PRINCE OF SONGKLA UNIVERSITY

FACULTY OF ENGINEERING

Final Examination: Semester 1

Academic Year: 2007-2008

Date: October 5, 2007

Time: 13:30 – 16:30

Subject Number: 240-462

Room: Robot Room

Subject Title: Multimedia Networking

Exam Duration: 3 hours

This paper has 5 pages (including this page).

- Write answers in the answer book provided.
- There are 100 marks total for this exam. This will contribute 50% of the course total.

Authorised Materials:

- Anything the student can carry (except mobile/cell phones.)

Instructions to Students:

- Attempt all 9 questions.
- **Clearly Number** the answers. It is **not** required that questions be answered in order.
- Anything illegible is incorrect.
- Show all calculations, not just the final result.
- Answer briefly where possible, essays are **not** required.
- Start the answer to each question on a new page.
- The marks allocated for each question are shown next to that question. There are 100 marks total for this examination.
- *Answer questions in English.* Good English is **not** required.

Question 1. *(20 marks)*

A packet arrives at a router and is about to be queued to be transmitted via some outgoing interface, when it is discovered that the queue is full.

Explain some (at least two or three) possible strategies that might be adopted at the router to handle this situation, including the effect upon real time traffic for each strategy you suggest.

Question 2. *(8 marks)*

An audio transmission application captures voice from the sound card and transmits audio packets using a G.729 codec. G.729 is a frame-based audio codec with a 10 ms (millisecond) frame size. Each audio frame will be compressed to be 10 data bytes. This application configures the packetisation interval to be 20 ms (that is, it builds and sends RTP packets every 20 ms.)

Find the bit rate of this audio transmission. This bit rate must include the IP (20 bytes), UDP (8 bytes) and RTP (12 bytes) header overhead.

Question 3. *(5 marks)*

Which of the following are **not** useful to assist with packet classification?

(There might be more than one answer)

A) The IPv6 flow label
B) The UDP destination port number
C) The packet length
D) The IPv4 header **Protocol** field
E) The IPv4 packet identifier
F) The source IP (v4 or v6) address

For each selected choice, explain why that data is not useful.

Question 4. *(15 marks)*

For each of the following parts of this question, select the most appropriate answer from those given, and **explain** why that answer is the appropriate answer for the question.

Simply selecting the answer that happens to be correct, without a good explanation why it is correct, will **not** result in full marks.

4. i) If you wanted to decrease the amount of jitter added to packet delivery times in a network under your control, your best strategy is likely to be:
   - A) increase network bandwidth
   - B) increase queue size in routers
   - C) decrease network bandwidth
   - D) decrease queue size in routers

4. ii) If your customers complain that too many packets are being dropped inside your network due to network congestion, and you want to reduce the loss rate, your best strategy is likely to be:
   - A) increase network bandwidth
   - B) increase queue size in routers
   - C) decrease network bandwidth
   - D) decrease queue size in routers

4. iii) Your network carries some data (packets) that must not be delayed any more than cannot be avoided. Which of the following will **NOT** assist in achieving the minimum delay?
   - A) use a priority queueing scheme
   - B) classify packets and set Traffic Class
   - C) implement Congestion Notification (the DEC bit)
   - D) provide the maximum possible bandwidth (link speed)

4. iv) We sometimes care about the minimum expected packet size when measuring or planning Quality of Service, because:
   - A) link layer requires at least N bytes/packet
   - B) small packets mean many packets, which increases the amount of time that is spent routing and queueing and then transmitting the data
   - C) need to calculate link layer overheads
   - D) more packets require more queue space

4. v) Integrated Services is not generally implemented on the Internet because:
   - A) it is too new and code is not yet written
   - B) the RSVP protocol consumes too much network bandwidth
   - C) service providers do not want to provide quality of service
   - D) there are too many packet flows in the Internet

Question 5. *(15 marks)*

Integrated services (with RSVP) has the receiver(s) of real time data request the service quality that is desired.

Explain whether you think this is better, worse, or makes no real difference, to the alternate method, more commonly used, of having the sender request the service desired. Give reasons for your opinion.

Question 6. *(5 marks)*

Explain why MPEG-4 can be used to compress video more effectively than M-JPEG.

Question 7. *(10 marks)*

Policing of packet flows at entry points to a network (ingress points) is essential to properly implementing any Quality of Service mechanism.

Do you believe this is **True** or **False** ?

Why?

Question 8. *(10 marks)*

Multicast is often used for multimedia data, because it is common to need to send the data to more than one recipient.

Does the use of multicast create any particular difficulties for a router attempting to give multimedia (real time multimedia) traffic good service quality?

If so, explain what extra problems are caused by the use of multicast traffic.

Otherwise (if not) explain why not.

Question 9. *(12 marks)*

During video transmission, the compressed video frames may be larger than MTU (Maximum Transmission Unit) of the communication link. To avoid IP fragmentation, we should divide each video frame into several small segments before putting them into RTP packets. The table below gives the values of three RTP header fields: **sequence number**, **timestamp** and **M** (the Marker Bit) of 18 packets that *machine X* receives from its peer (from a video communication application.)

Using this information, find the number of ***complete video frames*** that *machine X* receives.

For each video frame, you must show which packets are combined. (Identify each packet by its sequence number.)

Sequence Number	Timestamp	Marker Bit (M)
1	10000	0
2	10000	0
3	10000	1
4	10000	0
6	16000	0
7	16000	1
8	22000	0
9	22000	0
10	22000	1
12	28000	0
13	28000	0
14	28000	1
15	34000	0
16	34000	1
18	40000	0
19	40000	1
20	46000	0
21	46000	1