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

Second Semester M.Tech. Degree Examination, June/July 2016

Modern DSP

Time: 3 hrs. Max. Marks: 100

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

1 a. Derive an expression for SQNR of sinusoidal signals. **(06 Marks)**

b. A digital communication link carries binary coded words representing samples of an input signal $x_a(t) = 3\cos 600t - 2\cos 1800t$. The link is operated at 10000 bits/second and each input is quantized into 1024 different voltage levels.

i) What is the sampling frequency and folding frequency?

ii) What is the 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)**

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

2 a. Let x(n) = <1, -1, 2, 3, 0, 0>. Without computing IDFT, find y(n) whose 6-point DFT $Y(K) = W_3^{2K} X(K)$. **(08 Marks)**

b. Consider an FIR filter with impulse response $h(n) = \{\underset{\uparrow}{3}, 2, 1, 1\}$. If the input sequence x(n) = {1, 2, 3, 3, 2, 1, -1, -2, -3, 5, 6, -1, 2, 0, 2, 1}. Find the output using overlap-Add method assuming a block length of 7. **(12 Marks)**

3 a. The desired frequency response of an LPF is given by

$$H_d(e^{j\omega}) = H_d(\omega) = \begin{Bmatrix} e^{-j3\omega}, & |\omega| < \frac{3\pi}{4} \\ 0, & \frac{3\pi}{4} < |\omega| < \pi \end{Bmatrix}.$$

Determine the frequency response of the FIR filter if Hamming window is used with N = 7. **(10 Marks)**

b. Determine the coefficients of a linear-phase FIR filter of length M = 32, which has a symmetric unit sample response and a frequency response that satisfies the condition.

$$Hr\left(\frac{2\pi(K+\alpha)}{32}\right) = \begin{Bmatrix} 1, & K = 0,1,2,3,4,5 \\ T_1, & K = 6 \\ 0, & K = 7,8,\ldots 15 \end{Bmatrix}$$

where $T_1 = 0.378$ for $\alpha = 0$

$T_2 = 0.357$ for $\alpha = 1/2$ **(10 Marks)**

4 a. Design a 17 tap linear-phase FIR filter with a cutoff frequency $\omega_c = \pi/2$. The design is to be done based on frequency sampling technique. **(10 Marks)**

b. Determine the order and poles of a type I lowpass Chebyshev filter that has a 1 dB ripple in the passband, a cut-off frequency Ω_p = 1000 π, a stopband frequency of 2000 π and an attenuation of 40 dB or more for $\Omega \geq \Omega_s$. **(10 Marks)**

Important Note : 1. On completing your answers, compulsorily draw diagonal cross lines on the remaining blank pages.
2. Any revealing of identification, appeal to evaluator and /or equations written eg, 42+8 = 50, will be treated as malpractice.

5 a. Derive an expression for interpolation process for an integer by factor I. (10 Marks)

b. Derive an expression for decimation or down sampling process for an integer factor D. (10 Marks)

6 a. With a neat block diagram, explain the application of multi-rate DSP in sub-band coding of speech signals. (10 Marks)

b. Explain the two channel quadrature mirror filter band and how aliasing is eliminated. (10 Marks)

7 a. Explain the analysis and synthesis structure of UDFT filter bank with efficient realization structure. (10 Marks)

b. Explain the application of f adaptive filters in:

i) Suppression of narrow band interference in a wideband signal.

ii) Linear antenna array. (10 Marks)

8 a. Derive an expression for LMS algorithm. (10 Marks)

b. Derive an expression for RLS algorithm. (10 Marks)

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