Brunel University

Department of Electronic and Computer Engineering

Final Examination June 2001

EE3052B - MultiMedia Signal Processing

Time allowed 3 Hours

Answer *five* out of eight questions

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

Q1

(a) Traditional protection of artistic property has been undermined by advances in technology. In what specific way has the widespread use of each of the following prompted the need for electronic watermarks

(i) Image scanners
(ii) Computers
(iii) Internet

[3 marks]

(b) Explain with the aid of diagrams how an invisible complete watermark can be inserted in an image and extracted from an image.

[4 marks]

(c) Electronic watermarks may be weakened or obliterated by malicious attack.

(i) Craver defines four forms of attack: robustness, presentation, interpretation and legal attacks. Briefly describe what each of these attacks attempts to do.
(ii) Name three different types of robustness attack. In each case briefly describe the nature of the attack and a defence against it.

[4 marks]

(d) Describe two different uses of watermark labels for content description on the Internet and the associated benefits for surfers and web-site owners.

[2 marks]

(e) Record companies are very concerned by the prospect of 'free' downloads of MP3 format music off the Web.

(i) Explain with the aid of a diagram how watermark captions can prevent 'I-way robbery' by identifying both the owner and distributor of a work.
(ii) State what the concerns and problems are of using watermarks to protect recorded music. You may wish to consider technological, legal and social issues in your answer.

[7 marks]

Q 2

(a) Briefly state what is meant by non-stationary and stationary signal processes, and give an example of each type of process.

[2 marks]

(b) Write the equation for the relationship between autocorrelation function and power spectrum.
Write the relationship between autocorrelation, power and variance.
Obtain the power spectrum and autocorrelation of:

(i) A unit amplitude discrete-time impulse.
(ii) A white noise of unit variance.

[6 marks]

(c) Explain why probability is a useful function for measuring information and why a function of $\log_2$ of probability is usually employed.

[3 marks]

(d) State what is the Entropy of a random process, and write an equation for the entropy of an *N*-valued random process.

[3 marks]

(e) Speech is based on the use of about 40 basic acoustic symbols, known as phonemes (or phonetic units), these are used to construct words, sentences etc. Assuming that all phonetic units are equi-probable, and that the average speaking rate is 120 words per minute, and that the average word had 4 phonemes/word, calculate the minimum number of bits per second required to encode speech at the average speaking rate.

[6 marks]

Q3

(a) Describe the sources of line and acoustic echo in landline and mobile telephony, and state:

(i) What are the effects of a short-delay echo, and a long-delay echo, on the intelligibility of voice conversation. What is the accepted limit of 'tolerable' echo in terms of the round trip delay?

(ii) Estimate the total echo in a GSM-based mobile phone system if the main feedback is assumed to be from the speaker to the microphone with an assumed distance of 10 cm in length. What is the possible total echo if the call goes thorough a satellite system (assume a communication satellite distance from the earth of 22,223 miles and velocity of light is 186,000 miles per second).

[5 marks]

(b) Draw a block diagram for an acoustic echo suppression system and explain how echo suppression works and what are the drawbacks of echo suppression.

[5 marks]

(c) Draw a block diagram for an acoustic echo cancellation system and explain its operation. State and explain your choice of the followings:

(i) The filter structure for echo cancellation (i.e. IIR or FIR)

(ii) The adaptation algorithm selected.

[5 marks]

(d) Discuss, and list, the advantages of using a subband echo cancellation system in mobile phone and teleconference systems.

[5 marks]

Q4

(a) A Wiener filter is to be used for the restoration of a noisy signal modelled as

$$y(m) = x(m) + n(m)$$

where $y(m)$, $x(m)$ and $n(m)$ are the noisy signal, the original signal and the noise respectively.

(i) Rewrite the above Equation in frequency domain. [1 mark]

(ii) Briefly describe the principles on which the Wiener filter is based. [1 mark]

(iii) Prove that in frequency domain the Wiener filter is given by

$$W(f) = \frac{P_{XX}(f)}{P_{XX}(f) + P_{NN}(f)}$$

where $P_{XX}(f)$ and $P_{NN}(f)$ are the signal and the noise power spectra.

[8 marks]

(b) Figure Q4 shows a communication channel model followed by a channel equaliser.

(i) Write the time domain equation, and the equivalent frequency domain equation, describing the channel output in terms of the input signal, the channel response and the noise.

[2marks]

(ii) Derive the frequency domain Wiener filter for the equaliser.

[6 marks]

(iii) Describe a practical method for training of a Wiener equaliser for removal of the channel distortion.

[2 marks]

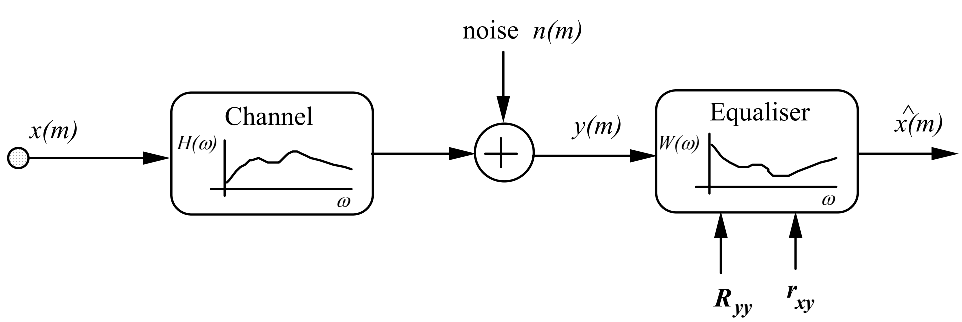

Figure Q4.

Q5

(a) Describe the factors that affect the choice of an adaptation algorithm in a given application.
What are the main differences between the following adaptation methods: (i) Wiener, (ii) the recursive least squared error (RLS), (iii) the steepest descent, and (iv) the LMS method.

[4 marks]

(b) Consider an adaptive finite-duration impulse response (FIR) filter with input $y(m)$ and filter coefficient vector $\boldsymbol{w} = [w_0, w_1, ..., w_{M-1}]$.
The steepest descent algorithm for adaptation of this filter is given by

$$\boldsymbol{w}(m+1) = \boldsymbol{w}(m) + \mu\left(-\frac{\partial E\left(e^2(m)\right)}{\partial \boldsymbol{w}(m)}\right)$$

Show that the equation describing the error in the filter coefficient vector is given by

$$\boldsymbol{v}(m+1) = [\mathbf{I} - \mu \boldsymbol{\Lambda}]\, \boldsymbol{v}(m)$$

where $\boldsymbol{\Lambda}$ is the eigen vector matrix of the correlation of the input signal.

[6 marks]

(c) Assuming that the maximum eigenvalue of a signal is 2 and the minimum eigenvalue is 0.2

(i) Calculate the eigenvalue spread.
(ii) The bounds on adaptation step size.
(iii) The decay factor of the error equations for the fastest and the slowest converging coefficients of the filter given that the adaptation stepsize is 0.4.

[5 marks]

(d) Derive the equation describing the least mean square (LMS) filter adaptation algorithm, and compare the performance of the LMS method at the point of convergence with that of the steepest descent method.

[5 marks]

Q6

(a) Figure Q6 is a block diagram illustration of a linear prediction model of signal $x(m)$.

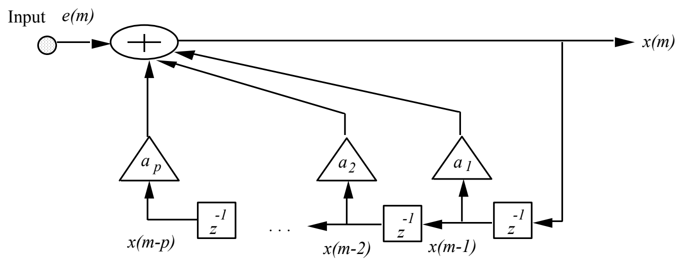

Figure Q6

(i) Write the time-domain equation describing the input-output relation of this model.

[1 Mark]

(ii) Derive the least squared error solution for the linear predictor coefficients.

[7 Marks]

(iii) Taking the z-transform of the input-output equation obtain the transfer function of the linear prediction model.

[2 Marks]

(b) The first three auto-correlation coefficients of a signal process are as follows

$$r(0) = 1.0,\ r(1) = 0.497,\ r(2) = -0.362.$$

(i) Obtain the coefficients of a second order linear prediction model for the process, and express the coefficients in polar form.

[6 Marks]

(ii) Use the model to write an expression for the frequency response of the process.

[2 Marks]

(iii) Write the inverse filter equation in the time domain, and sketch a pole-zero diagram showing the position of the poles and the zeros of the predictor and its inverse.

[2 Marks]

Q7

(a) A HiFi audio signal has significant audible energy up to a highest frequency of 20 kHz. It is required that after quantisation the signal should have a minimum quantisation signal to noise ratio of 90 dB.
Using this specification, calculate how many minutes of a stereo digital format audio signal can be stored on compact disc (CD) with a capacity of 650 Mega bytes.

[3 Marks]

(b)

(i) Using the Fourier transform show the effect of zero-insertion by a factor of L on the spectrum of a signal in both time and frequency domains.

[4 Marks]

(ii) 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 to a new (radio broadcast quality) sampling rate of 20 kHz.

[4 Marks]

(c) Using the window design technique and the inverse Fourier transform, design a bank of digital finite impulse response (FIR) filters for a telephony speech application to split a total bandwidth of 4 kHz into 4 equal bandwidth sub-bands. Write the impulse response of each sub-band filter. State how this filter can be made causal.

[9 Marks]

Q8

(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) From the DFT equation, using the orthogonality principle, derive the inverse DFT equation.

[5 Marks]

(ii) Describe what is meant by the time and the frequency resolutions in the discrete Fourier transform. Write an expression describing the relation between the time and the frequency resolutions. Hence, explain the Uncertainty Principle.

[3 Marks]

(iii) What is the typical time resolution used for analysis of audio signals and what would be the resulting frequency resolution.

[2 Marks]

(b) A DFT is used as part of a digital signal processing system for the analysis of a signal with significant frequency content of up to 100 kHz. Calculate:

(i) The number of time domain samples required to achieve a frequency resolution of 100 Hz at the minimum sampling rate.

[3 Marks]

(ii) A segment of N samples of this signal is padded with N zeros. Derive the Fourier transform equation for the zero-padded signal. Explain how the frequency resolution of the discrete Fourier transform of a signal changes by zero-padding.

[5 Marks]

(iii) Assuming that a segment of 4000 samples of this signal is zero padded with 4000 extra zeros, calculate the *actual* frequency resolution and the interpolated *apparent* frequency resolution.

[2 Marks]