TDDD66 – Mobile Networks
Final Examination: 8:00-12:00, Thursday, Oct. 31, 2013
Time: 240 minutes
Total Marks: 40
Grade Requirements: three (20/40); four (28/40); and five (36/40).
Assistance: None (closed book, closed notes, and no electronics)
Instructor: Niklas Carlsson

Instructions:

- Read all instructions carefully (including these)!!!!
- The total possible marks granted for each question are given in parentheses. The entire test will be graded out of 40. This gives you 10 marks per hour, or six minutes per mark, plan your time accordingly.
- This examination consists of a total of 9+1=10 questions. Check to ensure that this exam is complete.
- When applicable, please explain how you derived your answers. Your final answers should be clearly stated.
- Write answers legibly; no marks will be given for answers that cannot be read easily.
- Where a discourse or discussion is called for, be concise and precise.
- If necessary, state any assumptions you made in answering a question. However, remember to read the instructions for each question carefully and answer the questions as precisely as possible. Solving the *wrong* question may result in deductions! It is better to solve the *right* question incorrectly, than the *wrong* question correctly.
- Please write your AID number, exam code, page numbers (even if the questions indicate numbers as well), etc. at the top/header of each page. (This ensures that marks always can be accredited to the correct individual, while ensuring that the exam is anonymous.)
- Answers can be provided in either English or Swedish. (If needed, feel free to bring a dictionary from an official publisher. Hardcopy, not electronic!! Also, your dictionary is not allowed to contain any notes; only the printed text by the publisher.)
- Good luck with the exam.

1) Question: Encapsulation (6)

Show the link-layer frame for a *HTTP response* with a small Web page that fits in a single frame (i) when it arrives to the link layer at the mobile client using 802.11, and (ii) when it arrives at the link layer of the gateway router closest to the client. You do not have to show all the details of the different headers; however, you should (i) explain what protocols the different headers are associated with, and (ii) provide the address information associated with the source and destination fields for each of the different headers. You can assume that the client machine uses 802.11, have a MAC address AA:AA:AA:AA:AA:AA, and has obtained a dynamic IP address 111.222.111.222 from a DHCP server, which itself has IP address 111.222.111.001 and MAC address DD.DD.DD.DD.DD.DD. The client uses a local DNS server with IP address 111.222.001.001 and MAC address DD.EE.DD.EE.DD.EE. The closest gateway router to the client has three interfaces, with the interface closest to the client having MAC and IP addresses BB:BB:BB:AA:AA:AA and 111.222.111.111, and the interface on the path to/from the server having MAC and IP addresses BB:BB:BB:BB:BB:BB and 111.222.111.122. The MAC address of the access point that the client is associated is AA:AA:AA:CC:CC:CC. Finally, the MAC and IP addresses of the HTTP server are CC:CC:CC:CC:CC:CC and 222.222.222.222. (Note: As explained on the cover page, if the necessary address information is not explicitly provided in the question, you are expected to make reasonable assumptions, and carefully motivate these assumptions.)

2) Question: 802.11 collision avoidance (6)

The 802.11 protocol can handle some hidden-terminal problems using the RTS-CTS mechanism. (a) Please explain what is the hidden-terminal problem? When and how does it occur? (b) How does the RTS-CTS mechanism help towards solving the hidden-terminal problem? Please illustrate with the communication sequence when two nodes A and C both want to communicate with an intermediate node B. (c) Please discuss the performance tradeoffs of using RTS-CTS versus not using RTS-CTS. A concrete example figure that shows the communication sequence with and without RTS-CTS may be useful here.

3) Question: 802.11 rate adaptation (4)

The wireless link characteristics can greatly impact the quality of a transmission. Please explain the relationship between signal-to-noise ratio (SNR) and bit error rate (BER), as well as how 802.11 leverage multiple physical layer encodings to adapt to changing link conditions. Your answer should include an explanatory figure together with a clear example scenario in which the client benefit from the use of rate adaptation.

4) Question: Indirect routing in cell-phone network (4)

Explain and illustrate how indirect routing is done in the context of cell-phone networks (such as GSM) with a mobile client. Please draw and discuss a figure that illustrates what happens with the routing of the network traffic as a mobile user that is away from its home network moves along a road, for example. Consider a scenario with multiple base stations and mobile switching centers.

5) Question: Routing in ad-hoc networks (4)

Describe and explain how sequence numbers are used in the context of the table-driven routing protocol Destination Sequenced Distance Vector Algorithm (DSDV). Also, please describe the most significant advantages and disadvantages of table-driven protocols (relative to on-demand protocols).

6) Question: Power save mode (4)

Illustrate and explain how the power save mode in 802.11 can be used to save energy of the mobile nodes. What is the role of the access point? Also, sketch and explain the tradeoffs between latency (x-axis) and energy usage (y-axis), as well as latency (x-axis) and buffer size (y-axis) at the access point.

7) Question: HTTP, caches, packet losses, and throughput (6)

Please consider three mobile clients on the same wireless network in Sweden that are using HTTP-based downloading to download a (very) large file from a website hosted in the US. Assume that the link closest to the users has a high loss rate, but that there is a proxy cache placed close to the clients, on the path between the clients and the server. The first client downloads the file directly from the server without any proxy involvement. The second client downloads the file via the proxy, but the proxy does not have the file and must therefore download the file itself. Luckily, the proxy is able to download file content from the server in parallel with uploading file content to the client; of course, at a speed no greater than the speed with which the proxy itself downloads the content from the server. Finally, the third client (that made the request after the second client, for example) is served directly by the proxy cache. While the above downloads do not necessarily take place in parallel, for simplicity, please assume that the clients will obtain download throughput that is TCP fair (suggesting reasonable relationships between TCP throughput, RTTs, and loss rates).

- **Part A:** Please rank the download speed of the clients.

Assume that the round trip time (RTT) between the client and proxy is 5ms, the RTT between the proxy and server is 95ms, and the loss rate between the client and proxy is 9 times as high as between the proxy and server or the client and server. Furthermore, assume that the first client achieves a throughput of 250kbps.

- **Part B:** Please estimate the throughput each client.

For this question it will help to draw the topology, identify what the conditions of each part of the network is, and reason about how the average throughput that can be achieved for the different host-to-host connections may be. Please keep in mind what protocols are used in this question and make appropriate assumptions, if needed.

8) Question: BitTorrent (4)

Please explain what incentive BitTorrent provides peers to upload pieces to others *and* why this incentive may not work as well in the video-on-demand streaming context as in the context of large file downloads?

9) Question: Packet losses (2)

Give an example how interleaving can be used to handle packet losses in video streaming?

10) Bonus question: HTTP-based Adaptive Streaming (4)

In mobile environments, the network conditions experienced by a client may vary significantly over the duration of a session. In this context, please explain what the main difference between typical (non-adaptive) HTTP-based streaming (traditionally used by YouTube, for example) and HTTP-based adaptive streaming (HAS, used by Netflix, for example) when the available bandwidth goes from bad (low available bandwidth) to good (high available bandwidth). Your answer should clearly explain the actions taken by the two different types if players. Your answer should involve reasonable assumptions, and should clearly show the buffer occupancy and video playback quality as a function of time (using appropriate figures/graphs, for example).

Finally, consider a long duration HAS session between a client and a server, for which it was observed that the average round trip time (RTT) between the client and server was 200ms and the average TCP window size was 50 packets, each of which is 1.5kB. It was also measured that each video frame is buffered on average 10s at the player. Please estimate the average video encoding and buffer occupancy measured in bytes? **(Hint:** You may want to use Little's law twice.)

Good luck!