Kaferelsheikh University

Department of Electrical Engineering

Subject: Digital signal processing

Academic Number: ECE3211

Full Mark: 100

Date: 14/6/2021

Faculty of Engineering

Year: 3rd Electronics and

Electrical Communications

Final Exam: 2 pages

Time allowed: 3 h

This Exam measures the LOS related to competancies [A.1, A.2, B.2, B.3 and C.3]

Answer the following questions:

[1] Question One: (28 Mark)

- a- Assuming that an analog signal is given by $x(t) = 5\cos(4000\pi t) + 3\cos(6000\pi t)$ for $t \ge 0$ and it is sampled at the rate of 8,000 Hz,
 - i. Sketch the spectrum of the sampled signal up to 20 kHz.
 - ii. Sketch the recovered analog signal spectrum if an ideal low pass filter with a cutoff frequency of 4 kHz is used to filter the sampled signal to recover the original signal

[8marks]

.- Determine, analytically, whether these systems are linear, time invariant or causal

i-y(n)=
$$x(n) + 2x(n-2) - 3x(n-3)x(2n)$$
 ii- $y(n)=x(n) + nx(n+1)$

ii-
$$y(n) = x(n) + n x(n + 1)$$

[8marks]

c- A causal system is described by the difference equation

y(n) + 0.25y(n-1) - 0.125y(n-2) = x(n) + x(n-1). Draw a block diagram for this system , and write the matlab script to

i- Plot the pole-zero pattern

ii-compute and plot the impulse response of the system over $0 \le n \le 100$.

iii- Show how you can study the stability of this system from this impulse response

[12marks]

[2] Question Two: (27 Mark)

a- for a digital system having $H(z) = \frac{z+1}{z^2-0.9z+0.81}$, find

i- its difference equation representation ii- its impulse response representation.

Write the matlab script which can be used for calculations in part (ii)

[10 marks]

b- Using DTFT, determine the frequency response of a system characterized by

 $h(n) = (0.9)^n u(n)$. Plot the magnitude and the phase responses.

[9 marks]

Explain the difference between FFT and DFT and then compare between their ccomputational speeds

[8 marks]

[3] Question three: (25 Mark)

- **a-** An analog signal $xc(t) = 5 \cos(400\pi t) + 10 \sin(500\pi t)$ is to be processed by a digital signal processor in which the sampling frequency is 1 kHz.
- **i-** <u>Design</u> a minimum order FIR filter using one of the fixed windows that will pass the first component of xc(t) with attenuation of less than 1 dB but will attenuate the second component to at least 50 dB.
 - ii-Write the MATLAB script to ensure that the design achieves the required specifications

 [7 marks]
- **b** Design a digital band-pass IIR filter using pole-zero placement method with center frequency, $\pi/2$ radians and a bandwidth of $\pi/8$ radians.

[4] Question four: (20 Mark)

- **a-** write down the steps of LMS technique for adaptive filter coefficient determination [7] marks] **b-** Use Matlab functions and LMS algorithm to determine the coefficients of FIR filter which can be used to Identify the system described by the following difference equation: d(n)=d(n-1)-0.9d(n-2)+x(n)-2x(n-2)[7] marks]
- c- Explain, with aid of diagrams, the operation of adaptive channel equalizer. [6 marks]

Best Wishes

Committee of Correctors and Testers

Dr. Shamia Ghamry

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