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14ELD21

Second Semester M.Tech. Degree Examination, Dec.2018/Jan.2019
Modern DSP

Time: 3 hrs. Max. Marks:100

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

1 a. How are discrete – time signals and systems classified? **(08 Marks)**

b. Define N-point DFT and IDFT of a sequence. **(04 Marks)**

c. Find the 8-point DFT of x(n) = {1, 1, 0, 0, 0, 0, 0, 0} use the property of conjugate symmetry. **(08 Marks)**

2 a. Let x(n) = (1, 2, 0, 3, –2, 4, 7, 5). Evaluate the following :

i) X(0) ii) X(4) iii) $\sum_{k=0}^{7} X(k)$ iv) $\sum_{k=0}^{7} |x(k)|^2$. **(10 Marks)**

b. If DFT of x(n) is X(K), then prove the following properties.

i) $DFT\{w_N^{-\ell n} x(n)\} = X((K-\ell))_N$ **(03 Marks)**

ii) $DFT\{x((-n))_N\} = X((-K))_N$ **(03 Marks)**

iii) $DFT\{x((n-m))_N\} = w_N^{mk} X(K) \qquad 0 \le K \le N-1$. **(04 Marks)**

3 Design a digital FIR low-pass filter with

$$H_d(e^{jw}) = \begin{cases} e^{-jw}, & -\frac{\pi}{4} \le w \le \frac{\pi}{4} \\ 0, & \frac{\pi}{4} \le |w| \le \pi \end{cases}$$

using a Hamming window with N=7. **(20 Marks)**

4 a. Explain how an analog filter is mapped on to a digital using impulse invariance method. What are the limitations of the method? **(10 Marks)**

b. Convert the analog filter with transfer function :

$$H(s) = \frac{(s+a)}{(s+a)^2 + b^2}$$

into a digital filter using the impulse invariant transformation. **(10 Marks)**

5 a. Design a low-pass Butterworth filter using the bilinear transformation method for satisfying the following constraints.

Passband	: 0 – 400 Hz	stopband	: 2.1 KHz
passband ripple	: 2dB	stopband attenuation	: 20 dB
sampling frequency	: 10 KHz.		

(10 Marks)

b. Design a Chebyshev IIR digital low- pass filter to satisfy the constraints :

$$0.707 \le |H(\omega)| \le 1 \qquad 0 \le \omega \le 0.2\pi$$
$$|H(\omega)| \le 0.1 \qquad 0.5\pi \le \omega \le \pi$$

using bilinear transformation and assuming T = 1s. **(10 Marks)**

6 a. Explain the decimation process for an integer factor D. **(10 Marks)**

b. Consider the signal x(n) = n u(n)

i) Determine the spectrum of the signal

ii) The singal is applied to a decimator that reduces the sampling rate by a factor 3 determine the output spectrum. **(10 Marks)**

7 a. Describe the polyphase structure for decimation and interpolation filters. **(10 Marks)**

b. Explain clearly two applications of multi-note digital signal processing. **(10 Marks)**

8 a. Derive the expression of direct form Recursive least Square (RLS) algorithm. **(10 Marks)**

b. Explain four applications of adoptive filters. **(10 Marks)**

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