Roll No. .... Total Pages : 3

505202

May 2024

M.Tech. (ECE) - II SEMESTER

ADVANCED DIGITAL SIGNAL PROCESSING

(MEC-202)

Time : 3 Hours] [Max. Marks : 75

Instructions :

1. *It is compulsory to answer all the questions (1.5 marks each) of Part-A in short.*
2. *Answer any four questions from Part-B in detail.*
3. *Different sub-parts of a question are to be attempted adjacent to each other.*

PART–A

1. (a) Briefly explain the difference between time-domain and frequency-domain representations of signals. (1.5)

(b) What is an impulse invariant filter design? (1.5)

(c) Explain the role of Quadrature Mirror Filters (QMF) in multi-rate signal processing. (1.5)

(d) Define decimation and interpolation in the context of digital signal processing. (1.5)

(e) List *two* applications of AR Lattice filters. (1.5)

(f) What is the purpose of minimum-variance spectral estimation? (1.5)

(g) What is the principle behind the Least Mean Square (LMS) algorithm for adaptive filter adaptation? (1.5)

(h) What are wavelets, and how do they differ from traditional Fourier analysis? (1.5)

(i) What is the purpose of a phase shifter in signal processing? (1.5)

(j) Differentiate between parametric and non-parametric methods for power spectrum estimation. (1.5)

PART–B

2. (a) Explain the concept of the Fast Fourier Transform (FFT) and its advantages over the DFT. (7.5)

   (b) Describe the bilinear transformation technique for IIR filter design. (7.5)

3. (a) Discuss the advantages and disadvantages of using polyphase filters in multi-rate systems. (7.5)

   (b) Explain the working principle of a QMF bank for subband decomposition of a signal. (10)

4. (a) Derive the equation for the forward linear prediction filter for a stationary random process. (15)

   (b) Discuss the application of Wiener filters for noise cancellation in digital signals. (15)

5. (a) Explain the concept of the power spectral density (PSD) of a signal. Describe the principle behind the Maximum Entropy Method (MEM) for spectral estimation. (7.5)

   (b) A discrete-time signal is generated. Using different methods like the periodogram and Welch method, estimate and compare the power spectrum of the signal. Analyze the effects of windowing techniques on the estimated spectrum. (7.5)

6. (a) Design a simple adaptive filter using the LMS algorithm to cancel a sinusoidal noise component from a corrupted signal. (7.5)

   (b) An adaptive filter with an LMS algorithm is used to cancel noise from a communication signal. The desired signal has a constant amplitude, while the noise is a sinusoid with an unknown frequency. Simulate the behavior of the LMS algorithm and analyze its convergence properties. (7.5)

7. (a) Discuss the use of multi-rate DSP techniques in image compression applications. (7.5)

   (b) Describe the role of filter banks and other DSP algorithms in speech coding and feature extraction for speech recognition. (7.5)