TYPICAL QUESTIONS & ANSWERS

PART-I

OBJECTIVE TYPE QUESTIONS

Q.1. The number of point to point links required in a fully connected network for 50 entities is

(A) 1250 **(B)** 1225
(C) 2500 **(D)** 50

Ans: (C)

Q.2. For a non blocking cross bar configuration, taking N as the number of subscribers, there will be ______number of cross points and _______number of switches for establishing connections when all the subscribers are engaged.

(A) $N/2, N^2$ **(B)** N^2 , $N/2$
(C) $2N, N^2$ **(D)** $N/2, N^3$

Ans: (B)

Q.3. Echo suppressor is detrimental to full duplex operation because

(A) It disables one of the two pairs in a four-wire trunk line when a signal is detected on the other pair.
(B) It enables one of the two pairs in a four-wire trunk line when a signal is detected on the other pair.
(C) It activates both the pairs of a four-wire trunk line.
(D) It is independent of line conditions.

Ans: (A)

Q.4. Telephone companies normally provide a voltage of __________to power telephones.

(A) +24 volts DC **(B)** -24 volts DC
(C) +48 volts DC **(D)** -48 volts DC.

Ans: (D)

Q.5. The situation when both transmitter and receiver have to work in tandem is referred to as

(A) parallel **(B)** serial
(C) synchronous **(D)** asynchronous

Ans: (C)

Q.6. Common channel signalling______________

(A) Uses the speech or data path for signaling.
(B) Does not use the speech or data path for signaling.
(C) Needs no additional transmission facilities.
(D) Finds it difficult to handle signaling during speech.

Ans: (C)

Q.7. A large numbers of computers in a wide geographical area can be efficiently connected using
(A) twisted pair lines
(B) coaxial cables
(C) Communication satellites
(D) all of the above

Ans: (D)

Q.8. Which transmission mode is used for data communication along telephone lines?
(A) Parallel **(B)** Serial
(C) Synchronous **(D)** Asynchronous

Ans: (B)

Q.9. A sample rate of ________is required for a good quality representation of telephone conversation.
(A) 4500 times per second.
(B) 700 integer sample points per minute.
(C) 50 times per second per mile of distance travelled.
(D) 8000 times per second.

Ans: (C)

Q.10. The ________is a circuit-switched network, while the ______is a packet-switched network.
(A) Telephone, ATM **(B)** SONET and FDDI
(C) Satellite, Telephone **(D)** FDDI and SONET

Ans: (A)

Q.11. A Master group consists of
(A) 12 voice channels. **(B)** 24 voice channels
(C) 60 voice channels. **(D)** 300 voice channels

Ans: (D)

Q.12. Direct inward dialling is used as a feature in
(A) PSTN. **(B)** PBX.
(C) EPABX. **(D)** VPN.

Ans: (C)

Q.13. Trunks are the lines that run between
(A) Subscribers and exchange
(B) switching system and power plant
(C) Local area network.
(D) Switching stations.

Ans: (D)

Q.14. Traffic Capacity is given by
(A) Switching capacity × Theoretical maximum load
(B) Switching capacity / Theoretical maximum load
(C) Theoretical maximum load / switching capacity
(D) Theoretical maximum load × Switching capacity

Ans: (A)

Q.15. In a time multiplexed space switching system, one speech sample appears every
(A) 125 micro sec **(B)** **20** msec
(C) 125 msec **(D)** 1 sec

Ans: (A)

Q.16. ISDN handles data pertaining to
(A) All digital services **(B)** Speech and Video
(C) Computer data only **(D)** Speech only

Ans: (A)

Q.17. A star connected intermediate exchange is known as a
(A) Repeater exchange
(B) Hub exchange
(C) Private branch exchange
(D) Tandem exchange

Ans: (B)

Q.18. Time synchronization is necessary in
(A) FDM. **(B)** TDM.
(C) WDM. **(D)** Quadrature multiplexing

Ans: (B)

Q.19. In a frame transmission, CRC stands for
(A) Code Renewable Check
(B) Cyclic Redundancy Check
(C) Control and Refresh Code
(D) Cyclic Refreshing of CPU

Ans: (B)

Q.20. In a LAN network every system is identified by
(A) Name **(B)** MAC address
(C) IP Address **(D)** Serial number given by manufacturer

Ans: (C)

Q.21. An off-hook signal will repeat for a/an ________duration.

(A) finite
(B) infinite
(C) duration of 40 seconds
(D) duration of 80 seconds

Ans: (A)

Q 22. Typical human voice is centered around ________Hz.
(A) 200-400 (B) 280-3000
(C) 400-600 (D) 1400-1800

Ans: (B)

Q 23. Using_________each connected device is assigned a time slot whether or not the device has any thing to send.
(A) WDM (B) FDM
(C) TDM (D) STDM

Ans: (C)

Q.24. When a switch capacity is full, calls coming into that switch are said to be____.
(A) open (B) shorted
(C) blocked (D) shunted

Ans: (C)

Q.25. Using ______ARQ, a sending modem must wait for a return ACK for each sent block before sending the next block.
(A) discrete (B) efficient
(C) continuous (D) delivered

Ans: (A)

Q.26. A/An ______network is typically a company network that connects multiple company locations into a single network.
(A) local area (B) enterprise
(C) campus wide (D) protocol.

Ans: (B)

Q.27. Ethernet 10 Base 2 is an example of ______network topology.
(A) Bus (B) Ring
(C) Star (D) Mesh

Ans: (A)

Q.28. The______electro mechanical switch (developed in 1938) had fewer moving parts than earlier switches.
(A) No. 1ESS (B) Strowger
(C) Step-by-step (D) Crossbar

Ans: (D)

Q.29. Side tone is the speech heard by
(A) the receiving subscriber
(B) both the receiving and calling subscriber
(C) by on looker
(D) by calling subscriber

Ans: (D)

Q.30. Busy hour traffic is the
(A) maximum average simultaneous traffic.
(B) traffic during peak hour.
(C) traffic when all subscribers are engaged.
(D) the duration of maximum calls.

Ans: (B)

Q.31. The final selector is connected to the
(A) calling subscriber. **(B)** switching network.
(C) called subscriber. **(D)** line finder.

Ans: (C)

Q.32. In a DTMF phone a dialling of 8 generates
(A) 1336 Hz- 770 Hz **(B)** 1209 Hz - 1477 Hz
(C) 1209 Hz- 941 Hz **(D)** 1336 Hz-852 Hz

Ans: (D)

Q.33. SPC stands
(A) Standard Protocol Control
(B) Stored Program Control
(C) Signaling and switching Centre
(D) Signaling Process Center

Ans: (B)

Q.34. For two stage network the switching elements for M inlets with r blocks and N outlets with s blocks is given by
(A) Ms + Nr **(B)** Mr + Ns
(C) (M + N) (r + s) **(D)** (M + N)rs

Ans: (A)

Q.35. As per Nyquist criterion the sampling rate is
(A) 2fs **(B)** (1/2)fs
(C) (1/2fs) **(D)** (2/fs)
Where fs is the signal frequency

Ans: (A)

Q.36. Common channel signalling in SS7 is
(A) out band control channel.
(B) in band control channel.
(C) speech control channel.
(D) none of the above.

Ans: (B)

Q.37. Broad Band ISDN handles data rate of about
(A) 64 kbps **(B)** 100 mbps
(C) 5.4 mbps **(D)** 2.048 mbps

Ans: (A)

Q.38. MAC address helps in
(A) multimedia access control.
(B) media access control.
(C) mobile access control.
(D) master access point control

Ans: (B)

Q.39. Telex is a
(A) Telephone Service between various subscribers
(B) Tele printer Service between various subscribers
(C) Television Service between various subscribers
(D) Telegraph Service between various subscribers

Ans: (B)

Q.40. The bandwidth requirement of a telephone channel is
(A) 3 KHz **(B)** 15 KHz
(C) 5 KHz **(D)** 25 KHz

Ans: (A)

Q.41. Distortion caused on telephone line by an adjacent one is called
(A) Cross Fire **(B)** Inductive Disturbance
(C) Cross Talk **(D)** None of these

Ans: (C)

Q.42. Erlang is used to
(A) Measure busy period **(B)** Give total busy period in minutes
(C) Measure average call rate **(D)** Indicate total call period

Ans: (A)

Q.43. The grade of service is measured in
(A) Percentage **(B)** Number
(C) Fractional Number **(D)** Logarithmic Number

Ans: (C)

Q.44. Network with point-to-point link is known as
(A) Fully Connected Network **(B)** Half Connected Network
(C) Duplex Connected Network **(D)** None of these

Ans: (A)

Q.45. SPC is used for
(A) Carrying Exchange Control Functions
(B) Carrying Subscriber Control Functions
(C) Exchange Hardware
(D) Signalling Purpose

Ans: (A)

Q.46. Trunks are the lines that run between
(A) subscribers and exchange **(B)** switching system and power plant
(C) Local Area Network **(D)** switching systems

Ans: (B)

Q.47. Example of circuit switching and S&F (Stored and Forward) switching is
(A) Telephone and Post of Telegraph
(B) Video Signal Post or Telegraph
(C) Digital Signal Post or Telegraph
(D) None of above

Ans: (A)

Q.48. Network Layer is used for
(A) Breaking up the data in frames for transmission
(B) Deal with Error correction
(C) Automatic Recovery of Procedure
(D) Physical Architecture

Ans: (D)

Q.49. Call request signal is:
(A) Seize signal
(B) Idle state signal
(C) Line identification signal
(D) Called subscriber alert signal

Ans: (A)

Q.50. Telephone Traffic is measured in
(A) Seconds. **(B)** Hours.
(C) Erlang **(D)** Pulses per minute.

Ans: (C)

Q.51. In step by step switching line finders are connected to the
(A) Calling subscriber. **(B)** Switching network.
(C) Called subscriber. **(D)** Between exchanges.

Ans: (A)

Q.52. In a DTMF phone, digits are represented by:
(A) Orthogonal frequencies.
(B) Orthogonal Phases.
(C) Orthogonal codes.
(D) Orthogonal pulses.

Ans: (A)

Q.53. Companding helps in reducing __________ with respect to signal:
(A) Interference **(B)** Signal overloading
(C) Non linearity **(D)** Quantization noise

Ans: (D)

Q.54. SS7 Protocol uses:
(A) Out of band signalling.
(B) Associated signalling.
(C) Speech control signalling.
(D) No signalling.

Ans: (A)

Q.55. MAC is the abbreviation for:
(A) Multimedia access control
(B) Media access control
(C) Mobile access control
(D) Master access point control

Ans: (B)

Q.56. The CCITT standard bandwidth for speech is:
(A) 20000 Hz **(B)** 15000 Hz
(C) 7000 Hz **(D)** 3400 Hz

Ans: (D)

Q.57. Maximum channel utilization in a LAN is defined by frame time (t_f) and propagation time (t_p). It is defined by
(A) t_p/t_f **(B)** t_f/t_p
(C) $1+(t_f/t_p)$ **(D)** $t_f/(t_p+t_f)$

Ans: (D)

Q.58. The function of ARQ in a network protocol is to:

(A) Auto request **(B)** Acknowledge
(C) Address request **(D)** Abort

Ans: (A)

Q.59. Engaged tone is generated in the:
(A) Telephone instrument of calling subscriber
(B) Telephone instrument of called subscriber
(C) Exchange
(D) Repeater

Ans: (C)

Q.60. One Erlang is equal to
(A) 3600 CCS
(B) 36 CCS
(C) 60 CCS
(D) 24 CCS

Ans: (A)

Q.61. The analog signal needs to be sampled at a minimum sampling rate of:
(A) 2fs **(B)** 1/(2fs)
(C) fs/2 **(D)** 2/fs

Ans: (A)

Q.62. In a time division space switch the size of the control memory is N and its Width:
(A) $Log_{10}M$ **(B)** $Log_{e}M$
(C) $Log_{N}M$ **(D)** $Log_{2}M$
Where N are the outlets and M the number of data samples

Ans: (It should be 2 log_2[N]

Q.63. In a single stage network:
(A) There is no redundancy
(B) There is redundancy
(C) Alternative cross points are available
(D) Alternative paths are available

Ans: (B)

Q.64. Signalling transfer point (STP) exist in
(A) Strowger exchange **(B)** SS7
(C) Local area network **(D)** PABX

Ans: (B)

Q.65. ARQ is transmitted in the event of:

(A) Loss of signal **(B)** Error in received data
(C) Improve reliability **(D)** During time out

Ans: (B)

Q.66. Computer to computer communication is:
(A) Simplex **(B)** Duplex
(C) Half Duplex **(D)** Both Duplex and Half Duplex

Ans: (B)

Q.67. A distributed network configuration in which all data/information pass through a central computer is
(A) Bus network **(B)** Star network
(C) Ring network **(D)** Point to point network

Ans: (B)

Q.68. An important terminal that is required between DTE and PSTN is
(A) Server **(B)** MODEM
(C) Relay **(D)** Network card

Ans: (B)

Q.69. Traffic Handling Capacity is given by
(A) Switching capacity $\times$Theoretical maximum load
(B) Switching capacity / Theoretical maximum load
(C) Theoretical maximum load / Switching capacity
(D) Theoretical maximum load + Switching capacity

Ans: (B)

Q.70. Traffic Intensity can be measured in
(A) Erlangs **(B)** CCS
(C) CM **(D)** All of the above

Ans: (D)

Q.71. Trunks are the lines that run between
(A) Subscribers and exchange
(B) Switching system and power plant
(C) Local area network
(D) Switching stations

Ans: (D)

Q.72. Packet switching is used for
(A) Credit card verification
(B) Automated Teller Machine
(C) The internet and the World Wide Web
(D) All of the above

Ans: (D)

Q.73. Analog signals can be ________by combining them with a carrier frequency
(A) Carried **(B)** Transported
(C) Multiplexed **(D)** Mixed

Ans: (C)

Q.74. The Signalling connection control part (SCCP) and message transfer part (MTP) together are referred to as
(A) Signal Switching Points (SSPs)
(B) Signal Transfer Points (STPs)
(C) Signal Control Points (SCPs)
(D) Network service part (NSP)

Ans: (D)

State True or False

Q.75. A two stage non-blocking network requires twice the number of switching elements as the single stage non-blocking network.
(A) TRUE **(B)** FALSE

Ans: (A)

Q.76. The larger the Grade Of Service, the worse is the service given
(A) TRUE **(B)** FALSE

Ans: (A)

Q.77. A certain amount of side tone is essential in telephone communication
(A) TRUE **(B)** FALSE

Ans: (A)

Q.78. Sky wave Communication is prone to fading
(A) TRUE **(B)** FALSE

Ans: (A)

PART-II

DESCRIPTIVES

Q.1. With neat diagrams explain the configuration of a step-by-step switching system. **(10)**

Ans:
The schematic diagram for such an exchange is given in Fig. Each subscriber is connected to a single rotary pre-selector switch at the exchange, the outputs from this switch being connected to a bank of two-motion switches known as 'group selectors'. The out-puts from the pre-selector switches of a whole group of subscribers are connected together in parallel as that group of subscribers share a single bank of group selectors. When a subscriber lifts his telephone, the cradle switch causes a circuit to be completed back to the telephone exchange, signaling that the subscriber wishes to make a call. This causes the pre-selector switch to step around until it finds a free group selector. The pre-selector switch then stop in this position and the group selector is 'seized' by the subscriber wishing to make a call.

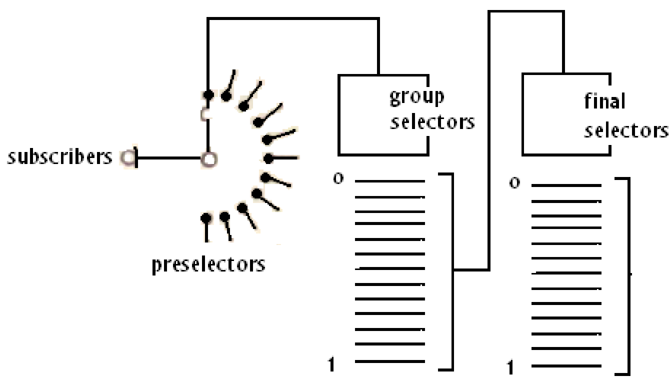


FIG – 1000 Line Exchange

On seizing the group selector, the subscriber dials his first digit and the selector switch moves up to the appropriate row on the switch contact array. Each final selector has the possibility of connection to 100 lines. The 1000 lines are therefore divided into 10 groups of 100 each, the group being identified by the first digit in the subscriber's number. The vertical motion of the group selector therefore selects a final selector in the group associated with the first digit dialed. Each individual row of contacts, or levels, of the group selector is connected to a bank of final selectors associated with a particular group of 100 line numbers. Having dialed the first digit to select the appropriate group, the group selector arm then automatically rotates in the vertical direction until it finds a free final selector. In the final selector, both directions of motion are under the control of the subscriber's dial and, after dialing two further digits, connection is established, providing the called subscriber's to answer his telephone.

Q.2 What is store program control (SPC)? Give the organization of centralized SPC. Discuss the advantages of SPC automation in telephone switching. **(10)**

Ans:

In **stored program control systems**, a program or set of instructions to the computer is stored in its memory and the instructions are executed automatically one by one by the processor. Carrying out the exchange control functions through programs stored in the memory of a computer led to this name.

There are two approaches to organizing stored program control:

1. **Centralized**: In this control, all the control equipment is replaced by a single processor which must be quite powerful.
2. **Distributed:** In this control, the control functions are shared by many processors within the exchange itself.

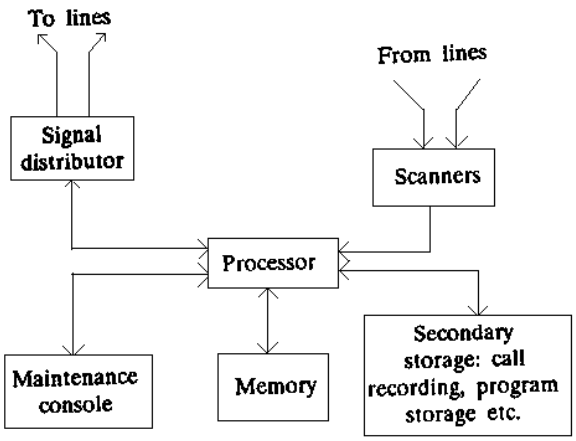


FIG - Typically Centralized SPC Organization

In centralized SPC, dual processor architecture may be configured to operate in one of three modes:

1. **Standby mode**: In this mode, one processor is active and the other is on standby, both hardware and software wise. The standby processor brought online when active processor fails. An important requirement of this configuration is the ability of the standby processor to reconstitute the state of the exchange system when it takes over the control.

2. **Synchronous duplex mode**: In synchronous duplex mode, hardware coupling is provided between the two processors which execute the same set of instructions and compare the results continuously. If a mismatch occurs, the faculty processor is identified and taken out of service immediately. When the system is operating normally, the two processors have the same data in their memories at all the times and receive all information from the exchange environment.

3. **Load Sharing mode;** In load sharing operation, an 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.

Advantages of SPC:

(i) Easy to control
(ii) Easy to maintain
(iii) Flexible
(iv) Wide range of services can be provided to customers.

(v) Increase level of automotive in switching

Q.3. What is time division switching? With the help of block diagram explain basic time division time switching method. **(10)**

Ans:

Time Division Switching: A switching element can be shared by number of simultaneously active speech circuits. This is the principle of time division switching. Obviously, with the way the switching elements are shared in time division switching, much greater savings can be achieved in the number of switching elements when compared to multistage space division switching.

Basic Time Division Switching: The functional blocks of a memory based time division switching switch is shown in Fig. and its equivalent circuit in Fig. In this organisation, the data coming in through the inlets are written into the data memory and later read out to the appropriate outlets. The incoming and outgoing data are usually in serial form whereas the data are written into and read out of the memory in parallel form. It, therefore, becomes necessary to perform serial-to-serial conversion and parallel-to-serial conversion at the inlets and outlets respectively. For convenience, the data-in and data-out parts of the MDR are shown separately for the data memory in Fig. 6.5 although in reality, MDR is a single register . Since there is only one MDR, a gating mechanism is necessary to connect the required inlet/outlet to MDR. This is done by the in-gate and out-gate units.

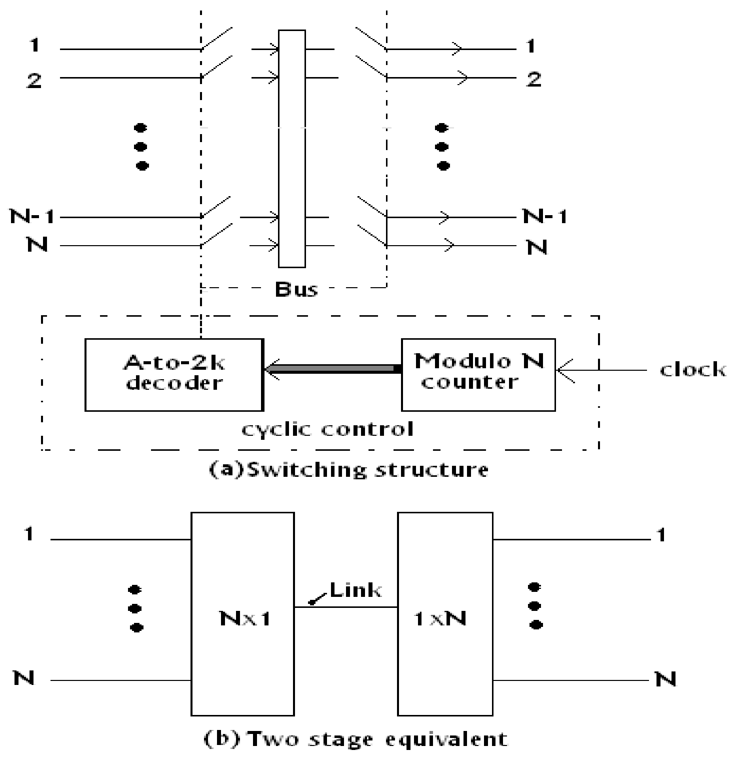


FIG – Simple PAM Time Division Switching.

Q.4. What are wired and wireless transmission systems? Explain the mechanisms of propagation of radio communication. **(6)**

Ans:

Transmission Systems: Modern long distance transmission systems can be placed under three broad categories:

1. Radio Systems
2. Coaxial cable systems
3. Optical fibre systems

Radio communication deals with electronic radiation of electromagnetic energy from one point to another through the atmosphere or free space. It is possible only in a certain position of the electromagnetic frequency spectrum. Presently, this portion includes frequencies from 9 kHz to 400 GHz. While there are international allocations for the radio spectrum up to 275 GHz, most of the commercial uses take place between 100 kHz and 20 GHz. Very few experimental systems have been operated beyond 100 GHz.

Different layers of the atmosphere play a role in propagating radio waves. The atmosphere consists of four layers. Of the four layers, the ionosphere and troposphere are useful for radio communication in certain frequency ranges. Certain other radio frequencies pass straight through the atmosphere and can be beamed towards satellites placed in the interplanetary space. Depending on the mechanism of propagation, radio communication can be placed under four categories:

1. Sky wave or ionosphere communication
2. Line-of-sight (LOS) microwave communication limited by horizon.
3. Troposphere scatter communication
4. Satellite communication.

Q.5. What are the end-to-end layers of OSI structure? Explain the transport layer in detail. **(8)**

Ans:

The layers 4-7 of ISO-OSI reference model communicate with peer entities in the end systems. There is no communication with entities in the intermediate systems. In this sense, layers 4-7 are often called end-to-end layers. These are Transport layer, Session layer, Presentation layer and Application layer respectively.

Transport Layer: The transport layer controls and ensures the end-to-end integrity of the data message propagated through the network between two devices, which provides for the reliable, transparent transfer of data between two end points . Transport layer responsibilities include message routing, segmenting, error recovery, and two types of basic services to an upper-layer protocol: connectionless oriented and connectionless. The transport layer is the highest layer in the OSI hierarchy in terms of communications and may provide data tracking, connection flow control, sequencing of data, error checking, and application addressing and identification.

Q 6. Explain the terms topology and access methods used in LANs. Discuss the CSMA/CD and CSMA/CA protocols. **(10)**

Ans:

(i) LAN topologies: Network topology is a physical schematic which shows interconnection of the many users. There are four basic topologies as under:

(a) Direct Connection or one to all topology
(b) Star topology
(c) Bus Topology
(d) Ring topology

(ii) Access methods used in LAN:

i. **Switched access**: It is used in LANs that are assigned around CBXs. Electronic switching is techniques are used to provide access.

ii. **Multiple access:** In multiple access schemes such as CSMA, whenever a station gets ready for transmission, it first listens to the bus to see if there is any ongoing transmission. If there is one, the new transmission is not initiated until the bus becomes free. This ensures that an ongoing transmission is not corrupted by a new transmission.

iii. **Token passing access:** In this scheme, a token packet is introduced to the network. This packet continues to circulate through the network as long as no user accepts it. When a user works to transmit he waits for the token packet to reach him and accepts the token, this acceptance removes the token packet from the network.

(iii) CSMA/CD: It is an access method used primarily with LANs configured in a bus topology. With CSMA/CD, any station (node) can send a message to any other station (or stations) as long as the transmission medium is free of transmissions from other stations. Stations monitor (listen to) the line to determine if the line is busy. If the station has a message to transmit but the line is busy, it waits for an idle condition before transmitting its message. If two stations transmit at the same time, a collision occurs. When this happens, the station first sensing the collision sends a special jamming signal to all other stations on the network. All stations then cease transmitting (back off) and wait a random period of time before attempting a retransmission. The random delay time for each station is different, and therefore, allows for prioritizing the stations on the network. If successive collisions occur, the back off period for each station is doubled. With CSMA/CD stations must contend for the network. A station is not guaranteed access to the network. To detect the occurrence of a collision, a station must be capable of transmitting and receiving simultaneously. CSMA/CD is used by most LANs configured in a bus topology. Ethernet is an example of a LAN that uses CSMA/CD.

CSMA/CA: It belongs to a class of protocols called multiple access methods. CSMA/CA stands for: Carrier Sense Multiple Access with Collision Avoidance. In CSMA, a station wishing to transmit has to first listen to the channel for a predetermined amount of time so as to check for any activity on the channel. If the channel is sensed "idle" then the station is permitted to transmit. If the channel is sensed as "busy" the station has to defer its transmission. This is the essence of both CSMA/CA and CSMA/CD. In CSMA/CA (Local Talk), once the channel is clear, a station sends a signal telling all other stations not to transmit, and then sends its packet. In Ethernet 802.3, the station continues to wait for a time, and checks to see if the channel is still free. If it is free, the station transmits, and waits for an acknowledgment signal that the packet was received

Q.7 Write short note on Associated vs. Common channel signaling. **(8)**

Ans:

Associated vs Common channel signalling:

The out band signalling suffers from the very limited bandwidth. Both are capable of having only a small signal repertoire. The trend in modern networking is to provide enhanced signalling facilities for the subscriber, the switching system and the telephone administration. Such a need is met by common channel signalling which may be common channel signalling which may be implemented in two ways: channel associated mode4 and channel non associated mode. In the former, the common signalling channel closely tracks the trunk groups along the entire length of a connection. In the letter, there is no close or simple assignment of control channels to trunk groups. These modes are illustrated in Fig. In the associated mode of operation shown in Fig. The signalling paths for the speech paths A-B, A-C-B and B-D are A-B, A-C-B and B-D respectively. The term 'associated signalling' in the CCS should not be confused with in channel signalling. The signalling in CCS associated mode is still done on a separate signalling channel; only that the signalling path passes through the same set of switches as does the speech path. Network topologies of the signalling network and the speech network are the same. The advantages of the scheme are the economic implementation and simple assignment of trunk groups to signalling channels.

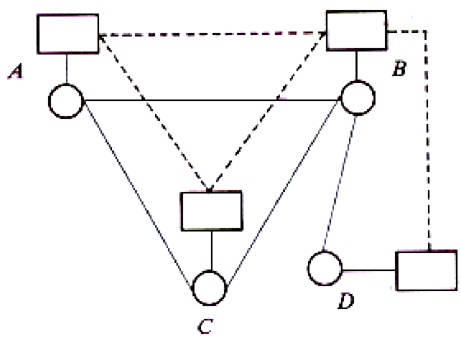


FIG – Channel Associated Signaling

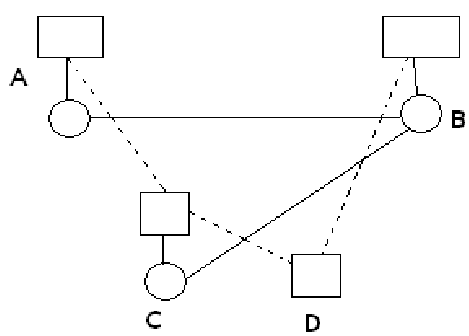


FIG – Non Associated Signaling.

Q.8. What are the different tones used in strowger telephony? Explain with the help of waveforms and the timings. **(8)**

Ans:
Dial Tone: This tone is used to indicate that the exchange is ready to accept dialed digits from the subscriber. The subscriber should start dialing only after hearing the dialing tone. Sometimes, however, a few seconds may elapse before the dial tone is heard. This happens particularly in common control exchanges which use shared resources for user interfaces. The dial tone is a 33 Hz or 50 Hz or 400 Hz

continuous tone as shown in Fig. The 400 Hz signal is usually modulated with Hz or 50 Hz.

When the called party line is obtained, the exchange control equipment sends out the ringing current to the telephone set of the called party. This ringing current has the familiar double-ring pattern. Simultaneously, the control equipment sends out a **ringing tone** to the calling subscriber, which has a pattern similar, that of the ringing current as shown in Fig. The two rings in the double ring pattern are separated by a time gap of 0.2 s and two double ring patterns by a gap of 2 s. The ring burst has duration of 0.4 s. The frequency of the ringing tone is 133 Hz or 400 Hz,

Busy tone pattern is shown in Fig. It is a bursty 400 Hz signal with silence period in between. The burst and silence duration have the same value of 0.75 s or 0.375 s. A busy tone is sent to the calling subscriber whenever the switching equipment or junction line is not available to put through the call or the called subscriber line is engaged. Fig. shows the **number unobtainable tone** which is continuous 400 Hz signal. This tone may be sent to the calling subscriber due to a variety of reasons such as the called party line is out of order or disconnected, and an error in dialing leading to the selection of a spare line. In some exchanges the number unobtainable tone is 400 Hz intermittent with 2.5 s on period and 0.5 s off period.

The **routing tone or call-in-progress tone** is a 400 Hz or 800 Hz intermittent pattern. In electromechanical systems, it is usually 800 Hz with 50 percent duty ratio and 0.5 s on/off period. In analog electronic exchanges it is a 400 Hz pattern with 0.5 s on period and 2.5 s off period. In digital exchanges, it has 0.1 s on/off periods at 400 Hz. When a subscriber call is routed through a number of different type of exchanges, one hears different call-in-progress tones as the call progresses through different exchanges. Fig. shows a routing tone pattern.

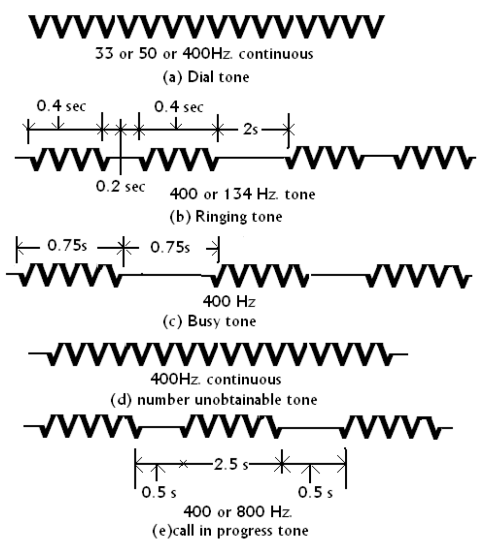


FIG – Signaling Tones in Automatic Exchange

Q.9. Using a combination of uniselectors and two motion selectors, draw a schematic of thousand line exchange and explain its working. **(8)**

Ans:

The schematic diagram for such an exchange is given in Fig. Each subscriber is connected to a single rotary pre-selector switch at the exchange, the outputs from this switch being connected to a bank of two-motion switches known as 'group selectors'. The out-puts from the pre-selector switches of a whole group of subscribers are connected together in parallel as that group of subscribers share a single bank of group selectors. When a subscriber lifts his telephone, the cradle switch causes a circuit to be completed back to the telephone exchange, signaling that the subscriber wishes to make a call. This causes the pre-selector switch to step around until it finds a free group selector. The pre-selector switch then stops in this position and the group selector is 'seized' by the subscriber wishing to make a call.

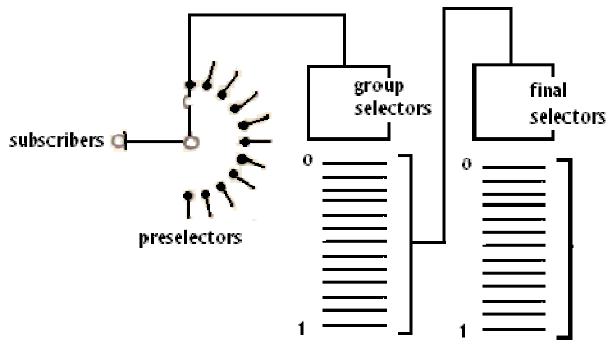


FIG – 1000 Line Exchange

In a 1000 line exchange, each subscriber has a 3 digit identification number starting from 000 to 999. On seizing the group selector, the subscriber dials his first digit and the selector switch moves up to the appropriate row on the switch contact array. Each final selector has the possibility of connection to 100 lines. The 1000 lines are therefore divided into 10 groups of 100 each, the group being identified by the first digit in the subscriber's number. The vertical motion of the group selector therefore selects a final selector in the group associated with the first digit dialed. Each individual row of contacts, or levels, of the group selector is connected to a bank of final selectors associated with a particular group of 100 line numbers. Having dialed the first digit to select the appropriate group, the group selector arm then automatically rotates in the vertical direction until it finds a free final selector. In the final selector, both directions of motion are under the control of the subscriber's dial and, after dialing two further digits, connection is established, providing the called subscriber's to answer his telephone.

Q.10. Name the switching schemes used in a digital exchange. How call processing takes place? **(8)**

Ans:

The different switching systems used are:

1. Strowger Switching System
2. Cross bar Switching.
3. Electronic Switching System.

Basic Call Procedure: Fig. Shows a simplification diagram illustrating how two telephone sets (subscribers) are interconnected through a central office dial switch. Each subscriber is connected to the switch through a local loop. The switch is most likely some sort of an electronic switching system (ESS machine). The local loop are terminated at the calling and called station s in telephone sets and at the central office ends to switching machines.

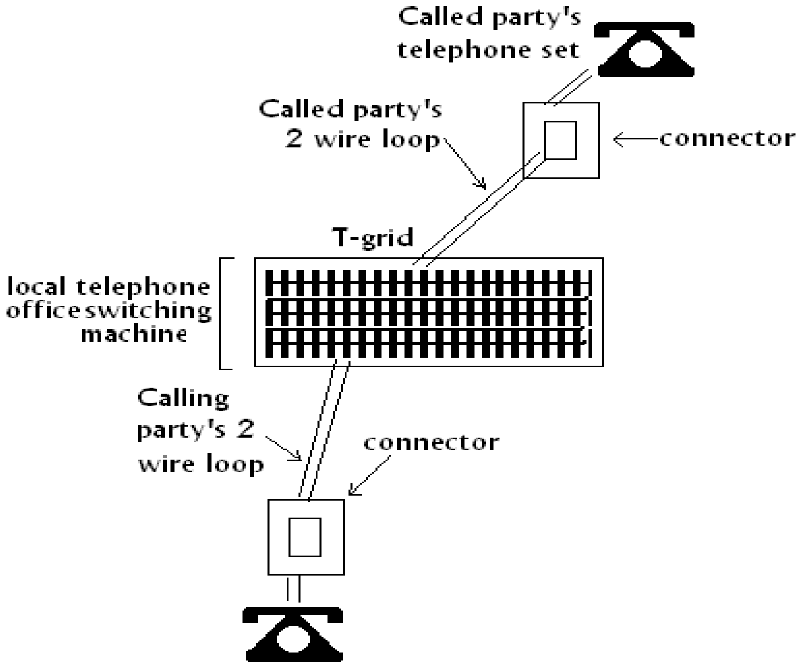


FIG - Telephone Call Procedure

When the calling party's telephone set goes off hook (i.e., lifting the handset off the cradle), the switch hook in the telephone set is released, completing a dc path between the tip and the ring of the loop trough the microphone .The ESS machine senses a dc current in the loop and recognizes this as an off-hook condition. Completing a local telephone call between two subscribers connected to the same telephone switch is accomplished through a standard set of procedure that includes the 10 steps listed next.

1. Calling station goes off hook.
2. After detecting a dc current flow on the loop, the switching machine returns an audible dial tone to the calling station, acknowledging that the caller has access to the switching machine.
3. The caller dials the destination telephone number using one of the two methods: Mechanical dial pulsing or, more likely, electronic dual-tone multi frequency (Touch-Tone) signals.
4. When the switching machine detects the first dialled number, it removes the dial tone from the loop.
5. The switch interprets the telephone number and then locates the loop for the destination telephone number.
6. Before ringing the destination telephone , the switching machine tests the destination loop for dc current to see if tt is idle (on hook) or in use (off hook). At the same time, the switching machine locates a signal path through the switch between the two local loops.
7. (a) If the destination telephone is off hook, the switching machine sends a station busy signal back to the calling station.
   (b) If the destination telephone is on hook, the switching machine sends a ringing signal to the destination telephone on the local loop and the same time sends a ring back signal to the calling station to give the caller some assurance that something is happening.
8. When the destination answers the telephone, it completes the loop, causing dc current to flow.

9. The switch recognizes the dc current as the station answering the telephone. At this time, the switch removes the ringing and ring-back signals and completes the path through the switch, allowing the calling and called parties to begin conversation.

10. When either end goes on hook, the switching machine detects an open circuit on that loop and then drops the connections through the switch.

Q.11. How speech is transmitted in a digital switching environment using PCM/TDM? **(8)**

Ans:

A digital carrier system is a communications system that uses digital pulses rather than analogue signals to encode information. Fig shows the block diagram for a digital carrier system. This digital carrier system T1 uses PCM-encoded samples from 24 voice band channels for transmission over a single metallic wire pair or optical fibre transmission line. Each voice-band channel has a band width of approximately 300 Hz to 3000 Hz. Again, the multiplexer is simply a digital switch with 24 independent inputs and one time division multiplexed output. The PCM output signals from the 24 voice-band channels are sequentially selected and connected through the multiplexer to the transmission line.

When a T1 carrier system, D-type (digital) channel banks perform the sampling, encoding and multiplexing of 24 voice-band channels, each channel contains an eight-bit PCM code and is sampled 8000 times a second. Each channel is sampled at the same rate. Fig. shows the channel sampling sequence for a 24-channel T1 digital carrier system. As the Fig. shows each channel is sampled once each frame but not at the same time. Each channel's sample is offset from the previous channel's sample by 1/24 of the total frame time. Therefore, one 64-kbps PCM-encoded sample is transmitted for each voice-band channel during each frame (a frame time of 1/8000 = 125 μs). Later, an additional bit (called the framing bit) is added to each frame. The framing bit occurs once per frame (8000-bps rate) and is recovered in the receiver, where it is used to maintain frame and sample synchronization between the TDM transmitter and receiver.

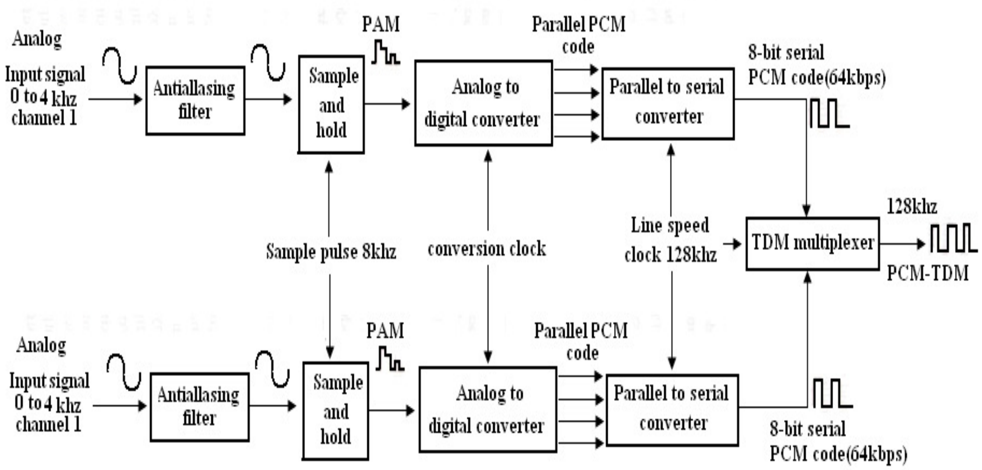


FIG – Two-channel PCM-TDM system

Q.12. How does one arrive at the probability of availability of free lines during the bu[sy] hour? How can this be improved? **(8)**

Ans:
One can arrive at the probability of free lines during busy hour by using the delay probability of the exchange. The delay probability indicates the delay that a caller may face before his call can be completed. In a delay system the caller is made to wait till a free line is made available for the completion of the call. This leads to call delay. Improvement can be made if the system is treated as a loss system instead of a delayed system. In a loss system, the caller has to repeat a call till the call is established. This reduces the delay by releasing a caller from the queue until there is repeated action.
Blocking probability can be useful. The blocking probability P is defined as the probability that all the servers in system are busy. When all the servers are busy, no further traffic can be carried by the system and the arriving subscriber's traffic is blocked. At the first instance, it may appear that the blocking probability is the same measure as the GOS. The probability that all the servers are busy may well represent the fraction of the calls lost, which is what the GOS is all about.

Q.13. What are the different types of distributing frames used in exchanges? Explain their importance. **(8)**

Ans:
The different distribution frames used in exchange are shown in figure. 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 this pair of wires. It is unwisely to run physically independent pairs from every subscriber premises to the exchange. It is far easier to lay cables containing a number of pairs of wires for different geographical locations and run individual pairs as required by the subscriber premises.

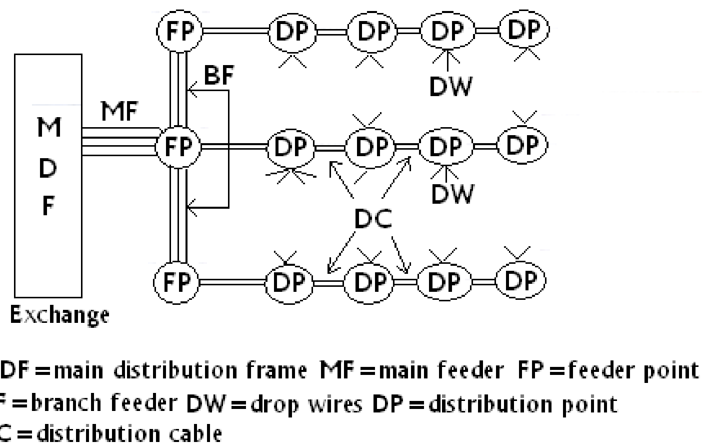


FIG – Cable Hierarchy For Subscriber Loops.

Generally four levels of cabling are used as shown in Fig. At the subscriber end, the drop wires are taken to a distribution point. The drop wires 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. Many distribution cables from nearby geographical locations are terminated on a feeder point where they are connected to branch feeder cables which, in turn, are connected to the main feeder cable. The main feeder cables carry a larger number of wire pairs, typically 100-2000,than the distribution cables which carry typically 10-500 pairs. The feeder cables are terminated on a main distribution frame (MDF) at the exchange. The subscriber cable pairs emanating from the exchange are also terminated on the MDF.

Q.14. What is meant by common control? Explain all the categories that are served by Common Control switching. **(8)**

Ans:
In some switching systems, the control subsystem may be an integral part of the switching network itself. Such system is known as **direct control** switching systems. Those systems in which the control subsystem is outside the switching network are known as **common control** switching system. Strowger exchanges are usually direct control systems, whereas crossbar and electronic exchanges are common control system. All stored program control systems are common control systems. Common control is also known as indirect control or register control.
Common Control Switching System: A functional block diagram of a common control switching system is shown in Fig. The control functions in a switching system may be placed under four broad categories:

(a) Event monitoring.
(b) Call processing.
(c) Charging.
(d) Operation and maintenance

Events occurring outside the exchange at the line units, trunk junctions and inter exchange signaling receiver/sender units are all monitored by control subsystem. Typical events include all request and call release signals at the line units. The occurrences of the events are signaled by operating relays which initiate control action. The control subsystem may operate relays in the junctions, receivers/senders and the line units, and thus command these units to perform certain functions. Events monitoring may be distributed. For examples, 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 market for dial tone, and the register finder is activated to seize a free register. Identity of the calling line is used to determine the line category and the class of service to which the subscriber belongs. A register appropriate to the line category is chosen, which then sends out the dial tone to the subscriber, in readiness to receive the dialing information. As soon as the initial digits (usually 2-5) which identify the exchange are received in the register, the register continues to receive the remaining digits.

The initial translator determines the route for the call through the network and decides whether a call should be put through or not. It also determines the charging methods and the rates applicable to the subscriber . Initial translation may also take into account instructions from the operating personnel and information regarding the status of the network.

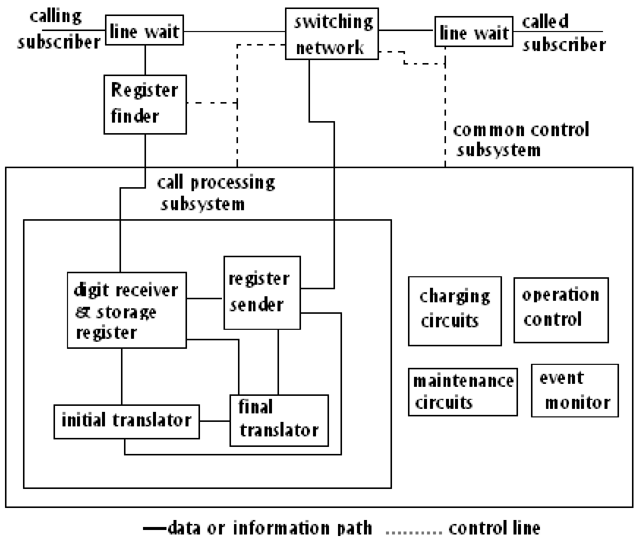


FIG - Common Control Switching System

If a call is destined to a number in an exchange other than the present one processing the digits, the initial translator generates the require routing digits and passes them on to the register sender. Here the digits corresponding to the subscriber identification are concatenated and the combined digit pattern is transmitted over the trunks to the external exchange. Register sender uses appropriate signaling technique, depending on the requirements of the destination exchange. If the call is destined to a subscriber within the same exchange, the digits are processed by the final translator. The translation of directory number to equipment number takes place at this stage. The final translator determines the line unit to which a call must be connected and the category of the called line. The category information may influence charging and connection establishment. In some practical implementations, both initial and final code translator functions are performed by a single translator.
Controlling the operation of the switching network is an important function of the common control subsystem. This is done by marking the switching elements at different stages in accordance with a set of binary data defining the path and then commanding the actual connection of the path. Path finding may be carried out at the level of the common control unit or the switching network.

Q.15 Explain the working of broad band ISDN. **(8)**

Ans:
BISDN Configuration: Fig. shows how access to the BISDN network is accomplished. Each peripheral device is interfaced to the access node of a BISDN network through a broadband distant terminal (BDT). The BDT is responsible for

electrical to optical conversion, multiplexing of peripherals, and maintenance of the subscriber's local system. Excess nodes concentrate several BDT's into high speed optical fiber line directed through a feeder point into a service node. Most of the control function for system excess is managed by the service node, such as call processing, administrative function and switching and maintenance functions. The functional modules are interconnected in a star configuration and include switching, administrative, gateway, and maintenance modules. The interconnection of the function module is shown in Fig. The central control hub acts as the end user interface for control signaling and data traffic maintenance. In essence, it oversees the operation of the modules.

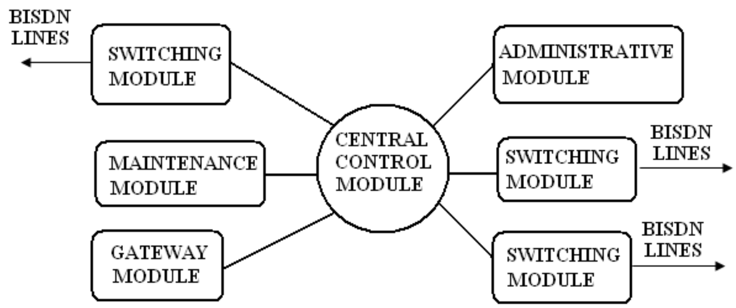


FIG - BISDN Functional Module Interconnection

Subscriber terminal near the control office may by pass the excess nodes entirely and the directly connected to the BISDN network through a service node. BISDN nodes that used optical fiber cables can utilize much wider band width and consequently, have higher transmission rates and offer more channel handling capacity than ISDN systems.

Q.16. What is the need of a hybrid in telephone networks? How does it work? **(8)**

Ans:
Digital exchanges require receive and transmit signals on separate two-wire circuits. This calls for two-wire to four-wire conversion. Such a conversion is normally required for trunk transmissions in analog exchanges. The circuit that performs 2-wire to 4-wire conversion is called Hybrid. A transformer based hybrid circuit is shown in Fig. The main function of a hybrid is to ensure that there is no coupling of signal from the input to the output in the 4-wire circuit. The operation of the circuit is as follows: The input signal is coupled to the B and F windings equally. Through the C winding, the input is coupled to the 2-wire circuit. The same signal when it flows through the balanced 2-wire couples the signal to winding D through winding C. The signal induced in B flows through E and induces a current in D that opposes the current induced by F. If the impedance Z_B exactly matches that of the 2-wire circuit, the effect of input signal on the output winding D is completely nullified. In a similar way, the input signal from the subscriber end is completely nullified from coupling into the winding A.

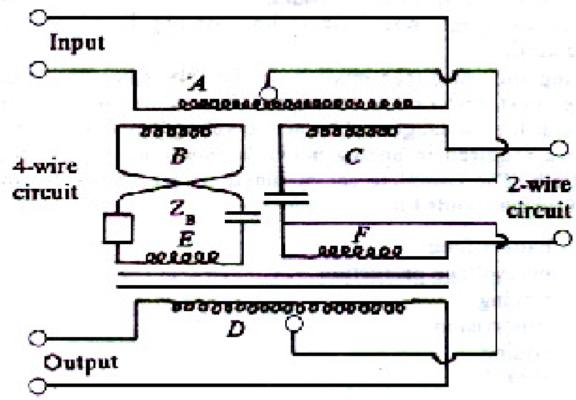


FIG – Two-wire to 4-wire Transformer Hybrid

Q.17 What layers are covered under end to end layer connectivity? Explain briefly the function of each one of them. **(8)**

Ans:

The layers 4-7 of ISO-OSI reference model communicate with peer entities in the end systems. There is no communication with entities in the intermediate systems. In this sense, layers 4-7 are often called end-to-end layers. These are Transport layer, Session layer, Presentation layer and Application layer respectively.

1. **Transport Layer**:
   - ➢ It is responsible for establishing a network independent communication path suitable for the particular terminal equipments (e.g. providing the appropriate data rate and end-to-end error control). It thus relieves the user from being concerned with such details.
   - ➢ In a packet switched network, the transport entity breaks up a long user message into packets to march the network capabilities. The packets are reassembled at the receiving end to reconstruct the user message.
   - ➢ End-to-end flow control & end-to-end error recovery are also the functions of transport layer.

2. Session Layer:
   - ➢ The session layer is used to allow users to identify themselves when waiting access to the network.
   - ➢ This is concerned with setting up and maintaining an operational session between terminals. E.g. "signing on" at the commencement of a task and "signing off" at its end.
   - ➢ The main function of the session layer is to organize different sessions between cooperating entities and perform all related functions like synchronization, failure management , control etc. for the successful execution of a session.
   - ➢ Another facility offered by the session layer is known as Activity management.

3. Presentation Layer:
   - ➢ This is concerned with the format of the data represented, in order to overcome difference in representation of the information as supplied to one terminal and required at the other. Its purpose is to make communication over the network machine independent.

➢ It resolves the syntax differences by encoding data into standard abstr notation that is valid throughout the network. Thus file format differences, data representations, data structure are resolved using a standard notation.

4. **Application Layer:**

➢ As the highest layer in the OSI reference model, the application layer provides services to the users of OSI environment. The layer provides following services:

1. Electronic mail or message handling service
2. Directory services
3. Cost allocation
4. Determination of quality of service
5. File transfer and management
6. Editors and terminal support services
7. Telematic services like videotext.

Q.18 Write short notes on:

i. Stored Program Control.
ii. Congestion.
iii. Common channel signaling.
iv. PSTN. **(4×4)**

Ans.

(i) Stored Program Control: 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. A typical control configuration of an ESS using centralized SPC is shown in Fig. A centralized SPC configuration may use more than one processor for redundancy purposes.

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:

(i) Standby mode
(ii) Synchronous duplex mode
(iii) Load Sharing mode

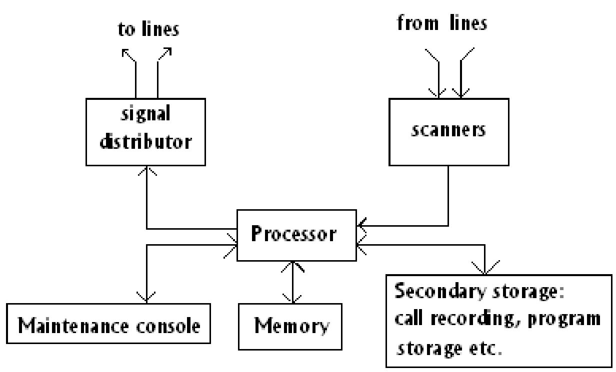


FIG - Typically Centralized SPC Organization

(ii) CONGESTION: It is uneconomic to provide sufficient equipment to carry all the traffic that could possibly be offered to a telecommunication system. In a telephone exchange it is theoretically possible for every subscriber to make a call simultaneously. A situation can therefore arise when all the trunks in a group of trunks are busy, and so it can accept further calls. This state is known congestion. In a message-switched system, calls that arrive during congestion wait in a queue until an outgoing trunk becomes free. Thus, they are delayed but not lost. Such systems are therefore called queuing systems or delay system. In a circuit-switched system, such as a telephone exchange, all attempts to make calls over a congested group of trunks are successful. Such systems are therefore called lost-call systems. In a lost-call system the result of congestion is that the traffic actually carried is less than the traffic offered to the system.

(iii) Common channel signalling. Signaling systems link the variety of switching systems, transmission systems and subscriber equipments in telecommunication network to enable the network to function as a whole. Three forms of signaling are involved in a telecommunication network:

1. Subscriber loops signaling.
2. Intra exchange or register signaling
3. Interexchange or inter register signaling

In a telephone network, subscriber loop signaling depends upon the type of a telephone instrument used. The intra exchange signaling is internal to the switching system and is heavily dependent upon the type and design of a switching system. It varies from one model to another even with the same manufacturer. This signaling does not involve signaling system of the type required on the switching network. When interexchange signaling takes place between exchanges with common control subsystems, it is called Inter register signaling. The main purpose of Inter register signaling is the exchange of address digits which pass from exchange to exchange on a link by link basis. Network wide signaling also involves end to end signaling between the originating exchange and the terminating exchange. Such a form of signaling is called line signaling. CCS does not use the speech or the data path for signaling. It uses a separate common channel for passing control signals for a group of trunks or information paths. It gives the following advantages:

(i) Information can be exchange between the processors much more rapidly than when channel associated signaling is used.
(ii) As a result, a much wider repertoire of signals can be used and this enables more services to be provided to customers.
(iii) Signals can be added or changed by software modification to provide new services.
(iv) There is no longer any need for line signaling equipments on every junction which results in a considerable cost saving.
(v) Since there is no line signaling, the junctions can be used for calls from B to A in addition to calls from A to B. Both way working requires fewer circuits to carry the traffic than if separate groups of junctions are provided from A to B and from B to A.
(vi) Signals relating to a call can be sent while the call is in progress. This enables customers to alter connections after they have been set up. For example a customer can transfer a call elsewhere, or request a third party to be connected in to an existing connection.

(iv) PSTN the Public Switched Telephone Network: The Public Switched Telephone Network (PSTN) accommodates two types of subscribers: public and private. Subscribers to the private sector are customers who lease equipment, transmission media (facilities), and service from telephone companies on a permanent basis. The leased circuits are designed and configured for their use only and are often referred to as private line circuits or dedicated line circuits. For example, large banks do not wish to share their communication network with other users, but it is not effective for them to construct their own networks. Therefore, banks lease equipment and facilities from public telephone companies and essentially operate a private telephone or data network within the PSTN. The public telephones companies are sometime called providers, as they lease equipment and provide service to other private companies, organizations, and government agencies. Most metropolitan area networks (MANs) and wide area networks (WANs) utilize private line data circuits and one or more service provider.

Subscribers to the public sector of the PSTN share equipment and facilities that are available to all the public subscribers to the network. This equipment is appropriately called common usage equipment, which includes transmission facilities and telephone switches. Anyone with a telephone number is a subscriber to the public sector of the PSTN. Since subscribers to the public network are interconnected only temporarily through switches, the network is often appropriately called the public switched telephone network (PSTN) and sometimes simply as the dial-up network. It is possible to interconnect telephones and modems with one another over great distance in fraction of a second by means of an elaborate network comprised of central offices, switches, cables (optical and metallic), and wireless radio systems that are connected by routing nodes (a node is a switching point). When someone talks about the public switched telephone network, they referring to the combination of lines and switches that from a system of electrical routes through the network.

Q.19 What is time multiplexed space switching? With a neat diagram explain its operation. **(8)**

Ans:
Time division switches where an inlet or an outlet corresponded to a single subscriber line with one speech sample appearing every 125 μs on the line. Such switches are used in local exchanges. We now consider switches that are required in transit exchanges. Here, the inlets and outlets are trunks which carry time division multiplexed data streams. We call such switches time multiplexed switches. A time multiplexed time division space switch is shown in Fig. There are N incoming trunks and N outgoing trunks, each carrying a time division multiplexed stream of M samples per frame. Each frame is of 125-μs time duration. In one frame time, a total of MN speech samples have to be switched. One sample duration, 125/M microseconds, is usually referred to as a time slot. In one time slot, N samples are switched. Fig shows an output-controlled switch. The output is cyclically scanned. There is a 1-to-M relationship between the outlets and the control memory locations, i.e. there are M locations in the control memory corresponding to each outlet.

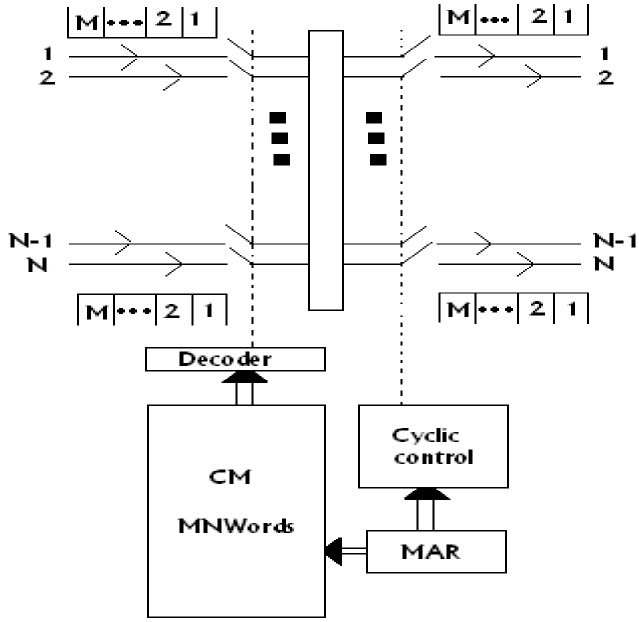


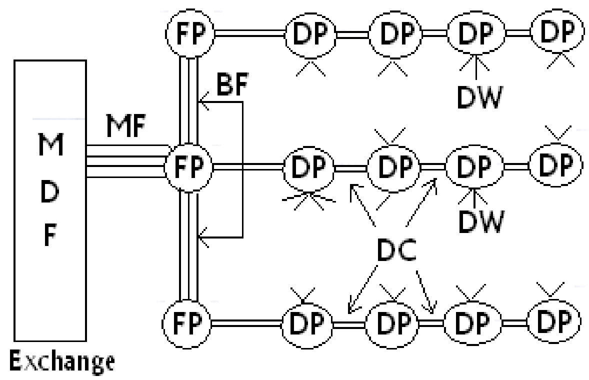
FIG – Time Multiplexing Space Switch

The control memory has MN words. If we view the control memory as M blocks of N words each, a location address may be specified in a two dimensional form,(i,j),where i is the block address and j is the word within the block. We have $1 \leq i \leq M$ and $1 \leq j \leq N$. The block address i corresponds to the time slot i and the word address j to the outlet j. The first N locations of the control memory correspond to the first time slot, the next N locations, i.e. locations N + 1 to 2N when addressed linearly, or locations (2,1) to (2,N) when addressed in a two dimensional form, correspond to the time slot 2 and so on. Therefore, if the location (i,j) contains an inlet address k, it implies that inlet k is connected to the outlet j during the time slot i. The number of trunks that can be supported on this switch is given by N =125/ Mt , Where t is the switching time including memory access time per inlet-outlet pair.

Q.20. What are the major systems of a telecommunication network? Discuss in detail the subscriber loop systems. **(8)**

Ans:
The major systems of any telecommunication network may consist of the following major systems:
1. Subscriber end instruments or equipments
2. Subscriber loop systems
3. Switching Systems
4. Transmission systems
5. Signalling systems

Subscriber Loop System: 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 this pair of wires. It is unwidely to run physically independent pairs from every subscriber premises to the exchange. It is far easier to lay cables containing a number of pairs of wires for different geographical locations and run individual pairs as required by the subscriber premises.



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

FIG – Cable Hierarchy For Subscriber Loops.

Generally four levels of cabling are used as shown in fig. At the subscriber end, the drop wires are taken to a distribution point. The drop wires 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. Many distribution cables from nearby geographical locations are terminated on a feeder point where they are connected to branch feeder cables which, in turn, are connected to the main feeder cable. The main feeder cables carry a larger number of wire pairs, typically 100-2000, than the distribution cables which carry typically 10-500 pairs. The feeder cables are terminated on a main distribution frame (MDF) at the exchange. The subscriber cable pairs emanating from the exchange are also terminated on the MDF.

Q.21. Classify data networks. Explain with the help of Nyquist theorem, the data rate limitations in PSTN's. Give an account of modems used in data transfer. Explain their importance and list some of the V-series recommendations. **(14)**

Ans:
Classifications of Data Network:
Data Networks are classified according to their geographical coverage**:**
- Wide area networks (WANs)
- Metropolitan area networks(MANs)
- Local area networks (LANs)

Data rates in PSTNs: A voice channel in a PSTN is band limited with a nominal bandwidth of 3.1 kHz. A first-cut estimate of this can be obtained from Nyquist's theorem which applies to noiseless channels and states:

$R = 2H \log V$ bps
Where
R = maximum data rate
H = bandwidth of the channel
V = number of discrete levels in the signal

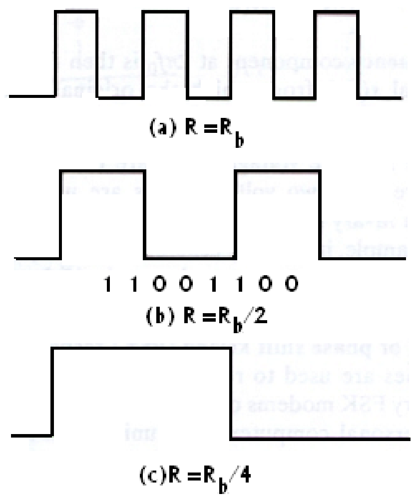


FIG – Baud Rates and Bit Rates

For a 3-kHz channel, and a binary signal, the maximum data rate works out to be 600 bps, if the channel is ideal. In a practical channel, the maximum rate would come down. By increasing the number of levels used to represent the signal, the bit rate may be increased arbitrarily in a noiseless channel. It is important to recognize that the actual number of signal transitions is still limited to the binary level limit; the effective bit rate goes up with more than two signal levels as each signal level can now represent a group of two or more bits. The maximum rate of signal transitions that can be supported by a channel is known as baud rate or symbol rate.
In a channel where noise is present, there is an absolute maximum limit for the bit rate. This limit arises because the difference between two signal levels becomes comparable to the noise level when the number of signal level is increased. Claude Shannon extended Nyquist's work to the case of noisy channels affected by random or thermal noise. Shannon's result states;

$$R = H \log (1+S/N)$$

Where R= the maximum bit rate obtainable
H= bandwidth of the channel
S/N = signal to noise ratio.

Modem : Modems are generally provided by network operators (Department of Telecommunication in India) or by vendors who are not necessarily the manufacturers of computer systems. Hence, there is a need to standardize the interface between the modem and the computer equipment. CCITT terminology for the modem is data circuit terminating equipment (DCE), and for the computer it is data terminal equipment (DTE). DCE is often referred to as data communication equipment, outside CCITT. It is important to recognize that DCE and DTE are generic terms and may be applied to a variety of equipments not necessarily only to modems or computers. For example, a DTE may be a terminal, workstation or a computer, and a DCE may be a modem or a computer based node in a data network.
A series of standards, known as V-series, has been defined by CCITT for interfacing DTEs to DCEs operating with PSTNs. The series also defines a variety of DCEs using different modulation techniques and operating at different speeds using either a leased PSTN or dial-up line Examples of DTE-DCE interface standards and other standards related to data transmission on PSTN are:
V.5 Standardization of data signaling rates for synchronous data transmission in the PSTN

V.24 DTE-DCE interface and control signals
V.28 DTE-DCE electrical characteristics for unbalanced double-current interchange circuits.
V.53 Limits for the maintenance of telephone type circuits used for data transmission.

Q.22. What is Grade of service and blocking probability? What are delay systems in telecommunication networks? **(8)**

Ans:
Grade of service: In loss systems, the traffic carried by the network is generally lower than the actual traffic offered to the network by the subscribers. The overload traffic is rejected and hence it is not carried by the network. The amount of traffic rejected by the network is an index of the quality of the service offered by the network and is termed as grade of service (GOS) and is defined as the ratio of lost traffic to offered traffic. Offered traffic is the product of the average number of calls generated by the users and the average holding time per call. GOS is given by

$$GOS = A-A_0 / A_0$$

Where
A= offered traffic
A_0 = carried traffic
$A-A_0$= lost traffic

Blocking Probability: The blocking probability P is defined as the probability that all the servers in system are busy. When all the servers are busy, no further traffic can be carried by the system and the arriving subscribers' traffic is blocked. At the first instance, it may appear that the blocking probability is the same measure as the GOS. The probability that all the servers are busy may well represent the fraction of the calls lost, which is what the GOS is all about.
Delay System: A class of telecommunication networks, such as data networks, places the call or message arrivals in a queue in the absence of resources, and services them as and when resources become available. Servicing is not taken up until the resources become available. Such systems are known as Delay Systems.
Delay Systems are analyzed using queuing theory which is sometimes known as waiting line theory. Delay Systems in telecommunications include the following:

- Message Switching
- Packet Switching
- Digit Receiver access
- Automatic Call distribution
- Call Processing

Q.23 What are single stage and multistage networks? Compare the strengths and weaknesses of each. **(6)**

OR

List the major difference in single stage, two stage and three stage Networks. Also discuss their blocking characteristics. **(8)**

Ans:

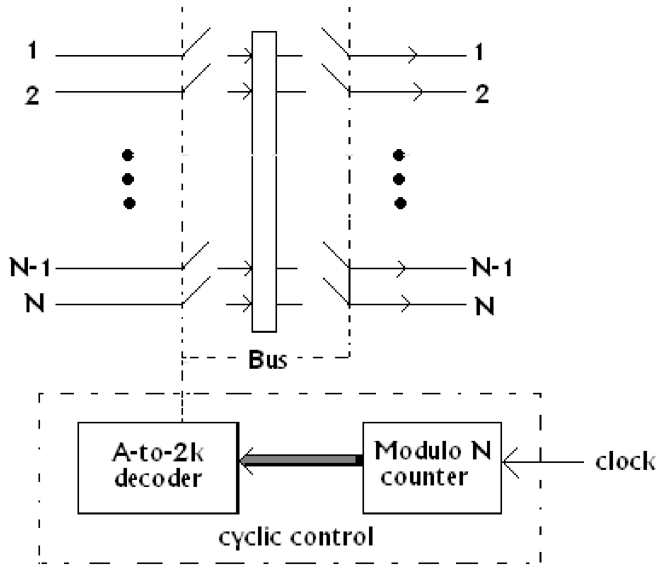
Single Stage Vs. Multistage Network.

Sr. No.	Single Stage	Multi Stage
1.	Inlet to outlet connection is through a single cross point.	Inlet to Outlet connection is through multiple cross points
2.	Use of single cross point per connection results in better quality link.	Use of multiple cross points may degrade the quality of a connection.
3.	Each individual cross point can be used for only one inlet/outlet pair connection.	Same cross point can be used establish connection between a number of inlet/outlet pairs.
4.	A specific cross point is needed for each specific connection.	A specific connection may be established by using sets of cross points.
5.	If a cross points fails, associated connection cannot be establish- There is no redundancy.	Alternative cross-points and paths are available.
6.	Cross points are inefficiently used. Only one cross point in each row or column of a square or triangular switch matrix is even in use , even if all the lines are active.	Cross points are used Efficiently
7.	Number of cross points is Prohibitive	Number of cross points is reduced significantly
8.	A large number of cross points in each inlet/outlet leads to capacitive loading.	There is no capacitive loading problem
9.	The network is non blocking in character	The network is blocking in character
10.	Time for establishing a call is less.	Time for establishing a call is more.

Q.24. Explain the basic architecture of digital switching systems. Explain in detail companding. **(8)**

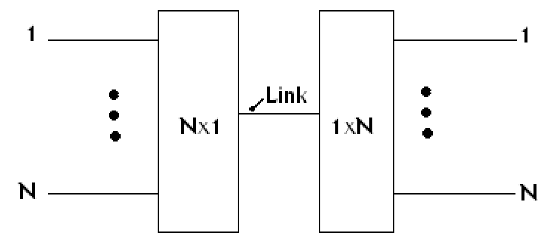
Ans:

A simple N X N time division space switch is shown in Fig. The switch can be represented in an equivalence form as a two-stage network with N X 1 and 1 X N switching matrices for the first and second stages respectively as shown in Fig. The network has one link interconnecting the two stages. Each inlet/outlet is a single speech circuit corresponding to a subscriber line. The speech is carried as PAM analogue samples or PCM digital samples, occurring at 125-µs intervals. When PAM samples are switched in a time division manner, the switching is known as analogue time division switching. If PCM binary samples switched, then the switching is known as digital time division switching. In Fig, the interconnected by a suitable control mechanism and the speech sample transferred from the inlet to the outlet.



(a)Switching structure

FIG – Switching Structure



(b) Two stage equivalent

FIG - Two – stage Equivalence

Companding: Companding is the process of compressing and expanding. It consists of compressing the signal at the transmitter and expanding it at the receiver. The modulating signal to be transmitted is passed through an amplifier which has correctly adjusted non linear transfer characteristic, favoring small amplitude signals. These are the artificially large when they are quantized and so the effect of quantizing noise upon them is reduced. The correct amplifier relations are restored at the receiver by expander. It is desirable to agree to some standard for companding in order to realize hardware efficiencies and to permit sending and receiving by a variety of users. The most common application of companding is in voice transmission. North America and Japan have adopted a standard compression curve known as **µ law companding**. The law µ companding formula is given by

$V_{out} = (V_{max} X \;\; In (1+ \mu V_{in} / V_{max}) / \; In (1+ \mu)$

Where

V_{max} = Maximum uncompressed modulating input amplitude.

V_{in} = Amplitude of the input signal at particular instant at time

μ = Parameter used to define the amount of compression
V_{out} = Compressed output voltage.

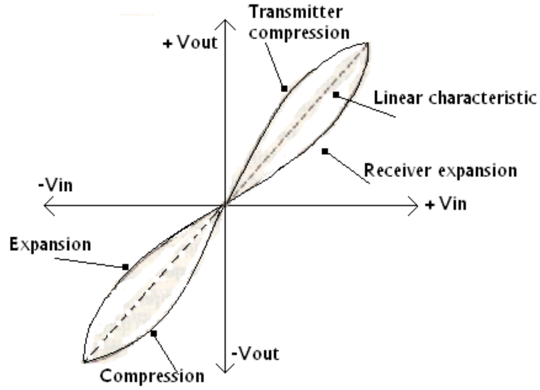


FIG – Companding Curves for PCM

Q.25. Describe the various signalling techniques. Compare in-channel signalling with common channel signaling, which is more advantageous. **(8)**

Ans:
Signaling systems link the variety of switching systems, transmission systems and subscriber equipments in telecommunication network to enable the network to function as a whole.
Three forms of signaling are involved in a telecommunication network:

1. Subscriber loop signaling.
2. Intra exchange or register signaling
3. Interexchange or inter register signaling

In a telephone network, **subscriber loop signaling** depends upon the type of a telephone instrument used.
The **intra exchange signaling** is internal to the switching system and is heavily dependent upon the type and design of a switching system .It varies from one model to another even with the same manufacturer. This signaling does not involve signaling system of the type required on the switching network.
When interexchange signaling takes place between exchanges with common control subsystems, it is called **Inter register signaling**. The main purpose of Inter register signaling is the exchange of address digits which pass from exchange to exchange on a link by link basis. Network wide signaling also involves end to end signaling between the originating exchange and the terminating exchange. Such a form of signaling is called line signaling. CCS does not use the speech or the data path for signaling. It uses a separate common channel for passing control signals for a group of trunks or information

In Channel	Common Channel
Trunks are held up during signaling.	Trunks are not required for Signaling
Signal repertoire is limited.	Extensive signal repertoire is possible.
Interference between voice and Control signals may occur.	No interference as the two Channels are physically separate.
Separate signaling	Only one set of signalling

equipment is required for each trunk and hence expensive.	equipments is required for a whole group of trunk Circuits and therefore CCS is economical
The voice channel being the control channel, there is a possibility of potential misuse by the customers.	Control Channel is in general in accessible to users.
Signaling is relatively slow.	Signaling is significantly fast.
Speech circuit reliability is assured.	There is no automatic test of the speech circuit.
It is difficult to change or add signals.	There is flexibility to change or add signals.
It is difficult to handle signaling during speech period.	Signals during speech. There is freedom to handle
Reliability of the signaling path is not Critical.	Reliability of the signaling Path is critical.
Common channel signaling is better than In-channel signaling.	

Q 26 Explain the following terms: progressive control, common control and stored program control. **(6)**

Ans:
(i) Progressive Control: Step by step system is an example of progressive control. The connection is set up in stages, in response to the digits dialed and each step in setting up the connection is controlled by relays mounted on the selector which operates at the input stage.
(ii) Common Control: Those systems in which the control subsystem is outside the switching network are known as common control switching system. Strowger exchanges are usually direct control systems, whereas crossbar and electronic exchanges are common control system. All stored program control systems are common control systems. Common control is also known as indirect control or register control.
(iii) Stored Program Control: Modern digital computers use the stored programmed concept. 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. Carrying out the exchange control functions through programs stored in the memory of a computer led to the nomenclature stored program control (SPC). An immediate consequence of program control is the full-scale automation of exchange functions and the introduction of a variety of new services to users. 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

Q.27. Explain what is DTMF signalling. **(6)**

Ans:
Dual Tone Multi Frequency (DTMF) was first introduced in 1963 with 10 buttons in Western Electric 1500 –type telephones. DTMF was originally called Touch-Tone. DTMF is a more efficient means than dial pulsing for transferring telephone numbers from a subscriber's location to the central office switching machine. DTMF is a simple two-of –eight encoding scheme where each digit is represented by the linear addition of two frequencies. DTMF is strictly for signaling between a subscriber's location and the nearest telephone office or message switching center. DTMF is sometimes confused with another two-tone signaling system called multi frequency signaling (Mf), which is a two-of-six code designed to be used only to convey information between two electronic switching machines.
Fig. shows the four-row-by-four column keypad matrix used with a DTMF keypad. As the figure shows, the keypad is comprised of 16 keys and eight frequencies. Most household telephones, however, are not equipped with the special-purpose keys located in the fourth column (i.e., the A, B, C, and D keys). Therefore, most household telephones actually use two-of-seven tone encoding scheme. The four vertical frequencies (called the low group frequencies) are 697 Hz, 852 Hz, and 941 Hz, and the four horizontal frequencies (called the high group frequencies) are 1209 Hz, 1336 Hz,1477 Hz, and 1633 Hz. The frequency tolerance of the oscillators is $\pm$ .5%. As shown in Figure, the digits 2 through 9 can also be used to represent 24 of the 26 letters (Q and Z are omitted). The letters were originally used to identify one local telephone exchange from another.

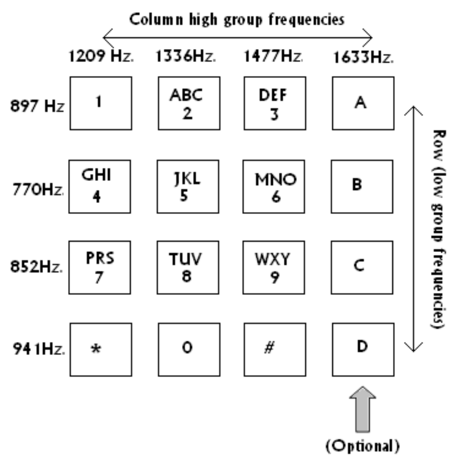


FIG - DTMF Keypad Layout.

Q.28 Write short notes
(i)Telephone hand set and it's working.
(ii) CPU based exchange **(8)**

Ans:
(i) A standard telephone set is comprised of a transmitter, a receiver, and electrical network for equalization, associated circuitry to control side tone levels and to regulate signal power, and necessary signalling circuitry. In essence, a telephone set

is an apparatus that creates an exact likeness of sound waves with an elect current. Fig 8-4 shows the functional block diagram of a telephone set. The essential components of a telephone set are the ringer circuit, on/off hook circuit, equalizer circuit, hybrid circuit, speaker, microphone, and a dialing circuit.

Ringer Circuit: The purpose of the ringer is to alert the destination party of incoming calls. The audible tone from the ringer must be load enough to be heard from a reasonable distance and offensive enough to make a person want to answer the telephone as soon as possible. In modem telephones, the bell has been placed with an electronic oscillator connected to the speaker..

On/off hook circuit: The on/off hook circuit (some times called a switch hook) is nothing more than a simple single-throw, double-pole (STDP) switch placed across the tip and ring. The switch is mechanically connected to the telephone handset so that when the telephone is idle (on hook), the switch is open. When the telephone is in use (off hook), the switch is closed completing and electrical path through the microphone between the tip and ring of the local loop.

Equalizer circuit: Equalizers are combination of passive components (resistors, capacitors and so on) that are used to regulate the amplitude and frequency response of the voice signals.

Speaker: In essence, the speaker is the receiver for the telephone. The speaker converts electrical signals received from the local loop to acoustical signals(sound waves) that can be heard and understood by a human being. The speaker is connected to the local loop through the hybrid network. The speaker is typically enclosed in the handset of the telephone along with the microphone.

Microphone: For all practical purposes, the microphone is the transmitter for the telephone. The microphone converts acoustical signals in the form of sound pressure waves from the caller to electrical signals that are transmitted into telephone net-work through the hybrid network. Both the microphone and the speaker are transducers, as they convert one form of energy into another form of energy. A microphone converts acoustical energy first to mechanical energy and then to electrical energy.

Hybrid network : The hybrid network (sometimes called a hybrid coil or duplex coil) in a telephone set is a special balanced transformer used to convert a two-wire circuit(the local loop) in to a four wire circuit(the telephone set) and the vice-versa, thus enabling full duplex operation over a two wire circuit. In essence, the hybrid network separates the transmitted signals from the received signals. Outgoing voice signals are typically in the 1-V to 2-V range, while incoming voice signals are typically half that value. Another function of the hybrid network is to allow a small portion of the transmit signal to be returned to the receiver in the form of a sidetone. In sufficient sidetone causes the speaker to raise his voice, making the telephone conversation seem unnatural. Too much sidetone causes the speaker to talk too softly, thereby reducing the volume that the listener receives.

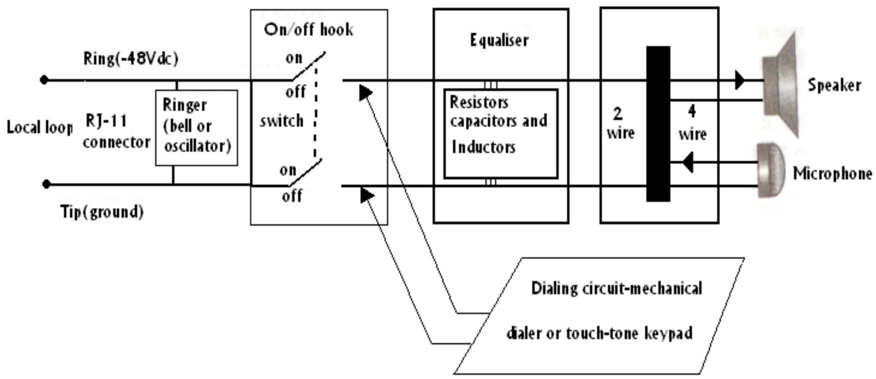


FIG – Functional Block Diagram Of a Standard Telephone Set.

Dialing Circuit: The dialing circuit enables the subscriber to output signals representing digits, and this enables the caller to enter the destination telephone number. The dialing circuit could be a rotary dialer, which is nothing more than a switch connected to a mechanical rotating mechanism that controls the number and duration of the on/off condition of a switch. However, more than likely, the dialing circuit is either an electronic dial-pulsing circuit or a Touch-Tone keypad, which sends various combinations of tones representing the called digits.

(**ii**) **CPU Based Exchange :** 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. A typical control configuration of an ESS using centralized SPC is shown in Fig. A centralized SPC configuration may use more than one processor for redundancy purposes. Although there are a large number of services which can be offered by CPU based exchange, they may be grouped under four broad categories.

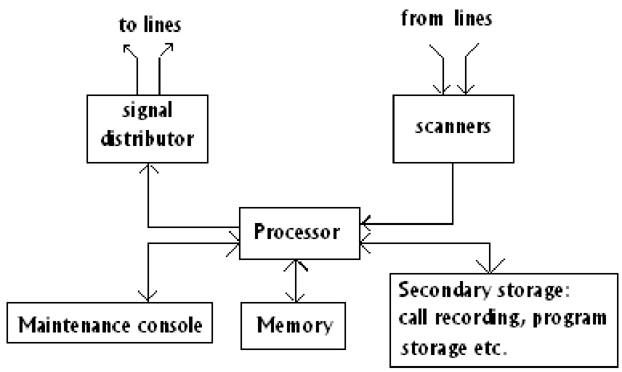


FIG - Typically CPU Based Exchange

1. Services associated with the calling subscribers and designed to reduce the time spent on dialing and the number of dialing errors.
2. Services associated with the called subscriber and designed to increase the call completion rate.
3. Services involving more than two parties.
4. Miscellaneous services.

These new services are known as supplementary services and some of the prominent ones are as follows:

Category 1:
- Abbreviated dialing
- Recorded number calls or no dialing calls
- Call back when free.

Category 2:
- Call forwarding
- Operator answer.

Category 3:
- Calling number record
- Call waiting
- Consultation hold
- Conference calls.

Category 4:
- Automatic alarm
- STD barring
- Malicious call tracing

A subscriber issues commands to an exchange to activate or deactivate a service, record or clear data in the subscriber line data area or solicit an acknowledgement from the exchange. As an example, a user may enable or disable STD facility on his line by using a command. A command may or may not have data associated with it. The number of digits in the data, when present, may vary depending upon the command. As a result, subscriber commands are designed to be of variable length necessitating the use of an end-of-command symbol

Q.29. Explain the terms.
(i) Register marker.
(ii) Conditional selection. **(6)**

Ans:
(i) Register marker: Strowger selectors perform counting and searching. However, the crossbar switch has no 'intelligence'. Something external to the switch must decide which magnets to operate. This is called a **Register Marker**. Since it takes less than a second to operate the switch, a marker can control many switches and serve many registers. Thus, even a large exchange needs few markers. This is further stage of common control, which we shall call centralized control.
(ii) Conditional selection: When a marker is instructed to set up a connection from a given incoming trunk to a given outgoing trunk, this also defines the link to be used and the select and bridge magnets to be operated to make the connection. The maker does not make the connection until it interrogated the busy/free condition of the outgoing trunk and of the relevant link. Only if both are found to be free does it operate the switches. This is called **Conditional Selection.**

Q.30. In a hundred-line exchange 24 two-motion selectors are used. Draw the schematic you suggest for this exchange and explain its working. How many simultaneous calls can be made during peak hour in this exchange? **(10)**

Ans:
The desired schematic for 100 lines exchange with 24 two motion selectors is shown in Fig.

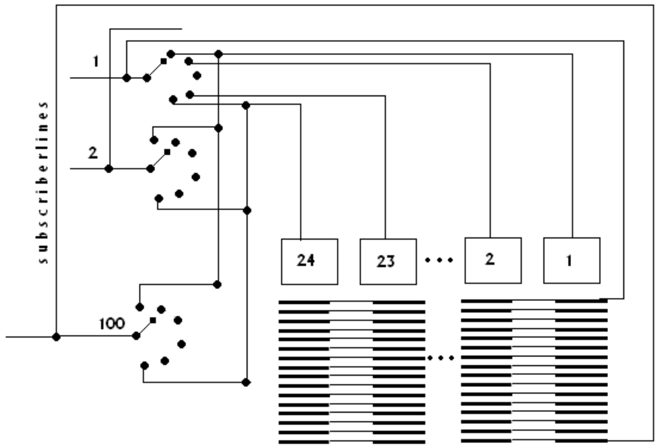


FIG – 100 Line – Exchange.

In the case, 24 simultaneous calls can be put through the switch . Typically, a 24-outlet uniselector is used as a selector hunter. Each of the 24 outlets is connected to one two-motion selector. Thus, a subscriber has access to all the 24 two-motion selectors in the system. The corresponding outlets of all the selector hunters are commoned and thus, all subscribers have access to all the two-motion selectors. This scheme is shown in Fig. The call establishment in this scheme takes place in two steps. Firstly, when the subscriber lifts his receiver handset, his uniselector hunts through the contact positions and seizes a free two-motion selector. At the next step, the two-motion selector responds to the dial pulses for appropriate connection.

Q.31. Compare electromechanical switching system with electronic switching system. **(6)**

Ans:

Sr. No.	**Electromechanical Switching System**	**Electronic Switching System**
1.	Makes use of electromechanical switches	Makes use of electronic switching elements such as diodes/ transistors.
2.	Require large space	Less space required and less bulks.
3.	Regular maintenance required	Less maintenance required
4.	Common control is used	Computer control is used
5.	Cannot provide additional services to customers	Can provide enhanced services to customers.

Q.32. What is centralized SPC, what are its modes of operation; explain the working of any one of these? **(8)**

Ans:
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. A typical control configuration of an ESS using centralized SPC is shown in Fig. A centralized SPC configuration may use more than one processor for redundancy purposes. In almost all the present day electronic switching systems using centralized control, only a two-processor configuration is used. Dual processor architecture may be configured to operate in one of three modes:

a. Standby mode
b. Synchronous duplex mode
c. Load Sharing mode

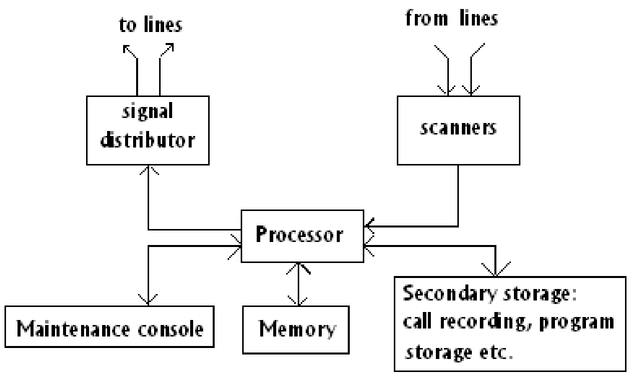


FIG - Typically Centralized SPC Organization.

Standby mode of operation is the simplest of dual processor configuration operations. Normally, one processor is active and the other is on standby, both hardware and software wise. The Standby processor is brought online only when the active processor fails. An important requirement of this configuration is the ability of the standby processor to reconstitute the state of the exchange system when it takes over the control, i.e. to determine which of the subscribers and trunks are busy or free, which of the paths are connected through the switching network etc. In small exchanges, this may be possible by scanning all the status signals as soon as the standby processor is brought into operation. In such a case, only the calls which are being established at the time of failure of the active processor are disturbed. In large exchanges, it is not possible to scan all the status signals within a reasonable time. Here, the active processor copies the status of the system periodically, say every five 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 and continues the operation. In this case, only the calls which changed status between the last update and the failure of the active processor are disturbed. Fig. shows a standby dual processor configuration with a common backup storage. The shared secondary storage need not be duplicated and simple unit level redundancy would suffice.

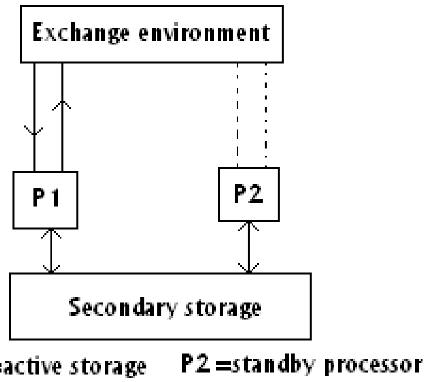


FIG - Standby Dual Processor Configuration.

Q.33. Explain the architecture of SS7 and compare with seven-layer OSI architecture. **(8)**

Ans:

A block schematic diagram of the CCITT no. 7 signaling system is shown in fig. Signal messages are passed from the central processor of the sending exchange to the CCS system. This consists of the microprocessor based subsystem.

The signaling control subsystems, the signaling termination subsystem and the error control subsystem. The signaling control subsystem structures the messages in the appropriate format and queues them for transmission. When there are no messages to send, it generates filler messages to keep the link active. Messages then passed to the signaling termination sub system, where complete signal units (SU) are assembled using sequence numbers and check bits generated by the error control subsystem. At the receiving terminal, the reverse sequence is carried out. The levels are as follows:

Level 1: The Physical Layer
Level 2: The Data Link Level
Level 3: The signaling network level
Level 4: The User Part

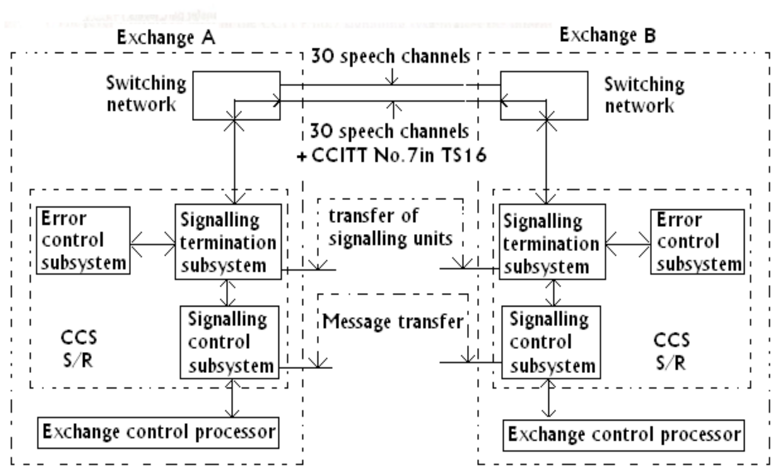


FIG – Block Schematic Diagram of CCITT No.7 Signally System

The relationship between these levels and the layers of the OSI model is shown in Fig. The user part encompasses layers 4 to 7 of the OSI model.

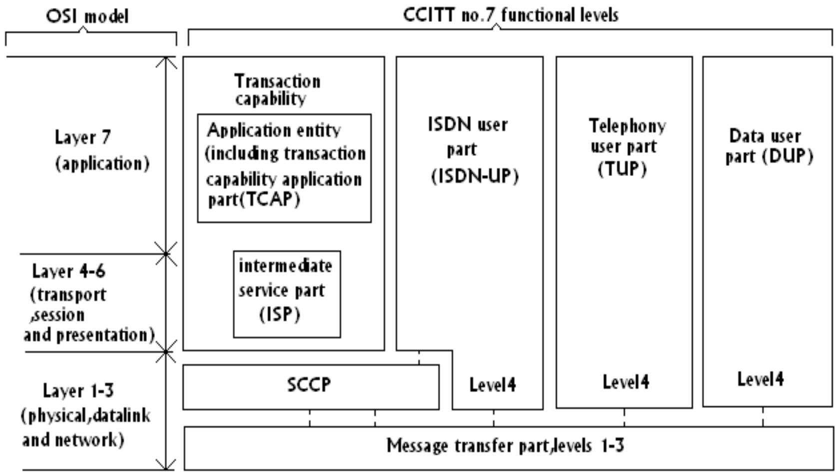


FIG – Relationship between CCITT No.7 Functional levels and OSI layers

Level 1 is the means of sending bit streams over a physical path. It uses times lot 16 of a 2 M bit/s PCM system or times slot 24 of a 1.5 M bit/s system.
Level 2 performs the functions of error control, link initialization, error rate monitoring, flow control and delineation of messages.
Level 3 provides the functions required for a signaling network. Each node in the network has a single point odd, which is a 14 bit address. Every message contains point codes of the originating and terminating nodes for those messages.
Levels 1 to 3 form the message transfer part (MTP) of CCITT no. 7 .
Level 4 is the user part. This consists of the processes for handling the service being supported by the signaling system. The message transfer part is capable of supporting many different user parts. So far, three have been defined: the telephone user part (TUE), the data user part(DUP) and the (ISDN) user part (ISDN-UP).

Q.34. What are different control function categories, explain, how they help in signalling and control. **(8)**

Ans:
In some switching systems, the control subsystem may be an integral part of the switching network itself. Such system is known as **direct control** switching systems. Those systems in which the control subsystem is outside the switching network are known as **common control** switching system. Strowger exchanges are usually direct control systems, whereas crossbar and electronic exchanges are common control system. All stored program control systems are common control systems. Common control is also known as indirect control or register control.
Common Control Switching System: A functional block diagram of a common control switching system is shown in Fig. The control functions in a switching system may be placed under four broad categories:

i. Event monitoring.
ii. Call processing.
iii. Charging.
iv. Operation and maintenance

Events occurring outside the exchange at the line units, trunk junctions and in exchange signaling receiver/sender units are all monitored by control subsystem. Typical events include all requests and call release signals at the line units. The occurrences of the events are signaled by operating relays which initiate control action. The control subsystem may operate relays in the junctions, receivers/senders and the line units, and thus command these units to perform certain functions. Events monitoring may be distributed. For examples, 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 market for dial tone, and the register finder is activated to seize a free register. Identity of the calling line is used to determine the line category and the class of service to which the subscriber belongs. A register appropriate to the line category is chosen, which then sends out the dial tone to the subscriber, in readiness to receive the dialing information. As soon as the initial digits (usually 2-5) which identify the exchange are received in the register, the register continues to receive the remaining digits. The initial translator determines the route for the call through the network and decides whether a call should be put through or not. It also determines the charging methods and the rates applicable to the subscriber. Initial translation may also take into account instructions from the operating personnel and information regarding the status of the network.

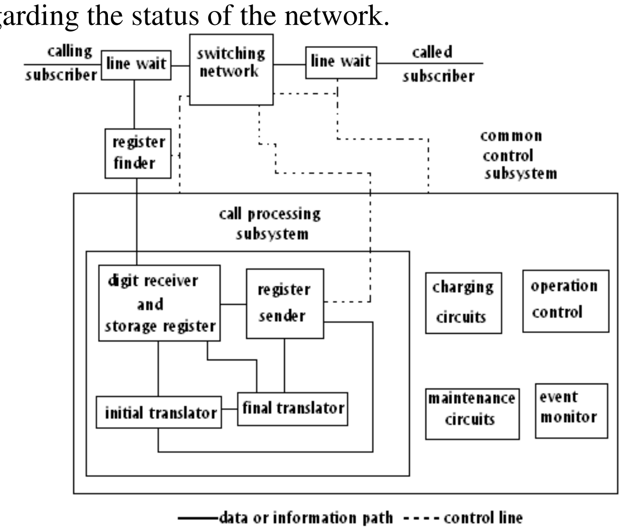


FIG - Common Control Switching System

If a call is destined to a number in an exchange other than the present one processing the digits, the initial translator generates the require routing digits and passes them on to the register sender. Here the digits corresponding to the subscriber identification are concatenated and the combined digit pattern is transmitted over the trunks to the external exchange. Register sender uses appropriate signaling technique, depending on the requirements of the destination exchange. If the call is destined to a subscriber within the same exchange, the digits are processed by the final translator. The translation of directory number to equipment number takes place at this stage. The final translator determines the line unit to which a call must be connected and the category of the called line. The category information may influence charging and connection establishment. In some practical implementations, both initial and final code translator functions are performed by a single translator.

Controlling the operation of the switching network is an important function of the common control subsystem. This is done by marking the switching elements at different stages in accordance with a set of binary data defining the path and then commanding the actual connection of the path. Path finding may be carried out at the level of the common control unit or the switching network.

Q.35. Show how finite state machine model helps in designing a switching system and give a typical example. **(8)**

Ans:
Switching system basically belongs to the class of finite state machines (FSM) which are asynchronous in nature and follows a sequential logic for their operation. They can be modeled by using a combinational part and a memory part as shown in Fig.. In FSM , the status of the output circuits not only depends upon the inputs but also upon the current state of the machine. Asynchronous sequential operation gives rise to many problems due to transient variations that may occur in the logic circuits and memory elements. Clocked synchronous operation shown in Fig. overcome such problems.

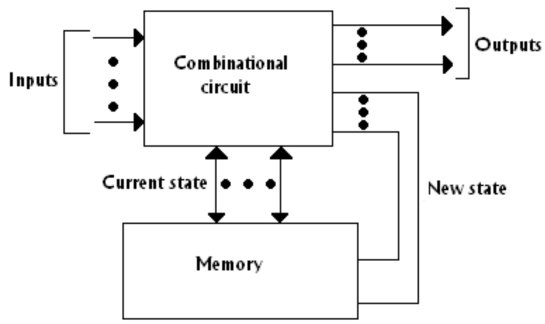


FIG – FSM Model.

Q.36. Through two block diagrams explain the difference between Space division and time division switching. **(8)**

Ans:
Space and Time Switching:

Space Switches: Connections can be made between incoming and outgoing PCM highways by means of a cross point matrix of the form shown in Fig. However, different channels of an incoming PCM frame may need to be switched by different cross points in order to reach different destinations. The cross point is therefore a two-input AND gate. One input is connected to the incoming PCM highway and the other to a connection store that produce a pulse at the required instant. A group of cross points gates can be implemented as an integrated circuit, for example by using a multiplexer chip.

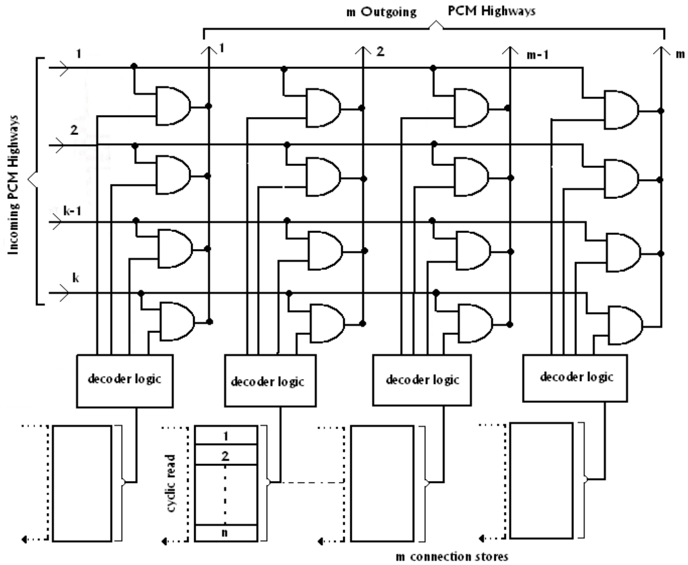


FIG – Space Switch.

Fig. shows a space switch with k incoming and m outgoing PCM highways, Each carrying n channels. The connections store for each column of cross points is a memory with an address location for each time-slot, which stores the number of the cross points to be operated in that time slot. This number is written into the address by the controlling processor in order to setup the connection. The numbers are read out cyclically, in synchronism with the incoming PCM frame. In each time slot, the number stored at the corresponding store address is read out and decoding logic converts this into a pulse or a single lead to operate the relevant cross point.
Since a cross point can make a different connection in each of the n time-slots, it is equivalent to n cross points in a space division network. The complete space switch is thus equivalent to n separate k x m switches in a space division switching network.

Time Switches: The principle of a time switch is shown in Fig. It connects an incoming n channel PCM highway to an outgoing n channel PCM highway. Since any incoming channel can be connected to any outgoing channel, it is equivalent to space division cross point matrix with n incoming and n outgoing trunks, as shown in Fig. Time-slot interchange is carried out by means of two stores, each having a storage address for every channel of the PCM frame. The speech store contains the data of each of the incoming time-slots (i.e. its speech sample) at a corresponding address. Each address of the connection store corresponds to a time slot on the outgoing highway. It contains the number of the time-slot on the incoming highways whose sample is to be transmitted in that outgoing time-slots. Information is read into the speech store cyclically in synchronism with the incoming PCM systems; however, random access read out is used .The connection store has cyclic read out, but writing in is non cyclic. To establish a connection, the number (X of the time-slot of an incoming channel is written into the connection store at the address corresponding to the selected outgoing channel (Y).During each cyclic scan of the speech store ,the incoming PCM sample from channel X is written into address X. During each cyclic scan of the connection store, the number X is read out at the beginning of time-slot Y. This is decoded to select address X of the speech store, whose contents are read out and sent over the outgoing highway.

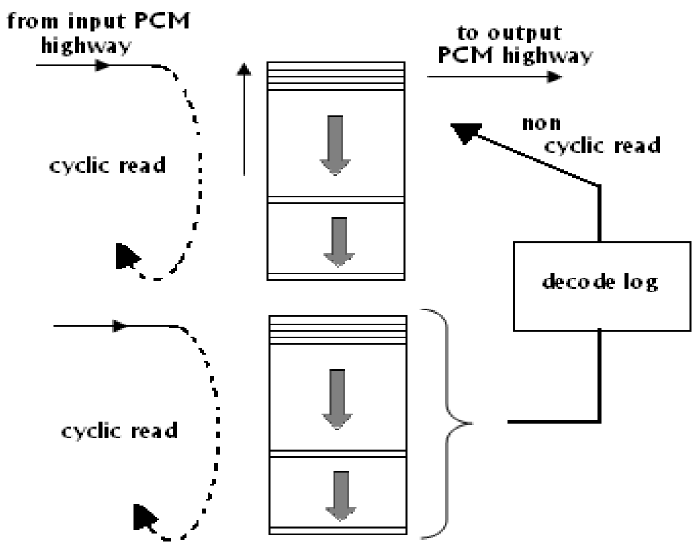
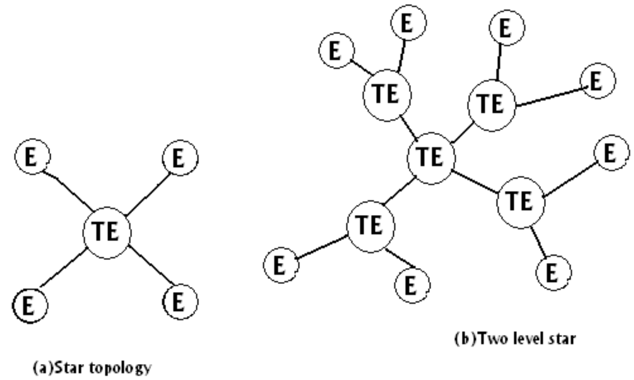


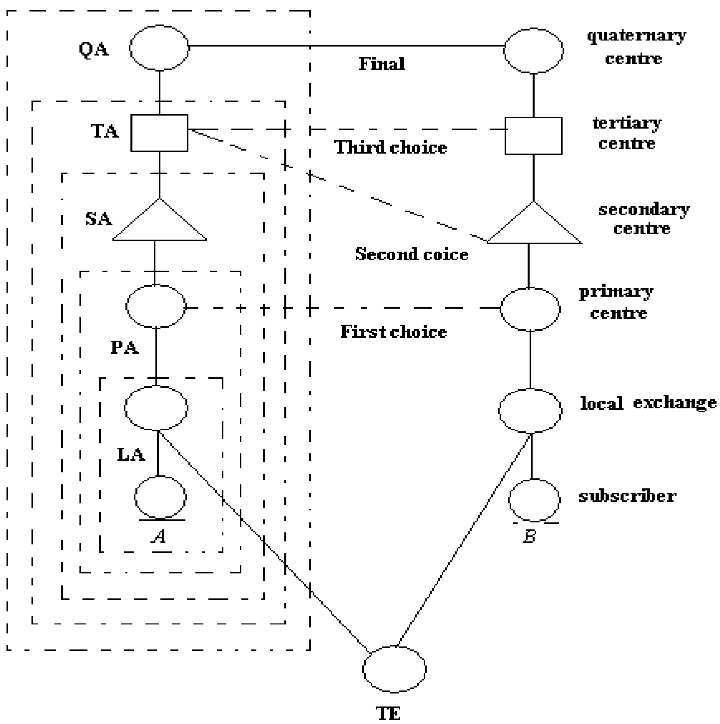
FIG – Time Switch

Q.37. Explain the CCITT hierarchical structure of switching and routing using block schematic. **(8)**

Ans:
Hierarchical network are capable of handing heavy traffic where required, and at the same time use minimal number of trunk groups. A 5-level switching hierarchy is recommended by CCITT as shown in Fig. In a strictly hierarchical network, traffic from subscriber A to subscriber B and vice-versa flows through the highest level of hierarchy, viz. Quaternary centres in Fig. A traffic route via the highest level of hierarchy is known as the final route. However, if there is a high traffic intensity between any pair of exchanges, direct trunk groups may be established between them as shown by dashed lines in Fig.. These direct routes are known as high usage routes. Wherever high usage routes exist, the traffic is primarily routed through them. Overflow traffic, if any, is routed along the hierarchical path. No overflow is permitted from the final route. In Fig. The first choice routing for traffic between subscribers A and B is via the high usage route across the primary centres. The second and the third choice route and the final route are also indicated in Fig. A hierarchical system of routing leads to simplified switch design. Three methods are commonly used for deciding on the route for a particular connection:

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





(e) CCITT hierarchical structure

In right-through routing the originating exchange determines the complete route from source to destination. No routing decisions are taken at the intermediate routes. In the absence of a computer, only a predetermined route can be chosen by the originating exchange. However, there may be more than one predetermined route and the originating node may select one out of these, based on certain like time of the day, even distribution of traffic etc.

Own-exchange routing or distributed routing allows alternative routes to be chosen at the intermediate nodes. Thus the strategy is capable of responding to changes in traffic loads and network configurations. Another advantage of distributed routing is that when new exchanges are added, modifications required in the switch are minimal.

Computers are used in network with common channel signaling (CCS) features. In CCS, there is a separate computer-controlled signaling network. With computers in position, a number of sophisticated route selection methods can be implemented. Computer based routing is a standard feature in data networks. A strictly hierarchical network suffers from serious drawback i.e. its poor fault tolerance future.

Q.38. What are the advantages of CCS over in-channel signalling? **(8)**

Ans:
The advantages of CCS over in-channel signalling are listed below:

a. Information can be exchange between the processors much more rapidly than when channel associated signaling is used.
b. As a result, a much wider repertoire of signals can be used and this enables more services to be provided to customers.
c. Signals can be added or changed by software modification to provide new services.
d. There is no longer any need for line signaling equipments on every junction which results in a considerable cost saving.
e. Since there is no line signaling, the junctions can be used for calls from B to A in addition to calls from A to B. Both ways working requires fewer circuit to carry the traffic than if separate groups of junctions are provided from A to B and from B to A.
f. Signals relating to a call can be sent while the call is in progress. This enables customers to alter connections after they have been set up.
g. Signals can be exchanged between processors for functions other than call processing, for example for maintenance or network management purposes.

Q 39. What is a Modem? What is the need of MODEM in data communication? Explain at least one modulation technique used for high speed modems. **(10)**

Ans:
Modems are essentially used to interface digital circuits to transmit information on analogue channels like telephone systems. Modem (from modulator-demodulator) is a device that modulates an analogue carrier signal to encode digital information, and also demodulates such a carrier signal to decode the transmitted information.
The goal is to produce a signal that can be transmitted easily and decoded to reproduce the original signal data.Modems can be used over any means of transmitting analog signals. Two main modulation schemes are currently being used to implement ADSL: carrier less amplitude/ phase (CAP) a single carrier modulation scheme based on quadrature amplitude modulation (OAM); and discrete multi-tone (DMT), a multichannel modulation scheme. The choice between them naturally depends on how well they perform in the presence of impairments on the existing copper twisted –pair access cabling (see side bar), because these can limit the transmission capacity. In addition, high bit rate services carried by ADSL must not interfere with other services. In essence, multicarrier modulation superimposes a number of carrier modulated waveforms to represent the input bit stream. The transmitted signal is the sum of these sub-channels (or tones), which have the same band width and equally spaced center frequencies. The number of tones must be large enough to ensure good performance. In practice, a value of 256 provides near optimum performance while ensuring manageable implementation complexity.

Need of Modem: Modems are used to interface computers, computer networks and other terminal equipment to telecommunication lines and radio channels. The word **Modem** is a contraction derived from the words modulator and demodulator. The modems at the transmitting station changes the digital output from the computer or other data terminal equipment to a form which can be easily sent via a communication circuit, while the receiving modem reverse the process. Modems differ in rate of data transmission, modulation methods and bandwidth and standards have been developed to provide compatibility between various manufacturers' equipment and systems.

FSK – Frequency Shift Keying: In this technique the frequency of the carrier signal is changed according to the data. The transmitter sends different frequencies for a "I" than for a "0" as shown in Fig. The disadvantages of this technique are that again (as it was with amplitude modulation) the rate of frequency changes is limited by the bandwidth of the line, and that distortion caused by the lines makes the detection even harder than amplitude modulation.

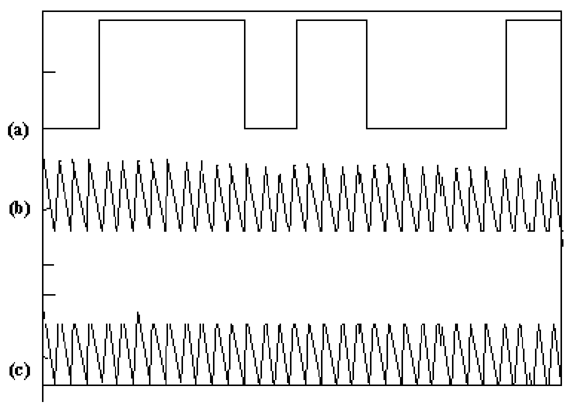


FIG – FSK

Q.40. What are the advantages and disadvantages of packet switching over circuit switching? **(6)**

Ans:
The comparison of packet switching and circuit switching showing advantages and disadvantages of packet switching over circuit switching is given below:

Circuit Switching	Packet Switching
Dedicated transmission path.	No dedicated transmission path
Transmission of data.	Transmission of packets.
Operate in real time.	Near real time.
Message not stored.	Message held for short time
Path established for entire message.	Route established for each packet
Call setup delay.	Packet transmission delay.
Busy signal if called party busy.	No busy signal.
Blocking may occur.	Blocking cannot occur
User responsible for message-loss	Network may be responsible for each

protection.	packet but not for entire message.
No speed or code conversion.	Speed and code conversion.
Fixed bandwidth transmission.	Dynamic use of bandwidth.
No overload bits after initial setup delay.	Overload bits in each packet.

Q.41. Explain crossbar exchange, with all call processing steps and diagrams **(8)**

Ans:

The basic idea of crossbar switching is to provide a matrix of n x m sets of contacts with only n + m activators or less to select one of the n x m sets of contacts. This form of switching is also known as coordinate switching as the switching contacts are arranged in a x-y-plane. A diagrammatic representation of a cross point switching matrix is shown in Fig. There is an 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 cross point switch and remains separated or open when not in use. The contact points are mechanically mounted (and electrically insulated) on a set of horizontal and vertical bars shown as dotted lines. The bars, in turn, are attached to a set of electromagnets.

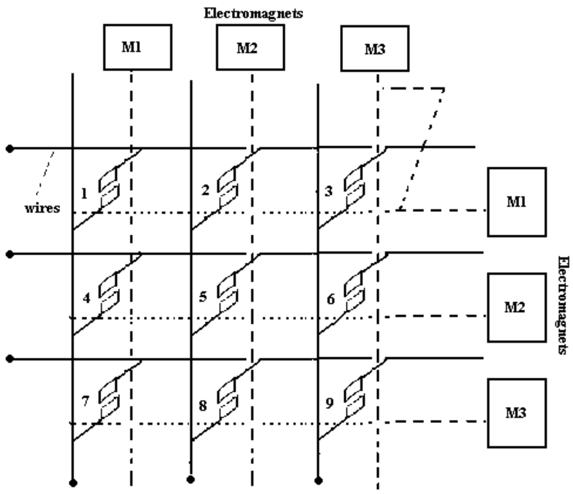


FIG - 3 X 3 Crossbar Switching

When an electromagnetic energized, the bar attached to it slightly rotates in such a way that the contact points attached to the bar move closer to its facing contact points but do not actually make any contact. Now if an electromagnet in the vertical direction is energized, the corresponding bar rotates causing the contact points at the intersection of the two bars to close. This happens because the contact points move towards each other. For example, if electromagnets M2 and M3' are energized, a contact is established at the cross point 6 such that the subscriber B is connected to the subscriber C.

Q.42. Explain simple telephone communication system with circuit and equation of current flow in microphone? **(8)**

Ans:
Simple Telephone Communication: In the simplest form of a telephone circuit, there is one way communication involving two entities, one receiving (listening) and the other transmitting (talking).

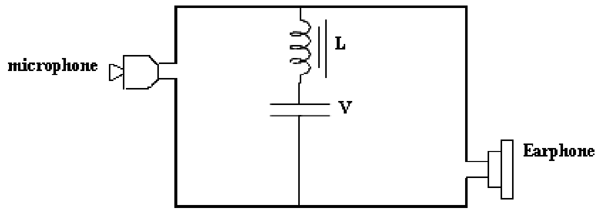


FIG - A Simplex Telephone Circuit

Simplex communication: The microphone and the earphone are the transducer elements of the telephone communication system. Microphone converts speech signal to electrical signals and the earphone converts the electrical signals into audio signals. Most commonly used microphone is carbon microphone. Carbon microphones do not produce high fidelity signals, but give out strong electrical signals at acceptable quality levels for telephone conversation. In carbon microphones, a certain quantity of small carbon granules is placed in a box. Carbon granules conduct electrically and the resistance offered by them is dependent upon the density with which they are packed. One side the box is covered is flexible and is mechanically attached to a diaphragm. When sound waves impinge on the diaphragm, it vibrates, causing the carbon granules to compress or expand, thus changing the resistivity offered by the granules. If a voltage is applied to the microphone, the current in the circuit varies according to the vibration of the diaphragm.
When the sound waves impinge on the diaphragm, the instantaneous resistance of the microphone is given by

$$r_i = r_0 - r \sin \omega t$$

Where
r_0 = Quiescent resistance of the microphone when there is no speech signal.
r = Maximum variation in resistance offered by the carbon granules, $r < r_0$.
The negative sign indicates that when the carbon granules are compressed the resistance decreases and vice versa. Ignoring impedances external to the microphone in the circuit given in Fig. without loss of generality, the instantaneous current in the microphone is given by

$$i = V/(r_0 - r \sin \omega t) = I_0(1 - m \sin \omega t)^{-1}$$

Where

$I_0 = V/r_0$ = Quiescent current in the microphone .

$m = r/r_0, \quad m < 1$

By binomial theorem, the Eq. may be expanded as

$$i = I_0 (1 + m \sin \omega t + m^2 \sin^2 \omega t + \ldots)$$

If the value of m is sufficient small, which is usually the case in practice, higher-order terms can be ignored in the above Eq. Giving thereby

$$i = I_0 (1 + m \sin \omega t)$$

Thus, the carbon granule microphone acts as a modulator of the direct current which is analogous to the carrier wave in AM system. The quantity m is equivalent to the modulation index in AM.

Q.43. Find the total number of link L having five entities? Explain differences between folded and non-folded network. **(8)**

Ans:
Folded network: When all the inlets/outlets are connected to the subscriber lines, the logical connection appears as shown in Fig. In this case, the output lines are folded back to the input and hence the network is called a folded network. Four types of connection may be established:

1. Local call connection between two subscribers in the system.
2. Outgoing call connection between a subscriber and an outgoing trunk.
3. Incoming call connection between an incoming trunk and a local subscriber.
4. Transit calls connection between an incoming trunk and an outgoing trunk.

In a folded network with N subscriber, there can be a maximum of N/2 simultaneous calls or information interchanges.

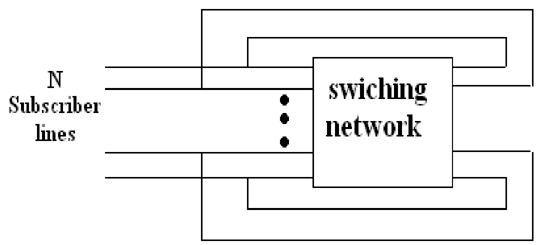


FIG – Folded Network

Non-Folded Network: In a switching network, all the inlet/outlet connection may be used for inter exchange transmission. In such a case, the exchange does not support local subscribers and is called a transit exchange. A switching network of this kind is shown in Fig. and is called a non-folded network. In non-folded network with N inlets and N outlets, N simultaneous information transfers are possible. Consequently, for a non-folded network to be non-blocking, the network should support N simultaneous switching paths.

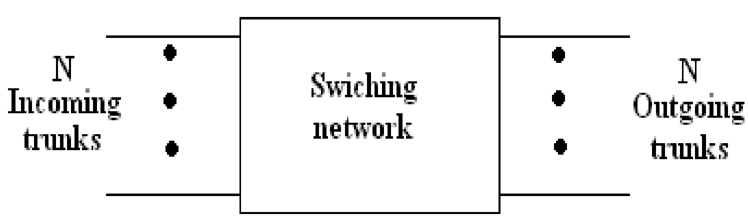


FIG – Non-Folded Network

Q.44. List all seven layer of OSI model and describe function of application layer. **(8)**

Ans:
The layers of OSI model are as follows:

(1) The Physical Layer: This defines an interface in terms of the connections, voltage levels and data rate, in order for signals to be transmitted bit by bit.

(2) The link Layer: This provides error detection and correction for a link to ensure that the exchange of data is reliable. It may require the data stream to be divided into blocks, called packets, for inserting error-checking bits or for synchronization. However, transparency is preserved for the data bits in these blocks.

(3) The network layer: This is concerned with the operation of the network between the terminals. It is responsible for establishing the corrections between the appropriate network nodes.

(4) The transport layer: This is responsible for establishing a network independent communication path suitable for the particular terminal equipments (e.g. providing the appropriate data rate and end-to-end error control). It thus relieves the user from being concerned with such details.

(5) The session layer: This is concerned with setting up and maintaining an operational session between terminals. For example, 'signing on' at the commencement of a task and 'signing off ' at its end.

(6) The presentation layer: This is concerned with the format of the data presented, in order to overcome difference in representation of the information as supplied to one terminal and required at the other. Its purpose is to make communication over the network machine-independent.

(7) The application layer: This defines the nature of the task to be performed. It supplies the user with the applications programs needed. Examples include electronic mail, word processing, banking transactions, etc.

Q.45. Explain topology method used in LAN technology in detail. **(8)**

Ans:
LAN Topologies: Network topology is a physical schematic which shows interconnection of the many users. There are four basic topologies as under:

(i) Direct Connection or one to all topology
(ii) Star topology
(iii) Bus Topology
(iv) Ring topology

(i) Direct connection or one to one topology: In the one to all topology, there is a path between every node and every other node. The number of paths required is defined by the equation $P = (n^2-n)/2$, where n is number of nodes and P, total number of paths. Thus number of paths increases considerably as the number of nodes increases. Each node must be physically able to connect to paths to every other node so a very large connector and associated interface support circuitry is needed at each node. Adding a new node requires major rewiring and affects the software that manages communications for the system.

(ii) Star Topology: In Star topology, all user nodes are connected to central node point that interconnects all the individual users links and nodes. Data flows from one node, through the star centre to the desired receiving node. The Central node is like a large switch which routes the data from the input line to the output line. Advanced central nodes can have multiple switching paths, so several paths exist at the same time. This topology is used in a telephone system central office. Addition of a new user node requires running only a single link from the star centre to the user without disturbing any other node.

(iii)Bus Topology: This topology shares a single link or path way among all users. This common single path way is known as bus. In this topology, the link serves as a high way for all data signals, and users connect on to the bus at their node location. In bus configurations, network control is not centralized to a particular node. Here control is distributed among all nodes connected to the LAN. Data transmission on a bus network is usually in the form of small packets containing user addresses and data. When one node/user desires to transmit data to another station, it monitors the bus to determine if it is currently being used. If no other nodes/users are communicating over the network, the monitoring node/user can start to transmit its data. Each node must monitor all transmission on the network and determine which are intended for them.

(iv) Ring Topology: In ring topology, all user nodes are connected with the physical path acting as links of a chain and the last user node is connected back to the first node. A signal going on to the next node must be processed by the first node, which then passes it through to the next node. Adding a new user requires breaking the ring temporarily, inserting the new node and then reestablishing the complete ring path.

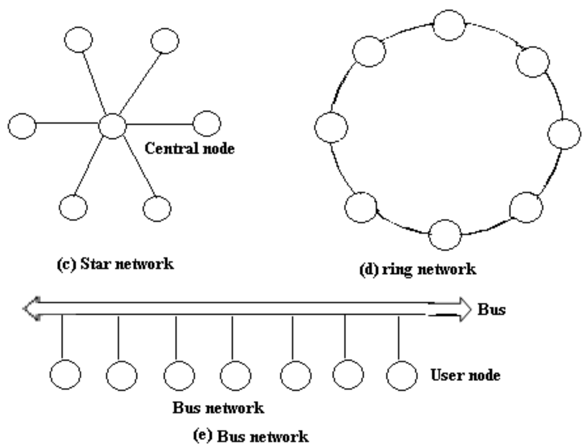


FIG - Networks Topologies

Q.46. What is Traffic Engineering? Define the term busy hour, traffic intensity and grade of service. **(8)**

Ans:

Traffic engineering provides the basis for analysis and design of telecommunication network to carry a given traffic at a particular loss probability. It provides a means to determine the quantum of common equipments required to provide a particular level of service.

Busy Hour: Continuous 1- hour period lying wholly in the time interval concerned, for which the traffic volume or the number of calls attempts is greatest.

Traffic Intensity: The traffic load on a given network may be on the local switching unit, interoffice trunk lines or other common subsystem. For analytical treatment in this text, all the common sub systems of a telecommunication network are collectively termed as servers. In other publications the term link or Trunk is used. The traffic on the network may then be measured in terms of the occupancy of the servers in the network. Such a measure is called the traffic intensity which defined as:

A = period for which a server is occupied/ Total period of observation

Grade of Service: In loss systems, the traffic carried by the network is generally lower than the actual traffic offered to the network by the subscribers. The overload traffic is rejected and hence it is not carried by the network. The amount of traffic rejected by the network is an index of the quality of the service offered by the network and is termed as grade of service (GOS) and is defined as the ratio of lost traffic to offered traffic. Offered traffic is the product of the average number of calls generated by the users and the average holding time per call. GOS is given by

$$GOS = A-A_0 / A_0$$

Where

A= offered traffic

A_0 = carried traffic

$A-A_0$= lost traffic

Q.47. Discuss briefly subscriber loop system. Give some technical specification for subscriber lines. **(8)**

Ans:

Subscriber Loop System: 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 this pair of wires. It is un widely to run physically independent pairs from every subscriber premises to the exchange. It is far easier to lay cables containing a number of pairs of wires for different geographical locations and run individual pairs as required by the subscriber premises.

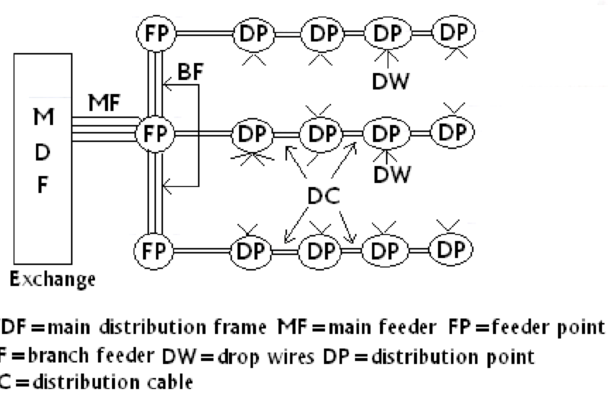


FIG – Cable Hierarchy For Subscriber Loops.

Generally four levels of cabling are used as shown in Fig. At the subscriber end, the drop wires are taken to a distribution point. The drop wires 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. Many distribution cables from nearby geographical locations are terminated on a feeder point where they are connected to branch feeder cables which, in turn, are connected to the main feeder cable. The main feeder cables carry a larger number of wire pairs, typically 100-2000, than the distribution cables which carry typically 10-500 pairs. The feeder cables are terminated on a main distribution frame (MDF) at the 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. The MDF thus provides a flexible interconnection mechanism which is very useful in reallocating cable pairs and subscriber numbers. The technical specifications for subscriber lines include:

1. Diameter of wire
2. D.C. resistance per Km
3. Attenuation per Km

Q.48. Discuss different Routing plan adopted in a Telephone network. **(8)**

Ans:

Hierarchical networks are capable of handing heavy traffic where required, and at the same time use minimal number of trunk groups. Three methods are used for deciding on the route for a particular connection:

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

In **right-through routing** the originating exchange determines the complete route from source to destination. No routing decisions are taken at the intermediate routes. In the absence of a computer, only a predetermined route can be chosen by the originating exchange. However, there may be more than one predetermined route and the originating node may select one out of these, based on certain like time of the day, even distribution of traffic etc.

Own-exchange routing or distributed routing allows alternative routes to be chosen at the intermediate nodes. Thus the strategy is capable of responding to changes in traffic loads and network configurations. Another advantage of distributed routing is that when new exchanges are added, modifications required in the switch are minimal.

Computers are used in network with common channel signaling (CCS) features. In CCS, there is a separate computer-controlled signaling network. With computers in position, a number of sophisticated route selection methods can be implemented. Computer based routing is a standard feature in data networks.

Q.49. How does a touch tone receiver differs from pulsed dial receiver? Explain with schematic. **(8)**

Ans:

Pulse dialing:

1. Generated through make and break contact.
2. DC Current pulse is generated.
3. Each number is separated by a short pause to prevent overlapping.
4. Codes are unary except for 0.
5. Dial is rotary.
6. Dial is slow moving.

DTMF Dialing

1. Uses push buttons
2. Uses 8 different frequencies in pairs.
3. 16 different characters can be represented.
4. The frequencies used prevent generating of harmonics.
5. Each number is represented by a high signal & low signal

Dual Tone Multi Frequency (DTMF) was first introduced in 1963 with 10 buttons in Western Electric 1500 –type telephones. DTMF was originally called Touch-Tone. DTMF is a more efficient means than dial pulsing for transferring telephone numbers from a subscriber's location to the central office switching machine. DTMF is a simple two-of –eight encoding scheme where each digit is represented by the linear addition of two frequencies. DTMF is strictly for signaling between a subscriber's location and the nearest telephone office or message switching center. DTMF is sometimes confused with another two-tone signaling system called multi frequency signaling (Mf), which is a two-of-six code designed to be used only to convey information between two electronic switching machines.
Fig. shows the four-row-by-four column keypad matrix used with a DTMF keypad. As the figure shows, the keypad is comprised of 16 keys and eight frequencies. Most household telephones, however, are not equipped with the special-purpose keys located in the fourth column (i.e. , the A,B,C, and D keys). Therefore, most household telephones actually use two-of-seven tone encoding scheme. The four vertical frequencies (called the low group frequencies) are 697 Hz, 852 Hz, and 941 Hz, and the four horizontal frequencies (called the high group frequencies) are 1209 Hz, 1336 Hz, 1477 Hz, and 1633 Hz. The frequency tolerance of the oscillators is $\pm$.5%. As shown in Fig., the digits 2 through 9 can also be used to represent 24 of the 26 letters (Q and Z are omitted). The letters were originally used to identify one local telephone exchange from another.

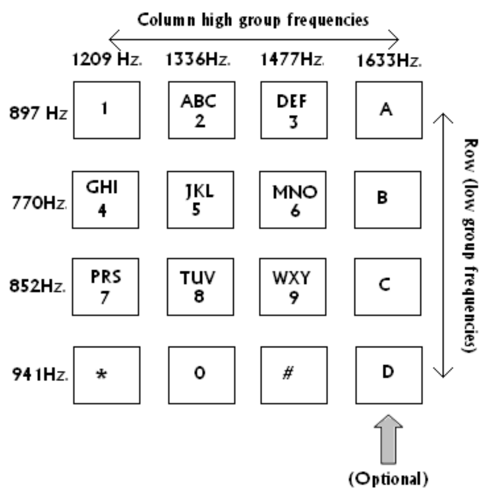


FIG - DTMF Keypad Layout

Dial Pulses: Dial pulsing (sometimes called rotary dial pulsing) is the method originally used to transfer digits from a telephone set to the local switch. Pulsing digits from a rotary switch began soon after the invention of the automatic switching machines. The concept of dial pulsing is quite simple and is depicted in Fig. The process begins when the telephone set is lifted off hook, completing a path for current through the local loop. When the switching machine detects the off-hook condition, it responds with dial tone. After hearing the dial tone, the subscriber begins the dial pulsing digits by rotating a mechanical dialing mechanism and then letting it returns to its rest position. As the rotary switch returns to its rest position, it outputs a series of dial pulses corresponding to digit dialed.

When a digit is dialed, the loop circuit alternately opens (breaks) and closes (makes) a prescribed number of times. The number of switch make/break sequences corresponds to the digit dialed (i.e., the digit 3 produces three switch openings and three switch closers).

Dial pulses occur at 10 make/break cycles per second (i.e., a period of 100 ms per pulse cycle). For example, the digit 5 corresponds to five make/break cycles lasting a total of 500ms. The switching machine senses and counts the number make/break pairs in the sequence.

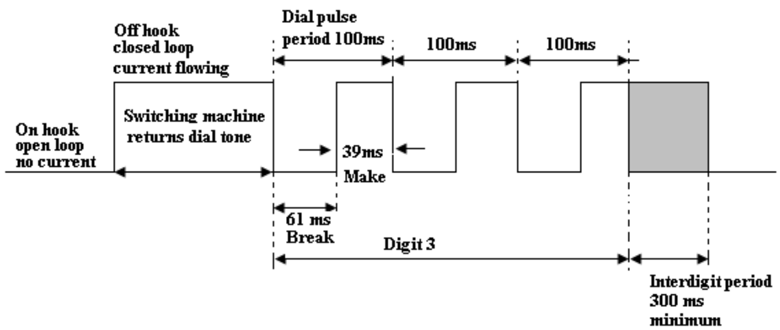


Fig. Dial Pulsing Sequence

The break time is nominally 61 ms, and the make time is nominally 39ms. Digits are separated by idle period of 300 ms called the inter digit time. It is essential that the switching machine recognize the inter digit time so that it can separate the pulses from successive digits. The central office switch incorporates a special time-out circuit to ensure that the break part of dialing pulse is not misinterpreted as the phone being returned to its on-hook (idle) condition.

Q.50. How numbering plan is achieved in modern telephony? Give the structure with example. **(8)**

Ans:

The objective of numbering plan is to uniquely identify every subscriber connected to a telecommunication network. A numbering plan may be open, semi-open or closed. An open-numbering plan permits wide variation in the number of digits to be used to identify a subscriber within a multi exchange area or within a country. This plan is used in countries equipped extensively with non-Director Strowger switching systems. In such cases the numbering scheme is usually an exact image of the network structure changes. A semi-open plan permits number lengths to differ by almost one or two digits. Today, this scheme is the most common and is used in many countries including India. In closed numbering plan or the Uniform numbering scheme, the number of digits in a subscriber number is fixed. An international numbering plan or world numbering plan has been defined by CCITT in its recommendations E.160-E.163. For numbering purposes, the world is divided into zones as shown in Fig. Each zone is given a single digit code. For the European zone two codes have been allotted because of the large number of countries within this zone. Every international telephone number consists of two parts as shown in Fig. The country code contains one, two or three digits, the first digit being the zone code in which the country lies.. In cases where an integrated numbering plan already covers an entire zone, the countries in tha zone are identified by the single digit zone code itself.

The existence of world numbering plan places restriction on the national numbering plan of each country. The number of digits in an international subscriber number is limited to an absolute maximum of 12. In practical, with a few exceptions, world numbers are limited to 11 digits. As a result, the number of digits available for a national numbering plan is 11-N, where N is the number of digits in the country code.

In general, a national number consist of three parts as shown in Fig. The area or the trunk code identify a particular numbering area or the multi exchange area of the called subscriber, and thus determine the routing for a trunk call and a charge for it. According to CCITT international usage, a numbering area is identified as that area in which any two subscriber use identical dialing procedure to reach any other subscriber in the network. An exchange code identifies a particular exchange within a numbering area. It determine the routing for incoming trunk call from another numbering area or for a call originating from one exchange and destined to another in the same numbering area. Subscriber line number is used to select the called subscriber line at the terminating exchange. In CCITT terminology, the combination of the exchange code and the subscriber line number is known as the subscriber number which is the number listed in the telephone directory.

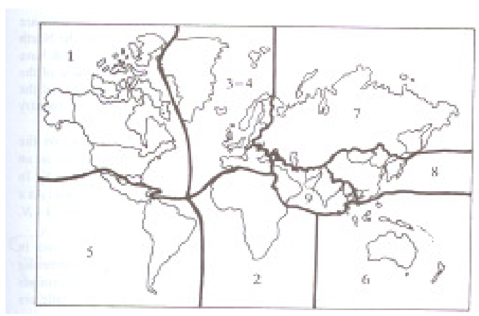


FIG – World Numbering Zones

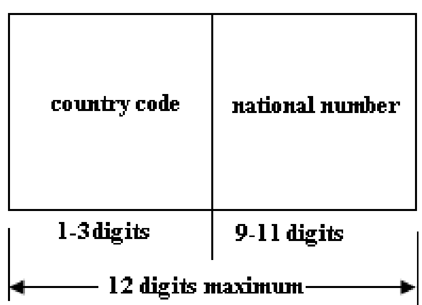


FIG – International Telephone Number

FIG – National Telephone Number

Q.51. What are the different ways of designing 100 line exchange using uni selector a two motion selectors? Show at least three variations. Which is the best option? **(8)**

Ans:
Here three different designing methods for 100 line exchange are discussed:
Design 1: Here**,** Strowger switching system is designed using one two-motion selector for each subscriber. A subscriber is assigned a number in the range 00-99, and the same number is used to identify the two-motion selector assigned to him. The 100 outlets of each two-motion selector are numbered as per the scheme given in table. The corresponding outlets in all the 100 two-motion selectors are commoned and folded back to the corresponding inlets. For example, a subscriber with 87 as his number is assigned the two-motion selector 87. The outlet 87 which corresponds to this subscriber is connected to the 7th contact in the 8th vertical position of all the two-motion selectors and folded back to his inlet. The arrangement is shown in Fig. If subscriber 23 dials 87, his two motion selector 23 would step vertically 8 times corresponding to the first digit and would step horizontally 7 times to reach the contact to which the subscriber 87 is connected. This switch is non- blocking and uses only one stage of switching elements. Since the two-motion selector is activated by the calling party, the call is terminated only when the calling party disconnects the line. If a two-motion selector goes out of order, the subscriber connected to it will not be able to make any outgoing calls but can receive incoming calls. The design parameters of this switch are:

$$S = 100,\ SC = 50,\ K = 1,\ TC = 1,$$
$$EUF = 0.5,\ C = 200,\ CCI = 25,\ P_B = 0.$$

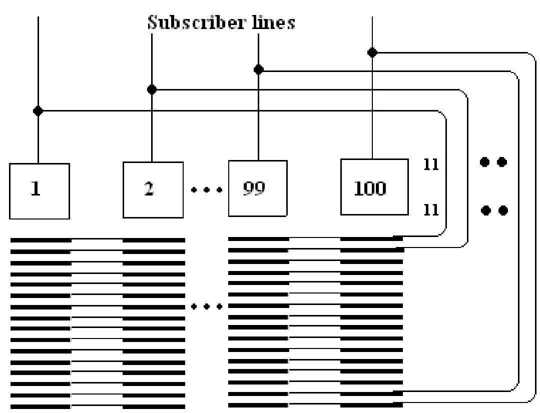


FIG - 100-Line exchange with one two-motion selector per subscriber

Design 2: Instead of 100 two-motion selectors as in the case of Design 3, let us consider only 24 two-motion selectors. In the case, 24 simultaneous calls can be put through the switch. The 24 two-motion selectors are shared by all the hundred users. The corresponding outlets of the two-motion selectors are commoned as in the previous case. It is implicitly assumed here that the average peak-hour traffic is 24 simultaneous calls. Typically, a 24-outlet uniselector is used as a selector hunter. Each of the 24 outlets is connected to one two-motion selector. Thus, a subscriber has access to all the 24 two-motion selectors in the system. The corresponding outlets of all the selector hunters are commoned and thus, all subscribers have access to all the two-motion selectors. This scheme is shown in Fig. The call establishment in this scheme takes place in two steps. Firstly, when the subscriber lifts his receiver handset, his uni selector hunts through the contact positions and seizes a free two-motion selector. At the next step, the two-motion selector responds to the dial pulses for appropriate connection. The design parameters of this system are:

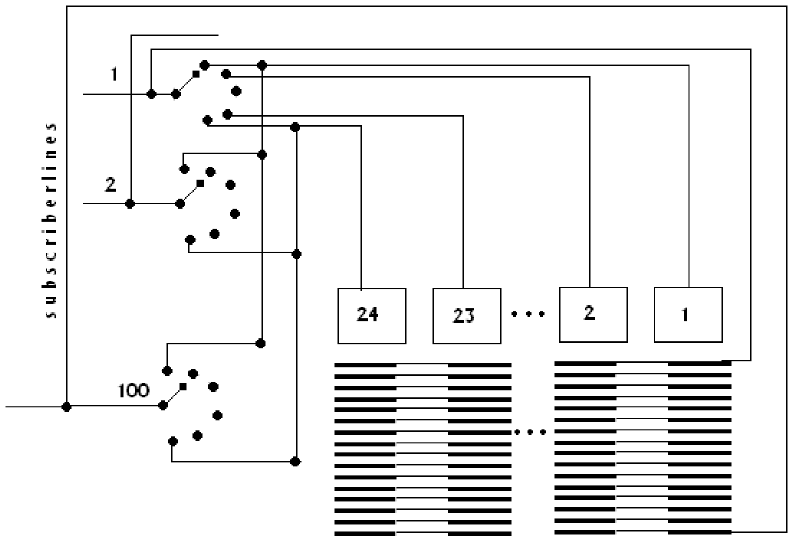


FIG – 100-Line exchange with selector finder

Design 3: In this design**,** there are 24 line finders. If any of the 100 subscribers has to get access to any of the 24 two-motion selectors, it is essential that every line finder is capable of reaching any of the 100 subscribers. In other words, each line finder must have 100 outlets. For this purpose, two motion selectors have to be used as line finders. The configuration is shown in Fig. The corresponding outlets of the line finders are commoned. Similarly, the outlets of the numerical selectors are also commoned.

When the start condition is received, the line finder is caused to hunt vertically until the wipers reach a marked level. The vertical hunting is then stopped and the horizontal hunt commences to find a particular marked contact in that level. It may be noted that in the extreme case, the line finder may have to take 20-steps -10 vertical and 10 horizontal –before a line is found. The line finders are made to step automatically, using interrupter contact mechanism. When the line finder locates the subscriber line, the start condition is removed, the allotter switch steps on to the next free line finder in readiness for further calls, and the numerical selector sends out the dial tone to the subscriber in readiness to receive dialing pulses. Thereafter the establishment of the connection proceeds in the usual manner.

Obviously, Design 3 is by far the best for a 100-line exchange. If we had used uni selectors as line finders, it would have been necessary to divide subscriber's line into small groups of, say 24 each. Such designs involving groupings, function efficiently only under certain specific traffic conditions and generally lead to high blocking probabilities.

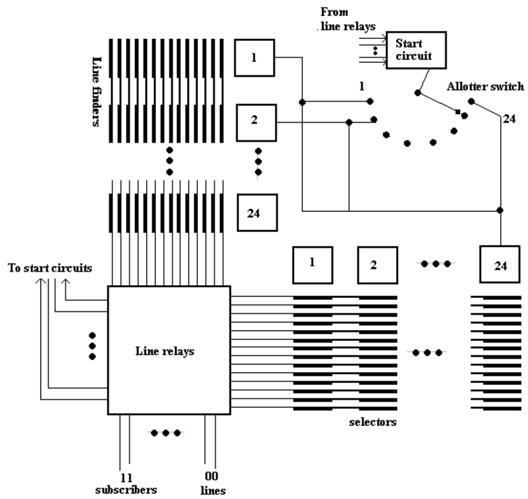


FIG – 100-Line exchange with two-motion line finders

Q.52. What are transmission bridges? How do they help in satisfying the connectivity? **(8)**

Ans:
A typical transmission bridge is shown in Fig. The series capacitance and the shunt inductances of the two relays provide a high-pass filter to transmit the AC speech signals, while the relays respond independently to the DC loop/disconnect signals from the calling and called customer.

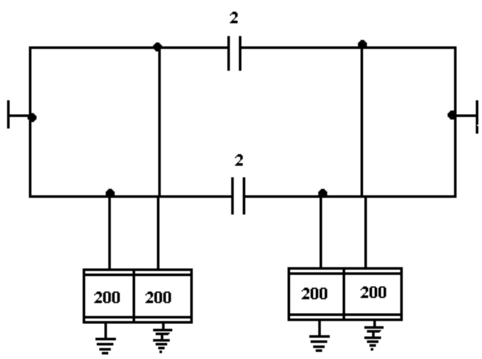


FIG – Transmission Bridge

Q.53. Explain FDM and show how CCITT standards help in building the base band? **(8)**

Ans:
Frequency Division Multiplexing: It is the process of combining several information channels by shifting their signals to different frequency groups within the spectrum so that they can be transmitted simultaneously over common transmission facility.
The bandwidth of a telephone speech signal is rather less than 4 kHz. Where as the available bandwidth on unloaded cables pair, is well above 100 kHz. It is therefore possible, using modulation techniques, to divide up the cable bandwidth so that a

number of telephone speech paths can be carried simultaneously along a single cable pair. The normal arrangement consists of 24 telephone channels per cable pair, the modulation into 24 channels being carried out in the two stages. In the first stage, 12 channels are multiplexed together to form what is commonly known as a basic group. The basic group arrangement is shown in Fig. Each of the 12 telephone signals are single side-band amplitude modulated on to carriers spaced at 4 kHz intervals from 64 kHz to 108 kHz. The lower side-band (LSB) is used in each case. The 12 base-band signals are therefore translated into the frequency band from 60 kHz 108 kHz as shown. The block diagram for the channel translating equipment is given in Fig. To form a 24-channel system, to basic groups are taken together. One basic group (B) is transmitted directly as it stands. The other basic group (A) is amplitude modulated on to a carrier are 120 kHz and the lower side-band is taken so as to occupy the frequency range from 12 kHz to 60 kHz as shown in Fig. by using two stages of modulation for basic group A, it is possible to reduce the physical size of the component required for the LSB filters of the Fig., since the lowest cut- off frequency required is at 64 kHz rather than 12 kHz if the whole block of 24 channels were assembled together in one stage of modulation.

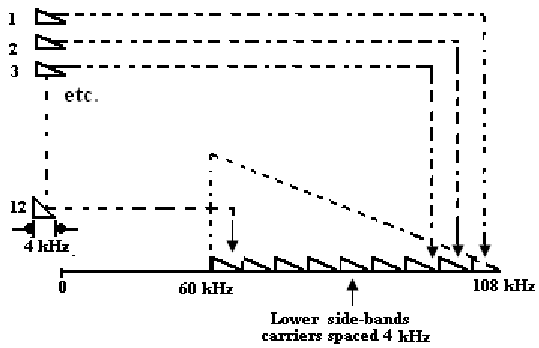


FIG – Basic Group Arrangement

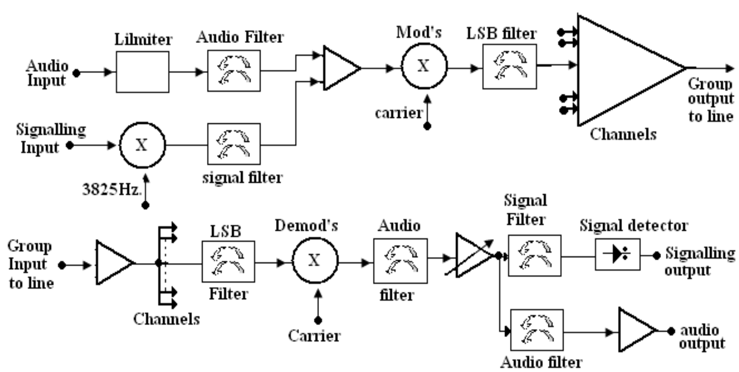


FIG – Channel Translating Equipment

Q.54. Draw a centralized SPC organization and explain how it works under load sharing operation. **(8)**

Ans:

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 ma... other ancillary tasks. A typical control configuration of an ESS using centralized SPC is shown in Fig. A centralized SPC configuration may use more than one processor for redundancy purposes. 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:

(i) Standby mode
(ii) Synchronous duplex mode
(iii) Load Sharing mode

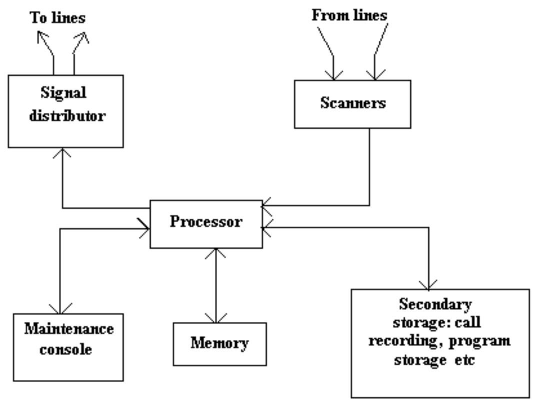


FIG - Typically Centralized SPC Organization

In load sharing operation, an 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.

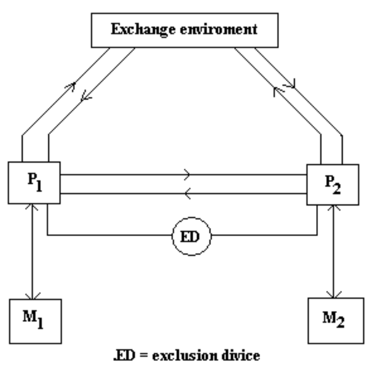


FIG - Load sharing Processor Configuration

Q.55. Explain the process of inter-register signalling. **(8)**

Ans:
Registers are used in common control exchanges to store and analyze routing data. They are provided on a common basis is a single register provides routing data for a number of speech circuits. Once a call has been setup, the register is then made

available to set up other calls. Inter register refers to signaling between t registers of different exchanging.

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

1. Subscriber loops signaling.
2. Intra exchange or register signaling
3. Interexchange or inter register signaling

In a telephone network, subscriber loop signaling depends upon the type of a telephone instrument used. The inter exchange signaling is internal to the switching system and is heavily dependent upon the type and design of a switching system. It varies from one model to another even with the same manufacturer. This signaling does not involve signaling system of the type required on the switching network. When interexchange signaling takes place between exchanges with common control subsystems, it is called Inter register signaling. The main purpose of Inter register signaling is the exchange of address digits which pass from exchange to exchange on a link by link basis. Network wide signaling also involves end to end signaling between the originating exchange and the terminating exchange. Such a form of signaling is called line signaling. CCS does not use the speech or the data path for signaling. It uses a separate common channel for passing control signals for a group of trunks or information paths.

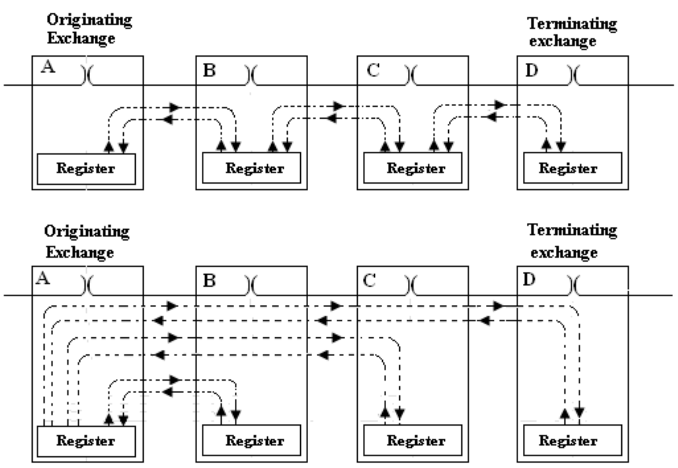


FIG – End-to-end Signaling

For a multi link connection in a network of register- controlled exchanges, a register in the originating exchange receives address information from the calling customer and sends out routing digits. Each succeeding register both receives and sends out routing digits, until the terminating exchange is reached. This sequence of operation introduces post-dialing delay. An inter-register signaling system cannot be used for seize, answer and clear signals. No register is connected when an incoming seize signal is received, since it is the signal which initiate a connection to a register. The register is released after it has set up a connection through its exchange and sent out routing digits; therefore, it cannot receive answer and clear signals. Either in-bloc or overlap signaling may be used. In en-bloc signaling, the complete address information is transferred from one register to the next as signal string of digit. Thus, no signal is sent out until the complete address information has been received. In overlap signaling, digits are sent out as soon as possible. Thus, some digits may be sent before the complete address has been received and signaling may take place before the complete address has been received and signaling may take place simultaneously on two links (i.e. the signal overlap). The enables subsequent registers to start digit to analyses earlier than is possible with en-bloc signaling and this reduces post-dialing delay. Either link-by-link signaling or end-to-end signaling may be employed. In link-by-link signaling, information exchange only between adjacent registers in a multi link connection as shown by Fig. In end-to-end signaling, the originating register controls the setting up of a connection until it reaches its final destination, as shown in Fig. Each transit register receives only the address information required to select the outgoing route to the next exchange in the connection. Having performed its task, it is released and the originating register signals to the next register.

Q.56. What do you mean by numbering and addressing? Draw the ISDN address structure and explain how the addressing works? **(8)**

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

1. It is based on, and is an enhancement of, the telephone numbering plan. It is independent of the nature of the services (e.g. voice, facsimile or data) or the performance characteristics of the connection (e.g. 32 kbps voice or 64 kbps voice).
2. It is independent of routing, i.e. the numbering or addressing does not specify the intermediate exchanges through which the services is to be put through.
3. It is a sequence of decimal digits. No alphabet or other characters are permitted as part of the address.
4. Its design is such that interworking between ISDNs requires only the use of ISDN number and no other additional digits or addressing signals.

Address Structure: The ISDN address structure is shown in Fig. ISDN number part has a maximum of 15 digits and the ISDN sub address part a maximum of 40 digits. National destination code is like an area code in telephony network and is of variable length. ISDN subscriber number is the one normally listed in the directories. It is the number to be dialed to reach a subscriber in the same numbering area. An ISDN number is a unique worldwide address and unambiguously identifies an end point connection. This end point may be:

1. single S or T reference point,

2. one to many T reference points at the same site, and
3. one of many S reference points using direct inward dialing feature.

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

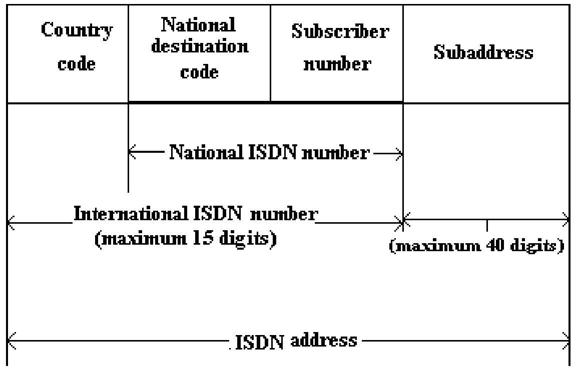


FIG – ISDN Address Structure

A sub address, although a part of the ISDN address, is not considered as an integral part of the numbering scheme. The sub address is carried in a separate field in the user-network interface message. It may or may not be present in a call setup message. The sub address is generally transparent to the network and it the equipment at the destination which analyses the sub address information for routing to the appropriate terminal. A typical address using both ISDN number and the sub address is shown in Fig. Here, an ISDN number identifies a T reference point and a sub address one of the many S reference points. Alternatively, as reference point may also be addressed by using direct inward dialing (DID) feature.

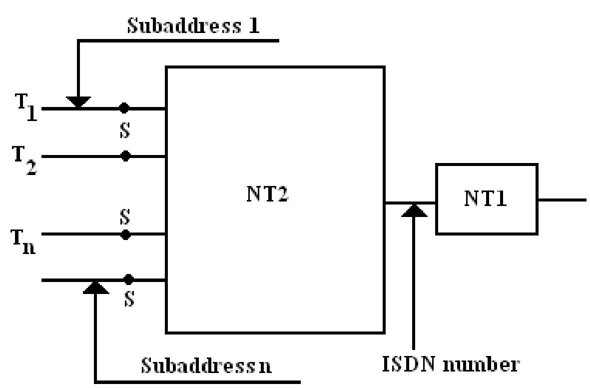


Fig – Example of ISDN Addressing.

Q.57. What are concentrators? Explain how it helps in connecting number of subscribers. **(8)**

Ans:

In rural areas, subscribers are generally dispersed. It is both unnecessary and expensive to provide a dedicated pair for every subscriber. Three techniques are used to gain on the number pairs:

1. Parity lines

2. Concentrators
3. Carrier Systems.

In the first technique, two or more subscribers are connected to the one line which is termed parity line. This scheme is not used commonly as it has a number of drawbacks. Only one subscriber at a time can use the line. Selective ringing is difficult and privacy is not maintained. Dialing between two subscribers on the same line is not possible.

In the second technique, a concentrator expander (CE) is used near the cluster of users and another one at the exchange end as shown in Fig.

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

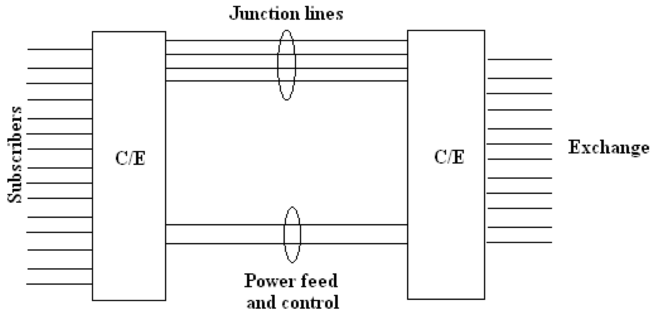


FIG – Concentrator-expander connection for dispersed subscribers.

Q.58. Give the operation of different topologies used in local area network bring out their advantages and disadvantages. **(8)**

Ans:

LAN topologies: Network topology is a physical schematic which shows interconnection of the many users. There are four basic topologies as under:

(i) Direct Connection or one to all topology
(ii) Star topology
(iii) Bus Topology
(iv) Ring topology

Direct connection or one to one topology: In the one to all topology, there is a path between every node and every other node. The number of paths required is defined by the equation $P = (n^2-n)/2$, where n is number of nodes and P, total number of paths. Thus number of paths increases considerably as the number of nodes increases. Each node must be physically able to connect to paths to every other node so a very large connector and associated interface support circuitry is needed at each node. Adding a new node requires major rewiring and affects the software that manages communications for the system.

Star Topology: In Star topology, all user nodes are connected to central node point that interconnects all the individual users links and nodes. Data flows from one node, through the star centre to the desired receiving node. The Central node is like a large switch which routes the data from the input line to the output line. Advanced central nodes can have multiple switching paths, so several paths exist at the same time. This topology is used in a telephone system central office. Addition of a new user node requires running only a single link from the star centre to the user without disturbing any other node.

Bus Topology: This topology shares a single link or path way among all users. The common single path way is known as bus. In this topology, the link serves as a high way for all data signals, and users connect on to the bus at their node location. In bus configurations, network control is not centralized to a particular node. Here control is distributed among all nodes connected to the LAN. Data transmission on a bus network is usually in the form of small packets containing user addresses and data. When one node/user desires to transmit data to another station, it monitor the bus to determine if it is currently being used. If no other nodes/users are communicating over the network, the monitoring node/user can start to transmit its data. Each node must monitor all transmission on the network and determine which are intended for them.

Ring Topology: In ring topology, all user nodes are connected with the physical path acting as links of a chain and the last user node is connected back to the first node. A signal going on to the next node must be processed by the first node, which then passes it through to the next node. Adding a new user requires breaking the ring temporarily, inserting the new node and then reestablishing the complete ring path. Rings also have simpler protocols than buses. In bus configurations, each node must monitor the bus constantly to see if there are any messages for it or to see if the bus is clear before transmitting. The ring uses a simpler scheme.

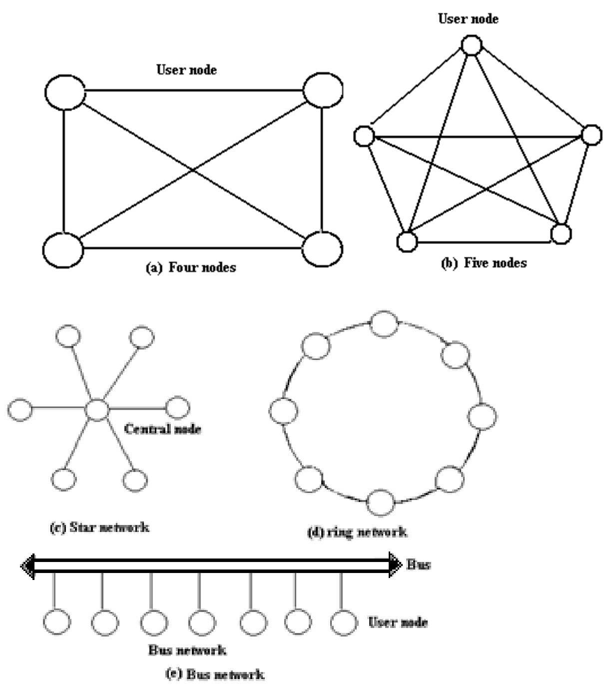


FIG - Networks Topologies

Q.59 Write short note on Quantization.

Ans:

Quantization: This is the first step in PCM. The total amplitude range of the modulating signal is divided into a number of standard levels known as quantization levels at equal intervals as shown in Fig. These levels are then transmitted in binary code. Hence, the actual number of these standard levels is a power of two such as 4, 8,16,32,64,128 etc. For the sake of simplicity, Fig shows only eight levels, however practical systems use 128 levels or even higher. A new signal is generated by producing, for each sample, a voltage level corresponding to the mid-point level of the standard level in which the sample falls. Thus if a range of 0-4 V were divided into four 1V standard levels and the signal was sampled when it was 2.8V , the quantizer will output a voltage of 2.5 V, and hold that level until the next sampling time 2.5 V corresponds to the midpoint of the third standard level. This result in a stepped wave form which follows the contour of the modulating signal. This discritisation of the modulating signal is known as Quantization. Fig. 11.10 shows the modulating signal m(t) and quantized signal ($m_q(t)$) when eight standard levels are used. The quantized wave forms are an approximation to the original signal. The difference between the two wave forms amounts to noise added to the signal by the quantization process. The error introduced so is known as quantizaton error.

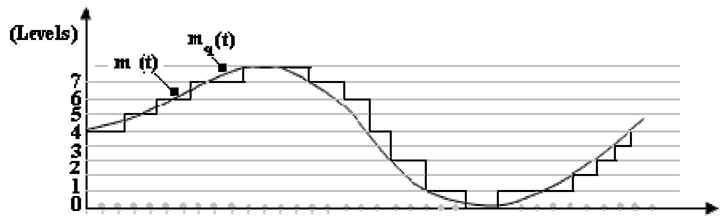


FIG - Modulating Signal and Quantized Signal

Q.60. Describe the nature of signals produced on the subscriber's loop by a pulse dialer, and a touch tone dialer. **(8)**

Ans:

DTMF is a simple two-of –eight encoding scheme where each digit is represented by the linear addition of two frequencies. DTMF is strictly for signaling between a subscriber's location and the nearest telephone office or message switching center. DTMF is sometimes confused with another two-tone signaling system called multi frequency signaling (Mf), which is a two-of-six code designed to be used only to convey information between two electronic switching machines. Fig. shows the four-row-by-four column keypad matrix used with a DTMF keypad. As the figure shows, the keypad is comprised of 16 keys and eight frequencies. Most household telephones, however, are not equipped with the special-purpose keys located in the fourth column (i.e. , the A,B,C, and D keys). Therefore, most household telephones actually use two-of-seven tone encoding scheme. The four vertical frequencies (called the low group frequencies) are 697 Hz, 852 Hz, and 941 Hz, and the four horizontal frequencies (called the high group frequencies) are 1209 Hz, 1336 Hz, 1477 Hz, and 1633 Hz. The frequency tolerance of the oscillators is $\pm$ .5%. As shown in Fig., the digits 2 through 9 can also be used to represent 24 of the 26 letters (Q and Z are omitted). The letters were origina[lly] used to identify one local telephone exchange from another.

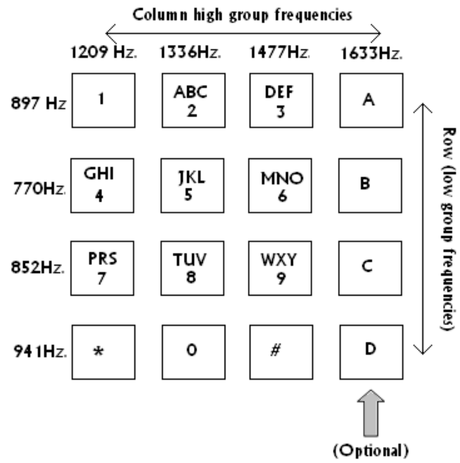


FIG - DTMF Keypad Layout

Dial Pulses: Dial pulsing (some times called rotary dial pulsing) is the method originally used to transfer digits from a telephone set to the local switch. Pulsing digits from a rotary switch began soon after the invention of the automatic switching machines. The concept of dial pulsing is quite simple and is depicted in Fig. The process begins when the telephone set is lifted off hook, completing a path for current through the local loop. When the switching machine detects the off-hook condition, it responds with dial tone. After hearing the dial tone, the subscriber begins the dial pulsing digits by rotating a mechanical dialing mechanism and then letting it returns to its rest position. As the rotary switch returns to its rest position, it outputs a series of dial pulses corresponding to digit dialed. When a digit is dialed, the loop circuit alternately opens (breaks) and closes (makes) a prescribed number of times. The number of switch make/break sequences corresponds to the digit dialed (i.e., the digit 3 produces three switch openings and three switch closers). Dial pulses occur at 10 make/break cycles per second (i.e., a period of 100 ms per pulse cycle). For example, the digit 5 corresponds to five make/break cycles lasting a total of 500ms. The switching machine senses and counts the number make/break pairs in the sequence.

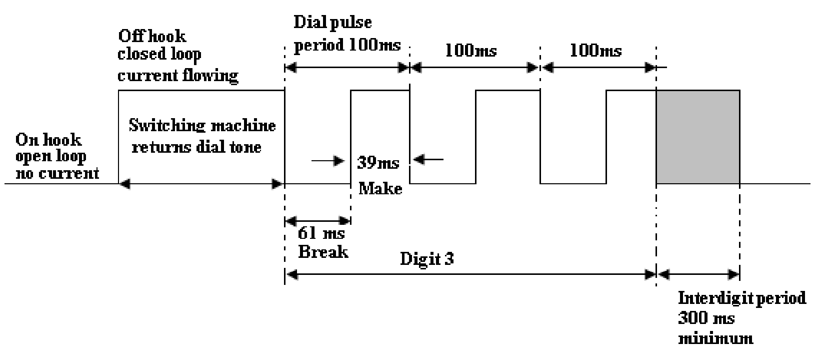


Fig. Dial Pulsing Sequence

The break time is nominally 61 ms, and the make time is nominally 39ms. Digits are separated by idle period of 300 ms called the inter digit time. It is essential that the switching machine recognize the inter digit time so that it can separate the pulses from successive digits. The central office switch incorporates a special time-out circuit to ensure that the break part of dialing pulse is not misinterpreted as the phone being returned to its on-hook (idle) condition.

Q.61. Draw a 100 line exchange using two motion selectors and explain, how switching takes place in it. **(8)**

Ans:
In a 100 line exchange, each subscriber is assigned a 2 digit telephone number between 00 and 99. This number identifies the group selector value and the final selector values. As the dialing of the telephone number is done the corresponding group selector and then the final selector levels are switched.The desired schematic for 100 lines exchange with 24 two motion selectors is shown in Fig.

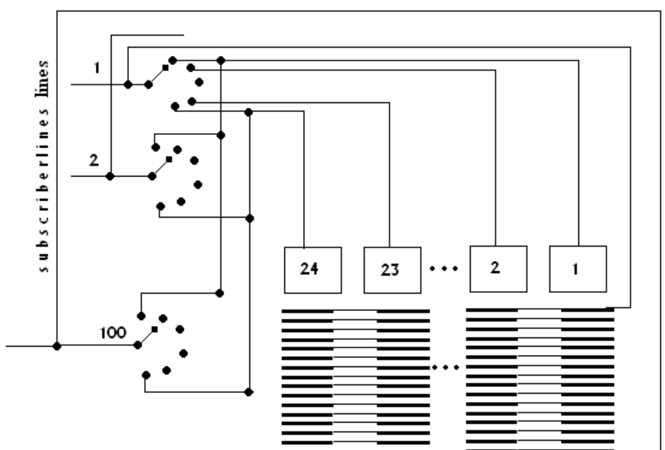


FIG – 100 Lines – Exchange

In the case, 24 simultaneous calls can be put through the switch. Typically, a 24-outlet uniselector is used as a selector hunter. Each of the 24 outlets is connected to one two-motion selector. Thus, a subscriber has access to all the 24 two-motion selectors in the system. The corresponding outlets of all the selector hunters are commoned and thus, all subscribers have access to all the two-motion selectors. This scheme is shown in Fig. The call establishment in this scheme takes place in two steps. Firstly, when the subscriber lifts his receiver handset, his uniselector hunts through the contact positions and seizes a free two-motion selector. At the next step, the two-motion selector responds to the dial pulses for appropriate connection.

Q.62. How time slot interchange switch works in time multiplexed time switching, explain using schematic. **(8)**

Ans:
The switches for which the inlets and outlets are trunks which carry time division multiplexed data streams. Such switches are called time multiplexed switches.

A time multiplexed time division space switch is shown in Fig... There are incoming trunks and N outgoing trunks, each carrying a time division multiplexed stream of M samples per frame. Each frame is of 125-µs time duration. In one frame time, a total of MN speech samples have to be switched. One sample duration, 125/M microseconds, is usually referred to as a time slot. In one time slot, N samples are switched. Fig shows an output-controlled switch. The output is cyclically scanned. There is a 1-to-M relationship between the outlets and the control memory locations, i.e. there are M locations in the control memory corresponding to each outlet.

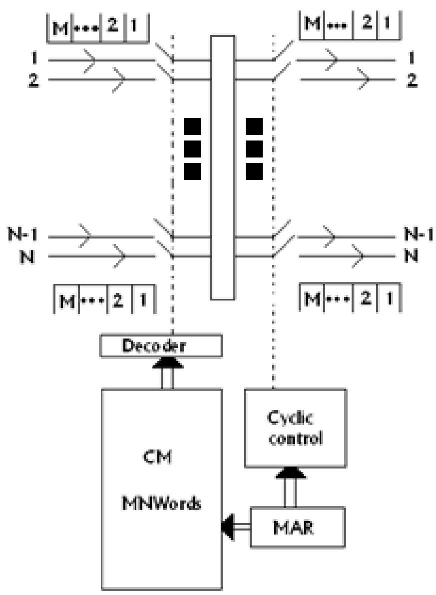


FIG – Time Multiplexing Space Switch

The control memory has MN words. If we view the control memory as M blocks of N words each, a location address may be specified in a two dimensional form,(i,j),where i is the block address and j is the word within the block. We have $1 \le i \le M$ and $1 \le j \le N$. The block address i corresponds to the time slot i and the word address j to the outlet j. The first N locations of the control memory correspond to the first time slot, the next N locations, i.e. locations N + 1 to 2N when addressed linearly, or locations (2,1) to (2,N) when addressed in a two dimensional form, correspond to the time slot 2 and so on. Therefore, if the location (i,j) contains an inlet address k, it implies that inlet k is connected to the outlet j during the time slot i. The number of trunks that can be supported on this switch is given by

$$N = \frac{125}{Mt^2}$$

Where t is the switching time including memory access time per inlet-outlet pair.

Q.63. What are different control function categories, explain how they help in signalling and control. **(8)**

Ans:

In some switching systems, the control subsystem may be an integral part of the switching network itself. Such system are known as **direct control** switching systems. Those systems in which the control subsystem is outside the switching network are known as **common control** switching system. Strowger exchanges are usually direct control systems, whereas crossbar and electronic exchanges are common control system. All stored program control systems are common control systems. Common control is also known as indirect control or register control.

Common Control Switching System: A functional block diagram of a common control switching system is shown in Fig. The control functions in a switching system may be placed under four broad categories:

(a) Event monitoring.
(b) Call processing.
(c) Charging.
(d) Operation and maintenance

Events occurring outside the exchange at the line units, trunk junctions and inter exchange signaling receiver/sender units are all monitored by control subsystem. Typical events include all requests and call release signals at the line units. The occurrences of the events are signaled by operating relays which initiate control action. The control subsystem may operate relays in the junctions, receivers/senders and the line units, and thus command these units to perform certain functions. Events monitoring may be distributed. For examples, 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 market for dial tone, and the register finder is activated to seize a free register. Identity of the calling line is used to determine the line category and the class of service to which the subscriber belongs. A register appropriate to the line category is chosen, which then sends out the dial tone to the subscriber, in readiness to receive the dialing information. As soon as the initial digits (usually 2-5) which identify the exchange are received in the register, the register continues to receive the remaining digits. The initial translator determines the route for the call through the network and decides whether a call should be put through or not. It also determines the charging methods and the rates applicable to the subscriber. Initial translation may also take into account instructions from the operating personnel and information regarding the status of the network.

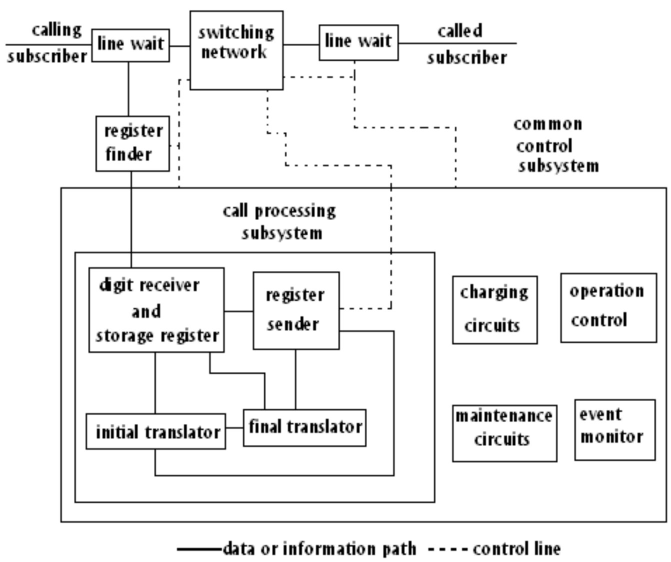


FIG - Common Control Switching System

If a call is destined to a number in an exchange other than the present o processing the digits, the initial translator generates the require routing digits and passes them on to the register sender. Here the digits corresponding to the subscriber identification are concatenated and the combined digit pattern is transmitted over the trunks to the external exchange. Register sender uses appropriate signaling technique, depending on the requirements of the destination exchange. If the call is destined to a subscriber within the same exchange, the digits are processed by the final translator. The translation of directory number to equipment number takes place at this stage. The final translator determines the line unit to which a call must be connected and the category of the called line. The category information may influence charging and connection establishment. In some practical implementations, both initial and final code translator functions are performed by a single translator. Controlling the operation of the switching network is an important function of the common control subsystem. This is done by marking the switching elements at different stages in accordance with a set of binary data defining the path and then commanding the actual connection of the path. Path finding may be carried out at the level of the common control unit or the switching network.

Q.64. Explain the following: **(8)**

(i) Busy Hour
(ii) Peak Busy Hour
(iv) Time consistent Busy Hour
(v) Traffic intensity

Ans:
(i) Busy Hour: Continuous 1-hour period lying wholly in the time interval concerned for which the traffic volume or the number of calls attempts is greatest.
(ii) Peak Busy Hour: The busy hour each day; it usually varies from day to day, or even a number of days.
(iii) Time consistent Busy Hour: The 1-hour period starting at the same time each day for which the average traffic volume or the number of call attempts is greatest over the days under consideration.
(iv) Traffic intensity: The traffic on the network may then be measured in terms of the occupancy of the servers in the network. Such a measure is called the traffic intensity which is defined as:

$$A_0 = \frac{\text{Period for which a server is occupied}}{\text{Total period of observation}}$$

Q.65. Explain the principles of operation of centralized SPC and distributed SPC and compare their performance. **(8)**

Ans:
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. A typical control configuration of an ESS using centralized SPC is shown in Fig. A centralized SPC configuration may use more than one processor for redundancy purposes there are two approaches to organizing stored program control:

Centralized: In this control, all the control equipment is replaced by a sin processor which must be quite powerful.

Distributed: In this control, the control functions are shared by many processors within the exchange itself.

In centralized SPC, dual processor architecture may be configured to operate in one of three modes:

Standby mode: In this mode, one processor is active and the other is on standby, both hardware and software wise. The standby processor brought online when active processor fails. An important requirement of this configuration is the ability of the standby processor to reconstitute the state of the exchange system when it takes over the control.

Synchronous duplex mode: In synchronous duplex mode, hardware coupling is provided between the two processors which execute the same set of instructions and compare the results continuously. If a mismatch occurs, the faculty processor is identified and taken out of service immediately. When the system is operating normally, the two processors have the same data in their memories at all the times and receive all information from the exchange environment.

Load Sharing mode; In load sharing operation, an 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.

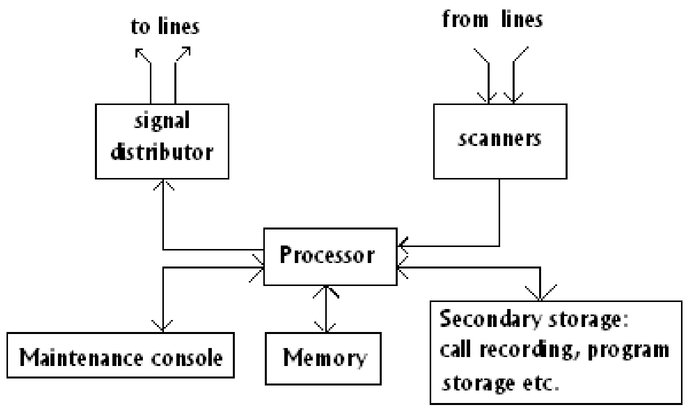


FIG - Typically Centralized SPC Organization .

Q.66. Draw the schematic of a CCS and explain, giving its advantages. **(8)**

Ans:

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

1. Subscriber loops signaling.
2. Intra exchange or register signaling
3. Interexchange or inter register signaling

In a telephone network, subscriber loop signaling depends upon the type of a telephone instrument used. The intra exchange signaling is internal to the switching system and is heavily dependent upon the type and design of a switching system. It varies from one model to another even with the same manufacturer. This signaling does not involve signaling system of the type required on the switching network. When interexchange signaling takes place between exchanges with common control subsystems, it is called Inter register signaling. The main purpose of Inter register signaling is the exchange of address digits which pass from exchange to exchange on a link by link basis. Network wide signaling also involves end to end signaling between the originating exchange and the terminating exchange. Such a form of signaling is called line signaling. CCS does not use the speech or the data path for signaling. It uses a separate common channel for passing control signals for a group of trunks or information paths.

It gives the following advantages:

1. Information can be exchange between the processors much more rapidly than when channel associated signaling is used.
2. As a result, a much wider repertoire of signals can be used and this enables more services to be provided to customers.
3. Signals can be added or changed by software modification to provide new services.
4. There is no longer any need for line signaling equipments on every junction which results in a considerable cost saving.
5. Since there is no line signaling, the junctions can be used for calls from B to A in addition to calls from A to B. Both way working requires fewer circuits to carry the traffic than if separate groups of junctions are provided from A to B and from B to A.
6. Signals relating to a call can be sent while the call is in progress. This enables customers to alter connections after they have been set up. For example a customer can transfer a call elsewhere, or request a third party to be connected in to an existing connection.
7. Signals can be exchanged between processors for functions other than call processing, for example for maintenance or network management purposes.

Q.67. What is the difference between message switching and packet switching, explain typical packet switching network configuration. **(8)**

Ans.
The main difference between message switching and packet switching is given below:

Message Switching	**Packet Switching**
Transmission of messages	Transmission of packets
Message Stored	Message held for short time
Route established for each message	Route established for each packet
Blocking cannot occur	Blocking cannot occur
Network responsible for lost messages	Network may be responsible for each packet but not for entire message.

Packet Switching: In packet switching the nodes handle much smaller data length than are found in message switching. The message is divided before transmission into a series of sections of data called data packets having a length of few thousand bits. This technique has a number of advantages. First, the short packets experience minimum delay in progress through the network. The method used is still a store and forward process but since the packets are small they are quickly copied by each node and require little memory space. Second by appending a sequence number to each packet as well as its destination address the nodes are able to interleave packets from several different sources and this leads to more efficient use of the transmission media. Fig. shows how this interleaving can function.

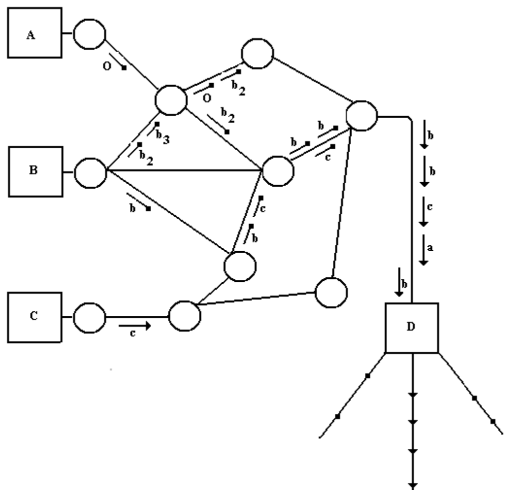


Fig. Interleaving of packets in network

Two approaches are applied to the way in which this stream of mixed packets is handled by nodes. These concepts are known as datagram and the virtual circuit.

Q.68. Explain CSMA/CD and CSMA/CA protocols used in LAN's, discuss its advantages and limitations. **(8)**

Ans:

CSMA/CD: It is an access method used primarily with LANs configured in a bus topology. With CSMA/CD, any station (node) can send a message to any other station (or stations) as long as the transmission medium is free of transmissions from other stations. Stations monitor (listen to) the line to determine if the line is busy. If the station has a message to transmit but the line is busy, it waits for an idle condition before transmitting its message. If two stations transmit at the same time, a collision occurs. When this happens, the station first sensing the collision sends a special jamming signal to all other stations on the network. All stations then cease transmitting (back off) and wait a random period of time before attempting a retransmission. The random delay time for each station is different, and therefore, allows for prioritizing the stations on the network. If successive collisions occur, the back off period for each station is doubled. With CSMS/CD stations must contend for the network. A station is not guaranteed access to the network. To detect the occurrence of a collision, a station must be capable of transmitting and receiving simultaneously. CSMA/CD is used by most LANs configured in a bus topology. Ethernet is an example of a LAN that uses CSMA/CD and is described later in this chapter. Another factor that could possibly cause collision with CSMA/CD is propagation delay. Propagation delay is the time it takes a signal to travel from a source to destination. Because of propagation delay, it is possible for the line to appear idle when, in fact, another station is transmitting a signal that has not yet reached the monitoring station.

CSMA/CA: It belongs to a class of protocols called multiple access metho CSMA/CA stands for: Carrier Sense Multiple Access with Collision Avoidance. In CSMA, a station wishing to transmit has to first listen to the channel for a predetermined amount of time so as to check for any activity on the channel. If the channel is sensed "idle" then the station is permitted to transmit. If the channel is sensed as "busy" the station has to defer its transmission. This is the essence of both CSMA/CA and CSMA/CD. In CSMA/CA (Local Talk), once the channel is clear, a station sends a signal telling all other stations not to transmit, and then sends its packet. In Ethernet 802.3, the station continues to wait for a time, and checks to see if the channel is still free. If it is free, the station transmits, and waits for an acknowledgment signal that the packet was received

Q.69. Describe the architecture of SS7 common channel signaling network with the help of a neat labeled diagram. **(8)**

Ans:
A block schematic diagram of the CCITT no. 7 signalling system is shown in fig. Signal messages are passed from the central processor of the sending exchange to the CCS system. This consists of the microprocessor based subsystem.
The signaling control subsystems, the signaling termination subsystem and the error control subsystem. The signaling control subsystem structures the messages in the appropriate format and queues them for transmission. When there are no messages to send, it generates filler messages to keep the link active. Messages then passed to the signaling termination sub system, where complete signal units (SU) are assembled using sequence numbers and check bits generated by the error control subsystem. At the receiving terminal, the reverse sequence is carried out. The levels are as follows:

Level 1: The Physical Layer
Level 2: The Data Link Level
Level 3: The signaling network level
Level 4: The User Part

The relationship between these levels and the layers of the OSI model is shown in Fig. The user part encompasses layers 4 to 7 of the OSI model.
Level 1 is the means of sending bit streams over a physical path. It uses times lot 16 of a 2 Mbit/s PCM system or times slot 24 of a 1.5 M bit/s system.
Level 2 performs the functions of error control, link initialization, error rate monitoring, flow control and delineation of messages.
Level 3 provides the functions required for a signaling network. Each node in the network has a single point odd, which is a 14 bit address. Every message contains a point code of the originating and terminating nodes for that messages.
Levels 1 to 3 form the message transfer part (MTP) of CCITT no. 7 .
Level 4 is the user part. This consists of the processes for handling the service being supported by the signaling system. The message transfer part is capable of supporting many different user parts. So far, three have been defined: the telephone user part(TUE), the data user part (DUP) and the (ISDN) user part (ISDN-UP).

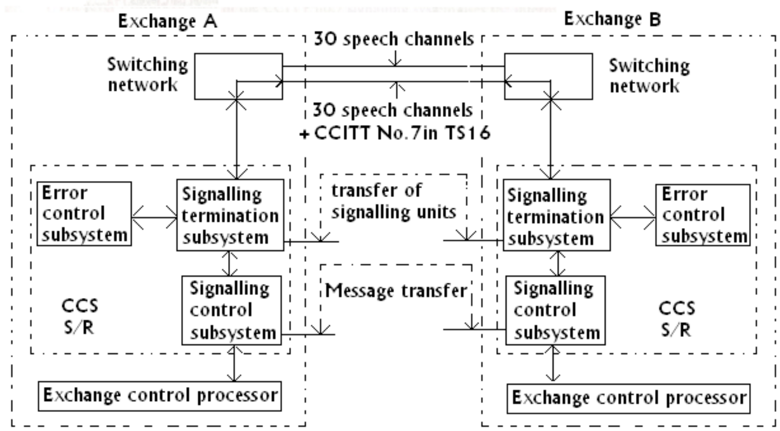


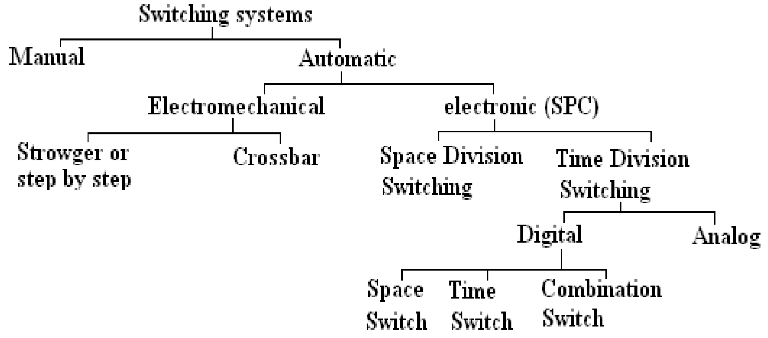
FIG – Block Schematic Diagram of CCITT No.7 Signally System

Q.70. Explain how presentation layer helps in establishing and processing data in End to End layers. **(8)**

Ans:
The purpose of the presentation layer is to represent information to the communicating application entities in a way that preserves the meaning while resolving syntax differences. Syntax differences are resolved by encoding application data into a standard abstract notation that is valid throughout the network. Thus, file format differences (e.g IBM format or DEC format) ,data representation differences(e.g ASCII or EBCDIC) or data structure differences are all resolved by using a standard notation. Data transformation and formatting may include data compression, encryption etc. There are two aspects associated with network wide handling of a variety of data in a standard form, and second, the transmission of data as a bit stream across the network.
This layer translates between different formats and protocols resenting functions data file formatting, encoding, encryption and decryption of data messages, dialogue procedures, data compression algorithms etc. This layer performs code and character set translation. And formatting information and determines the display mechanism for message.

Q.71. Discuss the classifications of switching systems. In what way is stored program control superior to hard wired control? **(8)**

Ans:
The classifications of switching systems are given in the block diagram below:



The SPC gains superiority over hard wired due to following points:

SPC	Hard Wired Control
Flexible	Not Flexible
Slower	Faster
More expensive for moderate processing functions	Less expensive for Moderate, simple and fixed Processing functions
Easier to implement, Complex Processing functions	Difficult to implement, Complex processing functions
Introducing new services is easy	Introducing new services is not easily possible
Easier to maintain	Difficult to maintain

Q.72. Discuss the various functions of telephone switching systems. **(8)**

Ans:
Functions of telephone switching systems are:

(i) Attending: The system must be continually monitoring all lines to detect call requests. The calling signal is sometimes called seize signal.

(ii) Information receiving: In addition to receiving calls and clear signals the system must receive information from the caller as to the called line required. This is called address signal.

(iii) Information Processing: The system must process the information received, in order to determine the action performed and to control these actions.

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

(v) Interconnection: For a call between two customers, three connection are made in the following sequence:

(a) A connection to the calling terminal
(b) A connection to the called terminal
(c) A connection between the two terminals

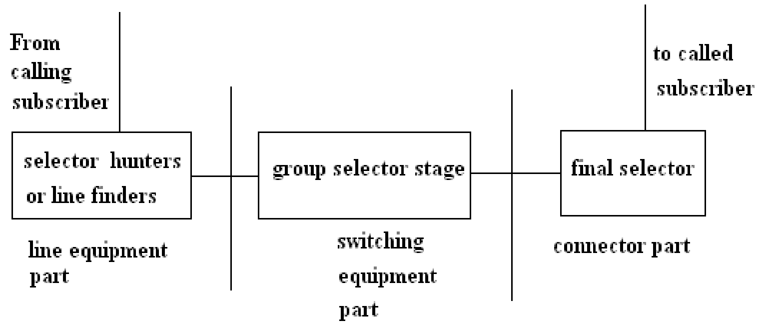
(vi) Alerting: having made the connections, system sends a signal to alert the called customer, e.g. by sending ringing current to a customer's telephone.

(vii) Supervision: 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.

(viii) Information sending: If the called customer's line is located on another exchange the additional function of information sending is required. The originating exchange must signal the required address to the terminating exchange (and possibly to intermediate exchanges if the call is to be routed through them)

Q.73. With the help of a neat diagram explain the configuration of a step by step switching system **(8)**

Ans:



Configuration of a step by step switching system:

A step by step switching system may be constructed using uni-selectors or two motion selectors or a combination of both. The wiper contacts of these selectors move in direct response to dialled pulses or other signals like off hook from the subscriber telephone. The wiper steps forward by one contact at a time and moves by as many contacts (takes as many steps) as the no of dialled pulses received or as required to satisfy certain signalling conditions Hence the name step by step switching is given to this method.

A step by step system has three major parts as shown in the figure:

1. **Line Equipment part**: It consists of selector hunters or line finders which represent two fundamental ways in which a subscriber gains access to common switching resources. A selector hunter searches and seizes a selector from the switching matrix part. There is one selector hunter for every subscriber. Line finders are associated with the first set of selectors in switching matrix part and there is one line finder for each selector in the set. As the name implies, a line finder searches and finds the line of a subscriber to be connected to the first selector associated with it.
2. **Switching matrix part**: It consists of one or more sets of two motion selectors known as first group selectors, second group selector, and so on. The larger the exchange size, the larger the no. of group selector stages.
3. **Connector part:** It comprises one set of two motion selectors known as final selectors.

Q.74. Discuss the advantages of automatic switching systems over manual switching system. **(8)**

Ans:

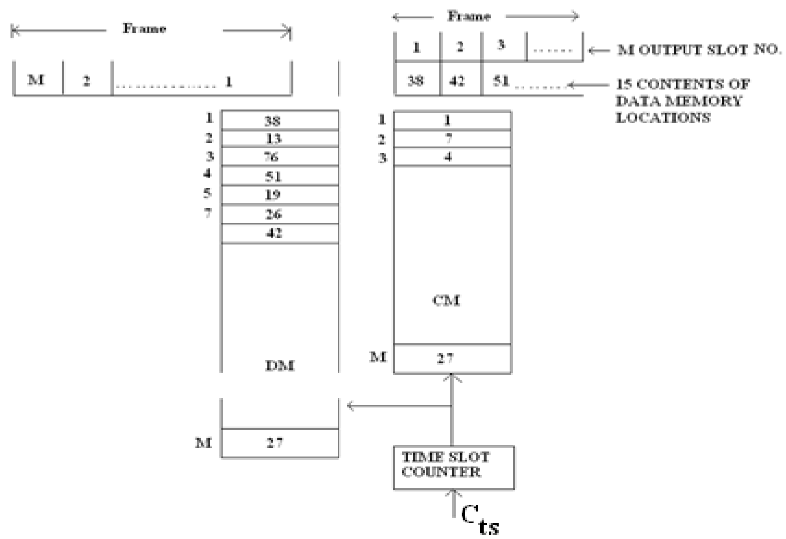
Automatic switching systems have a no. of advantages over the manual exchanges:

1. In a manual exchange, the subscriber needs to communicate with the operator and a common language becomes an important factor. On the other hand, the operation of automatic exchange is language independent.
2. A greater degree of privacy is obtained in automatic exchanges as no operator is normally involved in setting up and monitoring a call.
3. Establishment and release of calls are faster in automatic exchanges. In manual exchange, operator takes a few minutes to notice the end of conversation and release the circuits.

4. In an automatic exchange, the time required to establish a call remains more or less of same order irrespective of the load on the system or the time of the day. In manual systems this may not be true.

Q.75. Discuss the basic structure and principle of operation of Time Slot Interchange (TSI) switch with the help of a neat diagram. **(8)**

Ans:



Principle of time slot interchange

CM=Control memory DM= Data Memory

Time multiplexed time switch permits time slot inter changing. In TSI, a speech sample input during one time slot may be sent to the output during a different time slot. Such an operation necessarily implies a delay between the reception and the transmission of the sample.
This operation can be explained with ref. to above figure.

1. M channels are multiplexed on each trunk
2. The time slot duration is given by t_{TS} =125/M The time slot counter is incremented at the end of each time slot. The content of the counter provides location address for control memory and data memory.
3. Data memory and control memory accesses take place simultaneously in the beginning of the time slot. Thereafter , the contents of the control memory are used as the address of the data memory and the data read out to the output trunk
4. The input sample is available for reading in at the beginning of the time slot and the sample is ready to be clocked in on the output stream at the end of the time slot.

Q.76. Explain the following design parameters
S, SC, TC, C, CCI, EUF, K, T_S **(8)**

Ans:
S: Total no. of switching elements
A good design must attempt to minimize the no. of switching elements
SC: Switching capacity

Max. no. of simultaneous calls that can be supported by the switching system

TC: Traffic handling capacity

Traffic Handling Capacity is given by Switching capacity / Theoretical max. load

C: Cost of switching system

$C = S \times C_S + C_C + C_{Ch}$

where C_S, C_C, C_{Ch} are cost per switching element, cost of the common control system and cost of the common hardware respectively.

CCI: Cost capacity index

CCI = switching capacity/cost per subscriber line

EUF: Equipment utilization factor

EUF=No. of switching elements in operation when the SC is fully utilized/total no. of switching elements in the system

K: Number of switching stages

T_S**:** Call setup time

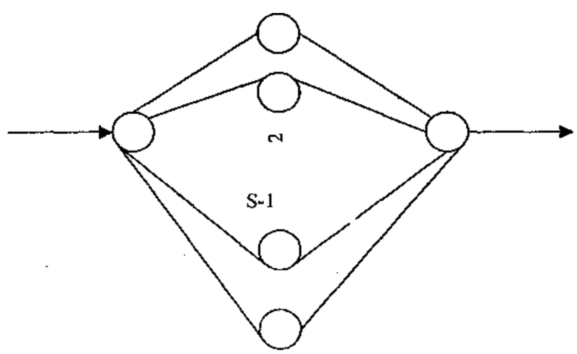
$T_S = T_{ST} \times K + T_O$

Where T_O is the time required for function other than switching and Tst is average switching time per stage.

Q.77 A three stage network is realized by using switching matrices of size **p x s** in stage 1, **r x r** matrices in stage 2 and **s x p** matrices in stage 3. Using the Lee's probability graph show that the Blocking Probability for the three stage network is given by

$P_B = \left[1 - (1 - \alpha / k)^2\right]^s$ Where $k = s/p$ and $\alpha =$ Inlet utilization factor **(8)**

Ans:



LEE'S Graph for a Three Stage Network

For a three stage network;

One path from input to output=2 links in series=link between stage 1 and stage 2+ link between stage 1 and stage 2

Let β = probability that a link is busy

So $1-\beta$ = probability that a link is **free**

i.e. the probability that a particular link (e.g. 1-2) from first stage to second stage is Available = $1-\beta$.

Similarly The probability that a particular link (e.g.2-3) from second stage to third stage is available = $1-\beta$

So probability that both of these links are available = $(1-\beta)(1-\beta) = (1-\beta)^2$

Probability that a particular path is busy = $[1-(1-\beta)^2]$

Now there are S parallel paths between input and output, so probability that all of them are busy = $[1-[1-\beta)^2]^S = P_B$

Therefore

$$P_B = \left[1-(1-\beta)^2\right]^5$$

If α = Inlet utilization factor, **i.e.** probability that an inlet at first stage is busy

Then $\beta = \frac{\alpha}{s}p$, putting $\frac{s}{p} = k$

$$\beta = \frac{\alpha}{k} \text{ so } P_B = \left[1-\left[1-\frac{\alpha}{k}\right]^2\right]^S$$

Q.78. Explain SPC. Also discuss the different modes of Centralized SPC. **(8)**

Ans:

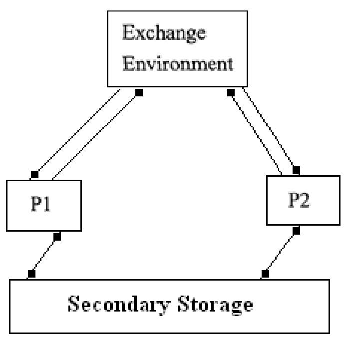
In stored program control systems, a program or set of instructions to the computer is stored in its memory and the instructions are executed automatically one by one by the processor. Carrying out the exchange control functions through programs stored in the memory of a computer led to this name. There are two approaches to organizing stored program control:

1. Centralised: In this control, all the control equipment is replaced by a single processor which must be quite powerful.

2. Distributed: In this control, the control functions are shared by many processors within the exchange itself.

In centralised SPC, dual processor architecture may be configured to operate in one of three modes:

a. Standby mode: In this mode, one processor is active and the other is on standby, both software and hardware wise. The standby processor brought online when active processor fails. An important requirement of this configuration is the ability of the standby processor to reconstitute the state of the exchange system when it takes over the control.

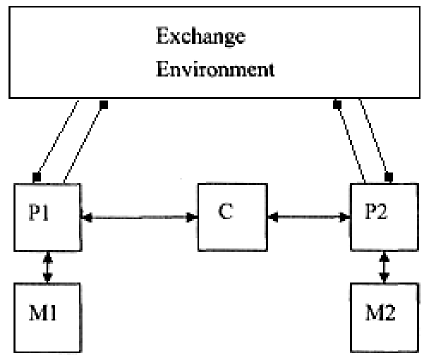


P_1 = Active processor

P_2 = Standby processor

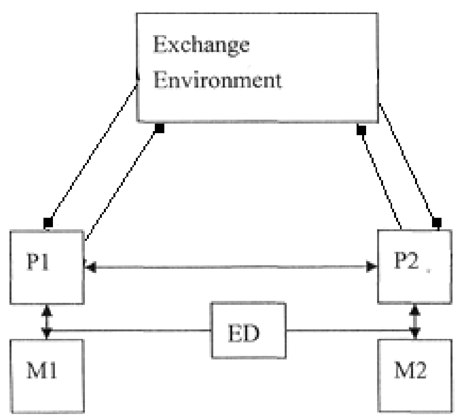
b. Synchronous duplex mode:

In synchronous duplex mode, hardware coupling is provided between the two processors which execute the same set of instructions and compare the results continuously. If a mismatch occurs, the faulty processor is identified and taken out of service immediately. When the system is operating normally, the two processors have the same data in their memories at all the times and receive all information from the exchange environment.



C = comparator
P = processor
M = memory

c. Load sharing mode: In load sharing operation, an 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.



Q.79 Discuss the various enhanced services that can be made available to the subscribers because of stored program control. **(8)**

Ans:
One of the immediate benefits of stored program control is that a host of new services can be made available to the subscriber. They can be grouped under four categories:

1. Services associated with the calling subscriber and designed to reduce the time spent on dialling and the number of dialling errors.
   - Abbreviated dialling
   - Recorded number calls or no dialing calls
   - Call back when free

2. Services associated with the called subscriber and designed to increase the call completion rate.
   • Call forwarding
   • Operator answer
3. Services involving more than two parties.
   • Calling number record
   • Call waiting
   • Consultation hold
   • Conference calls
4. Miscellaneous
   • Automatic alarm
   • STD Barring
   • Malicious call tracing

Q.80. With reference to telephone traffic, explain the following terms
GOS, BHCA, CCR, BHCR **(8)**

Ans:
GOS: in loss systems, the traffic carried by the network is generally lower than the actual traffic offered to the network by the subscribers. The overload traffic is rejected and hence not carried by the network. The amount of traffic rejected by the network is an index of the quality of the network. This is termed as GOS and is defined as the ratio of **lost traffic** to **offered traffic.**
BHCA: The number of call attempts in the busy hour is called Busy hour call attempts (BHCA), which is an imp. Parameter in deciding the processing capability of a common control or SPC system of an exchange
CCR: Call completion rate is defined as the ratio of the no. of successful calls to the no. of call attempts.
BHCR: Busy hour calling rate is defined as the average no. of calls originated by a subscriber during the busy hour.

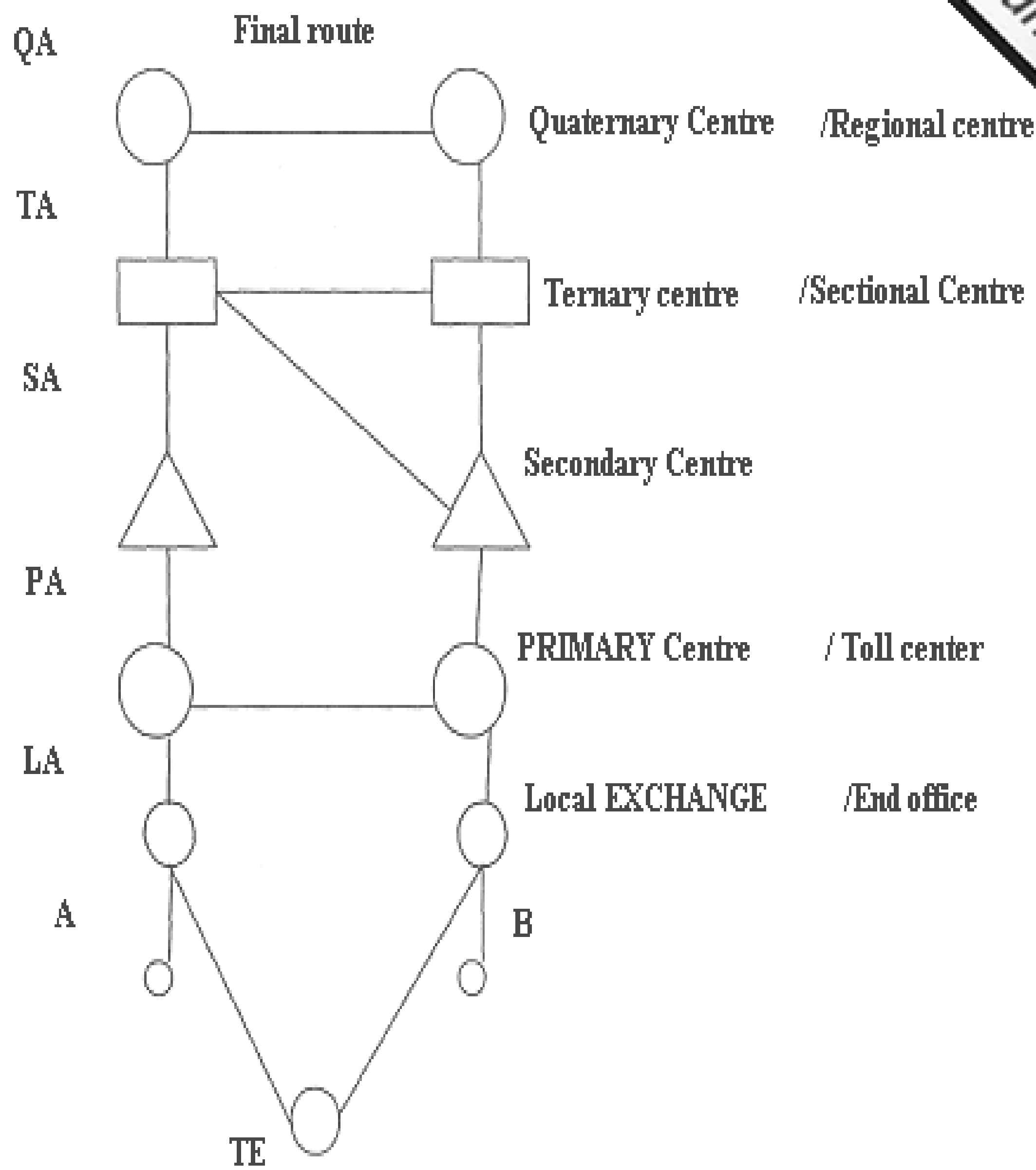
Q.81. What are the advantages of Hierarchical Networks? Discuss the 5-level switching hierarchy recommended by CCITT. **(8)**

Ans:
Hierarchical networks are capable of handling heavy traffic where required, and at the same time use minimal number of trunk groups. In a strictly hierarchical network, traffic from subscriber A to Subscriber B and vice versa flows through the highest level of hierarchy i.e. quaternary centres. A traffic route via the highest level of hierarchy is known as final route

01. End offices are the switching offices to which individual subscribers are directly connected

02. Toll offices are the switching offices used to interconnect the end offices. Intercity calls are routed via toll office.

03. A very populous area, such as a large city will have many end offices. Much of the traffic involves calls between subscribers in the city and hence between end offices. To facilitate this end office to end office traffic a **Tandem switching** office may be provided for one group of end offices.



CCITT Hierarchical structure

LA, PA, SA, TA, QA = Local, primary, secondary, tertiary, quarternary areas respectively

E=Exchange TE = tandem exchange

Q.82. What are End-to-End layers in ISO-OSI reference model? Explain briefly the function of each one of them. **(8)**

Ans:
The layers 4-7 of ISO-OSI reference model communicate with peer entities in the-end systems. There is no communication with entities in the intermediate systems. In this sense, layers 4-7 are often called end-to-end layers. These are Transport layer. Session layer, Presentation layer and Application layer respectively.

1. Transport Layer:

(i) It is responsible for establishing a network independent communication path suitable for the particular terminal equipments (e.g. providing the appropriate data rate and end-to-end error control). It thus relieves the user from being concerned with such details

(ii) In a packet switched network, the transport entity breaks up a long user message into packets to match the network capabilities. The packets are reassembled at the receiving end to reconstruct the user message.

(iii) End-to-end flow control & end-to-end error recovery are also the functions of transport layer.

2. Session Layer:

i. The session layer is used to allow users to identify themselves when wanting access to the network.
ii. This is concerned with setting up and maintaining an operational session between terminals. E.g. "signing on" at the commencement of a task and "signing off at its end.
iii. The main function of-the session layer is to organize different sessions between cooperating entities and perform all related functions like synchronization, failure management, control etc. for the successful execution of a session
iv. Another facility offered by the session layer is known as Activity management.

3. Presentation Layer;

i. This is concerned with the format of the data represented, in order to overcome difference in representation of the information as supplied to one terminal and required at the other. Its purpose is to make communication over the network machine independent.
ii. It resolves the syntax differences by encoding data into standard abstract notation that is valid throughout the network. Thus file format differences, data representations, data structure are resolved using a standard notation.

4. Application Layer:

As the highest layer in the OSI reference model, the application layer provides services to the users of OSI environment. The layer provides following services:

i. Electronic mail or message handling service
ii. Directory services
iii. Cost allocation
iv. Determination of quality of service
v. File transfer and management
vi. Editors and terminal support services
vii. Telematic services like videotext.

Q.83. What are the basic approaches to the design of subscriber access to Strowger systems? Describe them. **(8)**

Ans:

A step by step switching system has three major parts as shown in Fig . The line equipment part consists of selector hunters or line finders and the other two parts consist of selectors. The selector hunter and line finders represent two fundamental ways in which a subscriber gains access to common switching resources. The selector hunter scheme is sometimes called subscriber uniselector scheme as there is a dedicated uniselector for each subscriber in the system. Line finders are associated with the first set of selectors in the switching matrix part and there is one line finder for each selector in the set. The line equipment part is also known as preselector stage. The selector hunters and line finders are generically referred to as preselectors.

The switching matrix part consists of one or more sets of two-motion selectors known as first group selector, second group selector, and so on. The larger the exchange size, the larger is the number of group selector stages. The connectors part comprises one set of two-motion selectors known as final selectors.

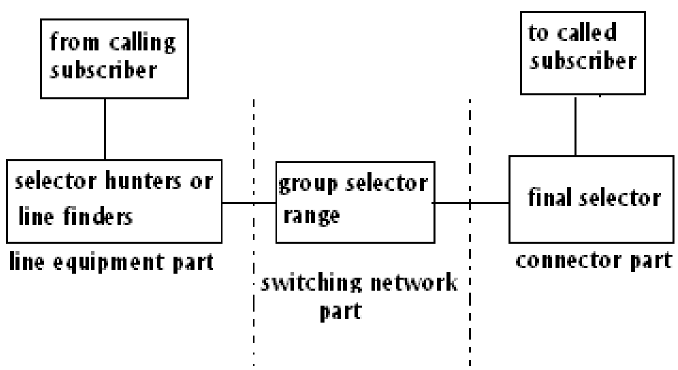


FIG - Configuration of a Step-by-step Switching System

The **selector hunter and line finder schemes** are illustrated in the trunking diagrams shown in Fig in selector hunter based approach, when a subscriber lifts his hand set, the interrupter mechanism in his selector hunter gets activated and the wiper steps until a free first group selector is found at the outlet. The status of the first group selector, free or busy, is known by a signal in one of the bank contacts of the selector hunter. Once a free first selector is sensed, the interrupter is disabled and the first selector is marked 'busy'. Then, the first selector sends out a dial tone to the subscriber via the selector hunter which simply provides an electrical path. The first selector is now ready to receive the dialing pulses from the subscriber. It is possible that two selector hunters land on the same free first selector simultaneously and attempt to seize it. This is resolved by suitable seizure circuit.

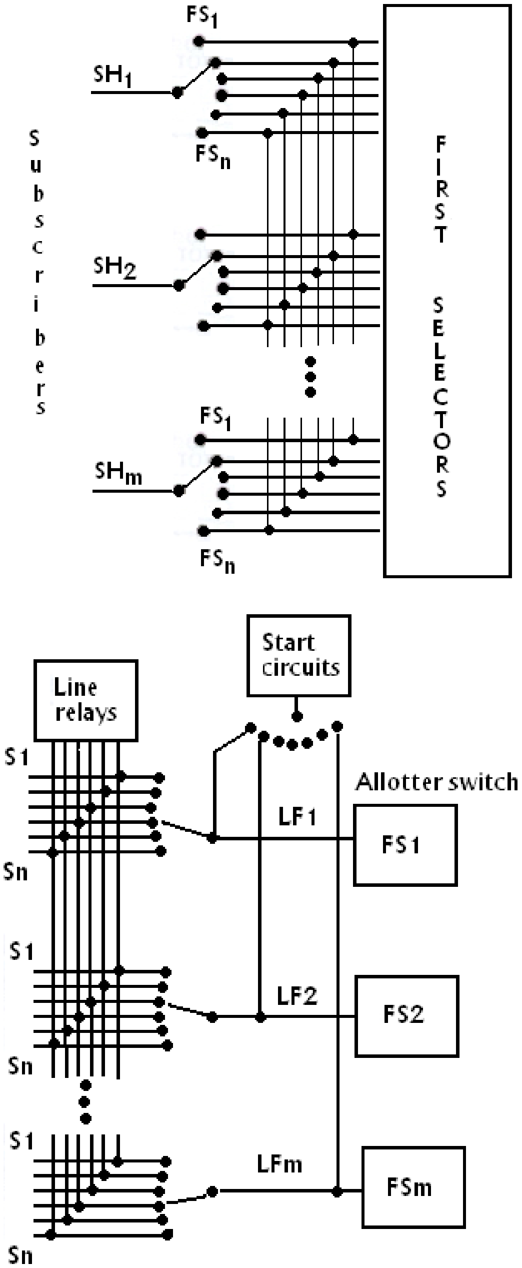


FIG - Subscriber Access to Strowger Switching System

In the case of **line finder based approach**, the off-hook signal is sensed by all the line finds us. Then the interrupter mechanism of one of the finders, whose associated first selector is free, gets activated and the line finder wiper steps until it reach the contact on to which the subscriber is terminated. On finding the line, the concerned first selector sends out the dial tone to the subscriber in readiness to receive the dial pulses. The selection of one of the line finders out of many free line finders, is achieved by means of an allotter switch in the start circuit of the line finders as shown in Fig. When a subscriber lifts his receiver, the start signal from his relay in passed to the particular line finders via the common start circuit and the allotter switch. The line finder then commences to hunt for the calling line. As soon as the calling line is found, the allotter switch steps to next free line finders. In effect, the line finder and the associated first selector to be used for the next future call is selected in advance by the allotter circuit. In practical designs, several allotter switch are provided in the system to serve calls that may originate in quick succession or simultaneously.

PART-III

NUMERICALS

Q.1. Calculate the maximum access time that can be permitted for the data and control memories in a TSI switch with a single input and single output trunk multiplexing 2500 channels. Also, estimate the cost of the switch and compare it with that of a single stage space division switch. **(6)**

Ans:

$$t_m = \frac{125 \times 10^3}{2500 \times 2} = 25 \text{ ns}$$

C= 2 X 2500 = 5000 units

This switch is non blocking and supports full availability. An equivalent single stage space division which uses a matrix of 2500 X 2500. Hence, the cost of such a switch is 6.25 million units

$$\text{Cost advantage of time switch} = \frac{6.25 \times 10^6}{5000} = 1250$$

Q.2 A subscriber makes three phone calls of 3 minutes, 4 minutes and 2 minutes duration in a one hour period. Calculate the subscriber traffic in erlangs, CCS and CM. **(6)**

Ans:

$$\text{Subscriber traffic in erlangs} = \frac{\text{busy period}}{\text{total period}} = \frac{3+4+2}{60} = 0.15 \text{ E}$$

$$\text{Traffic in CCS} = \frac{(3+4+2) \times 60}{100} = \frac{540}{100} = 5.4 \text{ CCS}$$

Traffic in CM = 3+4+2 = 9 CM.

Q.3. In a national transmission system, the characteristic impedances of the 4-wire circuit and the 2-wire circuit are 1200 Ω and 1000 Ω respectively. The average phase velocity of the signal in the circuit is 3×10^7 m/s. If the largest distance of a connection is 300 km, determine the return loss and round trip delay for echo. **(6)**

Ans:

$$RL = 20 \log \frac{Z_4 + Z_2}{Z_4 - Z_2} \text{ dB}$$

Where
RL = return loss
Z_4 = impedance of the 4-wire circuit
Z_2 = impedance of the 2-wire circuit

$$RL = 20 \log \frac{2200}{200} = 20.8 \text{ dB}$$

$$\text{Round trip delay for echo} = \frac{300 \times 10^3}{3 \times 10^7} = 10 \text{ ms}$$

Q.4.. A three stage network is designed with the following parameters: M=N=512, p= q = 16 and $\alpha = 0.65$. Calculate the blocking probability of the network, if s=16. Symbols carry their usual meanings. **(8)**

Ans:

The blocking probability

$P_B = [1-(1- \alpha /k)^2]^s$

Where

$\alpha = 0.65$

$k = p/s = 16/16 = 1$

$P_B = [1-(1- 0.65)^2]^{16}$

$[1-(0.35)^2]^{16} = 0.123$ **Ans.**

Q.5. Discuss grade of service. During busy hour, 1500 calls were offered to a group of trunks and 8 calls were lost. The average call duration was 120 seconds. Calculate the traffic offered, traffic lost, Grade of service and total duration of congestion. **(8)**

Ans:

(i) Grade of service: In loss systems, the traffic carried by the network is generally lower than the actual traffic offered to the network by the subscribers. The overload traffic is rejected and hence it is not carried by the network. The amount of traffic rejected by the network is an index of the quality of the service offered by the network. This is termed **Grade of Service (GOS)** and is defined as the ratio of lost traffic to offered traffic. Offered traffic is the product of the average number of calls generated by the users and the average holding time per call.. Accordingly , GOS is given by

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

Where

A= offered traffic

A_0 = carried traffic

$A-A_0$= lost traffic

(ii) we know that

Traffic offered $= A = \frac{Ch}{T} = \frac{1500 \times 2}{360} = 50$ E

Traffic lost $= \frac{8 \times 2}{6/30} = \frac{4}{15}$ E $= 0.26$ E

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

$= \frac{8}{1500}$

Duration of congestion = Grade of service x 1 h

$= \frac{8}{1500} \times 3600 = \frac{96}{5} = 19.2$ sec.

Q.6. In a 10000 line exchange, 0000 to 2999 is allotted to x group of subscribers, out of which 40% are active during busy hour. The remaining numbers are domestic numbers out of which 20% are active. Each group consists of 100 subscribers. Determine the number of final selectors required. **(8)**

Ans:
Number of simultaneous calls for X group = 40 per group.
Number of simultaneous calls for domestic subscriber group = 5 per group
Total number of final selectors required = 3 x 20 + 7 x 5 = 95.

Q.7. Calculate the number of trunks that can be supported on a time multiplexed space switch given that, 32 channels are multiplexed in each stream, while the control memory access time is 100 ns and the bus switching and transferring time is 100 ns per transfer. **(6)**

Ans:
We know that N =125/ Mt
Where N = number of incoming or outgoing trunks
M = sample per frame
t = switching time including memory access time per inlet outlet pair.
For M = 32
The total switching time = 100 + 100 = 200 ns.

$$\text{Therefore } N = \frac{125}{32 \times 200 \times 10} = 20$$

Q.8. An exchange uses a –40 V battery to drive subscriber lines. A resistance of 250 ohms is placed in series with the battery to protect it from short circuits. The subscribers are required to use a standard telephone set which offers a dc resistance of 50 ohms. The microphone requires 23mA for proper functioning given DC resistance of 133 ohms/km, find the farthest distance from the exchange at which the subscriber can be located **(6)**

Ans:
Let R be the line loop resistance, using the relation
I = V/R
The value of R can be calculated as

$$23 \times 10 = \frac{40}{(250+50+R)}$$

R = 1439
For 26 AWG wire R =133.89
Loop length= 1439/133.
Therefore, the farthest distance at which the subscriber can be located is
10.74/2 = 5.37 km

Q.9. A group of 20 servers carry traffic of 10 erlangs. If the average duration of a call is three minutes, calculate the number of calls put through by a single server and the group as a whole in a one hour period. **(6)**

Ans:
Traffic per server= 10/20 =0.5 E
i.e. a server is busy for 30 minutes in one hour
Number of calls put through by one server= 30/3 = 10 calls
Total number of calls put through by the group= 10 x 20 = 200 calls

Q.10. An amplifier has an input resistance of 600 ohms and a resistive load of 75 ohms. When it has an rms input voltage of 100 mV, the rms output current is 20mA. Find the gain in dB. **(6)**

Ans:

Input power is P = (100 x 10) /600 W = 16.7 W

Output Power is P = (20 x 10) x 75 W = 30 m W

Gain is $\frac{P}{P} = \frac{30 \times 10}{16.67 \times 10}$

10 log 1.8 +10 log = 2.6 + 30 = 32.6 dB.

Q.11. A three stage switching structure supports 100 inlets and 400 outlets. Find the number of cross points, and the number of primary and secondary switches used in the design. **(6)**

Ans:

We know that

$m = \frac{M}{M+N}$ and $n = \frac{N}{M+N}$

m = 100

1. if m=5, n=20, there are :
   20 primary switches of size 5 x 5
   5 secondary switches of size 20 x 20
   20 tertiary switches of size 5 x 20
2. if m = 4, n =16, there are:
   25 primary switches of size 4 x 4
   4 secondary switches of size 25 x 25
   25 tertiary switches of size 4 x 16

Q.12. Define congestion and grade of service. In a particular exchange during busy hour 1200 calls were offered to a group of trunks, during this time 6 calls were lost. The average call duration being 3 minutes Calculate

(i) traffic offered in erlangs'
(ii) traffic lost
(iii) grade of service and
(iv) period of congestion **(8)**

Ans:

(i) Congestion: It is uneconomic to provide sufficient equipment to carry all the traffic that could possibly be offered to a telecommunication system. In a telephone exchange it is theoretically possible for every subscriber to make a call simultaneously. A situation can therefore arise that all the trunks in a group of trunks are busy, and so it can accept further calls. This state is known as congestion. In a message-switched system, calls that arrive during congestion wait in a queue until an outgoing trunk becomes free. Thus, they are delayed but not lost. Such systems are therefore called queuing systems or delay system. In a circuit-switched system, such as a telephone exchange, all attempts to make calls over a congested group of trunks are successful. Such systems are therefore called lost-call systems. In a lost-call system the result of congestion is that the traffic actually carried is less than the traffic offered to the system. We may therefore write:

(ii) Grade of Service: In loss systems, the traffic carried by the network is generally lower than the actual traffic offered to the network by the subscribers. The overload traffic is rejected and hence it is not carried by the network. The amount of traffic rejected by the network is an index of the quality of the service offered by the network. This is termed **Grade of Service (GOS)** and is defined as the ratio of lost traffic to offered traffic. Offered traffic is the product of the average number of calls generated by the users and the average holding time per call. Accordingly, GOS is given by

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

Where

A= offered traffic

A_0 = carried traffic

$A-A_0$= lost traffic

(iii) Solution:

(a) traffic offered in erlangs'

$A = Ch/T = 1200 \times 3/60 = 60$ E

(b) traffic lost

$1194 \times 3/60 = 59.7$ E

(c) grade of service

$6 \times 3/60 = 0.3$ E

(d) period of congestion

$B = 6/1200 = 0.005$

$0.005 \times 3600 = 18$ seconds.

Q.13. In a two stage network there are 512 inlets and outlets, r=s=24. If the probability that a given inlet is active is 0.8, calculate:

(i) The switching elements

(ii) Switching capacity

(iii) Blocking probability **(6)**

Ans:

$N = M = 512$, $\alpha = 0.8$, $r = s = 24$

(i) The number of switching elements:

$S = M_s + N_r = 512 \times 24 + 512 \times 24$

(ii) Switching Capacity

$SC = rs = 24 \times 24$

(iii) Blocking Probability

$$P_B = \frac{M\,\alpha\,(s-1) - ((M/r-1))\,\alpha}{rs(s-1)}$$

$$= \frac{512 \times 0.8(23) - (512/24-1)0.8}{24 \times 24\,(23)} = 0.7$$

Q.14. In a subscriber loop that contains a series resistance of 300 ohms to protect the batteries in the exchange, a normalized telephone draws 10 mA and its standard input d.c. resistance is 50 ohms. Calculate the maximum distance at which a subscriber can get good speech reproduction if a cable of 52 ohms/km resistance is used. If a standard hand set of 30 mA current is used what will be the change. **(8)**

Ans:
Let R_L be the line loop resistance
Normalized Microphone current = 10 m A
Telephone set resistance = 50 Ω
Series resistance = 300 Ω
Battery voltage = 40 V
I =V/R

$$10 \times 10^{-3} = \frac{40}{(300+50+R_L)}$$

Hence 3500 + 10 R_L = 40,000
10 R_L = 3650 Ω
Maximum distance from exchanging = 3650/133.89 = 27.25 Km

(ii) When hand set current = 30 mA
(iii)

$$30 \times 10^{-3} = \frac{40}{(300+50+R_L)} = \frac{40}{(350+R_L)}$$

Hence 30 (350+ R_L) = 40,000
10500+30 R_L = 40,000
30 R_L = 29500/30 =983 Ω
For 26 AWG wire
Loop length = 983/133.89 = 7.3 km

Q.15. Calculate the blocking probably Pb in 100 line strowger switching system where 10 calls are in progress and 11th one arrives, probably that there is a call in a given decade = 1/10 and probably that another call is destined to same decade but not to same number = 9/98. **(8)**

Ans:
Probability that there is a call in a given decade =10/100
Probability that other call is destined to the same decade but not to same number = 9/98
Therefore, the blocking probability = (1/10)*(9/98)
= 0.009 Ans

Q.16. A CSMA/CD bus spans a distance of 1.5 Km. If data is 5 Mbps, What is minimum frame size where propagation speed in LAN cable is $200\,\text{m}/\mu\text{s}$. **(8)**

Ans:
Typical propagation speed in LAN cables = 200 m/µs
End-to-end propagation delay t = 1500/200 = 7.5 µs
Minimum frame size = 2x 7.5 x 10 x 5 x 10 = 75 bits.
Minimum frame size for CSMA/CD LAN may be expressed as
F = 10 dR

Where d= length of the cable in km.
R= data rate in Mbps
In arriving above, propagation delay is taken to be 5 µs/km

Q.17. Define congestion and grade of service. In a particular exchange during busy hour 900 calls were offered to a group of trunks, during this time 6 calls were lost. The average call duration being 3 minutes. Calculate:
(i) Traffic offered in erlangs (ii) Traffic lost
(iii) Grade of service (iv) Period of congestion **(8)**

Ans:
Congestion: It In a telephone exchange it is theoretically possible for every subscriber to make a call simultaneously. The cost of meeting this demand would be prohibitive, but the probability of it happening is negligible. This situation can therefore arise that all the trunks in a group of trunks are busy, and so it can accept further calls. This state is known as congestion. In a message-switched system, calls that arrive during congestion wait in a queue until an outgoing trunk becomes free. Thus, they are delayed but not lost. Such systems are therefore called queuing systems or delay system. In a circuit-switched system, such as a telephone exchange, all attempts to make calls over a congested group of trunks are successful. Such systems are therefore called lost-call systems. In a lost-call system the result of congestion is that the traffic actually carried is less than the traffic offered to the system.

Grade of service: In loss systems, the traffic carried by the network is generally lower than the actual traffic offered to the network by the subscribers. The overload traffic is rejected and hence it is not carried by the network. The amount of traffic rejected by the network is an index of the quality of the service offered by the network. This is termed **Grade of Service (GOS)** and is defined as the ratio of lost traffic to offered traffic. Offered traffic is the product of the average number of calls generated by the users and the average holding time per call. Accordingly, GOS is given by

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

Where
A= offered traffic
A_0 = carried traffic
$A-A_0$= lost traffic

(i) Traffic offered in erlangs Ch/T = 900X3/60 = 45 E

(ii) Traffic carried = $\frac{894 \text{ X } 3}{60}$ = 44.7 E

(iii)

(c) Traffic lost = $\frac{6\times3}{60} = 0.3$ E

(iv) B = 6/900 = 0.0066
Total duration of period of congestion = 0.0066 X 3600 =24 seconds

Q.18 A call processor in an exchange requires 120 ms to service a complete call. What is the BHCA rating for the processor? If the exchange is capable of carrying 700 Erlangs of traffic, what is the call completion rate? Assume an average call holding time of 2 minutes. **(8)**

Ans:

Call service time= 120ms

BHCA-(60X60)/(120X10^{-3})=30,000

t_h =holding time 2 minutes

Ao=mean effective traffic carried by the network=700 Erlangs

Ao==Co X t_h ,where Co=mean effective traffic rate

So **Co** = Ao/ t_h =700/2 per minute= 21,000 calls per hour

So total call attempts in Busy Hour= 30,000

And no. of successful calls= 21,000

Therefore CCR= 21,000/30.000 = 0.7