CSC 458/2209: Computer Networks, Fall 2016

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Handout #17: Problem Set 2 **Date:** Monday, November 7, 2016

1. **Network Address Port Translation (4 points).** Assume we have a home network that is connected by a wireless router that includes NAT capabilities as well as a DHCP server. On the wireless network we have a Desktop, a Laptop and a Video Game Console that are switched on and used in this order. The external IP address of the router is 1.2.3.4, the internal IP address is 10.0.0.1. The DHCP server on the router is programmed to give out IP addresses sequentially on the network 10.0.0.0/16 (*netmask* FF:FF:00:00).

   (a) Draw the topology of the network with the router after all devices are switched on. The topology should include IP addresses and netmasks where known.

   (b) After all three hosts on the wireless network are switched on they connect to a web server running on sky.cs.toronto.edu. What IP addresses will the web server record for the web requests coming from the Laptop, Desktop and Video Game Console?

   (c) Both the Laptop and Desktop have an SSH server running on port 22. Is it possible to connect from sky.cs.toronto.edu to the SSH servers? If yes, explain the steps involved in setting up the connection. If no explain why it is not possible.

   (d) Is there an upper limit for the maximum number of parallel UDP connections from hosts on the local network of the router to servers on the Internet? If yes, what is this maximum?

2. TCP Congestion Control (6 points). We'll use simplified rules for TCP congestion control:

(i) **Initial window size.** The window size is measured in packets, not bytes. We'll assume that all packets are the same length. Initially, the window size is set to 1 packet.

(ii) **Slow-start.** The connection is in slow-start from the time the connection starts until the first time a packet is dropped. Each time an acknowledgement is received, the window size is increased by one packet.

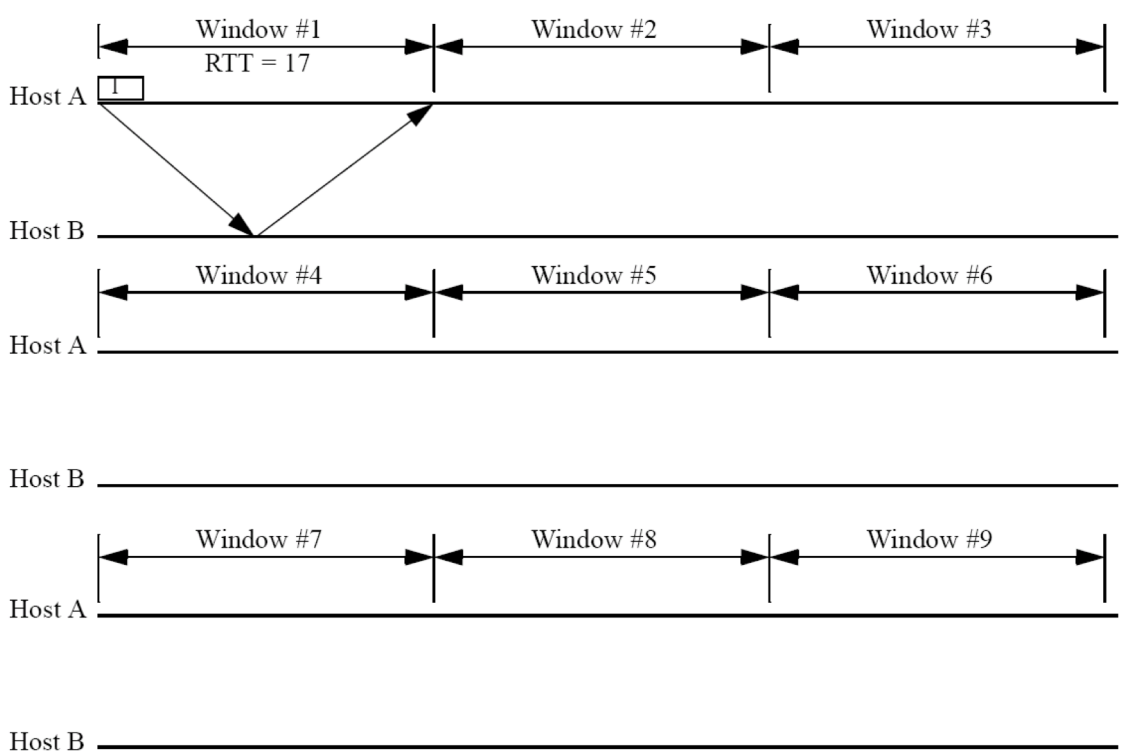
(iii) **Multiplicative decrease.** Each time a packet (or acknowledgement) is dropped, the window size is halved.

(iv) **Additive increase.** When not in slow-start, the window is increased by one packet when acknowledgements have been received for all the packets in the previous window.

(v) **Fast Retransmission and Fast Recovery.** This mechanism is not used in our simplified version of TCP.

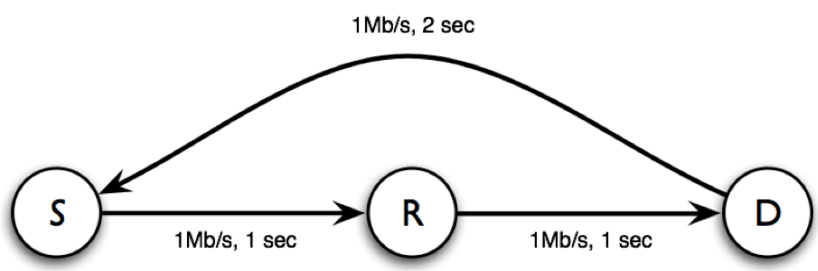
Host ***A*** has 100 packets to send to Host ***B*** using TCP. The two hosts have established a TCP connection, and Host ***A*** has sent its first data packet, as shown in the figure. We will assume that the RTT always equals the time taken by Host ***A*** to transmit 17 packets, and that the retransmission timeout is set to exactly RTT. In other words, there is no variability in the RTT and Host ***A*** retransmits unacknowledged packets after exactly RTT. We will further assume that the receiver has an infinite buffer size and so never invokes the flow-control mechanism. "Packet 4" is lost the first time it is transmitted; and when "Packet 18" is transmitted for the first time, its acknowledgement is lost. No other packets or acknowledgements are lost in the first 10 windows.

(a) Draw the sequence of packets that are sent by Host ***A*** up to and including the 9th window. Clearly mark each packet with its sequence number (starting from 1), showing which packets are transmitted (or retransmitted) in each window. Show how many packets have been sent by the end of Window #9. You can use the following figure for this part. (3 points)



(b) Plot the congestion window as a function of time. For simplicity, you can use 1 RTT as a unit of time. (3 points)

3. Buffer Sizing (4 points). Consider two end-hosts ***S*** and ***D*** connected via router ***R*** as in the following figure. The link connecting ***S*** to ***R*** and the link connecting ***R*** to ***D*** both have a delay of 1 second. The link connecting node ***D*** to node ***S*** has a delay of 2 seconds. All links operate at 1Mb/s.



Consider a single TCP flow starting at node ***S***, and sending packets over router ***R*** to destination node ***D***. The acknowledgement packets come back to ***S*** over the link directly connecting node ***D*** to node ***S***. We assume router ***R*** has no buffering and we have no processing delay at any of the nodes in this system. To simplify the analysis, we also assume that the TCP flow is always in congestion avoidance, i.e., it never enters slow-start.

(a) What is the maximum value for congestion window size (in packets) that would be achieved by this TCP flow? You can assume all packets are 1000 bytes. Remember that ***R*** has no buffering.

(b) Plot the variations in congestion window size as a function of time. Start at CWND = 0.

(c) From you graph in Part (b) what is the average congestion window size over time?

(d) What is the average throughput (bits per second) of this flow between ***S*** and ***D***?

4. Power (3 points). In class and the textbook, we used the somewhat arbitrary measure of "power" to characterize network performance. The usual definition of power, ***P***, is given by:

$$P = \frac{\text{load}}{\text{average delay}}$$

(a) Explain why this definition of power is commonly used, and in what sense it is "arbitrary". (1 point)

(b) A commonly used approximation to the relationship between normalized load (which can vary between 0 and 1 only) and average delay is:

$$d = \frac{1}{1 - \lambda}$$

Here **λ** is the normalized load offered to the network and ***d*** is the average delay of packets passing through the network. Find and sketch the value of power, ***P***, as a function of the offered load for this network. Be

sure to show the minimum and maximum values of power and load, as well as the value of offered load that maximizes the power. (2 points)