CS 421: Computer Networks **FALL 2004**

MIDTERM I
November 22, 2004
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) Assume you wish to transfer a B-byte file along a path composed of the source, five intermediate routers and the destination interconnected by six links. Suppose each link has a one-way propagation delay of 2 msec and a transmission rate of 4 Mbits/sec. The routers support both circuit and datagram packet switching. Thus you can either break up the file into 1 KB packets, or set up a circuit through the switches and send the file as one contiguous bit stream. With packet switching, packets have 24 bytes of packet header and 1000 bytes of data payload, and assume that the packet processing at each intermediate switch incurs a 1 msec delay after the packet has been **completely** received by the router. You can ignore all queuing delays in packet switching. Assume also that sender can transmit the packets continuously without waiting for acknowledgements. With circuit switching, circuit setup requires a 1 KB connection request message to travel from the source to the destination while incurring a 1 msec processing delay at each intermediate router after the connection request message has been **completely** received by the router. The connection acceptance message, which is also of 1 KB, is sent by the destination to the source after a 1msec processing delay upon the reception of the connection request message, but the acceptance message incurs no processing delay at the intermediate nodes. You may also assume that the file size, B, is a multiple of 1000 bytes for simplicity.

i. (7 pts) Determine the range of file sizes, B, such that the total number of bytes sent across the network is less with circuit switching than with packet switching.

ii. (8 pts) Determine the range of file sizes, B, such that the total delay from the beginning of the transaction until the entire file arrives at the destination is less with circuit switching than with packet switching.

b) (6 pts) A study on the traffic at root DNS name servers shows that 95% of the queries they receive are for non-existent top level domains, e.g., typos such as microsoft.co**n** instead of microsoft.com, even though a study of a representative cross-section of local DNS servers shows that only 0.1% of the queries they receive are for non-existent top level domains. How can you explain this discrepancy? **(Use at most 3 sentences.)**

c) (6 pts) What is the RCODE section in the DNS header used for? List possible interpretations for RCODE defined by the DNS specifications given in RFC 1035. **(Use at most 3 sentences.)**

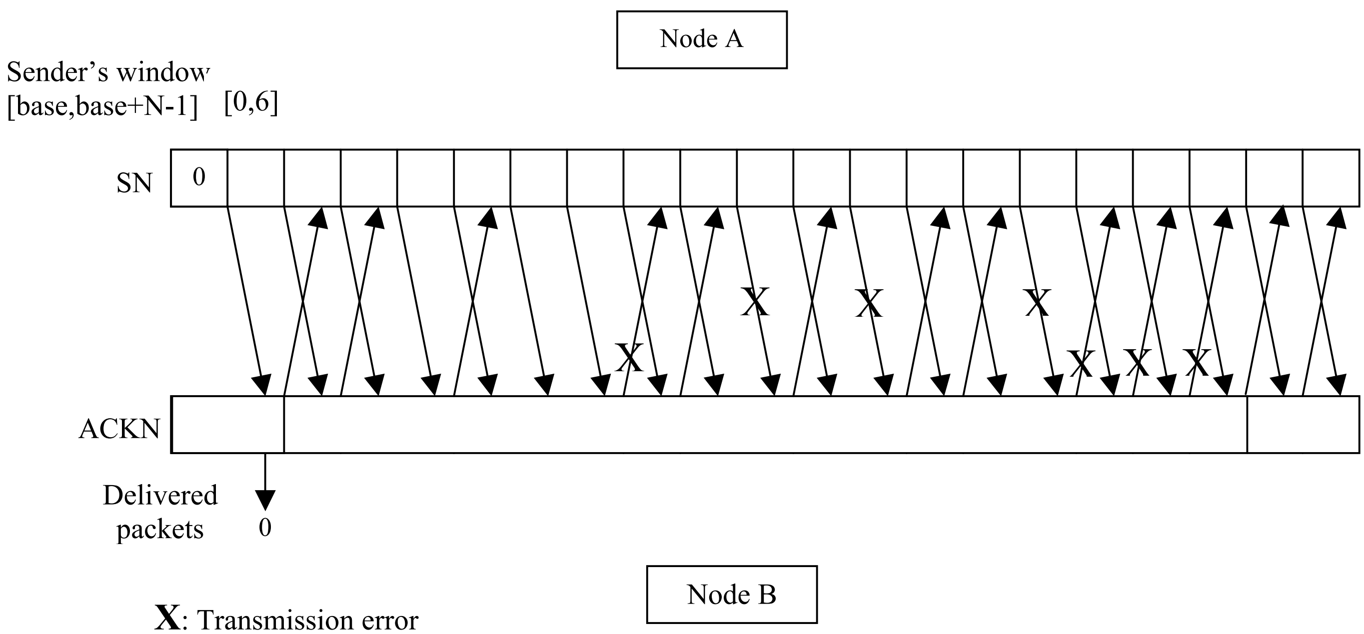
d) (6 pts) What is the purpose of using pointers in DNS reply messages? **(Use at most 3 sentences.)**

2)

a) (8 pts) Assume that a base html file containing 30 embedded images is requested by a client. Assume that the base html file and each of the images separately fit in a TCP segment. How many round trips are required to retrieve the base file and the images under the following settings? Assume that the round trip times dominate all other times.

i. HTTP 1.0 with no parallel connections
ii. HTTP 1.0 with up to 10 parallel connections
iii. HTTP 1.1 with no pipelining
iv. HTTP 1.1 with pipelining

b) (15 pts) Consider the use of Go-Back-7 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) (8 pts) Suppose that the sender and receiver are communicating using the Selective Repeat data transfer protocol with a window of size N packets. Suppose the sequence number of the segment at the base of the window at the receiver (rcv_base) is x. What are the possible ranges of sequence numbers in the sender's window? Justify your answer.

b) (6 pts) The conventional TCP/IP stack implements TCP protocol which provides reliable data delivery. Despite this, there are UDP based application layer protocols which provide reliability for real-time services such as streaming multimedia, Internet telephony, etc. Why is TCP not preferred for real-time streaming applications? **(Use at most 3 sentences.)**

c) (10 pts) Suppose a TCP connection has a window size of eight segments and a round trip time (RTT) of 800 ms. The sender sends segments at a regular rate of one segment every 100 ms, and the receiver immediately sends ACKs back to the sender upon arrival of a segment without any processing delay. The transmission times of the ACK packets are negligibly small. Suppose that a segment is lost, and the loss is detected by the sender using the fast retransmit mechanism on the receipt of the third duplicate ACK. At the point when the ACK of the retransmitted segment arrives, how much total time has the sender lost (compared to lossless transmission) if the sender waits for the ACK of the retransmitted lost packet once the whole window is exhausted?

4)

a) (6 pts) Assume that the congestion window is of size X segments just prior to a loss event occurring at time t. Assume that X is quite large. How does TCP Reno change the congestion window in response to each type of loss events? Give the size of the congestion window at times t, t + RTT, t + 2RTT, for each possible loss event, where RTT denotes the round trip time.

b) (6 pts) TCP has a relatively complicated algorithm for estimating the RTT and calculating the retransmission TimeOut. Give **two reasons** why it is very important for TCP to calculate the TimeOut correctly. **(Use at most 3 sentences.)**

c) (8 pts) Suppose that a TCP connection has a RTT (including all propagation, processing, queuing and store-and-forward delays) of 50 ms. The transmission time for each packet is 10 msec, and ACK packets have negligible transmission times. The TCP sender starts with the Slow Start phase, and the Threshold is initially set to 6 segments. How long does it take until the sender reaches the Congestion Avoidance phase?

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:

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

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

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

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*: The worst case scenario occurs when $\text{SampleRTT} \geq \text{EstimatedRTT}_{\text{old}}$.)