Saeed V. Vaseghi

Advanced Digital Signal Processing and Noise Reduction

Third Edition

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To my parents

With thanks to Peter Rayner, Ben Milner, Charles Ho and Aimin Chen

CONTENTS

PREFACE

Signal processing theory plays an increasingly central role in the development of modern telecommunication and information processing systems, and has a wide range of applications in multimedia technology, audio-visual signal processing, cellular mobile communication, adaptive network management, radar systems, pattern analysis, medical signal processing, financial data forecasting, decision making systems, etc. The theory and application of signal processing is concerned with the identification, modelling and utilisation of patterns and structures in a signal process. The observation signals are often distorted, incomplete and noisy. Hence, noise reduction and the removal of channel distortion is an important part of a signal processing system. The aim of this book is to provide a coherent and structured presentation of the theory and applications of statistical signal processing and noise reduction methods.

This book is organised in 15 chapters.

Chapter 1 begins with an introduction to signal processing, and provides a brief review of signal processing methodologies and applications. The basic operations of sampling and quantisation are reviewed in this chapter.

Chapter 2 provides an introduction to noise and distortion. Several different types of noise, including thermal noise, shot noise, acoustic noise, electromagnetic noise and channel distortions, are considered. The chapter concludes with an introduction to the modelling of noise processes.

Chapter 3 provides an introduction to the theory and applications of probability models and stochastic signal processing. The chapter begins with an introduction to random signals, stochastic processes, probabilistic models and statistical measures. The concepts of stationary, non-stationary and ergodic processes are introduced in this chapter, and some important classes of random processes, such as Gaussian, mixture Gaussian, Markov chains and Poisson processes, are considered. The effects of transformation of a signal on its statistical distribution are considered.

Chapter 4 is on Bayesian estimation and classification. In this chapter the estimation problem is formulated within the general framework of Bayesian inference. The chapter includes Bayesian theory, classical estimators, the estimate–maximise method, the Cramér–Rao bound on the minimum−variance estimate, Bayesian classification, and the modelling of the space of a random signal. This chapter provides a number of examples on Bayesian estimation of signals observed in noise.

Chapter 5 considers hidden Markov models (HMMs) for non-stationary signals. The chapter begins with an introduction to the modelling of non-stationary signals and then concentrates on the theory and applications of hidden Markov models. The hidden Markov model is introduced as a Bayesian model, and methods of training HMMs and using them for decoding and classification are considered. The chapter also includes the application of HMMs in noise reduction.

Chapter 6 considers Wiener Filters. The least square error filter is formulated first through minimisation of the expectation of the squared error function over the space of the error signal. Then a block-signal formulation of Wiener filters and a vector space interpretation of Wiener filters are considered. The frequency response of the Wiener filter is derived through minimisation of mean square error in the frequency domain. Some applications of the Wiener filter are considered, and a case study of the Wiener filter for removal of additive noise provides useful insight into the operation of the filter.

Chapter 7 considers adaptive filters. The chapter begins with the state-space equation for Kalman filters. The optimal filter coefficients are derived using the principle of orthogonality of the innovation signal. The recursive least squared (RLS) filter, which is an exact sample-adaptive implementation of the Wiener filter, is derived in this chapter. Then the steepest–descent search method for the optimal filter is introduced. The chapter concludes with a study of the LMS adaptive filters.

Chapter 8 considers linear prediction and sub-band linear prediction models. Forward prediction, backward prediction and lattice predictors are studied. This chapter introduces a modified predictor for the modelling of the short–term and the pitch period correlation structures. A maximum a posteriori (MAP) estimate of a predictor model that includes the prior probability density function of the predictor is introduced. This chapter concludes with the application of linear prediction in signal restoration.

Chapter 9 considers frequency analysis and power spectrum estimation. The chapter begins with an introduction to the Fourier transform, and the role of the power spectrum in identification of patterns and structures in a signal process. The chapter considers non–parametric spectral estimation, model-based spectral estimation, the maximum entropy method, and high–resolution spectral estimation based on eigenanalysis.

Chapter 10 considers interpolation of a sequence of unknown samples. This chapter begins with a study of the ideal interpolation of a band-limited signal, a simple model for the effects of a number of missing samples, and the factors that affect interpolation. Interpolators are divided into two categories: polynomial and statistical interpolators. A general form of polynomial interpolation as well as its special forms (Lagrange, Newton, Hermite and cubic spline interpolators) are considered. Statistical interpolators in this chapter include maximum a posteriori interpolation, least squared error interpolation based on an autoregressive model, time–frequency interpolation, and interpolation through search of an adaptive codebook for the best signal.

Chapter 11 considers spectral subtraction. A general form of spectral subtraction is formulated and the processing distortions that result form spectral subtraction are considered. The effects of processing-distortions on the distribution of a signal are illustrated. The chapter considers methods for removal of the distortions and also non-linear methods of spectral subtraction. This chapter concludes with an implementation of spectral subtraction for signal restoration.

Chapters 12 and 13 cover the modelling, detection and removal of impulsive noise and transient noise pulses. In Chapter 12, impulsive noise is modelled as a binary–state non-stationary process and several stochastic models for impulsive noise are considered. For removal of impulsive noise, median filters and a method based on a linear prediction model of the signal process are considered. The materials in Chapter 13 closely follow Chapter 12. In Chapter 13, a template-based method, an HMM-based method and an AR model-based method for removal of transient noise are considered.

Chapter 14 covers echo cancellation. The chapter begins with an introduction to telephone line echoes, and considers line echo suppression and adaptive line echo cancellation. Then the problem of acoustic echoes and acoustic coupling between loudspeaker and microphone systems are considered. The chapter concludes with a study of a sub-band echo cancellation system

Chapter 15 is on blind deconvolution and channel equalisation. This chapter begins with an introduction to channel distortion models and the ideal channel equaliser. Then the Wiener equaliser, blind equalisation using the channel input power spectrum, blind deconvolution based on linear predictive models, Bayesian channel equalisation, and blind equalisation for digital communication channels are considered. The chapter concludes with equalisation of maximum phase channels using higher-order statistics.

Saeed Vaseghi
June 2000

FREQUENTLY USED SYMBOLS AND ABBREVIATIONS

AWGN	Additive white Gaussian noise
ARMA	Autoregressive moving average process
AR	Autoregressive process
$\boldsymbol{A}$	Matrix of predictor coefficients
a_k	Linear predictor coefficients
$\boldsymbol{a}$	Linear predictor coefficients vector
a_{ij}	Probability of transition from state *i* to state *j* in a Markov model
$\alpha_i(t)$	Forward probability in an HMM
bps	Bits per second
$b(m)$	Backward prediction error
$b(m)$	Binary state signal
$\beta_i(t)$	Backward probability in an HMM
$c_{xx}(m)$	Covariance of signal $x(m)$
$c_{XX}(k_1,k_2,\cdots,k_N)$	k^{th} order cumulant of $x(m)$
$C_{XX}(\omega_1,\omega_2,\cdots,\omega_{k-1})$	k^{th} order cumulant spectra of $x(m)$
$\boldsymbol{D}$	Diagonal matrix
$e(m)$	Estimation error
$\mathcal{E}[x]$	Expectation of x
f	Frequency variable
$f_X(\boldsymbol{x})$	Probability density function for process $\boldsymbol{X}$
$f_{X,Y}(\boldsymbol{x},\boldsymbol{y})$	Joint probability density function of $\boldsymbol{X}$ and $\boldsymbol{Y}$
$f_{X\vert Y}(\boldsymbol{x}\vert\boldsymbol{y})$	Probability density function of $\boldsymbol{X}$ conditioned on $\boldsymbol{Y}$
$f_{X;\Theta}(\boldsymbol{x};\boldsymbol{\theta})$	Probability density function of $\boldsymbol{X}$ with $\boldsymbol{\theta}$ as a parameter
$f_{X\vert S,\mathcal{M}}(\boldsymbol{x}\vert\boldsymbol{s},\mathcal{M})$	Probability density function of $\boldsymbol{X}$ given a state sequence $\boldsymbol{s}$ of an HMM $\mathcal{M}$ of the process $\boldsymbol{X}$
$\boldsymbol{\Phi}(m,m-1)$	State transition matrix in Kalman filter
$\boldsymbol{h}$	Filter coefficient vector, Channel response
$\boldsymbol{h}_{\max}$	Maximum–phase channel response
$\boldsymbol{h}_{\min}$	Minimum–phase channel response
$\boldsymbol{h}^{\text{inv}}$	Inverse channel response
$H(f)$	Channel frequency response

$H^{\text{inv}}(f)$	Inverse channel frequency response
$\boldsymbol{H}$	Observation matrix, Distortion matrix
$\mathbf{I}$	Identity matrix
$\boldsymbol{J}$	Fisher's information matrix
$\lvert \boldsymbol{J} \rvert$	Jacobian of a transformation
$\boldsymbol{K}(m)$	Kalman gain matrix
LSE	Least square error
LSAR	Least square AR interpolation
λ	Eigenvalue
$\boldsymbol{\Lambda}$	Diagonal matrix of eigenvalues
MAP	Maximum a posterior estimate
MA	Moving average process
ML	Maximum likelihood estimate
MMSE	Minimum mean squared error estimate
m	Discrete time index
m_k	k^{th} order moment
$\mathcal{M}$	A model, e.g. an HMM
μ	Adaptation convergence factor
$\boldsymbol{\mu}_{\boldsymbol{x}}$	Expected mean of vector $\boldsymbol{x}$
$n(m)$	Noise
$\boldsymbol{n}(m)$	A noise vector of N samples
$n_i(m)$	Impulsive noise
$N(f)$	Noise spectrum
$N^*(f)$	Complex conjugate of $N(f)$
$\overline{N(f)}$	Time-averaged noise spectrum
$\mathcal{N}(\boldsymbol{x}, \boldsymbol{\mu}_{\boldsymbol{xx}}, \boldsymbol{\Sigma}_{\boldsymbol{xx}})$	A Gaussian pdf with mean vector $\boldsymbol{\mu}_{\boldsymbol{xx}}$ and covariance matrix $\boldsymbol{\Sigma}_{\boldsymbol{xx}}$
$O(\cdot)$	In the order of $(\cdot)$
P	Filter order (length)
pdf	Probability density function
pmf	Probability mass function
$P_X(\boldsymbol{x}_i)$	Probability mass function of $\boldsymbol{x}_i$
$P_{X,Y}(\boldsymbol{x}_i, \boldsymbol{y}_j)$	Joint probability mass function of $\boldsymbol{x}_i$ and $\boldsymbol{y}_j$
$P_{X\|Y}\left(\boldsymbol{x}_i \middle\| \boldsymbol{y}_j\right)$	Conditional probability mass function of $\boldsymbol{x}_i$ given $\boldsymbol{y}_j$
$P_{NN}(f)$	Power spectrum of noise $n(m)$
$P_{XX}(f)$	Power spectrum of the signal $x(m)$