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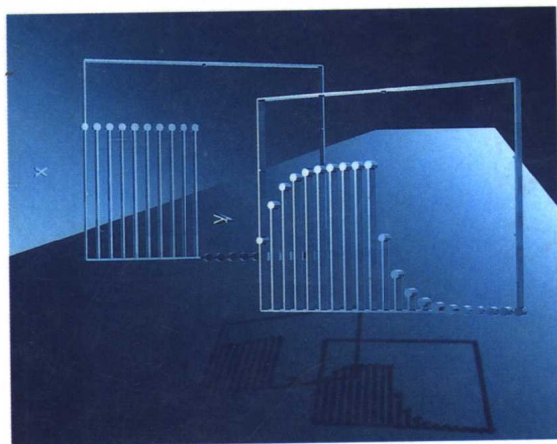
# 信号与系统基础

——应用 Web 和 MATLAB (第二版)

Fundamentals of Signals and Systems

Using the Web and MATLAB (Second Edition)

(英文影印版)



Edward W. Kamen Bonnie S. Heck 著

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## 内 容 简 介

本书为国外高校电子信息类优秀教材(英文影印版)之一。

本书综合介绍了连续时间和离散时间信号系统,并有 Web 演示和 MATLAB 实例贯穿全书各个章节。本书主要内容有由微分及差分方程定义的系统、卷积表示、傅里叶级数及变换、系统的频域分析、通信应用、离散时间信号的分析、数字滤波器和控制器的设计、状态表示等。

本书适用于高等院校电子工程、电子信息、机电类本科生,也可供一般工程技术人员参考。

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# 国外高校电子信息类优秀教材(英文影印版)

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11/34/07

*To our families*

# Preface

This book contains an introductory, yet comprehensive, treatment of continuous-time and discrete-time signals and systems with demos on the Web and MATLAB examples integrated throughout the text. The second edition contains modifications of the material in the first edition to improve the presentation, additional illustrative examples and homework problems, a new chapter on communication systems, and the use of numerous on-line demos that illustrate the concepts and techniques presented in the book. The demos are available at the Web site (<http://users.ece.gatech.edu/~bonnie/book>) that accompanies this text. The chapter on communications covers both analog and digital modulation with an emphasis on the digital case, including phase-shift keying, frequency-shift keying, quadrature amplitude modulation, and on-off keying. The background required for reading the book consists of the usual freshman/sophomore courses in calculus and elementary differential equations. It is also helpful, but not necessary, to have had some exposure to physics. The book is also intended to be used for self-study. Both authors have been teaching the material in the book to electrical engineering juniors for many years, and Bonnie Heck has been actively involved in the use of the Web for enhancing education in the fields of signals, systems, and controls.

A key feature of the text is the use of the on-line demos on the Web. In many of these demos, students may change various values to see what the result is. For example, the frequencies comprising a signal can be changed with the resulting effect on the signal displayed, and the parameters of a system's frequency response function (or transfer function) can be changed with the effect on system perfor-



Mass-  
Spring-  
Damper  
System

mance displayed. In some of the demos, the students can hear the sounds that correspond to the signals being considered. There is also a demo on a mass-spring-damper system that provides an animation of the output response resulting from the application of various inputs. Via this demo, students can actually see the characteristics of the response to an impulsive input, step input, and sinusoidal input. The reference to a demo in the text is given by an icon in the left-hand margin, as illustrated here.

Another key feature of the book is the use of MATLAB (Version 5.0 or higher) to generate computer implementations of the techniques for signal and system analysis and design covered in this work. Along with the on-line demos, the MATLAB implementations provide the reader with the opportunity to verify that the theory does work and allow the reader to experiment with the application of the techniques studied. Use of the various MATLAB commands is illustrated in numerous examples throughout the text. This includes a discussion in Chapter 1 on the use of MATLAB to plot signals. A large number of the homework problems also involve MATLAB. All of the MATLAB programs and M-files that are used in the examples are available from the Web site that accompanies the text.

The book includes a wide range of examples and problems on different areas in engineering including electrical circuits, mechanical systems, and biological systems. Other features of the book are a parallel treatment of continuous-time and discrete-time signals and systems, three chapters on communication systems, digital filtering, and feedback control that prepare students for senior electives in these topics, and a chapter on the state-space formulation of systems.

The book begins with the time-domain aspects of signals and systems in Chapters 1 through 3. This includes the basic properties of signals and systems, the input/output differential equation model, the input/output difference equation model, and the convolution models. Chapter 4 begins the treatment of signals and systems from the frequency-domain standpoint. Starting with signals that are a sum of sinusoids, the presentation then goes into the Fourier series representation of periodic signals and on to the Fourier transform of aperiodic signals. Then, the analysis of continuous-time systems via the Fourier transform is considered in Chapter 5, along with the application to sampling and signal reconstruction. In Chapter 6, the Fourier transform is applied to the study of the transmission and reception of signals in a communication system. In Chapter 7, the Fourier analysis of discrete-time signals and systems is pursued, including a brief treatment of the fast Fourier transform (FFT).

The second half of the book begins in Chapter 8 with the development of the Laplace transform and the transfer function representation for linear time-invariant continuous-time systems. In Chapter 9, the transfer function representation is used to study the response of a system to particular types of inputs, such as the unit-step function and sinusoidal functions. This leads to the notion of the frequency response function first considered in Chapter 5. The transfer function framework is then applied to the problem of control in Chapter 10. In Chapter 11, the z-transform and the transfer function representation of linear time-invariant discrete-time systems is considered, and in Chapter 12, this framework is applied to the design of digital fil-

ters and controllers. Finally, in Chapter 13 the fundamentals of the state description of continuous-time and discrete-time systems are presented.

The book can be used in a two-quarter or two-semester course sequence with Chapters 1 through 7 (or parts of these chapters) covered in one course and Chapters 8 through 13 (or parts thereof) covered in the other. By selecting appropriate sections and chapters from the book, an instructor can cover the continuous-time case in the first course and the discrete-time case in the second course. It is possible to cover the main portion of the material in this book in a one-semester course. A schedule for a one-semester course consisting of forty-two 50-minute lectures is given in the following table:

SCHEDULE OF COVERAGE FOR A ONE-  
SEMESTER COURSE

Text coverage	Number of 50-minute lectures
Chapter 1: All sections	4
Chapter 2: 2.1–2.3	4
Chapter 3: 3.1–3.4	4
Chapter 4: All sections	5
Chapter 5: All sections	5
Chapter 7: 7.1, 7.2, 7.4	4
Chapter 8: 8.1–8.5	6
Chapter 9: 9.1, 9.4, 9.5	4
Chapter 11: All sections	6
	<hr/> 42

The authors wish to thank our colleague Jim McClellan for his thorough review and numerous constructive comments on various drafts of the previous edition of the book. We would also like to thank Gary E. Ford, University of California, Davis, for his comments on the previous edition, and Tom Robbins for his comments on both the previous edition and this edition of the book. We are particularly indebted to Steve McLaughlin and John Barry in our department at Georgia Tech who provided many helpful comments on the material in Chapter 6. The second-named author wishes to thank her former students, John Finney and James Moan, who wrote preliminary versions of the MATLAB tutorial that is available on the web site, and Darren Garner, James Ho, Jason Meeks, Johnny Wang, and Brian Wilson for their efforts in generating the demos that are on the web site.

E.W.K  
B.S.H



# Contents

<b>Preface</b>	<b><i>xi</i></b>
<b>1 FUNDAMENTAL CONCEPTS</b>	<b>1</b>
1.1 Signals and Systems	1
1.2 Continuous-Time Signals	6
1.3 Discrete-Time Signals	17
1.4 Examples of Systems	25
1.5 Basic System Properties	36
Problems	47
<b>2 SYSTEMS DEFINED BY DIFFERENTIAL OR DIFFERENCE EQUATIONS</b>	<b>57</b>
2.1 Linear Input/Output Differential Equations with Constant Coefficients	57
2.2 System Modeling	61
2.3 Linear Input/Output Difference Equations with Constant Coefficients	66

- 2.4 Discretization in Time of Differential Equations 72
- 2.5 Systems Defined by Time-Varying or  
Nonlinear Equations 80
- Problems 88

### **3 CONVOLUTION REPRESENTATION**

**101**

- 3.1 Convolution Representation of Linear Time-Invariant  
Discrete-Time Systems 101
- 3.2 Convolution of Discrete-Time Signals 104
- 3.3 Convolution Representation of Linear Time-Invariant  
Continuous-Time Systems 114
- 3.4 Convolution of Continuous-Time Signals 118
- 3.5 Numerical Convolution 128
- 3.6 Linear Time-Varying Systems 132
- Problems 134

### **4 THE FOURIER SERIES AND FOURIER TRANSFORM**

**145**

- 4.1 Representation of Signals in Terms of Frequency  
Components 146
- 4.2 Fourier Series Representation of Periodic Signals 153
- 4.3 Fourier Transform 161
- 4.4 Properties of the Fourier Transform 175
- 4.5 Generalized Fourier Transform 190
- Problems 193

### **5 FREQUENCY-DOMAIN ANALYSIS OF SYSTEMS**

**202**

- 5.1 Response to a Sinusoidal Input 203
- 5.2 Response to Periodic Inputs 210
- 5.3 Response to Aperiodic Inputs 216
- 5.4 Analysis of Ideal Filters 221
- 5.5 Sampling 231
- Problems 238

<b>6</b>	<b>APPLICATION TO COMMUNICATIONS</b>	<b>251</b>
6.1	Analog Modulation	252
6.2	Demodulation of Analog Signals	260
6.3	Simultaneous Transmission of Signals	264
6.4	Digital Modulation	268
6.5	Baseband PAM	272
6.6	Passband PAM	276
6.7	Digital Communication Simulations	282
	Problems	289
<b>7</b>	<b>FOURIER ANALYSIS OF DISCRETE-TIME SIGNALS AND SYSTEMS</b>	<b>297</b>
7.1	Discrete-Time Fourier Transform	298
7.2	Discrete Fourier Transform	310
7.3	Properties of the DFT	322
7.4	System Analysis via the DTFT and DFT	331
7.5	FFT Algorithm	339
7.6	Applications of the FFT Algorithm	341
	Problems	349
<b>8</b>	<b>THE LAPLACE TRANSFORM AND THE TRANSFER FUNCTION REPRESENTATION</b>	<b>358</b>
8.1	Laplace Transform of a Signal	359
8.2	Properties of the Laplace Transform	363
8.3	Computation of the Inverse Laplace Transform	374
8.4	Transform of the Input/Output Differential Equation	394
8.5	Transfer Function Representation	401
8.6	Transfer Function of Block Diagrams	415
	Problems	423
<b>9</b>	<b>SYSTEM ANALYSIS USING THE TRANSFER FUNCTION REPRESENTATION</b>	<b>435</b>
9.1	Stability and the Impulse Response	436

9.2	Routh–Hurwitz Stability Test	439	
9.3	Analysis of the Step Response	443	
9.4	Response to Sinusoids and Arbitrary Inputs	459	
9.5	Frequency Response Function	465	
9.6	Causal Filters	484	
	Problems	498	
<b>10</b>	<b>APPLICATION TO CONTROL</b>		<b>509</b>
10.1	Introduction to Control	509	
10.2	Tracking Control	516	
10.3	Root Locus	527	
10.4	Application to Control System Design	535	
	Problems	543	
<b>11</b>	<b>THE <math>z</math>-TRANSFORM AND DISCRETE-TIME SYSTEMS</b>		<b>555</b>
11.1	$z$ -Transform of a Discrete-Time Signal	556	
11.2	Properties of the $z$ -Transform	560	
11.3	Computation of the Inverse $z$ -Transform	571	
11.4	Transfer Function Representation	581	
11.5	Stability of Discrete-Time Systems	593	
11.6	Frequency Response of Discrete-Time Systems	597	
	Problems	601	
<b>12</b>	<b>DESIGN OF DIGITAL FILTERS AND CONTROLLERS</b>		<b>612</b>
12.1	Discretization	613	
12.2	Design of IIR Filters	618	
12.3	Design of IIR Filters Using MATLAB	623	
12.4	Design of FIR Filters	630	
12.5	Design of Digital Controllers	643	
	Problems	649	

<b>13 STATE REPRESENTATION</b>	<b>655</b>
13.1 State Model	656
13.2 Construction of State Models	659
13.3 Solution of State Equations	667
13.4 Discrete-Time Systems	676
13.5 Equivalent State Representations	683
13.6 Discretization of State Model	689
Problems	692
<b>APPENDIX A BRIEF REVIEW OF COMPLEX VARIABLES</b>	<b>703</b>
<b>APPENDIX B BRIEF REVIEW OF MATRICES</b>	<b>708</b>
<b>BIBLIOGRAPHY</b>	<b>714</b>
<b>INDEX</b>	<b>717</b>

# Fundamental Concepts

## 1.1 SIGNALS AND SYSTEMS

The concepts of signals and systems arise in virtually all areas of technology, ranging from appliances or devices found in homes to very sophisticated engineering innovations. In fact, it can be argued that much of the development of high technology is a result of advancements in the theory and techniques of signals and systems. This section begins with a brief introduction to the concepts of signals and systems and the closely related field of signal processing. In Sections 1.2 and 1.3 the focus is on various fundamental aspects of continuous-time and discrete-time signals, and in Section 1.4 a number of specific examples of systems are given. Then in Section 1.5, the basic system properties of causality, linearity, finite dimensionality, and time invariance are defined.

### Signals

A signal  $x(t)$  is a *real-valued*, or *scalar-valued*, function of the time variable  $t$ . The term *real valued* means that for any fixed value of the time variable  $t$ , the value of the signal at time  $t$  is a real number. Common examples of signals are voltage or current waveforms in an electrical circuit, audio signals such as speech or music waveforms, bioelectric signals such as an electrocardiogram (ECG) or an electroencephalogram (EEG), forces or torques in a mechanical system, flow rates of liquids or gases in a chemical process, and so on.

Given a signal  $x(t)$  that is very complicated, it is often not possible to determine a mathematical function that is exactly equal to  $x(t)$ . An example is a speech signal, such as the 50-millisecond (ms) segment of speech shown in Figure 1.1. The segment of speech shown in Figure 1.1 is the “sh”-to-“u” transition in the utterance of the word “should.” Due to their complexity, signals such as speech waveforms are usually not specified in function form. Instead, they may be given by a set of sample values. For example, if  $x(t)$  denotes the speech signal in Figure 1.1, the signal can be represented by the set of sample values

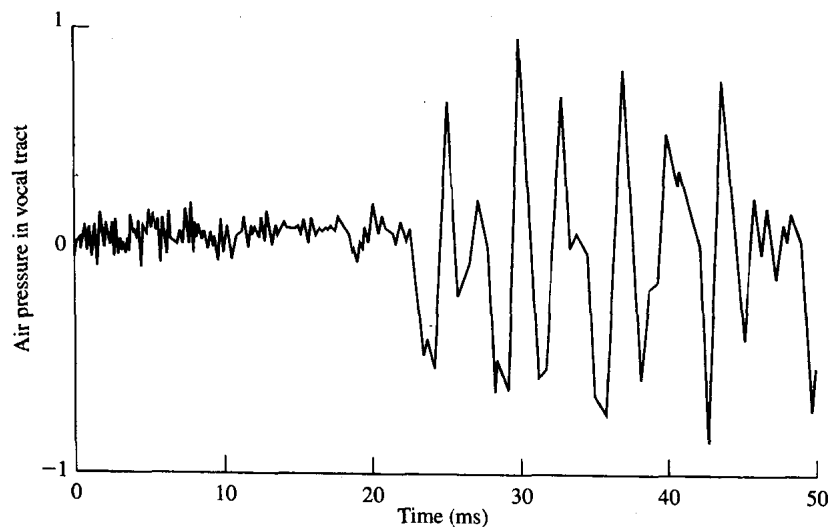
$$\{x(t_0), x(t_1), x(t_2), x(t_3), \dots, x(t_N)\}$$

where  $x(t_i)$  is the value of the signal at time  $t_i$ ,  $i = 0, 1, 2, \dots, N$ , and  $N + 1$  is the number of sample points. This type of signal representation can be generated by sampling the speech signal. Sampling is discussed briefly in Section 1.3 and then is studied in depth in later chapters.

In addition to the representation of a signal in function form or by a set of sample values, signals can also be characterized in terms of their “frequency content” or “frequency spectrum.” The representation of signals in terms of the frequency spectrum is accomplished by using the Fourier transform, which is studied in Chapters 4 to 7.

### Signal Processing

A very important component of technology is signal processing, that is, the processing of signals for various purposes, such as the extraction of the information carried in a signal. Determining the information contained in a signal may not be a simple problem; in particular, knowing the functional form or sample values of a signal does



**Figure 1.1** Segment of speech.

not directly reveal (in general) the information carried in a signal. An interesting example is the extraction of information carried in a speech signal. For example, it is a nontrivial matter to develop a speech processing scheme that is capable of identifying the person who is speaking from a segment of speech. Of course, to be able to identify the speaker correctly, the speech processor must have stored in its memory the “speech patterns” of a collection of people, one of whom is the speaker. The question here is what is an appropriate speech pattern. In other words, exactly what is it about one’s voice that distinguishes it from that of others? One way to answer this is to consider the characterization of speech signals in terms of their frequency spectrum. The concept of the frequency spectrum of a signal is studied in Chapters 4 to 7.

The extraction of information from signals is also of great importance in the medical field in the processing of bioelectric signals. For instance, an important problem is determining the health of a person’s heart from the information contained in a collection of ECG signals taken from surface electrodes placed on the person. A specific objective is to be able to detect if there is any heart damage that may be a result of coronary artery disease or from a prolonged state of hypertension. A trained physician may be able to detect heart disease by “reading” ECG signals, but due to the complexity of these signals, it is not likely that a “human processor” will be able to extract all the information contained in these signals. This is a problem area where signal processing techniques can be applied, and in fact, progress has been made on developing automated processing schemes for bioelectric signals.

Another important problem in signal processing is the reconstruction of signals that have been corrupted by spurious signals or noise. For example, suppose that a sensor provides measurements  $m(t)$  of a signal  $x(t)$  with the measurements given by  $m(t) = x(t) + n(t)$ , where  $n(t)$  is measurement noise or is a term resulting from distortion arising in the sensor. This type of situation arises in target tracking, where  $x(t)$  may be the distance (e.g., range) from some target to a radar antenna, and  $m(t)$  is the measurement of range provided by the radar. Since the energy reflected from a target is so small, radar measurements of a target’s position are always very noisy and usually are embedded in the “background noise.” The problem of determining a “good estimate” of a signal  $x(t)$  from measurements  $m(t) = x(t) + n(t)$  is referred to as an “estimation” or “filtering” problem. In some cases, the estimation of  $x(t)$  can be solved very effectively by considering the frequency spectra of  $x(t)$  and the corrupting signal  $n(t)$ .

It is important to note that estimators and filters, and other types of signal processors, can in fact be viewed as systems, and thus system-theoretic concepts and techniques can be applied to the analysis and design of signal processors. So the field of systems can be motivated in part by applications to signal processing.

## Systems

A system is an interconnection of components (e.g., devices or processes) with terminals or access ports through which matter, energy, or information can be applied or extracted. As illustrated in Figure 1.2, a common way of viewing a system is in



terms of a “black box” with input and output terminals. In the figure,  $x_1(t), x_2(t), \dots, x_p(t)$  are the signals applied to the  $p$  input terminals of the system and  $y_1(t), y_2(t), \dots, y_q(t)$  are the resulting responses appearing at the  $q$  output terminals of the system. In general,  $p$  is not equal to  $q$ ; in other words, the number of input terminals may not equal the number of output terminals. When  $p = q = 1$ , the system is a single-input single-output system. It should be noted that the input and output terminals shown in Figure 1.2 do not include “ground” connections. For example, if the  $x_i(t)$  and the  $y_i(t)$  are voltages relative to ground, the ground is not viewed as a terminal.

In this book the emphasis is on single-input single-output systems. The multi-input multi-output case is considered in Chapter 13 when the state model is presented.

A system may be subjected to different types of inputs. Common examples are control inputs, reference inputs, and disturbance inputs such as noise. Certain types of input signals, such as disturbance inputs, may not be directly measurable. In contrast, the output signals of a system are usually assumed to be measurable using sensing devices.

Some common examples of systems are listed below:

1. An electrical circuit with inputs equal to driving voltages and/or currents and with outputs equal to voltages and/or currents at various points in the circuit.
2. A communications system with inputs equal to the signals to be transmitted and with outputs equal to the received signals.
3. A biological system such as the human heart with inputs equal to the electrical stimuli applied to the heart muscle and with output equal to the flow rate of blood through the heart.
4. A robotic manipulator with inputs equal to the torques applied to the links of the robot and with output equal to the position of the end effector (hand).
5. An oil refinery with input equal to the flow rate of oil and with output equal to the flow rate of gasoline.
6. A manufacturing system with inputs equal to the flow rate of raw materials and with output equal to the rate of production of the finished product.

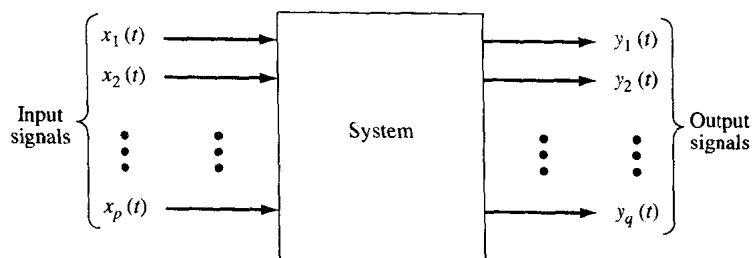


Figure 1.2 System with  $p$  inputs and  $q$  outputs.