Abstract: Audio signal processing systems have made considerable progress over the past 25 years due to increases in computational speed and memory capacity. These changes can be seen by examining the implementation of increasingly complex algorithms in less and less hardware. In this chapter, we will describe how machines have been designed to implement DSP algorithms. We will also show how progress in integration has resulted in the special purpose chips designed to execute a given algorithm.

5.1 INTRODUCTION

Audio signal processing systems have made considerable progress over the past 25 years due to increases in computational speed and memory capacity. These improvements are a direct result of the ever increasing enhancements in silicon processing technologies. These changes can be demonstrated by examining the implementation of increasingly complex algorithms in less and less hardware. In this chapter, we will describe how sound is digitized, analyzed and synthesized by various means. The chapter proceeds from input to output with a historical bent.
5.2 INPUT/OUTPUT

A DSP system begins at the conversion from the analog input and ends at the conversion from the output of the processing system to the analog output as shown in the figure 5.1:

Anti-aliasing filters (considered part of “Analog Conditioning”) are needed at the input to remove out of band energy that might alias down into baseband. The anti-aliasing filter at the output removes the aliases that result from the sampling theorem.

After the anti-aliasing filter, the analog/digital converter (ADC) quantizes the continuous input into discrete levels. ADC technology has shown considerable improvement in recent years due to the development of oversampling and noise-shaping converters. However, a look at the previous technologies [Blesser, 1978] [Blesser and Kates, 1978][Fielder, 1989] will help appreciate the current state-of-the-art.

After digital processing, the output of the system is given to a digital/analog converter (DAC) which converts the discrete levels into continuous voltages or currents. This output must also be filtered with a low pass filter to remove the aliases. Subsequent processing can include further filtering, mixing, or other operations. However, these shall not be discussed further.

5.2.1 Analog/Digital Conversion

Following the discussion in Bennett ([Bennett, 1948]), we define the Signal to Noise Ratio (SNR) for a signal with zero mean and a quantization error with zero mean as follows: first, we assume that the input is a sine wave. Next, we define the root mean square (RMS) value of the input as

$$\sigma_{RMS} = \frac{\Delta 2^{b-1}}{\sqrt{2}}$$

(5.1)