Experiment-7

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I. Aim of the experiment

To Analyse the impulse response and magnitude/Phase response of High Pass Filter(HPF) and Low Pass Filter(LPF)

II. Theory

A. Low Pass Filter (LPF)

A Low Pass Filter is a type of filter that allows signals with frequencies lower than a certain cutoff frequency to pass through while attenuating frequencies higher than the cutoff frequency. The transfer function of an ideal LPF is given by:

$$H_{\text{LPF}}(j\omega) = \begin{cases} 1 & \text{for } \omega < \omega_c \\ 0 & \text{for } \omega \geq \omega_c \end{cases}$$

Where ω_c is the cutoff frequency. Similarly, the impulse response of an LPF can be obtained by taking the inverse Fourier transform of its transfer function.

B. High Pass Filter (HPF)

A High Pass Filter is a type of filter that allows signals with frequencies higher than a certain cutoff frequency to pass through while attenuating frequencies lower than the cutoff frequency. The transfer function of an ideal HPF is given by:

$$H_{\text{HPF}}(j\omega) = \begin{cases} 0 & \text{for } \omega < \omega_c \\ 1 & \text{for } \omega \geq \omega_c \end{cases}$$

Where ω_c is the cutoff frequency. The impulse response of an HPF can be obtained by taking the inverse Fourier transform of its transfer function.

C. Magnitude and Phase Response

The magnitude response of a filter describes how the filter attenuates or amplifies different frequencies. It is given by the absolute value of the transfer function:

$$|H(j\omega)| = \sqrt{H(j\omega) \cdot H^*(j\omega)}$$

Where $H^*(j\omega)$ is the complex conjugate of $H(j\omega)$.
The phase response of a filter describes the phase shift introduced by the filter at different frequencies. It is given by the argument of the transfer function:

$$\angle H(j\omega) = \arg(H(j\omega))$$

Both magnitude and phase responses are crucial in understanding how a filter affects the input signal.

D. Relation between LPF and HPF

$$H_{\mathrm{HPF}}(j\omega) = 1 - H_{\mathrm{LPF}}(j\omega)$$

or,

$$h_{\mathrm{HPF}}(t) = \delta(t) - h_{\mathrm{LPF}}(t)$$

III. MATLAB Simulation

A. Code , Output and Plot of the simulations

```
function main()
N = 41;
wc = 2*pi/11;
fc = 1000;
fs = (2*pi*fc)/(wc);

hpf = hpf_func(fc,fs,N).*hw(N);
fvtool(hpf);

lpf = lpf_func(fc,fs,N).*hw(N);
%fvtool(lpf);
[d,r]= audioread('msmn1.wav');

df_hpf = convl(d,hpf);
%df_lpf = convl(d,lpf);

%specgram(d,1024,r);
specgram(df_hpf,1024,r);
%specgram(df_lpf,1024,r);

%Plot impulse response of LPF and HPF
% subplot(2,1,1);
% stem(lpf);
% xlabel('Time (samples)');
% ylabel('Amplitude');
% title('Impulse Response of Low-pass Filter');
% grid on;
% subplot(2,1,2);
% stem(hpf);
% xlabel('Time (samples)');
% ylabel('Amplitude');
% title('Impulse Response of High-pass Filter');
% grid on;

end

function y = lpf_func(fc,fs,N)
wc = (2*pi*fc)/(fs);
n = 1;
```

```
y = zeros(1,N−1);
for k = −(N−1)/2:(N−1)/2
    if k == 0
        y(n) = wc/pi;
    else
        y(n) = sin((wc∗k))/(pi∗k);
    end
    n = n + 1;
end
end
function y = hpf_func(fc, fs, N)
    wc = (2∗pi∗fc)/(fs);
    n = 1;
    y = zeros(1, N);
    for k = −(N−1)/2:(N−1)/2
        if k == 0
            y(n) = 1 − (wc)/pi;
        else
            y(n) = sin(pi∗k)/(pi∗k)−sin(wc∗k)/(pi∗k); % Adjustment for high−pass filter
        end
        n = n + 1;
    end
end
function y = hw(N) %window function
n = 1;
y = zeros(1,N);
for k = 0 : (N−1)
    if k >= 0 && k <= (N−1)
        y(n) = 0.54 − 0.46∗cos((2∗pi∗k)/(N−1));
    else
        y(n) = 0;
    end
    n = n + 1;
end
end
function y = convl(x,h)
  l = length(x) + length(h) − 1;
  y = zeros(1, l);
  for n = 1:l
    for k = 1:length(x)
      if (n − k + 1) >= 1 && (n − k + 1) <= length(h)
        y(n) = y(n) + x(k) ∗ h(n − k + 1); % This is just /sigma x(k)∗h(n−k)
      end
    end
  end
end
```

IV. LPF

A. Spectrogram and Magnitude/Phase Response

The following plots were generated using Matlab.

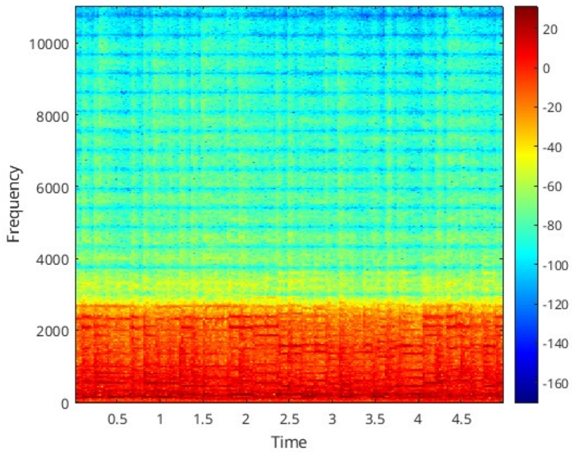

Fig. 0. Low pass filtered spectogram

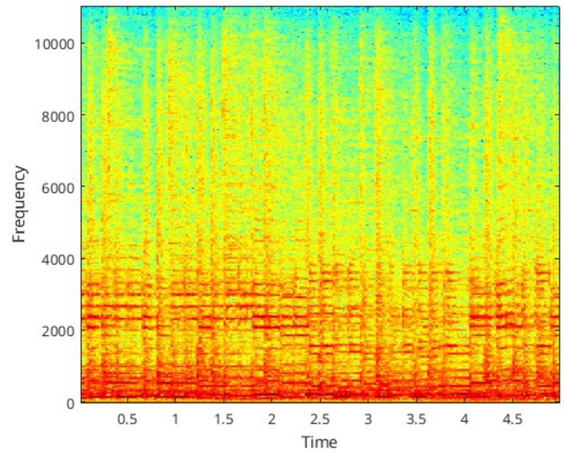

Fig. 0. Orginal spectogram

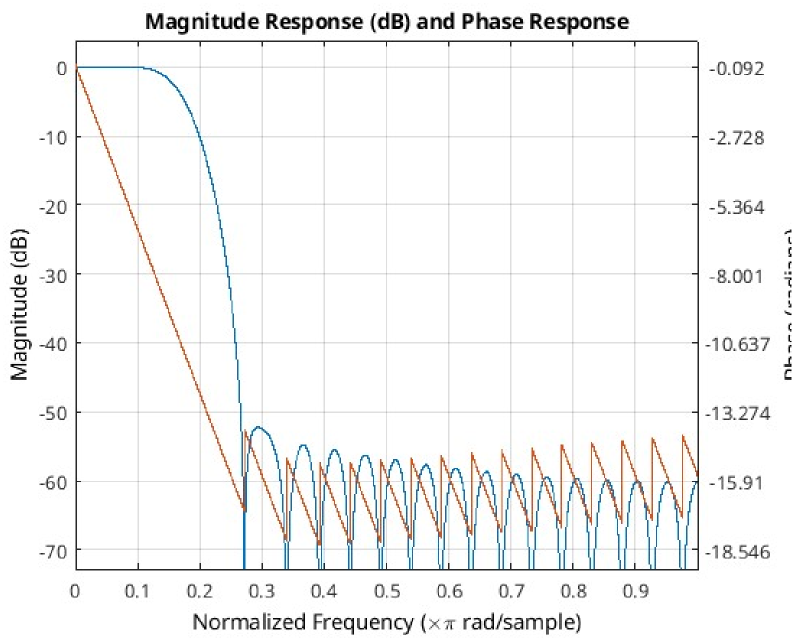

Fig. 0. Magnitude/Phase response for LPF

V. HPF

A. *Spectrogram and Magnitude/Phase Response*

The following plots were generated using Matlab.

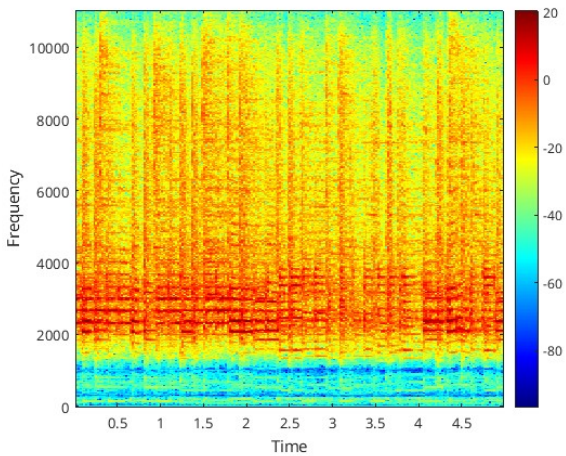

Fig. 0. High pass filtered spectogram

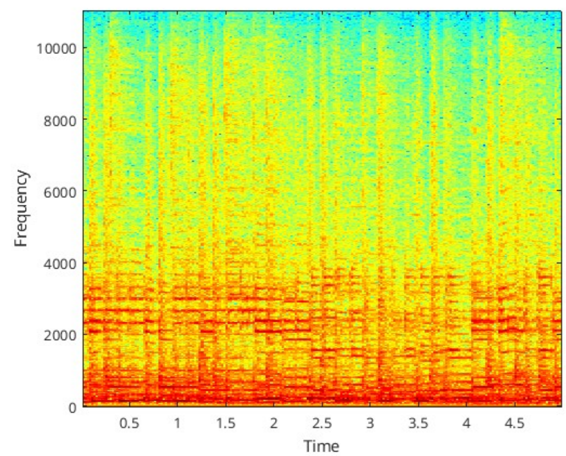

Fig. 0. Orginal spectogram

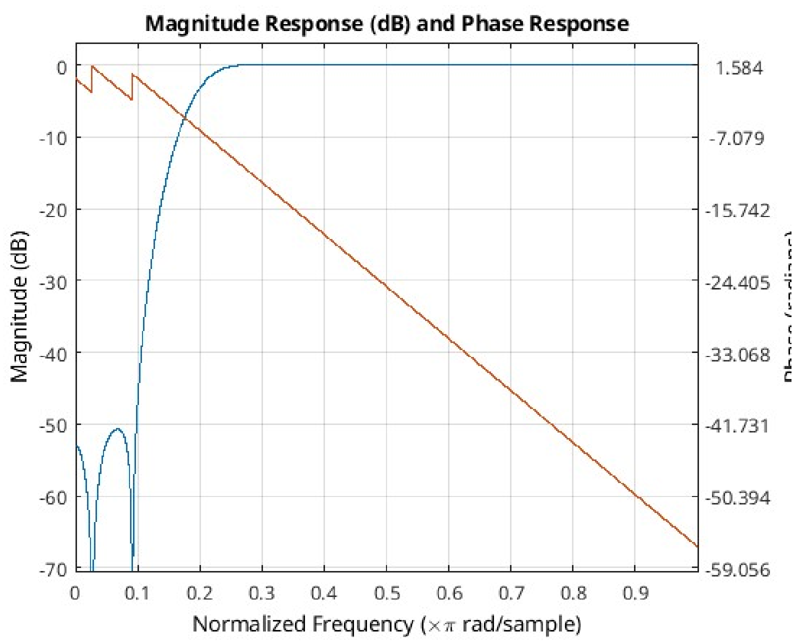

Fig. 0. Magnitude/Phase response for LPF

B. Impulse response of LPF and HPF

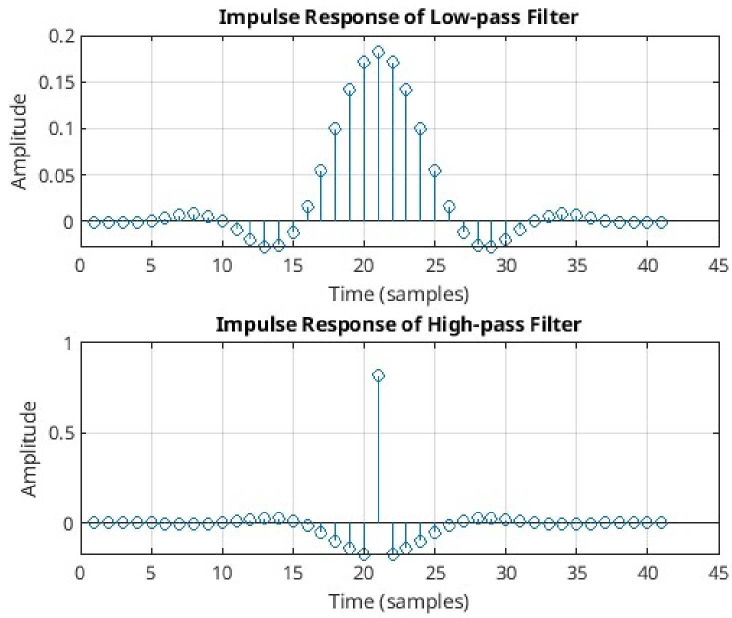

Fig. 0. Plotted in Time domain

VI. Observations

1) Comparing the original spectrogram with the filtered ones clearly illustrates the energy(frequency) content changes by change in color of the spectrum.
2) The HPF spectrogram sharpens the signal by reducing low frequencies(i.e spectrum at that part becomes dull seen as bluish) and emphasizing high ones.
3) The LPF spectrogram smoothens it by reducing high frequencies and enhancing low ones(i.e spectrum at that part becomes bright seen as deep red).
4) In the Impulse reponse, Low-pass filter (LPF), when converted into a High-pass filter (HPF), undergoes distinct changes in characteristics
5) Initially gradual in rise and fall, typical of LPF, the converted HPF impulse response exhibits a sharper onset and decay, reflecting its emphasis on higher frequencies.
6) Through this experiment, it becomes evident that HPF and LPF serve distinct purposes in signal processing, with HPF accentuating high-frequency details and LPF emphasizing low-frequency elements. The visual representations through spectrograms and frequency or, impulse responses offer comprehensive insights into their effects on the audio signal.

END OF REPORT