Reg. No. :

Question Paper Code : 40962

B.E./B.Tech. DEGREE EXAMINATION, APRIL/MAY 2018

Fifth/Sixth Semester

Electronics and Communication Engineering

EC 6502 – PRINCIPLES OF DIGITAL SIGNAL PROCESSING

(Common to Biomedical Engineering/Medical Electronics)

(Regulations 2013)

Time : Three Hours | Maximum : 100 Marks

Answer ALL questions

PART – A | (10×2=20 Marks)

1. Calculate the 4-point DFT of the sequence $x(n) = \{\underset{\uparrow}{1} \;\; 0 \;\; -1 \;\; 0\}$
2. What is the relationship between Fourier transform and DFT ?
3. What are the methods used for digitizing the analog filter into a digital filter ?
4. What is meant by frequency warping ?
5. Draw the direct form realization of FIR system.
6. How the zeros in FIR filter is located ?
7. Distinguish between fixed point arithmetic and floating point arithmetic.
8. Why is rounding preferred over truncation in realizing a digital filter ?
9. Show that the up sampler and down sampler are time invariant system.
10. Write the expression for the output y(n) as a function of the input x(n) for the given multirate system as in Figure 1.

$$x(n) \rightarrow \boxed{\uparrow 5} \rightarrow \boxed{\downarrow 10} \rightarrow \boxed{\uparrow 2} \rightarrow y(n)$$

Figure 1

For Notes, Syllabus, Question Papers: www.AllAbtEngg.com

PART – B (5×13=65 Marks)

11. a) i) State and prove any four properties of DFT. (8)
ii) Perform circular convolution of the following sequences $x_1(n) = \{1\ \ 1\ \ 2\ \ 1\}$; $x_2(n) = \{1\ \ 2\ \ 3\ \ 4\}$. (5)

(OR)

b) i) Mention the differences and similarities between DIT and DIF algorithms. (5)
ii) Compute 4 point DFT of a sequence $x(n) = \{0\ \ 1\ \ 2\ \ 3\}$ using DIF and DIT algorithms. (8)

12. a) i) Design an analog Butterworth filter for a given specifications. (7)
$0.9 \le |H(j\Omega)| \le 1$ for $0 \le \Omega \le 0.2\,\pi$.
$|H(j\Omega)| \le 0.2$ for $0.4\,\pi \le \Omega \le \pi$.
ii) Apply impulse invariant method and find H(z) for $H(s) = \dfrac{s+a}{(s+a)^2+b^2}$. (6)

(OR)

b) i) Apply bilinear transformation to $H(s) = \dfrac{2}{(s+1)\,(s+2)}$ with T = 1 sec and find H(z). (6)
ii) Explain the Lattice-Ladder structure with neat diagram. (7)

13. a) Write the expression for the frequency response of Rectangular window and Hamming window and explain. (7+6)

(OR)

b) Determine the filter coefficients h(n) obtained by sampling

$$H_d(e^{j\omega}) = e^{-j(N-1)\omega/2} \qquad 0 \le |\omega| \le \frac{\pi}{2}$$
$$= 0 \qquad \frac{\pi}{2} \le |\omega| \le \pi$$

for N = 7. (13)

14. a) The output signal of an A/D convertor is passed through a first order low pass filter, with transfer function given by $H(z) = \dfrac{(1-a)z}{z-a}$ for $0 \le a \le 1$. Find the steady state output noise power due to quantization at the output of the digital filter. (13)

(OR)

b) Briefly explain the following :
i) Coefficient quantization error. (4)
ii) Product quantization error. (4)
iii) Truncation and Rounding. (5)

15. a) Explain sampling rate conversion by a rational factor and derive input-output relation in both time and frequency domain. (13)

(OR)

b) With neat required diagrams explain any two applications of adaptive filtering. (6+7)

PART – C (1×15=15 Marks)

16. a) An FIR Filter is given by the difference equation

$$y(n) = 2x(n) + \frac{4}{5}x(n-1) + \frac{3}{2}x(n-2) + \frac{2}{3}x(n-3)$$

Determine its lattice form. (15)

(OR)

b) How is signal scaling used to prevent overflow limit cycle in the digital filter implementation ? Explain with an example. (15)