Preface

The aim of this book is to explain and present naturally in a didactic manner the principles and methods of signal analysis. It is intended both for students who have no prior knowledge of the topic as well as for those who, having received introductory training, have only retained a disheartening ensemble of mathematical formulas with little or no appreciation for their underlying scientific basis. The goal of the author is to lay the foundations and to develop logically and progressively the mathematical tools in order to associate knowledge, intuition, and understanding. By focusing at every stage of the presentation on the essential aspects, it is then easy to make progress in the establishment of the theory and to build upon those fundamentals to simply expose and derive the most current techniques of signal processing.

A prerequisite is a first-year multivariable calculus course at the university level with the basic concepts used to solve differential equations, perform integration, and solve problems in linear algebra.

Students will come away from the book equipped with the handling of Dirac distribution, integration in the complex plane, applications of linear algebra, and the opportunity to link the abstraction of mathematical formulas with practical applications, to conceive and perform the fundamental operations in deterministic and random signal processing.

The notions and techniques exposed in this book are essential in different engineering fields: telecommunications, teledetection, acoustics, imaging, nondestructive evaluation, and defence.

Such techniques are used in:

Spectral analysis
Design of analog and digital filters
Amplitude and phase modulations in telecommunications
Voice recognition and speech synthesis
Sonar and radar ranging
Signal detection in the presence of noise
Echo cancelation on transmission lines

Noise cancelation, reduction
Data compression by parametric modeling
Data compression by multi-resolution
Seismic exploration
Noise source identification
Noise reduction
Control of antimissile systems (Only first concepts are given here)
Detection of first signs of mechanical failure
Sound analysis, musical instruments, music synthesis
Audio noise reduction

Selected problems, many of them with worked solutions, at the end of each chapter support the content with examples.

To enable the transition to applications, an overview of the MATLAB programming language is given in Appendix 3 along with an example program. As they work through the book, students are strongly recommended to write programs to derive the results presented in the text. They will discover with wonder how the treatments whose notions they had spent several hours, maybe even days, to master, can be performed in a few tens of a second using preprogrammed functions embedded in the language. Browsing through the information given by the help command within MATLAB is a fascinating journey in the signal processing territory.

The first part of the book primarily discusses continuous-time systems and signals because they provide intuitive access to basic concepts. The nature of a signal is inseparable from that of the systems that create or receive it. We first show in Chap. 1 that, for the linear and time-invariant (LTI) systems that are often encountered in physics, the exponential signals e^{st} have a remarkable property: The action of LTI systems on these signals leaves their shape unchanged. Only the amplitude and the temporal location of these signals are affected. The action of an LTI system comes down to the multiplication of the input signals e^{st} by a function, called the transfer function, which depends only on the complex parameter s. This situation is encountered for filtering harmonic signals $e^{j\omega t}$ (monochromatic), which are a special form of exponential signals. The rule is simple: The frequency of the output signal of the LTI system is the same as the frequency of the input signal.

Using the case of R, L, C electrical circuits, a thorough analysis of the first- and second-order systems is given in Chap. 2. It is shown that their properties are completely conditioned by the position of the poles and zeros of their transfer functions in the complex plane. These two filters are the building blocks of the vast majority of filters.

In Chap. 3 on Fourier series, we find the first superposition of elementary signals, i.e., the sum of an exponential at a base frequency and the exponentials whose frequencies are multiples of the base frequency. This type of periodic sum signals is encountered particularly in music. We derive and discuss the rule of decomposition and reconstruction of these signals on the basis of harmonic functions. The theoretical aspects are deepened by the introduction of the concept of

Hilbert space. We define Hermitian operators in this space. We show that the eigenvalues of these operators are real and that the eigenvectors related to two non-equal eigenvalues are orthogonal. We show that the eigenvectors of the operator $i\frac{\mathrm{d}}{\mathrm{d}t}$ have the form $\mathrm{e}^{jn\omega_1 t}$ and may constitute a basis of the Hilbert space L^2 of periodic functions of period T_1. We also encounter the first example of optimal decomposition of a signal by a finite sum of functions. In that case, we show with an example the appearance of the Gibbs phenomenon on the reconstructed signal. In the last part of the chapter, we show the decisive advantage provided by Fourier analysis to characterize the physical properties of a signal.

The Dirac distribution plays an essential role in signal analysis. We define it in Chap. 4 as the infinite sum of monochromatic functions. This definition is best suited to signal theory because it leads naturally to the relationship between the impulse response of a system and its frequency response by Fourier transformation. This transformation is the cornerstone of signal analysis. It decomposes any signal into its monochromatic components as Newton's prism splits light. We simply deduce the response of a system to a signal of any shape.

The theoretical and practical aspects of Fourier transform of analog signals are developed in Chaps. 5–7. Chapter 5 introduces Fourier transform and its close relationship with the LTI systems. It is natural to decompose any signal in a series of harmonic components, to compute the action of the system on each component, and then reconstruct the results of those actions to recover the signal at the output of the system.

The discussion here emphasizes the essential nature of the Fourier integral, a key insight for students and practitioners: The projection of a function on sine functions. Simply put, it measures the proximity of this function with a sine wave according to the frequency of that sinusoid. This understanding then allows us to anticipate the effect of further treatments with a qualitative assessment of the situation.

Chapters 6 and 7 provide a range of detailed formulas and worked examples. It is strongly recommended that the reader work through these examples as an exercise. The ease of calculation that he will thereby acquire will be useful in a range of areas, from causal or analytical signals to modulations and time–frequency analysis, for example.

Chapter 8 is dedicated to the calculation of the impulse response of first- and second-order systems. The integration techniques in the complex plane used in these calculations are detailed in Appendix 1. We show that the causality of the system depends upon the position of the poles of its transfer function in the complex plane.

We explore in Chap. 9 the relationship between the two-sided Laplace transform and the Fourier transform. Attention is given to the domain of definition of the transfer function of a system and the consequence on stability and causality of that system. This property is an educational, striking example of the correspondence of a mathematical expression with a physical property.

Three main types of analog filters, Butterworth, Chebyshev and Bessel, are studied in Chap. 10. Their chief characteristics are given using the results of Chap. 2.

We explain qualitatively the differences in the properties of these three different classes of filters by the relative positions of the poles of their transfer functions in the Laplace plane. The study of these properties based on simple geometric arguments allows a general comprehension of the behavior filters. It is found in a slightly different form in the study of digital filters carried out in Chaps. 14 and 15 in the second part of the book.

In Chap. 11 we study the properties of causal and analytic signals. We demonstrate the formula giving the Fourier transform of the Heaviside function and prove the link by the Hilbert transform between the real part and the imaginary part of a signal in a domain when it is zero for negative values of the variable in the conjugate domain. Causal signals are the natural output of physical systems. In consequence, signal processing deals mainly with causal signals. Analytic signals have zero values at negative frequencies. They are a mathematical trick to allow an easy treatment of signal modulations.

While Fourier analysis is unrivaled to analyze the properties of linear systems and stationary signals, it is insufficient to account in an intelligible manner for the variation of signal properties over time. This is the case when dealing with the localization of echoes in radar or in seismic analysis. We are led to use a short-time Fourier transform and, more generally, to use the methods for time–frequency analysis developed in Chap. 12. A representation of the signals on alternative basis functions localized in time, as in continuous wavelet decomposition and in analysis with filter banks, is developed in this chapter.

Nowadays, signal recording and treatments are mainly digital. For this reason, the second part of the book is devoted to the presentation of digital processing methods. Claude Shannon has proven that we could sample a signal at each tick of a clock without loss of information. One can perfectly reconstruct the signal value at any time from the recorded samples if certain conditions are met. Of course, a condition on the frequency of the clock must be respected: The faster the signal variations are, the more frequent the samples will need to be in order to properly describe these variations, i.e., the greater the clock frequency must be. These notions are presented simply in Chap. 13 by qualitative arguments.

It is thus possible to sample a signal, process it digitally, and reconstruct the resulting processed analog signal, if desired. The prevalence of digital processing today is due to advances in electronics and computer technology, and to the algorithm of fast Fourier transform of Cooley and Tukey which has revolutionized signal processing. Because of this algorithm, it became possible to perform Fourier analysis in real time. It quickly became apparent to users that digital treatments were much more flexible and that they also allow treatment inapplicable in analog. In this second part, in parallel to the presentation of the analog processing, we define the numerical Fourier transform and the z-transform which is analogous to the Laplace transform for time-continuous signals. The eigenfunction z^n of digital LTI systems plays a role similar to the function e^{st} for analog systems developed in Chaps. 14 and 15. We define the digital moving average filters (MA).

Chapter 16 presents the Fourier transform of digital signals. The Shannon aliasing theorem and Shannon–Whittaker sampling theorem are demonstrated. Specific numerical transforms are discussed: the discrete Fourier transform and its use as the algorithm of fast Fourier transform (FFT). Fourier transform of time-limited signals is detailed, and the advantage of apodization windows is highlighted.

We find in Chap. 17 the properties of Autoregressive filters and ARMA filters. The pros and cons of these filters are compared to those of the MA filters encountered in Chap. 14.

Chapter 18 deals with minimum-phase filters and inverse filtering. The decisive advantage of numerical methods is also reflected in the calculation of inverse filters and in the treatment of nonstationary signals. The deconvolution techniques of a signal used in particular for the seismic signals are discussed.

We use the Haar transform as a first step for the description of nonstationary signals processing in Chap. 19. It allows a simple access to the concepts of filter banks and mirror filters. The Le Gall Tabatabai 5-3 filter used in the JPEG-2000 image compression standard is used to illustrate multiresolution methods. It becomes possible to decompose a signal using a simple filtering operation and return exactly to the signal using a second filter. The discrete wavelet transform is discussed using the example of the Daubechies wavelets. Their use is widespread today in signal processing and data compression of sound signals and images. The analogy between the filter bank processing and multiresolution analysis is emphasized.

Chapter 20 treats the parametric modeling of a signal as given by the impulse response of a digital system. The limits of Padé modeling are explained and the advantages of Prony's method are given. Prony's sytem of equations allows, for example, the modeling of a voice signal. It is called Linear Prediction Coding (LPC) in speech analysis. The chapter ends with the important concept of adaptive filters proposed by Widrow, which is a tracking algorithm in the least square sense that is able to subtract a spurious signal from the signal of interest. It provides an efficient noise canceler technique.

The third part of the book is devoted to the presentation of the properties of random signals and their treatments. After a refresher in the essential concepts of statistics on a single random variable and the normal law, Chap. 21 proceeds to an in-depth discussion of the statistics of two random variables.

The treatment of multiple r.v. is found in Chap. 22. The chi-square law used widely in statistics is presented and its use for the test of hypothesis of a probability law is illustrated by the example of testing the central limit theorem. The linear regression of a collection of data is studied by a simple method and by the use of results of linear algebra. We expose the Tikhonov regularization method which greatly improves the results when dealing with noisy data and ill-conditioned matrices. The maximum likelihood method of parameter estimation is discussed in several examples.

In Chap. 23 the correlation of two r.v., the correlation and covariance matrices, are defined. We show the optimality of Karhunen–Loève, principal components development, of a collection of random variables on a deterministic functions basis.

Chapter 24 is dedicated to the analysis of wide sense stationary signals (wss). We study the properties of their correlation functions, coherence, and power spectral densities. Filtering of random, digital, and analog signals is described. We study the role of filtering to improve the signal-to-noise ratio.

Spectral analysis of a random signal is often confronted with the fact that only one record of the signal is available which cannot claim to represent the statistical properties of the signal. However, when a signal is ergodic, it is possible to estimate the spectral properties from a single record using regularization methods. Different estimators of the autocorrelation function, the power spectral density, and methods to reduce the variance of these estimators are studied in Chap. 25.

Chapter 26 is dedicated to the parametric estimation of random signals. The Yule–Walker equations which enable the modeling of a regular process by an ARMA filter are established. Modeling a finite number of data is studied. The methods of extraction of significant components of Capon and Pisarenko are described.

Chapter 27, the final chapter in the book, develops the application of stochastic orthogonality on estimation and optimal filtering of random signals. The concepts have been established by Wiener. We present several Wiener filters for estimation and prediction using FIR, causal, and noncausal filters. In 1960 R. Kalman proposed a recursive algorithm for noise reduction and state system estimation. Its reach is beyond that of Wiener's filter as it is able to deal with nonstationary signals. It has the advantage of being highly computationally efficient which brings the possibility to make real-time estimations. We discuss its principle and provide a simple example of application.

Three appendices are included at the end of the book. The first two contain essential mathematical concepts. Appendix 1 is dedicated to integration in the complex plane and the residue theorem, which are used in the Fourier, Laplace, and z-transforms calculations.

Appendix 2 contains a review of matrices and linear algebra. The concepts discussed in this appendix are essential to the understanding of current digital processing methods.

Appendix 3 is devoted to the description of the MATLAB software and its use in signal analysis programming.

This book is translated, expanded, and updated from a book published in 2012 in French. I took the opportunity, while doing the translation in English, to bring improvements to the initial text and develop some aspects which seemed missing.

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