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

Connection Less:

-- > 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 includes no provision for sequencing or flow control.

>> 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 lose and can have a significant probability of error in these underlying networks.

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

>> Unacknowledged:

-- > This comes with a significantly simpler version and provides faster communication for networks that 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.

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

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 used to keep track of sequence numbers and timers.

>> Connectionless:

-- > Here, no prior context will be 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, it will return failure.

Câu 3: Explain the differences between PPP and HDLC.

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 different devices can use without any data format change.

Both use the same bit sequence 01111110 to mark the start and end of data frames.

A few significant differences are as below:

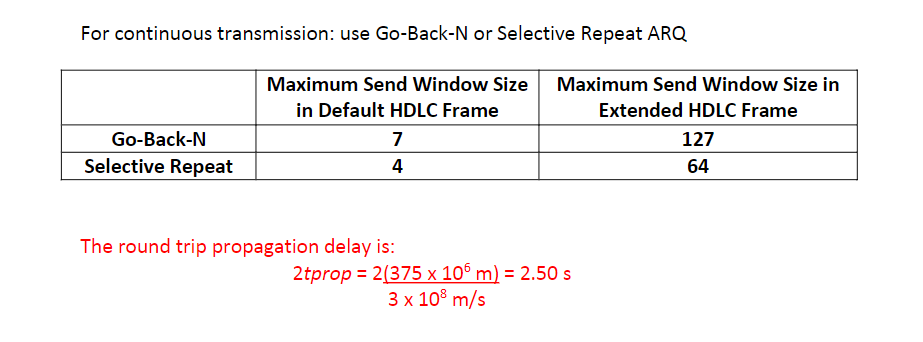
For communication through HDLC, a bit-oriented protocol is used for point-to-point links as well as for multipoint link channels. However, PPP uses a byte-oriented protocol for point-to-point links at the time of communication.

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.

Câu 4:

A 1.5 Mbps communications link is used 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 km, and the speed of light is 3 x 108 meters/second.



The round trip propagation delay is:

**Go-Back-N:**

If N = 7 : -> nf  535714 bits/frame

If N = 127 : -> nf  29528 bits/frame

**Selective Repeat:**

If N = 7 : -> nf = 973500 bits/frame

If N = 127 : -> nf  58594 bits/frame

**Câu 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?

R = 1.5 Mbps, and nf =250 bytes or 2000 bits (250 x 8). The distance that the information must travel is the earth-to-satellite distance, or d ≈ 36,000 km.

The speed of light c is 3 x 108.

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

Tprop = = = 120 ms

Tf =

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 = minimum time to transmit a group of N packets = Tf + 2tprop = 1.33 + 2x120 = 241.33 ms

n = no. of bits transmitted in a cycle = N x nf = 7x2000 = 14,000 bits

Rmax = = = 58 kbps

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:

Rmax = = = 1.052 Mbps.

**Câu 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.

Answer:

First, 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:

4.59

Now, the average number of packets received is 4.59 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:

$$= 6-4.59=1.41

Therefore, it will transmit 1.41 packets on average per T second from the second line.

**Câu 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.

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.

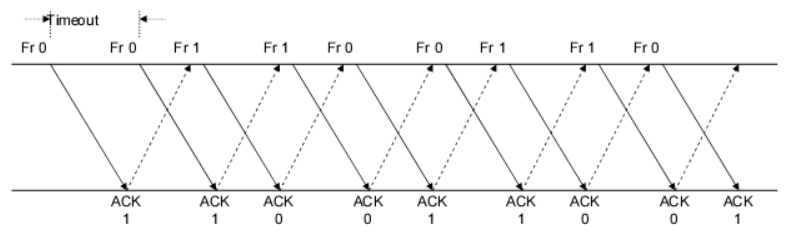
Privacy, integrity, and authentication have traditionally not been as important as the other issues discussed above.

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.

**Câu 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?

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.

Câu 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.

Bytes are each to be transmitted in several PDUs 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.

Câu 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 errorsWe 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?

1 MB=1024KB

1KB=1024 bytes

1 byte=8bits

a/Note for n large and p very small, (1 − p)n ≈ e-np.  
  
P[no error in the entire file] = (1 – p)n ≈ e-np , for n >> 1, p << 1

= e-8 = 3.35 10-4

We conclude that it is extremely unlikely that the file will arrive error free.

b/A subblock of length n/N is received without error with probability:

*P*[no error in subblock] = (1 – p)n/N

A block has no errors if all subblocks have no errors, so

P[no error in block] = P[no errors in subblock]N =((1 – p)n/N)N = (1 - p)n

So simply dividing the blocks does not help.

c/We assume the following:

* t0 = basic time to send a frame and receive the ACK/NAK ttimeout
* ttotal = total transmission time until success
* nf = number of bits/frame
* na = number of bits per ACK
* nt = number of transmissions
* Pf = probability of frame transmission error

T0 = tf + tACK = nf/R + na/R (tprop ≈ 0).  
  
P[nt = i ] = P[one success after i – 1 failure] = (1 – Pf) Pfi-1

Ttotal | i transmissions = i.t0

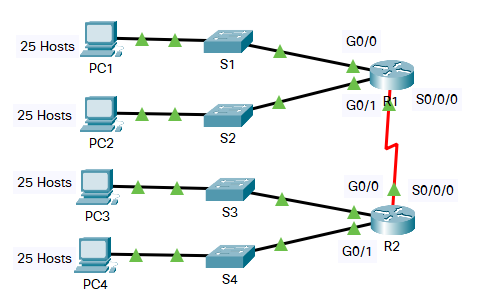
E[ttotal] = = t0 (1 − Pf) = t0(1–Pf )/(1–Pf)2 =t0/(1-Pf)

Here, nf = n >> na thus t0 ≈ tf = n/R ; and Pf = 1 – P[ no error] = 1 - e-np

E[total] = n/R (1 – Pf) = n/[Re-np] = 8 / (3.35 10-4) = 23,847 seconds = 6.62 hours!

The file gets through, but only after many retransmissions.

Câu 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?

Based on the topology, how many subnets are needed?

4

⦁ How many bits must be borrowed to support the number of subnets in the topology table?

2

⦁ How many subnets does this create?

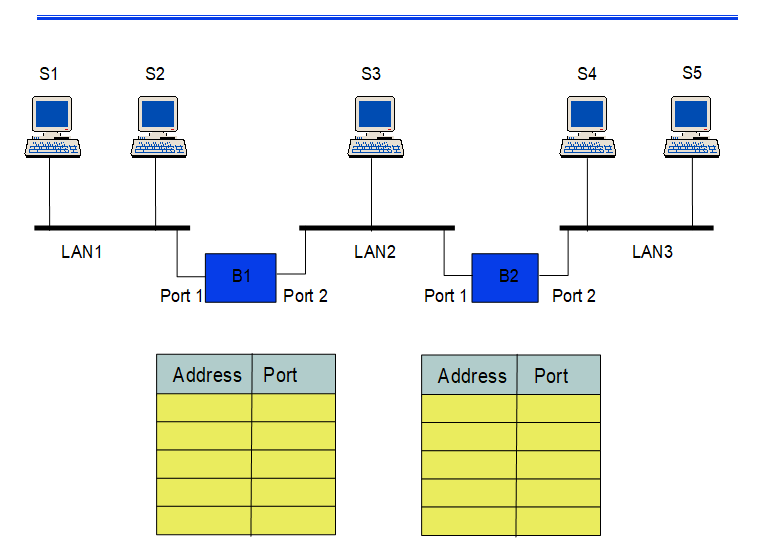
8

⦁ How many usable hosts does this create per subnet?

4

Câu 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.



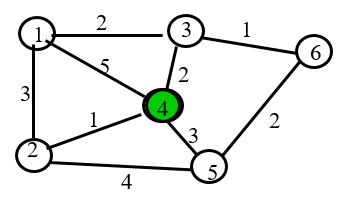
First, it must be understood that here we have 3 LANs, and each LAN is arranged in the form of a BUS, so when a machine transmits data, it will share in the broadcast mode (i.e, send to all devices, machine and network ports)

|  |  |  |  |  |
| --- | --- | --- | --- | --- |
| B1 | | | | |
|  |  | Address | Port | Explanation |
| Step 1 | S1 => S5 | S1 | 1 | Because there is S1 sent. So B1 will receive this packet. |
| Step 2 | S3 => S2 | S3 | 2 | Since S3 sends the broadcast, B1 receives it. |
| Step 3 | S4 => S3 | S4 | 2 | Because after step 2, B2 knows that S3 is on port 2, it will forward the packet to port 2. And of course, both S3 and port 2 of B1 will also receive it. |
| Step 4 | S2 => S1 | S2 | 1 | Since S2 sends the broadcast, B1 receives it. |
| Step 5 | S5 => S4 |  |  | B2 already knows that S4 is on network 3, so B2 will no longer forward the packet to networks 2 and 1. So this step B1 won't say where S5 is |

|  |  |  |  |  |
| --- | --- | --- | --- | --- |
| B2 | | | | |
|  |  | Address | Port | Explanation |
| Step 1 | S1 => S5 | S1 | 1 | Because in the beginning, there is S1 sending S5. But both B1 and B2 do not know where S5 is, so the packet that S1 sends is a broadcast. B1 will also forward the packet because it doesn't know where S5 is => B2 will receive it and know that S1 is on port 1 |
| Step 2 | S3 => S2 | S3 | 1 | Because there is a session sending S3 to S2, but since S3 doesn't know where S2 is initially, it sends it as a broadcast. And so, B2 can also receive the packet of S3, so B2 can also understand that S3 is on its port 1. |
| Step 3 | S4 => S3 | S4 | 2 | Because S4 sent to S3 |
| Step 4 | S2 => S1 |  |  | **Note:** After step 1, B1 already knows that S1 is on network 1, so it will not forward the packet anymore. So at this step, B2 doesn't know where S2 is. |
| Step 5 | S5 => S4 | S5 | 2 | Because S5 sends a broadcast |

Câu 13:

1. Consider the network in Figure.



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

|  |  |  |  |  |  |  |
| --- | --- | --- | --- | --- | --- | --- |
| Iteration | N | D1 | D2 | D3 | D5 | D6 |
| N | 0 | ~ | ~ | ~ | ~ | ~ |
| N,D2 | - | 5,N | 1,N | 2,N | 3,N | ~ |
| N,D2,D3 | - | 4,D2 | - | 2,N | 3,N | ~ |
| N,D2,D3,D5 | - | 4,D2 | - | - | 3,N | 3,D3 |
| N,D2,D3,D5,D6 | - | 4,D2 | - | - | - | 3,D3 |
| N,D2,D3,D5,D6,D7 | - | 4,D2 | - | - | - | - |

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

|  |  |  |
| --- | --- | --- |
| Destination | Cost | Next Hop |
| D1 | 4 | D2 |
| D2 | 1 | N |
| D3 | 2 | N |
| D5 | 3 | N |
| D6 | 3 | D3 |

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:

if(dist(u)+cost(u,v)<dist(v)

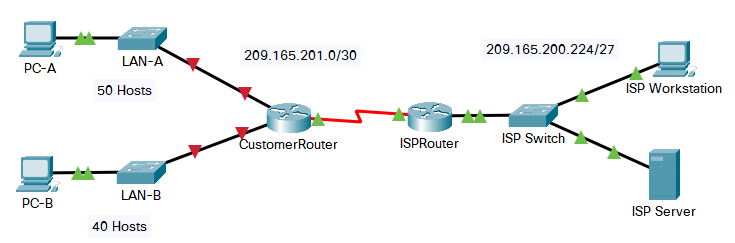
then update the dist(v) =dist(u)+cost(u,v)

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

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.

Questions:

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** |

answer

⦁ How many host addresses are needed in the largest required subnet?

Soln : 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 .

⦁ What is the minimum number of subnets required?

Soln : 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 .

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

Soln : 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.

⦁ 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.

Questions:

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

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

2.) 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.

Soln : 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

Câu 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:

a. How does the protocol need to be modified to accommodate this change?

b. What is the effect of the change on protocol performance?

GIVENTHAT :

a) How does the protocol need to be modified ?

2 things are needed to be changed:-

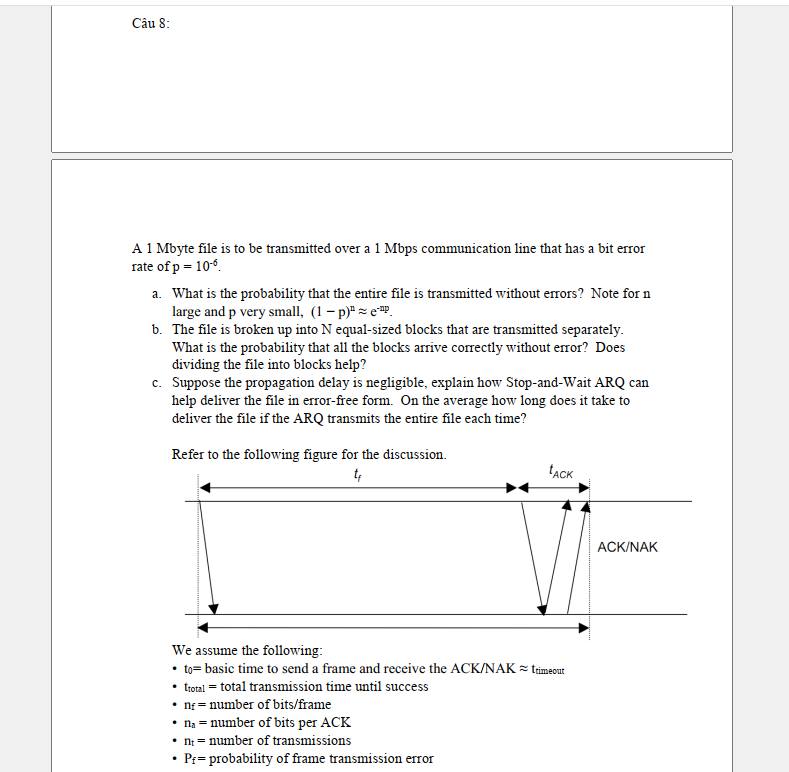
The frame header needs to be modified to recieve the list of frames and Since the reciever explicitly indicates which frames are needed to be transmitted.

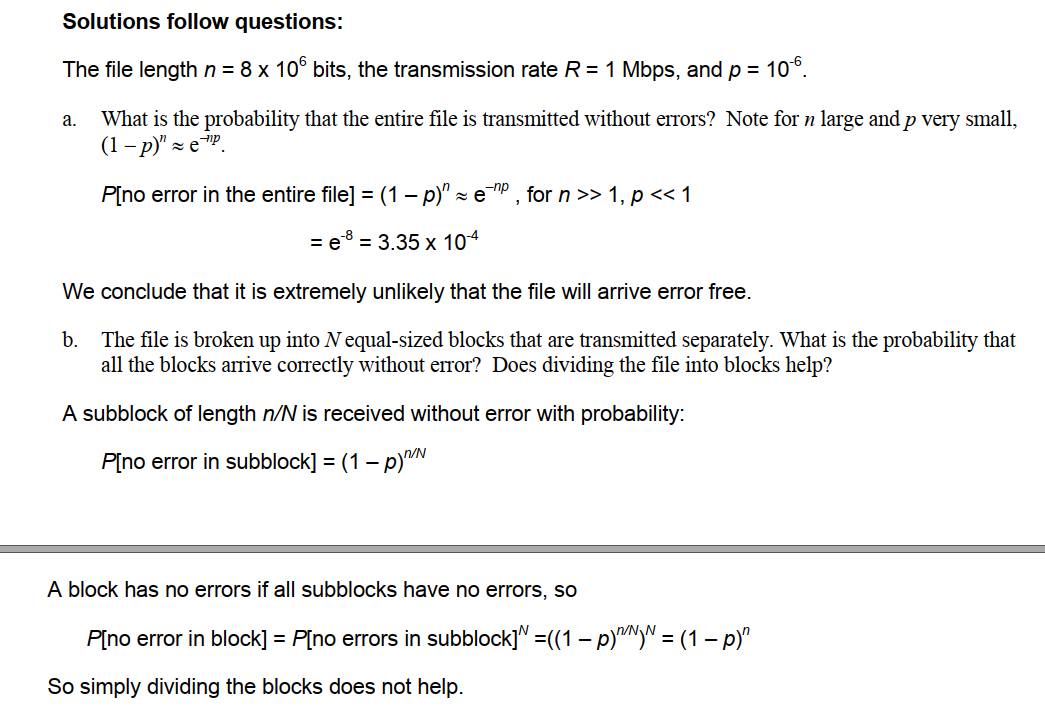
Change in transmitter operation is needed. If the recieved list contains m oldest frames that are yet to be recieved , then it can be used to skip retransmission of frames that have already been received.

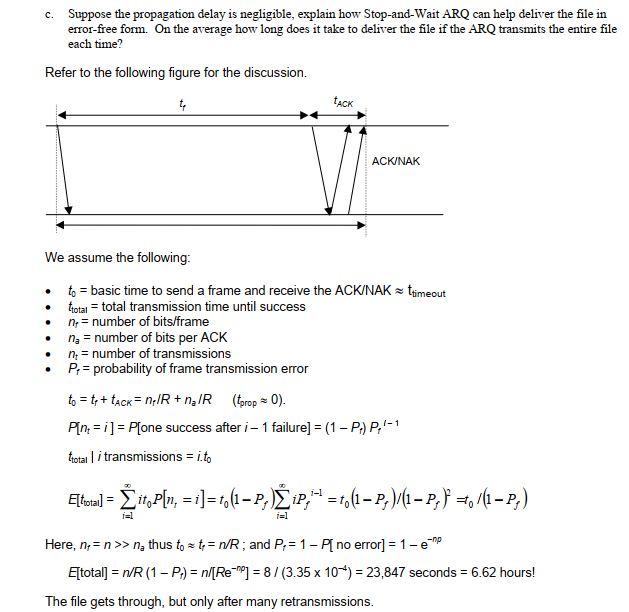
b) what is the effect of change on protocol performance?

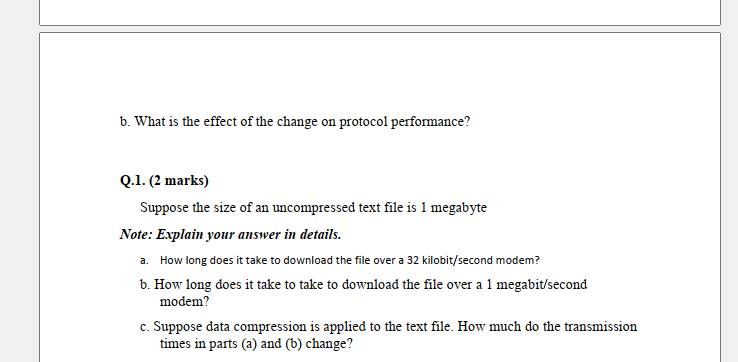
Performance will surely increase if the error rate is high or delay is high. A single frame can ask for the retransmission of several frames.

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









Sentence a)

Convert to bits and bits/second

Size text file = 1 x 1024 x 1024 x 8 (bits)

Speed = 32 x 1000 (bits/second)

=> T (32k) = (1 x 1024 x 1024 x 8 bits / (32 x 1000) = 262,144 (seconds)

Sentence b)

Convert to bits and bits/second

Size text file = 1 x 1024 x 1024 x 8 (bits)

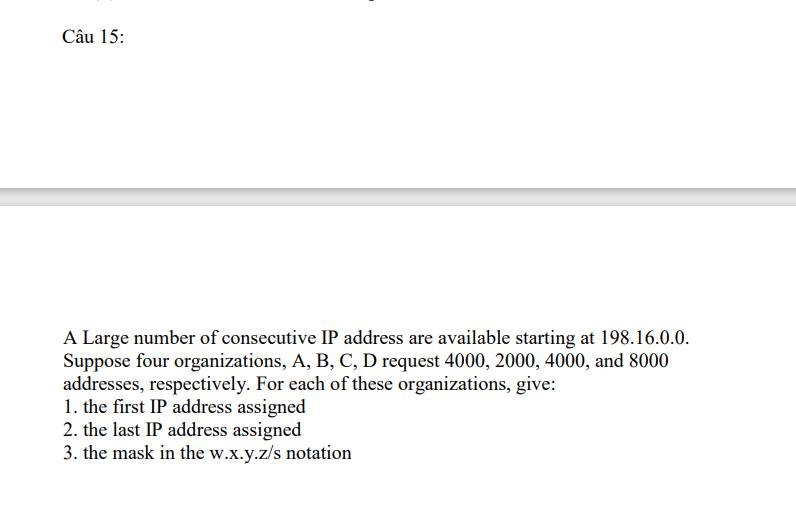
Speed = 1 x 1000 x 1000 (bits/second)

=> T (1M) = (1 x 1024 x 1024 x 8 bits / (1 x 1000 x 1000) = 8.38 (seconds)

Sentence c) The question calls for 1:6, then just multiply by 6 for the speed, if the school issues 1:10, then multiply by 10)

=> T (32k) = (1 x 1024 x 1024 x 8 / (32 x 1000 x 6) = 43.69 (seconds)

=> T (1M) = (1 x 1024 x 1024 x 8 / (1 x 1000 x 1000 x 6) = 1.4 (seconds)



IP addresses will be allocated in blocks of power of 2. So the four organizations will be allocated IPs as A-4096, B-2048, C-4096 and D-8192. Remaining unused IPs are wasted. IPs will be allocated to the organizations contiguously

A has 2^12 hosts. So lower order 12 bits will denote host ID and higher order 32-12=20 bits denotes network ID

A's first IP=198.16.0.0 (Host IP part contains all 0s)

A's last IP=11000110.00010000.00001111.11111111 (Host ID part contains all 1s

=198.16.15.255

B's Mask=198.16.15.255

\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_

B has 2^11 hosts. So lower order 11 bits will denote host ID and higher order 32-11=21 bits denotes network ID

B's first IP=198.16.16.0

B's last IP=11000110.00010000.00010111.11111111=198.16.23.255

B's Mask=198.16.16.0/21

\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_

C has 2^12 hosts. So lower order 12 bits will denote host ID and higher order 32-12=20 bits denotes network ID

C's first IP=198.16.24.0

C's last IP=11000110.00010000.00011111.11111111=198.16.31.255

C's Mask=198.16.24.0/20

\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_

D has 2^13 hosts. So lower order 13 bits will denote host ID and higher order 32-13=19 bits denotes network ID

D's first IP=198.16.32.0

D's last IP=11000110.00010000.00111111.11111111=198.16.63.255

D's Mask=198.16.32.0/19