Total No. of Questions :12] [Total No. of Pages :3

P1073

[3864] - 247

B.E. (Electronics)

ADVANCED DIGITAL SIGNAL PROCESSING

(2003 Course) (404205)

Time : 3 Hours] *[Max. Marks :100*

Instructions to the candidates:

1) ***Answer Q.1 or Q.2, Q.3 or Q.4, Q.5 or Q.6, Q.7 or Q.8, Q.9 or Q.10, Q.11 or Q.12.***
2) ***Answers to the two sections should be written in separate books.***
3) ***Neat diagrams must be drawn wherever necessary.***
4) ***Use of electronic pocket calculator is allowed.***
5) ***Figures to the right indicate full marks.***
6) ***Assume suitable data, if necessary.***

SECTION - I

Q1) a) Define the following terms. **[10]**

i) Random process.

ii) Stationary random process.

iii) Power density spectrum.

iv) Mean ergodic process.

b) Explain the difference between causality and stability. **[4]**

c) What do you mean by the term minimum phase system? **[2]**

OR

Q2) a) Explain the following terms: **[6]**

i) Decimation by a factor D.

ii) Interpolation by a factor U.

b) Explain the polyphase filter structure. **[4]**

c) Consider the signal $x(n) = a^n u(n), \mid a \mid < 1$ **[6]**

i) Determine the spectrum X(w).

ii) The signal $x(n)$ is applied to a decimator that reduces the rate by a factor of 2. Determine the O/P spectrum.

P.T.O.

Q3) a) Draw the block diagram of adaptive filters as noise canceller and explain. **[6]**

b) Explain how system modelling can be done with the help of adaptive filters. **[4]**

c) Explain the basic LMS adaptive algorithm. **[6]**

OR

Q4) a) Show that the adaptive filter turns itself off when there is no correlation between the interference signal x_R and the contaminated signal y_R. **[6]**

b) Explain adaptive filtering of ocular artifacts from the human EEG. **[6]**

c) Explain adaptive telephone echo cancellation. **[4]**

Q5) a) Explain forward linear prediction with the help of block diagram. **[8]**

b) The ARMA process is generated by the difference equation. **[10]**

$x(n) = 1.6\ x(n-1) - 0.63\ x(n-2) + w(n) + 0.9\ w(n-1)$.

i) Determine the system function of the whitening filter and its poles and zeros.

ii) Determine the power density spectrum of $\{x(n)\}$.

OR

Q6) a) Explain how Wiener filters can be used for filtering and prediction. **[8]**

b) Determine the Lattice Coefficients corresponding to FIR filter with system function. **[10]**

$$H(z) = A_3(z) = 1 + \frac{13}{24} z^{-1} + \frac{5}{8} z^{-2} + \frac{1}{3} z^{-3}.$$

SECTION - II

Q7) a) Explain the power spectrum estimation by the periodogram method. **[8]**

b) Explain the use of DFT in power spectrum estimation. **[4]**

c) What is the difference between parametric and non parametric method of power spectrum estimation. **[4]**

OR

Q8) a) Explain the Welch and the Bartlett method of power spectrum estimation. **[8]**

b) Determine the mean and the autocorrelation of the sequence $x(n)$ which is the output of a ARMA (1, 1) process described by the difference equation $x(n) = \frac{1}{2} x(n-1) + w(n) + w(n-1)$.

Where w(n) is a white noise process with variance σ_w^2. **[8]**

Q9) a) Explain the basic architecture for signal processing. **[6]**

b) Explain the principle of (VLIW) very long instruction word architecture and explain. **[6]**

c) Explain the difference between pipelined MAC configuration and non pipelined MAC configuration. **[6]**

OR

Q10) a) Explain the principles of superscalar architecture and dataflow in SHARC DSP processor. **[10]**

b) Explain how FIR digital filtering can be carried out on DSP processor. **[8]**

Q11) a) Define the following: **[6]**

i) Vowels. ii) Consonants.

iii) Formants. iv) Pitch.

b) Draw the diagram of a speech production model and explain how speech is produced. **[10]**

OR

Q12) a) What do you mean by the term speech digitization? **[3]**

b) What is the need for speech digitization? **[3]**

c) Compare the performance of various speech digitization techniques. **[6]**

d) Explain the following terms with respect to speech **[4]**

i) Cepstrum ii) MFCC.

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