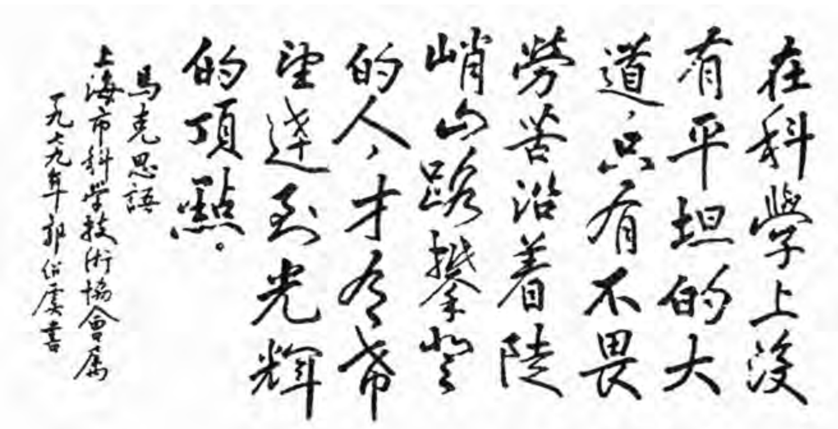
Preface

"There is no royal road to science, and only those who do not dread the fatiguing climb of its steep paths have a chance of gaining its luminous summits."

Karl Marx



The core of this book is taken from a series of lectures I gave in Shanghai in 1983 and 1985. On this occasion, the Chinese Association of Science and Technology (CAST) gave to me a calligraphic version of philosopher Karl Marx's famous quotation. I have always found the above quotation very pertinent and often used it to perk up my students in times of discouragement. I would like to thank Professor Yong Xiang Lu, President of the Chinese Sciences Academy, who encouraged me to use the quotation at the beginning of this book.

The performances of computers in general, and signal processors in particular, have been constantly evolving towards smaller and faster versions with higher and higher storage capabilities. This improved performance allows today's engineer to implement algorithms that are ever more complex[1].

Through this book, we wish to give the reader the significant results hitherto obtained in the field of parametric signal modeling, and to present some new approaches. To the best of our knowledge, these results are dispersed through various textbooks, and there is no single compendium grouping them all together. Moreover, a large part of these results are only presented in journal articles often inaccessible to the average student. This book attempts to fill this gap.

We will mostly consider signal estimation/identification, traditionally grouped in the domain of control engineering. Today, however, this parameter estimation/identification deserves a stature of its own in view of its recent maturity. One example where the deserved importance was given to it is the tri-annual conferences organized by the *International Federation of Automatic Control* (IFAC) and called the "Symposium on Identification and System Parameter Estimation". A recent trend during these conferences has been the increasing importance given to challenges lying in the field of signal processing. Over the past 15 years or so, identification in signal processing has undergone fervent activity, even a renaissance.

One of the fundamental differences between control engineering and signal/image processing comes from the nature of the input signal. In the former, the input is known whereas in the latter, it is always unknown. Several problems specific to signal processing have their roots in this difference.

For example, the rapid development of digital communications has led to a renewed interest in the identification of Single Input Multiple Output (SIMO) and Multiple Input Multiple Output (MIMO) systems. Additionally, equalization and blind-deconvolution issues are also increasingly important. This rise in interest is best attested by the number and quality of related articles in the IEEE Transactions on Signal Processing and the ICASSP conferences.

This book is organized into 9 chapters. Chapter 1 starts with a brief introduction and review of the basic theory of discrete linear models, notably the AR and ARMA models. We then analyze the shortcomings of these models and present an alternative composed of sinusoidal models and ARCOS models to characterize periodic signals.

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1 The term "complex" here includes both the computation cost and the abstraction level involved in the associated mathematical approaches.

Once we have chosen the model and its order, the estimation of its parameters has to be addressed. In Chapter 2, we present the least squares method and its variants in signal processing, namely the autocorrelation method and the covariance method for the AR model. For a non-recursive case, we then define the Normal (or Yule-Walker) equations. Thereafter, we take up the recursive forms of the least squares algorithm and consider the lower-complexity algorithms such as the Levinson and Durbin-Levinson methods. This latter topic also serves as the framework in which we introduce the reflection coefficients and the lattice algorithms. However, as we will note in the appropriate place, the least squares method gives biased estimations for correlated measurements. To get unbiased estimates, we introduce the generalized least squares method and the extended least squares method. Finally, we analyze the effect of an additive white measurement noise on the least squares estimation of AR parameters. We also present a review of existing methods used to compensate for the influence of the measurement noise.

In Chapters 3 to 5, we take up parameter estimation using optimal filters and adaptive filters such as the LMS, RLS and APA. For the discrete-time case, we present the relation linking the Wiener filter to the least squares method. To put R. E. Kalman's contribution into perspective, we first present N. Wiener's original derivation [1] for continuous signals, leading to the Wiener Hopf integral equation. The non-recursive nature of the Wiener filter led Kalman to propose an alternative recursive solution, the Kalman filter [2]. This alternative solution consisted of the transformation of the integral equation into a differential stochastic equation, for which he then found a recursive solution.

In Chapter 5, we derive the Kalman filter using an algebraic approach. Even though this algebraic approach may lack a certain elegance, it is based on fundamental notions of linear algebra. This presentation can form the starting point for the interested reader, leading him to the innovation-based presentation of the Kalman filter presented by Kailath *et al.* in their book *Linear Estimation* [3]. We then present the Extended Kalman Filter (EKF) for nonlinear cases. This extended filter is useful when we have to carry out the joint estimation of the desired signal and the corresponding model parameters associated with it.

The EKF is, however, not the only possible solution for nonlinear estimation cases. The purpose of the following chapters is to present other solutions, treating a case often seen in signal processing: when the covariance matrices of the driving process Q and the noise R are not known *a priori*.

Thus, in Chapter 6, we restate the classic methods proposed by Carew-Belanger and R. K. Mehra in the domain of control in the early 1970s. The use of sub-space approaches for identification frees us from the constraint of having known covariance matrices of Q and R; this allows us to see the signal enhancement problem as a realization issue. We then analyze the relevance of these approaches to enhance a signal.

As the test signal, we choose the speech signal because it combines features such as quasi-periodicity in the case of vowels and randomness for consonants. In addition, approaches such as Linear Predictive Coding (LPC), initially derived for speech signals, found widespread use as a generic technique in many other areas. This is also the case for the wavelets in the area of seismic signals.

Chapter 7 concerns parameter estimation techniques using instrumental variables. They are an alternative to the least squares methods and provide unbiased estimations. Instrumental variables techniques require the formulation of an intermediate matrix which is constructed, for example, using the system input in the case of control. However, information on the input is not available in speech processing, and we thus propose an alternative approach based on two interactive Kalman filters.

Moreover, to use the optimal Kalman filter, we have to rely on a number of assumptions which cannot always be respected in real cases.

The filter known as the H_∞ filter makes it possible to relax these assumptions. More specifically, this concerns the nature of the random processes and the necessity of knowing the variance matrices *a priori*. Thus, Chapter 8 is dedicated to this filter. We first recall the work done and results obtained so far, as concerns significant applications in signal processing, as well as some recent solutions. We then compare the H_2 and H_∞-based approaches in the field of signal enhancement. This comparison will moreover justify our decision to grade in this book the LMS algorithm in the category of optimal filters. For this justification, we use the results obtained at the Stanford school, wherein it was shown that this filter is H_∞-optimal.

To further ease the statistical assumptions, we present particle filtering as an alternative to Kalman filtering in Chapter 9.

The work presented in this book is the result of classes given in several universities and the work carried out by our research group. I have been fortunate to share my wishes with my PhD students and fellow professors. I would like to mention the following people for their contributions in writing this book: Eric Grivel, who has been involved in this adventure since its outset; without his continuous availability, I would not have been able to complete this book; Marcel Gabrea, presently at the Ecole de Technologie Supérieure de Montréal in Canada; and David Labarre to whom I am indebted for the material from his PhD dissertation, which he provided to write Chapters 8 and 9. I would also like to extend my gratitude to the following people for their constructive criticism and

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This book is the first in a new series being launched by ISTE, under my direction, concerning signal processing. The first part of this book is suitable as a textbook for students in the first year of Masters programs. The other chapters are the result of recent work performed on the different aspects of model parameter estimation in realistic scenarios.

Even though this second part is the core of this book, it is accessible to a wide readership, assuming that the fundamental theory is known.

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[1] N. Wiener, *Extrapolation, Interpolation and Smoothing at Stationary Time Series*, Wiley and Sons, New York, 1949.

[2] R. E. Kalman, "A new approach to linear filtering and prediction problems", ASME, Series D, *Journal of Basis Eng*., vol. 82, pages 35-45, March 1960.

[3] T. Kailath, A. Sayed and B. Hassibi, *Linear Estimation*, Prentice Hall, 2000.