7E4045

B. Tech. VII Semester (Main/Back) Examination 2014

ELECTRONICS & COMMUNICATION # 7EC2 Common with 7EI2

DIGITAL SIGNAL PROCESSING

Time : 3 Hours | Min. Passing Marks : 24 | Maximum Marks : 80

Instruction to Candidates :

Attempt any five questions, selecting one question from each unit. All questions carry equal marks. (Schematic diagrams must be shown wherever necessary. Any data you feel missing suitably be assumed and stated clearly. Units of quantities used/calculated must be stated clearly.)

Unit-I

1. Determine the Nyquist rate & Nyquist interval for the following signal

$$x(t) = \frac{1}{\pi t}\sin(500\,\pi t)$$ [4]

2. An analog system is defined as $x(t) = 6\cos 2000\pi t + 5\sin 6\times10^3\pi t + 10\cos 12\times10^3\pi t$

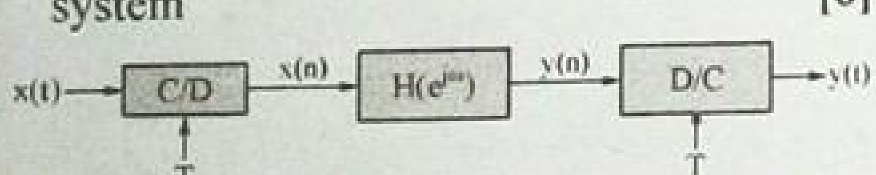
Determine the following

(a) What is the Nyquist rate of this signal?

(b) If the signal is sampled using a sampling rate of $F_S = 5000$ samples, what is the discrete time signal obtained after sampling.

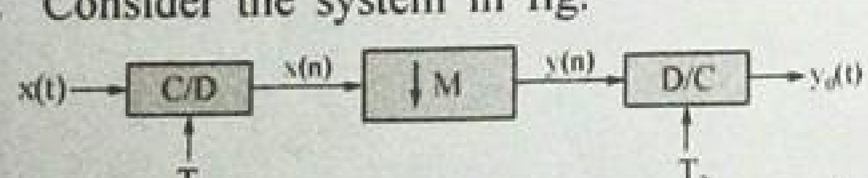
(c) What is the analog signal y(t) that can be reconstructed using ideal interpolation. [6]

3. Consider the system in figure below for implementing a continuous time system in terms of a discrete time system. Assume that the input to the C/D converter is band limited to $W_s = W_0/2$ and that the unit sample response of the discrete time system is $h(n) = \delta(n) - 0.9\delta(n-1)$. Find the over all frequency response of the system [6]



OR

1. Consider the system in fig.



Assume that the input is band limited, $X_c(jw) = 0$ for $|w| > 2\pi$. 1000

(a) What constraint must be placed on M, T_1 & T_2 in order for $y_a(t)$ to be equal to $X_a(t)$.

(b) If $f_1 = f_2 = 20$ kHz & M = 4, find the expression for $y_a(t)$ in terms of $x_a(t)$. [16]

Unit-II

1. A DSP system is described by the LCCD equation $y(n) = 0.2x(n) - 0.5x(n-1) + 0.4\,x(n-2)$ given the digital input sequence. $x(n) = \{-1, 1, 0, -1\}$ is applied to this system. Determine the digital output sequence. [4]

2. Determine the response of the LTI system with impulse response $h(n) = \left(\frac{1}{2}\right)^n 4(n)$ to the input signal $x(n) = 10 - 5\sin\left(\frac{\pi}{2}\right)n + 20\cos(\pi)n$. [12]

OR

1. Consider a system consisting of cascade of two LTI system with frequency responses

$$H_1(w) = \frac{2 - e^{-jw}}{1 + \frac{1}{2}e^{-jw}}$$

$$H_2(w) = \frac{1}{1 - \frac{1}{2}e^{-jw} + \frac{1}{2}e^{-2jw}}$$

(a) Determine the overall frequency response of the system.

(b) Find the difference equation describing the overall system. [8]

2. Decompose the given system function H(Z) in to the minimum phase system and all pass system $H(z) = \dfrac{1 + 3z^{-1}}{1 + \frac{1}{2}Z^{-1}}$ [4]

3. Explain linear phase system. [4]

Unit-III

1. Sketch the Direct form, cascade and parallel form network structure (SFG) for the system described by the LCCD equation as

$$Y(n) = \frac{3}{4}y(n-1) - \frac{1}{8}y(n-2) + x(n) + \frac{1}{3}x(n-1)$$ [16]

OR

1. Given a 3 stage lattice filter with reflection coefficients $K_1 = 1/4$, $K_2 = 1/4$ and $K_3 = 1/3$. Determine the FIR filter coefficients for the direct form structure. [8]

2. Determine FIR linear phase and cascade realization of the system functions which is expressed as $H(z) = \left(1 + \frac{1}{2}z^{-1} + z^{-2}\right)\left(1 + \frac{1}{4}z^{-1} + z^{-2}\right)$ [8]

Unit-IV

1. The system function of an analog filter is given as

$$H_0(S) = \frac{S + 0.1}{(S + 0.1)^2 + 16}$$

Obtain the system function of the digital filter using bilinear transformation which is resonant at $W_r = \frac{\pi}{2}$. [8]

2. Use bilinear transformation to convert low pass filter, $H(s) = \dfrac{1}{s^2 + \sqrt{2s} + 1}$ into a high pass filter with pass band edge at 100 Hz and Fs = 1 KHz. [8]

OR

1. Design a Chebyshev analog filter with maximum passband attenuation of 2.5 dB at $\Omega_p = 20$ rad/sec and stop band attenuation of 30 dB at $\Omega_s = 50$ rad/sec. [8]

2. A digital filter has the following frequency specifications

passband frequency = $W_p = 0.2\pi$

Stop band frequency = $W_s = 0.3\pi$

What are the corresponding specification for pass band and stop band frequencies in analog domain if

(i) Impulse variance technique is used for designing

(ii) Bilinear transformation is used for designing [8]

Unit-V

1. Find the linear convolution using circular convolution of the following sequence $x(n) = \{1, 2, 3, 4\}$, $u(n) = \{1, 2, 3\}$ [8]

2. Using Matrix method determine the 8-point DFT of sequence $X(n) = \{0,0,1,1,1,0,0,0\}$ [8]

OR

1. Determine the DFT of the following sequence using DIF-FFT algorithm $X(n) = \{1,1,1,0,0,1,1,1\}$. [8]

2. Explain technically Radix-2 DIT-FFT algorithm. [8]