Printed Pages—3 **EEC602**

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

PAPER ID : 131602 **Roll No.**

B.Tech.

(SEM. VI) THEORY EXAMINATION 2013-14

DIGITAL SIGNAL PROCESSING

Time : 3 Hours *Total Marks : 100*

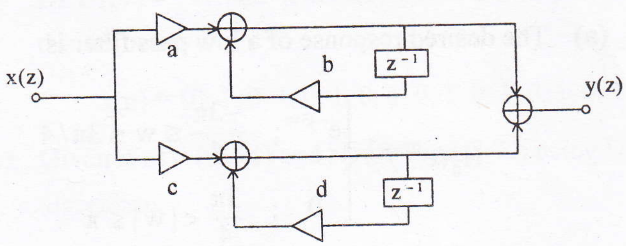
Note :– Attempt **all** questions. All questions carry equal marks.

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 :

$$0.9 \le |H(e^{jw})| \le 1 \;;\; 0 \le w \le 0.25\,\pi$$

$$|H(e^{jw})| \le 0.2 \;;\; 0.6\,\pi \le w \le \pi$$

using (i) Bilinear Transformation Technique and

(ii) Impulse Invariance Transformation Technique.

(b) Convert analog filter to digital filter whose system function is $H(s) = \dfrac{36}{\left[(s+0.1)^2 + 36\right]}$

The digital filter should have a resonant frequency of $w_r = 0.2\,\pi$. 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

$$H_d(e^{jw}) = \begin{cases} e^{-j3w}; & \dfrac{-3\pi}{4} \le w \le 3\pi/4 \\ 0 \;; & \dfrac{3\pi}{4} < |w| \le \pi \end{cases}$$

Determine $H(e^{jw})$ 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 Fs = 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.