

(Following Paper ID and Roll No. to be filled in your Answer Book)

PAPER ID : 131602

Roll No.

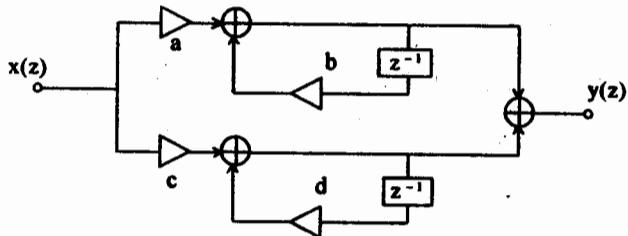
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B.Tech.**(SEM. VI) THEORY EXAMINATION 2013-14****DIGITAL SIGNAL PROCESSING***Time : 3 Hours**Total Marks : 100***Note :- Attempt all questions. All questions carry equal marks.**1. Attempt any two parts of the following : **(10×2=20)**

(a) The transfer function of a causal IIR filter is given by

$$H(z) = \frac{5z(3z - 2)}{(z + 0.5)(2z - 1)}$$

Determine the values of multiplier coefficients of the realization structure shown in Figure (1):



(b) Sketch the Ladder structure for the system :

$$H(z) = \frac{(1 - 0.6z^{-1} + 1.2z^{-2})}{(1 + 0.15z^{-1} - 0.64z^{-2})}$$

- (c) Obtain Direct form- I and Direct form - II and Cascade structure for following system :

$$y(n) = [-0.1 y(n-1) - 0.72 y(n-2) + 0.7 x(n) - 0.2 x(n-2)]$$

2. Attempt any two parts of the following : (10×2=20)

- (a) Design digital Butterworth filter to meet the constraints :

$$0.9 \leq |H(e^{j\omega})| \leq 1 ; 0 \leq \omega \leq 0.25 \pi$$

$$|H(e^{j\omega})| \leq 0.2 ; 0.6 \pi \leq \omega \leq \pi$$

using (i) Bilinear Transformation Technique and

(ii) Impulse Invariance Transformation Technique.

- (b) Convert analog filter to digital filter whose system function

$$\text{is } H(s) = \frac{36}{(s + 0.1)^2 + 36}$$

The digital filter should have a resonant frequency of $\omega_r = 0.2 \pi$. Use Bilinear Transformation Technique.

- (c) Why is Frequency transformation needed ? What are the different types of frequency transformations ?

3. Attempt any two parts of the following : (10×2=20)

- (a) The desired response of a low pass filter is

$$H_d(e^{j\omega}) = \begin{cases} e^{-j3\omega} ; & -\frac{3\pi}{4} \leq \omega \leq 3\pi/4 \\ 0 ; & \frac{3\pi}{4} < |\omega| \leq \pi \end{cases}$$

Determine $H(e^{j\omega})$ for $M = 7$ using Hamming Window.

- (b) What is a Kaiser Window ? In what way it is superior to another window function ? Explain the procedure for designing a FIR filter using Kaiser Window.

- (c) Design a FIR digital filter to approximate an ideal LPF with pass band gain of unity, cut off frequency of 850 Hz and working at a sampling frequency of $F_s = 5000$ Hz. The length of impulse response should be 5. Use a rectangular Window.

4. Attempt any two parts of the following : (10×2=20)

- (a) Compute the DFT Coefficients of a finite duration sequence (0, 1, 2, 3, 0, 0, 0, 0).

- (b) An Input Sequence $x(n) = \{2, 1, 0, 1, 2\}$ is applied to a DSP system having an impulse sequence $h(n) = \{5, 3, 2, 1\}$. Determine the output sequence by (i) Linear convolution and (ii) Verify the same through circular convolution.

- (c) Explain the difference between the DTFT and DFT and write the properties of DFT.

5. Attempt any two parts of the following : (10×2=20)

- (a) Develop a radix-4 DIT FFT algorithm for evaluating the DFT for $N = 16$ and hence determine the 16 point DFT of the sequence

$$x(n) = \{0, 1, 0, 1, 0, 1, 0, 1, 0, 1, 0, 1, 0, 1, 0, 1\}$$

- (b) Given $x(n) = (n + 1)$ and $N = 8$. Find $X(K)$ using DIF FFT algorithm.

- (c) Develop a DIT FFT algorithm for $N = 8$ using a 4 point DFT and a 2 point DFT. Compare the number of multiplications with the algorithm using only 2 point DFTs.