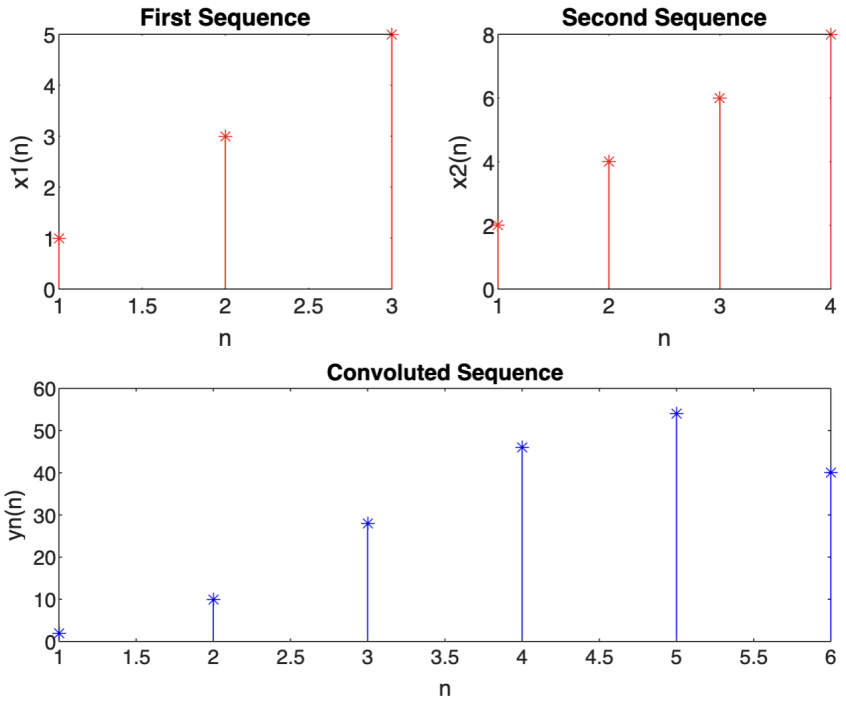
Digital Signal Processing Lab Manual (MATLAB)

B.Tech III Sem Electronics & Communication Engineering

1. Linear Convolution

```
% Input the sequences
x1n = input('Enter the first sequence: ');
x2n = input('Enter the second sequence: ');

% Perform convolution
yn = conv(x1n, x2n);

% Display the sequences and the convolution result
fprintf('First sequence is: %s', num2str(x1n));
fprintf('\nSecond sequence is: %s\n', num2str(x2n));
disp('Convoluted output is: ');
disp(yn);

% Plot the sequences and the convolution result
figure;
subplot(2,2,1);
stem(x1n, 'r*');
xlabel('n');
ylabel('x1(n)');
title('First Sequence');

subplot(2,2,2);
stem(x2n, 'r*');
xlabel('n');
ylabel('x2(n)');
title('Second Sequence');

subplot(2,1,2);
stem(yn, 'b*');
xlabel('n');
ylabel('yn(n)');
title('Convoluted Sequence');
```

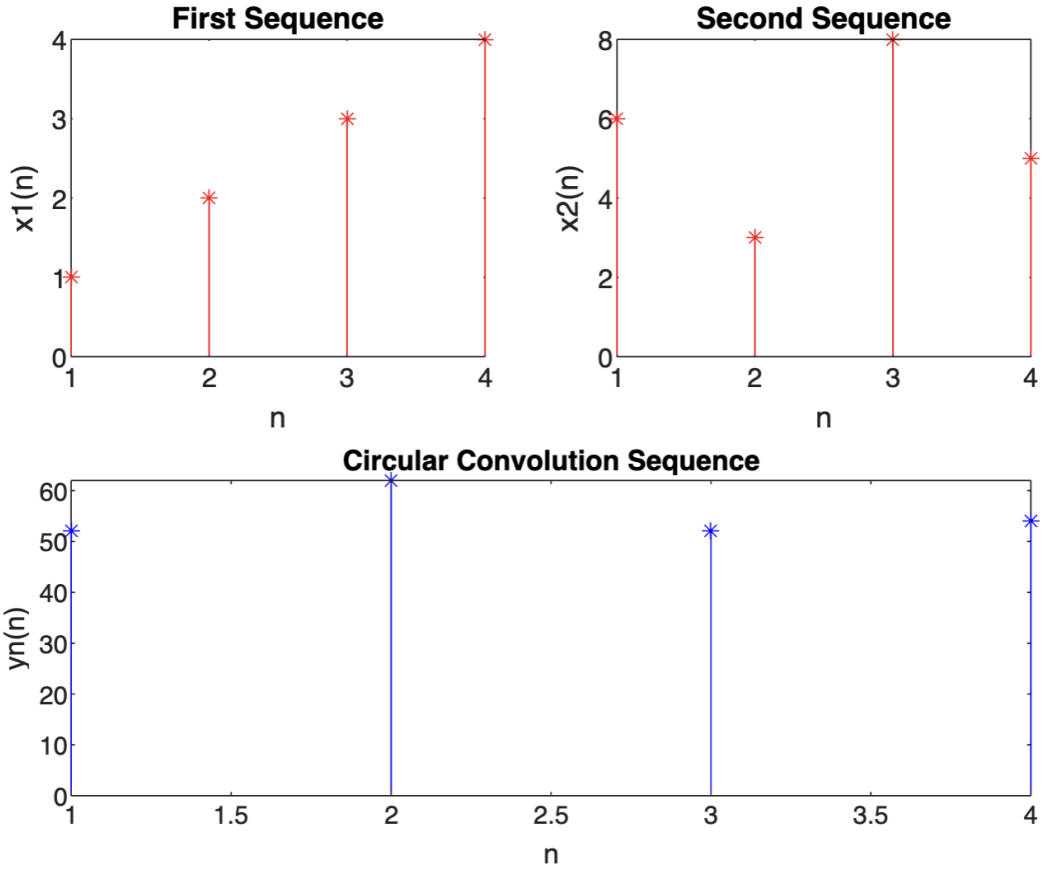
```
>> convol
Enter the first sequence (e.g., [1 2 3]):
[1 3 5]
Enter the second sequence (e.g., [4 5 6]):
[2 4 6 8]
First sequence is: 1  3  5
Second sequence is: 2  4  6  8
Convoluted output is:
     2    10    28    46    54    40
```



2. Circular Convolution

```
xn=input('Enter the first sequence x(n): ');
hn=input('Enter the second sequence h(n): ');
l1=length(xn);
l2=length(hn);
N=max(l1,l2);
xn=[xn, zeros(1, N-l1)];
hn=[hn, zeros(1, N-l2)];
for n=0:N-1
    y(n+1)=0;
   for k=0:N-1
       i=mod((n-k), N);
       i=i+1;
       y(n+1)=y(n+1)+hn(k+1)*xn(i);
   end
end
disp('circular convolution of the given sequence is: ');
disp(y);
disp('x(n): ');
disp(xn);
disp('h(n): ');
disp(hn);
subplot(2,2,1);
stem(xn);
xlabel('n');
ylabel('x(n)');
title('plot of first seq x(n)')
subplot(2,2,2);
stem(hn);
xlabel('n');
ylabel('h(n)');
title('plot of second seq h(n)')
subplot(2,1,2);
stem(y, 'r*');
xlabel('n');
ylabel('y(n)');
title('plot of circular convolution y(n)');
```

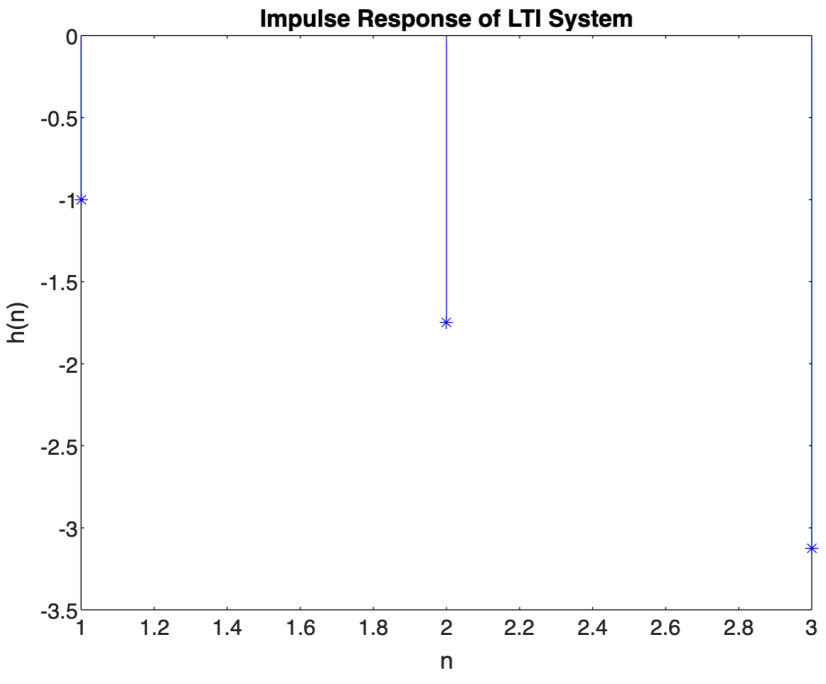
```
Enter the first sequence (e.g., [1 2 3]):
[1 2 3 4]
Enter the second sequence (e.g., [4 5 6]):
[6 3 8 5]
First sequence is: 1  2  3  4
Second sequence is: 6  3  8  5
Circularly Convoluted output is:
    52    62    52    54
```



3. Impulse Response

```
y = input('input sequence y(n) of the system: ');
x = input('input sequence x(n)of the system:');
N = input ('enter the length of the impulse response sequence:');
h = impz(x, y, N);
disp ('Impulse response of system h(n)');
disp (h);
% Plotting
n = 0:1:N-1;
stem (n,h, 'b*');
xlabel ('n');
ylabel ('h(h)');
title ('Impulse response');
```

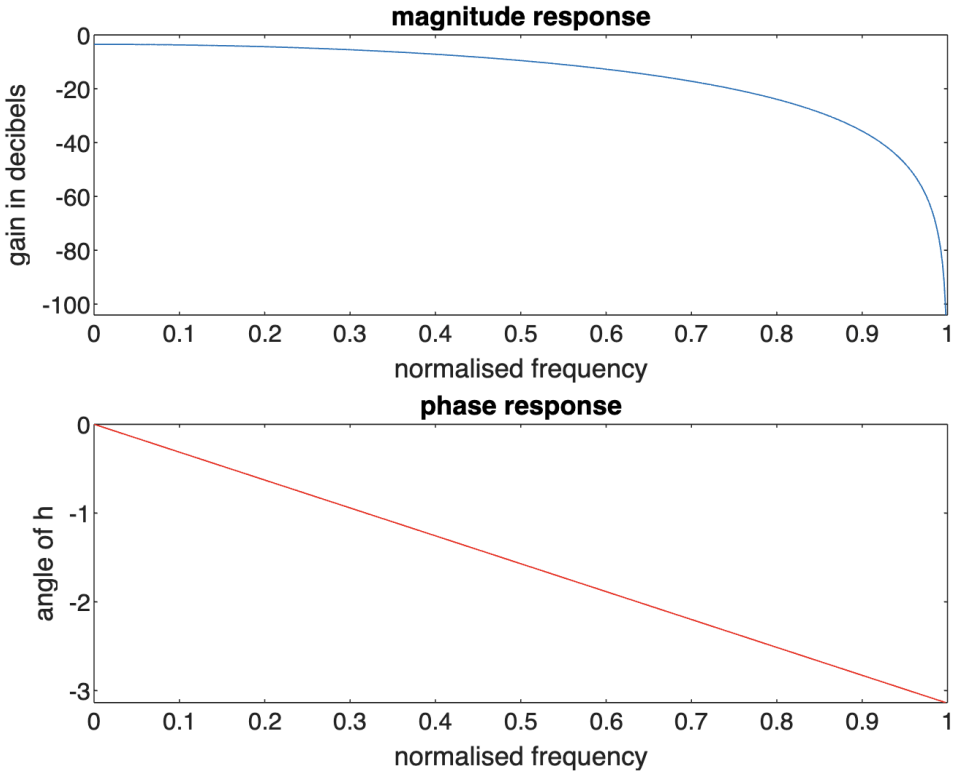
```
>> impez
Enter input sequence:
[-1 2]
Enter output sequence:
[1 -1/4 -3/8]
Length of impulse response:
3
Input sequence: -1  2
Input sequence: 1       -0.25      -0.375
Impulse response
   -1.0000
   -1.7500
   -3.1250
```



4. Frequency Response:

```
clc;
close all;
clear all;
num = input('enter the numerator coefficients: ');
den = input('enter the denominator coefficients: ');
[h, om] = freqz(num, den);
disp(h);

subplot(2,1,1);
plot(om/pi, 20*log10(abs(h)));
disp(abs(h));
xlabel('normalised frequency');
ylabel('gain in decibels');
title('magnitude response');

subplot(2,1,2);
plot(om/pi, angle(h));
disp(angle(h));
xlabel('normalised frequency');
ylabel('angle of h');
title('phase response');
```

```
enter the numerator coefficients:
[1/6 1/3 1/6]
enter the denominator coefficients:
[1]
   0.6667 + 0.0000i
   0.6666 - 0.0041i
   0.6666 - 0.0082i
   0.6665 - 0.0123i
   0.6664 - 0.0164i
   0.6662 - 0.0204i
   0.6660 - 0.0245i
   0.6657 - 0.0286i
```

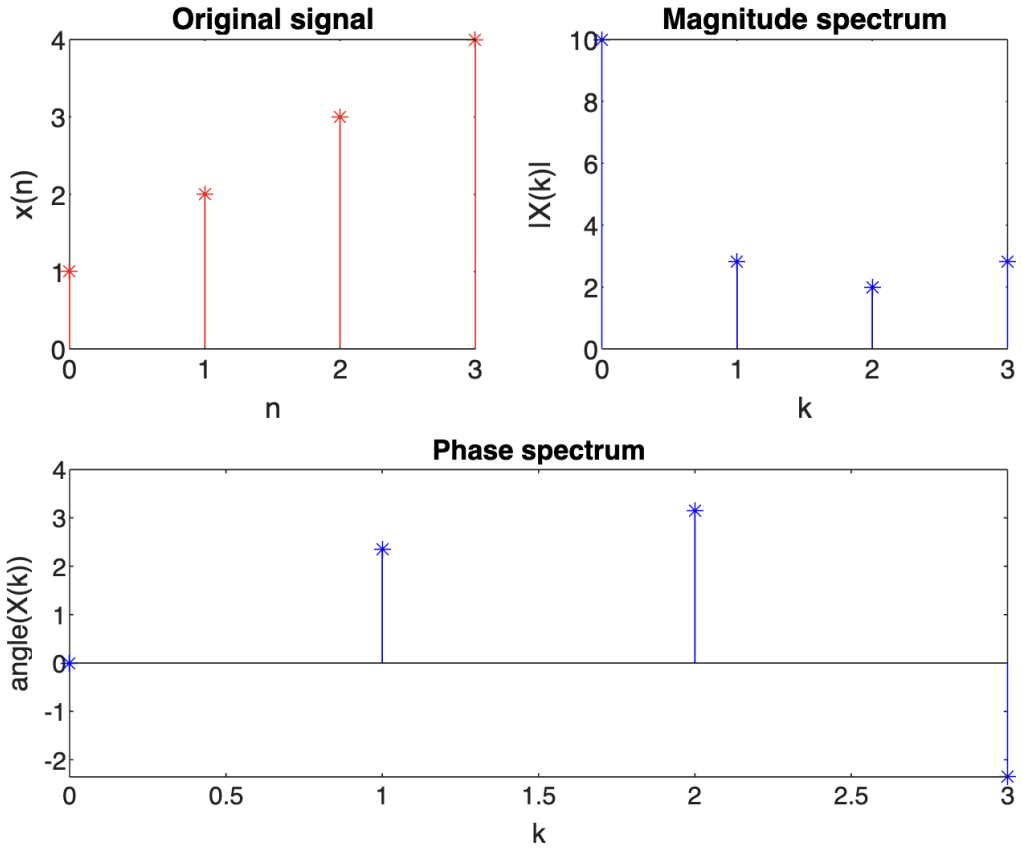

magnitude response
gain in decibels
0
-20
-40
-60
-80
-100
0
0.1
0.2
0.3
0.4
0.5
0.6
0.7
0.8
0.9
1
normalised frequency

phase response
angle of h
0
-1
-2
-3
0
0.1
0.2
0.3
0.4
0.5
0.6
0.7
0.8
0.9
1
normalised frequency

5. N-Point DFT

```
Clc;
Close all;
Clear all;
N = input('enter the N point: ');
Xn = input(enter the input sequence x(n):');
Xk = fft(Xn, N);
disp('N point DFT of x(n) is = ');
disp(Xk);

figure(1);
n = 0:1:length(Xn)-1;
stem(n, Xn);
xlabel('n');
ylabel('x(n)');
title('original signal');

figure(2);
k = 0:N-1;
stem(k, abs(Xk));
xlabel('k');
ylabel('|x(k)|');
title('Magnitude Spectrum');

figure(3);
stem(k, angle(Xk));
xlabel('k');
ylabel('<x(k)');
title('Phase Spectrum');
```

```
Enter number of samples N:
4
Enter the input sequence x(n):
[1 2 3 4]
N-point DFT is:
  10.0000 + 0.0000i  -2.0000 + 2.0000i  -2.0000 + 0.0000i  -2.0000 - 2.0000i
```


Original signal
x(n)
n
Magnitude spectrum
|X(k)|
k
Phase spectrum
angle(X(k))
k

6. Linear Convolution using DFT & IDFT

```
clc;
close all;
clear all;
xn = input('enter the sequence x(n): ');
hn = input('enter the sequence h(n): ');
N = length(xn) + length(hn)-1;
xk = fft(xn, N);
hk = fft(hn, N);
yk = xk.hk; % .* multiplies every element of xk to hk
yn = ifft(yk, N);
disp('Linear Convolution of x(n) & h(n): ');
disp(yn);

subplot(2,2,1);
stem(xn, 'b*');
xlabel('n'); ylabel('x(n)');
title('plot of x(n)');

subplot(2,2,2);
stem(hn, 'b*');
xlabel('n'); ylabel('h(n)');
title('plot of h(n)');

subplot(2,2,3);
stem(yn, 'r*');
xlabel('n'); ylabel('y(n)');
title('DFT IDFT convolution output');

yv = conv(xn, hn);
disp(yv);
subplot(2,2,4);
stem(xn, 'r*');
xlabel('n'); ylabel('yv(n)');
title('verified convolution');
```

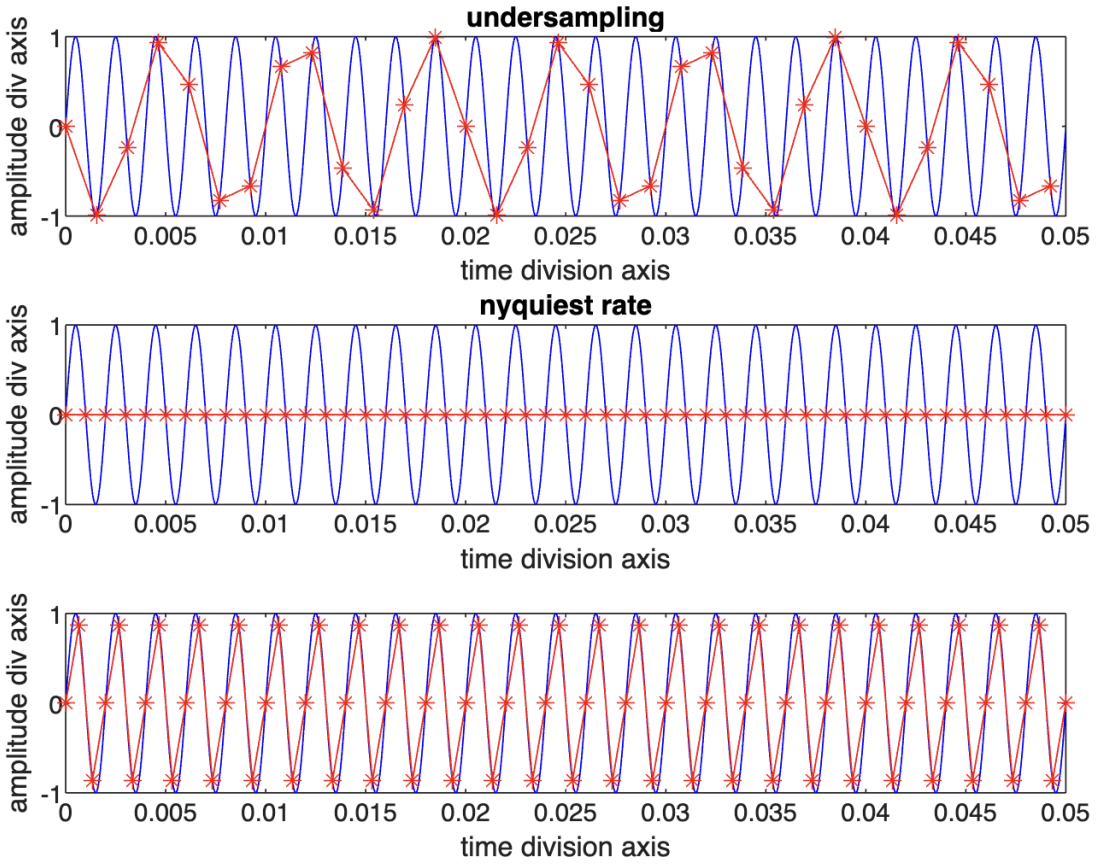
7. Circular Convolution using DFT IDFT

```
clc;
close all;
clear all;
xn = input('enter the sequence x(n): ');
hn = input('enter the sequence h(n): ');
N = max(length(xn), length(hn));
xk = fft(xn, N);
hk = fft(hn, N);
yk = xk.hk;
yn = ifft(yk, N);
disp('Circular Convolution of x(n) & h(n): ');
disp(yn);

subplot(2,2,1);
stem(xn, 'b*');
xlabel('n'); ylabel('x(n)');
title('plot of x(n)');

subplot(2,2,2);
stem(hn, 'b*');
xlabel('n'); ylabel('h(n)');
title('plot of h(n)');

subplot(2,1,2);
stem(yn, 'r*');
xlabel('n'); ylabel('y(n)');
title('DFT IDFT circular convolution output');
```

8. Sampling Theorem

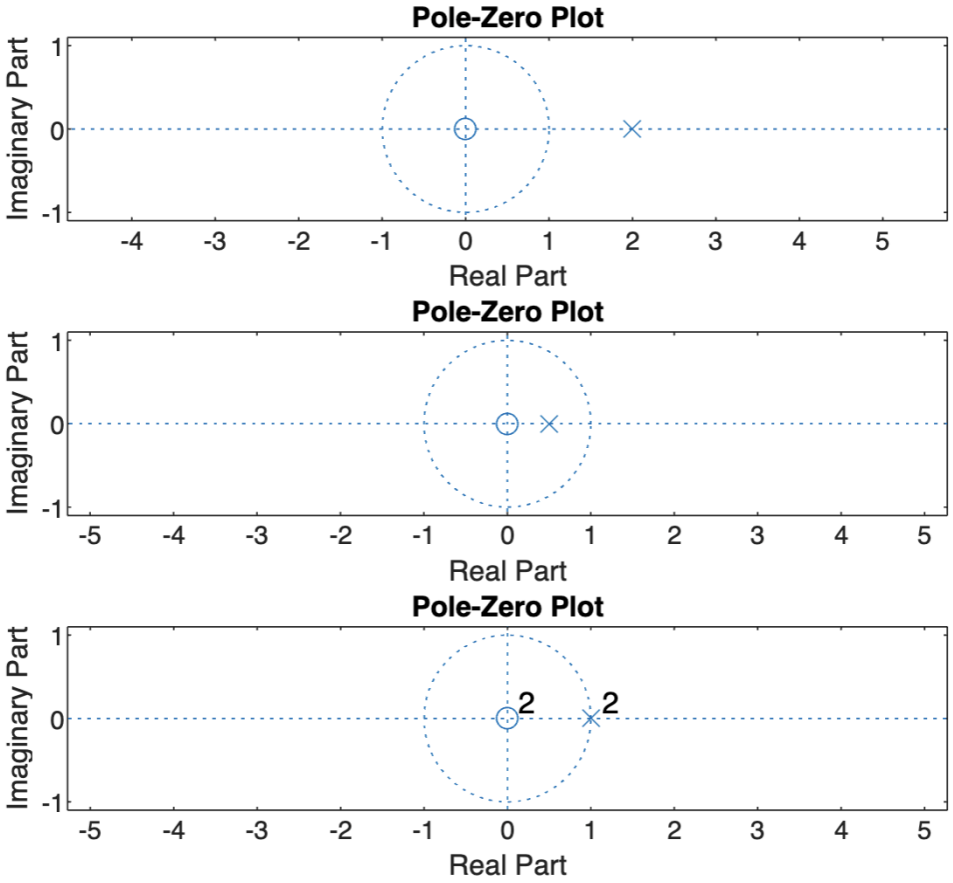
```
clc;
clear all;
close all;
final = 0.05;
t = 0:0.00005:final;
fd = input('Enter analog frequency: ');
xt = sin(2*pi*fd*t);

fs1 = 1.3 * fd;
n1 = 0:1/fs1:final;
xn = sin(2*pi*fd*n1);
subplot(3,1,1);
plot(t, xt, 'b', n1, xn, 'r*-');
xlabel('time division axis'); ylabel('amplitude div axis');
title('undersampling');

fs2 = 2.0 * fd;
n2 = 0:1/fs2:final;
xn = sin(2*pi*fd*n2);
subplot(3,1,2);
plot(t, xt, 'b', n2, xn, 'r*-');
xlabel('time division axis'); ylabel('amplitude div axis');
title('nyquiest rate');

fs3 = 3 * fd;
n3 = 0:1/fs3:final;
xn = sin(2*pi*fd*n3);
subplot(3,1,3);
plot(t, xt, 'b', n3, xn, 'r*-');
xlabel('time division axis'); ylabel('amplitude div axis');
title(oversampling);
```

```
Enter analog frequency:
500
```


undersampling
amplitude div axis
1
0
-1
0
0.005
0.01
0.015
0.02
0.025
0.03
0.035
0.04
0.045
0.05
time division axis
nyquiest rate
amplitude div axis
1
0
-1
0
0.005
0.01
0.015
0.02
0.025
0.03
0.035
0.04
0.045
0.05
time division axis
amplitude div axis
1
0
-1
0
0.005
0.01
0.015
0.02
0.025
0.03
0.035
0.04
0.045
0.05
time division axis

9. Z-Transform

```
clc;
close all;
clear all;
syms n;
a 2;
x = a^n;
X = ztrans(x);
disp('Z Transform of a^n, a > 1');
disp(X);

syms n;
a = 0.5;
x = a^n;
X1 = ztrans(x);
disp('Z Transform of a^n, 1 > a > 0');
disp(X1);

syms n;
x = n+1;
X2 = ztrans(x);
disp(Z transformation of 1+n');
disp(X2);

A = iztrans(X);
disp('Inverse Z Transform of a^n, a > 1: ');
disp(A);

B = iztrans(X1);
disp('Inverse Z Transform of a^n, 0 > a > 1: ');
disp(B);

C = iztrans(X2);
disp('Inverse Z Transform of 1+n:  ');
disp(C);

subplot(1,3,1);
zplane([1 0], [1 -2]);
subplot(1,3,2);
zplane([1 0], [1 -1/2]);
subplot(1,3,3);
zplane([1 0 0], [1 -2 1]);
```

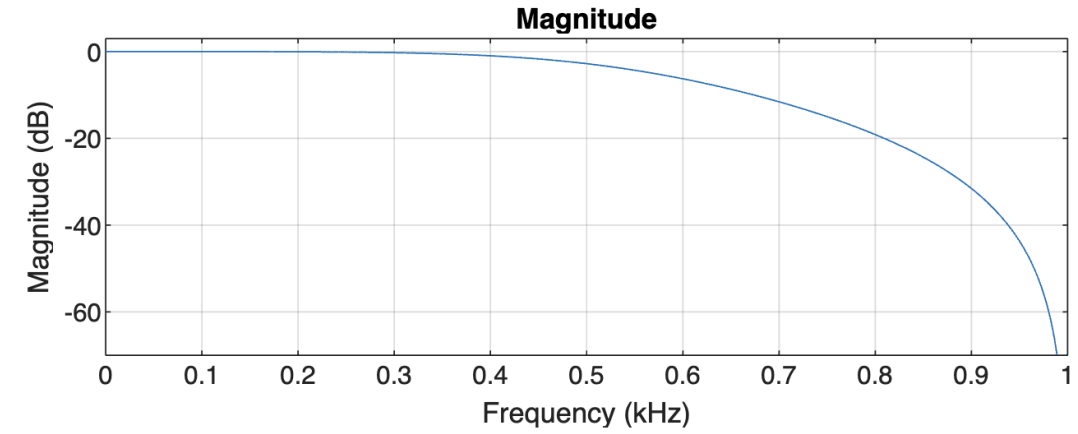

Pole-Zero Plot
Imaginary Part
Real Part
Pole-Zero Plot
Imaginary Part
Real Part
Pole-Zero Plot
2
2
Imaginary Part
Real Part

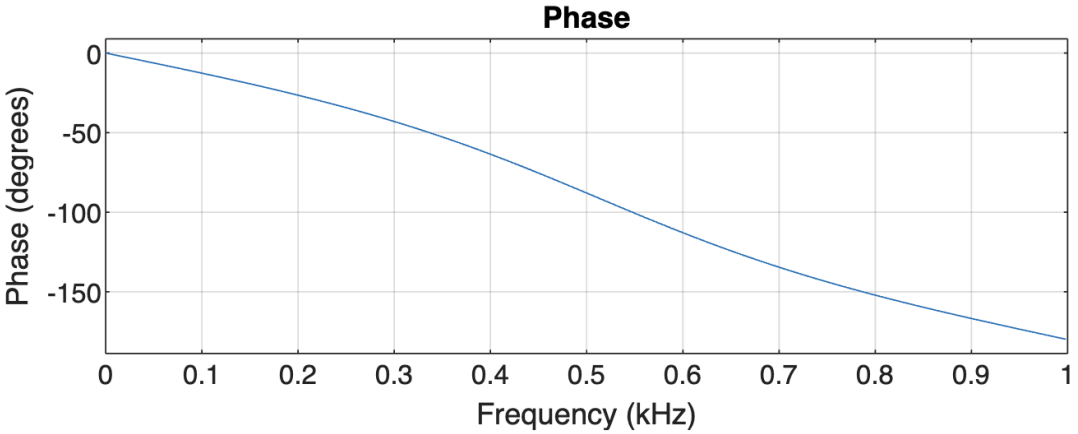
10. Design and Implementation of IIR Filter

10.1. Butterworth

```
clc;
clear all;
close all;

fp=input('enter passband frequency in Hz: ');
fs=input('enter stopband frequency in Hz: ');
ff=input('enter sampling frequency in Hz: ');
ap = input('passband ripple in dB: ');
as = input('stopband ripple in dB: ');
wp = 2*pi*(fp/ff);
ws = 2*pi*(fs/ff);
up = 2*tan(wp/2);
us = 2*tan(ws/2);
[n, wc]=buttord(up, us, ap, as, 's');

disp('order of filter is: ');
disp(n);
disp('Normalised frequency is: ');
disp(wc);

[num, den] = butter(n, wc, 's');
[b,a] = bilinear(num, den, 1);
freqz(b, a, 512, ff);
printsys(b,a,'z');
```

```
enter passband frequency in Hz:
500
enter stopband frequency in Hz:
750
enter sampling frequency in Hz:
2000
passband ripple in dB:
3.01
stopband ripple in dB:
15
order of filter is:
     2

Normalised frequency is:
    2.0526


num/den =

   0.30053 z^2 + 0.60106 z + 0.30053
   ---------------------------------
       z^2 + 0.030385 z + 0.17174
```

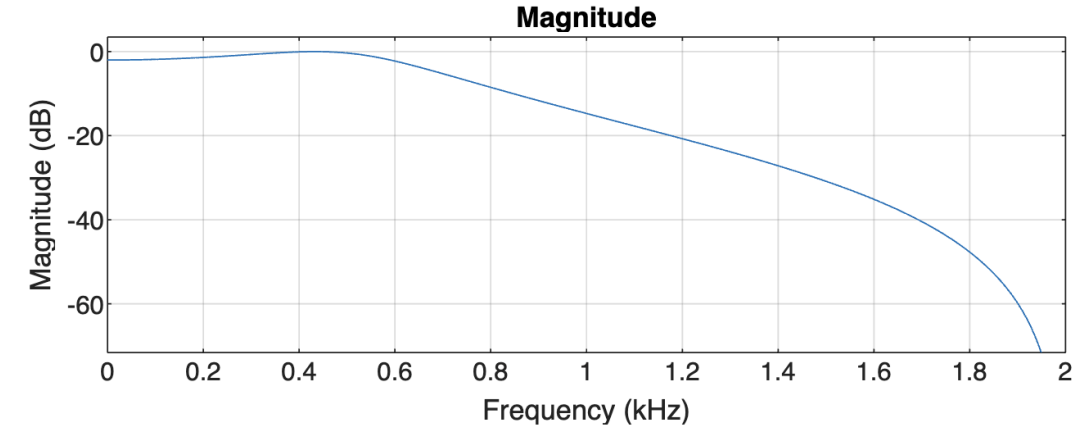


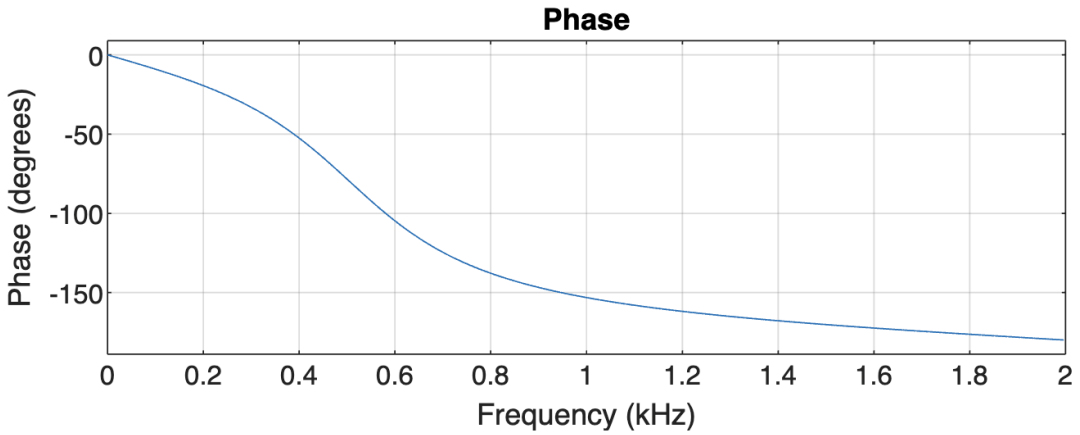


10.2. Chebyshev

```
clc;
clear all;
close all;

fp=input('enter passband frequency in Hz: ');
fs=input('enter stopband frequency in Hz: ');
ff=input('enter sampling frequency in Hz: ');
ap = input('passband ripple in dB: ');
as = input('stopband ripple in dB: ');
wp = 2*pi*(fp/ff);
ws = 2*pi*(fs/ff);
up = 2*tan(wp/2);
us = 2*tan(ws/2);

[n, wn]=cheb1ord(up, us, ap, as, 's');
disp('order of filter is: ');
disp(n);
disp('Normalised frequency is: ');
disp(wn);

[z,p,k]=cheb1ap(n,ap);       % Lowpass filter prototype
   [num,den]=zp2tf(z,p,k);     % Convert to transfer function form

 [b,a] = bilinear(num, den, 1);
freqz(b, a, 512, ff);
printsys(b,a,'z');
```

```
enter passband frequency in Hz:
100
enter stopband frequency in Hz:
500
enter sampling frequency in Hz:
4000
passband ripple in dB:
2
stopband ripple in dB:
20
order of filter is:
     2

Normalised frequency is:
    0.1574


num/den =

   0.10167 z^2 + 0.20333 z + 0.10167
   ---------------------------------
       z^2 - 0.98806 z + 0.50001
```



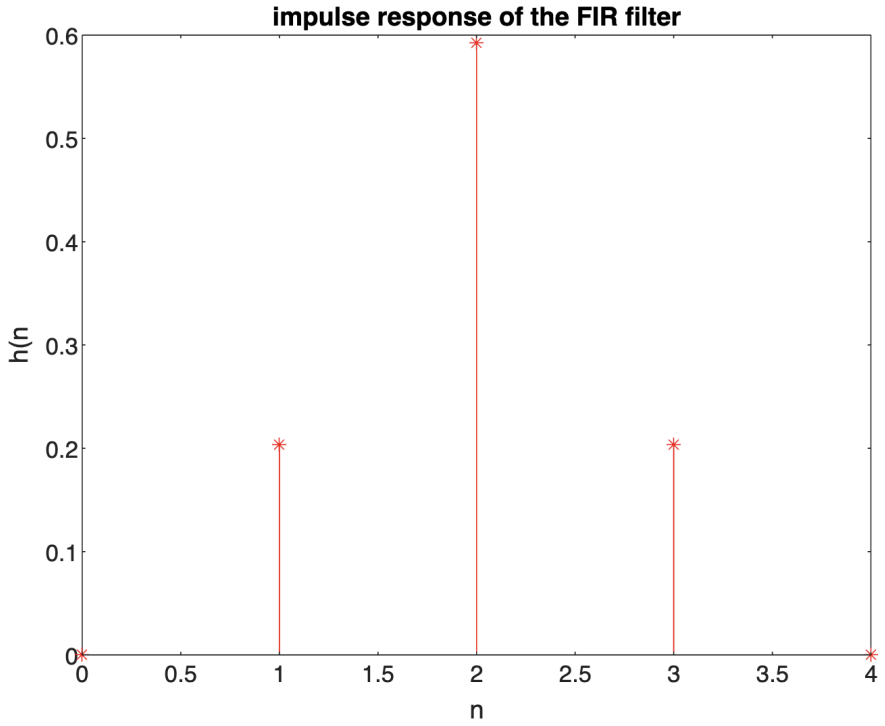


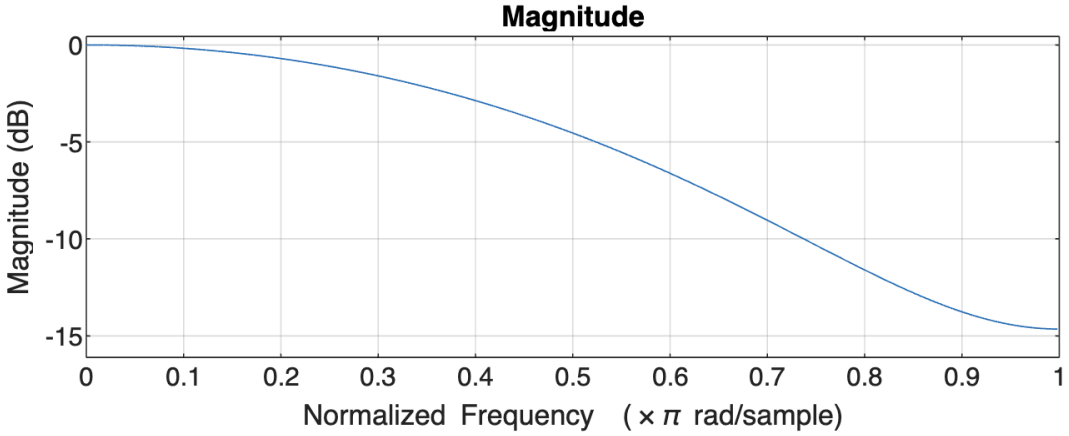
11. Design and Implementation of FIR Filter

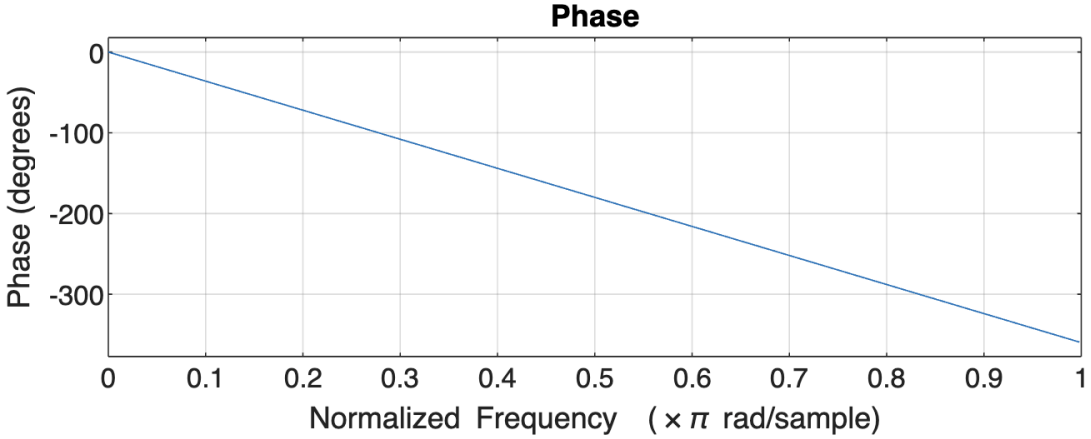
11.1. Hamming Window

```
clc;
clear all;
close all;
f = input('Enter frequency: ');
fs = input('Sampling Frequency: ');
N = input('Order of filter: ');
wc = 2*pi*(f/fs);
h= fir1(N-1, wc/pi, hamming (N));

disp('hamming window coefficients: ');
disp(hamming(N));
disp('impulse response of the FIR filter: ');
disp(h);

figure(1);
freqz(h);

figure(2);
n = 0:1:N-1;
stem(n,h, 'r*');
xlabel('n');
ylabel('h(n');
title('impulse response of the FIR filter');
```

```
Enter frequency:
1000
Sampling Frequency:
4000
Order of filter:
5
hamming window coefficients:
    0.0800
    0.5400
    1.0000
    0.5400
    0.0800

impulse response of the FIR filter:
    0.0000    0.2037    0.5926    0.2037    0.0000
```


impulse response of the FIR filter
h(n
n
0
0.1
0.2
0.3
0.4
0.5
0.6
0
0.5
1
1.5
2
2.5
3
3.5
4


Magnitude
Magnitude (dB)
Normalized Frequency (× π rad/sample)
0
-5
-10
-15
0
0.1
0.2
0.3
0.4
0.5
0.6
0.7
0.8
0.9
1


Phase
Phase (degrees)
Normalized Frequency (× π rad/sample)
0
-100
-200
-300
0
0.1
0.2
0.3
0.4
0.5
0.6
0.7
0.8
0.9
1