UNIVERSITY OF LONDON
IMPERIAL COLLEGE OF SCIENCE, TECHNOLOGY AND MEDICINE

EXAMINATIONS 1996

BEng Honours Degree in Computing Part III
BEng Honours Degree in Information Systems Engineering Part III
MEng Honours Degree in Information Systems Engineering Part III
MSc Degree in Computing Science
for Internal Students of the Imperial College of Science, Technology and Medicine

This paper is also taken for the relevant examinations for the
Diploma of Membership of Imperial College
Associateship of the City and Guilds of London Institute

PAPER 3.46 / I3.12

MULTIMEDIA SYSTEMS

Thursday, May 9th 1996, 10.00 - 12.00

Answer THREE questions

For admin. only: paper contains
5 questions
4 pages (excluding cover page)

Section A *(Use a separate answer book for this Section)*

1a Using a clearly labelled diagram, explain how the fractal transform can be used to compress a colour image.

When the fractal transform is used to compress an image, what determines

i) the compression ratio, and

ii) the quality of the decompressed image ?

b A 256x256 pixel, 16-bit grey scale image is to be compressed using the fractal transform. Clearly stating any assumptions you make, estimate the compression ratio that will be achieved.

c How could you extend the fractal transform technique to make it suitable for compression of moving images ?

The three parts carry, respectively, 50%, 30%, 20% of the marks.

2a A new "Song Memory" device is proposed which samples the sound from a single human singer, converts it into a MIDI data stream and stores the MIDI data in battery-backed RAM. The device can play the MIDI data back as sound. To playback, it uses repetitions of a short sample of the voice taken during the recording and it plays the sample back at different speeds for different pitches.

i) Draw a block diagram of the hardware required to capture the sound into the computer and replay it, explaining the function of each component. (You may assume that pitch information is extracted by software).

ii) Clearly showing your working and stating any assumptions, estimate the minimum viable sample rate for the system, assuming that a soprano singer can sing up to C_6 (two octaves above middle C).

Explain whether or not this sample rate would be a good choice for storing the short sample of voice waveform that is the basis of the playback system.

iii) Clearly stating your assumptions, estimate the compression achieved by storing the singing as MIDI data instead of as a compact-disc quality sampled waveform.

What data must be stored in addition to the MIDI data to play back the tune ?

What is lost by storing the singing as a MIDI data stream ?

b The software that extracts the pitch requires a formula to convert pitch in Hz to a MIDI note number. Carefully explaining your working, derive a formula that will turn pitches between 110Hz and 1760Hz into a corresponding MIDI note number.

(Middle C is MIDI note number 60 decimal and the A above it is 440Hz).

The two parts carry, respectively, 60%, 40% of the marks.

3a The MPEG audio standard achieves a high compression ratio partly because it does not store sounds we cannot hear.

Explain fully which sounds are discarded during MPEG audio compression.

Which, if any, sounds are discarded that we can hear ?

Are any sounds added that were not in the original signal ?

b The MPEG-1 video standard achieves a high compression ratio partly because it discards information that we cannot see. Explain how this occurs.

c A 256x256 pixel, 15-bit RGB, 25 frame per second movie is stored uncompressed in RAM with two bytes storing each RGB value. It has been suggested that the spare bit in each of the two bytes is used to store sound.

Explain whether or not this is feasible, estimating the sample rate, and sound quality (in bits per sample) that could be achieved.

Would this system be suitable for storing stereo orchestral-quality music ?

The three parts carry, respectively, 50%, 30%, 20% of the marks.

Turn over ...

Section B *(Use a separate answer book for this Section)*

4a Explain what is meant by QOS. What are the main QOS parameters that we look out for in multimedia transmission?

b What are the standard acceptable values of QOS parameters for a human being with regard to video conferencing ?

c Two networked computers are required to transmit multi-media information bidirectionally for purposes of video conferencing. The video conferencing software is supposed to give the best possible QOS characteristics that the network can cope with. By monitoring the transmission time for information to arrive from one computer to the other, the software automatically adjusts the quality of speech encoding and number of frames transmitted per second. You should assume that sound is transmitted at a maximum quality of 8 kbits/sec.

Using the following multi-media operating systems (MMOS) routines suggest a skeleton algorithm in a high-level language (C, Pascal, etc.) that can adapt based on network connection:

```
float MM_Get_Time_to_Transmit_Last_Video_Frame();

float MM_Get_Time_to_Transmit_One_Second_of_Sound();

void MM_Set_Video_Sampling_Rate(float frames_per_second);

void MM_Set_Sound_Sampling_Rate(float samples_per_second);

float MM_Get_Current_Sound_Sampling_Rate();

float MM_Get_Current_Frame_Rate();
```

N.B. Assume that all times returned and set are in seconds and that all rates are per second. If there is not enough bandwidth for video, then only transmit sound and set the video sampling rate to 0 frames per second. Assume that bandwidth is always sufficient to transmit sound.

d Given that the bandwidth of the internet connection of computer A is 64 kbits per second, comment on the possibilities of using such a bandwidth for video conferencing with a computer B that is connected to the internet. A list of comments is expected, rather than an essay answer.

The four parts carry, respectively, 15%, 20%, 50% , 15%, of the marks.

Turn over ...

5. GSM and PCN mobile telecommunications are emerging as one of the global standards for personal communications.

a Briefly describe the GSM communication model and bandwidths involved.

b Briefly explain the characteristics of the Short Message Service (SMS) and write a paragraph on possible applications using SMS.

c Explain the difference between direct connection (on-line) and store-and-forward mechanisms in communication.

d Data transmissions over GSM are currently conducted at 9.6 Kbits per second. Computer interfaces are available from several GSM handset manufacturers allowing mobile users to send and receive data and faxes.

   i) Using the information above estimate the time taken to transmit one A4-sized facsimile page of typed text using a standard GSM data terminal. You should use estimations of the amount of data to be transmitted and assume that the fax data is transmitted using run length encoding (RLE).

   ii) Comment on the prospects of using GSM transmission for video and how sufficient bandwidth might be made available.

The four parts carry, respectively, 15%, 20%, 15% , 50%, of the marks.

End of paper