CHAPTER

INTRODUCTION TO DIGITAL SIGNAL PROCESSING

1

CHAPTER OUTLINE

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 or video; digital recording; CD, DVD, MP3 players, iPhone, and iPad; digital cameras; digital and cellular telephones; digital satellite and TV; or wire and wireless networks. Medical instruments would be less efficient. It would be impossible to provide precise diagnoses if there were no digital electrocardiography (ECG), or digital radiography and other medical imaging modalities. We would also live in many different 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 data and perform their design, and so on.

The concept of DSP is illustrated by the simplified block diagram in Fig. 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.

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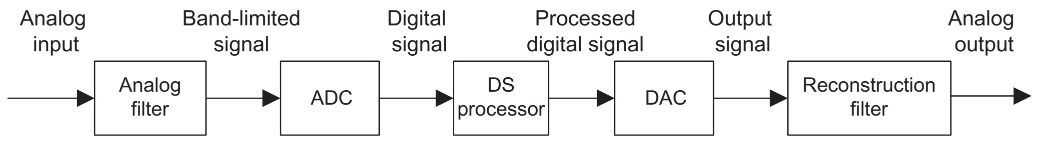

FIG. 1.1

A digital signal processing scheme.

As shown in the diagram, the analog input signal, which is continuous in time and amplitude, is generally encountered in our real life. 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 Chapter 2. 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 digital signal 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. Note that the digital signal processor unit is a special type of a 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 digital signal processor and corresponding software, a processed digital output signal is generated. This signal behaves in a manner based on the specific algorithm used. The next block in Fig. 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 Fig. 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 the real-world applications.

In general, analog signal processing does not require software, algorithm, ADC, and DAC. The processing relies entirely 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; therefore, they have more flexibility, less noise interference, and no signal distortion in various applications. However, as shown in Fig. 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 to digital form and back again to real-world information.

Note that there are many real-world DSP applications that do not require DAC, such as the 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 Fig. 1.2, depicting a digitized noisy signal obtained from digitizing analog voltages (sensor output) containing useful low-frequency signal and noise that occupy 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 low-frequency components, the high-frequency components above the cutoff frequency of our useful signal are considered as noise, which can be removed by using a digital lowpass filter. We set up the DSP block in Fig. 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 analog signal via DAC and the reconstruction filter.

The digitized noisy signal and clean digital signal, respectively, are plotted in Fig. 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 signal and enhancement of speech recording and others (Embree, 1995; Rabiner and Schafer, 1978; Webster, 2009).

FIG. 1.2

The simple digital filtering block.

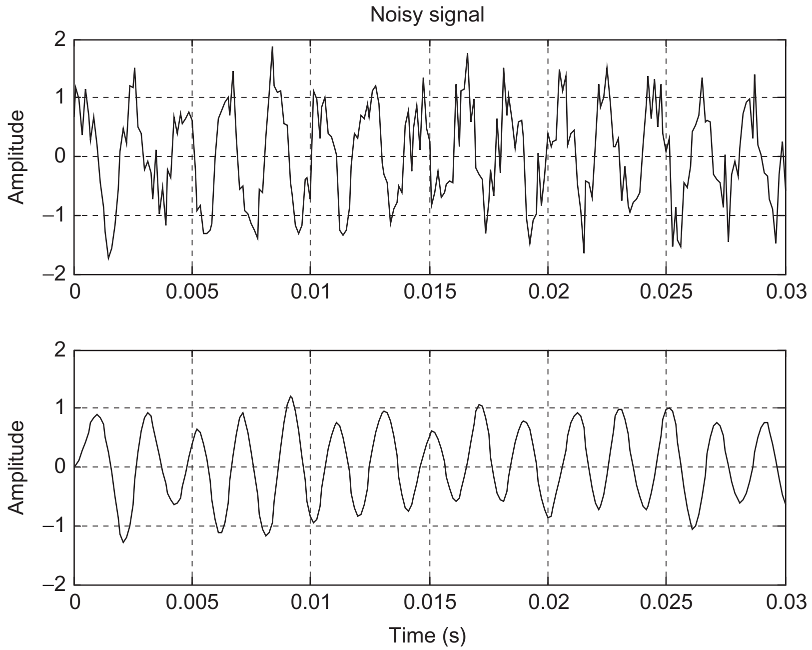

FIG. 1.3

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

1.2.2 SIGNAL FREQUENCY (SPECTRUM) ANALYSIS

As shown in Fig. 1.4, certain DSP applications often require that time domain information and the frequency content of the signal be analyzed. Fig. 1.5 shows a digitized audio signal and its calculated signal spectrum (frequency content), defined as the signal amplitude vs. its corresponding frequency for the time being via a DSP algorithm, called *fast Fourier transform* (FFT), which will be studied in Chapter 4. The plot in Fig. 1.5A is the time domain display of a recorded audio signal with a frequency of 1000 Hz sampled at 16,000 samples per second, while the frequency content display of plot (B) displays the calculated signal spectrum vs. frequencies, in which the peak amplitude is clearly located at 1000 Hz. Plot (C) shows a time domain display of an audio signal consisting of one signal

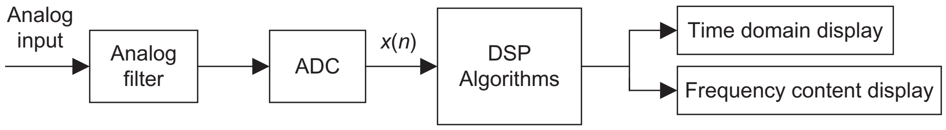

FIG. 1.4

Signal spectral analysis.

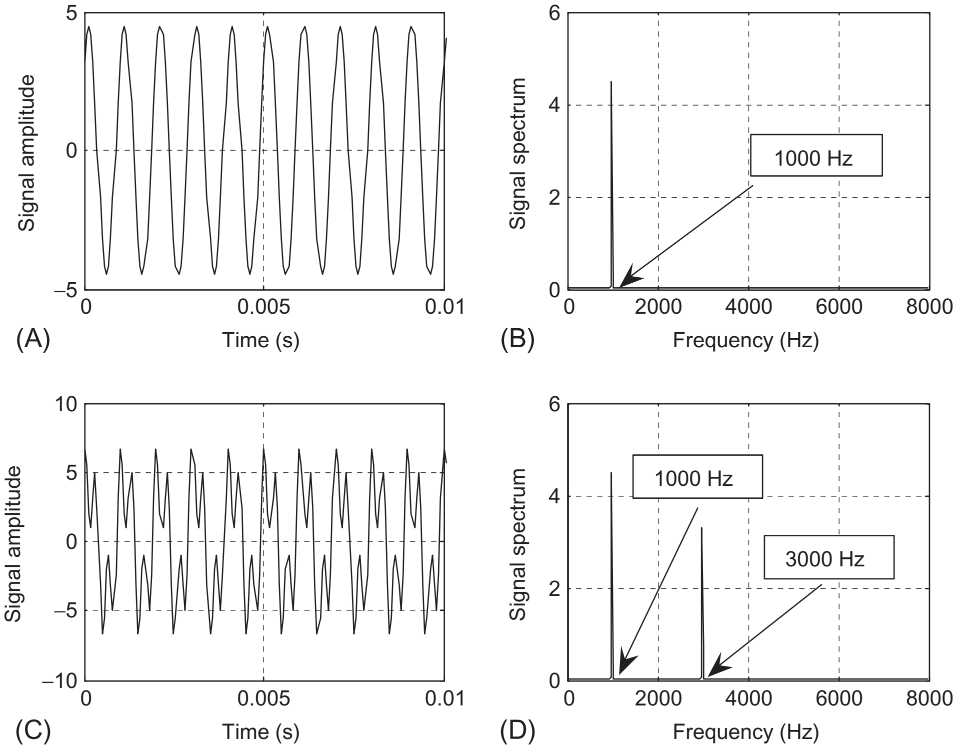

FIG. 1.5

Audio signals and their spectra. (A) 1000 Hz audio signal. (B) 1000 Hz audio signal spectrum. (C) Audio signal containing 1000 and 3000 Hz frequency components. (D) Audio signal spectrum containing 1000 and 3000 Hz frequency components.

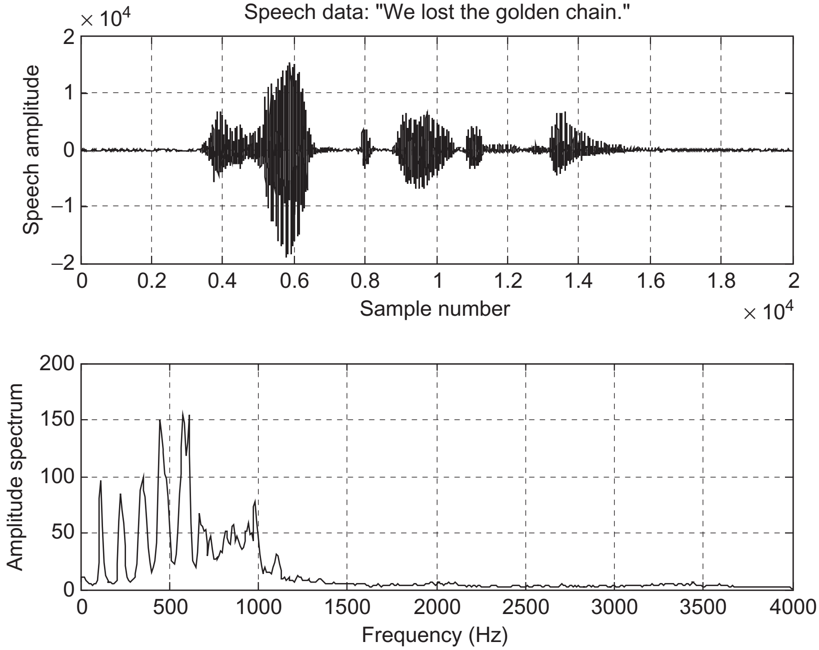

FIG. 1.6

Speech samples and speech spectrum.

of 1000 Hz and another of 3000 Hz sampled at 16,000 samples per second. The frequency content display shown in plot (D) gives two locations (1000 and 3000 Hz) where the peak amplitudes reside, hence the frequency content display presents clear frequency information of the recorded audio signal.

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 the spectral frequency details of speech information. Fig. 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 vs. its digitized sample number, while the bottom plot shows the frequency content information of speech for a range from 0 to 4000 Hz. We can observe that there are about 10 spectral peaks, called *speech formants,* in the range between 0 and 1500 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., 1999).

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, can be used to cover the full audio frequency range.

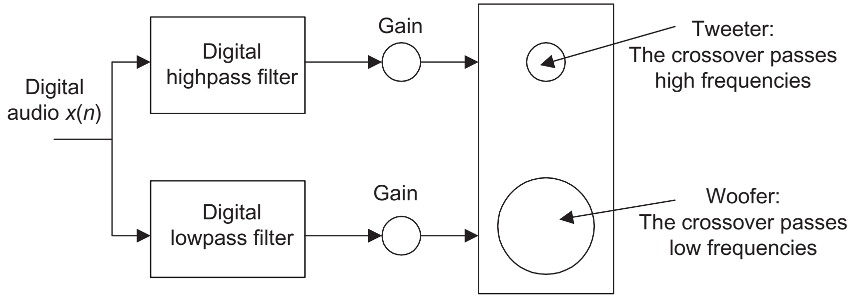

FIG. 1.7

Two-band digital crossover.

Fig. 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 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 programmable ability, flexibility, and high quality. This topic is taken up in Chapter 7.

1.3.2 INTERFERENCE CANCELLATION IN ELECTROCARDIOGRAPHY

In ECG recording, there often exists unwanted 60-Hz interference in the recorded data (Webster, 2009). The analysis shows that the interference comes from the power line and includes magnetic induction, 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 the use of a digital notch filter, which eliminates the 60-Hz interference while keeping all the other useful information. Fig. 1.8 illustrates a 60-Hz interference eliminator using a digital notch filter. As shown in Fig. 1.8, the acquired ECG signal containing the 60-Hz interference passes through the digital notch filter. The digital notch filter eliminates the 60-Hz interference and only outputs the clean ECG signal. With such enhanced ECG recording, doctors in clinics could provide accurate diagnoses for patients. This technique can also be used 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 Fig. 1.9 (top plot), describing the encoding process, while Fig. 1.9 (bottom plot) shows the decoding processing. As shown in Fig. 1.9 (top plot), the analog signal is first filtered by 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 digital signal processor. Next, the captured data are compressed using data compression rules to reduce the storage requirement. Finally, the compressed digital information is sent to storage

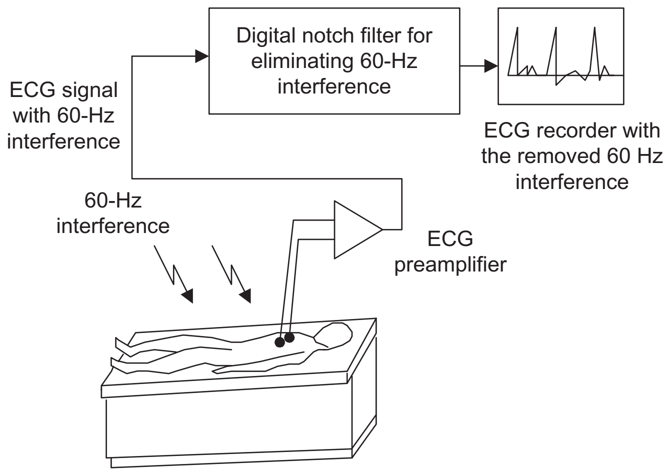

FIG. 1.8

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

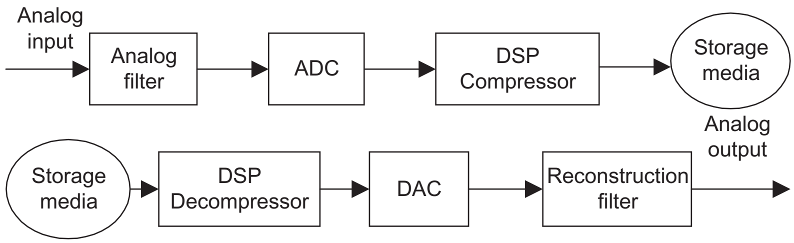

FIG. 1.9

(Top plot) Simplified data compressor. (Bottom plot) Simplified data expander (decompressor).

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., 1999; Li et al., 2014; Pan, 1995).

To retrieve the information, the reverse process is applied. As shown in Fig. 1.9 (bottom plot), the digital signal 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 a reconstruction filter.

1.3.4 COMPACT-DISC RECORDING SYSTEM

A compact-disc (CD) recording system is described in Fig. 1.10 (top plot). 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

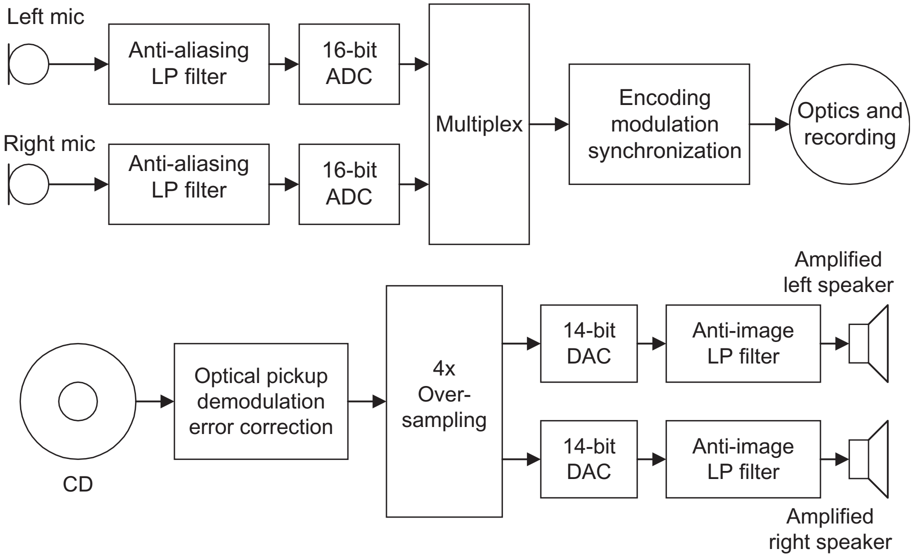

FIG. 1.10

(Top plot) Simplified encoder of the CD recording system. (Bottom plot) Simplified decoder of the CD recording system.

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 data bits are modulated for storage, and more synchronized bits are added for subsequent recovery of sampling frequency. The modulated signal is then applied to control a laser beam that illuminates the photosensitive layer of a rotating glass disc. When the laser turns on and off, the digital information is etched on the photosensitive layer as a pattern of pits and lands in a spiral track. This master disc forms the basis for mass production of the commercial CD from the thermoplastic material.

During playback, as illustrated in Fig. 1.10 (bottom plot), a laser optically scans the tracks on a CD to produce digital signal. The digital signal is then demodulated. The demodulated signal is further oversampled by a factor of 4 to acquire a sampling rate of 176.4 kHz for each channel and is then passed to the 14-bit DAC unit. For the time being, we consider the oversampling process as interpolation, that is, adding three samples between every two original samples in this case, as we shall see in Chapter 11. After DAC, the analog signal is sent to the anti-image analog filter, which is a lowpass filter to smooth the voltage steps from the DAC unit. The output from each anti-image filter is fed to its amplifier and loudspeaker. The purpose of the oversampling is to relieve the higher-filter-order requirement for the anti-image lowpass filter, making the circuit design much easier and economical (Ambardar, 1999).

Software audio players that play music from CDs, such as Windows Media Player and RealPlayer, installed on computer systems, are examples of DSP applications. The audio player has many advanced features, such as a graphical equalizer, which allows users to change audio with sound effects including boosting low-frequency content or emphasizing high-frequency content to make music sound more entertaining (Ambardar, 1999; Embree, 1995; Ifeachor and Jervis, 2002).

1.3.5 VIBRATION SIGNATURE ANALYSIS FOR DEFECTED GEAR TOOTH

Gearboxes are widely used in industry and vehicles (Spectra Quest, Inc.). During the extended service lifetimes, the gear teeth will inevitably be worn, chipped, or missing. Hence, with DSP techniques, effective diagnostic methods can be developed to detect and monitor the defected gear teeth in order to enhance the reliability of the entire machine before any unexpected catastrophic events occur. Fig. 1.11A shows the gearbox, in which two straight bevel gears with a transmission ratio of 1.5:1 inside the gearbox are shown Fig. 1.11B. The number of teeth on the pinion is 18. The gearbox input shaft

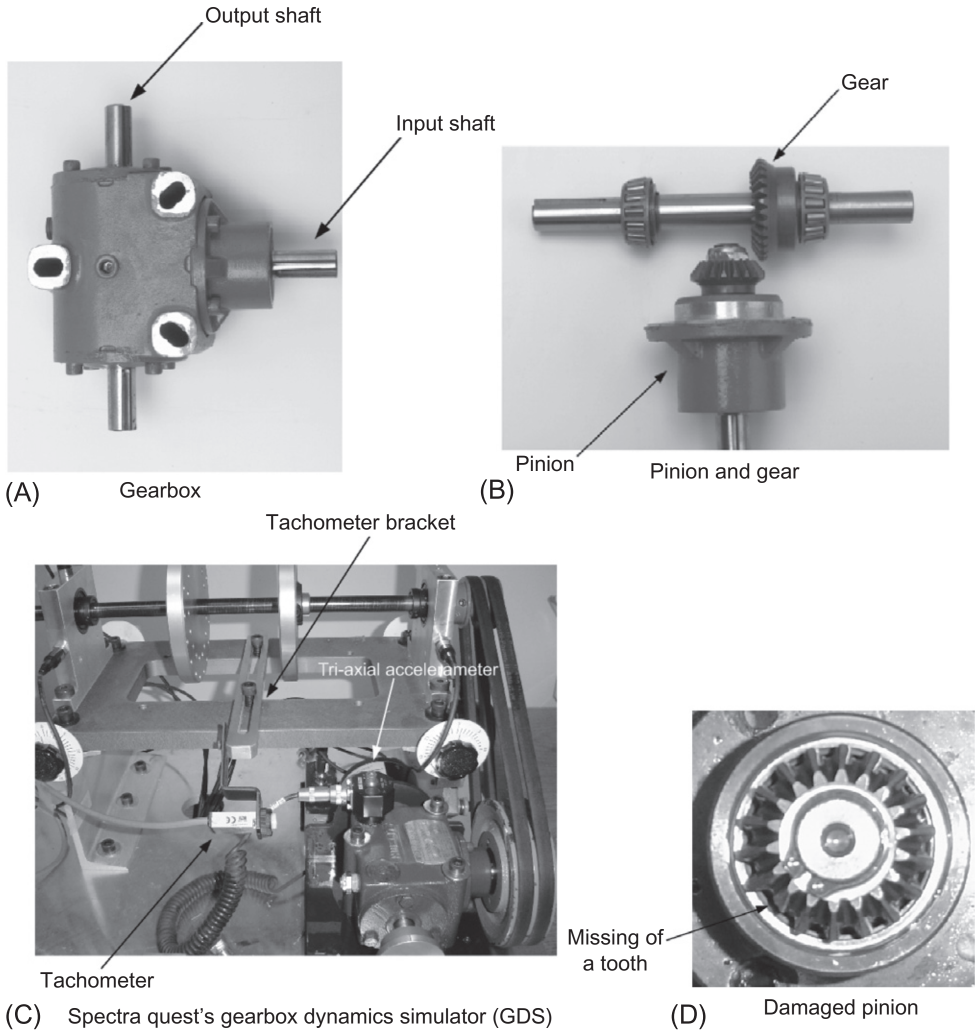

FIG. 1.11

Vibration signature analysis of the gearbox. (A) Gearbox, (B) Pinion and gear, (C) Spectra Quest's Gearbox Dynamics Simulator (GDS), (D) Damaged pinion.

Courtesy of SpectaQuest, Inc.

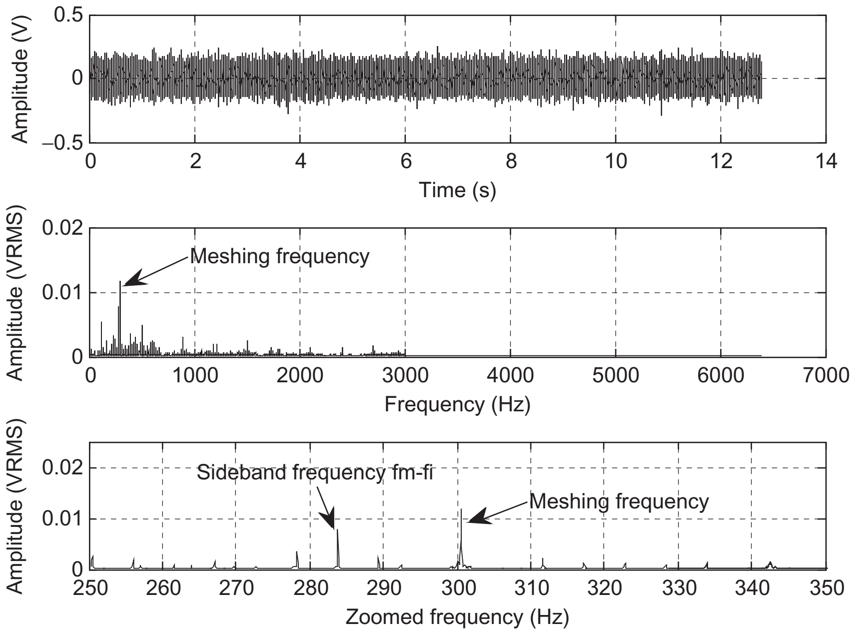

FIG. 1.12

Vibration signal and spectrum from the good condition gearbox.

Data provided by SpectaQuest, Inc.

is connected to a sheave and driven by a "V" belt drive. The vibration data can be collected by triaxial accelerometer installed on the top of the gearbox, as shown in Fig. 1.11C. The data acquisition system uses a sampling rate of 12.8 kHz. Fig. 1.11D shows that a pinion has a missing tooth. During the test, the motor speed is set to 1000 rpm (revolutions per minute) so the meshing frequency is determined as $f_m = 1000(\text{rpm}) \times 18/60 = 300$ Hz and input shaft frequency is $f_i = 1000(\text{rpm})/60 = 16.17$ Hz. The baseline signal and spectrum (excellent condition) from x direction of the accelerometer are displayed in Fig. 1.12, where we can see that the spectrum contains the meshing frequency component of 300 Hz and a sideband frequency component of 283.33 (300–16.67) Hz. Fig. 1.13 shows the vibration signature for the damaged pinion in Fig. 1.11D. For the damaged pinion, the sidebands ($f_m \pm f_i$, $f_m \pm 2f_i \ldots$) become dominant. Hence, the vibration failure signature is identified. More details can be found in Robert Bond Randall (2011).

1.3.6 DIGITAL IMAGE ENHANCEMENT

We can look at another example of signal processing in two dimensions. Fig. 1.14A shows a picture of an outdoor scene taken by a digital camera on a cloudy day. Due to this weather condition, the image is improperly exposed in natural light and comes out dark. The image processing technique called *histogram equalization* (Gonzalez and Wintz, 1987) can stretch the light intensity of an image using the digital information (pixels) to increase the image contrast, therefore, detailed information can easily be seen in the image, as we can see in Fig. 1.14B. We will study this technique in Chapter 13.

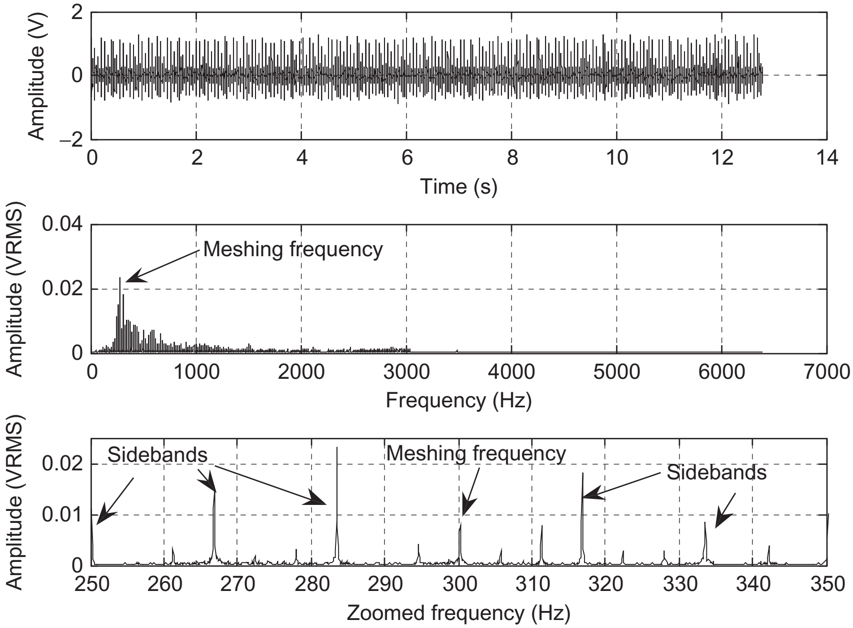

FIG. 1.13

Vibration signal and spectrum from the damaged gearbox.

Data provided by SpectaQuest, Inc.

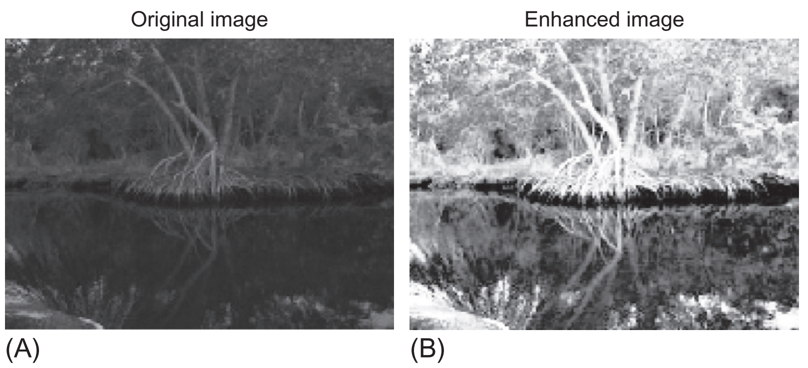

FIG. 1.14

Image enhancement. (A) Original image taken on a cloudy day. (B) Enhanced image using the histogram equalization technique.

Table 1.1 Applications of Digital Signal Processing

Digital Audio and Speech	Digital audio coding such as CD players, MP3 players, digital crossover, digital audio equalizers, digital stereo and surround sound, noise reduction systems, speech coding, data compression and encryption, speech synthesis and speech recognition
Digital telephone	Speech recognition, high-speed modems, echo cancellation, speech synthesizers, DTMF (dual-tone multifrequency) generation and detection, answering machines
Automobile Industry	Active noise control systems, active suspension systems, digital audio and radio, digital controls, vibration signal analysis
Electronic Communications	Cellular phones, digital telecommunications, wireless LAN (local area networking), satellite communications
Medical Imaging Equipment	ECG analyzers, cardiac monitoring, medical imaging and image recognition, digital X-rays and image processing
Multimedia	Internet phones, audio and video, hard disk drive electronics, iPhone, iPad, digital pictures, digital cameras, text-to-voice, and voice-to-text technologies

1.4 DIGITAL SIGNAL PROCESSING APPLICATIONS

Applications of DSP are increasing in many areas where analog electronics are being replaced by DSP chips, and new applications depend on DSP techniques. With the decrease in the cost of digital signal processors and increase in their performance, DSP will continue to affect engineering design in our modern daily life. Some application examples using DSP are listed in Table 1.1.

However, the list in the table by no means covers all the DSP applications. Many application areas are increasingly being explored by engineers and scientists. Applications of DSP techniques will continue to have profound impacts and improve our lives.

1.5 SUMMARY

1. An analog signal is continuous in both time and amplitude. Analog signals in the real world include current, voltage, temperature, pressure, light intensity, and so on. The digital signal contains the digital values converted from the analog signal at the specified time instants.
2. Analog-to-digital signal conversion requires an ADC unit (hardware) and a lowpass filter attached ahead of the ADC unit to block the high-frequency components that ADC cannot handle.
3. The digital signal can be manipulated using arithmetic. The manipulations may include digital filtering, calculation of signal frequency content, and so on.
4. The digital signal can be converted back to an analog signal by sending the digital values to DAC to produce the corresponding voltage levels and applying a smooth filter (reconstruction filter) to the DAC voltage steps.
5. DSP finds many applications in areas such as digital speech and audio, digital and cellular telephones, automobile controls, vibration signal analysis, communications, biomedical imaging, image/video processing, and multimedia.