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UNIVERSITI TUN HUSSEIN ONN MALAYSIA

FINAL EXAMINATION
SEMESTER I
SESSION 2021/2022

COURSE NAME	: ADVANCED DIGITAL SIGNAL PROCESSING
COURSE CODE	: MEE 10303
PROGRAMME	: MEE
EXAMINATION DATE	: JANUARY / FEBRUARY 2022
DURATION	: 3 HOURS
INSTRUCTION	: 1. ANSWER **ALL** QUESTIONS 2. THIS FINAL EXAMINATION IS AN **ONLINE** ASSESSMENT AND CONDUCTED VIA **OPEN BOOK**

THIS QUESTION PAPER CONSISTS OF **SIX (6)** PAGES

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Q1 **(a)** Convolution is a computation that proceeds on a sample-by-sample basis upon the input sequence. Considering the input sequence

$$x[n] = \{x[0], x[1], x[2], x[3], x[4]\}$$

and the impulse response

$$h[n] = \{h[0], h[1], h[2], h[3], h[4]\},$$

design the hardware implementation for the convolution computation in terms of adders, multipliers, delay and memory units.

(7 marks)

(b) Proof that discrete-time sinusoid whose frequencies are separated by an integer multiple of 2π are identical. Utilize $x(n) = cos(\omega n)$ as the example.

(4 marks)

(c) A digital communication link carries binary-coded words representing samples of an input signal

$$x_a(t) = 3cos(600\pi t) + 2cos(1800\pi t)$$

The link is operated at 10,000 bits/s and each input sample is quantized into 1024 different voltage levels. Estimate:

(i) the sampling frequency

(3 marks)

(ii) the folding frequency

(1 mark)

(iii) the Nyquist rate for the signal $x_a(t)$

(2 marks)

(iv) the frequencies in the resulting discrete-time signal $x(n)$

(3 marks)

(v) the resolution Δ

(2 marks)

(d) The first five points of the eight-point DFT of a real valued sequence are $\{0.25, 0.125 - j0.3018, 0, 0.125 - j0.0518, 0\}$. Suggest the remaining **THREE (3)** points.

(3 marks)

Q2 **(a)** Suggest the method to realize a system transfer function $H(z)$ as a practical digital filter

(4 marks)

(b) Convert the analog system with system function

$$H(s) = \frac{s + 0.1}{(s + 0.1)^2 + 9}$$

into a digital IIR filter by means of the impulse invariance method.

(5 marks)

(c) An IIR digital low-pass filter is required to meet the following specifications:

Passband ripple: $\leq$ 0.5 dB
Passband edge: 1.2 kHz
Stopband attenuation: $\geq$ 40 dB
Stopband edge: 2.0 kHz
Sample rate: 8.0 kHz

Estimate the required filter order for

(i) a digital Butterwoth filter
(ii) a digital Chebysev filter
(iii) a digital elliptic filter

(5 marks)

(d) Consider a random noise signal that is generated from an audio set of a car. Suggest the method to construct a probability density function (pdf) from this random signal.

(5 marks)

(e) A random experiment consists of observing the sum of the numbers showing up when two dices are thrown. Estimate:

(i) the sample space S of the experiment

(1 mark)

(ii) the probability of $A = \{sum = 7\}$

(1 mark)

(iii) the probability of $B = \{8 < sum \ll 11\}$

(2 marks)

(iv) the probability of $C = \{sum > 10\}$

(2 marks)

Q3 **(a)** Suppose a research student uses Bartlett's and Welch's method to estimate the power spectrum of his incoming input signal x(n) consisting of N = (First Student id)(Last Student id)00 samples.

(10 marks)

(i) Estimate the minimum FFT block length (M) required to obtain a relative variance of 0.01 in the student spectrum estimatation, for Bartlett method and Welch method (using 50% overlap and hanning window).

(ii) Formulate the frequency resolution (Δf) of the spectrum estimate for Bartlett method and Welch method (using 50% overlap and hanning window).

(b) An APPLE designer has developed a Weiner filter as illustrated in **Figure Q3(b)** for the latest Iphone 15 model. The given input signal and noise signal of the system are as follows:

$$x[n] = \{0.08186, -0.2926, -0.5408, -0.3086\}$$

$$v[n] = \{0.0157, -0.0874, -0.1972, 0.0555\}$$

Assume the FIR filter coefficient $b_0 = 1$ and $b_1 = 0.38$, predict this FIR filter coefficients using the Weiner filter scheme for the Iphone 15.

(15 marks)

Q4 **(a)** In signal processing, Wiener filter is a filter used to produce an estimation of a desired or target random process by linear time-invariant (LTI) filtering of an observed noisy process, by assuming the known stationary signal and noise spectra, with an additive noise. Derive the mean square error output of the system with an aid of a block diagram. Write the matrix representation of the system.

(7 marks)

(b) A signal $x[n] = S[n] + W[n]$, where $S[n]$satisfies the difference equation

$$S[n] = 0.6\, S[n-1] + V[n],$$

where $V[n]$is a white noise sequence with variance $\sigma_V^2 = 0.64$ and $W[n]$is a white noise sequence with variance $\sigma_W^2 = 1$. Design an IIR causal Wiener filter to estimated $S[n]$ if it is assumed that the autocorrelation and power spectral density of a stationary random process $X[n]$ may be represented by an equivalent innovation process $i[n]$ by passing $X[n]$ through a noise whitening filter with system function $\dfrac{1}{G(z)}$, where $G[z]$ is the minimum phase part obtained from the spectral factorization $\Gamma_{xx}(z) = \sigma_i^2 . G(z) . G(z^{-1})$. Given that $\sigma_i^2 = 1.8$ and

$$G(z) = \frac{1 - \frac{1}{3} z^{-1}}{1 - 0.(LastStudentIdNumber) z^{-1}}$$

(18 marks)

- END OF QUESTION

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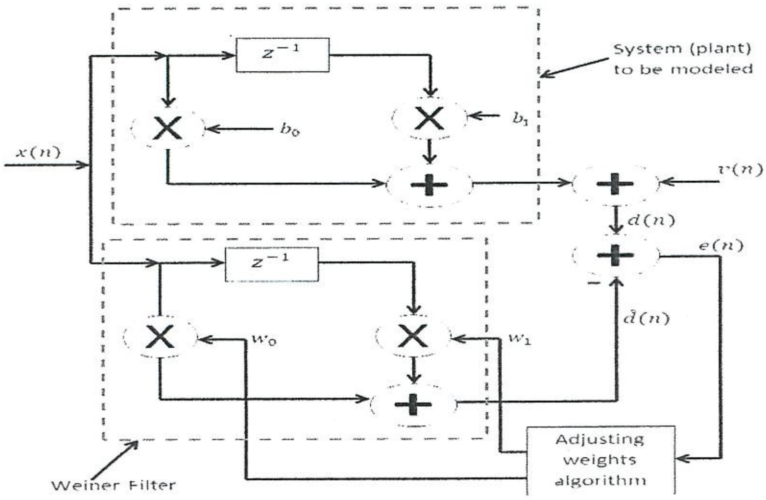

Figure Q3(b): Weiner Filter