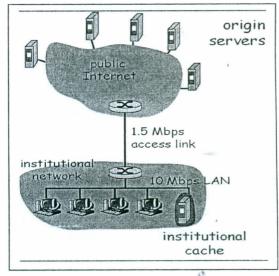
Faculty of Engineering	September 2013	M.Sc. Communications Eng.
Elect. & Comm. Eng. Dept.	Time: 3 Hours	Final Exam (M.Sc. preparation year)

## Attempt ALL questions. Assume any missing data. Books and Notes are ALLOWED.

## (7 Questions of 15 Marks Each)

- **1.a)** Using neat sketches, explain how do loss and delay occur in packet-switched networks and explain their sources and types.
  - b) During a test carried using Traceroute (which sent probes of 54 bytes packets each) to detect the path to www.eurecom.fr, the following measurements were found:
    - 1 jn1-at1-0-0-19.wor.vbns.net (204.147.132.129) 16 ms 11 ms 13 ms
    - 2 jn1-so7-0-0.wae.vbns.net (204.147.136.136) 21 ms 18 ms 18 ms□
    - 3 abilene-vbns.abilene.ucaid.edu (198.32.11.9) 22 ms $18 \mathrm{\ ms}$  22 ms $\Box$
    - 4 nycm-wash.abilene.ucaid.edu (198.32.8.46) 22 ms 22 ms 22 ms □
    - 5 62.40.103.253 (62.40.103.253) 104 ms 109 ms 106 ms
    - 6 de2-1.de1.de.geant.net (62.40.96.129) 109 ms 102 ms 104 ms□
    - i. Calculate the effective end-to-end throughput measured for each router/link.
    - ii. Is there any bottleneck in the path? Why?
    - iii. Is there any long link transtion on the path? If any: justify why and where?
- **2.** a) Discuss the reasons behind using web caching and explain how it affects the performance of web surfing.
  - b) An institution network access the Internet using a network topology as shown in figure. The requests to access objects at the origin servers was 15 object/sec, with average size of 1000000 bits each; the round trip delay from institutional router to any origin server and back to institution router = 2 sec. A web caching system is required to make the average delay per object at the client less than 1.5 second, if the LAN delay is ignored, calculate the required hit rate of the web caching system and the average bandwidth needed to pass through the uplink before and after appleving the original period.

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pass through the uplink before and after applaying the caching system.

- **3.a)** Using neat sketches, discuss the operating modes of wireless networks and explain how the wireless link charactristics are different from that of the wired links.
  - b) In a CDMA-based wireless network, two nodes (1 and 2) start sending a data of 3 bits at the same time to nodes 3 and 4 respectively. Node 1 is sending the bits {-1,1,-1} using code { 1,1,1,-1,-1,-1,-1} while Node 2 is sending {1,1,-1} using code { 1,-1,1,1,1,-1,1,1}. Draw a diagram illustrating the resulting signal sent over the communication channel and how the receiving nodes will decode the received signal and determine the output data at the decoder output of nodes 3 and 4.

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- **4.a)** Using neat sketches, discuss two schemes used for recovering from packet loss to get better performance when playing-out streamed multimedia over Internet at the receiving client.
  - **b)** Suppose the first scheme mentioned in part (a) generates a redundant chunk for every four original chunks while the second scheme uses a low-bit rate encoding whose transimission rate is 25% of the transimission rate of the nominal stream.
    - i. How much additional bandwidth and playback delay does each scheme add?
    - ii. If the first packet is lost in every group of five packets, which scheme will have better audio quality?
- **5.a)** Discuss, with aid of neat sketches, the operation of different types of priority queuing used in the Inernet to support different type of services that depend on packet classes and comare them with the operation of the original one FIFO.
  - b) Suppose that the WFQ scheduling policy is applied to a buffer that supports three classes with weights 0.5, 0.25, and 0.25;
    - i. If each class has a large number of packets in the buffer, in what sequence meight the three classes would be served?
    - ii. Suppose that classes 1 and 3 have a large number of packets in the buffer, and there are no class 2 packets, in what in what sequence meight the three classes would be served?
- **6.a)** Explain how RTCP can be used in conjunction with RTP in multicasting scenarios to control the streaming flow and discuss how nodes will divide their bandwidth between the two protocols to maintain the quality of streaming multimedia over the Internet.
  - **b**) Consider an RTP session consisting of four users, all of which are sending and receiving RTP packets into the same multicast address. Each user sends video at 100kbps.

i. What is the rate limit of RTCP traffic?

ii. How much RTCP bandwidth will be allocated by each sender and each receiver?

- **7.a)** Using neat sketches, discuss and compare the differences and similarities between the six computing paradigms, starting from dummy terminals/mainframe to the cloud computing, and discuss five new features offered by cloud computing more than other computing paradigms.
  - **b**) Explain the meaning of the following abbreviations and give examples to compare between their functions related to cloud computing services: i. SaaS, ii. PaaS, iii. IaaS, and iv. VM.

Good Luck Dr. *Hassan Soliman*