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10EE64

Sixth Semester B.E. Degree Examination, Jan./Feb. 2023
Digital Signal Processing

Time: 3 hrs. Max. Marks:100

Note: ***Answer FIVE full questions, selecting atleast TWO questions from each part.***

PART – A

1 a. State and prove the following properties of DFT.
i) Frequency shift
ii) Time shift. **(08 Marks)**

b. Evaluate the circular convolution of the two sequences.
$x_1(n) = (1, 2, 3, 1), x_2(n) = (4, 3, 2, 2)$. **(04 Marks)**

c. Find the 4-point DFT of the sequence,
$$x(n) = 6 + \sin\frac{2\pi n}{4},\ 0 \le n \le 3.$$ **(08 Marks)**

2 a. Evaluate the following function without computing the,
$$\text{DFT} : \sum_{k=0}^{11} e^{\frac{-j4\pi k}{6}} \cdot X(k) \text{ for a given 12 point sequence}$$
$x(n) = [8, 4, 7, -1, 2, 0, -2, -4, -5, 1, 4, 3]$. **(06 Marks)**

b. An FIR digital filter has an unit impulse response $h(n) = \{2, 2, 1\}$. Determine the output sequence y(n) in response to an input sequence $x(n) = \{3, 0, -2, 0, 2, 1, 0, -2, -1, 0\}$. Use overlap save fast convolution method, N = 5. Block sequence length. **(12 Marks)**

c. Explain the difference between linear convolution and circular convolution. **(02 Marks)**

3 a. Explain the advantages of FFT over DFT by direct method. **(05 Marks)**

b. What is in place computation? What is the total number of complex additions and multiplications required for N = 512 point, if DFT is computed directly and if FFT is used. **(05 Marks)**

c. Develop DIT – FFT algorithm for composite value of N = 6. Draw the corresponding single flow graph. **(10 Marks)**

4 a. Find the DFT of $x(n) = [1, 2, 3, 4, 4, 3, 2, 1]$ using the DIFFFT algorithm. **(12 Marks)**

b. Find the IDFT of $X(K) = \{0, 2 + 2j, -4j, 2 - 2j, 0, 2 + 2j, j4, 2 - 2j\}$ using Radix – 2 DIT – FFT algorithm. **(08 Marks)**

PART – B

5 a. Design an analog Butterworth that has a –2dB or better cut off frequency of 20rad/sec and atleast 10dB of attenuation at 30 rad/sec. **(10 Marks)**

b. Design a Chebyshev along low-pass filter that has –3dB cutoff frequency of 100rad/sec, and a stop band attenuation of 25dB or greater for all radian frequency past 250 rad/sec. **(10 Marks)**

6 a. Design a digital low-pass filter using Butterworth approaximation to meet the following specifications, passband edge = 500Hz, Pass band gain = 3.01dB stopband edge = 750Hz, stop band attenuation = 15dB. Assume sampling frequency of 2 KHz. Use bilinear transformation. **(10 Marks)**

b. Explain the impulse invariance method of transforming an analog filter into an equivalent digital filter. **(05 Marks)**

c. Explain the difference between the digital and analog filters. **(05 Marks)**

7 a. A filter is to be designed with the following desired frequency response :

$$H_d(\omega) = \begin{cases} 0, & \frac{-\pi}{4} < \omega < \frac{\pi}{4} \\ e^{-j2\omega}, & \frac{\pi}{4} < |\omega| < \pi \end{cases}$$

Find the frequency response of the FIR filter designed using a rectangular window defined below :

$$\omega_R(n) = \begin{cases} 1, & 0 \le n \le 4 \\ 0, & \text{otherwise} \end{cases}.$$ **(12 Marks)**

b. The frequency response of an FIR filter is given by,

$$H(\omega) = e^{-j3\omega}(1 + 1.8\cos 3\omega + 1.2\cos 2\omega + 0.5\cos\omega)$$

Determine the coefficients of the impulse response h(n) of the FIR filter. **(08 Marks)**

8 a. Obtain a parallel and cascade realization for the system described by,

$$H(z) = \frac{(1+z^{-1})(1+2z^{-1})}{(1+\frac{1}{2}z^{-1})(1-\frac{1}{4}Z^{-1})(1-\frac{1}{8}Z^{-1})}.$$ **(10 Marks)**

b. Sketch the direct form I and II realizations for the system.

$$H(z) = \frac{2z^2 + z - 2}{z^{-2} - 2}.$$ **(05 Marks)**

c. Explain linear phase realization. **(05 Marks)**

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