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Introductory Signal Processing

Roland Priemer

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World Scientific

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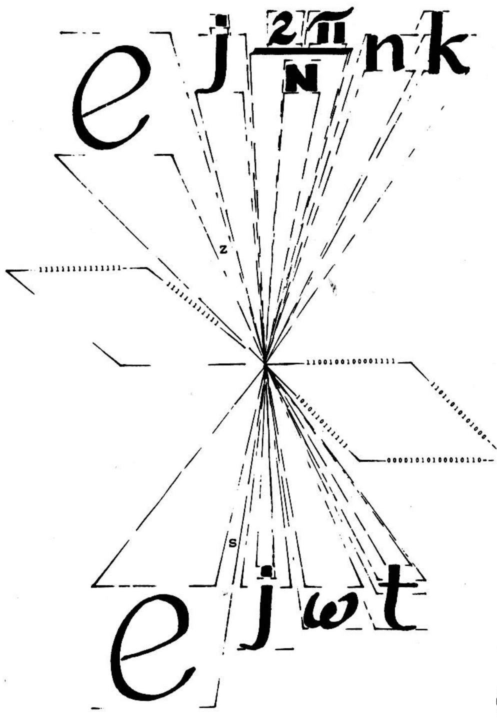

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To my parents, wife, and children

PREFACE

This book is intended to introduce both **time-continuous** and **time-discrete signal processing** to the undergraduate student of engineering and science. It was also designed to meet the needs of Electrical Engineering curricula, which now generally include transform analysis of both continuous and discrete time signals and systems and course work on digital signal processing. With the advent of high technology analog and digital hardware for signal processing, it is now necessary that the practicing engineer be familiar with techniques for time-discrete as well as time-continuous signal processing. A goal of this book is to present signal processing theory so that it can be used for course work in the beginning of an Electrical Engineering major curriculum and serve as a prerequisite for undergraduate course work in areas such computer control theory, digital communication, active RC and switched capacitor filter design, as well as the more traditional topics such as circuit analysis.

Another goal of this book is to develop, from basic principles, methods for time-continuous and time-discrete signal processing and to carefully examine the relationships, differences, and similarities between these methods. This goal is achieved by various means. The material is organized so that as time-continuous methods are presented, corresponding time-discrete methods are also developed. Utilizing the similarities between these methods, insight into the underlying principles can be more readily achieved. Often, simple examples are used to introduce mathematically oriented concepts. Although most of the examples in this book are concerned with electrical systems, some examples utilize systems from other fields. This is done to illustrate the diversity of applications for methods of signal processing. At the end of each chapter there are numerous homework problems with answers to selected problems in an appendix. There are computer projects at the end of most chapters. The intent of these projects is to provide, through guided computer based experimentation, the student reader with a sense of applying the theory. This book strives to point out the few basic principles upon which the theory of signal processing is based.

Prerequisites to the use of this book are a first course on differential equations and a sequence of general physics courses including electricity and magnetism. A suggested corequisite is an introductory course on electric circuits and devices. Aside from these requirements and with the exception of Chapter Eleven, which is concerned with microprocessors for signal processing, this book is self contained.

There is the issue of whether or not to present time-continuous and time-discrete methods separately, and which topic should first be presented. From an instructor's viewpoint, this may be a matter of personal preference. However, a student reader's background consists of real world experience and course work emphasizing continuous time signals and systems as in physics, calculus, and perhaps circuit theory. Therefore, I believe that it is better to build upon this background by developing time-discrete methods based on a foundation of methods for continuous time signal and system analysis. This is the approach taken in this book, where time-discrete methods are applied for the processing of continuous time signals. This approach better reflects engineering practice, where digital hardware, from a special purpose microprocessor to a mainframe computer, is often utilized for the processing of sampled continuous time signals. **Digital signal processing** techniques are widely applied in fields such as process control, bioengineering, speech and image processing, vibration analysis, robotics, seismology, and communications, to mention just a few. Recognizing that the real world is time-continuous, it is essential that the student reader understands the relationships, which are strongly emphasized in this book, between time-discrete methods for signal processing and time-continuous methods.

Emphasis has been placed on a development of **transform analysis**. The fundamental role that the **exponential function** plays in representing signals and in the **analysis of linear and time invariant (LTI) systems** is developed in Chapter One. LTI system concepts such as the **transfer function, frequency response** and **stability** are presented. This chapter concludes by developing the properties of the **impulse function** and **impulse response** of an LTI system, with which the **convolution operation** is introduced. In Chapter Two, the theme of signal representation by exponential functions is continued by deriving the **complex exponential Fourier series** representation of periodic signals. By characterizing a periodic signal in terms of its spectral content, the steady-state response of a time-continuous LTI system to a periodic input is then obtained. In Chapter Four, the notion of representing signals in terms of exponential functions is extended to nonperiodic and stable signals, resulting in the **Fourier transform**, the properties of which are then thoroughly developed. With the Fourier transform, relationships between the time domain and the frequency domain properties of time-continuous signals and LTI systems are investigated. Also, the convolution operation is examined in greater detail than in Chapter One.

In Chapter Eight, the **Laplace transform** is derived from the Fourier transform. With the Laplace transform both stable and unstable time-continuous signals of exponential order can be represented in terms of exponential functions. The material in Chapter Four is then readily extended to the Laplace transform. To facilitate system representation and analysis, **block diagram methods** and **state variable techniques** are also introduced in this chapter.

An important reason for the inclusion in this book of a chapter on the Laplace transform is that by investigating the sampling process with the aid of the Laplace transform, the ***Z*-transform** and inverse transform can be derived. This is done in the beginning of Chapter Nine. Then, the *Z*-transform and its properties are developed and used for the analysis of time-discrete LTI systems, which includes discrete time system properties such as the **transfer function, pulse response, convolution, stability**, and **frequency response**. Block diagram methods and state variable techniques for time-discrete LTI systems are also included in this chapter.

One of the main features of this book is that the complex Fourier series, Fourier transform, Laplace transform and *Z*-transform are developed in an evolutionary manner. These transforms are not merely defined. Instead, each transform is derived from the previous transform. With this approach to transform analysis, the reader will find that the properties of the exponential function and the superposition principle play the key role in the analysis of both time-continuous and time-discrete signals and LTI systems. Furthermore, considerable attention is given to expose the detailed similarities and differences between these transforms. This is done not so much for reasons of mathematical rigor, but rather to provide insight and to avoid unnecessary misconceptions.

The remainder if this book is concerned with discrete time and digital techniques for applying the material presented in Chapters One, Two, Four, Eight, and Nine. The development of these techniques is interlaced with these five chapters. This is done so that both time-continuous and time-discrete methods can be gradually developed in a complementary way.

A **numerical method** for the Fourier series calculations presented in Chapter Two is derived in Chapter Three. The result is the **discrete Fourier transform (DFT)**. Sampling rate and duration requirements and **aliasing and leakage error** due to using the DFT for the analysis of continuous time signals are also investigated in this chapter. Chapter Five more formally presents the discrete Fourier transform and its properties. This chapter starts by examining the sampling process to obtain the **impulse train sampler model** for the operation of sampling a signal. Then, the **sampling theorem** is derived, and the properties of the DFT are further investigated. At the same time, the sampling operation is placed in a practical context with the introduction of **data acquisition and reconstruction devices**. This chapter concludes by applying the DFT to signal spectrum estimation and convolution. The **fast Fourier transform (FFT)**, which is a computationally efficient algorithm for computing the DFT, is presented in Chapter Six.

Chapter Seven is concerned with **correlation analysis** of both continuous and discrete time signals. This important analytical tool is usually included as only an ancillary topic in introductory books on signal processing. Here however, this topic is presented to sufficient depth so that the reader will understand the use of the FFT for correlation function and power spectral density estimation of signals known only over a finite time interval.

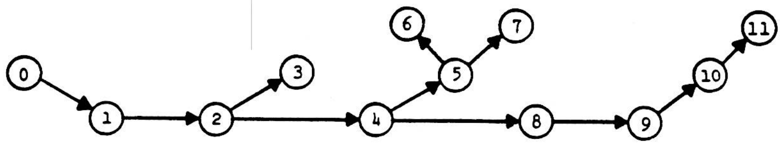
Chapter Ten presents methods for the **design of digital filters**, which are commonly classified according to the time duration of the filter's pulse response. The Fourier series (with and without windowing) and the frequency sampling methods for the design of digital filters, with a pulse response having a finite time duration, are presented first. Such digital filters can be designed to have a perfectly **linear phaseshift characteristic**. For the design of digital filters, which have a pulse response with an infinite time duration, the characteristics of a few analog filter types are examined along with appropriate frequency transformations. Various **discretization methods** are analyzed for their mapping and stability properties, and then several digital filter design techniques are presented and evaluated.

This book concludes with a chapter on **microprocessors for signal processing**. The intention of Chapter Eleven is to introduce the reader to real-time digital signal processing. Due to space limitations, it is not possible to extensively develop the hardware and software aspects of microprocessor based system design. Instead, the emphasis is on real-time processing as demonstrated with example digital systems and programs and results.

Implementing theoretical results with a computer is another feature of this book. Several computer subroutines (written in FORTRAN) for digital signal processing are listed in Appendix B. These subroutines were kept short so that students and instructors will find it convenient to employ them. Examples of their usage are given throughout this book. While FORTRAN is used in this book, the reader who is familiar with the BASIC programming language should have no difficulties in converting these programs into BASIC. From classroom experience, I have found that students gain considerable insight by numerically solving problems to obtain results that support theoretical developments. Besides, students enjoy such work.

The remaining appendices supply supportive material. Appendix A contains a review of complex number arithmetic. The operation of data acquisition and reconstruction devices is described in Appendix C. The basics of matrix algebra are given Appendix D. Appendix E lists a function minimization program, which is used in Chapter Ten for phase equalizer design.

This book can be used for various purposes. It can be used in its entirety as a textbook for a two semester or three quarter course sequence. The chapter dependencies are depicted in the following diagram.



Chapter Flow Diagram

There is considerable flexibility in the use of this book. For a two quarter course sequence, Chapters Six, Seven, and Ten can be omitted without loss of continuity. Or, depending on emphasis, Chapters Eight, Nine, and Ten can be omitted. For a one semester course, Chapters One through Five represent an introduction to continuous and discrete time signal analysis by frequency domain methods. An alternative is to use Chapters One, Two, Four, Eight, and Nine for a course on transform analysis of continuous and discrete time signals and systems, and then the remaining chapters will serve very well for an introductory course on digital signal processing.

Since many examples have been included in this book, it can be used for self-study by those having the required background. The practicing engineer, who is already familiar with the continuous time transform analysis, will find this book to be a useful introduction to digital signal processing. The large number of homework problems with answers in Appendix F and the computer projects will be an aid for attaining an understanding of the material.

Indeed, not one book, or one course, or one engineering curriculum can resolve the myriad of issues that can arise in engineering practice. The intention of this introductory book is to enable the reader to develop an understanding of the fundamentals for signal processing and to become prepared for further studies.

I am grateful to the many students who were instrumental in distilling the ways in which the topics are presented in this book. Also, I wish to express my appreciation to the many people who motivated me to write this book. It was not easy to get started, but then I found working on the manuscript to be a wonderful learning experience. It is difficult to be certain that there are no errors in this book. If there are mistakes, then I would be grateful to be informed about them.

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CHAPTER ZERO

SIGNALS AND SIGNAL PROCESSING

A **signal** is a function that conveys information about the behavior of a system or attributes of some phenomenon. Signals occur naturally and they are also synthesized. The term is used for referring to a wide variety of information. Consider: (1) an acoustic wave, which can convey speech or music information, (2) an electromagnetic wave, which is modulated to transport, for example, image information, (3) the output of a thermocouple, which conveys temperature, or (4) a pH meter's output, which conveys information about the acidity of a solution.

A signal is not necessarily an electrical quantity. However, to perform activities such as synthesizing, transporting, recording, analyzing, and modifying signals, it is often convenient to utilize a signal in the form of an electrical quantity. With electrical systems, a wide variety of signal processing activities can be achieved. For example, a microphone is a transducer of speech information that converts air pressure to a voltage as they both vary with time. Each of these two manifestations of speech information can be referred to with the term signal, and in the form of a time varying voltage signal, the speech information can be transported over electrical conductors, amplified, and converted back to a stronger acoustic wave with a speaker, which is another transducer. Or, the voltage representation for the speech signal, as it varies with time, can be converted into a sequence of numbers by an electronic device called an analog to digital (A/D) converter. The number sequence represents the signal from time to time, and it can be inputted into a computer. Thereby, a computer can be used to analyze speech.

We will consider a **signal** to be some function of an independent variable such as time. Methods for signal processing are not limited by the physical meaning of a signal, i.e., whether it is a microphone output that corresponds to speech information or a thermometer output that corresponds to the temperature of an engine, or signal produced by some other device or process. The mathematical theory for signal processing can be gainfully applied to a diversity of physical signals. Nevertheless, it is essential to understand as much as possible the origin and meaning of a signal, because this provides guidance for properly interpreting and utilizing the results of an analysis of a signal. We will study mathematics that can be applied for the representation (or description) and analysis of signals. One of our goals to quantify useful attributes of a signal.

Basically, **signal processing** is an activity that operates on an input signal to produce an output signal. For example, an amplifier performs a relatively simple signal processing activity. Its purpose is to amplify the power of a signal. In a short while, several examples will be given of some other signal processing activities.

Although we shall classify signals according to numerous properties, at this point let us classify signals into two categories: **time-continuous signals** and **time-discrete signals**. A continuous time signal is defined for all values of the time variable. The term **analog signal** is also used for a continuous time signal. A discrete time signal is only defined at discrete instants of time. The interval between these instants of time can be any time interval, e.g., a microsecond or a year, depending on the signal being considered.