Fundamentals of Data Acquisition

An Online Continuing Education Course for Engineers

Course Number: E-1011

Credit: 1 Hour / 1 PDH / 1 CPD

Fundamentals of Data Acquisition

Introduction

Data acquisition is a common tool in many areas of science and engineering; so common, in fact, that it is often assumed that the practitioner understands the fundamental concepts of data acquisition and appreciates the tradeoffs in performance characteristics. The aim of this course is to provide an introduction to the essentials of data acquisition, alert users to the potential pitfalls, and provide examples of selecting and configuring suitable data acquisition hardware. The emphasis is on concepts, rather than mathematical rigor or details of individual components. This course teaches the following specific knowledge and skills:

- Aliasing: cause and consequences
- Anti-aliasing filters
- Types of analog-digital converters

Figure 1 depicts the front end of a data acquisition system. The input is a continuous signal, one that can take on any value at any time. These are often real-world signals such as temperature, speed, etc. The analog to digital converter samples the input signal, typically at fixed time intervals. Each sample is approximated by a binary code. The quality of the approximations depends on the sampling intervals and the size of the binary code.

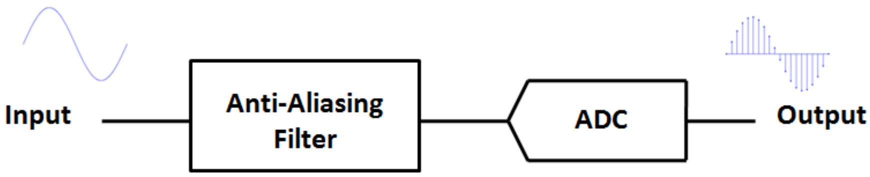


Figure 1. Data Acquisition Front End.

Aliasing: causes and consequences

Aliasing is the effect that causes different sampled signals to be indistinguishable. Figure 2 shows an example of aliasing. In the example, two different signals (blue and green) are sampled at regular intervals, indicated by the black lines. At the sampled points the blue and green signals are equal. Consequently, the digital representations of the two signals are identical and indistinguishable – aliasing has occurred.

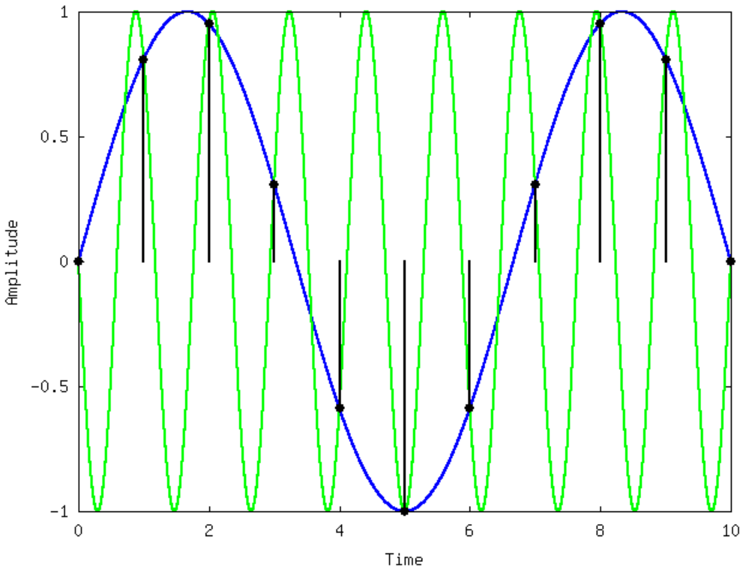


Figure 2. Two different signals, sampled to produce equivalent results.

The signals in Figure 2 are not the only possible signals that produce aliasing. There are, in fact, an infinite number of signals that would produce the same effect. To avoid aliasing, the sampling frequency must be at least twice the bandwidth of the input signal. Stated differently, any input signal above half of the sampling frequency will be indistinguishable from a particular signal below the sampling rate. For example, if an input signal occupies a bandwidth up to 10 kHz, a data acquisition system must sample faster than 20 kHz to avoid aliasing. This condition is known as the Nyquist Sampling Theorem.

As another example, typical human hearing does not extend beyond approximately 20 kHz. Consequently, high-fidelity audio systems sample at frequencies above 40 kHz (44.1 kHz is common).

Figure 3 shows frequency-domain plots of another example of aliasing. The top image shows the 6 kHz input signal. The bottom image shows the result when the 6 kHz input signal is sampled at 10 kHz (the minimum sampling frequency to avoid aliasing for a 6 kHz input signal is 12 kHz). The poor choice of sampling frequency aliases some of the input signal to 4 kHz. Without knowledge of the original signal, it is impossible to know that the peak at 4 kHz was caused by aliasing and not by a genuine 4 kHz input.

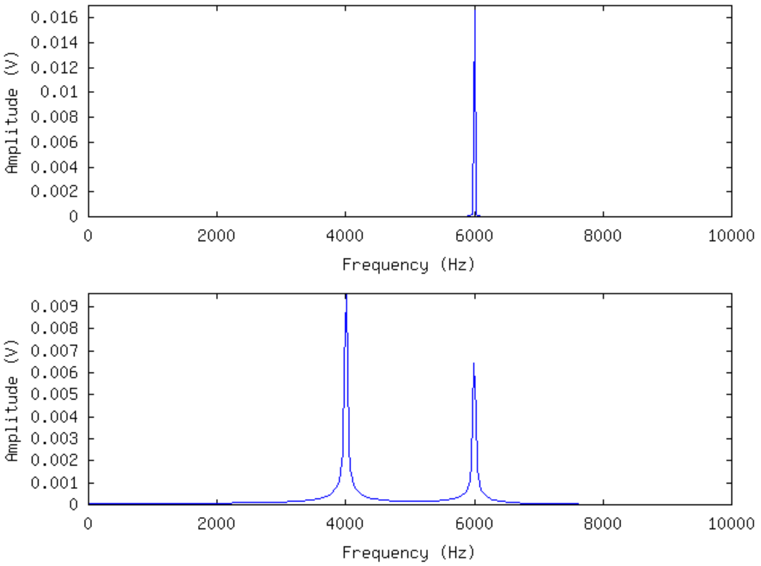


Figure 3. Spectra of Aliasing.

Filter Selection

Suppose that an input signal contains useful information up to 10 kHz. Beyond 10 kHz, the input signal contains noise (or possibly a separate data channel). In a data acquisition system, the signals beyond 10 kHz could alias to frequencies below 10 kHz, corrupting the useful information. To prevent aliasing, a filter is placed before the analog-to-digital converter. The purpose of the filter is to retain the signals of interest and eliminate those that could cause aliasing. Although an in-depth study of analog filters is beyond the scope of this document, an understanding of common filter types and their characteristics will be beneficial to anyone using data acquisition equipment.

In most applications the frequencies of interest are low and the anti-aliasing filter is selected to keep the low frequencies and remove the high. Ideally, when choosing a filter, the designer could specify a cutoff frequency, below which all signals would be reproduced exactly and above which all signals would be eliminated. The magnitude response of the ideal filter is shown in Figure 4. In the region below 1 rad/sec, called the pass band, all signals are passed without modification (0dB is a gain of 1). Above 1 rad/sec (the stop band or cutoff region), all signals are completely eliminated (a gain of 0).

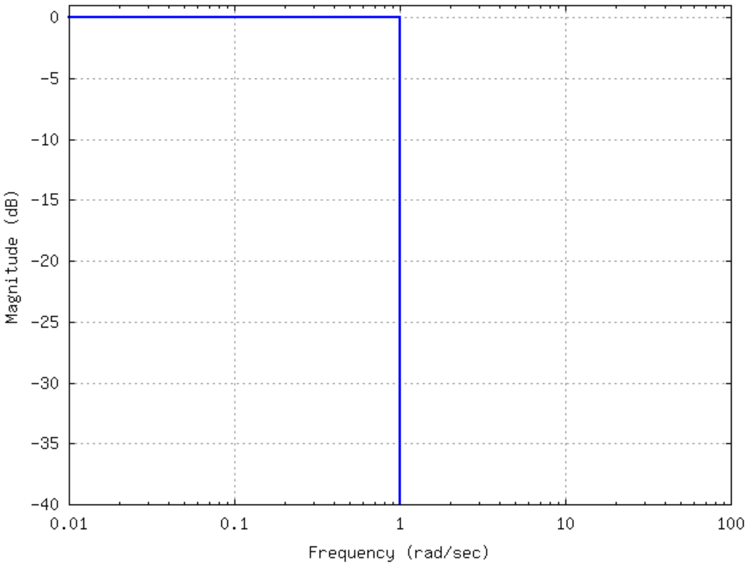
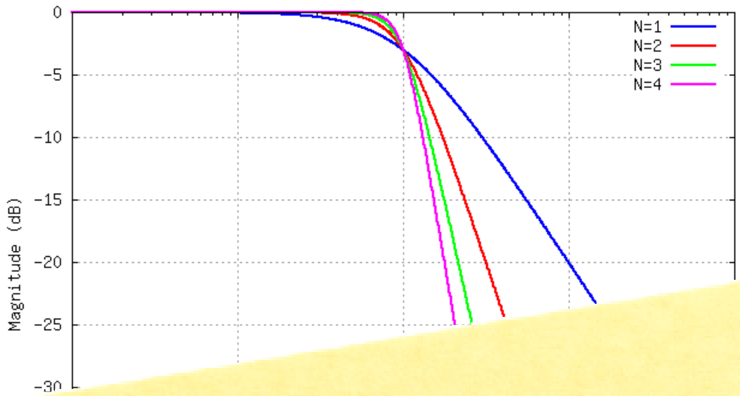


Figure 4. Magnitude response of an ideal low-pass filter.

In reality, the ideal filter depicted in Figure 4 cannot be constructed. First, real-world filters *attenuate*, but do not *eliminate*, input signals beyond the specified "cutoff" frequency. The order of the filter determines the rate at which out-of-band signals are attenuated. The higher the order, the more closely the filter approximates the ideal cutoff characteristic. However, higher order filters require more components and are more expensive to manufacture. Figure 5 shows the magnitude response of four low-pass filters of various orders. The slope of the filter's magnitude response (after the transition region) is $-20dB * N/$decade, where N is the filter order and a decade is the span of frequencies from f_1 to $10 * f_1$ for any frequency well above the cutoff frequency. For example, the slope of a 1^{st}-order filter is $-20dB * 1/dec = -20dB/dec$. It can be seen in Figure 5 that gain of the 1^{st}-order filter (blue curve) decreases by $20dB$ over the decade $10\ rad/sec$ to $100\ rad/sec$. Equivalently, the slope of filter attenuation can be calculated as $-6dB * N/$octave, where an octave is the span of frequencies from f_1 to $\mathbf{2} * f_1$ (again, for any frequency well above the cutoff frequency). For example, the slope of a 3^{rd}-order filter is $-18dB/$octave (or $-60dB/$decade).



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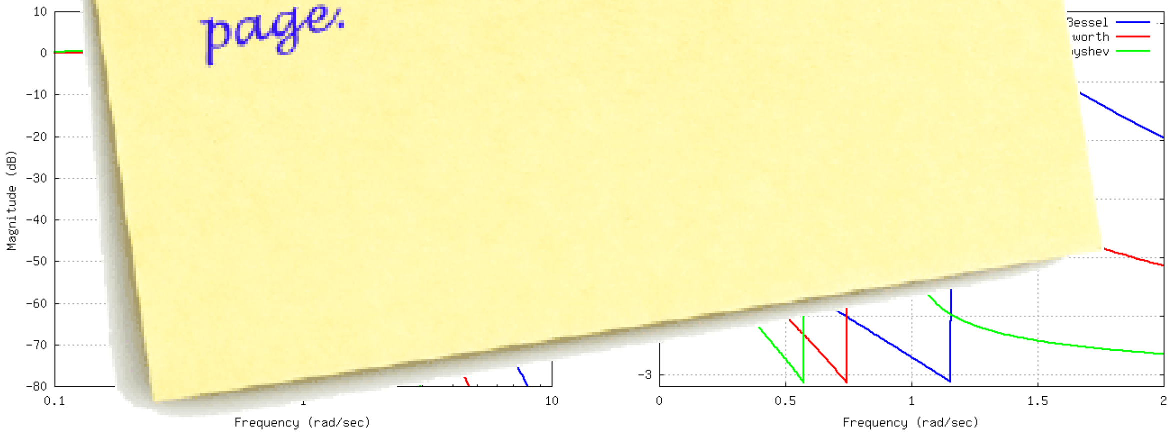

(a) Magnitude Response (b) Phase Response

Figure 6. Response of three common types of low-pass filters.

Figure 6(b) shows how each filter type affects the phase of input signals. It indicates that the