(DEEMED TO BE UNIVERSITY)

SIGNALS & COMMUNICATION SYSTEMS

24EC2105

STUDENT ID:

STUDENT NAME:

ACADEMIC YEAR: 2025-26

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A.Y. 2025-26 LAB CONTINUOUS EVALUATION

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S.No	Date	Experiment Name	Pre-Lab (10M)	In-Lab (25M)			Post-Lab (10M)	Viva Voce (5M)	Total (50M)	Faculty Signature
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Signals & Communication Systems - Lab-2

Generation of Elementary C.T. Signals: Time-Domain Representation

Introduction: The purpose of this lab experiment is to explore the continuous time signals using digital computers and the Matlab software environment. A continuous-time signal takes on a value at every point in time. The basic elementary signals place an important role in analyzing signals and testing the systems.

Objectives:

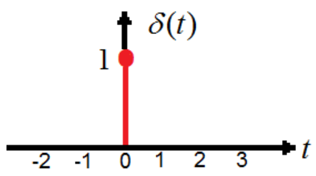
- To generate and display basic continuous-time signals.
- To generate and display Random signals.

Requirements: Digital Computer with MATLAB software.

Basic theory: The basic elementary signals are much useful in signal processing for analysis purpose. These signals are described both in mathematical and graphical representations as below.

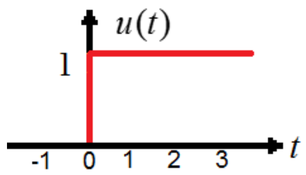
(a) The Singularity Functions:

The Unit Impulse Function: A continuous time **impulse signal** is defined as $\delta(t)=\begin{cases}1, & t=0\\ 0, & t\neq 0\end{cases}$



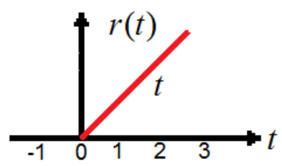
The Unit Step Function: A continuous time **unit signal** is defined as

$$u(t)=\begin{cases}1, & t\geq 0\\ 0, & t<0\end{cases}$$



The Unit Ramp Function: A continuous time **ramp signal** is defined as

$$r(t)=\begin{cases}t, & t\geq 0\\ 0, & t<0\end{cases}$$

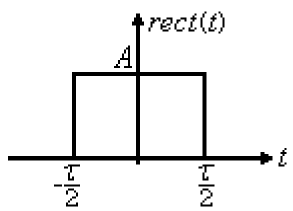


Example3: Develop Matlab code for display the unit ramp function.

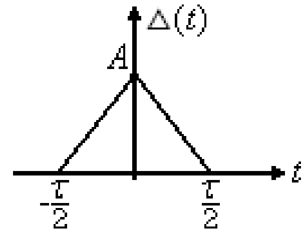
(b) Other useful signals

Gate function: A gate function is denoted by $rect(t)$ or $\Pi(t)$ of height A and width τ , centered at the origin. The mathematical expression is defined as

$$rect(t)=\begin{cases}A, for & -\frac{\tau}{2}<t<\frac{\tau}{2}\\ 0, & elsewhere\end{cases} \text{ or } rect(t)=\begin{cases}A, for & |t|\leq\tau/2\\ 0, for & |t|>\tau/2\end{cases}$$

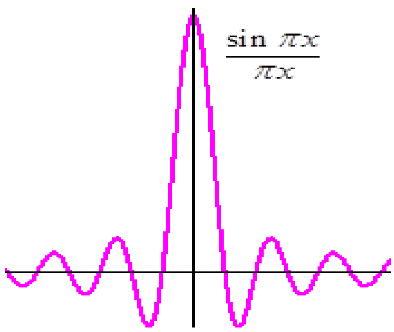


Triangular function: A triangular function is denoted by Δ(t), of height A and width τ , centered at the origin. The mathematical expression is defined as △



$$\Delta(t) = \begin{cases} \frac{2A}{\tau}t + A, for \ -\frac{\tau}{2} < t < 0 \\ -\frac{2A}{\tau}t + A, for \ 0 < t < \frac{\tau}{2} \end{cases}$$

Interpolation function: sinc(*x*) or Sa(x). The function $\frac{\sin x}{x}$ is the 'sine over argument' function denoted by sinc(*x*). It is also known as filtering or interpolating function.



- ❖ sinc(*x*) is an even function.
- ❖ Using L'Hospital' rule, sinc(0) = 1.
- ❖ sinc(*x*) is also denoted as Sa(x) in the literature. Some authors define as $\frac{\sin \pi x}{\pi x}$.

Examples:

Example 1: Develop Matlab code for display the unit impulse function.

```
clear all; close all; clc;
t = -3:0.0001:3;
% Define unit delta function
y = ud(t,0);
figure();
%%%% uncomment the following two lines and observe the results
% plot(t,zeros(size(t)),'k','LineWidth',1); hold on; %Horizontal line as time index
% plot(zeros(size(t)),t,'k','LineWidth',1); hold on; %Vertical line as magnitude index
plot(t, y,'b','LineWidth',3); grid on;
set(gca,'fontsize',14);
xlabel('time  ----->');ylabel('Amplitude ----->');
title('delta(t)');axis([-3 3 -0.1 1.2]);
```

Example 2: Develop Matlab code for display the unit step function.

```
clear all; close all; clc;
t = -3:0.0001:3;
% Define unit step function
y = us(t,0);
figure();
%%%% uncomment the following two lines and observe the results
```

```
% plot(t,zeros(size(t)),'k','LineWidth',1); hold on; %Horizontal line as time index
% plot(zeros(size(t)),t,'k','LineWidth',1); hold on; %Vertical line as magnitude index
plot(t, y,'r','LineWidth',3); grid on;
set(gca,'fontsize',14);
xlabel('time  ----->');ylabel('Amplitude ----->');
title('Unti step function');axis([-3 3 -0.1 1.2]);
```

Example 3: Develop Matlab code for display the unit ramp function.

```
clear all; close all; clc;
t = -2:0.0001:6;
% Define unit ramp function
y = ur(t,0);
figure();
%%%% uncomment the following two lines and observe the results
% plot(t,zeros(size(t)),'k','LineWidth',1); hold on; %Horizontal line as time index
% plot(zeros(size(t)),t,'k','LineWidth',1); hold on; %Vertical line as magnitude index
plot(t, y,'m','LineWidth',3); grid on;
set(gca,'fontsize',14);
xlabel('time  ----->');ylabel('Amplitude ----->');
title('Unti Ramp function');axis([-2 6 -0.2 6]);
```

Example 4: Develop Matlab code for display a rectangular pulse having unit height and unit width centered at origin.

```
clear all; close all; clc;
%======== Rectangular pulse ==========================
t = -3:0.001:3;
% Manual definition of rectangular pulse of width 1 centered at 0
y = double(abs(t) <= 0.5);
figure();
% Horizontal and vertical lines
plot(t, zeros(size(t)), 'k', 'LineWidth', 1); hold on;
plot(zeros(size(t)), t, 'k', 'LineWidth', 1); hold on;
% Rectangular pulse plot
plot(t, y, 'b', 'LineWidth', 3);
set(gca, 'fontsize', 14);
title('Rectangular Pulse'); grid on;
```

```
axis([-1 1 -0.1 1.2]);
xlabel('time  ----->');
ylabel('Amplitude ----->');
%========================================
```

Example 5: Develop Matlab code for display a triangular pulse having unit height and unit width centered at origin.

```
clear all; close all; clc;
t = -2:0.001:2;
y = tripuls(t, 1);  % 1 is the width of the triangle
figure;
plot(t, y, 'b', 'LineWidth', 2);
grid on;
title('Triangular Pulse');
xlabel('Time');
ylabel('Amplitude');
axis([-2 2 -0.1 1.1]);
```

Example 6: Develop Matlab code for display a sync signal.

```
clear all; close all; clc;
clear all; close all; clc;
t = (-pi:0.001:pi)/100;j = sqrt(-1);
A = 2; F0 = 100;
x = A*sin(2*pi*F0*t)./(2*pi*F0*t);
figure();
plot(t,x,'m','LineWidth',2.5);grid on
xlabel('time  ----->');ylabel('Amplitude ----->');
title('sinc(2*pi*f*t)');set(gca,'fontsize',14);
axis([-0.03, 0.03,-0.6 2.15]);
```

Pre lab work:

(a) Study and understand the basic theory required for this lab.

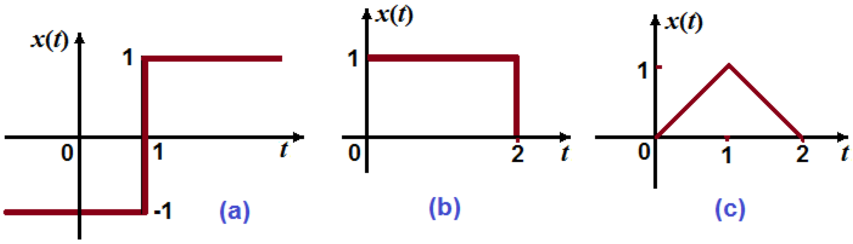
(b) Run the Matlab scripts given for examples and study the results.

(c) Perform the analytical solutions for the given laboratory exercises.

(d) Prepare the Matlab codes for the given laboratory exercises well before the commencement of lab scheduled time.

Laboratory Exercise 2

Q1. Write mathematical expressions in functional form for the signals shown below. Develop Matlab scripts, simulate and plot the results.



Q3. Explain the following

a) **What is the purpose of command clear all?**

b) **What is the purpose of command close all?**

c) **What is the purpose of command clc?**

d) **What is the purpose of command clf?**

e) **What is the purpose of command k = input('enter the samples = ');?**

f) **What is the purpose of symbol ;? If we remove this symbol what will happen?**

g) **What is the purpose of symbol :? Explain the meaning of the following (i) 1:10, (ii) 1:1:10 (iii) 1:0.1:10.**

h) **What is the meaning of pi? What is its value?**

i) **What is the purpose of command plot?**

j) **What is the purpose of command subplot? Differentiate from plot command.**

k) **What is the purpose of command figure?**

l) **What is the purpose of command stem? Differentiate from plot command.**

m) **What is the meaning of fill in the stem command?**

n) **What is the meaning of letters r, g, b, m, and c in the plot or stem command?**

o) **To plot the curve in black color, what is the letter in the plot command to be used?**

p) **What is the meaning of LineWidth, 1.5 in the plot or stem command? What is the meaning of numeric value 1.5? If this value changes, what will happen?**

q) **What are the purpose of commands xlabel and ylabel?**

r) **What is the purpose of command axis?**

s) **What is the purpose of command title?**

t) **What is the purpose of command legend? If we remove this command, what will happen?**

u) **What is the purpose of command grid on? If we remove this command, what will happen?**

v) **Explain the command axis([-1, k,-0.5 k]) in the ramp sequence.**

w) **Differentiate between the symbols * and .*.**

x) **Explain the following commands used for the real speech signals? wavread, wavwrite, auread, and auwrite, audioplayer and audiorecorder.**

y) **What are the purpose of commands sound and soundsc?**

z) **How do you display the part of the speech signal, explain?**

Ongoing Lab:

(a) Type the Matlab codes prepared for the exercises in the computer and observe the results.

(b) Compute the all the tasks given in exercises.

(c) Note the results and store the necessary figures and data for the lab report.

Post Lab:

(a) Complete the lab work in all aspects in the given specified lab time.

(b) Answer the given questions.

(c) Submit the lab report to the lab in-structure and get the signature in time.

(d) Type the complete description of commands used in the lab.

Due date:

Date of submission:

Student Id. No: Section:

Name of the student. Signature of student

Pre-lab Session work (15M) Lab experiment	In-Lab Session work (15M) Project	Post Lab session work (10M)	Viva (10M)	Total Marks 50M

Remarks:

Marks awarde

Date: **Signature of the Instructor**

Signals & Communication Systems -Lab-3

Manipulation / Operations of Continuous Time signals

Objectives:

- To compute and display basic operations on Continuous Time signals.
  - Amplitude scaling
  - Time shifting
  - Time scaling
  - Time reversal
  - Combined operations
- To determine the energy and power of given signals / sequences

Requirements: Digital Computer with MATLAB software.

Basic theory:

Basic Operations on signals: The operations performed on signals can be broadly classified into two types: Operations on dependent variables, and Operations on independent variables

(a) Operations on dependent variables: The operations of the dependent variable can be classified into five types: amplitude scaling, addition, multiplication, integration and differentiation.

Amplitude scaling: Amplitude scaling of a signal $x(t)$ given by equation

$$y(t) = ax(t)$$

results in amplification of $x(t)$ if a >1, and attenuation if a <1.

Signal Addition: The addition of signals is given by equation $y(t) = x_1(t) + x_2(t)$.

Signal Multiplication: The multiplication of signals is given by the simple equation

$$x(t) = x_1(t)x_2(t) .$$

(b): Operations on independent variables: There are various kind of operations on independent variables illustrated below.

Time scaling: Time scaling operation is given by equation $y(t) = x(at)$. This operation results in expansion in time for $a < 1$, and compression in time for $a > 1$.

Time reflection: Time reflection is given by equation $y(t) = x(-t)$.

Time shifting: The equation representing time shifting operation is given by $y(t) = x(t - t_0)$. If t_0 is positive constant, the operation is right shift. If t_0 is negative the operation is left shift.

Time shifting and scaling: The combined transformation of shifting and scaling is represented by an equation $y(t) = x(at - t_0)$. The time shifting operation is performed first and time scaling operation will be done in the second step.

Even (or Symmetric) and Odd (anti-symmetric) signals

$x(t) = x_e(t) + x_o(t)$, where $x_e(t) = \frac{1}{2}\{x(t) + x(-t)\}$ and $x_o(t) = \frac{1}{2}\{x(t) - x(-t)\}$

where again, $x_e(t)$ is an even signal and $x_o(t)$ is an odd signal.

Energy and Power of signals

The normalized energy content 'E' of a signal $x(t)$ is defined as $E = \int_{-\infty}^{\infty} |x(t)|^2 \, dt$.

The normalized average power 'P' of a signal is defined as $P = \underset{T\to\infty}{Lt} \frac{1}{T} \int_0^T |x(t)|^2 \, dt$.

Examples:

Ex 3.1: Scalar Multiplication: Amplification and Attenuation.

(a) Develop Matlab codes to sketch and label the following signals.

(i) $\delta(t)$ (ii) $1.2\delta(t)$ (iii) $-2\delta(t)$

```
clear all; close all; clc;
t = -3:0.0001:3;
y1 = ud(t,0);
y2 = 1.2*ud(t,0);
y3 =-2*ud(t,0);
figure();set(gca,'fontsize',12);
subplot(1,3,1); plot(t, y1,'b','LineWidth',2);title('delta(t)');axis([-3 3 -0.1 1.5]); axis square
subplot(1,3,2); plot(t, y2,'m','LineWidth',2);title('1.2delta(t)');axis([-3 3 -0.1 1.5]); axis square
subplot(1,3,3); plot(t, y3,'k','LineWidth',2);title('-2delta(t)');axis([-3 3 -2.2 0.5]); axis square
```

(b) Sketch and label the following signals using Matlab

(i) $r(t)$ (ii) $-r(t)$ (iii) $1.2r(t)$

```
clear all; close all; clc;
t = -3:0.0001:3;
y1 = ur(t,0);
y2 = -ur(t,0);
y3 = 1.2*ur(t,01);
figure();set(gca,'fontsize',12);
subplot(1,3,1); plot(t, y1,'b','LineWidth',2);
title('Ramp r(t)');axis([-3 3 -0.1 1.5]); axis square
subplot(1,3,2); plot(t, y2,'m','LineWidth',2);
title('-r(t)');axis([-3 3 -3 0.5]); axis square
subplot(1,3,3); plot(t, y3,'k','LineWidth',2);
title('1.2r(t)');axis([-3 3 -0.1 1.5]); axis square
```

Ex3.2: Time Shifting of CT Sequences:

(a) Sketch and label the following signals using Matlab

(i) $u(t)$ (ii) $u(t-2)$ (iii) $u(t+1)$

```
clear all; close all; clc;
t = -3:0.0001:3;
y1 = us(t,0);
y2 = us(t,2);
y3 = us(t,-1);
figure();
subplot(1,3,1);
%%%% uncomment the following two lines and observe the results
plot(t,zeros(size(t)),'k','LineWidth',1); hold on; %Horizontal line as time index
plot(zeros(size(t)),t,'k','LineWidth',1); hold on; %Vertical line as magnitude index
```

```
plot(t, y1,'r','LineWidth',2);
title('Unit step');axis([-3 3 -0.1 1.5]); axis square
subplot(1,3,2);
%%%% uncomment the following two lines and observe the results
plot(t,zeros(size(t)),'k','LineWidth',1); hold on; %Horizontal line as time index
plot(zeros(size(t)),t,'k','LineWidth',1); hold on; %Vertical line as magnitude index
plot(t, y2,'m','LineWidth',2);
title('Delayed step u(t-2)');axis([-3 3 -0.1 1.5]); axis square
subplot(1,3,3);
%%%% uncomment the following two lines and observe the results
plot(t,zeros(size(t)),'k','LineWidth',1); hold on; %Horizontal line as time index
plot(zeros(size(t)),t,'k','LineWidth',1); hold on; %Vertical line as magnitude index
plot(t, y3,'b','LineWidth',2);
title('Advanced step u(t+2)');axis([-3 3 -0.1 1.5]); axis square
```

(b) Consider a triangular pulse signal $x(t)$ having unit height and unit width centered at origin. Use Matlab built in function **`tripuls.m`**. Sketch and label the following signals using Matlab

(i) $x(2t)$ (ii) $x(t/4)$

```
clear all; close all; clc;
%======== Time scaling of Triangular pulse ==========================
t = -3:0.001:3;
y1 = tripuls(t,1);
y2 = tripuls(2*t,1);
y3 = tripuls(t/3,1);
figure();
```

```
plot(t, y1,'b','LineWidth',3); hold on;
plot(t, y2,'--r','LineWidth',3); grid on;
plot(t, y3,':m','LineWidth',3); grid on;
legend('Triangular Pulse', 'Compression','Expansion');
set(gca,'fontsize',14);
xlabel('time  ----->');ylabel('Amplitude ----->');
title('delta(t)');axis([-3 3 -0.1 1.2]);
```

Ex 3.3: Folding operation or Time Reversal operation

Sketch and label the time reversal operation of a ramp signal using Matlab

```
clear all; close all; clc;
t = -3:0.001:3;
x1 = ur(t,0).*us(t,0);        %  r(t)
% x2 = ur(-t,0).*us(-t,0);       % r(-t)
x2 = fliplr(x1);     % Using fliplr.m function

figure();
plot(t, x1.*us(-t,-1),'b','LineWidth',2);hold on;
plot(t, x2.*us(t,-1),'--m','LineWidth',2);
title('Time Reversal');axis([-2 2 -0.1 1.2]);
legend('r(t)', 'r(-t)');
set(gca,'fontsize',14);
xlabel('time  ----->');ylabel('Amplitude ----->');
```

Pre lab work:

(a) Study and understand the basic theory required for this lab.

(b) Run the Matlab scripts given for examples and study the results.

(c) Perform the analytical solutions for the given laboratory exercises.

(d) Prepare the Matlab codes for the given laboratory exercises well before the commencement of lab scheduled time.

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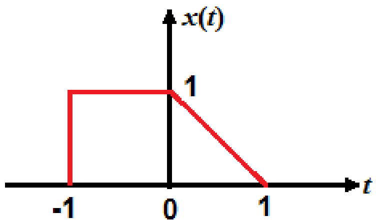
Laboratory Exercise-3

Q3.1. Consider a signal shown in figure:

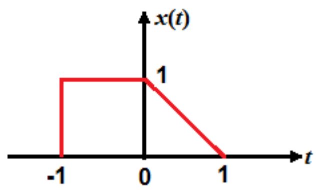
Determine analytically the following signals.

$(a)\ x(t-1)\quad (b)\ x(t+1)\quad (c)\ x(2t)\quad (d)\ x(-t)$

Then develop Matlab scripts and plot them.



Consider the signal shown in below:



(a). Express the signal in both (i) functional form and (ii) in-terms of unit step functions. Determine analytically the even and odd components of the signal

(b) Justify that $x(t) = x_e(t) + x_o(t)$

(c) Develop Matlab scripts for the above parts (a) and (b).

Ongoing Lab:

(a) Type the Matlab codes prepared for the exercises in the computer and observe the results.
(b) Compute the all the tasks given in exercises.
(c) Note the results and store the necessary figures and data for the lab report.

Post Lab:

(a) Complete the lab work in all aspects in the given specified lab time.
(b) Answer the given questions.
(c) Submit the lab report to the lab in-structure and get the signature in time.
(d) Type the complete description of commands used in the lab.

Viva Questions:

1. What do you mean by time shifting in continuous-time signals?
2. How does time scaling affect the shape of a signal?
3. Explain the significance of time reversal in signal processing.
4. What is signal folding, and how is it performed?
5. Differentiate between even and odd components of a signal.
6. Describe the procedure for compressing and expanding a continuous-time signal.
7. What happens when a signal is multiplied by a scalar?
8. Explain the effect of shifting a signal to the left versus to the right.
9. How would the graph of a signal x(t) change when you apply x(2t)?
10. In which real-world applications are time scaling operations used?

Due date:

Date of submission:

Student Id. No: Section:

Name of the student. Signature of student

Pre-lab Session work (15M) **Lab experiment**	**In-Lab Session work (15M)** **Project**	**Post Lab session work (10M)**	**Viva** **(10M)**	**Total Marks** **50M**

Remarks:

Marks awarde

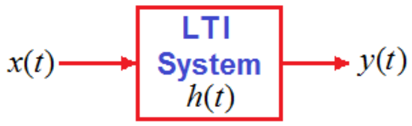
Date: **Signature of the Instructor**

Signals & Communication Systems - Lab-4

Linear Convolution

Introduction: Linear and time invariant LTI systems are particularly important class of systems has significant signal processing applications. These systems satisfy both the linearity and time-invariant properties. The processing of signals / sequences in time domain through discrete time systems are usually referred to as convolution or filtering operation. A mathematical operation that closely resembles convolution is called correlation. The correlation operation is used to measure the similarity between two sequences.

Convolution: The block schematic of CT system is illustrated in the following figure.



Let $x(t)$ is an input sequence applied to an CT, LTI system characterized by an impulse response $h(t)$. Then the output of the system is described mathematically by,

$$y(t) = \int_{-\infty}^{\infty} x(\tau)\, h(t-\tau)\, d\tau = \int_{-\infty}^{\infty} h(\tau)\, x(t-\tau)\, d\tau$$
$$y(t) = x(t) * h(t) = h(t) * x(t)$$

is known as convolution integral equation.

Examples:

Ex 4.1: Let $x(t)$ be the input to an LTI system with unit impulse response $h(t)$, where $x(t) = e^{-at}u(t)$ and $h(t) = u(t)$. Find the response of the system using convolution operation.

Solution: Given that $x(t)$ and $h(t)$. Then the response of the system is computed using convolution integral equation as below.

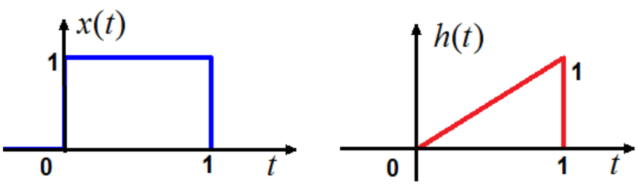
$$y(t) = \int_{-\infty}^{\infty} x(\tau)\, h(t-\tau)\, d\tau = \int_{-\infty}^{\infty} e^{-a\tau} u(\tau)\, u(t-\tau)\, d\tau$$
$$= \int_{0}^{t} e^{-a\tau}\, d\tau = -\frac{1}{a}\left[e^{-a\tau}\right]_0^t$$
$$= \frac{1}{a}(1 - e^{-at})\, u(t)$$

MAT Lab code: clear all; close all; clc;

t = -1:0.001:5;

```
x = exp(-2*t).*us(t,0);
h = us(t,0);h = double(h);
y = (0.002)*conv(x,h);
% y = us(t,0)-exp(-2*t).*us(t,0);
t2 = -2:0.001:10;
figure();
subplot(3,1,1);plot(t, x,'b','LineWidth',3);
xlabel('time  ----->');ylabel('Amplitude ----->');
title('x(t)'); axis([-1 5 -0.2 1.2]);
subplot(3,1,2);plot(t, h,'k','LineWidth',3);
xlabel('time  ----->');ylabel('Amplitude ----->');
title('h(t)');axis([-1 5 -0.2 1.2]);
subplot(3,1,3);plot(t2, y,'r','LineWidth',3);
xlabel('time  ----->');ylabel('Amplitude ----->');
title('y(t)=x(t)*h(t)');axis([-1 5 -0.2 1.2]);
```

Ex4.2: Determine the convolution analytically for the signals shown below. Then develop Matlab code for convolution and plot them.



Solution: The mathematical expressions for the given signals are

$$x(t)=\begin{cases}1, & 0\le t\le 1\\ 0, & Otherwise\end{cases} \text{ and } h(t)=\begin{cases}t, & 0\le t\le 1\\ 0, & Otherwise\end{cases}$$

```
clear all; close all; clc;
t = -1:0.001:5;
% Define signal x(t)
x = zeros(size(t));
i = find((t>0) & (t<1));
x(i) = 1;
% Define signal h(t)
h = zeros(size(t));
i = find((t>0) & (t<1));
h(i) = t(i);
y = (0.002)*conv(x,h);
t2 = -2:0.001:10;   %  Time index for y(t)
figure();set(gca,'fontsize',14);
subplot(3,1,1);plot(t, x,'b','LineWidth',3);
xlabel('time  ----->');ylabel('Amplitude ----->');
title('x(t)'); axis([-1 5 -0.2 1.2]);
subplot(3,1,2);plot(t, h,'k','LineWidth',3);
xlabel('time  ----->');ylabel('Amplitude ----->');
title('h(t)');axis([-1 5 -0.2 1.2]);
subplot(3,1,3);plot(t2, y,'r','LineWidth',3);
xlabel('time  ----->');ylabel('Amplitude ----->');
title('y(t)=x(t)*h(t)');axis([-1 5 -0.2 1.2]);
```

Pre lab work:

(a) Study and understand the basic theory required for this lab.

(b) Run the Matlab scripts given for examples and study the results.

(c) Perform the analytical solutions for the given laboratory exercises.

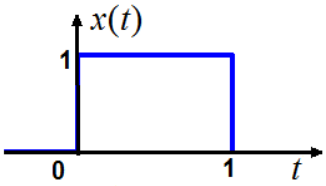
(d) Prepare the Matlab codes for the given laboratory exercises well before the commencement of lab scheduled time.

Lab Exercise-4

Exercise 4.1: Determine analytically the convolution of the following signals, and verify your answers using the Matlab.

(a) $x(t) = e^{-t}u(t)$ and $h(t) = u(t) - u(t-1)$

Exercise 4.2: Consider a continuous time signal shown below. Convolve the signal itself i.e., $y(t) = x(t) * x(t)$. Determine analytically the convolution and verify your answers using the Matlab.



Ongoing Lab:

(a) Type the Matlab codes prepared for the exercises in the computer and observe the results.

(b) Compute the all the tasks given in exercises.

(c) Note the results and store the necessary figures and data for the lab report.

Post Lab:

(a) Complete the lab work in all aspects in the given specified lab time.

(b) Answer the given questions.

(c) Submit the lab report to the lab in-structure and get the signature in time.

(d) Type the complete description of commands used in the lab.

Viva Questions:

1. What is linear convolution in the context of continuous-time signals?

2. How is the convolution integral defined mathematically for continuous-time signals?

3. What is the significance of the convolution operation in system analysis?

4. Describe the steps involved in performing graphical convolution between two signals.

5. What conditions must the signals satisfy for their convolution to be well-defined?

6. How does the duration of the output signal relate to the durations of the input signals?

7. What is the physical interpretation of convolution in terms of input-output behaviour?

8. How is convolution used in analysing Linear Time-Invariant (LTI) systems?

9. Explain the concept of the impulse response and its role in convolution.

10. How does the commutative property apply to convolution in continuous time?

Signals & Communication Systems - Lab-5
Fourier Transform of Continuous Time Aperiodic Signals

Introduction: The frequency representation of signals as well as the frequency response of systems are tools of great significance in signal processing, communi-cations, and control theory. In this lab we will explore the Fourier representation of signals by extending it to aperiodic signals. By a limiting process the harmonic represe-ntation of periodic signals is extended to the Fourier transform, a frequency-dense representation for non-periodic signals. The concept of spectrum introduced for periodic signals is generalized for both finite-power and finite-energy signals. Thus, the Fourier transform measures the frequency content of a signal, and unifies the representation of periodic and aperiodic signals.

Basic theory: An aperiodic, or nonperiodic, signal $x(t)$ can be thought of as a periodic signal $x_p(t)$ with an infinite period. Using the Fourier series representation of this signal and a limiting process we obtain a pair

$$x(t) \overset{\text{F.T}}{\longleftrightarrow} X(j\Omega)$$

where the signal $x(t)$ is transformed into a function $X(j\Omega)$ in the frequency domain by

the Fourier transform: $X(j\Omega) = \int_{-\infty}^{\infty} x(t)\, e^{-j\Omega t} dt$: Analysis equation

while $X(j\Omega)$ is transformed into a signal $x(t)$ in the time domain by the

Inverse Fourier transform: $x(t) = \dfrac{1}{2\pi}\int_{-\infty}^{\infty} X(j\Omega)\, e^{j\Omega t} d\Omega$: Synthesis equation

There are two important parameters associated with FT, known as magnitude spectrum and phase spectrum.

The magnitude spectrum is represented by $|X(j\Omega)|$, and similarly

The phase spectrum is represented by $\angle X(j\Omega)$.

Examples:

Ex 5.1: Find the FT of a signal $x(t) = e^{-at}u(t)$, where $a = 1$. Develop Matlab code to plot its magnitude and phase spectrum.

```
clear all; close all; clc;
% Time-domain signal
t = -1:0.01:6;
x = exp(-t) .* (t >= 0); % equivalent to e^(-t)u(t)
% Plot original signal
```

```
figure();
plot(t, x, 'LineWidth', 2);
title('Original Signal e^{-t} u(t)');
xlabel('t'); ylabel('x(t)');
axis([-1 6 0 1.2]);
grid on;
% ==========   Spectrum    ================
w = -pi:0.001:pi; % Frequency vector
X = 1 ./ (1 + 1i * w); % Fourier Transform of e^(-t)u(t)
Xm = abs(X);         % Magnitude spectrum
Xp = angle(X);       % Phase spectrum
% Plot magnitude and phase
figure();
subplot(2,1,1);
plot(w, Xm, 'r', 'LineWidth', 2);
title('Magnitude Spectrum');
xlabel('\omega'); ylabel('|X(\omega)|');
axis([-pi pi 0 1.2]); grid on;
subplot(2,1,2);
plot(w, Xp, 'b', 'LineWidth', 2);
title('Phase Spectrum');
xlabel('\omega'); ylabel('Phase of X(\omega)');
axis([-pi pi -pi/2 pi/2]); grid on;
```

Ex 5.2: Find the FT of a signal $x(t) = e^{-|t|}$. Develop Matlab code to plot its magnitude and phase spectrum.

```
clear all; close all; clc;
% Define time vector for numerical plot
t = -5:0.01:5;
x = exp(-abs(t)); % Original signal e^(-|t|)
% Plot the signal
figure();
plot(t, x, 'LineWidth', 2);
title('Original Signal e^{-abs(t)}');
xlabel('t'); ylabel('x(t)');
axis([-5 5 0 1.2]);
grid on;
% ==========   Spectrum (Known Fourier Transform) ================
w = -pi:0.001:pi; % Frequency vector
X = 2 ./ (w.^2 + 1); % Known Fourier transform of e^{-abs(t)}
Xm = abs(X);        % Magnitude spectrum
Xp = angle(X);      % Phase spectrum (should be zero for real and even signals)
% Plot magnitude and phase
figure();
subplot(2,1,1);
plot(w, Xm, 'r', 'LineWidth', 2);
title('Magnitude Spectrum');
xlabel('\omega'); ylabel('|X(\omega)|');
axis([-pi pi 0 2.2]); grid on;
subplot(2,1,2);
plot(w, Xp, 'b', 'LineWidth', 2);
title('Phase Spectrum');
xlabel('\omega'); ylabel('Phase of X(\omega)');
axis([-pi pi -pi/2 pi/2]); grid on;
```

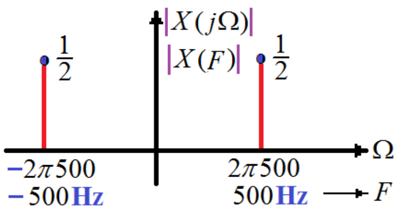
Ex 5.3: Find the FT of a unit step function $x(t) = u(t)$. Develop Matlab code to plot its magnitude and phase spectrum.

```
clear all; close all; clc;

% Numerical Heaviside function
t = -3:0.01:3;
x = (t >= 0);  % Equivalent to heaviside(t)

% Plot the step function
figure();
plot(t, x, 'LineWidth', 2);
title('Heaviside Function u(t)');
xlabel('t'); ylabel('u(t)');
axis([-3 3 -0.1 1.1]);
grid on;

% Fourier Transform not plotted due to singularities (πδ(w) + 1/(jω))
disp("Note: Fourier Transform of u(t) is πδ(ω) + 1/(jω), which includes a singularity and cannot be plotted directly.");
```

Ex 5.4: Fourier Transform (FT) of periodic sinusoidal signals $x(t) = \cos(2\pi 500t)$. Find the FT analytically and sketch its magnitude response and phase response. Develop Matlab code and verify the results.

Solution: Given that $x(t) = \cos(2\pi 500t)$, with frequency $\Omega_0 = 2\pi 500$ rad/sec, or ($\Omega_0 = 2\pi F_0 \Rightarrow F_0 = 500$ Hz or Cycles / sec). This signal is represented as

$$x(t) = \cos(2\pi 500t)$$
$$= \frac{1}{2}\left\{e^{j2\pi 500t} + e^{-j2\pi 500t}\right\}$$



The Fourier series coefficients for this signal are $c_1 = c_{-1} = \frac{1}{2}$ and remaining co-efficients are zero. The Fourier representation is shown in figure.

```
clear all; close all; clc;
fs=30000;%Sample frequency
N=5000;%Number of samples
```

```
Ts=1/fs;  % Sampling interval  ------------------------------------------
t = -(N/2)*Ts:Ts:((N/2)-1)*Ts;
% t = (-N/2:1:(N/2)-1)*Ts;
%%% 500 Hz Sinusoidal signal
fm = 500;         %Signal frequency  --------------------------------
m = cos(2*pi*fm*t);
%%% Spectrum of  Message signal  -----------------------------------------
f1 = (-N/2:1:N/2-1)*fs/N;
M = (2/N)*fftshift(fft(m));
Ma = abs(M);
Mp = angle(M);
% --------------------------------------------------------------------
%%%% 500 Hz Sinusoidal signal ----------
figure();
subplot(3,1,1);
plot(t,m/max(m), 'm', 'LineWidth',2);axis([-0.005 0.005 -1.2 1.2]);
xlabel('Time (seconds)');ylabel('Amplitude');title('500 Hz Sinusoidal signal');
grid on;
%%%%  Magnitude Spectrum of signal
subplot(3,1,2);
plot(f1,Ma/max(Ma),'r','Linewidth',2); axis([-800 800 -0.001 1.2]);
xlabel('frequency'); ylabel('Magnitude|'); title(' Magnitude Spectrum');
grid on;
%%%%  Phase Spectrum of signal
subplot(3,1,3);
plot(f1,Mp,'b','Linewidth',2); axis([-800 800 -4 4]);
xlabel('frequency'); ylabel('Phase anglee'); title('Phase Spectrum');
grid on;
```

Pre lab work:

(a) Study and understand the basic theory required for this lab.

(b) Run the Matlab scripts given for examples and study the results.

(c) Perform the analytical solutions for the given laboratory exercises.

(d) Prepare the Matlab codes for the given laboratory exercises well before the commencement of lab scheduled time.

Laboratory Exercise-5

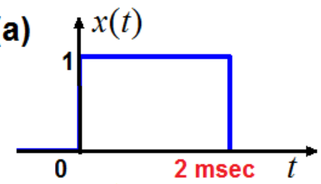
Exercise 5.1: Develop Matlab code to determine and plot the magnitude and phase spectrum for the following signals

(a) $x(t) = e^{|t|}\cos(10t)$

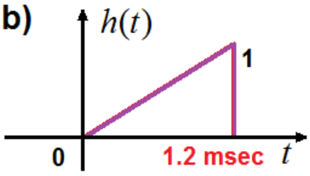
(b) $x(t) = 2\cos(2\pi 500t)\cos(2\pi 5000t)$

Exercise 5.2: Consider the following signals shown below. Write the mathematical expressions for these signals. Develop Matlab code to determine and plot the magnitude and phase spectrum of the signals.

(a)



(b)



Ongoing Lab:

(d) Type the Matlab codes prepared for the exercises in the computer and observe the results.

(e) Compute the all the tasks given in exercises.

(f) Note the results and store the necessary figures and data for the lab report.

Post Lab:

(e) Complete the lab work in all aspects in the given specified lab time.

(f) Answer the given questions.

(g) Submit the lab report to the lab in-structure and get the signature in time.

(h) Type the complete description of commands used in the lab.

Viva Questions:

1. What is the purpose of applying the Fourier Transform to aperiodic continuous-time signals?

2. How is the Fourier Transform of a continuous-time aperiodic signal defined mathematically?

3. What is the difference between Fourier Series and Fourier Transform?

4. What conditions must a signal satisfy to have a valid Fourier Transform?

5. Explain the physical significance of the magnitude and phase spectra of a signal.

6. How does time shifting affect the Fourier Transform of a signal?

7. What is the effect of time scaling on the Fourier spectrum?

8. Describe the convolution property of the Fourier Transform.

9. What is the duality property in Fourier Transform?

10. How do energy signals and power signals relate to the Fourier Transform?

Due date:

Date of submission:

Student Id. No: Section:

Name of the student. Signature of student

Pre-lab Session work (15M) Lab experiment	In-Lab Session work (15M) Project	Post Lab session work (10M)	Viva (10M)	Total Marks 50M

Remarks:

Marks awarde

Date: **Signature of the Instructor**

Signals & Communication Systems - Lab-6
Laplace Transform of Continuous Time Signals

Introduction: The Laplace transform is a generalization of the Continuous-Time Fourier Transform. It is used because the CTFT does not converge/exist for many important signals, and yet it does for the Laplace-transform (e.g., signals with infinite *l2*norm). The Laplace transform provide complementary representations of a signal to its own in the time domain, and an algebraic characterization of systems. The Laplace transform depends on a complex variable $s = \sigma + j\Omega$, composed of damping σ and frequency Ω, while the Fourier transform considers only frequency Ω. The LT is widely used to evaluate frequency response of systems that are all of great significance in signal processing, communications, and control theory. The location of the poles and the zeros of the transfer function relates to the dynamic characteristics of the system. Certain characteristics of CT systems can only be verified or understood via the Laplace transform. Such is the case of stability.

Basic theory: The analysis and synthesis equations of Laplace transform are illustrated as below.

$$x(t) \overset{\text{L.T}}{\longleftrightarrow} X(s)$$

Forward LT: $\text{L.T.}\{x(t)\},\ X(s) = \int_{-\infty}^{\infty} x(t)\, e^{-st} dt$: Analysis equation

Inverse LT: $\text{I. L.T. } \{X(s)),\ x(t) = \dfrac{1}{2\pi} \int_{\sigma - j\infty}^{\sigma + j\infty} X(s)\, e^{st} ds$: Synthesis equation

Examples:

Ex 6.1: Find the LT analytically for the following signals. Develop Matlab code and verify the results.

(a) $x(t) = 1 + \delta(t)$ (b) $x(t) = t\cos(2t)u(t)$

Solution: (a) $X(s) = 1 + \dfrac{1}{s}$ (b) $X(s) = \dfrac{2s^2}{(s^2+4)^2} - \dfrac{1}{(s^2+4)}$

```
clear all; close all; clc;

pkg load symbolic;  % Load symbolic package (required for laplace, dirac, etc.)
% ---------- Ex1(a) x(t) = u(t) + δ(t) ---------------------------
% Method 1: x(t) = 1 + δ(t)
syms t s;
```

```
x1 = dirac(t);      % δ(t)
x2 = 1;             % Constant 1
x = x1 + x2;
X = laplace(x, t, s);

disp('The Laplace transform of x(t) = 1 + δ(t) is X(s) = ');
disp(X);  % Expected: X(s) = 1/s + 1

% Method 2: x(t) = u(t) + δ(t)
x1 = dirac(t);          % δ(t)
x2 = heaviside(t);      % u(t)
x = x1 + x2;
X = laplace(x, t, s);

disp('The Laplace transform of x(t) = u(t) + δ(t) is X(s) = ');
disp(X);  % Expected: X(s) = 1/s + 1
```

```
clear all; close all; clc;
pkg load symbolic;

syms t s

x = t * cos(2*t);  % u(t) is implicit in Laplace
X = laplace(x, t, s);

disp('Laplace Transform of x(t) = t·cos(2t) is:');
disp(X);
```

Ex6.2: Given that $x_1(t) = e^{-t}u(t)$ and $x_2(t) = \cos(10t)$. Suppose these two signals are multiplied and the resultant signal is represented by $x(t) = x_1(t)x_2(t) = e^{-t}\cos(10t)u(t)$. Develop Matlab code to compute their LT. Further find poles and zeros and plot.

```
clear all; close all; clc;

pkg load symbolic;  % Load symbolic support

syms t s

% Define components
z1 = exp(-t);
z2 = heaviside(t);       % heaviside is built into symbolic pkg
x1 = z1 * z2;            % x1(t) = exp(-t)·u(t)
x2 = cos(10*t);          % x2(t) = cos(10t)
x  = x1 * x2;            % x(t) = exp(-t)·cos(10t)·u(t)

% Laplace Transforms
Z1 = laplace(z1, t, s); disp('Laplace of exp(-t):'); disp(Z1);
X1 = laplace(x1, t, s); disp('Laplace of exp(-t)·u(t):'); disp(X1);
X2 = laplace(x2, t, s); disp('Laplace of cos(10t):'); disp(X2);
X  = laplace(x, t, s);   disp('Laplace of exp(-t)·cos(10t)·u(t):'); disp(X);

% ------------------------ Plotting Signals ------------------------

t_vals = 0:0.01:5;
x1_vals = exp(-t_vals);
x_vals  = exp(-t_vals) .* cos(10 * t_vals);

figure(1);

% Plot x1(t) = exp(-t)·u(t)
subplot(2,2,1);
plot(t_vals, x1_vals, 'LineWidth', 2);
grid on; axis([0 5 0 1.1]);
```

```
title('x1(t) = exp(-t)·u(t)');
xlabel('t'); ylabel('x1(t)');

% Poles and Zeros of x1(t): Laplace = 1 / (s + 1)
nx1 = [1]; dx1 = [1 1];  % Transfer function numerator/denominator
[z1, p1, k1] = tf2zp(nx1, dx1);

% Plot pole-zero map manually
subplot(2,2,2);
plot(real(z1), imag(z1), 'ob', real(p1), imag(p1), 'xr');
grid on;
title('Pole-Zero Plot of x1(t)');
xlabel('Re'); ylabel('Im');
legend('Zeros', 'Poles');
axis equal;

% Plot x(t) = exp(-t)·cos(10t)·u(t)
subplot(2,2,3);
plot(t_vals, x_vals, 'LineWidth', 2);
grid on; axis([0 5 -1.1 1.1]);
title('x(t) = exp(-t)·cos(10t)·u(t)');
xlabel('t'); ylabel('x(t)');

% Poles and Zeros of x(t): Laplace = (s + 1) / [(s + 1)^2 + 100]
% Simplified: Numerator: s + 1; Denominator: s^2 + 2s + 101
nx = [1 1]; dx = [1 2 101];
[z2, p2, k2] = tf2zp(nx, dx);

% Plot pole-zero map for x(t)
subplot(2,2,4);
plot(real(z2), imag(z2), 'ob', real(p2), imag(p2), 'xr');
grid on;
title('Pole-Zero Plot of x(t)');
xlabel('Re'); ylabel('Im');
legend('Zeros', 'Poles');
axis equal;
```

Ex6.3: Find the inverse LT of the following signal

$$X(s) = \frac{2s+3}{(s^2+2s+4)}$$

```
clear all; close all; clc;

pkg load control;   % For tf, tf2zp, residue
pkg load symbolic;  % For symbolic math and ilaplace

% ---------------- Inverse Laplace Transform ----------------

syms t s
X = (2*s + 3)/(s^2 + 2*s + 4);
x = ilaplace(X, s, t);  % Inverse Laplace

% Numerical time vector for plotting
t_vals = linspace(0, 12, 1000);
x_vals = double(subs(x, t, t_vals));

figure();
plot(t_vals, x_vals, 'LineWidth', 2);
title('x(t) from Inverse Laplace');
xlabel('t'); ylabel('x(t)');
axis([0 12 -1 4]); grid on;

% ---------------- Transfer Function & Pole-Zero Plot ----------------

num = [0 2 3];     % Numerator coefficients (same as 2*s + 3)
```

```
den = [1 2 4];     % Denominator coefficients (s^2 + 2s + 4)

H = tf(num, den);  % Define transfer function

figure();
pzmap(H);          % Plot poles and zeros
title('Pole-Zero Map');

% ---------------- Roots (Poles and Zeros) ----------------

z = roots(num);
disp('Zeros:');
disp(z);

p = roots(den);
disp('Poles:');
disp(p);

% ---------------- Partial Fraction Expansion ----------------

[r, p, k] = residue(num, den);
disp('Poles from residue():');
disp(p);
disp('Residues:');
disp(r);

% ---------------- Custom s-plane Plot ----------------

figure();
plot(real(z), imag(z), 'ob', 'MarkerSize', 8); hold on;
plot(real(p), imag(p), 'xr', 'MarkerSize', 10);
title('Custom s-Plane Plot');
xlabel('Real Axis'); ylabel('Imag Axis');
```

```
legend('Zeros', 'Poles');
grid on; axis equal;
```

Pre lab work:

(a) Study and understand the basic theory required for this lab.

(b) Run the Matlab scripts given for examples and study the results.

(c) Perform the analytical solutions for the given laboratory exercises.

(d) Prepare the Matlab codes for the given laboratory exercises well before the commencement of lab scheduled time.

Laboratory Exercise - 6

Exercise 6.1: Find the LT of the following signals. Verify using Matlab.

(a) $x(t) = e^{-3t}u(t-3)$ **(b)** $x(t) = e^{-(t-2)}\cos(5t)u(t)$

Exercise 6.2: (a) Plot the pole zero diagram of the following transfer function.

$$H(s) = \frac{s+2}{s^2+2s+2}$$

Find the inverse Laplace transform x(t). use MATLAB to verify your result.

Ongoing Lab:

(a) Type the Matlab codes prepared for the exercises in the computer and observe the results.
(b) Compute the all the tasks given in exercises.
(c) Note the results and store the necessary figures and data for the lab report.

Post Lab:

(a) Complete the lab work in all aspects in the given specified lab time.
(b) Answer the given questions.
(c) Submit the lab report to the lab in-structure and get the signature in time.
(d) Type the complete description of commands used in the lab.

Viva Questions:

1. What is the definition of the Laplace Transform for a continuous-time signal?

2. How does the Laplace Transform differ from the Fourier Transform?

3. What is the significance of the Region of Convergence (ROC) in the Laplace Transform?

4. How does the Laplace Transform help in analyzing continuous-time systems?

5. Explain the time-shifting property of the Laplace Transform.

6. How is convolution represented in the Laplace domain?

7. What is the relationship between Laplace Transform and differential equations?

8. How do poles and zeros of a Laplace Transform affect the system behavior?

Due date:

Date of submission:

Student Id. No: Section:

Name of the student. Signature of student

Pre-lab Session work (15M) Lab experiment	In-Lab Session work (15M) Project	Post Lab session work (10M)	Viva (10M)	Total Marks 50M

Remarks:

Marks awarde

Date: **Signature of the Instructor**

Signals & Communication Systems - Lab-7
Sampling of Signals

Objective: The objective of this Lab is to understand concepts and observe the effects of periodically sampling a continuous signal at different sampling rates, changing the sampling rate of a sampled signal, aliasing, and anti-aliasing filters.

Introduction: The signals we use in the real world, such as our voices, are called "analog" signals. To process these signals in digital computers or digital systems, we need to convert the signals to "digital" form. While an analog signal is continuous in both time and amplitude, a digital signal is discrete in both time and amplitude. To convert a signal from continuous time to discrete time, a process called ***sampling*** is used. The value of the signal is measured at certain intervals in time. Each measurement is referred to as a sample. The analog signal is also quantized in amplitude, but that process is ignored in this demonstration.

Sampling Theorem: The sampling theorem can be defined in two ways as below.

Time domain Statement: A band limited signal having no frequency components higher than f_m Hz may be completely recovered from the knowledge of its samples taken at the rate of at least 2 f_m samples per second.

Frequency domain Statement: A band limited signal having no frequency components higher than f_m Hz is completely described by its sample values at uniform intervals less than or equal to 1 / 2 f_m seconds apart.

Basic theory: A simple process of sampling and reconstruction is illustrated in Fig1.

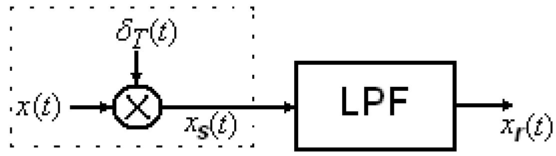
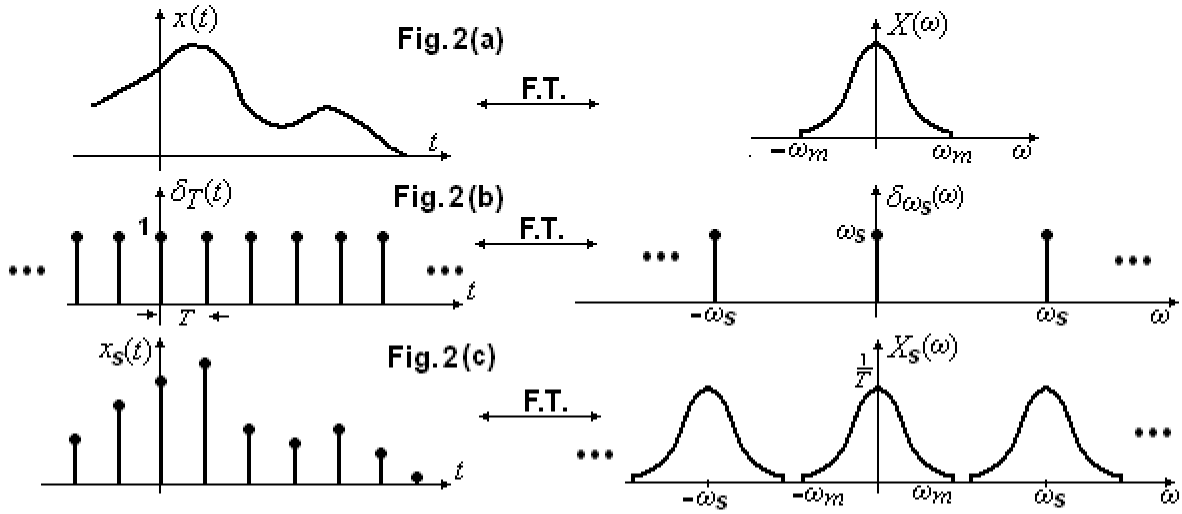


Fig.1. Sampling Process Reconstruction Process

The uniform sampling and reconstruction process is illustrated in Fig.1. Let us consider a band limited signal $x(t)$ having no frequency components beyond f_m Hz, i.e., $X(\omega)$ is zero for $|\omega| > \omega_m$, where $\omega_m = 2\pi f_m$. When this signal is multiplied by a periodic impulse function $\delta_T(t)$ (with period 'T'), the product yields a sequence of impulses located at uniform intervals of T seconds. The strength of resulting impulses is equal to the value of $x(t)$ at the corresponding instants.



The Fourier transforms of the band limited signal $x(t)$ and the impulse train $\delta_T(t)$ are $X(\omega)$ and $\delta_{\omega_s}(\omega)$ shown in Fig 2(a) and (b) respectively. The product of $x(t)$ and $\delta_T(t)$ yields a discrete time signal $x_s(t)$ as shown in Fig 2(c). The corresponding spectrum $X_s(\omega)$ can be determined by frequency convolution theorem as below.

We know that the Fourier Transform of periodic impulse train is also periodic function $\delta_{\omega_s}(\omega)$ and can be written as the sum of impulses located at $\omega = 0, \pm\omega_s, \pm\omega_s, ...$

$$\delta_T(t) = \sum_{n=-\infty}^{\infty} \delta(t - nT) \quad \xleftrightarrow{\text{F.T.}} \quad \omega_s \delta_{\omega_s}(\omega) = \omega_s \sum_{m=-\infty}^{\infty} \delta(\omega - m\omega_s)$$

Then the sampled signal is given by

$$\boxed{\begin{aligned} x_s(t) &= x(t)\delta_T(t) \\ &= \sum_{n=-\infty}^{\infty} x_n \delta(t - nT) \end{aligned}} \quad \xleftrightarrow{\text{F.T.}} \quad \boxed{\begin{aligned} X_s(\omega) &= \frac{1}{2\pi}\left[X(\omega) * \omega_s \delta_{\omega_s}(\omega)\right] \\ &= \frac{\omega_s}{2\pi}\left[X(\omega) * \delta_{\omega_s}(\omega)\right] = \frac{1}{T}\left[X(\omega) * \delta_{\omega_s}(\omega)\right] \end{aligned}}$$

Thus the spectrum $X_s(\omega)$ is obtained by convolving $X(\omega)$ and $\delta_{\omega_s}(\omega)$ represents $X(\omega)$ repeating every ω_s rad/sec. It is obvious from $X_s(\omega)$ shown in Fig 2(c), that $X(\omega)$ will repeat periodically without overlapping provided $\omega_s \geq 2\omega_m$ or $\frac{2\pi}{T} \geq 2(2\pi f_m)$. That is the sampling rate $f_s = \frac{1}{T}$ is given as $f_s \geq 2f_m$.

Hence the sampling rate should at least be equal to the twice of max frequency com-ponent present in the signal $x(t)$. This means that at least two samples per second are needed for a complete recovery of the signal from $x_s(t)$. Therefore the minimum sam-pling rate $f_s = 2f_m$. This minimum rate of sampling is known as Nyquist sampling rate.

Signal Recovery from its samples: It can be shown that the original signal $x(t)$ can be recovered by passing its sampled version through a LPF with a cut off frequency $f_c = f_m$ or $\omega_c = \omega_m$).

Let the function $x(t)$ is sampled at Nyquist rate ω_s , then we have $\omega_s = 2\omega_m$.

Therefore $X_s(\omega) = \dfrac{1}{T} \sum\limits_{m=-\infty}^{\infty} X(\omega - m\omega_s) = \dfrac{1}{T} \sum\limits_{m=-\infty}^{\infty} X(\omega - 2m\omega_m)$.

The LPF transfer function is given by $H(\omega) = \begin{cases} T, |\omega| < \omega_c \\ 0,\ |\omega| > \omega_c \end{cases}$

Then $X_r(\omega) = X_s(\omega)H(\omega)$ or

$$
\begin{aligned}
x_r(t) &= x_s(t) * \text{I.F.T}\{H(\omega)\} \\
&= x_s(t) * T\frac{\omega_s}{2\pi} Sa\{\omega_m t\} = x_s(t) * Sa\{\omega_m t\} \\
&= \sum_{n=-\infty}^{\infty} x_n \delta(t - nT) * Sa\{\omega_m t\} = \sum_{n=-\infty}^{\infty} Sa\{\omega_m (t - nT)\} = \sum_{n=-\infty}^{\infty} Sa(\omega_m t - n\pi)
\end{aligned}
$$

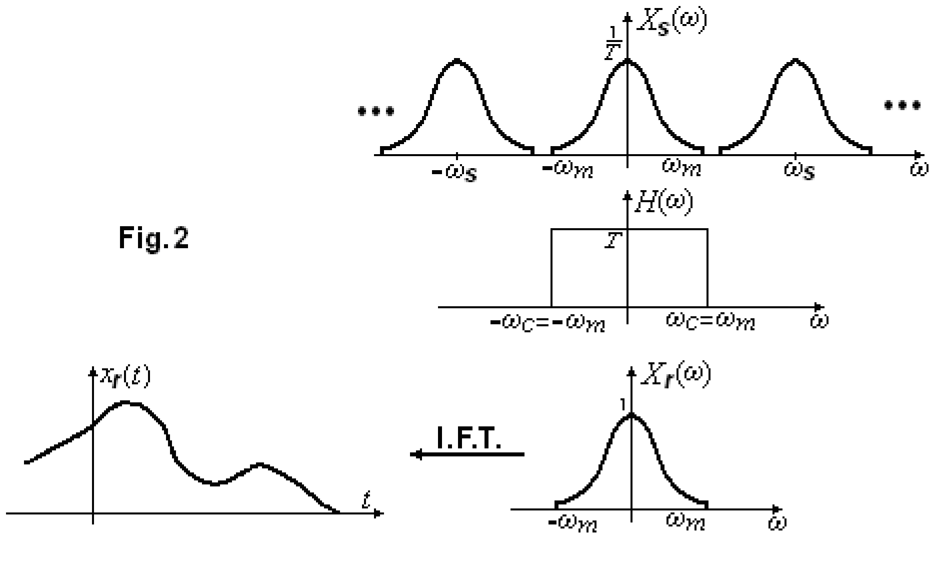


Fig.2

This equation represents the function $x_r(t)$ (or equivalent to $x(t)$) can be constructed by multiplying its samples x_n with a sampling function $Sa(\omega_m t - n\pi)$ and adding the multiplied values. This procedure is referred to as ideal band limited interpolation using the sync function. The construction of $x_r(t)$ (or equivalent to $x(t)$) is shown in Fig 2. and Fig 3.

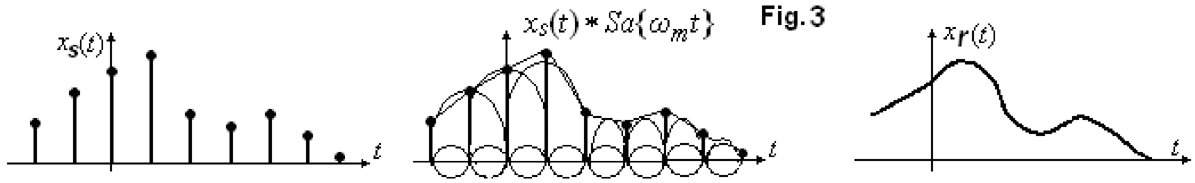
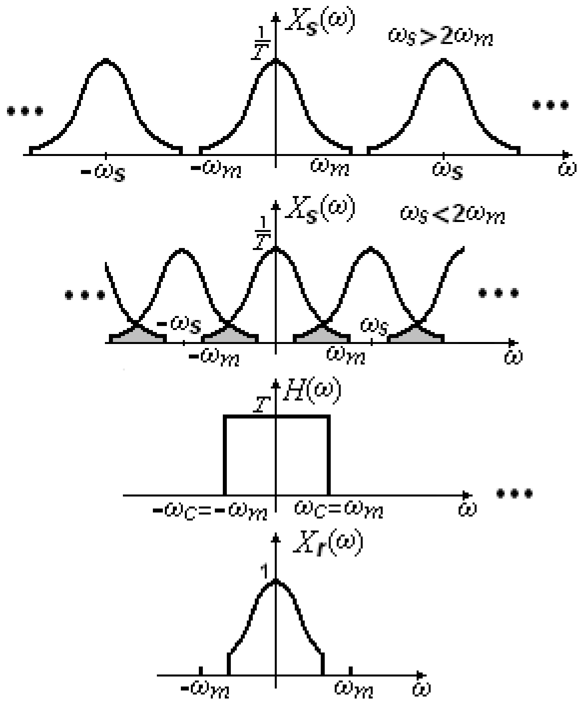


Fig.3

The effect of Under Sampling: Aliasing: As explained in the previous section, with $\omega_s \geq 2\omega_m$ (or $f_s \geq 2f_m$), the spectrum of sampled signal consists of scaled replications of the spectrum of $x(t)$ and thus forms the basis for the sampling theorem.



When $\omega_s < 2\omega_m$, $X(\omega)$ the spectrum of $x(t)$ is no longer replicated in $X_s(\omega)$ and thus no longer recoverable by low pass filtering. This effect in which the individual terms overlap is referred to as aliasing. It is also called ***frequency folding effect***. This is illustrated in the figure. The effect of under sampling, whereby higher frequencies are reflected into lower frequencies. Hence to avoid aliasing, it should be ensured that (i) $x(t)$ is strictly band limited and (ii) ω_s must be greater than $2\omega_m$.

Examples:

Ex 7.1: Time domain sampling demonstration

```
clear all; close all; clc;

f0=1000;%Freqyency of sin
fs1=10000;%Sampling Frequency Fs>2Fm
fs2=1500;%Sampling Frequency Fs<2Fm

n=0:1:20;
x=cos(2*pi*f0*n/fs1);
```

```
x1=cos(2*pi*f0*n/fs2);


figure ()

plot(n,x, 'm','LineWidth',2); title('Original Signal');


figure();

plot(n,x, 'b','LineWidth',2);hold on

stem(n,x,'--r','fill','LineWidth',2);

plot(n,x1,'k','LineWidth',2);

title('Sampling');

legend('Original Signal','Sampling with Fs>2Fm','Sampling With Fs<2Fm');
```

```
clear all; close all; clc;



clear all; clf

% sinusoids

omega_0 = 1;omega_s = 7;

T = 2* pi/omega_0;

t = 0:0.001:2*T; % a period of x1


x1 = cos(omega_0* t);

x2 = cos((omega_0 + omega_s)*t);


N = length(t);

Ts = 2*pi/omega_s; % sampling period


M = fix(Ts/0.001);

imp = zeros(1,N);


for k = 1:M:N-1.

   imp(k) = 1; % sequence of impulses

end

xs = imp.*x1; % sampled signal


plot(t,x1,'b',t,x2,'k'); hold on

stem(t,imp.*x1,'r','filled');

axis([0 max(t) -1.1 1.1]); xlabel('t'); grid
```

Pre lab work:

(a) Study and understand the basic theory required for this lab.

(b) Run the Matlab scripts given for examples and study the results.

(c) Perform the analytical solutions for the given laboratory exercises.

(d) Prepare the Matlab codes for the given laboratory exercises well before the commencement of lab scheduled time. Discuss with your lab instructors.

Laboratory Exercise-7

Exercise 7.1: Consider a multitone signal

$$s(t) = 1.2\sin(380\pi t + \tfrac{\pi}{3}) + 2\cos(840\pi t - \tfrac{\pi}{5}).$$

Suppose this signal is sampled with the following sampling frequencies.

(a) 600 sam/sec (b) 1200 sam /sec

Is there any aliasing effect? If these sampled signals are reconstructed, what are the frequency components are available? Develop Matlab codes, sketch and label them. Draw the corresponding spectra.

Ongoing Lab:

(a) Type the Matlab codes prepared for the exercises in the computer and observe the results.

(b) Compute the all the tasks given in exercises.

(c) Note the results and store the necessary figures and data for the lab report.

Post Lab:

(a) Complete the lab work in all aspects in the given specified lab time.

(b) Answer the given questions.

(c) Submit the lab report to the lab in-structure and get the signature in time.

(d) Type the complete description of commands used in the lab.

Viva Questions:

1. What is sampling in the context of continuous-time signals?

2. State and explain the Nyquist-Shannon sampling theorem.

3. What is aliasing, and how does it occur during sampling?

4. How can aliasing be prevented in practical sampling systems?

5. What is the significance of the sampling frequency in signal reconstruction?

6. Explain the difference between ideal and practical sampling.

7. What is meant by undersampling, and what are its consequences?

8. How is a continuous-time signal reconstructed from its samples?

9. What is the role of an anti-aliasing filter in a sampling system?

10. Describe the effect of over-sampling on the quality and size of sampled data.

Due date:

Date of submission:

Student Id. No: Section:

Name of the student. Signature of student

Pre-lab Session work (15M) Lab experiment	In-Lab Session work (15M) Project	Post Lab session work (10M)	Viva (10M)	Total Marks 50M

Remarks:

Marks awarde

Date: **Signature of the Instructor**

Signals & Communication Systems - Lab -8
Amplitude Modulation & Demodulation

Objectives:

1. To observe the operation of a linear integrated circuit function generator.
2. To construct and study the Amplitude Modulation using XR2206.
3. To examine the time displays of an AM signal.
4. To measure the percentage modulation, and the percentage of total power in both sidebands and in the carrier versus the modulation index.
5. To investigate the use (& limitation) of envelope detection in demodulating AM signals.

Pre-Lab Work:

1. Study the data sheet of XR2206 and understand the functions of each pin.
2. Study the basic theory of Amplitude modulation and envelope detection techniques. Time and Frequency analysis of AM waves.
3. Understanding the circuit diagrams of AM generation and envelope detection.
4. Understanding the data sheets of components used in the experiment.
5. Computer simulations (Multisim / pSpice) are performed and the objectives are obtained prior to the hardware experiment.

Output Amplitude versus Input Control Voltage Characteristics of XR2206

The amplitude versus input voltage characteristics of the XR-2206 linear integrated circuit function generator is examined here. The output amplitude of the XR-2206 function generator can be varied by applying a control voltage to pin1. The output amplitude varies linearly with the control voltage for values within ±4 Volts of $V_+/2$. The schematic diagram for the function generator circuit used in this section is shown in Fig 1.

Brief Theory:

AM Modulation: Amplitude Modulation is defined as a system of modulation in which the amplitude of the carrier wave $c(t) = A_C \cos\omega_C t$ is varied linearly with the instantaneous amplitude of the message signal $m(t)$. The terms A_C is the amplitude and f_C is the frequency of the carrier wave respectively. The standard form of amplitude modulated (AM) wave is defined by

To overcome this difficulty, the frequency spectrum of an aperiodic sequence is discretized. Such a frequency-domain represented leads to the discrete Fourier transform (DFT), which is a powerful computational tool for performing frequency analysis of discrete-time sequences.

$$s(t) = A_c[1 + k_a m(t)] \cos 2\pi f_c t \qquad (1)$$

where K_a = 1/ Ac is a constant is called amplitude sensitivity of the modulator. The term $A_c[1+ K_a m(t)]$ is referred to as envelope of the AM wave.

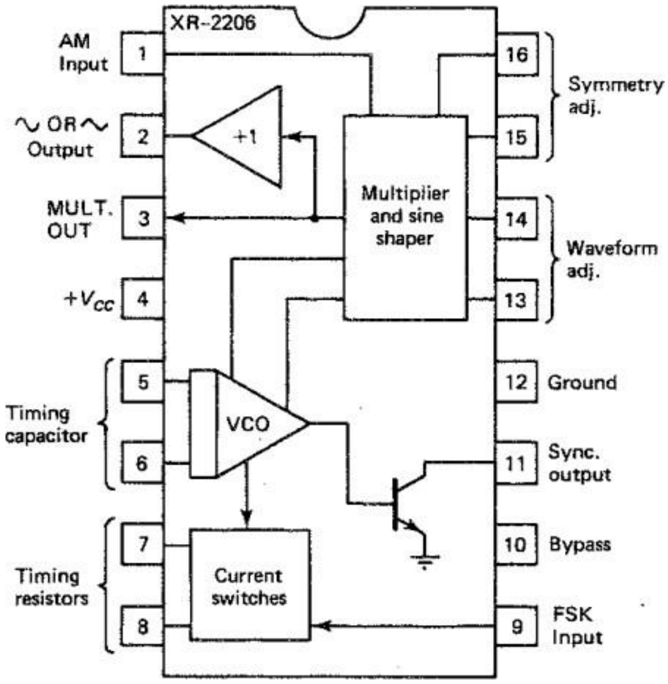


Fig.1: XR-2206 functional block diagram

Consider a modulating wave *m*(*t*) that consists of a single tone or frequency component. That is $m(t) = A_m \cos 2\pi f_m t$, where A_m is the amplitude and f_m is the frequency of the modulating wave respectively. Then the AM wave is described by

$$s(t) = A_c[1 + \mu \cos 2\pi f_m t] \cos 2\pi f_c t \qquad (2)$$

where μ = ka Am = Am / Ac is called modulation factor or percentage modulation. To avoid envelop distortion due to over modulation, μ must be kept below unity. Let Amax and Amin denote the maximum and minimum values of the envelop of the modulated wave,

$$Then\ \frac{A_{max}}{A_{min}} = \frac{A_c(1+\mu)}{A_c(1-\mu)} \Rightarrow \mu = \frac{A_{max} - A_{min}}{A_{max} + A_{min}} \qquad (3)$$

With this notation we can write the AM wave s(t) as

$$s(t) = A_c \cos 2\pi f_c t + \mu A_c \cos 2\pi f_m t \cos 2\pi f_c t$$

$$= A_c \cos 2\pi f_c t + \frac{\mu A_c}{2}\cos(\omega_c - \omega_m)\, t + \frac{\mu A_c}{2}\cos(\omega_c + \omega_m)\, t \qquad (4)$$

The total power in the AM modulated wave is given by $P_t = P_c + P_{LSB} + P_{USB}$, where

$$P_t = P_c\left(1 + \frac{\mu^2}{2}\right), P_c = \frac{{A_c}^2}{2}, \qquad P_{LSB} = P_{USB} = \frac{\mu^2 {A_c}^2}{8}$$

are the total power, carrier power, lower and upper sideband powers respectively. The transmission bandwidth of the AM wave is exactly equal to the twice of the message bandwidth. Transmission bandwidth = USB - LSB = (f+W)-(f-W)= 2W Hz, where 'W'' is message signal bandwidth. In the case if single tone signal m(t) = Am cos $2\pi f_m t$ the bandwidth is 2 f_m Hz. Fig.2 shows the circuit diagram of Amplitude Modulation using XR2206.

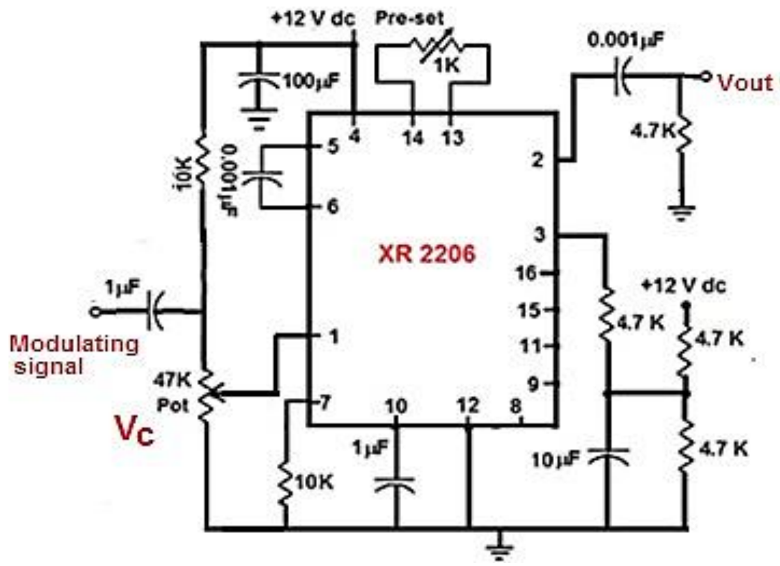


Fig 2. Circuit diagram of Amplitude Modulation

AM Demodulation: The process of extracting a base band signal from the modulated signal is known as de- modulation. AM signal with large carrier are detected by using the envelope detector. The envelope detector employs the circuit that extract the envelop of the AM wave. The envelope of the AM wave is the base band signal. However, a low level modulated signal can be detected by using square law detector in which a device operating in the non linear region is used to detect the base band signal. A diode operating in a linear region of its characteristics can extract the envelop detector. it is very simple and less expensive AM demodulation technique. Fig.3 illustrate the circuit diagram of envelope detector.

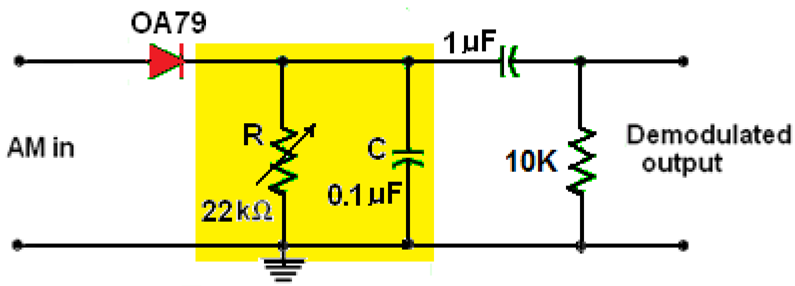


Fig 3. Circuit diagram of Envelope Detection.

Procedure:

1. Construct the circuit shown in Fig 2.
2. Adjust the amplitude of the control voltage VC to 0 V.
3. Reduce the amplitude of the audio signal generator output voltage to 0 V.
4. Calculate the function generator free running frequency.
5. Adjust preset (pin13 and 14) until a sine wave within minimum distortion is observed at Vout.
6. Increase the control voltage to + 5 V dc.
7. Increase the amplitude of signal generator output voltage to 3Vp-p at 1 kHz.
8. Adjust the amplitude of audio signal generator output voltage until an AM envelope with 100% modulation is observed at Vout.
9. Sketch the waveform observed in step 8.
10. Describe the waveform sketched in step 9 in terms of frequency content, amplitude, shape, and repetition rate.
11. Set the amplitude of audio signal generator output voltage to 1.5 Vp-p and determine the percentage modulation of output envelope using the following formula.

$$\mu = \frac{V_{max} - V_{min}}{V_{max} + V_{min}}$$

Where μ modulation index

Vmax = maximum peak-to-peak envelope amplitude (volts)

Vmin = minimum peak-to-peak envelope amplitude (volts)

12. Make the necessary connections between the AM modulator and the oscilloscope to display a trapezoidal pattern.
13. Vary the amplitude and frequency of the audio signal generator output voltage and describe with effect varying them has on the AM envelope.
14. For demodulation, connect the demodulation circuit as shown in Fig.3.
15. Observe the demodulated out for every modulating signal input as above. If necessary vary the 22K potentiometer for distortion less output.

Observations:

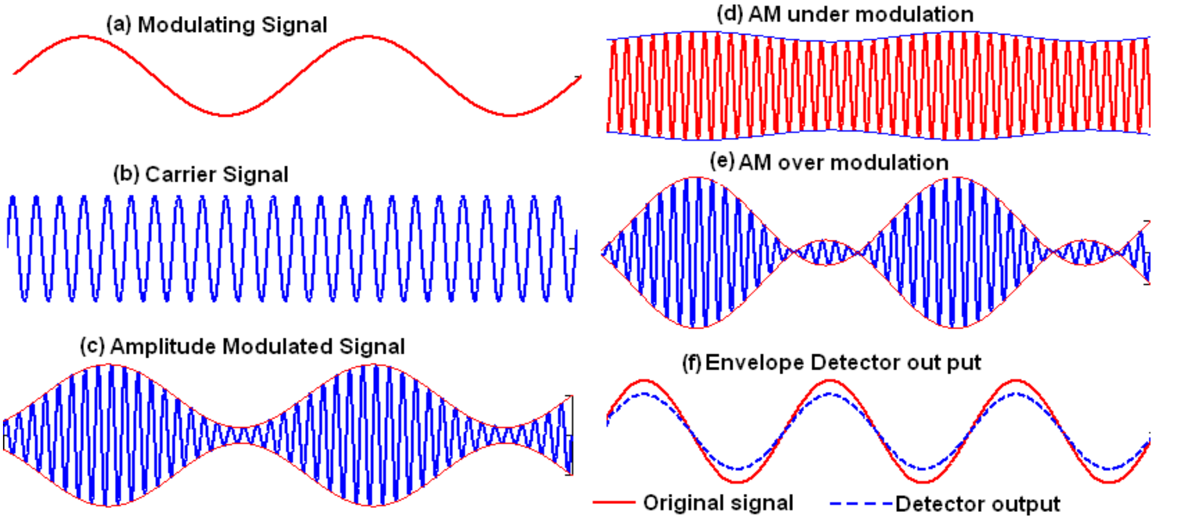


Fig 4. Amplitude Modulation waveforms with various modulation indices and Envelope detector output waveforms

Precautions:

1. Check the connections before switching ON the power supply
2. Observations should be done carefully.

Results:

Post-Lab Requirements

1. Create the illustration for Amplitude Modulation.
2. Compare the results are obtained in hardware lab with that of computer simulations.
3. Submit your illustration to the lab instructor at next week's lab.

Viva Questions:

1) Why modulation is necessary?

2) Define AM and draw its spectrum? What is its band width?

3) Why percentage modulation is always less than 100 % in case of A.M.?

4) Give the significance of modulation index?

5) What are the different degrees of modulation?

6) What will be the change in modulation index if there is change in amplitude of modulating signal ?

7) What will be the change in modulation index if there is change in frequency of modulating signal ?

8) Compare linear and nonlinear modulators?

9) Explain how AM wave is detected?

10) What are the different types of distortions that occur in an envelop detector? How can they be eliminated?

11) How many channels are contained in the AM broadcast band?

12) What is the bandwidth of each of the channels in the AM broadcast band?

13) Draw AM signal in which carrier signal is sinusoidal and modulating signal is triangular wave.

14) An audio signal of 7.5 KHz with a peak of 4.5 Volts modulates the carrier of 7.5 Volts peak with frequency 510 KHz. Find out the modulation index.

15) What is the bandwidth requirement for the AM signal when the frequencies of the modulating signals 200 Hz, 400 Hz and 800 Hz are transmitted simultaneously?

Due date:

Date of submission:

Student Id. No: Section:

Name of the student. Signature of student

Pre-lab Session work (15M) **Lab experiment**	**In-Lab Session work (15M)** **Project**	**Post Lab session work (10M)**	**Viva** **(10M)**	**Total Marks** **50M**

Remarks:

Marks awarde

Date: **Signature of the Instructor**

Signals & Communication Systems - Lab -9

DSB-SC Modulation & Demodulation

Objectives:

1. To construct and study the DSB-SC Modulation technique.
2. To examine the time displays of an DSB-SC signal.
3. To investigate the use (& limitation) of synchronous (envelope) detection for demodulating DSB-SC signals.

Pre-Lab Work:

1. Basic theory of DSB-SC modulation using balanced modulator and synchronous detection techniques.
2. Time and Frequency analysis of DSB-SC waves.
3. Understanding the circuit diagrams of DSB-SC generation using balanced modulator and envelope detection.
4. Understanding the data sheets of components used in the experiment.
5. Computer simulations (Multisim / pSpice) are performed and the objectives are obtained prior to the hardware experiment.

Equipment and Components:

Circuit Diagram:

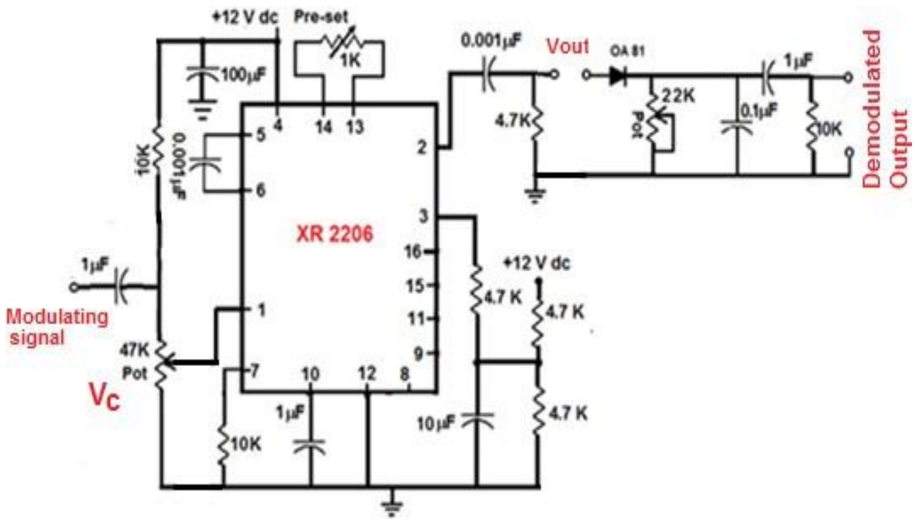


Fig 1. Circuit diagram of DSB-SC Modulation and Demodulation

Basic Theory: The block diagram of DSB-SC modulator is shown in Fig 2. A balanced modulator consists of two standard AM modulators arranged in a balanced configuration to suppress the carrier wave. We assume that the two modulators are identical except for the signal reversal of the modulating wave applied to the input of one of them. Thus the outputs of the two modulators are expressed as follows:

$$s_1(t) = A_c[1 + k_a m(t)] \cos 2\pi f_c t \quad and \; s_2(t) = A_c[1 - k_a m(t)] \cos 2\pi f_c t$$

By subtracting $s_2(t)$ from $s_1(t)$, We obtain

$$s(t) = s_1(t) - s_2(t) = [2k_a m(t) A_c \cos 2\pi f_c t] m(t).$$

Hence except for the scaling factor $2k_a$ the balanced modulator output is equal to the product of the modulating wave and carrier as required. An envelope detector is used for demodulation purpose.

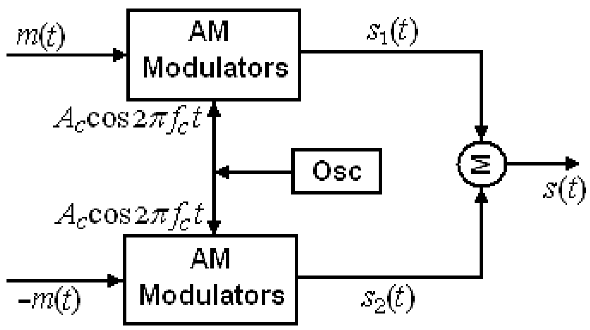


Fig. 2 Block diagram of Balanced Modulator

Procedure:

DSB-SC Modulation:

1. Construct the AM DSBSC modulator circuit shown in Figure 3. Set the signal generator output amplitude to 0V.
2. Set the dc control voltage Vc to 0 V and adjust R2 until a sine wave with minimum distortion is observed at Vout.
3. Increase the dc control voltage Vc until the output signal decrease to 0 V (approximately equal to +6V dc)
4. Set the amplitude of the audio signal generator output voltage to +2 Vp-p and adjust its frequency to 1 kHz.
5. Adjust the dc control voltage and signal generator output voltage until a symmetrical AM DSBSC waveform with maximum amplitude is observed at Vout.
6. Describe the waveform observed in step 5.
7. Vary the control voltage slightly above then slightly below +6 V Vdc and describe what effect it has on the output waveform.
8. Vary the frequency of the audio signal generator and describe what effect it has on the output waveform.

9. Vary the amplitude of the audio signal generator and describe what effect it has on the output waveform.

Demodulation:

1. Now assemble the demodulation circuit.
2. By varying the 22k potentiometer observe the demodulated signal.
3. By varying the modulating voltage in the DSB-SC modulation circuit, observe the demodulated signal.
4. Similarly, by varying the modulating signal frequency in the modulation circuit, observe the demodulated signal.

Observations:

Model Wave Forms:

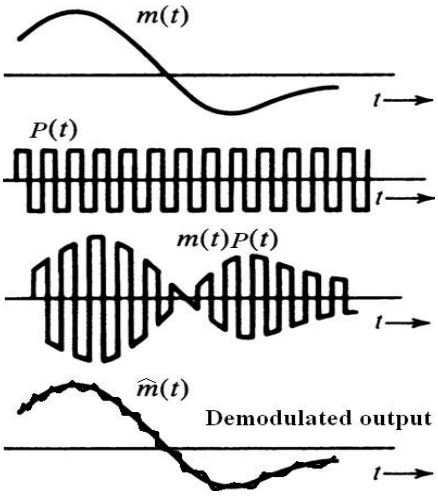


Fig 3. DSB-SC Modulation waveforms and Envelope detector output waveforms

Results:

Viva Questions

1. What are discrete frequencies in DSB-SC?

2. What is the advantage of DSB-SC over AM?

3. Mention the names of methods for DSB-SC generation?

4. What do you mean by coherence detection and non-coherent detection?

5. How a message signal recovered from DSBSC wave?

6. What is the disadvantage of DSB-SC?

7. What is the bandwidth of DSB-SC?

8. Why DSB-SC is not used for commercial broad casting?

9. Mention few applications for DSB-SC.

Due date:

Date of submission:

Student Id. No: Section:

Name of the student. Signature of student

Pre-lab Session work (15M) **Lab experiment**	**In-Lab Session work (15M)** **Project**	**Post Lab session work (10M)**	**Viva** **(10M)**	**Total Marks** **50M**

Remarks:

Marks awarde

Date: **Signature of the Instructor**

Signals & Communication Systems - Lab-10

Frequency Modulation and De-Modulation using PLL

Objectives:

1. To construct and study the Frequency Modulation technique.

2. To examine the time displays of an FM signal.

3. To examine the frequency deviations for various modulating signal voltages.

4. To measure the percentage of modulation, and total power in frequency modulation.

5. To investigate the use FM demodulation using PLL.

Pre-Lab Work:

1. Basic theory of Frequency Modulation techniques. Time and Frequency analysis of AM waves.

2. Basic theory of PLL FM demodulation.

3. Understanding the circuit diagrams of FM generation and detection.

4. Understanding the data sheets of components used in the experiment.

6. Computer simulations (Multisim / pSpice) are performed and the objectives are obtained prior to the hardware experiment.

Equipment and Components:

1. Signal generator
2. CRO
3. IC XR-2206, LM 565
4. Resistors
5. Capacitors.
6. Connecting wires &probes

Basic Theory:

Modulation is concerned with changing some characteristics of a high frequency carrier wave in accordance with the amplitude of the modulating signal to be transmitted. Frequency modulation is a system in which the frequency of the carrier is varied in accordance with the amplitude variations of the message signal; whereas the amplitude of the carrier remains unaltered. In FM the information is being carried by the carrier in its frequency variations and not in amplitude. This is a great advantage in FM because the noise generally affects the amplitudes of the waveform.

$$s_{FM}(t) = A_c \cos\left[\omega_c t + 2\pi k_f \int_{-\infty}^{t} m(\alpha)\, d\alpha\right]$$

where c(t) = $A_c \cos\omega_c t$ is RF carrier signal, m(t) is modulating signal and k_f is frequency modulation sensitivity constant. For a single-tone modulating signal, the FM wave is represented by

$$s_{FM}(t) = A_c \cos\left[\omega_c t + \beta_f \sin 2\pi f_m t\right]$$

Where $\beta_f = \frac{k_f A_m}{f_m} = \frac{\Delta f}{f_m}$ is called frequency modulation index, and where again $\Delta f = k_f A_m$ is known as frequency deviation. The bandwidth required to propagate an FM wave according to the Carson's rule is represented by $B.W_{FM} = 2(\Delta f + k_f)$ Hz. The circuit diagram of Frequency Modulation is shown in Fig.1.

Frequency Demodulation:

Frequency demodulation is the process that enables one to extract the original modulating from the frequency modulated wave. This can be achieved by a system which has a transfer characteristics just inverse of voltage controlled oscillator (VCO). In other words a frequency demodulator produces an output voltage whose instantaneous frequency of input FM signal. There are various kinds of FM demodulation techniques are available. In this experiment PLL based Frequency demodulation technique is used and the circuit is shown in Fig 2.

Circuit Diagram:

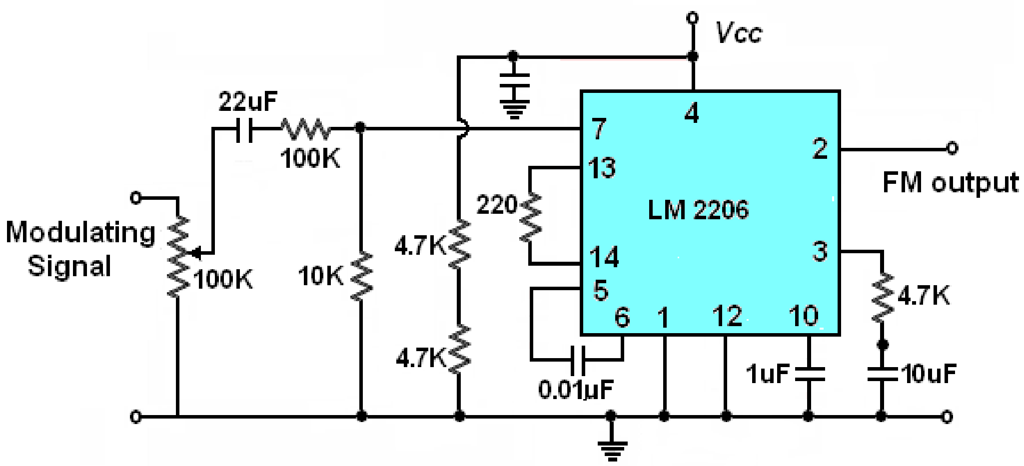


Fig.1 Circuit diagram of Frequency Modulation

Procedure:

Frequency Modulation.

1. Connect the Frequency Modulation circuit diagram shown in Fig.1.

2. Measure the frequency of the carrier signal at the FM output terminal with the modulating signal is zero and plot the same on graph.

3. Apply the modulating signal of 500HZ with 1Vp-p.

4. Observe the modulated wave on the C.R.O & plot the same on graph.

5. Find the modulation index by measuring minimum and maximum frequency deviations from the carrier frequency using CRO.

6. Determine the bandwidth of FM wave

7. Repeat the steps 5 and 6 by changing the amplitude and /or frequency of the modulating Signal.

FM Demodulation:

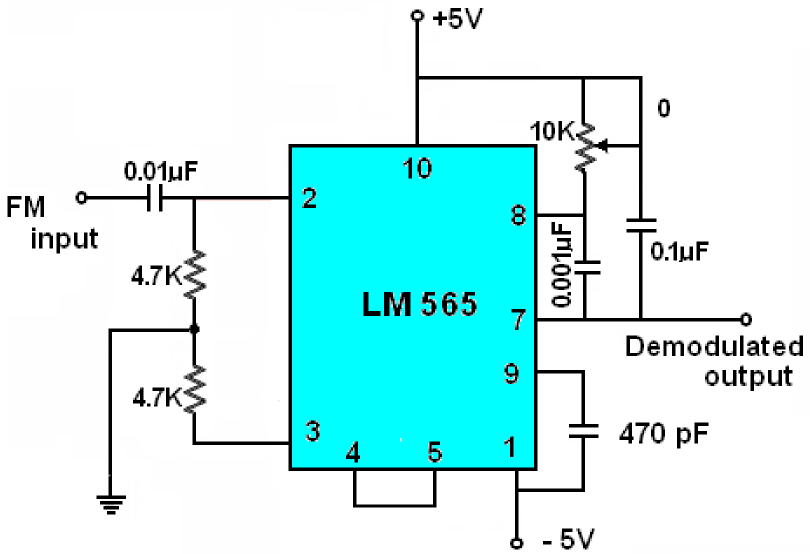


Fig.2 Circuit diagram of FM Demodulation

8. Now wire the circuit as per the FM demodulation circuit shown in Fig.2.

9. Initially lock the VCO of PLL to the carrier frequency of FM wave.

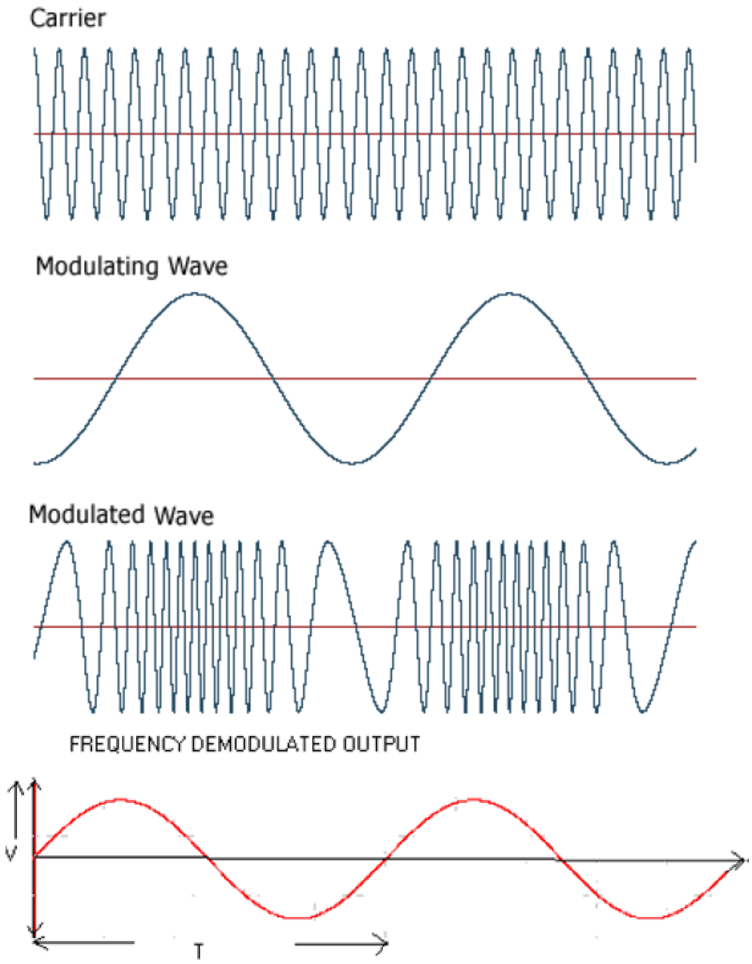
10. Now apply the modulated signal as an input to demodulator circuit and compare the demodulated signal with the input modulating signal & also draw the same on the graph.

11. Observe the demodulated output for changing the amplitude and /or frequency of the modulating Signal.

Observations:

S.No	A_m Volts	f_c Hz	f_m Hz	f_{max} Hz	f_{min} Hz	Freq. Deviation Δf	Modulation index
1							
2							

Model Wave Forms:



Results:

Viva Questions:

1. Define frequency modulation?

2. Mention the advantages of indirect method of FM generation?

3. Define modulation index and frequency deviation of FM?

4. What are the advantages of FM?

5. What is narrow band FM?

6. Compare narrow band FM and wide band FM?

7. Differentiate FM and AM?

8. How FM wave can be converted into PM wave?

9. State the principle of reactance tube modulator?

10. Draw the circuit of varactor diode modulator?

11. What is the bandwidth of FM system?

12. Want is the function of FM discriminator?

13. How does ratio detector differ from fosterseely discriminator?

14. What is meant by linear detector?

15. What are the drawbacks of slope detector?

Due date:

Date of submission:

Student Id. No: Section:

Name of the student. Signature of student

Pre-lab Session work (15M) **Lab experiment**	**In-Lab Session work (15M)** **Project**	**Post Lab session work (10M)**	**Viva** **(10M)**	**Total Marks** **50M**

Remarks:

Marks awarde

Date: **Signature of the Instructor**

Signals & Communication Systems - Lab-11

Spectrum analysis of AM and FM signals

Objectives:

1. To construct and study the Amplitude Modulation using XR2206.
2. To examine the spectrum displays of an AM signal.
3. To measure the bandwidth of both sidebands and AM.
4. To construct and study the Frequency Modulation using XR2206.
5. To examine the spectrum displays of an FM signal.
6. To measure the bandwidth of both sidebands and FM.

Requirements: Digital Computer with MATLAB software.

MAT Lab Code:

```
close all;

clear all;

clc;

% Sampling setup

Fs = 100;                    % Sampling frequency

t = [0:2*Fs+1]' / Fs;        % Time vector

Fc = 10;                     % Carrier frequency

x = sin(2 * pi * 2 * t);     % Message signal

Ac = 1;                      % Carrier amplitude

% -------- AM Modulation (conventional) --------

xam = (Ac + x) .* cos(2 * pi * Fc * t);  % AM with carrier

% Spectrum of AM

zam = fft(xam);

zam = abs(zam(1:floor(length(zam)/2)+1));

frqam = [0:length(zam)-1] * Fs / length(zam) / 2;
```

```
% -------- DSB-SC Modulation --------
x_dsb = x .* cos(2 * pi * Fc * t);       % DSB-SC
% Spectrum of DSB-SC
zdouble = fft(x_dsb);
zdouble = abs(zdouble(1:floor(length(zdouble)/2)+1));
frqdouble = [0:length(zdouble)-1] * Fs / length(zdouble) / 2;
% -------- SSB Modulation (Upper Sideband) --------
pkg load signal;                          % Load signal package for hilbert()
xh = hilbert(x);                          % Analytic signal
x_ssb = real(xh .* exp(1j * 2 * pi * Fc * t));  % Upper SSB
% Spectrum of SSB
zsingle = fft(x_ssb);
zsingle = abs(zsingle(1:floor(length(zsingle)/2)+1));
frqsingle = [0:length(zsingle)-1] * Fs / length(zsingle) / 2;
% -------- FM Modulation (Manual Implementation) --------
kf = 10;                                  % Frequency sensitivity
int_x = cumsum(x) / Fs;                   % Integral of message signal
xfm = cos(2 * pi * Fc * t + 2 * pi * kf * int_x);  % FM signal
% Spectrum of FM
zfm = fft(xfm);
zfm = abs(zfm(1:floor(length(zfm)/2)+1));
frqfm = [0:length(zfm)-1] * Fs / length(zfm) / 2;
% -------- Plotting --------
figure;
subplot(3,1,1);
plot(frqam, zam);
title('Spectrum of AM signal');
```

```
xlabel('Frequency (Hz)'); ylabel('Magnitude'); grid on;

subplot(3,1,2);

plot(frqdouble, zdouble);

title('Spectrum of DSB-SC signal');

xlabel('Frequency (Hz)'); ylabel('Magnitude'); grid on;

subplot(3,1,3);

plot(frqsingle, zsingle);

title('Spectrum of SSB (Upper) signal');

xlabel('Frequency (Hz)'); ylabel('Magnitude'); grid on;

figure;

plot(frqfm, zfm);

title('Spectrum of FM signal');

xlabel('Frequency (Hz)'); ylabel('Magnitude'); grid on;
```

Ongoing Lab:

(a) Type the Matlab codes prepared for the exercises in the computer and observe the results.

(b) Compute the all the tasks given in exercises.

(c) Note the results and store the necessary figures and data for the lab report.

Post Lab:

(a) Complete the lab work in all aspects in the given specified lab time.

(b) Answer the given questions.

(c) Submit the lab report to the lab in-structure and get the signature in time.

(d) Type the complete description of commands used in the lab.

Viva Questions:

1. What is the fundamental difference between AM and FM in terms of modulation technique?
2. How does the spectrum of an AM signal look, and what frequencies does it contain?
3. What is the bandwidth of an AM signal and how is it calculated?
4. How does the modulation index affect the spectrum of an AM signal?
5. Describe the spectral components present in a frequency modulated (FM) signal.
6. Why does an FM signal have a theoretically infinite bandwidth?
7. What is Carson's rule, and how is it used to estimate FM signal bandwidth?

8. How does the frequency spectrum of FM vary with the modulation index?

9. What are sidebands in AM and FM, and how do they differ?

10. How can you analyze or visualize the spectrum of AM and FM signals using Fourier Transform?

Due date:

Date of submission:

Student Id. No: Section:

Name of the student. Signature of student

Pre-lab Session work (15M) **Lab experiment**	**In-Lab Session work (15M)** **Project**	**Post Lab session work (10M)**	**Viva** **(10M)**	**Total Marks** **50M**

Remarks:

Marks awarde

Date: **Signature of the Instructor**

Signals & Communication Systems - Lab -12
Pre-Emphasis & De-Emphasis

Objectives:

1. To design a pre-emphasis circuit to boost the input signal level for a FM transmitter for a cut off frequency of 1KHz.
2. Attenuate the boosted high frequency signals at the receiver side using a deemphasis circuit with a cutoff frequency of 1.6KHz.
3. Analyse the frequency response characteristics of pre-emphasis and de-emphasis circuits.

Equipment and Components:

1. Pre-emphasis & De-emphasis trainer kits.
2. C.R.O (20 MHz)
3. Function generator (1MHz).
4. Patch chords and Probes.
5. PC with windows (95/98/XP/NT/2000)
6. MATLAB Software with communication toolbox

Basic Theory:

Frequency modulation is much immune to noise than amplitude modulation and significantly more immune than phase modulation. A single noise frequency will affect the output of the receiver only if it falls within its pass band.

The noise has a greater effect on the higher modulating frequencies than on lower ones. Thus, if the higher frequencies were artificially boosted at the transmitter and correspondingly cut at the receiver, improvement in noise immunity could be expected. This booting of the higher frequencies, in accordance with a pre-arranged curve, is termed pre-emphasis, and the compensation at the receiver is called de-emphasis.

If the two modulating signals have the same initial amplitude, and one of them is pre-emphasized to (say) twice this amplitude, whereas the other is unaffected (being at a much lower frequency) then the receiver will naturally have to de-emphasize the first signal by a factor of 2, to ensure that both signals have the same amplitude in the output of the receiver.

Before demodulation, i.e. while susceptible to noise interference the emphasized signal had twice the deviation it would have had without pre-emphasis and was thus more immune to noise. Alternatively, it is seen that when this signal is deemphasized any noise sideband voltages are de-emphasized with it and therefore have a correspondingly lower amplitude than they would have had without emphasis again their effect on the output is reduced.

Apart from that, it would be difficult to introduce pre-emphasis and de-emphasis in existing AM services since extensive modifications would be needed, particularly in view of the huge numbers is receivers in use.

Equipment and Components:

Circuit Diagram:

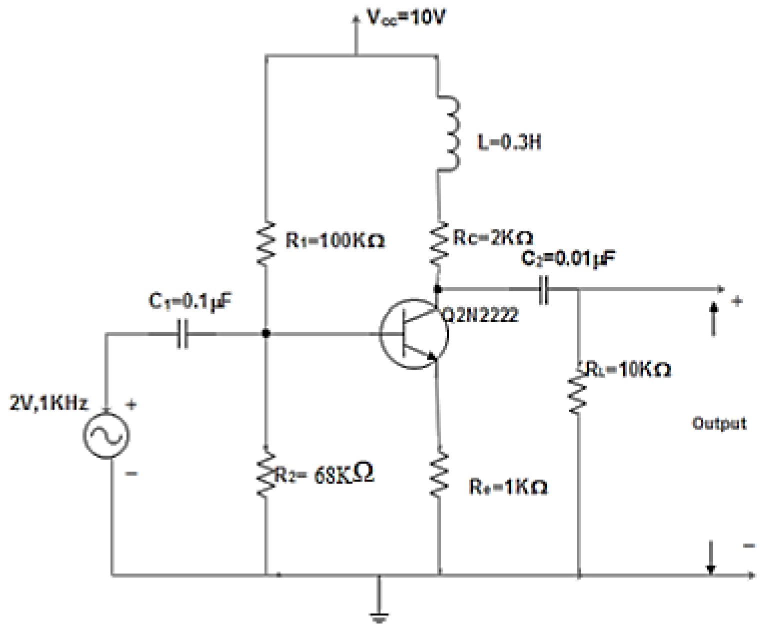


Fig. 1 Pre Emphasis Circuit

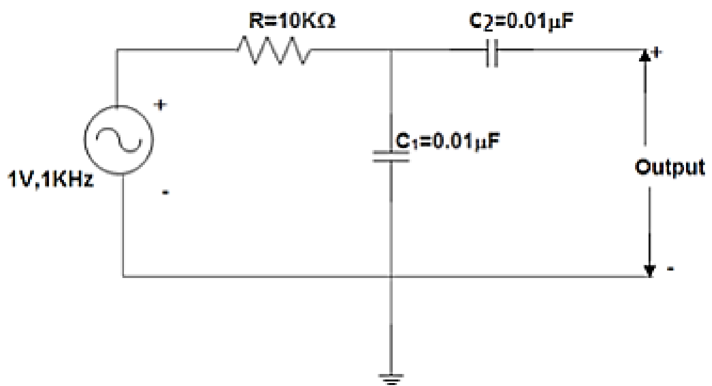


Fig. 2 De Emphasis Circuit

Procedure:

1. The circuit connections are made as shown in the circuit diagram for the pre-emphasis and de-emphasis circuits.
2. A power supply of 10V is given to the pre- emphasis circuit.
3. Set the input voltage at 2V, 1 KHz for pre-emphasis and 1V,1KHz for de-emphasis using AFO.
4. For this constant value of input voltage, the value of the frequency is varied and the output voltage is noted on the CRO.
5. A graph is plotted between gain and frequency in a semi log graph sheet for both pre-emphasis and de-emphasis outputs.

Model Graph:

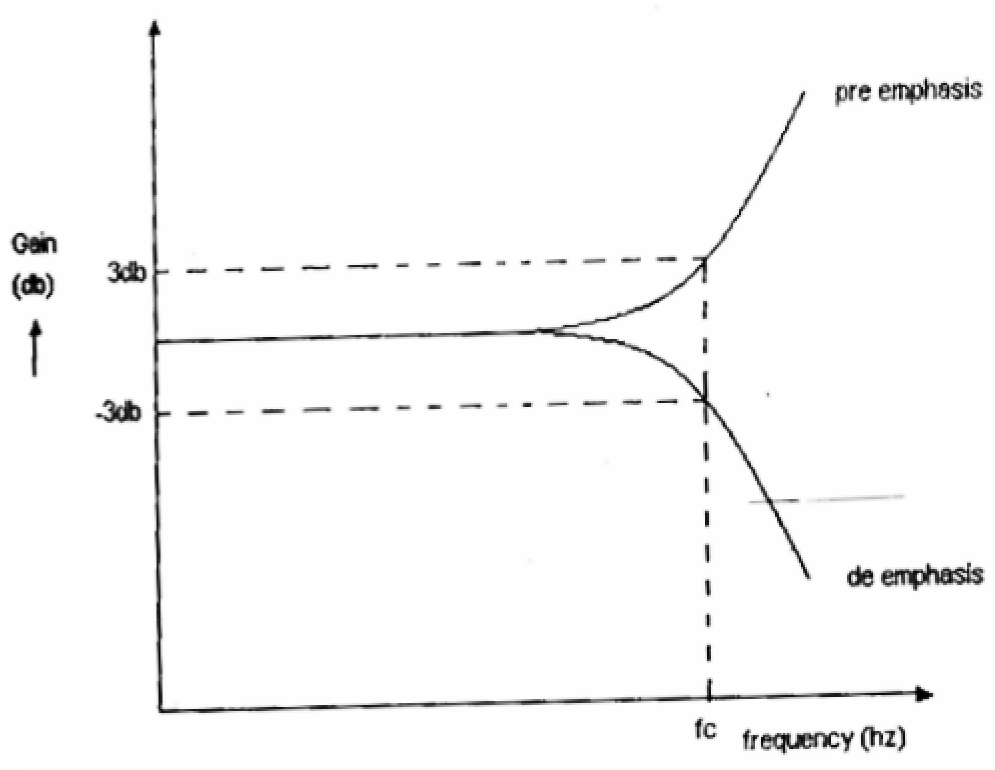


Fig. 3 Pre-emphasis and De-emphasis Characteristics Curve

Observation For Pre-Emphasis:

Vi=

Frequency (Hz)	Vo	Gain= Vo/Vi	Gain in dB=20log (Vo/Vi)

Observation For De-Emphasis:

Vi=

Frequency (Hz)	Vo	Gain= Vo/Vi	Gain in dB=20log (Vo/Vi)

Result:

Viva Questions:

1. What is de-emphasis?

2. How to reduce the noise during transmission in FM?

3. What should be the time constant for the de-emphasis circuit?

4. Why is pre-emphasis done after modulation?

5. List some applications of pre-emphasis circuit.

6. Design a circuit to boost the baseband signal amplitude in the FM transmitter for the cut off frequency fc =2KHz as shown in the figure below.

Due date:

Date of submission:

Student Id. No: Section:

Name of the student. Signature of student

Pre-lab Session work (15M) Lab experiment	In-Lab Session work (15M) Project	Post Lab session work (10M)	Viva (10M)	Total Marks 50M

Remarks:

Marks awarde

Date: **Signature of the Instructor**

Signals & Communication Systems - Lab -13

Pulse Amplitude Modulation

Objectives:

1. To construct and study the Pulse Amplitude Modulation technique.
2. To examine the time displays of PAM signal.
3. To study the effect of PAM for varying width of carrier pulse train.
4. To study the demodulating PAM signals.

Pre-Lab Work:

1. Basic theory of Pulse Amplitude modulation techniques. Time and Frequency analysis of PAM waves.
2. Basic theory of PAM demodulation techniques.
3. Understanding the data sheets of components used in the experiment.
4. Computer simulations (Multisim / pSpice) are performed and the objectives are obtained prior to the hardware experiment.

Equipment and Components:

Circuit Diagram:

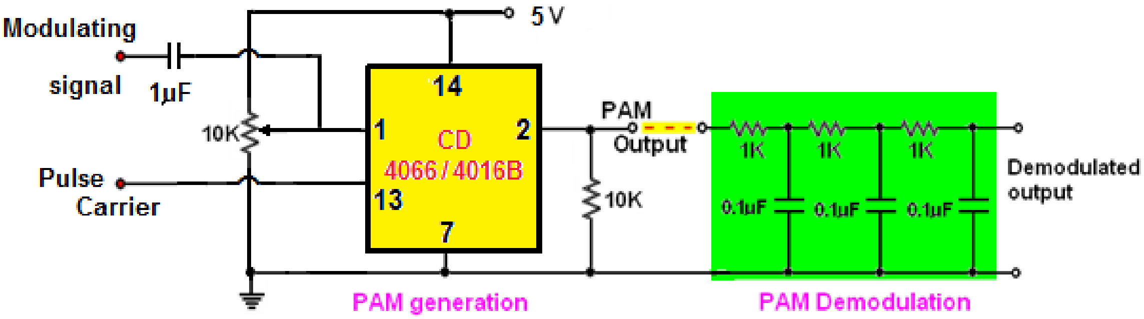


Fig.1 Circuit diagram of PAM modulation and demodulation.

Brief Theory:

PAM Modulation:

Pulse modulation is one form of pulse modulation used in the transmission of digital signals in a message processing format. In PAM, the RF carrier pulses amplitude is varied in accordance with the instantaneous values of a continuous message signal. The simplest form of the PAM modulator is an analog switch that is turned on and off at the RF carrier pulse rate. As the switch changes state, the intelligence signal is connected and disconnected from the output. Thus the output PAM signal is a sampled version of the input intelligence signal. The PAM wave contains upper and lower side band frequencies besides the modulating and pulse signals. Fig.1 shows the circuit diagram of PAM modulation.

PAM Demodulation:

A spectral analysis of complex PAM signal reveals that the intelligence signal frequency components are far removed in frequency from the other frequency components of the complex PAM waveform. This is true only if the sampling rate is considerably higher than the highest intelligence frequency being used. Thus if the sampling rate is kept enough in comparison to the intelligence signals being used, a simple low pass filter (LPF) can be used as a PAM demodulator to retrieve the original information.

In the PAM demodulated process, the PAM signal is passed through a low pass filter having a cut-off frequencies equal to the highest frequency in the modulating signal. At the output of the filter, the modulating signal along with the DC component is available. PAM has the same signal to noise ratio as AM and so it is not employed in practical circuits. Fig.1 shows the circuit diagram of PAM demodulation.

Procedure:

1. Connections are made as per the PAM circuit diagram shown in Fig.1.

2. Set the modulating frequency to 200HZ, magnitude 1Vp-p with offset 3.5V.

3. Set the sampling frequency to 1KHz, 0-4.5 V and 50 % duty cycle.

4. Observe the PAM wave output on CRO.

5. Measure the levels of Vmax and Vmin .

6. Observe the demodulated output. If necessary set the potentiometer to a value such that the demodulated output is reproduced without any distortion.

7. Now increase the frequency of the input sinusoidal (the amplitude and offset remains the same) to 400 Hz, 500 Hz (the Nyquist frequency) and 600 Hz. Comment on the PAM modulated waveforms.

8. Observe demodulation output for increase the frequency of the input sinusoid as in step 6.

9. Now vary the magnitude of the input sinusoidal signal (the frequency is remains constant).

10. Observe the PAM output and corresponding demodulated output in each case.

Model Wave Forms:

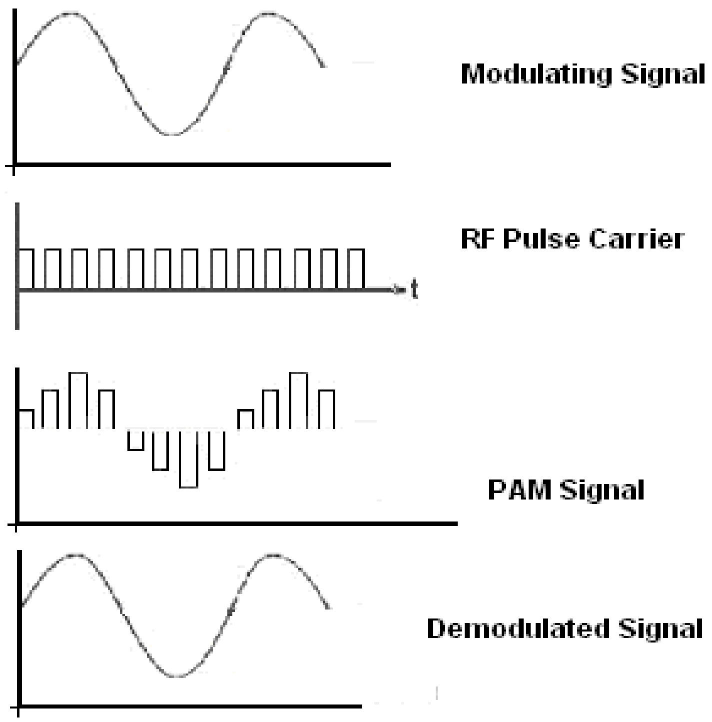


Fig 2. Pulse Amplitude Modulation and Demodulation

Precautions:

1. Connections should be made carefully.
2. The components (resistors, capacitors and ICs) must be identified properly before giving the circuit connections.
3. The components must be properly doped into the bread board.

Results:

Post-Lab Requirements

1. Create the illustration for Pulse Amplitude Modulation.
2. Compare the results are obtained in hardware lab with that of computer simulations.
3. Submit your illustration to the lab instructor at next week's lab.

Viva Questions:

1. What are the classifications of pulse modulation techniques?
2. What is the transmission bandwidth of Pulse amplitude modulation?
3. What are the limitations in Pulse amplitude modulated signal?
4. What do you mean by synchronization in PAM?
5. Write the standard equation of a PAM in frequency domain?
6. What is meant by Aperture effect?
7. Draw the frequency spectrum of a PAM signal?
8. What is the time domain representation of a PAM signal?
9. What are the major differences between PAM &PWM?
10. Which type of sampling technique is used in PAM signal?

Due date:

Date of submission:

Student Id. No: Section:

Name of the student. Signature of student

Pre-lab Session work (15M) Lab experiment	In-Lab Session work (15M) Project	Post Lab session work (10M)	Viva (10M)	Total Marks 50M

Remarks: **Marks awarde** **Date:** **Signature of the Instructor**

Signals & Communication Systems - Lab -14

Pulse Width Modulation

Objectives:

1. To construct and study the Pulse Width Modulation technique.
2. To examine the time displays of PWM signal.
3. To study the demodulating PWM signals.

Pre-Lab Work:

1. Basic theory of Pulse Width modulation techniques. Time and Frequency analysis of PAM waves.
2. Basic theory of PWM demodulation techniques.
3. Understanding the data sheets of components used in the experiment.
4. Computer simulations (Multisim / pSpice) are performed and the objectives are obtained prior to the hardware experiment.

Equipment and Components:

Circuit Diagram:

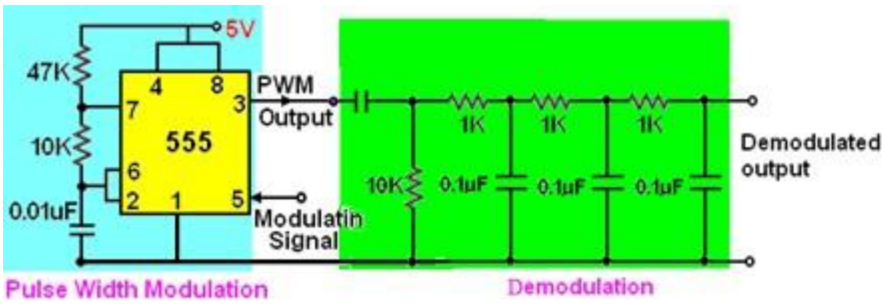


Fig.1 Pulse width Modulation and Demodulation

Basic Theory

Pulse Time Modulation is also known as Pulse Width Modulation or Pulse Length Modulation. In PWM, the samples of the message signal are used to vary the duration of the individual pulses. Width may be varied by varying the time of occurrence of leading edge, the trailing edge or both edges of the pulse in accordance with modulating wave. It is also called Pulse Duration Modulation. In Pulse width modulation, the amplitude of the pulses is constant. In generation of PWM, the input modulating signal is applied to non - inverting terminal of op-amp. The op-amp now compares this modulating signal with internally generated saw tooth signal. The output of the comparator is high only when instantaneous value of input modulating signal is grater then that of saw tooth waveform. When saw tooth waveform voltage is grater then input modulating signal at that instant the output of the comparator remains zero i.e. in negative saturation. Thus output of comparator is PWM signal. A low pass filter is used to demodulate the PWM modulated signal as shown in Fig.1.

Procedure:

1. Connections are made as per the PWM circuit diagram shown in Fig.1.

2. Set the modulating frequency to 200HZ, magnitude 2Vp-p.

3. Observe the PWM wave output on CRO.

4. Now wire the demodulation circuit as shown in Fig.1.

5. Observe the demodulated output. If necessary set the potentiometer to a value such that the demodulated output is reproduced without any distortion.

6. Now vary the magnitude of the input sinusoidal signal (the frequency is remains constant).

7. Observe the PWM output and corresponding demodulated output in each case.

8. Now increase the frequency of the input sinusoidal (the amplitude and offset remains the same) to 400 Hz, 500 Hz and 600 Hz. Comment on the PWM modulated waveforms.

9. Observe demodulation output for increase the frequency of the input sinusoid as in step 6.

Output Waveform:

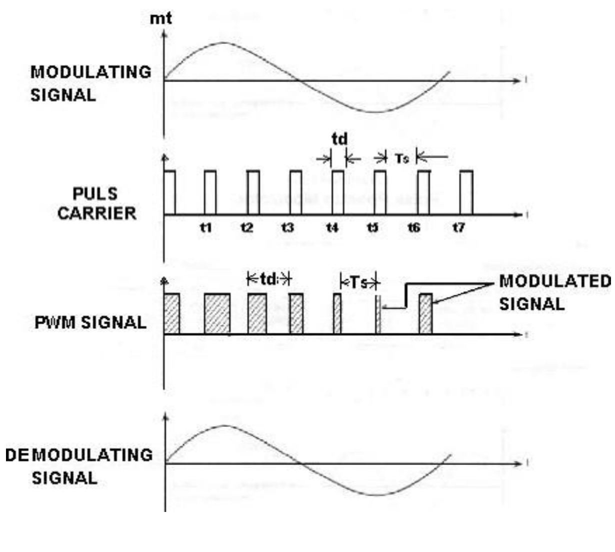


Fig.2. Expected waveforms

Precautions:

1. Connections should be made carefully.

2. The components (resistors, capacitors and ICs) must be identified properly before giving the circuit connections.

3. The components must be properly doped into the bread board.

Results:

Post-Lab Requirements

1. Create the illustration for Pulse Width Modulation

2. Compare the results are obtained in hardware lab with that of computer simulations.

3. Submit your illustration to the lab instructor at next week's lab.

Viva Questions:

1. What are the different types of PTM systems?

2. What is the other name of Pulse width modulation?

3. What do you mean by pulse time modulation?

4. What is the comparison between the PAM and PWM?

5. What is the definition of PWM?

6. What is the transmission bandwidth of PWM signal?

7. Which type sampling technique is used in PWM?

8. What are the applications of PWM modulation technique?

Due date:

Date of submission:

Student Id. No: Section:

Name of the student. Signature of student

Pre-lab Session work (15M) **Lab experiment**	**In-Lab Session work (15M)** **Project**	**Post Lab session work (10M)**	**Viva** **(10M)**	**Total Marks** **50M**

Remarks:

Marks awarde

Date: **Signature of the Instructor**

Signals & Communication Systems - Lab -15

Pulse Position Modulation

Objectives:

1. To construct and study the Pulse Position Modulation technique.

2. To examine the time displays of PPM signal.

3. To study the demodulating PPM signals.

Pre-Lab Work:

1. Basic theory of Pulse Position modulation techniques. Time and Frequency analysis of PPM waves.

2. Basic theory of PPM demodulation techniques.

3. Understanding the data sheets of components used in the experiment.

4. Computer simulations (Multisim / pSpice) are performed and the objectives are obtained prior to the hardware experiment.

Equipment and Components:

Circuit Diagram:

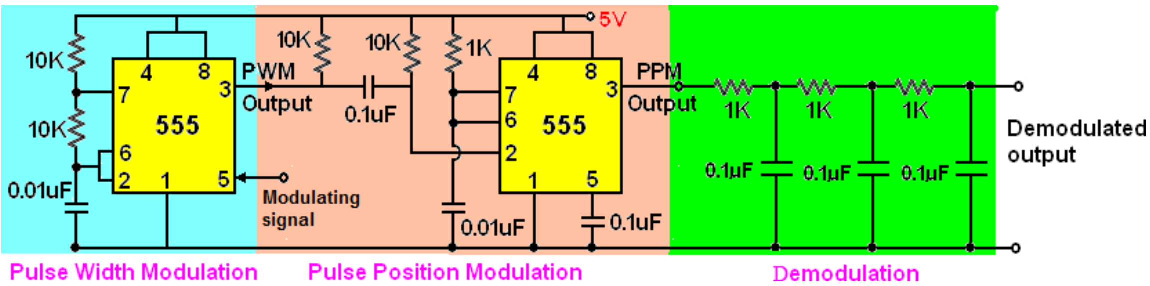
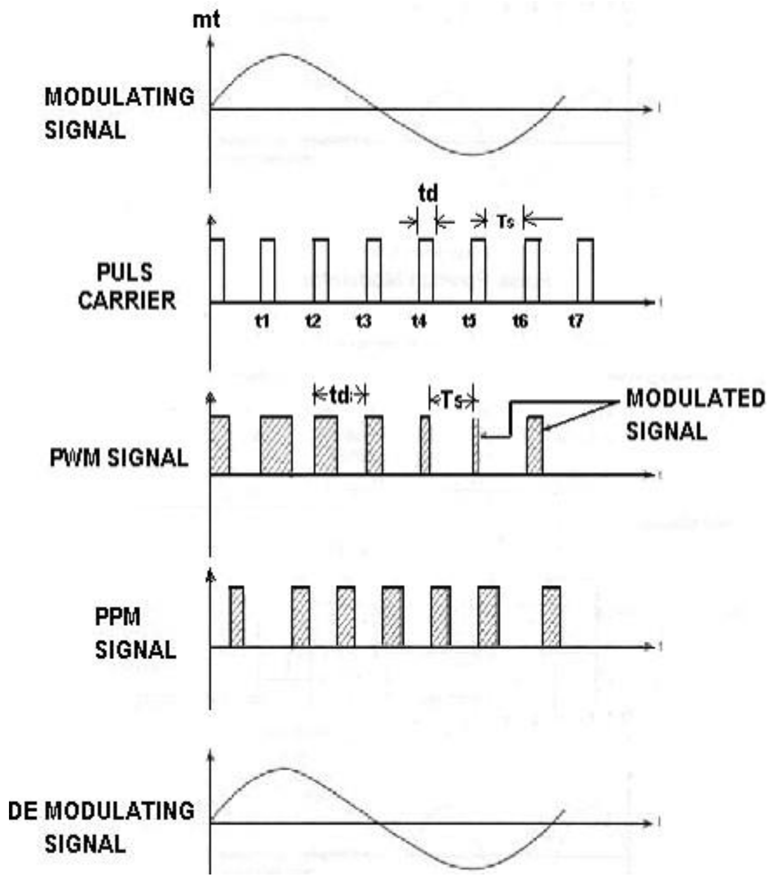


Fig1. The circuit diagram of Pulse Position Modulation using PWM and Demodulation.

Basic Theory:

In Pulse Position Modulation, both the pulse amplitude and pulse duration are held constant but the position of the pulse is varied in proportional to the sampled values of the message signal. Pulse time modulation is a class of signaling techniques that encodes the sample values of an analog signal on to the time axis of a digital signal and it is analogous to angle modulation techniques. The two main types of PTM are PWM and PPM. In PPM the analog sample value determines the position of a narrow pulse relative to the clocking time. In PPM rise time of pulse decides the channel bandwidth. It has low noise interference. Generation of PPM is carried out by giving the pulse width modulated output to mono-stable multi-vibrator as shown in Fig.1. A low pass filter is used to demodulate the PWM modulated signal as shown in Fig.1.

Output Waveform:



Precautions:

1. Connections should be made carefully.

2. The components (resistors, capacitors and ICs) must be identified properly before giving the circuit connections.

3. The components must be properly doped into the bread board.

Results:

Post-Lab Requirements

1. Create the illustration for Pulse Width Modulation

2. Compare the results are obtained in hardware lab with that of computer simulations.

3. Submit your illustration to the lab instructor at next week's lab.

Viva Questions:

1. Define PPM?

2. What are the differences between PPM & PWM?

3. Which type of sampling technique is used in PPM?

4. The Multivibrator used in PPM is?

5. Pulse duration of PPM is?

6. In PPM the position is proportionally varied in which parameter of the modulating signal?

Due date:

Date of submission:

Student Id. No: Section:

Name of the student. Signature of student

Pre-lab Session work (15M) Lab experiment	In-Lab Session work (15M) Project	Post Lab session work (10M)	Viva (10M)	Total Marks 50M
Remarks: Marks awarde Date: Signature of the Instructor				