MARRI LAXMAN REDDY
INSTITUTE OF TECHNOLOGY & MANAGEMENT
(Approved by AICTE-New Delhi, Accredited by NAAC with 'A' & Affliated to JNTU, Hyderabad)
Recognised Under Section 2(f) & 12(B) of The UGC act, 1956
Dundigal, Quthbullapur (M), Hyderabad-500043.

Department of Electronics & Communication Engineering

LABORATORY MANUAL

DIGITAL SIGNAL PROCESSING

III B.TECH -II Semester (ECE)

A.Y 2018-2019

MARRI LAXMAN REDDY
INSTITUTE OF TECHNOLOGY & MANAGEMENT
(Approved by AICTE-New Delhi, Accredited by NAAC with 'A' & Affliated to JNTU, Hyderabad)
Recognised Under Section 2(f) & 12(B) of The UGC act, 1956
Dundigal, Quthbullapur (M), Hyderabad-500043.

Department of Electronics & Communication Engineering

VISION AND MISSION OF THE INSTITUTE

INSTITUTE VISION:

To establish as an ideal academic institutions in the service of the nation, the world and the humanity by graduating talented engineers to be ethically strong, globally competent by conducting high quality research, developing breakthrough technologies, and disseminating and preserving technical knowledge.

INSTITUTE MISSION:

To fulfill the promised vision through the following strategic characteristics and aspirations:

1. **An atmosphere** that facilitates personal commitment to the educational success of students in an environment that **values diversity and community.**
2. Prudent and accountable **resource management**;
3. Undergraduate programs that integrate **global awareness, communication skills and team building** across the curriculum;
4. **Leadership and service** to meet society's needs;
5. Education and research **partnerships** with colleges, universities, and industries to graduate education and training that prepares students for **interdisciplinary engineering** research and advanced problem solving;
6. Highly **successful alumni** who contribute to the profession in the global society.

MARRI LAXMAN REDDY
INSTITUTE OF TECHNOLOGY & MANAGEMENT
(Approved by AICTE-New Delhi, Accredited by NAAC with 'A' & Affliated to JNTU, Hyderabad)
Recognised Under Section 2(f) & 12(B) of The UGC act, 1956
Dundigal, Quthbullapur (M), Hyderabad-500043.

Department of Electronics & Communication Engineering

VISION AND MISSION OF THE DEPARTMENT

DEPARTMENT VISION:

Imparting quality technical education through research, innovation and team work for a lasting technology development in the area of Electronics and Communication Engineering.

MISSION:

To develop a strong centre of excellence for education and research with excellent infrastructure and well qualified faculties to instill in them a passion for knowledge.

To achieve the Mission the department will

1. Establish a unique learning environment to enable the students to face the challenges of the Electronics and Communication Engineering field.
2. Promote the establishment of centre of excellence in niche technology areas to nurture the spirit of innovation and creativity among faculty and students.
3. Provide ethical and value based education by promoting activities addressing the societal needs.
4. Enable students to develop skills to solve complex technological problems of current times and also provide a framework for promoting collaborative and multidisciplinary activities.

PROGRAMME EDUCATIONAL OBJECTIVES

1. **PEO 1:** have successful **careers in Industry.**
2. **PEO 2:** show excellence in **higher studies/ Research.**
3. **PEO 3:** Show good competency towards **Entrepreneurship.**

MARRI LAXMAN REDDY
INSTITUTE OF TECHNOLOGY & MANAGEMENT
(Approved by AICTE-New Delhi, Accredited by NAAC with 'A' & Affliated to JNTU, Hyderabad)
Recognised Under Section 2(f) & 12(B) of The UGC act, 1956
Dundigal, Quthbullapur (M), Hyderabad-500043.

Department of Electronics & Communication Engineering

PROGRAM OUTCOMES

a	An ability to apply knowledge of Science, Mathematics, Engineering & Computing fundamentals for the solutions of Complex Engineering problems
b	An ability to identify, formulates, research literature and analyze complex engineering problems using first principles of mathematics and engineering sciences.
c	An ability to design solutions to complex process or program to meet desired needs.
d	Ability to use research-based knowledge and research methods including design of experiments to provide valid conclusions.
e	An ability to use appropriate techniques, skills and tools necessary for computing practice.
f	Ability to apply reasoning informed by the contextual knowledge to assess social issues, consequences & responsibilities relevant to the professional engineering practice.
g	Ability to understand the impact of engineering solutions in a global, economic, environmental, and societal context with sustainability.
h	An understanding of professional, ethical, Social issues and responsibilities.
i	An ability to function as an individual, and as a member or leader in diverse teams and in multidisciplinary settings.
j	An ability to communicate effectively on complex engineering activities within the engineering community.
k	Ability to demonstrate and understanding of the engineering and management principles as a member and leader in a team.
l	Ability to engage in independent and lifelong learning in the context of technological change.

PROGRAM SPECIFIC OUTCOMES	
PSO1	Analyze and design analog & digital circuits or systems for a given specification and function.
PSO2	Implement functional blocks of hardware-software co-designs for signal processing and communication applications.

MARRI LAXMAN REDDY
INSTITUTE OF TECHNOLOGY & MANAGEMENT
(Approved by AICTE-New Delhi, Accredited by **NAAC** with '**A**' & Affliated to JNTU, Hyderabad)
Recognised Under Section 2(f) & 12(B) of The UGC act, 1956
Dundigal, Quthbullapur (M), Hyderabad-500043.

Department of Electronics & Communication Engineering

COURSE OUTCOMES FOR DIGITAL SIGNAL PROCESSING LAB

Digital Signal Processing Lab	1. Generate different types of Signals using MATLAB and DSP Processor. 2. Plot DFT, FFT, and phase and Magnitude spectrums of various signals. 3. Implement low pass and high pass FIR and IIR Filters. 4. Generate DTMF Signals

MARRI LAXMAN REDDY
INSTITUTE OF TECHNOLOGY & MANAGEMENT
(Approved by AICTE-New Delhi, Accredited by NAAC with 'A' & Affliated to JNTU, Hyderabad)
Recognised Under Section 2(f) & 12(B) of The UGC act, 1956
Dundigal, Quthbullapur (M), Hyderabad-500043.

Department of Electronics & Communication Engineering

DIGITAL SIGNAL PROCESSING LAB

List of Experiments:

1. Generation of sinusoidal waveform /signal based on recursive difference equations.
2. To find DFT/IDFT of given DT signal.
3. To find frequency response of a given system (Transfer function /Differential equation form).
4. Implementation of FFT of given sequence.
5. Determination of power spectrum of a given signal.
6. Implementation of LP FIR filter for a given sequence.
7. Implementation of HP FIR filter for a given sequence.
8. Implementation of LP IIR filter for a given sequence.
9. Implementation of HP IIR filter for a given sequence.
10. Generation of sinusoidal signal through filtering.
11. Generation of DTMF signals.
12. Implementation of Decimation process.
13. Implementation of Interpolation process.
14 Implementation of I/D sampling rate converters.
15. Removal of noise by autocorrelation / cross correlation.
16. Extraction of periodic signal masked by noise using correlation.
17. Impulse response of first order and second order systems.

INSTRUCTIONS TO THE STUDENT

1. Students are required to attend all labs.
2. Students will work individually in hardware laboratories and in computer laboratories.
3. While coming to the lab bring the lab manual cum observation book, record etc.
4. Take only the lab manual, calculator (if needed) and a pen or pencil to the work area.
5. Before coming to the lab, prepare the pre-lab questions. Read through the lab experiment to familiarize yourself with the components and assembly sequence.
6. Utilize 3 hours time properly to perform the experiment (both in software and hardware) and note down the readings properly. Do the calculations, draw the graph and take signature from the instructor.
7. If the experiment is not completed in the prescribed time, the pending work has to be done in the leisure hour or extended hours.
8. You have to submit the completed record book according to the deadlines set up by your instructor.
9. For practical subjects there shall be a continuous evaluation during the semester for 25 sessional marks and 50 end examination marks.
10. Of the 25 marks for internal, 15 marks shall be awarded for day-to-day work and 10 marks to be awarded by conducting an internal laboratory test.

INDEX

EXPERMENT NO-1

GENERATION OF BASIC SIGNALS USING MATLAB

AIM: -

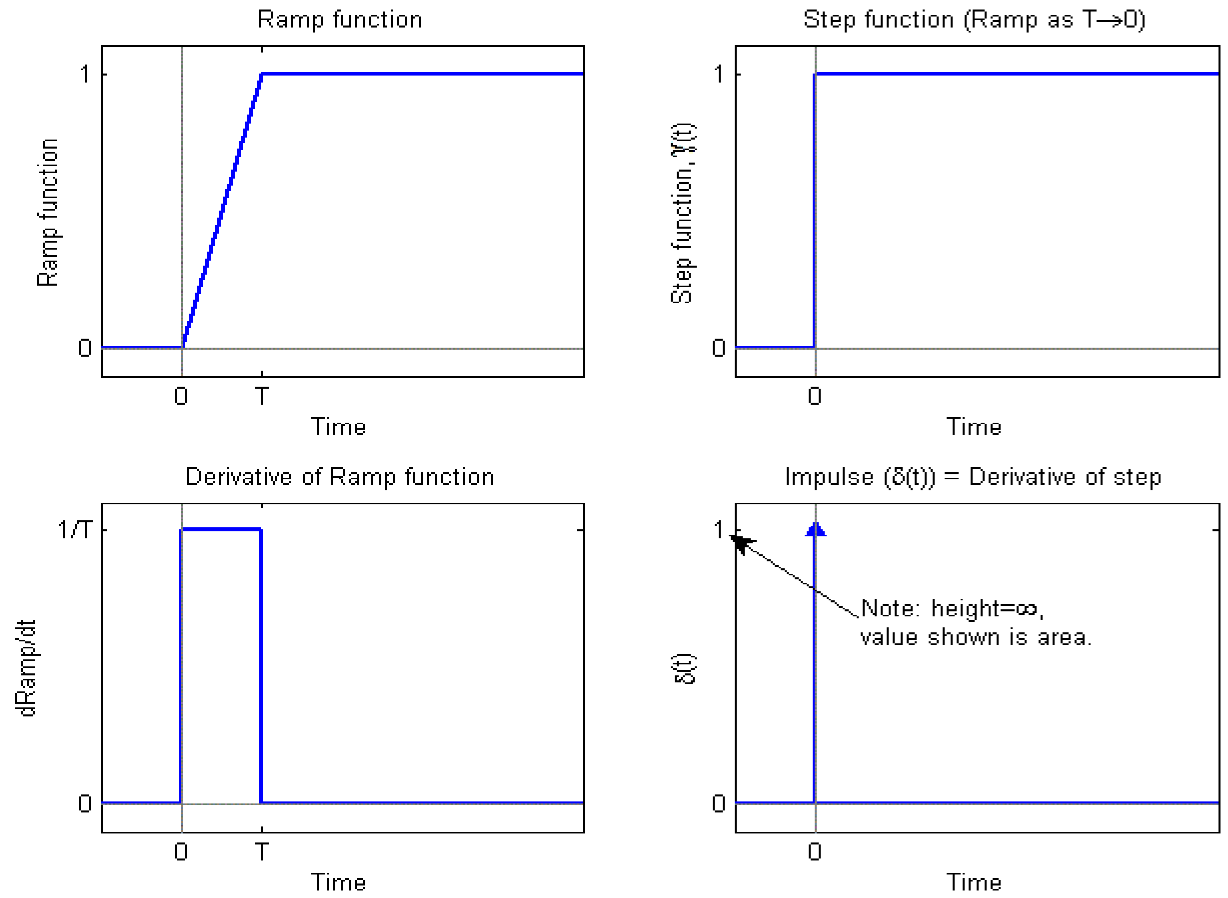
To write a "MATLAB" Program to generate various signals such as unit impulse, unit step, unit ramp, sinusoidal, exponential growing signal, exponential decaying signal, cosine signal.

SOFTWARE REQURIED:-

1. MATLAB R2010a.
2. Windows XP SP2.

THEORY:-

One of the more useful functions in the study of linear systems is the "unit impulse function." An ideal impulse function is a function that is zero everywhere but at the origin, where it is infinitely high. However, the *area* of the impulse is finite. This is, at first hard to visualize but we can do so by using the graphs shown below.



Key Concept: Sifting Property of the Impulse

If b>a, then

$$\int_a^b \delta(t-T)\cdot f(t)dt = \begin{cases} f(T), & a < T < b \\ 0, & \text{otherwise} \end{cases}$$

Example: Another integral problem

Assume a<b, and evaluate the integral

$$\int_a^b \delta(t) \cdot f(t-T)dt$$

Solution:

We now that the impulse is zero except at t=0 so

$$\delta(t) \cdot f(t-T) = \delta(t) \cdot f(0-T) = \delta(t) \cdot f(-T)$$

And

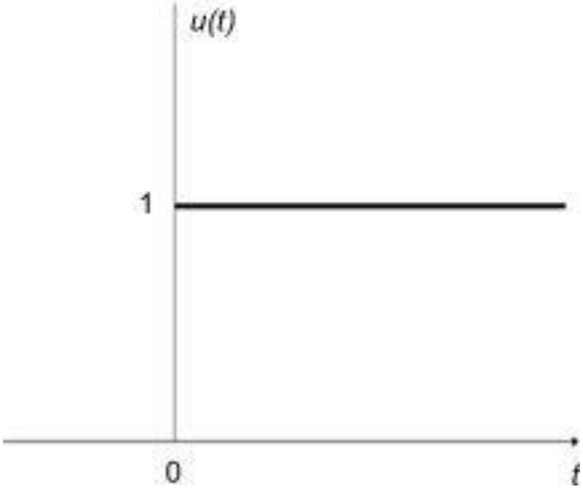
$$\begin{aligned} \int_a^b \delta(t) \cdot f(t-T)dt &= \int_a^b \delta(t) \cdot f(-T)dt \\ &= f(-T) \cdot \int_a^b \delta(t)dt \\ &= \begin{cases} f(-T), & a < 0 < b \\ 0, & \text{otherwise} \end{cases} \end{aligned}$$

Unit Step Function

The unit step function and the impulse function are considered to be fundamental functions in engineering, and it is strongly recommended that the reader becomes very familiar with both of these functions.

The unit step function, also known as the Heaviside function, is defined as such:

$$u(t) = \begin{cases} 0, & \text{if } t < 0 \\ 1, & \text{if } t > 0 \\ \frac{1}{2}, & \text{if } t = 0 \end{cases}$$



Sometimes, u(0) is given other values, usually either 0 or 1. For many applications, it is irrelevant what the value at zero is. u(0) is generally written as undefined.

Derivative

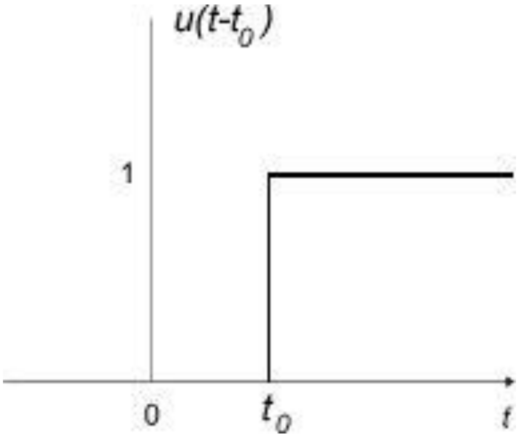
The unit step function is level in all places except for a discontinuity at t = 0. For this reason, the derivative of the unit step function is 0 at all points t, except where t = 0. Where t = 0, the derivative of the unit step function is infinite.

The derivative of a unit step function is called an **impulse function**. The impulse function will be described in more detail next.

Integral

The integral of a unit step function is computed as such:

$$\int_{-\infty}^{t} u(s)ds = \left\{ \begin{matrix} 0, & \text{if } t < 0 \\ \int_0^t ds = t, & \text{if } t \geq 0 \end{matrix} \right\} = tu(t)$$



Sinusoidal Signal Generation

The sine wave or sinusoid is a mathematical function that describes a smooth repetitive oscillation. It occurs often in pure mathematics, as well as physics, signal processing, electrical engineering and many other fields. Its most basic form as a function of time (t)

Where:
• A, the amplitude, is the peak deviation of the function from its center position.
• ω , the angular frequency, specifies how many oscillations occur in a unit time interval, in radians per second
• ϕ , the phase, specifies where in its cycle the oscillation begins at t = 0.
A sampled sinusoid may be written as:

$$x(n) = A\sin(2\pi \frac{f}{f_s} n + \vartheta)$$

PROCEDURE:-

- Open MATLAB
- Open new M-file
- Type the program
- Save in current directory
- Compile and Run the program
- For the output see command window\ Figure window

PROGRAM:-

%unit step signals%

```
clc;
clear all;
close all;
disp('unit step signals');
N=input('enter the no of samples');
x=ones(1,N);
stem(x);
xlabel('time');
ylabel('amplitude');
title('unit step sequence');
```

% sinusoidal signals%

```
clc;
clear all;
close all;
disp('sinusoidal signals');
N=input('enter the no of samples');
n=0:1:N;
x=sin(n);
stem(x);
xlabel('time');
ylabel('amplitude');
title('sinusoidal sequence');
```

% unit ramp signals%

```
clc;
clear all;
close all;
disp('unit ramp signals');
N=input('enter the no of samples');
n=0:1:N;
x=n;
stem(x);
xlabel('time');
ylabel('amplitude');
title('unit ramp sequence');
```

% unit impulse signal %

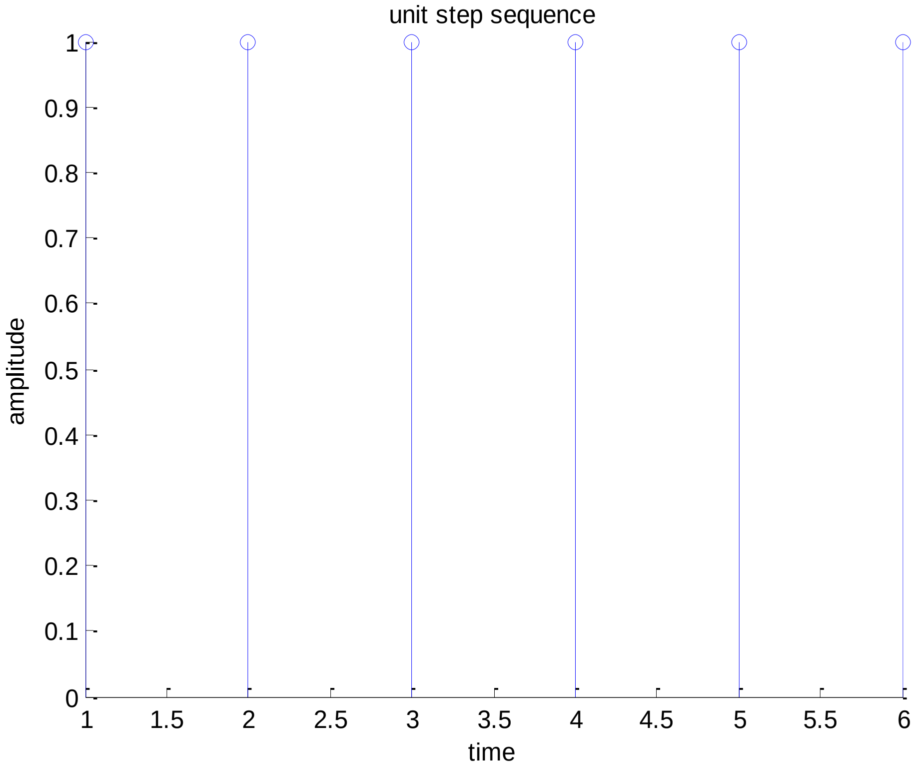
```
clc;
clear all;
close all;
disp('unit impuse signal');
N=input ('enter the no of samples');
n=-N:1:N;
x=[zeros(1,N) 1 zeros(1,N)];
stem(n,x);
xlabel('time');
ylabel('amplitude');
tittle('impulse sequence');
```

% exponentialsignals%

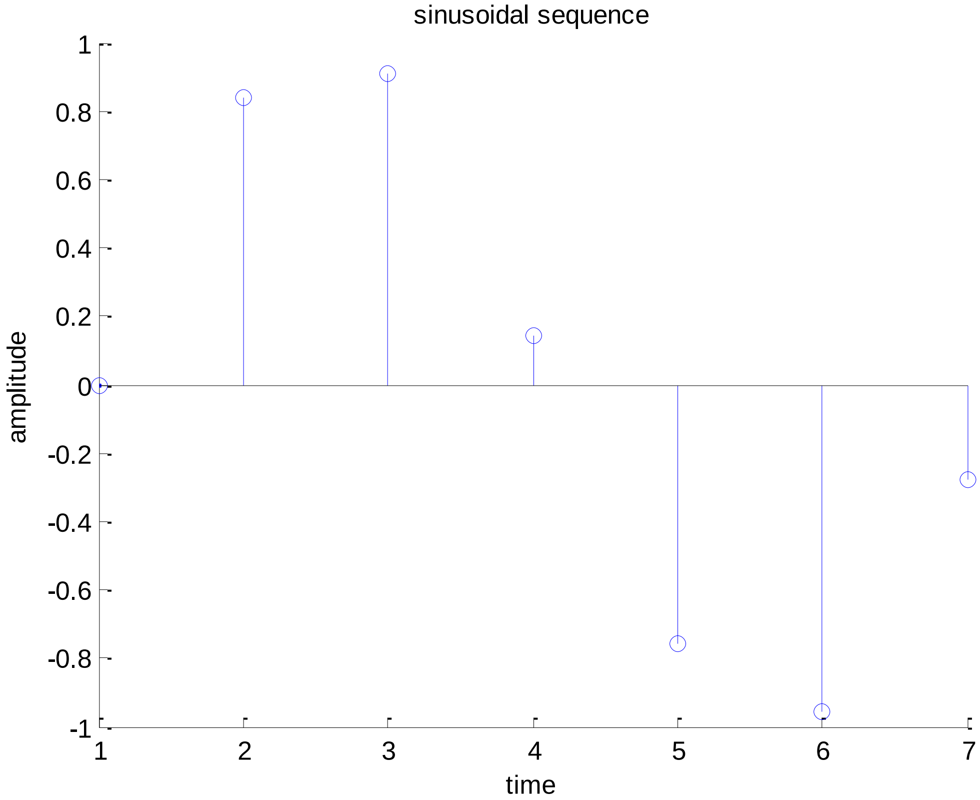
```
clc;
clear all;
close all;
disp('exponential signals');
N=input('enter the no of samples');
n=0:1:N;
n=-N:1:N;
a=0.5;
x=a.^n;
stem(x);
xlabel('time');
ylabel('amplitude');
title('exponential sequence');
```

OUTPUT:-

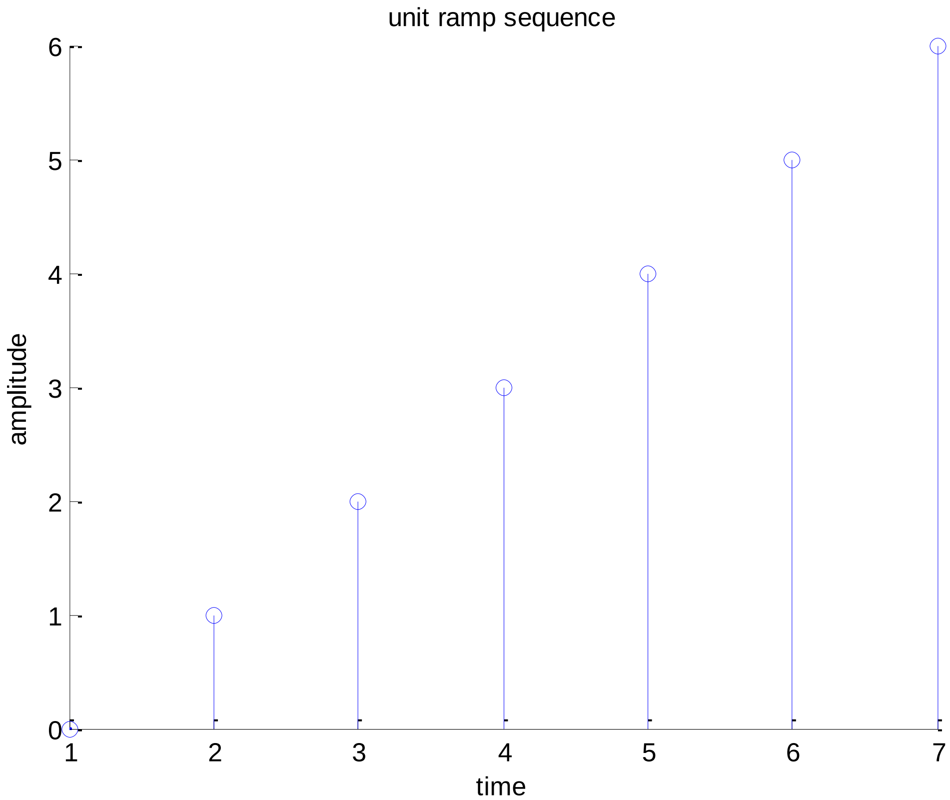
unit step signals
enter the no of samples6



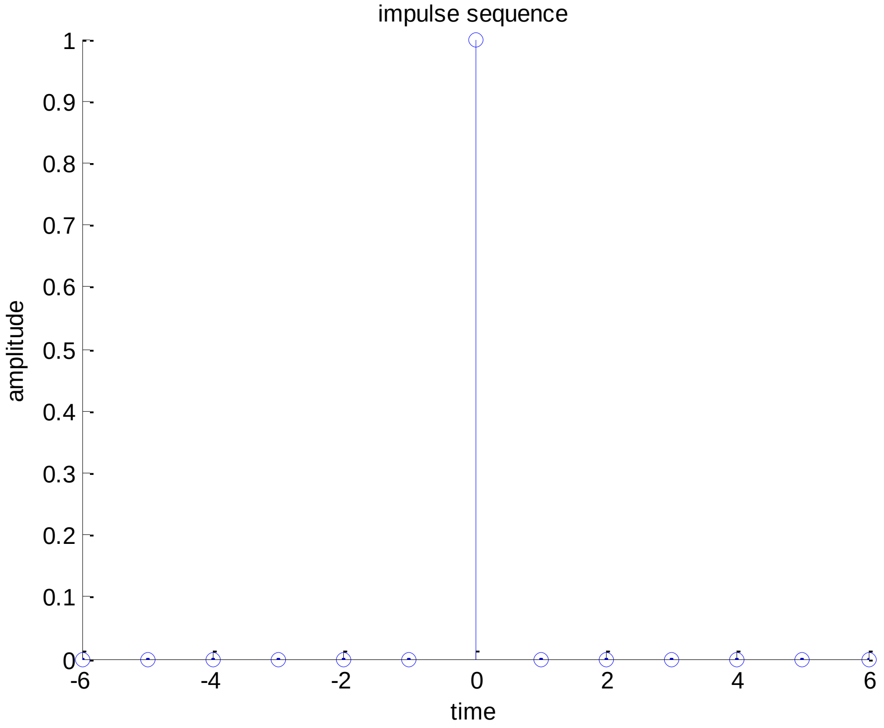
sinusoidal signals
enter the no of samples6



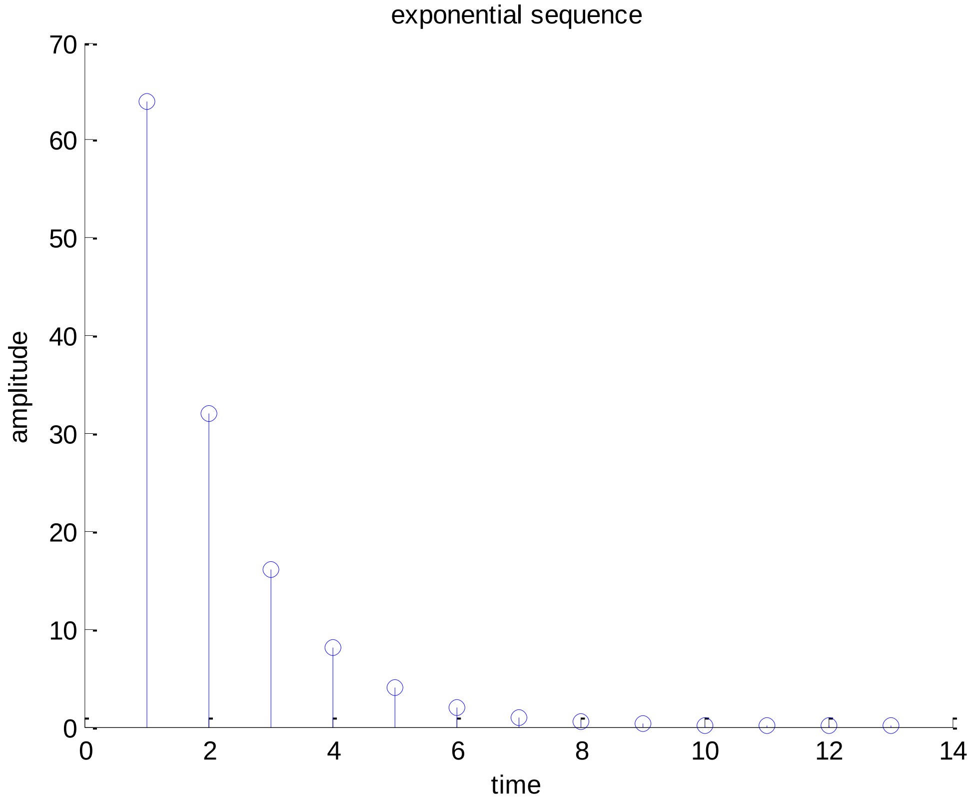
unit ramp signals
enter the no of samples6



unit impuse signal
enter the no of samples6



exponential signals
enter the no of samples6



RESULT:-

Thus the MATLAB program for generation of all signals was performed and the output was verified.

Real Life Applications

Trigonometry is especially important in architecture because it allows the architect to calculate distances and forces related to diagonal elements., for example on bridges and tall structures, the diagonal has to be strong and accurate to keep the structure standing. Architecture Space flight relies on calvulations and conversions to polat coordiates. because they help model orbital motions. Polar coordinates express a position on a two-dimensional plane using an angle from a fixed direction and a distance from a fixed point. Polar coordinates can be converted to Cartesian coordinates- the coordinate plane that we are used to seeing, and have been seeing since elementary. Polar coordinates can be converted to the Cartesian coordinates (x,y) by using sine and cosine functions. By multiplying the polar coordinates by cosine, the x coordinate can be obtained. By multiplying the polar coordinates by sine the y coordinate can be found. If a trumpet sounds at 440 Hz, at various amplitudes, the summation of sine waves or in other words Fourier series will be 440 Hz, 880 Hz, 1, 320 Hz, 1,760Hz. As we know sound travels in waves and frequencies.A French scientist and mathematician by the name of Jean Baptiste Fourier proved that any waveform that repeats itself after a period of time (such as a musical sound) can be expressed as the sum of an infinite set of sine curves. As we know sound travels in waves and frequencies. Ballisic trajectories rely on sin/cos.

EXERCISE PROGRAMS:-

1. Write program to get Discrete time Sinusoidal Signal?

2. Write program to get Fourier Transform of Sinusoidal Signal?

3. Write program to get Inverse Fourier Transform of Sinusoidal Signal?

4. Write Program for the following Function

   Y=exp (-2*∏*f*t)+exp(-8*∏*f*)

   Y= ((exp (-1.56∏f)*Sin(2∏f)+cos(2∏f)?
5. Write a mat lab program for generating u(n)-u(n-1)?
6. Write program to get Discrete time co-Sinusoidal Signal?
7. Write program to get Discrete time saw tooth Signal?
8. Write program to get Discrete time triangular Signal?
9. Write program to get addition of two sinusoidal sequences?
10. Write program to get exponential sequence?
11. Write program to get Discrete time Co-Sinusoidal Signal?
12. Write program to get Fourier Transform of Co-Sinusoidal Signal?
13. Write program to get Inverse Fourier Transform of Co-Sinusoidal Signal?
14. Write program to get exponential decaying sequence?
15. Write program to get exponential growing sequence?
16. Write program to get exponential decaying sequence?
17. Write program to get complex exponential sequence?
18. Write a mat lab program for generating sinc function?
19. Write a mat lab program for generating signum function?
20. Write a program to generate negative ramp signal?

VIVA QUESTIONS:-

1. Define Signal?
2. Define determistic and Random Signal?
3. Define Delta Function?
4. What is Signal Modeling?
5. Define Periodic and a periodic Signal?
6. Represent impulse signal in terms of unit step signal
7. Define Unit step signal
8. Define ramp signal
9. Define continuous time signal
10. Define discrete time signal.
11. Define impulse signal.
12. Define ramp signal
13. Define unit step signal
14. Define exponent ional signal
15. Define sinusoidal signal
16. Define C.T.S
17. Define D.T.S.
18. Compare C.T.S & D.T.S
19. Define Stem, Plot, Plot3,fplot, ezplot, linspace, flyplr, grid,mesh and legend.
20. Draw the C.T.S & D.T.S diagrams.
21. Define signal and signal processing
22. Differentiate digital and analog signals?
23. How the DSP processor will differ from conventional processors?
24. Expand the abbreviation TMS320C 5X/6X
25. What kind of processor is DSP processor?
26. What are the main building
27. blocks of DSP processor?
28. What is the main function of MAC unit?
29. Explain VLIW architecture?
30. What is meant by circular buffer?

EXPERMENT NO-2

DFT/IDFT OF A SEQUENCE WITHOUT USING THE INBUILT FUNCTIONS

AIM:-

To find the DFT& IDFT of a sequence without using the inbuilt functions.

SOFTWARE REQURIED:-

1. MATLAB R2010a.
2. Windows XP SP2.

THEORY:-

Given a sequence of N samples f(n), indexed by n = 0..N-1, the Discrete Fourier Transform (DFT) is defined as F(k), where k=0..N-1:

$$F(k) = \frac{1}{\sqrt{N}} \sum_{n=0}^{N-1} f(n)\, e^{-j2\pi kn/N}$$

F(k) are often called the 'Fourier Coefficients' or 'Harmonics'.

The sequence f(n) can be calculated from F(k) using the Inverse Discrete Fourier Transform (IDFT):

$$f(n) = \frac{1}{\sqrt{N}} \sum_{k=0}^{N-1} F(k)\, e^{+j2\pi nk/N}$$

In general, both f(n) and F(k) are complex.

The DFT is the most important discrete transform, used to perform Fourier analysis in many practical applications.[1] In digital signal processing, the function is any quantity or signal that varies over time, such as the pressure of a sound wave, a radio signal, or daily temperature readings, sampled over a finite time interval (often defined by a window function). In image processing, the samples can be the values of pixels along a row or column of a raster image. The DFT is also used to efficiently solve partial differential equations, and to perform other operations such as convolutions or multiplying large integers.

PROCEDURE:-

- Open MATLAB

- Open new M-file
- Type the program
- Save in current directory
- Compile and Run the program
- For the output see command window\ Figure window

PROGRAM:-

```
%DFT%
clc;
clear all;
close all;
a=input ('enter the input sequence');
N=length(a);
disp('length of input sequence is ');
N
for k=1:N;
x(k)=0;
for i=1:N;
x(k)=x(k)+a(i)*exp((-j*pi*2/N)*((i-1)*(k-1)));
end;
end;
k=1:N;
disp('the output is');
x(k)
subplot(2,1,1);
stem(k,abs(x(k)));
grid;
xlabel ('discrete frequency');
ylabel('magnitude');
title('magnitude response of dft');
subplot(2,1,2);
stem(angle(x(k))*180/(pi));
grid;
xlabel('discrete frequency');
ylabel('phase angle');
title('phase response of dft');
```

%IDFT%

```
clc;
clear all;
close all;
a=input('enter the input sequence');
disp('the length of input sequence is');
N=length(a);
```

```
N
for n=1:N;
x(n)=0;
for k=1:N;
x(n)=x(n)+a(k)*exp((j*pi*2*(n-1)*(k-1)/N));
end;
end;
n=1:N;
x=1/N*x(n);
disp('the output is');



x(n)
stem(n,abs(x));
grid;
xlabel('discrete time');
ylabel('magnitude');
title('magnitude response of the idft');
grid;
```

OUTPUT:-

enter the input sequence[1 2 3 4]
length of input sequence is

N =

4

the output is

ans =

10.0000 -2.0000 + 2.0000i -2.0000 - 0.0000i -2.0000 - 2.0000i

enter the input sequence[10 -2+2j -2 -2-2j]
the length of input sequence is

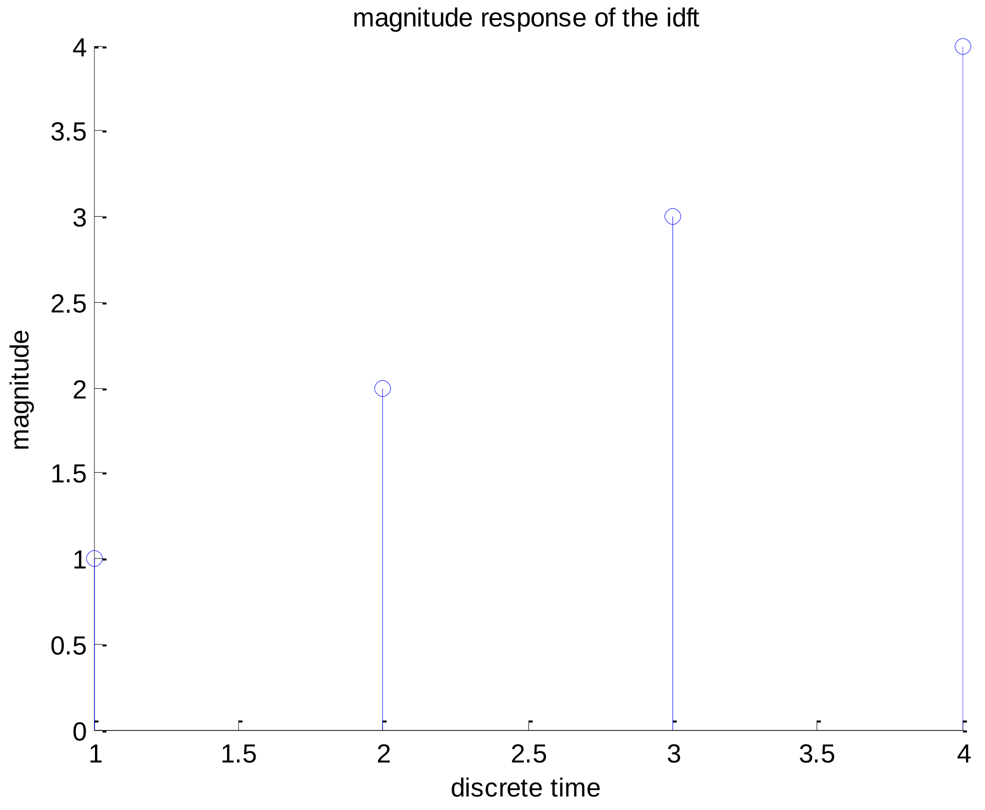
N =

4

the output is

ans =

1.0000 2.0000 + 0.0000i 3.0000 - 0.0000i 4.0000 - 0.0000i



RESULT:-

DFT&IDFT of a given discrete time signal are executed using mat lab software.

EXERCISE PROGRAM:-

1. Write a matlab program to find the circular convolution of two sequences?
2. Write a matlab program to find the circular convolution of x1(n)={2,3,-1,2};x2(n)={-1,2,-1,2}?
3. Write a matlab program to find the circular convolution of x1(n)={1,-1,2,3}; x2(n)={2,0,1,1}?
4. Write a matlab program to find the circular convolution of x1(n)={1,1,-1,2}; x2(n)={0,1,2,3}?
5. Write a matlab program to find the DFT of x(n) ={1 1 1 1 0 0 0 0}?
6. Write a matlab program to find the DFT of x(n) ={1 2 1 2}?
7. Write a matlab program to find the DFT of x(n) ={1 0 -1 0}?
8. Write a matlab program to find the IDFT of X(k) ={1,1,-2j,-1,1+2j }?
9. Write a matlab program to find the IDFT of X(k) ={1 0 1 0}?
10. Write a matlab program to find the to compare circular and linear convolution of two sequences?
11. Write a matlab program to find the DFT of x(n) ={1 0 1 0 1 0 1 0}?
12. Write a matlab program to find the DFT of x(n) ={1 1 0 0}?
13. Write a matlab program to find the IDFT of X(k) ={0 -1 0 -1}?
14. Write a matlab program to find the circular convolution of x1(n)={1,0,-1,0}; x2(n)={0,1,0,1}?
15. Write a matlab program to find the circular convolution of x1(n)={1,1,-1,-1}; x2(n)={1 2 3 4}?
16. Write a matlab program to find the DFT of x(n) ={1 0 1 0 0 0 0 0}?
17. Write a matlab program to find the DFT of x(n) ={1 1 1 2}?
18. Write a matlab program to find the DFT of x(n) ={1 0 -1 0}?
19. Write a matlab program to find the IDFT of X(k) ={1,1+2j,-1,1-2j }?
20. Write a matlab program to find the IDFT of X(k) ={0 0 1 1}?

VIVA QUESTIONS:-

1. Define Symmetric and Anti-Symmetric Signals?
2. Define Continuous and Discrete Time Signals?
3. What are the Different types of representation of discrete time signals?
4. What are the Different types of Operation performed on signals?
5. Define DFT
6. How DFT can be calculated in matrix form?
7. Write the standard formula of DFT
8. What is the relation between DFT and DFS
9. What is the relation between DFT and Z-Transform
10. What is the relation between DFT and Laplasetransform.
11. Whether DFT is a linear transform?
12. What is the difference between circular convolution & linear convolution?
13. Can you implement linear convolution using circular convolution?
14. How FFT algorithms are classified?
15. How to calculate output of DFT using MATLAB?
16. Where DFT is used?
17. What is the difference between DFT and IDFT?
18. How to compute maximum length N for a circular convolution using DFT and IDFT.
19. Explain the function of twiddle factor?
20. Give the practical application DFT & IDFT?
21. Explain the significance of convolution.
22. Define linear convolution.
23. Why linear convolution is called as a periodic convolution?
24. Why zero padding is used in linear convolution?
25. What are the four steps to find linear convolution?
26. What is the length of the resultant sequence in linear convolution?
27. How linear convolution will be used in calculation of LTI system response?
28. List few applications of linear convolution in LTI system design.
29. Give the properties of linear convolution.
30. How the linear convolution will be used to calculate the DFT of a signal

The Discrete Fourier Transform (DFT) is one of the most important tools in Digital Signal Processing. First, the DFT can calculate a signal's frequency spectrum. This is a direct examination of information encoded in the frequency, phase, and amplitude of the component sinusoids. For example, human speech and hearing use signals with this type of encoding. Second, the DFT can find a system's frequency response from the system's impulse response, and vice versa. This allows systems to be analyzed in the frequency domain, just as convolution allows systems to be analyzed in the time domain. Third, the DFT can be used as an intermediate step in more elaborate signal processing techniques. The classic example of this is FFT convolution, an algorithm for convolving signals that is hundreds of times faster than conventional methods.

EXPERMENT NO-3

STUDY FREQUENCY RESPONSE OF SECOND ORDER SYSTEM

AIM: -

To study frequency response of second order system using MATLAB.

SOFTWARE REQURIED:-

1. MATLAB R2010a.
2. Windows XP SP2.

THEORY:-

Second order systems are the systems or networks which contain two or more storage elements and have describing equations that are second order differential equations.

The frequency response of second order filters is characterised by three filter parameters: the gain k, the corner frequency and the quality factor Q.

A second order filter is a circuit that has a transfer function of the form:

$$H(s) = \frac{k \times \omega_o^2}{S^2 + \frac{\omega_o}{Q} S + \omega_o^2}$$

PROCEDURE:-

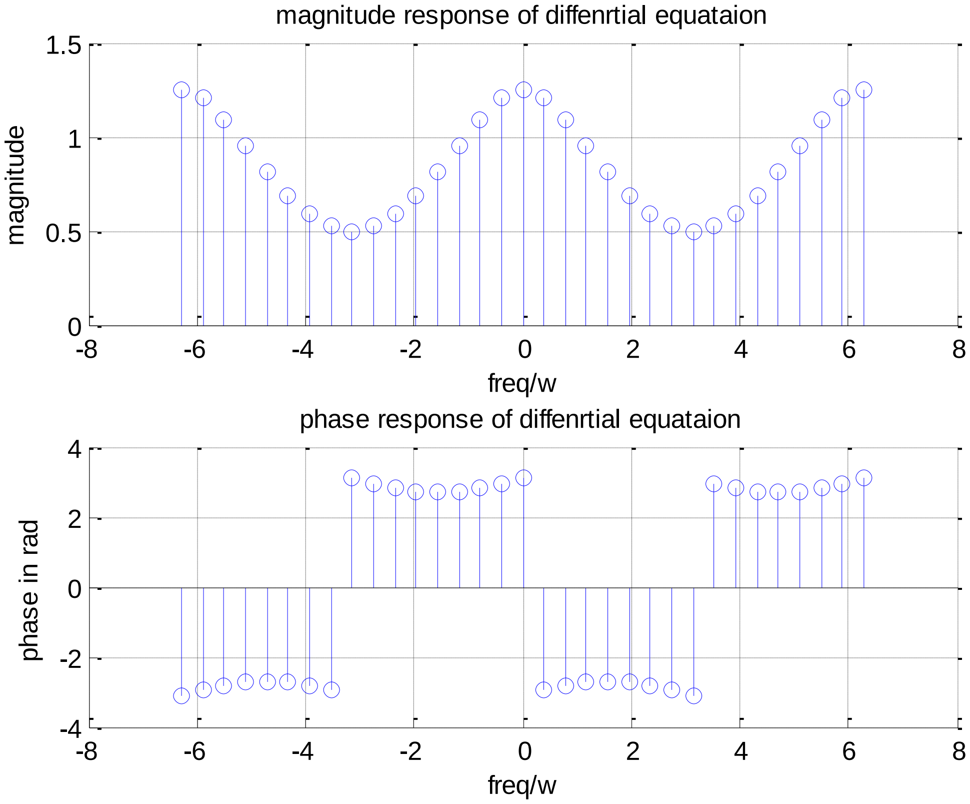
- Open MATLAB
- Open new M-file
- Type the program
- Save in current directory
- Compile and Run the program
- For the output see command window\ Figure window

PROGRAM:-

%frequency response of differential equation %

```
clc ;
clear all;
b=[1,4];
a=[1,-5];
w=-2*pi:pi/8:2*pi;
[h]=freqz(b,a,w);
subplot(2,1,1);
stem(w,abs(h));
xlabel('freq/w');
ylabel('magnitude');
grid;
title('magnitude response of diffenrtial equataion');
subplot(2,1,2);
stem(w,angle(h));
xlabel('freq/w');
ylabel('phase in rad');
grid;
title('phase response of diffenrtial equataion');
```

OUTPUT:-



RESULT:-

Hence the frequency response is executed by using MATLAB.

EXERCISE PROGRAM:-

1. Write a matlab program to find the frequency response of the following difference equation y(n)-7y(n-1)+9y(n-2)=3x(n)-2x(n-1)?
2. Write a matlab program to find the frequency response of the following difference equation 3y(n)+5y(n-1)=9x(n)?
3. Write a matlab program to find the frequency response of the following difference equation 9 y(n)-2y(n-1)+7y(n-2)-3y(n-3)=6x(n)+x(n-1)?
4. Write a matlab program to find the frequency response of the following difference equation 8y(n)+6y(n-1)=4x(n)+2x(n-1)?
5. Write a matlab program to find the frequency response of the following difference equation 3y(n)-8y(n-1)+9y(n-2)=9x(n)+5x(n-1) ?
6. Write a matlab program to find the frequency response of the following difference equation 6y(n)-5y(n-1)=9x(n)+5x(n-1) -7x(n-2)?
7. Write a matlab program to find the frequency response of the following difference equation 9y(n)-8y(n-1)+2y(n-2)=9x(n)-3x(n-1) ?
8. Write a matlab program to find the frequency response of the following difference equation 2y(n)-8y(n-1)=9x(n)+5x(n-1) ?
9. Write a matlab program to find the frequency response of the following difference equation 9y(n)-8y(n-1)+9y(n-2)=9x(n)+5x(n-1)-x(n-2) ?
10. Write a matlab program to find the frequency response of the following difference equation 3y(n)-8y(n-1)=7x(n)-3x(n-1) ?
11. Write a matlab program to find the frequency response of the following difference equation 11y(n-2)-5y(n-4)=6x(n)+5x(n-3) -9x(n-7)?
12. Write a matlab program to find the frequency response of the following difference equation 9y(n-1)-8y(n-5)+2y(n-7)=5x(n-1)-1x(n-3) ?
13. Write a matlab program to find the frequency response of the following difference equation y(n-3)-8y(n-7)=5x(n-1)+3x(n-5) ?
14. Write a matlab program to find the frequency response of the following difference equation 9y(n-2)-8y(n-4)+9y(n-6)=9x(n-1)+9x(n-3)-x(n-5) ?
15. Write a matlab program to find the frequency response of the following difference equation 3y(n+2)-8y(n+4)=7x(n-5)-3x(n-9) ?
16. Write a matlab program to find the frequency response of the following difference equation 11y(n-2)-5y(n-4)=6x(n-2)+5x(n-3) -9x(n-7)?
17. Write a matlab program to find the frequency response of the following difference equation 9y(n-1)-8y(n-4)+2y(n-7)=5x(n-1)-1x(n-3) ?
18. Write a matlab program to find the frequency response of the following difference equation y(n-2)-8y(n-7)=5x(n-1)+3x(n-5) ?
19. Write a matlab program to find the frequency response of the following difference equation 9y(n-2)-8y(n-4)+9y(n-6)=9x(n+1)+9x(n-3)-x(n-5) ?
20. Write a matlab program to find the frequency response of the following difference equation 3y(n+2)-8y(n+4)=7x(n+5)-3x(n-9) ?

VIVA QUESTIONS:-

1. What is the commend to find phase angle?
2. What is the commend to find frequency response?
3. What is transition band?
4. What is the formula for Z-transform?
5. What is the relationship b/w impulse response& frequency response?
6. How DFT can be calculated in matrix form?
7. What is the relation between DFT and Z-Transform.
8. Give any example of a FIR interpolator?
9. Define impulse response.
10. Give me one example for impulse response.
11. Write the Formula for impulse response.
12. What are major role in order & length?
13. Define power spectral Density?
14. What is the need for spectral estimation?
15. Determine the power spectrum density?
16. What is the relation between auto correlation & spectral density?
17. Give the estimation of auto correlation function & power density for random Signals?
18. Explain power spectrum estimation using the Bartlett window?
19. Give the formula for PSD.
20. What is filter?
21. Give mathematical definition of circular convolution.
22. Why circular convolution is called as periodic convolution?
23. Difference between linear convolution and circular convolution.
24. Explain the circular shift.
25. How circular convolution is used to calculate the Z
26. transform of a signal?
27. List few Applications of circular convolution
28. What are the different methods used to calculate circular convolution?
29. Explain properties of circular convolution?
30. Explain modulo N operation.

In electronics this stimulus would be an input signal. In the audible range it is usually referred to in connection with electronic amplifiers, microphones and loudspeakers. Radio spectrum frequency response can refer to measurements of coaxial cable, twisted-pair cable, video switching equipment, wireless communications devices, and antenna systems. Infrasonic frequency response measurements include earthquakes and electroencephalography (brain waves).Frequency response requirements differ depending on the application. In high fidelity audio, an amplifier requires a frequency response of at least 20–20,000 Hz, with a tolerance as tight as ±0.1 dB in the mid-range frequencies around 1000 Hz, however, in telephony, a frequency response of 400–4,000 Hz, with a tolerance of ±1 dB is sufficient for intelligibility of speech.

EXPERMENT NO-4

IMPLEMENTATION OF FFT OF GIVEN SEQUENCE

AIM: -

Implementation of FFT of given sequence.

SOFTWARE REQURIED:-

1. MATLAB R2010a.
2. Windows XP SP2.

THEORY:-

A fast Fourier transform (FFT) is an algorithm to compute the discrete Fourier transform (DFT) and its inverse. Fourier analysis converts time (or space) to frequency and vice versa; an FFT rapidly computes such transformations by factorizing the DFT matrix into a product of sparse (mostly zero) factors.

PROCEDURE:-

- Open MATLAB
- Open new M-file
- Type the program
- Save in current directory

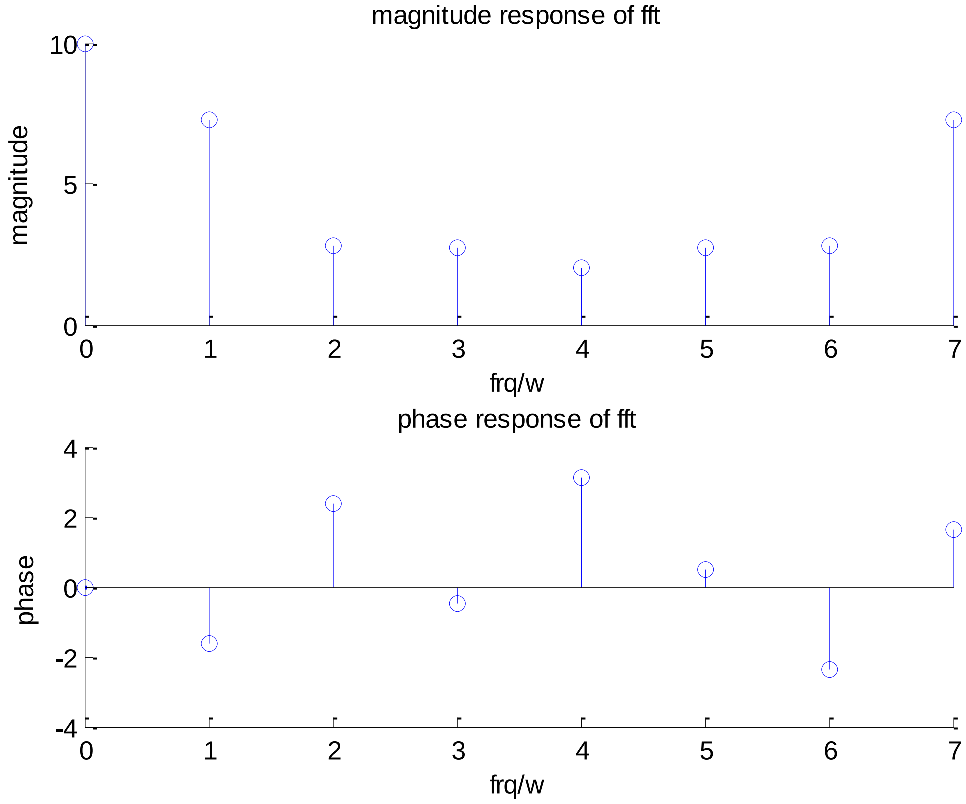
- Compile and Run the program
- For the output see command window\ Figure window

PROGRAM:-

```
%fft%
clc;
clear all;
close all;
xn=input('enter the input sequence');
N=input('enter the number of samples');
n=0:1:N-1;
xk=fft(xn,N);
k=0:1:N-1;
subplot(2,1,1);
stem(k,abs(xk));
xlabel('frq/w');
ylabel('magnitude');
title('magnitude response of fft');
subplot(2,1,2);
stem(k,angle(xk));
xlabel('frq/w');
ylabel('phase');
title('phase response of fft');
```

OUTPUT:-

```
enter the input sequence[1 2 3 4]
enter the number of samples8
```



RESULT:-

Hence the FFT of a given input sequence is performed & executed by using MATLAB.

EXERCISE PROGRAM:-

1. Write a matlab program to find the cross correlation using FFT?
2. Write a matlab program to find the DFT of x(n) ={1 1 0 0 0 0 0 0}?
3. Write a matlab program to find the DFT of x(n) ={1 1 1 1 1 0 0 0}?
4. Write a matlab program to find the DFT of x(n) ={1 0 1 0 1 0 1 0}?
5. Write a matlab program to find the IDFT of X(k) ={1,1+j,-2j,1+2j ,-j, +j}?

6. Write a matlab program to find the IDFT of X(k) ={1,0,-2j,-1,+2j,-7j }?
7. Write a matlab program to find the IDFT of X(k) ={1,1,-2j,-1,1+2j }?
8. Write a matlab program to find the IDFT of X(k) ={1,1+j,-2j,-1+-j,1+2j }?
9. Write a matlab program to find the DFT of x(n) ={1 1 0 0 1 1 0 0}?
10. Write a matlab program to find the DFT of x(n) ={1 0 0 0 1 0 0 0}?
11. Write a matlab program to find the IDFT of X(k) ={1+j,0,1-2j,-1,1+2j,1-7j }?
12. Write a matlab program to find the IDFT of X(k) ={1,2+5j,2-2j,-1,5+2j }?
13. Write a matlab program to find the IDFT of X(k) ={5+6j,1+6j,5-2j,-1+9j,1-2j }?
14. Write a matlab program to find the DFT of x(n) ={1 1 1 1 0 0 0 0}?
15. Write a matlab program to find the DFT of x(n) ={1 1 0 0 1 1 0 0}?
16. Write a matlab program to find the IDFT of X(k) ={2+j,0,7+2j,-1-5j,9+2j,6-7j }?
17. Write a matlab program to find the IDFT of X(k) ={1,2+5j,2-2j,-1,5+2j }?
18. Write a matlab program to find the IDFT of X(k) ={6j,1+6j,5-2j,-1-9j,1-2j }?
19. Write a matlab program to find the DFT of x(n) ={1 0 1 0 1 0 0 0 0 }?
20. Write a matlab program to find the DFT of x(n) ={1 0 0 1 0 0 0 0}?

VIVA QUESTIONS:-

1. Whether linear convolution equation is a difference equation?
2. Whether DFT is a linear transform?
3. What is the difference between circular convolution & linear convolution?
4. Can you implement linear convolution using circular convolution?
5. How FFT algorithms are classified?
6. How to calculate output of DFT using MATLAB?
7. Where DFT is used?
8. What is the difference between DFT and IDFT?
9. How to compute maximum length N for a circular convolution using DFT and IDFT.
10. Explain the function of twiddle factor?
11. Give the practical application DFT & IDFT?
12. Explain the role of DFT & IDFT when the signal converted from the time domain to frequency domain?
13. Differentiate between time variant and time invariant system. If x 1(n)={1,2,3,4} and x 2(n)={1,2,3} Find the convolution using tabular representation.
14. Draw all elementary standard discrete time signals.
15. Differentiate between causal and Non causal system.
16. If x 1(n)={1,2,3,4} and x 2(n)={5,6,7,8} Find the circular representation for the above sequences.
17. How can you compute Fourier transform form Z-transform ?
18. FFT is in complex domain how to use it in real life signals optimally?
19. What is the difference between FFT and IFFT?
20. Explain using convolution the effects of taking an FFT of a sample with no windowing .
21. What is a filter?
22. Differentiate analog filter and digital filter
23. Define FIR filter
24. What are the differences between recursive and non recursive systems?
25. List a few Applications of FIR filters.
26. Explain advantages of FIR filters over IIR filters.
27. Explain limitations of FIR filters.
28. What is the different method to design FIR filters?
29. Explain different window functions.
30. Differentiate rectangular, triangular and Kaiser windows.

The Discrete Fourier Transform (DFT) is one of the most important tools in Digital Signal Processing. First, the DFT can calculate a signal's frequency spectrum. This is a direct examination of information encoded in the frequency, phase, and amplitude of the component sinusoids. For example, human speech and hearing use signals with this type of encoding. Second, the DFT can find a system's frequency response from the system's impulse response, and vice versa. This allows systems to be analyzed in the frequency domain, just as convolution allows systems to be analyzed in the time domain. Third, the DFT can be used as an intermediate step in more elaborate signal processing techniques. The classic example of this is FFT convolution, an algorithm for convolving signals that is hundreds of times faster than conventional methods.

EXPERMENT NO-5

DETERMINATION OF POWER SPECTRUM

AIM: -

To obtain power spectrum of given signal using MATLAB.

SOFTWARE REQURIED:-

1. MATLAB R2010a.
2. Windows XP SP2.

THEORY:-

The power spectrum of a time-series x(t) describes how the variance of the data x(t) is distributed over the frequency components into which x(t) may be decomposed. This distribution of the variance may be described either by a measure μ or by a statistical cumulative distribution function S(f) = the power contributed by frequencies from 0 up to f.

PROCEDURE:-

- Open MATLAB
- Open new M-file
- Type the program
- Save in current directory
- Compile and Run the program
- For the output see command window\ Figure window

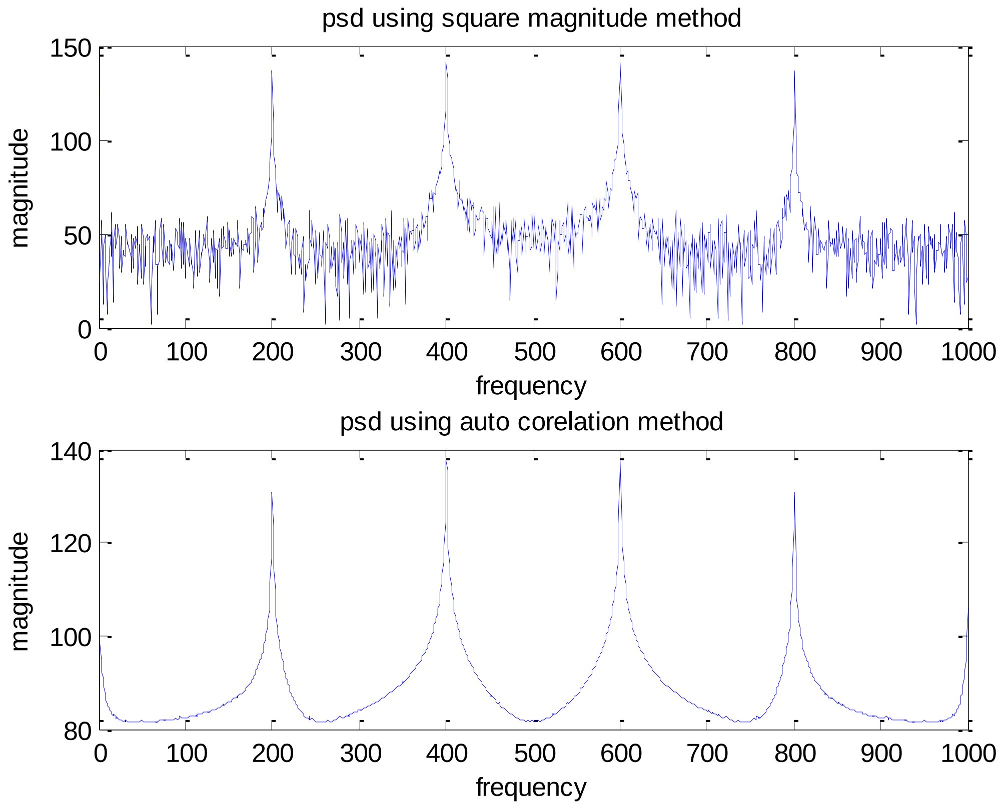
PROGRAM:-

```
% power spectrum %
clc;
clear all;
close all;
f1=input('enter the first frequencey f1=');
f2=input('enter the second frequencey f2=');
fs=input('enter the sampling frequencey fs=');
t=0:1/fs:1;
x=2*sin(2*pi*f1*t)+3*sin(2*pi*f2*t)+rand(size(t));
psd1=abs(fft(x).^2);
subplot(2,1,1);
plot(t*fs,10*log(psd1));
xlabel('frequency');
ylabel('magnitude');
title('psd using square magnitude method');
psd2=abs(fft(xcorr(x),length(t)));
subplot(2,1,2);
plot(t*fs,10*log(psd2));
xlabel('frequency');
```

```
ylabel('magnitude');
title('psd using auto corelation method');
```

OUTPUT:-

enter the first frequencey f1=200
enter the second frequencey f2=400
enter the sampling frequencey fs=1000



RESULT:-

Hence the power spectral density is performed & executed by using MATLAB.

EXERCISE PROGRAM:-

1. Write a matlab program for power spectrum estimate using Welch method?
2. Write a matlab program to plot the frequency response of a first order system?
3. Write a matlab program to plot the frequency response of the system?
4. Write a matlab program to generate the periodic sequence?
5. Write a matlab program to generate the aperiodic sequence?
6. Write a matlab program to demonstrate the property of digital frequency?
7. Write a matlab program to illustrate the concept of aliasing?
8. Write a matlab program to plot magnitude and phase response of first order lowpass filter?
9. Write a matlab program to plot magnitude and phase response of first order highpass filter?
10. Write a matlab program to plot magnitude and phase response of second order bandpass filter?
11. Write a matlab program to plot magnitude and phase response of second order bandstop filter?
12. Write a matlab program to plot magnitude and phase response of second order lowpass filter?
13. Write a matlab program to plot magnitude and phase response of second order highpass filter?
14. Write a matlab program to plot magnitude and phase response of first order bandpass filter?
15. Write a matlab program to plot magnitude and phase response of first order bandstop filter?
16. Write a matlab program to plot the frequency response of a first order system?
17. Write a matlab program to plot the frequency response of the system?
18. Write a matlab program to generate the periodic sinusoidal sequence?
19. Write a matlab program to generate the a periodic sinusoidal sequence?
20. Write a matlab program to demonstrate the property of digital signal?

VIVA QUESTIONS:-

1. Give the expressions for finding the Average power of a signal/sequence?
2. Give the expressions for finding the energy of a signal/sequence?
3. What is power spectrum?
4. Why there are two peaks in the magnitude spectrum of sine wave?
5. What is spectrogram? Which built in function is used to solve a given difference equation?
6. What is frequency response? Give equation for first order system andsecond order system?
7. .What is an LTI system?
8. What is steady state response?
9. Suppose we have a system with transfer function H(z) = 1 / ((z –1)*(z –0.9)). Is the system stable or unstable?
10. What is Auto Regressive Model? How is the order of auto regressive model is decided?
11. Differentiate between linear and circular convolution.
12. Determine the unit step response of the linear time invariant system with impulse response h(n)=a nu(n) a<1&-a<1
13. Determine the range of values of the parameter a for which linear time invariant system with impulse response h(n)=a u(n) is stable.
14. Consider the special case of a finite duration sequence given as X(n)={2 4 0 3}, resolve the sequence x(n) into a sum of weighted sequences.
15. Describe impulse response of a function?
16. Where to use command filter or impz, and what is the difference between these two?
17. How to calculate output of DFT using MATLAB?
18. Where DFT is used?
19. What is the difference between DFT and IDFT?
20. Explain the function of twiddle factor? List some a dvantages of digital filters over analog filters.
21. Write some differences between FIR and IIR filters
22. What are the different methods to design IIR filters.
23. Why IIR filters are not reliable What are different applications of IIR filters
24. What are advantages of IIR filters.
25. What are disadvantages of IIR filters.
26. Differentiate Butterworth and Chebyshev approximations.
27. What is meant by impulse response.
28. What is the importance of impulse response to calculate the o/p response of the filter?
29. Describe impulse response of a function?
30. Where to use command filter or impz, and what is the difference between these two?

It and the cumulative spectral density are very useful for identifying periodic components in time series. For instance, say you want to study weather cycles or even something like daily sales figures data for cycles. Cycles will show up pretty nicely as spikes in the PSD or CSD. You can also use the PSD or CSD to help diagnose whether your model fits well. If the spikes at identified frequencies are gone that's a sign that you have identified the periodic components.

EXPERMENT NO-6

IMPLEMENTATION OF LP FIR FILTERS

AIM: -

Implementation of Low Pass FIR filters for given sequence.

SOFTWARE REQURIED:-

1. MATLAB R2010a.
2.Windows XP SP2.

THEORY:-

A Finite Impulse Response (FIR) filter is a discrete linear time-invariant system whose output is based on the weighted summation of a finite number of past inputs. An FIR transversal filter structure can be obtained directly from the equation for discrete-time convolution.

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

In this equation, x(k) and y(n) represent the input to and output from the filter at time n. h(n-k) is the transversal filter coefficients at time n. These coefficients are generated by using FDS (Filter Design Software or Digital filter design package).

FIR – filter is a finite impulse response filter. Order of the filter should be specified. Infinite response is truncated to get finite impulse response. Placing a window of finite length does this. Types of windows available are Rectangular, Barlett, Hamming, Hanning, Blackmann window etc. This FIR filter is an all zero filter.

PROCEDURE:-

- Open MATLAB
- Open new M-file
- Type the program
- Save in current directory
- Compile and Run the program
- For the output see command window\ Figure window

PROGRAM:-

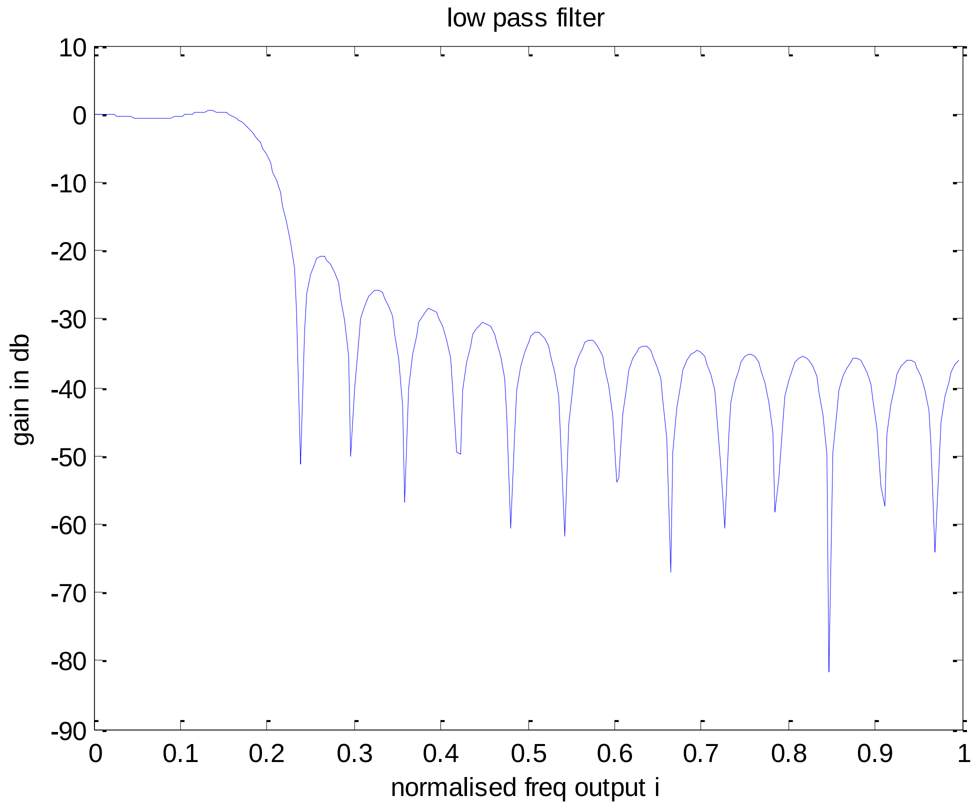
% LOW PASS FILTER %

```
clc;
clear all;
close all;
rp=input('enter the pass band ripple:rp=');
```

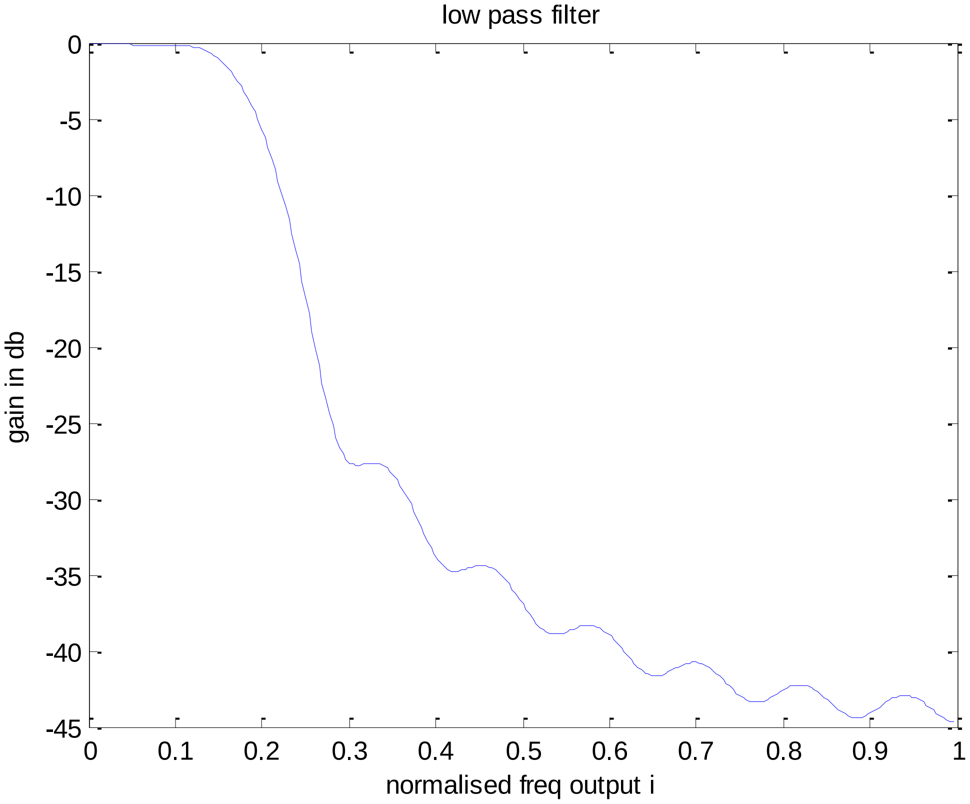
```
rs=input('enter the stop band ripple:rs=');
fp=input('enter the pass band freq :fp=');
fs=input('enter the stop band freq:fp=');
f=input('enter the sampling freq:f=');
wp=2*fp/f;
ws=2*fs/f;
num=-20*log10(sqrt(rp*rs))-13;
den=14.6*(fs-fp)/f;
n=ceil(num/den);
n1=n+1;
if(rem(n,2)~=0);
n1=n;
n=n-1;
end;
c=input('enter the type of window function 1.rectangular 2.trangular 3.kaiser:n=');
if(c==1);
y=rectwin(n1);
disp('rectangular window filter response');
end;
if(c==2);
y=triang(n1);
disp('triangular window filter response');
end;
if(c==3);
y=kaiser(n1);
disp('kaiser window filter response');
end;
% low pass filter %
b=fir1(n,wp,y);
[n,o]=freqz(b,1,256);
m=20*log10(abs(n));
plot(o/pi,m);
xlabel('normalised freq output i');
ylabel('gain in db');
title('low pass filter');
```

OUTPUT:-

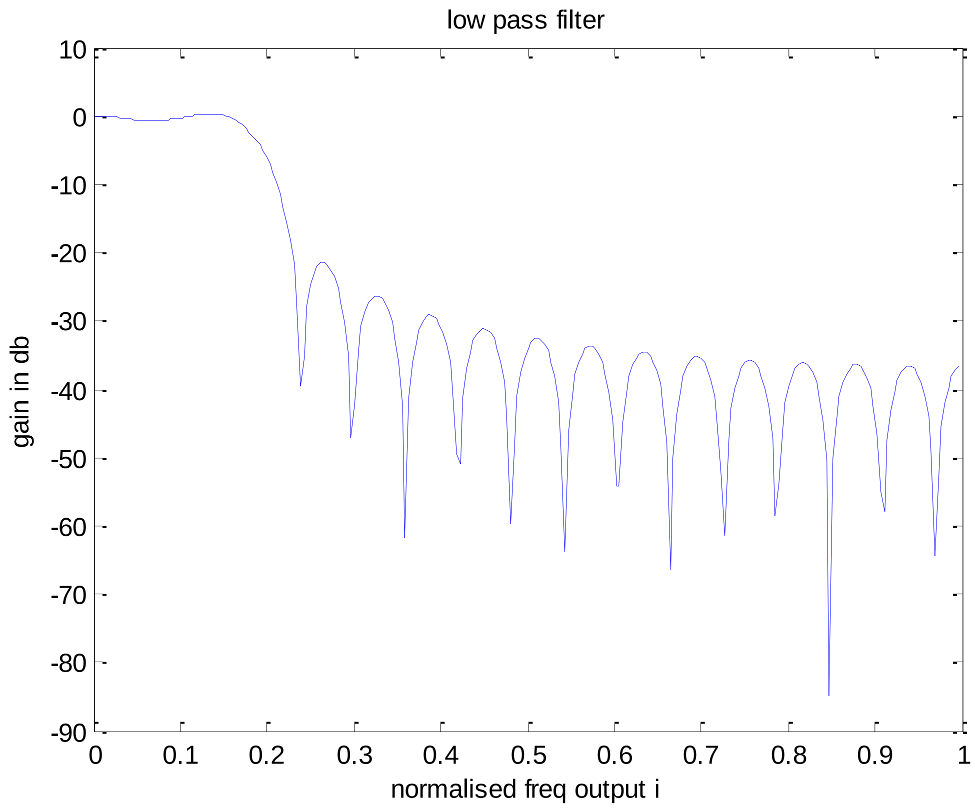
```
 enter the pass band ripple:rp=0.02
enter the stop band ripple:rs=0.01
enter the pass band freq :fp=1000
enter the stop band freq:fp=1500
enter the sampling freq:f=10000
enter the type of window function 1.rectangular 2.trangular 3.kaiser:n=1
rectangular window filter response
```



enter the pass band ripple:rp=0.02
enter the stop band ripple:rs=0.01
enter the pass band freq :fp=1000
enter the stop band freq:fp=1500
enter the sampling freq:f=10000
enter the type of window function 1.rectangular 2.trangular 3.kaiser:n=2
triangular window filter response



enter the pass band ripple:rp=0.02
enter the stop band ripple:rs=0.01
enter the pass band freq :fp=1000
enter the stop band freq:fp=1500
enter the sampling freq:f=10000
enter the type of window function 1.rectangular 2.trangular 3.kaiser:n=3
kaiser window filter response



RESULT:-

The implementation of Butterworth Low passes FIR Filters Completed.

EXERCISE PROGRAM:-

1. Write a matlab program to design FIR low pass filter using hamming/hanning window?
2. Write a matlab program to design FIR low pass filter using hamming& blackman window?
3. Write a matlab program to design FIR low pass filter with cutoff frequency .5pi using frequency sampling method?
4. Write a matlab program to plot the frequency response of low pass filter using Kaiser window for different values of beeta?
5. Write a matlab program to design FIR low pass filter for the given specifications using Kaiser window?
6. Write a matlab program to design a 25-tap Hilbert transformer using Bartlett and hamming windows and plot their frequency response?
7. Write a matlab program to design a 25-tap lowpass filter with cutoff frequency .5pi radians using rectangular and hamming windows and plot their frequency response?
8. Write a matlab program to design a 25-tap highpass filter with cutoff frequency .5pi radians using rectangular and Blackman windows and plot their frequency response?
9. Write a matlab program to design a 25-tap bandpass filter with cutoff frequency .25pi & .75pi radians using rectangular and hamming windows and plot their frequency response?
10. Write a matlab program to design a 25-tap bandstop filter with cutoff frequency .25pi & .75pi radians using rectangular and hamming windows and plot their frequency response?
11. Write a matlab program to design a 25-tap Hilbert transformer using Bartlett window and plot the frequency response?
12. Write a matlab program to design a 25-tap Hilbert transformer using hamming window and plot the frequency response?
13. Write a matlab program to design FIR low pass filter using hamming window?
14. Write a matlab program to design FIR low pass filter using blackman window?
15. Write a matlab program to design FIR high pass filter using blackman window?
16. Write a matlab program to design FIR high pass filter with cutoff frequency .5pi using frequency sampling method?
17. Write a matlab program to plot the frequency response of high pass filter using Kaiser window for different values of beeta?
18. Write a matlab program to design FIR high pass filter for the given specifications using Kaiser window?
19. Write a matlab program to design a 25-tap Hilbert transformer using Bartlett and hamming windows and plot their frequency response?
20. Write a matlab program to design a 25-tap high pass filter with cutoff frequency .5pi radians using rectangular and hamming windows and plot their frequency response?

VIVA QUESTIONS:-

1. Give the expression for finding the magnitude & phase response of FIR filter?
2. Compare different windows & their characteristics?
3. How FIR filters are designed using rectangular window?
4. What is Gibbs phenomenon? How it can be reduced?
5. Compare FIR&IIR filters?
6. What is "decimation-in-time" versus "decimation-in-frequency"?
7. What is "bit reversal"?
8. What do you mean by phase spectrum and magnitude spectrum/ give comparison?
9. How do you reduce spectral leakage?
10. What do you mean by spectral resolution?
11. What is FIR and IIR filter define, and distinguish between these two?
12. What is window method? How you will design an FIR filter using window method?
13. What are low-pass and band-pass filter and what is the difference between these two?
14. What is the matlab command for Hamming window? Explain.
15. What do you mean by built in function ‘abs’ and where it is used?
16. Explain how the FIR filter are stable?
17. Why is the impulse response "finite"?
18. What does "FIR" mean?
19. What are the advantages of FIR Filters (compared to IIR filters)?
20. What are the disadvantages of FIR Filters (compared to IIR filters)?
21. Define Gibb’s phenomena
22. What is meant by ringing effect?
23. Why do we need to represent a signal in frequency domain?
24. Why Fourier Series converges only for periodic signals?
25. How the ringing effect can be rectified?
26. What are the different forms of Fourier series?
27. What are the limitations of Fourier series?
28. Write few applications of Fourier series.
29. Explain the MATLAB functions ‘tic’ and ‘toc’
30. Explain the MATLAB function ‘legend’

The calculation of the digital filter function (DSP) is done by the CPU. For that a FIR algorithm of Infineon's TriLib is used. The coefficients for the filter can be calculated with an appropriate software program. For this application note the program "ScopeFIR" was used. Finally the GPTA of the TC1775 generates a PWM signal with the digital filtered.

EXPERMENT NO-7

IMPLEMENTATION OF HP FIR FILTERS

AIM: -

Implementation of High Pass FIR filter for given sequence.

SOFTWARE REQURIED:-

1. MATLAB R2010a.
2. Windows XP SP2.

THEORY:-

A Finite Impulse Response (FIR) filter is a discrete linear time-invariant system whose output is based on the weighted summation of a finite number of past inputs. An FIR transversal filter structure can be obtained directly from the equation for discrete-time convolution.

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

In this equation, x(k) and y(n) represent the input to and output from the filter at time n. h(n-k) is the transversal filter coefficients at time n. These coefficients are generated by using FDS (Filter Design Software or Digital filter design package).

FIR – filter is a finite impulse response filter. Order of the filter should be specified. Infinite response is truncated to get finite impulse response. Placing a window of finite length does this. Types of windows available are Rectangular, Barlett, Hamming, Hanning, Blackmann window etc. This FIR filter is an all zero filter.

PROCEDURE:-

- Open MATLAB
- Open new M-file
- Type the program
- Save in current directory
- Compile and Run the program
- For the output see command window\ Figure window

PROGRAM:-

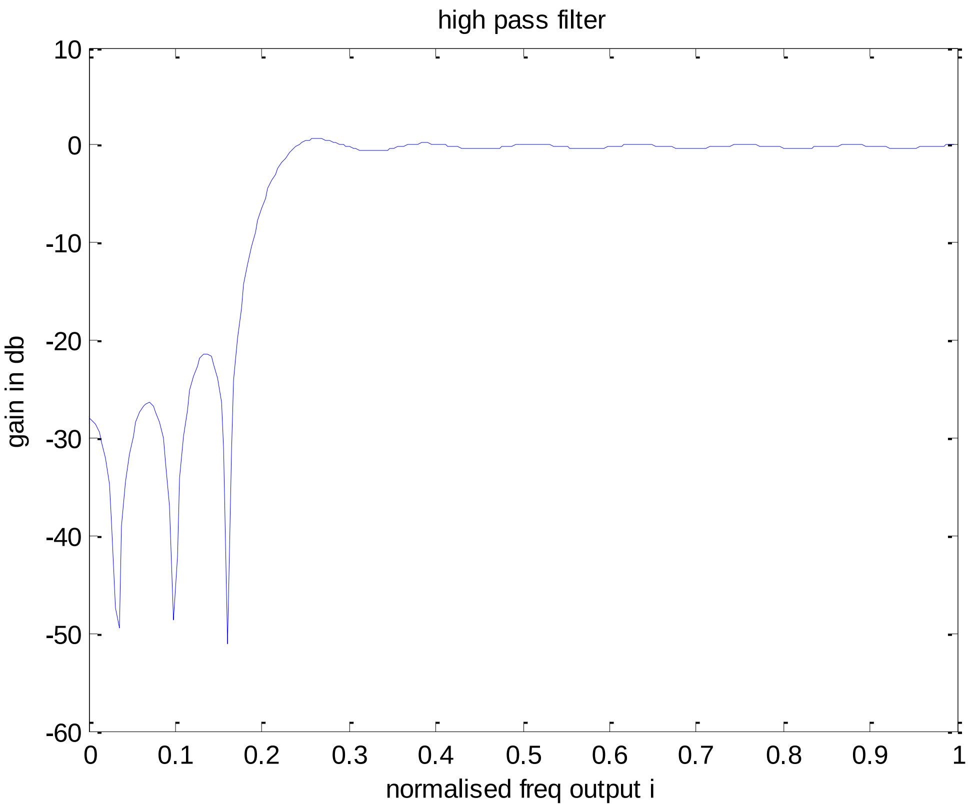
% high pass filter %

```
clc;
clear all;
close all;
rp=input('enter the pass band ripple:rp=');
rs=input('enter the stop band ripple:rs=');
fp=input('enter the pass band freq :fp=');
fs=input('enter the stop band freq:fp=');
f=input('enter the sampling freq:f=');
wp=2*fp/f;
ws=2*fs/f;
num=-20*log10(sqrt(rp*rs))-13;
```

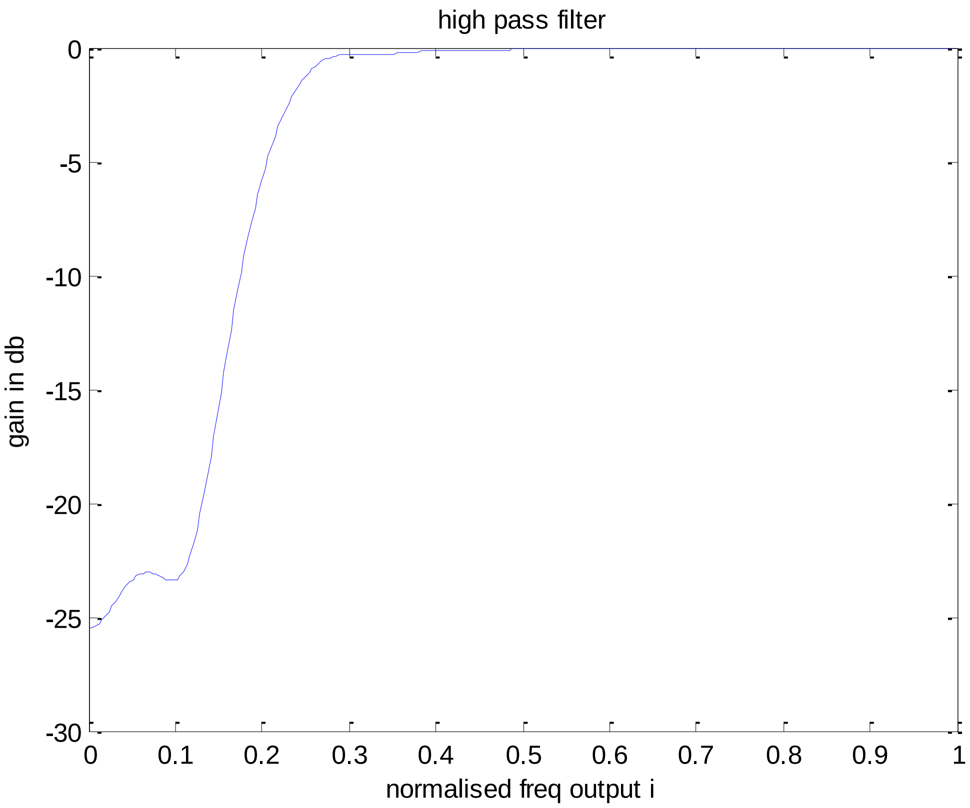
```
den=14.6*(fs-fp)/f;
n=ceil(num/den);
n1=n+1;
if(rem(n,2)~=0);
n1=n;
n=n-1;
end;
c=input('enter the type of window function 1.rectangular 2.trangular 3.kaiser:n=');
if(c==1);
y=rectwin(n1);
disp('rectangular window filter response');
end;
if(c==2);
y=triang(n1);
disp('triangular window filter response');
end;
if(c==3);
alpha=input('enter the alpha value=');
y=kaiser(n1,alpha);
disp('kaiser window filter response');
end;
% high pass filter %
b=fir1(n,wp,'high',y);
[n,o]=freqz(b,1,256);
m=20*log10(abs(n));
plot(o/pi,m);
xlabel('normalised freq output i');
ylabel('gain in db');
title('high pass filter');
```

OUTPUT:-

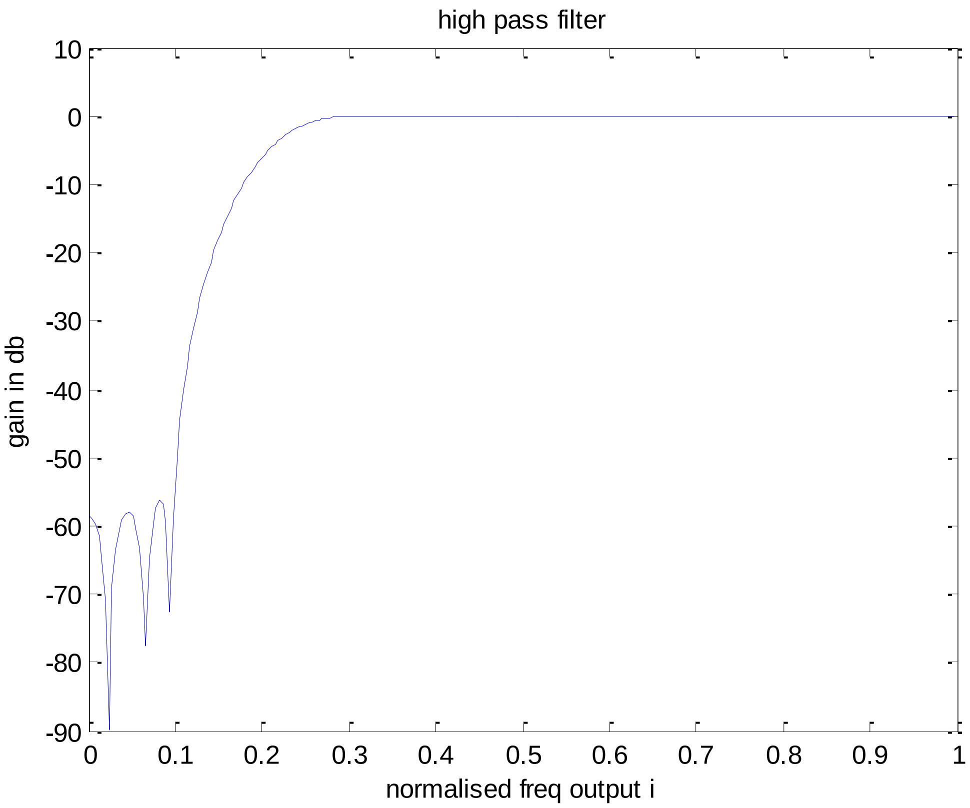
```
enter the pass band ripple:rp=0.02
enter the stop band ripple:rs=0.01
enter the pass band freq :fp=1000
enter the stop band freq:fp=1500
enter the sampling freq:f=10000
enter the type of window function 1.rectangular 2.trangular 3.kaiser:n=1
rectangular window filter response
```



enter the pass band ripple:rp=0.02
enter the stop band ripple:rs=0.01
enter the pass band freq :fp=1000
enter the stop band freq:fp=1500
enter the sampling freq:f=10000
enter the type of window function 1.rectangular 2.trangular 3.kaiser:n=2
triangular window filter response



enter the pass band ripple:rp=0.02
enter the stop band ripple:rs=0.01
enter the pass band freq :fp=1000
enter the stop band freq:fp=1500
enter the sampling freq:f=10000
enter the type of window function 1.rectangular 2.trangular 3.kaiser:n=3
enter the alpha value=5
kaiser window filter response



RESULT:-

The implementation of Butterworth High passes FIR Filters Completed.

EXERCISE PROGRAM:-

1. Write a matlab program to design FIR high pass filter using hamming/hanning window?
2. Write a matlab program to design FIR high pass filter using hamming& blackman window?
3. Write a matlab program to design FIR high pass filter with cutoff frequency .5pi using frequency sampling method?
4. Write a matlab program to plot the frequency response of high pass filter using Kaiser window for different values of beeta?
5. Write a matlab program to design FIR high pass filter for the given specifications using Kaiser window?
6. Write a matlab program to design a 25-tap Hilbert transformer using Bartlett and hamming windows and plot their frequency response?
7. Write a matlab program to design a 25-tap high pass filter with cutoff frequency .5pi radians using rectangular and hamming windows and plot their frequency response?
8. Write a matlab program to design a 25-tap high pass filter with cutoff frequency .5pi radians using rectangular and Blackman windows and plot their frequency response?
9. Write a matlab program to design a 25-tap band pass filter with cutoff frequency .25pi & .75pi radians using rectangular and hamming windows and plot their frequency response?
10. Write a matlab program to design a 25-tap Hilbert transformer using Bartlett window and plot the frequency response?
11. Write a matlab program to design a 25-tap Hilbert transformer using hamming window and plot the frequency response?
12. Write a matlab program to design FIR low pass filter using hamming window?
13. Write a matlab program to design FIR low pass filter using blackman window?
14. Write a matlab program to design FIR high pass filter using blackman window?
15. Write a matlab program to design a 25-tap bands top filter with cutoff frequency .25pi & .75pi radians using rectangular and hamming windows and plot their frequency response?
16. Write a matlab program to design a 25-tap Hilbert transformer using Bartlett window and plot the frequency response?
17. Write a matlab program to design a 25-tap Hilbert transformer using hamming window and plot the frequency response?
18. Write a matlab program to design FIR low pass filter using hamming window?
19. Write a matlab program to design FIR low pass filter using blackman window?
20. Write a matlab program to design FIR high pass filter using blackman window?

VIVA QUESTIONS:-

1. What is FIR filter?
2. State properties of FIR filters?
3. Why FIR filters are inherently stable?
4. How linear phase is achieved in FIR filters?
5. Given an approximate formula for order of FIR filter?
6. What is the delay of a linear-phase FIR?
7. What is the Z transform of a FIR filter?
8. What is the frequency response formula for a FIR filter?
9. How Can I calculate the frequency response of a FIR using (DFT)?
10. What is the DC gain of a FIR filter?
11. What do you meanby cut-off frequency?
12. Give the difference between analog and digital filter?
13. What is the difference between type 1 and type 2 filter structure?
14. what is the role of delay element in filter design?
15. Explain how the frequency is filter in filters?
16. Differences between Butterworth chebyshev filters?
17. Can IIR filters be Linear phase? how to make it linear Phase?
18. What is the special about minimum phase filter?
19. What is the special about minimum phase filter?
20. In signal processing, why we are much more interested in orthogonal transform?
21. What are the filter specifications required to design the analog filters?
22. What is meant by frequency response of filter?
23. What is meant by magnitude response?
24. What is meant by phase response?
25. Differentiate ideal filter and practical filter responses.
26. What are the different types of analog filter approximations?
27. Define order of the filter and explain important role it plays in designing of a filter.
28. Explain advantages and disadvantages of Butterworth filter
29. Explain advantages and disadvantages of Chebyshev filter
30. Chebyshev is better than Butterworth filter?

The calculation of the digital filter function (DSP) is done by the CPU. For that a FIR algorithm of Infineon's TriLib is used. The coefficients for the filter can be calculated with an appropriate software program. For this application note the program "ScopeFIR" was used. Finally the GPTA of the TC1775 generates a PWM signal with the digital filtered signal.

EXPERMENT NO-8

IMPLEMENTATION OF LP IIR FILTERS

AIM: -

Implementation of Low Pass IIR filters for given sequence.

SOFTWARE REQURIED:-

1. MATLAB R2010a.
2. Windows XP SP2.

THEORY:-

Infinite impulse response (IIR) is a property applying to many linear time-invariant systems. Common examples of linear time-invariant systems are most electronic and digital filters. Systems with this property are known as IIR systems or IIR filters, and are distinguished by having an impulse response which does not become exactly zero past a certain point, but continues indefinitely. This is in contrast to a finite impulse response in which the impulse response h(t) does become exactly zero at times t > T for some finite T, thus being of finite duration.

PROCEDURE:-

- Open MATLAB
- Open new M-file
- Type the program
- Save in current directory
- Compile and Run the program
- For the output see command window\ Figure window

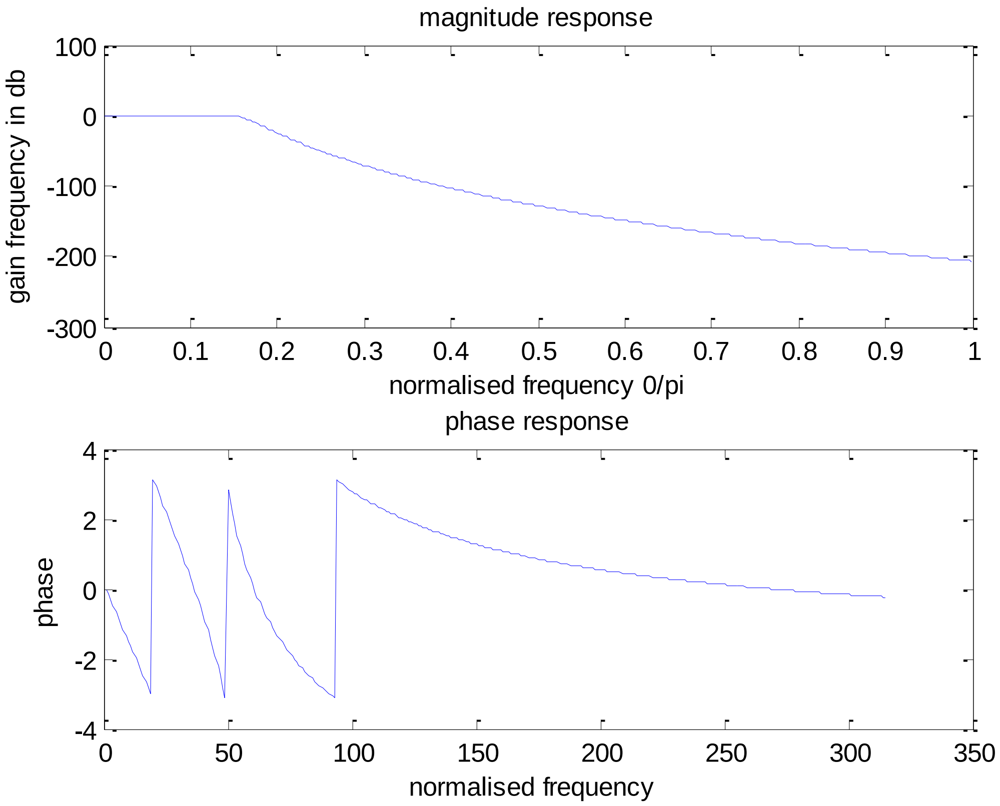
PROGRAM:-

% IIR LOW PASS FILTER %

```
clc;
clear all;
close all;
rp=input('enter the pass band ripple:rp=');
rs=input('enter the stop band ripple:rs=');
fp=input('enter the pass band frequency:fp=');
fs=input('enter the stop band frequency:fs=');
f=input('enter the sampling frequency:f=');
wp=2*fp/f;
ws=2*fs/f;
[N,wc]=buttord(wp,ws,rp,rs,'s');
[b,a]=butter(N,wc,'low','s');
w=0:0.01:pi;[n,o]=freqz(b,1,256);
[n,omega]=freq(b,a,w);
m=20*log10(abs(n));
subplot(2,1,1);
plot(omega/pi,m);

xlabel('normalised frequency 0/pi');
ylabel('gain frequency in db');
title('magnitude response');
subplot(2,1,2);
plot(angle,(n));
xlabel('normalised frequency ');
ylabel('phase');
title('phase response');
```

OUTPUT:-

enter the pass band ripple:rp=0.15
enter the stop band ripple:rs=60
enter the pass band frequency:fp=1500
enter the stop band frequency:fs=3000
enter the sampling frequency:f=7000



RESULT:-

Thus IIR lowpass filter is designed using MATLAB.

EXERCISE PROGRAM:-

1. Write a matlab program to generate IIR chebyshev analog lowpass filter?
2. Write a matlab program to design a Butterworth lowpass filter for the specifications?
3. Write a matlab program to design a Butterworth bandpass filter for the specifications?
4. Write a matlab program to design a Butterworth highpass filter for the specifications?
5. Write a matlab program to design a Butterworth bandreject filter for the specifications?
6. Write a matlab program to design a chebyshev -I lowpass filter for the specifications?
7. Write a matlab program to design a chebyshev -II lowpass filter for the specifications?
8. Write a matlab program to design a chebyshev -I bandpass filter for the specifications?
9. Write a matlab program to design a chebyshev -II bandpass filter for the specifications?
10. Write a matlab program to design a chebyshev -I high pass filter for the specifications?

11. Write a matlab program to design a chebyshev -II high pass filter for the specifications?
12. Write a matlab program to design a chebyshev -I bandreject filter for the specifications?
13. Write a matlab program to design a chebyshev -II bandreject filter for the specifications?
14. Write a matlab program to generate IIR chebyshev analog highpass filter?
15. Write a matlab program to design a Butterworth highpass filter for the specifications?
16. Write a matlab program to design a Butterworth bandpass filter for the specifications?
17. Write a matlab program to design a Butterworth highpass filter for the specifications?
18. Write a matlab program to design a Butterworth bandreject filter for the specifications?
19. Write a matlab program to design a chebyshev -II high pass filter for the specifications?
20. Write a matlab program to design a chebyshev -I bandreject filter for the specifications?

VIVA QUESTIONS:-

1. What is the difference b/w analog and digital filter?

2. State the advantages & disadvantages of digital filters?
3. What are the different types of digital filters?
4. What are the characteristics of Butterworth filters?
5. How the s-plane is mapped to z-plane in impulse invariant transformation?
6. How is the non-periodic nature of the input signal handled?
7. If a have two vectors how will i check the orthogonality of those vectors.
8. What is the importance of decimation for a given signal/sequence?
9. What do you mean Aliasing? What is the condition to avoid aliasing for sampling?
10. How does poly phasefiltering save computations in a decimation filter?
11. Give any practical application of decimation?
12. Which signals can be downsampled?
13. What happens if I violate the Nyquist criteria in down sampling or decimating?
14. Can we do decimate in multiple stages?
15. What are "decimation" and "downsampling"?
16. What isthe "decimation factor
17. How does polyphase filtering save computations in an interpolation filter?
18. Why do we need I&Q signals?
19. What is Interpolation and decimation filters and why we need it?
20. What are "upsampling" and "interpolation"?
21. Define power signal.
22. Define energy signal.
23. Define power spectral density of a signal
24. How the energy of a signal can be calculated?
25. Explain difference between energy spectral density and
26. power spectral density
27. Explain the PSD plot.
28. What is the importance of PSD?
29. What are the applications of PSD?
30. Explain MATLAB function randn(size(n))

IIR filter structure which is composed of cascaded sections of second order Direct Form I filters that use magnitude truncation. IIR filters are used in Small monitor loudspeaker. IIR filters are used electronic crossover for a 3-way loudspeaker.

EXPERMENT NO-9

IMPLEMENTATION OF HP IIR FILTERS

AIM: -
To implement the analog & digital High Pass IIR filter.

SOFTWARE REQURIED:-

1. MATLAB R2010a.
2. Windows XP SP2.

THEORY:-

Infinite impulse response (IIR) is a property applying to many linear time-invariant systems. Common examples of linear time-invariant systems are most electronic and digital filters. Systems with this property are known as IIR systems or IIR filters, and are distinguished by having an impulse response which does not become exactly zero past a certain point, but continues indefinitely. This is in contrast to a finite impulse response in which the impulse response h(t) does become exactly zero at times t > T for some finite T, thus being of finite duration.

PROCEDURE:-

- Open MATLAB
- Open new M-file
- Type the program
- Save in current directory
- Compile and Run the program

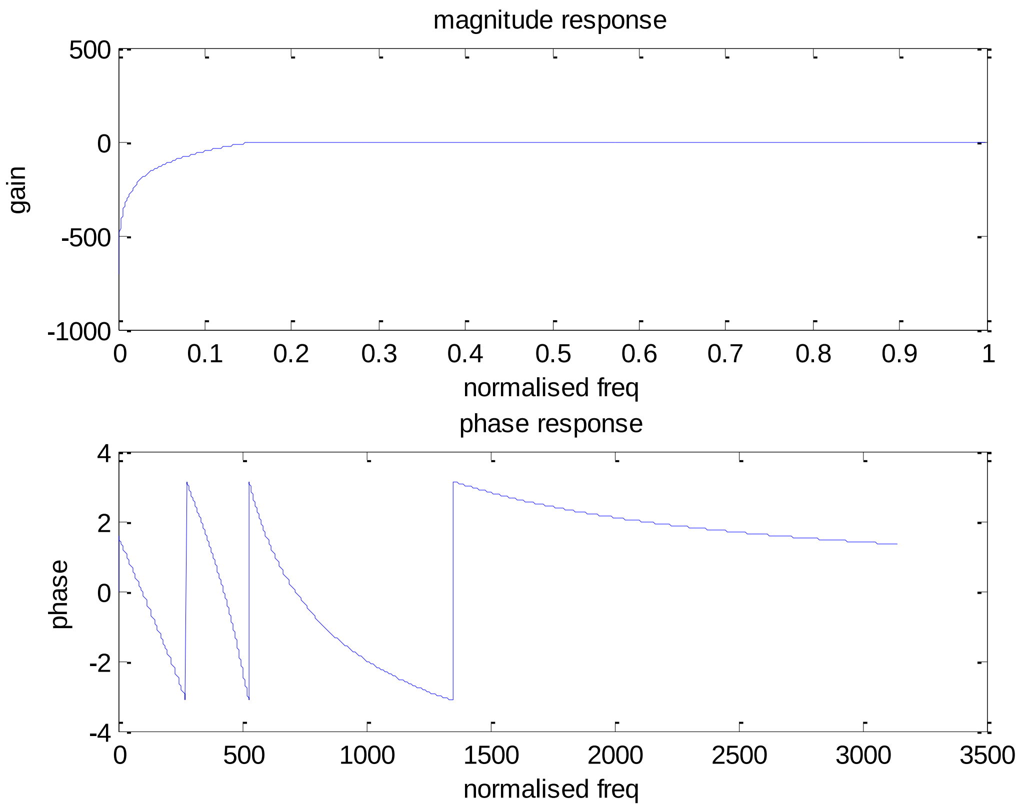
- For the output see command window\ Figure window

PROGRAM:-

```
clc;
clear all;
close all;
disp('enter the sepecifications of iir filter');
rp=input('enter the pass band ripple:rp=');
rs=input('enter the stop band ripple:rs=');
wp=input('enter the pass band freq:wp=');
ws=input('enter the stop band freq:ws=');
fs=input('enter the sampling freq fs=');
w1=2*wp/fs;
w2=2*ws/fs;
[N,wc]=buttord(w1,w2,rp,rs,'s');
disp('freq resp of iir high pass filter is:');
[b,a]=butter(N,wc,'high','s');
w=0:0.001:pi;
[n,omega]=freqs(b,a,w);
m=20*log10(abs(n));
subplot(2,1,1);
plot(omega/pi,m);




xlabel('normalised freq');
ylabel('gain');
title('magnitude response');
subplot(2,1,2);
plot(angle(n));
xlabel('normalised freq');
ylabel('phase');
title('phase response');
```

OUTPUT:-

```
enter the sepecifications of iir filter
enter the pass band ripple:rp=0.15
enter the stop band ripple:rs=60
enter the pass band freq:wp=1500
enter the stop band freq:ws=3000
enter the sampling freq fs=7000
freq resp of iir high pass filter is:
```



RESULT:-

Thus IIR lowpass filter is designed using MATLAB.

EXERCISE PROGRAM:-

1. Write a matlab program to generate IIR chebyshev analog highpass filter?
2. Write a matlab program to design a Butterworth highpass filter for the specifications?
3. Write a matlab program to design a Butterworth bandpass filter for the specifications?
4. Write a matlab program to design a Butterworth highpass filter for the specifications?
5. Write a matlab program to design a Butterworth bandreject filter for the specifications?
6. Write a matlab program to design a chebyshev -II high pass filter for the specifications?
7. Write a matlab program to design a chebyshev -I bandreject filter for the specifications?
8. Write a matlab program to design a chebyshev -II bandreject filter for the specifications?
9. Write a matlab program to design a chebyshev -I highpass filter for the specifications?
10. Write a matlab program to design a chebyshev -II highpass filter for the specifications?
11. Write a matlab program to design a chebyshev -I high pass filter for the specifications?
12. Write a matlab program to generate IIR chebyshev analog lowpass filter?
13. Write a matlab program to design a Butterworth lowpass filter for the specifications?
14. Write a matlab program to design a Butterworth bandpass filter for the specifications?
15. Write a matlab program to design a Butterworth highpass filter for the specifications?
16. Write a matlab program to design a Butterworth bandreject filter for the specifications?
17. Write a matlab program to design a chebyshev -I lowpass filter for the specifications?
18. Write a matlab program to design a chebyshev -II lowpass filter for the specifications?
19. Write a matlab program to design a chebyshev -I bandpass filter for the specifications?
20. Write a matlab program to design a chebyshev -II bandpass filter for the specifications?

VIVA QUESTIONS:-

1. What are the steps in designing the IIR filters?
2. State the disadvantages of impulse invariant transformation?
3. Why impulse invariant transformation is not suitable for design of high pass filters?
4. What is frequency relationship for bilinear transformation?
5. What is the frequency relationship for bilinear transformation?
6. Why interpolate,i needed for any signal/sequence?
7. What is the "interpolation factor"?
8. Which signals can be interpolated?
9. Can interpolate will happens in multiple stages? If yes give reason?
10. Give any example of a FIR interpolator?
11. Define impulse response.
12. Give me one example for impulse response.
13. Write the Formula for impulse response.

14. What are major role in order & length?
15. Define power spectral Density?
16. What is the need for spectral estimation?
17. Determine the power spectrum density?
18. What is the relation between auto correlation & spectral density?
19. Give the estimation of auto correlation function & power density for random Signals?
20. Explain power spectrum estimation using the Bartlett window?
21. What are the characteristics of Butterworth filters?
22. How the s-plane is mapped to z-plane in impulse invariant transformation?
23. How is the non-periodic nature of the input signal handled?
24. If a have two vectors how will i check the orthogonality of those vectors.
25. What is the importance of decimation for a given signal/sequence?
26. What do you mean Aliasing? What is the condition to avoid aliasing for sampling?
27. How does poly phasefiltering save computations in a decimation filter?
28. Give any practical application of decimation?
29. Which signals can be downsampled?
30. What happens if I violate the Nyquist criteria in down sampling or decimating?

IIR filter structure which is composed of cascaded sections of second order Direct Form I filters that use magnitude truncation. IIR filters are used in Small monitor loudspeaker. IIR filters are used electronic crossover for a 3-way loudspeaker.

EXPERMENT NO-10

SINUSOIDAL SIGNAL THROUGH FILTERING

AIM: -

Generation of Sine Wave & Illustration of the Sampling Process in the Time Domain.

SOFTWARE REQURIED:-

1. MATLAB R2010a.
2. Windows XP SP2.

THEORY:-

Sinusoidal Signal Generation

The sine wave or sinusoid is a mathematical function that describes a smooth repetitive oscillation. It occurs often in pure mathematics, as well as physics, signal processing, electrical engineering and many other fields. Its most basic form as a function of time (t) where:

• A, the amplitude, is the peak deviation of the function from its center position.
• ω, the angular frequency, specifies how many oscillations occur in a unit time interval, in radians per second
• ϕ, the phase, specifies where in its cycle the oscillation begins at t = 0.

A sampled sinusoid may be written as:

$$x(n) = A\sin(2\pi \frac{f}{f_s} n + \vartheta)$$

PROCEDURE:-

- Open MATLAB
- Open new M-file
- Type the program
- Save in current directory

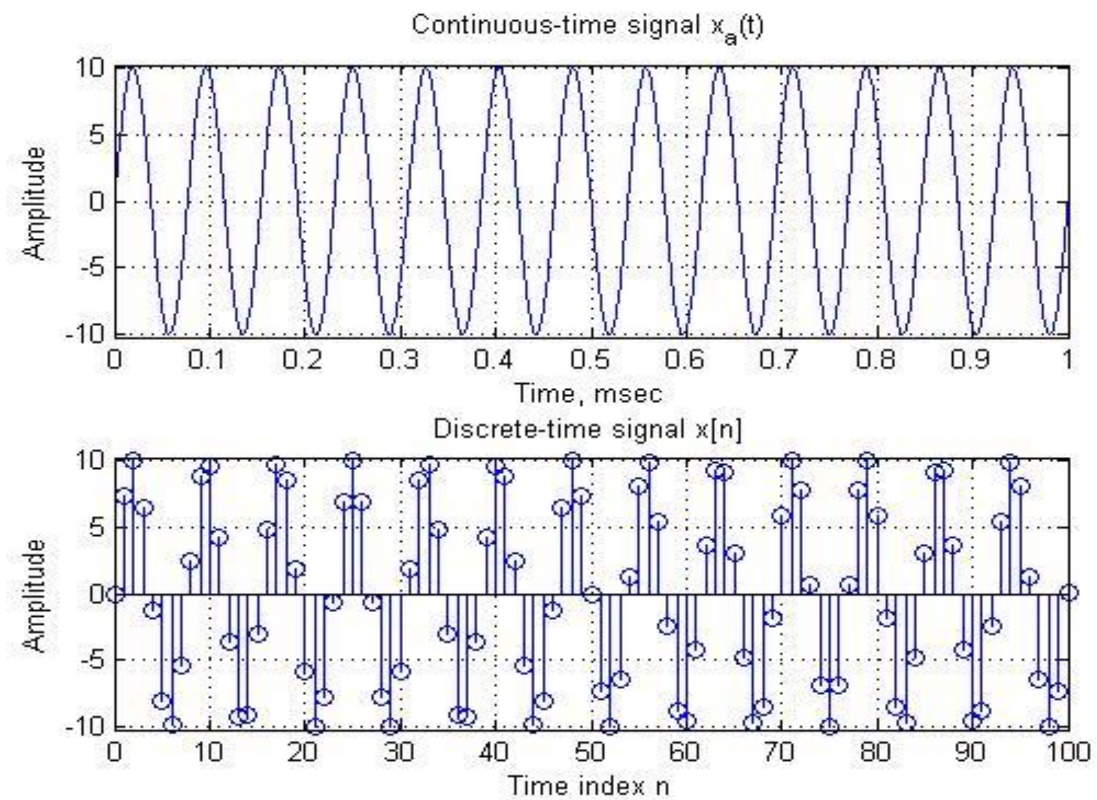
- Compile and Run the program
- For the output see command window\ Figure window

PROGRAM:-

% Generation of Sine Wave & Illustration of the Sampling Process in the Time Domain

```
clc;
t = 0:0.0005:1; a = 10
f = 13;
xa = a*sin(2*pi*f*t);
subplot(2,1,1)
plot(t,xa);grid xlabel('Time,
msec'); ylabel('Amplitude');
title('Continuous-time signal x_{a}(t)'); axis([0
1 -10.2 10.2])
subplot(2,1,2); T =
0.01;
n = 0:T:1;
xs = a*sin(2*pi*f*n); k =
0:length(n)-1; stem(k,xs);
grid
xlabel('Time index n');
ylabel('Amplitude'); title('Discrete-
time signal x[n]'); axis([0 (length(n)-1)
-10.2 10.2])
```

OUTPUT:-



RESULT:-

Sinusoidal signal is generated by using MATLAB.

EXERCISE PROGRAM:-

1. Write program to get Discrete time Sinusoidal Signal?
2. Write program to get Fourier Transform of Sinusoidal Signal?
3. Write program to get Inverse Fourier Transform of Sinusoidal Signal?
4. Write a matlab program for generating u(n)-u(n-1)?
5. Write program to get Discrete time co-Sinusoidal Signal?
6. Write program to get Discrete time saw tooth Signal?
7. Write program to get Discrete time triangular Signal?
8. Write program to get addition of two sinusoidal sequences?
9. Write program to get exponential sequence?
10. Write program to get Fourier Transform of Co-Sinusoidal Signal?
11. Write program to get Inverse Fourier Transform of Co-Sinusoidal Signal?
12. Write program to get exponential decaying sequence?
13. Write program to get exponential growing sequence?
14. Write program to get addition of two Co-sinusoidal sequences?
15. Write program to get continues time Square Signal?
16. Write program to get continues time Sinusoidal Signal?
17. Write program to get Fourier Transform of Sinusoidal Signal?
18. Write program to get Inverse Fourier Transform of Sinusoidal Signal?
19. Write a matlab program for generating u(n)+u(n-2)?
20. Write program to get continues time co-Sinusoidal Signal?

VIVA QUESTIONS:-

1. Define sinusoidal signal?
2. Define C.T.S?
3. Define D.T.S?
4. Compare C.T.S & D.T.S?
5. Draw the C.T.S & D.T.S diagrams?
6. Give the formula for PSD.
7. What is filter?
8. Define Stem, Plot, Plot3,fplot, ezplot, linspace, flyplr, grid,mesh and legend.
9. Draw the C.T.S & D.T.S diagrams.
10. Which built in function is used to solve a given difference equation?
11. What is frequency response? Give equation for first order system andsecond order system?
12. What is an LTI system?
13. What is steady state response?
14. What is Auto Regressive Model? How is the order of auto regressive model is decided?
15.
16. Differentiate between linear and circular convolution.
17. Determine the unit step response of the linear time invariant system with impulse response h(n)=a nu(n) a<1&-a<1
18. Determine the range of values of the parameter a for which linear time invariant system with impulse response h(n)=a u(n) is stable.
19. How is the non-periodic nature of the input signal handled?
20. If a have two vectors how will i check the orthogonality of those vectors.
21. Can IIR filters be Linear phase? how to make it linear Phase?
22. What is the special about minimum phase filter?

23. What is the special about minimum phase filter?
24. In signal processing, why we are much more interested in orthogonal transform?
25. What are the filter specifications required to design the analog filters?
26. What is meant by frequency response of filter?
27. What is meant by magnitude response?
28. What is meant by phase response?
29. Differentiate ideal filter and practical filter responses.
30. What are the different types of analog filter approximations?

Detection of QRS complexes in ECG signals is required to determine heart rate, and it is an important step in the study of cardiac disorders. ECG signals are usually affected by noise of low and high frequency. To improve the accuracy of QRS detectors several methods have been proposed to filter out the noise and detect the characteristic pattern of QRS complex. Most of the existing methods are at a disadvantage from relatively high computational complexity or high resource needs making them less optimized for its implementation on portable embedded systems, wearable devices or ultra-low power chips. We present a new method to detect the QRS signal in a simple way with minimal computational cost and resource needs using a novel non-linear filter.

EXPERMENT NO-11

DTMF SIGNAL GENERATION

AIM: -

The objective of this program is To Generate Dual Tone Multiple Frequency (DTMF) Signals.

SOFTWARE REQURIED:-

1. MATLAB R2010a.
2. Windows XP SP2.

THEORY:-

Dual Tone Multiple Frequency (DTMF) Signals.

PROCEDURE:-

- Open MATLAB
- Open new M-file
- Type the program
- Save in current directory
- Compile and Run the program
- For the output see command window\ Figure window

PROGRAM:-

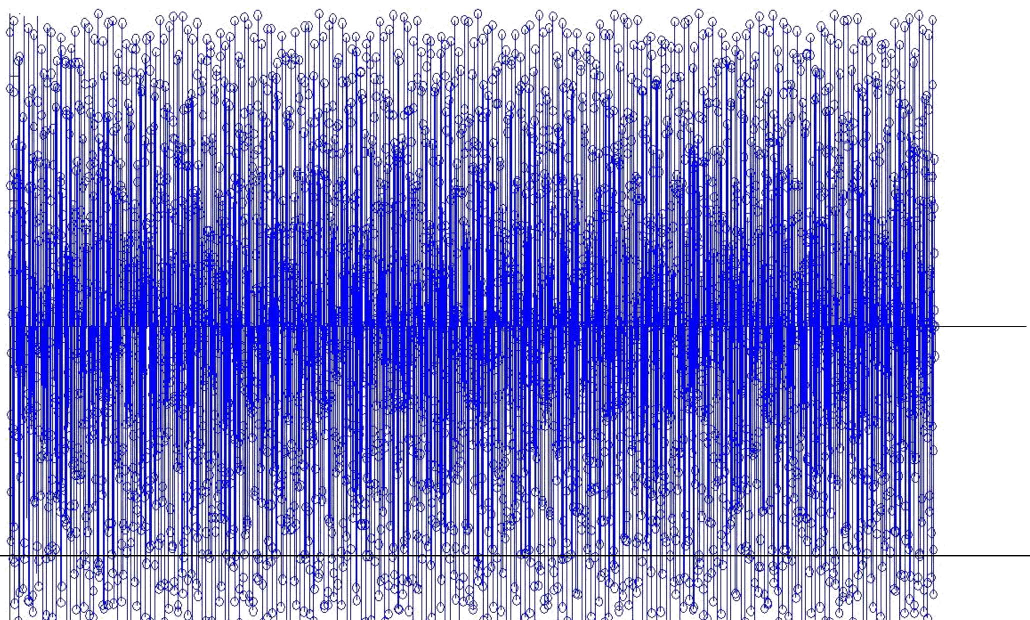
% Dual Tone Multiple Frequency (DTMF) Signals.

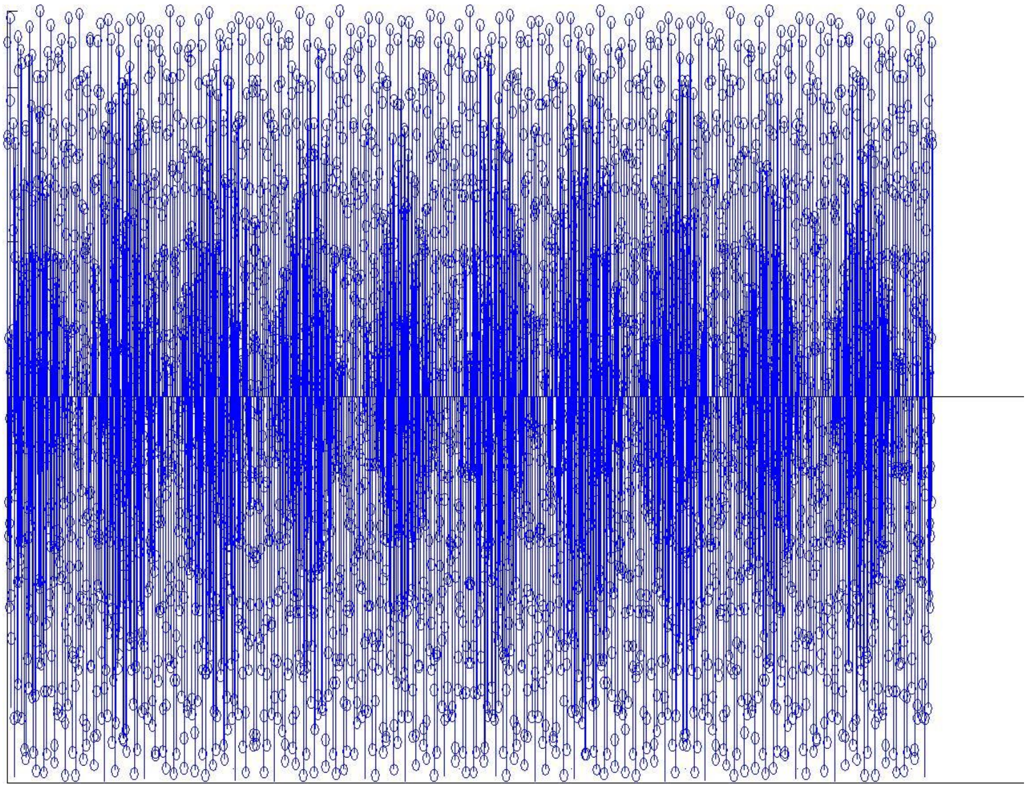
```
clc; clearall;

closeall;

number=input('enter a phone number with no spaces','s');
%number=1;

fs=8192;   % fs is the sampling Frequency

T=0.5;     % T stores how for how long a tone will be played

x= 2*pi*[697 770 852 941];

y= 2*pi*[1209 1336 1477 1633];
```

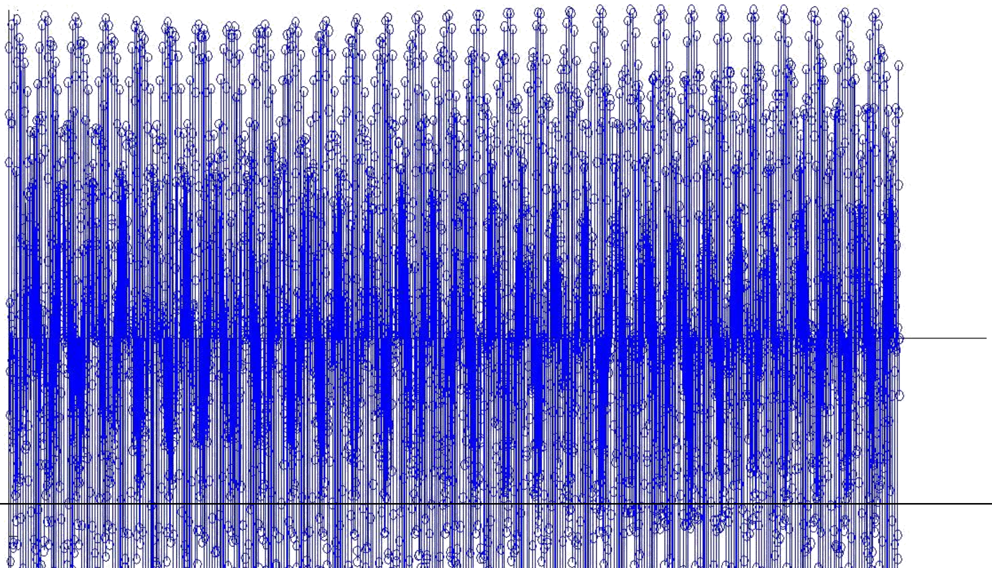
```
t=[0:1/fs:T]'

tx=[sin(x(1)*t),sin(x(2)*t),sin(x(3)*t),sin(x(4)*t)]/2;

ty=[sin(y(1)*t),sin(y(2)*t),sin(y(3)*t),sin(y(4)*t)]/2; for
k=1:length(number)
  switch number(k) case
    '1'
      tone = tx(:,1)+ty(:,1);
      sound(tone); stem(tone);
    case '2'
      tone = tx(:,1)+ty(:,2);
      sound(tone);
      stem(tone);
    case '3'
      tone = tx(:,1)+ty(:,3);
      sound(tone); stem(tone);
    case '4'
     tone = tx(:,2)+ty(:,1); sound(tone); stem(tone); otherwise
      disp('invalid number');
  end pause(2.70)
end;
```

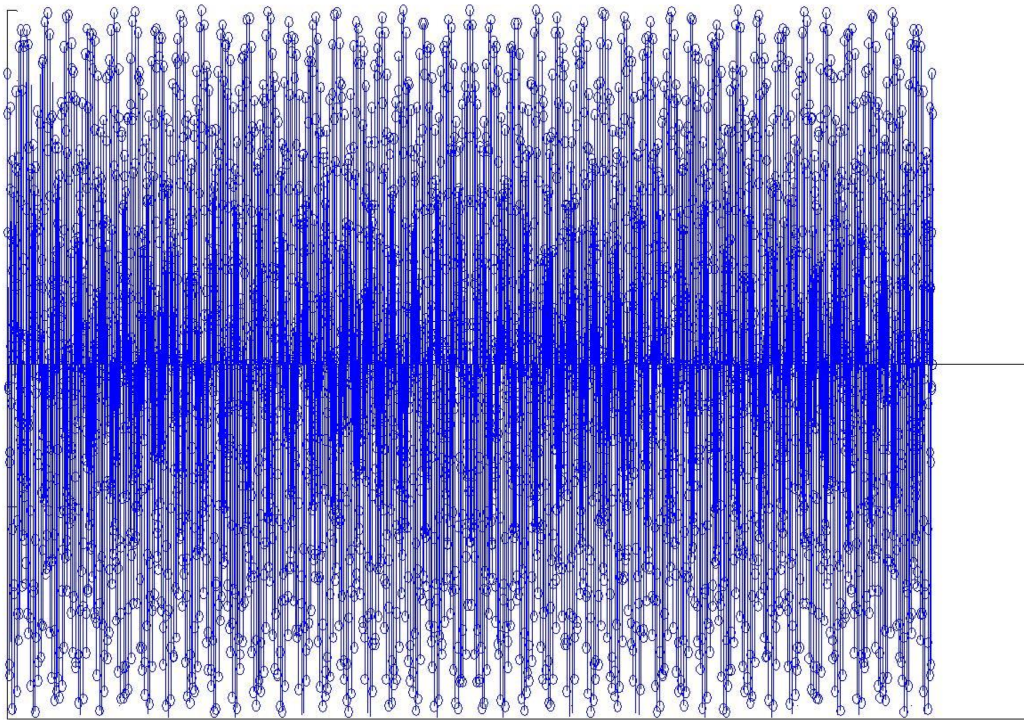
OUTPUT:-

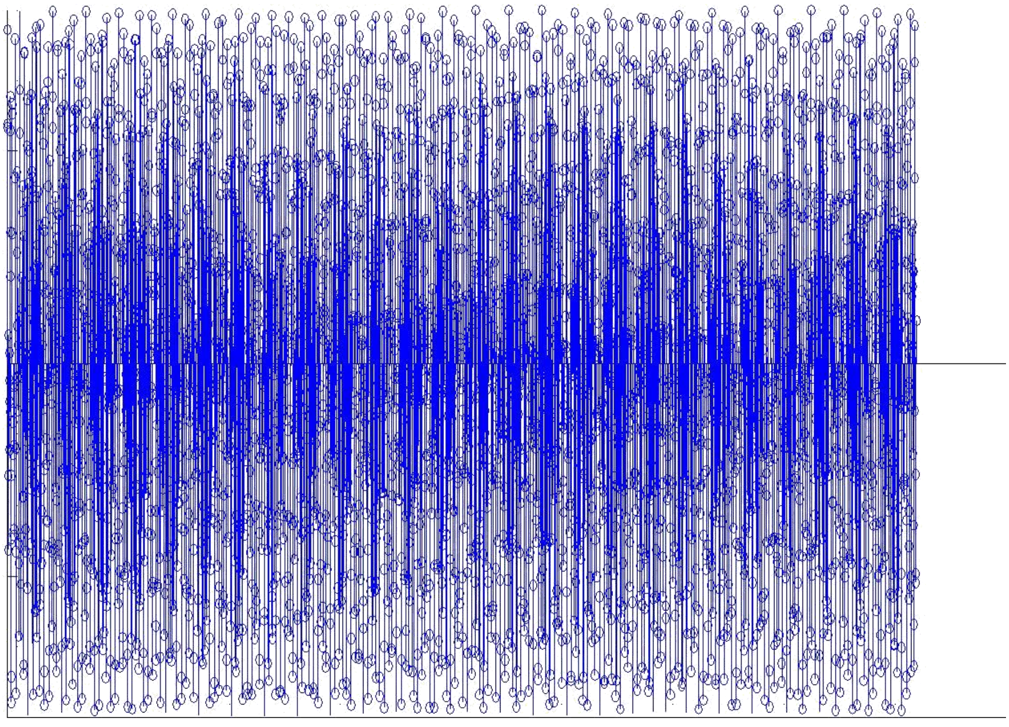
Input: 01234











RESULT:-

Dual Tone Multiple Frequency (DTMF) Signals are generated by using MAT LAB.

EXERCISE PROGRAM:-

1. Write a matlab program to generate a sine wave with amplitude = 3, frequency 20Hz?
2. Write a matlab program to generate a cos wave with amplitude = 3, frequency 20Hz?
3. Write a matlab program to generate a triangular wave with amplitude = 8, frequency 10Hz?
4. Write a matlab program to generate a square wave with amplitude = 2, frequency 10kHz?
5. Write a matlab program to generate a sinc wave with amplitude = -8, frequency5Khz?
6. Write a matlab program to generate a sine wave with amplitude = 7, frequency 29Hz.
7. Write a matlab program to generate a cos wave with amplitude = 9, frequency 50Hz.
8. Write a matlab program to generate a triangular wave with amplitude = 24, frequency 100Hz.
9. Write a matlab program to generate a square wave with amplitude = 12, frequency 10kHz.
10. Write a matlab program to generate a sinc wave with amplitude = 5, frequency5Khz.
11. Write a matlab program to generate a sine wave with amplitude = 17, frequency 29kHz.
12. Write a matlab program to generate a cos wave with amplitude = 19, frequency600kHz.
13. Write a matlab program to generate a triangular wave with amplitude = 24, frequency 100Hz.
14. Write a matlab program to generate a sawtooth wave with amplitude = 20, frequency 15kHz.
15. Write a matlab program to generate a sinc wave with amplitude = 8, frequency85Khz.
16. Write a matlab program to generate a triangular wave with amplitude = 10, frequency 20Hz?
17. Write a matlab program to generate a square wave with amplitude = 8, frequency 1kHz?
18. Write a matlab program to generate a sinc wave with amplitude = frequency5Khz?
19. Write a matlab program to generate a sine wave with amplitude = 7, frequency 29Hz?
20. Write a matlab program to generate a cos wave with amplitude = 9, frequency 50Hz?

VIVA QUESTIONS:-

1. Define Signal?
2. Define determistic and Random Signal?
3. Define Delta Function?
4. What is Signal Modeling?
5. Define Periodic and a periodic Signal?
6. Define impulse response.
7. Give me one example for impulse response.
8. Write the Formula for impulse response.
9. What are major role in order & length?
10. Define power spectral Density?
11.What is the need for spectral estimation?
12.Determine the power spectrum density?
13. What is the relation between auto correlation & spectral density?
14. Give the estimation of auto correlation function & power density for random Signals?
15. Explain power spectrum estimation using the Bartlett window?
16. Give the formula for PSD.
17. What is filter?
18. What do you mean by phase spectrum and magnitude spectrum/ give comparison?
19. How do you reduce spectral leakage?
20. What do you mean by spectral resolution?
21. Define sinusoidal signal
22. Define C.T.S .
23. Define D.T.S.
24. Compare C.T.S & D.T.S.
25. Define Stem, Plot, Plot3,fplot, ezplot, linspace, flyplr, grid,mesh and legend.
26. Draw the C.T.S & D.T.S diagrams.
27. Define signal and signal processing.
28. Differentiate digital and analog signals?
29. How the DSP processor will differ from conventional processors?
30. Expand the abbreviation TMS320C 5X/6X

In earlier days, our telephone systems were operated by manually in a telephone exchange room. The callers will pick up the phone and giving instruction to the operator to connect their destination line. The DTMF technology provides ultimate solutions for the telephone industries which is used to switch two lines automatically. The DTMF stands for ‘Dual Tone Multi-frequency’ which is one of the techniques for converting the analogue signal to digital using DTMF decoder. The DTMF decoder circuit mostly used in mobile communications system which recognizes the sequence of DTMF tones from the standard keypad of the mobile phone.

EXPERMENT NO-12

INTERPOLATION

AIM: -

The objective of this program is To Perform up sampling on the Given Input Sequence.

SOFTWARE REQURIED:-

1. MATLAB R2010a.
2. Windows XP SP2.

THEORY:-

Up sampling is interpolation, applied in the context of digital signal processing and sample rate conversion. When up sampling is performed on a sequence of samples of a continuous function or signal, it produces an approximation of the sequence that would have been obtained by sampling the signal at a higher rate (or density, as in the case of a photograph). For example, if compact disc audio is up sampled by a factor of 5/4, the resulting sample-rate increases from 44,100 Hz to 55,125 Hz.

PROCEDURE:-

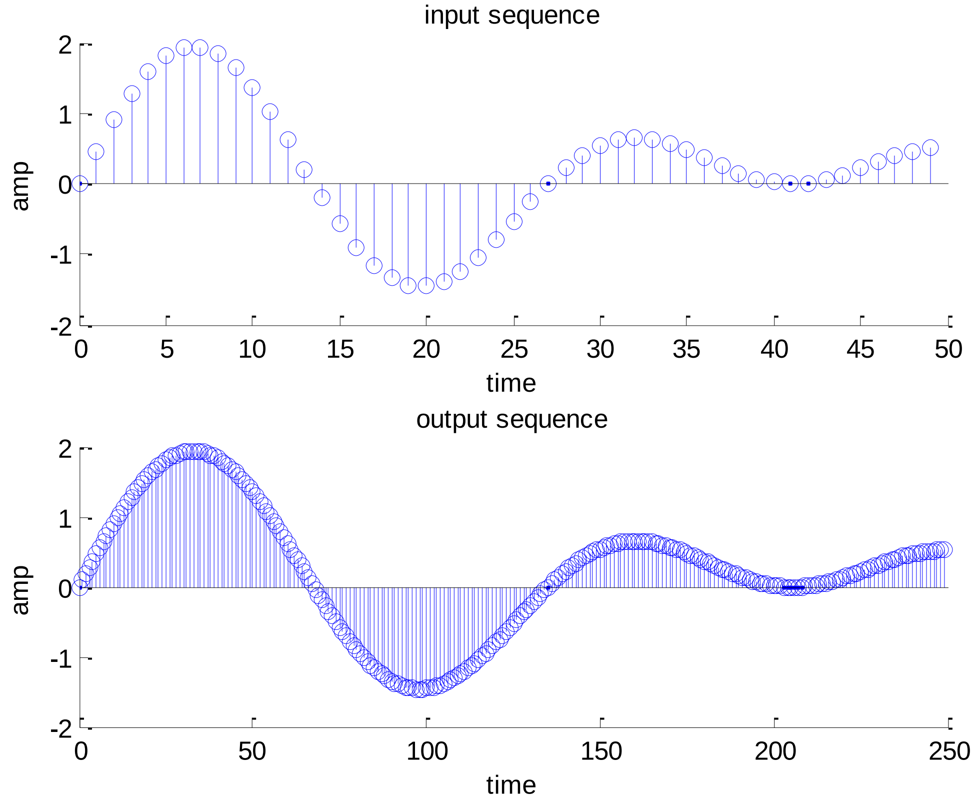
- Open MATLAB
- Open new M-file
- Type the program
- Save in current directory
- Compile and Run the program
- For the output see command window\ Figure window

PROGRAM:-

```
%interpolation%
clc;
clear all;
close all;
N=input('enter sample value');
n=0:N-1;
L=input('enter up sampling factor');
x=sin(2*pi*0.043*n)+sin(2*pi*0.031*n);
y=interp(x,L);
subplot(2,1,1);
stem(n,x(1:N));
xlabel('time');
ylabel('amp');
title('input sequence');
t=0:(N*L)-1;
subplot(2,1,2);
stem(t,y(1:N*L));
xlabel('time');
ylabel('amp');
title('output sequence');
```

OUTPUT:-

```
enter sample value50
enter up sampling factor
```



RESULT:-

This MATLAB program has been written to perform interpolation on the Given Input Sequence.

EXERCISE PROGRAM:-

1. Write a matlab program to illustrate the effect of anti-aliasing filter?
2. Write a matlab program to illustration of upsampling?
3. Write a matlab program to illustration of downsampling?
4. Write a matlab program to illustration of effect of upsampling in frequency domain?
5. Write a matlab program to illustration of effect of downsampling in frequency domain?

6. Write a matlab program to illustrate the concept of aliasing?
7. Write a matlab program to plot magnitude response of comb filter?
8. Write a matlab program to plot magnitude response of allpass filter?
9. Write a matlab program to plot magnitude & unwrapped phase response of two transfer functins?
10. Write a matlab program to design a filter that eliminates high frequency component in a CT signal?
11. Write a matlab program to illustrate the effect of aliasing filter?
12. Write a matlab program to illustration of up sampling?
13. Write a matlab program to illustration of down sampling?
14. Write a matlab program to illustration of effect of upsampling in frequency domain?
15. Write a matlab program to illustration of effect of downsampling in frequency domain?
16. Write a matlab program to illustrate the concept of aliasing?
17. Write a matlab program to plot magnitude response of comb filter?
18. Write a matlab program to plot magnitude response of bandpass filter?
19. Write a matlab program to plot magnitude response?
20. Write a matlab program to design a filter that eliminates low frequency component in a CT signa.

VIVA QUESTIONS:-

1. How aliasing can be avoided?
2. Which type of interpolation is used to reconstruct the signal?
3. What is aliasing?
4. Define interpolation?
5. What is pre-alias filter?
6. What is the importance of decimation for a given signal/sequence?
7. What do you mean Aliasing? What is the condition to avoid aliasing for sampling?
8. How does poly phasefiltering save computations in a decimation filter?
9. Give any practical application of decimation?
10. Which signals can be downsampled?
11. What happens if I violate the Nyquist criteria in down sampling or decimating?
12. Can we do decimate in multiple stages?
13. What are "decimation" and "downsampling"?
14. What is the "decimation factor
15. How does poly phase filtering save computations in an interpolation filter?
16. Why do we need I&Q signals?
17. What is Interpolation and decimation filters and why we need it?
18. What are "up sampling" and "interpolation"?
19. Why interpolate, needed for any signal/sequence?
20. What is the "interpolation factor"?
21. What kind of processor is DSP processor?
22. What are the main building
23. blocks of DSP processor?
24. What is the main function of MAC unit?
25. Explain VLIW architecture?
26. Explain the significance of convolution.
27. Define linear convolution.
28. Why linear convolution is called as a periodic convolution?
29. Why zero padding is used in linear convolution?
30. What are the four steps to find linear convolution?

Interpolation is a technique for obtaining new unknown data points within the range of discrete known data points and is often used to recover an image from its downsampled version, or to simply perform image expansion.

EXPERMENT NO-13

DECIMATION

AIM: -

The objective of this program is To Perform Decimation on the Given Input Sequence.

SOFTWARE REQURIED:-

1. MATLAB R2010a.
2. Windows XP SP2.

THEORY:-

In digital signal processing, decimation is the process of reducing the sampling rate of a signal. Complementary to interpolation, which increases sampling rate, it is a specific case of sample rate conversion in a multi-rate digital signal processing system. Decimation utilizes filtering to mitigate aliasing distortion, which can occur when simply down sampling a signal. A system component that performs decimation is called a decimator.

PROCEDURE:-

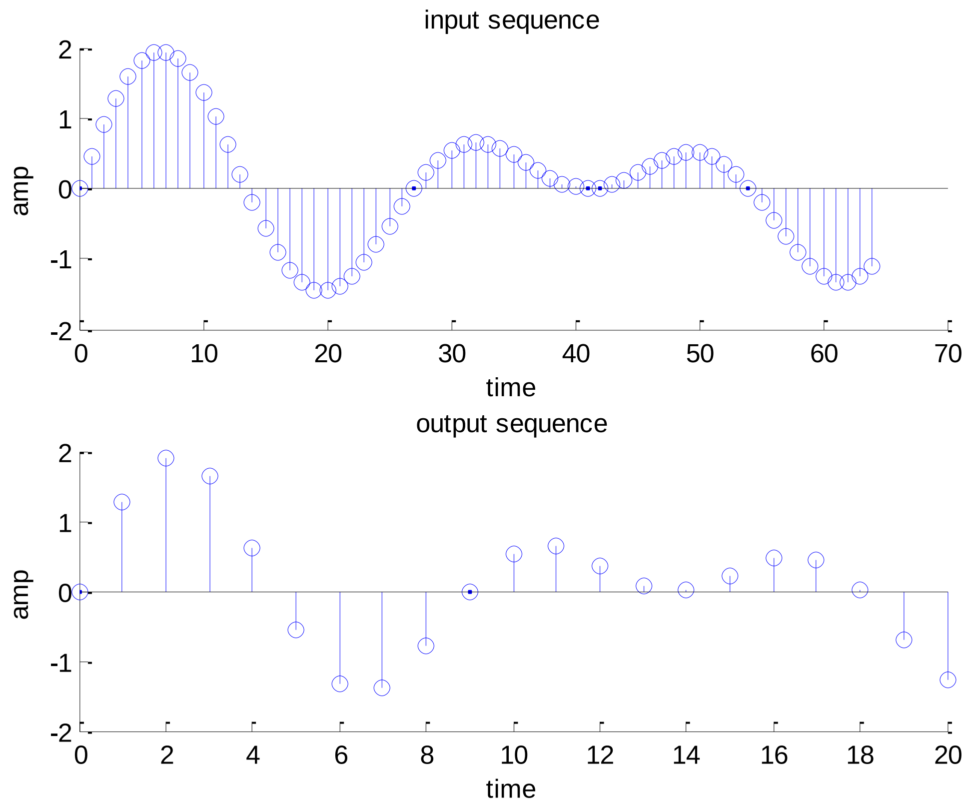
- Open MATLAB
- Open new M-file
- Type the program
- Save in current directory
- Compile and Run the program
- For the output see command window\ Figure window

PROGRAM:-

```
%decimation%
clc;
clear all;
close all;
N=input('enter sample value');
n=0:N-1;
m=input('enter down sampling factor');
x=sin(2*pi*0.043*n)+sin(2*pi*0.031*n);
y=decimate(x,m,'fir');
subplot(2,1,1);
stem(n,x(1:N));
xlabel('time');
ylabel('amp');
title('input sequence');
t=0:(N/m)-1;
subplot(2,1,2);
stem(t,y(1:N/m));
xlabel('time');
ylabel('amp');
title('output sequence');
```

OUTPUT:-

```
enter sample value65
enter down sampling factor3
```



RESULT:-

This MATLAB program has been written to perform Decimation on the Given Input Sequence.

EXERCISE PROGRAM:-

1. Write a matlab program to illustrate the effect of anti-aliasing filter?
2. Write a matlab program to illustration of upsampling?
3. Write a matlab program to illustration of downsampling?
4. Write a matlab program to illustration of effect of upsampling in frequency domain?
5. Write a matlab program to illustration of effect of downsampling in frequency domain?
6. Write a matlab program to illustrate the concept of aliasing?
7. Write a matlab program to plot magnitude response of comb filter?

8. Write a matlab program to plot magnitude response of allpass filter?
9. Write a matlab program to plot magnitude & unwrapped phase response of two transfer functins?
10. Write a matlab program to design a filter that eliminates high frequency component in a CT signal?
11. Write a matlab program to illustrate the effect of aliasing filter?
12. Write a matlab program to illustration of upsampling with sampling factor 5 ?
13. Write a matlab program to illustration of downsampling with sampling factor 5 ?
14. Write a matlab program to illustration of effect of upsampling in time domain?
15. Write a matlab program to illustration of effect of downsampling in time domain?
16. Write a matlab program to illustration of upsampling with sampling factor10?
17. Write a matlab program to illustration of downsampling with sampling factor 15 ?
18. Write a matlab program to illustration of upsampling with sampling factor 20?
19. Write a matlab program to illustration of downsampling with sampling factor 20?
20. Write a matlab program to illustration of downsampling with sampling factor 20?

VIVA QUESTIONS:-

1. Define decimation?
2. Define multi rate signal processing?
3. What are the effects of coefficient quantization in FIR filters?
4. What is quantization process?
5. What is transmultiplexer? What is its use?
6. What is the DC gain of a FIR filter?
7. What do you mean by cut-off frequency?
8. Give the difference between analog and digital filter?
9. What is the difference between type 1 and type 2 filter structure?
10. what is the role of delay element in filter design?
11. Explain how the frequency is filter in filters?
12. Differences between Butterworth chebyshev filters?
13. Can IIR filters be Linear phase? how to make it linear Phase?
14. What is the special about minimum phase filter?
15. What is the special about maximum phase filter?
16. In signal processing, why we are much more interested in orthogonal transform?
17. How is the non-periodic nature of the input signal handled?
18. If a have two vectors how will i check the orthogonality of those vectors.
19. What is the importance of decimation for a given signal/sequence?
20. What do you mean Aliasing? What is the condition to avoid aliasing for sampling?
21. Define impulse response.
22. Give me one example for impulse response.
23. Write the Formula for impulse response.
24. What are major role in order & length?
25. Define power spectral Density?
26. What is the need for spectral estimation?
27. Determine the power spectrum density?
28. What is the relation between auto correlation & spectral density?
29. Give the estimation of auto correlation function & power density for random Signals?
30. Explain power spectrum estimation using the Bartlett window?

It was originally developed for processing broadband, low frequency geophysical data in real time using a small digital processor. Sine and Cosine transforms are applied to sequences of data produced by successively applying a low pass digital filter and decimation by two operator to the original data. The resulting spectra are the average of estimates which are independent in time and represent constant percentage bandwidths. Thus, the technique is particularly well suited to applications where it is desireable to reject intervals of bad data or those which are naturally expressed on a log-frequency scale.

EXPERMENT NO-14

IMPLEMENTATION OF I/D SAMPLING RATE CONVERTERS

AIM: -

To study sampling rate conversion by a rational form using MATLAB.

SOFTWARE REQURIED:-

1. MATLAB R2010a.
2. Windows XP SP2.

THEORY:-

"Up sampling" is the process of inserting zero-valued samples between original samples to increase the sampling rate. (This is called "zero-stuffing".) Up sampling adds to the original signal undesired spectral images which are centered on multiples of the original sampling rate.

"Interpolation", in the DSP sense, is the process of up sampling followed by filtering. (The filtering removes the undesired spectral images.) As a linear process, the DSP sense of interpolation is somewhat different from the "math" sense of interpolation, but the result is conceptually similar: to create "in-between" samples from the original samples. The result is as if you had just originally sampled your signal at the higher rate.

PROCEDURE:-

- Open MATLAB
- Open new M-file
- Type the program
- Save in current directory
- Compile and Run the program
- For the output see command window\ Figure window

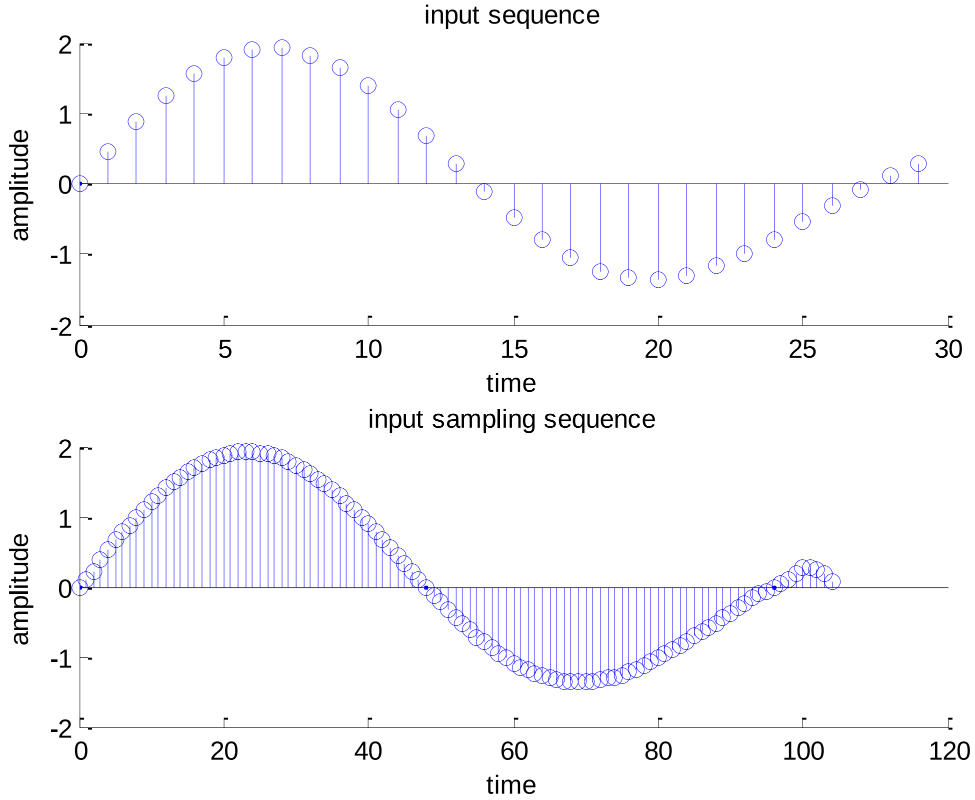
PROGRAM:-

% interpolation/dismation sampling %

```
clc; clear all;close all;
N=input('enter the sample value');
n=0:N-1;
l=input('enter up sampling factor');
m=input('enter down sampling factor');
x=sin(2*pi*0.043*n)+sin(2*pi*0.03*n);
y=resample(x,l,m);
subplot(2,1,1);
stem(n,x(1:N));
xlabel('time');
ylabel('amplitude');
title('input sequence');
t=0:(N*l/m)-1;
subplot(2,1,2);
stem(t,y(1:N*l/m));
xlabel('time');
ylabel('amplitude');
title('input sampling sequence');
```

OUTPUT:-

```
enter the sample value30
enter up sampling factor7
enter down sampling factor2
```



RESULT:-

Thus sampling rate conversion by a rational form is performed using MATLAB.

EXERCISE PROGRAM:-

1. Write a matlab program to illustrate the effect of anti-aliasing filter?
2. Write a matlab program to illustration of upsampling?
3. Write a matlab program to illustration of downsampling?
4. Write a matlab program to illustration of effect of upsampling in frequency domain?
5. Write a matlab program to illustration of effect of downsampling in frequency domain?
6. Write a matlab program to illustrate the concept of aliasing?
7. Write a matlab program to plot magnitude response of comb filter?
8. Write a matlab program to plot magnitude response of allpass filter?
9. Write a matlab program to plot magnitude & unwrapped phase response of two transfer functins?
10. Write a matlab program to design a filter that eliminates high frequency component in a CT signal?
11. Write a matlab program to illustrate the effect of aliasing filter?

12. Write a matlab program to illustration of I/D sampling with sampling factor 5 ?
13. Write a matlab program to illustration of I/D sampling with sampling factor 5 ?
14. Write a matlab program to illustration of effect of I/D sampling in time domain?
15. Write a matlab program to illustration of effect of I/D sampling in time domain?
16. Write a matlab program to illustration of I/D sampling with sampling factor10?
17. Write a matlab program to illustration of I/D sampling with sampling factor 15 ?
18. Write a matlab program to illustration of I/D sampling with sampling factor 20?
19. Write a matlab program to illustration of I/D sampling with sampling factor 20?
20. Write a matlab program to illustration of I/D sampling with sampling factor 20?

VIVA QUESTIONS:-

1. What is multi rate signal processing?
2. What is the need for anti-imaging filter after up sampling a signal?
3. What is the need for anti-imaging filter prior to down sampling?
4. Define down sampling?

5. What is meant by up sampling?
6. Give any practical application of decimation?
7. Which signals can be downsampled?
8. What happens if I violate the Nyquist criteria in down sampling or decimating?
9. Can we do decimate in multiple stages?
10. What are "decimation" and "downsampling"?
11. What isthe "decimation factor
12. How does polyphase filtering save computations in an interpolation filter?
13. Why do we need I&Q signals?
14. What is Interpolation and decimation filters and why we need it?
15. What are "upsampling" and "interpolation"?
16. Why interpolate,i needed for any signal/sequence?
17. What is the "interpolation factor"?
18. Which signals can be interpolated?
19. Can interpolate will happens in multiple stages? If yes give reason?
20. How is the non-periodic nature of the input signal handled?
21. What is "bit reversal"?
22. What do you mean by phase spectrum and magnitude spectrum/ give comparison?
23. How do you reduce spectral leakage?
24. What do you mean by spectral resolution?
25. What is FIR and IIR filter define, and distinguish between these two?
26. What is window method? How you will design an FIR filter using window method?
27. What are low-pass and band-pass filter and what is the difference between these two?
28. What is the matlab command for Hamming window? Explain.
29. What do you mean by built in function ‘abs’ and where it is used?
30. Explain how the FIR filter are stable?

Interpolation is a technique for obtaining new unknown data points within the range of discrete known data points and is often used to recover an image from its downsampled version, or to simply perform image expansion.

It was originally developed for processing broadband, low frequency geophysical data in real time using a small digital processor. Sine and Cosine transforms are applied to sequences of data produced by successively applying a low pass digital filter and decimation by two operator to the original data. The resulting spectra are the average of estimates which are independent in time and represent constant percentage bandwidths. Thus, the technique is particularly well suited to applications where it is desireable to reject intervals of bad data or those which are naturally expressed on a log-frequency scale.

EXPERMENT NO-15

AUDIO APPLICATIONS

AIM: -

To perform audio applications such as to plot a time & frequency display of microphone plus a cosine using DSP.

EQUIPMENTS:-

1. Operating system-windows XP.
2. Software –cc studio 3.
3. Software-mat lab 6.5.
4. DSK 6713DSP Trainer kit.
5. USB cable.
6. Power Supply.

PROGRAM:-

Spectrogram-rdts-mtl-frequency analysis of signals using RDTX-MATLAB

```
#include"dsk6713-aic23.h"                          //code-DSK support file
Unit32fs=dsk6713-aic23freq-8Khz;                   //set sampling rate
#include<rtdh.h>                                   //RTDX support file
#include<math.h>
#include"hamming.cof"                              //Hamming window coefficients
#define pts256                                     //# of points for FFT
#define pi3.1415926538979
typedef struct {flot real,imag;}COMPLEX;
Void FFT(COMPLEX*Y,int n);                         //FFT protype
float iobuffer{PTS],iobuffer 1[PTS],a[PTS];        //input and output buffer
float  x[PTS];                                     // intermediate buffer
short I;                                           //general purpose index variable
int j,k,l,curr_block=0;                            // index variables
short buffercount=0;                               //number of new samples in iobuffer
short flag=0;                                      //set to 1 by ISR when buffer is full
COMPLEX w[PTS];                                    //twiddle constants stored in w
COMPLEX samples [PTS];                             //primary working buffer
RTDX_ Create output Channel(ochan);                //create output channel C6x->PC

Main()
{
For(i=0;i<PTS;i++)                                 //set up twiddle constants in w
{
w[i].real=cos(2*PI*i/512.0);                       //Re component of twiddle constants
w[i].imagl=sin(2*PI*i/512.0);                      //Im component of twiddle constants
}
Comm._intr();                                      //init DSK,code,McBSP
While (!RTDX_is Outpu Enabled(&ochan))             //wait forPC to enable RTDX
Puts("\n\n Wating…");                              //while waiting
for(1=0;1<256;I++)
a[I]=cos(2*3.14*1500*1/8000);
for (k=0;k<5000;k++)                               //infinite loop

{
While (flag==0);                                   //wait until iobuffer is full
Flag=0;                                            //reset flag
For(i=0;I<PTS;i++)                                 //swap buffer
{iobuffer1[i]=iobuffer[i]+a[i];
Samples[i].real=h[i]*iobuffer1[i];        //multiply by Hamming window coeffs
iobuffer 1[i]=x[i];                       // process frame to iobuffer
}
```

```
For(i=0;i<PTS;i++)
Samples[i].imag=0.0                    //imag components=0
FFT(samples,PTS);                         //call C-coded FFT function
For(i=0;i<PTS;i++)                       //compute square of FFT magnitude
{
X[i]=(samples[i].real*samp0les[i].real
   +samples[i].imag*samples[i].imag)/16;//FFT data scaling
}
RTDX_write(&ochan,x1,sizeof(x1)/2);                 //send 128 samples to PC
}                                                              //end of infinite loop
}                                                              // end of main
Interrupt void c_int11()                                  //ISR
{
Output_ sample ((short)(iobuffer[buffercount]));           //out fromiobuffer
Iobuffer[[buffercount++]=(short)(input_sample());          //input to iobuffer
If(buffercount>=PTS)                                        //if iobuffer full
{
       Buffercount=0;                                          /reinit buffercount
          Flag=1;                                              //reset flag
```

RESULT:-

Thus audio application is performed & spectrogram of an input signal is plotted using MATLAB.

EXPERMENT NO-16

NOISE REMOVAL

AIM: -

To add noise above 3kHz &then remove.

EQUIPMENTS:-

1. Operating system-windows XP.
2. Software –cc studio 3.
3. Software-mat lab 6.5.
4. DSK 6713DSP Trainer kit.
5. USB cable.
6. Power Supply.
7. CRO.

8. Function Generator.

THEORY:-

Adaptive filters are best used in cases where signal conditions or system parameters are slowly Changing and the filter are to be adjusted to compensate for the change. The least mean squares (LMS) criterion is such algorithm that can be used to provide the strategy for adjusting the filter coefficient.

.

PROGRAM:-

```
#include "NCefg.h"
#include"dsk6713.h"
#include" dsk6713_aic23.h"
#define beta IE-13                    //rate of convergence
#define N 30                          //adaptive FIR filter length-vary this parameter & observe
float delay [N];
float w[N};
DSK6713_AIC23_Config config={\
0x0017,          /*0 DSK6713_AIC23_LEFTINVOL Left line input channel volume*/ \
0x0017,          /*1 DSK6713_AIC23_RIGHTINVOL Right line input channel volume*/ \
0x00d8,          /*2 DSK6713_AIC23_LEFTINVOL Left channel headphone volume*/ \
0x00d8,          /*3 DSK6713_AIC23_RIGHTINVOL Right channel headphone volume*/
\
0x0011,          /*4 DSK6713_AIC23_ANAPATH Analog audio path control*/ \
0x0000,          /*5 DSK6713_AIC23_DIGPATH Digital audio path control*/ \
0x0000,          /*6 DSK6713_AIC23_POWERDOWN Power down control */ \
0x0043,          /*7DSK6713_AIC23_DIGIF Digital audio interface format*/ \
0x0081,          /*8DSK6713_AIC23_SAMPLERATE Sample rate control*/ \
0x0001,          /*9DSK6713_AIC23_DIGACT Digital interface activation */ \
};
        /*main()-Main code routine, Initializes BSL and generates tone*/
Void main()
{
DSK6713_AIC23_codecHandle hCodec;
Int I_input,r_input,I_output,r_output,T;
```

```
                         /*Initialize the board support library,must be called first*/
DSK6713_init();
hCodec=DSK6713_AIC23_openCodec(0,&config);           /*start the code C*/
DSK6713_AIC23_setfreq(hCOde c,1);
For(T=0:T<30:T++)                      //Initialize  the adaptive FIR Coeffs=0
{        w[T]=0;                       //init buffer for weights
Delay[T]=0;                           ////init buffer for delay samples
```

```
}
While(1)
{/*Read a sample to the left channel */
While(!DSK6713_AIC23_read(hCodec.&l_input));
{/*Read a sample to the right channel */
While(!DSK6713_AIC23_read(hCodec.&r_input));
l_output=(short int) adaptive_filter(l_input,r_input);
l_output=r_output;
While(!DSK6713_AIC23_write (hCodec.&l_output));          /*send output to the Left channel*/
While(!DSK6713_AIC23_write(hCodec.&r_output));           /*send output to the Right channel*/
}
\DSK6713_AIC23_closecodec(hcodec);                        /*close the codec*/
}
```

RESULT:-

Thus noise signal cancellation using adaptive filters is verified**.**

EXPERMENT NO-17

IMPULSE RESPONSE

AIM: -

To find the impulse response of the given equation
y(n)-y(n-1+0.9y(n-2)=x(n)

SOFTWARE REQURIED:-

1.MATLAB R2010a.
2.Windows XP SP2.

THEORY:-

Second order systems are the systems or networks which contain two or more storage elements and have describing equations that are second order differential equations.

The frequency response of second order filters is characterised by three filter parameters: the gain k, the corner frequency and the quality factor Q.

PROCEDURE:-

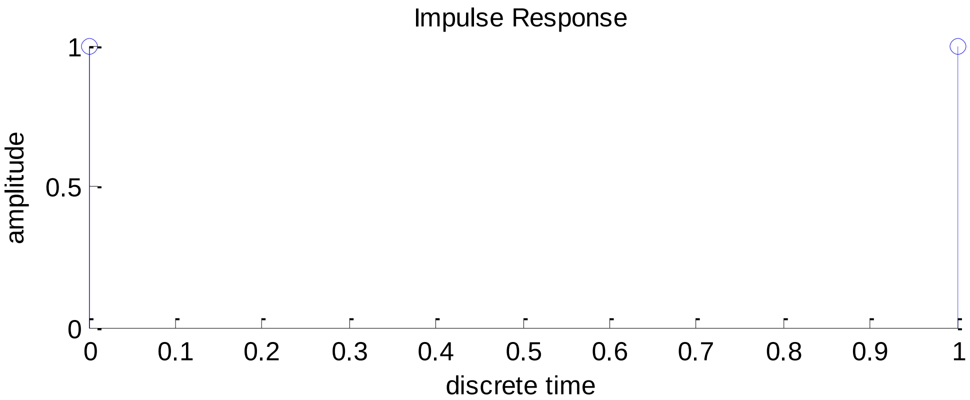
- Open MATLAB
- Open new M-file
- Type the program
- Save in current directory
- Compile and Run the program
- For the output see command window\ Figure window

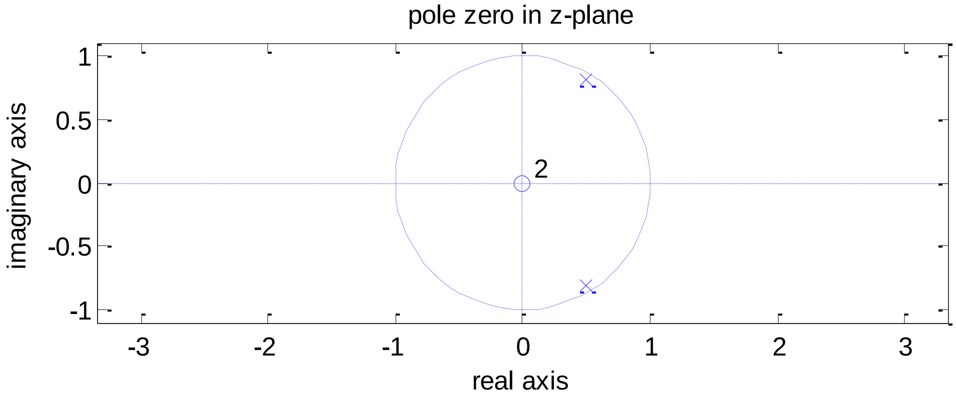
PROGRAM:-

```
%To find the impulse response of discrete time system
%y(n)-y(n-1)+0.9y(n-2)=xn)%
clc;
clear all;
close all;
b=input('Enter the coefficeints of x(n):b=');
a=input('Enter the coefficeints of x(n):a=');
N=input('Enter the order of N =');
h=impz(b,a,N);
n=0:N-1;
subplot(2,1,1);
stem(n,h);
xlabel('discrete time');
ylabel('amplitude');
title('Impulse Response');
subplot(2,1,2);
zplane(b,a);
xlabel('real axis');
ylabel('imaginary axis');
title('pole zero in z-plane');
```

OUTPUT:-

```
Enter the coefficeints of x(n):b=[1]
Enter the coefficeints of x(n):a=[1 -1 .9]
Enter the order of N =2
```





RESULT:-

Hence the impulse response of the given system is performed.

EXERCISE PROGRAM:-

1. Write a matlab program to find the frequency response of the following difference equation y(n)-7y(n-1)+9y(n-2)=3x(n)-2x(n-1)?
2. Write a matlab program to find the frequency response of the following difference equation 3y(n)+5y(n-1)=9x(n)?
3. Write a matlab program to find the frequency response of the following difference equation 9 y(n)-2y(n-1)+7y(n-2)-3y(n-3)=6x(n)+x(n-1)?
4. Write a matlab program to find the frequency response of the following difference equation 8y(n)+6y(n-1)=4x(n)+2x(n-1)?
5. Write a matlab program to find the frequency response of the following difference equation 3y(n)-8y(n-1)+9y(n-2)=9x(n)+5x(n-1) ?
6. Write a matlab program to find the frequency response of the following difference equation 6y(n)-5y(n-1)=9x(n)+5x(n-1) -7x(n-2)?

7. Write a matlab program to find the frequency response of the following difference equation 9y(n)-8y(n-1)+2y(n-2)=9x(n)-3x(n-1) ?
8. Write a matlab program to find the frequency response of the following difference equation 2y(n)-8y(n-1)=9x(n)+5x(n-1) ?
9. Write a matlab program to find the frequency response of the following difference equation 9y(n)-8y(n-1)+9y(n-2)=9x(n)+5x(n-1)-x(n-2) ?
10. Write a matlab program to find the frequency response of the following difference equation 3y(n)-8y(n-1)=7x(n)-3x(n-1) ?

VIVA QUESTIONS:-

1. INTRODUCTION TO DSP PROCESSORS

A signal can be defined as a function that conveys information, generally about the state or behavior of a physical system. There are two basic types of signals viz Analog (continuous time signals which are defined along a continuum of times) and Digital (discrete-time).

Remarkably, under reasonable constraints, a continuous time signal can be adequately represented by samples, obtaining discrete time signals. Thus digital signal processing is an ideal choice for anyone who needs the performance advantage of digital manipulation along with today's analog reality.

Hence a processor which is designed to perform the special operations(digital manipulations) on the digital signal within very less time can be called as a Digital signal processor. The difference between a DSP processor, conventional microprocessor and a microcontroller are listed below.

Microprocessor or General Purpose Processor such as Intel xx86 or Motorola 680xx family Contains - only CPU
-No RAM
-No ROM -
No I/O ports
-No Timer

Microcontroller such as 8051 family
Contains - CPU
- RAM
- ROM
-I/O ports
- Timer &
- Interrupt circuitry
Some Micro Controllers also contain A/D, D/A and Flash Memory

DSP Processors such as Texas instruments and Analog Devices

Contains
- CPU - RAM - ROM - I/O ports - Timer

Optimized for
- Fast arithmetic - Extended precision - Dual operand fetch - Zero overhead loop - Circular buffering

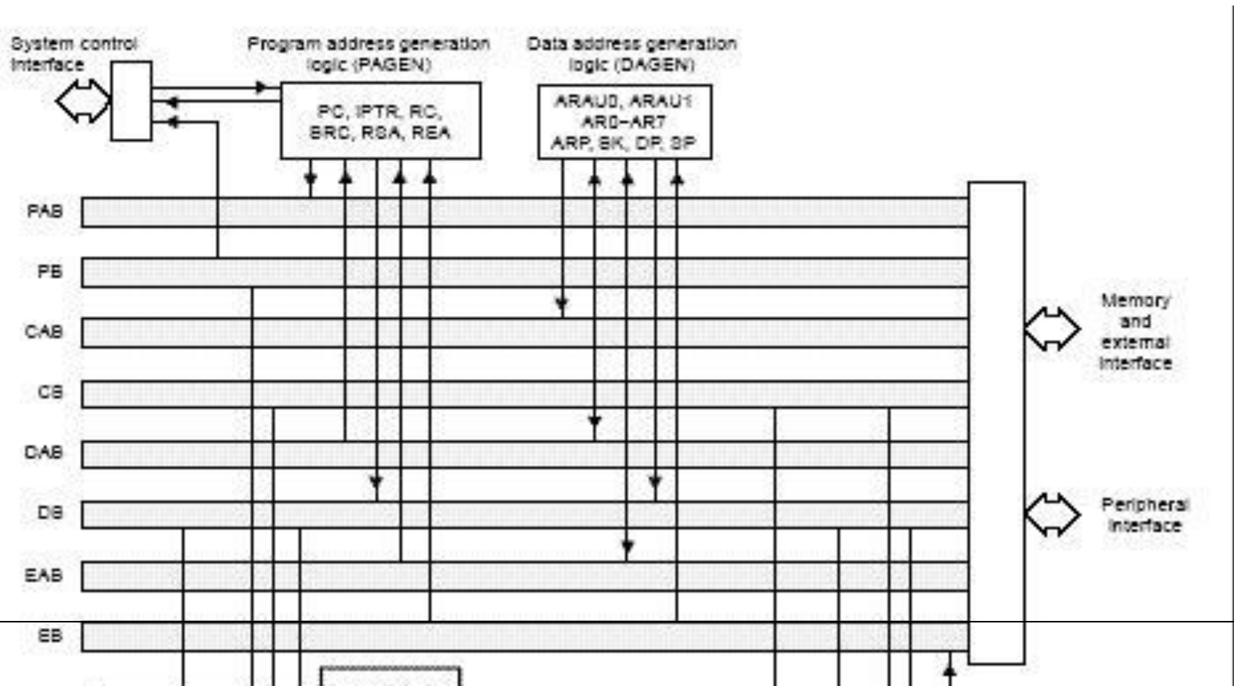
The basic features of a DSP Processor are

Feature	**Use**
Fast-Multiply accumulate	Most DSP algorithms, including filtering, transforms, etc. are multiplication- intensive
Multiple – access Memory Architecture	Many data-intensive DSP operations require reading a program instruction and multiple data items during each instruction cycle for best performance
Specialized addressing modes	Efficient handling of data arrays and first-in, first-out buffers in memory

Specialized program control	Efficient control of loops for many iterative DSP algorithms. Fast interrupt handling for frequent I/O operations.
On-chip peripherals and I/O Interfaces	On-chip peripherals like A/D converters allow for small low cost system designs. Similarly I/O interfaces tailored for common peripherals allow clean interfaces to off-chip I/O devices.

ARCHITECTUREOF6713 DSPPROCESSOR

This chapter provides an overview of the architectural structure of the TMS320C67xx DSP, which comprises the central processing unit (CPU), memory, and on-chip peripherals. The C67xE DSPs use an advanced modified Harvard architecture that maximizes processing power with eight buses. Separate program and data spaces allow simultaneous access to program instructions and data, providing a high degree of parallelism. For example, three reads and one write can be performed in a single cycle. Instructions with parallel store and application-specific instructions 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, the C67xx DSP includes the control mechanisms to manage interrupts, repeated operations, and function calling.



Bus Structure

The C67xx DSP architecture is built around eight major 16-bit buses (four program/data buses and four address buses):

_ The program bus (PB) carries the instruction code and immediate operands from program memory.

_ Three data buses (CB, DB, and EB) interconnect to various elements, such as the CPU, data address generation logic, program address generation logic, on-chip peripherals, and data memory.

_ The CB and DB carry the operands that are read from data memory.

_ The EB carries the data to be written to memory.

_ Four address buses (PAB, CAB, DAB, and EAB) carry the addresses needed for instruction execution.

The C67xx DSP can generate up to two data-memory addresses per cycle using the two auxiliary register arithmetic units (ARAU0 and ARAU1). The PB can carry data operands stored in program space (for instance, a coefficient table) to the multiplier and adder for multiply/accumulate operations or to a destination in data space for data move instructions (MVPD and READA). This capability, in conjunction with the feature of dual-operand read, supports the execution of single-cycle, 3-operand instructions such as the FIRS instruction. The C67xx DSP also has an on-chip bidirectional bus for accessing on-chip peripherals. This bus is connected to DB and EB through the bus exchanger in the CPU interface. Accesses that use this bus can require two or more cycles for reads and writes, depending on the peripheral’s structure.

Central Processing Unit (CPU)

The CPU is common to all C67xE devices. The C67x 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 C67x 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,
_ Two 16-bit words from data memory
_ 32-bit word from data memory
_ 40-bit word from either accumulator

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 for the other.

Barrel Shifter

The C67x 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).

2. VERIFY LINEAR CONVOLUTION

AIM: -

To perform linear convolution of two sequences

EQUIPMENT REQUIRED:-

Operating System – Windows XP
Constructor - Simulator
Software - CCStudio 3.3 & MATLAB 7.5

THEORY:-

Convolution is a formal mathematical operation, just as multiplication, addition, and integration. Addition takes two *numbers* and produces a third *number*, while convolution takes two *signals* and produces a third *signal*. Convolution is used in the mathematics of many fields, such as probability and statistics. In linear systems, convolution is used to describe the relationship between three signals of interest: the input signal, the impulse response, and the output signal.

$$y(n) = \sum_{k=0}^{N-1} x_1(k) x_2(n-k) \quad 0 < n < N-1 \qquad (1)$$

In this equation, x1(k), x2(n-k) and y(n) represent the input to and output from the system at time n. Here we could see that one of the input is shifted in time by a value every time it is multiplied with the other input signal. Linear Convolution is quite often used as a method of implementing filters of various types.

PROCEDURE:

1) Generate the first input sequence ‘x’.
2) Plot the sequence in discrete form. [Make use of stem( )]
3) Give some relevant names to x-axis and y-axis.
4) Generate second input sequence ‘y’.
5) Repeat steps (2) and (3) for second sequence ‘y’.
6) Use the inbuilt function ‘conv( )’ to compute linear convolution of ‘x’ and ‘y’.

   z = conv(x,y)

7) Plot the output sequence ‘z’. [Repeat steps 2 and 3]
8) Make use of subplot ( ) to plot the inputs and output sequences in a single window.

PROGRAM:

// Linear convolution program in c language using CCStudio

```
#include<stdio.h>
int x[15],h[15],y[15];
main()
{
int i,j,m,n;
printf("\n enter value for m");
scanf("%d",&m);
printf("\n enter value for n");
```

```
scanf("%d",&n);
printf("Enter values for i/p x(n):\n");
for(i=0;i<m;i++)
scanf("%d",&x[i]);
printf("Enter Values for i/p h(n)
\n"); for(i=0;i<n; i++)
scanf("%d",&h[i]);
// padding of zeros
for(i=m;i<=m+n-1;i++)
x[i]=0; for(i=n;i<=m+n-
1;i++)
h[i]=0;
/* convolution operation
*/ for(i=0;i<m+n-1;i++)
{
y[i]=0;
for(j=0;j<=i;j++)
{
y[i]=y[i]+(x[j]*h[i-j]);
}
}
//displaying the o/p
for(i=0;i<m+n-1;i++)
printf("\n The Value of output y[%d]=%d",i,y[i]);
```

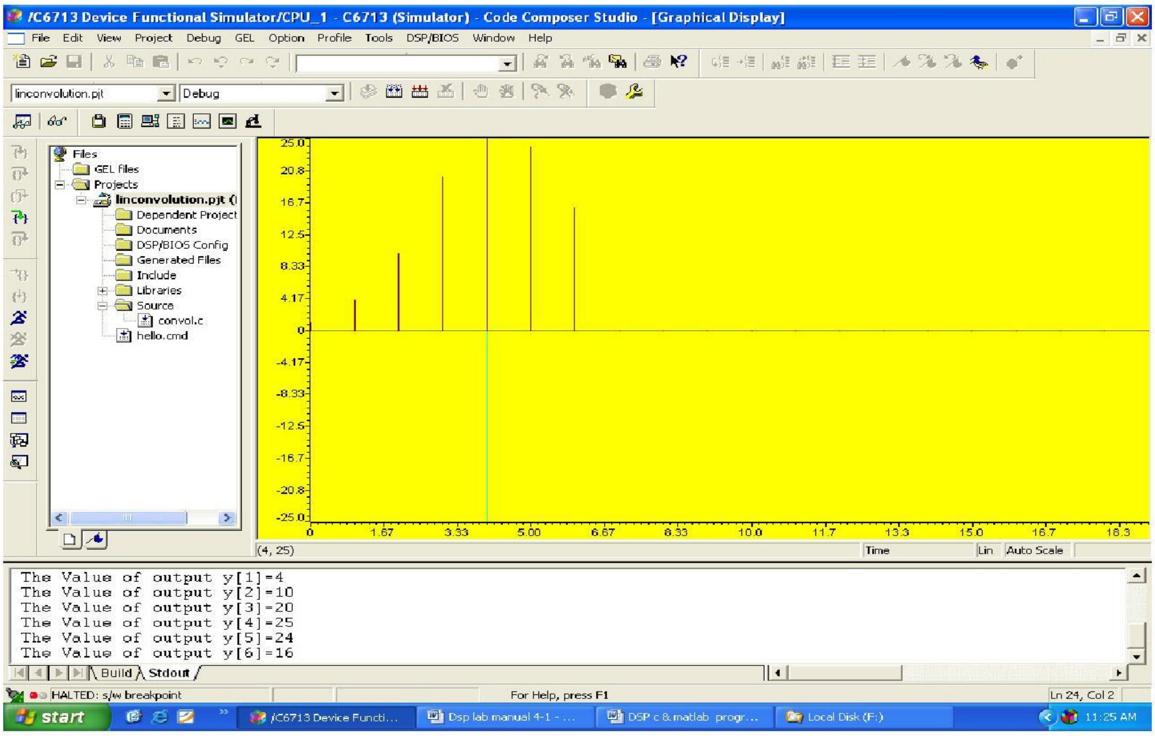
OUTPUT:

```
Enter value for m       4
Enter value for n       4
                        1 2 3
Enter values for i/p    4
                        1 2 3
Enter Values for n      4
The Value of output y[0]=1
The Value of output y[1]=4
The Value of output y[2]=10
The Value of output y[3]=20
```

The Value of output y[4]=25

The Value of output y[5]=24

The Value of output y[6]=16



RESULT:-

linear convolution of two sequences using CCStudio 3.3 is obtained.

VIVA QUESTIONS:-

1. What is the purpose of using convolution?
2. Give the formula for calculating linear convolution?
3. What are the properties of convolution?
4. What is meant by discrete convolution?
5. Define linear system and give example?

3. VERIFY CIRCULAR CONVOLUTION.

AIM:-

To perform circular convolution of two sequences using MATLAB & Code Composer Studio3.1.

EQUIPMENT REQUIRED:-

Operating System	– Windows XP
Constructor	- Simulator
Software	- CCStudio 3.3 & MATLAB 7.5

THEORY:-

Circular convolution is another way of finding the convolution sum of two input signals. It resembles the linear convolution, except that the sample values of one of the input signals is folded and right shifted before the convolution sum is found. Also note that circular convolution could also be found by taking the DFT of the two input signals and finding the product of the two frequency domain signals. The Inverse DFT of the product would give the output of the signal in the time domain which is the circular convolution output. The two input signals could have been of varying sample lengths. But we take the DFT of higher point, which ever signals levels to. For eg. If one of the signal is of length 256 and the other spans 51 samples, then we could only take

256 point DFT. So the output of IDFT would be containing 256 samples instead of 306 samples, which follows N1+N2 – 1 where N1 & N2 are the lengths 256 and 51 respectively of the two inputs. Thus the output which should have been 306 samples long is fitted into 256 samples. The
256 points end up being a distorted version of the correct signal. This process is called circular convolution.

PROCEDURE:-

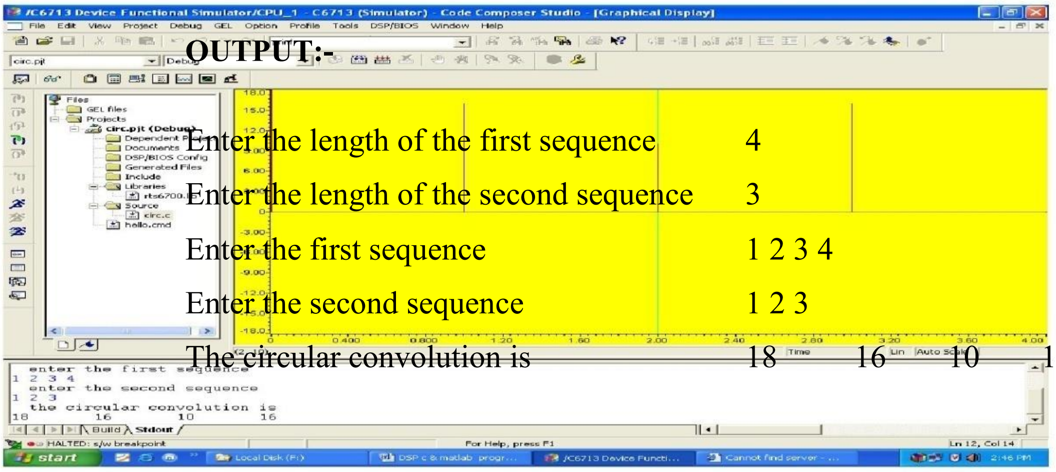
1. Generate the first input sequence ‘x’.
2. Plot the sequence in discrete form. [Make use of stem( )]
3. Give some relavent names to x-axis and y-axis.
4. Generate second input sequence ‘h’.
5. Repeat steps (2) and (3) for second sequence ‘h’.
6. Find out the length of first sequence and store it in variable ‘n1’. [Make use of length ( )]
7. Similarly find out the length of second sequence and store it in variable ‘n2’.
9. Make the length of smaller sequence equal to the larger sequence by padding zeros to it.
11. Display and plot the convolved sequence. [Repeat steps (2) and (3)]
12. Make use of subplot to plot the inputs and output sequences in a single window.
13. Compare theoretical and practical values.

PROGRAM USING CODE COMPOSER STUDIO 3.3

```
/* program to implement
circular convolution */
#include<stdio.h>
int
m,n,x[30],h[30],y[30],i,j,
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]);

}
```

OUTPUT:-



Enter the length of the first sequence 4

Enter the length of the second sequence 3

Enter the first sequence 1 2 3 4

Enter the second sequence 1 2 3

The circular convolution is 18 16 10 16

RESULT:-

Circular Convolution of two sequences using CC Studio3.3 is obtained.

VIVA QUESTIONS:-

1. What is the different between Circular and Linear convolution?
2. What is the function in MATLAB used for padding zeros to a sequence? If your sequence is, x = [1 2 3 4] and you want to pad zeros to it. How can you do that in MATLAB?
3. What is the use of following functions in MATLAB:
    i. length( )
    ii. max( )
    iii. min( )
4. Give the steps to get the result of linear convolution from the method of circular convolution?
5. What is the circular convolution?

4. DESIGN FIR FILTER (LP/HP) USING WINDOWING TECHNIQUE

FIR FILTERS (LP/HP) USING WINDOWING TECHNIQUE

a) Using Rectangular Window b) Using Bartlett Window c) Using Kaiser Window

AIM:-

To design of FIR(LP/HP) filters using rectangular window.

EQUIPMENT REQUIRED:-

Operating System – Windows XP Constructor
- Simulator

Software - CCStudio 3 & MATLAB 7.5

THEORY:

A Finite Impulse Response (FIR) filter is a discrete linear time-invariant system whose output is based on the weighted summation of a finite number of past inputs. An FIR transversal filter structure can be obtained directly from the equation for discrete-time convolution.

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

In this equation, x(k) and y(n) represent the input to and output from the filter at time n. h(n-k) is the transversal filter coefficients at time n. These coefficients are generated by using FDS (Filter Design Software or Digital filter design package).

FIR – filter is a finite impulse response filter. Order of the filter should be specified. Infinite response is truncated to get finite impulse response. placing a window of finite length does this. Types of windows available are Rectangular, Barlett, Hamming, Hanning, Blackmann window etc. This FIR filter is an all zero filter.

PROCEDURE:-

1. Enter the passband ripple (rp) and stopband ripple (rs).
2. Enter the passband frequency (fp) and stopband frequency (fs).
3. Enter the sampling frequency (f).
4. Calculate the analog passband edge frequency (wp) and stop band edge frequency (ws)
   wp=2*fp/f ws=2*fs/f
5. Calculate the order of the filter using the following formula,

$$n = \frac{(-20\log_{10}(rp.rs) - 13)}{(14.6\,(fs-fp)/f)}.$$

   [Use 'ceil( )' for rounding off the value of 'n' to the nearest integer] if 'n' is an odd number, then reduce its value by '1'.
6. Generate (n+1)th point window coefficients.For example boxcar(n+1) generates a rectangular window. y=boxcar(n+1)
7. Design an nth order FIR filter using the previously generated (n+1) length window function. b=fir1(n,wp,y)

8. Find the frequency response of the filter by using 'freqz( )' function. [h,o]=freqz(b,a,k) This function returns k-point complex frequency response vector 'h' and k-point frequency vector 'o' in radians/samples of the filter.

$$H(e^{iw}) = \frac{B(e^{jw})}{A(e^{jw})} = \frac{b(1)+b(2)e^{-jw}+\ldots\ldots\ldots\ldots b(m+1)e^{-jmw}}{a(1)+a(2)e^{-jw}+\ldots\ldots\ldots\ldots a(n+1)e^{-jnw}}$$

Where a, b are vectors containing the denominator and numerator coefficients. Here a=1.

9. Calculate the magnitude of the frequency response in decibels (dB). m= $20*\log_{10}(abs(h))$
10. Plot the magnitude response [magnitude in dB Vs normalized frequency (o/pi)]
11. Give relevant names to x- and y- axes and give an appropriate title for the plot.

PROGRAM USING CODE COMPOSER STUDIO 3.3:

```
#include<stdio.h>
#include<math.h>
#define pi 3.1415 int
n,N,c; float
wr[64],wt[64];
void main()
{
printf("\n enter no. of
samples,N= :");
scanf("%d",&N);

printf("\n enter choice of window function\n 1.rect \n 2. triang
\n c= :"); scanf("%d",&c);
printf("\n elements of window function
are:"); switch(c)
{
case 1:
for(n=0;n<=N-
1;n++)
{
wr[n]=1;
printf(" \n wr[%d]=%f",n,wr[n]);
}
break
; case
2:
for(n=0;n<=N-1;n++)
{
wt[n]=1-(2*(float)n/(N-1));
printf("\n
wt[%d]=%f",n,wt[n]);
}
break;
}
}
```

RESULT: -

By using windowing techniques (Bartlett, Rectangular, Blackman) the filters are designed.

VIVA QUESTIONS:-

1. What are the uses of function ceil and for?
2. Define boxcar
3. Define Kaiser
4. Define Bartlett
5. What is an FIR system? Compare FIR and IIR system?

5. IMPLEMENT IIR FILTER (LOW PASS & HIGH PASS)

IIR FILTER (LP/HP) USING BUTTER WORTH (ANLOG & DIGITAL) FILTERS & CHEBYSHEV (DIGITAL) TYPE – I & II FILTERS

AIM:-

To design of Butterworth Digital (low pass & high pass) filter.

EQUIPMENTS:-

Operating System – Windows XP Constructor
- Simulator

Software - CCStudio 3 & MATLAB 7.5

THEORY:-

The IIR filter can realize both the poles and zeroes of a system because it has a rational transfer function, described by polynomials in z in both the numerator and the denominator:

IIR filters can be expanded as infinite impulse response filters. In designing IIR filters, cutoff frequencies of the filters should be mentioned. The order of the filter can be estimated using butter worth polynomial. That's why the filters are named as butter worth filters. Filter coefficients can be found and the response can be plotted.

PROCEDURE:

1. Enter the pass band ripple (rp) and stop band ripple (rs).
2. Enter the pass band frequency (fp) and stop band frequency (fs).
3. Get the sampling frequency (f).
4. Calculate the analog pass band edge frequencies, w1 and w2.
   w1 = 2*fp/f w2 = 2*fs/f
5. Calculate the order and 3dB cutoff frequency of the analog filter. [Make use of the following function] [n,wn]=buttord(w1,w2,rp,rs,'s')
6. Design an nth order analog lowpass Butter worth filter using the following statement.
   [b,a]=butter(n,wn,'s')
7. Find the complex frequency response of the filter by using 'freqs( )' function

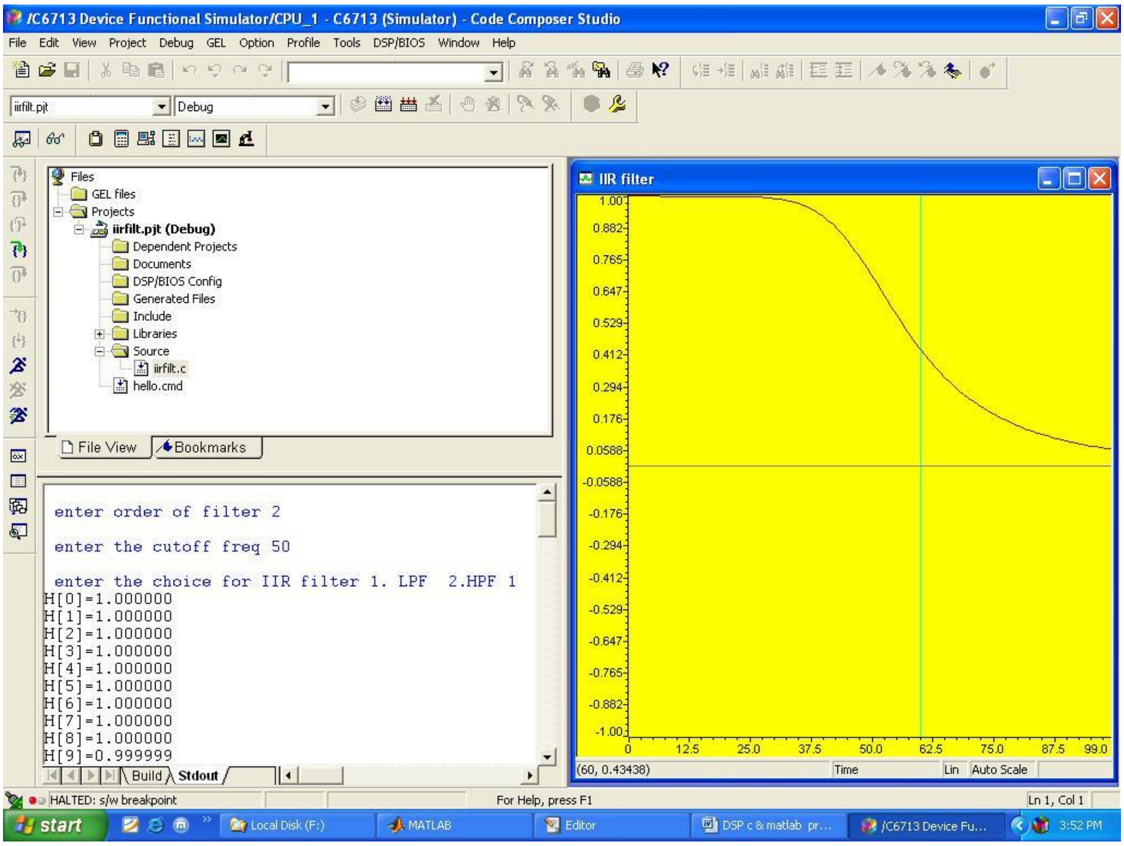
PROGRAM USING CODE COMPOSER STUDIO 3.3:

```
//iirfilters
#include<stdio.h>
#include<math.h>
int
i,w,wc,c,N
; float
```

```
H[100];
float
mul(float,
int);
void main()
{
printf("\n
enter order of
filter ");
scanf("%d",&
N);
printf("\n enter
the cutoff freq
");
scanf("%d",&
wc);
printf("\n enter the choice for IIR
filter 1. LPF 2.HPF ");
scanf("%d",&c);
switch(c)
{
case 1:
        for(w=0;w<100;w++)
        {
        H[w]=1/sqrt(1+mul((w/(float)wc),2*N));
        printf("H[%d]=%f\n",w,H[w]);
        }
        break;
case 2:
        for(w=0;w<=100;w++)
        {
        H[w]=1/sqrt(1+mul((float)wc/w,2*N));
        printf("H[%d]=%f\n",w,H[w]);
        }
        break;
}
```

```
}
float mul(float a,int x)
{
for(i=0;i<x-1;i++)
a*=a;
return(a);
}
```

OUTPUT:-



RESULT: -

Thus IIR (LP/HP) filter is designed using low pass Butterworth and chebyshev filters technique and verified using the DSP processor.

VIVA QUESTIONS:-

1. What are the properties of chebyshev filter?
2. Define signal flow graph?
3. Draw the signal flow graph of first order digital filter?
4. What is advantage of cascade realization?
5. What is the main disadvantage of direct-form realization?