Reg. No. :

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B.E./B.Tech. DEGREE EXAMINATIONS, NOVEMBER/DECEMBER 2019

Fifth/Sixth 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 (10×2=20 Marks)

1. State sampling theorem.
2. What is meant by correlation and mention its need in signal processing ?
3. The first 5 DFT coefficients of a 8 sample 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. Define DCT pair.
5. Discuss the need for prewarping.
6. What are the properties of Butterworth filter ?
7. What is the necessary and sufficient condition for the impulse response, h(n) of an FIR filter to be linear phase ?
8. Mention the desirable characteristics for choosing a window function for designing FIR filters.
9. Perform the addition of the decimal numbers (0.5 and 0.25) using binary fixed point representation.
10. Define deadband. How to compute the deadband of a recursive system ?

PART – B (5×13=65 Marks)

11. a) i) Determine the power and energy of the signal $x(n) = \left(\frac{1}{3}\right)^n u(n)$. (5)

ii) Determine whether the system described by the input-output relation y(n) = nx(n) linear, shift invariant, dynamic and causal. (8)

(OR)

b) i) Determine the z transform and plot its ROC for the signal, $x(n) = (a)^n u(n)$; $0 \le a \le 1$. **(5)**

ii) Perform the convolution given $h(n) = b^n u(n); 0 \le b \le 1$ and $x(n) = a^n u(n)$; $0 \le a \le 1$. **(8)**

12. a) Use DFT-IDFT method to compute the response of the system with impulse response $h(n) = \{1, 2, 1\}$ for the input $x(n) = \{3, 2\}$. **(13)**

(OR)

b) Compute the DFT of given sequence using DIF-FFT algorithm.

$x(n) = \{1, 2, 3, 4, 4, 3, 2, 1\}$. **(13)**

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 Butterworth digital filter using Bilinear Transformation technique. **(13)**

(OR)

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

ii) Obtain the direct form-II realization for the system $y(n) = -0.1\, y(n-1) + 0.2\, y(n-2) + 3x(n) + 3.6x(n-1) + 0.6x(n-2)$. **(5)**

14. a) Design a non-recursive HPF with cutoff frequency 1.2 radians of length N = 9 using Hamming window. **(13)**

(OR)

b) Using frequency sampling method, design a lowpass filter with the following specifications: cut off frequency, $\omega_c = \pi/4$ radians and order, N = 15 and plot the magnitude response of the designed filter. **(13)**

15. a) Find the steady state variance of the noise in the output, due to quantization of input for the first order filter $y(n) = ay(n-1) + x(n)$; $0 < a < 1$. **(13)**

(OR)

b) With relevant example, briefly discuss about the effect of coefficient quantization on the location of poles and zeros in the z-plane. **(13)**

PART – C (1×15=15 Marks)

16. a) Using Decimation in Time FFT algorithm, compute the circular convolution between the two given sequences, $x_1(n) = \{1, 2, 2, 1\}$ and $x_2(n) = \{4, 5\}$. **(15)**

(OR)

b) Use hanning window to design a 9 tap non-recursive LPF with cutoff frequency, $f_c = 2$ kHz. The filter should operate at the rate of 8000 samples/sec. Draw the linear phase realization structure for the designed filter. **(15)**