Cover Sheet for Examination Paper to be sat in March 2008

COMPGZ05: Multimedia Systems

Time allowed 2.5 hours

Calculators are allowed

Answer any THREE questions

Checked by First Examiner: Date:

Approved by External Examiner: Date:

COMPGZ05: Multimedia Systems, 2008

Answer any THREE questions

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

Calculators are permitted

Question 1

a) From a high-level viewpoint, MP3 works as follows:

1. Divide the raw audio samples into timeslices.
2. Apply a Modified Discrete Cosine Transform to each timeslice.
3. Use a psychoacoustic model to predict the masking thresholds for each frequency coefficient that was output by the MDCT.
4. Don't code the frequencies that won't be heard.
5. For the frequencies that are audible, re-scale the frequency coefficients so as to quantize them more coarsely.
6. Huffman code the quantized coefficients.

   i. What is the purpose of step 5?

   [4 marks]

   ii. How do steps 5 and 6 work together to reduce the bit rate?

   [4 marks]

b) Huffman encode the following sequence of quantized DCT coefficients:
   1,1,5,3,2,1,2,4,1,3,2,1,2,1,1,1,3,2,4,1,1,2,1,1,2,2,2,1,1,3,2,1
   Write down the first five encoded symbols of the resulting bitstream in binary and the Huffman table used to produce the bitstream.

   [10 marks]

c) The description in (a), if naively implemented as a single coding pass will produce a variable bitrate stream. Explain briefly why this is so.

   [4 marks]

d) When MP3 is stored in a file, many good-quality MP3 encoders give the option of generating a constant bitrate (CBR) MP3 or a variable bitrate (VBR) MP3 file. For CBR and VBR files of the same size, the quality of the VBR file will be better. Explain why this is so.

   [5 marks]

e) If a 128Kb/s MP3 file is decoded to an uncompressed WAV file, then later recoded again as a 128Kb/s MP3 file, the quality will degrade slightly. Explain why this is so.

   [6 marks]
   [Total 33 Marks]

Question 2

a) What is interlacing? Why does analog TV use interlacing?

[4 marks]

b) When conventional broadcast-quality analog video is captured and compressed using MPEG 2, interlacing needs to be taken into account. To deal with interlacing the MPEG 2 standard introduces three main changes compared to MPEG 1:
   a) A modified YUV format
   b) Modified motion vectors
   c) A modified way of compressing the DCT coefficients.
   Describe what each of these changes is and why it helps.

[9 marks]

c) MPEG 2 uses three different frame types:
   a) I frames
   b) P frames
   c) B frames
   Explain why the use of B frames usually results in the frames being transmitted over a network in a different order from the order in which they were captured.

[6 marks]

d) An MPEG 2 video stream is being transmitted over a constant bitrate channel of 4Mb/s with a one-way delay of 20ms. For simplicity, ignore coding delay and assume each I frame is 50,000 Bytes, each P frame is 25,000 bytes and each B frame is 10,000 bytes. The Group of Picture sequence is I,B,B,P,B,B (the sequence then repeats) and the frames are captured at 25fps.

   If the frames are to be correctly played out $1/25^{th}$ of a second apart, the frame suffering the largest delay will dictate the delay suffered by the stream from capture to playout. Copy and complete the following table to calculate the delay the stream experiences:

Frame	Type	Captured	Start sending	Time to send	Finish sending	Finishes arriving	Delay
1	I	0ms	0ms	100ms	100ms	120ms	120ms
2	B	40ms					
3	B	80ms					
4	P	120ms					
5	B	160ms					
6	B	200ms					
7	I	240ms					

[10 marks]

f) If an H.261 codec was operating at a similar data rate, what delay might the stream experience? Explain your reasoning.

[4 marks]

[Total 33 Marks]

Question 3

MyChat is a (fictional) suite of Internet telephony software which can switch between several audio codecs depending on the loss rate measured. MyChat uses RTP/RTCP over UDP to transmit audio data. To set up a session, a user merely enters the email address of the user they wish to call.

a) Explain why MyChat needs to use a signalling protocol to set up video calls. What functions does this signalling protocol need to perform before the call is established?

[6 marks]

b) What effect might a network address translator (NAT) have on this signalling?

[4 marks]

When MyChat's RTP packets are transmitted across the Internet they experience jitter.

c) What is jitter?

[2 marks]

d) Give two examples of ways that packets can be jittered in the Internet.

[4 marks]

The following table is a histogram of one-way delays measured using the RTP timestamps of the audio data and converted to milliseconds. As one-way delays cannot be measured without synchronized clocks, the packet that arrived with the lowest delay is listed as having 0 delay, and all other delays are measured relative to this.

Delay	00-09	10-19	20-29	30-39	40-49	50-59	60-69	70-79	80-89
Occurrences	28	33	17	7	3	1	0	0	1

The measured minimum round-trip time is 30ms and that the audio codec uses one 20ms frame per packet.

e) Explain how an adaptive playout buffer works for interactive audio.

[4 marks]

f) Given the data above, if you were the programmer for MyChat, what might you expect the audio round trip time be, from local microphone to the remote loudspeaker and back again. Assume the jitter is the same in both directions. Justify your answer.

[8 marks]

g) What would you propose to do to minimize the effect of packet loss in MyChat? Justify your answer.

[5 marks]

[Total 33 Marks]

Question 4

a) When you are sending compressed video data over a TCP connection, it doesn't greatly matter how the application writes data into the network socket. No additional packet headers need to be added by the application. When sending data over a UDP flow, things are somewhat different.

    i. Why does the application need to add extra headers to each UDP packet? What information is it most important to put in such headers?

[7 marks]

    ii. Why does it matter precisely how the application chooses to break the data up into separate calls to write() on the UDP socket? Use a diagram if it helps to explain your answer.

[5 marks]

b) In the context of MPEG Video Compression, describe how packet loss affects video decoding when video is streamed over UDP without using any form of packet loss repair or retransmission. What sort of artefacts would you expect to see, why would they occur, and how long would they persist? Give as much detail as you need to on how MPEG works to justify your answer.

[8 marks]

c) Given that the application in (a) is videoconferencing, what technique would you choose as the main way to improve the video quality? Explain your answer.

[4 marks]

d) YouTube streams video over TCP. What are the advantages of streaming over TCP?

[3 marks]

e) For Internet-based cable TV, why would the YouTube solution of streaming over TCP not be acceptable? You answer should discuss the constraints imposed by TCP streaming and how these constraints have to be handled by the streaming application.

[6 marks]

[Total 33 Marks]

Question 5

You have been employed by a start-up company to write a streaming video application for use by home users. The boss has suggested that H.264 would be a good codec and that one mode of operation is to stream video from stored files over DCCP. Your video server is to run on Linux, and the DCCP API is basically the same as non-blocking UDP:

- When the application calls write() on the DCCP socket, the packet is buffered by DCCP in the kernel for sending.
- DCCP determines when precisely to send the packet depending on its congestion control loop.
- If the sender's DCCP socket buffer is full, the call to write() will return an error and the packet will not be sent.

a) What (if any) advantage is there from using DCCP instead of TCP if the video data is stored at only a single data rate on the server?

[5 marks]

b) If the server stores multiple versions of the video coded at different data rates, how can it switch between them without causing predictor error?

[5 marks]

c) How might the server determine that it is time to switch encodings?

[5 marks]

d) The size of the DCCP kernel socket buffer can be set using a setsockopt() call. What advantages and disadvantages do you see in setting the size to:
   i. 3KBytes.

[4 marks]

   ii. 256KBytes.

[4 marks]

e) How do you plan to limit the damage caused by lost packets?

[5 marks]

f) Would your answer differ if the application were videoconferencing? Explain why or why not.

[5 marks]

[Total 33 Marks]

END OF PAPER