

EE 230
Fall 2006
Experiment 12

Data Converters – Inherent Performance Limitations

Purpose: The purpose of this laboratory experiment is to investigate some of the inherent fundamental limitations associated with the data conversion process. In particular, the issues of time quantization, amplitude quantization, and sampling rate will be explored.

Equipment:

Computer with MATLAB software and sound card
External speakers

Background:

Data converters are widely used as the interface between the analog environment and the digital world. Analog to Digital Converters (ADC) convert physical analog signals to digital form and Digital to Analog Converters (DAC) convert digital signals analog form. In most applications, it is expected that all information about the input signal to an ADC be preserved in the sampled outputs obtained from the ADC. Correspondingly, it is generally expected that the output of a DAC will create the desired analog signal.

Existing data converters can come very close to accomplishing these tasks provided that the right data converter is used and provided it is used properly. Even if data converters are ideal, some potential problems can occur if the resolution is not high enough or if the sampling rates are too low. In some applications, the phase of the sampling clock is also very important. If the data converter is not ideal, several additional problems can also occur.

In this experiment, emphasis will be placed upon the performance capabilities and limitations inherent in the data conversion process even if ideal data converters are used.

Time Quantization

Data converters are almost always used synchronously providing an output synchronously with a clock signal. That is, the output is updated once each clock period. For convenience, it will be assumed that update is coherent with the rising edge of the clock signal. Thus, although the input signal to an ADC may be a time-continuous signal or the de data sired output of a DAC may be a time-continuous signal, the actual outputs to any data converter will only occur at discrete points in time. This is termed time-quantization. The well-known sampling theorem, one of the most fundamental properties of the signal processing and communications fields, states that sufficient information exists in the sampled outputs to recover the input signal provided the signal is sampled at or above the Nyquist rate. Time quantization effects are inherent in the data conversion process and must be appropriately managed, even if the data converters are ideal.

If the requirements of the sampling theorem are not met, information is invariably lost making it difficult to interpret the results. Sampling at a rate lower than the Nyquist

rate will result in aliasing and aliasing will occur even with ideal data converters if the sampling rate is less than the Nyquist rate.

Amplitude Quantization

Any data converter will have finite resolution. This resolution could be as little as 1 or 2 bits to as many as 24 or more bits. Amplitude quantization introduces non-recoverable errors in the data conversion process, even if the data converters are ideal. Although theoretically deterministic, the differences between the actual output and the desired output associated with amplitude quantization behave much as if there were noise present. The term “quantization noise” is used to characterize the difference between the desired output and the actual output. Amplitude quantization effects can be managed, primarily by allocating an appropriate number of bits of resolution, but the cost of increasing the resolution beyond the minimum required is often very high. As such, the engineer invariably tries to achieve the desired outcome with as low of resolution as is viable.

Signal Reconstruction

A continuous-time signal must often be reconstructed from a sampled signal. One of the most common ways this is done is to sample and hold the discrete-time signal until the next signal is available. This is often termed a “zero-order sample and hold” operation. Often this signal is subsequently passed to a lowpass filter to further smooth the signal. In this application, the filter is termed a smoothing filter. The sound cards in most pcs can directly process the sampled signals by incorporating the sample and hold operation with a smoothing filter function.

Part 1 Time Quantization and Aliasing

Assume an ideal ADC of very high resolution is available.

- a) Using MATLAB, sample a 1KHz sinusoidal signal with an amplitude of 5V pp at 5 times the Nyquist rate. Plot the resultant samples and compare with the original signal.
- b) Repeat Part a) if sampled at only 2/3 the Nyquist rate
- c) Determine the aliased signal generated in part b) in the time domain.

Part 2 Amplitude Quantization

Assume an ADC is ideal with a finite resolution n .

- a) Using MATLAB, quantize the signal in Part 1a to 8 bits if V_{REF} is 10V (actually $\pm 5V$). Plot the input signal and the quantized signal and determine the quantization noise. Express the quantization noise in LSB.
- b) Repeat part a) if the ADC only has 3 bits of resolution.

Part 3 Signal Reconstruction

Generate samples of the signal in Part 1a for a 5 second interval for resolutions of n-bits for $3 < n < 16$. Store these samples in arrays.

- a) Send these samples to the sound card on your pc. At what resolution level can you hear the effects of reducing the resolution?
- b) Obtain a 10-second audio signal from your TA. Using MATLAB and the speaker on the sound card, determine the resolution level that you can first audibly notice the effects of quantization noise if the input is sampled at 20KHz..