Question Paper Code : 80596

B.E./B.Tech. DEGREE EXAMINATION, NOVEMBER/DECEMBER 2016.

Fifth Semester

Information Technology

IT 6502 — DIGITAL SIGNAL PROCESSING

(Common to Sixth Semester Computer Science and Engineering and Mechatronics Engineering)

(Regulations 2013)

Time : Three hours | Maximum : 100 marks

Answer ALL questions.

PART A — ($10 \times 2 = 20$ marks)

1. What do you mean by Signal and Signal Processing?

2. What do you mean by convolution?

3. Write N-point DFT for, $x(n)$ and IDFT of $X(k)$.

4. What is meant by radix-2 FFT?

5. Distinguish analog and digital filters.

6. What is meant by impulse invariant method?

7. What are advantages of FIR filter over IIR filter?

8. What condition on the FIR sequence $h(n)$ are to be imposed n order that this filter can be called a linear phase filter? Write the necessary and sufficient condition for the FIR filter to have linear phase.

9. Compare fixed point and floating point representations.

10. Define dead band.

PART B — ($5 \times 16 = 80$ marks)

11. (a) (i) Determine the power and energy of the signal $x(n) = \sin\left(\frac{\pi}{4}\right)n$. (8)

(ii) Determine whether the system described by the input output relation is time invariant or not

(1) $y(n) = x(n-1)$

(2) $y(n) = x(-n)$. (8)

Or

(b) (i) Determine the z transform and ROC of the signal $x(n) = (1/3)^n u(n)$. (8)

(ii) Find the cross correlation of $x(n) = \{1,2,1,1\}$ and $y(n) = \{1,1,2,1\}$. (8)

12. (a) Find the 8 point DFT of the sequence $x(n) = \{1, 1, 1, 1, 1, 1, 0, 0\}$. (16)

Or

(b) Compute the DFT for the sequence {2, 2, 2, 2, 1, 1, 1, 1}. Using radix -2 DIT - FFT algorithm. (16)

13. (a) Design a Butterworth low pass filter satisfying the following constraints.

$$\sqrt{0.5} \le \left|H(e^{jw})\right| \le 1, \quad 0 \le w \le \frac{\pi}{2}$$

Use Bilinear transformation

$$\left|H(e^{jw})\right| \le 0.2, \quad \frac{3\pi}{4} \le w \le \pi$$

Or

(b) Design an analog Chebyshev filter for the following specifications. Passband gain 0.89. Stop band attenuation 0.2, passband edge frequency 30Hz and stop band edge frequency 75Hz.

14. (a) Design a HPF with cut off frequency 1.2 radians of length N = 9 using Hamming window. (16)

Or

(b) Using frequency sampling method design a lowpass filter with the following specifications cut off frequency, $\omega_c = \pi/4$ and $N = 15$ and plot the magnitude response. (16)

15. (a) Derive the steady state output noise power and Find the steady state variance of the noise in the output due to quantization of input for the first order filter $y(n) = \alpha y(n-1) + x(n)$. (16)

Or

(b) State the need for Scaling and derive the scaling factor for a second order IIR filter. (16)