Signal and Systems

Experiment No. # 13

Objective:

The objective of this lab is to learn how to implement the basic types of filters in MATLAB and to apply them on an audio signal.

Theoretical Background:

Filters:

Filters, in general, are used to block some part of a signal and to pass some particular part of a signal. In frequency domain, we say that a filter will block a certain range of frequencies, while passing a specific range of frequencies. The three most common types of filters are:

1. Low pass Filters (LPF)

2. Band pass Filters (BPF)

3. High pass Filters (HPF)

1. Low Pass Filter:

A low pass filter, as its name suggests, is used to pass only the low frequency components of a signal, and block all higher frequency components.

2. Band Pass Filter:

A band pass filter is a filter that passes all frequencies of a signal within a certain range, (which neither includes zero frequency nor pi), and block all other frequencies outside that range.

3. High Pass Filter:

A high pass filter is used to pass all frequency components of a signal higher than a cutoff frequency, and stop all other frequency components which are lower than the cutoff frequency.

Tasks:

In order to create an understanding of passing signals through filters, the following tasks are to be performed by the students:

Task 1:

Using the filter design toolbox, (FDA tool), design three filters with the following specifications:

a) A Low Pass Filter with pass band ($f < 1500$ Hz)

b) A Band Pass Filter with pass band ($1500 < f < 3000$ Hz)

c) A High Pass Filter with pass band ($f > 3000$ Hz)

Task 2:

• Using audiorecorder, record a 5 second audio in MATLAB

• Export the filters into workspace, and save the filters and recorder object in a '.mat' file.

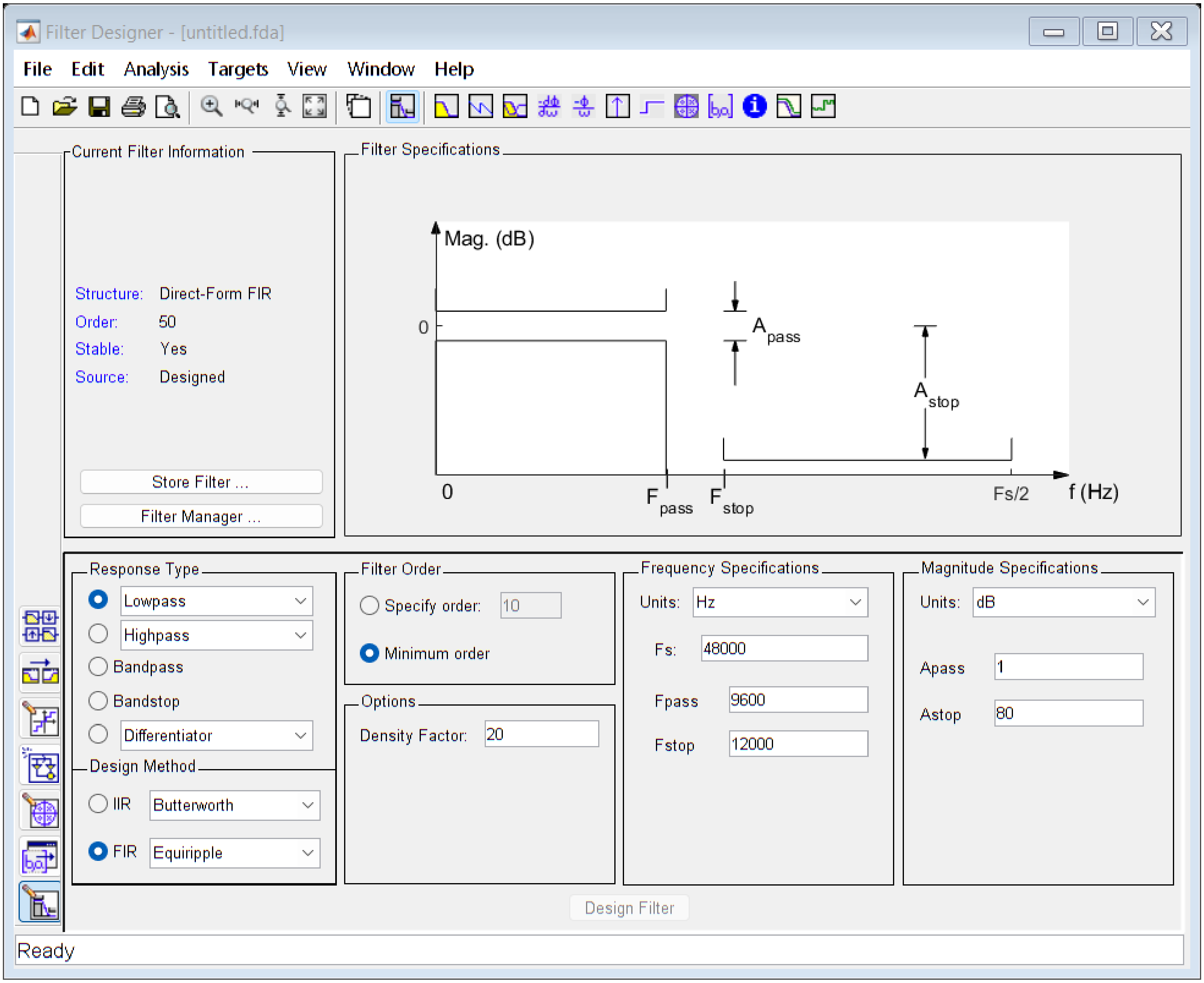
• In a new MATLAB script, load the '.mat' file into workspace. Now pass the signal through each of the three filters separately.

• Reconstruct the original signal by adding the outputs of all three filters

• Using subplot, show the original signal, the outputs of the three filters, and then the reconstructed signal.

Task 1:

```
filterDesigner;
```



Task 2:

```
% Create an audiorecorder object
recObj = audiorecorder;

% Record 5 seconds of audio
```

```
disp('Start speaking.');
```

```
Start speaking.
```

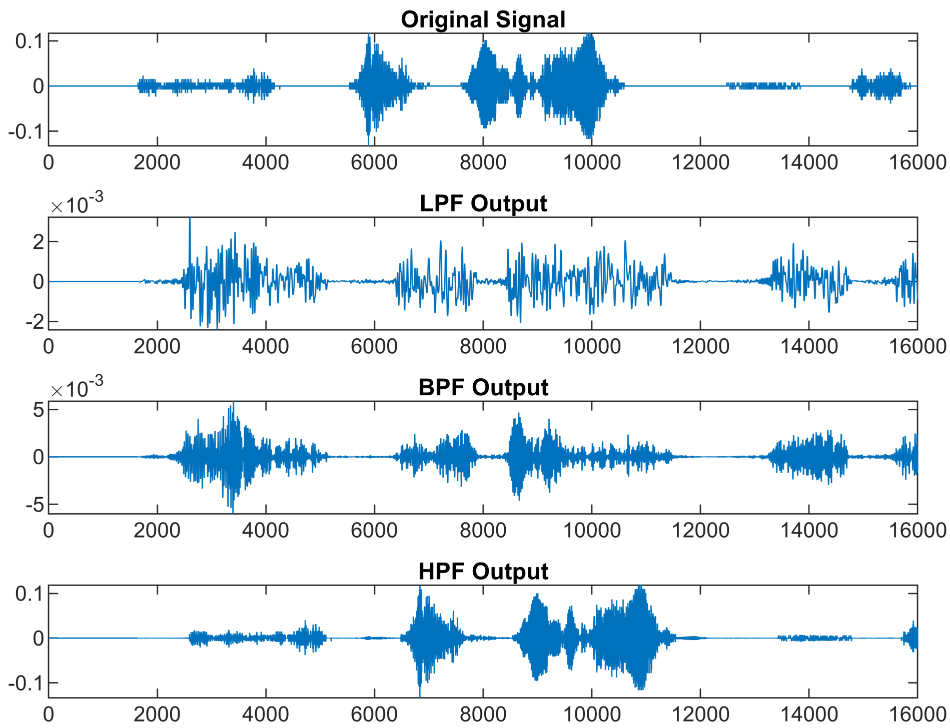
```
recordblocking(recObj, 2);
disp('End of Recording.');
```

```
End of Recording.
```

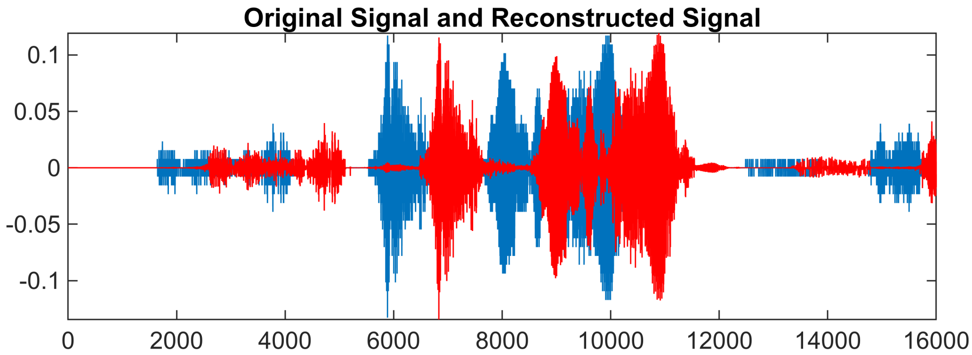
```
% Save the recorded audio to a variable
audioData = getaudiodata(recObj);
```

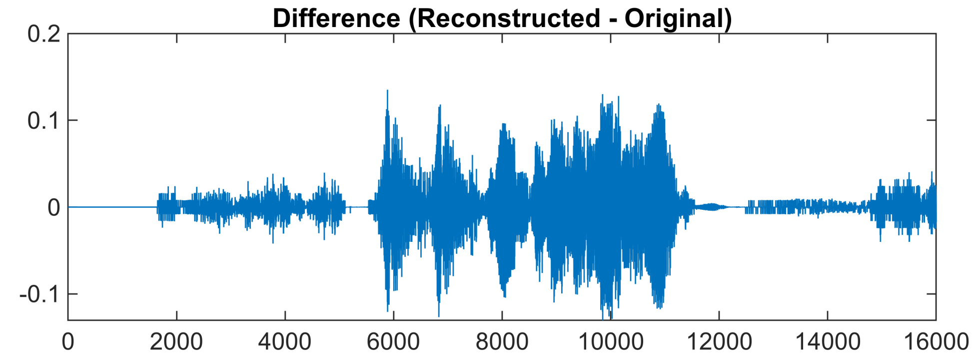
```
% Save the filters and recorder object in a MAT file
save('audio_processing_data.mat', 'LPF', 'BPF', 'HPF', 'recObj');
```

```
% Load the MAT file into the workspace
load('audio_processing_data.mat');

% Pass the signal through each filter separately
outputLPF = filter(LPF, audioData);
outputBPF = filter(BPF, audioData);
outputHPF = filter(HPF, audioData);

% Reconstruct the original signal by adding the outputs of all three filters
reconstructedSignal = outputLPF + outputBPF + outputHPF;

% Plot the signals using subplot
figure;

% Original Signal
subplot(4,1,1);
plot(audioData);
title('Original Signal');

% Filtered Signals
subplot(4,1,2);
plot(outputLPF);
title('LPF Output');

subplot(4,1,3);
plot(outputBPF);
title('BPF Output');

subplot(4,1,4);
plot(outputHPF);
title('HPF Output');
```




```
% Reconstructed Signal
figure;
subplot(2,1,1);
plot(audioData);
hold on;
plot(reconstructedSignal, 'r');
title('Original Signal and Reconstructed Signal');

subplot(2,1,2);
plot(reconstructedSignal - audioData);
title('Difference (Reconstructed - Original)');
```


Original Signal and Reconstructed Signal
0.1
0.05
0
-0.05
-0.1
0
2000
4000
6000
8000
10000
12000
14000
16000


Difference (Reconstructed - Original)
0.2
0.1
0
-0.1
0
2000
4000
6000
8000
10000
12000
14000
16000