

Introduction to Digital Signal Processing and Digital Filtering

1.1 Introduction

Digital signal processing (DSP) refers to anything that can be done to a signal using code on a computer or DSP chip. To reduce certain sinusoidal frequency components in a signal in amplitude, **digital filtering** is done. One may want to obtain the integral of a signal. If the signal comes from a tachometer, the integral gives the position. If the signal is noisy, then filtering the signal to reduce the amplitudes of the noise frequencies improves signal quality. For example, noise may occur from wind or rain at an outdoor music presentation. Filtering out sinusoidal components of the signal that occur at frequencies that cannot be produced by the music itself results in recording the music with little wind and rain noise. Sometimes the signal is corrupted not by noise, but by other signal frequencies that are of no present interest. If the signal is an electronic measurement of a brain wave obtained by using probes applied externally to the head, other electronic signals are picked up by the probes, but the physician may be interested only in signals occurring at a particular frequency. By using digital filtering, the signals of interest only can be presented to the physician.

1.2 Historical Perspective

Originally signal processing was done only on **analog or continuous time signals** using **analog signal processing (ASP)**. Until the late 1950s digital

computers were not commercially available. When they did become commercially available they were large and expensive, and they were used to simulate the performance of analog signal processing to judge its effectiveness. These simulations, however, led to digital processor code that simulated or performed nearly the same task on samples of the signals that the analog systems did on the signal. After a while it was realized that the simulation coding of the analog system was actually a DSP system that worked on samples of the input and output at discrete time intervals.

But to implement signal processing digitally instead of using analog systems was still out of the question. The first problem was that an analog input signal had to be represented as a sequence of samples of the signal, which were then converted to the computer's numerical representation. The same process would have to be applied in reverse to the output of the digitally processed signal. The second problem was that because the processing was done on very large, slow, and expensive computers, practical real-time processing between samples of the signal was impossible. Finally, as we will see in Chapter 9, even if digital processing could be done quickly enough between input samples in order to adequately represent the input signal, high sample rates require more bits of precision than slower ones.

The development of faster, cheaper, and smaller input signal samplers (ADCs) and output converters from digital data to analog data (DACs) began to make real-time DSP practical. Also, the processors were becoming smaller, faster, and cheaper and used more bits. Real-time replacements for analog systems may be just as small, cheap, and accurate and be able to process at a sample rate adequate for many analog signals.

However, testing and modification of the coding for DSP systems led to DSP systems that have no analog signal processing equivalents, yet sometimes perform the signal processing better than the DSP coding developed to replace analog systems. For digital filtering, these processing methods are discussed in Chapters 10 and 11.

1.3 Simple Examples of Digital Signal Processing

Digital signal processing entails anything that can be done to a signal using coding on a computer or DSP chip. This includes digital filtering

of signals as well as digital integration and digital correlation of signals. This text concentrates on constant rate digital filtering, with references to where the material is applicable to DSP in general. At the end of the text the techniques developed for digital filtering will be used for the DSP task of integration to show how the concepts and techniques are not limited to digital filtering.

The concepts are very simple. A signal is sampled in time at a constant rate in order to input its magnitude value at periodic intervals into the computer. The sample value of the analog magnitude is converted into a binary number. The sampling and conversion are done with an **analog to digital converter (ADC)**. Now the computer code can work on the signal. The computer code computes an output value, which is converted to an analog magnitude from a binary number and then held constant until a new output is computed to replace it. This is done by a **digital to analog converter (DAC)**. The basic DSP system described here is shown in Figure 1.1.

To illustrate the concept of DSP and to see where more study and analysis are needed, let's look at a few simple things that can be done to a sampled signal. If a signal is sampled every T seconds by an ADC and in the computer the sample value is just multiplied by a constant and then sent to the DAC, you have a digital amplifier. The gain of the amplifier is equal to the coded value of the constant. The following equation describes this digital amplifier, where x is the current input sample value from the ADC and y is the corresponding computer output to the DAC.

$$y = ax \quad \text{A simple digital amplifier}$$

If the sampled value of the input signal is multiplied by T , you have computed an approximation to the area under the signal between samples, as long as the signal doesn't change too much between samples. If this value is added to the previous input sample multiplied by T , you

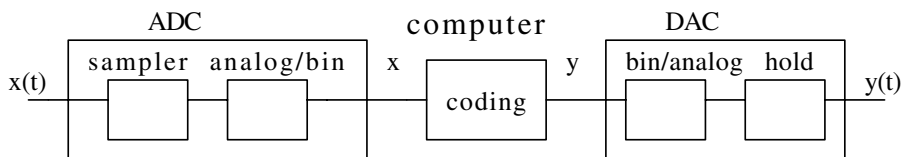


Figure 1.1. Basic DSP system

have approximated the area under the signal over two sample times. This could be repeated endlessly to approximate the area under the signal from when sampling started, as shown in Figure 1.2. The area under a signal or function is its integral. Thus you have performed very simple digital integration using the current sample of the input multiplied by T and then adding the result to the previous output. This process is described by the following equations after two input samples (the -1 subscripts indicate they are previous input or output values).

$$y_{-1} = Tx_{-1} \quad \text{Simple digital integration after one input sample} \\ \text{(the previous sample)}$$

$$y = y_{-1} + Tx \quad \text{Simple digital integration after two input samples} \\ \text{(the current sample)}$$

By using looping, such as a “While” or “For” loop, the preceding equations could be repeated endlessly by looping about one equation.

If the current input sample value is multiplied by one-half and added to half the previous sample value of the input, a current change in the input signal is reduced, while if the signal is changing slowly the output is very close to the input, since it is just the sum of two half values. Thus the computer is doing a very simple lowpass filtering of the input signal. This simple process is represented by the following equation. The result of using this equation on a string of input samples from the ADC is the input to the DAC shown in Table 1.1. As can be seen, the results, y , are smoothed or lowpass filtered versions of x , the ADC output.

$$y = 0.5x + 0.5x_{-1} \quad \text{Simple digital lowpass filtering}$$

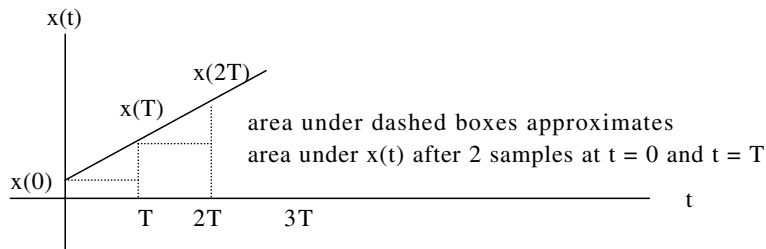


Figure 1.2. Example of digital integration