DIGITATL SIGNAL PROCESSING LABORATORY MANUAL

B.Tech

(III Year-II Sem)

(2020-2021)

Prepared by

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DEPARTMENT OF ELECTRONICS AND COMMUNICATIONS ENGG.

MALLA REDDY COLLEGE OF ENGINEERING & TECHNOLOGY

(Autonomous Institution – UGC, Govt. of India)

(Affiliated to JNTU, Hyderabad, Approved by AICTE - Accredited by NBA & NAAC – 'A' Grade - ISO 9001:2015 Certified)

Maisammaguda, Dhulapally (Kompally), Secunderabad – 500100, Telangana State, India

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ELECTRONICS & COMMUNICATION ENGINEERING

VISION

To evolve into a center of excellence in Engineering Technology through creative and innovative practices in teaching-learning, promoting academic achievement & research excellence to produce internationally accepted competitive and world class professionals.

MISSION

To provide high quality academic programmes, training activities, research facilities and opportunities supported by continuous industry institute interaction aimed at employability, entrepreneurship, leadership and research aptitude among students.

QUALITY POLICY

- **Impart up-to-date knowledge to the students in Electronics & Communication area to make them quality engineers.**
- **Make the students experience the applications on quality equipment and tools.**
- **Provide systems, resources and training opportunities to achieve continuous improvement.**
- **Maintain global standards in education, training and services.**

PROGRAMME EDUCATIONAL OBJECTIVES (PEOs)

PEO1: PROFESSIONALISM & CITIZENSHIP

To create and sustain a community of learning in which students acquire knowledge and learn to apply it professionally with due consideration for ethical, ecological and economic issues.

PEO2: TECHNICAL ACCOMPLISHMENTS

To provide knowledge based services to satisfy the needs of society and the industry by providing hands on experience in various technologies in core field.

PEO3: INVENTION, INNOVATION AND CREATIVITY

To make the students to design, experiment, analyze, interpret in the core field with the help of other multi disciplinary concepts wherever applicable.

PEO4: PROFESSIONAL DEVELOPMENT

To educate the students to disseminate research findings with good soft skills and become a successful entrepreneur.

PEO5: HUMAN RESOURCE DEVELOPMENT

To graduate the students in building national capabilities in technology, education and research.

PROGRAMME SPECIFIC OBJECTIVES (PSOs)

PSO1

To develop a student community who acquire knowledge by ethical learning and fulfill the societal and industry needs in various technologies of core field.

PSO2

To nurture the students in designing, analyzing and interpreting required in research and development with exposure in multi disciplinary technologies in order to mould them as successful industry ready engineers/entrepreneurs

PSO3

To empower students with all round capabilities who will be useful in making nation strong in technology, education and research domains.

PROGRAM OUTCOMES (POs)

Engineering Graduates will be able to:

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

MALLA REDDY COLLEGE OF ENGINEERING AND TECHNOLOGY

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III Year B.Tech. ECE-II Sem	**-**	**/ 3 /-**	**2**

(R18A0488) DIGITAL SIGNAL PROCESSING LAB

Note:

1. Minimum of 12 experiments has to be conducted.

2. The programs shall be implemented in software (Using MATLAB / Lab view / C programming/ Equivalent) and hardware (Using TI / Analog devices / Motorola / Equivalent DSP processors).

List of Experiments:

1. To find DFT / IDFT of given DT signal
2. Program to obtain Linear Convolution of two finite length sequences
3. Program for computing Auto-correlation.
4. To find frequency response of a given system (in Transfer Function/ Differential equation form).
5. Implementation of FFT of given sequence
6. Determination of power spectrum of a given signal(s).
7. Implementation of LP FIR filter for given sequence
8. Implementation of HP FIR filter for given sequence
9. Implementation of LP IIR filter for given sequence
10. Implementation of HP IIR filter for given sequence
11. Generation of sinusoidal signal through filtering
12. Generation of DTMF signals
13. Implementation of Decimation Process
14. Implementation of Interpolation Process
15. Implementation of I/D sampling rate converters.
16. Audio application such as to plot a time and frequency display of microphone plus a cosine using DSP. Read a .wav file and match with their respective spectrograms.
17. Impulse response of first order and second order systems.

COURSE OUTCOMES

1. Generate & Perform different operations on discrete time signals and systems.

2. Analyze and implement digital systems using the Discrete Fourier Transform and Fast Fourier Transform (FFT) techniques using MATLAB and signal processing toolboxes.

3. Use Z transforms to analyze a digital system finding the region of convergence using MATLAB and signal processing toolboxes.

4 Design and Implement digital FIR and IIR filters.

5 Design Up converter, down converter & Sample rate converter.

CONTENTS

PART A - LIST OF EXPERIMENTS USING MATLAB

INTRODUCTION TO MATLAB

MATLAB (MATrix LABoratory):

MATLAB is a software package for high-performance language for technical computing. It integrates computation, visualization, and programming in an easy-to-use environment where problems and solutions are expressed in familiar mathematical notation. Typical uses include the following

- Math and computation
- Algorithm development
- Data acquisition
- Modeling, simulation, and prototyping
- Data analysis, exploration, and visualization
- Scientific and engineering graphics
- Application development, including graphical user interface building

The name MATLAB stands for matrix laboratory. MATLAB was originally written to provide easy access to matrix software developed by the LINPACK and EISPACK projects. Today, MATLAB engines incorporate the LAPACK and BLAS libraries, embedding the state of the art in software for matrix computation.

MATLAB has evolved over a period of years with input from many users. In university environments, it is the standard instructional tool for introductory and advanced courses in mathematics, engineering, and science. In industry, MATLAB is the tool of choice for high-productivity research, development, and analysis.

MATLAB features a family of add-on application-specific solutions called toolboxes. Very important to most users of MATLAB, toolboxes allow learning and applying specialized technology. Toolboxes are comprehensive collections of MATLAB functions (M-files) that extend the MATLAB environment to solve particular classes of problems. Areas in which toolboxes are available include Image processing, signal processing, control systems, neural networks, fuzzy logic, wavelets, simulation, and many others.

The main features of MATLAB

1. Advance algorithm for high performance numerical computation ,especially in the Field matrix algebra
2. A large collection of predefined mathematical functions and the ability to define one’s own functions.
3. Two-and three dimensional graphics for plotting and displaying data
4. A complete online help system
5. Powerful, matrix or vector oriented high level programming language for individual applications.
6. Toolboxes available for solving advanced problems in several application areas

EXP.NO: 1
TO FIND DFT / IDFT OF GIVEN DT SIGNAL

AIM: To find Discrete Fourier Transform and Inverse Discrete Fourier Transform of given digital signal.

Software: MATLAB

THEORY:

Basic equation to find the DFT of a sequence is given below.

$$X(k) = \sum_{n=0}^{N-1} x(n) W_N^{nk}$$

$$where\ \ W_N^{nk} = e^{-j\frac{2\pi nk}{N}} \text{ [TWIDDLE FACTOR]}$$

Basic equation to find the IDFT of a sequence is given below.

$$x_n = \frac{1}{N}\sum_{k=0}^{N-1} X_k e^{\frac{2\pi i}{N}kn} \qquad n = 0, \ldots, N-1.$$

Algorithm:

Step I: Get the input sequence.

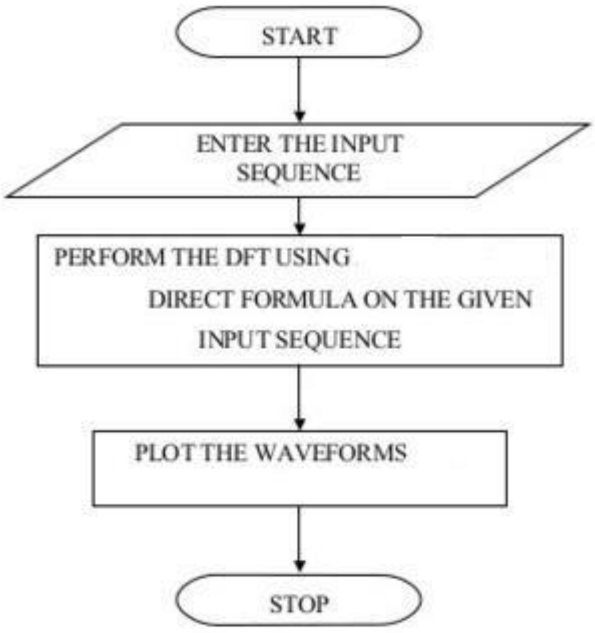
Step II: Find the DFT of the input sequence using direct equation of DFT.

Step III: Find the IDFT using the direct equation.

Step IV: Plot DFT and IDFT of the given sequence using matlab command stem.

Step V: Display the above outputs.

Flow chart:


START
ENTER THE INPUT SEQUENCE
PERFORM THE DFT USING DIRECT FORMULA ON THE GIVEN INPUT SEQUENCE
PLOT THE WAVEFORMS
STOP

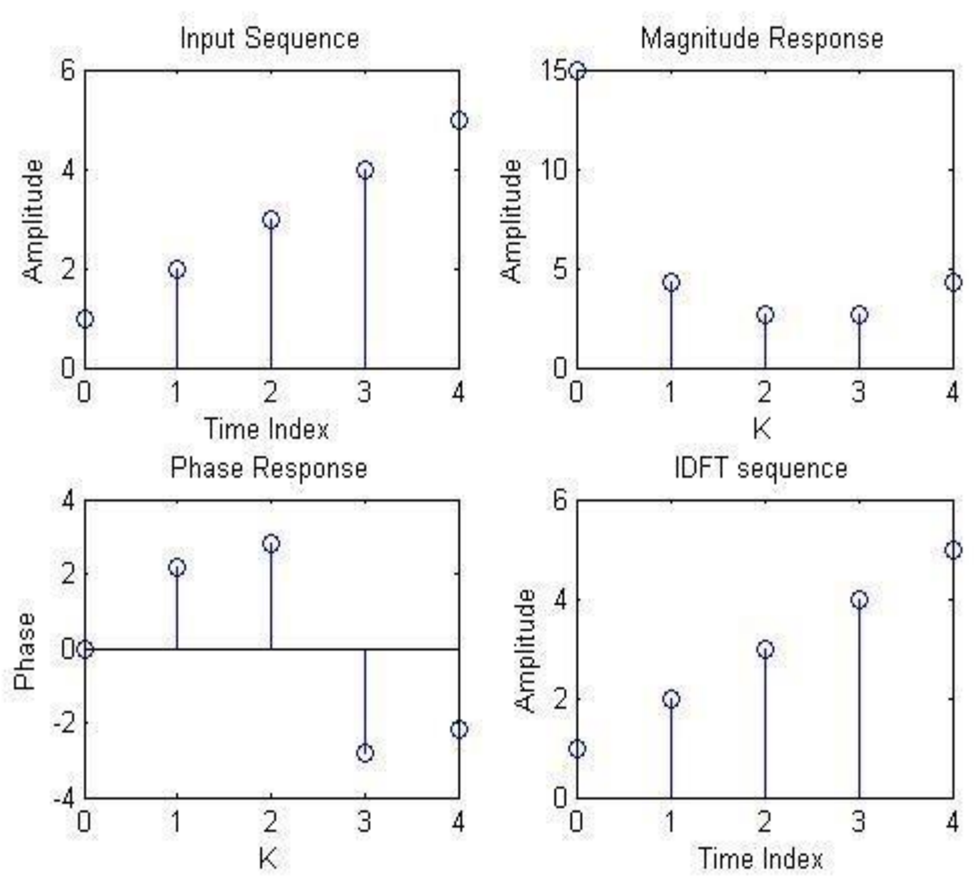
PROGRAM:

```
clc;
close all;
clear all;
xn=input('Enter the sequence x(n)'); %Get the sequence from user
ln=length(xn); %find the length of the sequence
xk=zeros(1,ln); %initialize an array of same size as that of input sequence
ixk=zeros(1,ln); %initialize an array of same size as that of input sequence
 %DFT of the sequence
 %-----------------------------------------------------
for k=0:ln-1
    for n=0:ln-1
        xk(k+1)=xk(k+1)+(xn(n+1)*exp((-i)*2*pi*k*n/ln));
    end
end
%-----------------------------------------------------
 %Plotting input sequence
 %-----------------------------------------------------
t=0:ln-1;
subplot(221);
stem(t,xn);
ylabel ('Amplitude');
xlabel ('Time Index');
title('Input Sequence');
%---------------------------------------------------------------
```

```
magnitude=abs(xk); % Find the magnitudes of individual DFT points
% plot the magnitude response
 %-----------------------------------------------------------
t=0:ln-1;
subplot(222);
stem(t,magnitude);
ylabel ('Amplitude');
xlabel ('K');
title('Magnitude Response');
%-----------------------------------------------------------
 phase=angle(xk); % Find the phases of individual DFT points % plot the magnitude
sequence
%-----------------------------------------------------------
t=0:ln-1;
subplot(223);
stem(t,phase);
ylabel ('Phase');
xlabel ('K');
title ('Phase Response');

%-----------------------------------------------------------
 %IDFT of the sequence
 %-----------------------------------------------------------
for n=0:ln-1
    for k=0:ln-1
        ixk(n+1)=ixk(n+1)+(xk(k+1)*exp(i*2*pi*k*n/ln));
    end
end
 ixk=ixk./ln;
%-----------------------------------------------------------
 %code block to plot the input sequence
 %-----------------------------------------------------------
t=0:ln-1;
subplot(224);
stem(t,ixk);
ylabel ('Amplitude');
xlabel ('Time Index');
title ('IDFT sequence');
%-----------------------------------------------------
```

Expected Waveforms:

Xn=[1 2 3 4 5]
Xk = 15,-2.50+3.44i,-2.50+0.81i,-2.49-0.81i,-2.49-3.44i



VIVA QUESTIONS:

1. Define signal, Give Examples for 1-D, 2-D, 3-D signals.
2. Define transform. What is the need for transform?
3. Differentiate Fourier transform and discrete Fourier transform.
4. Differentiate DFT and DTFT
5. Explain mathematical formula for calculation of DFT.
6. Explain mathematical formula for calculation of IDFT.
7. How to calculate FT for 1-D signal?
8. What is meant by magnitude plot, phase plot, power spectrum?
9. Explain the applications of DFT.
10. What are separable transforms?

Exercise:

1. Find 8-point DFT of the sequence x (n) = [1 2 3 4 4 3 2 1]
2. Find 4-point DFT of the sequence x (n) = [1,-2,3,2]

OBSERVATIONS

Output Waveforms:

EXP.NO: 2
LINEAR CONVOLUTION OF TWO SEQUENCES

AIM: To obtain Linear Convolution of two finite length sequences

Software: MATLAB

THEORY:

Convolution is a mathematical operation used to express the relation between input and output of an LTI system. It relates input, output and impulse response of an LTI system as

$$y(n)=x(n)*h(n)$$

Where y (n) = output of LTI

x (n) = input of LTI

h (n) = impulse response of LTI

Discrete Convolution

$$y(n)=x(n)*h(n)$$

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

By using convolution we can find zero state response of the system.

Algorithm:

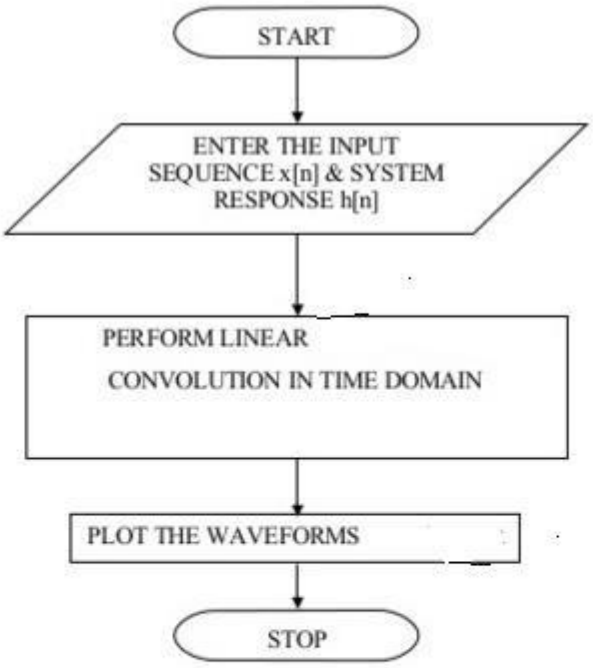
Step I: Give input sequence x[n].

Step II: Give impulse response sequence h(n)

Step III: Find the convolution y[n] using the matlab command conv.

Step IV: Plot x[n],h[n],y[n].

FLOWCHART:


START
ENTER THE INPUT SEQUENCE x[n] & SYSTEM RESPONSE h[n]
PERFORM LINEAR CONVOLUTION IN TIME DOMAIN
PLOT THE WAVEFORMS
STOP

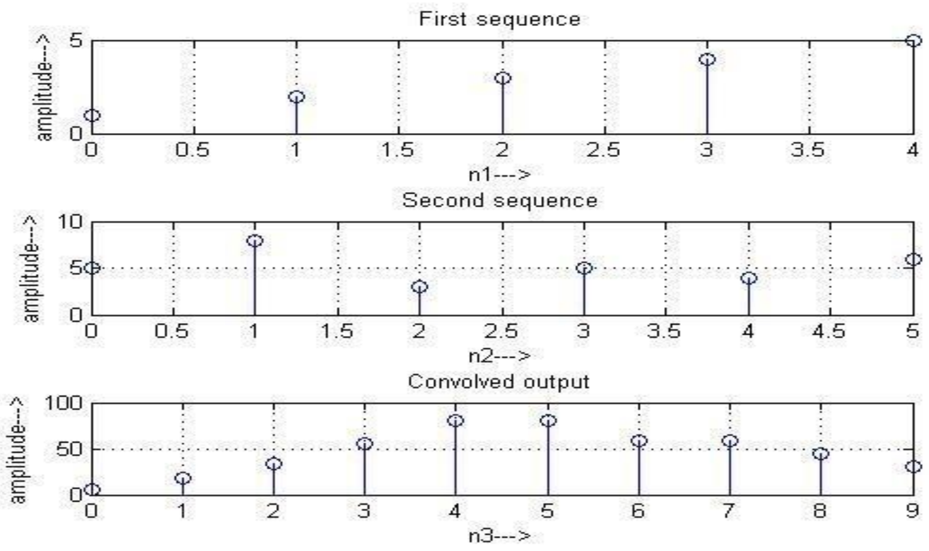
PROGRAM:

```
clc;
clear all;
close all;
x1=input('Enter the first sequence x1(n) = ');
x2=input('Enter the second sequence x2(n) = ');
L=length(x1);
M=length(x2);
N=L+M-1;
yn=conv(x1,x2);
disp(‘The values of yn are= ‘);
disp(yn);

n1=0:L-1;
subplot(311);
stem(n1,x1);
grid on;
xlabel('n1--->');
ylabel('amplitude--->');
title('First sequence');

n2=0:M-1;
subplot(312);
stem(n2,x2);
grid on;
xlabel('n2--->');
ylabel('amplitude--->');
```

```
title('Second sequence');

n3=0:N-1;
subplot(313);
stem(n3,yn);
grid on;
xlabel('n3--->');
ylabel('amplitude--->');
title('Convolved output');
```

Expected Waveforms:

```
Enter the first sequence x1(n) = [1 2 3 4 5]
Enter the second sequence x2(n) = [5 8 3 5 4 6]
The values of yn are=

    5    18    34    55    80    81   59    59    44    30
```



VIVA QUESTIONS:

1. Explain the significance of convolution.
2. Define linear convolution.
3. Why linear convolution is called as a periodic convolution?
4. Why zero padding is used in linear convolution?
5. What are the four steps to find linear convolution?
6. What is the length of the resultant sequence in linear convolution?
7. How linear convolution will be used in calculation of LTI system response?
8. List few applications of linear convolution in LTI system design.
9. Give the properties of linear convolution.
10. How the linear convolution will be used to calculate the DFT of a signal?

Exercise:

1. Find the linear convolution of x(n)=[7 5 4 0] and h(n)=[0 3 6 2 9]
2. Find the linear convolution of x(n)=[1 -3 0 5 6] and h(n)=[2 4 6 -2 -3]

OBSERVATIONS

Output Waveforms:

EXP.NO: 3
AUTO CORRELATION

AIM: To compute auto correlation between two sequences

SOFTWARE: MATLAB

THEORY:

Auto Correlation Function

Auto correlation function is a measure of similarity between a signal & its time delayed version. It is represented with R(k).

The auto correlation function of x(n) is given by

$$R11(k)=R(k)= \sum_{n=-\infty}^{\infty} x_1(n)x_2(n-k)$$

Algorithm:

Step I : Give input sequence x[n].

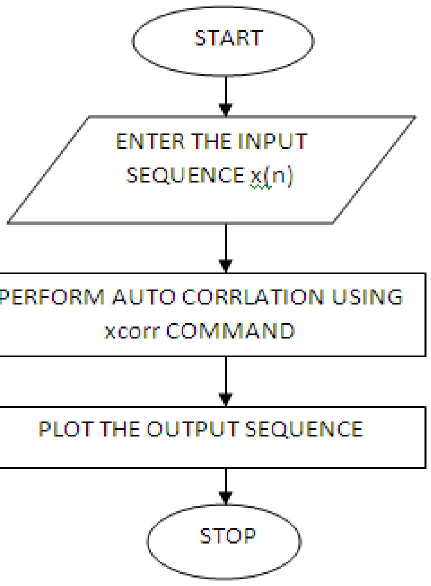
Step II : Give impulse response sequence h (n)

Step III: Find auto correlation using the matlab command xcorr.

Step IV: Plot x[n], h[n], z[n].

Step V: Display the results

Flow Chart:


START
ENTER THE INPUT SEQUENCE x(n)
PERFORM AUTO CORRLATION USING xcorr COMMAND
PLOT THE OUTPUT SEQUENCE
STOP

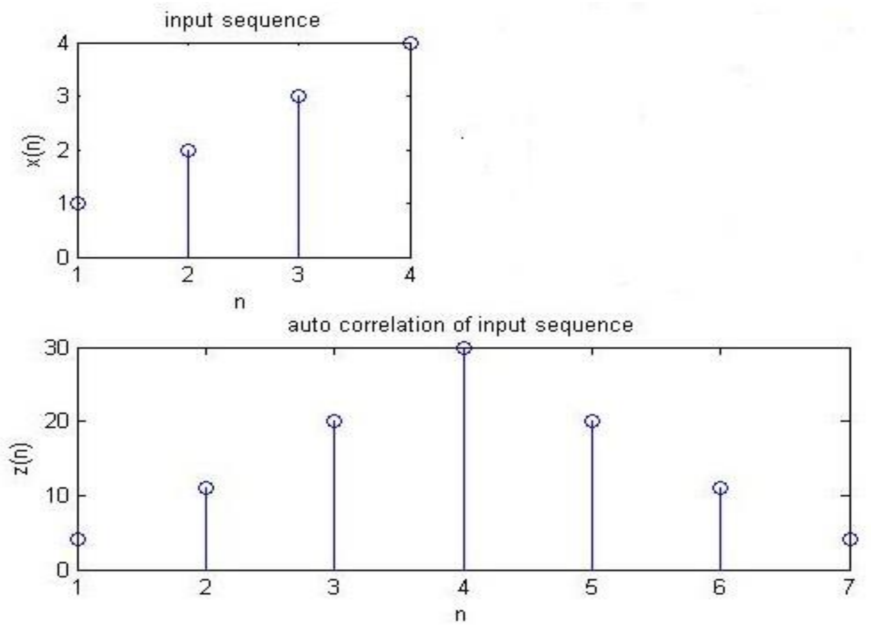
PROGRAM:

```
clc;
close all;
clear all;
% two input sequences
x=input('enter input sequence')
subplot(1,2,1);
stem(x);
xlabel('n');
ylabel('x(n)');
title('input sequence');
% auto correlation of input sequence
z=xcorr(x,x);
disp(‘The values of z are = ‘);disp(z);
subplot(1,2,2);
stem(z);
xlabel('n');
ylabel('z(n)');
title('auto correlation of input sequence');
```

Expected Wave forms:

Enter input sequence [1 2 3 4]

z = [3.99, 11, 20, 30, 20, 11, 3.99]



VIVA QUESTIONS:

1. Write mathematical formula to find auto correlation?
2. Define auto correlation?
3. Define correlation
4. Difference between Auto correlation and convolution?
5. Difference between Auto correlation and cross correlation?
6. Write mathematical formula to find cross correlation?
7. What is the length of the resultant sequence of auto correlation?
8. List few applications of correlation.
9. Give the properties of auto correlation.
10. Define cross correlation?

Exercise:

1. Find the auto correlation function of ramp sequence for $0 \le n \le 6$.
2. Find the auto correlation function of the input sequence x(n)=[4 5 6 8 2 3 4]

OBSERVATIONS

Output Waveforms:

EXP.NO: 4
TO FIND FREQUENCY RESPONSE OF A GIVEN SYSTEM GIVEN IN (TRANSFER FUNCTION/ DIFFERENTIAL EQUATION FORM)

AIM: To find frequency response of a given system in differential equation form.

Software: MATLAB

THEORY:

Systems respond differently to inputs of different frequencies. Some systems may amplify components of certain frequencies, and attenuate components of other frequencies. The way that the system output is related to the system input for different frequencies is called the *frequency response* of the system.

Since the frequency response is a complex function, we can convert it to polar notation in the complex plane. This will give us a magnitude and an angle. We call the angle the *phase*.

Amplitude Response:

For each frequency, the magnitude represents the system's tendency to amplify or attenuate the input signal.

$$A(\omega) = |H(j\omega)|$$

Phase Response:

The phase represents the system's tendency to modify the phase of the input sinusoids.

$$\phi(\omega) = \angle H(j\omega)$$

The phase response, or its derivative the group delay, tells us how the system delays the input signal as a function of frequency.

Given Difference equation is $y(n) - \frac{1}{6}y(n-1) - \frac{1}{6}y(n-2) = x(n)$

Algorithm:

Step I : Give numerator coefficients of the given transfer function or difference equation.

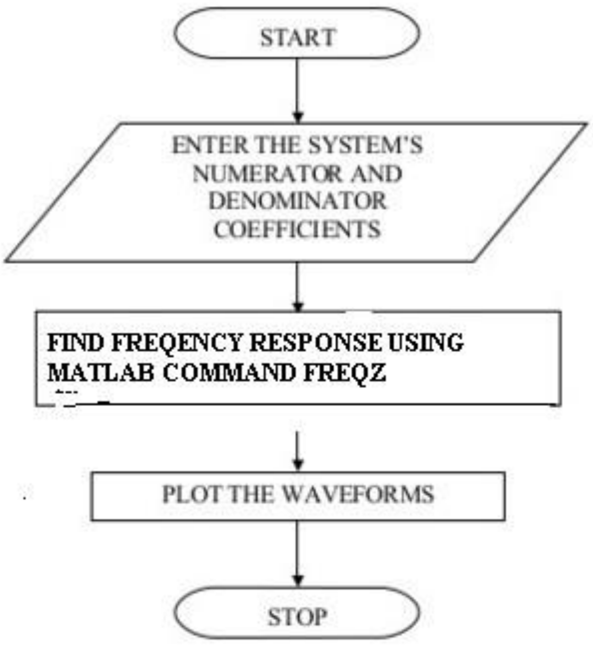
Step II : Give denominator coefficients of the given transfer function or difference equation

Step III : Pass these coefficients to matlab command freqz to find frequency response.

Step IV : Find magnitude and phase response using matlab commands abs and angle.

Step V : Plot magnitude and phase response.

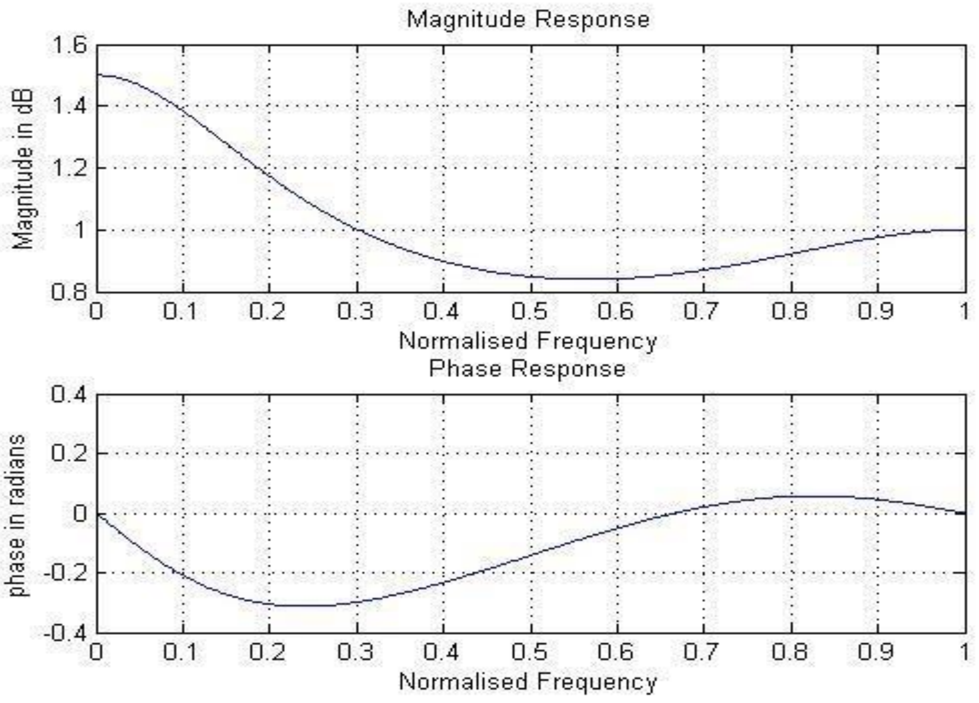
Flow Chart:


START
ENTER THE SYSTEM'S NUMERATOR AND DENOMINATOR COEFFICIENTS
FIND FREQENCY RESPONSE USING MATLAB COMMAND FREQZ
PLOT THE WAVEFORMS
STOP

PROGRAM:

```
%MATLAB program to plot the frequency response (magnitude and phase response)of
agiven difference equation.
clc;
clear all;
b=input('Enter the numerator coefficients:');          %coefficients of x(n)
a=input('Enter the denominator coefficients:');         %coefficients of y(n)
[h,w]=freqz(b,a);
subplot(2,1,1);
plot(w/pi,abs(h));
grid;
xlabel('Normalised Frequency');
ylabel('Magnitude in dB');
title('Magnitude Response');
subplot(2,1,2);
plot(w/pi,angle(h));
grid;
xlabel('Normalised Frequency');
ylabel('phase in radians');
title('Phase Response');
```

Expected Waveforms:
Enter the numerator coefficients: [1]
Enter the denominator coefficients: [1 -1/6 -1/6]



VIVA QUESTIONS:

1. Define Frequency response?
2. Define magnitude response?
3. Define Phase response?
4. Define transfer function.
5. State the significance of difference equations.
6. What are the classifications of system based on unit sample response?
7. What is zero input response?
8. Define impulse response?
9. Define System.
10. Write mathematical formula to find response of any given system.

Exercise:

1. Find the frequency response of the system described by the difference equation

   y(n)-3/4 y(n-1)+1/8y(n-3)=x(n)+2x(n-1).

OBSERVATIONS

Output Waveforms:

EXP.NO: 5

TO FIND THE FFT OF A GIVEN SEQUENCE

AIM: To find the FFT of a given sequence.

Software: MATLAB

THEORY:

DFT of a sequence

$$X[k] = \sum_{n=0}^{N-1} x[n] e^{-j(2\pi/N)kn}$$

Where N= Length of sequence.

K= Frequency Coefficient.

n = Samples in time domain.

FFT: -Fast Fourier transform.

There are two methods.

1. Decimation in time (DIT) FFT.
2. Decimation in Frequency (DIF) FFT.

Why we need FFT?

The no of multiplications in DFT = $N^{2.}$

The no of Additions in DFT = N (N-1).

For FFT.

The no of multiplication = $N/2 \log_2 N$.

The no of additions = $N \log_2 N$.

Algorithm:

Step I : Give input sequence x[n].

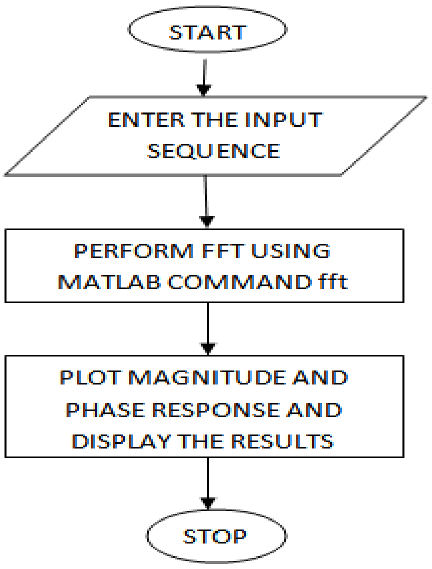
Step II : Find the length of the input sequence using length command.

Step III : Find FFT and IFFT using matlab commands fft and ifft.

Step IV : Plot magnitude and phase response

Step V : Display the results

Flow Chart:

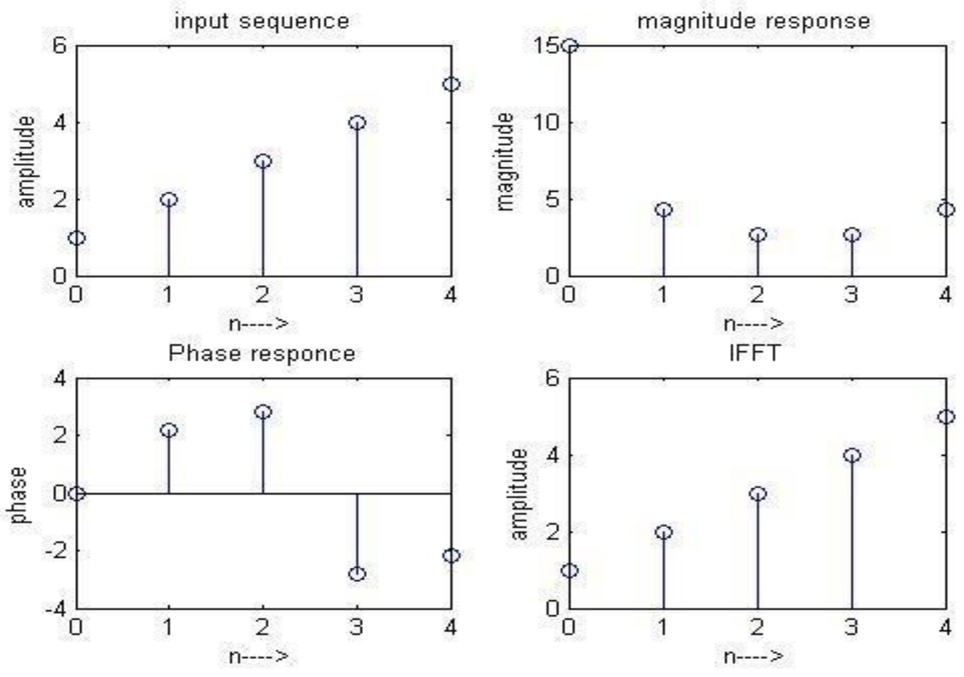

START
ENTER THE INPUT SEQUENCE
PERFORM FFT USING MATLAB COMMAND fft
PLOT MAGNITUDE AND PHASE RESPONSE AND DISPLAY THE RESULTS
STOP

PROGRAM:

```
clc;
clear all;
close all;
x=input('Enter the sequence : ')
N=length(x)
xK=fft(x,N)
xn=ifft(xK)
n=0:N-1;
subplot (2,2,1);
stem(n,x);
xlabel('n --->');
ylabel('amplitude');
title('input sequence');
subplot (2,2,2);
stem(n,abs(xK));
xlabel('n --->');
ylabel('magnitude');
title('magnitude response');
subplot (2,2,3);
stem(n,angle(xK));
xlabel('n --->');
ylabel('phase');
title('Phase responce');
subplot (2,2,4);
stem(n,xn);
xlabel('n --->');
ylabel('amplitude');
title('IFFT');
```

Expected Waveforms:

```
Enter the sequence: [1 2 3 4 5]
x =   1     2     3     4     5
N =  5
xK = 15.0000, -2.5000 + 3.4410i, -2.5000 + 0.8123i , -2.5000 - 0.8123i, -2.5000 - 3.4410i
xn = 1     2     3     4     5
```



VIVA QUESTIONS:

1. Define transform. What is the need for transform?
2. Differentiate Fourier transform and discrete Fourier transform.
3. Differentiate DFT and DTFT.
4. What are the advantages of FFT over DFT?
5. Differentiate DITFFT and DIFFFT algorithms.
6. What is meant by radix?
7. What is meant by twiddle factor and give its properties?
8. How FFT is useful to represent a signal?
9. Compare FFT and DFT with respect to number of calculation required?
10. How the original signal is reconstructed from the FFT of a signal?

Exercise:

1. Find 8-point DFT of sequence x(n)=[1 2 1 2 3 4 4 3] using FFT algorithm.

OBSERVATIONS

Output Waveforms:

EXP.NO: 6
DETERMINATION OF POWER SPECTRUM OF A GIVEN SIGNAL

AIM: Determination of Power Spectrum of a given signal.

Software: MATLAB

THEORY:

The power spectrum describes the distribution of signal power over a frequency spectrum. The most common way of generating a power spectrum is by using a discrete Fourier transform, but other techniques such as the maximum entropy method can also be used. The power spectrum can also be defined as the Fourier transform of auto correlation function.

Algorithm:

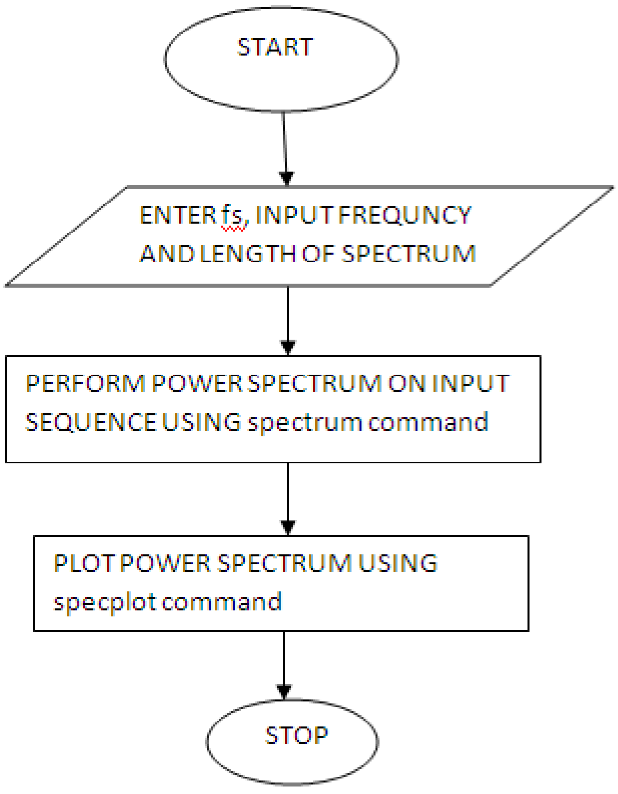
Step I : Give input sequence x

Step II : Give sampling frequency, input frequency and length of the spectrum.

Step III : Find power spectrum of input sequence using matlab command spectrum.

Step IV : Plot power spectrum using specplot.

Flow Chart:

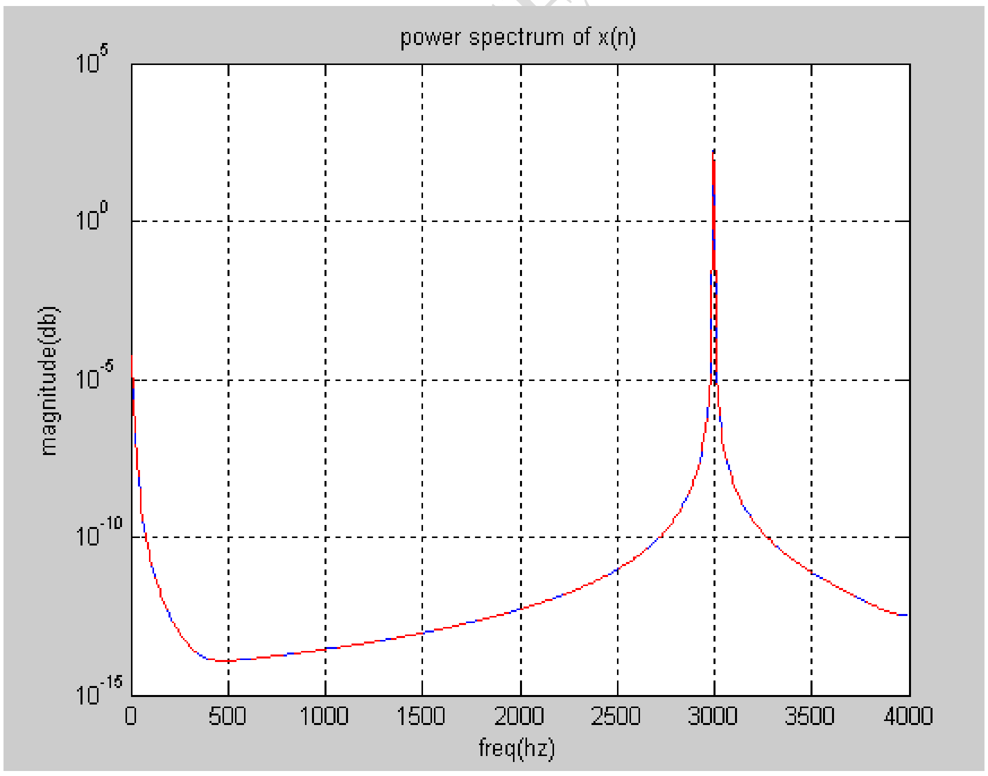

START
ENTER fs, INPUT FREQUNCY AND LENGTH OF SPECTRUM
PERFORM POWER SPECTRUM ON INPUT SEQUENCE USING spectrum command
PLOT POWER SPECTRUM USING specplot command
STOP

PROGRAM:

```
Clc;
clear all;
close all;
N=1024;
fs=8000;
f=input('enter the frequency[1 to 5000]:');
n=0:N-1;
x=sin(2*pi*(f/fs)*n)
pxx=spectrum(x,N);
specplot(pxx,fs);
grid on
xlabel('freq(hz)');
ylabel('magnitude(db)');
title('power spectrum of x(n)');
```

Expected Waveform:

Enter the frequency [1 to 5000]: 3000



VIVA QUESTIONS:

1. Define power signal.
2. Define energy signal.
3. Define power spectral density of a signal.
4. How the energy of a signal can be calculated?
5. Explain difference between energy spectral density and power spectral density.
6. Explain the PSD plot.
7. What is the importance of PSD?
8. What are the applications of PSD?
9. Explain MATLAB function randn(size(n)).
10. What is the need to represent the signal in frequency domain?

Exercise:

1. Find power spectrum of the signal x(n)=cos(2*pi*50*n)

OBSERVATIONS

Output Waveforms:

EXP. NO: 7
IMPLEMENTATION OF LP FIR FILTER

AIM: To implement LP FIR filter for a given sequence.

Software: MATLAB

THEORY:

FIR filters are digital filters with finite impulse response. They are also known as **non-recursive digital filters** as they do not have the feedback.

An FIR filter has two important advantages over an IIR design:

- Firstly, there is no feedback loop in the structure of an FIR filter. Due to not having a feedback loop, an FIR filter is inherently stable. Meanwhile, for an IIR filter, we need to check the stability.
- Secondly, an FIR filter can provide a linear-phase response. As a matter of fact, a linear-phase response is the main advantage of an FIR filter over an IIR design otherwise, for the same filtering specifications; an IIR filter will lead to a lower order.

FIR FILTER DESIGN

An FIR filter is designed by finding the coefficients and filter order that meet certain specifications, which can be in the time-domain (e.g. a matched filter) and/or the frequency domain (most common). Matched filters perform a cross-correlation between the input signal and a known pulse-shape. The FIR convolution is a cross-correlation between the input signal and a time-reversed copy of the impulse-response. Therefore, the matched-filter's impulse response is "designed" by sampling the known pulse-shape and using those samples in reverse order as the coefficients of the filter.

When a particular frequency response is desired, several different design methods are common:

1. Window design method
2. Frequency Sampling method
3. Weighted least squares design

WINDOW DESIGN METHOD

In the window design method, one first designs an ideal IIR filter and then truncates the infinite impulse response by multiplying it with a finite length window function. The result is a finite impulse response filter whose frequency response is modified from that of the IIR filter.

Window Name	**Transition Approximate**	**Width $\Delta\omega$ Exact values**	**Min. Stop band Attenuation**	**Matlab Command**
Rectangular	$\frac{4\Pi}{M}$	$\frac{1.8\Pi}{M}$	**21db**	**B = FIR1(N,Wn,boxcar)**
Bartlett	$\frac{8\Pi}{M}$	$\frac{6.1\Pi}{M}$	**25db**	**B = FIR1(N,Wn,bartlett)**
Hanning	$\frac{8\Pi}{M}$	$\frac{6.2\Pi}{M}$	**44db**	**B = FIR1(N,Wn,hanning)**
Hamming	$\frac{8\Pi}{M}$	$\frac{6.6\Pi}{M}$	**53db**	**B= FIR1(N,W_n,hamming)**
Blackman	$\frac{12\Pi}{M}$	$\frac{11\Pi}{M}$	**74db**	**B = FIR1(N,Wn,blackman)**

Algorithm:

Step I : Enter the pass band frequency (fp) and stop band frequency (fq).

Step II : Get the sampling frequency (fs), length of window (n).

Step III : Calculate the cut off frequency, fn

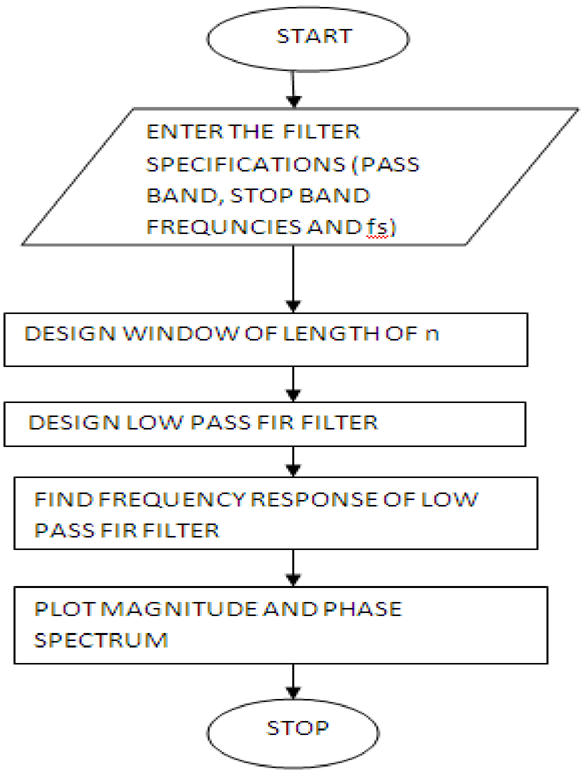
Step IV : Use boxcar, hamming, blackman Commands to design window.

Step V : Design filter by using above parameters.

Step VI : Find frequency response of the filter using matlab command freqz.

Step VII : Plot the magnitude response and phase response of the filter.

Flow Chart:

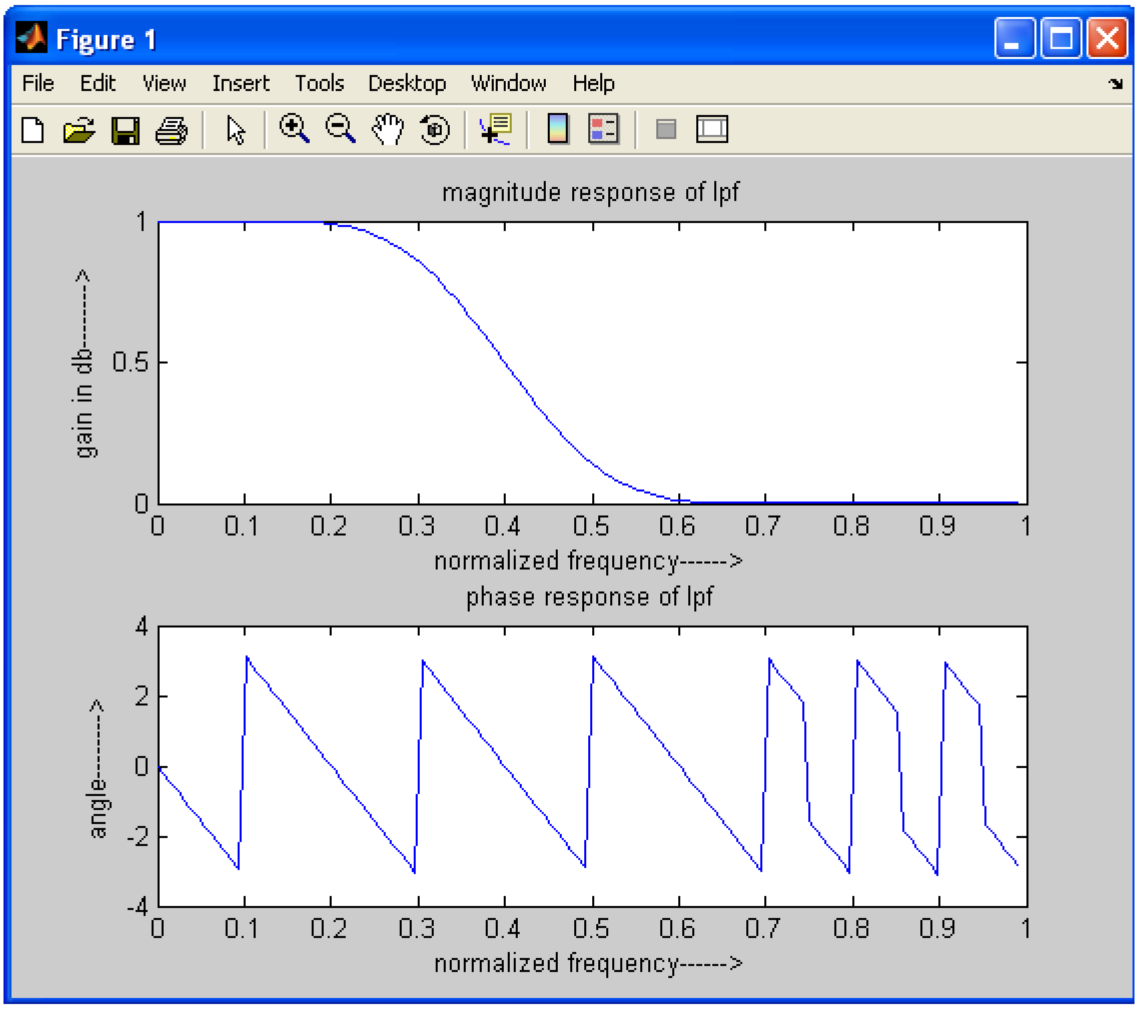

START
ENTER THE FILTER SPECIFICATIONS (PASS BAND, STOP BAND FREQUNCIES AND fs)
DESIGN WINDOW OF LENGTH OF n
DESIGN LOW PASS FIR FILTER
FIND FREQUENCY RESPONSE OF LOW PASS FIR FILTER
PLOT MAGNITUDE AND PHASE SPECTRUM
STOP

PROGRAM:

```
clc;
clear all;
close all;
n=20;
fp=200;
fq=300;
fs=1000;
fn=2*fp/fs;
window=blackman(n+1);
b=fir1(n,fn,window);
[H W]=freqz(b,1,128);
subplot(2,1,1);
plot(W/pi,abs(H));
title('magnitude response of lpf');
ylabel('gain in db ------->');
xlabel('normalized frequency ----->');
```

```
subplot(2,1,2);
plot(W/pi,angle(H));
title('phase response of lpf');
ylabel('angle ------ >');
xlabel('normalized frequency ----->');
```

Expected Wave forms:



VIVA QUESTIONS:

1. Define filter.
2. What are the different types of filters?
3. Why are FIR filters generally preferred over IIR filters in multirate (decimating and interpolating) systems/
4. Difference between IIR and FIR filters?
5. Differentiate ideal filter and practical filter responses.
6. What is the filter specifications required to design the analog filters?
7. What is meant by frequency response of filter?
8. What is meant by magnitude response?
9. What is meant by phase response?
10. Difference between FIR low pass filter and high pass filter.

Exercise:

1. Design LP FIR filter using Bartlett and hamming window.

OBSERVATIONS

Output Waveforms:

EXP. NO: 8
IMPLEMENTATION OF HP FIR FILTER

AIM: To implement HP FIR filter for a given sequence.

Software: MATLAB

Algorithm:

Step I : Enter the pass band frequency (fp) and stop band frequency (fq).

Step II : Get the sampling frequency (fs), length of window (n).

Step III : Calculate cut off frequency

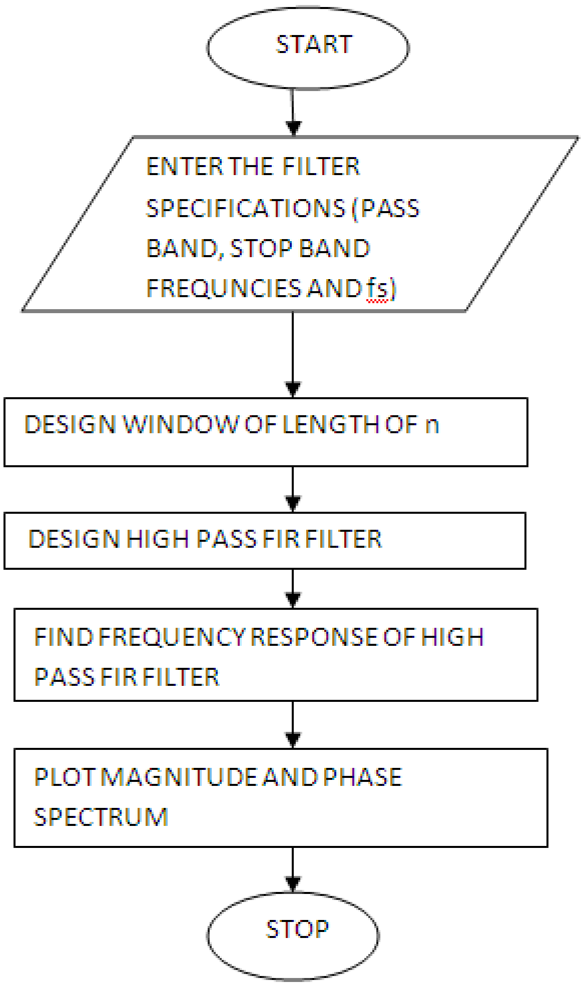
Step IV : Use boxcar, hamming, Blackman Commands to design window.

Step V : Design filter by using above parameters.

Step VI : Find frequency response of the filter using matlab command freqz.

Step VII : Plot the magnitude response and phase response of the filter.

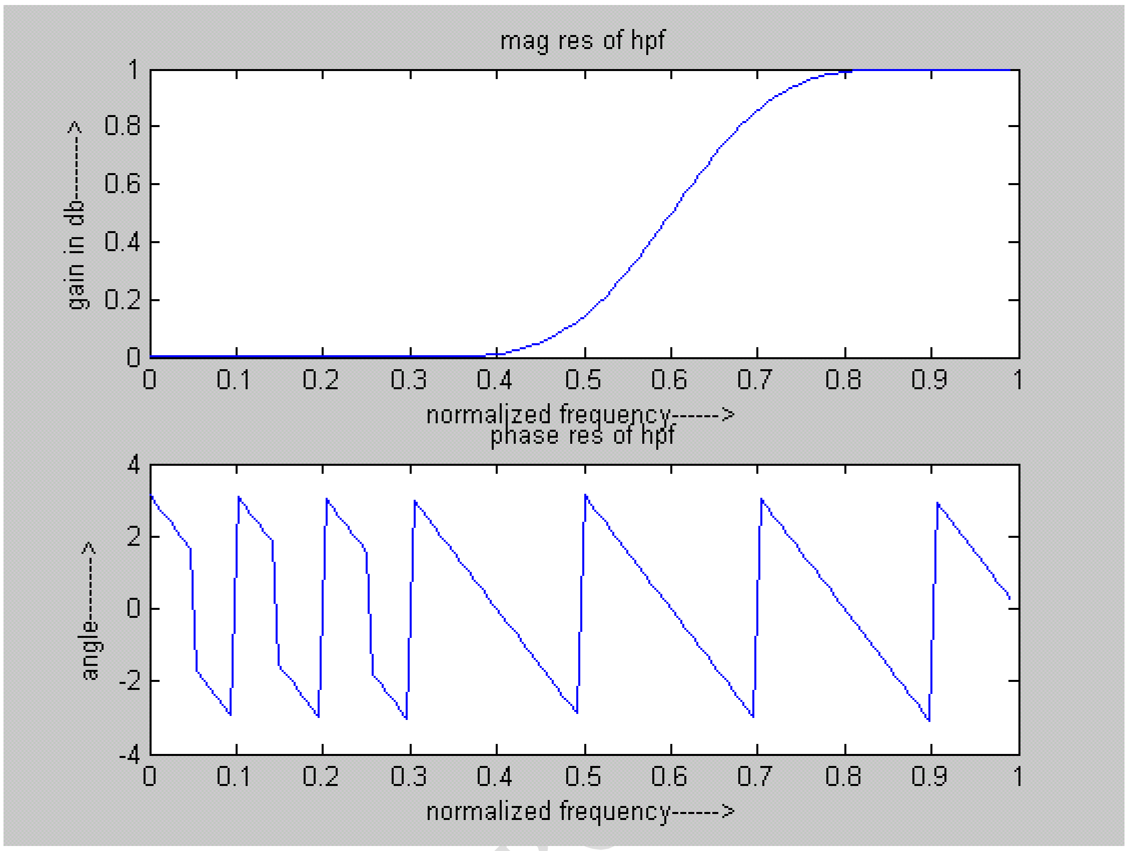
Flow Chart:


START
ENTER THE FILTER SPECIFICATIONS (PASS BAND, STOP BAND FREQUNCIES AND fs)
DESIGN WINDOW OF LENGTH OF n
DESIGN HIGH PASS FIR FILTER
FIND FREQUENCY RESPONSE OF HIGH PASS FIR FILTER
PLOT MAGNITUDE AND PHASE SPECTRUM
STOP

PROGRAM:

```
clc;
clear all;
close all;
n=20;
fp=300;
fq=200;
fs=1000;
fn=2*fp/fs;
window=blackman(n+1);
b=fir1(n,fn,'high',window);
[H W]=freqz(b,1,128);
subplot(2,1,1);
plot(W/pi,abs(H));
title('mag res of lpf');
ylabel('gain in db ------->');
xlabel('normalized frequency ----->');
subplot(2,1,2);
plot(W/pi,angle(H));
title('phase res of lpf');
ylabel('angle ------ >');
xlabel('normalized frequency ----->');
```

Expected Waveforms:



VIVA QUESTIONS:

1. What is a filter?
2. Differentiate analog filter and digital filter.
3. Define FIR filter.
4. What are the differences between recursive and non recursive systems?
5. List a few Applications of FIR filters.
6. Explain advantages of FIR filters over IIR filters.
7. Explain limitations of FIR filters.
8. What are the different methods to design FIR filters?
9. Explain different window functions.
10. Differentiate rectangular, triangular and Kaiser Windows.

Exercise:

1. Design HP FIR filter using Rectangular and hanning Window.

OBSERVATIONS

Output Waveforms:

EXP. NO: 9

IMPLEMENTATION OF LP IIR FILTER

AIM: To implement LP IIR filter for a given sequence.

Software: MATLAB

THEORY:

IIR filters are digital filters with infinite impulse response. Unlike FIR filters, they have the feedback (a recursive part of a filter) and are known as recursive digital filters therefore.

For this reason IIR filters have much better frequency response than FIR filters of the same order. Unlike FIR filters, their phase characteristic is not linear which can cause a problem to the systems which need phase linearity. For this reason, it is not preferable to use IIR filters in digital signal processing when the phase is of the essence. Otherwise, when the linear phase characteristic is not important, the use of IIR filters is an excellent solution.

There is one problem known as a potential instability that is typical of IIR filters only. FIR filters do not have such a problem as they do not have the feedback. For this reason, it is always necessary to check after the design process whether the resulting IIR filter is stable or not.

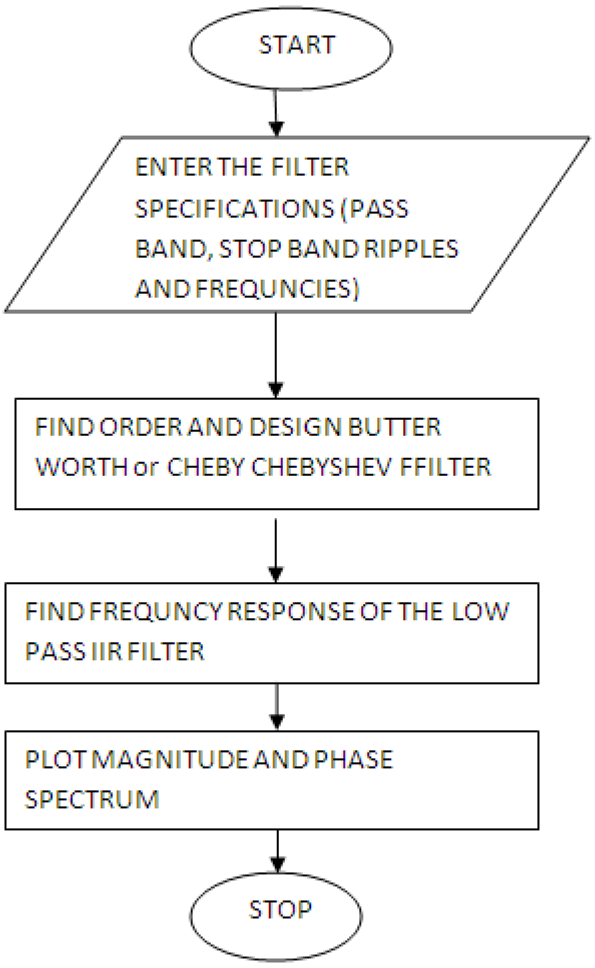
IIR FILTER DESIGN

For the given specifications to Design a digital IIR filter, first we need to design analog filter (Butterworth or chebyshev). The resultant analog filter is transformed to digital filter by using either “Bilinear transformation or Impulse Invariant transformation”.

Algorithm:

Step I	: Enter the pass band ripple (rp) and stop band ripple (rs).
Step II	: Enter the pass band frequency (wp) and stop band frequency (ws).
Step III	: Get the sampling frequency (fs).
Step IV	: Calculate normalized pass band frequency, and normalized stop band frequency w1 and w2 respectively. w1 = 2 * wp /fs w2 = 2 * ws /fs
Step V	: Make use of the following function to calculate order of filter Butterworth filter order [n,wn]=buttord(w1,w2,rp,rs) Chebyshev filter order [n,wn]=cheb1ord(w1,w2,rp,rs)
Step VI	: Design an nth order digital low pass Butterworth or Chebyshev filter using the following statements. Butterworth filter [b, a]=butter (n, wn) Chebyshev filter [b,a]=cheby1 (n, 0.5, wn)
Step VII	: Find the digital frequency response of the filter by using 'freqz( )' function
Step VIII	: Calculate the magnitude of the frequency response in decibels (dB) mag=20*log10 (abs (H))
Step IX	: Plot the magnitude response [magnitude in dB Vs normalized frequency]
Step X	: Calculate the phase response using angle (H)
Step XI	: Plot the phase response [phase in radians Vs normalized frequency (Hz)].

Flow Chart:


START
ENTER THE FILTER SPECIFICATIONS (PASS BAND, STOP BAND RIPPLES AND FREQUNCIES)
FIND ORDER AND DESIGN BUTTER WORTH or CHEBY CHEBYSHEV FFILTER
FIND FREQUNCY RESPONSE OF THE LOW PASS IIR FILTER
PLOT MAGNITUDE AND PHASE SPECTRUM
STOP

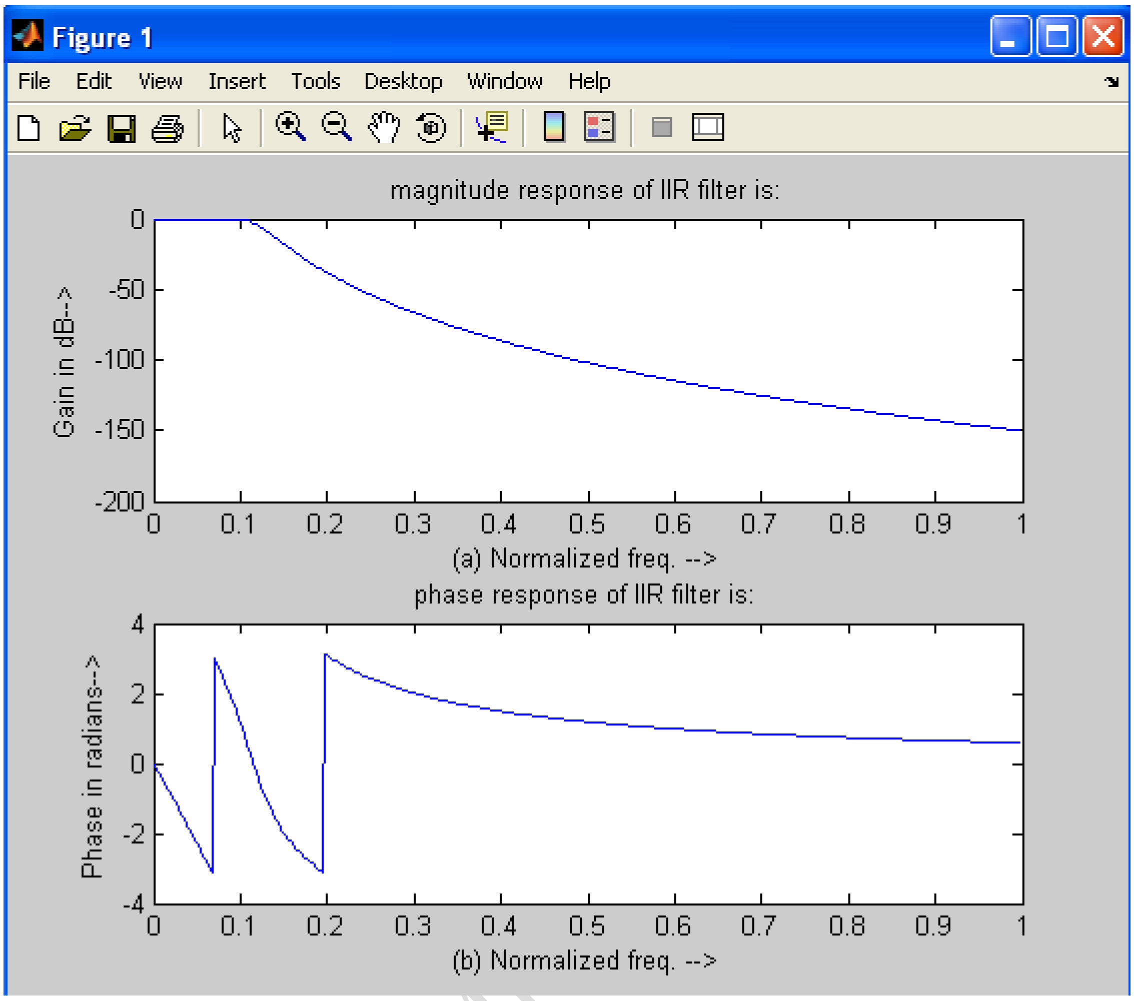
PROGRAM:

```
clc;
clear all;
close all;
disp('enter the IIR filter design specifications');
rp=input('enter the passband ripple:');
rs=input('enter the stopband ripple:');
wp=input('enter the passband freq:');
ws=input('enter the stopband freq:');
fs=input('enter the sampling freq:');
w1=2*wp/fs;w2=2*ws/fs;
[n,wn]=buttord(w1,w2,rp,rs,'s');
disp('Frequency response of IIR LPF is:');
[b,a]=butter(n,wn,'low','s');
```

```
w=0:.01:pi;
[h,om]=freqs(b,a,w);
m=20*log10(abs(h));
an=angle(h);
figure,subplot(2,1,1);plot(om/pi,m);
title('magnitude response of IIR filter is:');
xlabel('(a) Normalized freq. -->');
ylabel('Gain in dB-->');
subplot(2,1,2);plot(om/pi,an);
title('phase response of IIR filter is:');
xlabel('(b) Normalized freq. -->');
ylabel('Phase in radians-->');
```

Expected waveforms:

```
enter the IIR filter design specifications
enter the passband ripple:15
enter the stopband ripple:60
enter the passband freq:1500
enter the stopband freq:3000
enter the sampling freq:7000
```



VIVA QUESTIONS:

1. List some advantages of digital filters over analog filters.
2. Write some differences between FIR and IIR filters.
3. What are the different methods to design IIR filters?
4. Why IIR filters are not reliable?
5. What are different applications of IIR filters?
6. What are advantages of IIR filters?
7. What are disadvantages of IIR filters?
8. Differentiate Butterworth and Chebyshev approximations.
9. What is meant by impulse response?
10. Difference between IIR low pass and High pass filters.

Exercise:

1. Design low pass IIR filter using Chebyshev filter.

OBSERVATIONS

Output Waveforms:

EXP. NO: 10
IMPLEMENTATION OF HP IIR FILTER

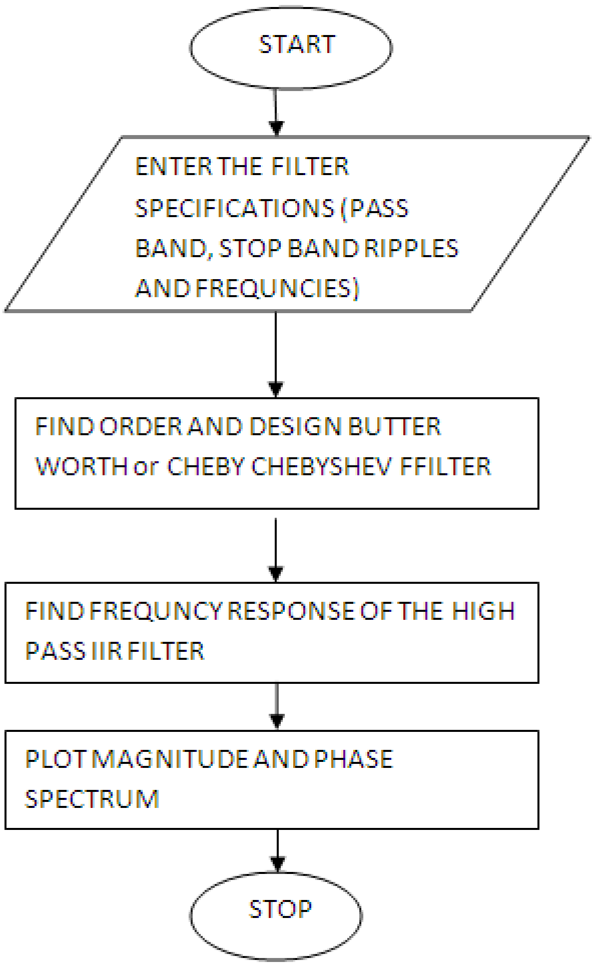
AIM: To implement HP IIR filter for a given sequence.

Software: MATLAB

Algorithm:

Step I	: Enter the pass band ripple (rp) and stop band ripple (rs).
Step II	: the pass band frequency (wp) and stop band frequency (ws).
Step III	: Get the sampling frequency (fs).
Step IV	: Calculate normalized pass band frequency, and normalized stop band frequency w1 and w2 respectively. w1 = 2 * wp /fs w2 = 2 * ws /fs
Step V	: Make use of the following function to calculate order of filter Butterworth filter order [n,wn]=buttord(w1,w2,rp,rs) Chebyshev filter order [n,wn]=cheb1ord(w1,w2,rp,rs)
Step VI	: Design an nth order digital high pass Butterworth or Chebyshev filter using the following statement. Butterworth filter [b,a]=butter (n, wn,’high’) Chebyshev filter [b,a]=cheby1 (n, 0.5, wn,'high')
Step VII	: Find the digital frequency response of the filter by using ‘freqz( )’ function
Step VIII	: Calculate the magnitude of the frequency response in decibels (dB) mag=20*log10 (abs (H))
Step IX	: Plot the magnitude response [magnitude in dB Vs normalized frequency]
Step X	: Calculate the phase response using angle (H)
Step XI	: Plot the phase response [phase in radians Vs normalized frequency (Hz)].

Flow chart:


START
ENTER THE FILTER SPECIFICATIONS (PASS BAND, STOP BAND RIPPLES AND FREQUNCIES)
FIND ORDER AND DESIGN BUTTER WORTH or CHEBY CHEBYSHEV FFILTER
FIND FREQUNCY RESPONSE OF THE HIGH PASS IIR FILTER
PLOT MAGNITUDE AND PHASE SPECTRUM
STOP

PROGRAM:

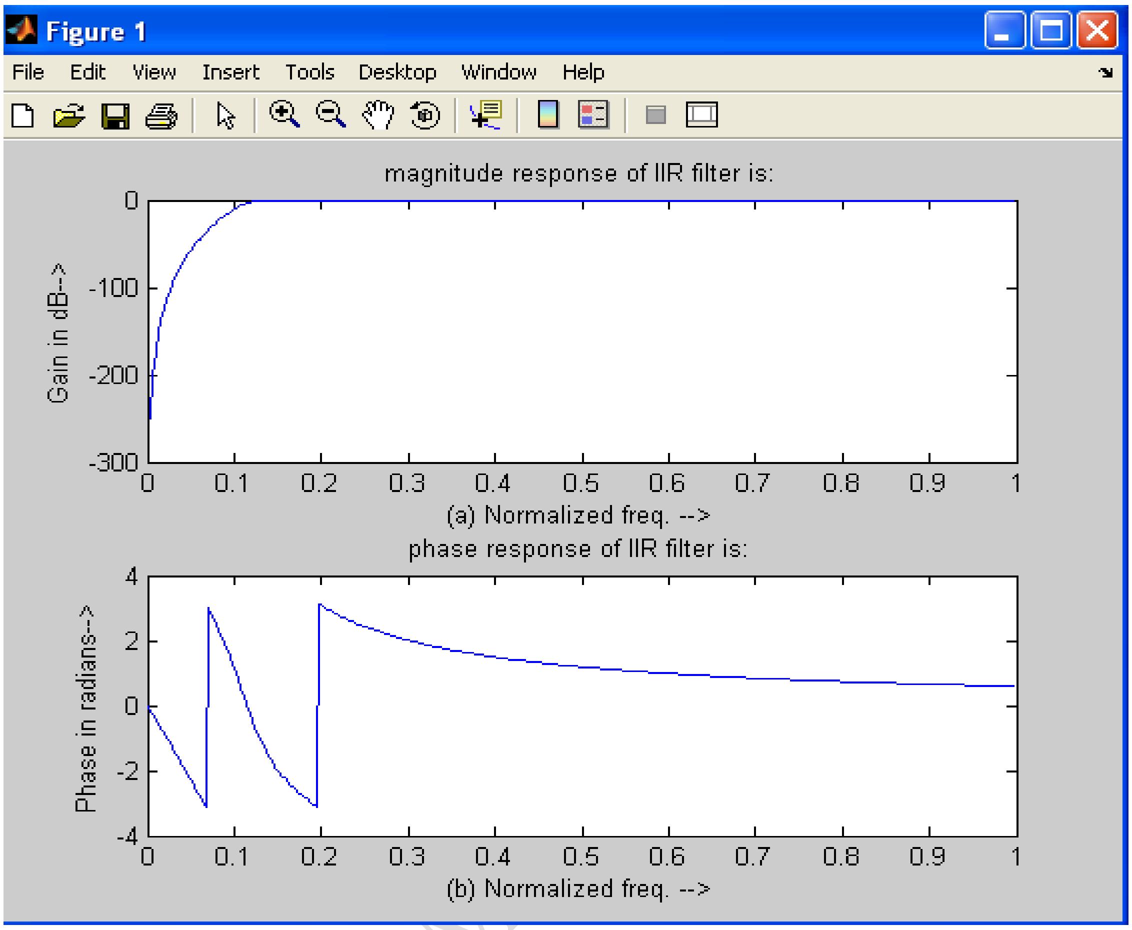
```
clc;
clear all;
close all;
disp('enter the IIR filter design specifications');
rp=input('enter the passband ripple');
rs=input('enter the stopband ripple');
wp=input('enter the passband freq');
ws=input('enter the stopband freq');
fs=input('enter the sampling freq');
w1=2*wp/fs;w2=2*ws/fs;
[n,wn]=buttord(w1,w2,rp,rs,'s');
```

```
disp('Frequency response of IIR HPF is:');
[b,a]=butter(n,wn,'high','s');
w=0:.01:pi;
[h,om]=freqs(b,a,w);
m=20*log10(abs(h));
an=angle(h);
figure,subplot(2,1,1);plot(om/pi,m);
title('magnitude response of IIR filter is:');
xlabel('(a) Normalized freq. -->');
ylabel('Gain in dB-->');
subplot(2,1,2);plot(om/pi,an);
title('phase response of IIR filter is:');
xlabel('(b) Normalized freq. -->');
ylabel('Phase in radians-->');
```

INPUT:

```
enter the IIR filter design specifications
enter the passband ripple15
enter the stopband ripple60
enter the passband freq1500
enter the stopband freq3000
enter the sampling freq7000
```

Expected Waveforms:



VIVA QUESTIONS:

1. What are the filter specifications required to design the analog filters?
2. What is meant by frequency response of filter?
3. What is meant by magnitude response?
4. What is meant by phase response?
5. Differentiate ideal filter and practical filter responses.
6. What are the different types of analog filter approximations?
7. Define order of the filter and explain important role it plays in designing of a filter.
8. Explain advantages and disadvantages of Butterworth filter
9. Explain advantages and disadvantages of Chebyshev filter 1
10. Why Chebyshev is better than Butterworth filter?

Exercise:

1. Design High pass IIR filter using Chebyshev filter.

OBSERVATIONS

Output Waveforms:

EXP. NO: 11
GENERATION OF SINUSOIDAL SIGNAL THROUGH FILTERING

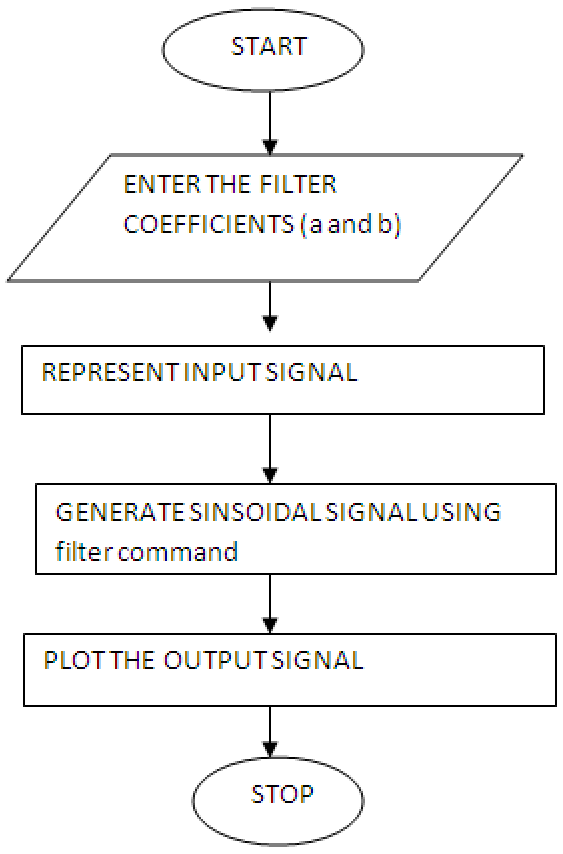
AIM: To generate a sinusoidal signal through filtering.

Software: MATLAB

Theory:

One application of an LTI discrete-time system is to pass certain frequency components in an input sequence without any distortion and block other frequency components. Such systems are called digital filters. The key to the filtering process is the inverse Discrete Fourier transform, which expresses an arbitrary input sequence as a linear weighed sum of an infinite number of exponential sequences, or equivalently, as a linear weighted sum of sinusoidal sequences. As a result, by appropriately choosing the values of magnitude function of the LTI digital filter at frequencies corresponding to the frequencies of the sinusoidal components of the input, some of these sinusoidal sequences can be selectively heavily attenuated or filtered with respect to the others.

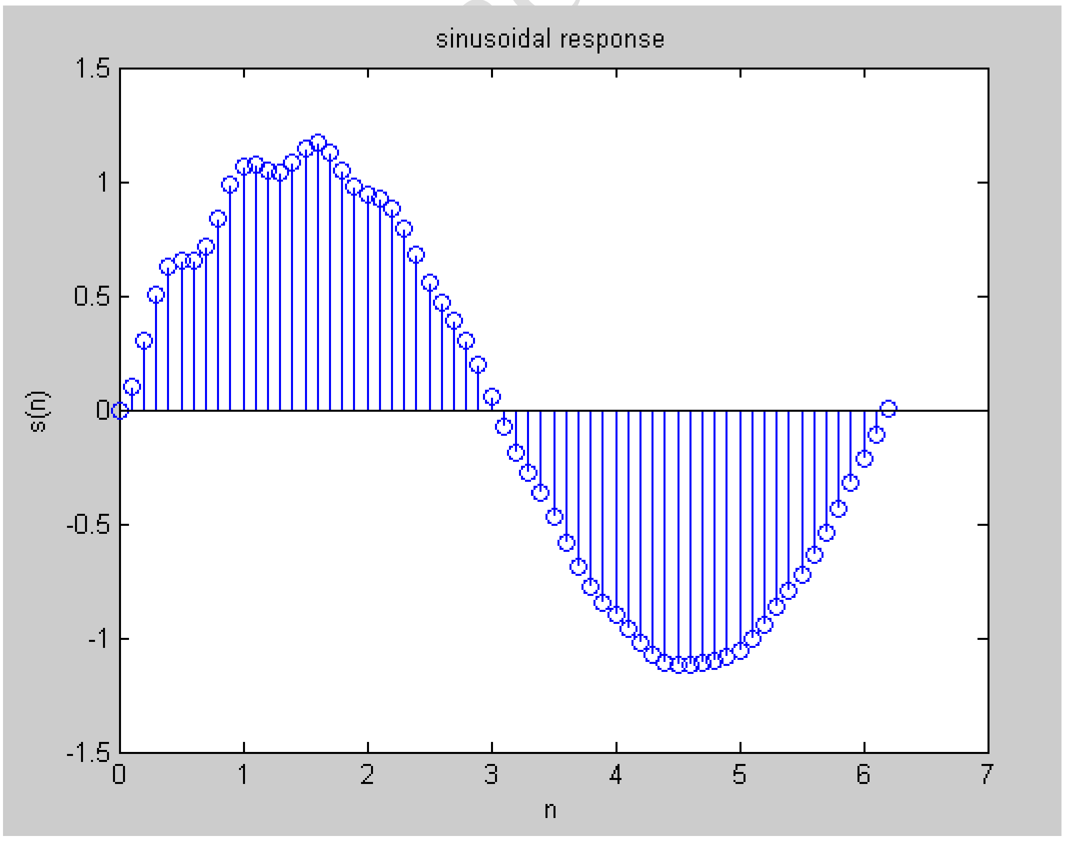
Flow Chart:


START
ENTER THE FILTER COEFFICIENTS (a and b)
REPRESENT INPUT SIGNAL
GENERATE SINSOIDAL SIGNAL USING filter command
PLOT THE OUTPUT SIGNAL
STOP

PROGRAM:

```
close all;
clear all;
clc;
b=[1];
 a=[1,-1,0.9];
  n=[-20:120];
  t=0:0.1:2*pi;
  x=sin(t);
  s=filter(b,a,x);
  stem(t,s);
  title('sinusoidal response');
  xlabel('n');
  ylabel('s(n)');
```

Expected Waveform:



VIVA QUESTIONS:

1. Define sequence and signal?
2. Differentiate periodic and non-periodic signals?
3. Define period of the signal?
4. Define LTI system.
5. What is filtering?

Exercise:
1. Generate sinusoidal signal using filtering with coefficients a= [1 3 7 8] & b= [0 1].

OBSERVATIONS

Output Waveforms:

EXP. NO: 12
GENERATION OF DTMF SIGNALS

AIM: To generate DTMF Signals using MATLAB Software.

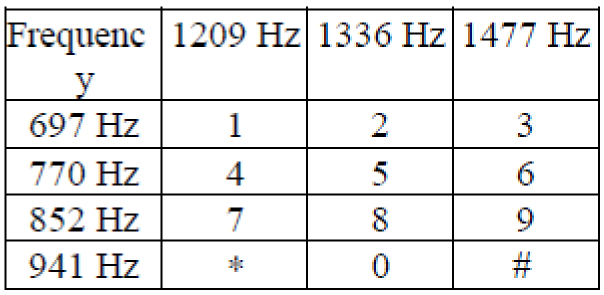
Software: MATLAB

THEORY:

The DTMF stands for "*D*ual *T*one *M*ulti *F*requency", and is a method of representing digits with tone frequencies, in order to transmit them over an analog communications network, for example a telephone line. In telephone networks, DTMF signals are used to encode dial trains and other information.

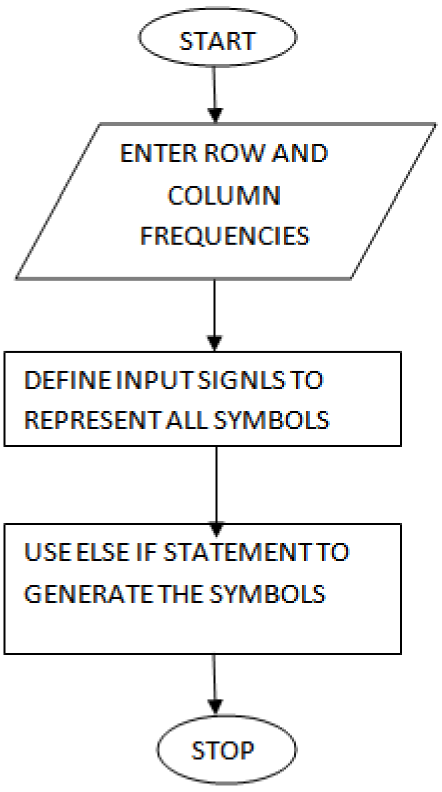
Dual-tone Multi-Frequency (DTMF) signaling is the basis for voice communications control and is widely used worldwide in modern telephony to dial numbers and configure switchboards. It is also used in systems such as in voice mail, electronic mail and telephone banking.

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



Frequenc y	1209 Hz	1336 Hz	1477 Hz
697 Hz	1	2	3
770 Hz	4	5	6
852 Hz	7	8	9
941 Hz	*	0	#

Flow Chart:


START
ENTER ROW AND COLUMN FREQUENCIES
DEFINE INPUT SIGNLS TO REPRESENT ALL SYMBOLS
USE ELSE IF STATEMENT TO GENERATE THE SYMBOLS
STOP

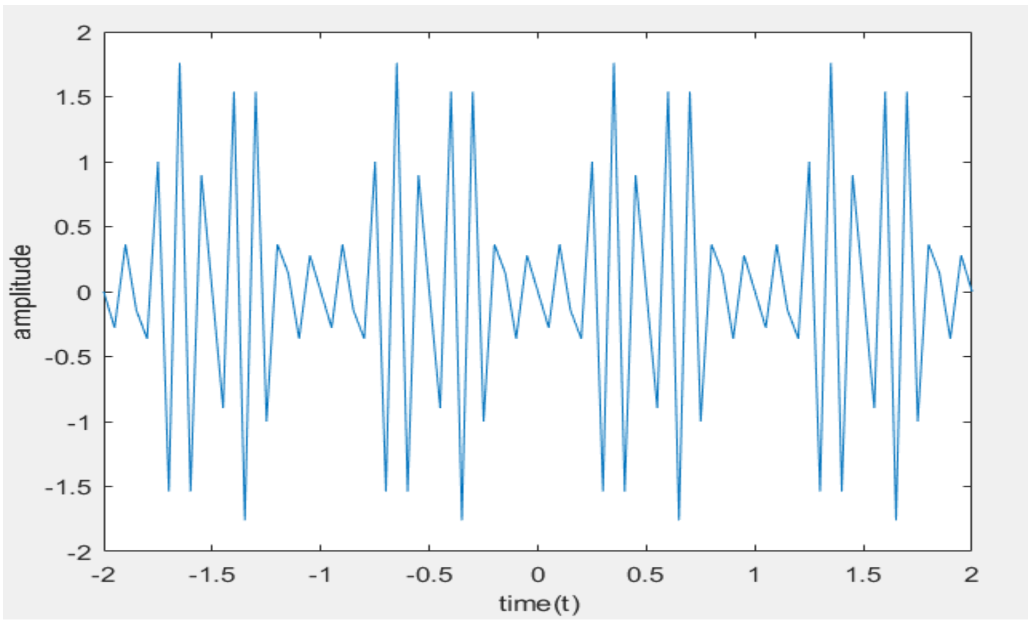
PROGRAM:

```
Clc;
Clear all;
Close all;
t = -2:0.05:2;
x=input('enter the input number');
 fr1=697;
 fr2=770;
 fr3=852;
 fr4=941;
 fc1=1209;
 fc2=1336;
 fc3=1477;
 fc4=1633;
y0  =  sin(2*pi*fr4*t)  +  sin(2*pi*fc2*t);   %  0
 y1  =  sin(2*pi*fr1*t)  +  sin(2*pi*fc1*t);  %  1
 y2  =  sin(2*pi*fr1*t)  +  sin(2*pi*fc2*t);  %  2
 y3 = sin(2*pi*fr1*t) + sin(2*pi*fc3*t); % 3
 y4  =  sin(2*pi*fr2*t)  +  sin(2*pi*fc1*t);  %  4
 y5  =  sin(2*pi*fr2*t)  +  sin(2*pi*fc2*t);  %  5
 y6  =  sin(2*pi*fr2*t)  +  sin(2*pi*fc3*t);  %  6
 y7  =  sin(2*pi*fr3*t)  +  sin(2*pi*fc1*t);  %  7
 y8  =  sin(2*pi*fr3*t)  +  sin(2*pi*fc2*t);  %  8
 y9 = sin(2*pi*fr3*t) + sin(2*pi*fc3*t); % 9
 y_start = sin(2*pi*fr4*t) + sin(2*pi*fc1*t);     % *
 y_canc  = sin(2*pi*fr4*t) + sin(2*pi*fc3*t);     % #
 if (x==1)
   plot(t,y1)
   xlabel('time(t)')
   ylabel('amplitude')
elseif (x==2)
   plot(t,y2)
   xlabel('time(t)')
   ylabel('amplitude')
elseif (x==3)
    plot(t,y3)
    xlabel('time(t)')
    ylabel('amplitude')
elseif (x==4)
    plot(t,y4)
    xlabel('time(t)')
    ylabel('amplitude')
elseif (x==5)
    plot(t,y5)
    xlabel('time(t)')
    ylabel('amplitude')
elseif (x==6)
    plot(t,y6)
    xlabel('time(t)')
    ylabel('amplitude')
elseif (x==7)
    plot(t,y7)
    xlabel('time(t)')
    ylabel('amplitude')
elseif (x==8)
    plot(t,y8)
    xlabel('time(t)')
    ylabel('amplitude')
elseif (x==9)
    plot(t,y9)
    xlabel('time(t)')
    ylabel('amplitude')
elseif (x==0)
    plot(t,y0)
    xlabel('time(t)')
    ylabel('amplitude')
elseif (x==11)
   plot(t,y_start)
```

```
  xlabel('time(t)')
  ylabel('amplitude')
  elseif (x==12)
  plot(t,y_canc)
  xlabel('time(t)')
  ylabel('amplitude')
else
   disp('enter the correct input')
end
```

Expected Waveform:

Enter the input number = 7



VIVA QUESTIONS:

1. Expand DTMF?
2. Define frequency groups for all symbols?
3. Define DTMF?
4. Give the applications of DTMF?
5. Define low frequency group and high frequency group?

Exercise:

1. Generate DTMF signal for the symbols 5, 8, 3 and #.

OBSERVATIONS

Output Waveforms:

EXP. NO: 13
IMPLEMENTATION OF DECIMATION PROCESS

AIM: program to verify the decimation of given sequence.

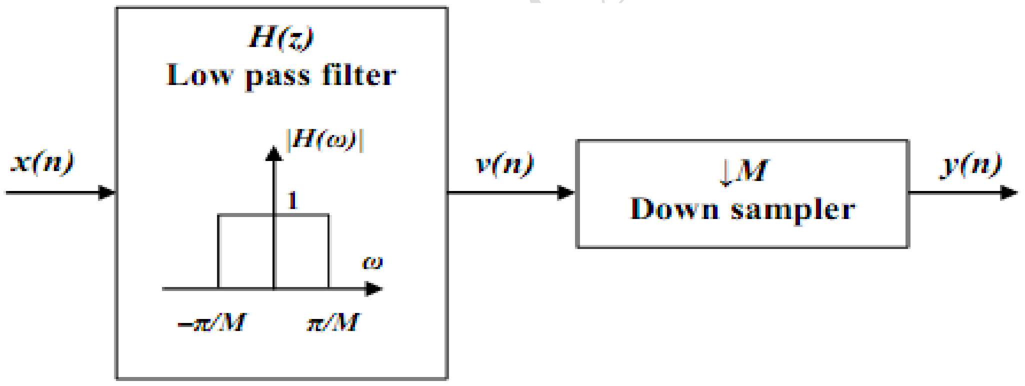
SOFTWARE: MATLAB

THEORY:

"Decimation" is the process of reducing the sampling rate.

"Downsampling" is a more specific term which refers to just the process of throwing away samples, without the lowpass filtering operation.

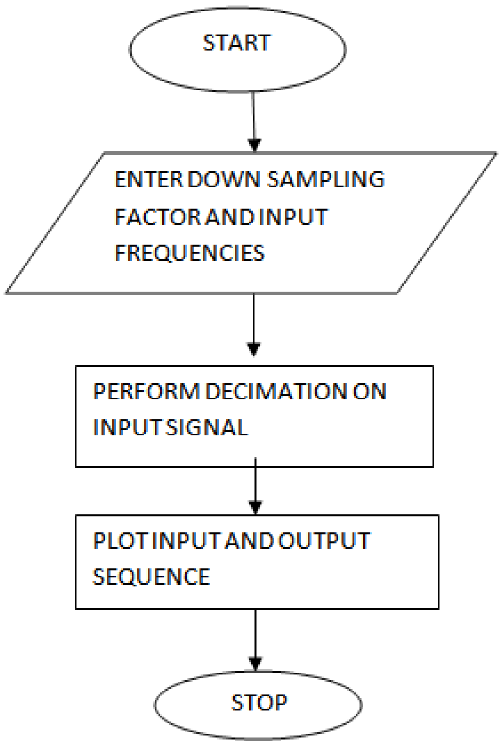
The most immediate reason to decimate is simply to reduce the sampling rate at the output of one system so a system operating at a lower sampling rate can input the signal. But a much more common motivation for decimation is to reduce the *cost* of processing: the *calculation* and/or *memory* required to implement a DSP system generally is proportional to the sampling rate, so the use of a lower sampling rate usually results in a cheaper implementation.



Algorithm:

Step I	: Define down sampling factor and input frequencies f1 and f2
Step II	: Represent input sequence with frequencies f1 and f2
Step III	: Perform the decimation on input signal using matlab command decimate.
Step IV	: Plot the input and output sequence

Flow Chart:

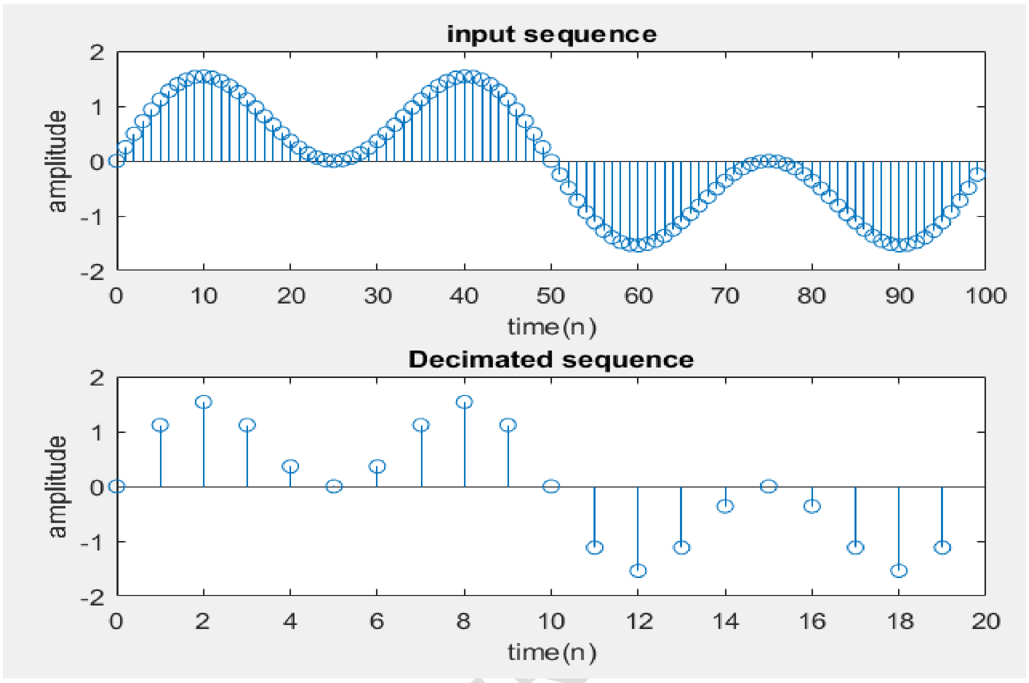

START
ENTER DOWN SAMPLING FACTOR AND INPUT FREQUENCIES
PERFORM DECIMATION ON INPUT SIGNAL
PLOT INPUT AND OUTPUT SEQUENCE
STOP

PROGRAM:

```
Clc;
Clear all;
Close all;
D=input('enter the downsampling factor');
L=input('enter the length of the input signal');
f1=input('enter the frequency of first sinusodal');
f2=input('enter the frequency of second sinusodal');
n=0:L-1;
x=sin(2*pi*f1*n)+sin(2*pi*f2*n);
y=decimate(x,D,'fir');
subplot(2,1,1);
stem(n,x(1:L));
title('input sequence');
xlabel('time(n)');
ylabel('amplitude');
subplot(2,1,2)
m=0:(L/D)-1;
stem(m,y(1:L/D));
title('Decimated sequence');
xlabel('time(n)');
ylabel('amplitude');
```

Expected Waveforms:

```
enter the downsampling factor = 5
enter the length of the input signal = 100
enter the frequency of first sinusoidal = 0.01
enter the frequency of second sinusoidal = 0.03
```



VIVA QUESTIONS:

1. Explain about multi rate digital signal processing.
2. List the Applications of multi rate digital signal processing.
3. Define interpolation.
4. Define decimation.
5. Define aliasing

Exercise:

1. Perform Decimation on the input signal x(n)=cos(2*pi*30*n).

OBSERVATIONS

Output Waveforms:

EXP. NO: 14
IMPLEMENTATION OF INTERPOLATION PROCESS

AIM: program to verify the decimation of given sequence.

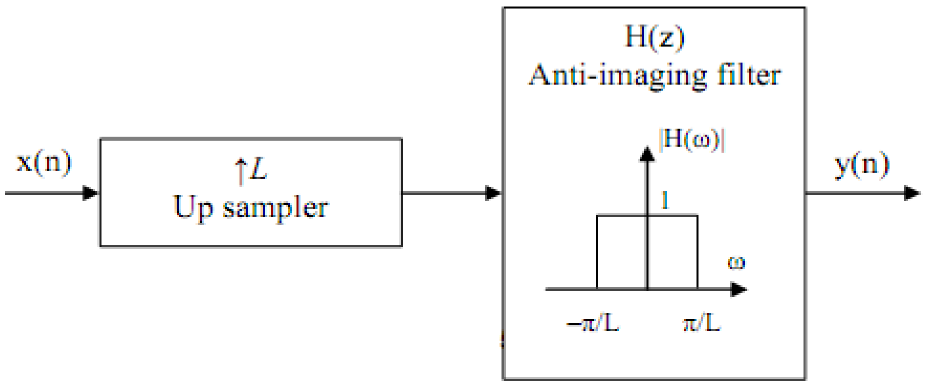
SOFTWARE: MATLAB

THEORY

"Upsampling" is the process of inserting zero-valued samples between original samples to increase the sampling rate. (This is called "zero-stuffing".)

"Interpolation", is the process of upsampling followed by filtering. The filtering removes the undesired spectral images.

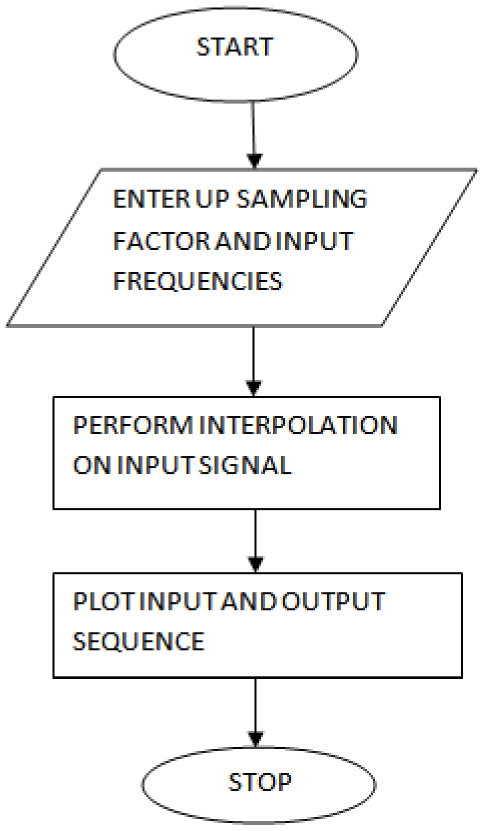
The primary reason to interpolate is simply to increase the sampling rate at the output of one system so that another system operating at a higher sampling rate can input the signal.



Algorithm:

Step I	: Define up sampling factor and input frequencies f1 and f2
Step II	: Represent input sequence with frequencies f1 and f2
Step II	: Perform the interpolation on input signal using matlab command interp.
Step IV	: Plot the input and output signal/sequence.

Flow Chart:


START
ENTER UP SAMPLING FACTOR AND INPUT FREQUENCIES
PERFORM INTERPOLATION ON INPUT SIGNAL
PLOT INPUT AND OUTPUT SEQUENCE
STOP

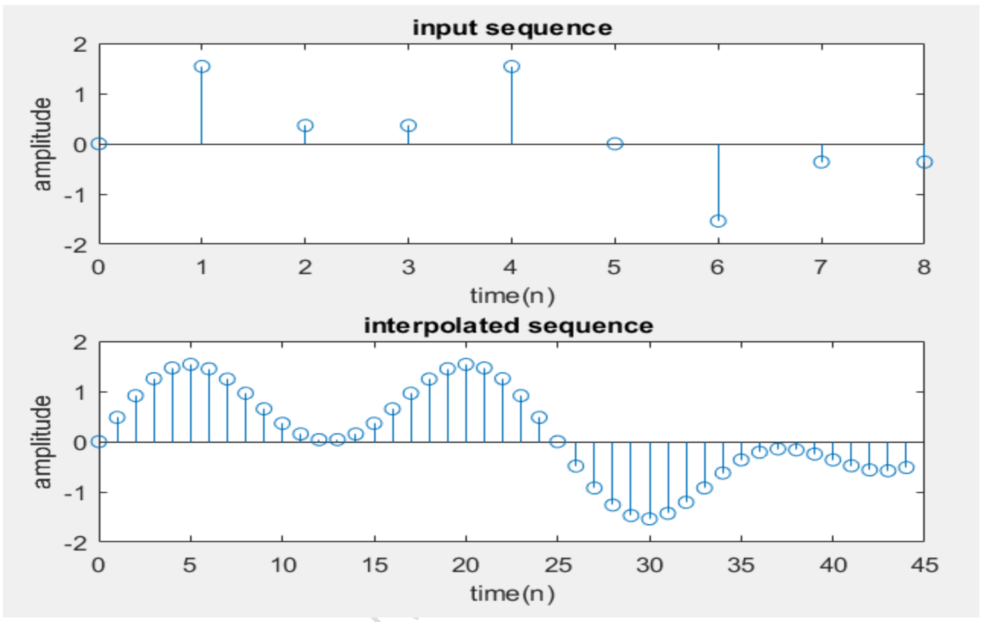
PROGRAM:

```
Clc;
Clear all;
Close all;
L=input('enter the upsampling factor');
N=input('enter the length of the input signal'); % Length should be greater than 8
f1=input('enter the frequency of first sinusodal');
f2=input('enter the frequency of second sinusodal');
n=0:N-1;
x=sin(2*pi*f1*n)+sin(2*pi*f2*n);
y=interp(x,L);
subplot(2,1,1)
stem(n,x(1:N))
title('input sequence');
xlabel('time(n)');
ylabel('amplitude');
subplot(2,1,2)
m=0:N*L-1;
stem(m,y(1:N*L))
title('output sequence ');
```

```
xlabel('time(n)');
ylabel('amplitude');
```

Expected Waveforms:

```
enter the upsampling factor =  5
enter the length of the input signal = 9
enter the frequency of first sinusoidal = 0.1
enter the frequency of second sinusoidal = 0.3
```



VIVA QUESTIONS:

1. Explain about multi rate digital signal processing.
2. List the Applications of multi rate digital signal processing.
3. Define interpolation.
4. Define decimation.
5. Define aliasing.

Exercise:

1. Perform interpolation on the input signal x(n)=cos(2*pi*30*n).

OBSEVATIONS

Output Waveforms:

EXP. NO: 15
IMPLEMENTATION OF I/D SAMPLING RATE CONVERTERS

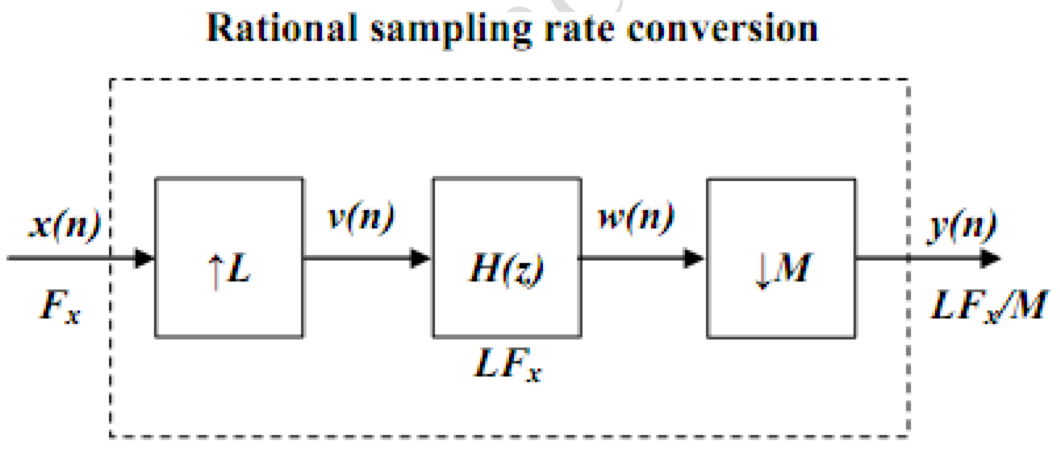
AIM: program to implement sampling rate conversion.

SOFTWARE: MATLAB

THEORY:

"Multirate" means "multiple sampling rates". A multirate DSP system uses multiple sampling rates within the system. Whenever a signal at one rate has to be used by a system that expects a different rate, the rate has to be increased or decreased, and some processing is required to do so. Therefore "Multirate DSP" refers to the art or science of changing sampling rates. "Resampling" means combining interpolation and decimation to change the sampling rate by a rational factor. Resampling is done to interface two systems with different sampling rates.

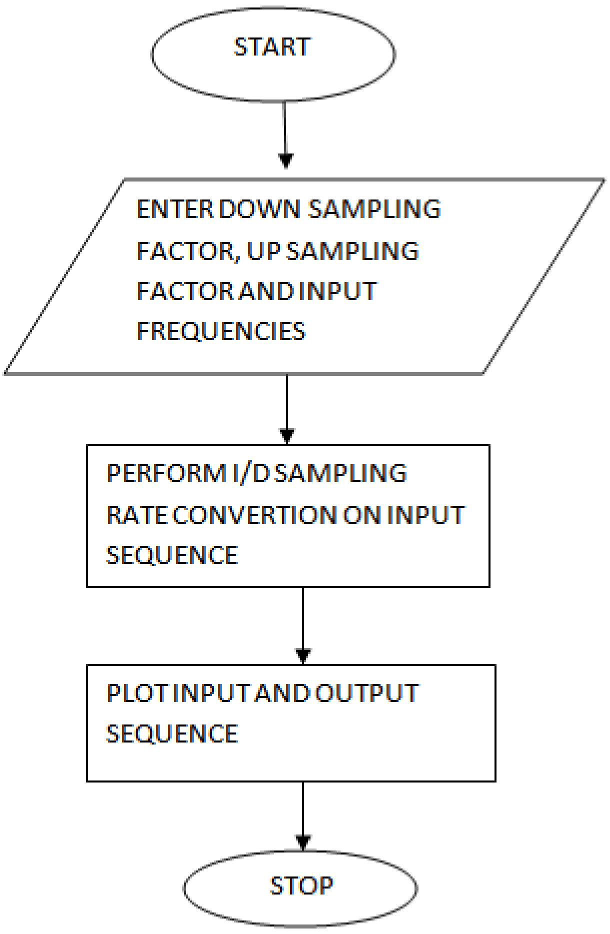
Block Diagram of Sampling rate conversion



Algorithm:

Step I	: Define up sampling factor, down sampling and input frequencies f1 and f2
Step II	: Represent input sequence with frequencies f1 and f2
Step III	: Perform I/D sampling rate conversion on input signal using matlab resample
Step IV	: Plot the input and output signal/sequence.

Flow Chart:


START
ENTER DOWN SAMPLING FACTOR, UP SAMPLING FACTOR AND INPUT FREQUENCIES
PERFORM I/D SAMPLING RATE CONVERTION ON INPUT SEQUENCE
PLOT INPUT AND OUTPUT SEQUENCE
STOP

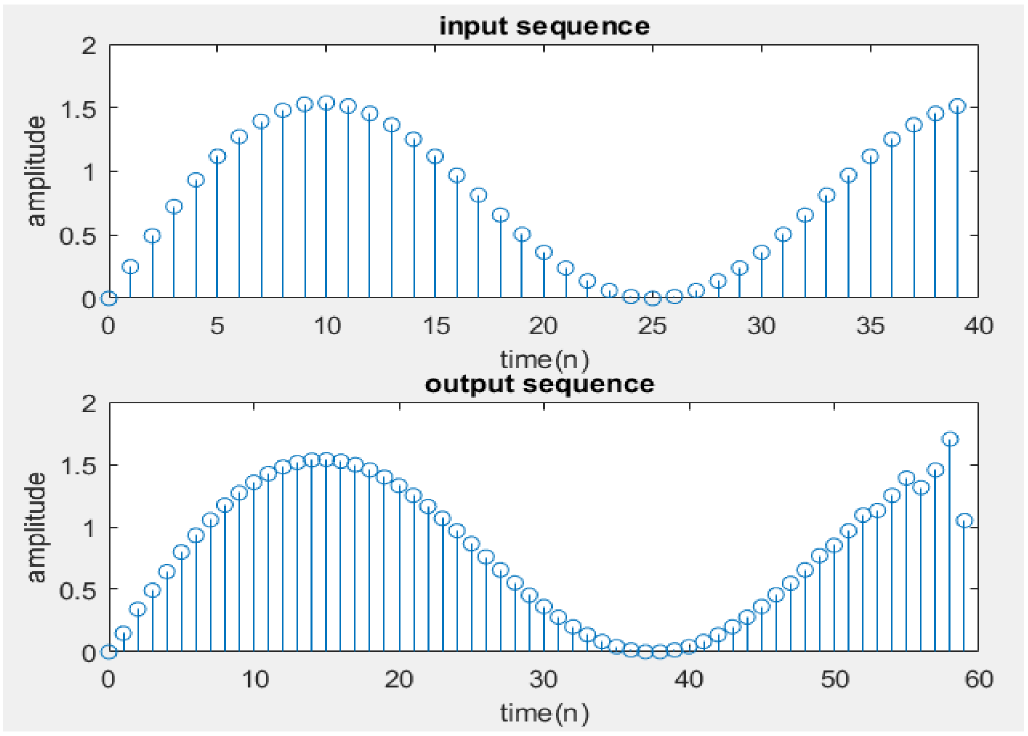
PROGRAM:

```
Clc;
Clear all;
Close all;
L=input('enter the upsampling factor');
D=input('enter the downsampling factor');
N=input('enter the length of the input signal');
f1=input('enter the frequency of first sinusodal');
f2=input('enter the frequency of second sinusodal');
n=0:N-1;
x=sin(2*pi*f1*n)+sin(2*pi*f2*n);
y=resample(x,L,D);
subplot(2,1,1)
stem(n,x(1:N))
title('input sequence');
xlabel('time(n)');
ylabel('amplitude');
subplot(2,1,2)
m=0:N*L/D-1;
```

```
stem(m,y(1:N*L/D));
title('output sequence ');
xlabel('time(n)');
ylabel('amplitude');
```

Expected Waveforms:

```
enter the upsampling factor = 3
enter the downsampling factor = 2
enter the length of the input signal = 40
enter the frequency of first sinusoidal = 0.01
enter the frequency of second sinusoidal = 0.03
```



VIVA QUESTIONS:
1. Explain about multi rate digital signal processing.
2. List the Applications of multi rate digital signal processing.
3. Define interpolation.
4. Define decimation.
5. Define aliasing.
6. What is meant anti aliasing?
7. What is the effect of anti imaging filter?
8. Define sampling rate.
9. What is the use of sampling rate convertors?
10. Explain advantages of anti aliasing filters

Exercise:

1. Perform I/D sampling rate conversion on the input signal x(n)=cos(2*pi*30*n).

OBSERVATIONS

Output Waveforms:

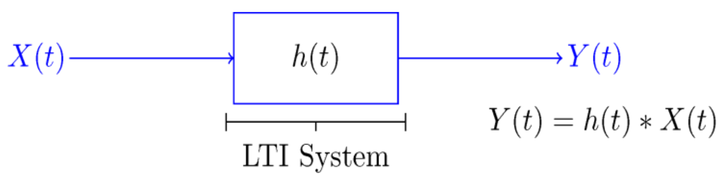
EXP. NO: 16
IMPULSE RESPONSE OF FIRST ORDER AND SECOND ORDER SYSTEMS

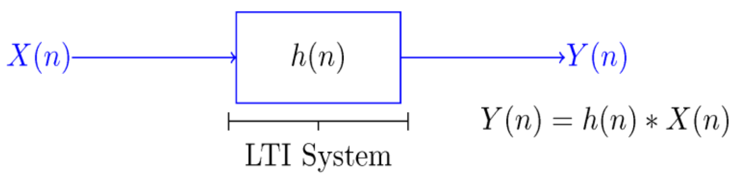
AIM: program to implement sampling rate conversion.

SOFTWARE: MATLAB

THEORY:

The response of an LTI system when an unit impulse sequence is applied at the input of the system. It is denoted by h(n).

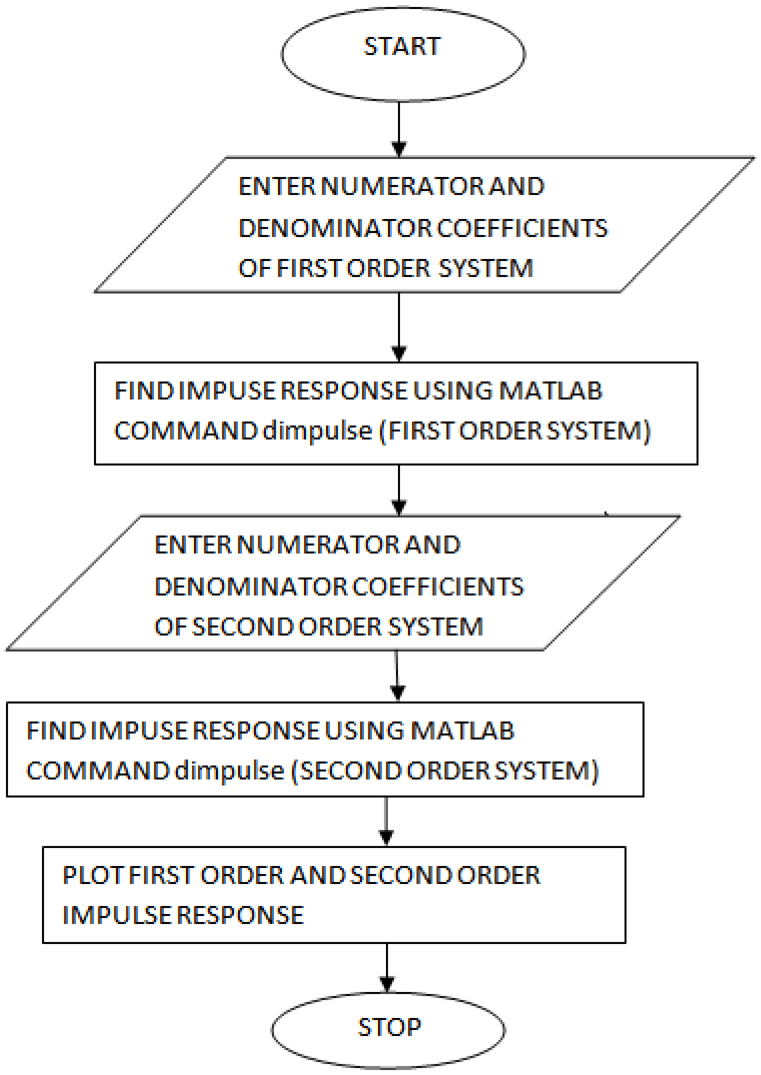




Algorithm:

Step I	: Give numerator and denominator coefficients of given first order system.
Step II	: Find impulse response of first order system using matlab command dimpulse.
Step III	: Give numerator and denominator coefficients of given second order system.
Step IV	: Find impulse response of second order system using matlab command dimpulse
Step V	: Plot the corresponding outputs.

Flow chart:

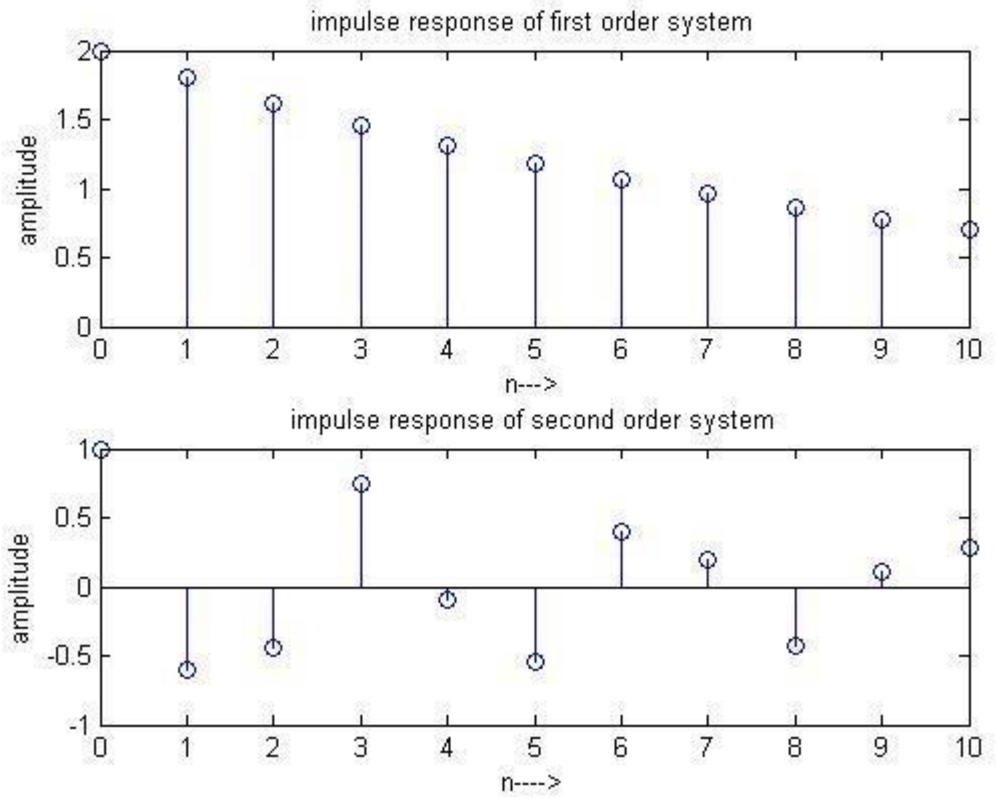

START
ENTER NUMERATOR AND DENOMINATOR COEFFICIENTS OF FIRST ORDER SYSTEM
FIND IMPUSE RESPONSE USING MATLAB COMMAND dimpulse (FIRST ORDER SYSTEM)
ENTER NUMERATOR AND DENOMINATOR COEFFICIENTS OF SECOND ORDER SYSTEM
FIND IMPUSE RESPONSE USING MATLAB COMMAND dimpulse (SECOND ORDER SYSTEM)
PLOT FIRST ORDER AND SECOND ORDER IMPULSE RESPONSE
STOP

PROGRAM:

```
clc;
close all;
clear all;
n=0:10;
%impulse response of first order system
b=[2 0 0];
a=[1 -0.9 0];
y=dimpulse(b,a,length(n));
subplot(2,1,1);
stem(n,y);
xlabel('n--->');
ylabel('amplitude');
title('impulse response of first order system');

%impulse response of second order system
```

```
b=[1 0 0];
a=[1 0.6 0.8];
y1=dimpulse(b,a,length(n));
subplot(2,1,2);
stem(n,y1);
xlabel('n --->');
ylabel('amplitude');
title('impulse response of second order system');
```

Expected Waveforms:



VIVA QUESTIONS:

1. Define impulse response?
2. Give example for first order difference equation?
3. Give example for second order difference equation?
4. Define LTI system?
5. Define convolution?

Exercise:

1. Find the impulse response of following difference equations.

   i) y(n-1)+5y(n)=0.5x(n) ii) 4y(n-2)+5y(n-1)+y(n) =0.5x(n)+2x(n-1)

OBSERVATIONS

Output Waveforms:

PART B - LIST OF EXPERIMENTS USING DSP PROCESSOR ARCHITECTURE AND INSTRUCTION SET OF DSPCHIP-TMS320C5515

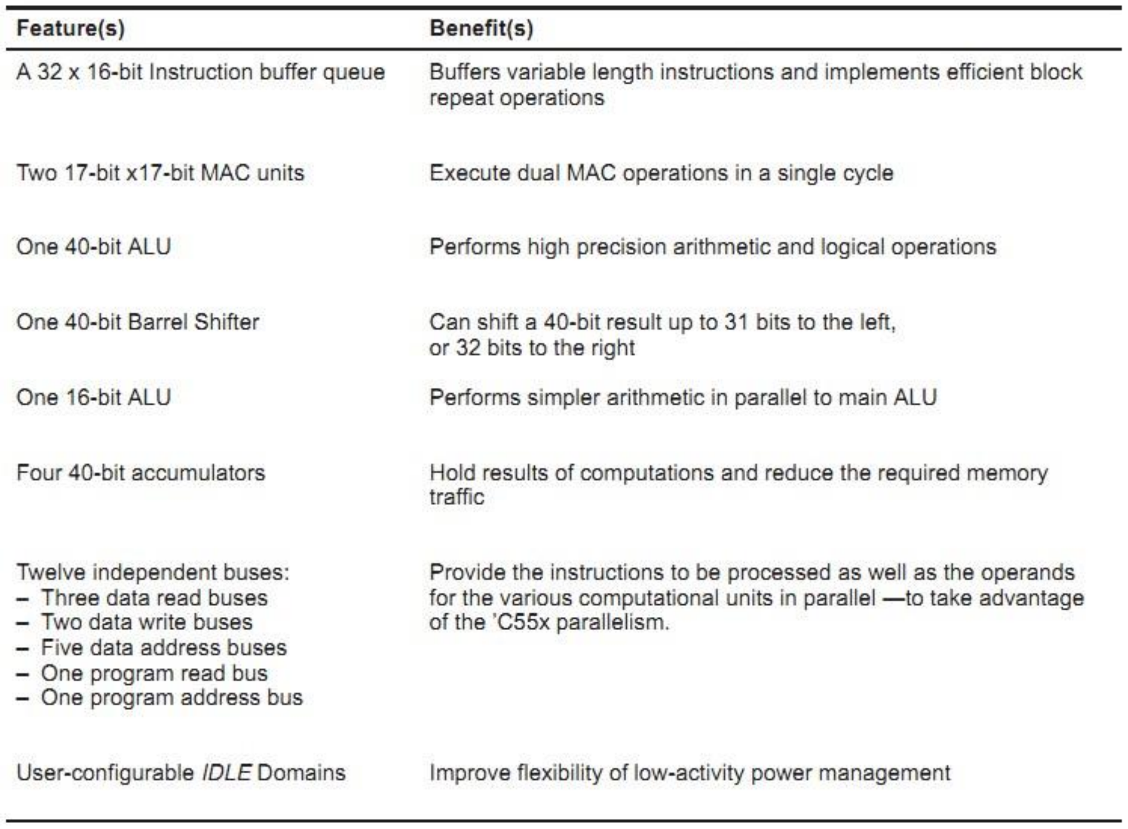
Introduction to the TMS320C55x:

The TMS320C55x digital signal processor (DSP) represents the latest generation of ’C5000 DSPs from Texas Instruments. The ’C55x is built on the proven legacy of the ’C54x and is source code compatible with the ’C54x, protecting the customer’s software investment. Following the trends set by the ’C54x, the ’C55x is optimized for power efficiency, low system cost, and best-in-class performance for tight power budgets. With core power dissipation as low as 0.05 mW/MIPS at 0.9V, and performance up to 800 MIPS (400 MHz), the TMS320C55x offers a cost-effective solution to the toughest challenges in personal and portable processing applications as well as digital communications infrastructure with restrictive power budgets. Compared to a 120-MHz ’C54x, a 300-MHz ’C55x will deliver approximately 5X higher performance and dissipate one-sixth the core power dissipation of the ’C54x. The ’C55x core’s ultra-low power dissipation of 0.05mW/MIPS is achieved through intense attention to low-power design and advanced power management techniques. The ’C55x designers have implemented an unparalleled level of power-down configurability and granularity coupled with unprecedented power management that occurs automatically and is transparent to the user.

The ’C55x core delivers twice the cycle efficiency of the ’C54x through a dual-MAC (multiply-accumulate) architecture with parallel instructions, additional accumulators, ALUs, and data registers. An advanced instruction set, a superset to that of the ’C54x, combined with expanded busing structure complements the new hardware execution units. The ’C55x continues the standard set by the ’C54x in code density leadership for lower system cost. The ’C55x instructions are variable byte lengths ranging in size from 8 bits to 48 bits. With this scalable instruction word length, the ’C55x can reduce control code size per function by up to 40% more than ’C54x. Reduced control code size means reduced memory requirements and lower system cost.

Key Features of the ’C55x

The ’C55x incorporates a rich set of features that provide processing efficiency, low-power dissipation, and ease of use. Some of these features are listed in Table



Feature(s)	Benefit(s)
A 32 x 16-bit Instruction buffer queue	Buffers variable length instructions and implements efficient block repeat operations
Two 17-bit x17-bit MAC units	Execute dual MAC operations in a single cycle
One 40-bit ALU	Performs high precision arithmetic and logical operations
One 40-bit Barrel Shifter	Can shift a 40-bit result up to 31 bits to the left, or 32 bits to the right
One 16-bit ALU	Performs simpler arithmetic in parallel to main ALU
Four 40-bit accumulators	Hold results of computations and reduce the required memory traffic
Twelve independent buses: – Three data read buses – Two data write buses – Five data address buses – One program read bus – One program address bus	Provide the instructions to be processed as well as the operands for the various computational units in parallel —to take advantage of the ’C55x parallelism.
User-configurable *IDLE* Domains	Improve flexibility of low-activity power management

Overview of the C5515 eZdsp USB Stick

The C5515 eZdsp USB Stick is an evaluation tool for the Texas Instruments TMS320C5515 Digital Signal Processor (DSP). This USB bus powered tool allows the user to evaluate the following items:

• The TMS320C5515 processor along with its peripherals

• The TLV320AIC3204 codec

• The Code Composer Studio IDETM software development tools

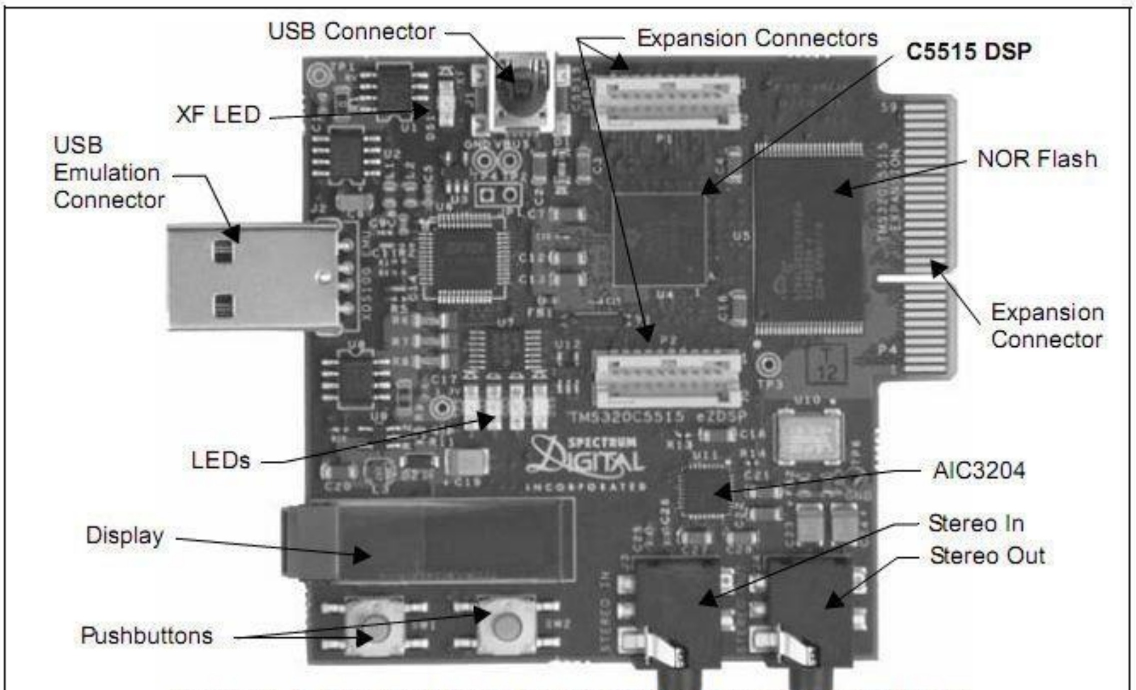


Figure 1-1, Key Features of the C5515 eZdsp USB Stick (top)

Key Features of the C5515 eZdsp USB Stick

The C5515 eZdsp USB Stick has the following features:

• Texas Instrument's TMS320C5515 Digital Signal Processor

• Texas Instruments TLV320AIC3204 Stereo Codec (stereo in, stereo out)

• Micro SD connector

• USB 2.0 interface to C5515 processor

• 32 Mb NOR flash

• I2C OLED display

• 5 user controlled LEDs

• 2 user readable push button switches

• Embedded USB XDS100 JTAG emulator

• Bluetooth board interface

• Expansion edge connector

• Power provided by USB interface

• Compatible with Texas Instruments Code Composer Studio v4

• USB extension cable

C5515 eZdsp USB Stick Block Diagram

The block diagram of the C5515 eZdsp USB Stick is shown below.

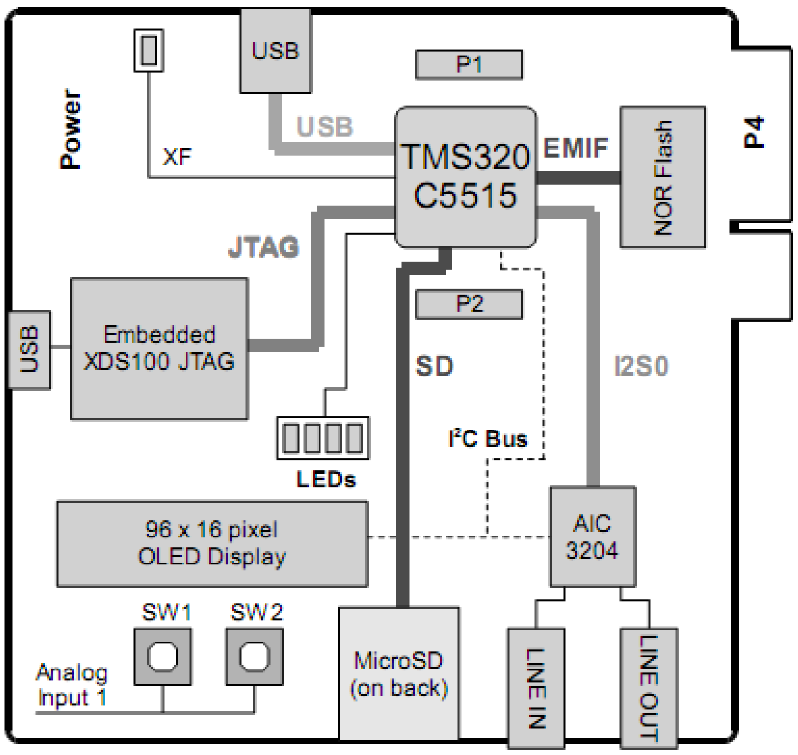


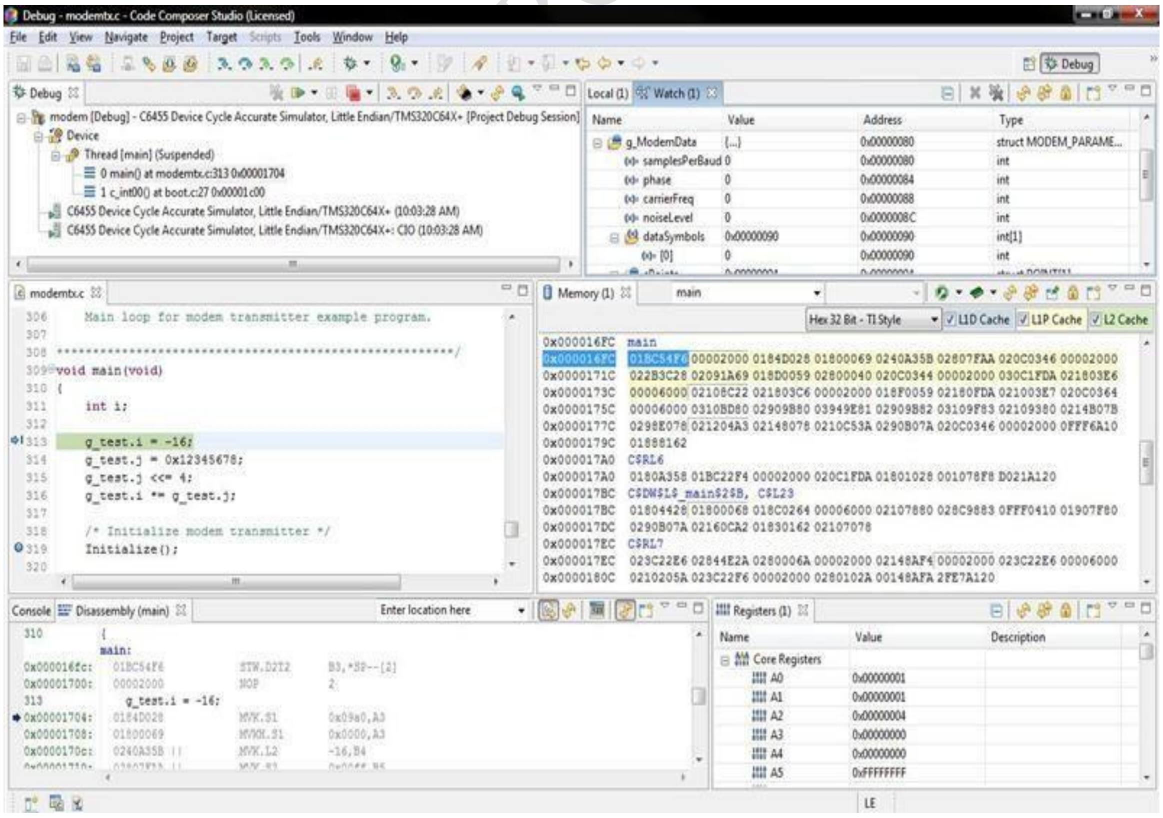
Figure 1-3, C5515 eZdsp USB Stick Block Diagram

CODE COMPOSER STUDIO

INTRODUCTION TO CODE COMPOSER STUDIO

Code Composer Studio™ (CCS or CCStudio) is the integrated development environment for TI's DSPs, microcontrollers and application processors. CCStudio includes a suite of tools used to develop and debug embedded applications. It includes compilers for each of TI's device families, source code editor, project build environment, debugger, profiler, simulators and many other features. CCStudio provides a single user interface taking users through each step of the application development flow. Familiar tools and interfaces allow users to get started faster than ever before and add functionality to their application thanks to sophisticated productivity tools.

CCStudio version 4 (CCSv4) is based on the Eclipse open source software framework. CCSv4 is based on Eclipse because it offers an excellent software framework for development environments a standard framework many embedded software vendors.
CCSv4 combines the advantages of the Eclipse software framework with advanced embedded debug capabilities from TI resulting in a compelling feature rich development environment for embedded developers.



```
modem [Debug] - C6455 Device Cycle Accurate Simulator, Little Endian/TMS320C64X+ [Project Debug Session]
  Device
    Thread [main] (Suspended)
      0 main() at modemtx.c:313 0x00001704
      1 c_int00() at boot.c:27 0x00001c00
  C6455 Device Cycle Accurate Simulator, Little Endian/TMS320C64X+ (10:03:28 AM)
  C6455 Device Cycle Accurate Simulator, Little Endian/TMS320C64X+: CIO (10:03:28 AM)
```

Name	Value	Address	Type
g_ModemData	{...}	0x00000080	struct MODEM_PARAME...
samplesPerBaud	0	0x00000080	int
phase	0	0x00000084	int
carrierFreq	0	0x00000088	int
noiseLevel	0	0x0000008C	int
dataSymbols	0x00000090	0x00000090	int[1]
[0]	0	0x00000090	int

```
    Main loop for modem transmitter example program.

******************************************************/
void main(void)
{
    int i;

    g_test.i = -16;
    g_test.j = 0x12345678;
    g_test.j <<= 4;
    g_test.i *= g_test.j;

    /* Initialize modem transmitter */
    Initialize();
```

```
0x000016FC  main
0x000016FC  01BC54F6 00002000 0184D028 01800069 0240A35B 02807FAA 020C0346 00002000
0x0000171C  022B3C28 02091A69 018D0059 02800040 020C0344 00002000 030C1FDA 021803E6
0x0000173C  00006000 02108C22 021803C6 00002000 018F0059 02180FDA 021003E7 020C0364
0x0000175C  00006000 0310BD80 02909B80 03949E81 02909B82 03109F83 02109380 0214B07B
0x0000177C  0298E078 021204A3 02148078 0210C53A 0290B07A 020C0346 00002000 0FFF6A10
0x0000179C  01888162
0x000017A0  C$RL6
0x000017A0  0180A358 01BC22F4 00002000 020C1FDA 01801028 001078F8 D021A120
0x000017BC  C$DW$L$_main$2$B, C$L23
0x000017BC  01804428 01800068 018C0264 00006000 02107880 028C9883 0FFF0410 01907F80
0x000017DC  0290B07A 02160CA2 01830162 02107078
0x000017EC  C$RL7
0x000017EC  023C22E6 02844E2A 0280006A 00002000 02148AF4 00002000 023C22E6 00006000
0x0000180C  0210205A 023C22F6 00002000 0280102A 00148AFA 2FE7A120
```

```
310          {
             main:
0x000016fc:   01BC54F6          STW.D2T2      B3,*SP--[2]
0x00001700:   00002000          NOP           2
313             g_test.i = -16;
0x00001704:   0184D028          MVK.S1        0x09a0,A3
0x00001708:   01800069          MVKH.S1       0x0000,A3
0x0000170c:   0240A35B ||       MVK.L2        -16,B4
```

Name	Value	Description
Core Registers		
A0	0x00000001	
A1	0x00000001	
A2	0x00000004	
A3	0x00000000	
A4	0x00000000	
A5	0xFFFFFFFF	

Features

Debugger

CCStudio's integrated debugger has several capabilities and advanced breakpoints to simplify development. Conditional or hardware breakpoints are based on full C expressions, local variables or registers. The advanced memory window allows you to inspect each level of memory so that you can debug complex cache coherency issues. CCStudio supports the development of complex systems with multiple processors or cores. Global breakpoints and synchronous operations provide control over multiple processors and cores.

Profiling

CCStudio's interactive profiler makes it easy to quickly measure code performance and ensure the efficient use of the target's resources during debug and development sessions. The profiler allows developers to easily profile all C/C++ functions in their application for instruction cycles or other events such as cache misses/hits, pipeline stalls and branches.

Profile ranges can be used to concentrate efforts on high-usage areas of code during optimization, helping developers produce finely-tuned code. Profiling is available for ranges of assembly, C++ or C code in any combination. To increase productivity, all profiling facilities are available throughout the development cycle.

Scripting

Some tasks such as testing need to run for hours or days without user interaction. To accomplish such a task, the IDE should be able to automate common tasks. CCStudio has a complete scripting environment allowing for the automation of repetitive tasks such as testing and performance benchmarking. A separate scripting console allows you to type commands or to execute scripts within the IDE.

Image Analysis and Visualization

CCStudio has many image analysis and graphic visualization. It includes the ability to graphically view variables and data on displays which can be automatically refreshed. CCStudio can also look at images and video data in the native format (YUV, RGB) both in the host PC or loaded in the target board.

Compiler

TI has developed C/C++ compilers specifically tuned to maximize the processor's usage and performance. TI compilers use a wide range of classical, application-oriented, and sophisticated device-specific optimizations that are tuned to all the supported architectures.

Some of these optimizations include:

• Common sub-expression elimination

• Software pipelining

• Strength Reduction

• Auto increment addressing

• Cost-based register allocation

• Instruction predication

• Hardware looping

• Function In-lining

• Vectorization

TI compilers also perform program level optimizations that evaluate code performance at the application level. With the program level view, the compiler is able to generate code similar to an assembly program developer who has the full system view. This application level view is leveraged by the compiler to make trade-offs that significantly increase the processor performance.

The TI ARM and Microcontroller C/C++ compilers are specifically tuned for code size and control code efficiency. They offer industry leading performance and compatibility.

Simulation

Simulators provide a way for users to begin development prior to having access to a development board. Simulators also have the benefit of providing enhanced visibility into application performance and behavior. Several simulator variants are available allowing users to trade off cycle accuracy, speed and peripheral simulation, with some simulators being ideally suited to algorithm benchmarking and others for more detailed system simulation. Hardware Debugging (Emulation)

TI devices include advanced hardware debugging capabilities. These capabilities include:

• IEEE 1149.1 (JTAG) and Boundary Scan

• Non-intrusive access to registers and memory

• Real-time mode which provides for the debugging of code that interacts with interrupts that must not be disabled. Real-time mode allows you to suspend background code at break events while continuing to execute time-critical interrupt service routines.

• Multi-core operations such as synchronous run, step, and halt. This includes crosscore triggering, which provides the ability to have one core trigger other cores to halt. Advanced Event Triggering (AET) which is available on selected devices, allows a user to halt the CPU or trigger other events based on complex events or sequences such as invalid data or program memory accesses. It can non-intrusively measure performance and count system events (for example, cache events).

CCStudio provides Processor Trace on selected devices to help customers find previously "invisible" complex real-time bugs. Trace can detect the really hard to find bugs – race conditions between events, intermittent real-time glitches, crashes from stack overflows, runaway code and false interrupts without stopping the processor. Trace is a completely nonintrusive debug method that relies on a debug unit inside the processor so it does not interfere or change the application's real-time behavior. Trace can fine tune code performance and cache optimization of complex switch intensive multi-channel applications. Processor Trace supports the export of program, data, timing and selected processor and system events/interrupts. Processor Trace can be exported either to an XDS560 Trace external JTAG emulator, or on selected devices, to an on chip buffer Embedded Trace Buffer (ETB).

Real time operating system support

CCSv4 comes with two versions of TI's real time operating system:

• DSP/BIOS 5.4x is a real-time operating system that provides pre-emptive multitasking services for DSP devices. Its services include ISR dispatching, software interrupts, semaphores, messages, device I/O, memory management, and power management. In addition, DSP/BIOS 5.x also includes debug instrumentation and tooling, including low-overhead print and statistics gathering.

• BIOS 6.x is an advanced, extensible real-time operating system that supports ARM926, ARM Cortex M3, C674x, C64x+, C672x, and 28x-based devices. It offers numerous kernel and debugging enhancements not available in DSP/BIOS 5.x, including faster, more flexible memory management, events, and priority-inheritance mutexes.

Note: BIOS 6.x includes a DSP/BIOS 5.x compatibility layer to support easy migration of application source code.

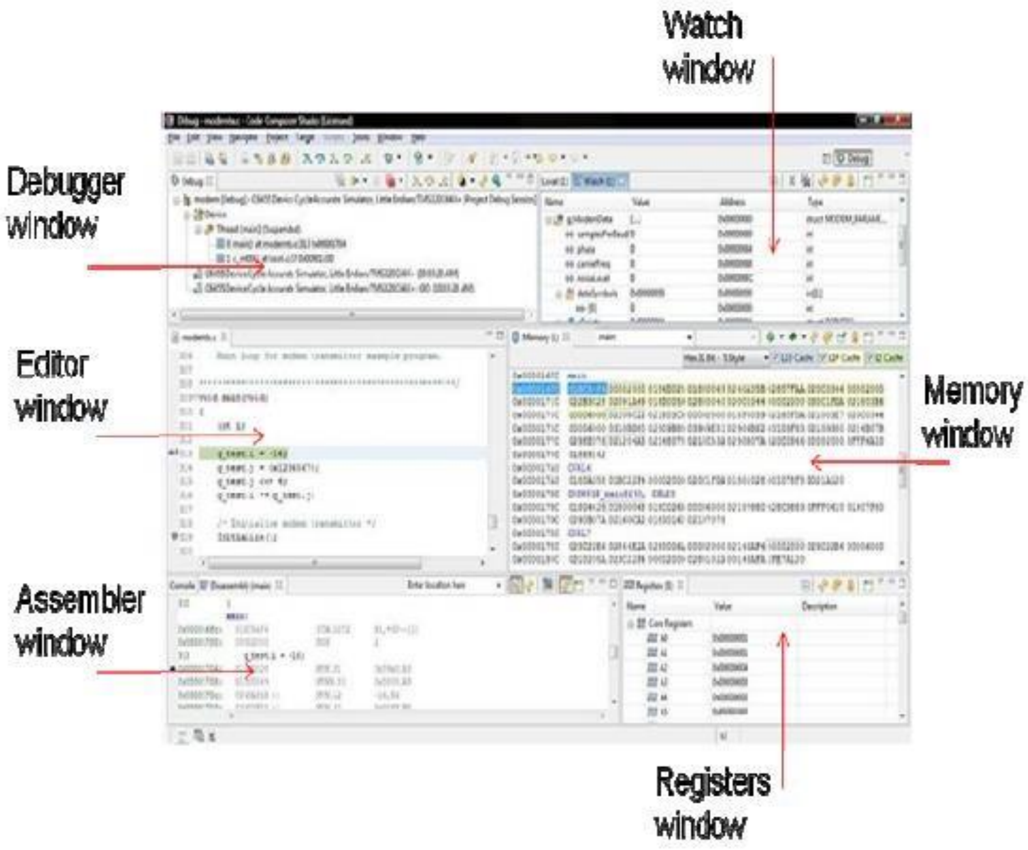
Step 1:

Open the code composer studio (CCSV4) and give a name to workspace and store it in the default path itself.

Note: don't assign other than default path unless you are familiar with eclipse frame work based CCSV4

Step 2:

Project windows overview



EXP.NO: 1
COMPUTATION OF N- POINT DFT OF A GIVEN SEQUENCE

Aim: To compute the N (=4/8/16) point DFT of the given sequence

EQUIPMENTS:

- Host (PC) with windows (95/98/Me/XP/NT/2000).
- TMS320C5515 DSP Starter Kit (DSK).

Theory:

The N point DFT of discrete time signal x[n] is given by the equation

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

Where N is chosen such that $N \geq L$, where L=length of x[n]. To implement using C program we use the expression $e^{\frac{-j2\pi kn}{N}} = \cos\left(\frac{2\pi kn}{N}\right) - j\sin\left(\frac{2\pi kn}{N}\right)$ and allot memory space for real and imaginary parts of the DFT X(k)

Program

```
//dft.c N-point DFT of sequence read from lookup table
#include <stdio.h>
#include <math.h>
#define PI 3.14159265358979
#define N 64
#define TESTFREQ 10000.0
#define SAMPLING_FREQ 64000.0
typedef struct
{
float real;
float imag;
} COMPLEX;
float x1[N],y1[N];
COMPLEX samples[N];
void dft(COMPLEX *x)
{
COMPLEX result[N];
int k,n,i;
for (k=0 ; k<N ; k++)
{
result[k].real=0.0;
result[k].imag = 0.0;
for (n=0 ; n<N ; n++)
{
result[k].real += x[n].real*cos(2*PI*k*n/N) + x[n].imag*sin(2*PI*k*n/N);
result[k].imag += x[n].imag*cos(2*PI*k*n/N) - x[n].real*sin(2*PI*k*n/N);
}
}
for (k=0 ; k<N ; k++)
{
x[k] = result[k];
}
printf("output");
for (i = 0 ; i < N ; i++) //compute magnitude
{
x1[i] = (int)sqrt(result[i].real*result[i].real + result[i].imag*result[i].imag);
printf("\n%d = %f",i,x1[i]);
}
}
void main() //main function
{
int n;
for(n=0 ; n<N ; n++)
{
y1[n] = samples[n].real = cos(2*PI*TESTFREQ*n/SAMPLING_FREQ);
samples[n].imag = 0.0;
printf("\n%d = %f",n,samples[n].real);
}
printf("real input data stored in array samples[]\n");
printf("\n"); // place breakpoint here
dft(samples); //call DFT function
printf("done!\n");
 }
```

Code Flow:

• Step 1 - Select no. of points for DFT(Eg: 64)

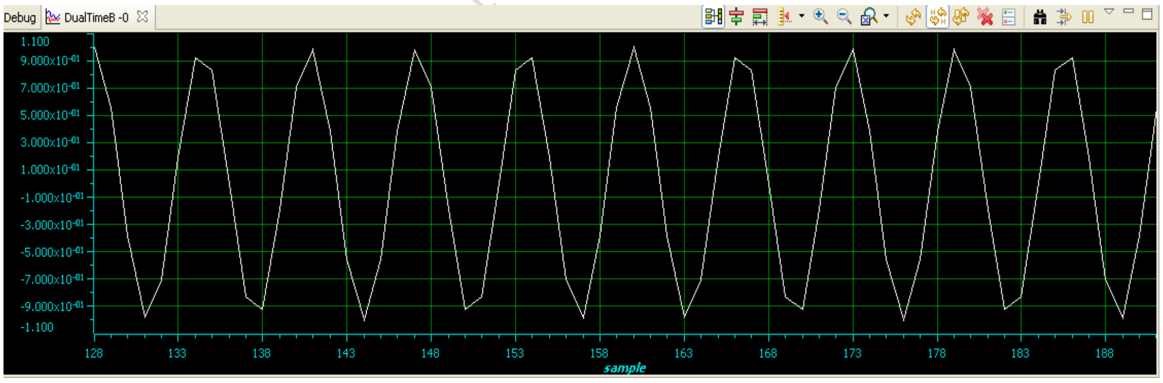
• Step 2 – Generate a sine wave of frequency ‘f ‘ (eg: 10 Hz with a sampling rate = No. of Points of DFT(eg. 64)) using math library function.

• Step 3 - Take sampled data and apply DFT algorithm.

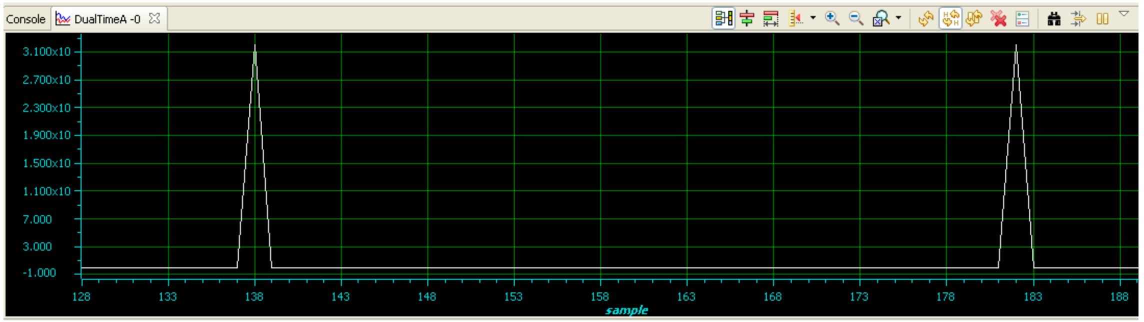
Execution Procedure:

- Open CCstudio setup
- Go to File Menu , select Import option.
- In the Import Window under CCs choose Existing CCS/CCE Eclipse project then next.
- In Select root Directory Browse for project file where it is located.
- Select DFT Project folder and Finish it.
- Now Connect the DSP Kit to PC and Launch it.(Follow the above given manual procedure from Step 46 to 51)
- Give Right Click on Your Dft.out file under Binaries and select Load program Option.
- Now Go to Target select Run.
- From Tools select Graph(Dual Time) , give properties and select OK.

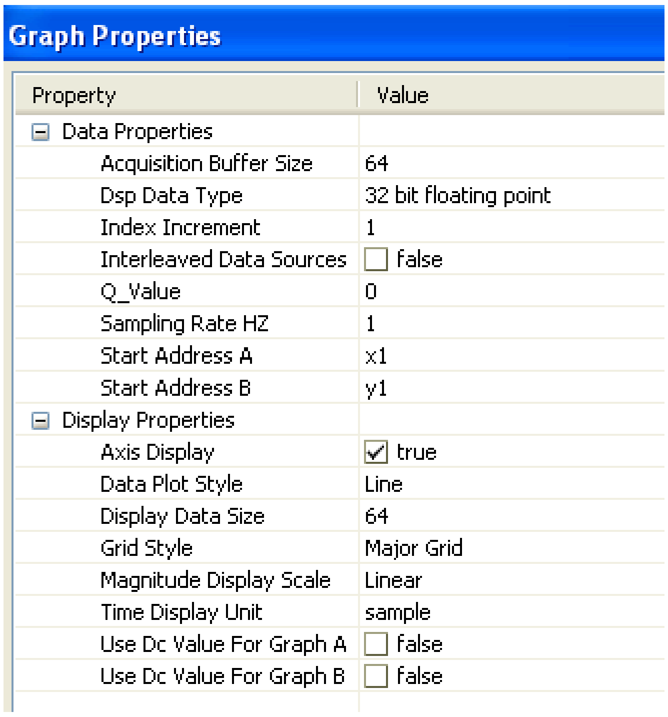
Input Signal:



Output Signal :



Graph Properties:



Graph Properties

Property	Value
⊟ Data Properties	
Acquisition Buffer Size	64
Dsp Data Type	32 bit floating point
Index Increment	1
Interleaved Data Sources	☐ false
Q_Value	0
Sampling Rate HZ	1
Start Address A	x1
Start Address B	y1
⊟ Display Properties	
Axis Display	☑ true
Data Plot Style	Line
Display Data Size	64
Grid Style	Major Grid
Magnitude Display Scale	Linear
Time Display Unit	sample
Use Dc Value For Graph A	☐ false
Use Dc Value For Graph B	☐ false

OBSERVATIONS

Output Waveforms:

EXP.NO: 2
IMPLEMENTATION OF FFT OF GIVEN SEQUENCE

AIM: To compute the FFT of the given sequence

EQUIPMENTS:

1. Host (PC) with windows (95/98/Me/XP/NT/2000).
2. TMS320C5515 DSP Starter Kit (DSK).

FFT Algorithm

The FFT has a fairly easy algorithm to implement, and it is shown step by step in the list below. This version of the FFT is the Decimation in Time Method

1. Pad input sequence, of N samples, with Zero's until the number of samples is the nearest power of two.

e.g. 500 samples are padded to 512 (2^9)

2. Bit reverse the input sequence.

e.g. 3 = 011 goes to 110 = 6

3. Compute (N / 2) two sample DFT's from the shuffled inputs. See "Shuffled Inputs"
4. Compute (N / 4) four sample DFT's from the two sample DFT's. See "Shuffled Inputs"
5. Compute (N / 2) eight sample DFT's from the four sample DFT's. See "Shuffled Inputs"
6. Until the all the samples combine into one N-sample DFT

PROGRAM:

Main.c

```
#include "usbstk5515.h"
#include <math.h>
#include <stdio.h>
#define PTS 64 //no of points for FFT
#define PI 3.14159265358979
typedef struct {float real,imag;} COMPLEX;
void FFT(COMPLEX *Y, int n); //FFT prototype
float iobuffer[PTS]; //as input and output buffer
float x1[PTS]; //intermediate buffer
short i; //general purpose index variable
short buffercount = 0; //number of new samples in iobuffer
short flag = 0; //set to 1 by ISR when iobuffer full
COMPLEX w[PTS]; //twiddle constants stored in w
COMPLEX samples[PTS]; //primary working buffer
void main(void)
{
for(i=0;i<PTS;i++)
{
iobuffer[i]=0;
x1[i]=0;
}
printf("\n input");
for (i = 0 ; i<PTS ; i++) // set up twiddle constants in w
{
w[i].real = cos(2*PI*i/(PTS*2.0)); //Re component of twiddle constants
w[i].imag =-sin(2*PI*i/(PTS*2.0)); //Im component of twiddle constants
}
for (i = 0 ; i < PTS ; i++) //swap buffers
{
iobuffer[i] = sin(2*PI*10*i/64.0);/*10- > freq, 64 -> sampling freq*/
printf("\n%d = %f",i,iobuffer[i]);
samples[i].real=0.0;
samples[i].imag=0.0;
}
for (i = 0 ; i < PTS ; i++) //swap buffers
{
samples[i].real=iobuffer[i]; //buffer with new data
}
for (i = 0 ; i < PTS ; i++)
samples[i].imag = 0.0; //imag components = 0
FFT(samples,PTS); //call function FFT.c
printf("\n output");
for (i = 0 ; i < PTS ; i++) //compute magnitude
{
x1[i] = sqrt(samples[i].real*samples[i].real +
samples[i].imag*samples[i].imag);
printf("\n%d = %f",i,x1[i]);
}
} //end of main
```

fft.c

```
#define PTS 64 //# of points for FFT
typedef struct {float real,imag;} COMPLEX;
extern COMPLEX w[PTS]; //twiddle constants stored in w
void FFT(COMPLEX *Y, int N) //input sample array, # of points
{
COMPLEX temp1,temp2; //temporary storage variables
int i,j,k; //loop counter variables
int upper_leg, lower_leg; //index of upper/lower butterfly leg
int leg_diff; //difference between upper/lower leg
int num_stages = 0; //number of FFT stages (iterations)
int index, step; //index/step through twiddle constant
i = 1; //log(base2) of N points= # of stages
do
{
num_stages +=1;
i = i*2;
}while (i!=N);
leg_diff = N/2; //difference between upper&lower legs
step = (PTS*2)/N; //step between values in twiddle.h
for (i = 0;i < num_stages; i++) //for N-point FFT
{
index = 0;
for (j = 0; j < leg_diff; j++)
{
for (upper_leg = j; upper_leg < N; upper_leg += (2*leg_diff))
{
lower_leg = upper_leg+leg_diff;
temp1.real = (Y[upper_leg]).real + (Y[lower_leg]).real;
temp1.imag = (Y[upper_leg]).imag + (Y[lower_leg]).imag;
temp2.real = (Y[upper_leg]).real - (Y[lower_leg]).real;
temp2.imag = (Y[upper_leg]).imag - (Y[lower_leg]).imag;
(Y[lower_leg]).real = temp2.real*(w[index]).real
-temp2.imag*(w[index]).imag;
(Y[lower_leg]).imag = temp2.real*(w[index]).imag
+temp2.imag*(w[index]).real;
(Y[upper_leg]).real = temp1.real;
(Y[upper_leg]).imag = temp1.imag;
}
index += step;
}
leg_diff = leg_diff/2;
step *= 2;
}
j = 0;
for (i = 1; i < (N-1); i++) //bit reversal for resequencing data
{
k = N/2;
while (k <= j)
{
j = j - k;
k = k/2;
}
j = j + k;
if (i<j)
{
temp1.real = (Y[j]).real;
temp1.imag = (Y[j]).imag;
(Y[j]).real = (Y[i]).real;
(Y[j]).imag = (Y[i]).imag;
(Y[i]).real = temp1.real;
(Y[i]).imag = temp1.imag;
}
}
return;
}
```

Code Flow:

• Step 1 - Select no. of points for FFT(Eg: 64)

• Step 2 – Generate a sine wave of frequency ‘f ‘ (eg: 10 Hz with a sampling rate = No. of Points of FFT(eg. 64)) using math library function.

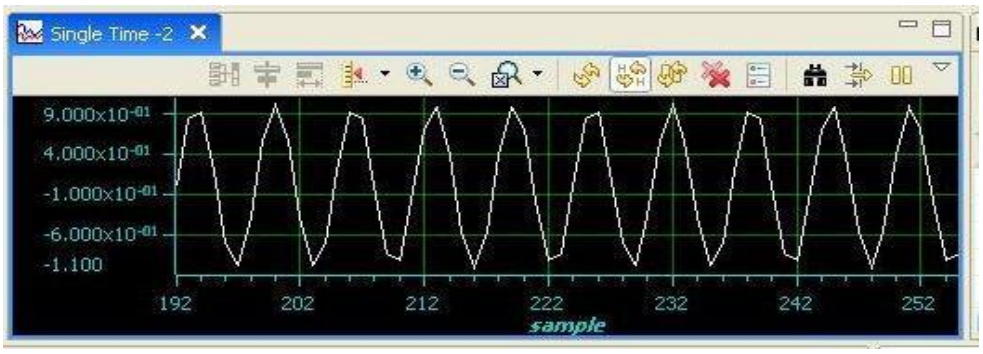
• Step 3 - Take sampled data and apply FFT algorithm.

Execution Procedure:

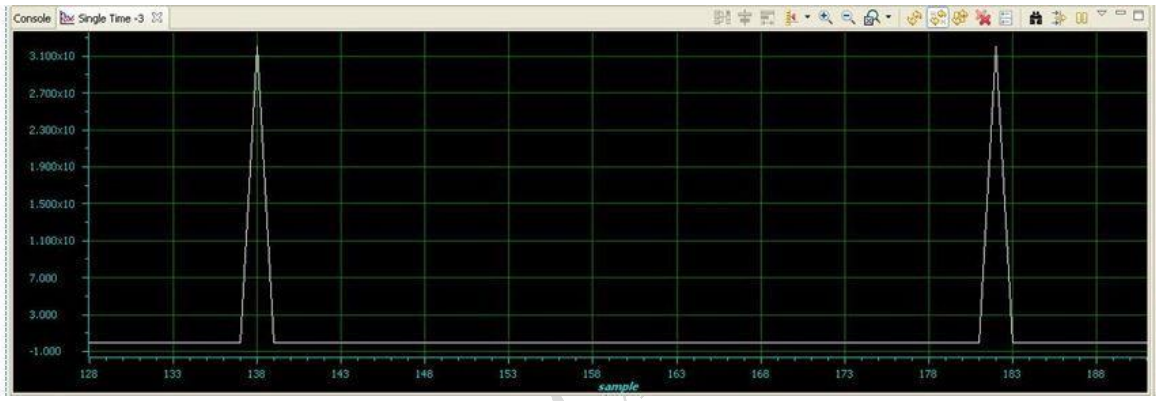
- Open CCstudio setup
- Go to File Menu , select Import option.
- In the Import Window under CCs choose Existing CCS/CCE Eclipse project then next.
- In Select root Directory Browse for project file where it is located.
- Select FFT Project folder and Finish it.
- Now Connect the DSP Kit to PC and Launch it.(Follow the above given manual procedure from Step 46 to 51)
- Give Right Click on Your fft.out file under Binaries and select Load program Option.
- Now Go to Target select Run.
- From Tools select Graph(Dual Time) , give properties and select OK.

Result :

Input Signal:



Output Signal:



OBSEVATIONS

Output Waveforms:

EXP.NO: 3
POWER SPECTRUM

Aim: To verify the power spectrum using DSP processor.

Equipments required:

1. Host (PC) with windows (95/98/Me/XP/NT/2000).
2. TMS320C5515 DSP Starter Kit (DSK).

Program:

Main.c

```
#include "usbstk5515.h"
#include <math.h>
#include <stdio.h>
#define PTS 64 //# of points for FFT
#define PI 3.14159265358979
typedef struct {float real,imag;} COMPLEX;
void FFT(COMPLEX *Y, int n); //FFT prototype
void apply_fft(void);
float iobuffer[PTS]; //as input and output buffer
float x1[PTS]; //intermediate buffer
float x[PTS];
short i; //general purpose index variable
short buffercount = 0; //number of new samples in iobuffer
short flag = 0; //set to 1 by ISR when iobuffer full
COMPLEX w[PTS]; //twiddle constants stored in w
COMPLEX samples[PTS]; //primary working buffer
void main(void)
{
float sum=0.0 ;
int n,k,i;
for (i = 0 ; i<PTS ; i++) // set up twiddle constants in w
{
w[i].real = cos(2*PI*i/(PTS*2.0)); /*Re component of twiddle constants*/
w[i].imag =-sin(2*PI*i/(PTS*2.0)); /*Im component of twiddle constants*/
}
/****************Input Signal X(n) *************************/
for(i=0;i<PTS;i++)
{
x[i] = sin(2*PI*5*i/PTS);
// Signal x(Fs)=sin(2*pi*f*i/Fs);
samples[i].real=0.0;
samples[i].imag=0.0;
}
/********************Auto Correlation of X(n)=R(t) ***********/
for(n=0;n<PTS;n++)
{
sum=0;
for(k=0;k<PTS-n;k++)
{
sum=sum+(x[k]*x[n+k]); // Auto Correlation R(t)
}
iobuffer[n] = sum;
}
/********************** FFT of R(t) ***********************/
for (i = 0 ; i < PTS ; i++) //swap buffers
{
samples[i].real=iobuffer[i]; //buffer with new data
}
for (i = 0 ; i < PTS ; i++)
samples[i].imag = 0.0; //imag components = 0
FFT(samples,PTS); //call function FFT.c
/******************** PSD ********************/
for (i = 0 ; i < PTS ; i++) //compute magnitude
{
x1[i] = sqrt(samples[i].real*samples[i].real + samples[i].imag*samples[i].imag);
}
}
```

FFT.c

Refer previous FFT experiment

Code Flow:

• Step 1 - Select no. of points for FFT(E.g.: 64)

• Step 2 – Generate a sine wave of frequency 'f '(e.g.: 10 Hz with a sampling rate = No. of Points of FFT (e.g. 64)) using math library function.

• Step 3 - Compute the Auto Correlation of Sine wave

• Step4 - Take output of auto correlation, apply FFT algorithm.

Execution Procedure:

_ Open CCstudio setup

_ Go to File Menu, select Import option.

_ In the Import Window under CCs choose Existing CCS/CCE Eclipse project then next.

_ In Select root Directory Browse for project file where it is located.

_ Select PSD Project folder and Finish it.

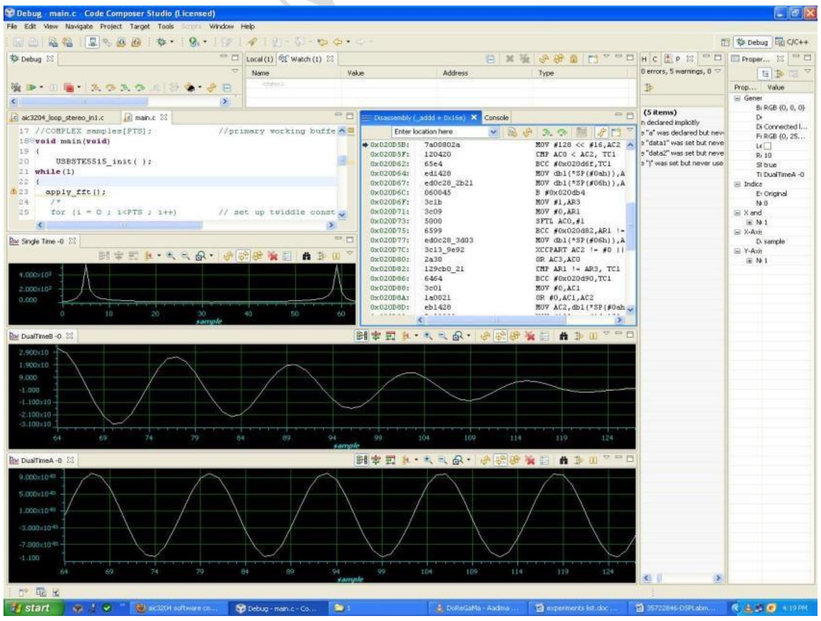
_ Now Connect the DSP Kit to PC and Launch it.(Follow the above given manual procedure from Step 46 to 51)

_ Give Right Click on Your psd.out file under Binaries and select Load program Option.

_ Now Go to Target select Run.

_ From Tools select Graph(Dual Time and single Time) , give properties and select OK.

Output:



Debug - main.c - Code Composer Studio (Licensed)
File Edit View Navigate Project Target Tools Scripts Window Help
Debug
Local (1) Watch (1)
Name Value Address Type
aic3204_loop_stereo_in1.c
main.c
Disassembly (_addd + 0x16e)
Console
Enter location here
Single Time -0
DualTimeB -0
DualTimeA -0
sample
Debug C/C++
Proper...
Prop... Value
start

```
//COMPLEX samples[PTS];                    //primary working buffe
void main(void)
{
    USBSTK5515_init( );
while(1)
{
  apply_fft();
   /*
   for (i = 0 ; i<PTS ; i++)           // set up twiddle const
```

```
0x020D5B:  7a00802a     MOV #128 << #16,AC2
0x020D5F:  120420       CMP AC0 < AC2, TC1
0x020D62:  65e4         BCC #0x020d6f,TC1
0x020D64:  ed1428       MOV dbl(*SP(#0ah)),A
0x020D67:  ed0c28_2b21  MOV dbl(*SP(#06h)),A
0x020D6C:  060045       B #0x020db4
0x020D6F:  3c1b         MOV #1,AR3
0x020D71:  3c09         MOV #0,AR1
0x020D73:  5000         SFTL AC0,#1
0x020D75:  6599         BCC #0x020d82,AR1 !=
0x020D77:  ed0c28_3d03  MOV dbl(*SP(#06h)),A
0x020D7C:  3c13_9e92    XCCPART AC2 != #0 ||
0x020D80:  2a30         OR AC3,AC0
0x020D82:  129cb0_21    CMP AR1 != AR3, TC1
0x020D86:  6464         BCC #0x020d90,TC1
0x020D88:  3c01         MOV #0,AC1
0x020D8A:  1a0021       OR #0,AC1,AC2
0x020D8D:  eb1428       MOV AC2,dbl(*SP(#0ah
```

0 errors, 5 warnings, 0

(5 items)
n declared implicitly
e "a" was declared but nev
e "data1" was set but neve
e "data2" was set but neve
e "j" was set but never use

Prop...	Value
Gener	
B	RGB {0, 0, 0}
D	
D	Connected l...
F	RGB {0, 25...
L	☐
R	10
S	true
T	DualTimeA -0
Indice	
E	Original
N	0
X and	
N	1
X-Axi	
D	sample
Y-Axi	
N	1

OBSEVATIONS

Output Waveforms:

EXP.NO: 4

IMPLEMENTATION OF LP FIR FILTER FOR GIVEN SEQUENCE & IMPLEMENTATION OF HP FIR FILTER FOR GIVEN SEQUENCE

Aim: The aim of this laboratory exercise is to design and implement a Digital FIR Filter & observe its frequency response. In this experiment we design a simple FIR filter so as to stop or attenuate required band of frequencies components and pass the frequency components, which are outside the required band.

EQUIPMENTS:

- Host (PC) with windows(95/98/Me/XP/NT/2000).
- TMS320C5515 DSP Starter Kit (DSK).

Finite Impulse Response (FIR) Filter: The FIR filters are of non-recursive type, where by the present output sample is depending on the present input sample and previous input samples.

The transfer function of a FIR causal filter is given by,

$$H(z) = \sum_{n=0}^{N-1} h(n)Z^{-n}$$

Where h(n) is the impulse response of the filter.

The Fourier transform of h(n) is

$$H(e^{jw}) = \sum_{n=0}^{N-1} h(n)e^{-jwn}$$

In the design of FIR filters most commonly used approach is using windows.

The desired frequency response Hd(ejw) of a filter is periodic in frequency and can be expanded in Fourier series. The resultant series is given by,

$$h_d(n) = (1/2\pi)\int_{-\pi}^{\pi} H(e^{jw})e^{jwn}\,dw$$

And known as Fourier coefficients having infinite length. One possible way of obtaining FIR filter is to truncate the infinite Fourier series at n = ± [(N-1)/2]

Where N is the length of the desired sequence.

The Fourier coefficients of the filter are modified by multiplying the infinite impulse response with a finite weighing sequence w(n) called a window.

Where w(n) = w(-n) ≠ 0 for |n| ≤ [(N-1)/2]

= 0 for |n| > [(N-1)/2]

After multiplying w(n) with hd(n), we get a finite duration sequence h(n) that satisfies the desired magnitude response,

h(n) = hd(n) w(n) for |n| ≤ [(N-1)/2]

= 0 for |n| > [(N-1)/2]

The frequency response H(ejw) of the filter can be obtained by convolution of Hd(ejw) and

W(ejw) is given by,

$$H(e^{jw}) = (1/2\pi)\int_{-\pi}^{\pi} H_d(e^{j\theta})e^{jw-\theta}\,d\theta$$

H(ejw) = Hd(ejw) * W(ejw).

DESIGNING AN FIR FILTER :

Following are the steps to design linear phase FIR filters Using Windowing Method.

I. Clearly specify the filter specifications.

Eg: Order= 30; Sampling Rate= 8000 samples/sec; Cut off Freq. = 400 Hz.

II. Compute the cut-off frequency Wc

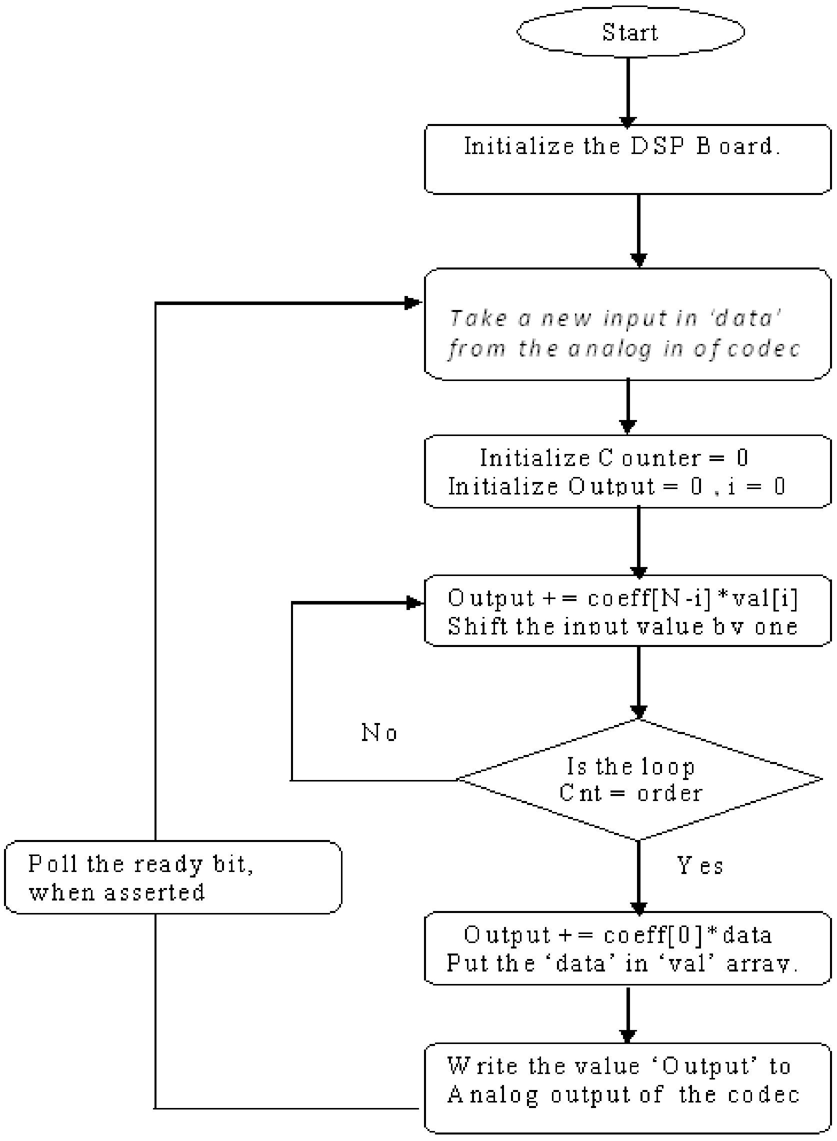
Eg: Wc = 2*pie* fc / Fs= 2*pie* 400/8000= 0.1*pie

III. Compute the desired Impulse Response h d (n) using particular Window.

Eg: b_rect1=fir1(order, Wc , 'high',boxcar(31));

IV. Convolve input sequence with truncated Impulse Response x (n)*h (n).

FLOW CHART TO IMPLEMENT FIR FILTER:


Start
Initialize the DSP Board.
Take a new input in 'data' from the analog in of codec
Initialize Counter = 0
Initialize Output = 0 , i = 0
Output += coeff[N-i]*val[i]
Shift the input value by one
No
Is the loop Cnt = order
Yes
Output += coeff[0]*data
Put the 'data' in 'val' array.
Write the value 'Output' to Analog output of the codec
Poll the ready bit, when asserted

Coefficients for FIR Low Pass Kaiser filter:

Cutoff freq: 8khz, sampling freq: 24khz

#define N 82 //length of filter

short h[N]= { 0, 1, -1, 0, 1, -1, 0, 1, -1, 0, 1, -1, 0, 1, -1, 0, 1, -1, 0, 2, -2, 0, 2, -2, 0, 2, -2, 0, 3, -3, 0, 4, -4, 0, 5, -6, 0, 10, -14, 0, 70, 70, 0, -14, 10, 0, -6, 5, 0, -4, 4, 0, -3, 3, 0, -2, 2, 0, -2, 2, 0, -2, 2, 0, -1, 1, 0, -1, 1, 0, -1, 1, 0, -1, 1, 0, -1, 1, 0, -1, 1, 0 };

Coefficients for FIR Low Pass rectangular filter:

Cutoff freq: 8khz, sampling freq: 24khz

#define N 82 //length of filter

short h[N]= { 0, 1, -1, 0, 1, -1, 0, 1, -1, 0, 1, -1, 0, 1, -1, 0, 1, -1, 0, 2, -2, 0, 2, -2, 0, 2, -2, 0, 3, -3, 0, 4, -4, 0, 5, -6, 0, 10, -14, 0, 70, 70, 0, -14, 10, 0, -6, 5, 0, -4, 4, 0, -3, 3, 0, -2, 2, 0, -2, 2, 0, -2, 2, 0, -1, 1, 0, -1, 1, 0, -1, 1, 0, -1, 1, 0, -1, 1, 0, -1, 1, 0 };

Coefficients for FIR Low Pass triangular filter:

Cutoff freq: 8khz, sampling freq: 24khz

#define N 82 //length of filter

short h[N]= { 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 1, -1, 0, 1, -1, 0, 1, -1, 0, 1, -2, 0, 2, -2, 0, 3, -3, 0, 5, -6, 0, 9, -13, 0, 70, 70, 0, -13, 9, 0, -6, 5, 0, -3, 3, 0, -2, 2, 0, -2, 1, 0, -1, 1, 0, -1, 1, 0, -1, 1, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0 };

Coefficients for FIR high Pass Kaiser filter:

Cutoff freq: 4khz, sampling freq: 12khz

#define N 82 //length of filter

short h[N]= { 1, 0, -1, 1, -1, -1, 1, -1, -1, 1, -1, -1, 1, -1, -1, 2, -1, -1, 2, -1, -1, 2, -1, -1, 2, -1, -1, 3, -2, -2, 4, -2, -2, 5, -3, -4, 9, -6, -8, 27, -41, 41, -27, 8, 6, -9, 4, 3, -5, 2, 2, -4, 2, 2, -3, 1, 1, -2, 1, 1, -2, 1, 1, -2, 1, 1, -2, 1, 1, -1, 1, 1, -1, 1, 1, -1, 1, 1, -1, 1, 0, -1 };

Coefficients for FIR high Pass rectangular filter:

Cutoff freq: 4khz, sampling freq: 12khz

#define N 82 //length of filter

short h[N]= { 1, -1, -1, 1, -1, -1, 1, -1, -1, 1, -1, -1, 1, -1, -1, 2, -1, -1, 2, -1, -1, 2, -1, -1, 2, -1, -1, 3, -2, -2, 4, -2, -2, 5, -3, -4, 9, -6, -8, 27, -41, 41, -27, 8, 6, -9, 4, 3, -5, 2, 2, -4, 2, 2, -3, 1, 1, -2, 1, 1, -2, 1, 1, -2, 1, 1, -2, 1, 1, -1, 1, 1, -1, 1, 1, -1, 1, 1, -1, 1, 1, -1 };

Coefficients for FIR high Pass triangular filter:

Cutoff freq: 4khz, sampling freq: 12khz

#define N 82 //length of filter

short h[N]= { 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 1, 0, 0, 1, 0, -1, 1, -1, -1, 1, -1, -1, 2, -1, -1, 3, -2, -2, 5, -3, -3, 8, -5, -8, 27, -41, 41, -27, 8, 5, -8, 3, 3, -5, 2, 2, -3, 1, 1, -2, 1, 1, -1, 1, 1, -1, 1, 0, -1, 0, 0, -1, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0 };

C-Program:

Main.c

```
#include "stdio.h"
#include "usbstk5515.h"
void main( void )
{
/* Initialize BSL */
USBSTK5515_init();
printf( "playing audio ::::: \n" );
while(1)
{
aic3204_test( );
}
}
```

Aic3204_test.c:

```
#define AIC3204_I2C_ADDR 0x18
#include "usbstk5515.h"
#include "usbstk5515_gpio.h"
#include "usbstk5515_i2c.h"
#include "stdio.h"
extern Int16 aic3204_tone_headphone( );
extern Int16 aic3204_loop_stereo_in1( );
Int16 AIC3204_rget( Uint16 regnum, Uint16* regval )
{
Int16 retcode = 0;
Uint8 cmd[2];
cmd[0] = regnum & 0x007F; // 7-bit Register Address
cmd[1] = 0;
retcode |= USBSTK5515_I2C_write( AIC3204_I2C_ADDR, cmd, 1 );
retcode |= USBSTK5515_I2C_read( AIC3204_I2C_ADDR, cmd, 1 );
*regval = cmd[0];
USBSTK5515_wait( 10 );
return retcode;
}
Int16 AIC3204_rset( Uint16 regnum, Uint16 regval )
{
Uint8 cmd[2];
cmd[0] = regnum & 0x007F; // 7-bit Register Address
cmd[1] = regval; // 8-bit Register Data
return USBSTK5515_I2C_write( AIC3204_I2C_ADDR, cmd, 2 );
}
```

```
Int16 aic3204_test( )
{
SYS_EXBUSSEL = 0x6100; // Enable I2C bus
USBSTK5515_I2C_init( ); // Initialize I2C
USBSTK5515_wait( 100 ); // Wait
if ( aic3204_loop_stereo_in1( ) )
return 1;
return 0;
}
aic3204_loop_stereo_in1.c:
#include "stdio.h"
#include "usbstk5515.h"
extern Int16 AIC3204_rset( Uint16 regnum, Uint16 regval);
#define Rcv 0x08
#define Xmit 0x20
#define N 82
Int16 data1, data2, data3, data4;
int sample,n,k,l;
Int16 dly0[N];
Int32 lyn,ryn;
/*//hpkaiser12-4
Int16 h[N]= {1, 0, -1, 1, -1, -1, 1, -1, -1, 1, -1, -1, 1, -1, -1, 2, -1, -1, 2, -1, -1, 2, -1, -1,
2, -1, -1, 3, -2, -2, 4, -2, -2, 5, -3, -4, 9, -6, -8, 27, -41, 41, -27, 8, 6, -9, 4, 3, -5, 2, 2, -4, 2,
2, -3, 1, 1, -2, 1, 1, -2, 1, 1, -2, 1, 1, -2, 1, 1, -1, 1, 1, -1, 1, 1, -1, 1, 1, -1, 1, 0, -1 };*/
/*//hprect12-4
Int16 h[N]= { 1, -1, -1, 1, -1, -1, 1, -1, -1, 1, -1, -1, 1, -1, -1, 2, -1, -1, 2, -1, -1, 2, -1, -1,
2, -1, -1, 3, -2, -2, 4, -2, -2, 5, -3, -4, 9, -6, -8, 27, -41, 41, -27, 8, 6, -9, 4, 3, -5, 2, 2, -4, 2,
2, -3, 1, 1, -2, 1, 1, -2, 1, 1, -2, 1, 1, -2, 1, 1, -1, 1, 1, -1, 1, 1, -1, 1, 1, -1, 1, 1, -1 };*/
/*//hptrig12-4
Int16 h[N]= { 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 1, 0, 0, 1, 0, -1, 1,  -1, -1, 1, -1, -1, 2, -
1, -1, 3, -2, -2, 5, -3, -3, 8, -5, -8, 27, -41, 41, -27, 8, 5, -8, 3, 3, -5, 2, 2, -3, 1, 1, -2, 1, 1, -
1, 1, 1,-1, 1, 0, -1, 0, 0, -1, 0, 0, 0, 0, 0,0, 0, 0, 0, 0, 0, 0, 0, 0, 0};*/
/*//lpkaiser24-8
```

```
Int16 h[N]= {0, 1, -1, 0, 1, -1, 0, 1, -1, 0, 1, -1,0, 1, -1, 0, 1, -1, 0, 2, -2, 0, 2, -2,0, 2, -2, 0,
3, -3, 0, 4, -4, 0, 5, -6, 0, 10, -14, 0, 70, 70, 0, -14, 10, 0, -6, 5, 0, -4, 4, 0, -3, 3, 0, -2, 2, 0,
-2, 2,0, -2, 2, 0, -1, 1, 0, -1, 1, 0, -1, 1,0, -1, 1, 0, -1, 1, 0, -1, 1, 0 };*/
/*//lprect24-8
short h[N]= {0, 1, -1, 0, 1, -1, 0, 1, -1, 0, 1, -1,0, 1, -1, 0, 1, -1, 0, 2, -2, 0, 2, -2,0, 2, -2, 0,
3, -3, 0, 4, -4, 0, 5, -6,0, 10, -14, 0, 70, 70, 0, -14, 10, 0, -6, 5, 0, -4, 4, 0, -3, 3, 0, -2, 2, 0, -
2, 2,0, -2, 2, 0, -1, 1, 0, -1, 1, 0, -1, 1,0, -1, 1, 0, -1, 1, 0, -1, 1, 0 };*/
/*//lptrig24-8
Int16 h[N]= { 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 1, -1, 0, 1, -1, 0, 1, -1, 0, 1, -2, 0, 2,
-2, 0, 3, -3, 0, 5, -6,0, 9, -13, 0, 70, 70, 0, -13, 9, 0, -6, 5,0, -3, 3, 0, -2, 2, 0, -2, 1, 0, -1, 1,
0, -1, 1, 0, -1, 1, 0, 0, 0, 0, 0, 0,0, 0, 0, 0, 0, 0, 0, 0, 0};*/
//lpkaiser48-8
Int16 h[N]= {-1, 0, 1, 1, 1, -1, -1, -1, 1, 1, 1, -1,-1, -1, 1, 2, 1, -1, -2, -1, 1, 2, 1, -1, -2, -1,
1, 3, 2, -2, -4, -2, 2, 5, 3, -4,-9, -6, 8, 27, 41, 41, 27, 8, -6, -9, -4, 3,5, 2, -2, -4, -2, 2, 3, 1, -
1, -2, -1, 1,2, 1, -1, -2, -1, 1, 2, 1, -1, -1, -1, 1,1, 1, -1, -1, -1, 1, 1, 1, 0, -1};
Int16 aic3204_loop_stereo_in1( )
{
Int16 j, i = 0;
/* Configure AIC3204 */
AIC3204_rset( 0, 0 ); // Select page 0
AIC3204_rset( 1, 1 ); // Reset codec
AIC3204_rset( 0, 1 ); // Point to page 1
AIC3204_rset( 1, 8 ); // Disable crude AVDD generation from DVDD
AIC3204_rset( 2, 1 ); // Enable Analog Blocks, use LDO power
AIC3204_rset( 0, 0 ); // Select page 0
/* PLL and Clocks config and Power Up */
AIC3204_rset( 27, 0x0d ); // BCLK and WCLK is set as o/p to AIC3204(Master)
AIC3204_rset( 28, 0x00 ); // Data ofset = 0
AIC3204_rset( 4, 3 ); // PLL setting: PLLCLK <- MCLK, CODEC_CLKIN <-PLL CLK
AIC3204_rset( 6, 8 ); // PLL setting: J=8
AIC3204_rset( 7, 15 ); // PLL setting: HI_BYTE(D)
AIC3204_rset( 8, 0xdc ); // PLL setting: LO_BYTE(D)
AIC3204_rset( 30, 0x88 ); // For 32 bit clocks per frame in Master mode ONLY
// BCLK=DAC_CLK/N =(12288000/8) = 1.536MHz = 32*fs
```

```
AIC3204_rset( 5, 0x91 ); // PLL setting: Power up PLL, P=1 and R=1
AIC3204_rset( 13, 0 ); // Hi_Byte(DOSR) for DOSR = 128 decimal or 0x0080 DAC
oversamppling
AIC3204_rset( 14, 0x80 ); // Lo_Byte(DOSR) for DOSR = 128 decimal or 0x0080
AIC3204_rset( 20, 0x80 ); // AOSR for AOSR = 128 decimal or 0x0080 for decimation
filters 1 to 6
AIC3204_rset( 11, 0x88 ); // Power up NDAC and set NDAC value to 8
AIC3204_rset( 12, 0x82 ); // Power up MDAC and set MDAC value to 2
AIC3204_rset( 18, 0x88 ); // Power up NADC and set NADC value to 8
AIC3204_rset( 19, 0x82 ); // Power up MADC and set MADC value to 2
/* DAC ROUTING and Power Up */
AIC3204_rset( 0, 0x01 ); // Select page 1
AIC3204_rset( 12, 0x08 ); // LDAC AFIR routed to HPL
AIC3204_rset( 13, 0x08 ); // RDAC AFIR routed to HPR
AIC3204_rset( 0, 0x00 ); // Select page 0
AIC3204_rset( 64, 0x02 ); // Left vol=right vol
AIC3204_rset( 65, 0x00 ); // Left DAC gain to 0dB VOL; Right tracks Left
AIC3204_rset( 63, 0xd4 ); // Power up left,right data paths and set channel
AIC3204_rset( 0, 0x01 ); // Select page 1
AIC3204_rset( 16, 0x06 ); // Unmute HPL , 6dB gain
AIC3204_rset( 17, 0x06 ); // Unmute HPR , 6dB gain
AIC3204_rset( 9, 0x30 ); // Power up HPL,HPR
AIC3204_rset( 0, 0x00 ); // Select page 0
USBSTK5515_wait( 500 ); // Wait
/* ADC ROUTING and Power Up */
AIC3204_rset( 0, 1 ); // Select page 1
AIC3204_rset( 0x34, 0x30 ); // STEREO 1 Jack
// IN2_L to LADC_P through 40 kohm
AIC3204_rset( 0x37, 0x30 ); // IN2_R to RADC_P through 40 kohmm
AIC3204_rset( 0x36, 3 ); // CM_1 (common mode) to LADC_M through 40 kohm
AIC3204_rset( 0x39, 0xc0 ); // CM_1 (common mode) to RADC_M through 40 kohm
AIC3204_rset( 0x3b, 0 ); // MIC_PGA_L unmute
AIC3204_rset( 0x3c, 0 ); // MIC_PGA_R unmute
AIC3204_rset( 0, 0 ); // Select page 0
```

```
AIC3204_rset( 0x51, 0xc0 ); // Powerup Left and Right ADC
AIC3204_rset( 0x52, 0 ); // Unmute Left and Right ADC
AIC3204_rset( 0, 0 );
USBSTK5515_wait( 200 ); // Wait
/* I2S settings */
I2S0_SRGR = 0x0;
I2S0_CR = 0x8010; // 16-bit word, slave, enable I2C
I2S0_ICMR = 0x3f; // Enable interrupts
/* Play Tone */
for(l=0;l<N;l++)
dly0[l]=0;
for ( i = 0 ; i < 5 ; i++ )
{
for ( j = 0 ; j < 1000 ; j++ )
{
for ( sample = 0 ; sample < N ; sample++ )
{
/* Read Digital audio input */
data3 = I2S0_W0_MSW_R; // 16 bit left channel received audio data
data1 = I2S0_W0_LSW_R;
data4 = I2S0_W1_MSW_R; // 16 bit right channel received audio data
data2 = I2S0_W1_LSW_R;
while((Rcv & I2S0_IR) == 0); // Wait for interrupt pending flag
lyn=0;
for(k=0;k<N;k++)
{
dly0[(N-1)-k]=dly0[(N-1)-k-1];
dly0[0]=data3;
}
for(k=0;k<N;k++)
lyn+=((h[k])*dly0[k]); //fir low pass filter
data1=lyn>>1;
data2=data1;
/* Write Digital audio input */
I2S0_W0_MSW_W = data1; // 16 bit left channel transmit audio data
I2S0_W0_LSW_W = 0;
I2S0_W1_MSW_W = data2; // 16 bit right channel transmit audio data
I2S0_W1_LSW_W = 0;
while((Xmit & I2S0_IR) == 0); // Wait for interrupt pending flag
}
}
}
/* Disble I2S */
I2S0_CR = 0x00;
return 0;
}
```

Execution Procedure:

- Open CCstudio setup
- Go to File Menu , select Import option.
- In the Import Window under CCs choose Existing CCS/CCE Eclipse project then next.
- In Select root Directory Browse for project file where it is located.
- Select FIR Project folder and Finish it.
- Now Connect the DSP Kit to PC and Launch it.(Follow the above given manual procedure from Step 46 to 51)
- Give Right Click on Your fir.out file under Binaries and select Load program Option.
- Now Go to Target select Run.
- Observe the audio (Give Input from Stereo in and listen Filtered Output from Stereo Out).

OBSEVATIONS

Output Waveforms:

EXP.NO: 5

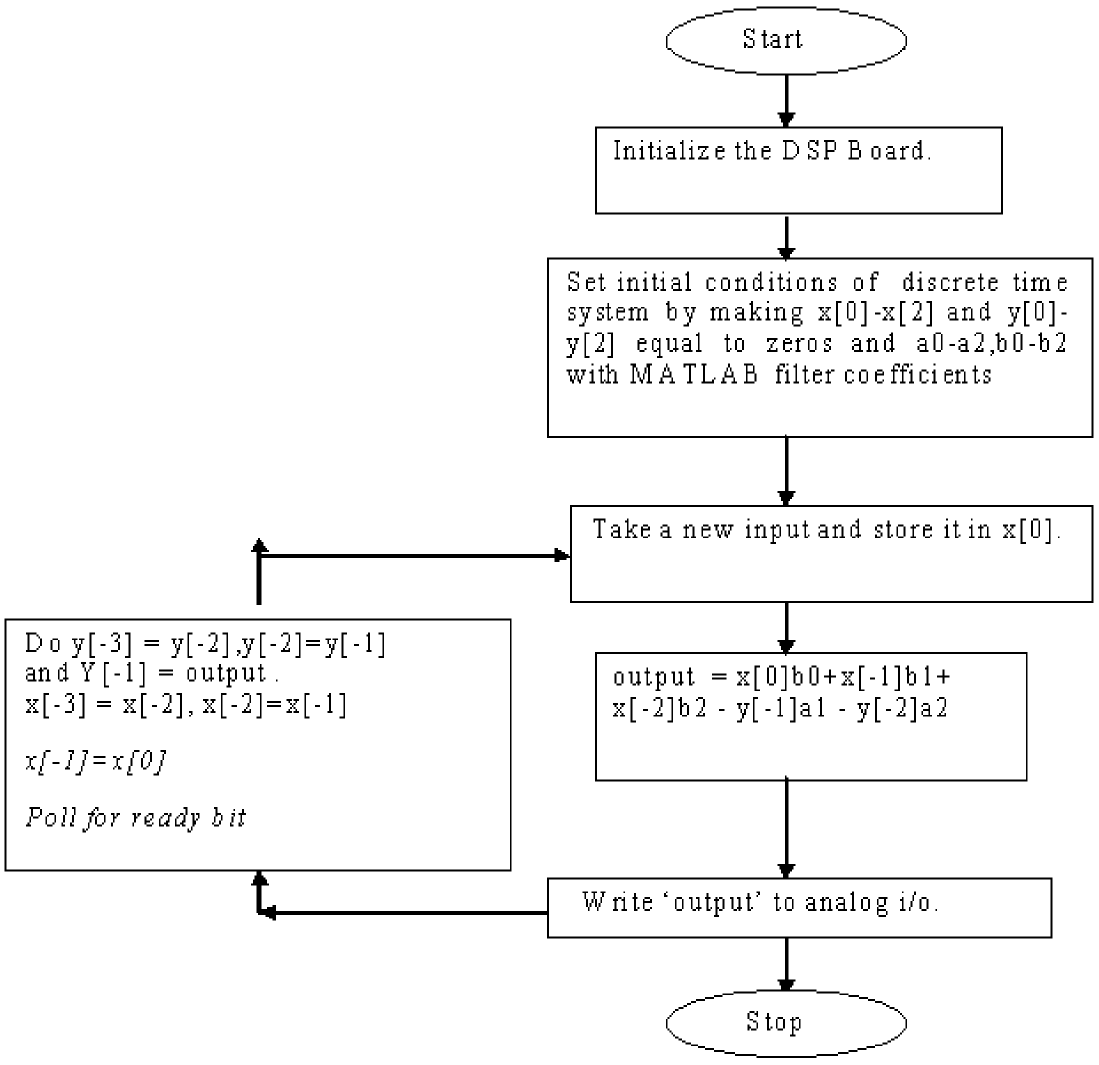
IMPLEMENTATION OF LP IIR FILTER FOR GIVEN SEQUENCE & IMPLEMENTATION OF HP IIR FILTER FOR GIVEN SEQUENCE

Aim: The aim of this laboratory exercise is to design and implement a Digital IIR Filter & observe its frequency response. In this experiment we design a simple IIR filter so as to stop or attenuate required band of frequencies components and pass the frequency components which are outside the required band

EQUIPMENTS NEEDED:

- Host (PC) with windows(95/98/Me/XP/NT/2000).
- TMS320C5515 DSP Starter Kit (DSK).
- Audio Jack Cable

FLOWCHART FOR IIR IMPLEMENTATION:


Start
Initialize the DSP Board.
Set initial conditions of discrete time system by making x[0]-x[2] and y[0]-y[2] equal to zeros and a0-a2,b0-b2 with MATLAB filter coefficients
Take a new input and store it in x[0].
output = x[0]b0+x[-1]b1+ x[-2]b2 - y[-1]a1 - y[-2]a2
Write 'output' to analog i/o.
Do y[-3] = y[-2],y[-2]=y[-1] and Y[-1] = output . x[-3] = x[-2], x[-2]=x[-1]
x[-1]=x[0]
Poll for ready bit
Stop

IIR high pass filter coefficients:

#define N 83 //length of filter

Short h[N]={1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,
3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1, 1,3,1,3,1,3,1,3,1,2,1};

IIR low pass filter coefficients:

```
#define N 83 //length of filter
Shorth[N]={1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,
3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,2,1 };
```

rogram:

main.c

```
#include "stdio.h"
#include "usbstk5515.h"
void main( void )
{
/* Initialize BSL */
USBSTK5515_init();
printf( "playing audio ::::: \n" );
while(1)
{
aic3204_test( );
}
}
```

Aic3204_test.c:

```
#define AIC3204_I2C_ADDR 0x18
#include "usbstk5515.h"
#include "usbstk5515_gpio.h"
#include "usbstk5515_i2c.h"
#include "stdio.h"
extern Int16 aic3204_tone_headphone( );
extern Int16 aic3204_loop_stereo_in1( );
Int16 AIC3204_rget( Uint16 regnum, Uint16* regval )
{
Int16 retcode = 0;
Uint8 cmd[2];
cmd[0] = regnum & 0x007F; // 7-bit Register Address
cmd[1] = 0;
retcode |= USBSTK5515_I2C_write( AIC3204_I2C_ADDR, cmd, 1 );
retcode |= USBSTK5515_I2C_read( AIC3204_I2C_ADDR, cmd, 1 );
*regval = cmd[0];
USBSTK5515_wait( 10 );
return retcode;
}
Int16 AIC3204_rset( Uint16 regnum, Uint16 regval )
{
Uint8 cmd[2];
cmd[0] = regnum & 0x007F; // 7-bit Register Address
cmd[1] = regval; // 8-bit Register Data
return USBSTK5515_I2C_write( AIC3204_I2C_ADDR, cmd, 2 );
}
Int16 aic3204_test( )
{
SYS_EXBUSSEL = 0x6100; // Enable I2C bus
USBSTK5515_I2C_init( ); // Initialize I2C
USBSTK5515_wait( 100 ); // Wait
if ( aic3204_loop_stereo_in1( ) )
return 1;
return 0;
}
```

aic3204_loop_stereo_in1.c:

```
#include "stdio.h"
#include "usbstk5515.h"
extern Int16 AIC3204_rset( Uint16 regnum, Uint16 regval);
#define Rcv 0x08
#define Xmit 0x20
#define N 83
Int16 data1, data2, data3, data4;
int sample, n, k, l;
Int16 dly0[N];
Int16 dly1[N];
Int32 lyn,lyn0,lyn1;
```

```
Int16                 h[N]={1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,
3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,   3,1,3,1,3,1,3,1,3,1,2,1
};
Int16 aic3204_loop_stereo_in1 ( )
{
/* Pre-generated sine wave data, 16-bit signed samples */
Int16 j, i = 0;
/* Configure AIC3204 */
AIC3204_rset( 0, 0 ); // Select page 0
AIC3204_rset( 1, 1 ); // Reset codec
AIC3204_rset( 0, 1 ); // Point to page 1
AIC3204_rset( 1, 8 ); // Disable crude AVDD generation from DVDD
AIC3204_rset( 2, 1 ); // Enable Analog Blocks, use LDO power
AIC3204_rset( 0, 0 ); // Select page 0
/* PLL and Clocks config and Power Up */
AIC3204_rset( 27, 0x0d ); // BCLK and WCLK is set as o/p to AIC3204(Master)
AIC3204_rset( 28, 0x00 ); // Data ofset = 0
AIC3204_rset( 4, 3 ); // PLL setting: PLLCLK <- MCLK, CODEC_CLKIN <-PLL
CLK
AIC3204_rset( 6, 8 ); // PLL setting: J=8
AIC3204_rset( 7, 15 ); // PLL setting: HI_BYTE(D)
AIC3204_rset( 8, 0xdc ); // PLL setting: LO_BYTE(D)
AIC3204_rset( 30, 0x88 ); // For 32 bit clocks per frame in Master mode ONLY
// BCLK=DAC_CLK/N =(12288000/8) = 1.536MHz = 32*fs
AIC3204_rset( 5, 0x91 ); // PLL setting: Power up PLL, P=1 and R=1
AIC3204_rset( 13, 0 ); // Hi_Byte(DOSR) for DOSR = 128 decimal or 0x0080 DAC
oversamppling
AIC3204_rset( 14, 0x80 ); // Lo_Byte(DOSR) for DOSR = 128 decimal or 0x0080
AIC3204_rset( 20, 0x80 ); // AOSR for AOSR = 128 decimal or 0x0080 for decimation
filters 1 to 6
AIC3204_rset( 11, 0x88 ); // Power up NDAC and set NDAC value to 8
AIC3204_rset( 12, 0x82 ); // Power up MDAC and set MDAC value to 2
AIC3204_rset( 18, 0x88 ); // Power up NADC and set NADC value to 8
AIC3204_rset( 19, 0x82 ); // Power up MADC and set MADC value to 2
```

```
/* DAC ROUTING and Power Up */
AIC3204_rset( 0, 0x01 ); // Select page 1
AIC3204_rset( 12, 0x08 ); // LDAC AFIR routed to HPL
AIC3204_rset( 13, 0x08 ); // RDAC AFIR routed to HPR
AIC3204_rset( 0, 0x00 ); // Select page 0
AIC3204_rset( 64, 0x02 ); // Left vol=right vol
AIC3204_rset( 65, 0x00 ); // Left DAC gain to 0dB VOL; Right tracks Left
AIC3204_rset( 63, 0xd4 ); // Power up left,right data paths and set channel
AIC3204_rset( 0, 0x01 ); // Select page 1
AIC3204_rset( 16, 0x06 ); // Unmute HPL , 6dB gain
AIC3204_rset( 17, 0x06 ); // Unmute HPR , 6dB gain
AIC3204_rset( 9, 0x30 ); // Power up HPL,HPR
AIC3204_rset( 0, 0x00 ); // Select page 0
USBSTK5515_wait( 500 ); // Wait
/* ADC ROUTING and Power Up */
AIC3204_rset( 0, 1 ); // Select page 1
AIC3204_rset( 0x34, 0x30 ); // STEREO 1 Jack
// IN2_L to LADC_P through 40 kohm
AIC3204_rset( 0x37, 0x30 ); // IN2_R to RADC_P through 40 kohmm
AIC3204_rset( 0x36, 3 ); // CM_1 (common mode) to LADC_M through 40 kohm
AIC3204_rset( 0x39, 0xc0 ); // CM_1 (common mode) to RADC_M through 40 kohm
AIC3204_rset( 0x3b, 0 ); // MIC_PGA_L unmute
AIC3204_rset( 0x3c, 0 ); // MIC_PGA_R unmute
AIC3204_rset( 0, 0 ); // Select page 0
AIC3204_rset( 0x51, 0xc0 ); // Powerup Left and Right ADC
AIC3204_rset( 0x52, 0 ); // Unmute Left and Right ADC
AIC3204_rset( 0, 0 );
USBSTK5515_wait( 200 ); // Wait
/* I2S settings */
I2S0_SRGR = 0x0;
I2S0_CR = 0x8010; // 16-bit word, slave, enable I2C
I2S0_ICMR = 0x3f; // Enable interrupts
/* Play Tone */
for(l=0;l<N;l++)
```

```
{
dly0[l]=0;
dly1[l]=0;
}
for ( i = 0 ; i < 5 ; i++ )
{
for ( j = 0 ; j < 1000 ; j++ )
{
for ( sample = 0 ; sample < N ; sample++ )
{
/* Read Digital audio input */
data3 = I2S0_W0_MSW_R; // 16 bit left channel received audio data
data1 = I2S0_W0_LSW_R;
data4 = I2S0_W1_MSW_R; // 16 bit right channel received audio data
data2 = I2S0_W1_LSW_R;
while((Rcv & I2S0_IR) == 0); // Wait for interrupt pending flag
dly0[sample]=data3;
lyn0=0;
lyn1=0;
for(k=0;k<=sample;k++)
lyn0+=((h[k])*dly0[sample-k]); //fir low pass filter
for(k=1;k<=sample;k++)
lyn1+=((h[k])*dly1[sample-k]); //fir low pass filter
lyn=lyn0-lyn1;
dly1[sample]=lyn;
data1=lyn<<1;
data2=lyn<<1;
/* Write Digital audio input */
I2S0_W0_MSW_W = data1; // 16 bit left channel transmit audio data
I2S0_W0_LSW_W = 0;
I2S0_W1_MSW_W = data2; // 16 bit right channel transmit audio data
I2S0_W1_LSW_W = 0;
while((Xmit & I2S0_IR) == 0); // Wait for interrupt pending flag
}
}
}
/* Disble I2S */
I2S0_CR = 0x00;
return 0;
}
```

Execution Procedure:

- Open CCstudio setup
- Go to File Menu , select Import option.
- In the Import Window under CCs choose Existing CCS/CCE Eclipse project then next.
- In Select root Directory Browse for project file where it is located.
- Select FIR Project folder and Finish it.
- Now Connect the DSP Kit to PC and Launch it.(Follow the above given manual procedure from Step 46 to 51)
- Give Right Click on Your fir.out file under Binaries and select Load program Option.
- Now Go to Target select Run.
- Observe the audio (Give Input from Stereo in and listen Filtered Output from Stereo Out).

OBSEVATIONS

Output Waveforms:

EXP.NO: 6
GENERATION OF SINUSOIDAL SIGNAL THROUGH FILTERING

Program:

Main.c

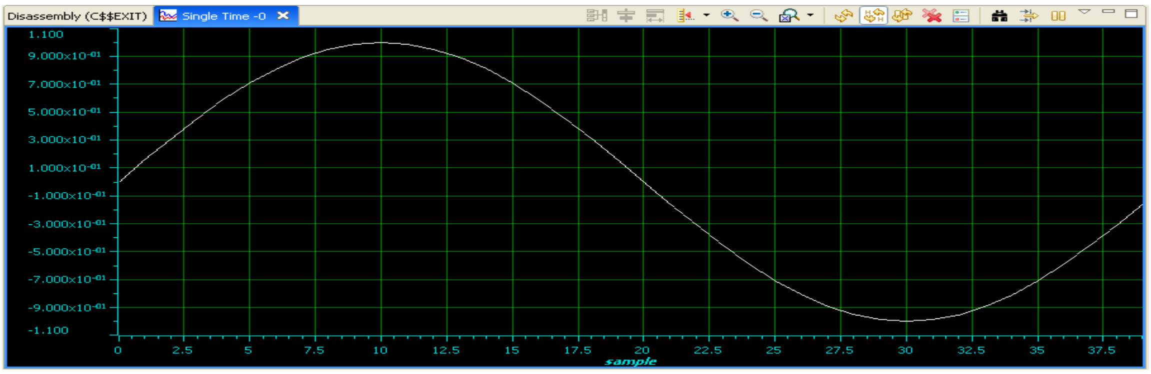
```
   #include <usbstk5515.h>
   #include<stdio.h>
   #include <math.h>
   #define PTS 40 //no of points for FFT
   #define PI 3.14159265358979
   float output;
   float y[40];
   void main()
   {
int n;
for (n=0; n<PTS; n++)
{
/*y[n]=A•y[n–1]+B•y[n–2] A=1.9754 and B=–1. Examining the behavior
of this IIR filter by its transfer function as below:
//y[n] = 1.9754•y[n–1] – y[n–2] */
if(n<2)
y[n]=sin(2*PI*n/PTS);
else
y[n]=(1.9754*y[n-1])-(y[n-2]);
printf("%f,",y[n]);
}
}
```

Execution Procedure:

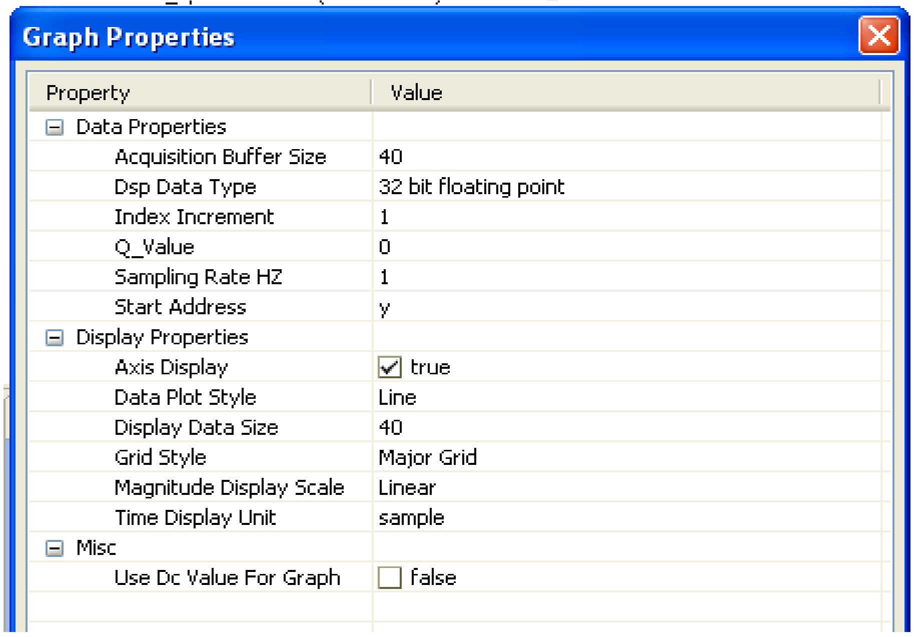
- Open CCstudio setup
- Go to File Menu , select Import option.
- In the Import Window under CCs choose Existing CCS/CCE Eclipse project then
- next.
- In Select root Directory Browse for project file where it is located.
- Select sinusoidal through Filtering Project folder and Finish it.
- Now Connect the DSP Kit to PC and Launch it.(Follow the above given manual

- procedure from Step 46 to 51)
- Give Right Click on Your sin.out file under Binaries and select Load program Option.
- Now Go to Target select Run.
- In Tools menu select Graph (single Time) set the properties and watch.

Result:



Graph Proeprties:



Property	Value
⊟ Data Properties	
Acquisition Buffer Size	40
Dsp Data Type	32 bit floating point
Index Increment	1
Q_Value	0
Sampling Rate HZ	1
Start Address	y
⊟ Display Properties	
Axis Display	✓ true
Data Plot Style	Line
Display Data Size	40
Grid Style	Major Grid
Magnitude Display Scale	Linear
Time Display Unit	sample
⊟ Misc	
Use Dc Value For Graph	☐ false

OBSEVATIONS

Output Waveforms:

EXP.NO: 7
GENERATION OF DTMF SIGNALS

Program:

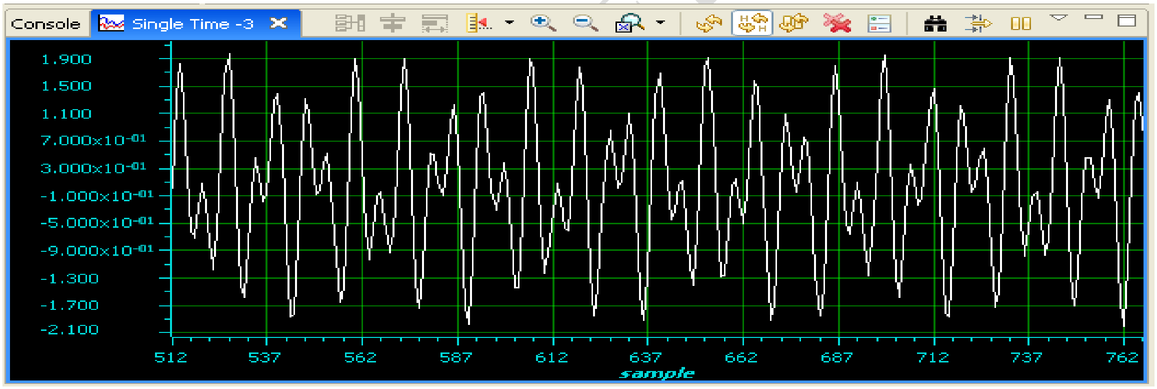
```
    #include <usbstk5515.h>
    #include<stdio.h>
    #include <math.h>
    #define PTS 256
    #define FS 8000 //sampling frequency of 8khz
    #define pi 3.14159265358979
    int lfg[ ]={697,770,852,941}; // Low frequency group
    int hfg[ ]={1209,1336,1477}; // High frequency group
    float num[12][PTS];
    float t[PTS];
    void main()
    {
    int n,tone_select=0,r=0,c=0;
    printf("DTMF Signal Generation:");
    for(n=0;n<PTS;n++) {
    t[n] = 2*pi*n/FS; }
    tone_select=0;
    for(r=0;r<4;r++) //loop for selection lfg index
    {
    for(c=0;c<3;c++) //loop for selection hfg index
    {
    for(n=0;n<PTS;n++)
    {
    num[tone_select][n]=sin(lfg[r]*t[n])+sin(hfg[c]*t[n]);
    //printf("%d %f \n",n,t[n],num[tone_select][n]);
    }
    tone_select++;
}
}
printf("Done");
}
```

Execution Procedure:

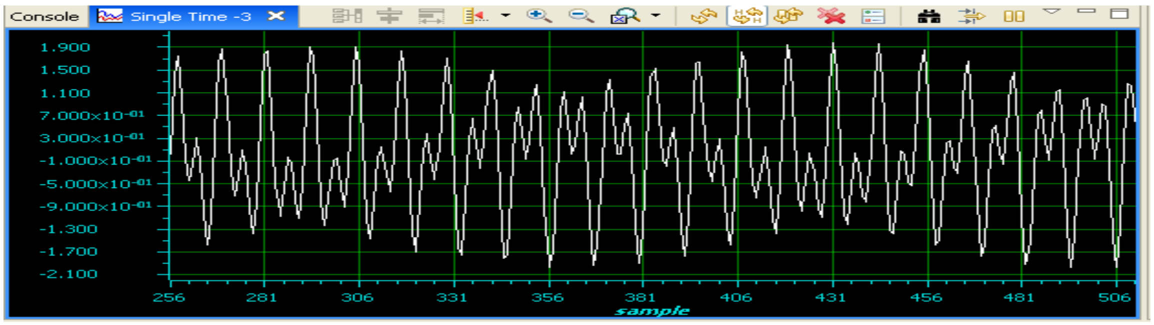
- Open CCstudio setup
- Go to File Menu, select Import option.
- In the Import Window under CCs choose Existing CCS/CCE Eclipse project then next.
- In Select root Directory Browse for project file where it is located.
- Select DTMF Project folder and Finish it.
- Now Connect the DSP Kit to PC and Launch it. (Follow the above given manual procedure from Step 46 to 51).
- Give Right Click on Your dtmf.out file under Binaries and select Load program Option.
- Now Go to Target select Run.
- In Tools menu select Graph (single Time) set the properties and watch.

Result:

DTMF 1:



DTMF 2:



OBSEVATIONS

Output Waveforms:

EXP.NO: 8
IMPLEMENTATION OF DECIMATION PROCESS

Program:

```
    #include <usbstk5515.h>
    #include<stdio.h>
    #include <math.h>
    #define PTS 256 //no of points for FFT
    #define PI 3.14159265358979
    float y1[PTS],y2[PTS],y3[PTS];
    main()
    {
    int n,m;
    printf("Enter Vaue for Decimation Factor\n");
    scanf("%d",&m);
    printf("Original signal samples:\n");
    for(n=0;n<=PTS;n++) //original signal
    {
    y1[n]=sin(2*PI*10*(n)/(PTS));
    printf("%d, %f\n",n,y1[n]);
}
printf("Decimated signal samples:\n");
for(n=0;n<=PTS;n++) //intrpol
{
y2[n]=sin(2*PI*10*n/(m*PTS));
printf("%d, %f\n",n,y2[n]);
}
} //end of main
```

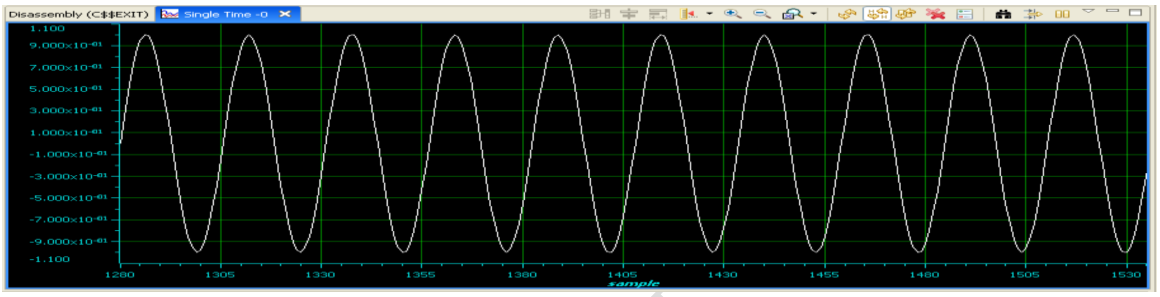
Execution Procedure:

- Open CCstudio setup
- Go to File Menu, select Import option.
- In the Import Window under CCs choose Existing CCS/CCE Eclipse project then next.
- In Select root Directory Browse for project file where it is located.
- Select Decimation Project folder and Finish it.

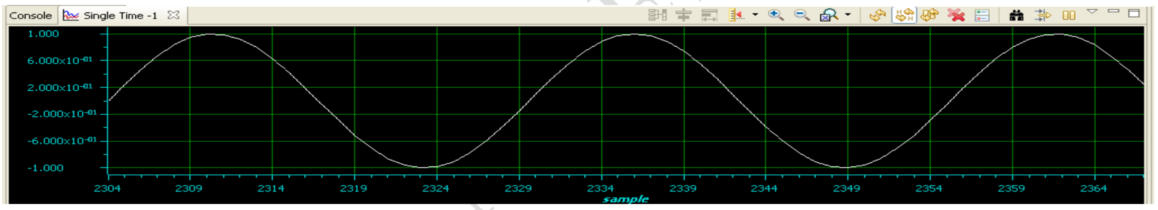
- Now Connect the DSP Kit to PC and Launch it. (Follow the above given manual procedure from Step 46 to 51).
- Give Right Click on Your decimation.out file under Binaries and select Load program Option.
- Now Go to Target select Run.
- In Tools menu select Graph (Dual Time) set the properties and watch input signal and decimated signal.

Result:

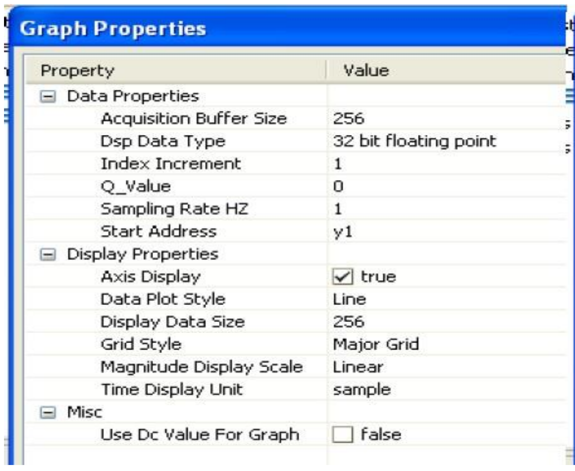
Input signal:



Output Signal:

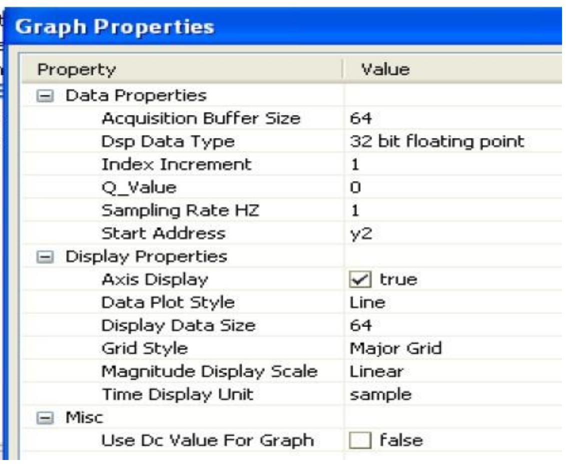


Graph Properties:



Graph Properties

Property	Value
⊟ Data Properties	
Acquisition Buffer Size	256
Dsp Data Type	32 bit floating point
Index Increment	1
Q_Value	0
Sampling Rate HZ	1
Start Address	y1
⊟ Display Properties	
Axis Display	✓ true
Data Plot Style	Line
Display Data Size	256
Grid Style	Major Grid
Magnitude Display Scale	Linear
Time Display Unit	sample
⊟ Misc	
Use Dc Value For Graph	☐ false



Graph Properties

Property	Value
⊟ Data Properties	
Acquisition Buffer Size	64
Dsp Data Type	32 bit floating point
Index Increment	1
Q_Value	0
Sampling Rate HZ	1
Start Address	y2
⊟ Display Properties	
Axis Display	✓ true
Data Plot Style	Line
Display Data Size	64
Grid Style	Major Grid
Magnitude Display Scale	Linear
Time Display Unit	sample
⊟ Misc	
Use Dc Value For Graph	☐ false

OBSEVATIONS

Output Waveforms:

EXP.NO: 9

IMPLEMENTATION OF INTERPOLATION PROCESS

Program:

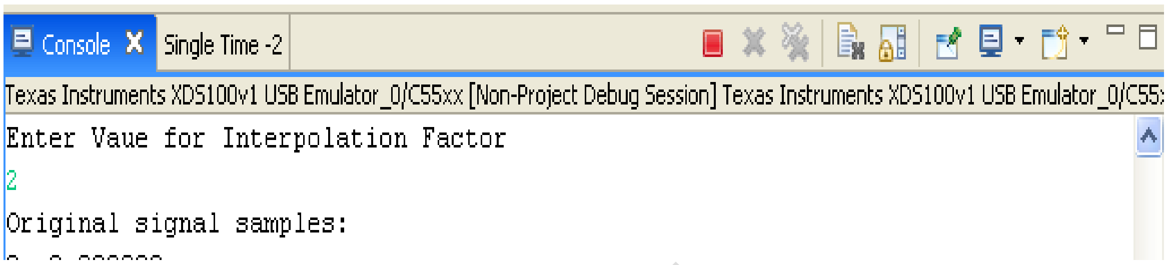
```
#include <usbstk5515.h>
#include<stdio.h>
#include <math.h>
#define PTS 256 //no of points for FFT
#define PI 3.14159265358979
float y1[PTS],y2[PTS],y3[PTS];
main()
{
int n,m;
printf("Enter Vaue for Interpolation Factor\n");
scanf("%d",&m);
printf("Original signal samples:\n");
for(n=0;n<=PTS;n++) //original signal
{
y1[n]=sin(2*PI*10*(n)/(PTS));
printf("%d, %f\n",n,y1[n]);
}
printf("Interpolated signal samples:\n");
for(n=0;n<=PTS;n++) //intrpol
{
y2[n]=sin(2*PI*10*(m*n)/(PTS));
printf("%d, %f\n",n,y2[n]);
}
}
```

Execution Procedure:

- Open CCstudio setup
- Go to File Menu, select Import option.
- In the Import Window under CCs choose Existing CCS/CCE Eclipse project then next.
- In Select root Directory Browse for project file where it is located.

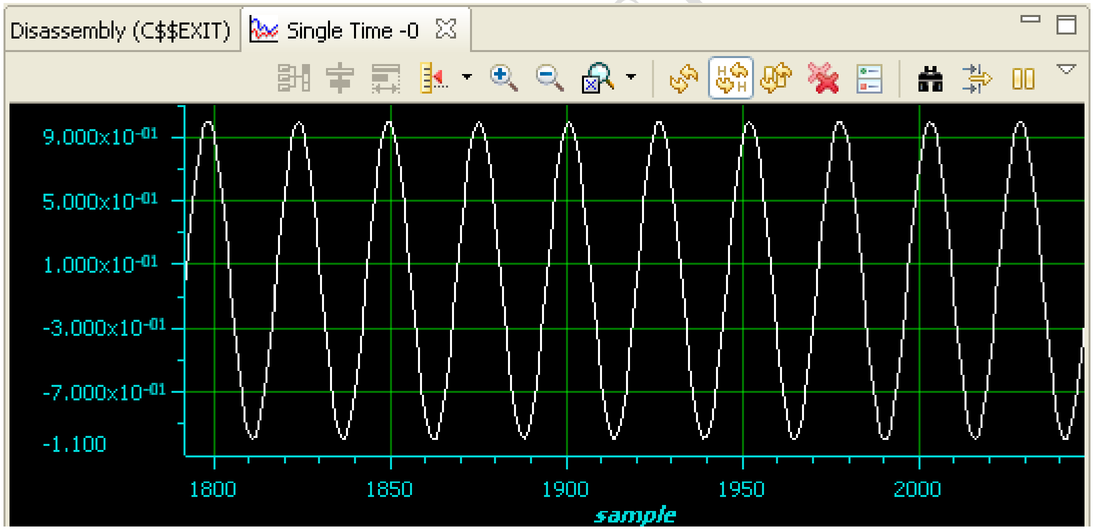
- Select Interpolation Project folder and Finish it.
- Now Connect the DSP Kit to PC and Launch it. (Follow the above given manual procedure from Step 46 to 51).
- Give Right Click on Your interpolation.out file under Binaries and select Load program Option.
- Now Go to Target select Run.
- In Tools menu select Graph (Dual Time) set the properties and watch input signal and decimated signal.

Result :

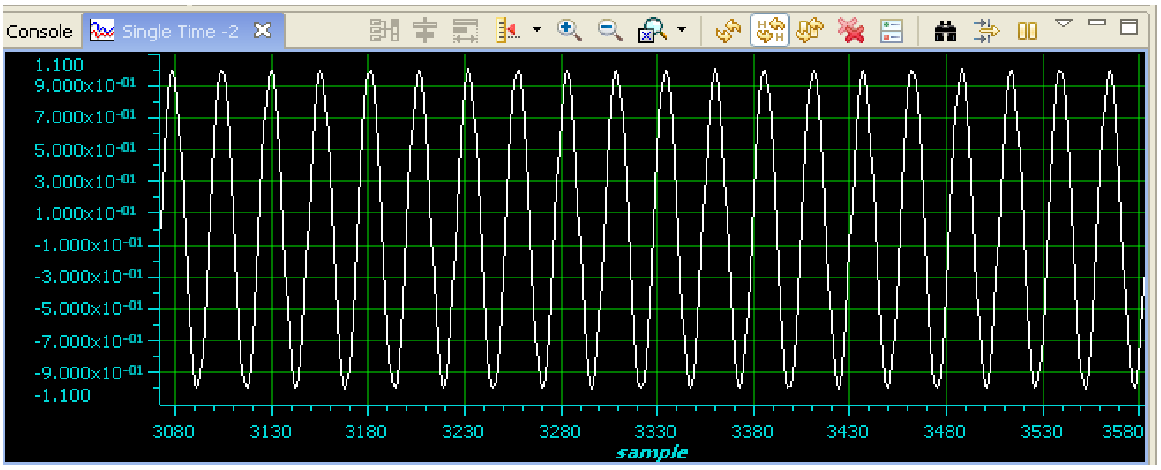


```
Texas Instruments XDS100v1 USB Emulator_0/C55xx [Non-Project Debug Session] Texas Instruments XDS100v1 USB Emulator_0/C55
Enter Vaue for Interpolation Factor
2
Original signal samples:
```

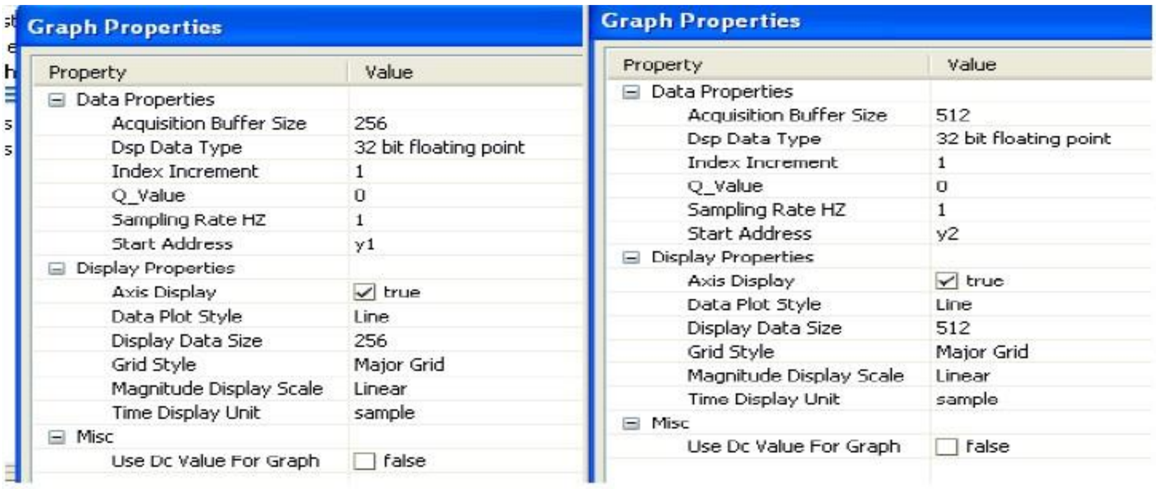
Input Signal:



Output Signal:



Graph Properties:


Graph Properties

Property	Value
⊟ Data Properties	
Acquisition Buffer Size	256
Dsp Data Type	32 bit floating point
Index Increment	1
Q_Value	0
Sampling Rate HZ	1
Start Address	y1
⊟ Display Properties	
Axis Display	✓ true
Data Plot Style	Line
Display Data Size	256
Grid Style	Major Grid
Magnitude Display Scale	Linear
Time Display Unit	sample
⊟ Misc	
Use Dc Value For Graph	☐ false

Graph Properties

Property	Value
⊟ Data Properties	
Acquisition Buffer Size	512
Dsp Data Type	32 bit floating point
Index Increment	1
Q_Value	0
Sampling Rate HZ	1
Start Address	y2
⊟ Display Properties	
Axis Display	✓ true
Data Plot Style	Line
Display Data Size	512
Grid Style	Major Grid
Magnitude Display Scale	Linear
Time Display Unit	sample
⊟ Misc	
Use Dc Value For Graph	☐ false

OBSEVATIONS

Output Waveforms:

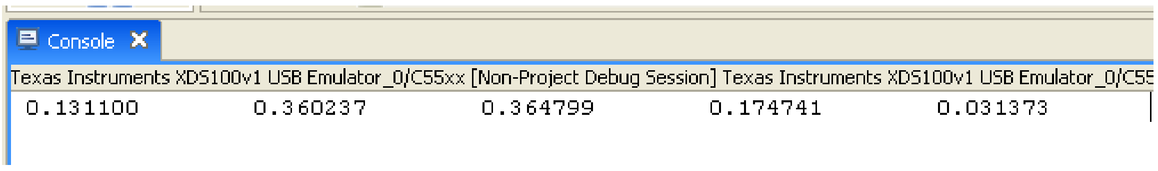
EXP.NO: 10
IMPULSE RESPONSE OF FIRST ORDER AND SECOND ORDER SYSTEMS

Program:

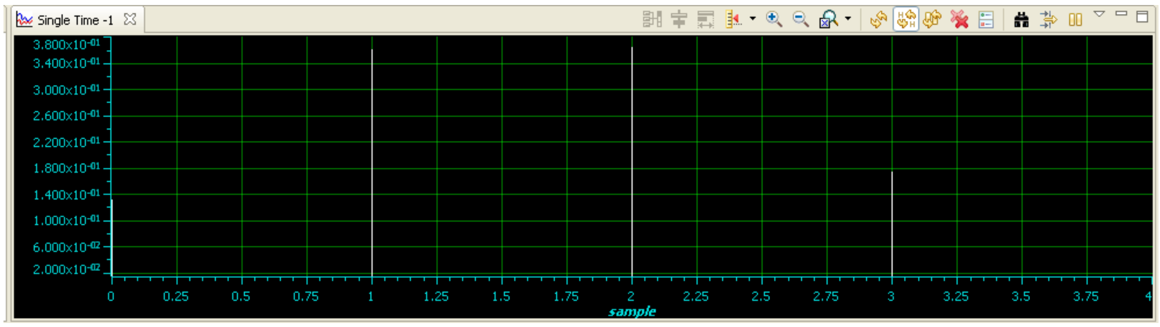
```
#include<stdio.h>
#define Order 2
#define Len 5
float h[Len] = {0.0,0.0,0.0,0.0,0.0},sum;
void main()
{
int j, k;
float a[Order+1] = {0.1311, 0.2622, 0.1311};
float b[Order+1] = {1, -0.7478, 0.2722};
for(j=0; j<Len; j++)
{
sum = 0.0;
for(k=1; k<=Order; k++)
{
if ((j-k) >= 0)
sum = sum+(b[k]*h[j-k]);
}
if (j <= Order)
h[j] = a[j]-sum;
else
h[j] = -sum;
printf (" %f ",h[j]);
}
}
```

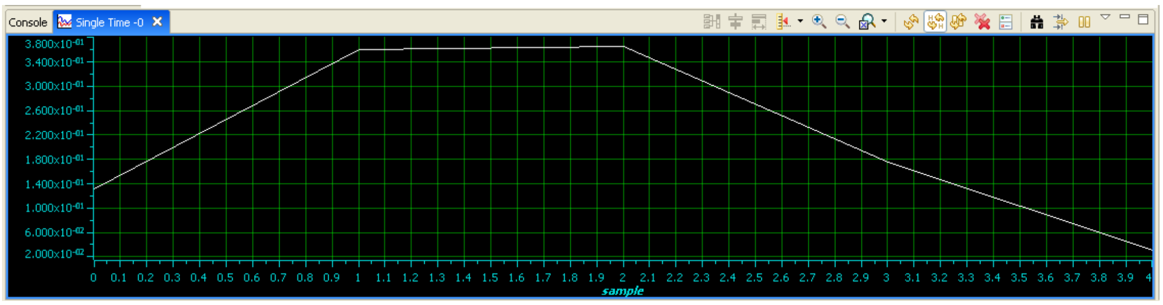
Output:

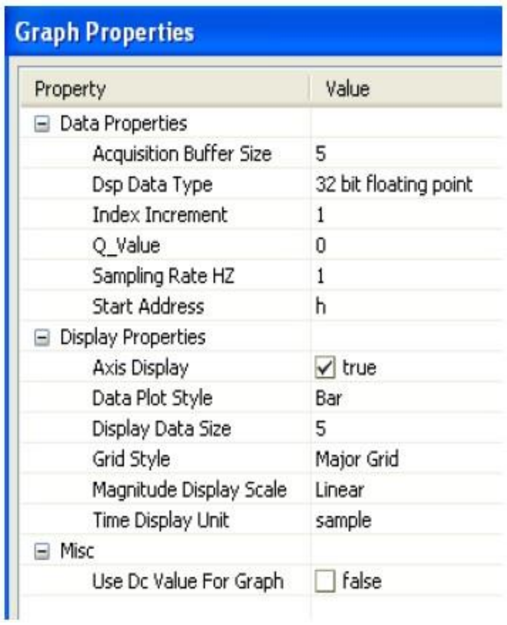
```
0.131100 0.360237 0.364799 0.174741  0.031373
```



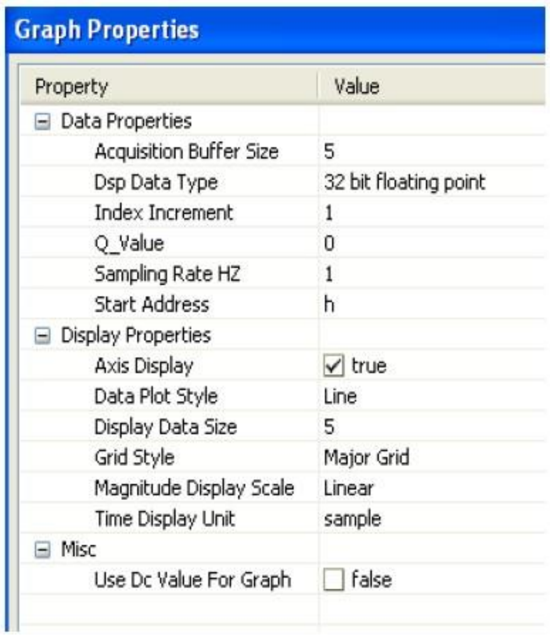
```
Texas Instruments XDS100v1 USB Emulator_0/C55xx [Non-Project Debug Session] Texas Instruments XDS100v1 USB Emulator_0/C55
 0.131100        0.360237        0.364799        0.174741        0.031373
```






Graph Properties

Property	Value
⊟ Data Properties	
Acquisition Buffer Size	5
Dsp Data Type	32 bit floating point
Index Increment	1
Q_Value	0
Sampling Rate HZ	1
Start Address	h
⊟ Display Properties	
Axis Display	✓ true
Data Plot Style	Bar
Display Data Size	5
Grid Style	Major Grid
Magnitude Display Scale	Linear
Time Display Unit	sample
⊟ Misc	
Use Dc Value For Graph	☐ false


Graph Properties

Property	Value
⊟ Data Properties	
Acquisition Buffer Size	5
Dsp Data Type	32 bit floating point
Index Increment	1
Q_Value	0
Sampling Rate HZ	1
Start Address	h
⊟ Display Properties	
Axis Display	✓ true
Data Plot Style	Line
Display Data Size	5
Grid Style	Major Grid
Magnitude Display Scale	Linear
Time Display Unit	sample
⊟ Misc	
Use Dc Value For Graph	☐ false

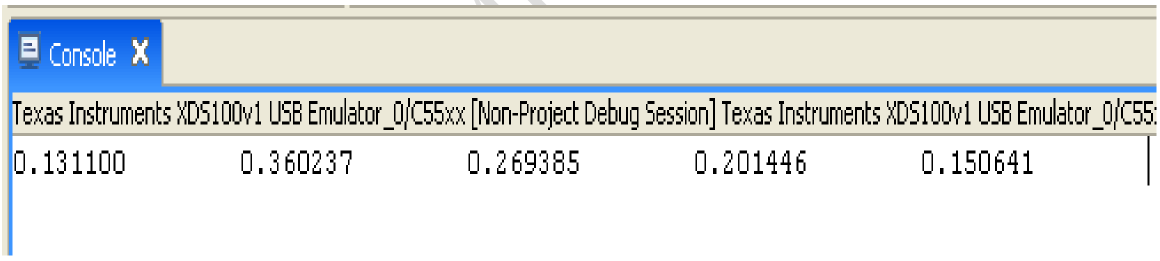
FOR FIRST ORDER DIFFERENCE EQUATION.

Program:

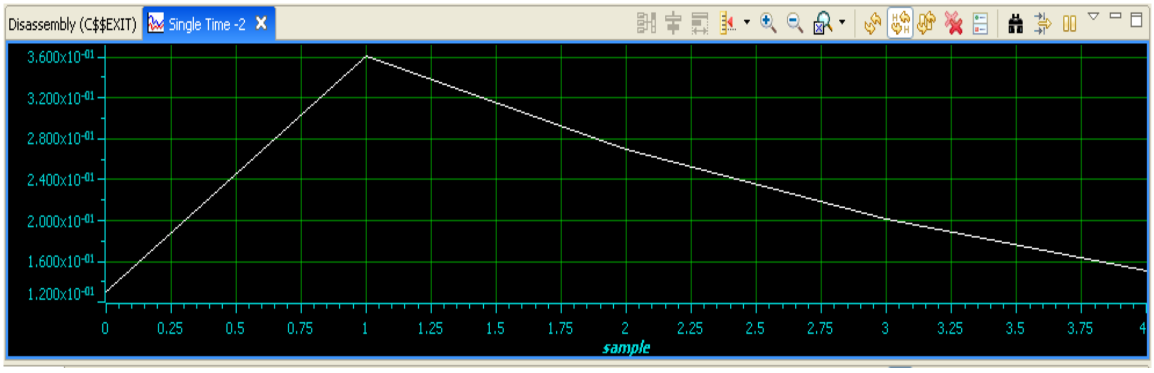
```
#include<stdio.h>

#define Order 1

#define Len 5

float h[Len] = {0.0,0.0,0.0,0.0,0.0},sum;

void main()

{

int j, k;

float a[Order+1] = {0.1311, 0.2622};
float b[Order+1] = {1, -0.7478};
for(j=0; j<Len; j++)
{
sum = 0.0;
for(k=1; k<=Order; k++)
{
if((j-k)>=0)
sum = sum+(b[k]*h[j-k]);
}
if(j<=Order)
h[j] = a[j]-sum;
else
h[j] = -sum;
printf("%f ", j, h[j]);
}
}
```

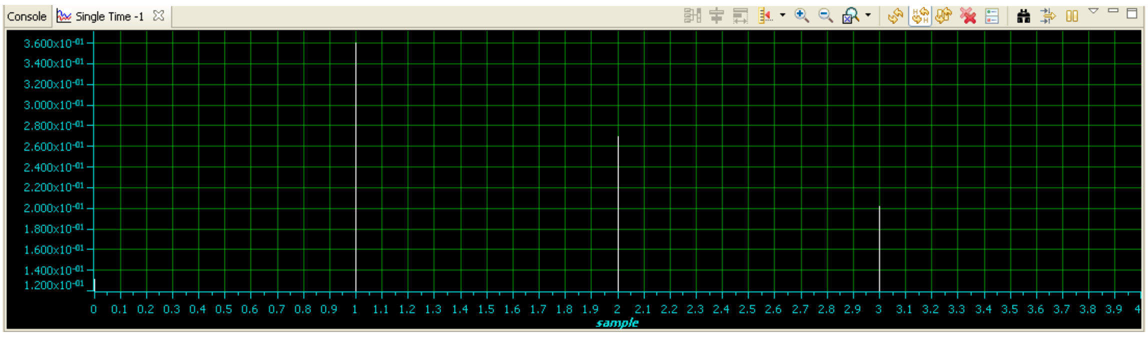
Output:

```
0.131100 0.360237 0.269385 0.201446  0.150641
```

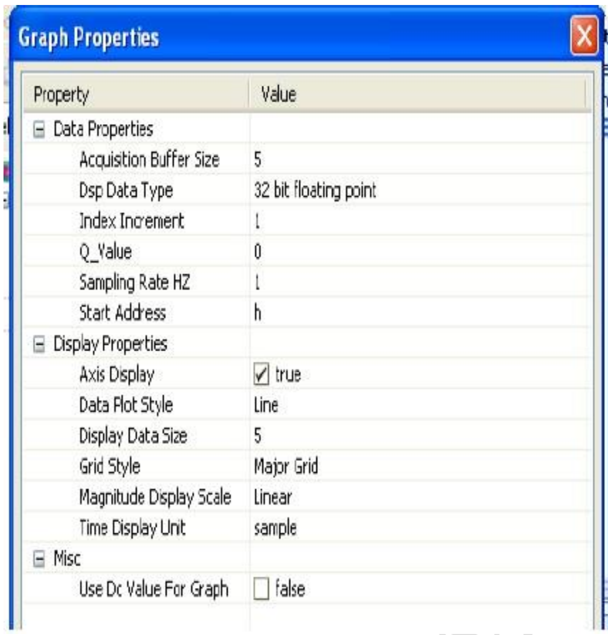


```
Texas Instruments XDS100v1 USB Emulator_0/C55xx [Non-Project Debug Session] Texas Instruments XDS100v1 USB Emulator_0/C55
0.131100        0.360237        0.269385        0.201446        0.150641
```



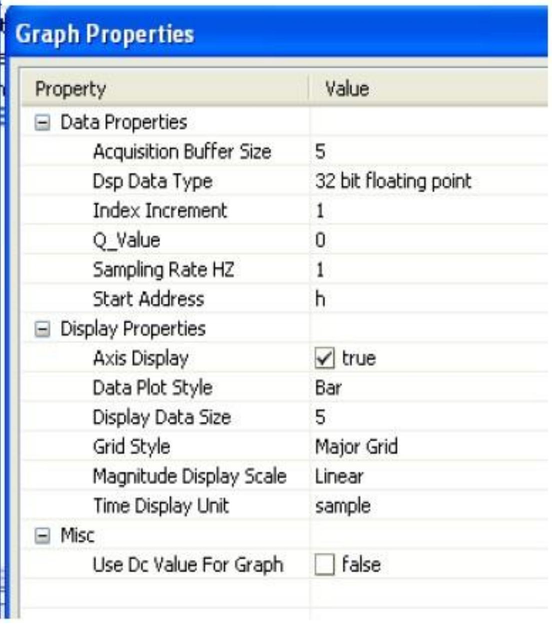


Graph Properties:



Graph Properties

Property	Value
⊟ Data Properties	
Acquisition Buffer Size	5
Dsp Data Type	32 bit floating point
Index Increment	1
Q_Value	0
Sampling Rate HZ	1
Start Address	h
⊟ Display Properties	
Axis Display	✓ true
Data Plot Style	Line
Display Data Size	5
Grid Style	Major Grid
Magnitude Display Scale	Linear
Time Display Unit	sample
⊟ Misc	
Use Dc Value For Graph	☐ false



Graph Properties

Property	Value
⊟ Data Properties	
Acquisition Buffer Size	5
Dsp Data Type	32 bit floating point
Index Increment	1
Q_Value	0
Sampling Rate HZ	1
Start Address	h
⊟ Display Properties	
Axis Display	✓ true
Data Plot Style	Bar
Display Data Size	5
Grid Style	Major Grid
Magnitude Display Scale	Linear
Time Display Unit	sample
⊟ Misc	
Use Dc Value For Graph	☐ false

OBSEVATIONS

Output Waveforms:

EXP. NO: 11
AUDIO APPLICATIONS

Aim: Audio applications such as to plot a time and frequency display of microphone plus a cosine using DSP. Read a wav file and match with their respective spectrograms.

Program:

```
#include <math.h>
#include <stdio.h>
#include "usbstk5515.h"
#define PTS 64
typedef struct {float real,imag;} COMPLEX;
void FFT(COMPLEX *Y, int n);
void cosine_plot(void);
Int16 aic3204_test( );
extern Int16 dly0[PTS];
COMPLEX data[PTS],samples[PTS];
COMPLEX w[PTS];
#define PI 3.14159265358979
float x1[PTS],y1[PTS],iobuffer[PTS];
int i;
void main( void )
{
/* Initialize BSL */
USBSTK5515_init();
for (i = 0;i<PTS;i++) // set up twiddle constants in w
{
w[i].real = cos(2*PI*i/(PTS*2.0)); //Re component of twiddle constants
w[i].imag =-sin(2*PI*i/(PTS*2.0)); //Im component of twiddle constants
}
printf( "Cosine Plot ::::: \n" );
cosine_plot();
printf( "playing audio ::::: \n" );
while(1)
{
```

```
aic3204_test( );
//for(i=0;i<PTS;i++)
// printf( "%d \n",dly0[i]);
for(i=0;i<PTS;i++)
{
data[i].real = dly0[i];
data[i].imag = 0.0;
}
FFT(data,PTS);
for (i = 0 ; i < PTS ; i++) //compute magnitude
{
x1[i] = sqrt(data[i].real*data[i].real + data[i].imag*data[i].imag);
printf("\n%d = %f",i,x1[i]);
}
}
}
void cosine_plot()
{
int i;
for (i = 0 ; i < PTS ; i++) //swap buffers
{
iobuffer[i] = cos(2*PI*2*i/64.0);/*10- > freq, 64 -> sampling freq*/
printf("\n%d = %f",i,iobuffer[i]);
samples[i].real=0.0;
samples[i].imag=0.0;
}
for (i = 0 ; i < PTS ; i++) //swap buffers
{
samples[i].real=iobuffer[i]; //buffer with new data
}
for (i = 0 ; i < PTS ; i++)
samples[i].imag = 0.0; //imag components = 0
FFT(samples,PTS); //call function FFT.c
printf("\n output");
```

```
for (i = 0 ; i < PTS ; i++) //compute magnitude
{
y1[i] = sqrt(samples[i].real*samples[i].real + samples[i].imag*samples[i].imag);
printf("\n%d = %f",i,x1[i]);
}
}
```

aic3204_loop_stereo_in1.C

```
#include "stdio.h"
#include "usbstk5515.h"
extern Int16 AIC3204_rset( Uint16 regnum, Uint16 regval);
#define Rcv 0x08
#define Xmit 0x20
#define N 64
Int16 data1, data2, data3, data4;
int sample,n,k,l;
Int16 dly0[N];
Int32 lyn,ryn;
Int16 aic3204_loop_stereo_in1( )
{
// Int16 j, i = 0;
/* Configure AIC3204 */
AIC3204_rset( 0, 0 ); // Select page 0
AIC3204_rset( 1, 1 ); // Reset codec
AIC3204_rset( 0, 1 ); // Point to page 1
AIC3204_rset( 1, 8 ); // Disable crude AVDD generation from DVDD
AIC3204_rset( 2, 1 ); // Enable Analog Blocks, use LDO power
AIC3204_rset( 0, 0 ); // Select page 0
/* PLL and Clocks config and Power Up */
AIC3204_rset( 27, 0x0d ); // BCLK and WCLK is set as o/p to AIC3204(Master)
AIC3204_rset( 28, 0x00 ); // Data ofset = 0
AIC3204_rset( 4, 3 ); // PLL setting: PLLCLK <- MCLK, CODEC_CLKIN <-PLL CLK
AIC3204_rset( 6, 8 ); // PLL setting: J=8
AIC3204_rset( 7, 15 ); // PLL setting: HI_BYTE(D)
AIC3204_rset( 8, 0xdc ); // PLL setting: LO_BYTE(D)
```

```
AIC3204_rset( 30, 0x88 ); // For 32 bit clocks per frame in Master mode ONLY
// BCLK=DAC_CLK/N =(12288000/8) = 1.536MHz = 32*fs
AIC3204_rset( 5, 0x91 ); // PLL setting: Power up PLL, P=1 and R=1
AIC3204_rset( 13, 0 );// Hi_Byte(DOSR) for DOSR = 128 decimal or 0x0080 DAC
oversamppling
AIC3204_rset( 14, 0x80 ); // Lo_Byte(DOSR) for DOSR = 128 decimal or 0x0080
AIC3204_rset( 20, 0x80 ); // AOSR for AOSR = 128 decimal or 0x0080 for decimation
filters 1 to 6
AIC3204_rset( 11, 0x88 ); // Power up NDAC and set NDAC value to 8
AIC3204_rset( 12, 0x82 ); // Power up MDAC and set MDAC value to 2
AIC3204_rset( 18, 0x88 ); // Power up NADC and set NADC value to 8
AIC3204_rset( 19, 0x82 ); // Power up MADC and set MADC value to 2
/* DAC ROUTING and Power Up */
AIC3204_rset( 0, 0x01 ); // Select page 1
AIC3204_rset( 12, 0x08 ); // LDAC AFIR routed to HPL
AIC3204_rset( 13, 0x08 ); // RDAC AFIR routed to HPR
AIC3204_rset( 0, 0x00 ); // Select page 0
AIC3204_rset( 64, 0x02 ); // Left vol=right vol
AIC3204_rset( 65, 0x00 ); // Left DAC gain to 0dB VOL; Right tracks Left
AIC3204_rset( 63, 0xd4 ); // Power up left,right data paths and set channel
AIC3204_rset( 0, 0x01 ); // Select page 1
AIC3204_rset( 16, 0x06 ); // Unmute HPL , 6dB gain
AIC3204_rset( 17, 0x06 ); // Unmute HPR , 6dB gain
AIC3204_rset( 9, 0x30 ); // Power up HPL,HPR
AIC3204_rset( 0, 0x00 ); // Select page 0
USBSTK5515_wait( 500 ); // Wait
/* ADC ROUTING and Power Up */
AIC3204_rset( 0, 1 ); // Select page 1
AIC3204_rset( 0x34, 0x30 ); // STEREO 1 Jack
// IN2_L to LADC_P through 40 kohm
AIC3204_rset( 0x37, 0x30 ); // IN2_R to RADC_P through 40 kohmm
AIC3204_rset( 0x36, 3 ); // CM_1 (common mode) to LADC_M through 40 kohm
AIC3204_rset( 0x39, 0xc0 ); // CM_1 (common mode) to RADC_M through 40 kohm
AIC3204_rset( 0x3b, 0 ); // MIC_PGA_L unmute
```

```
AIC3204_rset( 0x3c, 0 ); // MIC_PGA_R unmute
AIC3204_rset( 0, 0 ); // Select page 0
AIC3204_rset( 0x51, 0xc0 ); // Powerup Left and Right ADC
AIC3204_rset( 0x52, 0 ); // Unmute Left and Right ADC
AIC3204_rset( 0, 0 );
USBSTK5515_wait( 200 ); // Wait
/* I2S settings */
I2S0_SRGR = 0x0;
I2S0_CR = 0x8010; // 16-bit word, slave, enable I2C
I2S0_ICMR = 0x3f; // Enable interrupts
/* Play Tone */
for(l=0;l<N;l++)
dly0[l]=0;
for ( sample = 0; sample < N ; sample++ )
{
USBSTK5515_waitusec(25);
/* Read Digital audio input */
data3 = I2S0_W0_MSW_R; // 16 bit left channel received audio data
data1 = I2S0_W0_LSW_R;
data4 = I2S0_W1_MSW_R; // 16 bit right channel received audio data
data2 = I2S0_W1_LSW_R;
while((Rcv & I2S0_IR) == 0); // Wait for interrupt pending flag
// printf("data3.. %d \n",data3);
USBSTK5515_wait(5000);
dly0[sample] = data3;
/* Write Digital audio input */
I2S0_W0_MSW_W = data3; // 16 bit left channel transmit audio data
I2S0_W0_LSW_W = 0;
I2S0_W1_MSW_W = data3; // 16 bit right channel transmit audio data
I2S0_W1_LSW_W = 0;
while((Xmit & I2S0_IR) == 0); // Wait for interrupt pending flag
}
/* Disble I2S */
I2S0_CR = 0x00;
```

```
return 0;
}
aic3204_test.c
#define AIC3204_I2C_ADDR 0x18
#include "usbstk5515.h"
#include "usbstk5515_gpio.h"
#include "usbstk5515_i2c.h"
#include "stdio.h"
extern Int16 aic3204_tone_headphone( );
extern Int16 aic3204_loop_stereo_in1( );
Int16 AIC3204_rget( Uint16 regnum, Uint16* regval )
{
Int16 retcode = 0;
Uint8 cmd[2];
cmd[0] = regnum & 0x007F; // 7-bit Register Address
cmd[1] = 0;
retcode |= USBSTK5515_I2C_write( AIC3204_I2C_ADDR, cmd, 1 );
retcode |= USBSTK5515_I2C_read( AIC3204_I2C_ADDR, cmd, 1 );
*regval = cmd[0];
USBSTK5515_wait( 10 );
return retcode;
}
Int16 AIC3204_rset( Uint16 regnum, Uint16 regval )
{
Uint8 cmd[2];
cmd[0] = regnum & 0x007F; // 7-bit Register Address
cmd[1] = regval; // 8-bit Register Data
return USBSTK5515_I2C_write( AIC3204_I2C_ADDR, cmd, 2 );
}
Int16 aic3204_test( )
{
SYS_EXBUSSEL = 0x6100; // Enable I2C bus
USBSTK5515_I2C_init( ); // Initialize I2C
USBSTK5515_wait( 100 ); // Wait
```

```
if ( aic3204_loop_stereo_in1( ) )
return 1;
return 0;
}
```

FFT.C

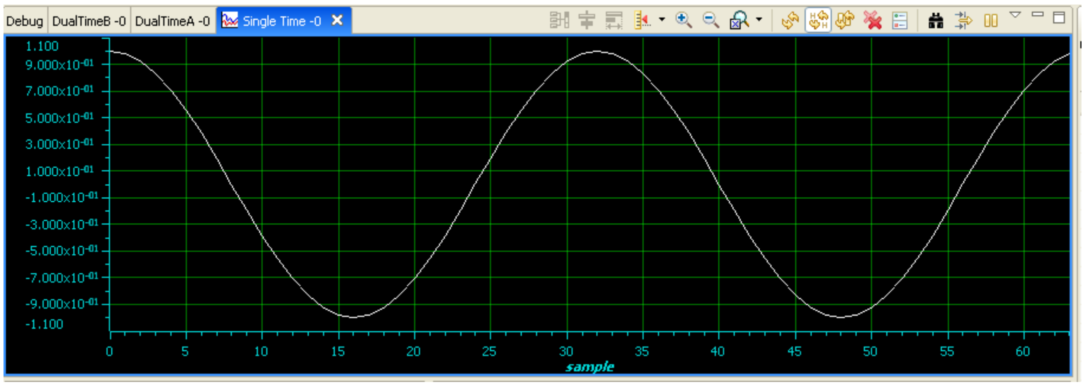
Refer previous FFT experiment

Execution Procedure:

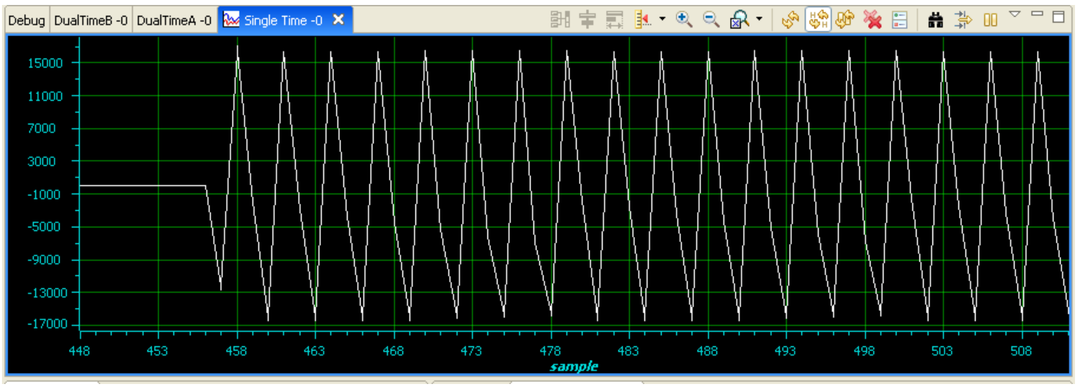
- Open CCstudio setup
- Go to File Menu , select Import option.
- In the Import Window under CCs choose Existing CCS/CCE Eclipse project then next.
- In Select root Directory Browse for project file where it is located.
- Select Audio Project folder and Finish it.
- Now Connect the DSP Kit to PC and Launch it.(Follow the above given manual procedure
- from Step 46 to 51)
- Give Right Click on Your Audio.out file under Binaries and select Load program Option.
- Now Go to Target select Run.
- Observe the audio (Give Input from Stereo in and listen Filtered Output
- from Stereo Out).
- In Tools menu select Graph (Dual Time) set the properties and watch.

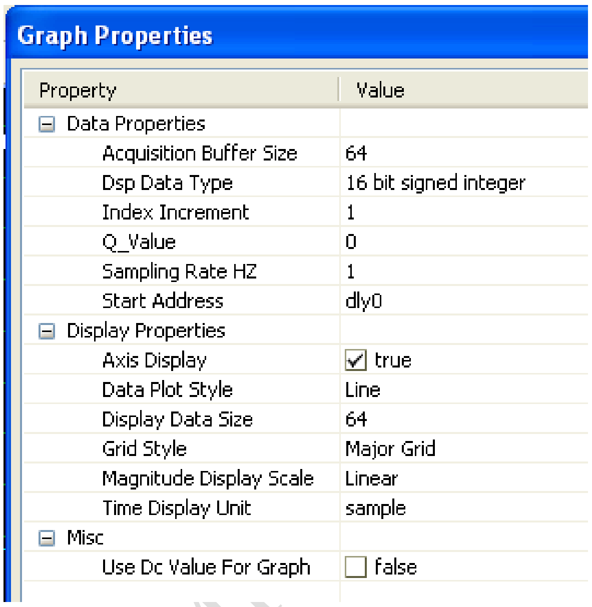
Result:

Cosine time plot



Audio time plot

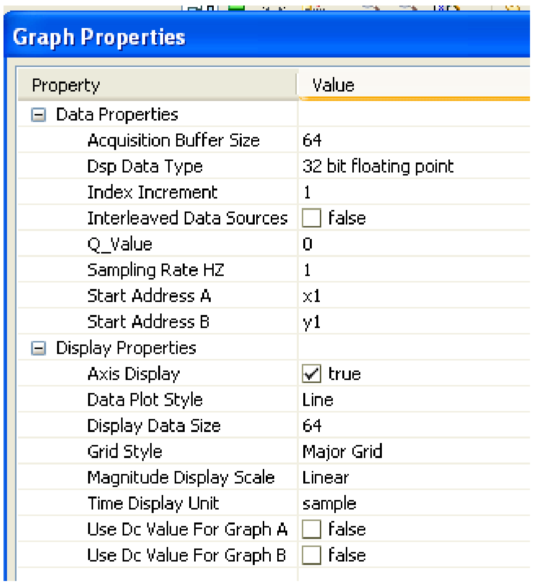




Graph Properties

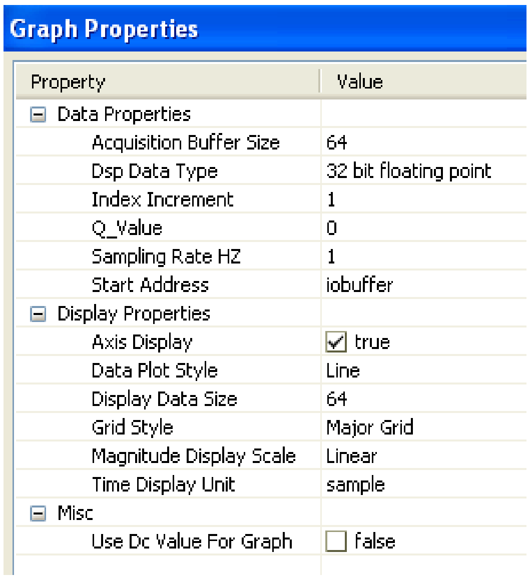
Property	Value
⊟ Data Properties	
Acquisition Buffer Size	64
Dsp Data Type	16 bit signed integer
Index Increment	1
Q_Value	0
Sampling Rate HZ	1
Start Address	dly0
⊟ Display Properties	
Axis Display	☑ true
Data Plot Style	Line
Display Data Size	64
Grid Style	Major Grid
Magnitude Display Scale	Linear
Time Display Unit	sample
⊟ Misc	
Use Dc Value For Graph	☐ false

Audio time plot **Frequency plot**



Graph Properties

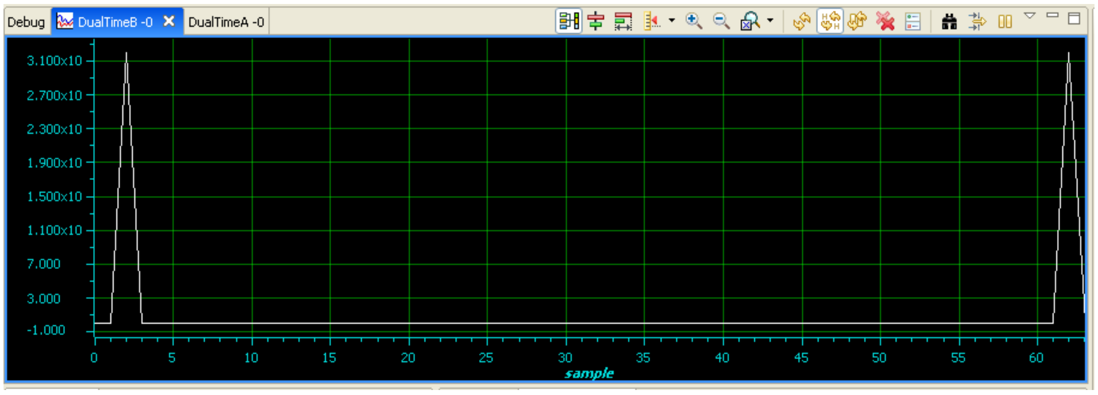
Property	Value
⊟ Data Properties	
Acquisition Buffer Size	64
Dsp Data Type	32 bit floating point
Index Increment	1
Interleaved Data Sources	☐ false
Q_Value	0
Sampling Rate HZ	1
Start Address A	x1
Start Address B	y1
⊟ Display Properties	
Axis Display	☑ true
Data Plot Style	Line
Display Data Size	64
Grid Style	Major Grid
Magnitude Display Scale	Linear
Time Display Unit	sample
Use Dc Value For Graph A	☐ false
Use Dc Value For Graph B	☐ false



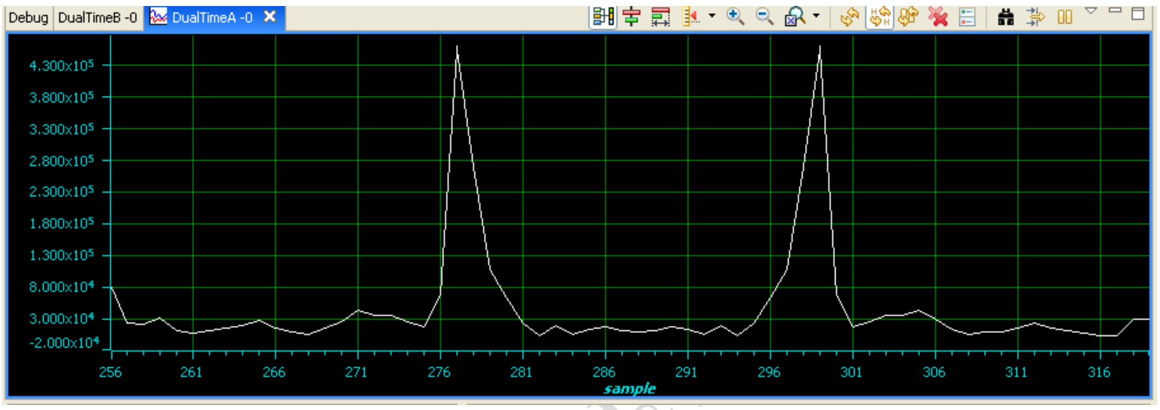
Graph Properties

Property	Value
⊟ Data Properties	
Acquisition Buffer Size	64
Dsp Data Type	32 bit floating point
Index Increment	1
Q_Value	0
Sampling Rate HZ	1
Start Address	iobuffer
⊟ Display Properties	
Axis Display	☑ true
Data Plot Style	Line
Display Data Size	64
Grid Style	Major Grid
Magnitude Display Scale	Linear
Time Display Unit	sample
⊟ Misc	
Use Dc Value For Graph	☐ false

Cosine time plot



Cosine Frequency plot



OBSERVATIONS

EXP. NO.12
NOISE REMOVAL

Noise removal programs:

(i) Add noise above 3kHz and then remove (using adaptive filters)

(ii) Interference suppression using 400 Hz tone

Program code:

Main.c

```
#include<usbstk5515.h>
#include<stdio.h>
Int16 aic3204_test( );
void main()
{
USBSTK5515_init();
while(1)
{
aic3204_test();
}
}
```

aic3204_loop_stereo_in1.c

```
#include "stdio.h"
#include "usbstk5515.h"
#include<math.h>
extern Int16 AIC3204_rset( Uint16 regnum, Uint16 regval);
#define Rcv 0x08
#define Xmit 0x20
#define beta 1.5E-8 /*rate of convergence */
#define N1 30 /*# of coefficients */
#define NS 128 /*# of output sample points*/
#define Fs 8000 /*sampling frequency */
#define pi 3.1415926
//#define DESIRED 1000*sin(2*pi*T*1000/Fs) /*desired signal */
#define ADDNOISE 1000*sin(2*pi*T*3000/Fs) /*additive noise */
#define REFNOISE 1000*cos(2*pi*T*3000/Fs) /*reference noise*/
Int16 data1, data2, data3, data4;
```

```
int sample,n,k,l;
Int16 dly0[NS];
Int32 lyn,ryn;
int I,T;
float W[N1+1];
float Delay[N1+1];
float Y, E, DPLUSN,IO_INPUT[NS],IO_OUTPUT[NS],a[NS],b[NS];
Int16 aic3204_loop_stereo_in1( )
    {
// Int16 j, i = 0;
/* Configure AIC3204 */
AIC3204_rset( 0, 0 ); // Select page 0
AIC3204_rset( 1, 1 ); // Reset codec
AIC3204_rset( 0, 1 ); // Point to page 1
AIC3204_rset( 1, 8 ); // Disable crude AVDD generation from DVDD
AIC3204_rset( 2, 1 ); // Enable Analog Blocks, use LDO power
AIC3204_rset( 0, 0 ); // Select page 0
/* PLL and Clocks config and Power Up */
AIC3204_rset( 27, 0x0d ); // BCLK and WCLK is set as o/p to AIC3204(Master)
AIC3204_rset( 28, 0x00 ); // Data ofset = 0
AIC3204_rset( 4, 3 ); // PLL setting: PLLCLK <- MCLK, CODEC_CLKIN <-PLL CLK
AIC3204_rset( 6, 8 ); // PLL setting: J=8
AIC3204_rset( 7, 15 ); // PLL setting: HI_BYTE(D)
AIC3204_rset( 8, 0xdc ); // PLL setting: LO_BYTE(D)
AIC3204_rset( 30, 0x88 ); // For 32 bit clocks per frame in Master mode ONLY
// BCLK=DAC_CLK/N =(12288000/8) = 1.536MHz = 32*fs
AIC3204_rset( 5, 0x91 ); // PLL setting: Power up PLL, P=1 and R=1
AIC3204_rset( 13, 0 );// Hi_Byte(DOSR) for DOSR = 128 decimal or 0x0080 DAC
oversamppling
AIC3204_rset( 14, 0x80 ); // Lo_Byte(DOSR) for DOSR = 128 decimal or 0x0080
AIC3204_rset( 20, 0x80 ); // AOSR for AOSR = 128 decimal or 0x0080 for decimation
filters 1 to 6
AIC3204_rset( 11, 0x88 ); // Power up NDAC and set NDAC value to 8
    AIC3204_rset( 12, 0x82 ); // Power up MDAC and set MDAC value to 2
```

```
AIC3204_rset( 18, 0x88 ); // Power up NADC and set NADC value to 8
AIC3204_rset( 19, 0x82 ); // Power up MADC and set MADC value to 2
/* DAC ROUTING and Power Up */
AIC3204_rset( 0, 0x01 ); // Select page 1
AIC3204_rset( 12, 0x08 ); // LDAC AFIR routed to HPL
AIC3204_rset( 13, 0x08 ); // RDAC AFIR routed to HPR
AIC3204_rset( 0, 0x00 ); // Select page 0
AIC3204_rset( 64, 0x02 ); // Left vol=right vol
AIC3204_rset( 65, 0x00 ); // Left DAC gain to 0dB VOL; Right tracks Left
AIC3204_rset( 63, 0xd4 ); // Power up left,right data paths and set channel
AIC3204_rset( 0, 0x01 ); // Select page 1
AIC3204_rset( 16, 0x06 ); // Unmute HPL , 6dB gain
AIC3204_rset( 17, 0x06 ); // Unmute HPR , 6dB gain
AIC3204_rset( 9, 0x30 ); // Power up HPL,HPR
AIC3204_rset( 0, 0x00 ); // Select page 0
USBSTK5515_wait( 500 ); // Wait
/* ADC ROUTING and Power Up */
AIC3204_rset( 0, 1 ); // Select page 1
AIC3204_rset( 0x34, 0x30 ); // STEREO 1 Jack
// IN2_L to LADC_P through 40 kohm
AIC3204_rset( 0x37, 0x30 ); // IN2_R to RADC_P through 40 kohmm
AIC3204_rset( 0x36, 3 ); // CM_1 (common mode) to LADC_M through 40 kohm
AIC3204_rset( 0x39, 0xc0 ); // CM_1 (common mode) to RADC_M through 40 kohm
AIC3204_rset( 0x3b, 0 ); // MIC_PGA_L unmute
    AIC3204_rset( 0x3c, 0 ); // MIC_PGA_R unmute
AIC3204_rset( 0, 0 ); // Select page 0
AIC3204_rset( 0x51, 0xc0 ); // Powerup Left and Right ADC
AIC3204_rset( 0x52, 0 ); // Unmute Left and Right ADC
AIC3204_rset( 0, 0 );
USBSTK5515_wait( 200 ); // Wait
/* I2S settings */
I2S0_SRGR = 0x0;
I2S0_CR = 0x8010; // 16-bit word, slave, enable I2C
I2S0_ICMR = 0x3f; // Enable interrupts
```

```
/* Play Tone */
for(l=0;l<NS;l++)
dly0[l]=0;
for(T=0;T<N1;T++)
{
W[T] = 0.0;
Delay[T] = 0.0;
}
for ( T = 0; T < NS ; T++ )
{ USBSTK5515_waitusec(25);
/* Read Digital audio input */
data3 = I2S0_W0_MSW_R; // 16 bit left channel received audio data
data1 = I2S0_W0_LSW_R;
data4 = I2S0_W1_MSW_R; // 16 bit right channel received audio data
data2 = I2S0_W1_LSW_R;
    while((Rcv & I2S0_IR) == 0); // Wait for interrupt pending flag
// printf("data3.. %d \n",data3);
USBSTK5515_wait(5000);
dly0[T] = data3;
Delay[0] = REFNOISE; /*adaptive filter's input*/
b[T]=DPLUSN = dly0[T]+ADDNOISE; /*desired + noise, d+n */
Y = 0;
for(I=0;I<N1;I++)
Y += (W[I] * Delay[I]); /*adaptive filter output */
E = DPLUSN-Y; /*error signal */
for(I=N1;I>0;I--)
{
W[I] = W[I]+(beta*E*Delay[I]); /*update weights */
if (I != 0)
Delay[I] = Delay[I-1]; /*update samples */
}
/* Write Digital audio input */
I2S0_W0_MSW_W = E; // 16 bit left channel transmit audio data
I2S0_W0_LSW_W = 0;
I2S0_W1_MSW_W = E; // 16 bit right channel transmit audio data
I2S0_W1_LSW_W = 0;
while((Xmit & I2S0_IR) == 0); // Wait for interrupt pending flag
}
/* Disble I2S */
    I2S0_CR = 0x00;
return 0;
}
/*
void apply_noise()
{
for(T=0;T<NS;T++) //# of output samples
{
a[T] = dly0[T];
IO_OUTPUT[T] = E; //overall output E
IO_INPUT[T]= DPLUSN; // store d + n
printf("%d %f %f\n",T,IO_INPUT[T],IO_OUTPUT[T]);
}
}*/
```

And one more .c file same as aic3204_test.c in the above experiment.

Execution Procedure:

- Open CCstudio setup
- Go to File Menu , select Import option.
- In the Import Window under CCs choose Existing CCS/CCE Eclipse project then next.
- In Select root Directory Browse for project file where it is located.
- Select FIR Project folder and Finish it.
- Now Connect the DSP Kit to PC and Launch it.(Follow the above given manual procedure
- from Step 46 to 51)
- Give Right Click on Your fir.out file under Binaries and select Load program Option.
- Now Go to Target select Run.
- Observe the audio (Give Input from Stereo in and listen Filtered Output
- from Stereo Out).

OBSEVATIONS

Output Waveforms: