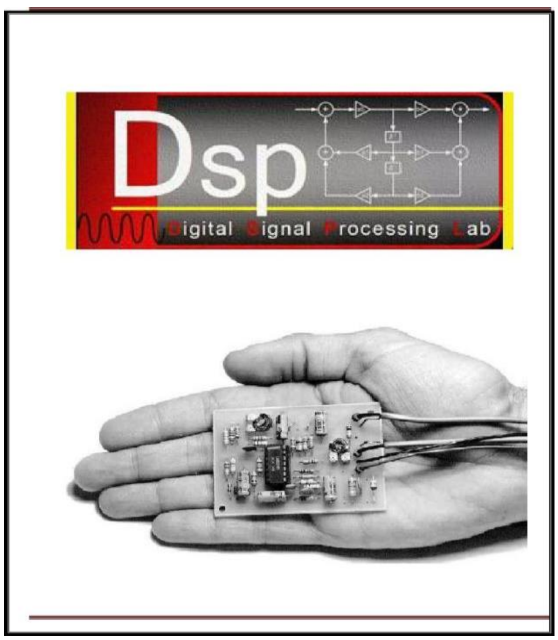
MARRI LAXMAN REDDY
INSTITUTE OF TECHNOLOGY AND MANAGEMENT
(AN AUTONOMOUS INSTITUTION)
(Approved by AICTE, New Delhi & Affiliated to JNTUH, Hyderabad)
Accredited by NBA and NAAC with 'A' Grade & Recognized Under Section2(f) & 12(B)of the UGC act,1956

Department of Electronics & Communication Engineering

LABORATORY MANUAL

DIGITAL SIGNAL PROCESSING

MLRS-R22



III B.TECH(ECE) - II Semester

Academic Year: 2025-2026

Prepared by
Mr.Ch.Nagababu, Asst. Professor
Mr.A.Anil Kumar, Asst. Professor

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Department of Electronics & Communication Engineering

VISION AND MISSION OF THE INSTITUTE

INSTITUTE VISION:

To be a globally recognized institution that fosters innovation, excellence, and leadership in education, research, and technology development, empowering students to create sustainable solutions for the advancement of society.

INSTITUTE MISSION:

- 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.

QUALITY POLICY

The management is committed in assuring quality service to all its stakeholders, students, parents, alumni, employees, employers, and the community.

Our commitment and dedication are built into our policy of continual quality improvement by establishing and implementing mechanisms and modalities ensuring accountability at all levels, transparency in procedures, and access to information and actions.

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VISION AND MISSION OF THE DEPARTMENT

DEPARTMENT VISION:

To provide quality technical education in Electronics and Communication Engineering through research, innovation, striving for global recognition in specified domain, leadership, and sustainable societal solutions.

MISSION:

DM1: To create a transformative learning environment that empowers students in electronics and communication engineering, fostering excellence intechnical skills and leadership.

DM2: To drive innovation through cutting-edge research, deliver a transformative education grounded in ethical principles, and nurture the development of professionals

DM3: To cultivate strong industry partnerships, and engaging actively with the community for societal and technological progress.

PROGRAMME EDUCATIONAL OBJECTIVES

1. **PEO 1:** have successful **careers in Industry.**
2. **PEO 2:** show excellence in **higher studies/ Research.**
3. **PEO 3:** Show good competency towards **Entrepreneurship.**

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PROGRAM OUTCOMES

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

PROGRAM SPECIFIC OUTCOMES	
PSO1	Analyze and design analog & digital circuits or systems for a given specification and function.
PSO2	Implement functional blocks of hardware-software co-design for signal processing and communication applications.

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Department of Electronics & Communication Engineering

COURSE STRUCTURE, OBJECTIVES

COURSE STRUCTURE

Laboratory subjects – Internal and external evaluation– Details of marks

Digital Signal Processing lab will have a continuous evaluation during 6th semester for 40 sessional marks and 60 end semester examination marks.

Out of the 40 marks for internal evaluation, day-to-day work in the laboratory shall be evaluated for 20 marks and internal practical examination shall be evaluated for 20 marks conducted by the laboratory teacher concerned.

The end examination will be evaluated for a maximum of 60 marks. The end semester examination shall be conducted with an external examiner and internal examiner. The external examiner shall be appointed by the principal / Chief Controller of examinations

COURSE OBJECTIVES

- Implementation of Linear and Circular Convolution.
- Implementation of FIR and IIR filters
- Predict time and frequency response of discrete-time systems using various techniques like Z Transform, DFT, FFT
- Study the architecture of DSP processor
- Demonstration of Finite word length effects

Department of Electronics & Communication Engineering

Course Outcomes:

Digital Signal Processing Lab	1.Understand the handling of discrete/digital signals using MATLAB 2.Learn the basic operations of Signal processing 3.Analyze the spectral parameter of window functions 4.Design IIR, and FIR filters for band pass, band stop, low pass and high pass filters 5.Design the signal processing algorithm using MATLAB

Course Outcomes (CO's)–Program Outcomes(PO's)Mapping

CO's Po's	PO1	PO2	PO3	PO4	PO5	PO6	PO7	PO8	PO9	PO10	PO11	PO12	PSO1	PSO2
CO1	3	3	1	3	3	-	-	-	-	1	-	-	2	2
CO2	3	3	1	3	3	-	-	-	-	1	-	-	2	-
CO3	3	3	1	3	3	-	-	-	-	1	-	-	2	-
CO4	3	3	1	3	3	-	-	-	-	1	-	-	2	2
CO5	3	3	1	3	3	-	-	-	-	1	-	-	2	-

Simple-1 **Moderate-2** **High-3**

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.

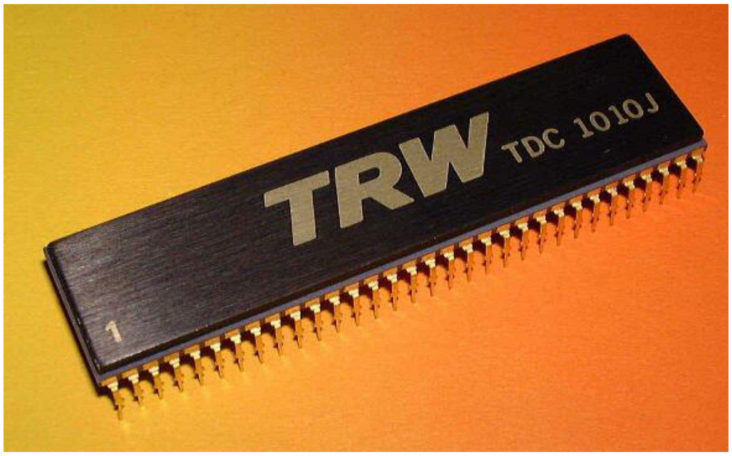
Intel's introduction of the first commercial microprocessor, the 4004, in 1971, was the first hint of what was to come. The Intel 4004 microprocessor could certainly multiply and add, but it could add only four bits at a time, and multiplication was a multi-step instruction sequence. The silicon was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

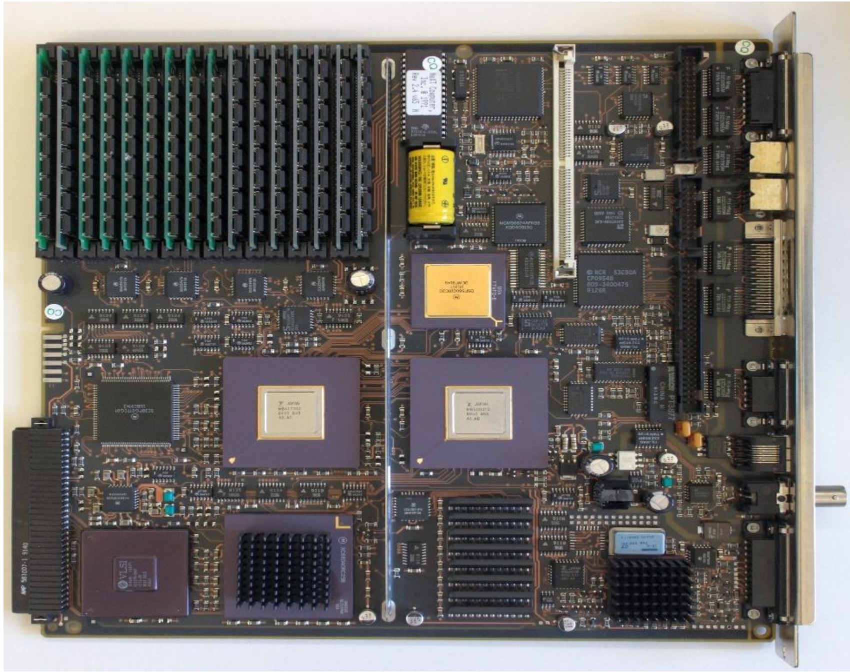


The same year that AMI introduced the S2811 Signal Processing Peripheral, TI introduced consumers to a toy based on DSP, the battery-powered "Speak & Spell," which implemented LPC as its core speech-encoding technology. The Speak & Spell toy incorporated a TI TMC0280 speech synthesizer chip, which implemented 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.

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DIGITAL SIGNAL PROCESSING 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.
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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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: -

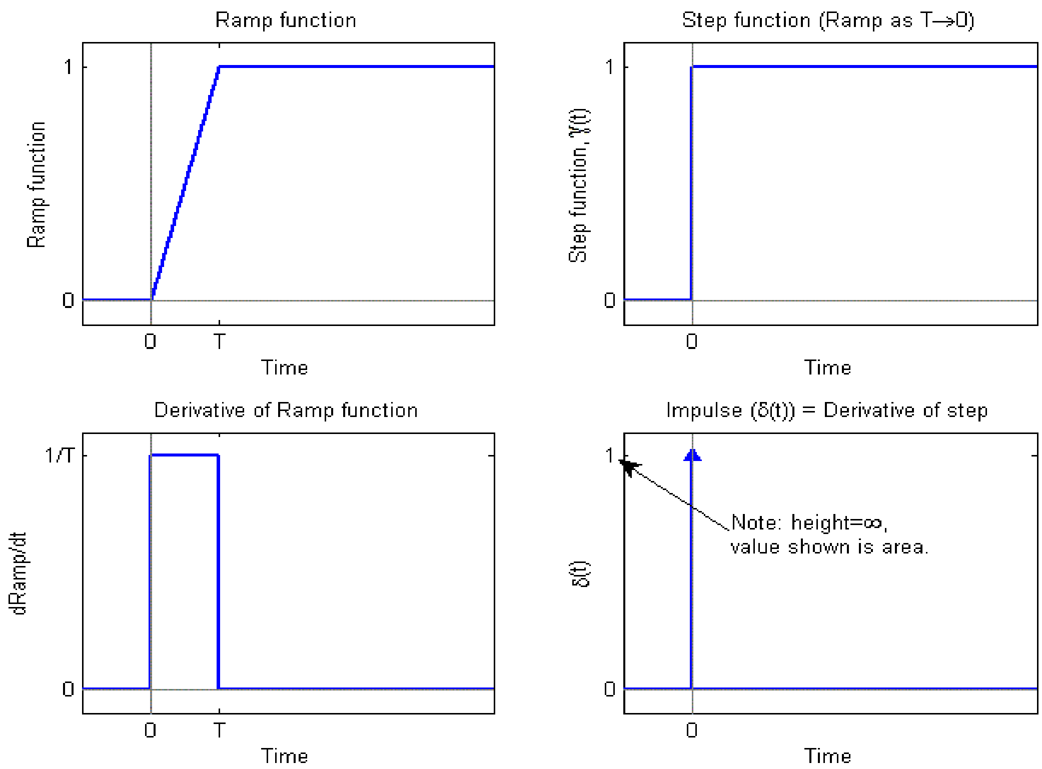
To write a "MATLAB" Program to generate various signals such as unit impulse, unit step, unit ramp, sinusoidal, exponential growing signal, exponential decaying signal, cosine signal and also generate sinusoidal signal based on recursive differential equation.

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

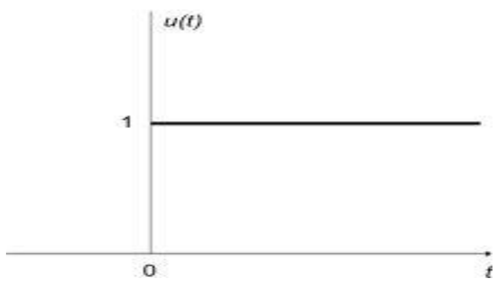
$$\begin{aligned}\int_a^b \delta(t) \cdot f(t - T)dt &= \int_a^b \delta(t) \cdot f(-T)dt \\ &= f(-T) \cdot \int_a^b \delta(t)dt \\ &= \begin{cases} f(-T), & a < 0 < b \\ 0, & \text{otherwise} \end{cases}\end{aligned}$$

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

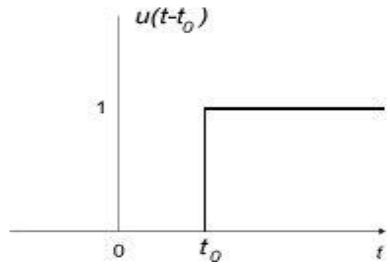
The unit step function is level in all places except for a discontinuity at t = 0. For this reason, the derivative of the unit step function is 0 at all points t, except where t = 0. Where t = 0, the derivative of the unit step function is infinite.

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:

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



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 %

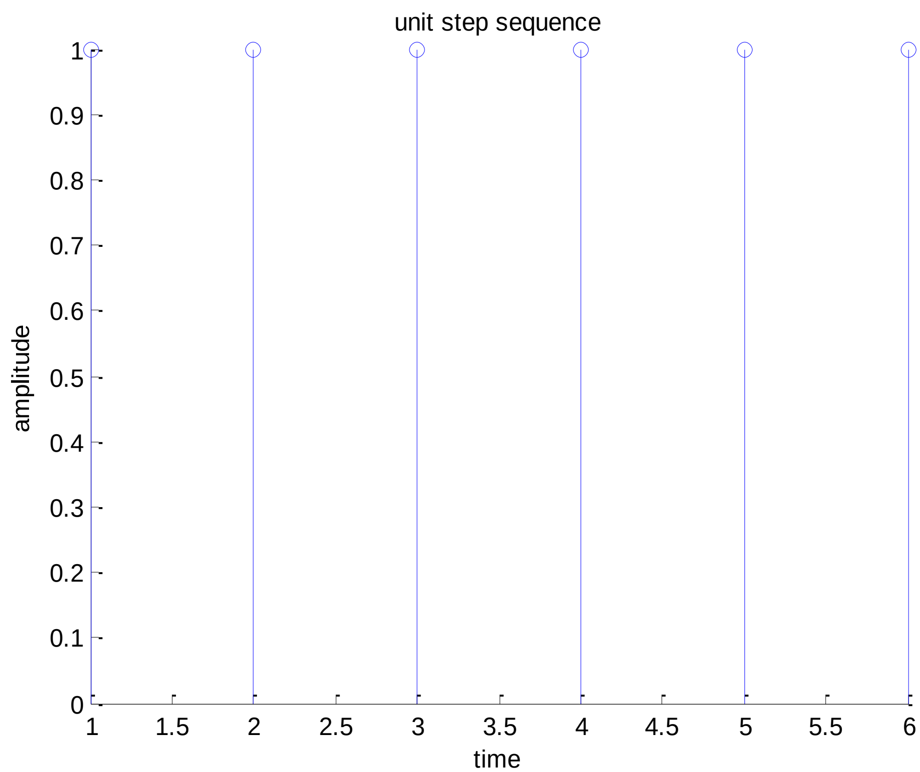
```
clc;
clear all;
close all;
disp('unit impuse signal');
N=input ('enter the no of samples');
n=-N:1:N;
x=[zeros(1,N) 1 zeros(1,N)];
stem(n,x);
xlabel('time');
ylabel('amplitude');
tittle('impulse sequence');
```

% exponentialsignals%

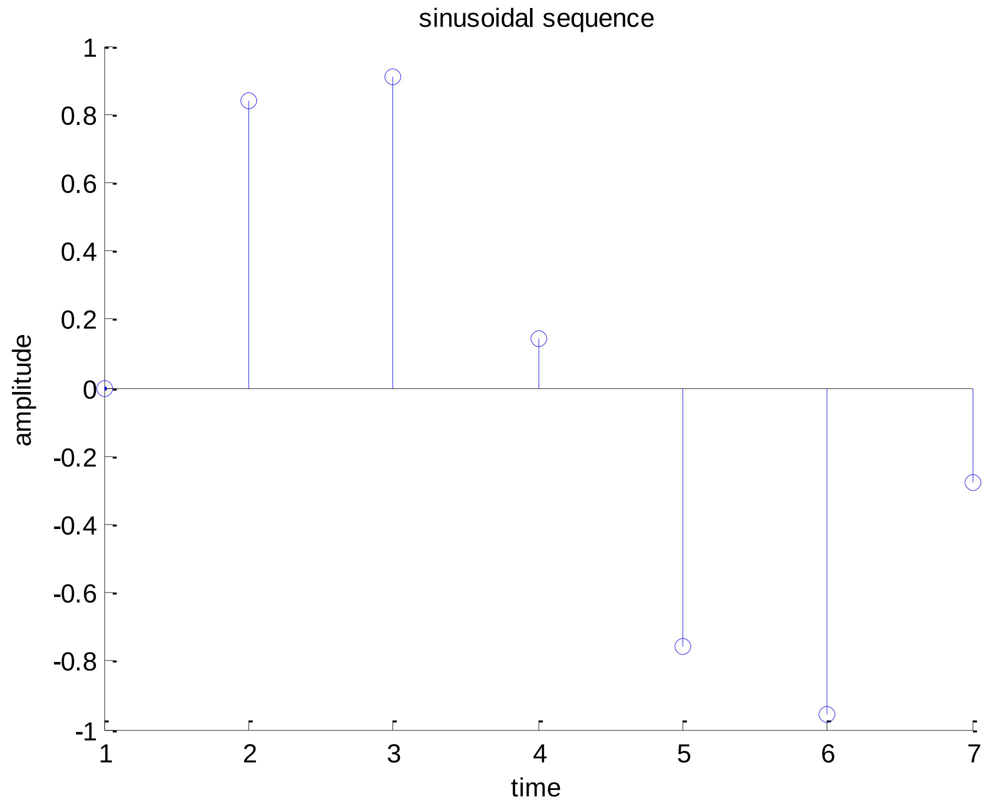
```
clc;
clear all;
close all;
disp('exponential signals');
N=input('enter the no of samples');
n=0:1:N;
n=-N:1:N;
a=0.5;
x=a.^n;
stem(x);
xlabel('time');
ylabel('amplitude');
title('exponential sequence');
```

OUTPUT:-

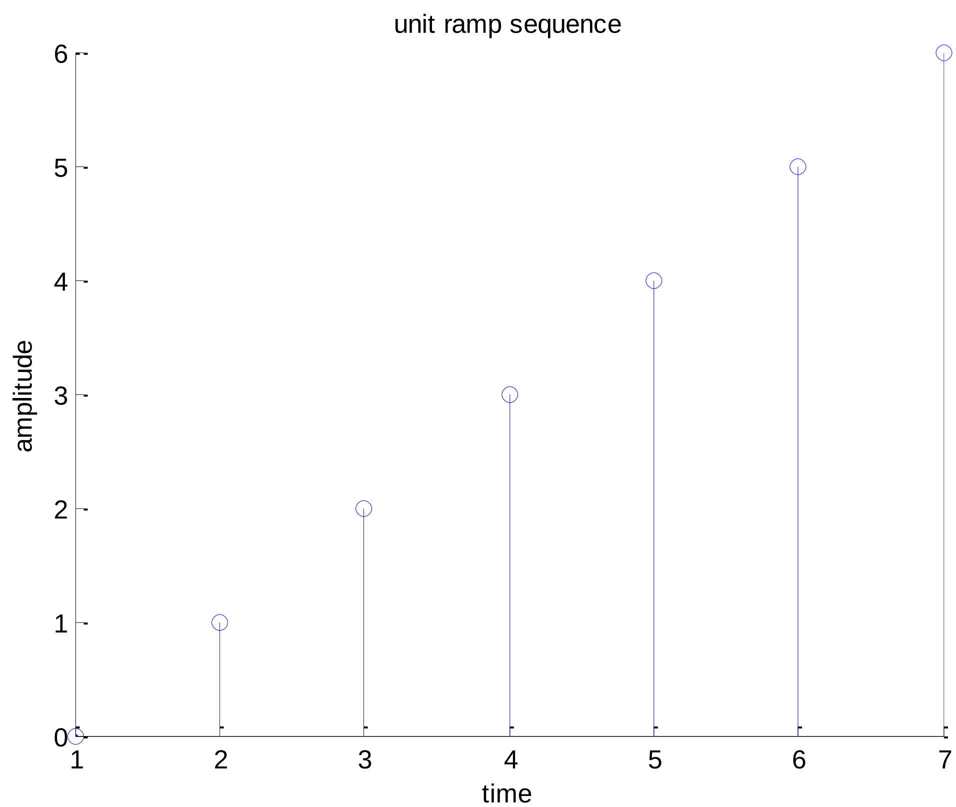
unit step signals
enter the no of samples6



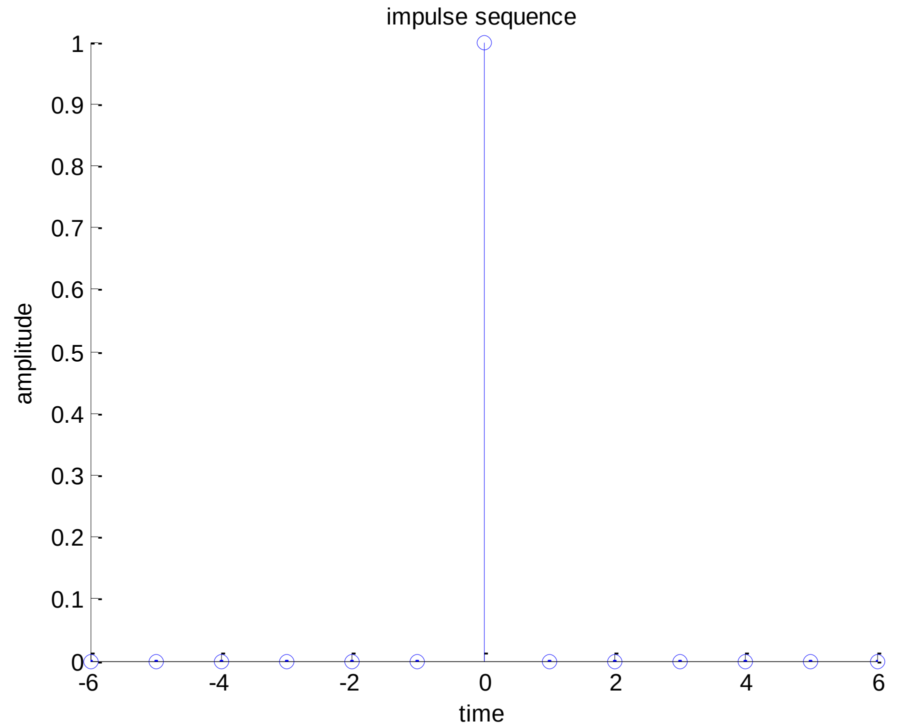
sinusoidal signals
enter the no of samples6



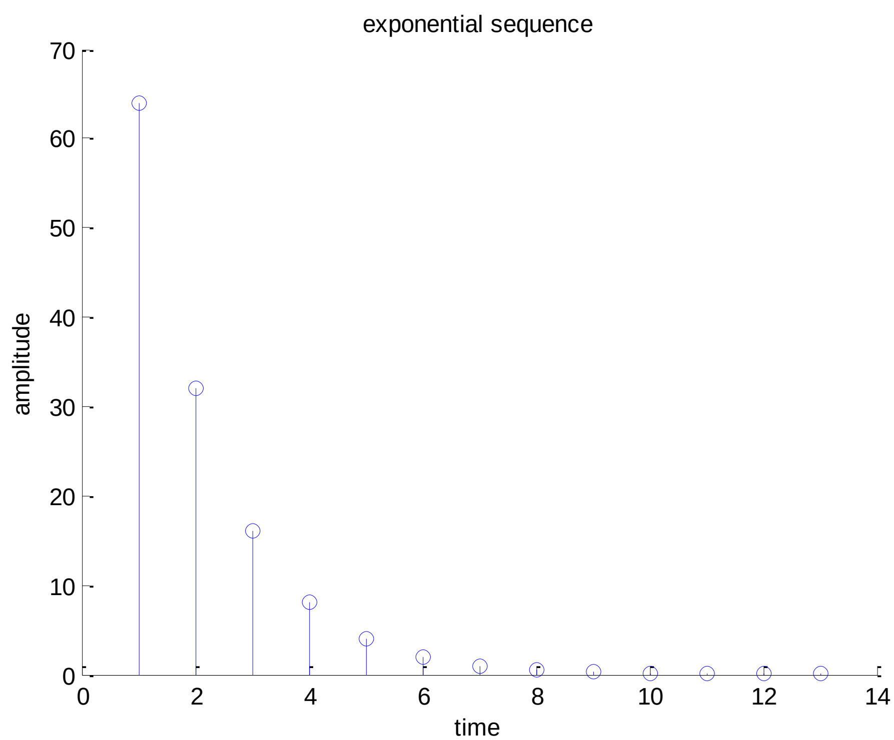
unit ramp signals
enter the no of samples6



unit impuse signal
enter the no of samples6



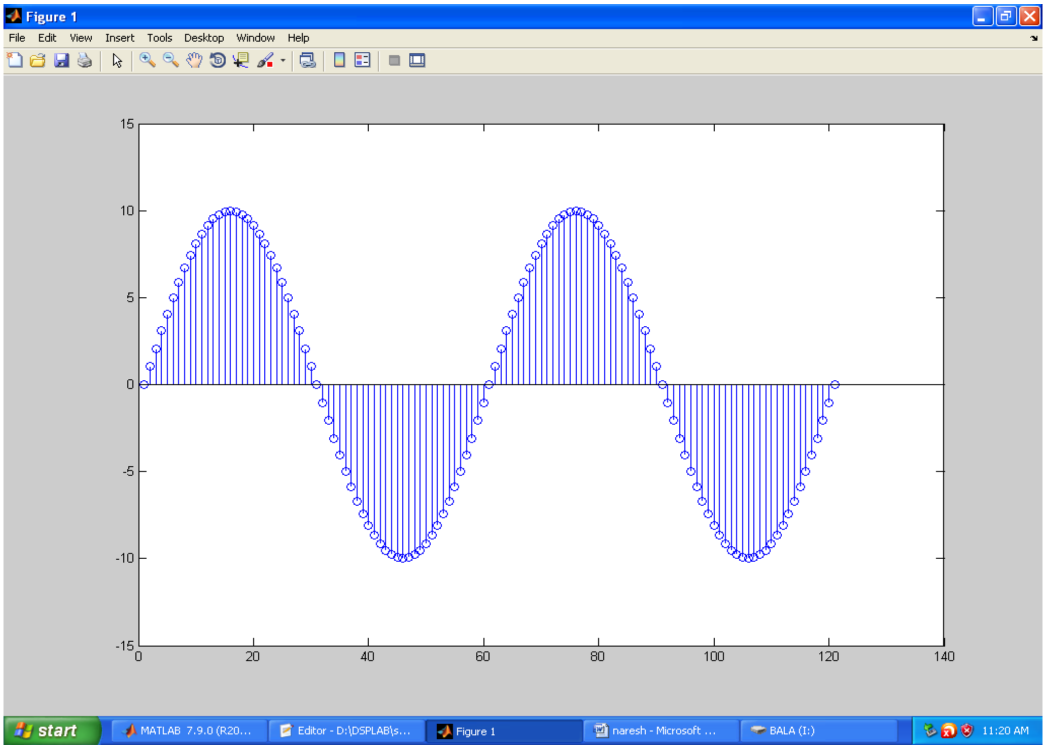
exponential signals
enter the no of samples6



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

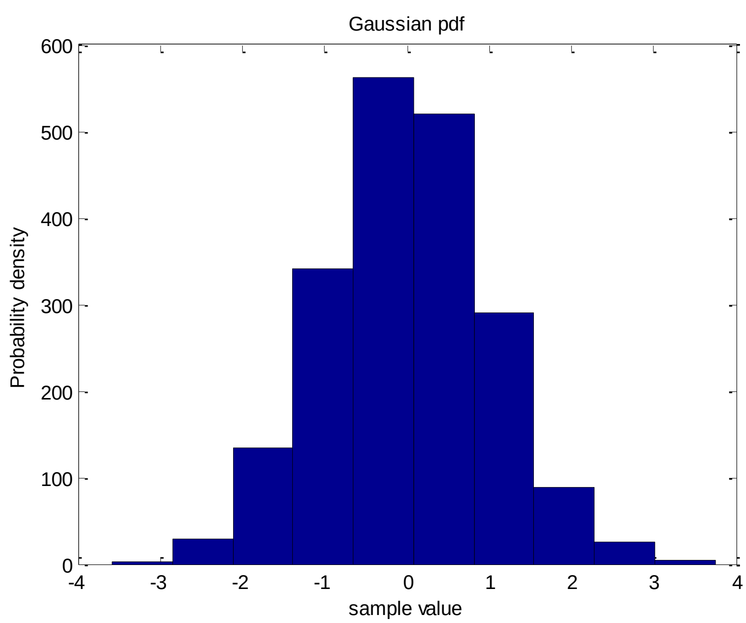
PROGRAM:-

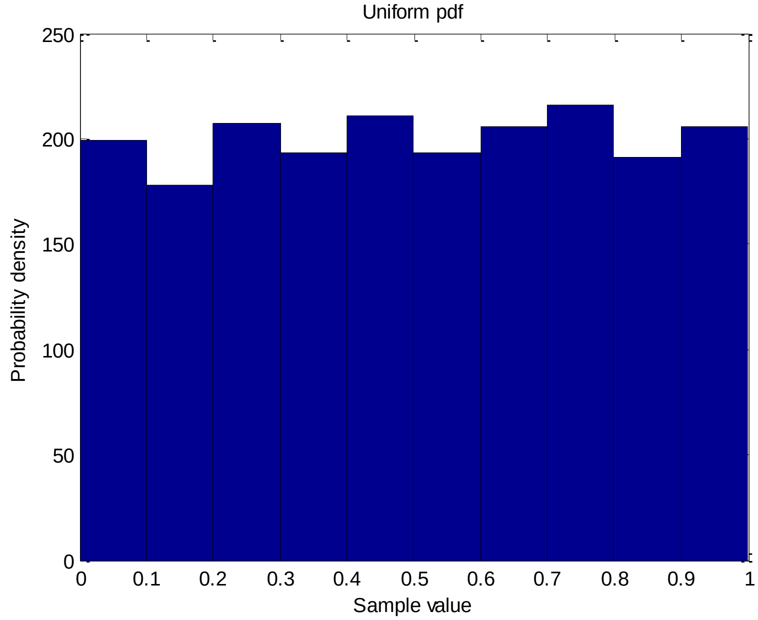
```
clc;
clear all;
close all;
x=randn(1,2000);
y=rand(1,2000);
hist(x);
xlabel('Sample value');
ylabel('Probability density');
title('Gaussian pdf');
figure;
hist(y);
xlabel('Sample value');
ylabel('Probability density');
title('Uniform pdf');
mx1=mean(x); display(mx1);
```

```
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:-

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%

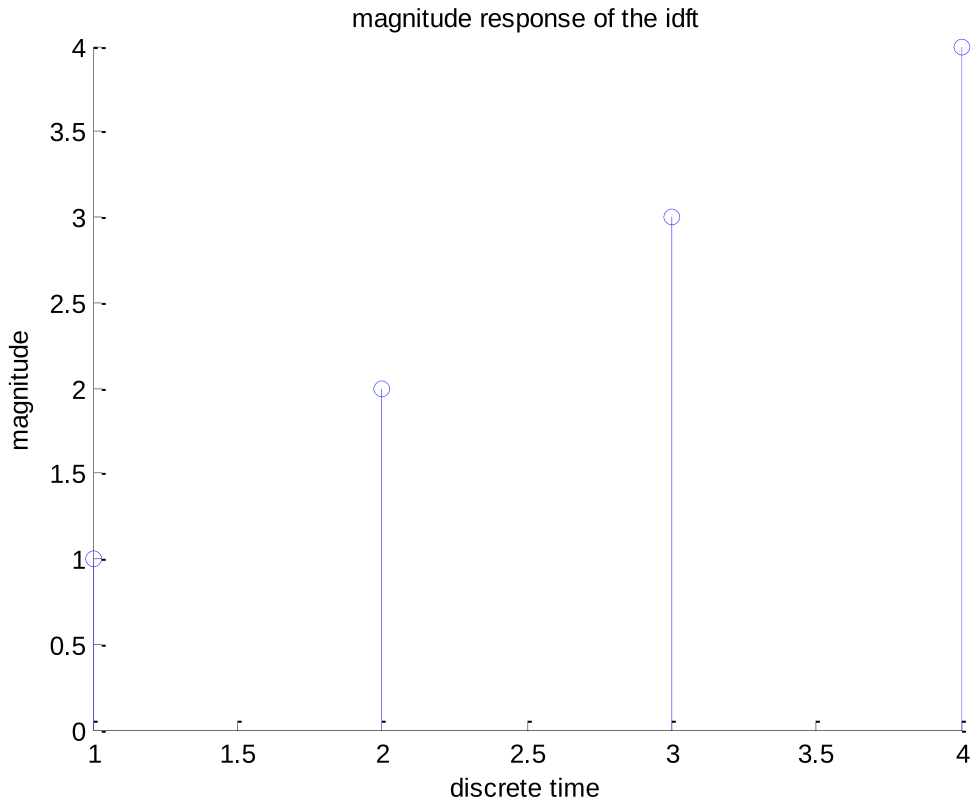
```
clc;
clear all;
close all;
a=input('enter the input sequence');
disp('the length of input sequence is');
N=length(a);
N
for n=1:N;
x(n)=0;
for k=1:N;
x(n)=x(n)+a(k)*exp((j*pi*2*(n-1)*(k-1)/N));
end;
end;
n=1:N;
```

```
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;
```

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

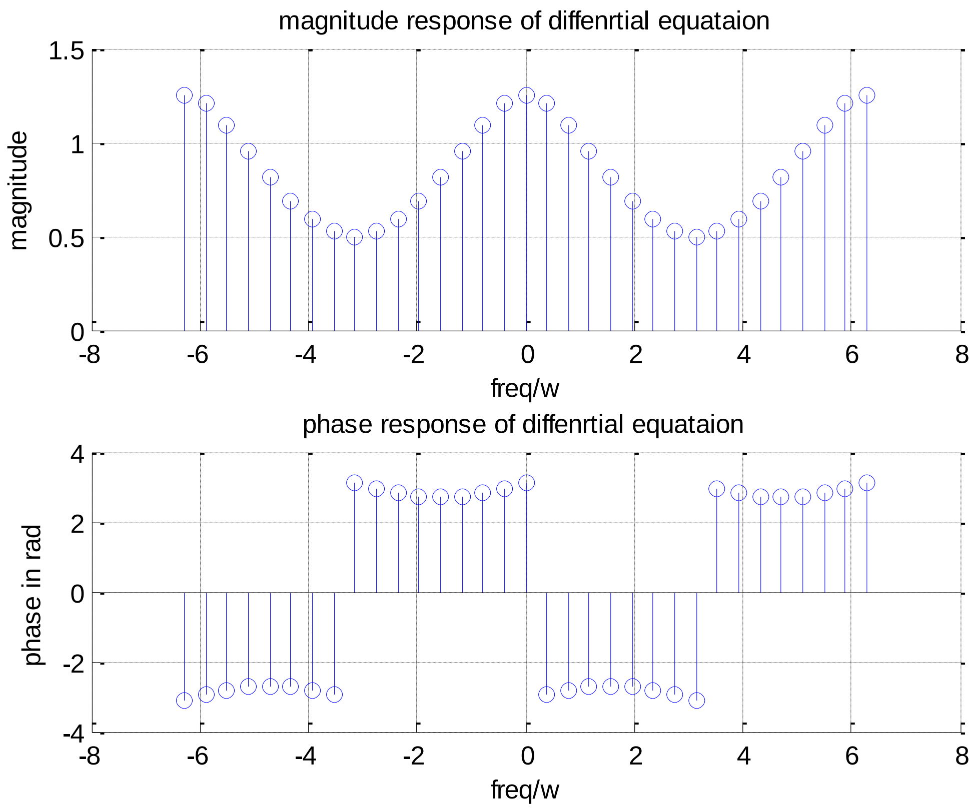
PROGRAM:-

%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.

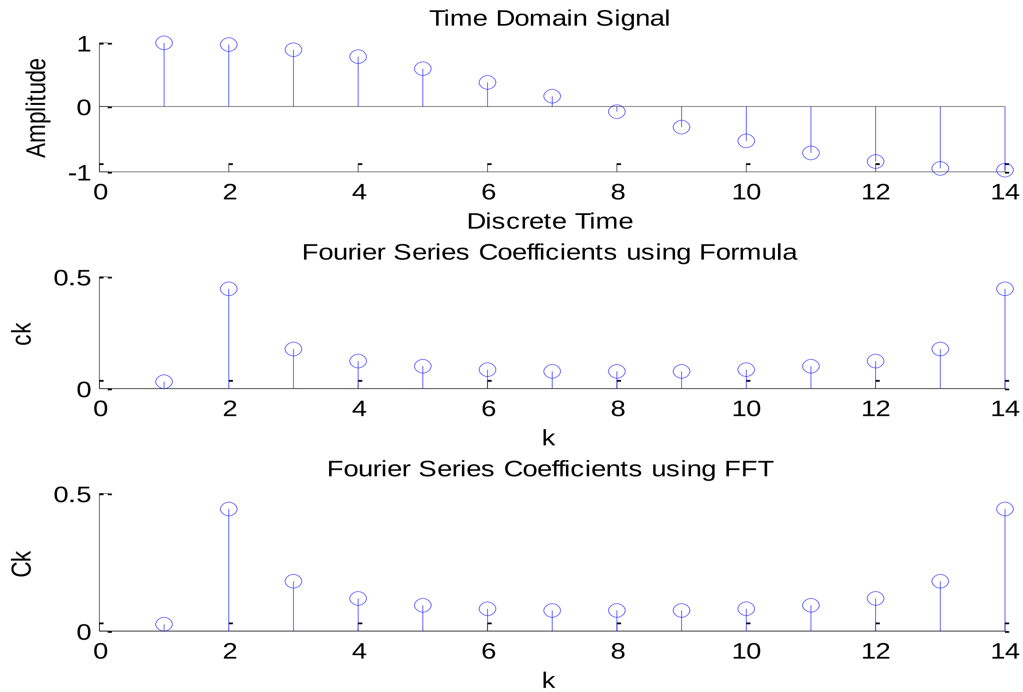
MATLAB Program:

```
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 l $f(t) = \sin t$?
12. Determine the Fourier series coefficients for a Saw-tooth Wave ?
13. Consider the function below $\{ 1-x \quad 0 < x \le 1;\ 0 \quad 1<x\le 2 \}$
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(5πt)
    x(t) = [1+sin(5πt)] [cos(14πt+π/9)] ?
20. Determine the Fourier series coefficients of the following signal
    x(t) = 1+cos(2πt)
    x(t) = [1+tan(2πt)] [cot(10πt+π/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:-

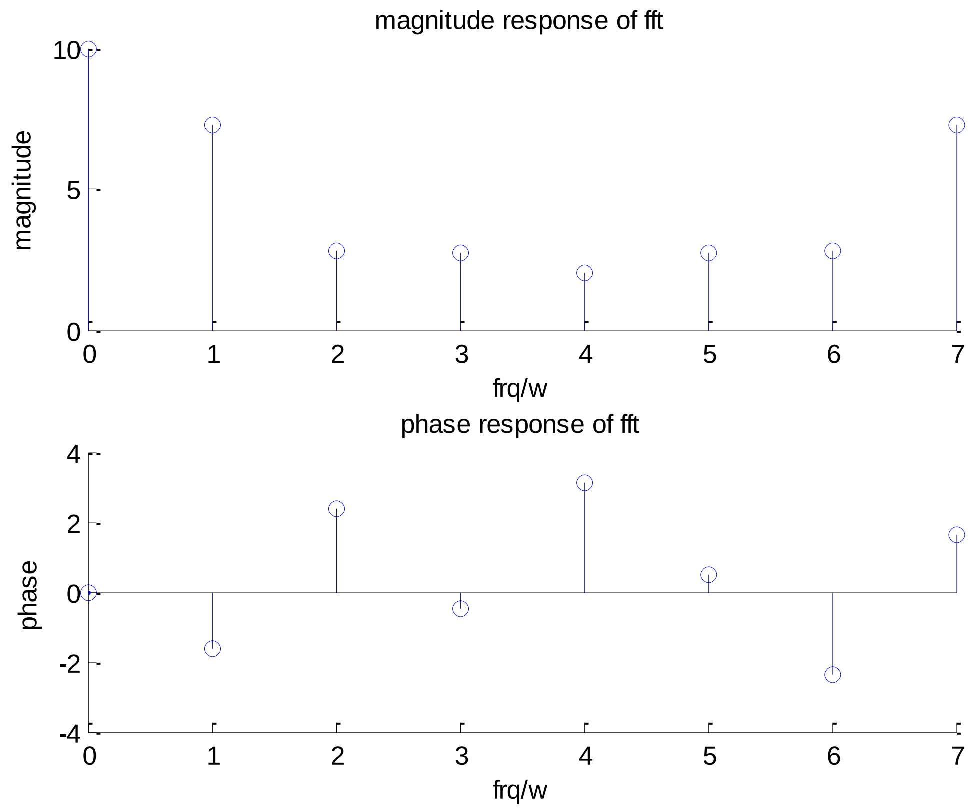
- Open MATLAB
- Open new M-file
- Type the program
- Save in current directory
- Compile and Run the program
- For the output see command window\ Figure window

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:-

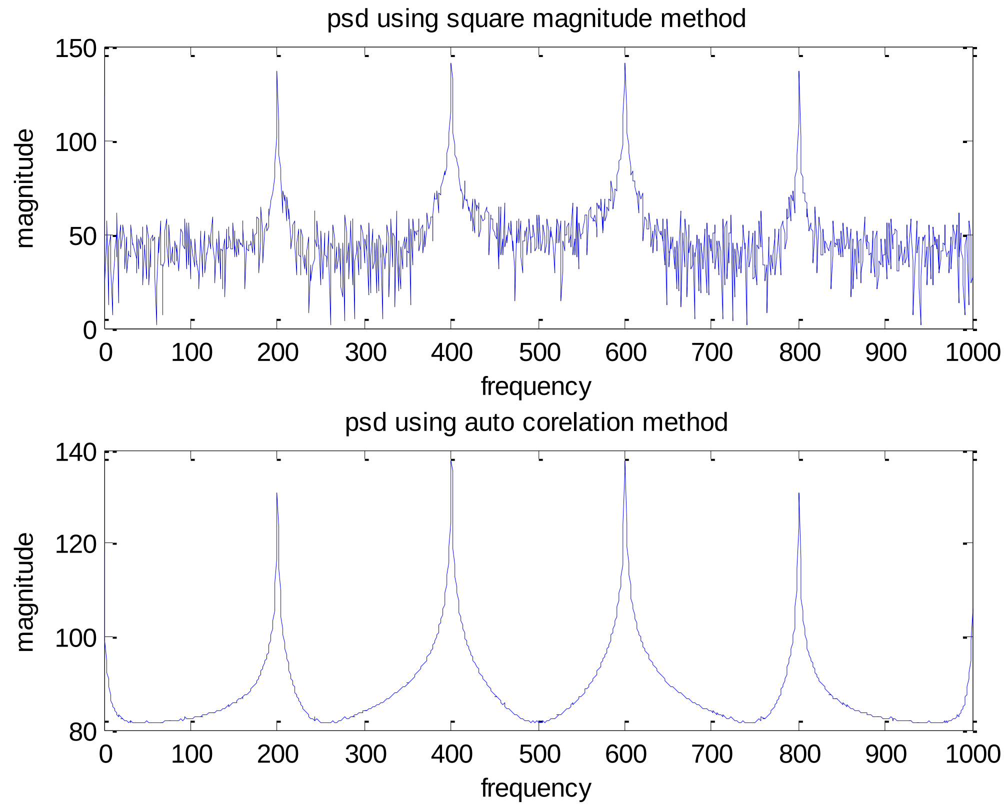
- Open MATLAB
- Open new M-file
- Type the program
- Save in current directory
- Compile and Run the program
- For the output see command window\ Figure window

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:-

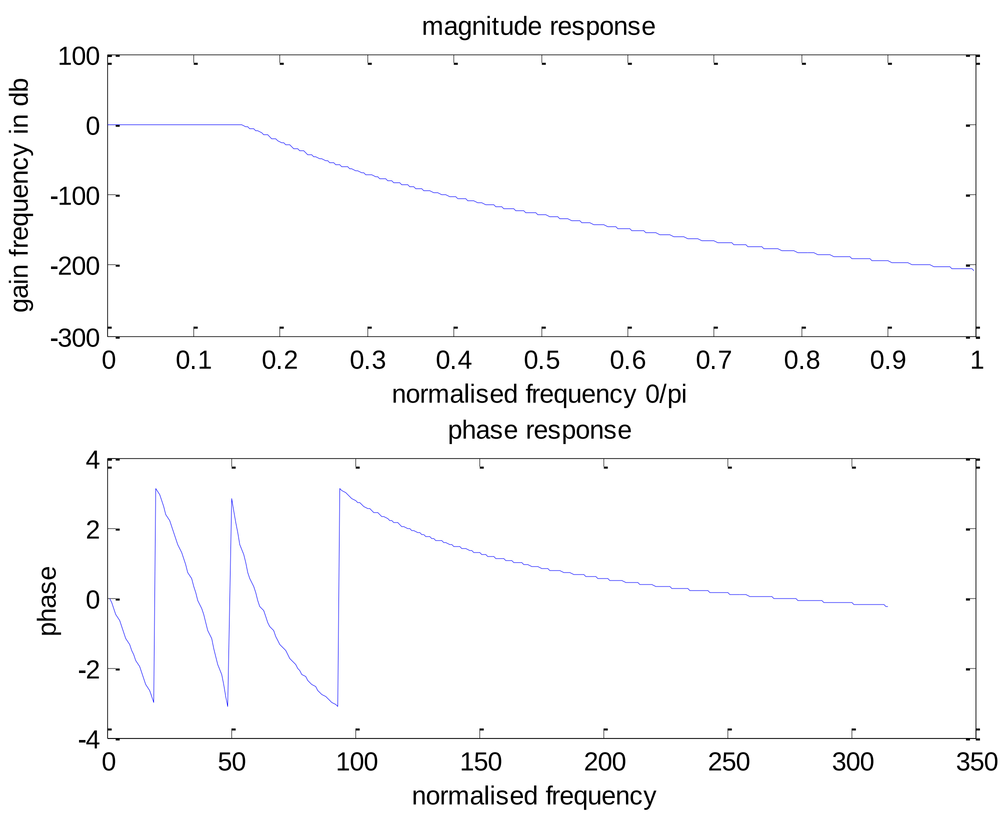
enter the pass band ripple:rp=0.15

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

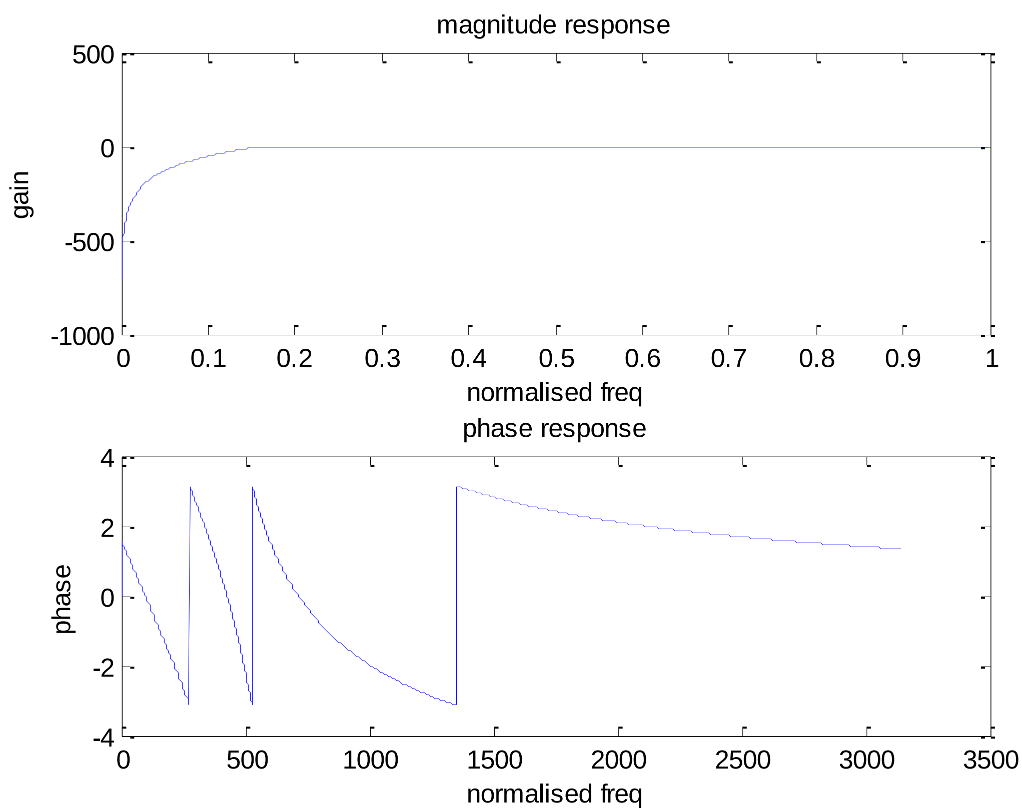
enter the stop band ripple:rs=60

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:-

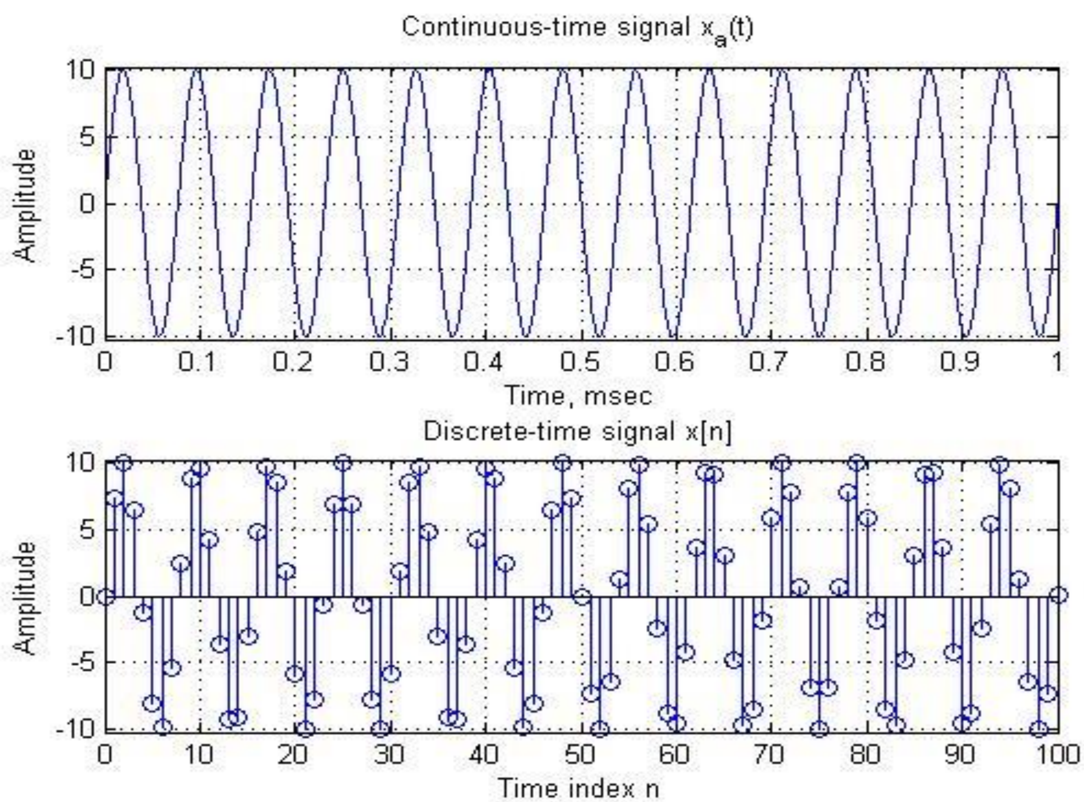
- Open MATLAB
- Open new M-file
- Type the program
- Save in current directory
- Compile and Run the program
- For the output see command window\ Figure window

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:-

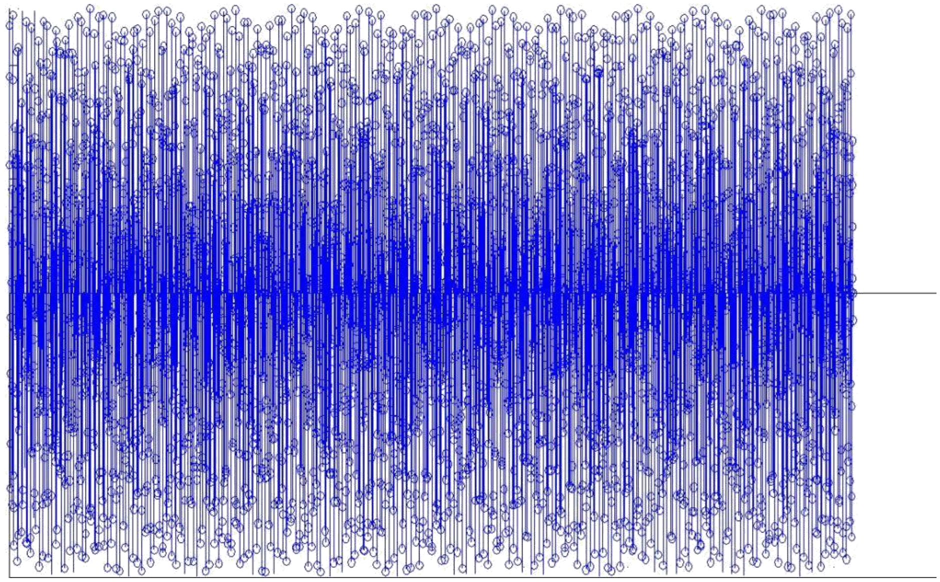
% Dual Tone Multiple Frequency (DTMF) Signals.

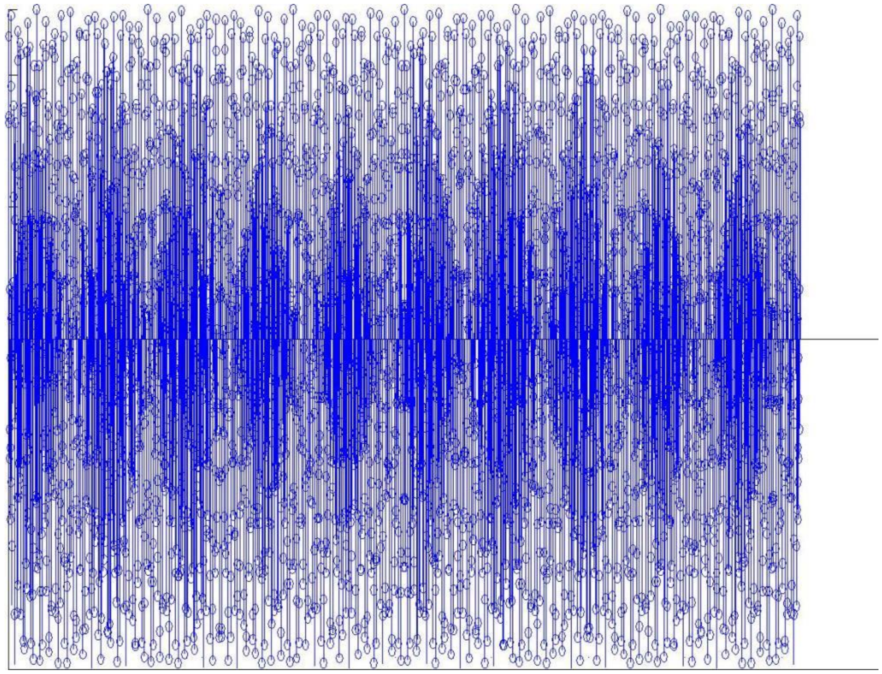
```
clc;

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)
```

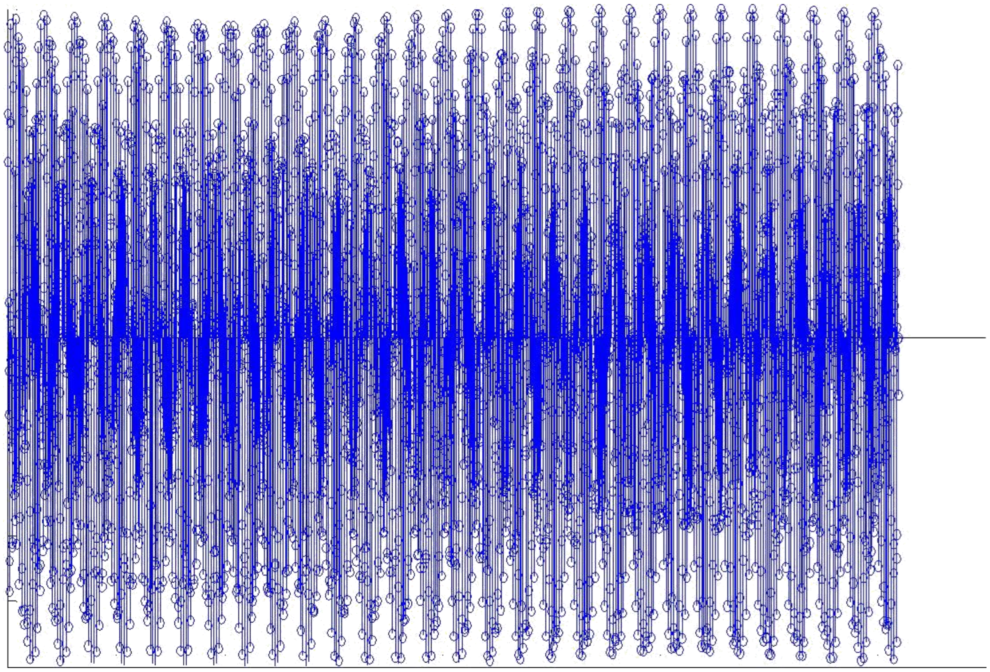
```
switch
    number(k)
    case '1'
      tone =
      tx(:,1)+ty(:,1);
      sound(tone);
      stem(tone);
    case '2'
      tone =
      tx(:,1)+ty(:,2);
      sound(tone);
      stem(tone);
    case '3'
      tone =
      tx(:,1)+ty(:,3);
      sound(tone);
      stem(tone);
    case '4'
   tone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

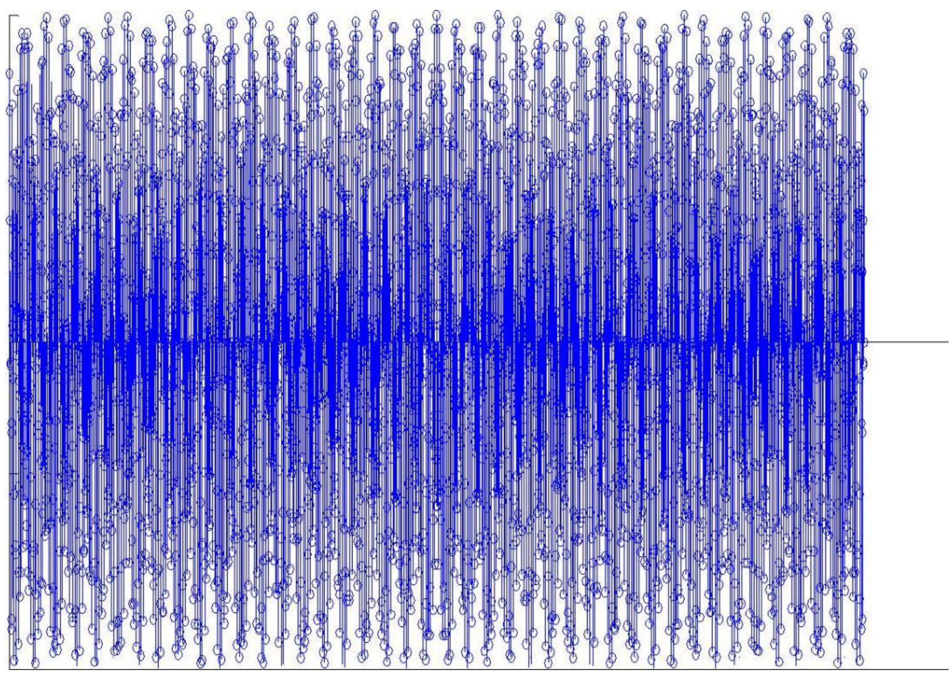
OUTPUT:-

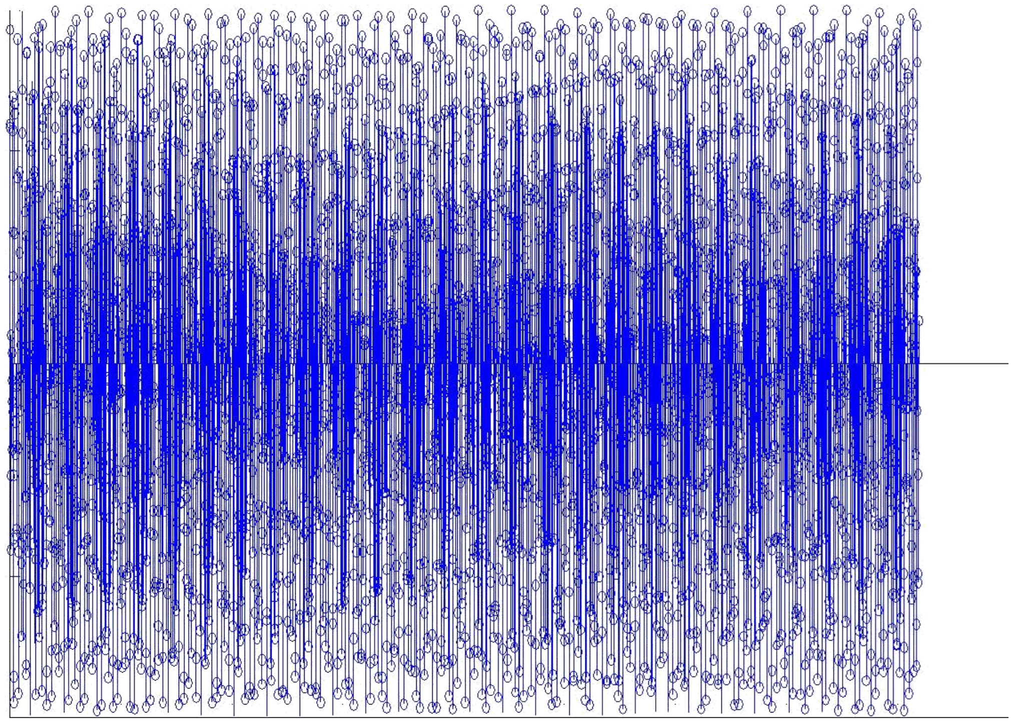
Input: 01234











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:-

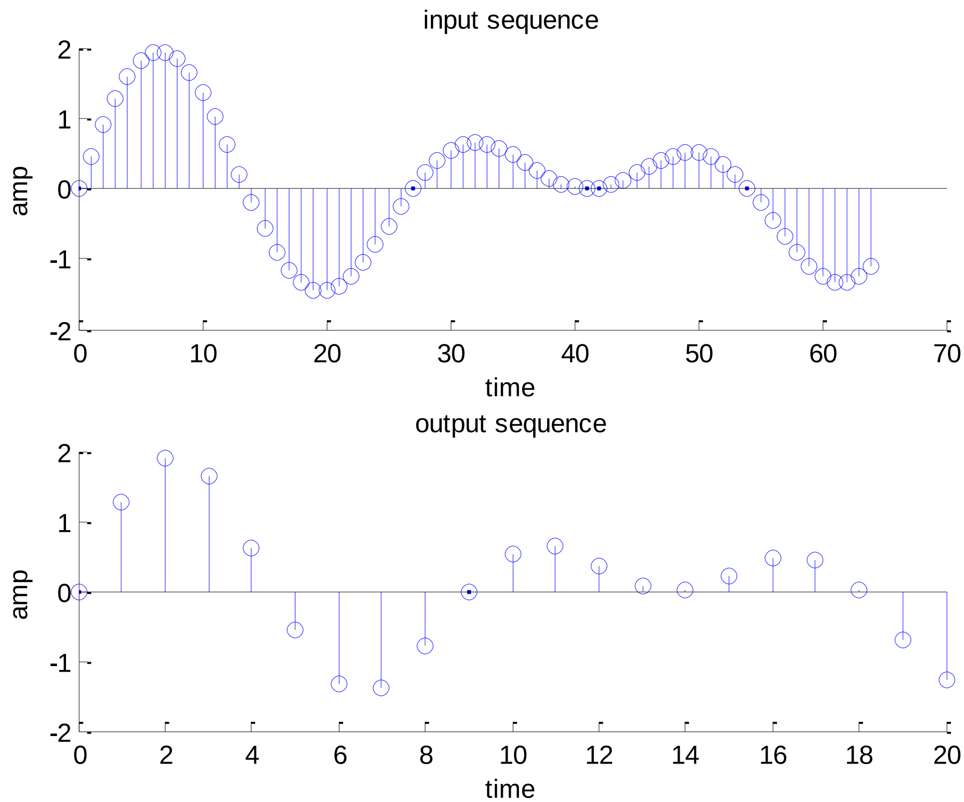
- Open MATLAB
- Open new M-file
- Type the program
- Save in current directory
- Compile and Run the program
- For the output see command window\ Figure window

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:-

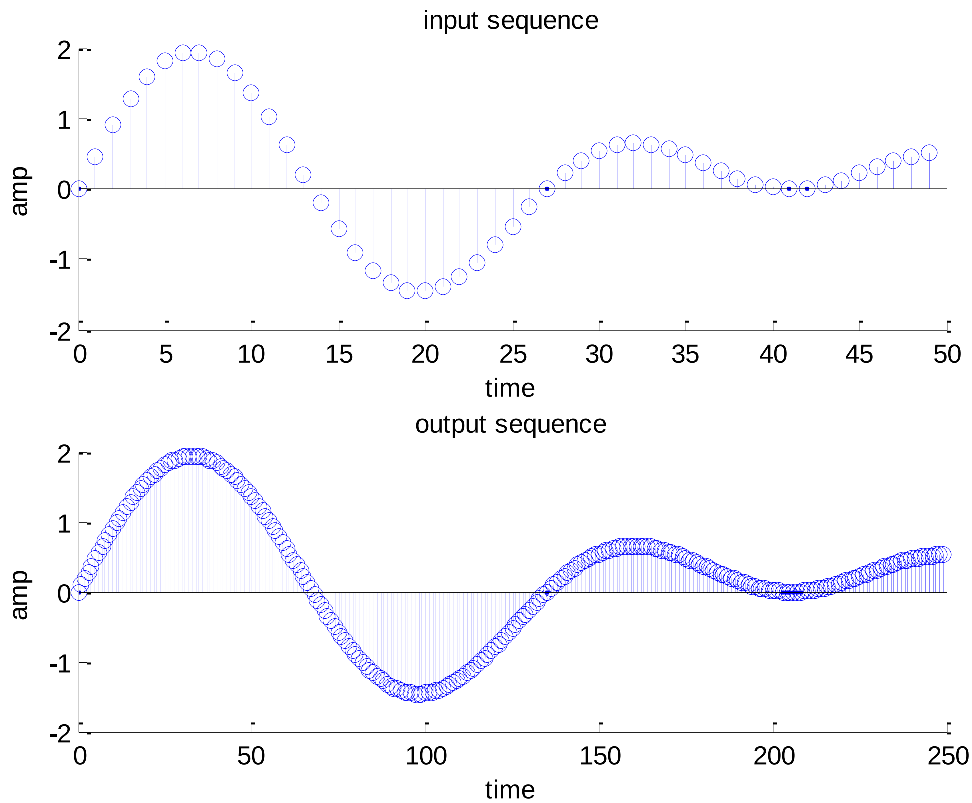
- Open MATLAB
- Open new M-file
- Type the program
- Save in current directory
- Compile and Run the program
- For the output see command window\ Figure window

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:-

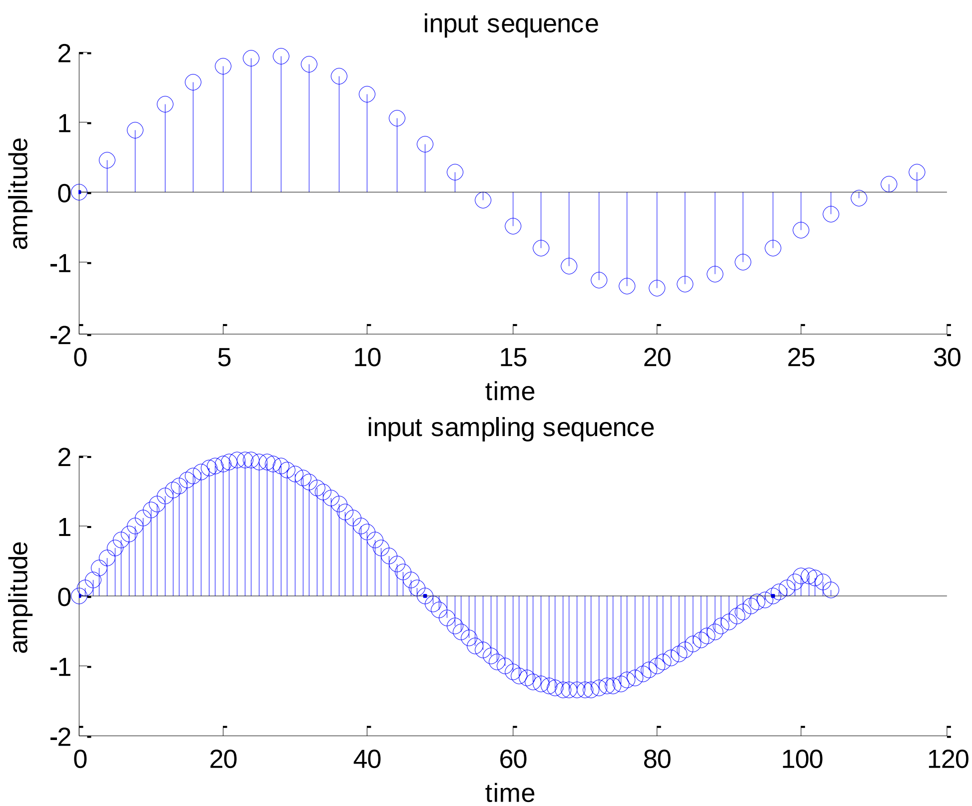
% interpolation/dismation sampling %

```
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

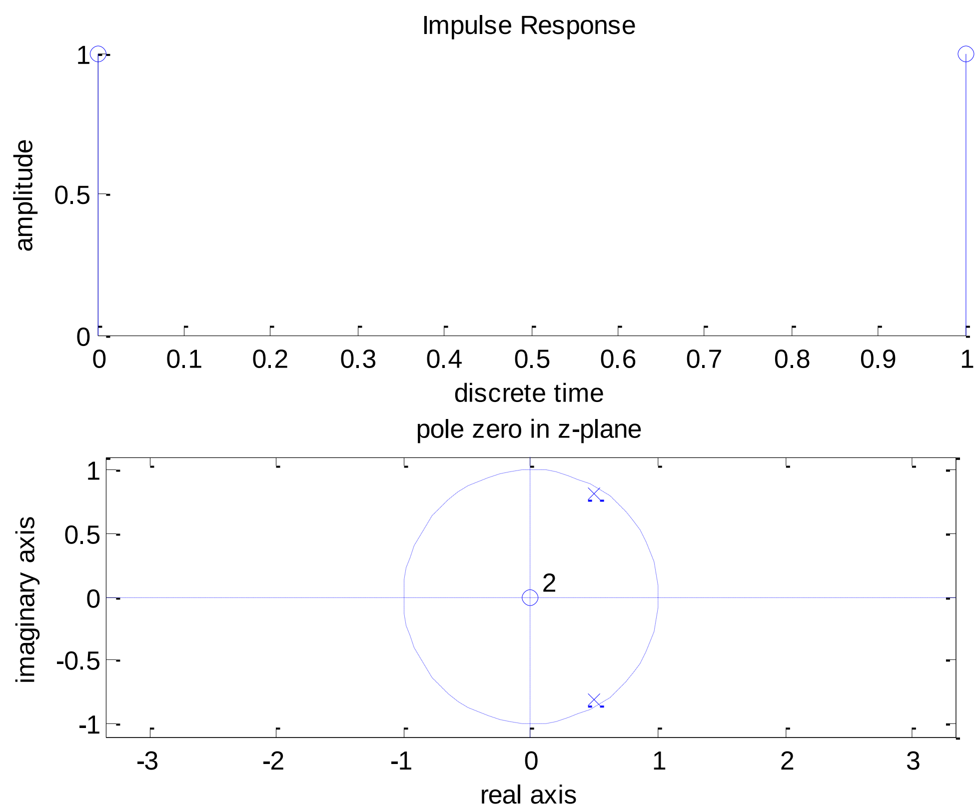
PROGRAM:-

```
%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( )