Network Systems (201300179), Test 4

April 4, 2014, 15:45–17:15

- This is an open-book exam: you are allowed to use the book by Peterson & Davie and the reader that belongs to this module. Furthermore, use of a dictionary is allowed. Use of a simple (non-graphical) calculator is allowed.
- Other written materials, and laptops, tablets, graphical calculators, mobile phones, etc., are not allowed. ***Please remove any such material and equipment from your desk, now!***
- Although the questions are stated in English, you may answer in English or Dutch, whichever you are more comfortable with.
- You should always explain or motivate your answers, with so much detail that the grader can judge whether you understand the material; so just saying "yes" or giving a formula without explanation is not enough.
- Visiting the toilet without explicit permission of the supervisor is not allowed. During the last 30 minutes of the exam, no toilet visits are allowed.

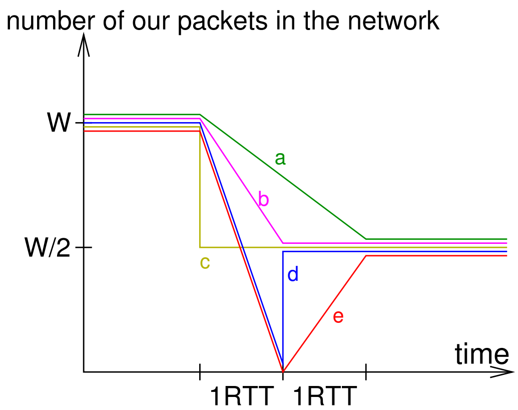
1. TCP congestion control

Two hosts, A and B, are communicating using a TCP connection. A and B are connected by a router, R, and two links, A-R and R-B. The link A-R has a very high bandwidth, and negligible propagation delay. The link R-B is limited in its data rate, and has a non-negligible propagation delay. A is transferring data to B, and is assumed to have always data in its buffer to transmit. There is no other traffic on the links. Transmission delay for ACK packets from B to A is negligible.

3 pt (a) In which two different ways can host A deduce that a packet was (probably) dropped? For each of the two ways of drop detection, explain how A will adapt its sending rate after the detection, in case of standard (new Reno) TCP.

2 pt (b) Suppose that the first data packet sent by A (after the 3-way handshake) is dropped by R. In which of the two ways mentioned by (a) will A detect the loss of this packet? Explain your reasoning.

Please look at the figure below. The figure displays the number of packets in transit from A to B as a function of time. W is the size of the congestion window (in packets of size MSS) just before the a (single) packet loss. The fast retransmit / fast recovery procedures in TCP have the macroscopic effect of halving the congestion control window after packet loss. The effect is that also the number of packets in the network is halved, as is indicated in the figure below. How the number of packets in transit evolves during the RTTs directly following the packet loss detection depends on the exact implementation of fast recovery.



3 pt (c) Suppose we would like to follow the purple line (labelled b): the number of packets in transit gradually decreases during 1 RTT, and after 1 RTT, it stabilizes at half of the congestion window before the loss.
Design an algorithm that TCP could follow to achieve this, by specifying what actions the sender should take upon receiving acknowledgements in the period from the packet loss detection until the number of packets in transit stabilizes at $W/2$.

2. Quality of Service

The company fly-by-night has an Internet connection that is only just capable of handling the entire company's data traffic. Their network administrator wants to ensure that the company's web servers have good upload speed to the Internet. He manages to install a QoS-enabled router that connects the company's internal networks to the Internet link. This router has functions implemented for packet classification, policing, admission control, and several queueing disciplines. The network administrator wants to use a certain queueing discipline to give the mentioned traffic a better treatment.

3 pt (a) How can the network administrator avoid that web traffic is completely occupying the Internet link, while starving the company's regular traffic?

Let us now look at the issue of QoS in a more general way. Suppose we have N sources, which each send packets satisfying a token bucket specification: their token buckets have rate r and bucket size B. The output of the sources must be transmitted over a link with a rate of S bytes per second ($N \cdot r < S$). For simplicity, we assume in the remainder of this exercise that all packets have a size of 1 byte.

2 pt (b) What is the maximum delay for a packet from any of the N sources with FIFO scheduling?

3 pt (c) Assume priority scheduling is used, with 2 sources using the high priority queue, and the remaining $N-2$ sources using the low priority queue. What is the maximum delay for a packet from one of the high priority sources?

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3. Security

3 pt (a) What are certificates used for, for example in the context of HTTPS? And what kind of attack would be possible if certificates were not used? Explain.

2 pt (b) The fly-by-night network administrator is afraid that people will send packets with "spoofed" IP addresses from his network (i.e., packets of which the source IP address does not actually belong to this network). In order to block such outgoing packets, the network administrator wants to use a firewall. Should he choose a *stateless* or a *stateful* firewall? Explain.

3 pt (c) Recall that WEP has a weakness which makes it easy to find the key after about 300 000 packets have been intercepted. Now suppose that switching to WPA2 is not an option. Propose a way to make WEP secure, by changing its key very frequently.

In your solution, keep in mind that it should be automated (the user must not have to do the key changing by hand), that all nodes should always know what the current key is, and that packets may occasionally get lost.

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4. Time synchronization & localization

2 pt (a) (1) What is the difference between accuracy and precision? (2) RBS is using reference broadcast beacons. Give two methods on how its precision be increased.

3 pt (b) In APIT there are fixed beacons and blind nodes. (1) How does APIT operate if there is only 1 blind node? (2) Can a position be estimated outside the convex hull of the beacons?

3 pt (c) Wifi localization on mobile phones using RSSI is not very accurate. Give three mechanisms on how to improve its accuracy.

End of this exam.