GLASGOW COLLEGE UESTC

Exam paper

Digital Signal Processing (UESTC4005)

Date: June 25th 2023
Time: 09:30-11:30AM

Attempt all PARTS. Total 100 marks

Use one answer sheet for each of the questions in this exam.
Show all work on the answer sheet.

Make sure that your University of Glasgow and UESTC Student Identification Numbers are on all answer sheets.

An electronic calculator may be used provided that it does not allow text storage or display, or graphical display.

All graphs should be clearly labelled and sufficiently large so that all elements are easy to read.

The numbers in square brackets in the right-hand margin indicate the marks allotted to the part of the question against which the mark is shown. These marks are for guidance only.

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Q1 A stable linear time invariant system has the transfer function

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

(a) Sketch its zero-pole plot, Is the system an FIR or IIR? [5]

(b) Find the frequency response $H(e^{j\omega})$ of this system. [5]

(c) Calculate the value of the frequency response $H(e^{j\omega})$ at $\omega = 0$, $\omega = 0.5\pi$ and $\omega=\pi$. [6]

(d) Is the system low-pass, high-pass or band-pass? Please give detailed reasons. [3]

(e) Find the output $y[n]$ produced by the input $x(n) = \cos(0.2n\pi)$. [3]

(f) To get zero output for an input analogue signal $\cos(2\pi \times 2000t)$ using this discrete system, choose the sampling rate and explain the detailed reason. [3]

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Q2 Consider the following 8-point signals, $0 \leq n \leq 7$.

$x_1[n] = \{1, 1, 1, 0, 0, 0, 1, 1\}$,
$x_2[n] = \{1, 1, 0, 0, 0, 0, -1, -1\}$,
$x_3[n] = \{0, 1, 1, 0, 0, 0, -1, -1\}$
$x_4[n] = \{0, 1, 1, 0, 0, 0, 1, 1\}$

(a) Which of these signals have a real-valued 8-point DFT? Please give detailed reasons. [5]

(b) Which of these signals have an imaginary-valued 8-point DFT? Please give detailed reasons. [5]

(c) Which of these signals have a minimum 8-point DFT square summation? Please give detailed reasons. [5]

(d) Which of these signals have a maximum 8-point DFT square summation? Please give detailed reasons. [5]

(e) Draw a diagram showing how these signals' DFTs may be calculated Using a radix-2 Decimation-in-Time Fast Fourier Transformation algorithm (DIT FFT) [3]

(f) Sometimes a sequence can be transmitted in its DFT form, to reduce the transmission rate (transmitted bits per second). Design a system which can transmit $x_1[n]$, $x_2[n]$, $x_3[n]$, $x_4[n]$. Explain how the system reduce the transmission rate. [2]

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Q3 Design a simple order-2 linear phase causal real discrete-time FIR LTI system with the following properties:

(a) The system should kill the signals $(-1)^n$ and the system should have low-pass unity gain. That is, $H(e^{j0}) = 1$. [5]

(b) Find the difference equation to implement the system. [5]

(c) Sketch the impulse response and zero-pole plot of the system. [5]

(d) Roughly sketch the frequency response magnitude $|H(e^{j\omega})|$. Clearly show the nulls of the frequency response. [5]

(e) Input a modulated signal $\cos(0.02\pi n) \times \cos(0.8\pi n)$ to the system you designed. What is the output signal? Explain what delay effect will the system lead to. [5]

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Q4 The spectrum of an analog signal is contained between DC and 10 Hz. The signal is corrupted by additive high-frequency noise band-limited between 20Hz and 25Hz. The noisy analog signal is sampled at 50 samples/second. Design an IIR discrete- time filter, operating at 50 samples/second, to attenuate the noise to 95% and keep the signal with distortion no more than 2%.

(a) Determine **the peak passband ripple** α_p and **the minimum stopband attenuation** α_s of this filter. [5]

(b) If an IIR LPF is employed and designed by **the bilinear transformation method**, determine the passband and stopband edge frequencies of **the analog prototype filter**. [6]

(c) If the Butterworth approximation is used to meet the analog filter specifications, determine the order N and the 3-dB cutoff frequency Ω_c of the Butterworth filter. [5]

(d) Based on **the bilinear transformation method**, design the discrete-time LPF and give out it's transfer function. [6]

(e) Design another scheme to satisfy same filter requirements but by FIR form. Compare the resources consumed in between two schemes. [3]

(Some normalized Butterworth analog transfer functions are given as:
2-ord: $\frac{1}{s^2+1.41421s+1}$;*3-ord:*$\frac{1}{s^3+2s^2+2s+1}$; 4-ord:$\frac{1}{s^4+2.6131s^3++3.41412s^2+2.6131s}$;
5-ord $\frac{1}{s^5+3.236\quad{}^4+5.2361s^3++5.2361s^2+3.2361s+1}$)

End of question paper