Experiment No: 2 Date: 12/08/2024

Verification of Sampling Theorem

Aim

1. To verify the sampling theorem and plot 4 signals
2. To verify the sampling theorem using cosine signal

Theory

SAMPLING THEORY

Sampling theorem states that "continues form of a time-variant signal can be represented in the discrete form of a signal with help of samples and the sampled (discrete) signal can be recovered to original form when the sampling signal frequency Fs having the greater frequency value than or equal to the input signal frequency Fm".

NYQUIST SAMPLING

The Nyquist–Shannon sampling theorem is an essential principle for digital signal processing linking the frequency range of a signal and the sample rate required to avoid a type of distortion called aliasing. The theorem states that the sample rate must be at least twice the bandwidth of the signal to avoid aliasing. In practice, it is used to select band-limiting filters to keep aliasing below an acceptable amount when an analog signal is sampled or when sample rates are changed within a digital signal processing function.

UNER SAMPLING

In signal processing, undersampling or bandpass sampling is a technique where one samples a bandpass-filtered signal at a sample rate below its Nyquist rate (twice the upper cutoff frequency), but is still able to reconstruct the signal.

OVER SAMPLING

In signal processing, oversampling is the process of sampling a signal at a sampling frequency significantly higher than the Nyquist rate. Theoretically, a bandwidth-limited signal can be perfectly reconstructed if sampled at the Nyquist rate or above it. The Nyquist rate is defined as twice the bandwidth of the signal. Oversampling is capable of improving resolution and signal-to-noise ratio, and can be helpful in avoiding aliasing and phase distortion by relaxing anti-aliasing filter performance requirements.

Program

1. To verify the sampling theorem and plot 4 signals

```
clc;
clf;
Am=input('Enter the amplitude of the signal=');
Cy=input('Enter the number of cycles=');
Fm=input('Enter the frequency=');
t=0:(1/(Fm*Fm)):Cy/Fm;
Amplitude=Am*sin(2*pi*Fm*t);
subplot(221);
plot(t,Amplitude)
grid on;
xlabel('Time');
ylabel('Amplitude');
title('Sampling');
Fs1=1.5*Fm;
t1=0:(1/(Fs1)):Cy/Fm;
Amplitude1=Am*sin(2*pi*Fm*t1);
subplot(222);
plot(t1,Amplitude1)
grid on;
xlabel('Time');
ylabel('Amplitude');
title('Under Sampling');
Fs2=3*Fm;
t2=0:(1/(Fs2)):Cy/Fm;
Amplitude2=Am*sin(2*pi*Fm*t2);
subplot(223);
plot(t2,Amplitude2)
grid on;
xlabel('Time');
ylabel('Amplitude');
title('Nyquist Sampling');
Fs3=20*Fm;
t3=0:(1/(Fs3)):Cy/Fm;
Amplitude3=Am*sin(2*pi*Fm*t3);
subplot(224);
plot(t3,Amplitude3)
grid on;
xlabel('Time');
ylabel('Amplitude');
title('Over Sampling');
```

2. To verify the sampling theorem using cosine signal

```
clc;
clf;
Am=input('Enter the amplitude of the signal=');
Cy=input('Enter the number of cycles=');
Fm=input('Enter the frequency=');
t=0:(1/(Fm*Fm)):Cy/Fm;
Amplitude=Am*cos(2*pi*Fm*t);
subplot(221);
plot(t,Amplitude)
grid on;
xlabel('Time');
ylabel('Amplitude');
title('Sampling');
Fs1=1.5*Fm;
t1=0:(1/(Fs1)):Cy/Fm;
Amplitude1=Am*cos(2*pi*Fm*t1);
subplot(222);
plot(t1,Amplitude1)
grid on;
xlabel('Time');
ylabel('Amplitude');
title('Under Sampling');
Fs2=3*Fm;
t2=0:(1/(Fs2)):Cy/Fm;
Amplitude2=Am*cos(2*pi*Fm*t2);
subplot(223);
plot(t2,Amplitude2)
grid on;
xlabel('Time');
ylabel('Amplitude');
title('Nyquist Sampling');
Fs3=20*Fm;
t3=0:(1/(Fs3)):Cy/Fm;
Amplitude3=Am*cos(2*pi*Fm*t3);
subplot(224);
plot(t3,Amplitude3)
grid on;
xlabel('Time');
ylabel('Amplitude');
title('Over Sampling');
```

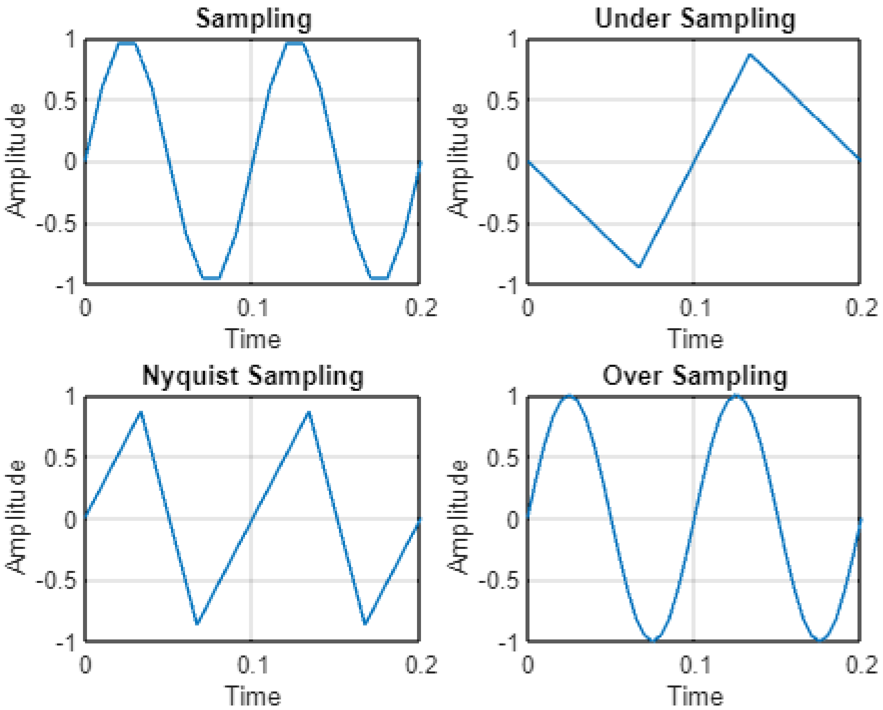
Result

Implemented and verified the theorem and generated the signals

https://github.com/Brindha192tkm/dspl-2/blob/EXP2OP/Experiment%202

Observation

1. **INPUT =** Am=1, Cy=2, Fm=10
   OUTPUT=



2. **INPUT=** Am=1, Cy=2, Fm=10
   OUTPUT=