NWC203c PE

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(Đây chỉ là giải lại theo góc nhìn khác, chứ hoàn toàn là sao chép. Tác giả tôn trọng bản quyền của các bài tập này.)

\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*

**Q.1:** Explain the difference between connectionless unacknowledged service and connectionless acknowledged service. How do the protocols that provide these services differ?

Solution:

Connectionless:

Connectionless service comes with a single free-standing data unit for all transmissions.

In this, each unit contains all of the protocols that control information necessary for delivery perspective, but this also contains no provision for sequencing or flow control.

(Dòng trên giải thích Connectionless)

>> Acknowledged:

This is achieved by the use of ACK and NAK control messages.

-- > These types of protocols are well suited for communication over the network, where high layers are very sensitive to loss and can have a significant probability of error in these underlying networks.

Example: HDLC, which offers for unnumbered acknowledgment service(setup and release).

(Dòng trên giải thích cho sự khác biệt của Acknowledged)

>> Unacknowledge:

This comes with a very simpler version and provides faster communication for networks, which are inherently reliable or provide service to a higher layer, that can tolerate loss in the information, or which has built-in error control/recovery feature.

(Dòng trên giải thích cho sự khác biệt của Unacknowledge)

**Q.2:** Explain the difference between connection-oriented acknowledged service and connectionless acknowledged service. How do the protocols that provide these services differ?

Solution:

Less & Oriented:

>> Connection-oriented:

In this type of service, a setup phase will be initialized between sender and receiver, to establish a context for transferring the information

This connection is provided to the sender for all SDUs.

This service requires a stateful protocol, which is used to keep track of sequence numbers, and timers.

(Dòng trên giải thích cho sự khác biệt của Connection-oriented)

>> Connectionless:

Here, there will be no prior context provided for transferring the information between sender and receiver.

The sender will pass its SDU to an underlying layer without any notice.

And in this, the sender requires an acknowledgment of SDU delivery.

The protocols are very different in these services

this service also does not require transmitting protocols to track the acknowledgment of PDU.

After receiving the PDU, the receiver needs to send acknowledgment, If not received in time, then it will return failure.

(Dòng trên giải thích cho sự khác biệt của Connectionless) **Q.3:** Explain the differences between PPP and HDLC. Solution:

HDLC is a short form of High-level Data Link Control that does the data encapsulation. PPP is an acronym for Point-to-Point Protocol that can be used by different devices without any data format change.

(Dòng trên định nghĩa HDLC và PPP)

A few major differences are as below:

For communication through HDLC, a bit-oriented protocol is used for point-to-point links as well as for multipoint link channels. However, PPP uses a byte-oriented protocol for point-to-point links at the time of communication.

HDLC does the encapsulation for synchronous media only whereas PPP can do the encapsulation for synchronous as well as for asynchronous media.

HDLC can be used only for CISCO devices whereas PPP can be easily used for other devices.

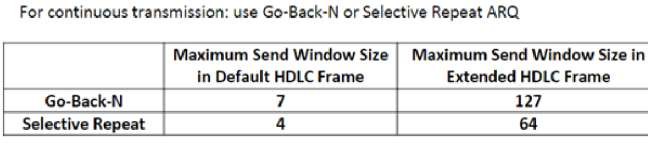
(Những dòng trên mô tả sự khác biệt của HDLC và PPP)

**Q.4:** A 1.5 Mbps communications link is to use HDLC to transmit information to the moon. What is the smallest possible frame size that allows continuous transmission? The distance between earth and the moon is approximately 375,000

3 x 108

km, and the speed of light is Solution:

meters/second.

We know that Default HDLC Frame and Extended HDLC Frame depend on Go-Back-N and Selective Repeat, and can be presented as below

(Đây là bảng tra cứu thông tin của Default và Extended, bảng này phải tự vẽ ra chứ đề không có sẵn)

D (Quãng đường = Distance) = 375,000 km = 375 x 106 m

c (Vận tốc ánh sáng = Speed of Light) = 3 x 108 m

Then, we can calculate Round Trip Propagation Delay (Trì hoãn do quãng đường) by this formula

2tprop=Dc=375 x 1063 x 108=2,5s

We know that

NnfR=2tpropnf=2tprop RN (\*)

In which, nf is Possible Frame Size (bits), Mbps is the number of Megabyte Per Second.

R = 1,5 Mbps so that R = 1,5 x 106 bps.

>> Go-back-N:

Default HDLC Frame: N = 7, substitute to (\*) then we have

2,5×1,5 ×1067=535714 (bits)

Extended HDLC Frame: N = 127, substitute to (\*) then we have

2,5×1,5 ×106127=29527 (bits)

>> Selective Repeat

Default HDLC Frame: N = 4, substitute to (\*) then we have

2,5×1,5 ×1064=937500 (bits)

Extended HDLC Frame: N = 64, substitute to (\*) then we have

2,5×1,5 ×10664= 58594 (bits)

**Q.5**: Suppose HDLC is used over a 1.5 Mbps geostationary satellite link. Suppose that 250-byte frames are used in the data link control. What is the maximum rate at which information can be transmitted over the link?

Solution:

We have

R = 1.5 Mbps or R = 1,5 x 106 bps, and nf = 250 bytes or nf=2000 bits (250 x 8). The distance that the information must travel is the Earth-to-Satellite Distance, or D ≈ 36000 km = 3,6 x 107 m.

The Speed of Light c = 3 x 108 m.

We can calculate the propagation delay and processing rate as follows:

tprop= Dc=3,6×1073×108=0,12 s=120 ms tf=nfR=20001,5×106 = 0,00133 s = 1,33 ms

In which, tprop is propagation delay, tf is processing rate.

We can use either Go-Back-N or Selective Repeat ARQ. The default window size is N = 7 (with a 3bit sequence number). 0 1 2 3 4 5 6 7 …

The maximum information rate is achieved with no error, and hence, no retransmission.

tcycle=tf+2 tprop=1,33+2×120=241,33 ms

In which, tcycle is the minimum time to transmit a group of N packets.

n=N×nf=7×2000=14,000 bits

Rmax=ntcycle=14000241,33=58,01 kbps

In which, n is number of bits transmitted in a cycle, Rmax is number of bits sent in a cycle / minimum cycle time.

If the extended sequence numbering option (7-bit) is used, the maximum send window size would be N = 27 – 1 = 127, and hence, the maximum information rate is:

n=N×nf=127×2000=254000 bits

Rmax=ntcycle=254000241,33=1052,5 kbps=1,0525 Mbps

**Q.6:** Suppose that a multiplexer receives constant-length packet from N = 60 data sources. Each data source has a probability p = 0.1 of having a packet in a given T- second period. Suppose that the multiplexer has one line in which it can transmit eight packets every T seconds. It also has a second line where it directs any packets that cannot be transmitted in the first line in a T-second period. Find the average number of packets that are transmitted on the first line and the average number of packets that are transmitted in the second line.

Solution:

Firstly, we find out the probability of the k packets that have reached the T - second. It can be computed with the help of binomial distribution that has parameters as N = 60 and shows the probability of p = 0.1.

The average number for the arrivals of the packets can be given as Np=6. Now, calculate the average number of packets received through the first line as below:

X= i=08iNipi1-pN-i

= i=08i60i0,1i×0,960-i= 4,595

Now, the average number of packets received is 4.595 that gets transmitted through the first line. The remaining will get transmitted by the second line. Now, the average number of packets transmitted through the second line per T second can be obtained as below:

Y=Np-X=6-4,595=1,405

Therefore, it will transmit 1.405 packets on average per T second from the second line.

**Q.7:** Consider the transfer of a single real-time telephone voice signal across a packet network. Suppose that each voice sample should not be delayed by more than 20 ms.

1. Discuss which of the following adaptation functions are relevant to meeting the requirements of this transfer: handling of arbitrary message size; reliability and sequencing; pacing and flow control; timing; addressing; and privacy, integrity and authentication.
2. Compare a hop-by-hop approach to an end-to-end approach to meeting the requirements of the voice signal.

Solution:

a, Message size is important because in real-time signals of voice it is necessary to transfer a fixed packet size of that holds no more than 20 ms of the speech signal. The handling of arbitrary message size is not as important as long as the desired packet size for voice can be handled.

Sequencing is essential because each packet needs to arrive in the same sequence that it was generated. Reliability is moderately important since voice transmission can tolerate a certain level of loss and error.

Pacing and flow control are not as important because the synchronous nature of the voice signal implies that the end systems will be matched in speed.

Timing, for real-time voice transfer, is important because this adaptation function helps to control the jitter in the delivered signal.

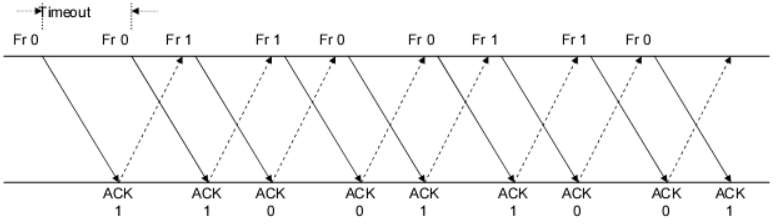
Addressing is only during the connection setup phase if we assume some form of virtual circuit packet switching method.

Privacy, integrity, and authentication have traditionally not been as important as the other issues discussed above.

b, If the underlying network is reliable then the end-to-end approach is better because the probability of error is very low so processing at the edge suffices to provide acceptable performance.

If the underlying network is unreliable then the hop-by-hop approach may be required. For example, if the probability of error is very high, as in a wireless channel, then error recovery at each hop may be necessary to make effective communication possible.

**Q.8:** Consider the Stop-and-Wait protocol as described. Suppose that the protocol is modified so that each time a frame is found in error at either the sender or receiver, the last transmitted frame is immediately resent.



1. Show that the protocol still operates correctly.
2. Does the state transition diagram need to be modified to describe the new operation?
3. What is the main effect of introducing the immediate-retransmission feature? Solution:

a, The sender in the stop-and-wait protocol described in the chapter retransmits a

frame when an acknowledgment is not received in time. The modified protocol says that the frame is retransmitted every time the sender or receiver sees an error.

Therefore, the only difference is that frames are retransmitted sooner. So, the protocol will work correctly.

b, No. The state transition diagram will stay the same.

c, The error recovery process will be faster with this modified protocol.

**Q.9:** Suppose that two peer-to-peer processes provide a service that involves the transfer of discrete messages. Suppose that the peer processes are allowed to exchange PDUs that have a maximum size of M bytes including H bytes of

header. Suppose that a PDU is not allowed to carry information from more than one message.

Solution:

Bytes each to be transmitted in several PDUs in order to exchange messages of any size. A single PDU must include all small messages.

Peer processes must exchange information that permits messages to be reassembled at the recipient. The message length, for example, could be included in the first PDU. A message end-of-message marker could be included in the last PDU. In connection- oriented networks, sequence numbers can be used to detect loss, while in connectionless networks, they can be used to aid in message reconstruction. Finally, because variable-size PDUs are allowed, the PDU size must be specified in the PDU header.

In this instance, each PDU must be identified with a stream ID in addition to all of the header information specified in (b), so that the receiver may treat each stream separately while reassembling messages.

**Q.10:** A 1 Mbyte file is to be transmitted over a 1 Mbps communication line that has a bit error rate of

p = 10-6.

1. What is the probability that the entire file is transmitted without errors. We conclude that it is extremely unlikely that the file will arrive error free.
2. The file is broken up into N equal-sized blocks that are transmitted separately. What is the probability that all the blocks arrive correctly without error? Does dividing the file into blocks help?
3. Suppose the propagation delay is negligible, explain how Stop-and-Wait ARQ can help deliver the file in error-free form. On the average how long does it take to deliver the file if the ARQ transmits the entire file each time?

Solution:

(Câu này tham khảo được trên mạng và chính xác, các bạn để ý cho tôi là

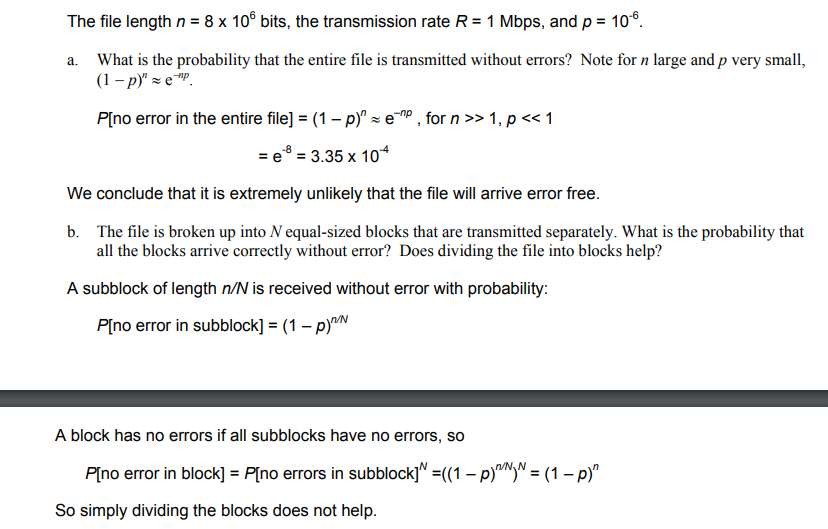
n = 8 x 108 bits là do 1 Mbyte = 106 byte = 8 x 108 bits do 1 byte = 8 bits,

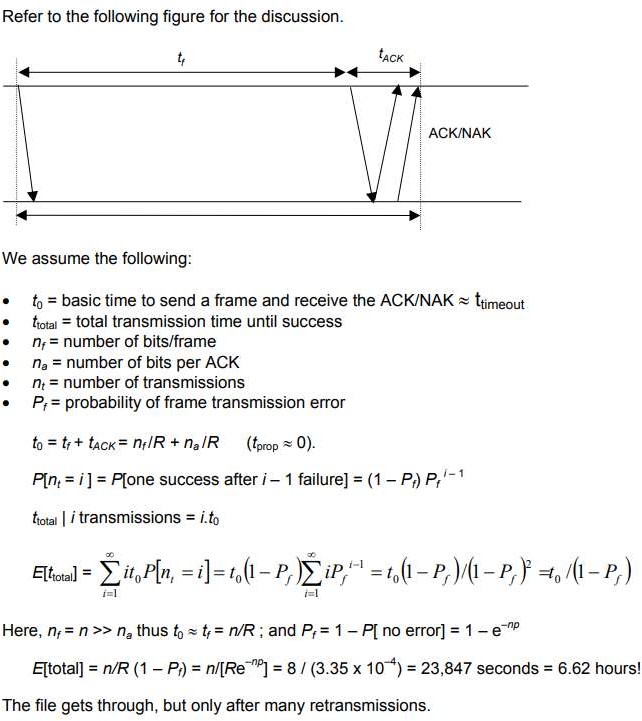
bên cạnh đó, nhớ rằng e = 3,35 là hằng số,

là hiển nhiên).

còn p = 10-6

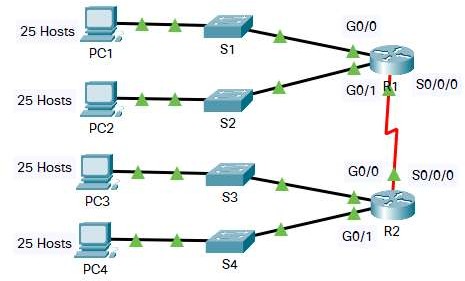
R = 1 Mbps = 106 bps,





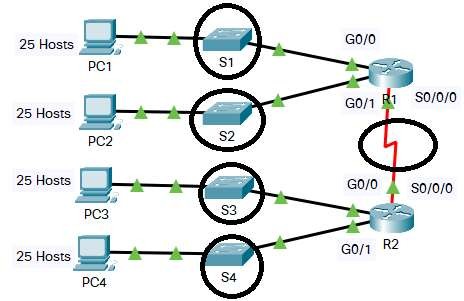
Câu c này khá khó, nhưng thay số thì dễ. Các bạn chú ý các phần nên thay là n,p và R.

# Q.11:



In this activity, you are given the network address of 192.168.100.0/24 to subnet and provide the IP addressing for the Packet Tracer network. Each LAN in the network requires at least 25 addresses for end devices, the switch and the router. The connection between R1 to R2 will require an IP address for each end of the link.

1. Based on the topology, how many subnets are needed?
2. How many bits must be borrowed to support the number of subnets in the topology table?
3. How many subnets does this create?
4. How many usable hosts does this create per subnet? Solution:

a,

We saw that S1, S2, S3, S4, S0/0/0 are the subnets needed, so that there is 5 subnets are needed.

b,

We call N is the number of bits, then N is the smallest number that satisfies

4×2N -2≥25

Then we got N = 3.

(Chú ý rằng 4 là số các subnet có thực – các subnet có hình hộp , là các subnet S1,S2,S3,S4, nó không tính subnet ảo – là subnet có hình sợi dây như là S0/0/0.)

c,

We saw that number of bits N = 3, then the number of subnets does this create is 23

=8.

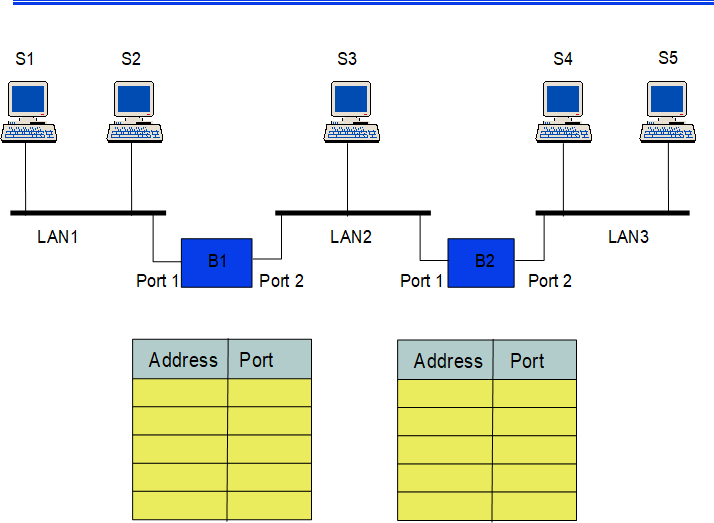
d, The number of host is 2N -2, where N is the smallest number that satisfies 2N- 2≥25,

we saw that N = 3, so that number of host is

2N -2=23-2=30

# Q.12:

Five stations (S1-S5) are connected to an extended LAN through transparent bridges (B1-B2), as shown in the following figure. Initially, the forwarding tables are empty. Suppose the following stations transmit frames: S1 transmits to S5, S3 transmit to S2, S4 transmits to S3, S2 transmits to S1, and S5 transmits to S4. Fill in the forwarding tables with appropriate entries after the frames have been completely transmitted.

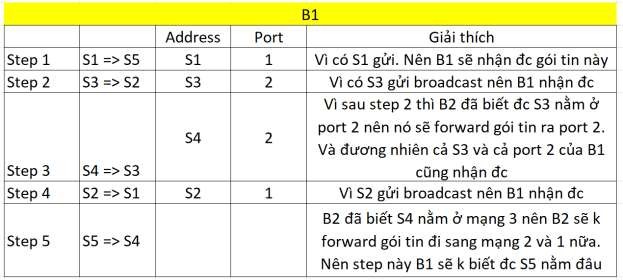
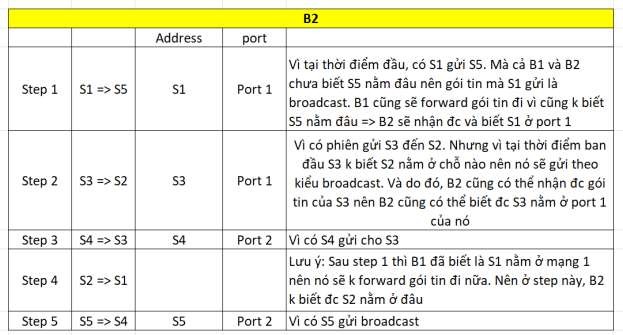


Solution:

(Câu này thầy đã giải thích khá kĩ, và tôi chỉ English hóa nó lại, để hiểu câu này, cần tham khảo hình và đọc kĩ giải thích là có thể hiểu được).

Trước tiên, phải hiểu là ở đây chúng ta có 3 mạng LAN, và mỗi 1 mạng LAN đều được bố trí theo dạng BUS nên khi 1 máy truyền dữ liệu, nó sẽ truyền theo kiểu broadcast (tức là truyền đi đến tất cả các máy và các cổng mạng)

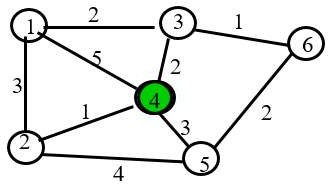
Firstly, we know that we have 3 types of LAN, and each LAN is arranged follow BUS. Then, if a device sends data, it will send according to broardcast type (send to any device and internet port).



Giải thích khá dễ nếu English hóa, các bạn tự cố gắng nha.

# Q.13:

1. Consider the network in Figure.



a. Use the Dijkstra algorithm to find the set of shortest paths from node 4 to other nodes.

We call that node that have number N is V(N) (i.e the green one is V(4))

b, Find the set of associated routing table entries (Destination, Next Hop, Cost)

|  |  |  |
| --- | --- | --- |
| Destination | Cost | Next Hop |

Solution:

a,

|  |  |  |  |  |  |  |
| --- | --- | --- | --- | --- | --- | --- |
|  | V(4) | V(1) | V(2) | V(3) | V(5) | V(6) |
| V(4) |  | 5, V(4) | 1,V(4) | 2,V(4) | 3,V(4) | ~ |
| V(4),V(2) |  | 4,V(2) |  | 2,V(4) | 3,V(4) | ~ |
| V(4),V(2),V(3) |  | 4,V(2)  4,V(3) |  |  | 3,V(4) | 3,V(3) |
| V(4),V(2),V(3),  V(5),V(6) |  | 4,V(2)  4,V(3) |  |  |  |  |

Then, we have

The shortest part from V(4) to V(1) is 4, and the path is V(4) -> V(2) -> V(1) or V(4) -> V(3) -> V(1)

The shortest part from V(4) to V(2) is 1, and the path is V(4) -> V(2) The shortest part from V(4) to V(3) is 2, and the path is V(4) -> V(3) The shortest part from V(4) to V(5) is 3, and the path is V(4) -> V(5)

The shortest part from V(4) to V(6) is 3, and the path is V(4) -> V(5) -> V(6)

b,

Dijkstras Algorithm:

Starting vertex is 4.

So the last row value is the shortest path from node 4 to all other nodes.

Each time it updates the path using relaxation:

distu+costu,v<distv

then update the

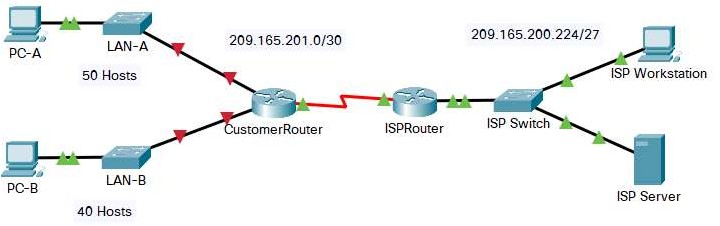
In which,

distv=distu+cost(u,v)

dist(u) is distance of vertex u from starting vertex ccost(u,v) is cost/weight of the edge uv

dist(v) is distance of vertex u from starting vertex

# Q.14:



You are a network technician assigned to install a new network for a customer. You must create multiple subnets out of the 192.168.12.0/24 network address space to meet the following requirements:

* The first subnet is the LAN-A network. You need a minimum of 50 host IP addresses.
* The second subnet is the LAN-B network. You need a minimum of 40 host IP addresses.
* You also need at least two additional unused subnets for future network expansion.

**Note**: Variable length subnet masks will not be used. All of the device subnet masks should be the same length.

Answer the following questions to help create a subnetting scheme that meets the stated network requirements:

1. How many host addresses are needed in the largest required subnet?
2. What is the minimum number of subnets required?
3. The network that you are tasked to subnet is 192.168.12.0/24. What is the /24 subnet mask in binary?
4. The subnet mask is made up of two portions, the network portion, and the host portion. This is represented in the binary by the ones and the zeros in the subnet mask.

In the network mask, what do the ones and zeros represent?

1. When you have determined which subnet mask meets all of the stated network requirements, derive each of the subnets. List the subnets from first to last in the table. Remember that the first subnet is 192.168.12.0 with the chosen subnet mask.

|  |  |  |
| --- | --- | --- |
| **Subnet Address** | **Prefix** | **Subnet Mask** |

Solution:

a, How many host addresses are needed in the largest required subnet? No of host addresses required = 2^6-2 = 62

Explanation : No. of subnets required = 4 which implies that the network is divided into four parts and

the given IP address i.e 192.168.12.0 is a Class C IP address . Therefore the max number of hosts possible in each subnet is 62 .

b, What is the minimum number of subnets required?

According to the question , two subnet are required for LAN-A and LAN-B and two subnets are needed to be left for future use Therefor the total number of subnets are 4 .

c, The network that you are tasked to subnet is 192.168.12.0/24. What is the /24 subnet mask in binary?

Subnet mask for any network is obtained by changing the net id bits to 1's and host id bits to 0's. Since the given network is a Class C network so the number of netid bits are 24 and the number of host id bits are 8 and there are two bits reserved for subnet

id bits . Therefore the subnet mask for network is 11111111.11111111.11111111.11000000 in binary.

d,

In the nerwork maslk the ones represent the net id bits and the zeroes represent the host id bits.

e, Subnets for LAN-A are :

a.) Starting network address is 192.168.12.00000000 b.) 192.168.12.00000001

c.) 192.168.12.00000010

d.) 192.168.12.00000011

and so on .........

Last network for the subnet will be 192.168.12.00111111 Subnets for LAN-B are :

a.) Starting network address is 192.168.12.01000000 b.) 192.168.12.01000001

c.) 192.168.12.01000010

d.) 192.168.12.01000011

and so on ......

Last network for the subnet will be 192.168.12.01111111 Other two subnets are :

Starting address of third subnet is : 192.168.12.10000000

Last address of the third subnet is : 192.168.12.10111111 Starting address of forth subnet is : 192.168.12.11000000 Last address of the forth subnet is : 192.168.12.11111111

# Q.15:

Suppose that Selective Repeat ARQ is modified so that ACK messages contain a list of the next m frames that it expects to receive.

Solutions follow questions:

1. How does the protocol need to be modified to accommodate this change?
2. What is the effect of the change on protocol performance? Solution:

a,

2 things are needed to be changed:-

The frame header needs to be modified to recieve the list of frames and Since the reciever explicitly indicates which frames are needed to be transmitted.

Change in transmitter operation is needed. If the recieved list contains m oldest frames that are yet to be recieved , then it can be used to skip retransmission of frames that have already been received.

b,

Performance will surely increase if the error rate is high or delay is high. A single frame can ask for the retransmission of several frames.

The complexity of the protocol will surely increase relative to the unchanged Selective repeat ARQ