Two hours

Question ONE is COMPULSORY

**UNIVERSITY OF MANCHESTER
SCHOOL OF COMPUTER SCIENCE**

Mobile Systems

Date: Monday 1st June 2015

Time: 09:45 - 11:45

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

Use a SEPARATE answerbook for EACH Question

This is a CLOSED book examination

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

1.   **Compulsory question**

Answer all ten of the following short questions (worth 2 marks each)

a)   What three steps are required to convert a physical signal such as sound, which is continuous in time and amplitude, into a form suitable for processing by the CPU in a mobile phone?

b)   What is the main difference between the pure ALOHA wireless protocol and the slotted ALOHA wireless protocol, and how does the latter increase the maximum potential throughput of the wireless medium?

c)   Describe briefly, giving the main advantages and disadvantages of each, the following three mechanisms for handling real-time data streams: i) polling; ii) interrupts; iii) DMA (direct memory access).

d)   Why does reducing the clock frequency of a CMOS circuit not directly improve the energy-efficiency of the circuit? What additional change in the circuit operating conditions does a reduced clock frequency allow that does improve energy-efficiency?

e)   What energy-efficiency trade-offs arise when the following techniques are considered to enhance radio data communications in a mobile system: i) data compression; ii) error detection and correction; iii) radio power modulation?

f)   What are the main goals of the '4G IMT-Advanced' standard for mobile telephony as proposed by the International Telecommunications Union (ITU)?

g)   What is meant by interleaving and why is it beneficial when using forward error correction (FEC)?

h)   What are the main properties of Huffman coding as used for psycho-acoustic music (mp3) encoding and also JPEG image encoding.

i)   Explain the difference between waveform coding & parametric coding as applied to narrow-band speech compression

j)   What is a spectrograph (or spectrogram) and how is such a graph normally displayed on a computer screen?

2. This question is about sound digitisation and compression

   a) What is the normal range of sound frequencies that a (young) human is assumed to be able to hear? How does this assumption determine the sampling rate used for audio compact discs (CDs)? (3 marks)

   b) Define the dB_SPL decibel scale commonly used for sound. What is the approximate dynamic range of human hearing, from the quietest sound that can be heard to the loudest sound that can be tolerated, in decibels? Approximately how many bits per sample would be required to digitally record sound with this dynamic range, using uniform quantisation? What is the approximate dynamic range of the sound that can be recorded with uniform quantisation and without compression on an audio CD? Why was this dynamic range adopted and why is it still considered acceptable for recordings on mobile phones and tablets? (5 marks)

   c) A 12-inch long-playing vinyl record could store about 40 minutes of stereo music in analogue form (using both sides of the vinyl). If an audio CD was to store the same amount of music in digital form, how many Mbytes of data must it hold (ignore any additional data required for error correction)? (2 marks)

   d) A solid-state telephone answering machine is designed to store up to 40 minutes of recorded ITU-G711 (A-Law) telephone-quality messages. How much solid-state memory does it require? (2 marks)

   e) Describe how mp3 compression works to reduce the amount of data required to store the music "ripped" from a CD on a mobile phone. Approximately how much compression can be achieved if there is required to be little audible loss of quality in the recorded music? (8 marks)

3. This question is about cellular mobile telephony

   a) Explain the principle of cellular spatial multiplexing as used by mobile telephony.
      (3 marks)

   b) How does the use of forward error correction (FEC) in cellular mobile systems increase the energy efficiency and effectiveness of spatial multiplexing by frequency re-use?
      (3 marks)

   c) Explain the terms 'circuit switching' and 'packet switching' as applied to mobile communications networks. What are the main advantages and disadvantages of each of these mechanisms. Which of these mechanisms are used for voice communications by second (2G) and third (3G) generation mobile telephony? How does 4G telephony deal with voice traffic currently and how may this change in future? (8 marks)

   d) Explain how the 'code division multiple access' (CDMA) multiplexing mechanism used by the third generation of cellular mobile telephony (3G) is able to share a radio spectral band in any given cell among many users. What do you consider to be the three main advantages of CDMA over the mechanism used by 2G-GSM telephony? (6 marks)

4. This question is about bit-rate compression for images and bit-error control for mobile transmission.

   a) How are the known characteristics of perception by the human eye exploited by mobile phones to achieve bit-rate reduction for images? (7 marks)

   b) To what extent would you expect a JPEG compressed image to be more sensitive to the effect of uncorrected bit-errors than an uncompressed image such as a bit-map? Explain your answer. (3 marks)

   c) State what is meant by 'Hamming Distance' and explain how it is calculated for two 7-bit numbers. Why is the concept of 'minimum Hamming distance' important for detecting and correcting bit-errors? (3 marks)

   d) 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. (7 marks)

END OF EXAMINATION