Department of Electronics& Computer Engineering

Mid-Session Test January 2001

EE3052B - MultiMedia Signal Processing

Time allowed 1 Hour

Answer *<u>three</u>* out of five questions

Ensure that your registration number is written clearly on the front cover.

Q. 1

(a) A signal with the highest frequency content of 20 kHz is sampled at a rate of 20 kHz, sketch the spectrum of the sampled signal and write an equation for the spectrum describing the spectral aliasing for this case.

(3 Marks)

(b) Write an expression describing the signal to quantisation noise ratio (SQNR) in an n-bit uniform quantiser and obtain the SQNR at the output of a 16 bit quantiser.

(3 Marks)

(c) With the aid of a block diagram, describe the outline of a system for re-sampling a HiFi digital audio signal $x(m)$ originally sampled at a rate of 44 kHz , at a new (broadcast quality) sampling rate of 16 kHz.

(4 Marks)

Q2.

(a) The discrete Fourier transform (DFT) is given by

$$X(k) = \sum_{m=0}^{N-1} x(m) e^{-j\frac{2\pi}{N}mk} \qquad k = 0, \ldots, N-1$$

(i) Describe what is meant by the time and the frequency resolutions in the discrete Fourier transform.

(ii) Write an expression describing the relation between the time and the frequency resolutions. Hence, explain the Uncertainty Principle.

(5 Marks)

(b) A DFT is used as part of a digital signal processing system for the analysis of an analog signal with significant frequency content of up to 3.5 kHz. Calculate:

(i) The minimum sampling rate F_s required, and

(ii) The number of time domain samples required to achieve a frequency resolution of 10 Hz at the minimum sampling rate.

(5 Marks)

Q.3

(a) Obtain an expression for and sketch the spectrum of the discrete-time impulse function given by

$$x(t) = \delta(t) = \begin{cases} 1 & t = 0 \\ 0 & t \neq 0 \end{cases}$$

Explain why the impulse function is commonly employed as a useful test signal.

(2 Marks)

(b) Show that the Laplace transform of a sampled signal is periodic with respect to the frequency axis ω of the complex variable of $s=\sigma+j\omega$.

(4 Marks)

(c) Derive the z-transform from the Laplace transform.

(4 Marks)

Q.4

(a) Describe the effect of complex pairs of poles and zeros on the frequency response of a filter.

(1 marks)

(b) The z-transfer function of a finite impulse response (FIR) system is given by

$$H(z) = \left(1 - 0.5z^{-1} + 0.25z^{-2}\right)\left(1 - 2z^{-1} + 4z^{-2}\right)$$

(i) Find the zeros of the filter. (3 marks)

(ii) Write the z-transfer function of this filter. (3 marks)

(iii) Hence, find an expression for the frequency response of this filter.

(3 marks)

Q.5

Using the inverse Fourier transform and the window design technique, design a bank of digital finite impulse response filters, for an audio graphic equaliser application, to split a total bandwidth of 20 kHz into 4 equal width sub-bands. Write the impulse response of each sub-band filter. State how can this filter be made causal.

(10 Marks)