AMERICAN INTERNATIONAL UNIVERSITY BANGLADESH

Faculty of Engineering

Laboratory Report Cover Sheet

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Laboratory Title: Study of Analog to Digital Conversion using MATLAB

Number: 05 Submission Date: 01/03/2023 Semester: Spring 2022 – 2023 Subject Code: COE 3201 Subject Name: Data Communication Section: K

Course Instructor: DR. SHUVRA MONDAL Degree Program: BSc CSE

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Group Number (if applicable): **01** ☐ Individual Submission ☒ Group Submission

No.	Student Name	Student ID	Contribution
1	SHEAKH, MOHAMMAD BIN AB. JALIL SHEAKH	20-42132-1	PerformanceTask (a) & (b)
2	AURTHY, MOST. LILUN NAHAR	20-43997-2	Abstract, Performance Task (a)
3	NISHAT, TARIKUL ISLAM	21-44632-1	Discussion, Conclusion, Performance Task (c)
4	MULLICK, IFTEKHAR UDDIN	21-44649-1	Performance Task (b) & (d)
5	ULLAH, MD ISMAIL JOBI	21-44747-1	Introduction, Performance Task (c)
6	ALANSAR, SADIAH	21-45612-3	

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Faculty comments____

Title:

Study of Analog to Digital Conversion using MATLAB

Abstract:
The objective of this experiment was to understand the use of MATLAB for solving communication engineering problems. It also developed an understanding of Analog to Digital Conversion using MATLAB

Introduction:
ANALOG TO DIGITAL CONVERSION: Analog signal is inferior to a digital signal. Nowadays, it is common practice to convert an analog signal to digital data. One of the line coding techniques can be used to transform digital data into a digital signal once they have been created digitally (digitization). We discuss the pulse code modulation and delta modulation approaches in this section.

PULSE CODE MODULATION (PCM): Pulse code modulation is the most used method for converting an analog signal to digital data (digitization) (PCM). Three operations take place in a PCM encoder: -The analog signal is sampled. The quantized sampled signal. Bitstreams are used to encode the quantized values.

SAMPLING: Sampling is the initial stage of PCM. Every Ts s, where Ts is the sample interval or period, the Bitstream signal is sampled. The sampling rate or sampling frequency is the inverse of the sampling interval and is represented by the symbol fs, where fs = 1/Ts. Ideal, natural, and flat-top sampling are the three available techniques. Pulses from the analog signal are sampled in perfect sampling. This is the best sampling technique; however, it is difficult to use. In natural sampling, a high-speed switch is only activated for the brief time that the sampling takes place. The technique is a series of samples that keep the analog signal's structure. However, the most used sampling technique, sample, and hold use a circuit to produce flat-top samples. Pulse amplitude modulation is another name for the sampling procedure (PAM). But it's important to keep in mind use the outcome is still an analog signal with nonintegral values. The Nyquist theorem states that the sampling rate must be at least twice as high as the signal's maximum frequency.

$fs = 2 \times fmax$

QUANTIZATION: This type of operation is carried out by analog-to-digital converters, who take the analog signal and turn it into a number of digital values. An analog signal is depicted in the following figure. To convert this signal to digital, sampling and quantization are required.

An analog signal is discretized with a variety of quantization levels before being quantized. A continuous amplitude sample is transformed into a discrete-time signal through quantization, which is the representation of sampled amplitude values by a limited set of levels.

$Vmax - Vmin$
$\Delta = L$

QUANTIZATION LEVELS: The range of the amplitudes of the analog signal and the degree of required signal recovery accuracy influence the choice of L or the number of levels.

QUANTIZATION ERROR: The number of quantization levels L, or the number of bits per sample nb, affects how much the quantization error contributes to the signal's SNRdB, as illustrated in the following formula:

$SNRdB = 6.02nb + 1.76$ dB

A simple method of quantization which can be implemented on MATLAB is given below:

Input Analog Signal = x

Number of Quantization Levels = L
Step Size, Δ = (xmax - xmin)/ (L-1)
Quantized Signal, xq = xmin + round ((x - xmin)/Δ)*Δ

Where, xmax and xmin are the maximum value and minimum values, respectively, of the analog input signal x. The symbol L denotes the number of quantization levels. The symbol Δ is the step size of the quantizer or the ADC resolution. Finally, xq indicates the quantization level, and i is an index corresponding to the binary code.

ENCODING: Encoding is PCM's final stage

. Each sample can be converted into an nb-bit code word once it has been quantized and the number of bits per sample has been chosen. It should be noted that the number of quantization levels determines the number of bits for each sample. The number of bits is L if there are L quantization levels.

$nb = log2\ (L)\ i = round\ ((x - xmin)/\Delta)$

The required bit rate (BR) for the encoding scheme can be determined as

$BR=fs\times n$

Performance Task:

The selected ID is the following:

2	0	-	4	3	9	9	7	-	2
A	**B**		**C**	**D**	**E**	**F**	**G**		**H**

Sig = a1*sin (2*pi*f1*t) + a2*cos (2*pi*f2*t) + a3*sin (2*pi*f3*t) + a4*sin (2*pi*f4*t)
= 11*sin (2*pi*12*t) + 12*cos (2*pi*14*t) + 11*sin (2*pi*8*t) + 13*sin (2*pi*9*t)

Performance Tasks 1&2:

Code:

```
clc
clear  all
close all
time_duration = 0.2;
a = [0.4 0.6 0.8];
Composite signalf = [5 12 20];
array     for     composite     signal     analog_t     =
0:0.0001:time_duration;
analog_sig=11*sin                   (2*pi*12*analog_t)+12*cos(2*pi*14*analog_t)                   +
11*sin(2*pi*8*analog_t + pi/4);figure
subplot (1, 2, 1)
plot(analog_t,
analog_sig,'linewidth',1.5)grid on
xlabel('time  in  seconds')
ylabel('amplitude         in
volts')          title('analog
signal')
fs = 250;

ts = 1/fs;
samp_t = 0:1/fs:time_duration;
samp_sig       =       9*sin(2*pi*8*samp_t)       +       11*cos(2*pi*10*samp_t)       +
10*sin(2*pi*4*samp_t + pi/4);subplot(1,2,2)
plot(samp_t,
samp_sig,'o','linewidth',1.5)grid on
xlabel('time  in  seconds')
ylabel('amplitude         in
volts')
title(['sampled signal for ',num2str(fs),' Hz sampling frequency'])
L = 8;
delta = (max(samp_sig) - min(samp_sig))/(L-1);
quant_sig = min(samp_sig) + round((samp_sig-min(samp_sig))/delta)*delta;
```

```
subplot(1,2,1)
plot(samp_t,
samp_sig,'o','linewidth',1.5)grid on
xlabel('time in seconds')
ylabel('amplitude in
volts') title('sampled
signal') subplot(1,2,2)
plot(samp_t,
quant_sig,'x','linewidth',1.5);
xlabel('time')
ylabel('amplitude')
title('quantized
samples')
Bits/Samplenb = log2(L);
i=round((samp_sig-
min(samp_sig))/delta); dig_data_matrix
= de2bi(i,nb);
dig_data = reshape(dig_data_matrix',1,[]);
disp(['The index values for encoding from quantization of the sampled signal are:
',num2str(i)]) disp(['The converted bits from the input analog signal are:
',num2str(dig_data)])
```

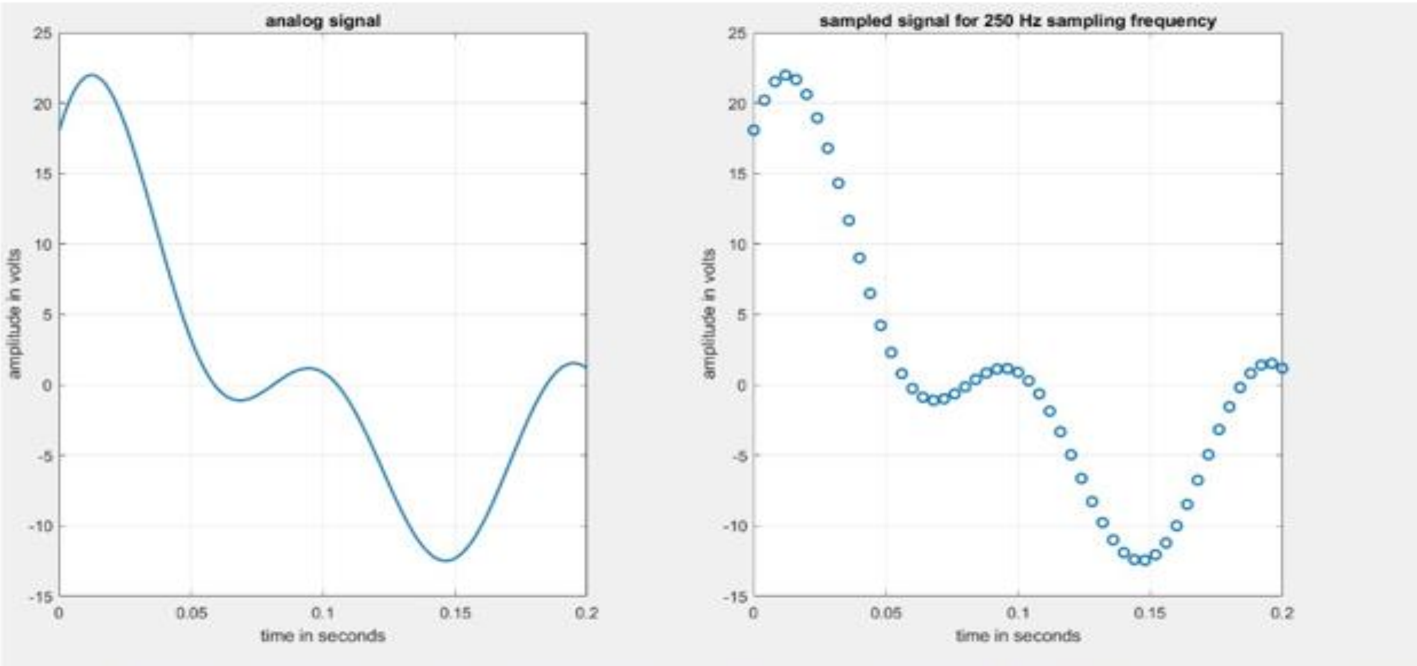

Fig 1: Analog Signal and Sampled signal for 250 Hz.

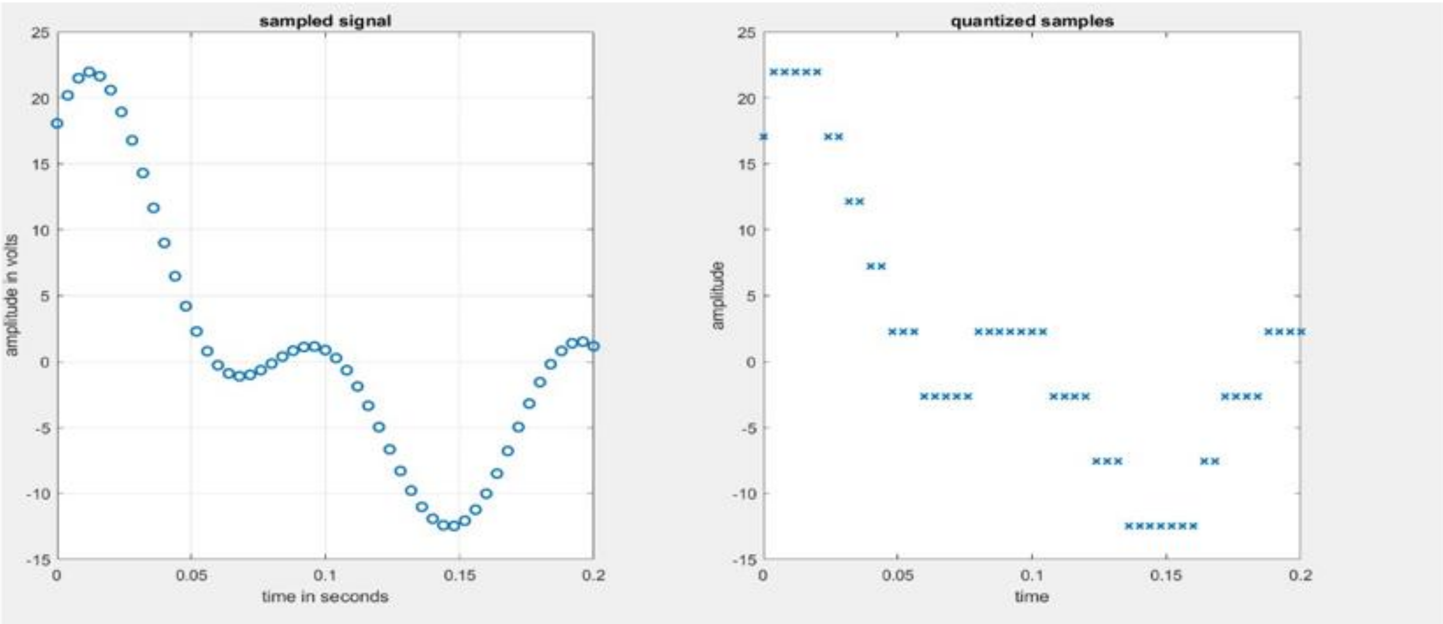

Fig 2: Sampled signal and quantized samples.

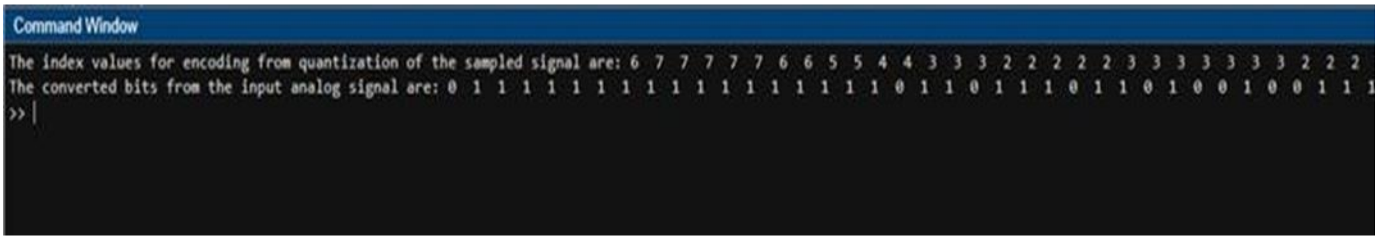

Fig 3: Digital data from an analog signal

Performance Task 3:

$SNR_{db} = SNR$

$=> 25 = 10\log_{10}SNR$

$=> \log_{10}SNR = \frac{25}{10} = 2.5$

$=> SNR = 10^{2.5} = 316.23$

Given,

$t_{max} = 150Hz$ $t_s = (2\times150) = 300Hz$

The minimum value at t_s is 300Hz.

$SNR_{db} = 6.02nb + 1.76$

$\Rightarrow 25 = 6.02nb + 1.76$

$\therefore nb = 3.86 \cong 4$

Data rate = $f_s nb = 300 \times 4 = 1200$ berCapacity = $25 \times \log 2\ (1+316.23)$

$= 238x$

$t_{max} = 150Hz$

$=> t_s \geq 150$

$=> SNR_{db} = 602nb + 1.76$

$\therefore$ Data rate $\leq$ Capacity

Discussion and Conclusion:

In this experiment, we learned different methods of Digital to Analog conversion. PULSE CODE MODULATION (PCM) is the most common technique to change an analog signal to digital data. We implemented the processes of PCM using MATLAB. Analog, quantized and sampled signals have been shown in the report.

References:

- Prakash C. Gupta, “Data communications”, Prentice Hall India Pvt.
- William Stallings, "Data and Computer Communications”, Pearson
- Forouzan, B. A. "Data Communication and Networking. Tata McGraw." (2005).
- AIUB Data Communication Engineering Lab Manual, Report 04