# MIEEC / MIEIC

Communication Services / System and Network Services

Winter Semester 2018/2019

Ana Aguiar

05.02.2019

Duration: 90 min

## Instructions

Please write your name and student number on all answer sheets.

This exam is open book.

Partial credit is possible, so give each question a try.

Show all your work and reasoning. This is the only way to be able to give partial credit to your answers.

If you get stuck in a question, leave it for later and go on to solve the others.

The use of communication devices (e.g., computer, smartphone, mobile phone, etc) during the exam is strictly forbidden.

### Voluntary Code of Ethics

Please sign below if you agree to comply with the following sentence.

I give my word of honour that I shall not use any unauthorised means to answer this exam.

**Good work!**

**Short Questions**

1. (2) Explain what zero rating is and comment, justifying, on whether in your opinion it is compliant with the European network neutrality guidelines.

2. (1) Identify 2 video QoE metrics, of which one depends on average end to end round trip time, and another does not.

3. (1) Explain why a high hit rate and low round trip time (closeness to a user) can be conflicting goals in a caching system.

4. (2) Consider that you want to download a webpage with 5 elements, 3 of them from a different server from where the page is hosted. Calculate broadly the savings in RTT with respect to HTTP/1.1 with persistent connections when HTTP/2.0 with request multiplexing but no server push is used. I suggest that you use sequence diagrams to support your answer, consider TCP connection establishment time, and consider an archetipal and equal RTT for both servers.

5. (1) Is flooding an adequate technique for content search in chunk-based peer-2-peer networks? Why/ why not?

6. (2) Consider a TCP flavor that alternates between periods when it measures the available bandwidth and latency, and periods during which it sends data to fill the pipe as measured in the most recent measurement period. How long would such a flow take to react to congestion/ packet drops? In your opinion is such a behaviour fair in the TCP fairness sense?

7. (2) Explain why a reduction in available bandwidth can lead to a excessive reduction in adaptive video streaming data rate. Here you should explain the unwanted behavior that can appear as a side effect of cross-layer interactions between application and TCP, and not the expected and necessary reduction in video bitrate.

8. (1) Suppose a router with capacity to forward 100 packets per second accepted flows with TSpecs shown in the following table, described in terms of token bucket filters with token rate r packets per second and bucket depth B packets.

| r [packets per second] | B [packets] |
|---|---|
| 10 | 10 |
| 50 | 1 |
| 20 | 5 |

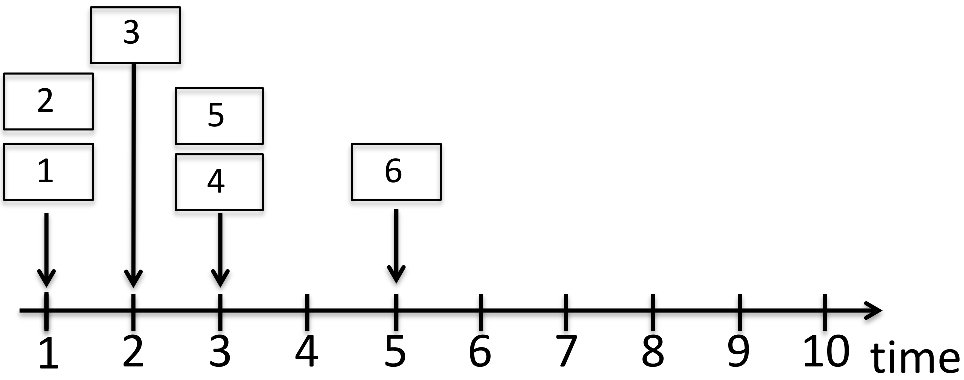
Should the router of the previous exercise, having only the initially available flows, and assuming it can use priority scheduling, accept a flow that requests a guaranteed delay of 0.01 seconds?

9. (1) Consider a chord ring with space $[0,2^7\text{-}1]$, and with the following set of nodes {4, 26, 44, 58, 77, 92, 105, 116}. Build the finger table for node 58.
10. (1) Considering an average inter-node delay of 50ms for the overlay in the previous case, in how much is the search time for node 8 reduced?

**Problems**

**Please show all your calculations and justify your options.**

1. (2) Please consider the following packet arrival sequence. Assume that one packet can be transmitted per time slot, and that packets arrive immediately before the indicated time.




Assume weighted fair queueing (WFQ) service. Assume that odd-numbered packets are from class 1, and even-numbered packets are from class 2. Class 1 has a WFQ weight of 1, while class 2 has a WFQ weight of 2. Indicate the order by which the packets leave the queue, showing your calculations to justify the answer.

2. Consider the following packet delays observed by the receiver of a video stream {76, 109, 134, 103, 57, 13, 40, 59, 114, 58} milliseconds.
    a. (1) If the video starts being played as soon as the first packet is received, how many of the 10 packets will arrive too late to be played? Say which packets will arrive too late.
    b. (1) Calculate the jitter for this sequence of packets.
    c. (1) What is the size of the playout buffer (in milliseconds) that will allow playing all 8 packets?
    d. (1) What is the size of the playout buffer (in milliseconds) if a loss rate of 10% is acceptable?