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

(a) Draw the flow graph of an 8 point DIF FFT algorithm and explain.

(b) Define Goertzel algorithm.

(c) Explain Fourier analysis of continuous time signals using DFT.

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

PAPER ID : 0300 **Roll No.** | | | | | | | | | | |

B.Tech.

(SEM. VII) ODD SEMESTER THEORY EXAMINATION 2010-11

FUNDAMENTAL OF DIGITAL SIGNAL PROCESSING

Time : 3 Hours *Total Marks : 100*

Note : Attempt all questions.

1. Attempt any four parts of the following : **(5×4=20)**

(a) Determine the response of the following system to the I/P signal :

$$x(n) = \begin{cases} |n| & -3 \le n \le 3 \\ 0 & \text{otherwise} \end{cases}$$

(i) $y(n) = x(n-1)$

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

(b) For each of the following impulse response of LTI systems indicate whether or not the system is causal :

(i) $h[n] = u(n+2) - u(n-2)$

(ii) $(1/2)^n u(n-1)$

(c) For the following impulse response of LTI system indicate whether or not the system is stable :

$$h[n] = \sin(n\pi/3)\, u(n)$$

(d) The given signal is periodic or not, if periodic calculate period :

$$x[n] = e^{j(2\pi n/5)}$$

(e) Find the DFT of the sequence :

$$x(n) = 1 \text{ for } 0 \le n \le 2$$
$$= 0 \text{ otherwise}$$

(f) State and explain "time reversal of a sequence" property of DFT.

2. Attempt any four of the following : **(5×4=20)**

(a) State sampling theorem. Draw the spectrum of a sampled signal and explain aliasing.

(b) Define all pass systems and minimum phase systems.

(c) Explain the need for multirate signal processing.

(d) Explain how sampling rate can be increased by an Integer factor.

(e) Explain the use of oversampling to simplify the process of analog-to-digital conversion in brief.

(f) With the help of block diagram explain Discrete time processing of continuous time signals.

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

(a) Obtain the cascaded realization for the following systems :

(i) $$H(z) = \frac{(1 + \tfrac{3}{2} z^{-1} + \tfrac{1}{2} z^{-2})\ (1 - \tfrac{3}{2} z^{-1} + z^{-2})}{(1 + z^{-1} + \tfrac{1}{4} z^{-2})\ (1 + \tfrac{1}{4} z^{-1} + \tfrac{1}{2} z^{-2})}$$

(ii) $$H(z) = \frac{(1 - \tfrac{1}{2} z^{-1})\ (1 - \tfrac{1}{2} z^{-1} + \tfrac{1}{4} z^{-2})}{(1 + \tfrac{1}{4} z^{-1})\ (1 + z^{-1} + \tfrac{1}{2} z^{-2})\ (1 - \tfrac{1}{4} z^{-1} + \tfrac{1}{2} z^{-2})}$$

(b) Develop Cascade and Parallel realisation structures for :

$$H(z) = \frac{z/6 + 5/24 + 5/24 z^{-1} + 1/24 z^{-2}}{1 - \tfrac{1}{2} z^{-1} + \tfrac{1}{4} z^{-2}}$$

(c) What is the effect of roundoff noise in digital filters ? Analyse the direct form IIR structure.

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

(a) Explain the procedure for designing an FIR filter using Kaiser Window.

(b) Discuss the Bilinear transformation design techniques for IIR filters.

(c) A filter is to be designed with the following desired frequency response :

$$Hd(e^{jw}) = \begin{cases} 0 & -\pi/4 \le w \le \pi/4 \\ e^{-j2w} & \pi/4 < |w| \le \pi \end{cases}$$

Determine the filter coefficients hd(n) if the window function is defined as :

$$w(n) = \begin{cases} 1 & 0 \le n \le 4 \\ 0 & \text{otherwise} \end{cases}$$

Also determine the freq response $H(e^{jw})$ of the designed filter.