JSS MAHAVIDYAPEETHA
JSS SCIENCE AND TECHNOLOGY UNIVERSITY
SRI JAYACHAMARAJENDRA COLLEGE OF ENGINEERING

- Constituent College of JSS Science and Technology University
- Approved by A.I.C.T.E
- Governed by the Grant-in-Aid Rules of Government of Karnataka
- Identified as lead institution for World Bank Assistance under TEQIP Scheme

DEPARTMENT OF ELECTRONICS AND COMMUNICATION ENGINEERING

DIGITAL SIGNAL PROCESSING LAB

(20EC57L)

V Semester

Room No.: AB208

Faculty in-charge:

Prof. B A Sujathakumari
Associate Professor

Dr. Shashidhar R
Assistant Professor

Vision statement of the JSS Science and Technology University

- Advancing JSS S&T University as a leader in education, research and technology on the International arena.
- To provide the students a universal platform to launch their careers, vesting the industry and research community with skilled and professional workforce.
- Accomplishing JSS S&T University as an epicenter for innovation, centre of excellence for research with state of the art lab facilities.
- Fostering an erudite, professional forum for researchers and industrialist to coexist and to work cohesively for the growth and development of science and technology for betterment of society.

Mission statement of the JSS Science and Technology University

- Education, research and social outreach are the core doctrines of JSS S&T University that are responsible for accomplishment of in-depth knowledge base, professional skill and innovative technologies required to improve the socio economic conditions of the country.
- Our mission is to develop JSS S&T University as a global destination for cohesive learning of engineering, science and management which are strongly supported with interdisciplinary research and academia.
- JSS S&T University is committed to provide world class amenities, infrastructural and technical support to the students, staff, researchers and industrial partners to promote and protect innovations and technologies through patents and to enrich entrepreneurial endeavors.
- JSS S&T University core mission is to create knowledge led economy through appropriate technologies, and to resolve societal problems by educational empowerment and ethics for better living.

Vision statement of the department of E&CE

Be a leader in providing globally acceptable education in electronics and communication engineering with emphasis on fundamentals-to-applications, creative-thinking, research and career- building.

Mission statement of the department of E&CE

- To provide best infrastructure and up-to-date curriculum with a conducive learning environment.
- To enable students to keep pace with emerging trends in Electronics and Communication Engineering.
- To establish strong industry participation and encourage student entrepreneurship.
- To promote socially relevant eco-friendly technologies and inculcate inclusive innovation activities.

Program Outcomes (POs)

1. **Engineering Knowledge**: Apply knowledge of mathematics, science, engineering fundamentals and an engineering specialization to the solution of complex engineering problems.

2. **Problem Analysis**: Identify, formulate, 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 specified needs with appropriate consideration for public health and safety, cultural, societal and environmental considerations.

4. **Conduct investigations of complex problems**: Using research based knowledge and research methods including design of experiments, analysis and interpretation of data and synthesis of 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 contextual knowledge to assess societal, health, safety, legal and cultural issues and the consequent responsibilities relevant to professional engineering practice.

7. **Environment and Sustainability**: Understand the impact of professional engineering solutions in societal and environmental contexts and demonstrate knowledge of and need for sustainable development.

8. **Ethics:** Apply ethical principles and commit to professional ethics and responsibilities and norms of 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. **Lifelong Learning**: Recognize the need for and have the preparation and ability to engage in independent and lifelong learning in the broadest context of technological change.

12. **Project Management and Finance**: Demonstrate knowledge and understanding of 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.

Program Specific Outcomes (PSOs)

1. Analyze, design and provide engineering solutions in the areas of electronic circuits and systems.

2. Demonstrate the mathematical modeling techniques, nurture analytical and computational skills to provide engineering solutions in the areas of electronics and communication.

3. Ability to address multidisciplinary research challenges and nurture entrepreneurship

Program Educational Objectives (PEOs)

1. To enable the graduates to have strong Engineering fundamentals in Electronics & Communication, with adequate orientation to mathematics and basic sciences.

2. To empower graduates to formulate, analyze, design and provide innovative solutions in Electronics & Communication, for real life problems.

3. To ensure that graduates have adequate exposure to research and emerging technologies through industry interaction and to inculcate professional and ethical values.

4. To nurture required skill sets to enable graduates to pursue successful professional career in industry, higher education, competitive exams and entrepreneurship.

Preface

This laboratory manual is prepared by the Department of Electronics and communication engineering of Sri Jayachamarajendra College of Engineering for Digital signal processing Laboratory. This lab manual is vital for studentsto complement theory with practical software and hardware applications in their curriculum. This lab manual can also be used as instructional book by staff and instructors to assist in performing and understanding the experiments.

The course aims at practical experience with the simulation and application of basicsignal processing algorithms, using standardized environments such as MATLAB and general-purpose DSP development kit such as TMS320C6713.

Experiments cover fundamental concepts of digital signal processing like samplingand aliasing, computation of discrete Fourier transform, verification of DFT properties, fast transforms, digital filter design and implementation, adaptive filtering, sampling-rate conversion and multi-rate processing.

The lab consists of 13 experiments of which 11 experiments are based on MATLAB simulation and two experiments on real time implementation using TMS Digital Signal Processing kit. Each lab session is of 3 hours' duration. This lab manual also provides a uniform evaluation scheme to be followed by the staff members handlingthis lab.

Course Title: Digital Signal Processing Lab	Course Code: 20EC57L
Credits : 1.5	Total Contact Hours (L:T:P): 0:0:30
Type of Course: Laboratory	Category: Professional Core Course
CIE Marks: 50	SEE Marks: 50

Pre-requisite: Signals and Systems, Digital Signal Processing

Course Objective: To make students familiar with the most important methods in DSP, including digital filter design, transform-domain processing and the importance of Signal Processors.

Course Outcomes: After completing this course, students should be able to

CO1:	Analyze and verify signal processing concepts and algorithms
CO2:	Design and demonstrate signal processing algorithms using simulation tool and/or Hardware platform

Expt. No.	Experiment Name	No. of Hours
1	Explore Digital Signal Processing Virtual Laboratory of Department of Electronics and Electrical Communication Engineering Indian Institute of Technology, Kharagpur http://www.digital.iitkgp.ernet.in/dsp/expts/index.php	3
2	a) Write a MATLAB code to illustrate the Nyquist sampling theorem. The program should illustrate the effects the sampling the signal at • Exactly the folding frequency • Frequency less than the folding frequency • Frequency greater than the folding frequency Plot the magnitude spectrum for all the above said cases b) Write a MATLAB code to compute the DTFT and DFT of a sequence x (n). Also plot the magnitude spectrum of both DTFT and DFT and provide the inference on the basis of results obtained. Further compute the IDTFT and IDFT.	3
3	Write a MATLAB code to verify the following properties of DFT a) Linearity b) Periodicity c) Circular shift and Circular symmetry of a sequence d) Symmetry property	3
4	Write a MATLAB code to verify the following properties of DFT a) Circular convolution and multiplication of two sequences. b) Time reversal of a sequence. c) Circular time shift and Circular frequency shift of a sequence. d) Parseval's theorem.	3
5	Write a MATLAB code to compute the DFT of a sequence x (n) using DIT and DIF algorithm. Also indicate the speed improvement factor in calculating the DFT of a sequence using direct computation and FFT algorithm (Use the same sequence as used in Program2).Further compute the IDFT using IDIT and IDIF	3

	algorithm.	
6	Write a MATLAB code to verify the Low pass and High Pass FIR linear phase filter design using Hamming and Hanning windows (with inbuilt and without using inbuilt commands). Plot the magnitude and phase response. Also, Provide the inference on the basis of results obtained for these to specifications. (To design should be verifiedmby convolving the input signal with the designed filter coefficients)	3
7	Write a MATLAB code to verify the Band pass and Band reject FIR linear phase filter design using Hamming and Hanning windows (with inbuilt and without using inbuilt commands). Plot the magnitude and phase response. Also, Provide the inference on the basis of results obtained for the set of specifications.	3
8	Write a MATLAB code to implement the Low pass Chebyshev (Type1) IIR filter design using bilinear transformation (BLT) method and Impulse Invariant Technique (IIT) method.	3
9	Write a MATLAB code to verify the Low pass Butterworth IIR filter design using bilinear transformation (BLT) method and Impulse Invariant Technique (IIT) method.	3
10	Write a MATLAB code to illustrate the effect of Decimation and Interpolation by an integer factor. Plot the magnitude spectrum. Design the necessary filter to overcome aliasing and image frequencies after decimating and inter-polating the signal respectively.	3
11	Read the data file named ecg2x60.datfrom http://people.ucalgary.ca/~ranga/enel563/SIGNAL_DATA_FILES/ That is corrupted with the 60Hz noise component. Write a MATLAB code to remove this 60Hz noise component from the signal using Notch filter and LMS adaptive filter. Plot the magnitude spectrum of the signal filtered using both Notch filter and LMS adaptive filter and provide the inference on the basis of results obtained.	3
	HardwareExperimentUsingTMS320C6713DSPKit	
12	a) Write a C code to obtain the impulse response of a given system and implement the same on TMS320C6713 DSK-kit. b) Write a C code to compute the linear and circular convolution and implement the same on TMS320C6713 DSK-kit.	3
13	a) Write a C code to compute the cross-correlation and auto-correlation and implement the same on TMS320C6713 DSK-kit. b) Write a C code to compute N-point DFT and IDFT of a sequence and implement the same on TMS320C6713 DSK-kit.	3

Reference Books:

1. **Sanjit K Mitra,**"*Digital Signal Processing Laboratory Using MATLAB*", McGraw Hill International Edition, 2002.
2. **Vinay K Ingle and John G Proakis,**"Digital *Signal Processing Laboratory Using MATLAB*", 3rd Edition, Cengage Learning, 2010.

Mapping - Course Outcomes with Program outcomes & Program Specific outcomes

	Program outcomes												Program specific outcomes		
	PO1	PO2	PO3	PO4	PO5	PO6	PO7	PO8	PO9	PO10	PO11	PO12	PSO1	PSO2	PSO3
CO1	3	3	2	2	2	0	0	0	2	3	0	2	3	3	0
CO2	3	3	3	2	3	0	0	0	2	3	0	2	3	3	0

0---No association, 1---Low association, 2--- Moderate association, 3---High association

Course Out comes

CO1: Analyze and verify signal processing concepts and algorithms

CO2: Design and demonstrate signal processing algorithms using simulation tool and/or Hardware platform

Evaluation Scheme

Sl. No	Scheme type	Weightage
1	CIE from experiments 1 to 13(A)	40 marks
2	Lab Test(B)	10 marks
3	Final CIE= A+B	50 marks

Continuous Internal Evaluation Scheme (CIE)

Sl. No	Evaluation Component	Duration	Marks
1	Preparedness	Continuous	08
2	Conduction	Continuous	08
3	Viva	Continuous	08
4	Report writing	Continuous	08
5	Result interpretation	Continuous	08

CIE Format

JSS MAHAVIDYAPEETHA: JSS SCIENCE AND TECHNOLOGY UNIVERSITY

RECORD OF CIE FOR PERFORMANCE IN THE LAB CLASSES

Section		Batch		Group Number	
Staff in Charge		Day		Timings	

Sl. No.	USN	Name Of The Student	AC1: Preparedness (8M)	SUBJECT:
1.			AC2: Conduction (8M)	20EC57L
2.			AC3: Viva (8M)	
3.			AC4: Report Writing (8M)	
4.			AC5: Result Interpretation (8M)	
			T: Total (40M)	

Sl. No.	Date	Experiment	Student-1						Student-2						Student-3						Student-4					
			AC 1	AC 2	AC 3	AC 4	AC 5	T	AC 1	AC 2	AC 3	AC 4	AC 5	T	AC 1	AC 2	AC 3	AC 4	AC 5	T	AC 1	AC 2	AC 3	AC 4	AC 5	T
1.																										
2.																										
3.																										
4.																										
5.																										

Sl. No.	Date	Experiment	Student-1						Student-2						Student-3						Student-4					
			AC 1	AC 2	AC 3	AC 4	AC 5	T	AC 1	AC 2	AC 3	AC 4	AC 5	T	AC 1	AC 2	AC 3	AC 4	AC 5	T	AC 1	AC 2	AC 3	AC 4	AC 5	T
6.																										
7.																										
8.																										
9.																										
10.																										
11.																										
12.																										
13.																										
CIE from Experiments 1 to 13 (40 Marks) (A)																										
	Lab test (10M) (B)																									
Final CIE = A + B																										
Attendance Percentage																										

Signature of the Staff in Charge: **Signature of the Lab in Charge:**

1.

2.

Experiment - 1

Explore Digital Signal Processing Virtual Laboratory of Department of Electronics and Electrical Communication Engineering Indian Institute of Technology, Kharagpur

http://www.digital.iitkgp.ernet.in/dsp/expts/index.php

Outcome: This experiment is to make student understand what virtual laboratory is and make them comfortable with the basic DSP concepts. The content of this website aims to provide a virtual laboratory platform for undergraduate Engineering students studying the course of Digital Signal Processing. The experiments designed will be two paced. For the relatively weaker students, it will dwell more on analysis part. For relatively stronger students, more challenging synthesis/design related experiments is made available.

Within each experiment, there will be many sub-experiments.

1. Study of sampling theorem, effect of under sampling.
2. Study of Quantization of continuous-amplitude, discrete-time analog signals.
3. Study of different types of Companding Techniques.
4. Study of properties of Linear time-invariant (LTI) system.
5. Study of convolution: series and parallel system.
6. Study of Discrete Fourier Transform (DFT) and its inverse.
7. Study of Transform domain properties and its use.
8. Study of FIR filter design using window method: Low pass and high pass filter.
9. Study of FIR filter design using window method: Band pass and Band stop filter.
10. Study of Infinite Impulse Response (IIR) filter.

Experiment - 2

a) Write a MATLAB code to illustrate the Nyquist sampling theorem. The program should illustrate the effects the sampling the signal at

- Frequency greater than or equal to the folding frequency
- Frequency less than the folding frequency
- Frequency greater than the folding frequency

Plot the magnitude spectrum for all the above said cases

Outcome: The student will be able to practically verify the concept of sampling theorem and apply sampling theorem on any given continuous input signal in order to reconstruct the signals exactly from its samples.

Theory Background:

Sampling theorem is a fundamental bridge between continuous-time signals (often called "analog signals") and discrete-time signals (often called "digital signals"). It establishes a sufficient condition for a sample rate that permits a discrete sequence of *samples* to capture all the information from a continuous-time signal of finite bandwidth.

A bandlimited signal can be reconstructed exactly if it is sampled at a rate atleast twice the maximum frequency component in it - Nyquist

Alternatively, the maximum frequency content of a signal (f_{max}) should be less than or equal to the folding frequency ($f_{s/2}$)

$$f_{max} \leq f_{s/2}$$

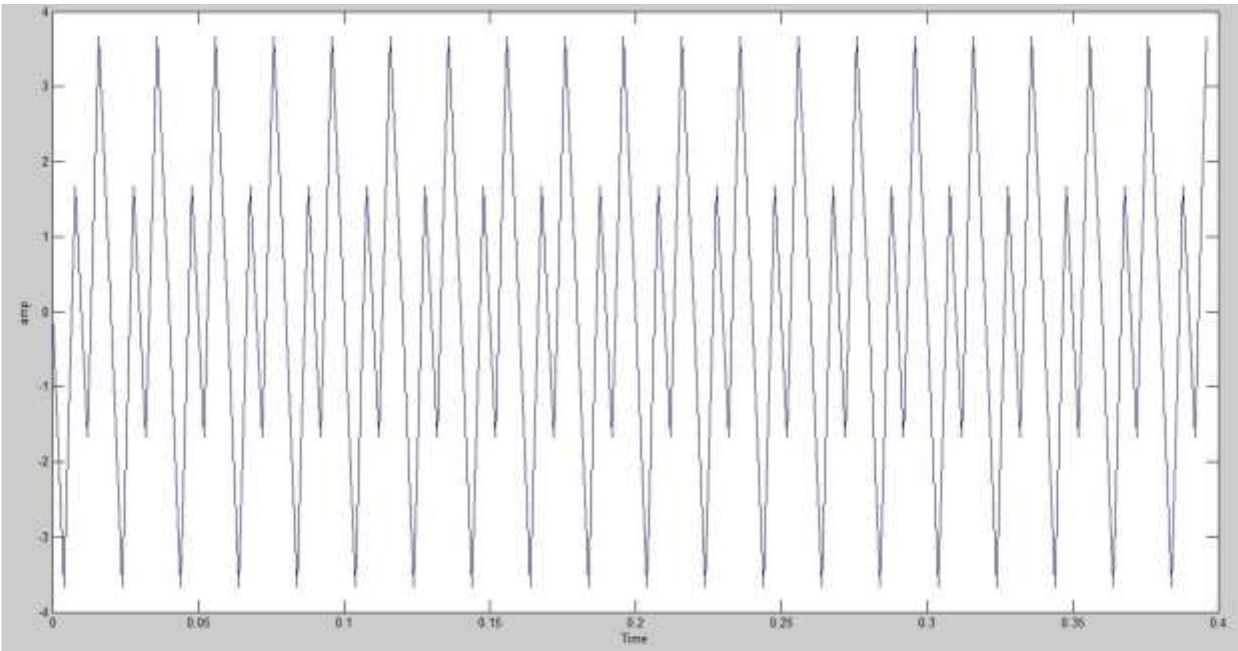
To be demonstrated:

Input:

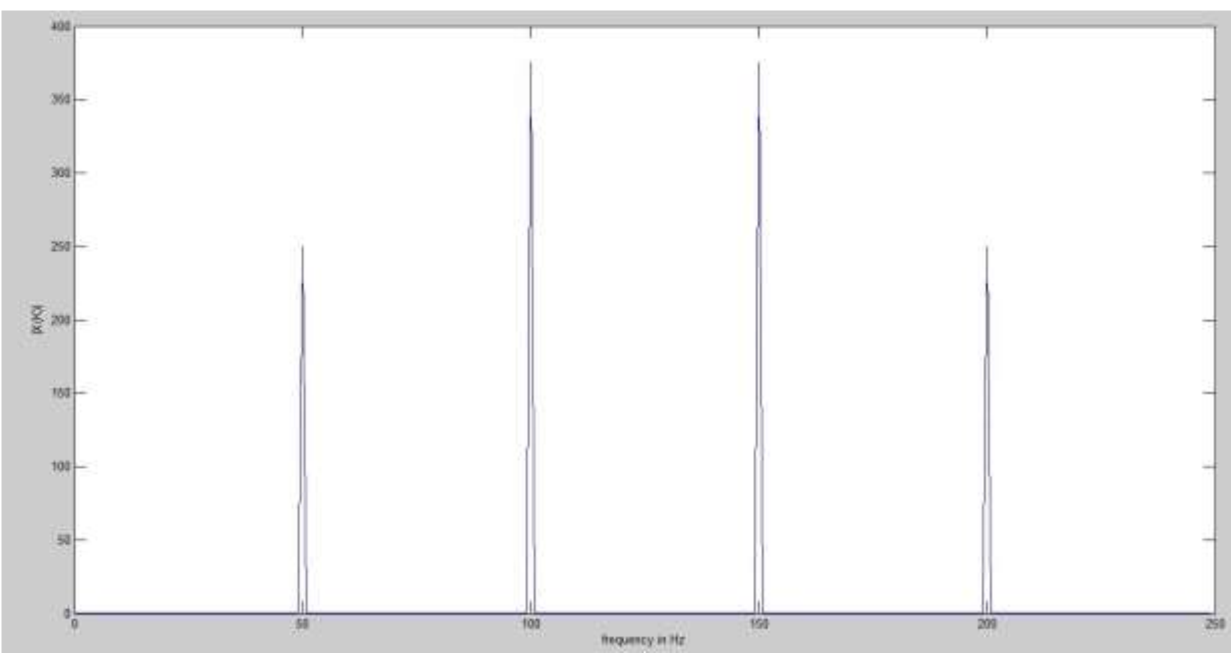
For the input signal **x (t) =2*sin (2*pi*f1*t) +3*sin (2*pi*f2*t) with f1=200Hz and f2=400Hz apply the sampling theorem to the signal to obtain each of the following outputs and provide your inference for each case.**

Case-1: Both the frequency components of the input signal are greater than folding frequency. (Case of aliasing)

Time domain plot

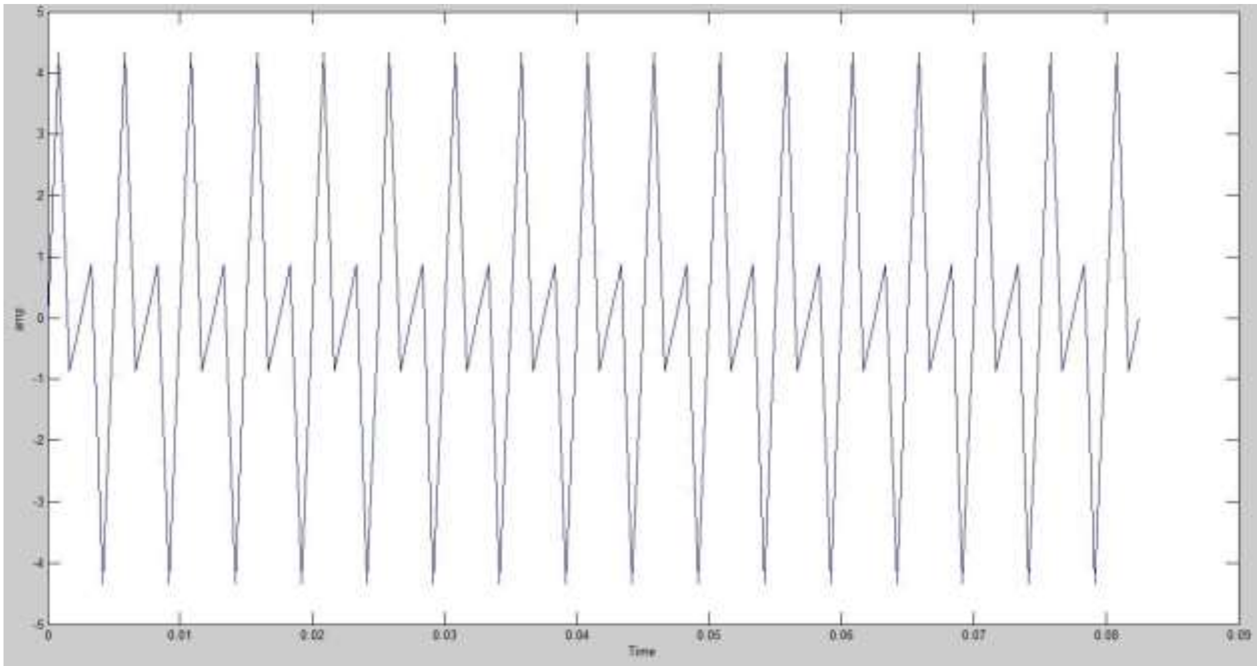


Frequency Plot

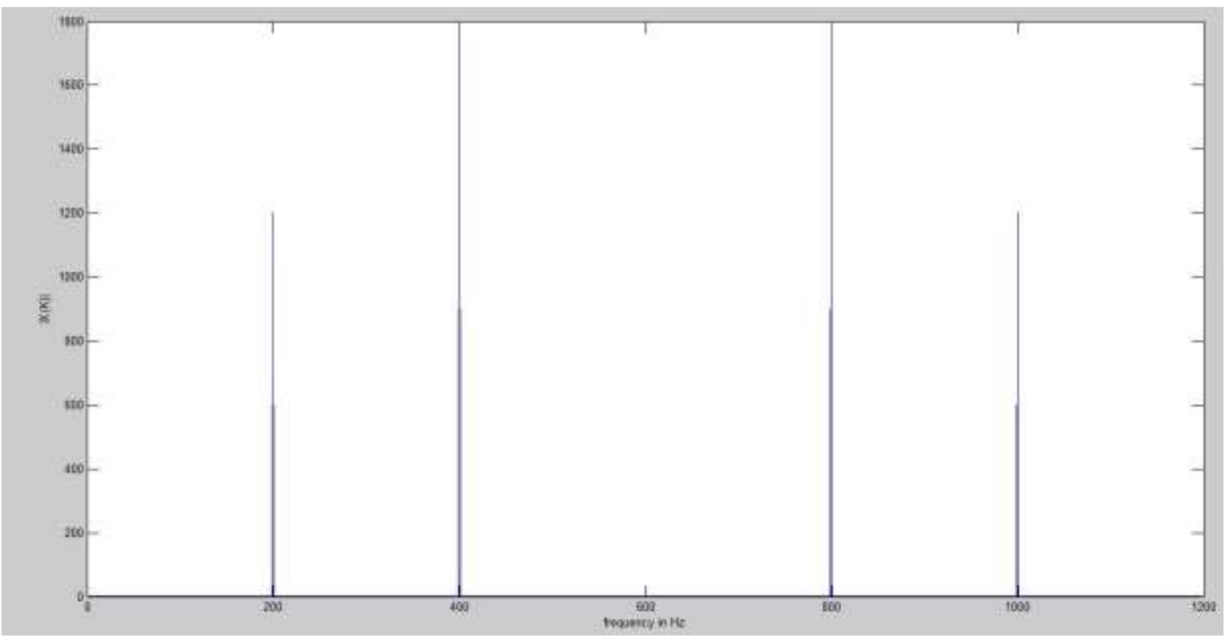


Case-2: Both the frequency components of the input signal are less than folding frequency. (Case of perfect reconstruction)

Time domain plot

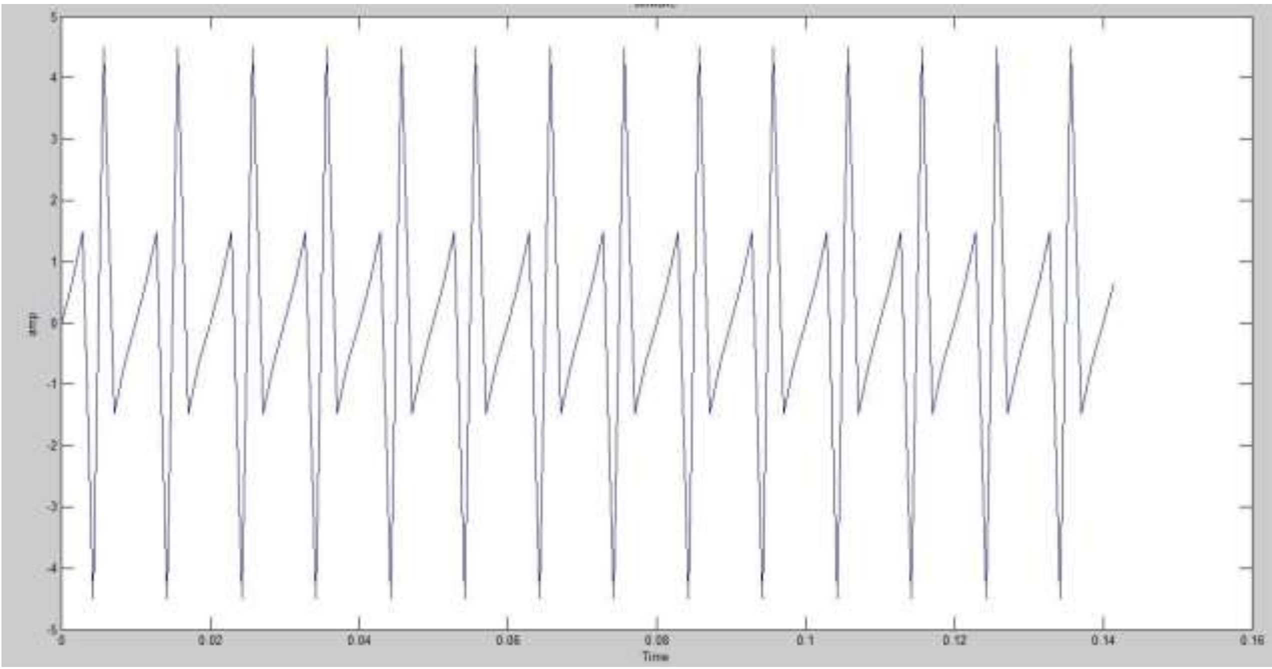


Frequency Plot

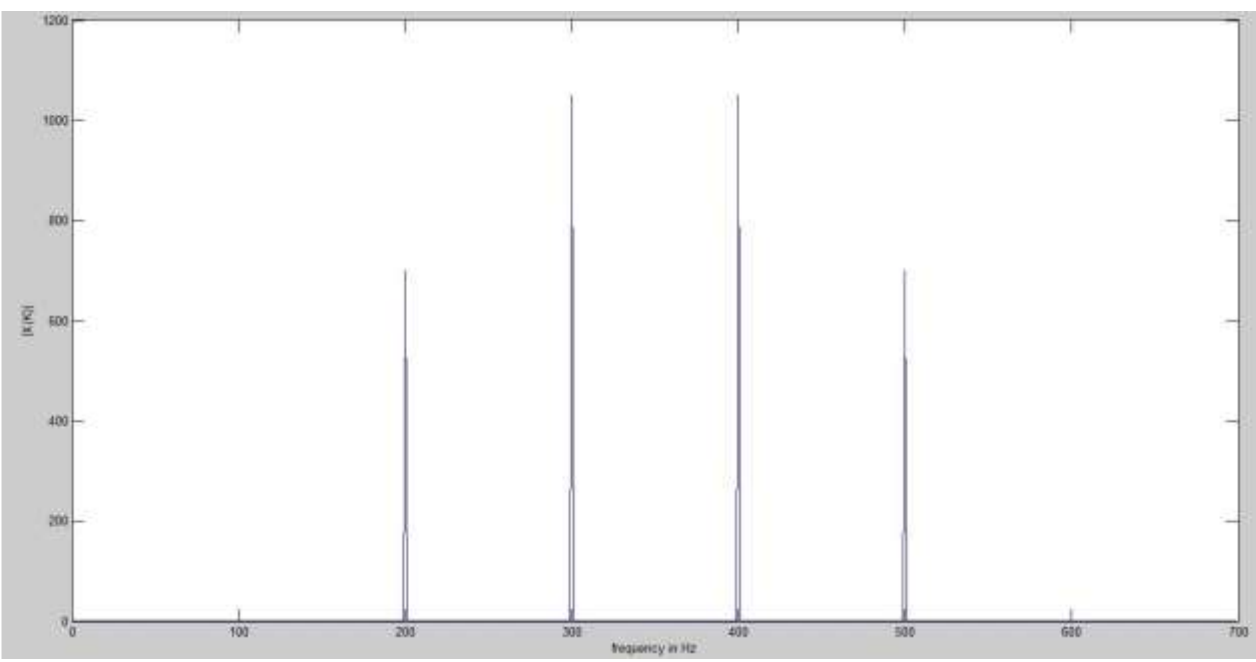


Case-3: One of the input frequency component is greater than folding frequency and one less than folding frequency.

Time domain plot



Frequency Plot



b) Write a MATLAB code to compute the DTFT and DFT of a sequence x (n). Also plot the magnitude spectrum of both DTFT and DFT and provide the inference on the basis of results obtained. Further compute the IDTFT and IDFT.

Outcome: The student will be able to practically compute the DFT and IDFT of a given sequence.

Theory Background:

Most time signals in practice are continuous and non-periodic, and their analytical expressions are not available in general. The spectrum of such a non-periodic and continuous signal can only be obtained numerically by a digital computer. To do so, the signal needs to be modified in two ways:

- First, we need to truncate the signal so that it has a finite time duration from 0 to T, with the underlying assumption that the signal repeats itself outside the interval 0<t<T, i.e., it becomes a periodic with period T. Correspondingly, its spectrum becomes discrete.
- Second, we need to sample the signal with some sampling frequency F so that it becomes discrete to be processed by a digital computer. Correspondingly, the spectrum of the signal becomes periodic.

The order can be reversed so that the continuous signal is sampled first and then truncated. In either case, only when the signal is both finite and discrete, can we apply the *discrete Fourier transform (DFT)* to find its spectrum, which is also discrete and finite.

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

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

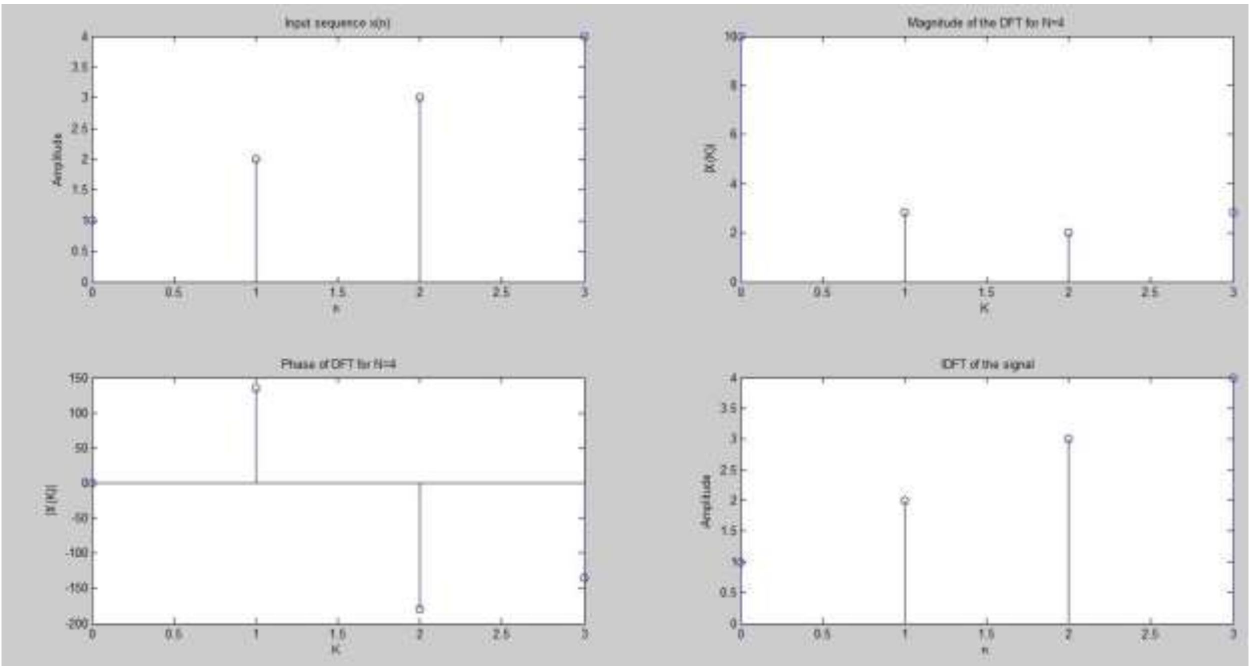
F(*k*) are often called the 'Fourier Coefficients' or 'Harmonics'.

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

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

To be demonstrated:

Input: For the input sequence x(n) = [1 2 3 4] compute the DFT of the sequence and then obtain the magnitude and phase plot as shown below. Also compute the IDFT of the DFT sequence. Compute the DFT for N=4, 8 & 16 and provide your inference.



Experiment - 3

Write a MATLAB code to verify the following properties of DFT

Outcome: The student will be able to appreciate the properties of the DFT and apply it to the practical problems.

a) Linearity

If x1(n) and x2(n) denote two periodic signals with period N and its Fourier Transform coefficients denoted by X1(k) and X2(k) respectively, then by linearity, the Fourier transform of x1(n)+x2(n) is given by X1(k)+X2(k).

To demonstrate:

Verify the property of linearity for the given input sequences:

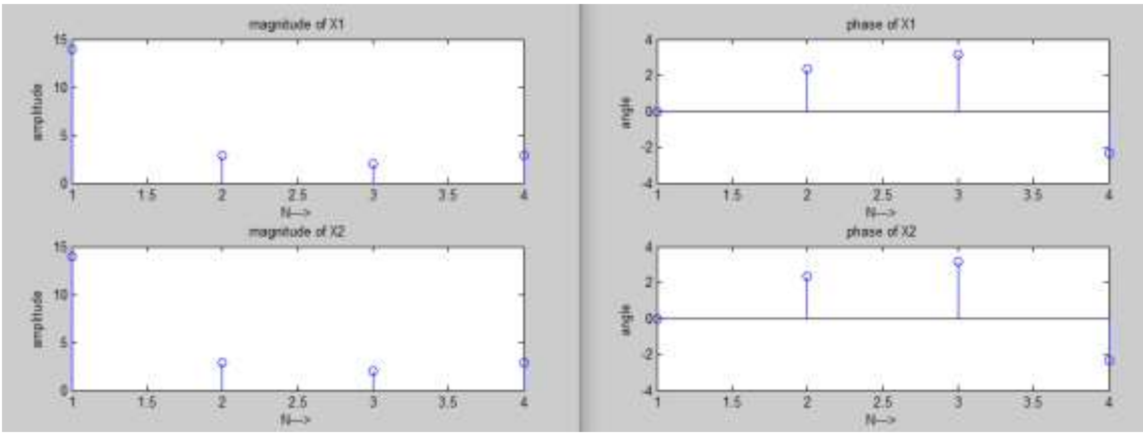
Sequence 1: [2 3 4 5]

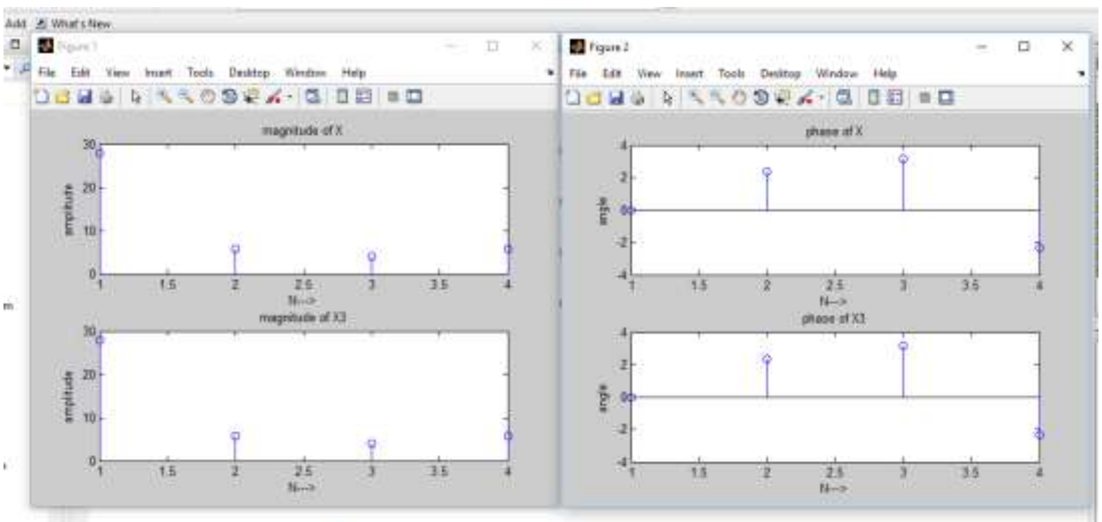
Sequence 2: [2 3 4 5]

Command Window

LINEARITY PROPERTY

```
enter sequence 1:
[2 3 4 5]
enter sequence 2:
[2 3 4 5]
 dft of sequence 1
  14.0000            -2.0000 + 2.0000i  -2.0000            -2.0000 - 2.0000i

 dft of sequence 2
  14.0000            -2.0000 + 2.0000i  -2.0000            -2.0000 - 2.0000i

 dft of x1(n)+x2(n) is
  28.0000            -4.0000 + 4.0000i  -4.0000            -4.0000 - 4.0000i

 sum of X1(k) and X2(k) is
  28.0000            -4.0000 + 4.0000i  -4.0000            -4.0000 - 4.0000i

fx >>
```





LINEARITY MAGNITUDE AND PHASE GRAPHS

b) Periodicity

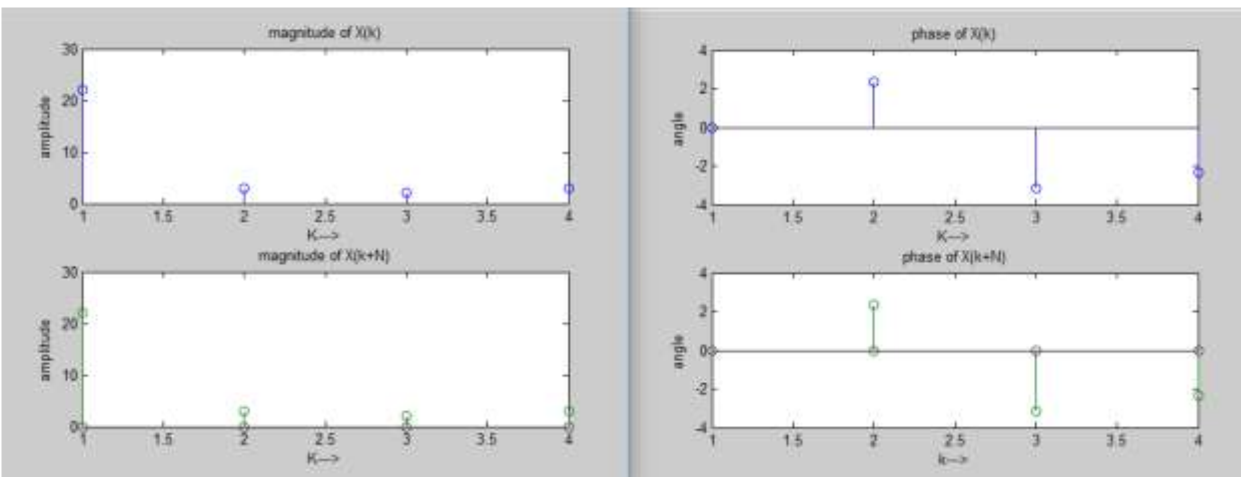
This property states that if X(k) is the DFT of the sequence x(n) then if x(n+N)=x(n) for all n, then ‘ X(k) = X(k + N) for all k.

To demonstrate:

Verify the property of periodicity for the given input sequence: [4 5 6 7]

```
periodicity property
enter the sequence [ 4 5 6 7]
 sequence you entered is x(n):
     4     5     6     7

 dft of x(n)  is X(k):
  22.0000            -2.0000 + 2.0000i  -2.0000 - 0.0000i  -2.0000 - 2.0000i

 X(N+k):
  Columns 1 through 6

        0            22.0000 + 0.0000i        0                  0                  0                  0
        0            -2.0000 + 2.0000i        0                  0                  0                  0
        0            -2.0000 - 0.0000i        0                  0                  0                  0
        0            -2.0000 - 2.0000i        0                  0                  0                  0

  Column 7

        0
        0
        0
        0
```

Periodicity property



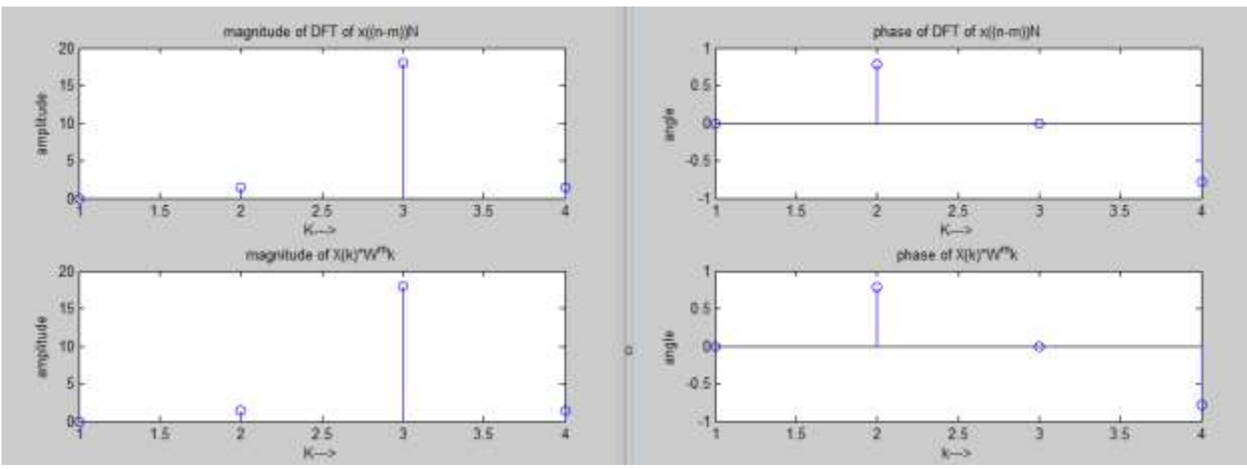
c) Circular time shift property of a sequence

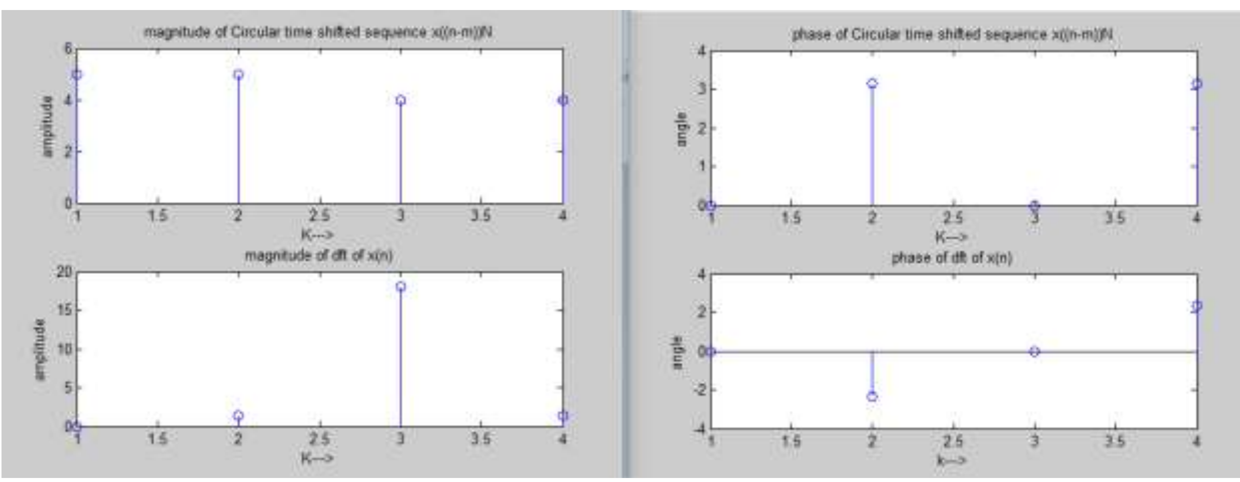
Circular time shift property states that if X(k) is the DFT of the sequence x(n) then shifting the sequence circularly by ‘m’ samples is equivalent to multiplying its DFT by e^(-j*2*pi*k*m / N)

To demonstrate:

Verify the property of Circular time shift for the given input sequence: [4 -4 5 -5] with number of shifts equal to 2.

```
Enter the sequence
[4 -4 5 -5]
Enter the number of shifts
2
DFT of the input sequence x(n) is
        0             -1.0000 - 1.0000i  18.0000            -1.0000 + 1.0000i

Circular time shifted sequence x((n-m))N
     5    -5     4    -4

DFT of x((n-m))N
        0              1.0000 + 1.0000i  18.0000             1.0000 - 1.0000i

X(k)*W^mk
        0              1.0000 + 1.0000i  18.0000 + 0.0000i   1.0000 - 1.0000i

Thus Circular-Time Shift Property is verified
>>
```





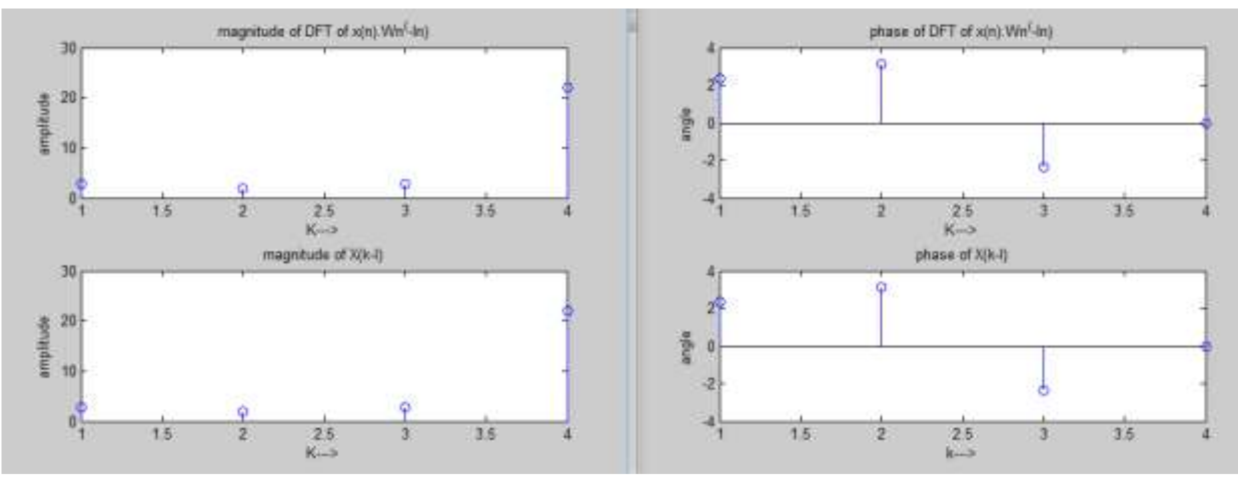
d) Circular frequency shift property of a sequence

Circular frequency shift property states that if X(k) is the DFT of the sequence x(n) then multiplying x(n) by e^(j*2*pi*l*n / N) is equivalent to shifting the DFT of the sequence x(n) circularly by ‘l’ samples as X((k − l))N.

To demonstrate:

Verify the property of Circular frequency shift for the given input sequence: [4 5 6 7] with number of shifts equal to 3.

```
Enter the sequence
[4 5 6 7]
Enter the delay
3
 DFT of the input sequence x(n)
  22.0000            -2.0000 + 2.0000i  -2.0000             -2.0000 - 2.0000i

Circular frequency shift of X(K) i.e DFT of x(n).e^(j*2*pi*l*n / N)
  -2.0000 + 2.0000i  -2.0000            -2.0000 - 2.0000i  22.0000

Resultant circular frequency shift i.e X((k − l))N
  -2.0000 - 2.0000i  22.0000 + 0.0000i  -2.0000 + 2.0000i  -2.0000 - 0.0000i

Thus Circular-Frequency Shift Property is verified
fx >>
```



Experiment - 4

Write a MATLAB code to verify the following properties of DFT

Outcome: The student will be able to appreciate the properties of the DFT and apply it to the practical problems.

a) Conjugate Symmetry property

The Conjugate symmetry property states that the DFT of '$x^*(n)$ ' is given by ' $X^*(N-k)$ ' which is also equal to $X^*((-k))N$.

To demonstrate:

Verify the property of Conjugate Symmetry for the given input sequence:

Real input sequence: [4 5 6 7]

Imaginary input sequence: [2i 3 4 5]

```
 Symmetric property
enter the real input sequence [4 5 6 7]

N =

     4

 it has symmetry
  22.0000             -2.0000 + 2.0000i  -2.0000             -2.0000 - 2.0000i

>>
```

Conjugate symmetry property for a real sequence

```
 Symmetric property
enter the real input sequence [2i 3 4 5]

N =

     4

 it doesnt have symmetry
  12.0000 + 2.0000i  -4.0000 + 4.0000i  -4.0000 + 2.0000i  -4.0000

>>
```

b)Circular convolution of two sequences

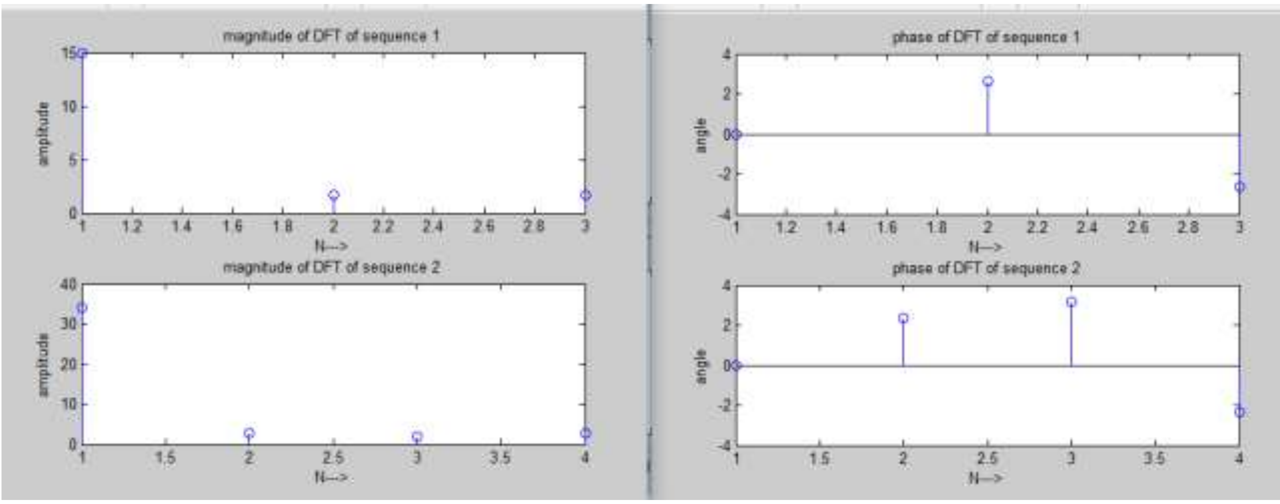
The circular convolution of two sequences x1(n) and x2(n) in time domain i.e. 'x1(n) ⊗ x2(n) ' is represented as multiplication i.e. ' X1(k).X2(k) ' in frequency domain.

To demonstrate:

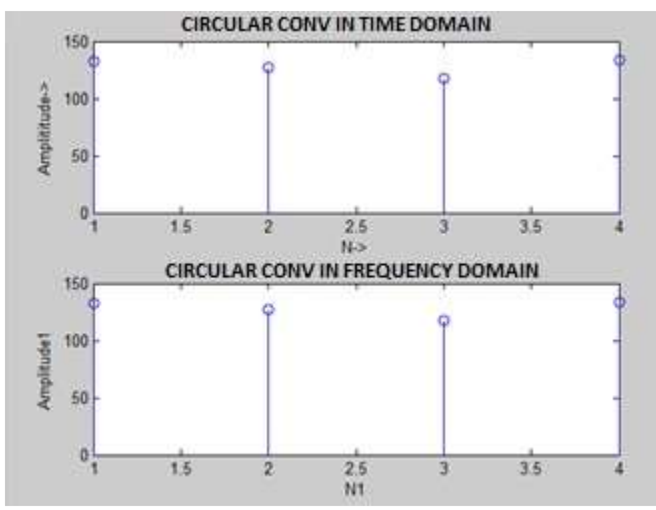
Verify the property of circular convolution for the given input sequences:

Sequence 1: [4 5 6]

Sequence 2: [7 8 9 10]




```
Enter the sequence 1:[4 5 6]
Enter the sequence 2:[7 8 9 10]
The resultant time domain convolution is
   132    127    118    133

conv in freq domain:
   132    127    118    133

DFT of sequence 1:
  15.0000              -1.5000 + 0.8660i  -1.5000 - 0.8660i

DFT of sequence 2:
  34.0000              -2.0000 + 2.0000i  -2.0000              -2.0000 - 2.0000i

>>
```



c) Time reversal of a sequence

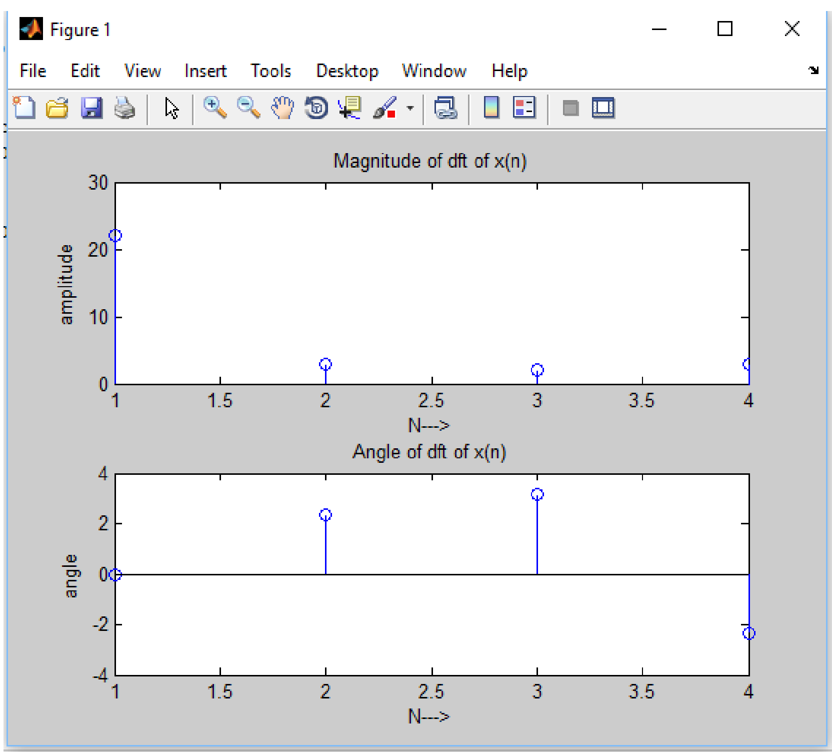
Time reversal of a sequence is demonstrated as: DFT of ‘x(N − n) ‘ is given by ‘ X(N − k) ‘.

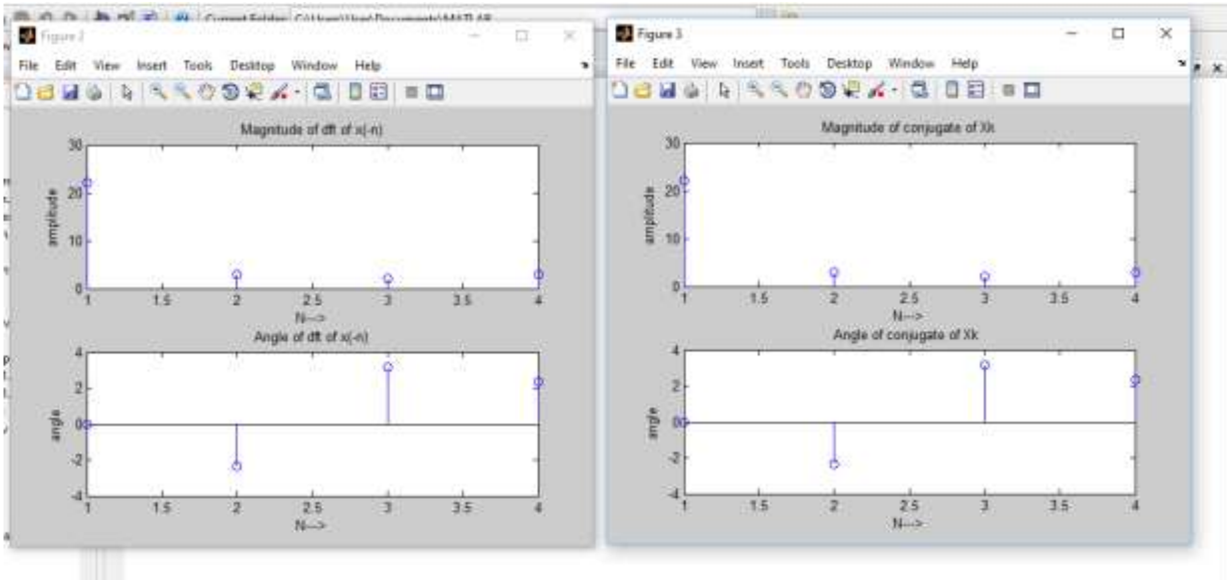
To demonstrate:

Verify the property of Time reversal for the given input sequence: [4 5 6 7]

```
Enter the sequence
[4 5 6 7]
DFT of the sequence x(n)
  22.0000             -2.0000 + 2.0000i  -2.0000             -2.0000 - 2.0000i

Time reversed sequence x(-n)
     4     7     6     5

DFT of time reversed sequence
  22.0000             -2.0000 - 2.0000i  -2.0000             -2.0000 + 2.0000i

Conjugate of X(k)
  22.0000             -2.0000 - 2.0000i  -2.0000             -2.0000 + 2.0000i

Thus Time-reversal Property is verified
>>
```

Time reversal property





d) Parseval's theorem

Parsevals theorem defines energy in time domain in the form $\sum_{n=0}^{N-1} x(n)y^*(n)$ which is equivalently represented in the frequency domain as $\frac{1}{N}\sum_{k=0}^{N-1} X(k)Y^*(k)$.

To demonstrate:

Verify Parseval's theorem for the given input sequence: [0 0.707 1 0.707]

```
Enter the sequence
[0 0.707 1 0.707]
Energy in time domain
    1.9997

Energy in frequency domain
    1.9997

Thus Parsevals theorem is verified
fx >>
```

Experiment - 5

Write a MATLAB code to compute the DFT of a sequence x (n) using DIT and DIF algorithm. Also indicate the speed improvement factor in calculating the DFT of a sequence using direct computation and FFT algorithm .Further compute the IDFT using IDIT and IDIF algorithm.

Outcome: The student will be able to apply computationally efficient algorithms for evaluating the DFT.

A fast Fourier transform (FFT) is any fast algorithm for computing the DFT. The development of FFT algorithms had a tremendous impact on computational aspects of signal processing and applied science. The DFT of an N-point signal

$$\{x[n], 0 \leq n \leq N-1\}$$

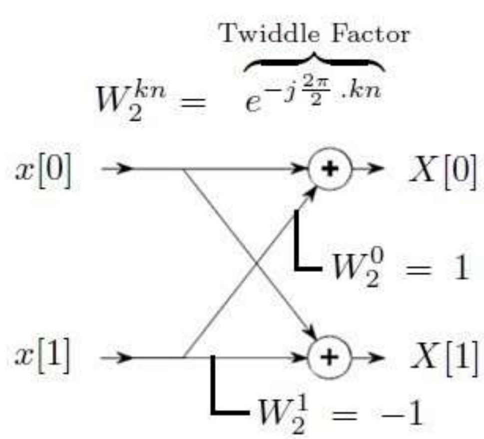
is defined as

$$X[k] = \sum_{n=0}^{N-1} x[n] \, W_N^{-kn}, \qquad 0 \leq k \leq N-1$$

where

$$W_N = e^{j\frac{2\pi}{N}} = \cos\left(\frac{2\pi}{N}\right) + j\,\sin\left(\frac{2\pi}{N}\right)$$

is the principal N-th root of unity.



I. DFT using DIT Algorithm

Consider an N-point signal $x[n]$ of even length. The derivation of the DIT radix-2 FFT begins by splitting the sum into two parts — one part for the even-indexed values $x[2n]$ and one part for the odd-indexed values $x[2n+1]$. Define two $N/2$-point signals $x_1[n]$ and $x_2[n]$ as

$$x_0[n] = x[2n]$$
$$x_1[n] = x[2n+1]$$

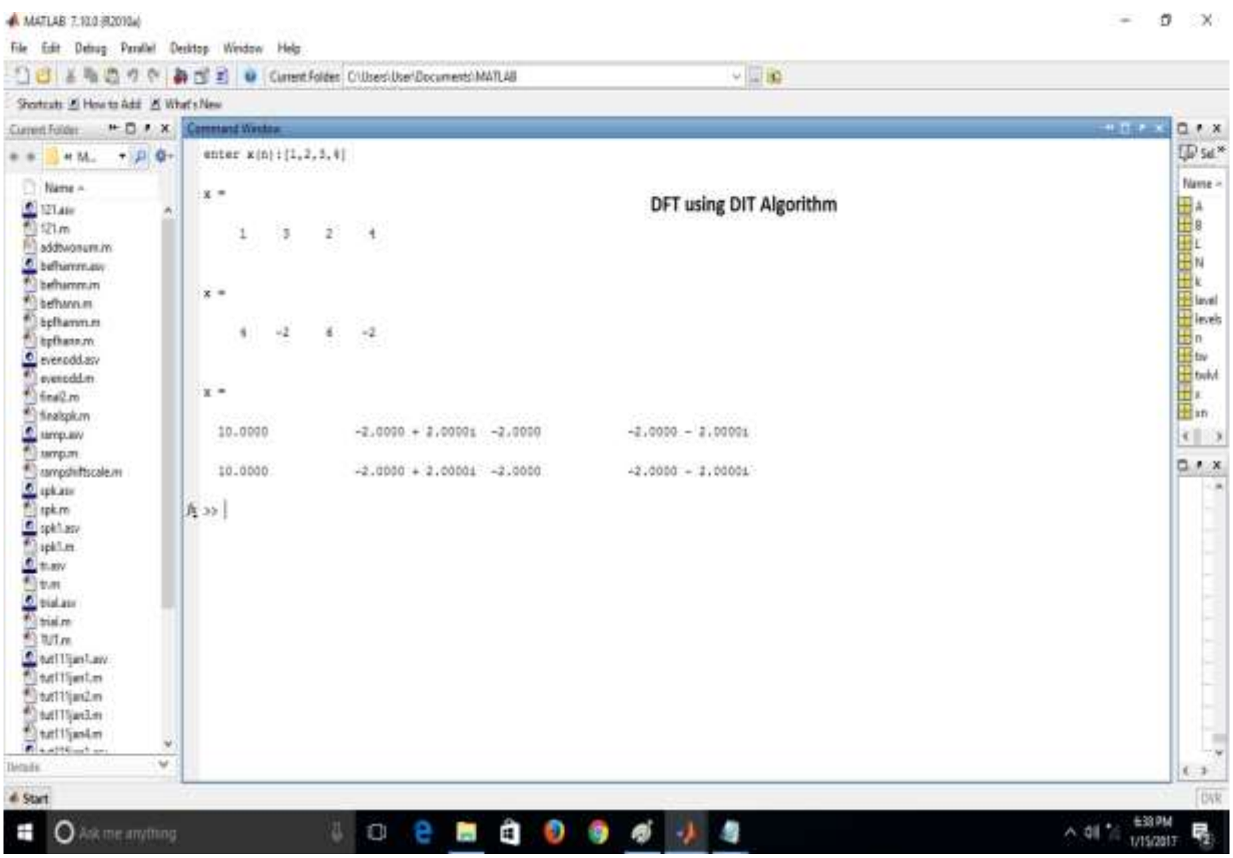
for $0 \le n \le N/2 - 1$. The DFT of the N-point signal $x[n]$ can be written as

$$X[k] = \sum_{\substack{n=0 \\ n \text{ even}}}^{N-1} x[n]\, W_N^{-nk} + \sum_{\substack{n=0 \\ n \text{ odd}}}^{N-1} x[n]\, W_N^{-nk}$$

To demonstrate:

To find the DFT of the given input sequence using DIT Algorithm

Input : [1 2 3 4]



II. DFT using DIF Algorithm

Consider an N-point signal $x[n]$ of even length. The derivation of the DIF radix-2 FFT begins by splitting the DFT coefficients $X[k]$ in to even- and odd- indexed values. The even values $X[2k]$ are given by:

$$\begin{aligned} X[2k] &= \sum_{n=0}^{N-1} x[n]\, W_N^{-2kn} \\ &= \sum_{n=0}^{N-1} x[n]\, W_{N/2}^{-kn}. \end{aligned}$$

Splitting this sum into the first $N/2$ and second $N/2$ terms gives

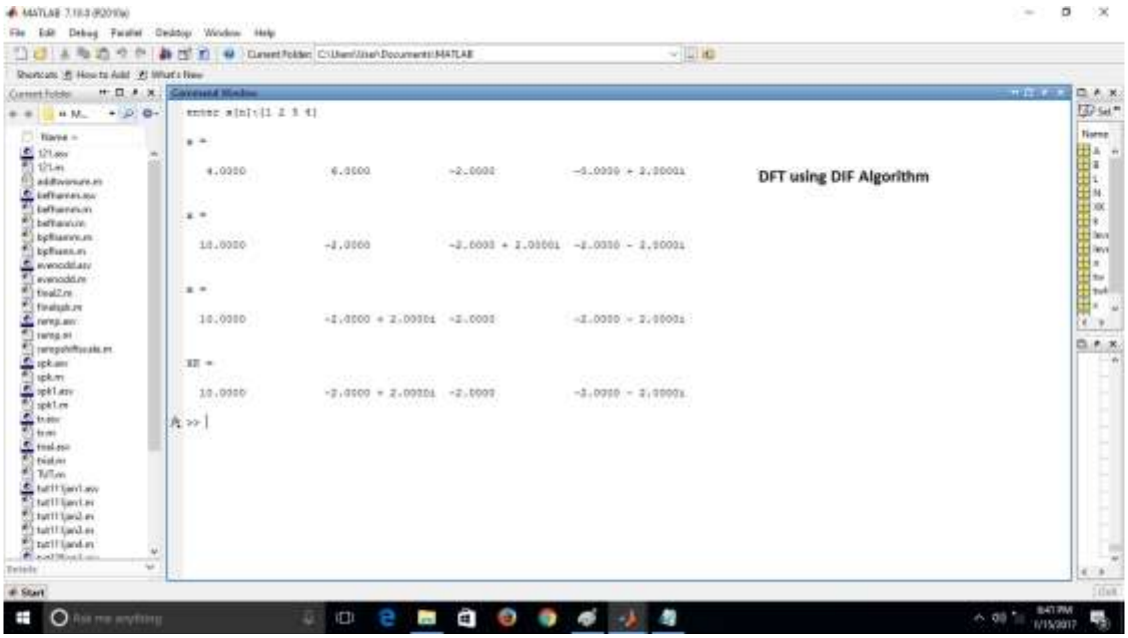
$$\begin{aligned} X[2k] &= \sum_{n=0}^{\frac{N}{2}-1} x[n]\, W_{N/2}^{-kn} + \sum_{n=\frac{N}{2}}^{N-1} x[n]\, W_{N/2}^{-kn} \\ &= \sum_{n=0}^{\frac{N}{2}-1} x[n]\, W_{N/2}^{-kn} + \sum_{n=0}^{\frac{N}{2}-1} x[n+\frac{N}{2}]\, W_{N/2}^{-k(n+\frac{N}{2})} \\ &= \sum_{n=0}^{\frac{N}{2}-1} x[n]\, W_{N/2}^{-kn} + \sum_{n=0}^{\frac{N}{2}-1} x[n+\frac{N}{2}]\, W_{N/2}^{-kn} \\ &= \sum_{n=0}^{\frac{N}{2}-1} \left(x[n] + x[n+\frac{N}{2}] \right) W_{N/2}^{-kn} \\ &= \mathrm{DFT}_{\frac{N}{2}} \left\{ x[n] + x[n+\frac{N}{2}] \right\}. \end{aligned}$$

That is, the even DFT values $X[2k]$ for $0 \le 2\,k \le N-1$ are given by the $\frac{N}{2}$-point DFT of the $\frac{N}{2}$-point signal $x[n] + x[n+N/2]$.

To demonstrate:

To find the DFT of the given input sequence using DIF Algorithm

Input: [1 2 3 4]

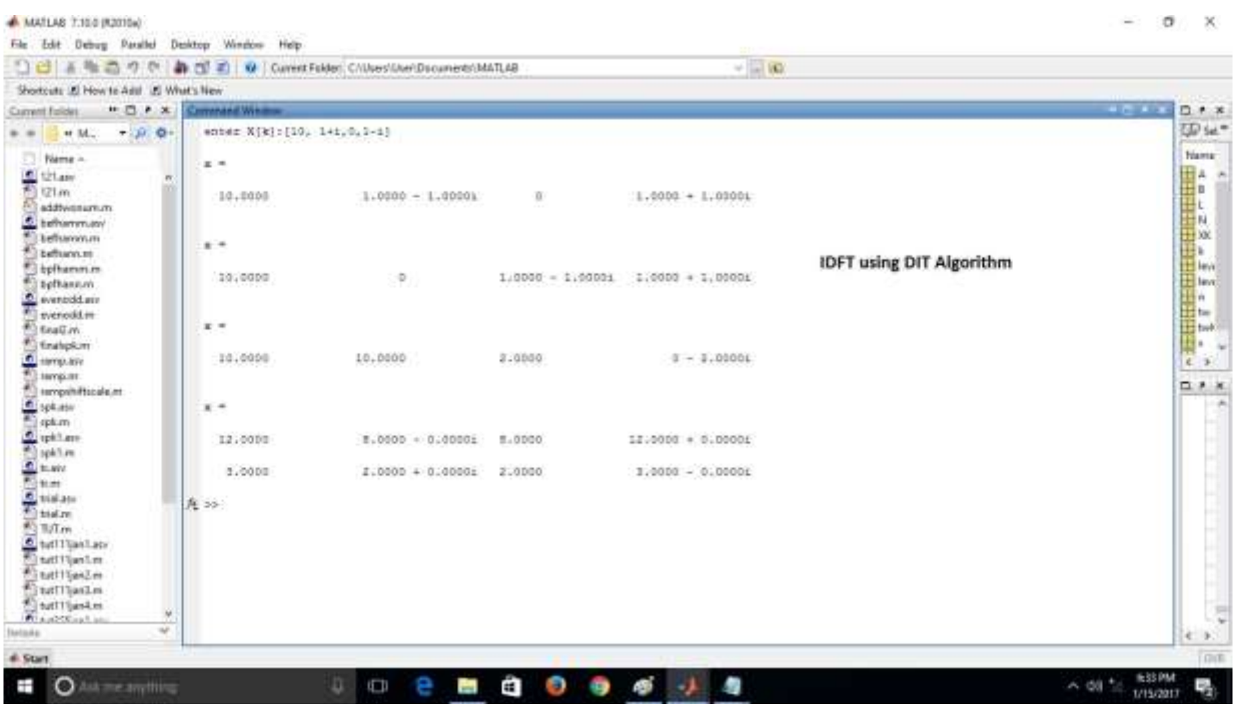


III. IDFT using DIT Algorithm

To demonstrate:

To find the IDFT of the given input sequence using DIT Algorithm

Input : [10 1+i 0 1-i]

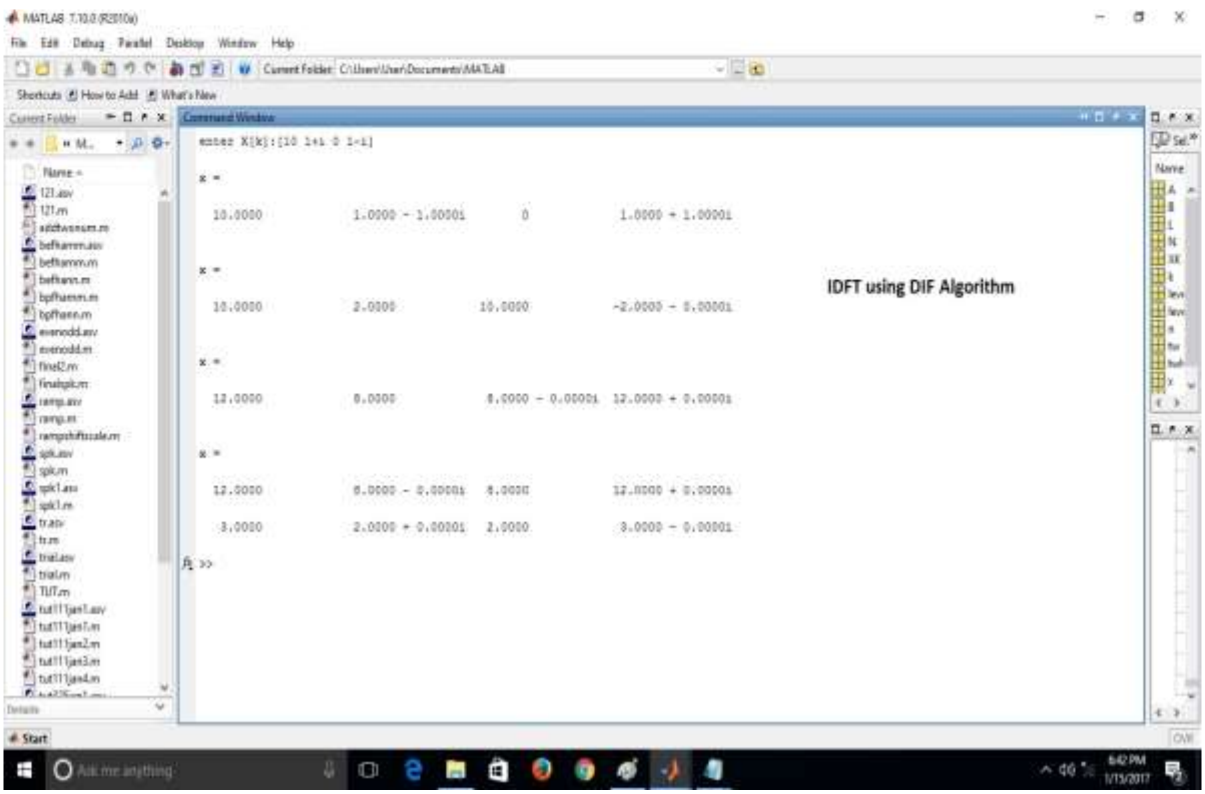


IV. IDFT using DIF Algorithm

To demonstrate:

To find the IDFT of the given input sequence using DIF Algorithm

Input : [10 1+i 0 1-i]



Experiment - 6

Write a MATLAB code to verify the Low pass and High Pass FIR linear phase filter design using Hamming and Hanning windows. Plot the magnitude and phase response. Also, Provide the inference on the basis of results obtained for the set of specifications. (To design should be verified by convolving the input signal with the designed filter coefficients).

Outcome: The student will be able to design and implement a linear phase FIR filter that best matches the application and satisfies the design requirements.

THEORY: A digital filter is a system which passes some desired signals more than others to reduce or enhance certain aspects of that signal. It can be used to pass the signals according to the specified frequency pass-band and reject the other frequency than the pass-band specification. The basic filter types can be divided into four categories:

1. Low-pass
2. High-pass
3. Band-pass
4. Band-stop.

On the basis of impulse response, there are two fundamental types of digital Filters:

1. Infinite Impulse Response (IIR) filters
2. Finite Impulse Response (FIR) filters.

FIR filter is described by the difference equation

$$y(n) = h(k)x(n-k)$$

Where, x(n) is input signal and h(n) is impulse response of FIR filter. The transfer function of a causal FIR filter is described by

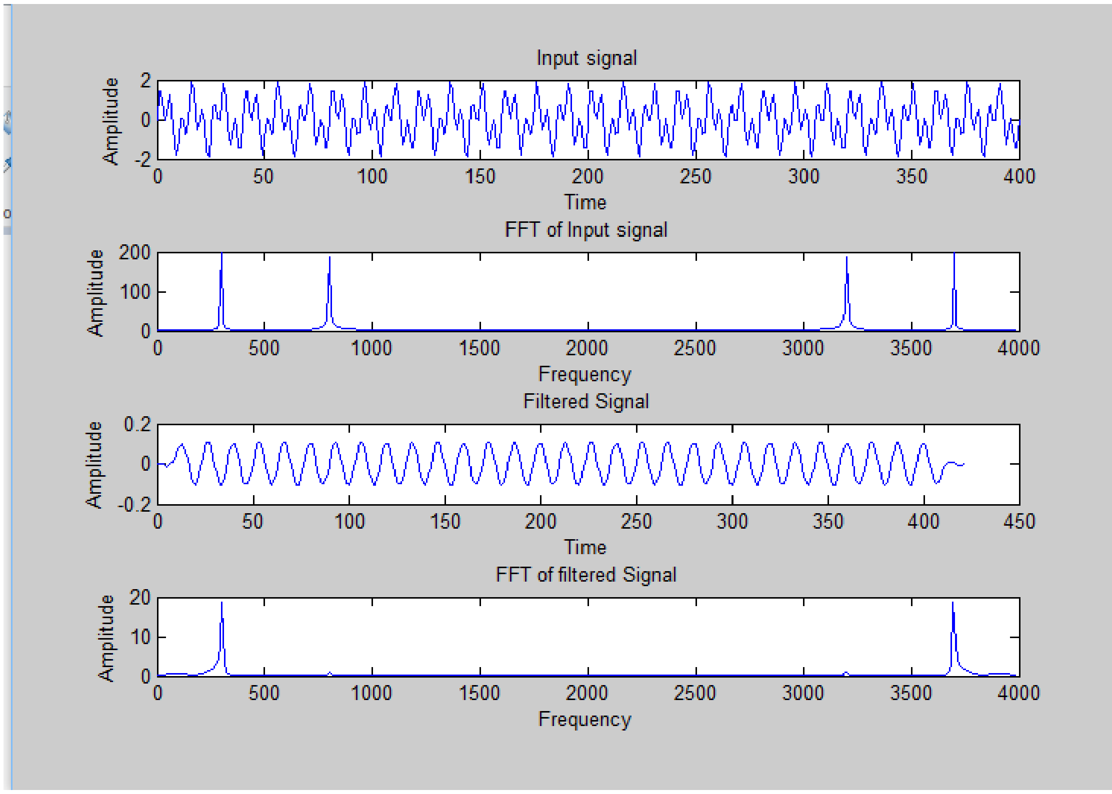
$$H(z) = \sum_{k=0}^{N-1} h(k) z^{-k}$$

A simple and effective way to design digital FIR filter is window method.
Window functions are used for designing a FIR filter.
(i). Hanning window function w (n) =0.5 − 0.5 cos (2nπ/N-1) , 0 ≤ n ≤ N – 1 0, otherwise
(ii). Hamming window function w (n) =0.54 − 0.46 cos (2nπ/N−1), 0 ≤ n ≤ N − 1 0, otherwise

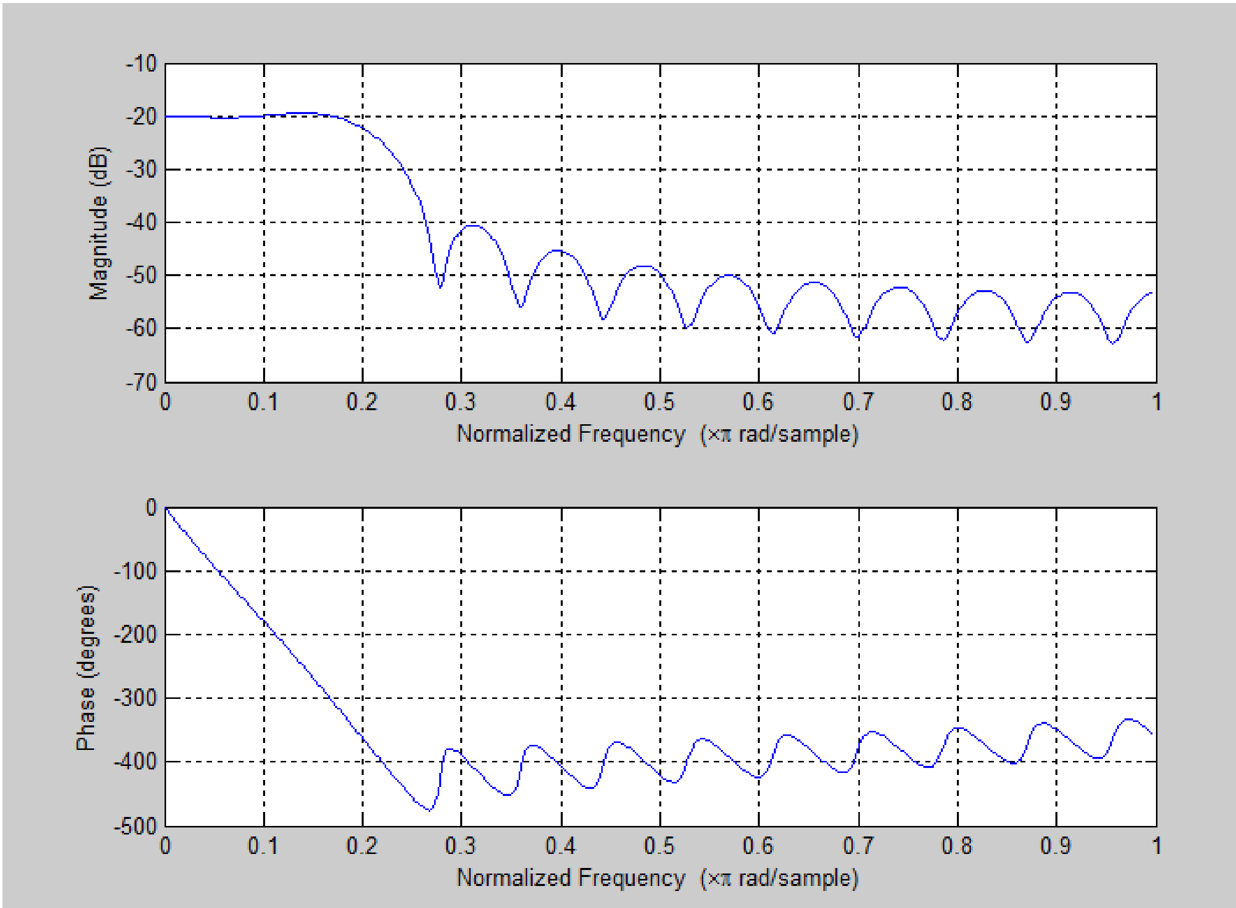
I. HAMMING LOW PASS FILTER

To be demonstrated:

For the given data below, construct a Hamming low pass filter and filter the input signal with frequencies as below:
Filter specifications:
Cut-off frequency – 450 Hz
Order of the filter – 23
Input signal:
First input frequency – 300 Hz
Second input frequency – 800 Hz



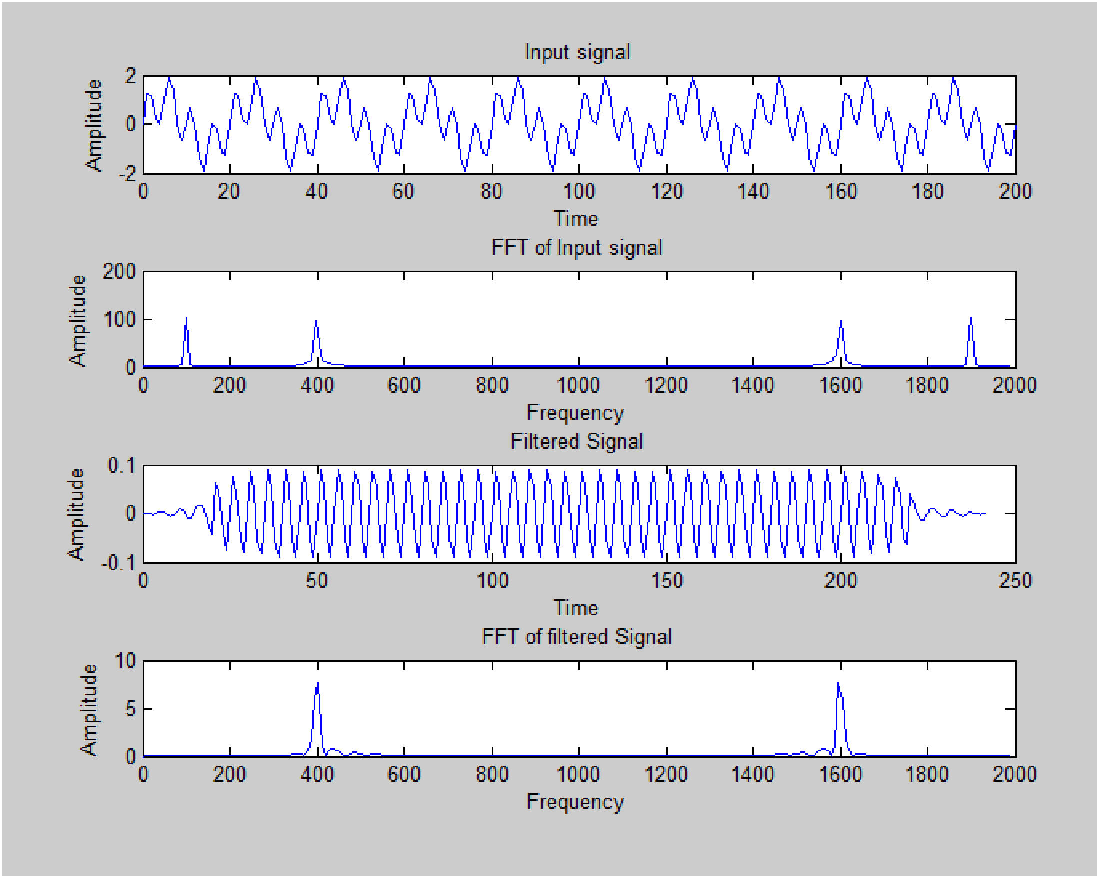

```
enter the order : 23
enter f1 : 300
enter f2 : 800
enter cut-off freq : 450
  Columns 1 through 6

    0.0022    0.0003   -0.0023   -0.0044   -0.0047   -0.0024

  Columns 7 through 12

    0.0024    0.0089    0.0155    0.0204    0.0222    0.0204

  Columns 13 through 18

    0.0155    0.0089    0.0024   -0.0024   -0.0047   -0.0044

  Columns 19 through 23

   -0.0023    0.0003    0.0022    0.0028    0.0021

fx >>
```

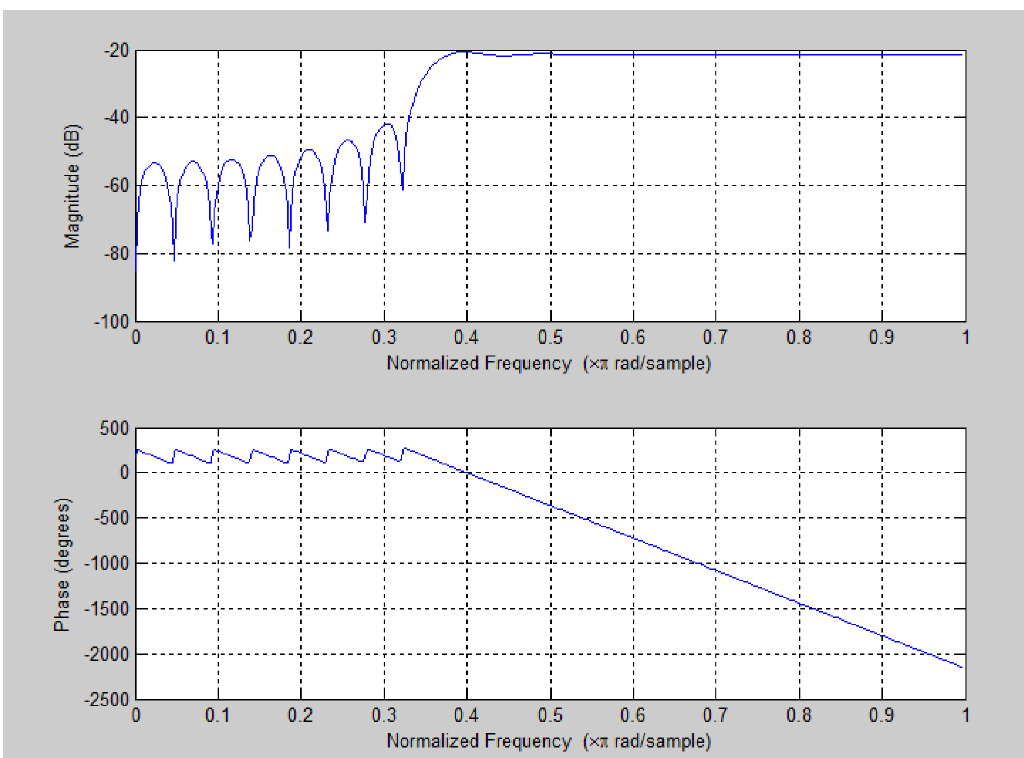


II. HAMMING HIGH PASS FILTER

To be demonstrated:

For the given data below, construct a Hamming high pass filter and filter the input signal with frequencies as below:
Filter specifications:
Cut-off frequency – 350 Hz
Order of the filter – 43
Input signal:
First input frequency – 100 Hz
Second input frequency – 400 Hz


Input signal
Amplitude
Time
FFT of Input signal
Amplitude
Frequency
Filtered Signal
Amplitude
Time
FFT of filtered Signal
Amplitude
Frequency

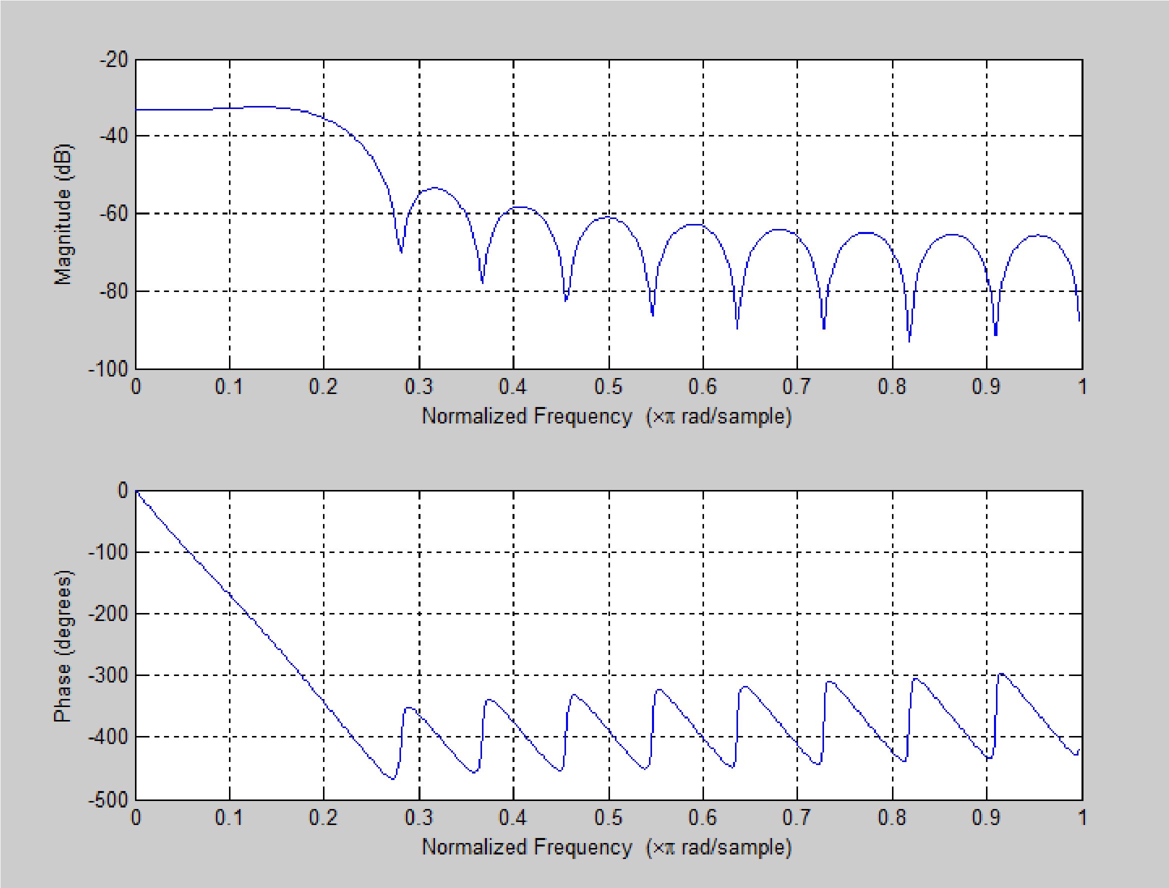

Magnitude (dB)
Normalized Frequency (×π rad/sample)
Phase (degrees)
Normalized Frequency (×π rad/sample)

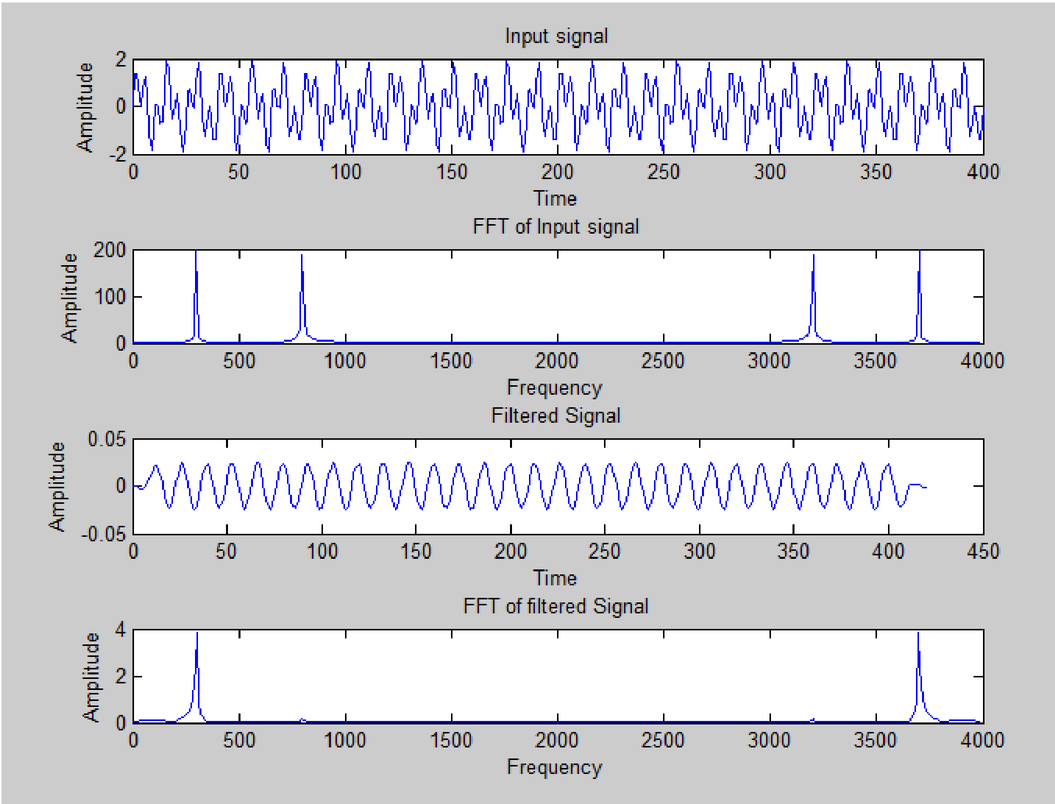
```
Command Window
h =

  Columns 1 through 6

   -0.0000   -0.0013   -0.0012    0.0002    0.0016    0.0013

  Columns 7 through 12

   -0.0006   -0.0021   -0.0013    0.0011    0.0027    0.0014

  Columns 13 through 18

   -0.0020   -0.0038   -0.0014    0.0038    0.0064    0.0014

  Columns 19 through 24

   -0.0110   -0.0241    0.0553   -0.0241   -0.0110    0.0014

  Columns 25 through 30

    0.0064    0.0038   -0.0014   -0.0038   -0.0020    0.0014

  Columns 31 through 36

    0.0027    0.0011   -0.0013   -0.0021   -0.0006    0.0013

  Columns 37 through 42

    0.0016    0.0002   -0.0012   -0.0013   -0.0000    0.0011

  Column 43

    0.0010
```

III. HANNING LOW PASS FILTER

To be demonstrated:

For the given data below, construct a Hanning low pass filter and filter the input signal with frequencies as below:
Filter specifications:
Cut-off frequency – 450 Hz
Order of the filter – 22
Input signal:
First input frequency – 300 Hz
Second input frequency – 800 Hz


Magnitude (dB)
Normalized Frequency (×π rad/sample)
Phase (degrees)
Normalized Frequency (×π rad/sample)

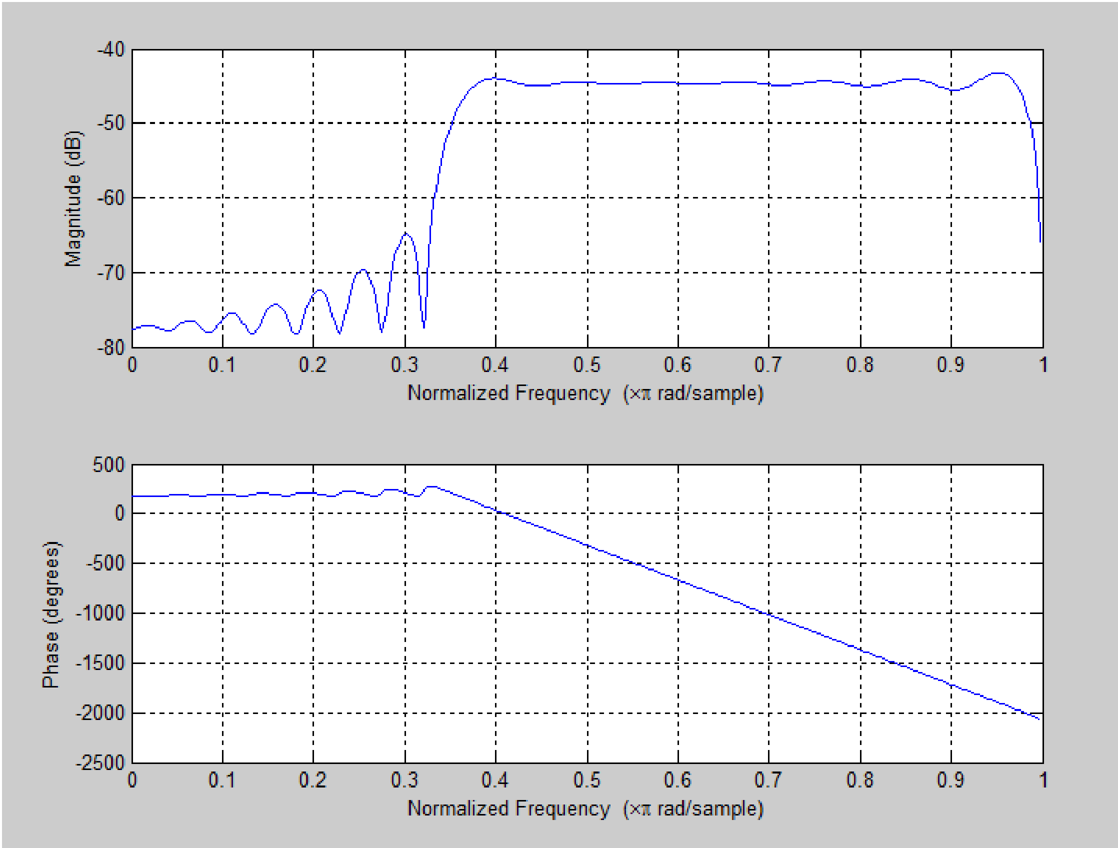

Input signal
Amplitude
Time
FFT of Input signal
Amplitude
Frequency
Filtered Signal
Amplitude
Time
FFT of filtered Signal
Amplitude
Frequency

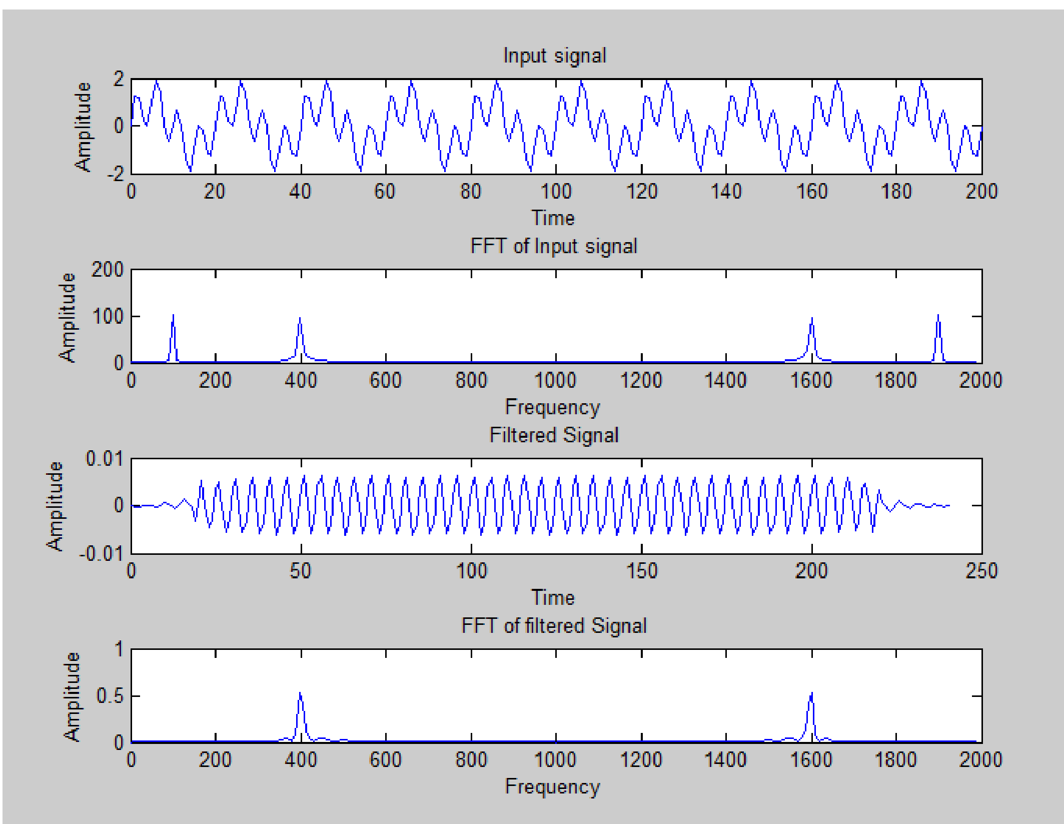
```
Command Window
enter the order : 22
enter f1 : 300
enter f2 : 800
enter cut-off freq : 450
  Columns 1 through 7

    0.0003   -0.0002   -0.0008   -0.0011   -0.0009   -0.0001    0.0013

  Columns 8 through 14

    0.0028    0.0041    0.0049    0.0049    0.0041    0.0028    0.0013

  Columns 15 through 21

   -0.0001   -0.0009   -0.0011   -0.0008   -0.0002    0.0003    0.0006

  Column 22

    0.0006

fx >>
```

IV. HANNING HIGH PASS FILTER

To be demonstrated:

For the given data below, construct a Hanning high pass filter and filter the input signal with frequencies as below:
Filter specifications:
Cut-off frequency – 350 Hz
Order of the filter – 42
Input signal:
First input frequency – 100 Hz
Second input frequency – 400 Hz


Magnitude (dB)
-40
-50
-60
-70
-80
0 0.1 0.2 0.3 0.4 0.5 0.6 0.7 0.8 0.9 1
Normalized Frequency (×π rad/sample)
Phase (degrees)
500
0
-500
-1000
-1500
-2000
-2500
0 0.1 0.2 0.3 0.4 0.5 0.6 0.7 0.8 0.9 1
Normalized Frequency (×π rad/sample)


Input signal
Amplitude
2
0
-2
0 20 40 60 80 100 120 140 160 180 200
Time
FFT of Input signal
Amplitude
200
100
0
0 200 400 600 800 1000 1200 1400 1600 1800 2000
Frequency
Filtered Signal
Amplitude
0.01
0
-0.01
0 50 100 150 200 250
Time
FFT of filtered Signal
Amplitude
1
0.5
0
0 200 400 600 800 1000 1200 1400 1600 1800 2000
Frequency

```
enter the order : 42
enter f1 : 100
enter f2 : 400
enter cut-off freq : 350

h =

  Columns 1 through 7

   -0.0001    0.0000   -0.0001    0.0002   -0.0000    0.0002   -0.0002

  Columns 8 through 14

    0.0000   -0.0002    0.0003   -0.0000    0.0002   -0.0005    0.0001

  Columns 15 through 21

   -0.0003    0.0008   -0.0002    0.0005   -0.0025    0.0018    0.0018

  Columns 22 through 28

   -0.0025    0.0005   -0.0002    0.0008   -0.0003    0.0001   -0.0005

  Columns 29 through 35

    0.0002   -0.0000    0.0003   -0.0002    0.0000   -0.0002    0.0002

  Columns 36 through 42

   -0.0000    0.0002   -0.0001    0.0000   -0.0001    0.0001   -0.0000

fx >>
```

Experiment - 7

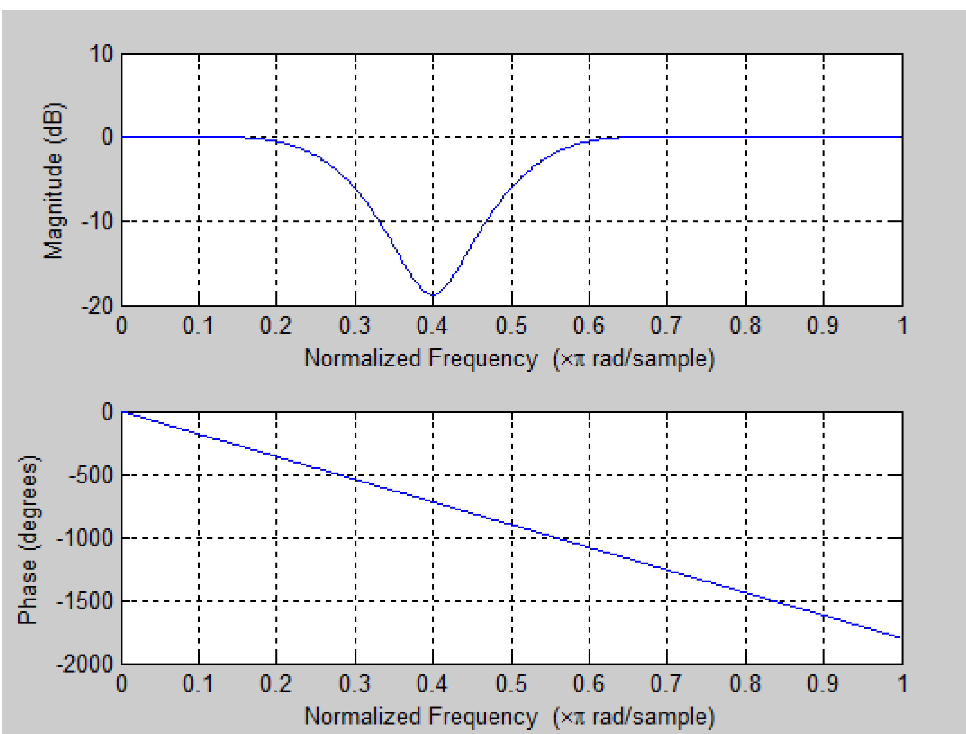
Write a MATLAB code to verify the Band pass and Band reject FIR linear phase filter design using Hamming and Hanning windows (with inbuilt and without using inbuilt commands). Plot the magnitude and phase response. Also, Provide the inference on the basis of results obtained for the set of specifications.

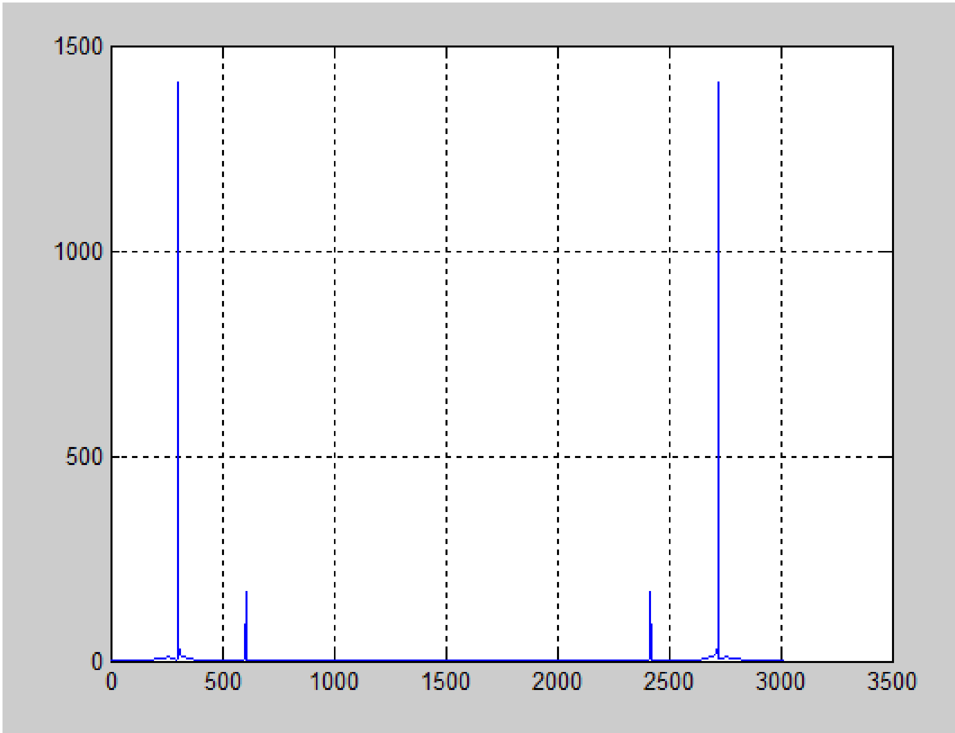
Outcome: The student will be able to design and implement a linear phase FIR filter that best matches the application and satisfies the design requirements.

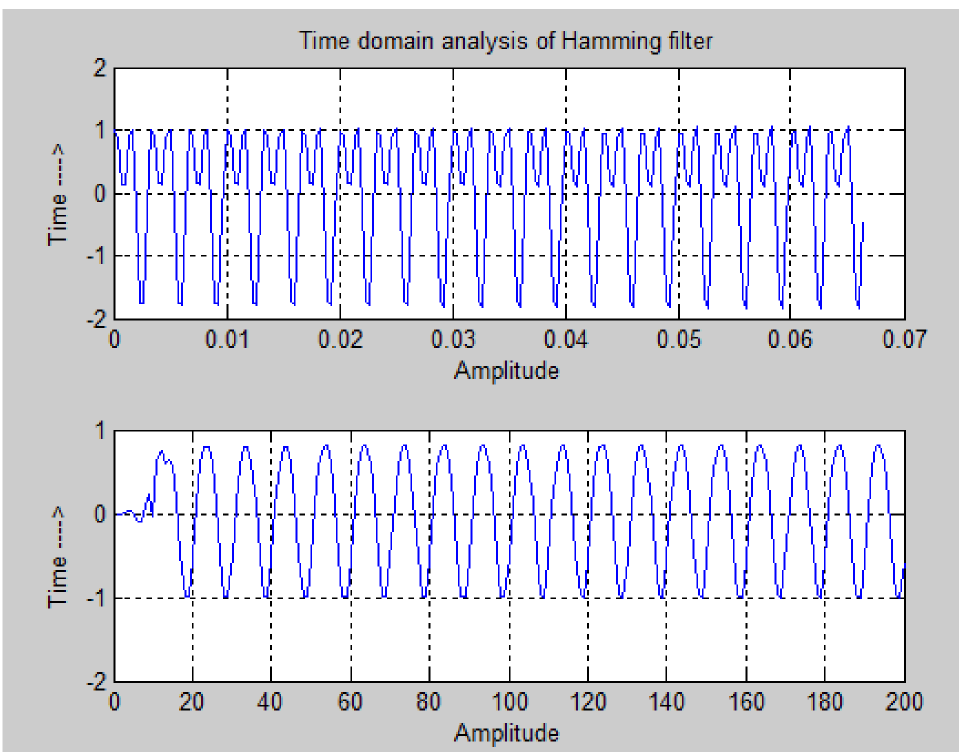
I. HAMMING BAND ELIMINATION FILTER

To be demonstrated:

For the given data below, construct a Hamming band elimination filter and filter the input signal with frequencies as below:
Filter specifications:
First cut-off frequency – 450 Hz
Second cut-off frequency – 750 Hz
Order of the filter – 23
Input signal:
First input frequency – 300 Hz
Second input frequency – 600 Hz




1500
1000
500
0
0
500
1000
1500
2000
2500
3000
3500


Time domain analysis of Hamming filter
Time ---->
Amplitude
0
0.01
0.02
0.03
0.04
0.05
0.06
0.07
2
1
0
-1
-2
Time ---->
Amplitude
0
20
40
60
80
100
120
140
160
180
200
1
0
-1
-2

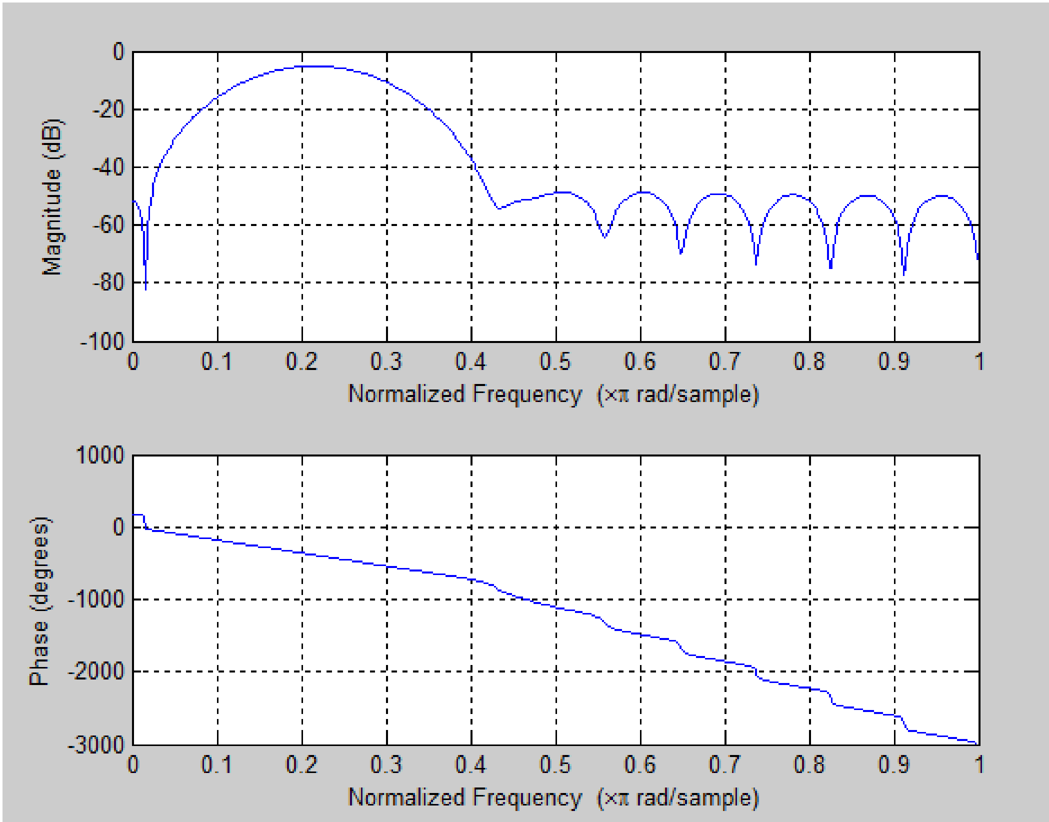
```
Command Window
Enter the 1st input frequency :300
Enter the 2nd input frequency :600
Enter the cutoff first frequency :450
Enter the cutoff second frequency :750
Enter the Order of the filter :23
  Columns 1 through 7

   -0.0000   -0.0010    0.0091    0.0207   -0.0150   -0.0771   -0.0339

  Columns 8 through 14

    0.1170    0.1401   -0.0598    0.8000   -0.0598    0.1403    0.1171

  Columns 15 through 21

   -0.0340   -0.0773   -0.0150    0.0208    0.0091   -0.0010   -0.0000

  Columns 22 through 23

    0.0004   -0.0025
```

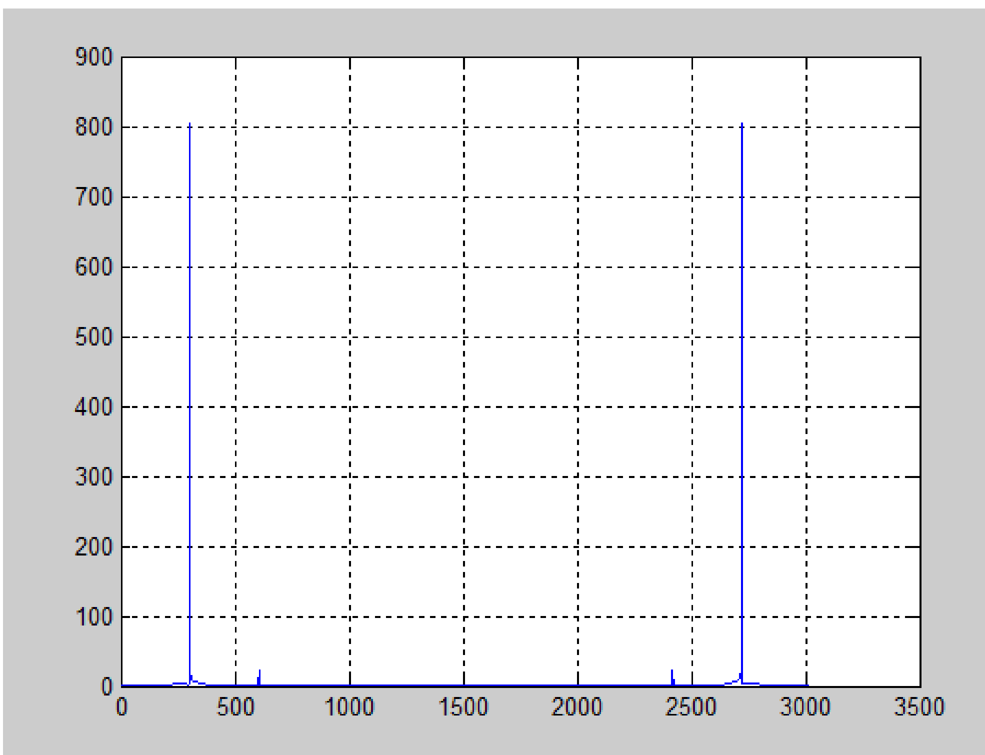
II. HAMMING BAND PASS FILTER

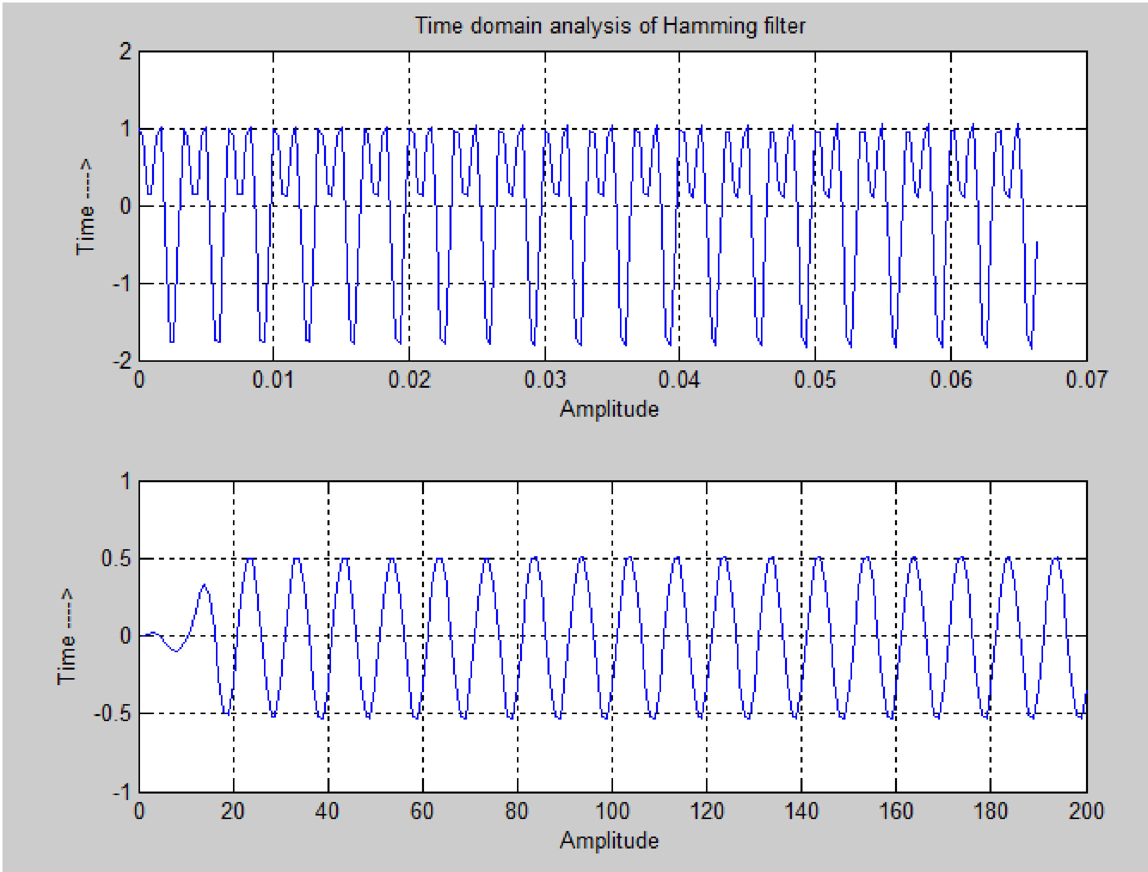
To be demonstrated:

For the given data below, construct a Hamming band pass filter and filter the input signal with frequencies as below:

Filter specifications:
First cut-off frequency – 250 Hz
Second cut-off frequency – 400 Hz
Order of the filter – 23
Input signal:
First input frequency – 300 Hz
Second input frequency – 600 Hz


Magnitude (dB)
0
-20
-40
-60
-80
-100
0
0.1
0.2
0.3
0.4
0.5
0.6
0.7
0.8
0.9
1
Normalized Frequency (×π rad/sample)
Phase (degrees)
1000
0
-1000
-2000
-3000
Normalized Frequency (×π rad/sample)


900
800
700
600
500
400
300
200
100
0
0
500
1000
1500
2000
2500
3000
3500



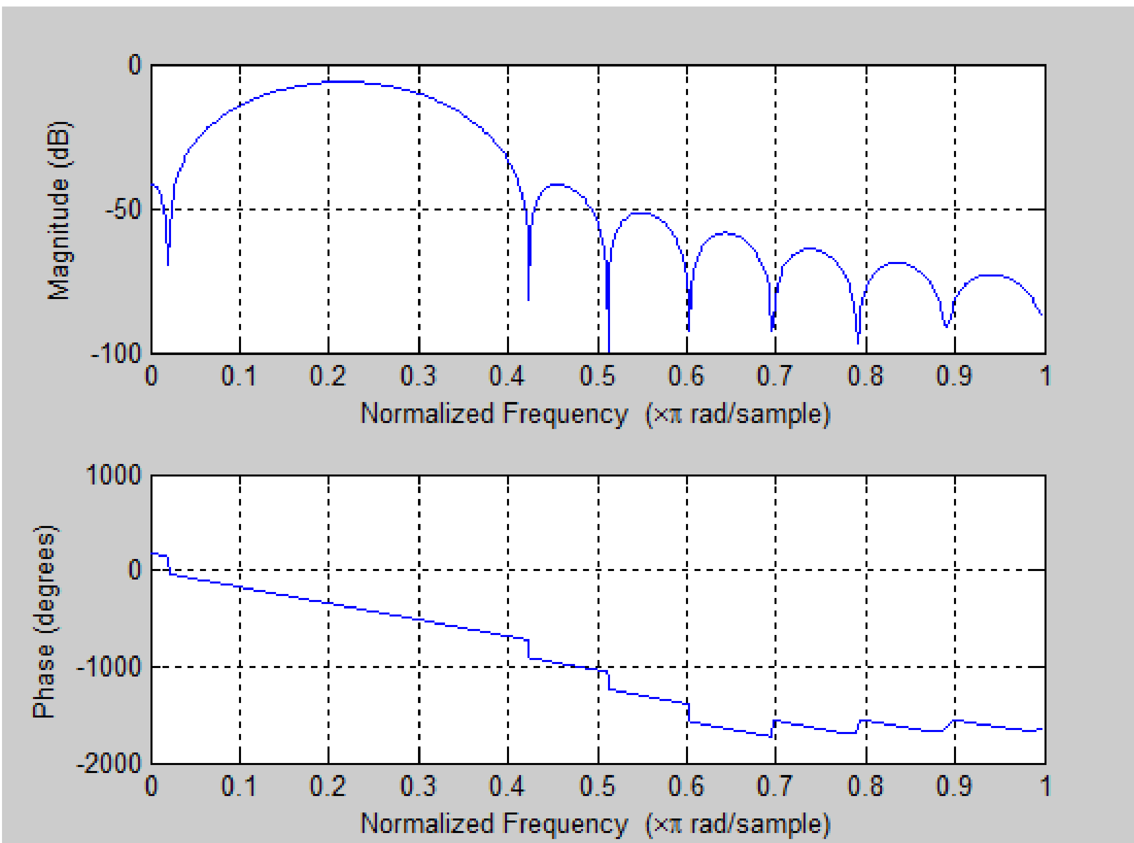

```
Command Window
Enter the 1st input frequency :300
Enter the 2nd input frequency :600
Enter the cutoff first frequency :250
Enter the cutoff second frequency :400
Enter the Order of the filter :23
  Columns 1 through 7

    0.0055    0.0106    0.0121    0.0014   -0.0240   -0.0526   -0.0624

  Columns 8 through 14

   -0.0367    0.0190    0.0760    0.1000    0.0760    0.0190   -0.0368

  Columns 15 through 21

   -0.0625   -0.0528   -0.0241    0.0014    0.0121    0.0106    0.0055

  Columns 22 through 23

    0.0017   -0.0015

fx >>
```

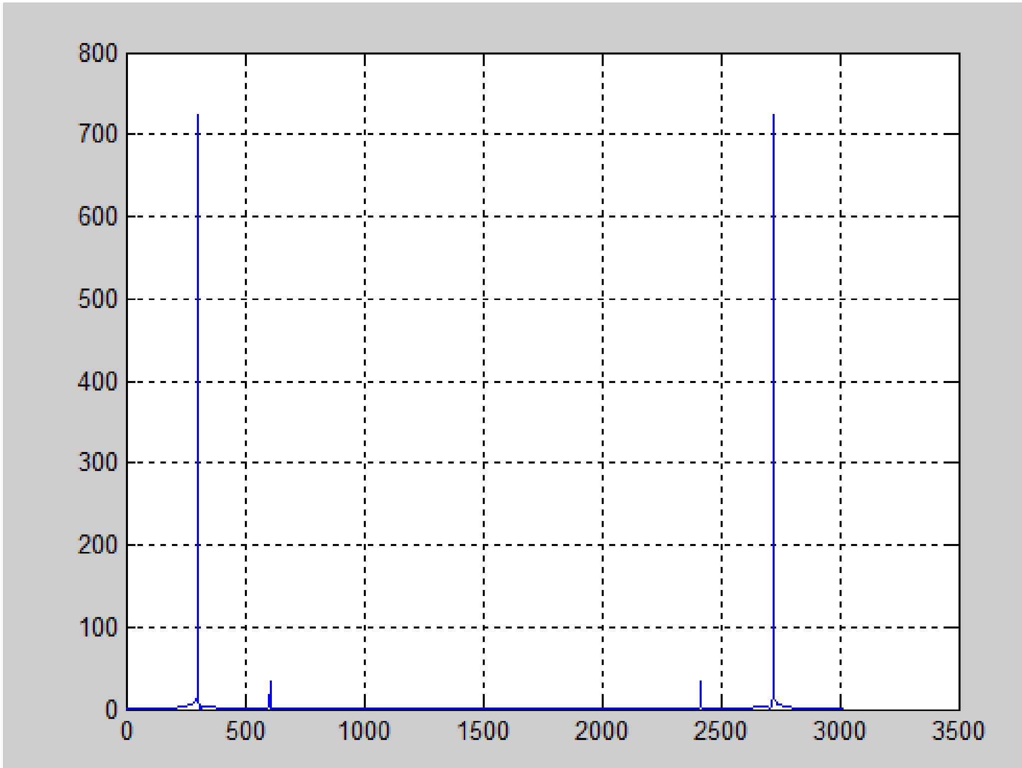
III. HANNING BAND PASS FILTER

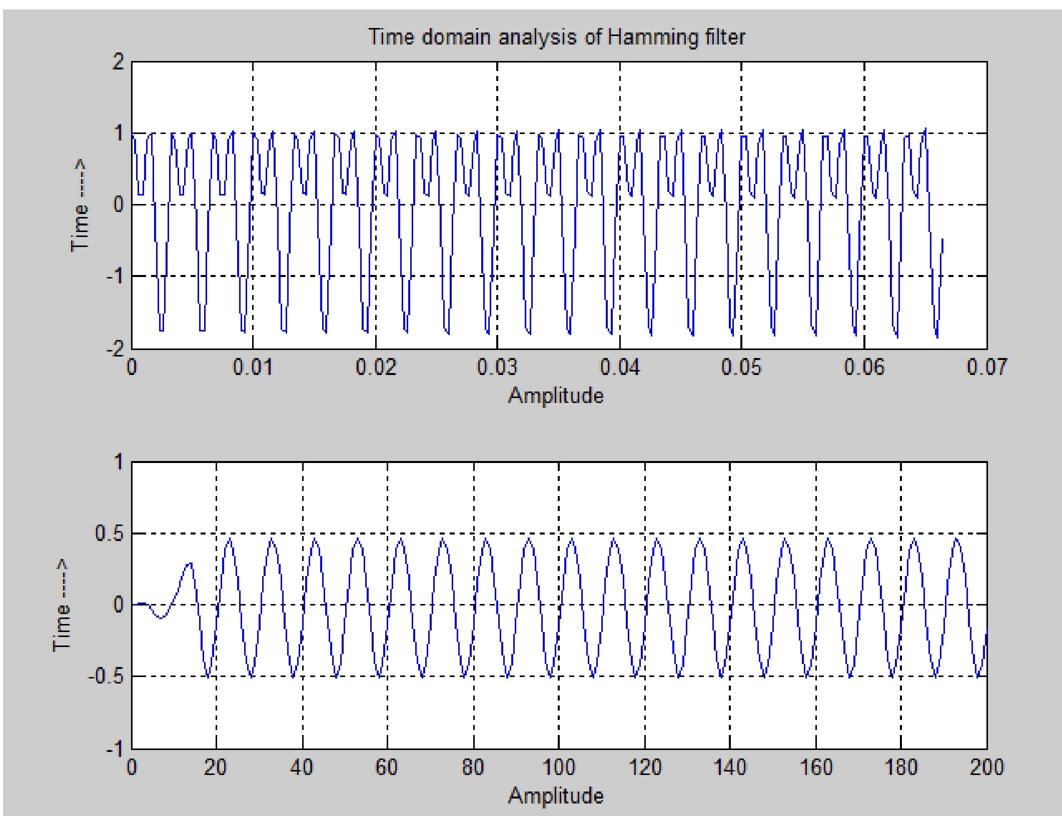
To be demonstrated:

For the given data below, construct a Hanning band pass filter and filter the input signal with frequencies as below:

Filter specifications:
First cut-off frequency – 250 Hz
Second cut-off frequency – 400 Hz
Order of the filter – 22
Input signal:
First input frequency – 300 Hz
Second input frequency – 600 Hz



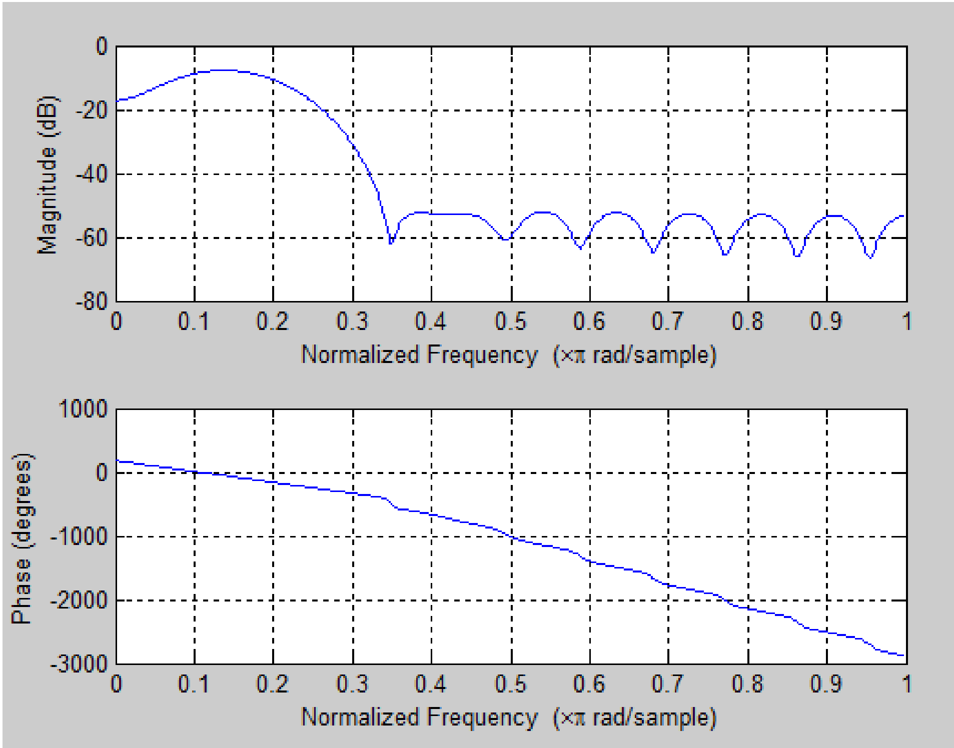

800
700
600
500
400
300
200
100
0
0
500
1000
1500
2000
2500
3000
3500


Time domain analysis of Hamming filter
Time ---->
Amplitude
2
1
0
-1
-2
0
0.01
0.02
0.03
0.04
0.05
0.06
0.07
Time ---->
Amplitude
1
0.5
0
-0.5
-1
0
20
40
60
80
100
120
140
160
180
200

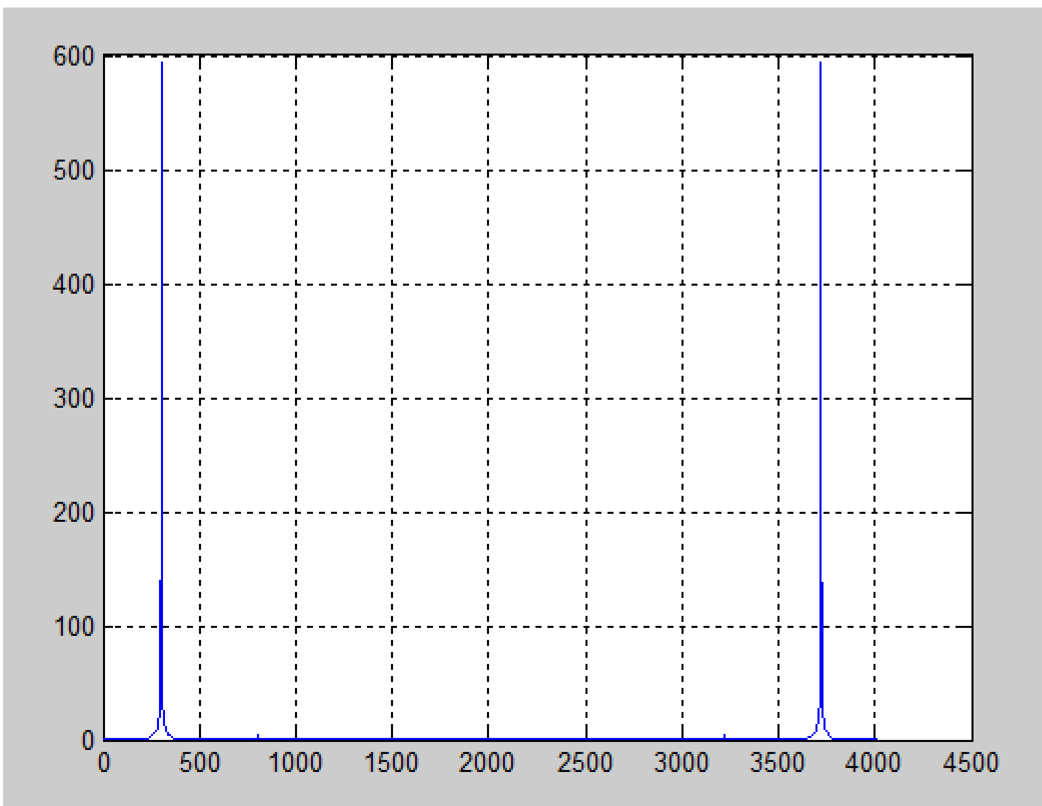
```
Command Window
Enter the 1st input frequency :300
Enter the 2nd input frequency :600
Enter the cutoff first frequency :250
Enter the cutoff second frequency :400
Enter the Order of the filter :22
  Columns 1 through 7

    0.0015    0.0055    0.0056   -0.0076   -0.0336   -0.0559   -0.0516

  Columns 8 through 14

   -0.0109    0.0492    0.0936    0.0937    0.0493   -0.0110   -0.0517

  Columns 15 through 21

   -0.0561   -0.0337   -0.0076    0.0056    0.0056    0.0015    0.0000

  Column 22

    0.0000
```

IV. HANNING BAND ELIMINATION FILTER

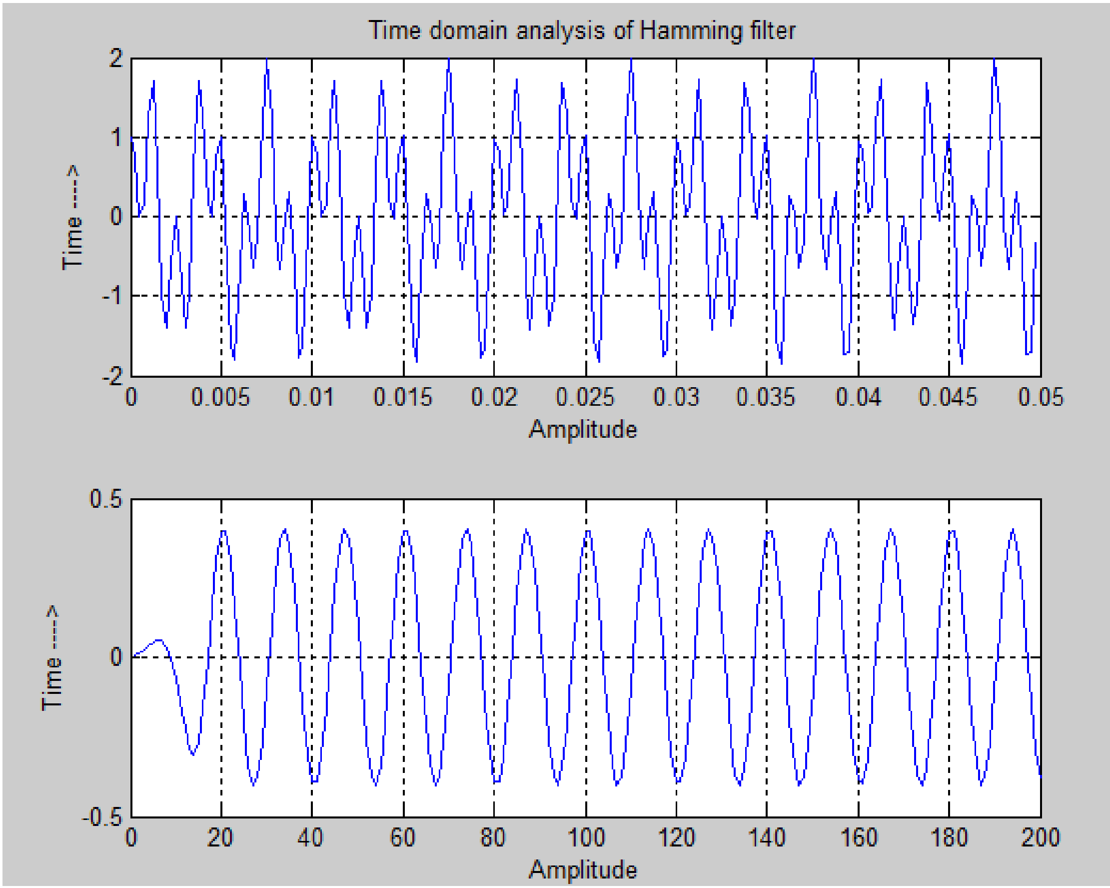
To be demonstrated: For the given data below, construct a Hanning band elimination filter and filter the input signal with frequencies as below:
Filter specifications:
First cut-off frequency – 200 Hz
Second cut-off frequency – 350 Hz
Order of the filter – 22
Input signal:
First input frequency – 300 Hz
Second input frequency – 800 Hz



Magnitude (dB)
0
-20
-40
-60
-80
0 0.1 0.2 0.3 0.4 0.5 0.6 0.7 0.8 0.9 1
Normalized Frequency (×π rad/sample)
Phase (degrees)
1000
0
-1000
-2000
-3000
0 0.1 0.2 0.3 0.4 0.5 0.6 0.7 0.8 0.9 1
Normalized Frequency (×π rad/sample)



600
500
400
300
200
100
0
0 500 1000 1500 2000 2500 3000 3500 4000 4500



Command Window

```
Enter the 1st input frequency :300
Enter the 2nd input frequency :800
Enter the cutoff first frequency :200
Enter the cutoff second frequency :350
Enter the Order of the filter :22
  Columns 1 through 7

    0.0035    0.0087    0.0165    0.0238    0.0254    0.0167   -0.0033

  Columns 8 through 14

   -0.0306   -0.0568   -0.0728   -0.0728   -0.0568   -0.0306   -0.0034

  Columns 15 through 21

    0.0167    0.0255    0.0239    0.0166    0.0088    0.0035    0.0008

  Column 22

   -0.0014
```

Experiment - 8

Write a MATLAB code to verify the Low pass Butterworth IIR filter design using bilinear transformation (BLT) method and Impulse Invariant Technique (IIT) method.

Outcome: The student will be able to design and implement an IIR filter that best matches the application and satisfies the design requirements.

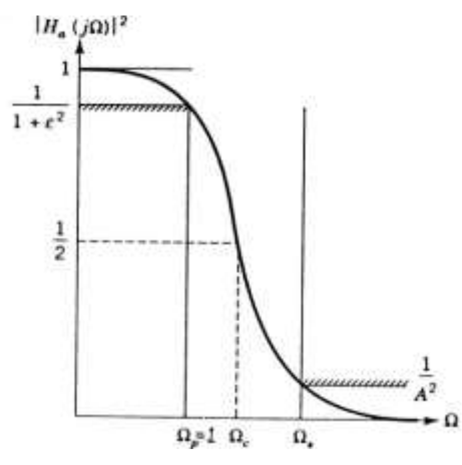
In the impulse invariance technique, the frequency response of analog and digital filters are similar only at low frequencies. This limits the application of impulse invariance method to only low pass filters. A better transformation that preserves frequency response and overcomes the limitations of impulse invariance method is called bilinear transformation method. The warping effect in impulse invariance between the analog and digital frequencies is eliminated by pre-warping the analog frequencies.

BUTTERWORTH FILTERS

- For the Nth-order Butterworth filter the squared-magnitude function is given by

$$|H_a(j\Omega)|^2 = \frac{1}{1+\epsilon^2\Omega^{2N}}.$$

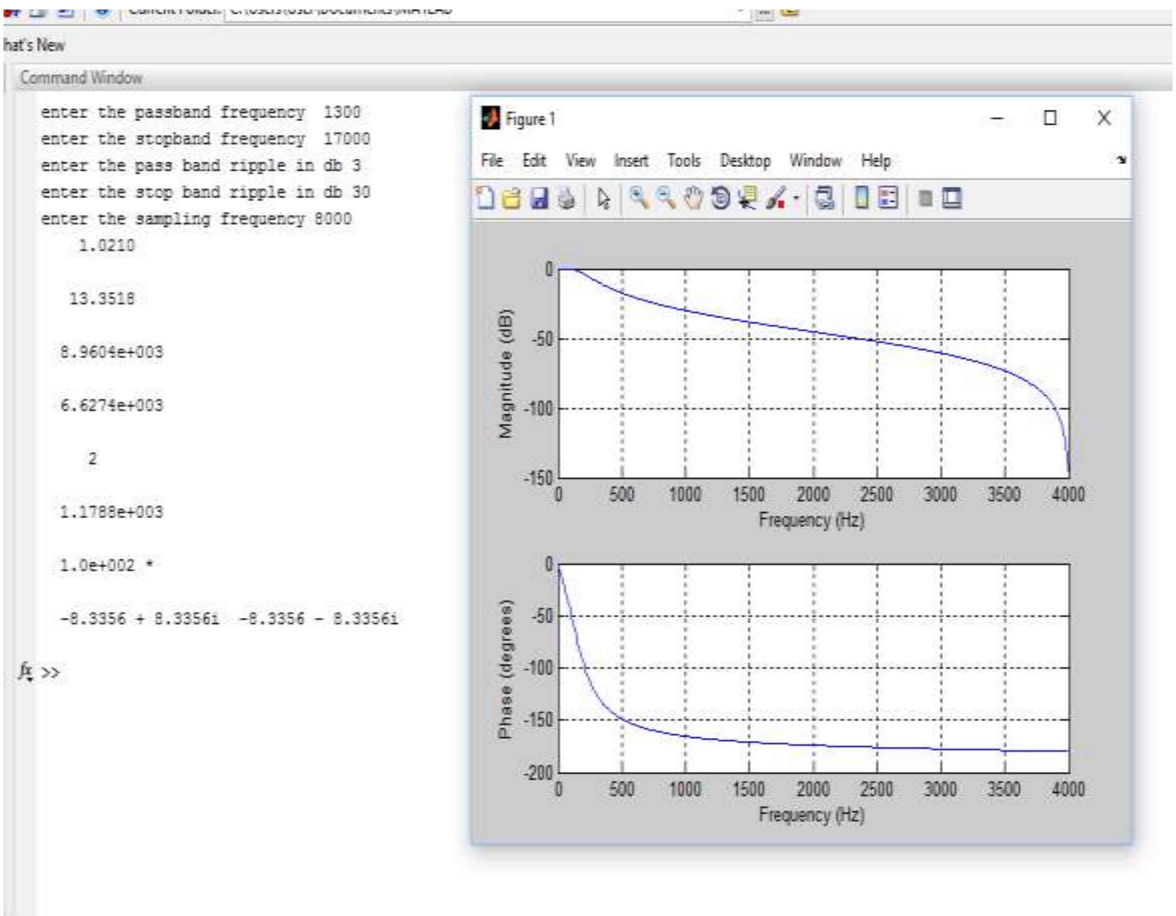
- This function achieves the value of unity at $\Omega = 0$ with the first $2N-1$ derivatives being zero at this point (**maximally flat passband**).
- At infinity, the value is zero and the first $2N-1$ derivatives are zero (**maximally flat stopband**).
  - The following figure gives the normalized specifications, where $\Omega_p = 1$, $|H_a(j\Omega_p)|^2 = 1/(1+\epsilon^2)$, and it is required that $|H_a(j\Omega_s)|^2 \leq 1/A^2$.



I. LOW PASS BUTTERWORTH IIR FILTER DESIGN USING BILINEAR TRANSFORMATION TECHNIQUE (BLT) METHOD

To be demonstrated: For the given data below, construct a Low pass Butterworth IIR filter design using Bilinear Transformation (BLT) method and filter the input signal with filter specifications as below:

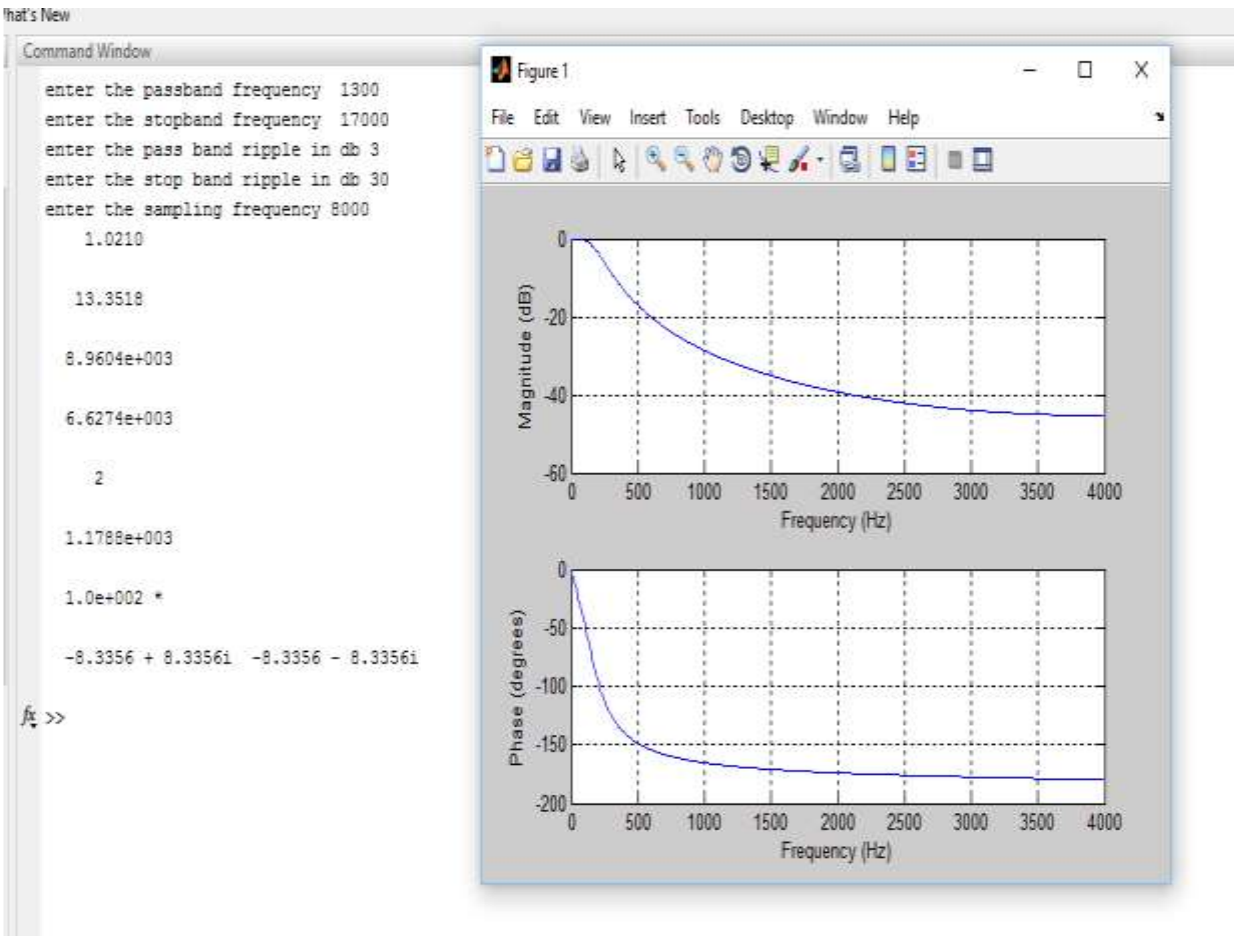
Filter specifications:
Pass band edge frequency – 1300 Hz
Stop band edge frequency – 17000 Hz
Pass band attenuation in db - 3
Stop band attenuation in db – 30
Sampling frequency – 8000 Hz



II. LOW PASS BUTTERWORTH IIR FILTER DESIGN USING IMPULSE INVARIANT TECHNIQUE (IIT) METHOD

To be demonstrated: For the given data below, construct a Low pass Butterworth IIR filter design using Impulse Invariant Technique (IIT) method and filter the input signal with filter specifications as below:

Filter specifications:
Pass band edge frequency – 1300 Hz
Stop band edge frequency – 17000 Hz
Pass band attenuation in db - 3
Stop band attenuation in db – 30
Sampling frequency – 8000 Hz



Experiment - 9

Write a MATLAB code to implement the Low pass Chebyshev (Type 1) IIR filter designusing bilinear transformation (BLT) method and Impulse Invariant Technique (IIT) method.

CHEBYSHEV FILTERS OR CHEBYSHEV TYPE I FILTERS

- The squared-magnitude function of this filter of order N is given by

$$|H_a(j\Omega)|^2 = \frac{1}{1+\epsilon^2 T_N^2(\Omega)}, \qquad (16a)$$

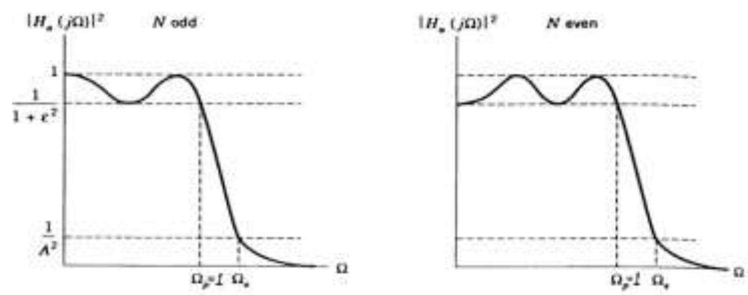
where

$$T_N(\Omega) = \begin{cases} \cos(N\cos^{-1}\Omega), & |\Omega| \le 1 \\ \cosh(N\cosh^{-1}\Omega) & |\Omega| > 1 \end{cases} \qquad (16b)$$

is the Nth-degree Chebyshev polynomial.

- In the normalized passband $0 \le \Omega \le \Omega_p = 1$, this function alternatingly achieves the values of 1 and $1/(1+\epsilon^2)$ at $N+1$ points such that $|H_a(j\Omega_p)|^2 = 1/(1+\epsilon^2)$. For N even, $|H_a(j0)|^2 = 1/(1+\epsilon^2)$ and for N odd, $|H_a(j0)|^2 = 1$ (**equiripple passband**).

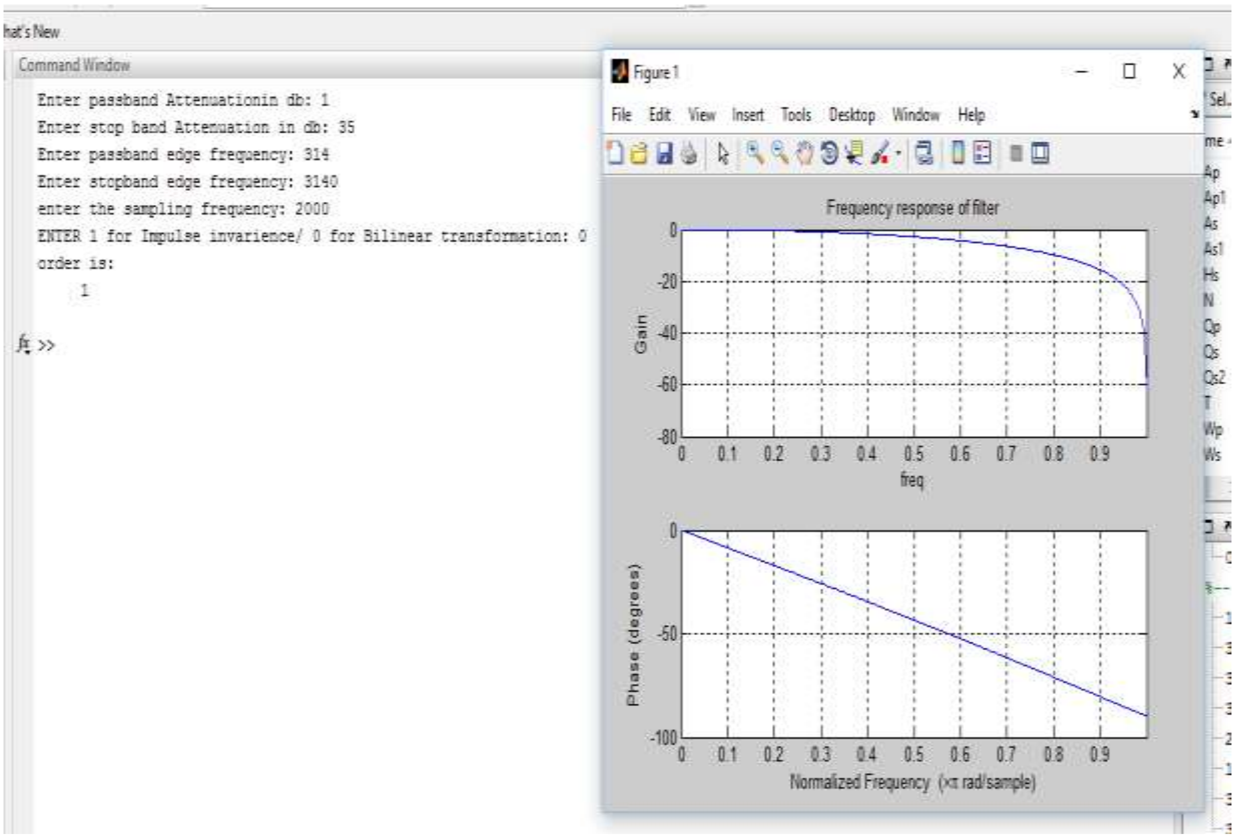
- At infinity, the value of $|H_a(j\Omega)|^2$ is zero and the first $2N-1$ derivatives are zero (**maximally flat stopband**) (see the figure shown below).



I. LOW PASS CHEBYSHEV IIR FILTER DESIGN USING BILINEAR TRANSFORMATION (BLT) METHOD

To be demonstrated: For the given data below, construct a Low pass Butterworth IIR filter design using bilinear transformation (BLT) method and filter the input signal with filter specifications as below:

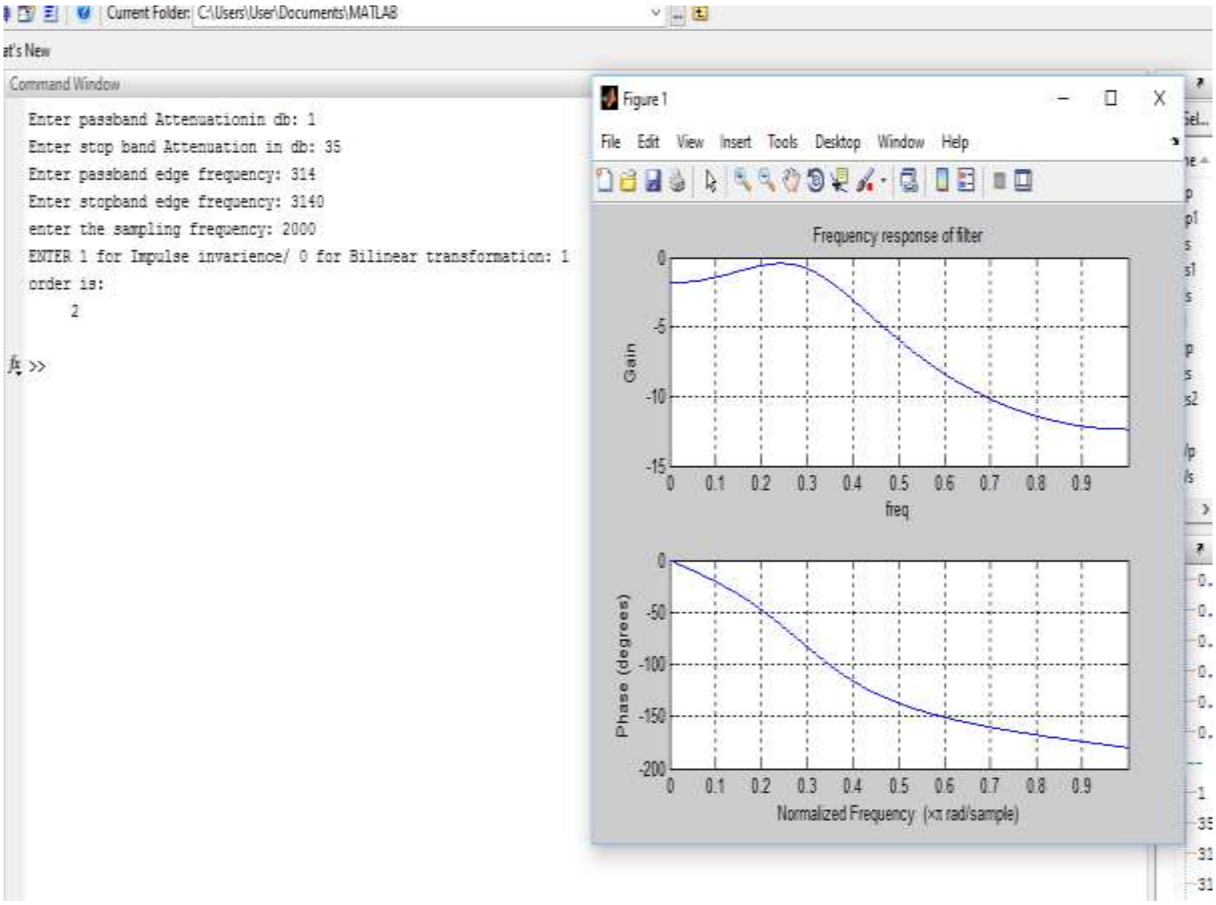
Filter specifications:
Pass band edge frequency – 314 Hz
Stop band edge frequency – 3140 Hz
Pass band attenuation in db - 1
Stop band attenuation in db – 35
Sampling frequency – 2000 Hz



II. LOW PASS CHEBYSHEV IIR FILTER DESIGN USING IMPULSE INVARIANT TECHNIQUE (IIT) METHOD

To be demonstrated: For the given data below, construct a Low pass Butterworth IIR filter design using Impulse Invariant Technique (IIT) method and filter the input signal with filter specifications as below:

Filter specifications:
Pass band edge frequency – 314 Hz
Stop band edge frequency – 3140 Hz
Pass band attenuation in db - 1
Stop band attenuation in db – 35
Sampling frequency – 2000 Hz


Current Folder: C:\Users\User\Documents\MATLAB
Command Window
Enter passband Attenuationin db: 1
Enter stop band Attenuation in db: 35
Enter passband edge frequency: 314
Enter stopband edge frequency: 3140
enter the sampling frequency: 2000
ENTER 1 for Impulse invarience/ 0 for Bilinear transformation: 1
order is:
2
Figure 1
File Edit View Insert Tools Desktop Window Help
Frequency response of filter
Gain
freq
Phase (degrees)
Normalized Frequency (×π rad/sample)

Experiment - 10

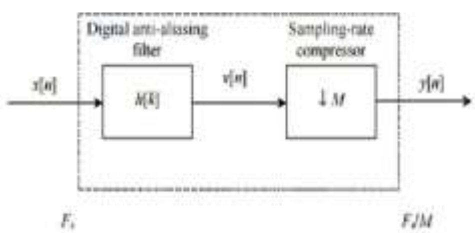
Write a MATLAB code to illustrate the effect of Decimation and Interpolation by an integer factor. Plot the magnitude spectrum. Design the necessary filter to overcome aliasing and image frequencies after decimating and interpolating the signal respectively.

Outcome: The student will be able to implement fundamental operations in multirate DSP.

I. DECIMATION

Decimation can be regarded as the discrete-time counterpart of sampling. Whereas in sampling we start with a continuous-time signal x(t) and convert it into a sequence of samples x[n], in decimation we start with a discrete-time signal x[n] and convert it into another discrete-time signal y[n], which consists of sub-samples of x[n]. Thus, the formal definition of M-fold decimation, or down-sampling, is defined by Equation 9.1. In decimation, the sampling rate is reduced from Fs to Fs/M by discarding M – 1 samples for every M samples in the original sequence.

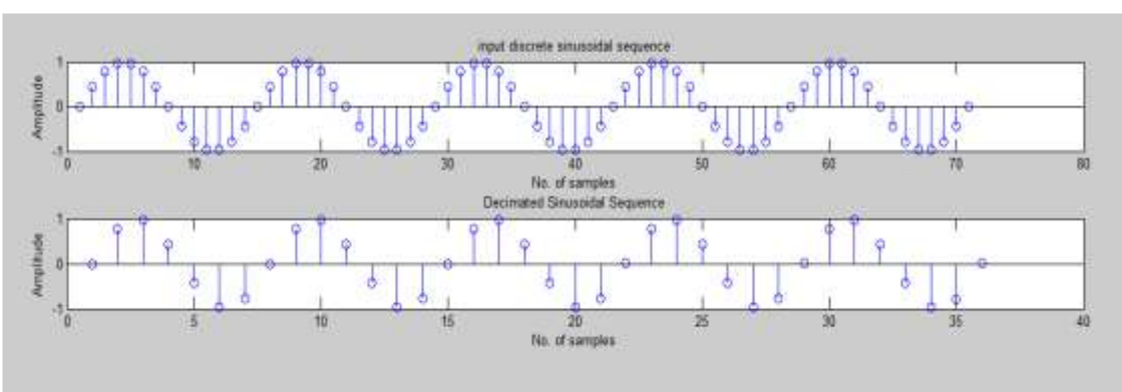
$$y[n] = v[nM] = \sum_{k=-\infty}^{\infty} h[k]x[nM-k]$$



Block diagram notation of decimation, by a factor of M.

To be demonstrated: Generate a sinusoidal signal of specifications given below and decimate it by the specified factor.

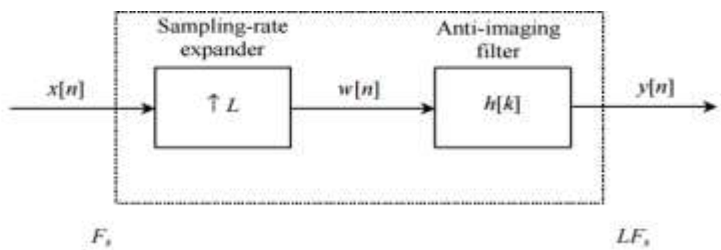
Signal specifications:
Input frequency – 10 Hz
Sampling frequency – 140 Hz
Decimation factor – 2



I. INTERPOLATION

Interpolation is the exact opposite of decimation. It is an information preserving operation, in that all samples of x[n] are present in the expanded signal y[n]. Interpolation works by inserting (L–1) zero-valued samples for each input sample. The sampling rate therefore increases from Fs to LFs. The expansion process is followed by a unique digital low-pass filter called an anti-imaging filter. Although the expansion process does not cause aliasing in the interpolated signal, it does however yield undesirable replicas in the signal's frequency spectrum.

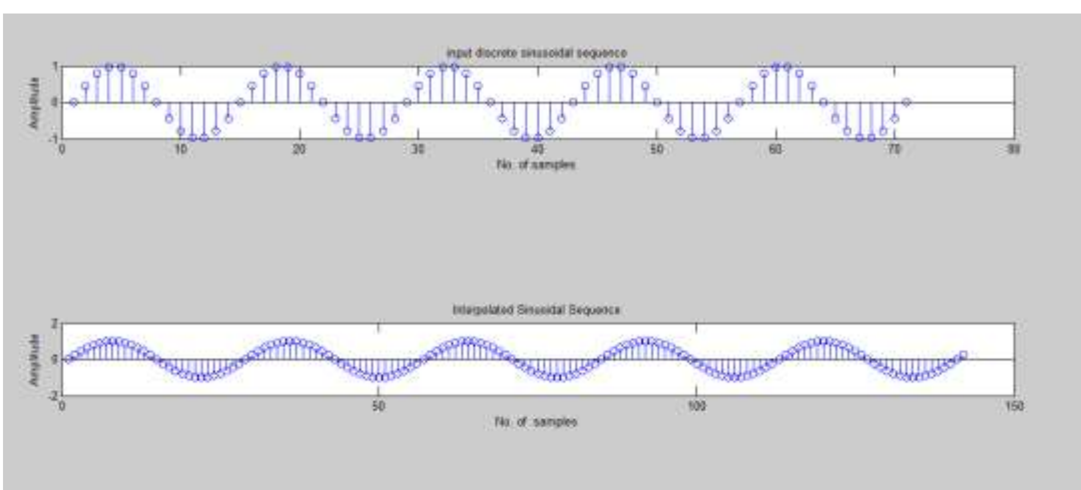
$$y[n] = L \sum_{k=-\infty}^{\infty} h[k]w[n-k]$$



Block diagram notation of interpolation, by a factor of L.

To be demonstrated: Generate a sinusoidal signal of specifications given below and decimate it by the specified factor.

Signal specifications:
Input frequency – 10 Hz
Sampling frequency – 140 Hz
Interpolation factor – 2

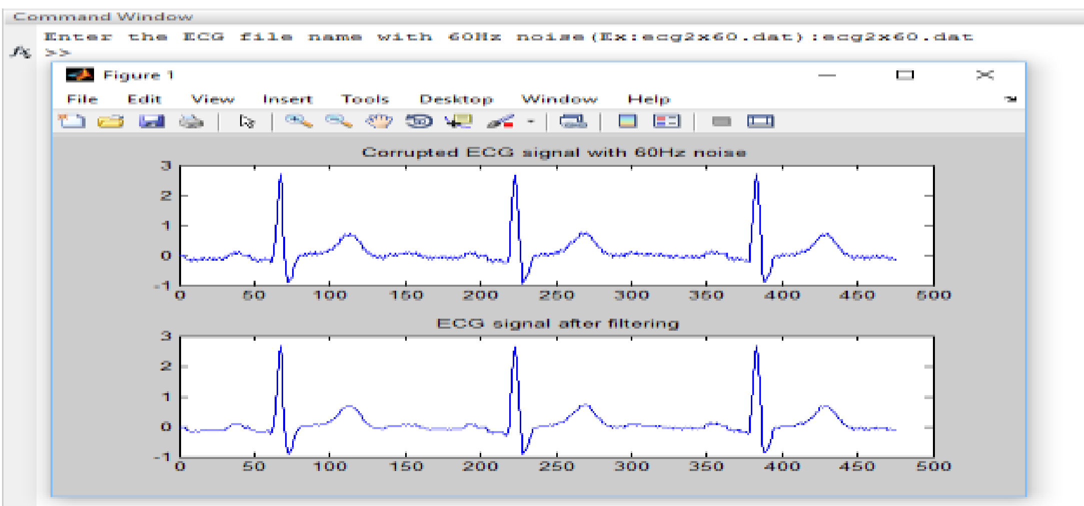


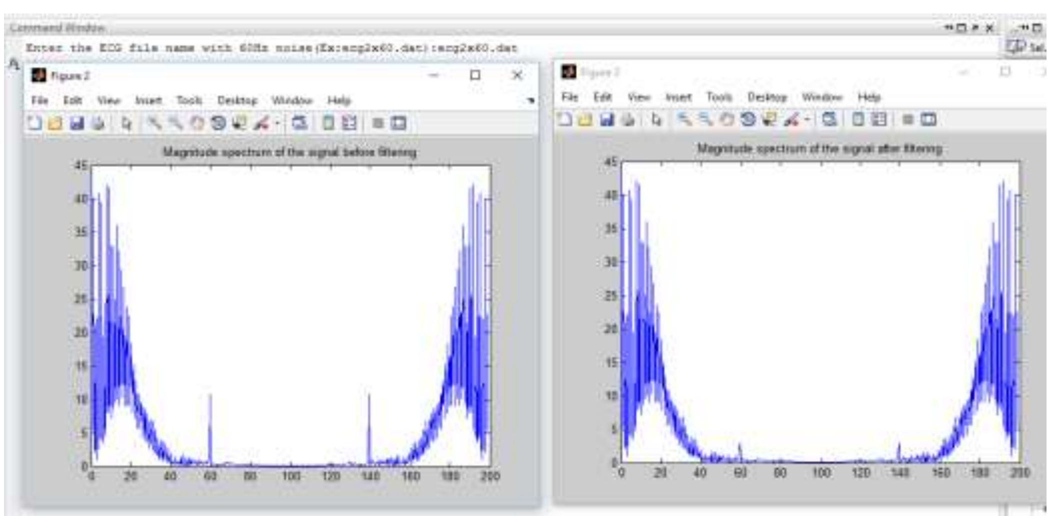
Experiment - 11

Read the data file named ecg2x60.dat from http://people.ucalgary.ca/~ranga/enel563/SIGNAL_DATA_FILES/) that is corrupted with the 60Hz noise component. Write a MATLAB code to remove this 60Hz noise component from the signal using Notch filter and LMS adaptive filter. Plot the magnitude spectrum of the signal filtered using both Notch filter and LMS adaptive filter and provide the inference on the basis of results obtained.

Outcome: The student will be able to apply adaptive filtering concept to remove the noise from the signal, whose statistics are not known in prior.

Least mean squares (**LMS**) algorithms are a class of adaptive filter used to mimic a desired filter by finding the filter coefficients that relate to producing the least mean square of the error signal (difference between the desired and the actual signal). It is a stochastic gradient descent method in that the filter is only adapted based on the error at the current time. It was invented in 1960 by Stanford University professor Bernard Widrow and his first Ph.D. student, Ted Hoff.

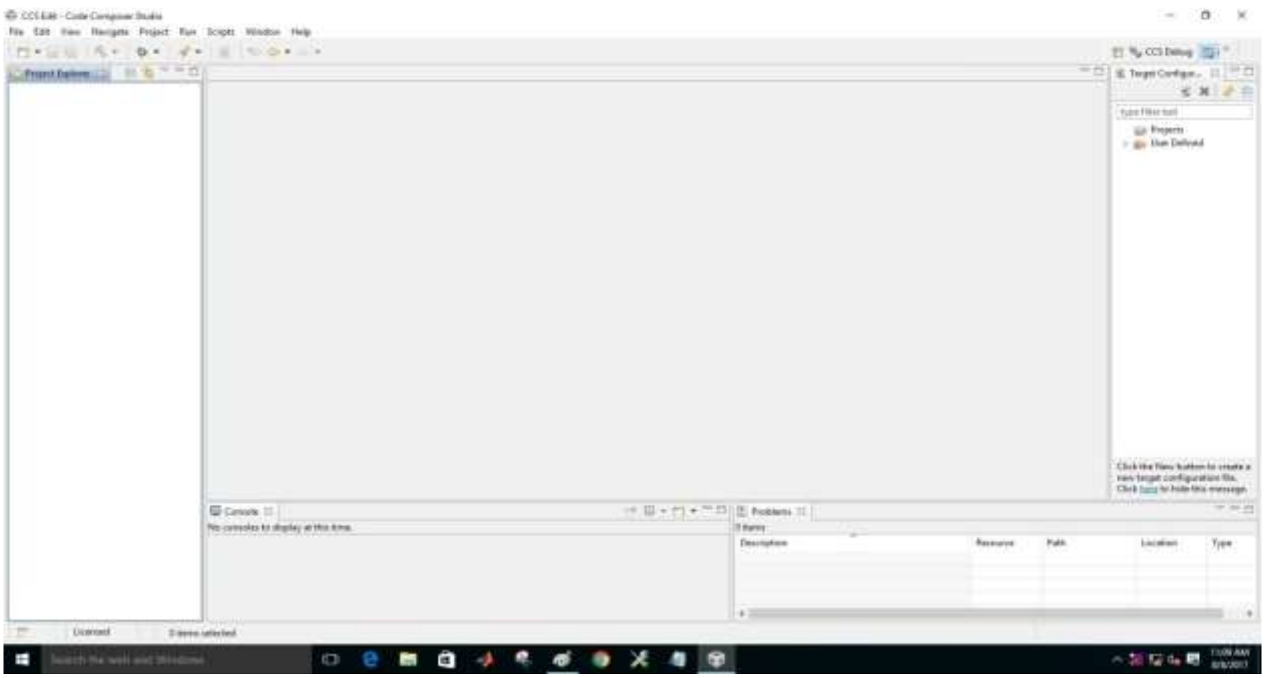




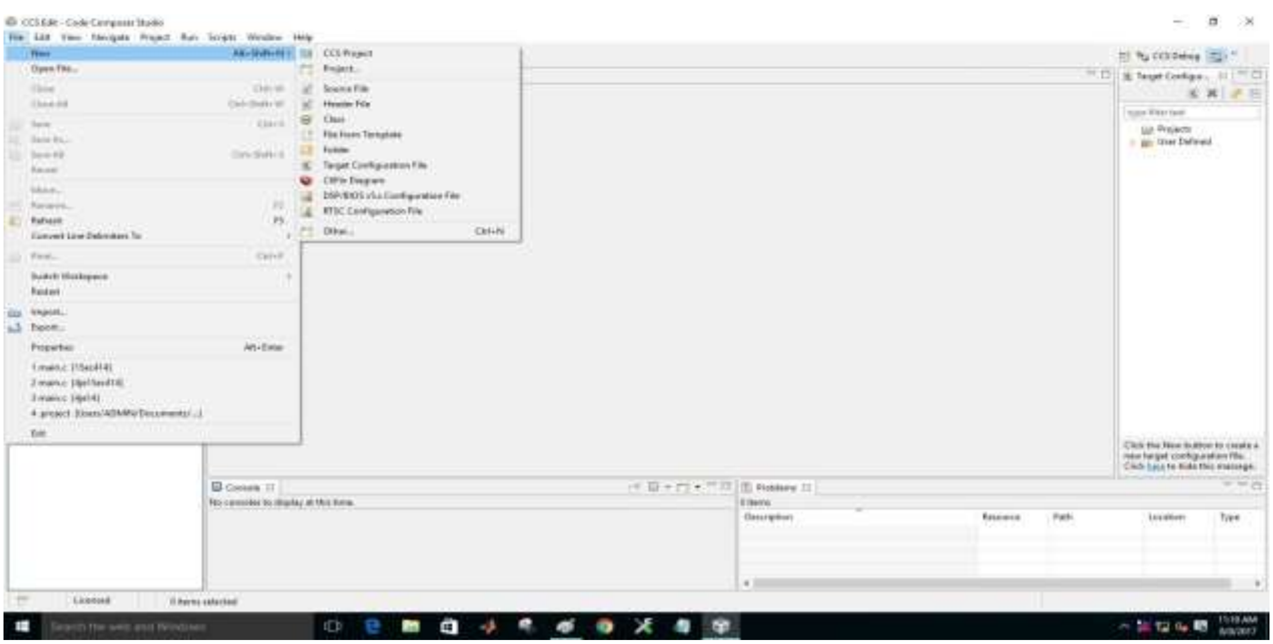
Experiment - 12

Procedure to use CCS Studio Version 5.0

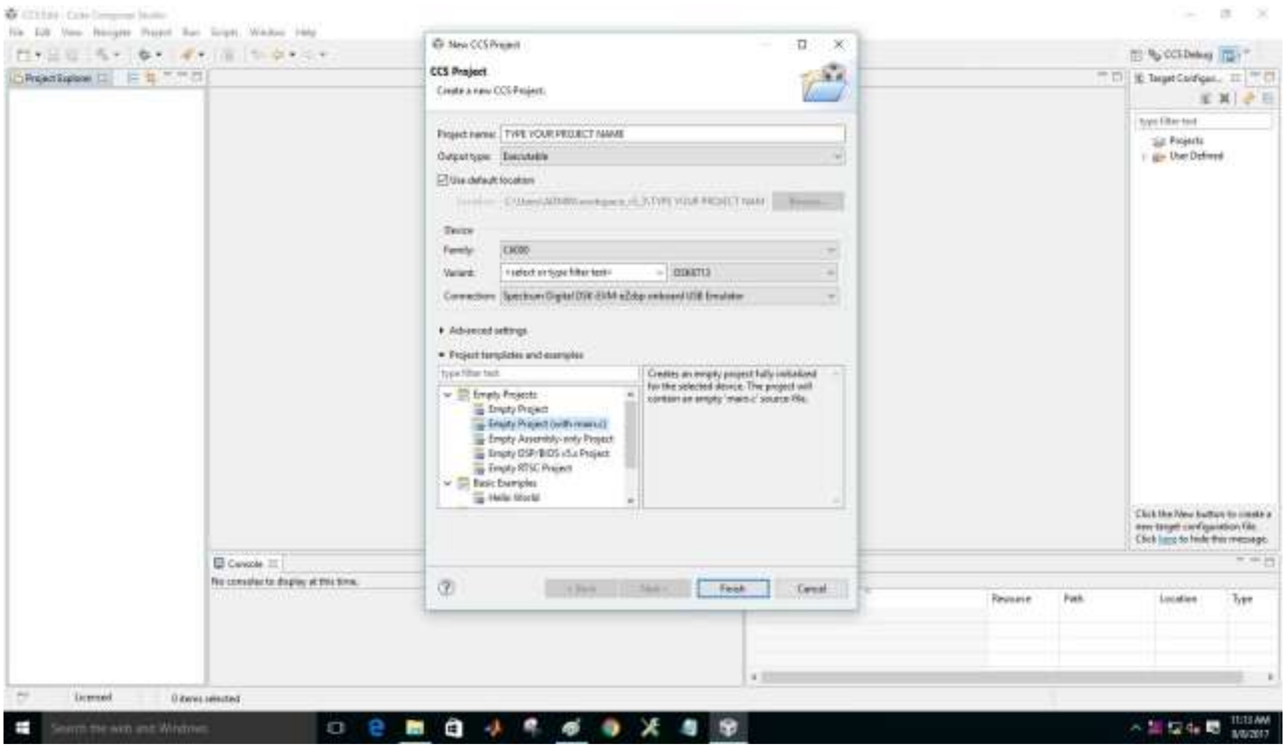
Step-1: Double click on CCS Studio icon and the following screen will appear. We can see the project explorer window towards the left, console at the middle and target configuration files towards the right side of the screen.



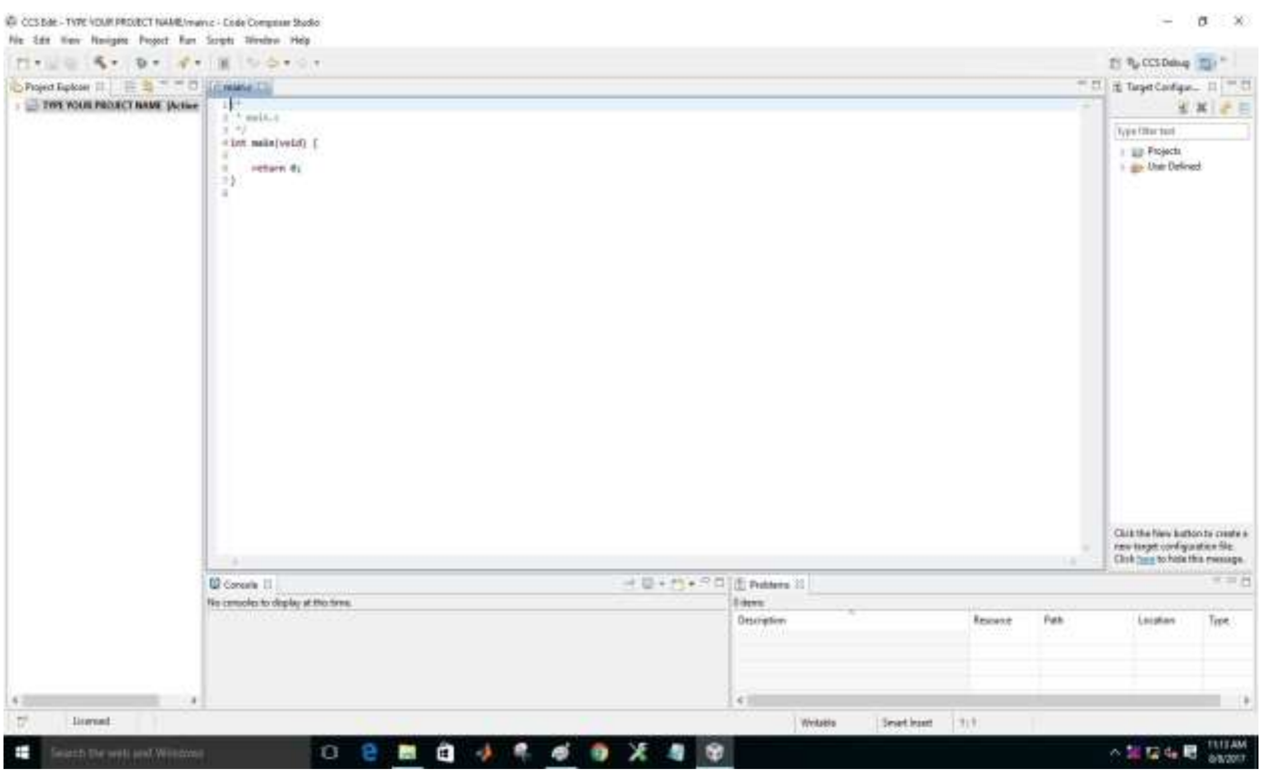
Step-2: Click on File → New → CCS Project



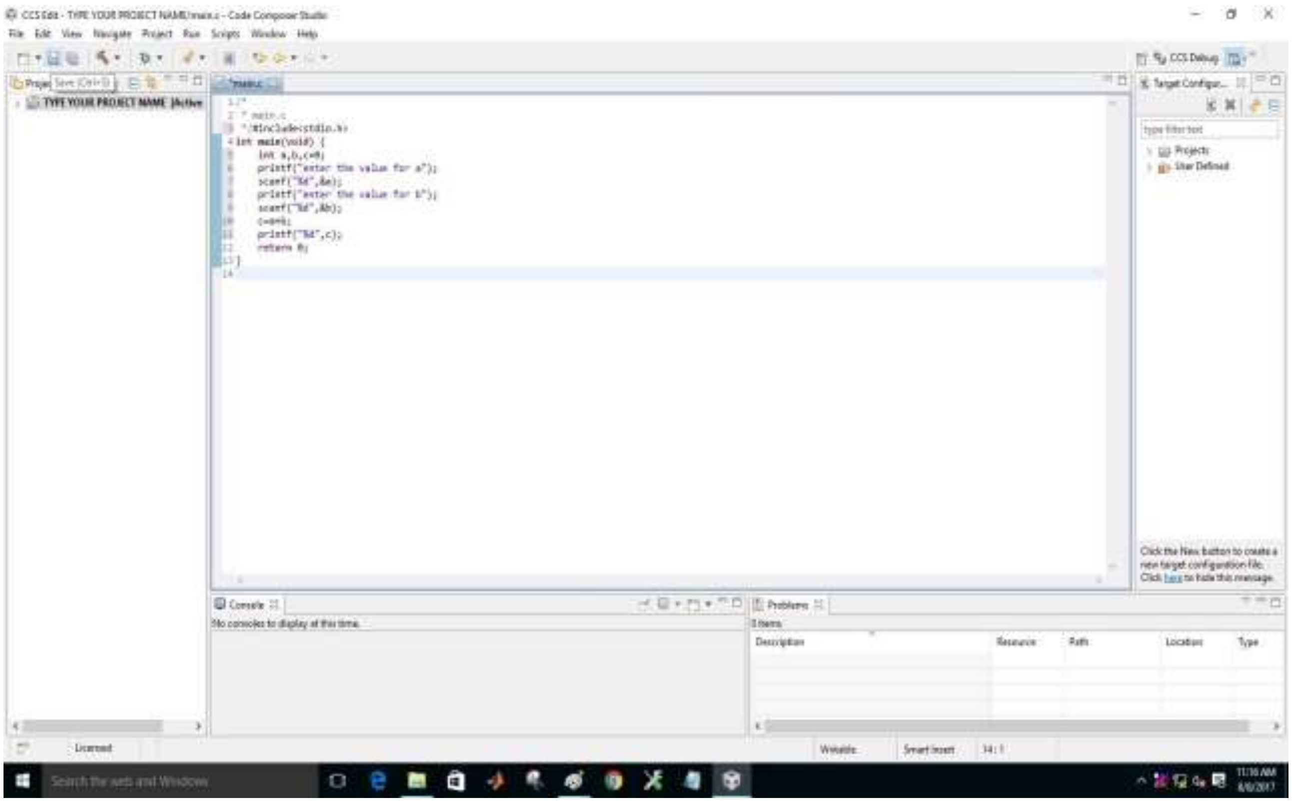
Step-3: Type your Project name, select Output type as Executable from the dropdown menu, Device Family as C6000, Variant as DSK6713, Connection as Spectrum Digital DSK EVM eZdsp USB Emulator, click on Empty projects (with main.c) & finish.



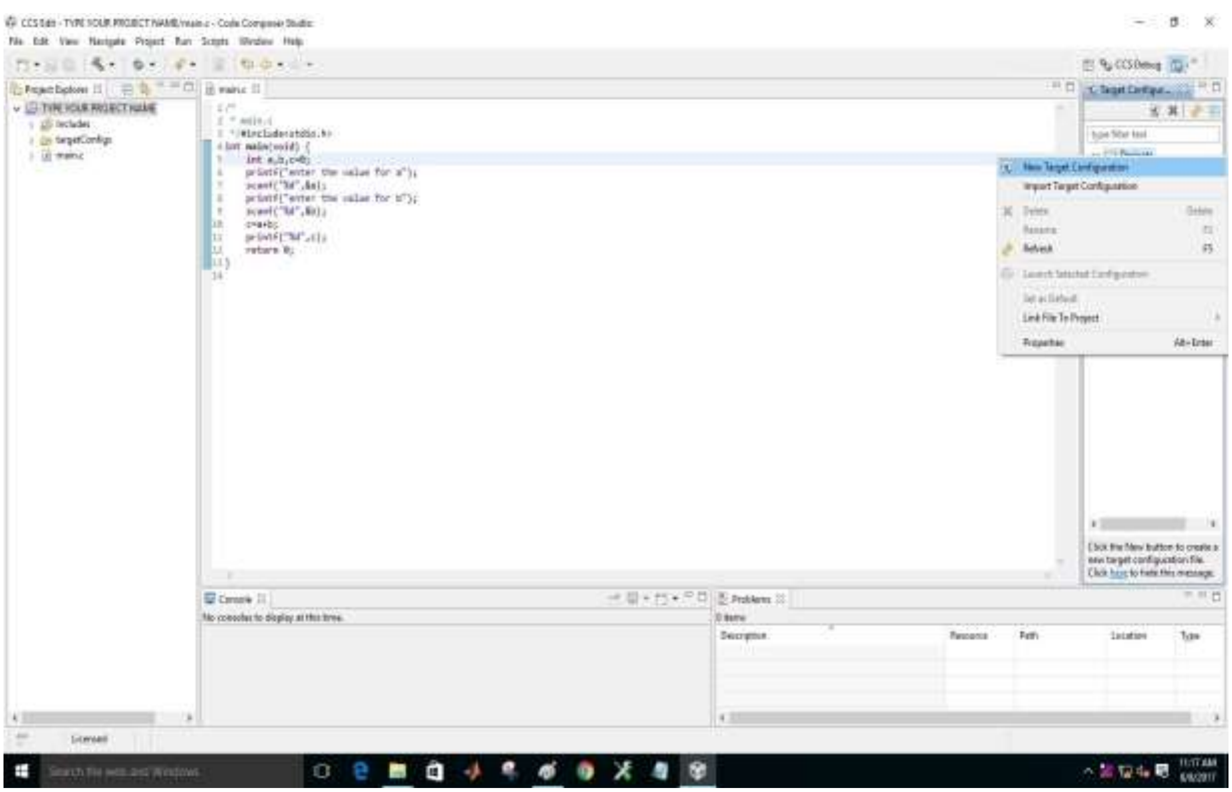
Step-4: Screen as shown below will appear with the Project Name.



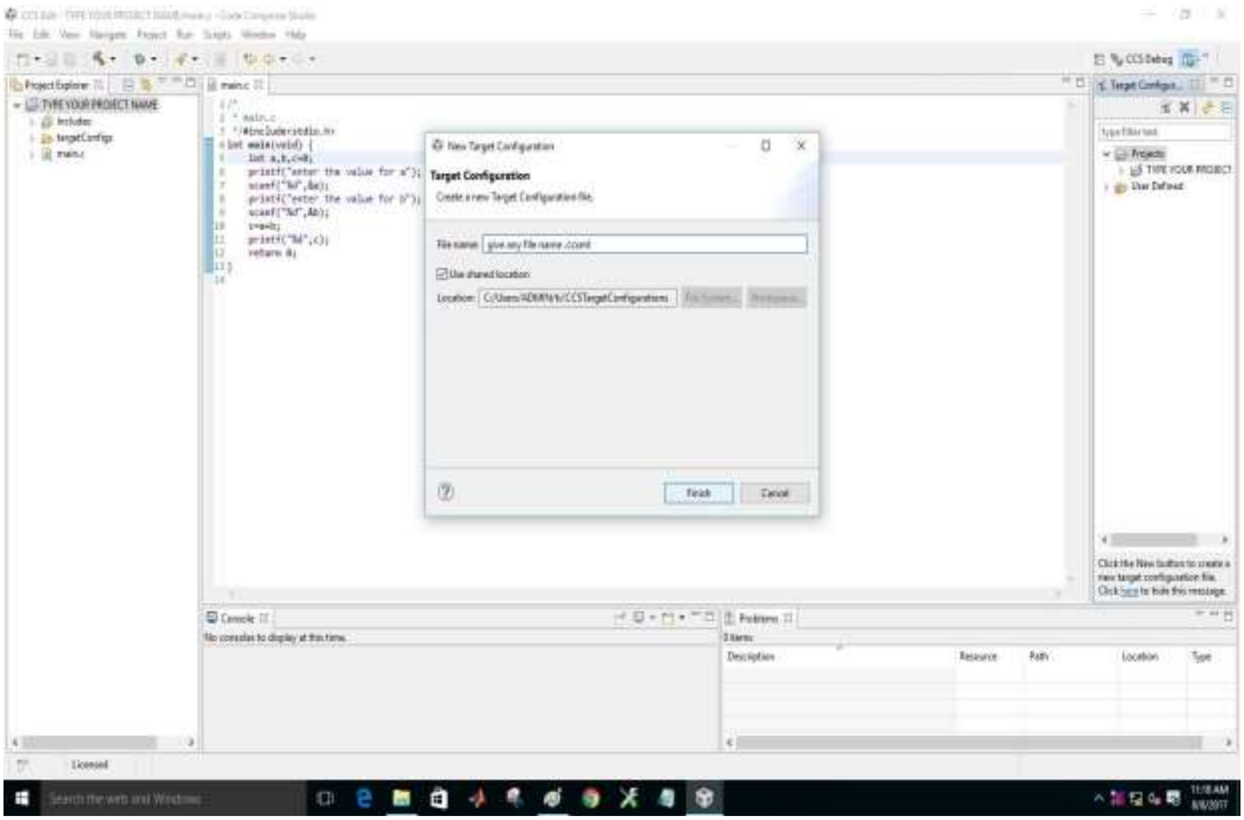
Step-5: Write your C code and then click on save.



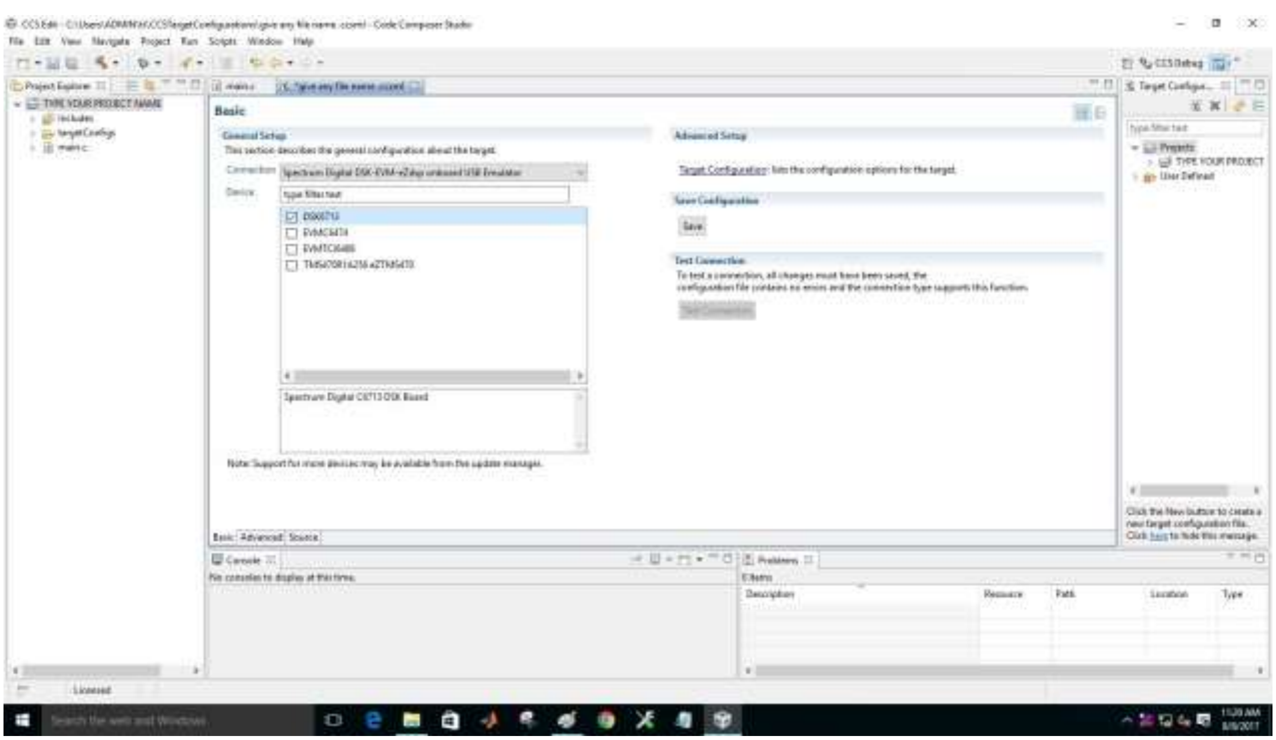
Step-6: Right click on projects and select New Target Configuration.



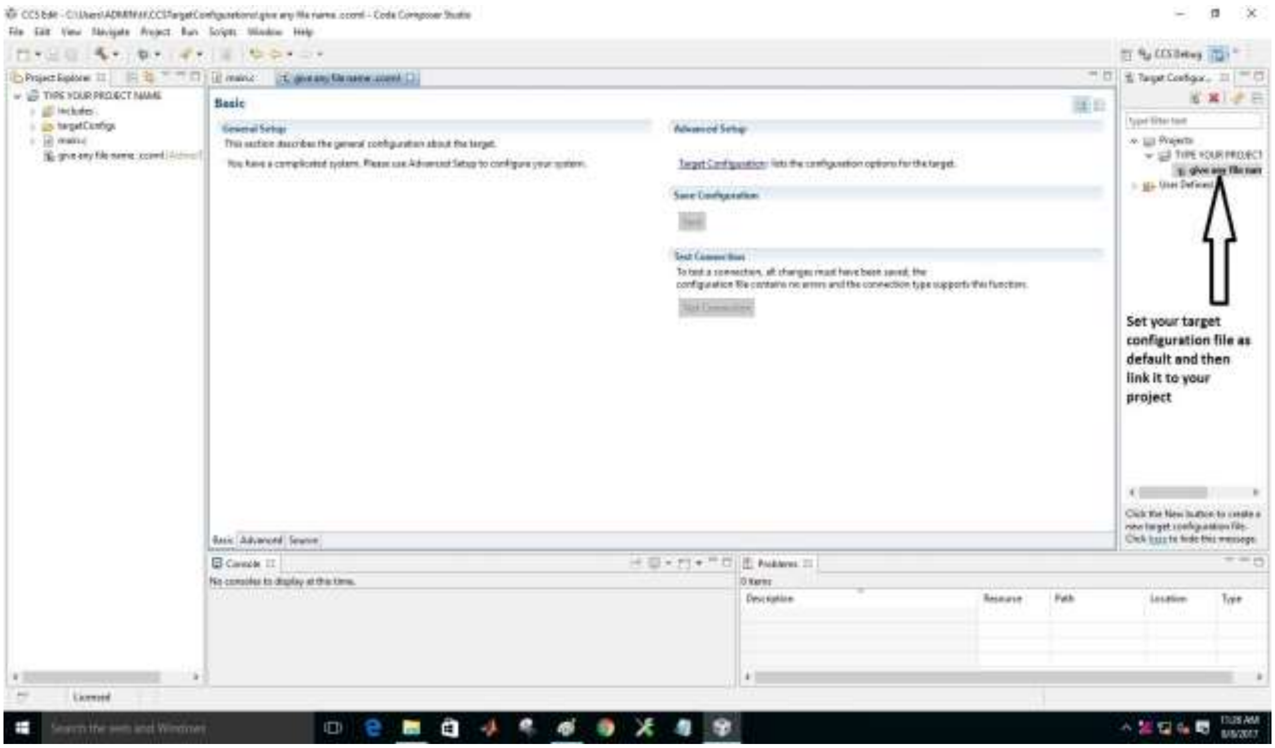
Step-7: Give the file name with .ccxml extension and click finish.



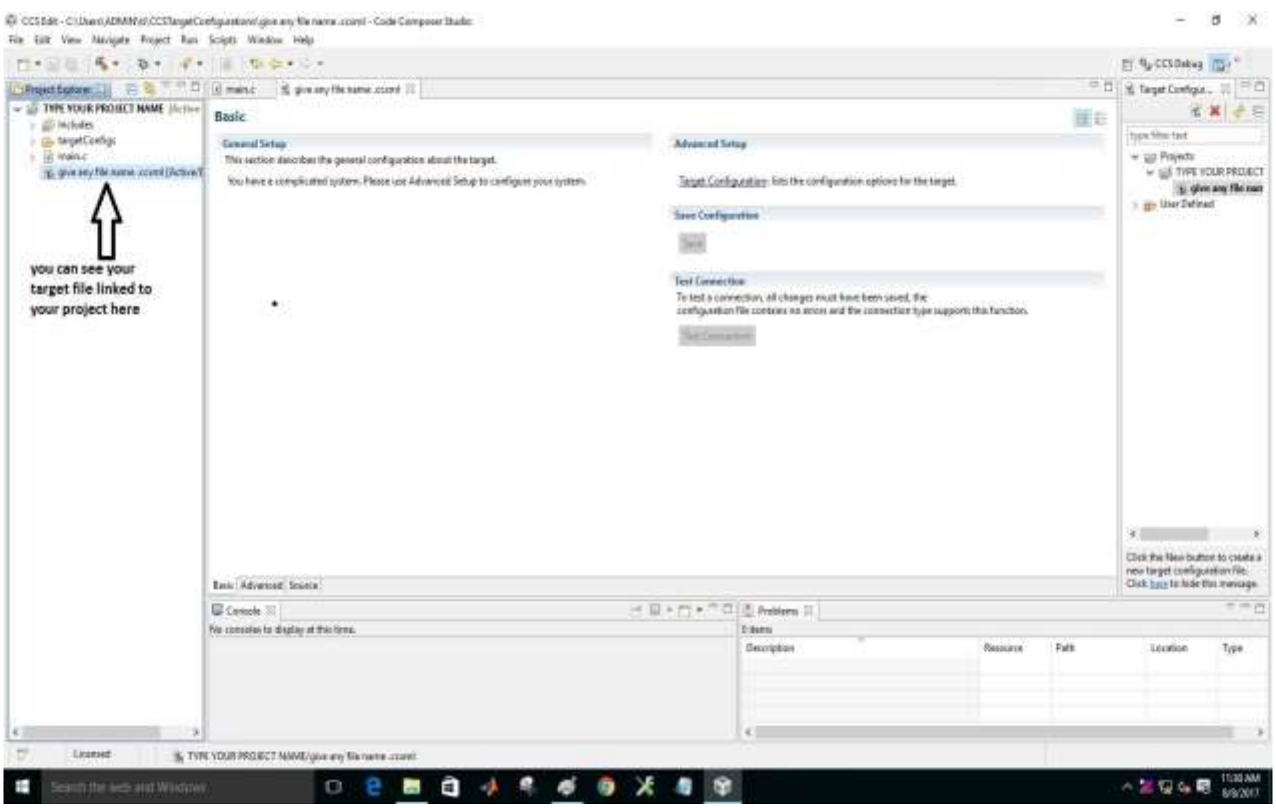
Step-8: Select the connection as Spectrum Digital DSK EVM eZdsp onboard USB Emulator and then select device as DSK6713 and click save.



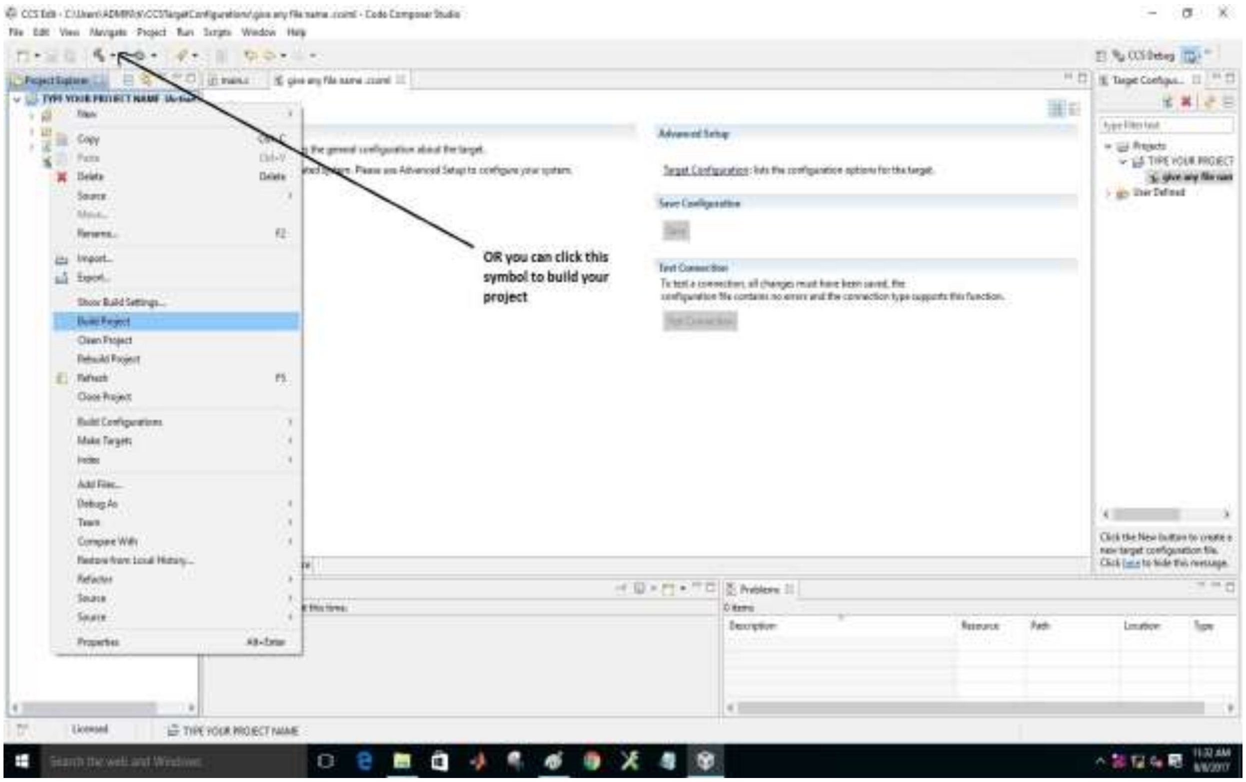
Step-9: Target Configuration file will appear as shown using the arrow symbol.

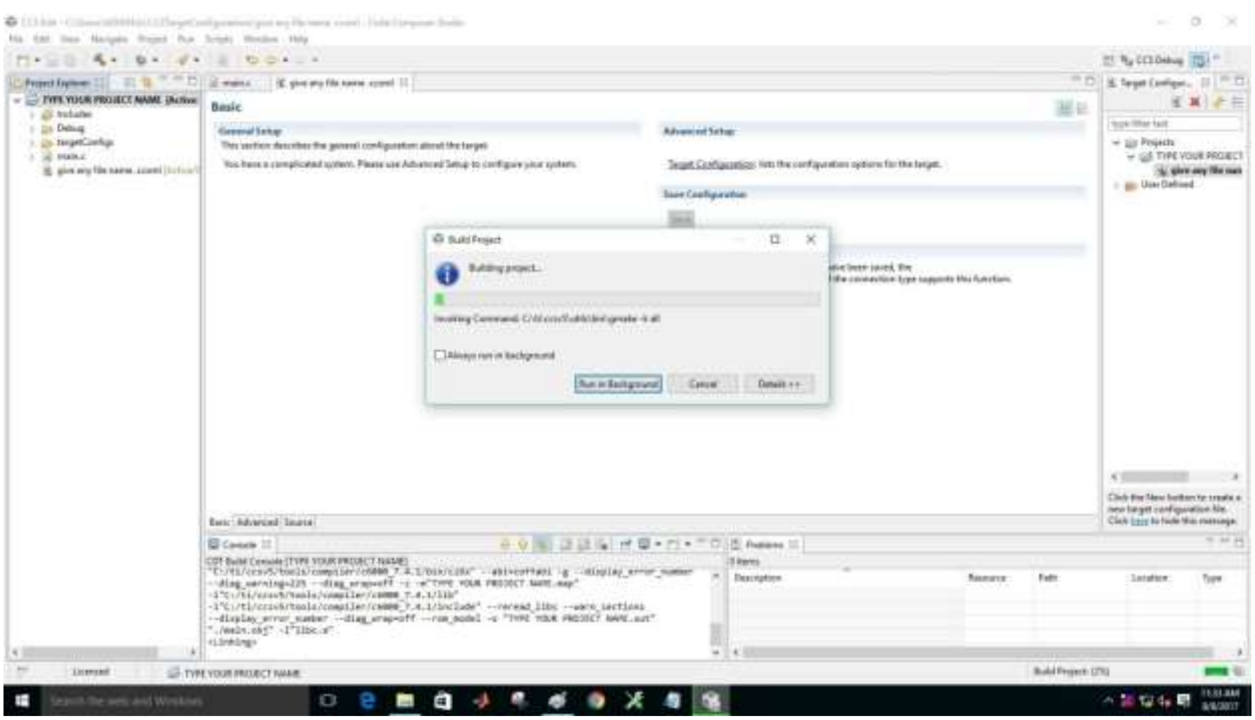


Step-10: Right click on the target configuration file, select link file to project and now you can see the target configuration file linked to your project as shown using arrow symbol

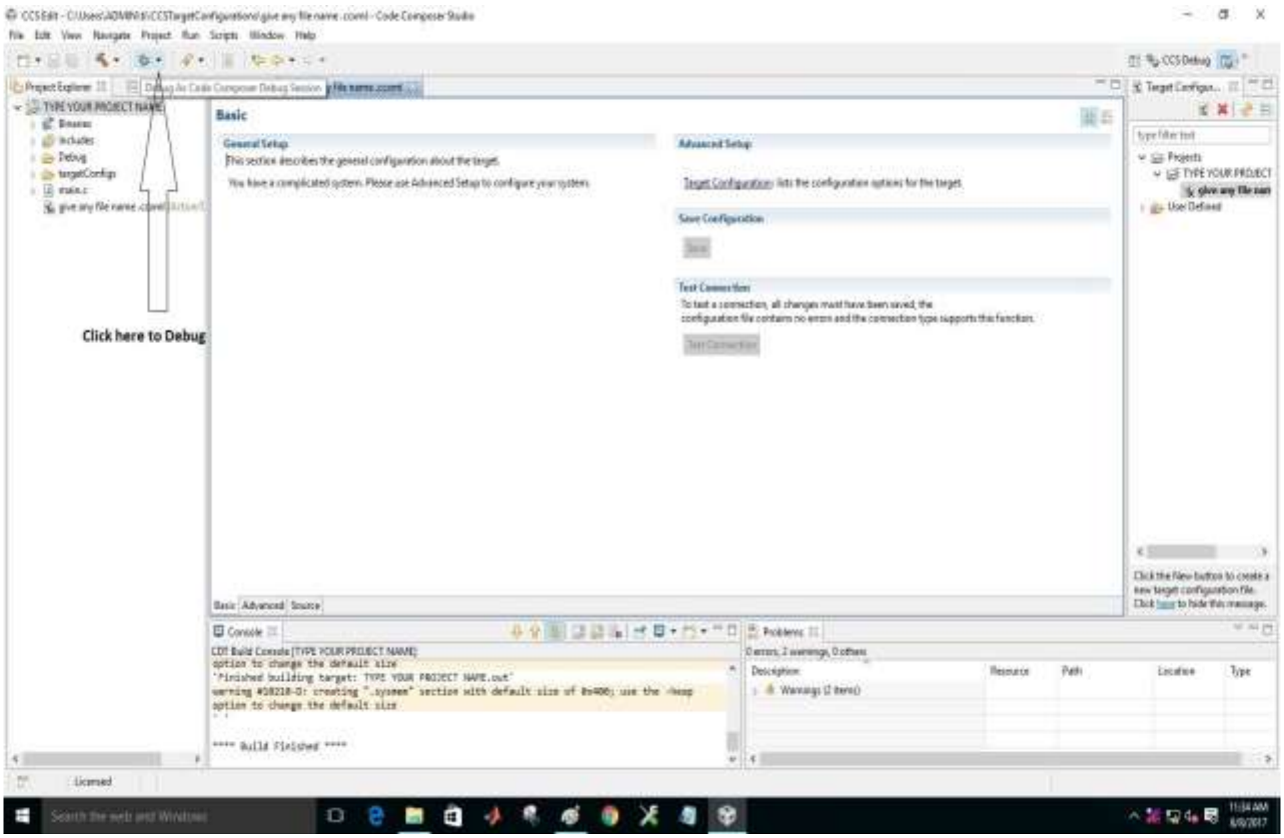


Step-11: Right click on the Project name and select Build project or click on the symbol as indicated by the arrow mark.

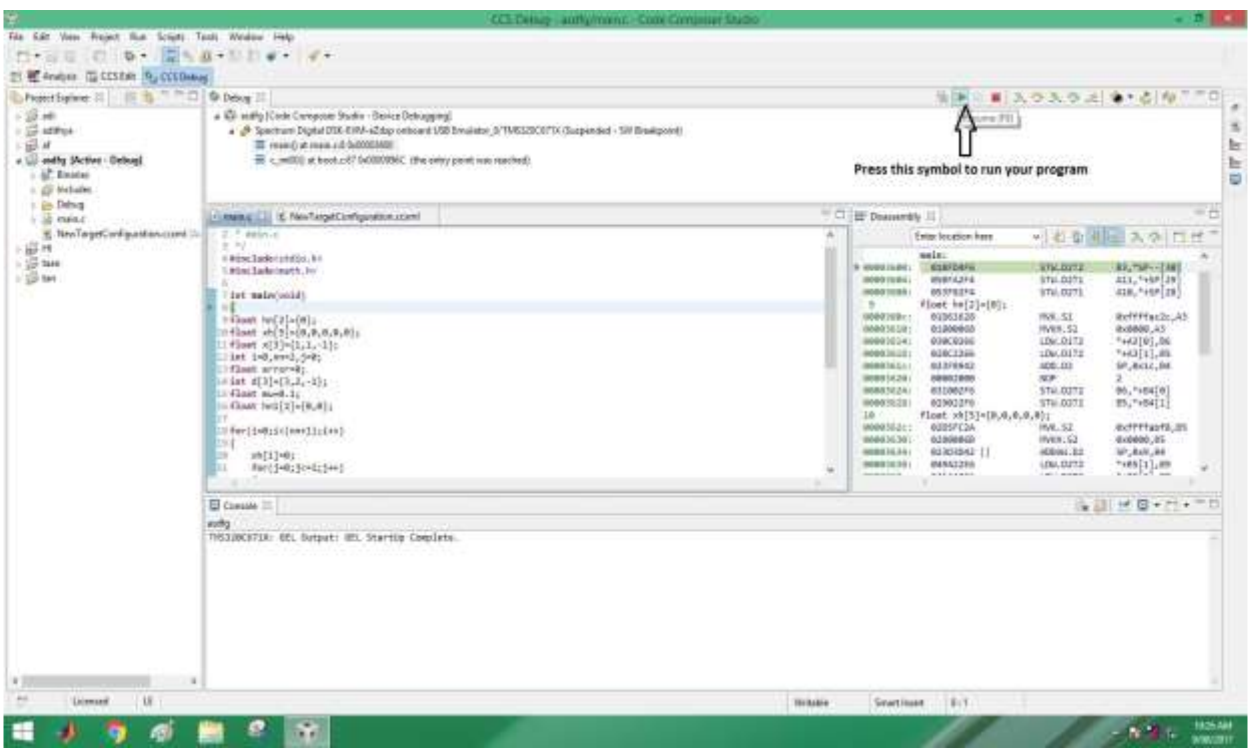




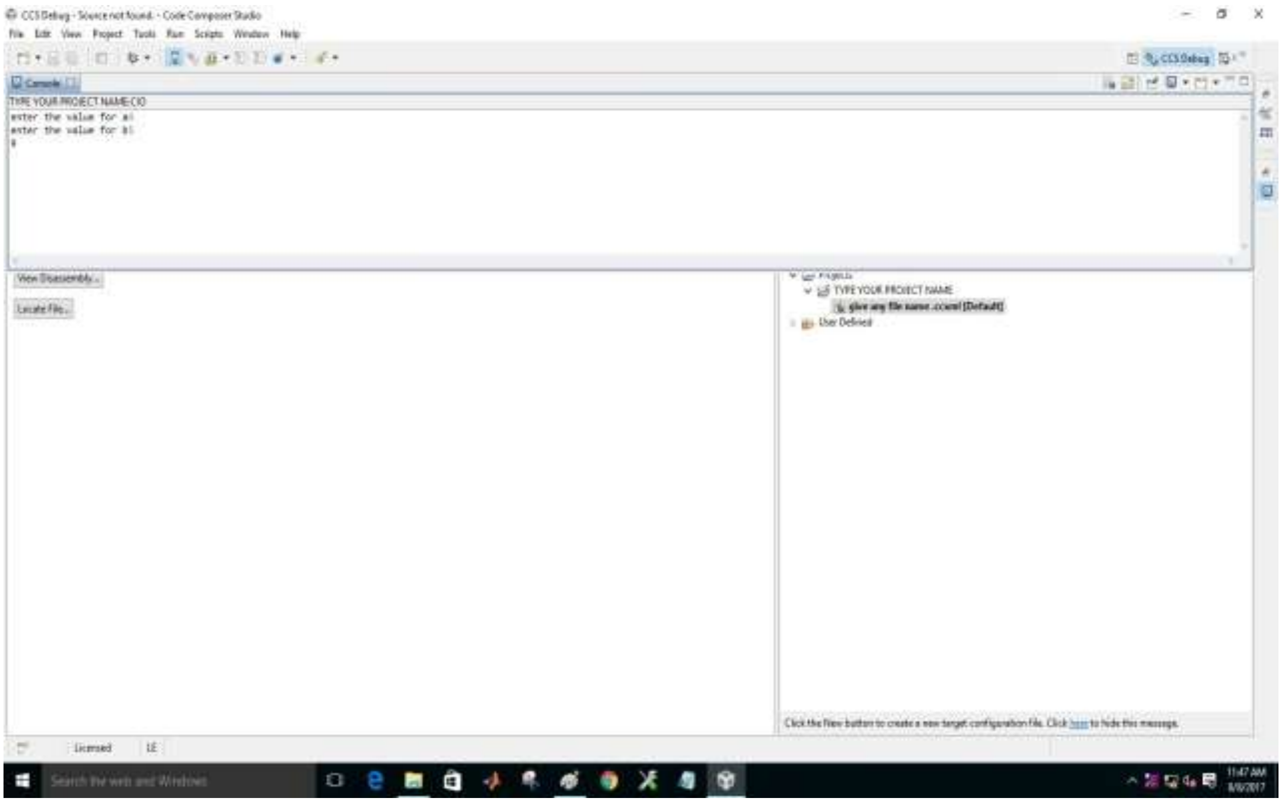
Step-12: Click on the symbol shown to debug your code.

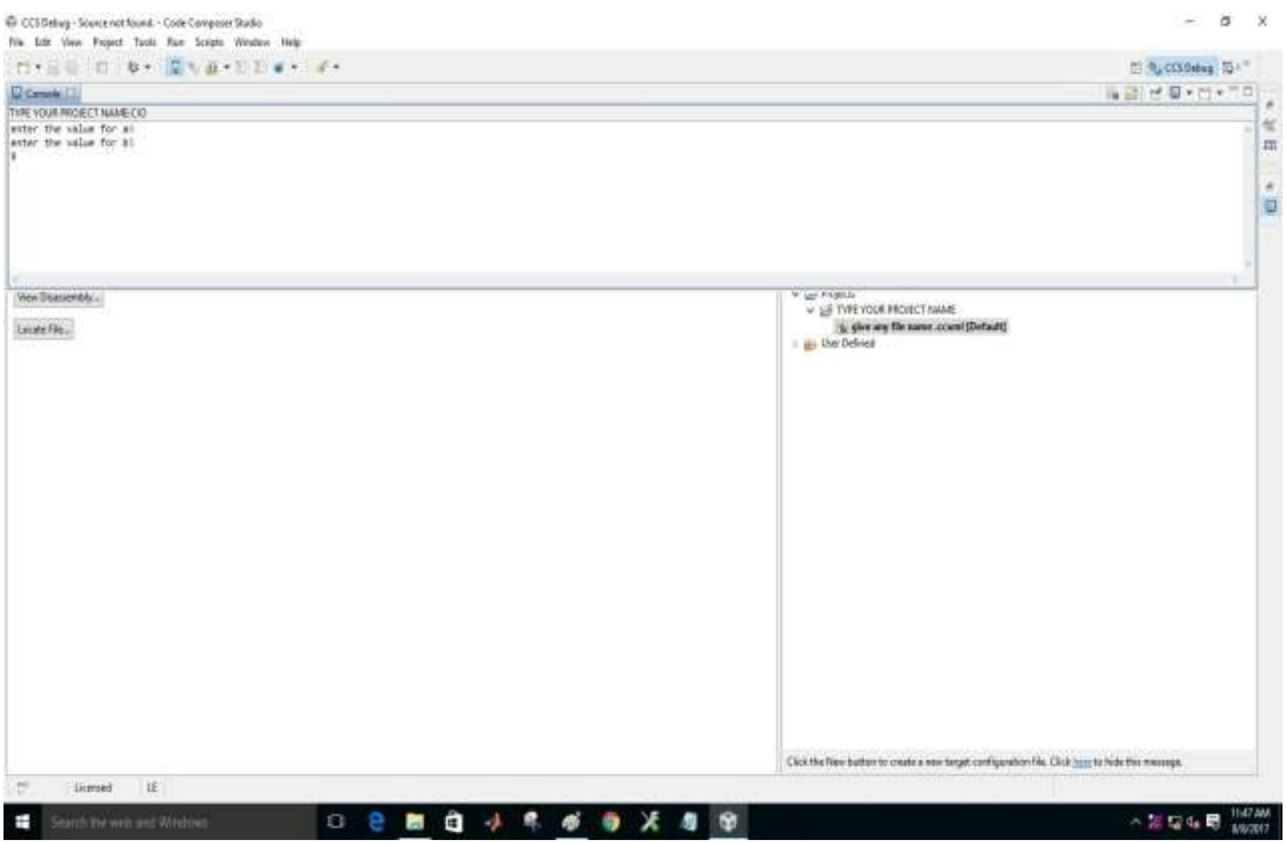


Step-13: Click on the Run symbol as depicted below to execute your code.



Step-13: After successful execution, output can be viewed in the console window as shown.





Outcome: The student will be able to implement the fundamental concepts of DSP using TMS320C6713 DSK.

For more information on 6713 DSK click here:
http://my.fit.edu/~vkepuska/ece3551/DSP%20Applications%20with%20the%20C6713%20and%20C6416%20DSK/ChassaingBook

8. a) Impulse Response: The response of a system to unit impulse input is called Impulse response, which can completely characterize a system.

To be demonstrated:

A discrete system LTI system is described by the following difference equation. Determine the first six samples of the system's impulse response.

$$(n) + (n - 1) = (\ \frac{1}{-})(n) + (3\frac{1}{3})(n - 3)$$

Output: h (n) = [1/3, 1/3, 1/3, 2/3, 2/3, 2/3]

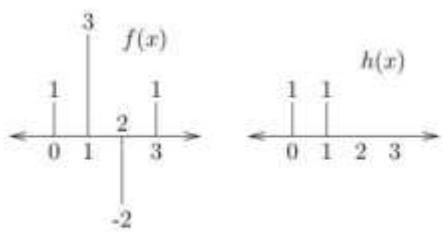
b) Convolution: It is the process through which we can obtain the output of an LTI system.

Linear Convolution

One dimensional linear discrete convolution is defined as:

$$g(x) = \sum_{s=-\infty}^{\infty} f(s)\,h(x-s) = f(x) * h(x)$$

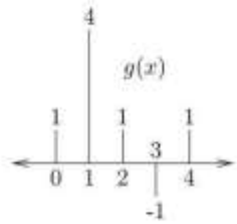
For example, consider the convolution of the following two functions:



This convolution can be performed graphically by reflecting and shifting $h(x)$, as shown in Figure 1. The samples of $f(s)$ and $h(s-x)$ that line up vertically are multiplied and summed:

$$g(0) = f(-1)h(1) + f(0)h(0) = 0 + 1 = 1$$
$$g(1) = f(0)h(1) + f(1)h(0) = 1 + 3 = 4$$
$$g(2) = f(1)h(1) + f(2)h(0) = 3 + -2 = 1$$
$$g(3) = f(2)h(1) + f(3)h(0) = -2 + 1 = -1$$
$$g(4) = f(3)h(1) + f(4)h(0) = 1 + 0 = 1$$

The result of the convolution is as shown below:



Notice that when $f(x)$ is of length 4, and $h(x)$ is of length 2, the linear convolution is of length $4 + 2 - 1 = 5$.

Circular Convolution

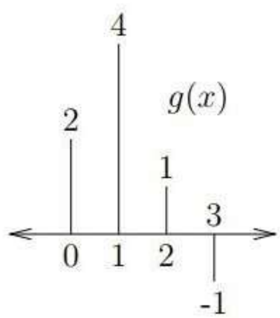
One dimensional circular discrete convolution is defined as:

$$g(x) = \sum_{s=0}^{M-1} f(s)\, h((x-s) \bmod M) = f(x) \circledast h(x)$$

For $M = 4$, the convolution can be performed using circular reflection and shifts of $h(x)$, as shown in Figure 2. The samples of $f(s)$ and $h((s-x) \bmod M)$ that line up vertically are multiplied and summed:

$$g(0) = f(3)h(1) + f(0)h(0) = 1 + 1 = 2$$
$$g(1) = f(0)h(1) + f(1)h(0) = 1 + 3 = 4$$
$$g(2) = f(1)h(1) + f(2)h(0) = 3 + -2 = 1$$
$$g(3) = f(2)h(1) + f(3)h(0) = -2 + 1 = -1$$

The result of the convolution is as shown below:



Notice that $f(x)$ and $h(x)$ are both treated as if they are of length 4, and the circular convolution is also of length 4.

Experiment - 13

Cross-correlation of two discrete-time sequences, x and y measures the similaritybetween x and shifted (lagged) copies of y as a function of the lag.

Auto-correlation is the linear dependence of a variable with itself at two points in time.

The cross correlation between $x(n)$ and $y(n)$ is defined as

$$r_{xy}(l) = \sum_{n=-\infty}^{\infty} x(n)y(n-l) \quad l = 0, \pm 1, \pm 2, ...$$

$$r_{xy}(l) = \sum_{n=-\infty}^{\infty} x(n+l)y(n) \quad l = 0, \pm 1, \pm 2, ... \quad \text{---- (1)}$$

The cross correlation between $y(n)$ and $x(n)$ is defined as

$$r_{yx}(l) = \sum_{n=-\infty}^{\infty} x(n-l)y(n) \quad l = 0, \pm 1, \pm 2, ...$$

$$r_{yx}(l) = \sum_{n=-\infty}^{\infty} x(n)y(n+l) \quad l = 0, \pm 1, \pm 2, ... \quad \text{---- (2)}$$

From (1) and (2)

$$r_{xy}(l) = r_{yx}(-l)$$

To be demonstrated:

Input signal: Obtain the cross correlation of the two sequences x(n)= [2 1 2 4] and y(n)= [2 1 2 4] and plot it.

Output: $(\boldsymbol{l}) = [\mathbf{8}, \mathbf{8}, \mathbf{12}, \mathbf{25}, \mathbf{12}, \mathbf{8}, \mathbf{8}]$

Correlation Coefficient=1

The autocorrelation of $x(n)$ is defined as

$$r_{xx}(l) = \sum_{n=-\infty}^{\infty} x(n+l)x(n) \quad l = 0, \pm 1, \pm 2, ...$$

$$r_{xx}(l) = \sum_{n=-\infty}^{\infty} x(n-l)x(n) \quad l = 0, \pm 1, \pm 2, ...$$

Input signal: Obtain the auto correlation of the sequence x(n)= [2 -1 1 3 5] and plot it.

Output: $(\boldsymbol{l}) = [\mathbf{10}, \mathbf{1}, \mathbf{4}, \mathbf{15}, \mathbf{40}, \mathbf{15}, \mathbf{4}, \mathbf{1}, \mathbf{10}]$

(b) As defined in Experiment 4 (a)