Second Edition

Li Tan
Jean Jiang

# DIGITAL SIGNAL PROCESSING

## Fundamentals and Applications

MATLAB® examples

# Digital Signal Processing

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## Fundamentals and Applications

**Second edition**

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# Contents

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# Preface

Technology such as microprocessors, microcontrollers, and digital signal processors have become so advanced that they have had a dramatic impact on the disciplines of electronics engineering, computer engineering, and biomedical engineering. Engineers and technologists need to become familiar with digital signals and systems and basic digital signal processing (DSP) techniques. The objective of this book is to introduce students to the fundamental principles of these subjects and to provide a working knowledge such that they can apply DSP in their engineering careers.

The book is suitable for a two-semester course sequence at the senior level in undergraduate electronics, computer, and biomedical engineering technology programs. Chapters 1 to 8 provide the topics for a one-semester course, and a second course can complete the rest of the chapters. This textbook can also be used in an introductory DSP course in an undergraduate electrical engineering program at traditional colleges. Additionally, the book should be useful as a reference for undergraduate engineering students, science students, and practicing engineers.

The material has been tested for two consecutive courses in a signal processing sequence at Purdue University North Central in Indiana. With the background established from this book, students will be well prepared to move forward to take other upper-level courses that deal with digital signals and systems for communications and control.

The textbook consists of 14 chapters, organized as follows:

- Chapter 1 introduces concepts of DSP and presents a general DSP block diagram. Application examples are included.
- Chapter 2 covers the sampling theorem described in the time domain and frequency domain and also covers signal reconstruction. Some practical considerations for designing analog anti-aliasing lowpass filters and anti-image lowpass filters are included. The chapter ends with a section dealing with analog-to-digital conversion (ADC) and digital-to-analog conversion (DAC), as well as signal quantization and encoding.
- Chapter 3 introduces digital signals, linear time-invariant system concepts, difference equations, and digital convolutions.
- Chapter 4 introduces the discrete Fourier transform (DFT) and digital signal spectral calculations using the DFT. Methods for applying the DFT to estimate the spectra of various signals, including speech, seismic signals, electrocardiography data, and vibration signals, are demonstrated. The chapter ends with a section dedicated to illustrating fast Fourier transform (FFT) algorithms.
- Chapter 5 is devoted to the z-transform and difference equations.
- Chapter 6 covers digital filtering using difference equations, transfer functions, system stability, digital filter frequency responses, and implementation methods such as direct-form I and direct-form II.
- Chapter 7 deals with various methods of finite impulse response (FIR) filter design, including the Fourier transform method for calculating FIR filter coefficients, window method, frequency sampling design, and optimal design. Chapter 7 also includes applications that use FIR filters for noise reduction and digital crossover system design.

- Chapter 8 covers various methods of infinite impulse response (IIR) filter design, including the bilinear transformation (BLT) design, impulse-invariant design, and pole-zero placement design. Applications using IIR filters include audio equalizer design, biomedical signal enhancement, dual-tone multifrequency (DTMF) tone generation, and detection with the Goertzel algorithm.
- Chapter 9 introduces DSP architectures, software and hardware, and fixed-point and floating-point implementations of digital filters.
- Chapter 10 covers adaptive filters with applications such as noise cancellation, system modeling, line enhancement, cancellation of periodic interferences, echo cancellation, and 60-Hz interference cancellation in biomedical signals.
- Chapter 11 is devoted to speech quantization and compression, including pulse code modulation (PCM) coding, mu-law compression, adaptive differential pulse code modulation (ADPCM) coding, windowed modified discrete cosine transform (W-MDCT) coding, and MPEG audio format, specifically MP3 (MPEG-1, layer 3).
- Chapter 12 covers topics pertaining to multirate DSP and applications, as well as principles of oversampling ADC, such as sigma-delta modulation. Undersampling for bandpass signals is also examined.
- Chapter 13 introduces a subband coding system and its implementation. Perfect reconstruction conditions for a two-band system are derived. Subband coding with an application of data compression is demonstrated. Furthermore, the chapter covers the discrete wavelet transform (DWT) with applications to signal coding and denoising.
- Finally, Chapter 14 covers image enhancement using histogram equalization and filtering methods, including edge detection. The chapter also explores pseudo-color image generation and detection, two-dimensional spectra, JPEG compression using DCT, image coding using the DWT, and the mixing of two images to create a video sequence. Finally, motion compensation of the video sequence is explored, which is a key element of video compression used in MPEG.

MATLAB programs are listed whenever they are possible. Therefore, a MATLAB tutorial should be given to students who are new to the MATLAB environment.

- Appendix A serves as a MATLAB tutorial.
- Appendix B reviews key fundamentals of analog signal processing. Topics include Fourier series, Fourier transform, Laplace transform, and analog system basics.
- Appendixes C, D, and E review Butterworth and Chebyshev filters, sinusoidal steady-state responses in digital filters, and derivation of the FIR filter design equation via the frequency sampling method, respectively.
- Appendix F details the derivations of wavelet analysis and synthesis equations.
- Appendix G offers general useful mathematical formulas.

In this new edition, MATLAB projects dealing with practical applications are included in Chapters 2, 4, 6, 7, 8, 10, 12, and 13.

Instructor support, including solutions, can be found at http://textbooks.elsevier.com. MATLAB programs and exercises for students, plus Real-time C programs can be found at booksite.elsevier.com/9780124158931.

Thanks to all the faculty and staff at Purdue University North Central in Westville, Indiana, for their encouragement. In particular, the authors wish to thank Professors Thomas F. Brady, Larryl Matthews, Christopher J. Smith, Alain Togbe, Edward Vavrek, Nuri Zeytinoglu, and Shengyong Zhang for their support and suggestions. We are also indebted to all former students in our DSP classes at Purdue University North Central for their feedback over the years, which helped refine this edition.

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The book has benefited from many constructive comments and suggestions from the following reviewers and anonymous reviewers. The authors take this opportunity to thank them for their significant contributions. We would like to thank the following reviewers for the second edition:

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CHAPTER

# Introduction to Digital Signal Processing

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## CHAPTER OUTLINE



## OBJECTIVES:

This chapter introduces concepts of digital signal processing (DSP) and reviews an overall picture of its applications. Illustrative application examples include digital noise filtering, signal frequency analysis, speech and audio compression, biomedical signal processing such as interference cancellation in electrocardiography, compact-disc recording, and image enhancement.

## 1.1 BASIC CONCEPTS OF DIGITAL SIGNAL PROCESSING

Digital signal processing (DSP) technology and its advancements have dramatically impacted our modern society everywhere. Without DSP, we would not have digital/Internet audio and video; digital recording; CD, DVD, and MP3 players; iPhone and iPad; digital cameras; digital and cellular telephones; digital satellite and TV; or wired and wireless networks. Medical instruments would be less efficient or unable to provide useful information for precise diagnoses if there were no digital electrocardiography (ECG) analyzers, digital X-rays, and medical image systems. We would also live in many less efficient ways, since we would not be equipped with voice recognition systems, speech synthesis systems, and image and video editing systems. Without DSP, scientists, engineers, and technologists would have no powerful tools to analyze and visualize the data necessary for their designs, and so on.

Digital Signal Processing. http://dx.doi.org/10.1016/B978-0-12-415893-1.00001-9

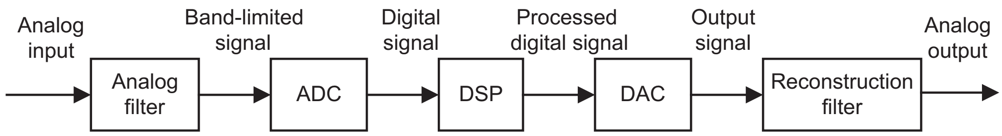


**FIGURE 1.1**

A digital signal processing scheme.

The basic concept of DSP is illustrated by the simplified block diagram in Figure 1.1, which consists of an analog filter, an analog-to-digital conversion (ADC) unit, a digital signal (DS) processor, a digital-to-analog conversion (DAC) unit, and a reconstruction (anti-image) filter.

As shown in the diagram, the analog input signal, which is continuous in time and amplitude, is generally encountered in the world around us. Examples of such analog signals include current, voltage, temperature, pressure, and light intensity. Usually a transducer (sensor) is used to convert the nonelectrical signal to the analog electrical signal (voltage). This analog signal is fed to an analog filter, which is applied to limit the frequency range of analog signals prior to the sampling process. The purpose of filtering is to significantly attenuate *aliasing distortion*, which will be explained in the next chapter. The band-limited signal at the output of the analog filter is then sampled and converted via the ADC unit into the digital signal, which is discrete both in time and in amplitude. The DS processor then accepts the digital signal and processes the digital data according to DSP rules such as lowpass, highpass, and bandpass digital filtering, or other algorithms for different applications. Notice that the DS processor unit is a special type of digital computer and can be a general-purpose digital computer, a microprocessor, or an advanced microcontroller; furthermore, DSP rules can be implemented using software in general.

With the DS processor and corresponding software, a processed digital output signal is generated. This signal behaves in a manner according to the specific algorithm used. The next block in Figure 1.1, the DAC unit, converts the processed digital signal to an analog output signal. As shown, the signal is continuous in time and discrete in amplitude (usually a sample-and-hold signal, to be discussed in Chapter 2). The final block in Figure 1.1 is designated as a function to smooth the DAC output voltage levels back to the analog signal via a reconstruction (anti-image) filter for real-world applications.

In general, the analog signal process does not require software, an algorithm, ADC, and DAC. The processing relies wholly on the electrical and electronic devices such as resistors, capacitors, transistors, operational amplifiers, and integrated circuits (ICs).

DSP systems, on the other hand, use software, digital processing, and algorithms; thus they have a great deal of flexibility, less noise interference, and no signal distortion in various applications. However, as shown in Figure 1.1, DSP systems still require minimum analog processing such as the anti-aliasing and reconstruction filters, which are musts for converting real-world information into digital form and digital signals back into real-world information.

Note that there are many real-world DSP applications that do not require DAC, such as data acquisition and digital information display, speech recognition, data encoding, and so on. Similarly, DSP applications that need no ADC include CD players, text-to-speech synthesis, and digital tone generators, among others. We will review some of them in the following sections.

## 1.2 BASIC DIGITAL SIGNAL PROCESSING EXAMPLES IN BLOCK DIAGRAMS

We first look at digital noise filtering and signal frequency analysis, using block diagrams.

### 1.2.1 Digital Filtering

Let us consider the situation shown in Figure 1.2, depicting a digitized noisy signal obtained from digitizing analog voltages (sensor output) containing a useful low-frequency signal and noise that occupies all of the frequency range. After ADC, the digitized noisy signal $x(n)$, where $n$ is the sample number, can be enhanced using digital filtering.

Since our useful signal contains the low-frequency component, the high-frequency components above that of our useful signal are considered noise, which can be removed by using a digital lowpass filter. We set up the DSP block in Figure 1.2 to operate as a simple digital lowpass filter. After processing the digitized noisy signal $x(n)$, the digital lowpass filter produces a clean digital signal $y(n)$. We can apply the cleaned signal $y(n)$ to another DSP algorithm for a different application or convert it to the analog signal via DAC and the reconstruction filter.

The digitized noisy signal and clean digital signal, respectively, are plotted in Figure 1.3, where the top plot shows the digitized noisy signal, while the bottom plot demonstrates the clean digital signal obtained by applying the digital lowpass filter. Typical applications of noise filtering include acquisition of clean digital audio and biomedical signals and enhancement of speech recording, among others (Embree, 1995; Rabinar and Schafer, 1978; Webster, 1998).


**FIGURE 1.2**

The simple digital filtering block.

### 1.2.2 Signal Frequency (Spectrum) Analysis

As shown in Figure 1.4, certain DSP applications often require that time domain information and the frequency content of the signal be analyzed. Figure 1.5 shows a digitized audio signal and its calculated signal spectrum (frequency content), that is, the signal amplitude versus its corresponding frequency for the time being, obtained from a DSP algorithm, called the *fast Fourier transform* (FFT), which will be studied in Chapter 4. The plot in Figure 1.5(a) is a time domain display of the recorded audio signal with a frequency of 1,000 Hz sampled at 16,000 samples per second, while the frequency content display of plot (b) displays the calculated signal spectrum versus frequency, in which the peak amplitude is clearly located at 1,000 Hz. Plot (c) shows a time domain display of an audio signal consisting of one signal of 1,000 Hz and another of 3,000 Hz sampled at 16,000 samples per second. The frequency content display shown in plot (d) gives two locations (1,000 Hz and 3,000 Hz) where the peak amplitudes reside, hence the frequency content display presents clear frequency information of the recorded audio signal.

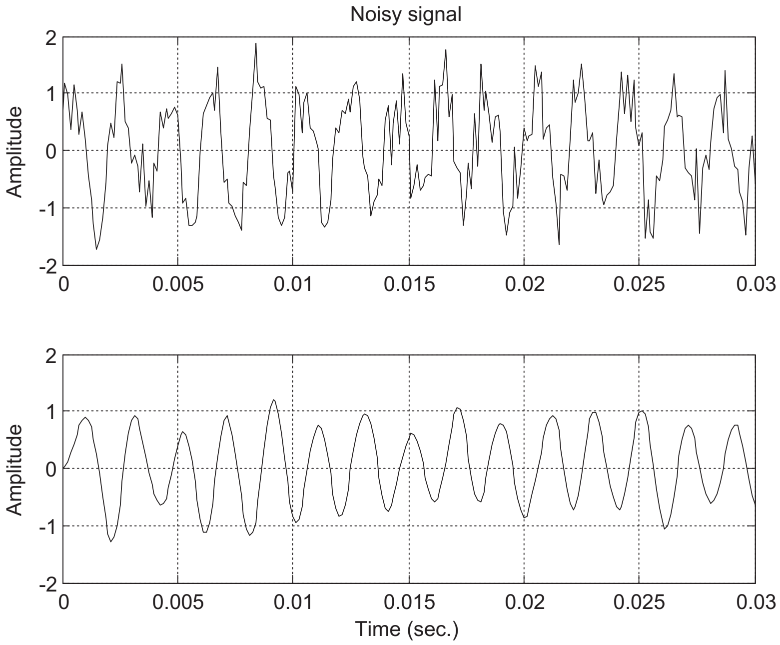


**FIGURE 1.3**

(Top) Digitized noisy signal. (Bottom) Clean digital signal using the digital lowpass filter.

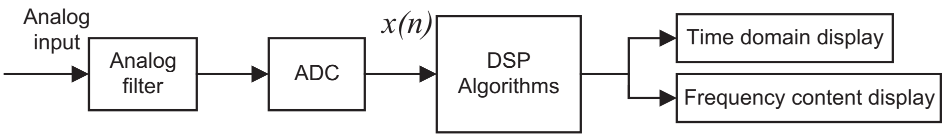


**FIGURE 1.4**

Signal spectral analysis.

As another practical example, we often perform spectral estimation of a digitally recorded speech or audio (music) waveform using the FFT algorithm in order to investigate spectral frequency details of speech information. Figure 1.6 shows a speech signal produced by a human in the time domain and frequency content displays. The top plot shows the digital speech waveform versus its digitized sample number, while the bottom plot shows the frequency content information of speech for a range from 0 to 4,000 Hz. We can observe that there are about ten spectral peaks, called *speech formants,* in the range between 0 and 1,500 Hz. Those identified speech formants can be used for applications such as speech modeling, speech coding, speech feature extraction for speech synthesis and recognition, and so on (Deller et al., 1993).

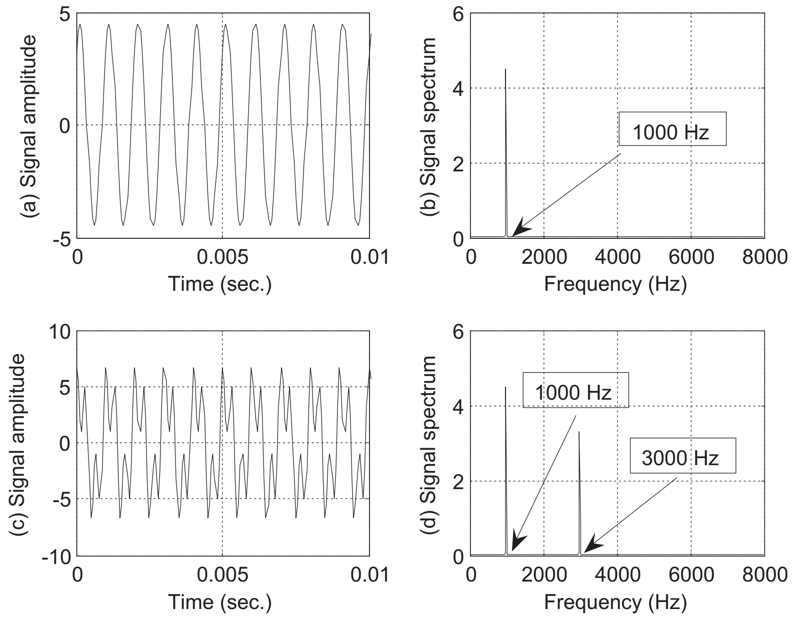


**FIGURE 1.5**

Audio signals and their spectrums.

## 1.3 OVERVIEW OF TYPICAL DIGITAL SIGNAL PROCESSING IN REAL-WORLD APPLICATIONS

### 1.3.1 Digital Crossover Audio System

An audio system is required to operate in an entire audible range of frequencies, which may be beyond the capability of any single speaker driver. Several drivers, such as the speaker cones and horns, each covering a different frequency range, are used to cover the full audio frequency range.

Figure 1.7 shows a typical two-band digital crossover system consisting of two speaker drivers: a woofer and a tweeter. The woofer responds to low frequencies, while the tweeter responds to high frequencies. The incoming digital audio signal is split into two bands by using a digital lowpass filter and a digital highpass filter in parallel. Then the separated audio signals are amplified. Finally, they are sent to their corresponding speaker drivers. Although the traditional crossover systems are designed using the analog circuits, the digital crossover system offers a cost-effective solution with programmability, flexibility, and high quality. This topic is taken up in Chapter 7.

### 1.3.2 Interference Cancellation in Electrocardiography

In ECG recording, there often is unwanted 60-Hz interference in the recorded data (Webster, 1998). The analysis shows that the interference comes from the power line and includes magnetic induction,

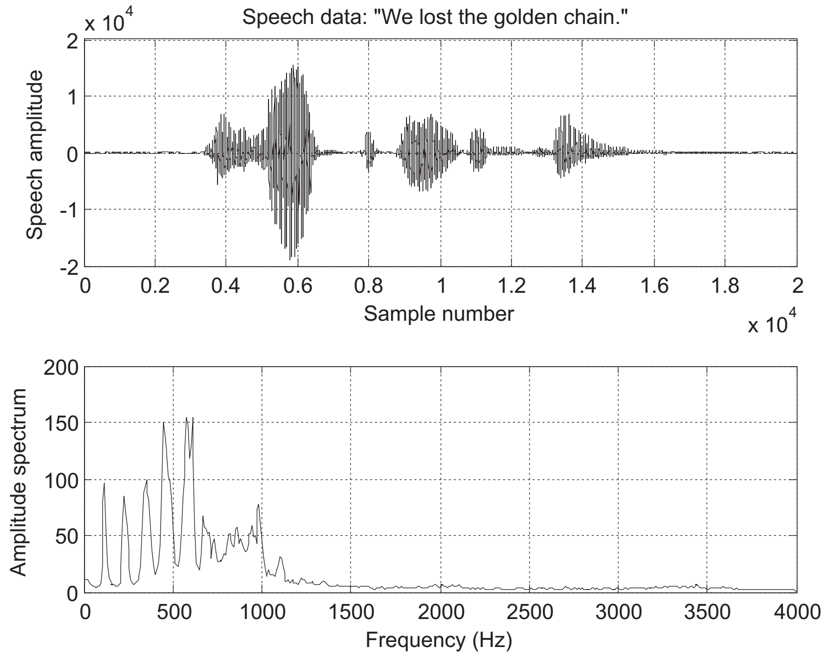


**FIGURE 1.6**

Speech samples and speech spectrum.

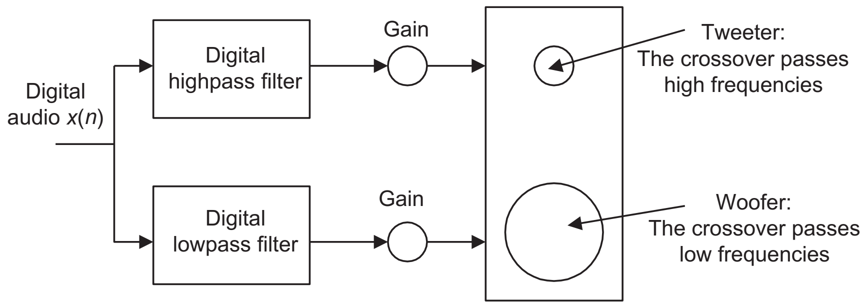


**FIGURE 1.7**

Two-band digital crossover.

displacement currents in leads or in the body of the patient, effects from equipment interconnections, and other imperfections. Although using proper grounding or twisted pairs minimizes such 60-Hz effects, another effective choice can be use of a digital notch filter, which eliminates the 60-Hz interference while keeping all the other useful information. Figure 1.8 illustrates a 60-Hz interference eliminator using a digital notch filter. With such enhanced ECG recording, doctors in clinics could give accurate diagnoses for patients.

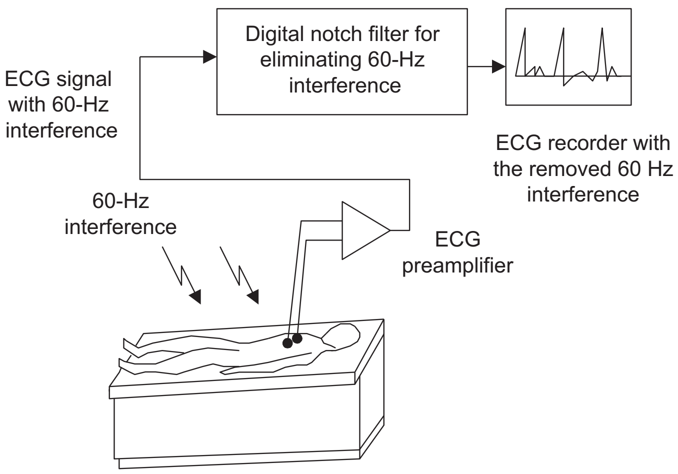


**FIGURE 1.8**

Elimination of 60-Hz interference in electrocardiography (ECG).

This technique can also be applied to remove 60-Hz interference in audio systems. This topic is explored in depth in Chapter 8.

### 1.3.3 Speech Coding and Compression

One of the speech coding methods, called *waveform coding*, is depicted in Figure 1.9A, describing the encoding process, while Figure 1.9B shows the decoding processing. As shown in Figure 1.9A, the analog signal is first sent through an analog lowpass filter to remove high frequency noise components and is then passed through the ADC unit, where the digital values at sampling instants are captured by the DS processor. Next, the captured data are compressed using data compression rules to reduce the storage requirements. Finally, the compressed digital information is sent to storage media. The compressed digital information can also be transmitted efficiently, since compression reduces the original data rate. Digital voice recorders, digital audio recorders, and MP3 players are products that use compression techniques (Deller et al., 1993; Li and Drew, 2004; Pan 1985).

To retrieve the information, the reverse process is applied. As shown in Figure 1.9B, the DS processor decompresses the data from the storage media and sends the recovered digital data to DAC. The analog output is acquired by filtering the DAC output via the reconstruction filter.

### 1.3.4 Compact-Disc Recording System

A compact-disc (CD) recording system is described in Figure 1.10A. The analog audio signal is sensed from each microphone and then fed to the anti-aliasing lowpass filter. Each filtered audio signal is sampled at the industry standard rate of 44.1 kilo-samples per second, quantized, and coded to 16 bits for each digital sample in each channel. The two channels are further multiplexed and encoded, and extra bits are added to provide information such as playing time and track number for the listener. The encoded sample content of Digital Signal Processing: Fundamentals and Applications

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