B.E./B.Tech. DEGREE EXAMINATION, APRIL/MAY 2017.

Fifth/Sixth Semester

Information Technology

IT 6502 – DIGITAL SIGNAL PROCESSING

(Common to Computer Science and Engineering/Mechatronics Engineering)

(Regulations 2013)

Time : Three hours — Maximum : 100 marks

Answer ALL questions.

PART A — (10 × 2 = 20 marks)

1. What is meant by aliasing? How can it be avoided?

2. Find the energy of $(1/4)^n u(n)$.

3. The first 5 DFT coefficients of a sequence x(n) are X(0) = 2, X(1) = 0.5 – j1.206, X(2) = 0, X(3) = 0.5-j0.206, X(4) = 0.Determine the remaining DFT coefficients.

4. Calculate % saving in computing through radix -2, DFT algorithm of DFT coefficients. Assume N = 512.

5. What does "frequency warping" mean? What is the effect on magnitude and phase response?

6. Given the Transfer function of LPF, $H(s)=\frac{1}{s+1}$, find the Transfer function of HPF having a cutoff frequency of 10 rad/sec.

7. State the advantages and disadvantages of FIR filter over IIR filter.

8. Define Gibbs phenomenon.

9. What is zero input limit cycle oscillation?

10. Define truncation error for sign magnitude representation and for 2's complement representation?

PART B — (5 × 16 = 80 marks)

11. (a) (i) Check whether the systems described by the following equations are (1) $y(n) = x(n) \cos \omega n$ (2) $y(n) = |x(n)|$

Static or Dynamic, Causal or non causal, Linear or nonlinear, Time variant or invariant, Stable or Unstable. (8)

(ii) Find the response of the system for the input signal, $x(n) = \{1,2,2,3\}$ and $h(n) = \{1,0,3,2\}$. (8)

Or

(b) Determine the inverse Z-transform of $X(z) = 1 / (1-1.5z^{-1}+0.5z^{-2})$ if

(i) ROC : $|Z|>1$ (ii) ROC : $|Z|<0.5$ (iii) ROC : $0.5<|Z|<1$. (16)

12. (a) Explain the filtering methods based on DFT and FFT.

Or

(b) Determine the response of LTI system when input sequence $x(n) = \{-1, 1, 2, 1\}$ and impulse response $h(n) = \{-1, 1, -1, 1\}$ by radix-2 DIT FFT. (16)

13. (a) The specification of the desired low pass digital filter is

$0.8 \le |H(e^{j\omega})| \le 1.0;\ 0 \le \omega \le 0.2\pi$

$|H(e^{j\omega})| \le 0.2;\ 0.6\pi \le \omega \le \pi$. Design a Chebyshev digital filter using impulse Invariant Transformation. (16)

Or

(b) (i) Determine the system function of the IIR digital filter for the analog transfer function $H(s) = \frac{10}{s^2+7s+10}$ with T = 0.2 sec using impulse invariance method. (8)

(ii) Obtain the direct form-I and direct form-II realization for the system

$y(n) = -0.1\,y(n-1) + 0.2\,y(n-2) + 3x(n) + 3.6x(n-1) + 0.6x(n-2)$ (8)

14. (a) Design an FIR filter for the ideal frequency response using Hamming window with N=7.

$H_d(\omega) = e^{-j3\omega}$ for $-\frac{\pi}{8} \le \omega \le \frac{\pi}{8}$,

0 for $\frac{\pi}{8} \le |\omega| \le \pi$. (16)

Or

(b) Determine the filter coefficient $h(n)$ of length $M=15$ obtained by sampling and its frequency response as

$$H\left(\frac{2\pi k}{15}\right)=1 \quad ; K=0, 1, 2, 3, 4$$
$$=0.4 \ ; \ K=5$$
$$=0 \quad ; \ K=6, 7.$$
(16)

15\. (a) (i) Explain the characteristics of a limit cycle oscillation w.r.to the system described by the equation $y(n) = 0.95y(n-1)+x(n)$. Determine the dead band of the filter. (12)

(ii) Bring out the differences between fixed-point and floating-point arithmetic. (4)

Or

(b) (i) Explain in detail about finite word length effects in Digital filter. (8)

(ii) Determine the variance of the round of noise power at the output of cascade realization of the filter is as described by the transfer function $H(z) = H_1(z) \ H_2(z)$. Where $H_1(z)=\frac{1}{1-0.5z^{-1}}$ and $H_2(z)=\frac{1}{1-0.25z^{-1}}$. (8)