

Introduction

EXAR's family of high resolution, high speed analog-to-digital converters sample the input signals then convert these samples to a digital format for processing. The sampling of analog signals is subject to restrictions and limitations. The choice of sampling frequency and the effect of aperture uncertainty are two important considerations in sampled data systems. These are discussed below.

Choosing a Sampling Frequency

The Nyquist sampling theorem defines the limitation on sampling frequency of an analog signal: In order to sample an analog signal for future reconstruction, the sampling frequency, f_s , must be greater than twice the maximum frequency component of the analog signal being sampled.

An anti-aliasing filter must be placed in front of the sampler if the the signal bandwidth is greater than half the sampling rate. The tradeoffs to be considered in selecting the filter and sampling frequency are:

- a) A steep cutoff filter permits the use of a lower sampling frequency but requires a high order filter.
- b) Gradual filter rolloff (lower order filter) implies a higher sample rate (faster A/D).
- c) Phase linearity of the filter must sometimes be considered which increases filter complexity.

Undersampling (sampling at a rate lower than double the maximum frequency) is a technique that can be usefully applied in particular applications. It can for example be employed for down conversion of band limited signals.

Aperture Uncertainty

Consider a continuous analog signal, $v(t)$, having a maximum bandwidth of f_{MAX} . When converting this signal, a periodic sampler, such as a high speed A/D converter can introduce errors due to sampling time uncertainty.

The clock signal, which dictates when samples are taken, deviates from a perfectly timed periodic signal by some amount and the actual system sample time is characterized by a mean and standard deviation called aperture delay, t_d , and aperture uncertainty, t_a .

When sampling an analog signal, the maximum allowable signal bandwidth can be related to the aperture uncertainty of the sampler & clock.

Consider a sinusoidal signal of maximum amplitude V :

$$v(t) = V \cdot \sin(2 \cdot \pi \cdot f \cdot t)$$

its maximum slope, occurs at $V(t)=0$ and is:

$$\frac{dv}{dt} = 2 \cdot \pi \cdot f \cdot V$$

An analog signal which has a upper spectral component of f_{MAX} will thus have a maximum voltage slope:

$$\frac{dv}{dt} = 2 \cdot \pi \cdot f_{MAX} \cdot V$$

Sampling this slope with an rms time uncertainty of t_a will generate an rms amplitude uncertainty of:

$$V_{ERR} = 2 \cdot \pi \cdot V \cdot t_a \cdot f_{MAX}$$

Amplitude quantization by an n-bit A/D converter typically yields a 1/2 LSB to 1LSB integral nonlinearity. If one allows the sampler to add an 1/2LSB rms error due to the sampling inaccuracy:

$$\frac{V_{ERR}}{2V} = \frac{1}{2^{n+1}}$$

then by substitution:

$$f_{MAX} = \frac{1}{2 \cdot \pi \cdot t_a \cdot 2^{n+1}}$$

As an example, an $n = 8$ -bit high speed A/D converter with $t_a = 40$ ps can convert an analog input signal with a highest full-amplitude spectral component of $f_{MAX} = 15.5$ MHz, assuming a 1/2 LSB rms amplitude error is allowable.

Notes