Datakommunikation I - HT05

Theoretical Exercise 2

Deadline Friday September 16, 2005, 12:00. Preferably, hand it in via mailbox 73 outside room P1457.

Realisation You can work in groups of two and answer in either English or Swedish.

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Exercise 1: Error Control

Efficiency is the ratio between the number of messages delivered at the receiver side to the application and the number of messages sent by the sender. Let's further define A as the ration between propagation time and transmission time.

For each of the following statements, please state if it is true or false and explain in a few sentences why.

a) Assume the sender does not apply flow control, that it uses timers to detect the loss of messages, and that it applies Go-Back-N as retransmission strategy. Assume that the round-trip time is constant, message losses are random, the message loss rate is 10^{-6} and $A = 100$. Statement: If the sender increases the value for the timeouts, the efficiency stays the same. (3P)

b) Assume $A > 100$. Statement: The efficiency of the Go-Back-N retransmission strategy will increase for a constant loss rate when the number of messages that are lost consecutively increases. (3P)

c) Assume that the Selective Repeat strategy is used with cumulative ACKs: At a timeout for message i, (1) message i is retransmitted, and (2) all timers j for $i \leq j \leq k$, where k is the last transmitted message, are reset. $A > 100$. Statement: The efficiency of the Selective Repeat strategy will decrease for a constant loss rate when the number of messages that are lost consecutively increases. (3P)

d) (Optional) Calculate the throughput for the stop and wait (selective ACK), Go-Back-N (cumulative ACK), and Selective Repeat (selective ACK) strategies under the assumption that the loss rate p is low but that losses always occur with k (e.g., k=2) consecutive messages missing. Compare it with the throughput calculated during the lesson for uncorrelated losses.

Exercise 2: Throughput of a Broadband Satellite Internet Access

In this exercise we compare three flow control protocols and estimate the maximum throughput of a satellite access link. Satellite links are characterised by long delays and high error probability. We assume a geostationary satellite (altitude $h = 36000$ km) with a bit rate of 512 Kbit/s, i.e., $512 \cdot 10^3$ bit/s. The bit error rate of the link is $\epsilon = 10^{-6}$ due to atmospheric disorder. The bit errors are uncorrelated.

To estimate the maximum throughput we assume no packet loss on the satellite link, i.e., there is no queue overflow in either of the involved equipment. Moreover, we make the following technical assumptions:

- A checksum is employed to detect corrupted packets that include bit errors. We idealistically assume that the checksum itself is never affected by corruption and is such that it can always correctly detect any corruptions(s).
- The receiver treats packets that arrive in corrupted state the same way as lost packets. In particular this means that the receiver does not issue negative acknowledgements (NAKs) for corrupted packets.
- Each packet is acknowledged separately. The sender assumes packet loss when the acknowledgement that corresponds to a packet does not arrive before a time budget expires. To measure this time budget, the sender employs a timer. Hence a timeout on this timer indicates a packet loss. We assume that the timer is started after sending the packet to make the time budget independent of the packet's size. For this case study, we set the timeout to $t^* = 600$ms.
- Any processing delay on end systems as well as the time to receive packets can be neglected.
- Packet size is 512 Bytes.
- The velocity of signal propagation is c$= 3 \cdot 10^8$m/s.

You should now persue the following approach to estimate the maximum throughput (net bit rate) of the satellite access link:

a) **Stop and Wait**: Calculate the delay induced by the satellite downlink in case of a successful packet transmission as well as the delay induced by a unsuccessful transmission attempt. Derive the probability of packet errors from bit errors. Quantify the delay induced by fixed number of packet errors in order to determine the expectation of the packet transmission time. Use this expectation to derive the net bit rate. (3P)

b) **Go Back N; cumulative ACK**: Draw a diagram for the interactions of this protocol. Reevaluate the delay induced by a packet error. Redetermine the expectation of the packet transmission time. (4P)

c) **Selective Repeat; selective ACK**: see (b) (4P)

d) Compare the results of (a) to (c). Which impact on the results has the choice of the timeout? Which of the data link protocols would you recommend to use on this satellite link? Do you need to account for criteria other than the throughput when giving you recommendation? (3P)

Exercise 3: TCP

a) For this exercise you can think of UDP to be a (simple) protocol that does not provide reliable communication. However, if a packet is delivered to the application, the packet is correct. From an applications point of view - when is it suitable to choose UDP rather than TCP as transport protocol? (1P)

b) TCP distinguishes between *loss events*: Timeout and duplicate acknowledgements. They are (among other things) used to estimate the congestion in the network. How severe is the congestion when either of these loss events occur? Assume that the round-trip-time is large compared with the transmission time for one segment. (2P)

c) The TCP sender in this problem is configured to never use more than 64 Kbyte of buffer space for the sending window. It is connected to a network where the bottleneck link capacity is 80 Mbit/s, the average signal propagation speed in the network between the sender and the receiver is $2 \cdot 10^8$ m/s and the one-way delay added by intermediate routers, end-node processing etc. is as large as the propagation delay. How far apart can the end-nodes be without having the maximum buffer size limit the throughput of the connection? (5p)

d) TCP continuously collects RTT samples from incoming ACK:s. What are these samples used for? (you don't have to describe how). (2P)

e) Consider TCP is used over a link layer technology that has optional retransmission. The error rate on the channel is fluctuating. Explain in a qualitative way the consequences for TCP having retransmission on the link layer switched on and switched off. Hint: Think about the consequences of the different loss events (see question b). (4P)

Exercise 4 (Optional): Connection Management

The task of connection management is to ensure that a message only has a meaning for at most one connection (at-most-once semantics).

a) In handshake-based connection management schemes the handshake (i.e.the exchange of set-up messages during connection establishment) ensures that no old or duplicate set-up message will be accepted.

Timer-based connection management schemes do not need to handshake. Explain how timer- based connection management schemes can ensure that no old or duplicate set-up messages will be accepted.

b) TCP uses a 3-way handshake for connection management, and TCP does not use a separate connection identifier to distinguish connections. How does TCP uniquely identify connections? What are the limitations of the approach used?