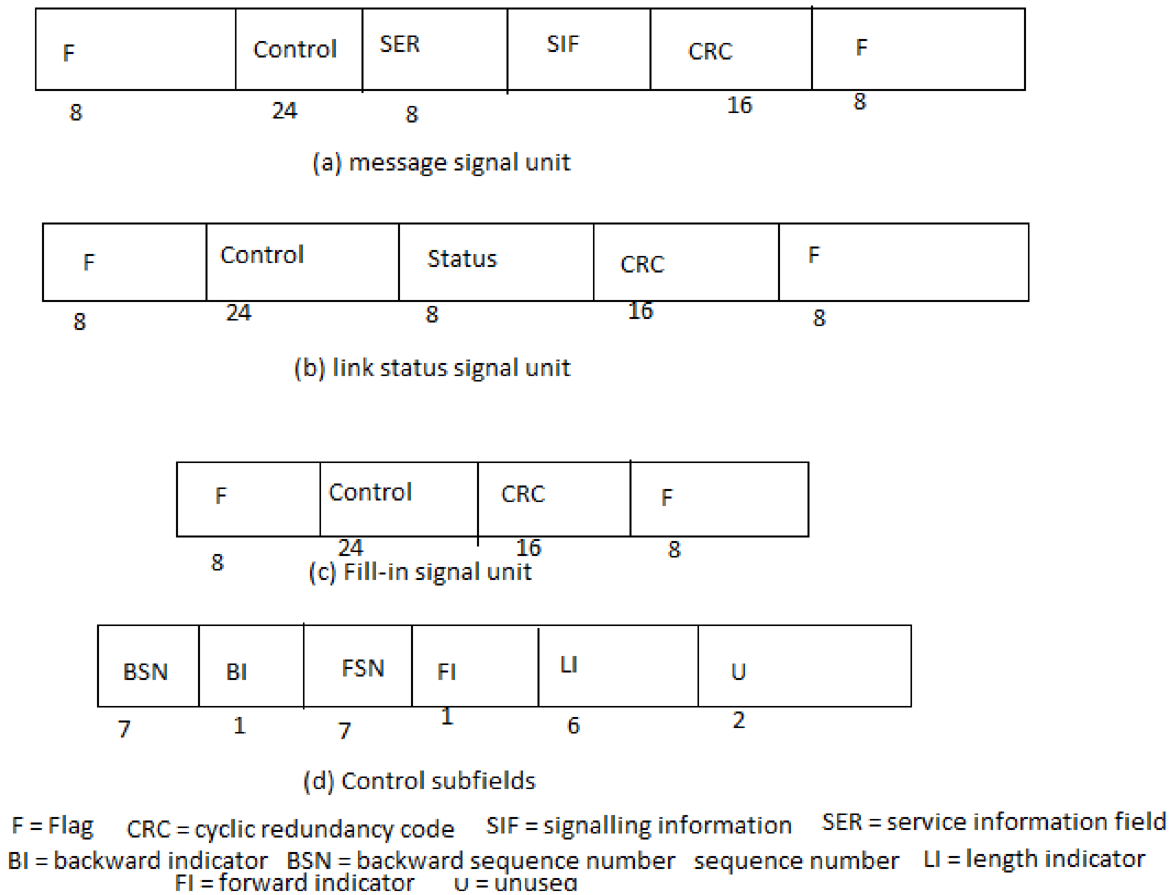


Fig.39: Formats of signalling units [1]

Objective of the Module- III:

- **Transmission of different types of data through public switched telephone network.**
- **Architecture of data communication.**
- **Data networks through satellite.**
- **Overview of standards that use data network.**
- **Overview of standards that use integrated service digital network (ISDN).**
- **Concepts of broadband ISDN.**
- **Concepts of voice and data integration.**
- **Quality of service requirements in ISDN network.**
- **Transmission channels used in ISDN**
- **Different types of signalling that are used in ISDN network.**
- **Different types of service characteristics that are used in ISDN network.**

MODULE-III

22. DATA NETWORKS:

When a network is to transfer a stream of data from a source to destination, it must assign to the data stream a route, that is, a sequence of links or channels connecting the source to the destination. The network should also allocate the data stream a portion of the capacity or BW in each channel along the route to be used. These decisions are performed by switches (or sometimes routers) in telephone exchanges. The process is called switching.

Till 1980's, OSI model was widespread and dominated the entire networking, commercially as well as by architecture. In 1990's TCP/IP has become firmly established as the dominant commercial architecture. Now the TCP/IP is the protocol of choice in many LAN-to- WAN environments.

22.1 DATA TRANSMISSION IN PSTN

The transmission medium is the physical foundation for all the data communications. The amount of data carried across the networks crossed the voice traffic level. The data is growing at a rate of 30 percent per year.With public switched telephone network, there is a possibility of carrying signals at higher speeds. Public switched telephone networks and electronic PABX's are designed to carry analog voice signals. They can be used for data transmission by employing suitable interfaces.

22.2 Data Rates in PSTN

Baud rate: The maximum rate of signal transitions that can be supported by a channel is known as baud rate. Baud rate is a close measure of information throughput, or the effective information data transfer rate from sender to receiver. Thus, baud rate is one that can be supported in a noiseless channel.

We know, a voice channel in a PSTN is band limited with a nominal bandwidth of 3.1 kHz. A maximum data rate that a noiseless or ideal voice channel can support can be obtained from the Nyquist theorem.

$$D = 2\,B \log_2 L \text{ bps}$$

where D = Maximum data rate (in Baud or bps)

B = Bandwidth of the channel

L = Number of discrete levels in the signals.

For a 3 kHz channel, and a binary signal, the maximum data rate is 6000 bps, if the signal level is two.

For higher data rates, we translate information rate into symbols per second. A symbol is any element of an electrical signal that can be used to represent one or more binary data bits. The rate at which symbols are transmitted is the symbol rate. This rate may be represented as a systems baud rate.

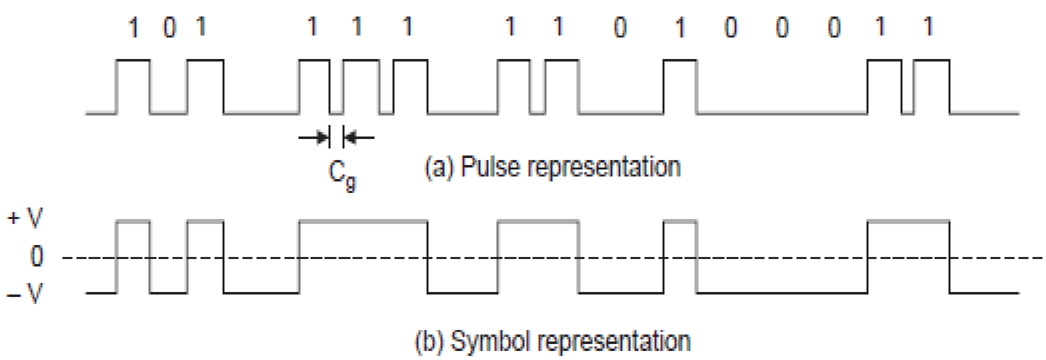


Fig.40: A symbol representation [1]

Bit rate: In the noisy channel, there is an absolute maximum limit for the bit rate. This limit arises because the difference between two adjacent signal levels becomes comparable to the noise level when the number of signal level is increased. For noisy channel, data rate is calculated by:

$$D_b = B \log_2 (1 + S/N)$$

Where D_b = Data rate in noisy channel (in bps)

B = Bandwidth of the channel

S/N = Signal to noise ratio.

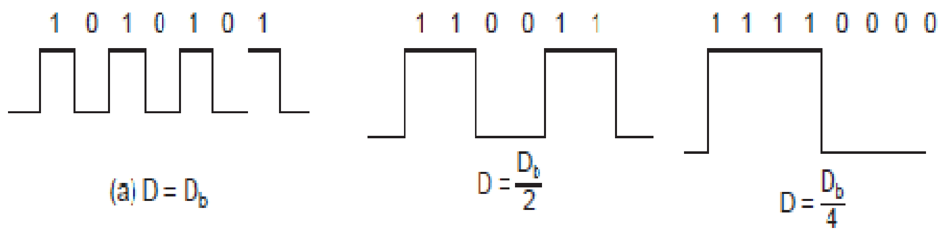
For S/N of 30 dB and 3 kHz Bandwidth noisy channel, D_b is 30000 bps

Relation between baud rate (or symbol rate) and bps : The baud rate and bit rate are related as

$$D_b = D \times n$$

where n = number of bits required to represent signal levels.

In the example considered for baud rate explanation n is assumed as one. Hence baud rate is equal to bps. Fig. 11.2 illustrates the relation between baud and bit rates.



For low-speed applications, the difference between baud and bit rate are insignificant. Thus 300 and 1200 bps modems originally used with personal computers were frequently referred to as 300 or 1200 baud modems.

22.3 Data Communications Link

In order to communicate from a terminal, computer or any equipment, the following six parts have to be put together in proper order.

1. The transmission medium that carries the traffic between source and destination.

2. Data communication equipment or data circuit terminating equipment (DCE).

3. Data terminal equipment (DTE).

4. Communication protocols and software.

5. Terminal devices.

6. Interface.

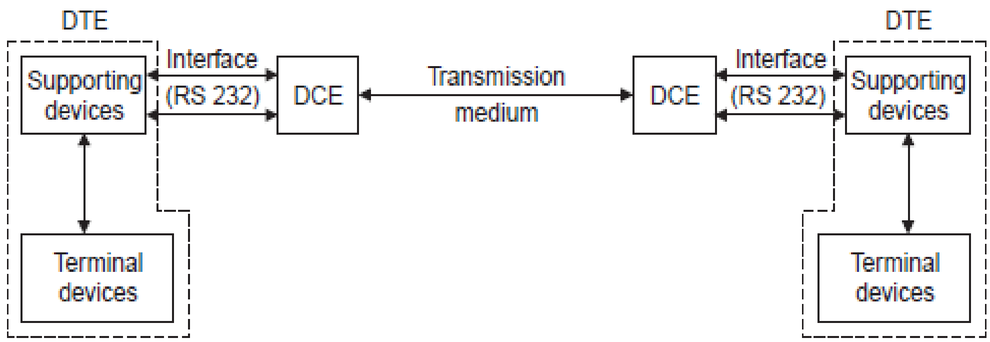


Fig. 41: Data communication link. [1]

Transmission medium. The transmission medium includes communication channels, path, links, trunks and circuits. The transmission medium may be a telephone lines, coaxial cable, twisted pair, Fiber cable, radio waves (free space), microwave link or satellite link.

Terminal devices. These are the end points in a communication link. Terminal devices are also called as nodes. For the two point network, the node points are the primary station and the remote or secondary station. A primary station is responsible for establishing and maintaining the data link between it and a secondary station. The terminal devices includes main frame computer, personal computer, peripherals such as printers, keyboards, FAX machines and data display terminals.

22.4 Data terminal equipment (DTF): The terminal devices, communication station, UART, and line control unit (LCU) grouped together and named as DTE.

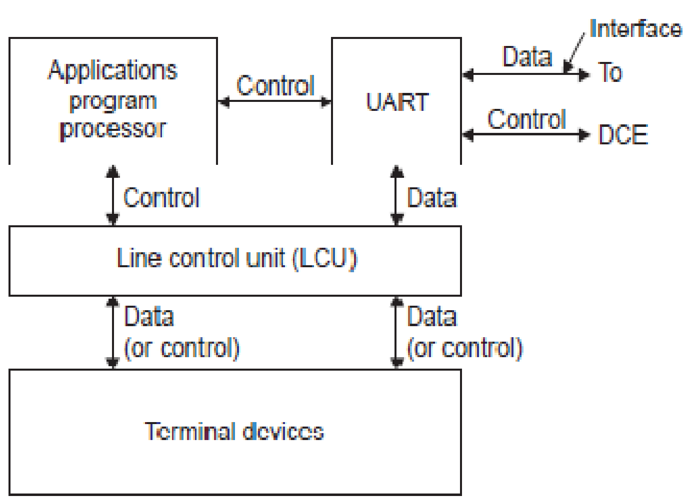


Fig.42: Data terminal equipment [1]

UART. The universal asynchronous receiver transmitter (UART) and the universal synchronous/asynchronous receiver transmitter (USART) perform the parallel to serial conversion (and vice versa at the receiving station).

Application program processor. An application program used by the DTE, called a protocol, defines a set of rules that determine requirements for the successful establishment of a data link and the transfer of actual information between stations. Protocols are key components of communication architectures. Protocols provide the rules for communication between counterpart components on different devices. The application programs also direct control information to the line control unit and UART to allow data flow from the peripheral currently serviced by the LCU to the UART and out to the DCE.

Line control unit (LCU). Data sent from one station to another usually originates in parallel binary form from one or more peripheral devices connected to that station through a LCU. The unit acts as an interface between terminal devices and UART and the application programme processor.

Interface.RS 232 interface is used to connect UART and the DCE. The RS-232 interface defines the electrical function of the pins and the mechanical function of the connector. The Electrical Industry Association (EIA) revised RS-232 C in 1989 and called the revision RS-232 D (connector with 25 pins). RS-232 is a standard connection for serial communication. All modems use RS 232 connections and all PCS have a RS 232 port.

Data communication equipment (DCE). The DCE is a modem. This device is used to convert the serial data stream into a form which is suitable for transmission. This serial data stream transferred through a transmission medium. At the receiver side, the serial data stream are converted back to digital and sent to DTE. DCE may be a modem or a computer based node in a data network.

23. SWITCHING TECHNIQUES:

This section describes various techniques used to establish connections between user exchanges.

Switches are hardware and/or software devices used to connect two or more users temporarily. Message switching, circuit switching and packet switching are the most important switching methods. The terminals of the message switching systems are usually teleprinters. In this switching, delays are incurred but no calls are lost as each messageis queued for each link.

Thus much higher link utilization is achieved. The reason for the delay is that the system is designed to maximize the utilization of transmission links by queuing message awaiting the use of a line. This switching is also called store and forward switching. The circuit switching sets up connection between the telephones, telex networks etc. which interchange information directly. If a subscriber or system to which connection to be made as engaged with other connection, path setup cannot be made. Thus circuit switching is also referred as lost call system. The modified form of message switching is called packet switching. Packet switching system carries data from a terminal or computer as short packets of information to the required destination.

23.1 Message Switching

In message switching, the messages are stored and relayed from secondary storage. So, message switching is best known by the term store and forward. In message switching, there is no direct link between the sender and the receiver. A message delivered to the destination is rerouted along any path before it reaches the destination. It was common in 1960's and 1970's.

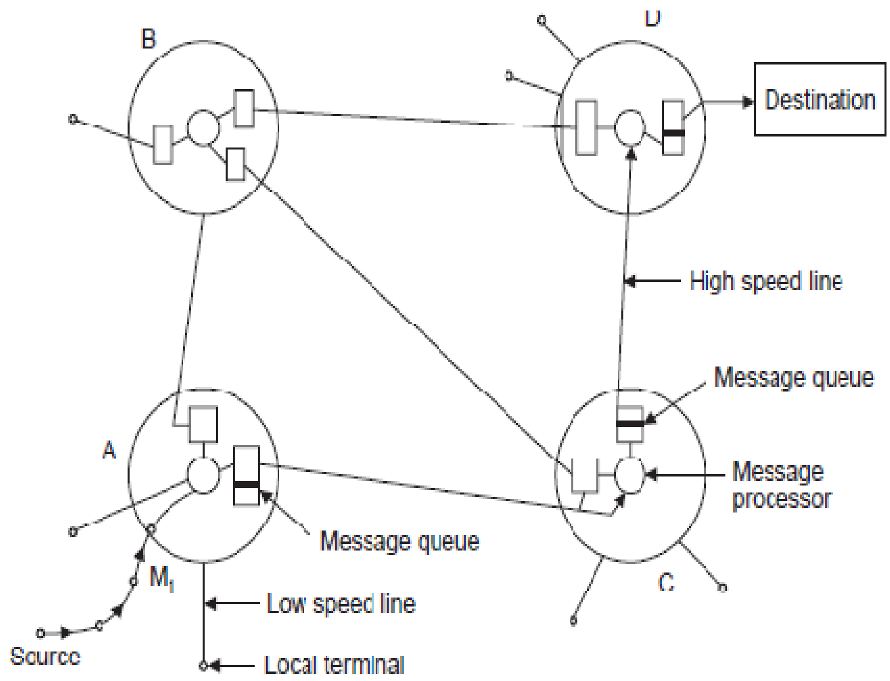
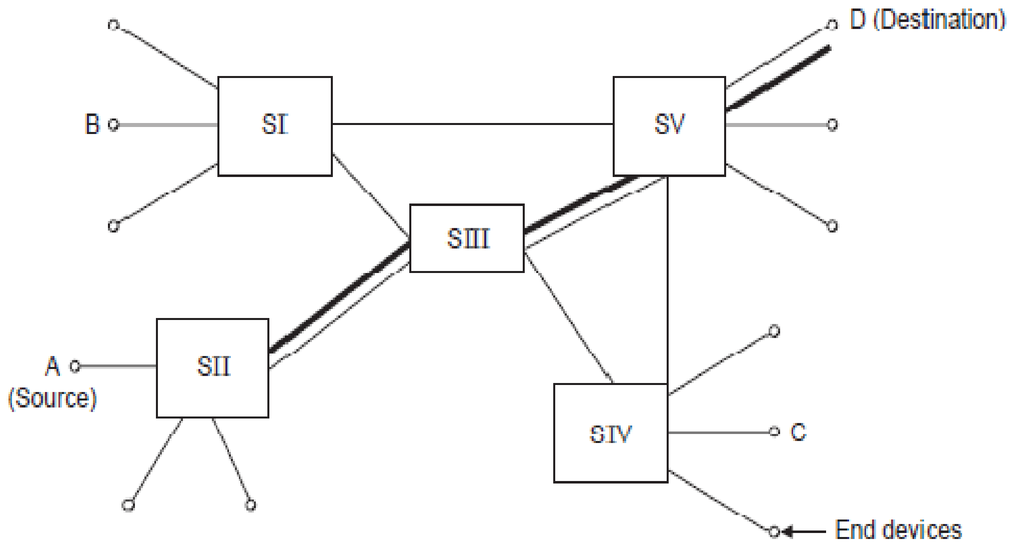


Fig.43: Message switching network.[1]

Message switching offers the possibility of greatly improved economy. The working of message switching is as follows. Source sends message M1 to the destination. Suppose that the transmission path selected is A—C—D. In message switching no complete connection is required. Thus the each message includes a header contains the destination address, routing information and priority information (for special cases).

23.2 Circuit Switching:

Circuit switching creates a direct physical connection between two devices such as phones or computers. In order to setup a direct connection over many links it is necessary that each link to be simultaneously free. This implies that the average utilization of the links must be low if the probability of demand for connection is more. It is therefore used in voice networks mainly and not in networks designed for data transfer. A circuit switch is a device with *n* inputs and *m* outputs that creates a temporary connection between source and destination. The inputs *n* and outputs *m* need not be equal. In order to transmit information, a circuit switched network finds a route along which it has free circuits. The network connects the circuits together and reserves them for the transmission.



S1-SN → Switches, —Path between switches, ▬Path setup between end device A&D,

Fig.44: Circuit switching network.[1]

As the data transfer takes place in three phases, the time taken for the data transfer (T) is expressed as

$$T = T_p + T_d + T_r$$

where T_p = Path setup time $(N-1)\,T_{rs}$

T_d = data transfer time = M/R

T_{rs} = average route selection time

T_r = data release time = NT_n

N = Number of switches in the path

M – Message length in bits

R = data rate in bits per sec

T_h = house keeping entries time.

Thus $T = (N-1)\,T_{rs} + M/R = NT_h$

The propagation time is not considered as it is comparatively very small. In our case N = 3, If T_{rs} = 2 sec, T_n = 2 sec, R = 2400 bps and the message is 300 bytes long, the time for the data transfer is T = (3 – 1) × 2 + 300 × 8/2400 + 3 × 0.2 = 4 + 1 + 0.6 = 5.6 sec.

Comparison of message and circuit switching :

Message switching	Cirtuit switching
The source and destination do not interact in real time	The source and destination are connected temporarily during data transfer.
Message delivery is on delayed basis if destination node is busy or otherwise unable to accept traffic.	Before path setup delay, may be there due to busy destination node. Once the connection is made, the data transfer takes place with negligible propagation time.
Destination node status is not required before sending message.	Destination node status is necessary before setting up a path for data transfer.
Message switching network normally accepts all traffic but provides longer delivery time because of increased queue length.	A circuit switching network rejects excess traffic, if all the lines are busy.
In message switching network, the transmission links are never idle.	In circuit switching, after path setup, if the users denied service, the line will be idle. Thus, the transmission capacity will be less, if the lines are idle.

23.3. Packet Switching:

There are three types of switching used in PSTN network.Circuit switching was designed for voice communication. Circuit switching creates temporary (dialed) or permanent (leased) dedicated links that are well suited to this type of connection. The circuit switching also limits the flexibility and not suitable for connecting variety of digital devices. More efficient utilization of the network requires greater control channel bandwidth and increased call processing capacities in the switches. But the circuit switching not providing these capabilities. Message switching overcomes the limitations of circuits switching. This switching stores the incoming messages into a computer memory and forwards it to the destination when available. This causes delay in switching. The packet switching overcomes all the limitations of message and circuit switching. Thus it is highly suitable for the data communication.

Packet Switching Principles

The data stream originating at the source is divided into packets of fixed or variable size. The data communication system typically hasbursty traffic. Thus, the time interval between consecutive packets may vary, depending on the burstiness of the data stream.

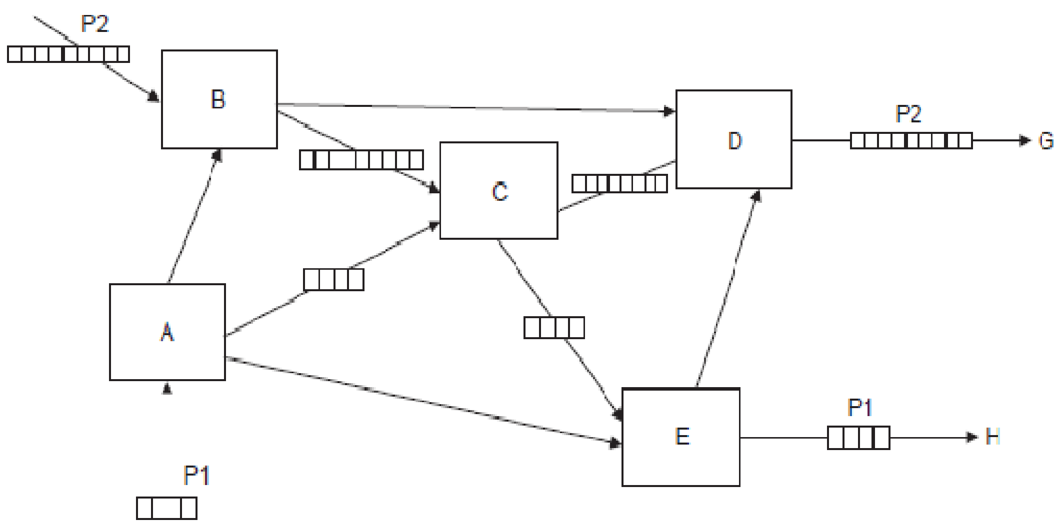


Fig.45: packet switching. [1]

24. Routing Control

From the previous section, it is clear that in packet switching, messages are broken into packets and sends one at a time to the network. Routing control decides how the network will handle the stream of packets as it attempts to route them through the network and deliver them to the intended destination. The routing decision is determined in one of two ways. They are

1. Datagram and

2. Virtual circuit.

Datagram. In datagram, each packet within a stream is independently routed. A routing table stored in the router (switch) specifies the outgoing link for each destination. The table may be static or it may be periodically updated. In the second case, the routing depends on the router's estimate of the shortest path to the destination. Since the estimate may change with time, consecutive packets may be routed over different links. Therefore each packet must contain bits denoting the source and destination. Thus may be a significant overhead

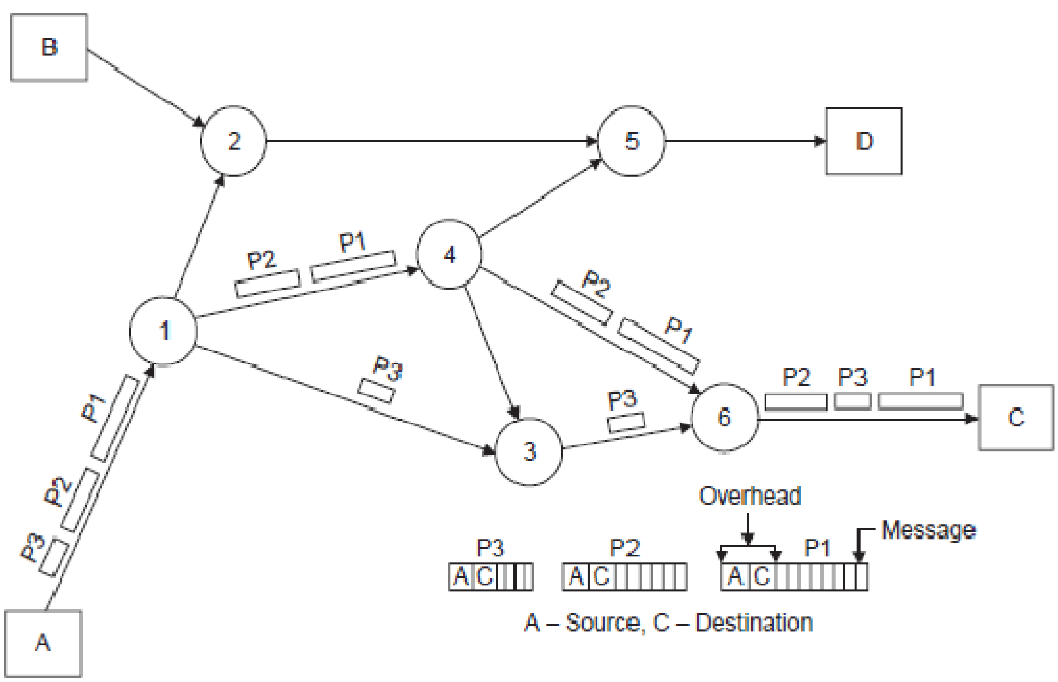


Fig.46: Concept of Data gram. [1]

The station A is assumed to send three packets of message namely P1, P2 and P3 (for explanation purpose named so). At first, A transmits these packets to node 1. Node 1 makes decision on routing of these packets. Node 1 finds node 4 as shortest compared to node 3. Thus it passes P1 and P2 to node 4. Accidently, if node 4 is not accessible, node 1 finds node 3 as shortest and sends packet P3 to node 3.

Node 3 and 4 sends its received messages to the destination C through node 6. It is shown that the order of the packet is changed due to the different routing of the packets. Thus in datagram, it is the responsibility of destination station to reorder the packets in proper sequence. Also if a packet crashes in a switching node, the destination C may not receive, all packets. In such a case also, it is the responsibility of station C to recover the lost packet.

Virtual circuit. In virtual circuit, a fixed route is selected before any data is transmitted in a call setup phase similar to circuit switched network. All packets belonging to the same data stream follow this fixed route called a virtual circuit. Packet must now contain a virtual circuit identifier. This bit string is usually shorter than the source and destination address identifiers needed for datagram. Once the virtual circuit is established, the message is transmitted in packets.

Suppose that end station B has two messages to send to the destination D. First B sends a control packet referred as call-request packet to node 2, requesting logical connection to D. Node 2 decides to route the request and the subsequent message packets through node 3 and 4 to destination D. If D prepared to accept the connection, it sends a call-accept packet to node 4. Node 4 sends the call-accept packet to B through node 3 and 2. Because the route is fixed for the duration of the logical connection, it is somewhat similar to a circuit switching network and is referred to as a virtual circuit.

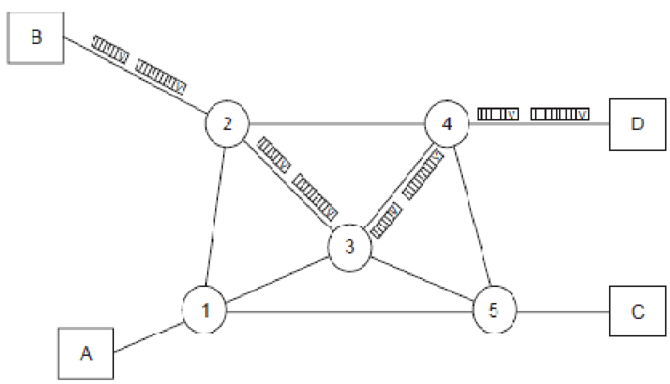


Fig.47: concept of virtual circuit. [1]

Packet size. If an organization has large amounts of data to send, then the data can be delivered to a packet assembler/disassembler (PAD). The PAD (software package) receives the data and breaks it down into manageable packets. In the data communication, a packet can be a variable length. Usually up to 128 bytes of data is in one packet. X-25 services have created packets upto 512 bytes, but the average is 128. The 128 byte capability is also referred to as fast select. There is a significant relationship between packet size and transmission time. The process of using more smaller packets (for example 30 byte information may be sent as a single packet with header of 3 byte or two packets with 15 byte each plus the header in each packet or 5 packets with 6 bytes plus header) increases the speed of transmission.

24.1 Comparison of Circuit Switching and Packet Switching:

There are two types of approaches in packet switching. Datagram and virtual circuit. The circuit switching is compared with these two approaches.

Datagram switching achieves higher link utilization than circuit switching especially when the traffic is busty. No dedicated path is required as circuit switching. But the datagram have the disadvantage over virtual circuit wire.

1. End to end delay may be so large or so random as to preclude applications that demant guaranteed delay.

2. The overhead due to source and destination identifiers and bits needed to delimit packets may waste a significant fraction of the transmission capacity if the packet are very short.

3. A datagram switch does not have the state information to recognize if a packet belongs to a particular application. Hence the switch cannot allocate resources (bandwidth and buffers) that the application may require.

Virtual circuits are more advantages and currently the packet switching network uses the virtual circuit approach. The overhead is comparable to circuit switching. As the packets arrives in sequence, no re-sequencing is needed. Statistical multiplexing of packets at the router or switch can achieve better utilization than in circuit switching. Since packets contains their virtual circuit identifiers (VCI), the switch can allocate resources depending on the VCI. During the connection setup phase, the switches may be notified that a particular VCI should be given extra resources.

24.2 Packet Formats

The format of a packet in packet switching network can vary significantly from one network to another. Generally header includes all related control information. In some cases, control information is communicated through special control packets.

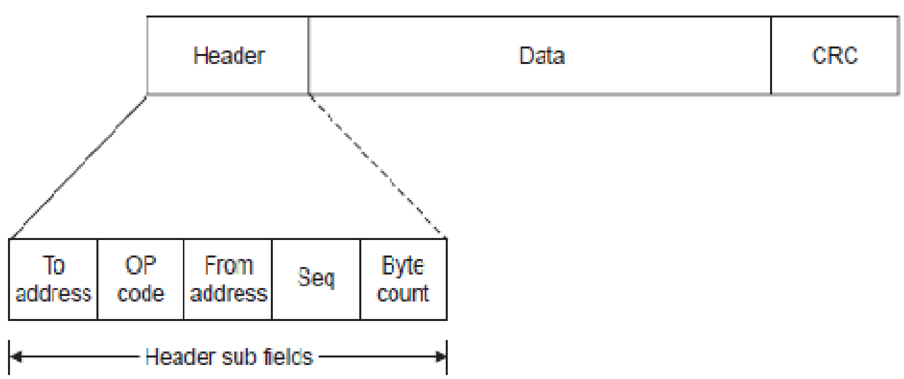


Fig.48: typical packet format [1]

A packet contains 3 major fields.

1.**Header.**It contains sub fields in addition to the necessary address fields. Other than the to and from address field, the following are the useful control information.

(*a*) **Op code.**It designates whether the packet is a message (text) packet or control packet.

(*b*) A **sequence number (Seq)**to reassemble messages at the destination node, detect faults and facilitates recovery procedures.

(*c*) **Byte count.**Used to indicate the length of a packet.

2. **Data.**A portion of a data stream to be transferred in the data field. Some packets may not contain a message field if they are being used strictly for control purposes.

3. **CRC.**The cyclic redundancy checks (CRC) field contains a set of parity bits that cover overlapping fields of message bits. The fields overlap in such a way that small numbers of errors are always detected. The probability of not detecting the occurrence of 2 large numbers of errors is 1 in 2M, where M is the number of bits in the cheek code.

25. OSI MODEL

The Open System Inter connection (OSI) model was developed by the International organization for standardization (ISO). The ISO developed OSI for networking. An open system is a set of protocols that allows two computers to communicate with each other regardless of their design, manufacturer or CPU type. Open system architectures are flexible structures set into fixed frame works. The concept of an open system approach to networking allows any device or system operating with any protocol to communicate with another device or system using its own protocol. The OSI model defines seven distinct levels in its communication model. In the following paragraphs all this levels are explained in detail.

25.1 OSI Network Architecture

The OSI model divides network communication into seven layers, with each layer performing specific tasks. Each level has a set of specifications and functions that it performs. Any number of communications protocols can operate within a specified level. Related header, trailer information, error detection capabilities and other overhead type are added to the message. The entire message with its overhead denoted as payload. The pay load is then encapsulated into the data portion of the next layer’s message format and transported using that level’s protocol rules. In the table 3 shown, the specification/functions of the layer are given briefly.

The detailed explanation of each layer is given in the following sections.

LAYER	SPECIFICATIONS
7	**APPLICATION LAYER:** Performs information processing such as file transfer, e-mail and teletext. Detailed and application specific information about data being exchanged.
6	**PRESENTATION LAYER:** Defines the format of data to be sent : ASCII, data encryption, data compression and EBCDIC.
5	**SESSION LAYER:** Management of connections between programs. Sets up a session between two applications by determining the type of communication such as duplex, half duplex, synchronization etc.
4	**Transport layer:** Delivery of sequence of packets. Ensures data gets to destination. Manages error control, flow control and quality of service.
3	**NETWORK LAYER:** Format of individual data packets. Sets up connection, disconnects connection, provides routing and multiplexing.
2	**DATA LINK LAYER:** Manages framing, error detection, and retransmission of message. Access to and control of transmission medium.
1	**PHYSICAL LAYER:** Medium and signal formed of raw bit information. Electrical interface (type of signal), Mechanical interface (type of connector), converts electrical signal to bits, transmits and receives electrical signals.

The seven layers of OSI are grouped into three layers. The layer 1, 2 and 3 are called network support layers. The layers 5, 6 and 7 are called support layers and layer 4 is transport layer.

Let computer A sends a data stream of bits to computer B. Communication must move from higher layer down through the lower layers on computer A. Each layer in sending machine adds its own information to the message. In receiving computer B, communication must move from lower layer up through the higher layers. At the receiving machine, the message is unwrapped layer by layer.

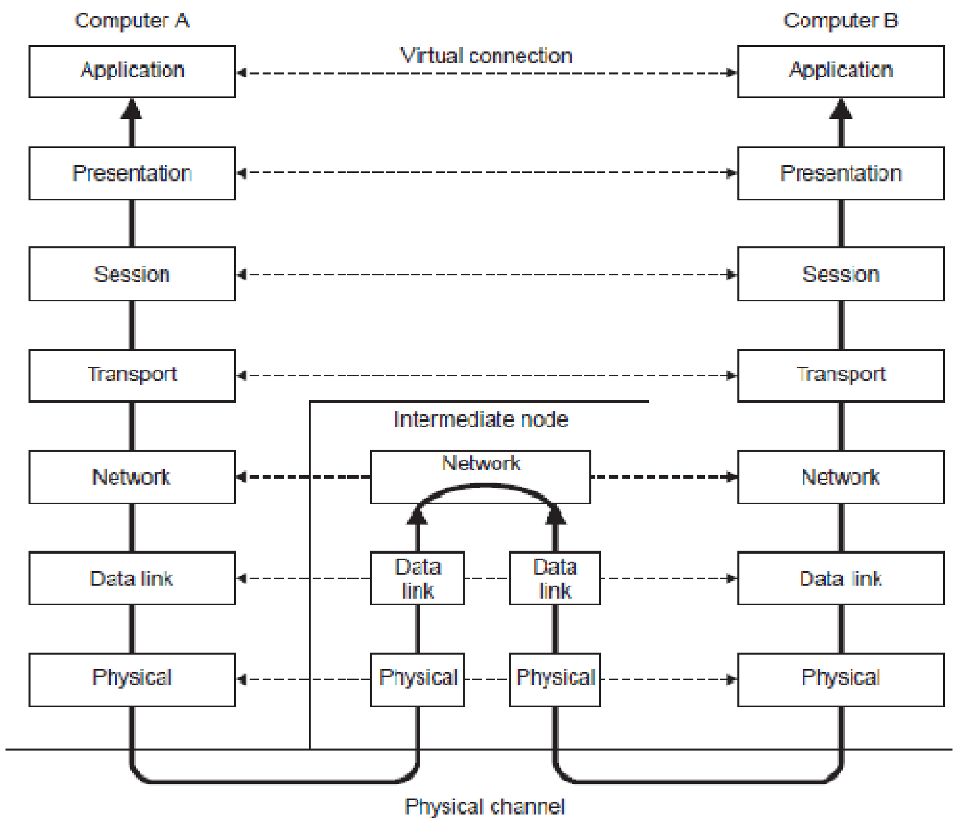


Fig.49: OSI Network architecture [1]

PHYSICAL LAYER. The physical layer is the lowest layer of the OSI model. It defines the mechanical, electrical, functional and procedural aspects of the physical link between networks. Physical layer standards have been widely used for years in point-to-point wide area network applications. (CCITT/ITU has established X-21–X-24 to specify the functions at the physical level of the based circuits. Numerous other standards such as EIA-232 and V-21– V-34 are widely used for various purpose at the physical layers.The physical layer implements an unreliable bit link. A link consists of a transmitter, a receiver and a medium over which signals are propagated. The physical layer data consists of stream bits. The physical layer defines the type of encoding to convert the bit stream into electrical or optical signal to transmit in the medium. At receiver the physical layer converts back into bit stream. The receiver must be in synchronism with transmitter to receive the specific bit pattern. To assist synchronization, the physical layer adds a specific bit pattern called preamble at the beginning of the packet.

DATA LINK LAYER. The data link layer defines the frame format such as start of frame, end of frame, size of frame and type of transmission. The principal service provided by the data link layer to higher layer is that of error detection and control. This layer is the first software protocol layer of the OSI model. It specifies the data format, sequence, and acknowledgement process and error detection methods. The data link layer accepts information from the network layer and breaks the information into frames. It then adds the destination address, source address, frame check sequence (FCS) field and length field to each frame and passes each frame to the physical layer for transmission on the receiving side, the data link layer accepts the bits from the physical layer and from them into frame, performing error detection. If the frame is free of error, the data link layer passes the frame up to the network layer. It performs frame synchronization, that is, it identifies the beginning and end of each frame. Existing protocols for the data link layer are :

1. Synchronous Data Link Control (SDLC). Developed by IBM as link access for System Network Architecture.

2. High Level Data Link Control (HDLC). It is a version of SDLC modified by the ISO for use in the OSI model.

3. Link Access Procedure Balanced (LAPB). The modified version of HDLC is LAPB.

The data link layer also performs the flow control and access control. By flow control, if the rate at which the data are absorbed by the receiver is less than the rate produced in the sender, the data link layer imposes a flow control mechanism to prevent overwhelming the receiver. By access control, when two or more devices are connected to the same link, the protocol of data link layer determines which device has control over the link at any given time. Two sub layers defined in the data link are the media access control (MAC) and the logical link control (LLC) layer. MAC performs address management function. LLC manages flow and error control, automatic requests for retransmission (APQ) and handshake processes

NETWORK LAYER. If two systems are attached to different networks (links) with connecting devices between the networks (links), there is often a need for the network layer to accomplish source to destination delivery. Thus the function of the network layer is to perform routing. The network layer checks the logical address of each frame and forwards the frame to the next router based on a look up table. The network layer is responsible for translating each logic address (name address) to a physical address (MAC address).

TRANSPORT LAYER

The transport layer provides a mechanism for the exchange of data between end systems. It optimizes the use of network services with providing a requested quality of service to session entities. The size and complexity of a transport protocol depend on how reliable or unreliable the underlying network and network layer services are. Essentially, this layer is responsible for the reliable data transfer between two end nodes and is sometimes referred to as host-tohost layer.

SUPPORT LAYERS

The session layer, presentation layer and application layer are considered as support layers. All these layers explained in the following paragraphs.

SESSION LAYER. The session layer establishes a logical connection between the applications of two computers that are communicating with each other. The session layer concerns file management and overall networking functions. Access availability and system time allocations are included in this layer. The session layer can partition a transfer of a large number of messages by inserting synchronous points. Specific responsibilities of the session layers are authentication of user access, fault recovery if a break in service occurs, permitting multiple applications to share a virtual circuit, connection and disconnection of any node from the network.

PRESENTATION LAYER. The presentation layer receives information from theapplication layer and converts into ASCII or Unicode or encrypts or decrypts. This layer isconcerned with the syntax and semantics of the information exchanged between two systems.

The three basic forms of protocols used in the presentation layer are

1. Virtual terminal protocol, which is used to allow different types of terminals to support different applications.

2. Virtual file protocol, which handles code conversions' within files, file communication and file formatting.

3. Job transfer and manipulating protocols which controls the structure of jobs andrecords.

A separate protocol function known as abstract syntax notation (ASN) specifies file datastructure.

APPLICATION LAYER. This layer enables users to access the network with applications such as e-mail, FTP and Telnet. It provides user interfaces and support for various services. Specific services are (*a*) Network virtual terminal (*b*) File transfer access and management (FTAM), (*c*) Mail services (*d*) Directory services (*e*) Specific User Service Element (SUSE) — deals with actual user requirements for access and use of the network (*f*) Common application service element (CASE) — This sets guidelines for the applications required quality of service and (*g*) Specific application service element (SASE) — deals with large amount of data, including database access.

26. INTEGRATED SERVICE DIGITAL NETWORK (ISDN)

26.1 INTRODUCTION

ISDN has been most important development to emerge in the field of computer communications. ISDN is a well-conceived and planned area of development in the field of telecommunication.ISDN –An integrated digital network in which the same digital switches and digital paths are used to establish different services for e.g. telephony, data.

Six conceptual concepts on which ISDN standards laid are:

1. ISDN will evolve from telephony IDN by incorporating additional functions and network feature including other dedicated network to provide for existing and new services.

2. New services should be compatible with 64kbps switched digital connections.

3. The transition from existing network to comprehensive ISDN may require one or two decades.

4. Arrangement must be made for internetworking of services and services on other networks during transition period.

5. ISDN contains the intelligent of providing service features,maintenance and network management functions.

6. Various access arrangements to ISDN require layered functional set of protocols.

So ISDN can be defined as an ISDN is network evolving from telephony IDN that provides end to end digital connectivity to support wide range of services including voice and non-voice services, to which users have access by a limited set of standard multipurpose user network interface.

26.2 NEW SERVICES:

ISDN will support variety of services including the existing voice and data services and host of new services. Short lists of some important new services are:

1. Videotext
2. Electronic Mail
3. Digital Facsimile
4. Teletext
5. Database access
6. Electronic Fund Transfer
7. Image and graphics exchange
8. Document storage and transfer

9. Automatic alarm services, e.g., smoke,fire,police,medical
10. Audio and video conferencing

VIDEOTEX

Videotex is a generic term for systems that provide easy to use, low cost computer based services via communication facilities. Three forms of videotext exist:

1. View data
2. Teletext
3. Open Channel teletext

View data is fully interactive videotex. This means that request for information and service from a user are actually sent to, received by, and acted on by a centralized computer.

Teletex is broadcast or pseudo-interactive videotext service. Teletext users may select the information to be seen, the pace at which the information to be displayed and, often sequence of display. The information cast in the form of frames and set of frames which is called a magazine is recycled continuously. Teletext is a one way communication system and there is no real interaction between the user and the computer.

Open channel teletext is totally non-interactive one way videotext. With this form of videotext, the user receives pre-selected information in predetermined order. There is no interaction either real or apparent. The user has no control over the pace or sequence of display. Open channel text may classify into three categories according to the way of preselected information displayed and the way of display channel is used:

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

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

ELECTRONIC MAIL

Electronic Mail is popularly known as email, may be defined as the communication of textual messages via electronics means. Even the telex communication is electronic in nature but the differences are telex communication is terminal to terminal, electronic mail communication is user-to-user. In telex message destined to no. of users are sent to the same terminal form where it is distributed by an operator or messenger. On the other hand electronic mail delivered to the mail boxes of individuals. Telex works on a circuit switched mode, where electronic mail is store and forward(S&F) services. Electronic mail is computer based message system where telex is generally not.Advantage of Electronic mail is first of all security then its ability to reduce the consumption of paper in the office. Being a computer based messaging system, files are prepared like automation packages like word, spreadsheet etc. easily interchanged as electronic mail. This facility improves efficiency for office work.

Early electronic mail systems were organized around the single time sharing or multiuser computer system,where electronic mail was exchanged among the user of the system.The typical configuration of electronic mail is given below which was established in 1970s.

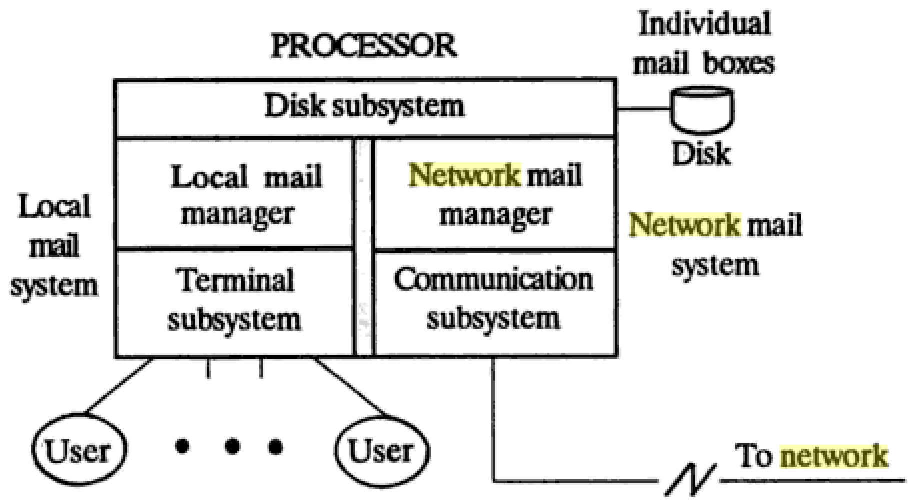


Figure.50: A typical configuration of electronic mail system [1]

Component of the system are: one to handle within the system another to handle mail over the network. Both share common disk storage where mail boxes are maintained. Thus, real time exchanges of messages are possible in an electronic mail system, if the two concerned parties are logged onto the same machine at the same time. Electronic mail being S&F on a network, real time exchange may not possible. Some are the well-knownnetwork who is providing the electronic mail services are UUNET,BITNET,CSNET, and JANET.

In the context of Open System Interconnection (OSI) networks, electronic mail is considered as an application process running on the seventh layer. Standard electronic mail service components have been defined and approved by 1984 CCITT Plenary Assembly. These are known as X.400 family of standards of message handling systems (MHS).List of X.400 family is given below.

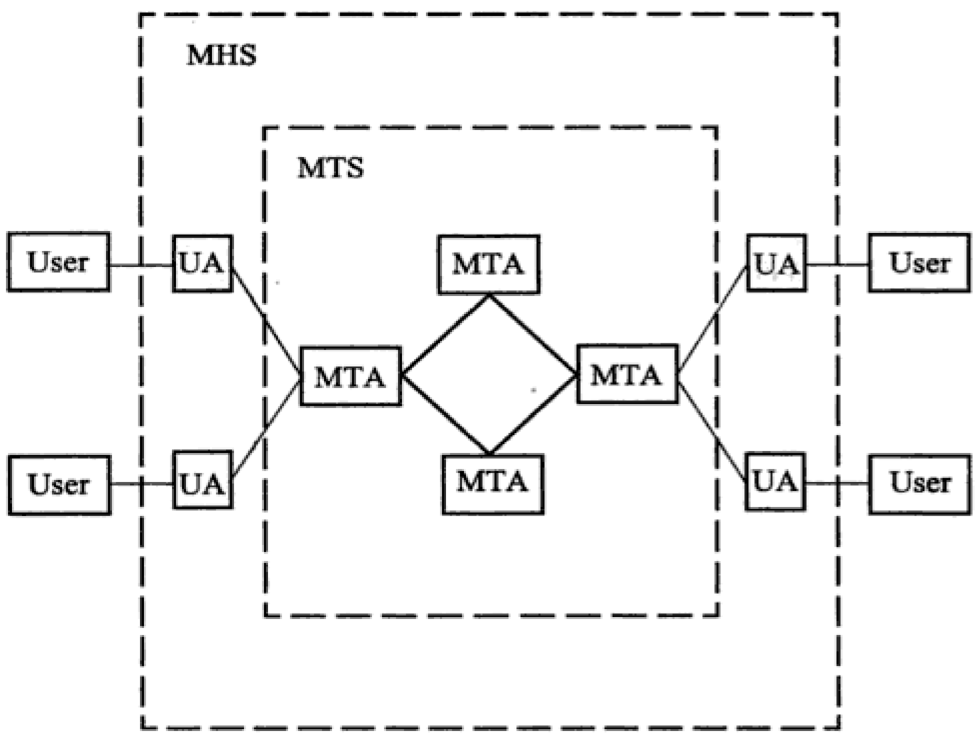
Number	Subject dealt with
X.400	System model — service elements
X.401	Basic service elements and optional user facilities
X.408	Encoded information-type conversion rules
X.409	Presentation transfer syntax and notation
X.410	Remote operations and reliable transfer server
X.411	Message transfer layer
X.420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

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

User agent entity (UAE)

Message transfer agent entity (MATE)

The figure of X.400 model is given below. User agent performs functions relating the preparation, submission and delivery of messages.



MHS-message handling system MTA-message transfer agent

MTS – message transfer system UA-user agent

Fig.51: X.400 message handling system model [1]

It also assists user in other message function such as filing,replying, retrieving and forwarding. Message transfer agent is concerned with transfer of message across of network and functions in an environment designated as message transfer system (MTS). It obtains the message from the source UA and deliveries the same to the destination UA. On accepting message the MTA performs either a delivery function or a routing function. If the destination UA is in the same as the MTA or is attached with MTA directly, then the MTA performs the delivery function, otherwise it performs a routing function.Depending upon the physical location of the MTA and UAs, a no. of different physical realization of emails systems are possible which are listed below.

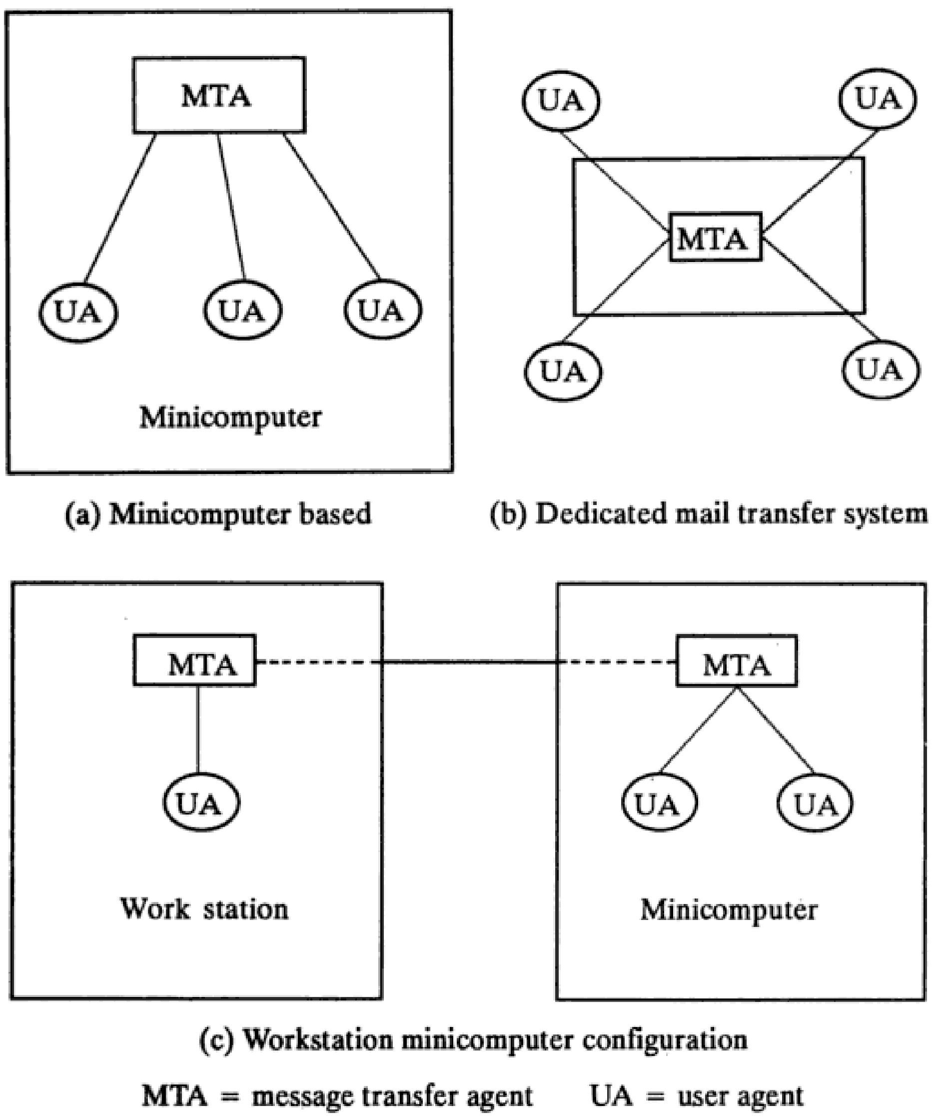


Fig.52: Email system configuration [1]

In the context of OSI reference model, X.400 MHS is part of the application layer. The UA and MTA functions are treated as sub layers as shown in figure below. In systems where only the UA function is implemented, the message transfer sub layer comprises a submission and delivery entity (SDE). The entity acts the interface between a local UA and remote MTA. In the sense, SDE implements a remote procedure call protocol. In the intermediate system, only the message transfer agent entity (MTAE) plays a role acts as a relay. Zero or most intermediate system may be involved in transferring a message from the source MTAE to the destination MTAE. A message may be destined to one or more UAs & its responsibility of message transfer sub layer to deliver the message to all intended recipients.

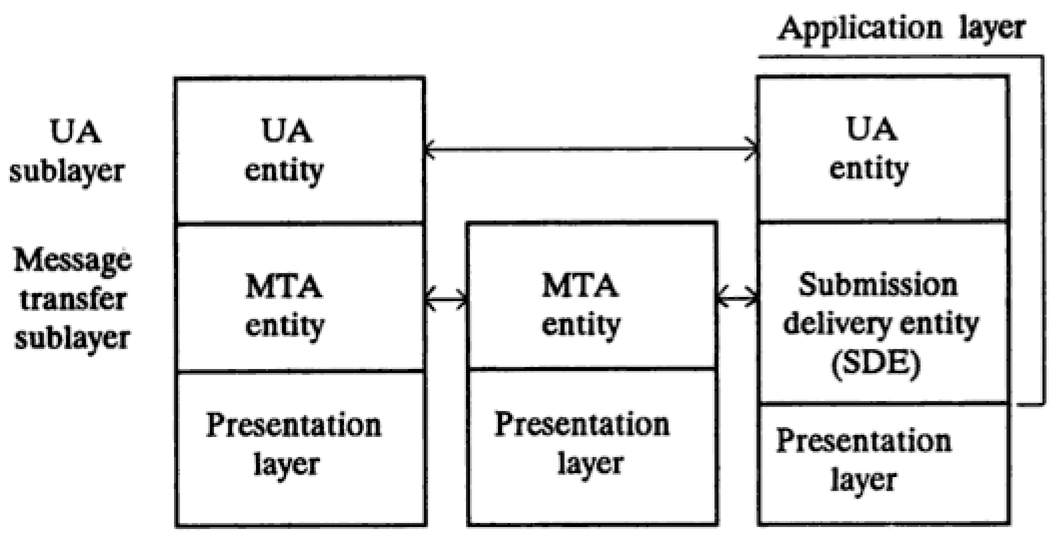


Fig.53: X.400 in the context of OSI model [1]

The services provided by UAEs are known as interpersonal messaging services. Two types of service are supported:

- ➔ Send/receive user message
- ➔ Send / receive status report

Corresponding to two types services two types of protocol data units (PDU) are used. The PDU structure is shown in below.

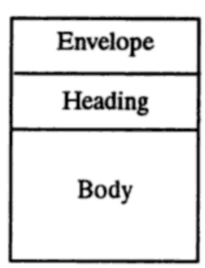


Fig.54: X.400 user message format [1]

The message consisting of three parts the body of the message is the actual user information, which may considered as user data unit in the OSI parlance

DIGITAL FACSIMILE

Digital Facsimile which is process that digitally encoded the picture signal, i.e. encodes the baseband signal resulting from scanning the object. The facsimile equipment output may be either analog, as defined by CCITT group 3 protocol, or digital defined by CCITT group 4 , STANAG 5000 type I and STANAG 5000 Type II protocols.

Two types of facsimile systems are exists:

- ➔ Photographic facsimile
- ➔ Document facsimile

In photographic facsimile, the gray level information is transmitted and printed in addition to black and whitei.e. Typically 8 or 16 gray levels that can be recognized by the system.Document facsimile system handles only black and white levels, i.e. only two gray levels. Document facsimile system is more popular than the photographic system. The receiver / transmitter functions are applicable to both type of facsimile systems are shown in the figure below

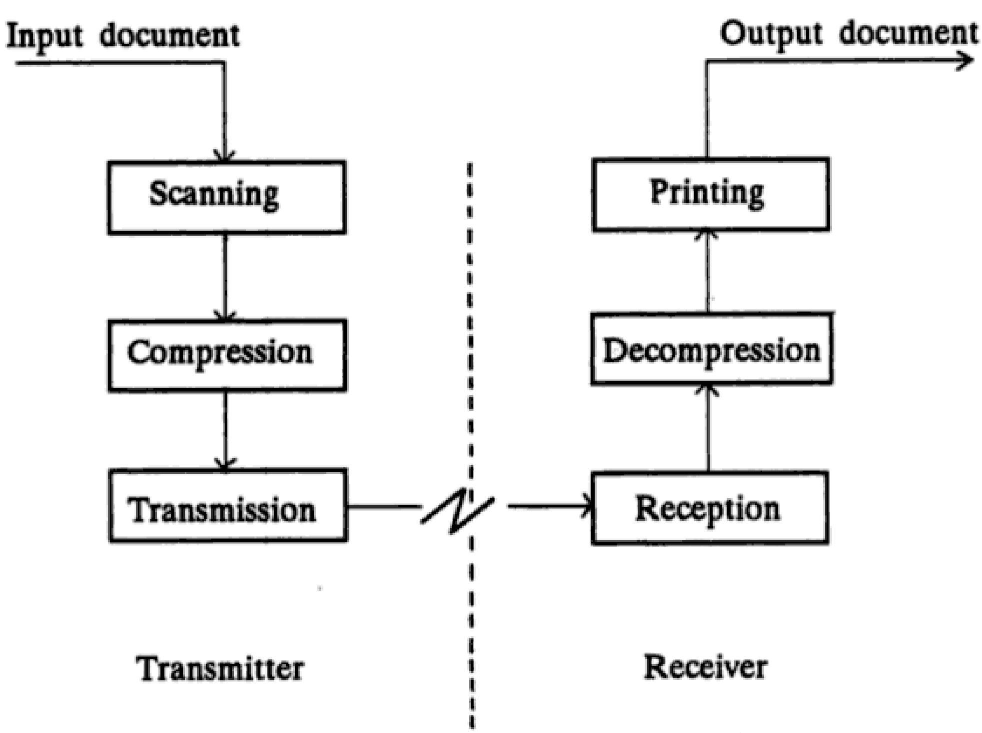


Fig.55: Function of facsimile system [1]

Facsimile inputs or scanned data are required to compress before it transmitting. This is the second step in facsimile transmission. There are two types of compression techniques

- ➔ Information preserving techniques
- ➔ Approximate techniques

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

→Modified Hoffman Technique (MH)

→Modified READ (MR) Technique

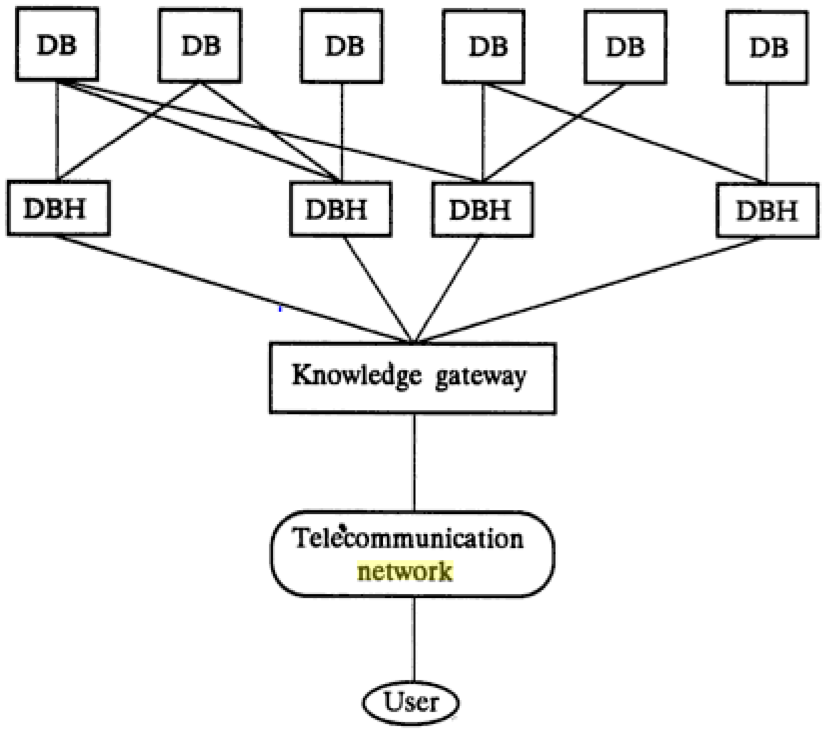
Before applying Huffman or relative element address designate (READ) technique, the scanned information is coded using a basic technique known as run-length coding. Huffman coding is based on principle that are more efficient coding technique that can be evolved by using short code words for frequently occurring symbols and long code words for sparingly occurring symbols, instead of using a uniform size for all symbols. Huffman code is modified to view the run lengths into two parts and code them independently, taking into account of their probability of occurrence. The two parts of the run length are known as **made up codepart (<64 letters)** and **terminating code part(> 64 letters)**.

Relative element address designate (READ) code is based on the principle that further code efficiency may be gained by coding the relative position of changing elements. There is a strong correlation between black-white patterns two adjacent scanned lines in a document. This fact is exploited in modified READ (MR) coding. A changing element is coded in terms of distance in preceding changing element on the same or on the previous line.

TELETEX

Teletex is an upgrade to the conventional telex service. The terminal to terminal communication service of telex will be turned into office-to office document transmission system by teletex. Teletex envisages direct communication between electronic typewriters, word processor and personal computers. In teletex system the transmission and reception of messages should proceed in the background without affecting the work which is the user may carrying out the foreground with the equipment.

How to access data from the database host?



DB-database DBH- Database host

Fig.56: Electronic access of information [1]

Considering the database access will emerge as a major application in the 1990s, ISO has initiated standardization efforts for **search and retrieval (SR)** of databases. The SR service modeled as pair of application processes. Within each application process, there are two types of functions i.e. local processing functions and OSI related functions.

There are six service elements defined in SR application.

INTIALISE- Initializes communication with a database provider for subsequent program.

SEARCH-permits a database user to search a database at a remote site; outcome is a result set.

PRESENT- permits a database user to retrieve records from a result set.

DELETE- permits a database user to delete records set at result set.

SR-RELEASE- permits a database user to orderly terminate an association.

SR-ABORT- permits a database user or database provider to request abrupt termination of the association.

27. NETWORK AND PROTOCOL ARCHITECTURE:

Network architecture of ISDN is defined to follow an evolutionary path. It is natural that an evolutionary approach is taken with regard to ISDN.As a first step the existing analog telephone networks should be converted to digital networks. These networks should then be operated along with other existing data and signaling networks. Such networks are termed **integrated digital networks (IDN).** An ISDN exchange and a suitable user-network interface permit an integrated access to network facilities. As the economics and technology dictate, the segregated infrastructure will be replaced by an integrated infrastructure as shown in figure below.

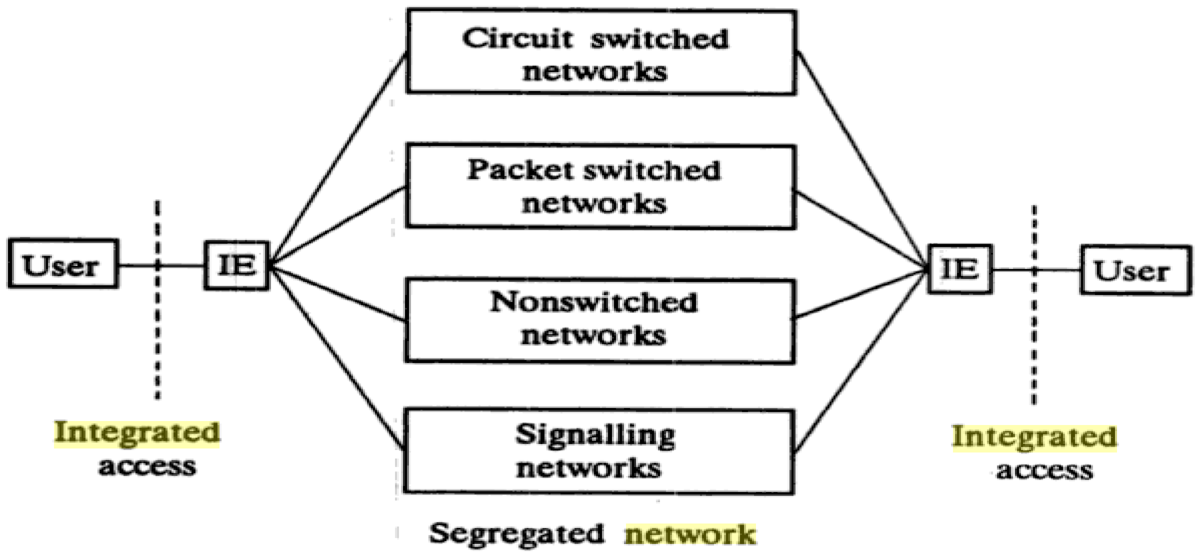


Fig.57 (a): Segregated ISDN Architecture [1]

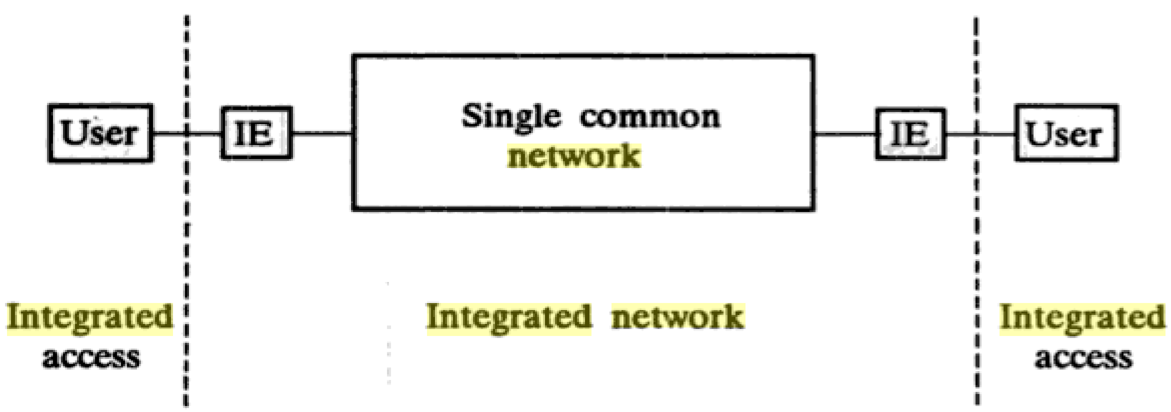


Fig.57 (b): Integrated ISDN Architecture [1]

The motivation for moving into an integrated infrastructure is based on factors like operational issues and management complexities and economics of realization. We model the segregated network as M/M/c and integrated network as M/M/1 queuing systems respectively and analyze their performance. The models used are shown in the figure below.

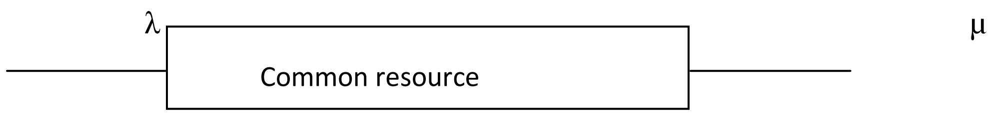


Fig.58 (a): Common resource queuing model.

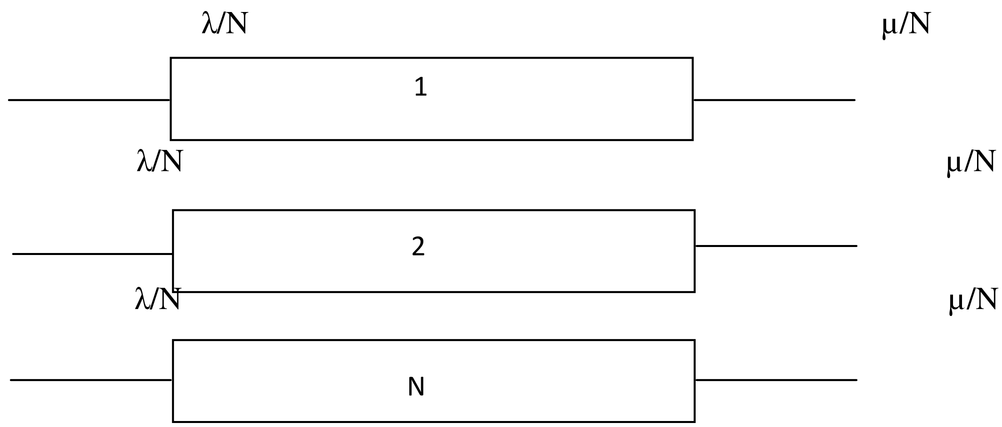


Fig.58 (b): N independent resource queuing model. [1]

The dynamics of the queuing systems are governed by the steady state birth death equations.

Let us consider the following parameters:

λ = total mean arrival rate

μ = total mean service rate

N= no. of services for the case of segregated system

The total arrival rate and the total service capacity remain the same in both the cases. For the sake of simplicity, we assume that the service capacity and the arrivals are uniformly distributed over the N servers in the segregated model. Arrivals and service models assumed to be constant and independent of the state of the system.Then the required equation will be

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

$p_1\mu = p_0\lambda$For k= 0

For different values of k, we get

$p_1\mu = p_0\lambda$or$p_1 = \rho p_0$

$p_2\mu = p_1\lambda$or$p_2 = p_0\rho^2$

$p_3\mu = p_2\lambda$or$p_3 = p_0\rho^3$

.

.

$p_k\mu = p_{k-1}\lambda$or$p_k\mu = p_0\rho^k$

Where $\rho = \frac{\lambda}{\mu}$ is known as traffic intensity. To eliminate p_0 from eqn.we use the fact that the probability sums to 1.

$\sum_{k=0}^{\infty} \rho^k \rho_0$ =1

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

Therefore,$p_0 = 1 - \rho$, ρ_k=$(1 - \rho)\rho_k$

Let us now the mean waiting time of customer T, as a measure of performance of the system. To determine T we first determine the mean no. of customers in the system M, by using the state probabilities.

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

Now, differentiating both sides of the equation

$\sum_{k=0}^{\infty} \rho^k$=$\frac{1}{(1-\rho)}$ with respect to ρ ,we get

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

Multiplying both sides by ρ ,we obtain

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

Therefore M=$\frac{\rho}{1-\rho}$

Now we use the results the mean number of customers in any queuing system is equal to the product of the mean waiting time for a customer and the mean arrival rate. The result is valid for all queuing systems irrespective of arrival of time or service time distributions.

Hence, M=λT, or T=M/λ

For the integrated model ,we have T1=$\frac{\rho}{1-\rho} \times \frac{1}{\lambda}$

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

$T1=\frac{\lambda}{\mu(1-\lambda/\mu)} \times \frac{1}{\lambda} = \frac{1}{\mu-\lambda}$

For the segregated system, we have

$Ts=\frac{1}{(\mu/N)-(\lambda/N)} = \frac{N}{\mu-\lambda} = NT1$

Shows the delay performance of segregated system deteriorates by N times if the integrated system is broken into N segments with 1/Nth of the capacity of the integrated system.

1. The total capacities of the integrated and segregated networks are identical.
2. Input services are identical requirements in both the cases.
3. Services rate distributions are identical.

The protocol architecture of ISDN layer is according to the OSI reference model.ISDN is largely concerned with the lower layers 1-3 Layers 4-7 concern of service providers. Accordingly lower three layers are known as bearer service functions. Service offered by service providers are known as teleservices.

Layer 1(Physical Layer)

The physical layer is the lowest layer of the OSI model. It defines the mechanical, electrical, functional and procedural aspects of the physical link between networks. Physical layer standards have been widely used for years in point-to-point wide area network applications. (CCITT/ITU has established X-21–X-24 to specify the functions at the physical level of the based circuits. Numerous other standards such as EIA-232 and V-21– V-34 are widely used for various purposes at the physical layers. The physical layer implements an unreliable bit link. A link consists of a transmitter, a receiver and a medium over which signals are propagated. The physical layer data consists of stream bits. The physical layer defines the type of encoding to convert the bit stream into electrical or optical signal to transmit in the medium. At receiver the physical layer converts back into bit stream. The receiver must be in synchronism with transmitter to receive the specific bit pattern. To assist synchronization, the physical layer adds a specific bit pattern called preamble at the beginning of the packet.

Layer 2(Data Link Layer)

1. Establishing and clearing the data links
2. Error, flow and congestion control
3. Synchronization

The data link layer defines the frame format such as start of frame, end of frame, size of frame and type of transmission. The principal service provided by the data link layer to higher layer is that of error detection and control. This layer is the first software protocol layer of the OSI model. It specifies the data format, sequence, and acknowledgement process and error detection methods. The data link layer accepts information from the network layer and breaks the information into frames. It then adds the destination address, source address, frame check sequence (FCS) field and length field to each frame and passes each frame to the physical layer for transmission on the receiving side, the data link layer accepts the bits from the physical layer and from them into frame, performing error detection. If the frame is free of error, the data link layer passes the frame up to the network layer. It performs frame synchronization, that is, it identifies the beginning and end of each frame. Existing protocols for the data link layer are :

1. Synchronous Data Link Control (SDLC). Developed by IBM as link access for System Network Architecture.

2. High Level Data Link Control (HDLC). It is a version of SDLC modified by the ISO for use in the OSI model.

3. Link Access Procedure Balanced (LAPB). The modified version of HDLC is LAPB.

The data link layer also performs the flow control and access control. By flow control, if the rate at which the data are absorbed by the receiver is less than the rate produced in the sender, the data link layer imposes a flow control mechanism to prevent overwhelming the receiver. By access control, when two or more devices are connected to the same link, the protocol of data link layer determines which device has control over the link at any given time. Two sub layers defined in the data link are the media access control (MAC) and the logical link control (LLC) layer. MAC performs address management function. LLC manages flow and error control, automatic requests for retransmission (APQ) and handshake processes

Layer 3 (Network layer)

1. Addressing and routing
2. Establishing and clearing network level connection
3. User-to-user signaling
4. Network level multiplexing
5. Internetworking Multiplexing

If two systems are attached to different networks (links) with connecting devices between the networks (links), there is often a need for the network layer to accomplish source to destination delivery. Thus the function of the network layer is to perform routing. The network layer checks the logical address of each frame and forwards the frame to the next router based on a look up table. The network layer is responsible for translating each logic address (name address) to a physical address (MAC address).

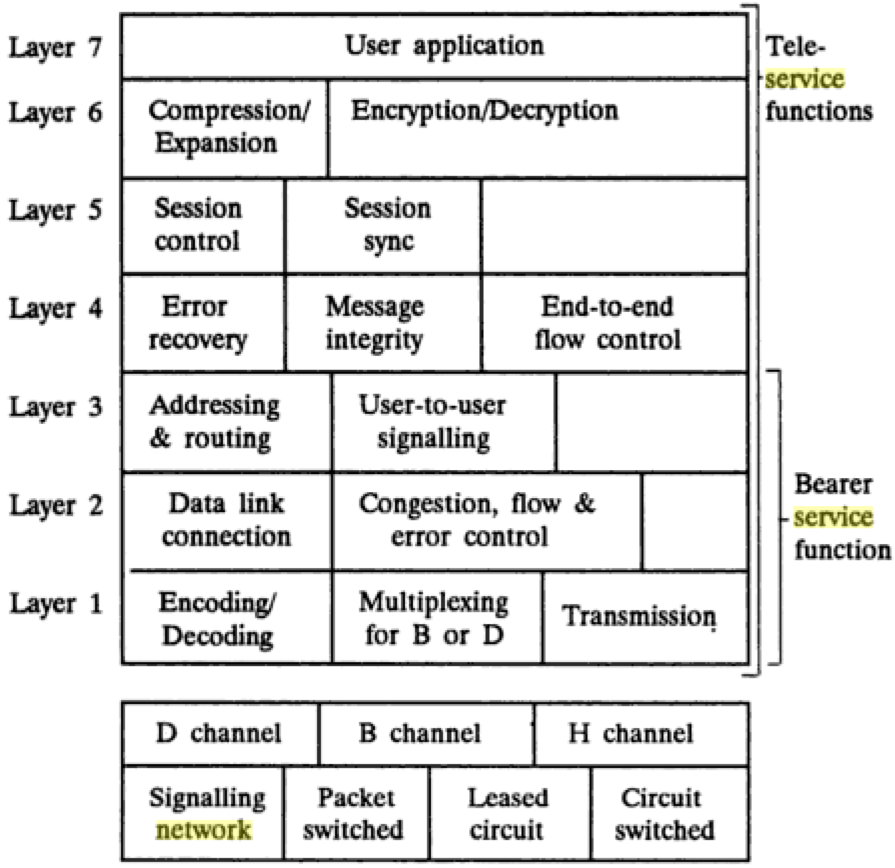


Fig.59: ISDN protocol Architecture [1]

Transmission channel B, D, H will be discussed in the Transmission channel part.Channel level multiplexing is also dealt with that section. Basic and primary rates are discussed, which also deals with some layers 3 functionalities. User-to user level signaling is covered in section. Addressing structure is discussed with interworking of networks. Further details of bearer and teleservices are presented in the section below.

Layer 1 protocol definitions apply uniformly to all transmission channels and services. But, from layer 2 upwards, the protocol structure differs for different channels and services. For example in layer 2, a new protocol, LAP-D has been defined for handling D channel information. Level 2 of X.25 is used for packet switched connection over B channel. LAP-D protocol is modeled after LAP-B protocol of X.25.

Transport Layer

The transport layer provides a mechanism for the exchange of data between end systems. It optimizes the use of network services with providing a requested quality of service to session entities. The size and complexity of a transport protocol depend on how reliable or unreliable the underlying network and network layer services are. Essentially, this layer is responsible for the reliable data transfer between two end nodes and is sometimes referred to as host-to-host layer.

Support Layers

The session layer, presentation layer and application layer are considered as support layers. All these layers explained in the following paragraphs.

Session Layers

The session layer establishes a logical connection between the applications of two computers that are communicating with each other. The session layer concerns file management and overall networking functions. Access availability and system time allocations are included in this layer. The session layer can partition a transfer of a large number of messages by inserting synchronous points. Specific responsibilities of the session layers are authentication of user access, fault recovery if a break in service occurs, permitting multiple applications to share a virtual circuit, connection and disconnection of any node from the network.

Presentation Layer

The presentation layer receives information from theapplication layer and converts into ASCII or Unicode or encrypts or decrypts. This layer isconcerned with the syntax and semantics of the information exchanged between two systems.

The three basic forms of protocols used in the presentation layer are

1. Virtual terminal protocol, which is used to allow different types of terminals to supportdifferent applications.
2. Virtual file protocol, which handles code conversions' within files, file communicationand file formatting.
3. Job transfer and manipulating protocols which controls the structure of jobs andrecords.

A separate protocol function known as abstract syntax notation (ASN) specifies file datastructure.

Application Layer.

This layer enables users to access the network with applications such as e-mail, FTP and Telnet. It provides user interfaces and support for various services. Specific services are (*a*) Network virtual terminal (*b*) File transfer access and management (FTAM), (*c*) Mail services (*d*) Directory services (*e*) Specific User Service Element (SUSE) — deals with actual user requirements for access and use of the network (*f*) Common application service element (CASE) — This sets guidelines for the applications required quality of service and (*g*) Specific application service element (SASE) — deals with large amount of data, including database access.

28. TRANSMISSION CHANNELS

In simple word transmission channel is nothing but a path between two nodes in the network. It may refer to the physical cable, the signal transmitted within the cable or to a sub channel within the carrier frequency. There are three types of fundamental channels in ISDN around which the entire information transmission is organized. These are

- ➔ Basic information channel or B channel,64 kbps
- ➔ Signaling channel or D channel, 16 or 64 kbps
- ➔ High speed channel or H channel

These three ISDN channels are described below.

B channel (Bearer Channel):B channels are logical digital ''pipes'' which exist on a single ISDN line. B channel carry data and services at 64 kbps. It carries data in full duplex mode. Each B channels provide a 64 kbps clear channel, clear meaning that the entire bandwidth is available for data, B channels typically form circuit switched connections. B channel connection is an end-to-end physical circuit that is temporarily dedicated to transferring data between two devices. The circuit switched nature of B channel connections; combined with their reliability and relatively high bandwidth makes ISDN suitable for a range of applications including voice, video, fax and data. B channels are normally used for on-demand connection. As B channel operation based on circuit switching, it can be configured as semi-permanent or ''nailed up'' connections.

D channel (Delta Channel): D channel can be either 16 or 64 kbps, depending on the needs of the user. The primary function of the D channel is to carry control signaling and administrative information for B channels to set up and tear down the calls. The D channel uses packet switched connection. The packet switched connection are best adapted to the intermittent but latency sensitive nature of signalling traffic, accounting for the highly reduced call setup time of 1 to 2 seconds on ISDN calls. Unlike the B-channel, which can function as a simple 'pipe', the D channel is associated with higher level protocols at layers 2 and 3 of OSI model which form the packet switched connections. The D channel provides the signalling information that is required for caller identification. It also includes low-rate data transfer and applications such as telemetry and alarm transmission.

H channels(Hybrid Channel).H channels are suitable for high data rate applications such as video, teleconferencing and so on. Table 4 gives ISDN channel and its specifications.

Channel	Bit rate (kbps)	Interface	Purpose
B	64	BRI	Bearer services
H0	384	PRI	6 B channels
H11	1536	PRI	24 B channels
H12	1920	PRI	30 B channels
D	16	BRI	Administrative and control signalling
D	64	PRI	,,

Table 4: ISDN Channel specification [1]

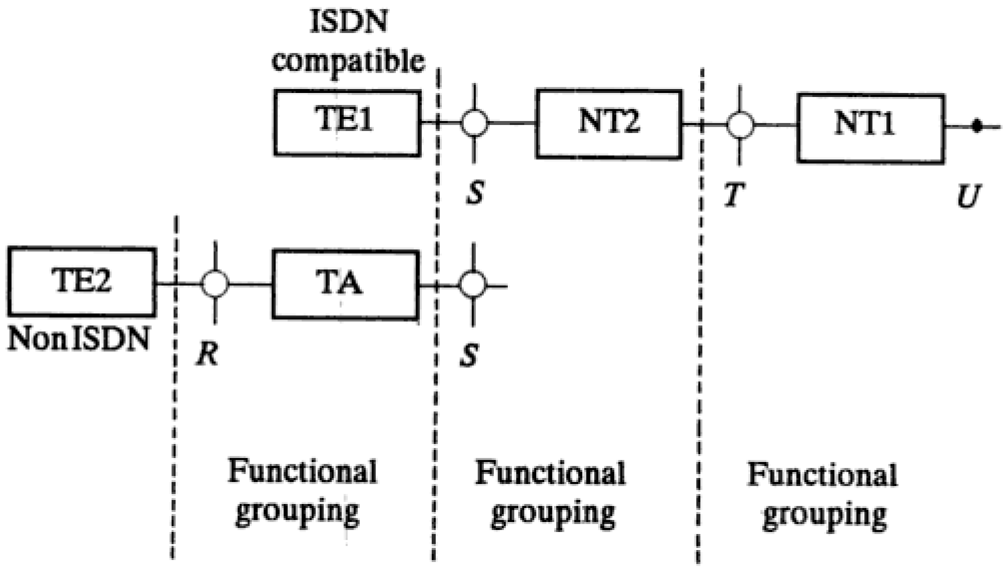
27.1 USER NETWORK INTERFACES

Comprehensive user network interface definitions are key to ensuring worldwide ISDN compatibility.ISDN categories to variety of services such as voice, data, telemetry and image. Two information rate access interfaces have been standardized for ISDN.

- ➔ Basic rate access
- ➔ Primary rate access

Functional grouping and reference points

Taking into account these regulatory factors, ISDN users network interfaces are functionally grouped and the associated access points at different functional levels are known as **reference points.** Figure 56 shows the functional groupings and the points for ISDN user network interfaces. There are three CCITT official reference points R, S,and T. There is a fourth reference point U which has come about on account of current regulations in the USA.



R,S,T -CCITT reference points U-Reference points in USA
NT1- Network termination 1 NT2- Network Termination 2
TE1- Terminal equipment type 1 TE2- Terminal equipment type 2
TA- Terminal adapter

Fig.60: Functional groupings and ISDN Reference points [1]

T reference point will have to concern itself only with the function of layers 2-6.Network termination 2 (NT2) functional grouping includes the function of ISDN layers 2 and 3 in addition to NT1 functions. Consequently, equipment connected at the S reference points is concerned only with the function of higher level layers 4-7. Reference points S,T and U are ISDN compatible points and ISDN compatible equipment can be connected to these points. ISDN compatible equipment are known as terminal equipment type1 (TE1).There is a functional grouping that would support the non-existing ISDN equipment. This is achieved by using terminal adapter (TA) between the S reference points and non ISDN terminals. The access points for non ISDN equipment is the reference point of R. The non ISDN terminals belong to the terminal equipment type 2(TE 2). After having defined the reference points and functional groupings, ISDN permits considerable flexibility in the physical realization and implementation of functions. There are different types of switched applications networks listed below.

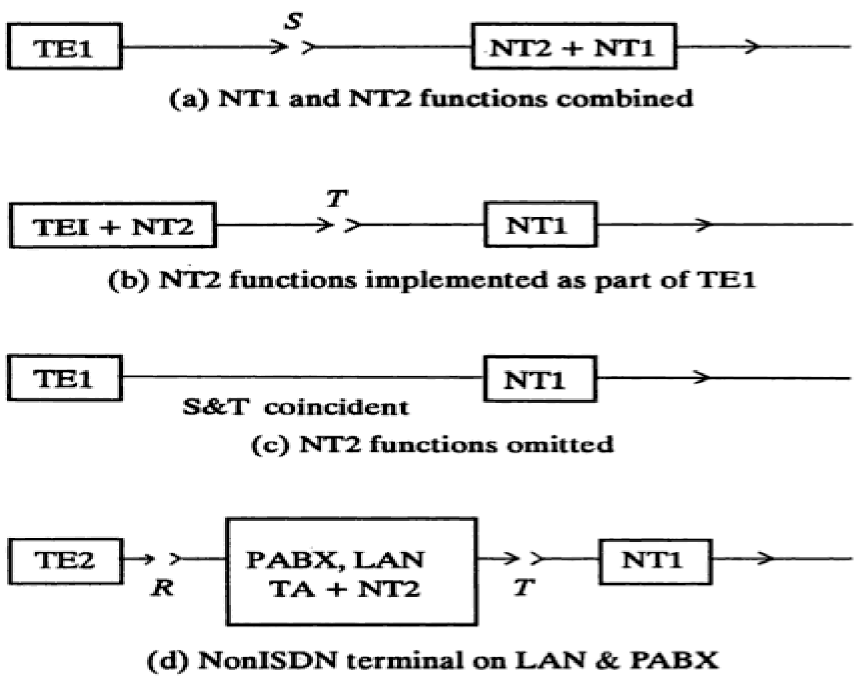


Fig.61: Some physical implementations of ISDN functional groupings [1]

28. SIGNALLING

ISDN uses a common channel signaling scheme. The signaling is done over the D channel which acts as the common signaling channel for B and H channels which carry the user information channel is also used for carrying some user information,if there is spare capacity. In such cases also the required signal is done on the D channel. The concept of common channel signaling and the CCITT's signaling systems (SS7) have been discussed here.

Signalling in ISDN grouped into two distinct categories.

- ➔ User level signaling
- ➔ Network level signaling

28.1 User Level Signalling:

User level signaling in ISDN permits a user to

1. Establish, control and terminate circuit switched connections in B channel,
2. Carry out user to user signaling
3. Establish, control and terminate packet switched connection in B or D channels.

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

- ➔ Message based signaling
- ➔ Stimulus signaling

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

1. Call establishment messages
2. Call control messages
3. Call disconnect messages
4. Miscellaneous messages

Call establishment group includes set up, call proceeding, alert, connect and connect acknowledge messages. Alert signal corresponds to ring back signal and is used when a non automatic answering terminal is used at the receiving end. If the auto answering facility is available, the terminal responds with connect signal directly and the alert signal is skipped.

Call control groups include suspend and resume messages and also user to user messages.

Call disconnect groups includes disconnect, release, and release complete messages.

The primary function of miscellaneous messages is to negotiate network facilities to support additional service features like call forwarding, direct inward dialing, reverse charging etc.

All user level messages have a common message format. These fields are mandatory for all messages.

1. Protocol discriminator
2. Call reference
3. Message type

As the D channel may carry computer and telemetry data etc. In addition to signaling messages, it is necessary to have a mechanism for differentiating packets and their associated protocols. The protocol discriminator field is provided for this purpose at present two message protocols are supported the ISDN signaling message protocol and X.25 level 3 packets protocols.

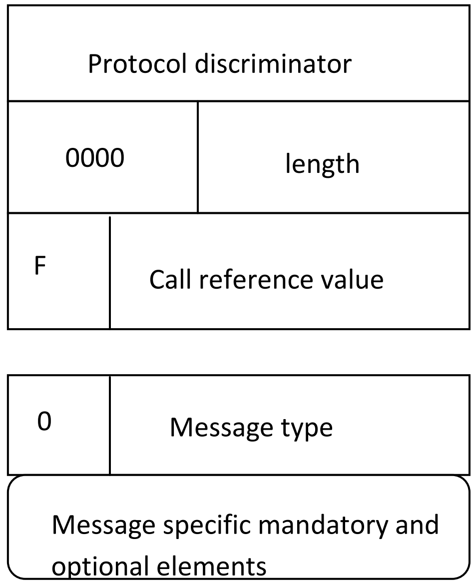


Fig.62: User level signaling message structure [1]

The field has three sub fields: length subfield, flag and reference value. The call reference field gives reference to the B, H, or D channel information transfer activity to which a signaling packet pertains. Depending upon the service and channel used, the length of the call reference value may vary.1-bit flag is used to indicate which end of the connection initiated the call.

Stimulus signaling is used when the user end equipment are dumb device with no intelligence e.g. digitaltelephone. As the device don't have functional capabilities, stimulus signaling messages are generated as a direct result of actions by terminal user. The signals send by a network to an intelligent device are in the nature of inducing specific events at the terminal end. Stimulus signaling procedures are defined as a compatible subset of a message based on signaling procedures in order to facilitate functional expansion for simple terminals.

28.3 NETWORK LEVEL SIGNALLING

Network level signaling in ISDN is concerned with interoffice signaling. Circuit suspension call supervision messages are examples of network level of signaling. The procedure for network level signaling is defined as the ISDN user part (ISUP) of the signaling system 7.One of the main aim of the context of the ISUP has been evolve a flexible design for signaling system to accommodate new services and connection type that may come about in the future to be supported on ISDN.

About 40 network level message have been standardized so far these messages are broadly categorized into 9 units.

1. Forward address
2. General setup

3. Backward setup
4. Call supervision
5. Circuit supervision
6. Circuit group supervision
7. In call modification
8. End-to end user
9. User-to-user

Messages belonging to 1-4 above are used to support the call setup process initiated by user and start accounting and charging functions. Circuit and circuit group supervision messages are permit blocking and de-blocking of circuits and circuit groups respectively. End-to end signaling or node-to-node signaling in between the originating and terminating ISDN exchanges and in accomplished in two ways. The pass along service of ISUP enables end –to end signaling. Another way of doing end-to-end signaling is to use services of the signaling connection control part(SCCP) of the SS7.This method is uniformly applicable and unlike pass along method is independent of the presence or absence of circuit connection between message originating and terminating exchanges.

A common format for all messages is defined for ISUP.The format messages consists of six steps.

1. Routing level
2. Circuit identification code
3. Message type
4. Mandatory fixed part
5. Mandatory variable part
6. Optional Part
   Routing level indicates the source and destination exchanges of message include the link selection subfield.

29. NUMBERING AND ADDRESSING:

In telephone and data networks, the end equipment are more often single units than multiple device units like PABX or LAN. Primarily a telephone, a computer, or a terminal has been the predominant end equipment. The numbering system is generally required single equipment end points.

But, in ISDN multiple device at the end points are more of the normal than single units. so it requires specific equipment end points. Specific equipment is a two level process; first the end point is identified as in the case of telephone or data network and then equipment at the end point. The component of the ISDN address which is used to identify the end point is known as the **ISDN number,** and the component for identifying the specific equipment at the end point is called as **ISDN sub address.**

The numbering plan for ISDN is as follows:

1. It is based on, and it is attachment of, the telephone numbering plan. In particular the country code is evolved for the telephone numbering is defined as CCITT standard E.163 is adopted in to for ISDN.
2. It is dependent of nature of the service (e.g. voice facsimile data) or the performance characteristics of the connection.
3. It is dependent of routing, i.e. the numbering or addressing does not specify the immediate exchanges through which the service is to be put through. Some addressing schemes in data network demand that the complete route from source to destination be specified as a part of address. E.g. UNIX.
4. It is the sequence of decimal digits. No alphabet or other characters are permitted as part of the address.
5. Its design is such that interworking between ISDNs require only the use of ISDN number and no other additional digits or addressing signals.

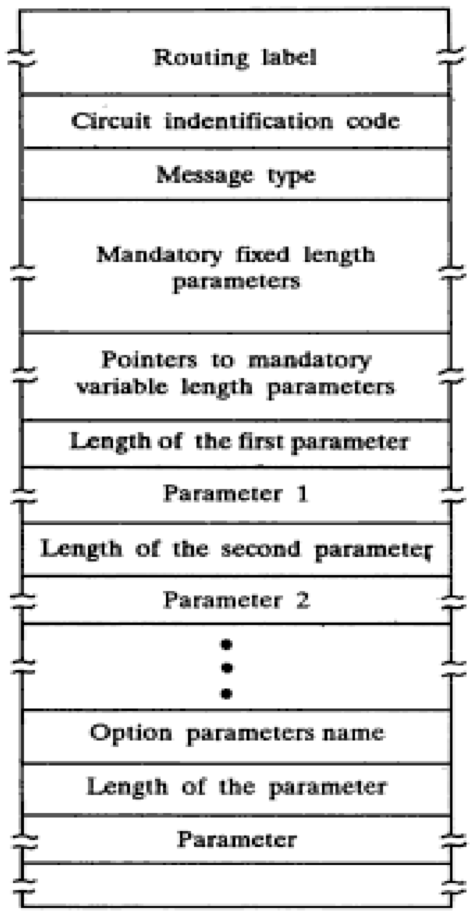


Fig.63: Message format for ISDN user part [1]

29.1 Address Structure

In ISDN address the number part is maximum of 15 digits and the ISDN sub address part a maximum of 40 digits.

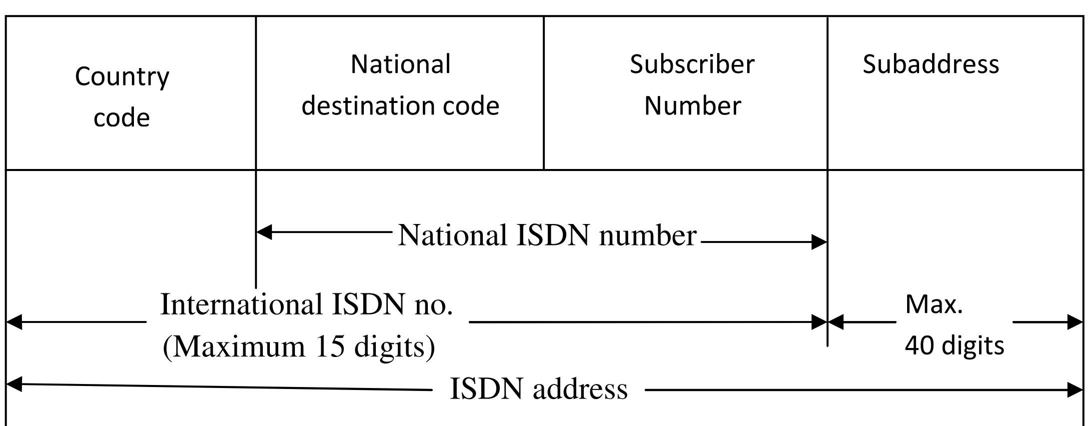


Fig.64: ISDN Address structure [1]

National destination code is like an area code in telephony network and is of variable length. ISDN subscriber number is the only normally listed in the directories. An ISDN number is a unique worldwide address and unambiguously identified an end point connection. The end point may be

1. Single S or T reference point
2. One of many T reference points at the same site
3. One of many S reference points using direct inward dialing feature.

A single S or T reference points may also be addressed by multiple ISDN numbers. This feature generally used in internetworking.

A sub address is a part of ISDN address, is carried in separate field in the user network interface message. The typical address using both ISDN number and the sub address as shown figure below.

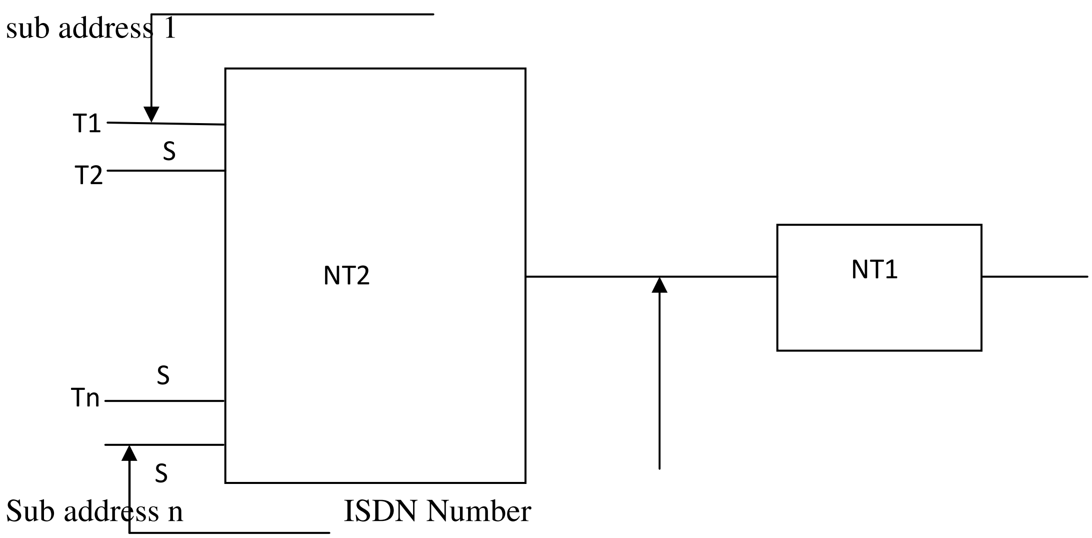


Fig.65: Example of ISDN addressing [1]

30. SERVICE CHARACTERISATION

ISDN services placed under two categories:

- ➔ Bearer service
- ➔ Teleservices

Bearer services are accessible at T and S reference points for ISDN compatible equipment. For non ISDN terminals the bearer service is accessible at R reference points of the user network interface. A packet data transmission is an example of bearer service, whereas telephony,teletex,videotext and facsimile are under class of teleservices.

Figure 62 shows the ISDN service possibilities.

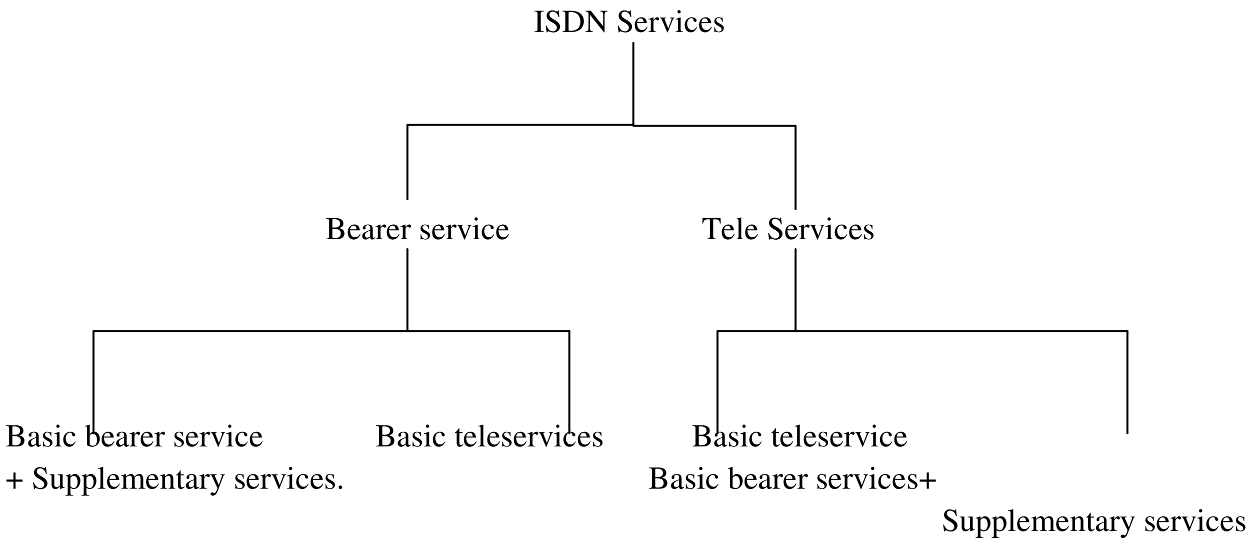


Fig.66: ISDN service categories [1]

Supplementary services call for additional functionalities both in lower and upper layers, depending on whether they supplement a basic bearer service or a basic teleservice.
Bearer service or lower level service attributes are under three categories:

1. Information transfer attributes
2. Access attributes
3. General attributes

Information transfer attributes describe the network capabilities for digital information transfer from S/T reference point of one customer to one or more S/T reference points of other customers. The attributes used for the purpose

1. Transfer mode- circuit switched mode or packet switched mode
2. Transfer rate- one of many standard rates
3. Structure-This type of transmission implies that the integrity of data is maintained by conveying timing information along with the data transmitted.
4. Transfer capability- It specifies the ability of network to carry structured data or unstructured data, restricted or unrestricted data, speech,audio and so on.

Some typical examples of transfer capabilities are:

➔ 8KHZ structured unrestricted

➔ 8KHZ structured speech

➔ 8KHZ structured ,3.1KHZ audio

The information transfer attributes 1-4 are referred to as **dominant attributes**.

5. Establishment communication-It can take three values demand, reserved, or permanent connection. In demand communication, a set communication is setup and released on demand. In reserve mode setup and release time is fixed in advance by the customer. In permanent a connection is pre-established.
6. Establishment of connection-It may take three values: switch, semi-permanent, permanent. Semi-permanent connection is a switched connection but provided for an indefinite period. Permanent connection is non switched connection, like a leased line bypassing the ISDN exchange.
7. Communication configuration-The configuration of communication may be point-to-point, multipoint or broadcast.
8. Connection configuration-The attribute connection configuration is of three sub attributes:
   → Topology
   → Uniformity

→ Dynamics

Topology may be simple with one connection element or tandem with two or more connections being formed with two or more elements put in parallel. **Uniformity** specifies the homogeneity of elements involved in connection. The **dynamics** connection deals with temporal aspect of establishing or releasing connections.

Symmetry deals with the information flow characteristics. Information flow may be unidirectional or bidirectional symmetric or bidirectional asymmetric. The information transfer attributes 5-9 are referred to as **secondary attributes.**

The third category of bearer service attributes, i.e.**general attributes** including the followings.

1. Supplementary services
2. Quality of services
3. Connection performance
4. Interworking
5. Commercial or operational attributes

A List of supplementary services in ISDN

Abbreviated dialing	closed user group
Do not disturb	reverse charging
Call waiting display	call forwarding
Call bearing	conference calls
Three party services	directdialing
City wide Centrexcredit card calling	

A number of parameters are associated with quality of services:-

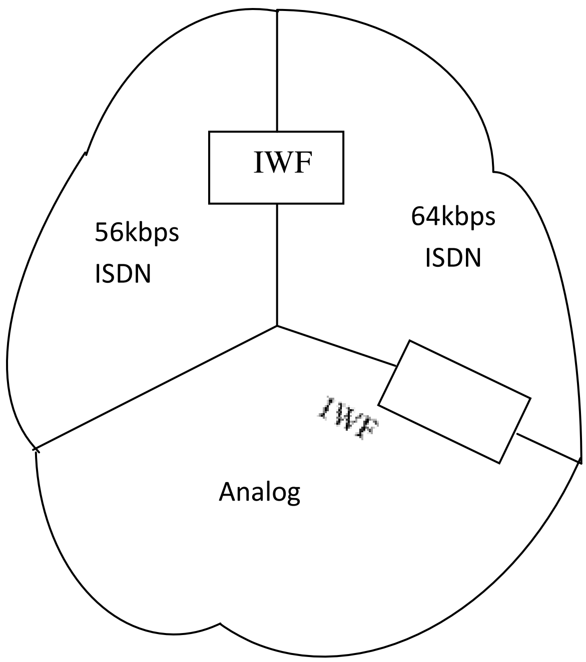
Flexibility-Ability to choose different services or terminal equipment and to interwork services
Cost/Charging-Ability to give correct billing and appropriate billing/ charging information desired by the user.
User friendliness- Provision of prompts and voice announcements
Ergonomicity-Aesthetic and ergonomic aspects of terminal equipment
Waiting time- Waiting time for the user to obtain service or terminal equipment
Enhance ability-Progressive introduction of new facility and technology upgrades.
Reliability-Mean time between failures, mean time to repair.
Promptness- Delay in establishment /release in connection
Fidelity-quality of production
Backup assistance-User training, maintenance
Transfer speed-End to end information transfer time or through put
Universality-ability to interwork other private or public network

The attributes characterizing teleservices are also placed under three broad categories.:
1. Low layer attributes
2. High layer attributes
3. General attributes

Low layer attributes is same as the attributes for bearer services and in particular the information transfer and the access attributes. High layer attributes deal with different communication aspects such as voice, audio, text, facsimile and picture transmission. They are defined in the forms of protocol confirming the OSI model.

31. INTERWORKING

ISDN interworking network based on three types of networks. i.e. the analog network, the 56kbps digital network for voice or data, and the ISDN with 64 kbps clear channel. Interoperability is achieved through interworking function (IWF) are different for different network and interface details.



IWF=Interworking Function

Fig.67: Coexistence of ISDN with other network [1]

Separate reference points have been identified and their interface details have been defined to enable interworking with different networks.

Reference points	Interworking With
K	Analog or IDN (voice or data) when the IWF is implemented in ISDN
L	Analog or IDN when the IWF is implemented as part of existing network
M	Specialized service providers or value added networks(VANs)
N	Another ISDN
P	A specialized resource with ISDN

Interworking functions of ISDN with other networks having following objectives

1. Determine if the network resources of the nonISDN network are adequate to meet the ISDN service demand.

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

The above interworking functions point to need the parameter exchange among the existing nonISDN terminals, ISDN terminals, and the strategically located IWF modules.

31.1 Numbering Interworking

The international PSTN numbering scheme used a 12 digit numbers with a two part format of country code and national significant number. Other ISDN also uses 15 digit numbers.ISO defined its own numbering scheme in the context of OSI reference model. However two basic methods for interworking between the ISDN numbering plan and other have been proposed:

- ➔ Two-stage or port method
- ➔ Single stage or integrated numbering method.

In the port method the numbering plan of the network to which the called party is attached is treated as calling party's network and a two stage dialing is used.

The main disadvantages of two stage dialing are:

- ➔ The user dials two sets of numbers.
- ➔ Dialing is dependent on the relative numbering plans of the called and calling terminal.
- ➔ A delimiter or pause is necessary between two stages e.g to obtain the dial tone of second network.
- ➔ The dialing becomes even more cumbersome if more than two networks are evolved in establishing a connection.

An important advantage of two stage scheme that is simple to implement and is universally acceptable.

In a single stage or integrated numbering scheme, a prefix is used to identify a particular network and thereafter the numbering scheme of the destination network is dialed.

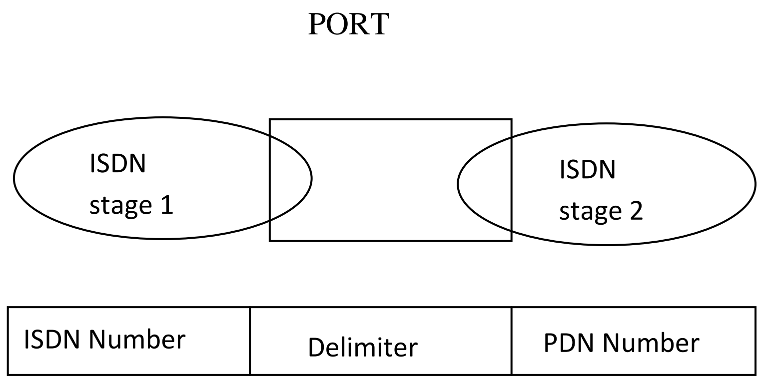


Fig.68 (a): Two stage dialing [1]

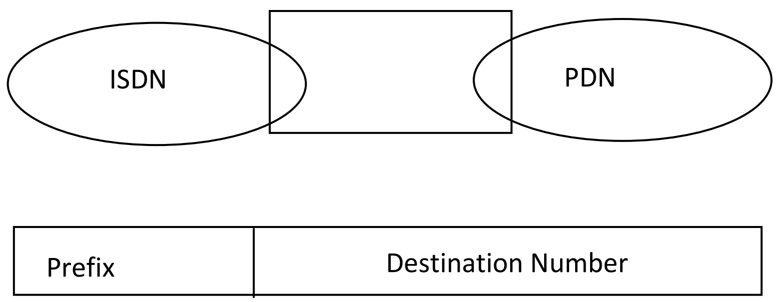


Fig.68 (b): Single stage dialing [1]

32.BROADBAND ISDN (B-ISDN)

B-ISDN provides the needs (High speed and large data handling) of the next generation technology. B-ISDN is a digital service with speed above 1.544 Mbps. The original ISDN is called narrow band ISDN (N-ISDN). B-ISDN uses fiber at all levels of telecommunications. BISDN provides two types of services. They are interactive (conversational or messaging or retrieval). The interactive service is bidirectional. The distributive services are unidirectional (with user control or without user control). Several forms of B-ISDN exist. Some are listed below:

Frame relay service:-Frame relay is considered to be a B-ISDN service. Frame relay is a packet switching protocol service offered by telephone corporations to replace the X-25 protocol. It is a WAN network.

Switched Multimegabit Digital Service (SMDS):- SMDS is a digital service that provides a high speed digital path. The transport speed of SMDS is usually 155 Mbps.

ATM:-The transport speed of most ATM applications is 155 Mbps. B-ISDN uses the same functional groupings of N-ISDN. But, named as B-NT1, B-NT2, B-TE1, B-TE2 and B-TA. B-

ISDN uses the same R, S, T and U reference points. However BISDN uses three different access methods to satisfy the user needs. They are symmetrical 155.52 Mbps, asymmetrical 155.520 Mbps and symmetrical 622.080 Mbps.

32.1 VOICE DATA INTEGRATION

Generally in segregated infrastructure, we have collection of different types of networks; like packet switched circuit switched, no switched and so on. Depending on the nature of the traffic of the service, the service is carried on. For example voice traffic is carried on circuit switched network, whereas data traffic is packet switched network. In this topic we have considered about the traffic i.e. of two types: digitized voice and data. In this section the part is discussed about voice data integration on a single channel.

Table: Parameters relating to Digitized voice and data traffic

Digitized Voice	Data
Periodic busty in nature	A periodic busty in nature
Fixed length bursts	Variable length bursts
Small packet size	Large packet switch
Packetisation time critical	Packetisation time is not critical
Hard bound on delay	Soft bound delay
Hard bound on variance of delay	soft bound on the variance of delay
Loss of parts of speech acceptable	Loss of part of data unacceptable
Low overhead as there is no error recovery	High overhead due to error detection and recovery

Data traffic is aperiodic whereas voice traffic is periodic. The traffic for digitized voice may be considered to be busty, through periodic as the transmission may be organized on sample by sample basis i.e. one byte in every 125 µs.If packetized voice transmission is used, and then the packet size is to be small to maintain the real time characteristics of the service. Large packet sizes would lead to be broken speech transmission. For similar reason the packetisation time is critical for voice packets. Voice packets should be reached the destination within the specified time or else speech would discontinuous. A basic model for integrating voice and data is to considered a TDM frame that is capable of carrying both circuit and packet switched data. A portion of frame Fc bits is occupied by the circuit switched data and the remaining portion Fp bits by packet switched data as shown in figure below.

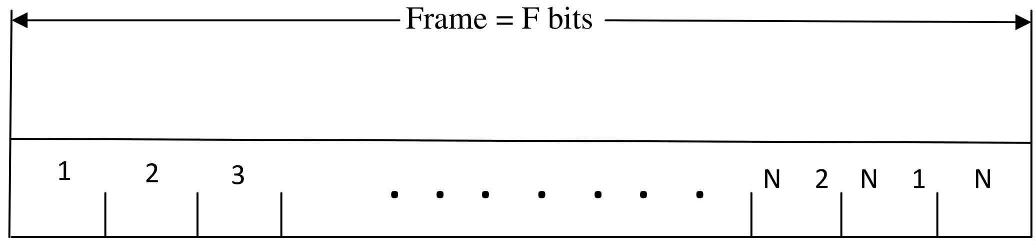


Fig.69: An integrated TDM network [1]

Circuit switched traffic= Fc bits

Packet switched traffic= Fp bits

Fc+ Fp=F

A number of circuit switched sources may constitute the Fc bits of the frame. Each source may require different bandwidth and have different arrival time and holding time characteristics. A TDM frame may considered containing N slots of b bits each, as a group of b bits likely to be integral in any traffic.e.g. Voice samples of 8bits or a byte in data.

F=Nb slot ,One slot may be considered as a basic bandwidth unit(BBU) .A source is assigned one or more BBU's depending upon the bandwidth requirements. A circuit switched source is blocked if adequate no. of BBUs are not available for the assignment and a packet switched source is queued awaiting free BBUs. For voice data integration we restrict the circuit switched traffic to voice. i.e. all sources have equal bandwidth requirement of one BBU and the same arrival time and the holding time characteristics. Similarly all packet switched sources may be considered to have the same arrival and service time characteristics for simplicity.

Voice data Integration schemes followed by may be placed under two characteristics:

- Random allocation of BBUs
- Contiguous allocation of BBUs

In random allocation, any available BBU is allocated to either voice or data using some criterion. The usual criteria include First-come-first-serve basis (FCFS) and pre-emptive scheduling. In contiguous allocation schemes attempts are made to gather together all BBUs allotted to voice and so also BBUs allotted to data traffic. The contiguous allocation schemes can be classified as:

- Fixed boundary scheme
- Movable boundary scheme

In Fixed boundary scheme, the N slots in the frame are divided into two parts containing N1 & N2 =N-N1, slots respectively as shown figure 65(a).

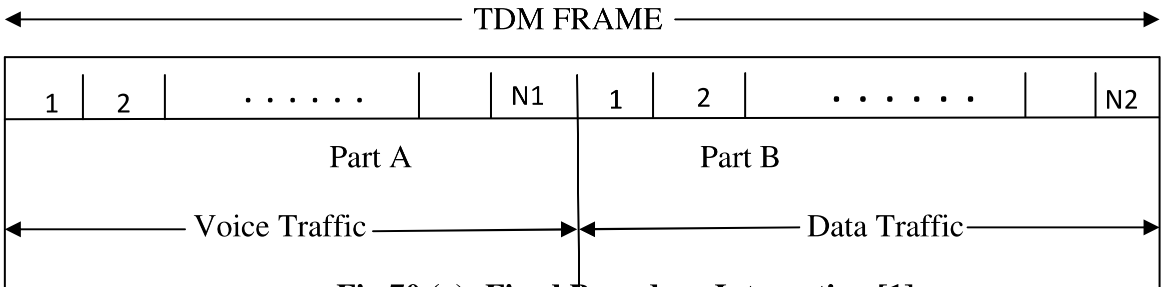
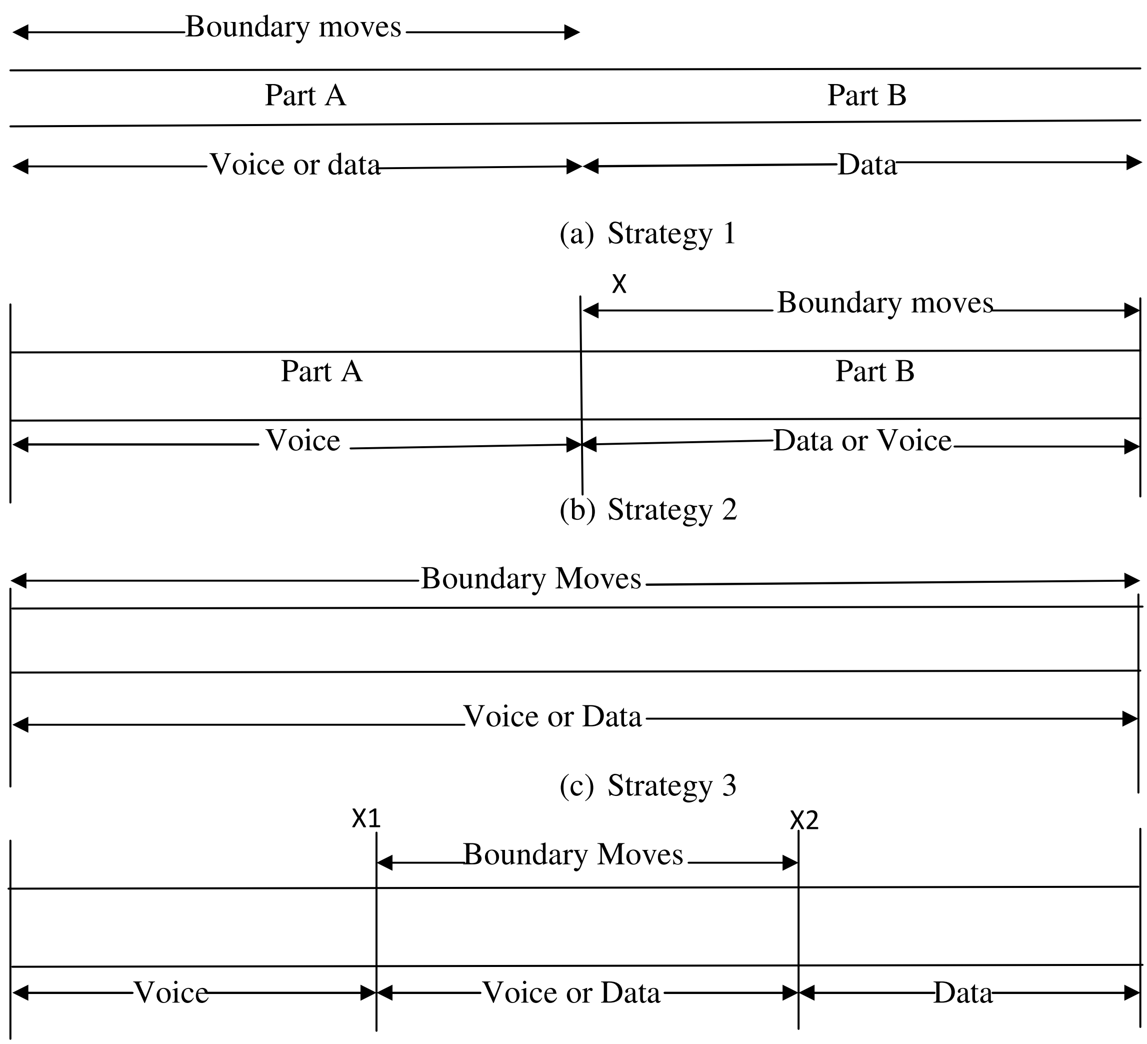


Fig.70 (a): Fixed Boundary Integration [1]

It has two parts part A for **voice traffic** and part B for **data traffic**. Voice traffic is always scheduled in part A, and if there are no free BBUs in that part, the call is blocked, even though free BBUs may be available in part B. Similarly data traffic will be queued if there are no free BBUs in part B,even though free BBUs available in part A. The performance is measured in terms of blocking probability for voice sources and mean delay for data sources.In movable boundary scheme, four strategies are possible as shown in figures below.



(a) Strategy 1

(b) Strategy 2

(c) Strategy 3

(d) Strategy 4**Fig.70 (b): Movable Boundary Schemes [1]**

In strategy 1 and 2, there is a national boundary shown at X. In the former data traffic is allowed to use free BBUs. If a voice source arrives, then a data source occupying the voice area is pre-empted and the BBUs are allotted to the voice source .In this sense the boundary moves back and forth to the left of X. If all the BBUs in part A occupied by voice sources ,further voice arrivals are blocked even if free BBUs are available in part B. This strategy gives the same blocking probability performance as the fixed boundary scheme for voice sources;but it provides better delay performance for data sources.

In strategy 2, the boundary is allowed to move according to figure shown above, thereby providing additional bandwidth to voice sources when available. Consequently if the voice sources hold up BBUs in the data traffic for a long time, the delay performance for the data traffic may be affected adversely.

In strategy 3, the boundary shifts dynamically throughout the frame, depending on the traffic load. Pre-empting may be permitted in this strategy for voice traffic. In strategy voice may be completed domination the channel, hence forth some researchers referred to **strategy 4** as shown in figure, where a minimum bandwidth allocation is always assured for both voice or data traffic. Such traffic is also known as **min-max scheduling** as there is a minimum and maximum bandwidth bound for either of traffic.

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LECTURE NOTES ON "DIGITAL SWITCHING AND TELECOM NETWORK"

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