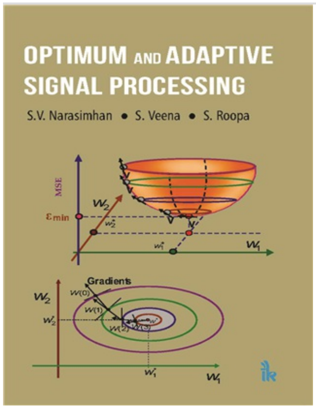
Book Information Sheet



Optimum and Adaptive Signal Processing

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About the Book

Optimum and Adaptive Signal Processing has been developed in a simple, logical and directed way. The ease of understanding is facilitated by providing all the intermediate steps in a mathematical expression derivation and its physical meaning, preserving the rigour and depth of the subject. Chapters 2 to 5, covering the background, make the book self-contained. The detailed treatment of both optimum and adaptive processing (unlike in some of the other books), enables enhanced understanding. Also, chapters 1-8 and 9-17, can cater as books on optimum signal processing and adaptive signal processing (for those with optimum signal processing background), respectively.

A CD containing MATLAB programs for most of the illustrations is provided.

Salient Features

The nonparametric estimation approach limitation, the trade-off between variance and frequency resolution presented emphasises the importance of optimum approach, for spectrum and bispectrum. **A group delay approach that reduces this limitation for spectrum and recovers the system phase from bispectrum**, are given.
In the optimum approach for the AR modelling, **warped** versions of linear prediction, Burg and escalator predictor are covered. Among the ARMA methods, for the preferred simultaneous pole-zero modelling, its limitations and the necessary solutions required to overcome them, are dealt. The ARMA modelling **by pole-zero decomposition in group delay** domain and its use in different scenarios are indicated. Further, the bispectrum estimation via AR model is presented.
Different versions of LMS adaptive algorithms, like, leaky, momentum, sign-sign, median, filtered input LMS (FX-LMS), Griffiths LMS and variable stepsize LMS; are dealt. A computationally efficient algorithm which uses **both variable stepsize and variable gradient which are robust to white/colour observation noise and its transform domain version which further improves convergence rate**, are addressed. Compared to forward error LMS, the **forward-backward error LMS algorithm** which reduces the misadjustment by 6 dB both in time and transform domain and also its warped version; are considered. Also, **a bispectrum based LMS adaptive algorithm for nonminimum phase system identification, is presented.**

Computationally efficient **Joint Laguerre escalator lattice adaptive algorithm** appropriate for nonstationary signals, the lattice and cascade predictors which have a faster convergence; are presented. Computationally efficient **Frequency domain partitioned block LMS approach** and its delayless version; are dealt.
The **conjugate gradient (CG)** method for the adaptive filter which uses single CG iteration for every sample both **in time and transform domain**; are dealt.
In delayless subband/wavelet transform domain adaptive approach, the trade-off between aliasing and eigenvalue spread and the complex weight transform; which are solved by using **DCT harmonic wavelet transform**, is covered.
For IIR equation error adaptive filters, the bias for the colour observation noise is removed by estimating the system output and using it as input to the adaptive filter which estimates the denominator polynomial, known as **bias freen approach and its extension to. Steiglitz-Mc bride method; are introduced. An adaptive notch filter realized by Steiglitz-Mc bride approach using variable stepsize LMS algorithm** is also introduced. The Recursive Least Square (RLS) algorithm and its fast transversal and lattice versions; are derived in detail. Also the affine projection and Kalman filter, are briefly treated.
The nonlinear adaptive filters based on back propagation neural network and on 2nd order Volterra series realized by transversal, lattice structures and **by transform approach**; are given. Further, the applications of adaptive filter for communication, audio, speech and acoustics; biomedical signals and Inverse control; are presented.

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