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No. of Printed Pages—4

EC–801

B. TECH.

EIGHTH SEMESTER EXAMINATION, 2003-2004

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 from the following :— **(5×4)**

   (a) What are the advantages of digital signal processing as compared to analog signal processing ? Show that any arbitrary sequence can be represented in terms of scaled and delayed unit samples.

   (b) For each of the following discrete-time systems, determine whether or not the system is :

   (i) linear, *(ii)* causal, *(iii)* stable, and *(iv)* shift invariant :—

   (i) $y(n) = n^2 x(n)$

   (ii) $y(n) = \alpha \; x(-n)$

   (c) Define Z transform and region of convergence. Find region of convergence $W(Z) = X(Z)\,Y(Z)$.

   Given that $x(n) = u(n)$, and
   $y(n) = a^n \, u(n)$.

   (d) Define system function $H(Z)$. What is meant by canonical form of realization of discrete systems ? For the following discrete system, determine the system function :—

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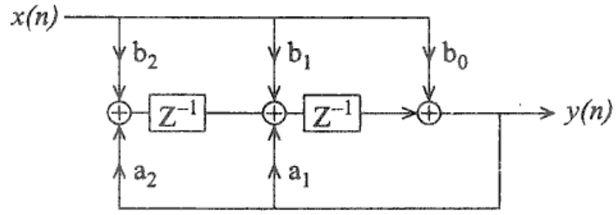

(e) Consider a periodic sequence $\tilde{w}(n)=\tilde{x}(n)+\tilde{y}(n)$, where $\tilde{x}(n)$ is periodic sequence with period N and $\tilde{y}(n)$ is also a periodic sequence with period M. Find the period of periodic sequence $\tilde{w}(n)$.

(f) What do you mean by DIT FFT algorithm for computation of DFT ? Derive mathematical expression and hence draw the flow graph of D.I.T. decomposition of an N point DFT computation into two $\frac{N}{2}$ point DFT computations with N = 8.

2. Attempt any *FOUR* parts of the following :— **(5×4)**

(a) Compare FIR and IIR filters. Show that impulse response of a linear phase FIR filter satisfies —

$$h(n)=\pm h(M-1-n) \quad n=0,1,\ldots\ldots, M-1.$$

(b) What do you mean by Gibb's phenomenon ? What is the cause of Gibb's phenomenon ? How can it be minimized ?

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

(d) What are different window functions ? Write mathematical expression for three window functions. How are these windows used in the design of FIR filter ?

(e) How are the linear phase FIR filters classified ? Draw and explain the unit sample response of each type.

(f) Write a short note on Effect of Finite Register Length in FIR filter design.

3. Attempt any *TWO* parts of the following :— **(10×2)**

(a) Derive the expression for Bilinear Transformation of Design of IIR filter. What is meant by frequency warping and how can it be removed ?

(b) Enumerate the types of Chebyshev filter. Draw and explain the characteristics of each type with mathematical expressions. Determine the order and poles of a type I low pass Chebyshev filter that has a 1dB ripple in the pass band, a cut off frequency $\Omega p = 1000\pi$, a stop band frequency of 2000π, and an attenuation of 40 dB or more for $\Omega \geq \Omega_S$.

(c) Derive mathematical expression to find system function from analog transfer function using impulse invariance technique. Convert the following analog filter transfer function given by :—

$$H_a(S) = \frac{S + 0{\cdot}1}{(S + 0{\cdot}1)^2 + 9},$$

into a discrete system function, using impulse invariance method.

4. Attempt any *TWO* parts of the following :— **(10×2)**

(a) What do you mean by multi-rate digital signal processing ? Enumerate areas of application of multi-rate digital signal processing. Find the expression for the spectra of response sequence $y(n)$ obtained by down sampling sequence $x(n)$ by a factor D.

(b) For a stationary random process, define mean value, time average mean, variance, auto covariance sequence and power density spectrum. A linear system is described by the difference equation —

$$y(n) = 0{\cdot}8y\,(n-1) + x(n) + x(n-1),$$

where $x(n)$ is stationary random processes with zero mean and auto correlation —

$$\gamma_{xx}(m) = \left(\tfrac{1}{2}\right)^{|m|}.$$

Determine :

(i) Power density spectrum,

(ii) Variance of output.

(c) Explain Bartlett power estimation method and show that the variance of the Bartlett power spectrum estimate is reduced by a factor K (No. of data segments).

5. Attempt any *TWO* parts of the following :— **(10×2)**

(a) Define Chirp Ƶ Transform. Explain with the help of suitable mathematical expressions, how Chirp Ƶ Transform can be used for radar signal spectrum analysis.

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

(c) What is meant by Cepstrum ? Draw the block diagram of cepstrum analysis for extracting spectral envelop and fundamental period. Explain each block of your diagram with suitable mathematical expressions.

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