Experiment - 1

RAISONI GROUP
a vision beyond

Aim :- To write a MATLAB program to generate discrete time signal.

Algorithm :-

- Get the amplitude & Frequency of signal
- Use 'sin', 'cos', 'square' matlab built in functions
- Using 'plot' function plot the signal.

Theory :-

Discrete-time signals.

- In signal processing, a discrete time signal is sequence of value that are defined only at discrete point in time. Each value of signal correspds to specific discrete time index. Discrete-time signals are commonly represented in MATLAB using arrays or vectors, where each element of array represents the value of signal at specific time index.

Types of Discrete time signals

- There are various types of discrete-time signals commonly encountered in signal processing.

1) Step function - A step function, also known as a unit step function, is a signal that abruptly changes its value from one constant le

to another at specific time index.

2) Ramp function - A ramp function is a signals that linearly increase or decrease its value over time.

$

3) Sinusoidal function - A sinusoidal function is signal that oscillates sinusoidally over time. It is characterized by parameters such as amplitude, frequency, phase & sometimes a DC offset.

Conclusion - Hence, we have studied the how to write MATLAB program to generate discrete time signals.

Experiment - 2

Aim - To write a MATLAB program to find the impulse response of system defined by difference equation.

Algorithm :-

- Create a matrix a for coefficient of y[n].
- Create a matrix b for coefficient of x[n].
- Generate an impulse signal.
- Find the response h[n] of system defined by a & b coefficient.
- To the impulse signal using 'filter' command.

Theory :-

Impulse response

In signal processing & systems theory, the impulse response of system is output of system when its input is an impulse function, typically denoted as δ[n] in discrete-time systems. An impulse function is a signal that is zero everywhere except at one point, where it is ~~dit~~ infinitely tall.

Difference equation

A difference equation describes the relationship between the input and output of discrete-time system. It expresses how the current and past input and output of system are related to produce th

current output. A typically linear-invariant discrete-time system can be represented by linear constant coefficient difference eq^n.
The general form of linear constant coefficient difference equation is

$$y[n] = \sum_{k=0}^{N} b_k \cdot x[n-k] - \sum_{m=1}^{M} a_m \cdot y[n-m]$$

Finding Impulse response from Difference equation

~~y[n] is~~ To find the impulse response of system defined by difference equation, you can exploit the fact that when, the input is impulse function ($\delta[n]$), the output is simply the impulse response of system. By setting the input ($x[n]$) to $\delta(n)$ in the difference equation & solving for output ($y[n]$) you obtain the impulse response of system.

Conclusion - Hense, we have studied how to write MATLAB program to find impulse response of system defined by difference equation.

RAISONI GROUP
a vision beyond

Experiment - 3

Aim :- To perform following basic operations on discreate signal

1. Addition
2. Substraction
3. Multiplication
4. Shifting signal
5. Reversing a signal

Theory :-

Addition - perform addition of $x(n) = [1\ 2\ 3\ 4]$
$x_1(n) = [1\ 1\ 1\ 1]$

Step 1 :- After the 2 signals are defined and the duration of input signal using min and max function

Step 2 :- Generating a zero matrix of 1 row having elements with length = durration of output.

- Making the length of input signal equivalent to that of output. we have to generate two signals S1 & S2
- S1 & S2 are generated as a zero matrix using zero function.
- An array of 1 is created in the respective position which satistics the condition statement is S1 (find (len3) > min(m1)) & S3 <= max(n1)) = x
- Signal ~~s1~~ with duration of output signal 3 2 find (+n3) >= min(n2)) & (n 3 < = max(n2) == 1) = 1

RAISONI GROUP
a vision beyond

Function used:-

1) min & max():

- min() returns smallest element in an array if I and y are array.
- min(x, y) return smallest element with same size of x & y element of corresponding x & y must be same length.
- max() returns largest element in an array if x is an array.
- max(x, y) return an array with same size of x & y.

2) zero(): Syntax - zero(), zero(min)
zero(min, p...)

3) find(): syntax
- find(x,y) return linear indices of zero element in an array x..

e.g. if x = [0, 4, 0, 5, 6]
find x: output = 2 4 5

4) axis(): syntax
- Axis ([x min, x max, y min, y max]), sets limits for x & y axis.

5) stem(): used to discrete time pointing signal.

2) Substraction :- Perform substraction of $x_1(n)$=[12 and $x_2(n)$= [1111] is plot output

3) Multiplication - Perform multiplication of $x_1(n) = [1\ 3\ 5\ 6]$
$x_2(n) = [4\ 5\ 7\ 8]$ and plot output

4) Shifting the signal : matlab can be used to perform shifting of signals. A signal can be delayed as well as advanced.

5) Reversing a Signal : The inbuilt function can be used to perform reversing or folding a signal.

Syntax :- fliplr(a) :- row vector results a vector with same size.

Syntax

(1) conv (a,b) : convolves the vector a & b.

(2) 'f' variable used in the code is length of output, As you may be aware that total length of convoluted output will be | l-m-1|

Conclusion : Hence, we have studied the basic operations on discreate signals.

RAISONI GROUP
a vision beyond

Experiment - 4

Aim :- Find out z-transform & inverse z-transform using matlab.

Tool :- Matlab Software

Theory :-

z- transform :-

It is a transformation that map discreate time signal x(n) into a function is the complex variable z namely. It can be considered as a discreate time equivalence of laplase transform. This similarity is explored in the theory of time scale calculus.

The expression for calculating z-transform is given as.

$$x(z) = \sum_{n=-\infty}^{\infty} x(n) z^{-n}$$

$$H(z) = \frac{\sum^{N-1} bk - z^{-k}}{\sum^{N-1} ak\, z^{-k}}$$

$$H(z) = \frac{b_0}{a_0} z^{-MN} \frac{(z-z_1)(z-z_2)\ldots(z-z}{(z-p_1)(z-p_2)\ldots(z-p}$$

$$H(z) = H z^{N-M} \frac{[\,]^{na}_{k>1} (z-z_k)}{(\,)^{N}_{k<1} (z-z_k)}$$

conclusion - Hence, we have studied z-transform & Inverse z-transform

RAISONI GROUP
a vision beyond

Experiment - 5

Aim - To write a MATLAB program to find the poles, zeros & to plot pole-zero map in z-plane.

Software required :- MATLAB

Procedure :-
- open MATLAB software
- Open new M-file
- Type the program
- Save in current directory
- Run the program
- For output see command window/ figure window.

Theory -
poles and zeros
- In the context of discrete-time systems poles and zeros are fundamental concepts that describes the behavior of system in z-domain

- zeros :- zeros of discrete-time system are various of z- for which system's transfer function becomes zero.

- Poles :- poles of discrete-time system are the values of z for which system's transfer function becomes infinite.

Pole - zero map in z-plane

A pole-zero map, also known as pole-zero plot, is graphical representation of poles & zeros of system in complex z-plane. The positions of poles & zeros in z plane provide valuable insights into stability & frequency.

- Poles :- The poles of system are typically denoted by x marks in pole-zero plot. The duration of poles in z-plane indicates stability indiates in z-plane of system.
- zeros :- The zeros of system are typically denoted by 'o' marks in pole-zero plot. zeros affect the system's frequency response by cancelling out certain frequencies.

Conclusion - Hence, we have studied How to write MATLAB program to find poles, zeros & to plot-zero map in z-plane.

Experiment - 6

RAISONI GROUP
a vision beyond

Aim :- write a MATLAB program to find DFT of sequence.

Algorithm :-

- Enter the input sequence $x[n]$
- Enter length of sequence, N
- Use the MATLAB function 'fft'.
- plot the input & output sequence.

Theory :-

Discrete - Fourier transform

- DFT is mathematical tenrique used to analyze the frequency content of discrete-time signal. It converts a sequence of complex numbers, which represents the value of signal sampled at equally spaced point in time, into another sequence of complex numbers, which represent the signal's frequency components.

DFT equation

- The formula for DFT of sequence $x[n]$ of length N is given by

$$x[K] = \sum_{n=0}^{N-1} x[n] \cdot e^{-j2\pi \frac{kn}{N}}$$

RAISONI GROUP
a vision beyond

Where,
- X[k] is k-th element of DFT sequence.
- x[n] is input signal.
- $e^{-j2\pi k\frac{n}{N}}$ represents the complex exponential function.
- N is the length of input sequence.
- k is frequency index.

Conclusion - Hence, we have studied how to write MATLAB program to find DFT of sequence.

RAISONI GROUP
a vision beyond

Experiment - 7

Aim: To write a MATLAB program to find IDFT of sequence.

Algorithm:-

- Enter the output sequence y[n]
- Enter the length of sequence, N
- Use the matlab function 'ifft'.
- plot the input & output sequence.

Theory -

Inverse Discrete - fourier Transform

- The inverse discrete fourier transform is the process of converting a sequence of complex numbers, which represent the frequency component of signal, back into original time domain sequence.

It is inverse operation of discrete Fourier transform (DFT). The Inverse discrete fourier transform allows us to recover the time-domain signal from its frequency-domain representation.

Experiment - 8.

Aim:- To write MATLAB program to plot magnitude response & phase response of digital butter worth Low pass filter.

Algorithm:-

- Get the passband and stopband ripples
- Get the passband & stop band edge frequencies.
- calculate the order of filter using 'buttord' function.
- Find the filter coefficients, using 'butter' function.
- Draw the magnitude & phase response.

Theory:-

- Butterworth low-Pass filter:-
  - The butterworth filter is a type of analog or digital filter characterized by maximally flat frequency response in the pass band. In the digital domain, a butterworth filter is commonly used for various signal processing task such as smoothing, noise reduction & anti-aliasing.

RAISONI GROUP
a vision beyond

Magnitude Response:-

- The magnitude response of digital filter describes how the filter affects amplitude of difference frequencies in the input signal. for low - pass filters, the magnitude response shows the attenuation of higher frequencies while allowing lower frequencies to pass through.

Phase response:-

The phase response of digital filter describes the phase shift introduced by filter at different frequencies. It indicates the time delay or advance experienced by each frequency component of input signal as it passes through filter.

Conclusion:- Hence, we studied how to plot magnitude response & phase response of digital butter worth Low pass filter.

Experiment - 9

RAISONI GROUP
a vision beyond

Aim :- Write a MATLAB program to plot magnitude response and phase response of digital butter worth high pass filter.

Algorithm :-

- Get the passband & Stopband ripples
- Get the passband & stopband edge frequencies.
- Calculate the order of filter using 'buttord' function.
- Find the filter coefficient, using 'butter' function.
- Draw the magnitude & phase response.

Theory -

Butterworth - high pass filter.

- Similar to low-pass filter, the butter-worth high-pass filter is another type of digital filter characterized by maximally flat frequency response However, unlike the low-pass filter, the high-pass filter attenuates lower frequencies while allowing higher frequencies to pass through.

RAISONI GROUP
a vision beyond

Magnitude response-

The magnitude response of butterworth high-pass filter shows the attenuation of lower frequencies & the passband behavior of higher frequencies. It is typially plotted on logarithmic scale to visualize the filter's frequency selectivity.

Phase response:

The phase response of butterworth high pass filter describes the phase shift introduced by the filter at different frequencies. It indicates the time delay or advance experienced by each frequency component of input signal as it passes through filter.

Conclusion - hence, we studied how to plot magnitude response & phase response of digital butter worth high pass filter.

Experiment - 10.

Aim:- To write a MATLAB program to plot magnitude response & phase response of digital Butterworth Band pass filter.

Algorithm:-

- Get the passband & Stopband ripples
- Get the passband & Stopband edge frequencies
- Calculate the order of filter using buttord function
- Find the filter coefficient, using 'butter' function.
- Draw the magnitude & phase response

Theory:-

Butterworth Band-Pass filter

A Butterworth band-pass filter is a type of digital filter that selectively passes a range of frequencies while attenuating frequencies outside this range. It combines the characteristics of low-pass & high-pass filters to create a band-pass response.

magnitude Response:
- The magnitude response of Butter-worth band pass filter shows the amplitude of output signal as function of frequency within the passband. It is typically exhibits a peak or peaks frequency within passband and attenuates frequencies outside the passband

Phase response:
- The phase response of Butterworth band-pass filter describes the phase shift at different frequencies within the passband. The phase response can exhibit nonlinear behaviour, specially near the edge of passband.

Conclusion - hence, we have studied to plot magnitude response & phase response of digital Butterworth Band pass filter.