Introduction

Over the last decade, digital signal processing has matured; thus, digital signal processing techniques have played a key role in the expansion of electronic products for everyday use, especially in the field of audio, image and video processing. Nowadays, digital signal is used in MP3 and DVD players, digital cameras, mobile phones, and also in radar processing, biomedical applications, seismic data processing, etc.

This book aims to be a text book which presents a thorough introduction to digital signal processing featuring the design of digital filters. The purpose of the first part (Chapters 1 to 9) is to initiate the newcomer to digital signal and image processing whereas the second part (Chapters 10 and 11) covers some advanced topics on stability for 2-D filter design. These chapters are written at a level that is suitable for students or for individual study by practicing engineers.

When talking about filtering methods, we refer to techniques to design and synthesize filters with constant filter coefficients. By way of contrast, when dealing with adaptive filters, the filter taps change with time to adjust to the underlying system. These types of filters will not be addressed here, but are presented in various books such as [HAY 96], [SAY 03], [NAJ 06].

Chapter 1 provides an overview of various classes of signals and systems. It discusses the time-domain representations and characterizations of the continuous-time and discrete-time signals.

Chapter 2 details the background for the analysis of discrete-time signals. It mainly deals with the z-transform, its properties and its use for the analysis of linear systems, represented by difference equations.

Chapter 3 is dedicated to the analysis of the frequency properties of signals and systems. The Fourier transform, the discrete Fourier transform (DFT) and the fast Fourier transform (FFT) are introduced along with their properties. In addition, the well-known Shannon sampling theorem is recalled.

As we will see, some of the most popular techniques for digital infinite impulse response (IIR) filter design benefit from results initially developed for analog signals. In order to make the reader's task easy, Chapter 4 is devoted to continuous-time filter design. More particularly, we recall several approximation techniques developed by mathematicians such as Chebyshev or Legendre, who have thus seen their names associated with techniques of filter design.

The following chapters form the core of the book. Chapter 5 deals with the techniques to synthesize finite impulse response (FIR) filters. Unlike IIR filters, these have no equivalent in the continuous-time domain. The so-called windowing method, as a FIR filter design method, is first presented. This also enables us to emphasize the key role played by the windowing in digital signal processing, e.g., for frequency analysis. The Remez algorithm is then detailed.

Chapter 6 concerns IIR filters. The most popular techniques for analog to digital filter conversion, such as the bilinear transform and the impulse invariance method, are presented. As the frequency response of these filters is represented by rational functions, we must tackle the problems of stability induced by the existence of poles of these rational functions.

In Chapter 7, we address the selection of the filter structure and point out its importance for filter implementation. Some problems due to the finite-precision implementation are listed and we provide rules to choose an appropriate structure while implementing filter on fixed point operating devices.

In comparison with many available books dedicated to digital filtering, this title features both 1-D and 2-D systems, and as such covers both signal and image processing. Thus, in Chapters 8 and 9, 2-D filtering is investigated.

Moreover, it is not easy to establish the necessary and sufficient conditions to test the stability of 2-D signals. Therefore, Chapters 10 and 11 are dedicated to the difficult problem of the stability of 2-D digital system, a topic which is still the subject of many works such as [ALA 2003] [SER 06]. Even if these two chapters are not a prerequisite for filter design, they can provide the reader who would like to study the problems of stability in the multi-dimensional case with valuable clarifications. This contribution is another element that makes this book stand out.

The field of digital filtering is often perceived by students as a "patchwork" of formulae and recipes. Indeed, the methods and concepts are based on several specific optimization techniques and mathematical results which are difficult to grasp.

For instance, we have to remember that the so-called Parks-McClellan algorithm proposed in 1972 was first rejected by the reviewers [PAR 72]. This was probably due to the fact that the size of the submitted paper, i.e., 5 pages, did not enable the reviewers to understand every step of the approach [McC 05].

In this book we have tried, at every stage, to justify the necessity of these approaches without recalling all the steps of the derivation of the algorithm. They are described in many articles published during the 1970s in the IEEE periodicals i.e., *Transactions on Acoustics Speech and Signal Processing*, which has since become *Transactions on Signal Processing* and *Transactions on Circuits and Systems*.

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[ALA 2003] ALATA O., NAJIM M., RAMANANJARASOA C. and TURCU F., "Extension of the Schur-Cohn Stability Test for 2-D AR Quarter-Plane Model", *IEEE Trans. on Information Theory*, vol. 49, no. 11, November 2003.

[HAY 96] HAYKIN S., *Adaptive Filter Theory*, 3rd edition, Prentice Hall, 1996.

[McC 05] McCLELLAN J.H. and PARKS W. Th., "A Personal History of the Parks-McClellan Algorithm" *IEEE Signal Processing Magazine*, pp 82-86, March 2005.

[NAJ 06] NAJIM M., *Modélisation, estimation et filtrage optimale en traitement du signal*, forthcoming, 2006, Hermes, Paris.

[PAR 72] PARKS W. Th. and McCLELLAN J.H., "Chebyshev Approximation for Nonrecursive Digital Filters with Linear Phase," *IEEE Trans. Circuit Theory*, vol. CT-19, no. 2, pp 189-194, 1972.

[SAY 03] SAYED A., *Fundamentals of Adaptive Filtering*, Wiley IEEE Press, 2003.

[SER 06] SERBAN I., TURCU F., NAJIM M., "Schur Coefficients in Several Variables", *Journal of Mathematical Analysis and Applications*, vol. 320, issue no. 1, August 2006, pp 293-302.