2nd Sem D.E

USN | | | | | | | | | | **12EC123**

M.Tech. Degree Examination, June/July 2014
Modern DSP

Time: 3 hrs. Max. Marks:100

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

1 a. Derive an expression for SQNR in case of sinusoidal quantization. **(06 Marks)**

b. Determine whether each of the following signals are periodic, if the signals are periodic find its fundamental period?

i) $x_a(t) = 3\cos\left(5t + \frac{\pi}{6}\right)$

ii) $x(n) = \cos\left(\frac{n\pi}{2}\right) - \sin\left(\frac{n\pi}{8}\right) + 3\cos\left(\frac{n\pi}{4} + \frac{\pi}{3}\right)$ **(06 Marks)**

c. A digital communication link carries binary coded words representing samples of an input signal $x_a(t) = 3\cos 600\pi t + 2\cos 1800\pi t$. The link is operated at 10,000 bits/sec and each input signal is quantized in to 1024 different voltage levels:

i) What is the sampling frequency and folding frequency?
ii) What is he Nyquist rate for the signal $x_a(t)$?
iii) What are the frequencies in the resulting discrete-time signal x(n)?
iv) What is the resolution Δ? **(08 Marks)**

2 a. Compute the N point DFT of the signal

$$x(n) = \begin{cases} 1, & n \text{ even} \quad 0 \le n \le N-1 \\ 0, & n \text{ odd} \end{cases} \quad N \text{ is odd.}$$ **(06 Marks)**

b. State and prove circular time shift property of DFT. **(06 Marks)**

c. Consider the finite length sequence $x(n) = \delta(n) - 2\delta(n-5)$. Find:

i) The 10 point DFT of x(n)
ii) The sequence y(n) that has a DFT $Y(K) = e^{-\frac{j4\pi k}{10}} X(K)$ where X(K) is the 10 point DFT of x(n).
iii) The 10 point sequence y(n) that has a DFT Y(K) = X(K)W(K) where X(K) is the 10 point DFT of X(n) and W(K) is the 10 point DFT of U(n) – U(n – 6). **(08 Marks)**

3 a. Consider a FIR filter with impulse response h(n) = {3, 2, 1, 1}. If the input is x(n) = {1, 2, 3, 3, 2, 1, -1, -2, -3, 5, 6, -1, 2, 0, 2, 1}. Find the output y(n) using overlap add method and using 7 point circular convolution. **(08 Marks)**

b. Calculate the filter coefficients of an FIR filter with passband edge frequency of 1.5 kHz, stopband edge frequency 2 kHz, sampling frequency of 8 kHz. Use Hamming window. **(08 Marks)**

c. Compare IIR and FIR filters. **(04 Marks)**

4 a. Design a 17-tap linear phase FIR filter with cut-off frequency $W_c = \frac{\pi}{2}$. The design is to be done based on frequency sampling technique. **(08 Marks)**

b. Explain how an analog filter is mapped on to a digital filter using impulse invariance method. What are the limitations of the method? How the mapping is improved with bilinear transformation? **(12 Marks)**

5 a. Design a Chebyshev 1 filter to meet the following specifications:
i) Passband ripple ≤ 2 dB
ii) Passband edge frequency : 1 rad/sec
iii) Stopband attenuation ≥ 20 dB
iv) Stopband edge frequency : 1.3 rad/sec. **(10 Marks)**

b. Design a IIR filter using Butterworth approach for the following specifications. Use bilinear transformation

$$0.8 \leq |H(e^{jw})| \leq 1 \quad \text{for} \quad 0 \leq w \leq 0.2\pi$$
$$|H(e^{jw})| \leq 0.2 \quad \text{for} \quad 0.6\pi \leq w \leq \pi$$
(10 Marks)

6 a. Explain the frequency domain characterization of down sampling and up sampling. **(10 Marks)**

b. Explain the sampling rate conversion by a rational factor I/D. **(10 Marks)**

7 a. Explain the polyphase decomposition of a linear filter for down sampling and upsampling. **(10 Marks)**

b. With a neat block diagram and equations, explain 2 channel quadrature mirror filter bank. How it eliminates aliasing? Also explain the perfect reconstruction of a 2 channel QMF bank. **(10 Marks)**

8 a. Explain adaptive channel equalization. **(10 Marks)**

b. Explain linear predictive coding of speech signals. **(10 Marks)**

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