Lab Sheet

IA 3203 – DIGITAL SIGNAL PROCESSING

Department of Instrumentation and Automation Technology
University of Colombo

DSP 304 – Analog to Digital Conversion

Quantization

The process of converting an analog signal to a digital signal by expressing each sample valueas a finite number of digits is defined as quantization. The error for quantization could be byeither truncation or rounding off.

Sampling Theorem

The process of converting a continuous analog signal into a discrete digital signal is definedas sampling. The sampling theorem or the Nyquist theorem specifies a minimum sampling rate required to capture all the information from the original continuous time signal by taking a discrete sequence of uniform samples if the sampling frequency satisfies the condition stated below.

$$f_{sampling} \geq 2\, f_{max}$$

If the sampling frequency is lower than the maximum frequency component in the frequency spectrum of the original signal, acquired samples will not reconstruct the signal accurately. This effect is known as the aliasing effect.

Anti-aliasing Filters

Anti-aliasing filters are used to prevent the above-mentioned aliasing effect specially when sampling high frequencies with a low sampling rate. Therefore, an appropriate low-pass filterwill be used to cut off higher frequencies.

Exercises

This practical is designed to give you a basic understanding of analog to digital conversion. You will do all the implementations for exercises using MATLAB or Octave.

Note that you are expected to give titles, label axes appropriately and provide legends specifying results if more than one result is plotted in the same sheet.

1. Suppose that you are given a sinusoidal signal $x(t) = \frac{1}{3}[\sin\left(\frac{t}{11}\right) + \sin\left(\frac{t}{31}\right) + \cos(\frac{t}{67})]$ to do a rounding operation to quantize. Let the number of samples be 50 andnote that the signal should cover the entire sampling space.

   Hint: You may use $'floor(\ )'$ and $'stem(\ )'$ when quantizing and plotting discrete signals.

   a. Plot the original sinusoidal signal.

b. Get the quantized signal by rounding off the given signal with respect to 2 bits and plot your result.
c. Find the quantization error.
d. Repeat part b for 6-bit quantization.
e. Do a comparison for the 2 versions of quantization and state your observations.

2. Suppose that you are given a sinusoidal signal $x(t) = \frac{1}{3}\sin(2\pi \times 20t)$ do a truncation operation to quantize.
Hint: You may use $'floor(\ )'$ and $'stem(\ )'$ when quantizing and plotting discrete signals.

    a. Plot the given sinusoidal signal given signal in the range $0 \le t \le 0.1$ s. take the sampling rate as 1000 Hz.
    b. Using MATLAB/Octave, calculate the number of bits required if asked to quantify the signal into 7 levels.
    c. Quantize the signal using the information given in part c and plot the quantized in the same plot you created in part a.
    d. Find the mean quantization error.

3. Generate a sinusoidal signal with an amplitude of 3 V and a 50 Hz frequency. Set the time from 0 to 0.05 s along the time axis.

    a. Display the signal you generated. (Use a very small step size such that the generated signal will be smooth and a close approximation to the actual analog signal).
    b. Use simple mathematical operations and your knowledge on analog signals to determine the minimum sampling rate required to avoid aliasing.
    c. Suppose that the signal was sampled at 80 Hz. Plot the discrete time signal obtained after sampling.
    d. If the sampling was done at 100 Hz, plot the new signal in a separate graph in the same panel along with the original signal and the signal sampled in part c.
    e. If the sampling was done at 500 Hz, plot the new signal in a separate graph in the same panel along with the original signal and 80 Hz and 100 Hz signals.

4. Suppose two sinusoid signals, $x(t) = 2\cos[(2\pi F_1) t + \pi/3]$ and $y(t) = 2\cos[(2\pi F_2) t + \pi/3]$ where F1=100 Hz and F2=600 Hz. Given the sampling frequency to be 500 Hz.
Hint: You may use $'stem(\ )'$ when plotting discrete signals.

    a. Plot the given signals in the same plot from t=0 to t=0.02 with an appropriate step size.
    b. Use a sampling index from 1 to 20 to plot the sampled signals of given sinusoids and show that results are similar.
    c. Briefly explain how your results in parts a. and b. demonstrate the aliasing effect. If necessary, use plots to graphically justify your answer.