Saeed V. Vaseghi

Advanced Digital Signal Processing and Noise Reduction

Third Edition

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Saeed V. Vaseghi
Professor of Communications and Signal Processing
Department of Electronics and Computer Engineering
Brunel University, UK

John Wiley & Sons, Ltd

Telephone (+44) 1243 779777

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Preface

The applications of DSP are numerous and include multimedia technology, audio signal processing, video signal processing, cellular mobile communication, adaptive network management, radar systems, pattern analysis, pattern recognition, medical signal processing, financial data forecasting, artificial intelligence, decision making systems, control systems and information search engines.

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 and interference are important parts of a signal processing system.

Since the publication of the first edition of this book in 1996, digital signal processing (DSP) in general and noise reduction in particular, have become even more central to the research and development of efficient, adaptive and intelligent mobile communication and information processing systems. The third edition of this book has been revised extensively and improved in several ways to take account of the recent advances in theory and application of digital signal processing. The existing chapters have been updated with new materials added. Two new chapters have been introduced; one for speech enhancement in mobile noisy conditions and the other for modelling and combating noise and fading in wireless communication systems.

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 and is organised in 17 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, nonstationary 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 Cramer–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 nonstationary signals. The chapter begins with an introduction to the modelling of nonstationary 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 square (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 nonparametric 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) is considered. Statistical interpolators in this chapter include maximum *a posteriori* interpolation, least square error interpolation based on an autoregressive model, time–frequency interpolation, and interpolation through the 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 from 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 nonlinear 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 nonstationary 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 is considered. The chapter concludes with a study of a sub-band echo cancellation system.

Chapter 15 covers 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.

Chapter 16 covers speech enhancement methods. Speech enhancement in noisy environments improves the quality and intelligibility of speech for human communication and increases the accuracy of automatic speech recognition systems. Noise reduction systems are increasingly important in a range of applications such as mobile phones, hands-free phones, teleconferencing systems and in-car cabin communication systems. This chapter provides an overview of the main methods for single-input and multiple-input speech enhancement in noise.

Chapter 17 covers the issue of noise in wireless communication. Noise, fading and limited radio bandwidth are the main factors that constrain the capacity and the speed of communication on wireless channels. Research and development of communications systems aim to increase the spectral efficiency, defined as the data bits per second per Hertz bandwidth of a communication channel. For improved efficiency, modern mobile communications systems rely on signal processing methods at almost every stage from source coding to the allocation of time bandwidth and space resources. In this chapter we consider how communications signal processing methods are employed for improving the speed and capacity of communications systems.

As an additional resource, this book is supported by a companion website on which lecturers and instructors can find electronic versions of the figures. Please go to ftp://ftp.wiley.co.uk/pub/books/vaseghi3e.

Saeed V. Vaseghi

Symbols

$\boldsymbol{A}$	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
$b(m)$	Backward prediction error
$b(m)$	Binary state signal
$\beta_i(t)$	Backward probability in an HMM
$c_{xx}(m)$	Covariance of signal $x(m)$
$c_{XX}(k_1, k_2, \cdots, k_N)$	kth-order cumulant of $x(m)$
$C_{XX}(\omega_1, \omega_2, \cdots, \omega_{K-1})$	kth-order cumulant spectra of $x(m)$
$\boldsymbol{D}$	Diagonal matrix
$e(m)$	Estimation error
$\mathcal{E}[x]$	Expectation of x
f	Frequency variable
Fs	Sampling frequency
$f_X(\boldsymbol{x})$	Probability density function for process $\boldsymbol{X}$
$f_{X,Y}(\boldsymbol{x}, \boldsymbol{y})$	Joint probability density function of $\boldsymbol{X}$ and $\boldsymbol{Y}$
$f_{X\|Y}(\boldsymbol{x}\|\boldsymbol{y})$	Probability density function of $\boldsymbol{X}$ conditioned on $\boldsymbol{Y}$
$f_{X;\Theta}(x; \boldsymbol{\theta})$	Probability density function of $\boldsymbol{X}$ with $\boldsymbol{\theta}$ as a parameter
$f_{X\|S,\mathcal{M}}(\boldsymbol{x}\|\boldsymbol{s}, \mathcal{M})$	Probability density function of $\boldsymbol{X}$ given a state sequence $\boldsymbol{s}$ of an HMM $\mathcal{M}$ of the process $\boldsymbol{X}$
$\Phi(m, m-1)$	State transition matrix in Kalman filter
G	Filter gain factor
$\boldsymbol{h}$	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
$\boldsymbol{I}$	Identity matrix
$\boldsymbol{J}$	Fisher's information matrix
$\lvert\boldsymbol{J}\rvert$	Jacobian of a transformation
$\boldsymbol{K}(m)$	Kalman gain matrix
λ	Eigenvalue
Λ	Diagonal matrix of eigenvalues
m	Discrete time index
m_k	kth-order moment
$\mathcal{M}$	A model, e.g. an HMM
$\boldsymbol{\mu}$	Adaptation convergence factor
$\boldsymbol{\mu}_x$	Expected mean of vector $\boldsymbol{x}$
$n(m)$	Noise
$\boldsymbol{n}(m)$	A noise vector of N samples
$n_{\mathrm{i}}(m)$	Impulsive noise
$N(f)$	Noise spectrum
$N^*(f)$	Complex conjugate of $N(f)$
$\overline{N(f)}$	Time-averaged noise spectrum
$\mathcal{N}(\boldsymbol{x}, \mu_{xx}, \boldsymbol{\Sigma}_{xx})$	A Gaussian pdf with mean vector $\boldsymbol{\mu}_{xx}$ and covariance matrix $\boldsymbol{\Sigma}_{xx}$
$O(\cdot)$	In the order of $(\cdot)$
P	Filter order (length)
$P_X(\boldsymbol{x}_i)$	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 $\boldsymbol{y}_j$
$P_{X\mid Y}(\boldsymbol{x}_i \mid \boldsymbol{y}_j)$	Conditional probability mass function of $\boldsymbol{x}_i$ given $\boldsymbol{y}_j$
$P_{\mathrm{NN}}(f)$	Power spectrum of noise $n(m)$
$P_{XX}(f)$	Power spectrum of the signal $x(m)$
$P_{XY}(f)$	Cross-power spectrum of signals $x(m)$ and $y(m)$
$\boldsymbol{\theta}$	Parameter vector
$\hat{\boldsymbol{\theta}}$	Estimate of the parameter vector θ
r_k	Reflection coefficients
$r_{xx}(m)$	Autocorrelation function
$\boldsymbol{r}_{xx}(m)$	Autocorrelation vector
$\boldsymbol{R}_{xx}$	Autocorrelation matrix of signal $\boldsymbol{x}(m)$
$\boldsymbol{R}_{xy}$	Cross-correlation matrix
$\boldsymbol{s}$	State sequence
$\boldsymbol{s}^{\mathrm{ML}}$	Maximum-likelihood state sequence
σ_{n}^2	Variance of noise $n(m)$
$\boldsymbol{\Sigma}_{\mathrm{nn}}$	Covariance matrix of noise $\boldsymbol{n}(m)$
$\boldsymbol{\Sigma}_{\mathrm{xx}}$	Covariance matrix of signal $\boldsymbol{x}(m)$
σ_x^2	Variance of signal $x(m)$
σ_n^2	Variance of noise $n(m)$
$x(m)$	Clean signal
$\hat{x}(m)$	Estimate of clean signal
$\boldsymbol{x}(m)$	Clean signal vector
$X(f)$	Frequency spectrum of signal $x(m)$
$X^*(f)$	Complex conjugate of $X(f)$

$\overline{X(f)}$	Time-averaged frequency spectrum of the signal $x(m)$
$X(f, t)$	Time-frequency spectrum of the signal $x(m)$
$\boldsymbol{X}$	Clean signal matrix
$\boldsymbol{X}^{\mathrm{H}}$	Hermitian transpose of $\boldsymbol{X}$
$y(m)$	Noisy signal
$\boldsymbol{y}(m)$	Noisy signal vector
$\hat{y}(m \mid m-i)$	Prediction of $y(m)$ based on observations up to time m–i
$\boldsymbol{Y}$	Noisy signal matrix
$\boldsymbol{Y}^{\mathrm{H}}$	Hermitian transpose of $\boldsymbol{Y}$
Var	Variance
w_k	Wiener filter coefficients
$\boldsymbol{w}(m)$	Wiener filter coefficients vector
$W(f)$	Wiener filter frequency response
z	z-transform variable

Abbreviations

AR	Autoregressive process
ARMA	Autoregressive moving average process
AWGN	Additive white Gaussian noise
bps	Bits per second
cdf	Cumulative density function
CELP	Code excited linear prediction
dB	Decibels: $10\log_{10}$(power ratio)
DFT	Discrete Fourier transform
DSP	Digital signal processing
EM	Estimate–maximise
ESPIRIT	Estimation of signal parameters via rotational invariance techniques
FFT	Fast Fourier transform
FIR	Finite impulse response
GMM	Gaussian mixture model
GSM	Global system for mobile communications
HMM	Hidden Markov model
Hz	Hertz, unit of frequency in cycles per second
IFFT	Inverse fast Fourier transform
IID	Independent identically distributed
IIR	Infinite impulse response
ISD	Itakura–Saito distance
ISI	Inter symbol interference
LMS	Least mean squared error
LP	Linear prediction model
LPSS	Spectral subtraction based on linear prediction model
LS	Least square
LSAR	Least square AR interpolation
LSE	Least square error
LTI	Linear time invariant
MA	Moving average process
MAP	Maximum *a posteriori* estimate

M-ary	Multilevel signalling
MAVE	Minimum absolute value of error estimate
MIMO	Multiple-input multiple-output
ML	Maximum likelihood estimate
MMSE	Minimum mean squared error estimate
ms	Milliseconds
MUSIC	Multiple signal classification
NLMS	Normalised least mean squared error
pdf	Probability density function
pmf	Probability mass function
psd	Power spectral density
QRD	Orthogonal matrix decomposition
RF	Radio frequency
RLS	Recursive least square
SINR	Signal-to-impulsive noise ratio
SNR	Signal-to-noise ratio
STFT	Short-time Fourier transform
SVD	Singular value decomposition
Var	Variance

Contents