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14ELD21

Second Semester M.Tech. Degree Examination, Dec.2017/Jan.2018

Modern DSP

Time: 3 hrs. Max. Marks:100

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

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.

1 a. Define signals and system. Explain classifications of signals with example. **(14 Marks)**

b. Consider the analog signal $x_a(t) = 3\cos 2000\pi t + 5\sin 6000\pi t + 10\cos 12000\pi t$:

i) What is the Nyquist rate for this signal?

ii) Assume now that we sample this signal using a sampling rate Fs = 5000 samples/s. What is the discrete time signal obtained after sampling? **(06 Marks)**

2 a. Consider the signal $x(n) = a^n u(n)$, $0 < a < 1$. The spectrum of this signal is sampled at frequencies $\omega_k = 2\pi k/N$, k = 0, 1, ….N – 1. Determine the reconstructed spectra for a = 0.8 when N = 5 and N = 50. **(08 Marks)**

b. State and prove the following properties:

i) Circular time shift of a sequence.

ii) Circular correlation. **(06 Marks)**

c. Explain the use of the DFT in linear filtering. **(06 Marks)**

3 a. Explain characteristics of practical frequency-selective filters. **(06 Marks)**

b. Explain the design of linear phase FIR filter using windows. **(06 Marks)**

c. 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

$$Hr\left(\frac{2\pi k}{15}\right) = \begin{cases} 1, & K = 0,1,2,3 \\ 0.4, & K = 4 \\ 0, & K = 5,6,7 \end{cases}$$ **(08 Marks)**

4 a. Determine the order and the poles of a type 1 lowpass Chebyshev filter that has a 1 dB ripple in the passband, a cutoff frequency $\Omega_p = 1000\pi$, a stop band frequency of 2000π, and an attenuation of 40 dB or more for $\Omega \geq \Omega_s$. **(12 Marks)**

b. Explain the frequency transformation in digital domain. **(08 Marks)**

5 a. Explain the design and implementation for sampling rate conversion using polyphase filter structure. **(10 Marks)**

b. Explain the sampling rate conversion of bandpass signals. **(10 Marks)**

6 a. Briefly explain the practical applications of multirate signal processing. **(08 Marks)**

b. Explain the analysis and synthesis of uniform DFT filter banks. **(12 Marks)**

7 a. Write a note on two-channel quadrature mirror filter bank. **(10 Marks)**

b. Define adaptive filter. Explain steps used to implement LMS algorithm. **(10 Marks)**

8 a. Describe the applications of adaptive filter. **(08 Marks)**

b. Derive the filter coefficient updating equation using RLS algorithm. **(12 Marks)**

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