### **Exercise 8**

### December 20th, 2012



## **Chapter 4 outline**

- 3.1 Transport-layer services
- 3.2 Multiplexing and demultiplexing
- 3.3 Connectionless transport: UDP
- 3.4 Principles of reliable data transfer

- 3.5 Connection-oriented transport: TCP
  - segment structure
  - o reliable data transfer
  - flow control
  - connection management
- 3.6 Principles of congestion control
- 3.7 TCP congestion control



## Flow vs. congestion control

#### **Question 7:**

 What is the difference between flow control and congestion control?





#### **Flow Control**:

• Prevent overwhelming the receiver by sending too much data.



#### LastByteSent - LastByteAcked <= rwnd</pre>



## Flow vs. congestion control

#### Answer 7:

### **Flow Control:**

 Prevent overwhelming the receiver by sending too much data.

### **Congestion Control:**

 React on congestion in the network (on the path to the receiver).



#### **Question 1:**

 Suppose that in TCP, the sender window is of segment size N = 200, the base of the window is at sequence number 600, and the sender has just sent a complete window size of segments. Let RTT be the senderto-receiver-to-sender round trip time of 200 ms and Maximum Segment Size MSS = 1 000 bytes.

a). Assuming no loss, what is the **throughput** (in terms of MSS and RTT and in terms of Megabit/s) of this message exchange?

b). Suppose TCP is in its congestion avoidance phase. Assuming no loss, what is the **window size** (in terms of segment) after the N = 200 segments are acknowledged?



Three questions raised by TCP congestion control approach

- 1. How does a TCP sender **limit its send rate** (at which it sends traffic into its connection)?
- 2. How does a TCP sender **perceive** that there is **congestion** on the path between itself and the destination?
- 3. What algorithm should the sender use to change its send rate as a function of perceived end-to-end congestion?



1. How does a TCP sender **limit its send rate** (at which it sends traffic into its connection)?



#### LastByteSent - LastByteAcked <= min{cwnd, rwnd}</pre>



2. How does a TCP sender **perceive** that there is **congestion** on the path between itself and the destination?

• Timeout

 $_{\odot}\,$  Three duplicated ACK's



3. What algorithm should the sender use to change its send rate as a function of perceived end-to-end congestion?



## **TCP** sender congestion control



N-E-T»

WORKS

## Why Slow start is quick?





#### Answer 1:

N=200, RTT=200ms, MSS=1000 bytes, sender just sent a complete window!

a). Assuming no loss, what is the throughput (in terms of MSS and RTT and in terms of Megabit/s) of this message exchange?

$$throughput = \frac{segments \cdot MSS}{RTT} = \frac{200 \cdot 8000Bit}{0.2s} = 8000000 \frac{Bit}{s} = 8\frac{MBit}{s}$$



# TCP congestion control cont'd

#### Answer 1:

b). Suppose TCP is in its congestion avoidance phase. Assuming no loss, what is the window size (in terms of segment) after the N = 200 segments are acknowledged?

#### • Congestion Avoidance, in one RTT:

Here we define cwnd := Congestion Window.

$$cwnd = cwnd + MSS \cdot \left(\frac{MSS}{cwnd}\right)$$

 Each ack increases the cwnd by MSS/cwnd, which is 8000Bit/200=40Bit. As 200 acks arrive, the window is increased by 8000Bit which is exactly 1MSS, therefore cwnd=200+1! Note cwnd = 12/20/2011 MSS!



## **TCP-Reno and Tahoe**

#### **Question 2:**

 What is the difference between the two congestion control algorithms TCP-Tahoe and TCP-Reno?



### **TCP-Reno and Tahoe**

#### Answer 2:

- o Difference in handling timeouts and triple duplicate acks!
  - Tahoe always down to 1MSS,
  - **Reno** distinguishes:
    - 3 duplicate ACKs -> go down to 50% then CA,
    - timeout

-> go down to 1MSS





## **Selective Repeat**

#### **Question 3**:

 Please explain the selective repeat dilemma and name a solution to prevent its occurrence.







## **Selective Repeat**

#### Answer 3:

 Dilemma occurs on a limited sequence range and large window size. Solution: Window size should be maximally half of the sequence range!



### TCP vs. UDP

#### **Question 4:**

Please name at least three differences between UDP and TCP.



## TCP vs. UDP

#### Answer 4:

- 1. TCP is connection oriented, UDP is not
- 2. TCP is a reliable data transfer protocol, UDP is not reliable
- 3. TCP enables in-order delivery, UDP does not guarantee in-order deliver
- 4. UDP has less overhead (lightweight) compared to TCP (heavy load due to ordering, window maintenance etc...)
- 5. TCP uses flow control, UDP does not
- 6. TCP uses congestion control, UDP does not



# **Choosing a protocol**

#### **Question 5:**

 If you would like to transfer a file, which transport protocol would you use? Which protocol would you use for voice traffic?



# **Choosing a protocol**

#### Answer 5:

- File: TCP as it is reliable, in-order delivery.
  Receiver can directly pipe data contents into file
- Voice: UDP as it is lightweight, small inorders cannot be heard and reliability has no advantage if delivery takes to long



## **TCP** fast retransmit

**Question 6:** 

• Please explain TCP fast retransmit.



## **TCP** fast retransmit

#### **Answer 6**:

- Time-out period often 0 relatively long:
- Detect lost segments via three duplicate ACKs.
- Fast retransmit: resend 0 segment before timer expires, directly after receiving three duplicate acks





## **Estimated vs. sampled RTT**

#### **Question 8:**

 Why is an EstimatedRTT used to calculate the TCP timeout instead of the recently sampled RTT?



# **Estimated vs. sampled RTT**

RTT: gaia.cs.umass.edu to fantasia.eurecom.fr

### Answer 8:

- Exponential weighted moving average
- influence of past sample decreases exponentially fast



SampleRTT fluctuates
 too much. EstimatedRTT + safety margin is a safer guess to set the timer.



### • That's all and thanks for your attention!

