Telecommunication (3-1-2)

Evaluation:

	Theory	Practical	Total
Sessional	30	20	50
Final	50	-	50
Total	80	20	100

Objective:
To provide students with broad knowledge of principles of transmission, switching, signaling and networking aspects of modern telecommunication systems.

1. Introduction **(4 hrs)**

1.1 Public-switched telephone network (PSTN)
1.2 Network topology
1.3 Central office switch
1.4 Subscriber telephone
1.5 Subscriber loop
1.6 Telephone conversation, hierarchical networks
1.7 Comparison between analog and digital transmission
1.8 Transmission impairments (distortion, noise, interference, crosstalk, echo, singing, jitter)

2. Multiplexing and multiple access techniques **(2hrs)**

2.1 Multiplexing and concentration
2.2 Space-Division Multiplexing (SDM)
2.3 Wavelength-Division Multiplexing (WDM)
2.4 Time-Division Multiple Access (TDMA)
2.5 Code- Division Multiple Access (CDMA)
2.6 Space- Division Multiple Access (SDMA)
2.7 ALOHA, slotted-ALOHA, CSMA/CD

3. Pulse code modulation (PCM) **(4hrs)**

3.1 PCM transmission format (T1, and E1 lines)
3.2 Frame and multiframe
3.3 Frame and multiframe alignment strategy
3.4 Higher order PCM
3.5 Plesiochronous Digital Hierarchy (PDH), Synchronous Digital Hierarchy (SDH) and SONET

4. Switching techniques and system **(6 hrs)**

4.1 Manual switching, circuit switching, packet switching & Message switching
4.2 Electro-mechanical switching (Strowger, step-by-step, crossbar)
4.3 Electronic switching, stored control program (Centralized SPC and Distributed SPC)
4.4 Space-division switching, time-division switching, space-time division switching
4.5 Multiple stage switching (Two-Stage Networks, Three-Stage Networks, n- Stage Networks)
4.6 Digital cross connect, private branch exchange

6. **Signaling in telephone networks** **(4hrs)**

6.1 Signaling system, types of signaling (in-channel signaling and common channel signaling)

6.2 CCITT Signaling System No. 7 (Block diagram, signaling units, comparison with OSI model)

6.3 Dual Tone Multi Frequency (DTMF) and pulse dialing.

6.4 Numbering Plan and Charging Plan

7. **Synchronization and network management** **(4hrs)**

7.1 Synchronization principle and mode of operation

7.2 Synchronizer circuits

7.3 Sampling time recovery & frame time recovery

7.4 Timing inaccuracies and elastic stores

7.5 Routing control and flow control in network management

7.6 Network management using SNMP

8. **Diversity techniques** **(2 hrs)**

8.1 Multipath propagation (line-of-sight LOS and non-LOS models)

8.2 Fading models (flat and frequency selective fading)

8.3 Diversity system (space, time, frequency, polarization, angle)

9. **Traffic theory** **(5hrs)**

9.1 Poisson process

9.2 Little's theorem

9.3 Characterization of queues

9.4 Measurement of telephone traffic, Blockage, Lost calls and Grade of Service

9.5 Smooth, Rough and Random Traffic

9.6 Loss systems, Lost call Cleared Models, Congestion, Erlang's B-calculation and Dimensioning

9.7 Queuing Theory

10. **Protocols in telecommunications** **(8 hrs)**

10.1 OSI model

10.2 X.25 packet switched networks

10.3 Frame relay

10.4 Integrated Services Digital Network, ISDN features, service and application, architecture and data rate, protocols like link access procedure for D channel, B channel data link protocol and ISDN layer three protocol, broadband-ISDN

10.5 ATM services and application, ATM network access, ATM header and payload, ATM signaling

11. **Next Generation Network (NGN)** **(6 hrs)**

11.1 Convergence of media

11.2 NGN definition, advantages, challenges and its scope

11.3 Transmission of real time traffic using packets, packet length, buffering, delay, jitter, Quality of Service (QoS)

11.4 Integrated Service Architecture, Differentiated Service Architecture

11.5 Multi-Protocol Label Switching(MPLS)
11.6 Resource Reservation Protocol
11.7 Real time transport protocol
11.8 Session Initiation Protocol
11.9 Megaco signaling protocol

Laboratory (Field Visit):
Visiting industries and preparation of report on various topics.

References:

1. J. Bellamy, Digital Telephony
2. B. Carson, Communication Systems.
3. W. Stallings, Data Communication and Computer Networks
4. J. E. Flood, Telecommunication switching, networks and traffic
5. T. Vishwanathan, Telecom technology

INTRODUCTION

Development of Telecommunication:

- **Early** means of communication over a distance included:

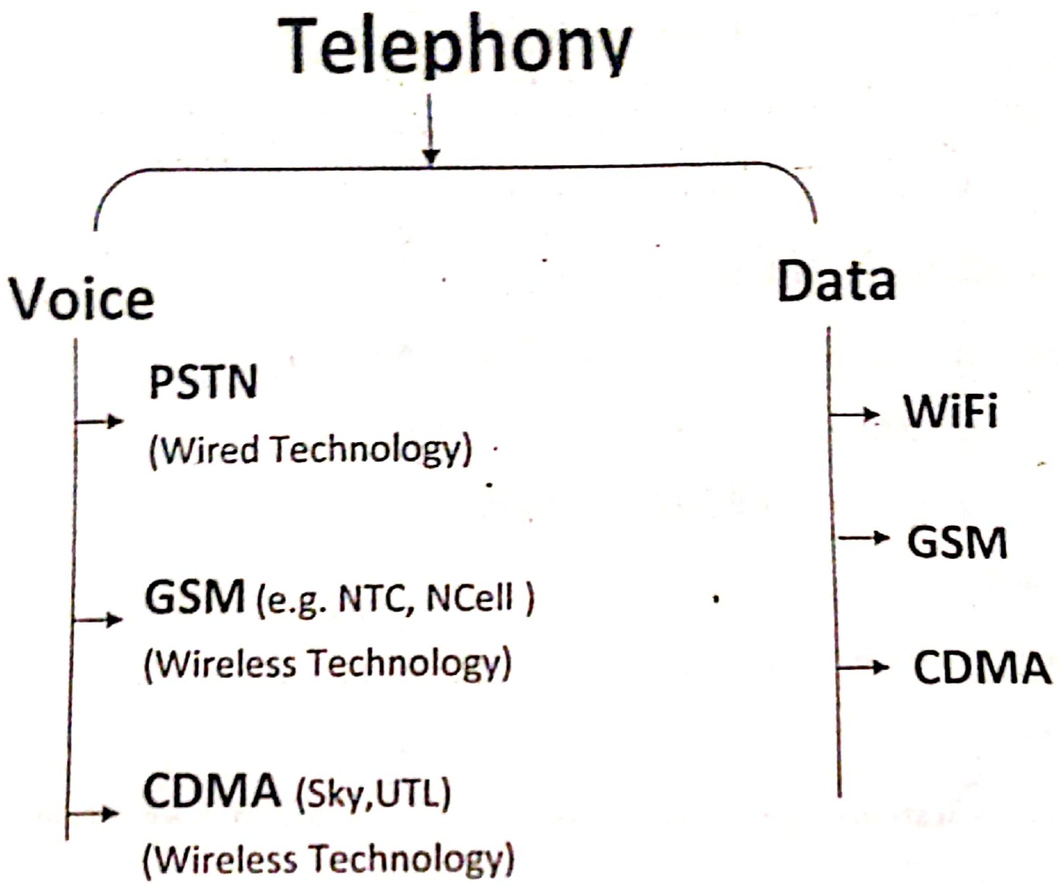
  (i.) **visual signals**

  - Beacons (for navigation, defense, vehicles),
  - smoke signals (war, alphabet)
  - Semaphore telegraphs (shutter/blade position in tower)
  - Signal flags (different shape, color code)
  - Heliograph (flashes of sunlight & mirror used)

  (ii.) **Audio signals**

  - Long-blown horns,
  - Loud whistles

- **Electrical communications** began with the invention of the **telegraph** independently by Wheatstone & Morse in 1837.
- Telegraph system consisted mainly of separate point-to-point lines, sending information in one direction across both continents and oceans almost instantly at a time, using number of wires
- Two-way communication for the first time demonstrated by Alexander Graham Bell, in March 1876.
- He demonstrated his telephone set and the possibility of telephony i.e. long distance voice transmission.
- He also gave the concept of '**central office**' or telephone exchange center for the first time.
- Modern mobile wireless concepts, generations, NGN, Apps (like SayHi, Big data (every data intelligent), google glass, etc.)



<u>Telecommunication:</u>

- Exchange of information between two entities over a significant distances by using electrical signals, either over a physical medium (such as signal cables) or in the form of electromagnetic waves.

<u>Telecommunication Network:</u>

- Can be defined as a set of devices (nodes), mechanisms and procedures by which the end user equipment in the network can exchange information meaningfully.
- E.gs. Computer networks, the internet, the telephone networks, etc.

Public Switched Telephone Network (PSTN)

- PSTN is the network of world's public circuit switched telephone networks.
- The term Public Switched Telephone Network (PSTN) describes the various equipment and interconnecting facilities that provide phone service to the public.
- Also referred to as POTS (Plain Old Telephone Service)
- It consists of :

  - Telephone lines,
  - Fiber-optic cables,
  - microwave transmission links,
  - Communication satellite, and
  - Undersea telephone cable

  Interconnected by switching centers.
- PSTN began in US in 1878 with a manual mechanical switchboard that connected different parties and allowed them to carry on conversation.
- Today, the PSTN is a network of computers and other electronic equipment that converts speech into digital data and provides a multitude of sophisticated phone features, data services and mobile wireless access.
- At the core of the PSTN are digital switches.
- The PSTN is well known for providing reliable communications to its subscribers.
- The phrase "five nines reliability," representing network availability of 99.999 percent for PSTN equipment, has become ubiquitous within the telecommunications industry.

Network Topology

- The topology of a network describes the various network nodes and how they interconnect(physically & logically both)
- Regulatory policies play a major role in exactly how voice network topologies are defined in each country.
- Depending on geographical region, PSTN nodes are sometimes referred to by different names. The three node types are:

End Office (EO)

- Also called as Local Exchange.
- The End Office provides network access for the subscriber.

- It is located at the bottom of the network hierarchy.

Tandem

- Connects EOs together, providing an aggregation point for traffic between them.
- In some cases, the Tandem node provides the EO access to the next hierarchical level of the network.

Transit

- Provides an interface to another hierarchical network level.
- Transit switches are generally used to aggregate traffic that is carried across long geographical distances.

➢ There are different types of Network topologies:

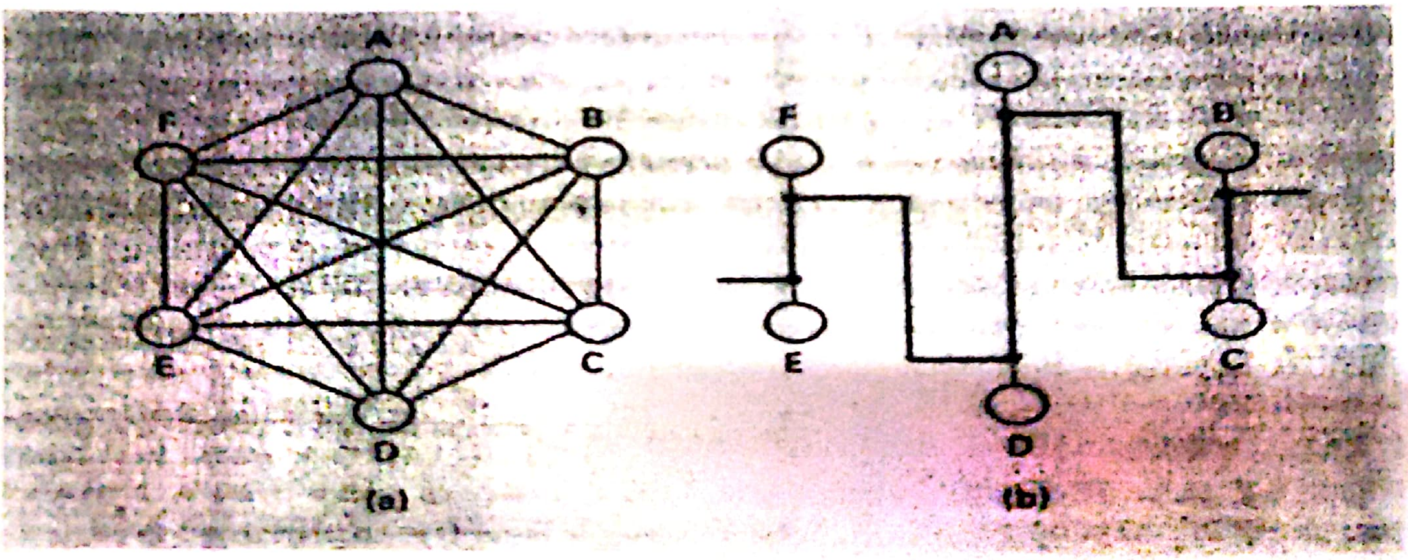
a) Mesh Topology

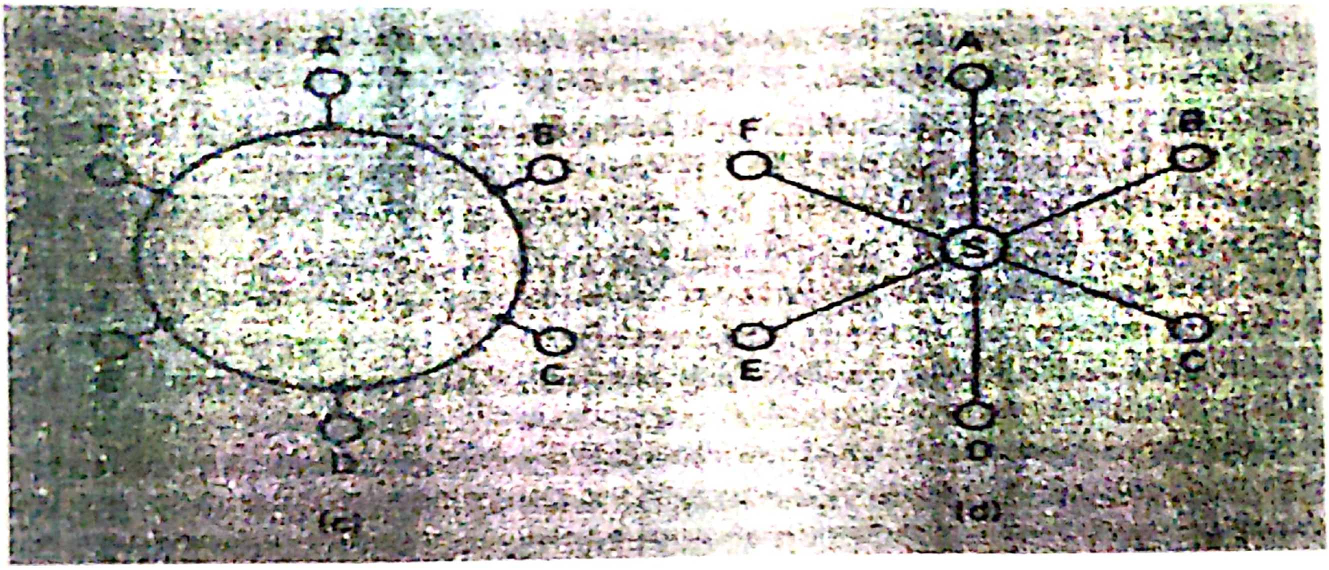
b) Bus Topology

c) Ring Topology

d) Star Topology

e) Hierarchical Tree Topology





➤ Although **Bus** and **Ring** networks, can be used for high speed data communication but are useless for normal telephony, since only one conversation at a time is only possible, so inefficient.

Mesh Topology:

- Mesh is a fully connected network.
- In this network, each users' station is connected to other by establishing a single separate line.
- So, each station needs lines to 'n-1' others, if communication is required between 'n' users' station
- Thus, total number of lines required is,

  $N = \frac{1}{2} * n * (n-1)$

  If $n >> 1$, then N is approximately proportional to n^2
- Practical if 'n' is small and lines are short.
- Used only when there is a heavy traffic among exchanges, as may happen in metropolitan area.

Disadvantages

- As 'n' increases and the lines become longer, the arrangement becomes much expensive.
- E.g. a system serving 10,000 users' station would need nearly 50 million lines

Star Topology:

- For telephony, two-way communication is required, on demand between any pair of stations and must be possible for many conversations to take place at the same time.
- These requirements can be met by providing a lines from each user's station to central switching center (e.g. telephone exchange) which connects the lines together as required.
- The number of lines is only, N = n.
- If N is large, the cost of providing the switching center is far more outweighed by the saving in-line costs.

Disadvantages

- As the area covered by star network and number of stations served by it grow, line costs increase.
- Services stop if the central switching exchange or hub damages, i.e. no alternate route

Hierarchical Tree Topology:

- Many star networks may be interconnected using additional tandem exchange, leading to two-level star network.
- An orderly construction of multilevel star network leads to hierarchical networks.
- Line cost decrease but cost of providing exchange increases.
- For large area, customers on each exchange wish to converse with customers on other exchanges
- So, **junction circuits** and thus junction networks required to provide connection between exchanges.
- Junction network has **mesh** configuration.
- As subscribers increase, cost of junction circuits increase and uneconomic to connect customer's local exchange. Cheaper to make connections via a central switching center, called a **tandem exchange.**
- In practice, the network of area is a mixture of a star network, joining all local exchanges to tandem exchange and mesh networks connecting some of the local exchanges together.

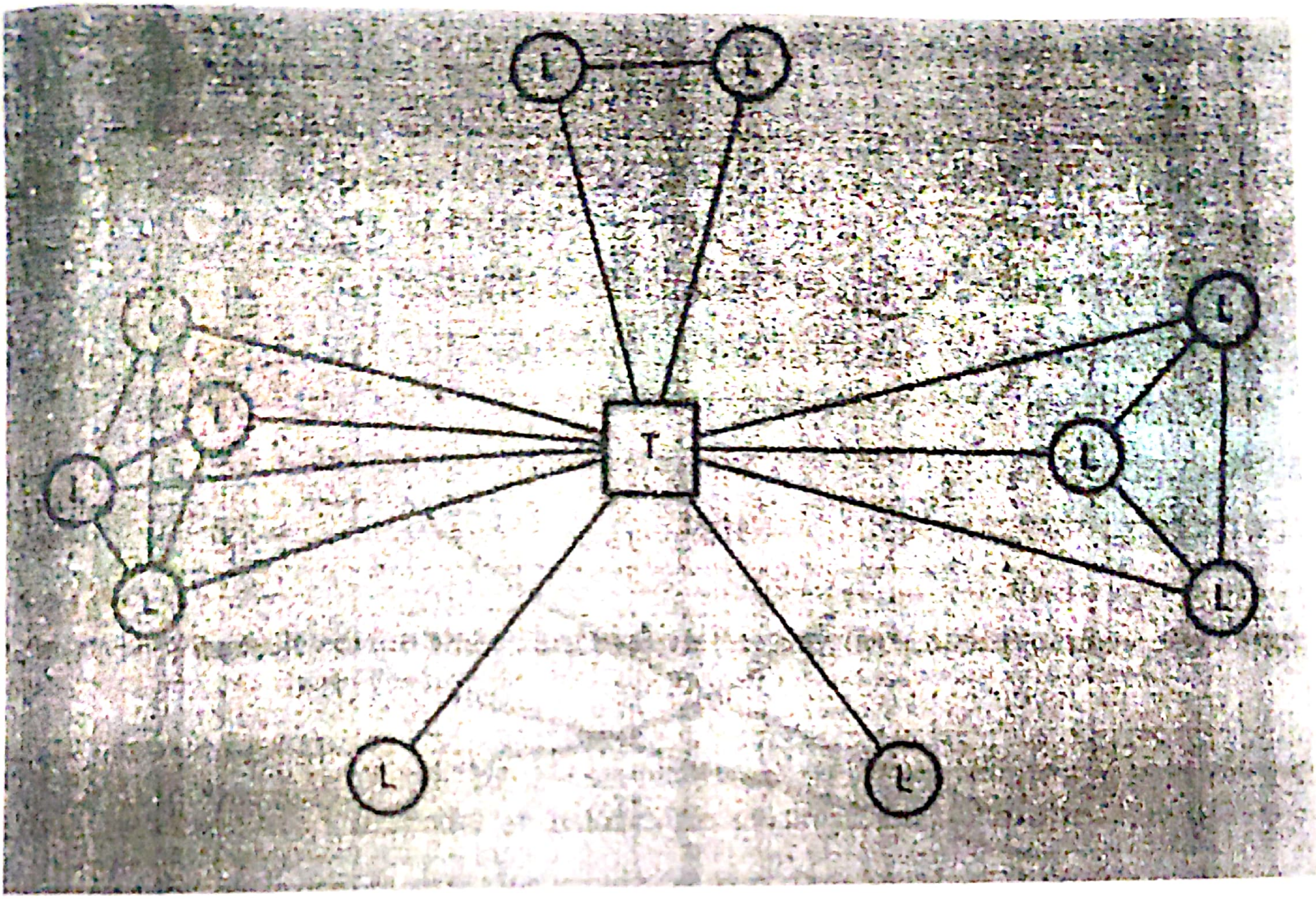


Fig. Multi-exchange area

L = Local Exchange

T = Tandem Exchange

PSTN Hierarchy:

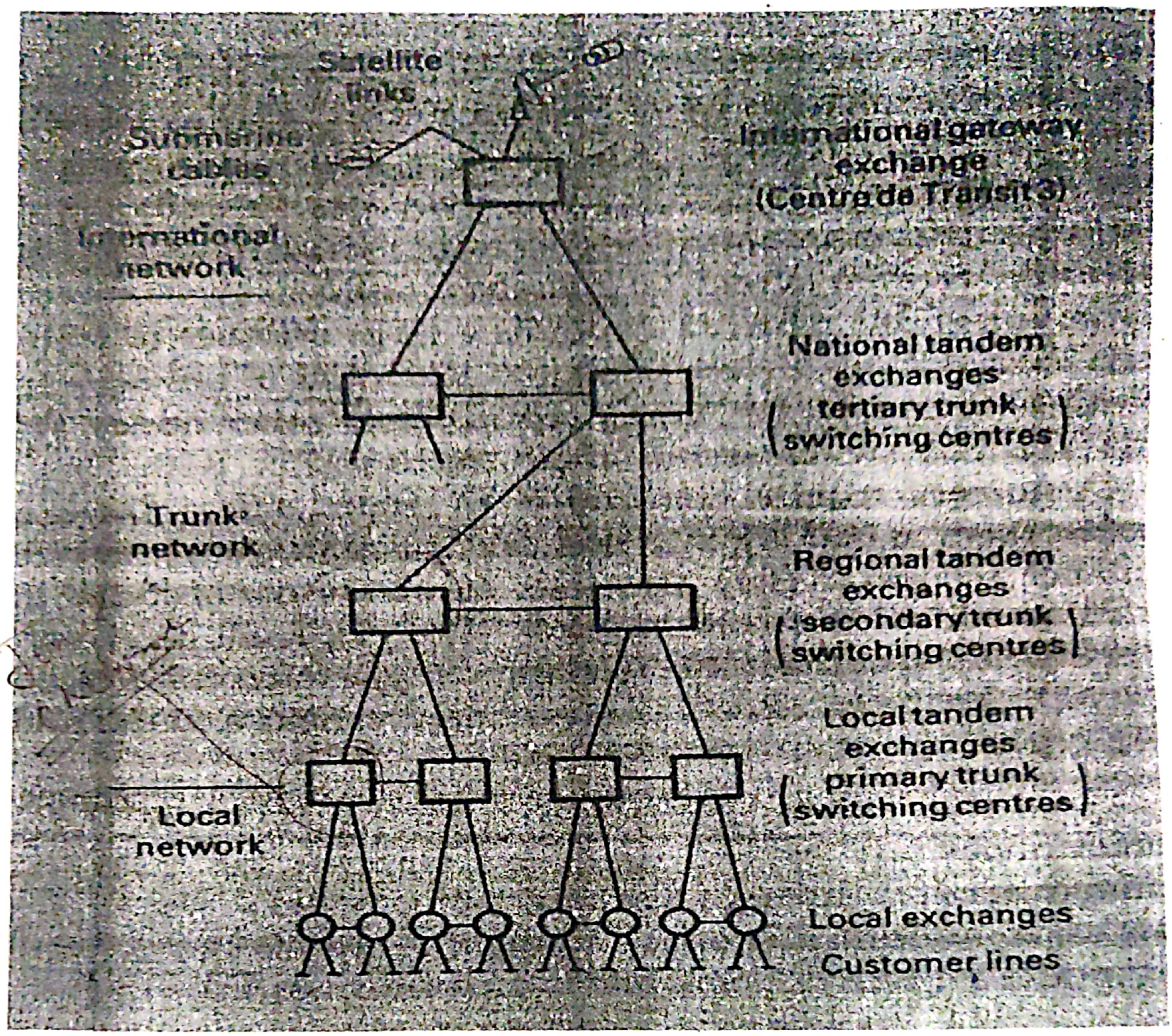


Fig. PSTN Hierarchy

PSTN Hierarchy Explanation:

✓ **Local Exchange (Class 5 office):**

- The exchange that is directly connected to the subscriber's telephone line or customer line, using the Subscriber loop is called local exchange.
- Lies at the bottom of PSTN Hierarchy.

- The main roles of Local exchange are

    - to provide PSTN network access to the subscriber (may be individual subscriber or corporate using Private Branch Exchange (PBX)),
    - to provide overall maintenance services,
    - to switch local calls,
    - To forward trunk calls to other successive exchanges in the hierarchy.
    - Send different signaling tones like- dial tone, busy tone (for local calls) to the subscriber.

- E.gs. Naxal Exchange, Gongabu Exchange, etc.
- The network which connects customers' stations to their local exchanges is called **local network**. This network is also called subscribers' distribution network or customer access network or the customer loop or local loop.

Tandem (Trunk) Exchange:

- The exchange which interconnects the local exchange of certain geographical region is called as tandem exchange.
- Tandem exchange is also called as backbone/core switch.
- Exchange where local exchanges are connected in tandem is known as Local Tandem Exchange.
- Similarly, regional tandem exchange interconnects the local tandem exchanges of certain local area and national tandem exchange interconnect the regional tandem exchanges of certain geographical regions and provide access to the international network to the subscriber.
- The networks which interconnect a group of local exchanges serving an area and a tandem or trunk exchange is called **junction network.**
- The **trunk network** or **toll network** provides long-distance circuits between local areas throughout the country.
- The totality of **junction network** and **trunk or toll network is called core network, the** inner core consisting of trunk network and the outer one consisting of the junction network.

International Gateway Exchange:

- The national network is connected to the international network by one or more international gateway exchanges.

- This exchange handles and switches the international calls of the subscriber.
- Provides access to satellite links or submarine cables as per the need of the subscriber call.
- The circuit that links the national networks of different countries is called **international network.**

- In the PSTN network of the kind as shown in fig. above, when there is a direct route between two exchanges at the same level, there is also a possible **alternative route** between them via an exchange at the next higher level.

- Thus, if the direct circuit is not available (e.g. because of cable breakdown), it is possible to divert traffic to the indirect route.

- In older switching systems, such changes must be made by manual arrangement. However, modern switching systems provide **Automatic Alternative Routing (AAR).**

- With AAR, if an originating exchange is unable to find a free circuit on the direct route, may be due to **cable breakdowns or traffic overloads,** it automatically routes the call through the higher level exchange.

Drawback:

- A strictly hierarchical network suffers from one serious drawback, i.e. its **poor fault tolerance** feature.

Central Office Switch:

- The central office is an office in a locality to which subscriber home and business lines are connected on a local loop.
- The central office (CO) houses the digital switching equipment that terminates subscribers' lines and trunks and switch calls.
- The switching center is called an 'exchange' in the UK and 'central office (CO)' in North America.
- Originally 'switch' referred to the manual switchboards, when call connections were manually created using cords to connect lines on a plug board.
- Electro-mechanical switches replaced manual switchboards, and those eventually evolved into the computer-driven digital switches of today's networks.
- 'DMS (Digital Multiplex System) - 100 digital switch' is one example of popular central office.

- Now, switching between calls is done electronically, under software control, and such systems are called as Stored Program Control (SPC) systems.

Terminology:

- Different names for the various networks and their switching centers are used not only in different languages but also in different English- speaking countries.
- For example, a switching center is called an '**exchange**' in the UK, but a '**central office**' in North America.
- The different names for Telecommunication terms in North American English and British English are as follows:

North America	British
Customer's loop	Local network / Access network
Central Office	Exchange
End Office / Class 5 office	Local exchange
Inter-office trunk	Junction
Junctor	Trunk
Toll office	Trunk exchange
Toll network	Trunk network

- Now, any telephone network may be viewed as consisting of the following major systems:

1. Subscriber end instruments or equipment,
2. Subscriber loop systems,
3. Switching Systems,
4. Transmission Systems, and
5. Signalling Systems

Subscriber Telephone:

- In a simplest form of a telephone circuit i.e. **simplex communication circuit,** there is a one way communication link involving two entities, one receiving (listening) and the other transmitting (talking).
- This form of one way communication is as shown in figure below:

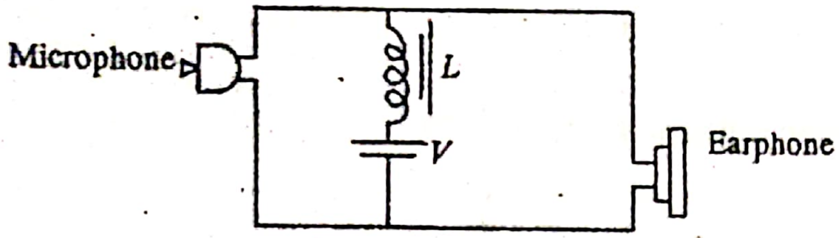


Fig. A simplex telephone circuit.

- Most commonly used microphone is a carbon microphone. Microphone & Earphone are transducer elements.

Microphone:

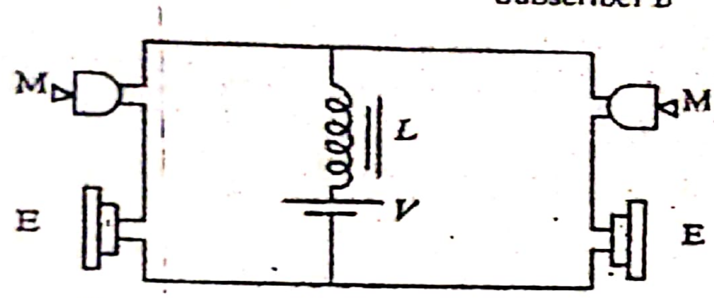
- In carbon microphones, a certain quantity of small carbon granules is placed in a box.
- Carbon granules conduct electricity and the resistance offered by them is dependent upon the density with which they are packed.
- One side of the box cover is flexible and is mechanically attached to a diaphragm.
- When sound wave impinge on the diaphragm, it vibrates, causing the carbon granules to compress or expand, thus changing the resistivity offered by the granules.
- If the voltage is applied to the microphone, the current in the circuit varies according to the vibrations of the diaphragm.

Earphone:

- Earphone is electromagnet with magnetic diaphragm, which is positioned such that there is an air gap between it and the poles of the electromagnet
- When the electromagnet is energized by passing a current, a force is exerted on the diaphragm.
- The voice frequency current from the microphone causes variation in the force exerted by the electromagnet, thus vibrating the diaphragm and producing sound waves.

- In a normal telephone communication system, information is transferred both ways.
- The form of communication in which information transfer takes place both ways but not simultaneously is known as, **half-duplex communication** and if information transfer takes place both directions simultaneously, then it is called **full-duplex communication.**

- Figure above can be modified to achieve half-duplex communication by the introduction of a transmitter and receiver at both ends of the circuit as shown in figure below:



E = earphone M = microphone

Fig. A half-duplex telephone circuit.

- In this circuit, the speech of A is heard by B as well as in A's own earphone. This audio signal heard at the generating end, is called **sidetone**.
- Human speech and hearing system is a feedback system in which the volume of speech is automatically adjusted, based on the sidetone heard by the ear. If no sidetone is heard, person tends to shout, and if too much of sidetone is present, there is a tendency to reduce the speech to a very low level.
- In above circuit for half-duplex communication, the entire speech intensity is heard as sidetone, which is not desirable.
- The circuit given below, gives a circuit where a small level of sidetone and full speech signal from the other party are coupled to the receiver.

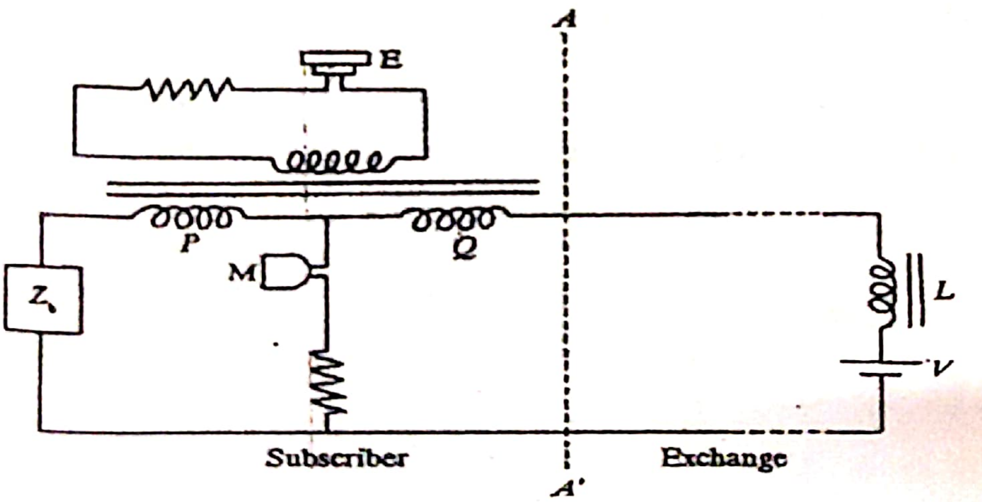


Fig. A telephone circuit with sidetone coupling.

- In above circuit, the impedance Z_b is chosen to be more or less equal to the impedance seen by the circuit to the right of section AA'
- As a consequence with proper sidetone coupling the speech signal from the microphone M divides more or less equally in the two windings P and Q.
- Since the signals in these two windings are in the opposite direction, only a small induced voltage appears in the receiver circuit providing the appropriate sidetone.

Subscriber loop :

- Every subscriber in a telephone network is connected generally to the nearest switching office by means of a dedicated pair of wires. Subscriber loop refers to these pair of wires.
- So, the dedicated pair of wires that connect the every subscriber in a telephone network to the nearest local exchange, are called as <u>subscriber loop</u>.

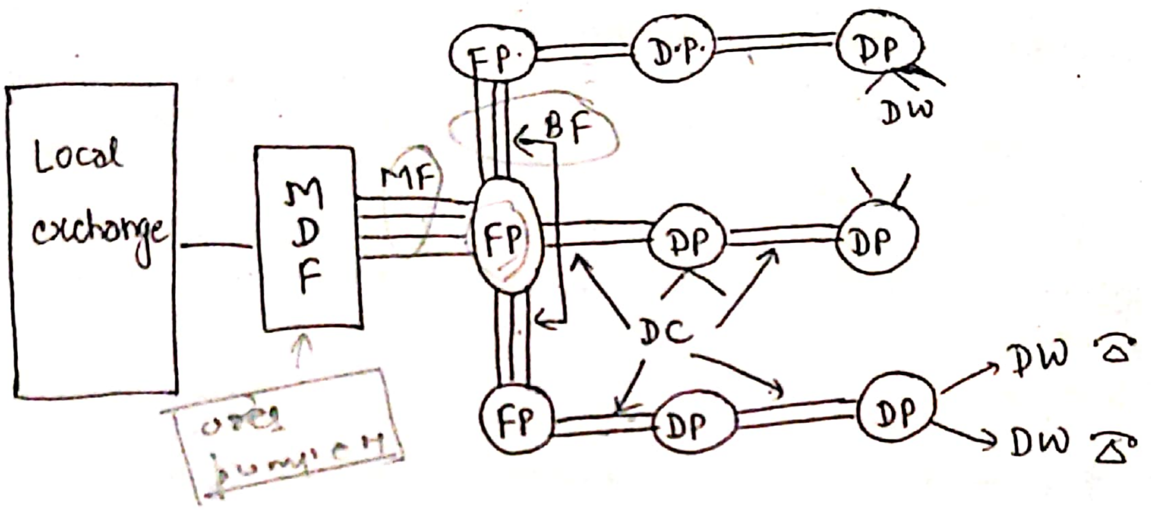
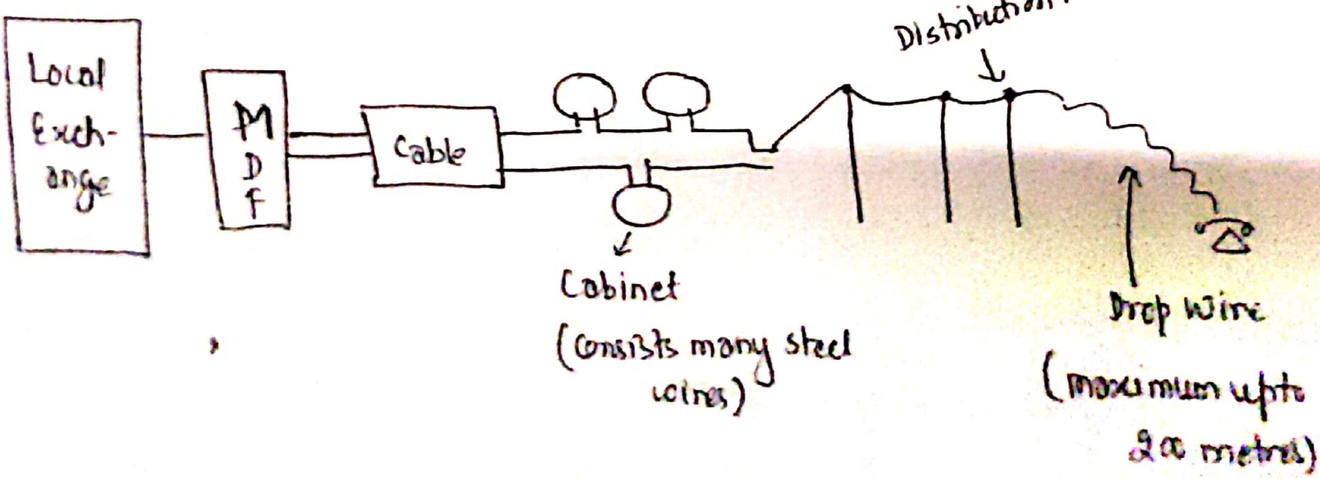


Fig.(1.4) Cable Hierarchy for Subscriber loop

where,

MDF = Main Distribution Frame.
MF = Main Feeder
BF = Branch Feeder
FP = Feeder Point

DP = Distribution Point
DC = Distribution Cable
DW = Drop Wires.



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End office or Class 5 office	Local exchange.
Inter-office trunk	Junction.
Junctor	trunk.
Toll office	Trunk exchange
Toll network	Trunk network.

→ It is far easier to lay cables containing a number of pair of wires for different geographical locations and run individual pairs as required by the subscriber premises.

→ Generally four levels of cabling are used as shown in fig. 1.4.

→ At the subscriber end, the drop wires (DW) are taken to a distribution point (DP).

→ The Drop Wires (DW) are the individual pairs that run into the subscriber premises.

→ At the distribution point, the drop wires are connected to wire pairs in the distribution cables (DC).

→ Many Distribution Cables (DC) from nearby geographical locations are terminated on a feeder Point (FP), where they are connected to Branch Feeder (BF) cables which, in turn, are connected to the main feeder cable.

→ The Main Feeder (MF) cables carry a larger number of wire pairs, typically 100 - 2000, than the distribution cables which carry 100 - 500 pairs.

→ The feeder cables are terminated on a Main Distribution Frame (MDF) at the local exchange.

→ The subscriber cable pairs emanating from the exchange are also terminated on the MDF.

→ Subscriber pairs and exchange pairs are interconnected at the MDF by means of jumpers.

Advantages of Jumper Settings:

① The jumper settings in MDF provides a flexible interconnection mechanism, which is very useful in reallocating cable pairs and subscriber numbers.

(Contd...)

→ For example, if a subscriber moves his house to a nearby ~~street~~ area served by the same exchange but a different distribution point, s/he can be permitted to retain the same telephone number by a suitable connection at the MDF. Similarly, a wire pair released by him/her can be given to a new number and assigned to another subscriber.

→ (II) In a newer installations, distribution and feeder points also have flexible crosspoint connection capability.
This enables a subscriber drop wire to be ~~easily~~ easily reconnected to any pair in the distribution cables, and similarly any pair from the distribution cable can be connected to any other pair in the feeder cable at the feeder point (FP).

(III) Above arrangement permits efficient utilization of the cable pairs as well as helps in cable management during faults. For example, if a particular cable is faulty, important subscribers assigned to this cable may be reassigned to free pairs in other cables.

Numbering System: (* details not required)

→ ITU-T Standard.
→ ITU-T divides the whole world into 9 zone codes.
→ A call is directed through the telephone network by telephone number

Types:

(I) National (STD) → Subscriber Trunk Dialing.

3 digits (more people can have same number)

Area/Trunk Code	Exchange Code	Link Code

eg. Ktm-01

$\underbrace{\text{Exchange Code} \quad \text{Link Code}}_{\text{Subscriber number.}}$

For example, if a subscriber moves his house to a nearby area served by the same exchange but a different distribution point, s/he can be permitted to retain the same telephone number by a suitable connection at the MDF. Similarly, a wire pair released by him/her can be given to a new number and assigned to another subscriber.

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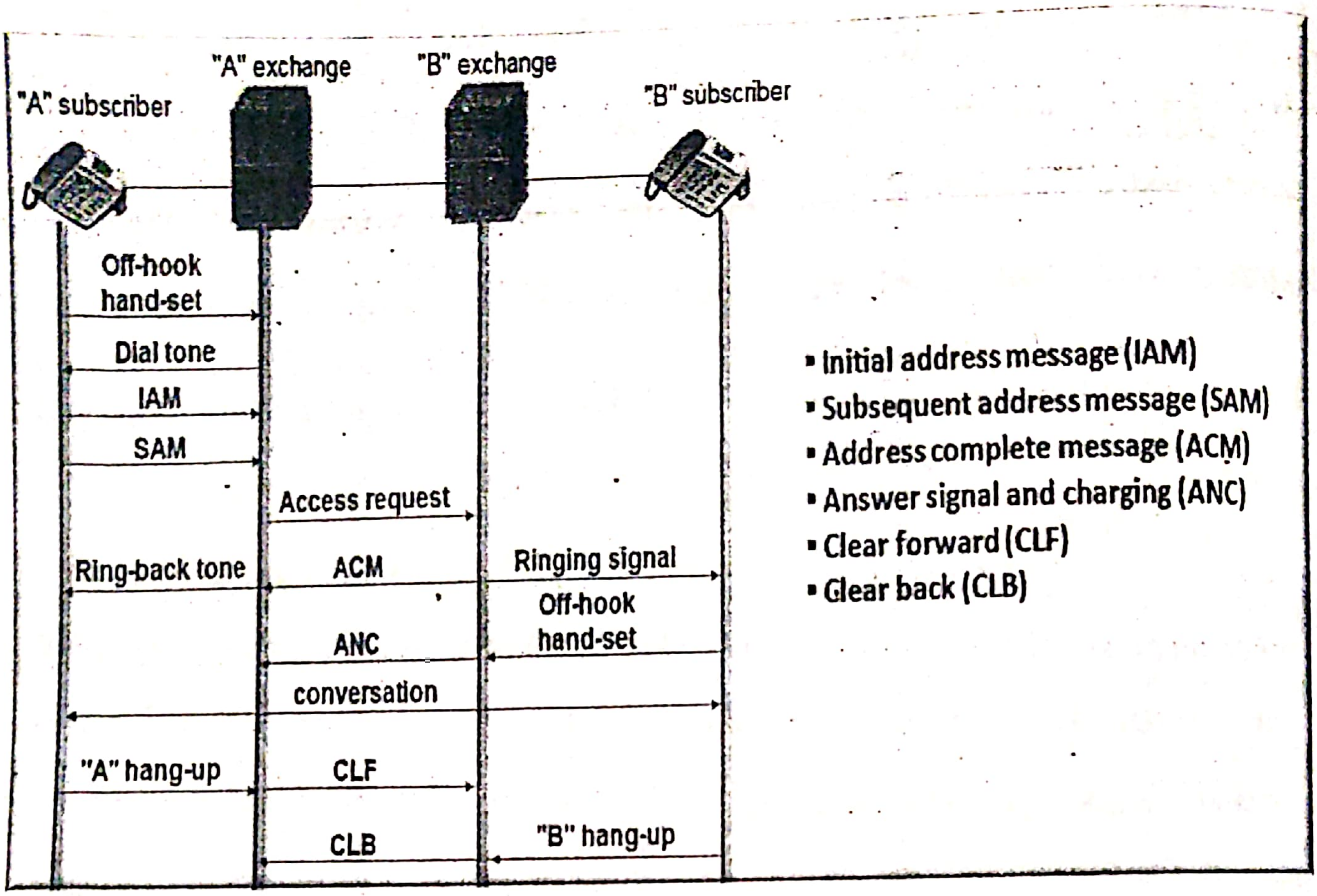
For example, if a particular cable is faulty, important subscribers assigned to this cable may be reassigned to free pairs in other cables.

Factors that limit the length of Subscriber loop:

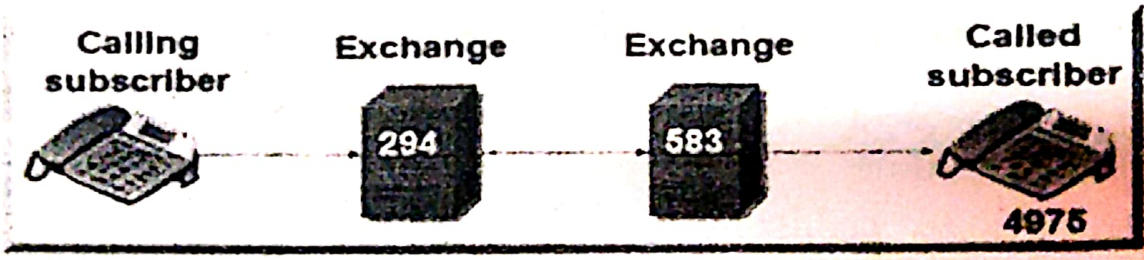
- Signalling limits.
- Attenuation limits (-5dB)

Telephone Conversation:



Call setup process

Example : Suppose the calling subscriber dialed "5834975"



- At first the exchange(294) which the calling subscriber is directly connected to, examines the dialed digits "583-4975"
- Secondly it acts upon the first three digits and access its look up table to rout the call to the "583" exchange
- Then the "583" exchange acts upon the information
- It identifies the dialed number and connects the correct subscriber loop which matches the "4975" number
- Then ring current is applied to the loop to alert the called subscriber and when the call is answered conversation begins

Transmission:

- The purpose of communication system is to transport an information-bearing signal from source to a user destination via a communication channel.
- Types of Communication System:–
  (I) Analog, (II) Digital.
- Analog and digital signals are used to transmit information usually through electric signals.
- An analog waveform (signal) is characterized by being continuously variable along amplitude and frequency
- Analog transmission could mean that the transmission is a transfer of an analog source signal which uses an analog modulation method (eg FM, AM)
- Digital transmission represents the information in two binary format (zero or one) usually by square wave where each bit is representative of two distinct amplitudes.

Advantages of Digital Transmission Over Analog:

- Increased immunity to channel noise & external interference
- Flexible operation of the system & ~~compute~~ compatibility & reliability.
- A common format for the transmission of different kinds of message signals (eg. voice, signals, video signals, data, etc)
- Improved security of communication through the use of encryption.
- Regeneration, error detection & correction, ease in hardware implementation, etc

Disadvantages of Digital Networks:

- Increased Bandwidth (T1 system requires 8 times more BW than analog)
- Need for time synchronization,
- Topologically restricted multiplexing
- Incompatibility with analog facilities.

Comparison of analog and digital transmission:

Feature	Analog transmission	Digital transmission
Signal and Waves	Analog signal is a continuous signal which represents physical measurements & are denoted by sine wave	Digital signals are discrete time signals generated by digital modulation & are denoted by square wave.
Representation	Uses continuous range of values to represent information eg Human voice in air, analog electronic devices, etc.	Uses discrete or discontinuous values to represent information eg. Computer CDs, DVDs etc
Traffic measurement	Hz (for example, a telephone channel is 4 KHz.)	Bits per second eg. a T-1 line carries 1.544 mbps & E1 transports at 2.048 mbps data rate
ndwidth	Low bandwidth (4 KHz) which means low data transmission rates (upto 33.6 Kbps) because of limited channel bandwidth	High bandwidth that can support high-speed data & emerging applications that involve video & multimedia
Network apacity	Low; one conversation per telephone channel	High; multiplexers enable multiple conversations to share a communications channel & hence to achieve greater transmission efficiencies.

Feature	Analog transmission	Digital transmission
Power requirement	High because the signal contains a wide range of frequencies and amplitudes.	Low because only two discrete signals - the one and zero - need to be transmitted
Security	Poor; when you tap into an analog circuit, you hear the voice stream in its native form, and it is difficult to detect an intrusion.	Good; encryption can be used.
Error rates	High; 10^{-5} bits (i.e. 1 in 100,000 bits) is guaranteed to have an error.	Low; with twisted-pa 10^{-7} (i.e. 1 in 10 millio bits per second) will have an error, with sate. Lite, 10^{-9} (i.e 1 in 1 billi on per second) will have an error, and with fiber 10^{-11} (i.e only 1 in 10 trillion bits per second) will have an error.

→ Communication media is defined as anything that can carry information from a source to destination.

Communication media
- Guided (wired)
  - Twisted Pair
  - Co-axial cable
  - optical fibre
- Unguided (wireless)
  - Radio wave
  - Micro wave
  - Infrared wave

Guided Media:

→ Physical media is present during transmission.

① Twisted Pair Cable:

→ It consists of 2 conductors wrapped each with its own plastic insulation.

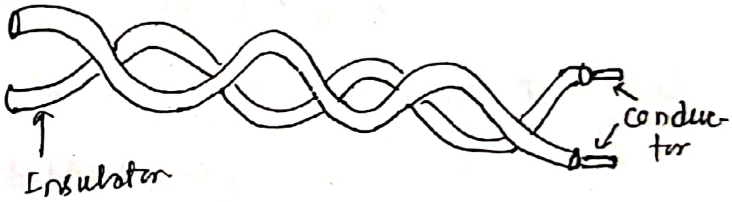
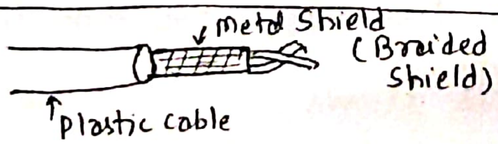
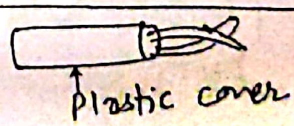


fig Twisted pair Cable

→ One of the wire is used to carry signals to the receiver and the other for ground reference.

STP (Shielded Twisted Pair)	UTP (Unshielded Twisted Pair)
	
→ More expensive	→ Less expensive
→ Bulkier	→ Simple
→ metal casing improves the quality of cable.	→ Quality not good as STP.
→ Hardly used.	→ Mostly used.

Note:

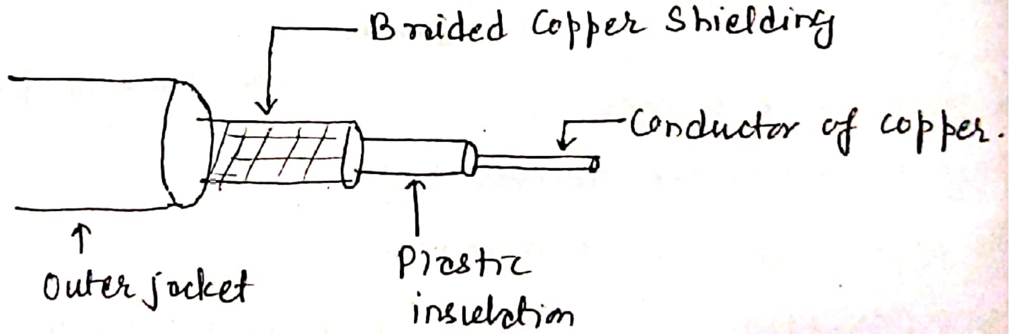
→ STP provides better noise protection than UTP Cabling.

→ For many years, STP was the cabling structure specified for use in Token Ring network installations. With the use of Token Ring declining, its demand has decreased

→ UTP consists of 4 pairs of color-coded wires that have been twisted together, external electromagnetic field create the same interference in each wire.

→ UTP cabling, commonly used in LAN cabling environments. eg Category 5 (Cat 5) Cable is used commonly in 100BASE-TX Fast Ethernet Installations.
Other categories include Enhanced Category 5 (Cat 5e) cable and Category 6 (Cat 6).

Co-axial Cable:



→ It carries signal of higher frequency range than in twisted pair.

→ Instead of 2 wire, it has a single central wire.
(As all the layers share the same axis, this construction is called co axial or coax for short.)

→ Much higher bandwidth, data rate higher, so the signal weaken rapidly & requires the use of repeaters
(at every 5 km).

→ Attenuation higher

eg. RG-59 → Used in Cable TV.

© **Optical Fibre** : (*Construction, modes & functions read self)

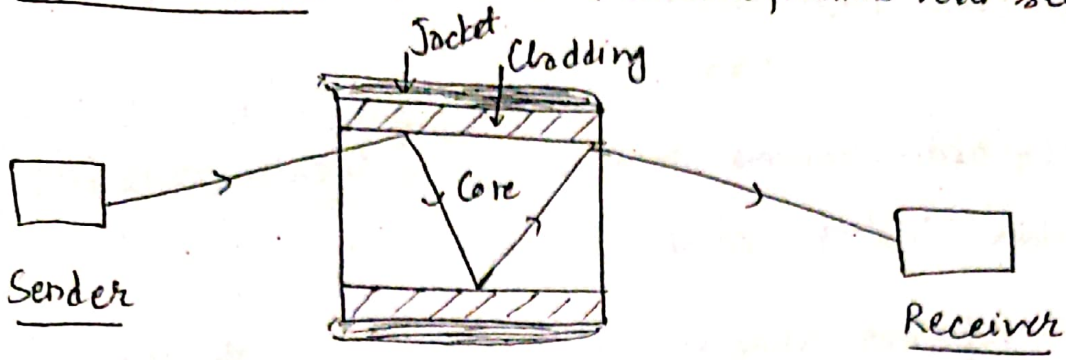


fig Optical Fibre

→ Made up of glass (for ~~short~~ long distance) or plastic (short distance) and transmit signals in the form of light.

Advantages :

(i) Higher Bandwidth and hence higher data rate. A single optical fiber operating at 1300 or 1550 nm of wavelength has a potential bandwidth of 20THz $(20 \times 10^{12}$ Hz), which is enough for 312 million 64-kbps channels.

(ii) Less signal attenuation as a signal can run for 50km without requiring repeaters. (while 5km for co-axial and twisted pair cables).

(iii) Much lighter (small size and weight) than copper wire.

(iv) Electromagnetic noise can't affect it and optical fibres have electrical isolation and immunity to interference.

(V) Greater immunity tapping (lightening also doesn't affect it).

(VI) Resistance to corrosive materials as it is made up of glass.

(VII) Highly secured communication media as lights from optical fiber doesn't radiate significantly.

(VIII) Greater repeater spacing as there is much less transmission loss, with losses as low as 0.2dB per km.

Disadvantages:

(I) Its intallation and maintenance requires expertise that is not yet ~~a~~ cheaply available everywhere

(II) There may be losses from joints.

Unguided Media

→ The unguided media ~~donot use a~~ is that media for ~~messa~~ signal ~~data~~ transmission through air, wave etc, without using a physical conductor.

(I) Radiowaves:

→ Electromagnetic waves ranging in frequencies between 3 KHz – 1 GHz, are called radiowaves.

→ Omnidirectional ie propagated in all direction.

→ Can travel long distance, so used in broadcasting such as AM radio.

→ Used in TV, cordless phones.

Microwaves:

- Electromagnetic waves with frequency 1 GHz - 300 GHz
- Unidirectional, when an antenna transmits microwaves they can be narrowly focussed.
- microwave propagation is a line-of-sight (LOS) propagation and can't overcome Earth's and other obstacles.

Applications:

↳ Cellular phones
↳ Satellite network

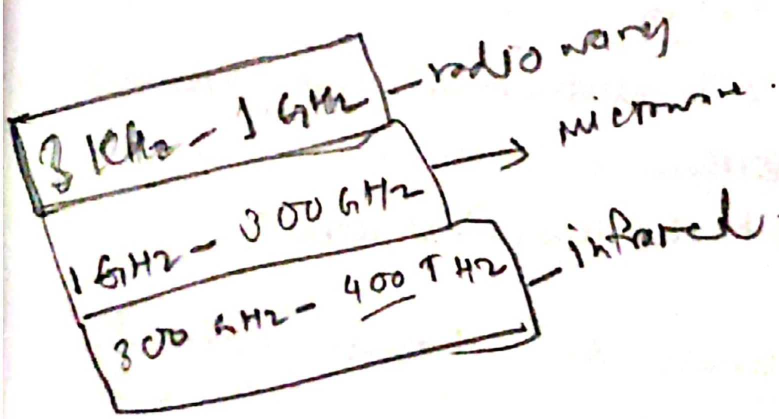
(III) Infrared waves:

- frequency range is 300GHz - 400THz
- can be used for short range communication.
- can't penetrate walls, this property prevents interference between one system and the other.

Applications:

TV remote

* IR band has an excellent potential for data transmission with a very high data rate.



Transmission Impairments:

Q. Discuss the transmission impairments in telecommunication.

Q. How crosstalk, echo and singing occurs in communication system?

Q. Write about talker's echo and listener's echo? Which is more harmful?

Ans?

→ One of the most difficult aspects of designing an analog telephone network is determining how to allocate transmission impairments to individual subsystems within the network.

The various transmission impairments are as follows:-

① Distortion:

→ If the output signal or the ~~transmitted~~ received signal is not exact replica of the input signal or the transmitted signal, then the signal is said to be distorted.

→ There are many different types and sources of distortion within the telephone network.

→ Some distortions arise from the nonlinearities in the network such as carbon microphones, saturating voice-frequency amplifiers, and unmatched companders.

→ Other distortions are linear in nature and are usually characterized in the frequency domain as either amplitude distortion or phase distortion.

→ Amplitude distortion refers to attenuating some frequencies in the voice spectrum more than others.

→ Phase distortion is related to the delay characteristics of the transmission medium.

→ Any deviation from a linear phase characteristics is referred to as phase distortion.

→ In contrast to noise and interference, distortion is deterministic; it is repeated every time the same signal is sent through the same path in the network.

→ Thus, distortions can be controlled or compensated for once the nature of the distortion is understood.

→ Harmonic distortions add overtones that are whole number multiples of a sound wave's frequencies.

→ Non-flat frequency response is a form of distortion that occurs when different frequencies are amplified by different amounts caused by filters.

→ Phase distortion mostly occurs due to the reactive component such as, capacitive, ~~reactive~~ reactance and inductive reactance. Here, all the components of the input signal are not amplified with the same phase shift, hence making some parts of output signal ~~of~~ out of phase with the rest of the output.

→ As the system output is given by, $y(t) = F(x(t))$, then if the inverse function F^{-1} can be found, and used intentionally to distort either the input or output of the system, then the distortion is corrected.

Noise:-

→ Noise is defined in electrical terms as any unwanted introduction of energy tending to interfere with the proper reception.

Types:

(a) External noise:

→ The various forms of noise created outside the receiver.
eg: Atmospheric noise, extraterrestrial noise, industrial noise, etc.

(b) Internal Noise:

- Noise created by any of the active or passive devices found in the receiver. eg: thermal noise (noise created by resistance or resistive components), shot noise (caused by random variations in arrival of electrons (or holes) at output electrode of an amplifying device), transit-time noise (If electron takes significant time to travel from emitter to collector), flicker noise (at low audio frequencies, a poorly understood form of noise), etc

(3) Interference:

→ Arises from unwanted coupling from other signals in the network.
→ It occurs most often in the communication system whose receiving antenna usually intercepts several signals at the same time.
→ The different types of interferences that may occur in communication are as follow :-
  - Electromagnetic interference (EMI),
  - Co-channel interference (CCI),
  - Adjacent-channel interference (ACI),
  - Intersymbol Interference (ISI)
  - Inter-Carrier Interference (ICI).

→ Interference in communication can be reduced by filtering.

(4) Crosstalk:

→ It is any phenomenon by which a signal transmitted by one circuit or channel of a transmission system, ~~creates~~ creates an undesired effect in another circuit or channel.
→ Crosstalk is usually caused by undesired capacitive, inductive or conductive coupling from one circuit, part of a circuit or channel to another.

- In telecommunication, crosstalk is often distinguishable as pieces of speech or signaling tones leaking from other people's connections.
- A common example of crosstalk is hearing pieces of other people's conversations on a telephone, or picking up part of a broadcast from a different radio station when listening to a radio show.

Sources of Crosstalk:

- Coupling between wires in a cable.
- Inadequate filtering.
- Effects of non-linear components on FDM signals.
- Improper shielding.

- 2 basic forms of ~~cabling~~ crosstalk are as follows:-

① NEXT (Near End Crosstalk):

- It refers to coupling from a transmitter into a receiver at a common location.
- NEXT is more troublesome because of a large difference in power levels between the transmitted and received signals.

② FEXT (Far End Crosstalk):

- It refers to unwanted coupling into received signal from a transmitter at a distant location.

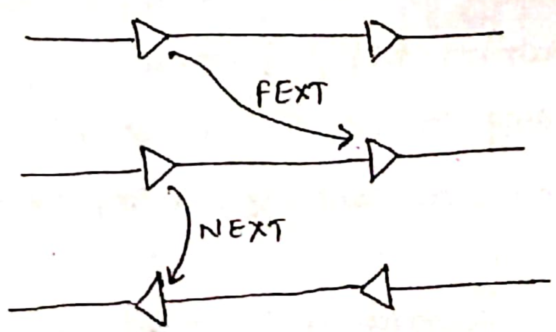


Fig. Near-end and far-end crosstalk.

(5) Echoes and Singing:

- Echoes and singing both occur as a result of transmitted signals being coupled into a return path and fed back to the respective sources.
- 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 interfaces.
- Due to long distances involved, the bearer circuits need amplifiers or repeaters at appropriate intervals to boost the signals.
- The amplifiers are almost invariably one-way devices and cannot handle bidirectional signals.
- Since the telephone conversation calls for signal transmission, both ways, long distance trunks require separate circuits for each direction leading to 4-wire circuits.
- Hence, the need for 2-wire to 4-wire conversion in long distance connections. The conversion is done by hybrids.
- An important function of hybrid is to ensure that the received signal is not coupled.
- The coupling is zero only when the 2-wire circuit and 4-wire circuit impedances are perfectly matched.
- While it is relatively easy to control the impedances of the trunk circuits, the subscriber loop impedances vary from subscriber to subscriber, depending on the distance at which the subscriber is located from the exchange, so impedance mismatch occurs.
- The effect of such a mismatch is to reflect a part of incoming speech signal on to the outgoing circuit, which returns to the speaker as echo.

→ If only one reflection occurs, the situation is referred to as 'talkers echo'.

→ If a second reflection occurs, "listener echo" results.

→ When the returning signal is repeatedly coupled back to the forward path to produce oscillations, then it is called "singing".

→ Singing occurs if the loop gain at some frequency is greater than unity.

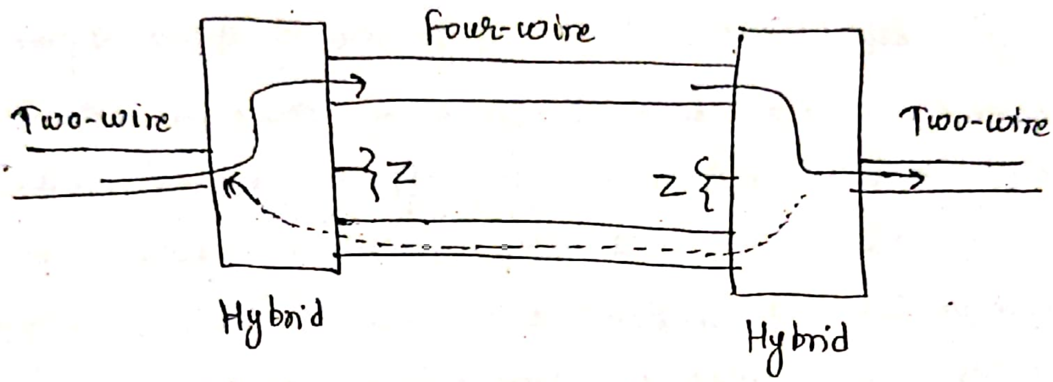


Fig Generation of echoes at two-wire to four-wire interface

Q. Explain different types of devices used to control echo.

Ans:

① Echo suppressor:

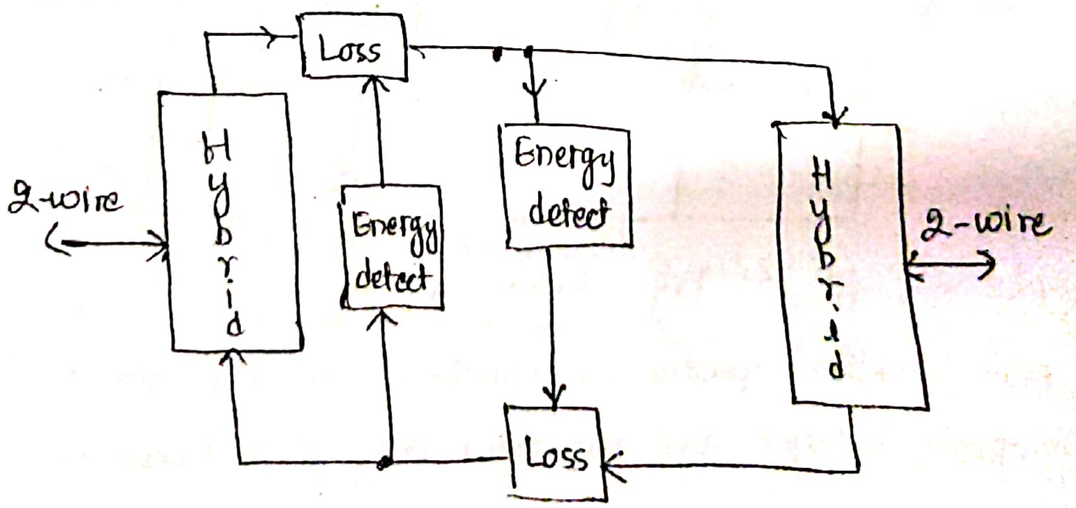


Fig Echo suppressor.

→ An echo suppressor operates on four-wire circuits by measuring the speech power in each leg and inserting a large amount of loss (35 dB) typically in the opposite leg when the power level exceeds a threshold.

→ Thus, a returning echo is essentially blocked by the high level of attenuation.

Drawbacks:

→ Echo suppressor converts a full-duplex circuit into a half-duplex circuit.

→ They might ~~stop~~ clip the beginning portions of speech segments.

→ With several milliseconds of interruption while an echo suppressor in one direction is turned off and the one in other direction is turned on, it is very difficult to organize data transmission. So, echo suppressors are usually disabled while the circuits are used for data transmissions.

Echo Canceller:

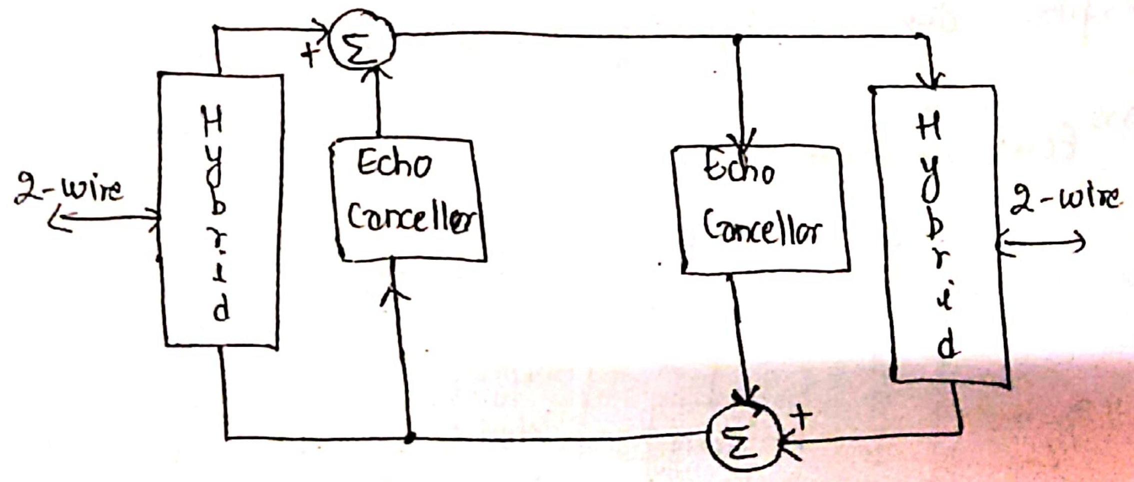


Fig Echo Canceller.

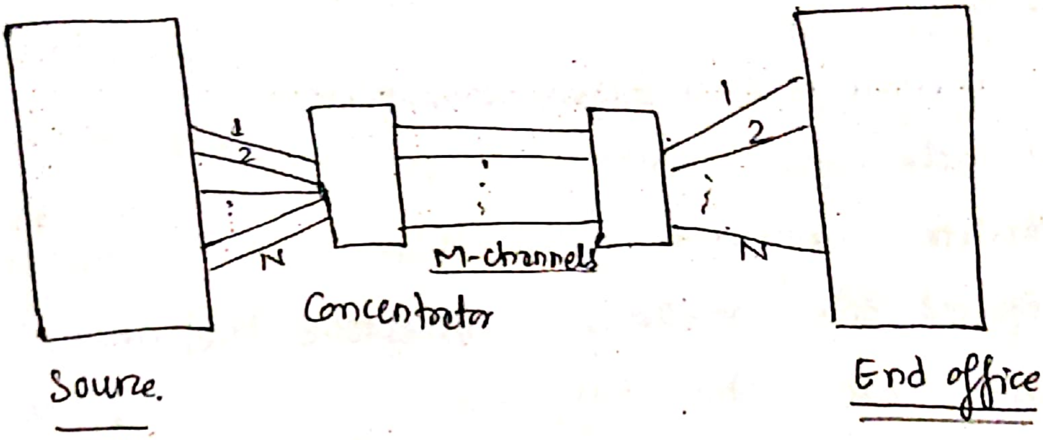
→ An echo canceller operates by simulating the echo path to subtract a properly delayed and attenuated copy of a transmitted signal from the receive signal to remove (cancel) echo components.

P.T.O →

Chapter- 2

Multiplexing and Concentration:

Concentration:



→ multiple sources are switched by concentrating the active source to a smaller number of shared output lines.

→ In data transmission, a functional unit that permits a common path to handle more data sources than there are channels currently available with the path, is called concentrator.

→ A concentrator usually provides communication capability between many low-speed, usually asynchronous channels and one or more high speed, usually synchronous channels.

→ Concentration introduces some amount of blocking and requires transfer of information between concentrator & expander

→ At the other end, expansion occurs by switching from shared lines to individual lines.

→ Thus, Echo Cancellation requires ~~characteristics of the circuit~~ training to determine how much delay and attenuation are needed to simulate the echo characteristics of the circuit.

Advantages:

→ It maintains a full duplex communication.
→ Eliminate Speech Clipping.
→ Satellite circuits with greater than 500ms of roundtrip delay required echo cancellers for acceptable performance.
→ Better than echo suppressor.

6 Jitter:

→ Jitter is the undesired deviation from true periodicity of an assumed periodic signal, in relation to a reference clock source.
→ Jitter may be observed in characteristics such as the frequency of successive pulses, the signal amplitude or phase of periodic signals.
→ Jitter is a significant, and usually undesired, factor in the design of almost all communication links (eg USB, SATA etc)
→ In digital transmission, it is possible to use regenerative repeaters instead of analog amplifiers.
→ A regenerative repeaters samples the received waveform at intervals corresponding to the digit rate.
→ If the received voltage at the sampling instant exceeds a threshold voltage, this triggers a pulse generator which sends a pulse to the next section of line.
→ The instants at which pulses are retransmitted by regenerative repeaters are determined by local oscillator synchronized to the digit rate, which must be extracted from received waveform.
→ Variations in the extracted frequency can cause a periodic variations of times of the regenerated pulses, is known as ~~jitter~~ jitter

Multiplexing

→ multiplexing (also known as muxing) is a method by which multiple analog message signals or digital data streams are combined into one signal over a shared medium.

→ The aim is to share an expensive resource

→ For example, in telecommunications, several telephone calls may be carried using one wire.

→ Thus, whenever the Bandwidth of a medium linking two devices, is greater than Bandwidth needs of the devices, the links can be shared.

→ multiplexing is the technique that allows the simultaneous transmission of multiple signals, across a single data link.

Types:

(I) Frequency Division multiplexing (FDM),

(II) Time Division multiplexing (TDM),

(III) wavelength Division multiplexing (WDM), and

(IV) Space Division multiplexing (SDM).

(I) Frequency Division multiplexing (FDM):

→ ~~an~~ FDM system, divides the available Bandwidth into number of subchannels.

→ Individual signals are inserted into the subchannels.

→ In FDM, signals generated by each sending device modulate different carrier frequency.

→ These modulated signals are then combined into a single composite signal that can be transported by a channel.

→ Thus, in FDM transmission, a number of baseband channels are sent over a common wideband transmission path by using each channel to modulate a different carrier frequency.

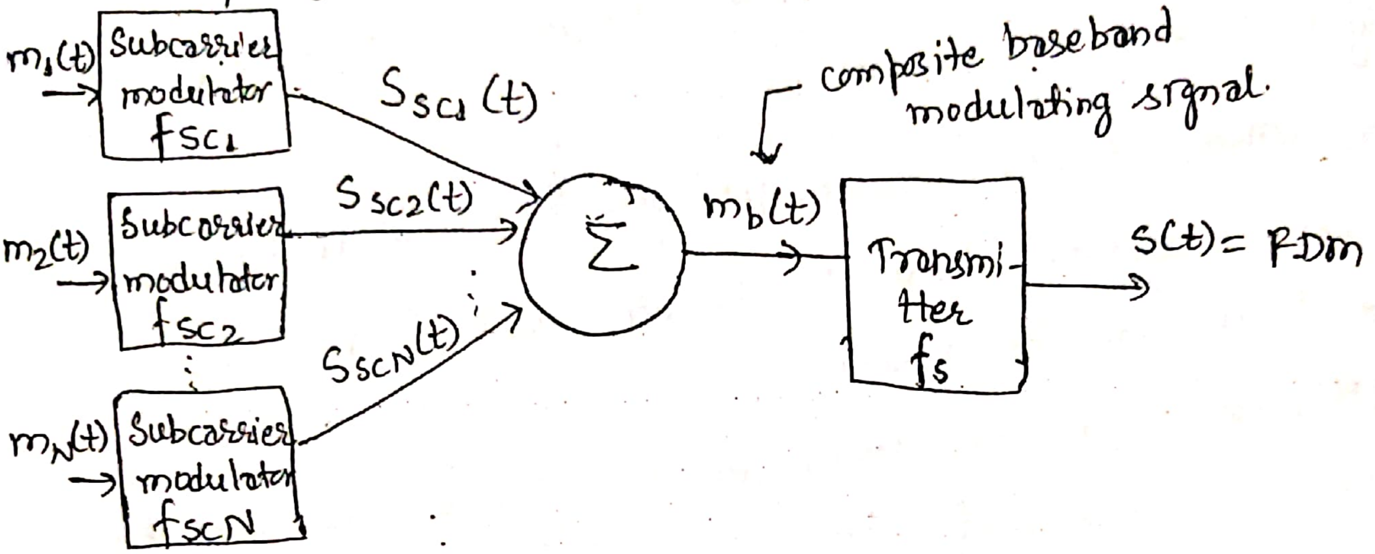


Fig (a) Transmitter of a FDM System

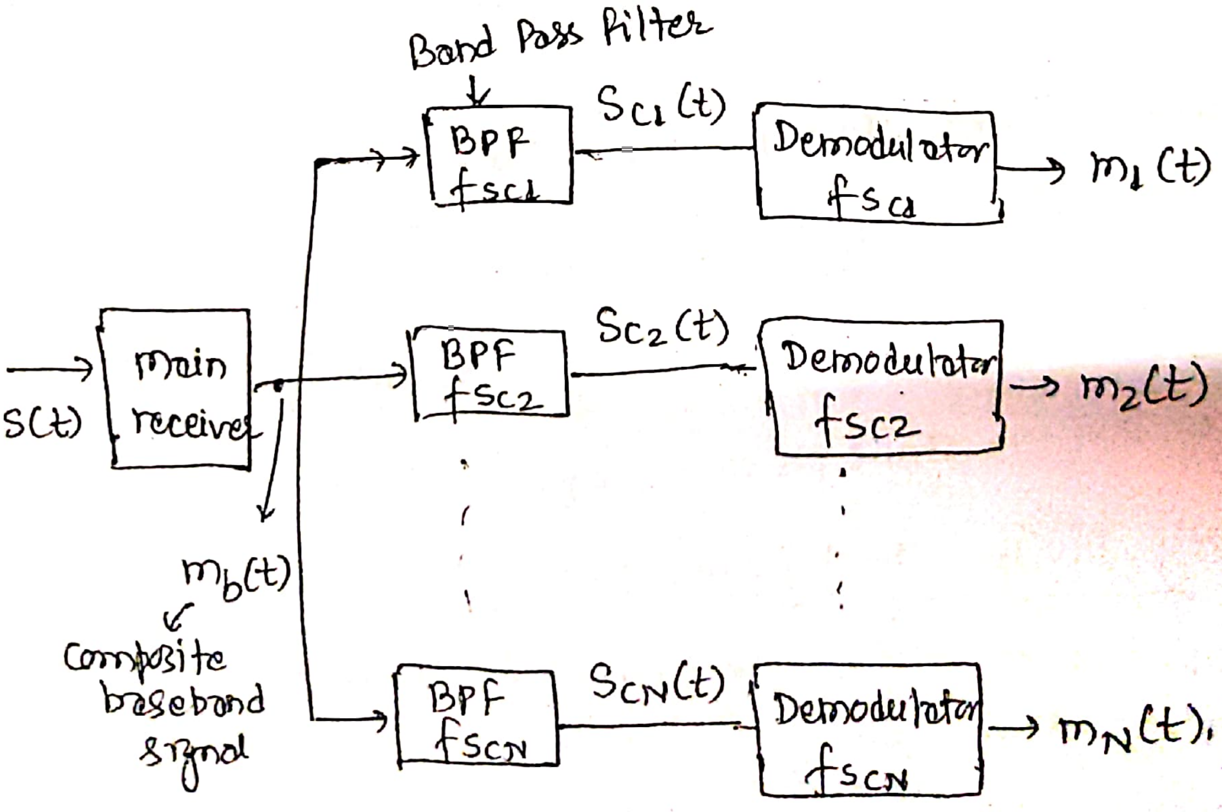


Fig B(b) Receiver of a FDM signal

→ Above fig (a) and (b). shows the transmitter and receiver of TDM system.

→ At the sending end, each incoming baseband signal from an audio frequency circuit is applied to a balanced modulator supplied with the appropriate carrier (f_c)

→ The output of this modulator is a double-sideband suppressed-carrier signal

→ This signal is applied to a bandpass filter which suppresses the upper sideband.

→ The outputs of these filters are commoned to give a composite output signal containing the signal of each telephone channel translated to a different portion of the frequency spectrum.

→ At the receiving end, the incoming signal is applied to a bank of band-pass filters (BPFs), each of which selects the frequency band containing the signal of one channel.

→ This signal is applied to a modulator supplied with the appropriate carrier (f_c) and the output of this modulator consists of baseband signal and unwanted high-frequency components.

→ The unwanted components are suppressed by low-pass filter and the baseband signal is transmitted to the audio-frequency circuit at the ~~curren~~ correct level by means of an amplifier.

• FDM in Telephony:

→ FDM provides a way of keeping a number of individual telephone signals separate while transmitting them simultaneously over a common transmission line circuit.

→ Each baseband telephone signal is modulated onto a separate sub-carrier, and all the upper or all the lower side bands, are combined to form the frequency-multiplexed signal

→ To facilitate interconnection among the different telecommunication system in use worldwide, CCITT has recommended a standard modulation plan for FDM.

→ The standard group in the plan consists of 12 voice channels.

→ The carrier spacing is 4 kHz, thus 12 channels occupy the band from 60 to 108 kHz.

→ Each channel has a baseband from 300 Hz to 3.4 kHz

→ The frequency guard band between adjacent channels is only 900 Hz, so crystal filters are used to obtain the necessary sharp transitions between pass and stop bands.

→ FDM is used in AM & FM radio & 1st generation cellular telephone

~~for a.~~

Time-Division multiplexing (TDM):

→ TDM involves sharing a transmission media by establishing a sequence of time slots during which individual source can transmit signals.

→ Thus, the entire Bandwidth of the facility is periodically available to each source for a restricted time interval.

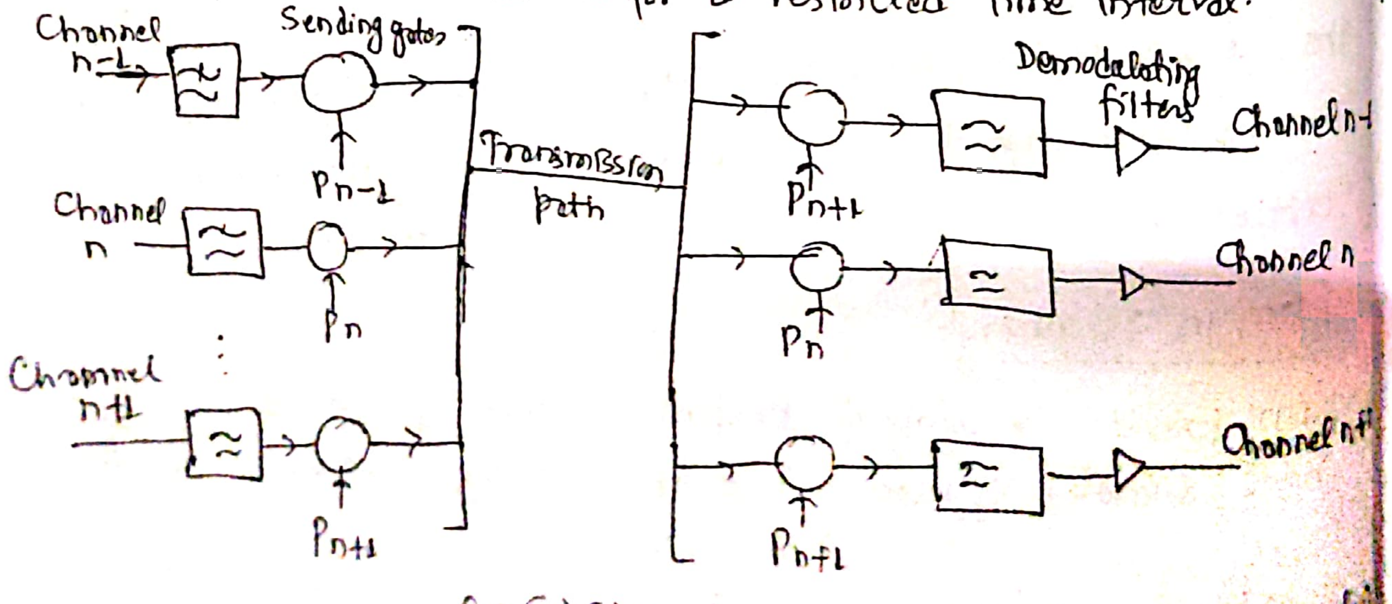


Fig (a) Elementary TDM system (one direction only)

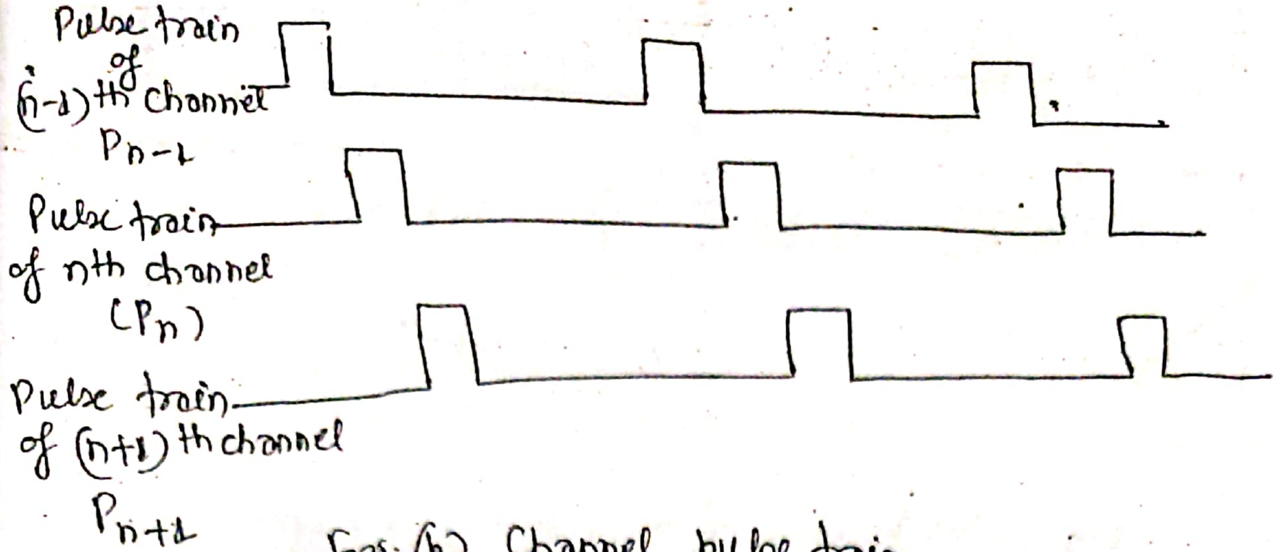


Fig. (b) Channel pulse train.

→ In TDM, each baseband channel is connected to the transmission path by a sampling gate which is opened for short intervals by means of a train of pulses.

→ In this way, samples of baseband signal are sent at regular intervals by means of amplitude-modulated pulses.

→ Thus, the common transmission path receives interleaved trains of pulses modulated by the signals of different channels.

Interleaving:

→ TDM can be visualized as the two fast moving switches at mux and demux side

→ The switches are synchronized and rotate at the same speed but in opposite direction.

→ On mux side, as switch opens in front of connection that connection has the opportunity to send a unit onto the path. This process is called interleaving

→ On demux side, the connection has opportunity to receive a unit

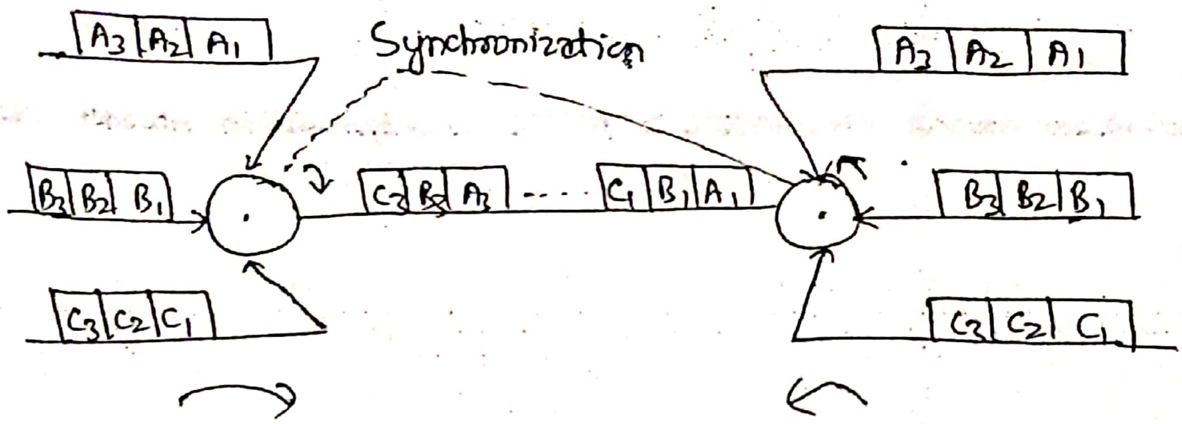


Fig Interleaving of message bits through TDM System.

Wavelength Division multiplexing (WDM):

- Same as FDM except that the multiplexing and demultiplexing involve optical signals.
- WDM is designed to use the high data rate capability of fiber optic cable.
- Using a fiber optic cable for one single line wastes the available Bandwidth.

multiplexing allows ~~several~~ to combine several lines to one.

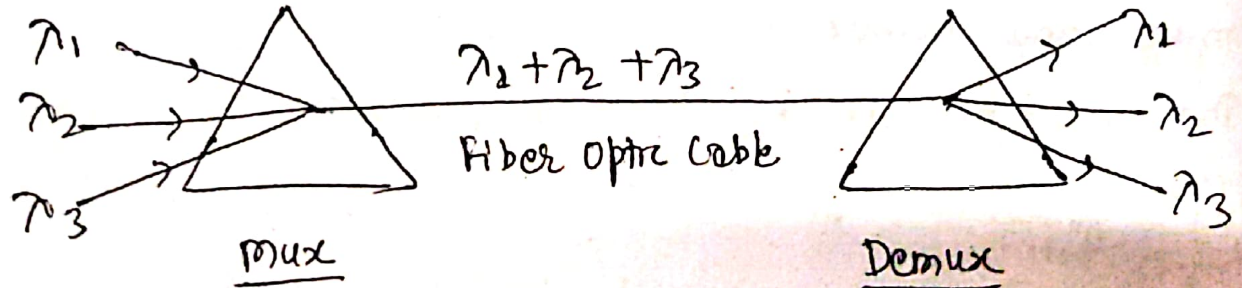


Fig WDM

- The combining and splitting of light sources are easily handled by a Prism.
- Prism bends a beam of light based on angle of incidence and frequency.

→ Using this technique, a 'mux' can be made to combine several input beams of light based on angle of incidence and frequency to give one output ~~of~~ beam of wider band of frequency.

→ Demux operates reversely.

Space Division multiplexing (SDM):

→ The introduction of cable systems into the transmission plant to increase the circuit packing density of open wire is one instance of multiplexing in the telephone network. This form of multiplexing is known as Space Division multiplexing.

→ It involves nothing more than building more than one pair of wires into a single cable.

→ SDM is the provision of multiple fixed bandwidth channels by multiple physical paths (i.e pairs of wires or optical fibres)

→ A good example of SDM is the use of a 25-pair cable to carry the conversations of 25 individual users from the customer's premises to the local telephone company's central office location.

Differences between TDM and FDM:

FDM	TDM
(i) The signals which are to be multiplexed are added in the time domain. But they occupy different slots in the frequency domain.	(i) The signals which are to be multiplexed can occupy the entire bandwidth in the time domain.
(ii) FDM is usually preferred for the analog signals.	(ii) TDM is preferred for digital signals.
(iii) Synchronization is not required	(iii) Synchronization is required
(iv) The FDM requires a complex circuitry at Transmitter & Receiver.	(iv) TDM circuitry is not very complex
(v) FDM suffers from the problem of crosstalk due to imperfect Band Pass Filter (BPF)	(v) In TDM the problem of crosstalk is not severe
(vi) Due to slow narrowband fading the transmission channel may be affected in FDM.	(vi) Due to slow narrowband fading all the TDM channels may get wiped out

Differences between 'multiplexing' and 'multiple Access Techniques':

multiplexing	multiple Access
(I) In telecommunications and computer networks, multiplexing is a process where multiple analog message signals or digital data streams are combined into one signal over a shared medium.	(I) multiple access techniques are used in telecommunications and computer networks, to share the available bandwidth of a channel.
(II) A device that performs the multiplexing is called a multiplexer (MUX), and a device that performs the reverse process is called demultiplexer (DEMUX)	(II) A channel-access scheme is also ~~scheme~~ based on multiple access ~~control~~ protocol and control mechanism also known as media access control (MAC). This protocols deals with issues such as addressing, assigning multiplex channels to different users, and avoiding collisions.
(III) It works on physical layer of OSI model	(III) It works on the Data Link layer of OSI model.
(IV) Classification of multiplexing (w.r. to channelization methods): - Time Division multiplexing (TDM), - Frequency Division multiplexing (FDM) - Wavelength Division multiplexing (WDM).	(IV) Classification of multiple Access (w.r. to channelization methods): - Time Division multiple Access (TDMA), - Frequency Division " " (FDMA), - Wavelength Division " " (WDMA),

Multiple Access Techniques:

→ Multiple access schemes are used to allow many mobile users to share simultaneously a finite amount of radio spectrum.

→ The sharing of spectrum is required to achieve high capacity by simultaneously allocating the available bandwidth (or the available amount of channels) to multiple users.

- ## Duplexing:

→ In wireless communication systems, it is often desirable to allow the subscriber to send simultaneously information to the base station while receiving information from the base station.

For example, in conventional telephone systems, it is possible to talk, and listen simultaneously, and this effect is called duplexing.

→ Duplexing may be done using frequency or time domain techniques. So, types of duplexing are as follows :-

① Frequency Division Duplexing (FDD):

→ It provides two distinct band of frequencies for every user.

→ The forward band provides traffic from the base station to the mobile, and the reverse band provides traffic from the mobile to the base station.

→ In FDD, any duplex channel actually consists of two simplex channels (a forward and reverse), and a device called duplexer is used inside each subscriber unit and base station to allow simultaneous bidirectional radio transmission and reception for both the subscriber unit and the base station on the duplex channel pair.

(ii) Time Division Duplexing (TDD):

→ It uses time instead of frequency to provide both a forward and reverse link.

→ In TDD, multiple users share a single radio channel by taking turns in the time domain.

→ Individual users are allowed to access the channel in assigned time slots, and each duplex channel has both a forward time slot and a reverse time slot to facilitate bidirectional communication.

→ Because of the rigid timing required for time slotting, TDD generally is limited to cordless phone or short range portable access.

* FDD is geared toward radio communications systems that allocate individual radio frequencies for each user.

Multiple Access:

→ multiple access techniques are used to share the available bandwidth in a wireless communication system.

→ There are different access techniques. Among them, the major multiple access techniques are:-

1. Frequency Division multiple Access (FDMA),
2. Time Division multiple Access (TDMA),
3. Code Division multiple Access (CDMA), and
4. Space Division multiple Access (SDMA).

→ These multiple access techniques can be grouped as narrowband and wideband systems, depending upon how the available bandwidth is allocated to the users.

Narrowband Systems:

→ In narrowband multiple access system, the available radio spectrum is divided into a large number of narrowband channels.

→ The channels are usually operated using FDD.

→ The term narrowband is used to relate the bandwidth of a single channel to the expected coherence bandwidth of the channel.

→ FDMA/FDD or TDMA/TDD access systems are ~~used~~ described as narrowband systems.

Wideband Systems:

→ In wideband systems, the transmission bandwidth of a single channel is much larger than the coherence bandwidth of the channel.

→ In wideband multiple access systems, a large number of transmitters are allowed to transmit on the same channel.

→ TDMA allocates time slots to the many transmitters on the same channel and allows only one transmitter to access the channel at any instant of time, whereas spread spectrum CDMA allows all of the transmitters to access the channel at the same time.

Frequency Division Multiple Access (FDMA):

- Frequency Division Multiple Access (FDMA) assigns individual channels to individual users.
- These channels are assigned on demand to users who request service
- During the period of the call, no other user can share the same channel.

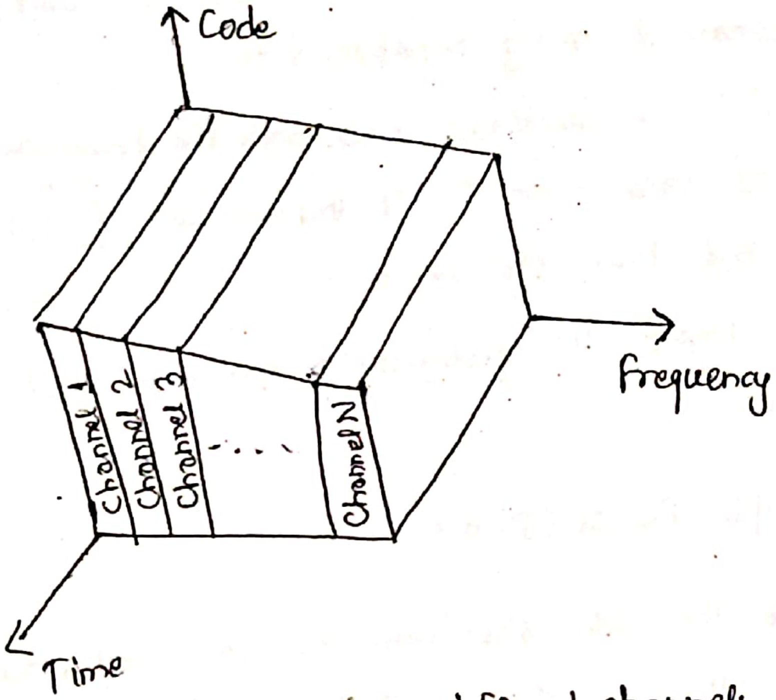


Fig FDMA where different channels are assigned different frequency bands.

The features of FDMA are as follows:

(i) The FDMA channel carries only one phone circuit at a time.

(ii) If an FDMA channel is not in use, then it sits idle and cannot be used by other users to increase or share capacity. It is essentially a wasted resource

(iii) The bandwidths of FDMA channels are relatively narrow (30 kHz in AMPS) and FDMA is usually employed in narrow-band systems.

(iv) As the amount of intersymbol interference is low in narrowband so, little or no equilization is required in FDMA narrowband systems.

(v) The complexity of FDMA mobile systems is ~~lay~~ lower as compared to TDMA systems.

(vi) Since, FDMA is a continuous transmission scheme, fewer bits are needed for overhead purposes (such as synchronization & framing bits) as compared to TDMA.

(vii) FDMA systems have higher cell site system costs as compared to TDMA systems, because of costly bandpass filters.

(viii) FDMA mobile unit uses duplexers, since both the transmitter and receiver operate at same time. So, it increases cost of FDMA subscriber unit and base stations.

(ix) FDMA requires tight RF filtering to minimize adjacent channel interference.

Time Division Multiple Access (TDMA):

→ TDMA systems divide the radio spectrum into time slots, and in each slot only one user is allowed to either transmit or receive.

→ Each user occupies a cyclically repeating time slot, so a channel may be thought of as a particular time slot that reoccurs every frame, where N time slots comprise a frame

→ TDMA systems transmit data in buffer-and-burst method, thus the transmission for any user is noncontinuous.

→ This implies that unlike in FDMA systems which accommodate analog FM, digital data and digital modulation must be used with TDMA.

→ The transmission from various users is interlaced into a repeating frame structure.

→ It can be seen that the frame consists of a number of slots,

→ Each frame is made up of preamble, an information message, and tail bits as shown in fig below:

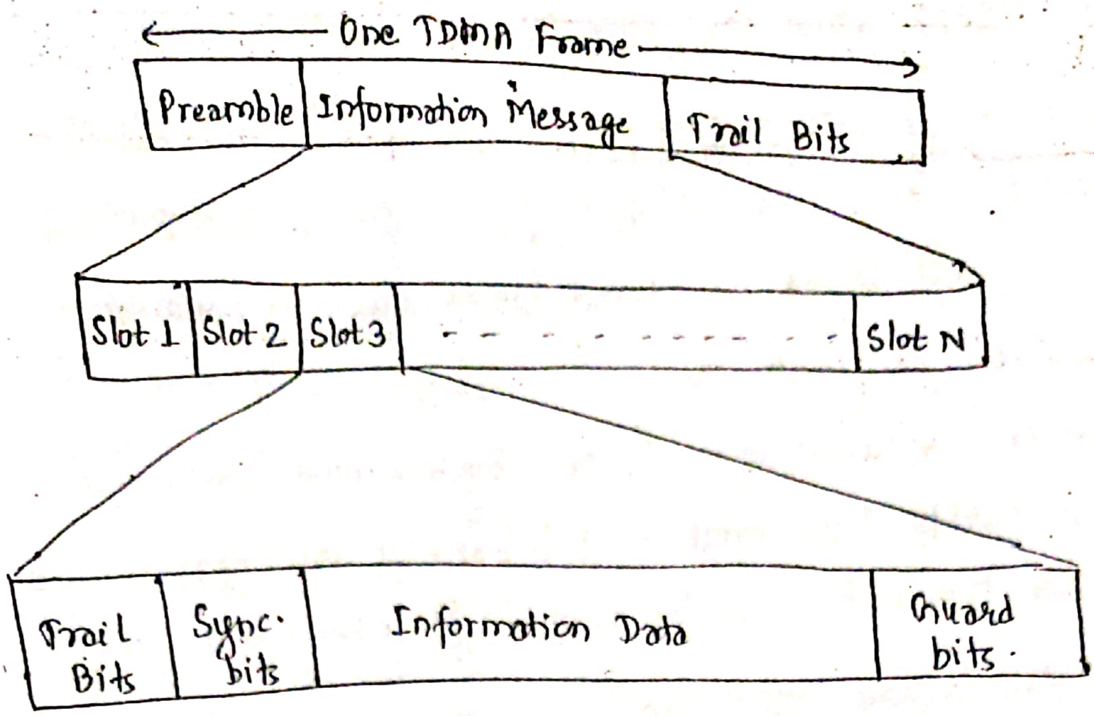


Fig: TDMA Frame Structure. The frame is cyclically repeated over time

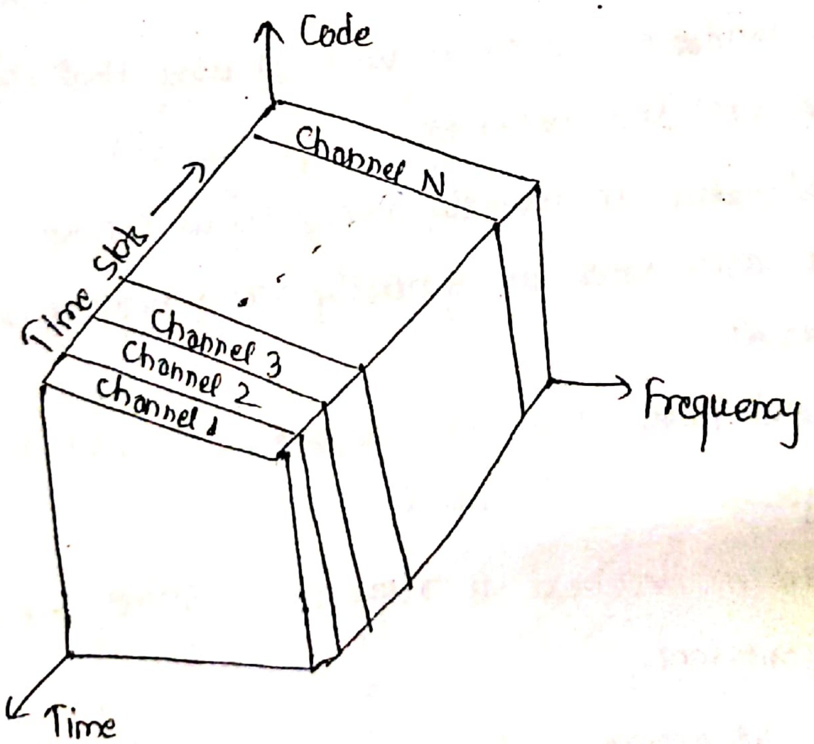


Fig. TDMA Schemes where each channel occupies a cyclically repeating time slot

The features of TDMA include the following:-

(I) TDMA shares a single carrier frequency with several users, where each user makes use of nonoverlapping time slots. The number of time slots per frame depends on several factors, such as modulation technique, available bandwidth, etc

(II) Data transmission for users of a TDMA system is not continuous, but occurs in bursts. This results in low battery consumption, since the subscriber transmitter can be turned off when not in use

(III) Because of discontinuous transmissions in TDMA, the handoff process is much simpler for subscriber unit,

(IV) TDMA uses different time slots for transmission and reception, thus duplexers are not required. Even if FDD is used, a switch rather than duplexer inside the subscriber unit is used that switches between transmitter and receiver

(V) Adaptive equalization is usually necessary in TDMA systems, since the transmission rates are generally very high as compared to FDMA channels.

(VI) In TDMA guard times (time slot separation) should be minimized, to avoid adjacent channel interference

(VII) High synchronization overhead is required in TDMA systems because of burst transmissions,

(VIII) TDMA has an advantage in that it is possible to allocate different numbers of time slots per frame to different users. Thus, bandwidth can be supplied on demand to different users by concatenating or reassigning time slots based on priority

Code Division Multiple Access (CDMA):

- In CDMA, the narrowband message signal is multiplied by a very large bandwidth signal, called the spreading signal.
- The spreading signal is a pseudonoise code sequence that has a chip rate which is of orders of magnitude greater than the data rate, of the message
- All users in CDMA system, use the same carrier frequency and may transmit simultaneously.
- Each user has its own pseudorandom codeword which is approximately orthogonal to all other codewords.
- The receiver performs a time correlation operation to detect only the specific desired codeword. All other codewords appear as noise due to decorrelation.
- For detection of message signals, the receiver needs to know the codeword used by transmitter.

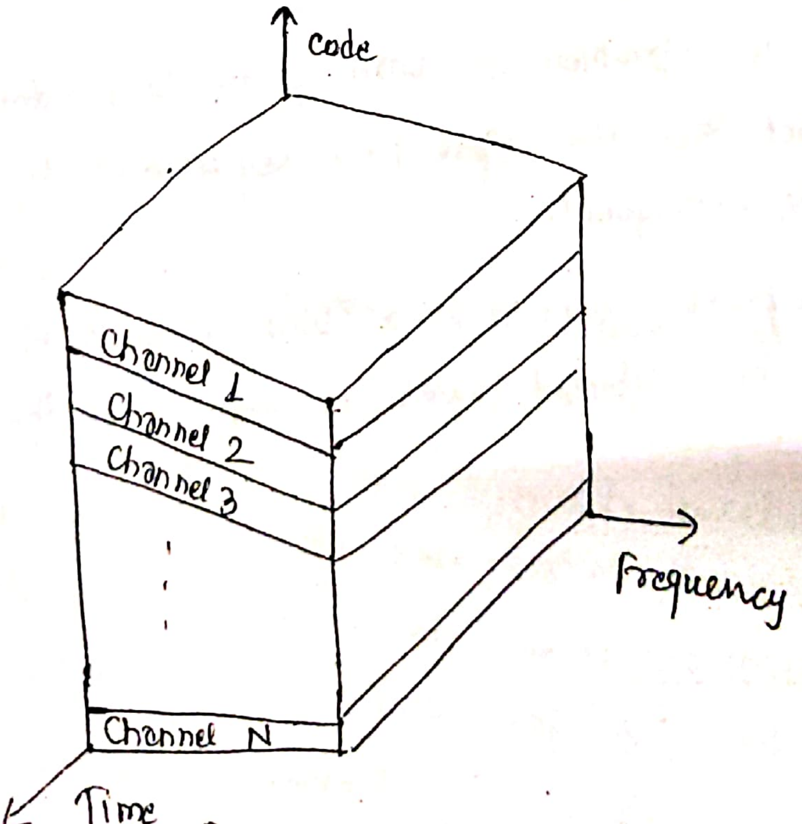


Fig. CDMA System with each channel assigned a unique PN code

The features of CDMA are as follows :-

(i) many users of a CDMA system share the same frequency. Either TDD or FDD may be used

(ii) Unlike TDMA or FDMA, CDMA has soft capacity limit. Increasing the number of users in a CDMA system raises the noise floor in linear manner. So, there is no absolute limit on the number of users in CDMA. Rather, system performance degrades as number of user increases and ~~deg~~ improves as number of users is decreased

(iii) multipath fading may be substantially reduced because the signal is spread over a large spectrum.

(iv). Channel data rates are very high in CDMA systems.

(v) Since CDMA uses co-channel cells, it can use macroscopic spatial diversity to provide soft handoff. Soft handoff is performed by MSC (mobile Switching Centre)

(vi) Self-jamming is a problem in CDMA System. Self jamming arises from the fact that the spreading sequences of different users arenot exactly orthogonal.

(vii) The near-far problem occurs at a CDMA receiver, if an undesired user has a high detected power as compared to the desired user

Space Division Multiple Access (SDMA):

- SDMA controls the radiated energy for each user in space
- SDMA serves different users by using spot beam antennas.
- These different areas covered by the antenna beam may be served by the same frequency (in TDMA or CDMA system) or different frequencies (in FDMA system).
- Sectorized antenna may be thought of a primitive application of SDMA.

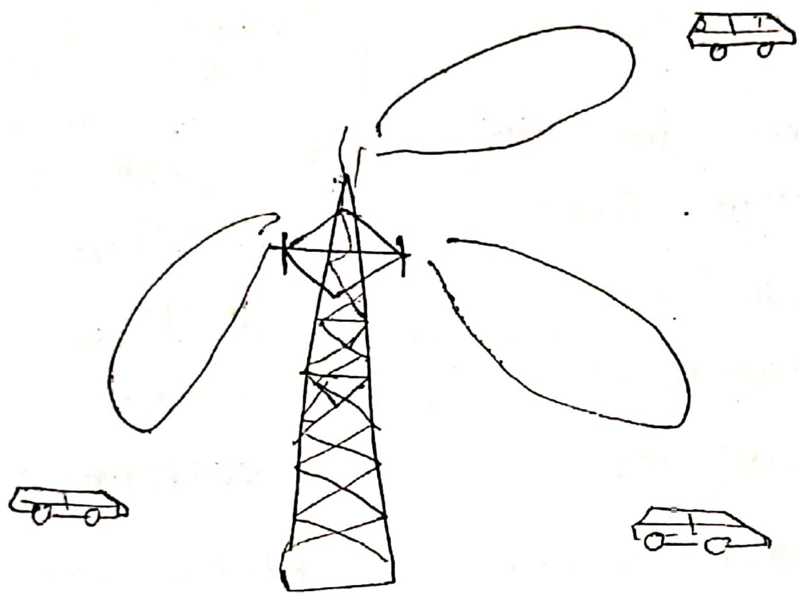


Fig. A spatially filtered base station antenna serving different users by using spot beams.

- The reverse link presents the most difficulty in cellular systems for several reasons. If the base station is made to spatially filter each desired user so that more energy is detected from each subscriber, then the reverse link for each user is improved and less power is required.

Adaptive antennas implement optimal SDMA, thereby providing a unique channel that is free from the interferences of all other users in the cell.

→ With SDMA, all users within the system would be able to communicate at the same time using the same channel.

→ Moreover, a perfect adaptive antenna system would be able to track the individual multipath components for each user and combine them in an optimal manner to collect all of the available signal energy from each user.

Multiple Access Techniques Used in Different Wireless Communication Systems :

Cellular System	Multiple Access Techniques
Advanced Mobile Phone System (AMPS)	FDMA/FDD
Global System for Mobile (GSM)	TDMA/FDD
US Digital Cellular (USDC)	TDMA/FDD
Pacific Digital Cellular (PDC)	TDMA/FDD
CT2 (Cordless Telephone)	FDMA/TDD
W-CDMA (3GPP)	→ CDMA/FDD & CDMA/TDD
cdma 2000 (3GPP2)	→ CDMA/FDD & CDMA/TDD

Multiple-access protocols

- **Random-access protocols**
  - ALOHA
  - CSMA
  - CSMA/CD
  - CSMA/CA
- **Controlled-access protocols**
  - Reservation
  - Polling
  - Token Passing
- **Channelization protocols**
  - FDMA
  - TDMA
  - CDMA

→ The lower sublayer of data link layer ie media access control (MAC) layer is responsible for multiple access resolution.

→ When nodes or stations are connected and use a common link, called a multipoint broadcast link, we need a multiple-access protocol, to co-ordinate access to the link.

→ Random-access protocols are used in Packet radio (PR) access techniques, so radom-access protocols are also called as packet radio protocols.

→ The different packet-radio protocols are as follows:-

① **ALOHA**:

→ ALOHA protocols were ~~of~~ developed for early satellite systems.

→ Allows subscriber to transmit whenever they have data to send.

→ The transmitter then listens for a time interval to determine feedback from the receiver.

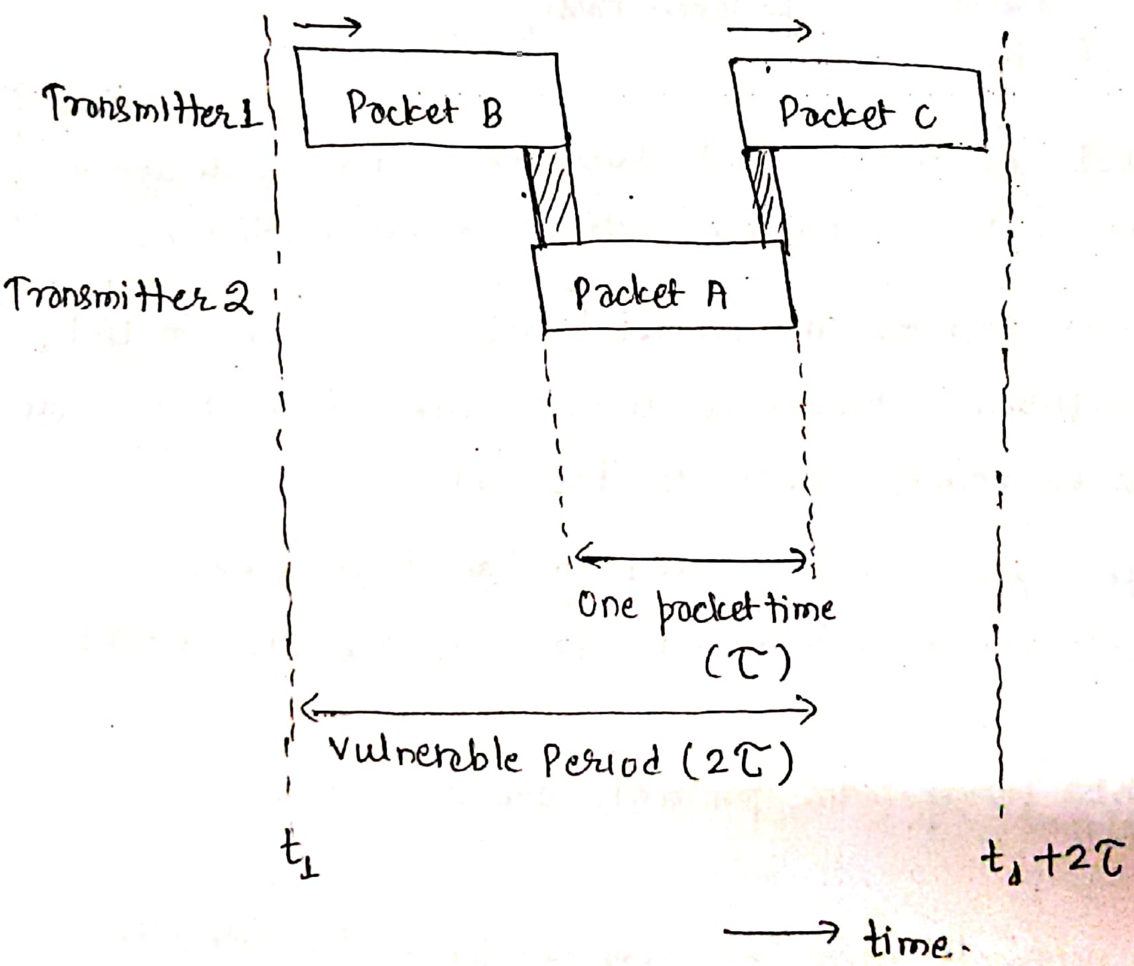
→ The station waits for an amount of time equal to maximum possible roundtrip propagation delay.

→ If an ACK is received, the transmitter knows that the data has been received successfully by the receiver.

→ If NACK is received, the ~~data~~ terminal waits for a random period of time and retransmits the message

→ As the number of users increase, a greater delay occurs because the probability of collission increases.

→ For the ALOHA protocol, the vulnerable period is double the packet duration.



Packet A will collide with packets B and C because of overlap in transmission time.

Fig. Vulnerable period for a packet using the ALOHA protocol.

Disadvantages:

(i) The frame may be invalid because of other data transmitted at the same time. This causes collision.

(ii) Causes degradation in performance as traffic increases due to collision.

(2) Slotted ALOHA:

→ modification of ALOHA.

→ In slotted ALOHA, time is divided into equal time slots of lengths greater than the packet duration τ.

→ The subscribers each have synchronized clocks and transmit a message only at the beginning of a new time slot, thus resulting in a discrete distribution of packets.

→ This prevents partial collisions, where one packet collides with a portion of another.

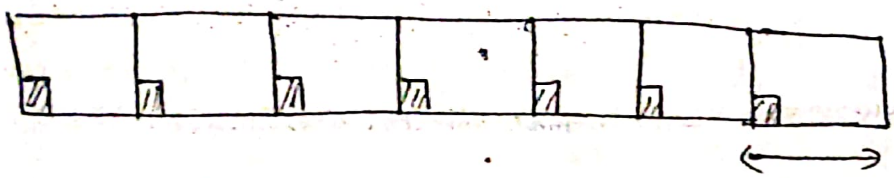
→ As the number of users increase, a greater delay will occur due to complete collisions and the resulting repeated transmissions of those packets originally lost.

→ The number of slots which a transmitter waits prior to retransmitting also determines the delay characteristics of the traffic

→ The vulnerable period for slotted ALOHA is only one packet duration, since partial collisions are prevented through synchronization.

Advantages:

- Efficiency increased as a beginning of each user's data transmission is different. So, collision reduces.



→ To avoid collision,

$$\boxed{\text{frame size} \geq \text{packet duration}}$$

Disadvantages:

→ Collision occurs since it has no carrier sense mechanism.
→ Still collision occurs if,

$$\boxed{\text{Propagation delay} > \text{Packet Duration}}$$

Carrier Sense Multiple Access (CSMA) Protocols:

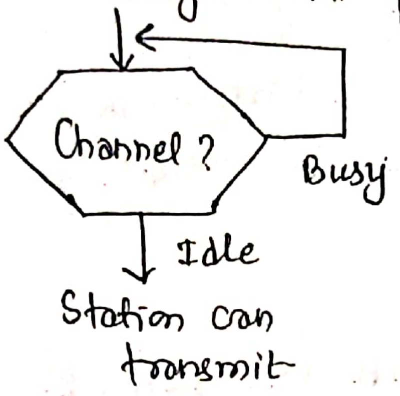
→ ALOHA protocols donot listen to the channel before transmission, and therefore donot explot information about the other users.
→ By listening to the channel before engaging in transmission, greater efficiencies may be achieved.
→ CSMA protocols are based on the fact that each terminal on the network is able to monitor the status of the channel before transmitting information.
→ If the channel is idle (ie no carrier is detected), then the user is allowed to transmit a packet based on a particular algorithm which is common to all transmitters on the network.

There exists several variation of CSMA strategy:-

① **1-persistent CSMA:**

→ The terminal listens to the channel and wait for transmission until it finds the channel idle.

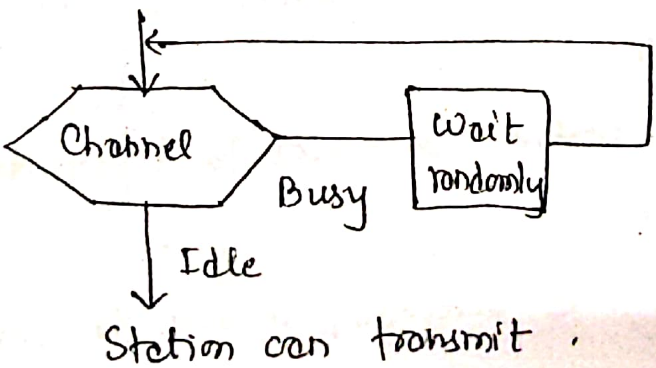
→ As soon as the the channel is idle, the terminal transmits its message with probability one



② **non-persistent CSMA:**

→ In this type of CSMA strategy, after receiving a negative acknowledgement the terminal waits a ~~fed~~ random time before transmission of the packet.

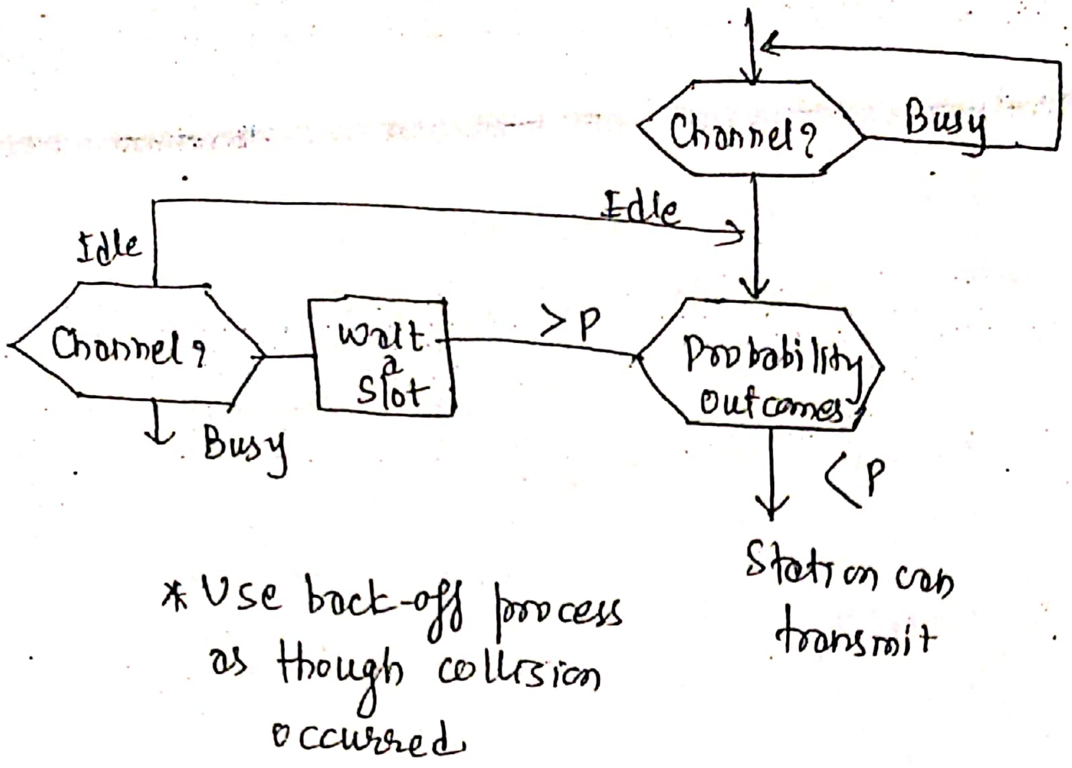
→



③ **p-persistent CSMA:**

→ It is applied to slotted channels.

→ When a channel is found to be idle, the packet is transmitted in the first available slot with probability 'p' or ~~the~~ in the next slot with probability '1-p'



Q. Describe CSMA/CD with its key features.

Ans.

→ CSMA/CD stands for Carrier Sense Multiple Access/ Collision Detection.

→ In CSMA/CD, the user monitors its transmission for collisions

→ CSMA/CD was developed to detect collision along with carrier sense

→ In this, the station continues to listen to the medium while transmitting.

→ If a collision is detected, during transmission, a brief jamming signal is transmitted to assure all stations about the collision occurred recently and cease transmission

→ After transmitting the jamming signals, the station waits for a certain amount of time, then attempts to retransmit.

→ The flowchart to describe the procedures involved in CSMA/CD are as follows:-

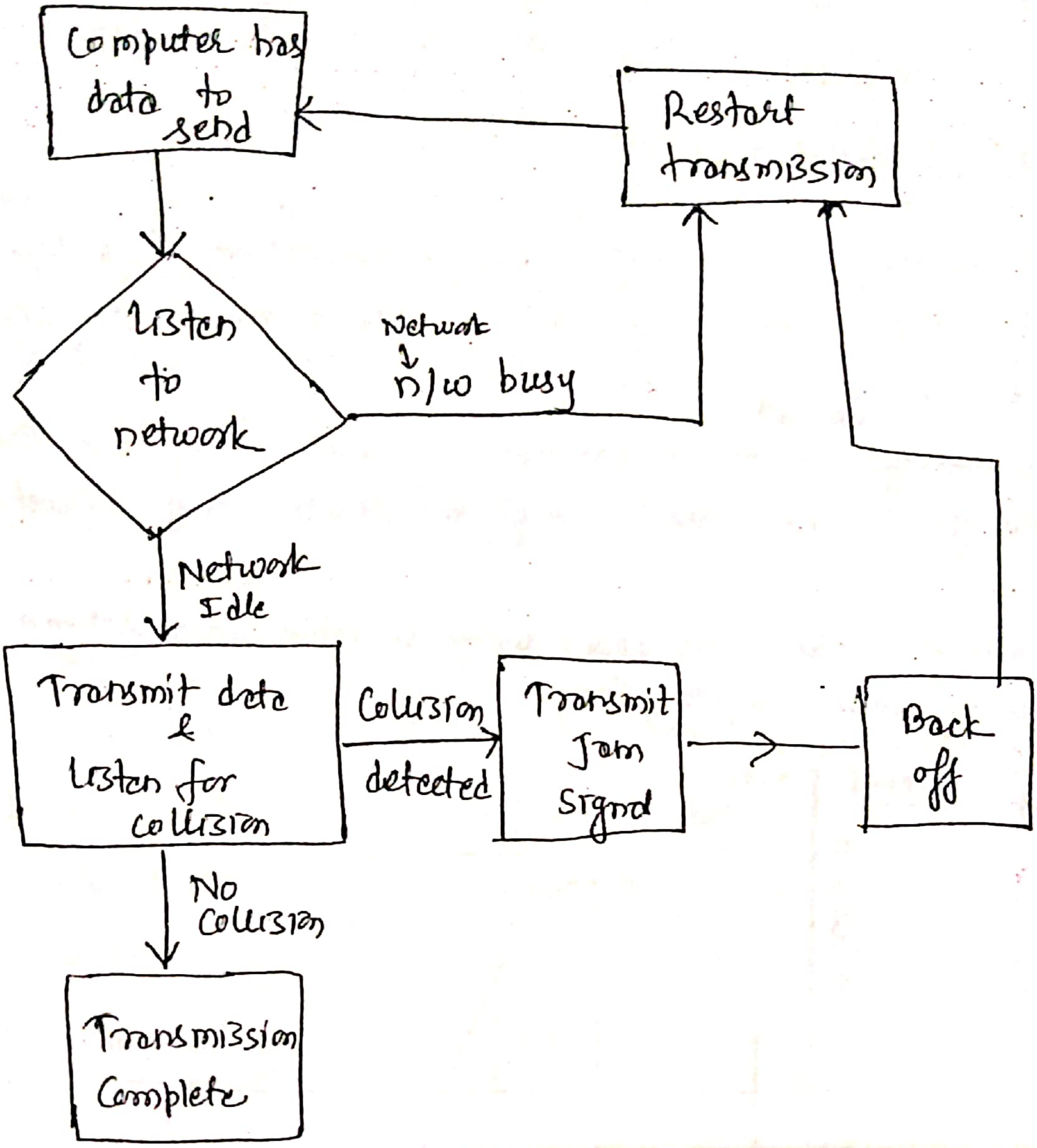


Fig. CSMA/CD

Read self

→ Sampling

→ Quantization

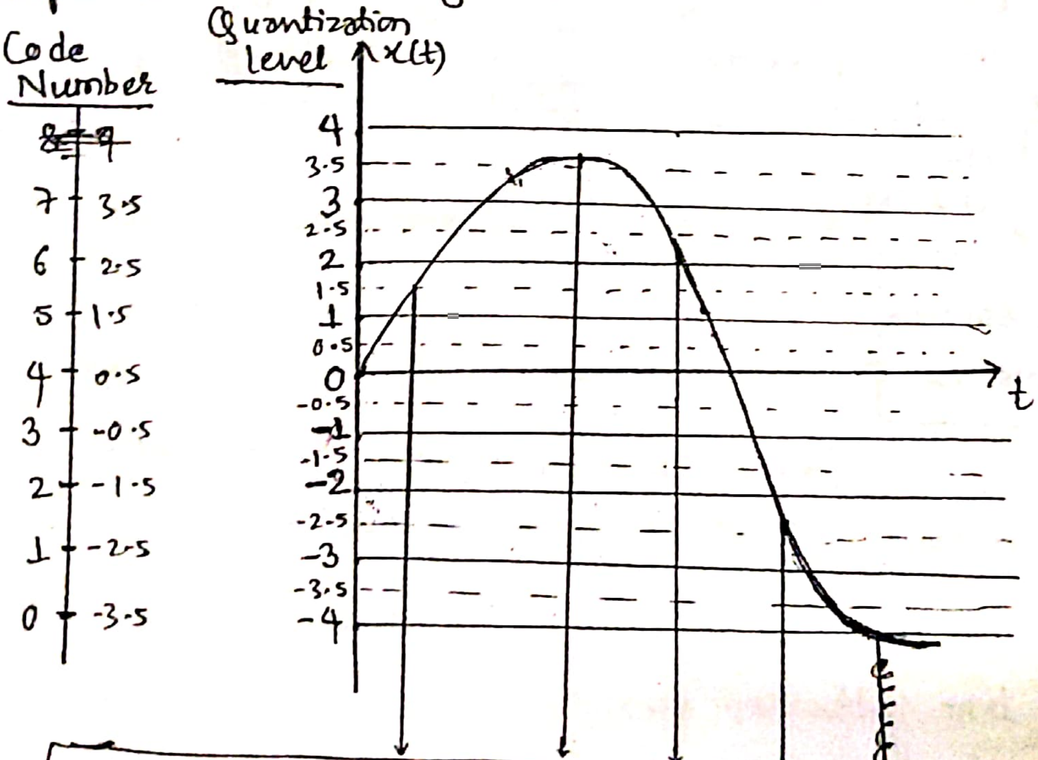
→ Pulse Amplitude modulation (PAM), Pulse width modulation (PWM), Pulse Position modulation (PPM)

Pulse Code modulation (PCM)

→ It is essentially the analog to digital conversion of a special type, where the amplitude of each sample of an analog signal is represented by digital word.

→ In PCM, message signal is sampled and amplitude of each sample is rounded-off to the nearest-one of the finite sets of discrete levels.

→ The quantized version is coded using suitable coding scheme. Thus, output is in coded digital form.



Sample Values:	1.5	3.6	2.3	-2.3	~~-4~~
Nearest Quantization level	1.5	3.5	2.5	-2.5	~~-4~~
Code Number	5	7	6	1	0
Binary Representation	101	111	110	001	

Explanation:

- Assume that an analog signal, $x(t)$ is limited in excursion to range $-4V$ to $4V$.
- Step size between quantization levels has been set at $1V$.
- Thus, 8 quantization levels are employed These are located at

$$-3.5, -2.5, \dots +3.5V.$$

- We assign code number 0 to the level at $-3.5V$, 1 to level -2.5 and so on 7 to $+3.5V$
- Each code number has its representation in binary arithmetic ranging from 000 for 0 to 111 for code number 7.
- The ordinate (ie. y-axis) is labelled with quantization level and their code numbers.
- Each sample of the analog signal is assigned to the quantization level closest to the value of the sample
- Quantization level corresponds to code number
- These code numbers are converted to binary equivalence, generating PCM Sequence

Block Diagram of PCM System:

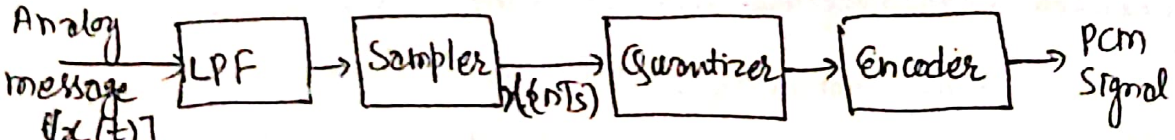


Fig. PCM Transmitter

Fig. Transmission Path.

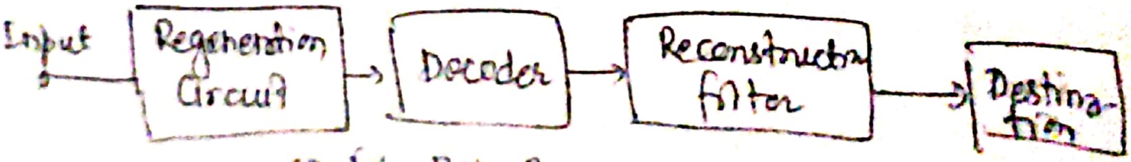


Fig. PCM Receiver

- In PCM generator, the signal $x(t)$ is first passed through LPF of cut-off frequency 'f_m' Hz.
- This LPF blocks all the frequency components which are lying above 'f_m' Hz
- This means that now the signal $x(t)$ is bandlimited to 'f_m' Hz
- The sample and hold circuit then samples this signal at the rate of f_s, $f_s \geq 2f_m$
- The output of sample and hold circuit is denoted by $x(nT_s)$.
- This signal $x(nT_s)$ is discrete in time and continuous in amplitude

Why to use PCM?

- inexpensive digital circuitry may be used in the system
- PCM supports time division multiplexing, where analog sources may be merged with data signal and transmitted over a common high-speed communication system.
- Due to digital nature of the signal, repeaters can be placed betn the transmitter and the receiver. The repeater actually regenerates the received PCM signals.
- Very high noise immunity
- It is possible to store PCM due to digital nature
- Provides high security to data as various coding techniques may be used

Non-Uniform Quantization (Companding) ④

- If the quantizer characteristics is nonlinear and the step size is not constant instead if it is variable dependent on the amplitude of input signal then the quantization is known as non-uniform quantization.
- Some signals (eg speech) contain both low and high amplitudes, but small amplitudes are more likely. Thus, uniform quantization is inappropriate for speech
- Non-uniform quantization provides more quantization levels for low level signals than uniform quantization with the same number of bits, and in telephony this means both quiet and loud talkers can be readily accommodated
- In practice, non-uniform quantization is achieved via sample compression, followed by uniform quantization

Q. Why non-uniform quantization is better than uniform quantization?

Ans.
- Actually uniform quantization uses constant stepsize over all input range
- So, it causes higher quantization error probability, specially to the lower amplitude signals.
- SNR is worst for low level signals than higher level we having quantization noise, $P_q = \frac{\Delta^2}{12}$
- Signal power will be small for weak signals but quantization noise power is constant
  So, SQNR for weak signals is very poor
- So, we use non-uniform quantization (Companding)

Companding:

Companding = Compression + Expanding.

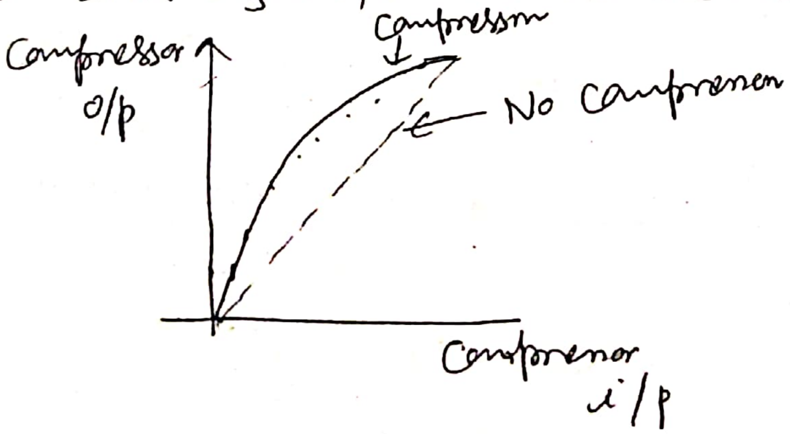
Fig. Compander

→ The weak signals are amplified and strong signals are attenuated before applying them to a uniform quantizer. This is achieved by compressor

→ At receiver exactly opposite is followed, called expansion, done by expander

Compressor:

- Provides a higher gain to weak signals.
- Smaller gain to strong input signals
- Present at PCM transmitter
- Improves ~~SNR~~ SQNR, as it boosts weak signals.



Expander:

- present in receiver of PCM.
- performs exactly the inverse operation of compressor
- all artificially boosted signal by the compressor, are brought back to their original amplitudes at the receiver end

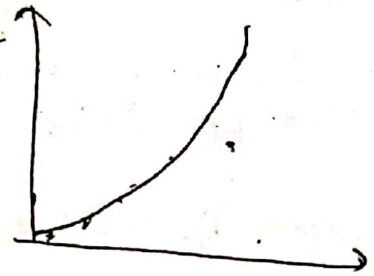


Fig Expander Characteristics

Compander:

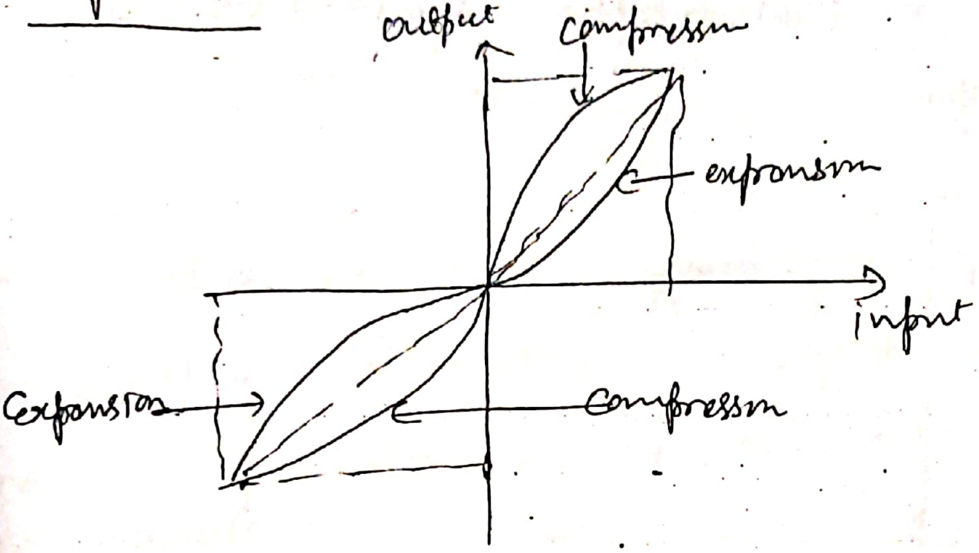


Fig Companding curve for PCM

→ Due to inverse nature of compressor & expander, the overall characteristics of compander is a straight line.

μ-law and A-law Companding:

μ-law → USA, & Canada

A-law → Europe (in Nepal also)

→ Compressor characteristics is linear.

→ It is approximately linear for smaller values of input levels and logarithmic for high input levels.

$$|y| = \frac{\log\left(1 + \mu|x|/x_{max}\right)}{\log(1+\mu)} \quad \text{where } 0 \le \frac{|x|}{x_{max}} \le 1$$

→ Practical value of $\mu = 255$ with 7 bits (128 levels)

→ For $\mu = 0$, characteristics is uniform quantization

→ The μ-law companding is used for speech and music signals.

→ It is used for PCM telephone systems in US, Canada, Japan

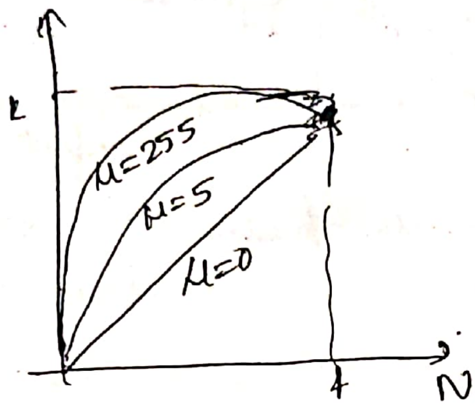


Fig: Compressor Characteristics for μ-law compressor

A-law Companding:

→ $$y = \begin{cases} \dfrac{A \cdot |x|/x_{max}}{1+\log A}, & \text{for } 0 \le \dfrac{|x|}{x_{max}} \le 1/A \\ \dfrac{1+\log\left(A\,|x|/x_{max}\right)}{1+\log A}, & \text{for } 1/A \le \dfrac{|x|}{x_{max}} \le 1 \end{cases}$$

→ Practical value of $A = 87.56$

→ For $A=1$ characteristics is linear (uniform quantization)

→ Used for PCM in Europe

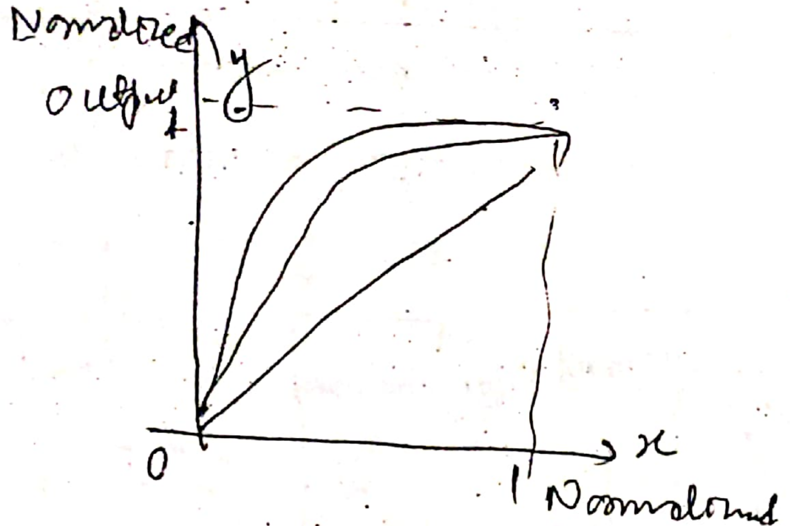


Fig A-law Companding.

→ In practice, non-linear signal processing at the transmitting end together with the inverse nonlinear processing at the receiving end presents considerable problem

→ So, piecewise linear segment approximation is used in practice

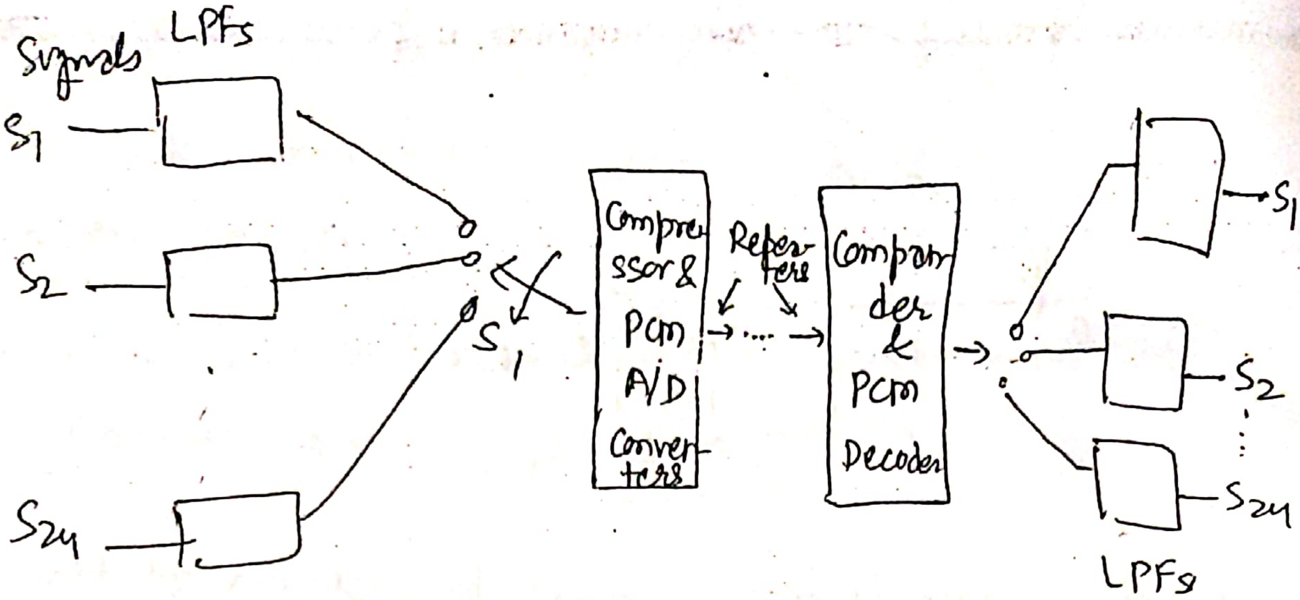
→ A-law → 8 segments

→ 13 effective segment in the curve
So, A-law → 13 segments companding law.

μ-law → 15 effective segment.

→ 8-bit words.

T1 Digital System



→ 24 analog signals, → Uses 'μ' law of companding

→ each signal is band-limited to 3.3 kHz,

→ sampled at the rate of 8 kHz

→ Repeaters are placed periodically at 6000 ft. intervals.

→ The commutator sweeps continuously from S_1 to S_{24} & back to S_1 at the rate of 8000 revolutions per sec and thereby providing 8000 samples per second of each signal

→ Each sample is encoded to 8 bits

→ The digital signal generated will be of

$$24 \times 8 = 192 \text{ bits}$$

→ 1 bit used for synchronization

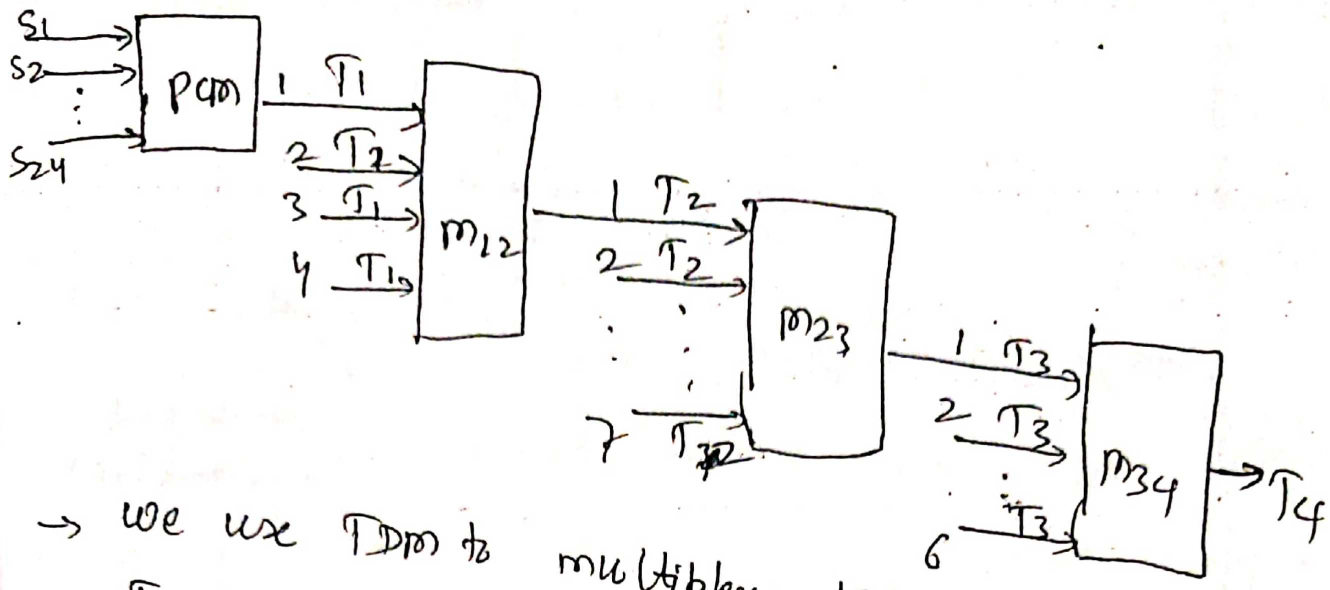
→ So, frame is of $192 + 1 = 193$ bits.

→ ~~Each Sampled~~ Each signal sampled at,

$$T_p = \frac{1}{8000} = 125\mu s$$

→ Bit rate $= f_b(T_1) = \frac{193}{125\mu s} = 1.544$ mbps.

multiplexing T1 lines - T1, T2, T3 and T4 lines:



→ we use TDM to multiplex various T_1, T_2, T_3 & T_4 lines. (4, 7, 6)

Bit Rate:

- M12 adds 17 bits for frame synchronization.
  So. No. of bits per frame
  $= (193 \times 4) + 17 = 789$ bits/frame

  ∴ T2 bit rate:
  $f_b(T_2) = 789$ bits/frame $\times \frac{1}{8000}$
  $= 6.312$ mb/s

→ M23 mux adds 69 bits for frame synchronization
so, no. of bits per frame,
$= (789 \times 7) + 69 = 5592$ bits/frame, and
$f_b(T_3) = 5592 \times \frac{1}{8000} = 44.736$ mb/s

→ M34 mux adds 720 bits for frame synchronization, so, no. of bits per frame
$= (5592 \times 6) + 720 = 34272$ bits/fram
and $f_4(T_4) = 34272 \times \frac{1}{8000} = 274.176$ mb/s

PCM Transmission Formats:

- Pulse generator at receiving terminal must be synchronized with that of sending terminal.
- PAM, PPM, PWM are not used for line transmission because attenuation and delay distortion cause dispersion of the transmitted pulses. They spread in time and interfere with the pulses of adjacent channels, thus causing inter-channel crosstalk.
- PCM systems were first developed for telephone transmission over cables originally designed for audio-frequency transmission. It is found that these are satisfactory, using suitable bipolar coding for transmitting up to 2 Mbit/s.
- Consequently, telephone channels are combined by time-division multiplexing to form an assembly of 24 or 30 channels. This is known as the **primary multiplex group.**
- It is also used as a building block for assembling larger numbers of channels in higher-order multiplex systems.
- Types of PCM primary multiplex group:
  - T1 Digital System (24-Channel PCM)
  - E1 Digital System (30-Channel PCM)

24-Channel Frame Format:

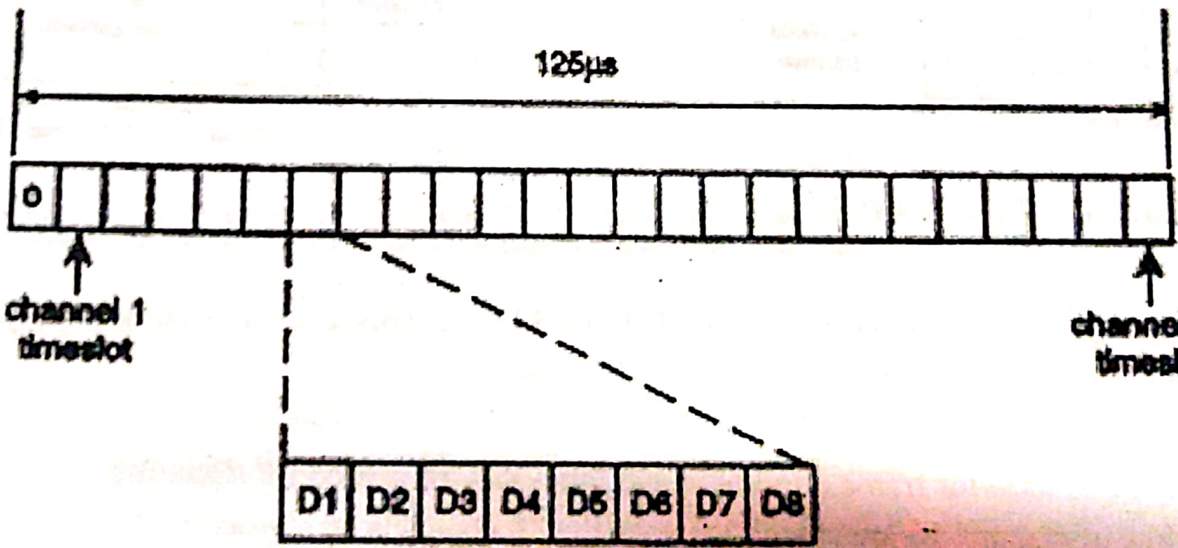


Figure 24-channel frame format

- The frame format of the 24-channel system is as shown in figure above.
- The basic frame consists of 193 bits, thus, the digit rate is 193 x 8 KBit/s = 1.544 Mbit/s.

- The first bit is used for framing and is called F bit, the others form 24 8-bit time slots for speech channels.
- On odd-numbered frames, the F bit takes on the alternating pattern '1, 0, 1, 0...' which is the pattern for frame alignment.
- On even frames, the pattern is '0, 0, 1, 1, 1, 0...' which defines a 12-frame multiframe.
- On frames 6 and 12 of multiframe, bit D8 of each channel time-slot is used for signaling for that channel.
- This bit stealing causes a small degradation in quantizing distortion.

30-channel PCM Frame Format

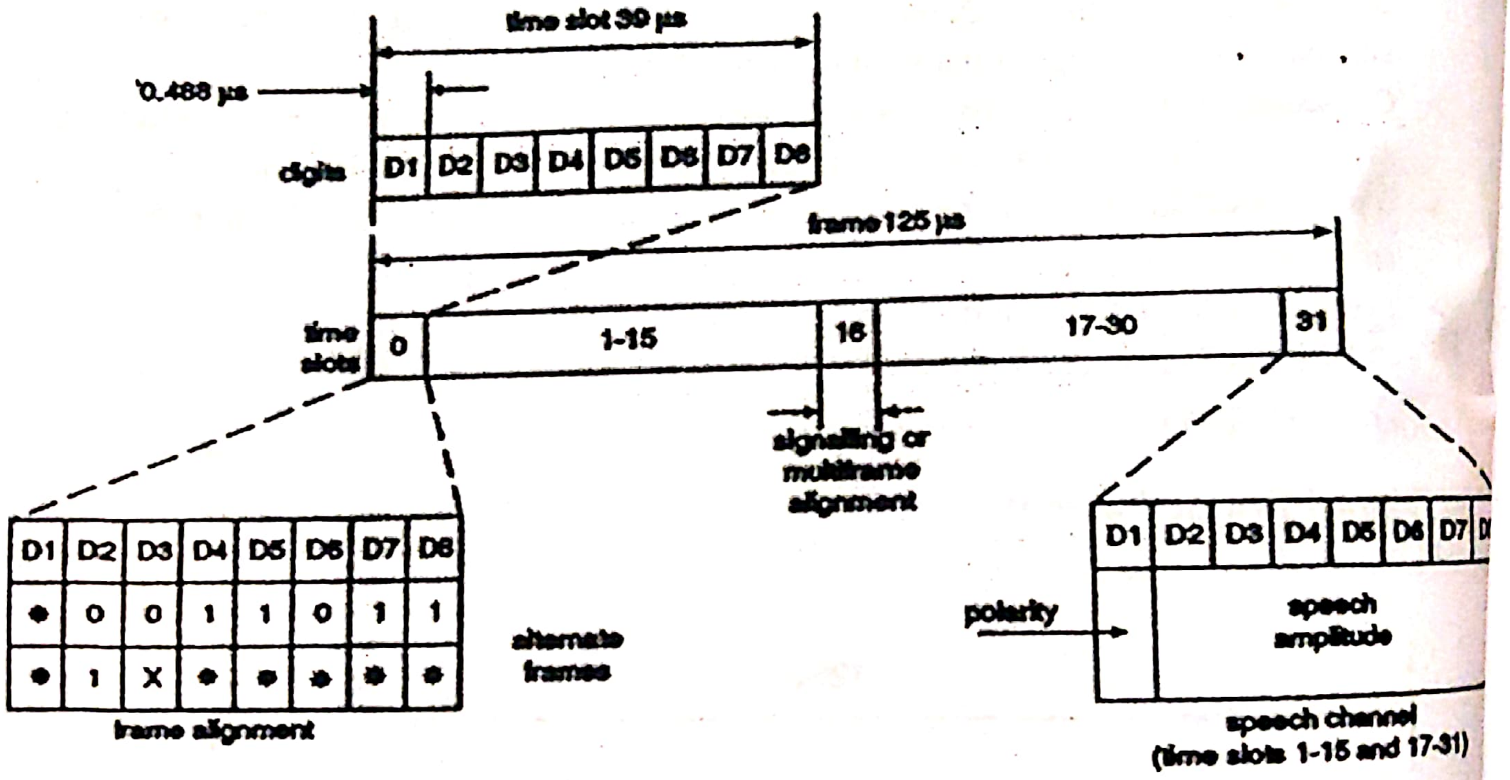


Figure 30-channel frame format

- The frame of 30- channel system is divided into 32 time-slots, each of 8 digits.
- Thus, total bit rate is 8 kHz x 8 x 32 = 2.048 Mbps.
- Time slots 1 to 15 and 17 to 31 are each allotted to a speech channel.
- Time slot 0 is used for frame alignment and time- slot 16 is used for signaling.
- The 8-bits of channel 16 are shared between the 30 channels by a process of *multiframing*. 16 successive appearances of channel 16 form a multiframe of 8-bit time-slots.
- The first contains a multiframe alignment signal and each of the subsequent 15 time-slots contains four bits for each of two channels.

Framing Alignment Strategies

1. **Added- digit Framing:**

- Periodically insert a framing bit with an identifiable data sequence.
- Usually framing bit is added once for every frame and then alternatives in value.
- Used in T1 digital system.
- Continuously monitor all bit positions, when misframe is detected, a new frame position is established.

2. **Added- channel Framing:**

- Framing digits are added in group such that extra channel is established. E.g. E1 digital system

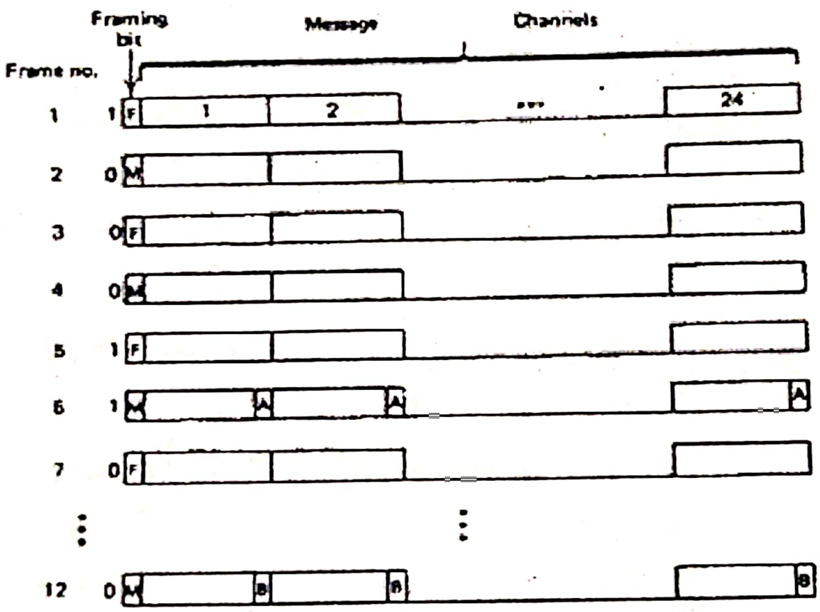


Figure 4.34 Twelve-frame superframe structure of DS1 signal with robbed digit signaling. Frame alignment signal (F) = 101010; multiframe alignment signal (M) = 001110.

Unique Line Code Framing:

- Bipolar coding manages to shape the spectrum (remove the dc component) of the line code by adding extra signal levels to provide more flexibility in selecting signals.
- E.g. ISDN S/T bit rate interface.

Statistical Framing:

- Relies on statistics of data within individual bits of transmission schemes.
- E.g. bit is MSB of PCM. etc. used in ADPCM 7-kHz audio

E1 Telephony / 30 Channel PCM / Co-channel PCM

- It uses A law companding
- Contains 30 voice channel plus 2 channels for frame alignment and signaling
- Each channel divided into 8 bits,

So,

Total no. of bits $(n) = (30+2) \times 8 = 256$ bits

- message signal $(f_m) = 3400\text{Hz}$
- Sampling freq. $(f_s) = 2f_m \simeq 8\text{kHz}$
- T_s (total duration) $= \frac{1}{f_s} = 125\mu s$
- Data rate $= nf_s = 256 \times 8000$

$= 2.048$ mbps.

- Each bit duration $= \frac{T_s}{n} = \frac{125}{256} = 0.488\ \mu s$.

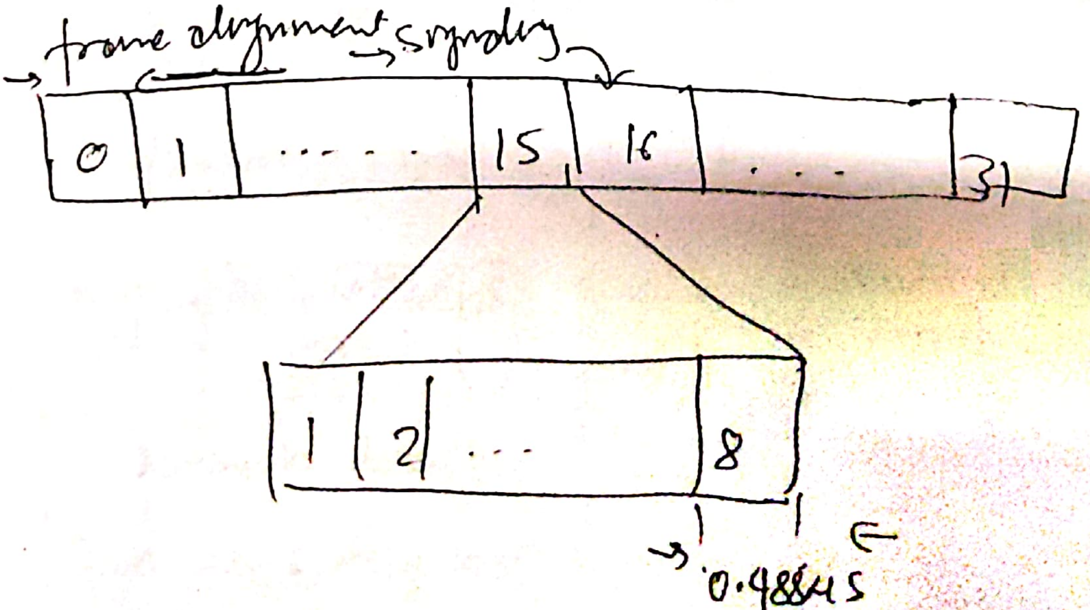


fig E1 Telephony.

→ To overcome, this problem pulse-code modulation (PCM) is used

→ In PCM, each analog sample is applied to an analog-to-digital (A/D) converter, which produces a group of pulses that represents its voltage in a binary code

→ And decoded at receiving end.

→ Since the coder used for A/D conversion and the decoder used for D/A conversion are required to perform their operations within the duration of time slot of one channel, they can be common to all the channels of a TDM system

30-channel PCM frame format:

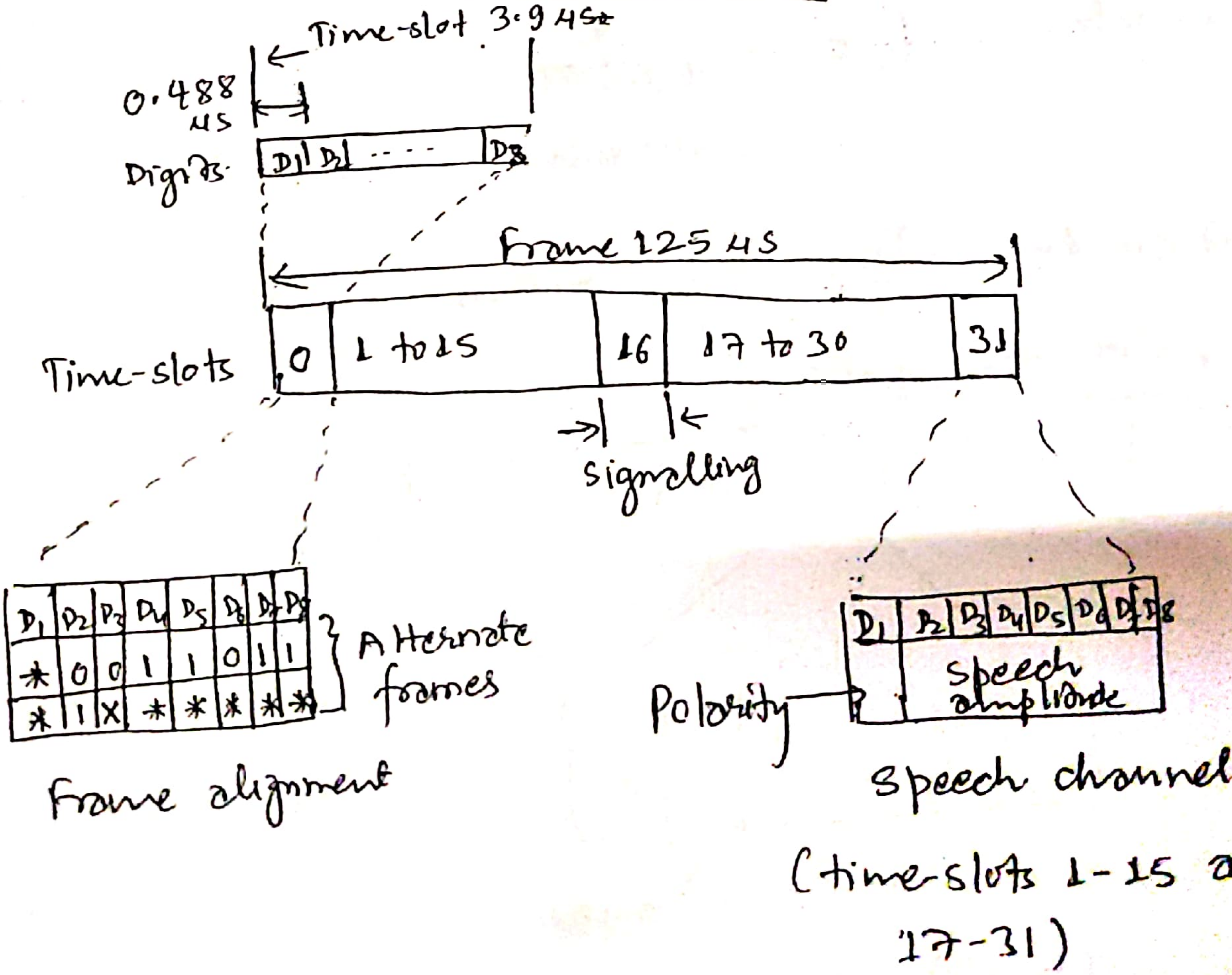


Fig. 30-channel PCM frame format

Framing:

- To identify individual time slots, within a TDM frame, certain amount of transmission overhead is required to establish and maintain frame synchronization

Frame Alignment Strategy:

(i) Added Digital Framing:

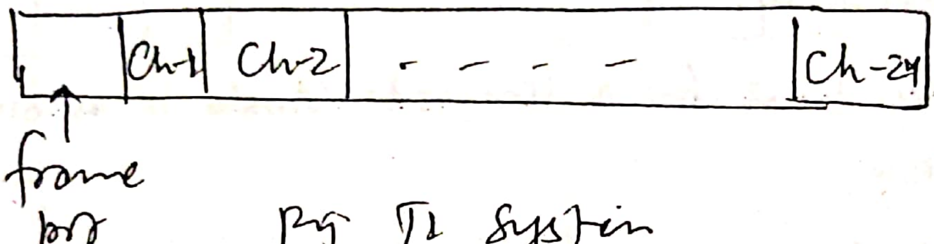


Fig: T1 system

(ii) Added Channel Framing:

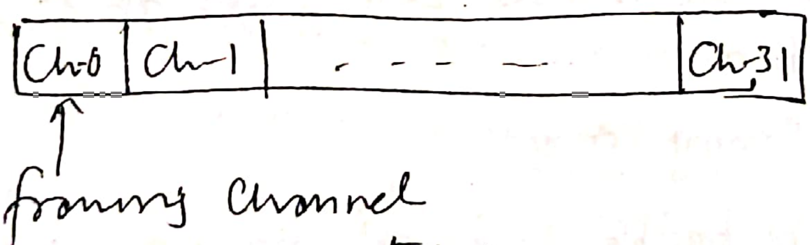


Fig: E1 system

(iii) Unique Line Coding Framing:

- frame synchronization is established using bipolar coding.
- Bipolar violation can be used to identify the framing boundaries uniquely & rapidly.

(iv) Statistical Framing:

- Based on statistics of data
- Only applicable to special appⁿ. eg ADPCM.

Line Coding:

→ It is simply the mapping of data bits to signal elements.

→ The waveform pattern of voltage or current used to represent the 1s and 0s of digital data on transmission link is called as line coding.

→ A line code must have the following properties:

① **Transmission Bandwidth:**

- For line code, the transmission bandwidth must be as small as possible

② **Power efficiency:**

- Transmitted power for a line code should be as small as possible

③ Error detection and correction capability

④ **Favourable Power Spectral Density:**

- It is desirable to have zero power spectral density (PSD) at $\omega = 0$.

⑤ **Adequate Timing Content:**

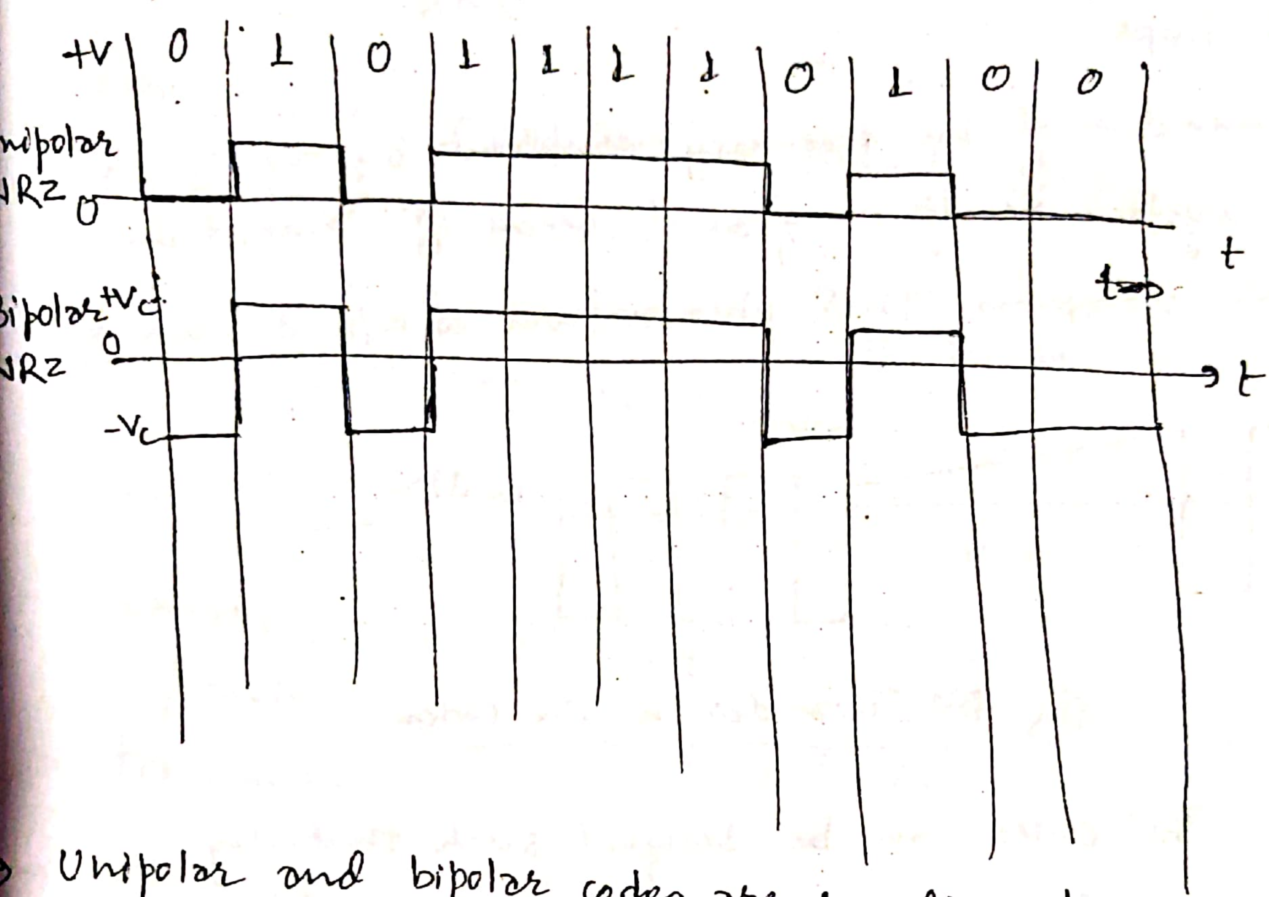
- It must be possible to extract timing or clock information from the signal.

⑥ **Transparency:**

- It must be possible to transmit a signal correctly regardless of the patterns of 1's and 0s.

Line Coding (…remaining).

→ Binary digital information is usually represented by either unipolar or bipolar codes using positive or negative logic systems.



- Unipolar and bipolar codes are sometimes k/a balanced and unbalanced codes respectively.
- Bipolar codes are more efficient from the point of view of power consumption.
- In NRZ (non return-to-zero) code, the voltage levels are maintained for entire duration of the signal
- In return-to-zero (RZ) codes, there is a voltage transition for every '1' state of the signal
- So, RZ codes are better suited for clock synchronization, than NRZ.

→ However, long strings of zeros have no transition in RZ coding.

→ ~~They~~ Above signals use non-zero dc, but most transmission lines donot pass d.c signals as they are a.c coupled using transformers or capacitors to eliminate d.c ground loops.

→ The elimination of low frequency components by the transmission system results in gradual decay of amplitudes of NRZ waveforms. This phenomenon ~~is~~ k/a. d.c wander

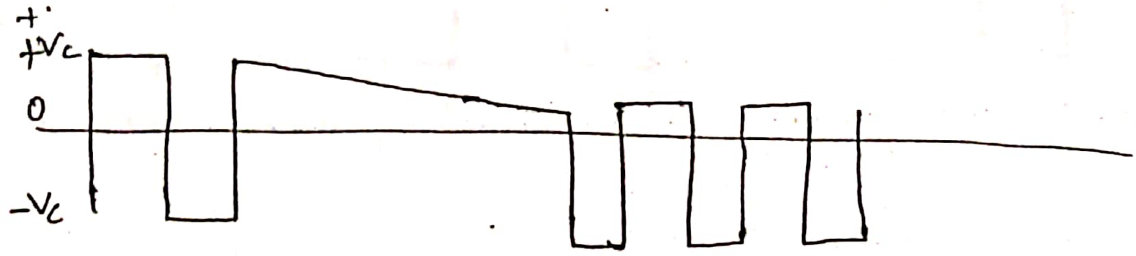


Fig. DC wander in line codes

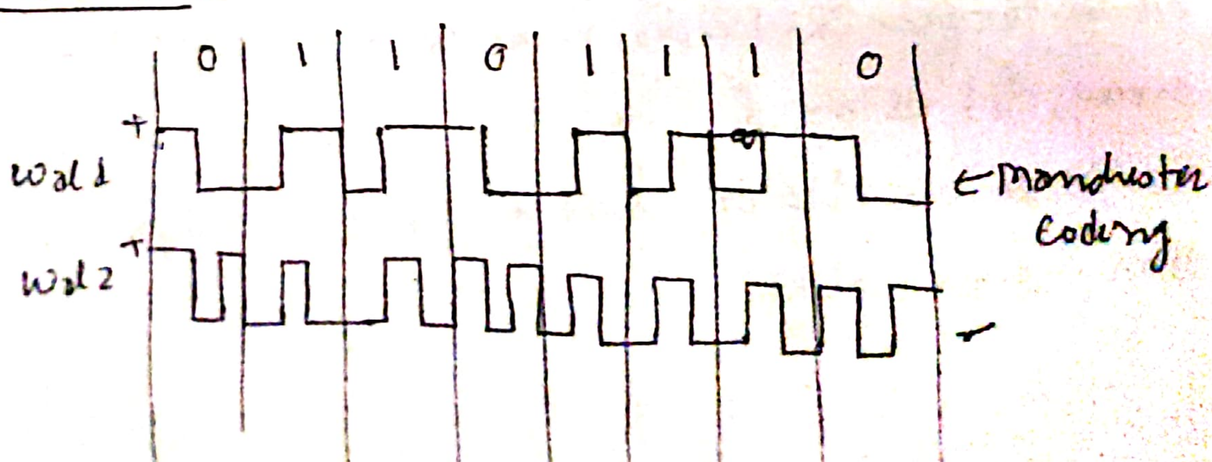
→ The line codes can be designed such that they donot contain any dc energy.

→ Two bipolar line codes, based on Walsh function donot contain any d.c component.

→ Code contain large number of transitions ~~and~~ from which timing information can be recovered easily.

Walsh code:

→



Input 0 1 0 0 1 1 0 1 1 1

AMI

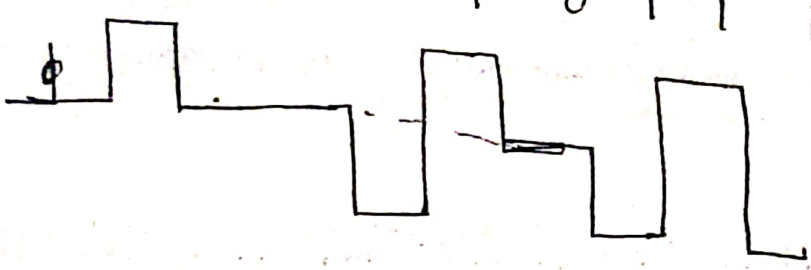


Fig AMI Coding

→ AMI coding resolves two important problems encountered by many other bipolar codes

- no dc component in transmitted signal and hence no phenomenon of <u>dc wander</u>.
- Excellent timing information

– However, the code fails to provide timing information when a long series of zeros are transmitted

→ Modification to AMI coding k/a <u>binary N zeroes substitution</u> breaks up long runs of zeros (N zeroes) by substituting a group of pulses that violate the AMI alternating pulse polarity rule

→ A widely used code in this category is HDB3 (high density binary) code recommended by CCITT.

High Density Bipolar (HDB) Signalling:

- ~~fo~~ In case of bipolar NRZ or AMI signal, the transmitted signal is equal to zero when a binary 0 is to be transmitted
- This is true even for the unipolar RZ and unipolar NRZ signals.
- The absence of transmitted signal can cause problems in synchronization at the receiver, if the long sequence of binary 0s are being transmitted
- This problem can be solved by adding (transmitting) pulses when long strings of 0s exceeding a number 'n' are being transmitted
- This type of coding is called as High Density Bipolar Coding, denoted by HDBN, where, N = 1, 2, 3, ----
- The most widely used HDB format is with N = 3 i.e HDB3.

B8ZS : 000V B0VB

HDB3 : $\underset{\text{Odd}}{\underset{\downarrow}{000V}}$ $\underset{\text{Even}}{\underset{\downarrow}{B00V}}$ where, B = V = 1

Techniques for HDB3 coding:

- Always replace initial four zeros (irrespective of even or odd 1s) with 000V.
- If 5 zeros then, neglect last zero and only replace first 4 zeroes as per the odd/even rule.
- In 8 zeroes (employing HDB3), always replace last 4 zeroes by B00V.

B8ZS (Bipolar 8 zeros substitution)

→ At least continuous 8 zeros are required

→ Substitution by, 000V BOVB

where, V = Violation Bit

B = Bipolar Bit

and, B = V = 1.

Q. Explain PDH and SDH.

Q. Why number of bits in the Higher Order multiplex system is not exact multiple of the number of times the bits in the tributary.

near

Plesiochronous Digital Hierarchy (PDH):

→ Almost synchronous but not synchronous.

→ multiplexing is used to lift up in the hierarchy.

→ The primary multiplex group is used as building block for larger number of channels in higher order multiplex system.

→ They originate from different crystal oscillators and vary within the clock tolerance, so they are said to be plesio-chronous.

→ At each level in the hierarchy, several bit streams, known as tributaries are combined by a multiplexer.

→ The output from a multiplexer may serve as a tributary to a multiplexer at the next higher levels in the hierarchy.

→ There are two methods of interleaving digital signals from tributaries to the multiplexers :-

(I) Bit interleaving, and (II) word interleaving

(I) Bit interleaving:

- In bit interleaving, one bit is taken from each tributary in turn.
- If there are N input signals, each with a rate of 'f_t' bit/s, then the combined rate will be 'Nf_t' bit/s and each element of the combined signal will have a duration equal to 1/N of an input digit.

(11) Word interleaving:

- In word interleaving, group of bits are taken from each tributary in turn and this involves the use of storage at each input to hold the bits waiting to be sampled.

→ ~~Bit~~ Since bit interleaving is simpler, it was chosen for PDH.

→ There are 3 incompatible sets of standards for PDH centered on Europe (30-channel), North America and Japan (24-channel primary multiplex).

→ The frame length is same as for primary multiplex i.e. 125μs, since this is determined by the basic channel sampling rate of 8 kHz.

→ However, when N tributaries are combined, the number of digits contained in higher-order frame is greater than N times the number of digits in the tributary frame. This is because it is necessary to add extra 'overhead' digits for: frame alignment and justification.

Frame alignment:

- A higher-order demultiplexer must recognize the start of each frame in order to route subsequent digits to the correct outgoing tributaries, just as a primary demultiplexer must ~~recognize~~ route received digits to the correct outgoing channels.
- A unique code is sent as a frame alignment word (FAW), which is recognized by demultiplexer and used to maintain its operation in synchronism ~~is~~ with incoming signal.

Justification:

- This process is used to enable the multiplexer and demultiplexer to maintain correct operation, although the input signals of the tributaries entering the multiplexer may drift relative to each other

- If an input tributary is slow, a dummy digit (i.e justification digit) is added to maintain the correct output digits rate.

- If the input tributary speeds up, no justification digit is needed.

- This justification digit must be removed by demultiplexer, inorder to send correct sequence of signal digits to the output tributary.

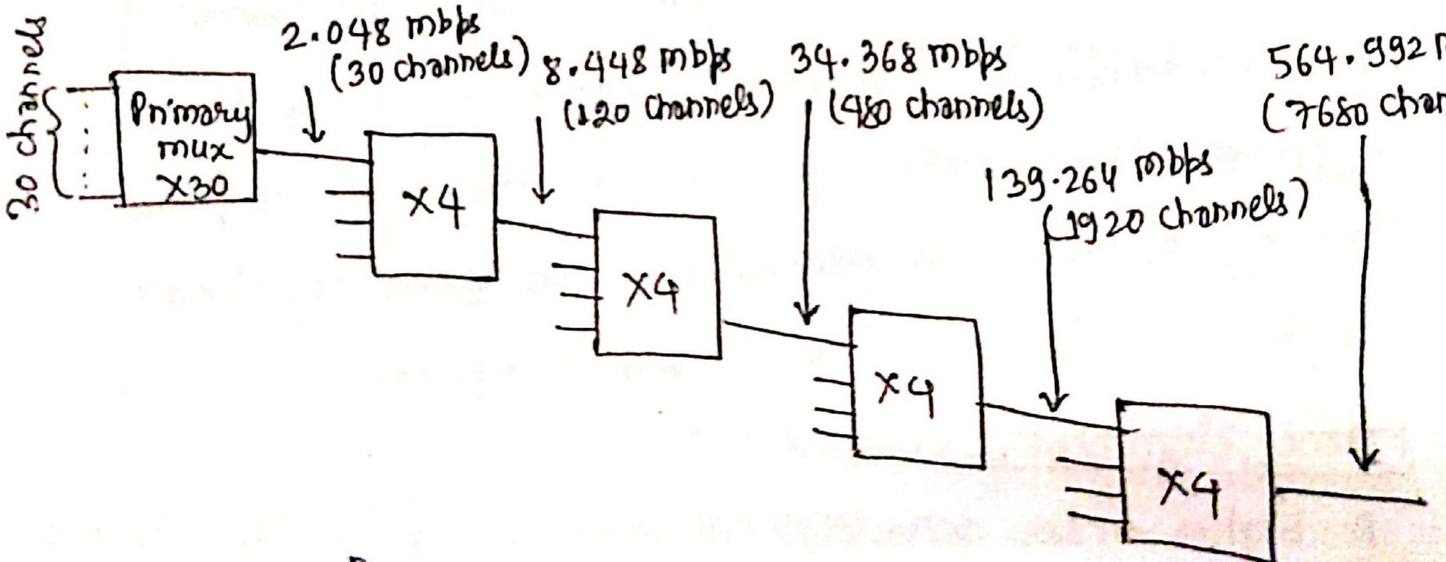


Fig: European Plesiochronous Digital Hierarchy

SDH (Synchronous Digital Hierarchy)

- Networks are becoming fully digital, operating synchronously, using high-capacity optical-fiber transmission system and time-division switching.
- SDH is full synchronous operation hierarchy.
- In USA, this is called SONET (Synchronous Optical Network).
- The basic SDH signal is known as Synchronous Transport module at level 1 (STM-1).
- The STM-1 frame format is as shown below:

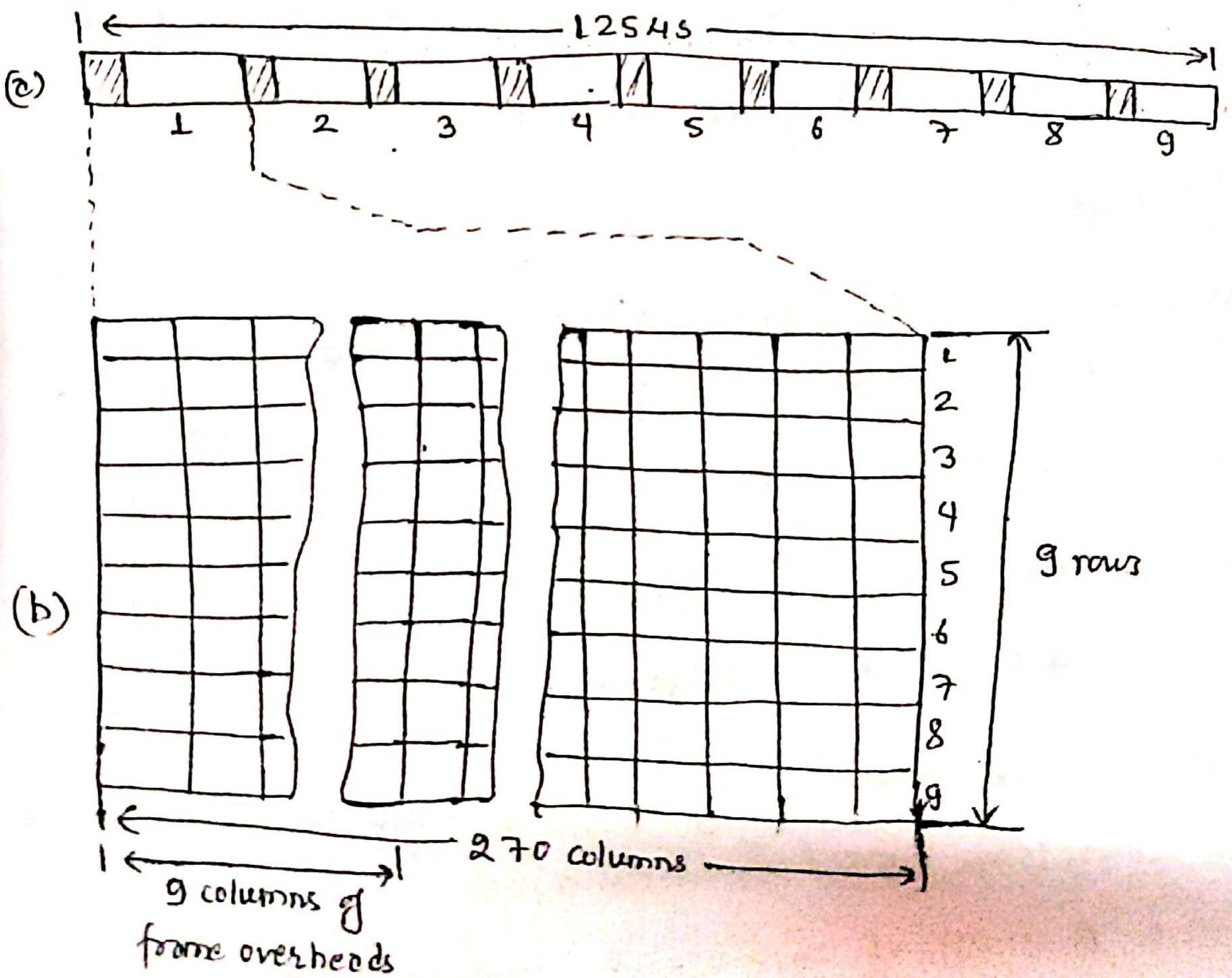


Fig. SDH frame structure (STM-1),
(a) outline frame structure,
(b) frame structure shown in rows and columns.

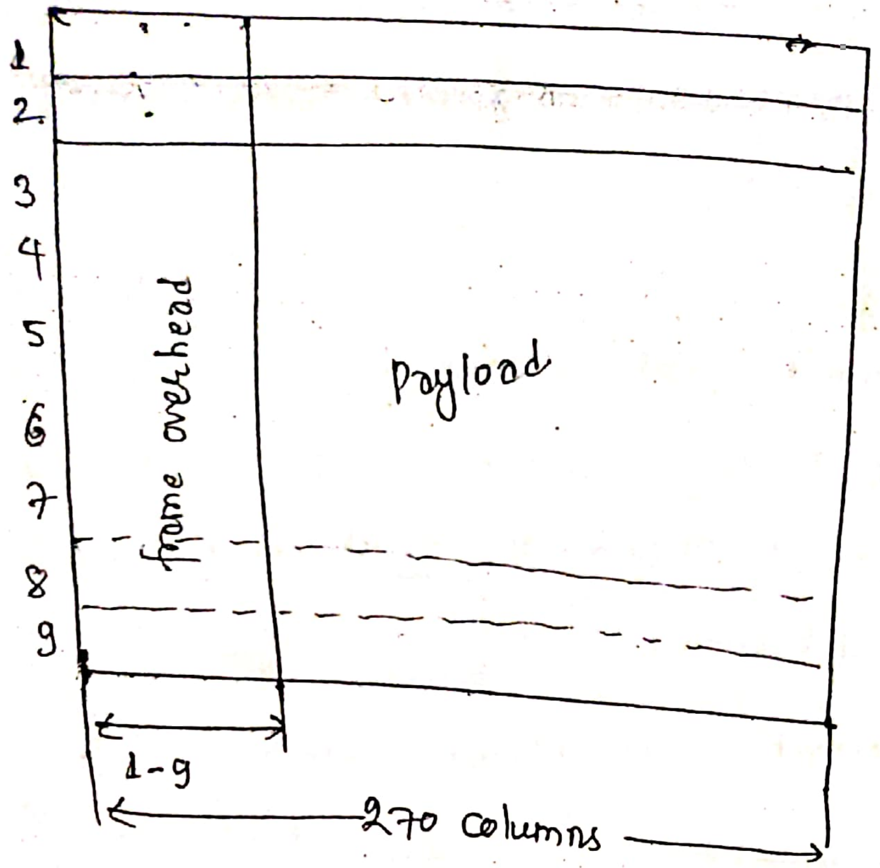


Fig. STM-1 frame format (simplified)

Bit. Rate:

$STM\text{-}1 = 63 E_1 = 63 \times 2.048$ mbps + Overhead
$= 155.52$ mbps.

$STM\text{-}4 = 4 \times STM\text{-}1 = 622.08$ mbps.

$STM\text{-}16 = 4 \times STM\text{-}4 = 2.5$ Gbps

$STM\text{-}64 = 4 \times STM\text{-}16 = 10$ Gbps.

→ Contains special overhead bytes and pointer for error detection error correction, NMS, maintenance, etc.

→ STM-1 consists of nine equal segments, with 'overhead' bytes at start of each.

- The total length is 2430 bytes, with each overhead using nine bytes.

- STM-1 frame is usually represented as nine rows and 270 columns of 8-bit bytes, as shown in fig above.
- The first nine columns are for Section overhead (SOH), such as frame alignment, error monitoring and data.
- The remaining 261 columns comprise the payload into which a variety of signals can be mapped.
- Each tributary to the multiplex has its own payload area, known as tributary unit (TU). In North America a TU is called a Virtual Tributary (VT).
- Each column contains 9 bytes (one from each row); with each byte having 64 kbit/s capacity.
- Three columns (ie. 27 bytes) can hold 1.5 mbit/s PCM signal, with 24 channels and some overheads.
- Four columns (ie 36 bytes) can hold a 2 mbit/s PCM Signal with 32 time-slots. ie. 32 channels.
- In this way, SDH accommodates E_1 and T_1 Lines.
- In the multiplexing process, the bytes from a tributary are assembled into a container and a path overhead is added to form a virtual container (VC).

  In North America, the VC is known as Virtual Tributary Synchronous Payload Envelope.
- The VC travels through the network as a complete package until it is demultiplexed.

Q. Explain STM-1 frame format of SDH. How T1 and E1 lines are accommodated in STM-1 frame? What is Virtual Container?

Imp

Differences between PDH and SDH:

SDH	PDH
(i) Synchronous multiplexing technique	(i) Asynchronous multiplexing technique
(ii) High data rate (upto 10Gbps)	(ii) Low data rate (only upto ~~564~~ 564.992 mbps)
(iii) Use optical Fibre Transmission System.	(iii) Use Cable Transmission System.
(iv) Establish standard multiplexing format.	(iv) Does not establish any standard multiplexing format
(v) Support network management.	(v) Doesnot support network management
(vi) Flexible format	(vi) Not flexible
(vii) Can handle large traffic	(vii) Handle less traffic
(viii) Advanced System of Digital Hierarchy	(viii) Primitive System of Digital Hierarchy
(ix) Accommodates both E_1 and T_1 lines.	(ix) Only accommodate either E_1 or T_1 at a time
(x) Less bit stuffing is used.	(x) Very high bit stuffing is used, so complex multiplexing

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Chapter - 4 (Telecommunication)
Switching Techniques and System

→ The primary function of a switching system is to establish an electrical path between the given inlet (set of input circuits) - outlet (set of output circuits) pairs.

→ The hardware used for establishing such a connection is called a switching matrix or the switching network.

Imp.
Functions of a Switching System

The basic functions that all switching systems must perform are as follows:-

① Attending:
- The system must be continually monitoring all lines to detect call requests.
- The 'calling' signal is sometimes known as 'seize' signal because it obtains a resource from the exchange.

② Information receiving:
- In addition to receiving call and clear signals, the system must receive information from the caller as to the called line required. This is called address signal.

③ Information processing:
- The system must process the information received, in order to determine the actions to be performed and to control these actions.

~~Since~~

) Busy testing:
- Having processed the received information to determine the required outgoing circuits, the system must make a busy test to determine whether it is free or already engaged on another call

⑤ Interconnection:

→ For a call between two customers, three connections are made in the following sequence:-

ⓐ A connection to the calling terminal,

ⓑ A connection to the called terminal.

ⓒ A connection between the two terminals.

⑥ Alerting:

- Having made the connection, the system sends a signal to alert the called customer to the call, eg. by sending ringing current to a customer's telephone.

⑦ Superivision:

- After the called terminal has answered, the system continues to monitor the connection in order to be able to clear it down when the call has ended.

⑧ Information Sending:

- If the called customer's line is located on another exchange the additional function of information sending is required.

Data Switching:

① Circuit Switching.
② Message Switching.
③ Packet Switching.

① Circuit Switching:

→ In circuit switching, the dedicated electrical path is established between the source and the destination before any data transfer takes place.

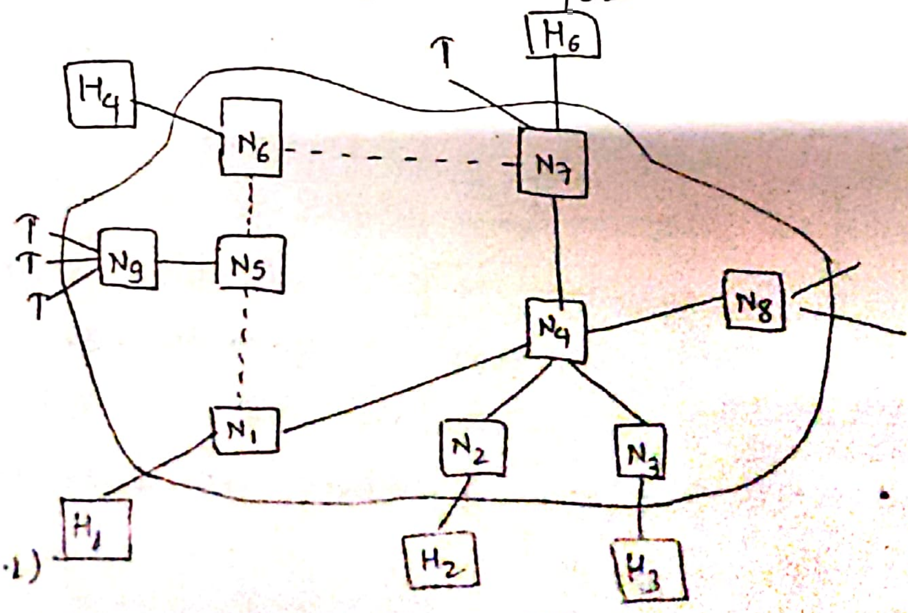
→ The electrical path may be realized by physical wires or coaxial cables or radio or satellite links.

→ The circuit switching establishes a dedicated connection (a radio channel between the base and mobile stations, and a dedicated phone line between the MSC and the PSTN) for the entire duration of a call; irrespective of whether the data is actually transferred or not.

→ No other potential user can use the path even if it is idle.

→ The connection is released only when specifically signalling is signalled by either of the communicating entities.

→ Data transmission using a PSTN connection is a typical example of Circuit switched Data transfer



(fig 5.1)

H = Host
N = Node
T = Terminal

M = Message lengths in bits.

R = data transfer in bps.

T_h = time taken by a node to make host entries.

Example:

Q. A circuit switched connection involves 5 ~~stot~~ switching node. Each node takes 2 seconds and 0.2 second for establishing and releasing connections respectively. If the data transfer rate is 2.4 kb/s., Compute the data transfer time for a message that is 300 bytes long.

Soln:

$$T_{cs} = (N-1)T_m + \frac{M}{\cancel{2.4}} + NT_h$$

$$= (5-1)\times 2 + \frac{300\times 8}{2400} + 5\times 0.2$$

$$= 10 \text{ second}$$

//

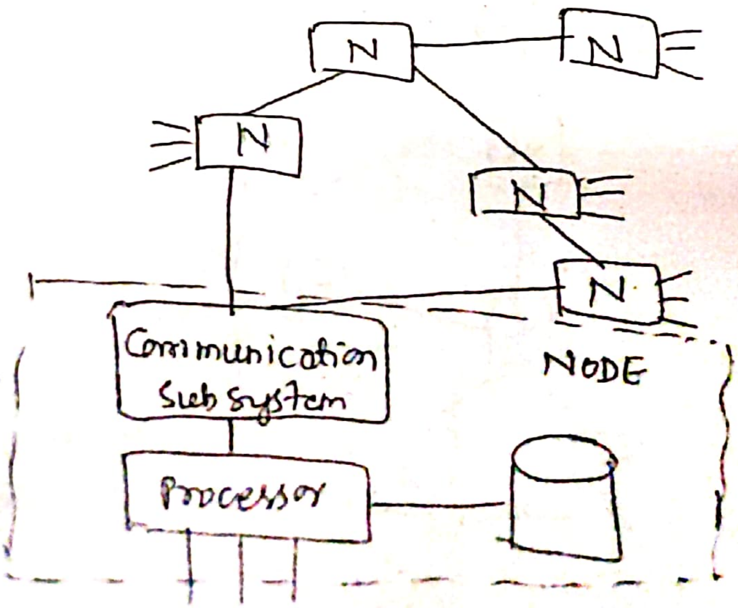
(2) Message Switching:

→ It is often called store and forward switching.

→ It accepts messages from the originating nodes, store the messages and then forward each messages to the next node or destination, when circuit becomes available.

eg. postal system, telex.

→



→ There is usually some delay in message delivery.
→ It uses efficiently, the expensive transmission links.
→ A well designed message switching ~~sw~~ system keeps a uniform load throughout the time.
→ In a message-switching centre an incoming message is not lost when the required outgoing route is busy.
→ It is stored in a queue with any other messages for the same route and retransmitted when the required circuit becomes free.
→ Message switching is an example of <u>delay system</u> or a <u>queuing system</u>.
→ The node processor, in case of message switching performs the following functions:-

(I) Receive the full user message & store the same

(II) Check message for data transmission errors and perform error recovery if required

(III) Determine the destination address from the user message

(IV) Choose an appropriate link towards destination based on certain routing criterion.

(V) Forward the message to the next node on the chosen link

→ The transfer of data files between computers results in very long messages.
→ The VDU (Visual Display Unit) operator will not obtain the desired quick response to a message, if it has to wait for the completion of a large file transfer.
→ This problem is solved by dividing long messages into smaller units, known as <u>packets</u>, so modern data communication replaces message switching by packet switching.

③ Packet Switching:

- This type of switching utilizes some of the advantages of circuit and message switching and mitigates some of the disadvantages of both.
- In the packet switching, the long messages are divided into subsets of equal length called "packets". (few Kbytes).
- Every packet contains some control information in its header, which is required for routing and other purposes.

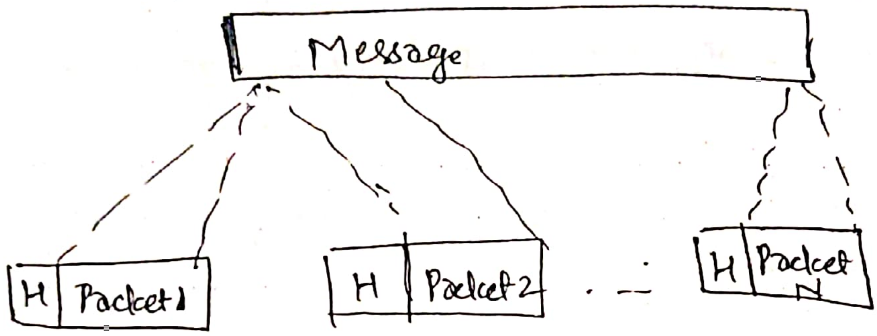


Fig. messages divided into number of equal length short packets.

- In Packet Switching, different messages (and even different packets) can pass through different routes, and when there is a "dead time" in the communication between the source and destination, the lines can be used by other sources.

Header					User
Destination ID	Source ID	message ID	Packet ID	Control	Data

Fig. Packet format

- A packet switch is a message processor.
- Packets are forwarded over optimum routes based on route condition delay and congestion.
- It provides error control and notifies the originator of packet received at the destination.

→ As shown in fig. above 5.1, when the Host Computer, H_1 wants to transfer data to the host computer H_6, a connection request is made to the switching node N_1, which in turn, selects neighbouring node through which the desired connection may be established eg. $N_5 - N_6 - N_7 - H_6$.

→ The path selection is generally based on a routing algorithm that may take into account the network traffic, path length, etc

→ There are three explicit phases involved in circuit switched data transfer :

(I) Circuit Establishment :

- Before any signals can be transmitted, an end-to-end circuit must be established.

(II) Data Transfer:

- The data may be analog or digital, depending on the nature of the network.

(III) Circuit Disconnect:

- After the duration of data transfer, the connection is terminated usually by the action of one of the two stations -

→ In circuit switching, the call cannot be stored, so it is lost.

→ Thus, circuit switching is an example of lost-call system.

Numerical:

$$T_{cs} = T_e + T_t + T_r$$

$$\therefore \quad T_{cs} = (N-1)T_m + \frac{M}{R} + T_h$$

where,

T_e = time for path setup / path establishment.

T_t = time for data transmission.

T_r = time for path release

N = Number of switching node in path

T_m = average route selection time in each node for a given network load.

→ Two basic approaches are commonly used ~~to~~ in packet switching: virtual circuit packet switching and datagram packet switching.

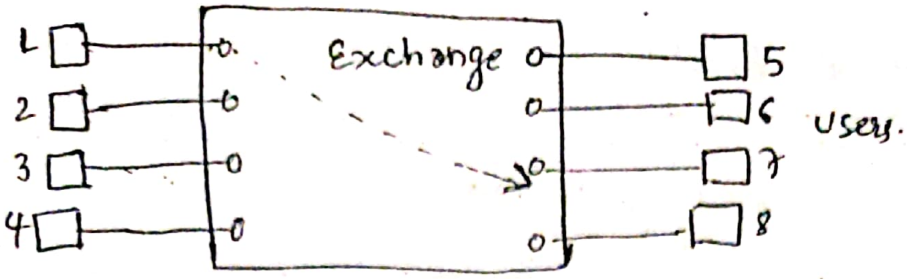
→ In virtual-circuit packet switching, a virtual circuit is made before actual data is transmitted.

→ In datagram packet switching, packets can follow different routes to the destination, and delivery is not guaranteed.

Summary of Data Switching System:

Switching method	Advantages	Disadvantages
Circuit Switching	(i) mature technology. (ii) Near real time connectivity (iii) Excellent for inquiry & response. (iv) leased service attractive	(i) High cost of switch. (ii) Lower Link (Tv) utilization. (iii) Privately owned service can only be justified with high traffic volume
message Switching	(i) Efficient trunk utilization. (ii) Cost-effective for low-volume leased services.	(i) Delivery delay may be faced (ii) Not viable for enquiry & response (iii) Requires large storage buffer
Packet Switching	(i) High efficiency (ii) Approaches near real time connectivity. (iii) Highly reliable, survivable (iv) low traffic volume attractive for leased service.	(i) multiple route & nod network expensive (ii) Processing intensive (iii) Large traffic volume justifies private services.

Manual Switching:



→ In manual switching, connections are setup manually.

→ For example, if User1 is to connect to 7, then manually 1 and 7 are connected in the exchange office

→ Thus, interconnection function is done by jacks by operators, subscriber line jacks and jacks for incoming and outgoing trunks.

→ The cords are always less than half the number of jacks appearing on the board, because one interconnecting cord occupies two two ~~pages~~ jacks as interconnection is made by double ended connecting cords.

→ The operator is alerted by the lamp, when there is an incoming call requiring connection. This is the attending, alerting function.

→ Then after testing the busy-condition, if the outlet are free, the operator connects to the idle destination plug.

Disadvantages of manual Switching:

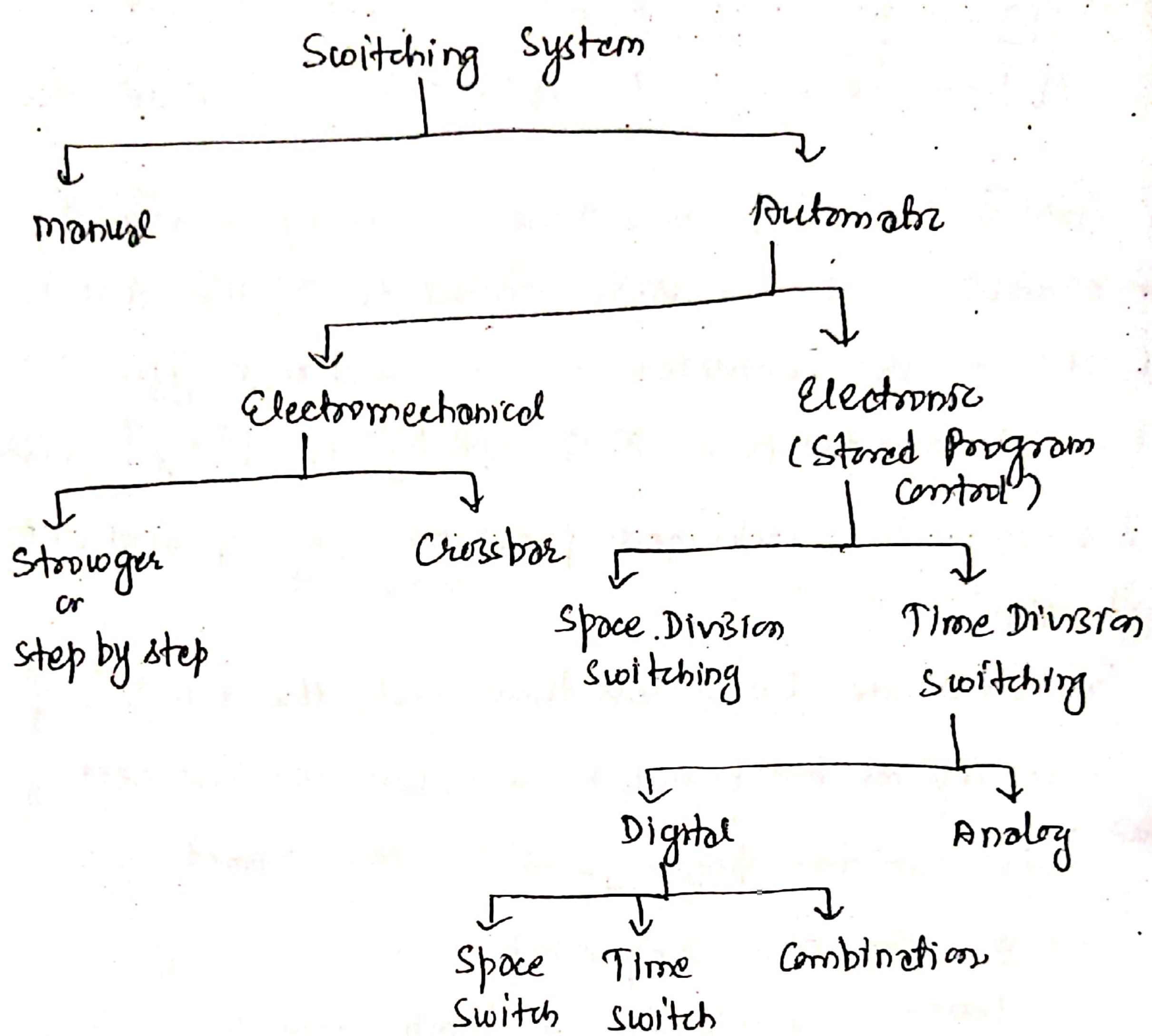
→ In manual exchange, the operator must be communicated to subscriber, before making connection, so, in multilingual areas, common language becomes an important factor.

→ No privacy, as operators supervises every calls.

→ Call establishment and release time are much longer in manual exchanges.

Electromechanical Switching:

→



→ Strowger switching system was the first automatic switching system developed by Almon B. Strowger in 1889.

Advantages of automatic switching:

→ Language independent,

→ Privacy,

→ Call setup is faster

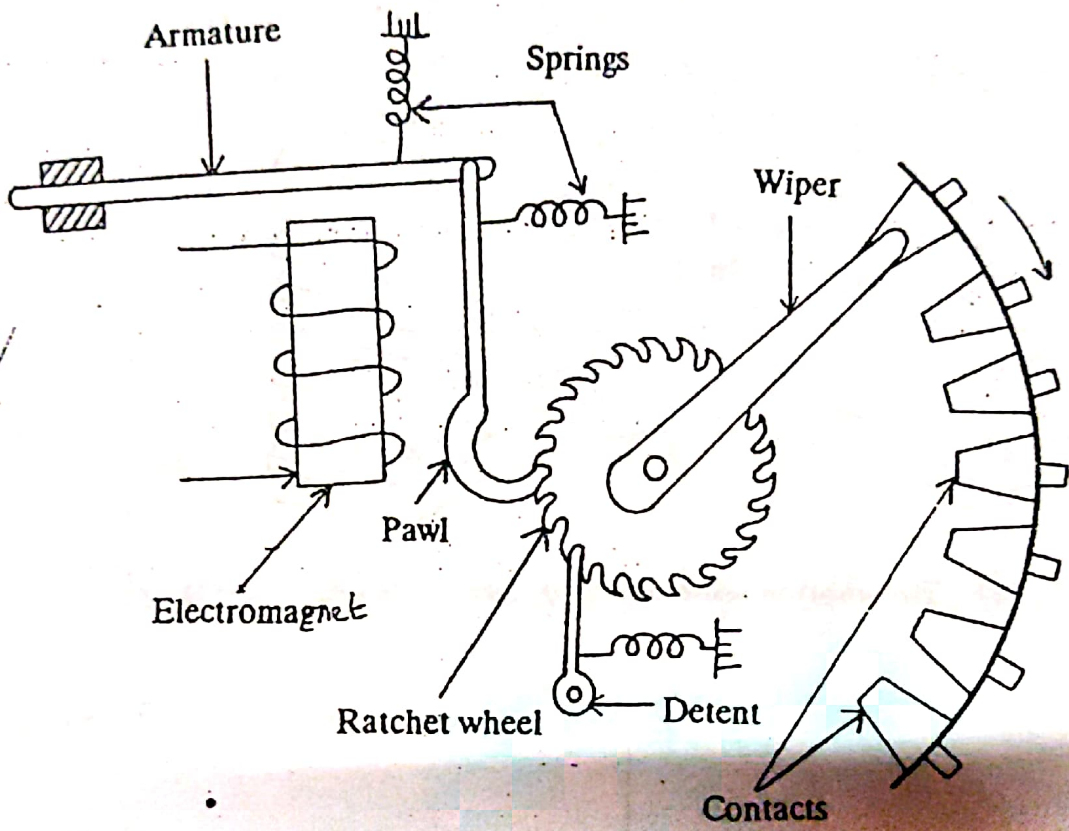
→ time to set/release a call is same

2.3 Strowger Switching Components

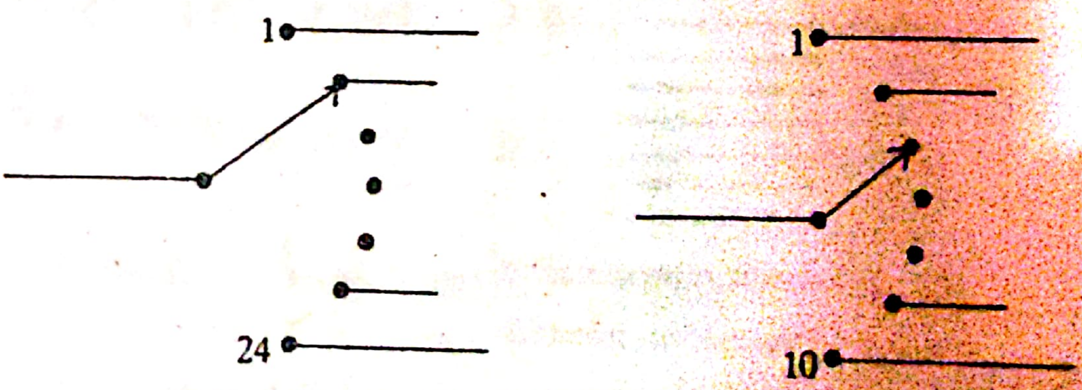
In the Strowger system, there are two types of selectors which form the building blocks for the switching system;

- Uniselector
- Two-motion selector.

These selectors are constructed using electromechanical rotary switches. The drive mechanism of a rotary switch is shown in Fig. 2.5(a).

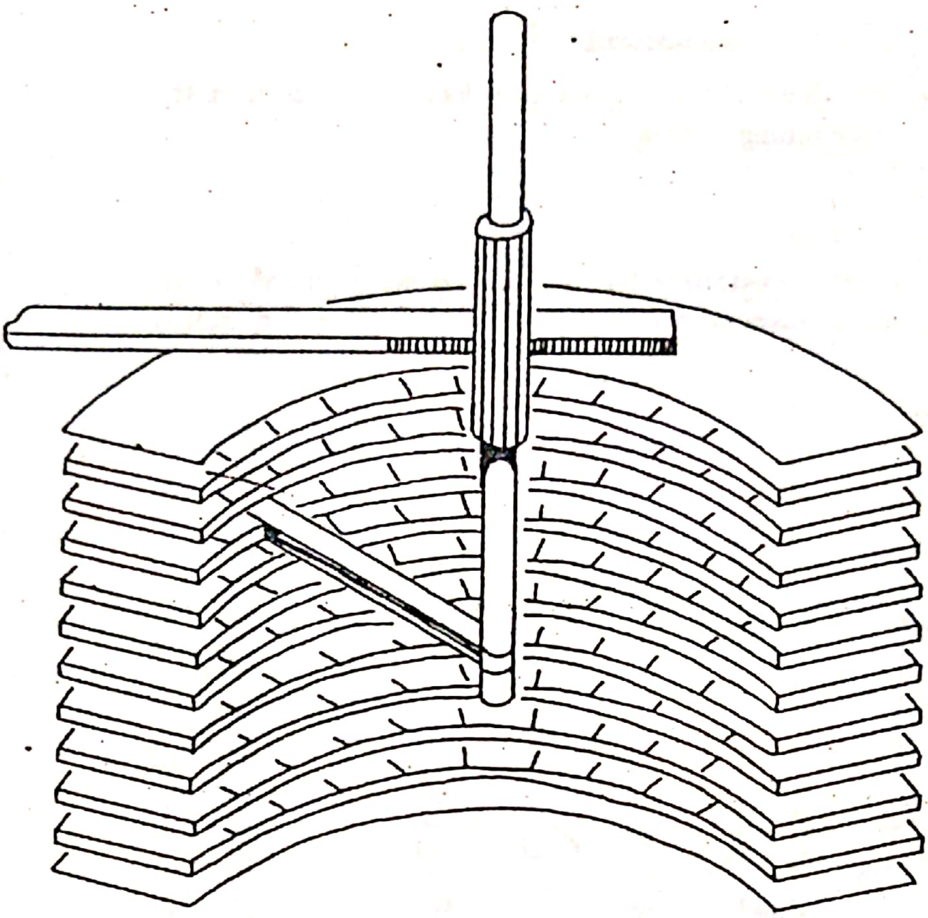


(a) Drive mechanism of a rotary switch

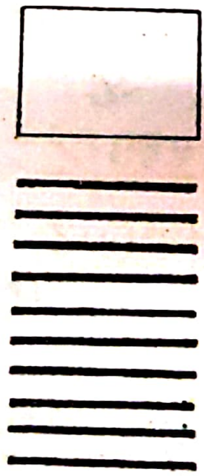


(b) Schematic representation of uniselectors

Fig. 2.5 Uniselectors.



(a) Two-motion selector arrangement



(b) Schematic representation

Fig. 2.6 Two-motion selectors.

net and the second impulse train drives the horizontal rotary switch. In such a case, it follows that the bank contacts are so numbered as to correspond to

→ Whenever the electromagnet is energized, the 'armature' is attracted to it and 'pawl' falls 1 position below the present tooth position.

→ When the electromagnet is de-energized, the armature is released and returns to rest position.

→ During this, the pawl moves the rachet wheel 1 position up which is known as Reverse drive type

→ Thus, if electromagnet is energized and de-energized 5 times by applying 5 pulses, the wiper moves 5 contacts.

Uniselector:

- Has a single rotary switch with a bank of contacts.
- Typically, there are 4 banks of which 3 are used for switching and the 4th one is used for homing
- The switching banks have 25 or 11 contacts each.
- 1st contact in each bank is known as home contact and the remaining as switching contacts
- Homing bank has only two contacts:
    - 1 at 1st position corresponding to home contacts of the other banks, and
    - other extending as an arc from 2nd position to last position.
- Arc contact is often called homing arc
- Depending upon number of switching contacts, uniselectors are identified as 10-outlet or 24-outlet uniselectors.

Two-motion Selector:

- capable of horizontal as well as vertical stepping movement
- Has 2 rotary switches, one providing vertical motion for the wiper assembly,
    - other providing horizontal movement for the wipers.
- The horizontal movement rotary switch of a two-motion selector has a <u>interrupter contact</u>
- In two-motion selector normally there are 11 vertical positions and 11 horizontal contacts in each vertical position.
- The lowest vertical position & first horizontal contact in each vertical level are home positions and the remaining ones are the actual switching positions.
- Thus, the wiper in 2 motion selector has access to 100 switching contacts.

2008

Disadvantages of ~~Switching~~ Strowger Switching

(i) It is too expensive because each time has its own Strowger selector

(ii) Its capacity is limited to 100 lines only,

(iii) Dependent on moving parts and contacts which subject to wear and tear.

③ Step-by-step Switching:

→ May be constructed using uniselectors or two-motion selectors or combination of both.

→ The wiper contacts of these selectors move in direct response to dial pulse or other signals like - off-hook from the subscriber telephone

→

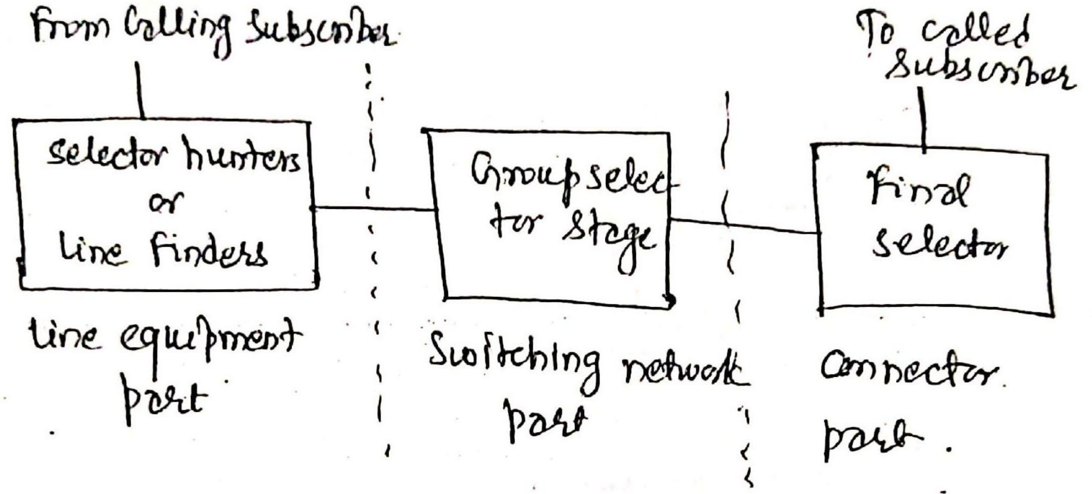


Fig. Configuration of a step-by step switching system

→ In this type of switching system, the wiper steps forward by one contact at a time and moves by as many contacts (takes as many steps) as the number of dial pulses received or as required to satisfy certain signaling conditions. So, the name given as, step-by-step.

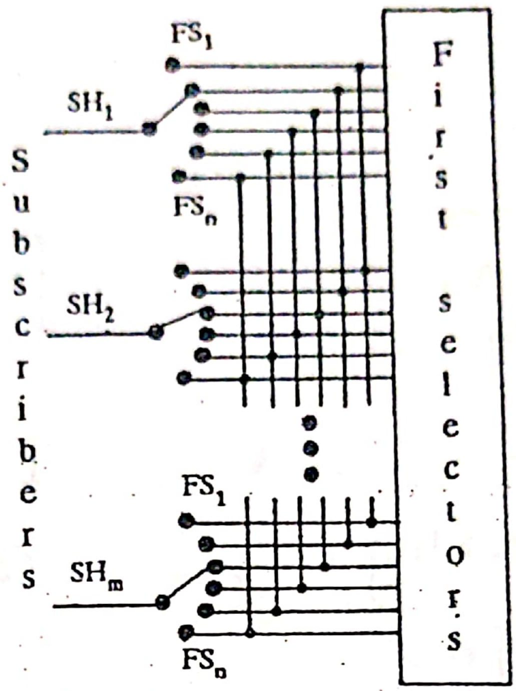
ⓐ Line Equipment part:

→ It consists of selector hunter or Line finder, other two parts consist of selector

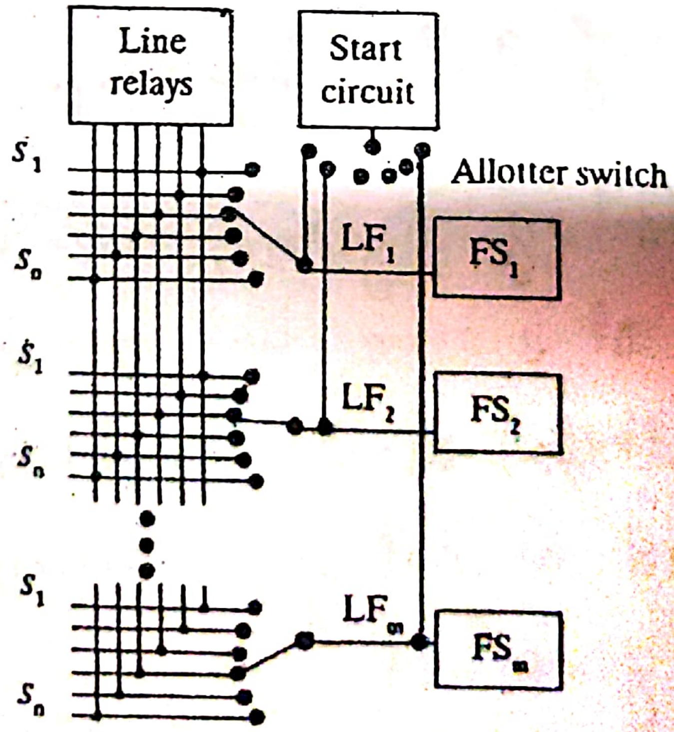
→ Selector Hunter (S.H.) is a way in which a subscriber gain access to common switching resources

→

line finders out of many free line finders, is achieved by means of an allotter switch in the start circuit of the line finder as shown in Fig. 2.8 (b). The circuit arrangements are such that the wiper of the allotter switch normally stands



(a) Selector hunter based access



(b) Line finder based access

FS = first selector LF = line finder SH = selector hunter

Fig. 2.8 Subscriber access to Strowger switching system.

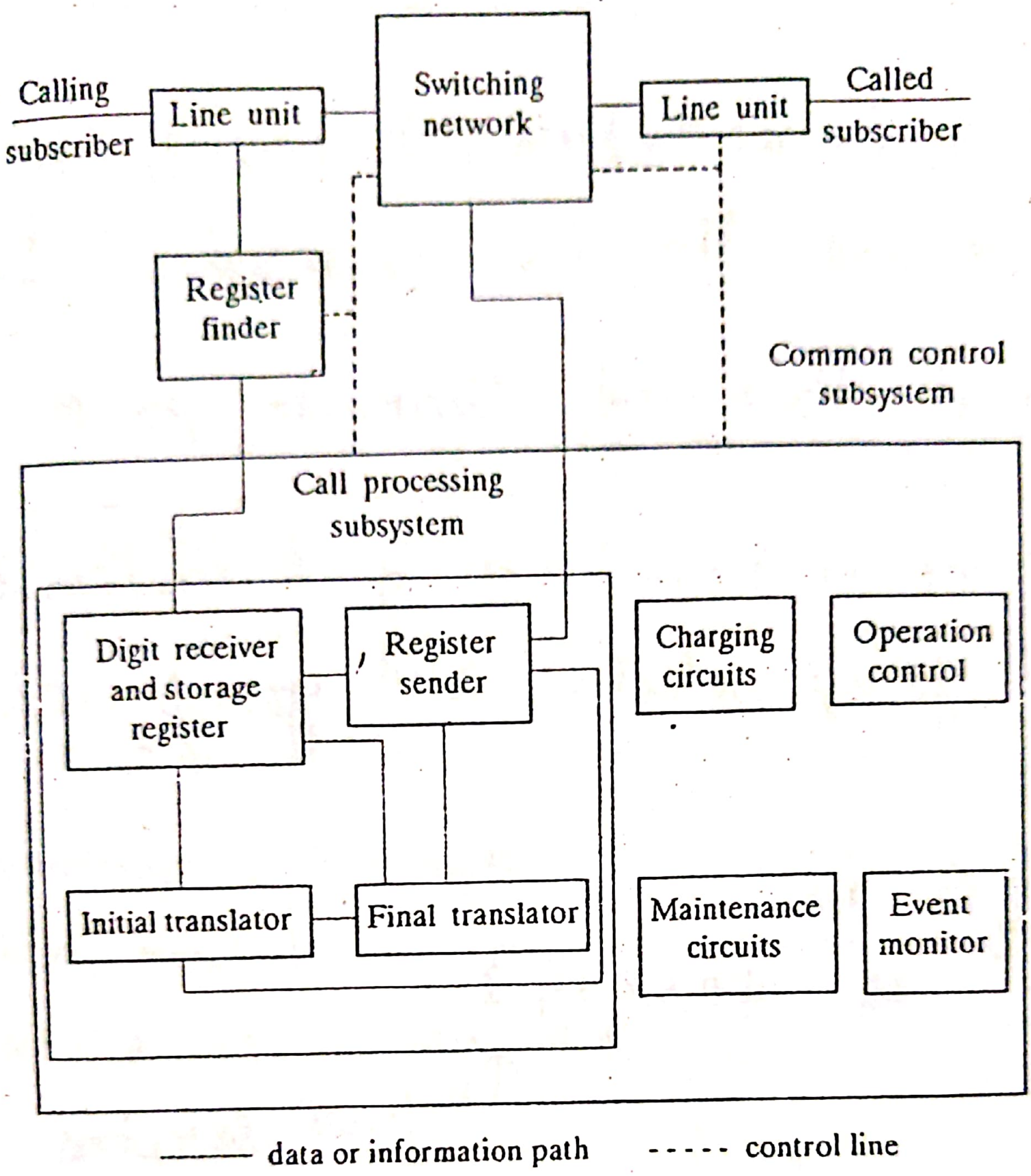


Fig. 3.2 Common control switching system.

relays which initiate control action. The control subsystem may operate relays in the junctors, receivers/senders and the line units, and thus command these units to perform certain functions. Event monitoring may be distributed. For example, the line units themselves may initiate control actions on the occurrence of certain line events.

When a subscriber goes off-hook, the event is sensed, the calling location is determined and marked for dial tone, and the register finder is activated to seize a free register. Identity of the calling line is used to determine line category and the class of service to which the subscriber belongs. As mentioned in Section 2.1, a subscriber telephone may use either pulse dialling or multifrequency dialling, and the line is categorised based on this. A register appropriate to the line category is chosen, which then sends out the dial tone

→ When a subscriber lifts his hand set, the interrupt mechanism in his selector hunter gets activated and the wiper ~~set~~ steps until a ~~fir~~ free first group selector is found at the outlet

→ The status of first group selector free or busy, ~~it~~ is known by a signal in one of the bank contacts of S.H.

→ Once a free F.S. (First Selector) is found, the interrupter is disabled and the F.S. is marked busy.

(b) Switching Network Part and Connector Part:

→ It consists of one or more sets of 2 motion selector known as 1st group selector, 2nd group selector and so on.

In 7 digit no.

$$\text{eg } \underbrace{40327}_{\text{group selector}}\underbrace{41}_{\text{final selectors.}}$$

→ When subscriber starts dialing, the first selector cuts off the dial tone and receives pulse trains.

→ Its wiper assembly steps vertically as many steps as the number of dial pulses

→ The wiper then moves in horizontal plane to which free 2nd group selector is connected

→ Each group selector performs in similar fashion.

→ The last 2 digits of detailed number are processed by final selector.

- It responses to the last 2 digits for the movement of vertical and horizontal wipers respectively.
- If the subscriber is free, then F.S. sends out ringing current to the called subscriber and ringing tone to calling subscriber
- If busy, the final selector (F.S.) sends out a busy tone to calling subscriber.

(3) Cross-bar Switching:

- Major disadvantage of Strowger switching system is its dependence on moving parts and contacts that are subject to wear and tear
- Such mechanical systems require regular maintenance & ~~terminating a connection~~ adjustment and for this purpose they must be located in places that are easily & speedily accessible by skilled manpower
- This demerits led to Crossbar switching.
- The crossbar switching was introduced with small mechanical motions.
- The basic idea of crossbar switching is to provide 'n×m' set of contacts with only 'n+m' activators or less to select one of the 'n×m' sets of contacts
- There is an array of horizontal & vertical wires shown by solid lines in fig.

3.3 Principles of Crossbar Switching

The basic idea of crossbar switching is to provide a matrix of $n \times m$ sets of contacts with only $n + m$ activators or less to select one of the $n \times m$ sets of contacts. This form of switching is also known as coordinate switching as the switching contacts are arranged in a xy-plane. A diagrammatic representation of a crosspoint switching matrix is shown in Fig. 3.6. There is an

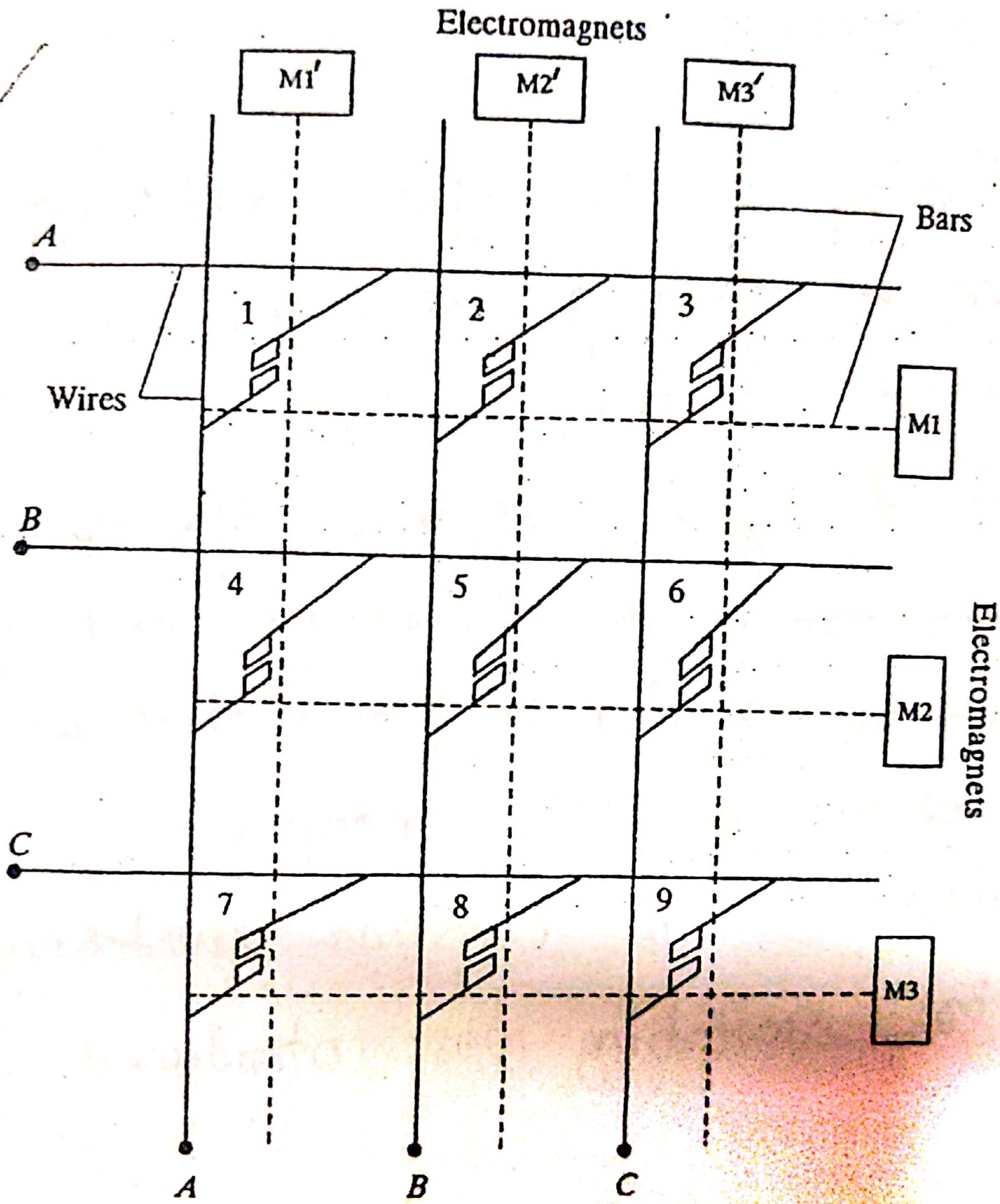


Fig. 3.6 3 × 3 crossbar switching.

array of horizontal and vertical wires shown by solid lines. A set of vertical and horizontal contact points are connected to these wires. The contact points form pairs, each pair consisting of a bank of three or four horizontal and a corresponding bank of vertical contact points. A contact point pair acts as a crosspoint switch and remains separated or open when not in use. The

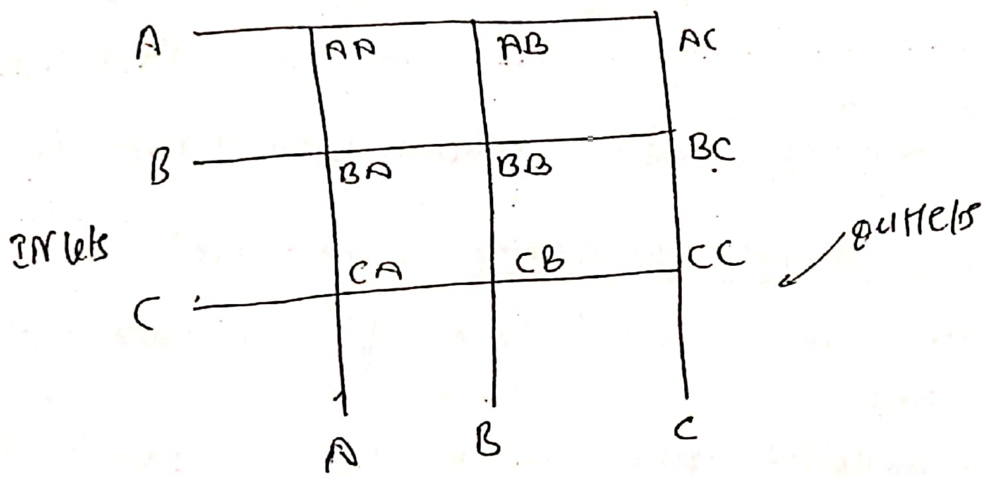
→ Contact point are mechanically mounted on bars as shown by dotted lines.

→ The contact point pair act as a cross point switch & remains open when not in use.

→ when an electromagnet, say in horizontal direction is energised, the bar attached to it slightly rotates and move closer to its facing contact point but donot actually make any contact.

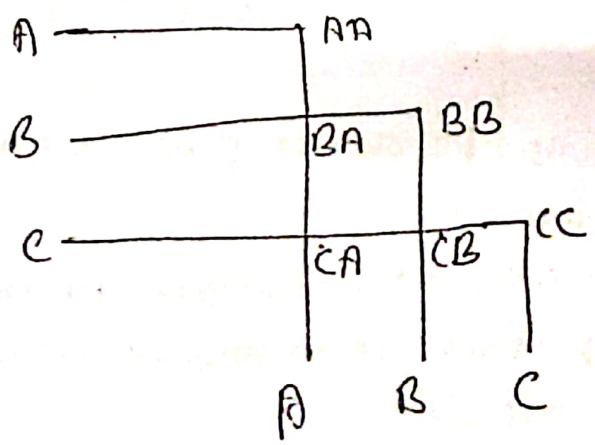
→ Now if electromagnet in vertical direction is energized; the corresponding bar rotates causing the contact point at the intersection of 2 bars too close.

eg: if m_2 & m_3' are energised, contact is established at ~~corre~~ crosspoint 6 such that subscriber B is connected to subscriber C'.



→ when all subscriber are energized only N/2 switches are used.

→ this can be replaced diagonal crossbar switching.



1000 × 10000

≈ 1,000,000

1 million

→ $N^2/2$

→ $\frac{N(N-1)}{2}$

- In space division switching the path in the ckt. are separated from one another spatially.
- Crossbar switching is eg of single stage space division switching.

Limitations of Space division Switching

- The major limitation is the no. of crosspoint required.
- To connect n i/ps to m o/ps we require n x m crosspoint.
- For eg: To connect 1000 i/p & 1000 o/p 1000000 crosspoints are required & only 25% crosspoints are in used at any time. Rest are idle.

Thus the system is inefficient.

- In digital Tx, sampled values of speech are sent as PCM signals with 8 KHZ sampling rate & occurs every 125 us.
- In space division, the dedicated cross point is maintained throughout the communication.
- As a result, switching element remains unused most of time.

2018 For this, time division switching is introduced.

In this, switching is introduced to no. of inlet - outlet - pair for a few us each.

- Thus, a single switching element can be used to transmit speech samples from a no. of inlets to corresponding outlets.
- Thus Time division switching is superior to space division.

2018/ Time Division Switching / Time Slot Interchange (TSI)

- Time division switching uses time division multiplexing (TDM) inside a switch.
- It involves sharing of cross points for shorter period of time.
- A time slot represents voice channel.
- The time slot interchange involves moving the data contained in each time slot from the incoming bit stream to an outgoing bitstream according to the destination of each time slot.

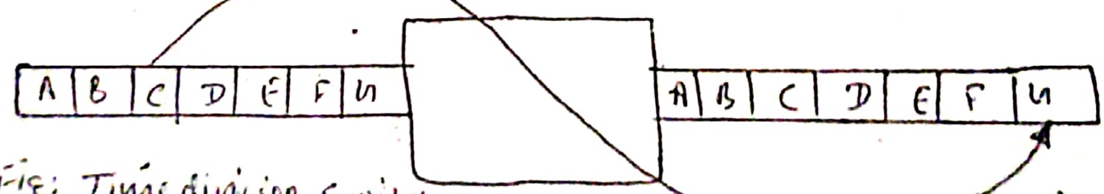


Fig: Time division switch showing connectivity from user C to user G

Space division switching

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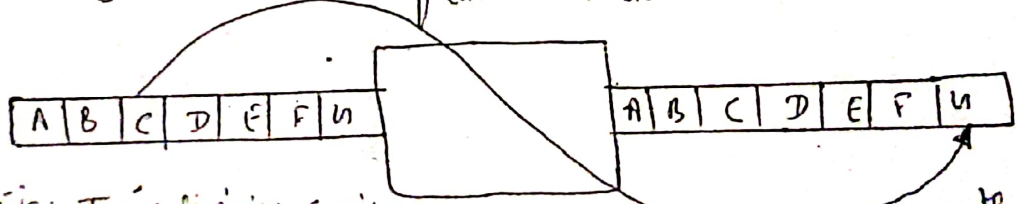


Fig: Time division switch showing connectivity from user C to user G.

i) sequential write, random read
ii) Random write, sequential read

i) Sequential Write, random read (2005)

→ The time slots are written into memory as they appear in incoming bit-stream. They are read out of memory in the correct order for outgoing bit stream as mapped by Control memory.

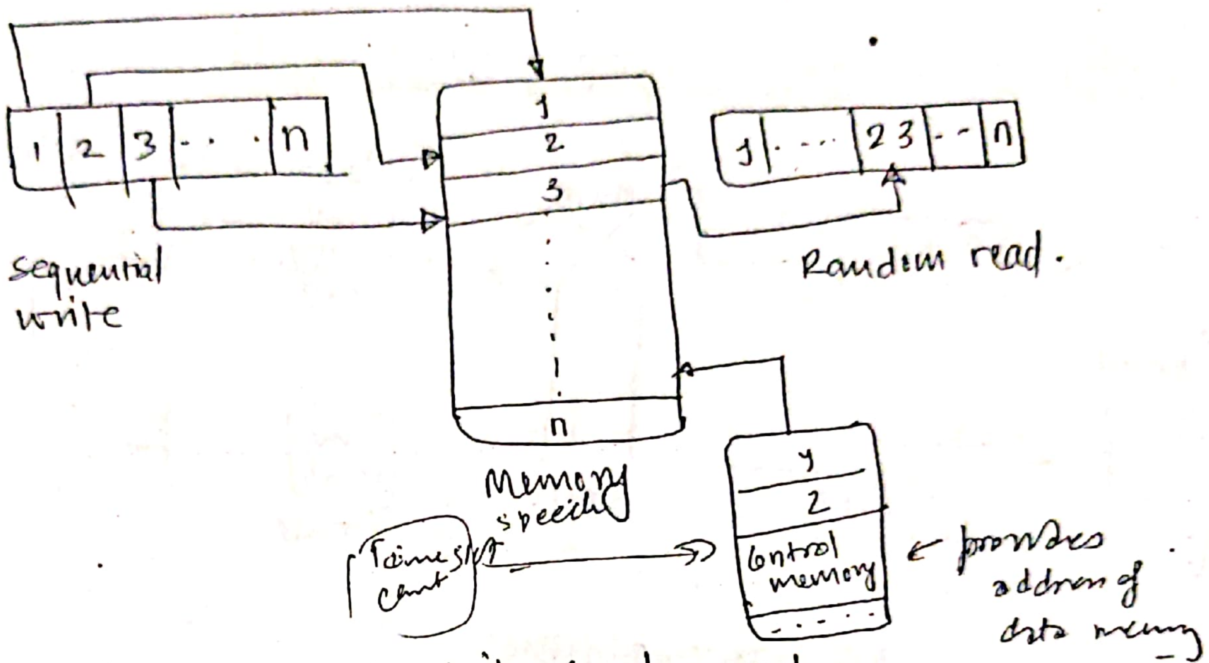


Fig: Sequential Write, Random read.

ii) Random Write Sequential read

→ In random write sequential read, the incoming time slot are written into memory in the order of appearance in outgoing bit-stream.

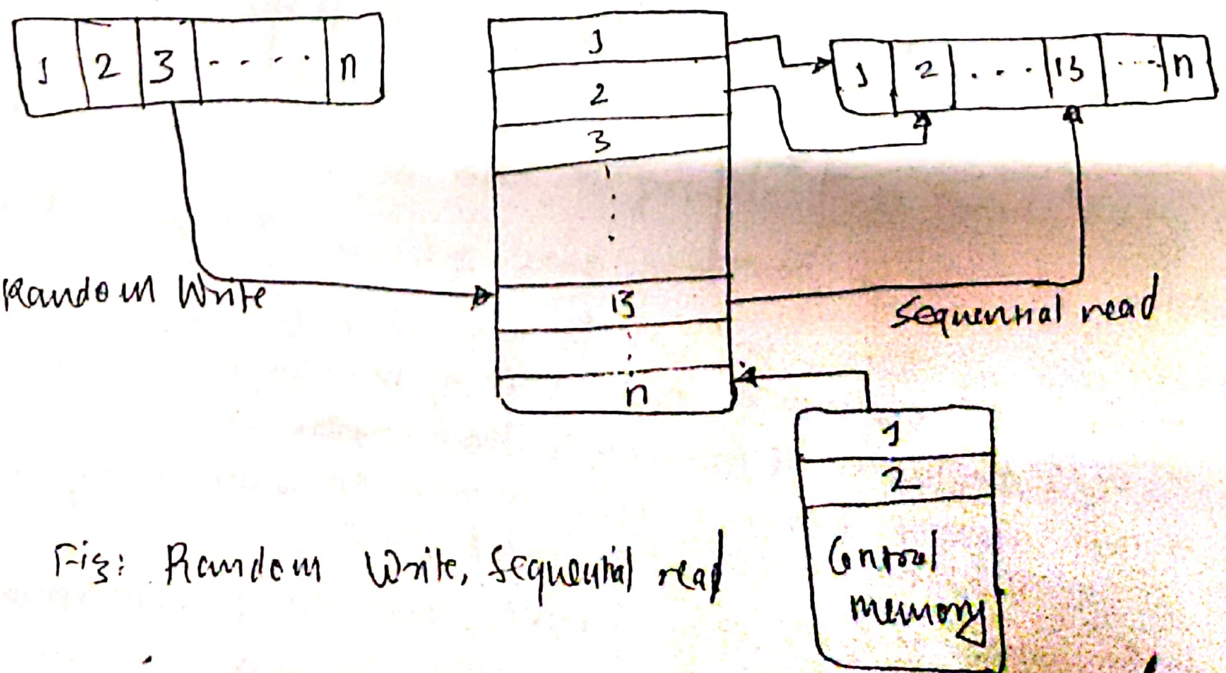


Fig: Random Write, Sequential read

8/ # * Combination Switching / Multistage Switching.

→ Space Switch Can't perform time slot interchange & time Switch are not Capable of switching the values across spatial (Space) domain.

→ Thus a Combination of time & space switches leads to a Configuration that achieves both Capabilities.

→ The basic configuration includes TS, ST, TST, STS, TSSST etc

06/07/ ## Time Space Time switch (TST)

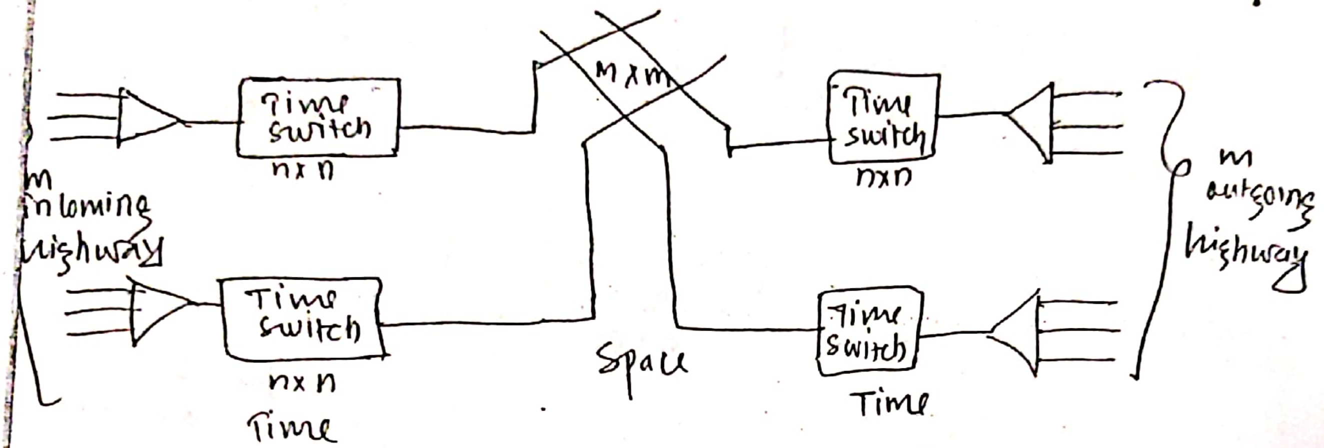
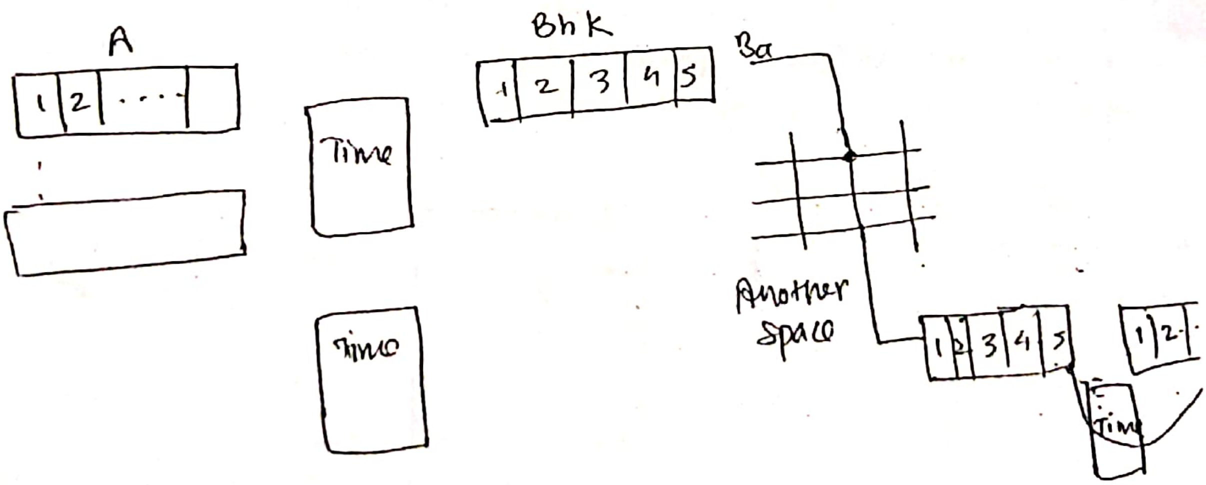


Fig: TST Switch

→ TST switch consists of sequence of time, space & Time switch.



→ In TST switching n/w, each of m incoming & m outgoing PCM highways is connected to time switch.

→ The incoming & outgoing time switches are connected by the space switch.

→ To make a connection between time slot X of incoming highway & time slot Y of outgoing highway, it is necessary

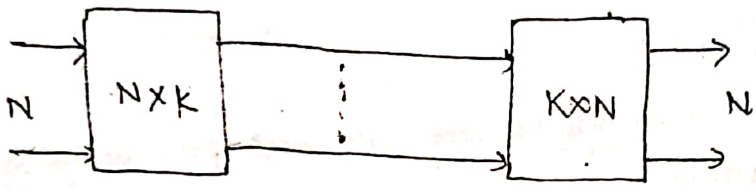
Multi-stage Networks:

Two-Stage Networks:

Theorem:

"For any single-stage telecommunication switching network there exists an equivalent multistage network".

→ A 'N×N' single stage network with a switching capacity of 'K' connections can be realised by a two-stage network of 'N×K' and 'K×N' stages as shown in fig(a) below:



Fig(a) A two-stage representation of an N×N network

→ A connection needs two switching elements.

→ Any of the N inlets can be connected to any of the 'K' outputs of the first stage. Similarly, any of the 'K' inputs of the second stage can be connected to any of the 'N' outlets.

→ As a result, there are 'K' alternative paths for any inlet/outlet pair connection.

→ Thus, the network is said to provide full connectivity or full availability, in the sense that any of the 'N' inlets can be connected to any of the 'N' outlet in the network.

→ For large N, the switching matrix 'N×K' may still be difficult to realise practically. It is necessary to consider architectures that use smaller sized switching matrices.

→ Let us consider the two-stage realisation of an M×N switch using a number of smaller switching matrices as shown in fig below:

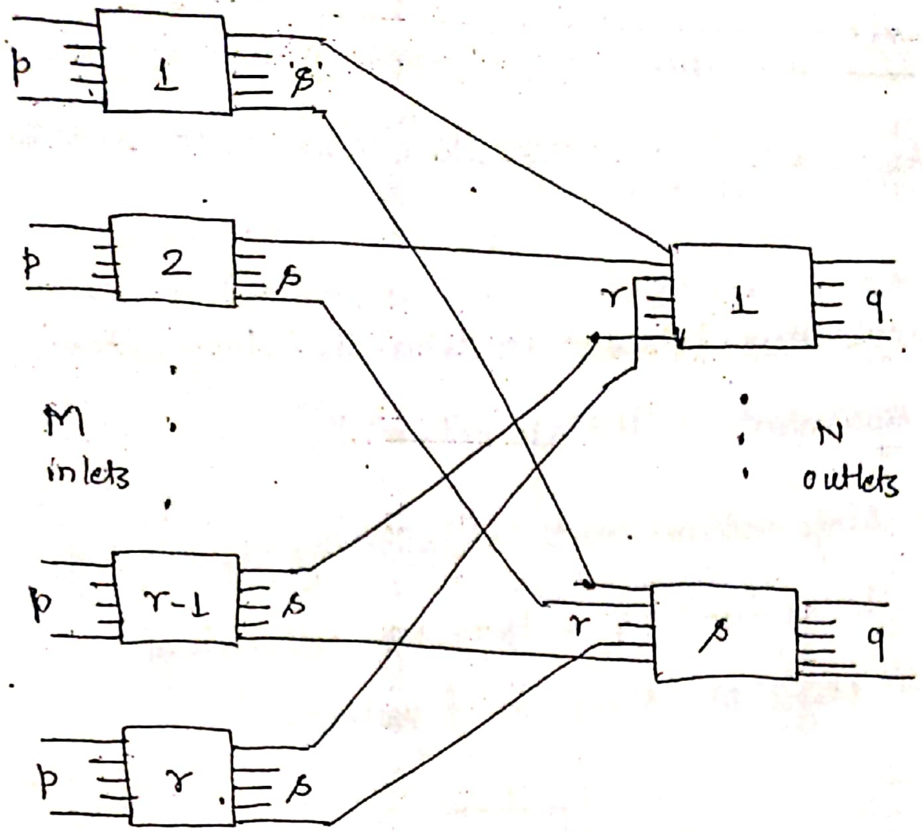


Fig-(b) Two-stage network with multiple switching matrices in each stage.

→ In the fig(b) above, as shown, 'M' inlets are divided into 'r' blocks of 'p' inlets each such that $M = pr$ —— (i)
Similarly, the N outlets are divided into 's' blocks of 'q' outlets, each such that $N = qs$ —— (ii)

→ In order to ensure full availability, there must be at least one outlet from each block in the first stage terminating as inlet on every block of the second stage.

→ This determines the block sizes as '$p \times s$' and '$r \times q$' for the first and second stages respectively.
Thus, total number of switching elements 'S' is given by,

$$S = psr + qrs.$$

From (i) and (ii);

$$S = Ms + Nr$$

→ The switching capacity, SC is equal to the number of links between the first and second stage. i.e, $SC = rs$

This two-stage network is blocking in nature, and the blocking may occur under two conditions:-

(i) The calls are uniformly distributed, and there are '$r \times s$' calls in progress and $(rs+1)$-th call arrives.

(ii) The calls are not uniformly distributed, 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 originates in the I-th block destined to the J-th block.

→ Often, square switching matrices can be used as building blocks for switching networks, modifying above fig (b) architecture as,

$$p = r = s = q = \sqrt{N}$$

→ This new architecture of two-stage network that supports N simultaneous connections but under restricted traffic distribution conditions are known as <u>baseline networks</u>

→ This square network, can be made nonblocking considering $K = \sqrt{N}$, Then, we have, $S = 2N^2$, $SC = N$

→ Thus, two stage non-blocking network requires twice the number of switching element as the single stage nonblocking network.

→ Thus, the purpose of using two-stage network offers no distinct advantage over a single stage network except that it provides 'N' alternative paths for establishing a connection.

→ Further, the real advantages of multi-stage networks become evident, when we consider networks of three or more stages.

Differences between singe stage and multistage networks:

S.No.	Single Stage	Multistage
1.	Inlet to outlet connection is through a single crosspoint.	Inlet to outlet connection is through multiple cross-points.
2.	Use of a single crosspoint per connection results in better quality link.	Use of multiple crosspoints may degrade the quality of a connection.
3.	Each individual crosspoint can be used for only one inlet/outlet pair connection.	Same crosspoint can be used to establish connection between a number of inlet/outlet pairs.
4.	If a crosspoint fails, associated connection cannot be established. There is no redundancy.	Alternative cross-points and paths are available
5.	Crosspoints are inefficiently used. Efficiency of active lines is not used.	Crosspoints are used efficiently.
6.	Number of crosspoints is prohibitive	Number of crosspoints is reduced significantly.
7.	A large number of crosspoints in each inlet/outlet leads to capacitive loading	There is no capactive loading problem
8.	The network is non-blocking in nature	The network is blocking in nature
9.	Time for establishing a call is less.	Time for establishing a call is more

Three - Stage Networks:

- The blocking probability and the number of switching elements can be reduced significantly by adopting a three-stage structure in place of two-stage networks.
- The general structure of an 'N X N' three-stage blocking network is as shown in fig. (c).
- The network is realised by using switching matrices of size 'p X s' in stage 1, 'r X r' in stage 2, and 's X p' in stage 3.
- Here, any arbitrary inlet in the first stage has 's' alternative paths to reach any arbitrary outlet in the third stage.
- Thus, total number of switching elements is given by;

$$S = rps + sr^2 + spr = 2Ns + sr^2 = s(2N + r^2).$$

If we use square matrices, in first and third stages, we have, $p = s = (N/r)$, so,

$$S = \frac{2N^2}{r} + Nr.$$

- The probability graph of three-stage network can be drawn as ~~ether~~ shown in fig (d) to study the probability of alternative paths and thus blocking probability.

n-stage Networks:

- Further, reductions in the number of switching elements is possible by using even higher number of stages than three.
- The typical five-stage switching network as shown in fig. (e), can ~~providing~~ provide switching with less number of switching elements with further reduction of blocking probability.

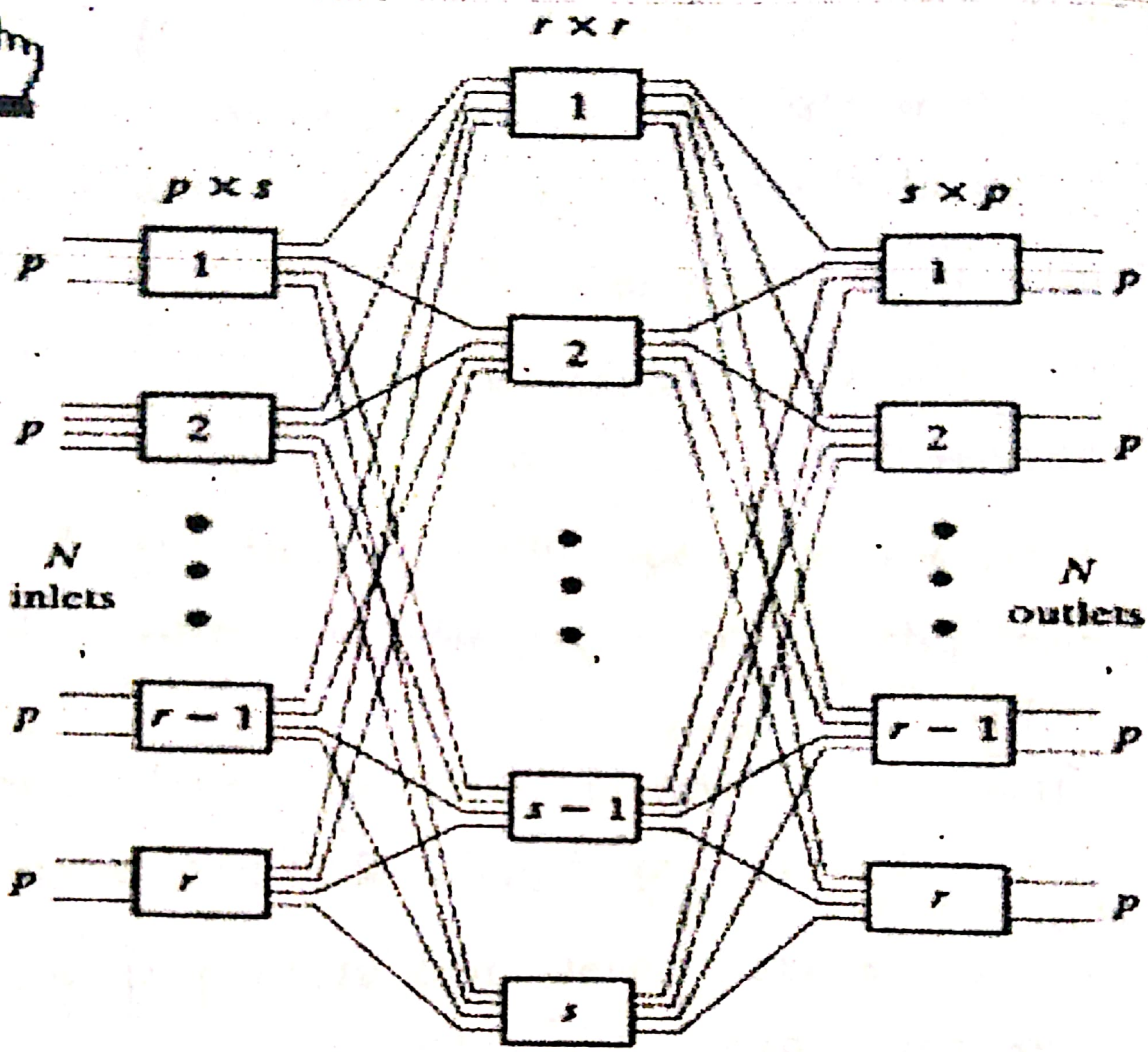


Fig. 4.24 $N \times N$ three-stage switching network.

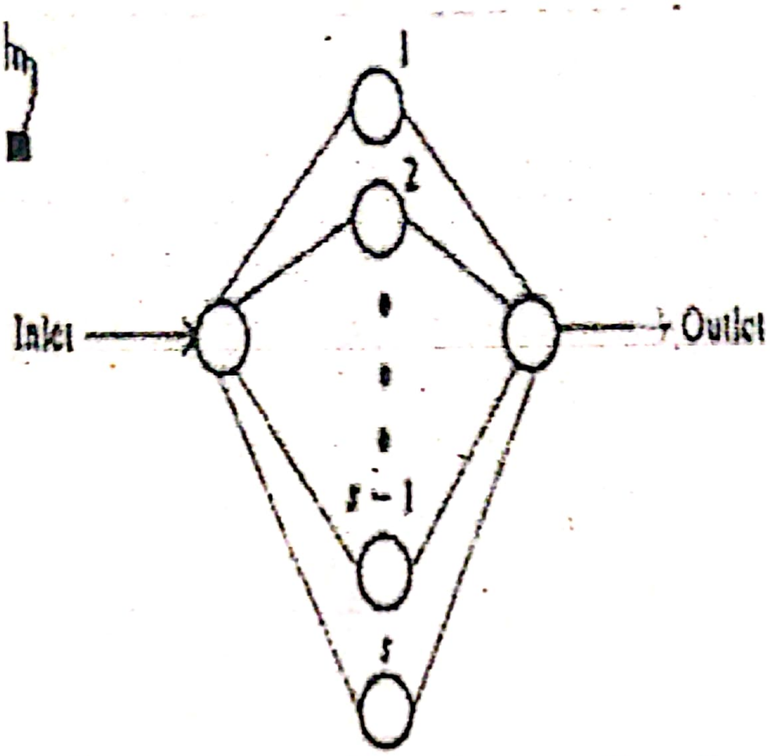


Fig. 4.25 Lee's graph for a three-stage network.

Construction of five-stage ~

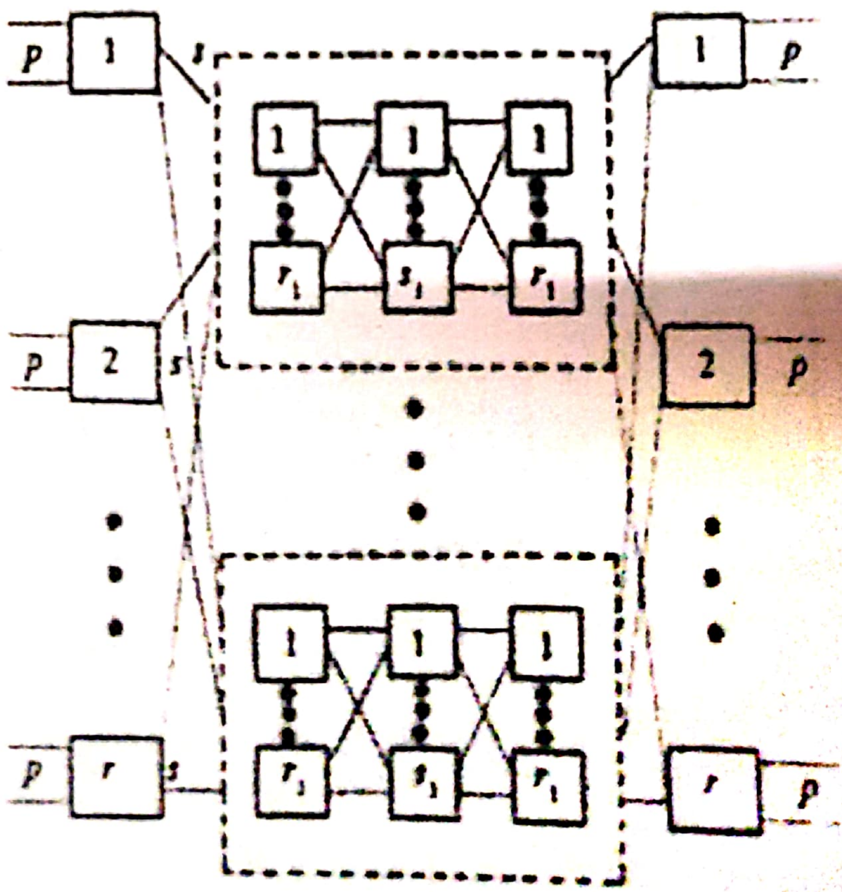


Fig. 4.27 Five-Stage switching network.

it is necessary to choose a time slot Z which is free in Connection store.

→ The connection is established by setting the incoming time switch shift from X to Z, setting the outgoing time switch to shift from Z to Y & operating the appropriate crosspoint at time Z in each frame.

(5 marks) **STS switching (2007)**

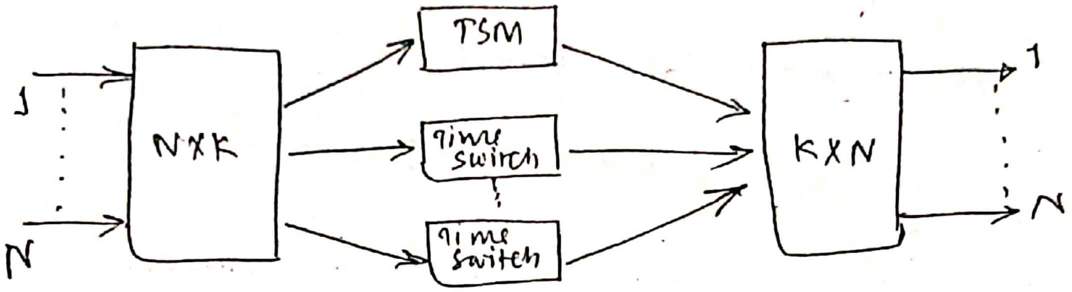


Fig: STS switching structure

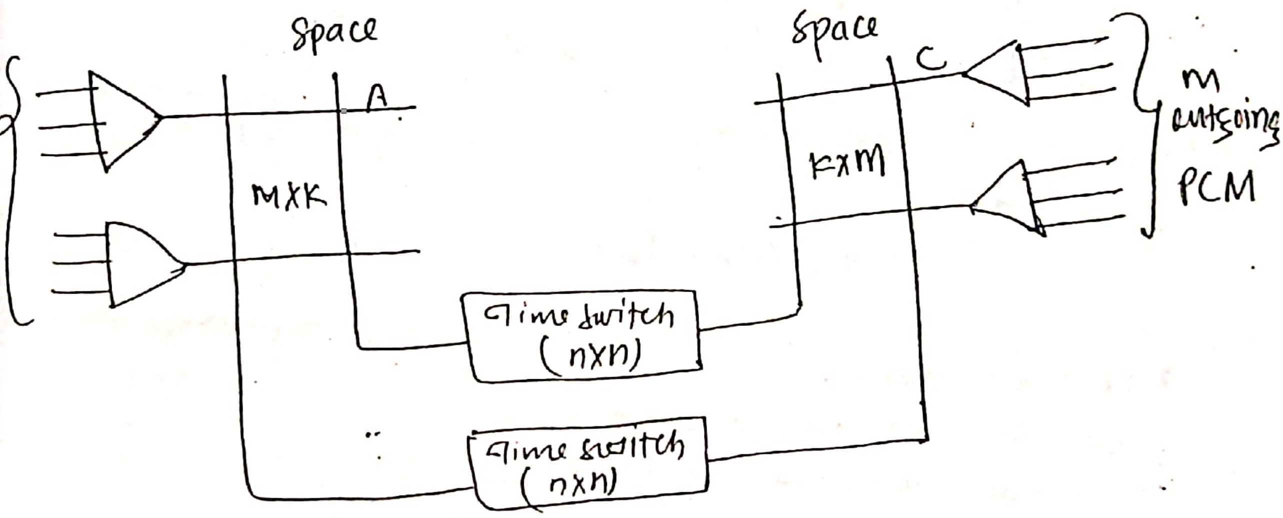


Fig: STS switching Structure

→ The STS switch consists of a space crosspoint matrix at I/P followed by an array of time switch which ultimately fed to the another crosspoint matrix at the O/P.

→ Each m incoming channel is connected to k links by crosspoints in A switch.

→ The other end of the links are connected to m outgoing channel by crosspoints in C switch.

→ Each link contains time switch
→ To make connection between time slot X of incoming channel & time slot Y of outgoing channel, appropriate crosspoint in switch A is selected at time X and appropriate crosspoint in switch C is selected at time Y.

Time switch produce shift from X to Y.

- For small traffic STS is preferred because STS is simple.
- When large amount of traffic is to be handled TST switch have advantage.
- TST is more cost effective.

Time Space Switch (TS switch)

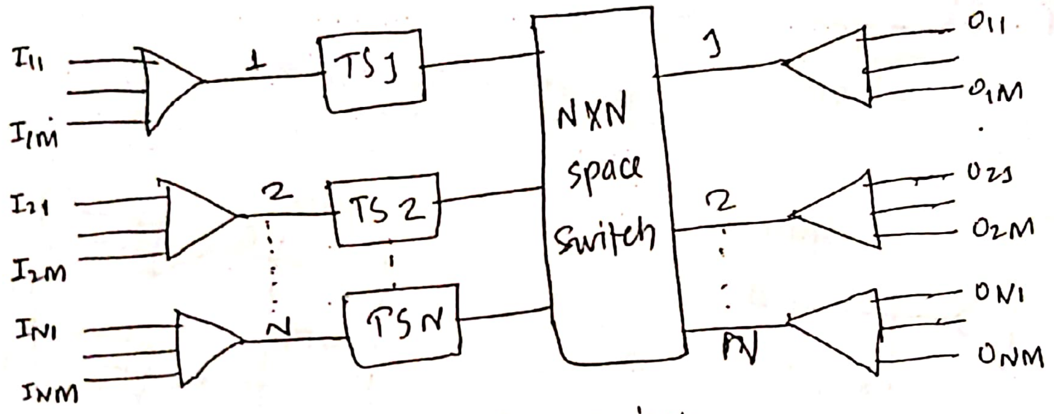
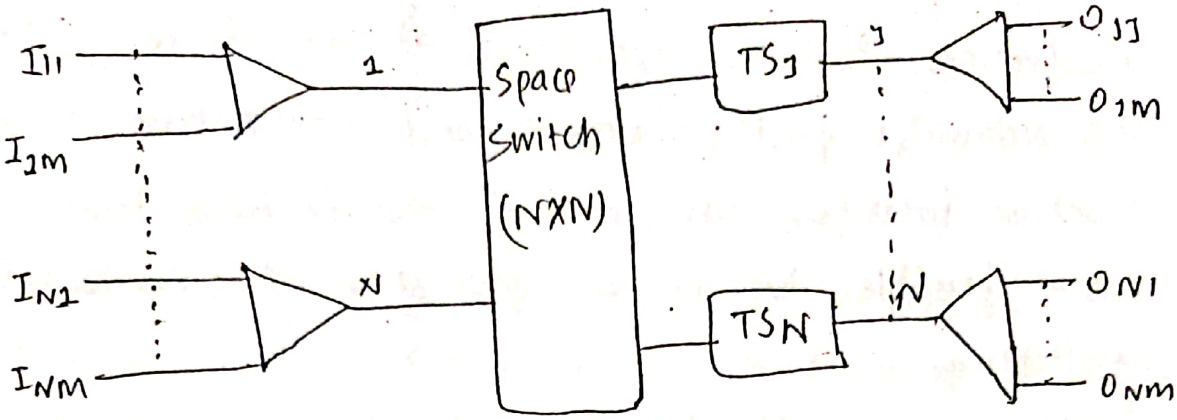


Fig: Time Space switch

→ It is the combination of time switch in first stage and space switch in second stage.
→ Each time multiplexed inlet/outlet stream carries M channels.
→ A subscriber on the i/p side is assigned to one of the inlet & a time slot in that inlet.
→ An i/p subscriber assigned to line 4 at time slot 7 is identified by the label I_{47}.
→ Similarly, a subscriber connected to the outlet 5 and time slot 6 is identified by O_{56}.
→ Suppose if connection is to be established between these 2 subscriber $I_{47} \rightarrow O_{56}$
→ The i/p sample from I_{S47} is first moved to I_{S46} at the o/p of TS_1 switch.

→ During the time slot 6, a connection is established between the inlet and the outlet 5 at the space switch.

Space - Time Switch (ST Switch)



→ In ST switch, the first stage is space switch and then Time switch is followed by space switch.

→ Consider a connection between I_{47} & O_{69}. During the time slot 7, the i/p sample is switch from inlet 4 to outlet 6 by the space switch.

→ It is then switched to time slot 9 by the time switch.

5marks

Private Branch Exchange (PBX)

→ The term PBX refers generically to any switching system owned or leased by a business or organisation to provide both internal switching function & access to public n/w.

→ PBX may use either manual or automatic control.

→ The automatic controlled PBX is K/a PABX

→ Large PBX may use a switching n/w similar to that of a public exchange.

Features:

① Multiple classes of service with priorities and access restriction to area codes.

② Traffic monitoring & analysis to determine the utilization of existing ckts. & blocking probability.

③ Extension lines & exchange lines.

imp Stored Program Control: (SPC):

- Carrying out the exchange control ~~functions for~~ functions through programs stored in the memory of a computer led to the ~~name~~ technology Stored Program Control.
- All common channel signaling (CCS), centralized maintenance and automatic fault diagnosis and interactive human machine interface are some of the features that have become possible due to the application of SPC to telephone switching.
- There are basically two approaches to organizing stored program control: (a) Centralized, and (b) Distributed

(a) Centralized SPC:

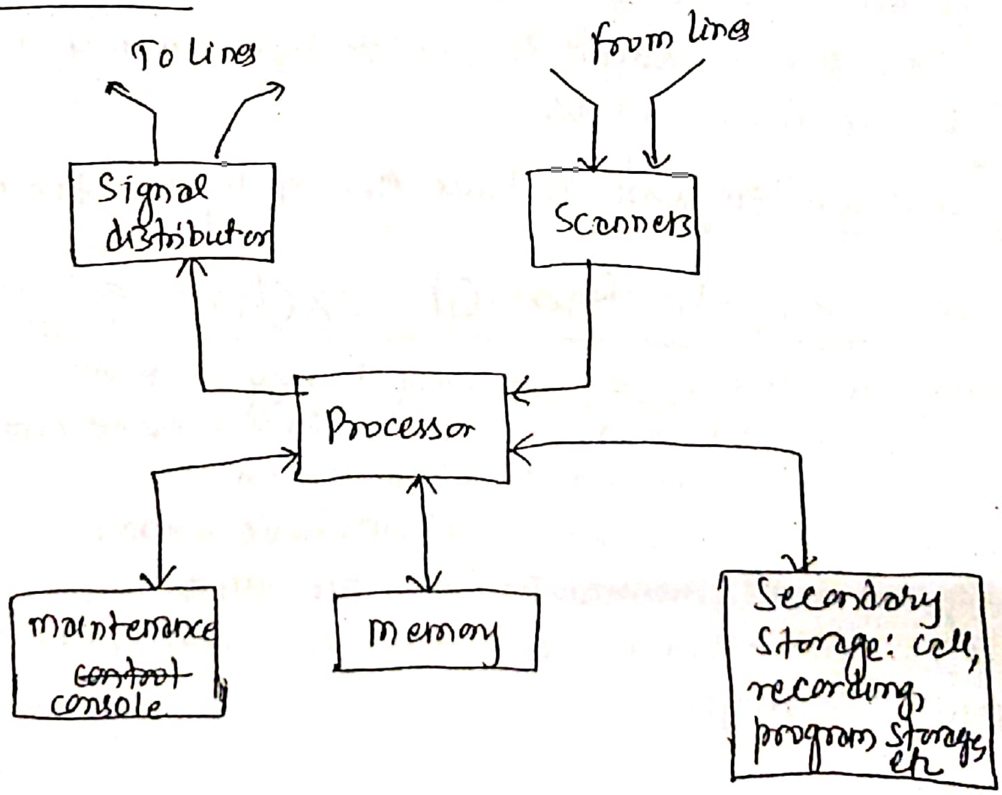
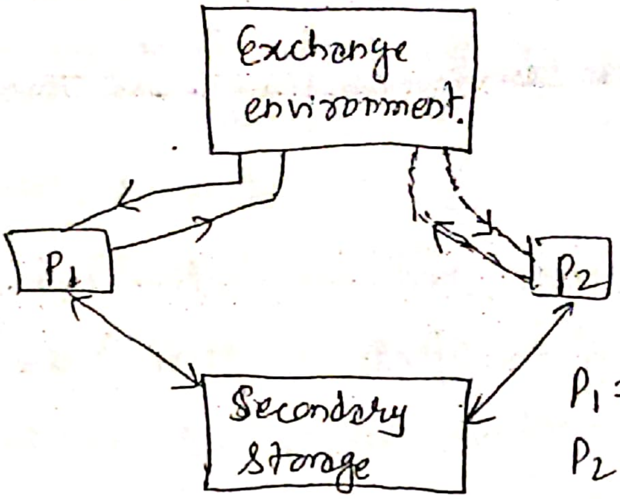


Fig. Typical ~~SPC~~ centralised SPC organization

- Here a program or a set of instructions to the computer is stored in its memory and the instructions are executed automatically one by one by the processor.
- Common Channel Signaling (CCS), centralized maintenance & automatic fault diagnosis, and interactive human machine interface are some of the features that have become possible due to the application of SPC to switching system.
- In centralized control, all the control equipment is replaced by a single processor which must be quite powerful.
- It must be capable of processing 10 to 100 calls per second depending on the load on the system, and simultaneously performing many other ancillary tasks.
- In almost all the present day electronic switching systems using centralized control, only a two processor configuration is used
  - A dual processor architecture may be configured to operate in one of three modes:
    1. Standby mode,
    2. Synchronous duplex mode,
    3. Load Sharing mode

(I) Stand by mode:

- It is the simplest of dual processor configuration operations.
- Normally, one processor is active and the other is on standby, both hardware & software wise



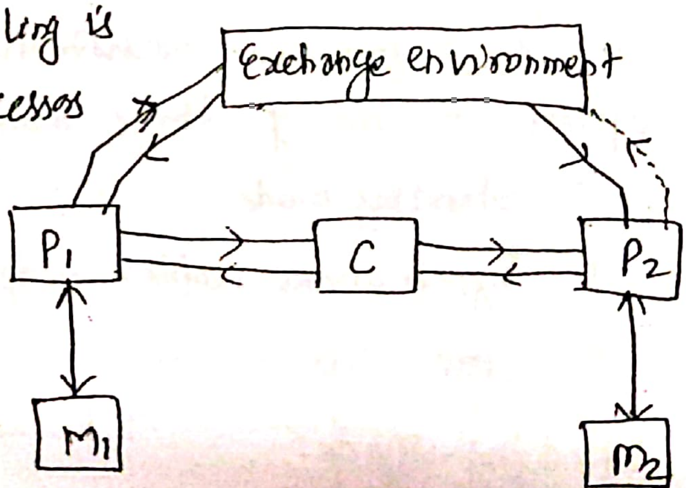
P_1 = active processor
P_2 = Standby processor

Fig. Standby dual processor configuration.

- The standby processor is brought ~~only~~ online only when the active processor fails.
- Here, the active processor copies the status of the system periodically, say every 5 seconds, into a secondary storage
- When a switch-over occurs, the online processor loads the most recent update of the system status from the secondary storage & continues the operation.

② Synchronous Duplex mode:

- In this mode, hardware coupling is provided between the two processors which execute the same set of instructions and compare the results continuously.
- If the mismatch occurs, the faulty processor is identified and taken out of the service within a few milliseconds.

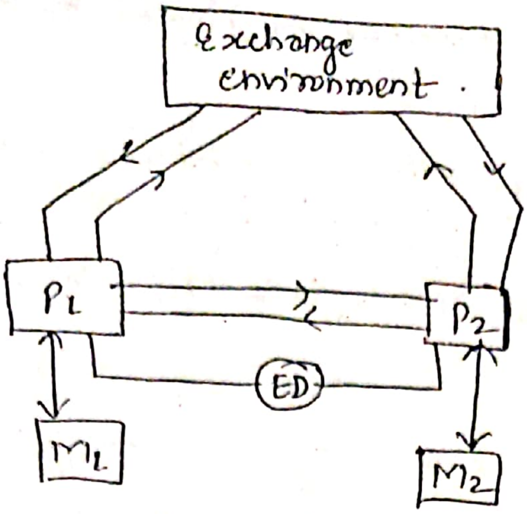


C = Comparator, M = memory, P = Processor

Fig: Synchronous Duplex operation

- When the system is operating normally the two processors have the same data in their memories at all times & simultaneously receive all data from exchange environment
- One of the processor actually controls the switch.

③ Load Sharing mode:



ED = Exclusion Device

Fig: Load Sharing Configuration

- In this mode, an ~~car~~ incoming call is assigned randomly or in a predetermined order to one of the processors which then handles the call right through completion.
- Thus, both the processors are active simultaneously and share the load and the resources dynamically.
- As shown in fig., both the processors have access to the entire exchange environment which is sensed as well as controlled by these processors.
- Since, the calls are handled independently by the processors, they have separate memories for storing temporary call data.
- There is an interprocessor link through which the processors exchange information needed for mutual coordination and verifying the 'state of health' of the other.
- If the exchange of information fails, one of the processors which detects the same takes over the entire load including the calls that are already set up by the failing processors.
- Under normal condition, each processor handles one-half of the calls on a statistical basis.
- One of the main purposes of redundant configuration is to increase the overall availability of the system.

(2) Distributed SPC:

- In this SPC, the control functions are shared by many processors within the exchange itself.
- This type of structure owes its existence to the low cost microprocessors.
- This structure offers better availability and reliability than the centralized SPC.
- 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 the processor that performs all control functions related to that block of equipments.
- In horizontal decomposition, each processor performs only one or some of the exchange control functions.

Digital Cross Connect:

- One of the first electronic device to be tried out as a crosspoint is the cold cathode diode
- This was soon abandoned because of the practical difficulties in implementation and inadequate transmission characteristics.
- With the advances in the semiconductor, technology transistorised crosspoints were developed in the 1960's.
- They offered better performance than reed relays (electromagnetic relay) at that time, but were not economically competitive
- With the advent of IC, many ~~to~~ PABX were designed using IC crosspoints.

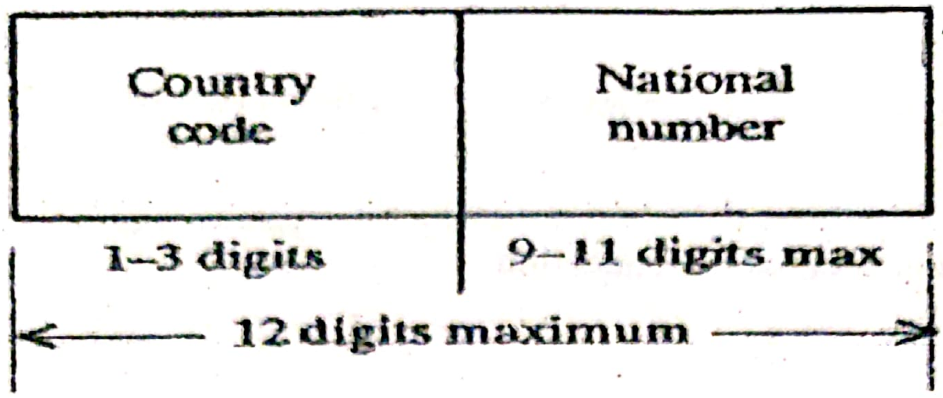
Numbering Plan:

- **National numbering plan** – Subscriber Trunk Dialling (STD) or direct distance dialing (DDD) for intercity or intertown, where multiexchange area are identified by unique code.

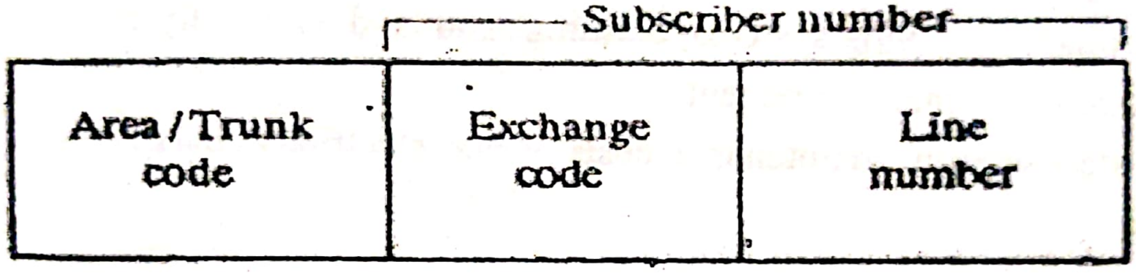
- **International numbering plan** –International Subscriber dialing (ISD).

- Numbering plan may be – Open (e.g. non-Director Strowger Switching), Semi-open (number length differ by 1 or 2 digit), Closed (number of digits in a subscriber number is fixed)
- An **international numbering plan** or **world numbering plan** defined by CCITT in its recommendations E.160 – E.163
- For numbering purposes, the world is divided into 9 zones.
- Each zone is given a single digit code (1-9).
- Since first digit of country codes can be any 1-9, **0** prefix is used to distinguish between a local call and national call.
- A two digit '**00**' or three-digit '**010**' prefix is used to differentiate between national and international calls, First 0 routes call to trunk exchange and following digits to International Gateway Exchange.
- Special services like, fire, ambulance, police, etc. are given short subscribers numbers in the range 100-199.(112, 911 emergency call number in Nepal)
- Remote maintenance and maintenance exchanges are identified by number range 900-999.
- Zone Code table:

Codes	Zones
1	North America (US, Canada)
2	Africa
3-4	European Countries
5	Latin America
6	Australia
7	Russia
8	East Asia, Japan
9	South East Asia (e.g.Nepal, India, etc.)



(a) International telephone number



(b) National telephone number

Fig. 9.27 Telephone number structure.

Charging Plan:

- A charging plan for a telecommunication service levies mainly 3 different charges on a subscriber:
    - An initial charge for providing a network connection.
    - A rental or leasing charge
    - Charges for individual calls made
- Capital costs and operating cost
- Capital cost include line plant, switching systems, buildings and land. Generally covered in initial connection charge and rental component.
- Operating costs include staff salaries, maintenance costs, water, electricity charges and miscellaneous expenses.
- Metering, count pulses
- Charging methods for individual calls fall under two broad categories:
    - Duration independent charging (e.g. local calls)
    - Duration dependent charging
- Coin operated box, special signaling provisions between exchange and coin box.

Compiled By: Ramu Pandey
SP, NCIT

Chapter-5

Signaling in telephone networks:

2 marks

Signaling System:

(1) → Signaling system links the variety of switching systems, transmission system and subscriber equipments in a telecommunication network to enable the network to function as a whole

Signaling functions:

mainly 2 types:

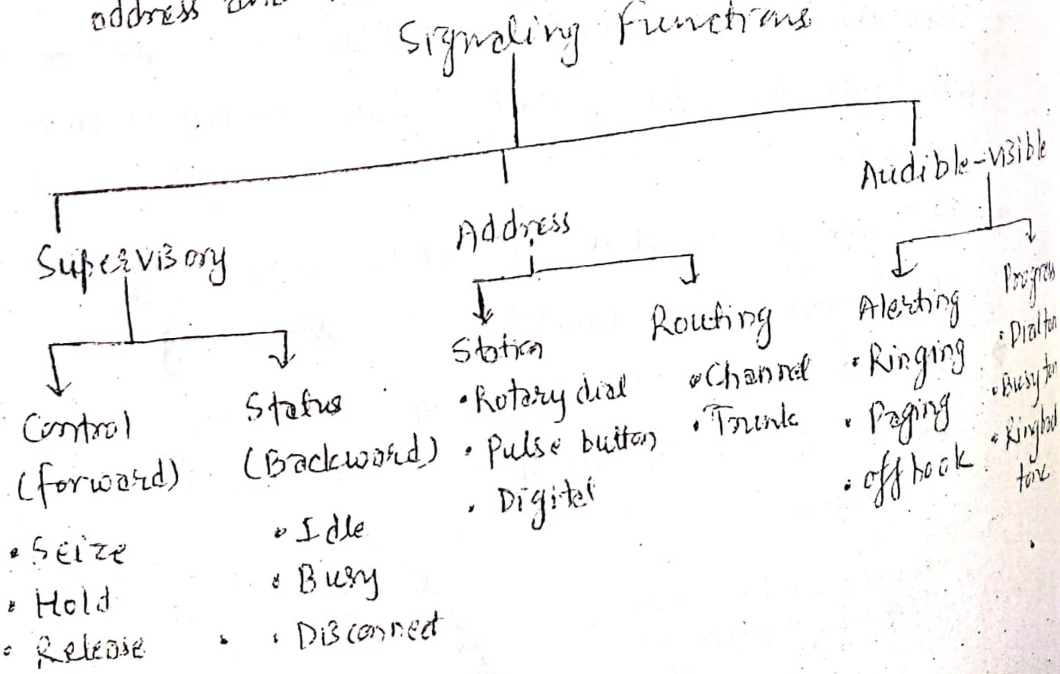
(i) Supervisory:

→ It conveys status or control of network elements

egs. request for service, call alerting, call termination, called party ringing, called party busy tones,

(ii) Information bearing:

→ Signals include called party address, calling party address and toll charges.



Three forms of signaling are involved in a telecommunication network :-

(i) Subscriber loop signaling (depends on type of telephone set)

(ii) ~~Inter~~ Intraexchange or register signaling (Internal to the switching system depends on the type of switching equipments)

(iii) Interexchange or interregister signaling or line signaling (communications between two or more exchange).

imp ✓ Types of Signaling [2006, 2005, 2010]:

Signaling
- In channel
- Common channel

(i) Inchannel Signaling :

→ Uses the same channel to carry user voice ~~and~~ or data and to pass control signals related to that connection.

→ Its merit is that, it doesnot need any additional transmission facilities for signaling.

→ In this type of signaling busy tone is returned from terminating exchange.

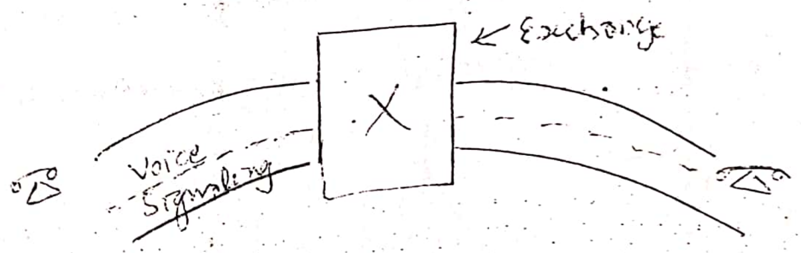


Fig: Inchannel Signaling

DISadvantages:

→ Interference between voice and control signal may occur

② Common Channel Signaling (CCS):

→ CCS doesnot use the speech or data path for signaling.
→ It uses a separate common channel for passing control signals for a group of trunks or information path.

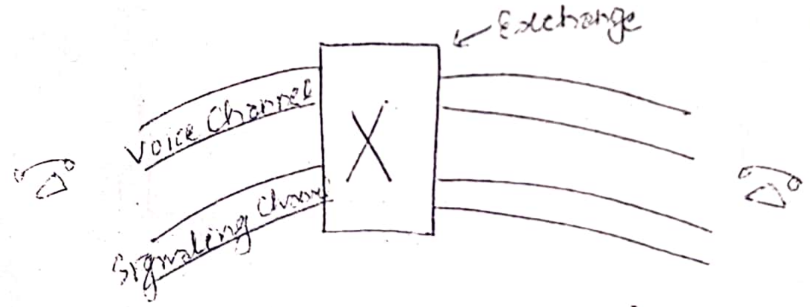


Fig Common Channel Signaling

Common Channel Signaling is further divided into two types :-
ⓐ Channel Associated CCS
ⓑ Non-associated CCS

ⓐ Channel Associated Mode:

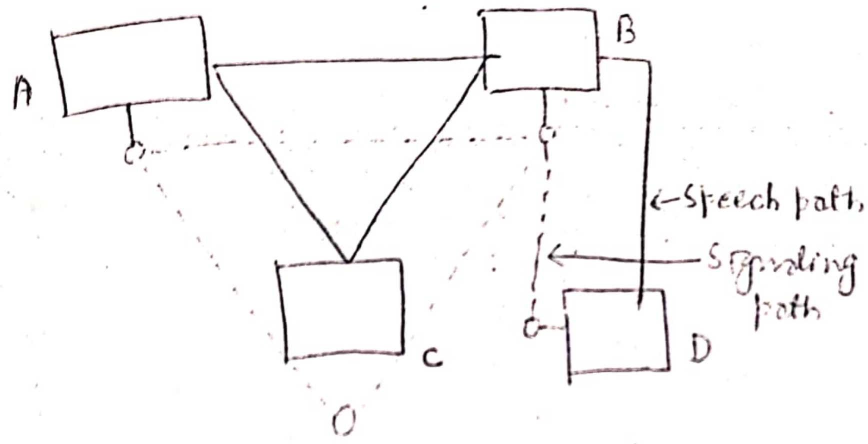
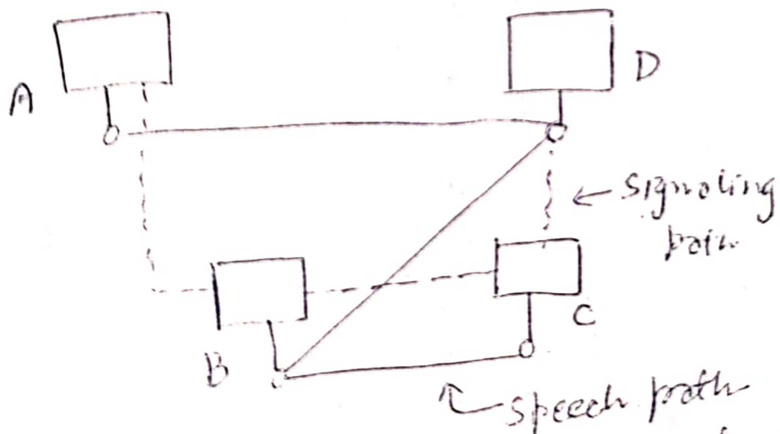


Fig Channel associated mode

→ In this mode, the signaling path passes through the same set of switches as does the speech path.

Advantages: ① Simple, ② Economical

ⓑ Non-associated Signaling:



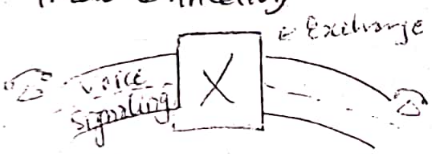
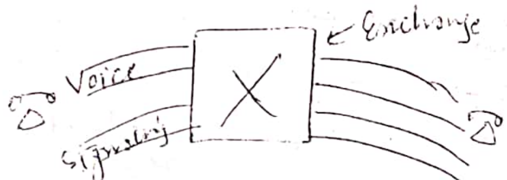
→ In this signaling, the signaling information may follow route i.e. different from speech path.

Advantage: offers flexibility.

Signaling

- Inchannel
  - D.C. (use non-amplified ckts.)
  - Low freq. (a.c.)
  - Voice freq.
    - Inband (300–3400 Hz)
    - Outband (< ~~4000Hz~~ 4000Hz)
  - PCM
- Common channel
  - Associated
  - Non-associated

2006, 2007, 2015. 5-7 marks

Inchannel	Common Channel
(i) Use same channel to carry user voice or data & to pass control signals related to that connection.	(i) Uses a separate common channel for passing control signals for a group of trunks or information path
 Fig. Inchannel Signaling	 Fig. Common channel Signaling
(ii) Trunks are held up during signaling. supports prevent South from falling	(ii) Trunks are not required for signalling.
(iii) Interference between voice and control signals may occur	(iii) No interference as the two channels are physically separate.
(iv) Signal repertoire (repert) is limited.	(iv) ~~Signal~~ Extensive signal repertoire is possible.
(v) Separate signaling equipment is required for each trunk and hence it is expensive.	(v) Only one set of signaling equipment is required for whole group of trunks and so CCS is economical.

145

Inchannel	Common Channel
(vi) As the voice channel being the control channel, there is a possibility of potential misuse by customers.	(vi) Control channel is in general inaccessible to users.
(vii) Signaling is relatively slow.	(viii) Signaling is significantly fast.
(viii) Speech circuit reliability is assured.	(viii) There is no automatic test of speech circuit
(ix) It is difficult to change or add signals.	(ix) There is flexibility to add or change signals.
(x) Difficult to handle signaling during speech period	(x) Freedom to handle signals during speech.

CCITT Signaling System No. 7

→ CCITT SS7 was developed to meet the advanced signaling requirements of all the digital network based on the 64kbps channel.

→ SS1 - SS7, SS1 - SS5 → Inchannel Signaling
SS6 - SS7 → Common Channel Signaling.

→ In SS7, common channel signaling, a phase-equalized voice channel is capable of supporting a bit rate of 2.4 kb/s with acceptable error rates for signaling. At this bit rate, one CCS link can carry signals for 1500-2000 speech circuits.

Inchannel - difficult to add

difficult to change or add signals

→ Specific attention has been given to the requirement of ISDN while designing SS7.

→ The internal control and network intelligence to an ISDN are produced by SS7.

→ SS7 is suitable for operation over both terrestrial and satellite links.

Architecture of SS7:

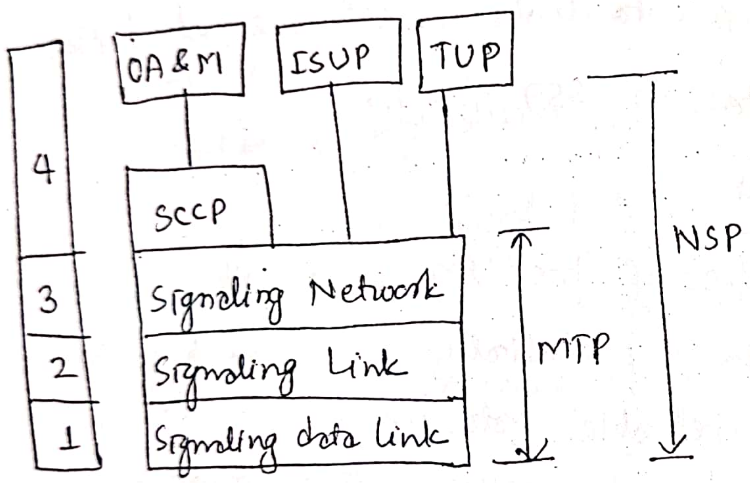


Fig Block diagram of architecture of SS7.

Where,

OA & M → Operation, Administration & Maintenance

MTP → Message Transfer Part

TUP → Telephone User Part

NSP → Network Service Part

ISUP → ISDN User Part

SCCP → Signaling connection Control Part

→ The protocol architecture of SS7 has four levels.
→ The lower three levels are referred to as the message transfer part (MTP) that provides the reliable service for routing messages through SS7 network.

(i) Signaling Data link:

→ It is the lowest layer that is concerned with the physical and electrical characteristics of signaling link.
→ All signaling data links in SS7 are full duplex links dedicated to SS7 traffic.

(ii) Signaling link:

→ The main purpose of the signaling link i.e second layer is turn a potentially unreliable physical link into a reliable data link.
→ The signaling link must ensure that:-
  - there are no losses or duplication of control messages.
  - messages are delivered in the same order in which they originate
  - there is a match between the receiver capacity and the transmission rates (i.e receiver is capable of exercising flow control over the sender).

Scanned with CamScanner

(III) Signaling network:

→ This relate to message handling and network management.

Message handling: involves:

- discrimination,
- routing (based on signaling link information)
- distribution of message

Network management:

- monitor signaling links and overcome link failures or degradation.

(IV) Signaling Connection Control Part (SCCP):

→ The main purpose of SCCP is to enhance the limited routing, distribution & addressing capabilities of 3rd layer

→ SCCP and MTP together are referred to as NSP.

2009

Signaling Unit (SU)

(I) → Information to be sent is structured by the Signaling control unit (level-2) into a signal unit.

→ There are 3 types of signaling units defined in SS7:-

(I) Message Signaling Unit (MSU),
(II) Link Status Signal Unit (LSSU)
(III) Fill in signal Unit (FISU).

149 There are 3 types of signaling units defined in SS7: (i) message signal

(1) Message Signal Unit (MSU):

→ This transfers information supplied by a user part (level-4) through level 3 (signaling network level)

F	Control	SER	SIF	CRC	F

Fig. MSU Packet.

Note:
*→ All the SUs begin and end with a flag (F) field which has the unique bit pattern 01111110.
- The flags act as delimiters for the SUs.
- The 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.

→ Here CRC = Cyclic Redundancy ~~Check~~ code
(Used for error ~~handling~~ checking)

SIF (Signaling Information):
→ contains level 3 and level 4 information.
- contains user data from TUP, ISUP, OA&M.

SER(Service Information field):
→ This information field specifies the type of message and whether the message is related to a national or international network

→ CRC is of 16-bit and is used for error checking

(iii) Link Status Signal Unit (LSSU):

F	Control	Status	CRC	F

Fig LSSU.

→ LSSU is used for link initialization & flow-control.

(iv) Fill-in Signal Unit (FISU):

F	Control	CRC	F

Fig. FISU.

→ This is ~~used~~ sent to maintain alignment when there is no signal traffic
- for maintaining data rate

* The control field consists of 5 subfields:

BSN	BI	FSN	FI	LI	U
7	1	7	1	6	2

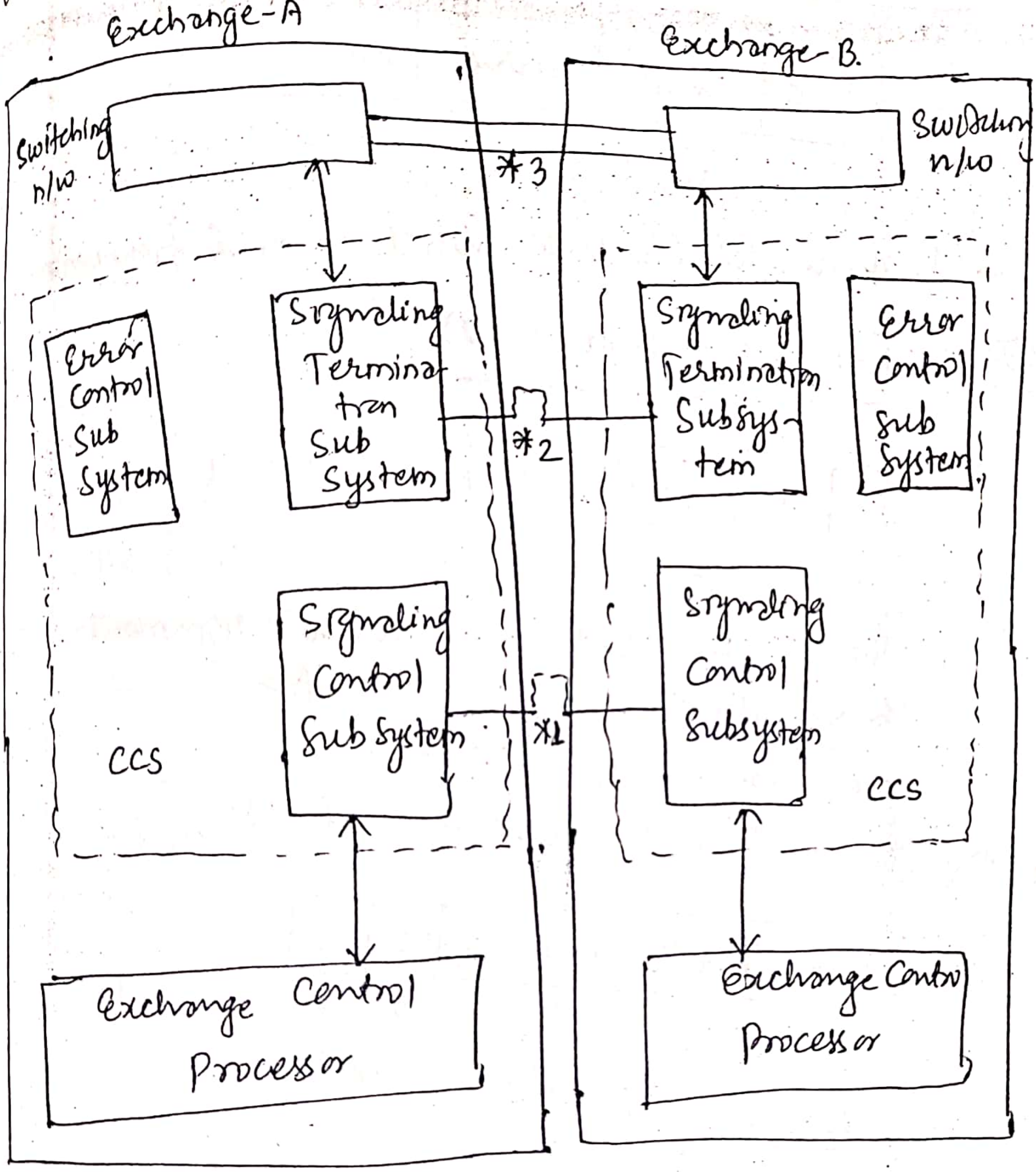
Fig. Control subfields

Where

BSN → Backward Sequence number } provide acknowledgement
BI → Backward Indicator

FSN → Forward Sequence Number
FI → Forward Indicator
LI → Length Indicator
↳ 0 → FISU
1 → LSSU
3 to 63 → MSU.

Block Diagram of SS7



where

*1 → message Transfer

*2 → Transfer of Signaling Units,

*3 → Speech + Signaling Information

Fig: Block Diagram of SS7.

→ Signal messages are passed from the central processor of the sending exchange to the CCS system.

→ This consists of 3 microprocessor based subsystems :-

1. Signaling Control Subsystem,
2. Error Control Subsystem,
3. Signaling Termination Subsystem.

① Signaling Control Subsystem:

(i) → It structures the message in the appropriate format and queues them for transmission

(ii) → when there are no message to send, it generates filter message to keep the link active

② Signaling Termination Subsystem:

(i) → message is then passed to signaling termination subsystem.

(ii) → Here, complete signal units are assembled using sequence number & check-bits, generated by the error control subsystem

③ Error Control Subsystem:

(i) → It generates the checkbits in sender

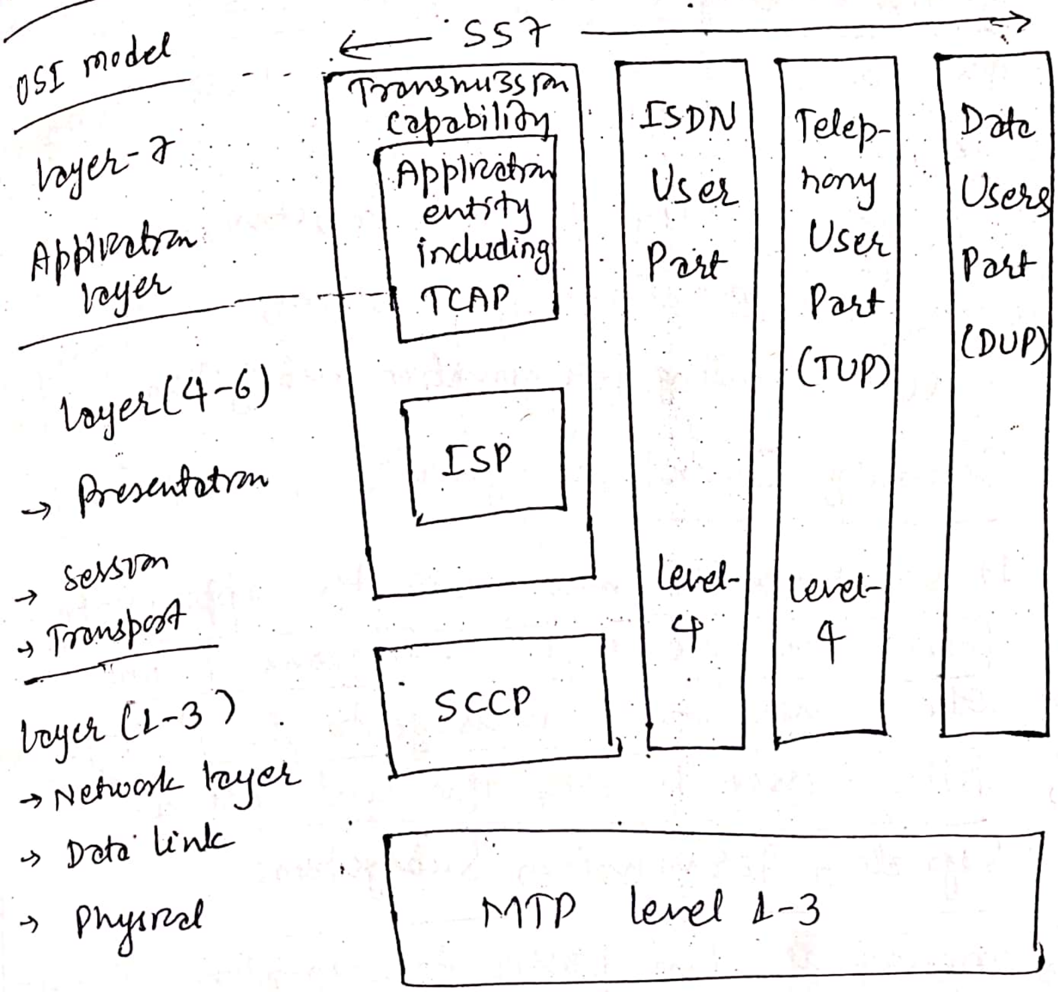
(ii) → The message is then passed to switching, which transmits the voice and other signaling information through channel.

(iii) → At the receiving terminal, the reverse sequence is carried out.

when there are no ... generates it the

it generates

Relationship between SS7 and OSI layer model:



where,

TCAP → Transaction Capability Application Part

ISP → Intermediate Service Part

MTP → Message Transfer Part

SCCP → Signaling Control Connection Part

1) → The relationship between levels of SS7 with the layer of OSI model is as shown in fig above

2) → Level 1 is the means of sending bit streams over a physical path

(3) → Level-2 performs the function of error-control, link initialization, flow control, etc

(4) → Level-3 provides the function required for a signaling network

(5) → Thus, level 1-3 form the MTP of SS7.

(6) → Level 4 is the User Part. This consists of the process for handling the service being supported by the Signaling System

(7) → Since SS7 was developed after the OSI model, its protocols were specified to confirm with OSI.

(8) → SCCP has been added to level-3 to make it fully compatible with level-4 of OSI model.

(9) → The ISP performs the function of layer 4-6 of OSI.

(10) → TCAP provides the feature of layer-7.

Dual Tone Multi Frequency (DTMF) & pusle Dialing

Pulse Dialing:

- Pulse dialing originated in 1895 and is used extensively even today.
- In this form of dialing, a train of pulses is used to represent a digit in the subscriber number

→ The number of pulses in a train is equal to the digit value it represents except in case of zero which is represented by 10 pulses.

→ Successive digits in a number are represented by a series of pulse trains:

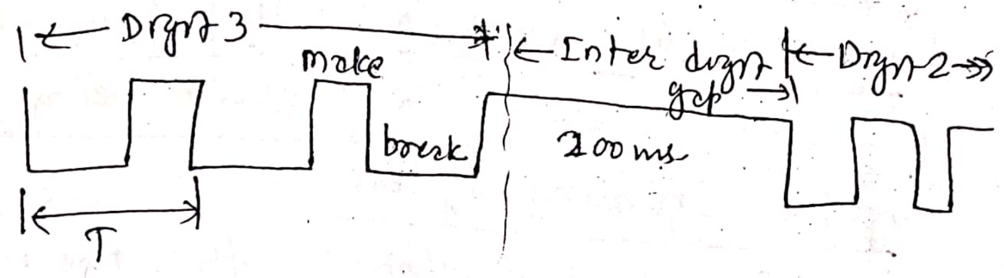


Fig. Pulse Dialing

→ Two successive trains are distinguished from one another by a pause in between them known as inter-digit gap.

→ The pulse rate is usually 10 pulses per second with a 10% tolerance

→ The inter digit gap is at least 200 ms although in some designs it may be 400-500 ms.

→ In introducing dial pulsing mechanism in the telephone set, the following points have to be considered:-

① Since the pulses are produced by make & break of the subscriber loop, there is likelihood of sparking inside the telephone instrument

(ii) The transmitter, receiver and the bell circuits of the telephone set may be damaged, if the dialing pulse are passed through them.

(iii) The dialing habits of the users vary widely and hence all timing aspects should be independent of user action.

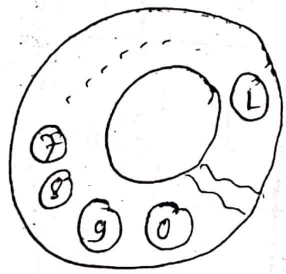


Fig. Rotary Telset

Finger Plate

Dual Tone multi frequency (DTMF)

→ In a rotary dial telephone, it takes about 12 secs to dial a 7-digit number

→ From the subscriber point of view, a faster dialing rate is desirable Strowger system cannot respond to rate higher than 10-12 pulses per second.

→ Pulse dialing has many limitations which led to touch tone dial telephone

→ A push-button (key phone or touch tone) telephone uses dual tone multi frequency signaling.

→ It sends each digit by means of a combination of two frequencies one from each 2 groups.

Hz/Hz	1209	1336	1477	1638
697	1	2	3	Spare
770	4	5	6	Spare
852	7	8	9	Spare
941	*	0	#	Spare

Fig: Frequency Coding Used by Push button

→ For example, pressing the push button '9' transmits 852 Hz and 1477 Hz

→ Since each digit uses 2 frequencies and these arenot harmonically related, there is much less chance of the combination being produced by speech or room noise picked by the telephone transmitter than if a single frequency were used

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Chapter 8 (7-10 marks)

Synchronization and Network Management

2006, 2008

Synchronization

Principle and Mode of operation:

(1) → When a PCM bit stream is transmitted over a telecommunication link, there must be synchronization at three different levels :-

1. Bit level,
2. Time slot, and
3. frame.

(2) → Bit synchronization refers to the need for the transmitter (coder) and receiver (decoder) to operate at the same bit rate. It also refers to the requirement that the receiver decision point be exactly at the mid-position of the incoming bit. Bit synchronization assures that the bits will not be misread by the receiver.

→ Obviously, the ~~time~~ digital receiver must also know when a time slot begins and ends. If we can synchronize a frame, time slot synchronization can be assured. ensure

telecommunication
assured
ensure
refer

→ Frame synchronization assumes that bit synchronization has been achieved. We know where a frame begins and ends by some kind of marking device, with T1, it is framing bit and E1 has a separate framing and synchronization channel, namely 0.

In this case, the recovery looks in channel 0 for framing sequence in bits 2 to 8 (bit 1 reserved) of every other frame. The framing sequence is 0011011. Once the frame sequence is acquired, the receiver exactly knows where frame boundaries are. It also has time slot aligned.

(4) Synchronization is required in communication network for interconnecting various digital transmission.

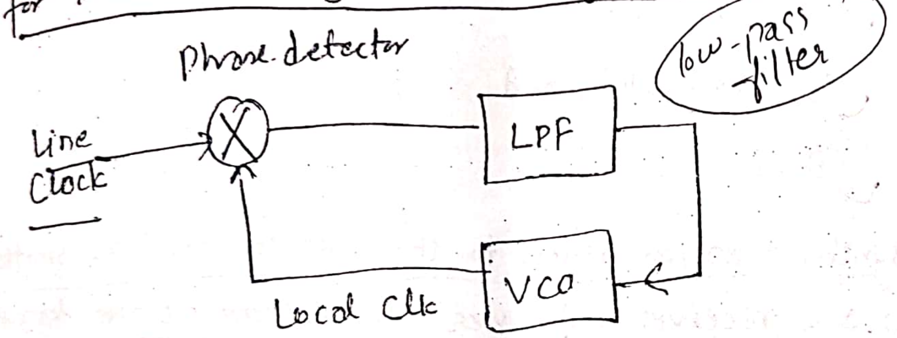


Fig: PLL Clock recovery ckt

1. Phase detector measures the phase difference between the incoming clk and locally generated clk.
2. If the zero crossing of the line clk proceeds → begin or continue a course of action. the local clk, a +ve voltage is generated, else -ve voltage is generated.
3. The output of the phase detector is filtered to remove noise and hence the phase measurement adjusts the frequency of VCO to reduce phase difference.

Clock Instability:

→ It occurs due to variations in the output frequency of VCO.

→ Important aspect of 'Clk' instability is its frequency i.e. the rate at which the clock frequency changes goes from being high to low.

→ If the variation is too slow, it is known as Clock wander

→ When the variations are more rapid, the variations are referred to as jitter.

→ Main causes of clk instability are:

(i) Noise and interference,

(ii) Change in length of transmission medium

(iii) Change in velocity of propagation

(iv) Irregular timing information,

(v) Doppler shift.

Elastic Store [short note, 2006]

(1) → When transmission link is interfaced ~~with~~ with 'Clk' the difference between a received & relatively fixed clk, is ~~recomme~~ reconsiled using elastic store.

(2) → Elastic store is considered as a data buffer i.e. written rate by 1 clk and read from another clk.

⑤ If instability occur in either clk, the elastic store absorbs the difference as the amount of data transmitted and amount of data received

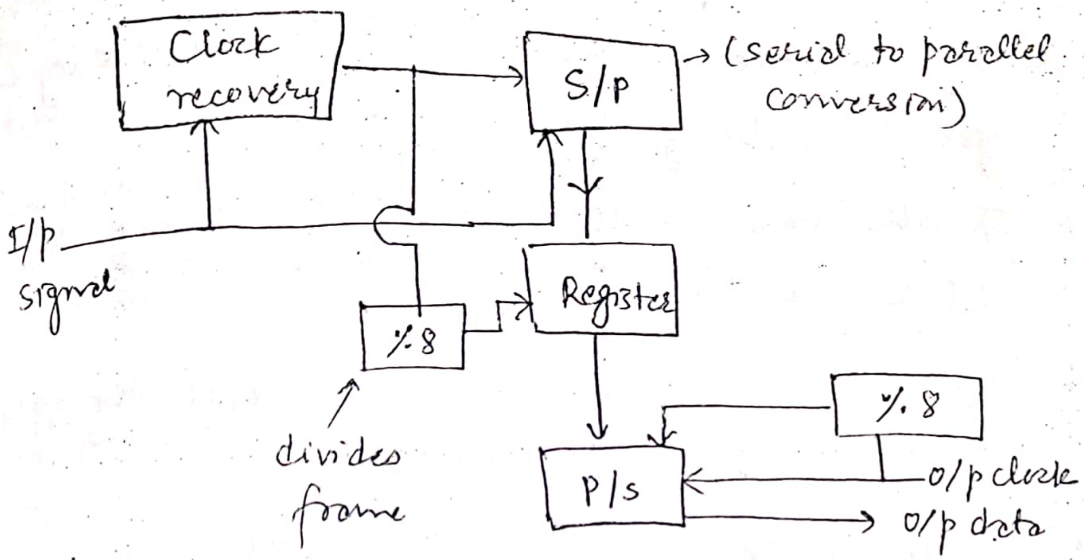


Fig Basic Implementation of an Elastic Store

④ → Elastic store implementation using S/P and P/s (where, P = Parallel, S = Serial) converter

⑤ → Incoming data are transmitted into the registers as soon as each word is accumulated in S/P ~~converter~~ converter

Serial to parallel Conversion

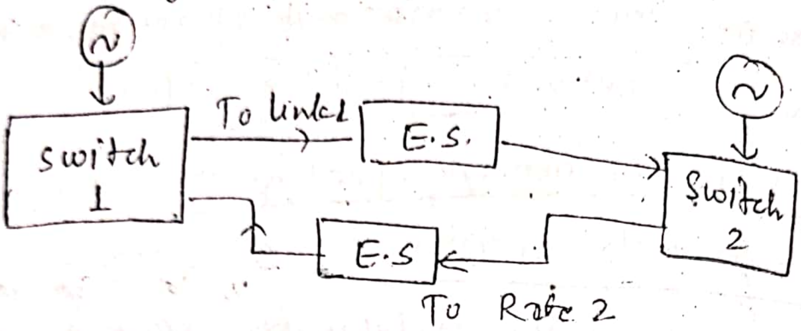
⑥ → Data in register are transferred to the o/p P/s converter

⑦ → Transfer a P/s Converter is independent of incoming clk.

⑧ → No data loss and ~~jtt~~ jitter observed by varying delays through the elastic store.

Timing accuracy:

→ In digital communication equipment, using autonomous frequency sources are connected, then the clock rates of the two system are never exactly the same.



→ The elastic store at the 1st digital switch is written into by the recovered link clk R_2 but read from the local rate R_1.

(R_1 and R_2 are clock rates).

→ If average rate of R_1 and R_2 are different, the elastic store will overflow or underflow.

→ If R2 is greater than $R_2 > R_1$, the elastic store at 1st switch will overflow, causing loss of data

→ If $R_1 > R_2$, the elastic store (E.S.) underflow causing extra data to be inserted

→ Disruption in the data stream, caused by underflow & overflow of an elastic store is referred to as slip.

→ Uncontrolled 'slip', causes loss of frame synchronization

→ 'Slip' are allowed to occur only in prescribed manner, that donot upset framing

→ One approach to control the slip is to ensure that they occur only in the form of repetition or deletion of entire frame such that the framing logic remains synchronized.

Network management [2009].

(1) Basic goal of network management is to maintain efficient operation during equipment failure and traffic overflow.

(2) Effective network management optimizes a telecommunication network's operational capabilities :-

(i) It keeps the network operating at peak performance

(ii) It informs the operator of impending deterioration
आसन्न => be about to happen.

(iii) It provides easy alternative routing & work arounds when deterioration or failure takes place

(iv) It provides the tools for ~~pinpoin~~ pinpointing causes of performance deterioration or failure

(v) It serves as the frontline command post for network survivability.

(vi) It informs in quasi-real time regarding network performance

(vii) It maintains & enforces network security eg encryption & password use

(viii) It gathers and files data on network usage.

(ix) It performs a configuration management function

(x) It also performs an administrative management function

→ Network management tasks:

① Fault management
② Configuration management
③ Performance management
④ Security ,, &
⑤ Accounting ,,

deterioration

Operational Capabil.

Effective Network management

→ ~~In data link control protocol, err~~

→ To see the need for data link control, we list some of the requirements and objectives for effective data communication between two directly connected transmitting / receiving stations :-

1. Frame Synchronization: Data are sent in blocks called frames, The beginning & end of each frame must be recognizable.
2. Flow Control: The sending station must not send frames at a rate faster than receiving station can absorb them
3. Error Control: Bit errors introduced by the transmission system should be corrected
4. Addressing: On a multipoint line, such as LAN the identity of the two stations involved in a transmission must be specified
5. Control & Data on Same Link: It is not desirable to have a physically separate communication path for control information. In addition, the receiver must be able to distinguish control information from data being transmitted
6. Link management: The ~~initiance~~ initiation, maintenance & termination of a sustained data exchange require a fair amount of co-ordination & co-operation among stations.

recognizable

180
160
480

180
160
160
Reading

165

recognizable

find frames

Scanned with CamScanner

main considerations for network management are:-

Imp. [07, 08, 09, 010]

(I) Routing Control, &
(II) Flow Control

(I) Routing Control:

1. It refers to procedure that determines which path in the network are assigned to particular connections.
2. If possible connections should use the most direct routes at lowest levels of the network.
3. The direct routes are obviously desirable because they use fewer network facilities and generally provide better transmission quality.
4. However, economic considerations often limit the capacities of the direct routes so that alternate routes are needed to maintain suitably low blocking probabilities.
5. If a trunk group between two switching machine contain enough circuits to provide low blocking probability, while a significant number of circuits in group are idle during average traffic load.
6. A more economical design allocates a limited number of heavily utilized trunks in direct route & provides alternate routes for overflow traffic.

Procedure that

→ In this manner, users are able to share larger portion of network

Flow Control:

(1) - Routing algorithm are concerned only with the utilization of paths or direction of travel within a network.

(2) - Another requirement of network management is to control the amount of traffic in a network.

(3) → Managing the rate at which traffic enters a network is referred to as flow control.

(4) → Flow control is a technique for assuring that a transmitting entity doesnot overwhelm a receiving entity with data.

निराशा

(5) → Receiving entity typically allocates a data buffer of some maximum length for a transfer. When data are received, the receiver must do a certain amount of processing before passing the data to the higher level software.

(6) → In the absence of flow control, the receiver's buffer may ~~fall~~ fill up and overflow while it is processing old data. So, the network will be inefficient or may ceases the function of network entirely in heavy traffic conditions.

at which other control the

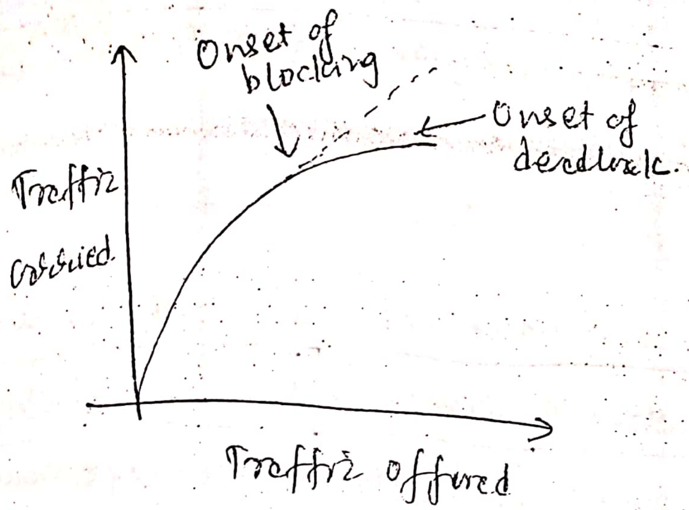


Fig. Carried traffic Vs traffic offered with no flow Control.

- As indicated in above fig., when light traffic condition exists, the network carries all requests.
- As the load increases, some of the offered traffic is rejected because no appropriate circuits are available i.e blocking exists.
- If further input load increases, then the network may even cease to carry any traffic at all
- Network flow control is greatly simplified with common channel signalling support for centralized network control.
- There are two main types of flow control methods: (I) Stop & wait flow Control, and
  (II) Sliding window Flow Control.

(i) Stop & wait flow Control:

(i) → The simplest form of flow control in which a source entity transmits a frame, after the destination entity receives the frame.

(ii) → The destination sends 'back' signal to receiver as it wants ~~to~~ another frame. ← acknowledgement

(iii) → So, ~~the~~ after sending one frame, the source waits for 'back' signal from the receiver to transmit next frame.

(ii) Sliding Window flow Control:

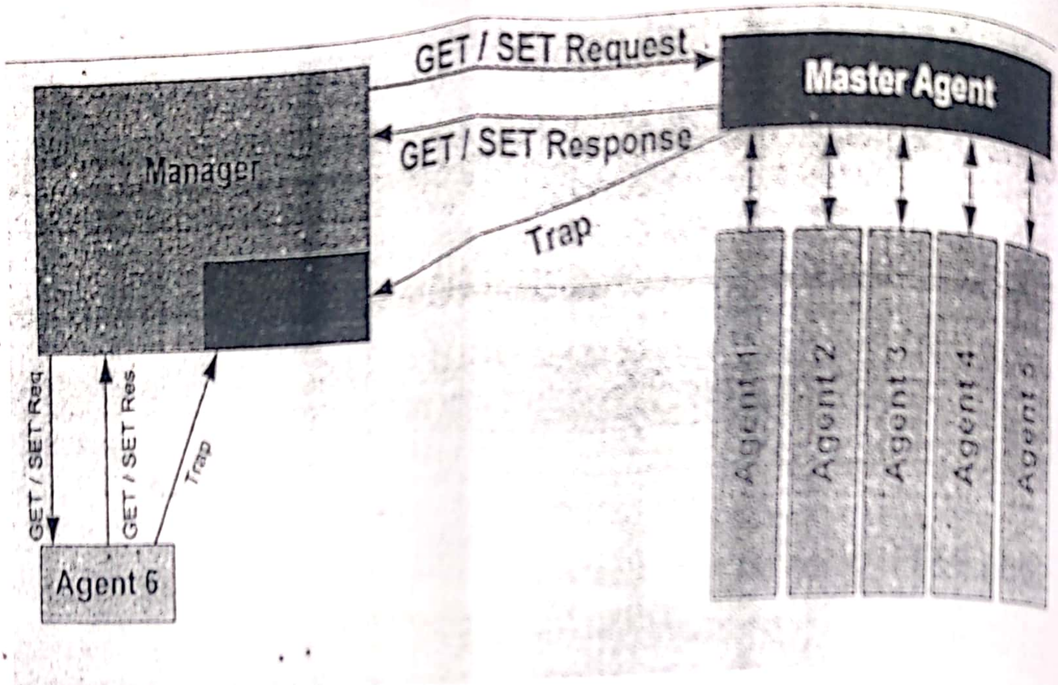
(i) → Efficiency can be greatly improved by allowing multiple frames to be in transit at the same time.

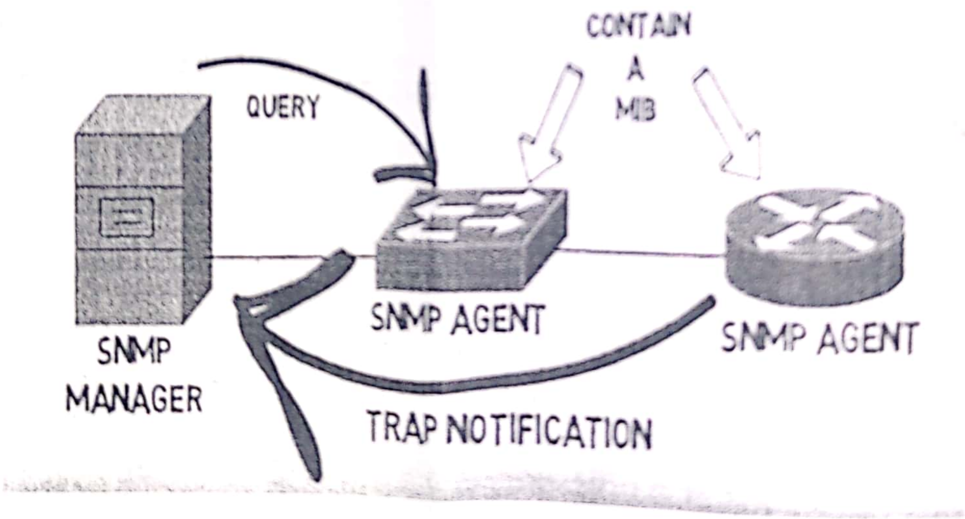
(ii) → Transmit end maintains a list of sequence numbers that it is allowed to send and receiver maintains a list of sequence numbers that it is prepared to receive.

(iii) → Each of these lists can be thought of as a window, of frames. The operation is ~~refere~~ referred to as sliding window flow control.

window of frames the operation is referred participants and adding or deleting media streams.

- other applications include video conferencing, instant messaging, file transfer, online games, etc.
- The SIP protocol is an application layer protocol.
- A motivating goal for SIP was to provide a signaling and call setup protocol for IP-based communications that can support a superset of the call processing functions and features present in PSTN.
- SIP is a text based protocol with syntax similar to that of HTTP.


GET / SET Request
Master Agent
Manager
GET / SET Response
Trap
Agent 1
Agent 2
Agent 3
Agent 4
Agent 5
GET / SET Req.
GET / SET Res.
Trap
Agent 6


CONTAIN
A
MIB
QUERY
SNMP AGENT
SNMP AGENT
SNMP
MANAGER
TRAP NOTIFICATION

Network management using SNMP

(i) SNMP is an internet-standard protocol for collecting and organizing information about managed devices on IP networks and for modifying that information to change device behavior.

(ii) Devices that support SNMP include routers, switches, servers, workstations, printers, modem racks and more.

(iii) SNMP managed network consists of 3 main components:

i. Managed device,

ii. Agent – software that runs on managed devices,

iii. Network Management Station (NMS) – software which runs on the manager.

- A **Managed device** is a network node that implements an SNMP interface that allows unidirectional (read-only) or bidirectional (read and write) access to node-specific information, e.g. routers, access servers, switches, IP telephones, etc.
- An **agent** has local knowledge of management information and translates that information to or from an SNMP-specific form.
- **A network management station (NMS)** executes applications that monitor and control managed devices. They provide bulk processing and memory resources required. 1 or more NMSes may exist on any managed network.
- **Management Information Base (MIB):** The variables accessible via SNMP are organized in hierarchies. These hierarchies, and other metadata (such as type and description of variable) are described by MIBs. A component of a MIB containing a collection of variables, which can be queried by or configured from an SNMP manager.

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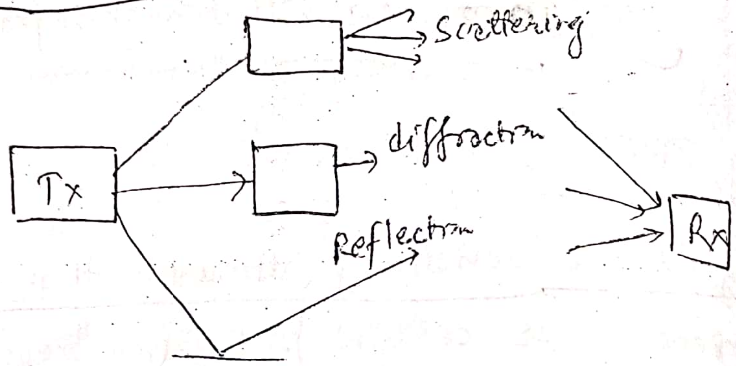
Chapter ~~6~~ 7 (5-7 marks).

Diversity Technique:

2 marks.

Multipath propagation:

→ Multipath propagation is a ~~technique~~ phenomenon that results in signal, reaching the receiver's antenna by two or more paths.



→ Waves may propagate in more than 1 path.

→ Multipath signals result from obstacles in the path of signal propagation eg. large buildings, woods, rock, hills, etc.

→ They attenuate, bend or reflect the transmitted signal

→ Causes of multipath:-

- Atmospheric ducting,
- Ionospheric reflection & refraction,
- Reflection from mountains & buildings.

L.O.S. (Line of Sight):

(I) → Even when L.O.S. exists multipath propagation still occurs due to reflections from the ground & surrounding structure.

(2) The incoming radio waves arrive from different direction with different propagation delays.

Non-Line-of-Sight (N-L.O.S):

→ It is due to:

(I) Atmospheric ducting → a system of ducts.

(II) Reflection from high mountains

(III) ~~Io~~ Ionospheric reflection or refraction

Short note

Fading model:

→ fading is the deviation of attenuation that signal experiences over certain propagation media.

Factors influencing fading are:- प्रभावित

(1) multipath Propagation:

→ The presence of reflecting objects and scatterers in the channel creates a constantly changing environment that dissipates the signal energy in amplitude, phase and time. These effects result in multiple versions of transmitted signal that arrive at the receiving antenna, displaced with one another with respect to time and spatial orientation.

→ The random phase and amplitudes of different multipath components cause fluctuations in signal strength, thereby ~~indu~~ inducing fading, signal distortion or both.

→ Each signal from different path experience different attenuation, delay and phase shift.

② Speed of the Receiver:

→ The relative motion between the base station and the receiver station results in random frequency modulation due to different Doppler shift on each of the multipath components.

→ Doppler shift will be positive or negative depending on whether the receiver is moving toward or away from the base station or transmitter.

③ Speed of Surrounding Objects:

→ If objects in the radio channel are in motion, they induce a time varying Doppler shift on multipath components.

④ Transmission Bandwidth of a Signal:

→ The transmission bandwidth of a signal & channel also ~~result in~~ affect the fading.

→ Two types of fading resulted due to variation in features of transmission bandwidth of signal are as follows:

(i) Flat fading and

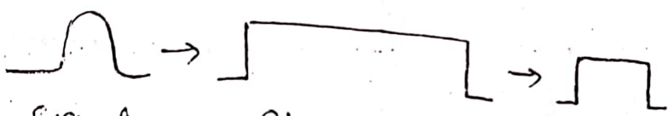
(ii) Frequency Selective Fading

Fading based on multipath time delay spread:

① ~~Flat~~ fading:

(i) It occurs if the coherence Bandwidth (maximum frequency difference for which signals are still strongly correlated in amplitude) of the channel is greater than Bandwidth of signal.

(ii) All frequency component will experience same magnitude of fading



Signal Channel

② Frequency Selective Fading:

(i) It occurs if the Bandwidth of the signal is greater than the coherence Bandwidth of channel.

(ii) In this different frequency component of signal experience different fading

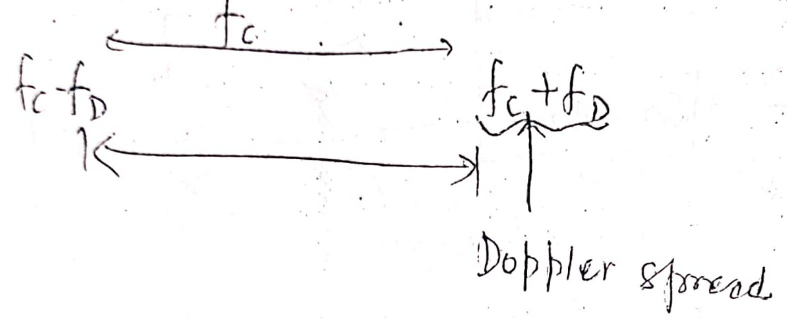
multipath time delay spread

Fading Based on Doppler's spread:

(i) Doppler spread is a parameter which describes time-varying nature of channel.

(ii) This provides information about time varying nature of channel, caused by either relative motion between receiver and transmitter or by movement of objects in channels.

(iii) → Doppler's spread - f_D is a measure of spectral broadening caused by motion & is defined as range of frequency over which received Doppler spectrum is essentially non-zero.



Diversity.

(i) → It is one of the technique to minimize fading.

(ii) → Diversity is a powerful communication receiver technique that provides wireless link improvement at low cost.

(iii) → If one radio path undergoes a deep fade, another independent path may have a strong signal.

(iv) → Diversity decisions are made by the receiver and are unknown to the transmitter.

(1) Time Diversity?

→ Time diversity repeatedly transmits information at time spacing.

→ multiple repetition of signal will be received with independent fading condition thereby providing diversity.

→ multiple version of same signal are transmitted at different instants.

→ one modern implementation of time diversity involves use of RAKE receiver for CDMA.

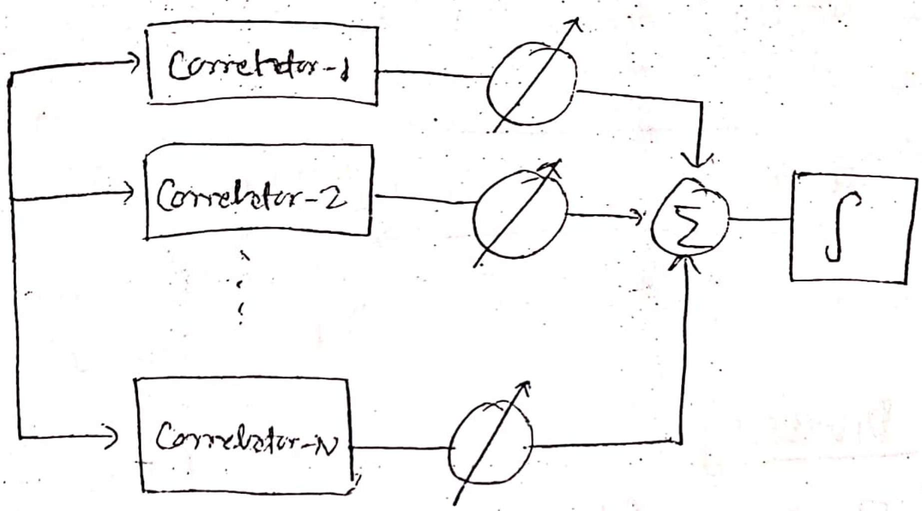


Fig Rake Rx

→ In 'correlators', received signal is multiplied by time-shifted version of locally generated code sequence

→ An 'attenuator' is an electronic device that reduces the amplitude or power of a signal without distorting its waveform.

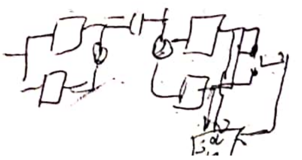
2006, 2009, 2010

Frequency Diversity (5 marks)

(i) → Frequency diversity is one of the means of providing backups to overcome signal fading

(ii) → frequency diversity is implemented by transmitting information on more than one carrier frequency.

Providing backups to overcome fading



from alm

(iii) → The logic behind this is that frequencies separated by more than one coherence bandwidth of channel will not experience same fading.

(iv) → Thus, when 1 carrier fades, it is unlikely that another carrier fades simultaneously.

(v) → Frequency diversity involves the use of spare Transmitter (Tx) and receiver (Rx) operating in a normally unused channel.

(vi) → Since separate hardware is used, frequency diversity also provides protection against hardware failures.

Separate

Unused Channel

Simultaneously

Thus, when 1 carrier fades

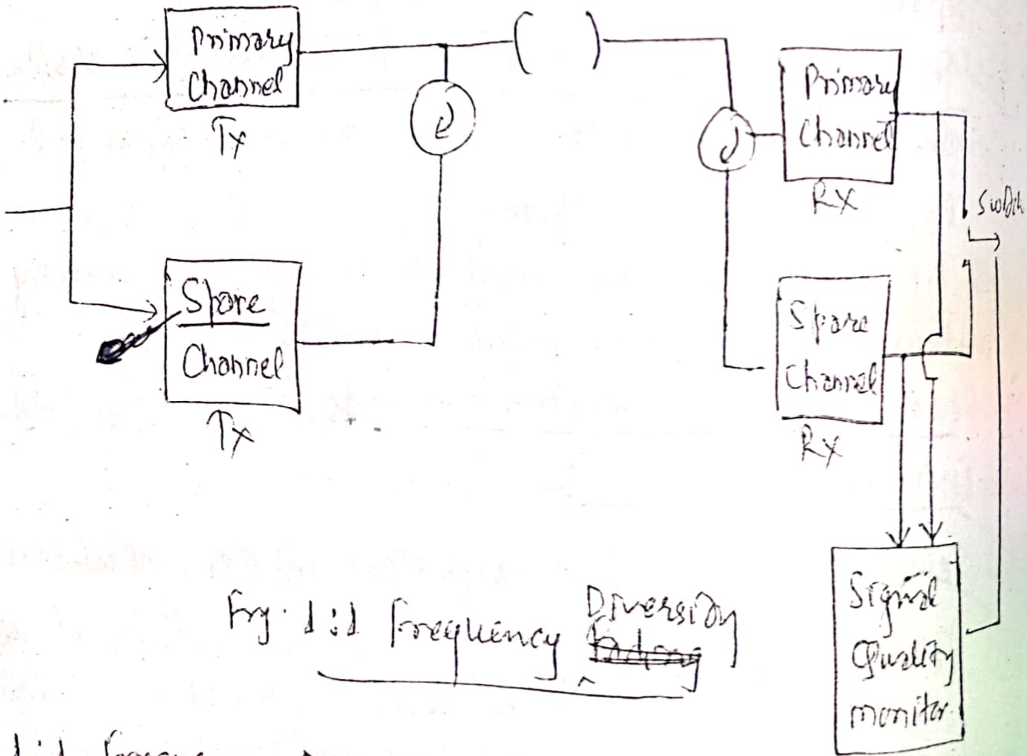


Fig. 1:1 Frequency Diversity ~~fading~~

(vii) → 1:1 Frequency Diversity implies that 1 spare Channel is provided for each assigned message Channel.

frequencies will not experience same fading

2021/2020/2008

Space Diversity / Antenna Diversity :

1. → In space diversity, the signal is transmitted over several different propagation paths.
2. In case of wired transmission, this can be received by transmitting through multiple wires.
3. → In case of wireless transmission, it can be achieved by antenna diversity using multiple transmitter antennas & receiving antennas.
4. → The resulting difference in two paths is normally sufficient to provide independent fading at the two antennas. Received by

wire transmission

5. → Space diversity is most expensive one.
6. → Space diversity is effective than angle diversity.

→ The concept of antenna space diversity is also used in base station design.

→ At each cell site multiple base station receiving antennas are used to provide diversity reception.

7. → Space diversity reception methods can be classified into four categories :-

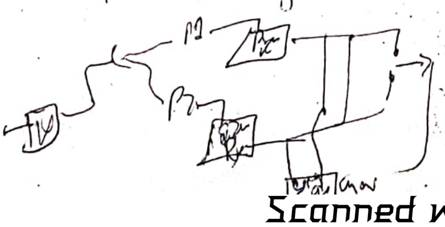
(i) Selection diversity (best signal selected & sent to demodulator)

(ii) Feedback diversity (m signals are scanned in fixed sequence until one is found above predefined threshold)

(iii) Maximal ratio combining (SNR of each signal is tested & selected)

(iv) Equal gain diversity (signals from each branch are co-phased & gain measured)

Can be greatly classified into four categories



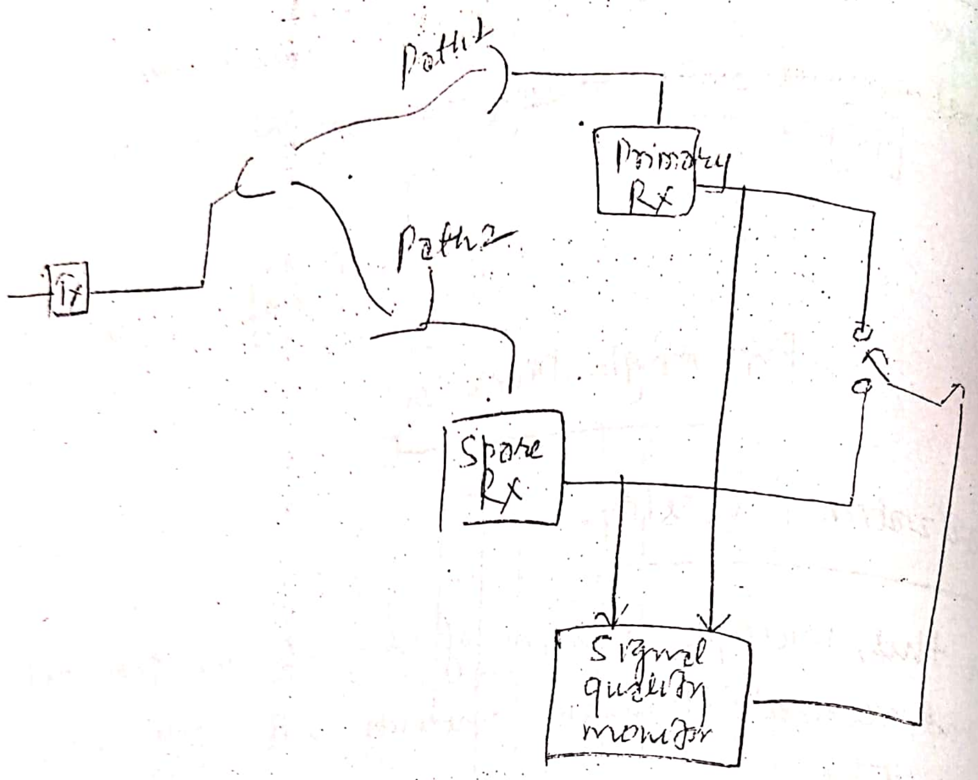


Fig. Space Diversity

2009
Angle Diversity:

→ Because multipath fading is produced by multiple incident rays arriving at slightly different angles, protection from fading can be achieved by discriminating on the angle of arrival.

→ Angle diversity utilizes 2 side by side receiving antennas with slightly different angle elevation to provide the discrimination.

→ Not as effective as space diversity but low cost than space diversity.

→ Implemented where the tower requirements of space diversity are impractical or disallowed

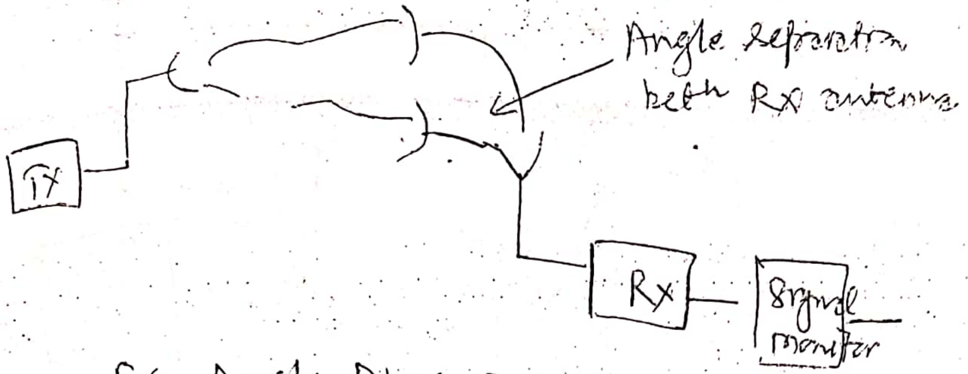


Fig Angle Diversity!

Polarization Diversity!

- In this, multiple version of signals are transmitted and received through antennas with different polarizations.
- The comparatively high cost of using space diversity at the base station prompts using orthogonal polarization to explicit polarization diversity.
- Polarization diversity combines a pair of antenna with orthogonal polarization (horizontal/vertical)
- By pairing two complementary polarization, this scheme can immunize a system from polarization mismatch that would otherwise have signal fading
- It is accomplished through use of separate vertically & horizontally polarized receiving antenna.

Compiled By: Ramu Pandey
J.P. N...

Chapter-8

Traffic Theory:

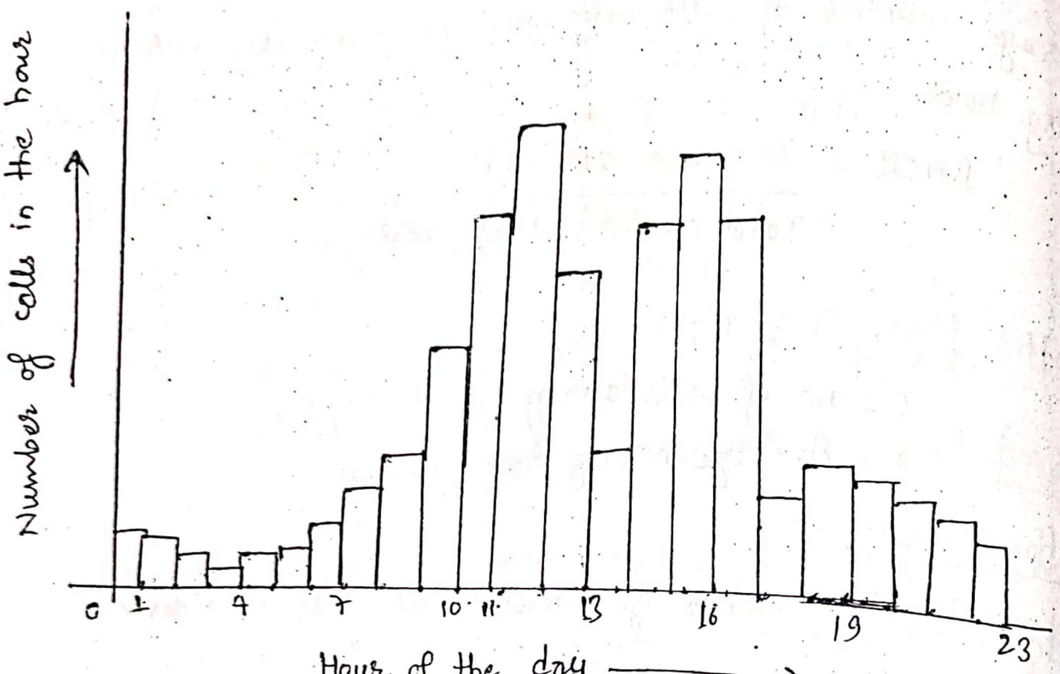


Fig. Typical telephone traffic pattern on a working day

(I) Busy Hour (BH):

→ The period of 1 hr. in a day corresponding to the highest traffic peak.

(II) Peak busy hour:

→ The busy hour of each day maynot be same over the

(III) Time Consistent Busy Hour:

→ The 1 hour period starting at the same time each day for which number of call attempts is greatest.

(IV) Call Completion Rate (CCR):

→ The ratio of the number of successful calls to the call attempts.

$$CCR = \frac{\text{No. of successful calls}}{\text{No. of calls attempt}}$$

(V). Busy hours Call Attempt (BHCA):

→ No. of calls attempt during Busy hour

(VI) Busy Hour Calling Rate (BHCR):

→ Average number of calls originated by subscriber during busy hour

$$BHCR = \frac{\text{Average Bits calls}}{\text{Total number of subscribers}}$$

(VII) Traffic flow, $A = C \times T$

C = no. of calls during 1 hr : calls/hr.

T = Average holding time ; hr/call.

(VIII) Traffic Density:

- Represents the number of simultaneous calls at a given movement.

(IX) Traffic intensity:

- Represents the average traffic density during 1 hr. period. This is used for dimensioning switches traffic

(X) Traffic Unit:

- The traffic intensity more often called simply traffic is defined as the number of calls in progress or occupancy of the server during the observation period

It is denoted by 'A'.

$$\boxed{A = \frac{\text{occupied duration}}{\text{Total duration}}} \quad E$$

→ Although the traffic intensity is dimensionless but the unit of traffic is taken as Erlang, named after A.K Erlang.

Named ? or during 1hr. period.

→ Erlang of traffic indicates that a trunk is ~~capable~~ of occupied for entire period of time.

Also, $\boxed{1 E = 36\ CCS}$ → Century call second

$$\boxed{1 E = 36 CCS = 3600 CS = 60 CM}$$

where, CS = Call Second,
CM = Call minute.

1 CCS = 1 calls for 100 s, or 100 calls for 1 s.

→ One erlang represents a circuit occupied for 1 hour

→ If we know that a group of 10 circuits have a call intensity of 5 Erlangs, we would expect half of the circuits to be busy at the time of measurement

Eg. ①: An exchange serves 2000 subscribers. If the average BHCA is 10,000 and CCR is 60%. Calculate the busy hour calling rate

Soln: Here,

Average busy hour calls = BHCA × CCR
= 6000 calls.

$$\text{Busy hour calling Rate} = \frac{\text{average busy hour calls}}{\text{total no. of subscribers}}$$

$$= \frac{6000}{2000} = 3$$

2019 Spring

Q.② In a group of 10 servers or trunks, each is occupied for 30 minutes in an observation interval of two hours. Calculate the traffic carried by the groups.

Soln! Here:

$$\text{Traffic carried per trunk} = \frac{\text{occupied duration}}{\text{total duration}}$$

$$= \frac{30}{120} = 0.25 \text{ Erlang}$$

$$\text{Total traffic carried by the group} = 10 \times 0.25$$

$$= 2.5 \text{ E}$$ //

Q.③ A group of 20 servers carry a traffic of 10 Erlangs. If the average duration of a call is 3 min. Calculate the number of calls put through by a single server and the group as a whole in a one-hour period

Soln: Here,

$$\text{Traffic per trunk} = \frac{10}{20} = 0.5 \text{E} = 30 \text{ mins/hr}$$

$$\text{No. of calls by a single trunk} = 30/3 = 10 \text{ calls.}$$

$$\text{No. of '' by. 20 trunks} = 10 \times 20 = 200 \text{ calls.}$$

Q.④ A subscriber makes 3 phone calls of 3 mins, 4 mins and 2 mins duration in a 1hr period. Calculate the subscriber traffic in Erlangs, CCS (Centum call second) & CM (Call minutes)

9 ccs

Soln: Here,

we have,

$1E = 36\,CCS = 3600\,CS = 60\,CM.$

Subscriber traffic in Erlangs

$$= \frac{\text{busy period}}{\text{total period}}$$

$$= \frac{3+4+2}{60}$$

$$= 0.15E$$

Traffic in CCS $= 36 \times 0.15 = 5.4\,CCS.$

" " CM $= 60 \times 0.15 = 9\,CM.$

Eg ⑤ Over a 20 mins observation interval, 40 subscribers initiate calls. Total duration of the calls is 4800 seconds. Calculate the load offered to the network by the subscribers and the average subscriber traffic

Soln: Here:

Mean arrival rate, $C = \frac{40}{20} = 2$ calls/min

Mean holding time, $t_h = \frac{4800\,secs}{40} = \frac{4800}{40 \times 60}$ min

$= 2$ min/call

offered traffic, $A = C \times t_h$

$= 2 \times 2 = 4E.$

And,

Average subscriber traffic, $= \frac{4}{40} = 0.1E.$

Thus, we have
$$\text{Traffic carried } (A) = \lambda \times t_H \quad \text{or } \lambda = C$$
where,

λ or C = Average call arrival rate
= calls/sec

t_H = Avg holding time periods
= sec/calls.

Thus,

Traffic = ① traffic generated by the subscriber
② observation of busy servers in the network

Congestion:

→ The situation that arise in telephone exchange where all the trunks are busy so that it can't accept further calls, is known as congestion

→ There are two ways in which overall traffic can be handled:

(I) LCC (Lost Call Cleared System)

(II) LCD (Lost Call Delay System)

(I) LCC [e.g. circuit switching]

→ In a circuit switched systems such as telephone exchange all attempts to make a call over a constant group of trunks are unsuccessful i.e they are rejected or dropped

Prev, Jun (4.05)

(ii) LCD (eg message switched network)

→ In a message switched system calls that are arrived during congestion wait in a queue until an outgoing trunks become free. Thus, they are delay not lost. Such system is called queuing system

* - Parameters to measure loss systems → Grade of Service (G.O.S.)

- Parameter to measure delayed system → Blocking Probability.

Grade of Service (G.O.S.)

(i) → The proportions of the calls that is lost or delayed due to congestion is a measure of service provided. It is ~~also~~ called Grade of Service (G.O.S.)

→ For a lost call system, Grade of Service is defined as,

$$B = \frac{\text{No. of calls lost}}{\text{No. of calls offered}}$$

or,

$$B = \frac{\text{Traffic lost}}{\text{Traffic offered}} = \frac{A - A_0}{A}$$

where $A \rightarrow$ offered traffic, $A_0 \rightarrow$ carried traffic

$A - A_0$ = lost traffic.

→ Lower the G.O.S., better is the performance

→ If a system or server has G.O.S. as 0.002 means that 2 calls are lost in total of 1000 calls.

$\frac{2}{1000} = 0.002$

Eg (6): During the busy hour, 1200 calls were offered to a group of trunks and 6 calls were lost. The average call duration was 3 minutes. Find:

2007

(I) Traffic offered,

(II) " carried

(III) " lost.

(IV) G.O.S.

Soln; Here:

(I) Traffic offered $(A) = C \times t_h$

$$= \frac{1200}{60 \times 60} \times 3 \times 60 = 60E$$

(II) Traffic lost $= \frac{6}{60 \times 60} \times 3 \times 60 = 0.3E$

(III) Traffic carried = Traffic offered − traffic lost

$$= (60 - 0.3)\,E = 59.7\,E$$

(IV) $G.O.S. = \frac{\text{Traffic lost}}{\text{Traffic offered}} = \frac{0.3}{60} = 0.005.$

Smooth, Rough and Random Traffic:

→ Traffic probability distributions can be divided into three distinct categories:-

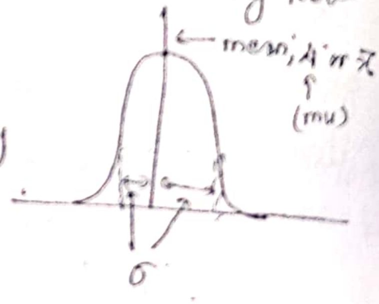
1. Smooth, $\alpha < 1$
2. Rough, $\alpha > 1$
3. Random, $\alpha = 1$, Poission

where, α = VMR = Variance to mean ratio.

→ Telephone-call originations in any particular area are random in nature. We find that originating calls or call arrivals at an exchange closely fit a family of probability distribution curves following a Poisson distribution.

→ The Poisson distribution is fundamental to traffic theory.

→ Most of the common probability-distribution curves are two-parameter curves, ie they may be described by two-parameters, 'mean' and 'variance'.



→ The mean is a point on probability distribution curve, where an equal number of events occur to the right of the point as to the left of the point.

→ The second parameter used to describe a distribution curve is the dispersion, and the dispersion of population or traffic can be measured by standard deviation where standard deviations 's' of sample 'n' observations, $x_1, x_2 - x_i$ is,

$$\text{'}\sigma\text{' or 's'} = \sqrt{\frac{1}{n-1}\sum_{i=1}^{n}(x_i-\bar{x})^2}$$

And, variance 'V' is given by, $V = s^2$ or σ^2

Thus, variance to mean ratio (VMR) is called the coefficient of over dispersion, $\boxed{VMR = \alpha = \frac{\sigma^2}{\mu}}$

→ Rough traffic tends to be peakier than random or smooth traffic. For a given grade of service (G.O.S.), more circuits are required for rough traffic because of the greater spread of the distribution curve.

→ Smooth traffic behaves like random traffic that has been filtered. This filtering is done at local exchanges.

→ Smooth traffic is characterised by Bernoulli's distribution.

→ If we assume subscribers make calls independently of each other and that each has a probability 'p' of being engaged in conversation, then if subscribers are examined, then the probability that 'x' of them will be engaged is,

$$B(x) = C_x^n p^x (1-p)^{n-x}; \quad 0 < x < n \quad \text{—— (1)}$$

its mean = np

its variance = $np(1-n)$.

→ Smooth traffic is assumed in dealing with small groups of subscriber number. The number 200 is often used as a breakpoint.

→ The smooth and rough traffic can be given by; Bernoulli (B) theorem as;

$$B'(x, s, h) = C_s^x h^x (1-h)^{s-x}$$

where, C_s^x = no. of combinations of 's' calls taken 'x' at a time

h = probability of finding 1st line of an exchange busy.

$1-h$ = " " " " " " an exchange idle

s = no. of subscribers.

→ The Poisson probability function can be derived from the binomial distribution, assuming that the number of subscribers 's' is very large and the calling rate per line 'h' is low such that the product 'sh' = 'm' remains constant and letting 's' increases to infinity in the limit.

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

where, $x = 0, 1, 2 \ldots$

→ Generally, we consider, call-holding times to have a negative exponential distribution in the form,

$$P = e^{-t/h}$$

where, t/h = average holding time

P = Probability of call lasting longer than, some arbitrary time interval

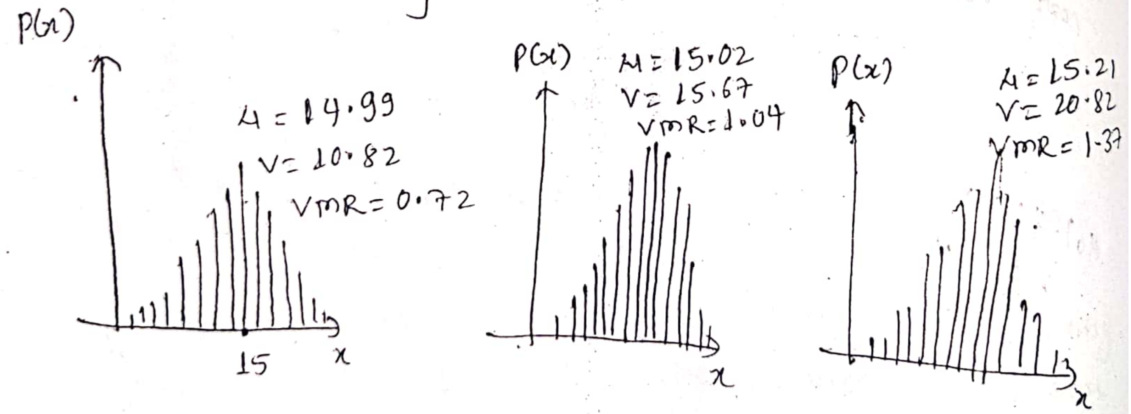


Fig: Smooth Random Rough

Eg. (7) A switching system serves 10,000 subscribers with a traffic intensity of 0.1E per subscriber. If there is a sudden ~~supurt~~ spurt in the traffic, increasing the average traffic by 50%, what is the effect in the arrival rate?

Soln: Here

No. of active subscribers $= 10,000 \times 0.1 = 1000$

Increase in traffic $= 1000 \times 1.5 = 1500$.

No. of available subscribers for generating ~~network~~ new traffic,
Normal traffic $= 9000$
Increased " $= 8500$

Change in arrival rate,

$$= \frac{9000 - 8500}{9000} \times 100$$

$$= \frac{500}{9000} \times 100$$

$$= 5.6\%$$ //

Little's theorem:

→ States that the mean number of customers in any queuing system is equal to the product of the mean waiting time for a customer and the mean arrival rate. The result is valid for all queuing system irrespective of the arrival time or service time distribution. Hence $m = \lambda \times T$

$$T = m/\lambda$$ //

Mathematical Model:

→ In order to obtain analytical solutions in tele-traffic problems, it is necessary to have mathematical model of the traffic offered to telecommunication system.

→ A simple model is based on the following assumptions:-

① Pure Chance Traffic

② Statistical Equilibrium.

→ The assumption of pure chance traffic means that call arrivals and call terminations are independent random events.

→ The assumption of statistical equilibrium means that the generation of traffic is stationary random process, i.e probability do not change during the period being considered.

→ Consequently, the mean no. of calls in progress remains constant. The assumptions lead to poisson's distribution:-

$$\boxed{P(x) = \frac{\mu^x e^{-\mu}}{x!} = \frac{(\lambda t)^k e^{-\lambda t}}{k!}}$$

where, $\mu = \lambda t$ is the average value of samples.

1 call/s
t = 10

1 call/5s

$\mu = \lambda t$

~~Eg. (8)~~

Eg. (8) A rural telephone exchange normally experiences four call originations per minute. What is the probability that exactly eight calls occur in an arbitrarily chosen interval of 30 sec.

Soln; Here,

$$\lambda = 4 \text{ calls/min} = \frac{4}{60} \text{ calls/sec}$$

$$= 1/15 \text{ calls/sec}$$

$t = 30 \text{ sec}$

$k = 8$

$$\therefore \lambda t = \frac{1}{15} \times 30 = 2$$

$$\therefore P_8(30) = \frac{2^8 e^{-2}}{8!} = 0.00086$$

Q. (9). On average 1 call arrives every 5 sec during a period of 10 sec what is the probability that:

(i) No call arrives, $\lambda = 2$, $p(0)$, $p(1)$, $\mu = 2$

(ii) 1 call arrive,

(iii) 2 call arrive,

(iv) more than 2 calls arrive

Soln: Here, $\mu = 2$ (for 10 sec)

Now, we have $P(x) = \frac{\mu^x e^{-\mu}}{x!}$

Sol.

(i) Probability that no call arrives;

$$P(0) = \frac{2^0 e^{-2}}{0!}$$

$$= \frac{1 \cdot e^{-2}}{1} = e^{-2} = 0.135.$$

(ii) Probability that 1 call arrives,

$$P(x=1) = \frac{2^1 e^{-2}}{1!} = \frac{2 \cdot e^{-2}}{\cancel{\ }}$$

$$= 0.270$$

(iii) Probability that 2 calls arrive,

$$P(x=2) = \frac{2^2 e^{-2}}{2!} = 2 \cdot e^{-2} = 0.270$$

(iv) Probability that more than 2 calls arrive,

$$P(x>2) = 1 - P(x \leq 2)$$

$$= 1 - \{P(x=0) + P(x=1) + P(x=2)\}$$

$$= 1 - \{0.135 + 0.270 + 0.270\}$$

$$= 0.325 //$$

1 5

1 5.

Poisson's distribution:

→ For a group of 'n' trunks, the number of calls in progress varies randomly.

The no of calls in progress is always between 0 & n.

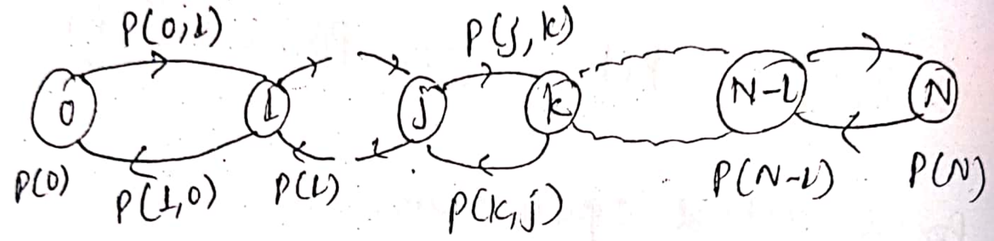


Fig. State transition diagram for N trunks

Here,
P(j) = Probability of state j.
P(k) = " " next higher state k.

→ P(j,k) is the probability of a state increase to 'k' given that present state is 'j'.

→ P(k,j) is the probability of a decrease to j' given that present state is 'k'.

→ The probabilities P(0), P(1), ---P(N) are called state probabilities and the conditional probabilities P(j,k) and P(k,j) are called transitional probabilities of Markov's chain.

→ If we consider the arrival call in 'A' for the holding time 'h' then probability of one call arrival during δ(t) is, $\boxed{P(0) = \frac{A\,\delta(t)}{h}}$

→ Similarly, if we consider no. of call termination is 'k' during holding period 'h', then

$$\boxed{P(e) = \frac{k\,\delta(t)}{h}}$$

Now,

$$P_{j\to k} = P(j)\,P(o) = P(j) \times \frac{A\,\delta(t)}{h}$$

$$P_{k\to j} = P(k) \times P(e) = P(k) \times \frac{k\,\delta(t)}{h}$$

For statistical equilibrium,

$$P_{j\to k} = P_{k\to j}$$

or $P(j)\,\frac{A\,\delta\cancel{(t)}}{\cancel{h}} = P(k)\,k\,\frac{\delta\cancel{(t)}}{\cancel{h}}$

$$\therefore \boxed{P(k) = \frac{A}{k}\,P(j)}$$

For $k=1$,

$$P(1) = \frac{A}{1}\,P(0)$$

$$P(2) = A/2\;P(1) = A/2 \times A\,P(0) = \frac{A^2}{2!}\,P(0)$$

$$P(3) = A/3\;P(2) = \frac{A^3}{3\times 2!}\,P(0)$$

$$\therefore \boxed{P(x) = \frac{A^x}{x!}\,P(0)}$$

→ 'x' can have values 0 to ∞ and the sum of probability must be 1.

$$1 = \sum_{x=0}^{\infty} P(x).$$

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$$\text{or, } 1 = \sum_{x=0}^{\infty} \frac{A^x}{x!} P(0)$$

$$\text{or, } 1 = e^{A} P(0)$$

$$\therefore P(0) = \frac{1}{e^{A}} = e^{-A}$$

$$\therefore \boxed{P(x) = \frac{A^x}{x!} \cdot e^{-A}}$$ //

2006-2010

LOST-CALL SYSTEM / Erlang B / LCC Model

(I) → It provides the probability of blockage at switch due to congestion

(2) → It determines G.O.S. of lost call system having 'n' trunks.

Assumptions: (I) Pure chance traffic

(II) Statistical equilibrium

(III) Full availability.

(3) → It means that every calls that arrive can be connected to any outgoing trunk which is free

(IV) calls which encounter congestion are lost, i.e finite no. of servers (x).

→ If there are 'x' calls in progress then, we have,

$$\boxed{P(x) = \frac{A^x}{x!} P(0)}$$

* If more marks then prove this as well from above

However, there can't be -ve no. of calls and there can't be more than 'N' servers.

$0 \le x \le N$.

$$\therefore \sum_{x=0}^{N} P(x) = 1$$

or $$\sum_{x=0}^{N} \frac{A^x}{x!} P(0) = 1$$

or $$P(0) = \frac{1}{\sum_{x=0}^{N} \frac{A^x}{x!}}$$

Now, Putting the value of $P(0)$.

$$P(x) = \frac{A^x}{x!} \times \frac{1}{\sum_{x=0}^{N} \frac{A^x}{x!}}$$

& the blocking probability is,

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

N serve

$0 \le n \le N$

$\sum_{n=0}^{N} p(n) = 1$

$\frac{A^n}{n!} p(0) = 1$

$p(0) = \frac{1}{\sum_{n=0}^{N} \frac{A^n}{n!}}$

Now, pu

$p(n) = \frac{A^n}{n!} \times$

Lost call Delay (LCD) System / Queuing System / 2nd Erlang Distribution / Erlang C

→ Erlang determined the probability of encountering delay when traffic A is offered to a queuing system with 'N' trunks.

Assumptions:

(i) Pure Chance traffic

(ii) Statistical equilibrium.

(iii) Full availability,

(iv) Calls which encounter congestion enter a queue and are stored there until a server becomes free

If $x \leq N$;

→ There is no queue and behaviour of system is the same as that of lost call system in absence of congestion

$$\boxed{P(x) = \frac{A^x P(0)}{x!}} \quad \text{for } 0 \leq x \leq N.$$

If $x \geq N$ (N served, $x-N$ in queue)

→ The probability of a call arrival in a very short period of time 'δt' is given by, $P(a) = \frac{A\,\delta(t)}{h}$, where 'h' is mean service time.

→ The probability of transition from $(x-1)$ to x calls during 'δt' is given by,

$$P((x-1) \to x) = P(x-1)\, A \frac{\delta(t)}{h}$$

→ The probability of transition from x to $(x-1)$ calls is given by,

$$P(x \to (x-1)) = P(x) \times N \frac{\delta(t)}{h}.$$

For statistical equilibrium,

$$P((x-1) \to x) = P(x \to (x-1))$$

call arrival = call termination

or $P(x)\, N \frac{\delta(t)}{h} = P(x-1)\, A \frac{\delta(t)}{h}$

or $P(x) = P(x-1) \frac{A}{N}$

for, $P(N+1) = P(N) \cdot A/N$

But, $P(N) = \frac{A^N}{N!} P(0)$.

or $P(N+1) = \frac{A^N}{N!} P(x) \frac{A}{N} = \frac{P(0)\, A^{N+1}}{N \cdot N!}$

Similarly,

$$P(N+2) = P(N+1)\, A/N$$

$$= \frac{P(0)\, A^{N+1}}{N N!} \times A/N.$$

or, $P(N+2) = \dfrac{P(0)\, A^{N+2}}{N^2 N!}$

In general, for $x \geq N$.

$$P(x) = \frac{P(0)\, A^x}{N^{x-N} N!}$$ (Here, $x = N+2$) $\Rightarrow x - N = 2$

$$= \frac{P(0) A^x}{N^x N^{-N} N!} = \frac{N^N}{N!} \cdot \left(\frac{A}{N}\right)^x \cdot P(0).$$

$$\therefore \boxed{P(x) = \frac{N^N}{N!} \left(\frac{A}{N}\right)^x P(0)} \quad //$$

This is Erlang C distribution

2007/2010.

Q. (10) In a busy hour, 100 subscribers initiate calls. The total duration of calls is 9600 sec

Calculate:

(I) the offered traffic

(II) the avg subscribers traffic

(III) the carried trunk traffic if GoS need to be maintained at 0.1

Sol^n Here

(i) Traffic offered (A) = $\lambda \times t_w$

$$= \frac{100}{60 \times 60} \times \frac{9600}{100}$$

$$= \frac{96}{36}$$

$$= 2.667\ E$$

(ii) Avg. subscriber traffic $= \frac{2.667\ E}{100}$

$\cancel{= 0.26}$

$= 0.02667\ E$

(iii) G.O.S. = 0.1

or $\frac{\text{Lost traffic}}{\text{offered traffic}} = 0.1$

or Lost traffic $= 0.1 \times 2.667$

$= 0.2667\ E$

So,
Carried trunk traffic

$= (2.667 - 0.2667)\ E$

$= 2.4003\ E$ //

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2007
Q. (11) During busy hour 900 calls were offered to a group of trunks & 90 calls were lost. The average call duration was 4 min. Find the lost traffic & G.O.S

Soln, Here

$$\text{traffic offered} = \frac{900}{60 \times 60} \times 4 \times 60$$

$$= 60E$$

$$\text{traffic lost} = \frac{90}{60 \times 60} \times 4 \times 60$$

$$= \frac{360}{60} = 6E$$

$$\therefore \ G.O.S = 6/60 = 0.1 \ //$$

2005
Q. (12) A telephone exchange normally experiences 4 calls per minute. What is the probability that:

(i) 8 calls occur in 30 sec interval

(ii) At least 5 calls arrive in 3 sec interval

Soln, Here

$$\mu = 4 \text{ calls per min}$$

$$= 4 \text{ calls}/60 \text{ sec} = 2 \text{ calls}/30 \text{ sec}$$

Thus,

$$\text{(i) } P(x=8) = \frac{e^{-\mu} \times \mu^x}{x!} = \frac{e^{-2} \times 2^8}{8!} = 0.000859 \ //$$

(ii) $P(x \geq 5) = 1 - P(x<5)$

$= 1 - [P(x=0) + P(x=1) + P(x=2) + P(x=3) + P(x=4)]$

$$= 1 - \left[\frac{0.2^0 \times e^{-0.2}}{0!} + \frac{0.2^1 \times e^{-0.2}}{1!} + \frac{0.2^2 \times e^{-0.2}}{2!} + \frac{0.2^3 \times e^{-0.2}}{3!} + \frac{0.2^4 \times e^{-0.2}}{4!} \right]$$

$= 2.258 \times 10^{-6}$ //

Imp.

Q. (13). A group of 5 trunk is offered 2 Erlang of traffic. Find:

(i) G.O.S.

(ii) Probability that only 1 trunk is busy

(iii) " " " " " free

(iv) " " at least 1 trunk is free

Soln: Here, $N=5$, $A=2E$

$$G.O.S. = P(N) = \frac{A^N/N!}{\sum_{k=0}^{N} \frac{A^k}{k!}} = \frac{2^5/5!}{\sum_{k=0}^{5} \frac{2^k}{k!}}$$

1 free means 4 busy trunk

$P(x \geq 4) = \cancel{P(x=\ldots)}$

$= P(x=5) + P(x=4)$

$=$

$\therefore$ G.O.S $= \dfrac{4/15}{1+2+2+8/6+\frac{2^4}{4!}+2^5/5!}$

$= 0.037$ //

(ii) $P(1) = \dfrac{2^1/1!}{\sum_{k=0}^{1} 2^k/k!} = 2/3 = 0.67$

(iii) 1 free in 5 $\xrightarrow{\text{means}}$ 4 busy.

So, $P(4) = \dfrac{2^4/4!}{1+2+2+8/6+2^4/4!} = 0.08333$ #

(iv) 1 trunk free $\xrightarrow{\text{means}}$ 4 busy

2 " " $\rightarrow$ 3 busy

3 " " $\rightarrow$ 2 busy

4 " " $\rightarrow$ 1 busy

5 " " 0 busy.

So, $P(x<5) = 1 - P(x \geq 5)$

$= 1 - P(x=5)$

$= -$

16/24

20/04/2009

Q (14) A group of 20 trunks provide a G.O.S. 0.01 when offered 12 E of traffic.

(i) How much is G.O.S. improved if 1 extra trunk is added to the group?

(ii) How much does G.O.S. deteriorate if 1 trunk is out of service?

Soln: Here $N = 20$, $G.O.S. = 0.01$, $A = 12E$

So
$$\underbrace{E_1, N(A)}_{G.O.S.} = \frac{A \times [E_1, N-1 (A)]}{N + A \times [E_1, (N-1) (A)]}$$

(i)
$$E_1\ 21 (12) = \frac{12 \times [E_1, 20 (12)]}{21 + 12 \cdot E_1, 20 (12)}$$

$$= \frac{12 \times 0.01}{21 + (12 + 0.01)} = 0.0057.$$

(ii) Now
$$E_1, 20 (12) = \frac{12 \times (E_1, 19 (12))}{20 + 12 \times [E_1, 19(12)]}$$

or
$$0.01 = \frac{12 \times [E_1, 19(12)]}{20 + 12 \times [E, 19(12)]}$$

or $E_1, 19(12) = 0.0165$ //

2005

Q (15) A group of 40 trunks ckts with G.O.S. 0.01 has 18 E of traffic

(i) What will be G.O.S. of if 2 additional trunk are added in group.

(ii) What will be G.O.S. if 2 additional trunk become faulty.

Protocols in Telecommunication

Chapter 9

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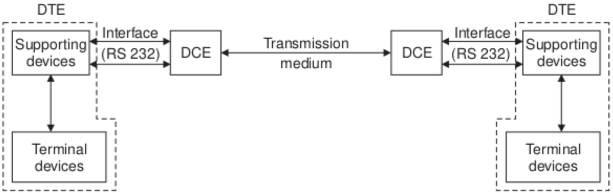
Data Communications Link

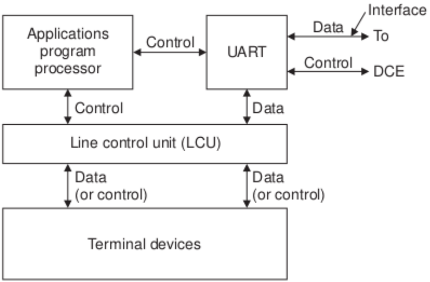
- In order to communicate from a terminal, computer or any equipment, the following:
  1. The transmission medium that carries the traffic between source and destination
  2. Data communication equipment or data circuit terminating equipment (DCE)
  3. Data terminal equipment (DTE)
  4. Communication protocols and software
  5. Terminal devices
  6. Interface



Data Communications Link

- **Transmission medium:**
  - The transmission medium include 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/nodes:**
  - End points in a communication link
  - Examples: Computer, eripherals such as printers, keyboards, FAX machines and data display terminals
- **Data terminal equipment (DTE):**
  - DTE: equipment which acts as source or destinations in digital communication
    - Capable of converting information to signals and also reconverting received signals
  - DTE usually do not communicate between each other, which is usually done by data communications equipment (DCE)
  - The terminal devices, communication station, UART, and line control unit (LCU) grouped together and named as DTE

Data terminal equipment (DTE)



- **UART:** The universal asynchronous receiver transmitter (UART) and the universal synchronous/asynchronous receiver transmitter (USART) performs the parallel to serial conversion (and vice versa at the receiving station)
- **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

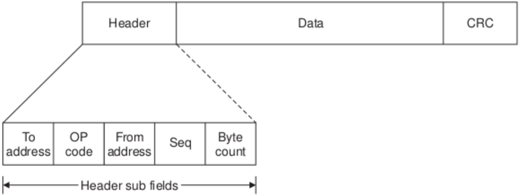
Data Communications Link

- **Application programme 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 provide the rules for communication between counterpart components on different devices
- **Interfaces:**
  - RS 232 interface is used to connect UART and the DCE
  - 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)**
  - DCE is used to convert the serial data stream into a form which is suitable for transmission
  - 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

Packet Switching

- The datastream originating at the source is divided into packets of fixed or variable size
- The data communication system typically have bursty traffic
  - the time interval between consecutive packets may vary, depending on the burstiness of the data stream
- Upper bound on packet length is 512 octets (bytes)
- Each packet contains a portion of the user's data plus some control information

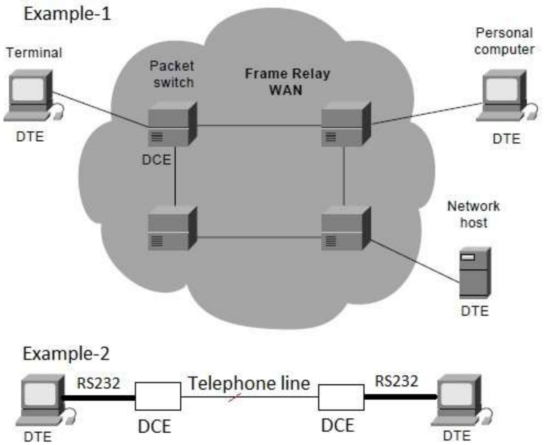
Packet Formats



A packet contains 3 major fields

1. **Header:** contains sub fields in addition to the necessary address fields
   1. **Op code:** designates whether the packet is a message (text) packet or control packet
   2. **sequence number (Seq)** to reassemble messages at the destination node, detect faults and facilitates recovery procedures
   3. **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
3. **Cyclic redundancy checks (CRC):** field contains a set of parity bits to detect errors

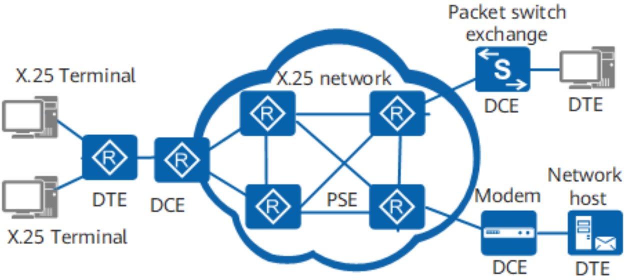
Data Communication



X.25

- X.25 is an ITU standard, well known and most widely used protocol
- An interface between DTE and DCE for terminal operation at the packet mode on public data network
  - how DTE's communicates with network
  - how packets are sent over the network using DCEs
- X.25 is a Connection-Oriented protocol: used in wide area network
- A packet-switched network consists of point-to-point connections
- Typically bandwidth offered is 2.4/9.6 kbps
- X.25 standard for packet switching is a lower three layer equivalent of the OSI model
  - a physical layer, a link layer, and packet layer

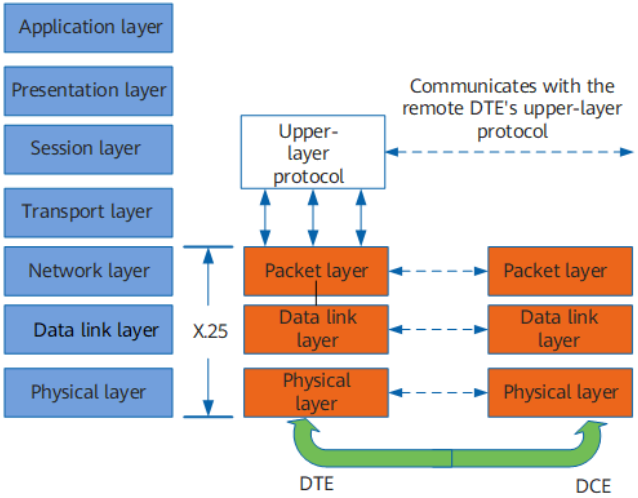
X.25 Network Model



X.25 Network Model

1. **Data terminal equipment (DTE):**
   - a communication terminal such as a telephone, personal computer, or network host on the X.25 network
   - DTE devices are used at the user side
2. **Data circuit-terminating equipment (DCE):**
   - a communication device such as a modem or packet switch on the X.25 network
   - DCE devices connect DTE devices and packet switching exchanges (PSEs), and are often used at the carrier side
3. **Packet switching exchange (PSE):**
   - a device used to transmit data from DTE devices

X.25 Protocol Layers



X.25 Protocol Layers

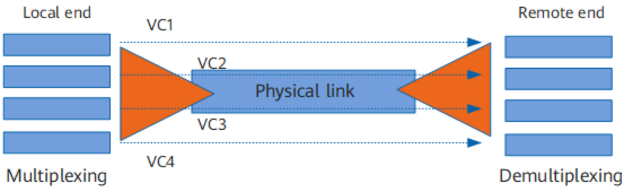
- **Physical layer:**
  - defines the specifications of electrical interfaces connecting DTE and DCE devices
  - X.21, X.21 bis and V.24 physical interfaces are being used
- **Data link layer:**
  - also known as Frame Layer
  - The data link layer of X.25 is link access procedure balanced (LAPB) using high level data link control (HDLC)
  - It also provides a communication link and transmission that is error-free among any two physically connected nodes or X.25 nodes
  - LAPB also allows DTE (Data Terminal Equipment) or DCE (Data Circuit-Terminating Equipment) simply to start or end a communication session or start data transmission

X.25 Protocol Layers

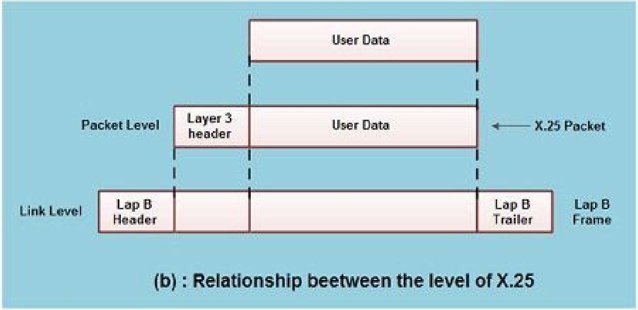
- **Packet layer:**
  - defines the packet format and packet exchange process between entities at the packet layer
  - defines how to implement procedures such as flow control and error processing
    - establishes, maintains, and terminates user sessions, handles addressing and carries out fault-management
  - provides external virtual circuit (VC) service
    - PVCs (Permanent Virtual Circuits): is a fixed, network assigned virtual circuit. Data transfer takes place as with virtual calls, but no call set up or clearing is required
    - SVCs (Switched Virtual Circuits): is established dynamically when needed through call set up procedure, and the circuit is relinquished through call clearing procedure

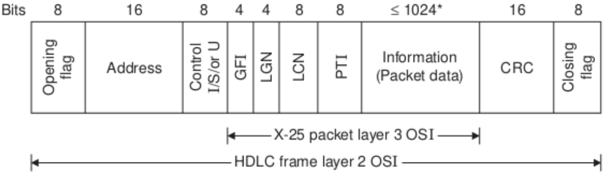
X.25 Protocol Layers: Packet layer

- A VC is a logical channel that is set up between the local and remote DTE devices
- A VC can pass through any number of intermediate nodes, such as data circuit-terminating equipment (DCE) devices and packet switching exchanges (PSEs)
- A VC consists of several logical channels
    - Each logical channel has an independent number, which is called a logical channel number (LCN)



X.25 packet format





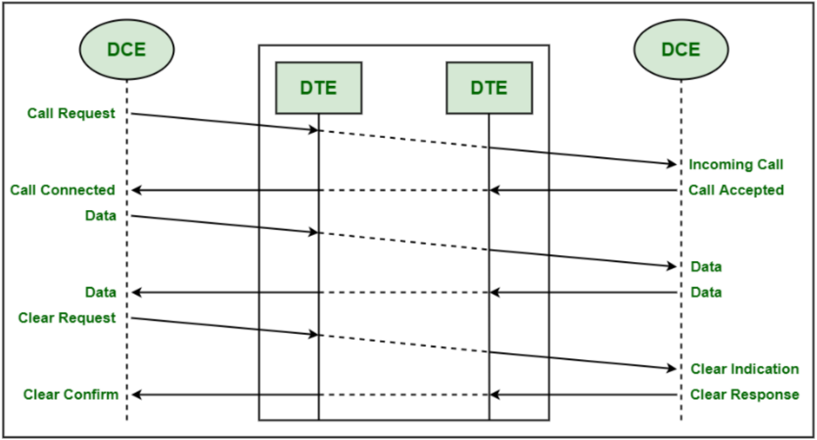
X.25 packet format

- Opening flag and closing flag are made up of 8 bit information
  - Packets are delimited by a starting and an ending flag (01111110)
- **Address:**
  - Typically one octet but can extended in increments of 8 bits (total 16 bits)
  - The value varies, depending on the direction of data flow, and on whether this is a single or multilink operation
- **Control I/S/ or U:**
  - consists of 8 bits of data describing the type of HDLC frame
  - **Information (I):** used to transfer data across the link at a rate determined by the receiver and with error detection and correction
  - **Supervisory (S):** used to determine the ready state of the devices receiver is ready (RR), receiver is not ready (RNR) or reject (REJ)
  - **Unnumbered (U):** used to dictate parameters, such as set modes, disconnect

X.25 packet format

- **Packet information:**
  - **General format identifier (GFI):** 4 bit of information that describes how the **data in the packet is being used**, from/to an end user, from/to a device controlling the end user device
  - **Logical channel group number (LGN):** 4 bits of information that describe the **grouping of channels**
  - **Logical channel number (LCN):** 8 bit of information of the actual channel being used
- **Packet type identifier (PTI):**
  - 8 bit sequence that describers the type of packet being sent across the network
  - Six types of packets are used in X-25
    - call request, call accept, clear request, interrupt request, reset request and restart request
- **Packet data:** contains encapsulated data from an upper-layer protocol like TCP/IP
- **Cyclic Redundancy Check (CRC):** The 16 bit sequence is used for error detection and/or correction

Virtual call procedure



Virtual call procedure: Call Setup Mode

- Only done using SVC's but not with PVC's (connection are already built)
- X.21 addressing scheme to usually set up the virtual circuit
- **Call Request Packet:**
  - The calling DTE sends this request as it is used to establish a connection on an SVC's
  - Includes the address of the remote DTE to be contacted
- **Call Accept Packet:**
  - called DTE returns packet when call is accpeted
  - network delivers this to the calling terminal as a 'call connected' packet
- **Clear Request Packet:**
  - packet is used to clear a call
  - either terminal may initiate clearing down of the call using packet sequence: 'clear request', 'clear request', 'clear indication' and 'clear confirmation'

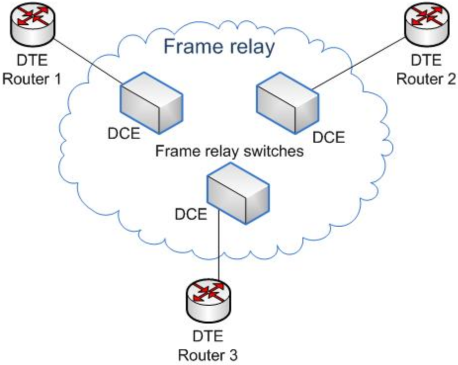
Virtual call procedure

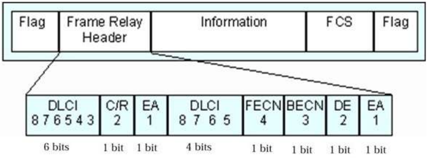
- **Data Transfer Mode:**
  - managed by packet layer protocol (PLP) among two DTE stations across an SVC's and also across PVC's
  - error and flow control are managed and handled
- **Idle Mode:**
  - basically used when a virtual call is established but there is no transferring of data
- **Restarting Mode:**
  - used for transmission synchronization among two DTE stations
  - mode is used prior to entering data transfer mode and also to re-establish synchronization whenever it is lost

Frame Relay

- packet-switching network protocol that is designed to work at the data link layer of the network
  - by dividing the data into packets known as frames and transmitting these packets across the network
- used to connect Local Area Networks (LANs) and transmit data across Wide Area Networks (WANs)
  - LAN to WAN
  - WAN to WAN
- technology called fast packet in which error checking does not occur in any intermediate node of the transmission but is done at the ends
- More efficient than X.25, and a higher process speed is achieved (it can transmit over 2.044 Mbps)
- Evolved from X.25 packet switching and the objective is to reduce network delays, protocol overheads and equipment cost

Frame relay





Standard Frame Relay Frame

Frame Relay Frame

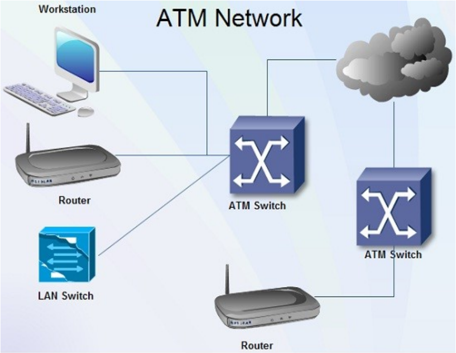
- **Flag:** Delimits the beginning and end of the frame. Consists of 1 byte (01111110)
- **Frame Relay Header:**
  - **Data-Link Connection Identifier (DLCI):**
    - Frame Relay virtual circuits are identified by unique DLCIs usually assigned by the Frame Relay service provider (for example, the telephone company)
    - excatly same as Logical channel number used in X.25
    - Frame Relay DLCIs have local significance, which means that their values are unique in the LAN, but not necessarily in the WAN
  - **Command/response (C/R):** not currently defined
  - **Extended address (EA):**
    - indicates whether the current byte is the final byte of the address
    - If EA = 1 it indicates that the current byte is the final one but if EA = 0, then it tells that another address byte is going to follow
  - **Congestion control:** a 3-bit field that contains the FECN, BECN and DE bits
  - **Data:** up to 16,000 bytes of user data
  - **Frame check sequence (FCS):** used for error detection

Frame Relay: Congestion control

- Due to high data transfer: congestion control is required
- 3 bits that control the Frame Relay congestion-notification mechanisms
- Forward-explicit congestion notification (FECN)
  - single-bit field that can be set to a value of 1 by a switch to indicate that congestion was experienced in the direction of the frame transmission from source to destination
- Backward-explicit congestion notification (BECN)
  - single-bit field that, when set to a value of 1 by a switch, indicates that congestion was experienced in the network in the direction opposite of the frame transmission from source to destination
- Discard eligibility (DE)
  - set by the DTE or by any switch in the network to value 1, then that frame can be discarded in the event of congestion

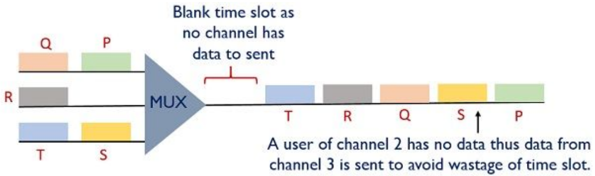
Asynchronous Transfer Mode (ATM)

- ATM switching technique standardized by ITU and ANSI
- ATM is universal switching system capable of handling all existing and expected future services, ranging from voice to video and all sorts of data: delay tolerant and intolerant, high and low speed



Asynchronous Transfer Mode (ATM)

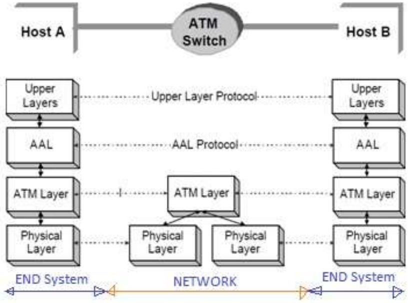
- ATM is a packet-switched technology
  - The packets that are switched in an ATM network are of a fixed length, 53 bytes (5 bytes header and 48 bytes payload), and are called cells
- Cells are transmitted asynchronously and the network is connection-oriented
  - requires connection setup before transmitting data i.e, virtual circuit
- Technique utilizes asynchronous time-division multiplexing to encode data into tiny and fixed-sized cells
- **Asynchronous** represents that the packets obtained from various users are not required to be transmitted at periodic intervals



ATM Operation

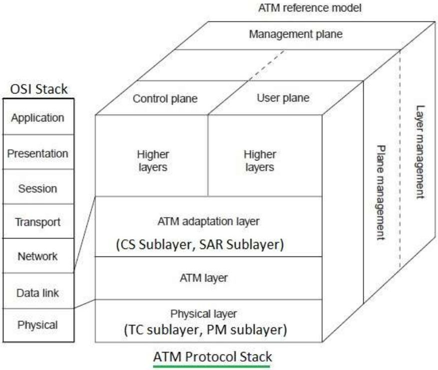
Electronics Desk

ATM End to End Protocol



ATM End to End Protocol Layers

ATM Protocol Architecture



- **User plane:** for user traffic including flow and error control
- **Control plane:** for connection control
- **Management plane:** manages the system as a whole and coordinates the planes and layers

ATM Architecture

- **Physical layer:**
  - Converts cells into a bitstream
  - Concerned with specifications of the transmission medium and signal encoding
  - Data rates specified include 155 and 622 Mbps with other data rates possible
- **ATM layer:**
  - Defines transmission of data in fixed size cells (Cell header generations)
  - Also defines the logical connections (Virtual circuits and virtual paths)
  - Responsible for simultaneously sharing the virtual circuits over the physical link known as cell multiplexing
  - Handles transmission, switching, congestion control, cell header processing
- **ATM Adaptation Layer (AAL):**
  - AAL accepts higher layer packets and segments them into fixed sized ATM cells (48 byte payload) before transmission via ATM

ATM Networks and Interfaces

1. **User-Network Interface (UNI):**
   - Interfacing is done between an end-user or subscriber and a specific ATM switch
   - Public and private in nature
2. **Network-Network Interface (NNI):**
   - Interfacing is done between two separate ATM switches
   - Public and private in nature

ATM Cell format

Header (5 octets)	Payload (48 octets)

Header

- **Generic flow control (GFC):**
  - 4 bits allocated for controlling the flow of data within the network but controlling is done before the data enters the network
    - maintain quality of service
  - only retained at the user-network interface (UNI)
- **Virtual channel identifier (VCI):**
  - 16 bits used for routing to and from an end user
  - virtual channel has a stream of cells carrying voice, data, video signals
- **Virtual Path Identifier (VPI):**
  - used with VCI to identify the next destination of a cell as it passes through a series of ATM switches
    - a routing field for the network
  - 8-bits at the UNI and 12-bits at the NNI

ATM Cell format

- **Payload type (PT):**
  - 3 bits indicate type of information in the cell payload
  - first bit indicates whether the payload is data or control data
  - second bit indicates congestion
  - third bit indicates whether the cell is the last cell in the series
- **Congestion loss priority (CLP):**
  - 1 bit field indicates whether a cell should be discarded if it encounters extreme congestion
  - If CLP $= 1$ indicates a cell that is subject to discard (lower priority cell)
  - bit is used for QOS
- **Header Error Control (HEC):**
  - 8 bit CRC detect all single errors and certain multiple bit errors

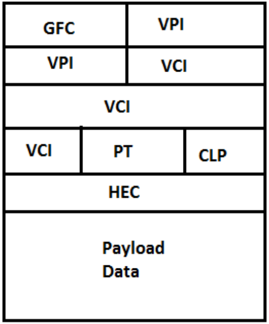
Payload: The payload consists of 48 octets which are specifically allocated for information (data, image, video, audio)

ATM frame format

GFC: Generic Flow Control

VPI: Virtual Path Identifier

VCI: Virtual Channel Identifier

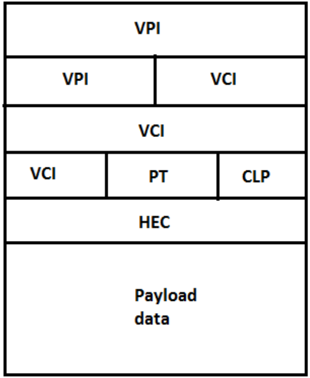


UNI Frame format

PT: Payload type

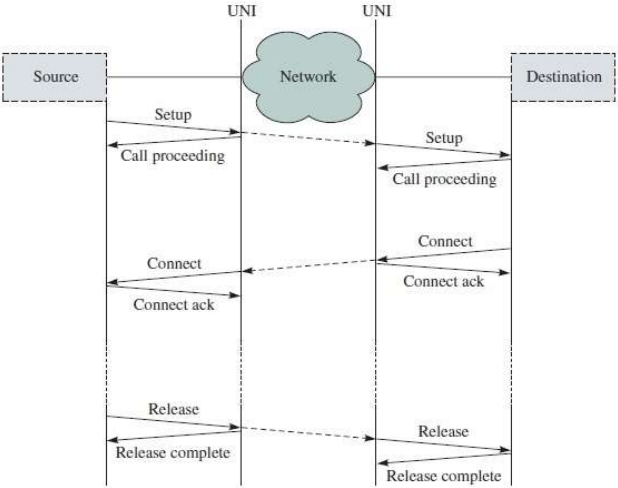
CLP: Cell loss priority

HEC: Header Error control



NNI Frame format

ATM UNI Signaling



ATM UNI Signaling Messages

ATM services

1. **Constant bit rate (CBR)**
   - CBR traffic is derived from the source, where the information is transmitted at a constant rate
   - Examples: Voice, Video, Television
2. **Variable Bit Rate (VBR)**
   - Service allows users to send traffic at a rate that varies with time depending on the availability of user information
   - Non-real time: Multimedia email
   - Real-time: voice with speech activity detection (SAD) and interactive compressed video (sensitive to cell delay variation)
3. **Available Bit Rate (VBR)**
   - When a carrier has allocated the necessary bandwidth on the links to carry CBR traffic and minimum VBR is guaranteed
   - ABR is the mechanism to share the remaining bandwidth fairly between the links
   - Example: data traffic (file transfer and e-mail)
4. **Unspecified Bit Rate (UBR)**
   - no guarantee about the bandwidth traffic delay and loss
   - control of flow in UBR can be provided from the end device

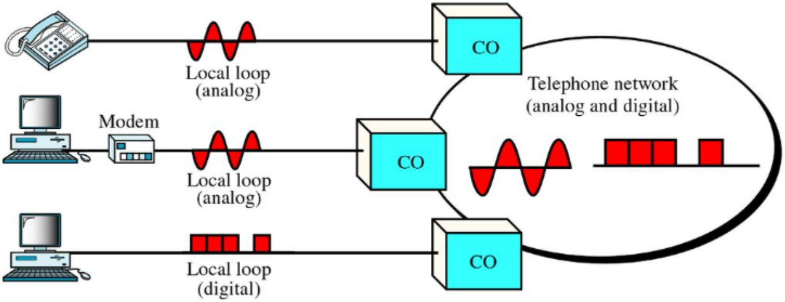
ATM Application

1. **ATM WAN:**
   - ATM can be used as a WAN to send cells over long distances, a router serving as an end-point between ATM network and other networks, which has two stacks of the protocol
2. **Multimedia virtual private networks and managed services:**
   - It helps manage ATM, LAN, voice, and video services and is capable of full-service virtual private networking, including integrated multimedia access
3. **Frame relay backbone:**
   - Frame relay services are a networking infrastructure for a range of data services and enable frame-relay ATM service to Internetworking services
4. **Residential broadband networks:**
   - ATM by choice provides the networking infrastructure for establishing residential broadband services in search of highly scalable solutions
5. **Carrier infrastructure for telephone and private line networks**
   - To make more effective use of SONET/SDH fiber infrastructures by building the ATM infrastructure for carrying the telephonic and private-line traffic

Integrated Services Digital Network (ISDN)

- A set of communication standards for simultaneous digital transmission of voice, video, data, and other network services over the traditional circuits of the PSTN
- ISDN is a circuit-switched telephone network system, but it also provides access to packet-switched networks that allows digital transmission of voice and data
- The goal is to form a WAN that provides universal end-to-end connectivity over digital media

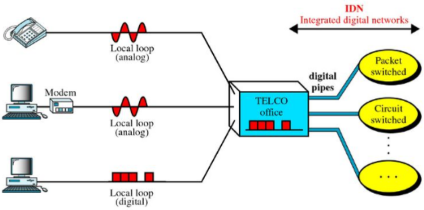
Before ISDN



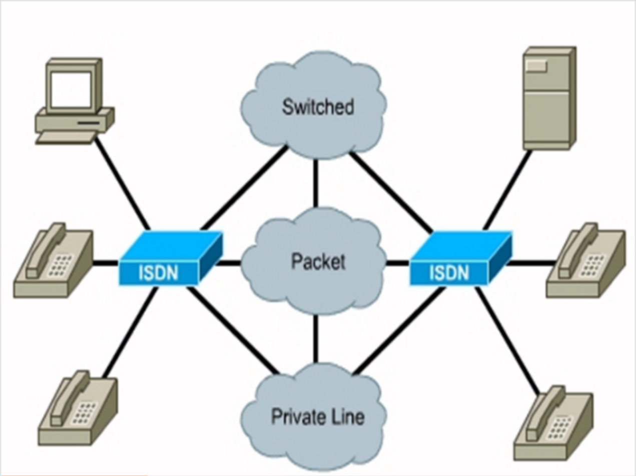
- Problem: how to transport digital services across a telephony infrastructure based on copper wiring was originally intended to carry analog signals only

Integrated Digital Network (IDN)

- Created Integrated Digital Networks (IDN): a combination of networks available for different purposes
- Access to these networks is by digital pipes which are time-multiplexed channels sharing very high speed paths



Integrated Services Digital Network (ISDN)



First set of ISDN standards

1. ISDN will be **based** on and will **evolve** from the **telephony IDN** by progressively incorporating additional functions and network features including those of any other dedicated networks so as to provide for existing and new services
2. New services introduced into the ISDN should be so arranged as to be **compatible with 64 kbps** switched digital connections
3. The transition from the existing networks to a comprehensive ISDN may require a period of time extending over one or two decades.
4. During the **transition period**, arrangements must be made for the interworking of services on ISDNs and services on other networks
5. The ISDN will contain intelligence for the purposes of providing service features, maintenance and network management functions. This intelligence may not be sufficient for some new services and may have to be supplemented by either additional intelligence within the network or possibly compatible intelligence in the customer terminals.
6. A layered functional set of protocols appears desirable for the various access arrangements to ISDN. Access from the customer to ISDN resources may vary, depending upon the service required and on the status of the evolution of national ISDNs

Features of ISDN

1. ISDN can handle a large range of voice (telephone) and non-voice (digital data) applications on the same network by utilizing a small number of defined facilities
2. ISDN connections can be switched or non-switched application
3. An ISDN will have intelligence for delivering service features, maintenance, and network management tasks.
4. The 64-Kbps digital link is the foundation of ISDN. New ISDN services should be compatible with 64 Kbps switched digital connections.
5. OSI (open system interconnection) standards can be utilized for ISDN.
6. Depending on national regulations, ISDN can be deployed in a variety of ways

ISDN channels

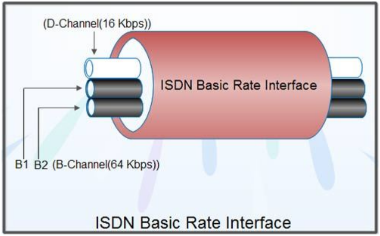
1. **Bearer channel (B channel)**
   - Basic user channel carries data and services at 64 kbps
   - Can carry any type of digital information in full-duplex modes as long as the required transmission rate does not exceed 64 kbps
   - Uses both circuit-switch and packet-switch connection
2. **Delta channel (D 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 signalling and administrative information for B channels to set up and tear down the calls
   - The D channel uses a packet-switched connection
3. **Hybrid channels (H channel)**
   - Carry user information at higher bit rates 384 kbps or 1536 kbps or 1920 kbps
   - Suitable for high data rate applications such as video, teleconferencing

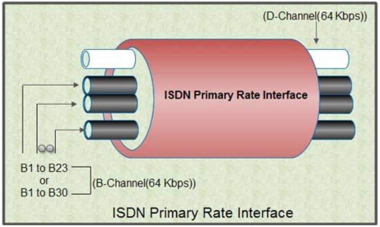
ISDN Channels

B Channel	D Channel	H Channel
64 Kbps	16/64 Kbps	384/1536/1920 Kbps
Digital Voice	Signalling (SS7)	High-speed trunk

ISDN Interfaces

1. **Basic Rate Interface (BRI)**
2. **Primary Rate Interface (PRI)**





ISDN Interfaces: Basic Rate Interface (BRI)

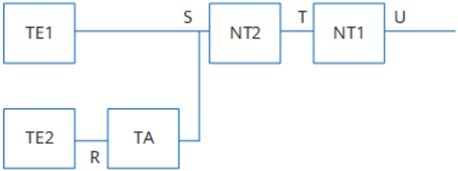
- Line that delivers two Bearer Channels (B-Channels) and one Delta Channel (D-Channel) over an ordinary telephone line
  - B-channels for voice, data and video
  - D-channels for signalling
- Get 144Kbps plus requires 48 Kbps operating overhead
  - BRI requires a digital pipe of 192 Kbps
- Intended primarily for home and small business use

ISDN Interfaces: Primary Rate Interface (PRI)

- Intended for users with higher data rate requirements, such as large business establishments, offices with a digital PBX or a LAN
- In North America 23 B channels and one 64 Kbps D channel and get 1.536 Mbps rate (similar to T1)
- In Europe 30 B channels and one 64 Kbps D channel and get 2.048 Mbps rate (similar to E1)
- Support voice and data

ISDN Reference Model

- ISDN provides users with a set of user-network interfaces (UNIs) to connect different terminals such as phones, fax machines and computers to ISDN networks
- Different terminals connect to the ISDN network through the same interface



Function groups in the ISDN reference model have the following functions:

- **Network Termination 1 (NT1):** implements physical layer functions, including subscriber line transmission, loop detection, and D channel preemption.

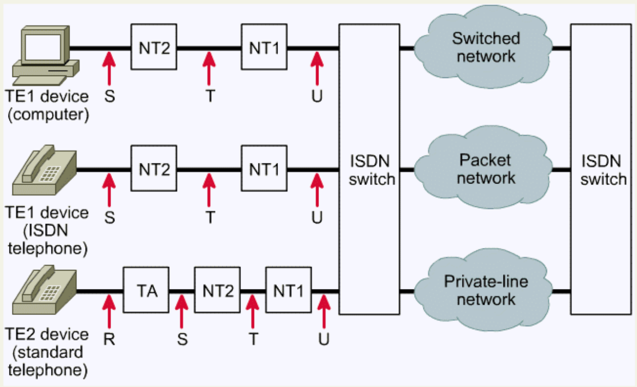
ISDN Reference Model

- **Network Termination 2 (NT2):** is an intelligent network terminal such as a private branch exchange (PBX) or a LAN router. Directs traffic to and from different subscriber devices and the NT1.
- **Terminal Equipment Type 1 (TE1):** is an ISDN standard terminal complying with the ISDN interface standard, such as a digital phone.
- **Terminal Equipment Type 2 (TE2):** is a non-ISDN standard terminal, which does not comply with the ISDN interface standard.
- **Terminal Adapter (TA):** implements the adaptation function to enable a TE2 to access a standard ISDN interface.

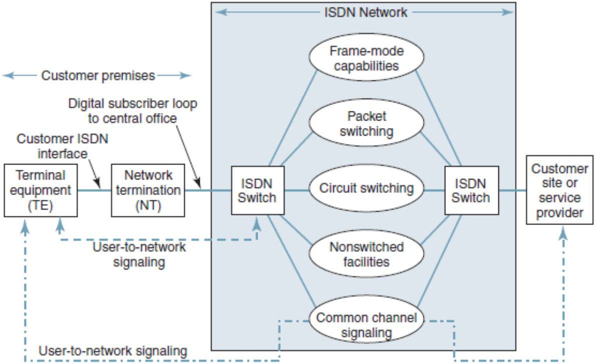
UNI interfaces support the following reference points:

- R: a reference point between a TE2 (non-ISDN standard) and a TA
- S: a reference point between a TE1 or TA and an NT
- T: a reference point between an NT1 and an NT2
- U: a reference point between an NT and an ISDN network

ISDN Reference Model



ISDN Architecture



- ISDN network is designed to support an entirely new physical connector for the customer, a digital subscriber loop, and a variety of transmission services
- A single interface will be used for telephones, computer terminals, and video equipment

ISDN Architecture

- Allow the exchange of control information between the customer's device and the ISDN network
- There are three basic types of ISDN channels:
  1. B channel: 64 kbps for voice, data and video
  2. D channel: 16 kbps or 64 kbps for signalling
  3. H channel: 384 kbps (H0), 1536 kbps (H11), or 1920 kbps (H12) for High speed trunk
- Residential users of the network be provided a basic access consisting of three full-duplex (BRI $2B + D$), TDM digital channels
- Primary service interface (PRI) that will provide multiple 64-kbps channels intended to be used by the higher-volume subscribers to the network
  - In North America $23B + D$ for a combined bit rate of 1.544 Mbps
  - In Europe $30B + D$ for a combined bit rate of 2.048 Mbps
- ISDN provide a circuit-switched B channel with the existing telephone system
- However, packet-switched B channels for data transmission at nonstandard rates would have to be created

ISDN Architecture

- Residential users of the network (i.e., the subscribers) be provided a basic access consisting of three full-duplex (BRI $2B + D$), time-division multiplexed digital channels
- Primary service interface (PRI) that will provide multiple 64-kbps channels intended to be used by the higher-volume subscribers to the network
  - In North America $23B + D$ for a combined bit rate of 1.544 Mbps
  - In Europe $30B + D$ for a combined bit rate of 2.048 Mbps
- ISDN provide a circuit-switched B channel with the existing telephone system
- However, packet-switched B channels for data transmission at nonstandard rates would have to be created

ISDN Protocol Architecture

	D Channel	B Channel
Layer3	DSS1(Q.931)	IP/IPX
Layer2	LAPD(Q.921)	PPP/HDLC/FR
Layer1	I.430/I.431	

- **Physical layer protocol:** B channels and D channels are multiplexed on the same physical interface
  - ISDN B channels and D channels at the physical layer use the same protocols: ITU-T I.430 (BRI) and ITU-T I.431 (PRI)
- **Data link layer protocol:**
  - ISDN does not define Layer 2 protocols dedicated to B channels
  - Link Access Procedure on the D-channel (LAPD) defined in Q.921 for D channels
  - Establishing and clearing data links
  - Error, flow and congestion control
  - Synchronization

ISDN Protocol Architecture

- **Network layer protocol:**
  - Does not define Layer 3 protocols dedicated to B channels
  - Defines Layer 3 protocols in the Q.931 standard for D channels
  - Addressing and routing
  - Establishing and clearing network-level connections

Link Access Protocol D-channel (LAPD): Layer 2

ISDN Layer 2 protocols comply with the Q.921 standard, which defines Link Access Procedures on the D-channel (LAPD). LAPD has the following functions:

- Transmits signaling packets over ISDN D channels.
- Allocates and manages terminal endpoint identifiers (TEIs).
- Controls the packet sequence.
- Detects and rectifies faults.
- Controls traffic to prevent links from being overloaded.

Format A

Flag	Address Field	Control Field	FCS	Flag

Format B

Flag	Address Field	Control Field	Information	FCS	Flag

- Format A: does not contains information fields
- Format B: contains information fields

LAPD frame format

- **Flag field:** is 1 byte in length and has a fixed value of 0x7E. A frame starts and ends with the Flag field.
- **Address field:** is 2 bytes in length

8	...	3	2	1
SAPI			C/R	EA1
TEI				EA2

  - Divided into two main Identifiers fields i.e. Service Access Point Identifier (SAPI) 6 bits and Terminal End Point Identifier (TEI) 7bits
  - **Command/Response field bit** (0 $\rightarrow$ command, 1 $\rightarrow$ response)
  - **EA1:** is the first address extension and has a value of 0
  - **EA2:** is the second address extension and has a value of 1

LAPD frame format

- **Control field:** is 1 byte or 2 bytes in length according to the frame type
  - **I:** indicates information frames, which transmit valid information or data
  - **S:** indicates supervisory frames, which are used for error control and flow control. Supervisory frames help to detect error data so that error data can be retransmitted
  - **U:** indicates unnumbered frames, which are used to establish, tear down, and control links
- **Information:** consists of an integer number of bytes and transmits TEI management messages or Layer 3 call management messages
- **Frame check sequence (FCS):** 2 bytes in length and ensures the validity of received frames. It provides detection by receiver of any errors that might have arisen or occurred during frame transmission.

ISDN Layer 3 Protocols

- ISDN Layer 3 protocols comply with the Q.931 standard
- The device currently uses Digital Subscriber Signaling System No.1 (DSS1) as the Layer 3 protocol
- The Q.931 standard defines methods to establish, maintain and terminate network connections (data calls and voice calls) between user-side devices and network-side ISDN switches.

Functions performed by layer 3

1. The processing of primitives for communicating with the data-link layer.
2. Generation and interpretation of layer 3 messages for peer level communications.
3. Administration of timers and logical entities (e.g., call references) used in call control procedures.
4. Administration of access resources, including B-channels and packet ayer logical channels (e.g., ITU-T X.25).
5. Checking to ensure that services provided are consistent with user requirements, such as compatibility, address, and service indicators.
6. Network Connection: mechanisms for providing network connections making use of data-link connections provided by the data-link layer.
7. Routing and Relaying: Network connections may involve intermediate systems which provide relays to other interconnecting subnetworks and which facilitate interworking with other networks. Routing functions determine an appropriate route between layer 3 addressees

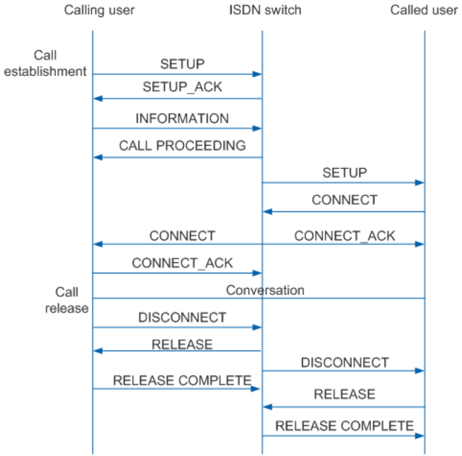
Functions performed by layer 3

8. Conveying User Information: may be carried out with or without the establishment of a circuit-switched connection.
9. Network Connection Multiplexing: Layer 3 provides multiplexing of call control information for multiple calls onto a single data-link connection.
10. Segmenting and Reassembly (SAR): Layer 3 may segment and reassemble layer 3 messages to facilitate their transfer across user–network interface.
11. Error Detection: Error detection functions are used to detect procedural errors in the layer 3 protocol. Error detection in layer 3 uses, among other information, error notification from the data-link layer.
12. Error Recovery: mechanisms for recovering from detected errors.
13. Sequencing: mechanisms for providing sequenced delivery of layer 3 information over a given network connection when requested.

Functions performed by layer 3

14. Congestion Control and User Data Flow Control: Layer 3 may indicate rejection or unsuccessful indication for connection establish requests to control congestion within a network. Typical is the congestion control message to indicate the establishment or termination of flow control on the transmission of User Information messages.
15. Restart: This function is used to return channels and interfaces to an idle condition to recover from certain abnormal conditions.

ISDN Layer 3 Protocols specification: call control process



Call establishment phase

- The calling user sends a SETUP message to initiate call establishment. The SETUP message contains all information required by the network to process this call.
- If the SETUP message contains no or incomplete called user information, an ISDN switch replies with a SETUP_ACK message and requests the remaining called user information
- The calling user sends one or more INFORMATION messages carrying the remaining called number and additional information to the ISDN switch.
- After the ISDN switch receives all required called user information, it replies with a CALL PROCEEDING message to indicate that requested call establishment has been initiated and sends a SETUP message to the called user to establish a call.
- If the called user answers the call, it sends a CONNECT message to the calling user to indicate that the call has been accepted.
- The calling user sends a CONNECT_ACK message.
- The call is established between the calling user and called user.

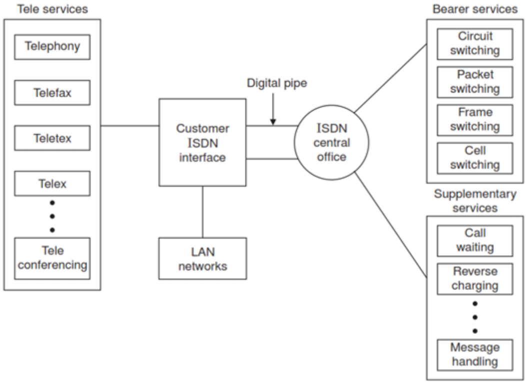
Call release phase

- The calling user sends a DISCONNECT message to terminate a connection.
- The ISDN switch sends a DISCONNECT message to the called user and sends a Release message to the calling user.
- The calling user releases the call and sends a RELEASE_COMPLETE to the ISDN switch.
- The called user receives the DISCONNECT message and sends a RELEASE message to the ISDN switch.
- The ISDN switch sends a RELEASE_COMPLETE message to the called user.
- The call is released.

ISDN Services

1. Bearer services
   - ISDN works on the principle of transport services known as bearer services.
   - The basic operation of the bearer service is the 64 kbps channel capacity.
   - Bearer services provide the means to transfer information (voice, data and video) between users. The network does not need to process the information.
   - Bearer service belongs to the first three layers of the OSI model.
   - These services can be provided with circuit-switched, packet-switched, frame-switched or cell-switched networks
2. Teleservices
   - In this service, the network may change or process the contents.
   - This service corresponds to layers 4–7 of the OSI model.
   - Tele services include telephony, telefax, video fax, telex and teleconferencing

3. Supplementary services
   - It provides additional functionality to the bearer service and teleservices.
   - Supplementary services include call waiting, Reverse charging, and message handling.



Broadband ISDN

- Broadband ISDN (B-ISDN) is defined as a network capable of supporting data rates greater than the primary rate (1.544 or 2.048 Mbps) supported by ISDN
- In the context of B-ISDN, the original ISDN concept is often termed narrowband ISDN (N-ISDN)
- The main aim of B-ISDN is to support video and image services.
- B-ISDN services are broadly classified as
  1. Interactive services
     1. Conversational services
     2. Messaging services
     3. Retrieval services
  2. Distribution services
     1. Broadcast services
     2. Cyclic services

Interactive services

Conversational services

- support end-to-end information transfer on real time, bidirectional basis
- In this service, the telephone instrument has the capability to transmit, receive and display video signals
- A dial-up connection brings about both video and audio transmission
- Other applications include video conferencing and video surveillance

Messaging services

- Offer store and forward communication
- Examples: voice mail, video mail and document mail containing text, graphics

Retrieval services

- Offer the capability to retrieve sound passages, high resolution images, graphics, short video scenes, animated pictures, etc. from centralised or distributed databases.
- Enhancement of videotex services in N-ISDN

Distribution services

Broadcast distribution services

- Provide support for broadcasting video, facsimile and graphical images to subscribers.
- Examples of such applications include television broadcasting over the network and electronic newspaper distribution.
- In broadcast services, the user has no control over what is being received on his/her screen

Cyclic distribution services

- offer some control to the user in the presentation of information on the screen
- an enhancement of the conventional teletext service.
- Unlike teletext where only textual information is transmitted, in cyclic distribution services, text, images, graphics and video and audio passages may be transmitted

B-ISDN support

- narrowband and broadband signals,
- interactive and distributive services,
- point-to-point, point-to-multipoint and broadcast connections,
- different traffic patterns (e.g. for voice, data and video),
- value-added services,
- multirate switched and nonswitched connections, and
- channel bandwidths up to 140 Mbps per service

Broadband channel rates

1. H2 channel, 30-45 Mbps
2. H3 channel, 60-70 Mbps
3. H4 channel, 120-140 Mbps.

Next Generation Network

Chapter 11

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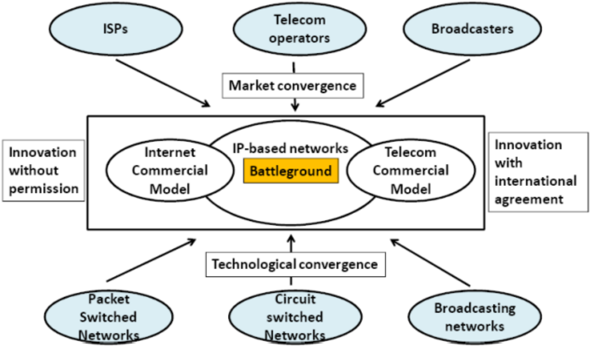
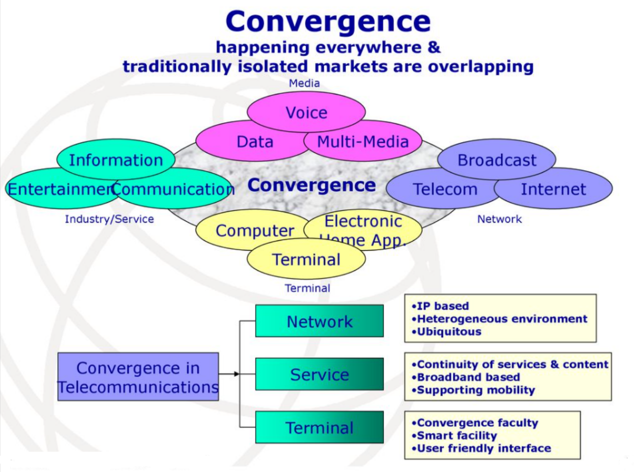
Convergence of the Media

- **Convergence:** represents the shift from traditional architecture to innovative models
- The path towards convergence was led mainly by the increasing digitalization of content, the shift towards IP-based networks, the diffusion of high-speed broadband access, and the availability of multimedia communication and computing devices
- **Convergence of media:** integration of various forms of communication and media, such as television, radio, print, and the Internet, into a single digital platform
- Convergence of media: significant changes in the way that people consume media and access information, and made easier to access a wide range of content from a single source.

Convergence Model



Convergence of the Media



Next Generation Network

- defined by ITU-T is a packet-based network able to provide services including telecommunication services and able to make use of multiple broadbands, QoS-enabled transport technologies and in which service-related functions are independent of underlying transport-related technologies.
- refers to a packet-based network and it can be used for both telecommunication services as well as data and it supports mobility.
- main goal of NGN is to serve/work as a replacement of PSTN and ISDN
- idea behind the NGN is that one network which transports all type of data and provides services in the form of packets similar to those which is used on the internet
    - NGNs are built around Internet Protocol (IP)

Features of NGN

- NGN works on Packet based transferring.
- There is an automatic separation of control functions among bearer capabilities, call/session and application/service.
- It supports a wide range of services, applications and mechanisms based on service building blocks.
- The network has Broadband capabilities with end-to-end QoS and transparency.
- This network also has a feature of interworking with legacy networks via open interfaces.
- It provides the advantage of general mobility.
- It provides unrestricted access by users to different service providers.
- It also provides a variety of identification schemes which can be resolved to IP addresses for the purpose of routing in IP network.
- It is composed of Unified service characteristics for the same services as perceived by the user.

Advantages of NGN

- Flexibility: multiple services can access
- Scalability: NGNs are designed to be scalable, allowing them to easily expand to meet the growing demand for communication services.
- Cost-effectiveness: NGNs can be more cost-effective than traditional circuit-switched networks, as they can support multiple services over a single network and do not require dedicated circuits for each service.
- Quality of service: NGNs can offer improved QoS, as they can prioritize different types of traffic and adjust the routing of packets based on the needs of the service being provided.
- It generates additional revenue streams for new IP/Ethernet services.
- It fulfils customer's demand for high bandwidth, Ethernet/ IP solutions.
- It gives End of Life/ End of Service vendor notification.
- Users can choose multiple service providers to take maximum advantage of competitive offers but may get single bill.

Challenge of NGN

- **Security:** NGN networks have a large attack surface, which makes them vulnerable to cyber threats. Protecting against these threats is critical.
- **Quality of Service (QoS):** NGN networks must be able to provide a high level of QoS to ensure that different types of traffic, such as voice and video, are delivered with minimal delays and disruptions.
- **Scalability:** NGN networks must be able to scale up or down as needed to accommodate changes in demand.
- **Cost:** Building and maintaining NGN networks can be expensive, which can be a challenge for service providers
- Migration complexities
- Not all legacy services can be replaced with new alternatives
- Not all existing infrastructure can be shut down
- Regulatory restrictions for critical services

Applications of NGN

- Voice Telephone services
- Multimedia services
- Data services
- Push to talk over NGN (PoN)
- Content delivery services
- Global mobility services
- Virtual Private Services (VPNs)
- Broadcasting/Multicast services
- e- commerce and m- commerce
- Session Controller based Internet services
- Third party/OSA based services
- 3D Imaging
- Machine to Machine communication
- Data Augmentation

Scope of NGN

1. **Support for multiple services:** NGNs can support a variety of communication services (voice, data, and video), over a single network.
2. **Packet-switched architecture:** which allows them to transmit data in small packets that are routed through the network to their destination
3. **Quality of service:** NGNs can offer improved QoS, as they can prioritize different types of traffic and adjust the routing of packets based on the needs of the service being provided.
4. **Scalability:** NGNs are designed to be scalable, allowing them to easily expand to meet the growing demand for communication services.
5. **Support for new technologies:** NGNs are designed to be modular and flexible, allowing new technologies and services to be easily integrated into the network.

Network Traffic

- Network traffic is the amount of data moving across a network at any given time.
- Network traffic/ data traffic is broken down into small data packets (transmitted efficiently) and sent over a network before being reassembled by the receiving device
- To better manage bandwidth, network administrators decide how certain types of traffic are to be treated by network devices like routers and switches
- Data traffic is categorised into two
  1. **Real-time Traffic:**
  2. **Non-real-time Traffic:**

Network Traffic

1. **Real-time Traffic:**
   - used to describe the movement of data packets across the network in which the packets are delivered as quickly as possible
   - packets are delivered in a best-effort manner, ie., packets may be delayed or lost due to congestion or other factors
   - packets are transmitted over the network using a protocol, such as UDP, which is optimized for real-time traffic
   - Examples of real-time network traffic include VoIP, videoconferencing, and online live streaming
2. **Non-real-time Traffic:**
   - Any data that is not required to be transmitted immediately, and can be stored and forwarded at the earliest opportunity.
   - Examples: email, web browsing, and file transfers.

Quality of Service (QoS)

- Quality of service (QoS) refers to any technology that manages data traffic to reduce packet loss, latency and jitter on a network.
- QoS controls and manages network resources by setting priorities for specific types of data on the network.

Benefits of Qos

- **Increased efficiency** of the network by optimizing the use of network resources
- **Improved performance** of critical applications by prioritizing their traffic and allocating the necessary resources
- **Better user experience** by ensuring that real-time applications such as voice and video receive a higher level of performance

Architecture used in QoS

1. Integrated Service
2. Differentiated Service

Quality of Service (QoS): Integrated Service

- a flow-based QoS model, which means that a user needs to create a flow, a kind of virtual circuit, from the source to the destination and inform all routers of the resource requirement
  - refers to an architecture that ensures the Quality of Service (QoS) on a network
- allows the receiver to watch and listen to video and sound without any interruption.
- each router in the network implements integrated services
- let every application requires some kind of guarantee to make an individual reservation.
- requires admission control and resource reservation protocol signalling
- Drawbacks:
  - scalability problem (maintain per-flow state in router)
  - implementation complexity (management, accounting)

Quality of Service (QoS): Differentiated Service

- developed to provide prioritized service mechanisms without requiring connection level information to be maintained at routers
- divide traffic into multiple classes, and treat them differently, especially when there is a shortage of resources
- a multiple service model that can satisfy many requirements
- minimizes signaling and concentrates on aggregated flows and per hop behaviour (PHB) applied to a network-wide set of traffic classes.
- Arriving flows are classified according to pre-determined rules, which aggregate many application flows into a limited and manageable set (perhaps 2 to 8) of class flows
- Advantages over Integrated Service
  - scalability
  - network management similar to IP neyworks
- Drawbacks
  - lack of demand and lack of support from router vendors

INTEGRATED SERVICES

VERSUS

DIFFERENTIATED SERVICES

INTEGRATED SERVICES	DIFFERENTIATED SERVICES
Architecture that specifies the elements to guarantee Quality of Service (QoS) on network	Architecture that specifies a simple and scalable mechanism for classifying and managing network traffic and providing QoS on modern IP networks
Involve prior reservation of resources before sending to achieve the required Quality of Service	Mark the packets with priority and send it to the network and do not require prior reservation
Also called IntServ	Also called DiffSer
Not scalable	Scalable
Involve per flow setup	Involve long term setup
Involve end to end service scope	Involve domain service scope

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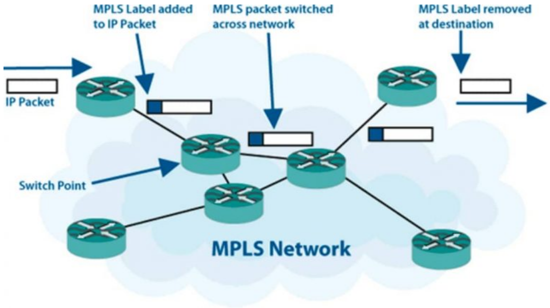
Multiprotocol Label Switching (MPLS)

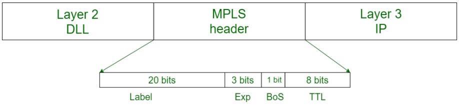
- is a networking technology that routes traffic using the shortest path based on "labels," rather than network addresses, to handle forwarding over WAN
- In an MPLS network, incoming packets are assigned a "label" by a "label edge router (LER)".
- Packets are forwarded along a "label switch path (LSP)" where each "label switch router (LSR)" makes forwarding decisions based solely on the contents of the label.
- At each hop, the LSR strips off the existing label and applies a new label which tells the next hop how to forward the packet.
- Label Switch Paths (LSPs) are established by network operators for a variety of purposes, such as
  - to guarantee a certain level of performance
  - to route around network congestion
  - to create IP tunnels for network-based virtual private networks
- Protocol-independent networking technology

MPLS

- network is Layer 2.5, operates between Layers 2 and 3 of the OSI model
  - Runs over existing Layer 2 ATM, Frame Relay and Ethernet networks
  - Also runs over Layer 3 IP networks
- MPLS is the ability to create end-to-end circuits, across any type of transport medium, eliminating the need for Layer 2 only control mechanisms



MPLS header



- Label – This field is 20 bit long
- Exp – They are 3 bits long and used for Quality of Service(QoS).
- Bottom of stack (BoS) – It is of size 1 bit. MPLS labels are stacked one over other. If there is only one label remained in MPLS header, then its value is 1 otherwise 0.
- Time to Live (TTL) – It is 8 bit long and its value is decreased by one at each hop to prevent packet to get stuck in network.

Advantages of MPLS

- **Cost:** Network resources can be easily shared in MPLS since it is a layer 3 technology. Network resources can be easily shared in MPLS since it is a layer 3 technology. Besides that, all of the customers data can be routed privately using MPLS
- **Scalability:** MPLS can be scaled up and down easily. Even if there is a requirement for thousands of sites.
- **Efficiency:** MPLS offers much higher quality connections without packet loss and jitter. Using it along with VoIP may lead to increased efficiency.
- **Reliability:** MPLS uses labels for forwarding packets, so it can be assured that the packets will be delivered to the right destination. Moreover, it is possible to assign network traffic according to priority.
- **Bandwidth:** MPLS allows multiple traffics to pass through the network of various data types, which means that bandwidth is optimally utilized.

Disadvantages of MPLS

- **Security:** security of MPLS solutions are totally in the hands of the user. There is no any inherent security features offered by the MPLS provider.
- **Maintenance:** The maintenance needs to be dependent on an ISP.
- **Control:** The configuration is taken care entirely by the service provider. The only control the user has through MPLS is dynamic routing
- **Accessibility:** MPLS is not optimized for cloud applications. It is made exclusive for point-to-point connectivity.
- **Deployment:** If the offices are present on different locations, then MPLS will take a long time to deploy.

Resource reservation protocol (RSVP)

- a transport layer protocol that is used to reserve resources in a network to get different quality of services (QoS) while accessing Internet applications
- used in real-time systems for an efficient quality band transmission to a particular receiver
- generally used by the receiver side for the fast delivery of the transmission packets from the sender to the receiver
- RSVP is not a routing protocol but a signaling protocol
- merely used to reserve resources along the existing route set up by whichever underlying routing protocol is in place
- RSVP is quite straightforward in packet format and operations, and so is relatively low cost in terms of implementation in end systems and routers

RSVP

- each RSVP operation only applies to packets of a particular session and as such every RSVP message must include details of the session to which it applies
- RSVP identifies a communication session by the combination of destination address, transport layer protocol type and destination port number.

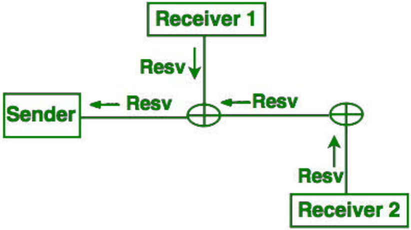
Features of RSVP

- RSVP is a receiver oriented signalling protocol. The receiver initiates and maintains resource reservation.
- It is used both for unicasting (sending data from one source to one destination) and multicasting (sending data simultaneously to a group of destination computers).
- RSVP is simplex (unidirectional). The receiver node just receives the packets and does not want to send any data.
- Quality of Service is provided by RSVP protocol

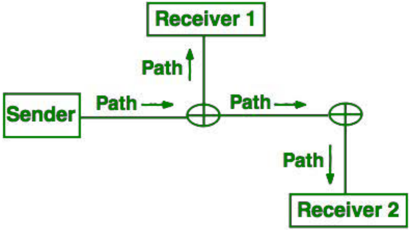
RSVP Messages

1. **Reservation Message (resv):** The receiver sends the Reservation Message (resv) to the sender, which specifies all the required resources and parameters for the reservation to establish
2. **Path Message (path):**
    - Upon receiving the reservation message from the receiver, the sender records all the necessary resources to be reserved and records the path.
    - The sender multicasts a Path Message (path) to all the receivers, which specifies the routing details of the packet.
    - It also contains all the necessary specifications about the reservation to be made for the receiver.
    - The data sent by the sender in Resource Reservation Protocol is encrypted to prevent the data sent to the receiver from breach.

RSVP Messages



Resv Messages



Path messages

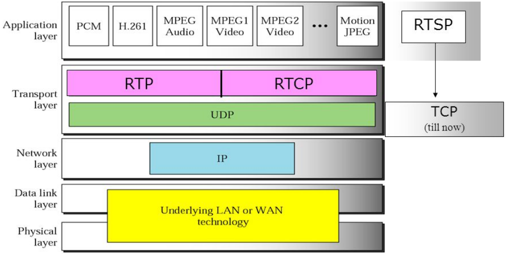
Real time transport protocol (RTP)

- a network protocol used to deliver streaming audio and video media over the internet, thereby enabling the Voice Over Internet Protocol (VoIP).
- RTP is generally used with a signaling protocol, such as Session initiation protocol (SIP), which sets up connections across the network
- RTP is used in conjunction with the RTP Control Protocol (RTCP)
    - RTP carries the media streams (e.g., audio and video)
    - RTCP is used to monitor transmission statistics and quality of service (QoS) and aids synchronization of multiple streams
- RTP has end-to-end transport capabilities for real-time applications on multicast or unicast network services.
    - widely used for interactive audio and video conferencing
- Information provided by RTP include timestamps (for synchronization), sequence numbers (for packet loss and reordering detection) and the payload format which indicates the encoded format of the data.

RTP

- Real-time multimedia streaming applications require timely delivery of information and often can tolerate some packet loss to achieve this goal.
  - For example, loss of a packet in audio application may result in loss of a fraction of a second of audio data, which can be made unnoticeable

Protocol stack for multimedia services



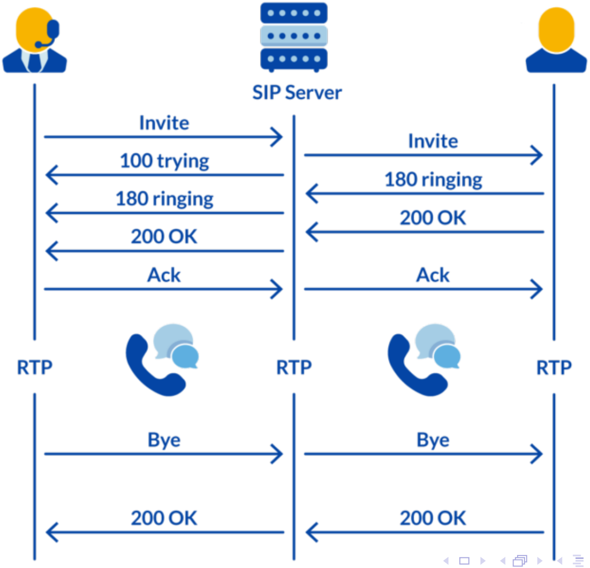
Session initiation protocol (SIP)

- an IETF (Internet Engineering Task Force)-defined signaling protocol, widely used for controlling multimedia communication sessions such as voice and video calls over IP
- protocol can be used for creating, modifying and terminating two-party (unicast) or multiparty (multicast) sessions consisting of one or several media streams.
  - The modification can involve changing addresses or ports, inviting more participants, and adding or deleting media streams.
- SIP takes the help of SDP (Session Description Protocol) which describes a session and RTP used for delivering voice and video over IP network.

Session initiation protocol (SIP)

- The SIP Protocol is text-based, and bears significant resemblance to the HTTP protocol
- The messages are text-based, and the request-response mechanism makes for easier troubleshooting.
- The actual data transmission is done by the TCP or the UDP on layer 5 of the OSI model
- Examples: Voice over IP (most common), Video conferencing, instant messaging, online gaming.

SIP call session



SIP call session

- The calling phone sends out an INVITE.
- The called phone sends an information response 100 – Trying – back.
- When the called phone starts ringing a response 180 – Ringing – is sent back.
- When the caller picks up the phone, the called phone sends a response 200 – OK.
- The calling phone responds with ACK – acknowledgement.
- Now the actual conversation is transmitted as data via RTP.
- When the person calling hangs up, a BYE request is sent to the calling phone.
- The calling phone responds with a 200 – OK.

Media gateway control (Megaco) signalling protocol

- introduced by the IETF and ITU to help control and manage the increasing volume of VoIP traffic
- requires the use of softswitches for call control and more resembles the telephony model of the circuit-switched PSTN than do SIP
- softswitch is aware of the entire call throughout its duration (it manages state) and enables operator intervention like the PSTN
- MEGACO protocol employs the master/slave architecture, where the Media Gateway Controller (MGC) acts as a master server, while the Media Gateway (MG) behaves like a slave device

Media Gateway Controller

- central point of intelligence for call signaling
- It maintains the states of each MG and responds appropriately to any event notification
- For instance, upon receiving an off-hook event from an MG, the MGC instructs the MG to play the dial tone and listen for the dual tone multi-frequency (DTMF) tones.

MEGACO Architecture

Media Gateway

- The master/slave architecture was designed to eliminate processor-intensive functionalities from the MG
- Due to the reduced complexity, the cost of MG is much lower than the cost of MGC, making it more affordable to the commercial and residential markets
- MG is a dumb terminal awaiting commands from the MGC for its next actions
- Upon the successful creation of a connection, the MG is also responsible for streaming the voice packets over the IP backbone using various encoding/compression algorithms.

MEGACO Architecture