DSP LAB - Experiment 7

Audio signal processing using HPF

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EE22BTECH11003

April 2, 2024

Aim

To investigate the effects of applying both low-pass and high-pass filters on audio signals. Through this experiment, the aim is to explore how these filters alter the frequency content and perception of sound in order to gain insights into their practical applications in audio processing and signal manipulation.

Theory

Low Pass Filter

A low-pass filter is a filter that passes signals with a frequency lower than a selected cutoff frequency and attenuates signals with frequencies higher than the cutoff frequency. The exact frequency response of the filter depends on the filter design.

High Pass Filter

A high-pass filter is a filter that passes signals with a frequency higher than a certain cutoff frequency and attenuates signals with frequencies lower than the cutoff frequency. The exact frequency response of the filter depends on the filter design.

Design

The LPF and HPF are designed using the following specifications:

- LPF: Cutoff frequency = 10 kHz
- HPF: Cutoff frequency = 10 kHz

MATLAB Code

The function LPF for the low-pass filter is given below:

```
function [h_n] = LPF(fc,fs,N)
    n = (-N + 1)/2 : (N - 1)/2;
    omega_c = 2 * pi * fc / fs;
    hd_n = sin(omega_c * n) ./ (pi * n);
    hd_n(N - (N - 1)/2) = omega_c / pi;
    w_n = zeros(1, N);
    for i = 1:N
        if i >=1 && i <= N
        w_n(i) = 0.54 - 0.46 * cos(2 * pi * (i - 1) / (N - 1));
        end
    end
    h_n = hd_n .* w_n;
    end
```

The function HPF for the high-pass filter is given below:

```
function [h_n] = HPF(fc,fs,N)
n = (-N + 1)/2 : (N - 1)/2;
omega_c = 2* pi * fc / fs;
hd_n = (sin(pi*n)- sin(omega_c * n)) ./ (pi * n);
hd_n(N - (N - 1)/2) =1- (omega_c / pi);
w_n = zeros(1, N);
for i = 1:N
    if i >=1 && i <= N
        w_n(i) = 0.54 - 0.46 * cos(2 * pi * (i - 1) / (N - 1));
    end
end
h_n = hd_n .* w_n;
end
```

The script to generate the plots is given below:

```
% Read the audio file
[d,r] = audioread('msmn1.wav');

% plot the spectrogram of the original signal
figure(1);
specgram(d,1024,r);
title('Spectrogram of the original signal');

N = 21;
fc = 1000;

% plot the impulse response of the low pass filter
figure(2);
stem(0:N-1,LPF(fc,r,N));
xlabel('n');
ylabel('h(n)');
```

```matlab
title('Impulse response of the low pass filter')

% plot the impulse response of the high pass filter
figure(3);
stem(0:N-1,HPF(fc,r,N));
xlabel('n');
ylabel('h(n)');
title('Impulse response of the high pass filter')

% plot the magnitude response of the low pass filter
% figure(4);
fvtool(LPF(fc,r,N));

% plot the magnitude response of the high pass filter
% figure(5);
fvtool(HPF(fc,r,N));

% apply the low pass filter to the signal
LPF_output = conv(d,LPF(fc,r,N));

% plot the spectrogram of the output signal
figure(6);
specgram(LPF_output,1024,r);
title('Spectrogram After applying the low pass filter');

% apply the high pass filter to the signal
HPF_output = conv(d,HPF(fc,r,N));

% plot the spectrogram of the output signal
figure(7);
specgram(HPF_output,1024,r);
title('Spectrogram After applying the high pass filter');
```

Outputs

The specgram plot of the original input wave is given in Figure 1.

Low Pass Filter (LPF)

The impulse response of the LPF is given in Figure 2.

The magnitude response of the LPF is given in Figure 3.

The specgram plot of the output wave after passing through the LPF is given in Figure 4.

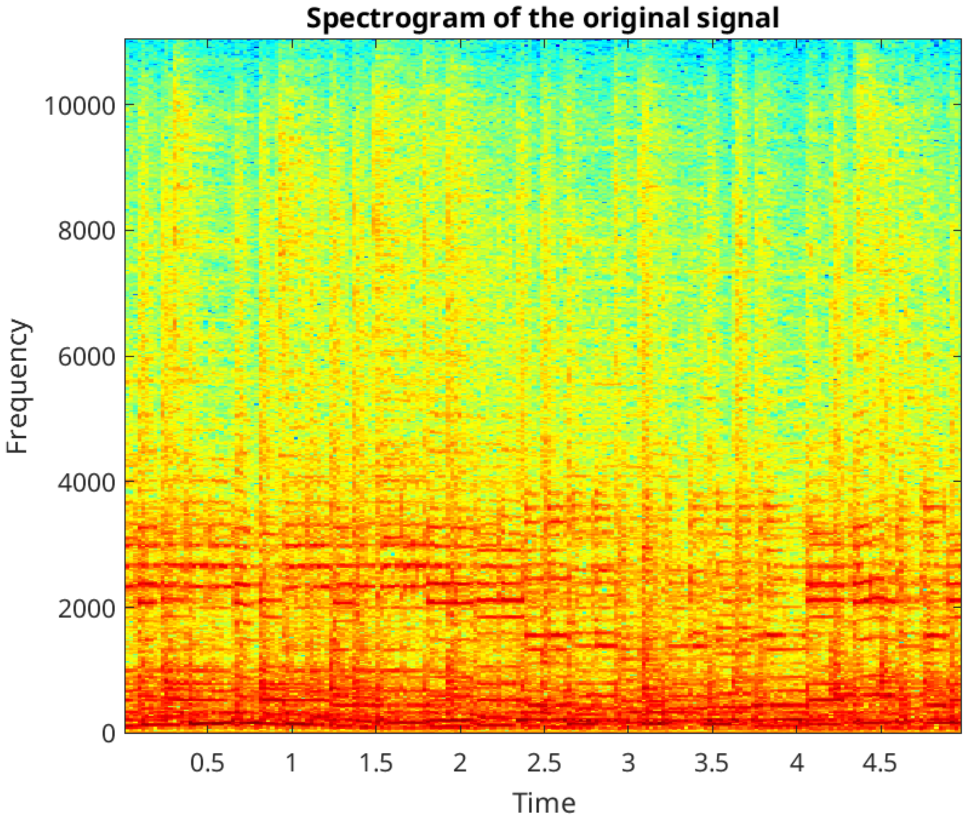

Figure 1: Specgram of the original wave

High Pass Filter (HPF)

The impulse response of the HPF is given in Figure 5.

The magnitude response of the HPF is given in Figure 6.

The specgram plot of the output wave after passing through the HPF is given in Figure 7.

Observations

- HPF can be obtained from an LPF by shifting the frequency response by π.
- The LPF attenuates the high frequency components of the audio signal, resulting in a smoother and less sharp sound.
- The HPF attenuates the low frequency components of the audio signal, resulting in a sharper and more pronounced sound.
- The LPF output has a more muffled and less distinct sound compared to the original signal, while the HPF output has a more crisp and clear sound.
- As the order of the filter increases, the sharpness of the filter increases and the specgram becomes more clear.

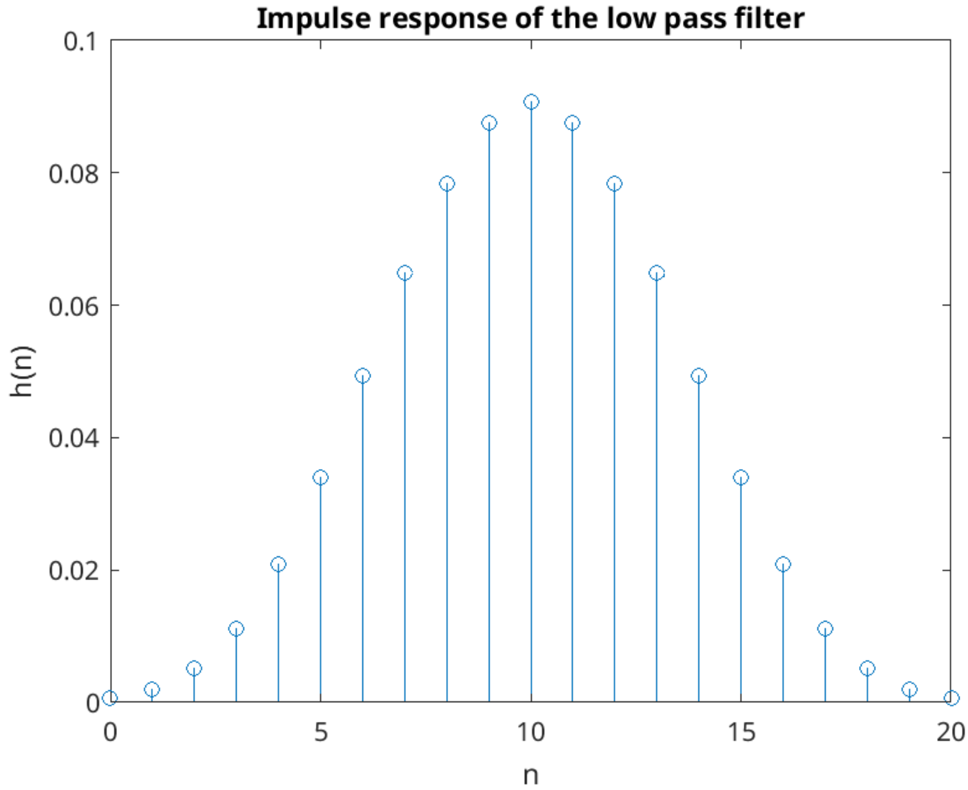

Figure 2: Impulse response of the LPF

- The original wave has a lot of low frequency components, which are attenuated by the HPF, resulting in a more pronounced high frequency sound.

Conclusion

The experiment demonstrates the effects of applying low-pass and high-pass filters on audio signals. The LPF attenuates high frequency components, resulting in a smoother sound, while the HPF attenuates low frequency components, resulting in a sharper sound. The LPF output is more muffled and less distinct compared to the original signal, while the HPF output is more crisp and clear. The experiment provides insights into the practical applications of LPF and HPF in audio processing and signal manipulation. It also shows that HPF can be obtained from an LPF by shifting the frequency response by π.

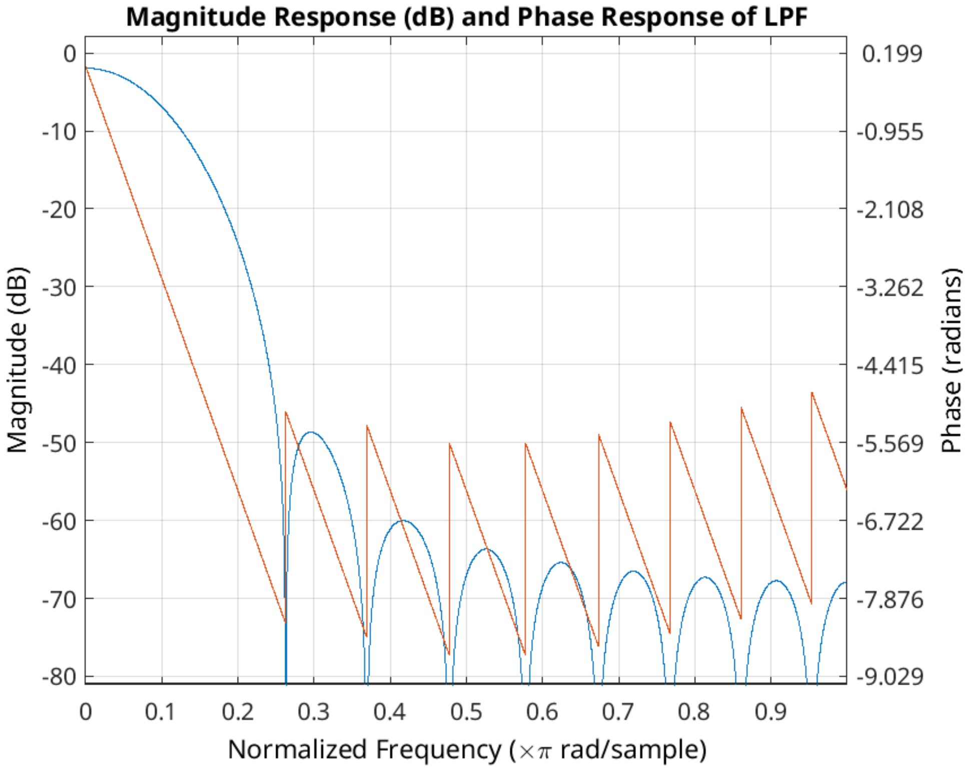

Figure 3: Magnitude response of the LPF

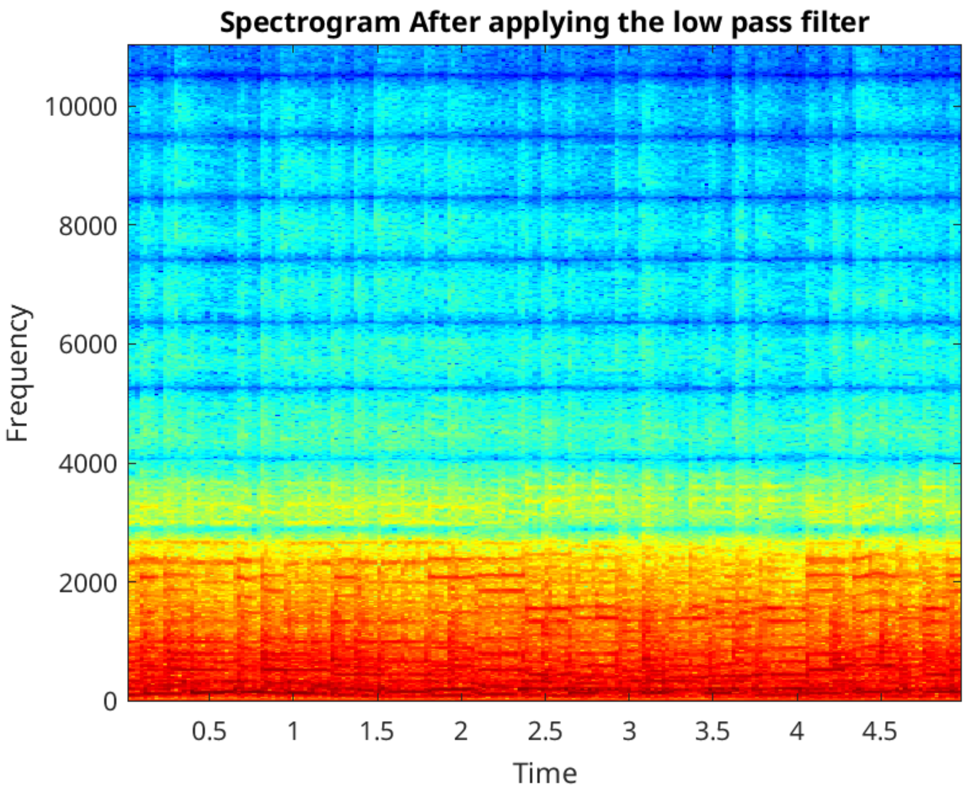

Figure 4: Specgram of the LPF output

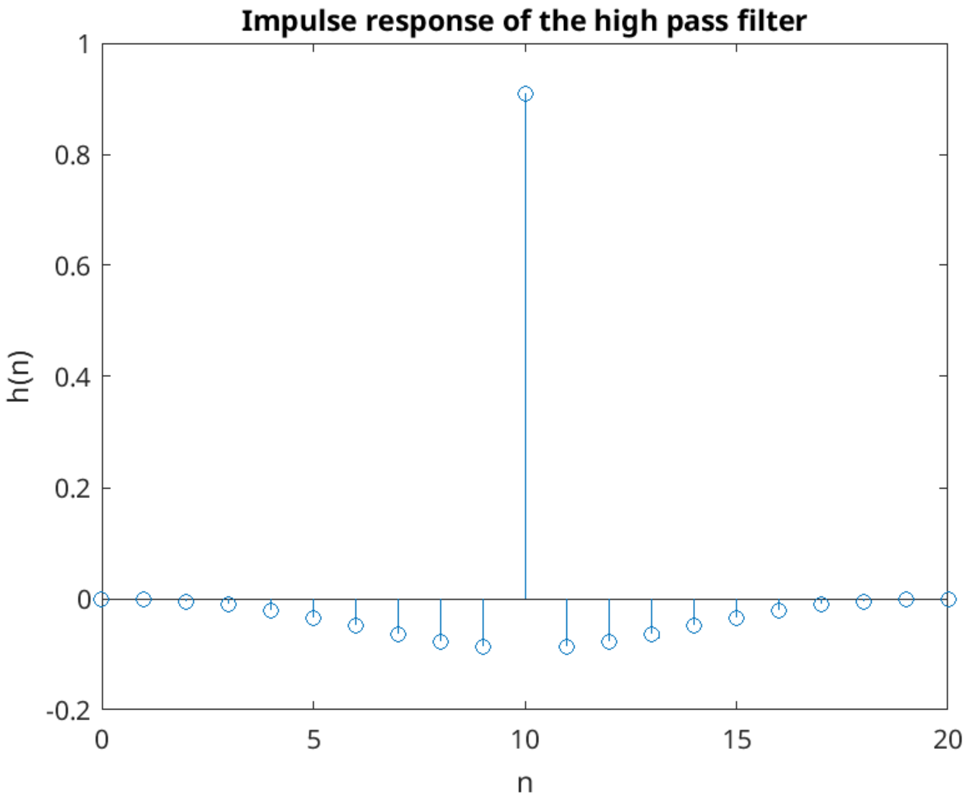

Figure 5: Impulse response of the HPF

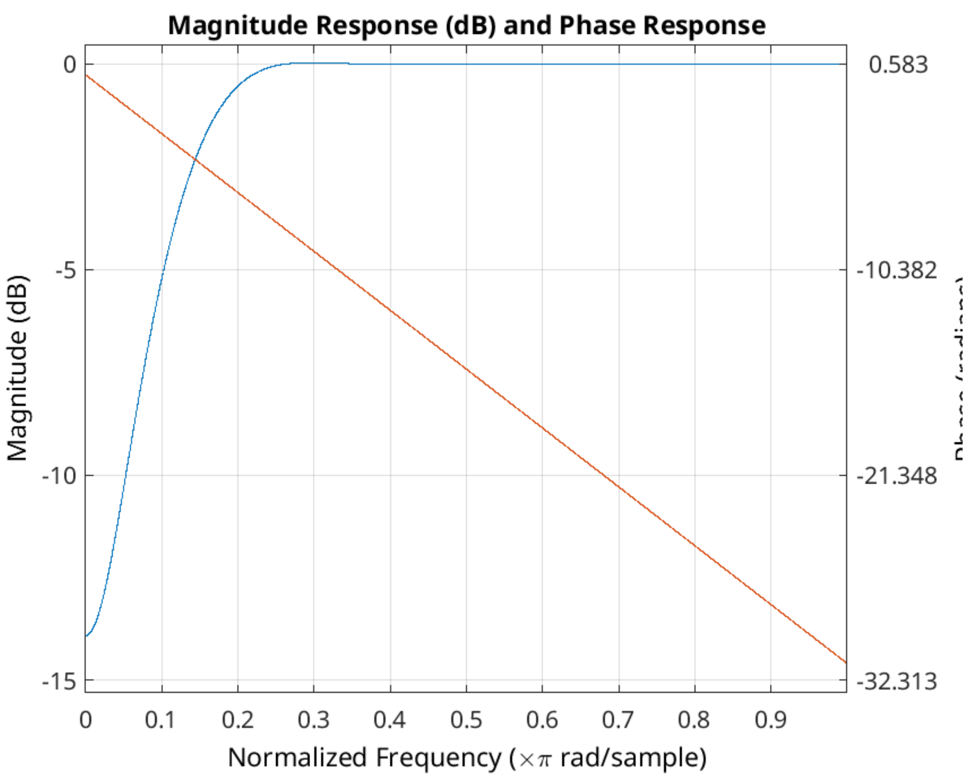

Figure 6: Magnitude response of the HPF

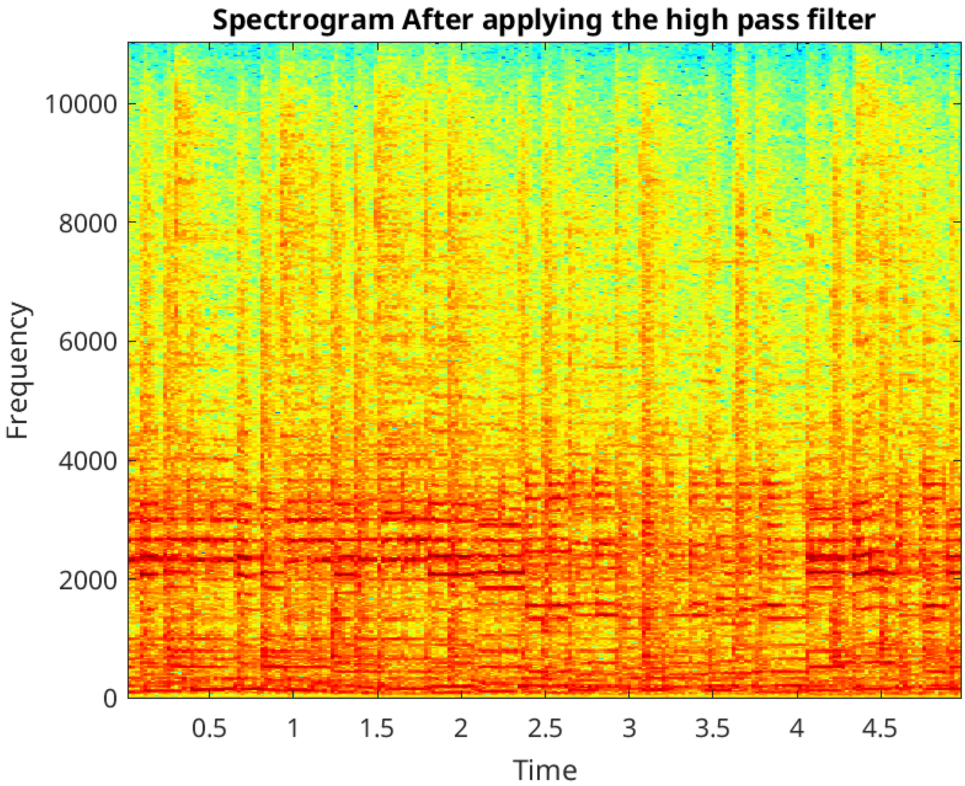

Figure 7: Specgram of the HPF output