

Two hours

**UNIVERSITY OF MANCHESTER  
SCHOOL OF COMPUTER SCIENCE**

Mobile Systems

Date: Tuesday 28th May 2013

Time: 09:45 - 11:45

---

**Please answer Question ONE and also TWO other Questions  
from the remaining THREE Questions provided**

---

This is a CLOSED book examination

The use of electronic calculators is permitted provided they  
are not programmable and do not store text

**[PTO]**

**1. Compulsory question**

Answer all of the following parts of this question.

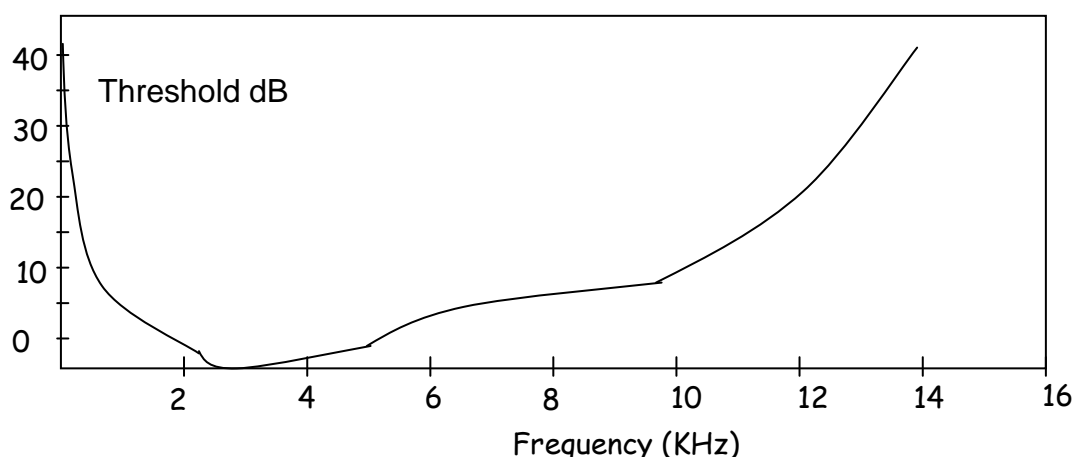
(2 marks each)

- a) What is the Nyquist criterion, and how should an audio signal from a microphone be processed to ensure that this criterion is met?
- b) What is a Fourier Transform, and what does it do to a time-domain signal?
- c) How much memory would be required to store 1 hour of ITU-G11 (A-law) encoded telephone-quality speech?
- d) What is the difference between “soft” and “hard” real-time systems?
- e) Sketch the organisation in memory of a double-buffered DMA real-time data output process.
- f) What is meant by interleaving and why is it beneficial when using forward error correction (FEC)?
- g) What is a spectrograph (or spectrogram) and how can such a graph be conveniently displayed on a computer screen?
- h) What does ‘white’ mean when describing electrical or acoustical noise and what does ‘white’ acoustical noise sound like?
- i) What are the main properties of Huffman coding as used for psycho-acoustic music (mp3) encoding and also JPEG image encoding.
- j) Explain the difference between waveform coding & parametric coding as applied to narrow-band speech compression.

[PTO]

2. **This question is about the use of bit-rate compression when digitising speech and music.**

- a) How does 'quantisation noise' arise when digitising narrow-band speech?  
 If a narrow-band speech coder has a 10 bit uniformly quantising analogue to digital converter and a sampling rate of 16 kHz, estimate the maximum achievable signal-to-quantisation noise ratio (SQNR) assuming the speech to be approximately sinusoidal. What is the 'dynamic range' of the encoded speech given that the SQNR must be at least 30 dB for acceptable speech quality?  
 State what assumptions it is reasonable to make about the nature of the quantisation noise. (6 marks)
- b) Figure 1 shows a commonly assumed 'masking contour in quiet' for the psycho-acoustical threshold of a person's hearing.
- Explain what this graph tells us that is important for mp3 music coding. (2 marks)
  - How could a simple experiment be performed to verify the shape of this graph for a given listener? (2 marks)
  - How could simple experiments be performed to verify the effects of simultaneous (frequency) masking and temporal masking. (4 marks)
- c) With the aid of a block-diagram, explain how the psycho-acoustical properties of hearing mentioned above are exploited by mp3 encoders to allow high quality music to be recorded (or transmitted) at bit-rates considerably lower than is used for compact disk recordings. (6 marks)



**Figure 1: Masking contour 'in quiet'**

[PTO]

3. **This question is mainly about bit-error control**

- a) Why is the use of forward error correction (FEC) much more important with mobile systems using wi-fi and cellular radio than with systems that use wired connections? (4 marks)
- b) A 4-bit integer ' $B_3 B_2 B_1 B_0$ ' is Hamming coded by appending three additional bits  $P_1$ ,  $P_2$  and  $P_3$  to allow the correction of any single bit-error that may result from its radio transmission by a mobile system. Show how a suitable set of three additional bits may be derived for the given 4-bit integer and explain how the correction would be done, if necessary, at the receiver. (8 marks)
- c) Explain the principle of a cyclic redundancy check (CRC). Why are CRC checks often used in conjunction with convolutional coders and Viterbi decoders? (4 marks)
- d) According to the Shannon-Hartley Law, what is meant by the capacity  $C$  of a communication channel? Estimate the channel capacity for a mobile radio channel of band-width 4 kHz, where the reception is affected by 'additive white Gaussian noise' (AWGN) and the signal-to-noise ratio at the receiver is 60 dB. To what extent can the use of FEC increase the channel capacity? (4 marks)

[PTO]

4. **This question is concerned with still and moving image compression and transmission, together with high-level mobile system design issues.**

- a) Describe the steps involved in compressing a still RGB image using the JPEG algorithm. (6 marks)
- b) What controls the degree of compression achieved by the JPEG algorithm? (2 marks)
- c) Why is it inefficient to compress a moving image by simply using JPEG to compress each image in the sequence, and how is this inefficiency overcome in MPEG compression? (2 marks)
- d) A museum is considering deploying a mobile audio visual system based on a PDA with a built-in digital camera, using WiFi (802.11) radio communications to a central server. The idea is to display a graphical icon next to each exhibit that identifies the exhibit, and to use the camera to get an image of the icon so that the server can supply relevant audio and visual information about the exhibit.

Discuss the strengths and weaknesses of the proposed mechanisms for interpreting the icons:

- i) capture the raw image and send it to the server for analysis
  - ii) compress the raw image and send a JPEG file to the server for analysis
  - iii) analyze the image on the PDA and send the exhibit ID to the server. (2 marks each)
- e) It is realised that the PDAs are likely to be stolen unless there is a good security mechanism in place. A non-technical museum manager suggests leaving the camera on the PDA on so that the server can tell where it is at all times.
  - i) How do you explain, in non-technical terms, why this suggestion is likely to be impractical? (3 marks)
  - ii) Suggest a better security mechanism. (1 mark)

**END OF EXAMINATION**