USN | | | | | | | | | | | **14ELD21**

Second Semester M.Tech. Degree Examination, June/July 2015
Modern DSP

Time: 3 hrs. Max. Marks: 100

Note: ***Answer any FIVE full questions.***

1 a. Write the properties of discrete time sinusoidal signal and investigate. **(04 Marks)**

b. Consider the simple signal processing system shown in Fig.Q.1(b) below. The sampling periods of the A/D and D/A converters are T = 5ms and T′ = 1ms respectively. Determine the output $y_a(t)$ of the system if the input is, $x_a(t) = 3\cos 100\pi t + 2\sin 250\pi t$ [t in seconds]. The post filter removes any frequency component above Fs/2. **(06 Marks)**

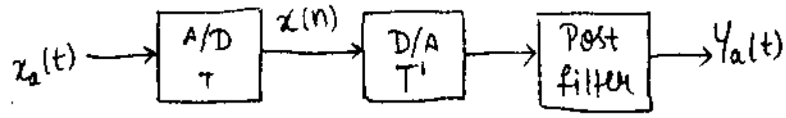

Fig.Q.1(b)

c. Derive the expression for signal to quantization noise ratio of an analog sinusoidal signal $x_a(t) = A\cos\Omega_0 t$. **(05 Marks)**

d. The discrete time signal $x(n) = 6.35\cos\left[\frac{\pi}{10}\right]n$ is quantized with a resolution $\Delta = 0.1$ and $\Delta = 0.02$. How many bits are required in the A/D converter in each case? **(05 Marks)**

2 a. Compute convolution using DFT and IDFT, the two sequences are given by

$h(n) = \{\underset{\uparrow}{1}, 2, 3\}$

$x(n) = \{\underset{\uparrow}{1}, 2, 2, 1\}$ for N = 8. **(08 Marks)**

b. State and prove the following properties of DFT:
i) Circular time reversal of a sequence.
ii) Multiplication of two DFT's.
iii) Circular time shift of a sequence. **(12 Marks)**

3 a. With necessary diagram, describe the characteristics of practical frequency selective filters. **(04 Marks)**

b. Write the expression of frequency response and phase characteristics for linear phase FIR filter with symmetric and asymmetric conditions. **(04 Marks)**

c. Design an FIR linear phase digital filter approximating the ideal frequency response

$$H_d(w) = \begin{cases} e^{-jw12} & \text{for } |w| \le \pi/6 \\ 0 & \text{for } \pi/6 < |w| \le \pi \end{cases}$$

Determine the coefficients of a 25 tap filter based on the window method with a rectangular window and Hanning window. **(06 Marks)**

d. Determine the coefficients of a linear phase FIR filter of length M = 15 which has a symmetric unit sample response and a frequency response that satisfies the conditions:

$$H_r\left(\frac{2\pi k}{15}\right) = \begin{cases} 1 & k = 0,1,2,3 \\ 0.4 & k = 4 \\ 0 & k = 5,6,7 \end{cases}.$$ **(06 Marks)**

4 a. Derive the system function for IIR filter using impulse invariant technique. **(06 Marks)**
b. Derive the mapping formula for IIR filter using bilinear transformation. **(08 Marks)**
c. Design a single pole low pass digital filter with a 3dB bandwidth of 0.2π, using the bilinear transformation applied to the analog filter $H(s) = \frac{\Omega_c}{s + \Omega_c}$ where, Ω_c is the 3dB bandwidth of the analog filter. **(06 Marks)**

5 a. Explain the decimation process for an integer factor D. **(10 Marks)**
b. Explain the interpolation process for an integer factor I. **(10 Marks)**

6 a. Describe the sampling rate conversion by a rational factor (I/D). **(10 Marks)**
b. Describe the polyphase structure for decimation and interpolation filters. **(10 Marks)**

7 With neat block diagram, explain the application of adaptive filters in
i) System modeling
ii) Adaptive channel equalization and
iii) Echo cancellation in data transmission (Telephone channel and digital communication system). **(20 Marks)**

8 a. Derive that expression of Least Mean Square (LMS) algorithm with fixed step size. **(10 Marks)**
b. Derive the expression of direct form Recursive Least Squares (RLS) algorithm. **(10 Marks)**

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