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Reg. No. : ....

Name : ....

Fourth Semester B.Tech. Degree Examination, February 2016
(2013 Scheme)
13.404 : DIGITAL SIGNAL PROCESSING (AT)

Time : 3 Hours Max. Marks : 100

PART – A

Answer **all** questions. **2** marks **each**.

M1 1. Determine the DFT of the sequence $x(n) = a^n u(n)$ for $a < 1$.

M1 2. The first five points of the eight-point DFT of a real-valued sequence are {0.25, 0.125 – j0.3018, 0, 0.125 – j0.0518, 0}. Determine the remaining three points.

M4 3. Determine the order of the analog Butterworth filter that has a –2 dB pass band attenuation at a frequency of 20 rad/sec and at least – 10 dB stop band attenuation at 30 rad/sec.

M3 4. Give the equation specifying Hanning and Blackman windows.

M4 5. A digital filter with a 3 dB bandwidth of 0.25π is to be designed from the analog filter whose system response is $H(s) = \Omega_c / (s + \Omega_c)$. Use bilinear transformation to obtain H(z).

M6 6. What is product quantization error ?

M6 7. Why rounding is preferred to truncation in realizing digital filter ?

M6 8. What is co-efficient inaccuracy ?

M6 9. What is the use of anti-aliasing filters in decimator ?

M6 10. State the advantages of Multirate Signal Processing.

(10×2=20 Marks)

P.T.O.

PART – B

Answer **one** question from **each** Module.

Module – 1

M1 11. a) By means of DFT and IDFT, determine the sequence $x_3(n)$ corresponding to the circular convolution of the sequences $x_1(n) = \{\underset{\uparrow}{2}, 1, 2, 1\}$ and $x_2 = \{\underset{\uparrow}{1}, 2, 3, 4\}$. 12

M1 b) Explain in detail about the filtering of long data sequences in DFT. 8

M1 12. a) State and prove any two properties of DFT. 6

M2 b) Compute 8-point DFT of the sequence $x(n) = \{1, -1, 1, -1, -1, 0, 0\}$ using DIT FFT algorithm. 10

c) Explain DCT as a orthogonal transform. 4

Module – 2

M3 13. a) Derive the frequency response of linear phase FIR filter cosidering the symmetrical impulse response when N is odd. 8

b) Determine H(Z) for a Butterworth filter satisfying the following specifications. 12

M4

$$0.8 \le |H(e^{j\omega})| \le 1, \text{ for } 0 \le \omega \le \pi/4$$

$$|H(e^{j\omega})| \le 0.2, \text{ for } \pi/2 \le \omega \le \pi$$

Assume T = 0.1 sec. Apply bilinear transformation method.

14. a) Design a Chebychev analog high pass filter that will pass all radian frequencies greater than 200 rad/sec with no more than 2 dB attenuation and have a stop band attenuation of greater than 20 dB for all ω less than 100 rad/sec. Convert the analog filter into a digital filter using impulse invariance method. 12

b) Derive mapping formula for bilinear transformation in design of IIR filters. 8

M4

Module – 3

15. a) How to avoid overflow using scaling inputs ? Explain. **8**

b) For the system with the system function $H(z) = (1 + 0.75 Z^{-1}) / (1 - 0.4 Z^{-1})$, find the scale factor S0 to avoid overflow in the input adder. **8**

c) Derive the expression for Signal to Quantization Noise ratio. **4**

16. a) Obtain the direct form I, direct form II, Cascade and parallel realization for the following system : $y(n) = 0.1\, y(n-1) + 0.2y\,(n-2) + 3x\,(n) + 3.6x\,(n-1) + 0.6x\,(n-2)$. **16**

M5

b) Express the decimal values : 6/8 and 9/8 in i) Sign Magnitude form and ii) Two's complement form. **4**

M6

Module – 4

17. a) Derive the frequency domain characterization of decimator and interpolator. **12**

M6

b) Explain about the programming tools used for DSP processors. **8**

18. a) The analysis filters in a three-channel QMF bank have the transfer functions $H_0(z) = 1 + z^{-1} + z^{-2}$; $H_1(z) = 1 - z^{-1} + z^{-2}$; $H_2(z) = 1 - z^{-2}$. **8**

Determine the synthesis filters $G_0(z)$, $G_1(z)$ and $G_2(z)$ that results in perfect reconstruction.

b) Explain in detail about Trans-Multiplexers. **12**