SIDDHARTHA
INSTITUTE OF TECHNOLOGY & SCIENCES
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LABORATORY MANUAL

For

DIGITAL SIGNAL PROCESSING

(III B. Tech ECE- II Semester- R18 .AY:2021 - 22)

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SIDDHARTHA INSTITUTE OF TECHNOLOGY & SCIENCES

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SIDDHARTHA INSTITUTE OF TECHNOLOGY & SCIENCES

VISION & MISSION OF THE INSTITUTE

VISION:

To be a Centre of Excellence in Technical Education and to become an epic center of Research for creative solutions.

MISSION:

To address the Emerging Needs through Quality Technical Education with an emphasis on practical skills and Advanced Research with social relevance.

OBJECTIVES:

- To translate our vision into action and accomplish our mission, we strive to provide state-of-art infrastructure.
- Recruit, Motivate and develop faculty of high caliber and with multiple specialization.
- Continuously review, innovate and experiment teaching methodologies and learning processes.
- Focus on research, training and consultancy through an Integrated Institute-Industry symbiosis.

SIDDHARTHA INSTITUTE OF TECHNOLOGY & SCIENCES

VISION & MISSION OF THE DEPARTMENT

VISION :

To provide innovative teaching and learning methodologies for excelling in a high-value career, higher education and research to the students in the field of Electronics and Communication Engineering to meet the needs of the industry and to be a part of the advancing technological revolution.

MISSION :

- To create engineers of high quality on par with international standards by providing excellent infrastructure and well qualified faculty.
- To establish centers of excellence to enhance collaborative and multidisciplinary activities to develop human and intellectual qualities.
- To provide technical expertise to carry out research and development.

PROGRAM EDUCATIONAL OBJECTIVES (PEOS) :

Graduates shall apply the fundamental, advanced and contemporary knowledge of

1. Electronics, Communication and allied Engineering, to develop efficient solutions and systems, to meet the needs of the industries and society.
2. Graduates will get employed or pursue higher studies or research.
3. Graduates will have team spirit, good communication skills and ethics with lifelong learning attitude.

SIDDHARTHA INSTITUTE OF TECHNOLOGY & SCIENCES

PROGRAM OUTCOMES:

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 multidisciplinary environments.
12. **Life-long learning:** Recognize the need for, and have the preparation and ability to engage in independent and life-long learning in the broadest context of technological change.

RULES AND REGULATIONS OF LAB

All students must observe the Dress Code while in the laboratory.

- All bags must be placed at rack.
- The lab timetable must be strictly followed.
- Be PUNCTUAL for your laboratory session.
- Program/experiment must be executed within the given time.
- Workspace must be kept clean and tidy at all time.
- Handle the systems and interfacing kits with care.
- All students are liable for any damage to the accessories due to their own negligence.
- All interfacing kits connecting cables must be RETURNED if you taken from the lab supervisor.
- Students are strictly PROHIBITED from taking out any items from the laboratory.
- Students are NOT allowed to work alone in the laboratory without the Lab Supervisor
- USB Ports have been disabled if you want to use USB drive consult lab supervisor.
- Report immediately to the Lab Supervisor if any malfunction of the accessories, is there

Before leaving the lab

- Place the chairs properly.
- Turn off the system properly
- Turn off the monitor.
- Please check the laboratory notice board regularly for updates.

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.

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

PROGRAM:

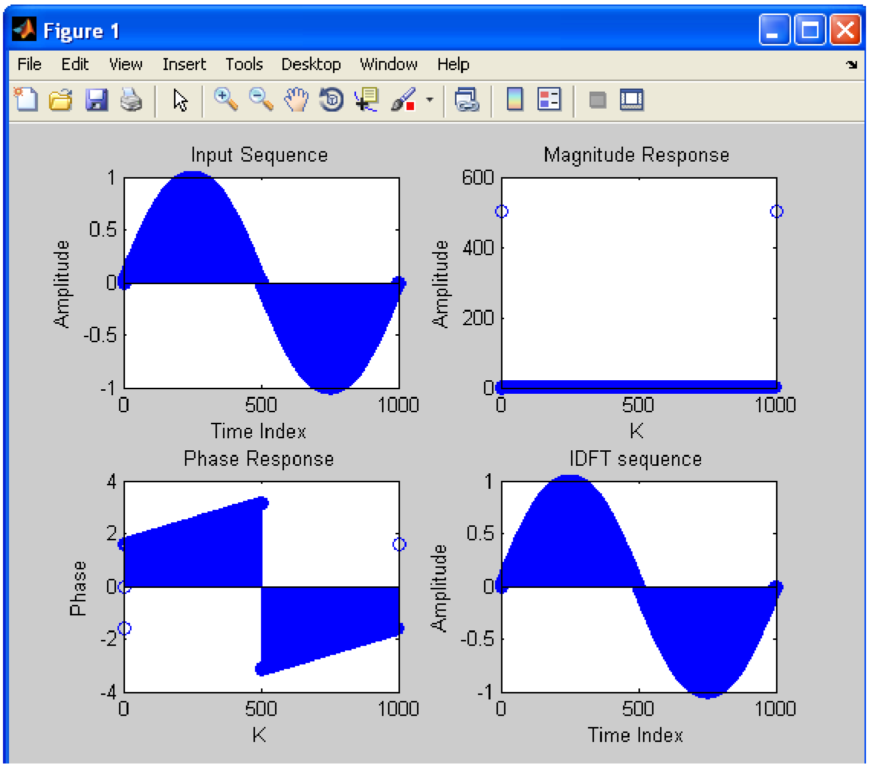
```
%% To find DFT/IDFT of a given signal/sequence
 clc;
close all;
clear all;
ts = 0:1e-3:1;
in = input('Enter 1 for signal and 2 for sequence');
if(in==1)
    xn = sin(2*pi*ts);
elseif (in==2)
    xn = randint(1,10,[1 10]);
else
    disp('Wrong value entered');
    return;
end
%xn=input('Enter the sequence x(n)'); %Get the sequence from user
```

```
ln=length(xn); %find the length of the sequence
xk=zeros(1,ln); %initilise an array of same size as that of input sequence
ixk=zeros(1,ln); %initilise 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;
```

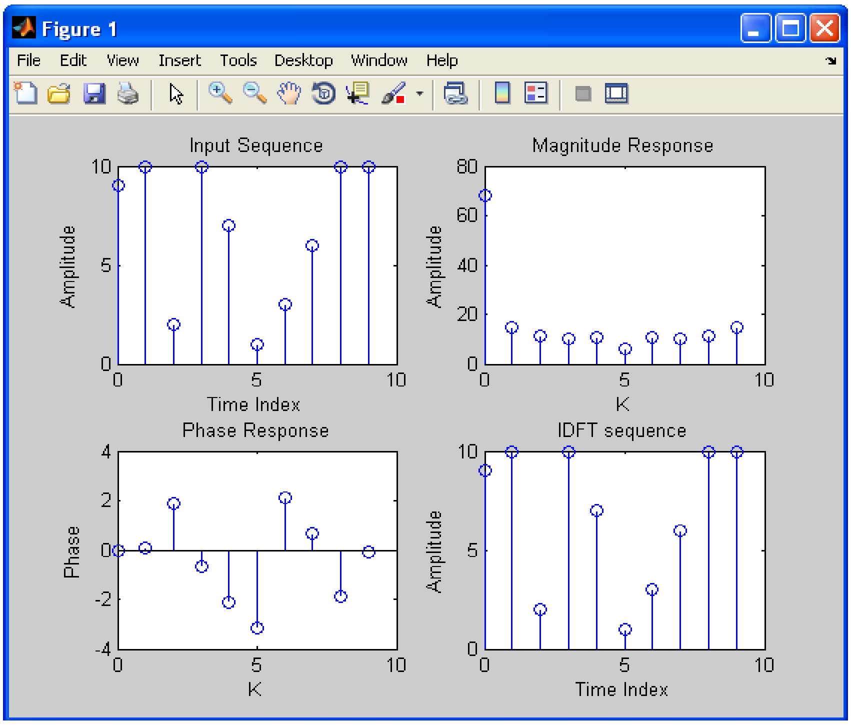
```
subplot(223);
stem(t,phase);
ylabel ('Phase');
xlabel ('K');
title ('Phase Response');
%.....................................................................................
 %IDFT of the sequence
 %.....................................................................................
for n=0:n-1
    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,xn);
ylabel ('Amplitude');
xlabel ('Time Index');
title ('IDFT sequence');
%.....................................................................................
```

SIMULATION RESULT

- **DFT and IDFT of Sinosoidal Signal**



- **DFT and IDFT of a random sequence**



EXP.NO: 2
TO FIND FREQUENCY RESPONSE OF A GIVEN SYSTEM GIVEN IN (TRANSFER FUNCTION/ DIFFERENTIAL EQUATION FORM).

AIM: To obtain the impulse response/step response of a system described by the given difference equation

Software: MATLAB

THEORY:

- A difference equation with constant coefficients describes a LTI system. For example the difference equation **y[n] + 0.8y[n-2] + 0.6y[n-3] = x[n] + 0.7x[n-1] + 0.5x[n-2]** describes a LTI system of order 3. The coefficients 0.8, 0.7, etc are all constant i.e., they are not functions of time (n). The difference equation y[n]+0.3ny[n-1]=x[n] describes a time varying system as the coefficient 0.3n is not constant.
- The difference equation can be solved to obtain y[n], the output for a given input x[n] by rearranging as y[n] = x[n] + 0.7x[n-1]+0.5x[n-2]- 0.8y[n-2]- 0.6y[n-3] and solving.
- The output depends on the input x[n]
    - With **x[n]= δ[n],** an impulse, the computed output y[n] is the **impulse response**.
    - If **x[n]=u[n]**, a **step response** is obtained.
    - If **x[n] = cos(wn)** is a sinusoidal sequence, a **steady state response** is obtained (wherein y[n] is of the same frequency as x[n], with only an amplitude gain and phase shift-refer Fig.7.3).
    - Similarly for any arbitrary sequence of x[n], the corresponding output response y[n] is computed.
- The difference equation containing past samples of output, i.e., y[n-1], y[n-2], etc leads to a recursive system, whose impulse response is of infinite duration (IIR). For such systems the impulse response is computed for a large value of n, say n=100 (to approximate n=∞). The MATLAB function filter is used to compute the impulse response/ step response/ response to any given x[n]. Note: The filter function evaluates the convolution of an infinite sequence (IIR) and x[n], which is not possible with conv function (remember conv requires both the sequences to be finite).

- The difference equation having only y[n] and present and past samples of input (x[n], x[n-k]), represents a system whose impulse response is of finite duration (FIR). The response of FIR systems can be obtained by both the „conv" and „filter" functions. The filter function results in a response whose length is equal to that of the input x[n], whereas the output sequence from conv function is of a longer length (xlength + hlength-1).

ALGORITHM:

1. Input the two sequences as a and b representing the coefficients of y and x.
2. If IIR response, then input the length of the response required (say 100, which can be made constant).
3. Compute the output response using the „filter" command.
4. Plot the input sequence & impulse response (and also step response, etc if required).

MATLAB Implementation:

MATLAB has an inbuilt function „***filter***" to solve difference equations numerically, given the input and difference equation coefficients (b,a).

y=filter(b,a,x)

where x is the input sequence, y is the output sequence which is of same length as x.

Given a difference equation $a_0y[n]+a_1y[n-1]+a_2y[n-2]=b_0x[n]+b_2x[n-2]$, the coefficients are written in a vector as $b=[b_0\ 0\ b_2]$ and $a=[a_0\ a_1\ a_2]$. Note the zero in b (x[n-1] term is missing).

Also remember a_0, the coefficient of y[n] should always be 1.

For impulse response $x[n] = \{\underset{\uparrow}{1},0,0,0,....\}$the number of zeros = the length of the IIR response required (say 100 implemented by function zeros(1,99)).

For step response $x[n] = \{\underset{\uparrow}{1},1,1,1,1,........\}$the number of ones = the length of the IIR response required-1 (say 100 implemented by function ones(1,100).

Similarly for any given x[n] sequence, the response y[n] can be calculated.

Given Problem

1)Difference equation y(n) - y(n-1) + 0.9y(n-2) = x(n);

Calculate impulse response h(n) and also step response at n=0,…..,100

2)Plot the steady state response y[n] to $x[n]=\cos(0.05\pi n)u(n)$, given y[n]-0.8y[n-1]=x[n]

MATLAB PROGRAM:

```
clc;
clear all;
close all;
%To find Impulse Response
N=input('Length of response required=');
b=[1];                      %x[n] coefficient
a=[1,-1,0.9];               %y coefficients
x1=[1,zeros(1,N-1)];        %impulse input
n=0:1:N-1;                  %time vector for plotting
h=filter(b,a,x1);           %impulse response
disp('Impulse response');disp(h)
subplot(2,1,1);             %plot the waveforms
stem(n,x1);
title('impulse input');
xlabel('n');
ylabel('x(n)');
subplot(2,1,2);
stem(n,h);
title('impulse response');
xlabel('n');
ylabel('h(n)');

%To find step response
N=input('Length of response required=');
x2=[ones(1,N)];             %step input
y=filter(b,a,x2);           %step response
disp('step response');disp(y);
figure;
subplot(2,1,1);             %plot the waveforms
stem(n,x2);
title('step input');
xlabel('n');
ylabel('u(n)');
subplot(2,1,2);
stem(n,y);
title('step response');
xlabel('n');
ylabel('y(n)');

%To find steady state response:
%y[n]-0.8y[n-1]=x[n]
N=input('Length of response required=');
b=[1];                      % x[n] coefficient
a=[1,-0.8];                 % y coefficients
n=0:1:N-1;                  % time vector
x=cos(0.05*pi*n);           % sinusoidal input
y=filter(b,a,x);            % steady state response
disp('steady state response');disp(y);
figure;
```

```
subplot(2,1,1);                % plot the waveforms
stem(n,x);
title('steady input');
xlabel('n');
ylabel('x(n)');
subplot(2,1,2);
stem(n,y);
title('steady state response');
xlabel('n');
ylabel('y(n)');
```

Output:
Length of response required=10
Impulse response
1.0000 1.0000 0.1000 -0.8000 -0.8900 -0.1700 0.6310 0.7840 0.2161 -0.4895

Length of response required=10
step response
1.0000 2.0000 2.1000 1.3000 0.4100 0.2400 0.8710 1.6550 1.8711 1.3816

Length of response required=35
steady state response
Columns 1 through 12

1.0000 1.7877 2.3812 2.7960 3.0458 3.1437 3.1028 2.9362 2.6580 2.2828 1.8263 1.3046

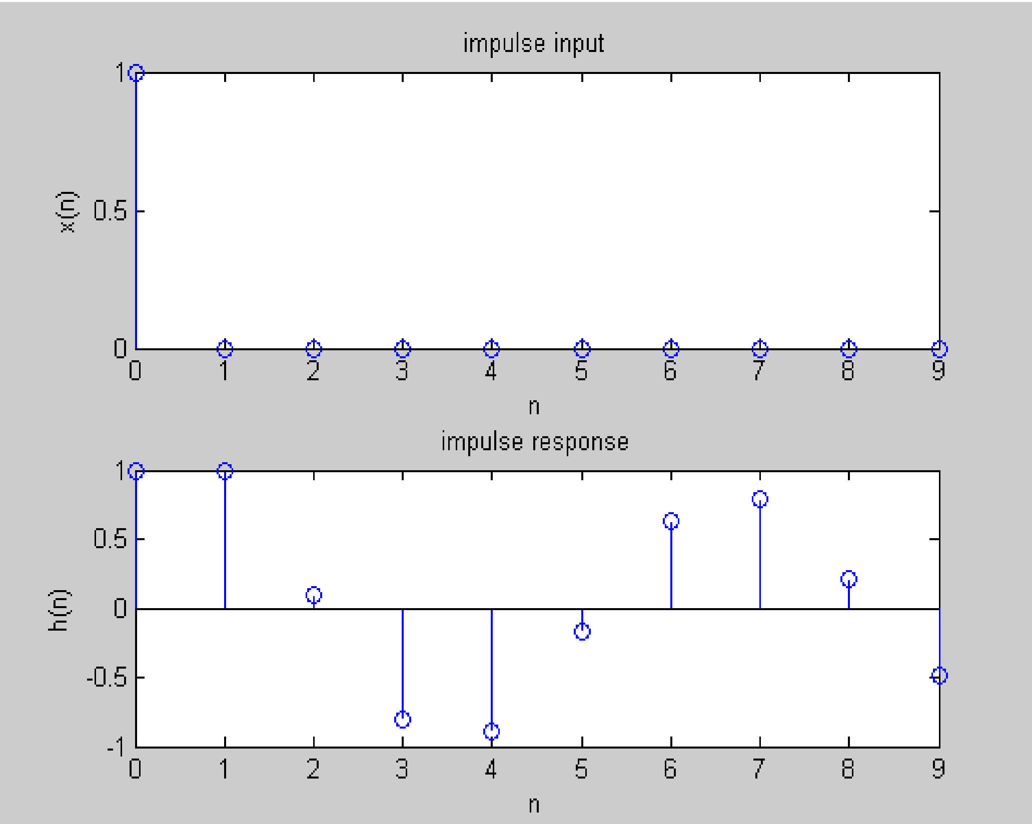
Columns 13 through 24

0.7346 0.1337 -0.4808 -1.0918 -1.6824 -2.2369 -2.7406 -3.1802 -3.5441 -3.8230 -4.0095 -4.0986

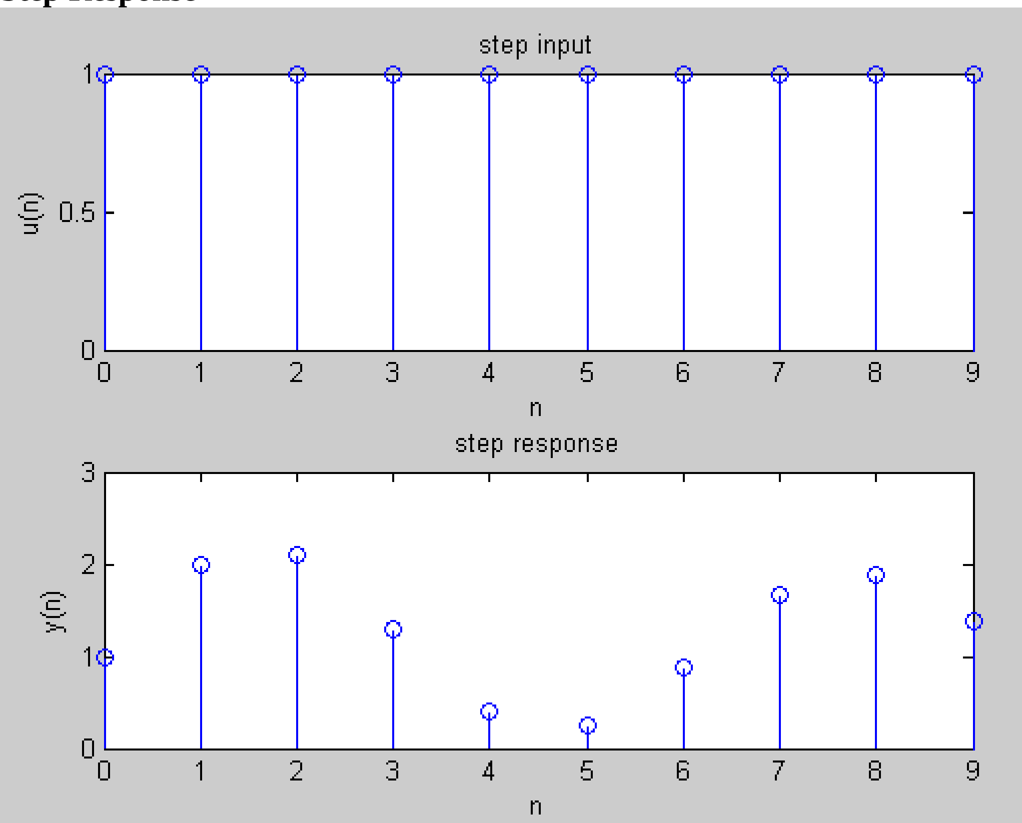
Columns 25 through 35

-4.0879 -3.9774 -3.7697 -3.4698 -3.0848 -2.6243 -2.0994 -1.5231 -0.9095 -0.2736 0.3689

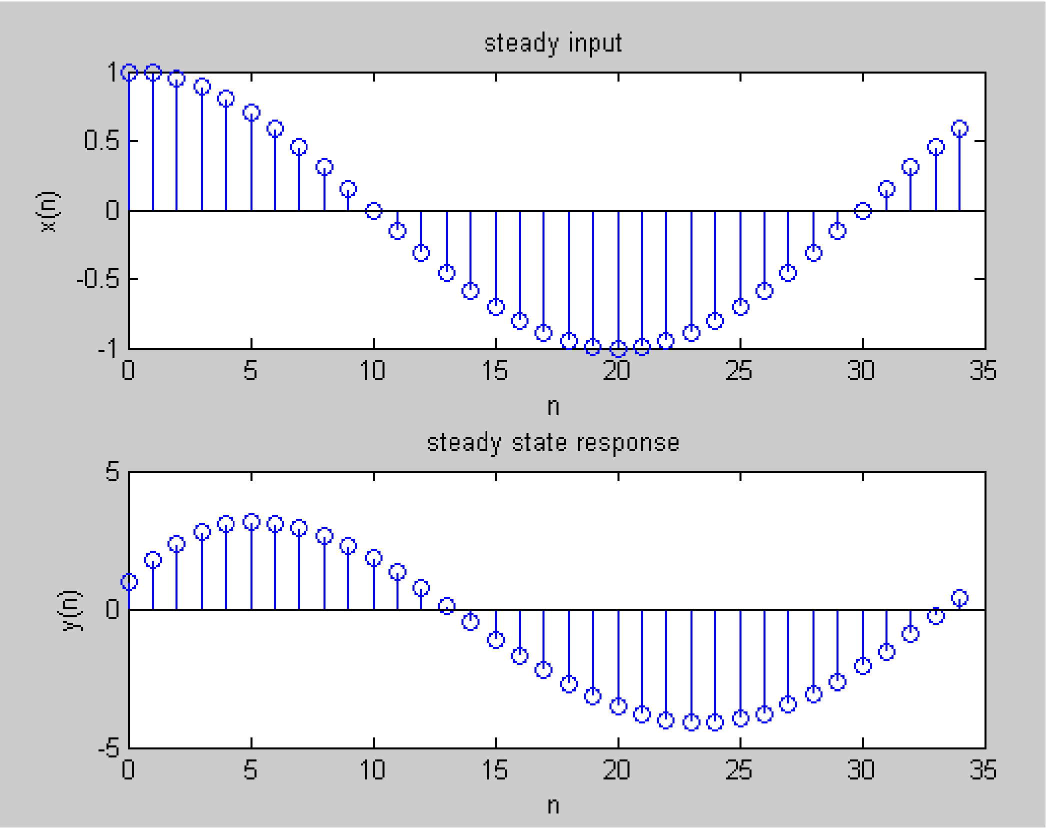
Impulse Response



Step Response



Steady State Response:



EXP.NO: 3

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 transformer .

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.

PROGRAM:

%Finding the Fourier Transform of a given signal and plotting its

%magnitude and phase spectrum.

```
clc;
clear all;
close all;
x=input('Enter the sequence : ')
N=length(x)
xK=fft(x,N)
xn=ifft(xK)
subplot(3,1,1)
```

```
stem(abs(xK))
title('Magnitude of Fourier Spectrum')
xlabel('frequency')
ylabel('Magnitude')
subplot(3,1,2)
stem(angle(xK))
title('Phase of Fourier Spectrum')
xlabel('frequency')
ylabel('phase')
subplot(3,1,3)
stem(xn)
title('Inverse FFT')
xlabel('time')
ylabel('amplitude')
```

Output:

Enter the sequence : [1 2 4 6 8 3 -2 0 7]

x =

1 2 4 6 8 3 -2 0 7

N =

9

xK =

Columns 1 through 6

29.0000 -3.7476 - 9.3636i 5.2306 +13.6981i -7.0000 + 3.4641i -4.4829 + 2.2771i -4.4829 - 2.2771i

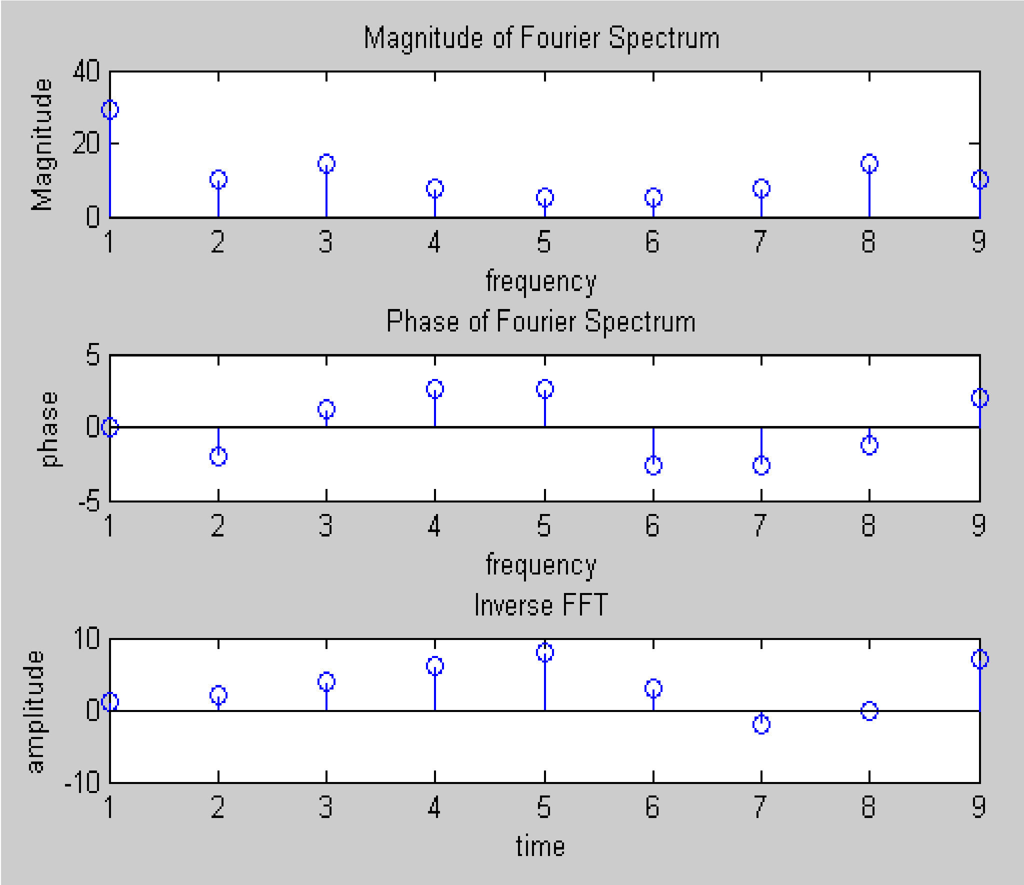
Columns 7 through 9

-7.0000 - 3.4641i 5.2306 -13.6981i -3.7476 + 9.3636i

xn =

1.0000 2.0000 4.0000 6.0000 8.0000 3.0000 -2.0000 0 7.0000

Plotting Magnitude and Phase spectrum



EXP.NO: 4
DETERMINATION OF POWER SPECTRUM OF A GIVEN SIGNAL

AIM: Determination of Power Spectrum of a given signals

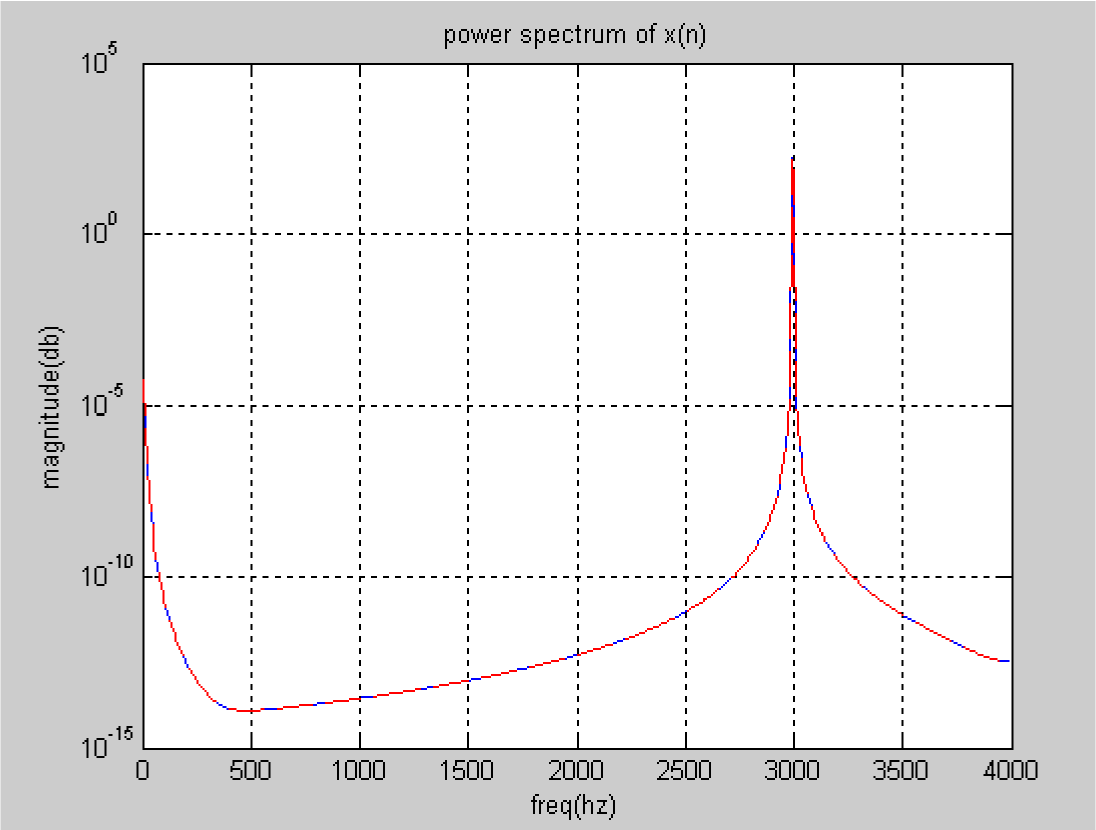
Software: MATLAB

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)');
```

OUTPUT:

enter the frequency[1 to 5000]:3000


power spectrum of x(n)
magnitude(db)
freq(hz)
10^5
10^0
10^-5
10^-10
10^-15
0
500
1000
1500
2000
2500
3000
3500
4000

EXP. NO: 5
IMPLEMENTATION OF LP FIR FILTER FOR A GIVEN SEQUENCE

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

Software: MATLAB

THEORY:

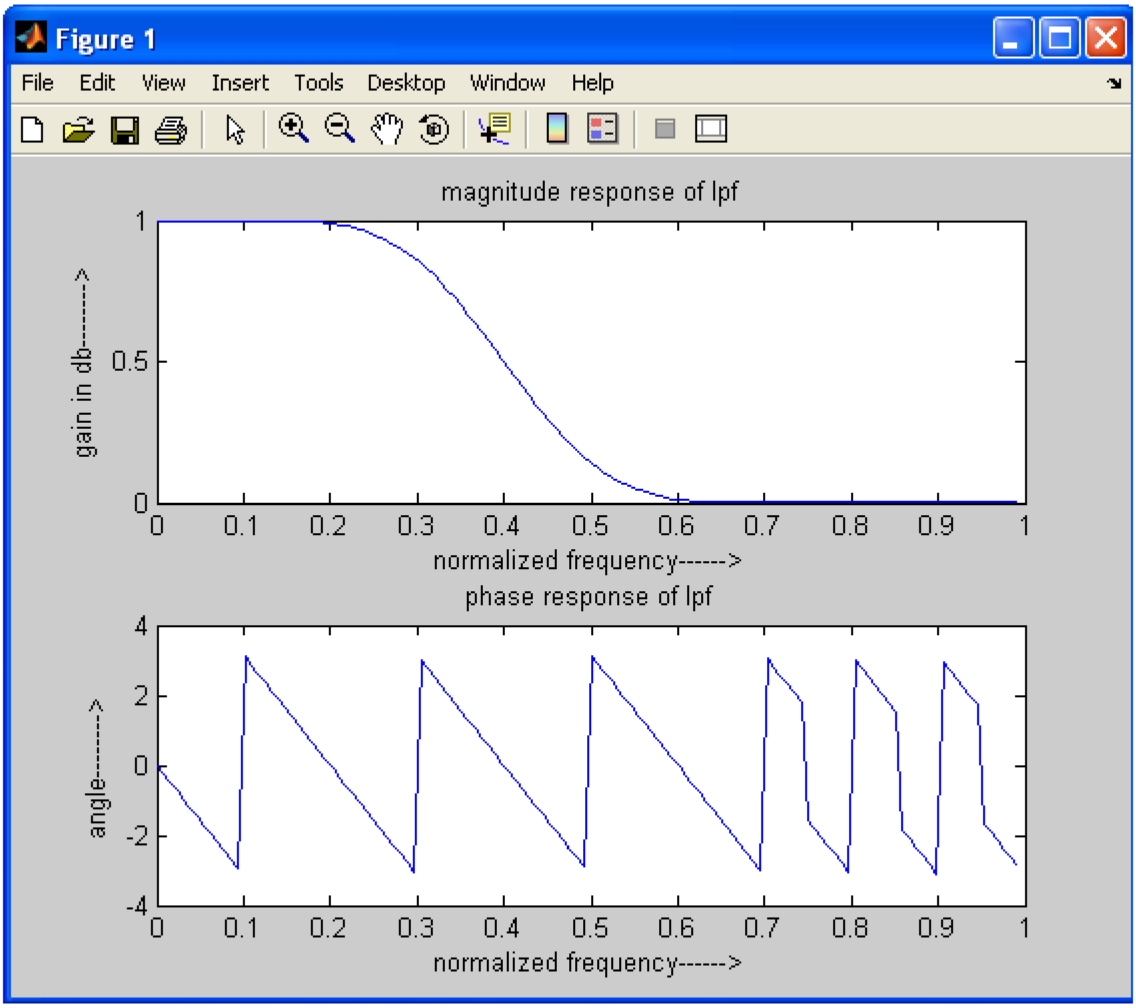
Window Name	Transition Width $\Delta\omega$ Approximate	Exact values	Min. Stop band Attenuation	Matlab Command
Rectangular	$\frac{4\Pi}{M}$	$\frac{1.8\Pi}{M}$	**21db**	**B = FIR1(N,Wn,boxcar)**
Bartlett	$\frac{8\Pi}{M}$	$\frac{6.1\Pi}{M}$	**25db**	**B = FIR1(N,Wn,bartlett)**
Hanning	$\frac{8\Pi}{M}$	$\frac{6.2\Pi}{M}$	**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)**

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 ----->');
```

Result:



window =

-0.0000 0.0092 0.0402 0.1014 0.2008 0.3400 0.5098 0.6892 0.8492

0.9602 1.0000 0.9602 0.8492 0.6892 0.5098 0.3400 0.2008 0.1014

0.0402 0.0092 -0.0000

b = 0.0000 -0.0003 -0.0009 0.0027 0.0101

-0.0000 -0.0386 -0.0430 0.0794 0.2906

0.3999 0.2906 0.0794 -0.0430 -0.0386

-0.0000 0.0101 0.0027 -0.0009 -0.0003

0.0000

EXP. NO: 6
IMPLEMENTATION OF HP FIR FILTER FOR A GIVEN SEQUENCE

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

Software: MATLAB

THEORY:

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 $H_d(e^{jw})$ 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 $n = \pm [(N-1)/2]$

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 $w(n) = w(-n) \neq 0$ for $|n| \leq [(N-1)/2]$

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

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

$h(n) = h_d(n)\, w(n)$ for $|n| \leq [(N-1)/2]$

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

The frequency response $H(e^{jw})$ of the filter can be obtained by convolution of $H_d(e^{jw})$ and $W(e^{jw})$ is given by,

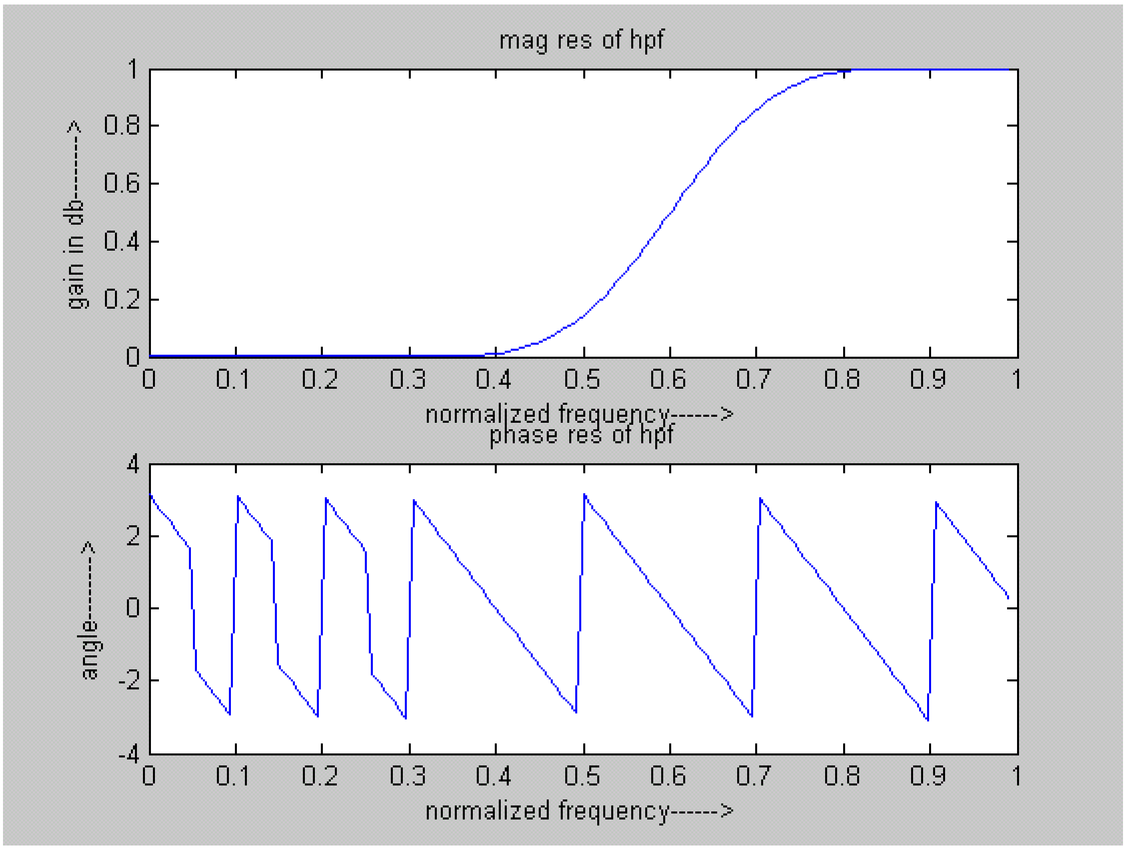
$$H(e^{jw}) = (1/2\pi) \int_{-\pi}^{\pi} H_d(e^{j\theta})\, W(e^{j(w-\theta)})\, d\theta$$

$$H(e^{jw}) = H_d(e^{jw}) * W(e^{jw}).$$

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 ----->');
```

RESULT:



fn= 0.6

window = -0.0000 0.0092 0.0402 0.1014 0.2008 0.3400
0.5098 0.6892 0.8492 0.9602 1.0000 0.9602 0.8492
0.6892 0.5098 0.3400 0.2008 0.1014 0.0402 0.0092
-0.0000

b =

0.0000 0.0003 -0.0009 -0.0027 0.0101
0.0000 -0.0386 0.0430 0.0794 -0.2906
0.3999 -0.2906 0.0794 0.0430 -0.0386
0.0000 0.0101 -0.0027 -0.0009 0.0003
0.0000

EXP. NO: 7
IMPLEMENTATION OF LP IIR FILTER FOR A GIVEN SEQUENCE

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

Software: MATLAB

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-->');
```

OUTPUT:

enter the IIR filter design specifications enter the passband ripple:15

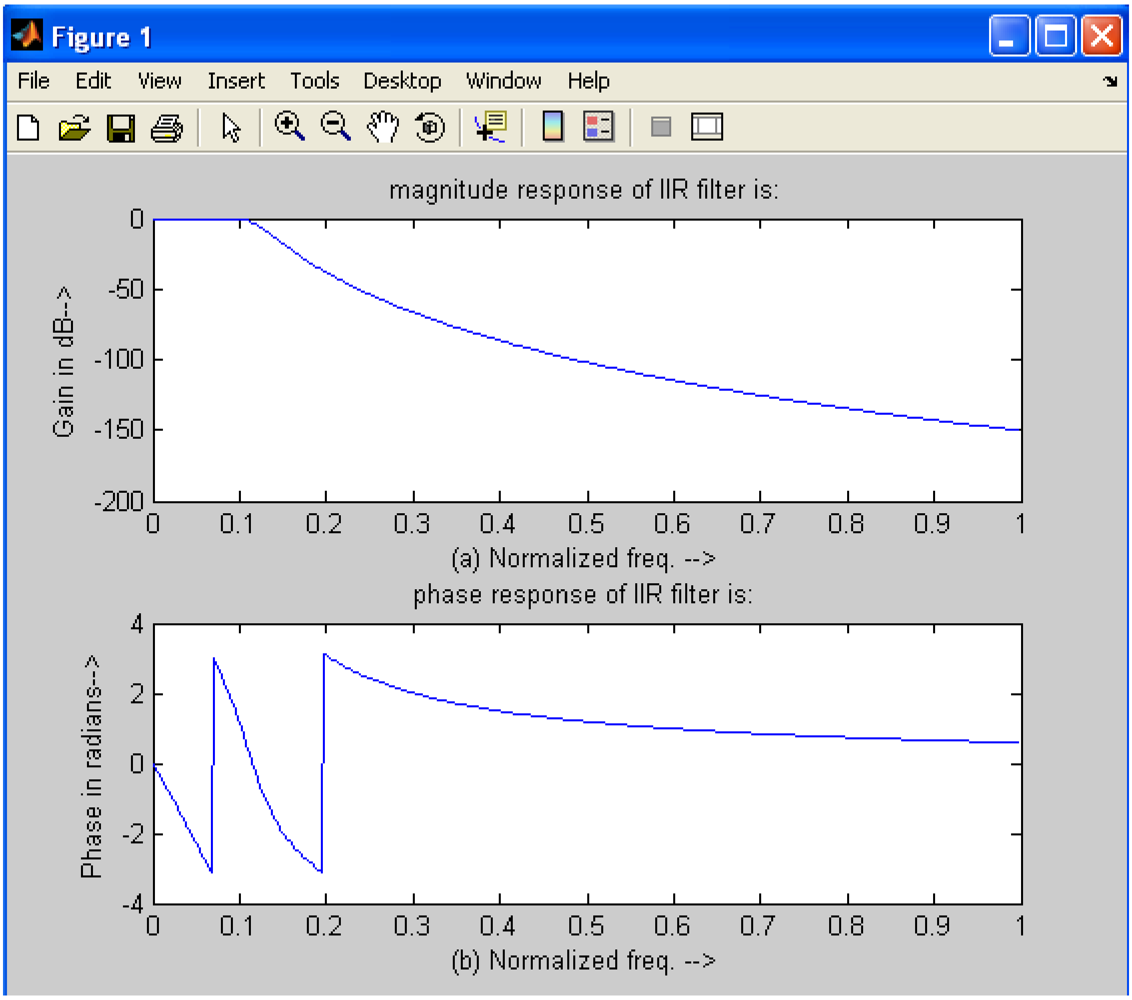
enter the stopband ripple:60

enter the passband freq:1500

enter the stopband freq:3000

enter the sampling freq:7000

Frequency response of IIR LPF is:



EXP. NO: 8
IMPLEMENTATION OF HP IIR FILTER FOR A GIVEN SEQUENCE

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

Software: MATLAB

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-->');
```

OUTPUT:

enter the IIR filter design specifications enter the passband ripple15

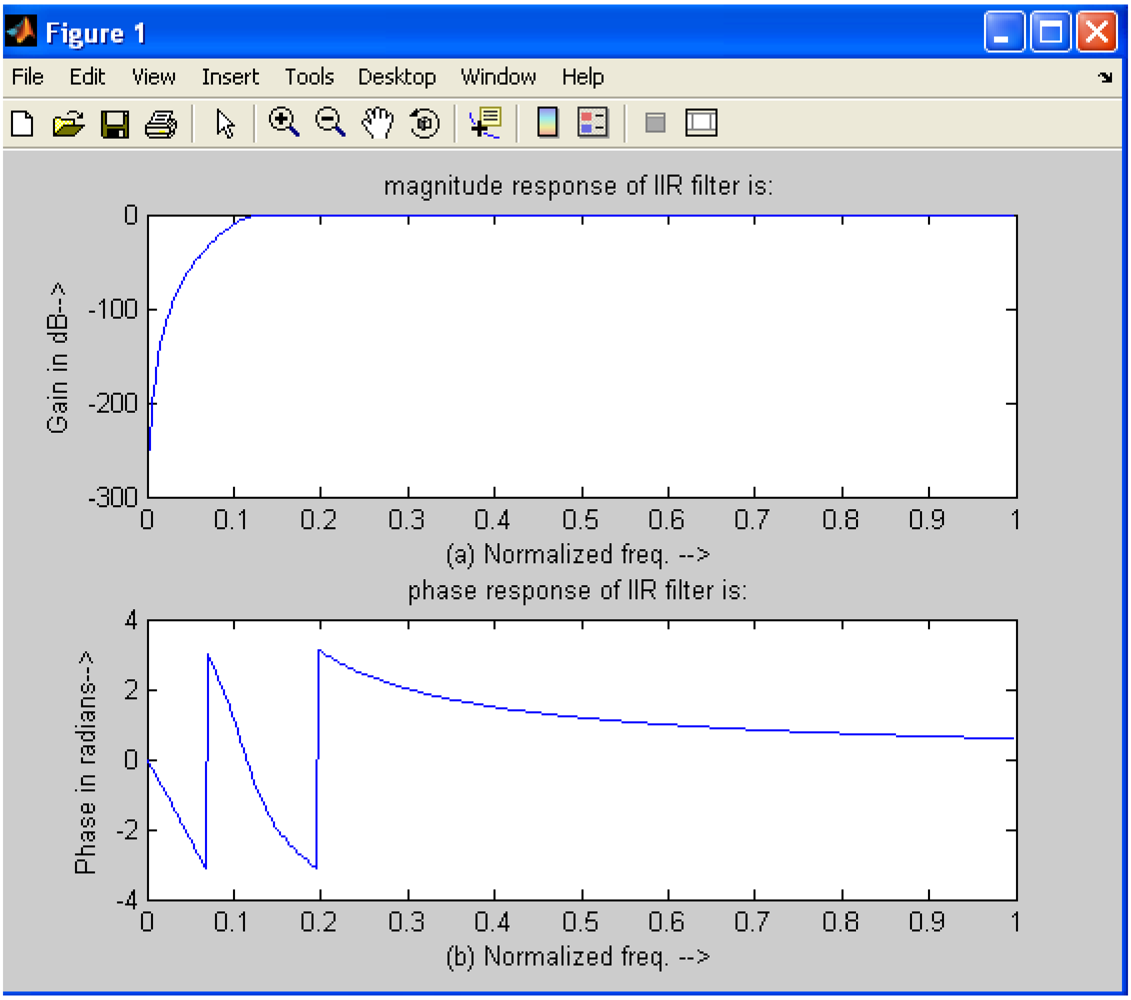
enter the stopband ripple60

enter the passband freq1500

enter the stopband freq3000

enter the sampling freq7000

Frequency response of IIR HPF is:



EXP. NO: 9

GENERATION OF SINUSOIDAL SIGNAL THROUGH FILTERING

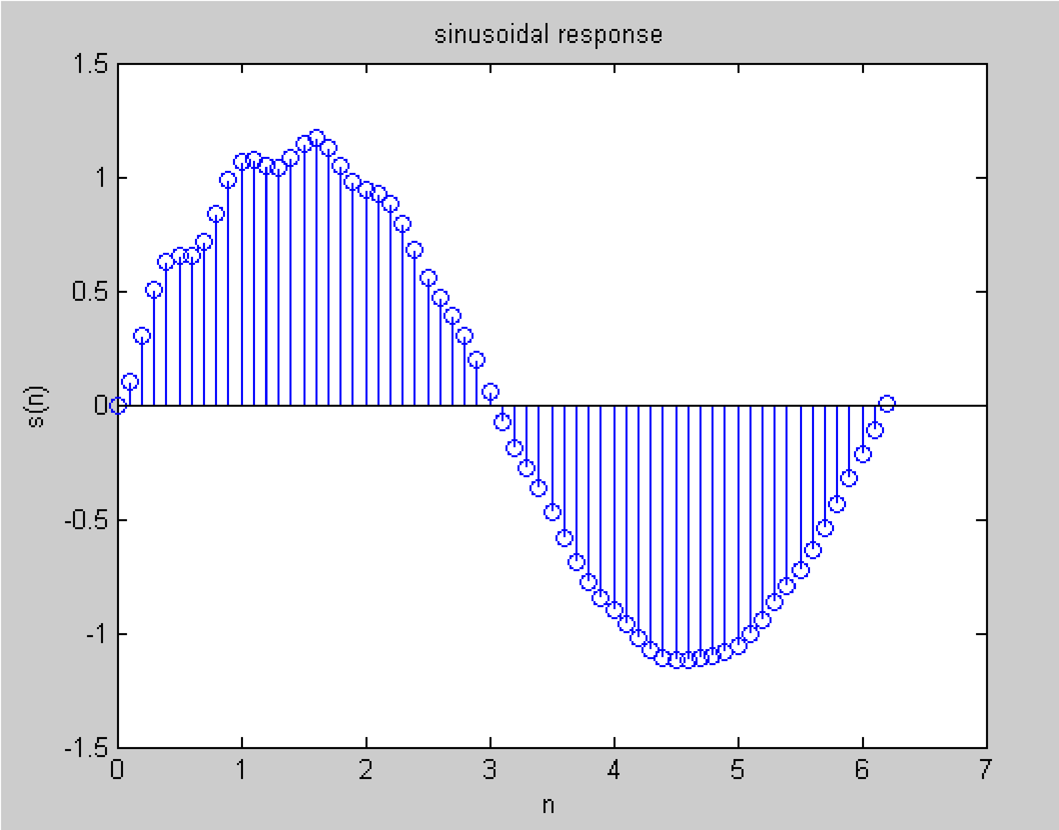
AIM: To generate a sinusoidal signal through filtering.

Software: MATLAB

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)');
```

OUTPUT:


sinusoidal response
s(n)
n
1.5
1
0.5
0
-0.5
-1
-1.5
0
1
2
3
4
5
6
7

EXP. NO: 10
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.

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

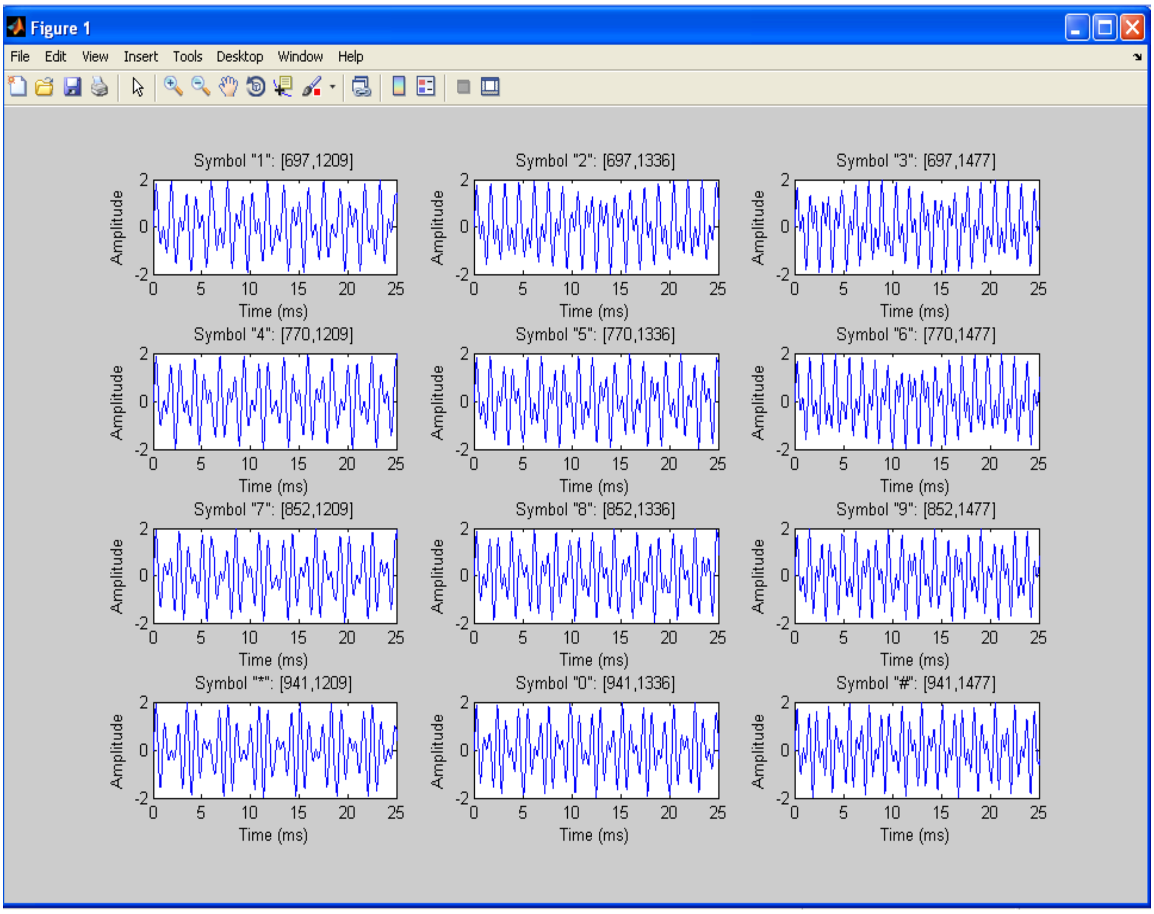
Program:

```
clc
close all
clear all
 %generate the twelve frequency pairs
symbol = {'1','2','3','4','5','6','7','8','9','*','0','#'};
lfg = [697 770 852 941]; % Low frequency group
hfg = [1209 1336 1477]; % High frequency group
f = [];
```

```
for c=1:4,
    for r=1:3,
        f = [ f [lfg(c);hfg(r)] ];
    end
end
f';
 %generate and visualize the DTMF tones
 Fs  = 8000;       % Sampling frequency 8 kHz
N = 800;          % Tones of 100 ms
t = (0:N-1)/Fs; % 800 samples at Fs
tones = zeros(N,size(f,2));
for toneChoice=1:12,
    % Generate tone
    tones(:,toneChoice) = sum(sin(f(:,toneChoice)*2*pi*t))';
    % Plot tone
    subplot(4,3,toneChoice),plot(t*1e3,tones(:,toneChoice));
    title(['Symbol "', symbol{toneChoice},'":
[',num2str(f(1,toneChoice)),',',num2str(f(2,toneChoice)),']'])
    set(gca, 'Xlim', [0 25]);
    ylabel('Amplitude');
     xlabel('Time (ms)');
end
```

Generation of DTMF Signals:

Output



EXP. NO: 11
IMPLEMENTATION OF DECIMATION PROCESS

AIM: program to verify the decimation of given sequence.

SOFTWARE: MATLAB 7.0 and above

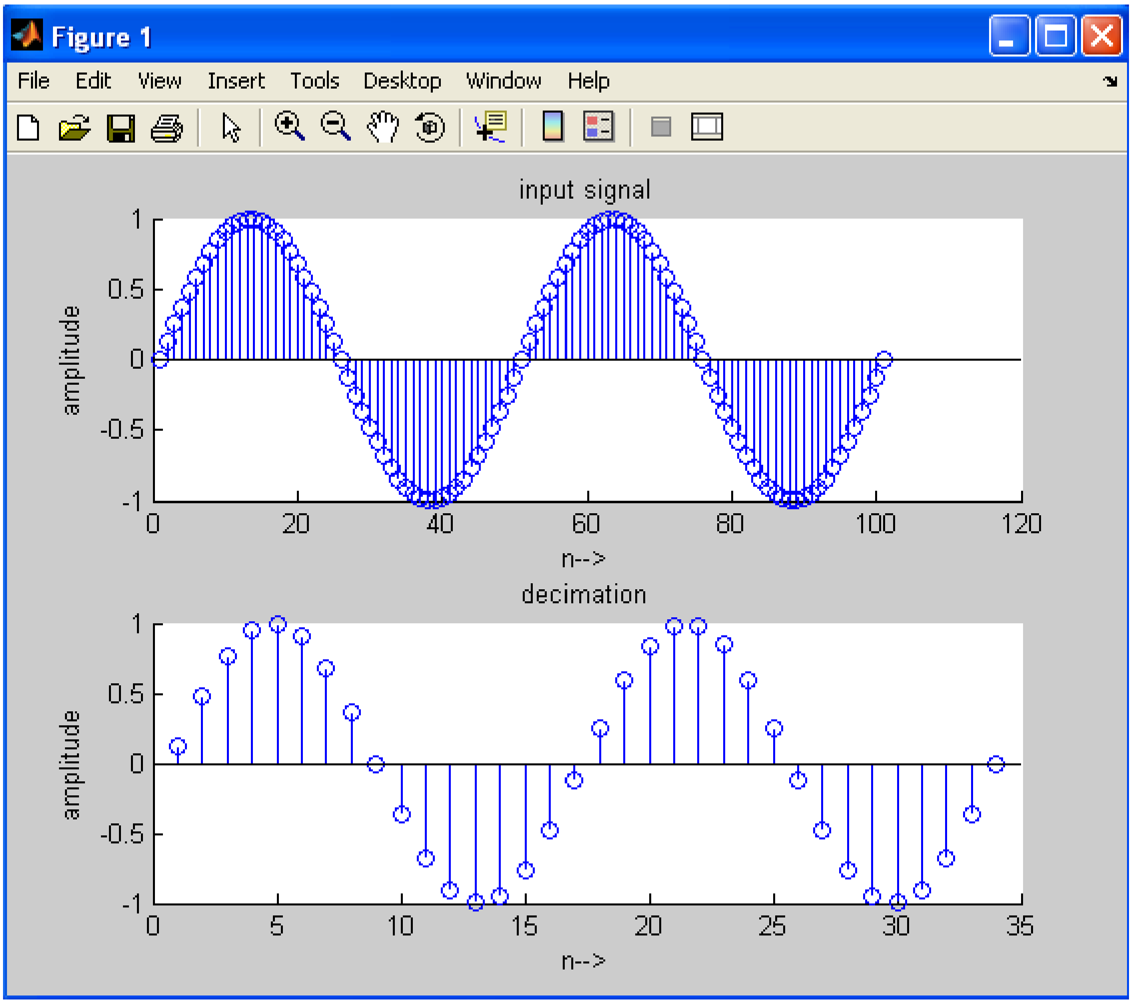
Signal Processing Toolbox

PROGRAM:

```
clc;
clear all;
close all;
%~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~
%input signal
%~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~
fm=20;
fs=1000;
t=0.1;
dt=1/fs;
n=0:dt:t;
m=sin(2*pi*fm*n);
subplot(2,1,1);
stem(m);
xlabel('n-->');
ylabel('amplitude');
title('input signal');
%~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~
%decimation
%~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~
r=input(' enter the length of decimation factor r= ');
d=decimate(m,r);
subplot (2,1,2);
stem(d);
xlabel('n-->');
ylabel('amplitude');
title('decimation');
```

OUTPUT:

enter the length of decimation factor b= 3



EXP. NO: 12
IMPLEMENTATION OF INTERPOLATION PROCESS

AIM: program to verify the decimation of given sequence.

SOFTWARE: MATLAB 7.0 and above

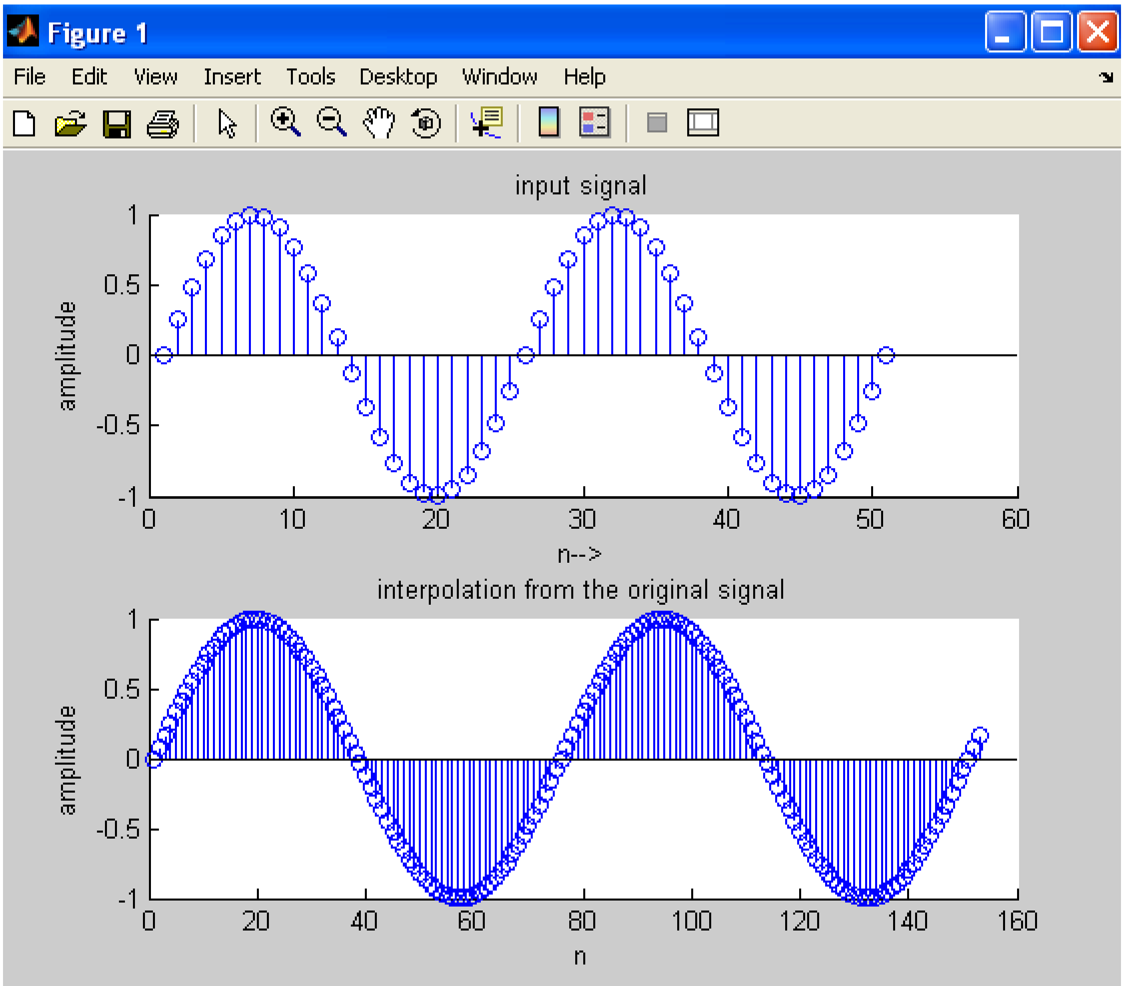
Signal Processing Toolbox

PROGRAM:

```
clc;
clear all;
close all;
%~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~
%input signal
%~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~
fm=20;
fs=500;
t=0.1;
dt=1/fs;
n=0:dt:t;
m=sin(2*pi*fm*n);
subplot(2,1,1);
stem(m);
xlabel('n-->');
ylabel('amplitude');
title('input signal');
%from the origina signal interpolation
b=input('enter the length of interpolation factor b= ');
i=interp(m,b);
subplot(2,1,2);
stem(i);
xlabel('n');
ylabel('amplitude');
title('interpolation from the original signal');
```

OUTPUT:

enter the length of interpolation factor b= 3



EXP. NO: 13
IMPLEMENTATION OF I/D SAMPLING RATE CONVERTERS

AIM: program to implement sampling rate conversion.

SOFTWARE: MATLAB 7.0 and above

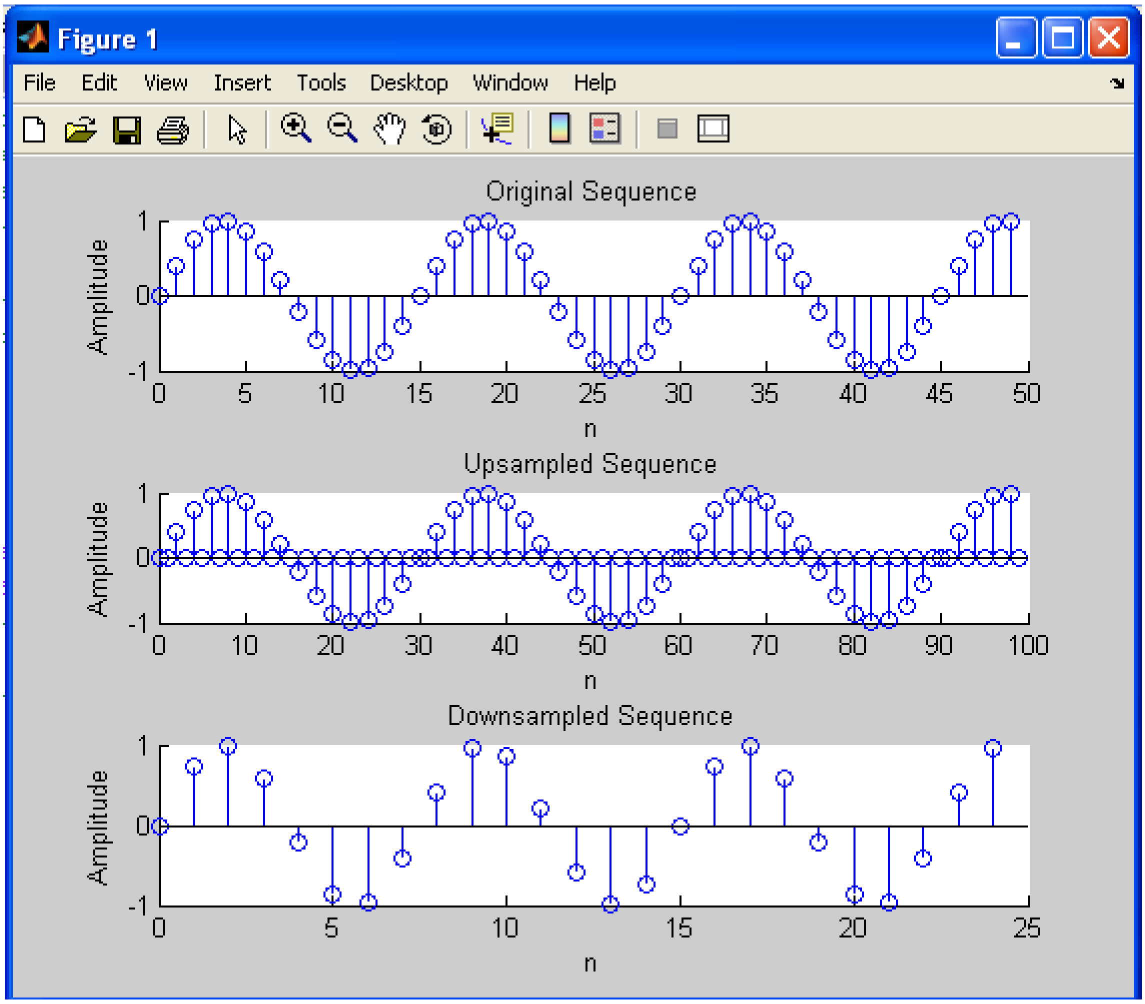
Signal Processing Toolbox

PROGRAM:

```
clc;% clear comand window
clear all; % clear work space
close all; %close all figure windows
%input signal
N=50 ;% no of samples
n=0:1:N-1;
x=sin(2*pi*n/15);% input signal
subplot(3,1,1)
stem(n,x);
xlabel('n');
ylabel('Amplitude');
title('Original Sequence');
%up sampling
L=2;% upsampling factor
x1=[zeros(1,L*N)];
n1=1:1:L*N;
j =1:L:L*N;
x1(j)=x;
subplot(3,1,2)
stem(n1-1,x1);
xlabel('n');
ylabel('Amplitude');
title('Upsampled Sequence');
% down sampling
M=2;
x2=x(1:M:N);
n2=1:1:N/M;
subplot(3,1,3)
```

```
stem(n2-1,x2);
xlabel('n');
ylabel('Amplitude');
title('Downsampled Sequence');
```

Output waveforms for Sampling Rate Conversion:



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

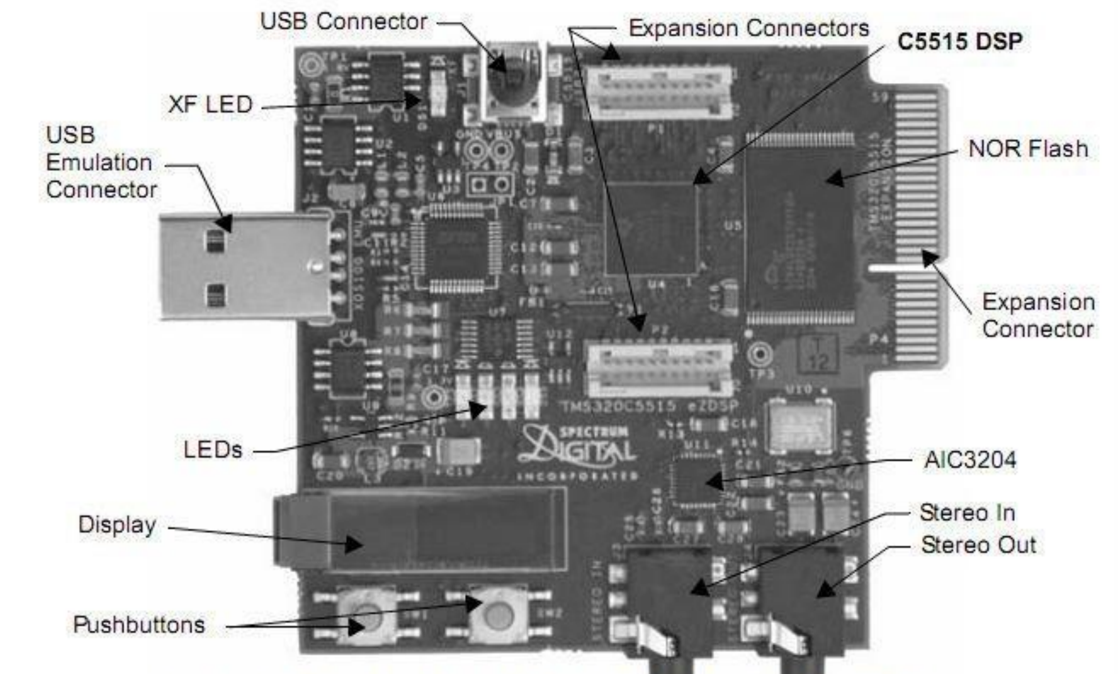


Figure 1-1, Key Features of the C5515 eZdsp USB Stick (top)

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.

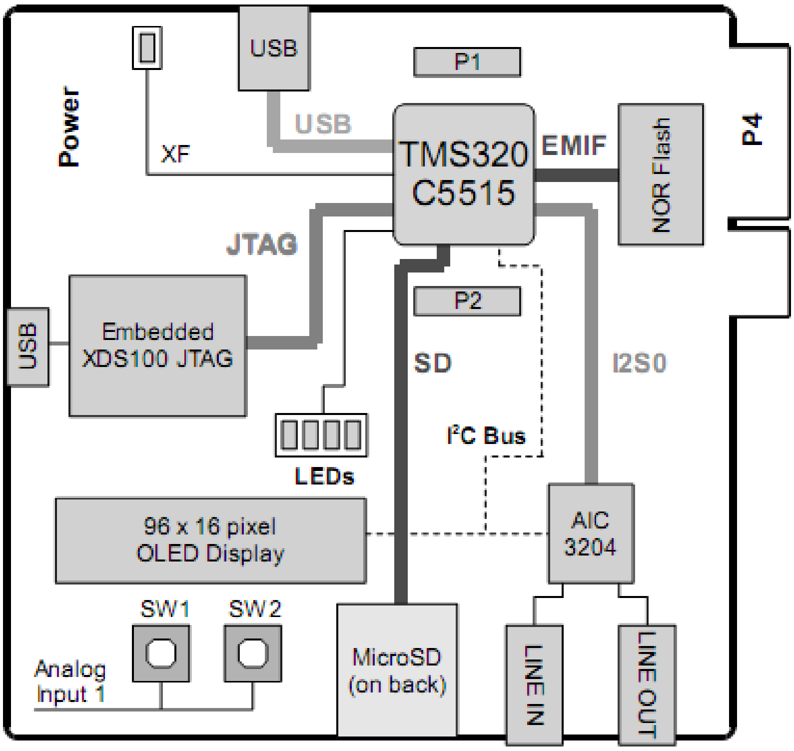


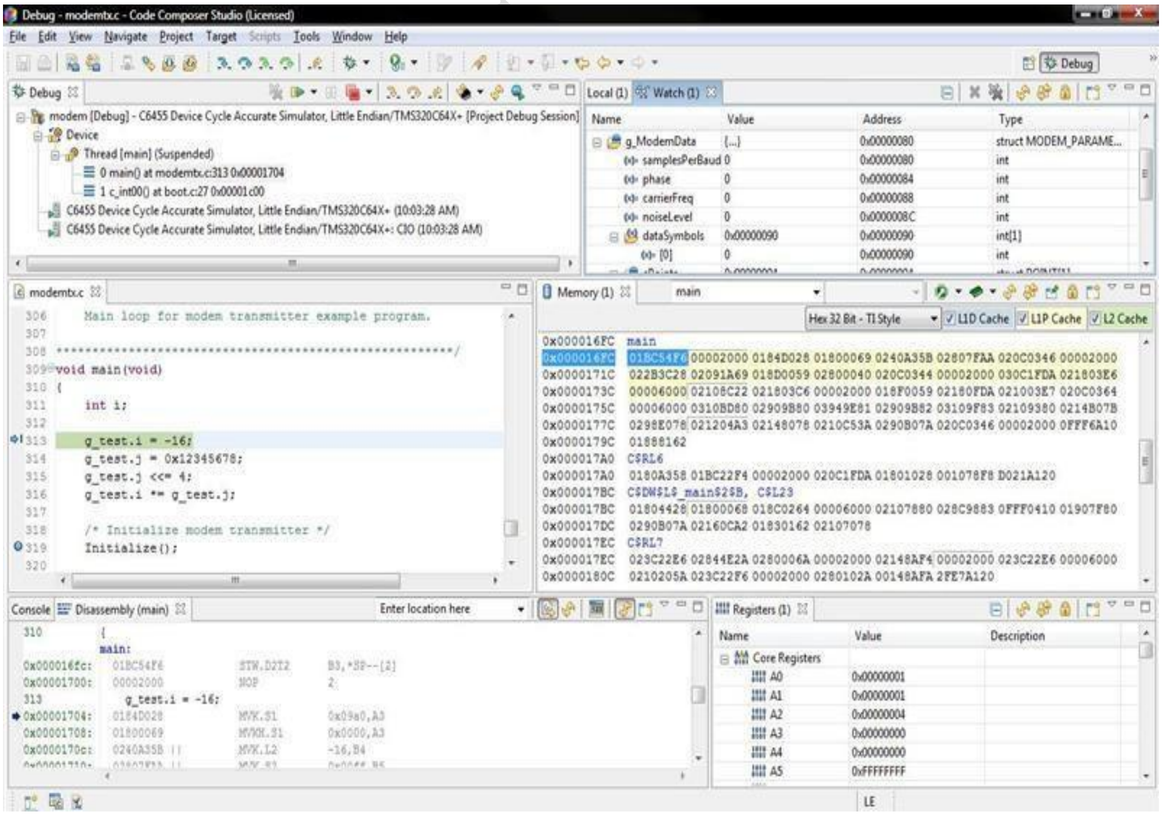
Figure 1-3, C5515 eZdsp USB Stick Block Diagram

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.



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.

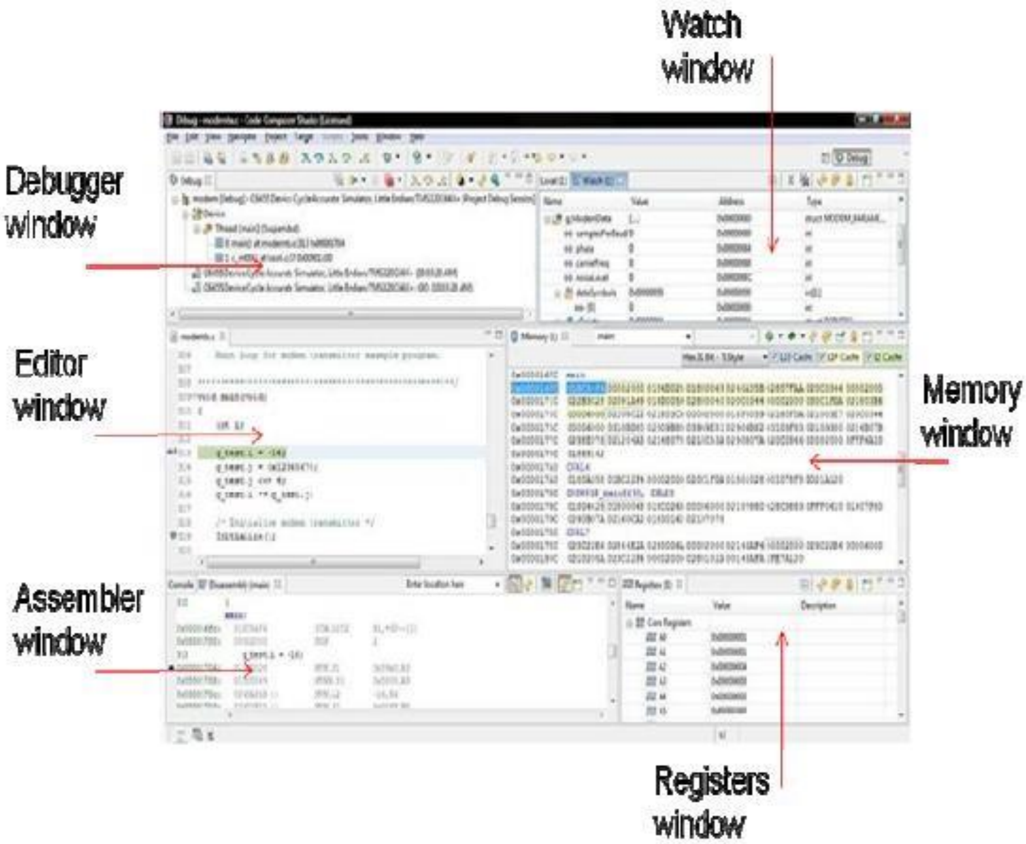
Step 1:

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: 14
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.

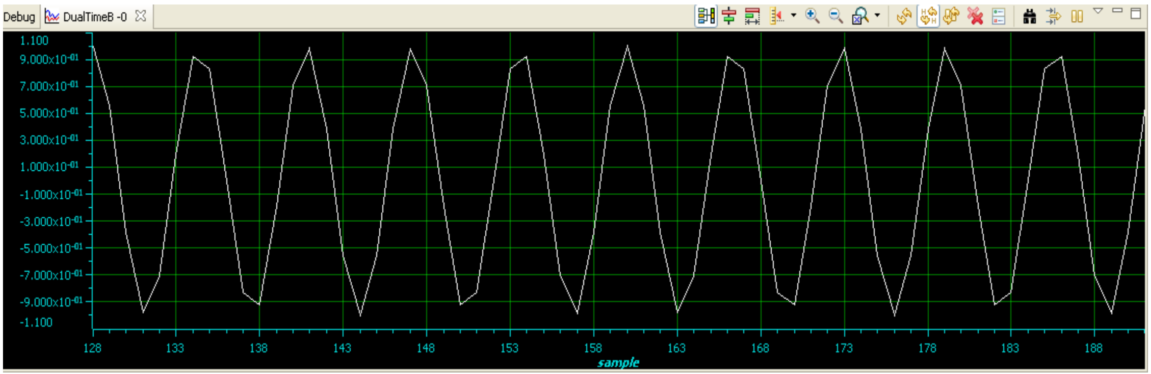
• Step 3 - Take sampled data and apply DFT algorithm.

Execution Procedure:

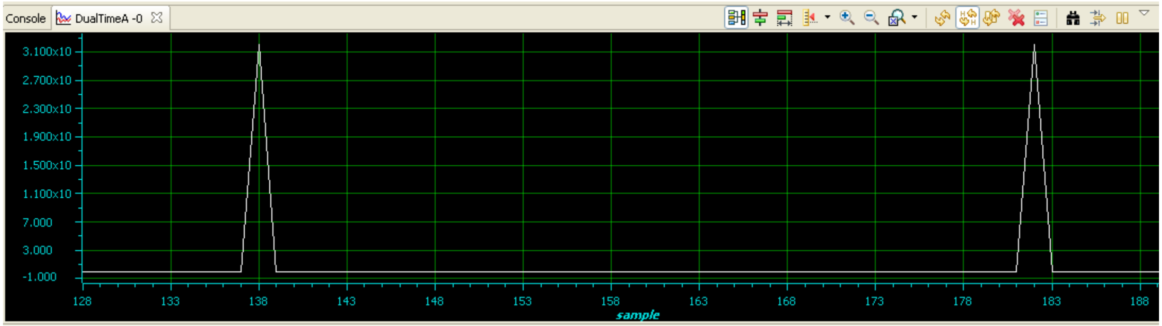
- Open CCstudio setup
- Go to File Menu , select Import option.
- In the Import Window under CCs choose Existing CCS/CCE Eclipse project then next.
- In Select root Directory Browse for project file where it is located.
- Select DFT Project folder and Finish it.
- Now Connect the DSP Kit to PC and Launch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

Result:

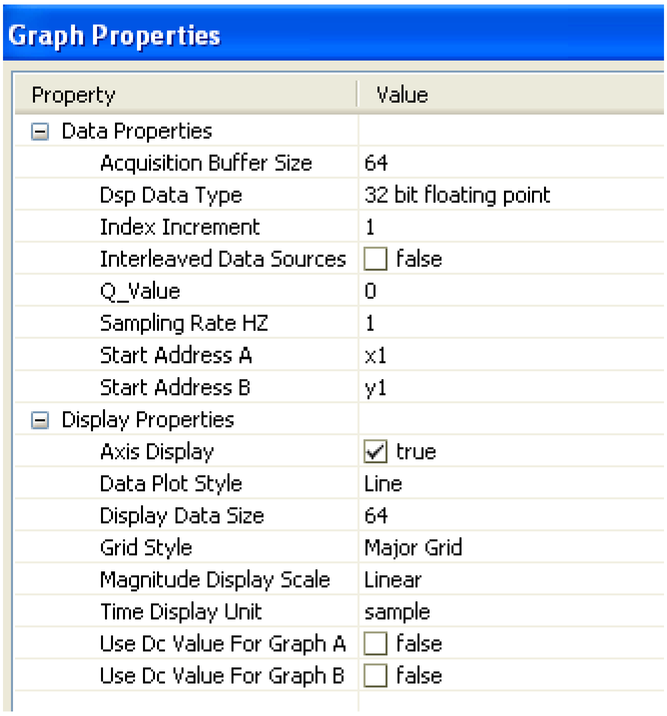
Input Signal:



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

EXP.NO: 15
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.

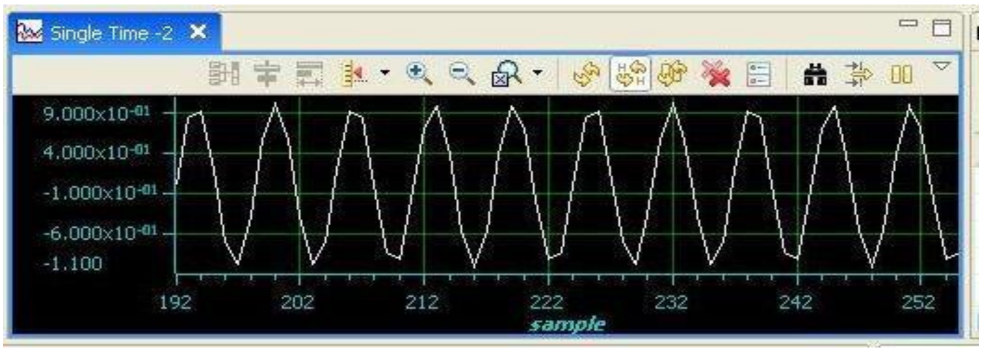
• Step 3 - Take sampled data and apply FFT algorithm.

Execution Procedure:

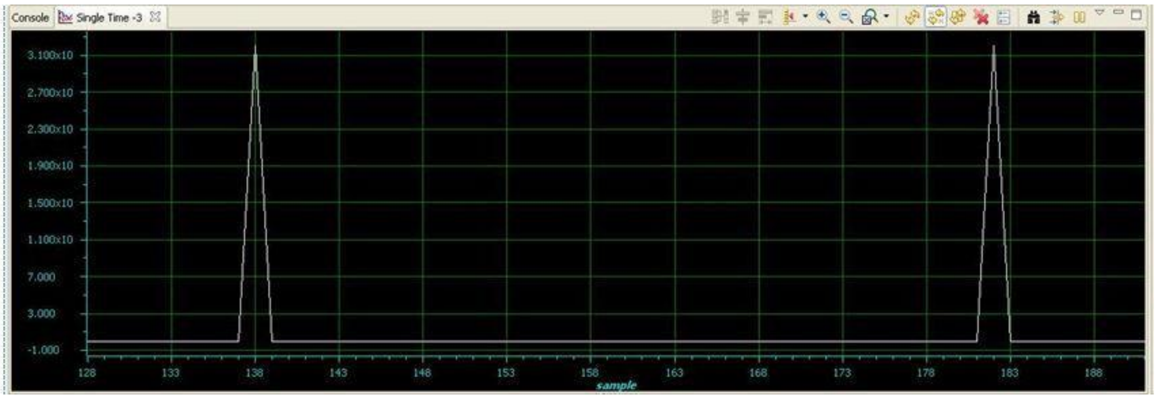
- Open CCstudio setup
- Go to File Menu , select Import option.
- In the Import Window under CCs choose Existing CCS/CCE Eclipse project then next.
- In Select root Directory Browse for project file where it is located.
- Select FFT Project folder and Finish it.
- Now Connect the DSP Kit to PC and Launch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



EXP.NO: 16
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.

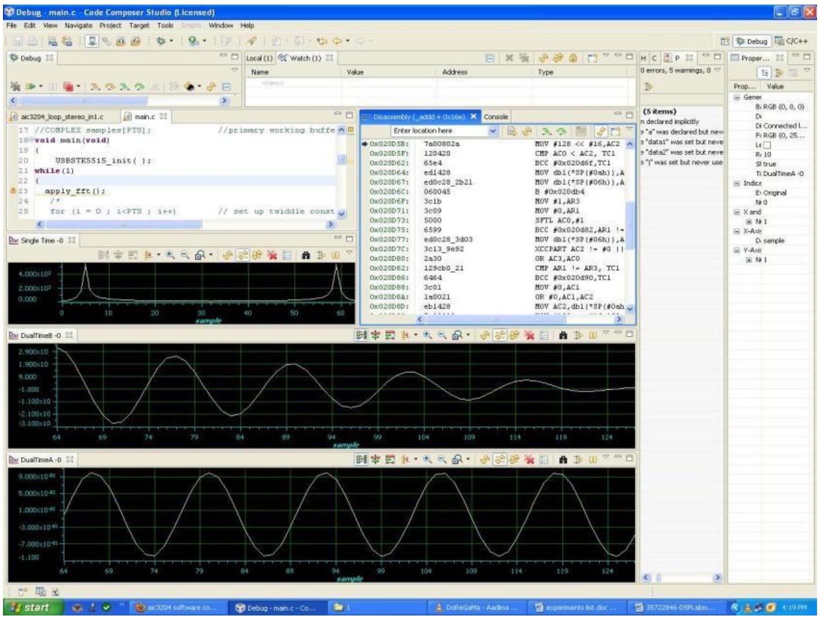
_ Now Connect the DSP Kit to PC and Launch it.(Follow the above given manual procedure from Step 46 to 51)

_ 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:



EXP.NO: 17

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

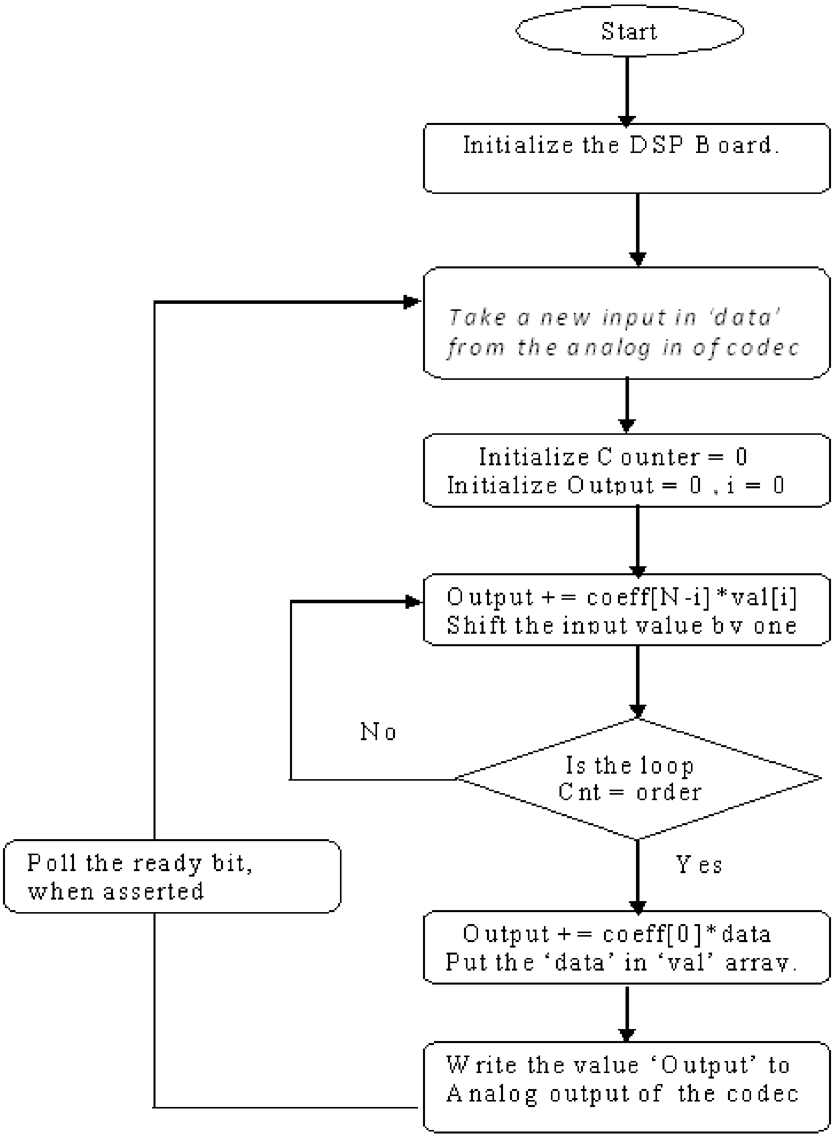
Eg: Wc = 2*pie* fc / Fs= 2*pie* 400/8000= 0.1*pie

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:



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 };

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};*/
/*//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, 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).

EXP.NO: 18

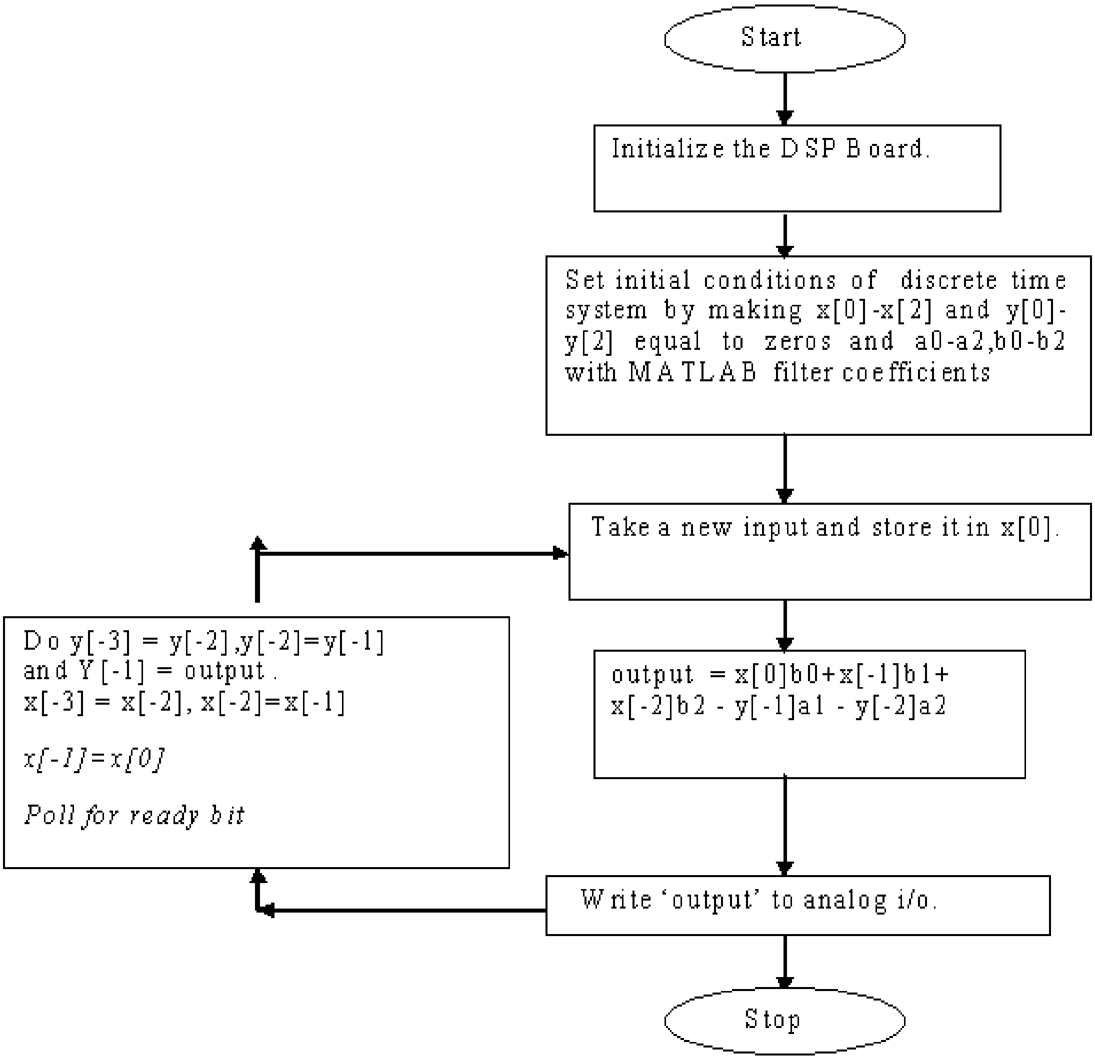
IMPLEMENTATION OF LP IIR FILTER FOR GIVEN SEQUENCE & IMPLEMENTATION OF HP IIR FILTER FOR GIVEN SEQUENCE

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:



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,
3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,3,1,2,1 };

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 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,  3,1,3,1,3,1,3,1,3,1,2,1
};
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).

EXP.NO: 19
GENERATION OF SINUSOIDAL SIGNAL THROUGH FILTERING

Program:

Main.c

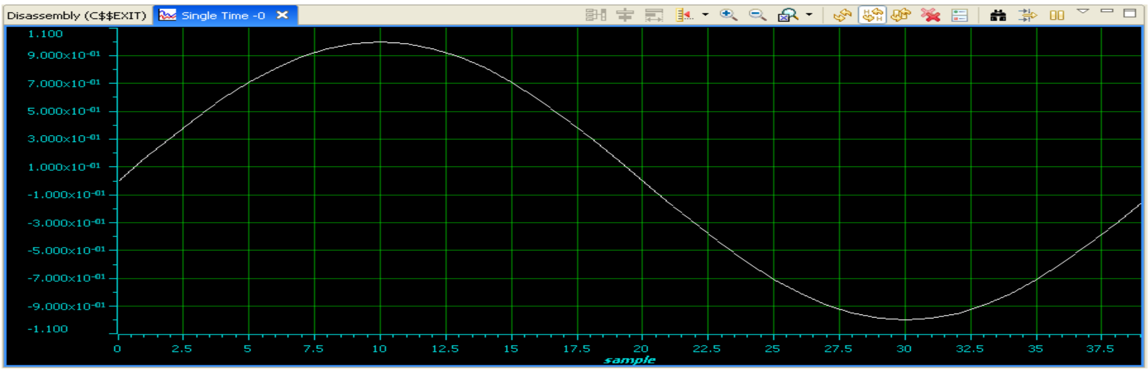
```
#include <usbstk5515.h>
#include<stdio.h>
#include <math.h>
#define PTS 40 //no of points for FFT
#define PI 3.14159265358979
float output;
float y[40];
void main()
{
int n;
for (n=0; n<PTS; n++)
{
/*y[n]=A•y[n–1]+B•y[n–2] A=1.9754 and B=–1. Examining the behavior
of this IIR filter by its transfer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

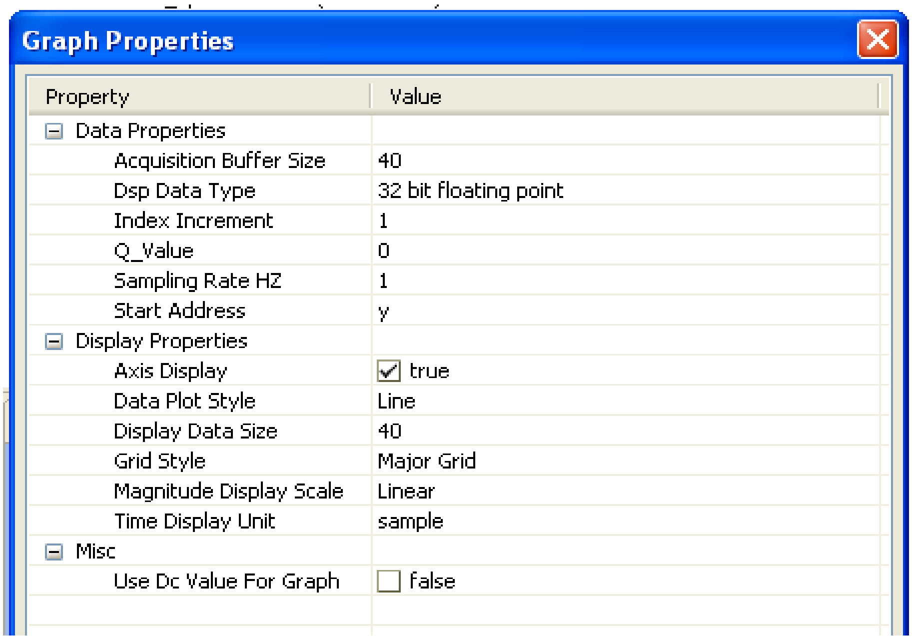
- Open CCstudio setup
- Go to File Menu , select Import option.
- In the Import Window under CCs choose Existing CCS/CCE Eclipse project then
- next.
- In Select root Directory Browse for project file where it is located.
- Select sinusoidal through Filtering Project folder and Finish it.
- Now Connect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



EXP.NO: 20
GENERATION OF DTMF SIGNALS

Program:

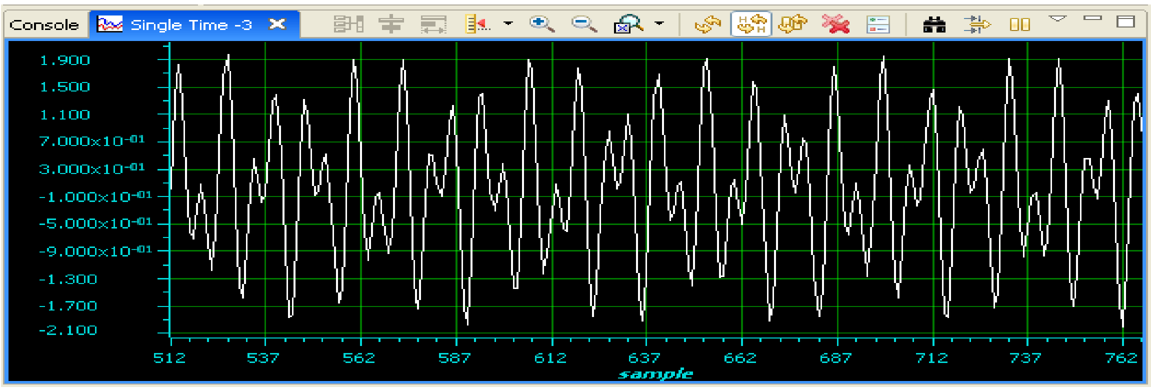
```
#include <usbstk5515.h>
#include<stdio.h>
#include <math.h>
#define PTS 256
#define FS 8000 //sampling frequency of 8khz
#define pi 3.14159265358979
int lfg[ ]={697,770,852,941}; // Low frequency group
int hfg[ ]={1209,1336,1477}; // High frequency group
float num[12][PTS];
float t[PTS];
void main()
{
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:

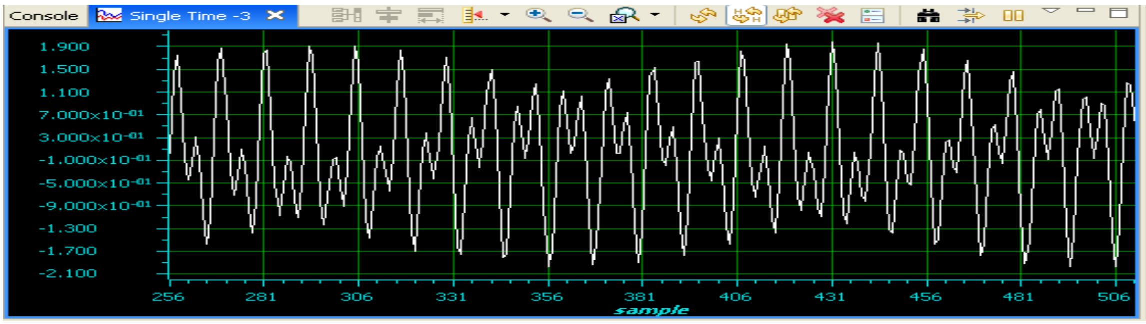
- Open CCstudio setup
- Go to File Menu, select Import option.
- In the Import Window under CCs choose Existing CCS/CCE Eclipse project then next.
- In Select root Directory Browse for project file where it is located.
- Select DTMF Project folder and Finish it.
- Now Connect the DSP Kit to PC and Launch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



EXP.NO: 21
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
```

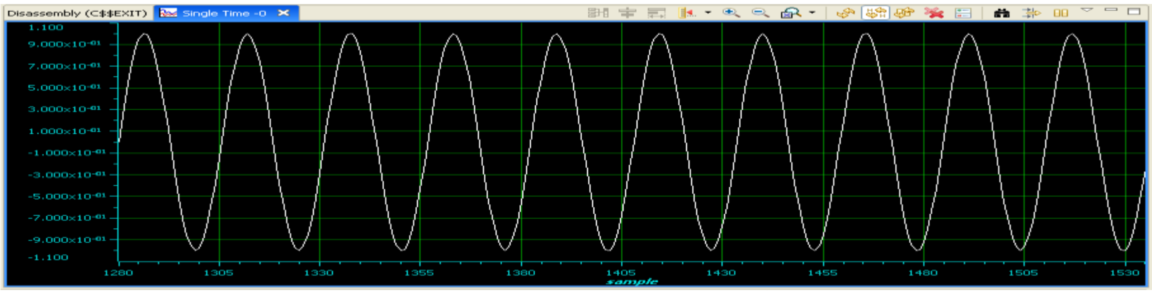
Execution Procedure:

- 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.

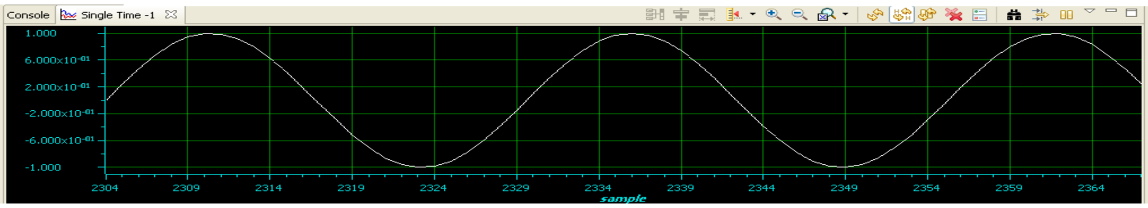
- Now Connect the DSP Kit to PC and Launch it. (Follow the above given manual procedure from Step 46 to 51).
- Give Right Click on Your decimation.out file under Binaries and select Load program Option.
- Now Go to Target select Run.
- In Tools menu select Graph (Dual Time) set the properties and watch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

Result:

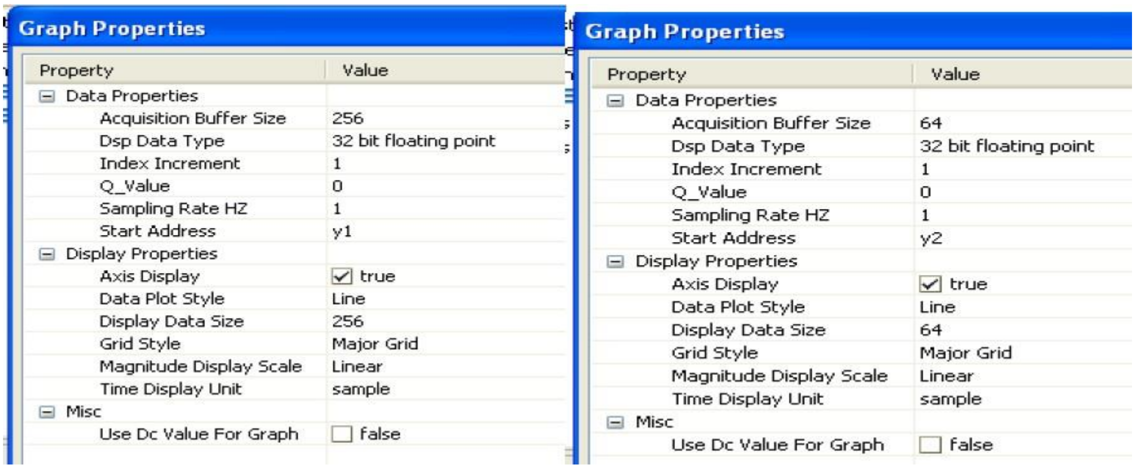
Input signal:



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

EXP.NO: 22

IMPLEMENTATION OF INTERPOLATION PROCESS

Program:

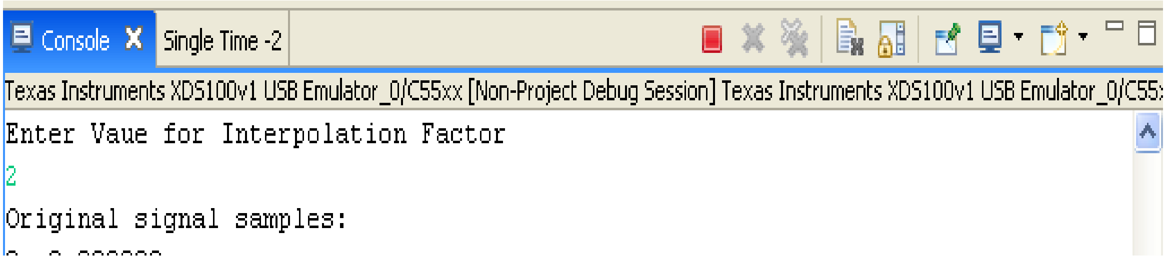
```
#include <usbstk5515.h>
#include<stdio.h>
#include <math.h>
#define PTS 256 //no of points for FFT
#define PI 3.14159265358979
float y1[PTS],y2[PTS],y3[PTS];
main()
{
int n,m;
printf("Enter Vaue for Interpolation Factor\n");
scanf("%d",&m);
printf("Original signal samples:\n");
for(n=0;n<=PTS;n++)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

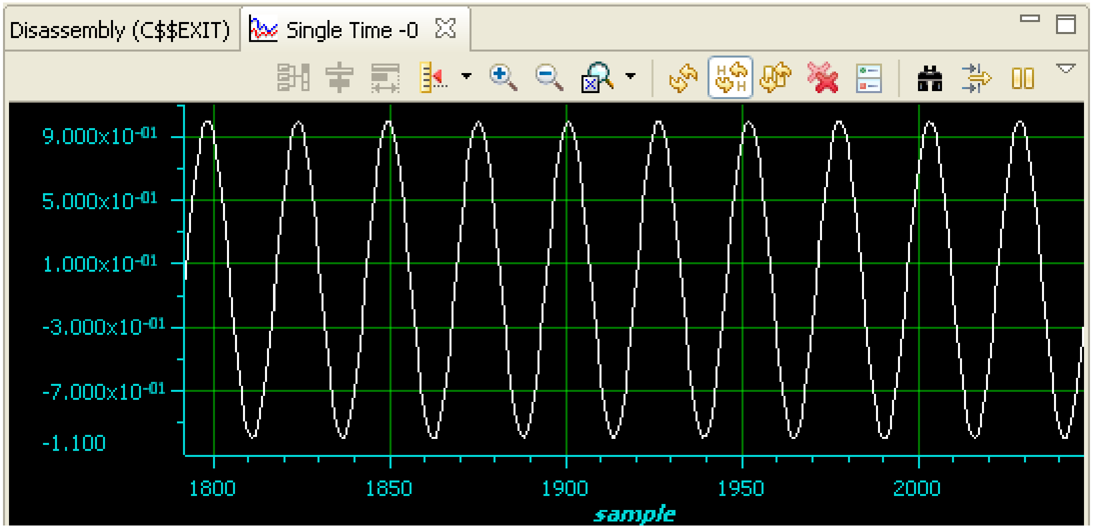
- Open CCstudio setup
- Go to File Menu, select Import option.
- In the Import Window under CCs choose Existing CCS/CCE Eclipse project then next.
- In Select root Directory Browse for project file where it is located.

- 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.

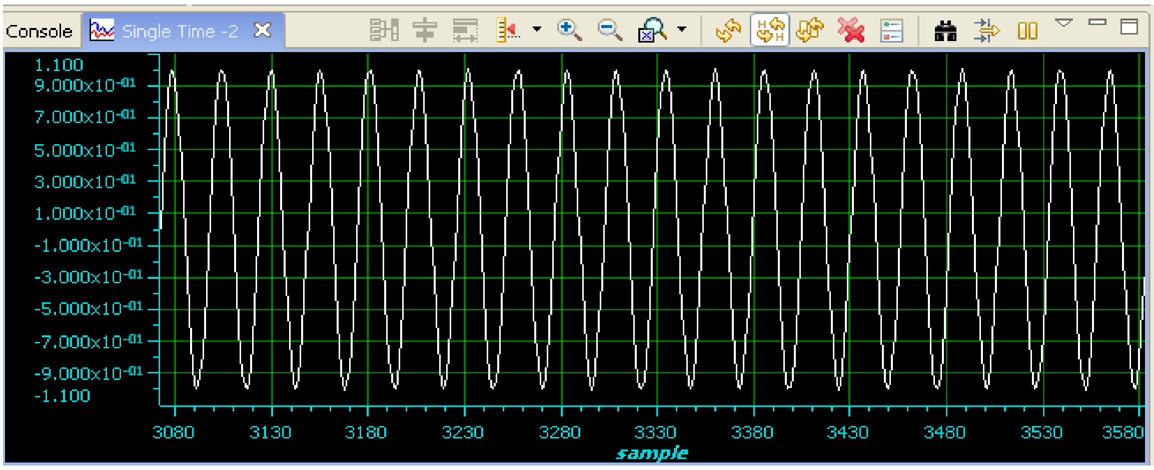
Result :



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

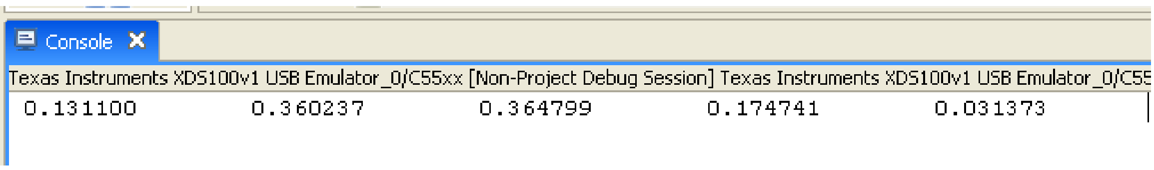
EXP.NO: 23
IMPULSE RESPONSE OF FIRST ORDER AND SECOND ORDER SYSTEMS

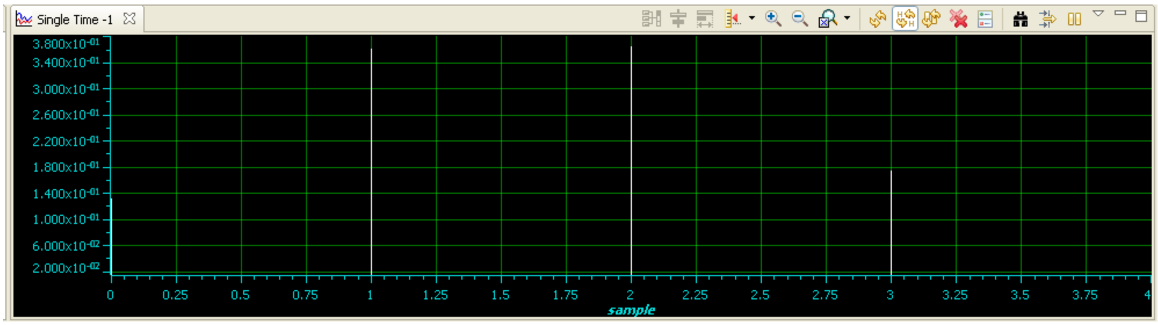
Program:

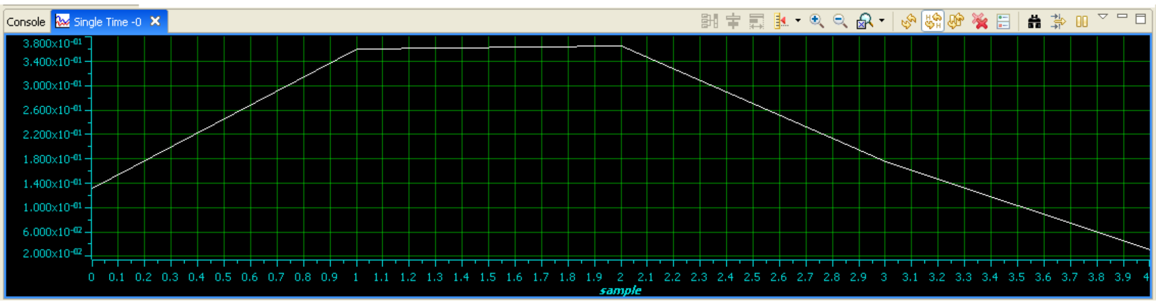
```
#include<stdio.h>
#define Order 2
#define Len 5
float h[Len] = {0.0,0.0,0.0,0.0,0.0},sum;
void main()
{
int j, k;
float a[Order+1] = {0.1311, 0.2622, 0.1311};
float b[Order+1] = {1, -0.7478, 0.2722};
for(j=0; j<Len; j++)
{
sum = 0.0;
for(k=1; k<=Order; k++)
{
if ((j-k) >= 0)
sum = sum+(b[k]*h[j-k]);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

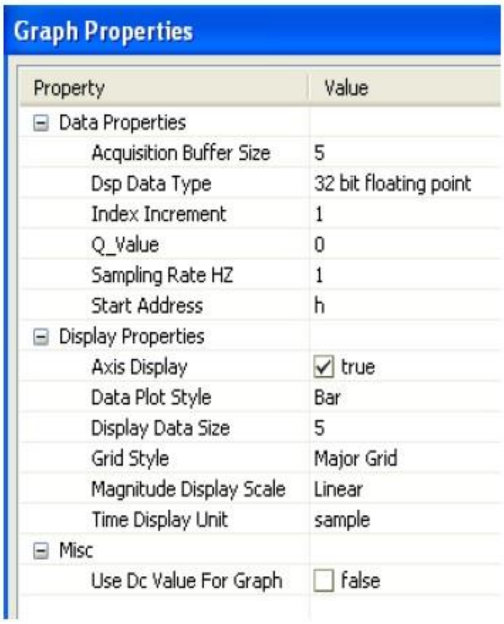
Output:

0.131100 0.360237 0.364799 0.174741 0.031373

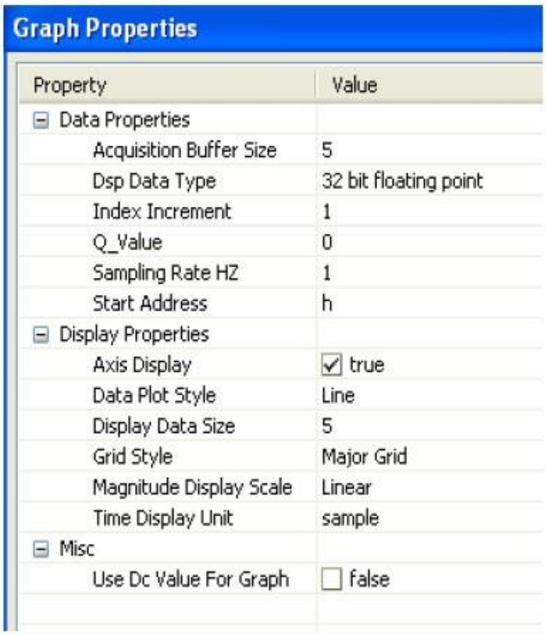







Graph Properties

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.

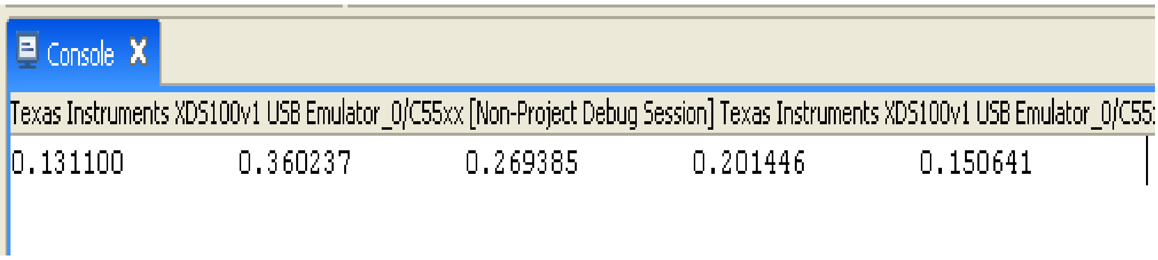
Program:

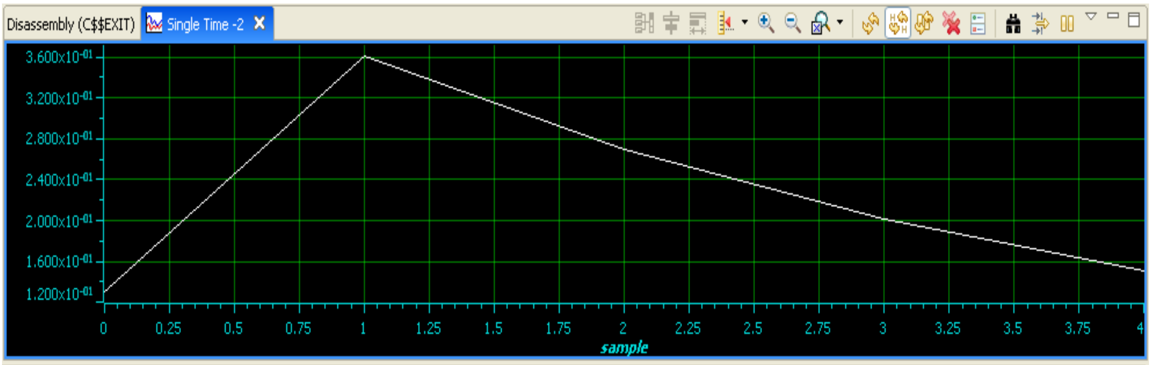
```
#include<stdio.h>
#define Order 1
#define Len 5
float h[Len] = {0.0,0.0,0.0,0.0,0.0},sum;
void main()
{
int j, k;
float a[Order+1] = {0.1311, 0.2622};
```

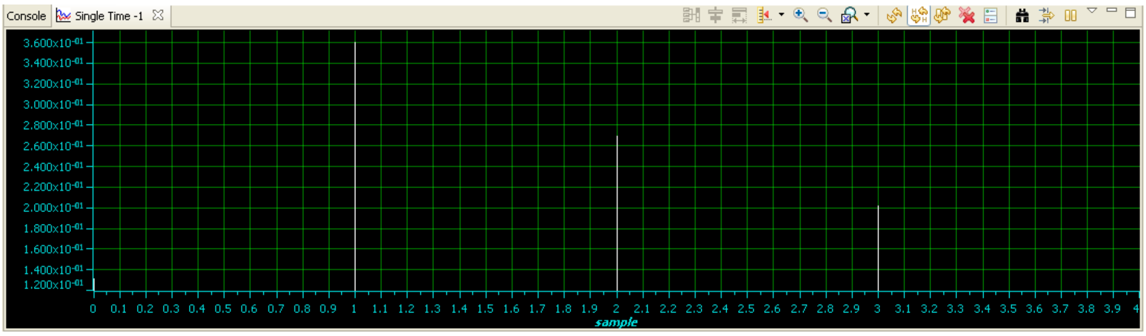
```
float b[Order+1] = {1, -0.7478};
for(j=0; j<Len; j++)
{
sum = 0.0;
for(k=1; k<=Order; k++)
{
if((j-k)>=0)
sum = sum+(b[k]*h[j-k]);
}
if(j<=Order)
h[j] = a[j]-sum;
else
h[j] = -sum;
printf("%f ", j, h[j]);
}
}
```

Output:

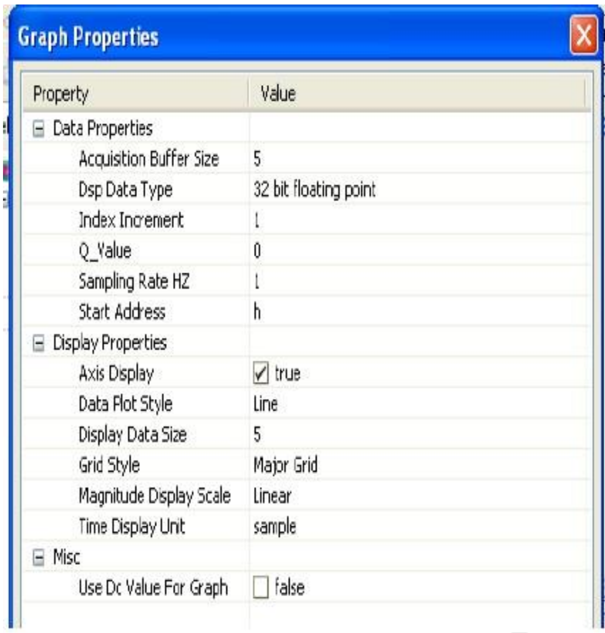
0.131100 0.360237 0.269385 0.201446 0.150641

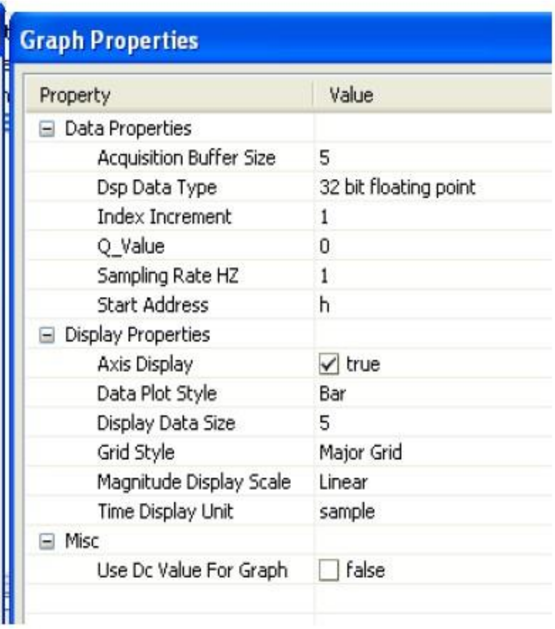






Graph Properties:





EXP. NO 24
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

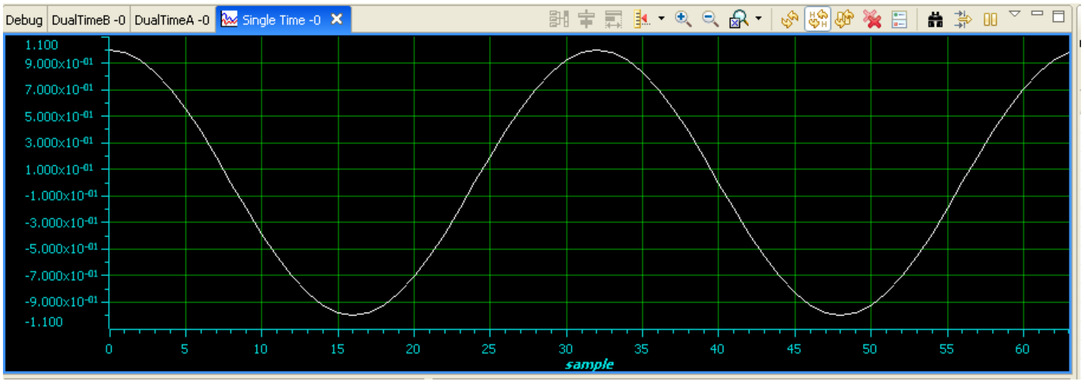
Refer previous FFT experiment

Execution Procedure:

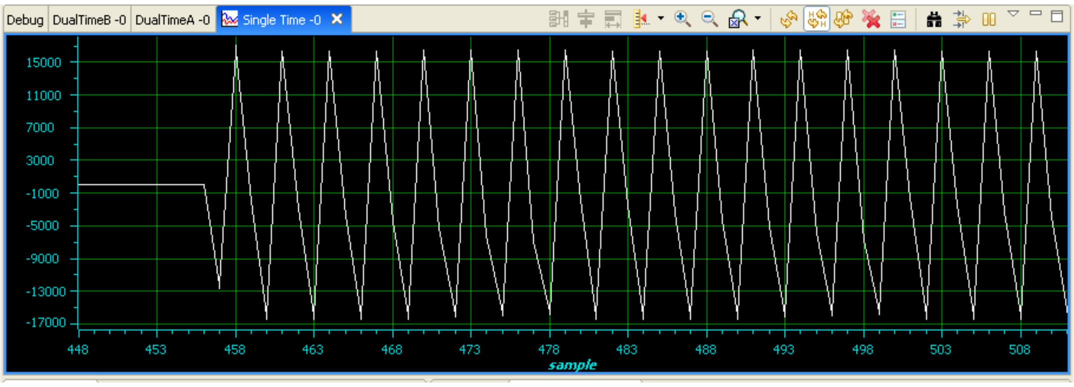
- Open CCstudio setup
- Go to File Menu , select Import option.
- In the Import Window under CCs choose Existing CCS/CCE Eclipse project then next.
- In Select root Directory Browse for project file where it is located.
- Select Audio Project folder and Finish it.
- Now Connect the DSP Kit to PC and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

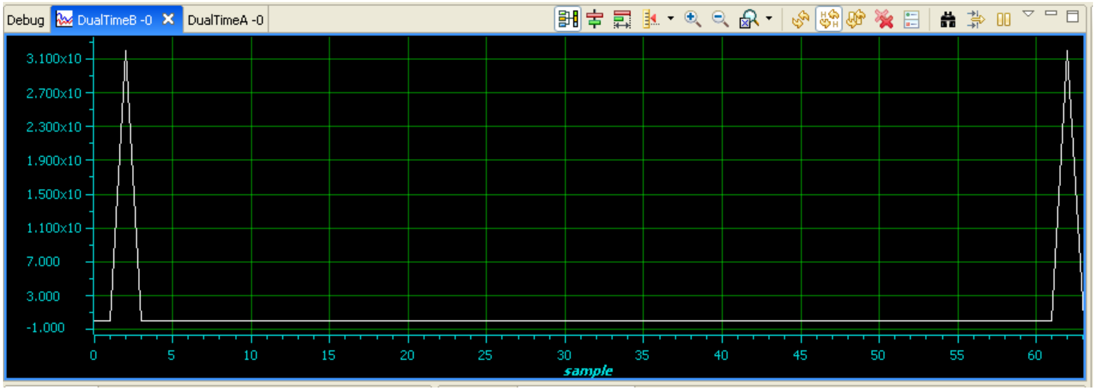
Graph Properties

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

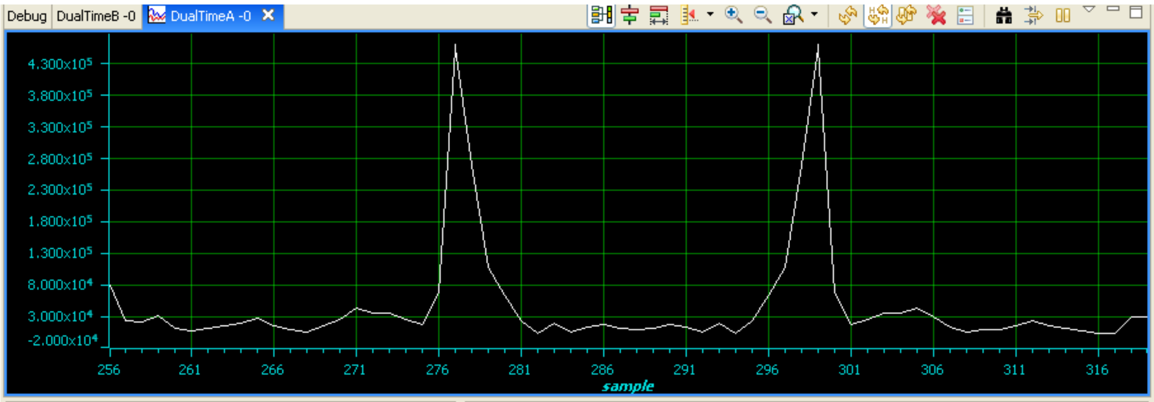
Graph Properties

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



EXP. NO.25
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).