Lec:1 Introduction to Digital Signal Processing

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, and MP3 players; digital cameras; digital and cellular telephones; digital satellite and TV; or wire 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 or 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 data and perform their design, and so on.

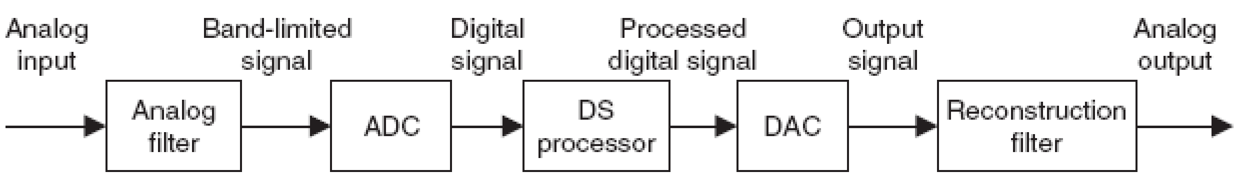

FIGURE 1.1 **A digital signal processing scheme.**

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.

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 non-electrical 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.

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 DAC unit converts the processed digital signal to an analog output signal. The signal is continuous in time and discrete in amplitude (usually a sample-and-hold signal). 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 real-world applications.

1.2 Basic Digital Signal Processing

1.2.1 Digital Filtering

Consider the situation shown in Fig. (1.2), of a digitized noisy signal 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 as noise, which can be removed by using a digital lowpass filter.

After processing the digitized noisy signal x(n), the digital lowpass filter produces a clean digital signal y(n). The cleaned signal y(n) is applied 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 Fig. (1.3), where the top plot shows the digitized noisy signal, x(n), while the bottom plot demonstrates the clean digital signal ,y(n), obtained by applying the digital lowpass filter.

FIGURE 1.2 **The simple digital filtering block.**

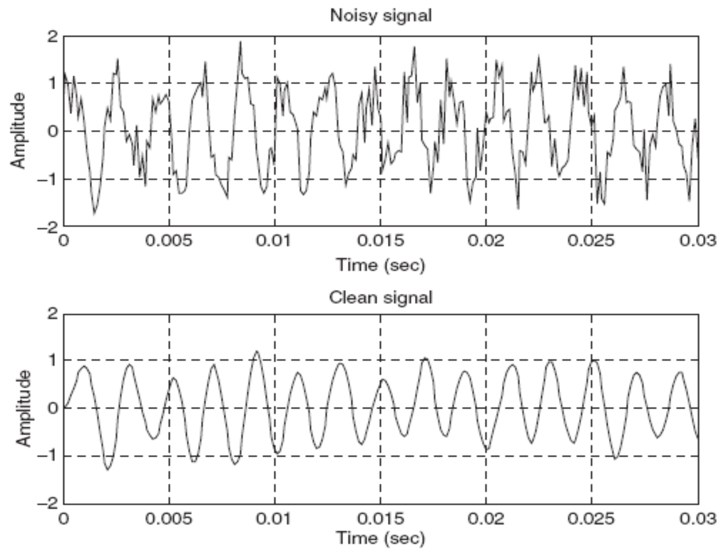

FIGURE 1.3 (*Top*) Digitized noisy signal. (*Bottom*) Clean digital signal using the digital lowpass filter.

1.2.2 Signal Frequency (Spectrum) Analysis

As shown in Figure 1.4, certain DSP applications often require that time domain information and the frequency content of the signal be analyzed.

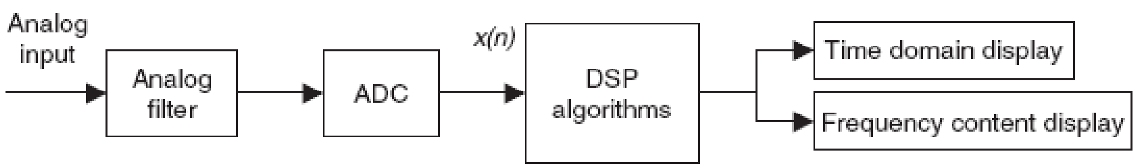

FIGURE 1.4 Signal spectral analysis.

Figure 1.5 shows a digitized audio signal and its calculated signal spectrum (frequency content), defined as the signal amplitude versus its corresponding frequency. It is also called fast Fourier transform (FFT).

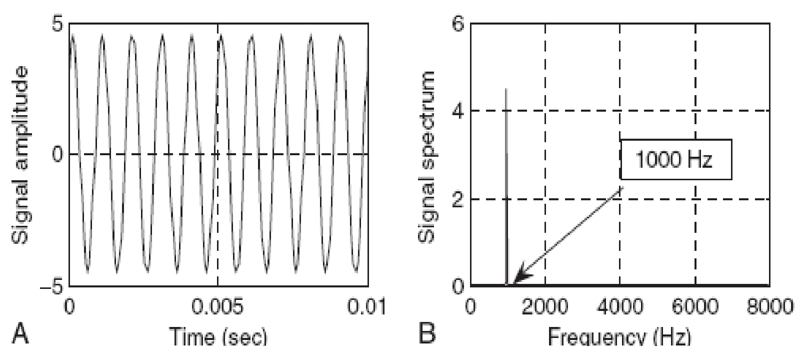

Figure 1.5 Audio signal and its spectrum

The plot in Figure 1.5 (a) is a time domain display of the recorded audio signal with 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 frequencies, in which the peak amplitude is clearly located at 1,000 Hz.

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, and speech feature extraction for speech synthesis and recognition,

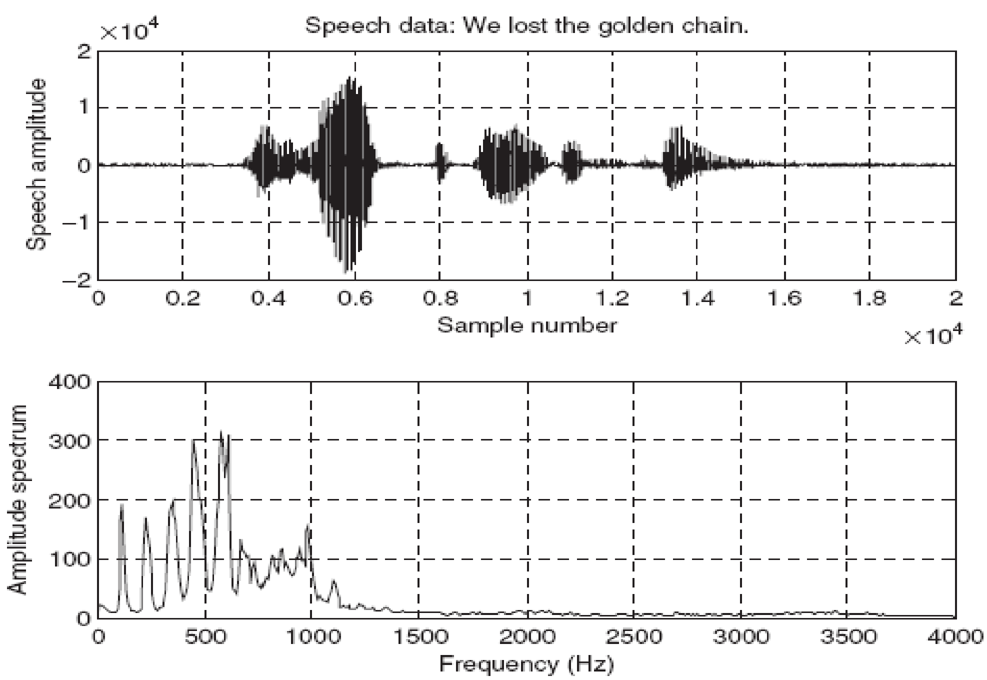

Figure 1.6 Speech sample and speech spectrum

1.3 Digital Signal Processing Applications

The list below by no means covers all DSP applications. Many more areas are increasingly being explored by engineers and scientists. Applications of DSP techniques will continue to have profound impacts and improve our lives.

1. ***Digital audio and speech***: Digital audio coding such as CD 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.
2. ***Digital telephone***: Speech recognition, high-speed modems, echo cancellation, speech synthesizers, DTMF (dual-tone multi frequency) generation and detection, answering machines.
3. ***Automobile industry***: Active noise control systems, active suspension systems, digital audio and radio, digital controls.
4. ***Electronic communications***: Cellular phones, digital telecommunications, wireless LAN (local area networking), satellite communications.
5. ***Medical imaging equipment***: ECG analyzers, cardiac monitoring, medical imaging and image recognition, digital x-rays and image processing.
6. ***Multimedia***: Internet phones, audio, and video; hard disk drive electronics; digital pictures; digital cameras; text-to-voice and voice-to-text technologies