

Department of Mechatronics Engineering
Rajshahi University of Engineering and Technology



Course No.: **MTE-2206.**

Course Title: **Sensor and Instrumentations Sessional.**

Experiment No: 03

Experiment Name: Verification of sampling theorem and analyse the effect of aliasing

Remarks

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Session: 2018-19

Experiment Name: Verification of sampling theorem and analyze the effect of aliasing.

Objectives:

1. To know about sampling theorem and how sampling work.
2. To know about the effect of aliasing.
3. To know about signal processing for converting continuous analog signal to discrete signal.

Theory:

Voltage or amplitude, frequency, and phase are three properties of a signal. Where digital technology is not available, the signals are only represented in analog form. Analog signals are continuous in time and have varying voltage levels at different times. The biggest disadvantage is that the amplitude of the signal changes with the period of the transmission. The digital form of signal representation can circumvent this. The sampling technique can be used to convert an analog signal to a digital signal in this case. [1]

A continuous signal or an analog signal can be represented in the digital version in the form of samples. Here, these samples are also called as discrete points. The sampling theorem can be defined as the conversion of an analog signal into a discrete form by taking the sampling frequency as twice the input analog signal frequency.

The sampling theorem states that a signal must be sampled at least twice as frequently as it was originally. If a continuous time signal has no frequency components greater than f Hz, uniform samples taken at a rate of f_s Hz can be used to determine it completely. [2] Here,

$$f_s \geq 2f$$

or, in term of the sampling period,

$$T = 1/f_s$$

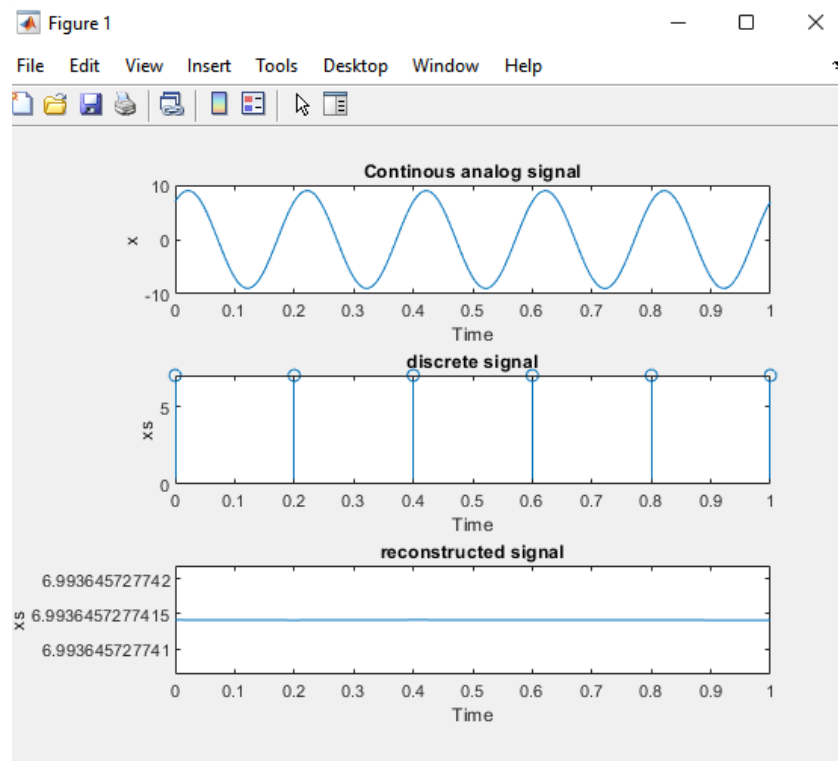
Aliasing is a different phenomenon in signal processing. When two completely separate signals with different frequencies are compared in the course of analog-to-digital conversion, the sampled copies of them appear to be similar. As a result, they're aliases for one another. [3] Aliasing occurs when sampling a signal. Aliasing is when a continuous-time frequency appears as a discrete-time frequency with multiple frequencies. The sampling theorem establishes conditions that prevent aliasing so that a continuous-time signal can be uniquely reconstructed from its samples.[4]

When $f_s=f$,

Code:

```
signalsampling.m
1 - clc;
2 - clear all;
3 - %Continuous analog signal
4 - A= 9;
5 - f=5;
6 - theta=0.89;
7 - t=0:0.005:1;
8 - x=A*sin(2*pi*f*t+theta);
9 - subplot(3,1,1)
10 - plot(t,x)
11 - xlabel("Time")
12 - ylabel("x")
13 - title("Continuous analog signal")
14 - %discrete signal
15 - fs=f;
16 - ts=1/fs;
17 - t1=0:ts:fs*ts;
18 - xs=A*sin(2*pi*f*t1+theta);
19 - subplot(3,1,2)
20 - stem(t1,xs)
21 - xlabel("Time")
22 - ylabel("xs")
23 - title("discrete signal")
24 - %reconstructed signal
25 - subplot(3,1,3)
26 - plot(t1,xs)
27 - xlabel("Time")
```

Output:

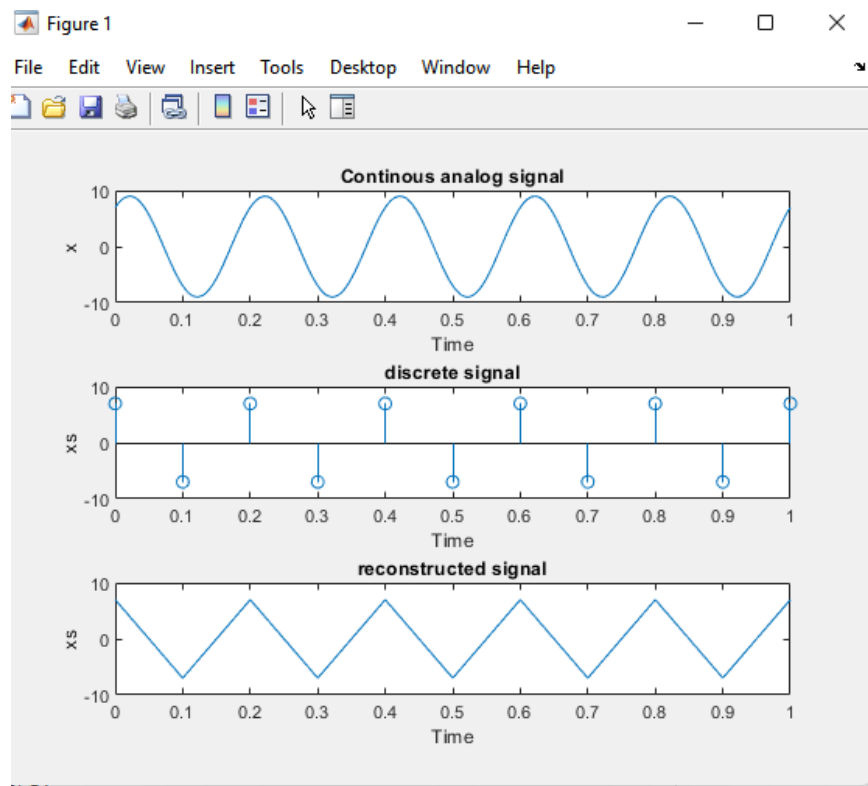


When $f_s=2f$,

Code:

```
signalsampling.m
1 - clc;
2 - clear all;
3 - %Continuous analog signal
4 - A= 9;
5 - f=5;
6 - theta=0.89;
7 - t=0:0.005:1;
8 - x=A*sin(2*pi*f*t+theta);
9 - subplot(3,1,1)
10 - plot(t,x)
11 - xlabel("Time")
12 - ylabel("x")
13 - title("Continuous analog signal")
14 - %discrete signal
15 - fs=2*f;
16 - ts=1/fs;
17 - tl=0:ts:fs*ts;
18 - xs=A*sin(2*pi*f*tl+theta);
19 - subplot(3,1,2)
20 - stem(tl,xs)
21 - xlabel("Time")
22 - ylabel("xs")
23 - title("discrete signal")
24 - %reconstructed signal
25 - subplot(3,1,3)
26 - plot(tl,xs)
27 - xlabel("Time")
```

Output:

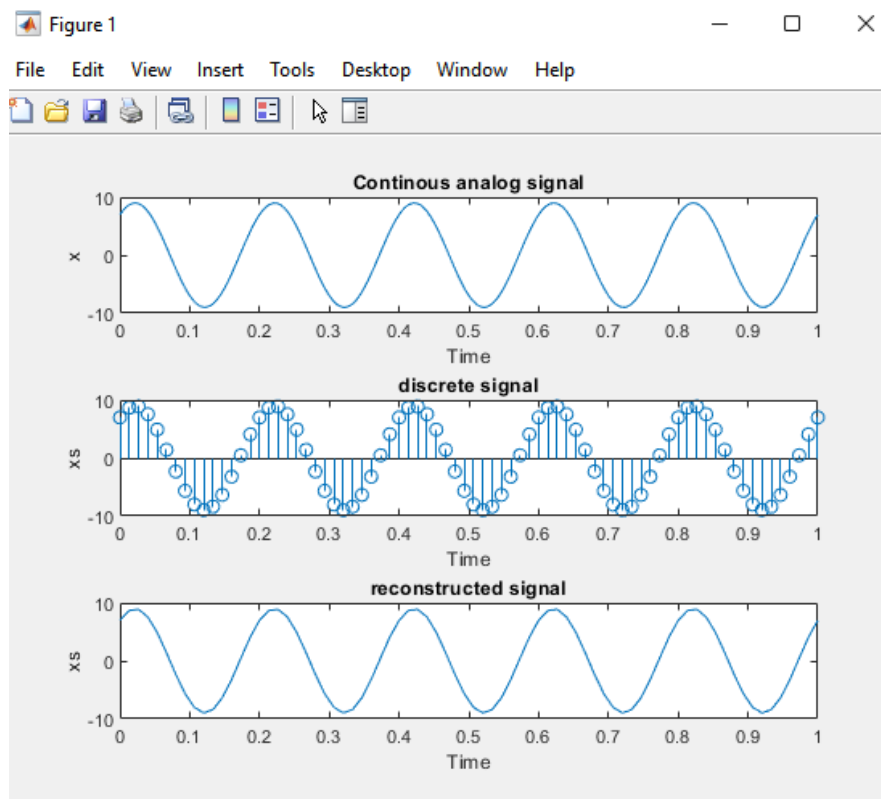


When $fs=15f$,

Code:

```
signalsampling.m
1 - clc;
2 - clear all;
3 - %Continuous analog signal
4 - A= 9;
5 - f=5;
6 - theta=0.89;
7 - t=0:0.005:1;
8 - x=A*sin(2*pi*f*t+theta);
9 - subplot(3,1,1)
10 - plot(t,x)
11 - xlabel("Time")
12 - ylabel("x")
13 - title("Continuous analog signal")
14 - %discrete signal
15 - fs=15*f;
16 - ts=1/fs;
17 - t1=0:ts:fs*ts;
18 - xs=A*sin(2*pi*f*t1+theta);
19 - subplot(3,1,2)
20 - stem(t1,xs)
21 - xlabel("Time")
22 - ylabel("xs")
23 - title("discrete signal")
24 - %reconstructed signal
25 - subplot(3,1,3)
26 - plot(t1,xs)
27 - xlabel("Time")
```

Output:



Discussion and Conclusion: We can see from the signal results section that the discrete signal is fairly closer to the original signal as the number of sample signals are increased. The amount of inaccuracy can be reduced by increasing the value of f_s . So, from this experiment, we get a very clear idea of how sampling theorem work and how aliasing affects the signals. It is very important to understand as it has a huge impact on signal processing and transmission.

References:

1. "What is Sampling Theorem Statement and Its Applications." Available:
<https://www.elprocus.com/sampling-theorem-statement-and-its-applications/>
2. Lai, E. (n.d.). *Sampling Theorem*. Sampling Theorem - an overview | ScienceDirect Topics. <https://www.sciencedirect.com/topics/computer-science/sampling-theorem>.
3. Basics of aliasing in sampled waveforms. Available:
<https://www.testandmeasurementtips.com/basics-of-aliasing-in-sampled-waveforms/>
4. Creative, 3200. (2019, February 8). *Aliasing and the Sampling Theorem Simplified*. ALLSIGNALPROCESSING.COM. <https://allsignalprocessing.com/lessons/aliasing-and-the-sampling-theorem-simplified/>.