Department of Computer Science
University College London

Cover Sheet for Examination Paper to be sat in March 2005

COMPZ24: Multimedia Systems

Time allowed 2.5 hours

Calculators are NOT allowed

Answer THREE questions

Checked by First Examiner: Date:

Approved by External Examiner: Date:

COMPZ24: Multimedia Systems, 2005

Answer any THREE questions

Marks for each part of each question are indicated in square brackets.

Calculators are permitted

1.   Replace with question 1   [x marks]

                                                                [Total 33 Marks]

2.   Replace with question 2   [x marks]

                                                                [Total 33 Marks]

Question 3:

a) The Internet provides *best effort* service. Explain briefly what this means in terms of what might happen to packets travelling through the network.

[4 marks]

A constant stream of packets is sent over the best effort Internet from a well connected computer in San Francisco, California to a home computer connected over a 512Kb/s ADSL line in London (a distance of approximately 10,000km). One 160 byte UDP packet is sent every 20ms.

b) Describe what the packet arrival pattern at the computer in London might look like under typical (non-failure) network conditions when no other network traffic is traversing the ADSL line. Give approximate numbers where appropriate. Explain the causes for the arrival pattern look you describe?

[9 marks]

c) What might be the expected one-way end-to-end delay (to the nearest 10ms)? State any assumptions you make. Assume the speed of light in fibre is 200,000 km/s.

[5 marks]

Windows Update now starts running automatically, downloading the latest Windows security bug fixes over same the ADSL line as the UDP traffic. Windows Update uses HTTP over TCP to perform these large downloads.

d) Describe the effects this might have on the UDP packet stream, and the process by which these effects are caused. Again, give approximate numbers where appropriate.
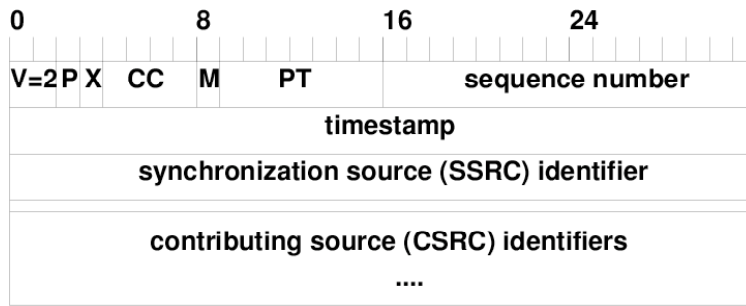
[9 marks]

e) The UDP packet stream is being used for an Internet telephony call. Describe what the effects observed by the users of the telephone call might be during the minute after Windows Update starts its download. Assume no redundant transmission is being used, that simple PCM encoding is used, and that the receiver has an adaptive playout buffer.
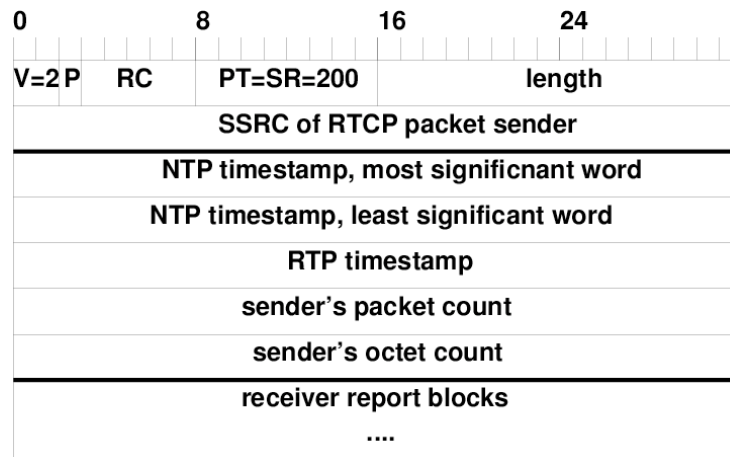
[6 marks]

[Total 33 Marks]

Question 4: RTP

```
0               8               16              24
|  |  |  |  |  |  |  |  |  |  |  |  |  |  |  |  |  |  |  |  |  |  |  |
| V=2 | P | X | CC | M |   PT   |        sequence number          |
|                          timestamp                             |
|              synchronization source (SSRC) identifier          |
|              contributing source (CSRC) identifiers            |
|                              ....                              |
```

The figure above shows the RTP (Realtime Transport Protocol) packet header for multimedia data packets. PT is the payload type field. CC is the CSRC count. (The V, P, C, and M bits can be ignored for this question).

```
0               8               16              24
|  |  |  |  |  |  |  |  |  |  |  |  |  |  |  |  |  |  |  |  |  |  |  |
| V=2 | P |   RC    |  PT=SR=200  |            length             |
|                  SSRC of RTCP packet sender                    |
|              NTP timestamp, most significnant word             |
|              NTP timestamp, least significant word             |
|                        RTP timestamp                           |
|                   sender's packet count                        |
|                    sender's octet count                        |
|                   receiver report blocks                       |
|                              ....                              |
```

The figure above shows an RTCP Sender Report (SR) message (the V, P, and RC fields and the receiver report blocks can be ignored for this question).

a) Why are both a sequence number and a timestamp field needed in the RTP data packet header? Why is one or the other not sufficient by itself?

[5 marks]

b) With reference to the RTP packet header and the use of RTCP SR (Sender Report) messages, explain how RTP might be used in the following scenarios. Note that in some cases some of the fields don't serve much purpose.

   i)   A two-way internet telephony session.

[9 marks]

ii)   A three-way audio conference using unicast data channels and an audio bridge to mix the audio.

[9 marks]

iii)   A one-way unicast audio/video stream, where the audio and video are synchronized to permit lip synchronization, but sent on separate RTP sessions on different UDP ports.

[10 marks]
[Total 33 Marks]

Question 5

Streaming media applications such as Windows Media Player and Real Player can use either UDP or TCP to transmit audiovisual streams over the Internet.

a) Discuss the relative merits of TCP and UDP for streaming media? Why do such applications need to support both protocols?

[6 marks]

b) DCCP is a new alternative to TCP or UDP for transporting multimedia data. If it were widely deployed in operating systems, how might it make the application designer's job easier?

[5 marks]

c) Discuss three options for handling packet loss in a streaming media application that uses UDP. What are their relative advantages and disadvantages?

[12 marks]

d) How appropriate are the three options you listed in (c) if the application were Internet telephony rather than streaming media? Explain your reasoning.

[10 marks]
[Total 33 marks]

END OF PAPER