

## Discrete-Time Fourier Transform

### Table of Contents

<b>A.</b>	<b><i>Discrete-time Fourier transform (DTFT) .....</i></b>	<b>2</b>
<b>B.</b>	<b><i>Time convolution .....</i></b>	<b>4</b>
<b>C.</b>	<b><i>FIR filter design by frequency Sampling .....</i></b>	<b>8</b>

## A. Discrete-time Fourier transform (DTFT)

```
% A.1

n = 0:127;
u = @(n) (n >= 0) * 1.0 .* (mod(n,1)==0);
% Signal x[n]
x = @(n) (1-n/7).*(u(n)-u(n-7));
% Compute DTFT of x[n] using fft command
X = fft(x(n));
% Center the zero frequency component
X = fftshift(X);

Omega = linspace(-pi,pi,128);

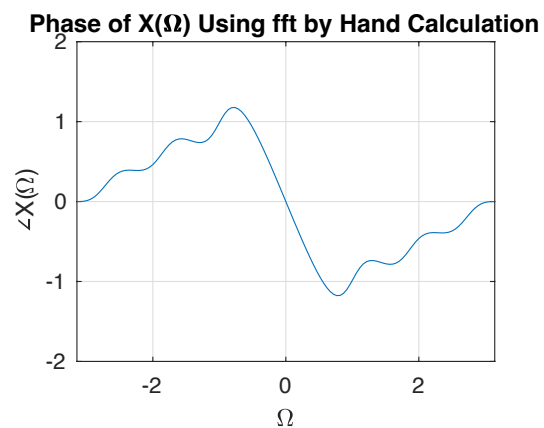
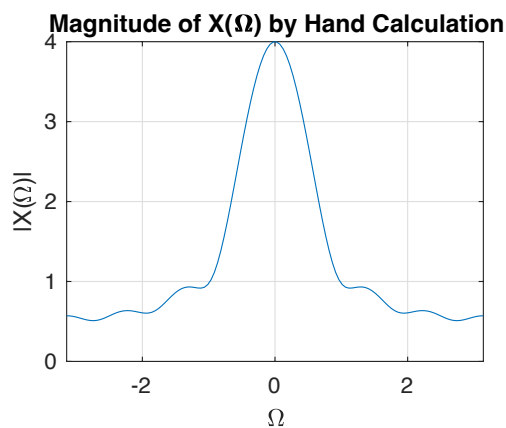
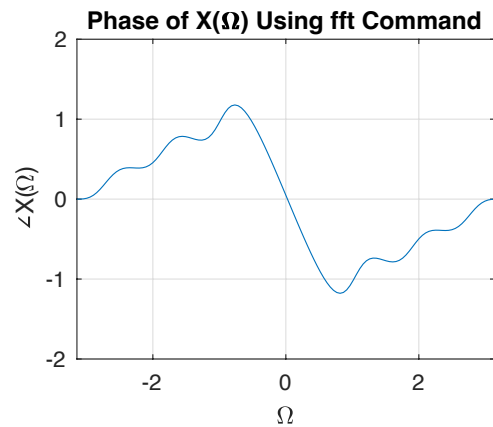
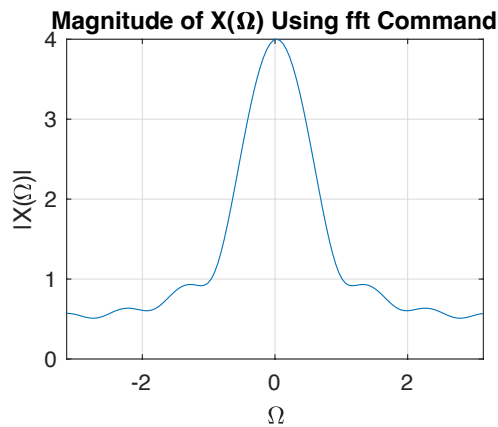
% A.2
% Compute DTFT of x[n] by hand calculation
X1 = @(Omega) (1 ...
    + (6/7).*exp(-1*j*Omega)...
    + (5/7).*exp(-2*j*Omega)...
    + (4/7).*exp(-3*j*Omega)...
    + (3/7).*exp(-4*j*Omega)...
    + (2/7).*exp(-5*j*Omega)...
    + (1/7).*exp(-6*j*Omega));

figure;
subplot(2,2,1);
plot(Omega,abs(X));
grid;
title("Magnitude of X(\Omega) Using fft Command");
xlabel('\Omega');
ylabel('|X(\Omega)|');
axis([-pi pi 0 4]);

subplot(2,2,2);
plot(Omega,angle(X));
grid;
title("Phase of X(\Omega) Using fft Command");
xlabel('\Omega');
ylabel('\angle X(\Omega)');
axis([-pi pi -2 2]);

subplot(2,2,3);
plot(Omega,abs(X1(Omega)));
grid;
title("Magnitude of X(\Omega) by Hand Calculation");
xlabel('\Omega');
ylabel('|X(\Omega)|');
axis([-pi pi 0 4]);

subplot(2,2,4);
plot(Omega,angle(X1(Omega)));
grid;
title("Phase of X(\Omega) Using fft by Hand Calculation");
xlabel('\Omega');
ylabel('\angle X(\Omega)');
axis([-pi pi -2 2]);
```



% The magnitude and phase of DTFT obtained by hand calculation give  
 % the same result as DTFT computed from fft command from Matlab.

% A.3

% Inverse zero-frequency shift

X = ifftshift(X);

% Compute x[n] from inverse DTFT using ifft

x\_n = ifft(X);

figure;

subplot(2, 1, 1);

stem(n, x(n));

grid;

title('Original Signal x[n]');

xlabel('n');

ylabel('x[n]');

axis([0 127 0 1]);

subplot(2, 1, 2);

stem(n,x\_n);

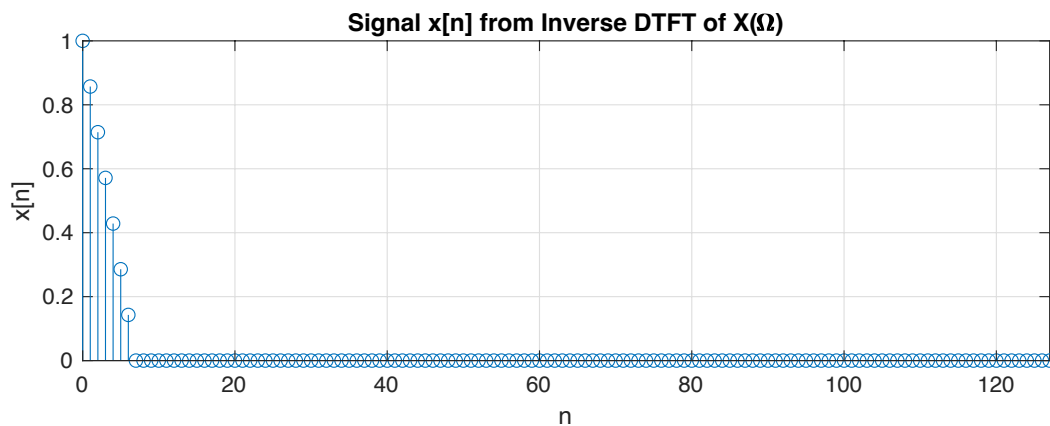
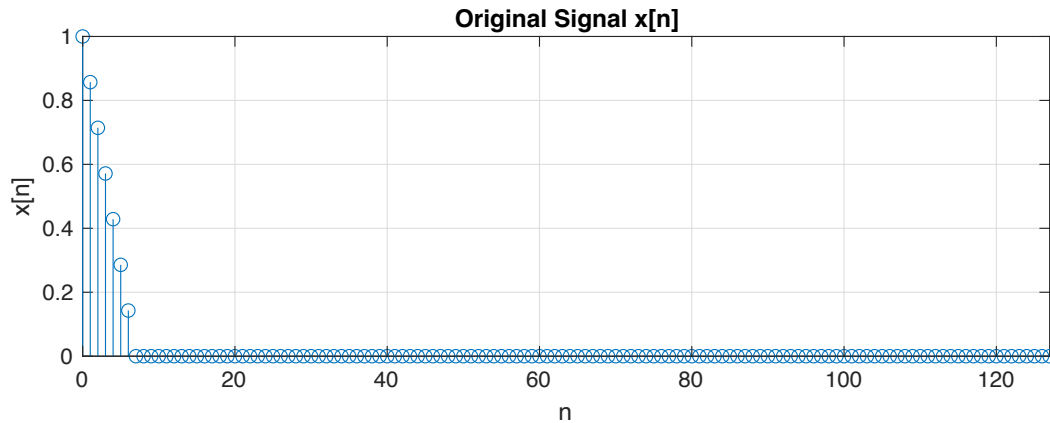
grid;

title('Signal x[n] from Inverse DTFT of X(\Omega)');

xlabel('n');

ylabel('x[n]');

axis([0 127 0 1]);



% The obtained result is the same as signal  $x[n]$  since `ifft` command  
 % is used to computed the inverse DTFT of  $X(\Omega)$ .

## B. Time convolution

% B. Time Convolution

% B.1

`n = 0:1000;`

`u = @(n) (n >= 0) * 1.0 .* (mod(n,1)==0);`

% Signal  $x[n]$

`x = @(n) sin(2*pi*n/10).*(u(n)-u(n-10));`

% Compute DTFT of signal  $x[n]$

`omega = linspace(-pi, pi, 1001);`

`W_omega_x = exp(-j).^((0:length(x(n))-1)'*omega);`

`X = (x(n)*W_omega_x);`

`figure;`

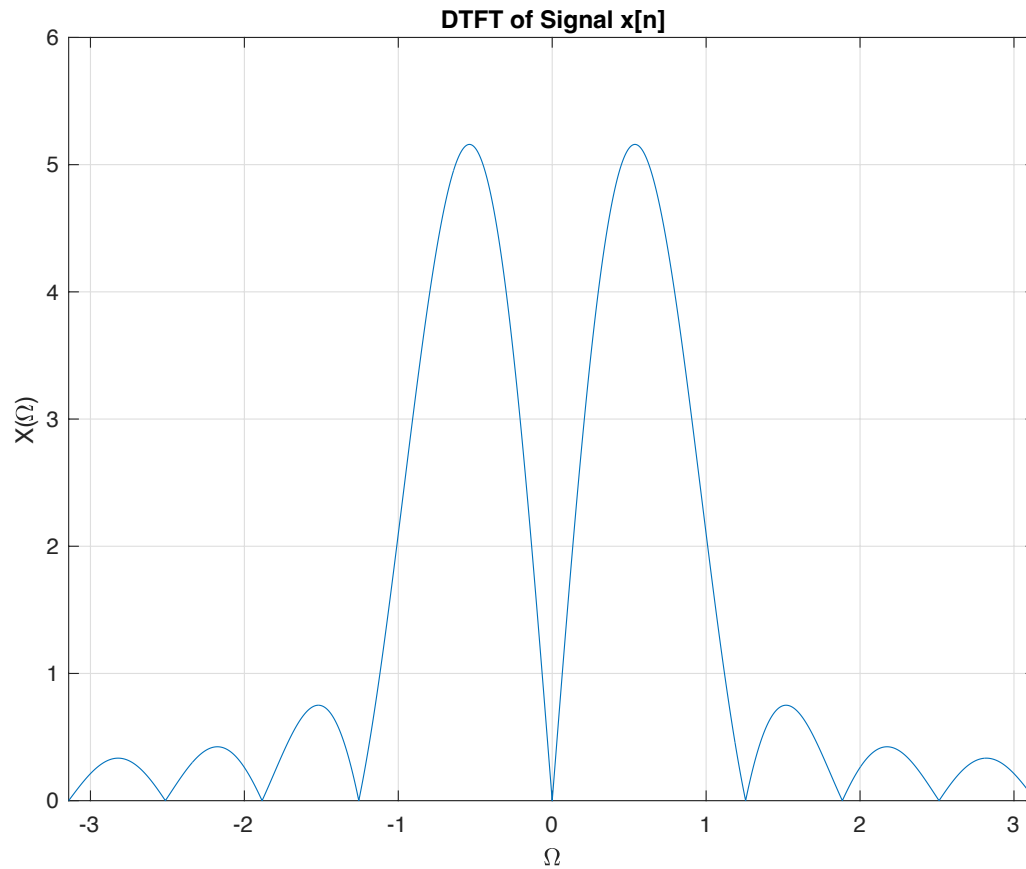
`plot(omega,abs(X));`

`grid;`

`title('DTFT of Signal  $x[n]$ ');`

`xlabel('\Omega');`

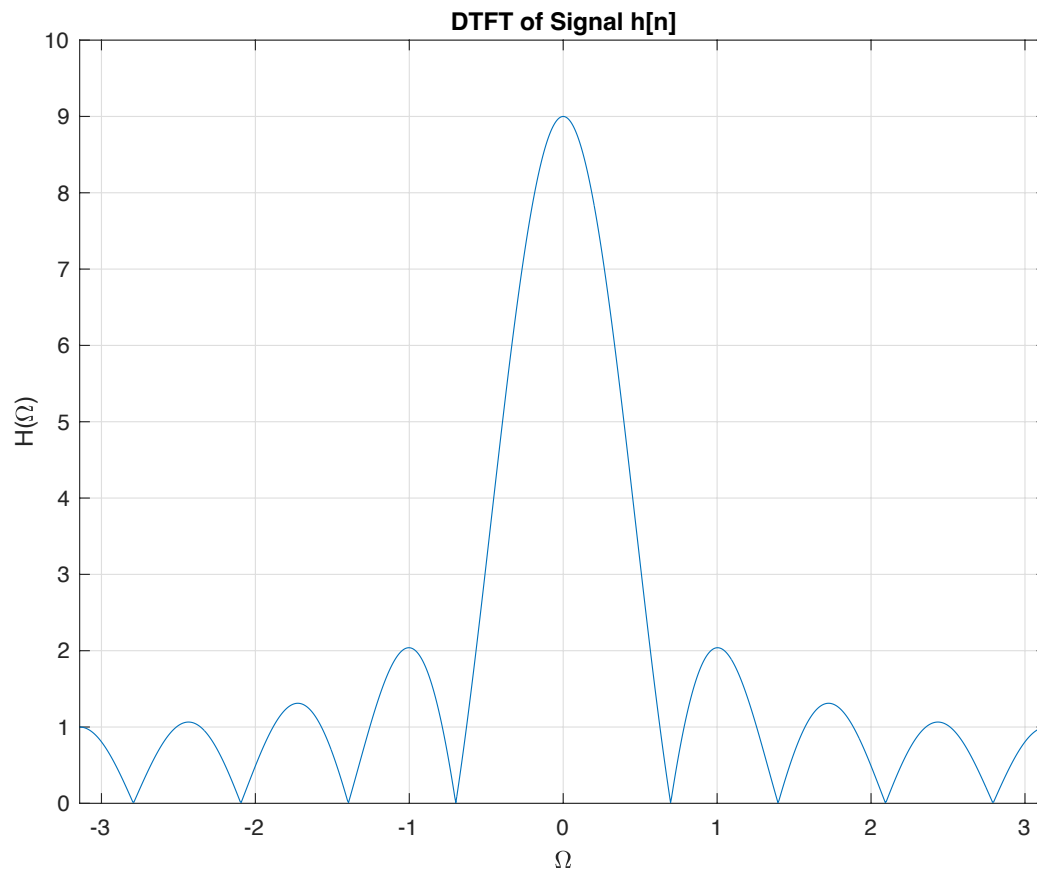
```
ylabel('X(\Omega)');
axis([-pi pi 0 6]);
```



**% B.2**

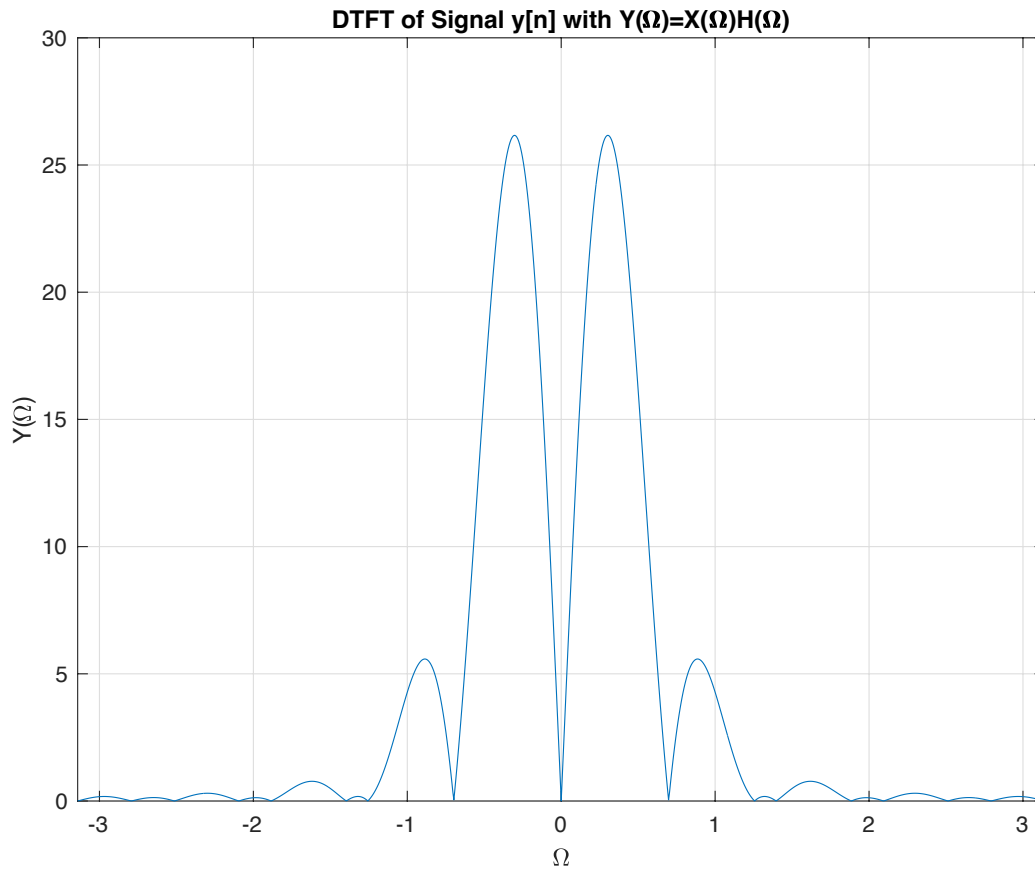
```
% Signal h[n]
h = @(n) u(n)-u(n-9);
% Compute DTFT of signal h[n]
W_omega_h = exp(-j).^((0:length(h(n))-1)*omega);
H = (h(n)*W_omega_h);
```

```
figure;
plot(omega,abs(H));
grid;
title('DTFT of Signal h[n]');
xlabel('\Omega');
ylabel('H(\Omega)');
axis([-pi pi 0 10]);
```



```
% B.3
Y1 = X.*H;

figure;
plot(omega,abs(Y1));
grid;
title('DTFT of Signal y[n] with Y(\Omega)=X(\Omega)H(\Omega)');
xlabel('\Omega');
ylabel('Y(\Omega)');
axis([-pi pi 0 30]);
```



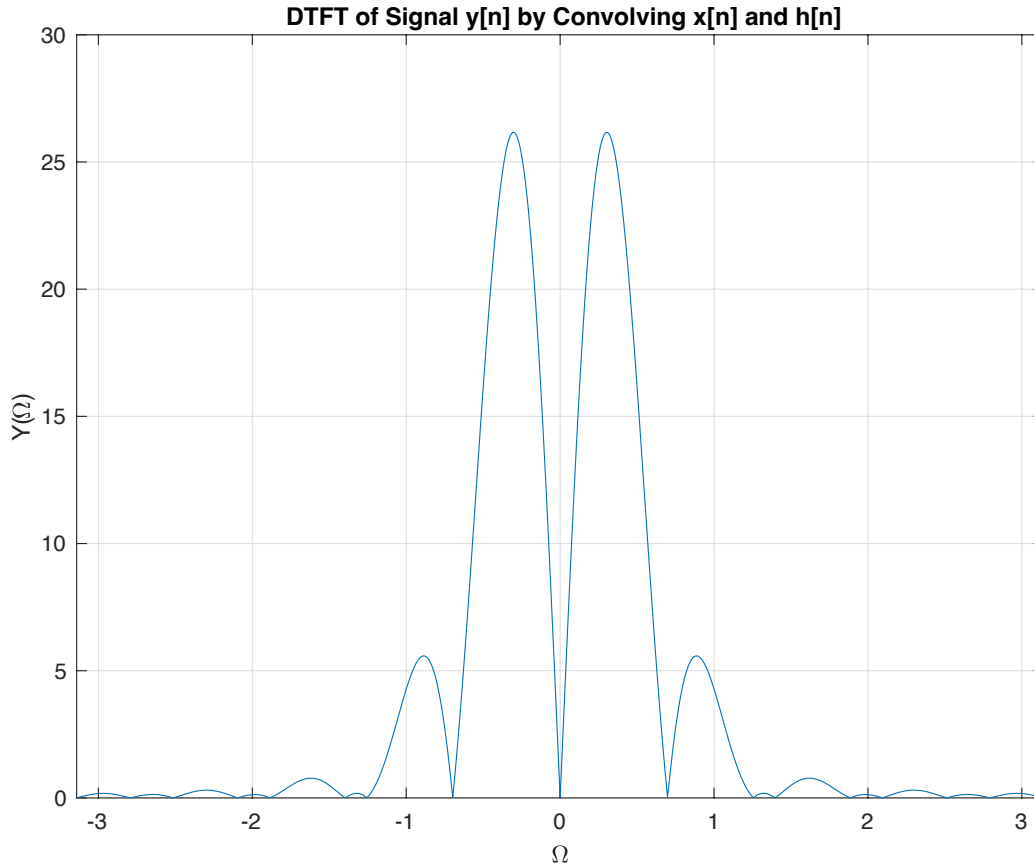
% B.4

```
% Compute y[n] by convolving x[n] and h[n]
y = @(n) conv(x(n),h(n));
```

% B.5

```
% Compute DTFT of signal y[n]
W_omega_y = exp(-j).^((0:length(y(n))-1)'*omega);
Y2 = (y(n)*W_omega_y);
```

```
figure;
plot(omega,abs(Y2));
grid;
title('DTFT of Signal y[n] by Convolving x[n] and h[n]');
xlabel('\Omega');
ylabel('Y(\Omega)');
axis([-pi pi 0 30]);
```



% B.6

% The results in part 3 and 5 are the same as it shows the time  
 % convolution property of the DTFT. Specifically, the convolution  
 % of signals in the time domain will be transformed into the  
 % multiplication of their Fourier transforms in the frequency domain.

## C. FIR filter design by frequency Sampling

% C.1

N = 35; % filter length

% Create N equally spaced frequency samples

Omega = linspace(0,2\*pi\*(1-1/N),N)';

% Design of a high pass FIR filter

H\_d\_35 = @(Omega) (mod(Omega,2\*pi)>2\*pi/3) & (mod(Omega,2\*pi)<2\*pi-2\*pi/3);

% Sample the desired magnitude response and create h[n]

H\_35 = H\_d\_35(Omega);

% Define phase to shift h[n] by (N - 1)/2

H\_35 = H\_35.\*exp(-j\*Omega\*((N-1)/2));

H\_35(fix(N/2)+2:N,1) = H\_35(fix(N/2)+2:N,1)\*((-1)^(N-1));

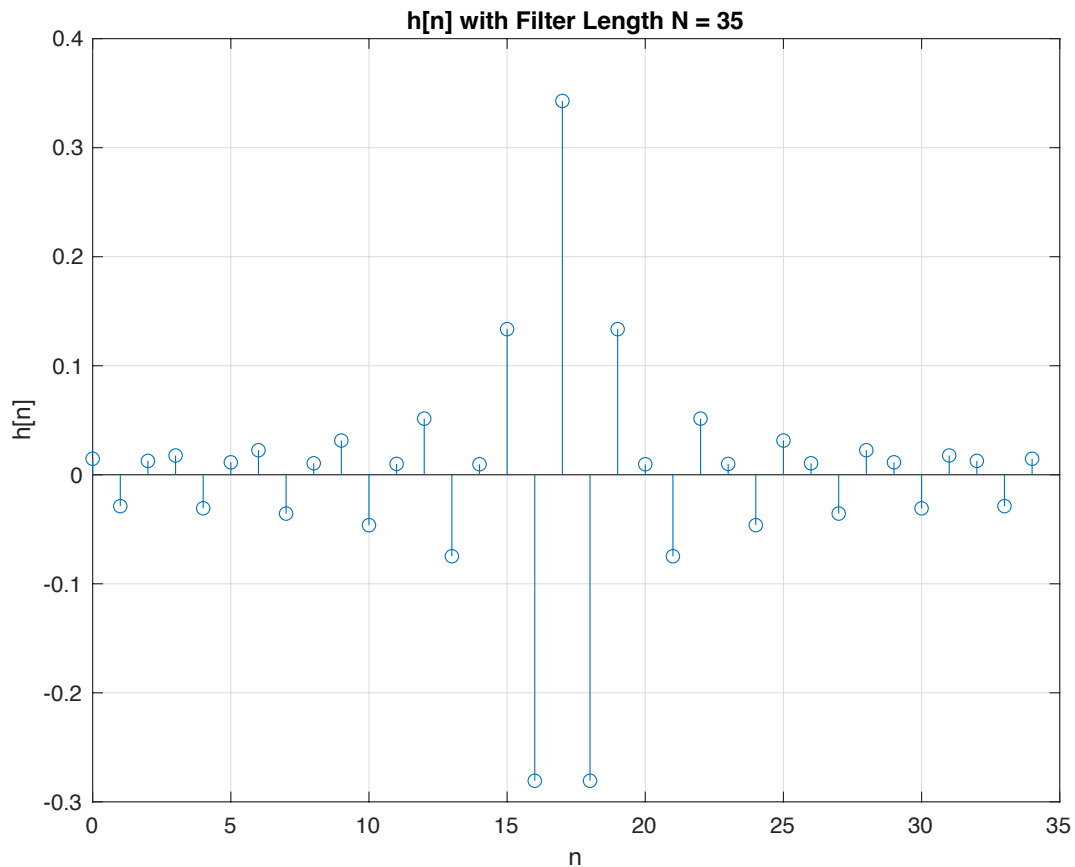


```

% Create h[n]
h_35 = real(ifft(H_35));

figure;
stem(0:N-1,h_35);
grid;
title('h[n] with Filter Length N = 35');
xlabel('n');
ylabel('h[n]');

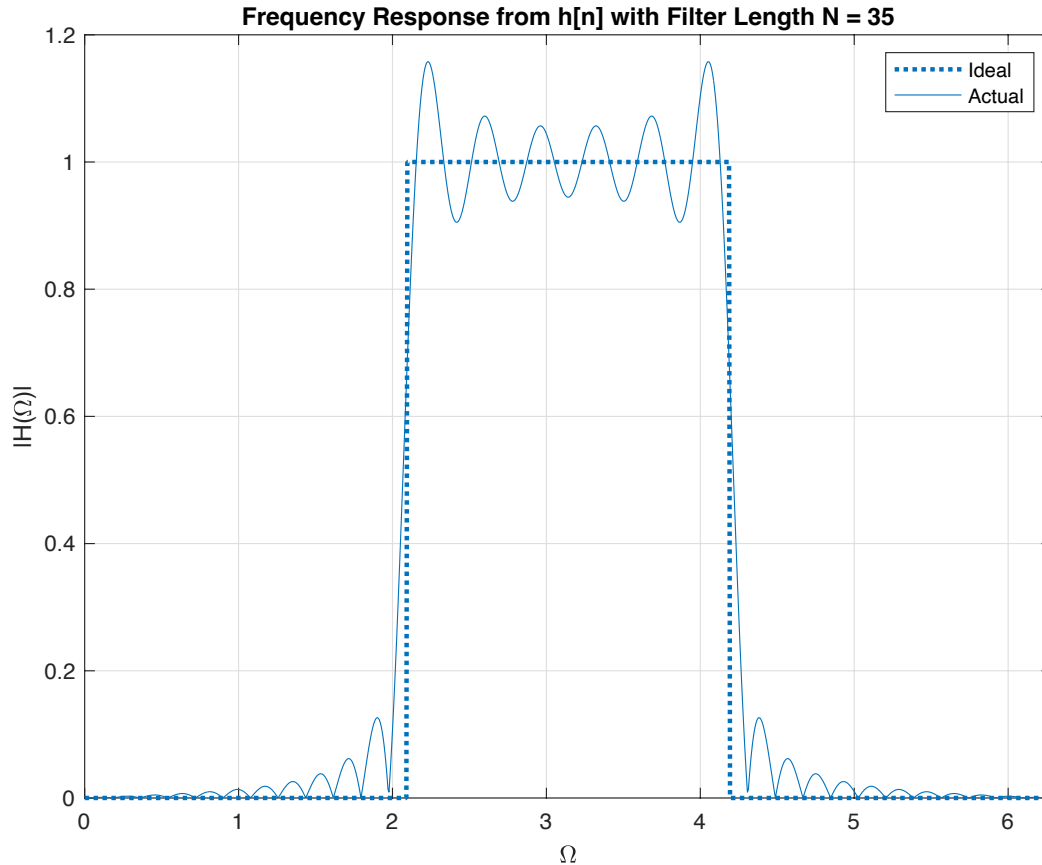
```



```

% C.2
Omega = linspace (0, 2*pi, 1002);
H_35 = freqz(h_35,1,0:2*pi/1001:2*pi);
figure;
plot(Omega,H_d_35(Omega),':','LineWidth',2);
hold on;
plot(Omega,abs(H_35),'color',[0.00 0.45 0.74]);
hold off;
grid;
title('Frequency Response from h[n] with Filter Length N = 35');
xlabel('\Omega');
ylabel('|H(\Omega)|');
axis([0 2*pi 0 1.2]);
legend('Ideal','Actual');

```



% C.3

% The frequency response with filter length  $N = 35$  has general  
 % shape which is close to the ideal filter with occurrence of ripple.

% C.4

$N = 71$ ; % filter length

% Create  $N$  equally spaced frequency samples

$\Omega = \text{linspace}(0, 2\pi \cdot (1 - 1/N), N)'$ ;

% Design of a high pass FIR filter

$H\_d\_71 = @( \Omega ) (\text{mod}(\Omega, 2\pi) > 2\pi/3) \ \& \ (\text{mod}(\Omega, 2\pi) < 2\pi - 2\pi/3)$ ;

% Sample the desired magnitude response

$H\_71 = 1.0 \cdot H\_d\_35(\Omega)$ ;

% Define phase to shift  $h[n]$  by  $(N - 1)/2$

$H\_71 = H\_71 \cdot \exp(-j \cdot \Omega \cdot ((N - 1)/2))$ ;

$H\_71(\text{fix}(N/2) + 2:N, 1) = H\_71(\text{fix}(N/2) + 2:N, 1) \cdot ((-1)^{(N - 1)})$ ;

% Create  $h[n]$

$h\_71 = \text{real}(\text{ifft}(H\_71))$ ;

figure;

stem(0:N-1,  $h\_71$ );

grid;

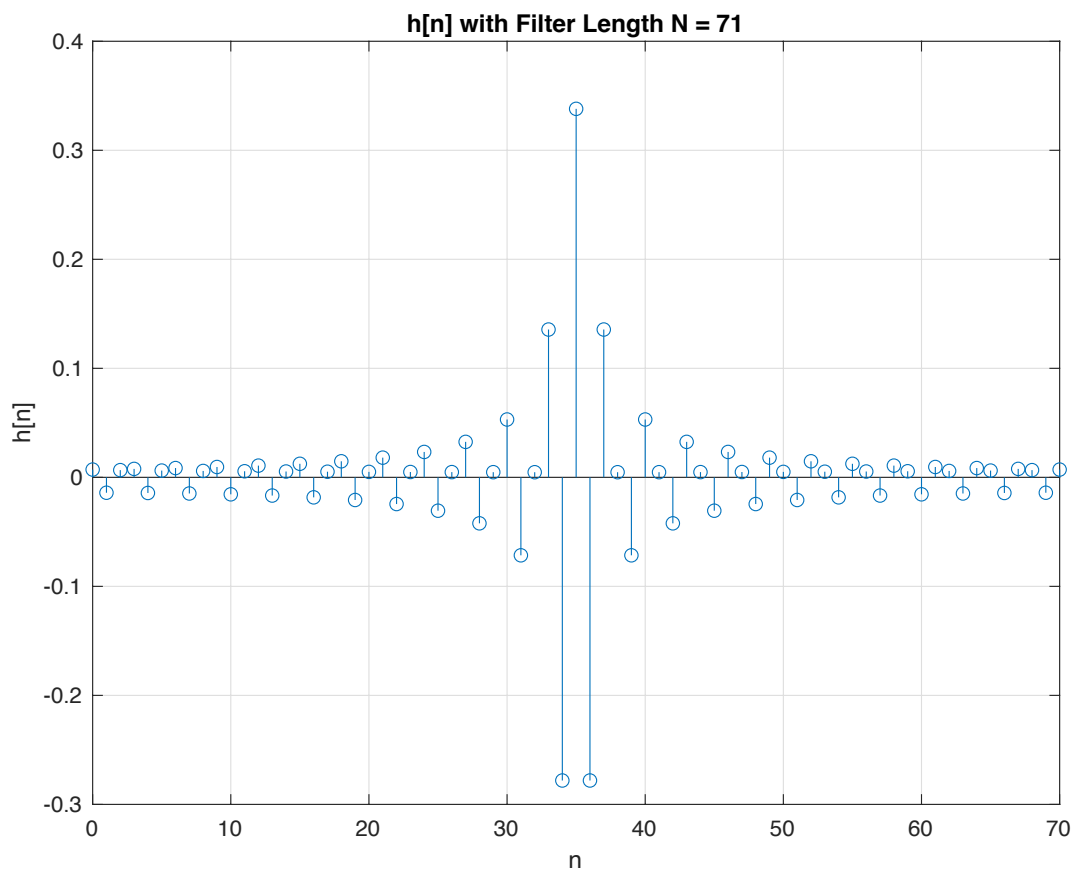
title('h[n] with Filter Length  $N = 71$ ');

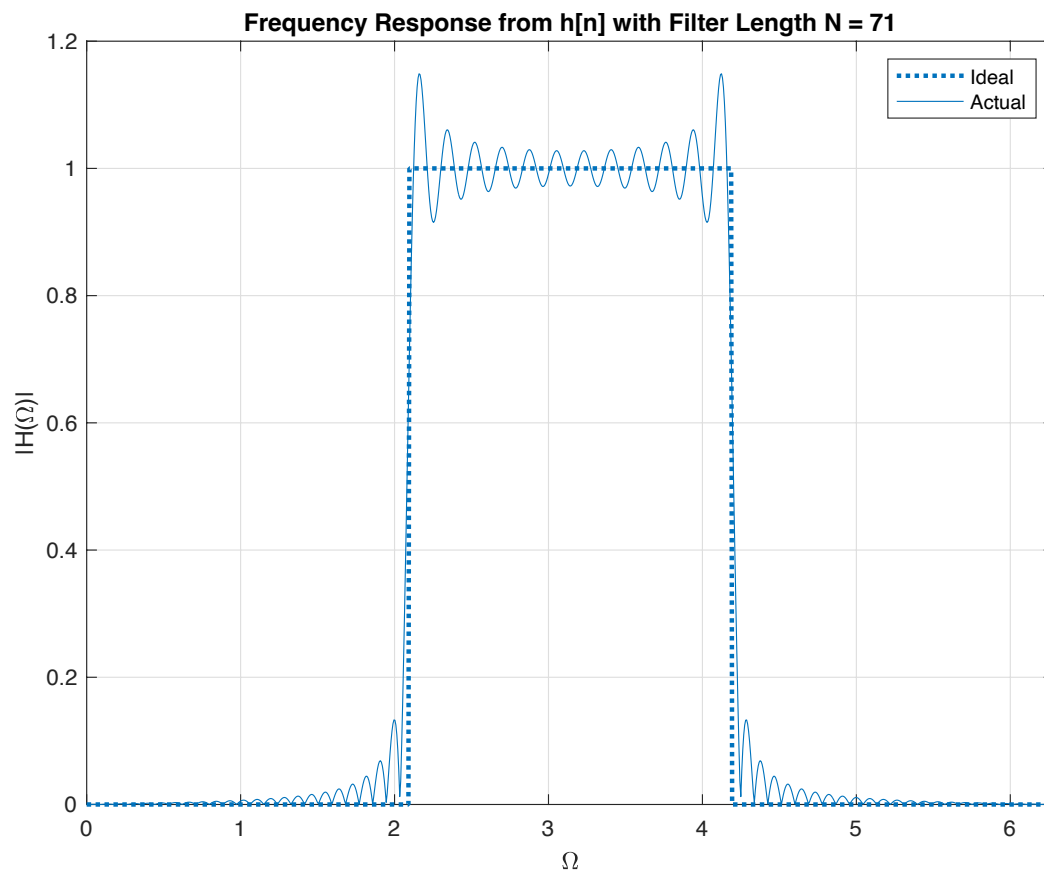
```

xlabel('n');
ylabel('h[n]');

Omega = linspace (0, 2*pi, 1002);
H_71 = freqz(h_71,1,0:2*pi/1001:2*pi);
figure;
plot(Omega,H_d_71(Omega),':','LineWidth',2);
hold on;
plot(Omega,abs(H_71),'color',[0.00 0.45 0.74]);
hold off;
grid;
title('Frequency Response from h[n] with Filter Length N = 71');
xlabel('\Omega');
ylabel('|H(\Omega)|');
axis([0 2*pi 0 1.2]);
legend('Ideal','Actual');

```





% C.5

% When the length of filter increased to 71, the ripple in the  
 % frequency response is significantly reduced as it has more samples.  
 % Therefore, the quality of the filter is improved and the response is  
 % closer to the ideal filter.