Universiti Tun Hussein Onn Malaysia

UNIVERSITI TUN HUSSEIN ONN MALAYSIA

FINAL EXAMINATION
SEMESTER I
SESSION 2022/2023

COURSE NAME	:	DIGITAL SIGNAL PROCESSING
COURSE CODE	:	BEJ30603
PROGRAMME CODE	:	BEJ
EXAMINATION DATE	:	FEBUARY 2023
DURATION	:	3 HOURS
INSTRUCTION	:	1. ANSWER **ALL** QUESTIONS 2. THIS FINAL EXAMINATION IS CONDUCTED VIA **CLOSED BOOK** 3. STUDENTS ARE **PROHIBITED** TO CONSULT THEIR OWN MATERIAL OR ANY EXTERNAL RESOURCES DURING THE EXAMINATION CONDUCTED VIA CLOSED BOOK

THIS QUESTION PAPER CONSISTS OF **FOUR (4)** PAGES

Q1 (a) In digital signal processing technique, an analog signal must not only be sampled in time but also quantized the signal amplitudes to a finite set of values. Based on this quantization technique, define the following term:

(i) Quantization signal to noise ratio
(ii) Dynamic range
(iii) Quantization step size

(5 Marks)

(b) Based on fact that all discrete signals can be separated into even and odd part, discuss on this fact using mathematical approach.

(10 Marks)

(c) A digital signal processing engineer has designed a system as shown in **Figure Q1(c)**. Calculate output signal of the system if input signal of the system is given as:

$$a(n) = \{ \overset{\Downarrow}{1}, -1, 3, 4, -3, 6\}$$

(10 Marks)

Q2 Digital filter is a system used to compress any unwanted component in a signal. In digital signal processing technique, FIR filter can be designed with linear phase, hence no phase distortions occur. Referring to FIR filter design technique,

(a) Design a low pass FIR digital filter using a sampling frequency of 10 Hz with a cut-off frequency of 2 Hz. In this filter design, consider to use Hamming window and set the filter length, $n = 5$.

(10 marks)

(b) Consider a digital signal, $x(n) = \{ \overset{\Downarrow}{M1}, M4, 5, M5, -2\}$ where $M1$, $M4$ and $M5$ are based on your matrix number. Calculate the output signal of the FIR filter system designed in **Q2(a)**.

(15 marks)

NOTE: if your matrix number is CE123456, *then M1=1, M2=2, M3=3, M4=4, M5=5 and M6=6*

Q3 Analysis of a system in frequency domain can be made in z-plane representation. As an electronic engineer, you have been given a system as, $h(n) = \{\overset{\Downarrow}{1}, -1, 0, -1, 2\}$. Analyse the given system using z-transform technique. Show the magnitude and phase response and determine the stability of the system.

(25 marks)

Q4 Implementation of an IIR filter in digital signal processing technique yield a much smaller filter order for a given application. Considering to this advantage,

(a) Design an IIR high pass filter with cut-off frequency at 2 Hz using sampling frequency of 10 Hz. The filter order should be 5 and the window used is Boxcar window.

(10 marks)

(b) Consider a digital signal, $x(n) = \{\overset{\Downarrow}{M1}, M4, 5, M5, -2\}$ where *M1*, *M4* and *M5* are based on your matrix number. Calculate the output signal of the IIR filter system designed in Q4(a).

(15 marks)

NOTE: if your matrix number is CE123456, *then M1=1, M2=2, M3=3, M4=4, M5=5 and M6=6*

- END OF QUESTIONS -

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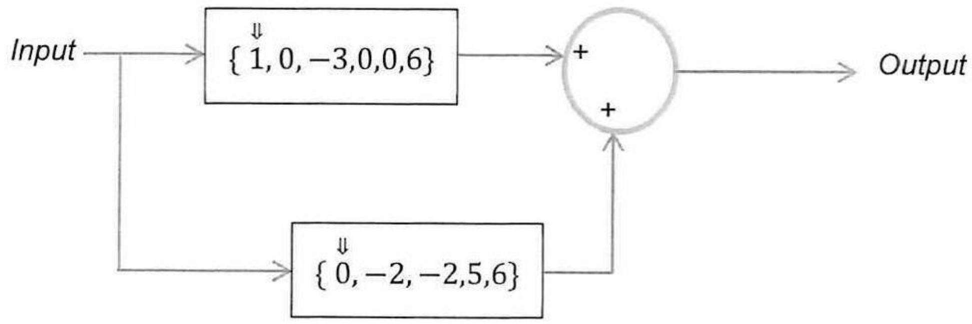

Figure Q1(c)