R13

Code No: 126EK

JAWAHARLAL NEHRU TECHNOLOGICAL UNIVERSITY HYDERABAD

B. Tech III Year II Semester Examinations, May - 2017

DIGITAL SIGNAL PROCESSING

(Common to ECE, EIE)

Time: 3 hours **Max. Marks: 75**

Note: This question paper contains two parts A and B.
Part A is compulsory which carries 25 marks. Answer all questions in Part A. Part B consists of 5 Units. Answer any one full question from each unit. Each question carries 10 marks and may have a, b, c as sub questions.

PART - A

(25 Marks)

1.a) What is an LTI system? [2]
b) Define the frequency response of a discrete-time system. [3]
c) Define discrete Fourier series. [2]
d) Obtain the circular convolution of the sequence x(n)={1,2,1}; h(n)={1,-2,2}. [3]
e) What is meant by bilinear transformation? [2]
f) Prove that physically realizable and stable IIR filters cannot have linear phase. [3]
g) What are the disadvantages of Fourier Series Method? [2]
h) What is the desirable characteristics of the Window? [3]
i) What is the need for anti-imaging filter after up sampling a signal? [2]
j) What are the effects of Dead band? [3]

PART - B

(50 Marks)

2.a) Determine whether each of the following systems defined below is (i) Causal (ii) Linear (iii) Dynamic (iv) Time invariant (v) Stable.

(I) $y(n) = \sum_{k=n-3}^{n} e^{x(k)}$ (II) $y(n) = x(-n-2)$

b) For each impulse response listed below, determine whether the corresponding system is (i) causal (ii) stable. [5+5]

(I) $h(n) = 2^n u(-n)$ (II) $h(n) = e^{2n} u(n-1)$

OR

3.a) If x(n) is a causal sequence, find the z- transform of the following sequences.
(i) $x(n) = nu(n)$ (b) $x(n) = nu(n-1)$

b) Find the response of $y(n) + y(n+1) - 2y(n-2) = u(n-1) + 2u(n-2)$ due to $y(-1) = 0.5; y(-2) = 0.25$. [5+5]

4.a) Compute the DFT of the square-wave sequence

$$x(n) = \begin{cases} 1 & 0 \le n \le \frac{N}{2} - 1 \\ -1 & \frac{N}{2} \le n \le N-1 \end{cases}$$ Where N is even.

b) Find 4-point DFT of the following sequence $x(n) = \left(\frac{1}{4}\right)^n$. [5+5]

OR

5.a) An 8-point sequence is given by $x(n) = \{2,2,2,2,1,1,1,1\}$. Compute 8-point DFT of x(n) by radix-2 DIT-FFT.

b) Compute the DFT of the sequence $x(n) = \cos\frac{n\pi}{2}$, where N=4 using DIF FFT algorithm. [5+5]

6. Design a chebyshev filter for the following specifications using (a) bilinear transformation. (b) Impulse Invariance method. $z = e^{sT}$ [10]

$$0.8 \le \left|H\left(e^{jw}\right)\right| \le 1 \quad 0 \le w \le 0.2\pi$$
$$\left|H\left(e^{jw}\right)\right| \le 0.2 \quad 0.6\pi \le w \le \pi$$

OR

7.a) Design a lowpass filter that will operate on the sampled analog data such that the cutoff frequency is 200Hz and at 400Hz, the attenuation is atleast 20dB with a monotonic shape past 200Hz. Take T = $\frac{1}{2000}$ secs and use normalized lowpass filter.

b) A third-order Butterworth low pass filter has the transfer function:
$H(s) = \frac{1}{(s+1)(s^2+s+1)}$. Design H(z) using Impulse Invariance method. [5+5]

8.a) Design an ideal Hilbert transformer having frequency response

$$H\left(e^{jw}\right) = \begin{cases} j & for \quad -\pi \le w \le 0 \\ -j & for \quad 0 \le w \le \pi \end{cases}$$; using rectangular window.

b) For the desired frequency response given by

$$H_d(w) = \begin{cases} e^{-j3w}, & |w| < \frac{3\pi}{4} \\ 0, & \frac{3\pi}{4} < |w| < \pi \end{cases}$$

Find H(ω) for N=7 using Hamming window for truncating $h_d(n)$. [5+5]

OR

9.a) Design an FIR digital filter H(z) that when used in the prefilter A/D -H(z) – D/A structure will satisfy the following equivalent analog specifications.
i) Low pass filter with -1dB cutoff at 100π rad/sec.
ii) Stop band attenuation of 35dB or greater at 1000π rad/sec.
iii) Sampling rate of 2000 samples/sec.
iv) The phase must be linear.

b) Draw the magnitude response, $|W(\omega)|$ versus ω, for nine-term windows of the following
i) Rectangular window ii) Hanning window. [6+4]

10.a) Explain the application of sampling rate conversion in subband coding.
b) Discuss in detail the down sampling with a neat diagram. [5+5]

OR

11.a) Explain the multistage implementation of sampling rate conversion.
b) Explain the finite word length effects in digital filter. [5+5]

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