Code for Assignment-3

Importing Sound File and Extracting Input x[n]

Parameters

```
% Sampling Frequency
Fs = 22050;

% Term in Impulse Response or BandWidth of Low-Pass Filter
k = 1;
Wo = 0.5*Fs;

% Reading Audio File
[Data,Fs] = audioread("msmn1.wav");

% Size of Input Array
s = size(Data);
s = s(1)
```

```
s = 110250
```

```
% Ns = No.of Samples taken during Fourier Transform
% If x = k*s is not an integer Ns = floor(x) or Ns = ceil(x) is considered such that
% Ns is an Even Integer

x = floor(1*s);     % Considering all points in input audio file to attain maximum outp
x = x(1);
if rem(x,2) == 1
    Ns = x+1;
else
    Ns = x
end
```

```
Ns = 110250
```

Listening to Audio File

```
%soundsc(Data,Fs);
```

Impulse Response h[n] or Low-Pass Filter

h_values = h[n]

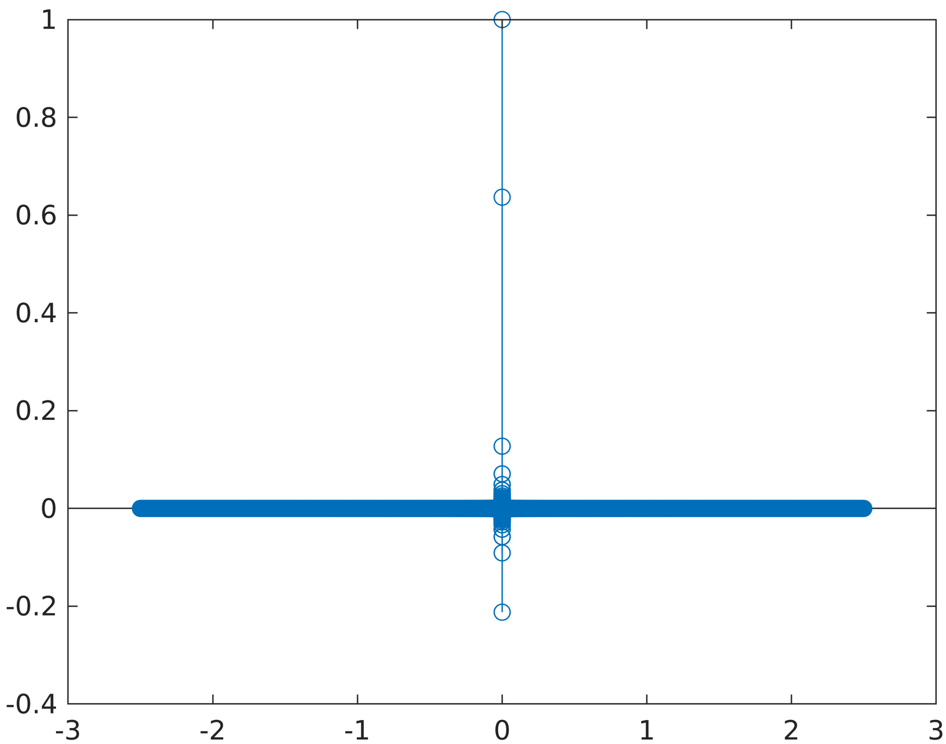
h[n] = h(t) | t=nTs

```
Ts = 1/Fs;
n = [-s/2:s/2];
```

```
t = (Ts*n)';
h_values = sinc(Wo*t);
```

Plotting Impuse Response of System

```
stem(t,h_values)
hold off
```



H_w = N-point Fourier Transform of h[n] = H(jw)

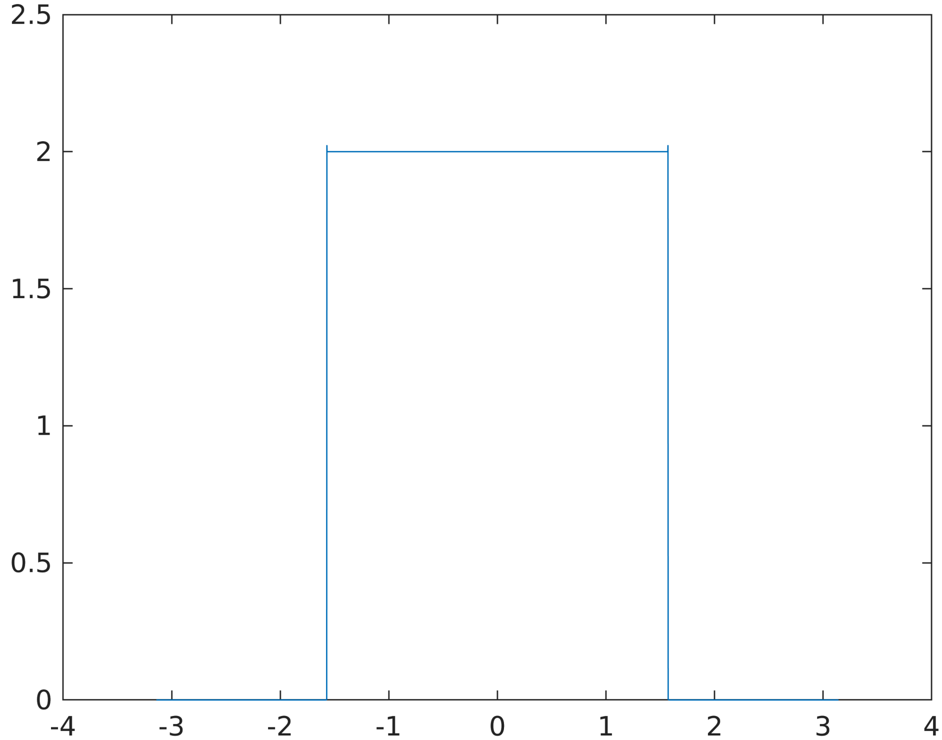
```
H_w = abs(fft(h_values,Ns));
```

Visualising Fourier Transform of Impulse Response h[n]

```
H_w_m_plot = abs(fftshift(H_w));
size(H_w_m_plot);
%t = (s-Ns)/2;
%H_w_plot = padarray(H_w_m_plot,t,'both');
%size(H_w_plot);
```

Plotting Fourier Transform of Impuse Response

```
%f = 2*pi*(1*(-s/2:(s/2)-1)/s)';
%size(f);
fm = 2*pi*(1*(-Ns/2:(Ns/2)-1)/Ns)';
plot(fm,H_w_m_plot)
hold off
```




```
% plot(f,H_w_plot);
```

Low Frequency Components of x[n]

y = y[n] and Data = x[n]

Y_w = Y(jw) and X_w = X(jw)

Y(jw) = X(jw).H(jw)

```
X_w = fft(Data,Ns);
size(X_w)
```

```
ans = 1×2
      110250           1
```

Output of Signal when passed through Low-Pass Filter

```
Y_w = X_w.*H_w;
y = ifft(Y_w);
```

Listen to High Freqency Components

```
%audiowrite('Low_Pass_Audio.wav', y, 22050);
%soundsc(y,Fs);
```

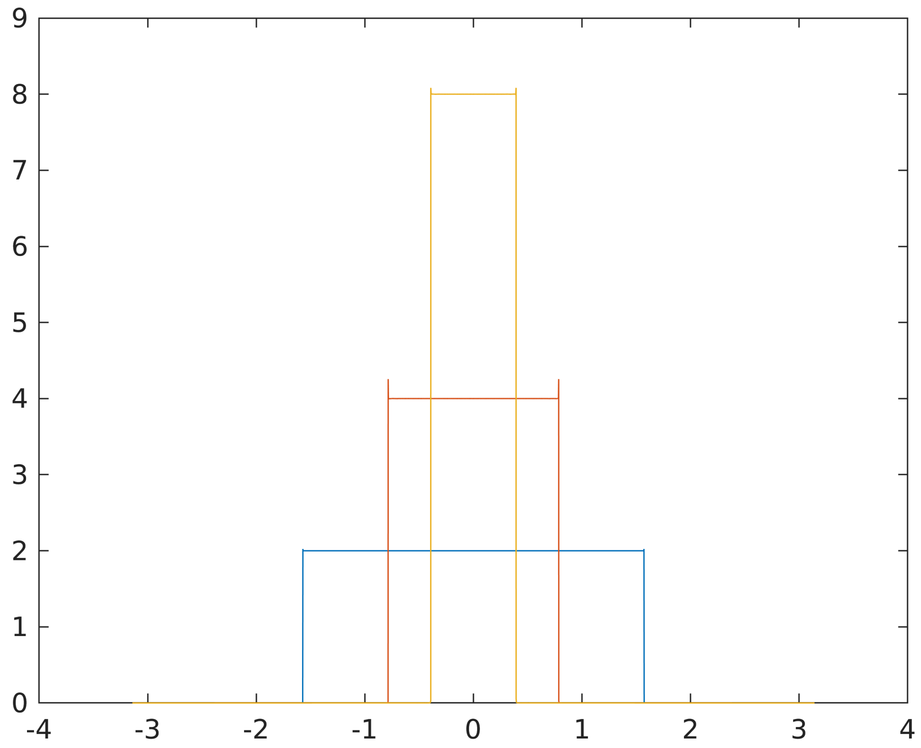
Function to Pass Audio Files through Low-Pass Filter

Testing for Different values of Wo

```
for w = [0.5*Fs 0.25*Fs 0.125*Fs]
    signal = ProcessLowPassAudio(Data,w,Ns,Fs);
end
```

```
filename =
'Low_Pass_Audio_with_W=11025.wav'
filename =
'Low_Pass_Audio_with_W=5512.5.wav'
Warning: Data clipped when writing file.
filename =
'Low_Pass_Audio_with_W=2756.25.wav'
Warning: Data clipped when writing file.
```

```
hold off
```



High Pass Filter

G_w = N-point Fourier Transform of High-Pass Filter = G(jw)

Using Low Pass to High Pass Transformation to get the Transfer Function of High Pass Filter.

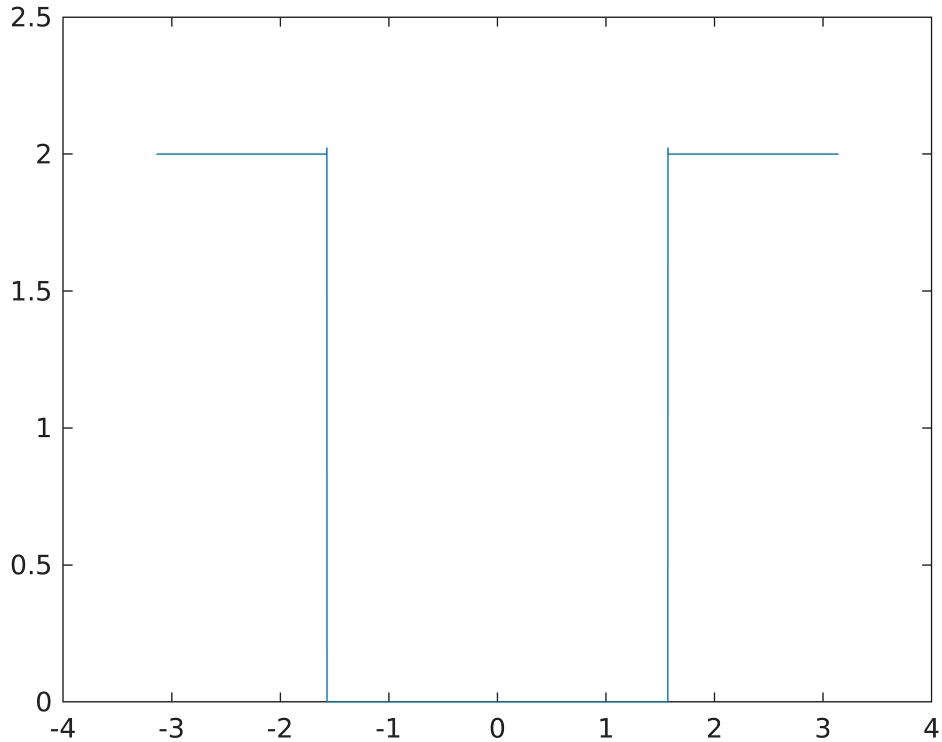
Note: On using Transformation Low-Pass and High-Pass Filters dont have same Cut-off Frequency.

G(jw) = H(j(w+pi))

```
% Frequency Response of High-Pass Filter
% fftshift command is used to shift H(jw) by pi
G_w = fftshift(H_w);
```

Plotting High-Pass Filter

```
G_w_m_plot = abs(fftshift(G_w));
plot(fm,G_w_m_plot)
hold off
```



High Frequency Components of x[n]

z = z[n] and Data = x[n]

Z_w = Z(jw) and X_w = X(jw)

Z(jw) = X(jw).G(jw)

Output of Signal when Passed through High Pass Filter

```
Z_w = X_w.*G_w;
z = ifft(Z_w);
```

Listen to High Freqency Components

```
%soundsc(g,Fs)
%audiowrite('High_Pass_Audio.wav', z, 22050);
```

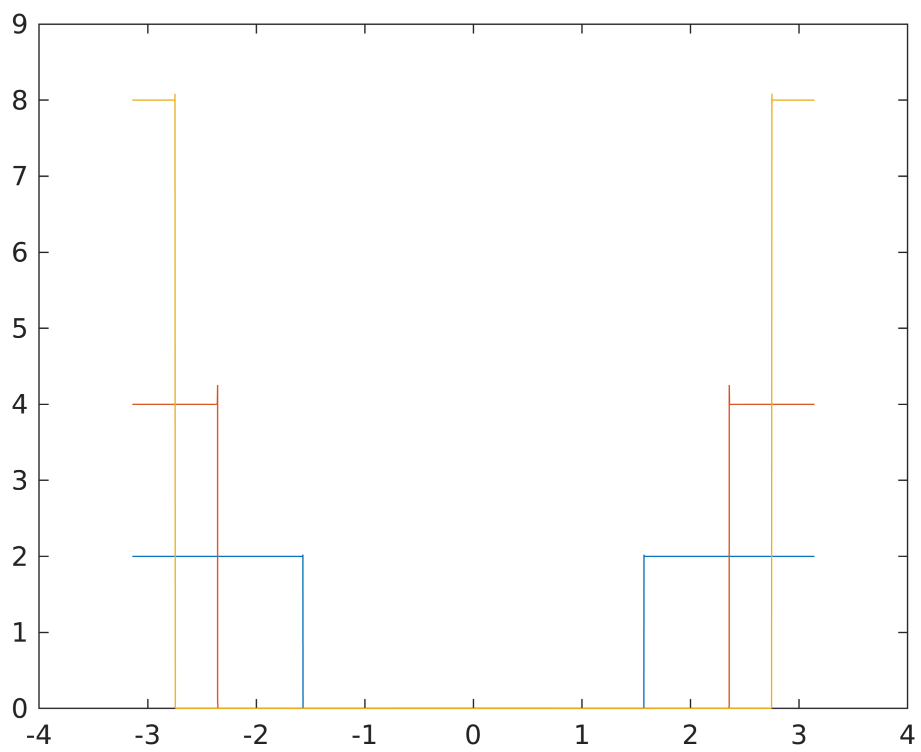
Function to Pass Audio Files through High-Pass Filter

Testing for Different values of Wo

```
for w = [0.5*Fs 0.25*Fs 0.125*Fs]
    signal = ProcessHighPassAudio(Data,w,Ns,Fs);
end
```

```
filename =
'High_Pass_Audio_with_Cutoff_Freq=5512.5.wav'
filename =
'High_Pass_Audio_with_Cutoff_Freq=8268.75.wav'
filename =
'High_Pass_Audio_with_Cutoff_Freq=9646.875.wav'
```

```
hold off
```



H(z)

Parameters

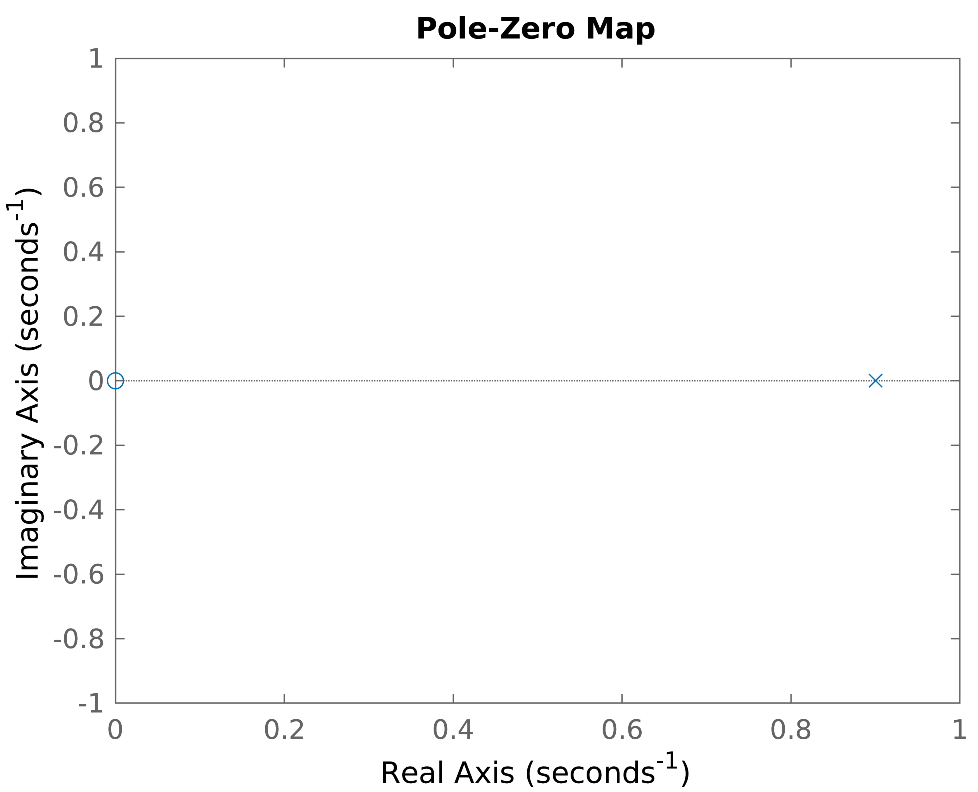
```
Fs = 22050;
c = 0.5;
Nz = 1*s;

% a in H(z)
a = 0.9;

% H(z)
Hz = @(z) (1-a)/(1-a*(z^-1));
```

Pole and Zero Plot of H(z). By rearranging terms and using a built in function

```
TF = tf([1-a 0],[1 -a]);
pzmap(TF)
```



Substituting z = r*e^jw in H(z) where r = 1 which satisfies the condition |z| > |a|

`Hz_w = H(e^jw) --- Low Pass Filter`

`Xz_w = X(jw)`

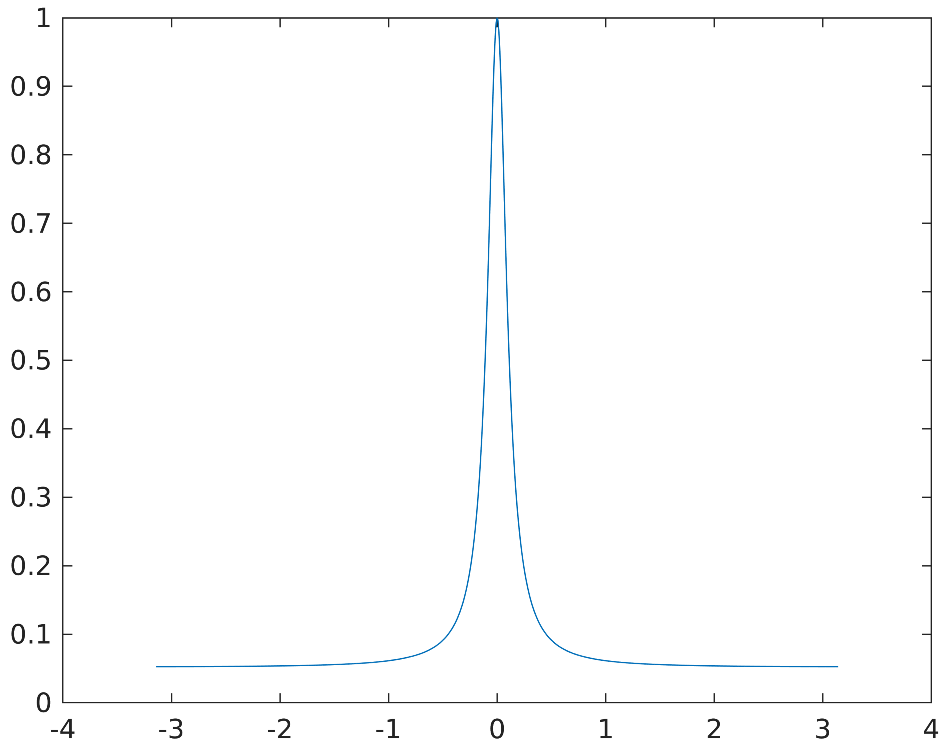
```
r = 1;
w = linspace(-pi,pi,Nz);
Hz_w = arrayfun(Hz,(r * exp(1j*w))');
```

```
Xz_w = fft(Data,Nz);
```

Plotting Low-Pass Filter H(w)

```
plot(w,Hz_w)
```

```
Warning: Imaginary parts of complex X and/or Y arguments ignored
```



Output of Signal when Passed through Low Pass Filter

Yz_w = Y(jw) = X(jw).H(jw)

```
Yz_W = Xz_w.*Hz_w;
yz = real(ifft(Yz_W));
```

Listening to Low Frequency Components

```
%soundsc(yz,Fs);
audiowrite('H(z)_Low_Pass_Audio.wav', yz, 22050);
```

Gz_w = G(e^jw) --- High Pass Filter

Using Low Pass to High Pass Transformation to get the Transfer Function of High Pass Filter

Note: On using Transformation Low-Pass and High-Pass Filters dont have same Cut-off Frequency.

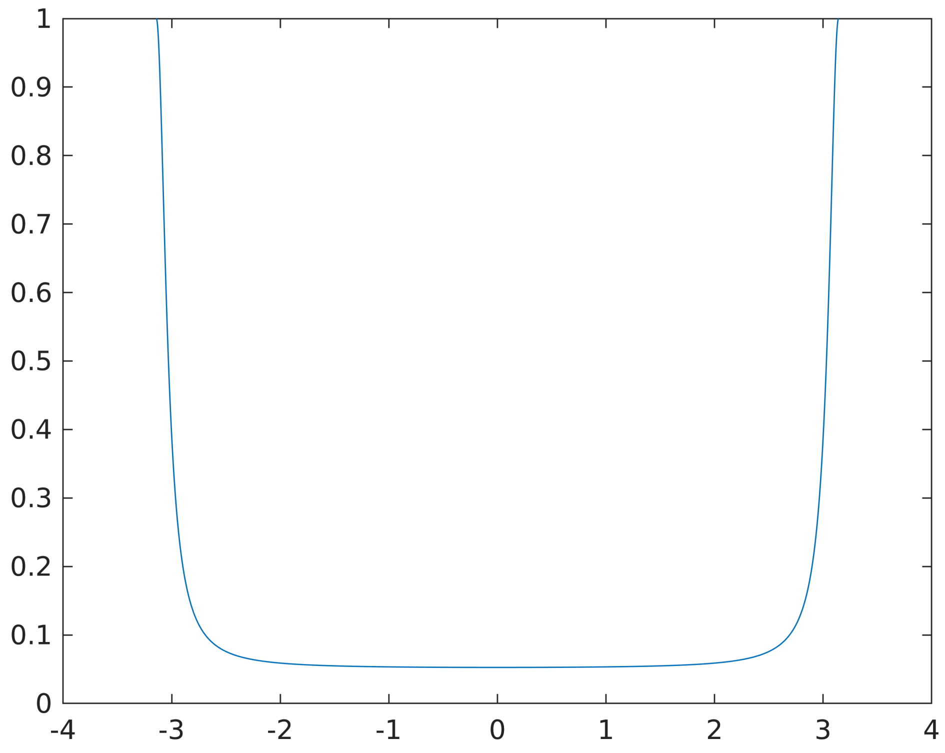
G(jw) = H(j(w+pi))

```
Gz_w = fftshift(Hz_w);
```

Plotting High Pass Filter G(w)

```
plot(w,Gz_w)
```

```
Warning: Imaginary parts of complex X and/or Y arguments ignored
```



`Zz_w = Z(jw) = X(jw).G(jw)`

Output of Signal when Passed through High Pass Filter

```
Zz_w = Xz_w.*Gz_w;
zz = real(ifft(Zz_w));
```

Listening to High Frequency Components

```
%soundsc(zz,Fs);
audiowrite('H(z)_High_Pass_Audio.wav', zz, 22050);
```

Functions used in the Code

```
function signal = ProcessLowPassAudio(data,w,m,Fs)
    % t = nTs substitution
    Ts = 1/Fs;
    n = [-m/2:m/2];
    t = (Ts*n)';
```

```
    % Impulse Response of Low-Pass Filter
    h_values = sinc(w*t);

    % Fourier Transform of Impulse Response
    H_w = abs(fft(h_values,m));

    % Fourier Transform of Input
    X_w = fft(data,m);

    % Multiplication in Frequency Domain
    Y_w = X_w.*H_w;

    % Inverse Fourier Transform or Reconstruction of Signal
    signal = ifft(Y_w);

    H_w_m_plot = abs(fftshift(H_w));
    fm = 2*pi*(1*(-m/2:(m/2)-1)/m)';
    plot(fm,H_w_m_plot)
    hold on

    a = 'Low_Pass_Audio_with_W=';
    b = num2str(w);
    c = '.wav';
    filename = strcat(a,b,c)
    %soundsc(y,Fs)
    audiowrite(filename, signal, Fs);
end

function signal = ProcessHighPassAudio(data,w,m,Fs)
    % t = nTs substitution
    Ts = 1/Fs;
    n = [-m/2:m/2];
    t = (Ts*n)';

    % Impulse Response of Low-Pass Filter
    h_values = sinc(w*t);

    % Fourier Transform of Impulse Response
    H_w = abs(fft(h_values,m));

    % Fourier Transform of Input
    X_w = fft(data,m);

    % Low-Pass to High Pass Transformation and Multiplication in Frequency Domain
    G_w = fftshift(H_w);
    Z_w = X_w.*G_w;

    % Inverse Fourier Transform or Reconstruction of Signal
    signal = ifft(Z_w);

    G_w_m_plot = abs(fftshift(G_w));
    fm = 2*pi*(1*(-m/2:(m/2)-1)/m)';
    plot(fm,G_w_m_plot)
```

```
    hold on

    a = 'High_Pass_Audio_with_Cutoff_Freq=';
    b = num2str((Fs-w)/2);
    c = '.wav';
    filename = strcat(a,b,c)

    %soundsc(y,Fs)
    audiowrite(filename, signal, Fs);
end
```