COMP3721 – Introduction to Data Communications
Assignment Three – Fall 2006

General Instructions

- You may work with one partner for this assignment. Your partner may be from your set or another full-time CST set.
- You and your partner may discuss any and all details of each question freely. You may also discuss questions in broad terms with others, particularly in lab, but ultimately your answers should show sufficient individuation from others' answers reflecting your work in answering the questions.
- All work submitted is subject to the standards of conduct as specified in BCIT Policy 5002.

Submissions

- This assignment is due Friday, November 17, 2006 by 1630 hrs at the latest. Late assignments will not be accepted.
- Submit you assignment to your **lab instructor's** assignment box in the SW2 connector.
- Your submissions must include a cover page clearly specifying your name, student number and set. If working with a partner, this information should be provided for each partner.

Marking

The assignment consists of 8 questions totaling 40 marks.

1. Ethernet uses CSMA/CD for media access.

   a. Explain the purpose of the CSMA/CD protocol, or more generally, media access.

      In broadcast and multi-point networks, where a number of nodes share a common channel, there is a requirement for a protocol to control access to the channel.

      CSMA/CD is a media access protocol where the node wishing to transmit a frame first senses the line for carrier. In the absence of carrier it will transmit its frame. During the transmission the mode will monitor the line for collisions by performing a bit-by-bit comparison of the bits on the channel and the bits in its internal buffer. In the event of a collision the node will retry its previous transmission.

   b. Consider the effectiveness of this protocol as more and more nodes are added to the network. What limits exists in terms of the number of nodes attached to a single Ethernet LAN? How can these limits be overcome?

      Given that the nodes on a single link are contending for access to a common channel when they have data to transmit, collisions will occur – these users are said to belong to a single collision domain. As the number of users in that collision domain increases, so too do the number of collisions – too many users in a collision domain and network throughput will collapse due to the massive number of collisions. While the retransmission algorithm used in Ethernet CSMA/CD allows it to scale better than many other forms of CSMA/CD, in practical terms putting more than 50-75 users into a single collision domain can often result in throughput collapse.

      The solution to this is to separate users into separate collision domains. A switch does just this – it is a data-link layer device providing each connected device a unique link and thus a unique collision domain. By separating devices into separate collision domains, scalability is greatly increased.

2. Consider a reliable sliding window protocol where frame numbers are represented by *n* bits. For Go-Back-N ARQ, the sending windows size (SWS) must be strictly less than 2^n. For Selective Repeat, the SWS must be less than or equal to 2^{n-1}. Show a worst-case scenario (in terms of window positioning) for each ARQ strategy and how the above rules above ensure the no overlap of frame numbers.

In a fixed sequence space, sequence numbers will inevitably wrap-around at some point. That is, with n-bit sequence numbers, frames will be numbered

$$0, 1, 2, .., (2^n-1), 0, 1, \ldots$$

where the second frame 0 is distinct from the first despite the fact that the sequence number is the same. For the sake of clarity, from here on we shall refer to the first frame 0 simply as 0, the second frame 0 as 0', the third as 0'', and so on.

It is clearly important that frame 0 and 0' are never confused during data communications – delivery would not be considered reliable if we accepted one in place of the other. One of the key design criterion then for a sliding-window communication protocol is to ensure the sending and receiving windows cannot ever bring into consideration both 0 and 0' simulateously (or any other similar sequence pair.

Consider then a Go-Back-N protocol. The receiving window size is 1 (RWS=1) as the receiver is only ever willing to accept the next frame in order. If n-bit sequence numbers are used, then one of the 2^n possible sequence numbers is used by the receiver and 2^n-1 unique sequence numbers remain for the sender. Thus the relationship SWS ≤ 2^n-1 for Go-Back-N. If the sending window size (SWS) were greater than this, then there would be more than 2^n frames (potentially) outstanding at any one time and by the pigeon hole principle, some frames will map to the same sequence number. In that case, the potential exists for 0 and 0' to be confused. Figure 1 below illustrates a case where the above limit is properly observed. Figure 2 shows a case where the limit is exceeded.

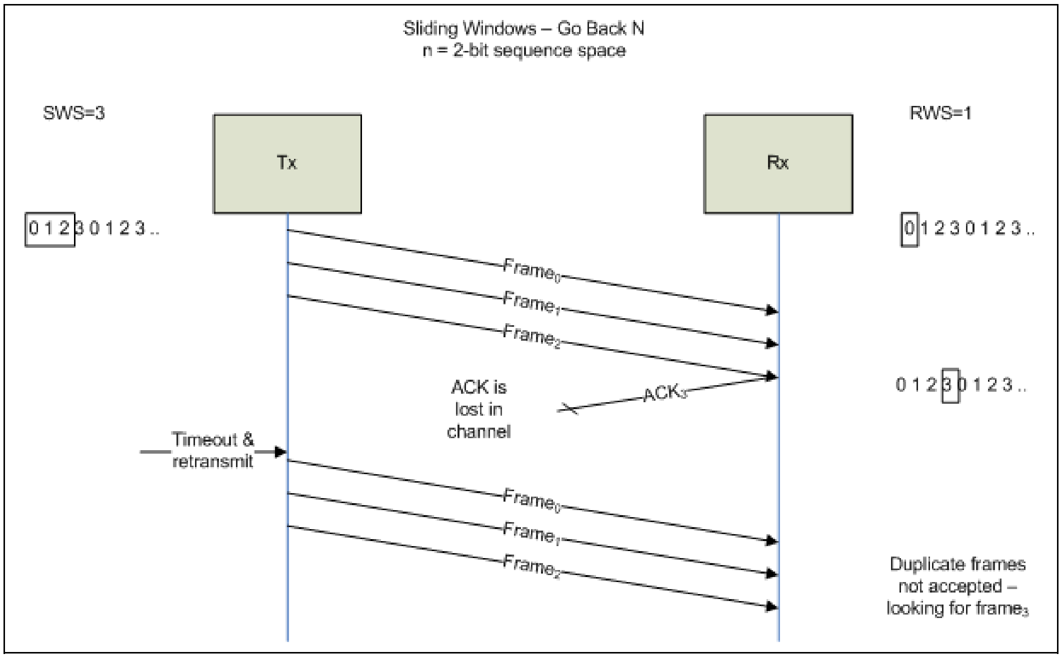


Figure 1 - Go-Back-N ARQ respecting SWS limit

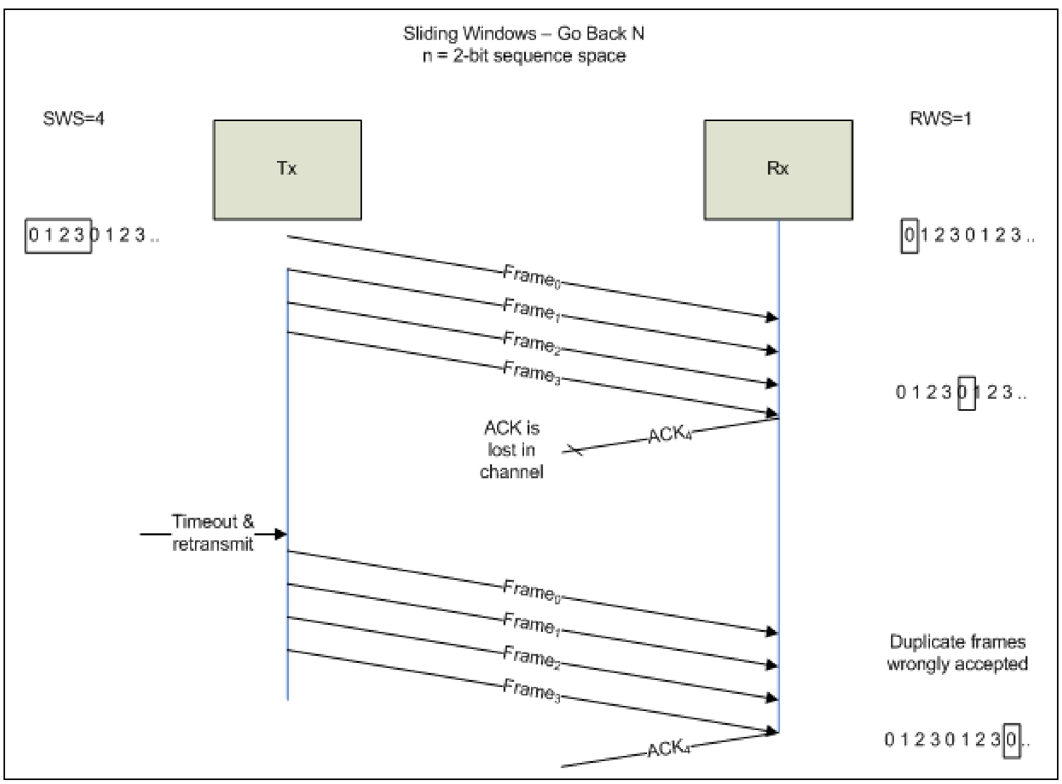


Figure 2 - Go-Back-N exceeding SWS limit

Selective Repeat ARQ requires that the receiver be willing to buffer any of the frames transmitted by the sender. Thus the receiving window size must be equal to the sending window size. Using similar logic to the Go-Back-N argument above, we can see that SWS ≤ 2^{n-1}. Figure 3 demonstrates where this rule is observed versus figure 4 where the limit is exceeded.

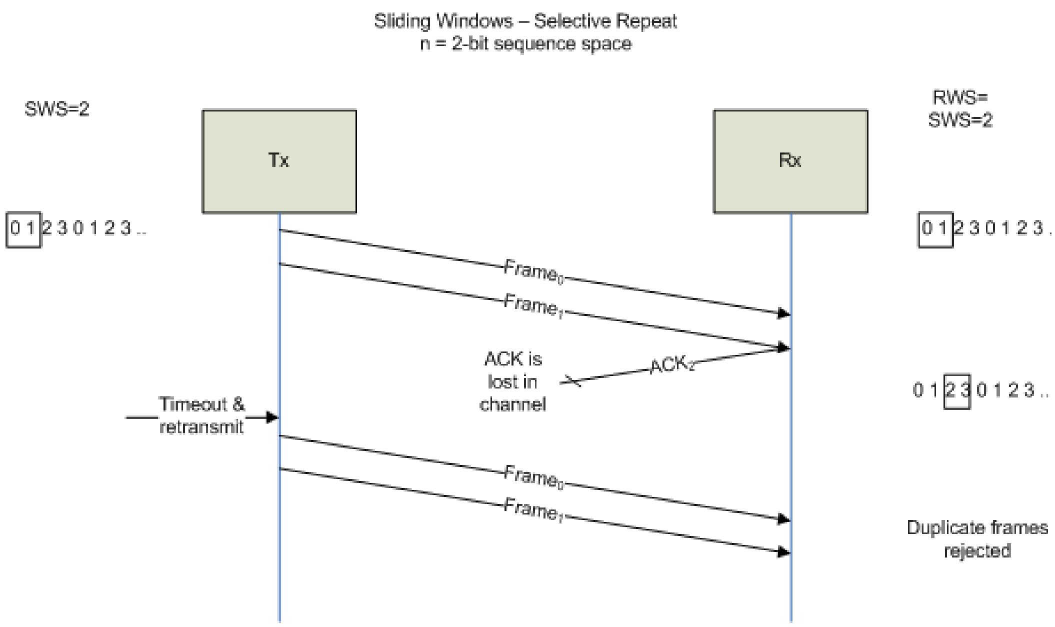


Figure 3 - Selective Repeat ARQ respecting SWS limit

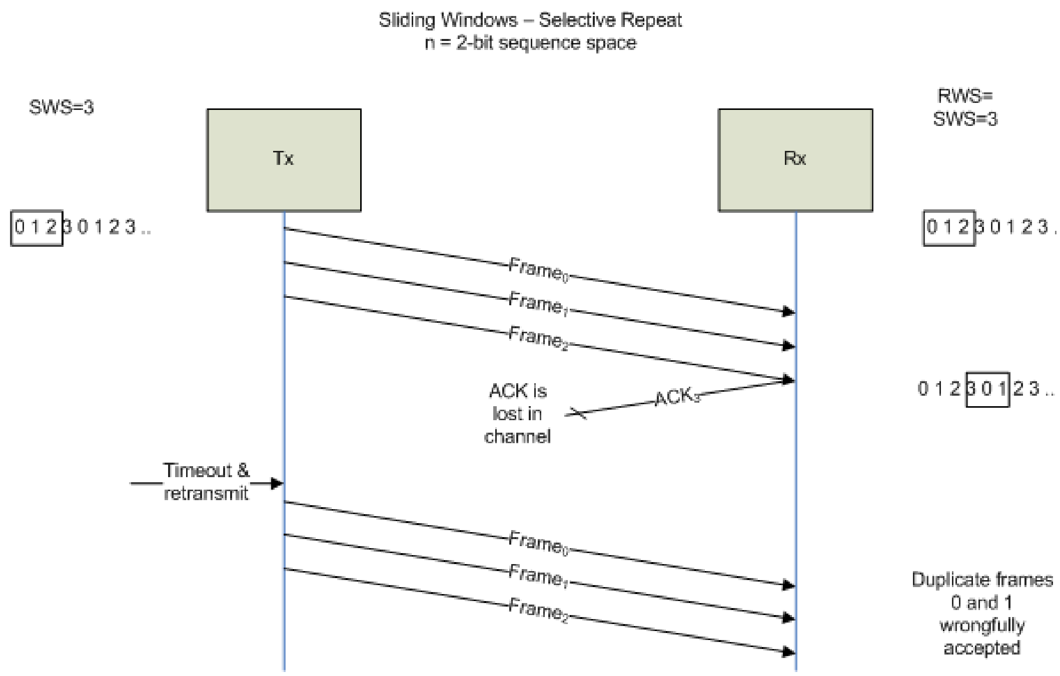


Figure 4 - Selective Repeat ARQ exceeding SWS limit

3. Suppose that 80 percent of the traffic generated in the LAN is for stations in the LAN, and 20 percent is for stations outside the LAN. Is an Ethernet Hub preferable to an Ethernet switch? Does the answer change if the percentages are reversed?

In the above scenario a hub is not preferable to a switch. An n-port switch running at X Mbps provides a total of n * X Mbps full-duplex capacity. In contrast, an n-port hub running at X Mbps provides only X Mbps total across all ports. So in the worst case, if all of the traffic on the network is destined for a single physical link, that link is still at an advantage as it is not sharing its X Mbps at all.

4. A 5000-km long, full-duplex T1-trunk is used to transmit 128-byte frames using the go-back-n sliding window protocol. If the propagation speed is 6 microseconds/km, how many bits should the sequence numbers be? Repeat the calculation for selective-repeat? Assume ACKs are piggybacked, frames are always ready to be transmitted over the reverse channel and are also 128-bytes in size.

a. Sliding Windows using Go-Back-N?

$\mathbf{T_{frame} = 128\ bytes * 8\ bits/byte / 1.544\ Mbps = 663\ \mu s}$

$\mathbf{T_{prop} = 5000\ km * 6\ \mu s/km = 30ms}$

$\mathbf{T_{ack} = T_{frame} = 663\ \mu s}$

$\mathbf{T_{rtt} = T_{frame} + T_{ack} + 2T_{prop} = 61.326\ ms}$

$\mathbf{SWS_{min} = ceiling(T_{rtt} / T_{frame}) = 93\ frames}$

$\mathbf{SWS = 93 \leq 2^n - 1}$

n ≥ 7-bit sequence space

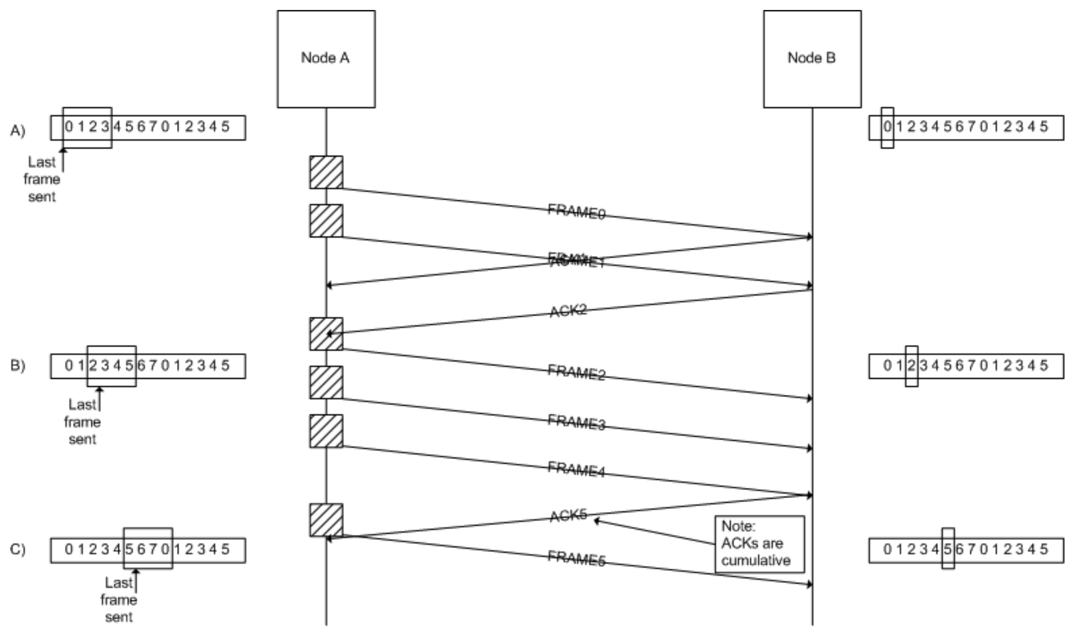
b. Sliding Windows using Selective Repeat?

$\mathbf{SWS = 93 \leq 2^{n-1}}$

n ≥ 8-bit sequence space

5. Two neighboring nodes (A and B) use a sliding window protocol with a 3-bit sequence number. As the ARQ mechanism, Go-Back-N is used with a window size of 4. Assume A is transmitting and B is receiving, show the window positions (for sender and receiver) for the following succession of events:

   a. before A sends any frames
   b. after A sends frames 0, 1, 2 and B acknowledges 0, 1 and the ACKS are received by A
   c. after A sends frames 3, 4, and 5 and B acknowledges 4 and the ACK is received by A



6. What role does the MAC address play in the delivery of a frame? Is it necessary if the link is multi-point? What if the link is point-to-point?

In a multi-point network, nodes require some means of identifying the source and destination for a frame. The MAC address fulfills this role in Ethernet network. Manufacturers of Ethernet cards are provided unique 3-byte MAC prefixes – they then add an additional 3-byte value to create a unique 6-byte address. Each Ethernet frame transmitted over a link contains two MAC addresses – that of the sender and the destination. Switches and bridges maintain tables of these addresses to determine the appropriate physical port to transmit on when delivering a given unicast frame.

A point-to-point link explicitly involves two hosts – thus there is never any confusion at the data-link layer as to the transmitter of a given frame.

7. A primary responsibility of the data link layer is to provide reliable delivery of a frame over a link. Why then is inter-network delivery of a packet (the primary network-layer service) not already reliable? Stated otherwise, why must the transport layer be added to provide reliable end-to-end packet delivery?

The data link layer provides the transmitter assurance that a packet is reliably delivered over a link to the receiver. Inter-networks however involve delivery over multiple links with routers interconnecting those separate networks. Everything would be reliable if each router, upon receipt of a frame, reliably forwarded out towards the ultimate destination. However, there are numerous cases where the router cannot do so and simply drops the frame instead. Then, despite reliable delivery over a link, a frame can still get lost in an inter-network.

As an example of a router needing to drop a packet, consider two links into a router, both with a sustained inbound rate of 100Mbps. If all of the capacity from both of those inbound links is destined for a single 100Mbps outbound link, the router will be forced to buffer some frames. Given that buffer space is always finite, if the above situation persists for too long a period, the router will eventually need to drop some frames. This particular problem is one of

congestion – too much traffic is feeding through a limited resource (in this case the overwhelmed router and the single 100Mbps outbound channel).

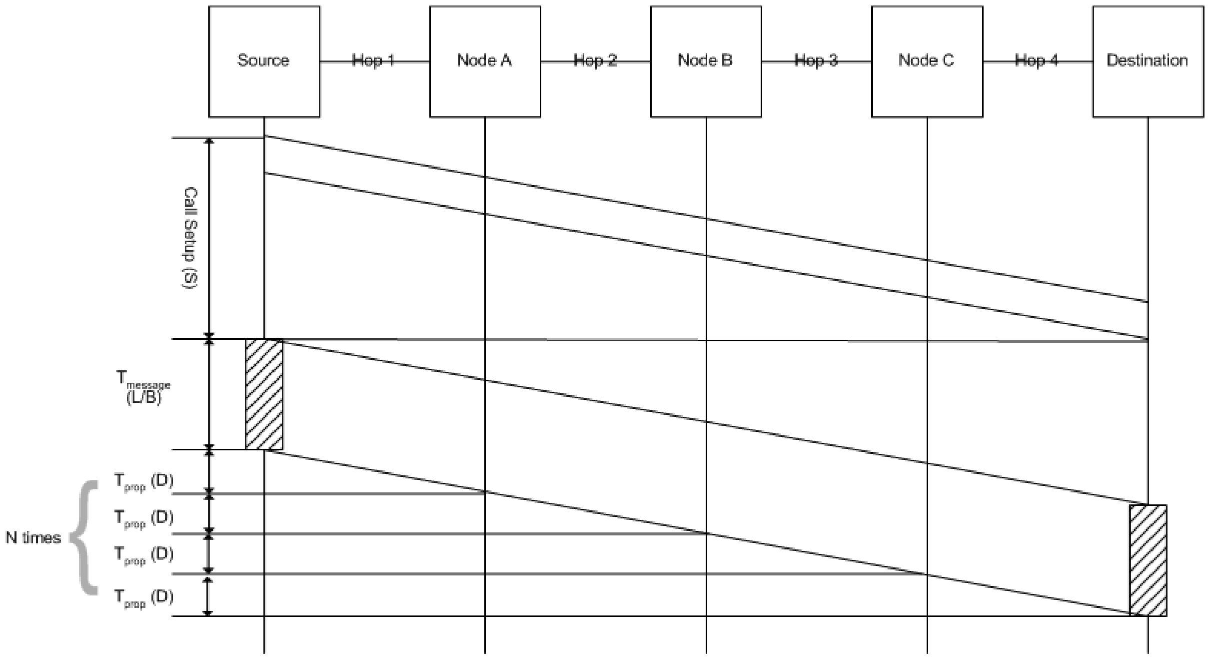
Adding reliability (notification of lost packets would be a form of reliability) to the routing responsibility (network layer) would both complicate network-layer protocols and potential impose overhead where it is not needed. The OSI model therefore dictates that reliable inter-network delivery be handled by an additional layer, the transport layer, built on top of best-effort delivery by the network layer.

8. Two stations wish to communicate over a network. There are 4 hops required to get from station to station. The data rate on all links is 1.544 Mbps. Message length is 8192 bits. If packet switching is used, each packet size is 1024 bits which includes 16 header bits. Call setup time is 0.25 seconds. Propagation delay per hop is 1 msec. Calculate the total end-to-end delay for:

   a. Circuit switching
   b. Message switching
   c. Packet switching

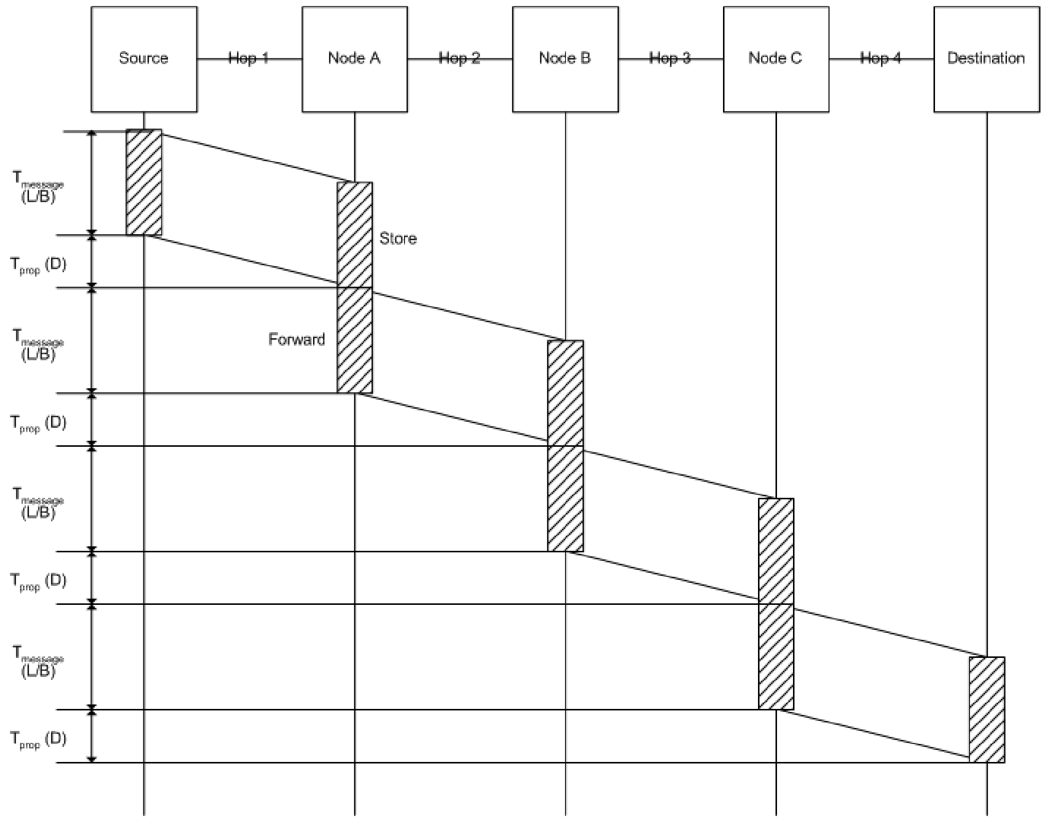
   Assume no buffering delay at the nodes. Use the following variable names for deriving your delay equations:

   N : # of hops between two given stations.
   L : Message length in bits.
   B : Data rate, in bps, on all links.
   P : Packet size in bits.
   H : Overhead (header) in bits.
   S : Call setup time (CS or VC) in sec's.
   D : Propagation delay/hop in sec's.



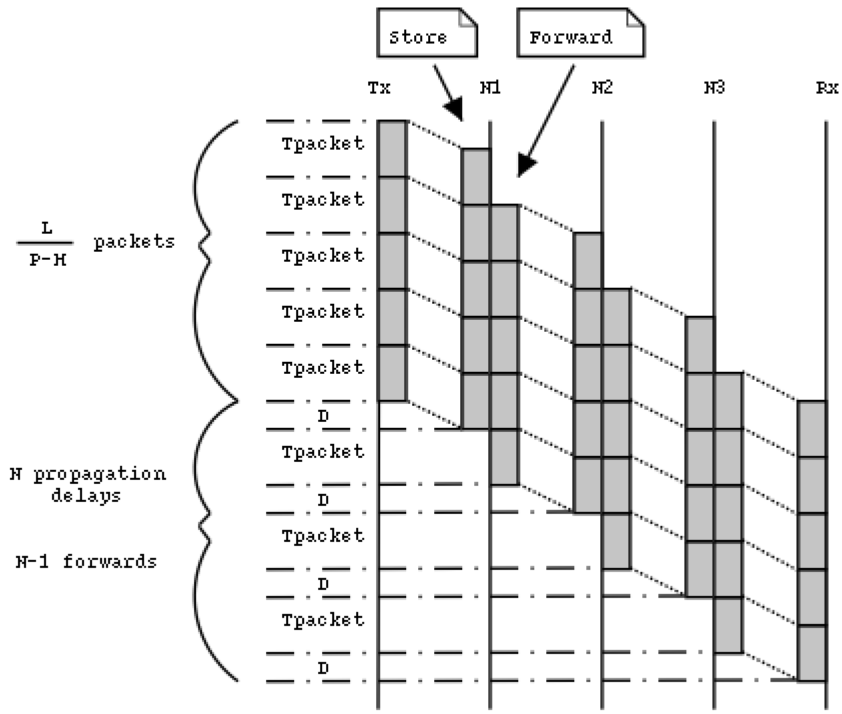
a. **Circuit Switching**

$$T_{total} = S + N \times D + \frac{L}{B} = 250ms + 4hops \times 1ms / hop + \frac{8192bits}{1.544Mbps} = 259.3ms$$


Source
Hop 1
Node A
Hop 2
Node B
Hop 3
Node C
Hop 4
Destination
T message (L/B)
T prop (D)
Store
Forward
T message (L/B)
T prop (D)
T message (L/B)
T prop (D)
T message (L/B)
T prop (D)

b. **Message Switching**

$$T_{total} = N\left(D + \frac{L}{B}\right) = 4(1 + \frac{8192bits}{1.544Mbps}) = 25.2ms$$



c. **Packet Switching**

$$T_{total} = (\left\lceil \frac{L}{P-H} \right\rceil \times \frac{P}{B}) + (N \times D) + ((N-1) \times \frac{P}{B})$$

$$= (\left\lceil \frac{8192bits}{(1024-16)bits/\ frame} \right\rceil \times \frac{1024bits}{1.544Mbps}) + (4hops \times 1ms/hop) + ((4-1) \times \frac{1024bits}{1.544Mbps})$$

$$= 5.97ms + 4ms + 1.99ms = 11.96ms$$

Note: the above assumes fixed-sized packets – fractional packets are padded to full-size (thus the ceiling operator for the first term of the equation). So for this particular case, we would require 9 packets (due to the header bits).