EE522: ADVANCED DIGITAL SIGNAL PROCESSING (3-0-0: 3)

Parametric Methods for Power Spectrum Estimation

Discrete random signals, power spectral density and properties, spectral estimation methods.
Relationship between the auto correlation and the model parameters – The Yule – Walker method for the AR Model Parameters – The Burg Method for the AR Model parameters – unconstrained least-squares method for the AR Model parameters – sequential estimation methods for the AR Model parameters – selection of AR Model order.

Filter Design

Digital filter design techniques, Basic concepts of IIR and FIR filters, difference equations, design of Butterworth IIR analog filter using impulse invariant and bilinear transform, design of linear phase FIR filters, transformation of digital filters, FIR filter design using windows, MATLAB based examples.

Adaptive Signal Processing

FIR adaptive filters – steepest descent adaptive filter – LMS algorithm – convergence of LMS algorithms – Application: noise cancellation – channel equalization – adaptive recursive filters – recursive least squares.

Digital Signal Processor

Elementary idea about the architecture and important instruction sets of TMS320C 5416/6713 processor, writing of small programs in assembly Language.

FPGA

Architecture, different sub-systems, design flow for DSP system design, mapping of DSP alrorithms onto FPGA.

Multirate DSP

Introduction to multirate DSP, decimation and interpolation, polyphase decomposition, uniform DFT filter banks, quadrature mirror filters and perfectreconstruction, introduction to

Speech Signal Processing

Digital models for speech signal : Mechanism of speech production – model for vocal tract, radiation and excitation – complete model – time domain processing of speech signal:- Pitch period estimation – using autocorrelation function – Linear predictive Coding: Basic Principles – autocorrelation method – Durbin recursive solution.

Wavelet Transforms

Fourier Transform : Its power and Limitations – Short Time Fourier Transform – The Gabor Transform - Discrete Time Fourier Transform and filter banks – Continuous Wavelet Transform – Wavelet Transform Ideal Case – Perfect Reconstruction Filter Banks and wavelets – Recursive multi-resolution decomposition – Haar Wavelet – Daubechies Wavelet.

Text Books and References

1. S. K. Mitra, “Digital Signal Processing :A computer Based Approach”, TMH.
2. J. G. Proakis, D. G. Manobakis, “Digital Signal Processing, Principles, Algorithms and Applications”, PHI.

3. M. H.Hayes, “Statistical Digital Signal Processing and Modeling”, Wiley.
4. L.R.Rabiner and R.W.Schafer, “Digital Processing of Speech Signals”, Pearson Education.
5. R. Crist, “Modern Digital Signal Processing”, Thomson Brooks/Cole.
6. R. M. Rao, A. S.Bopardikar, “Wavelet Transforms, “Introduction to Theory and applications”, Pearson Education, Asia.
7. A. V. Oppenheim and R. W. Schafer, “Discrete-Time Signal Processing”, Pearson.