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.

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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.
- 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 students 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 basic signal 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 <u>http://www.digital.iitkgp.ernet.in/dsp/expts/index.php</u> | 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 |
| | a) Write a C code to obtain the impulse response of a given system and | |
| 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.
- 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

| | | | | | P | rograi | n outco | omes | | | | | Pro | gram spe outcom | |
|-----|-----|-----|-----|-----|-----|--------|------------|------|-----|------|------|------|------|--------------------|------|
| | 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

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RECORD OF CIE FOR PERFORMANCE IN THE LAB CLASSES

| Section | Batch | Group Number | |
|-----------------|-------|--------------|--|
| Staff in Charge | Day | Timings | |

| SI. | USN | Name Of The Student | AC1: Preparedness (8M) | SUBJECT: |
|-----|-----|---------------------|---------------------------------|----------|
| No. | | | AC2: Conduction (8M) | |
| 1. | | | AC3: Viva (8M) | 2000000 |
| 2. | | | | 20EC57L |
| 3. | | | AC4: Report Writing (8M) | |
| 3. | | | AC5: Result Interpretation (8M) | |
| 4. | | | T: Total (40M) | |

| SI. | Date | Experiment | | 5 | stude | nt-1 | | | | 5 | stude | nt-2 | | | | 5 | Stude | ent-3 | | | | 5 | štude | nt-4 | | |
|-----|------|------------|----|----|-------|------|----|---|----|----|-------|------|----|---|----|----|-------|-------|---|---|----|----|-------|------|---|---|
| No. | | | AC | AC | AC | AC | AC | т | AC | AC | AC | AC | AC | т | AC | AC | AC | AC | | Т | AC | AC | AC | AC | | т |
| | | | 1 | 2 | 3 | 4 | 5 | | 1 | 2 | 3 | 4 | 5 | | 1 | 2 | 3 | 4 | 5 | | 1 | 2 | 3 | 4 | 5 | |
| 1. | | | | | | | | | | | | | | | | | | | | | | | | | | |
| 2. | | | | | | | | | | | | | | | | | | | | | | | | | | |
| 3. | | | | | | | | | | | | | | | | | | | | | | | | | | |
| 4. | | | | | | | | | | | | | | | | | | | | | | | | | | |
| 5. | | | | | | | | | | | | | | | | | | | | | | | | | | |

| SI. | Date | Experiment | | | Stude | | | | | | Stude | | | | | | Stude | | | | Student-4 | | | | | | |
|------|----------|--|---------|---------|---------|---------|----------|----------|---------|---------|---------|---------|---------|----------|---------|---------|---------|----------|---------|------------|-----------|---------|---------|---------|---------|---|--|
| No. | | | AC 1 | AC 2 | AC 3 | AC 4 | AC 5 | т | AC 1 | AC 2 | AC 3 | AC 4 | AC 5 | т | AC 1 | AC 2 | AC 3 | AC 4 | AC 5 | т | AC 1 | AC 2 | AC 3 | AC 4 | AC 5 | Т | |
| 6. | | | | | | | | | | | | | | | | | | | | | | | | | | Γ | |
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| 11. | | | | | | | | \vdash | | | | | | \vdash | | | | | | \vdash | | | | | | ┢ | |
| 12. | | | | | | | | ╞ | | | | | | ╞ | | | | | | \uparrow | | | | | | ┢ | |
| 13. | | | | | | | | | | | | | | | | | | | | | | | | | | F | |
| 1 | CI | E from Experiments 1 to 13 (40 Marks) (A) | | I | 1 | 1 | 1 | L | | I | | | I | I | | | | 1 | 1 | _ | | 1 | 1 | 1 | 1 | L | |
| | | Lab test (10M) (B) | | | | | | | | | | | | | | | | | | | | | | | | | |
| | | Final CIE = A + B | | | | | | | | | | | | | | | | | | | | | | | | | |
| | | Attendance Percentage | | | | | | | | | | | | | | | | | | | | | | | | | |
| Sign | ature of | the Staff in Charge: | | | | | | | | | | Sig | matu | ire | of th | e La | b i n | Cha | rge: | | - | | | | | | |

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 $(f_{s/2})$

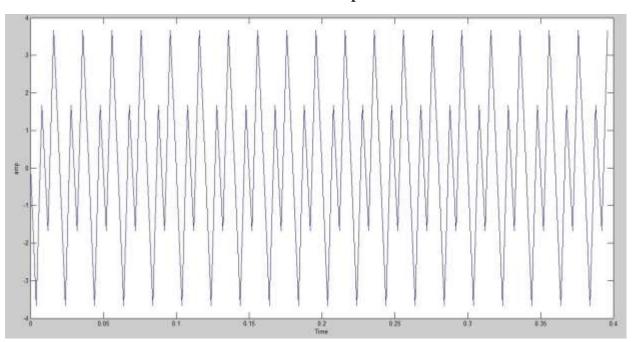
$$\mathbf{f}_{\max} \leq \mathbf{f}_{s/2}$$

To be demonstrated:

Input:

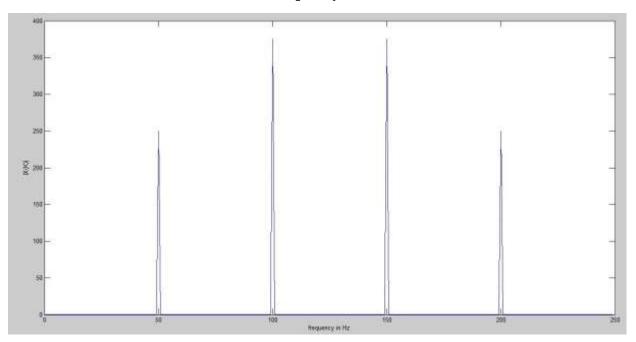
For the input signal x (t) =2*sin (2*pi*f1*t) + 3*sin (2*pi*f2*t) with f1=200Hz and f2=400Hz apply the sampling theorem to the signal to obtain each of the following outputs and provide your inference for each case.

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

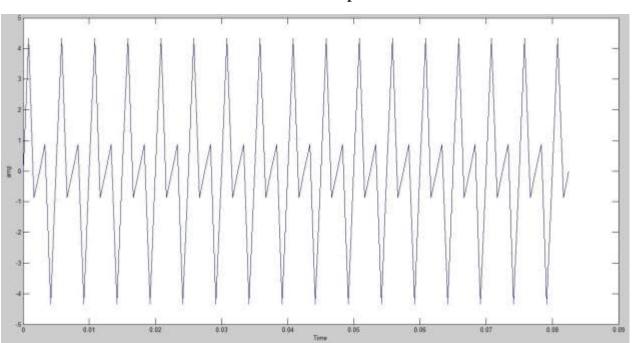


Time domain plot

Frequency Plot

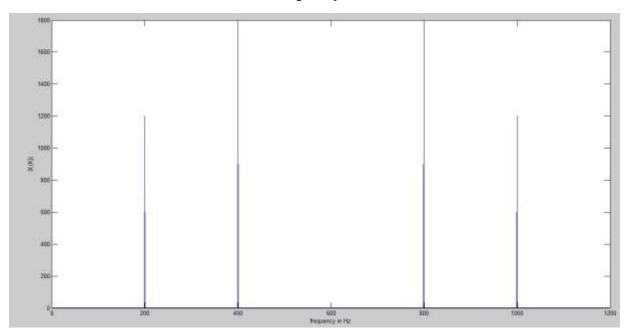


Case-2: Both the frequency components of the input signal are less than folding frequency. (Case of perfect reconstruction)

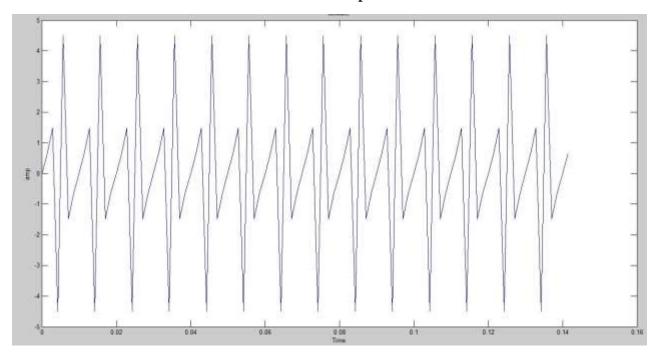


Time domain plot

Frequency Plot

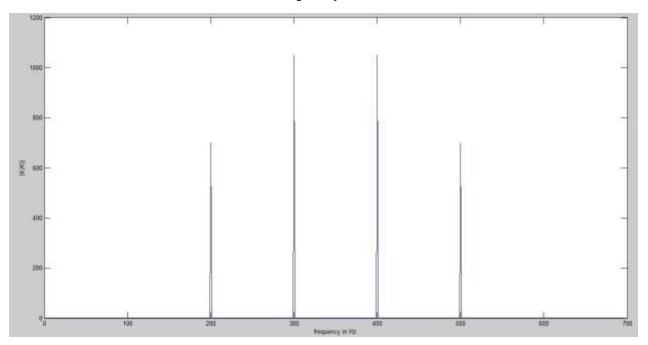


Case-3: One of the input frequency component is greater than folding frequency and one less than folding frequency.



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:

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

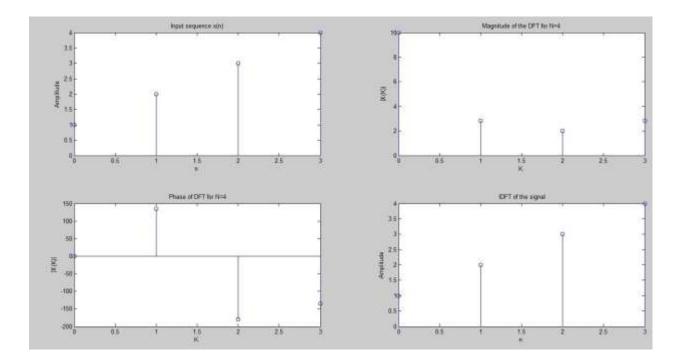
F(k) are often called the 'Fourier Coefficients' or 'Harmonics'.

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

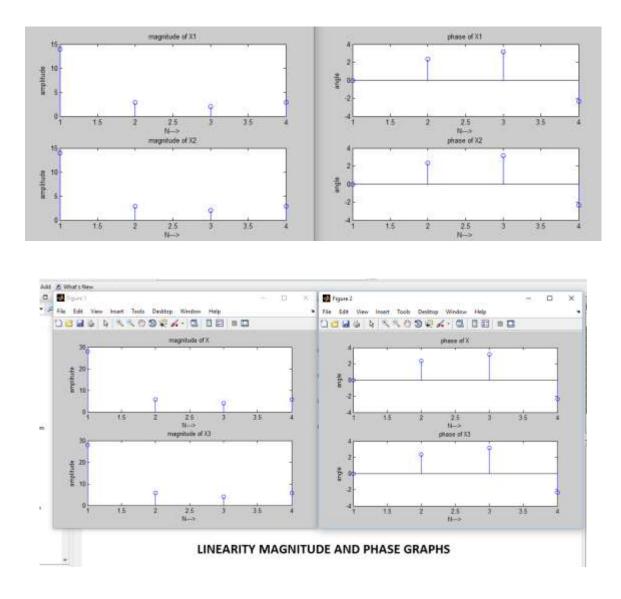
If x1(n) and x2(n) denote two periodic signals with period N and its Fourier Transform coefficients denoted by X1(k) and X2(k) respectively, then by linearity, the Fourier transform of x1(n)+x2(n) is given by X1(k)+X2(k).

To demonstrate:

Verify the property of linearity for the given input sequences:

```
Sequence 1: [2 3 4 5]
Sequence 2: [2 3 4 5]
```

| enter sequence 1: [2 3 4 5] enter sequence 2: [2 3 4 5] | LINEARITY | Y PROPERTY | |
|--|-------------------|------------|-------------------|
| 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 | | |
| | -4.0000 + 4.0000i | -4.0000 | -4.0000 - 4.0000i |
| sum of X1(k) and X | ()()r) is | | |
| Sum OF AT(K) and A | -4.0000 + 4.0000i | 4 0000 | -4.0000 - 4.0000i |



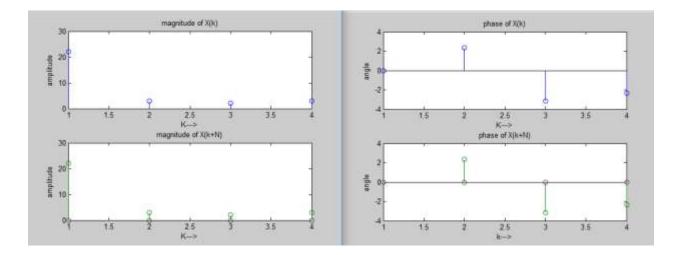
b) Periodicity

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 \mathbf{x}(n):
           5
                6
     4
                         7
 dft of x(n) is X(k):
  22.0000
                       -2.0000 + 2.00001 -2.0000 - 0.00001 -2.0000 - 2.00001
 X(N+k):
  Columns 1 through 6
                      22.0000 + 0.0000i 0
-2.0000 + 2.0000i 0
-2.0000 - 0.0000i 0
-2.0000 - 2.0000i 0
         0
                                                                        0
                                                                                             0
                                                                                                                   0
         0
                                                                        0
                                                                                             0
                                                                                                                   0
         0
                                                                        0
                                                                                             0
                                                                                                                   0
         0
                                                                        0
                                                                                              0
                                                                                                                   0
  Column 7
         0
         0
                                       Periodicity property
         0
         0
```



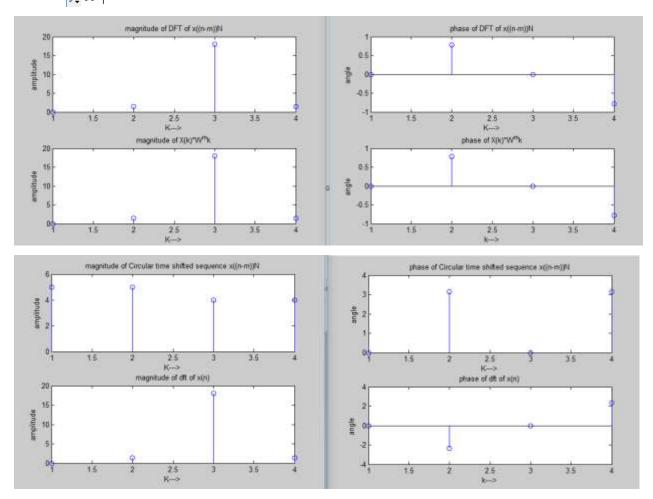
c) Circular time shift property of a sequence

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 \mathbf{x}(n) is
                       -1.0000 - 1.0000i 18.0000
                                                               -1.0000 + 1.0000i
          0
  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
f_{\underline{x}} >>
```



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: [4567] 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 + 2.00001 -2.0000 -2.0000 - 2.00001

Circular frequency shift of X(K) i.e DFT of x(n).e<sup>(</sup>(j*2*pi*l*n/N)

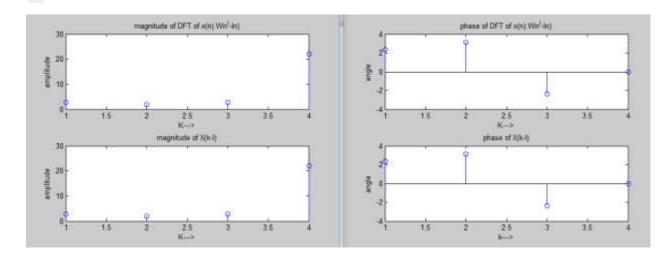
-2.0000 + 2.00001 -2.0000 -2.0000 - 2.00001 22.0000

Resultant circular frequency shift i.e X((k - 1))N

-2.0000 - 2.00001 22.0000 + 0.00001 -2.0000 + 2.00001 -2.0000 - 0.00001

Thus Circular-Frequency Shift Property is verified

f<sub>k</sub> >>
```



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 -2.0000i
```

 $f_{\frac{x}{2}} >>$

Conjugate symmetry property for a real sequence

b)Circular convolution of two sequences

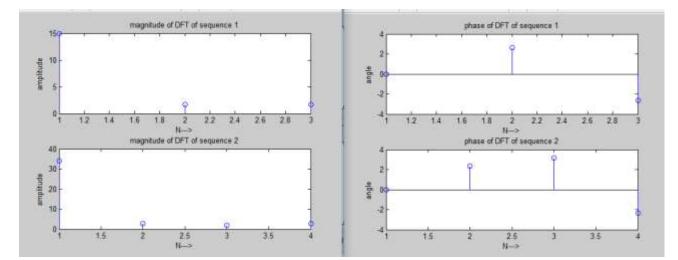
The circular convolution of two sequences x1(n) and x2(n) in time domain i.e. ' $x1(n) \otimes x2(n)$ ' is represented as multiplication i.e. 'X1(k).X2(k) ' in frequency domain.

To demonstrate:

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

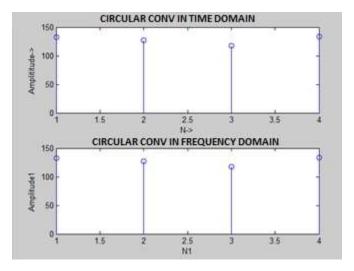
Sequence 1: [4 5 6]

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 + 0.86601 -1.5000 - 0.86601
DFT of sequence 2:
34.0000 -2.0000 + 2.00001 -2.0000 -2.00001
```

```
f_{\star} >>
```



c) Time reversal of a sequence

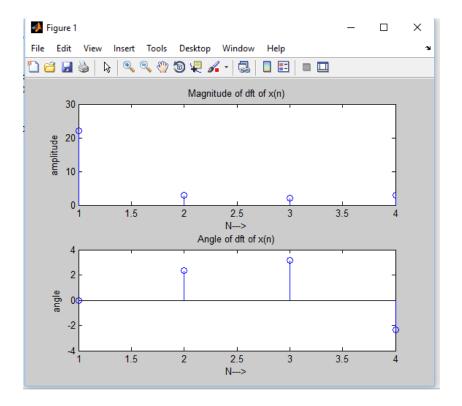
Time reversal of a sequence is demonstrated as: DFT of 'x(N - n) ' is given by ' X(N - k) '.

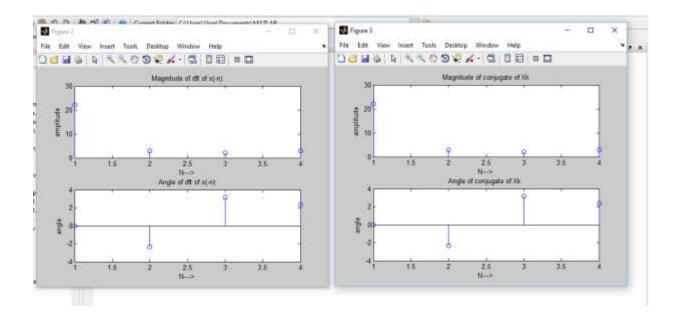
To demonstrate:

Verify the property of Time reversal for the given input sequence: [4567]

```
Enter the sequence
 [4 5 6 7]
 DFT of the sequence \mathbf{x}(n)
                      -2.0000 + 2.0000i -2.0000
                                                           -2.0000 - 2.0000i
   22.0000
 Time reversed sequence x(-n)
            7
                  6
       4
                        5
 DFT of time reversed sequence
   22.0000
                     -2.0000 - 2.0000i -2.0000
                                                            -2.0000 + 2.0000i
 Conjugate of X(k)
                      -2.0000 - 2.0000i -2.0000
   22.0000
                                                       -2.0000 + 2.0000i
 Thus Time-reversal Property is verified
f_{\underline{x}} >>
```

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: [00.707 10.707]

```
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

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

is defined as

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

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

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

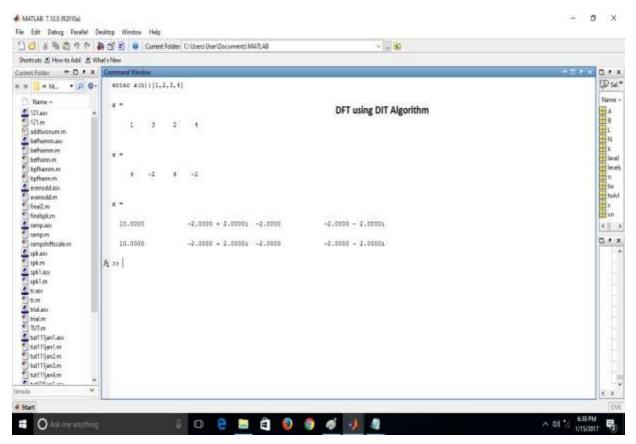
for $0 \le n \le N/2 - 1$. The DFT of the N-point signal x[n] can be written as

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

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:

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

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

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

That is, the even DFT values X[2k] for $0 \le 2k \le N-1$ 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]

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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]

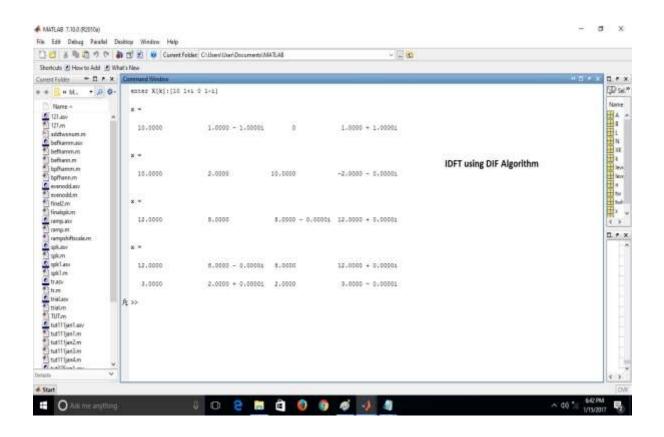
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| ntfuldet + C + x | Convenient Window | | | | | + C + + D |
| - M. + 19 0- | enter X[k]:[10, | 141,0,1-1] | | | | 14 |
| Name - | | | | | | Th |
| 121aw e 121m addiwsnumm | 10,0000 | 1.0000 - 1.00001 | 0 | 1.0000 + 1.00001 | | |
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IV. 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 (2n π /N-1), $0 \le n \le N - 1$ 0, otherwise

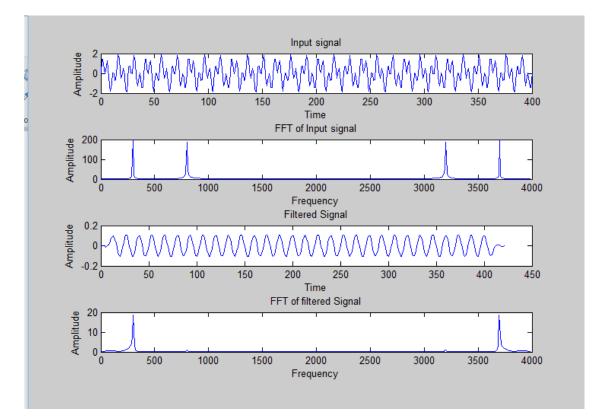
(ii). Hamming window function w (n) =0.54 – 0.46 cos ($2n\pi/N-1$), $0 \le n \le N - 1$ 0, otherwise

I. HAMMING LOW PASS FILTER

To be demonstrated:

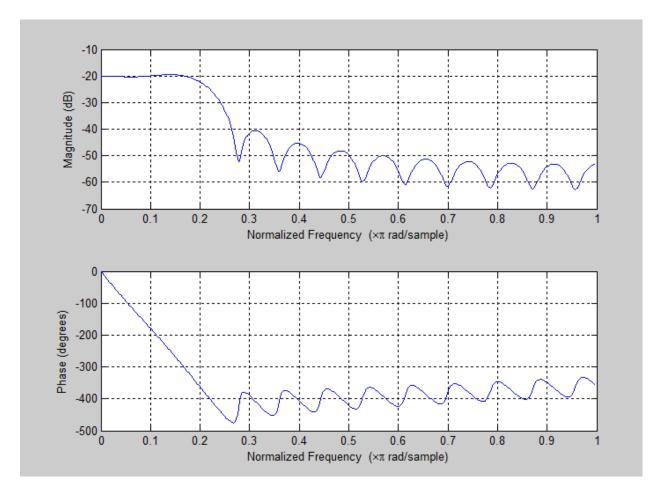
For the given data below, construct a Hamming low pass filter and filter the input signal with frequencies as below: Filter specifications:

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
                         0.0155
                                   0.0204
                                             0.0222
               0.0089
                                                       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 >>
```

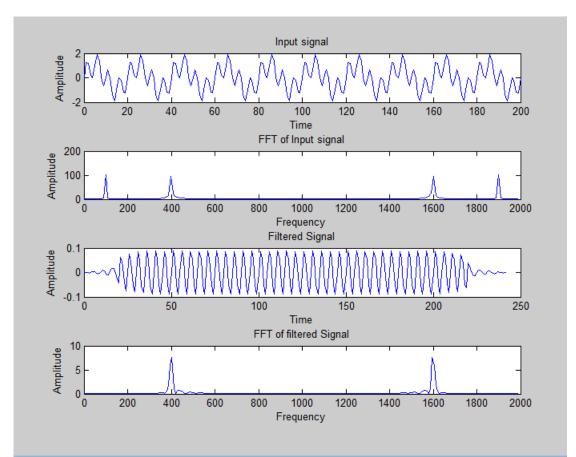
```
34
```

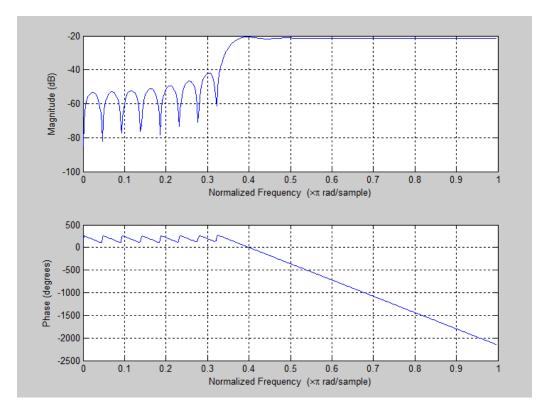


II. HAMMING HIGH PASS FILTER

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





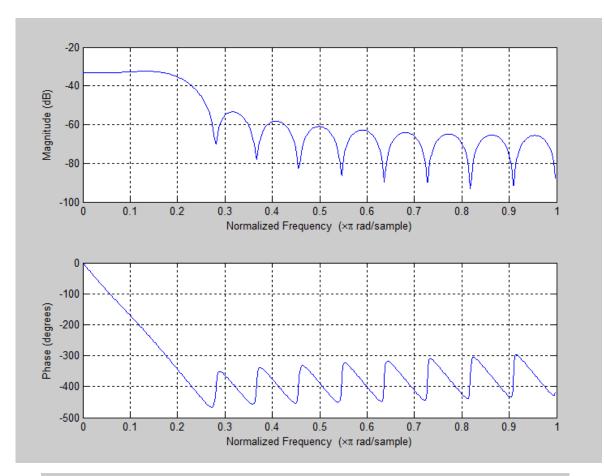
```
→1 <u></u>
```

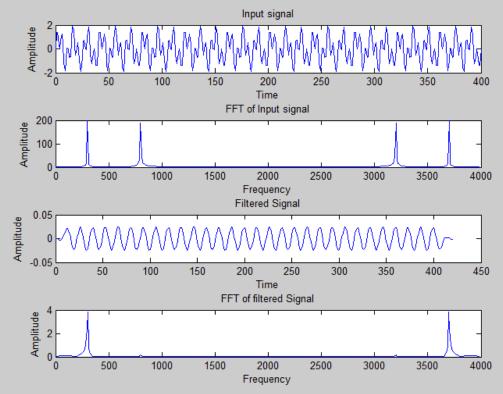
```
Command Window
  h =
   Columns 1 through 6
            -0.0013 -0.0012 0.0002
    -0.0000
                                          0.0016
                                                   0.0013
   Columns 7 through 12
    -0.0006
            -0.0021
                     -0.0013 0.0011
                                          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
fx
     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





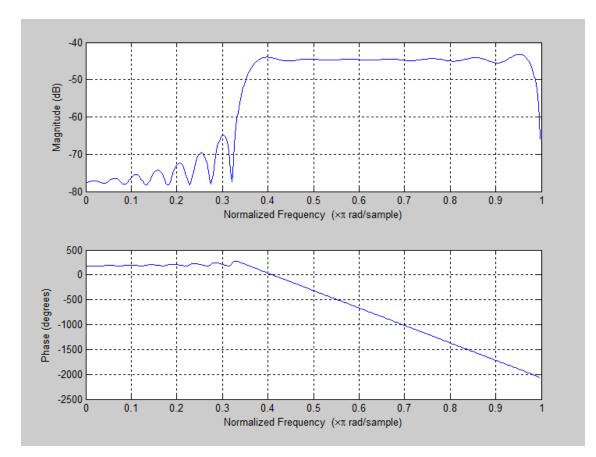
```
Command Window
```

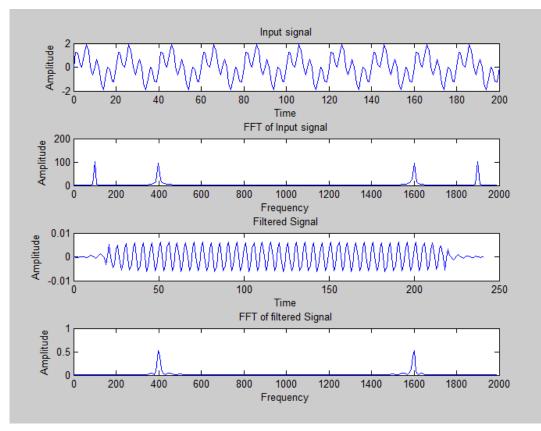
```
→ 🗖 🖉 🗙
   enter the order : 22
   enter f1 : 300
   enter f2 : 800
   enter cut-off freq : 450
     Columns 1 through 7
       0.0003
                -0.0002
                          -0.0008 -0.0011
                                               -0.0009
                                                          -0.0001
                                                                     0.0013
     Columns 8 through 14
                                      0.0049
       0.0028
                 0.0041
                           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
f_{\underline{x}} >>
```

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 -42Input signal: First input frequency - 100 Hz Second input frequency – 400 Hz





```
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 >>
```

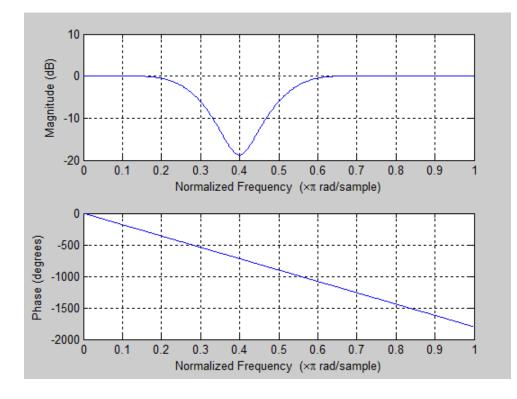
Write a MATLAB code to verify the Band pass and Band reject FIR linear phase filter design using Hamming and Hanning windows (with inbuilt and without using inbuilt commands). Plot the magnitude and phase response. Also, Provide the inference on the basis of results obtained for the set of specifications.

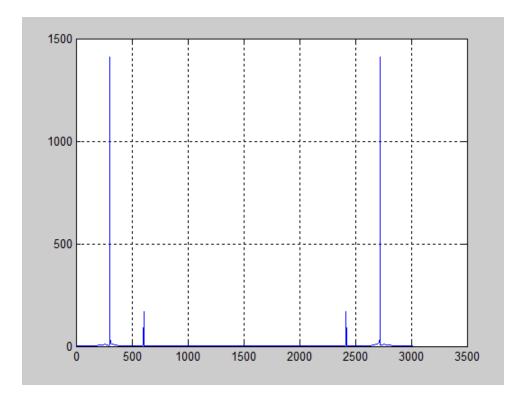
Outcome: The student will be able to design and implement a linear phase FIR filter that best matches the application and satisfies the design requirements.

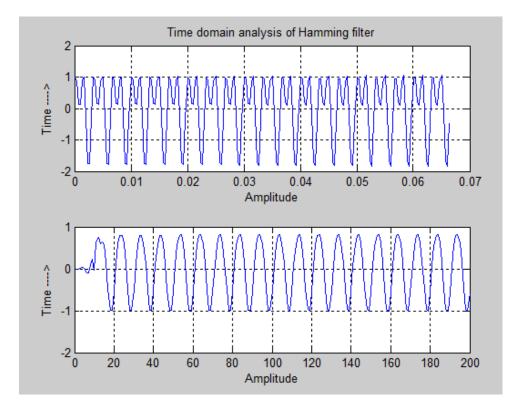
I. HAMMING BAND ELIMINATION FILTER

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







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

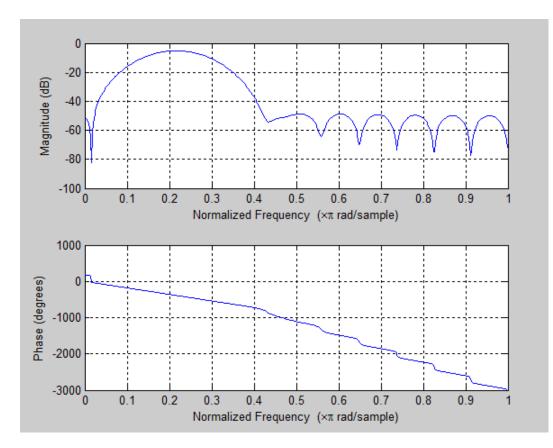
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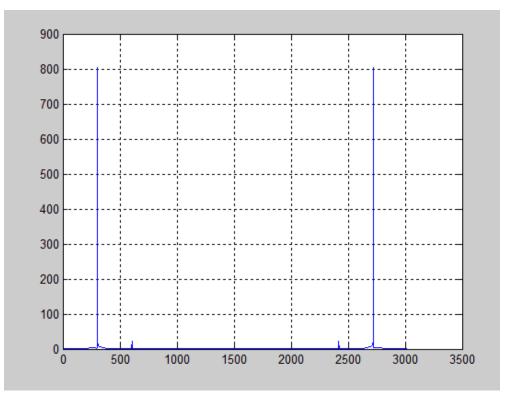
II. HAMMING BAND PASS FILTER

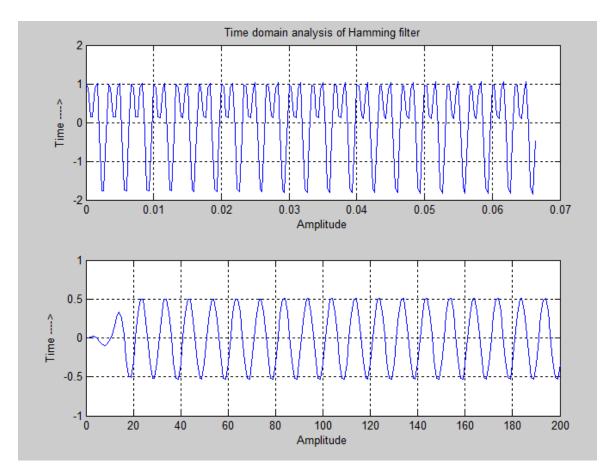
To be demonstrated:

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







```
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
             -0.0528 -0.0241
                                   0.0014
                                             0.0121
                                                      0.0106
     -0.0625
                                                                0.0055
    Columns 22 through 23
      0.0017 -0.0015
```

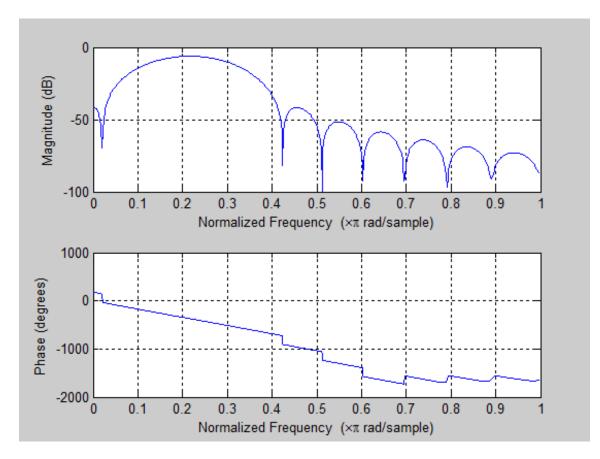
 $f_{\frac{x}{2}} >>$

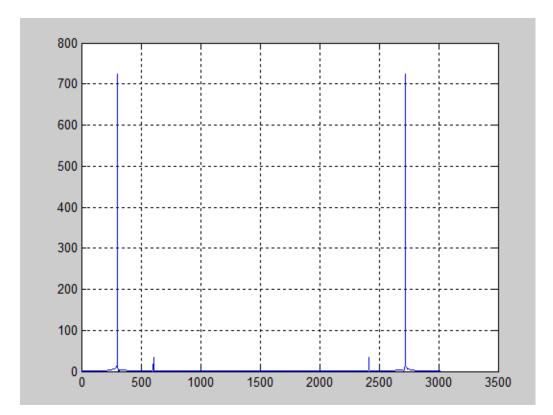
III. HANNING BAND PASS FILTER

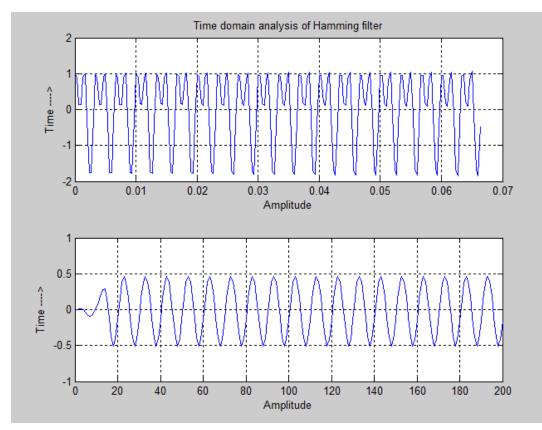
To be demonstrated:

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



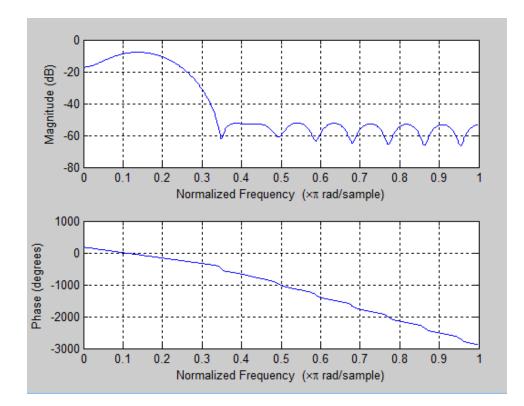


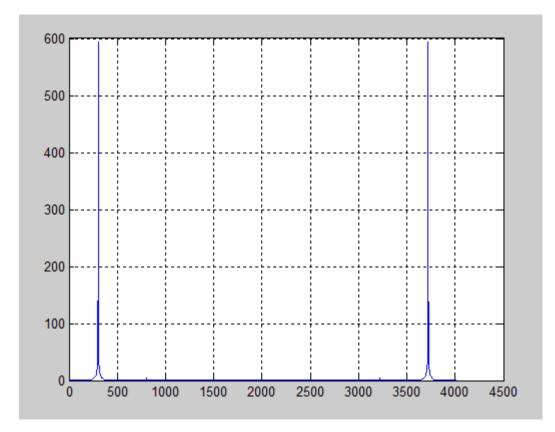


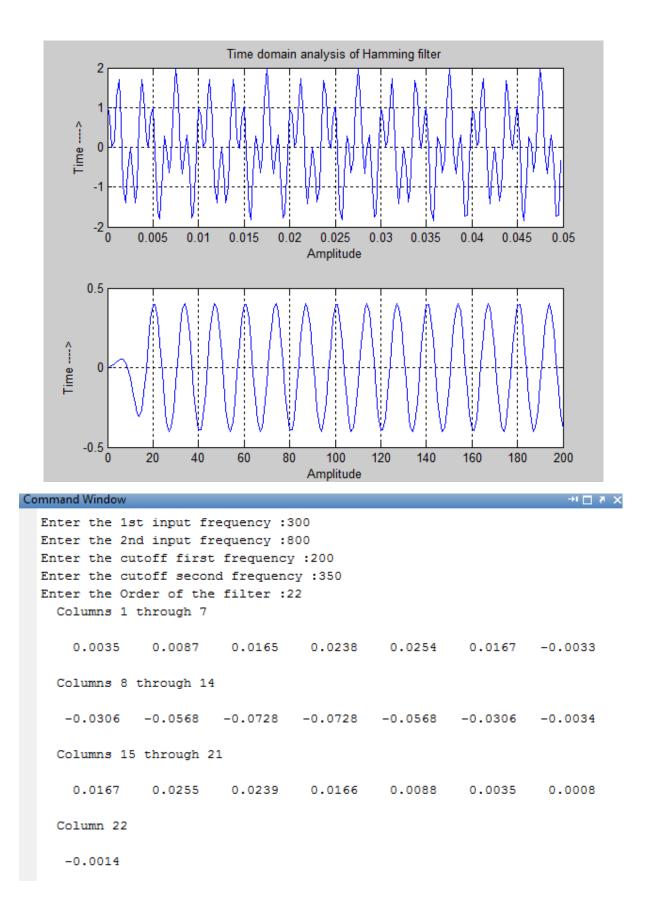
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IV. HANNING BAND ELIMINATION FILTER

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







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.

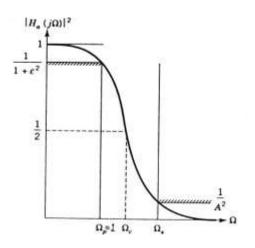
In the impulse invariance technique, the frequency response of analog and digital filters are similar only at low frequencies. This limits the application of impulse invariance method to only low pass filters. A better transformation that preserves frequency response and overcomes the limitations of 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 squaredmagnitude function is given by

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

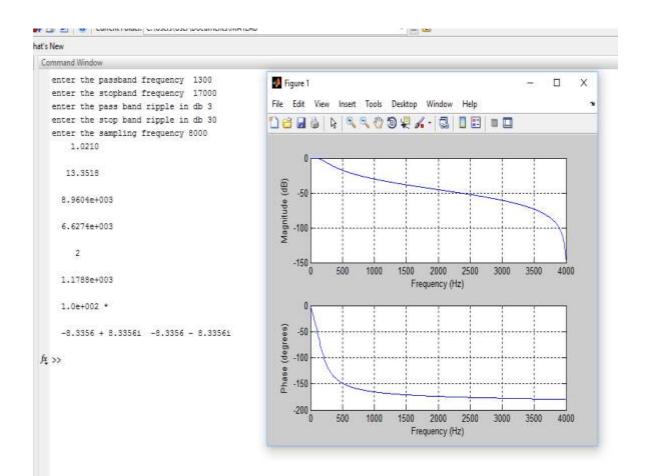
- This function achieves the value of unity at $\Omega = 0$ with the first 2N 1 derivatives being zero at this point (maximally flat passband).
- At infinity, the value is zero and the first 2N-1 derivatives are zero (maximally flat stopband).
 - The following figure gives the normalized specifications, 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:

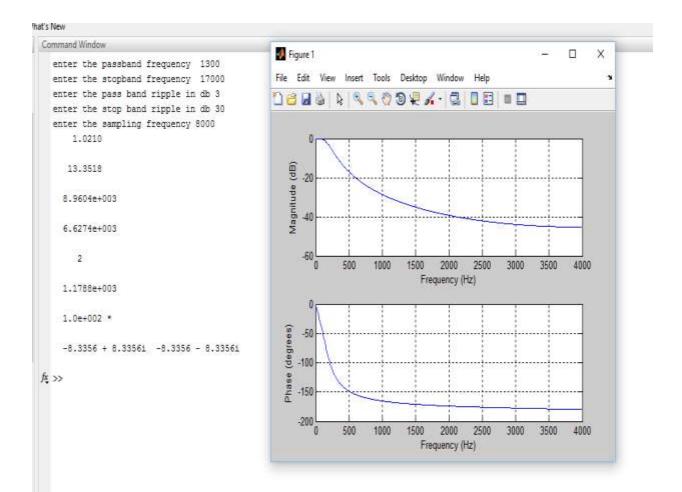
Filter specifications: Pass band edge frequency – 1300 Hz Stop band edge frequency – 17000 Hz Pass band attenuation in db - 3 Stop band attenuation in db – 30 Sampling frequency – 8000 Hz



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



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

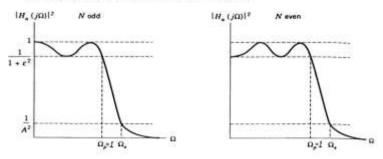
$$|H_a(j\Omega)|^2 = \frac{1}{1 + \epsilon^2 T_N^2(\Omega)},$$
 (16a)

where

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

is the Nth-degree Chebyshev polynomial.

- In the normalized passband 0 ≤ Ω ≤ Ω_p = 1, this function alternatingly achieves the values of 1 and 1/(1+ε²) at N + 1 points such that |H_a(jΩ_p)|² = 1/(1 + ε²). For N even, |H_a(j0)|² = 1/(1 + ε²) and for N odd, |H_a(j0)|² = 1 (equiripple passband).
 - At infinity, the value of |H_a(jΩ)|² is zero and the first 2N 1 derivatives are zero (maximally flat stop-band) (see the figure shown below).

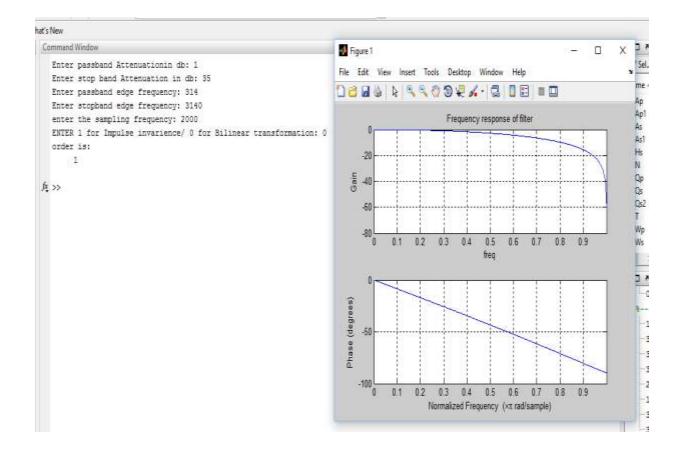


I. LOW PASS CHEBYSHEV IIR FILTER DESIGN USING BILINEAR TRANSFORMATION (BLT) METHOD

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 -1Stop band attenuation in db -35Sampling frequency -2000 Hz



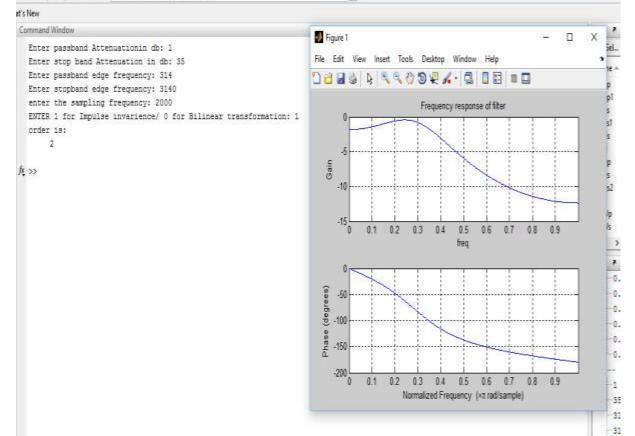
II. LOW PASS CHEBYSHEV IIR FILTER DESIGN USING IMPULSE INVARIANT TECHNIQUE (IIT) METHOD

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 -1Stop band attenuation in db -35Sampling frequency -2000 Hz 🕴 🛐 📃 🦉 Current Folder: C-(Users),User\Docurrents\MATLA8

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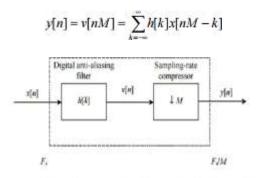


Write a MATLAB code to illustrate the effect of Decimation and Interpolation by an integer factor. Plot the magnitude spectrum. Design the necessary filter to overcome aliasing and image frequencies after decimating and interpolating the signal respectively.

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

I. DECIMATION

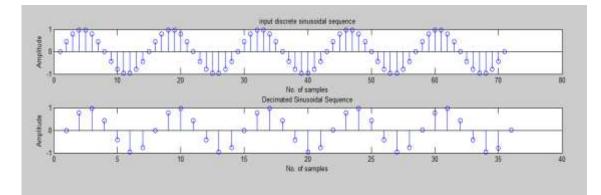
Decimation can be regarded as the discrete-time counterpart of sampling. Whereas in sampling we start with a continuous-time signal x(t) and convert it into a sequence of samples x[n], in decimation we start with a discrete-time signal x[n] and convert it into another discrete-time signal y[n], which 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.



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.

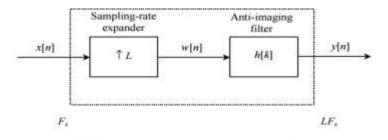
Signal specifications: Input frequency – 10 Hz Sampling frequency – 140 Hz Decimation factor – 2



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.

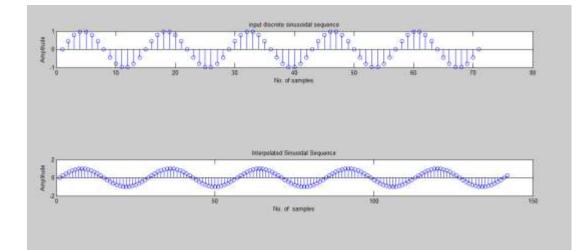
$$y[n] = L \sum_{k=-\infty}^{\infty} h[k] w[n-k]$$



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.

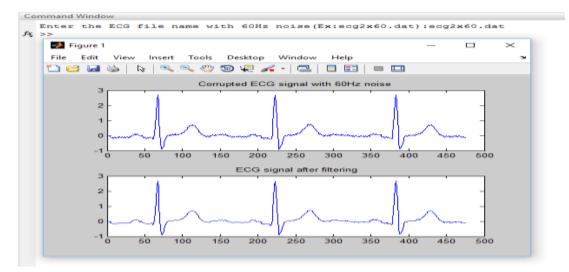
Signal specifications: Input frequency – 10 Hz Sampling frequency – 140 Hz Interpolation factor – 2

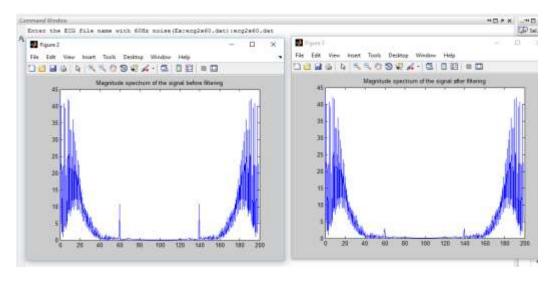


Read the data file named ecg2x60.dat from <u>http://people.ucalgary.ca/~ranga/enel563/SIGNAL_DATA_FILES/</u> that is corrupted with the 60Hz noise component. Write a MATLAB code to remove this 60Hz noise component from the signal using Notch filter and LMS adaptive filter. Plot the magnitude spectrum 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.

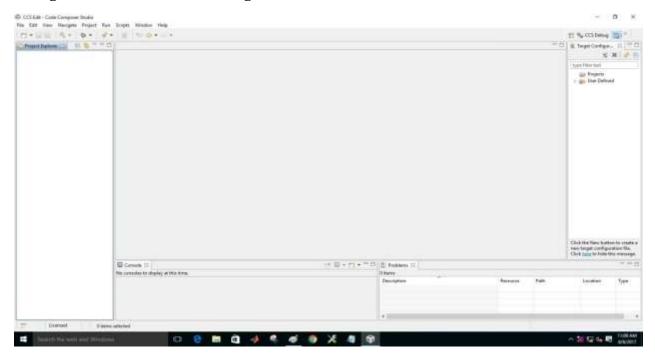
Least mean squares (LMS) algorithms are a class of adaptive filter used to mimic a desired filter by finding the filter coefficients that relate to producing the least mean square of the error signal (difference between the desired and the actual signal). It is a stochastic gradient descent method in 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.





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.



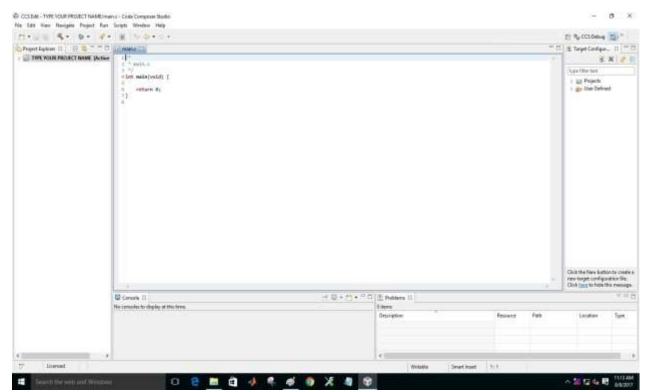
Step-2: Click on File \rightarrow New \rightarrow CCS Project

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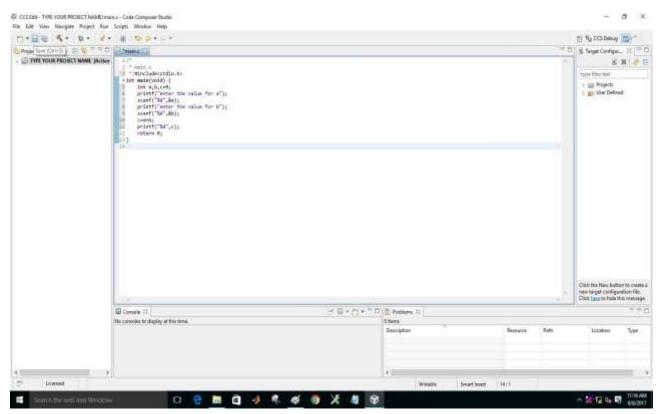
Step-3: Type your Project name, select Output type as Executable from the dropdown menu, Device Family as C6000, Variant as DSK6713, Connection as Spectrum Digital DSK EVM eZdsp USB Emulator, click on Empty projects (with main.c) & finish.

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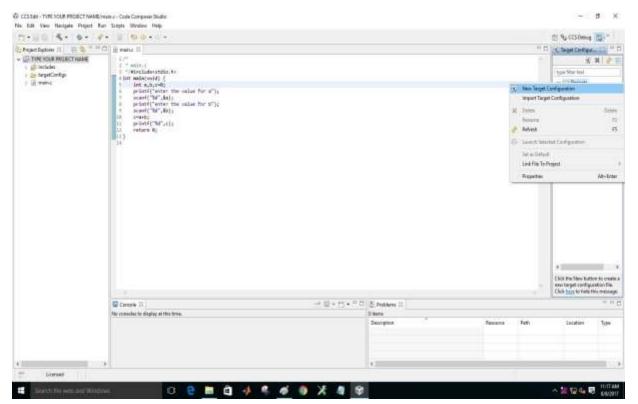
Step-4: Screen as shown below will appear with the Project Name.



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



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



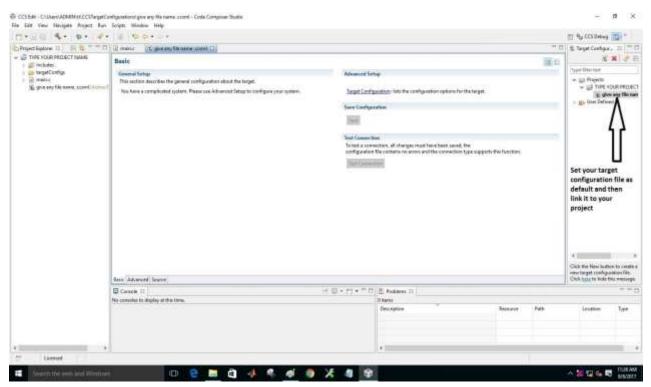
Step-7: Give the file name with .ccxml extension and click finish.

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

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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

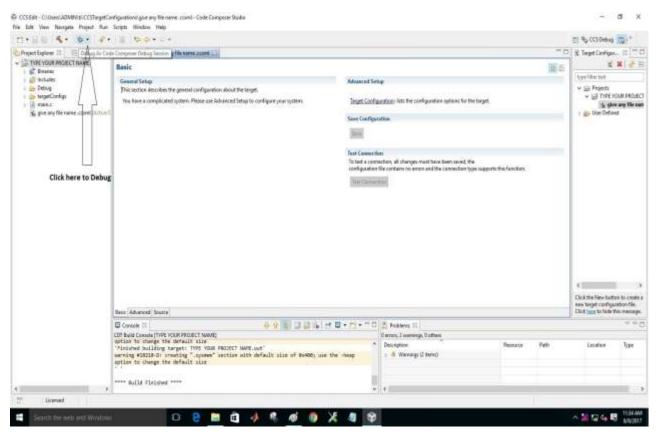
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Step-11: Right click on the Project name and select Build project or click on the symbol as indicated by the arrow mark.

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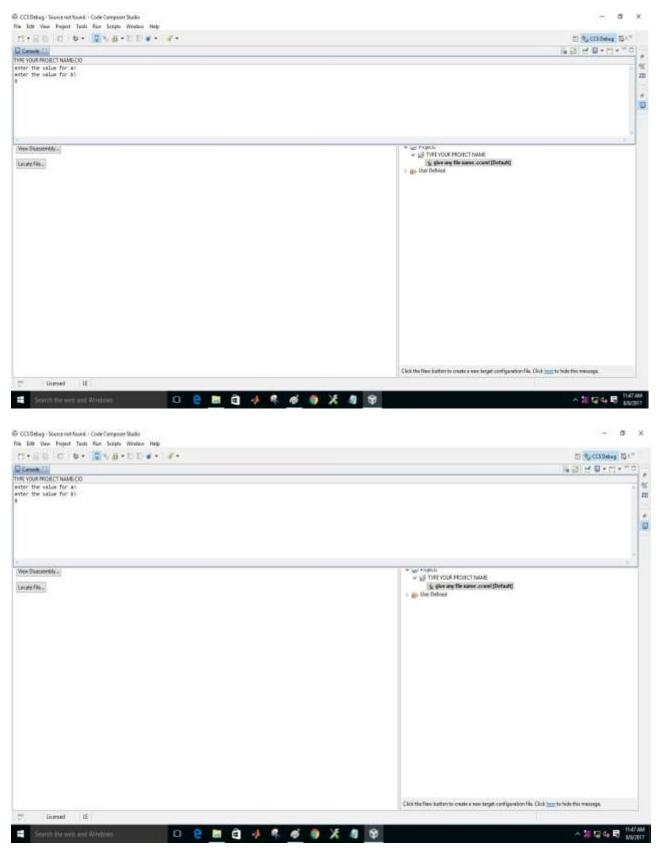
Step-12: Click on the symbol shown to debug your code.



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

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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:

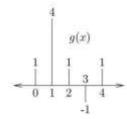
$$g(x) = \sum_{s=-\infty}^{\infty} f(s) h(x-s) = f(x) * h(x)$$

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:

 $\begin{array}{l} g(0)=f(-1)h(1)+f(0)h(0)=0+1=1\\ g(1)=f(0)h(1)+f(1)h(0)=1+3=4\\ g(2)=f(1)h(1)+f(2)h(0)=3+-2=1\\ g(3)=f(2)h(1)+f(3)h(0)=-2+1=-1\\ g(4)=f(3)h(1)+f(4)h(0)=1+0=1 \end{array}$

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:

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

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 $h((s - x) \mod M)$ that line up vertically are multiplied and summed:

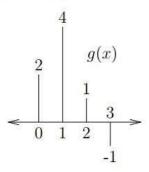
$$g(0) = f(3)h(1) + f(0)h(0) = 1 + 1 = 2$$

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

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

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

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.

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

$$\begin{aligned} r_{xy}(l) &= \sum_{n=-\infty}^{\infty} x(n)y(n-l) & l = 0, \pm 1, \pm 2, \dots, \\ r_{xy}(l) &= \sum_{n=-\infty}^{\infty} x(n+l)y(n) & l = 0, \pm 1, \pm 2, \dots, \end{aligned}$$
The cross correlation between $y(n)$ and $x(n)$ is defined as

$$\begin{aligned} r_{yx}(l) &= \sum_{n=-\infty}^{\infty} x(n-l)y(n) & l = 0, \pm 1, \pm 2, \dots, \\ r_{yx}(l) &= \sum_{n=-\infty}^{\infty} x(n)y(n+l) & l = 0, \pm 1, \pm 2, \dots, \end{aligned}$$
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) = [2124] and y(n) = [2124] and plot it.

Output: (l) = [8, 8, 12, 25, 12, 8, 8]

Correlation Coefficient=1

The autocorrelation of x(n) is defined as

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

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

Output: (l) = [10, 1, 4, 15, 40, 15, 4, 1, 10]

(b) As defined in Experiment 4 (a)