Code No: **R32043** **R10** **Set No. 1**

III B.Tech II Semester Supplementary Examinations, April - 2017

DIGITAL SIGNAL PROCESSING

(Common to, Electronics and Communications Engineering**,**
Electronics and Computer Engineering)

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

Answer any FIVE Questions
All Questions carry equal marks
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1 a) For each of the following impulse responses given below, determine if the corresponding system is causal and stable with appropriate reasons [7M]

i) $h(n)=\sin\left(\frac{n\pi}{2}\right)$ ii) $h(n)=\rho^{2n}u(n-1)$

b) Determine the impulse response of the following causal systems [8M]

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

ii) $y(n)-\frac{3}{4}y(n-1)+\frac{1}{8}y(n-2)=x(n)$

2 a) Find the Z transforms of $x(n)=\left(\frac{1}{4}\right)^n u(n)+\left(\frac{1}{5}\right)^n u(-n-1)$ [7M]

b) Let x[n] be a discrete periodic signal with period N whose Fourier series coefficients are k a with period N. Determine the Fourier series coefficients of the signal $y(n)=x(n)-x(n-1)$ [8M]

3 a) Find the 8 point DFT of the following sequence using Decimation In Time(DIT) FFT algorithm $x(n)=\cos(2\pi n)$ [7M]

b) Find the 10-point inverse DFT of $X(k)=\begin{cases} 3 & k=0 \\ 2 & k=3,7 \\ 1 & else \end{cases}$ [8M]

4 a) Realize the following filter function using the direct form-I and II realizations [7M]

$y(n)-\frac{2}{5}y(n-1)+\frac{3}{7}y(n-2)=2x(n)+\frac{2}{3}x(n-1)$

b) Explain about lattice structure with appropriate equations and diagrams. Determine the FIR filter coefficients for the direct form structure having three stage latter filter coefficients given by $K_1=\frac{1}{4}$ $,K_2=\frac{1}{4},K_3=\frac{1}{3}$ [8M]

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5 a) Convert the following analog filter with system equation $H(s)=\frac{s+0.1}{(s+0.1)^2+9}$ into a digital IIR filter using impulse invariance method. The resultant digital filter should have a resonant frequency of $\omega_r = \frac{\pi}{4}$ [8M]

b) Using bilinear transformation, design a digital Butterworth filter with the following specifications i) sampling frequency F_s = 8KHz, [7M]
ii) $\alpha_p = 2dB$ in the passband $800\,Hz \le f \le 1000\,Hz$
iii) $\alpha_s = 20dB$ in the stopband $0 \le f \le 400\,Hz$ and $2000\,Hz \le f \le \infty$

6 a) Design a linear phase highpass filter using the Hamming window for the following desired frequency response [8M]

$$H_d(w) = \begin{cases} e^{-j3\omega} & for\ \frac{\pi}{6} \le |\omega| \le \pi \\ 0 & |\omega| < \frac{\pi}{6} \end{cases} \quad \text{and} \quad \omega(n) = 0.54 - 0.46\cos\left(\frac{2\pi n}{N-1}\right)$$

where N is the length of the Hamming window.

b) What is a Kaiser window? Explain the design of a FIR filter using Kaiser window. [7M]

7 a) What is multirate signal processing? Explain different applications of multirate signal processing. [8M]

b) Design a 24kHz to 16kHz sample rate converter and show how this converter can be efficiently realised. [7M]

8 a) Compare a general purpose processor and digital signal processor. [8M]

b) Explain the key features of a digital signal processor with neat diagrams. [7M]

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