Department of Computer Science
University College London

Cover Sheet for Examination Paper to be sat in March 2006

COMPZ24: Multimedia Systems

Time allowed 2.5 hours

Calculators are allowed

Answer THREE questions

Checked by First Examiner: Date:

Approved by External Examiner: Date:

COMPZ24: Multimedia Systems, 2006

Answer any THREE questions

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

Calculators are permitted

Question 1

a) H.261 can compress video to a small fraction of its original data rate, giving data rates of 64Kb/s to 2Mb/s. At 2Mb/s, very few compression artifacts are visible. For CIF video (352x288 pixels) at 25fps in YUV 4:2:0 format, what is the compression ratio when the output data rate is 2Mb/s? State any assumptions you need to make.

[5 marks]

b) Describe the basic process of H.261 video compression. Include in your description the role of the Discrete Cosine Transform (you don't need to remember the formula), the role of quantization, the difference between inter-frame compression and intra-frame compression, and any other details you consider important. Identify which steps in this process actually contribute to the final compression rate you calculated in (a).

[20 marks]

c) MPEG-1 uses similar technology to H.261, and achieves similar compression performance. Why are there two different standards? What advantages does each have over the other?

[8 marks]

[Total 33 Marks]

Question 2

a) Normal telephony audio captures analog frequencies up to 3.5KHz and has a digital data rate of 64Kb/s. Explain how this data rate came about, and the basic principles involved, given the range of frequencies to be captured. Your answer should cover both the analog-to-digital conversion process and the final digital encoding used.

[9 marks]

b) LPC codecs can achieve very low bitrates such as 2.4Kb/s, but this compression comes at the expense of synthetic sounding voice, and a lack of robustness to background noise. Explain, with reference to how LPC encoding works why LPC is not robust to background noise.

[11 marks]

c) MP3 encoding consists of transforming the audio into the frequency domain using 32 frequency bands, and using a psychoacoustic model to predict which sounds in each of these bands will be inaudible. It then uses this model to allocate the available bits such that the quantization noise in each band will be as inaudible as possible given the total number of bits available for that frame.

If you encode music with one MP3 encoder at 128Kb/s, decode it, and re-encode it with another MP3 encoder at 128Kb/s, the audio quality will suffer. With reference to how MP3 works, explain the reasons for this quality degradation.

[13 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]

b) You are charged with designing a video-telephony application for use over the public Internet. Much of your target market will be using ADSL lines. Your boss (who is technically competent but knows little about multimedia) has suggested using an MPEG-2 video codec because he has heard that DVDs use MPEG-2, and he would like to provide "DVD-quality video".

Write a brief report to your boss, commenting on the problems of using MPEG-2 for this application, and on his goal of "DVD-quality video" (you do not need to be tactful). Possible points you might wish to cover in your report are bandwidth, video quality, delay, and the effects of packet loss.

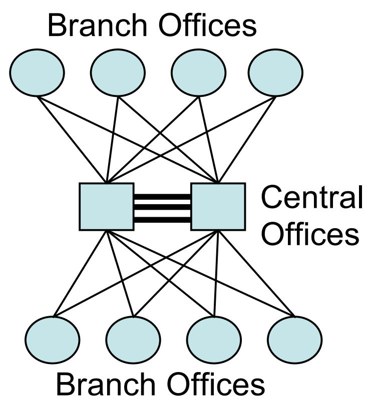
[13 marks]

c) In light of your report, your boss changes his mind, and decides to use H.263 instead for video and G.729 for audio. Give three possible options for protecting a media stream against packet loss, and explain the degree to which they are suitable for this videotelephony application. Which would you choose for audio and which for video? Justify your choices.

[16 marks]

[Total 33 Marks]

Question 4



You have just taken charge of network operations for a large bank with two central offices and branch offices in many cities, as shown in the diagram.

The bank uses its IP-based internal network to exchange business-critical data using TCP (many small connections and a few large ones) and it also uses the same network for internal VOIP telephony traffic and for distribution of multicast video feeds of Bloomberg News, BBC News 24, training materials, and speeches by the CEO. The telephony and multicast video are not congestion controlled.

On a normal day the most loaded inter-office network links run at about 45% utilization, and the network is arranged so that each branch office is linked to both central offices with the goal that failure of a network link does not result in overload.

You are revising the bank's disaster planning, and you have noted that it is common in disaster scenarios for network traffic to increase significantly. As a result, you have become concerned that the internal network could suffer congestion collapse in a disaster.

a) Discuss what might happen to the network and the applications if the network traffic increased by 30-40% at the same time as one of the two central offices going offline completely. This would move all the traffic onto the remaining links via the other central office.

[5 marks]

b) Discuss how each of the following technologies might (or might not) be used to ensure that the bank's network functions as well as possible under the circumstances:

i. Congestion control for the telephony traffic.

[5 marks]

ii. Fair queuing.

[5 marks]

iii. RSVP with Intserv

[5 marks]

iv. Differentiated Services (diffserv)

[5 marks]

v. Buying more bandwidth.

[4 marks]

c) What combination of these solutions would you choose?

[4 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) With both interactive media (such as Internet video-telephony) and streaming media, a receiver playout buffer is used to temporarily hold data that has arrived at the receiver before decoding and playout.

   i. Explain the roles of receiver buffering for streaming media and for videotelephony, and how they differ. Consider both TCP and UDP for streaming, and UDP for videotelephony.

   [7 marks]

   ii. How much data (measured in seconds or milliseconds) might be buffered in each case? What user constraints bound the amount of receiver buffering? Again, consider both TCP and UDP for streaming, and UDP for videotelephony.

   [6 marks]

c) For an Internet videoconferencing application, what are the main sources of end-to-end delay other than receiver buffering? Give approximate numbers (with justification for the numbers) for how much delay might be incurred. [Note that these delay sources may differ for the audio and video streams.]

[14 marks]

[Total: 33 marks]

END OF PAPER