FR. CONCEICAO RODRIGUES COLLEGE OF ENGINEER
Department of Electronics and Computer Science

Expt 6: Verification of Sampling Theorem

Course, Subject & Experiment Details

Timeline (3)	Understanding (3)	Self Efforts (4)	Total (10)

Student's Name		Roll No.	
Academic Year	2021 – 22	Estimated Time	2 Hours
Course & Semester	T.E. (ECS) Sem. V	Subject Name	Communication Engineering Laboratory
Unit No.	4	Chapter Title	Pulse modulation
Experiment Type	Software Performance	Subject Code	ECL 501

1. Aim of the Experiment:

To Study the effect of sampling frequency on the reconstructed signal.(use Matlab)

2. Apparatus:

Matlab software

3. Expected Outcome of Experiment

Students will be able to investigate the effect of the sampling frequency on the reconstructed signal.

4. Theoretical Description

An analog signal can be converted to a discrete time signal by a process called sampling. The sampling theorem specifies the minimum-sampling rate at which a continuous-time signal needs to be uniformly sampled so that the original signal can be completely recovered or reconstructed by these samples alone and can be stated as,

A continuous time signal can be represented in its samples and can be recovered back when sampling frequency f_s is greater than or equal to the twice the highest frequency component of message signal. i. e.

$$fs \geq 2fm.$$

If we reduce the sampling frequency fs less than fm, the side bands and the information signal will overlap and we cannot recover the information signal simply by low pass filter. This phenomenon is called fold over distortion or aliasing.There are two methods of sampling. (1) Natural sampling (2) Flat top sampling. Sample & Hold circuit holds the sample value until the next sample is taken. Sample & Hold technique is used to maintain reasonable pulse energy.

The process of sampling can be understood with the help of figure 1.

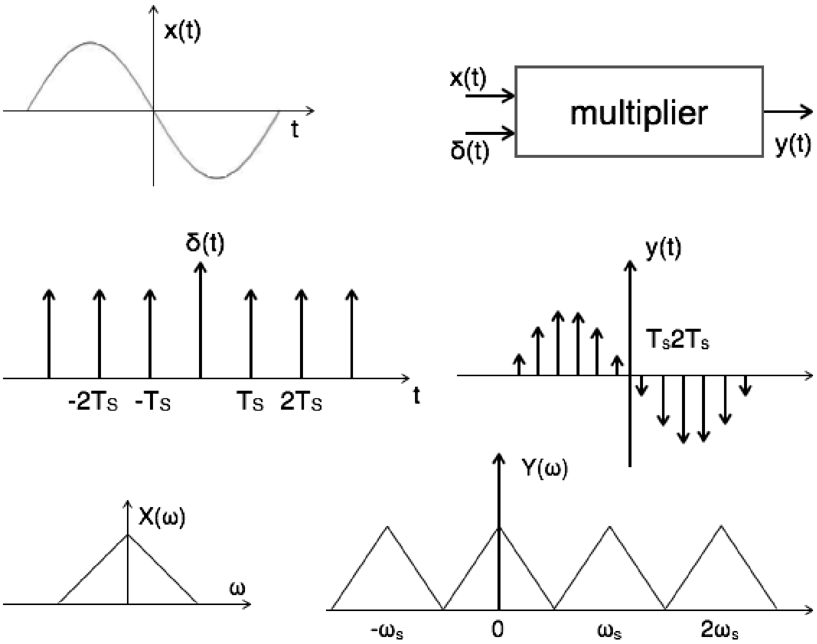


Figure 1: Sampling process

Consider a continuous time signal x(t). The spectrum of x(t) is a band limited to f_m Hz i.e. the spectrum of x(t) is zero for $|\omega|>\omega_m$.

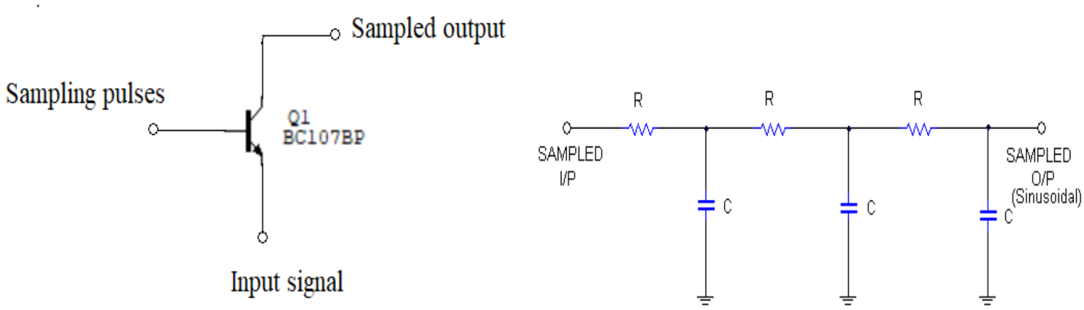
Sampling of input signal x(t) can be obtained by multiplying x(t) with an impulse train δ(t) of period T_s. The output of multiplier is a discrete signal called sampled signal which is represented by y(t).

$$y(t)= x(t).\delta(t)= \frac{1}{T_s}[x(t)+2\cos\omega_s t.x(t)+2\cos 2\omega_s t.x(t)+2\cos 3\omega_s t.x(t)......] \qquad (1)$$

In frequency domain,

$$Y(\omega)=\frac{1}{T_s}[X(\omega)+X(\omega-\omega_s)+X(\omega+\omega_s)+X(\omega-2\omega_s)+X(\omega+2\omega_s)+...]. \qquad (2)$$

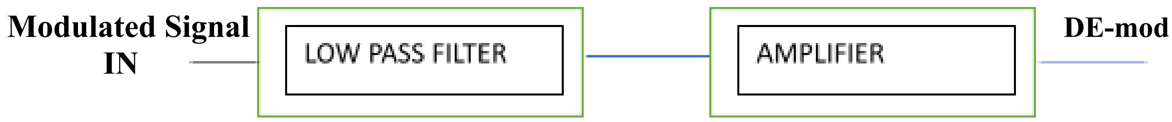
The spectrum of y(t) is shown in figure 1. From equation (1), it is clear that if y(t) is passed through a low pass filter having cut-off frequency slightly above f_m, x(t) can be reconstructed properly. Hence the reconstruction circuit consists of a simple low pass filter. The circuit diagram for sampling and reconstruction are shown in figure 2.



(a) (b)

Figure 2: (a) sampling circuit, (b) reconstruction circuit

DEMODULATION:-



6. Procedure:

1. Connect the sampling using Circuit Maker as per the circuit diagram shown in Fig.1(a).
2. Apply sinusoidal signal of 1kHz frequency and amplitude 12 Vp as the input signal, and sampling signal of frequency 2, 4, 8, and 16kHz and amplitude 10 Vp.
3. Note down the sampled waveforms for each sampling frequency.
4. To design the low pass filter, find the value of R from $f_{cutoff} = \frac{1}{2\pi RC}$, taking C = 0.01µF, and f_{cutoff} = 1.5kHz.
5. To reconstruct x(t) with minimum distortion, connect the cascade of low pass filters to the output of the sampling circuit and note down the reconstructed waveforms for all sampling frequencies.

7. Simulation Steps

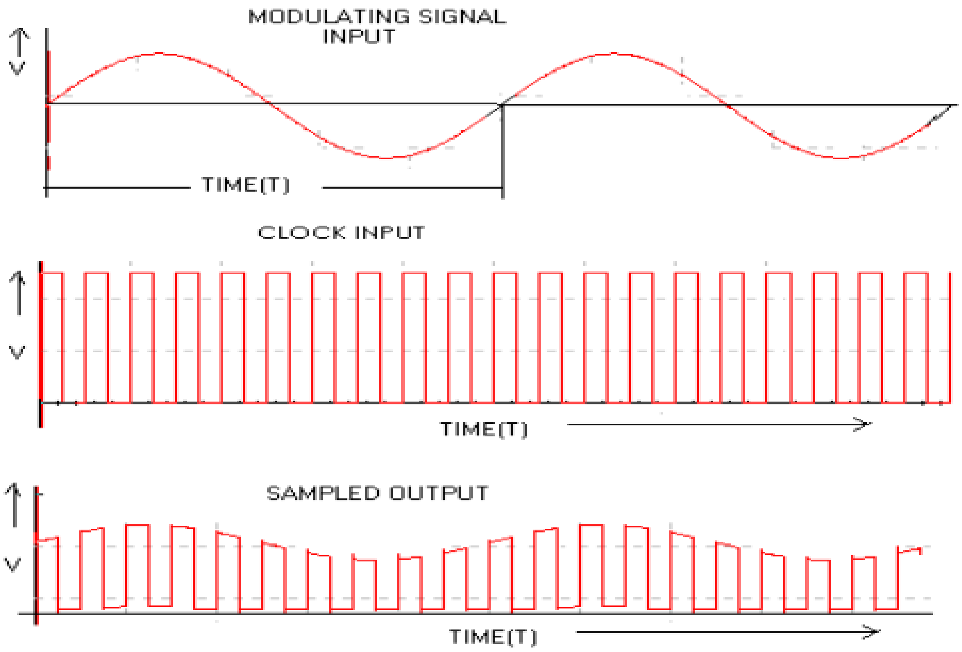
1. Create a pulse train x(t)with desired sampling time period(As per Nyquist rate) ;
2. Generate a information signal sine/cos wave m(t).
3. change sampling frequency fs to different value and observe sampled output

8. GRAPHS:

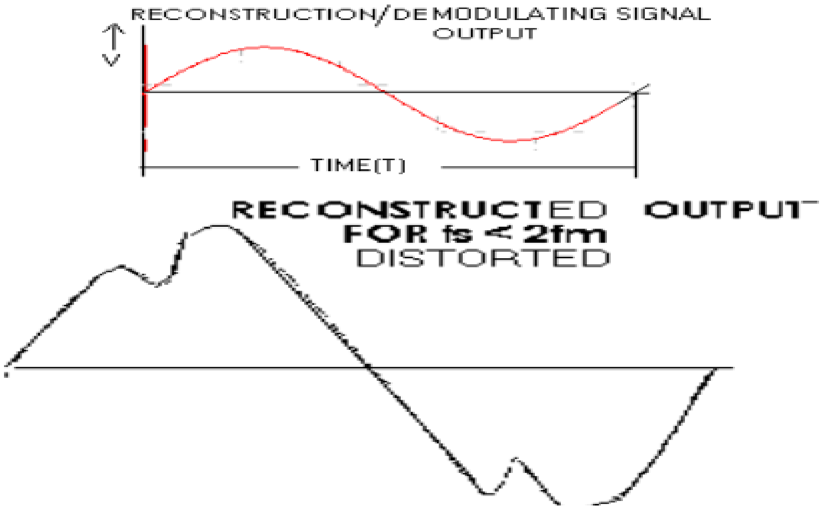
1. Input signal waveform
2. The sampled waveforms for 2, 4, 8, and 16kHz sampling frequencies.
3. The reconstructed waveforms for these sampling frequencies.

Expected OUTPUT of SIMULATION:-

Below Wavforms for **fs > 2fm**



Demodulated Output



9. Conclusion

Program

```
clc;clear all;
fm=10;
ts=[1:20]/1000;
%fs=2fm
t=[1:20]/20;
m=sin(2*pi*fm*t);
subplot(4,1,1);
stem(ts,m);
hold on
plot(ts,m);
%fs=4fm
t=[1:20]/40;
m=sin(2*pi*fm*t);
subplot(4,1,2);
stem(ts,m);
hold on
plot(ts,m);
%fs=8fm
t=[1:20]/80;
m=sin(2*pi*fm*t);
subplot(4,1,3);
stem(ts,m);
hold on
plot(ts,m);
%fs=16fm
t=[1:20]/160;
m=sin(2*pi*fm*t);
subplot(4,1,4);
stem(ts,m);
hold on
plot(ts,m);
```

Output waveforms