SRM VALLIAMMAI ENGINEERING COLLEGE

(An Autonomous Institution)

SRM Nagar, Kattankulathur – 603 203

DEPARTMENT OF MEDICAL ELECTRONICS

LAB MANUAL

Regulation – 2023

MD3565 DIGITAL SIGNAL PROCESSING LABORATORY

ACADEMIC YEAR: 2025 - 2026 (ODD SEM)

III YEAR MDE
V SEMESTER

Prepared by
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SRM VALLIAMMAI ENGINEERING COLLEGE

(An Autonomous Institution)
SRM Nagar, Kattankulathur – 603 203

DEPARTMENT OF MEDICAL ELECTRONICS

VISION OF THE INSTITUTE

Educate to excel in social transformation

MISSION OF THE INSTITUTE

- To contribute to the development of human resources in the form of professional engineers and managers of international excellence and competence with high motivation and dynamism, who besides serving as ideal citizen of our country will contribute substantially to the economic development and advancement in their chosen areas of specialization.
- To build the institution with international repute in education in several areas at several levels with specific emphasis to promote higher education and research through strong institute-industry interaction and consultancy.

VISION OF THE DEPARTMENT

- To provide quality education for improving the healthcare and well-being of humankind.

MISSION OF THE DEPARTMENT

- **M1:** To inculcate students with fundamental knowledge, interdisciplinary problem solving skills and confidence required to excel in Medical Electronics
- **M2:**. To up skill the students with the current technological trends and carryout quality research to meet the expectation of healthcare service sectors.
- **M3:** To instil creativity, responsibility, commitment and leadership qualities with professional ethics and moral values.

SRM VALLIAMMAI ENGINEERING COLLEGE

(An Autonomous Institution)
SRM Nagar, Kattankulathur – 603 203

PROGRAM OUTCOMES

1. **Engineering knowledge:** Apply the knowledge of mathematics, science, engineering fundamentals, and an engineering specialization to the solution of complex engineering problems.
2. **Problem analysis:** Identify, formulate, review research literature, and analyze complex engineering problems reaching substantiated conclusions using first principles of mathematics, natural sciences, and engineering sciences.
3. **Design/development of solutions:** Design solutions for complex engineering problems and design system components or processes that meet the specified needs with appropriate consideration for the public health and safety, and the cultural, societal, and environmental considerations.
4. **Conduct investigations of complex problems:** Use research-based knowledge and research methods including design of experiments, analysis and interpretation of data, and synthesis of the information to provide valid conclusions.
5. **Modern tool usage:** Create, select, and apply appropriate techniques, resources, and modern engineering and IT tools including prediction and modeling to complex engineering activities with an understanding of the limitations.
6. **The engineer and society:** Apply reasoning informed by the contextual knowledge to assess societal, health, safety, legal and cultural issues and the consequent responsibilities relevant to the professional engineering practice.
7. **Environment and sustainability:** Understand the impact of the professional engineering solutions in societal and environmental contexts, and demonstrate the knowledge of, and need for sustainable development.
8. **Ethics:** Apply ethical principles and commit to professional ethics and responsibilities and norms of the engineering practice.
9. **Individual and team work:** Function effectively as an individual, and as a member or leader in diverse teams, and in multidisciplinary settings.
10. **Communication:** Communicate effectively on complex engineering activities with the engineering community and with society at large, such as, being able to comprehend and write effective reports and design documentation, make effective presentations, and give and receive clear instructions.
11. **Project management and finance:** Demonstrate knowledge and understanding of the engineering and management principles and apply these to one's own work, as a member and leader in a team, to manage projects and in multidisciplinary environments.
12. **Life-long learning:** Recognize the need for, and have the preparation and ability to engage in independent and life-long learning in the broadest context of technological change.

PROGRAM SPECIFIC OUTCOME(PSOs)

PSO1: Ability to apply the acquired knowledge of basic skills, mathematical foundations, principles of electronics, modeling and design of electronics based systems in solving engineering Problems.

PSO2: Ability to understand and analyze the interdisciplinary problems for developing innovative sustained solutions with environmental concerns.

PSO3: Ability to update knowledge continuously in the tools like MATLAB, NS2, XILINIX and technologies like VLSI, Embedded, Wireless Communications to meet the industry requirements.

PSO4: Ability to manage effectively as part of a team with professional behavior and ethics.

SYLLABUS

OBJECTIVES:

The student should be made:

- To understand the fundamentals of discrete-time signal processing and its applications.
- To analyze and implement signal processing operations like convolution, correlation, and frequency domain analysis.
- To design and implement digital filters (FIR and IIR) for signal filtering and processing.
- To explore and practice signal sampling techniques like up-sampling and down-sampling.
- To gain practical knowledge of DSP processor architecture and its addressing modes.

LIST OF EXPERIMENTS: MATLAB / EQUIVALENT SOFTWARE PACKAGE

1. Generation of elementary Discrete-Time sequences.
2. Linear and Circular convolutions.
3. Auto correlation and Cross Correlation.
4. Frequency Analysis using DFT.
5. Design of FIR filters (LPF/HPF/BPF/BSF) and demonstrates the filtering operation.
6. Design of Butterworth IIR filters (LPF/HPF/BPF/BSF) and demonstrate the filtering operations.
7. Implement an Up-sampling and Down-sampling operation.

DSP PROCESSOR BASED IMPLEMENTATION

1. Study of architecture of Digital Signal Processor.
2. Perform MAC operation using various addressing modes.
3. Generation of various signals and random noise.
4. Design and demonstration of FIR Filter for Low pass, High pass, Band pass and

Band stop filtering.

5. Design and demonstration of Butter worth IIR Filters for Low pass, High pass, Band pass and Band stop filtering.

TOTAL PERIODS: 45

COURSE OUTCOMES:

At the end of the course, the student should be able to:

CO1: Generate and visualize elementary discrete-time sequences such as impulse, step, and sinusoidal signals.

CO2: Analyze signal properties using auto-correlation and cross-correlation techniques.

CO3: Design and implement FIR and IIR filters for various applications like low-pass, high-pass, band-pass, and band-stop filtering.

CO4: Perform up-sampling and down-sampling operations for multi-rate signal processing.

CO5: Understand the architecture and functionalities of DSP processors for real time signal processing applications.

LIST OF EXPERIMENTS

CYCLE I

1. Generation of elementary Discrete-Time sequences.
2. Linear and Circular convolutions.
3. Auto correlation and Cross Correlation.
4. Frequency Analysis using DFT.
5. Design of FIR filters (LPF/HPF/BPF/BSF) and demonstrates the filtering operation.
6. Design of Butterworth IIR filters (LPF/HPF/BPF/BSF) and demonstrate the filtering operations.
7. Implement an Up-sampling and Down-sampling operation.

CYCLE II

8. Study of architecture of Digital Signal Processor.

9. Perform MAC operation using various addressing modes.

10. Generation of various signals and random noise.

11. Design and demonstration of FIR Filter for Low pass, High pass, Band pass and Band stop filtering.

12. Design and demonstration of Butter worth IIR Filters for Low pass, High pass, Band pass and Band stop filtering.

<u>TOPIC BEYOND SYLLABUS</u>

13. Study of Image processing tool box.

14. Generation of Dual Tone Multi Frequency Signals (DTMF) using DFT.

Ex No: **Date:**

1. Generation of sequences (Functional &Random) & correlation

Aim:

To write a MATLAB program to generate unit step, unit impulse, unit ramp, exponential, sine waveform, cosine waveform, random signal and correlation.

Prerequisite Lab Questions:

1. Define Continuous Time and Discrete Time Signal?
2. How to represent Unit step, impulse and ramp function?
3. Compare stem & Plot functions.
4. Define Energy & Power signals.
5. Give an example for real time signals.

Description:

Input : Enter the sequence length for the generation of signals.

(a)The unit sample sequence (Unit Impulse) is

$\delta(n) = 1$ for $n = 0$

$= 0$ for $n \neq 0$

(b)The unit step sequence is

$u(n) = 1$ for $n \geq 0$

$= 0$ for $n < 0$

(c) The unit ramp sequence is denoted as $u_r(n)$ is

$u_r(n) = n$ for $n \geq 0$

$= 0$ for $n < 0$

(d)The exponential signal is a sequence of the form

$x(n) = e^n$ for all 'n'

(e)Sine signal $x(t) = \sin\omega t$, for all t values.

(f) Sine sequensce $x(n) = \sin\omega n$, for all n values.

(g)Cosine signal $x(t) = \cos\omega t$, for all t values.

(h)Cosine signal $x(t) = \cos\omega n$, for all n values.

Output: Plot of all the above mentioned signals.

Algorithm:

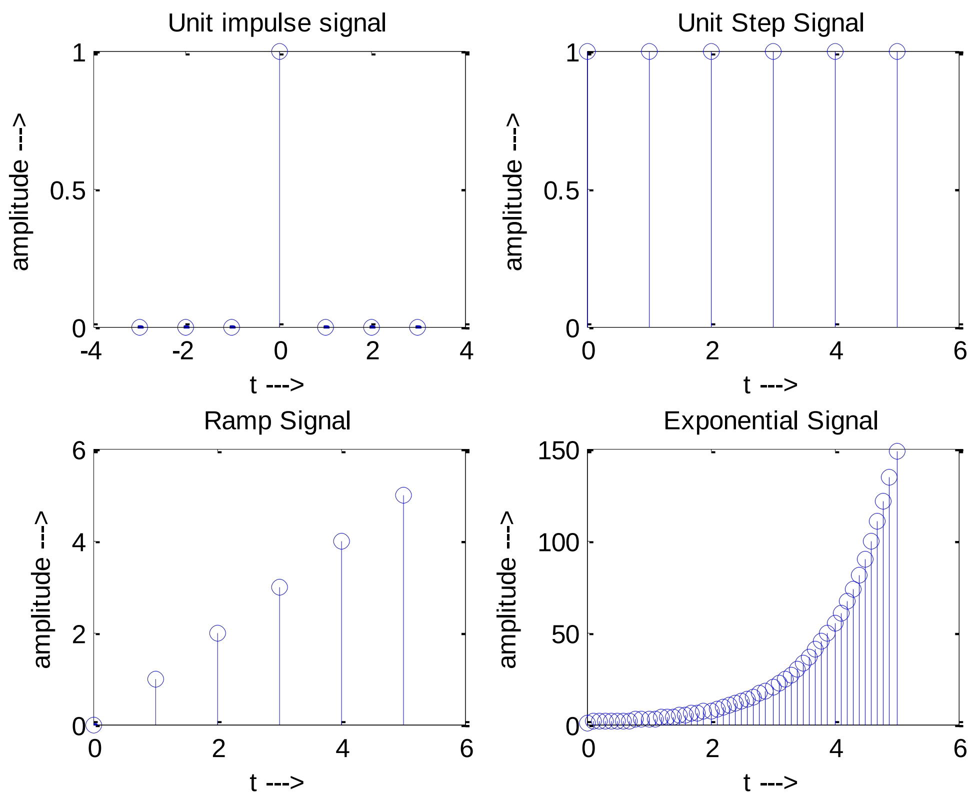
1. Start the program.
2. Generate unit step, unit impulse, unit ramp, exponential, sine waveform, cosine waveform, random signal and correlation.
3. Execute the program.
4. Plot the output for each sequence.
5. Stop the program.

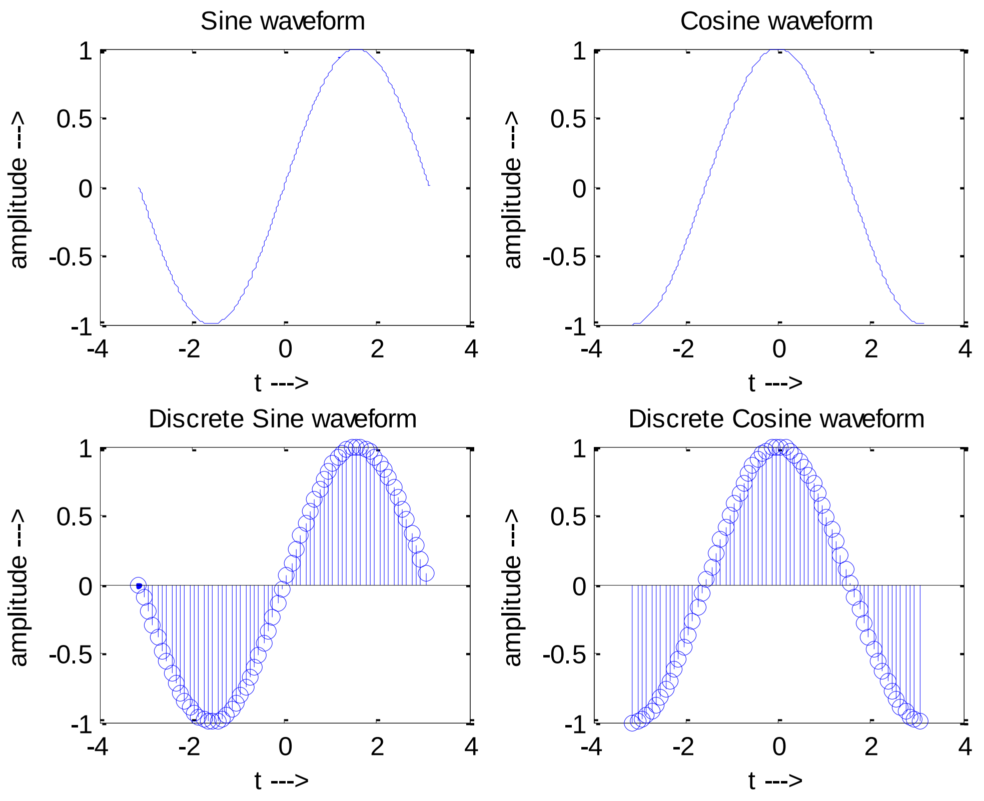
Program:

```
%Program for Unit Impulse Signal
clc;
clear all;
t=-3:1:3;
y=[zeros(1,3),ones(1,1),zeros(1,3)];
figure(1);
subplot(2,2,1);
stem(t,y);
ylabel('amplitude --->');
xlabel('t --->');
title('Unit impulse signal');

%Program for Unit Step Signal
t=0:5;
y=[ones(1,6)];
subplot(2,2,2);
stem(t,y);
ylabel('amplitude --->');
xlabel('t --->');
title('Unit Step Signal');

%Program for Ramp Signal
t=0:5;
y=t;
subplot(2,2,3);
stem(t,y);
ylabel('amplitude --->');
xlabel('t --->');
title('Ramp Signal');

%Program for Exponential Signal
t=0:0.1:5;
y=exp(t);
subplot(2,2,4);
stem(t,y);
ylabel('amplitude --->');
xlabel('t --->');
title('Exponential Signal');
figure(2);
%Program for Sine waveform
t=-pi:0.01:pi;
y=sin(t);
subplot(2,2,1);
plot(t,y);
```

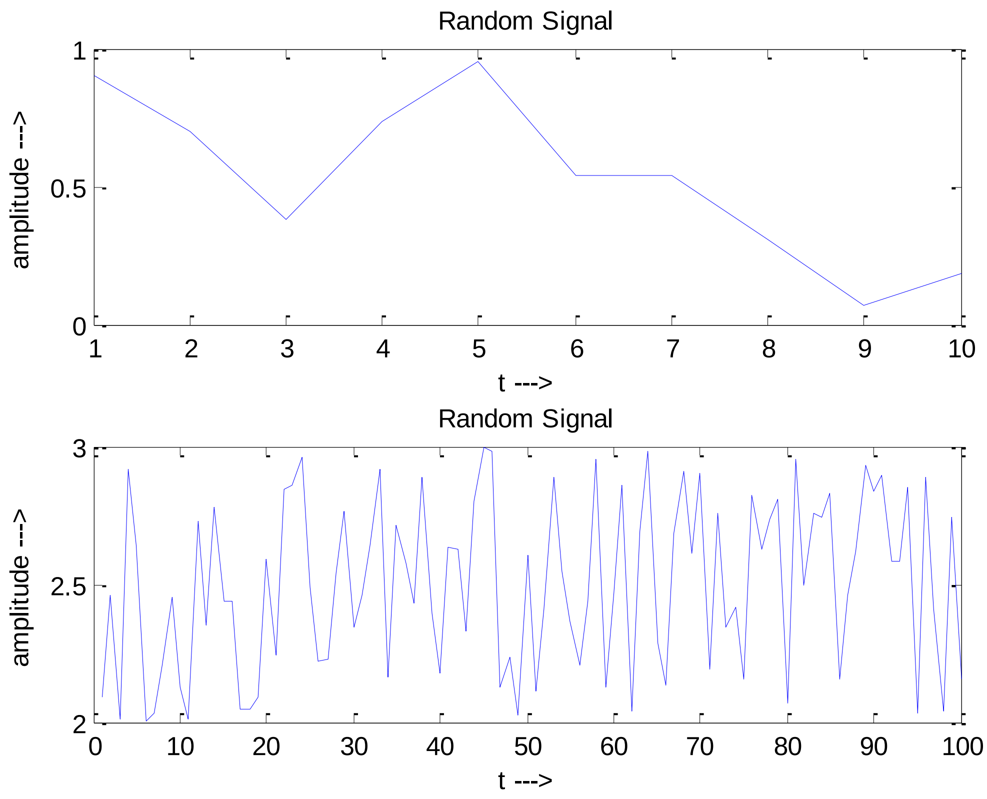
```
ylabel('amplitude --->');
xlabel('t --->');
title('Sine waveform');


%Program for Cosine waveform
t=-pi:0.01:pi;
y=cos(t);
subplot(2,2,2);
plot(t,y);
ylabel('amplitude --->');
xlabel('t --->');
title('Cosine waveform');

%Program for Discrete Sine waveform
t=-pi:0.1:pi;
y=sin(t);
subplot(2,2,3);
stem(t,y);
ylabel('amplitude --->');
xlabel('t --->');
title('Discrete Sine waveform');

%Program for Discrete Cosine waveform
t=-pi:0.1:pi;
y=cos(t);
subplot(2,2,4);
stem(t,y);
ylabel('amplitude --->');
xlabel('t --->');
title('Discrete Cosine waveform');
```

Model Graph:


Unit impulse signal
Unit Step Signal
Ramp Signal
Exponential Signal
amplitude --->
t --->


Sine waveform
Cosine waveform
Discrete Sine waveform
Discrete Cosine waveform
amplitude --->
t --->

```
%Program for Random Signal
t = rand(1,10);
subplot(2,1,1);
plot(t);
ylabel('amplitude --->');
xlabel('t --->');
title('Random Signal');
disp('t=');
disp(t);
a=2;
b=3;
t1 = a + (b-a).*rand(100,1);
subplot(2,1,2);
plot(t1)
ylabel('amplitude --->');
xlabel('t --->');
title('Random Signal');
disp('t1=');
disp(t1);
```



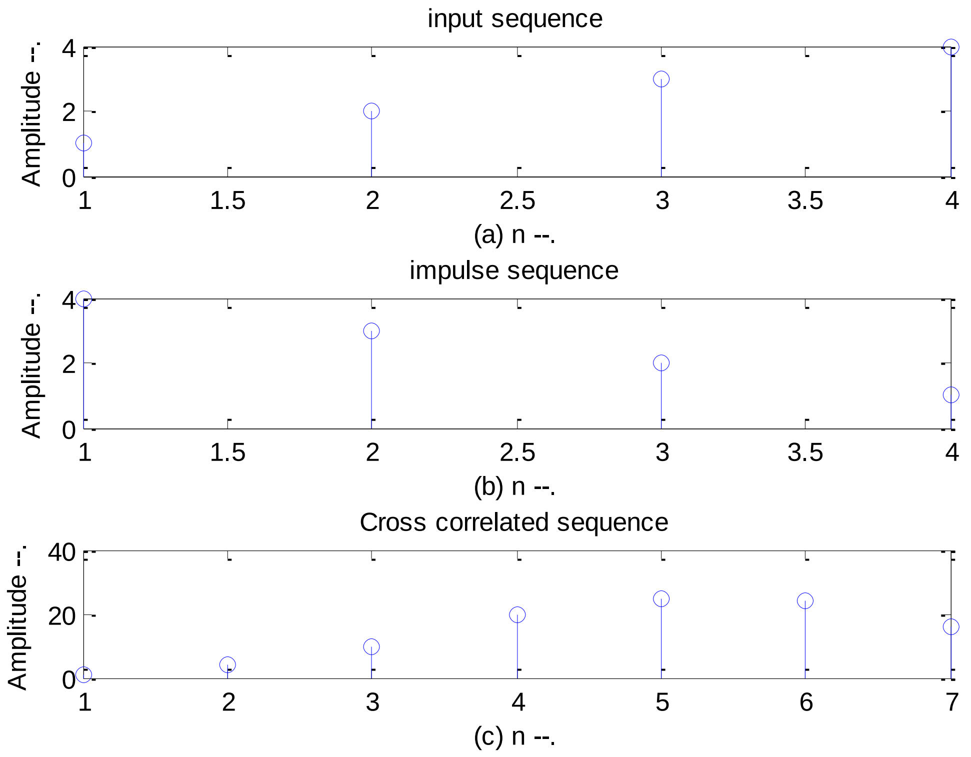
% Program for computing cross-correlation of the sequences x[1 2 3 4] and h[4 3 2 1]

```
clc;
clear all;
close all;
x=input('enter the 1st sequence');
h=input('enter the 2nd sequence');
y= xcorr(x,h);
subplot(3,1,1);
stem(x);
ylabel('Amplitude --.');
xlabel('(a) n --.');
title('input sequence');
subplot(3,1,2);
stem(h);
ylabel('Amplitude --.');
xlabel('(b) n --.');
title('impulse sequence');
subplot(3,1,3);
stem(y);
ylabel('Amplitude --.');
xlabel('(c) n --.');
title('Cross correlated sequence');
disp('The resultant signal is y=');
disp(y)
```

Output:

```
enter the 1st sequence[1 2 3 4]
enter the 2nd sequence[4 3 2 1]
The resultant signal is y=
   1.0000    4.0000   10.0000   20.0000   25.0000   24.0000   16.0000
```

Model Graph:

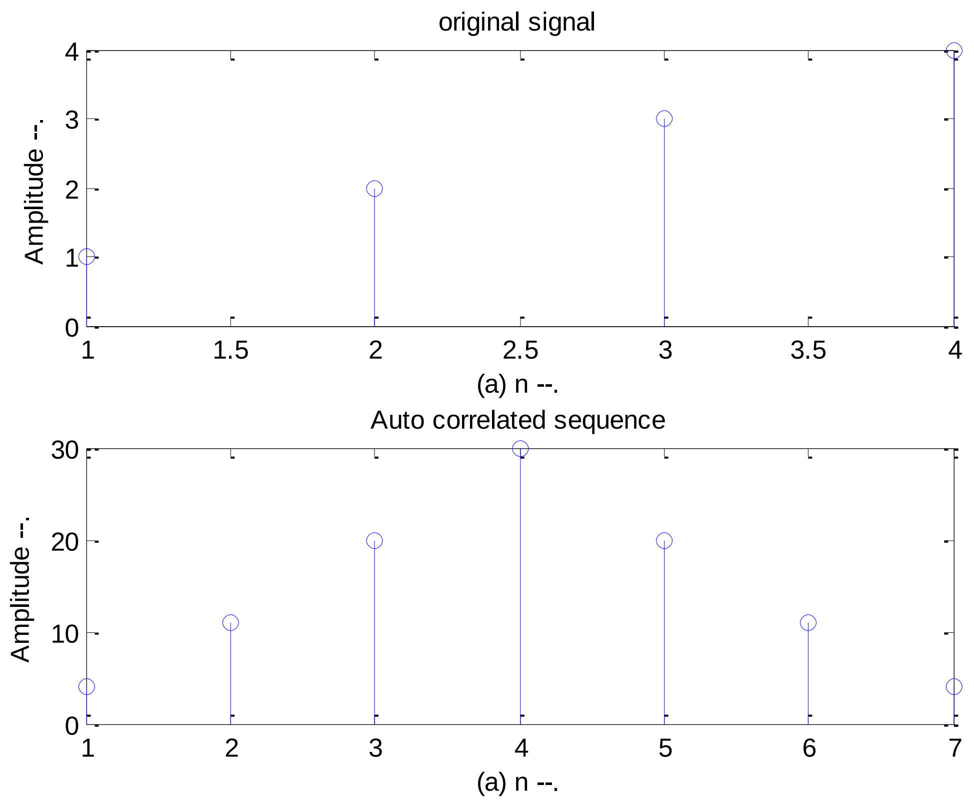


```
%Program for computing autocorrelation function x[1 2 3 4]
x=input('enter the sequence');
y=xcorr(x,x);
subplot(2,1,1);
stem(x);
ylabel('Amplitude --.');
xlabel('(a) n --.');
title('original signal');
subplot(2,1,2);
stem(y);
ylabel('Amplitude --.');
xlabel('(a) n --.');
title ('Auto correlated sequence');
disp('The resultant signal is y=');
disp(y)
```

Output:

```
enter the sequence[1 2 3 4]
The resultant signal is y=
   4.0000   11.0000   20.0000   30.0000   20.0000   11.0000    4.0000
```

Model Graph



<u>Postlab Questions:</u>

1. Modify Program (b) to generate a delayed unit step sequence u[n] with an advance of “n” samples. Run the modified program and display the sequence generated.
2. What is the subplot command used to plot 4 figures in 2 rows and 2 columns?
3. How can you compute the average power of the generated sinusoidal sequence?
4. Write the syntax to determine auto correlation of the given signal x[n].
5. Give the examples for 1-D, 2-D & 3-D signals.

Result:

Thus a program to generate sequence for unit step, unit impulse, unit ramp, exponential, sine, cosine and correlation signal using MATLAB was written and executed and the corresponding simulation output is plotted.

Ex No: **Date:**

2a.LINEAR CONVOLUTION OF TWO SEQUENCES

Aim:

To write a program to find out the linear convolution of two sequence using Matlab.

Prerequisite Lab Questions:

1. Define Convolution.
2. What is zero padding ?
3. List the properties of linear convolution.
4. What is the MATLAB function used to perform linear & circular convolution?
5. How will you determine the maximum length of the convolution of the sequences?

Description :

Input : Give the input x(n) and h(n) in matrix form.

Output : y(n), using the formula

$$y(n) = \sum_{k=-\infty}^{\infty} x(k)h(n-k)$$

Algorithm:

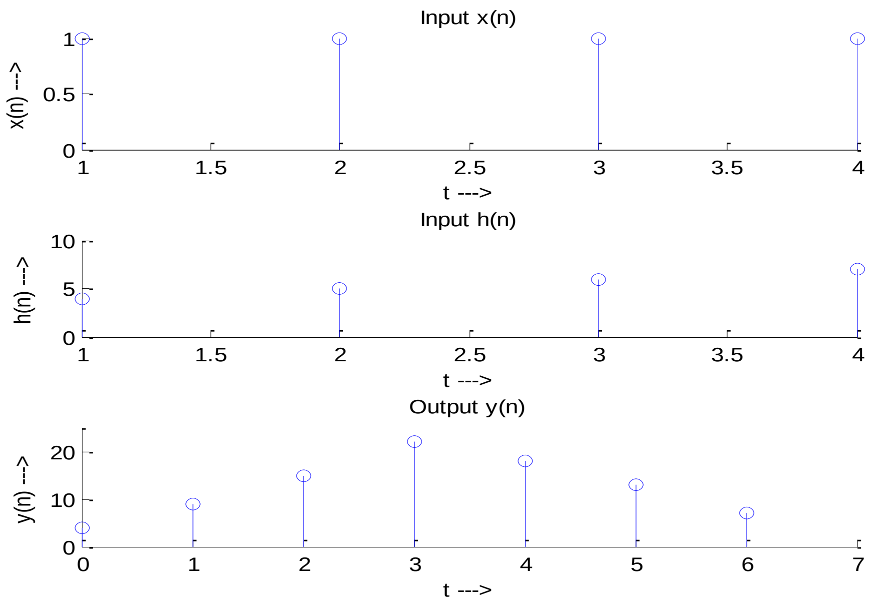
1. Start the program.
2. Get the two input sequence x(n) and h(n).
3. Flip the h vector to the right or left direction.
4. Calculate the length of the input and output sequence.
5. Append zeros to the input to get the same length as output.
6. Sum the product of x(n) and shifted sequence of h(n-k).
7. Execute the program, display the result and verify theoretically.
8. Plot the graph for the input and output sequence.
9. Stop the pogram.

Program:

```
% Program for Linear Convolution
clc;
clear all;
close all;
%x=[1 1 1 1];
%h=[4 5 6 7];
x=input('enter the value x=');
h=input('enter the value h=');
subplot(3,1,1);
stem(x);
title('Input x(n)');
ylabel('x(n) --->');
xlabel('t --->');
subplot(3,1,2);
stem(h);
```

```
title('Input h(n)');
ylabel('h(n) --->');
xlabel('t --->');
h1=fliplr(h);
n1=length(x);
n2=length(h);
n=n1+n2-1;
x=[zeros(1,n2-1),x];
h1=[h1,zeros(1,n1-1)];
for i=1:n
   new=[zeros(1,i-1),h1(1:n-i+1)];
out(i)=x*new';
end
disp('output=');
disp(out);
subplot(3,1,3);
t=0:1:n-1;
stem(t,out);
title('Output y(n)');
ylabel('y(n) --->');
xlabel('t --->');
```

MODEL GRAPH



THEORETICAL CALCULATION

$$y(n) = \sum_{k=-\infty}^{\infty} x(k)h(n-k)$$

Enter the value x=[1 1 1 1]

```
Enter the value h=[4 5 6 7]
Output= 4     9    15    22    18    13     7
```

EX NO: 2b CIRCULAR CONVOLUTION

Aim:

To write a program to find out the circular convolution of two sequences using MATLAB.

Description:

Input : Give the input x(n) and h(n) in matrix form.

Output : y(n), using the formula

$y(n) = x(n) \otimes h(n)$

$$y(n) = \sum_{k=0}^{N-1} x(k)h(n-k)$$

Algorithm:

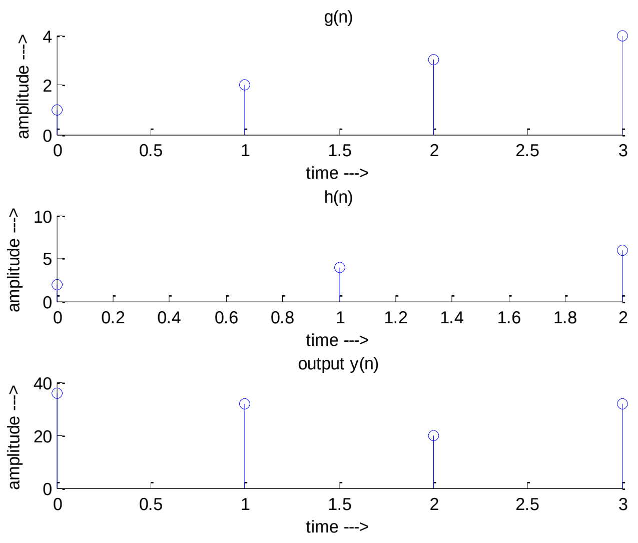
1. Start the program.
2. Get the two input sequence x(n) and h(n) and compute the length of the sequence.
3. Append zeros to x(n) or h(n) to make both sequence of same length.
4. Rotate the sequence h(-k), 'n' times. If 'n' is positive shift the sequence in anticlockwise direction else in the clockwise direction.
5. Determine the product of sequence x(n) and shifted h(n) sequence and sum it.
6. Execute the program, display the result and verify it theoretically.
7. Plot the output graph for each sequence.
8. Stop the process.

Program:

```
% Program for Circular Convolution
clc;
clear all;
close all;
%g=[1,2,3,4];
%h=[2,4,6];
x=input('enter the sequence');
h=input('enter the sequence');
N1=length(x);
N2=length(h);
subplot(3,1,1);
n=0:1:N1-1;
stem(n,x);
title('x(n)');
ylabel('amplitude --->');
xlabel('time --->');
subplot(3,1,2);
n=0:1:N2-1;
```

```
stem(n,h);
title('h(n)');
ylabel('amplitude --->');
xlabel('time --->');
N=max(N1,N2);
if(N1<=N);
  x1=[x,zeros(1,N-N1)];
end
if(N2<=N);
h1=[h,zeros(1,N-N2)];
end
for n=1:N
   y(n)=0;
for i=1:N
   j=n-i+1;
if(j<=0)
  j=N+j;
end
  y(n)=y(n)+x1(i)*h1(j);
end
end
disp('y=');
disp(y);
subplot(3,1,3);
n=0:1:N-1;
stem(n,y);
title('output y(n)');
ylabel('amplitude --->');
xlabel('time --->');
```

MODEL GRAPH



THEORITICAL CALCULATION

Using graphical or Matrix method

$$y(n) = x(n) \otimes h(n)$$

$$y(n) = \sum_{k=0}^{N-1} x(k)h(n-k)$$

Enter the sequence g=[1,2,3,4]
Enter the sequence h=[2,4,6]

Output Sequence is = 36 32 20 32

Matrix Method:

$$\begin{pmatrix} x(0) & x(N-1) & x(n-2)\,\ldots\ldots & x(1) \\ x(1) & x(0) & x(N-1)\ldots & x(2) \\ x(2) & x(1) & x(0)\ldots & x(N-1) \\ . & & & \\ . & & & \\ . & & & \\ x(N-1) & x(N-2) & x(N-3).. & x(0) \end{pmatrix} \begin{pmatrix} h(0) \\ h(1) \\ h(2) \\ . \\ . \\ . \\ h(N-1) \end{pmatrix} = \begin{pmatrix} y(0) \\ y(1) \\ y(2) \\ . \\ . \\ \\ y(N-1) \end{pmatrix}$$

Postlab Questions :

1. What is the difference between Linear convolution & circular convolution?
2. State convolution property of a signal.
3. What are sectioned convolutions & what is the need of it?
4. Write the applications of convolution.
5. What is the relation between convolution in time domain and frequency domain ?

Result:

Thus a program to find out the linear and circular convolution of two sequences using Matlab was written and executed.

Ex No: **Date :**

3(a). SPECTRUM ANALYSIS USING DFT

Aim:

To analyze the amplitude and phase response of the DFT and IDFT signalusing MATLAB.

Prerequisite Lab Questions:

1. Write the analysis and synthesis equation for DFT?
2. What do you mean by the property of shifting of DFT?
3. Find the value of twiddle factor for N=8, K=2 .
4. How many multiplications and additions are required to compute N point DFT ?
5. What is the relationship between DFT and Z-transform?

Algorithm:

1. Start the program.
2. Get the two input sequence x (n) and type of the DFT.
3. Compute the Discrete Fourier Transform (DFT) of the given sequence.
4. Find out the magnitude and phase response of the given sequence.
5. Execute the program, display the result and verify it.
6. Plot the output graph for magnitude and phase.
7. Compute the Inverse Discrete Fourier Transform (IDFT) of the given sequences.
8. Display the results verify it and plot the result.
9. Stop the process.

Program:

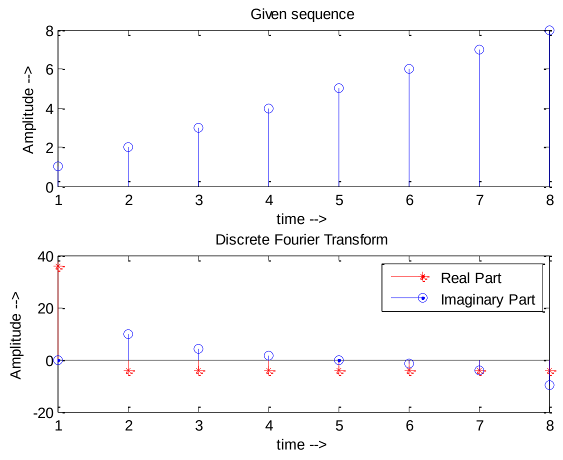
```
% Program for Discrete Fourier Transform
clc;
clear all;
close all;
%x=[1 2 3 4 5 6 7 8];
%N=[8];
x=input('enter the input x=');
N=input('enter the order of dft N=');
k=0:1:N-1;
disp('Input Sequence x=');
disp(x);
 XK=fft(x,N);
disp('XK=');
disp(XK);
R=real(XK);
disp('Real part=');
disp(R);
I=imag(XK);
disp('Imaginary part=');
disp(I);
 mag=abs(XK);
disp('Magnitude=');
disp(mag);
 ang=angle(XK);
disp('Angle=');
disp(ang);
xn=ifft(XK,N);
disp('IDFT xn=');
disp(xn);

figure(1);
subplot(2,1,1);
stem(x);
xlabel('time -->');
ylabel('Amplitude -->');
title('Given sequence');

subplot(2,1,2);
stem(R,'r*');
hold on;
stem(I);
legend('Real Part','Imaginary Part');
xlabel('time -->');
ylabel('Amplitude -->');
title('Discrete Fourier Transform')

figure(3);
subplot(2,2,1);
stem(k,mag);
xlabel('time -->');
ylabel('x(k) --->');
title('Magnitude response');

subplot(2,2,2);
stem(k,ang);
xlabel('time -->');
ylabel('ang x(k) --->');
title('Phase response');

subplot(2,2,3);
stem(k,xn);
xlabel('time -->');
ylabel('x(n)');
title('Inverse Discrete Fourier transform');
 %Spectrum analysis
Fs=[10];
xdft = XK(1:N/2+1);
```

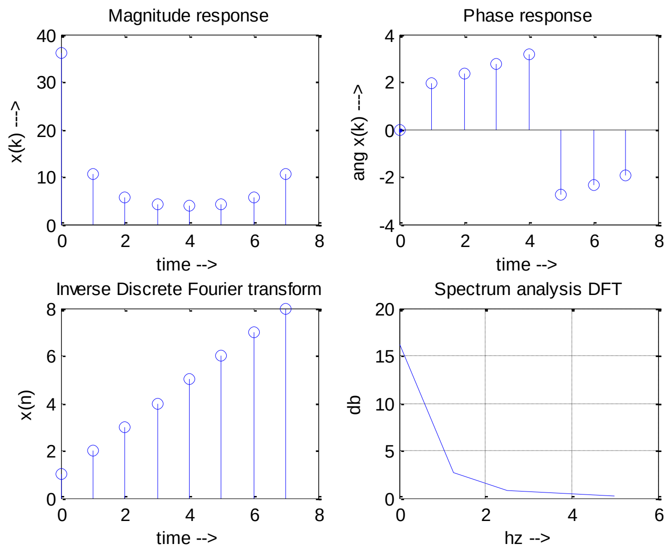
```
psdx = (1/(Fs*N)) * abs(xdft).^2;
psdx(2:end-1) = 2*psdx(2:end-1);
freq = 0:Fs/N:Fs/2;
subplot(2,2,4);
plot(freq,psdx);
grid on;
xlabel('hz -->');
ylabel('db');
title('Spectrum analysis DFT');
disp('Spectrum analysis DFT=');
disp(psdx);
```

Output:

```
enter the input x=[1 2 3 4 5 6 7 8]
enter the order of dft N=[8]
Input Sequence x=1    2    3    4    5    6    7    8

XK=
  36.0000 + 0.0000i
  -4.0000 - 9.6569i
  -4.0000 - 4.0000i
  -4.0000 - 1.6569i
  -4.0000 + 0.0000i
  -4.0000 + 1.6569i
  -4.0000 + 4.0000i
  -4.0000 + 9.6569i

Real part=   36    -4    -4    -4    -4    -4    -4    -4

Imaginary part=     0    9.6569    4.0000    1.6569         0   -1.6569   -4.0000   -9.6569

Magnitude= 36.0000   10.4525    5.6569    4.3296    4.0000    4.3296    5.6569   10.4525

Angle= 0    1.9635    2.3562    2.7489    3.1416   -2.7489   -2.3562   -1.9635

IDFT xn= [1     2     3     4     5     6     7     8]
Spectrum analysis DFT= 16.2000    2.7314    0.8000    0.4686    0.2000
```

Model Graph





Postlab Questions :

1. Mention the applications of DFT?
2. What is the drawback in Fourier transform and how it can be avoided?
3. Write the 4-point DFT matrix representation.
4. How many multiplications and additions are required to compute DFT for N=512?
5. Distinguish between DFT and DTFT.

Result:

Thus a MATLAB program for Spectrum analysis of the DFT and IDFT signal was written, executed and the graphs were plotted.

Ex : No: **Date:**

3(b).FAST FOURIER TRANSFORM AND INVERSE FAST FOURIER TRANSFORM (FFT & IFFT)

Aim:

To write a program to find out the Fast Fourier transform and Inverse Fast Fourier transform of the sequence using MATLAB and display the magnitude and phase response.

Prerequisite Lab Questions:

1. What do you mean by the term " bit reversal" as applied to FFT?
2. How many multiplications and additions are required to compute N point DFT using radix 2 FFT?
3. What are the advantages of FFT algorithm over direct computation of DFT?
4. State and periodicity and symmetry property of FFT?
5. Define in place computation?

Description

Input : Enter input x (n) as a sequence
Output **:** X (k) can be obtained by using formula,

FFT

$$X(k) = \sum_{n=0}^{N-1} x(n)\,\omega_N^{kn}$$

IFFT

$$x(n) = \frac{1}{N}\sum_{k=0}^{N-1} X(k)\,\omega_N^{-kn}$$

$$\omega_N^{kn} = e^{-j\frac{2\pi kn}{N}}$$

Algorithm:

1. Start the program.
2. Get the two input sequence x (n) and type of the FFT.
3. Compute the fast Fourier transform (FFT) of the given sequence.
4. Find out the magnitude and phase response of the given sequence.
5. Execute the program, display the result and verify it.
6. Plot the output graph for magnitude and phase.
7. Compute the inverse fast Fourier transform (FFT) of the given sequences.
8. Display the results verify it and plot the result.
9. Stop the process.

Program:

```
% Program for FFT & IFFT
clc;
clear all;
close all;
%x=[2 2 2 2 1 1 1 1];
%N=[8];
x=input('enter the input x=');
N=input('enter the order of fft N=');
k=0:1:N-1;

disp('Input Sequence x=');
disp(x);

XK=fft(x,N);
disp('XK=');
disp(XK);

R=real(XK);
disp('Real part=');
disp(R);

I=imag(XK);
disp('Imaginary part=');
disp(I);

mag=abs(XK);
disp('Mag=');
disp(mag);

ang=angle(XK);
disp('angle=');
disp(ang);

xn=ifft(XK,N);
disp('IFFT xn=');
disp(xn);

figure(1);
stem(R,'r*');
hold on;
stem(I);
legend('Real Part','Imaginary Part');
xlabel('time -->');
ylabel('Amplitude -->');
title('Fast Fourier Transform')
```

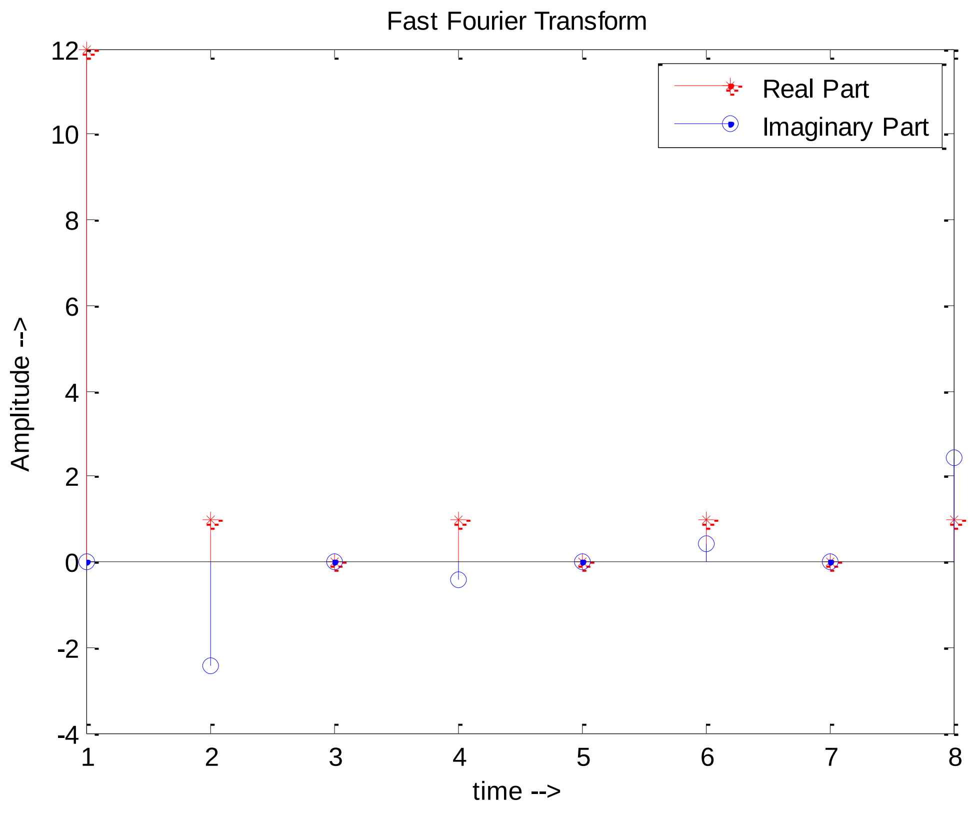
```
figure(2);
subplot(2,2,1);
stem(x);
xlabel('time -->');
ylabel('Amplitude -->');
title('Given sequence');
subplot(2,2,2);
stem(k,mag);
xlabel('time -->');
ylabel('x(k) --->');
title('Magnitude response');

subplot(2,2,3);
stem(k,ang);
xlabel('time -->');
ylabel('ang x(k) --->');
title('Phase response');

subplot(2,2,4);
stem(k,xn);
xlabel('time -->');
ylabel('x(n)');
title('Inverse Fast Fourier transform');
```

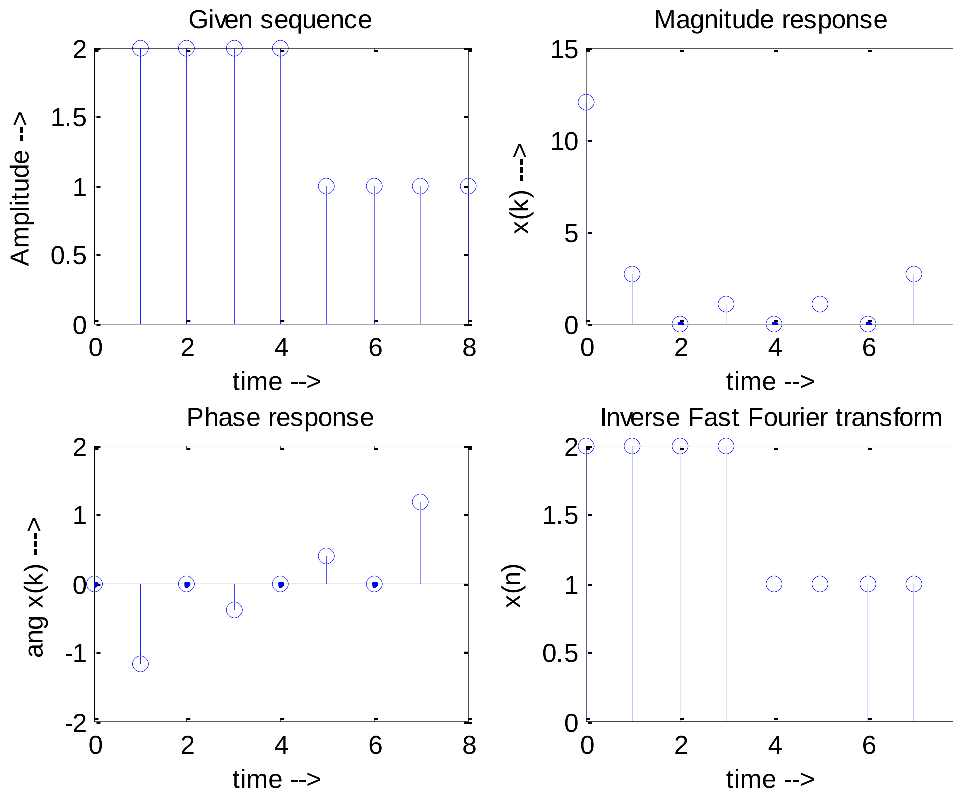
Output:

```
enter the input x=[2 2 2 2 1 1 1 1]
enter the order of fft N=[8]
Input Sequence x=    2    2    2    2    1    1    1    1
XK=
  12.0000 + 0.0000i
   1.0000 + 2.4142i
   0.0000 + 0.0000i
   1.0000 + 0.4142i
   0.0000 + 0.0000i
   1.0000 - 0.4142i
   0.0000 + 0.0000i
   1.0000 - 2.4142i

Real part=    12    1    0    1    0    1    0    1

Imaginary part=        0   -2.4142        0   -0.4142        0    0.4142        0    2.4142

Mag=   12.0000    2.6131        0    1.0824        0    1.0824        0    2.6131

Angle=       0   -1.1781        0   -0.3927        0    0.3927        0    1.1781
```

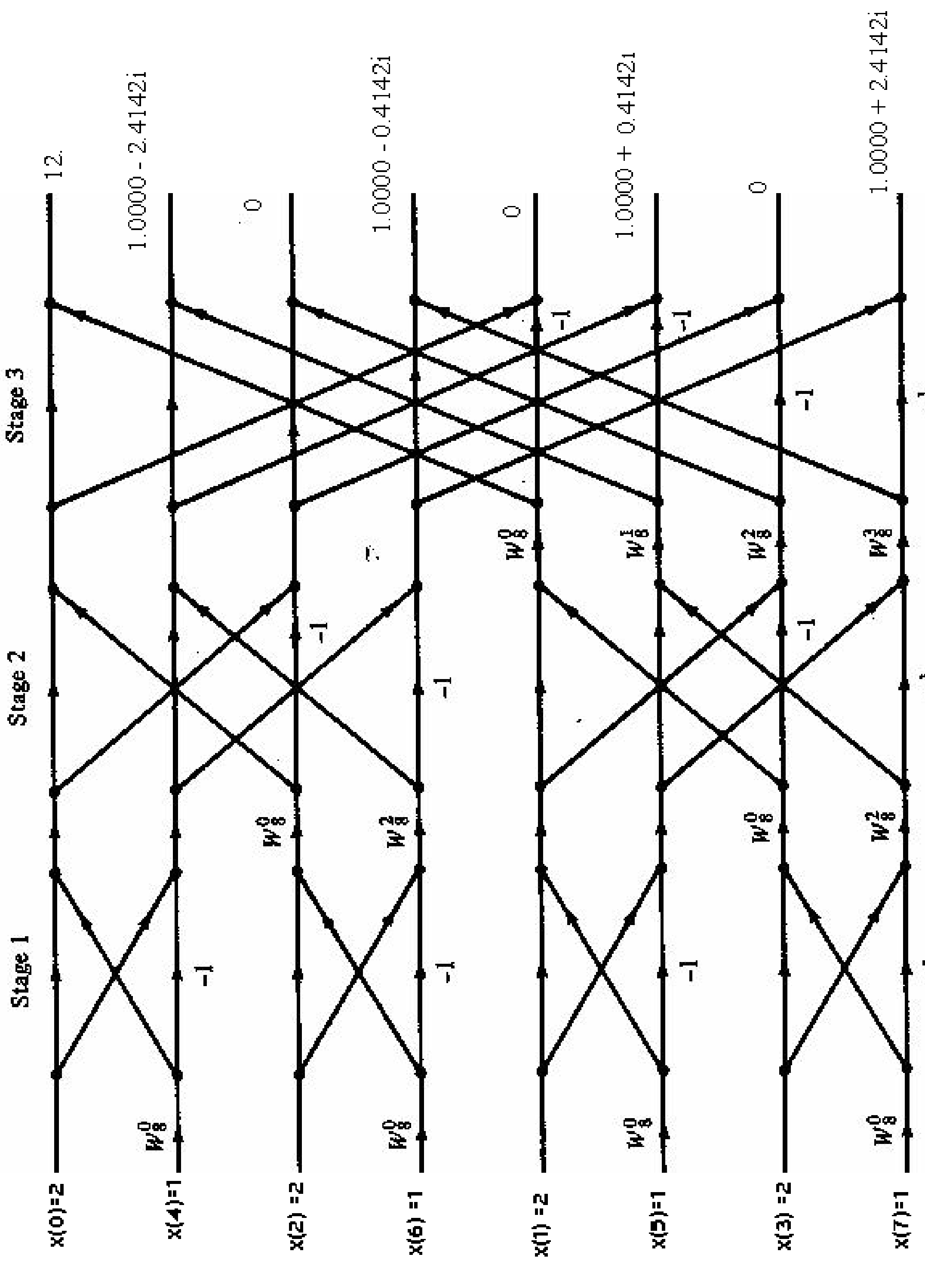
```
IFFT xn=    2    2    2    2    1    1    1    1
```

Model Graph:





Given sequence
Amplitude -->
time -->
Magnitude response
x(k) ---->
time -->
Phase response
ang x(k) ---->
time -->
Inverse Fast Fourier transform
x(n)
time -->



Stage 1
Stage 2
Stage 3
x(0)=2
x(4)=1
x(2) =2
x(6) =1
x(1) =2
x(5)=1
x(3) =2
x(7)=1
W_8^0
W_8^2
W_8^1
W_8^3
−1
12.
1.0000 - 2.4142i
0
1.0000 - 0.4142i
0
1.0000 + 0.4142i
0
1.0000 + 2.4142i

PostLab Questions:

1. How many multiplications and additions are involved in radix-2 FFT for N=1024 when compared to DFT?
2. Draw the 2-point DIF-FFT butterfly structure.
3. Compare DIF and DIT FFT algorithm.
4. Why FFT is needed?
5. Mention the applications of FFT algorithm?

Result:

Thus a program to find out the fast Fourier (FFT) and inverse fast Fourier transform (IFFT) of the sequence using Matlab was written and executed.

Ex:No:
Date:

4.FIR FILTER DESIGN

Aim:

To write a program to design the FIR low pass, high pass, band pass and band stop filters and obtain thefrequency response of the filter using MATLAB.

Prerequisite Lab Questions:

1. Mention the advantages and disadvantages of FIR filters.
2. What are the classifications of filters?
3. List the techniques for designing FIR filters.
4. What are the features of windowing method?
5. What are the types of windows in FIR filter design?

Description:

Input: Give the cutoff frequency and order of the filter.

Output: 1) Find the impulse response hd (n) for the given cut-off frequency and Order of the filter.
2) Decide the window function w (n) and compute the window Coefficients.
3) Find the filter coefficients h (n) =hd (n)*w (n).
4) Plot the frequency response for Low pass, High pass, Band pass and Band stop filters

Algorithm:

1. Start the program.
2. Get the cutoff frequency and order of the filter as input...
3. Find the ideal impulse response of the filter hd (n).
4. Select a window function w (n).
5. Find out the filter coefficients h (n) by multiplying hd (n) and w (n).
6. Compute the frequency response of the filter.
7. Repeat the process for high pass, band pass and band stop filters.
8. Execute the program, display the result and verify it.
9. Plot the output graph for hd(n),w(n),h(n)and frequency response of LPF,HPF,BPF and BSF.
10. Stop the process.

Program:

```
%   Ideal LowPass filter computation
%   hd = ideal impulse response between 0 to M-1
%   wc = cutoff frequency in radians
%   M = length of the ideal filter
%   Save as ideal_lp
```

% Function:

```
function [hd]=ideal_lp(wc,M)
alpha=(M-1)/2;
n=0:1:M-1;
N=n-alpha+eps;  % add smallest number to avoid divide by zero
hd=sin(wc*N)./(pi*N);
```

Program:

```
%PROGRAM FOR LOW PASS FILTER
clc;
clear all;
close all;
M=7;
wc=1.2;
n=0:1:M-1;
hd=ideal_lp(wc,M);
w=hamming(M);
h=hd.*w';
disp('Filter coefficients for LPF=');

[M1,P]=freqz(h,1,2000);
mag=20*log10(abs(M1));

subplot(2,2,1);
stem(n,hd);
xlabel('n--->');
ylabel('hd(n)--->');
title('Ideal Impulse Response');
disp('hd=');
disp(hd);
subplot(2,2,2);
stem(n,w);
xlabel('n--->');
```

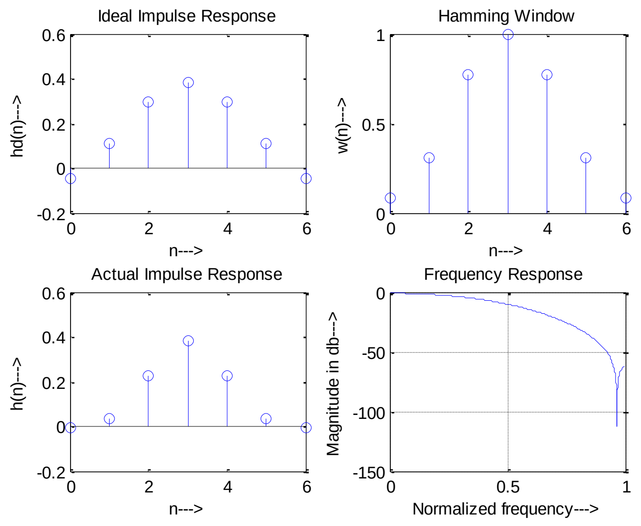
```
ylabel('w(n)--->');
title('Hamming Window');
disp('w=');
disp(w);
subplot(2,2,3);
stem(n,h);
xlabel('n--->');
ylabel('h(n)--->');
title('Actual Impulse Response');
disp('h=');
disp(h);
subplot(2,2,4);
plot(P/pi, mag);
grid on;
xlabel('Normalized frequency--->');
ylabel('Magnitude in db--->');
title('Frequency Response ');
```

Output:

```
Filter coefficients for LPF=

hd= -0.0470    0.1075    0.2967    0.3820    0.2967    0.1075   -0.0470

w =  0.0800    0.3100    0.7700    1.0000    0.7700    0.3100    0.0800

h = -0.0038    0.0333    0.2284    0.3820    0.2284    0.0333   -0.0038
```

Model Graph:

LOW PASS FILTER



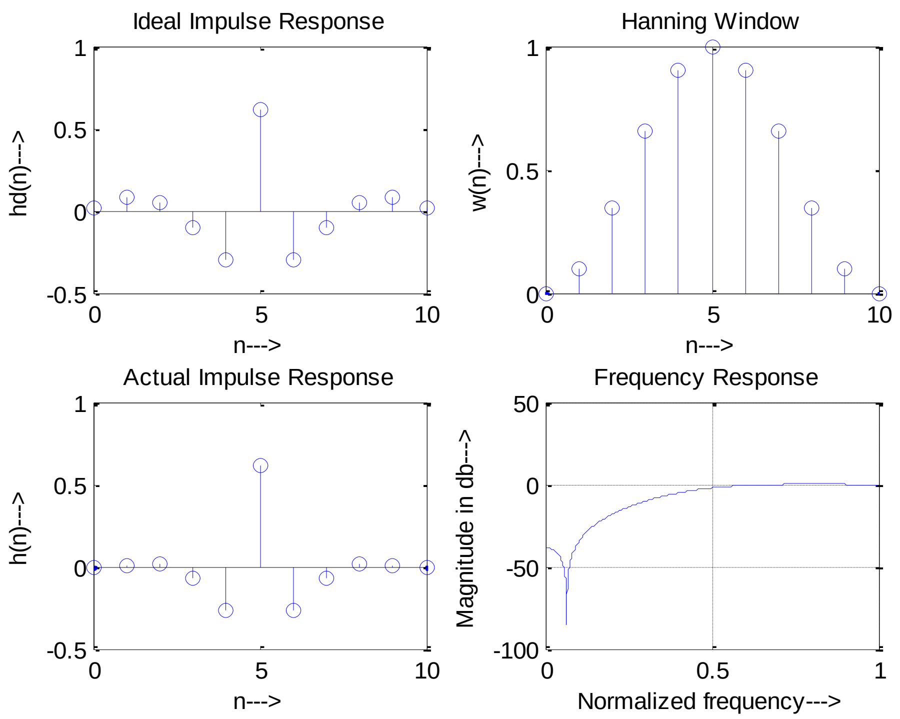
```
%PROGRAM FOR HIGH PASS FILTER
        clc;
        clear all;
        close all;
        M=11;
        wc=1.2;
        n=0:1:M-1;
        hd=ideal_lp(pi,M)-ideal_lp(wc,M);
        w=hann(M);
        h=hd.*w';
        disp('Filter coefficients for HPF=');

        [M1,P]=freqz(h,1,2000);
        mag=20*log10(abs(M1));

        subplot(2,2,1);
        stem(n,hd);
        xlabel('n--->');
        ylabel('hd(n)--->');
        title('Ideal Impulse Response');
        disp('hd=');
        disp(hd);

        subplot(2,2,2);
        stem(n,w);
        xlabel('n--->');
        ylabel('w(n)--->');
        title('Hanning Window');
        disp('w=');
        disp(w);

        subplot(2,2,3);
        stem(n,h);
        xlabel('n--->');
        ylabel('h(n)--->');
        title('Actual Impulse Response');
        disp('h=');
        disp(h);

        subplot(2,2,4);
        plot(P/pi, mag);
        grid on;
        xlabel('Normalized frequency--->');
        ylabel('Magnitude in db--->');
        title('Frequency Response ');
```

Output:

```
Filter coefficients for HPF=

hd=0.0178    0.0793    0.0470   -0.1075   -0.2967    0.6180   -0.2967   -0.1075    0.0470
 0.0793    0.0178

w = 0    0.0955    0.3455    0.6545    0.9045    1.0000    0.9045    0.6545    0.3455    0.0955
 0

h = 0    0.0076    0.0162   -0.0704   -0.2683    0.6180   -0.2683   -0.0704    0.0162    0.0076
 0
```

Model Graph:

HIGH PASS FILTER



```
%PROGRAM FOR BAND PASS FILTER
                clc;
                clear all;
                close all;
                M=11;
                wc1=1.2;
                wc2=2;
```

```
n=0:1:M-1;
hd=ideal_lp(wc2,M)-ideal_lp(wc1,M);
w=hann(M);
h=hd.*w';
disp('Filter coefficients for BPF=');

[M1,P]=freqz(h,1,2000);
mag=20*log10(abs(M1));

subplot(2,2,1);
stem(n,hd);
xlabel('n--->');
ylabel('hd(n)--->');
title('Ideal Impulse Response');
disp('hd=');
disp(hd);

subplot(2,2,2);
stem(n,w);
xlabel('n--->');
ylabel('w(n)--->');
title('Hanning Window');
disp('w=');
disp(w);

subplot(2,2,3);
stem(n,h);
xlabel('n--->');
ylabel('h(n)--->');
title('Actual Impulse Response');
disp('h=');
disp(h);

subplot(2,2,4);
plot(P/pi, mag);
grid on;
xlabel('Normalized frequency--->');
ylabel('Magnitude in db--->');
title('Frequency Response ');
```

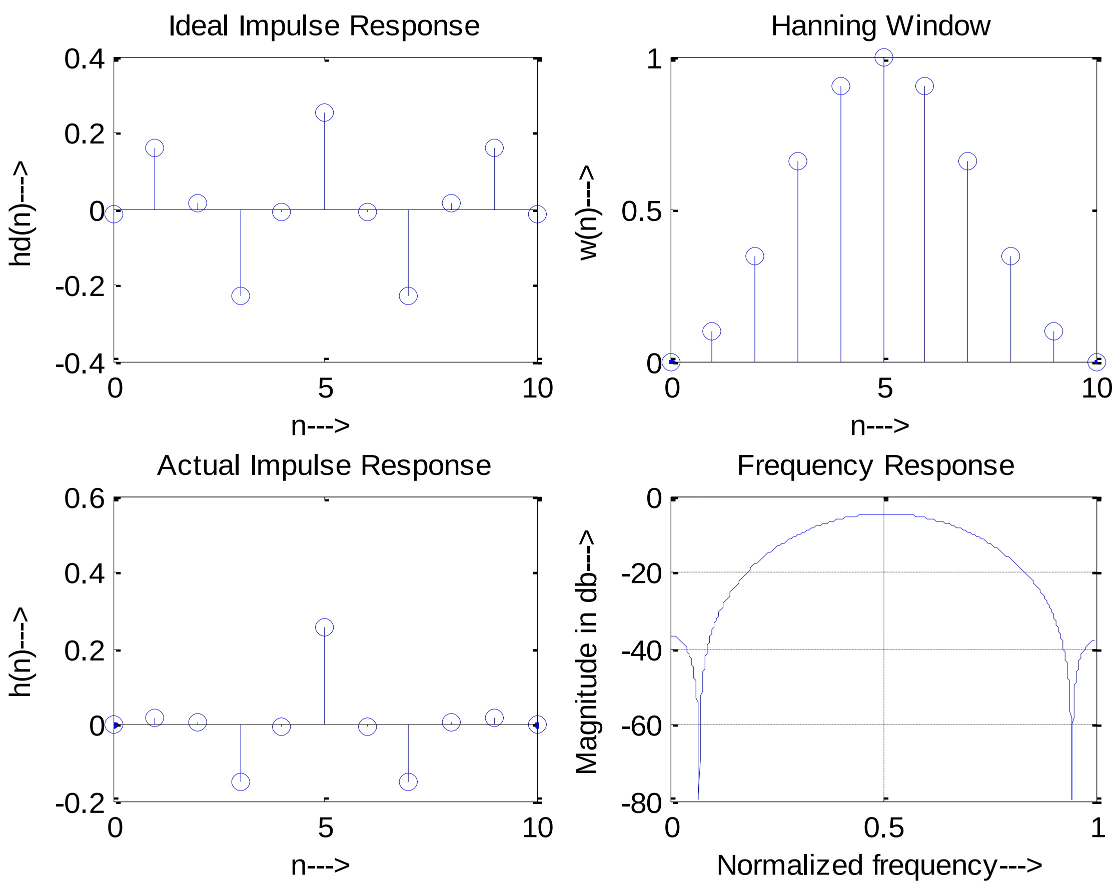
Output:

```
Filter coefficients for BPF=

hd=-0.0168   0.1580   0.0173  -0.2280  -0.0072   0.2546  -0.0072  -0.2280   0.0173
0.1580  -0.0168
```

```
w= 0    0.0955    0.3455    0.6545    0.9045    1.0000    0.9045    0.6545    0.3455    0.0955
0

h= 0    0.0151    0.0060   -0.1492   -0.0065    0.2546   -0.0065   -0.1492    0.0060
0.0151   0
```

Model Graph:

BANDPASS FILTER



```
%PROGRAM FOR BAND STOP FILTER
                    clc;
                    clear all;
                    close all;
                    M=7;
                    wc1=1.2;
                    wc2=2;
```

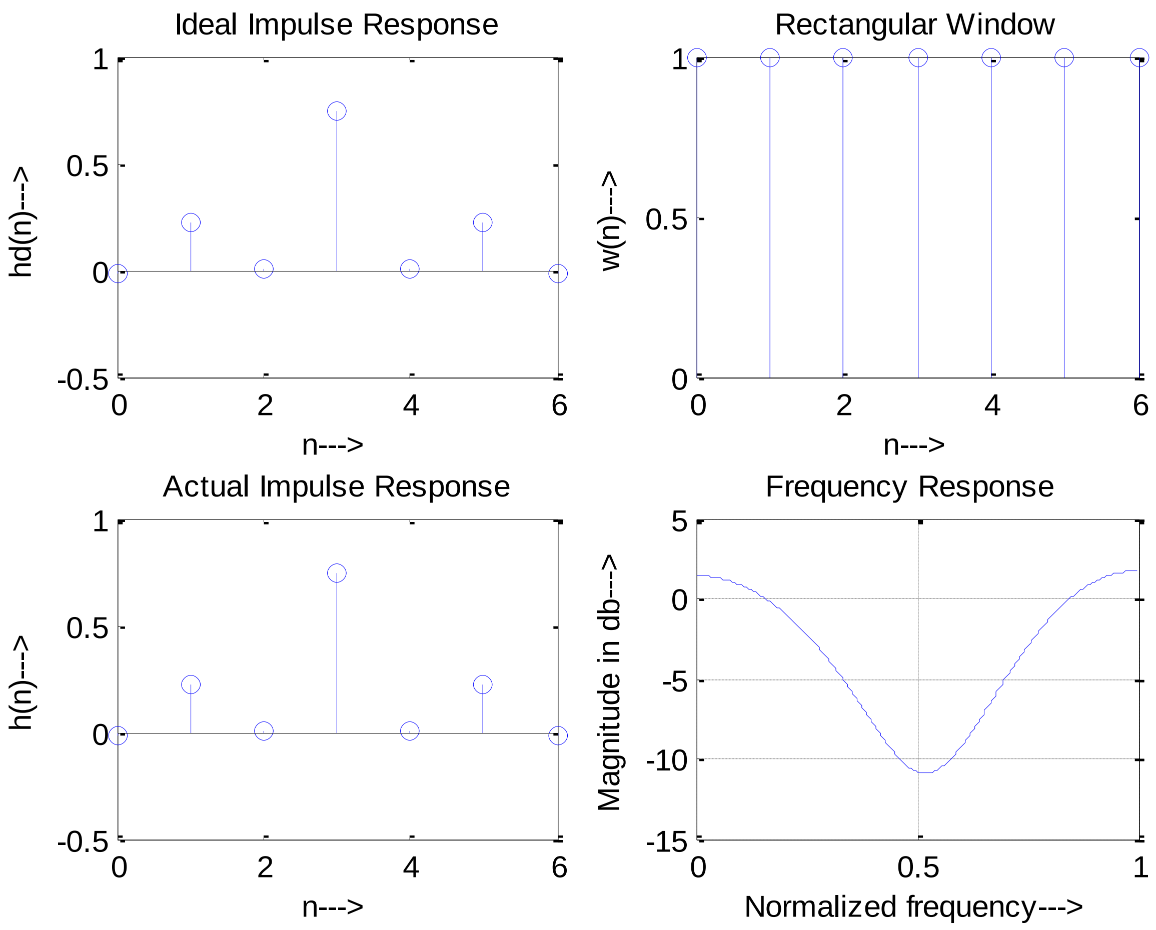
```
n=0:1:M-1;
hd=ideal_lp(pi,M)-ideal_lp(wc2,M)+ideal_lp(wc1,M);
w=rectwin(M);
h=hd.*w';
disp('Filter coefficients for BSF=');

[M1,P]=freqz(h,1,2000);
mag=20*log10(abs(M1));

subplot(2,2,1);
stem(n,hd);
xlabel('n--->');
ylabel('hd(n)--->');
title('Ideal Impulse Response');
disp('hd=');
disp(hd);

subplot(2,2,2);
stem(n,w);
xlabel('n--->');
ylabel('w(n)--->');
title('Rectangular Window');
disp('w=');
disp(w);

subplot(2,2,3);
stem(n,h);
xlabel('n--->');
ylabel('h(n)--->');
title('Actual Impulse Response');
disp('h=');
disp(h);

subplot(2,2,4);
plot(P/pi, mag);
grid on;
xlabel('Normalized frequency--->');
ylabel('Magnitude in db--->');
title('Frequency Response ');
```

Output:

```
Filter coefficients for BSF=

hd=   -0.0173    0.2280    0.0072    0.7454    0.0072    0.2280   -0.0173
```

```
w =    1    1    1    1    1    1    1

h =   -0.0173    0.2280    0.0072    0.7454    0.0072    0.2280   -0.0173
```

Model Graph:
<u>BANDSTOP FILTER</u>



<u>Windows:</u>
<u>Theoretical calculation:</u>

1. hd (n)=1/2π ∫ Hd(w)e^{jwn}dw
2. Select w (n)
3. Find h (n) = hd (n)w(n)

<u>Rectangular Window</u>

$$w(n) = \begin{cases} 1, & 0 \le n \le M-1 \\ 0, & otherwise \end{cases}$$

<u>Hanning Window</u>

$$w(n) = \begin{cases} 0.5*\left[1-\cos\left(\frac{2\pi n}{M-1}\right)\right], & 0 \le n \le M-1 \\ 0, & otherwise \end{cases}$$

<u>Hamming Window</u>

1. $$w(n) = \begin{cases} 0.54 - 0.46*\cos\left(\frac{2\pi n}{M-1}\right), & 0 \le n \le M-1 \\ 0, & otherwise \end{cases}$$

<u>Kaiser window</u>

$$w(n) = \begin{cases} I_0\left(\beta\left(1-((n-\alpha)/\alpha)^2\right)^{1/2}\right)/I_0(\beta) & 0 \le n \le M-1 \\ 0 & otherwise \end{cases}$$

Where α = (*M*-1)/2 and I_0 = the zeroth order Modified Bessel Function

If ω_p, ω_s, R_p, and A_s are giventhen the following equations are needed for design.

$$\text{Transition width} = \Delta f = \frac{\omega_s - \omega_p}{2\pi}$$

$$\text{Filter order } M \cong \frac{A_s - 7.95}{14.36\Delta f} + 1$$

$$Parameter\,\beta = \begin{cases} 0.1102(A_s - 8.7), & A_s \ge 50 \\ 0.5842(A_s - 21)^{0.4} + 0.07886(A_s - 21), & 21 > A_s > 50 \\ 0 & A_s < 21 \end{cases}$$

PostLab questions:

1. What is meant by linear phase filters?
2. What is the condition for the impulse response of FIR filter to satisfy for constant and Phase delay and for only constant group delay?
3. What are Gibbs oscillations?
4. Give the equation for rectangular, hamming and Kaiser Windows.
5. State the condition for linear phase characteristics in FIR.

Result:

Thus a program to design the FIR low pass, high pass, band pass and band stop filters was Written and response of the filter using MATLAB was executed.

Ex : No: **Date:**

5.IIR FILTER DESIGN

Aim:

To write a program to design the Butterworth low pass, high pass filters and find out the response of the filter using MATLAB.

Prerequisite Lab Questions:

1. What are the requirements for an analog filter to be stable and causal?
2. What are the types of analog filters?
3. State the conditions to design a stable digital filters.
4. Write the relationship between analog and digital frequency in impulse invariant method?
5. Define Bilinear transformation.

Description:

Input :(a) Enter the pass band ripple and stop band attenuation
(b)Enter the passband and stop band frequency

Output **:** (a) Find the order and cutoff frequency using bilinear and impulse Invariant transformation
(b) Calculate the Analog filter coefficients H(s)
(c) Transform the Analog filter to digital filter and find H (z)
(d) Plot the responses for all the filters.

Algorithm:

1. Start the program.
2. Get the input sequences for pass band ripple, stop band ripple, pass band frequency, stop band frequency and sampling frequency.
3. Convert the frequency into radians/sec.
4. Compute the order and cutoff frequency of the filter and display it.
5. Design the analog filter and find the coefficient.
6. Convert the analog filter into digital using bilinear or impulse invariant transform.
7. Find out the frequency response of the filter.
8. Compute the magnitude and phase response of the filter.
9. Execute the program, display the result and verify it.
10. Plot the output graph for each sequence for low pass and high pass filters.
11. Stop the process.

Program:

```
% function to find order of butterworth filter using impulse invariant transformation
% Save as butt_impord
function [N,Wc]= butt_impord(w1,w2,a1,a2)
l=sqrt((1/a2^2)-1)
e=sqrt((1/a1^2)-1)
num=log10(l/e)
den=log10(w2/w1)
N=ceil((num/den))
```

```
Wc=(w1)/((e)^(1/N))
%function to find order of butterworth filter using bilinear transformationSave is as
butt_biord.
function [N1,Wc1]= butt_biord(w1,w2,a1,a2)
T=1;
wp1=(2/T)*tan(w1/2)
ws1=(2/T)*tan(w2/2)
lam=sqrt((1/a2^2)-1)
ep=sqrt((1/a1^2)-1)
num1=log10(lam/ep)
den1=log10(ws1/wp1)
N1=ceil((num1/den1))
Wc1=wp1/((ep)^(1/N1))

% Program for Butterworth Low Pass Filter
clc;
clear all;
close all;
%w1=0.2*pi;w2=0.6*pi;a1=0.8;a2=0.2;
w1=input('Enter the Passband frequency w1=');
w2=input('Enter the Stopband frequency w2=');
a1= input('Enter the Passband ripple a1=');
a2=input('Enter the Stopband ripple a2=');
[N,Wc]=butt_impord(w1,w2,a1,a2);
disp('LPF ANALOG FILTER TRANSFER FUNCTION');
[num,den]=butter(N,Wc,'s')
disp('LPF DIGITAL FILTER TRANSFER FUNCTION');
[b,a]=impinvar(num,den)
[mag,angle]=freqz(b,a,512);
magnitude=20*log10(abs(mag));
figure(1);
plot(angle/pi,magnitude);
grid on;
title('Low Pass Butterworth Filter');
xlabel('frequency --->');
ylabel('magnitude --->');

%Program for Butterworth High Pass Filter
[N1,Wc1]=butt_biord(w2,w1,a1,a2);
disp(' HPF ANALOG FILTER TRANSFER FUNCTION');
[num1,den1]=butter(N1,Wc1,'high','s')
disp('HPF DIGITAL FILTER TRANSFER FUNCTION');
[b1,a1]=bilinear(num1,den1,1)
[mag1,angle1]=freqz(b1,a1,512);
magnitude=20*log10(abs(mag1));
figure(2);
plot(angle1/pi,magnitude);
grid on;
```

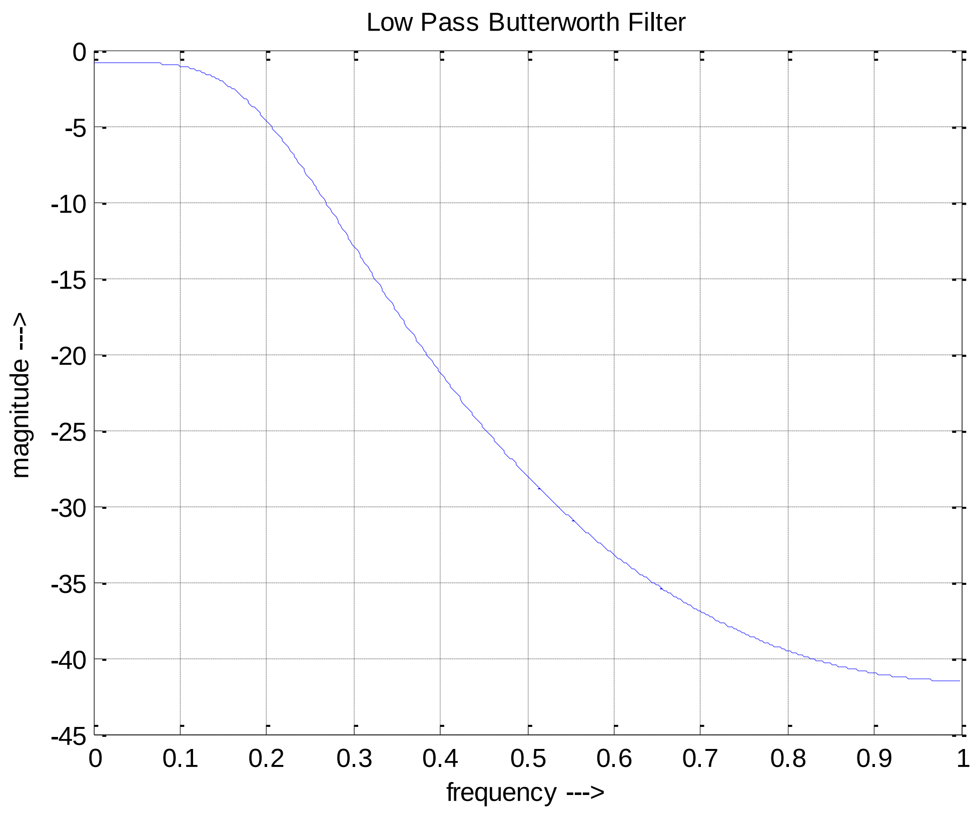
```
title('High Pass Butterworth Filter');
xlabel('frequency --->');
ylabel('magnitude --->');
```

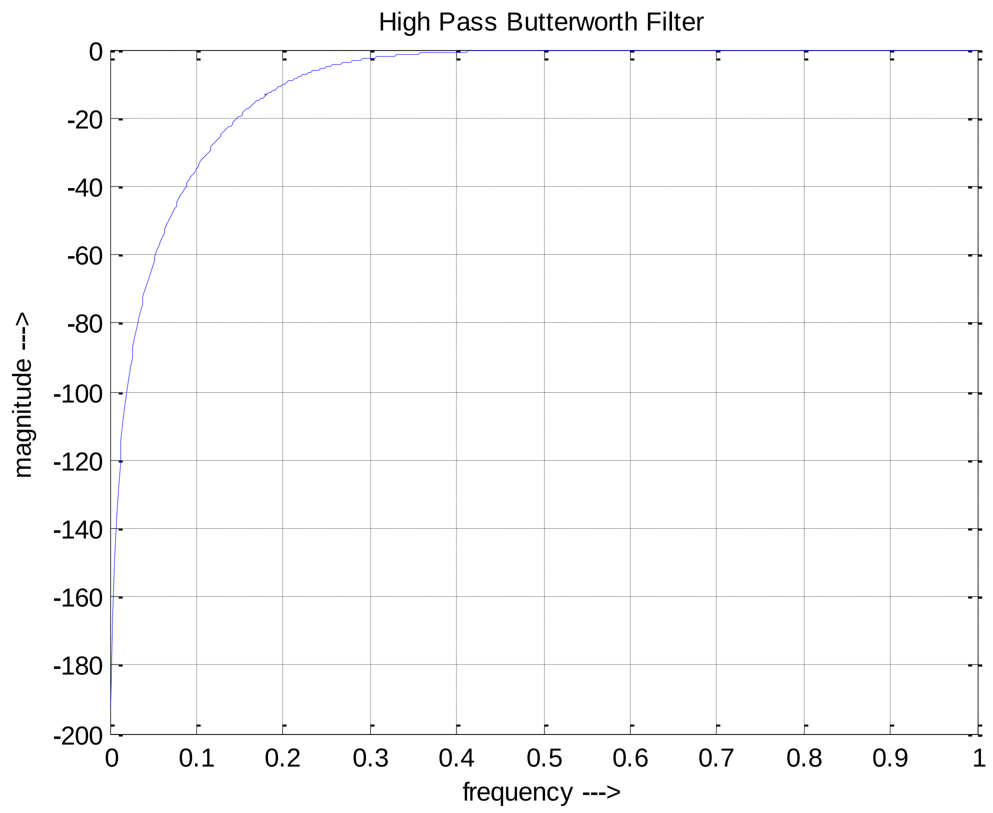
Output:

```
Enter the Passband frequency w1=[0.2*pi]
Enter the Stopband frequency w2=[0.6*pi]
Enter the Passband ripple a1=[0.8]
Enter the Stopband ripple a2=[0.2]

l = 4.8990 ,e = 0.7500

num =   0.8150 ,den =  0.4771

N = 2 ,Wc =   0.7255

LPF ANALOG FILTER TRANSFER FUNCTION

num =   0         0    0.5264

den =  1.0000    1.0260    0.5264

LPF DIGITAL FILTER TRANSFER FUNCTION

b =   0    0.3015         0

a =   1.0000   -1.0432    0.3584

T =   1,  wp1 = 0.6498

ws1 = 2.7528,  lam =  4.8990

ep =  0.7500

num1 =   0.8150,  den1 = 0.6270

N1 =   2

Wc1 =   0.7504

 HPF ANALOG FILTER TRANSFER FUNCTION

num1 = 1     0     0

den1 =   1.0000    1.0612    0.5631

HPF DIGITAL FILTER TRANSFER FUNCTION
```

```
b1 =   0.5983   -1.1966    0.5983

a1 =   1.0000   -1.0282    0.3651
```

Model graph





Theoretical calculation

Bilinear Transformation

```
wp1=(2/T)*tan(w1/2)
ws1=(2/T)*tan(w2/2)
lam=sqrt((1/a2^2)-1)
ep=sqrt((1/a1^2)-1)
N=log10(lam/ep) /  log10(ws1/wp1);
Wc1=wp1/((ep)^(1/N1));
```

N-Even

$$H(s) = \prod_{k=1}^{\frac{N}{2}} \frac{\Omega_c^2}{s^2 + 2\Omega_c\, b_k s + \Omega_c^2};\cdots k = 1,2,\ldots\frac{N}{2}$$

N- Odd

$$H(S) = \frac{\Omega_c}{S + \Omega_c} \prod_{k=1}^{\frac{N}{2}-1} \frac{\Omega_c^2}{s^2 + 2\Omega_c\, b_k s + \Omega_c^2};\cdots k = 1,2,\ldots\frac{N}{2} - 1$$

b_k = 2 sin(2k-1)*pi/4

To find H (Z) replace‘s’ by

S= 2(Z-1)
T(Z+1)

Impulse Invariant Transformation

l=sqrt((1/a2^2)-1)
e=sqrt((1/a1^2)-1)
N=log10(l/e) / log10(w2/w1); ;
Wc=(w1)/((e)^(1/N));
H(S) is same

$$H_c(s) = \sum_{k=1}^{N} \frac{A_k}{s - s_k} H(z) = \sum_{k=1}^{N} \frac{A_k}{1 - e^{s_k T} z^{-1}}$$

PostLab questions:

1. What are the properties of Butterworth low pass filters?
2. Give the magnitude function of Butterworth filter.
3. What is meant by conformal mapping?
4. Define warping effect?
5. What are the advantages and disadvantages of bilinear transformations?

Result:

Thus a program to design the Butterworth low pass and high pass filter using Matlab was written and response of the filter was executed.

Ex : No: **Date:**

6. MULTIRATE FILTERS

Aim:

Write a program to perform the decimation and interpolation on the given input sequence operation using Matlab.

Prerequisite Lab Questions:

1. What are the need for multirate signal processing?
2. Define Decimator and Interpolator.
3. How aliasing effect is overcome in decimation?
4. Write the time domain relationship for upsampling and downsampling.
5. List the advantages of MDSP.

Description:

Input: Give the input signal

Output: Find the decimated and interpolated output signal for the given number of samples.

Algorithm:

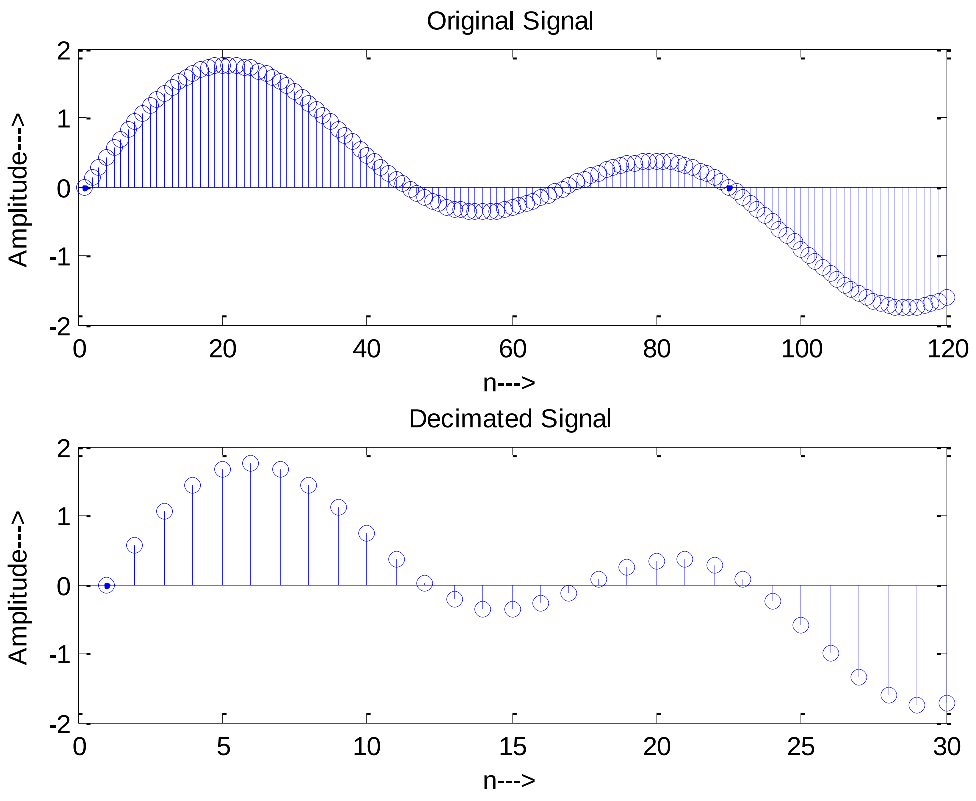
1. Start the program.
2. Get the amplitude of the input signal.
3. Get the decimation and interpolation factor.
4. Perform the decimation and interpolation operations using Matlab functions.
5. Plot the output signals.
6. Stop the process.

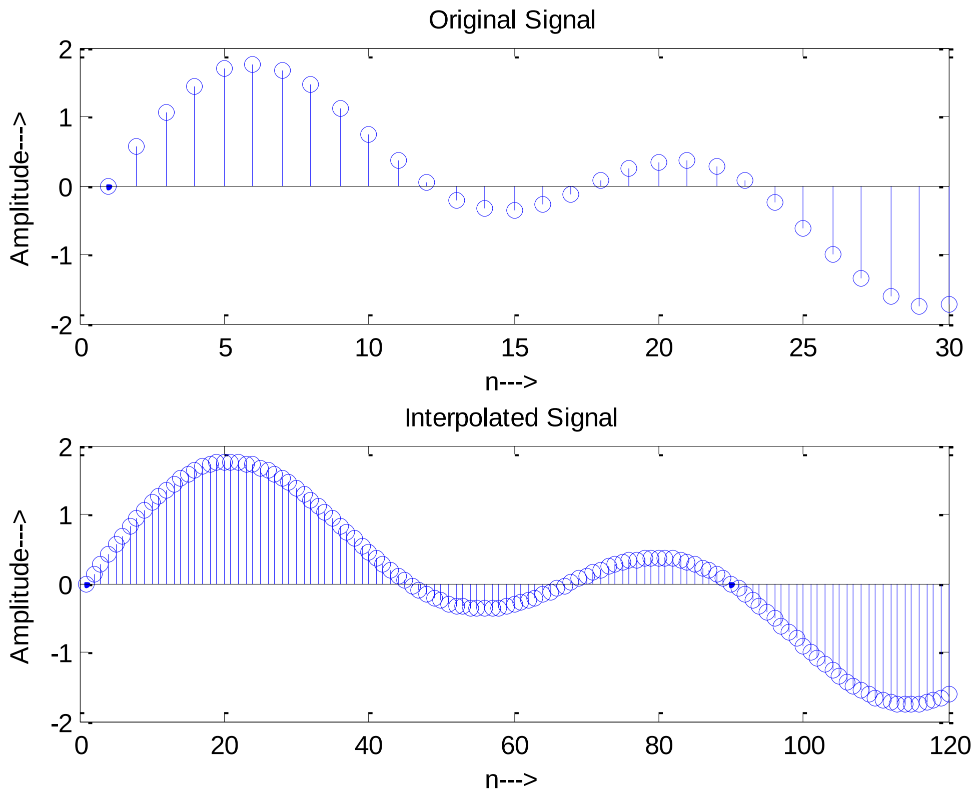
Program:

```
%Program for Decimate signal

clc;
clear all;
close all;
t=0:0.00025:1;    % Time vector
x=sin(2*pi*30*t)+sin(2*pi*60*t);
r=4;              % Factor of 4
y=decimate(x,r);

%View the original signal
figure(1);
subplot(2,1,1);
stem(x(1:120));
xlabel('n--->');
ylabel('Amplitude--->');
```

```
title('Original Signal');



%View the decimated signal
subplot(2,1,2);
stem(y(1:30));
xlabel('n--->');
ylabel('Amplitude--->');
title('Decimated Signal');

%Program for Interpolated signal

t=0:0.001:1;     % Time vector
x=sin(2*pi*30*t)+sin(2*pi*60*t);
r=4;            % Factor of 4
y=interp(x,r);
 %View the Original signal
figure(2);
subplot(2,1,1);
stem(x(1:30));
xlabel('n--->');
ylabel('Amplitude--->');
title('Original Signal');

%View the Interpolated Signal
subplot(2,1,2);
stem(y(1:120));
xlabel('n--->');
ylabel('Amplitude--->');
title('Interpolated Signal');
```

Model graph:





PostLab Questions:

1. What are the effects of upsampling?
2. List the applications of multirate signal processing?
3. What are sampling rate converters?
4. Define sub band coding of signals.
5. For the given signal x[n]= {1,3,4,5,6,7,8,9,1,-2} . Decimate and interpolate by a factor of 2.

Result:

Thus a program to perform decimation and interpolation the input signal was written using MATLAB and plotted.

Ex : No: **Date:**

7.EQUALIZATION

Aim:

To Design an Least Mean Square (LMS) channel equalizer using MATLAB.

Prerequisite Lab Questions:

1. Define adaptive filtering?
2. What are the two steps in adaptive filtering?
3. Whether FIR or IIR filters are used for adaptive filtering?
4. List the classifications of algorithms for adaptive filtering.
5. What is the need for adaptive equalization?

THEORY:

The Least Mean Square algorithm or non-recursive adaptive filter is routinely used in filtering applications that range from adaptive equalizers to adaptive noise control systems. The various reasons for its priority, the stability are controlled. There are simple and efficient algorithms to adjust the filter coefficients. Finally the performance of these algorithms is well understood in terms of convergence and stability. For an LMS algorithm having p+1 coefficient require P+1 multiplications and p additions to calculate the output. The simplicity of this algorithm comes from the update equation. And most importantly the data and LMS weight vector are statistically independent. The LMS algorithm is the most commonly used nonrecursive channel equalizer technique.

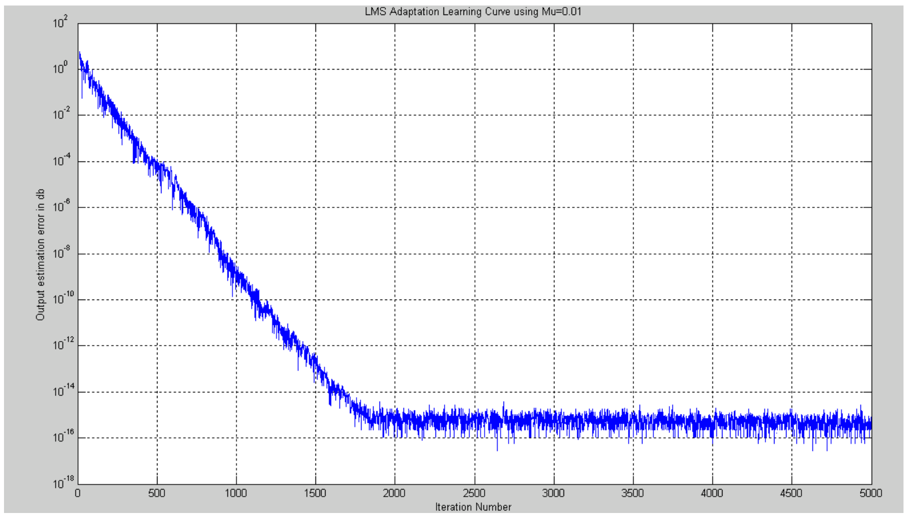
ALGORITHM:

1. The Input values are all set
2. For input generate normally distributed random matrix.
3. For filter generate uniform distributed pseudo random matrix.
4. The filter coefficients are determined.
5. The input vector is filtered with the specified vector and stored in another vector.
6. Now all the outputs and vectors are initialized.
7. Now the LMS adaptation curve is determined using the equation
w=w+mu(Re(e(n)*conj(in))-iImg(e(n)*conj(in)))
8. Finally the generated output waveform is plotted in a semilog graph sheet

```
% ****************************************************
%        LEAST MEAN SQUARE ALGORITHM
% ****************************************************
clear all;
close all;
randn('state',sum(100*clock)); % Returns a matrix with pseudorandom values in normal
distribution
rand('state',sum(100*clock));  % Returns a matrix with pseudorandom values in uniform
distribution
numpoints=5000;             % Sampling points
numtaps = 10;               % No of filter coefficients
Mu=0.01;
x=randn(numpoints,1)+i*randn(numpoints,1); % Complex Random Input Signal
```

```
h=rand(numtaps,1);                    % filter transfer function
h=h /max(h);
d = filter(h,1,x); % digital filter with num coeff h and den coeff 1 and vector x
w=[];
y=[];
in=[];
e=[];
w=zeros(numtaps+1,1)+i*zeros(numtaps+1,1); % initialize the w matrix
for n=numtaps+1:numpoints
   in=x(n:-1:n-numtaps);
   y(n)=w'*in;
   e(n)=d(n)-y(n); % Calculates the error value
   w=w+Mu*(real(e(n)*conj(in))-i*imag(real(e(n)*conj(in))));
end
figure(10);
semilogy(abs(e));grid on;
title(['LMS Adaptation Learning Curve using Mu=',num2str(Mu)]);
xlabel('Iteration Number');
ylabel('Output estimation error in db');
```

OUTPUT



PostLab Questions:

1. List the applications of adaptive filtering.
2. What do you mean by adaptive noise cancellation?
3. Define mean square error criterion.
4. What is in-phase and quadrature signals?
5. What are the characteristics of LMS algorithm?

RESULT :

Thus the Least Mean Square (LMS) channel equalizer was designed and the generated output sequence was plotted using MATLAB software.

Ex : No: **Date:**

8. STUDY OF ARCHITECTURE OF DIGITAL SIGNAL PROCESSOR

Aim:

To Study the architecture of TMS320C5416 Digital Signal Processor Starter Kit (DSK).

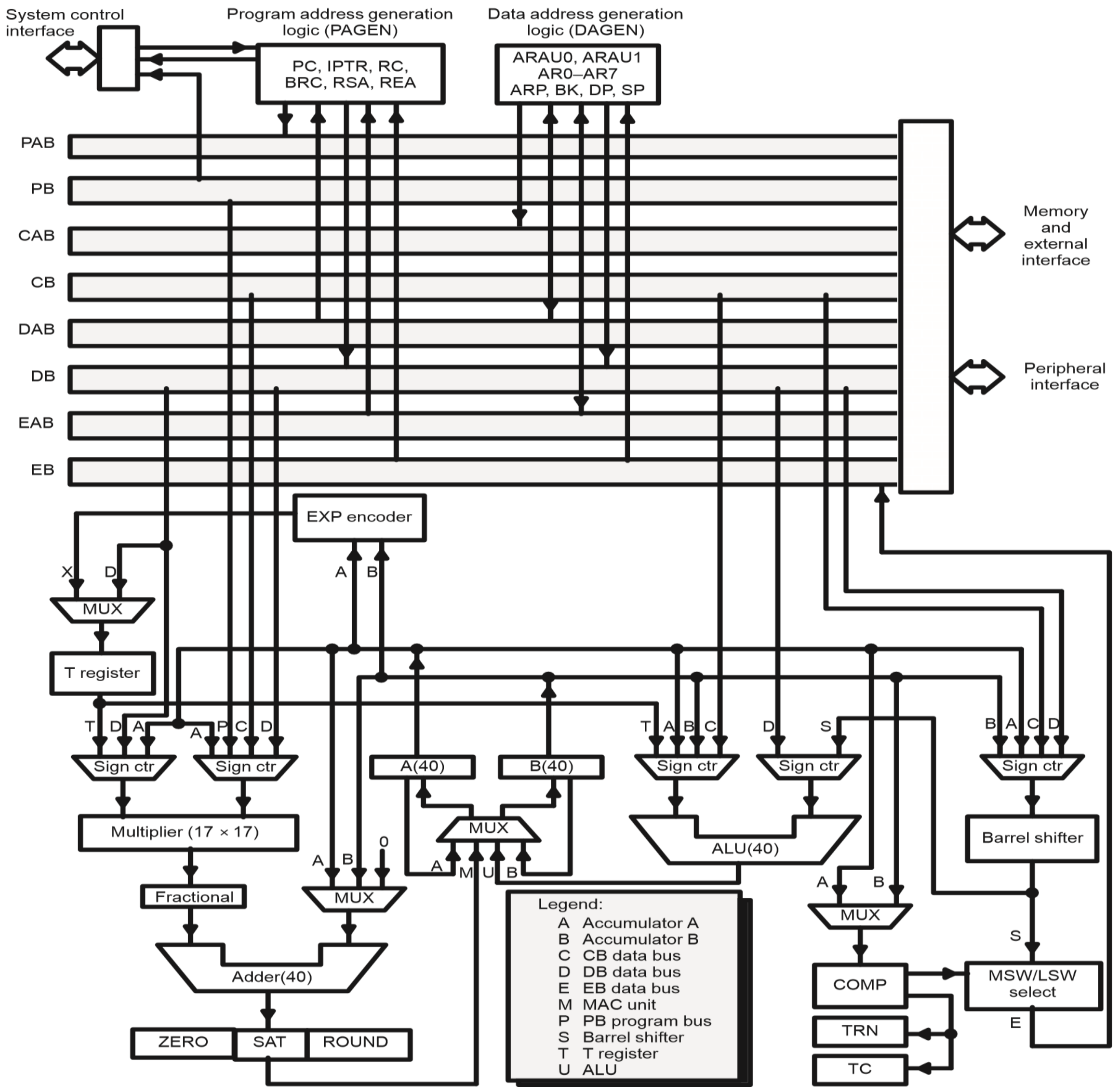
Description:

The TMS320VC5416 fixed-point, digital signal processor (DSP) (hereafter referred to as the device unless otherwise specified) is based on an advanced modified Harvard architecture that has one programMemory bus and three data memory buses. This processor provides an arithmetic logic unit (ALU) with aHigh degree of parallelism, application-specific hardware logic, on-chip memory, and additional on-chipPeripherals. The basis of the operational flexibility and speed of this DSP is a highly specialized instructionSet. Separate program and data spaces allow simultaneous access to program instructions and data, providing, a high degree of parallelism. Two read operations and one write operation can be performed in a single cycle. Instructions with parallel store and application-specific instructions can fully utilize this architecture. In addition, data can be transferred between data and program spaces. Such parallelism supports a powerful set of arithmetic, logic, and bit-manipulation operations that can all be performed in a single machine cycle. The device also includes the control mechanisms to manage interrupts, repeated
Operations, and function calls.

Architecture:

The "54x DSPs use an advanced, modified Harvard architecture that maximizes processing power by maintaining one program memory bus and three data memory buses. These processors also provide an arithmetic logic unit (ALU) that has a high degree of parallelism, application-specific hardware logic, on-chip memory, and additional on-chip peripherals. These DSP families also provide a highly specialized instruction set, which is the basis of the operational flexibility and speed of these DSPs. Separate program and data spaces allow simultaneous access to program instructions and data, providing the high degree of parallelism. Two reads and one write operation can be performed in a single cycle. Instructions with parallel store and application-specific instructions can fully utilize this architecture. In addition, data can be transferred between data and program spaces. Such parallelism supports a powerful set of arithmetic, logic, and bit-manipulation operations that can all be performed in a single machine cycle. Also included are the control mechanisms to manage interrupts, repeated operations, and function calls.

Architecture of TMS320VC5416 Fixed-Point Digital Signal Processor:



Central Processing Unit (CPU)

The CPU is common to all C54x™ devices. The C54x CPU contains:
40-bit arithmetic logic unit (ALU)
Two 40-bit accumulators
Barrel shifter
17 × 17-bit multiplier
40-bit adder
Compare, select, and store unit (CSSU)
Data address generation unit
Program address generation unit

Arithmetic Logic Unit (ALU)

The C54x DSP performs 2s-complement arithmetic with a 40-bit arithmetic logic unit (ALU)and two 40-bit accumulators(accumulators A and B). The ALU can also perform Boolean operations. The ALU uses these inputs:

16-bit immediate value
16-bit word from data memory
16-bit value in the temporary register, T
Two 16-bit words from data memory
32-bit word from data memory
40-bit word from either accumulator

The ALU can also function as two 16-bit ALUs and perform two 16-bit operations simultaneously.

Accumulators

Accumulators A and B store the output from the ALU or the multiplier/adder block. They can also provide a second input to the ALU; accumulator A can be an input to the multiplier/adder. Each accumulator is divided into three parts:

Guard bits (bits 39–32)
High-order word (bits 31–16) Low-order word (bits 15–0)

Instructions are provided for storing the guard bits, for storing the high- and the low-order accumulator words in data memory, and for transferring 32-bit accumulator words in or out of data memory. Also, either of the accumulators can be used as temporary storage.

Barrel Shifter

The C54x DSP barrel shifter has a 40-bit input connected to the accumulators or to data memory (using CB or DB), and a 40-bit output connected to the ALU or to data memory (using EB). The barrel shifter can produce a left shift of 0 to 31 bits and a right shift of 0 to 16 bits on the input data. The shift requirements are defined in the shift count field of the instruction, the shift count field (ASM) of status register ST1, or in temporary register T (when it is designated as a shift count register).

The barrel shifter and the exponent encoder normalize the values in an accumulator in a single cycle. The LSBs of the output are filled with 0s, and the MSBs can be either zero filled or sign extended, depending on the state of the sign-extension mode bit (SXM) in ST1. Additional shift capabilities enable the processor to perform numerical scaling, bit extraction, extended arithmetic, and overflow prevention operations.

Multiplier/Adder Unit

The multiplier/adder unit performs 17 17-bit 2s-complement multiplication with a 40-bit addition in a single instruction cycle. The multiplier/adder block consists of several elements: a multiplier, an adder, signed/unsigned input control logic, fractional control logic, a zero detector, a rounder (2s complement), overflow/saturation logic, and a 16-bit temporary storage register (T). The multiplier has two inputs: one input is selected from T, a data-memory operand, or accumulator A; the other is selected from program memory, data memory, accumulator A, or an immediate value.

The fast, on-chip multiplier allows the C54x DSP to perform operations efficiently such as convolution, correlation, and filtering. In addition, the multiplier and ALU together execute multiply/accumulate (MAC) computations and ALU operations in parallel in a single instruction cycle. This function is used in determining the Euclidian distance and in implementing symmetrical and LMS filters, which are required for complex DSP algorithms.

Compare, Select, and Store Unit (CSSU)

The compare, select, and store unit (CSSU) performs maximum comparisons between the accumulator's high and low word, allows both the test/control flag bit (TC) in status register ST0 and the transition register (TRN) to keep their transition histories, and selects the larger word in the accumulator to store into data memory. The CSSU also accelerates Viterbi-type butterfly computations with optimized on-chip hardware.

Status Registers (ST0, ST1)

The status registers ST0 and ST1 contain the status of the various conditions and modes for the C54x devices. ST0 contains the flags (OVA, OVB, C, and TC) produced by arithmetic operations and bit manipulations, in addition to the DP and the ARP fields. ST1 reflects the status of modes and instructions executed by the processor.

Auxiliary Registers (AR0–AR7)

The eight 16-bit auxiliary registers (AR0–AR7) can be accessed by the CPU and modified by the auxiliary register arithmetic units (ARAUs). The primary function of the auxiliary registers is to generate 16-bit addresses for data space. However, these registers can also act as general-purpose registers or counters.

Transition Register (TRN)

The 16-bit transition (TRN) register holds the transition decision for the path to new metrics to perform the Viterbi algorithm. The CMPS (compare select max and store) instruction updates the contents of TRN register on the basis of the comparison between the accumulator high word and the accumulator low word.

Temporary Register (T)

The temporary (T) register has many uses. For example, it may hold:

One of the multiplicands for multiply and multiply/accumulate instructions (For more details about the T register and the processes of multiplication.

A dynamic (execution-time programmable) shift count for instructions with shift operation such as the ADD, LD, and SUB instructions

A dynamic bit address for the BITT instruction

Branch metrics used by the DADST and DSADT instructions for ACS operation of Viterbi decoding

In addition, the EXP instruction stores the exponent value computed into T register, and then the NORM instruction uses the T register value to normalize the number.

Stack-Pointer Register (SP)

The 16-bit stack-pointer register (SP) contains the address of the top of the system stack. The SP always points to the last element pushed onto the stack. The stack is manipulated by interrupts, traps, calls, returns, and the PSHD, PSHM, POPD, and POPM instructions. Pushes and pops of the stack predecrement and postincrement, respectively, the 16-bit value in the stack pointer.

Circular-Buffer Size Register (BK)

The ARAUs use16-bit circular-buffer size register (BK) in circular addressing to specify the data block size. For information on BK and circular addressing.

Block-Repeat Registers (BRC, RSA, REA)

The 16-bit block-repeat counter (BRC) register specifies the number of times a block of code is to repeat when a block repeat is performed. The 16-bit blockrepeat start address (RSA) register contains the starting address of the block of program memory to be repeated. The 16-bit block-repeat end address (REA) register contains the ending address of the block of program memory to be repeated.

Memory-Mapped Registers

The 64K words of data memory space include the device’s memory-mapped registers, which reside in data page 0 (data addresses 0000h–007Fh).

The peripheral registers are used as control and data registers in peripheral circuits. These registers reside within addresses 0020h–005F and reside on a dedicated peripheral bus structure. For a list of peripherals on a particular C54x device.

The scratch-pad RAM block (60h–7Fh in data memory) includes 32 words of DARAM for variable storage that helps avoid fragmenting the large RAM block.The data memory space contains memory-mapped registers for the CPU and the on-chip peripherals. Simplifying access to them. The memory-mapped access provides a convenient way to save and restore the registers for context switches and to transfer information between the accumulators and the other registers.

Data Memory

The data memory space addresses up to 64K of 16-bit words. The device automatically accesses theOn-chip RAM when addressing within its bounds. When an address is generated outside the RAM bounds,the device automatically generates an external access.

The advantages of operating from on-chip memory are as follows:

· Higher performance because no wait states are required

· Higher performance because of better flow within the pipeline of the central arithmetic logic unit

(CALU)

· Lower cost than external memory
· Lower power than external memory
The advantage of operating from off-chip memory is the ability to access a larger address space.

Program Memory

Software can configure their memory cells to reside inside or outside of the program address map. When the cells are mapped into program space, the device automatically accesses them when their addressesare within bounds. When the program-address generation (PAGEN) logic generates an address outside itsBounds, the device automatically generates an external access.

The advantages of operating from on-chip memory are as follows:
· Higher performance because no wait states are required
· Lower cost than external memory
· Lower power than external memory
The advantage of operating from off-chip memory is the ability to access a larger address space.

Overview of the TMS320VC5416 DSK

The TMS320VC5416 DSK is a stand-alone development and evaluation module. It allows evaluators to examine certain characteristics of the C5416 digital signal processor (DSP) to determine if it meets their application requirements. Furthermore, the module is an excellent platform to develop and run software for the TMS320VC5416 family of processors.

The DSK allows full speed verification of VC5416 code. With 64K words of on board RAM memory, 256K words of on board Flash ROM, and a Burr Brown PCM 3002 stereo codec, the board can solve a variety of problems as shipped. Three expansion connectors are provided for interfacing to evaluation circuitry not provided on the as shipped configuration.

To simplify code development and shorten debugging time, a special version of Code Composer Studio is shipped with the board.

Key Features of the TMS320VC5416 DSK

The VC5416 DSK has the following features:
• VC5416 operating at 16-160 MHz
• On board USB JTAG controller with plug and play drivers
• 64K words of on board RAM
• 256K words of on board Flash ROM
• 3 Expansion Connectors (Memory Interface, Peripheral Interface, and Host Port Interface)
• On board IEEE 1149.1 JTAG Connection for Optional Emulation Debug
• Burr Brown PCM 3002 Stereo Codec
• +5 volt operation

Code Composer Studio - 3.1v

INTRODUCTION TO CODE COMPOSER STUDIO

Code Composer is the DSP industry's first fully integrated development environment (IDE) with DSP-specific functionality. With a familiar environment liked MS-based C++TM, Code Composer lets you edit, build, debug, profile and manage projects from a single unified environment. Other unique features include graphical signal analysis, injection/extraction of data signals via file I/O, multi-processor debugging, automated testing and customization via a C-interpretive scripting language and much more.

Initial setup to work on Code Composer Studio:

- To execute using Simulator OpenSetup CCStudio_v3.1 add C5416 Device Simulator Save and Quit.

(OR)

- To Execute using DSP Processor Kit -→DSK Diagnostic Utility→Start
  Check for PASS in the display and continue the Procedure
- Select 5416 dskCCStudio_v3.1→Debug→Connect

Procedure to work on Code Composer Studio:

To create the New Project

Project → New → (C:\CCStudio_v3.1\MyProjects\projectname**.**pjt)

To Create a Source file

File → New → Type the code (Save & give file name, Eg: filename**.c**or filename**.asm**).
(C:\CCStudio_v3.1\MyProjects\projectname**.**pjt\filename.c)

To Add Source files to Project

Project → Add files to Project →filename**.c**

To add library &cmd:

Project → Add files to Project →rts_EXT.lib
Library files: (Path: C:\ CCStudio\C5400\cgtools\lib\rts_EXT.lib)
Note: Select Object & Library from all files
Project → Add files to Project →c5416dsk.cmd
CMD file – Which is common for all non-real time programs.
(Path: Path: C:\ CCStudio\tutorial\dsk5416\shared\c5416dsk.cmd)
Note: Select Linker Command file (*.cmd) in Type of files

To Compile:

Project → Compile

To Build:

Project→ Build
If there is more number of warnings after build, then
Project→ Build Options → Advanced (in Category) →Use Far Calls (- mf)

Procedure to Load and Run program:

Load the program to DSK: **File**→ Load program →project name. Out

(C:\CCStudio_v3.1\MyProjects\projectname**.**pjt\Debug\projectname.out)

To execute project:

Debug→ Run.

To Plot Graph:

View → Graph (OR Right click in the program →Insert Graph)
Select proper Properties and specify the address.

Result:

Thusthe architecture of TMS320C5416 Digital Signal Processor Starter Kit (DSK) and programming steps for Code Composer Studio was studied.

Ex : No: **Date:**

9. MAC OPERATION USING VARIOUS ADRESSING MODES

Aim:

To perform MAC operation usingvarious addressing modes of TMS320C5416 DSP processor.

Prerequisite Lab Questions:

1. What are the classification of DSP Processors?
2. Compare fixed and floating point Processors ?
3. Brief the different types of addressing mode present in the TMS320C5416?
4. Define program counter.
5. What is the architectural difference between conventional and DSP proceesors?

Algorithm:

1. Start the program.
2. Store the first number from X memory to AR2.
3. Store the second number from Y memory to AR3.
4. Store the immediate address to AR4 to store the output.
5. Add both the numbers and store the result in accumulator A.
6. Store the lower order word to the memory location 1500 and higher order word to 1501 and increment the memory location
7. Load the content pointed by AR2 to the temporary register
8. Multiply the Content pointed by AR2 with the temporary register and store the result in B register.
9. Store the lower order word and higher order word of B register to the memory location pointed by AR4
10. Stop the program execution.

Program:

```
; Program for Addition

              .global _main
              .mmregs
              .data
X:            .word 50h
Y:            .word 20h
              .text
_main
              STM X,AR2
              STM Y,AR3
              STM #1500h,AR4
              ADD *AR2,*AR3,A
              STL A,*AR4+
     STH A,*AR4
```

```
        .end
```

Output:

```
0x1500: 0x0000
0x1501: 0x0070
```

```
; Program for Subtraction

               .global _main
               .mmregs
               .data
X:             .word 60h
Y:             .word 20h
               .text
_main
               STM X,AR2
               STM Y,AR3
               STM #1500h,AR4
               SUB *AR2,*AR3,A
               STL A,*AR4+
               STH A,*AR4
               .end
```

Output:

```
0x1500: 0x0000
0x1501: 0x0040
```

```
; Program for Multiplication

               .global _main
               .mmregs
               .data
X:             .word 30h
Y:             .word 20h
               .text
_main
               STM X,AR2
               STM Y,AR3
               STM #1500h,AR4
               MPY *AR2,*AR3,A
               STL A,*AR4+
               STH A,*AR4
               .end
```

Output:

```
0x1500: 0x0600
```

```
; Program for Division

                .global _main
                .mmregs
                .data
X:              .word 90h
Y:              .word 30h
                .text
_main
                STM X,AR2
                STM Y,AR3
                STM #1500h,AR4
                LD *AR2+,A
                RPT #15
                SUBC *AR3,A
                STL A,*AR4+
                STH A,*AR4
                .end
```

SUBC performs binary division like long division. For 16-bit by 16-bit integer division, the dividend is stored in low part accumulator A. The program repeats the SUBC command 16 times to produce a 16-bit quotient in low part accumulator A and a 16-bit remainder in high part accumulator B. For each SUBC subtraction that results in a negative answer, you must left-shift the accumulator by 1 bit. This corresponds to putting a 0 in the quotient when the divisor does not go into the dividend. For each subtraction that produces a positive answer, you must left shift the difference in the ALU output by 1 bit, add 1, and store the result in accumulator A. This corresponds to putting a 1 in the quotient when the divisor goes into the dividend.

Output:

```
0x1500: 0x0003
```

PostLab Questions:

1. Give an example for immediate and indirect addressing modes?
2. What do you mean by bit reversal addressing mode?
3. What are the registers available for indirect addressing mode?
4. What are barrel shifters?
5. What is arithmetic and logical shift in DSP Processors?

Result:

Thus the assembly coding for MAC operation usingvarious addressing modes of DSP processor TMS320C5416 was written and executed.

Ex :No: **Date:**

10.LINEAR CONVOLUTION

Aim:

Write a program to find out the linear convolution of two sequence using TMS320C5416 processor.

Prerequisite Lab Questions:

1. Define Convolution.
2. What is zero padding ?
3. List the properties of linear convolution.

Description:

Input: Give the input x (n) and h (n) in matrix form.

Output: y (n), using the formula

$$y(n) = \sum_{k=-\infty}^{\infty} x(k)h(n-k)$$

Algorithm:

1. Start the program.
2. Get the two input sequence x (n) and h (n) with zero appended.
3. Sum the product of x (n) and shifted sequence of h (n-k).
4. Execute the program, display the result and verify theoretically.
5. Stop the process.

PROGRAM FOR LINEAR CONVOLUTION

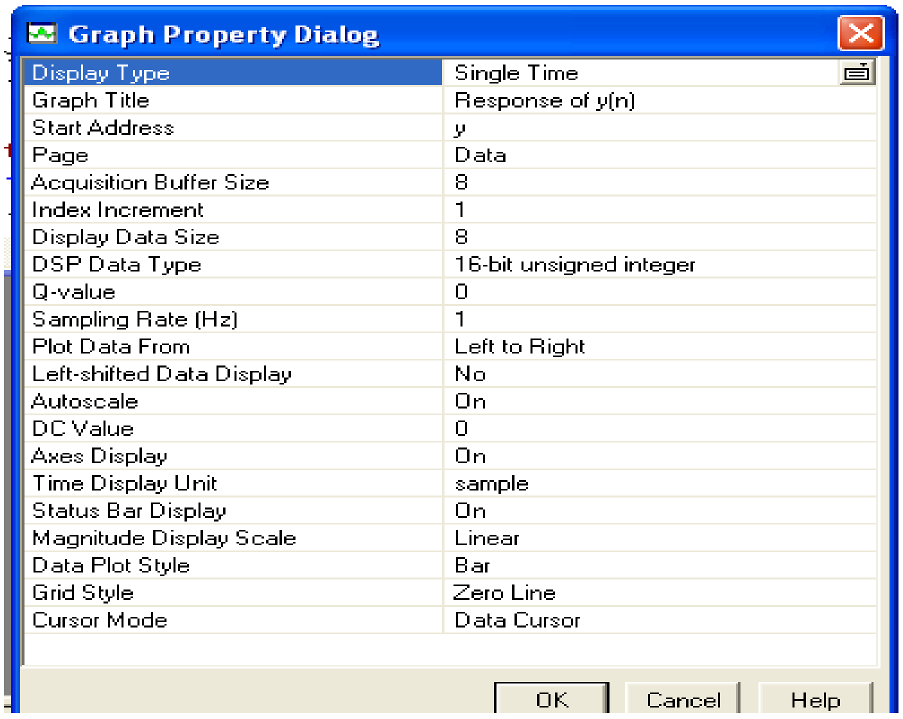
```
//PROGRAM FOR LINEAR CONVOLUTION

#include<stdio.h>
int x[10];              /*={1 2 3 4 0 0 0}     Input Signal Samples*/
int h[10];              /*={1 2 3 4 0 0 0}  Impulse Response     */
/*At the end of input sequences pad 'M' and 'N' no. of zero's*/
int y[10];
int m=4;                /*Lenght of input samples sequence x(n)*/
int n=4;                /*Lenght of impulse response h(n) */
int i,j;
main()
{
printf("  enter the  first sequence\n");
for(i=0;i<m+n-1;i++)
scanf("%d",&x[i]);
```

```
        printf("  enter the second sequence\n");
        for(j=0;j<m+n-1;j++)
        scanf("%d",&h[j]);
        for(i=0;i<m+n-1;i++)
        {
        y[i]=0;
        for(j=0;j<=i;j++)
                y[i]+=x[j]*h[i-j];
        }
        printf("output\n");
        for(i=0;i<m+n-1;i++)
        printf("%d\n",y[i]);
        }
```

Result:

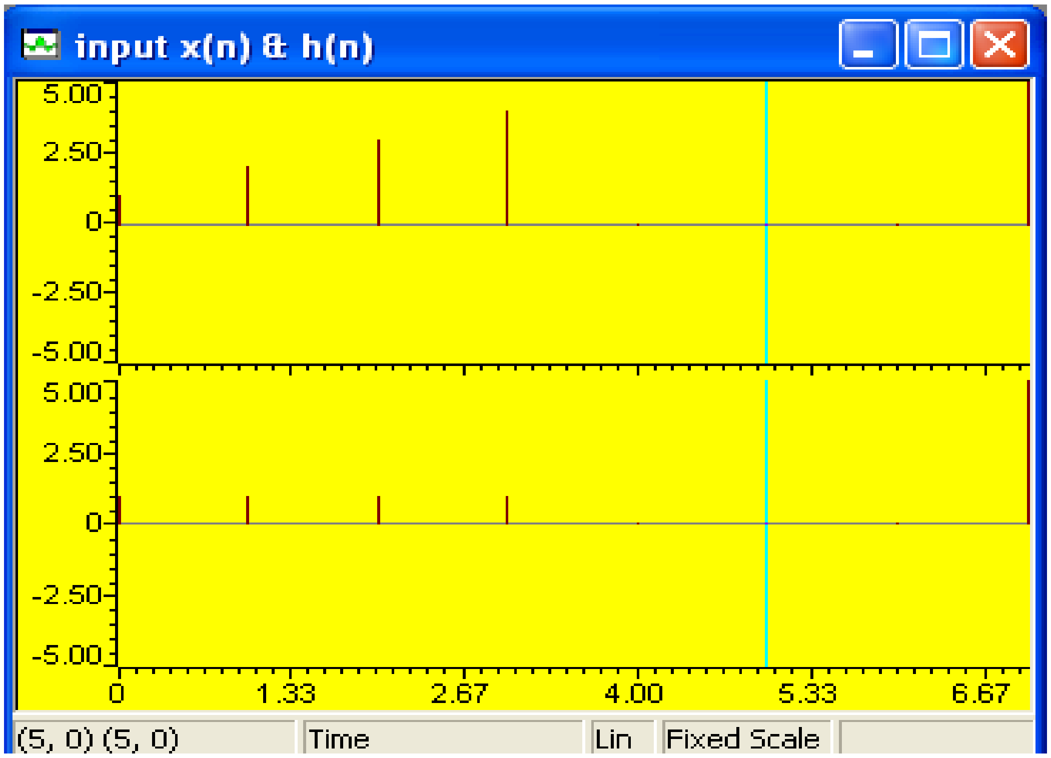
```
Enter the first sequence
1 2 3 4 0 0 0
Enter the second sequence
1 1 1 1 0 0 0
Output:
1
3
6
10
9
7
4
```

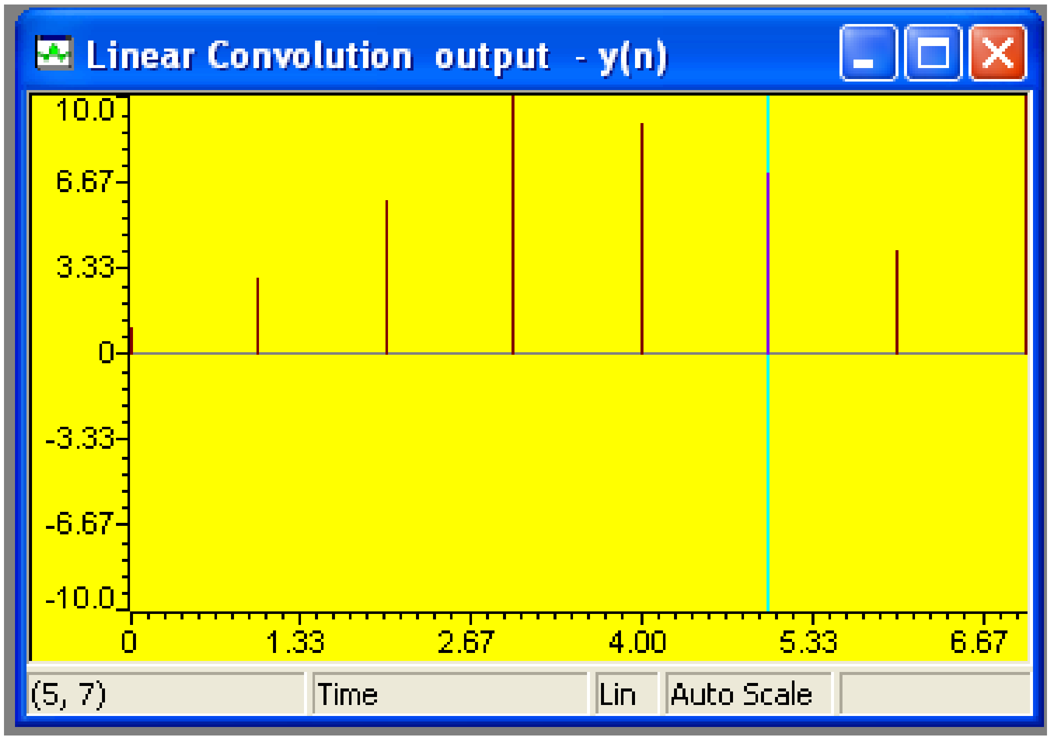
Procedure for Plot:



Display Type	Single Time
Graph Title	Response of y(n)
Start Address	y
Page	Data
Acquisition Buffer Size	8
Index Increment	1
Display Data Size	8
DSP Data Type	16-bit unsigned integer
Q-value	0
Sampling Rate (Hz)	1
Plot Data From	Left to Right
Left-shifted Data Display	No
Autoscale	On
DC Value	0
Axes Display	On
Time Display Unit	sample
Status Bar Display	On
Magnitude Display Scale	Linear
Data Plot Style	Bar
Grid Style	Zero Line
Cursor Mode	Data Cursor

MODEL GRAPH


input x(n) & h(n)
5.00
2.50
0
-2.50
-5.00
5.00
2.50
0
-2.50
-5.00
0
1.33
2.67
4.00
5.33
6.67
(5, 0) (5, 0)
Time
Lin
Fixed Scale


Linear Convolution output - y(n)
10.0
6.67
3.33
0
-3.33
-6.67
-10.0
0
1.33
2.67
4.00
5.33
6.67
(5, 7)
Time
Lin
Auto Scale

Postlab Questions :

1. State convolution property of a signal.
2. What are sectioned convolutions & what is the need of it?
3. Write the applications of convolution.

Result:

Thus a program to find out the linear convolution of two sequences using TMS320C5416 processor was written and executed.

Ex:No: **Date:**

11.CIRCULAR CONVOLUTION

Aim:

Write a program to find out the circular convolution of two sequences using TMS320C5416 processor.

Prerequisite Lab Questions:

1. What is the MATLAB function used to perform linear & circular convolution?
2. How will you determine the maximum length of the convolution of the sequences?
3. List the methods used to find the circular convolution.

Description:

Input: Give the input x (n) and h (n) in matrix form.

Output: y (n), using the formula

$y(n) = x(n) \otimes h(n)$

$$y(n) = \sum_{k=0}^{N-1} x(k)h(n-k)$$

Algorithm:

1. Start the program.
2. Get the two input sequence x (n) and h (n) and compute the length of the sequence.
3. Append zeros to x (n) or h (n) to make both sequence of same length.
4. Rotate the sequence h (-k), 'n' times. If 'n' is positive shift the sequence in anticlockwise direction else in the clockwise direction.
5. Determine the product of sequence x (n) and shifted h (n) sequence and sum it.
6. Execute the program, display the result and verify it theoretically.
7. Plot the output graph for each sequence.
8. Stop the process.

PROGRAM FOR CIRCULARCONVOLUTION

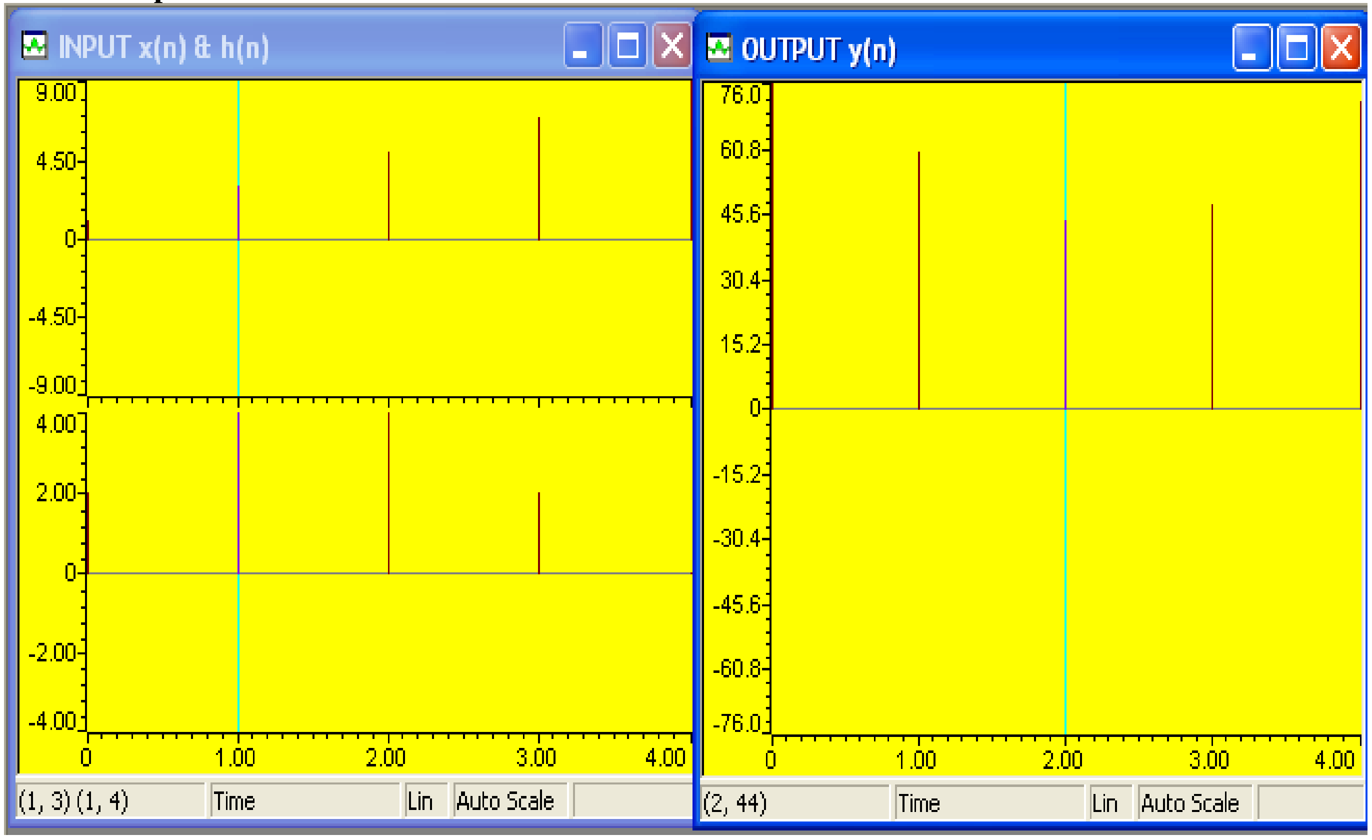
```
#include <stdio.h>

int N1,N2,N,x[30],h[30],y[30],i,j,k;

void main()
{
    printf("enter the length of first sequence\n");
    scanf("%d",&N1);
    printf("enter the length of second sequence\n");
    scanf("%d",&N2);
    printf("Enter first sequence elements\n");
    for(i=0;i<N1;i++)
        scanf("%d",&x[i]);
    printf("Enter second sequence elements\n");
    for(i=0;i<N2;i++)
        scanf("%d",&h[i]);
    if(N1>N2)
        {
            N=N1;
        }
    else
        {
            N=N2;
        }
    for(i=N1;i<N;i++)
        x[i]=0;
    for(j=N2;j<N;j++)
        h[j]=0;
    for(k=0;k<N;k++)
    {
        y[k]=0;
        for(i=0;i<N;i++)
        {
            j=k-i;
            if(j<0)
            {
                j=N+j;
            }
            y[k]=y[k]+(x[i]*h[j]);
        }
    }
    printf("The circular convolution is \n");
    for(k=0;k<N;k++)
        printf("%d\t",y[k]);

}
```

//PROGRAM FOR CIRCULAR CONVOLUTION

```
            #include<stdio.h>
            int m,n,x[30],h[30],y[30],i,j,temp[30],k,x2[30],a[30];
            void main()
```

```
{
printf("  enter the length of the first sequence\n");
scanf("%d",&m);
printf("  enter the length of the second sequence\n");
scanf("%d",&n);
printf("  enter the first sequence\n");
for(i=0;i<m;i++)
scanf("%d",&x[i]);
printf("  enter the second sequence\n");
for(j=0;j<n;j++)
scanf("%d",&h[j]);
if(m-n!=0)                      /*If length of both sequences are not equal*/
{
if(m>n)                         /* Pad the smaller sequence with zero*/
{
for(i=n;i<m;i++)
h[i]=0;
n=m;
}
for(i=m;i<n;i++)
x[i]=0;
m=n;
}
y[0]=0;
a[0]=h[0];
for(j=1;j<n;j++)                                        /*folding h(n) to h(-n)*/
a[j]=h[n-j];
 /*Circular convolution*/
for(i=0;i<n;i++)
y[0]+=x[i]*a[i];
for(k=1;k<n;k++)
{
y[k]=0;
/*circular shift*/
for(j=1;j<n;j++)
x2[j]=a[j-1];
x2[0]=a[n-1];
for(i=0;i<n;i++)
{
a[i]=x2[i];

y[k]+=x[i]*x2[i];
}
}
/*displaying the result*/
printf("  the circular convolution is\n");
for(i=0;i<n;i++)
printf("%d \t",y[i]);
}
```

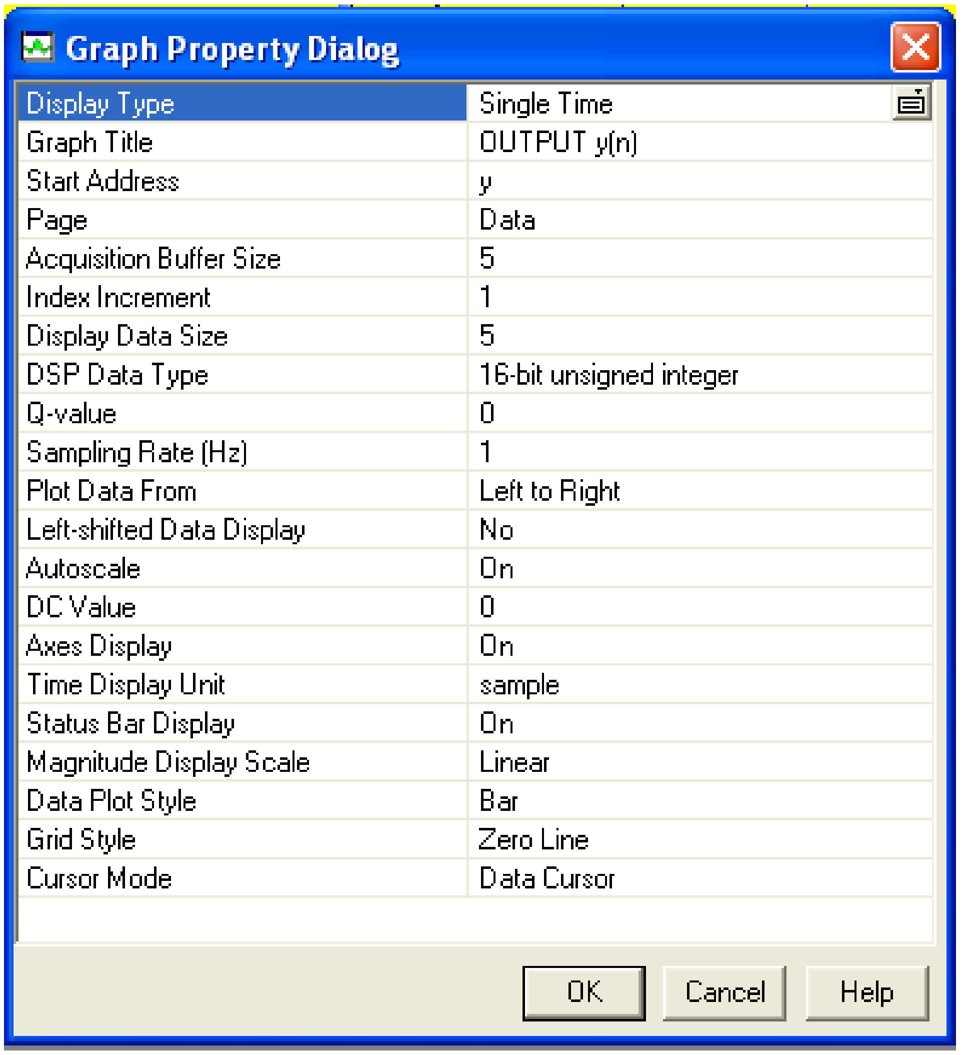
Model Graph:



Output:

```
enter the length of the first sequence
5
enter the length of the second sequence
4
enter the first sequence
1 3 5 7 9
enter the second sequence
2 4 4 2
the circular convolution is
76      60      44      48      72
```

Properties to Plot Graph:



Graph Property Dialog

Display Type	Single Time
Graph Title	OUTPUT y(n)
Start Address	y
Page	Data
Acquisition Buffer Size	5
Index Increment	1
Display Data Size	5
DSP Data Type	16-bit unsigned integer
Q-value	0
Sampling Rate (Hz)	1
Plot Data From	Left to Right
Left-shifted Data Display	No
Autoscale	On
DC Value	0
Axes Display	On
Time Display Unit	sample
Status Bar Display	On
Magnitude Display Scale	Linear
Data Plot Style	Bar
Grid Style	Zero Line
Cursor Mode	Data Cursor

PostLab Questions:

1. What is the difference between Linear convolution & circular convolution?
2. What is the relation between convolution in time domain and frequency domain ?
3. Give the significance of using convolution process in digital signal processing.

Result:

Thus a program to find out the circular convolution of two sequences using TMS320C5416 processor was written and executed.

Ex: No: **Date:**

12. FFT IMPLEMENTATION

Aim:

To write a program to perform Discrete Fourier transform using TMS320C5416 DSP Processor.

Prerequisite Lab Questions:

1. Write the analysis and synthesis equation for DFT?
2. What do you mean by the property of shifting of DFT?
3. Find the value of twiddle factor for N=8, K=2 .
4. How many multiplications and additions are required to compute N point DFT ?
5. What is the relationship between DFT and Z-transform?

Algorithm:

1. Start the program.
2. Declare the variables
3. Generate the input sequence
4. Using DFT, calculate the real and imaginary part of X (k) for k=0, 1... N-1
5. Calculate the Magnitude of X (k).
6. Print the Result.
7. Stop the Process.

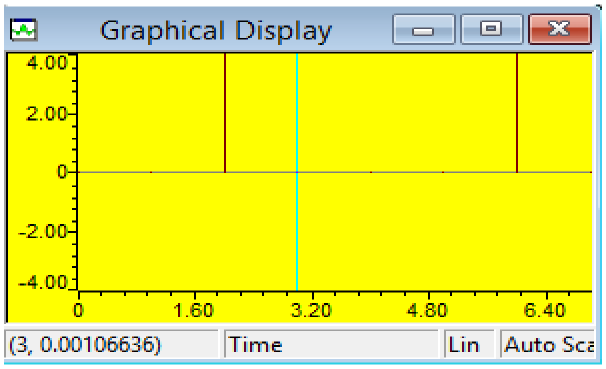
Program:

```
//Program for DFT
#include<stdio.h>
#include<math.h>
#define pi 3.1415
#define N 8
float X[K];
int i,k,n;
float xr[i],xi[i];
float XR[K],XI[K];
main()
{

 for (n=0;n<=N-1;n++)
{
xr[n]=sin(2*pi*10*n/8.0);
 xi[n]=0;
 }
 for (k=0;k<N;k++)
{
```

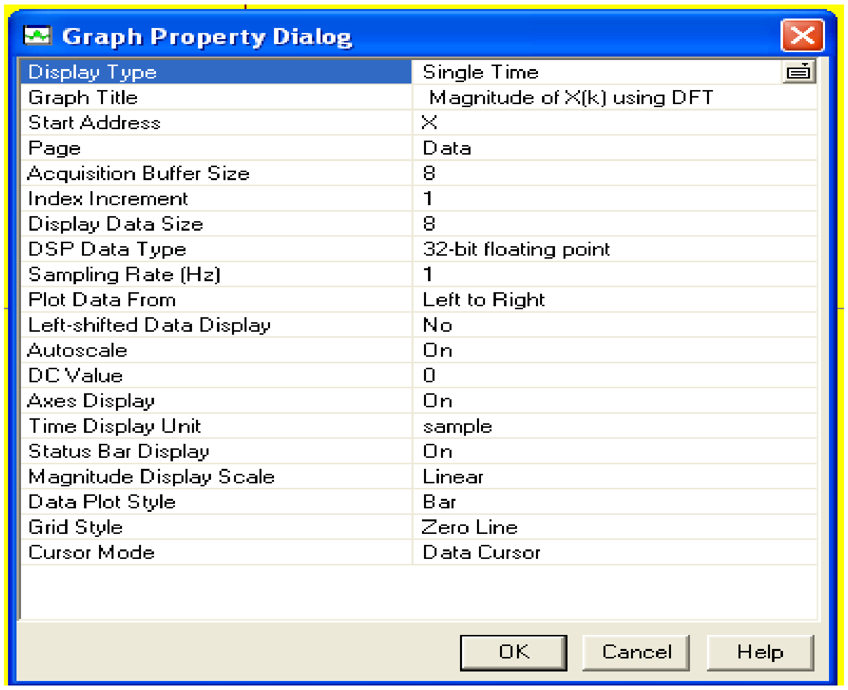
```
XR[k]=0;
 XI[k]=0;

for (n=0;n<N; n++)
{
XR[k]+=(xr[n]*cos(2*pi*k*n/N))+(xi[n]*sin(2*pi*k*n/N));
XI[k]+=(xr[n]*sin(2*pi*k*n/N))-(xi[n]*cos(2*pi*k*n/N));
}

X[k]=sqrt((XR[k]*XR[k])+(XI[k]*XI[k]));
 }
printf("X(k) and  magnitude of X(k)\n");

for (k=0;k<N;k++)
{
printf("\n X[%d] = %f + j %f   \n",k,XR[k],XI[k]);
printf(" magnitude of X(%d) = %f\n",k,X[k]);
}
}
   //Program for DFT
   #include<stdio.h>
   #include<math.h>
   #define pi 3.1415
   #define N 8
          int i,k,n;
   float X[N];
   float xr[N],xi[N];
   float XR[N],XI[N];
   main()
   {

    for (n=0;n<=N-1;n++)
   {
   xr[n]=sin(2*pi*10*n/8.0);
    xi[n]=0;
    }
    for (k=0;k<N;k++)
   {

XR[k]=0;
    XI[k]=0;

   for (n=0;n<N; n++)
   {
   XR[k]+=(xr[n]*cos(2*pi*k*n/N))+(xi[n]*sin(2*pi*k*n/N));
   XI[k]+=(xr[n]*sin(2*pi*k*n/N))-(xi[n]*cos(2*pi*k*n/N));
   }
```

```
        X[k]=sqrt((XR[k]*XR[k])+(XI[k]*XI[k]));
         }
        printf("X(k) and  magnitude of X(k)\n");

        for (k=0;k<N;k++)
        {
        printf("\n X[%d] = %f + j %f   \n",k,XR[k],XI[k]);
        printf(" magnitude of X(%d) = %f\n",k,X[k]);
        }
        }
```

Model Graph:



Graph Properties:



Graph Property Dialog

Display Type	Single Time
Graph Title	Magnitude of X(k) using DFT
Start Address	X
Page	Data
Acquisition Buffer Size	8
Index Increment	1
Display Data Size	8
DSP Data Type	32-bit floating point
Sampling Rate (Hz)	1
Plot Data From	Left to Right
Left-shifted Data Display	No
Autoscale	On
DC Value	0
Axes Display	On
Time Display Unit	sample
Status Bar Display	On
Magnitude Display Scale	Linear
Data Plot Style	Bar
Grid Style	Zero Line
Cursor Mode	Data Cursor

Sample Values:

X(k) and magnitude of X(k)

```
X[0] = 0.000926 + j 0.000000
 magnitude of X(0) = 0.000926

X[1] = 0.000927 + j -0.000795
 magnitude of X(1) = 0.001221

X[2] = -0.002035 + j 3.999998
 magnitude of X(2) = 3.999999

X[3] = 0.000925 + j 0.000531
 magnitude of X(3) = 0.001066

X[4] = 0.000924 + j -0.000373
 magnitude of X(4) = 0.000997

X[5] = 0.000923 + j -0.001582
 magnitude of X(5) = 0.001831

X[6] = -0.005004 + j -3.999995
 magnitude of X(6) = 3.999999

X[7] = 0.000927 + j 0.001844
 magnitude of X(7) = 0.002064
```

<u>Calculation:</u>
<u>Formula:</u>

$$X(k) = \sum_{k=0}^{M} x(n) \quad e^{-j2\pi\frac{nk}{n}} ;0 < k < N-1$$

<u>Postlab Questions :</u>

1. Mention the applications of DFT?
2. What is the drawback in Fourier transform and how it can be avoided?
3. Write the 4-point DFT matrix representation.
4. How many multiplications and additions are required to compute DFT for N=512?
5. Distinguish between DFT and DTFT.

Result:

Thus a program was written to perform Discrete Fourier Transform and executed using DSP processor.

Ex : No: **Date:**

13. WAVEFORM GENERATION

Aim:

To write a program to generate Sine wave and Ramp wave form using TMS320C5416 DSP Processor.

Algorithm:

1. Start the program.
2. Declare the output buffer to store the result.
3. Assign 0 to accumulator
4. Increment the variable by one and store the value in the output buffer
5. After the counter is incremented to 40, then decrement the value of the variable one by one from a negative going ramp
6. Again decrement the value from a negative going ramp in negative scale until the value becomes maximum negative value.
7. Decrement the maximum negative value one by one until the value of the variable becomes zero.

Program:

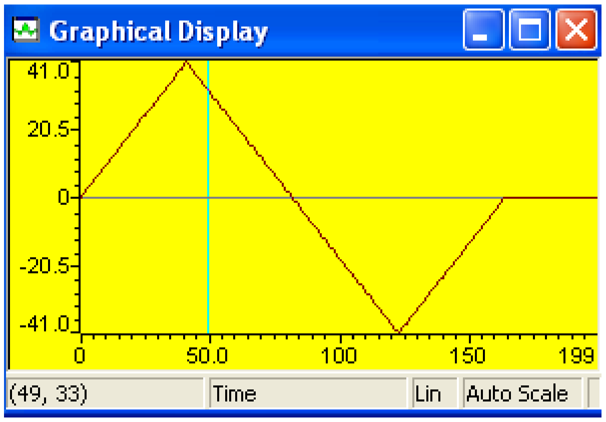
```
; Program for Triangular Wave Generation

                    .global _main
                    .mmregs
                    .def start
                    .data
                    .text

_main

start               STM #2000h,AR4
                    LD #0h,0,A

BEGIN               STM #40,BRC
                    RPTB RAMP
                    STL A,*AR4+
                    LD #1h,0,B

RAMP                ADD B,0,A
                    STM #40,BRC
                    RPTB RAMP1
                    STL A,*AR4+
                    LD #1h,0,B
```

```
RAMP1           SUB B,0,A
                STM #40,BRC
                RPTB RAMP2
                STL A,*AR4+
                LD #1h,0,B

RAMP2           SUB B,0,A
                STM #40,BRC
                RPTB RAMP3
                STL A,*AR4+
                LD #1h,0,B

RAMP3           ADD B,0,A
                STL A,*AR4+
                LD #1h,0,A
                SUB #1h,0,A
                BC BEGIN,ANEQ
                .end
```

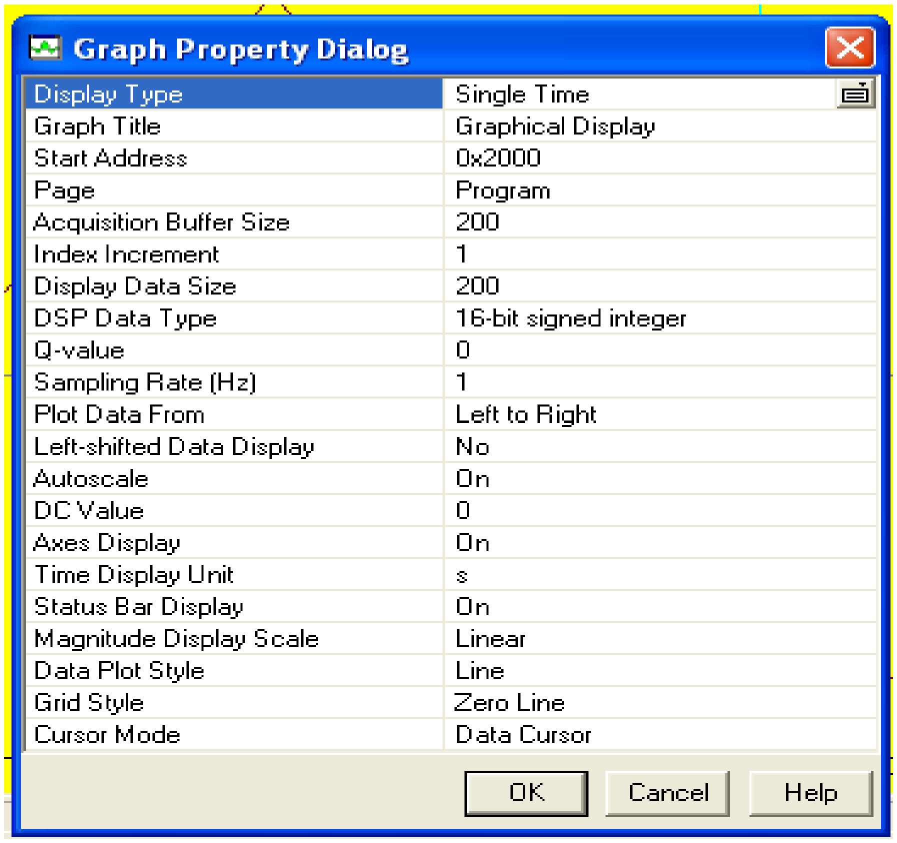
The repeat-block instructions are used to repeat a block of code N + 1 times, where N is the value loaded into the block-repeat counter register (BRC).

BRC contains the value N, which is one less than the number of times the block is to be repeated. The block-repeat start address register (RSA) holds the address of the first instruction of the block of code to be repeated. The block-repeat end address register (REA) holds the address of the last instruction word of the block of code to be repeat

Model Graph:



Graph Property:

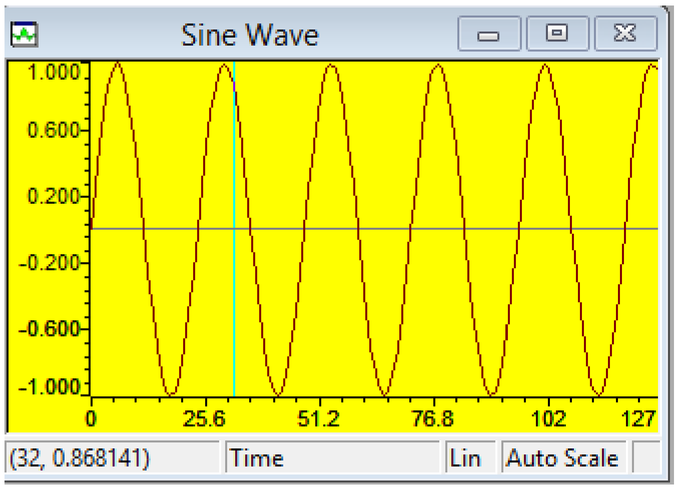


Display Type	Single Time
Graph Title	Graphical Display
Start Address	0x2000
Page	Program
Acquisition Buffer Size	200
Index Increment	1
Display Data Size	200
DSP Data Type	16-bit signed integer
Q-value	0
Sampling Rate (Hz)	1
Plot Data From	Left to Right
Left-shifted Data Display	No
Autoscale	On
DC Value	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

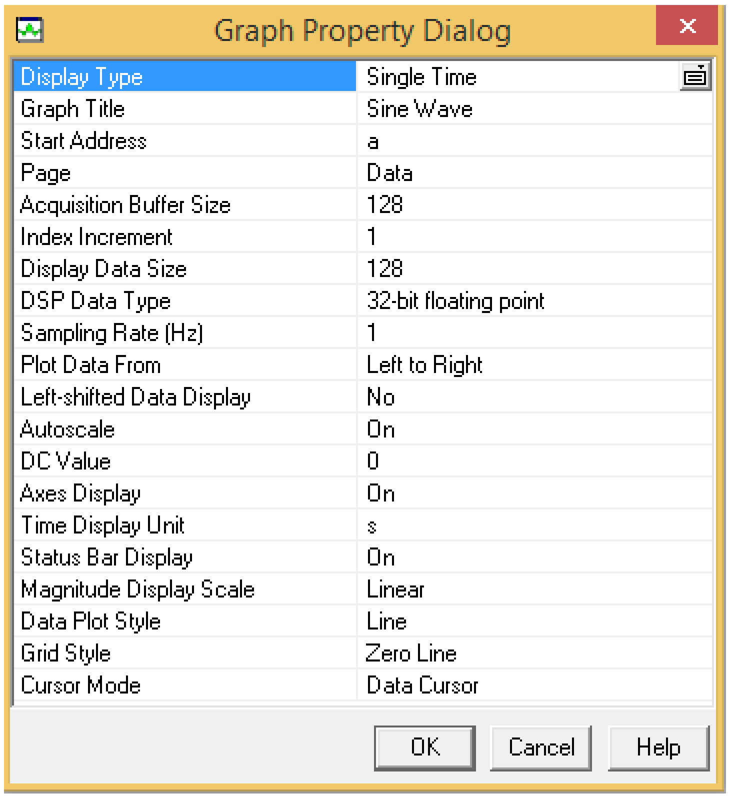
Program:

```
//Program for Sine Wave Generation

#include <stdio.h>
#include<math.h>
#define FREQ 1000

float a[128];
main()
{
 int i=0;

 for(i=0;i<128;i++)
          {
           a[i]=sin(2*3.14*FREQ*i/24000);
          printf("%f",a[i]);
          }
}
```

Model Graph:



Graph Property:



Graph Property Dialog

Display Type	Single Time
Graph Title	Sine Wave
Start Address	a
Page	Data
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	No
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

Result:

Thus the program was written to generate a Ramp signal, Sine signal and executed using DSP processor.

Ex: No: **Date:**

14.IIR AND FIR IMPLEMENTATION

Aim:

To write a program to implement Low pass filter using Finite Impulse Response in TMS320C5416 DSP Processor.

Prerequisite Lab Questions:

1. What are the properties of FIR filter?
2. When cascade from realization is preferred in FIR filters?
3. What is the reason that FIR filter is always stable?
4. What are the desirable characteristics of the windows?

Description:

Input: Give the cutoff frequency and order of the filter

Output: 1. Find the ideal impulse response hd (n) of the filter order.
2. Find the window coefficients w (n) using the appropriate window Function.
3. Find the filter coefficients h (n) =hd (n)*w (n).
4. Plot hd (n), w (n) and h (n).

Algorithm:

1. Start the program.
2. Get the cutoff frequency and order as an input...
3. Select a window function w (n).
4. Find the ideal impulse response of the filter hd (n).
5. Find the filter coefficients h (n) by multiplying hd (n) and w (n).
6. Execute the program, display the result and verify it.
7. Plot the graph for hd (n), w (n) and h (n) for the low pass filter.
8. Stop the process.

Program:

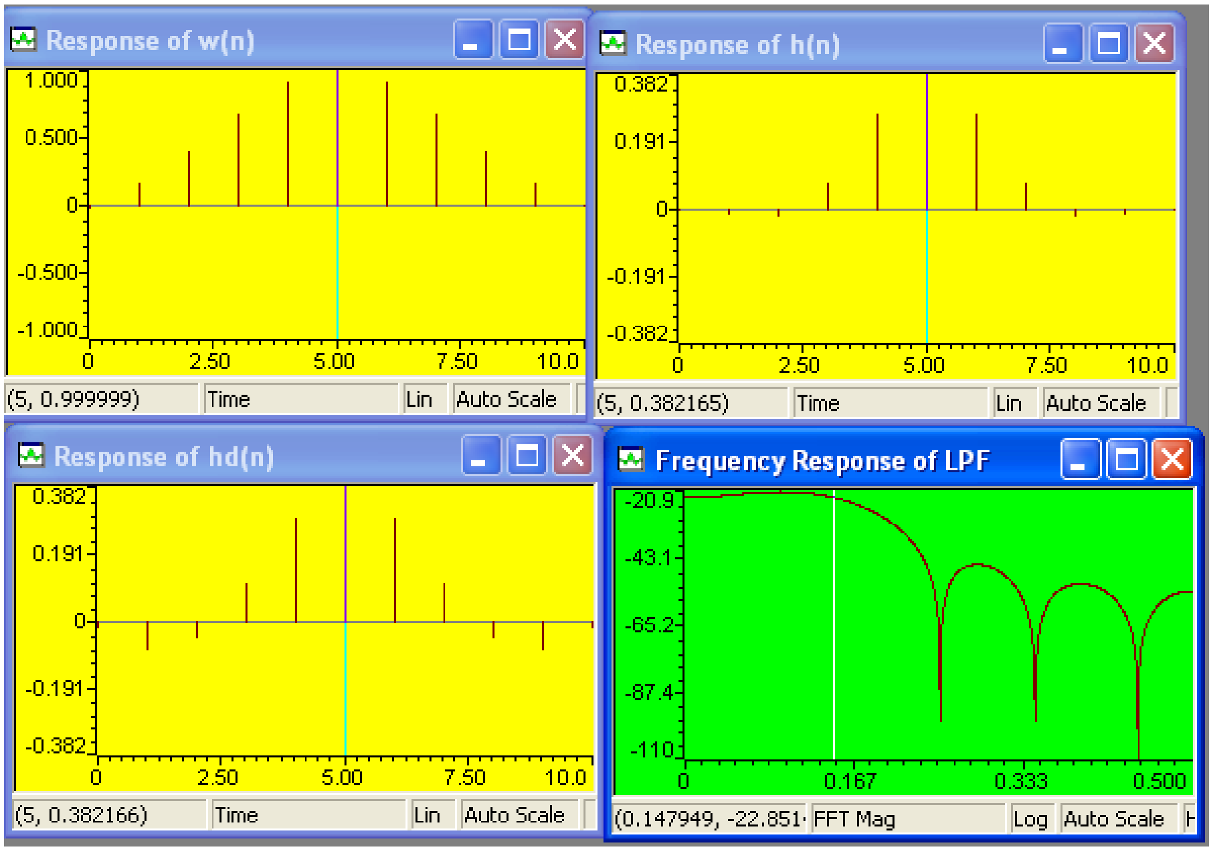
```
//Program for Fir

#include<stdio.h>
#include<math.h>
#define pi 3.14
int n,N;
float  hd[10],alpha,w[10],h[10],wc;
main()
{
printf("enter N value");
```

```
scanf("%d",&N);
printf("enter wc value");
scanf("%f",&wc);
for(n=0;n<N;n++)
{
/*Hamming window function*/
w[n]=0.54-0.46*(cos((2*pi*n)/(N-1)));
printf("w[%d]= %f \n",n,w[n]);
}
alpha=(N-1)/2;
/* Impulse function */
for(n=0;n<N;n++)
{
if(n!=alpha)
hd[n]=sin(wc*(n-alpha))/(pi*(n-alpha));
else
hd[n]=wc/3.14;
printf("hd[%d]= %f\n",n,hd[n]);
}
/*Filter coefficients*/
for(n=0;n<N;n++)
{
h[n]=hd[n]*w[n];
printf("h[%d]= %f\n",n,h[n]);
}
```

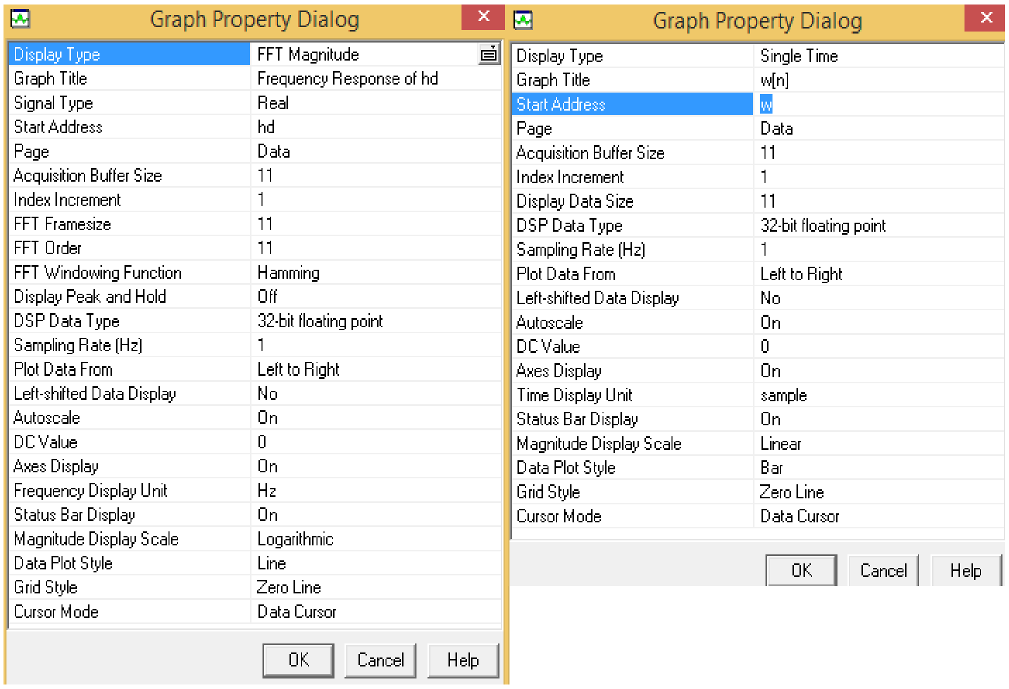
Output:

```
enter N value11
enter wc value1.2

 window coefficients:
w[0]= 0.080000       w[1]= 0.167766       w[2]= 0.397574       w[3]= 0.681730
w[4]= 0.911803       w[5]= 0.999999       w[6]= 0.912664       w[7]= 0.683123
w[8]= 0.398967        w[9]= 0.168629      w[10]= 0.080002

Ideal impulse response:
hd[0]= -0.017797     hd[1]= -0.079312     hd[2]= -0.046977     hd[3]= 0.107558
hd[4]= 0.296828      hd[5]= 0.382166      hd[6]= 0.296828      hd[7]= 0.107558
hd[8]= -0.046977     hd[9]= -0.079312     hd[10]= -0.017797


Low pass filter coefficients:
h[0]= -0.001424      h[1]= -0.013306      h[2]= -0.018677      h[3]= 0.073325
h[4]= 0.270648 h[5]= 0.382165       h[6]= 0.270904       h[7]= 0.073475
h[8]= -0.018742      h[9]= -0.013374      h[10]= 0.000025
```

Model Graph:



Graph Property:



Display Type	FFT Magnitude
Graph Title	Frequency Response of hd
Signal Type	Real
Start Address	hd
Page	Data
Acquisition Buffer Size	11
Index Increment	1
FFT Framesize	11
FFT Order	11
FFT Windowing Function	Hamming
Display Peak and Hold	Off
DSP Data Type	32-bit floating point
Sampling Rate (Hz)	1
Plot Data From	Left to Right
Left-shifted Data Display	No
Autoscale	On
DC Value	0
Axes Display	On
Frequency Display Unit	Hz
Status Bar Display	On
Magnitude Display Scale	Logarithmic
Data Plot Style	Line
Grid Style	Zero Line
Cursor Mode	Data Cursor

Display Type	Single Time
Graph Title	w[n]
Start Address	w
Page	Data
Acquisition Buffer Size	11
Index Increment	1
Display Data Size	11
DSP Data Type	32-bit floating point
Sampling Rate (Hz)	1
Plot Data From	Left to Right
Left-shifted Data Display	No
Autoscale	On
DC Value	0
Axes Display	On
Time Display Unit	sample
Status Bar Display	On
Magnitude Display Scale	Linear
Data Plot Style	Bar
Grid Style	Zero Line
Cursor Mode	Data Cursor

Theoretical calculation:

1. hd(n)=1/2π ∫ Hd(w)e^{jwn}dw
2. Select w (n)
3. Find h (n) = hd (n) w (n)

Rectangular Window

$$w(n) = \begin{cases} 1, & 0 \le n \le M-1 \\ 0, & otherwise \end{cases}$$

Hanning Window

$$w(n) = \begin{cases} 0.5 * \left[1 - \cos\left(\frac{2\pi n}{M-1}\right)\right], & 0 \le n \le M-1 \\ 0, & otherwise \end{cases}$$

Hamming Window

2. $$w(n) = \begin{cases} 0.54 - 0.46 * \cos\left(\frac{2\pi n}{M-1}\right), & 0 \le n \le M-1 \\ 0, & otherwise \end{cases}$$

Post Lab Questions:

1. What is the necessary and sufficient condition for linear phase characteristics in FIR filter?
2. What is the principle of designing FIR filter using frequency sampling method?
3. What is the principle of designing FIR filter using frequency sampling method?
4. What are the disadvantages of' Fourier series method?

Result:

Thus a program was written to implement Low pass filter using Finite Impulse Response and executed using DSP processor.

Ex No : **Date :**

15. FINITE WORD LENGTH EFFECT

Aim:

To study the functions of finite word length effect in fixed point DSP systems..

Apparatus Required:

Hardware : Personal Computer & TMS320C67XX kit
Software : Code Composer Studio version4

Prerequisite Lab Questions:

1. What is the difference between rounding and tradeoff?
2. Define finite word length effect
3. What is Limit cycle oscillation?
4. What do you mean by dead band?
5. What is the need for scaling?

Program:

```
function ADCNoiseGain=ADCNoise(b,a,n,FM) [B,A] = sos2tf([b a]);
%form A(z) and B(z) [h,t] = impz(B,A,n);
ADCNoiseGain = sum(h.^2)/12.0;
fprintf('ADC noise gain is %f\n\n',ADCNoiseGain);
if FM~=1
fprintf('ADC noise is %g^2*%g*q^2\n',[FM ADCNoiseGain]);
else
fprintf('ADC noise is %g*q^2\n',ADCNoiseGain);
 end function
CoeffQuantizeErr(b,a,maxbits,ftype,f,Fs)
%COEFFICIENT QUANTIZATION ERROR ANALYSIS
n=256;
for nbits=2:maxbits [B,A]=QuantizeCoeff(b,a,nbits);
[B,A] = sos2tf([B A]);
[h,w] = freqz(B,A,n);
amag = abs(h);
amag = amag/max(amag);
fprintf('\n\nnbits\tstage\tA1\tA2\tradius1\tangle1\tradiu s2\tangle2\n');
for nbits=2:maxbits [B,A]=QuantizeCoeff(b,a,nbits);
for i=1:size(b,1) r1 = sqrt(abs(A(i,3)));
angle1 = 180/pi*acos(A(i,2)/(-2.0*r1));
r2 = sqrt(abs(a(i,3)));
angle2 = 180/pi*acos(a(i,2)/(-2.0*r2));
fprintf('%d\t%d\t%-7.4f\t%-7.4f\t%-7.4f\t%-7.2f\t%- 7.4f\t%-7.2f\n', nbits, i,A(i,2) ,
A(i,3), r1,angle1,r2,angle2);
end
end format;
```

```
function ScaleFactor(b,a,nstep,size,structure) norm0 = DirectScale(b,a,0,nstep);
norm1 = DirectScale(b,a,1,nstep);
norm2 = DirectScale(b,a,2,size);
else norm0 = CanonicScale(b,a,0,nstep);
norm1 = CanonicScale(b,a,1,nstep);
norm2 = CanonicScale(b,a,2,size);
end disp('L1-norms of the second order sections:');
disp(norm0); disp('L2-norms of the second order sections:');
disp(norm1); disp('Loo-norms of the second order sections:');
disp(norm2);
function s = DirectScale(b,a,iopt,n)
if(iopt>=3) s = ones(1,size(b,1));
 return;
else
A = 1; B = 1;
for i=1:size(b,1)
%loop for each stage
A = conv(A,a(i,:));
B = conv(B,b(i,:));
s(i) = GetScaleFactor(B,A,iopt,n);
end
end function
s = CanonicScale(b,a,iopt,n)
if(iopt>=3)
s = ones(1,size(b,1));
 return;
else
A = 1;
B = 1;
for i=1:size(b,1)
A = conv(A,a(i,:));
if i>1 B = conv(B,b(i-1,:));
 end
s(i) = GetScaleFactor(B,A,iopt,n);
 end
end
```

Post Lab Questions:

1. What are the advantages of floating point arithmetic?
2. What are the methods to prevent overflow?
3. What are the two kinds of limit cycles?
4. What is the relationship between truncation error e and the bits b for representing a decimal into binary?
5. What are the three quantization errors to finite word length registers in digital filters?

Result:

Thus the function of finite word length effect in fixed point DSP processor is studied

Ex :No: **Date:**

16. INTRODUCTION TO IMAGE PROCESSING TOOLBOX

Aim:

To study the basic operations in image processing using MATLAB program.

Prerequisite Lab Questions:

1. Define image processing.
2. What is a digital image?
3. List the applications of image processing techniques.
4. How will you load and display an image?
5. How will you define 2-D DFT of an image?

Description:

Input: Give input images from MATLAB edit window.

Output: Perform various image processing techniques given and display it in different Figure window.

Algorithm:

Load an image in edit window and perform the following operations:

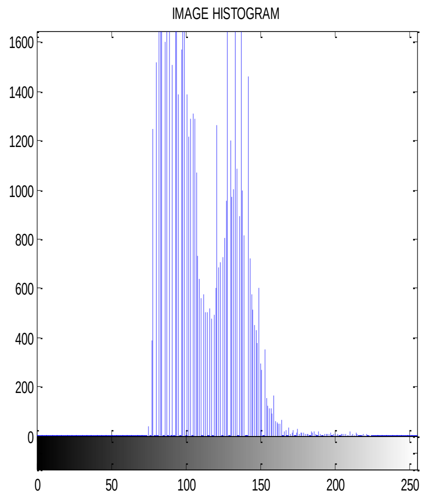
1. Reading and displaying an image
2. Improving the contrast of an image
3. Addition of images
4. Displaying an color image
5. Converting a color image to Gray scale image

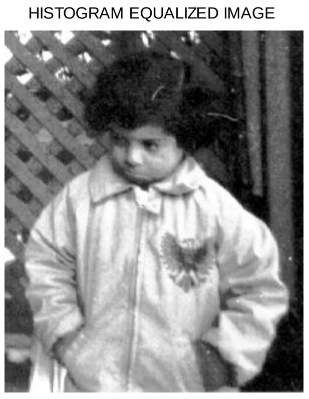
Program:

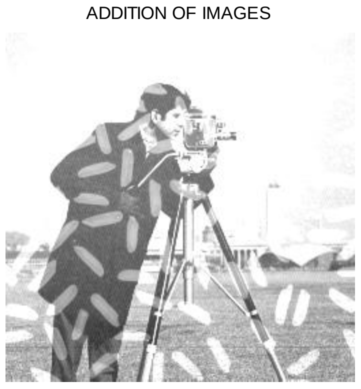
```
% PROGRAM FOR READING AND DISPALYING AN IMAGE
clc;
clear all;
close all;
I = imread('pout.tif');
figure, imshow(I)

% PROGRAM FOR IMPROVING THE CONTRAST OF AN IMAGE
figure, imhist(I)
I2 = histeq(I);
figure, imshow(I2)

%PROGRAM FOR ADDING THE IMAGES
```

```
 I = imread('rice.png');
 J = imread('cameraman.tif');
 K = imadd(I,J);
 figure, imshow(K)
% PROGRAM FOR  CONVERTING COLOR IMAGE TO GRAY SCALE
I = imread('board.tif');
J = rgb2gray(I);
figure, imshow(I), figure, imshow(J);
```

Sample Output

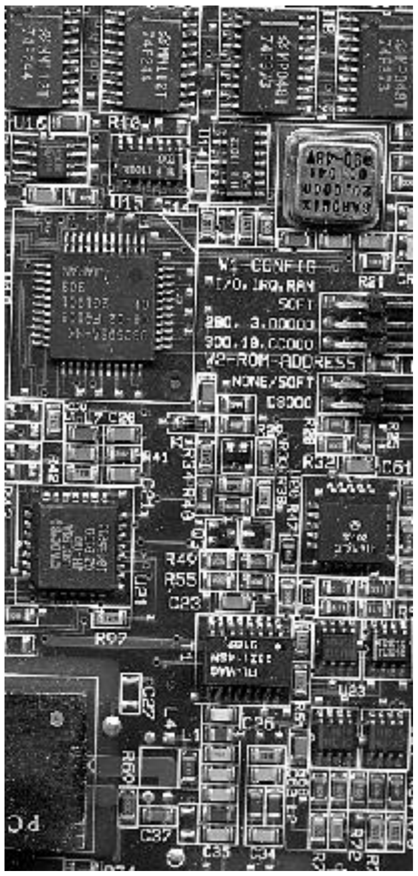






COLOR IMAGE

GRAY SCALE IMAGE



PostLab Questions:

1. Mention the applications of image processing techniques.
2. How will you improve the contrast of an image?
3. What are the various mathematical operations done in an image?
4. Define spatial and frequency domain?
5. How to perform 2-D DFT of an image?

Result:

Thus a MATLAB program for studying the image processing tool box was written, executed and the graphs were plotted.

Ex No : **Date :**

17.GENERATION AND DETECTION OF DTMF SIGNAL USING DFT

Aim:

To study the applications of Dual Tone Multi- Frequency (DTMF) signals in telephones and to generate and visualize the DTMF signals using MATLAB program.

Prerequisite Lab Questions:

1. What are the applications of DTMF signals?
2. List the various low and high frequency signals used for DTMF generation.
3. How DFT is used for DTMF generation?
4. Draw the basic symbols in a telephone key pad.
5. What do you mean by DTMF generation and detection?

Description:

Input: Generate sine input signal for low and high frequencies.

Output:

i. Represent the generated low and high frequency signals using DFT.
ii. Generate the Matlab code for the generation and visualization of the DTMF tones

Algorithm:

1. Start the program.
2. Generate the sinusoidal signal with low frequency group f = 697 Hz, 770 Hz, 852 Hz, 941 Hz and high frequencies h= 1209 Hz, 1336 Hz, 1477Hz.

3Set the sampling frequency 8 KHz,N = 800 , No of samples at a spacing of 100ms.

4. Generate and visualize the DFT of the corresponding low and high frequency signal.
5. Stop the program.

Theory :

Dual-tone Multi-Frequency (DTMF) signaling is the basis for voice communications control and is widely used worldwide in modern telephony to dial numbers and configure switchboards. It is also used in systems such as in voice mail, electronic mail and telephone banking.

Generating DTMF Tones. A DTMF signal consists of the sum of two sinusoids - or tones - with frequencies taken from two mutually exclusive groups. These frequencies were chosen to prevent any harmonics from being incorrectly detected by the receiver as some other DTMF frequency. Each pair of tones contains one frequency of the low group (697 Hz, 770 Hz, 852 Hz, 941 Hz) and one frequency of the high group (1209 Hz, 1336 Hz, 1477Hz) and represents a unique symbol. The frequencies allocated to the push-buttons of the telephone pad are shown below:

Matlab Program

i. **Generation of the twelve frequency pairs**

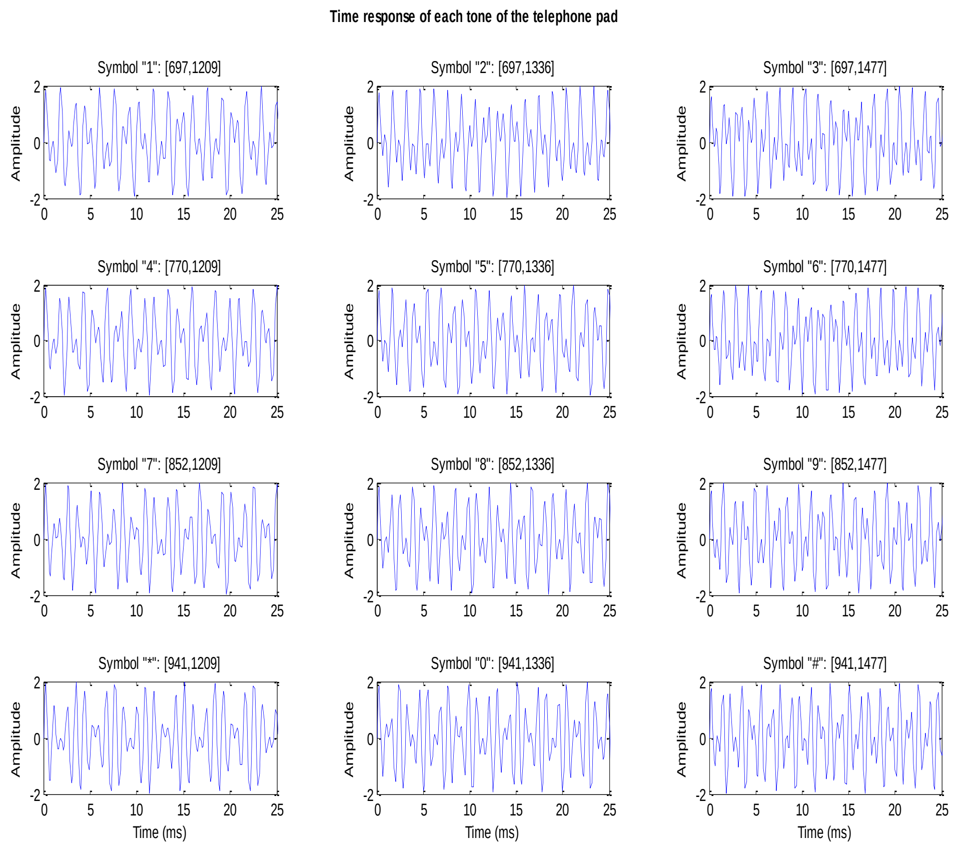
```
Clear all;
Close all;
symbol = {'1','2','3','4','5','6','7','8','9','*','0','#'};
lfg = [697 770 852 941]; % Low frequency group
hfg = [1209 1336 1477];  % High frequency group
f  = [];
for c=1:4,
    for r=1:3,
        f = [ f [lfg(c);hfg(r)] ];
    end
end

ans =

        697       1209
        697       1336
        697       1477
        770       1209
        770       1336
        770       1477
        852       1209
        852       1336
        852       1477
        941       1209
        941       1336
        941       1477
```

Matlab code for the generation and visualization of the DTMF tones

```
Fs  = 8000;       % Sampling frequency 8 kHz
N = 800;          % Tones of 100 ms
t   = (0:N-1)/Fs; % 800 samples at Fs
pit = 2*pi*t;

tones = zeros(N,size(f,2));
for toneChoice=1:12,
    % Generate tone
    tones(:,toneChoice) = sum(sin(f(:,toneChoice)*pit))';
    % Plot tone
    subplot(4,3,toneChoice),plot(t*1e3,tones(:,toneChoice));
title(['Symbol "', symbol{toneChoice},'":
[',num2str(f(1,toneChoice)),',',num2str(f(2,toneChoice)),']'])
```

```
set(gca, 'XLim', [0 25]);
ylabel('Amplitude');
   if toneChoice>9, xlabel('Time (ms)'); end
end
set(gcf, 'Color', [1 1 1], 'Position', [1 1 1280 1024])
annotation(gcf,'textbox', 'Position',[0.38 0.96 0.45 0.026],...
   'EdgeColor',[1 1 1],...
   'String', '\bf Time response of each tone of the telephone pad', ...
   'FitBoxToText','on');
```

MODEL GRAPH



PostLab Questions:

1. What are the applications of DSP systems?
2. List the procedure for generation of DTMF signals.
3. Define Goertzel algorithm for DFT estimation.
4. What are the types of multiple access in communication systems?
5. Define OFDM.

Result:
Thus a program to generate and visualize DTMF tones for telephone communication using Matlab was written and executed.