

## Faculty of Engineering, Mathematics and Science School of Computer Science & Statistics

Integrated Engineering

Semester 2 2019

Year 3

Computer Networks

30th April 2019

**RDS Simmonscourt** 

09:30 - 11:00

Prof Ciarán Mc Goldrick

## Instructions to Candidates:

You may not start this examination until you are instructed to do so by the Invigilator.

Attempt all questions.

Use diagrams as and where appropriate.

Be brief and concise in your answers.

## Materials permitted for this examination:

Non-programmable calculators are permitted for this examination — please indicate the make and model of your calculator on each answer book used.

Two (2) double sided A4 cheat-sheets allowed (i.e. no more than 2 A4-sized sheets in total). You MUST submit your cheat sheets in your answer book(s) at the end of the examination.

- 1. Determine if each of the following statements is *TRUE* or *FALSE* **AND** provide a *one* (1) sentence rationale for each answer.
  - (a) Closing a TCP connection takes at least 2 RTTs and includes a timeout mechanism.
  - (b) With the Distance Vector algorithm, each node talks only to its directly connected neighbours.
  - (c) IP fragmentation is required if the underlying Ethernet payload (data field) size is larger than the IP datagram size.
  - (d) If a UDP checksum adds up to all 1's at the receiver, we can be certain that the packet was not corrupted during transmission.
  - (e) Suppose two nodes start to transmit (at the same time) a packet of length L over a broadcast channel of rate R. Denote the propagation delay between the two nodes as  $d_{prop}$ . There is a collision if  $d_{prop} < L/R$ .
  - (f) Consider congestion control in TCP. When the timer expires at the sender, the value of ssthresh is set to one half of its previous value.
  - (g) The major services provided by every transport layer protocol include multiplexing/demultiplexing, error checking, and flow control.
  - (h) Frame collisions are completely eliminated in TDM based media access protocols.

 $[8 \times 2 : 16 \text{ marks}]$ 

- 2. Answer the following questions. Be brief and concise.
  - (a) Why isn't the DNS database simply located on a large, centralized server?

    Give at least 3 reasons.

    [4 marks]
  - (b) Consider the GBN protocol with a sender window size of 4 and a sequence number range of 1,024. Suppose that, at time t, the next in-order packet that the receiver is expecting has a sequence number of k. Assume that the medium does not reorder messages. Answer the following questions:
    - i. What are the possible sets of sequence numbers inside the sender's window at time t? Justify your answer.
    - ii. What are all possible values of the ACK field in all possible messages currently propagating back to the sender at time t? Justify your answer.

[6 marks]

(c) UDP and TCP use 1s (ones) complement for their checksums. Suppose you have the following three 8-bit bytes: 01010011, 01100110, 01110100. What is the 1s complement of the sum of these 8-bit bytes? (Note that although UDP and TCP use 16-bit words in computing the checksum, for this problem you are being asked to consider 8-bit sums.) Show all work. Why is it that UDP takes the 1s complement of the sum; that is, why not just use the sum?

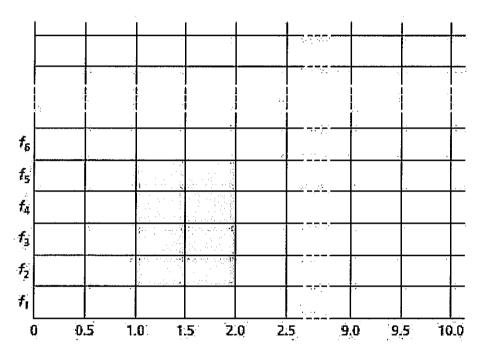
With the 1s complement scheme, how does the receiver detect errors?

Is it possible that a 1-bit error will go undetected? How about a 2-bit error?

[6 marks]

[4+6+6: 16 marks]

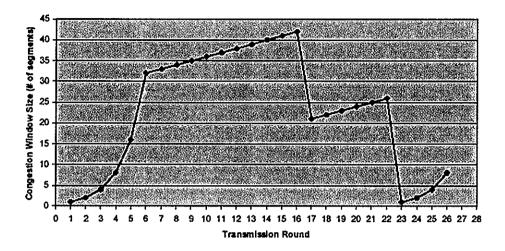
3. Consider the following idealized LTE scenario. The downstream channel (see figure below) is slotted in time, across F frequencies. There are four nodes, A, B, C, and D, reachable from the base station at rates of 10 Mbps, 5 Mbps, 2.5 Mbps, and 1 Mbps, respectively, on the downstream channel. These rates assume that the base station utilizes all time slots available on all F frequencies to send to just one station. The base station has an infinite amount of data to send to each of the nodes, and can send to any one of these four nodes using any of the F frequencies during any time slot in the downstream sub-frame.



- (a) What is the maximum rate at which the base station can send to the nodes, assuming it can send to any node it chooses during each time slot? Is your solution fair? Explain and define what you mean by "fair."
- (b) If there is a fairness requirement that each node must receive an equal amount of data during each one second interval, what is the average transmission rate by the base station (to all nodes) during the downstream sub-frame? Explain how you arrived at your answer.
- (c) Suppose that the fairness criterion is that any node can receive at most twice as much data as any other node during the sub-frame. What is the average transmission rate by the base station (to all nodes) during the subframe? Explain how you arrived at your answer.

[5 x 3: 15 marks]

4. Consider the following plot of TCP window size (measured by number of segments) as a function of time (measured in transmission rounds). Assuming we are using TCP Reno.



- (a) Identify the intervals of time when TCP slow start is operating.
- (b) Identify the intervals of time when congestion avoidance is operating.
- (c) After the 16th transmission round, is segment loss detected by a triple duplicate ACK or by a timeout?
- (d) After the 22nd transmission round, is segment loss detected by a triple duplicate ACK or by a timeout?
- (e) What is the initial value of the threshold variable in the first transmission round?
- (f) What is the value of the threshold variable at the 18th transmission round?
- (g) What is the value of the threshold variable at the 24th transmission round?
- (h) During what transmission round is the 70th segment sent?
- (i) Assuming a packet loss is detected after the 26th round by the receipt of a triple duplicate ACK, what will be the values of the congestion window size and of the threshold variable?

[9 x 2 : 18 marks]