CS 421: Computer Networks **SPRING 2005**

MIDTERM I
March 31, 2005
120 minutes

Name: ______________________________

Student No: ___________________________

Show all your work very clearly. Partial credits will only be given if you carefully state your answer with a reasonable justification.

1)

a) (6 pts) An Internet traffic analysis study found that the traffic (measured in bytes) going to destination port 25 (SMTP) exceeds by an order of magnitude the traffic coming from port 25, i.e., with source port 25, whereas the traffic (measured in bytes) coming from port 80 (HTTP) exceeds by an order of magnitude the traffic going to port 80. Explain these findings. **(Use at most 3 sentences.)**

b) (6 pts) How does a web server distinguish between incoming segments belonging to different connections but having the same destination port number 80? Explain. **(Use at most 3 sentences.)**

c) (6 pts) Consider an HTTP client that wants to retrieve a Web document at a given URL. The IP address of the HTTP server is initially unknown. The Web document at the URL has 5 embedded JPEG image objects that reside at the same server as the original document. List below the **transport** and **application-layer** protocols that are used by the client in this scenario.

Application Layer	Transport Layer

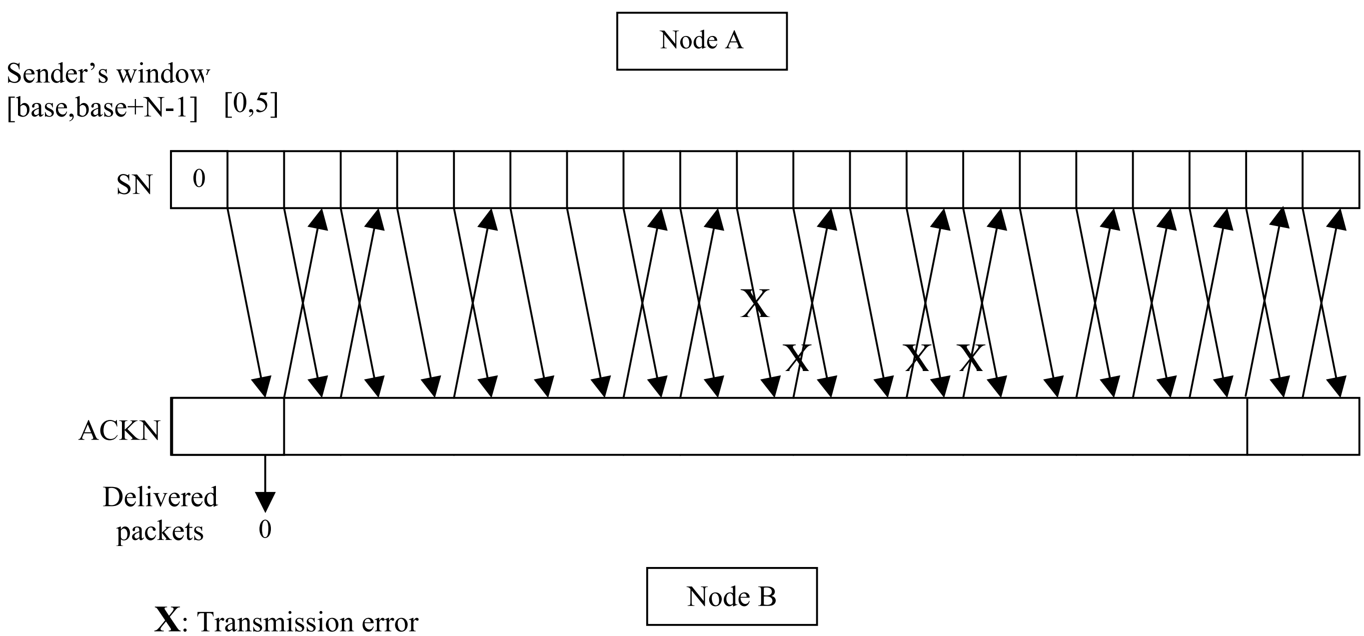
d) (6 pts) Give two reasons why different packets belonging to the same session, i.e., all going from the same source to the same destination may experience different delays. **(Use at most 2 sentences.)**

e) (6 pts) One of the features used by the KazaA P2P file sharing system is the parallel downloading. What is the HTTP method used in the HTTP request message which is sent in order to find out the size of the file to be downloaded? **(Just simply give the name of the specific method, you do not need to describe the procedure.)**

2)

a) (10 pts) Consider a 4 Mbits/sec channel with a 5 msec one-way propagation delay, i.e., 10 msec roundtrip propagation delay. We want to transfer a file of size 9000 Bytes. Each packet, including the ACK packets, has a total size of 1000 Bytes including the 40 Bytes of header. When there is data to be transmitted, each packet contains the maximum number of bytes. Assume that there is a processing delay of 1 msec after a packet is fully received at the receiver until the transmission of the corresponding ACK is started. Selective Repeat protocol is used with a window size of N = 4 packets. Assume that the 5th packet crossing the forward channel is lost while all other packets (including ACKs) are not lost or corrupted. Assume that the sender sets the Timeout for each packet to a value 2 msec larger than the minimum necessary period so that the sender will not timeout prematurely even if the packet is not lost. How much time is required to complete the transfer of the whole file and receive the final ACK at the sender?

b) (15 pts) Consider the use of Go-Back-6 protocol for communication from Node A to Node B. Assume that when the sender reaches at the end of the window, i.e., all packets in the window are sent but not ACK'd, the sender goes to the beginning of the window and retransmits all packets in the window, as if the timeout occurred. In the following figure, indicate the sequence number (SN) for packets sent from A to B, the ACK number (ACKN) for packets sent from B to A, the times and SN of the packets at B delivered to the application layer, and the window kept at A. **Note that packets received during the transmission of another packet will be immediately processed, but the corresponding action, e.g., update of SN/ACKN, will take effect with the start of transmission of the next packet.**



3)

a) (9 pts) Assume that the congestion window of a TCP flow was 14 segments long when a *timeout* occurred. Assume that there are no segments or acknowledgments of this flow that were in transit when the timeout occurred. The round trip delay for the flow is fixed and is equal to 50 msec and the transmission time for a segment is 5 msec. The receive window is fixed at 100 segments for the entire duration of the connection. What is the minimum time the flow will spend in the Slow Start phase after the timeout before reaching the Congestion Avoidance phase?

b) (8 pts) Consider a TCP connection with the following parameters:

- RTT = 30 msec
- Transmit Buffer size = 10 KB
- Receive Buffer size = 5 KB
- MSS = 1 KB
- Transmission time for a TCP segment with MSS = 5 msec
- Number of segments to be transmitted = 1000
- Initial Threshold = 100 segments, i.e., 100 KB

Assume that there are no lost or errored segments and the application process immediately reads the data from the receive buffer. What is the maximum window size, in Bytes, (considering both **congestion** and **flow** control mechanisms) achieved during this connection? **Carefully justify your answer.**

c) (6 pts) What is the difference between flow control and congestion control? **(Use at most 3 sentences.)**

4)

a) (5 pts) There are two loss events in TCP Reno: three duplicate acknowledgements (fast retransmission) and timeout. Which one of these two loss events is an indication of a heavier congestion? Justify your answer in **at most 3 sentences**.

b) (5 pts) Explain why during TCP connection termination the client stays in the TIME_WAIT state for a fixed amount of time before moving to the CLOSED state. How is this fixed time determined? **(Use at most 3 sentences.)**

c) (6 pts) Assume that a TCP sender measures the round trip time for the first segment after the connection is established as 20 ms, and thus sets its estimated round trip time to 20 ms. The next round trip time that the sender measures is 45 ms. In response, the sender increases its estimated round trip to 25 ms. The next round trip time that the sender measures is 40 ms. What is the next estimated round trip computed by the sender? Justify your answer.

d) (6 pts) Why does the Congestion Avoidance phase use the additive increase instead of the multiplicative increase such as the one used in the Slow Start phase? **(Use at most 3 sentences.)**

5) (Bonus 10 pts, **no partial credits, only completely correct answers will be graded.**)

Consider the RTT estimation algorithm for setting the retransmission TimeOut used by TCP as we discussed in the class:

$$\text{EstimatedRTT}_{new} \leftarrow (1 - \alpha)\ \text{EstimatedRTT}_{old} + \alpha\ \text{SampleRTT}$$

$$\text{DevRTT}_{new} \leftarrow (1 - \beta)\ \text{DevRTT}_{old} + \beta\ |\text{EstimatedRTT}_{old}\text{-SampleRTT}|$$

$$\text{TimeOut} \leftarrow \text{EstimatedRTT}_{new} + \gamma\ \text{DevRTT}_{new}$$

One of the requirements of this algorithm, although not explicitly stated, is that the TimeOut should be at least equal to the most recent measured RTT, i.e.,

$$\text{TimeOut} \geq \text{SampleRTT}$$

Obtain a relation between α, β and γ such that this requirement is always satisfied. The following values are used in TCP: $\alpha = 1/8$, $\beta = 1/4$ and $\gamma = 4$. Determine whether these values satisfy the expression you obtained. (*Hint*: Consider the case when we measure an RTT which is larger than the current value of the RTT estimate, i.e., $\text{SampleRTT} \geq \text{EstimatedRTT}_{old}$.)