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Question Paper Code : 57293

B.E./B.Tech. DEGREE EXAMINATION, MAY/JUNE 2016

Fifth Semester

Medical Engineering

EC 6502 – PRINCIPLES OF DIGITAL SIGNAL PROCESSING

(Common to Electronics and Communication Engineering)

(Regulations 2013)

Time : Three Hours **Maximum : 100 Marks**

Answer ALL questions.

PART – A (10 × 2 = 20 Marks)

1. Is $h(n) = \frac{-1}{4}\delta(n+1) + \frac{1}{2}\delta(n) - \frac{1}{4}\delta(n-1)$ is stable and causal ? Justify.

2. What is the smallest no. of DFTs and IDFTs needed to compute the linear convolution of a length 50 sequence with a length of 800 sequence is to be computed using 64 pt DFT & IDFT ?

3. What is known as warping effect ?

4. Why impulse invariant method is not preferred in the design of IIR filter other than LPF ?

5. What are the two kinds of limit cycle behaviour in DSP ?

6. List out the advantages of FIR filters.

7. Define Dead band.

8. What are the methods used to prevent overflow ?

9. What is the need for antialiasing filter ?

10. If the spectrum of a sequence $x(n)$ is $X(e^{jw})$, then what is the spectrum of the signal down sampled by 2 ?

PART – B (5 × 16 = 80 Marks)

11. (a) (i) State and prove if $x_3(K) = x_1(K)\, x_2(K)$, then $x_3(n) = \sum_{m=0}^{N-1} x_1(m)\, x_2((n-m))_N$. **(6)**

(ii) Using the equation given in 11(a)(i), for the 8 point DFT of the sequence

$x(n) = 1, 0 \le n \le 3$

$0, 4 \le n \le 7$, compute the

DFT of $x_1(n) = 1, n = 0$

$0, 1 \le n \le 4$

$1, 5 \le n \le 7$. **(10)**

OR

(b) (i) Compute the 8 point circular convolution

$x_1(n) = \{1, 1, 1, 1, 0, 0, 0, 0\}$

$x_2(n) = \sin\frac{3\pi n}{8}, 0 \le n \le 7$

using matrix method. **(12)**

(ii) State the differences between (a) overlap-save (b) overlap-add. **(4)**

12. (a) If $H_a(S) = \frac{1}{(S+1)(S+2)}$, find the corresponding H(z) using impulse invariant method for sampling frequency of 5 samples/second. **(16)**

OR

(b) Write down steps to design digital filter using bilinear transform technique and using this design a HPF with a pass band cutoff frequency of 1000 Hz & down 10 dB at 350 Hz the sampling frequency is 5000 Hz. **(16)**

13. (a) Design a filter with $H_d(e^{jw}) = e^{-j3w}, -\frac{\pi}{4} \le w \le \frac{\pi}{4}$

$$= 0, \frac{\pi}{4} < |w| \le \pi$$

Using a Hamming window with N = 7. **(16)**

OR

(b) Consider the transfer function $H(z) = H_1(z) \cdot H_2(z)$ where $H_1(z) = \frac{1}{1 - \alpha_1 z^{-1}}$ and $H_2(z) = \frac{1}{1 - \alpha_2 z^{-1}}$. Find the output round off noise power by assuming $\alpha_1 = 0.5$, $\alpha_2 = 0.6$. **(16)**

14. (a) Draw the quantization noise model for a second order system $H(z) = \frac{1}{1 - 2r\cos\theta\, z^{-1} + r^2 z^{-2}}$ and find the steady state output noise variance. **(16)**

OR

(b) Explain the characteristics of limit cycle oscillation with respect to the system described by the difference equation $y(n) = 0.95\, y(n-1) + x(n)$. Determine the dead band of the filter. **(16)**

15. (a) For the signal $x(n)$, obtain the spectrum of down sampled signal $x(Mn)$ and upsampled signal $x\left(\frac{n}{L}\right)$ **(16)**

OR

(b) Discuss in detail about any two applications of adaptive filtering with a suitable diagram.
