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信息与通信技术教程

英文版

SIGNAL PROCESSING ADVANCES IN WIRELESS AND MOBILE COMMUNICATIONS

VOLUME 1: TRENDS IN CHANNEL ESTIMATION AND EQUALIZATION

无线与移动通信中的信号处理新技术

第1册：信道估计与均衡

GEORGIOS B.GIANNAKIS　YINGBO HUA　PETRE STOICA　LANG TONG
编著

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内 容 提 要

《无线与移动通信中的信号处理新技术》丛书，介绍了近年来无线与移动通信中使用的信号处理（SP）工具的最新的重要进展，以及世界范围内该领域的领先者的贡献。本书是两本书中的第 1 册。本丛书的内容涵盖了范围广泛的技术和方法论，包括噪声与干扰消除、调制解调器设计、移动互联网业务、下一代音频/视频广播、蜂窝移动电话和无线多媒体网络等。

本书（第 1 册）重点阐述单用户点对点链路的信道识别与均衡的关键技术。由于信息承载信号是在衰落介质中传播的，所以现代的均衡器必须充分考虑移动无线信道的可变性，减小符号间干扰和同（共）信道干扰，并抑制在单个或多个传感器的接收机中的噪声。本书介绍了最近提出的带宽节省（半）盲算法与性能分析，以及线性预编码技术，这些技术利用发射冗余使基于训练序列的系统获得明显的改善。本书内容包括：

- 盲识别与反卷积的子空间方法
- 有色信号驱动的信道的盲识别与均衡
- 最优子空间方法；多信道均衡的线性预测算法
- FIR 多信道估计的半盲方法
- 盲判决反馈均衡等

本书还介绍了在世界范围内各种期刊中的研究成果，全面汇集了用于优化单用户点对点链路的先进信号处理技术。本书对于通信工程师、研究人员、管理人员、通信系统设计人员和参与最新通信系统设计或构造的同行将是极其有价值的。

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PREFACE

Signal processing has always played a critical role in the research and development of wireless communication systems. As the demand for high capacity and high reliability systems increases, signal processing has an even more important role to play. This two-volume book, comprising chapters written by leading experts, provides an easy access to recent and important research findings in this area.

The first volume of this book focuses on channel estimation and equalization. Since the physical environment surrounding the propagation path of a radio signal is generally difficult to control, the channel characteristics in wireless communications are often time varying. This time-varying nature is a major obstacle to increasing the capacity and reliability of wireless communication systems. Fast real-time channel estimation and equalization are essential. The traditional techniques for channel estimation and equalization use training data, which not only consumes a significant portion of available bandwidth but also requires a perfect cooperation between the transmitter and receiver. In recent years, the so-called blind techniques have been explored intensively in the literature. The blind techniques do not use any training data except for certain prior information inherent in the original strings of symbols, which hence saves the bandwidth and relaxes the relationship between the transmitter and receiver. Consequently, the blind techniques have a clear potential to increase the capacity and reliability of wireless systems. As a subject area, the blind techniques have had in recent years a very fast growing rate in the general field of signal processing for communications. For this reason, the first volume of this book is devoted to the blind (or semi-blind) techniques.

The volume begins with a chapter by Tugnait on higher-order statistics methods. The higher-order statistics approach is among the earliest blind approaches and has a relatively long history. The second chapter by Stoica and Ng derives the Cramer-Rao bound for blind channel estimation problems. In the blind context, the estimated channel response is ambiguous up to (at least) a scalar. The traditional Cramer-Rao bound on estimation variance is generalized in this chapter. The third chapter by Loubaton, Moulines and Regalia introduces a versatile approach known as the subspace approach. The subspace approach often provides a good trade-off between computation and accuracy. The fourth chapter by Hua on blind identification of multiple input–multiple output (MIMO) channels driven by colored signals presents one of the latest advances in the theory of channel estimation. This theory is readily applicable to wireless communications (interfaces) between human and "smart" microphones, where the voice signals are naturally colored. The fifth chapter by Kristensson and Ottersten addresses the optimization of subspace based methods. The results shown prove that the statistical performance of this class of methods can be enhanced by careful design. The sixth chapter by Ding on linear prediction introduces a number of ways to exploit the whiteness of spectrum spread signals. This exploitation significantly increases the robustness of estimation as long as the whiteness is present. The seventh chapter by Carvalho and Slock on semi-blind methods explores the compromise between a fully trained system and a fully blind system. The highest capacity of a wireless system may be achievable via a semi-blind method. The eighth chapter by Tong, Gu and Kung on the geometrical approach provides a new look at some of the classical techniques for symbol estimation. The ninth chapter by Scaglione, Giannakis and Barbarossa on linear precoding is a tutorial presentation of this relatively new concept. Linear precoding is a versatile coding approach to combat signal distortions by unknown channels. The tenth chapter by Manton and Hua on blind channel identifiability with an arbitrary linear precoder explores in depth the implications of linear precoding by making extensive use of abstract algebra. The final chapter by Casas, Endres, Touzni, Johnson and Treichler on decision feedback equalization brings the reader back to a classic equalization approach for which there is a rich body of theory and applications. This "feedback" will ensure continued renewal, cross-fertilization and further development of many useful concepts and techniques for wireless communications.

On a closing note, the co-editors wish to thank all the contributors to this fine collection, as well as their students, mentors and collaborators that introduced them and continue to spark their interest in signal processing and communications-related research. The first author wishes to extend special thanks to his graduate student and co-author Zhengdao Wang for his extra help with the typesetting and compiling of all the chapters in both volumes.

G. B. Giannakis, Y. Hua, P. Stoica and L. Tong
May 2000

CONTENTS

Chapter 1

CHANNEL ESTIMATION AND EQUALIZATION USING HIGHER-ORDER STATISTICS

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1.1 Introduction

Two major sources of impairments of digital communications signals as they propagate through channels are multipaths and limited bandwidth. This leads to intersymbol interference (ISI) at the receiver which, in turn, may lead to high error rates in symbol detection. Equalizers are designed to compensate for these channel distortions. One may directly design an equalizer given the received signal, or one may first estimate the channel impulse response and then design an equalizer based on the estimated channel. The received signals are sampled at the baud (symbol) or higher (fractional) rate before processing them for channel estimation and/or equalization. Depending upon the sampling rate, one has either a single-input/single-output (SISO) (baud rate sampling), or a single-input/multiple-output (SIMO) (fractional sampling), complex discrete-time equivalent baseband channel in the single user case, and multiple-input/multiple-output (MIMO), complex discrete-time equivalent baseband channel in the multi-user case.

A training sequence (known to the receiver) may be transmitted during start-up (acquisition mode). In the operational stage, the receivers typically switch to a decision-directed mode where the previously equalized and detected symbols are used as a (pseudo-)training sequence together with the received data to update the channel or the equalizer coefficients. The various issues involved and the trade-offs among various competing approaches (linear, decision-feedback, maximum-likelihood sequence estimation, least mean-square vs. recursive least-squares, baud rate vs. fractional rate, etc.) are fairly well-understood and documented; see the well-known text [12] and references therein. More recently, there has been much interest in blind (self-recovering) channel estimation and blind equalization where no training sequences are available or used. In multipoint networks, whenever a channel from the master to one of the tributary stations goes down, it is clearly not feasible (or desirable) for the master to start sending a training sequence to reboot a particular line. In digital communications over fading/multipath channels, a restart is required following a temporary path loss due to a severe fade. In in-service transmission impairment monitoring, the training sequences are obviously not supplied by the transmitter.

As in the trained case, various approaches to blind channel estimation and equalization have been developed. When sampled at the baud rate, the received signal is discrete-time stationary and typically nonminimum-phase. When sampled at higher than baud rate (typically an integer multiple of baud rate), the signal is discrete-time scalar cyclostationary and equivalently, it may be represented as a discrete-time vector stationary sequence with an underlying SIMO model. With baud rate sampling, one has to exploit the higher-order statistics of the received signal either implicitly (as in [5] and [18] where direct design of equalizers is considered) or explicitly (as in [3], [6], [10], [11], [21] and [25]-[29] where the focus is on first estimating the channel impulse response using higher-order cumulants of the received signal). Higher-order statistics provide an incomplete characterization of the underlying non-Gaussian process. Joint channel and data estimation using maximum-likelihood and related approaches may be found in [9], [17] and references therein where a complete (non-Gaussian) probabilistic characterization of the signal is exploited. Computational complexity of these algorithms (explicit higher-order statistics and joint channel-data estimation) is large when the ISI spans many symbols (as in telephone channels) but they are relatively simple when ISI span is short (as in mobile radio channels). However, they may suffer from local convergence problems.

When there is excess channel bandwidth, baud rate sampling is below the Nyquist rate leading to aliasing and depending upon the symbol timing phase, in certain cases, causing deep spectral notches in sampled, aliased channel transfer function [4]. This renders the equalizer performance quite sensitive to symbol timing errors. Initially, in the trained case, fractional sampling was investigated to robustify the equalizer performance against timing error. More recently, in the blind context, it was discovered (see [24] and references therein) that oversampling provides some new information regarding the channel, which can be exploited for blind channel estimation and equalization as long as some technical conditions are satisfied (the notorious "no common subchannel zeros" condition, also called channel disparity, for the underlying equivalent SIMO model). A similar SIMO model results if multiple sensors are used with or without fractional sampling. The work of [24] has spawned intense research activity in the use of second-order statistics for blind identification and equalization. It should be noted that the requisite technical conditions for applicability of these approaches are not always satisfied in practice; some examples are in [26].

In this chapter, we discuss higher-order statistics based approaches to blind channel estimation and equalization, with focus on estimation of the channel impulse response, for both single-user as well as multi-user systems, and both baud-rate as well as fractional-rate sampling. We consider a baseband equivalent discrete-time model with finite impulse response (although precise knowledge of the channel length is not crucial). Given the mathematical model, there are three broad classes of approaches to channel estimation, the distinguishing feature among then being the choice of the optimization criterion. All of the approaches involve (more or less) a least-squares error measure. The error definition differs, however, as follows:

- *Inverse Filter Error:* Filter (equalize) the data by an inverse filter (inverse to the channel in the noise-free, single-user case) and then minimize (or maximize) some function of the inverse filter output. The approaches of [5], [18] and [19] fall in this category. Typically the cost function is optimized with respect to the inverse filter coefficients (equalizer taps) and the optimized inverse filter output is the equalized output. This class of solutions results in a nonlinear optimization problem. Convergence to a local, non-desirable extremum is a possibility if (practical) finite length equalizers are used although schemes utilizing variable length equalizers exist to alleviate this problem [19]. On the other hand, precise knowledge of the channel length is not needed to apply these approaches. Inverse filter criteria based approaches utilize higher-order statistics implicitly only. They are treated in more detail elsewhere in this book and, therefore, will not be considered in any detail in this chapter except for the multi-user case discussed in Sec. 1.4.

- *Fitting Error:* Match the model-based statistics to the estimated (data-based) statistics in a least-squares sense to estimate the channel impulse response, as in [27], for example. This approach allows consideration of noisy observations. In general, it results in a nonlinear optimization problem. It requires availability of a good initial guess to prevent convergence to a local minimum. It yields estimates of the channel impulse response.

- *Equation Error:* It is based on minimizing an "equation error" in some equation which is satisfied ideally. The approaches of [3], [6], [21], [22] and [23] fall in this category. In general, this class of approaches results in a closed-form solution for the channel impulse response so that a global extremum is always guaranteed provided that the channel length (order) is known. These