1. What advantage does a circuit-switched network have over a packet-switched Network?

Circuit Switched Network has a **fixed and dedicated channel** either between two end points in the network. On the other hand, packet switched networks don't have a fixed on dedicated connection. In packet switching the data packets are sent independent of each other over a wireless medium.

Probability of data packets getting lost or getting dropped is much higher in a packet-switched network compared to a circuit switched network. Circuit switch network is said to be great in terms of **low loss of data** due to reliable connection between the two end points established.

Along with low loss of data, circuit switched networks are much more reliable compared to packet switched networks as **dedicated paths provide no chance for delays**.

Circuit switched network provides **secure data transfe**r due to dedicated physical connection. Packet switched networks lack proper security in terms of data transmission compared to circuit switched networks.

2. What advantage does TDM have over FDM in a circuit switched network?

In FDM since signals are given different frequency slots, it leads to probability of crosstalk during data transmission. However in TDM signal transmission takes place in different time slots as a result of which the **possibility of cross talk is negligible compared to FDM**.

Entire fundamentals/logic of FDM are open to interference during data transmission while efficiency of data transmission in **TDM offers less chances of interference during.** So in simpler words **TDM is less prone to interferences** during data transmission.

The **infrastructure to implement FDM is more complicated compared to TDM** where only little wiring and simple embedded systems can be used to implement a simple TDM infrastructure. TDM infrastructure is simple and less complicated than FDM.

3. We consider sending real-time voice from Host A in the US to Host B in Singapore over a packet-switched network. Host A converts analog voice to a digital 65kbps bit stream and send these bits into 56-byte packets. There is one link between the host in US and the host in Singapore and the transmission rate is 1 Mbps and its propagation delay is 20 msec. As soon as Host A gathers a packet, it sends it to Host B. As soon as Host B receives an entire packet, it converts the packet's bits into an analog signal. How much time elapses from the time a bit is created (from the original analog signal at Host A) until the bit is decoded (as part of the analog signal at Host B)?

Given,
Since Host A converts the analog voice to a 64 kbps bit stream,
Then, total time required for packet generation would be (total no of bits / digital bit stream) i.e.
(56*8)/(64 * 103) = 6.89230 * 10^3 sec = 6.89230 msec. .... (1)

7 msec is the time required for the Host A to create a packet with given data and bit stream.

The packet transmission of this data is calculated as :
(Total no of bits / transmission rate) i.e.
(56 * 8) / (1 * 10 6) = 4.48 * 10 -4 sec = 448 µsec **........** (2)

Given, propagation delay is 20 msec. ....(3)

The total time until the bit is decoded is :

6.89230 μsec.+ 448 μsec + 20 msec. = **0.02734 sec**

Solution : It will take 27.34 msec for host A to send the data to Host B till it is decoded.

4. (9pts) Consider a Go-Back-N sliding window algorithm (1 packet is 1500 bytes long) running over a 100km point-to-point fibre link with bandwidth of 100 Mbps.
a. Compute the one-way propagation delay for this link, assuming that the speed of light is 2 x 108 m/s in the fiber.
b. Suggest a suitable timeout value for the algorithm to use. List factors you need to consider.
c. Suggest N to achieve 100% utilization in this link.

Given,
Bandwidth = 100 Mbps.
Distance = 100 km = 100 * 1000 meter.
Propagation Speed = 2 * 10^8 m/s.

a. Propagation delay = Distance / Speed = (100 * 1000)/ (2 * 10^8) = 5 *10^4 seconds= 0.5msec
One way propagation delay = 0.5 msec
Total round trip = 0.5 * 2 = 1 msec.
Transmission time for one packet
= (packet size) / BW
= 1500 * 8 / 100 mbps
= 1500*8/ (100*10^6) = 1.2 * 10-4 sec
= 1.2 * 10^4 sec

b. Timeout Time = Transmission time for one packet + Total Round trip time
= (1.2 * 10^4 sec) + (1 msec)
= (1.2 * 10^4 sec) + (1 * 10^3sec)
= 1.12 * 10^3sec
Suitable timeout time = 1.12 * 10^3sec

c. N (for 100% utilization) = {(Bandwidth * one way delay) / Packet Size}
= {((100*10^6) * 0.5 msec)/ 1500 * 8}
= 50000 / (1500 * 8)
= 4.1667
N = 4 for 100% utilization.

Q5: Suppose a 1-Gbps point-to-point link is being set up between the Earth and a new mars colony. The distance from the mars to the Earth is approximately 93.45 million mi, and data travels over the link at the speed of light—3×10^8 m/s.
a. Calculate the minimum RTT for the link.
b. Using the RTT as the delay, calculate the delay × bandwidth product for the link.
c. What is the significance of the delay × bandwidth product computed in (b) ?

Given, Distance = 93.45 million miles = 93,450,000 miles = 150393196.8 km= 150393196800 meters

a. Minimum RTT is twice the time required to travel from earth to mars colony. I.e.

Minimum RTT = propagation delay * 2 where
Propagation delay = Distance/Time
Distance = 150393196800 meter and Time = 501.310656 sec

Time required to travel from earth to mars colony,
(150393196800 meter)/ 3*10^8 meter/sec = 501.310656 sec

So minimum RTT = 501.310656 * 2 secs = 1002.62 sec
Minimum RTT = 1002.62 sec

b. Delay * Bandwidth Product = 1002.62 sec * 1Gbps
= 1002.62 * (1* 10^9 bits/sec)
= 1.00262 * 10^12 bits/sec or 1002.62 Mbits/sec or
= 125.3275 Megabytes/sec
delay × bandwidth product = 1.00262 * 10^12 bits/sec

c. 125.3275 Megabytes/sec obtained in (b) is a delay bandwidth product indicating the maximum amount of data at any given time on the network. Its the product of network data link capacity. Its means the system can transmit a maximum of 125.3275 Megabytes/sec before receiving any acknowledgement. Sender can transmit a maximum of 125.3275 Megabytes/sec data on the line before receiving the notification or acknowledgement from the receiver that the first bit has been acknowledged.

Q6 : Host A wants to send a 1,000 MB file to Host B. The Round Trip Time (RTT) of the Duplex Link between Host A and B is 160ms. Packet size is 1KB. A handshake between A and B is needed before data packets can start transferring which takes 2xRTT. Calculate the total required time of file transfer in the following cases. The transfer is considered complete when the acknowledgement for the final packet reaches A.
a. The bandwidth of the link is 4Mbps. Data packets can be continuously transferred on the link.
b. The bandwidth of the link is 4Mbps. After sending each packet, A need to wait one RTT before the next packet can be transferred.
c. Assume we have "unlimited" bandwidth on the link, meaning that we assume transmit time to be zero. After sending 50 packets, A need to wait one RTT before sending next group of 50 packets.
d. The bandwidth of the link is 4Mbps. During the first transmission A can send one (21-1) packets, during the 2nd transmission A can send 22-1 packets, during the 3rd transmission A can send 23-1 packets, and so on. Assume A still need to wait for 1 RTT between each transmission.

a. Transmit time per packet = Packet Size / BW
= 1KB / 4 Mbps
= (1*1024* 8) bits / 4000000 bits/sec
= 2.048 msec
No of packets = File Size / packet Size
= 1000 MB / 1KB
= 1000000 KB / 1KB
= 1000000 packets
Total transmit time is no of packets per packet transfer time i.e.
= (1000000 packets) * (2.048 * 10 ^3 sec)
= 2048 secs
Total time = 2 RTT + Total transmit time + PD
= (2* 0.16 sec) + (2048 sec) + (0.16 sec / 2)
= 2048.4 sec

Total required time of file transfer = 2048.4 sec

b. Total packets here are 1000000 packets as seen in (a) so total time =
= 2048 secs * (999999 * 0.16 sec)
= 2048 sec * 159999.84
= 327679672.3 sec
. **Total required time of file transfer =327679672.3 sec**

c. We have to send 1000000 packets, so 1000000/50 = 20000 RTTs for total data transfer
But last 50 packets only need RTT/2 = 0.16/2 = 0.08 sec
= 2 * RTT + (20000- 0.08) * RTT
= (2 * 0.16) + (19999* 0.16)
= 3200.16 sec
Total required time of file transfer= 3200.16 sec

d. As per question the format in which packets are sent are as follows:
1000000 = 1 + 2 + 4 + 8 + 16 + 32 + 64 + 128 + 256 + 489 +
1024+2048+4096+8192+16384+32786+65536+131072+262144+524288

Total 20 times !

Hence total time = Handshake time + Transmission time for 1000000 packets + 20 RTT + PD
= 2 * 0.16 + 2048 sec + 20 * 0.16 + 2 * 0.16
= 2051.84 sec
Total required time of file transfer = 2051.84 sec

Q7 : Determine the width of a bit on a 10 Gbps link. Assume a copper wire, where the speed of propagation is 2.3 ∗ 108 m/s.

Answer :

Given,
Speed of link i.e. Data Rate = 10 Gbps = 10 * 10^9 bits/sec
Speed of propagation = 2.3 * 10^8 m/s

Width of a bit = S/r = (2.3 * 10^8 m/s) / (10 * 10 ^9 b/s) = 0.023 meters

Hence width of a bit on a 10 Gbps link of copper wire with speed of propagation as 2.3 * 10^8 m/s is **0.023 m/bit**

Q8 : Suppose two hosts, A and B, are separated by 20,000 kilometres and they are connected by a direct link of R=1Gbps. Suppose the propagation speed over the link is 2.5 x 108 meters/sec.
a. Calculate the bandwidth delay product (BDP) of the link.
b. Consider sending a file of 800,000 bits from Host A to Host B as one large message. What is the maximum number of bits that will be in the link at any given time?
c. What is the width (in meters) of a bit in the link?
d. Suppose now the file is broken up into 20 packets with each packet containing 40,000 bits. Suppose that each packet is acknowledged by the receiver and the transmission time of an acknowledgement packet is negligible. Finally, assume that the sender cannot send a packet until the preceding one is acknowledged. How long does it take to send the file?

Given,
Distance = 20000 km = 20000 * 1000 m
Rate = 1 Gbps = 10^9 bits/sec
Speed = 2.5 * 10^8 m/s

a. BDP is "Rate * (Distance / Speed)"
BDP = (1 * 10^9 b/s) *((20000 * 1000) / (2.5 * 10^8 m/s))
= 80000000 bits
BDP is 80000000 bits i.e. 80 Mb

b. Maximum bits on the link at any given time is calculated as total no of bits plus gain bandwidth product
Max bits on link = total bits + BDP
= 80000 bits + 80000000 bits
= 80800000 bits
Maximum bits on the link at any given time = 80800000 bits

c. Speed of link i.e Data Rate = 1 Gbps = 1* 10^9 bits/sec
Speed of propagation 2.5 * 10^8 m/s

Width of a bit = S/r = (2.5 * 10^8 m/s) / (1 * 10^9 b/s) = **0.25 meters**

d. With new proposed method for transmission for file broken into 20 packets,
With each packet continuing 40,000 bits
PD = (20000 * 10^3 m) / (2.5 * 10^8 m/sec)
= 0.08 sec
Transmission time for one packet = 40000 / 1*10^9 = 0.00004 sec
Total transmission time = 20 * (Transmission time for one packet + 2 * PD)
= 20 *(0.08 sec *2 + 0.00004 sec)
= 20 * (0.16004)
= 3.2008 sec
It takes 3.2008 sec for file transfer

Q 9 : Suppose there is a 10 Mbps microwave link between a geostationary satellite and its base station on Earth. Every minute the satellite takes a digital photo and sends it to the base station. Assume a propagation speed of 2.4 x 108 meters/sec. Geostationary satellite is 36,000 kilometres away from earth surface
a. What is the propagation delay of the link?
b. What is the bandwidth-delay product, R x (propagation delay)?
c. Let x denote the size of the photo. What is the minimum value of x for the microwave link to be continuously transmitting?

Given,
Distance = 36000 km = 36000 * 1000 meter
Speed = 2.4 * 10^8 m/sec
Rate = 10 Mbps = 10 * 10 ^6 bits/sec

a. Propagation delay is the result of distance to be covered per speed
i.e. 36000 * 1000 meter / 10* 10^6 bits/sec
= 0.15 sec

b. Gain bandwidth product is Propagation delay in sec multiplied by Rate
BDP = Rate * Propagation delay
= (10 * 10^6) * (0.15)
= 1500000 bits

c. Minimum value to transmit a photo of size x is
Speed of microwave link * time interval (ideally 60 sec)
Minimum value of x = 10 * 10^6 * 60
= 6 * 10^8 bits
OR
Given there is a 10 Mbps link between satellite and earth station
Hence, 10 Mbits per seconds
So, 600 Mbits per 60 seconds
Therefore, 600 Mbits / 8 = 75 MBytes

Q10 : Explain collision domain and broadcast domain with respect to a hub, switch, and a router.

HUB : Operating at a physical level, it gets signal from one node and forwards that signal to other nodes connected, A hub either creates or expands a collision domain. In an ethernet or TCP IP network collision domain is a group of nodes that can hear each other. One node if it's using a hub others must wait. If collisions get too large then we may run into dozens of problems and the collision domain will increase. The more the number of hubs we use the larger the chances of collision domains occurring.

SWITCH : Operating at layer 2 i.e. data link. It separates collision domains due to smartness. It gives every conversation a full bandwidth of the network. For example if node A is talking it gives full bandwidth to node B to hear it apart from any other node. Both A and B can talk without collision.

Router : Operating at the network layer, routers only forward IP address based packets. They separate collision domains as well as broadcast domains. If multiple LAN are connected to a router, inter LAN communication will not be heard or read by other LAN. A router thus separates both collision and broadcast domain as it creates definite boundaries between its networked LAN acres and does not allow communication between them.

To summarize, Hub creates and expands both collision domain and broadcast domain.
Switch separates collision domain but creates broadcast domain. Router separates both collision and broadcast domain.

Q11 : . (5pts) Consider the following networked computers connected by Bridge X and Y. Bridge X has interface 1,2 and 3. Bridge Y has interface 1 and 2. Assume at the beginning the address tables of Bridge X and Y are all empty. Write down the address tables of Bridge X and Y after the following communication finished (Assume that the receiver does not respond to the packet sent by the sender.).

1. A send a packet to C
2. B send a packet to D
3. C send a packet to E
4. E send a packet to B
5. D send a packet to A

Bridge X :

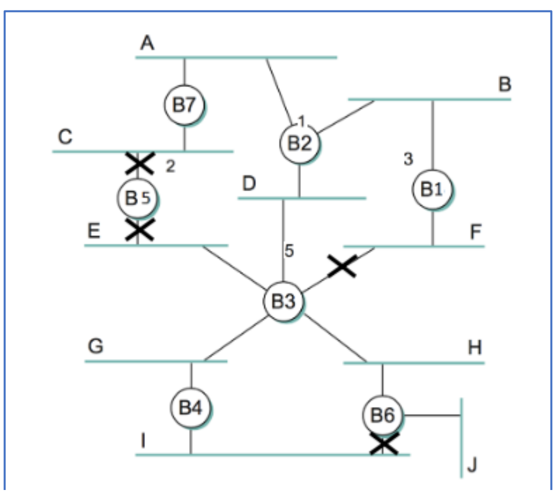
Address	Interface

A	1
B	1
C	2
E	3
D	3

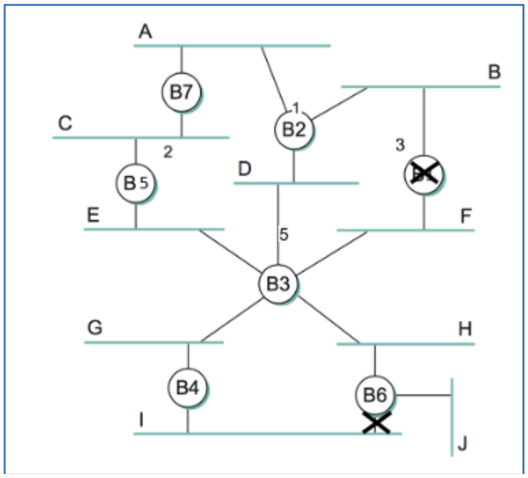
Bridge Y :

Address	**Interface**
A	1
B	1
C	1
E	2
D	1

Q12 : 5pts) Given the extended LAN shown in Figure 2, indicate which ports are not selected by the spanning tree algorithm. Note that the bridge with the smallest ID becomes a root.



Q13 :Still considering Figure 2. If Bridge B1 suffers catastrophic failure. Again indicate which ports are not selected by the spanning tree algorithm.



Q14 Consider the GBN protocol with a sender window size of N=4 and a sequence number range of 1,024. Suppose that at time t, the next in-order packet that the receiver is expecting has a sequence number of k. Assume that the medium does not reorder messages. Answer the following questions:
(a) What are the possible sets of sequence numbers inside the sender's window at time t? Justify your answer. (2 pts)
(b) What are all possible values of the ACK field in all possible messages currently propagating back to the sender at time t? Justify your answer. (2 pts)
(c) With the Go-Back-N protocol, is it possible for the sender to receive an ACK for a packet that falls outside of its current window? Justify your answer with an example. (2 pts)

a) Given, window size N = 4. Let's assume that sender sent up to K-1 packets, currently accepting packet K. Additionally let's say that all of the previous packets has been acknowledged. Once the sender got all the acknowledged packets [K, K+N-1] can be the correct representation of the senders window.

   Instead if we assume, none of the ACK's has been received on the receiver side, same senders window will be indicated by [K-N, K+N-1-N] i.e. [K-N , K-1].

   Therefore, based on given conditions that sender's window size is of 4 (N=4), ranges somewhere [K-4, K].

b) It can be noted that receiver is waiting for packet K then it has received K-1 packet successfully with ACKs too. Along with ACKs it may have also received N-1 packets prior. If by any chance still none of the ACKs has been received then [K-N,K-1] ACK's is in still in queue at the received.

   Now since sender already have sent [K-N, K-1] packets, so K-N-1 ^{tH} acknowledgement must have been also received already. Since ACK for k − N − 1 has been sent by receiver, it will never send an ACK that is less that k − N − 1.

   Thus the range of in-flight ACK values can range from k − N − 1 to k − 1.

c) Yes, it is possible. Consider the following example scenario. Suppose the sender has a window size of 3 and sends packets 1, 2 and 3 at time t0. At t1, the receiver ACKs packets 1, 2 and 3. At t2 , the sender times out and resends 1, 2 and 3. At t3, the receiver receives the duplicates and re-acknowledges 1, 2 and 3. At t4 , the sender receives the ACKs that the receiver sent at and advances its window to packets 4, 5 and 6. Finally, at t5, the sender receives the ACKs that the receiver sent at t2 . These ACKs are outside its window.