VIVEKANAND EDUCATION SOCIETY'S INSTITUTE OF TECHNOLOGY

CHEMBUR, MUMBAI 400071

PRINCIPLES OF COMMUNICATION ENGINEERING

(ELX 405)

COURSE PROJECT

"VERIFICATION OF SAMPLING THEOREM USING MATLAB"

SUBMITTED BY

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1. INTRODUCTION

Sampling Theorem: In the field of digital signal processing, Sampling theorem is a fundamental principle that connects Analog signals with Digital signals. It describes the rate at which a continuous or analog signal needs to be sampled such that all the information from the signal is captured without any information being lost during sampling. The analog signal should be reconstructed back from the sampled signal.

The theorem states that, *"For a faithful representation of analog signal, the Sampling rate or Nyquist rate (f_s) should be at least twice the rate of analog signal (f_m)."* i.e.

$$\boxed{\boldsymbol{f_s \geq 2f_m}}$$

To perfectly reconstruct the original signal, we need to sample it as many times as possible. The basic idea of this theorem is that we record multiple instantaneous values of analog signal (Samples) at such a rate that will allow us to accurately reconstruct the original signal from the sampled signal.

2. USE OF MATLAB IN VERIFICATION OF SAMPLING THEOREM

MATLAB is a programming-based simulation software that allows many operations like Matrix multiplication, plotting functions and data, implementing algorithms, etc.

Here MATLAB (Version R2018a) has been used as a tool to generate an analog signal and to sample it, in order to study and verify the Sampling theorem for the three conditions (cases) on the sampling rate (f_s) which are as follows:

Case 1: $\boldsymbol{f_s < 2f_m}$

In this case, the side frequencies from one harmonic fold over into the sideband of other harmonic. This results in "aliasing" i.e. inability to distinguish between the two signals.

Case 2: $\boldsymbol{f_s = 2f_m}$

As long as this case is valid, none of the side frequency from one harmonic folds over into the other harmonic. No aliasing effect is seen.

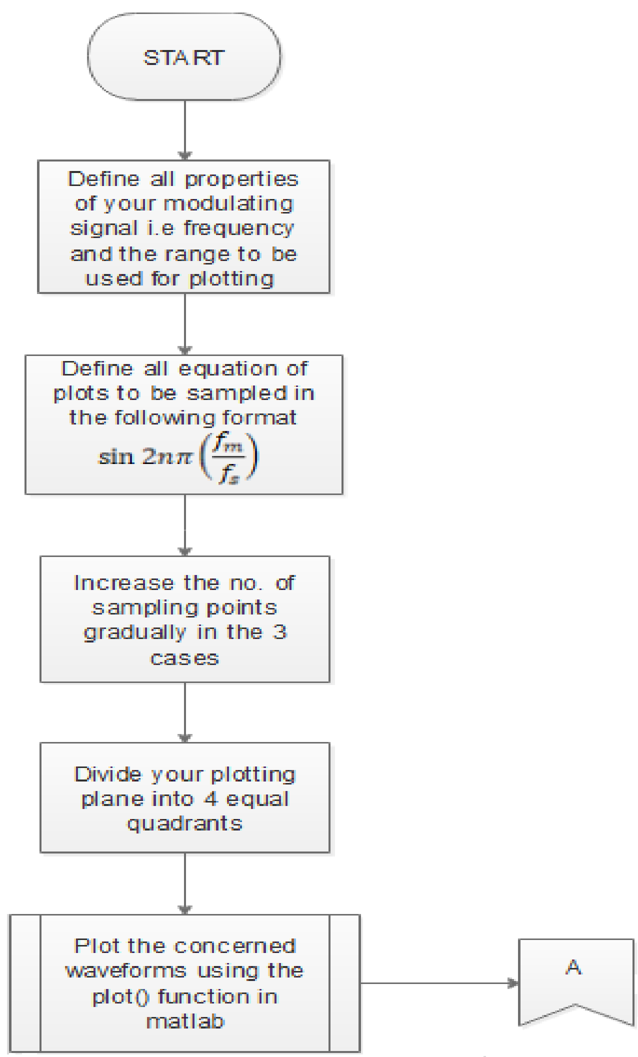
Case 3: $\boldsymbol{f_s > 2f_m}$

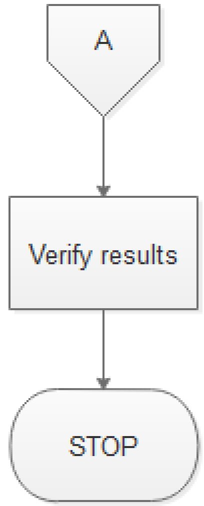
No aliasing effect is seen. A large gap is formed between two successive harmonics. It is called as "Guard Band".

3. ALGORITHM

1. Generate Modulating (analog) signal and define all its properties like nature of signal, frequency, range, etc.
2. Give equations of all the plots to be sampled in the standard form i.e. sinusoidal form.
3. Define the sampling intervals (no. of sampling points) for all 3 cases. As a thumb rule keep the number of sampling points
4. Divide the plotting plane into 4 equal parts to show the Modulating signal along with the 3 cases of sampling.
5. Plot all the waveforms and verify results.

4. FLOWCHART





5. CODE IN MATLAB

```
%Define properties for modulating signal
fm=0.02;
t=-100:0.1:100;
subplot(2,2,1);
plot(t,sin(2*pi*fm*t));
title('Modulating Signal')
xlabel('Time')
ylabel('Amplitude')

%Plot various cases

%Case 1: fs<2fm
n=-2:2;                 %no. of sampling points=5
subplot(2,2,2);
stem(n,sin(2*pi*n));    %fm/fs=1
hold on;
subplot(2,2,2);
plot(n,sin(2*pi*n));
title('Case 1: f_s < 2f_m')
xlabel('Time')
ylabel('Amplitude')
```

```
%Case 2: fs=2fm
n=-4:4;                  %no. of sampling points=9
subplot(2,2,3);
stem(n,sin(0.5*pi*n));  %fm/fs=0.5
hold on;
subplot(2,2,3);
plot(n,sin(0.5*pi*n));
title('Case 2: f_s = 2f_m')
xlabel('Time')
ylabel('Amplitude')

%Case 3: fs>2fm
n=-10:10;               %no. of sampling points=21
subplot(2,2,4);
stem(n,sin(.25*pi*n)); %fm/fs=0.25
hold on;
subplot(2,2,4);
plot(n,sin(.25*pi*n));
title('Case 3: f_s > 2f_m')
xlabel('Time')
ylabel('Amplitude')
```

6. RESULT

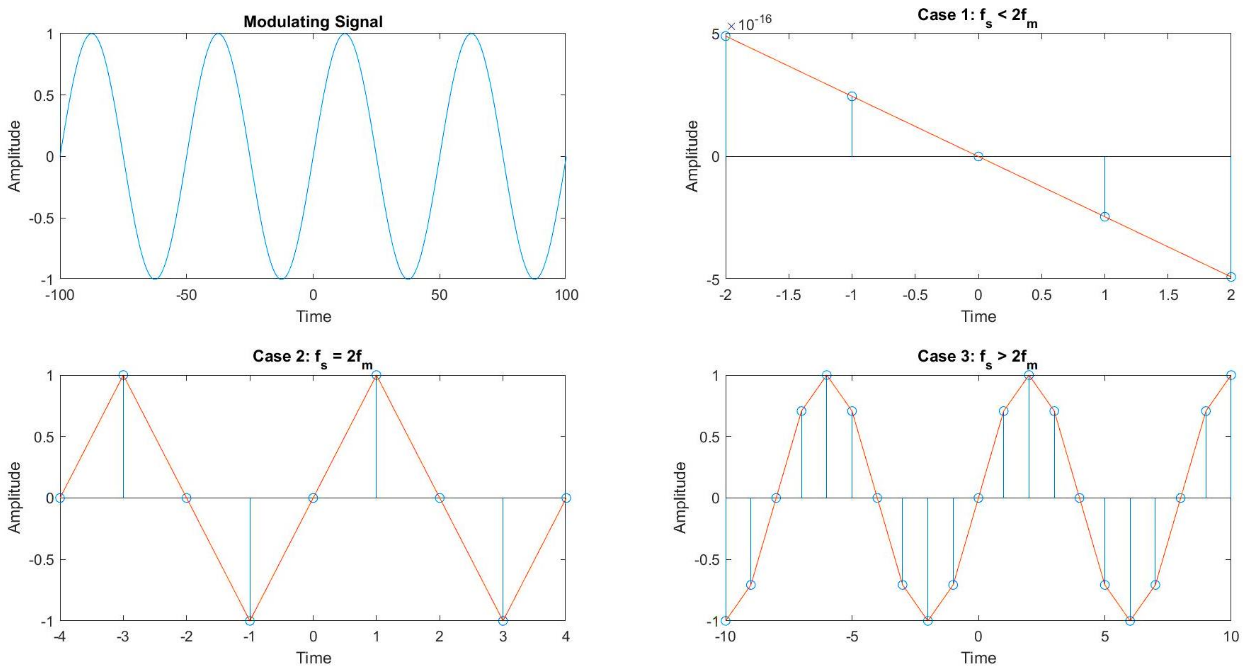


Figure 6.1 (Starting from top left) (a) Modulating Signal (b) $f_s < 2f_m$ with five samples (c) $f_s = 2f_m$ with nine samples and (d) $f_s > 2f_m$ with twenty one samples

7. APPLICATIONS OF SAMPLING THEOREM

a) Used extensively in music industry for maintaining the quality of sound recording.

b) In television broadcasting, the contents of a Standard Definition(SD) channels and High Definition(HD) channel are recorded at different pixel quality under the criteria of Sampling theorem. Here, the frame rate is taken as a measure of sampling rate.

c) Used in volume rendering technique to display 2D projection of a 3D content. Here, the “voxels” of a 3D image are sampled for accurate projection.