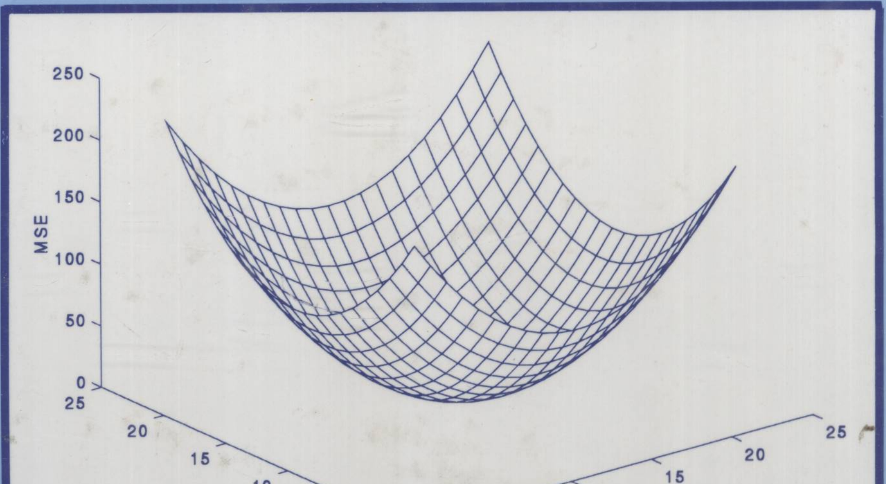
ADAPTIVE FILTERING

Algorithms and Practical Implementation

Paulo S. R. Diniz



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by

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PREFACE

The field of *Digital Signal Processing* has developed so fast in the last two decades that it can be found in the graduate and undergraduate programs of most universities. This development is related to the growing available technologies for implementing digital signal processing algorithms. The tremendous growth of development in the digital signal processing area has turned some of its specialized areas into fields themselves. If accurate information of the signals to be processed is available, the designer can easily choose the most appropriate algorithm to process the signal. When dealing with signals whose statistical properties are unknown, fixed algorithms do not process these signals efficiently. The solution is to use an adaptive filter that automatically changes its characteristics by optimizing the internal parameters. The adaptive filtering algorithms are essential in many statistical signal processing applications.

Although the field of adaptive signal processing has been subject of research for over three decades, it was in the eighties that a major growth occurred in research and applications. Two main reasons can be credited to this growth, the availability of implementation tools and the appearance of early textbooks exposing the subject in an organized form. Presently, there is still a lot of activities going on in the area of adaptive filtering. In spite of that, the theoretical development in the linear-adaptive-filtering area reached a maturity that justifies a text treating the various methods in a unified way, emphasizing the algorithms that work well in practical implementation. This text concentrates on studying on-line algorithms, those whose adaptation occurs whenever a new sample of the environment signals is available. The so-called block algorithms, those whose adaptation occurs when a new block of data is available, are not directly presented here in our view this subject requires a book for itself. Besides, block algorithms require implementation resources that are distinct of the on-line algorithms. The theory of nonlinear adaptive filters based on high-order statistics is probably the most important complement to the subject treated in this book. Although this subject is not treated here, the understanding of the material presented is fundamental for studying this still growing field.

The idea of writing this book started while teaching the adaptive signal processing course at the graduate school of the Federal University of Rio de Janeiro (UFRJ). The request of the students to cover as many algorithms as possible made me think how to organize this subject such that not much time is lost in adapting notations and derivations related to different algorithms. Another common question was which algorithms really work in a finite-precision implementation. These issues made me believe that a new text on this subject could be written with these objectives in mind. Also, considering that most graduate and undergraduate programs include a single adaptive filtering course, this book should not be lengthy. Another objective to seek is to provide an easy access to the working algorithms for the practicing engineer.

It was not until I spent a sabbatical year and a half at University of Victoria, Canada, that this project actually started. In the leisure hours, I slowly started this project. Parts of the early chapters of this book were used in short courses on adaptive signal processing taught in different institutions, namely: Helsinki University of Technology, Espoo, Finland; University Menendez Pelayo in Seville, Spain; and at the Victoria Micronet Center, University of Victoria, Canada. The remaining parts of the book were written based on notes of the graduate course in adaptive signal processing taught at COPPE (the graduate engineering school of UFRJ).

The philosophy of the presentation is to expose the material with a solid theoretical foundation, while avoiding straightforward derivations and repetition. The idea was to keep the text with a manageable size, without sacrificing clarity and without omitting important subjects. Another objective is to bring the reader up to the point where implementation can be tried and research can begin. A number of references are included in the end of the chapters in order to aid the reader to proceed on learning the subject.

It is assumed the reader has previous background on the basic principles of digital signal processing and stochastic processes, including: discrete-time Fourier- and $\mathcal{Z}$-transforms, finite impulse response (FIR) and infinite impulse response (IIR) digital filter realizations, random variables and processes, first- and second-order statistics, moments, and filtering of random signals. Assuming that the reader has this background, I believe the book is self contained.

Chapter 1 introduces the basic concepts of adaptive filtering and sets a general framework that all the methods presented in the following chapters fall under. A brief introduction to the typical application of adaptive filtering is also presented.

In Chapter 2, the basic concepts of discrete-time stochastic processes are reviewed with special emphasis to the results that are useful to analyze the behavior of adaptive filtering algorithms. In addition, the Wiener filter is presented, establishing the optimum linear filter that can be sought in stationary environments. The concept of mean-square error surface is then introduced, another useful tool to analyze adaptive filters. The classical Newton and steepest-descent algorithms are briefly introduced. Since the use of these algorithms would require a complete knowledge of the stochastic environment, the adaptive filtering algorithms introduced in the following chapters come into play. Practical applications of the adaptive filtering algorithms are revisited in more detail at the end of Chapter 2.

Chapter 3 presents the analysis of the LMS algorithm in some depth. Several aspects are discussed, such as convergence behavior in stationary and nonstationary environments, and quantization effects in fixed- and floating-point arithmetics.

Chapter 4 deals with some algorithms that are in a sense related to the LMS algorithm. In particular, the algorithms introduced are the quantized-error algorithms, the LMS-Newton algorithm, the transform-domain algorithm, and the normalized LMS algorithm. Some properties of these algorithms are also discussed in Chapter 4.

Chapter 5 introduces the conventional recursive least-squares (RLS) algorithm. This algorithm minimizes a deterministic objective function, differing in this sense from the LMS-based algorithms. Following the same pattern of presentation of Chapter 3, several aspects of the conventional RLS algorithm are discussed, such as convergence behavior in stationary and nonstationary environments, and quantization effects in fixed- and floating-point arithmetics. The results presented, except for the quantization effects, are also valid to the RLS algorithms presented in the following chapters.

In Chapter 6, a family of fast RLS algorithms based on the FIR lattice realization is introduced. These algorithms represent an interesting alternative to the computationally complex conventional RLS algorithm. In particular, the unnormalized, the normalized and the error-feedback algorithms are presented.

Chapter 7 deals with the fast transversal RLS algorithms, which are very attractive due to their low computational complexity. However, these algorithms are known to face stability problems in practical implementation. As a consequence, special attention is given to the stabilized fast transversal RLS algorithm.

Chapter 8 is devoted to a family of RLS algorithms based on the QR decomposition. The conventional and two fast versions of the QR-based algorithms are presented in this chapter.

Chapter 9 addresses the subject of adaptive filters using IIR digital filter realizations. The chapter includes a discussion of how to compute the gradient and how to derive the adaptive algorithms. The cascade, the parallel, and the lattice realizations are presented as interesting alternative to the direct-form realization for the IIR adaptive filter. The characteristics of the mean-square error surface, for the IIR adaptive filtering case, are also discussed in this chapter. Algorithms based on alternative error formulations, such as the equation-error and Steiglitz-McBride methods are also introduced.

I decided to use some standard examples to present a number of simulation results, in order to test and compare different algorithms. This way a lot of repetition was avoided while allowing the reader to easily compare the performance of the algorithms.

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Prof. Paulo S. R. Diniz

Niterói, Brazil

To: My Parents,
Mariza,
Paula,
and Luiza.

CONTENTS

1

INTRODUCTION TO ADAPTIVE FILTERING

1.1 INTRODUCTION

In this section we define the kind of signal processing systems that will be treated in this text.

In the last thirty years significant contributions have been made in the signal processing field. The advances in digital circuit design have been the key technological development that sparkled a growing interest in the field of digital signal processing. The resulting digital signal processing systems are attractive due to their reliability, accuracy, small physical sizes, and flexibility.

One example of a digital signal processing system is called *filtering*. Filtering is a signal processing operation whose objective is to process a signal in order to manipulate the information contained in the signal. In other words, a filter is a device that maps its input signal in another output signal facilitating the extraction of the desired information contained in the input signal. A digital filter is the one that processes discrete-time signals represented in digital format. For time-invariant filters the internal parameters and the structure of the filter are fixed, and if the filter is linear the output signal is a linear function of the input signal. Once prescribed specifications are given, the design of time-invariant linear filters entails three basic steps, namely: the approximation of the specifications by a rational transfer function, the choice of an appropriate structure defining the algorithm, and the choice of the form of implementation for the algorithm.

An adaptive filter is required when either the fixed specifications are unknown or the specifications cannot be satisfied by time-invariant filters. Strictly speaking an adaptive filter is a nonlinear filter since its characteristics are dependent on the input signal and consequently the homogeneity and additivity conditions are not satisfied. However, if we freeze the filter parameters at a given instant of time, the adaptive filter considered in this text is linear in the sense that its output signal is a linear function of its input signal.

The adaptive filters are time-varying since their parameters are continually changing in order to meet a performance requirement. In this sense, we can interpret an adaptive filter as a filter that performs the approximation step on-line. Usually the definition of the performance criterion requires the existence of a reference signal that is usually hidden in the approximation step of fixed-filter design. This discussion brings the feeling that in the design of fixed (nonadaptive) filters a complete characterization of the input and reference signals is required in order to design the most appropriate filter that meets a prescribed performance. Unfortunately, this is not the usual situation encountered in practice, where the environment is not well defined. The signals that compose the environment are the input and the reference signals, and in cases where any of them is not well defined, the design procedure is to model the signals and subsequently design the filter. This procedure could be costly and difficult to implement on-line. The solution to this problem is to employ an adaptive filter that performs on-line updating of its parameters through a rather simple algorithm, using only the information available in the environment. In other words, the adaptive filter performs a data-driven approximation step.

The subject of this book is adaptive filtering, which concerns the choice of structures and algorithms for a filter that has its parameters (or coefficients) adapted, in order to improve a prescribed performance criterion. The coefficient updating is performed using the information available at a given time.

The development of digital very large scale integration (VLSI) technology allowed the widespread use of adaptive signal processing techniques in a large number of applications. This is the reason why in this book only discrete-time implementations of adaptive filters are considered. Obviously, we assume that continuous-time signals taken from the real world are properly sampled, i.e., they are represented by discrete-time signals with sampling rate higher than twice their highest frequency. Basically, it is assumed that when generating a discrete-time signal by sampling a continuous-time signal, the Nyquist or sampling theorem is satisfied [1]-[8].