COMP28512

<u>Two hours</u>

UNIVERSITY OF MANCHESTER
SCHOOL OF COMPUTER SCIENCE

Mobile Systems

Date: Friday 25th May 2012

Time: 14:00 - 16:00

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

For full marks your answers should be concise as well as accurate
Marks will be awarded for reasoning and method as well as being correct

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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**

Answer all of the following parts of this question. (2 marks each)

a) In 1948, the Manchester "Baby" computer used 3.5 kW of electrical power while executing 700 instructions per second. A processor in a modern mobile system typically uses 20 mW while executing 200 million instructions per second (MIPS). How much more efficient than the Baby is the modern processor?

b) An analogue signal can be converted into a form suitable for processing by computer by (i) passing it through an anti-alias filter, then (ii) sampling it and finally (iii) quantizing the samples. Explain why each of these three processes is necessary.

c) CD-quality sound is sampled in stereo at 44.1 kHz with 16-bit per sample uniform quantisation. What is the data capacity of a CD that can store one hour of CD-quality sound? (Express your answer in bytes.)

d) What is meant by the terms 'time-domain' representation and 'frequency-domain' representation as applied to a digitised audio signal. How can one of these representations be converted to the other?

e) Describe the characteristic features of an event-driven system. Why does designing a real-time digital system to be 'event-driven' help to minimise its power consumption?

f) A mobile communication system uses a radio channel of bandwidth 4000 Hz. The reception is affected by 'additive white Gaussian noise' (AWGN) whose constant level is such that the signal-to-noise ratio is 36 dB. According to the Shannon-Hartley Law, what is the maximum bit-rate that can be conveyed with arbitrary low bit-error probability over this radio channel?

g) A narrow-band speech coder has a 10-bit uniformly quantising analogue-to-digital converter and a sampling rate of 8 kHz. Estimate the maximum achievable signal-to-quantisation noise ratio (SQNR) assuming the speech to be approximately sinusoidal. What are the likely characteristics of the quantisation noise?

h) What is the maximum bit-rate achievable over a channel of bandwidth 0 Hz to 4 kHz using binary signalling?

i) What are the essential similarities and differences between the Discrete Fourier Transform (DFT), the Fast Fourier Transform (FFT) and the Discrete Cosine transform (DCT)?

j) What is usually meant by 'zero-padding' and why is it often applied in the time domain before applying a Fast Fourier Transform (FFT)?

[PTO]

2 This question is about spectral analysis, the Fast Fourier Transform (FFT) and time-domain speech coding for mobile telephony

a) The FFT magnitude spectrum plotted in figure 1 was obtained by applying a 512 point FFT to a segment of male voiced speech sampled at 8 kHz. Point out the important features of this graph and estimate the fundamental frequency and the frequencies of any 'formants'. (6 marks)

b) In human speech production, what determines the fundamental frequency and what causes the formants? How are these characteristics of speech perceived by the listener? (2 marks)

c) What features of speech signals and speech perception by human listeners are exploited by:
(i) the G711 64 kb/s A/MU-Law PCM standard (5 marks)
(ii) linear prediction based speech coders (7 marks)
to reduce the bit-rate necessary to transmit acceptable 'telephone quality' speech.

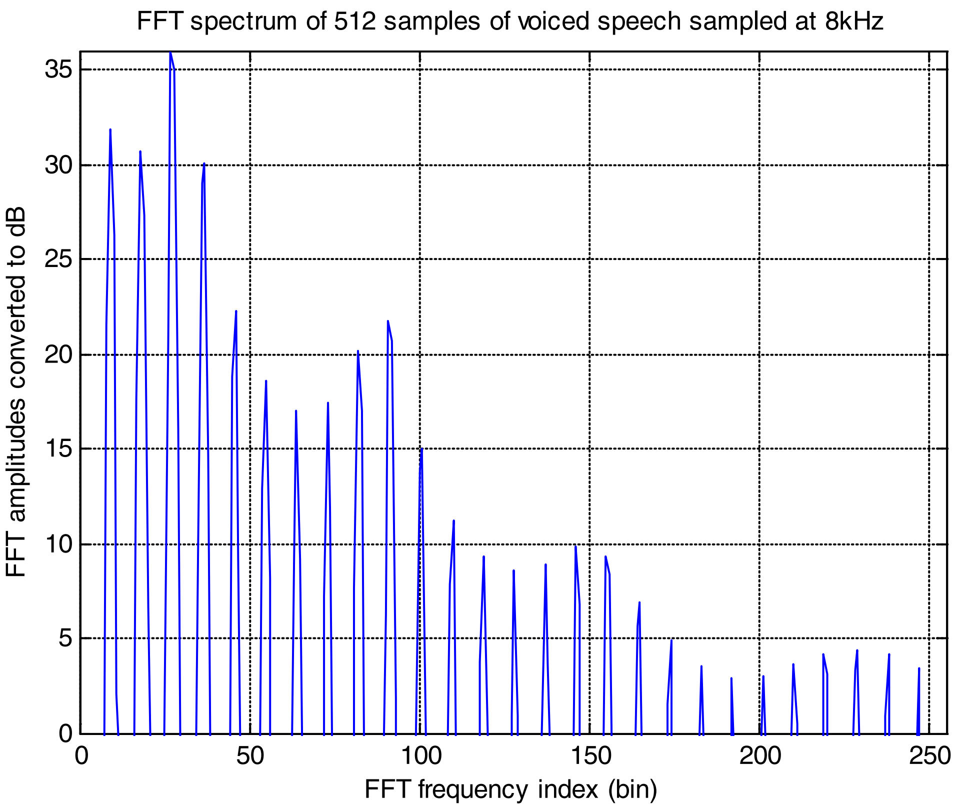

Figure 1 : FFT magnitude spectrum of a segment of voiced speech sampled at 8 kHz

3. This question is about image compression for mobile transmission, the need for Huffman coding and the derivation of a Huffman code

   a) How is bit-rate compression achieved for images according to the JPEG standard? (7 marks)

   b) Why is run-length coding and Huffman coding required by JPEG? (4 marks)

   c) Symbols A, B, C, D, E and F represent six possible values of quantised luminance and have probabilities: 0.3, 0.1, 0.15, 0.15, 0.2 and 0.1 respectively. Devise a Huffman code for these six symbols and consider how it would be decoded. (7 marks)

   d) Why would you expect a JPEG compressed image to be more sensitive to the effect of bit-errors than an uncompressed image such as a bit-map? (2 marks)

4. This question is concerned with multimedia communication, together with high-level mobile system design issues.

   a) Sketch and describe the buffering mechanism used to smooth out network jitter when streaming audio or similar real-time media data over the Internet. Take particular care to indicate and describe the function of the buffer high- and low-water marks. (6 marks)

   b) How can the adverse effect of lost packets be reduced when streaming real-time media over the internet? (2 marks)

   c) A mobile system is required to transmit and receive real-time video information over a wireless network link. Discuss the trade-offs that must be made when choosing the video quality, data compression, error correction and battery life of the mobile system. (6 marks)

   d) What are burst errors as occur with digital communications over radio channels? What techniques can be used to reduce the impact of burst errors on the quality of received media signals? (3 marks)

   e) What techniques may be used to reduce the highly-variable bit-error rates that could occur with the radio communication channels that are used by mobile systems? (3 marks)

END OF EXAMINATION