MUFFAKHAM JAH COLLEGE OF ENGINEERING & TECHNOLOGY

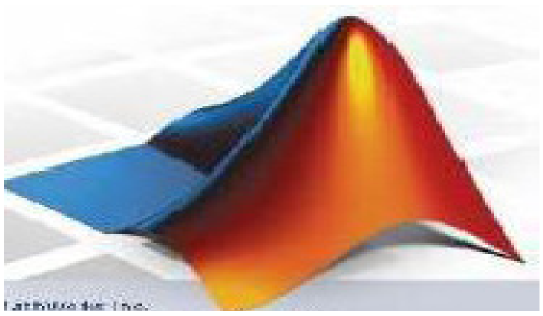
Banjara Hills Road No 3, Hyderabad- 34

www.mjcollege.ac.in

ELECTRICAL ENGINEERING DEPARTMENT

LABORATORY MANUAL

DIGITAL SIGNAL PROCESSING LAB



For

IV/IV B.E II SEM EEE/EIE

Prepared by

Mohd. Abdul Muqeet
Assoc. Professor, EED

WITH EFFECT FROM THE ACADEMIC YEAR 2013-2014

EE 481

DSP LAB (COMMON TO EEE & IE)

Instruction	3 Periods per week
Duration of University Examination	3 Hours
University Examination	50 Marks
Sessional	25 Marks

1. **Waveform generation -Square, Triangular and Trapezoidal.**
2. **Verification of Convolution Theorem-comparison Circular and Linear Convolutions.**
3. **Computation of DFT, IDFT using Direct and FFT methods.**
4. **Verification of Sampling Theorem**
5. **Design of Butterworth and Chebyshev of LP & HP filters.**
6. **Design of LPF using rectangular and Hamming, Kaiser Windows.**
7. **16 bit Addition, Integer and fractional multiplication on 2407 DSP Trainer kit.**
8. **Generation of sine wave and square wave using DSP trainer kit.**
9. **Response of Low pass and High pass filters using DSP trainer kit.**
10. **Linear convolution using DSP trainer kit.**
11. **PWM Generation on DSP trainer kit.**
12. **Key pad interfacing with DSP.**
13. **LED interfacing with DSP.**
14. **Stepper Motor Control using DSP.**
15. **DC Motor 4- quadrant speed control using DSP.**
16. **Three phase 1M speed control using DSP.**
17. **Brushless DC Motor Control.**

At least ten experiments should be completed in the semester

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[1]. Waveform generation -Square, Triangular and Trapezoidal.
[2]. Verification of Convolution Theorem-comparison Circular and Linear Convolutions.
[3]. Computation of DFT, IDFT using Direct and FFT methods.
[4]. Verification of Sampling Theorem
[5]. Design of Butterworth and Chebyshev of LP & HP filters.
[6]. Design of LPF using rectangular and Hamming, Kaiser Windows.

Cycle –II

[7]. 16 bit Addition, Integer and fractional multiplication on DSP Trainer kit.
[8]. Generation of sine wave and square wave using DSP trainer kit.
[9]. LED interfacing with DSP.
[10]. DC Motor 4- quadrant speed control using DSP.
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Cycle-I

INTRODUCTION

MATLAB, which stands for **MAT**rix **LAB**oratory, is a state-of-the-art mathematical software package for high performance numerical computation and visualization provides an interactive environment with hundreds of built in functions for technical computation, graphics and animation and is used extensively in both academia and industry. It is an interactive program for *numerical* computation and data visualization, which along with its programming capabilities provides a very useful tool for almost all areas of science and engineering.

At its core ,MATLAB is essentially a set (a “toolbox”) of routines (called “m files” or “mex files”) that sit on your computer and a window that allows you to create new variables with names (e.g. voltage and time) and process those variables with any of those routines (e.g. plot voltage against time, find the largest voltage, etc).
It also allows you to put a list of your processing requests together in a file and save that combined list with a name so that you can run all of those commands in the same order at some later time. Furthermore, it allows you to run such lists of commands such that you pass in data.

MATLAB Windows:
MATLAB works with through these basic windows

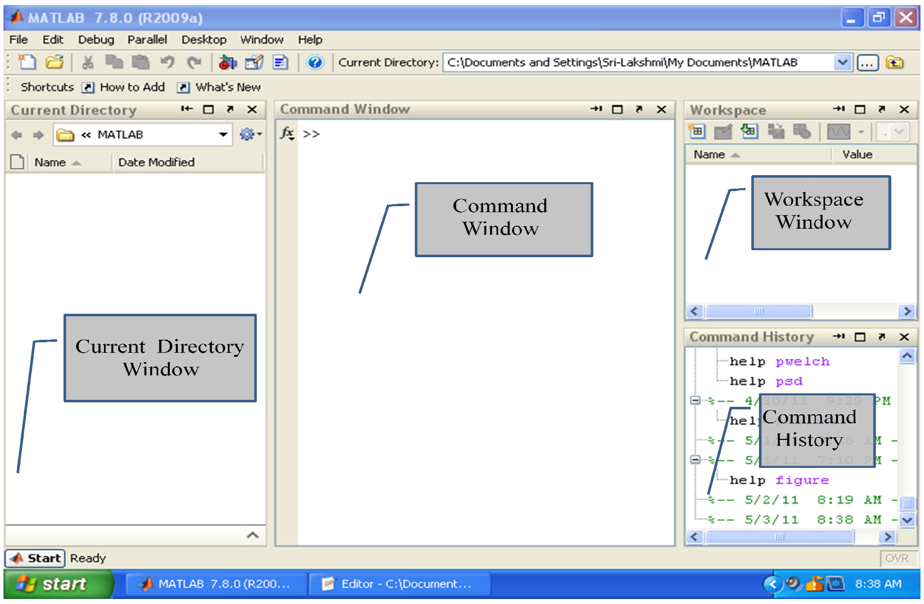
Command Window
This is the main window .it is characterized by MATLAB command prompt >> when you launch the application program MATLAB puts you in this window all commands including those for user-written programs ,are typed in this window at the MATLAB prompt

The Current Directory Window
The Current Directory window displays a current directory with a listing of its contents. There is navigation capability for resetting the current directory to any directory among those set in the path. This window is useful for finding the location of particular files and scripts so that they can be edited, moved, renamed, deleted, etc. The default current directory is the Work subdirectory of the original MATLAB installation directory

The Command History Window
The Command History window, at the lower left in the default desktop, contains a log of commands that have been executed within the Command window. This is a convenient feature for tracking when developing or debugging programs or to confirm that commands were executed in a particular sequence during a multistep calculation from the command line.

Graphics Window
The output of all graphics commands typed in the command window are flushed to the graphics or figure window, a separate gray window with white background color the user can create as many windows as the system memory will allow.

Edit Window
This is where you write edit, create and save your own programs in files called M files.



```
help pwelch
help psd
help figure
%-- 5/2/11  8:19 AM --
%-- 5/3/11  8:38 AM --
```

Input-output

MATLAB supports interactive computation taking the input from the screen and flushing, the output to the screen. In addition it can read input files and write output files

Data Type

The fundamental data –type in MATLAB is the array. It encompasses several distinct data objects- integers, real numbers, matrices, character strings, structures and cells. There is no need to declare variables as real or complex, MATLAB automatically sets the variable to be real.

Dimensioning

Dimensioning is automatic in MATLAB. No dimension statements are required for vectors or arrays .we can find the dimensions of an existing matrix or a vector with the size and length commands.

Where to work in MATLAB?

All programs and commands can be entered either in the
a) Command window
b) As an M file using MATLAB editor

Note: Save all M files in the folder 'work' in the current directory. Otherwise you have to locate the file during compiling.
Typing quit in the command prompt>> quit, will close MATLAB Development Environment.
For any clarification regarding plot etc, which are built in functions type help topic i.e. help plot

Basic Instructions in MATLAB

1. **T = 0: 1:10** This instruction indicates a vector T which as initial value 0 and final value 10 with an increment of 1 Therefore

```
T = [0 1 2 3 4 5 6 7 8 9 10]
```

2. **F= 20: 1: 100**

```
F = [20 21 22 23 24 ……… 100]
```

3. **T= 0:1/ pi: 1**

```
T= [0, 0.3183, 0.6366, 0.9549]
```

4. **zeros (1, 3)** The above instruction creates a vector of one row and three columns whose values are zero Output= [0 0 0]

5. **Transpose a vector**

Suppose `T= [1 2 3],`

Then transpose

```
T’= 1
    2
       3
```

6. **Empty vector**

```
Y = []
Y =
        []
```

6. **Matrix Operation**

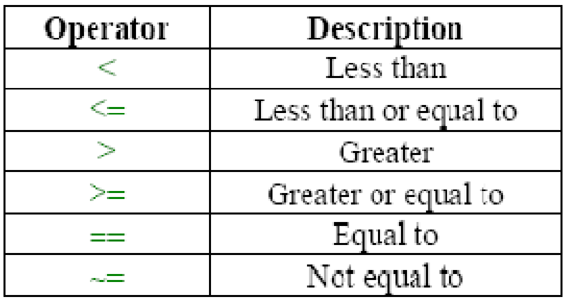
```
a)If a = [ 1 2 3] b = [4 5 6]
     a.*b = [4 10 18]
b)If v = [0:2:8]
          v = [0 2 4 6 8]
  v(1:3)
     ans [0 2 4]
  v(1:2:4)
     ans[ 0 4]
c) A = [1 2 3; 3 4 5; 6 7 8]
    A =
          1 2 3
          3 4 5
          6 7 8
    A(2,3)
       ans 5
    A(1:2,2:3)
        ans =
               2 3
               4 5
    A(:,2)
        ans =
               2
               4
               7
   A(3,:)
       ans =
               6 7 8
```

Operations on vector and matrices in MATLAB

MATLAB utilizes the following arithmetic operators;

+	Addition
-	Subtraction
*	Multiplication
/	Division
^	Power Operator
‘	transpose

Relational operators in MATLAB



Operator	Description
<	Less than
<=	Less than or equal to
>	Greater
>=	Greater or equal to
==	Equal to
~=	Not equal to

Control Flow in MATLAB

1) Syntax of the **for loop** is shown below

```
for k = array
commands
end
```

The commands between **for** and **end** statements are executed for all values stored in the **array**.

2) Syntax for the **if loop**

```
if expression
      commands
end
```

This construction is used if there is one alternative only.
Two alternatives requires the following construction

```
if expression
      commands (evaluated if expression is true)
else
      commands (evaluated if expression is false)
end
```

3) Syntax of the switch-case construction is

```
switch expression (scalar or string)
      case value1 (executes if expression evaluates to value1)
            commands
      case value2 (executes if expression evaluates to value2)
            commands
      ...
      otherwise
            statements
end
```

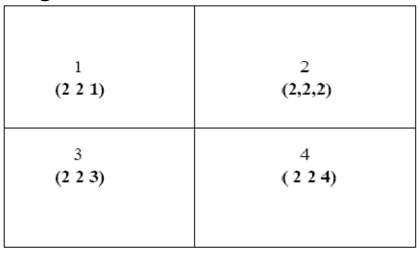
Switch compares the input expression to each case value. Once the match is found it executes the associated commands.

Basic Functions in MATLAB

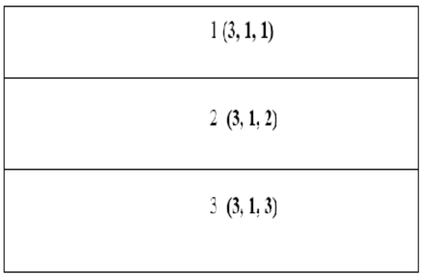
1) **Plot** Syntax: plot (x,y)
Plots vector y versus vector x. If x or y is a matrix, then the vector is plotted versus the rows or columns of the matrix.

2) Stem Syntax: stem(Y)
Discrete sequence or "stem" plot.
Stem (Y) plots the data sequence Y as stems from the x axis terminated with circles for the data value. If Y is a matrix then each column is plotted as a separate series.

3) Subplot Syntax: Subplot (2 2 1)
This function divides the figure window into rows and columns.
Subplot (2 2 1) divides the figure window into 2 rows and 2 columns 1 represent number of the figure.



Subplot (3 1 2) divides the figure window into 3 rows and 1 column 2 represent number of the figure



4) **Disp** Syntax: disp(X)
Description: disp(X) displays an array, without printing the array name. If X contains a text string, the string is displayed.Another way to display an array on the screen is to type its name, but this prints a leading "X=," which is not always desirable.Note that disp does not display empty arrays.

5) **xlabel** Syntax: xlabel('string') Description: xlabel('string') labels the x-axis of the current axes.

6) **ylabel** Syntax : ylabel('string')
Description: ylabel('string') labels the y-axis of the current axes.

7) **Title** Syntax : title('string')
Description: title('string') outputs the string at the top and in the center of the current axes.

8) **grid on** Syntax : grid on
Description: grid on adds major grid lines to the current axes.

Experiment – 1

Aim :- To generate the waveform for the following signals using MATLAB.

1) Sine Wave signal
2) Cosine Wave signal
3) Saw Tooth Wave signal
4) Square Wave signal
5) Triangular Wave signal
6) Trapezoidal Wave signal

Apparatus: Matlab Software, PC

Algorithm:-

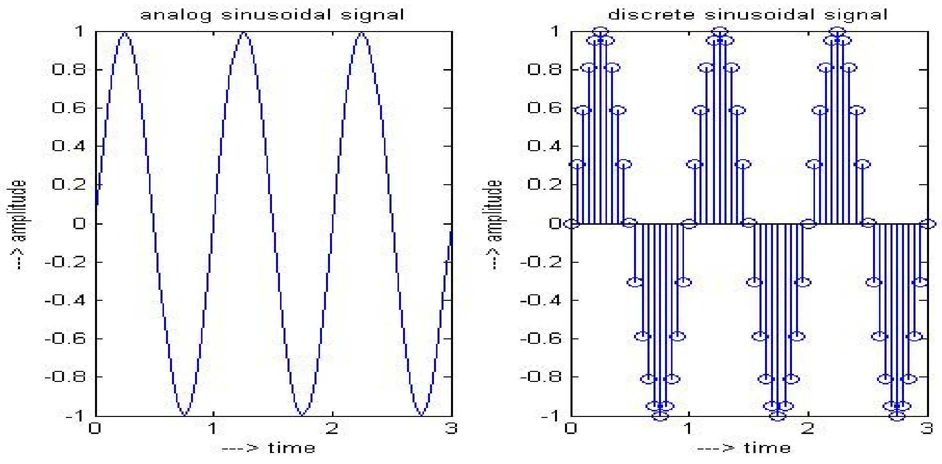
1) Enter the number of cycles, period and amplitude for respective waves.
2) Generate the signals using corresponding general formula.
3) Plot the graph.

Program:

```
1)% To generate a sinusoidal signal
clear all;
close all;clc;
N = input('enter the number of cycles....');
t = 0:0.05:N;
x = sin(2*pi*t);
subplot(121);
plot(t,x);
xlabel('---> time');
ylabel('---> amplitude');
title('analog sinusoidal signal');
subplot(122);
stem(t,x);
xlabel('---> time');
ylabel('---> amplitude');
title('discrete sinusoidal signal');
```

Results:

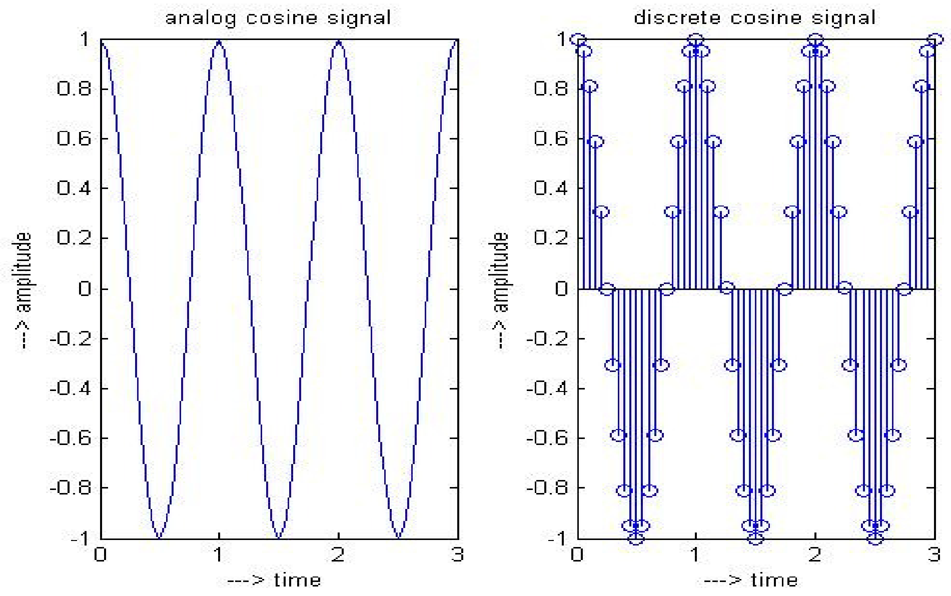
enter the number of cycles....3



```
2)% To generate a Cosine Wave signal

clear all;
close all;
clc;
N = input('enter the number of cycles....');
t = 0:0.05:N;
x = cos(2*pi*t);
subplot(121);
plot(t,x);
xlabel('---> time');
ylabel('---> amplitude');
title('analog cosine signal');
subplot(122);
stem(t,x);
xlabel('---> time');
ylabel('---> amplitude');
title('discrete cosine signal');
```

Results:

```
enter the number of cycles....3
```

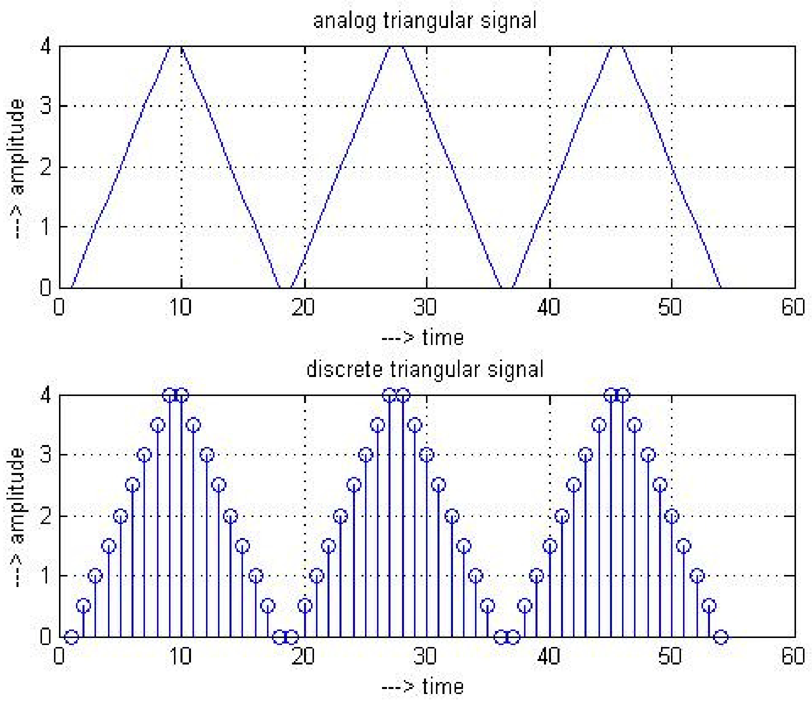


```
3) % To generate a triangular signal
clc;
clear all;
close all;
N = input('enter the number of cycles....');
M = input('enter the amplitude....');
t1 = 0:0.5:M;
t2 = M:-0.5:0;
t = [];
for i = 1:N,
    t = [t,t1,t2];
end;
subplot(211);
```

```
plot(t); grid on;
xlabel('---> time');
ylabel('---> amplitude');
title('analog triangular signal');
subplot(212);
stem(t); grid on;
xlabel('---> time');
ylabel('---> amplitude');
title('discrete triangular signal');
```

Results:

```
enter the number of cycles....3
enter the amplitude....4
```

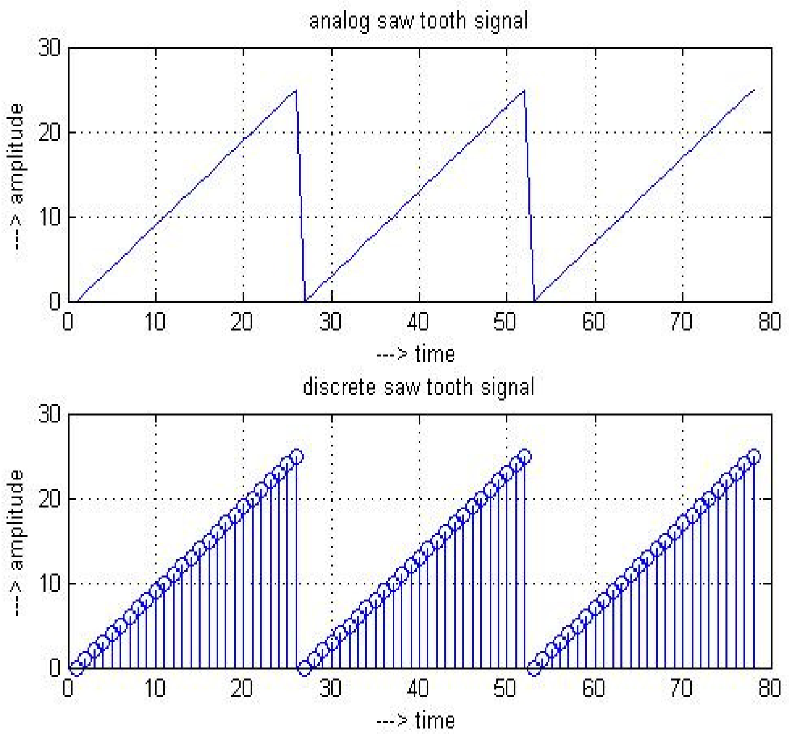


```
4) % To generate a saw tooth signal
clear all;
close all;
clc;
N = input('enter the number of cycles....');
t1 = 0:25;
t = [];
for i = 1:N,
    t = [t,t1];
end;
subplot(211);
plot(t); grid on;
xlabel('---> time');
ylabel('---> amplitude');
title('analog saw tooth signal');
subplot(212);
stem(t); grid on;
```

```
xlabel('---> time');
ylabel('---> amplitude');
title('discrete saw tooth signal');
```

Results:

```
enter the number of cycles....3
```

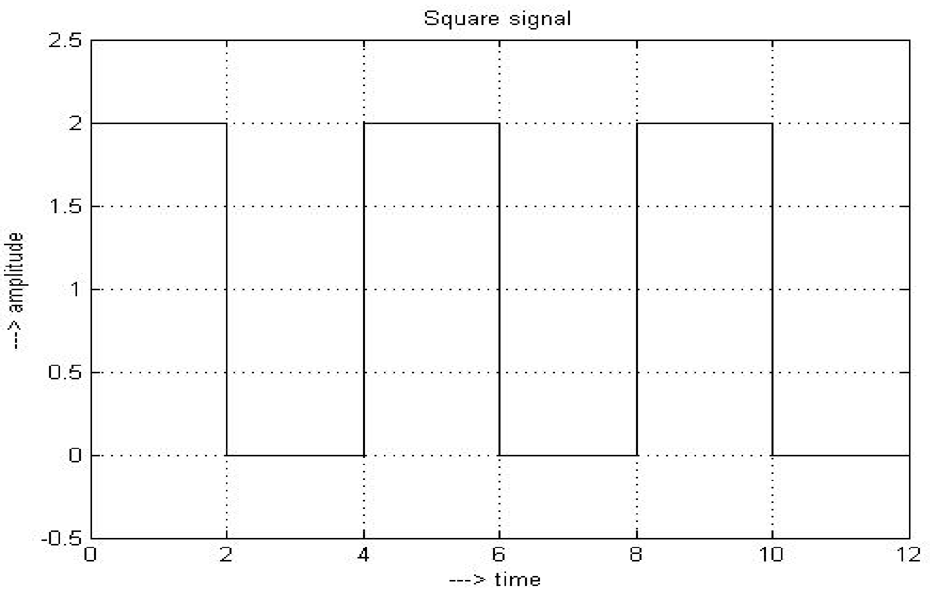


```
5)% To generate a square signal
clear all;
close all; clc;
N = input('enter the number of cycles....');
M = input('enter the period....');
y = 0:0.001:2;
for j = 0:M/2:M*N;
    x = y;
    plot(j,x,'k'); grid on;
    hold on;
end;
for k = 0:M:M*N;
    x = k+y;
    m = 2;
    plot(x, m, 'k'); grid on;
    hold on;
end;
for k =2:M:M*N;
    x = k+y;
    m =0;
    plot(x, m, 'k'); grid on;
    hold on;
```

```
end;
hold off;

axis([0 12 -0.5 2.5])
xlabel('---> time');

ylabel('---> amplitude');
title('Square signal');
```

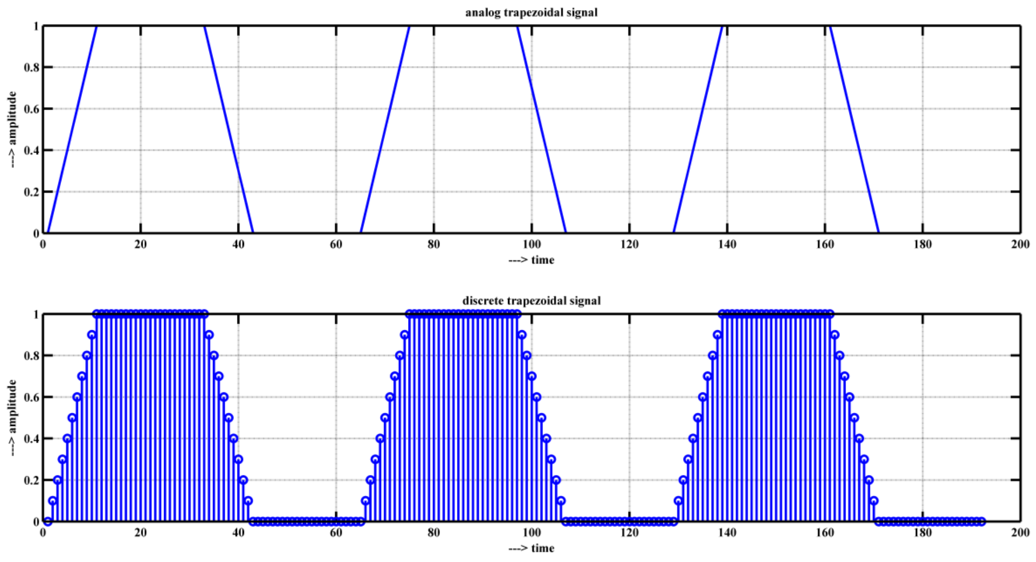
Results:

```
enter the number of cycles....4
enter the period....4
```



```
5)% To generate a Trapezoidal signal

clear all;
close all;
clc;
N = input('enter the number of cycles....');
LN=1;
x=0:0.1:LN; % 'x' is meant for linear rise %
a=length(x);
y=ones(1,a+10); % 'y' is meant for constancy %
z=LN:-0.1:0; % 'z' is meant for linear fall  %
y3=[x y z ];
%y4=[y3 y3 y3 y3];
y4=[];
for i = 1:N,
    y4=[y4,y3];
end;
subplot(211);
plot(y4); grid on;
xlabel('---> time');
ylabel('---> amplitude');
title('analog trapezoidal signal');
```

```
subplot(212);
stem(y4); grid on;
xlabel('---> time');
ylabel('---> amplitude');

title('discrete trapezoidal signal');


Results:
enter the number of cycles....3
```



Discussions on results:

Thus different waveforms have been generated in Matlab and plotted with respect to time.

By performing the experimentation the student will be to

1. Discuss the effect of change in number of cycles on waveform.
2. Discuss the effect of change in time duration on the waveform
3. Discuss the application and significance of each waveform in digital signal processing.

Experiment – 2

Aim: Write a Matlab program to verify Convolution Theorem-comparison Circular and Linear Convolutions.

a) Write a Matlab program to implement and verify Linear Convolution.

Apparatus: Matlab Software, PC

Theory:

The mathematical definition of convolution in discrete time domain

$$y[n]=x[n]*h[n]=\sum_{k=-\infty}^{\infty}x[k]\cdot h[n-k]$$

where *x*[n] is input signal, *h*[n] is impulse response, and *y*[n] is output. * denotes convolution. Here we multiply the terms of *x*[k] by the terms of a time-shifted *h*[n] and add them up.
In this equation, x(k), h(n-k) and y(n) represent the input to and output from the system at time n. Here one of the input is shifted in time by a value every time it is multiplied with the other input signal. Linear Convolution is quite often used as a method of implementing filters of various types.

Algorithm:

1) Give input sequence x[n].
2) Give impulse response sequence h[n].
3) Find the convolution y[n] using the matlab command CONV.
4) Plot x[n],h[n],y[n].

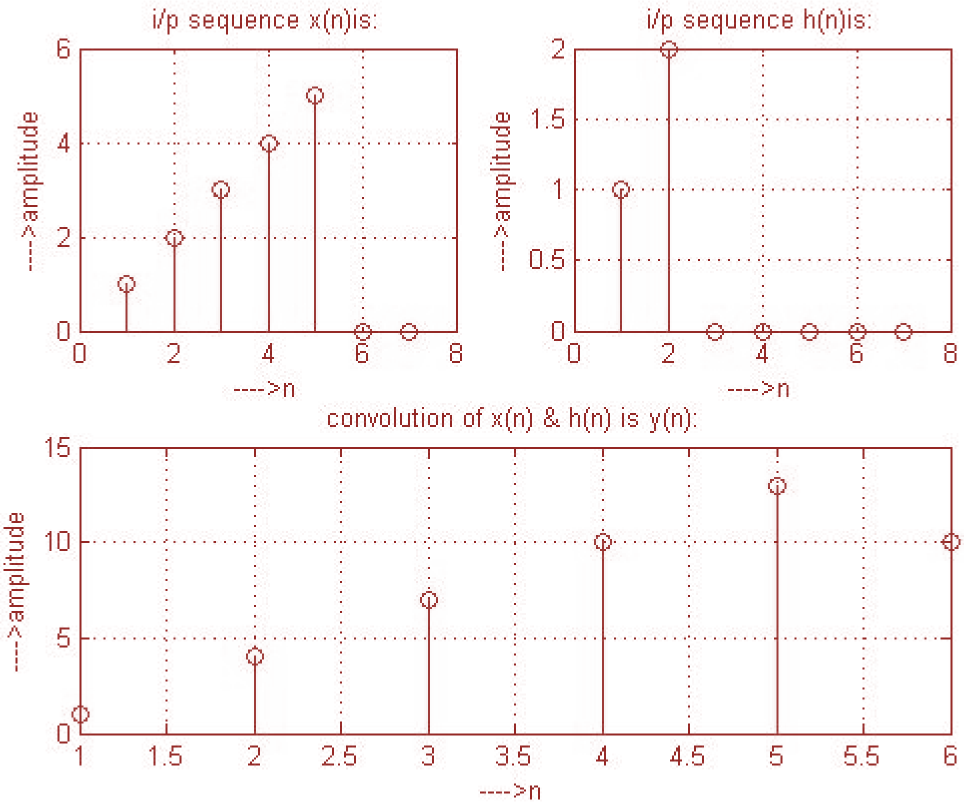
Program:

```
% MATLAB program for linear convolution
clc;
clear all;
close all;
disp('linear convolution program');
x=input('enter i/p x(n):');
m=length(x);
h=input('enter i/p h(n):');
n=length(h);
x=[x,zeros(1,n)];
subplot(2,2,1), stem(x);
title('i/p sequence x(n)is:');
xlabel('---->n');
ylabel('---->amplitude');grid;
h=[h,zeros(1,m)];
subplot(2,2,2), stem(h);
title('i/p sequence h(n)is:');
xlabel('---->n');
ylabel('---->amplitude');grid;
disp('convolution of x(n) & h(n) is y(n):');
y=zeros(1,m+n-1);
for i=1:m+n-1
```

```
y(i)=0;

for j=1:m+n-1
if(j<i+1)
     y(i)=y(i)+x(j)*h(i-j+1);
end
end
end
y
subplot(2,2,[3,4]),stem(y);
title('convolution of x(n) & h(n) is y(n):');
xlabel('---->n');
ylabel('---->amplitude');grid;
```

Results:

```
linear convolution program
enter i/p x(n):[1 2 3 4 5]
enter i/p h(n):[ 1 2]
convolution of x(n) & h(n) is y(n):

y =

     1     4     7    10    13    10
```



b) Write a Matlab program to implement and verify Circular convolution of two given sequences.

Apparatus: Matlab Software, PC

Theory:

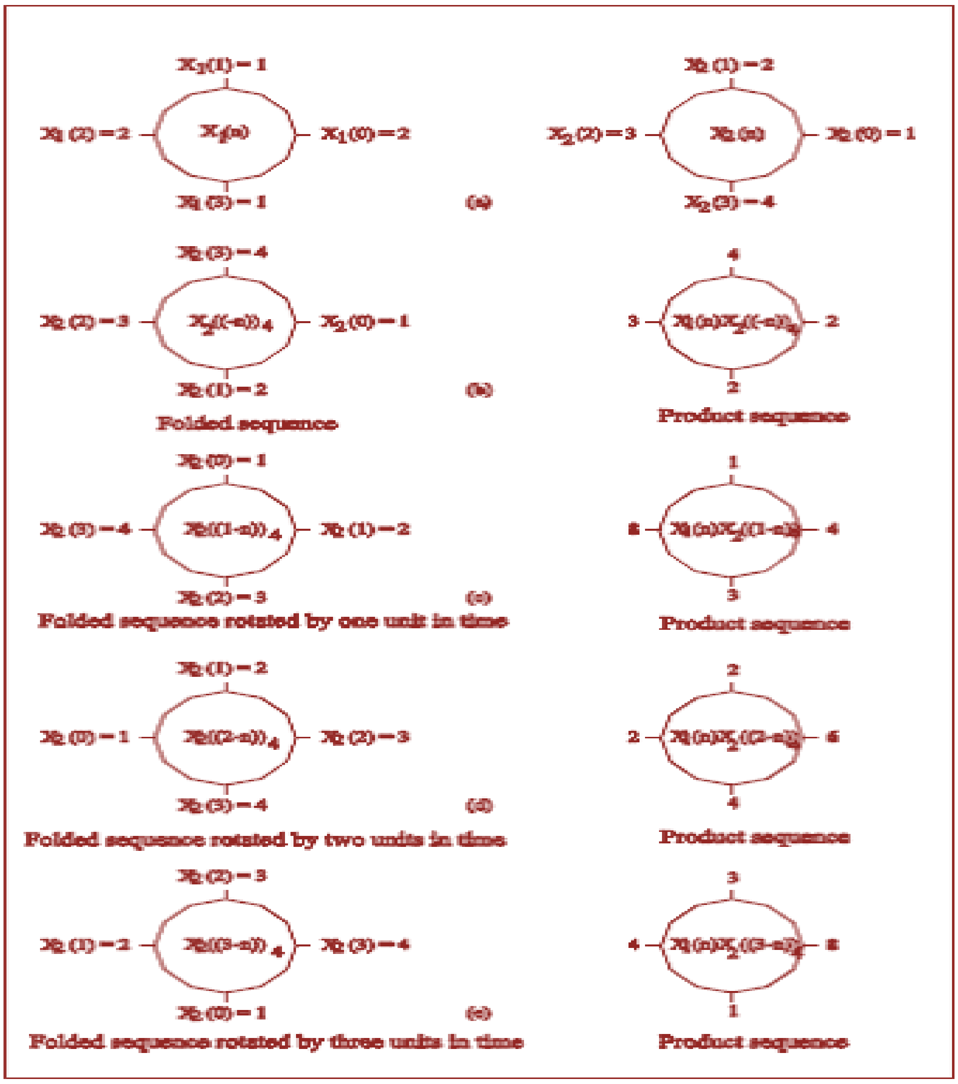
Circular convolution is another way of finding the convolution sum of two input signals. It resembles the linear convolution, except that the sample values of one of the input signals is folded and right shifted before the convolution sum is found. Also note that circular convolution could also be found by taking the DFT of the two input signals and finding the product of the two frequency domain signals. The Inverse DFT of the product would give the output of the signal in the time domain which is the circular convolution output. The two input signals could have been of varying sample lengths. But we take the DFT of higher point, which ever signals levels to. For eg. If one of the signal is of length 256 and the other spans 51 samples, then we could only take 256 point DFT. So the output of IDFT would be containing 256 samples instead of 306 samples, which follows N1+N2 – 1 where N1 & N2 are the lengths 256 and 51 respectively of the two inputs. Thus the output which should have been 306 samples long is fitted into 256 samples. The 256 points end up being a distorted version of the correct signal. This process is called circular convolution.
Circular convolution is explained using the following example.
The two sequences are

$$x1\ (n) = \{2,1,2,1\}$$
$$x2\ (n) = \{1,2,3,4\}$$

Each sequence consists of four nonzero points. For purpose of illustrating the operations involved in circular convolution it is desirable to graph each sequence as points on a circle. Thus the sequences x1 (n) and x2 (n) are graphed as illustrated in the **fig**.We note that the sequences are graphed in a counterclockwise direction on a circle.This stablishes the reference direction in rotating one of sequences relative to the other. Now, y (m) is obtained by circularly convolving x (n) with h (n).

Algorithm:

1) Give input sequence x[n].
2) Give impulse response sequence h[n].
3) Find the Circular Convolution y[n] using the DFT method.
4) Plot x[n],h[n],y[n].



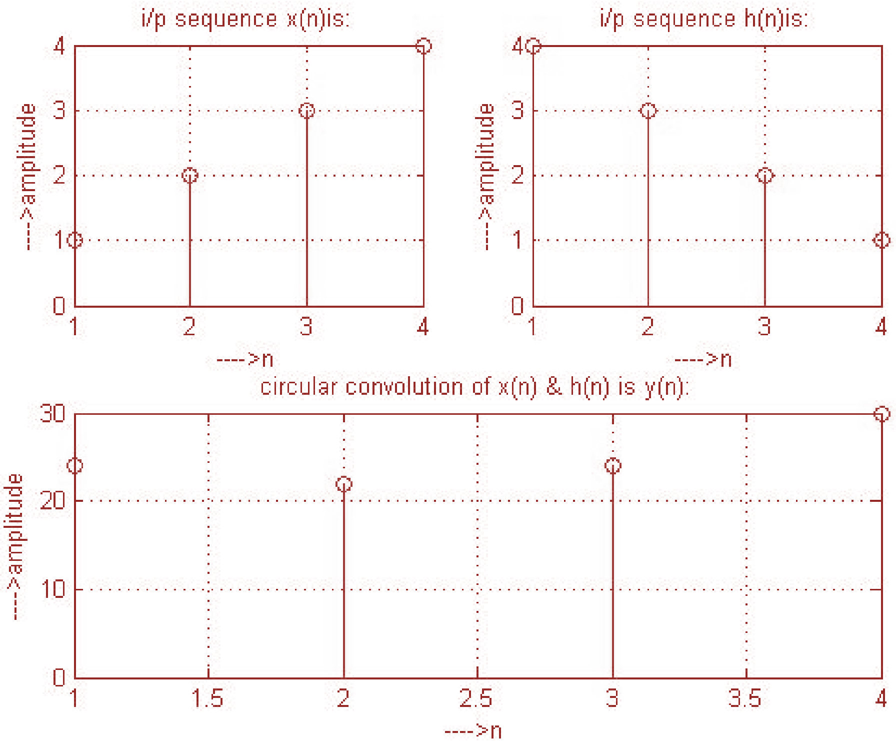
Program:

```
clc;
clear all;
close all;
disp(`Circular convolution program');
x=input('enter i/p x(n):');
m=length(x);
h=input('enter i/p h(n):');
n=length(h);
subplot(2,2,1), stem(x);
title('i/p sequence x(n)is:');
xlabel('---->n');
ylabel('---->amplitude');grid;
subplot(2,2,2), stem(h);
title('i/p sequence h(n)is:');
xlabel('---->n');
ylabel('---->amplitude');grid;
disp('circular convolution of x(n) & h(n) is y(n):');
y1=fft(x,n);
y2=fft(h,n);
y3=y1.*y2;
```

```
y=ifft(y3,n);
y
subplot(2,2,[3,4]),stem(y);
title('circular convolution of x(n) & h(n) is y(n):');
xlabel('---->n');
ylabel('---->amplitude');grid;
```

Result:

```
Circular convolution program
enter i/p x(n):[1 2 3 4]
enter i/p h(n):[4 3 2 1]
circular convolution of x(n) & h(n) is y(n):

y =

    24    22    24    30
```



Discussions on results:

Thus the Linear convolution and circular convolution for discrete time signals are obtained mathematically and graphically .Through this experiment student will be able to

1) Discuss the effect on results if zero padding is used in the program.
2) Discuss the effect on results if zero padding is not used in the program.
3) Discuss the results in obtaining the circular convolution without using frequency domain technique.
4) Discuss the difference between linear convolution and circular convolution.

Experiment – 3

Aim: Write a Matlab program for computation of DFT and IDFT using Direct and FFT method.

Apparatus: Matlab Software, PC

Theory:

DFT:

Discrete Fourier Transform (DFT) is used for performing frequency analysis of discrete time signals. DFT gives a discrete frequency domain representation whereas the other transforms are continuous in frequency domain. The N point DFT of discrete time signal x[n] is given by the equation

$$X(k) = \sum_{n=0}^{N-1} x[n] e^{\frac{-j2\pi kn}{N}} ; \quad k = 0,1,2,..N-1$$

The inverse DFT allows us to recover the sequence x[n] from the frequency samples

$$x[n] = \frac{1}{N} \sum_{k=0}^{N-1} x[n] e^{\frac{j2\pi kn}{N}} ; \quad n = 0,1,2,..N-1$$

FFT:

A fast Fourier transform (FFT) is an efficient algorithm to compute the discrete Fourier transform (DFT) and its inverse. FFTs are of great importance to a wide variety of applications, from digital signal processing and solving partial differential equations to algorithms for quick multiplication of large integers. Evaluating the sums of DFT directly would take O(*N* 2) arithmetical operations. An FFT is an algorithm to compute the same result in only O(*N* log *N*) operations. In general, such algorithms depend upon the factorization of *N*, but there are FFTs with O(*N* log *N*) complexity for all *N*, even for prime *N*. Since the inverse DFT is the same as the DFT, but with the opposite sign in the exponent and a 1/*N* factor, any FFT algorithm can easily be adapted for it as well.

Algorithm:

1) Get the input sequence
2) Find the DFT of the input sequence using direct equation of DFT.
3) Find the IDFT using the direct equation.
4) Find the FFT of the input sequence using MATLAB function.
5) Find the IFFT of the input sequence using MATLAB function.
4) Display the above outputs using stem function.

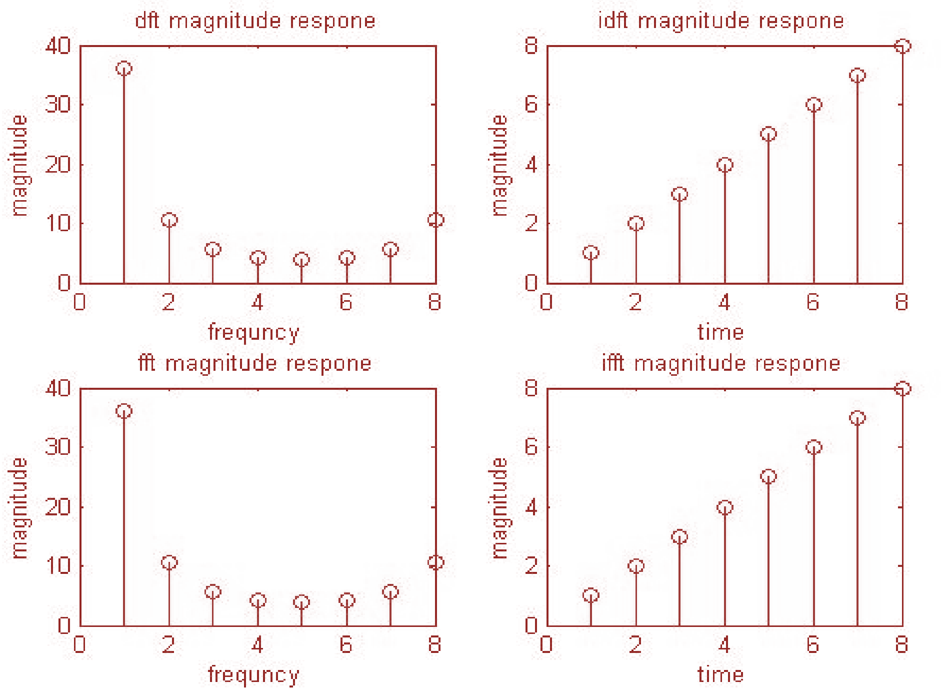
Program:

```
%********** Direct DFT ***********
clc;close all;clear all;
xn=input('enter 8 inputs');
N=length(xn);
n=0:N-1;
k=0:N-1;
wn=exp((-1i*2*pi*n'*k)/N);
xf=wn*xn';
subplot(2,2,1);
stem(abs(xf));
title('dft magnitude respone');
ylabel('magnitude');
xlabel('frequncy');
```

```
% ******* Direct IDFT **********
WN=exp((1i*2*pi*n'*k)/N);
pn=WN*xf/N;
subplot(2,2,2);
stem(abs(pn));
title('idft magnitude respone');
ylabel('magnitude');
xlabel('time');
%******* FFT Method**********
xp=fft(xn,N);
subplot(2,2,3);
stem(abs(xp));
title('fft magnitude respone');
ylabel('magnitude');
xlabel('frequncy');
%******** IFFT method *********
xw=ifft(xp,N);
subplot(2,2,4);
stem(abs(xw));
title('ifft magnitude respone');
ylabel('magnitude');
xlabel('time');
```

Results:

```
enter 8 inputs[1 2 3 4 5 6 7 8]
```



Discussions on results:

Thus from the results students will be able to

1. Discuss that the Fourier transform of a discrete time signal is also called as Signal Spectrum.
2. Discuss the changes in the results due to more number of inputs in the given sequences in finding the DFT and FFT.
3. Discuss that FFT performs faster and take less computational time compared to DFT.

Experiment – 4

Aim: Write a Matlab program to verify Sampling Theorem

Apparatus: Matlab Software, PC

Theory:

Sampling Theorem: The sampling theorem, attributed to Nyquist, Shannon, Kotelnikov and Whittaker, is useful when calculating the sampling frequency required for use in the Analog-to-Digital converter.
The theorem states that a band limited signal can be reconstructed exactly if it is sampled at a rate at least twice the maximum frequency component in it.
The maximum frequency component of g(t) is fm. To recover the signal g(t) exactly from its samples it has to be sampled at a rate fs=2fm. The minimum required sampling rate fs = 2fm is called Nyquist rate.
Sampling is also a process of converting a continuous time signal (analog signal) x(t) into a discrete time signal x[n],which is represented as a sequence of numbers. (A/D Converter)
Converting back x[n] into analog (resulting in) x(t) is the process of reconstruction.(D/A Converter)

Algorithm:

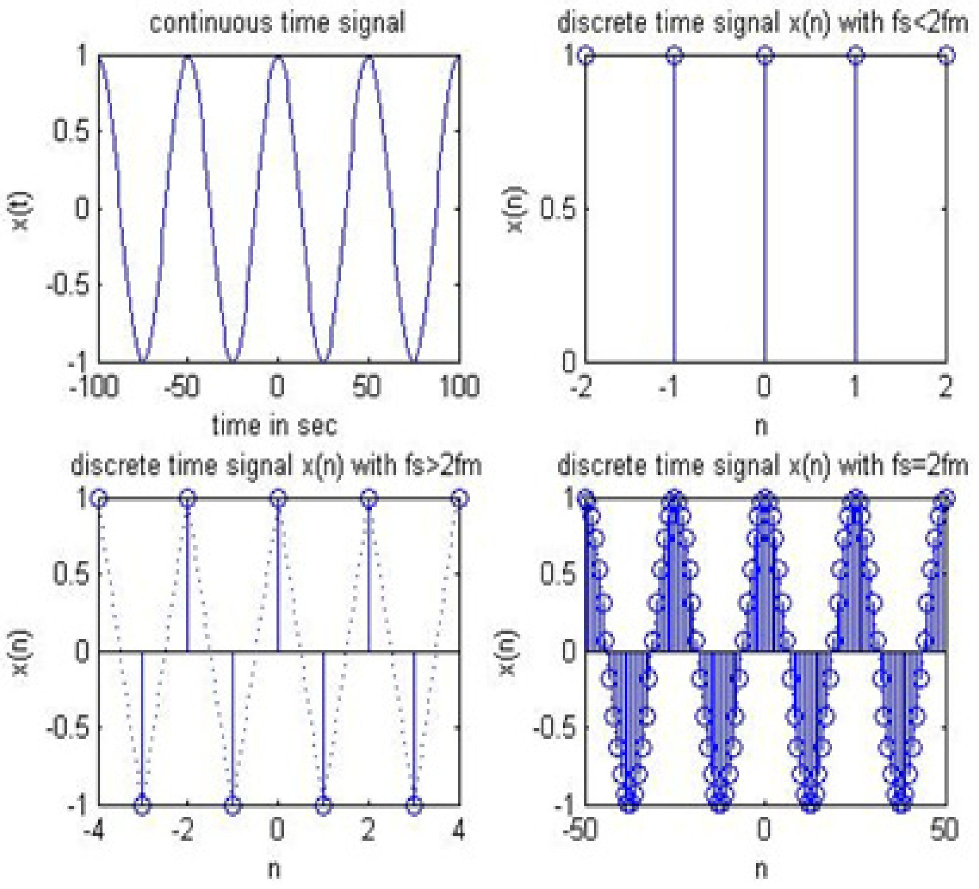
1) Input the desired frequency f_m (for which sampling theorem is to be verified)
2) Generate the cosine wave, i.e a continuous-time signal given mathematically as, $x(t) = \cos(2\pi f_m t)$ where **f** represents the frequency and *t* the time.
3) Generate the discrete-time signals for Undersampling, Nyquist sampling andoversampling conditions.
   oversampled & under sampled conditions after sampling at instants n1, n2, n3 which are given as, ,.
   a. To do this for **under sampling**, choose sampling frequency **fs1<2*fm.** For this sampling rate T1=1/fs1,
   b. For **Nyquist Sampling**, choose sampling frequency **fs2=2*fm.** For this sampling rate T2=1/fs2.
   c. For **Over Sampling**, choose sampling frequency **fs2>fd.**
4) Plot the waveforms and hence prove sampling theorem.

Program:

```
clc;
clear all;
%define analog signal for comparison
t=-100:01:100;
fm=0.02;
x=cos(2*pi*t*fm);
subplot(2,2,1);
plot(t,x);
xlabel('time in sec');
ylabel('x(t)');
title('continuous time signal');
```

```
%simulate condition for undersamplingi.e.,fs1<2*fm
fs1=0.02;
n=-2:2;
x1=cos(2*pi*fm*n/fs1);
subplot(2,2,2);
stem(n,x1);
hold on
subplot(2,2,2);
plot(n,x1,':');
title('discrete time signal x(n) with fs<2fm');
xlabel('n');
ylabel('x(n)');
%condition for Nyquist plot
fs2=0.04;
n1=-4:4;
x2=cos(2*pi*fm*n1/fs2);
subplot(2,2,3);
stem(n1,x2);
hold on
subplot(2,2,3);
plot(n1,x2,':');
title('discrete time signal x(n) with fs>2fm');
xlabel('n');
ylabel('x(n)');
%condition for oversampling
n2=-50:50;
fs3=0.5;
x3=cos(2*pi*fm*n2/fs3);
subplot(2,2,4);
stem(n2,x3);
hold on
subplot(2,2,4);
plot(n2,x3,':');
xlabel('n');
ylabel('x(n)');
title('discrete time signal x(n) with fs=2fm');
```

Results:



Discussions on results

This experiment verifies the sampling theorem in Matlab for undersampling, Nyquist sampling and oversampling.

Thus from the results students will be able to

1) Discuss the effect of undersampling for the given signal
2) Discuss the effect of Nyquist sampling for the given signal
3) Discuss the effect of oversampling for the given signal.

Experiment – 5

Aim: -To Design and generate IIR Butterworth/ Chebyshev LP/HP Filter using MATLAB

Apparatus Required: - MATLAB Software, PC

Theory:

The Digital Filter Design problem involves the determination of a set of filter coefficients to meet a set of design specifications. These specifications typically consist of the width of the passband and the corresponding gain, the width of the stopband(s) and the attenuation therein; the band edge frequencies (which give an indication of the transition band) and the peak ripple tolerable in the passband and stopband(s).

The design of IIR filters is closely related to the design of analog filters, which is a widely studied topic. An analog filter is usually designed and a transformation is carried out into the digital domain. Two transformations exist – the **impulse invariant** transformation and the **bilinear** transformation.

Analog to Digital Domain Mapping Techniques

Digital Filters are designed by using the values of both the past outputs and the present input, an operation brought about by convolution. If such a filter is subjected to an impulse then its output need not necessarily become zero. The impulse response of such a filter can be infinite in duration. Such a filter is called an *Infinite Impulse Response* filter or **IIR** filter. The infinite impulse response of such a filter implies the ability of the filter to have an infinite impulse response. This indicates that the system is prone to feedback and instability.
The experiment studies two different types of IIR filters Butterworth Filter, and Chebyschev I type Filters.
IIR filters are designed essentially by the Impulse Invariance or the Bilinear Transformation method.

1) Impulse Invariance

This procedure involves choosing the response of the digital filter as an equi-spaced sampled version of the analog filter.

1. Decide upon the desired frequency response
2. Design an appropriate analogue filter
3. Calculate the impulse response of this analogue filter
4. Sample the analogue filter's impulse response
5. Use the result as the filter coefficients

2) Bilinear Transformation:

The Bilinear Transformation method overcomes the effect of aliasing that is caused to due the analog frequency response containing components at or beyond the *Nyquist* Frequency. The bilinear transform is a method of compressing the infinite, straight analogue frequency axis to a finite one long enough to wrap around the unit circle once only. This is also sometimes called frequency warping. This introduces a distortion in the frequency. This is undone by pre-warping the critical frequencies of the analog filter (cut-off frequency, center frequency) such that when the analog filter is transformed into the digital filter, the designed digital filter will meet the desired specifications.

Filter Types

Butterworth Filters

Butterworth filters are causal in nature and of various orders, the lowest order being the best (shortest) in the time domain, and the higher orders being better in the frequency domain. Butterworth or maximally flat filters have a monotonic amplitude frequency response which is maximally flat at zero frequency response and the amplitude frequency response decreases logarithmically with increasing frequency.
A Butterworth filter is characterized by its magnitude frequency response,

$$|H(j\Omega)| = \frac{1}{\left[1+(\frac{\Omega}{\Omega_c})^{2N}\right]^{\frac{1}{2}}}$$

Where N is the order of the filter and Ωc is defined as the cutoff frequency where the filter magnitude is 1/√2 times the dc gain (Ω=0).

Chebyshev Filters

Chebyshev filters are equiripple in either the passband or stopband. Hence the magnitude response oscillates between the permitted minimum and maximum values in the band a number of times depending upon the order of filters. There are two types of chebyshev filters. The chebyshev I filter is equiripple in passband and monotonic in the stopband, where as chebyshev II is just the opposite.
The Chebyshev low-pass filter has a magnitude response given by

$$|H(j\Omega)|^2 = \left(1+\varepsilon^2 T_N^2(\frac{\Omega}{\Omega_c})\right)^{-1}$$

where ϵ is a parameter related to the ripple present in the passband
$T_N(x)$ is given by

$$C_N(x) = \begin{Bmatrix} \cos(N\cos^{-1}x)\, for\ |x| \le 1,\ passband \\ \cos(N\cosh^{-1}x)\, for\ |x| \le 1,\ stopband \end{Bmatrix}$$

The magnitude response has equiripple pass band and maximally flat stop band. By increasing the filter order N, the Chebyshev response approximates the ideal response. The phase response of the Chebyshev filter is more non-linear than the Butter worth filter for a given filter length N.

Algorithm:

1) Enter the pass band ripple (rp) and stop band ripple (rs).
2) Enter the pass band frequency (wp) and stop band frequency (ws).
3) Get the sampling frequency (fs).
4) Calculate normalized pass band frequency, and normalized stop band frequency w1 and w2 respectively.

w1 = 2 * wp /fs
w2 = 2 * ws /fs

5) Make use of the following function to calculate order of filter

Butterworth filter order

```
[n,wn]=buttord(w1,w2,rp,rs)
```

Chebyshev filter order

[n,wn]=cheb1ord(w1,w2,rp,rs)

6) Design an nth order digital lowpass Butterworth or Chebyshev filter using the following statements.

Butterworth filter

[b, a]=butter (n, wn)

Chebyshev filter

[b,a]=cheby1(n,0.5,wn)

OR

Design an nth order digital high pass Butterworth or Chebyshev filter using the following statement.

Butterworth filter

[b,a]=butter (n, wn,’high’)

Chebyshev filter

[b,a]=cheby1 (n, 0.5, wn,'high')

7) Find the digital frequency response of the filter by using ‘freqz( )’ function

[H,w]=freqz(b,a,512,fs)

8) Calculate the magnitude of the frequency response in decibels (dB)

mag=20*log10 (abs (H))

9) Plot the magnitude response [magnitude in dB Vs normalized frequency (Hz]]

10) Calculate the phase response using an = angle (H)

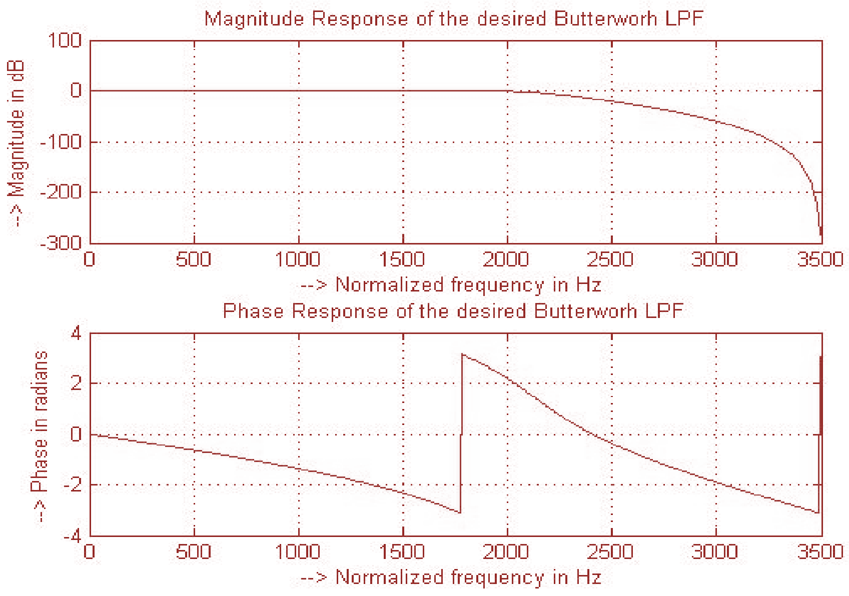
11) Plot the phase response [phase in radians Vs normalized frequency (Hz)].

Program:

```
% IIR filters
clc; clear all; close all;
warning off;
disp('enter the IIR filter design specifications');
rp=input('enter the passband ripple');
rs=input('enter the stopband ripple');
wp=input('enter the passband freq');
ws=input('enter the stopband freq');
fs=input('enter the sampling freq');
w1=2*wp/fs;%normalized pass band frequency
w2=2*ws/fs;%normalized stop band frequency
[n,wn]=buttord(w1,w2,rp,rs);% Find the order n and cut-
off frequency
ch=input('give type of filter 1:LPF,2:HPF');
switch ch
case 1
 disp('Frequency response of Butterworth IIR LPF is:');
 [b,a]=butter(n,wn); % Find the filter coefficient of LPF
 [H,w]=freqz(b,a,512,fs);% to get the transfer function
of the filter
 mag=20*log10(abs(H));
 phase=angle(H);
 subplot(211);
 plot(w,mag);grid on;
 ylabel('--> Magnitude in dB');
 xlabel('--> Normalized frequency in Hz');
 title('Magnitude Response of the desired Butterworh
LPF');

subplot(212);

plot(w,phase);grid on;
 ylabel('--> Phase in radians');
 xlabel('--> Normalized frequency in Hz');
 title('Phase Response of the desired Butterworh LPF');
case 2
 disp('Frequency response of IIR Butterworth HPF is:');
 [b,a]=butter(n,wn,'high'); % Find the filter co-
efficients of HPF
 [H,w]=freqz(b,a,512,fs);% to get the transfer function
of the filter
 mag=20*log10(abs(H));
 phase=angle(H);
 subplot(211);
 plot(w,mag);grid on;
 ylabel('--> Magnitude in dB');
 xlabel('--> Normalized frequency in Hz');
 title('Magnitude Response of the desired Butterworh
HPF');
 subplot(212);
 plot(w,phase);grid on;
 ylabel('--> Phase in radians');
 xlabel('--> Normalized frequency in Hz');
 title('Phase Response of the desired Butterworh HPF');
end
```

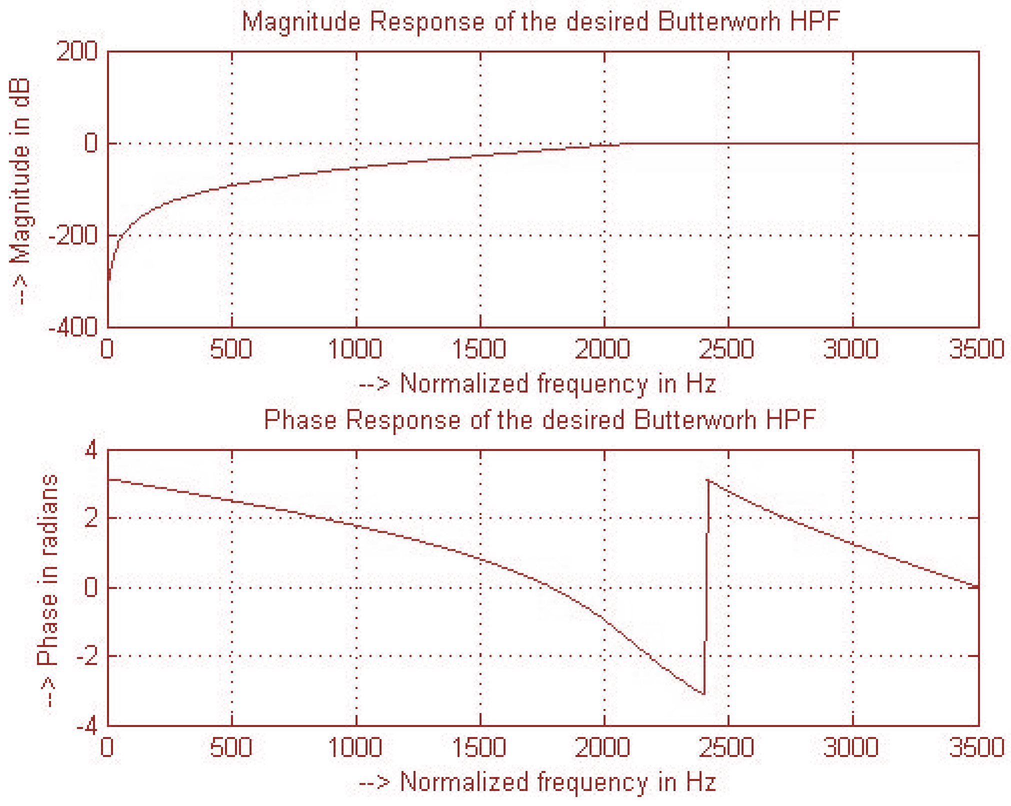
Results:

```
enter the IIR filter design specifications
enter the passband ripple   0.15
enter the stopband ripple    60
enter the passband freq     1500
enter the stopband freq     3000
enter the sampling freq     7000
give type of filter 1:LPF,2:HPF
1
Frequency response of Butterworth IIR LPF is:
```



IIR HIGH PASS FILTER

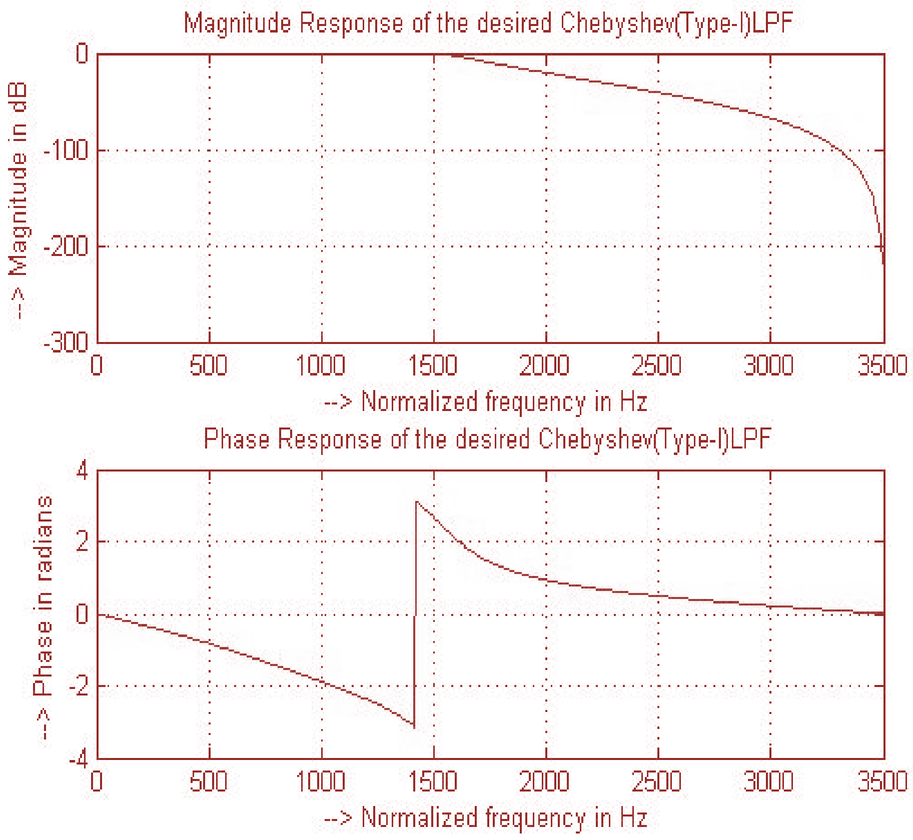
```
enter the IIR filter design specifications
enter the passband ripple   0.15
enter the stopband ripple    60
enter the passband freq     1500
enter the stopband freq     3000
enter the sampling freq     7000
give type of filter 1:LPF,2:HPF
2
Frequency response of Butterworth IIR HPF is:
```



```
%To design a Chebyshev (Type-I) Low/High Pass Filter for the
given specifications
clc; clear all; close all;
disp('enter the IIR filter design specifications');
rp=input('enter the passband ripple');
rs=input('enter the stopband ripple');
wp=input('enter the passband freq');
ws=input('enter the stopband freq');
fs=input('enter the sampling freq');

w1=2*wp/fs;%to get normalized pass band frequency
w2=2*ws/fs;% to get normalized stop band frequency

ch=input('give type of filter 1:LPF,2:HPF');
 % to get the order and cut-off frequency of the filter
[n,wn]=cheb1ord(w1,w2,rp,rs);
switch ch
 case 1

disp('Frequency response of Chebyshev IIR LPF is:');
  [b,a]=cheby1(n,0.5,wn);% to get the filter coefficients
  % to get the transfer function of the filter
  [H,w]=freqz(b,a,512,fs);
  mag=20*log10(abs(H));
  phase=angle(H);
  subplot(211);
  plot(w,mag);grid on;
  ylabel('--> Magnitude in dB');
  xlabel('--> Normalized frequency in Hz');
  title('Magnitude Response of the desired Chebyshev Type -I)
LPF');
  subplot(212);
  plot(w,phase);grid on;
  ylabel('--> Phase in radians');
  xlabel('--> Normalized frequency in Hz');
  title('Phase Response of the desired Chebyshev(Type-I)LPF');
 case 2
  disp('Frequency response of Chebyshev IIR HPF is:');
  % to get the filter coefficients
  [b,a]=cheby1(n,0.5,wn,'high');
  % to get the transfer function of the filter
  [H,w]=freqz(b,a,512,fs);
  mag=20*log10(abs(H));
  phase=angle(H);
  subplot(211);
  plot(w,mag);grid on;
  ylabel('--> Magnitude in dB');
  xlabel('--> Normalized frequency in Hz');
  title('Magnitude Response of the desired Chebyshev(Type-
I)HPF');
  subplot(212);
  plot(w,phase);grid on;
  ylabel('--> Phase in radians');

  xlabel('--> Normalized frequency in Hz');
  title('Phase Response of the desired Chebyshev(Type-I)HPF');
end
```

Results:

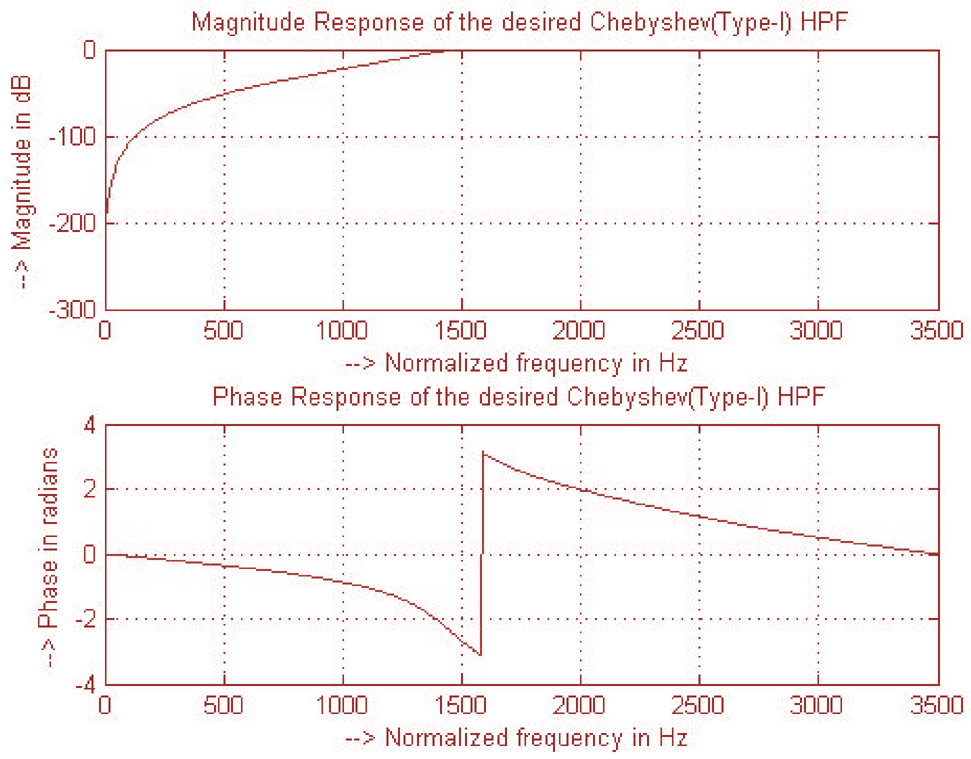
```
enter the IIR filter design specifications
enter the passband ripple     0.15
enter the stopband ripple     60
enter the passband freq       1500
enter the stopband freq       3000
enter the sampling freq       7000
give type of filter 1:LPF,2:HPF
1

Frequency response of Chebyshev IIR LPF is:
```



High Pass Filter

Result:

```
enter the IIR filter design specifications
enter the passband ripple     0.15
enter the stopband ripple     60
enter the passband freq       1500
enter the stopband freq       3000
enter the sampling freq       7000
give type of filter 1:LPF,2:HPF
2
Frequency response of Chebyshev IIR HPF is:
```



Discussions on results:

By this experiment we have studied the LP/HP IIR digital filter designing.

From the obtained results the students will be able to

1) Discuss the effect of order of the filer on magnitude response.
2) Discuss the effect of variation in pass band ripple, stop band ripple, pass band frequency, stop band frequency and sampling frequency respectively in designing the IIR Butterworth digital filter.
3) Discuss the effect of variation in pass band ripple, stop band ripple, pass band frequency, stop band frequency and sampling frequency respectively in designing the IIR Chebyshev digital filter.

Experiment – 6

Aim: - Design and implementation of FIR Filter (LP/HP) to meet given specifications Using Windowing technique

a. Rectangular window
b. Hamming window
c. Kaiser window

Apparatus: Matlab Software, PC

Theory:

A linear-phase is required throughout the passband of the filter to preserve the shape of the given signal in the passband. A causal IIR filter cannot give linear-phase characteristics and only special types of FIR filters that exhibit center symmetry in its impulse response give the linear-space. A Finite Impulse Response (FIR) filter is a discrete linear time-invariant system whose output is based on the weighted summation of a finite number of past inputs.
A zero-phase frequency response of an ideal filter is given as

$$H_{LP}(e^{j\omega}) = \begin{cases} 1, |\omega| \le \omega_c, \\ 0, \omega_c < \omega \le \pi. \end{cases}$$

Hence time domain impulse response is

$$h_d[k] = \frac{1}{2\pi}\int_{-\pi}^{\pi} H_d\left(e^{j\omega}\right)e^{j\omega k}\,d\omega = .. = \alpha\frac{\sin(\omega_c k)}{\omega_c k}$$

so the impulse response is doubly infinite, not absolutely summable, and therefore unrealizable.
By setting all impulse response coefficient outside the range $-M \le n \le M$ equal to zero, we arrival at a finite-length noncausal approximation of length $N = 2M + 1$ which when shifted to the right yield the coefficients of a causal FIR lowpass filter:

$$h_{LP}\left[n\right] = \begin{cases} \dfrac{\sin(\omega_c(n-M))}{\pi(n-M)} & ,0 \le n \le N-1 \\ 0, otherwise \end{cases}$$

Gibbs phenomenon

The causal FIR filter obtained by simply truncating the impulse response coefficients of the ideal filters exhibit an oscillatory behavior in their respective magnitude responses which is more commonly referred to as the Gibbs phenomenon
Cause of Gibbs phenomenon:
The FIR filter obtained by truncation can be expressed as

$$h[n] = h_d[n] \cdot \omega[n]$$

$$H(e^{j\omega}) = \frac{1}{2\pi}\int_{-\pi}^{\pi} H_d(e^{j\phi})\psi(e^{j(\omega-\phi)})d\phi$$

The window used to achieve simple truncation of the ideal filter is rectangular window

$$w_R[n] = \begin{cases} 1, 0 \le |n| \le M \\ 0, otherwise \end{cases}$$

Thus by applying windowing functions we can obtain FIR filters.
Available Fixed window functions Rectangular, Bartlett, Hamming, Hanning, Blackmann etc.
Hamming window function

$$w[n] = 0.54 + 0.46\cos(\frac{2\pi n}{2M+1}), -M \le n \le M$$

In Adjustable Window Functions, windows have been developed that provide control over ripple by means of an additional parameter.
Like Kaiser Window

$$w[n] = \frac{I_0\left\{\beta\sqrt{1-(n/M)^2}\right\}}{I_0(\beta)}, -M \le n \le M$$

Where β is an adjustable parameter and $I_0(\beta)$ is a zero order Bessel function
To design a FIR filter order of the filter should be specified or can be calculated from the following equation

$$N = \frac{-20\log_{10}\left(\sqrt{r_p r_s}\right) - 13}{14.6\left(w_s - w_p\right)/2\pi}$$

rp=passband ripple, rs=stopband ripple, wp=passband frequency
ws=stopband frequency
Then from order of the filter we can find the length by which a window function can be applied.

Algorithm:

FIR Low Pass Filter design

1) Enter the pass band ripple (rp) and stop band ripple (rs).
2) Enter the pass band frequency (wp) and stop band frequency (ws).
3) Get the sampling frequency (fs), beta value for Kaiser window.
4) Calculate the analog pass band edge frequencies, w1 and w2.
   i. w1 = 2*wp/fs
   ii. w2 = 2*ws/fs
5) Calculate the order of the filter using the order equation.
6) Use switch condition and ask the user to choose either Rectangular Window or Hamming window or Kaiser window.

7) Use rectwin, hamming, Kaiser Commands
   Command **fir1** uses the window method of FIR filter design, If w(n) denotes a window, where 1 ≤ n ≤ N, and the impulse response of the ideal filter is h(n), where $h_d(n)$ is the inverse Fourier transform of the ideal frequency response.
   $$h[n] = h_d[n] \cdot \omega[n]$$
8) Calculate the digital frequency response using the command 'freqz()'
9) Calculate the magnitude of the frequency response in decibels
   m=20*log10 (abs(h))
10) Plot the magnitude response [magnitude in dB Vs normalized frequency (om/pi)]

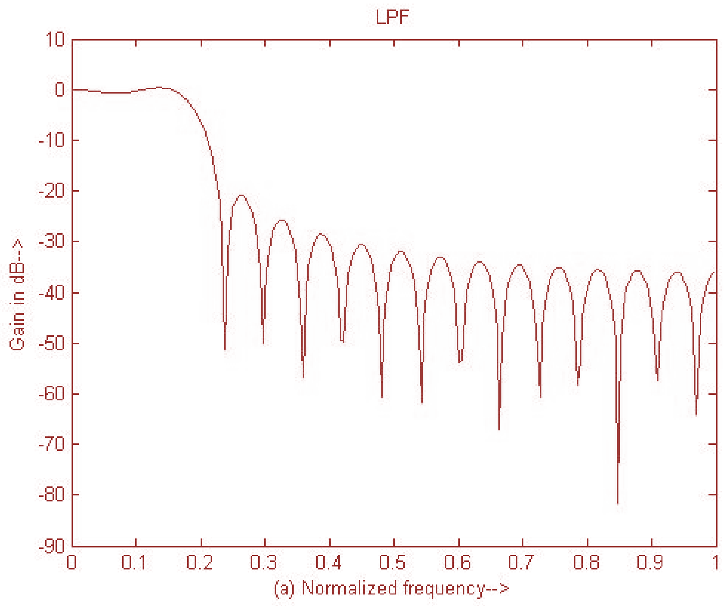
Program:

```
%FIR Low Pass/High pass filter design using
Rectangular/Hamming/Kaiser window
clc; clear all; close all;
rp=input('enter passband ripple');
rs=input('enter the stopband ripple');
wp=input('enter passband freq');
ws=input('enter stopband freq');
fs=input('enter sampling freq ');
beta=input('enter beta value');
w1=2*wp/fs;
w2=2*ws/fs;
num=-20*log10(sqrt(rp*rs))-13;
dem=14.6*(ws-wp)/fs;
n=ceil(num/dem);
n1=n+1;
if(rem(n,2)~=0)
    n1=n; n=n-1;
end
c=input('enter your choice of window function 1. rectangular
2. Hamming 3.kaiser: \n ');
if(c==1)
    y=rectwin(n1);
    disp('Rectangular window filter response');
end
if (c==2)
    y=hamming(n1);
    disp('Hamming window filter response');
end
if(c==3)
    y=kaiser(n1,beta);
    disp('kaiser window filter response');
end

ch=input('give type of filter 1:LPF,2:HPF');
switch ch
    case 1
      b=fir1(n,w1,y);
       [h,o]=freqz(b,1,256);
      m=20*log10(abs(h));
      plot(o/pi,m);
      title('LPF');
      xlabel('(a) Normalized frequency-->');
      ylabel('Gain in dB-->');

    case 2
      b=fir1(n,w1,'high',y);
       [h,o]=freqz(b,1,256);
      m=20*log10(abs(h));
      plot(o/pi,m);
      title('HPF');
      xlabel('(b) Normalized frequency-->');
      ylabel('Gain in dB-->');
end
```

Results:

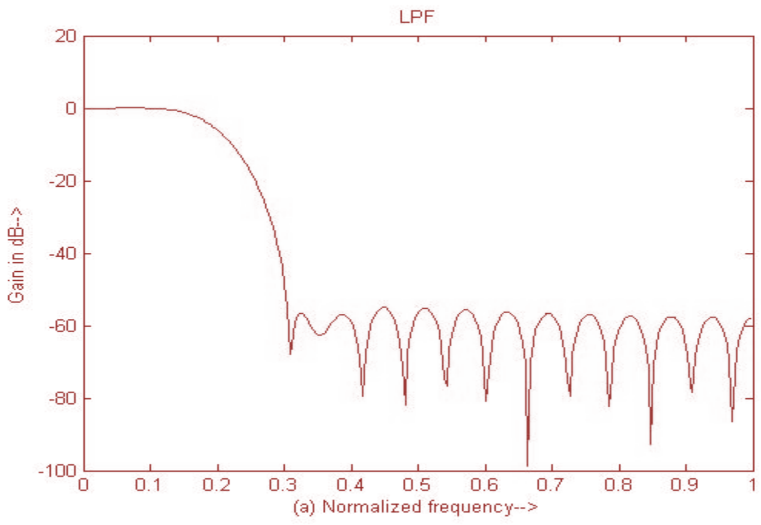
```
enter passband ripple         0.02
enter the stopband ripple     0.01
enter passband freq           1000
enter stopband freq           1500
enter sampling freq           10000
enter beta value                 5
enter your choice of window function 1. rectangular 2.
Hamming 3.kaiser:
 1
Rectangular window filter response
give type of filter 1:LPF,2:HPF
1:LPF
```

Low pass FIR filter using Rectangular Window



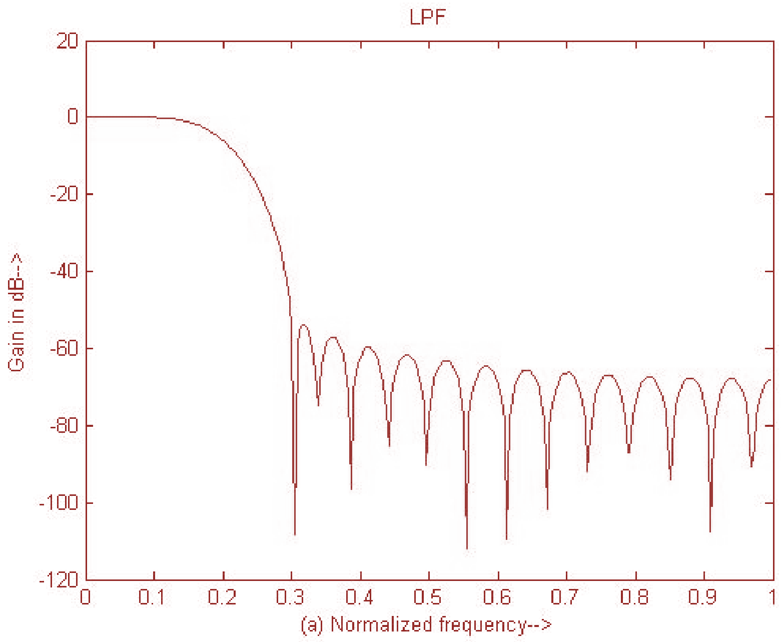
```
enter your choice of window function 1. rectangular 2.
Hamming 3.kaiser:
 2
Hamming window filter response
give type of filter 1:LPF,2:HPF
1:LPF
```

Low pass FIR filter using Hamming Window



```
enter your choice of window function 1. rectangular 2.
Hamming 3.kaiser:
 3
kaiser window filter response
give type of filter 1:LPF,2:HPF
1:LPF
```

Low pass FIR filter using Kaiser Window

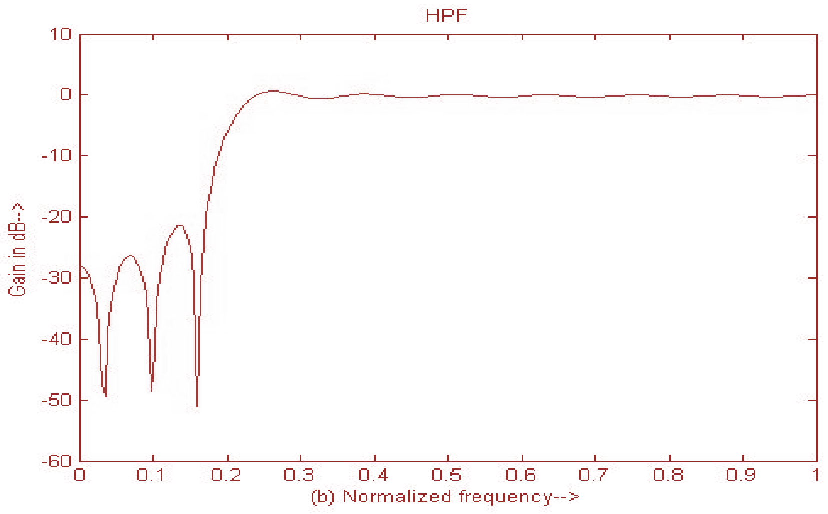


FIR High pass Filter design

Results:

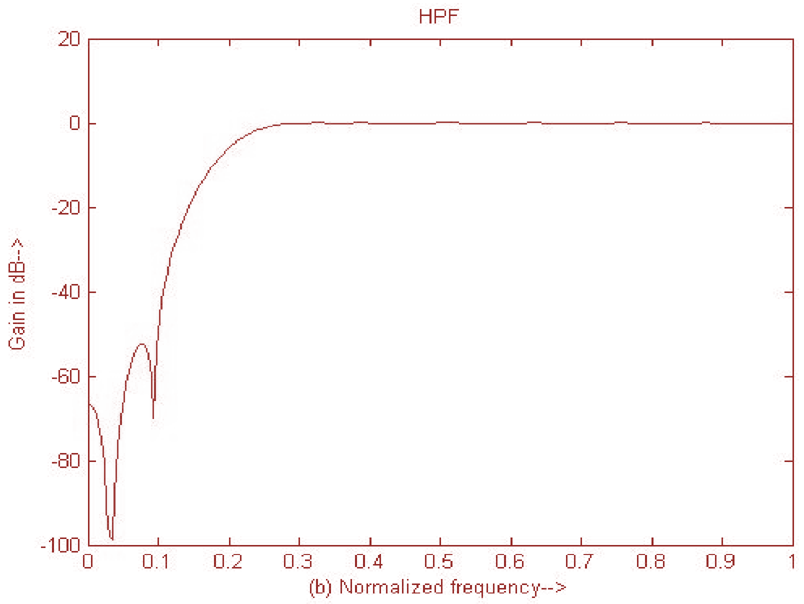
```
enter passband ripple         0.02
enter the stopband ripple     0.01
enter passband freq           1000
enter stopband freq           1500
enter sampling freq           10000
enter beta value                 5
enter your choice of window function 1. rectangular 2.
Hamming 3.kaiser:
 1
Rectangular window filter response
give type of filter 1:LPF,2:HPF
2:HPF
```

High pass FIR filter using Rectangular Window



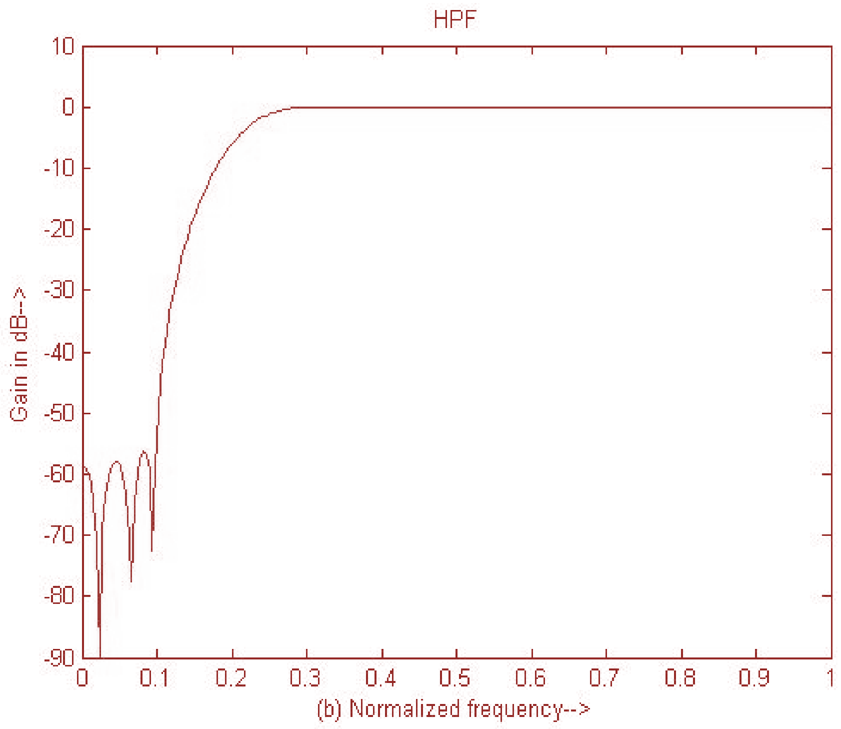
```
enter your choice of window function 1. rectangular 2.
Hamming 3.kaiser:
2
Hamming window filter response
give type of filter 1:LPF,2:HPF
2:HPF
```

High pass FIR filter using Hamming Window



```
enter your choice of window function 1. rectangular 2.
Hamming 3.kaiser:
 3
kaiser window filter response
give type of filter 1:LPF,2:HPF
2: HPF
```

High pass FIR filter using Kaiser Window



Discussions on results:

Thus FIR digital filter designing is experimented using Matlab software.

Thus from the results students will be able to

1) Discuss the effect of order of the filer on magnitude response.
2) Discuss the effect of variation in pass band ripple, stop band ripple, pass band frequency, stop band frequency and sampling frequency respectively in designing the FIR digital filter.
3) Discuss the difference between the Rectangular, Hamming and Kaiser Window functions.
4) Discuss the performance of FIR digital filter designed using Kaiser window over FIR digital filter designed with Rectangular and Hamming window functions.

Cycle-II

TMS320C50 Architecture Overview

1. Introduction:

It is needless to say that in order to utilize the full feature of the DSP chip TMS320C50, the DSP engineer must have a complete knowledge of the DSP device. This chapter is an introduction to the hardware aspects of the TMS320C50. The important units of TMS320C50 are discussed.

2. The DSP Chip TMS320C50:

The TMS320C50 is a 16-bit fixed point digital signal processor that combines the flexibility of a high speed controller with the numerical capability of an array processor, thereby offering an inexpensive alternative to multichip bit-slice processors. The highly paralleled architecture and efficient instruction set provide speed and flexibility capable of executing 10 MIPS (Million Instructions per Second). The TMS320C50 optimizes speed by implementing functions in hardware that other processors implement through microcode or software. This hardware intensive approach provides the design engineer with processing power previously unavailable on a single chip.

The TMS320C50 is the third generation digit l signal processor in the TMS320 family. Its powerful instruction set, inherent flexibility, high-speed number-crunching capabilities, and innovative architecture have made this high-performance, cost-effective processor the ideal solution to many telecommunications, computer, commercial, industrial, and military applications.

3. Key Features of TMS320C50:

The key features of the Digital Signal Processor TMS320C50 are:

* 35-/50-ns single-cycle fixed-point instruction execution time (28.6/20 MIPS)
* Upward source-code compatible with all `C1X and `C2x devices
* RAM-based memory operation (`C50)
* 9K x 16-bit single-cycle on-chip program/data RAM (`C50)
* 2K x 16-bit single-cycle on-chip boot ROM (`C50)
* 1056 x 16-bit dual-access on-chip data RAM
* 224K x 16-bit maximum addressable external memory space (64K program, 64K data, 64K I/O, and 32K global)
* 32-bit arithmetic logic unit (ALU), 32-bit accumulator (ACC), and 32-bit accumulator buffer (ACCB)
* 16-bit parallel logic unit (PLU)
* 16 x 16-bit parallel multiplier with a 32-bit product capability.
* Single-cycle multiply/accumulate instructions
* Eight auxiliary registers with a dedicated auxiliary register arithmetic unit for indirect addressing.
* Eleven context-switch registers (shadow registers) for storing strategic CPU controlled registers during an interrupt service routine
* Eight-level hardware stack
* 0- to 16-bit left and right data barrel-shifters and a 64-bit incremental data shifter
* Two indirectly addressed circular buffers for circular addressing
* Single-instruction repeat and block repeat operations for program code
* Block memory move instructions for better program/data management
* Full-duplex synchronous serial port for direct communication between the `C5x and another serial device
* Time-division multiple-access (TDM) serial port

* Interval timer with period, control, and counter registers for software stop, start, and reset
* 64K parallel I/O ports, 16 of which are memory mapped
* Sixteen software programmable wait-state generators for program, data, and I/O memory spaces.

4. Architecture:

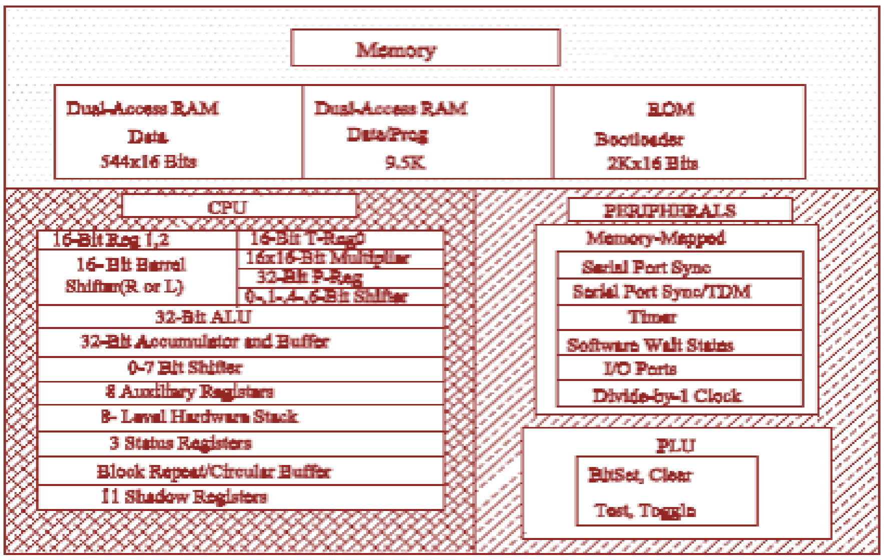
A detailed architectural block diagram of TMS320C50 is illustrated in Figure 1-1. The TMS320C50 utilizes a modified Harvard architecture for speed and flexibility. In a strict Harvard architecture, program and data memory are in two separate spaces, permitting a full overlap of instruction fetch and execution. The TMS320 family's modification of the Harvard architecture allows transfer between program and data spaces, thereby increasing the flexibility of the device. This modification permits coefficients stored in program memory to be read into the data RAM, eliminating the need for a separate coefficient ROM. It also makes available immediate instructions and subroutines based on computed values

.

4.1 32-Bit Accumulator:

The TMS320C50 contains a 32-bit ALU and accumulator for support of double-precision, two's complement arithmetic. The ALU is a general purpose arithmetic unit that operates on 16-bit words taken from the data RAM or derived from immediate instructions. In addition to the usual arithmetic instructions, the ALU can perform Boolean operations, providing the bit manipulation ability required of a high-speed controller. The accumulator stores the output from the ALU and is often an input to the ALU. Its word length is 32-bit. The accumulator is divided into a highorder
word (bits 31 through 16) and a low-order word (bits 15 through 0). Instructions are provided for storing and loading the high and lower order accumulator words to memory.

TMS320C50 BLOCK DIAGRAM



4.2 16 x 16-Bit Parallel Multiplier:

The multiplier performs a 16 x 16-bit two's complement multiplication with a 32-bit result in a single instruction cycle. The multiplier consists of three units: the T-

Register, P-Register, and multiplier array. The 16-bit T-Register temporarily stores the multiplicand and the P-Register stores the 32-bit product. Multiplier values either

come from the data memory or are derived immediately from the MPY (multiply immediate) instruction word. The fast on-chip multiplier allows the device to perform fundamental operations such as convolution, correlation, and filtering. Two multiply/accumulate instructions in the instruction set fully utilize the computational bandwidth of the multiplier, allowing both operands to be processed simultaneously.

4.3 Shifters:

A 16-bit scaling shifter is available at the accumulator input. This shifter produces a left shift of 0 to 16-bits on the input data to accumulator.
TMS320C50 also contains a shifter at the accumulator output. This shifter provides a left shift of 0 to 7, on the data from either the ACCH or ACCL register, right, before transferring the product to accumulator.

4.4 Date and Program Memory:

Since the TMS320C50 uses Harvard architecture, data and program memory reside in two separate spaces. Additionally TMS320C50 has one more memory space called I/O memory space. The total memory capacity of TMS320C50 is 64KW each of Program, Data and I/O memory. The 64KW of data memory is divided into 512 pages with each page containing 128 words. Only one page can be active at a time. One data page selection is done by setting data page pointer. TMS320C50 has 1056 words of dual access on chip data RAM and 9K words of single access Data/Program RAM. The 1056 words of on chip data memory is divided as three blocks B0, B1 & B2, of which B0 can be configured as program or data RAM.
Out of the 64KW of total program memory, TMS320C50 has 2K words of on-chip program ROM.

The TMS320C50 offers two modes of operation defined by the state of the MC/MP pin: the microcomputer mode (MC/MP = 1) or the microprocessor mode (MC/MP = 0). In the microcomputer mode, on-chip ROM is mapped into the memory space with upto 2K words of memory available. In the microprocessor mode all 64K words of program memory are external.

4.5 Interrupts and Subroutines:

The TMS320C50 has three external maskable user interrupts available for external devices that interrupt the processor.
The TMS320C50 contains an eight-level hardware stack for saving the contents of the program counter during interrupts and subroutine calls. Instructions are available for saving the device's complete context. PUSH and POP instructions permit a level of nesting restricted only by the amount of available RAM.

4.6 Serial Port:

A full-duplex on-chip serial port provides direct communication with serial devices such as codecs, serial A/D converters and other serial systems. The interface signals are compatible with codecs and many others serial devices with a minimum of external hardware.

4.7 Input and Output:

The 16-bit parallel data bus can be utilised to perform I/O functions in two cycles. The I/O ports are addressed by the four LSBs on the address lines, allowing 16 input and 16 output ports. In addition, polling input for bit test and jump operations (BIO) and three interrupt pins (INT0 - INT2) have been incorporated for multitasking.

Software Overview

This chapter illustrates the use of program and execution mainly in the **standalone mode**.
The Micro-50 EB has 3 software development tools namely

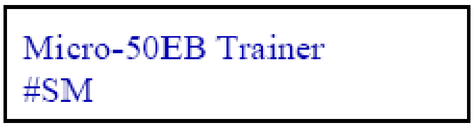
1. Standalone Mode
2. Monitor program
3. Serial Mode

In "Standalone Mode" the **Micro-50 EB** works with a 104 keys keyboard and 16x2 LCD display and line assembler. With this configuration, the student can enter his program through the keyboard and edit and display it on the LCD display. The user can enter the Mnemonics using the Line Assembler, and debug the program to run it on **Micro-50 EB.**
"Monitor Program" is used to enter data directly into Data or Program memory, display the data etc. It has several commands to enter the user program, for editing and debugging.
In "Serial Mode", it works with a IBM PC computer and program entry and debugging is done at the PC level. The Chapter-6 gives details about this operation.

Program and Execution:

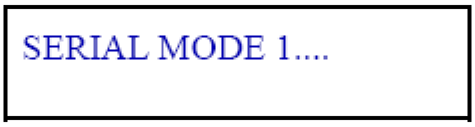
1. Serial Monitor Mode:

Connect the serial monitor cable from Serial port of Micro-50 EB kit to the serial port COM1 or COM2 of PC XT/AT (prefer COM1 for default selection). Now execute communication software (XTALK.EXE) in PC.
Power on the Micro-50 EB Kit with all its set up ready and enter the following command at the prompt.



```
Micro-50EB Trainer
#SM
```

Press the enter key to enter into serial monitor mode and the screen displays the followingbMessage



```
SERIAL MODE 1....
```

Now the following menu will appear on the monitor.

```
Micro-50 EB Serial Monitor, Ver.1.0
(C) Copyright 1996 by Vi Microsystems (P) Ltd. Chennai.
#
```

Now enter "HE" to view the help menu of the serial monitor.
To assemble the program given in the example enter "AS" at the prompt and press the Enter Key, the screen displays the following message.

#Micro-50 EB Line Assembler, Version 2.0
Enter Address:

Now enter the program memory starting address "C000H" and press Enter Key. Now the screen display next consequent message as

C000H

Enter the mnemonics of the program sequentially viewing opcodes of the respective mnemonics after pressing enter key.
On completion of assembling enter dot (.) and press the enter key to come out to prompt.
To execute the program use the command "GO C000" and press the Enter Key, where C000 is program memory starting address.
To abort the execution, press the reset switch once to reset the Micro-50 Kit. Now enter "SM" in Micro-50 EB trainer to re-enter into serial mode. To verify the execution, dump the data memory from 8000H to 9000H using "DD" command. This operation is same as Line assembler in standalone mode.

Note:

1) At any occasion to abort the serial monitor from the execution of the program press the reset switch of Micro-50 EB kit.
2) To quit of serial monitor mode enter "QU" at the prompt and press enter key.

Experiment – 7

Aim: To perform 16 bit addition and Integer Multiplication using DSP Trainer Kit (TMS 320C50PQ57).

Apparatus: PC, DSP Trainer Kit (TMS 320C50PQ57).

Theory:

This experiment mainly described about how to construct the TMS320C50 assembly program using line assembler in the Monitor mode for Arithmetic manipulations and usage of various instruction of TMS320C50 processor. Examples of Arithmetic manipulations and usage of Instruction of TMS320C50 are given here. Numerical analysis, floating point computations or other operations may require arithmetic to be executed with more than 32-bits of precision. Since, the TMS320C50 is 16-bit fixed-point processor; software is required for the extended precision of arithmetic operations. For proper operation, the overflow mode bit should be reset (OVM = 0) so that the accumulator result will not be loaded with the saturation value.

Algorithm:

Addition:

In this example, two 16-bit numbers will be added as shown below:

```
        X0
+       Y0
   ----------------
        W0
```

Where all the numbers X0 & Y0 are 16-bit numbers. TMS320C50 has provision to add two 16-bit numbers.
First load X0 in higher accumulator using LACC instruction and add Y0 with accumulator by using ADD instruction.

Integer Multiplication:

The TMS320C50 hardware multiplier normally performs two's complement 16-bit by 16-bit multiplies and produces a 32-bit result in a single processor cycle. To multiply two operands, one operand must be loaded into the T-register. The second operand is moved by the multiply instruction to the multiplier, which then produces the product in the P-register. Before another
multiply can be performed, the content of the P-register should be moved to the accumulator or should be stored in data memory.

The following program stores the data 37A at data memory 8000 and the data 12E at the data memory address 8001. The instruction LT OP1 (0000h) loads the first operand from dma 8000 to T register. The instruction MPY OP2 (0001h) multiplies the content of dma 8001 with the T register. The result is stored in P register. The instruction PAC transfers the content of P register to accumulator. The SACL and SACH instructions store the lower and higher order results in dma 8002 and 8003.

Procedure:

1) Open the C50 Debugger software from the desktop.

2) Click the serial port settings and select auto detect.

Serial>Port Settings>AutoDetect

3) It won't detect at first attempt then reset the kit using reset button and Type **SM** using kit's keyboard and press Enter.

4) Again Select auto detect then it will detect.

5) Select New Project in C50 Debugger Software, give name to project.

6) Create New Assembly File from the File Tab of C50 Debugger.
File>New>Assembly File

7) Type the Program in the above file and save it as **filename.asm**
*Select filename as Project Name

8) Go to Project Folder and right click on Assembly folder and add the filename.asm to it.

9) Right Click on the CMD Files folder and add Micro50.cmd to it.

10) Save the Project, Build the Project.

11) Load the Program using Serial Tab.
Load Program>Browse Corresponding ascii file name whose name is same as name of program.

12) Reset the Kit and Type **SM** on kit using Kit's Keyboard.

13) Now access the Program Memory and Data Memory by using Communication Window.
Select **Serial>Communication Window**

14) After successfully downloading read the Message.
PI C000-Downloading from Host in Communication Window.

15) Type **SD 8000** and press enter then enter the Inputs.
Ex: SD 8000:1234-3333 where 1234 is old data and 3333 is new data.

16) For two values press enter 3 times.

17) Then type dot. to come out.

18) Type **GO C000** and press enter. See the message **Executing….**

19) Reset the Kit and Type **SM** using the Kits Keyboard.

20) Type **SD 8002**, press enter and observe the outputs.

16-Bit Addition

Program:

Addition

```
                  .MMREGS
                  .TEXT
        C000       START: LDP  #100H
        C001                   LACC 0000H  ;ACC=X0
        C002                   ADD 0001    ;ACC=X1 +X0

        C003                   SACL 02H    ;ACCL =W0
        C004                   SACH 03H    ;ACCH=W1

        C005        HERE:    B   HERE
```

Results:

```
                    Communication Window
#SD 8000
Substitute data 8000:0003-0004
Substitute data 8001:0004-0002
Substitute data 8002:0007

# G0 C000
Executing….
Micro-50 pt Serial Monitor,Ver-2.0

#SD 8000
Substitute data 8000:0004-
Substitute data 8001:0002-
Substitute data 8002:0006
```

Interger Multiplication

```
ADDRESS    MNEMONICS
                .MMREGS
                .TEXT
C000  START: LDP #100H
C001          LACC #037AH,0
C002          037A
C003          SACL 0000, 0
C004          LACC #012EH, 0
C005          012E
C006          SACL 0001,0
C007          LT 0000      ; T=37A
C008          MPY 0001   ; P=37A*12E=0004 19EC
C009          PAC          ; ACC=0004 19EC
C00A          SACL 0002, 0
C00B          SACH 0003, 0
C00C    HERE  :  B HERE
```

Results:

```
#SD 8000
Substitute data 8000:0003-037A
Substitute data 8001:0004-012E
Substitute data 8002:0007

# G0 C000
Executing….
Micro-50 pt Serial Monitor,Ver-2.0

#SD 8000
Substitute data 8000:037A-
Substitute data 8001:012E-
Substitute data 8002:19EC-
Substitute data 8003:0004-
```

Discussions on results:

Thus using DSP Trainer Kit (TMS 320C50PQ57) we have performed the 16 bit addition and multiplication. Thus from the results students will be able to

1) Discuss the different integer arithmetic operations performed using the DSP Trainer Kit (TMS 320C50PQ57).
2) Discuss the usage of DSP Trainer Kit (TMS 320C50PQ57) for floating point arithmetic operations.

Experiment – 8

Aim: To generate Triangular, and Square Waveforms using DSP Trainer Kit (TMS 320C50PQ57).

Apparatus: PC, DSP Trainer Kit (TMS 320C50PQ57).

Theory:

The SPLK instruction allows a full 16-bit pattern to be written into any memory location. The parallel logic unit (PLU) supports this bit manipulation independently of the ALU so that the ACC is un affected. In this programme, LDP instruction is used to load data memory pointer from 8000.

Algorithm:

Triangular Waveform:

1) Load data pointer with the current page number in which we want to store local variable.
2) Use SPLK to load the value of the temporary register.
3) Under the label CONT1, load the auxiliary register ar2 with the contents of frequency.
4) Under the label CONT1 initialize count as 04h.
5) Load accumulator register with temporary register value.
6) Add it with value of amplitude.
7) And store back it to temporary register.
8) Check current auxiliary register value for zero, if not equal to zero go back to label COUNT, otherwise go to next instruction.
9) Again load the auxiliary register ar2 with the contents of frequency.
10) Again out count as 04h.
11) Load accumulator register with temporary register value.
12) Subtract it with value of amplitude.
13) And again store back it to temporary register.
14) Multiply the auxiliary register with the temp register value.
15) Check current auxiliary register value for zero, if not equal to zero go back to label COUNTX, otherwise go to next instruction.
16) Go back to label COUNT1 to repeat the algorithm continuously.

Square Waveform:

1) Load data pointer with the current page number in which we want to store local variable.

2) Use SPLK to load the value of the temporary register with value #7FFFH
3) Output the valued through DAC port.
4) Call DELAY label.
5) Use SPLK to load the value of the another temporary register with value #0FFFH
6) Output the valued through DAC port.
7) Again Call DELAY label.
8) In label DELAY load the Auxiliary Register AR3 with a value FF.
9) Multiply this auxiliary register AR3 value with the same value
10) In label BACK decrement the resultant value of above multiplication by one and check for not equal to zero. Repeat this procedure till the result goes to zero, and then return from the loop.

Procedure:

1) Open the C50 Debugger software from the desktop.

2) Click the serial port settings and select auto detect.
Serial>Port Settings>AutoDetect

3) It won't detect at first attempt then reset the kit using reset button and Type **SM** using kit's keyboard and press Enter.

4) Again Select auto detect then it will detect.

5) Select New Project in C50 Debugger Software, give name to project.

6) Create New Assembly File from the File Tab of C50 Debugger.
File>New>Assembly File

7) Type the Program in the above file and save it as **filename.asm**
*Select filename as Project Name

8) Go to Project Folder and right click on Assembly folder and add the filename.asm to it.

9) Right Click on the CMD Files folder and add Micro50.cmd to it.
10) Save the Project, Build the Project.

11) Load the Program using Serial Tab.
Load Program>Browse Corresponding ascii file name whose name is same as name of program.

12) Reset the Kit and Type **SM** on kit using Kit's Keyboard

13) After successfully downloading the program observe the output waveform on CRO and calculate Time Period, amplitude and frequency.

Programs

Triangular Waveform Generation

```
                .MMREGS
                .TEXT
C000      START:   LDP #100H
C001               SPLK #0, 0
C002      CONT1:   LAR 2, #43; Value of frequency
C003      CONT  :  OUT 00, 4
C004               LACC 00
C005               ADD #15    ; Value of amplitude (added)
C006               SACL 00
C007               MAR *,2
C008               BANZ C004,*-
C009               LAR 2, #43    ; Value of frequency
C00a      CONTX:   OUT 00, 4
C00b               LACC 00

C00c              SUB #15      Value of amplitude (subtracted)
C00d              SACL 00
C00e              MAR *,2
C00f              BANZ C00D
C010              B COUNT1
```

Square Waveform Generation

```
                .MMREGS
                .TEXT
C000            START:  LDP #100H
C001                     SPLK #7FFH,00
C002                     OUT 00,4
C003                     CALL DELAY
C004                     SPLK #0FFFH,01
C005                     OUT 01,4
C006                     CALL DELAY

C007        DELAY: LAR AR3, #FF
C008        BACK    :  MAR *, AR3
C009                     BANZ BACK,*-
C00a                     RET
```

Discussions on results:

This experiment not only provides the software implementation of the waveform generation but also enables the students to perform real time interfacing with external hardware like CRO. Thus from the results students will be able to

1) Discuss the different parameter settings in DSP Trainer Kit (TMS 320C50PQ57) for obtaining the different waveforms.
2) Discuss how the assembly language programs are written using line assembler in the Monitor mode.

Experiment – 9

Aim: To verify and perform LED interfacing using DSP Trainer Kit (TMS 320C50PQ57).

Apparatus: PC, DSP Trainer Kit (TMS 320C50PQ57).

Algorithm:

1) Load data pointer with the current page number in which we want to store local variable.
2) Load accumulator with contents of address 00h.
3) Start a label BACK and store the low accumulator with 00h.
4) Call DELAY label.
5) Initialize count 0ah and compare it with zero.
6) And Call DELAY
7) After coming from Call DELAY compliment the values
8) Again go back to BACK
9) Repeat the procedure.
10) In label DELAY load the Auxiliary Register AR7 with a value FF. Multiply this auxiliary register AR7 value with the same value.
11) In label HERE decrement the resultant value of above multiplication by one and check for not equal to zero. Repeat this procedure till the result goes to zero, and then return from the loop

Procedure:

1) Open the C50 Debugger software from the desktop.

2) Click the serial port settings and select auto detect.

Serial>Port Settings>AutoDetect

3) It won't detect at first attempt then reset the kit using reset button and **Type SM** using kit's keyboard and press Enter.

4) Again Select auto detect then it will detect.

5) Select New Project in C50 Debugger Software, give name to project.

6) Create New Assembly File from the File Tab of C50 Debugger.

File>New>Assembly File

7) Type the Program in the above file and save it as **filename.asm**

*Select filename as Project Name

8) Go to Project Folder and right click on Assembly folder and add the filename.asm to it.

9) Right Click on the CMD Files folder and add **Micro50.cmd** to it.

10) Save the Project, Build the Project.

11) Load the Program using Serial Tab.

Load Program>Browse Corresponding ascii file name whose name is same as name of program.

12) Reset the Kit and Type **SM** on kit using Kit's Keyboard

13) After successfully downloading the program observe the output i.e LEDs on the DSP kit.

Program:

```
          .MMREGS
          .TEXT
START:    LDP #100H
          LACC #0H
BACK:     SACL 00H
          OUT 00H,0AH
          CALL DELAY
          CMPL
          B BACK

DELAY:    LAR AR7,#07FFH
          MAR *,AR7
HERE:     BANZ HERE,*-
          RET
```

Discussions on results:

This experiment not only provides the software implementation of the LED interfacing but also enables the students to understand the programming concept to initialize hardware like LEDs.

From the results students will be able to

1) Discuss the different parameter settings in DSP Trainer Kit (TMS 320C50PQ57) for performing LED interfacing.
2) Discuss how the assembly language programs are written using line assembler in the Monitor mode.
3) Discuss the effect of avoiding the delay routine in the program.
4) Discuss the effect of increasing the delay duration in the program.

INTRODUCTION TO MICRO – 2407 Trainer Kit

1. Introductions

Micro-2407 is a 16-bit (data lines) fixed point DSP trainer, based on texas instruments TMS320LF2407A DSP Processor. This trainer enables the user to learn the basics of digital signal processing & digital control along with basic DSP functions like filtering, PWM generation, and calculation of spectral characteristics of input analog signals. The trainer helps to perform real time implementation of very complex algorithms, such as adaptive control, Motor control etc.

The TMS320LF2407A contains a C2xx DSP core along with useful peripherals such as ADC, Timer, PWM Generation are integrated onto a single piece of silicon.

The Micro-2407 trainer can be operated in two modes.

In the mode: 1(serial mode) the trainer is configured to communicate with PC through serial port.

In the mode: 2 (stand alone mode), the user can interact with the trainer through the IBM PC keyboard and 16 × 2 LCD display.

Specifications

1. PROCESSOR

CPU	: Texas Instruments TMS320LF2407A,
Crystal Frequency	: 10MHz.
Clock Frequency	: 40 MHz
Wait States	: 2 for EPROM, 0 for On-Chip RAM, 2 for external RAM and 6 for LCD DISPLAY.
2. MONITOR (EPROM)	: 0x0000 - 0xBFFF for 48kwords
3. MEMORY	
Program RAM	: 0xC000 - 0xFFFF for 16kwords.
Data RAM	: 0x9000 - 0xFFFF for 32kwords (0x8000 to 0x8FFF reserved for monitor).
4. SERIAL	: One RS232C Serial Interface using on-chip SERIAL COMMUNICATION INTERFACE (SCI) module
5. TIMER	: On-chip timer can be used.
6. INTERRUPTS	: 6 Interrupt lines of TMS320LF2407A are available to users.
7. IBM AT KEY BOARD	: CD 4015-101 KEY keyboard controller
8. DISPLAY	: 16x2 LCD display (For Mode-2).
9 ON-BOARD BATTERY BACKUP	: 3.6V, Ni-Cd Battery

10. POWER SUPPLY (LPOW-001A) SPECIFICATIONS

Mains	: 230 Volts AC at 50 Hz
Outputs	: 1. + 5 Volts, 3.5 Amps Regulated 2. + 12 Volts, 150 mA Regulated 3. - 12 Volts, 150 mA Regulated 4. +5 Volts, 500 mA Regulated

2. Hardware Description of MICRO 2407

2.1. Front Panel Description

This chapter gives a brief description about Front Panel of Micro-2407 Trainer board.

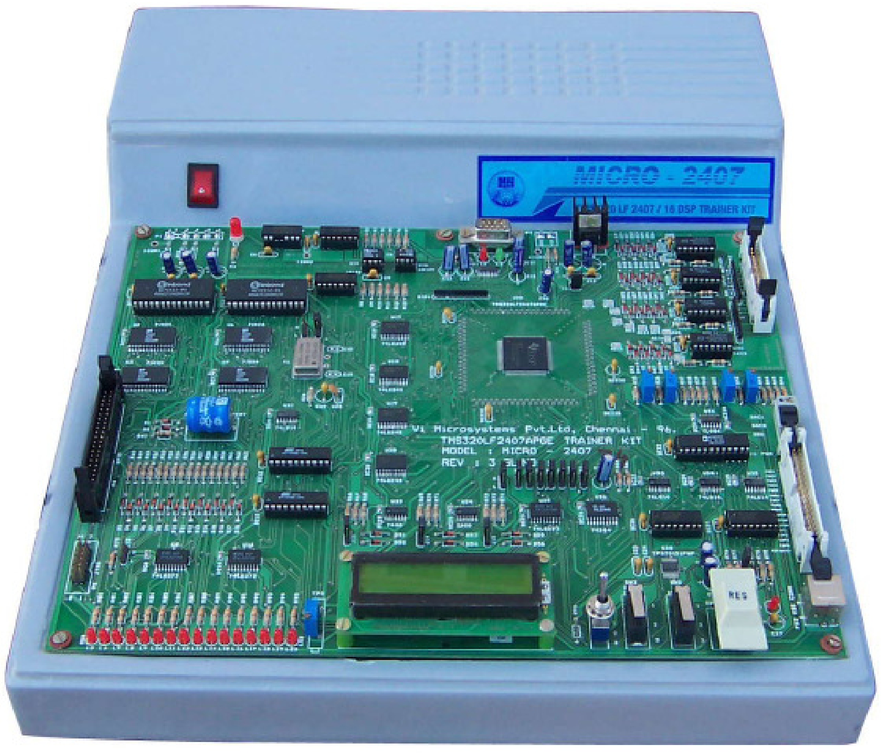

MICRO - 2407
LF 2407 / 16 DSP TRAINER KIT
Vi Microsystems Pvt.Ltd, Chennai - 96.
TMS320LF2407APGE TRAINER KIT
MODEL : MICRO - 2407
REV : 3

Figure -Pictorial View of Micro-2407

2.2 *Connector Details*

1. Introduction

Following are the shortlist of connectors available on Micro-2407 trainer board.

P1 - 5 Pin Unicon Connector
P2 - 9 Pin Serial port Connector
P3 - 2 Pin J801 Connector
P4 - 40 Pin FRC Connector
P5 - 14 Pin JTAG Connector
P6 - 26 Pin FRC Connector
P7 - J801 3 Pin Connector
P8 - 34 Pin FRC Connector
P9 - 6 Pin Keyboard Connector

2. Power Connector: (P1)

Connector Used

Single row 5 Pin UNICON Male Connector
- Spacing between pins 2,3,4,5 = 5mm

- Spacing between pins 1&2 = 7.5mm

Pin Details

PIN	DETAILS
1	GND
2	-12V
3	+12V
4	NC
5	VCC

Where,
NC - No Connection
VCC - +5V Power Supply
GND - 0V Reference Ground

Mating Connector
Single row 5-pin UNICON Female Connector
- With the same spacing as said above

3. Serial Port Connector: (P2)

Connector Used

9 Pin D type Male Connector
- Pins arranged in two rows of 5 and 4 pins
- Grid pitch is 2.76 mm * 2.84 mm
- The connector is AMPHENOL standard

PIN	DETAILS
1	NC
2	RxD
3	TxD
4	NC
5	GND
6	NC
7	RTS
8	CTS
9	NC

Signal Definitions

TxD	Transmit Data
RxD	Receive Data
RTS	Ready to send
CTS	Clear to send

Mating Connector
9 Pin D type Female Connector with the same specifications.

4. J801 2 Pin Connector (P3)

Pin Configuration

1 - GND
2 - +5v

5. General Purpose Input / Output Connector (P4)

Connector Used: 40 Pin Dual row male header
Mating Connector: 40 Pin Dual row female Socket

6. 14 PIN JTAG Connector (P5)

Connector Used: 14 Pin Double row male connector
Mating Connector: 14 Pin Double row female Socket

7. ADC Input FRC Connector (P6)

Connector Used
26 Pin Double Row Header (13*2)
Mating Connector
26 Pin Double row socket

8. J801 3 Pin Connector (P7)

Pin Configuration
1 - GND
2 - DAC1 OUTPUT
3 - DAC2 OUTPUT

9. PWM Output 34 Pin FRC Connector (P8)

Connector Used: 34 Pin Double Row Headers

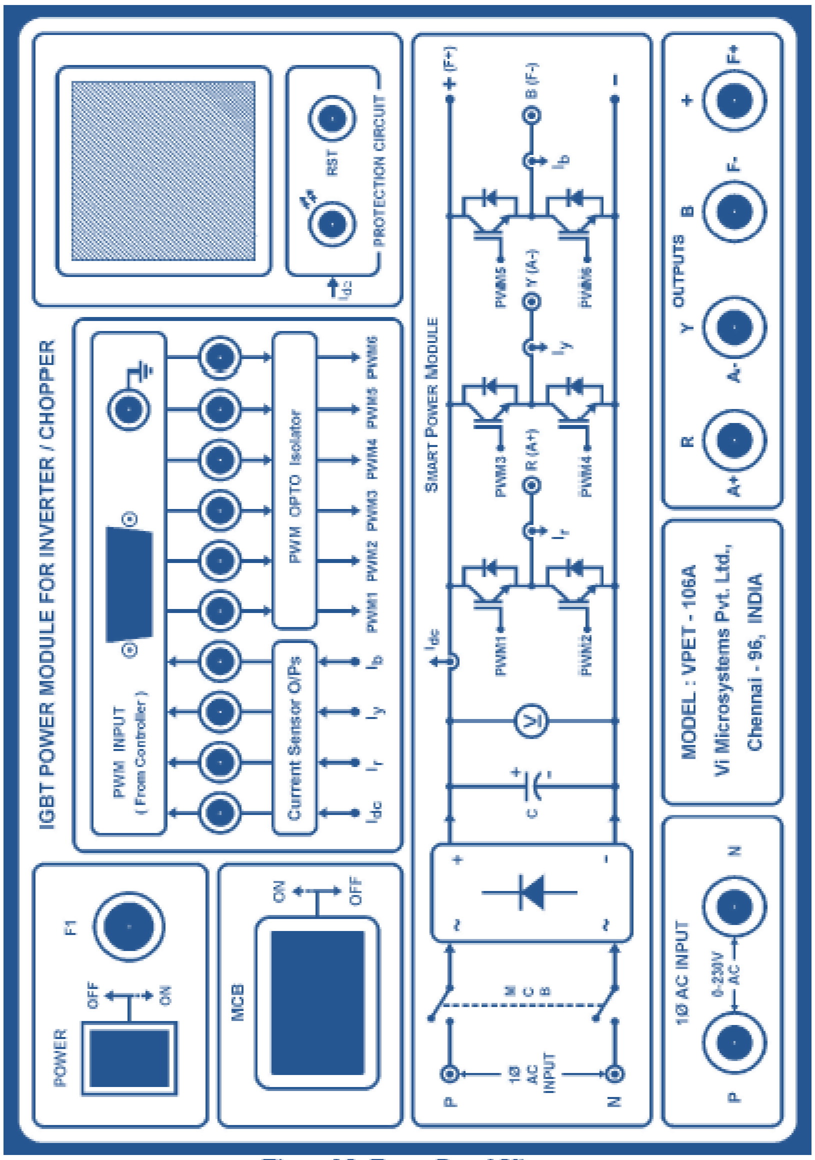
***Mating* Connector**
34 Pin Double Row Socket

10. IBM PC Keyboard Connector: (P9)

Connector *Used*
6 Pin PS2 Female Connector

Signal Definitions
CLK - Keyboard Clock
DATA - Serial Data from keyboard
***Mating* Connector**
6 Pin PS2 Male Connector
(Available in the IBM PC Keyboard itself)

2.3 Front Panel View



Front Panel Description

Power ON/OFF switch	- To switch ON/OFF power to the module.
MCB	- To control input AC voltage to the power circuit.
PWM INPUT	- To connect the PWM input / feedback signal from / to the controller unit
R, Y, B	- To connect 3· output to the load.
Voltmeter	- To display DC link voltage.
P, N	- To connect AC input to the module.
A+, A-, F+, (+)	- To connect the DC output to the load.

RST - To reset the module.

PWM1, PWM 2...PWM 6 - Test points to view the PWM signal to the switches with respect to ground point.

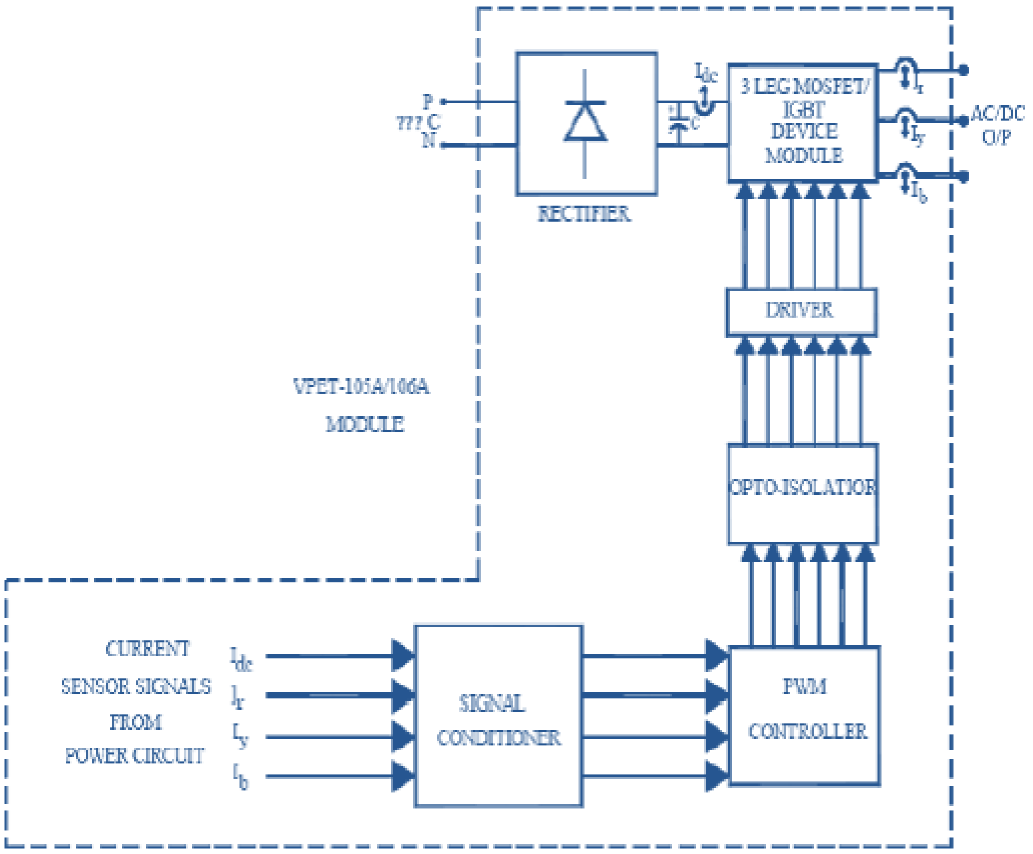
I_{dc}, I_r, I_y, I_b - Test points to view the sensed input DC current & output R, Y and B phase currents.

PWM Isolation - To isolate the gate signals.

2.3 Hardware Description

Block Diagram

Figure 16 Block diagram for IGBT power module



The block diagram for IGBT power module consists of the following.

1. Rectifier & Filter
2. Power circuit
3. Controller
4. Optoisolator
5. Gate Driver
6. Current Sensor
7. Signal conditioner
8. Protection Circuit.

Block Diagram Description

1. Rectifier & Filter

The input AC voltage is rectified by the Diode Bridge Rectifier circuit. The rectified DC voltage is fed to the power circuit through a filter capacitor. The filter capacitor **eliminates the unwanted ripple** in the Dc input.

2. Power Circuit

The Power circuit consists of three leg IGBT circuit. IGBT switch is used as a switching device in the power circuit. The PWM signal from the driver IC is fed to the gate of the switch. The output from the power circuit is given to the load. The output may be either AC/DC depending on the inverter/chopper mode of operation.

3. Controller

The PWM signal to the IGBT switches is generated by the controller unit. The controller may be any processor. The PWM signal from the controller is fed to the module through the connector provided on the front panel. In this implementation, controller is used PWM generation.

4. Optoisolator (6N137)

The function of Optoisolator is to isolate the control circuit from power circuit.PWM signal from the controller is not directly fed to the power circuit in order to protect the PWM signal it is essential to provide isolation circuit between power circuit and control circuit or else the high power components may damage the low power PWM circuit components.

5. Gate Driver (IR2110)

An IGBT drive circuit is designed **to connect the gate directly to a voltage bus** with no intervening resistance other than the impedance of the drive circuit switch. Gate driver acts as a **high-power buffer stage** between the PWM output of the control device and gates of the primary power switching IGBT.

6. Current Sensor

IGBT power module output current is not directly fed to control (Protection) circuits. The sensor used for sensing current works on the principle of Hall Effect. Hence these sensors are called Hall Effect transducer. Hall Effect transducer output current depends upon transducer primary and secondary winding ratio. The turn's ratio represents the ratio of the number of primary turns to the number of secondary turns. A Hall effect current transducer senses the current I_{DC}, I_r(R), I_y(Y) I_b(B).

7. Signal Conditioner

The signal from the current sensor is fed back to the controller unit. The gain of the current signals has to be improved to achieve controller requirements. So the signal needs to be conditioned in the signal conditioner circuits.

8. Protection Circuit

In the protection circuit the DC current I_{dc} is sensed. If the I dc value increases than the limit, the protection circuit activates and cuts the PWM to the switches in the module. The LED glows to indicate the system shut down condition. The system can be reset by the RST press button provided on the front panel. Now the LED switches OFF. Again the PWM signal should be applied to the system.

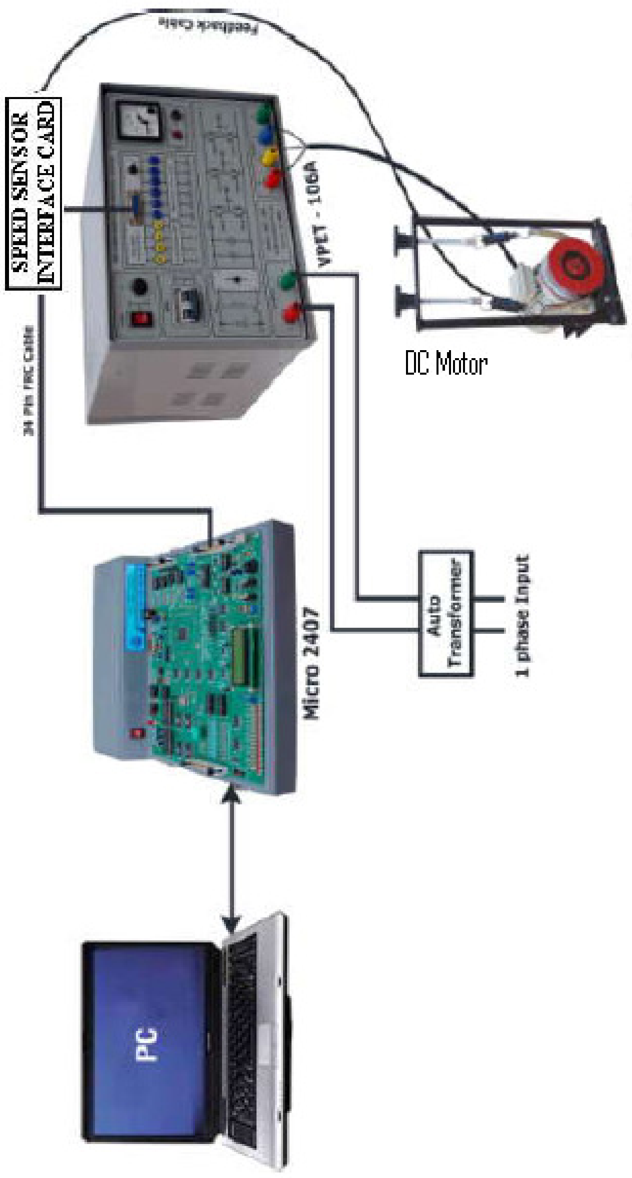
Experiment – 10

Aim: To develop a Program to control the speed of Separately Excited DC motor by implementing 4 quadrant choppers and using Micro – 2407 & VPET-106A.

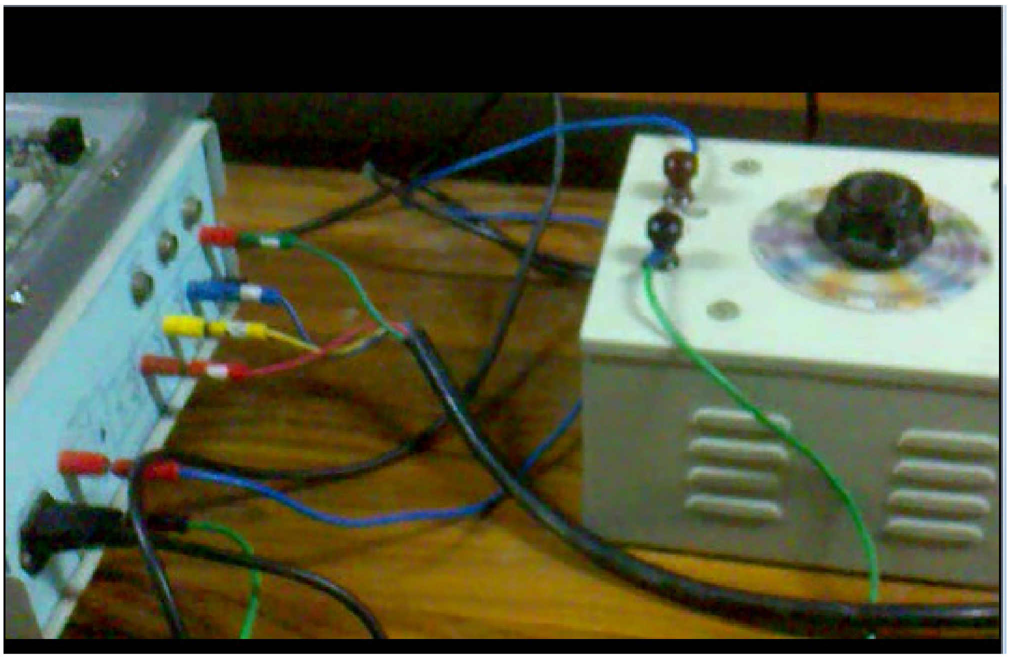
Equipment Required:

1. VPET-106A Module
2. Separately Excited DC motor
3. Micro-2407 Trainer
4. PC-PC serial port cable.
5. Patch Chords
6. 26 Pin FRC cable
7. 34 Pin FRC cable (dual type)
8. DC Regulated Power Supply (0-30V)

Connection Diagram



Connection View



Procedure:

a) Connection Procedure:

1. Connect the 26 pin FRC cable one end to Micro-2407 Trainer and the other end to "ADC input to DSP" placed in ASIPM Power Module.

2.Connect the 34 pin FRC 1to 1 to cable one end to MICRO-2407 Trainer and ,the other end to "PWM output from DSP" and the remaining end to QEP Driver.

3. Connect the Motor terminals A and AA to the U and V terminals of "ASIPM Power Module"

4. Connect the motor terminal F and FF to the W, +ve terminals in "ASIPM Power Module".

5. If speed of the motor is to be sensed using optical encoder or proximity switch connect, the 9 pin D cable from the motor into the speed feedback connector placed at the back side of "ASIPM Power Module"

6. If the speed of the motor is to be sensed using QEP, connect the 9 pin D cable from the signal conditioner. Connect the 34 pin FRC from the QEP Signal Conditioner unit into the 34 pin header placed in the Micro-2407 Trainer.

7. Connect the serial port of PC to the serial port connector (P2) in the Micro-2407 Trainer.

8. Set the SW1 Switch in the Micro-2407 Trainer downward direction, it works in serial monitor mode.

b) Experiment Procedure

1. Verify the connection as per the connection procedure and wiring diagram.
2. Switch ON the Micro-2407 Trainer.
3. Power ON the "ASIPM Power Module" power ON power switch.
4. Check whether shut down LED "SD" LED glows, press the Reset switch, the LED gets OFF.

Open Loop Control of Dc Motor

1. Select C2407 from your target directory or from your desktop in PC.
2. Now the 'configure the port open, select the port to which Micro-2407 trainer and press enter key.
3. Select the open loop Menu in that select DC motor followed by Chop\v/up.
4. The plot of 'speed and current' in now available in PC.
5. Select the send, now transmission completed message appears given ok.
6. Select the Execute, now "Kp and Ki parameter settings window appears. Enter the Kp and Ki values and given OK.
7. Now the current value of speed and current is displayed.
8. Vary the speed of motor by pressing SW2 and SW3 key placed in the Micro-2407 trainer and note down the corresponding output voltage.

Closed Loop Control of DC Motor.

1. Select the "C2407" from the desktop in PC.
2. Now the "Configure port open "open, select the port to which Micro-2407 Trainer and press "Enter Key".
3. Select the Closed Loop Menu in that select DC Motor followed by Chop\v/up.
4. The plot of "Speed and current" is now available in PC.
5. Select the SEND, now transmission completed message appears given ok.
6. Select the EXECUTE; now the "K_p and K_i Parameter Settings" windows appears. Enter the K_p and K_i values and given OK.
7. Now the current value of speed and current is displayed in PC.
8. Vary the speed of motor by pressing SW2 or SW3 key placed in the Micro-2407 Trainer and note down the corresponding voltage.

Results: Use the following Table for both Open Loop and Closed Loop for calculation of the

Table for closed Loop Mode of operation

Sr. No	Input DC Voltage V_{in}	Measured Output voltage From Power Module V_o	T_{on}	T_{off}	T $T_{on}+T_{off}$	Calculted Output Voltag $V_0 = \frac{T_{on}}{T} V_{in}$	Set Speed (rpm) N_{set}	Rated Speed (rpm) N_{rated}	Proportional Gain K_p	Integral Gain K_I
1										
2										
3										
4										
5										

Or we can use this Table for Open Loop mode

Table for Open Loop Mode of operation

S. No	Vin (v)	Iin (A)	S1 Kg	S2 Kg	Speed RPM	Torque N-m	I/P Power (w)	O/P Power (w)	% Efficiency η
1									
2									
3									
4									
5									

Torque = $(s_1 - s_2).r.g$
Where r=drum radius = 0.071m
Input Power: $= V_{in}.I_{in}$
Output Power: $2\pi NT / 60$
Efficiency η : (output power/Input Power)*100

Discussions on results:

Thus we have studied the speed of Separately Excited DC motor by implementing 4 quadrant choppers and using Micro – 2407 & VPET-106A.
From the results students will be able to

1) Discuss the open loop mode operation for calculating the efficiency of the motor in load variation.
2) Discuss the closed loop mode operation for calculating the efficiency of the motor in load variation.
3) Discuss the effect of Kp value on the offset error in speed.
4) Discuss the effect of Ki value on the offset error in speed.

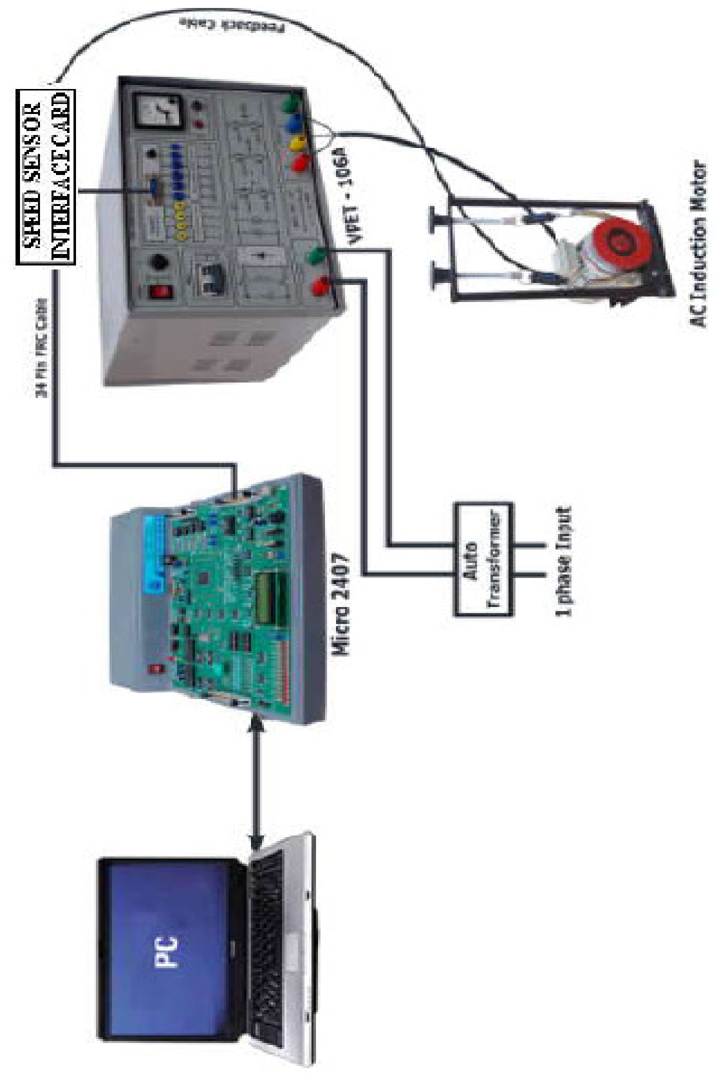
Experiment – 11

Aim: To study the Open Loop/Closed Loop Speed control of 3 phase AC Induction Motor with spring balance Load using VF Control (Sine PWM) using VPET-106A Module and Micro-2407 trainer kit.

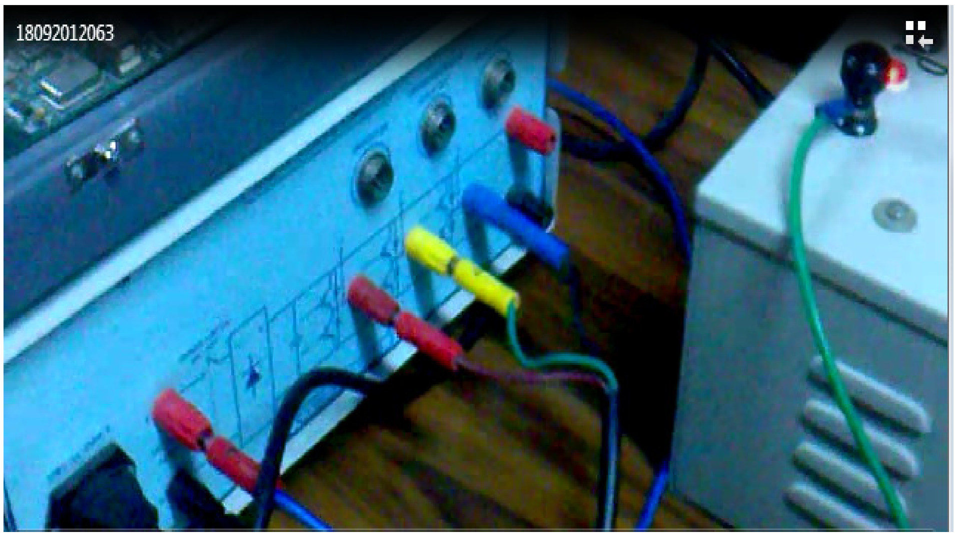
Equipments Required

1. VPET-106A Module
2. 3 phase AC induction motor
3. Micro-2407 Trainer
4. PC-PC serial port cable.
5. Patch Chords
6. 26 Pin FRC cable
7. 34 Pin FRC cable (dual type)
8. DC Regulated Power Supply (0-30V)

Connection Diagram



Connection View



Open Loop Speed Control:

Connection Procedure:

1. Connect the 1 phase AC supply terminals to the “1 phase 230V AC input” terminals of the VPET-106A Module through the 1 phase -VARIAC.

2. Connect the VPET-106A module output terminals U, V and W to the input terminals of 3 phase AC induction motor.

3. Connect Micro-2407 trainer module to power supply.

4. Connect the 34 pin FRC cable one end to the 34 Pin FRC headers in Micro – 2407 Trainer, one end to speed sensor interface card and the other end to “IGBT- PWM INPUTS” in the VPET-106A Power Module.

5. Connect the speed feedback signal from the motor to speed sensor interface card through 9-pin ‘D’ type 1:1 cable.

6. Connect Serial Port connector of PC to the 9-pin serial connector of Micro-2407 trainer using PC-PC serial port cable.

Experiment Procedure

1. Verify the connections as per the connection procedure and wiring diagram.

2. Switch ON the Micro-2407 Trainer.

3. Keep the auto transformer in minimum position, and then switch ON the power ON/OFF switch in the VPET-106A. Check whether shut down LED “SD” glows or not. If ‘SD’ LED glows press the Reset switch in the Micro-2407 and then reset your VPET-106A power module.

4. Download and execute the program as per the following **“Program Download Procedure”** given below.

5. Switch ON the MCB, then slowly increases the 3 phase AC input to the module through the auto transformer and set the DC link voltage at 350V.

6. Switch ON the PC and then press Reset Switch of the Micro-2407 Trainer.

Program Download Procedure

1. Double Click C2407 debugger icon from the target directory or from the desktop. Now the window shows

2. Now configure the serial port settings and press ‘OK’ button. Now the window shows.

3. Select the required program to be downloaded. For open loop AC motor with Sine/Tri or space vector Modulation select **Open Loop —> AC Motor —> Sine/Tri or Space vector.**

4. Ensure whether the motor is connected or not. Then click ‘OK’.

6. Click ‘Send’.

7. Click ‘OK’ (the particular file will be download to controller).

8. Now the transmission completed message appears, click ‘OK’ and then click ‘EXECUTE’.

9. Check the all PWMs by connecting CRO. Now apply AC voltage though single phase variac upto 300VDC.

10. Now, the speed value is plotted then speed and frequency numerical values are displayed.

11. To change the speed of the motor, use the increment/decrement switch in the 2407 kit.

12. To measure i/p voltage of the motor, connect Ac voltmeter (0- 450 V) across U and V terminals of the VPET-106A and frequency is obtained from the front end software.

Now check the V/F ratio of the motor.

13. By applying load to the motor, motor speed is not remains constant and it is not equal to the set speed.

14. To measure the load current of the motor, externally connect one AC ammeter in series with any one phase.

Closed Loop Speed Control:

Connection Procedure

1. Connect the 1 phase AC supply terminals to the “1 phase 230V AC input” terminals of the VPET-106A Module through the 1 phase -VARIAC.

2. Connect the VPET-106A module output terminals U, V and W to the input terminals of 3 phase AC induction motor.

3. Connect Micro-2407 trainer module to power supply.

4. Connect the 34 pin FRC cable one end to the 34 Pin FRC header in Micro – 2407 Trainer, one end to speed sensor interface card and the other end to "IGBT- PWM INPUTS" in the VPET-106A Power Module.

5. Connect the speed feedback signal from the motor to speed sensor interface card through 9-pin 'D' type 1:1 cable.

6. Connect Serial Port connector of PC to the 9-pin serial connector of Micro-2407 trainer using PC-PC serial port cable.

Experiment Procedure

1. Verify the connections as per the connection procedure and wiring diagram.

2. Switch ON the Micro-2407 Trainer.

3. Keep the auto transformer in minimum position, and then switch ON the power ON/OFF switch in the VPET-106A. Check whether shut down LED "SD" glows or not. If 'SD' LED glows press the Reset switch in the Micro-2407 and then reset your VPET-106A power module.

4. Download and execute the program as per the following **"Program Download Procedure"** given below

5. Switch ON the MCB, then slowly increases the 3 phase AC input to the module through the auto transformer and set the DC link voltage at 350V.

6. Switch ON the PC and then press Reset Switch of the Micro-2407 Trainer.

Program Downloading Procedure:

1. Double Click C2407 icon from the target directory or from the desktop. 2. Now configure the serial port settings and press 'OK' button. Now the window shows.

2. Select the required program to be downloaded. For close loop AC motor with Sine/Tri or space vector Modulation select **Close Loop —> AC Motor —> Sine/Tri or Space vector.**

3. Ensure whether the motor is connected or not. Then click 'OK'.

4. Click 'Send' then the window shows below.

5. Click 'OK' (the particular file will be download to controller)

6. Now the transmission completed message appears, click 'OK' and then click 'EXECUTE'.

7. Check the all PWMs by connecting CRO. Now apply AC voltage though single phase variac upto 300VDC.

8. Now, the speed value is plotted then speed and frequency numerical values are displayed.

9. To change the speed of the motor, use the increment/decrement switch in the 2407 kit.

10. To measure i/p voltage of the motor, connect Ac voltmeter (0- 450 V) across U and V terminals of the VPET-106A and frequency is obtained from the front end software. Now check the V/F ratio of the motor.

11. By applying load to the motor, motor speed is remains constant and it is equal to the set speed.

12. To measure the load current of the motor, externally connect one AC ammeter in series with any one phase. (Ref: connection diagram).

Results: Use the following Table for Closed Loop mode for operation

Sr. No	Input Voltage V_{in}	Measured Output voltage From Power Module V_o	T_{on}	T_{off}	T T_{o}	Calculted Output Voltag	Set Speed (rpm) N_{set}	Rated Speed (rpm) N_{rated}	K_p	K_I
1										
2										
3										
4										
5										

Discussions on results:

Thus the real time interfacing of the Induction motor with Micro-2407 trainer kit is experimented and speed control of 3 –ph Induction Motor with spring balance Load using the VPET-106A Module and Micro-2407 trainer kit is performed.
From the results students will be able to

1) Discuss the open loop mode operation for calculating the efficiency of the motor in load variation.
2) Discuss the closed loop mode operation for calculating the efficiency of the motor in load variation.
3) Discuss the effect of Kp value on the offset error in speed.
4) Discuss the effect of Ki value on the offset error in speed.

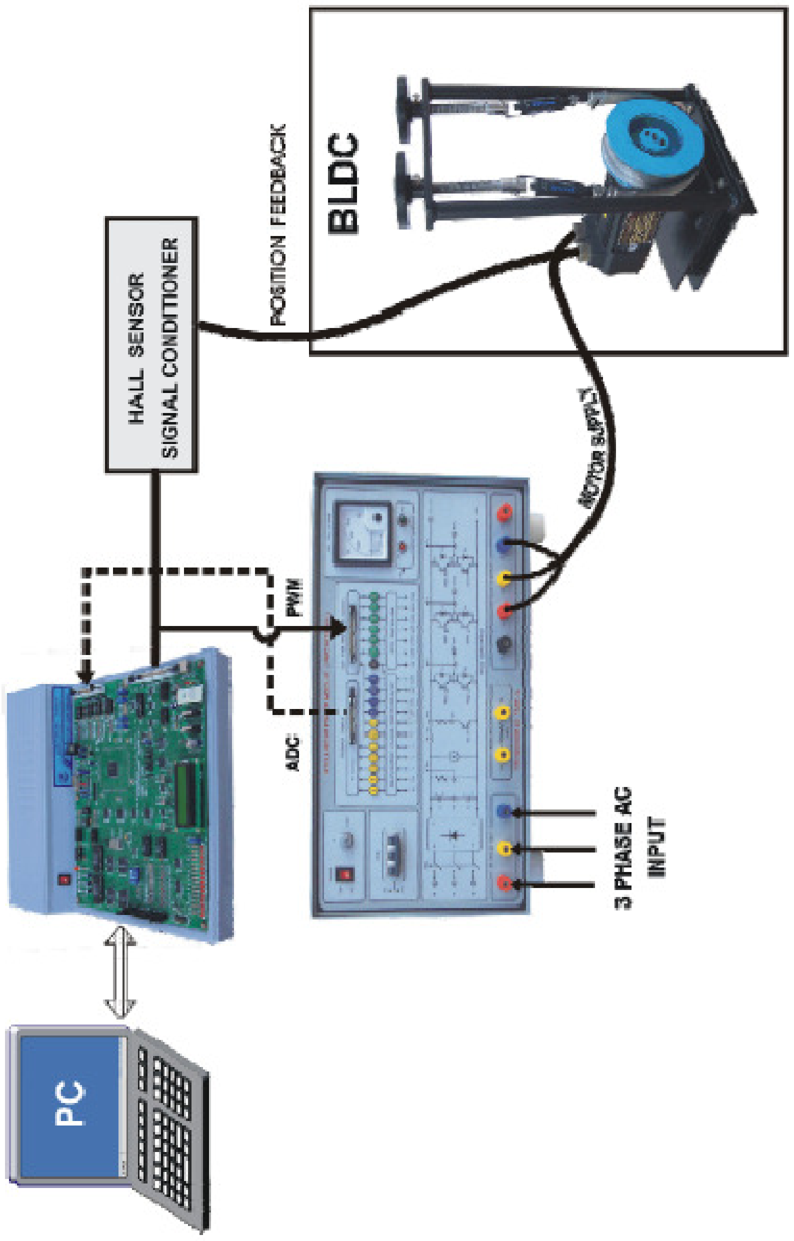
Experiment – 12

Aim: To study the open and closed loop speed control of BLDC Motor using IPM and Micro-2407.

Equipments Required

1. PEC16DSMO1 Power module
2. Micro - 2407 Trainer kit
3. BLDC Motor
4. Hall Sensor signal conditioner
5. Cables
    i. 34 pin FRC 1 to 1 to 1(dual type).
    ii. 26 pin FRC 1 to 1.
    iii. PC to PC Serial port cable

Connection Diagram



Connection Procedure

1. Connect the three-phase AC supply (MAINS) to i/p of 3NVARIAC.Then 3-phase VARIAC o/p connect to R, Y and B (3N INPUTS) terminals of PEC16DSMO1 Power Module.
2. Connect the U, V, W terminals (through switching output connector) in the PEC16DSMO1 Power Module to the 7 - pin supply connector of the Motor (refer connection diagram)
3. Connect the 17 pin Feedback connector from the Motor into Hall sensor signal conditioner (Through 9 Pin "D Connector").
4. Connect the 34 pin FRC cable one end to the 34 Pin FRC header in Micro - 2407 Trainer, one end to Hall Sensor signal conditioner and the other end to "IGBT-PWM INPUTS" in the PEC16DSMO1 Power Module.
5. Connect the serial port PC-PC cable between PC and Micro - 2407 Trainer.
6. Connect one end of 26 Pin FRC cable to 26 pin connector placed in Micro - 2407 Trainer and the other end to "FEEDBACK INPUTS" in PEC16DSMO1 Power Module.

Experiment Procedure

1. Verify the connections as per the connection procedure and Wiring Diagram.
2. Switch ON the Micro-2407 DSP Trainer.
3. Power ON the Intelligent Power Module (PEC16DSMO1) and MCB.
4. Check whether shut down LED "SD" glows in the power module or not. If 'SD' LED glows press the Reset switch in the front panel, the LED gets OFF.
5. After ensuring all the connections, using VARIAC apply the input voltage slowly 450V DC (DC rail Voltage) which is shown in the power module's voltmeter.
6. Switch ON the PC and set the switch SW1 in down position (Serial Mode) of Micro- 2407 DSP Trainer, then press Reset switch.
7. Select the "BLDC Motor Controller" icon in the Desktop.
8. To open port, Select "File" and click "OPEN PORT" as shown below.
9. On Port open the window asks for Port No, Baud Rate, etc., as shown below.
10. In the displayed window, select the connected port (COM 1, 2, 3, or 4) and other settings are default, port is open for communication
11. Now select the "Motor Controller –> BLDC" and select the desired controller "OPEN LOOP" or "CLOSED LOOP".

Open Loop Control

*Select the OPEN LOOP controller option.

12. From the Graphical display, click “DOWNLOAD” button.
13. Reset the 2407 kit once & select the “HSBLDCOL.ASC” File from the respective path.
14. After selecting the file, “DOWNLOADED SUCCESSFULLY” appears in the screen, click “EXECUTE” to execute the downloaded file.
15. Now check the PWM output & Hall sensor output, which are all terminated in the power module.
16. To change the speed of the motor, use the up/down switch in the 2407 kit.
17. To run the motor in Forward/Reverse direction, or to Run/ Brake, use the options given in the software.
18. The motor speed is shown in the PC with graphical representations.
19. The Phase current, DC current waveforms are plotted in the plotting section.

Closed Loop Control

*Select the CLOSED LOOP controller option.

20. From the Graphical display, click “DOWNLOAD” & reset the 2407 kit once & select the “HSBLDCCL.ASC” File from the respective path
21. After selecting the file, “DOWNLOADED SUCCESSFULLY” message appears in the screen. Now, click “EXECUTE” to execute the downloaded file.
22. Now check the PWM output & Hall sensor output, which are all terminated in the power module.
23. To change the speed of the motor, use the up/down switch in the 2407 kit or click “ENTER SPEED” option in the software, then enter the required speed.
24. To run the motor in Forward/Reverse direction, or to Run/ Brake, use the options given in the software.
25. The set speed & motor speed are shown in the PC with graphical representations.
26. The Phase current, DC link current waveforms are plotted in the plotting section.
27. To measure the load current of the motor, Externally connect one AC ammeter in series with any one phase.(ref connection diagram)

Results: Use the following Table for closed Loop mode of operation.

Sr. No	**Input Voltage** V_{in}	**Measured Output voltage From Power Module** V_o	T_{on}	T_{off}	**T** T_{on}	**Calculted Output Voltag**	**Set Speed (rpm)** N_{set}	**Rated Speed (rpm)** N_{rated}	K_p	K_I
1										
2										
3										
4										
5										

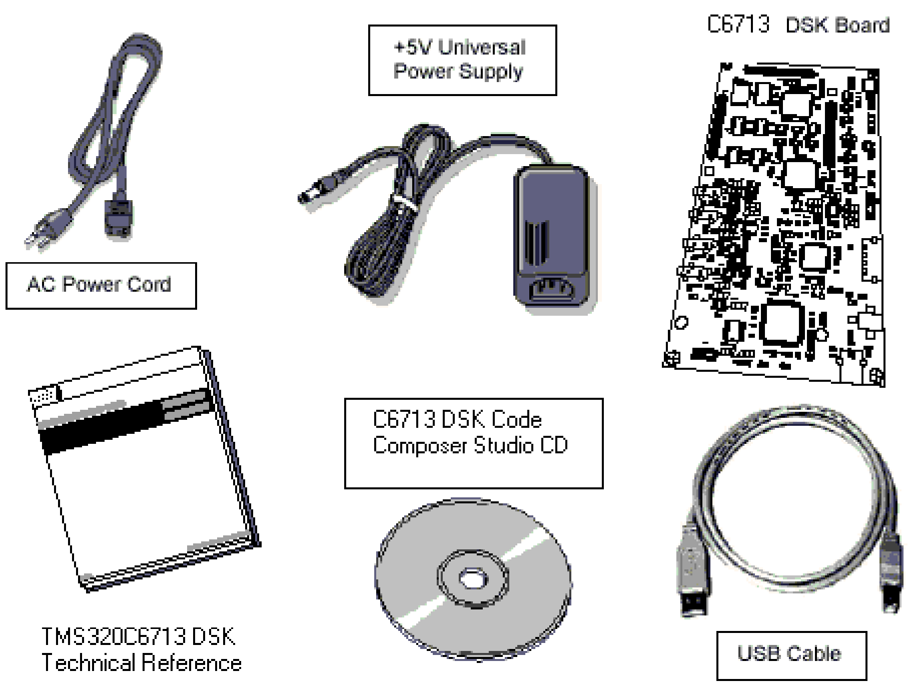
Discussions on results:

Thus the real time interfacing of the Induction motor with Micro-2407 trainer kit is experimented and speed control of BLDC motor in open and closed loop mode is experimented using IPM and Micro-2407.

From the results students will be able to

1) Discuss the open loop mode operation for calculating the efficiency of the motor in load variation.
2) Discuss the closed loop mode operation for calculating the efficiency of the motor in load variation.
3) Discuss the effect of Kp value on the offset error in speed.
4) Discuss the effect of Ki value on the offset error in speed.

Introduction to TMS320C6713 DSK[Courtesy: Texas Instrument]


AC Power Cord
+5V Universal Power Supply
C6713 DSK Board
TMS320C6713 DSK Technical Reference
C6713 DSK Code Composer Studio CD
USB Cable

DSP Starter Kit (DSK) for the TMS320C6713 (Courtesy Spectrum Digital)

The C6713™ DSK builds on TI's industry-leading line of low cost, easy-to-use DSP Starter Kit (DSK) development boards. The high-performance board features the TMS320C6713 floating-point DSP. Capable of performing 1350 million floating-point operations per second (MFLOPS), the C6713 DSP makes the C6713 DSK the most powerful DSK development board.
The DSK is USB port interfaced platform that allows to efficiently develop and test applications for the C6713. The DSK consists of a C6713-based printed circuit board that will serve as a hardware reference design for TI's customers' products. With extensive host PC and target DSP software support, including bundled TI tools, the DSK provides ease-of-use and capabilities that are attractive to DSP engineers.

The following checklist details items that are shipped with the C6711 DSK kit.

➢ **TMS320C6713 DSK**	**TMS320C6713 DSK development board**
➢ **Other hardware**	**External 5VDC power supply**
	IEEE 1284 compliant male-to-female cable
➢ **CD-ROM**	**Code Composer Studio DSK tools**

The C6713 DSK has a TMS320C6713 DSP onboard that allows full-speed verification of code with Code Composer Studio. The C6713 DSK provides:

- A USB Interface
- SDRAM and ROM
- An analog interface circuit for Data conversion (AIC)

- An I/O port
- Embedded JTAG emulation support

Connectors on the C6713 DSK provide DSP external memory interface (EMIF) and peripheral signals that enable its functionality to be expanded with custom or third party daughter boards.

The DSK provides a C6713 hardware reference design that can assist you in the development of your own C6713-based products. In addition to providing a reference for interfacing the DSP to various types of memories and peripherals, the design also addresses power, clock, JTAG, and parallel peripheral interfaces.

The C6713 DSK includes a stereo codec. This analog interface circuit (AIC) has the following characteristics:

High-Performance Stereo Codec

- 90-dB SNR Multibit Sigma-Delta ADC (A-weighted at 48 kHz)
- 100-dB SNR Multibit Sigma-Delta DAC (A-weighted at 48 kHz)
- 1.42 V – 3.6 V Core Digital Supply: Compatible With TI C54x DSP Core Voltages
- 2.7 V – 3.6 V Buffer and Analog Supply: Compatible Both TI C54x DSP Buffer Voltages
- 8-kHz – 96-kHz Sampling-Frequency Support

Software Control Via TI McBSP-Compatible Multiprotocol Serial Port

- I 2 C-Compatible and SPI-Compatible Serial-Port Protocols
- Glueless Interface to TI McBSPs

Audio-Data Input/Output Via TI McBSP-Compatible Programmable Audio Interface

- I 2 S-Compatible Interface Requiring Only One McBSP for both ADC and DAC
- Standard I 2 S, MSB, or LSB Justified-Data Transfers
- 16/20/24/32-Bit Word Lengths

The C6713DSK has the following features:

The 6713 DSK is a low-cost standalone development platform that enables customers to evaluate and develop applications for the TI C67XX DSP family. The DSK also serves as a hardware reference design for the TMS320C6713 DSP. Schematics, logic equations and application notes are available to ease hardware development and reduce time to market.

The DSK uses the 32-bit EMIF for the SDRAM (CE0) and daughtercard expansion interface (CE2 and CE3). The Flash is attached to CE1 of the EMIF in 8-bit mode.

An on-board AIC23 codec allows the DSP to transmit and receive analog signals. McBSP0 is used for the codec control interface and McBSP1 is used for data. Analog audio I/O is done through four 3.5mm audio jacks that correspond to microphone input, line input, line output and headphone output. The codec can select the microphone or the line input as the active input. The analog output is driven to both the line out (fixed gain) and headphone (adjustable gain) connectors. McBSP1 can be re-routed to the expansion connectors in software.

A programmable logic device called a CPLD is used to implement glue logic that ties the board components together. The CPLD has a register based user interface that lets the user configure the board by reading and writing to the CPLD registers. The registers reside at the midpoint of CE1.

The DSK includes 4 LEDs and 4 DIP switches as a simple way to provide the user with interactive feedback. Both are accessed by reading and writing to the CPLD registers.

An included 5V external power supply is used to power the board. On-board voltage regulators provide the 1.26V DSP core voltage, 3.3V digital and 3.3V analog voltages. A voltage supervisor monitors the internally generated voltage, and will hold the board in reset until the supplies are within operating specifications and the reset button is released. If desired, JP1 and JP2 can be used as power test points for the core and I/O power supplies.

Code Composer communicates with the DSK through an embedded JTAG emulator with a USB host interface. The DSK can also be used with an external emulator through the external JTAG connector.

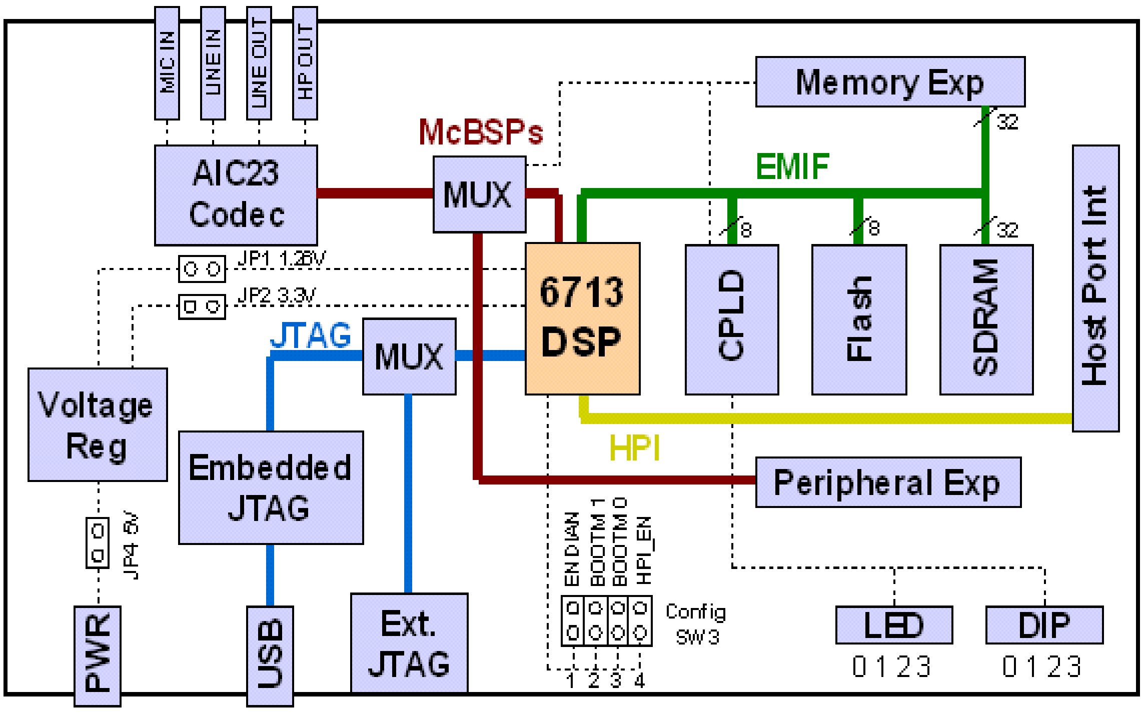
TMS320C6713 DSP Features

- Highest-Performance Floating-Point Digital Signal Processor (DSP):
  - Eight 32-Bit Instructions/Cycle
  - 32/64-Bit Data Word
  - 300-, 225-, 200-MHz (GDP), and 225-, 200-, 167-MHz (PYP) Clock Rates
  - 3.3-, 4.4-, 5-, 6-Instruction Cycle Times
  - 2400/1800, 1800/1350, 1600/1200, and 1336/1000 MIPS /MFLOPS
  - Rich Peripheral Set, Optimized for Audio
  - Highly Optimized C/C++ Compiler
  - Extended Temperature Devices Available
- Advanced Very Long Instruction Word (VLIW) TMS320C67x™ DSP Core
  - Eight Independent Functional Units:
    - Two ALUs (Fixed-Point)
    - Four ALUs (Floating- and Fixed-Point)
    - Two Multipliers (Floating- and Fixed-Point)
  - Load-Store Architecture With 32 32-Bit General-Purpose Registers
  - Instruction Packing Reduces Code Size
  - All Instructions Conditional
- Instruction Set Features
  - Native Instructions for IEEE 754
    - Single- and Double-Precision
  - Byte-Addressable (8-, 16-, 32-Bit Data)
  - 8-Bit Overflow Protection
  - Saturation; Bit-Field Extract, Set, Clear; Bit-Counting; Normalization
- L1/L2 Memory Architecture

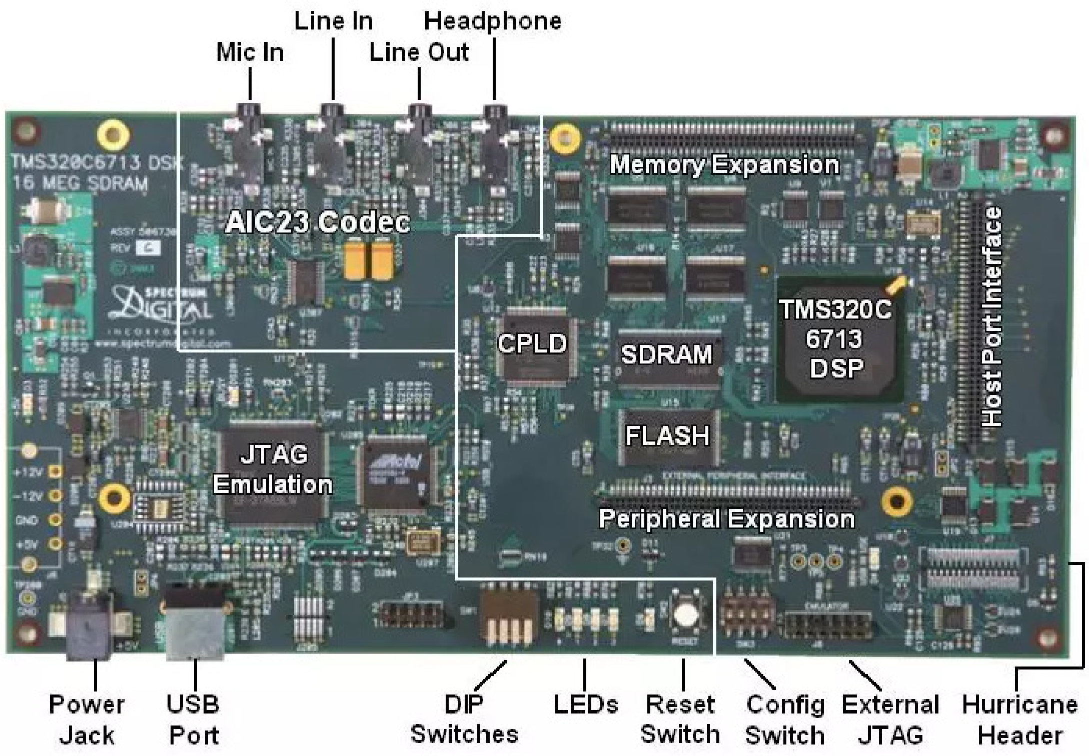
- 4K-Byte L1P Program Cache (Direct-Mapped)
- 4K-Byte L1D Data Cache (2-Way)

- 256K-Byte L2 Memory Total: 64K-Byte L2 Unified Cache/Mapped RAM, and 192K-Byte Additional L2 Mapped RAM

- Device Configuration
  - Boot Mode: HPI, 8-, 16-, 32-Bit ROM Boot
  - Endianness: Little Endian, Big Endian
- 32-Bit External Memory Interface (EMIF)
  - Glueless Interface to SRAM, EPROM, Flash, SBSRAM, and SDRAM
  - 512M-Byte Total Addressable External Memory Space
- Enhanced Direct-Memory-Access (EDMA) Controller (16 Independent Channels)
- 16-Bit Host-Port Interface (HPI)
- Two Multichannel Audio Serial Ports (McASPs)
  - Two Independent Clock Zones Each (1 TX and 1 RX)
  - Eight Serial Data Pins Per Port:
    Individually Assignable to any of the Clock Zones
  - Each Clock Zone Includes:
    - Programmable Clock Generator
    - Programmable Frame Sync Generator
    - TDM Streams From 2-32 Time Slots
    - Support for Slot Size:
      8, 12, 16, 20, 24, 28, 32 Bits
    - Data Formatter for Bit Manipulation
  - Wide Variety of I2S and Similar Bit Stream Formats
  - Integrated Digital Audio Interface Transmitter (DIT) Supports:
    - S/PDIF, IEC60958-1, AES-3, CP-430 Formats
    - Up to 16 transmit pins
    - Enhanced Channel Status/User Data
  - Extensive Error Checking and Recovery
- Two Inter-Integrated Circuit Bus (I2C Bus™) Multi-Master and Slave Interfaces
- Two Multichannel Buffered Serial Ports:
  - Serial-Peripheral-Interface (SPI)
  - High-Speed TDM Interface
  - AC97 Interface
- Two 32-Bit General-Purpose Timers
- Dedicated GPIO Module With 16 pins (External Interrupt Capable)
- Flexible Phase-Locked-Loop (PLL) Based Clock Generator Module
- IEEE-1149.1 (JTAG †) Boundary-Scan-Compatible
- Package Options:
  - 208-Pin PowerPAD™ Plastic (Low-Profile) Quad Flatpack (PYP)
  - 272-BGA Packages (GDP and ZDP)
- 0.13-µm/6-Level Copper Metal Process
  - CMOS Technology
- 3.3-V I/Os, 1.2-V Internal (GDP & PYP)
- 3.3-V I/Os, 1.4-V Internal (GDP)(300 MHz only)



TMS320C6713 DSK Overview Block Diagram [Courtesy: Texas Instrument]



DSK 6713 Peripheral Description [Courtesy dsprelated.com]

Experiment -13

Aim: To verify Linear Convolution using TMS320C6713 DSK

Equipments Required: PC, TMS320C6713 DSK, Code composer Studiov3.0

Theory: These operations can be represented by a Mathematical Expression as follows:

$$y[n] = \sum_{k=-\infty}^{\infty} x[k].h[n-k]$$

x[ ]= Input signal Samples
h[ ]= Impulse response co-efficient.
y[ ]= Convolution output.
n = No. of Input samples
h = No. of Impulse response co-efficient.

Algorithm:

Algorithm to implement 'C' program for Convolution:

Eg: **x[n] = {1, 2, 3, 4}**
h[k] = {1, 2, 3, 4}

Where: n=4, k=4. ;Values of n & k should be a multiple of 4.
If n & k are not multiples of 4, pad with zero's to make multiples of 4
r= n+k-1 ; Size of output sequence.
= 4+4-1
= 7.

r=	0	1	2	3	4	5	6
n= 0	x[0]h[0]	x[0]h[1]	x[0]h[2]	x[0]h[3]			
1		x[1]h[0]	x[1]h[1]	x[1]h[2]	x[1]h[3]		
2			x[2]h[0]	x[2]h[1]	x[2]h[2]	x[2]h[3]	
3				x[3]h[0]	x[3]h[1]	x[3]h[2]	x[3]h[3]

Output: **y[n] = { 1, 4, 10, 20, 25, 24, 16}.**

Procedure:

1) Open Code Composer Studio; make sure the DSP kit is turned on.
2) Start a new project using 'Project-new ' pull down menu, save it in a separate directory(c:\ti\myprojects) with name lconv.pjt.
3) Add the source files conv.c to the project using 'Project→add files to project' pull down menu.
4) Add the linker command file hello.cmd .
(Path: c:\ti\tutorial\dsk6713\hello1\hello.cmd)
5) Add the run time support library file rts6700.lib
(Path: c:\ti\c6000\cgtools\lib\rts6700.lib)
6) Compile the program using the 'Project-compile' pull down menu or by clicking the shortcut icon on the left side of program window.

7) Build the program using the 'Project-Build' pull down menu or by clicking the shortcut icon on the left side of program window.
8) Load the program (lconv.out) in program memory of DSP chip using the 'File-load program' pull down menu.
9) To View output graphically
   Select view → graph → time and frequency.

Program:

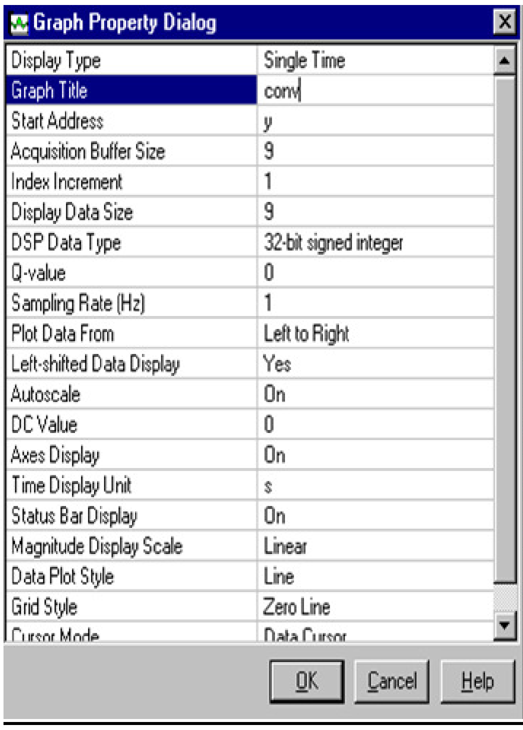
```
/* program to implement linear convolution */

#include<stdio.h>
#define LENGHT1 6   /*Lenght of i/p samples sequence*/
#define LENGHT2 4   /*Lenght of impulse response Co-
efficients */

int x[2*LENGHT1-1]={1,2,3,4,5,6,0,0,0,0,0};  /*Input
Signal Samples*/
int h[2*LENGHT1-1]={1,2,3,4,0,0,0,0,0,0,0};  /*Impulse
Response Co-efficients*/

int y[LENGHT1+LENGHT2-1];

main()
{
     int i=0,j;

     for(i=0;i<(LENGHT1+LENGHT2-1);i++)
     {
     y[i]=0;
     for(j=0;j<=i;j++)

          y[i]+=x[j]*h[i-j];

     }
     for(i=0;i<(LENGHT1+LENGHT2-1);i++)
     printf("%d\n",y[i]);

}
```

Results:

Thus, the Linear Convolution of two given discrete sequence has been performed. The input sequences are given in the program and the output will be displayed in the CCS software.

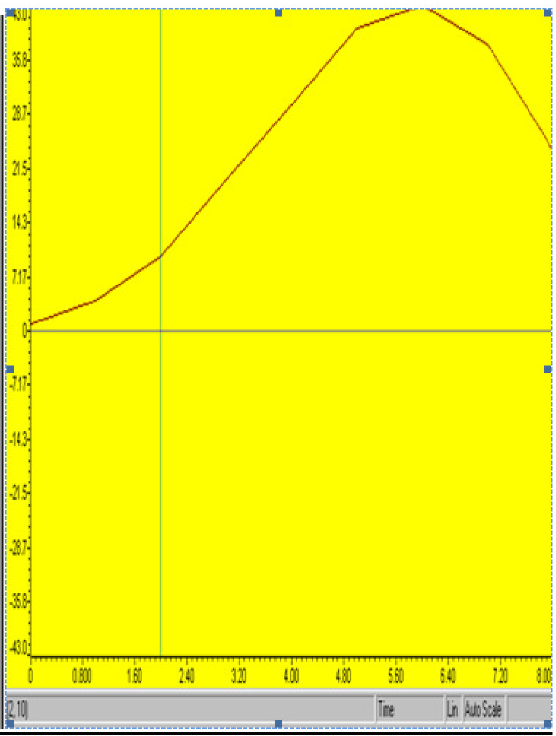
Input	**x[n] = {1, 2, 3, 4}**
	h[n] = {1, 2, 3, 4}
Output:	**y[n] = { 1, 4, 10, 20, 25, 24, 16}.**

Now configure the Graphical window as shown below



Graph Property Dialog

Display Type	Single Time
Graph Title	conv
Start Address	y
Acquisition Buffer Size	9
Index Increment	1
Display Data Size	9
DSP Data Type	32-bit signed integer
Q-value	0
Sampling Rate (Hz)	1
Plot Data From	Left to Right
Left-shifted Data Display	Yes
Autoscale	On
DC Value	0
Axes Display	On
Time Display Unit	s
Status Bar Display	On
Magnitude Display Scale	Linear
Data Plot Style	Line
Grid Style	Zero Line
Cursor Mode	Data Cursor



Discussions on results:

Thus we have verified the Linear Convolution in Code composer studio environment by writing a C program.

From the results students will be able to

1) Discuss the steps required to interface TMS320C6713 Kit with Code composer studio environment.
2) Discuss the changes in the program to get the input sequences from user.
3) Discuss the steps required in graphical property dialog of the Code composer studio for graphical visualization of linear convolution output

Experiment -14

Aim: Generation of Sine wave and square wave using TMS320C6713 DSK and Code Composer Studio.

Equipments Required: PC, TMS320C6713 DSK, Code composer Studiov3.0

Theory:

Sinusoidal Wave: The sine wave or sinusoidal wave is a mathematical curve that describes smooth repetitive oscillations. It can be represented in mathematical form as

$$y(t) = A\sin t(\omega t + \phi)$$

Where

A=Amplitude in V.

ω=angular frequency

ϕ=Phase in radins.

Square Wave: The square wave is a non sinusoidal periodic wave form which is represented as infinite summation of sinusoidal waves in which the amplitude alternates at a steady frequency between fixed minimum and maximum values with the same duration at maximum and minimum.

Algorithm:

1) Define frequency in C program.
2) Generate the signals using corresponding general formula.
3) Plot the graph in Code Composer Studio.

Procedure for Sinusoidal wave form generation

1) Open Code Composer Studio; make sure the DSP kit is turned on.
2) Start a new project using 'Project-new ' pull down menu, save it in a separate directory(c:\ti\myprojects) with name lconv.pjt.
3) Add the source files sinewave.c to the project using 'Project→add files to project' pull down menu.
4) Add the linker command file hello.cmd .
   (Path: c:\ti\tutorial\dsk6713\hello1\hello.cmd)
5) Add the run time support library file rts6700.lib
   (Path: c:\ti\c6000\cgtools\lib\rts6700.lib)
6) Compile the program using the 'Project-compile' pull down menu or by clicking the shortcut icon on the left side of program window.
7) Build the program using the 'Project-Build' pull down menu or by clicking the shortcut icon on the left side of program window.
8) Load the program(sinewave.out) in program memory of DSP chip using the 'File-load program' pull down menu.
9) To View output graphically
   Select view → graph → time and frequency.

Program:

```
/* program for sinewave generation */

#include<stdio.h>
#include<math.h>

#define freq 400
 float m[128];

main()
{
     int n=0;
     for(n=0;n<127;n++)
     {
        m[n]=sin(2*3.14*freq*n/24000)
        printf("%f",m[n]);
     }

}
```

Procedure for Square waveform generation

1) Open Code Composer Studio; make sure the DSP kit is turned on.
2) Start a new project using 'Project-new ' pull down menu, save it in a separate directory(c:\ti\myprojects) with name lconv.pjt.
3) Add the source files squarewave.c to the project using 'Project→add files to project' pull down menu.
4) Add the linker command file hello.cmd .
   (Path: c:\ti\tutorial\dsk6713\hello1\hello.cmd)
5) Add the run time support library file rts6700.lib
   (Path: c:\ti\c6000\cgtools\lib\rts6700.lib)
6) Compile the program using the 'Project-compile' pull down menu or by clicking the shortcut icon on the left side of program window.
7) Build the program using the 'Project-Build' pull down menu or by clicking the shortcut icon on the left side of program window.
8) Load the program(squarewave.out) in program memory of DSP chip using the 'File-load program' pull down menu.
9) To View output graphically
   Select view → graph → time and frequency.

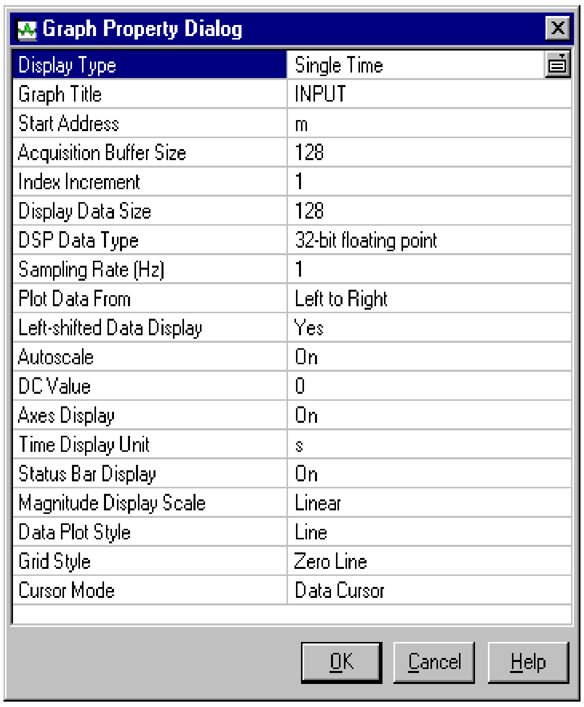
Program for Square wave Generation

```
/* program for Squarewave generation */

#include<stdio.h>
#include<math.h>

#define freq 500
 float m[81];

main()
{
     int  n=0;
     for(n=0;n<21;n++)
     {
        m[n]=5.0
     }
     for(n=21;n<41;n++)
     {
        m[n]=-5.0
     }
     for(n=41;n<61;n++)
     {
        m[n]=5.0
     }
     for(n=61;n<81;n++)
     {
        m[n]=-5.0
     }
}
```

Results:

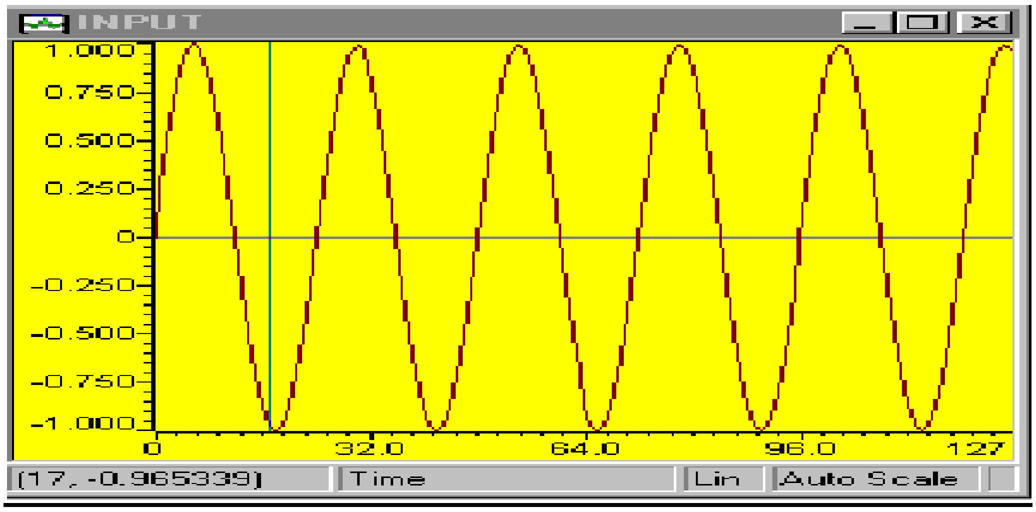
Thus, the waveform generation on sine wave and square wave is performed in Code composer environment by using a C program. Output will be displayed in the graphical window of CCS software**.**

Now configure the graphical window as shown below for Sine wave generation .the same procedure can be followed for square wave generation.

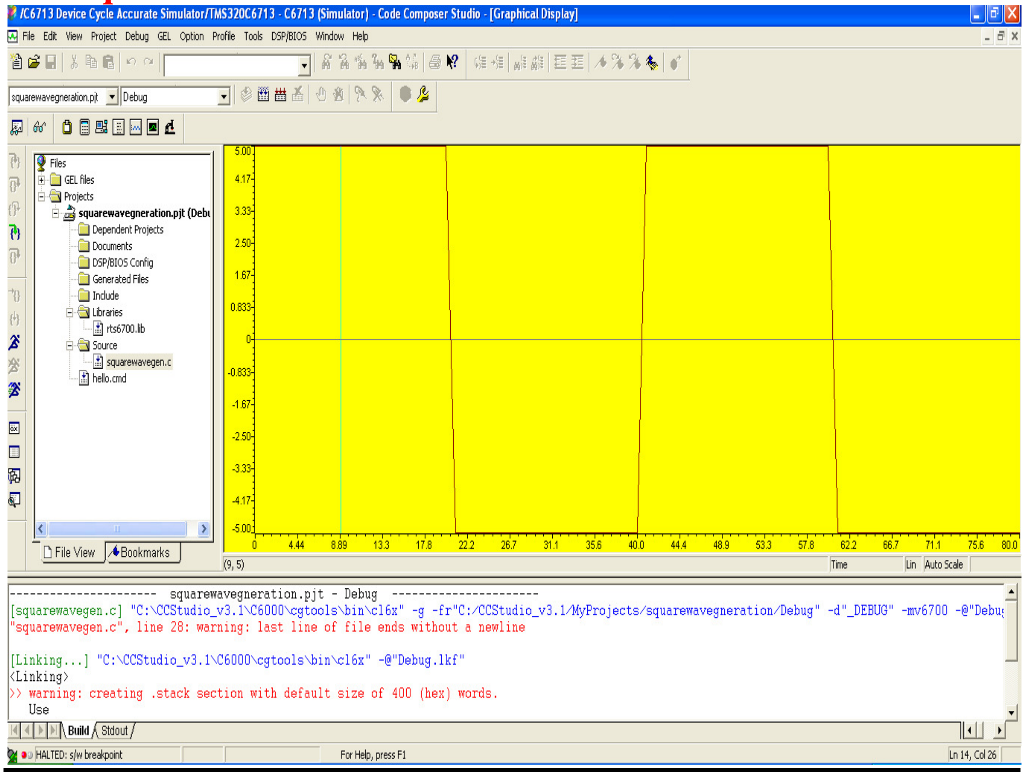


Display Type	Single Time
Graph Title	INPUT
Start Address	m
Acquisition Buffer Size	128
Index Increment	1
Display Data Size	128
DSP Data Type	32-bit floating point
Sampling Rate (Hz)	1
Plot Data From	Left to Right
Left-shifted Data Display	Yes
Autoscale	On
DC Value	0
Axes Display	On
Time Display Unit	s
Status Bar Display	On
Magnitude Display Scale	Linear
Data Plot Style	Line
Grid Style	Zero Line
Cursor Mode	Data Cursor

Plot of sine waveform



Plot of Square wave form



```
--------------------  squarewavegneration.pjt - Debug  ---------------------
[squarewavegen.c] "C:\CCStudio_v3.1\C6000\cgtools\bin\cl6x" -g -fr"C:/CCStudio_v3.1/MyProjects/squarewavegneration/Debug" -d"_DEBUG" -mv6700 -@"Debu
"squarewavegen.c", line 28: warning: last line of file ends without a newline

[Linking...] "C:\CCStudio_v3.1\C6000\cgtools\bin\cl6x" -@"Debug.lkf"
<Linking>
>> warning: creating .stack section with default size of 400 (hex) words.
   Use
```

Discussion on Results:

Thus we have performed the waveform generation in Code composer studio environment by writing a C program.

From the results the student will be able to

1) Discuss the steps required to interface TMS320C6713 Kit with Code composer studio environment.
2) Discuss the changes in the program to get the input sequences from user.
3) Discuss the steps required in graphical property dialog of the Code composer studio for graphical visualization of the sine wave and square wave output.

Experiment -15

Aim: Generating the Responses of Low Pass and High Pass filters using DSP Trainer Kit (TMS320C6713)

Equipments Required: PC Host (PC) with windows (95/98/Me/XP/NT/2000), TMS320C6713 DSP Starter Kit (DSK).Oscilloscope and Function generator, Code Composer Studio v3.0

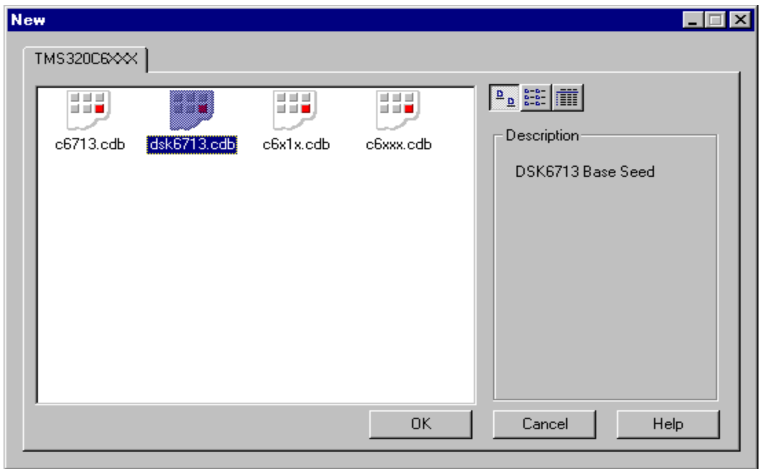
Algorithm:

We need to realize the Butter worth band pass IIR filter by implementing the difference equation y[n] = b0x[n] + b1x[n-1]+b2x[n-2]-a1y[n-1]-a2y[n-2] where b0 – b2, a0-a2 are feed forward and feedback word coefficients respectively [Assume 2nd order of filter].These coefficients are calculated using MATLAB.A direct form I implementation approach is taken.

1) Initialize the McBSP, the DSP board and the on board codec. “Kindly refer the Topic Configuration of 6713Codec using BSL”
2) Initialize the discrete time system, that is , specify the initial conditions. Generally zero initial conditions are assumed.
3) Take sampled data from codec while input is fed to DSP kit from the signal generator. Since Codec is stereo, take average of input data read from left and right channel. Store sampled data at a memory location.
4) Perform filter operation using above said difference equation and store filter Output at a memory location.
5) Output the value to codec (left channel and right channel) and view the output at Oscilloscope.
6) Step 6 - Go to step 3.

Procedure for Real time Programs:

1. Connect CRO to the Socket Provided for LINE OUT.
2. Connect a Signal Generator to the LINE IN Socket.
3. Switch on the Signal Generator with a sine wave of frequency 500 Hz. and Vp-p=1.5v
4. Now Switch on the DSK and Bring Up Code Composer Studio on the PC.
5. Create a new project with name codec.pjt.
6. From the File Menu ➔ new ➔ DSP/BIOS Configuration ➔select “dsk6713.cdb” and save it as “xyz.cdb”



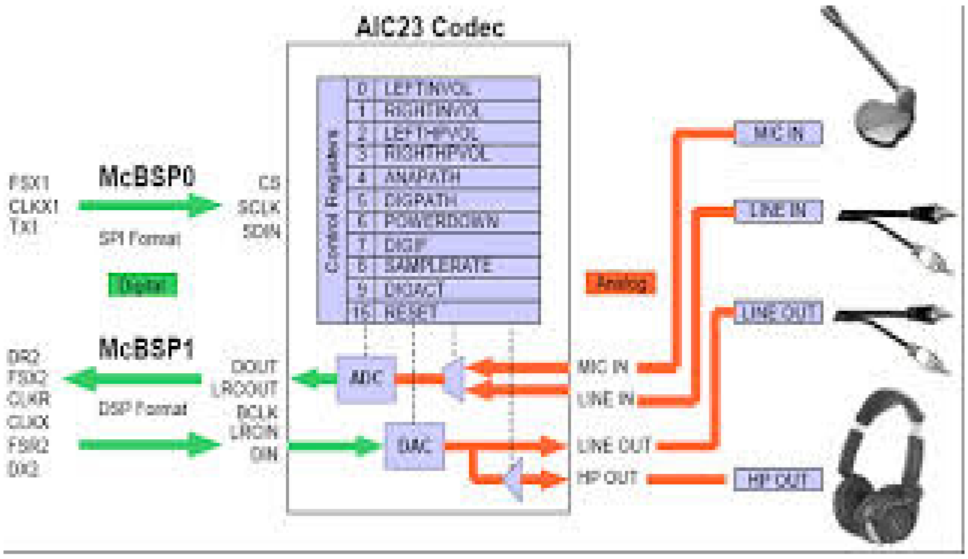
DSK6713 Base Seed

7. Add “xyz.cdb” to the current project.
8. Add the given “codec.c” file to the current project which has the main function and calls all the other necessary routines.
9. Add the library file “dsk6713bsl.lib” to the current project
   Path → “C:\CCStudio\C6000\dsk6713\lib\dsk6713bsl.lib”
10. Copy files “dsk6713.h” and “dsk6713_aic23.h” from C:\CCStudio\C6000\dsk6713\include and paste it in current project.
11. Build, Load and Run the program.
12. You can notice the input signal of 500 Hz. appearing on the CRO verifying the codec configuration.
13. You can also pass an audio input and hear the output signal through the speakers.
14. You can also vary the sampling frequency using the DSK6713_AIC23_setFreq Function in the “codec.c” file and repeat the above steps.

Procedure to execute IIR Filter Program

1) Switch on the DSP board.
2) Open the Code Composer Studio.
3) Create a new project
   Project → New (File Name. pjt , Eg: IIR.pjt)
4) Initialize on board codec.
5) Add the given above ‘C’ source file to the current project (remove codec.c source file from the project if you have already added).
6) Connect the speaker jack to the input of the CRO.
7) Build the program.
8) Load the generated object file (*.out) on to Target board.
9) Run the program
10) Observe the waveform that appears on the CRO screen.
11) Vary the frequency on function generator to see the response of filter.

AIC23 Codec Connections



[Courtesy Texas Instruments]

Program:

```
#include "xyzcfg.h"
#include "dsk6713.h"
#include "dsk6713_aic23.h"

const signed int filter_Coeff[] =
{
     //12730,-12730,12730,2767,-18324,21137 /*HP 2500 */
     //312,312,312,32767,-27943,24367       /*LP 800 */
     //1455,1455,1455,32767,-23140,21735  /*LP 2500 */
     //9268,-9268,9268,32767,-7395,18367  /*HP 4000*/
       7215,-7215,7215,32767,5039,6171,         /*HP 7000*/
} ;

/* Codec configuration settings */
DSK6713_AIC23_Config config = { \
    0x0017,  /* 0 DSK6713_AIC23_LEFTINVOL  Left line input channel volume */
\
    0x0017,  /* 1 DSK6713_AIC23_RIGHTINVOL Right line input channel volume
*/\
    0x00d8,  /* 2 DSK6713_AIC23_LEFTHPVOL  Left channel headphone volume */
\
    0x00d8,  /* 3 DSK6713_AIC23_RIGHTHPVOL Right channel headphone volume */
\
    0x0011,  /* 4 DSK6713_AIC23_ANAPATH    Analog audio path control */
\

    0x0000,  /* 5 DSK6713_AIC23_DIGPATH    Digital audio path control */
\
    0x0000,  /* 6 DSK6713_AIC23_POWERDOWN  Power down control */
\
    0x0043,  /* 7 DSK6713_AIC23_DIGIF      Digital audio interface format */
\
    0x0081,  /* 8 DSK6713_AIC23_SAMPLERATE Sample rate control */
\
    0x0001   /* 9 DSK6713_AIC23_DIGACT     Digital interface activation */
\
};

/*
 *  main() - Main code routine, initializes BSL and generates tone
 */
void main()
{
    DSK6713_AIC23_CodecHandle hCodec;

    int l_input, r_input, l_output, r_output;

    /* Initialize the board support library, must be called first */
    DSK6713_init();

    /* Start the codec */
    hCodec = DSK6713_AIC23_openCodec(0, &config);

    DSK6713_AIC23_setFreq(hCodec, 3);

       while(1)
        {           /* Read a sample to the left channel */
             while (!DSK6713_AIC23_read(hCodec, &l_input));

             /* Read a sample to the right channel */
             while (!DSK6713_AIC23_read(hCodec, &r_input));

                    l_output=IIR_FILTER(&filter_Coeff ,l_input);
                    r_output=l_output;

             /* Send a sample to the left channel */
             while (!DSK6713_AIC23_write(hCodec, l_output));

             /* Send a sample to the right channel */
             while (!DSK6713_AIC23_write(hCodec, r_output));
        }

    /* Close the codec */
    DSK6713_AIC23_closeCodec(hCodec);
}

signed int IIR_FILTER(const signed int * h, signed int x1)
{
      static signed int x[6] = { 0, 0, 0, 0, 0, 0 };  /* x(n), x(n-1), x(n-
2). Must be static */
      static signed int y[6] = { 0, 0, 0, 0, 0, 0 };  /* y(n), y(n-1), y(n-
2). Must be static */
       int temp=0;

       temp = (short int)x1; /* Copy input to temp */

      x[0] = (signed int) temp; /* Copy input to x[stages][0] */

      temp =  ( (int)h[0] * x[0]) ;   /* B0 * x(n)   */

      temp += ( (int)h[1] * x[1]);    /* B1/2 * x(n-1) */
      temp += ( (int)h[1] * x[1]);    /* B1/2 * x(n-1) */
      temp += ( (int)h[2] * x[2]);    /* B2 * x(n-2) */

      temp -= ( (int)h[4] * y[1]);    /* A1/2 * y(n-1) */
      temp -= ( (int)h[4] * y[1]);    /* A1/2 * y(n-1) */

      temp -= ( (int)h[5] * y[2]);   /* A2 * y(n-2) */


    /* Divide temp by coefficients[A0] */

       temp >>= 15;

    if ( temp > 32767 )
      {
        temp = 32767;
      }
     else if ( temp < -32767)
      {
        temp = -32767;
      }
 y[0] =  temp ;

     /* Shuffle values along one place for next time */

     y[2] = y[1];   /* y(n-2) = y(n-1) */
     y[1] = y[0];   /* y(n-1) = y(n)   */

     x[2] = x[1];   /* x(n-2) = x(n-1) */
     x[1] = x[0];   /* x(n-1) = x(n)   */

     /* temp is used as input next time through */

  return (temp<<2);
}
```

Results:

Thus the result can be observer on the CRO for various frequencies.

Discussions on results:

The designing of IIR Low pass and High Pass Filters requires initialization of BSL codec.

From the results students will be able to

1) Discuss the real time interfacing of the TMS320C6713 Kit by using the full functionality of the board support library files and AIC23 Codec.
2) Discuss the effect of changing the value of coefficients for filters.
3) Discuss the steps required to do the connection of CRO and function generator to TMS320C6713 Kit.

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