G 463 (Pages : 3) Reg. No.................

Name.....................

B.TECH. DEGREE EXAMINATION, MAY 2014

Sixth Semester

Branch : Electronics and Communication Engineering/Information Technology/Applied Electronics and Instrumentation/Electronics and Instrumentation

DIGITAL SIGNAL PROCESSING (L,T,A,S)

(Old Scheme—Prior to 2010 Admissions)

[Supplementary/Mercy Chance]

Time : Three Hours Maximum : 100 Marks

Part A

Answer **all** *questions briefly.*
Each question carries 4 *marks.*

1. Explain the impulse invariance method of transforming analog filter to digital filter.
2. Draw the cascade realization of the system function $H(z)=\left(1+\frac{1}{2}z^{-1}+z^{-2}\right)\left(1+\frac{1}{4}z^{-1}+z^{-2}\right)$.
3. What are the conditions for the impulse response of FIR filter to satisfy for constant group and phase delay and for only constant group delay ?
4. What is a Hann window function ? Obtain its frequency domain characteristics ?
5. What are the advantages of FFT algorithm when compared to direct computation of DFT ?
6. Explain the method of overlap-add to obtain linear convolution of a finite and infinite sequence.
7. Explain zero input limit cycle oscillations ?
8. Compare fixed point and floating point arithmetic.
9. What are the different methods of speech coding ? Explain briefly.
10. What is a homomorphic vecoder ? Explain with a block diagram ?

(10 × 4 = 40 marks)

Part B

Answer **all** *questions.*
Each full question carries 12 *marks.*

11. (a) Obtain the cascade form realization for the system $H(z)=\dfrac{1+\left(3/4\right)z^{-1}+\left(\frac{1}{8}\right)z^{-2}}{1-\frac{5}{8}z^{-1}+\frac{1}{16}z^{-2}}$.

Turn over

(b) Obtain the parallel form structure of a digital filter $H(z) = \frac{\left(1 - \frac{1}{2} z^{-1}\right)}{\left(1 - \frac{1}{4} z^{-1}\right)\left(1 - \frac{1}{5} z^{-1}\right)}$.

Or

12. Design a digital Butterworth filter to meet the following constraint :

$$\frac{1}{\sqrt{2}} \leq |H(w)| \leq 1 \text{ for } 0 \leq w \leq \pi/5.$$

$$0 \leq |H(w)| \leq 0.1 \text{ for } \pi/2 \leq w \leq \pi.$$

13. Design a high-pass filter using Hamming window, with a cut-off frequency of 1.2 radians/sec and N = 9.

Or

14. Consider an FIR lattice filter with coefficients $k_1, = \frac{1}{2}, k_2 = \frac{1}{3}, k_3 = \frac{1}{4}$. Determine the FIR filter coefficient for the direct form structure.

15. Compute the 8-point DFT of the sequence $x(n) = \begin{cases} 1, & 0 \leq n \leq 7 \\ 0, & \text{otherwise} \end{cases}$ using DIT and DIF algorithms.

Or

16. (a) Find the circular convolution of 2 sequences $x_1(n) = \{\underset{\uparrow}{2}, 1, 2, 1\}$, $x_2(n) = \{1, 2, 3, 4\}$.

(6 marks)

(b) Consider the following 8 point sequences defined for $0 \leq n \leq 7$. Which of these sequences have a real valued 8 point DFT ?

(i) $x_1(n) = \{1, 2, 3, -1, 0, -1, 3, 2\}$.

(ii) $x_2(n) = \{0, 2, 3, 4, 0, -4, -3, -2\}$.

(6 marks)

17. Given the system $y(n) = \frac{1}{2} y(n-1) + x(n)$.

   (i) Calculate the response to the input $x(n) = \left(\frac{1}{4}\right)^n u(n)$ assuming infinite precision arithmetic.

   (ii) Calculate the response $y(n), 0 \le n \le 5$ to the same input assuming finite precision with 5 bits, one sign bit plus four fractional bits. The quantization is performed by truncation. Discuss the results.

Or

18. (a) For the system with system function $H(z) = \frac{1 + 0.75 z^{-1}}{1 - 0.04 z^{-1}}$, draw the signal flow graph and find scale factor S_0 to avoid overflow in the input adder.

(8 marks)

   (b) Write a note on finite word length effects in digital filters. (4 marks)

19. Describe channel vocoder in two separate blocks : (a) channel vocoder analyser ; (b) channel vocoder synthesizer.

Or

20 Explain with the help of block diagram, the radar system and signal processing in the radar system.

[5 × 12 = 60 marks]