IMPORTANT QUESTION BANK

ECS-601 Computer Networks

YEAR / SEM: III/VI

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Part – A (2 Marks)

General 'Fill in the Blanks' Questions **(**may have some duplicates**)**

Q.1. The connection points between the Network Layer and the Transport Layer is the IP number(s) whereas the connection points between the Transport Layer and the upper layers are called ***Ports*** which are represented by ***16*** bits in the TCP segment header.

Q.2. Unlike ***TCP*** the Transport Layer Protocol ***UDP*** is a connectionless protocol.

Q.3. The programmatic connection between the Network Layer and the upper (application) layers is usually established by small software running in Transport Layer. These are usually called ***Sockets***.

Q.4. In a 802.3 LAN, IP to ethernet address translation is performed by/with ***ARP*** which stands for ***Address Resolution Protocol***.

Q.5. In an IP address with CIDR standard a quad can range from ***0*** to ***255***.

Q.6. Bluetooth uses ***Time Division*** Multiplexing.

Q.7. A Bluetooth network with 7 active slaves is called a ***Piconet***.

Q.8. Bluetooth standard specifies ***13*** applications which are called ***profiles***.

Q.9. As modulation technology, Bluetooth employs ***FSK*** with ***1M*** bps.

Q.10. The number of necessary OSI layers in a bridge is ***two***

Q.11. ***Routers*** and ***Gateways*** are internetworking-connecting devices

Q.12. A networked computer must minimally have a ***host-IP***, ***a netmask*** and ***default gateway-IP*** numbers set up in order to do IP-networking properly.

Q.13. An IP-address with all host bits set to 1 is called ***Broadcast address***.

Q.14. A Class-B IP-range may have about ***65000*** hosts in it.

Q.15. IEEE-802.3 defines ***broadcast*** networks standards.

Q.16. The word 'Base' in 10BaseT identifies the media as ***baseband***.

Q.17. The letter 'T' in 10BaseT indicates that ***it is a twisted-pair cable***.

Q.18. An ethernet address with all bits set to 1 is called ethernet ***broadcast*** address.

Q.19. CIDR addressing scheme allows us to create ***subnets*** for efficient IP use.

Q.20. ***ARP*** is a process of obtaining ethernet addresses from IP-addresses.

Q.21. Performance of a network is usually measured by ***the amount of data transferred per unit time***.

Q.22. Reliability of a network is usually measured by ***the frequency of failure (inverse of it)***.

Q.23. Logical to physical addressing conversion is done in ***the Network Layer***.

Q.24. In ***stop and wait*** flow control method receiver ACKs each data frame received. The alternative to this is called ***sliding window***.

Q.25. CSMA stands for ***"Carrier Sense Multiple Access"*** which is standardized in IEEE 802.3.

Q.26. The 10 Mbps twisted pair ethernet is usually designated by ***10BaseT***.

Q.27. ***Ethernet or NIC addresses*** are 48 bit unique numbers used to identify physical devices in CSMA/CD networks.

Q.28. ***Bridges*** divide the networks into smaller segments in order to reduce traffic.

Q.29. ***Repeaters*** and ***Bridges*** are networking connecting devices.

Q.30. In class-B IP-addresses ***16*** and ***16*** bits are reserved for host and network identification respectively.

Q.31. Topologically fully connected networks have ***the highest*** performance and ***the highest*** cost among the possible.

Q.32. In ***full-duplex*** transmission mode devices can transmit and receive simultaneously.

Q.33. For bit-rate to be three times the baud rate we need at least ***8*** constellations.

Q.34. Physical Layer deals with the ***physical and electrical*** specifications.

Q.35. The term ***point-to-point*** indicates the dedicated links between two nodes.

Q.36. ***Bus topology*** is the simplest and cheapest topology to implement in small networks.

Q.37. ***Full-Duplex*** transmission mode can be characterized simply as "two way simultaneous transmission/reception".

Q.38. Performance of a network is usually measured by ***bits per second***

Q.39. ***Switching/Routing*** is the job of Network Layer.

Q.40. ***Sliding Window*** line discipline where only some of the enquiries are acknowledged is more efficient then ***Stop-and-Wait***.

Q.41. 10BaseT can have a maximum segment length of ***100*** meters.

Q.42. All NICs are manufactured having unique ***Hardware Address***.

Q.43. Physical addressing, error control and access (to media) control are managed by the ***Data Link*** layer.

Q.44. An advantage of 10BaseT over 10Base2 is that ***10Base2 is maintained easier than 10Base2***.

Q.45. 'Preamble' field at the beginning of an ethernet frame is used for ***synchronization***.

Q.46. Bridges must have the following layers; ***Physical Layer 2 Data Link Layer***.

Q.47. 'Time To Live' field in an IP packet determines the ***number of hops (routers)*** it passes through before it is ***discarded***.

Q.48. A host running in an IP-network on the ethernet must be assigned these three numbers properly in order for the networking software to operate correctly; ***IP address, 2. Netmask, 3. Gateway address.***

Q.49. ARP is used on IEEE-802.3 networks in order to ***obtain ethernet addresses using the IP addresses***.

Q.50. The field named as 'Window Size' in a TCP segment header indicates ***the sliding window size***.

Q.51. Standard TCP services use some standard TSAP numbers known as ***well known ports numbers***.

Q.52. TCP is a ***connection*** oriented protocol while ***UDP*** is not.

Q.53. The three problems which limit a communication line are ***attenuation, distortion*** and ***noise***.

Q.54. In order for a 2400 baud modem to achieve 9.6 kbps the constellation diagram must have at least ***16*** distinct points.

Q.55. The ***filter*** with ***a bandwidth of 300-3400*** Hz at the end office for an ordinary voice phone line is removed for DSL operation.

Q.56. An ADSL modem and a ***splitter*** are the required equipment residing at the customer's premises for the ADSL system.

Q.57. Three of the BlueTooth profiles are ***Service Discovery, Serial Port, Cordless Telephony.***

Q.58. In piconets of BlueTooth, ***Frequency Hopping Spread Spectrum technique*** is used with ***1600***

Q.59. hops/sec and hop sequence is dictated by the master.

Q.60. Pseudoternary encoding technique is used on T and S interfaces of ISDN system in order to ***maintain synchronization during the long sequences of zeros***.

Q.61. PRI ISDN service carries ***23*** bearer and 1 data channels with a total of ***1544*** kbps.

Q.62. Bus topology usually requires ***terminators*** at both ends of the cable.

Q.63. In ***half-duple*****x** transmission mode both station can transmit and receive but not at the same time.

Q.64. Logical addressing and routing is the function of ***network*** layer.

Q.65. Mail services are being made available by ***application*** layer.

Q.66. Manchester coding is one of the ***polar*** encoding techniques which effectively eliminate DC component of the signal.

Q.67. In Diff. Manchester, the transition in the middle is used for ***synchronization***.

Q.68. The number of signal units per second is called ***baud-rate***.

Q.69. In QAM both ***amplitude*** and ***phase*** of the carrier signal are varied.

Q.70. In ***stop-and-wait*** flow control technique, every frame is acknowledged.

Q.71. Physical Layer deals with the ***physical and electrical*** specifications.

Q.72. ***Full-duplex*** transmission mode can be characterized simply as "two way simultaneous transmission/reception".

Q.73. All NICs are manufactured having unique ***ethernet address/MAC number***.

Q.74. 10BaseT can have a maximum segment length of ***100*** meters.

Q.75. The words and numbers '10', 'Base' and 'T' in 10BaseT respectively indicate ***10 Mbps baseband*** and ***twisted pair.***

Q.76. ***Carrier extension*** and ***Frame bursting*** are features added to 802.3 by the gigabit ethernet standards.

Q.77. 1000Base-T uses ***4*** pairs of Cat-5 UTP.

Q.78. ***Flooding*** is a packet routing method in which incoming packet is sent to every neighbor except where it came from.

Q.79. In ***Distance Vector Routing*** a router receives routing information from all of its neighbors and by using the knowledge about its distance to its neighbors it constructs its own routing table which in turn used by the router and distributed to the neighbors.

Q.80. ***Hierarchical Routing*** reduces the memory requirements at some penalty on the path optimality in large networks with large number of routers.

Q.81. If packets from a live audio/video source are to be distributed to multiple destinations we need to talk about ***Broadcast Routing.***

Q.82. The 802.11 configuration in which no central coordination is used for is called ***Distributed***

Q.83. ***Coordination Function***

Q.84. ***Multipath Reception*** is a problem in wireless systems, which deteriorates the received signal at the receiver.

Q.85. When there is no central coordination is employed in 802.11, channel access privileges (who transmits when) are determined by a protocol called ***Carrier Sense Multiple Access / Collision Avoidance***.

Q.86. In wireless ethernet, when stations directly talk to each other, a station wanting to transmit data when the channel is idle transmits ***Request To Send (RTS)*** frame first.

Q.87. The small clusters of stations communicating using Bluetooth are called ***Piconets***.

Q.88. There can be only ***seven*** active slaves in a Bluetooth station cluster.

Q.89. The master station in a Bluetooth cluster employs ***Time Division Multiplexing*** in order to communicate with slaves and send commands to them.

Q.90. The destination address field in a Bluetooth frame is ***three*** bits

Q.91. Bluetooth operates on ***2.4 G***Hz ISM band and its range is about ***10*** meters.

Q.92. The PSTN term ***Local Loop*** refers to the wiring between the customer and end office of the telecom company.

Q.93. In PSTN, in order for a modem customer to achieve ***56 kbps*** transmission rate, the connection to ISP must be ***digital***.

Q.94. QAM stands for ***Quadrature Amplitude Modulation*** which means changing both ***Amplitude*** and ***Phase*** of the carrier.

Q.95. In 128-QAM, one change in carrier transmits ***seven*** bits including parity.

Q.96. In DMT which is employed in DSL, of ***256*** frequency channels ***5*** are not used to prevent interference between voice and data signals.

Q.97. Original antialiasing filter is removed in DSL, but a pair of ***Low Pass Filter*** is still used to split voice and data channels in end office and in customer premises.

a. ISDN stands for ***Integrated Services Digital Network.***

Q.98. In ISDN two standard services are provided; ***BRI*** and ***PRI***.

Q.99. A B channel in ISDN service has the data rate of ***64 k***bps whereas a D channel carries ***16 k***bps.

Q.100. 23B+D configuration in ISDN is capable of carrying ***1544 k***bps.

Q.101. Two features brought by gigabit ethernet, ***Carrier Extension***, ***Frame Bursting*** are required to achieve 200 m in hub based networks.

Q.102. Fast ethernet on Cat-5 cable runs at most ***100*** meters.

Q.103. ***Hubs/Repeaters*** just repeat the incoming data and send it through all other connected lines.

Q.104. Different modes of light travel different total distances resulting in ***dispersion*** which causes distortion in ***multimode*** fibers at long distances.

Q.105. Multistation Access Unit is a term which refers to devices used to form a star topology in ***Token Ring***.

Q.106. FDDI stands for ***Fiber Distributed Data Interface***.

Q.107. In Token Ring there is at least one frame traveling around the ring, and it is called the ***Token***.

Q.108. The communication speed in FDDI is ***100 M***bps.

Q.109. Two common transmission rates in ATM are ***155.52 Mbps*** and ***622 Mbps***.

Q.110. In ATM systems ***delivery*** is not guaranteed but the ***order*** is.

Q.111. Since the ATM cell size is ***53*** bytes ATM routers can be designed to be entirely hardware (no software).

Q.112. In ATM, after the connection establishment between the source and destination, data always follow the same route which is called ***Virtual Circuit***.

Q.113. Fiber, among other common transmission media, has the best performance thanks to its ***high bandwidth*** and ***low EMI***.

Q.114. OSI stands for ***Open Systems Interface***.

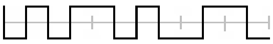
Q.115. Service-point addressing (different number for each program) is the task of ***transport*** layer in OSI 7- layer model.

Q.116. Signaling standards is in the interest of ***physical*** layer.

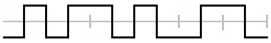
Q.117. Ethernet bridges consist of ***physical*** and ***data-link*** layer(s).

Q.118. In an ethernet network the interfaces are identified/addressed by their ***ethernet address/hardware address/MAC address***.

Q.119. The number of pins on a RJ-45 connector at the end of a cat-5 cable is ***8***.

Q.120. In Differential Manchester  represents the binary stream **001011**.

Q.121. In an ethernet frame specified by IEEE 802.3, preamble is used for ***synchronization***.

Q.122. In Manchester encoding  represents the binary stream **001101**.

Q.123. The destination address in an ethernet frame is ***48*** bits.

Q.124. The '***Time-To-Live***' field in IP-datagram determines the number of hops.

Q.125. ***Netmask*** is used to separate the network address from the IP address.

Q.126. The service that converts hostnames into host IP numbers is called ***Name Service***.

Q.127. The interface IP-numbers of a router between IP networks shall be selected from the ranges of the ***connected networks***.

Q.128. ***Routing table*** defines where to deliver the IP-packets when the destination is not in the same network.

Q.129. ***Exponential Back-off*** algorithm is used to determine the time to wait for another transmission attempt in CSMA-CD networks.

Q.130. Fast ethernet on twisted pair cable is referred as ***100***Base-***TX.***

Q.131. Fast ethernet on fiber runs at most ***2000*** meters.

Q.132. Bit rate is ***bits*** per second whereas the baud rate is ***symbols/changes*** per second.

Q.133. ADSL stands for ***Asymmetric Digital Subscriber Line***.

Q.134. Channel 0 in DMT of DSL is used for ***basic telephony service***.

Q.135. In DSL, a filter called ***splitter*** must be used in both customers' and Telco's premises.

Q.136. 23B+D configuration in ISDN is capable of carrying ***1544*** kbps.

Q.137. ***CSMA/CD*** is not used in wireless ethernet because most radio devices can not listen and transmit simultaneously.

Q.138. The RF communication technique in wireless ethernet, in which the communication frequency is periodically switched within a set of predetermined sequence, is called ***Frequency Hopping Spread Spectrum***.

Q.139. A Bluetooth device is expected to support the applications ***Generic Access*** and ***Service Discovery*** and the other profiles are optional.

Q.140. A collection of Bluetooth piconets is called ***scatternet***.

Q.141. In Differential Manchester encoding technique, the transition in the middle helps ***synchronization*** but reduces the ***transmission rate***.

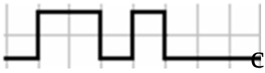
Q.142. In ***Link State Routing*** routing, a router obtains the distances of its neighbors and shares this information with all other routers it knows using specially designed packets.

Q.143. ARP is used on IEEE-802.3 networks in order to obtain ***ethernet address*** from ***IP address***

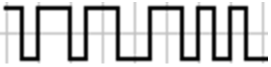
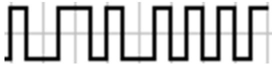
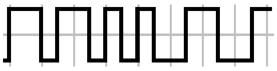
Q.144. In class-A IP-addresses ***24*** bits are reserved for host identification.

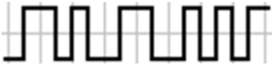
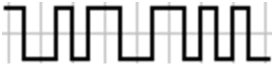
Multiple Choice Questions:--

Part 1

1. The binary stream which NRZ-I waveform  corresponds to is ……..
a) 01101111 b) 11010000 c) 10010111 **d)** 01011100 e) 01101000

2. The diff. Manchester waveform corresponding to the binary stream 01101000 is ………

a)  **b)**  c) 

d)  e) 

3. Repeaters operate similar to ……… which also just create electronic nodes to extend the cable length.
a) Ethernet hubs b) Bridges c) Ethernet switches d) Connectors e) Hoppers

4. Fast ethernet ……..
a) uses only optical-fiber so it is really fast
b) is not backward compatible with old ethernet.
c) runs 2000 meters with twisted pair cable **d)** switches are full-duplex
e) hubs are 10 Mbps

5. Which one of the following is incorrect?
a) Full-duplex means simultaneous transmission and reception.
b) Simplex device cannot talk back.
c) One of the reliability measures of networks is the robustness in catastrophe.
d) On a dedicated link there can only be two devices.
e) Half-duplex means only half of the cable is duplex.

4. ………. layer handles the physical addressing of communicating devices. a) Physical b) Logical **c)** Data link d) Network e) Transport

5. Gateways are similar to routers …………………………..
a) in the way they route incoming packets.
b) that they both use the same OSI layers. c) requiring ethernet addresses on all ports. d) except when working with IP packets.
e) which can be thought of advanced switches.

6. Class of an IP address can be determined from ………. and determines the ………. a) the first 3 quads -- network part.
b) the number of netmask bits -- network address.
c) the routing table -- destination IP address.
d) the first 4 bits -- number of hosts in the network.
e) the host bits -- netmask.

Part 2

1. What is the name of the algorithm used in CSMA/CD networks in order to avoid repeated collisions?

a) collision avoidance b) crash prevent **c)** exponential back-off
d) exponential collisions e) collision detection

2. In which of the following, are the received cells guaranteed to be in order?

a) ATM b) Wireless ethernet c) TCP/IP d) ISDN e) ADSL

3. Multimode fibers have more ……….. than single mode fibers at the receiving end. a) light **b)** dispersion c) speed d) wavelength e) cable length

4. What is the signal that travels around the token-ring and carries data called?

a) Packet b) cell c) frame **d)** token e) bus

5. What is the name of the fixed route established at the time of initial connection setup in ATM networks?

a) connection network b) data route **c)** virtual circuit d) ring e) VPN

6. What is the maximum length of Cat-5 cable in a fast ethernet network?

a) 180 m b) 200 m c) 1000 m d) 1200 m **e)** 100 m

7. Which one of the following is the lowest bit rate in ATM networks?

a) 155 Mbps b) 622 Mbps c) 1544 kbps d) 1000 Mbps e) 10 Mbps

8. QAM uses different ……… in order to transmit multiple bits in one signal change. a) frequencies b) pulses c) bit-rates **d)** amplitude-phase e) bits

9. Higher capacity channel in BRI service of ISDN is called ……. channel?

a) DSL **b)** B c) PRI d) NT1 e) D

10. What is the lowest discrete channel in discrete multitone signaling of DSL used for?

a) control **b)** voice c) data d) channel allocation e) unused

11. What does A in ADSL stand for?

a) Alternative b) Asynchronous c) Attenuated **d)** Asymmetric e) Additive

12. What is the type of operation in wireless ethernet when a management/arbitration device exists?

a) point coordination b) distributed coordination c) managed arbitration
d) management point e) access point

13. What does FHSS stand for in wireless communication?

a) Fairly Harmonic Signal Sink b) Frequency Halving Signal Spectrum
c) Frequency Hopping Spread Spectrum d) Fast High Simple Solution
e) Full Harmonic Signal Spectrum

14. Concatenation of multiple frames for a single transmission in gigabit ethernet is called ….

a) carrier extension b) combination c) flow control d) full frame

e) frame bursting

15. How many bits are used for addressing in gigabit ethernet?

a) 64 bits b) 48 bits c) 32 bits d) variable e) 128 bits

16. What is the maximum length of 100Base-FX segment?

a) 2000 m b) 100 m c) 185 m d) 5000 m e) 200 m

17. What is the netmask of the gateway interface in a sub-C network where only 14 hosts may exist and IP address of one of the hosts is 193.145.122.76?

a) 255.255.255.224 b) 193.145.122.15 c) 255.255.255.0

d) 255.255.255.240 e) 193.145.122.240

18. What is the name of routing type in which every incoming packet is sent to every neighbor router except the one from which the packet came?

a) distribution b) multicast c) link-state d) shortest-path e) flooding

19. The shortest routes from a router to all other routers make up a …….. because of the optimality principle.

a) short-node b) cut-tree c) sink-tree d) binary-tree e) tentative-route

20. What is the main difference of dynamic routing compared to static one?

a) real-time connection establishment b) dynamic routing table-updates

c) dynamic behavior against flooding d) multiple output routes

e) dynamic change of hardware position

Part 3

1. Which of the following does not have a Data-Link Layer?

a) Router b) Gateway c) Switch d) Bridge e) Repeater

2. How many network bits does125.140.128.16 have?

a) 8 b) 16 c) 24 d) 32 e) 1

3. How many cable segments does a fully connected network of 10 hosts have?

a) 100 b) 45 c) 90 d) 10 e) 81

4. Which specifications the Physical Layer deals with?

a) Physical-Electrical b) Logical-Electrical c) Capacity-Load

d) Routing-Switching e) Delivery-CSMA/CD

5. How many bytes are ethernet addresses in CSMA/CD networks?

a) 16 b) 5 c) 6 d) 48 e) 4

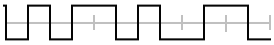
6. IP address of a host is found using its name by the service named ………………

a) Routing service b) Encription service c) Network Layer

d) Name service e) ARP service

7. How many pins of RJ-45 connectors are required when used in 10Base-T networks?

a) 8 **b)** 4 c) 2 d) 6 e) 3

8. Which binary stream is represented by  in NRZ-I?

a) 110100 b) 010110100110 **c)** 111101110101 d) 001011 e) 010110100110

9. What is a broadcast IP address?

a) IP address of a broadcasting host b) IP address with all hosts bits set to zero

c) IP address with all network bits set to 1 **d)** IP address with all host bits set to 1 e) IP address with last byte set to 255

10. Which OSI layer deals with physical addressing of the device?

a) Ethernet L. b) IP L. c) Address L. d) Physical L. **e)** Data Link L.

11. Which of the following can be a measure for the reliability of a computer network?

a) Mean time between failures b) Implemented protocol

c) Type of hosts d) Connections per unit time e) Protection software used on hosts

12. A dedicated physical link between two hosts is said to be?

a) Mesh Lin b) Shared link c) Simplex **d)** Point-to-point e) Duplex

13. Which of the following is a disadvantage for networks with bus topology?

a) Less cabling than ring b) Needs a central device c) Cheaper connectors

d) It is collision-free **e)** Cable faults down entire network

Which of the following is not an advantage of fiber-optic media for communication?

a) Has high bandwidth b) Has low EMI c) Has high security

d) Has low attenuation **e)** Requires high-tech equipment

15. In a router, what is the name of the structure that tells where to deliver IP-packets?

a) Router address **b)** Routing table c) Delivery table

d) Status table e) Default entry table

16. Which of the following is not one of the seven OSI layers?

a) Adaptation b) Session c) Network d) Transport e) Presentation

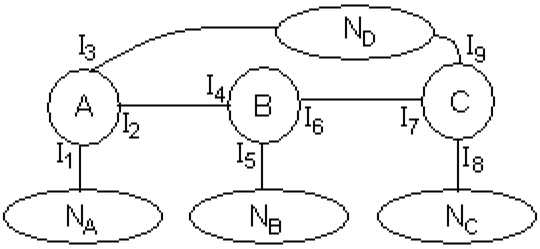
17. What is the purpose of preamble bits in an ethernet frame?

a) Initialization of ARP b) Pre-bit counting **c)** Synchronization

d) Error checking e) Destination address

Part 4

Answer the following according to the network given



18. What are the routing table entries in B, other than a possible default?

a) N_B ->I_5, N_C -> I_6, N_A -> I_4 b) A -> I_2, C -> I_7

c) N_A -> I_2, N_C -> I_7, N_D -> I_7 d) N_D -> I_3, N_A -> I_1, N_C -> I_8

e) N_D -> I_7, N_C -> I_8, N_A -> I_2, N_B -> I_5

19. Which one of the following is probably incorrect?

a).GW of a host in N_D is I_3 b) GW of a host in N_D is I_9 c) GW of a host in N_B is I_5

d) I_3 and I_6 are same   e) Netmasks of I_2-I_4 and I_6-I_7 are same

20. N_A is a sub class-C network. Which of the following is possible?

a) I_1=10.1.1.1   **b)** I_1=220.140.141.x   c) Netmask of N_A = 255.255.255.254

d) A host IP in N_A=72.16.141.19   e) A host in N_A has GW = 190.16.128.1

21. The use of "Spread Spectrum" techniques is pointless in ………?

a) baseband systems   b) 802.11x   c) wireless systems

d) satellite communications   e) between two radio stations   f) bluetooth

22. What is the name of the fixed route established at the time of initial connection setup in ATM networks?

a) VPN   b) virtual circuit   c) connection network

d) passage   e) data route   f) cell network

23. In ethernet networks, physical addressing is dealt with ……….

a) ethernet connector   b) logical layer   c) network layer

d) data-link layer   e) physical layer   f) physical address layer

24. What is the netmask of the gateway interface in a sub-network where only 14 hosts may exist and IP address of one of the hosts is 193.145.122.76?

a) 255.255.255.224   b) 193.145.122.15   c) 255.255.255.0

d) 193.145.122.240   e) 255.255.255.240   f) 255.255.0.0

25. What was the bit rate of IEEE 802.5 (token-ring) when it was first standardized?

a) 4.16 Mbps b) 10.2 Mbps   c) 100 Mbps   d) 12 Mbps   e) 144.5 Mbps f) 512 kbps

26. What are small bluetooth networks called?

a) chirpnets   b) minibus   c) BLAN   d) smalltalk   e) scatternet   f) piconet

27. What is the algorithm to handle collisions in ethernet networks?

a) Collision Pooling b) Time Division Mux c) exponential-backoff

d) ARP e) CSMA-CD f) CSMA-CA

28. The number of allowed hops for an IP packet is kept in the ……… field.

a) AHF b) IHL c) Frag. Off. d) TTL e) ToS f) PID

29. A fully connected network of 6 hosts requires a total of ………….. interfaces.

a) 30 b) 36 c) 35 d) 15 e) 18 f) 25

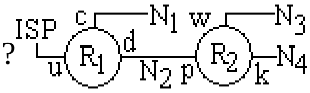
30. Which of the following is most important disadvantage of bus networks?

a) have many collisions b) expensive cables used c) one fault downs entire net.

d) less cabling required e) lower international standard f) higher traffic

31. How many channels are reserved for voice in DMT technique used in ADSL?

a) 1 b) 5 c) 251 d) 256 e) 4 f) 0

32. What are the router settings for N3 in R1 and default route in R2? 

a) N_3−>d, def−>p b) N_3−>w, def−>d c) N_3−>p, def−>u

d) N_3−>p, def−>d e) N_3−>N_2, def−>ISP f) N_3−>d, def−>d

33. Problems encountered in IP networks are usually reported with this protocol.

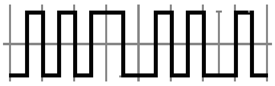
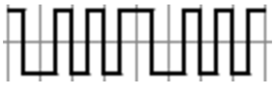
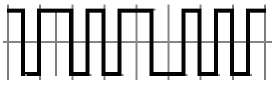
a) TCP b) SNMP c) SMTP d) HTTP e) ARP f) ICMP

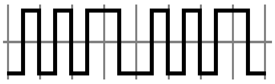
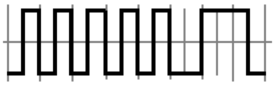
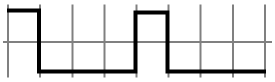
34. What are VLAN tags?

a) Error indicator attached to IP packets b) Routing information on ethernet adapters. c) A number in ethernet frame used to group switched hosts. d) Special MAC numbers

e) A special number on ethernet switches f) A special group number on networked hosts

35. Which of the following does represent 10001000 in Manchester coding?

a)  b)  c) 

d)  e)  f) 

36. What does C in DHCP stand for?

a) Connection b) Contention c) Collision d) Count e) Configuration f) Control

37. Which of the following is name of a digital channel used in ISDN?

a) 2B1Q b) TE1 c) BRI d) PBX e) B8ZS f) OFDM

38. What are the splitters used for in DSL?

a) Use two telephony device b) separate voice and data signals

c) use multiple telco lines d) use multiple computers

e) reject low frequency components for data f) for splitting power

39. Distance Vector Routing uses …….. to determine distances to its own neighbors. a) incoming vectors b) outgoing packets c) leaky bucket

d) shortest path e) flooding f) ping

40. Class of an IP address can be determined from …………

a) first 8 bits b) first 3 bytes c) last 8 bits d) first 3 bits e) first 4 bits f) netmask

41. A sequence of events in a TCP session between two hosts (H1,H2) is given below. Determine what is wrong with it.

1) H2: Connect, 2) H1: ConnectionRequest, 3) H1: Accept, 4) H1: SendData,

5) H2: DataArrival, 6) H2: Accept, 7) H1: Close

a) Host-1 cannot send data in step-4 b) Host-1 should not close yet in step-7

c) Accept is incorrect in step-6 d) Accept is incorrect in step-3 e) DataArrival is not called in step-5

42. In ----- packets/frames/cells guaranteed to be in order?

a) ATM b) Wireless ethernet c) TCP/IP d) ISDN e) ADSL

413. Multimode fibers have more ----- than single mode fibers at the receiving end.

a) light b) dispersion c) speed d) wavelength e) cable length

44. Higher capacity channel in BRI service of ISDN is called ----- channel?

a) DSL b) B c) PRI d) NT1 e) D

45. A in ADSL stands for ----- ?

a) Alternative b) Asynchronous c) Attenuated d) Asymmetric e) Additive

46. When a management/arbitration device exists in wireless ethernet networks, the type of operation is called ----- ?

a) access point b) distributed coordination c) managed arbitration

d) management point e) point coordination

47. VLAN tags are ---- ?

a) Error indicators attached to IP packets b) Routing information on ethernet adapters.

c) Numbers in ethernet frame used to group switched hosts.

d) Special numbers on ethernet switches

e) Special group numbers on networked hosts

48. Splitters used on telephony wires connected to DSL modems are used to ----- ?

a) Use two telephony device b) reject high frequency data signals

c) use multiple telco lines d) use multiple computers

e) reject low frequency components for data f) for splitting power

49. In ISDN 23B+D configuration is capable of carrying ----- kbps.

a) 10000 b) 100000 c) 52000 d) 1544 e) 622

50. Distance Vector Routing uses ----- to determine elements of its own distance vector. a) incoming vectors b) outgoing packets c) flooding d) shortest path e) echo packets

51. Manchester coding is one of the encoding techniques used to eliminate ----- ?

a) DC component of the signal b) High frequency components

c) common mode signals d) redundancy e) echo signals

52. ATM routers can be entirely hardware as ATM frame/packet/cell size is ----- bytes. a) 64 b) 72 c) 53 d) 1544 e) 46

53. ----- is a collection of Bluetooth piconets.

a) nanonets b) micronets c) picosubnets d) bluenets e) scatternet

54. Transmitting a packet on every route except the one where it came from is called -----. a) flooding b) broadcast c) multicast d) replicast e) routing

55. The filters used in telephony end offices limit high frequency components on telephone lines. What is its cut-off frequency when ADSL modems are used on customer lines?

a) 3000 Hz b) 3400 Hz c) 4500 Hz d) no filter e) 9600 Hz

56. Which of the following is name of a digital channel used in ISDN?

a) 2B1Q b) TE1 c) BRI d) PBX e) OFDM

57. FDDI stands for -----.

a) Fiber Distributed Data Interface b) Fully Distributed Device Interface c) Frequency Data Digital Interface d) Fiber Data Device Interconnection e) Faulty Data Dynamic Interface

58. Gigabit ethernet uses ----- bit hardware address.

a) 53 b) 48 c) 16 d) 32 e) 128

59. In IP networking, network and host addresses are separated using ----- ?

a) ARP b) TCP c) gateway d) netmask e) separator

60. In TCP sockets, DataArrival and ConnectionRequest are -----.

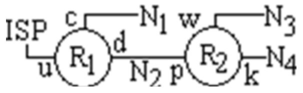
a) Messages b) Signals c) Properties d) Methods e) Events

61. Eight hosts making up a fully connected network has a total of ………….. dedicated lines. a) 8 b) 28 c) 64 d) 56 e) 49

62. DMT divides freq. band into 256 channels in ADSL, of which …. are used for voice. a) 1 b) 5 c) 251 d) 255 e) 4

63. What are the default route entries for R1 and R2 respectively?

a) ISP, ISP b) p, d c) ISP, d



d) u, p e) u, u

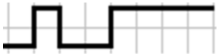
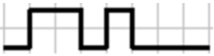
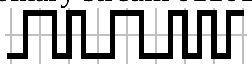
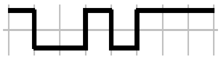
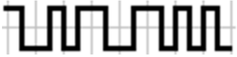
64. What is the netmask of the gateway interface in a sub-network where maximum of 25 hosts exist and IP address of one of the hosts is 192.168.1.1?
a) 192.168.1.0 b) 192.168.1.224 c) 192.168.1.27
d) 255.255.255.0 e) 255.255.255.224

65. In OSI systems, IP-routing is dealt with ……….
a) application layer b) data-link layer c) logical layer
d) network layer e) physical layer

66. Which of the following is not one of the seven OSI layers?
a) Network b) Session c) Routing d) Transport e) Presentation

67. What is a broadcast IP address for 193.140.141.128 / 26 ?
a) 193.140.141.128 b) 193.140.141.127 c) 255.255.255.63
d) 255.255.255.191 e) 193.140.141.191

68. The diff. NRZ-I waveform corresponding to the binary stream 01101000 is ………

a) 
b) 
c) 
d) 
e) 

69. Which one of the following is a job of the Physical Layer?
a) Switching, routing b) Signaling standards c) Segmentation-reassembly
d) Encryption, Compression e) Physical addressing

70. Gigabit ethernet uses ….. bit physical addresses.
a) 16 b) 32 c) 24 d) 48 e) 64

71. How many differential pairs do 10BaseT UTP interfaces need?
a) 6 b) 1 c) 2 d) 4 e) 8

72. ADSL stands for ….?
a) Asymmetric Digital Subscriber Line b) Asynchronous Digital Subscriber Line
c) Asynchronous Digital Serial Line d) Advanced Digital Serial Line
e) Asynchronous Delivery Serial Line

73. ARP finds out ….. from ……
a) IP, Hostname b) Hostname, IP c) Ethernet Addr, name
d) Gateway-IP, IP e) Ethernet Addr, IP

74. The word “splitter” usually refers to …. in networking?
a) A high-pass filter that extracts data signals in ADSL lines
b) A low-pass filter that extracts voice signal in ADSL lines
c) A multiport device that duplicates packets towards all connected lines
d) A connector box that splits customers’ premises from Telco’s
e) A BNC T connector for splitting coaxial lines.

75. What does the initial part of Ethernet frame that is designed for synchronization called?
a) Preample b) Syncpulses c) Header d) Synchronizer e) Preloader

76. Repeaters operate similar to … that extend the cable length.

a) Bridges b) Ethernet hubs c) Ethernet switches d) Connectors e) Routers

77. A typical socket-server application responds user requests using TCP over a specified port? What is the typical sequence in terms of socket functions on server side?
a) Listen, DataArrival, Accept, SendData, Close
b) Listen, ConnectionRequest, Accept, DataArrival, GetData, SendData, Close
c) Open, Connect, Accept, SendData, DetData, Close
d) ConnectionRequest, DataArrival, GetData, Accept, SendData, Close, Listen
e) Open, SendData, Listen, GetData, DataArrival, RetrieveData, Close
f) Listen, Open, GetData, SendData, Close

78. What are the DHCP pool types?
a) Automatic, Backed-up b) Configured, Free c) Large-pool, Limited
d) Dynamic, Recursive e) Static, Dynamic f) Switched, Bridged

79. What are the minimal OSI layers of a switching router?
a) Physical, DataLink, Network b) Transport, Network, DataLink
c) Application, VLAN, Switching d) Switching, Routing
e) Bridging, Routing, Physical f) DataLink, Physical

80. Connection between VLANs can be provided by …..?
a) VLAN-tags b) fiber cable c) router d) bridge e) switch f) VLAN-server

81. IEEE 8092.3 covers mostly …..?
a) CSMA/CD on Ethernet b) Token-Ring c) Token-Bus
d) ATM e) Network Layer f) DoD Layers

82. Four routers are connected as a ring. Including broadcast and network addresses, what is the minimum number of IP addresses needed/wasted for inter-router connections?
a) 4 b) 8 c) 6 d) 64 e) 32 f) 16

83. Which of the following is not a part of the IP datagram?
a) Fragment offset b) Packet identifier c) Type of service
d) TTL e) Dest. ethernet address f) Header length

84. FDDI stands for?
a) Fiber Digital Data Interconnect b) Full Duplex Data Integration
c) Fiber Data Domain Internet d) Fiber Distributed Data Interface
e) Fast Digital Data Internetwork f) Fully Distributed Domain Interface

85. How does source host know if a destination host is on the same IP network?
a) Looks up from a hosts list b) DHCP server tells it c) Via ARP
d) From network part of IP addr. e) If it responds then it is local f) it doesn't

86. Name server is a ….?
a) host with full of names b) short of an HTTP-server c) routing application
d) program for administrators. e) network protocol [✓ f)] program answering name queries

87. Which of the following is a well-known routing method?
a) multilink routing b) selective routing [✓ c)] multicast routing
d) link stared routing. e) short-test routing f) vector-metric routing

88. In DMT of ADSL how many channels are used for analog voice communication?
a) 5 [✓ b)] 1 c) 6 d) 2 e) 8 f) 256

89. ATM cells are ….. bytes?
[✓ a)] 53 b) 64K c) 1524 d) 2048 e) 48 f) 64

90. Which of the following is a common wireless communication technique?
a) Direct Synchronous Serial Shift [✓ b)] Frequency Hopping Spread Spectrum c)
Fully Hierarchical Shifted Spectrum d) Orthogonal Fast Direct Maintenance
e) Spectrum Balanced Orthogonal Shift f) Frequency Adaptive Discrete Multiplexing

91. Wireless ethernet and BlueTooth interfere as their operating frequencies coincide at ….. ?
a) 900MHz b) 5.GHz c) 1.9GHz d) 2.11GHz e) 4.9GHz [✓ f)] 2.4GHz

92. Bridges do not separate logical networks since …. ..
a) they are used to connect logical nets. b) this is done by network administrators.
c) what seems logical may not really be. d) CompNet students do not know how to. [✓ e)]
they work on physical addresses. f) logical addressing is costly

93. In a network with several switches bridges and routers, usually the routers are the bottlenecks as ………..
a) fiber cables do not work with routers. b)
they are fast and reliable.
[✓ c)] they have software layers to do the job and software is slow. d)
they require additional money to operate adequately.
e) they are the most disliked by the administrators.
f) bottles usually have narrow necks for TCP/IP packets.

94. What is ‘default route’ refers to for routers?
a) They usually end up in poorer departments and later in dusty shelves.
b) Branches and leaves of the network tree.
c) The most reliable route among all connected interfaces when the network is congested.
[✓ d)] The interface to send through when the destination is not known for an IP packet. e)
It is the administrators’ traffic monitoring software.
f) All traffic is routed to ‘default route’.

95. What is the nominal bit rate of ‘fast ethernet’?
a) 1000 MB/s b) 10 GB/s c) 10 MB/s d) 200 MB/s e) 1 Gb/s [✓ f)] 100 MB/s

Subjective Questions

Question No. 1

Define computer networks? Discuss various types of networks topologies in computer network. Also discuss various advantages and disadvantages of each topology.

Answer:-

‘‘Computer network" to mean a collection of autonomous computers interconnected by a single technology. Two computers are said to be interconnected if they are able to exchange information. The old model of a single computer serving all of the organization's computational needs has been replaced by one in which a large number of separate but interconnected computers do the job. These systems are called computer networks.

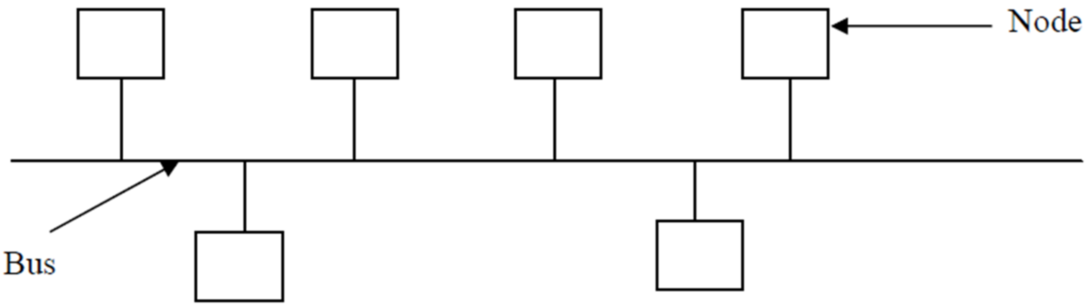
Network topologies:

Network topology defined as the logical connection of various computers in the network.

The six basic network topologies are: bus, ring, star, tree, mesh and hybrid.

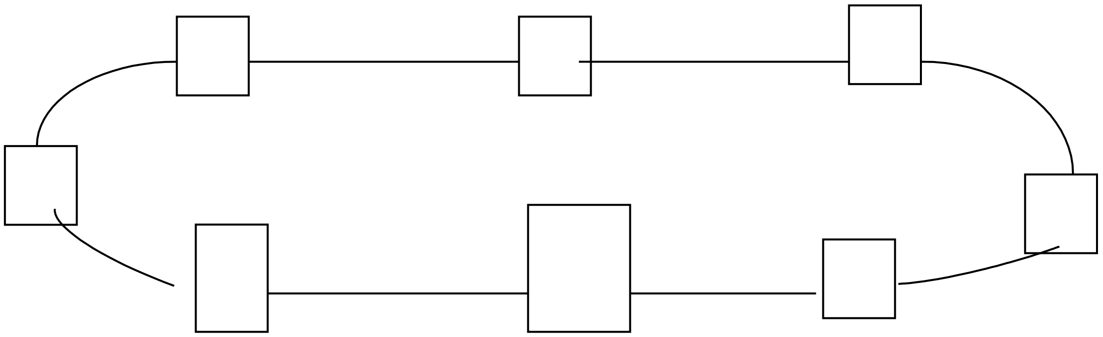
1. Bus Topology:

In bus topology all the computers are connected to a long cable called a bus. A node that wants to send data puts the data on the bus which carries it to the destination node. In this topology any computer can data over the bus at any time. Since, the bus is shared among all the computers. When two or more computers to send data at the same time, an arbitration mechanism are needed to prevent simultaneous access to the bus.



BUS TOPOLOGY

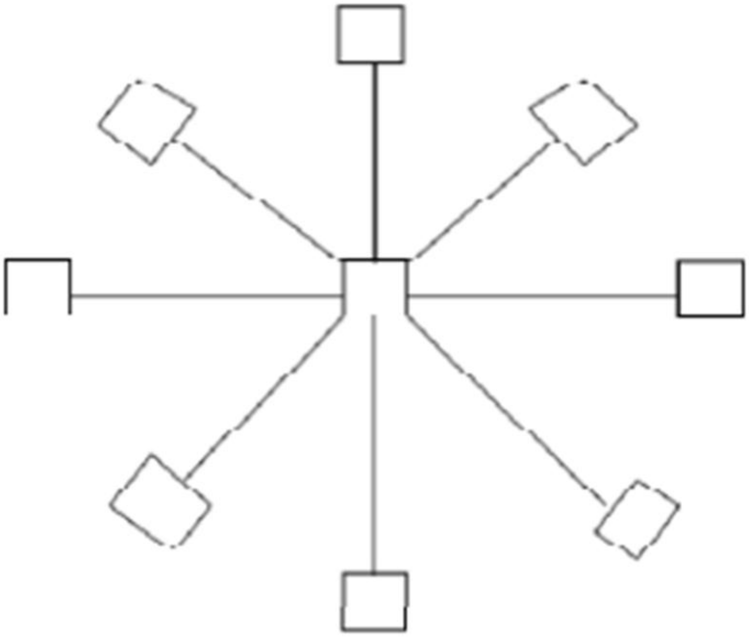
A bus topology is easy to install but is not flexible i.e., it is difficult to add a new node to bus. In addition to this the bus stops functioning even if a portion of the bus breaks down. It is also very difficult to isolate fault.



2. Ring Topology:
In ring topology, the computers are connected in the form of a ring. Each node has exactly two adjacent neighbors. To send data to a distant node on a ring it passes through many intermediate nodes to reach to its ultimate destination.

A ring topology is as to install and reconfigure. In this topology, fault isolation is easy because a signal that circulates all the time in a ring helps in identifying a faulty node. The data transmission takes place in only one direction. When a node fails in ring, it breaks down the whole ring. To overcome this drawback some ring topologies use dual rings. The topology is not useful to connect large number of computers.

3. Star Topology:
In star topology all the nodes are connected to a central node called a hub. A node that wants to send some six data to some other node on the network, send data to a hub which in turn sends it the destination node. A hub plays a major role in such networks.



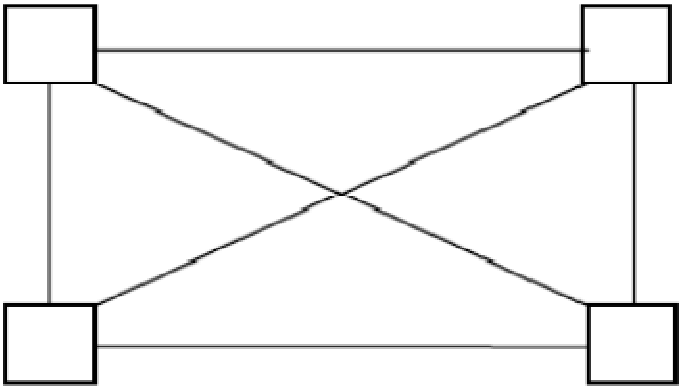
STAR TOPOLOGY

Star topology is easy to install and reconfigure. If a link fails then it separates the node connected to link from the network and the network continues to function. However, if the hub goes down, the entire network collapses.

4. Tree Topology:
Tree topology is a hierarchy of various hubs. The entire nodes are connected to one hub or the other. There is a central hub to which only a few nodes are connected directly. The central hub, also called active hub, looks at the incoming bits and regenerates them so that they can traverse over longer distances. The secondary hubs in tree topology may be active hubs or passive hubs. The failure of a transmission line separates a node from the network.

5. Mesh Topology:
A mesh topology is also called complete topology. In this topology, each node is connected directly to every oilier node in the network. That is if there are n nodes then there would be n(n — 1)/2 physical links in the network.

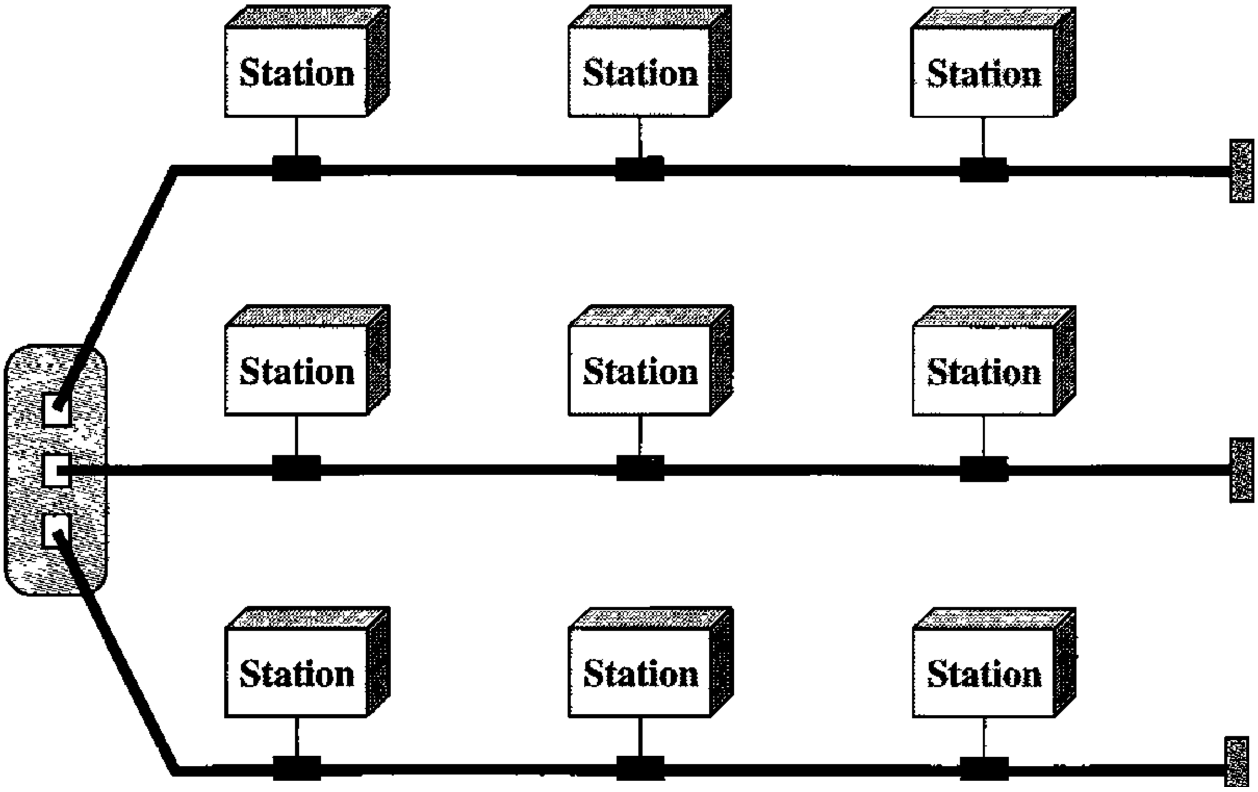
As there are dedicated links, the topology does not have congestion problems. Further it does not need a special Media Access Control (MAC) protocol to prevent simultaneous access to the transmission media since links are dedicated, not shared. The topology also provides data security. The network can continue to function even in the failure of one of the links. Fault identification is also easy. The main disadvantage of mesh topology is the complexity of the network and the cost associated with the cable length. The mesh topology is not useful for medium to large networks.



MESH TOPOLOGY

6. Hybrid Topology:

Hybrid topology is formed by connecting two or more topologies together. For example, hybrid topology can be created by using the bus, star and ring topologies



HYBRID TOPOLOGY

Question No. 2
What are the applications of Computer Networks?

Answer:-

1. Information:
One of the applications of computer networks is the ability to provide access to remote information.

- Pay bills; carry out transactions on bank accounts etc.
- Shop from home by inspecting the catalogs of thousands of companies available online.
- Ask the newspaper for full information about your interesting topics such as corrupt politicians, big fires, football and so on.
- Access information about health, science, art, business, cooking, sports, travel, and government and so on. All this is available on the information systems like the World Wide Web (WWW).

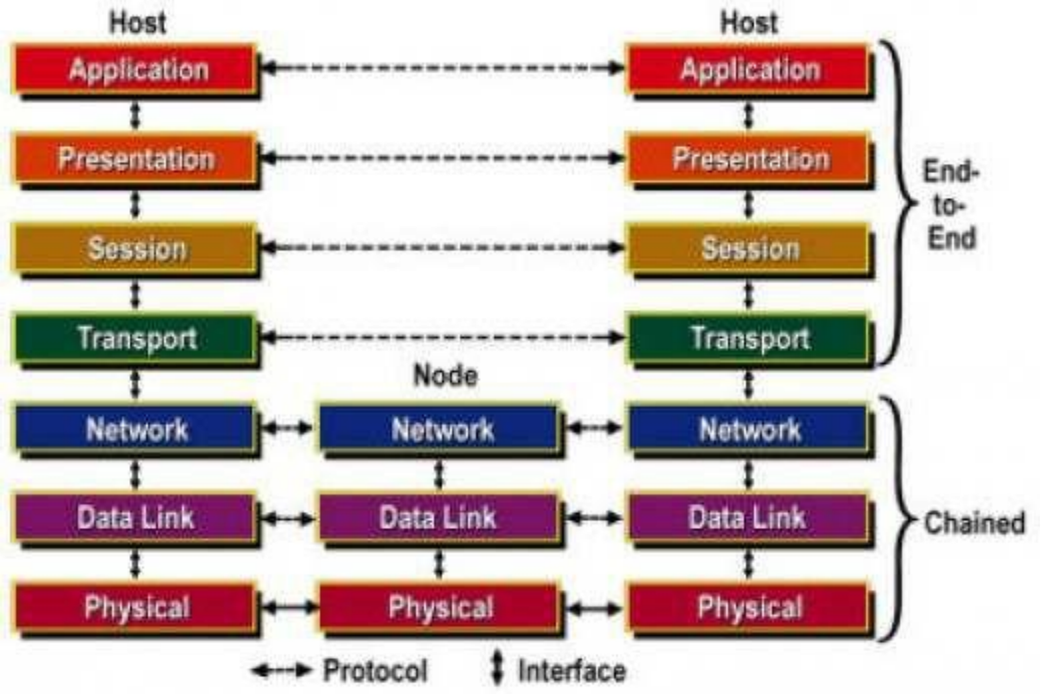
2. Communication:
The popular application of computer networks is electronic mail or e-mail which widely used by millions of people to send and receive text messages. With real-time e-mail, remote users can Communicate even by see and hear each other at the same time. It is also possible to have virtual meetings called videoconference on-line among remote users.

3. Entertainment:
A huge and growing application is entertainment. It entertains people by allowing video demand, and has multiple real-time games etc.

Question No. 3
What is OSI Model? Explain the functions and protocols and services of each layer?

Answer: - The OSI Reference Model:
The OSI model (minus the physical medium) is shown in Fig 4. This model is based on a proposal developed by the International Standards Organization (ISO) as a first step toward international standardization of the protocols used in the various layers (Day and Zimmermann, 1983). It was revised in 1995(Day, 1995). The model is called the ISO-OSI (Open Systems Interconnection) Reference Model because it deals with connecting open systems—that is, systems that are open for communication with other systems. The OSI model has seven layers. The principles that were applied to arrive at the seven layers can be briefly summarized as follows:

1. A layer should be created where a different abstraction is needed.
2. Each layer should perform a well-defined function.
3. The function of each layer should be chosen with an eye toward defining internationally standardized protocols.
4. The layer boundaries should be chosen to minimize the information flow across the interfaces.
5. The number of layers should be large enough that distinct functions need not be thrown together in the same layer out of necessity and small enough that the architecture does not become unwieldy.



THE OSI REFERENCE MODEL:

1. The Physical Layer:
The physical layer is concerned with transmitting raw bits over a communication channel. The design issues have to do with making sure that when one side sends a 1 bit, it is received by the other side as a 1 bit, not as a 0 bit.

2. The Data Link Layer:
The main task of the data link layer is to transform a raw transmission facility into a line that appears free of undetected transmission errors to the network layer. It accomplishes this task by having the sender break up the input data into data frames (typically a few hundred or a few thousand bytes) and transmits the frames sequentially. If the service is reliable, the receiver confirms correct receipt of each frame by sending back an acknowledgement frame. Another issue that arises in the data link layer (and most of the higher layers as well) is how to keep a fast transmitter from drowning a slow receiver in data. Some traffic regulation mechanism is often needed to let the transmitter know how much buffer space the receiver has at the moment. Frequently, this flow regulation and the error handling are integrated.

3. The Network Layer:
The network layer controls the operation of the subnet. A key design issue is determining how packets are routed from source to destination. Routes can be based on static tables that are "wired into" the network and rarely changed. They can also be determined at the start of each conversation, for example, a terminal session (e.g., a login to a remote machine). Finally, they can be highly dynamic, being determined anew for each packet, to reflect the current network load. If too many packets are present in the subnet at the same time, they will get in one another's way, forming bottlenecks. The control of such congestion also belongs to the network layer. More generally, the quality of service provided (delay, transit time, jitter, etc.) is also a network layer issue. When a packet has to travel from one network to another to get to its destination,

many problems can arise. The addressing used by the second network may be different from the first one. The second one may not accept the packet at all because it is too large. The protocols may differ, and so on. It is up to the network layer to overcome all these problems to allow heterogeneous networks to be interconnected. In broadcast networks, the routing problem is simple, so the network layer is often thin or even nonexistent.

4. The Transport Layer:

The basic function of the transport layer is to accept data from above, split it up into smaller units if need be, pass these to the network layer, and ensure that the pieces all arrive correctly at the other end. Furthermore, all this must be done efficiently and in a way that isolates the upper layers from the inevitable changes in the hardware technology. The transport layer also determines what type of service to provide to the session layer, and, ultimately, to the users of the network. The most popular type of transport connection is an error-free point-to-point channel that delivers messages or bytes in the order in which they were sent. However, other possible kinds of transport service are the transporting of isolated messages, with no guarantee about the order of delivery, and the broadcasting of messages to multiple destinations. The type of service is determined when the connection is established. The transport layer is a true end-to-end layer, all the way from the source to the destination. In other words, a program on the source machine carries on a conversation with a similar program on the destination machine, using the message headers and control messages. In the lower layers, the protocols are between each machine and its immediate neighbors, and not between the ultimate source and destination machines, which may be separated by many routers.

5. The Session Layer:

The session layer allows users on different machines to establish sessions between them. Sessions offer various services, including dialog control (keeping track of whose turn it is to transmit), token management (preventing two parties from attempting the same critical operation at the same time), and synchronization (check pointing long transmissions to allow them to continue from where they were after a crash).

6. The Presentation Layer:

The presentation layer is concerned with the syntax and semantics of the information transmitted. In order to make it possible for computers with different data representations to communicate, the data structures to be exchanged can be defined in an abstract way, along with a standard encoding to be used "on the wire." The presentation layer manages these abstract data structures and allows higher-level data structures (e.g., banking records), to be defined and exchanged.

7. The Application Layer:

The application layer contains a variety of protocols that are commonly needed by users. One widely-used application protocol is HTTP (Hypertext Transfer Protocol), which is the basis for the World Wide Web. When a browser wants a Web page, it sends the name of the page it wants to the server using HTTP. The server then sends the page back. Other application protocols are used for file transfer, electronic mail, and network news.

Question No. 4

Explain the following:-

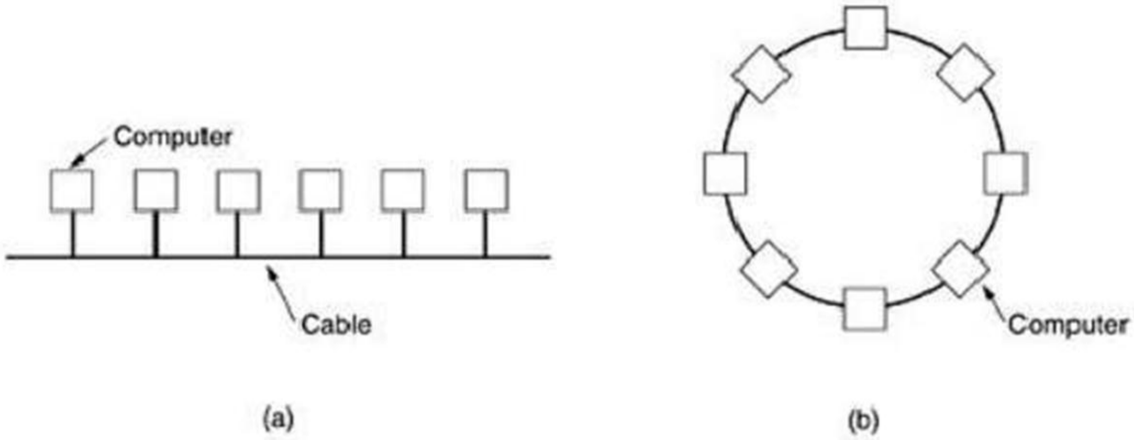
a) LAN
b) MAN
c) WAN
d) ARPANET

Answer: - a) LAN-Local Area Networks

Local area networks, generally called LANs, are privately-owned networks within a single building or campus of up to a few kilometers in size. They are widely used to connect personal computers and workstations in company offices and factories to share resources (e.g., printers) and exchange information. LANs are distinguished from other kinds of networks by three characteristics:

(1) Their size,
(2) Their transmission technology, and
(3) Their topology.

LANs are restricted in size, which means that the worst-case transmission time is bounded and known in advance. Knowing this bound makes it possible to use certain kinds of designs that would not otherwise be possible. It also simplifies network management. LANs may use a transmission technology consisting of a cable to which all the machines are attached, like the telephone company party lines once used in rural areas. Traditional LANs run at speeds of 10 Mbps to 100 Mbps, have low delay (microseconds or nanoseconds), and make very few errors.



Newer LANs operate at up to 10 Gbps various topologies are possible for broadcast LANs. Figure1 shows two of them. In a bus (i.e., a linear cable) network, at any instant at most one machine is the master and is allowed to transmit. All other machines are required to refrain from sending. An arbitration mechanism is needed to resolve conflicts when two or more machines want to transmit simultaneously. The arbitration mechanism may be centralized or distributed. IEEE 802.3, popularly called Ethernet, for example, is a bus-based broadcast network with decentralized control, usually operating at 10 Mbps to 10 Gbps. Computers on an Ethernet can transmit whenever they want to; if two or more packets collide, each computer just waits a random time and tries again later.

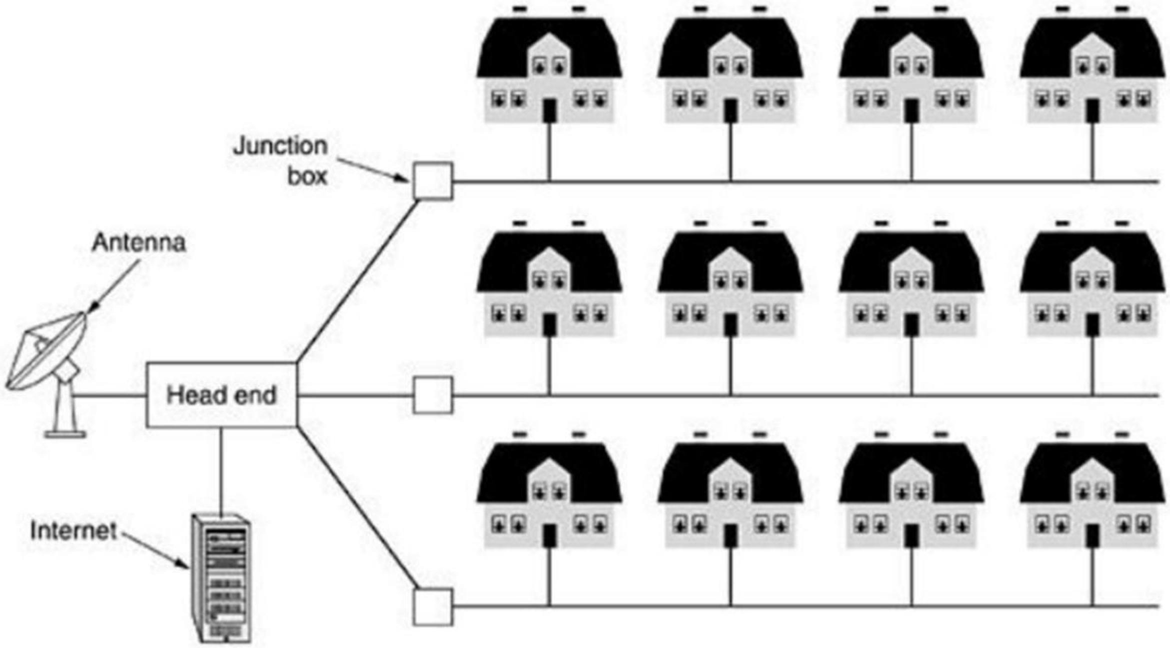
A second type of broadcast system is the ring. In a ring, each bit propagates around on its own, not waiting for the rest of the packet to which it belongs. Typically, each bit circumnavigates the entire ring in the time it takes to transmit a few bits, often before the complete packet has even been transmitted. As with all other broadcast systems, some rule is needed for arbitrating simultaneous accesses to the ring. Various methods, such as having the machines take turns, are in use. IEEE 802.5 (the IBM token ring), is a ring-based LAN operating at 4 and 16 Mbps. FDDI is another example of a ring network.

b) Metropolitan Area Network:

A metropolitan area network, or MAN, covers a city. The best-known example of a MAN is the cable television network available in many cities. This system grew from earlier community antenna systems used in areas with poor over-the-air television reception. In these early systems, a large antenna was placed on top of a nearby hill and signal was then piped to the subscribers' houses.

At first, these were locally-designed, ad hoc systems. Then companies began jumping into the business, getting contracts from city governments to wire up an entire city. The next step was television programming and even entire channels designed for cable only. Often these channels were highly specialized, such as all news, all sports, all cooking, all gardening, and so on. But from their inception until the late 1990s, they were intended for television reception only.

To a first approximation, a MAN might look something like the system shown in Fig.2. In this figure both television signals and Internet are fed into the centralized head end for subsequent distribution to people's homes.



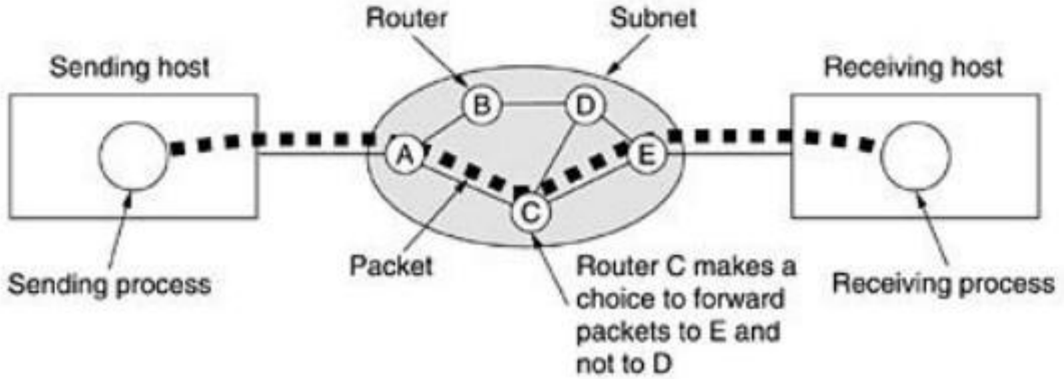
Cable television is not the only MAN. Recent developments in high-speed wireless Internet access resulted in another MAN, which has been standardized as IEEE 802.16. A MAN is implemented by a standard called DQDB (Distributed Queue Dual Bus) or IEEE 802.16. DQDB has two unidirectional buses (or cables) to which all the computers are attached.

c) Wide Area Network:

A wide area network, or WAN, spans a large geographical area, often a country or continent. It contains a collection of machines intended for running user (i.e., application) programs. These machines are called as hosts. The hosts are connected by a communication subnet, or just subnet for short. The hosts are owned by the customers (e.g., people's personal computers), whereas the communication subnet is typically owned and operated by a telephone company or Internet service provider. The job of the subnet is to carry messages from host to host, just as the telephone system carries words from speaker to listener. Separation of the pure communication aspects of the network (the subnet) from the application aspects (the hosts), greatly simplifies the complete network design.

In most wide area networks, the subnet consists of two distinct components: transmission lines and switching elements. Transmission lines move bits between machines. They can be made of copper wire, optical fiber, or even radio links. In most WANs, the network contains numerous transmission lines, each one connecting a pair of routers. If two routers that do not share a transmission line wish to communicate, they must do this indirectly, via other routers. When a packet is sent from one router to another via one or more intermediate routers, the packet is received at each intermediate router in its entirety, stored there until the required output line is free, and then forwarded. A subnet organized according to this principle is called a store-and-forward or packet-switched subnet. Nearly all wide area networks (except those using satellites) have store-and-forward subnets. When the packets are small and all the same size, they are often called cells.

The principle of a packet-switched WAN is so important. Generally, when a process on some host has a message to be sent to a process on some other host, the sending host first cuts the message into packets, each one bearing its number in the sequence. These packets are then injected into the network one at a time in quick succession. The packets are transported individually over the network and deposited at the receiving host, where they are reassembled into the original message and delivered to the receiving process. A stream of packets resulting from some initial message is illustrated. In this figure, all the packets follow the route ACE, rather than ABDE or ACDE. In some networks all packets from a given message must follow the same route; in others each packed is routed separately. Of course, if ACE is the best route, all packets may be sent along it, even if each packet is individually routed.

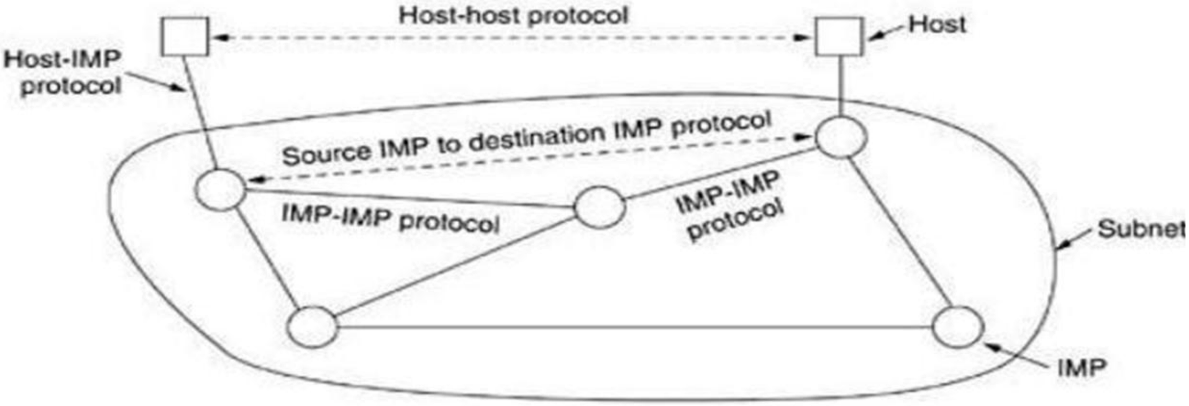


Not all WANs are packet switched. A second possibility for a WAN is a satellite system. Each router has an antenna through which it can send and receive. All routers can hear the output from the satellite, and in some cases they can also hear the upward transmissions of their fellow routers to the satellite as well. Sometimes the routers are connected to a substantial point-to-point subnet, with only some of them having a satellite antenna. Satellite networks are inherently broadcast and are most useful when the broadcast property is important.

d) ARPANET:

The subnet would consist of minicomputers called IMPs (Interface Message Processors) connected by 56-kbps transmission lines. For high reliability, each IMP would be connected to at least two other IMPs. The subnet was to be a datagram subnet, so if some lines and IMPs were destroyed, messages could be automatically rerouted along alternative paths.

Each node of the network was to consist of an IMP and a host, in the same room, connected by a short wire. A host could send messages of up to 8063 bits to its IMP, which would then break these up into packets of at most 1008 bits and forward them independently toward the destination. Each packet was received in its entirety before being forwarded, so the subnet was the first electronic store-and-forward packet-switching network.



ORIGINAL ARPANET DESIGN

Each node of the network was to consist of an IMP and a host, in the same room, connected by a short wire. A host could send messages of up to 8063 bits to its IMP, which would then break these up into packets of at most 1008 bits and forward them independently toward the destination. Each packet was received in its entirety before being forwarded, so the subnet was the first electronic store-and-forward packet-switching network.

ARPA then put out a tender for building the subnet. Twelve companies bid for it. After evaluating all the proposals, ARPA selected BBN, a consulting firm in Cambridge, Massachusetts, and in December 1968, awarded it a contract to build the subnet and write the subnet software. BBN chose to use specially modified Honeywell DDP-316 minicomputers with 12K 16-bit words of core memory as the IMPs. The IMPs did not have disks, since moving parts were considered unreliable. The IMPs were interconnected by 56-kbps lines leased from telephone companies. Although 56 kbps is now the choice of teenagers who cannot afford ADSL or cable, it was then the best money could buy.

The software was split into two parts: subnet and host. The subnet software consisted of the IMP end of the host-IMP connection, the IMP-IMP protocol, and a source IMP to destination IMP protocol designed to improve reliability. The original ARPANET design is shown in Fig.10. Outside the subnet, software was also needed, namely, the host end of the host-IMP connection, the host-host protocol, and the application software. It soon became clear that BBN felt that when it had accepted a message on a host-IMP wire and placed it on the host-IMP wire at the destination, its job was done.

Question No. 5

What is TCP/IP Model? Explain the functions and protocols and services of each layer? Compare it with OSI Model.

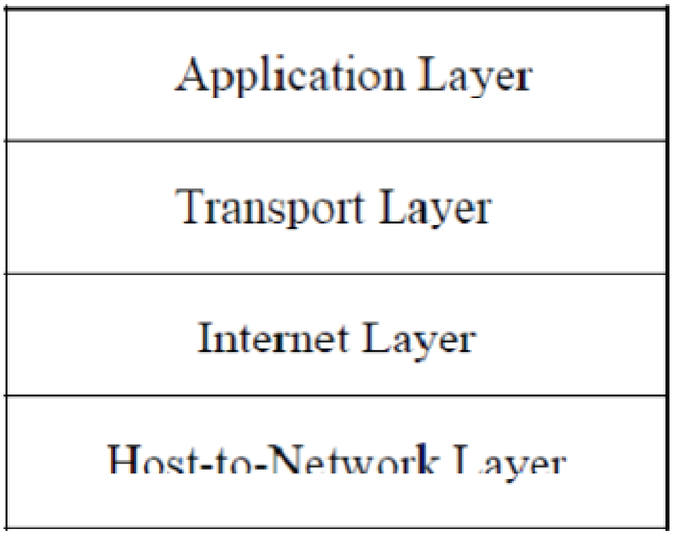
Answer: - The TCP/IP MODEL:-

The TCP/IP reference model was developed prior to OSI model. The major design goals of this model were,

1. To connect multiple networks together so that they appear as a single network.
2. To survive after partial subnet hardware failures.
3. To provide a flexible architecture.

Unlike OSI reference model, TCP/IP reference model has only 4 layers. They are,

1. Host-to-Network Layer
2. Internet Layer
3. Transport Layer
4. Application Layer



TCP/IP Reference model

1. Host-to-Network Layer:

The TCP/IP reference model does not really say much about what happens here, except to point out that the host has to connect to the network using some protocol so it can send IP packets to it. This protocol is not defined and varies from host to host and network to network.

2. Internet Layer:

This layer, called the internet layer, is the linchpin that holds the whole architecture together. Its job is to permit hosts to inject packets into any network and have they travel independently to the destination (potentially on a different network). They may even arrive in a different order than they were sent, in which case it is the job of higher layers to rearrange them, if in-order delivery is desired. Note that "internet" is used here in a generic sense, even though this layer is present in the Internet.

The internet layer defines an official packet format and protocol called IP (Internet Protocol). The job of the internet layer is to deliver IP packets where they are supposed to go. Packet routing is clearly the major issue here, as is avoiding congestion. For these reasons, it is reasonable to say that the TCP/IP internet layer is similar in functionality to the OSI network layer.

3. The Transport Layer:

The layer above the internet layer in the TCP/IP model is now usually called the transport layer. It is designed to allow peer entities on the source and destination hosts to carry on a conversation, just as in the OSI transport layer. Two end-to-end transport protocols have been defined here. The first one, TCP (Transmission Control Protocol), is a reliable connection-oriented protocol that allows a byte stream originating on one machine to be delivered without error on any other machine in the internet. It fragments the incoming byte stream into discrete messages and passes each one on to the internet layer. At the destination, the receiving TCP process reassembles the received messages into the output stream. TCP also handles flow control to make sure a fast sender cannot swamp a slow receiver with more messages than it can handle.

The second protocol in this layer, UDP (User Datagram Protocol), is an unreliable, connectionless protocol for applications that do not want TCP's sequencing or flow control and wish to provide their own. It is also widely used for one-shot, client-server-type request-reply queries and applications in which prompt delivery is more important than accurate delivery, such as transmitting speech or video. Since the model was developed, IP has been implemented on many other networks.

4. The Application Layer:

The TCP/IP model does not have session or presentation layers. On top of the transport layer is the application layer. It contains all the higher-level protocols. The early ones included virtual terminal (TELNET), file transfer (FTP), and electronic mail (SMTP), as shown in Fig.6.2. The virtual terminal protocol allows a user on one machine to log onto a distant machine and work there. The file transfer protocol provides a way to move data efficiently from one machine to another. Electronic mail was originally just a kind of file transfer, but later a specialized protocol (SMTP) was developed for it. Many other protocols have been added to these over the years: the Domain Name System (DNS) for mapping host names onto their network addresses, NNTP, the protocol for moving USENET news articles around, and HTTP, the protocol for fetching pages on the World Wide Web, and many others.

Comparison of the OSI and TCP/IP Reference Models:
The OSI and TCP/IP reference models have much in common. Both are based on the concept of a stack of independent protocols. Also, the functionality of the layers is roughly similar. For example, in both models the layers up through and including the transport layer are there to provide an end-to-end, network-independent transport service to processes wishing to communicate. These layers form the transport provider. Again in both models, the layers above transport are application-oriented users of the transport service. Despite these fundamental similarities, the two models also have many differences.

Three concepts are central to the OSI model:
1. Services.
2. Interfaces.
3. Protocols.

Probably the biggest contribution of the OSI model is to make the distinction between these three concepts explicit. Each layer performs some services for the layer above it. The service definition tells what the layer does, not how entities above it access it or how the layer works. It defines the layer's semantics. A layer's interface tells the processes above it how to access it. It specifies what the parameters are and what results to expect. It, too, says nothing about how the layer works inside.

Finally, the peer protocols used in a layer are the layer's own business. It can use any protocols it wants to, as long as it gets the job done (i.e., provides the offered services). It can also change them at will without affecting software in higher layers. The TCP/IP model did not originally clearly distinguish between service, interface, and protocol, although people have tried to retrofit it after the fact to make it more OSI-like. For example, the only real services offered by the internet layer are SEND IP PACKET and RECEIVE IP PACKET.

As a consequence, the protocols in the OSI model are better hidden than in the TCP/IP model and can be replaced relatively easily as the technology changes. Being able to make such changes is one of the main purposes of having layered protocols in the first place. The OSI reference model was devised before the corresponding protocols were invented.

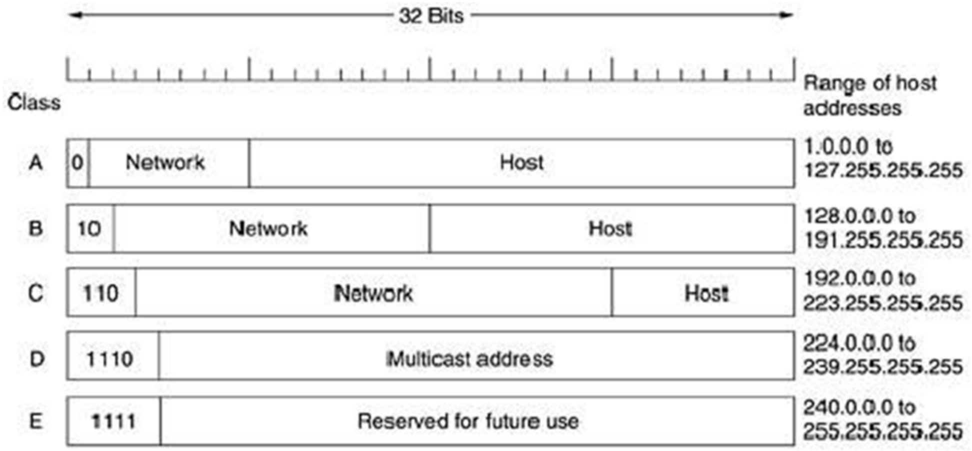
This ordering means that the model was not biased toward one particular set of protocols, a fact that made it quite general. The downside of this ordering is that the designers did not have much experience with the subject and did not have a good idea of which functionality to put in which layer.
Another difference is in the area of connectionless versus connection-oriented communication. The OSI model supports both connectionless and connection-oriented communication in the network layer, but only connection-oriented communication in the transport layer, where it counts (because the transport service is visible to the users). The TCP/IP model has only one mode in the network layer (connectionless) but supports both modes in the transport layer, giving the users a choice. This choice is especially important for simple request-response protocols.

Question No. 6

What is IP addressing? How it is classified? How is subnet addressing is performed?

Answer: -

Every host and router on the Internet has an IP address, which encodes its network number and host number. The combination is unique: in principle, no two machines on the Internet have the same IP address. All IP addresses are 32 bits long and are used in the Source address and Destination address fields of IP packets. It is important to note that an IP address does not actually refer to a host. It really refers to a network interface, so if a host is on two networks, it must have two IP addresses. However, in practice, most hosts are on one network and thus have one IP address.


32 Bits
Class
Range of host addresses
A
0
Network
Host
1.0.0.0 to 127.255.255.255
B
10
Network
Host
128.0.0.0 to 191.255.255.255
C
110
Network
Host
192.0.0.0 to 223.255.255.255
D
1110
Multicast address
224.0.0.0 to 239.255.255.255
E
1111
Reserved for future use
240.0.0.0 to 255.255.255.255

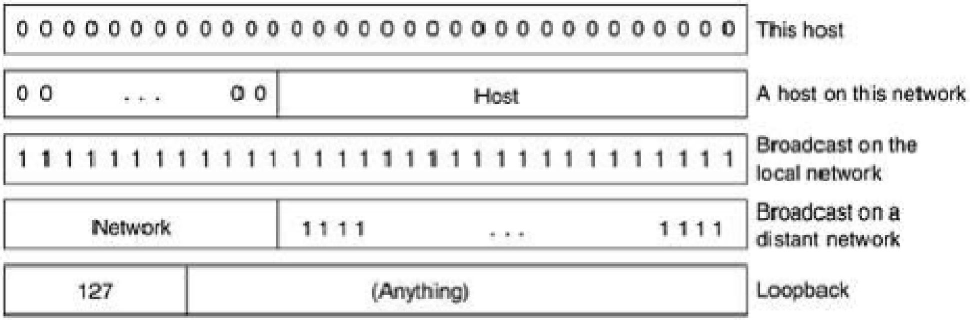
For several decades, IP addresses were divided into the five categories

The class A, B, C, and D formats allow for up to 128 networks with 16 million hosts each, 16,384 networks with up to 64K hosts, and 2 million networks (e.g., LANs) with up to 256 hosts each (although a few of these are special). Also supported is multicast, in which a datagram is directed to multiple hosts. Addresses beginning with 1111 are reserved for future use. Over 500,000 networks are now connected to the Internet, and the number grows every year. Network numbers are managed by a nonprofit corporation called ICANN (Internet Corporation for Assigned Names and Numbers) to avoid conflicts. In turn, ICANN has delegated parts of the address space to various regional authorities, which then dole out IP addresses to ISPs and other companies.

Network addresses, which are 32-bit numbers, are usually written in dotted decimal notation. In this format, each of the 4 bytes is written in decimal, from 0 to 255. For example, the 32-bit hexadecimal address C0290614 is written as 192.41.6.20. The lowest IP address is 0.0.0.0 and the highest is 255.255.255.255. The values 0 and -1 (all 1s) have special meanings, as shown in Fig. 5-56. The value 0 means this network or this host. The value of -1 is used as a broadcast address to mean all hosts on the indicated network.

The IP address 0.0.0.0 is used by hosts when they are being booted. IP addresses with 0 as network number refer to the current network. These addresses allow machines to refer to their own network without knowing its number (but they have to know its class to know how many 0s to include). The address consisting of all 1s allows broadcasting on the local network, typically a

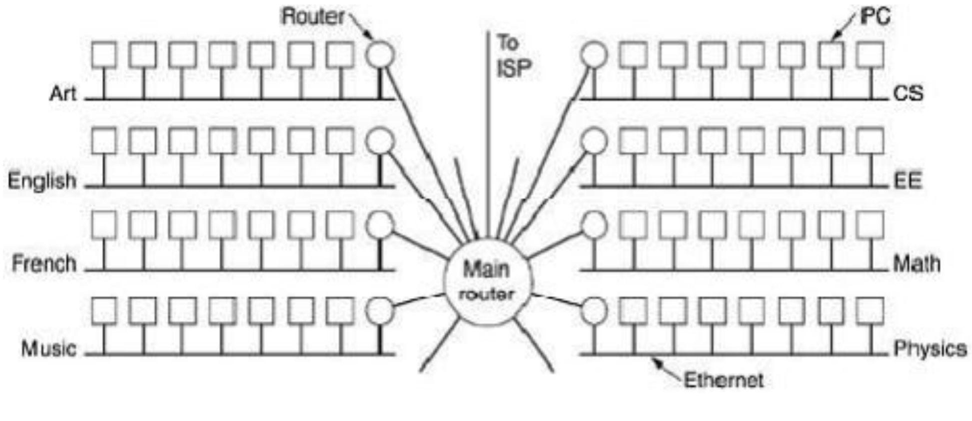
LAN. The addresses with a proper network number and all 1s in the host field allow machines to send broadcast packets to distant LANs anywhere in the Internet (although many network administrators disable this feature). Finally, all addresses of the form 127.xx.yy.zz are reserved for loopback testing. Packets sent to that address are not put out onto the wire; they are processed locally and treated as incoming packets. This allows packets to be sent to the local network without the sender knowing its number.



SOME SPECIAL IP ADDRESS

Sub netting

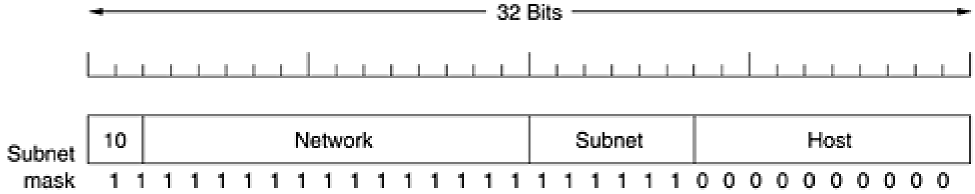
As we have seen, all the hosts in a network must have the same network number. This property of IP addressing can cause problems as networks grow. For example, consider a university that started out with one class B network used by the Computer Science Dept. for the computers on its Ethernet. A year later, the Electrical Engineering Dept. wanted to get on the Internet, so they bought a repeater to extend the CS Ethernet to their building. As time went on, many other departments acquired computers and the limit of four repeaters per Ethernet was quickly reached. A different organization was required. Getting a second network address would be hard to do since network addresses are scarce and the university already had enough addresses for over 60,000 hosts. The problem is the rule that a single class A, B, or C address refers to one network, not to a collection of LANs. As more and more organizations ran into this situation, a small change was made to the addressing system to deal with it. The solution is to allow a network to be split into several parts for internal use but still act like a single network to the outside world. A typical campus network nowadays might look like that of Fig. 5-57, with a main router connected to an ISP or regional network and numerous Ethernets spread around campus in different departments. Each of the Ethernets has its own router connected to the main router (possibly via a backbone LAN, but the nature of the inter router connection is not relevant here).



When a packet comes into the main router, how does it know which subnet (Ethernet) to give it to? One way would be to have a table with 65,536 entries in the main router telling which router to use for each host on campus. This idea would work, but it would require a very large table in the main router and a lot of manual maintenance as hosts were added, moved, or taken out of service.

Instead, a different scheme was invented. Basically, instead of having a single class B address with 14 bits for the network number and 16 bits for the host number, some bits are taken away from the host number to create a subnet number. For example, if the university has 35 departments, it could use a 6-bit subnet number and a 10-bit host number, allowing for up to 64 Ethernets, each with a maximum of 1022 hosts (0 and -1 are not available, as mentioned earlier). This split could be changed later if it turns out to be the wrong one.

To implement subnetting, the main router needs a subnet mask that indicates the split between network + subnet number and host, as shown in Fig. 5-58. Subnet masks are also written in dotted decimal notation, with the addition of a slash followed by the number of bits in the network + subnet part. For the example of Fig. 5-58, the subnet mask can be written as 255.255.252.0. An alternative notation is /22 to indicate that the subnet mask is 22 bits long.



A class B network subnetted into 64 subnets.

Question No. 7

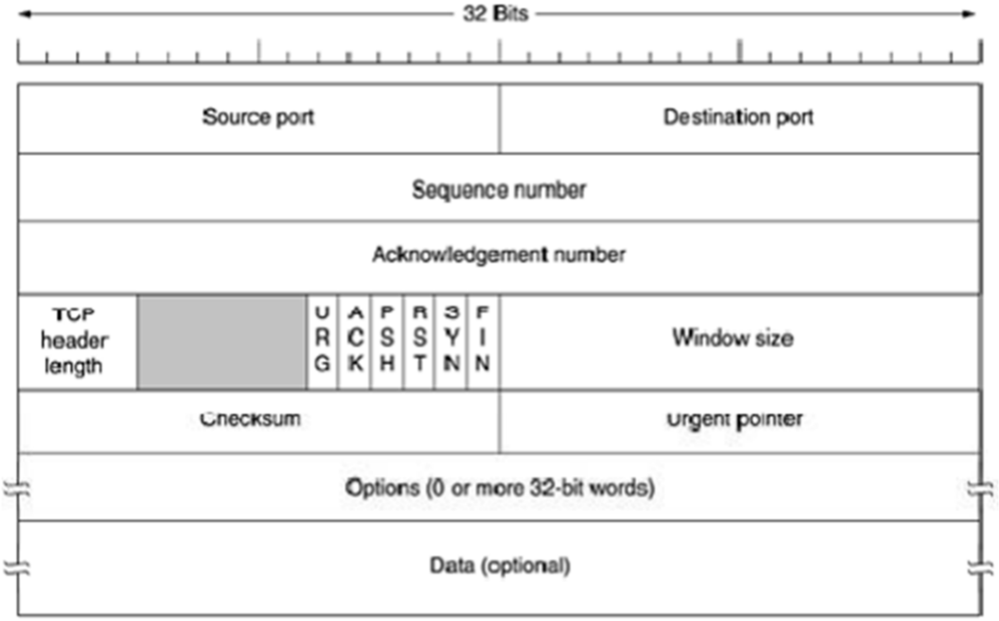
Explain the following:

i. TCP
ii. UDP

Answer: - I. TCP- Transmission Control Protocol

TCP (Transmission Control Protocol) was specifically designed to provide a reliable end-to- end byte stream over an unreliable internetwork. An internetwork differs from a single network because different parts may have wildly different topologies, bandwidths, delays, packet sizes, and other parameters. TCP was designed to dynamically adapt to properties of the internetwork and to be robust in the face of many kinds of failures.TCP was formally defined in RFC 793. As time went on, various errors and inconsistencies were detected, and the requirements were changed in someareas. These clarifications and some bug fixes are detailed in RFC 1122. Extensions are given in RFC 1323.

Each machine supporting TCP has a TCP transport entity, either a library procedure, a user process, or part of the kernel. In all cases, it manages TCP streams and interfaces to the IP layer. A TCP entity accepts user data streams from local processes, breaks them up into pieces not exceeding 64 KB (in practice, often 1460 data bytes in order to fit in a single Ethernet frame with the IP and TCP headers), and sends each piece as a separate IP datagram. When datagrams containing TCP data arrive at a machine, they are given to the TCP entity, which reconstructs the original byte streams. For simplicity, we will sometimes use just "TCP" to mean the TCP transport entity (a piece of software) or the TCP protocol (a set of rules). From the context it will be clear which is meant. For example, in "The user gives TCP the data," the TCP transport entity is clearly intended.



TCP HEADER

The IP layer gives no guarantee that datagrams will be delivered properly, so it is up to TCP to time out and retransmit them as need be. Datagrams that do arrive may well do so in the wrong order; it is also up to TCP to reassemble them into messages in the proper sequence. In short, TCP must furnish the reliability that most users want and that IP does not provide. Figure 8 shows the layout of a TCP segment. Every segment begins with a fixed-format, 20-byte header. The fixed header may be followed by header options. After the options, if any, up to 65,535 - 20 - 20 = 65,495 data bytes may follow, where the first 20 refer to the IP header and the second to the TCP header. Segments without any data are legal and are commonly used for acknowledgements and control messages.

The Source port and Destination port fields identify the local end points of the connection. The source and destination end points together identify the connection. The sequence number and Acknowledgement number fields perform their usual functions. Note that the latter specifies the next byte expected, not the last byte correctly received. Both are 32 bits long because every byte of data is numbered in a TCP stream. The TCP header length tells how many 32-bit words are contained in the TCP header.

This information is needed because the Options field is of variable length, so the header is, too. Technically, this field really indicates the start of the data within the segment, measured in 32-bit words, but that number is just the header length in words, so the effect is the same. Next comes a 6-bit field that is not used. The fact that this field has survived intact for over a quarter of a century is testimony to how well think out TCP is. Lesser protocols would have needed it to fix bugs in the original design. Now comes six 1-bit flags. URG is set to 1 if the Urgent pointer is in use. The Urgent pointer is used to indicate a byte offset from the current sequence number at which urgent data are to be found. This facility is in lieu of interrupt messages. As we mentioned above, this facility is a bare-bones way of allowing the sender to signal the receiver without getting TCP itself involved in the reason for the interrupt.

The ACK bit is set to 1 to indicate that the Acknowledgement number is valid. If ACK is 0, the segment does not contain an acknowledgement so the Acknowledgement number field is ignored.

The PSH bit indicates PUSH ed data. The receiver is hereby kindly requested to deliver the data to the application upon arrival and not buffer it until a full buffer has been received (which it might otherwise do for efficiency).

The RST bit is used to reset a connection that has become confused due to a host crash or some other reason. It is also used to reject an invalid segment or refuse an attempt to open a connection. In general, if you get a segment with the RST bit on, you have a problem on your hands. The SYN bit is used to establish connections. The connection request has SYN = 1 and ACK = 0 to indicate that the piggyback acknowledgement field is not in use. The connection reply does bear an acknowledgement, so it has SYN = 1 and ACK = 1. In essence the SYN bit is used to denote CONNECTION REQUEST and CONNECTION ACCEPTED, with the ACK bit used to distinguish between those two possibilities.

The FIN bit is used to release a connection. It specifies that the sender has no more data to transmit. Both SYN and FIN segments have sequence numbers and are thus guaranteed to be processed in the correct order.

Flow control in TCP is handled using a variable-sized sliding window. The Window size field tells how many bytes may be sent starting at the byte acknowledged. A Window size field of 0 is legal and says that the bytes up to and including Acknowledgement number - 1 have been received, but that the receiver is currently badly in need of a rest and would like no more data for the moment. The receiver can later grant permission to send by transmitting a segment with the same Acknowledgement number and a nonzero Window size field.

II) UDP:

The Internet protocol suite supports a connectionless transport protocol, UDP (User Datagram Protocol). UDP provides a way for applications to send encapsulated IP data grams and send them without having to establish a connection. UDP is described in RFC 768. UDP transmits segments consisting of an 8-byte header followed by the payload. The header is shown BELOW. The two ports serve to identify the end points within the source and destination machines. When a UDP packet arrives, its payload is handed to the process attached to the destination port. This attachment occurs when BIND primitive or something similar is used, for TCP (the binding process is the same for UDP). In fact, the main value of having UDP over just using raw IP is the addition of the source and destination ports. Without the port fields, the transport layer would not know what to do with the packet. With them, it delivers segments correctly.

◄———— 32 Bits ————►

SOURCE PORT	DESTINATION PORT
UDP LENGTH	UDP CHECKSUM

UDP HEADER

The source port is primarily needed when a reply must be sent back to the source. By copying the source port field from the incoming segment into the destination port field of the outgoing segment, the process sending the reply can specify which process on the sending machine is to get it.

The UDP length field includes the 8-byte header and the data. The UDP checksum is optional and stored as 0 if not computed (a true computed 0 is stored as all 1s). Turning it off is foolish unless the quality of the data does not matter (e.g., digitized speech). It is probably worth mentioning explicitly some of the things that UDP does not do. It does not do flow control, error control, or retransmission upon receipt of a bad segment. All of that is up to the user processes. What it does do is provide an interface to the IP protocol with the added feature of demultiplexing multiple processes using the ports. That is all it does. For applications that need to have precise control over the packet flow, error control, or timing, UDP provides just what the doctor ordered. One area where UDP is especially useful is in client-server situations. Often, the client sends a short request to the server and expects a short reply back. If either the request or reply is lost, the client can just time out and try again. Not only is the code simple, but fewer messages are required (one in each direction) than with a protocol requiring an initial setup.

Question No. 8

What is pure ALOHA and slotted ALOHA? Consider the delay of both at low load. Which one is less? Explain your answer.

Answer:-

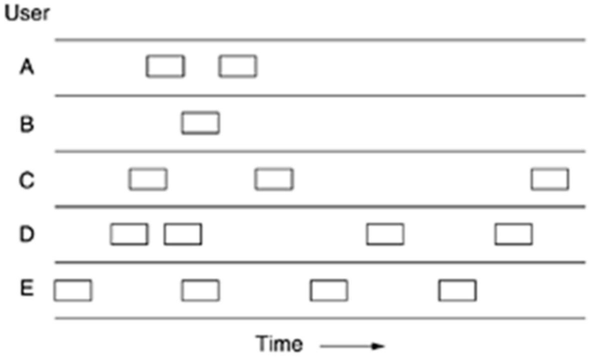
ALOHA:

In the 1970s, Norman Abramson and his colleagues at the University of Hawaii devised a new and elegant method to solve the channel allocation problem. Their work has been extended by many researchers since then (Abramson, 1985). Although Abramson's work, called the ALOHA system, used ground-based radio broadcasting, the basic idea is applicable to any system in which uncoordinated users are competing for the use of a single shared channel. There are two versions of ALOHA: pure and slotted. They differ with respect to whether time is divided into discrete slots into which all frames must fit. Pure ALOHA does not require global time synchronization; slotted ALOHA does.

Pure ALOHA:

The basic idea of an ALOHA system is simple: let users transmit whenever they have data to be sent. There will be collisions, of course, and the colliding frames will be damaged. However, due to the feedback property of broadcasting, a sender can always find out whether its frame was destroyed by listening to the channel, the same way other users do. With a LAN, the feedback is immediate; with a satellite, there is a delay of 270 msec before the sender knows if the transmission was successful. If listening while transmitting is not possible for some reason, acknowledgements are needed. If the frame was destroyed, the sender just waits a random amount of time and sends it again. The waiting time must be random or the same frames will collide over and over, in lockstep. Systems in which multiple users share a common channel in a way that can lead to conflicts are widely known as contention systems.

We have made the frames all the same length because the throughput of ALOHA systems is maximized by having a uniform frame size rather than by allowing variable length frames. Whenever two frames try to occupy the channel at the same time, there will be a collision and both will be garbled. If the first bit of a new frame overlaps with just the last bit of a frame almost finished, both frames will be totally destroyed and both will have to be retransmitted later. The checksum cannot (and should not) distinguish between a total loss and a near miss.



PURE ALOHA

In 1972, Roberts published a method for doubling the capacity of an ALOHA system (Robert, 1972). His proposal was to divide time into discrete intervals, each interval corresponding to one frame. This approach requires the users to agree on slot boundaries.

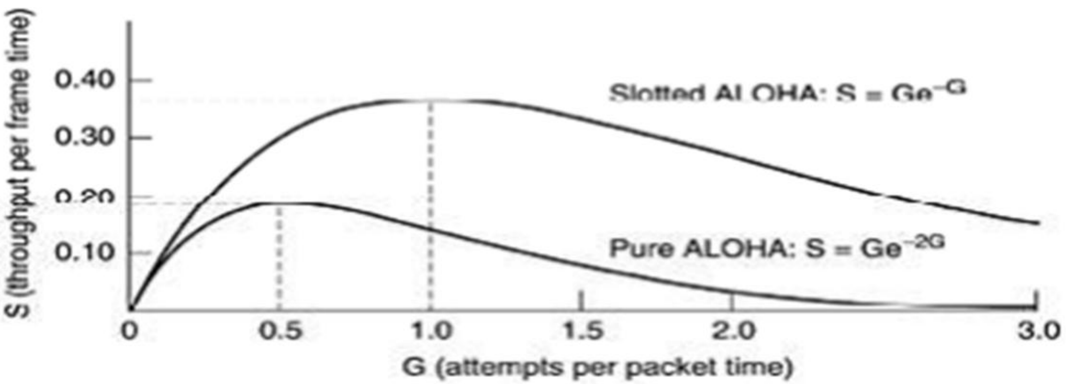
One way to achieve synchronization would be to have one special station emit a pip at the start of each interval, like a clock. In Roberts' method, which has come to be known as slotted ALOHA, in contrast to Abramson's pure ALOHA, a computer is not permitted to send whenever a carriage return is typed. Instead, it is required to wait for the beginning of the next slot. Thus, the continuous pure ALOHA is turned into a discrete one. Since the vulnerable period is now halved, the probability of no other traffic during the same slot as our test frame is e-G which leads to

Equation

$$S=Ge^{-G}$$

As you can see from Fig.3.3, slotted ALOHA peaks at G = 1, with a throughput of S =1/e or about 0.368, twice that of pure ALOHA. If the system is operating at G = 1, the probability of an empty slot is 0.368. The best we can hope for using slotted ALOHA is 37 percent of the slots empty, 37 percent successes, and 26 percent collisions. Operating at higher values of G reduces the number of empties but increases the number of collisions exponentially.

To see how this rapid growth of collisions with G comes about, consider the transmission of a test frame. The probability that it will avoid a collision is e-G, the probability that all the other users are silent in that slot. The probability of a collision is then just 1 – e-G. The probability of a transmission requiring exactly k attempts, (i.e., k – 1 collisions followed by one success).



SLOTTED ALOHA

Question No. 9
Explain in detail CSMA Protocol in detail.

Answer:-

(i) 1-persistance CSMA

(ii) Non-persistence CSMA

(iii) P-persistence CSMA.

Carrier Sense Multiple Access Protocols:

With slotted ALOHA the best channel utilization that can be achieved is 1/e. This is hardly surprising, since with stations transmitting at will, without paying attention to what the other stations are doing, there are bound to be many collisions. In local area networks, however, it is possible for stations to detect what other stations are doing, and adapt their behaviour accordingly. These networks can achieve a much better utilization than 1/e. In this section we will discuss some protocols for improving performance. Protocols in which stations listen for a carrier (i.e., a transmission) and act accordingly are called carrier sense protocols. A number of them have been proposed. Kleinrock and Tobagi (1975) have analysed several such protocols in detail. Below we will mention several versions of the carrier sense protocols.

1.1-persistent CSMA:

The first carrier sense protocol that we will study here is called 1-persistent CSMA (Carrier Sense Multiple Access). When a station has data to send, it first listens to the channel to see if anyone else is transmitting at that moment. If the channel is busy, the station waits until it becomes idle. When the station detects an idle channel, it transmits a frame. If a collision occurs, the station waits a random amount of time and starts all over again. The protocol is called 1-persistent because the station transmits with a probability of 1 when it finds the channel idle. The propagation delay has an important effect on the performance of the protocol.

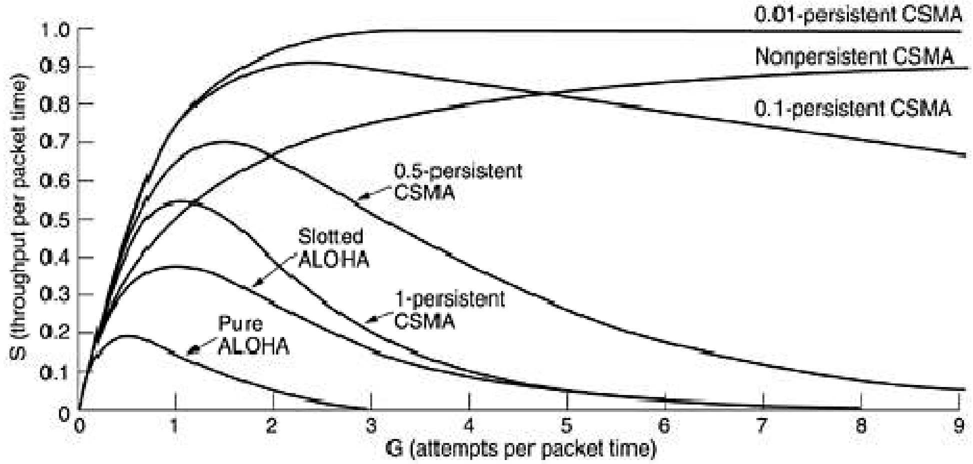
There is a small chance that just after a station begins sending, another station will become ready to send and sense the channel. If the first station's signal has not yet reached the second one, the latter will sense an idle channel and will also begin sending, resulting in a collision. The longer the propagation delay, the more important this effect becomes, and the worse the performance of the protocol.

Even if the propagation delay is zero, there will still be collisions. If two stations become ready in the middle of a third station's transmission, both will wait politely until the transmission ends and then both will begin transmitting exactly simultaneously, resulting in a collision. If they were not so impatient, there would be fewer collisions. Even so, this protocol is far better than pure ALOHA because both stations have the decency to desist from interfering with the third station's frame. Intuitively, this approach will lead to a higher performance than pure ALOHA. Exactly the same holds for slotted ALOHA.

2. Non-persistent CSMA:

A second carrier sense protocol is non-persistent CSMA. In this protocol, a conscious attempt is made to be less greedy than in the previous one. Before sending, a station senses the channel. If no one else is sending, the station begins doing so itself. However, if the channel is already in use, the station does not continually sense it for the purpose of seizing it immediately upon detecting the end of the previous transmission. Instead, it waits a random period of time and then repeats the algorithm. Consequently, this algorithm leads to better channel utilization but longer delays than 1-persistent CSMA.

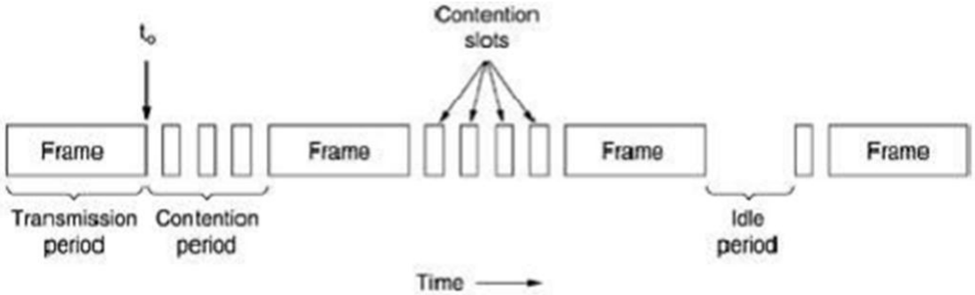
3. P-persistent CSMA:
The last protocol is p-persistent CSMA. It applies to slotted channels and works as follows. When a station becomes ready to send, it senses the channel. If it is idle, it transmits with a probability p. With a probability q = 1 - p, it defers until the next slot. If that slot is also idle, It either transmits or defers again, with probabilities p and q. This process is repeated until either the frame has been transmitted or another station has begun transmitting. In the latter case, the unlucky station acts as if there had been a collision (i.e., it waits a random time and starts again). If the station initially senses the channel busy, it waits until the next slot and applies the above algorithm. Figure 4 shows the computed throughput versus offered traffic for all three protocols, as well as for pure and slotted ALOHA.



Question No. 10
Explain in detail CSMA/CD Protocol in detail. How it detects collision.

Answer:-
Persistent and non-persistent CSMA protocols are clearly an improvement over ALOHA because they ensure that no station begins to transmit when it senses the channel busy. Another improvement is for stations to abort their transmissions as soon as they detect a collision. In other words, if two stations sense the channel to be idle and begin transmitting simultaneously, they will both detect the collision almost immediately. Rather than finish transmitting their frames, which are irretrievably garbled anyway, they should abruptly stop transmitting as soon as the collision is detected. Quickly terminating damaged frames saves time and bandwidth.

This protocol, known as CSMA/CD (CSMA with Collision Detection) is widely used on LANs in the MAC sub layer. In particular, it is the basis of the popular Ethernet LAN, so it is worth devoting some time to looking at it in detail. CSMA/CD, as well as many other LAN protocols, uses the conceptual model of Fig.5. At the point marked t_0, a station has finished transmitting its frame. Any other station having a frame to send may now attempt to do so. If two or more stations decide to transmit simultaneously, there will be a collision. Collisions can be detected by looking at the power or pulse width of the received signal and comparing it to the transmitted signal.



After a station detects a collision, it aborts its transmission, waits a random period of time, and then tries again, assuming that no other station has started transmitting in the meantime. Therefore, our model for CSMA/CD will consist of alternating contention and transmission periods, with idle periods occurring when all stations are quiet (e.g., for lack of work).

Now let us look closely at the details of the contention algorithm. Suppose that two stations both begin transmitting at exactly time t_0. How long will it take them to realize that there has been a collision? The answer to this question is vital to determining the length of the contention period and hence what the delay and throughput will be. The minimum time to detect the collision is then just the time it takes the signal to propagate from one station to the other.

Question No. 11

What is IPv6? Explain its advantages over IPv4. Also explain its frame format.

Answer:-

IPv4 provides the host-to-host communication between systems in the Internet. Although IPv4 is well designed, data communication has evolved since the inception of IPv4 in the 1970s. IPv4 as some deficiencies that make it unsuitable for the fast-growing Internet.

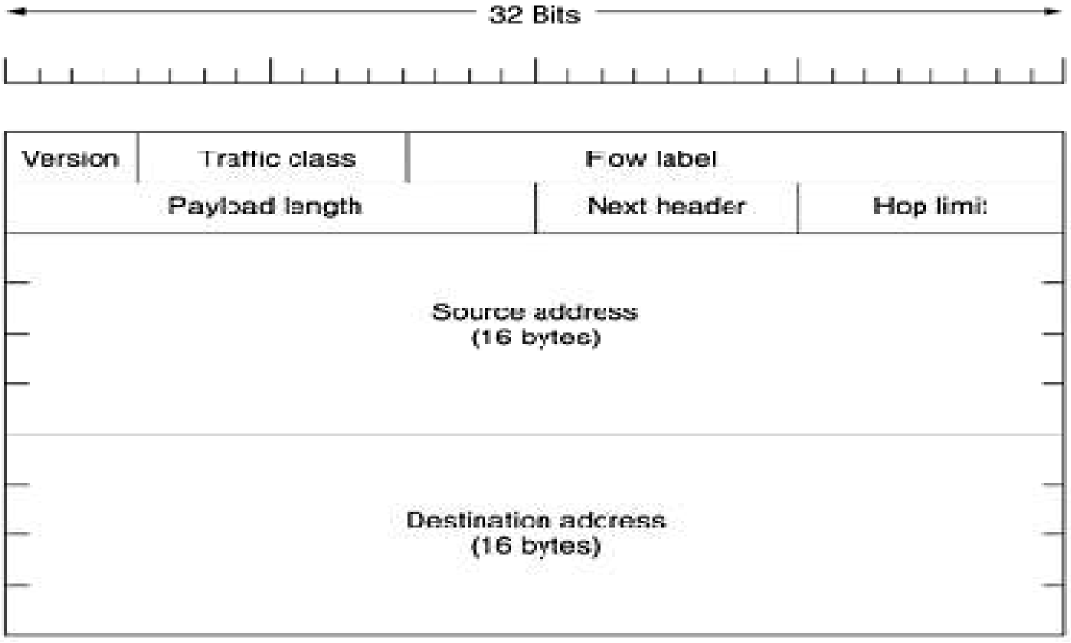
- Despite all short-term solutions, such as subnetting, classless addressing, and NAT, address depletion is still a long-term problem in the Internet.
- The Internet must accommodate real-time audio and video transmission. This type of transmission requires minimum delay strategies and reservation of resources not provided in the IPv4 design.
- The Internet must accommodate encryption and authentication of data for some applications. No encryption or authentication is provided by IPv4.

To overcome these deficiencies, IPv6 (Internetworking Protocol, version 6), also known as IPng (Internetworking Protocol, next generation), was proposed and is now a standard. In IPv6, the Internet protocol was extensively modified to accommodate the unforeseen growth of the Internet. The format and the length of the IP address were changed along with the packet format. Related protocols, such as ICMP, were also modified. Other protocols in the network layer, such as ARP, RARP, and IGMP, were either deleted or included in the ICMPv6 protocol (see Chapter 21). Routing protocols, such as RIP and OSPF (see Chapter 22), were also slightly modified to accommodate these changes. Communications experts predict that IPv6 and its related protocols will soon replace the current IP version. In this section first we discuss IPv6. Then we explore the strategies used for the transition from version 4 to version 6. The adoption of IPv6 has been slow. The reason is that the original motivation for its development, depletion of IPv4 addresses, has been remedied by short-term strategies such as classless addressing and NAT. However, the fast-spreading use of the Internet, and new services such as mobile IP, IP telephony, and IP-capable mobile telephony, may eventually require the total replacement of IPv4 with IPv6.

Advantages

The next-generation IP, or IPv6, has some advantages over IPv4 that can be summarized as follows:

1. Larger address space: An IPv6 address is 128 bits long, Compared with the 32-bit address of IPv4, this is a huge (296) increase in the address space.
2. Better header format. IPv6 uses a new header format in which options are separated from the base header and inserted, when needed, between the base header and the upper-layer data. This simplifies and speeds up the routing process because most of the options do not need to be checked by routers.
3. New options. IPv6 has new options to allow for additional functionalities.
4. Allowance for extension. IPv6 is designed to allow the extension of the protocol if required by new technologies or applications.
5. Support for resource allocation. In IPv6, the type-of-service field has been removed, but a mechanism (called jlow label) has been added to enable the source to request special handling of the packet. This mechanism can be used to support traffic such as real-time audio and video.
6. Support for more security. The encryption and authentication options in IPv6 provide confidentiality and integrity of the packet.



IPv6 Packet Format

1. Version (4 bits)
   The constant 6 (bit sequence 0110).
2. Traffic Class (8 bits)
   The bits of this field hold two values. The 6 most-significant bits are used for DSCP, which is used to classify packets. The remaining two bits are used for ECN, priority values subdivide into ranges: traffic where the source provides congestion control and non-congestion control traffic.
3. Flow Label (20 bits)
   Originally created for giving real-time applications special service. Flow Label specifications and minimum requirements are described, and first uses of this field are emerging.
4. Payload Length (16 bits)
   The size of the payload in octets, including any extension headers. The length is set to zero when a Hop-by-Hop extension header carries a Jumbo Payload option.
5. Next Header (8 bits)
   Specifies the type of the next header. This field usually specifies the transport layer protocol used by a packet's payload, when extension headers are present in the packet this field indicates which extension header follows. The values are shared with those used for the IPv4 protocol field, as both fields have the same function (see List of IP protocol numbers).
6. Hop Limit (8 bits)
   Replaces the time to live field of IPv4. This value is decremented by one at each intermediate node the packet visits. When the counter reaches 0 the packet is discarded.
7. Source Address (128 bits)
   The IPv6 address of the sending node.
8. Destination Address (128 bits)
   The IPv6 address of the destination node(s)

Some Additional Questions With Solutions

1. Which of the following protocols are examples of TCP/IP transport layer protocols?

a. Ethernet
b. HTTP
c. IP
d. UDP
e. SMTP
f. TCP

2. Which of the following protocols are examples of TCP/IP network access layer protocols?

a. Ethernet
b. HTTP
c. IP
d. UDP
e. SMTP
f. TCP
g. PPP

3. The process of HTTP asking TCP to send some data and make sure that it is received correctly is an example of what?

a. Same-layer interaction
b. Adjacent-layer interaction
c. The OSI model
d. All the other answers are correct.

4. The process of TCP on one computer marking a segment as segment 1, and the receiving computer then acknowledging the receipt of segment 1, is an example of what?

a. Data encapsulation
b. Same-layer interaction
c. Adjacent-layer interaction
d. The OSI model
e. None of these answers are correct.

5. The process of a web server adding a TCP header to a web page, followed by adding an IP header, and then a data link header and trailer is an example of what?

a. Data encapsulation
b. Same-layer interaction
c. The OSI model
d. All of these answers are correct.

6. Which of the following terms is used specifically to identify the entity that is created when encapsulating data inside data link layer headers and trailers?

a. Data
b. Chunk

c. Segment
d. Frame
e. Packet
f. None of these—there is no encapsulation by the data link layer.

7. Which OSI layer defines the functions of logical network-wide addressing and routing?
a. Layer 1
b. Layer 2
c. Layer 3
d. Layer 4
e. Layer 5
f. Layer 6
g. Layer 7

8. Which OSI layer defines the standards for cabling and connectors?
a. Layer 1
b. Layer 2
c. Layer 3
d. Layer 4
e. Layer 5
f. Layer 6
g. Layer 7

9. Which OSI layer defines the standards for data formats and encryption?
a. Layer 1
b. Layer 2
c. Layer 3
d. Layer 4
e. Layer 5
f. Layer 6
g. Layer 7

10. Which of the following terms are not valid terms for the names of the seven OSI layers?
a. Application
b. Data link
c. Transmission
d. Presentation
e. Internet
f. Session

Fill In The Blanks:

1. Number of links to connect n nodes in a mesh topology is = _______.

2. Mesh Topology is _______ flexible and has a ________ expandability

3. In BUS topology, at each end of the bus is a ____________, which absorbs any signal, removing it from the bus.

4. In BUS topology, One can easily add any new node or delete any node with-out affecting other nodes; this makes this topology easily _____________.

5. _________ and ______________ will force a maximum length of shared medium which can be used in BUS topology.

6. The two alternatives for the operation of the central node in STAR topology are:

__________ and ___________.

7. In Ring Topology, the links are _____________; that is, data are transmitted in __________ direction only and all are oriented in the same way

8. In Ring Topology, Repeater works in 3 modes: ____________, _____________ and ______________.

9. ___________ topology can be considered as an extension to BUS topology.

10. ______________ is suitable for use in star and ring topologies

11. Coaxial cable is suitable for use in __________ topology.

Solutions:

1. n(n-1)/2

2. not, poor

3. terminator

4. expandable.

5. Delay, signal unbalancing

6. repeater, switch

7. unidirectional, one

8. Listen, Transmit, By-Pass

9. Tree

10. Twisted pair

11. BUS

Short Answer Questions:

Q-1. List out the advantages and drawbacks of bus topology.

Ans: Advantages:

i) Easy to implement

ii) It is very cost effective because only a single segment required

iii) It is very flexible

iv) Moderate reliability.

v) Can add new station or delete any station easily (scalable)

Disadvantages:

i) Required suitable medium access control technique.

ii) Maximum cable length restriction imposed due to delay and signal

unbalancing problem.

Q-2. List out the advantages and drawbacks of ring topology.

Ans: Advantages:

i) Data insertion, data reception and data removal can be provided by repeater

ii) It can provide multicast addressing.

iii) Point-to-point links to its adjacent nodes (moderate cost)

Disadvantages:

i) The repeater introduces a delay

ii) The topology fails if any link disconnects or a node fails.

iii) Direct link not provided

iv) It provides complex management

Q-3. Why star topology is commonly preferred?

Ans: It gives high reliability, more flexible and higher bandwidth. Since there is a central control point, the control of network is easy and priority can be given to selected nodes.

Q-4. Is there any relationship between transmission media and topology?

Ans: Yes, medium should be selected based on the topology. For example, for bus topology coaxial cable medium is suitable, and for ring/star topology twisted-pair or optical fiber can be used.

Fill In The Blanks:

1. The basic question which has to be answered by the medium-access control techniques is “How Goes ________”?

2. In ________________ technique, each node gets a chance to access the medium by rotation.

3. The key issues involved in MAC protocol are - Where and ______ the control is exercised.

4. ‘Where’ refers to whether the control is exercised in a ______________ or _______________ manner.

5. The ________________ techniques can be broadly categorized into three types; Round-Robin, Reservation and____________.

6. _______________ is an example of centralized control and _______________ is an example of distributed control

Version 2 CSE IIT, Kharagpur 7. In Polling technique, if there is no data, usually a _______________ message is sent back.

8. In pure ALOHA, channel utilization, expressed as throughput S, in terms of the offered load G is given by __________

9. In slotted ALOHA, a maximum throughput of ________ percent at 100 percent of offered load can be achieved, while it is ______ percentage for pure ALOHA.

10. __________________ is abbreviated as CSMA/CD and is also known as .__________

11. To achieve stability in CSMA/CD back off scheme, a technique known as _______________ is used

Solutions:

1. Next

2. token passing

3. How

4. centralized, distributed

5. asynchronous, Contention

6. Polling, token passing

7. poll reject

8. S=Ge-2G

9. 37, 18

10. Carrier Sensed Multiple Access with Collision Detection, Listen-While-Talk .

11. binary exponential back off

Short Answer Questions:

Q-1. In what situations contention based MAC protocols are suitable?

Ans: Contention based MAC protocols are suitable for bursty nature of traffic under light to moderate load. These techniques are always decentralized, simple and easy to implement.

Q-2. What is vulnerable period? How it affects the performance in MAC protocols?

Ans: The total period of time when collision may occur for a packet is called vulnerable period. Let, all packets have a fixed duration λ. Then vulnerable period is 2λ in pure ALOHA scheme and λ in slotted ALOHA scheme. If vulnerable period is long, probability of the occurrence collision increases leading to reduction in throughput.

Q-3. How throughput is improved in slotted ALOHA over pure ALOHA?

Ans: In pure ALOHA vulnerable period is 2λ.

So, S/G = e-2G or throughput S = G e-2G , where G is the total number of packets.

Maximum value of G = 0.5 or maximum throughput Smax = 1/2e.

In slotted ALOHA, vulnerable period is λ and S/G = e-G or throughput S = G e-G . Here,

maximum value of G is 1 and maximum throughput Smax = 1/e.

Q-4. What is the parameter ‘a’? How does it affect the performance of the CSMA protocol?

Ans: The efficiency of CSMA scheme depends on propagation delay, which is represented by a parameter ‘a’ as defined below.

propagation delay

a = --------------------------------

packet transmission time

Smaller the value of propagation delay, lower is the vulnerable period and higher is the efficiency. If propagation delay is zero, collision cannot occur in CSMA scheme. But in practice, there is some delay and depending on the value of 'a' collision occurs.

Q-5. How performance is improved in CSMA/CD protocol compared to CSMA protocol?

Ans: In CSMA scheme, a station monitors the channel before sending a packet. Whenever a collision is detected, it does not stop transmission leading to some wastage of time. On the other hand, in CSMA/CD scheme, whenever a station detects a collision, it sends a jamming signal by which other station comes to know that a collision occurs. As a result, wastage of time is reduced leading to improvement in performance.

Fill In The Blanks :

1. The 802.2 standard describes the ____________, which is the upper part of the data link layer.

2. LLC offers three types services: Unreliable datagram service, ________________ and ________________.

3. IEEE 802 bundle also includes a MAN standard IEEE 802.6 which is also known as _______________.

4. 100Base-T2 means _________________________________

5. 100 Mbps, baseband, long wavelength over optical fiber cable will be abbreviated as ________________

6. Ethernet uses __________ encoding

Answers:

1. LLC (logical link layer)

2. Acknowledged datagram service, Reliable connection oriental service

3. Distributed Queue Dual Bus (DQDB)

4. 100 Mbps, baseband, over two twisted-pair cables

5. 1000Base F

6. Bi-phase Manchester

Short question Answers

Q-1 What are the goals in mind of IEEE 802 committee?

Ans: IEEE 802 committee has few goals in mind, namely

– To promote compatibility

– Implementation with minimum efforts

– Accommodate diverse applications

Q-2. List the functions performed by the physical layer of 802.3 standard?

Ans. Functions of physical layer are:

i) Data encoding/decoding (To facilitate synchronization and efficient transfer of signal through the medium).

ii) Collision detection (It detects at the transmit side)

iii) Carrier sensing (Channel access senses a carrier on the channel at both the transmit and receive sides)

iv) Transmit/receive the packets (Frame transmitted to all stations connected to the channel)

v) Topology and medium used (Mediums are co-axial cable, twisted pair and fiber optic cable)

Q-3. Why do you require a limit on the minimum size of Ethernet frame?

Ans. To detect collision, it is essential that a sender continue sending a frame and at the same time receives another frame sent by another station. Considering maximum delay with five Ethernet segments in cascade, the size of frame has been found to be 64 bytes such that the above condition is satisfied.

Q-4. What are the different types of cabling supported by Ethernet standard?

Ans. Types of cabling are:

i) 10 BASE 5 - Maximum cable length is 500 meters using 4” diameter coaxial cable.

ii) 10 BASE 2 - Maximum cable length is 185 meters using 0.25” diameter CATV cable.

iii) 10 BASE T - Maximum cable length is 100 meters using twisted-pair cable (CAT-3 UTP).

iv) 10 BASE FL - Maximum cable length is 2 Km using multimode fiber optic cable (125/62.5 micrometer).

Fill In The Blanks:

1. Originally, ______________ developed Token Ring network in the ________.

2. A disadvantage of this topology is that it is vulnerable to _________ or _____ failure.

3. Unlike CSMA/CD networks (such as Ethernet), token-passing networks are ________________, which means that it is possible to calculate the maximum time that will pass before any end station will be capable of transmitting.

4. Token Ring frames have two fields that control priority: ___________ and the ___________field.

Version 2 CSE IIT, Kharagpur 5. In Token Ring inside the wire center, _____________ are used to isolate a broken wire or a faulty station.

6. The Mac sublayer in Token BUS consists of four major functions: ____________, the access control machine (ACM), ______________________and .____________

7. ___________ determines when to place a frame on the bus, and responsible for the maintenance of the logical ring including the error detection and fault recovery.

Answers:

1. IBM, 1970

2. link, station

3. deterministic

4. the priority field, reservation

5. bypass relays

6. the interface machine (IFM), the receiver machine (RxM), the transmit machine (TxM).

7. Access control machine (ACM)

Short question Answers:

Q-1. What is the advantage of token passing protocol over CSMA/CD protocol?

Ans. Advantage of token passing protocol over CSMA/CD protocol:

The CSMA/CD is not a deterministic protocol. A packet may be delivered after many (up to 15) collisions leading to long variable delay. An unfortunate packet may not get delivered at all. This feature makes CSMA/CD protocol unsuitable for real-time applications. On the other hand, token passing protocol is a

deterministic approach, which allows a packet to be delivered within a known time frame. It also allows priority to be assigned to packets. These are the two key advantages of token passing protocol over CSMA/CD protocol.

Q-2. What are the drawbacks of token ring topology?

Ans. Token ring protocol cannot work if a link or a station fails. So, it is vulnerable to link and station failure.

Q-3. How the reliability of token ring topology can be improved?

Ans. Reliability of the ring network can be improved by implementing the ring topology using a wiring concentrator. This allows not only to detect fault, but also to isolate the faulty link/station with the help of a bypass relay.

Q-4. What role the active token monitor performs?

Ans. Token ring is maintained with the help of active token monitor. Any one of the stations has the capability to act as active token monitor, but at a particular instant only one acts as active token monitor. It monitors various error situations such as multiple token, orphan packet, etc, and takes appropriate action to come out of the error situation.

Fill In The Blanks:

1. The high speed LANs that have emerged can be broadly categorized into three types_______________, successors of Ethernet and ____________.

2. ATM, fiber channel and the Etherswitches comes under high speed LANs based on ________________.

3. ________________________ is abbreviated as FDDI.

4. FDDI over copper is referred to as ____________________________.

5. The basic topology for FDDI is ______________________.

6. An ______________ provides continuous dual-ring operation if a device on the dual ring fails

7. Each data frame in FDDI carries up to _____________ bytes.

8. FDDI gives fair and equal access to the ring by using a __________ protocol.

9. FDDI implements MAC using three timers namely: _________________, Token Rotation Timer (TRT) and ____________________

10. Token holding Timer (THT), which determines .______________________________

11. The frame status (FS) byte is set by the _____________ and checked by the ________________ station which removes its frame from the ring and generates another token.

12. When the frame returns to the sending station, that station removes the frame from the ring by a process called ___________.

13. The physical layer is divided into two sub layers - _______ and ______

Solutions.

1. based on token passing, based on switching technology.

2. based on switching technology.

3. Fiber Distributed Data Interface

4. Copper-Distributed Data Interface (CDDI).

5. dual counter rotating rings

6. optical bypass switch

7. 4500

8. timed token

9. Token holding Timer (THT), Valid Transmission Timer (VTT)

10. how long a station may continue once it has captured a token

11. destination, source

12. stripping

13. PMD, PHY

Short Questions:

Q-1. In what way the MAC protocol of FDDI differs from that of token ring?

Ans: In the frame format of FDDI protocol, preamble is eight bytes instead of one byte in token ring. Also token has one additional byte. FDDI can have multiple frames simultaneously, which cannot be present in token ring. Here, the access method is timed token passing. Multiple frames can be transmitted after capturing a token. First, the entire token is captured and then the data frames are introduced, whereas token ring follows token passing protocol and beginning of token is converted to the header of a frame. In case of token ring token is released after receiving the acknowledgement (as the data frame returns after circulating the ring). On the other hand, in case of FDDI, token is released immediately after sending data frame, which is known as early token release.

Q-2. How FDDI offers higher reliability than token ring protocol?

Ans: Token ring protocol is applicable in a single ring. Disadvantage of this protocol is that, if one segment of wires fails or a node fails, the protocol cannot work. To increase reliability, dual counter ring topology used in FDDI protocol, where there are two rings, called primary ring and secondary ring. In case of failure of a node or a fiber link, the ring is restored the by wrapping up the primary ring to the secondary ring. Further improvement in reliability can achieve by using dual ring of trees and dual homing mechanism. It will provide multiple paths and if one path fails, another path will be available for passing token or data.

Q-3 What are the functionalities of a Optical Bypass Switch?

Ans: An optical bypass switch provides continuous dual-ring operation if a device on the dual ring fails. This is used both to prevent ring segmentation and to eliminate failed stations from the ring. The optical bypass switch performs this function using optical mirrors that pass light from the ring directly to the DAS (dual-attachment station) device during normal operation. If a failure of the DAS device occurs, such as a power-off, the optical bypass switch will pass the light through itself by using internal mirrors and thereby will maintain the ring's integrity. When using the OB, you will notice a tremendous digression of your network as the packets are sent through the OB unit.

Q-4 What are the functionalities provided by SMT standard?

Ans: The Station Management (SMT) standard provides services that monitor and control a FDDI station. SMT include facilities for connection management, node configuration, recovery from error condition, and encoding of SMT frames.

Q-5 Describe various fields in frame format of FDDI?

Ans: Let us have a look at the various fields:

SD: The first byte, after the preamble, of the field is the frame's starting flag. As in

Token ring these bits are replaced in physical layer by the control codes.

FC: it identifies the frame type i.e. token or a data frame.

Address: the next 2 fields are destination and source addresses. Each address consists of

2-6 bytes.

Data: Each data frame carries up to 4500 bytes.

FCS: FDDI uses the standard IEEE four-byte cyclic redundancy check.

ED: this field consists of half a byte in data frame or a full byte in token frame. This

represents end of the Token.

FS: FDDI FS field is similar to that of Token Ring. It is included only in data/Command frame and consists of one and a half bytes.

Fill In The Blanks:

1. Switched Ethernet gives dedicated 10 Mb/s bandwidth on __________ of its ports.

2. In Ethernet (IEEE 802.3) the topology, though physically is _________ but logically is BUS. i.e. the collision domain of all the nodes in a LAN is ._________

3. In Switched Ethernet, collision domain is separated. Hub is replaced by a .________

4. There are two techniques used in the implementation of Ethernet switches: _____________ and ________

5. IEEE has designed two categories of Fast Ethernet: ____________ and .__________

6. 100-Base-X itself is divided into two: __________ and __________

7. The Gigabit Ethernet Alliance was formed in ___________ by ______ companies.

8. The GMII is the interface between the ________ layer and the ___________ layer.

9. ________, a sublayer of GMII provides a medium-independent means for the PCS to support various serial bit-oriented physical media.

10. Packet Bursting is an extension of ______________. Packet Bursting is "Carrier Extension plus a ____________".

Solutions:

1. each

2. star, common

3. switch

4. store-and-forward, cut-through

5. 100Base-X, 100Base-T4

6. 100Base-TX, 100base-FX.

7. May 1996, 11

8. MAC, Physical

9. PMA (Physical Medium Attachment)

10. Carrier Extension, burst of packets

Short Questions:

Q-1. Explain the basic difference between IEEE 802.3 and switched Ethernet, as far as implementation is concerned.

Ans: In Ethernet (IEEE 802.3) the topology, though physically is start but logically is BUS. i.e. the collision domain of all the nodes in a LAN is common. In this situation only one frame can send the frame, if more than one station sends the frame, there is a collision.

In Switched Ethernet, this collision domain is separated. Hub is replaced by a switch, a device that can recognize the destination address and can route the frame to the port to which the destination station is connected, the rest of the media is not involved in the transmission process. The switch can receive another frame from another station at the same time and can route this frame to its own final destination.

Q-2. Explain the two techniques for implementing Ethernet switches.

Ans: There are two techniques used in the implementation of Ethernet switches: storeand-forward and cut-through. In the first case, the entire frame is captured at the incoming port, stored in the switch's memory, and after an address lookup to determine the LAN destination port, forwarded to the appropriate port. The lookup table is automatically built up. On the other hand, a cut-through switch begins to transmit the frame to the destination port as soon as it decodes the destination address from the frame header.

Store-and-forward approach provides a greater level of error detection because damaged frames are not forwarded to the destination port. But, it introduces longer delay of about 1.2 msec for forwarding a frame and suffers from the chance of loosing data due to reliance on buffer memory. The cut-through switches, on the other hand, has reduced latency but has higher switch cost.

Q-3. What are the different categories of Fast Ethernet?

Ans: IEEE has designed two categories of Fast Ethernet: 100Base-X and 100Base-T4. 100Base-X uses two cables between hub and the station while 100Base-T4 uses four. 100-Base-X itself is divided into two: 100Base-TX and 100base-FX. * 100 BASE-T4: This option is designed to avoid overwriting. It is used for half-duplex communication using four wire-pairs of the existing category 3 UTP cable, which is already available for telephone services in homes/offices. Two of four pairs are bi-directional; other two are

unidirectional. This means that there are 3 pairs to be used for carrying data, in each direction (2 bi-directional and 1 unidirectional). Because 100Mbps data cannot be handled by voicegrade UTP, this specification splits the 100 Mbps flow into three 33.66Mbps flow.

*100 BASE TX: This option uses two category 5 UTP or two shielded (STP) cable to connect a station to hub. One pair is used to carry frames from the hub to the station and other to carry frames from station to hub. Encoding is 4B/5B to handle 100 Mbps; signaling is NRZ-I. The distance between station and hub should be less than 100 meters.

*100 BASE FX: This option uses two Fiber optic cables, one carry frames from station to hub and other from hub to station. The encoding is 4B/5B and signaling in NRZ-I. the distance between station and hub should be less than 2000 meters.

Q-4. What are the Objectives of The Gigabit Ethernet Alliance?

Ans: The objectives of the alliance are:

- supporting extension of existing Ethernet and Fast Ethernet technology in response to demand for higher network bandwidth.

- developing technical proposals for the inclusion in the standard

- establishment of inter-operability test procedures and processes

Q-5. Explain GMII (Gigabit Media Independent Interface) in brief.

Ans: The GMII is the interface between the MAC layer and the Physical layer. It allows any physical layer to be used with the MAC layer. It is an extension of the MII (Media Independent Interface) used in Fast Ethernet. It uses the same management interface as MII. It supports 10, 100 and 1000 Mbps data rates. It provides separate 8-bit wide receive and transmit data paths, so it can support both full-duplex as well as half-duplex operation.

The GMII provides 2 media status signals: one indicates presence of the carrier, and the other indicates absence of collision. With the GMII, it is possible to connect various media types such as shielded and unshielded twisted pair, and single-mode and multi mode optical fiber, while using the same MAC controller. It has three sub-layers namely: PCS (Physical Coding Sublayer), PMA (Physical Medium Attachment) and PMD (Physical Medium Dependent).

Sample Questions

1. What are the major advantages of STP over UTP?
2. Describe the components of fibre optic cable. Draw a picture.
3. Why are slots used in DQDB?
4. What is the difference between network layer delivery and transport layer delivery?
5. How can a device have more than one IP address?
6. Which control bit is involved in setting up a TCP session?
7. What factors effects the data rate of a link?
8. What are the advantages of FDDI over a basic token ring?
9. What 3 functions can SNMP perform to manage network devices?
10. What is the purpose of the timer at the sender in systems using ARQ?
11. Is there any drawback of using piggybacking?
12. Which of the following address does not belong to the same network(no subnetting)? Explain why?
    1. 130.31.23.31
    2. 130.31.24.22
    3. 130.32.23.21
    4. 130.31.21.23

1. What are the two reasons for using layered protocols?
2. What do you mean by link to link layers of OSI reference model? Explain their functions briefly?
3. Write a short note on ISDN?
4. What is the difference between boundary level masking and non- boundary level masking? Give examples
5. Draw the IP datagram header format. “IP datagram has a checksum field still it is called an unreliable protocol”. Justify?
6. What are the principles that were applied to arrive at the seven layers in OSI model?
7. Explain the working of 3 bit sliding window protocol with suitable example.
8. Explain the following ARQ techniques in detail
    1. Stop and wait ARQ
    2. Selective repeat ARQ

9. What are the reasons for using layered protocols ?
10. 10 Enumerate the main responsibilities of data link layer ?
11. Is the nyquist theorem true for optical fibre or only for copper wire ? Explain.
12. Why do data link layer protocols position the checksum in the trailer and not in the header ?
13. Compare the maximum window size in go-back-N and selective-repeat ARQs.
14. Why does ATM use the cell of small and fixed length ?
15. Give the equivalent binary word for the polynomial x^8+x^2+x+1.
16. In which of the 7 layers of OSI will a service handling conversion of characters is from EBCDIC to ASCII be normally implemented ?
17. Where is the special IP address 127.0.0.0. used ?
18. Convert the IP address 197.228.17.56 into binary ?
19. Compare satellite with fiber as a communication medium and enumerate the application areas where satellite still holds a niche(or special)marker.
20. A binary signal is sent over a 3-khz channel whose signal-to-noise ratio is 20 db.Calculate the maximum achievable data rate ?
21. What does ‘data transparency’ mean ? With the help of a flow chart, explain the process of bit de-stuffing at the receiver’s end.
22. Assuming classful addressing, find the no of subnets and the no of hosts per subnet foe the following blocks:

(i) 122.45.77.32/20

(ii) A class B block having mask of 255.255.192.0

23. Write short note on any four of the following:

(a) The ATM reference model

(b) HDLC

(c) Salient difference between ISO-OSI and TCP/IP models.

(d) Network Topologies and their uses.

(e) Wireless networks.

24. (a) Differentiate between static and dynamic channel allocation.

(b) List out the main responsibilities of the network layer.

(c) Give two examples of a 'collision-free' protocol ?

(d) Why is IP called 'best-effort delivery' protocol?

(e) What is a transparent bridge?

(f) what are the two sub layers of data link layer called ?

(g) What are the other names of IEEE 802.11 protocol or standard?

(h) What is the baud rate of a standard 10 mbps Ethernet LAN ?

(i) What is a minimum data size of an Ethernet frame ?

25. Explain distance vector routing . What are its limitations and how are they overcome?

26.Explain pure-ALOHA and slotted- ALOHA systems.Give the expression for throughout for each,clearly explaining the various terms.

27. Explain 1-persistent, p-persistent and 0-persistent CSMA giving strong and weak points of each.

28. Explain network equipment used in wired-LANS and explain the function of Hub, Switch, and bridge.

29. Write short note on any four of the following:

(a) Token ring

(b) Various fields in Ethernet frame format

(c) Difference between congestion control and QoS(or Quality of service)

(d) FDDI

(e) Traffic shaping using token bucket algorithm

(f) CSMA/CD

30. If the transmitted code word is 10011000 and the received code word is 11001001. What is the error word ? Write transmitted code word , received code word and error word as polynomials.

31.Why transport layer protocols like TCP and UDP are called end-to-end protocols. What is the difference between them?

32. Differentiate between:

(i) Baseband co-axial cable and broad band coaxial cable

(ii) Optical fibre and twisted pair

(iii) SMTP and SNMP

33. What are the basic functions of the data link layer? Write down the basic features of HDLC protocol? Could HDLC be used as a data link protocol for a LAN? Justify your answer.

34. The physical layer service is a non-confirmed service. Assume that some bits of data are lost during transmission over physical media,which layer will detect the loss and take some remedial measures. Explain any one method clearly depicting how this operation is performed.

35. What are the advantages of cell switching that is used in ATM?

36. Outline and discuss the main fields in Ethernet IEEE 802.3 frame. What are the main objectives of preamble ?

37. What is the average number of transmission required to send a frame of length 1600 bytes correctly, if the bit error rate is 1 x 10^{-6}.

38. Explain what is meant by the term ‘integrated service digital network’. Give three reasons a company might choose an ISDN link in preference to a leased line.

39. Subnet the class C network address 198.67.25.0 into eight subnets. Why are the ‘all ones’ and ‘all zeroes’ subnets not used ?

40.What do you understand by the term ‘structured cabling’. State the main rules that should be used when installing a cable. Show that maximum cabling area for LAN for horizontal cabling runs is approximately 200m.

41. What are the various classes of IP addressing? Calculate the maximum number of class A, B and C network Ids.

42. Why is a data link layer switch preferred over a hub ?

43. Which device is needed to connect two LANs with different network Ids ?

44. When is a translating bridge used ?

45. Can a switch be used to connect two LANs with different network IDs ?

46. Write two ways in which computer applications differ from network applications ?

47. What is count to infinity problem ?

48. What was the reason for selecting a speed of 155.52 Mbps in the original ATM standard ?

49. Contrast link state and distance vector routing protocols, giving an example of each.

50. What is ISO-OSI reference model ? Compare it with TCP/IP reference model. Why TCP/IP reference model is more popular than OSI model ? Which layer is used for the following :

(i) to route packets

(ii) to convert packets to frame

(iii) to detect and correct errors

(iv) to run services like FTP, Telnet etc.

51. Discuss Shannon’s capacity. What implications does it have ?

52. Discuss how satellite network differs from traditional networks such as Ethernet, Tokenbus.

53. What is packet switching ? Explain two different approaches of packet switching. ?

54. Doscuss the different factors affecting congestion control algorithms. ?

55. How does a token ring network work ? In what way is it different from Ethernet ?

56. Describe and distinguish between FDMA, TDMA, and CDMA.

57. Discuss the following terms with respect to ATM: VPI, UNI, asynchronous, AAL, Cell , PVC.

58. What is sliding window protocol ? Differentiate between stop-and wait ARQ and Go-back-N protocol.

59. Differentiate between ISO-OSI and TCP/IP reference model.

60. Explain leaky bucket algorithm and compare it with token bucket algorithm.

61. Explain ATM reference model.

62. Explain different kinds of Switching techniques.

63. Differentiate between Link state and Distance Vector Routing algorithm.

64. Explain network layer in ATM,

65. Differentiate between IEEE 802.3, IEEE 802.4 and IEEE 802.5 standards.

66. Explain any three error detection and correction techniques.

67. Explain various cabling techniques used in IEEE 802.3 standard,

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