Project Report

Team Members:

- Harikrishnan K.P (150260026)
- Pranjal R.S (150260030)

AIM

Write the code for convolution in MATLAB and using it pass different signals (voice signal and music) through the given filters and make observations about the output signal.

PROCEDURE

A function for convolution was written, which takes the input signal and the filter as input parameters and returns the convoluted signal. This file titled '*convolution.m*' has been uploaded. We then set about passing different signals through the filter and observed what happens.

The 2 filters given are:

- h1 = [1 1 1 1 1 1 1]/7
- h2 = [-1 -1 -1 2 1 1 1]/2

CODE

This function contains code for the convolution of x & h, where x is the input and h is the filter. This file has been uploaded, titled ‘*convolution.m*’.

```
function z=convolution(x,h)
    lx=length(x);           % lx is length of input signal
    lh=length(h);           % lh is length of filter
    z=zeros(lx+lh,1);       % defining a column vector for the output
    % Convolution
    for n=2:lh              % component values of output from '2' to 'lh'
        for k=1:n-1
            z(n,1)=z(n,1)+x(k,1)*h(1,n-k);
        end
    end

    for n=lh+1:lx+1         % component values of output from 'lh-1' to
'lx+1'
        for k=n-lh:n-1
            z(n,1)=z(n,1)+x(k,1)*h(1,n-k);
        end
    end

    for n=lx+2:lx+lh        % component values of output from 'lx+2' to
'lx+lh'
        for k=n-lh:lx
            z(n,1)=z(n,1)+x(k,1)*h(1,n-k);
        end
    end
end
```

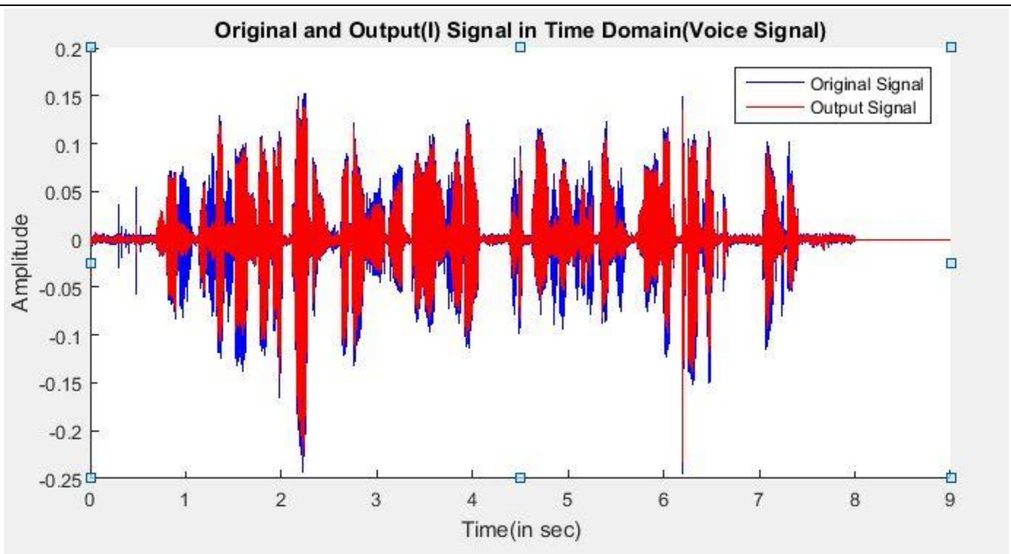
The following code reads an audio file and using the ‘*convolution’* function defined above, obtains the output when it is passed through the filter one or two times. The Fourier transform of the output is also taken. Plots in both time and frequency domain are obtained as output.
This file has been uploaded titled ‘*signal_analysis.m*’.

```
% In this file the audio used is the voice signal, please correct the file
path to get output.
% The filter defined is h2, for analysing h1, please change the filter
definition
% For obtaining the output after signal is passed through filter twice,
replace y2 with y.
```

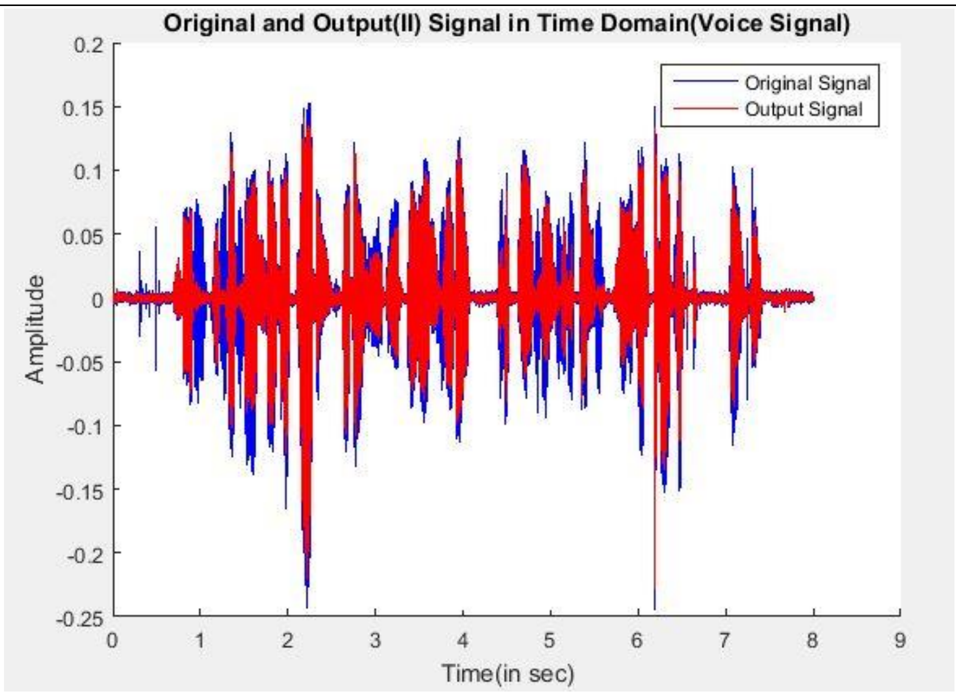
```
% If filter is changed please swap the order of plotting of input and
output to avoid masking.


Fs = 44100;                         %Sampling Frequency
x = audioread('C:\Users\Harikrishnan\Documents\MATLAB\Voice\voice.wav');
%input signal
h = [-1 -1 -1 2 1 1 1]/2;            %Impulse response

y = convolution(x,h);               %Finding output using convolution,
convolution function has been defined in a separate matlab file
y2 = convolution(y,h);              %Finding output after filtering twice
l1 = length(x);                     %length of signal
n_x = 2^nextpow2(l1);               %converting length of signal to the next
highest power of 2
X = fft(x,n_x);                     %fourier transform
X = X(1:n_x/2);                     %removing half of the signal
mag_X = abs(X) ;

l2 = length(y);                     %length of signal
n_y = 2^nextpow2(l2);               %converting length of signal to the next
highest power of 2
Y = fft(y,n_y);                     %fourier transform
Y = Y(1:n_y/2);                     %removing half of the signal
mag_Y = abs(Y) ;


figure(1);
plot((Fs*(0:(n_y/2)-1))/n_y,mag_Y,'color','g');   %plot the output Signal
in Frequency Domain
hold on
plot(Fs*(0:((n_x/2)-1))/n_x,mag_X,'color','b');   %plot the input Signal in
Frequency Domain


xlabel('Frequency(in Hz)')                        %Add x label
ylabel('Power')                                   %Add y label
title('Original and Output Signals in Frequency Domain(Music Signal)')
%Add Title
legend('Output Signal','Original Signal')
out = y;

figure(2);
plot((1:length(out))/44100,out,'color','r')         %Plot Output Signal in
Time Domain
hold on
plot((1:length(x))/44100,x,'color','b')  %Plot Input Signal in Time Domain


xlabel('Time(in sec)')                              %Add X label
ylabel('Amplitude')                                 %Add Y label
title('Original and Output Signal in Time Domain(Music Signal)')
%Add Title
legend('Output Signal','Original Signal')           %Add Legend
```

OBSERVATIONS

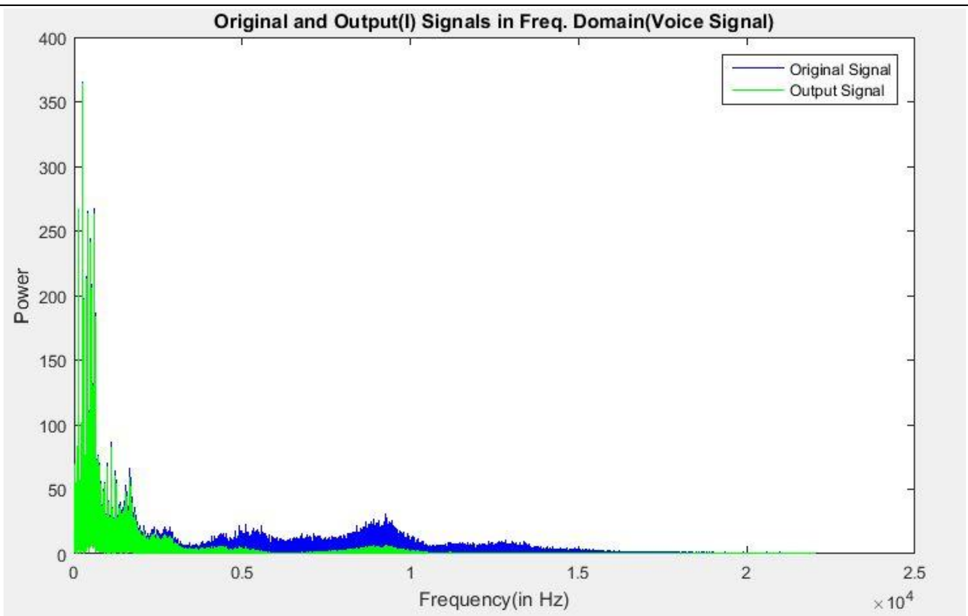
On passing a voice audio *'voice.wav'* through h1 filter, the following observations were made:



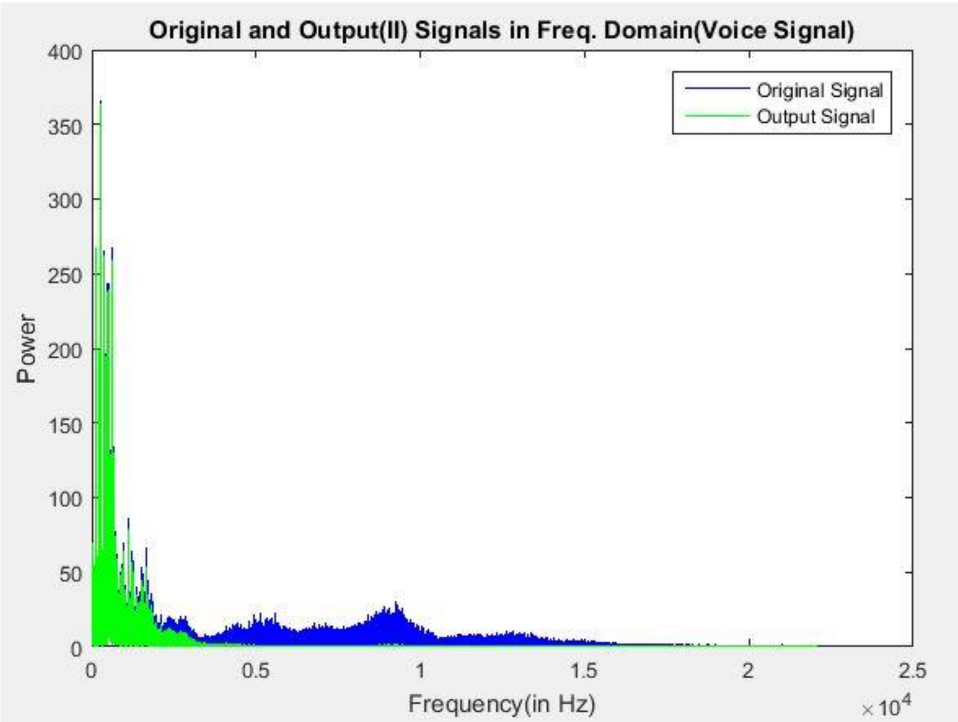
After passing through filter once



After passing through filter twice

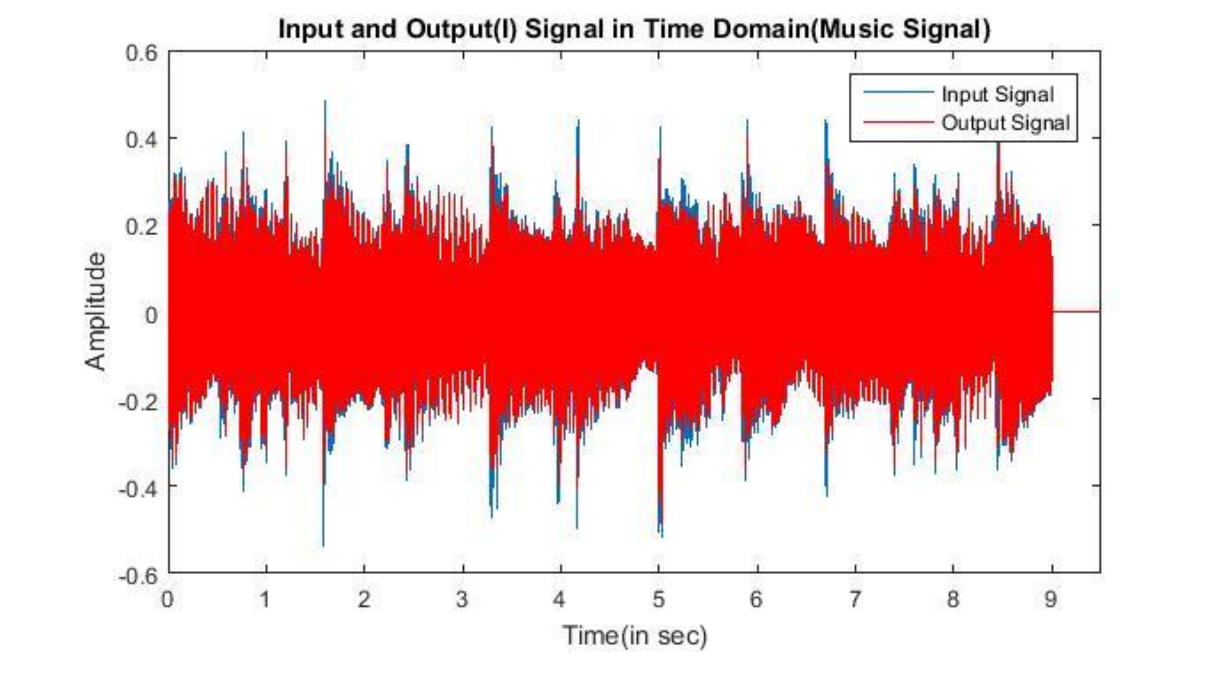


After passing through filter once

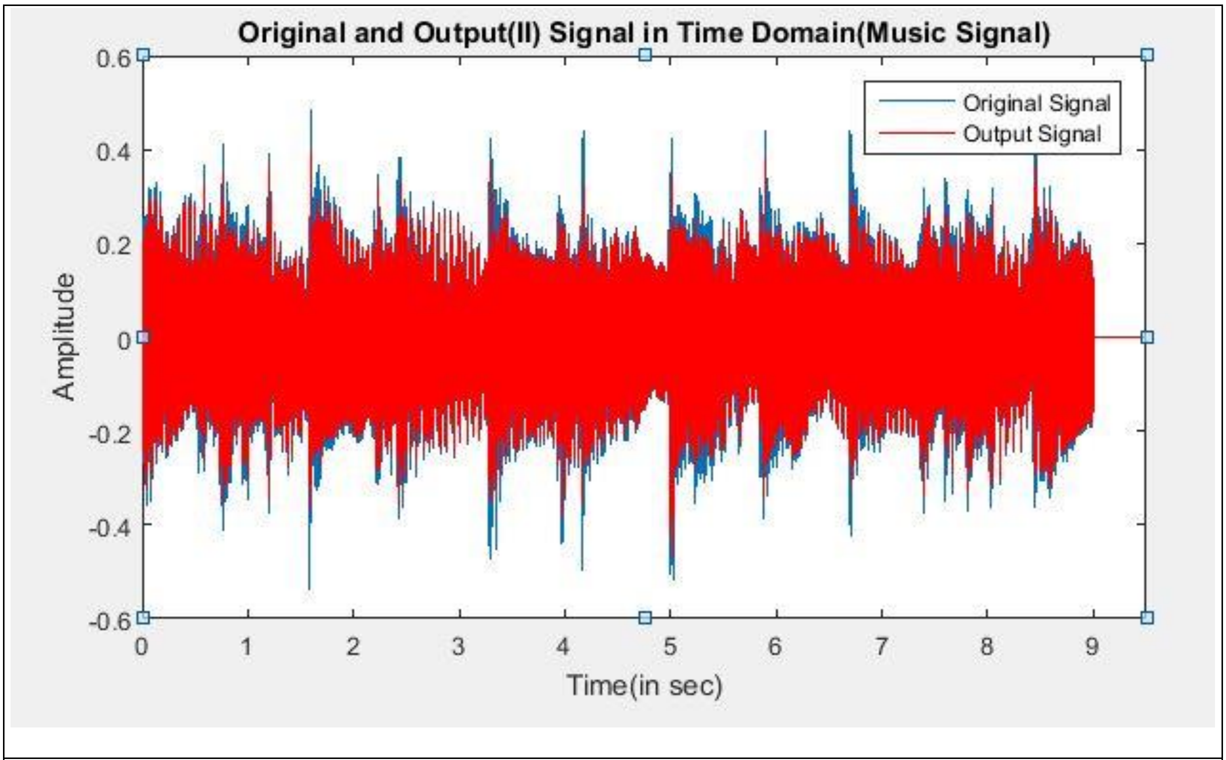


After passing through filter twice

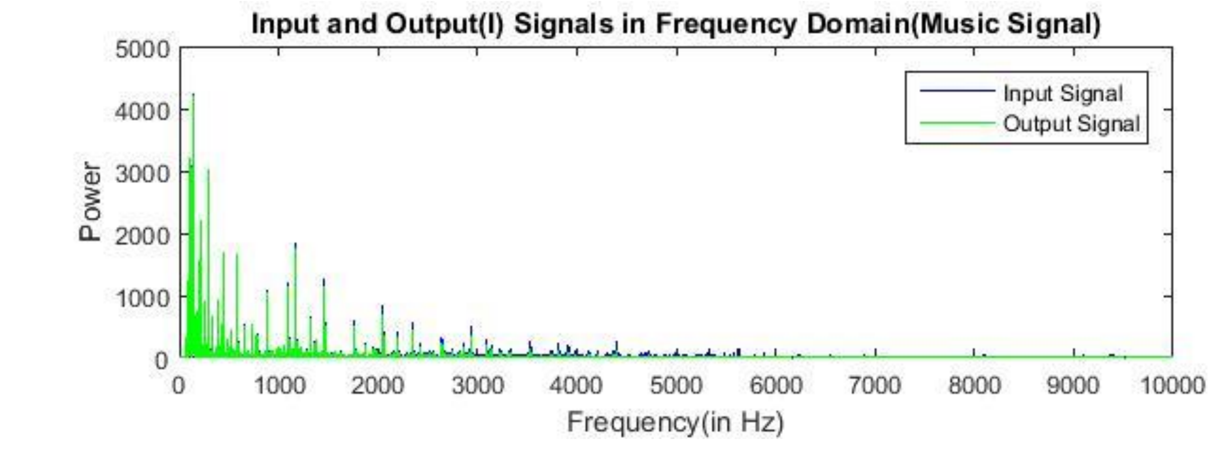
On passing a music audio *‘Guitar.wav’* through filter h1, we observed:



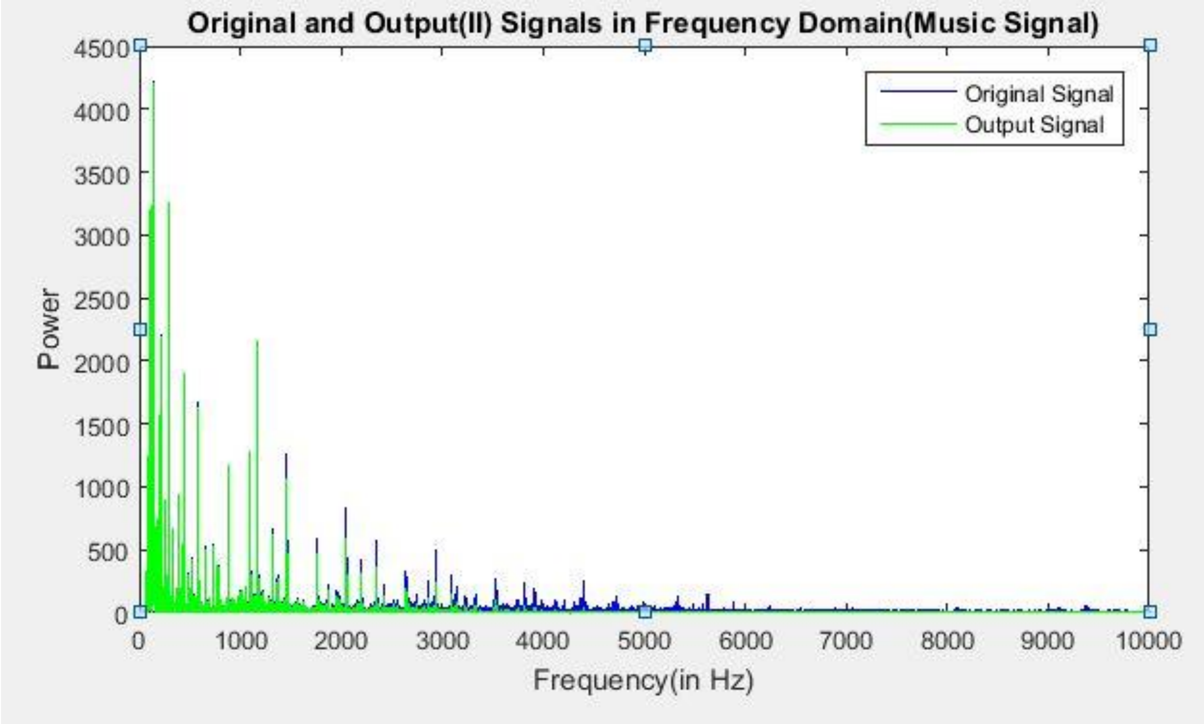
After passing through filter once



After passing through filter twice



After passing through filter once



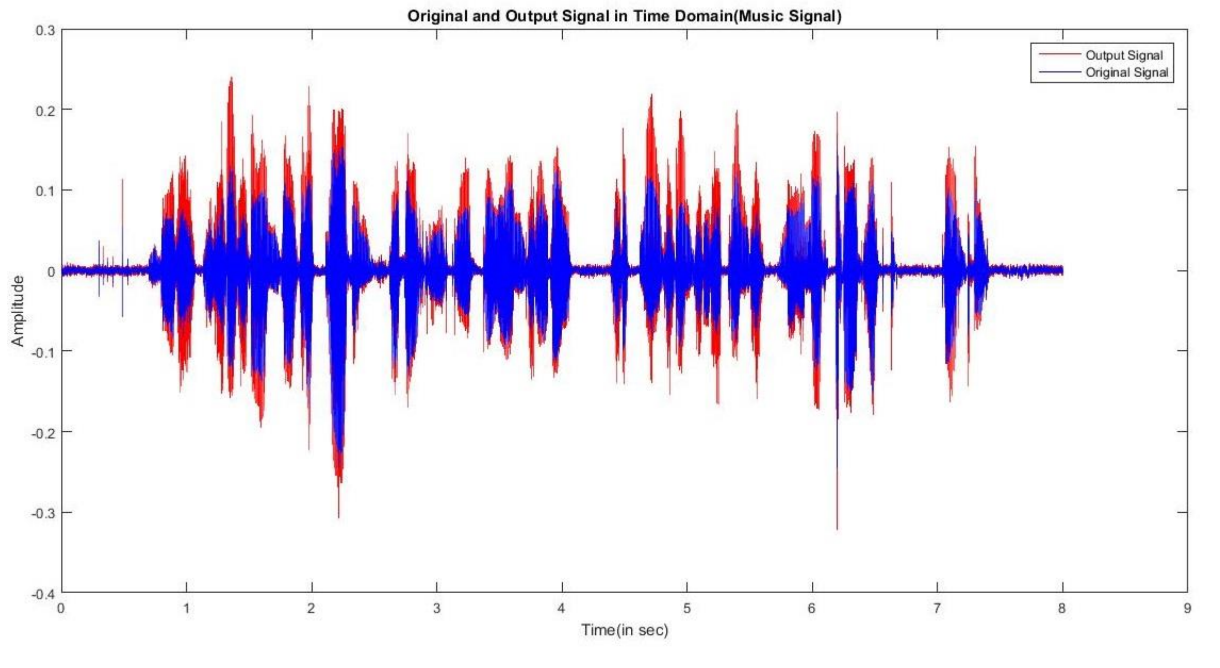
After passing through filter twice

It can be clearly observed that the higher frequencies are being cut off by the filter, hence h1 would be a *low pass filter*.

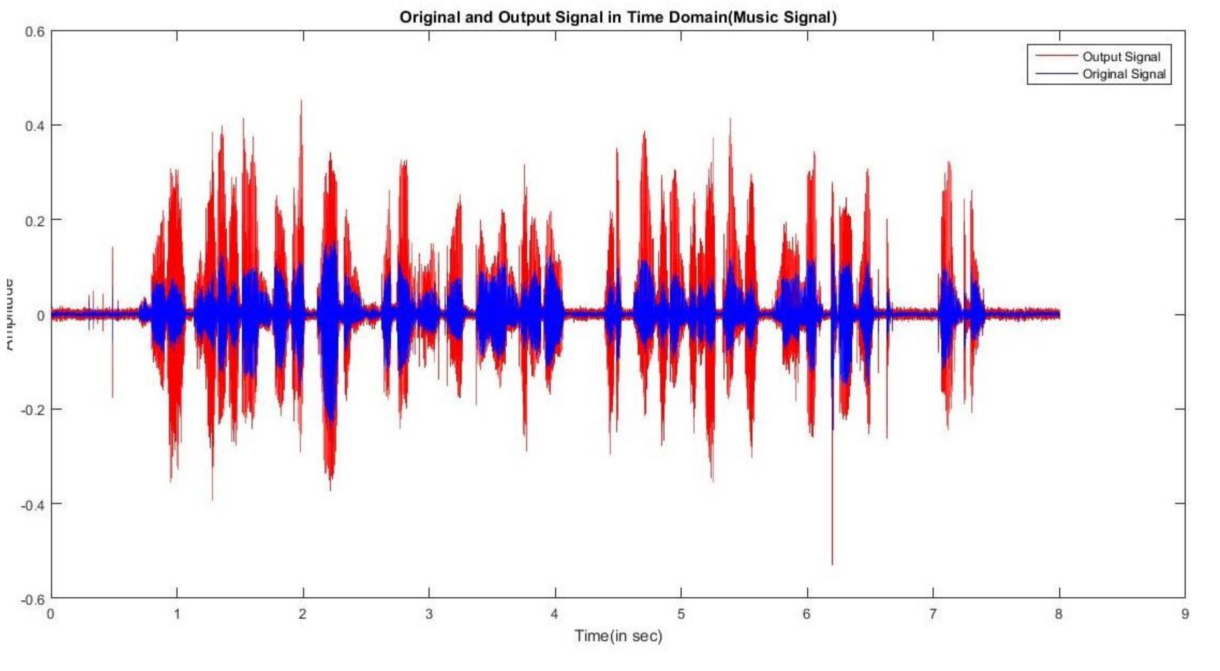
From the plots, the cut-off frequency seems to be around 4000 Hz.

Analysis with filter h2:

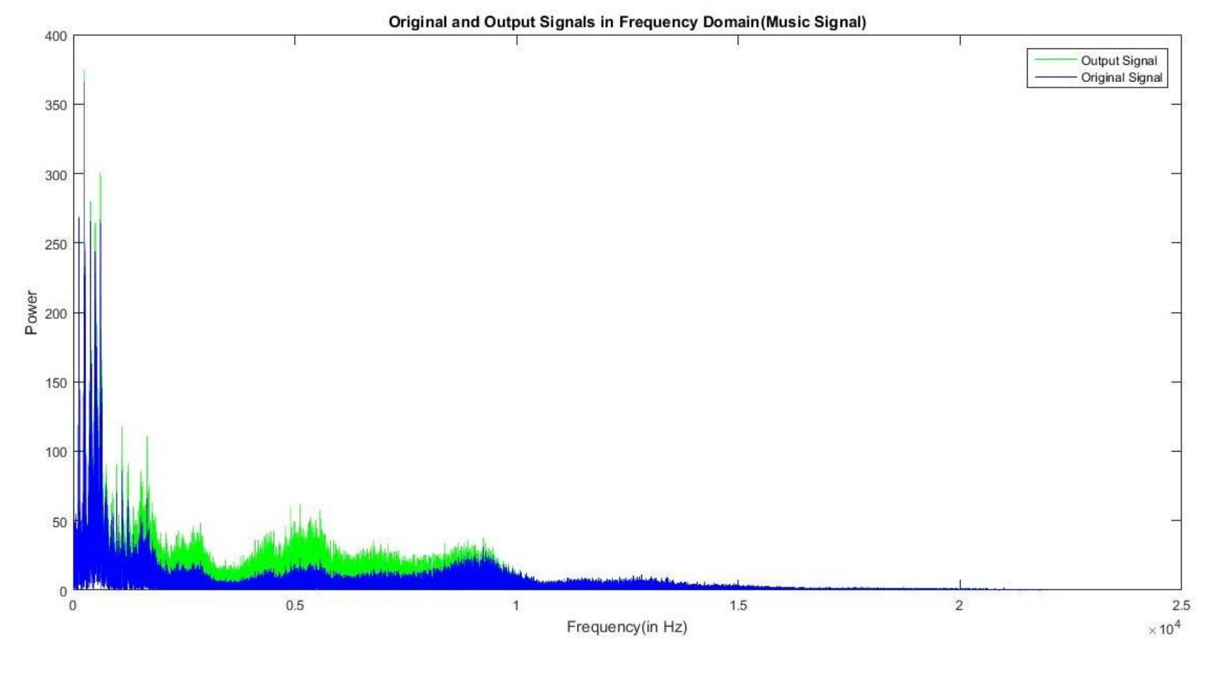
On passing the voice signal through h2, we obtained these plots.



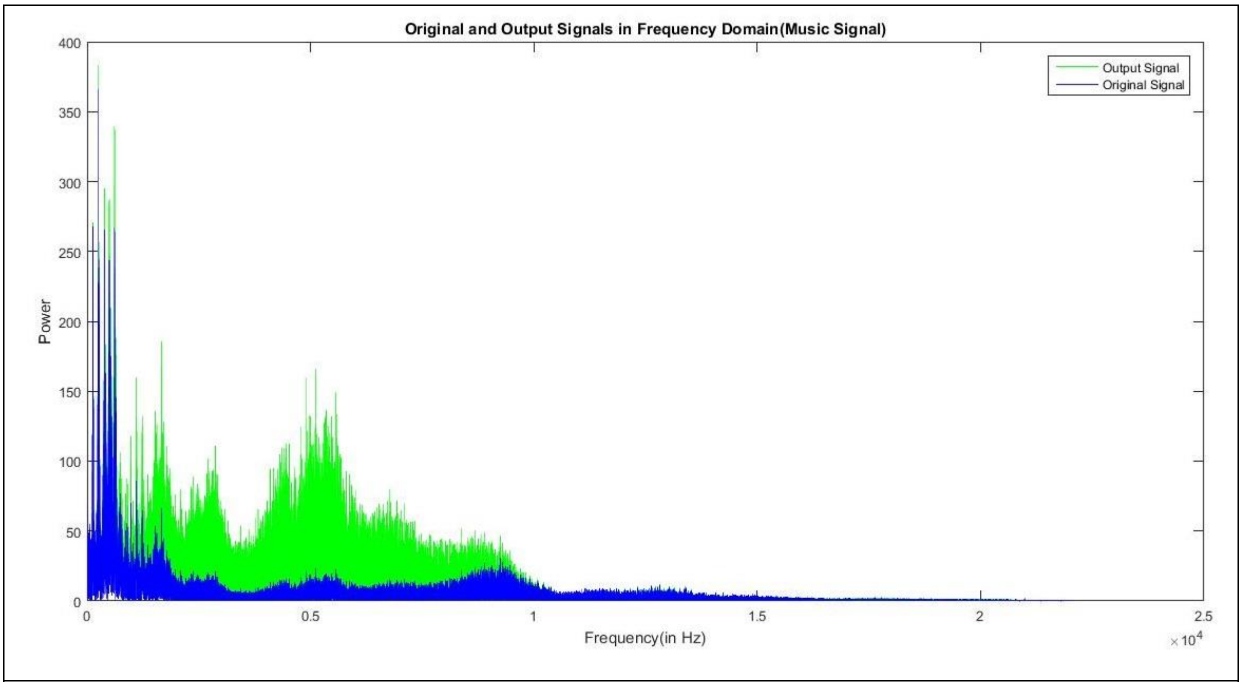
After passing through filter once



After passing through filter twice

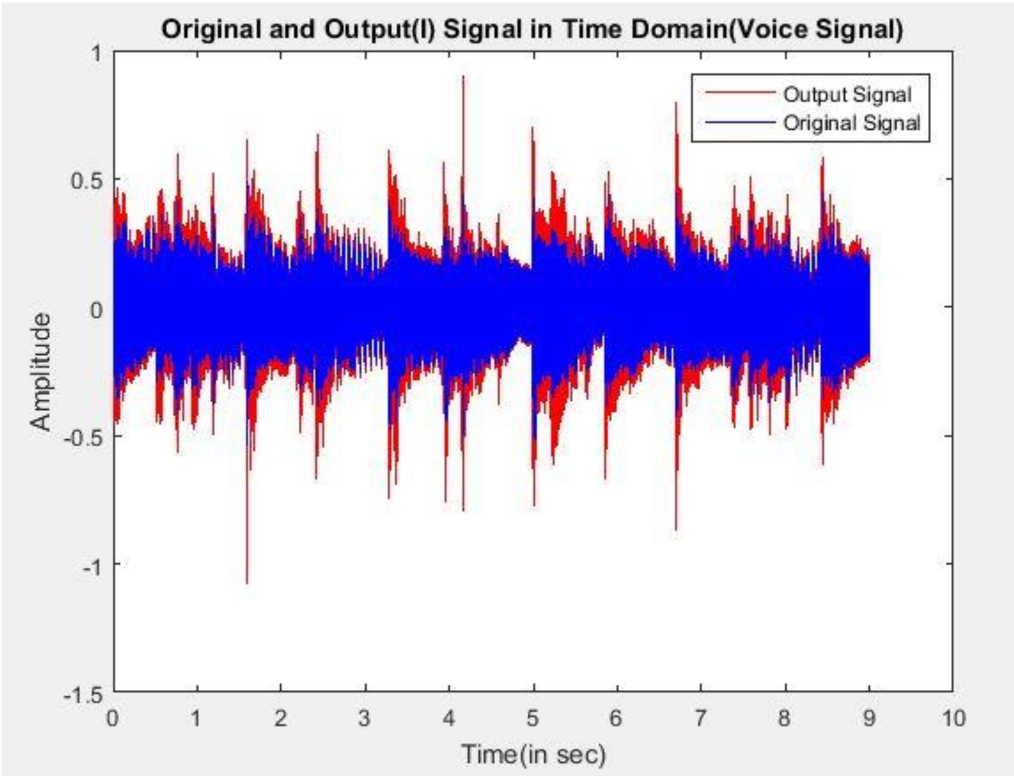


After passing through filter once

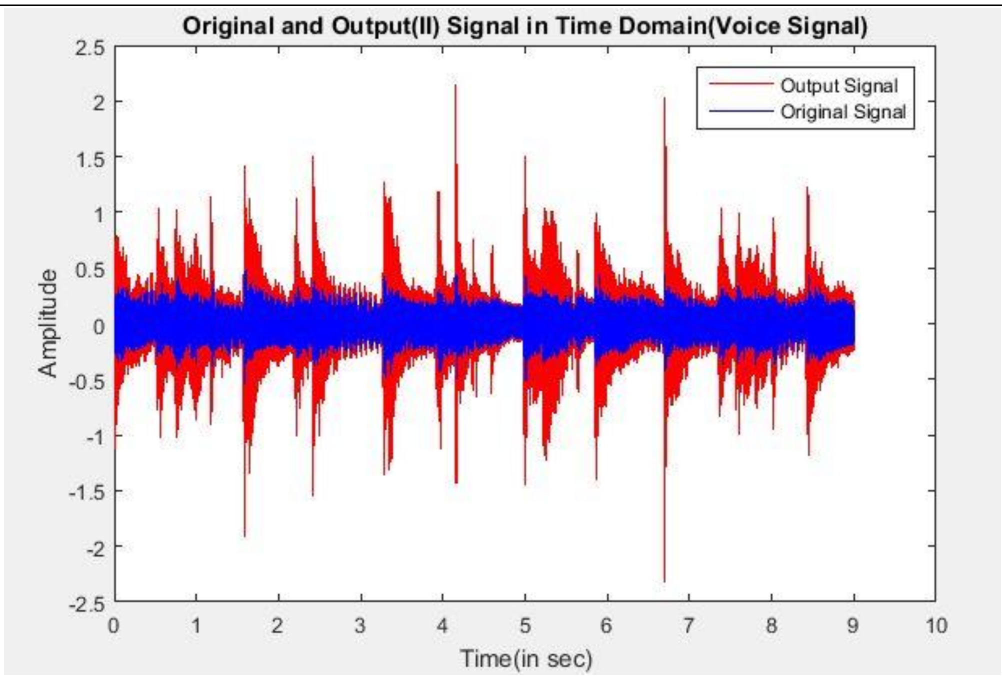


After passing through filter twice

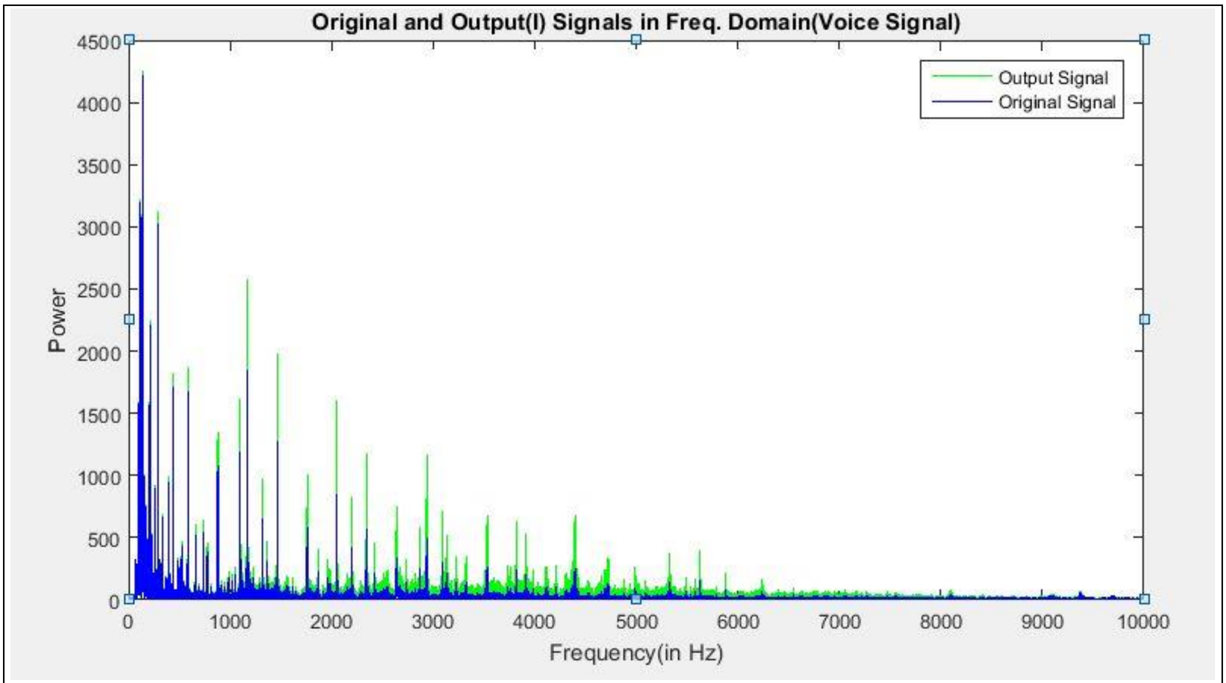
Doing the analysis with the music signal, we get:



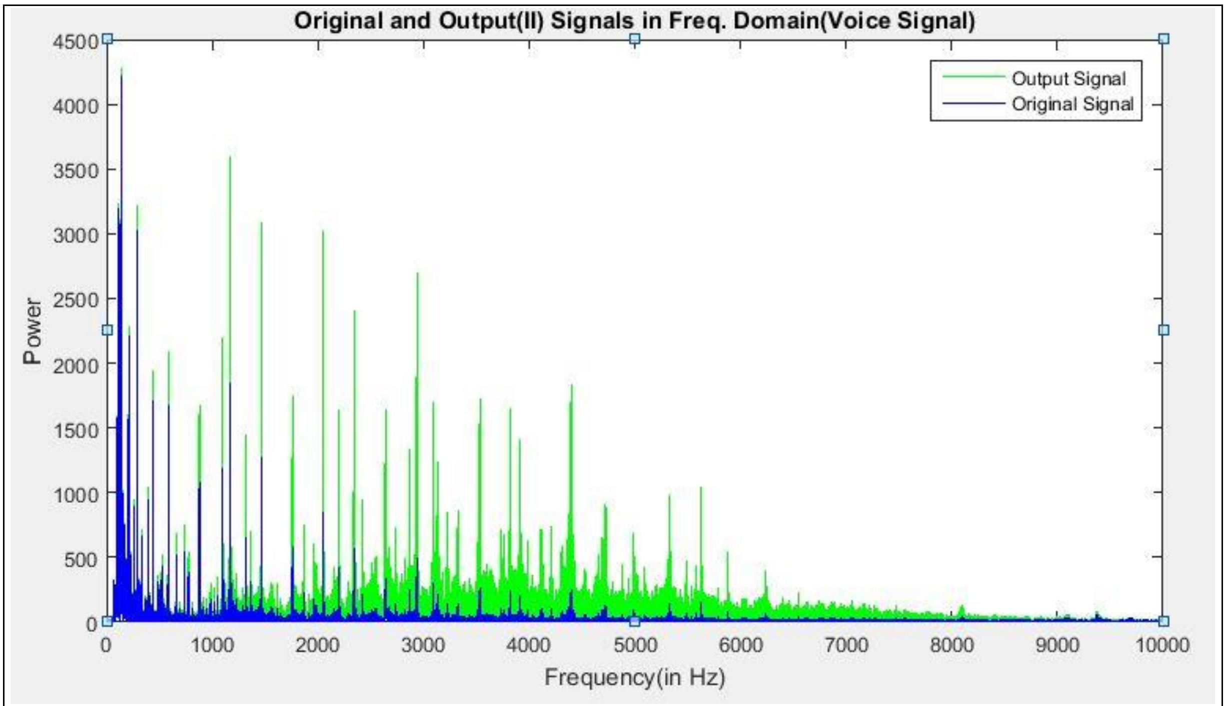
Passing through filter once



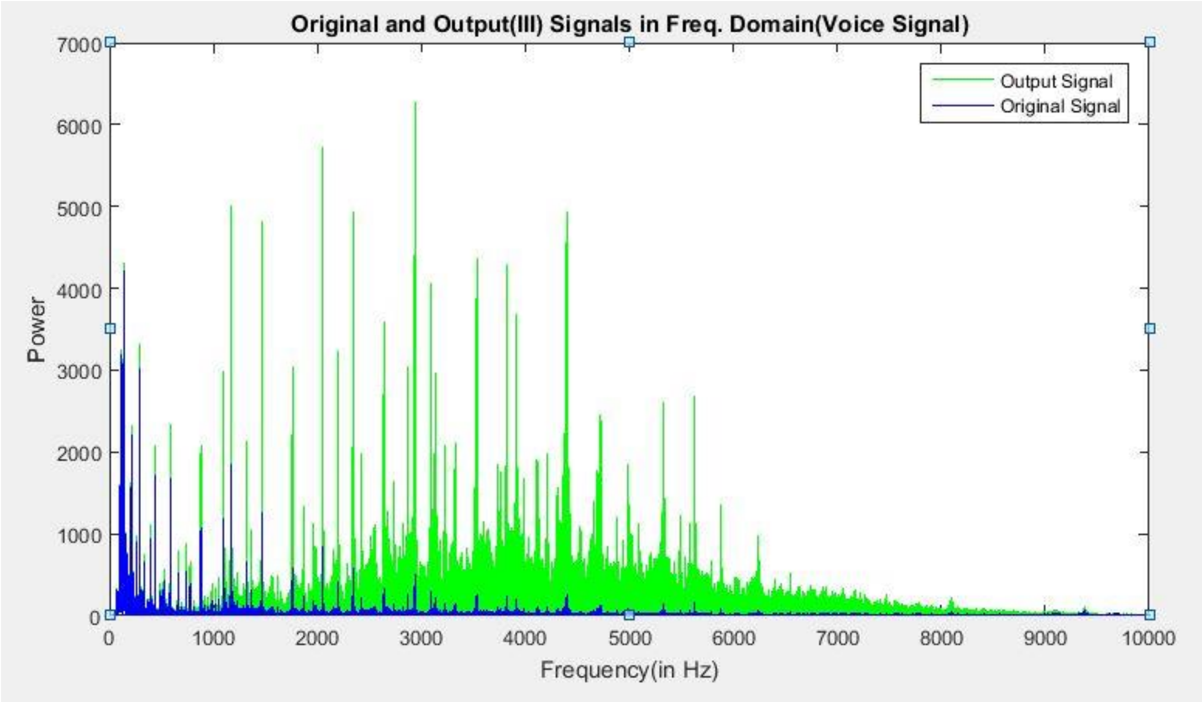
Passing through filter twice



Passing through filter once



Passing through filter twice



After passing through filter thrice

The signals are getting amplified and this is pronounced for frequencies in an approximate range of 1000 Hz to 8000 Hz.

Hence h2 must be a *bandpass filter*.

RESULTS

From the analysis of the output signal using the time domain plots and the frequency domain plots after Fourier Transform, we can conclude that the amplitude of the signal is reduced and higher frequencies cut off in case of the filter [1 1 1 1 1 1 1]/7. It is a low pass filter.

For the filter [-1 -1 -1 2 1 1 1]/2 the amplitude has increased considerably for a particular range of frequencies. It is likely to be a band pass filter.