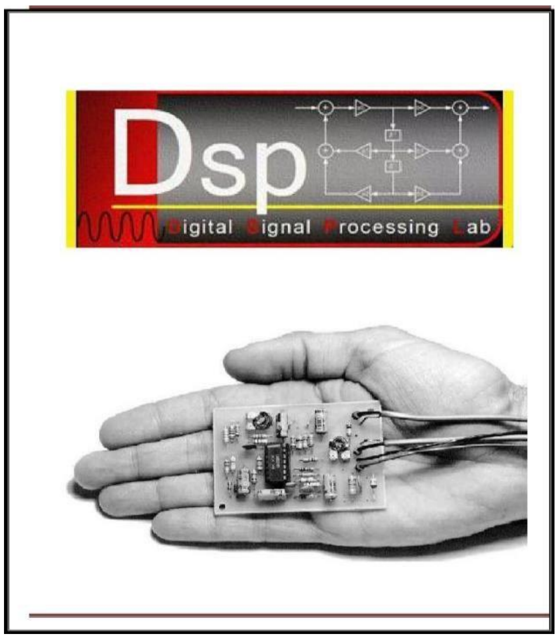
MARRI LAXMAN REDDY
INSTITUTE OF TECHNOLOGY AND MANAGEMENT
(AN AUTONOMOUS INSTITUTION)
(Approved by AICTE, New Delhi & Affiliated to JNTUH, Hyderabad)
Accredited by NBA and NAAC with 'A' Grade & Recognized Under Section2(f) & 12(B)of the UGC act,1956

Department of Electronics & Communication Engineering

LABORATORY MANUAL
DIGITAL SIGNAL PROCESSING

MLRS-R24



III B.TECH(ECE) - II Semester

Academic Year: 2026-2027

Prepared by
Mrs.P.Sandhya, Asst. Professor
Mr.A.Anil Kumar, Asst. rofessor

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DEPARTMENT OF ELECTRONICS AND COMMUNICATION ENGINEERING

CERTIFICATE

This is to certify that this manual is a bonafide record of practical work carried out in the Digital Signal Processing **Design Laboratory** for the **B.Tech (Electronics and Communication Engineering) VI Semester** Programme during the academic year **2025–2026.**

This manual has been prepared by **Mr. A.Anil Kumar (Assistant Professor), Mrs. P.Sandhya(Assistant Professor),** Department of Electronics and Communication Engineering, with my/our own efforts and to the best of our knowledge.

Signature of Lab Faculty **Signature of HOD**

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DEPARTMENT OF ELECTRONICS AND COMMUNICATION ENGINEERING

PREFACE

This laboratory lays the foundation for the Electronics and Communication Engineering students during Fourth year of their course.

Design of digital circuits in ECAD Lab can be divided into 2 groups: Cycle 1 and Cycle 2. In Cycle -I, students will know how to write Verilog programming of digital circuits and simulate using Xilinx software tool.

In cycle -II, students will design the circuits of digital system and implement them in Cadence tool. After performing all the experiments included in this Laboratory, it is hoped the student receives good training to handle any electronic equipment available in electronics field.

By,
Mrs.P.Sandhya
Mr. A.Anil Kumar

DEPARTMENT OF ELECTRONICS AND COMMUNICATION

ACKNOWLEDGEMENT

It was really a good experience, working at Digital Signal Processing Lab. First, I would like to thank Dr. N. Srinivas, Professor, Department of Electronics and Communication Engineering, Marri Laxman Reddy Institute of technology & Management for giving the technical support in preparing the document.

I express my sincere thanks to Dr. N. Srinivas, Head of the Department of Electronics and Communication Engineering, Marri Laxman Reddy Institute of technology & Management, for his concern towards me and gave me opportunity to prepare Concrete Technology laboratory manual.

I am deeply indebted and gratefully acknowledge the constant support and valuable patronage of Dr.Ravi Prasad, Dean Academics, Marri Laxman Reddy Institute of technology & Management. I am unboundedly grateful to him for timely corrections and scholarly guidance.

I express my heartful thanks to Dr.P.Sridhar, Director, and Dr. R.Murali Prasad, Principal, Marri Laxman Reddy Institute of technology & Management, for giving me this wonderful opportunity for preparing the Concrete Technology laboratory manual.

At last, but not the least I would like to thank the entire ECE Department faculties those who had inspired and helped me to achieve my goal.

By,
Mrs.P.Sandhya
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DEPARTMENT OF ELECTRONICS AND COMMUNICATION
GENERAL INSTRUCTIONS

1. Students are instructed to come to DIGITAL SIGNAL PROCESSING LABORATORYLABORATORYon time. Late comers are not entertained in the lab.

2. Students should be punctual to the lab. If not, conducted experiments will not be repeated.

3. Students are expected to come prepared at home with the experiments which are going to performed.

4. Students are instructed to display their identity cards and apron before entering into the lab.

5. Students are instructed not to bring mobile phones to the lab.

6. The equipment's and other accessories used in Digital Signal Processing lab should be handled with care and responsibility.

7. Any damage to the equipment's during the lab session is student's responsibility and penalty or fine will be collected from the student.

8. Students should update the records and lab observation books session wise. Before leaving the lab, the student should get his lab observation book signed by the faculty.

9. Students should submit the lab records 2/3 days in advance to the concerned faculty members in the staffroom for their correction and return.

10. Students should not move around the lab during the lab session.

11. If any emergency arises, the student should take the permission from faculty member concerned in written format.

12. The faculty members may suspend any student from the lab session on disciplinary grounds.

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SAFETY MEASURES

1. While working in the laboratory suitable precautions should be observed to prevent accidents.

2. Always follow the experimental instructions strictly.

3. Use the first aid box in case of any accident/mishap.

4. Never work in the laboratory unless a demonstrator or teaching assistant in present.

5. When the experiment is completed, students should disconnect the setup made by them, and should return all the components/instruments taken for the purpose.

DEPARTMENT OF ELECTRONICS AND COMMUNICATION

VISION & MISSION OF THE INSTITUTE

Vision of the Institute:

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

Mission of the Institute:

To foster a transformative learning environment that empowers students to excel in engineering, innovation, and leadership.

To produce skilled, ethical, and socially responsible engineers who contribute to sustainable technological advancements and address global challenges.

To shape future leaders through cutting-edge research, industry collaboration, and community engagement.

VISION & MISSION OF THE DEPARTMENT

Vision of the Department:

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

Mission of the Department:

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

To achieve the Mission the department will:

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

Program Educational Objectives (PEOs)

PEO 1	**Professional Excellence** Analyze, design, build, maintain, or improve ELECTRONICS AND COMMUNICATION ENGINEERING-based systems, considering environmental, economic, and societal requirements.
PEO 2	**Multidisciplinary Approach** Develop a strong educational foundation to design and conduct experiments, meeting needs within multidisciplinary constraints such as economic, environmental, social, political, ethical, health and safety, manufacturability, and sustainability, while analyzing and interpreting data.
PEO 3	**Continued Self-Learning** Identify, formulate, and solve engineering problems, and engage in lifelong learning in advanced areas of ELECTRONICS AND COMMUNICATION ENGINEERING and related fields.
PEO 4	**Effective Contribution to Society** Utilize modern engineering techniques, skills, and tools necessary for ELECTRONICS AND COMMUNICATION ENGINEERING practice, serving the community as ethical and responsible professionals.

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DIGITAL SIGNAL PROCESSING LABORATORYLABORATORYLABORATORY

Virtual lab details

Name of the Virtual Lab:
Virtual Lab Host Institute:
URL/Link to Lab
Academic Year
Semester

List of Experiments Available in Virtual Lab
List of Experiments:

1. Generation of sinusoidal waveform /signal based on recursive difference equations.
2. Histogram of white gaussian noise and uniformly distributed noise.
3. Find DFT/IDFT of given DT signal.
4. Find frequency response of a given system in Transfer function /Differential equation form.
5. Obtain Fourier series coefficients by formula and compare for half sine wave.
6. Implement of FFT of given sequence.
7. Determine of power spectrum of a given signal.
8. Implement of LP FIR filter for a given sequence.
9. Implement of HP FIR filter for a given sequence.

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DEPARTMENT OF Electronics and Communication ENGINEERING
DIGITAL SIGNAL PROCESSING

LAB PLANNER

S.No	Experiment	CO	Virtual Lab Availabilty	Date planned	Date conducted
1	Generation of sinusoidal waveform /signal based on recursive difference equations.	CO1		03.12	04)12
2	Histogram of White Gaussian Noise and Uniformly Distributed Noise.	CO1		10.12	
3	To find DFT/IDFT of given DT signal.	CO2		17.12	
4	To find frequency response of a given system (Transfer function /Differential equation form).	CO2		24.12	
5	Implementation of FFT of given sequence.	CO2		31.12	
6	Determination of power spectrum of a given signal.	CO3		07.01	
7	LAB INTERNAL-1			21.01	
8	Implementation of LP FIR filters for a given sequence.	CO3		28.01	
9	Implementation of HP IIR filters for a given sequence	CO3		11.02	
10	Implementation of Decimation process.	CO4		18.02	
11	Implementation of Interpolation process.	CO4		25.02	
12	Implementation of I/D sampling rate converters.	CO5		11.03	
13	Impulse response of first order and second order systems.	CO5		18.03	
14	MID-II			25.03	

DEPARTMENT OF ELECTRONICS AND COMMUNICATION ENGINEERING

DIGITAL SIGNAL PROCESSING

LAB PLANNER

Date planed	**10/7**			**17/7**			**24/7**																									
Date conducted																																
Roll Number	Exp No	CO	VL*	Exp No	CO	VL*	Exp No	CO	VL*	Exp No	CO	VL*	Exp No	CO	VL*		Exp No	CO	VL*	Exp No	CO	VL*	Exp No	CO	VL*	Exp No	CO	VL*	Exp No	CO	VL*	
237Y1A 0401	1	1	Y	2	1	N	3	2	Y	4	2	N	5	3	Y		6	4	Y	7	4	Y	8	4	Y	9	5	N	10	5	Y	
237Y1A 0402	2	1	N	3	2	Y	4	2	N	5	3	Y	6	4	Y	M	7	4	Y	8	4	Y	9	5	N	10	5	Y	1	1	Y	**M**
237Y1A 0403	3	2	Y	4	2	N	5	3	Y	6	4	Y	7	4	Y	I	8	4	Y	9	5	N	10	5	Y	1	1	Y	2	1	N	**I**
237Y1A 0104	4	2	N	5	3	Y	6	4	Y	7	4	Y	8	4	Y	D	9	5	N	10	5	Y	1	1	Y	2	1	N	3	2	Y	D
237Y1A 0405	5	3	Y	6	4	Y	7	4	Y	8	4	Y	9	5	N	-	10	5	Y	1	1	Y	2	1	N	3	2	Y	4	2	N	**-**
237Y1A 0406	6	4	Y	7	4	Y	8	4	Y	9	5	N	10	5	Y	I	1	1	Y	2	1	N	3	2	Y	4	2	N	5	3	Y	**I I**

Note:VL*-Virtual Lab Availabilty

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DIGITAL SIGNAL PROCESSING

RUBRICS USED TO ASSESS LEARNINGS IN LABORATORIES

1. RUBRICS FOR DAY TO DAY EVALUATION

Parameter	Max Marks	Level-1 (Very Poor)	Level-2 (Poor)	Level-3 (Average)	Level-4 (Good)	Level-5 (Excellent)
Observation Book	05	No observations or irrelevant data. (0–1)	Incomplete or incorrect data. (2)	Basic values with some errors. (3)	Mostly correct with good format. (4)	Fully correct, clear, and well-formatted. (5)
Record Writing	05	Not submitted. (0-1)	Submitted but mostly incomplete. (2)	Submitted with some missing/wrong parts. (3)	Submitted with minor issues. (4)	Fully complete, correct algorithm & flowchart. (5)
Result	05	No result or major errors. (0–1)	Result partially obtained. (2)	Acceptable result with limited error. (3)	Near-correct result and reasonable error. (4)	Accurate result. (5)
Viva-Voce	05	Did not answer any questions. (1)	Answered very few questions. (2)	Answered some questions with help. (3)	Answered most questions correctly. (4)	Answered all questions accurately. (5)

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DIGITAL SIGNAL PROCESSING

2.RUBRICS FOR INTERNAL EVALUATION

Criterion	Max Marks	Level-1 *(Very Poor)*	Level-2 *(Poor)*	Level-3 *(Average)*	Level-4 *(Good)*	Level-5 *(Excellent)*
Design/Tool/Apparatus Selection	2 Marks	Incorrect tool/design and no reasoning. **(0)**	Tool/design selection attempted with unclear logic. **(0.5)**	Satisfactory selection with partial justification. **(1)**	Correct selection and proper analysis with few errors. **(1.5)**	Smart selection with accurate, relevant analysis. **(2)**
Execution (Code/Debug/Run) /Analysis/Method Used	4 Marks	Did not attempt or completely failed to execute. **(0)**	Attempted but unable to proceed or with major errors. **(1)**	Partial execution with some logic/syntax errors. **(2)**	Mostly correct execution with minimal help. **(3)**	Fully correct and independently executed program. **(4)**
Results& Documentation	2 Marks	Incomplete or poorly presented. **(0)**	Basic structure but lacks clarity or formatting. **(0.5)**	Complete but generic or with formatting issues. **(1)**	Well-structured and mostly clear. **(1.5)**	Well-organized, professional, and engaging documentation. **(2)**
Viva-Voce (Understanding of Concepts)	2 Marks	No understanding; could not answer questions. **(0)**	Answered a few with difficulty. **(0.5)**	Answered half the questions with basic clarity. **(1)**	Good understanding with confident answers. **(1.5)**	Answered all questions with clarity and depth. **(2)**

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DIGITAL SIGNAL PROCESSING

3. RUBRICS FOR SEMESTER END EXAMINATIONS

Criterion	Max Marks	Level-1 (Very Poor (*0–2 marks)*	Level-2 (Poor) *(3–4 marks)*	Level-3 (Average) *(5–6 marks)*	Level-4 (Good) *(7–9 marks)*	Level-5 (Excellent) *(10–12 marks)*
Preparedness for the Experiment	12 marks	No clarity on objective or procedure. Unable to explain basics.	Limited idea of the objective/procedure. Needed prompting.	Has basic understanding; minor gaps in concept or preparation.	Well-prepared, with clear understanding of steps and background.	Fully prepared with strong conceptual clarity and confident explanation.
Performance in the Laboratory	12 marks	Unable to perform experiment. Relied entirely on examiner's help.	Performed with multiple errors and constant support.	Performed with some errors; required occasional help.	Performed mostly independently with minimal support.	Performed independently, efficiently, and with precision.
Calculations & Graphs	12 marks	No or incorrect calculations. Graphs missing or irrelevant.	Multiple calculation errors. Graphs/plots inaccurate or poorly labeled.	Calculations partially correct. Graphs present but with some flaws.	Correct calculations and graphs with minor errors.	Accurate calculations and well-labeled graphs with proper interpretation.
Results & Error Analysis	12 marks	No result or invalid result. No error analysis attempted.	Incorrect result with vague or no error discussion.	Acceptable result. Error analysis attempted but limited.	Correct result with sound error discussion.	Accurate result with detailed and relevant error analysis.
Viva-Voce (Subject Knowledge)	12 marks	Unable to answer any questions. No conceptual	Answered few questions with poor logic.	Answered half of the questions with average understanding.	Answered most questions with clarity	Answered all questions with depth, clarity, and reasoning.

		understandin g.			and confidence.	

EXPERMENT NO-3

DFT/IDFT OF A SEQUENCE

AIM:-

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

SOFTWARE REQURIED:-

1. MATLAB R2010a.
2. Windows XP SP2.

THEORY:-

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

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

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

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

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

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

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

PROCEDURE:-

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

PROGRAM:-

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

%IDFT%

```
clc;
clear all;
close all;
a=input('enter the input sequence');
disp('the length of input sequence is');
N=length(a);
N
for n=1:N;
```

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

```
x=1/N*x(n);
disp('the output is');
x(n)
stem(n,abs(x));
grid;
xlabel('discrete time');
ylabel('magnitude');
title('magnitude response of the idft');
grid;
```

Validation of the Results :

VIVA QUESTIONS

S.No	Question	CO	Blooms Taxonomy
1	Define Signal	CO1	Understand
2			
3			
4			
5			
6			
7			
8			
9			
10			
11			
12			
13			
14			
15			
16			
17			
18			
19			
20			

Note :Each experiment should contain Minimum 20 Viva Questions

History and fundamental background of the lab:

DSP dates back to the very beginnings of the digital age, perhaps even a little bit before. If the construction of the first digital computer, ENIAC, in 1946, marks the beginning of the digital age in 1946, then DSP popped up a scant two years later. The IEEE published a monograph in 1998 titled "Fifty Years of Signal Processing: The IEEE Signal Processing Society and its Technologies 1948-1998," which marks the start of the DSP age in 1948 by calling it the DSP *annus mirabilis*. That's the year that Claude Shannon at Bell Telephone Laboratories published his landmark paper titled "A Mathematical Theory of Communication," that carved in stone the relationship between achievable bit rate, channel bandwidth, and signal-to-noise ratio.

It's also the year that Shannon, Bernard M. Oliver, and John R. Pierce – all at Bell Labs – published "The Philosophy of PCM," documenting the practical nature of pulse code modulation and putting the stamp of practicality on PCM, first envisioned by Alec Reeves in 1937. (Bernard Oliver is perhaps better known in wider circles as Barney Oliver, the brilliant man who founded HP Labs in 1966, but that's a different story entirely.) Shannon, Oliver, and Pierce were documenting some of the PCM concepts used to build the top secret SIGSALY secure speech system, a room-sized, 50-ton behemoth that encoded and encrypted the most important speech communications for the Allied forces during World War II.

Coincidentally, Bell Labs announced the development of the transistor on June 30, 1948, the same year it published the two landmark papers that sparked the DSP revolution. (The actual development of the transistor occurred the year before.) The transistor and solid- state electronics would be needed to transform the concepts in the papers published by Shannon, Oliver, and Pierce into practical technologies inexpensive enough to change the world of electronics, so 1948 was truly DSP's *annus mirabilis*.

After 1948, not much happened with DSP technology for a very long time. Digital electronics was too nascent a field for DSP to become practical, at least not for real-time signal processing. During that period, a lot of DSP involved manual entry of numbers into Friedan and Marchant mechanical calculators, which was wildly impractical for audio or video communications. The budding world of DSP awaited a critical development. Actually, several critical developments.

This is the story of how DSP and single-chip DSPs managed to take over the entire world of signal processing. It parallels the history of digital electronics itself, spanning the development of integrated circuits (ICs), microprocessors, DSPs, and FPGAs. Spoiler alert: FPGAs win big in the end.

A Few Shaky Steps

The first critical development required to make DSP practical was the invention of the IC. Nearly simultaneously, Jack Kilby at Texas Instruments (TI) and Robert Noyce at Fairchild Semiconductor envisioned two wildly different ways to build the first integrated circuits. Kilby at TI filed for a patent first, in February 1959. Kilby had envisioned building multiple electronic components onto one bar of silicon and then using small gold bond wires to hook them together. He actually did build such a circuit in 1958 before filing for the patent. However, Kilby's intricate and terribly manual assembly process was completely impractical and unlikely to scale up for commercial volume production.

Noyce's idea, developed early in 1959, was to use photolithography, which Fairchild was already using to make silicon transistors, to image multiple electronic components on one die and then interconnect the components with a metal interconnect layer using the same photolithographic techniques. He left the details to Jean Hoerni, who developed the planar process that's been used to make ICs ever since. Noyce and Fairchild filed for patents on these ideas later than Kilby, but still in 1959.

A practical manufacturing method for making ICs was only the first of many critical developments needed. Early digital ICs were far too primitive and incorporated far too few transistors to seriously consider using them for practical DSP. That's because DSP involves an extremely esoteric concept called math. In particular, you need two critical mathematical operators – multiplication and addition – and you need to use lots and lots of these operations to perform DSP. Some of us became digital engineers so we could forget all about math. Not so with DSP engineering. When working with DSP, there's no escaping the math.

While the electronics world was awaiting sufficient semiconductor technology advancement to make DSP a practical technology, the rest of the world couldn't wait. The Bell System needed to develop methods to cram more voice capacity through its immense installed base of wires, and PCM was clearly the first step. In addition, the military's use of radar and sonar blossomed after World War II, and DSP was clearly the path to refining and improving the capabilities of those systems. Communications satellites, first envisioned in a paper written by Arthur C. Clarke in 1945, were going to need digital communications to punch through some horrendous signal-to-noise problems involved in sending signals to and receiving signals from earth orbit.

The World Was Ready, But The ICs Were Not.

While the DSP world waited for semiconductor technology to catch up, the signal-processing theoreticians did not. Binshu Atal and Manfred Schroeder at Bell Labs developed Adaptive Predictive Coding (APC) in 1967, making it possible to get moderately decent audio from a 4.8kbps bit stream.

Then Atal developed Linear Predictive Coding (LPC) for speech compression. Nearly simultaneously, Fumitada Imakura of Nagoya University and Shuzo Saito of NTT developed partial correlation (PARCOR) coding, which is a very similar algorithm. These new speech-processing algorithms naturally needed more computation – more multiplications and additions – making it increasingly apparent that specialized ICs would be needed to make DSP practical and cost-effective.

But speech running through bandwidth-limited telephone channels were not the only signals crying out for DSP. Radar and sonar signal-processing algorithms needed it too. Television signals, which are real bandwidth hogs, needed it. Every signal being generated and received could benefit from DSP, if only the technology were practical. If only it didn't require racks and racks of circuit boards stuffed with the medium-scale ICs that TI and a host of other vendors were selling in the 1960s.

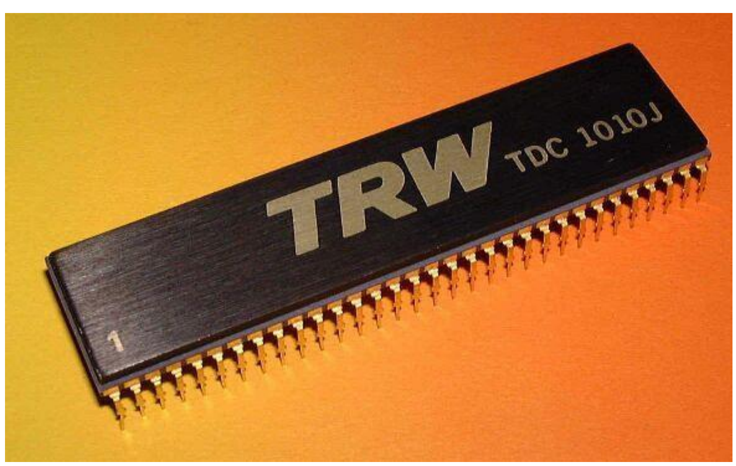
Intel's introduction of the first commercial microprocessor, the 4004, in 1971, was the first hint of what was to come. The Intel 4004 microprocessor could certainly multiply and add, but it could add only four bits at a time, and multiplication was a multi-step instruction sequence. The silicon was willing, but the ALU and bit width were weak.

The First DSP Chips Didn't Quite Cut It

TRW managed to create and market a 16×16-bit, single-chip digital multiplier – the MPY016H – in 1976, manufactured with a 1-micron bipolar process technology. The TRW MPY016H could multiply two 16-bit numbers to produce a 32-bit result in 45nsec (40nsec for the dash-1 part) but it couldn't add. You needed to add extra ICs to attach an accumulator to the multiplier. In addition, you could not extract the 32-bit result in one operation. You got the result in two chunks through the IC's 16-bit output port. So this product really wasn't a DSP. It was just part of a DSP. In addition, with two 16-bit input ports and one 16-bit output port, the TRW MPY016H had to be packaged in a wide, 64-pin DIP. It ran on 5V, but it needed nearly an amp to power up. At 5 watts, it needed a bit of cooling as well.

AMI introduced the S2811 Signal Processing Peripheral in 1978. It was a DSP with a 12-bit hardware multiplier, a 16-bit ALU, and a 16-bit output, but it was not designed as a single-chip DSP. AMI designed the S2811 as a memory-mapped peripheral device for the 8- bit 6800 microprocessor, which AMI also manufactured as an alternate source to the microprocessor's originator, Motorola Semiconductor. AMI's version of the 6800 microprocessor was called the S6800.

The 6800 microprocessor configured and accessed the AMI S2811 through one small and three larger on-chip, multiport RAMs. Although announced in 1978, the AMI S2811 was based on a difficult-to-manufacture VMOS process technology that delayed its arrival by several years. By then, several single-chip DSPs had been announced; the 16-bit microprocessor generation had arrived with the introduction of the Intel 8088, the Zilog Z8000, and the Motorola 68000; and the market for 6800 microprocessor peripherals began to shrink rapidly. Consequently, the obsolete AMI S2811 never achieved commercial success.

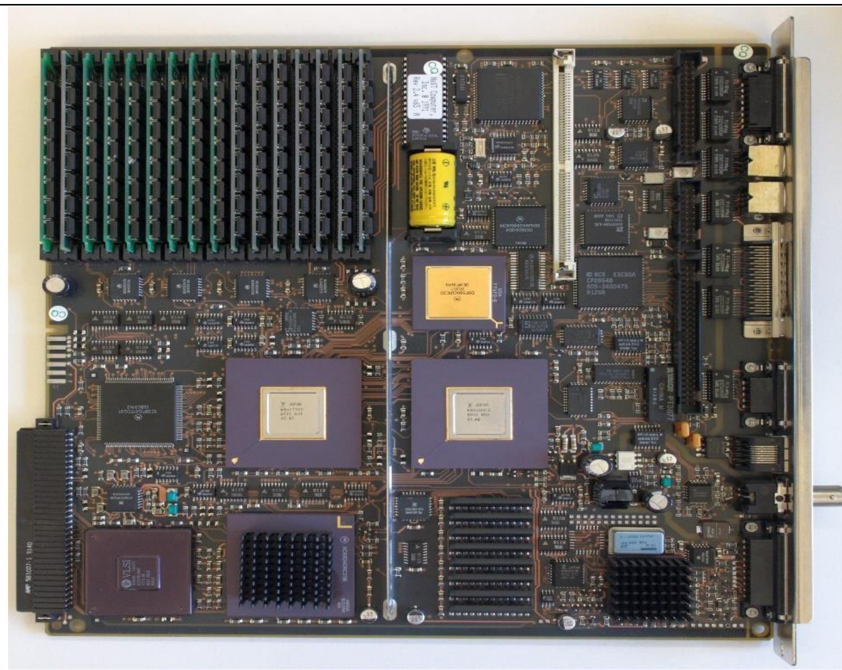


The same year that AMI introduced the S2811 Signal Processing Peripheral, TI introduced consumers to a toy based on DSP, the battery-powered "Speak & Spell," which implemented LPC as its core speech-encoding technology. The Speak & Spell toy incorporated a TI TMC0280 speech synthesizer chip, which implemented Binshu Atal's LPC algorithm in hardware. The TO TMC0280 was essentially a dedicated DSP.

Although the semiconductor technology of the day limited the TI Speak & Spell's vocabulary to 165 words, the toy's sparse vocabulary was a giant technological leap for a child's toy, even at the steep (for the time) $50 retail price. Although the TI TMC0280 was a specialized, dedicated speech DSP, its low cost and its ability to run for quite a while on a battery pointed the way to DSP ICs soon to come.

In February, 1979, Intel attempted to say "Yeah, we can do that" by announcing the Intel 2920 "Analog Signal Processor." This oddball integrated DSP had a 9-bit ADC (8 bits plus sign) and a four-input analog multiplexer on the front end, a 9-bit DAC with an 8-channel analog sample-and-hold circuit and analog multiplexer on the back end, and a digital ALU in the middle capable of performing addition, subtraction, and absolute-value operations to produce 25-bit results. A lack of multiplication and division instructions forced the use of multiple-instruction sequences to perform these required DSP math operations. On the order of 12 instructions were required per multiplication operation, and 14 instructions were needed for a divide operation. Each Intel 2920 instruction needed about half a microsecond to execute, so multiplication and division operations took microseconds to execute.

The Intel 2920 was intended for signal-filtering applications, but its slow execution speed, limited data path, unique instruction set, lack of a hardware multiplier, limited analog input and output voltage range, and other severe limitations doomed the IC to commercial failure. Consequently, few people remember the Intel 2920, but it too was a harbinger of DSPs to come.



As the 1970s ended, the world was clearly ready, hungry even, for real single-chip DSPs. Thanks to the theoreticians, the algorithms were developed and ready. Many signal- processing applications were begging for capable DSP silicon. All that remained was to develop the chip designs and the process technologies that could support the requirements. AMI, AT&T, Intel, Matsushita, Motorola, NEC, TI, Analog Devices, and others were all working feverishly on the problem. An explosion of DSP chips was imminent.

MARRI LAXMAN REDDY
INSTITUTE OF TECHNOLOGY AND MANAGEMENT
(AN AUTONOMOUS INSTITUTION)
(Approved by AICTE, New Delhi & Affiliated to JNTUH, Hyderabad)
Accredited by NBA and NAAC with 'A' Grade & Recognized Under Section2(f) & 12(B)of the UGC act,1956

Department of Electronics & Communication Engineering

DIGITAL SIGNAL PROCESSING LABORATORYLABORATORYLAB

List of Experiments:

1. Generation of sinusoidal waveform /signal based on recursive difference equations.
2. Histogram of white gaussian noise and uniformly distributed noise.
3. Find DFT/IDFT of given DT signal.
4. Find frequency response of a given system in Transfer function /Differential equation form.
5. Obtain Fourier series coefficients by formula and compare for half sine wave.
6. Implement of FFT of given sequence.
7. Determine of power spectrum of a given signal.
8. Implement of LP FIR filter for a given sequence.
9. Implement of HP FIR filter for a given sequence.
10. Generate Narrow Band Signal through filtering.
11. Generate DTMF signals.
12. Implement Decimation process.
13. Implement Interpolation process.
14. Implementation of I/D sampling rate converters.
15. Impulse response of first order and second order systems.

NOTE: Minimum of 12 experiments to be conducted.

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Department of Electronics & Communication Engineering

INSTRUCTIONS TO THE STUDENT

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

INDEX

EXPERMENT NO-1

GENERATION OF SINUSOIDAL WAVEFORM /SIGNAL BASED ON RECURSIVE DIFFERENCE EQUATIONS

AIM: -

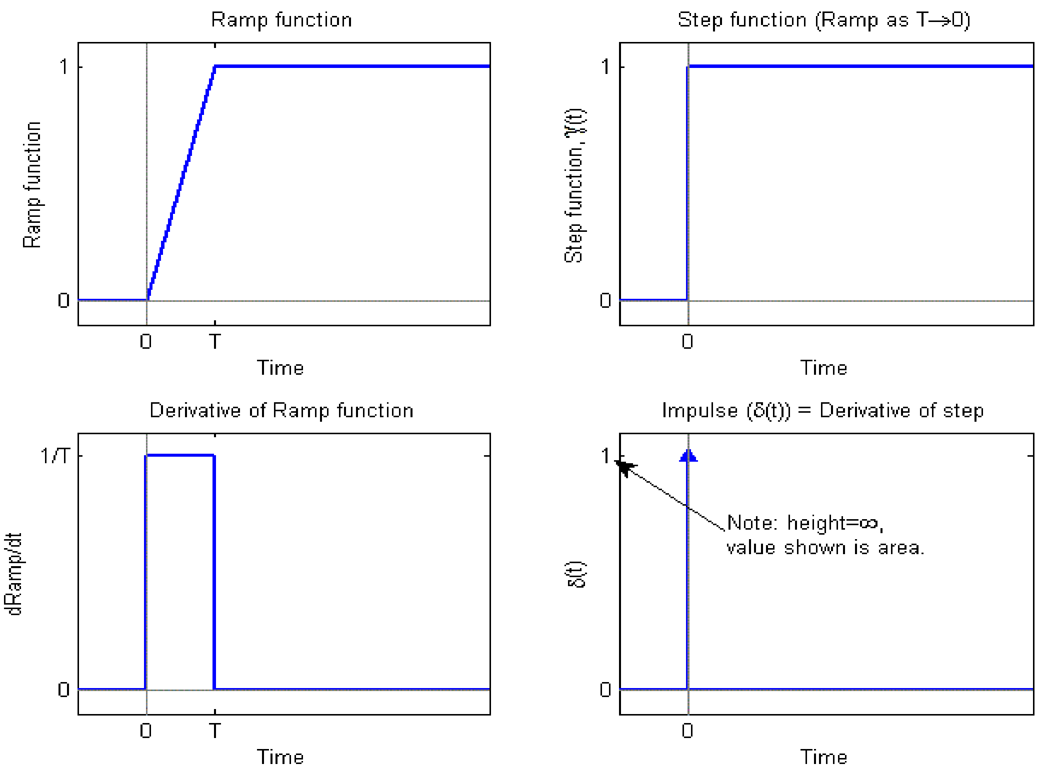
To write a "MATLAB" Program to generate various signals such as unit impulse, unit step, unit ramp, sinusoidal, exponential growing signal, exponential decaying signal, cosine signal and also generate sinusoidal signal based on recursive differential equation.

SOFTWARE REQURIED: -

1. MATLAB R2010a.
2. Windows XP SP2.

THEORY: -

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



Key Concept: Sifting Property of the Impulse

If b>a, then

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

Example: Another integral problem

Assume a<b, and evaluate the integral

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

Solution:

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

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

And

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

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

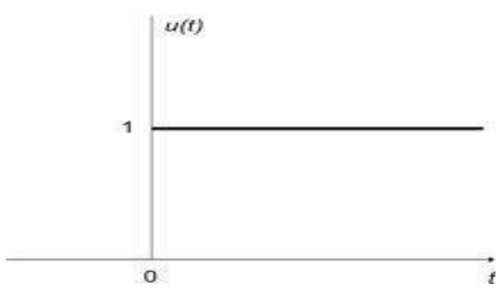
$$= \begin{cases} f(-T), & a < 0 < b \\ 0, & \text{otherwise} \end{cases}$$

Unit Step Function

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

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

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



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

Derivative

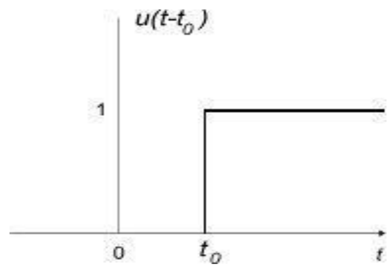
The unit step function is level in all places except for a discontinuity at t = 0. For this reason, the derivative of the unit step function is 0 at all points t, except where t = 0. Where t = 0, the derivative of the unit step function is infinite.

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

Integral

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

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



Sinusoidal Signal Generation

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

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

A sampled sinusoid may be written as:

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

PROCEDURE:-

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

PROGRAM:-
I. generate various signals such as unit impulse, unit step, unit ramp, sinusoidal, exponential growing signal,

%unit step signals%

```
clc;
clear all;
close all;
disp('unit step signals');
```

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

% sinusoidal signals%

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

% unit ramp signals%

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

% unit impulse signal %

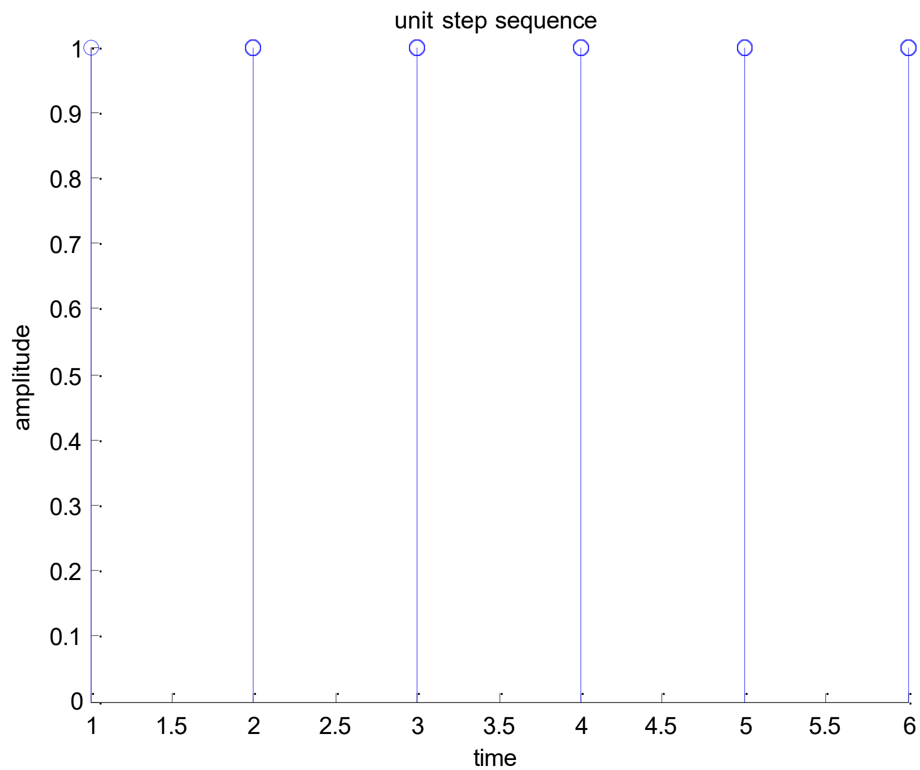
```
clc;
clear all;
close all;
disp('unit impuse signal');
N=input ('enter the no of samples');
n=-N:1:N;
x=[zeros(1,N) 1 zeros(1,N)];
stem(n,x);
xlabel('time');
ylabel('amplitude');
tittle('impulse sequence');
```

% exponentialsignals%

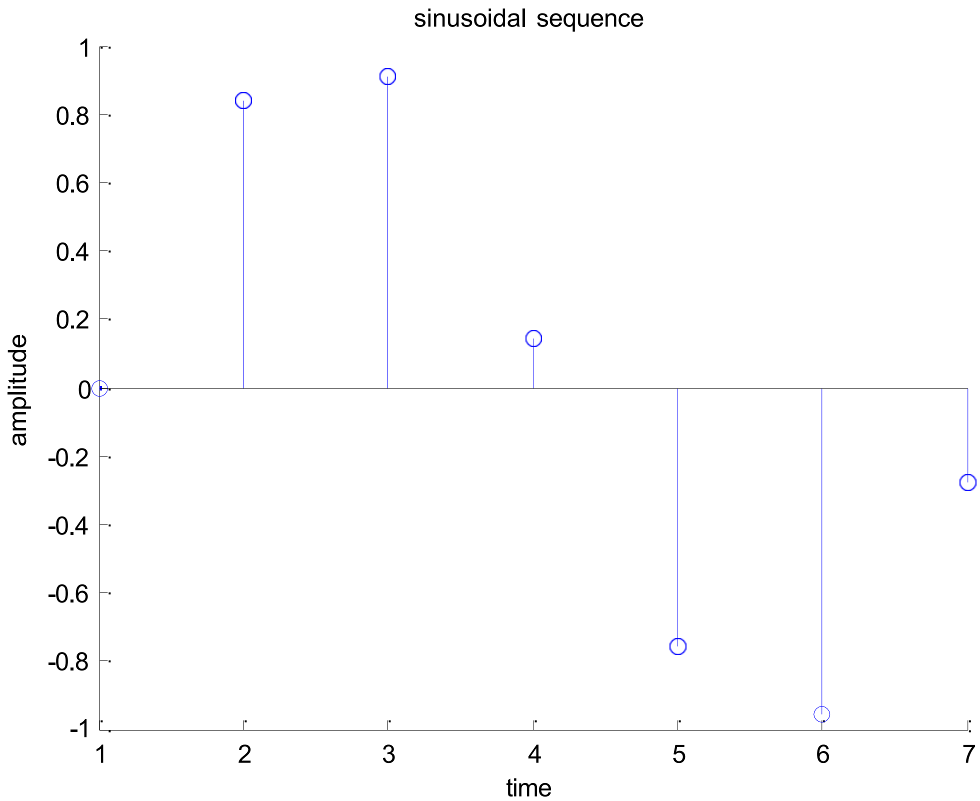
```
clc;
clear all;
close all;
disp('exponential signals');
N=input('enter the no of samples');
n=0:1:N;
n=-N:1:N;
a=0.5;
x=a.^n;
stem(x);
xlabel('time');
ylabel('amplitude');
title('exponential sequence');
```

OUTPUT:-

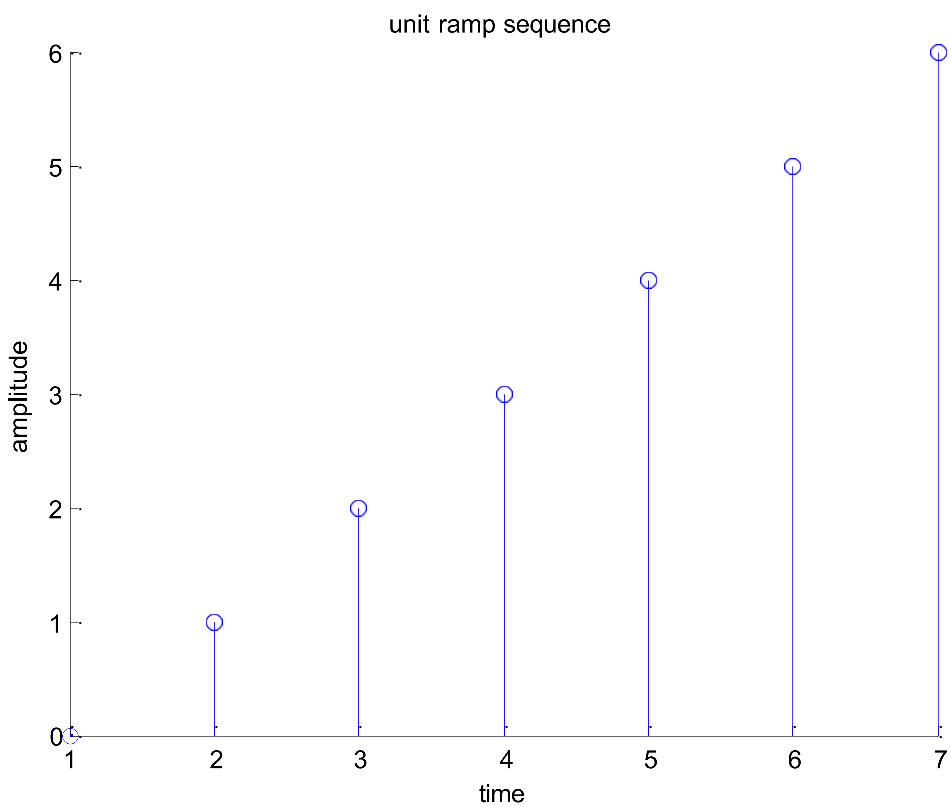
unit step signals
enter the no of samples6



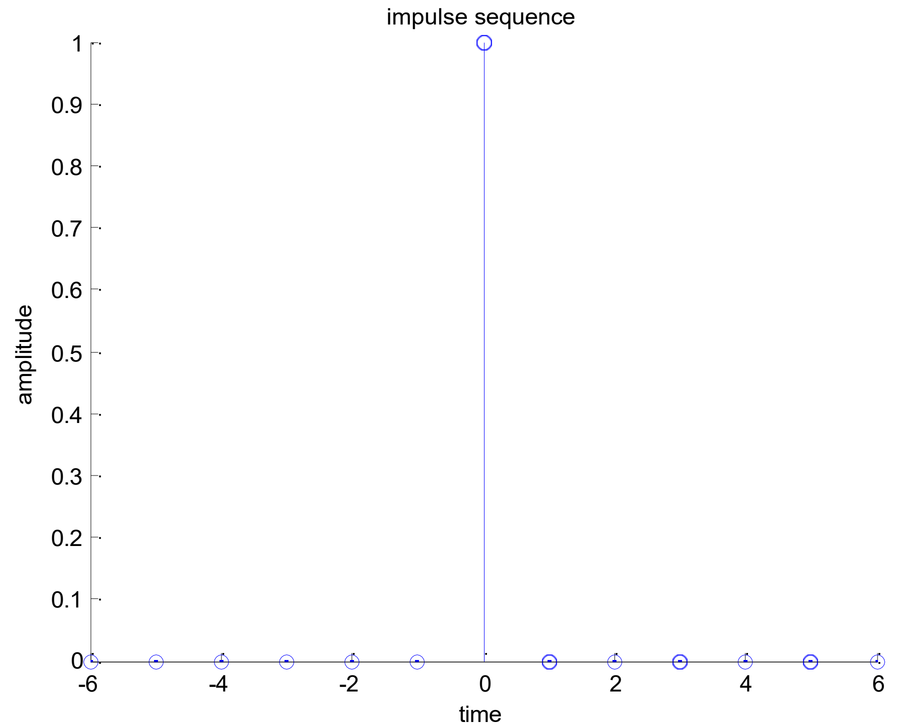
sinusoidal signals
enter the no of samples6



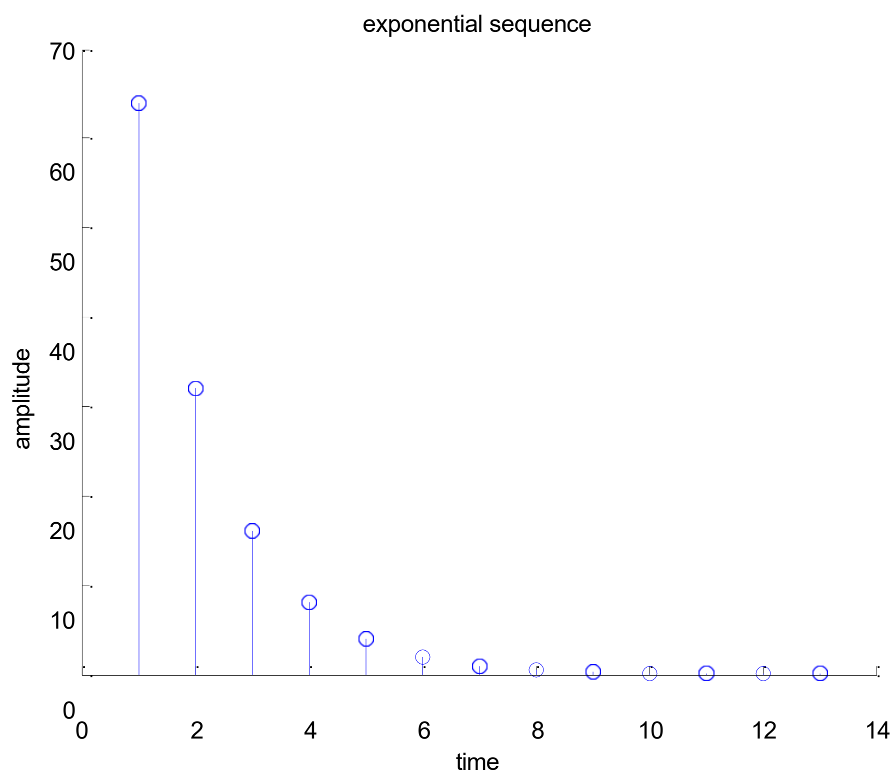
unit ramp signals
enter the no of samples6



unit impuse signal
enter the no of samples6



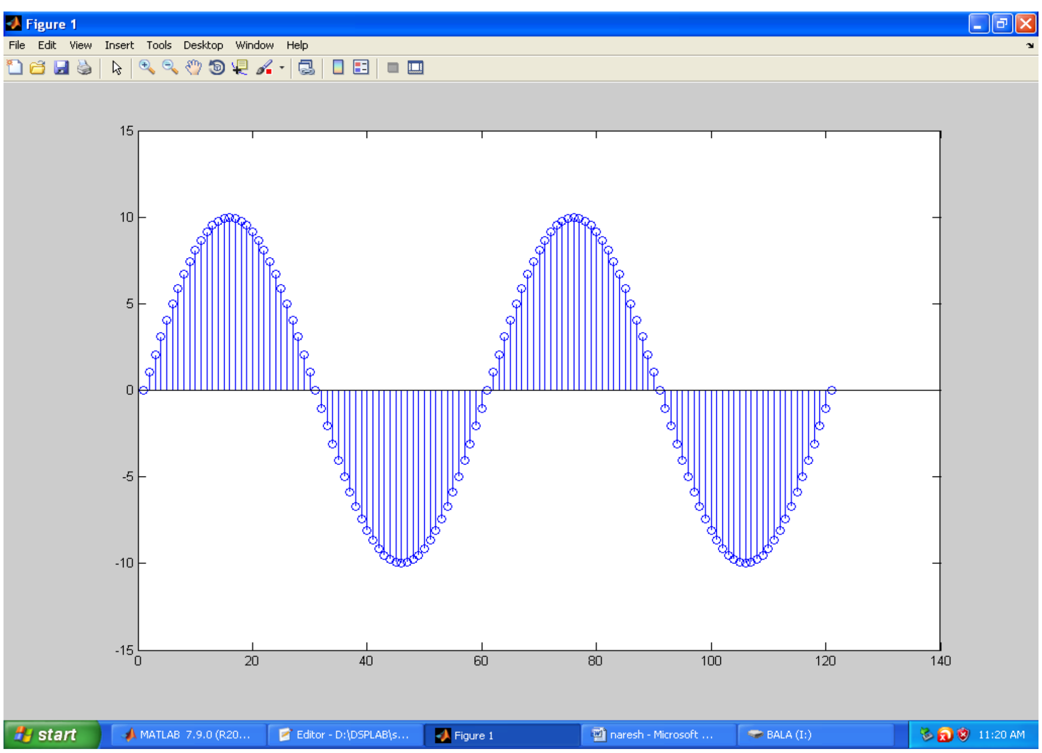
exponential signals
enter the no of samples6



II) Signal Based on Recursive Difference equation

```
clc;
clear all;
n=0:3;
y=10*sin(2*pi*n/60);
disp(y);
a=y(3)/y(2);
b=(y(4)-a*y(3))/y(2);
disp(a);
disp(b);
%a=[1.9754,-1];
%disp(a(1));

%y=[0 0.1564];
disp(y(3));
disp(y(4));
for k=1:1:119
   p(1)=y(1);
   p(2)=y(2);
      p(k+2)=a*p(k+1)+b*p(k);
      end
disp(p);
stem(p);
```

RESULT:



RESULT:-

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

Real Life Applications

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

EXERCISE PROGRAMS: -

1. Write program to get Discrete time Sinusoidal Signal?

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

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

4. Write Program for the following Function

   Y=exp (-2*∏*f*t)+exp(-8*∏*f*)

   Y= ((exp (-1.56∏f)*Sin(2∏f)+cos(2∏f)?
5. Write a mat lab program for generating u(n)-u(n-1)?
6. Write program to get Discrete time co-Sinusoidal Signal?
7. Write program to get Discrete time saw tooth Signal?
8. Write program to get Discrete time triangular Signal?
9. Write program to get addition of two sinusoidal sequences?
10. Write program to get exponential sequence?
11. Write program to get Discrete time Co-Sinusoidal Signal?
12. Write program to get Fourier Transform of Co-Sinusoidal Signal?
13. Write program to get Inverse Fourier Transform of Co-Sinusoidal Signal?
14. Write program to get exponential decaying sequence?

15. Write program to get exponential growing sequence?
16. Write program to get exponential decaying sequence?
17. Write program to get complex exponential sequence?
18. Write a mat lab program for generating sinc function?
19. Write a mat lab program for generating signum function?
20. Write a program to generate negative ramp signal?

VIVA QUESTIONS:-

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

EXPERMENT NO-2

HISTOGRAM OF WHITE GAUSSIAN NOISE AND UNIFORMLY DISTRIBUTED NOISE.

AIM:-

To write a MATLAB program for computation of statistical parameters of Gaussian distribution and uniform distribution.

SOFTWARE REQURIED:-

1. MATLAB R2010a.
2. Windows XP SP2.

THEORY:-

In random variable or random process if all the samples are clustered at center value, then it is Gaussian distribution. Instead, if all the samples are equally repeated then it is called uniform distribution. The histogram is analogous to probability density function. It plots the sample values with their iteration number. The Gaussian and uniform distributions differ in statistical averages such as mean, variance, standard deviation, skew, kurtosis, etc.,.

PROCEDURE:-

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

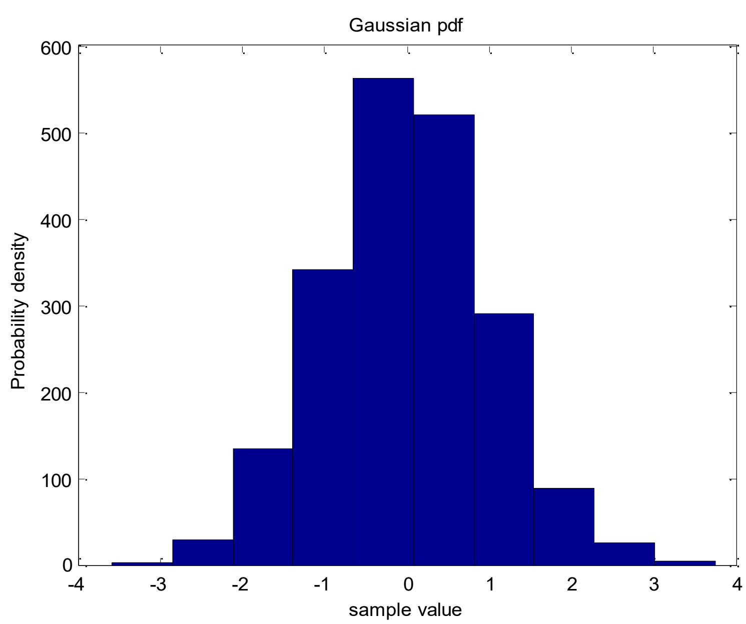
PROGRAM:-

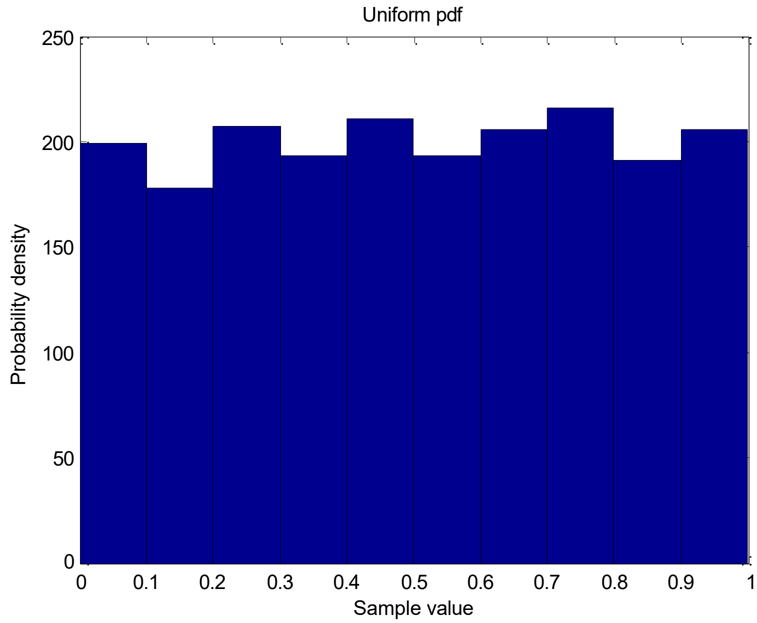
```
clc;
clear all;
close all;
x=randn(1,2000);
y=rand(1,2000);
hist(x);
xlabel('Sample value');
ylabel('Probability density');
title('Gaussian pdf');
figure;
hist(y);
xlabel('Sample value');
ylabel('Probability density');
title('Uniform pdf');
mx1=mean(x); display(mx1);
```

```
my1=mean(y); display(my1);
ux2=var(x); display(ux2);
uy2=var(y); display(uy2);
ux3=skewness(x); display(ux3);
uy3=skewness(y); display(uy3);
ux4=kurtosis(x); display(ux4);
uy4=kurtosis(y); display(uy4);
```

OUTPUT:-

```
mx1 =
   0.0220
my1 =
   0.5042
ux2 =
   1.0159
uy2 =
  0.0832
ux3 =
  -0.0329
uy3 =
  -0.0071
ux4 =
   3.1472
uy4 =
   1.8301
```





EXERCISE PROGRAMS

1. Write a MATLAB program to plot the co-sinusoidal signal.
2. Write a MATLAB program to subtract two sinusoidal signals.
3. Write a MATLAB program to subtract and multiply twosinusoidal signals.
4. Write a MATLAB program to right shift the signal to 5 times of the original signal.
5. Write a MATLAB program to left shift the signal to 8 times of the original signal.
6. Write a MATLAB program to add two different signals with 2 <t<5
7. Write a MATLAB program to shift a positive time line signal to negative timeline signal.
8. Write a MATLAB program to subtract co-sinusoidal signals.
9. Write a MATLAB program to subtract two sinusoidal signals
10. Write a MATLAB program to division and multiply two co-sinusoidal signals.
11. Write a MATLAB program to generate time scaling of a sequence.
12. Write a MATLAB program to generate time shifting of a sequence.
13. Write a MATLAB program to generate time folding of a sequence.
14. Write a MATLAB program to generate amplitude scaling of a sequence with amplitude 5.
15. Write a MATLAB program to generate time scaling of a sequence with time 2sec.
16. Write a MATLAB program to add two different signals with 4 <t<8
17. Write a MATLAB program to shift a negative time line signal to positive timeline signal.
18. Write a MATLAB program to subtract sinusoidal signals.
19. Write a MATLAB program to subtract and divide two sinusoidal signals
20. Write a MATLAB program to add and multiply two co-sinusoidal signals.
21. Write a MATLAB program to generate time scaling of a signal.
22. Write a MATLAB program to generate time shifting of a signal.
23. Write a MATLAB program to generate time folding of a signal.
24. Write a MATLAB program to generate amplitude scaling of a sequence with amplitude 10
25. Write a MATLAB program to generate time scaling of a sequence with time 5sec.

VIVA QUESTIONS:-

1. What is a noise ?
2. What is gaussion noise?
3. What is correlation ?
4. State Paeseval's energy theorem for a periodic signal?
5. What is Signum function?
6. How many types of correlation are there?
7. how many types of noises are there?
8. what is auto correlation?
9. what are the types of noise?
10. what is ramp signal?
11. what is signal
12. what is white noise?
13. what is periodic signal?
14. what is non periodic signal?
15. what is system bandwidth?
16. what is signal bandwidth?
17. what is causal system?
18. what is non causal system?
19. what is periodic signal?
20. what is non periodic signal?
21. what is signum function?
22. what is sinc function?
23. what are the types of correlation?
24. what is ramp function?
25. what is correlation ?
26. application of periodic function?
27. what is gaussian noise?
28. what is continuous time signal?
29. what is descrete time signal?
30. what is plot?
31. what is stem?
32. what is time delay?
33. what is xalbel?
34. what is y label?
35. what is psd?
36. what is random signal?
37. what is signal?
38. what are the types of signals?
39. what is bandwidth?
40. what is correlation?
41. what are the correlation types?
42. what is y label?
43. what is mean square?

44. what is periodi?
45. what is non periodic signal?
46. .what is gaussian noise?
47. what is continuous time signal?
48. what is descrete time signal?
49. what is plot?
50. what is time delay?

Realtime Applications:

- Stream processing
- Block processing
- Vector processing
- Digital processing applications

EXPERMENT NO-3

DFT/IDFT OF A SEQUENCE

AIM:-

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

SOFTWARE REQURIED:-

3. MATLAB R2010a.
4. Windows XP SP2.

THEORY:-

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

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

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

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

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

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

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

PROCEDURE:-

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

PROGRAM:-

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

%IDFT%

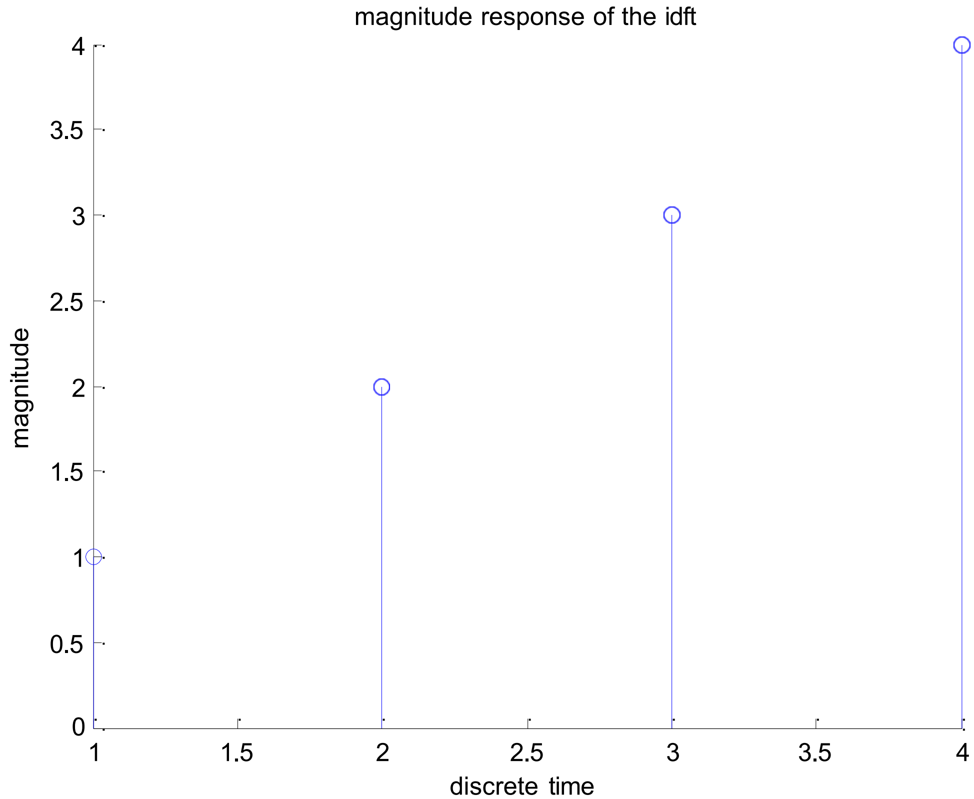
```
clc;
clear all;
close all;
a=input('enter the input sequence');
disp('the length of input sequence is');
N=length(a);
N
for n=1:N;
x(n)=0;
for k=1:N;
x(n)=x(n)+a(k)*exp((j*pi*2*(n-1)*(k-1)/N));
end;
end;
n=1:N;
```

```
x=1/N*x(n);
disp('the output is');
x(n)
stem(n,abs(x));
grid;
xlabel('discrete time');
ylabel('magnitude');
title('magnitude response of the idft');
grid;
```

OUTPUT:-

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

N =

    4

the output is

ans =

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



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

N =

    4

the output is

ans =
```

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



RESULT:-

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

EXERCISE PROGRAM:-

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

VIVA QUESTIONS:-

1. Define Symmetric and Anti-Symmetric Signals?
2. Define Continuous and Discrete Time Signals?
3. What are the Different types of representation of discrete time signals?
4. What are the Different types of Operation performed on signals?
5. Define DFT
6. How DFT can be calculated in matrix form?
7. Write the standard formula of DFT
8. What is the relation between DFT and DFS
9. What is the relation between DFT and Z-Transform
10. What is the relation between DFT and Laplasetransform.
11. Whether DFT is a linear transform?
12. What is the difference between circular convolution & linear convolution?
13. Can you implement linear convolution using circular convolution?
14. How FFT algorithms are classified?
15. How to calculate output of DFT using MATLAB?
16. Where DFT is used?
17. What is the difference between DFT and IDFT?
18. How to compute maximum length N for a circular convolution using DFT and IDFT.
19. Explain the function of twiddle factor?
20. Give the practical application DFT & IDFT?

21. Explain the significance of convolution.
22. Define linear convolution.
23. Why linear convolution is called as a periodic convolution?
24. Why zero padding is used in linear convolution?
25. What are the four steps to find linear convolution?
26. What is the length of the resultant sequence in linear convolution?
27. How linear convolution will be used in calculation of LTI system response?
28. List few applications of linear convolution in LTI system design.
29. Give the properties of linear convolution.
30. How the linear convolution will be used to calculate the DFT of a signal

REAL TIME APPLICATIONS:

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

EXPERMENT NO-4

FREQUENCY RESPONSE OF A GIVEN SYSTEM (TRANSFER FUNCTION /DIFFERENTIAL EQUATION FORM).

AIM: -

To study frequency response of second order system using MATLAB.

SOFTWARE REQURIED:-

1. MATLAB R2010a.
2. Windows XP SP2.

THEORY:-

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

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

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

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

PROCEDURE:-

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

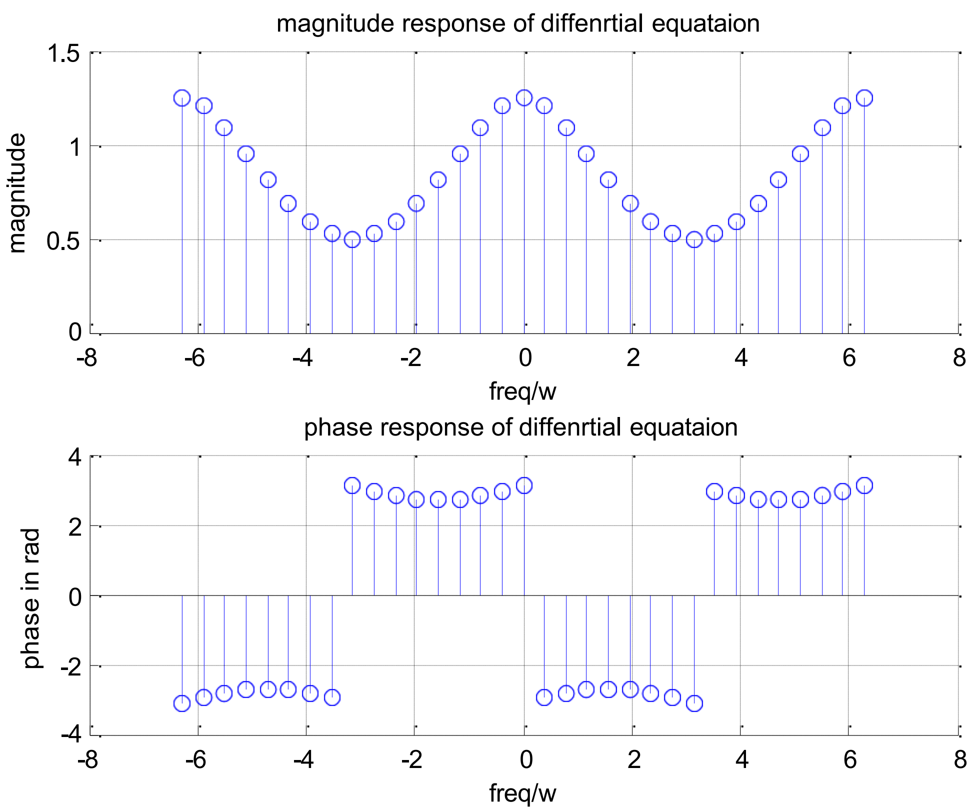
PROGRAM:-

%frequency response of differential equation %

```
clc ;
clear all;
b=[1,4];
a=[1,-5];
w=-2*pi:pi/8:2*pi;
[h]=freqz(b,a,w);
subplot(2,1,1);
```

```
stem(w,abs(h));
xlabel('freq/w');
ylabel('magnitude');
grid;
title('magnitude response of diffenrtial equataion');
subplot(2,1,2);
stem(w,angle(h));
xlabel('freq/w');
ylabel('phase in rad');
grid;
title('phase response of diffenrtial equataion');
```

OUTPUT:-



RESULT:-

Hence the frequency response is executed by using MATLAB.

EXERCISE PROGRAM:-

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

VIVA QUESTIONS:-

1. What is the commend to find phase angle?
2. What is the commend to find frequency response?
3. What is transition band?
4. What is the formula for Z-transform?
5. What is the relationship b/w impulse response& frequency response?

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

REAL TIME APPLICATIONS:

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

EXPERMENT NO-5

FOURIER SERIES COEFFICIENTS

AIM: -

To write a MATLAB program for obtaining Fourier series coefficients by formula and using Fast Fourier Transform algorithm, to verify the same in Code composer studio, and to plot all the corresponding graphs.

SOFTWARE REQURIED:-

1. MATLAB R2010a.
2. Windows XP SP2.

THEORY:-

The fast Fourier transform represents a given signal in frequency domain by reducing the number of complex multiplication to $(N/2)\log_2 N$ and complex additions to $N\log_2 N$ in DFT algorithm. The time taken by fft to find Fourier transform is lesser than dft. It is implemented in decimation in time (DIT FFT) and decimation in frequency (DIF FFT) by using radix-based butterfly structures.

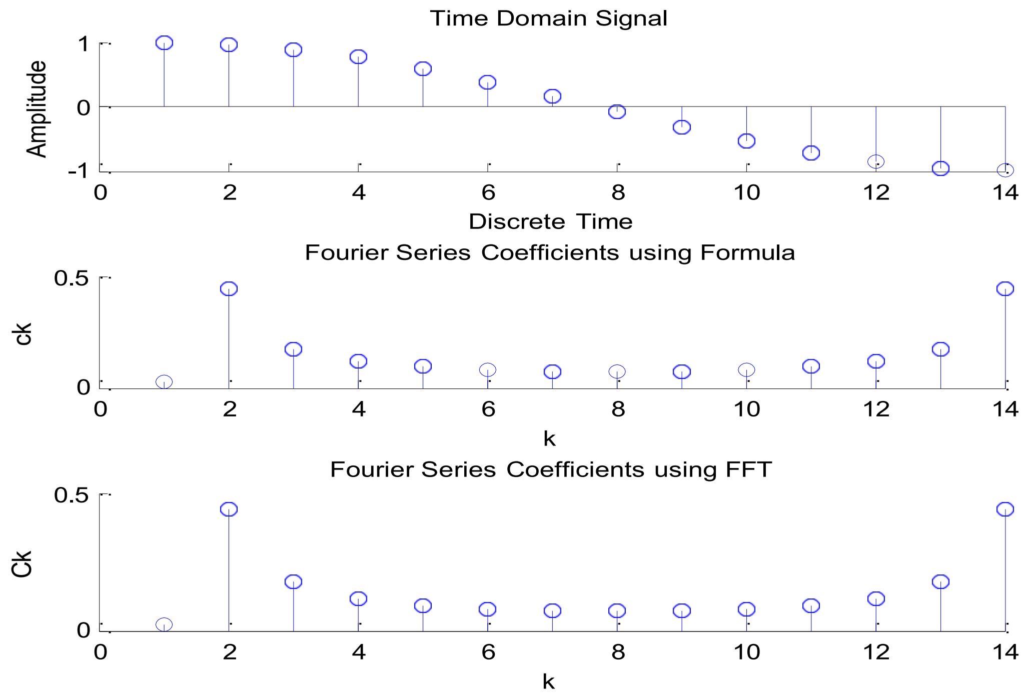
MATLAB Program:

```
clc;
close all;
clear all;
n =0:0.075:1;
x=cos(2*pi*0.5*n);
N=length(x);
for k=1:N
   y=0;
   for n=1:N
      y=y+x(n).*exp(-j*2*pi*(k-1)*(n-1)/N);
   end
C1(k)=y/N;
end
display(C1);
C2=ifft(x);
m1=abs(C1);
m2=abs(C2);
display(C1);
subplot(3,1,1);
stem(x);
xlabel('Discrete Time')
ylabel('Amplitude')
title('Time Domain Signal')
```

```
subplot(3,1,2)
stem(m1);
xlabel('k')
ylabel('ck')
title('Fourier Series Coefficients using Formula');
subplot(3,1,3);
stem(m2);
xlabel('k')
ylabel('Ck')
title('Fourier Series Coefficients using FFT');
```

Output:

```
C1 =
  Columns 1 through 4
   0.0238              0.0893 - 0.4314i   0.0747 - 0.1591i   0.0727 - 0.0923i
  Columns 5 through 8
   0.0721 - 0.0579i   0.0718 - 0.0348i   0.0717 - 0.0164i   0.0717 + 0.0000i
  Columns 9 through 12
   0.0717 + 0.0164i   0.0718 + 0.0348i   0.0721 + 0.0579i   0.0727 + 0.0923i
  Columns 13 through 14
   0.0747 + 0.1591i   0.0893 + 0.4314i
C1 =
  Columns 1 through 4
   0.0238              0.0893 - 0.4314i   0.0747 - 0.1591i   0.0727 - 0.0923i
  Columns 5 through 8
   0.0721 - 0.0579i   0.0718 - 0.0348i   0.0717 - 0.0164i   0.0717 + 0.0000i
  Columns 9 through 12
   0.0717 + 0.0164i   0.0718 + 0.0348i   0.0721 + 0.0579i   0.0727 + 0.0923i
  Columns 13 through 14
   0.0747 + 0.1591i   0.0893 + 0.4314i
```



Result: The MATLAB program for obtaining Fourier series coefficients by formula and using Fast Fourier Transform algorithm has been written, the same is verified.

EXERCISE PROGRAM:-

1. Determine the Fourier series coefficients ak for x1(t) shown below. x1(t)= x1(t + 10) ?
2. Determine the Fourier series coefficients bk for x2(t) shown below. x2(t)= x2(t + 10) ?
3. Determine the Fourier series coefficients ck for x3(t) shown below. x3(t)= x3(t + 10)?
4. Determine the Fourier series coefficients for a cos wave by using formula method?
5. Determine the Fourier series coefficients for a cosine wave using FFT ?
6. Determine the Fourier series coefficients of the following signal

   $x(t) = 1+\cos(2\pi t)$

   $x(t) = [1+\cos(2\pi t)]\ [\sin(10\pi t+\pi/6)]$?

7. Find the Fourier series coefficients for (periodic extension of)

   f(t) = 1 − t 2 , t ∈ [−1, 1] ?

8. . Find the cosine Fourier series coefficients for (periodic extension of)

   f(t) = ½ 1, t ∈ [0, 1); 0, t ∈ [1, 4). ?

9. Find the Fourier series for (periodic extension of) f(t) = ½ 1, t ∈ [0, 2); −1, t ∈ [2, 4). ?
10. Determine the Fourier series coefficients for a given even signal $f(t)=2\cos\pi t$?
11. Determine the Fourier series coefficients for a odd function 1 $f(t) = \sin t$?
12. Determine the Fourier series coefficients for a Saw-tooth Wave ?
13. Consider the function below { 1-x 0 < x ≤1; 0 1<x≤2 }
14. Determine the Fourier series coefficients ck for xn(t) shown below. xn(t)= xn(t + 50)?
15. Determine the Fourier series coefficients of the following signal

    $x(t) = 1+\cos(2\pi t)$

    $x(t) = [1+\sin(2\pi t)]\ [\cos(10\pi t+\pi/6)]$?

16. Find the Fourier series coefficients for (periodic extension of)
    f(t) = 1+t 2 , t ∈ [1, 2] ?
17. Find the cosine Fourier series coefficients for (periodic extension of)
    f(t) = 2/31, t ∈ [0, 1); 0, t ∈ [1, 4). ?
18. Determine the Fourier series coefficients ck for xn(t) shown below.
    xn(t)= xn(t+ 350)?
19. Determine the Fourier series coefficients of the following signal
    $x(t) = 1+\cos(5\pi t)$
    $x(t) = [1+\sin(5\pi t)] [\cos(14\pi t+\pi/9)]$?
20. Determine the Fourier series coefficients of the following signal
    $x(t) = 1+\cos(2\pi t)$
    $x(t) = [1+\tan(2\pi t)] [\cot(10\pi t+\pi/6)]$?

VIVA QUESTIONS:-

1. What is ment by fourier series?
2. Difine Fourier transforms?
3. Give the differences between fourier series and fourier transforms?
4. What is the formula for Fourier series?
5. What is the formula for Fourier transforms?
6. What is the formula for Z-transforms?
7. What is the formula for Laplace transforms?
8. Give the differences between Z-transform and laplace transforms?
9. Applications of Fourier Series and Fourier Transforms?
10. What are the Fourier Coefficents?
11. Do exponential fourier series also have fourier coefficients to be evaluated.?
12. What is the polar form of the fourier series?
13. Give any example of a FIR interpolator?
14. What is a line spectrum?
15. Fourier series is not true in case of discrete time signals.
16. What is the disadvantage of exponential Fourier series?
17. Fourier series uses which domain representation of signals?.
18. How does Fourier series make it easier to represent periodic signals?
19. If transfer function of a system is $H(z) = 6 + z^{-1} + z^{-2}$ then system is ?
20. List few Applications of circular convolution
21. What are the different methods used to calculate circular convolution?
22. Explain properties of circular convolution?
23. Explain modulo N operation.
24. Difference between linear convolution and circular convolution.
25. Explain the circular shift.
26. How circular convolution is used to calculate the Z
27. transform of a signal?
28. List few Applications of circular convolution
29. What are the different methods used to calculate circular convolution?
30. Explain properties of circular convolution?

REAL TIME APPLICATIONS:

Fourier series are of great importance in both theoretical and applied mathematics. For orthonormal families of complex valued functions {φn}, Fourier Series are sums of the φn that can approximate periodic, complex valued functions with arbitrary precision. This paper will focus on the Fourier Series of the complex exponentials. Of the many possible methods of estimating complex valued functions, Fourier series are especially attractive because uniform convergence of the Fourier series (as more terms are added) is guaranteed for continuous, bounded functions. Furthermore, the Fourier coefficients are designed to minimize the square of the error from the actual function. Finally, complex exponentials are relatively simple to deal with and ubiquitous in physical phenomena. This paper first defines generalized Fourier series, with an emphasis on the series with complex exponentials. Then, important properties.

EXPERMENT NO-6

IMPLEMENTATION OF FFT OF GIVEN SEQUENCE

AIM: -

Implementation of FFT of given sequence.

SOFTWARE REQURIED:-

1. MATLAB R2010a.
2. Windows XP SP2.

THEORY:-

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

PROCEDURE:-

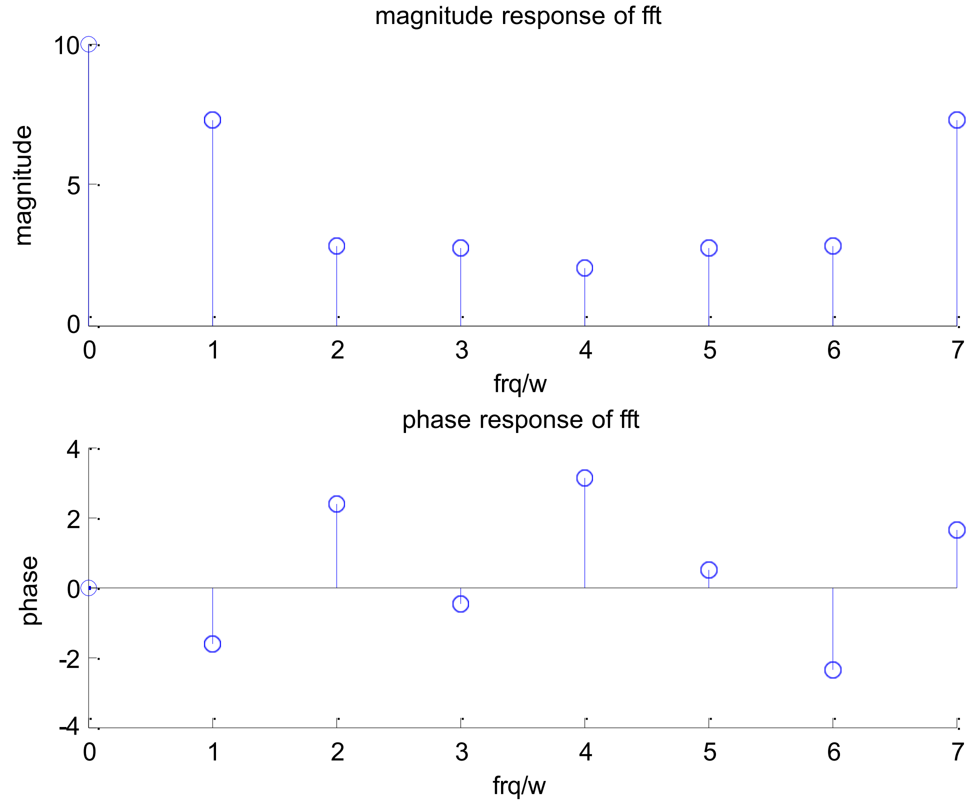
- Open MATLAB
- Open new M-file
- Type the program
- Save in current directory
- Compile and Run the program
- For the output see command window\ Figure window

PROGRAM:-

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

OUTPUT:-

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



RESULT:-

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

EXERCISE PROGRAM:-

1. Write a matlab program to find the cross correlation using FFT?
2. Write a matlab program to find the DFT of x(n) ={1 1 0 0 0 0 0 0}?
3. Write a matlab program to find the DFT of x(n) ={1 1 1 1 1 0 0 0}?
4. Write a matlab program to find the DFT of x(n) ={1 0 1 0 1 0 1 0}?
5. Write a matlab program to find the IDFT of X(k) ={1,1+j,-2j,1+2j ,-j, +j}?
6. Write a matlab program to find the IDFT of X(k) ={1,0,-2j,-1,+2j,-7j }?
7. Write a matlab program to find the IDFT of X(k) ={1,1,-2j,-1,1+2j }?
8. Write a matlab program to find the IDFT of X(k) ={1,1+j,-2j,-1+-j,1+2j }?

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

VIVA QUESTIONS:-

1. Whether linear convolution equation is a difference equation?
2. Whether DFT is a linear transform?
3. What is the difference between circular convolution & linear convolution?
4. Can you implement linear convolution using circular convolution?
5. How FFT algorithms are classified?
6. How to calculate output of DFT using MATLAB?
7. Where DFT is used?
8. What is the difference between DFT and IDFT?
9. How to compute maximum length N for a circular convolution using DFT and IDFT.
10. Explain the function of twiddle factor?
11. Give the practical application DFT & IDFT?
12. Explain the role of DFT & IDFT when the signal converted from the time domain to frequency domain?
13. Differentiate between time variant and time invariant system. If x 1(n)={1,2,3,4} and x 2(n)={1,2,3} Find the convolution using tabular representation.
14. Draw all elementary standard discrete time signals.
15. Differentiate between causal and Non causal system.
16. If x 1(n)={1,2,3,4} and x 2(n)={5,6,7,8} Find the circular representation for the above sequences.
17. How can you compute Fourier transform form Z-transform ?

18. FFT is in complex domain how to use it in real life signals optimally?
19. What is the difference between FFT and IFFT?
20. Explain using convolution the effects of taking an FFT of a sample with no windowing .
21. What is a filter?
22. Differentiate analog filter and digital filter
23. Define FIR filter
24. What are the differences between recursive and non recursive systems?
25. List a few Applications of FIR filters.
26. Explain advantages of FIR filters over IIR filters.
27. Explain limitations of FIR filters.
28. What is the different method to design FIR filters?
29. Explain different window functions.
30. Differentiate rectangular, triangular and Kaiser windows.

REAL TIME APPLICATIONS:

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

EXPERMENT NO-7

DETERMINATION OF POWER SPECTRUM

AIM: -

To obtain power spectrum of given signal using MATLAB.

SOFTWARE REQURIED:-

1. MATLAB R2010a.
2. Windows XP SP2.

THEORY:-

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

PROCEDURE:-

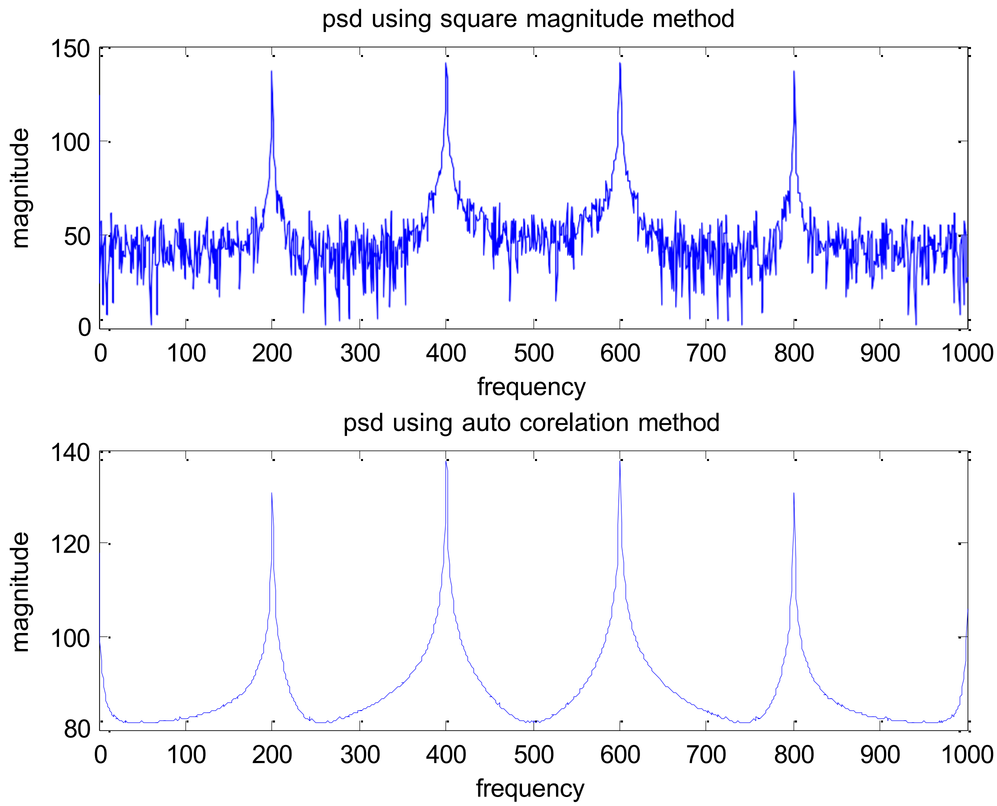
- Open MATLAB
- Open new M-file
- Type the program
- Save in current directory
- Compile and Run the program
- For the output see command window\ Figure window

PROGRAM:-

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

OUTPUT:-

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



RESULT:-

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

EXERCISE PROGRAM:-

1. Write a matlab program for power spectrum estimate using Welch method?
2. Write a matlab program to plot the frequency response of a first order system?
3. Write a matlab program to plot the frequency response of the system?
4. Write a matlab program to generate the periodic sequence?
5. Write a matlab program to generate the aperiodic sequence?
6. Write a matlab program to demonstrate the property of digital frequency?
7. Write a matlab program to illustrate the concept of aliasing?
8. Write a matlab program to plot magnitude and phase response of first order lowpass filter?
9. Write a matlab program to plot magnitude and phase response of first order highpass filter?

10. Write a matlab program to plot magnitude and phase response of second order bandpass filter?
11. Write a matlab program to plot magnitude and phase response of second order bandstop filter?
12. Write a matlab program to plot magnitude and phase response of second order lowpass filter?
13. Write a matlab program to plot magnitude and phase response of second order highpass filter?
14. Write a matlab program to plot magnitude and phase response of first order bandpass filter?
15. Write a matlab program to plot magnitude and phase response of first order bandstop filter?
16. Write a matlab program to plot the frequency response of a first order system?
17. Write a matlab program to plot the frequency response of the system?
18. Write a matlab program to generate the periodic sinusoidal sequence?
19. Write a matlab program to generate the a periodic sinusoidal sequence?
20. Write a matlab program to demonstrate the property of digital signal?

VIVA QUESTIONS:-

1. Give the expressions for finding the Average power of a signal/sequence?
2. Give the expressions for finding the energy of a signal/sequence?
3. What is power spectrum?
4. Why there are two peaks in the magnitude spectrum of sine wave?
5. What is spectrogram? Which built in function is used to solve a given difference equation?
6. What is frequency response? Give equation for first order system andsecond order system?
7. .What is an LTI system?
8. What is steady state response?
9. Suppose we have a system with transfer function H(z) = 1 / ((z –1)*(z –0.9)). Is the system stable or unstable?
10. What is Auto Regressive Model? How is the order of auto regressive model is decided?
11. Differentiate between linear and circular convolution.

12. Determine the unit step response of the linear time invariant system with impulse response h(n)=a nu(n) a<1&-a<1
13. Determine the range of values of the parameter a for which linear time invariant system with impulse response h(n)=a u(n) is stable.
14. Consider the special case of a finite duration sequence given as X(n)={2 4 0 3}, resolve the sequence x(n) into a sum of weighted sequences.
15. Describe impulse response of a function?
16. Where to use command filter or impz, and what is the difference between these two?
17. How to calculate output of DFT using MATLAB?
18. Where DFT is used?
19. What is the difference between DFT and IDFT?
20. Explain the function of twiddle factor? List some a dvantages of digital filters over analog filters.
21. Write some differences between FIR and IIR filters
22. What are the different methods to design IIR filters.
23. Why IIR filters are not reliable What are different applications of IIR filters
24. What are advantages of IIR filters.
25. What are disadvantages of IIR filters.
26. Differentiate Butterworth and Chebyshev approximations.
27. What is meant by impulse response.
28. What is the importance of impulse response to calculate the o/p response of the filter?
29. Describe impulse response of a function?
30. Where to use command filter or impz, and what is the difference between these two?

REAL TIME APPLICATIONS:

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

EXPERMENT NO-8

IMPLEMENTATION OF LP IIR FILTERS

AIM: -

Implementation of Low Pass IIR filters for given sequence.

SOFTWARE REQURIED:-

1. MATLAB R2010a.
2. Windows XP SP2.

THEORY:-

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

PROCEDURE:-

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

PROGRAM:-

% IIR LOW PASS FILTER %

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

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

OUTPUT:-

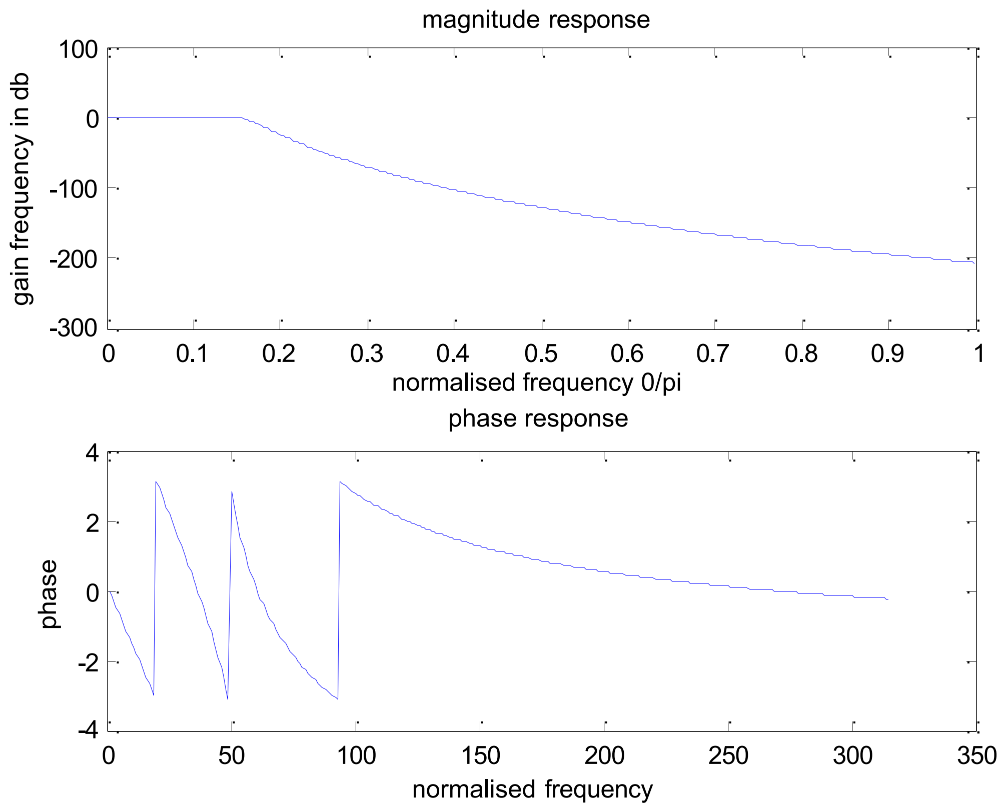
enter the pass band ripple:rp=0.15

enter the stop band ripple:rs=60

enter the pass band frequency:fp=1500

enter the stop band frequency:fs=3000

enter the sampling frequency:f=7000



RESULT:-

Thus IIR lowpass filter is designed using MATLAB.

EXERCISE PROGRAM:-

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

20. Write a matlab program to design a chebyshev -I bandreject filter for the specifications?

VIVA QUESTIONS:-

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

REAL TIME APPLICATIONS:

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

EXPERMENT NO-9

IMPLEMENTATION OF HP IIR FILTERS

AIM: -

To implement the analog & digital High Pass IIR filter.

SOFTWARE REQURIED:-

1. MATLAB R2010a.
2. Windows XP SP2.

THEORY:-

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

PROCEDURE:-

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

PROGRAM:-

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

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

OUTPUT:-

enter the sepecifications of iir filter

enter the pass band ripple:rp=0.15

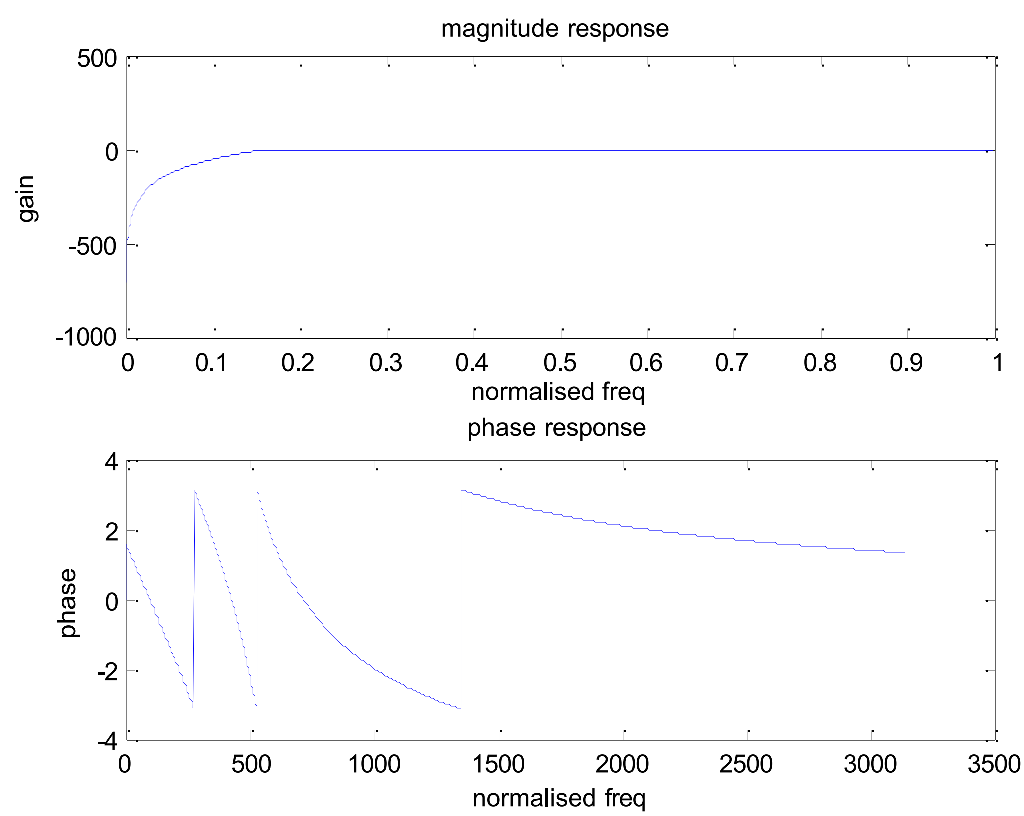
enter the stop band ripple:rs=60

enter the pass band freq:wp=1500

enter the stop band freq:ws=3000

enter the sampling freq fs=7000

freq resp of iir high pass filter is:



RESULT:-

Thus IIR lowpass filter is designed using MATLAB.

EXERCISE PROGRAM:-

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

VIVA QUESTIONS:-

1. What are the steps in designing the IIR filters?
2. State the disadvantages of impulse invariant transformation?
3. Why impulse invariant transformation is not suitable for design of high pass filters?

4. What is frequency relationship for bilinear transformation?
5. What is the frequency relationship for bilinear transformation?
6. Why interpolate,i needed for any signal/sequence?
7. What is the "interpolation factor"?
8. Which signals can be interpolated?
9. Can interpolate will happens in multiple stages? If yes give reason?
10. Give any example of a FIR interpolator?
11. Define impulse response.
12. Give me one example for impulse response.
13. Write the Formula for impulse response.
14. What are major role in order & length?
15. Define power spectral Density?
16. What is the need for spectral estimation?
17. Determine the power spectrum density?
18. What is the relation between auto correlation & spectral density?
19. Give the estimation of auto correlation function & power density for random Signals?
20. Explain power spectrum estimation using the Bartlett window?
21. What are the characteristics of Butterworth filters?
22. How the s-plane is mapped to z-plane in impulse invariant transformation?
23. How is the non-periodic nature of the input signal handled?
24. If a have two vectors how will i check the orthogonality of those vectors.
25. What is the importance of decimation for a given signal/sequence?
26. What do you mean Aliasing? What is the condition to avoid aliasing for sampling?
27. How does poly phasefiltering save computations in a decimation filter?
28. Give any practical application of decimation?
29. Which signals can be downsampled?
30. What happens if I violate the Nyquist criteria in down sampling or decimating?

REAL TIME APPLICATIONS:

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

EXPERMENT NO-10

GENERATE NARROWBAND SIGNAL THROUGH FILTERING

AIM: -

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

SOFTWARE REQURIED:-

1. MATLAB R2010a.
2. Windows XP SP2.

THEORY:-

Sinusoidal Signal Generation

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

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

A sampled sinusoid may be written as:

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

PROCEDURE:-

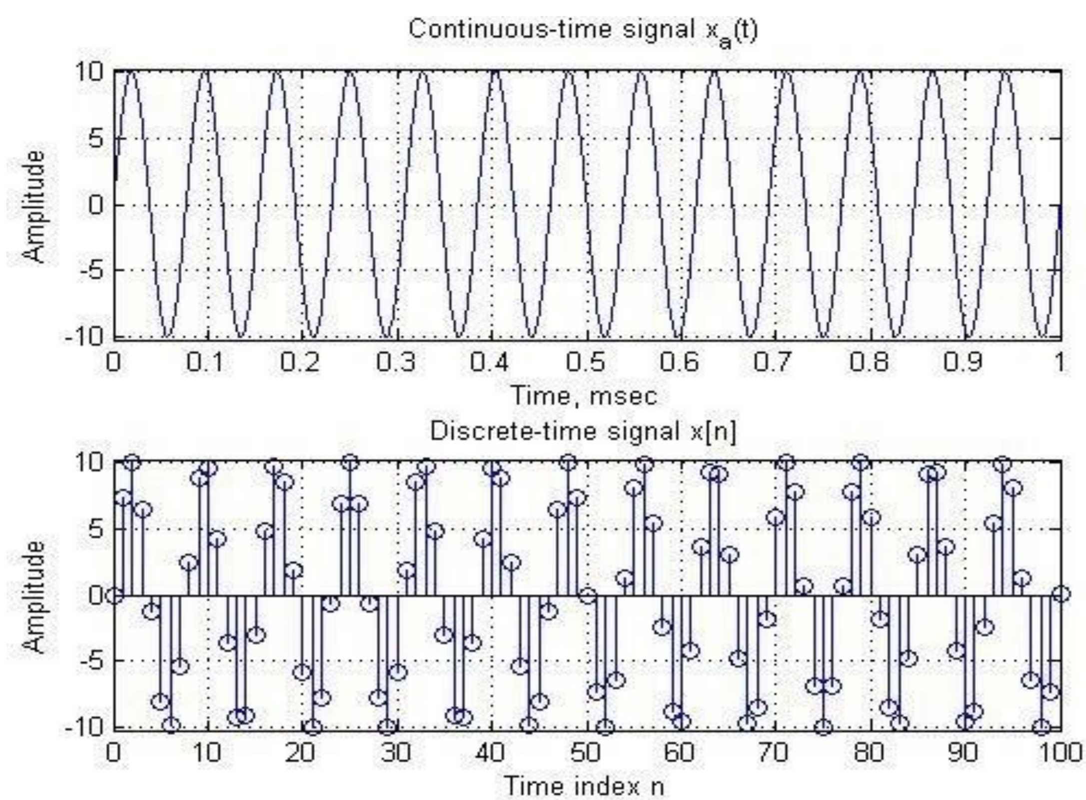
- Open MATLAB
- Open new M-file
- Type the program
- Save in current directory
- Compile and Run the program
- For the output see command window\ Figure window

PROGRAM:-

```
clc;
t=0:0.0005:1;
a = 10
f = 13;
xa=a*sin(2*pi*f*t)
subplot(2,1,1)
plot(t,xa);
grid
xlabel('Time,msec')
;ylabel('Amplitude'
```

```
);

title('Continuous-time signal
axis([01-10.2 10.2])

subplot(2,1,2
); T = 0.01;
n = 0:T:1;

xs=a*sin(2*pi*f*n)
; k = 0:length(n)-1;
stem(k,xs);
grid

xlabel('Time index n');
ylabel('Amplitude');
title('Discrete-time signal
x[n]'); axis([0 (length(n)-1)
10.2 10.2])
```

OUTPUT:-



RESULT:-

Sinusoidal signal is generated by using MATLAB.

EXERCISE PROGRAM:-

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

VIVA QUESTIONS:-

1. Define sinusoidal signal?
2. Define C.T.S?
3. Define D.T.S?
4. Compare C.T.S & D.T.S?
5. Draw the C.T.S & D.T.S diagrams?
6. Give the formula for PSD.
7. What is filter?
8. Define Stem, Plot, Plot3,fplot, ezplot, linspace, flyplr, grid,mesh and legend.
9. Draw the C.T.S & D.T.S diagrams.

10. Which built in function is used to solve a given difference equation?
11. What is frequency response? Give equation for first order system andsecond order system?
12. What is an LTI system?
13. What is steady state response?
14. What is Auto Regressive Model? How is the order of auto regressive model is decided?
15. Differentiate between linear and circular convolution.
16. Determine the unit step response of the linear time invariant system with impulse response h(n)=a nu(n) a<1&-a<1
17. Determine the range of values of the parameter a for which linear time invariant system with impulse response h(n)=a u(n) is stable.
18. How is the non-periodic nature of the input signal handled?
19. If a have two vectors how will i check the orthogonality of those vectors.
20. Can IIR filters be Linear phase? how to make it linear Phase?
21. What is the special about minimum phase filter?
22. What is the special about minimum phase filter?
23. In signal processing, why we are much more interested in orthogonal transform?
24. What are the filter specifications required to design the analog filters?
25. What is meant by frequency response of filter?
26. What is meant by magnitude response?
27. What is meant by phase response?
28. What is steady transient response?
29. Differentiate ideal filter and practical filter responses.
30. What are the different types of analog filter approximations?

Realtime Applications:

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

EXPERMENT NO-11

DTMF SIGNAL GENERATION

AIM: -

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

SOFTWARE REQURIED:-

1. MATLAB R2010a.
2. Windows XP SP2.

THEORY:-

Dual Tone Multiple Frequency (DTMF) Signals.

PROCEDURE:-

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

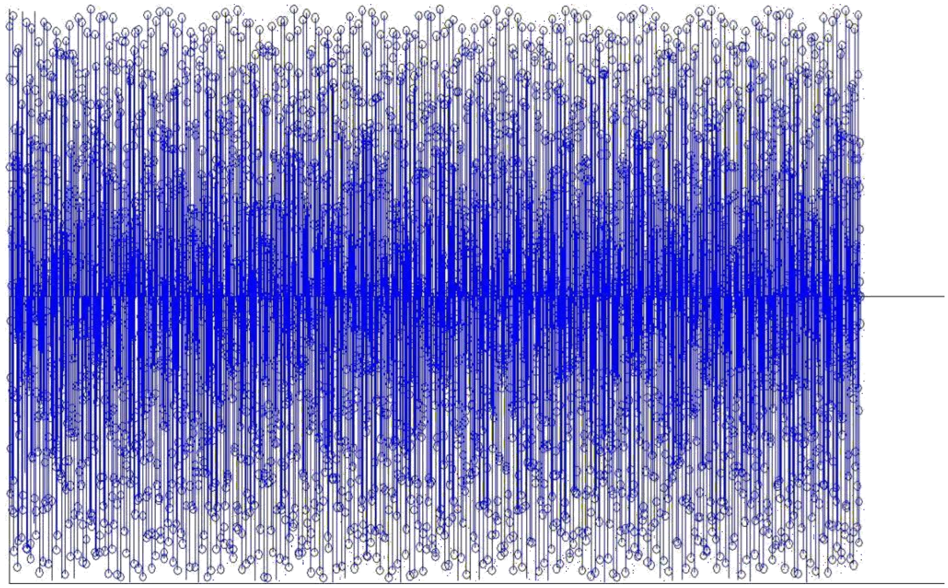
PROGRAM:-

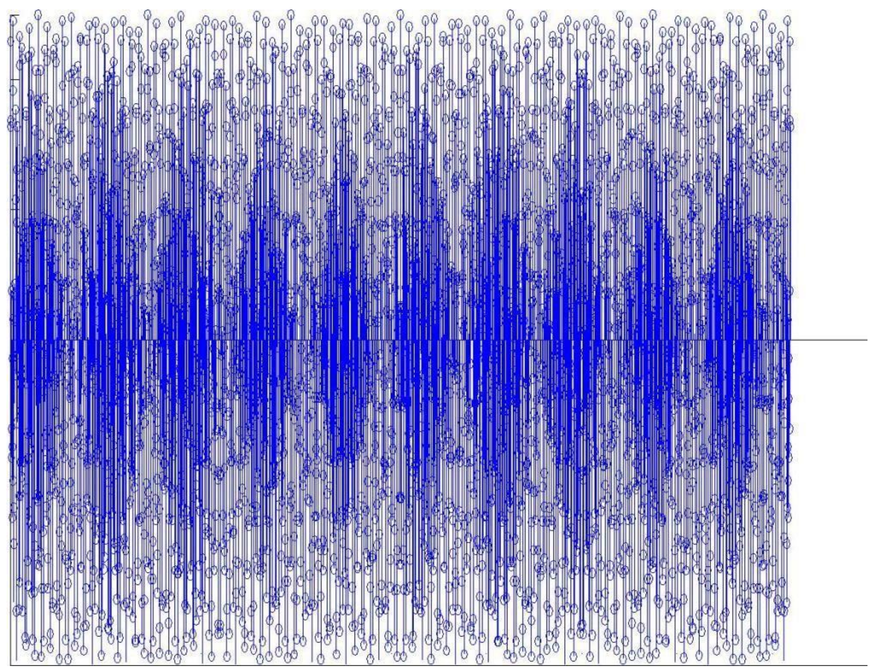
```
% Dual Tone Multiple Frequency (DTMF) Signals.
clc;

clearall;

closeall;

number=input('enter a phone number with no

spaces','s'); %number=1;

fs=8192;   % fs is the sampling Frequency

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

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

y= 2*pi*[1209 1336 1477
1633]; t=[0:1/fs:T]'

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


ty=[sin(y(1)*t),sin(y(2)*t),sin(y(3)*t),sin(y(4)*t)

]/2; for k=1:length(number)
```

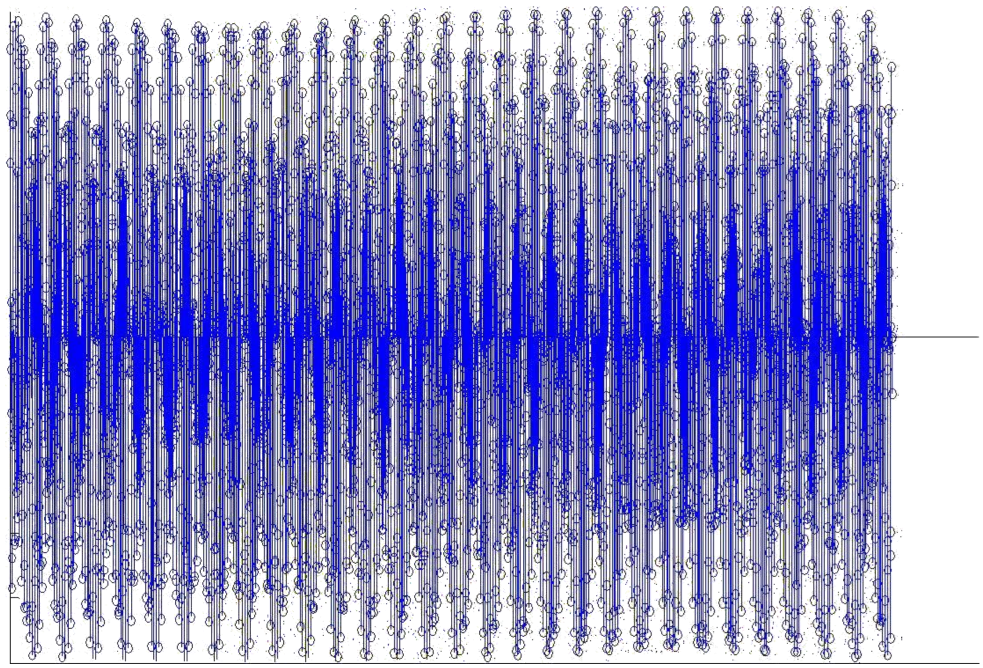
```
switch
    number(k)
    case '1'
      tone =
      tx(:,1)+ty(:,1);
      sound(tone);
      stem(tone);
    case '2'
      tone =
      tx(:,1)+ty(:,2);
      sound(tone);
      stem(tone);
    case '3'
      tone =
      tx(:,1)+ty(:,3);
      sound(tone);
      stem(tone);
    case '4'
   tone = tx(:,2)+ty(:,1); sound(tone); stem(tone); otherwise
      disp('invalid number');
end
pause(2.70
)
end;
```

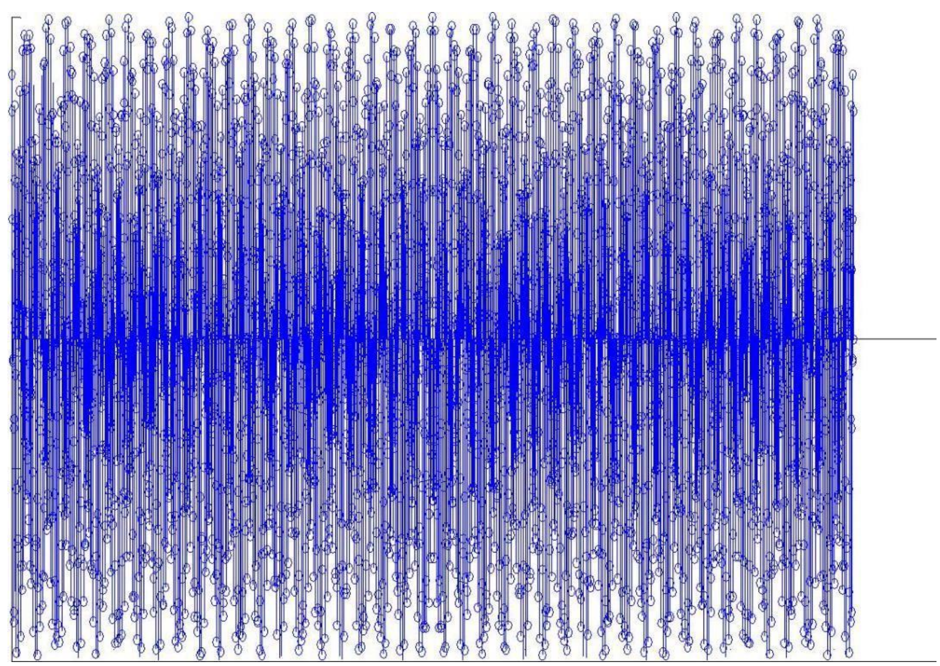
OUTPUT:-

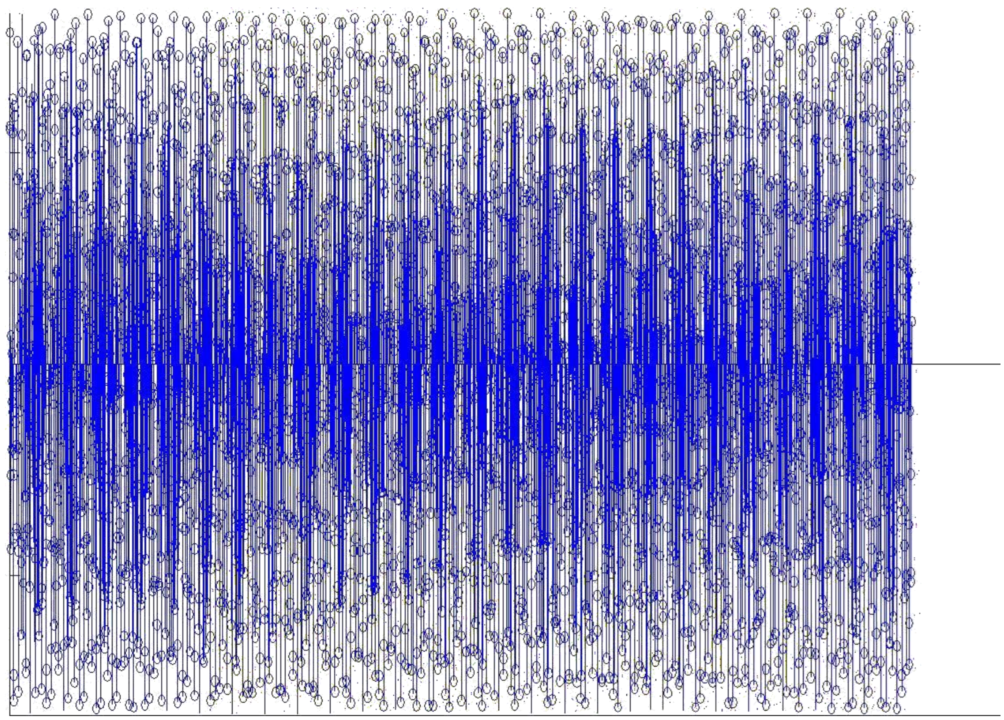
Input: 01234











RESULT:-

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

EXERCISE PROGRAM:-

1. Write a matlab program to generate a sine wave with amplitude = 3, frequency 20Hz?
2. Write a matlab program to generate a cos wave with amplitude = 3, frequency 20Hz?
3. Write a matlab program to generate a triangular wave with amplitude = 8, frequency 10Hz?
4. Write a matlab program to generate a square wave with amplitude = 2, frequency 10kHz?
5. Write a matlab program to generate a sinc wave with amplitude = -8, frequency5Khz?
6. Write a matlab program to generate a sine wave with amplitude = 7, frequency 29Hz.
7. Write a matlab program to generate a cos wave with amplitude = 9, frequency 50Hz.
8. Write a matlab program to generate a triangular wave with amplitude = 24, frequency 100Hz.
9. Write a matlab program to generate a square wave with amplitude = 12, frequency 10kHz.
10. Write a matlab program to generate a sinc wave with amplitude = 5, frequency5Khz.

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

VIVA QUESTIONS:-

1. Define Signal?
2. Define determistic and Random Signal?
3. Define Delta Function?
4. What is Signal Modeling?
5. Define Periodic and a periodic Signal?
6. Define impulse response.
7. Give me one example for impulse response.
8. Write the Formula for impulse response.
9. What are major role in order & length?
10. Define power spectral Density?
11. What is the need for spectral estimation?
12. Determine the power spectrum density?
13. What is the relation between auto correlation & spectral density?
14. Give the estimation of auto correlation function & power density for random Signals?
15. Explain power spectrum estimation using the Bartlett window?
16. Give the formula for PSD.
17. What is filter?

18. What do you mean by phase spectrum and magnitude spectrum/ give comparison?
19. How do you reduce spectral leakage?
20. What do you mean by spectral resolution?
21. Define sinusoidal signal
22. Define C.T.S .
23. Define D.T.S.
24. Compare C.T.S & D.T.S.
25. Define Stem, Plot, Plot3,fplot, ezplot, linspace, flyplr, grid,mesh and legend.
26. Draw the C.T.S & D.T.S diagrams.
27. Define signal and signal processing.
28. Differentiate digital and analog signals?
29. How the DSP processor will differ from conventional processors?
30. Expand the abbreviation TMS320C 5X/6X

REAL TIME APPLICATIONS:

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

EXPERMENT NO-12

DECIMATION

AIM: -

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

SOFTWARE REQURIED:-

1. MATLAB R2010a.
2. Windows XP SP2.

THEORY:-

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

PROCEDURE:-

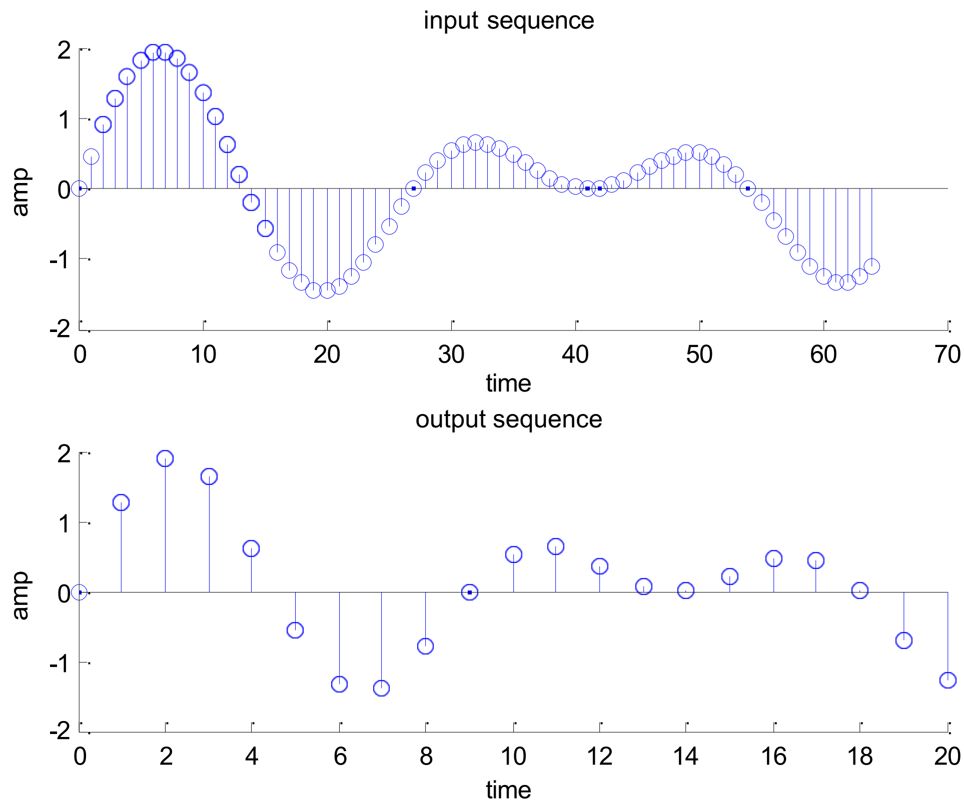
- Open MATLAB
- Open new M-file
- Type the program
- Save in current directory
- Compile and Run the program
- For the output see command window\ Figure window

PROGRAM:-

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

OUTPUT:-

enter sample value65
enter down sampling factor3



RESULT:-

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

EXERCISE PROGRAM:-

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

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

VIVA QUESTIONS:-

1. Define decimation?
2. Define multi rate signal processing?
3. What are the effects of coefficient quantization in FIR filters?
4. What is quantization process?
5. What is transmultiplexer? What is its use?
6. What is the DC gain of a FIR filter?
7. What do you mean by cut-off frequency?
8. Give the difference between analog and digital filter?
9. What is the difference between type 1 and type 2 filter structure?
10. what is the role of delay element in filter design?
11. Explain how the frequency is filter in filters?
12. Differences between Butterworth chebyshev filters?
13. Can IIR filters be Linear phase? how to make it linear Phase?
14. What is the special about minimum phase filter?
15. What is the special about maximum phase filter?
16. In signal processing, why we are much more interested in orthogonal transform?
17. How is the non-periodic nature of the input signal handled?
18. If a have two vectors how will i check the orthogonality of those vectors.

19. What is the importance of decimation for a given signal/sequence?
20. What do you mean Aliasing? What is the condition to avoid aliasing for sampling?
21. Define impulse response.
22. Give me one example for impulse response.
23. Write the Formula for impulse response.
24. What are major role in order & length?
25. Define power spectral Density?
26. What is the need for spectral estimation?
27. Determine the power spectrum density?
28. What is the relation between auto correlation & spectral density?
29. Give the estimation of auto correlation function & power density for random Signals?
30. Explain power spectrum estimation using the Bartlett window?

REAL TIME APPLICATIONS:

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

EXPERMENT NO-13

INTERPOLATION

AIM: -

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

SOFTWARE REQURIED:-

1. MATLAB R2010a.
2. Windows XP SP2.

THEORY:-

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

PROCEDURE:-

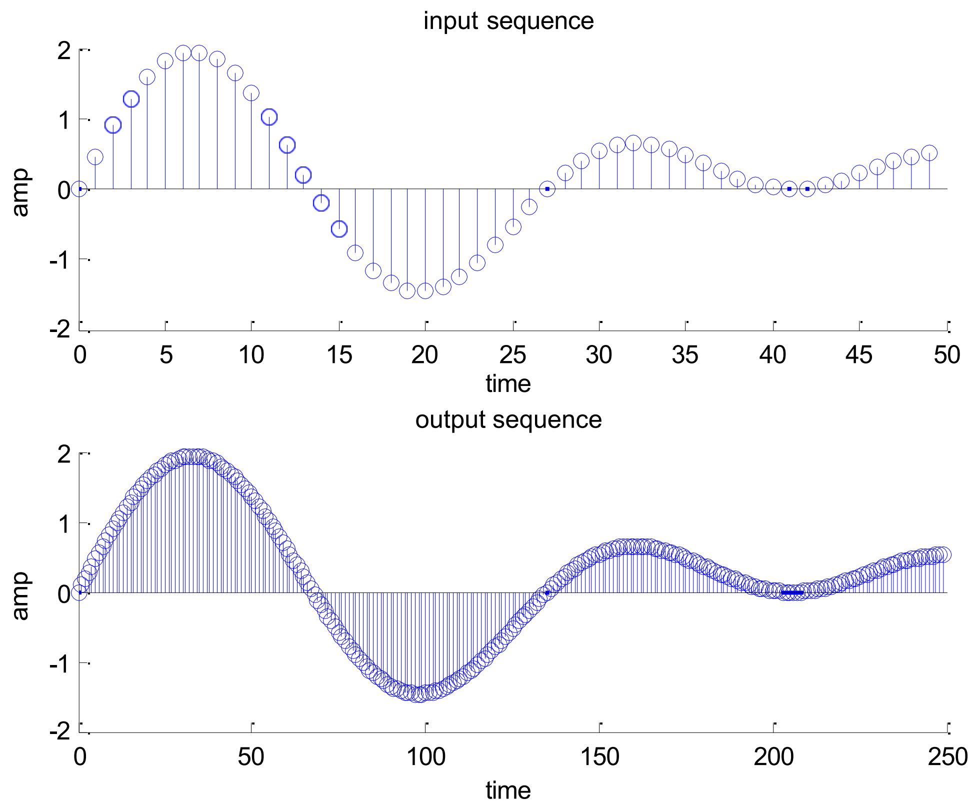
- Open MATLAB
- Open new M-file
- Type the program
- Save in current directory
- Compile and Run the program
- For the output see command window\ Figure window

PROGRAM:-

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

OUTPUT:-

enter sample value50
enter up sampling factor



RESULT:-

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

EXERCISE PROGRAM:-

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

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

VIVA QUESTIONS:-

1. How aliasing can be avoided?
2. Which type of interpolation is used to reconstruct the signal?
3. What is aliasing?
4. Define interpolation?
5. What is pre-alias filter?
6. What is the importance of decimation for a given signal/sequence?
7. What do you mean Aliasing? What is the condition to avoid aliasing for sampling?
8. How does poly phasefiltering save computations in a decimation filter?
9. Give any practical application of decimation?
10. Which signals can be downsampled?
11. What happens if I violate the Nyquist criteria in down sampling or decimating?
12. Can we do decimate in multiple stages?
13. What are "decimation" and "downsampling"?
14. What is the "decimation factor
15. How does poly phase filtering save computations in an interpolation filter?

16. Why do we need I&Q signals?
17. What is Interpolation and decimation filters and why we need it?
18. What are "up sampling" and "interpolation"?
19. Why interpolate, needed for any signal/sequence?
20. What is the "interpolation factor"?
21. What kind of processor is DSP processor?
22. What are the main building
23. blocks of DSP processor?
24. What is the main function of MAC unit?
25. Explain VLIW architecture?
26. Explain the significance of convolution.
27. Define linear convolution.
28. Why linear convolution is called as a periodic convolution?
29. Why zero padding is used in linear convolution?
30. What are the four steps to find linear convolution?

REAL TIME APPLICATIONS:

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

EXPERMENT NO-14

IMPLEMENTATION OF I/D SAMPLING RATE CONVERTERS

AIM: -

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

SOFTWARE REQURIED:-

1. MATLAB R2010a.
2. Windows XP SP2.

THEORY:-

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

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

PROCEDURE:-

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

PROGRAM:-

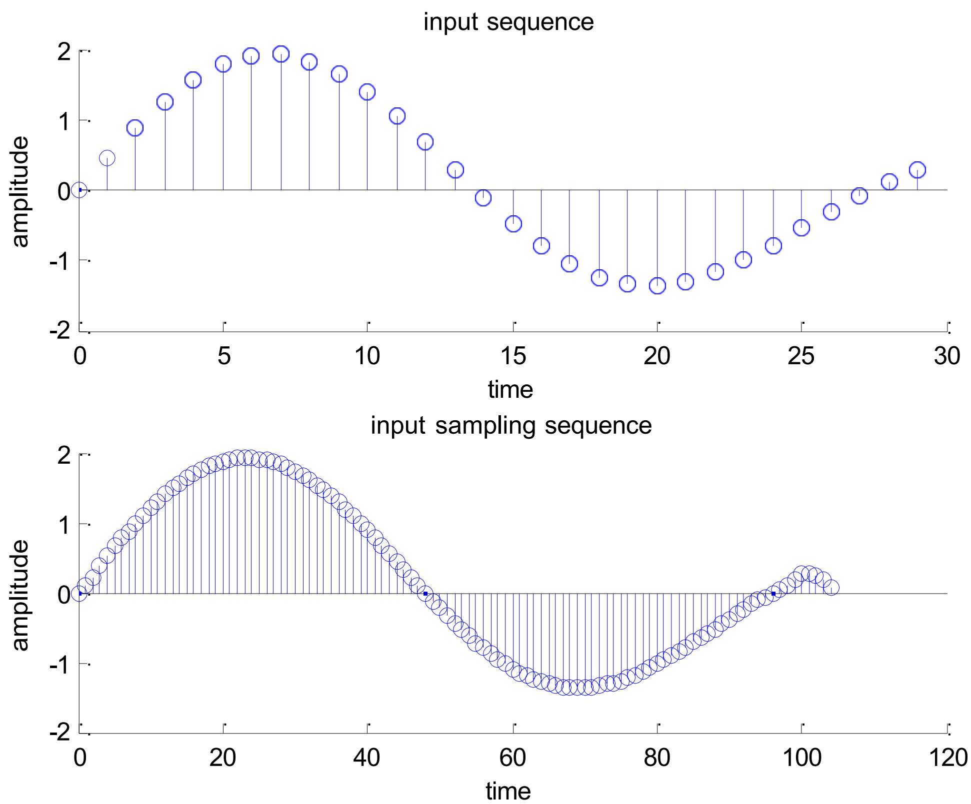
% interpolation/dismation sampling %

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

```
subplot(2,1,2);
stem(t,y(1:N*l/m));
xlabel('time');
ylabel('amplitude');
title('input sampling sequence');
```

OUTPUT:-

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



RESULT:-

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

EXERCISE PROGRAM:-

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

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

VIVA QUESTIONS:-

1. What is multi rate signal processing?
2. What is the need for anti-imaging filter after up sampling a signal?
3. What is the need for anti-imaging filter prior to down sampling?
4. Define down sampling?
5. What is meant by up sampling?
6. Give any practical application of decimation?
7. Which signals can be downsampled?
8. What happens if I violate the Nyquist criteria in down sampling or decimating?
9. Can we do decimate in multiple stages?
10. What are "decimation" and "downsampling"?
11. What isthe "decimation factor
12. How does polyphase filtering save computations in an interpolation filter?
13. Why do we need I&Q signals?
14. What is Interpolation and decimation filters and why we need it?

15. What are "upsampling" and "interpolation"?
16. Why interpolate,i needed for any signal/sequence?
17. What is the "interpolation factor"?
18. Which signals can be interpolated?
19. Can interpolate will happens in multiple stages? If yes give reason?
20. How is the non-periodic nature of the input signal handled?
21. What is "bit reversal"?
22. What do you mean by phase spectrum and magnitude spectrum/ give comparison?
23. How do you reduce spectral leakage?
24. What do you mean by spectral resolution?
25. What is FIR and IIR filter define, and distinguish between these two?
26. What is window method? How you will design an FIR filter using window method?
27. What are low-pass and band-pass filter and what is the difference between these two?
28. What is the matlab command for Hamming window? Explain.
29. What do you mean by built in function 'abs' and where it is used?
30. Explain how the FIR filter are stable?

REAL TIME APPLICATIONS:

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

EXPERMENT NO-15

IMPULSE RESPONSE

AIM: -

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

SOFTWARE REQURIED:-

1. MATLAB R2010a.
2. Windows XP SP2.

THEORY:-

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

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

PROCEDURE:-

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

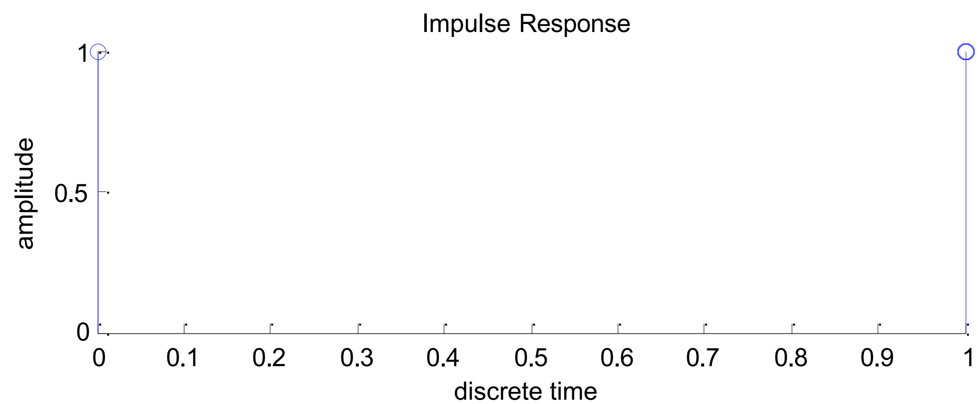
PROGRAM:-

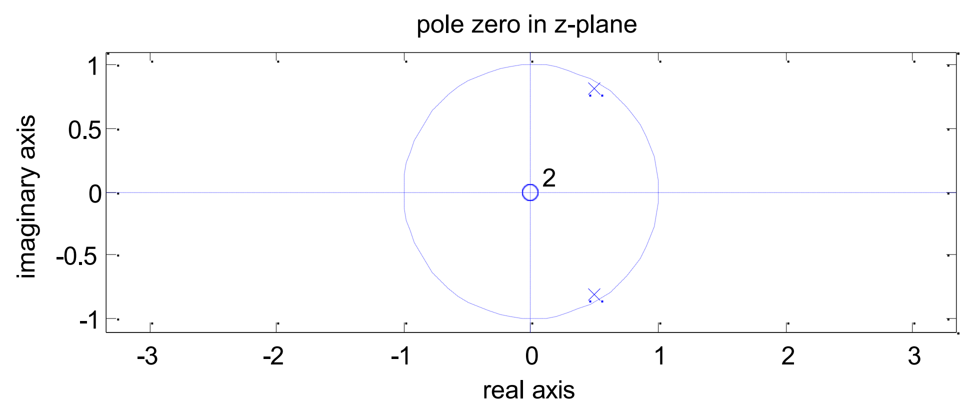
```
%To find the impulse response of discrete time system
%y(n)-y(n-1)+0.9y(n-2)=xn)%
clc;
clear all;
close all;
b=input('Enter the coefficeints of x(n):b=');
a=input('Enter the coefficeints of x(n):a=');
N=input('Enter the order of N =');
h=impz(b,a,N);
n=0:N-1;
subplot(2,1,1);
stem(n,h);
xlabel('discrete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

```
zplane(b,a);
xlabel('real axis');
ylabel('imaginary axis');
title('pole zero in z-plane');
```

OUTPUT:-

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





RESULT:-

Hence the impulse response of the given system is performed.

EXERCISE PROGRAM:-

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

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

VIVA QUESTIONS:-

1. What is the purpose of using convolution?
2. Give the formula for calculating linear convolution?
3. What are the properties of convolution?
4. What is meant by discrete convolution?
5. Define linear system and give example?
6. What is the different between Circular and Linear convolution?
7. Give the steps to get the result of linear convolution from the method of circular convolution?
8. What is the circular convolution?
9. What are the uses of function ceil and for?
10. Define boxcar
11. Define Kaiser
12. Define Bartlett
13. What is an FIR system? Compare FIR and IIR system?
14. What are the properties of chebyshev filter?
15. Define signal flow graph?
16. Draw the signal flow graph of first order digital filter?

17. What is advantage of cascade realization?
18. What is the main disadvantage of direct-form realization?
19. What is the function in MATLAB used for padding zeros to a sequence? If your sequence is, x = [1 2 3 4] and you want to pad zeros to it. How can you do that in MATLAB?
20. What is the use of following functions in MATLAB:
    i. length( )
    ii. max( )
    iii. min( )