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Printed Pages : 4

EC – 801

(Following Paper ID and Roll No. to be filled in your Answer Book)

PAPER ID : 3048 Roll No.

B. Tech.

(SEM. VIII) EXAMINATION, 2006-07

DIGITAL SIGNAL PROCESSING

Time : 3 Hours] *[Total Marks : 100*

Note : *Attempt **all** questions. All questions carry **equal** marks. Assume missing data, if any, suitably. Notations have their usual meaning unless otherwise stated.*

1 Attempt any **four** parts of the following : 5×4

(a) What are the basic elements of DSP systems? Write the limitations of it.

(b) Write the process of reconstruction of analog signals. What is role of A/D converter in a digital signal processing system?

(c) A system has the unit sample response h (n) given by

$$h(n) = -\frac{1}{4}\delta(n+1) + \frac{1}{2}\delta(n) - \frac{1}{2}\delta(n-1)$$

(i) Is the system BIBO stable?

(ii) Is the filter causal?

(iii) Find the frequency response $H\left(e^{jw}\right)$

(d) Use convolution to find x(n) if X(z) is given by

$$X(z) = \frac{1}{\left(1-\frac{1}{2}z^{-1}\right)\left(1+\frac{1}{4}z^{-1}\right)}$$

(e) Define z – transform and region of convergence. Establish the relation between DFT and z – transform.

(f) Compute 8-point DFT of the following sequence using (i) DIT algorithm (ii) DIF algorithm

$$X(n) = \{1, 2, 3, 2, 1, 2, 3, 2\}$$

2 Attempt any **four** parts of the following : 5×4

(a) Distinguish between FIR and IIR filters. Write their merits/demerits.

(b) Discuss the Park-McClellan method for the design of equi-ripple linear phase FIR filter.

(c) Use the window method with a Hamming window to design a 13-tap differentiator (N=13)

(d) Realize the following FIR system in :

(i) Cascade form

(ii) Lattice form

$$H(z) = 1 + 3z^{-1} + 2z^{-2}$$

(e) What do you understand by effect of finite – register length in the FIR filter design? Explain how this effect affects the filter performance ?

(f) Design an approximation to an ideal high-pass filter with magnitude response.

$$H\left(e^{jw}\right) = 0;\ 0 \le |w| \le \lambda| 3$$

$$= 1;\ \text{otherwise}$$

by the Fourier series method. Take N = 11.

3 Attempt any **two** of the following : **10×2**

(a) Design a digital Butterworth filter satisfying the constraints

$$0.75 \le \left|H\left(e^{jw}\right)\right| \le 1; \; 0 \le W \le \pi/2$$

$$\left|H\left(e^{jw}\right)\right| \le 0.2; \; \frac{3\pi}{4} \le w \le \pi$$

with T = 1 sec using impulse invariance.

(b) Define the Chekysher filters in terms of the Chekysher polynomials. Give the recursive formula to generate the Chekysher formula. Explain the difference between type I and type II Chekysher filters.

(c) Using bilinear transformation, design a digital Butterworth filter with the following specifications

Sampling frequency F = 8 kHz

$\alpha_p = 2\ dB$ in the pass band $800\ Hz \le f \le 1000\ Hz$

$\alpha_s = 20\ dB$ in the stop band $0 \le f \le 400\ Hz$

and $2000\ H_2 \le f \le \infty$

4 Attempt any **two** parts of the following : **10×2**

(a) What is the need for spectral estimation? How can the energy density spectrum be determined?

What do you mean by a multi-rate digital signal processing? Enumerate areas of applications of multi-rate digital signal processing.

(b) Differentiate among following non parametric methods of power spectrum estimation

(i) The Barlett Method

(ii) The Welch Method

(iii) The Blademan and Tukey Method.

(c) Explain the following :

(i) Decimator and Decimation filter

(ii) Interpolator and interpolation filter

(iii) Poly-phase digital filter structure.

5 Attempt any **two** of the following : **10×2**

(a) Why DSP hardware/algorithms are becoming popular in signal processing? Explain the DSP subsystem used in radar system.

(b) What do you mean by acoustic characteristics of speech signal? Draw the block diagram for speech analysis procedure indications parameters. Why is short term spectrum of speech preferred ?

(c) Write short notes on the following :

(i) Speech synthesizer

(ii) Adaptive filter.