# Jawaharlal Nehru Engineering College

*Laboratory Manual*

# DIGITAL SIGNAL PROCESSING

**For**

THIRD YEAR ENGINEERING

Manual made by

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# MGM'S

# Jawaharlal Nehru Engineering College

**N-6, CIDCO, Aurangabad**

# Department of Electronics &Telecommunication

## Vision of the Department:

To develop **GREAT** technocrats and to establish centre of excellence in the field of **Electronics and Telecommunications.**

- ***G***lobal technocrats with human values
- ***R***esearch and lifelong learning attitude,
- ***E***xcellent ability to tackle challenges
- ***A***wareness of the needs of society
- ***T***echnical expertise

## Mission of the Department:

1. To provide good technical education and enhance technical competency by providing good infrastructure, resources, effective teaching learning process and competent, caring and committed faculty.
2. To provide various platforms to students for cultivating professional attitude and ethical values.
3. Creating a strong foundation among students which will enable them to pursue their career choice.

# Jawaharlal Nehru Engineering College

## Technical Document

This technical document is a series of Laboratory manuals of Electronics & Telecommunication and is a certified document of Jawaharlal Nehru Engineering College. The care has been taken to make the document error free but still if any error is found kindly bring it to the notice of subject teacher and HOD.

Recommended by,

HOD

Approved by,

Principal

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**FOREWARD**

It is my great pleasure to present this laboratory manual for third year engineering students for the subject of Digital signal processing keeping in view the vast coverage required for visualization of concepts of basic electronic circuits.

As a student, many of you may be wondering with some of the questions in your mind regarding the subject and exactly that has been tried to answer through this manual.

Faculty members are also advised that covering these aspects in initial stage itself will greatly relieve them in future, as much of the load will be taken care by the enthusiastic energies of the students, once they are conceptually clear. Students are advised to thoroughly go through this manual rather than only topics mentioned in the syllabus as practical aspects are the key to understanding and conceptual visualization of theoretical aspects covered in the books.

Good Luck for your Enjoyable Laboratory Sessions.

Prof. A.P.PHATALE

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5. Design butterworth & chebyshev filters using IIR filters.
6. FIR filters design using linear phase & windows.
7. Illustration of decimation process & interpolation process.
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3. Quiz for the subject.

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5. Evaluation and marking scheme.

## 1. DO's and DON'Ts in Laboratory:

1. Do not handle kit without reading the instructions/Instruction manuals.
2. Refer Help for debugging the program.
3. Go through Demos of Signal Processing tool box.
4. Strictly observe the instructions given by the teacher/Lab Instructor.

## 2 Instruction for Laboratory Teachers:

1. Lab work completed during prior session should be corrected during the next lab session.
2. Students should be guided and helped whenever they face difficulties.
3. The promptness of submission should be encouraged by way of marking and evaluation patterns that will benefit the sincere students.

## **EXPERIMENT NO. 1**:

AIM: Generation of standard signals.

THEORY: Write theory for standard signals.

PROGRAM:

**%generate step signal**

```
n=0:10;                              % Specify the range of n=0 to 10.
d=(n>=0);                            % condition for step signal
stem(n,d);                           % Plot DTS waveform
title('step signal');                % Give title to graph
xlabel('n');                         % Give label to x-axis
ylabel('magnitude');grid on;         % Give label to y-axis
```

**%generate sine signal**

```
x=0:0.05:10;                         % Specify the range of x=0 to 10.
y=sin(x);                            % Function of sine wave.
xlabel('x');                         % Give label to x-axis.
ylabel('sin(x)');                    % Give label to y-axis.
plot(x,sin(x));                      % Plot continuous sine waveform.
```

**%generate exponential signal**

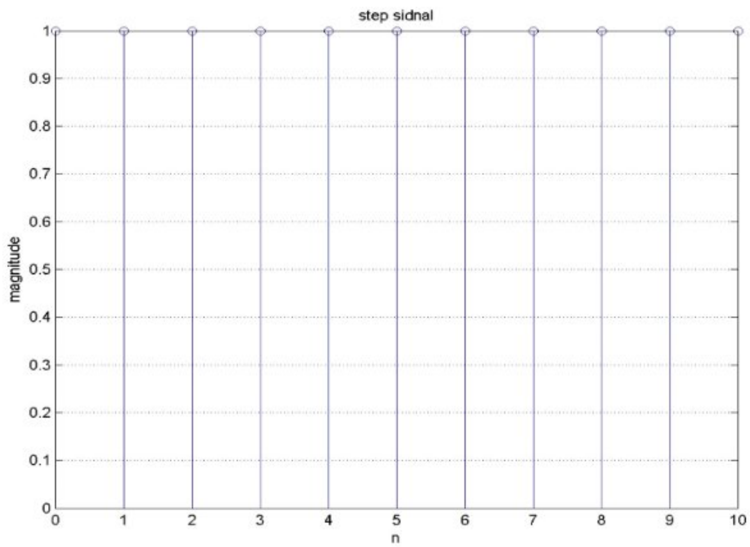
```
x=[0:10];a=5;                        % Specify the range of x=0 to 10.
y=(a.^n);                            % condition for exponential waveform.
stem(x,y);                           % Plot DTS waveform of exponential.
title('exponential signal');         % Give title to graph.
xlabel('n');                         % Give label to x-axis.
ylabel('magnitude');                 % Give label to y-axis.
```

**%generate ramp signal**

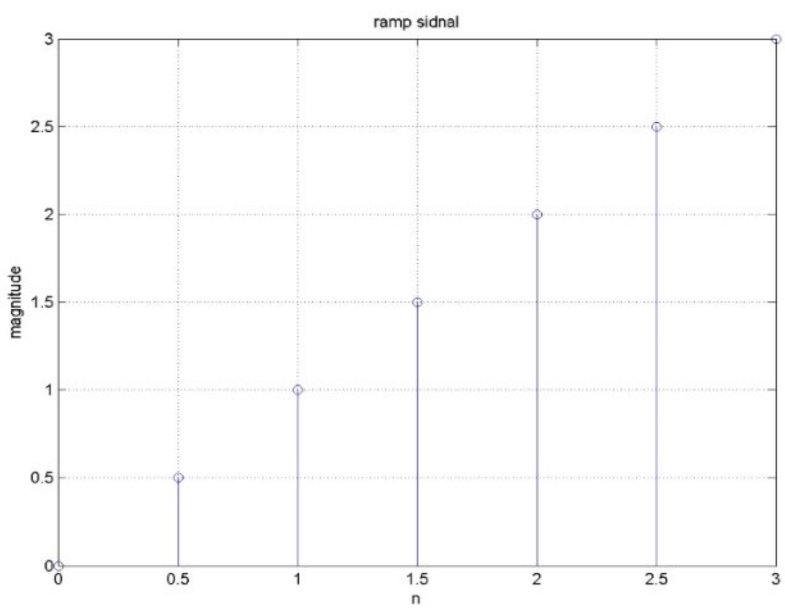
```
n=0:0.5:3;                           % Specify the range of n=0 to 3.
```

```
d=(n);                              % Condition for ramp signal.
stem(n,d);                          % Plot DTS waveform of ramp.
title('ramp signal');               % Give title to graph.
xlabel('n');                        % Give label to x-axis.
ylabel('magnitude');grid on;        % Give label to y-axis.
```

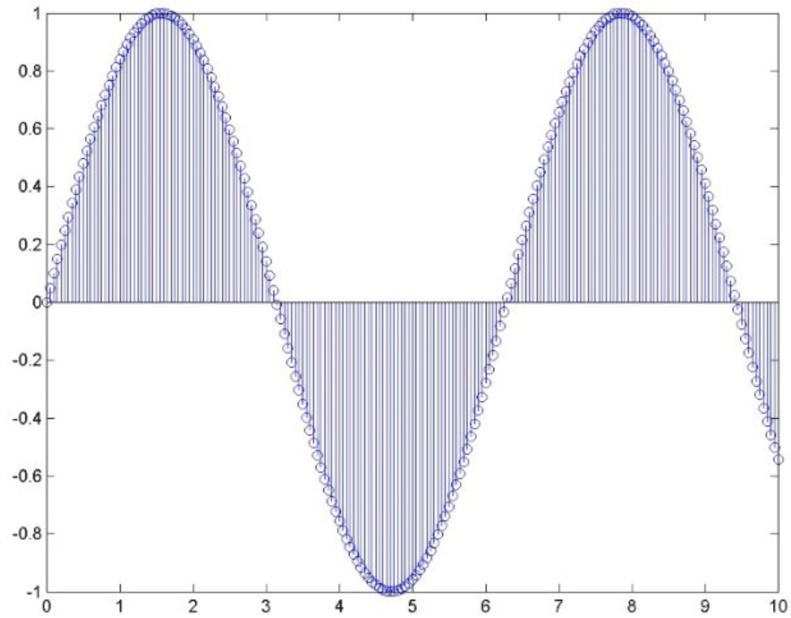
**OUTPUT:STEP SIGNAL**



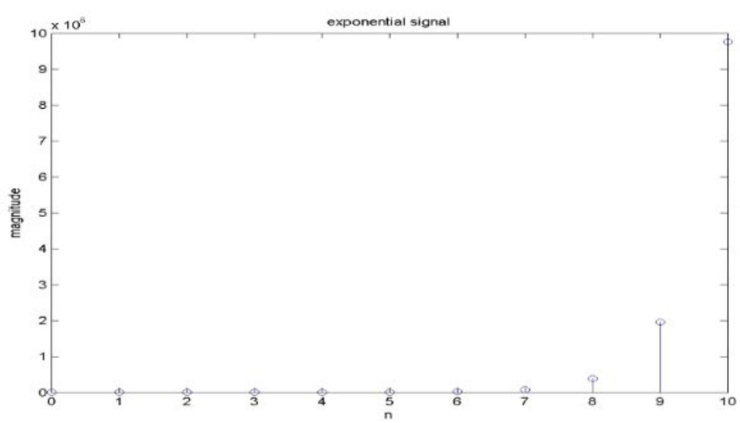

**RAMP SIGNAL**



## SINE SIGNAL



## EXPONENTIAL SIGNAL



# EXPERIMENT NO.2

**AIM:-** Verification of zero pole analysis using transfer function,zero pole gain. Express the following Z transform in factored form, plot its poles & zeros & then determine its ROC's.
G(Z)=[2z4+16z3+44z2+56z+32]/[3z4+3z3-15z2+18z-12]

**THEORY:** Write theory for pole zero analysis.

**PROGRAM:**

```
num = input('type in numerator coefficient=');       % Enter the value of num.
den = input('type in denominator coefficient=');     % Enter the value of den.
[z,p,k]=tf2zp(num,den);                              % Convert T.F. into zero,
                                                     pole analysis.
m=abs(p);
disp('zeros are at');disp(z);                        % Display zeros.
disp('poles are at');disp(p);                        % Display poles.
disp('gain constant');disp(k);                       % Display gain.
disp('radius of poles');disp(m);                     % Display radius.
sos=zp2sos(z,p,k);                                   % Convert zero pole to
                                                     second order section.
disp('second order section');disp(real(sos));        % Display SOS.
zplane(num,den);                                     % Plot zero pole in Z-
                                                     Plane
```

**INPUT:**

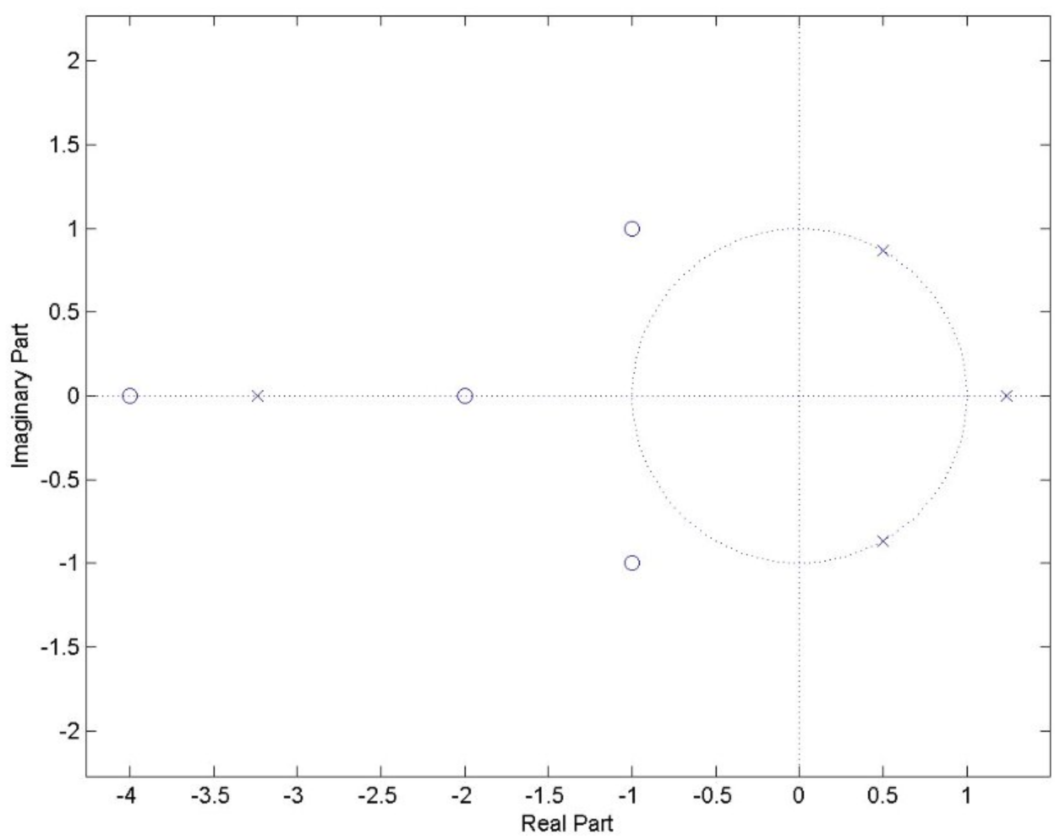
type in numerator coefficient =[2 16 44 56 32]
type in denominator coefficient =[3 3 -15 18 -12]

**OUTPUT:**

zeros are at
-4.0000
-2.0000
-1.0000+1.0000

-1.0000-1.0000
Poles are at
-3.2361
1.2361
0.5000+0.8660i
0.5000-0.8880i

Gain constant
0.6667



# EXPERIMENT NO.3

**AIM:** To verify the partial fraction expansion of the Z-transform.
G(Z)=18Z3/(18Z3+3Z2-4Z-1)

**THEORY:** Write theory for partial fraction expansion of the z-transform.

**PROGRAM:**

```
num = input('type in numerator coefficient=');    % Enter numerator
                                                    coefficient
den = input('type in denominator coefficient='); % Enter denominator
                                                    coefficient.
[r,p,k]=residuez(num,den);                          % Convert numerator
                                          & denominator residues. pole.
disp('residues are');                               % Display residue and poles.

disp(r);
disp('poles are at');
disp(p);
```

**INPUT:**
type in numerator coefficient =[18]
type in denominator coefficient =[18 3 -4 -1]

**OUTPUT:**

Residues
0.3600
0.2400
0.4000

Poles
0.5000
-0.3333
-0.3333

# EXPERIMENT NO.4

**AIM:** Spectral analysis using DFT.

**THEORY:** Write theory for spectral analysis using DFT.

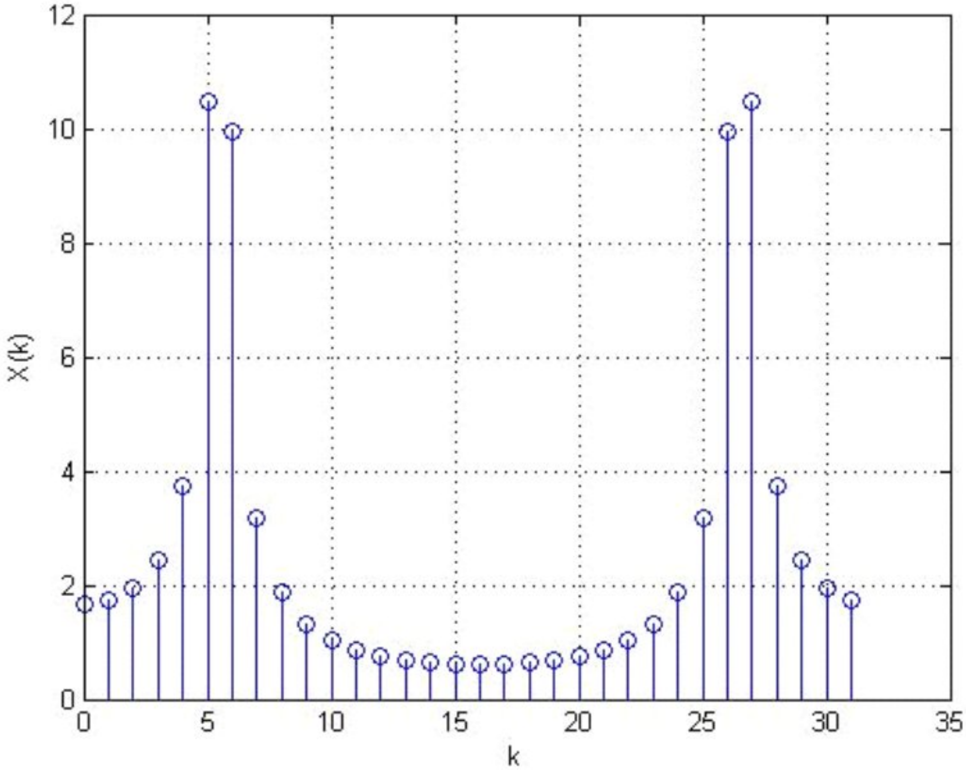
**PROGRAM:**

```
N=input('type length of DFT= ');          % Enter length of DFT.
T=input('type sampling period= ');        % Enter value of T.
freq=input('type the sinusoidal freq= '); % Enter value of sinusoidal frequency
k=0:N-1;                                  % Specify range of K=0 to N-1;
f=sin(2*pi*freq*1/T*k);
F=fft(f);                                 % Convert into s/o function of sine wave
stem(k,abs(F));                           % Plot DTS of DFT
grid on;
xlabel('k');
ylabel('X(k)');
```

**INPUT:**

type length of DFT=32
type sampling period=64
type the sinusoidal freq=11

OUTPUT:



# EXPERIMENT NO-5

**AIM:** Computation of linear convolution using DFT.

**THEORY:** Write theory for linear convolution.

**PROGRAM:**

```
g=input('Type in first sequence');            % Enter the first sequence.
h=input('Type in second sequence');           % Enter the second sequence.
ga=[g zeros(1,length(h)-1)];
ha=[h zeros(1,length(g)-1)];
G=fft(ga);                                    % Convert into DFT of ga.
H=fft(ha);                                    % Convert into DFT of ha.
Y=G.*H;                                       % Multiply G and H.
y=ifft(Y);                                    % Convert into inverse FFT.
disp('New sequence');
disp(real(y));
```

**Inputs:**

Type in first sequence [1  2  0  1]

Type in second sequence [2  2  1  1]

**Output:**

New sequence

2.0000    6.0000    5.0000    5.0000    4.0000    1.0000    1.0000

## EXPERIMENT NO-6

**AIM:** Design butterworth & chebyshev filters using IIR filters.

**THEORY:**

An IIR filter is represented by the equation:
Y(n)=-∑ak y(n-k)+∑bk x(n-k).
For N=7,F1=500 Hz,∂1=-3db,F2=1000Hz,∂2=40db.
Design h(n)by butterworth and chebyshev method and plot in frequency domain the response of the filter.
Write theory for butterworth and chebyshev filters.

**PROGRAM:**

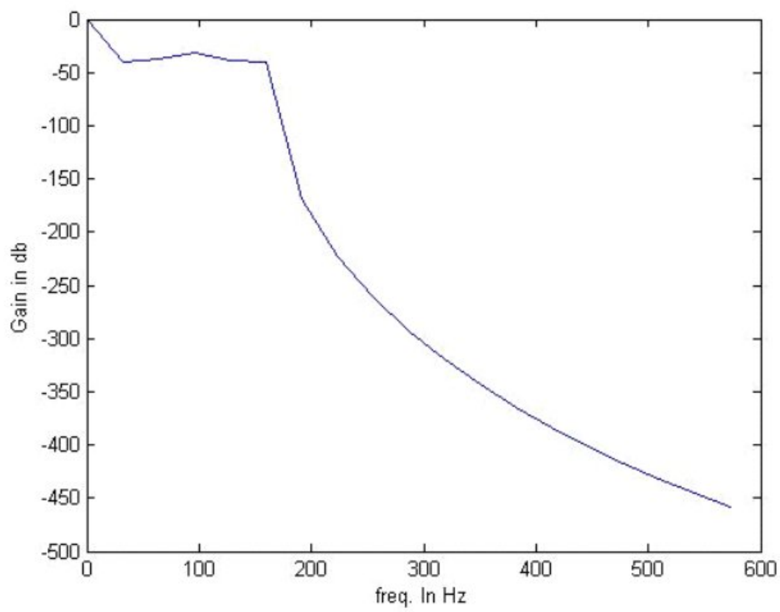
```
%For chebyshev lowpass filter

N=input('type filter order');                          % Enter Type of filter.
fp=input('passband edge frequency in Hz= ');    % Enter pass band frequency.
rp=input('passband ripple in db= ');              % Enter ripple in dB.

%Determine coefficients of the transfer function

[num,den]=cheby1(N,rp,fp,'s');             % Determine Coefficient of T.F.


%Compute and plot the frequency response

omega=[0:200:1200*pi];                     %  Specify the range of omega.
h=freqs(num,den,omega);                    %  Specify frequency number, den
                                           coe.
plot(omega/(2*pi),20*log10(abs(h)));       % Plot continuous waveform of
                                           C.L.P.F.
xlabel('freq. In Hz');                     % Give label to x-axis.
ylabel('Gain in db');                      % Give label to y-axis.
```

**PROGRAM:**

%For butterworth lowpass filter

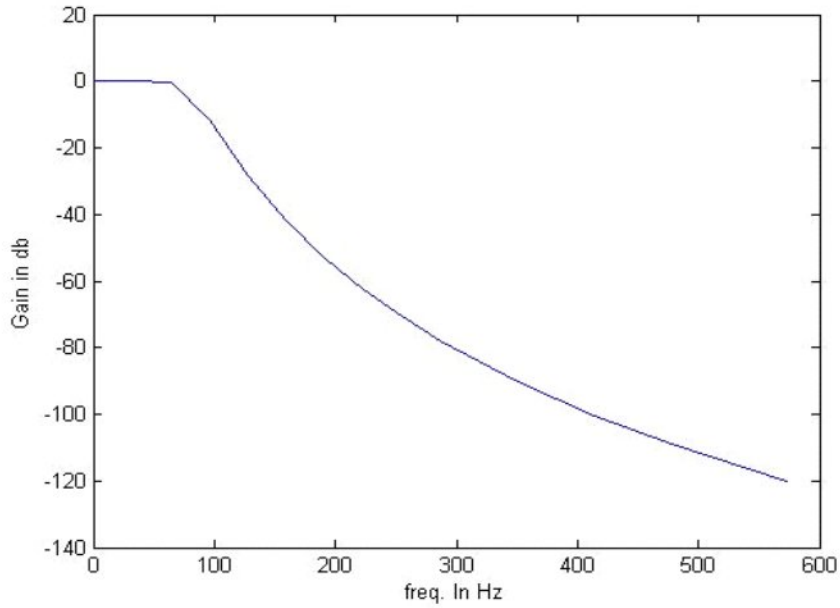
```
N=input('type filter order');
wn=input('3 db cutoff frequency');
[num,den]=butter(N,wn,'s');
omega=[0:200:1200*pi];
h=freqs(num,den,omega);
plot(omega/(2*pi),20*log10(abs(h)));
xlabel('freq. In Hz');
ylabel('Gain in db');
```

INPUT:
Type filter order=25
Passband edge frequency in Hz=1000
Passband ripple in db=40




For butterworth filter
Type filter order=7

3 db cut off frequency=500



**EXPERIMENT NO-7**

**AIM:** FIR filters design using linear phase & windows.

**THEORY:** Write theory for design of FIR filters

**PROGRAM:**

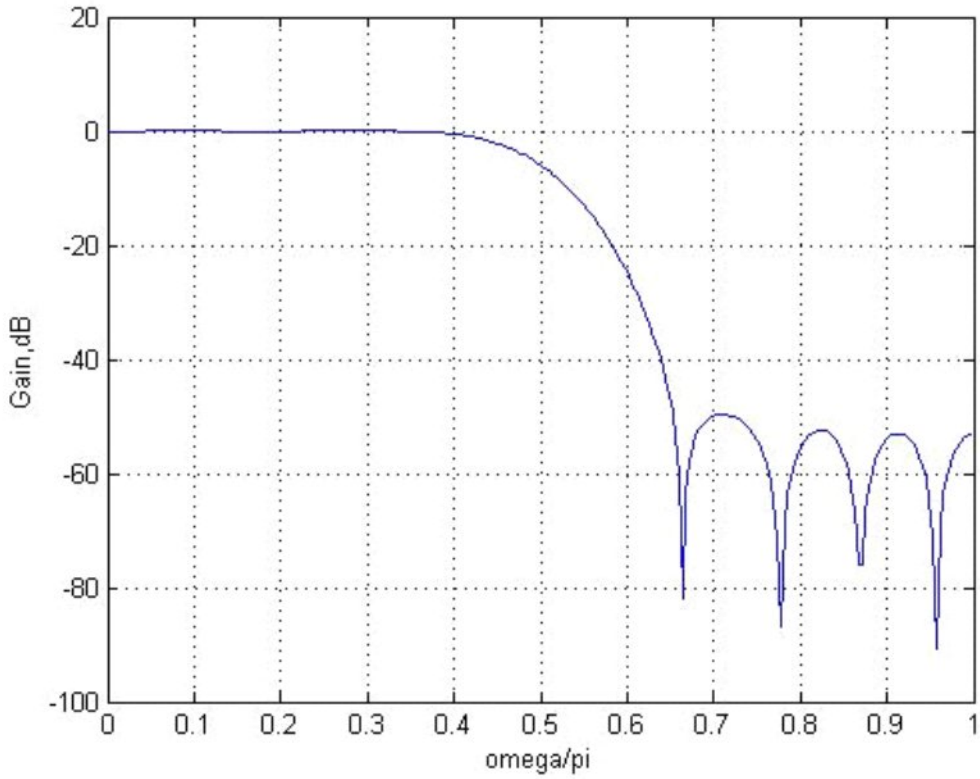
```
N=input('Type in the filter length=');        % Enter the filter length.
L=input('Type in the  value of L=');          % Enter value of L.
K=(N-1)/2;                                    % Find K.
n=-K:K;                                       % Specify range of K.

% Generate the truncated impulse response of the ideal low pass filter.

b=sinc(n/L)/L;
%Generate the window sequence
win=hamming(N);
%Generate the coefficients of the windowed filter
c=b.*win';

% Plot the gain response of the windowed filter

[h,w]=freqz(c,1,256);           % Specify range of frequency convert.
g=20*log10(abs(h));             % Convert h into db.
plot(w/pi,g);                   % Plot continuous waveform.
grid on;
xlabel('omega/pi');
ylabel('Gain,dB');
```

**Inputs:**

Type in the filter length=23

Type in the  value of L=2

**Output:**



## EXPERIMENT NO.8

**AIM:-** Illustration of decimation process and interpolation process.

**THEORY:** Write theory for interpolation & decimation process.

**PROGRAM:**

```
%program for decimation process

N=input('length of input signal=');        % Enter the length of input signal.
M=input('down sampling factor=');          % Enter down sampling factor.
f1=input('freq of 1st sinusoid=');         % Enter frequency of 1st sinusoid.
f2=input('freq of 2nd sinusoid=');         % Enter frequency of 2nd sinusoid.
n=0:N-1;                                   % Specify the range of n=0 to N-1.
x=sin(2*pi*f1*n)+sin(2*pi*f2*n);           % Add both sinusoidal waveform.
y=decimate(x,M,'fir');                     % Use decimation.
Subplot(2,1,1);                            % Plot 1st figure in first window.
stem(n,x(1:N));                            % Plot D.T.S
title('time index n');
xlabel('input sequence');
ylabel('amplitude');
subplot(2,1,2);                            % Plot 2nd figure in second
window.
m=0:N/M-1;                                 % Specify range of m from 0 to
                                                      N/(M-1).
stem(m,y(1:N/M));                          % Plot DTS.
title('output sequence');
xlabel('time index n');
ylabel('magnitude');
```

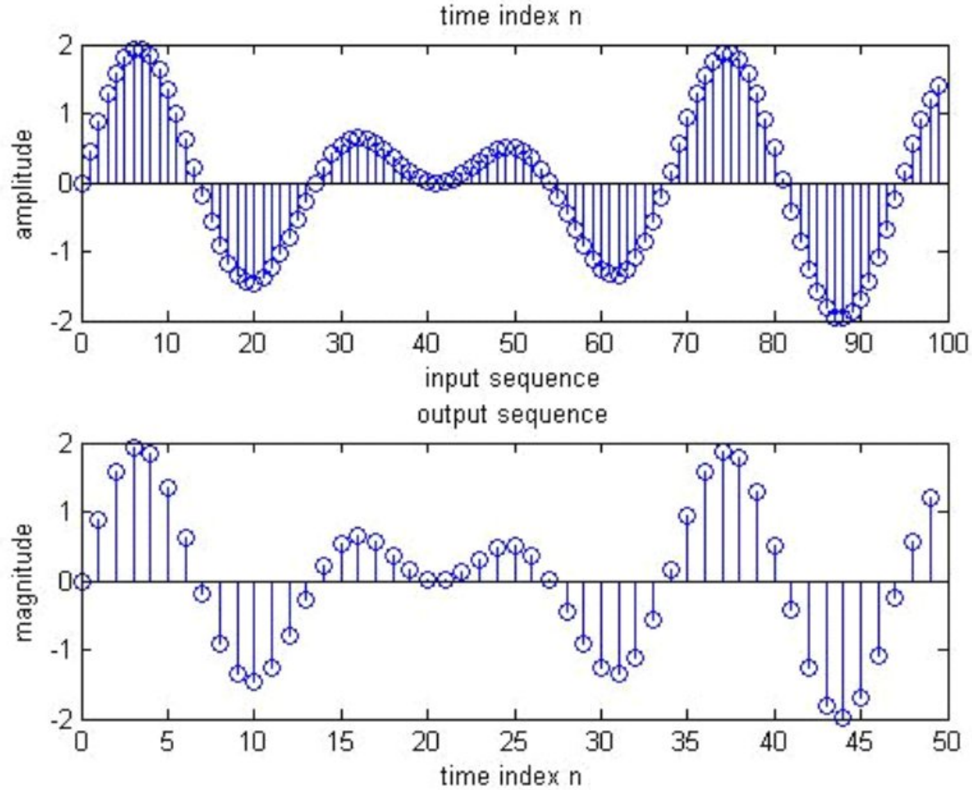
**INPUT:-** Length of input signal=100
Down-sampling factor=2
Freq.of first sinusoidal=0.043
Freq, of second sinusoidal=0.031

**PROGRAM:**

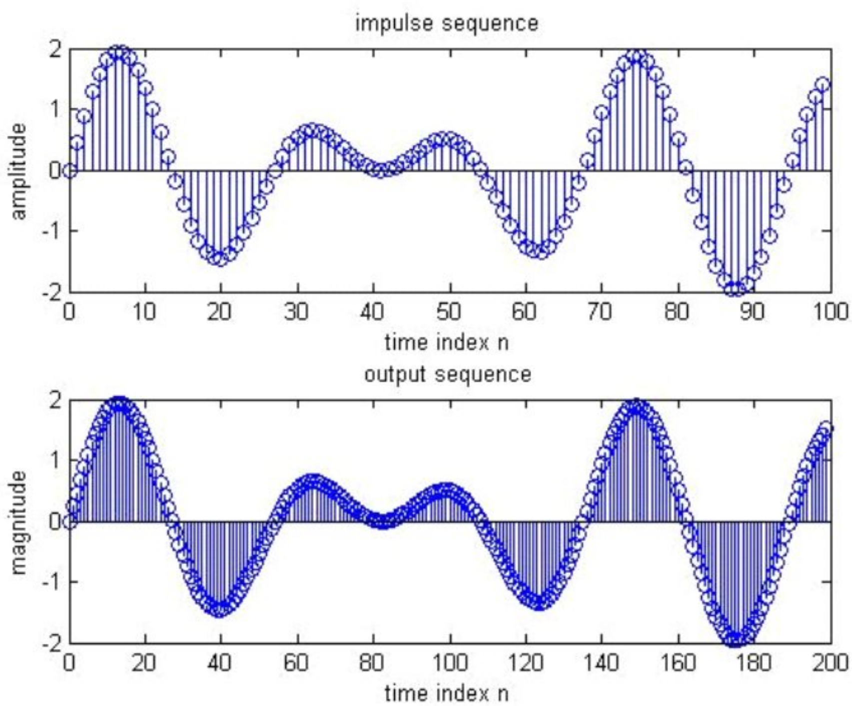
```
%program for interpolation process

N=input('length of input signal=');
L=input('up sampling factor=');
f1=input('freq of 1st sinusoid=');
f2=input('freq of 2nd sinusoid=');
n=0:N-1;
x=sin(2*pi*f1*n)+sin(2*pi*f2*n);
y=interp(x,L);
subplot(2,1,1);
stem(n,x(1:N));
title('impulse sequence');
xlabel('time index n');
ylabel('amplitude');
subplot(2,1,2);
m=0:N*L-1;
stem(m,y(1:N*L));
title('output sequence');
xlabel('time index n');
ylabel('magnitude');
```

**INPUT:-** Length of input signal=100
UP-sampling factor=2
Freq.of first sinusoidal=0.043
Freq, of second sinusoidal=0.031

**OUTPUT FOR A) DECIMATION**




OUTPUT FOR B) INTERPOLATION



# EXPERIMENT NO-9

**AIM:** Computation of the output noise variance based on partial fraction approach.

**THEORY:** Write theory for output noise variance.

**PROGRAM:**

```
num=input('Type in the numerator=');
den=input('Type in the denominator=');
[r,p,k]=residue(num,den);
R=size(r,1);
R2=size(K,1);
if R2>1
disp('Cannot continue...');
return;
end
if R2==1
nvar=K^2;
else
nvar=0;
end
% Compute output noise varience
for k=1:R,
for m=1:R,
integral=r(k)*conj(r(m))/(1-p(k)*conj(p(m)));
nvar=nvar+integral;
end
end
disp('Output Noise Varience');
disp(real(nvar));
```

**INPUTS:**

Type in the numerator=

[0.06891875 0.13808186 0.18636107 0.13808186 0.06891875]

Type in the denominator=

[1 -1.30613249 1.48301305 -0.77709026 0.2361457]

**Output:**

Output Noise Variance= 0.4026

## EXPERIMENT NO-10

AIM: Program for Dual-tone multi frequency tone detection.

THEORY: Write theory for Dual-tone multi frequency Signal detection.

PROGRAM:

```
%Dual-tone multi frequency tone detection using DFT.
d=input('Type in the telephone digit=','s');
symbol=abs(d);
tm=[49 50 51 65;52 53 54 66;55 56 57 67;42 48 35 68];
for p=1:4;
for q=1:4;
if tm(p,q)==abs(d);break,end
end
if tm(p,q)==abs(d);break,end
end
f1=[697 770 852 941];
f2=[1209 1336 1477 1633];
n=0:204;
x=sin(2*pi*n*f1(p)/8000)+sin(2*pi*n*f2(q)/8000);
k=[18 20 22 24 31 34 38 42];
val=zeros(1,8);
for m=1:8;
```

```
Fx(m)=ifft(x,205,k(m));

end

val=abs(Fx);

stem(k,val);

grid;

xlabel('k');

ylabel('X[k]');

limit=80;

for s=5:8;

if val(s)>limit,break,end

end

for r=1:4;

if val(r)>limit,break,end

end

disp(['Touch-tone symbol=',setstr(tm(r,s-4))]);
```

## *4. Quiz on the subject:*

1).--------Signals are the signals repeating after specific period.

a] Energy b)Digital c] Periodic

2) The system is said to be ------- or to have memory.

a] Dynamic b] Recursive c] Stable

3) If $y(n) = nx(n)$ then the system is

a] Causual b] Recursive c] dynamic

4) If the signal is infinite duration and both sided then its ROC is

a] an annular ring b] entire z plane except $z=0$

c] entire z plane except z=

5) If the sequence is real and even then the DFT consists of

a] real and even parts b] purely imaginary parts

6)Appending zeros to a sequence in order to increase its length is called

a)zero padding b)convolution c)oversampling

7)In 8 point DFT by Radix -2 FFT there are ______ stages with ______ butterflies Diagram.

a) three,three b) four, three c) two ,three

8)The frequency response of a digital filter is periodic in the range

a) $0<w< 2\pi$ b) $-\pi<w<\pi$ c) $0<w<\pi$

9)In FIR filters the Gibbs oscillations are due to

a) non linear magnitude characteristic b)non linear phase characteristic

c) sharp transition from pass band to stop band

10) If wc is cutoff frequency of LPF then the response lies only in the range of

a) $-wc<w<\pi$ b) $-wc<w<wc$ c) $-\pi<wc<-wc$

11)The width of main lobe in Rectangular window spectrum is a)$4\pi/N$ b)$16\pi/N$ c)$8\ \pi/N$

12) Symmetric impulse response having odd number of samples N=7 with center of symmetry $\alpha$ is equal to

a)2 b)5 c)3

13) IIR filters are designed by considering all the

a) Infinite sample of frequency response b)finite sample of impulse response

c) Infinite sample of impulse response

14)In Impulse invariant transformation the digital frequency 'w' for a given analog frequency ,$\Omega$ is given by

a)$w = \Omega T$ b)$w = \Omega /T$ c)$w = T/\Omega$

15) In bilinear transformation the 's' is replaced by

a)$s= 2\ 1+z^{-1}/T\ \ 1+ z^{-1}$ b)$s-2 /T$ c) $s= wT$

*5. Conduction of Viva-Voce Examinations:*

Teacher should conduct oral exams of the students with full preparation. Normally, the objective questions with guess are to be avoided. To make it meaningful, the questions should be such that depth of the students in the subject is tested. Oral examinations are to be conducted in co-cordial environment amongst the teachers taking the examination. Teachers taking such examinations should not have ill thoughts about each other and courtesies should be offered to each other in case of difference of opinion, which should be critically suppressed in front of the students.

*6. Evaluation and marking system:*

Basic honesty in the evaluation and marking system is absolutely essential and in the process, impartial nature of the evaluator is required in the examination system. It is a wrong approach or concept to award the students by way of easy marking to get cheap popularity among the students, which they do not deserve. It is a primary responsibility of the teacher that right students who are really putting up lot of hard work with right kind of intelligence are correctly awarded.

The marking patterns should be justifiable to the students without any ambiguity and teacher should see that students are faced with just circumstances.