Decimation and Interpolation of Audio Signals

Multirate Digital Signal Processing- EE6133

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Experiment 1

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Link to the git repository hosting the code, input and output files: github link

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Part 1: Downsampling without Anti-Alias Filtering

1.1 Read input files, play back and plot Magnitude Response

```
[audio16k, Fs_audio] = audioread('inp/music16khz.wav');
[speech8k, Fs_speech] = audioread('inp/speech8khz.wav');
disp(['Original Audio Signal Sampling Rate: ', num2str(Fs_audio), 'Hz']);
```

```
Original Audio Signal Sampling Rate: 16000Hz
```

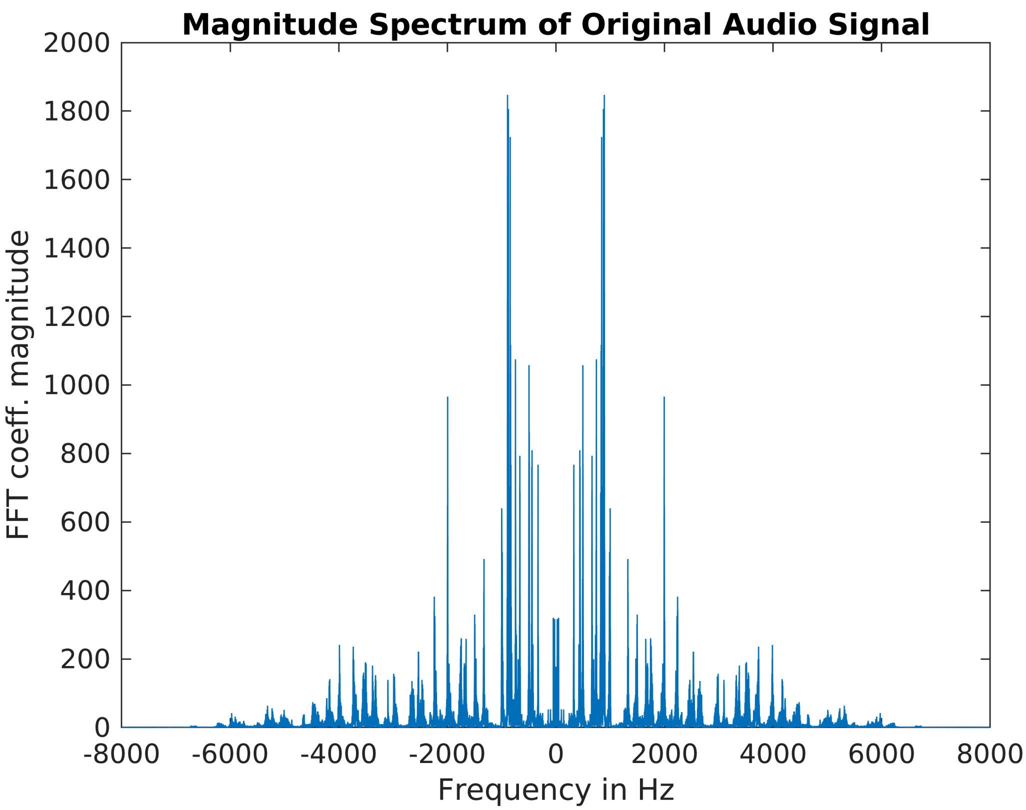
```
disp(['Original Speech Signal Sampling Rate: ', num2str(Fs_speech), 'Hz']);
```

```
Original Speech Signal Sampling Rate: 8000Hz
```

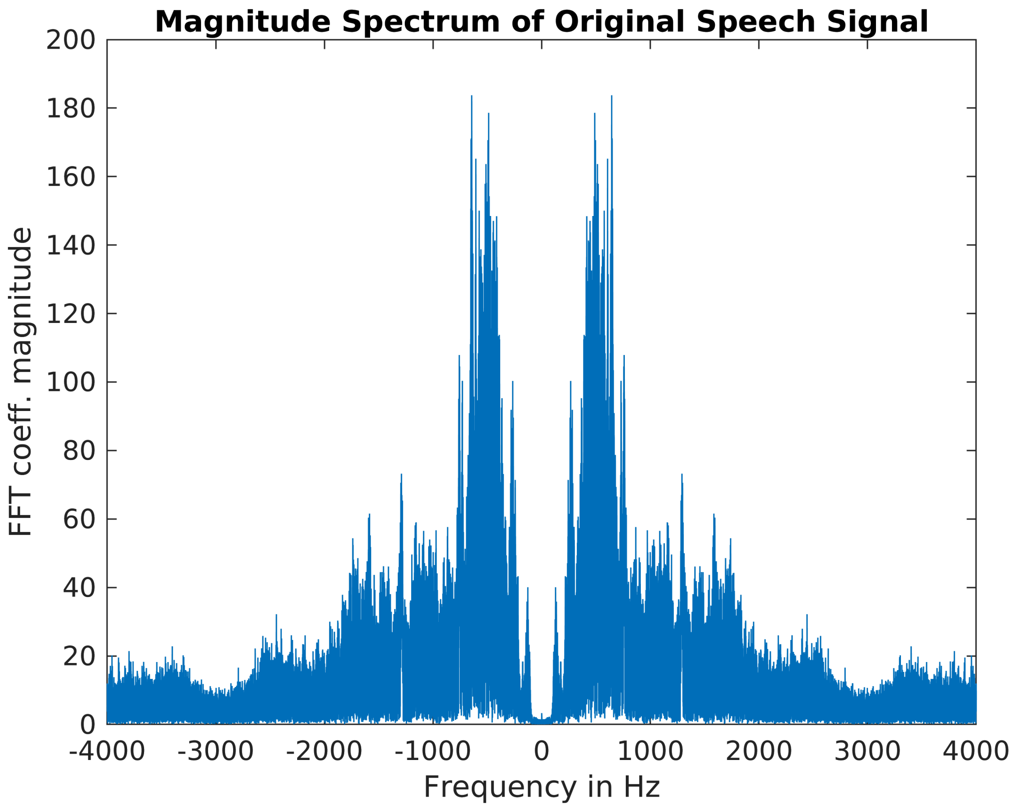
```
% Playing the .wav files at original Sampling Rates
player1 = audioplayer(audio16k, Fs_audio);
player2 = audioplayer(speech8k, Fs_speech);
playblocking(player1);
playblocking(player2);
```

```
%Magnitude Response of Original Signals
Naudio = 2^(ceil(log2(length(audio16k))));
```

```
Nspeech = 2^(ceil(log2(length(speech8k))));
fft_audio = fftshift(fft(audio16k, Naudio));
fft_speech = fftshift(fft(speech8k, Nspeech));
wAudio = linspace(-Fs_audio/2, Fs_audio/2, Naudio);
wSpeech = linspace(-Fs_speech/2, Fs_speech/2, Nspeech);
figure();
plot(wAudio, abs(fft_audio));
xlabel("Frequency in Hz");
ylabel("FFT coeff. magnitude");
title("Magnitude Spectrum of Original Audio Signal");
```




```
figure();
plot(wSpeech, abs(fft_speech));
xlabel("Frequency in Hz");
ylabel("FFT coeff. magnitude");
title("Magnitude Spectrum of Original Speech Signal");
```



1.2 Downsample by 2, play back and generate Magnitude Response

The generated audio files are saved to the location out/<originalFileName>-D2.wav

The signal {x[n]} is downsampled as

$\{x_d[n]\} = x[Mn]$, where M is the downsampling factor

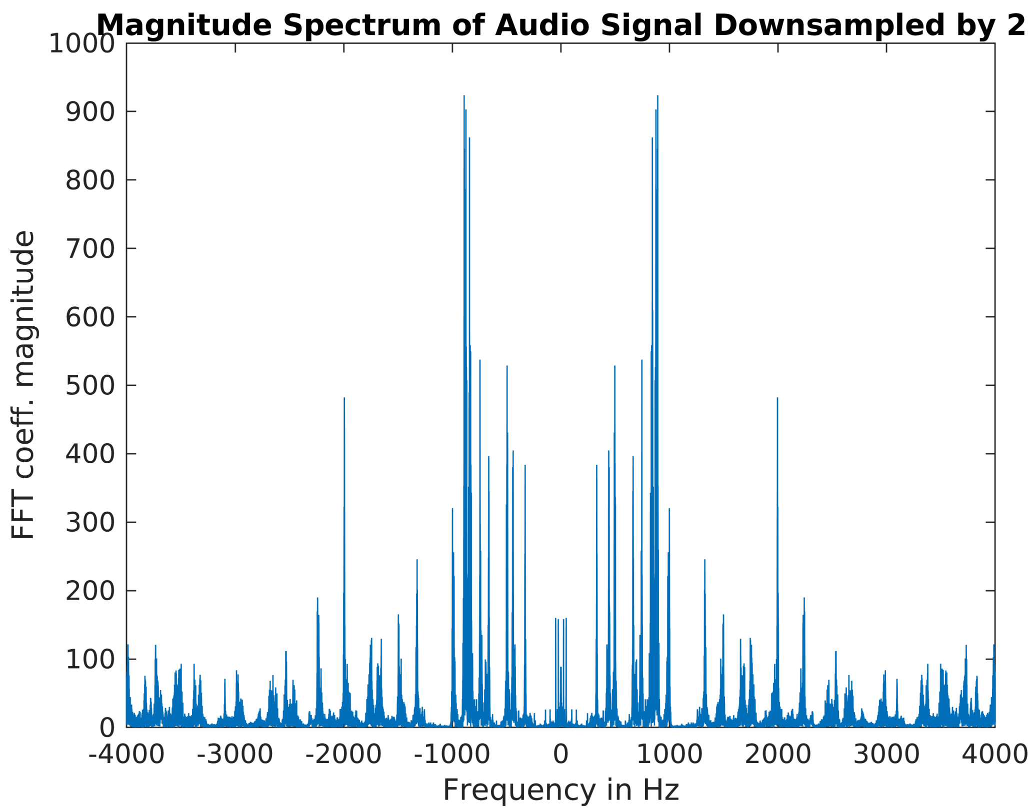
```
%Downsampling by 2
M = 2;
%M = 4;
audioDown = audio16k(1:M:end);
speechDown = speech8k(1:M:end);
Fs_audioD = Fs_audio/M;
Fs_speechD = Fs_speech/M;
player1 = audioplayer(audioDown, Fs_audioD);
player2 = audioplayer(speechDown, Fs_speechD);
playblocking(player1);
playblocking(player2);
audiowrite('out/music16k-D2.wav', audioDown, Fs_audioD);
audiowrite('out/speech8k-D2.wav', speechDown, Fs_speechD);
```

Inference:

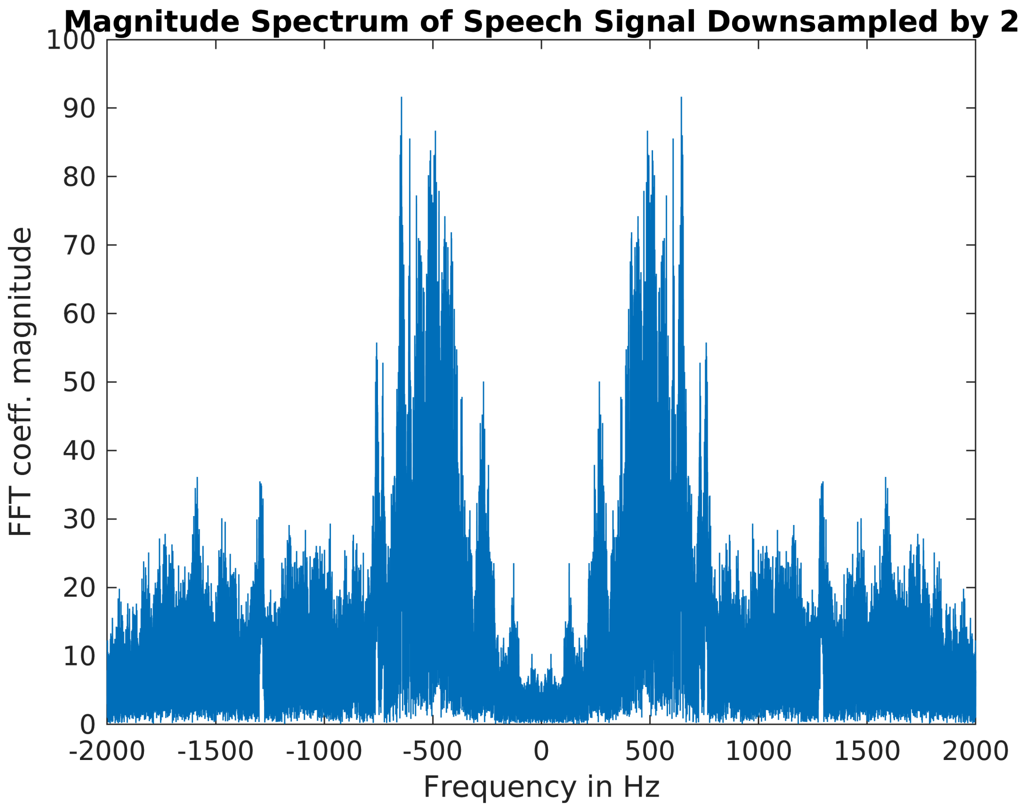
There is a clear reduction in the 'quality' of the audio signals (both music and speech) when the original signal is downsampled. For M=2, the reduction in quality due to aliasing is more pronounced for the speech signal while for M = 4, the distortions in both music and speech are evident even to the untrained ear.

The following cell plots the magnitude spectra of the downsampled signal (default: M =2). To generate the spectrum for M = 4, uncomment the "%M = 4;" line in the previous cell.

```
%Magnitude Spectrum of Downsampled Signals
NaudioD = 2^(ceil(log2(length(audioDown))));
NspeechD = 2^(ceil(log2(length(speechDown))));
fft_audioD = fftshift(fft(audioDown, NaudioD));
fft_speechD = fftshift(fft(speechDown, NspeechD));
wAudio = linspace(-Fs_audioD/2, Fs_audioD/2, NaudioD);
wSpeech = linspace(-Fs_speechD/2, Fs_speechD/2, NspeechD);
figure();
plot(wAudio, abs(fft_audioD));
xlabel("Frequency in Hz");
ylabel("FFT coeff. magnitude");
title("Magnitude Spectrum of Audio Signal Downsampled by 2");
```




```
figure();
plot(wSpeech, abs(fft_speechD));
xlabel("Frequency in Hz");
ylabel("FFT coeff. magnitude");
title("Magnitude Spectrum of Speech Signal Downsampled by 2");
```

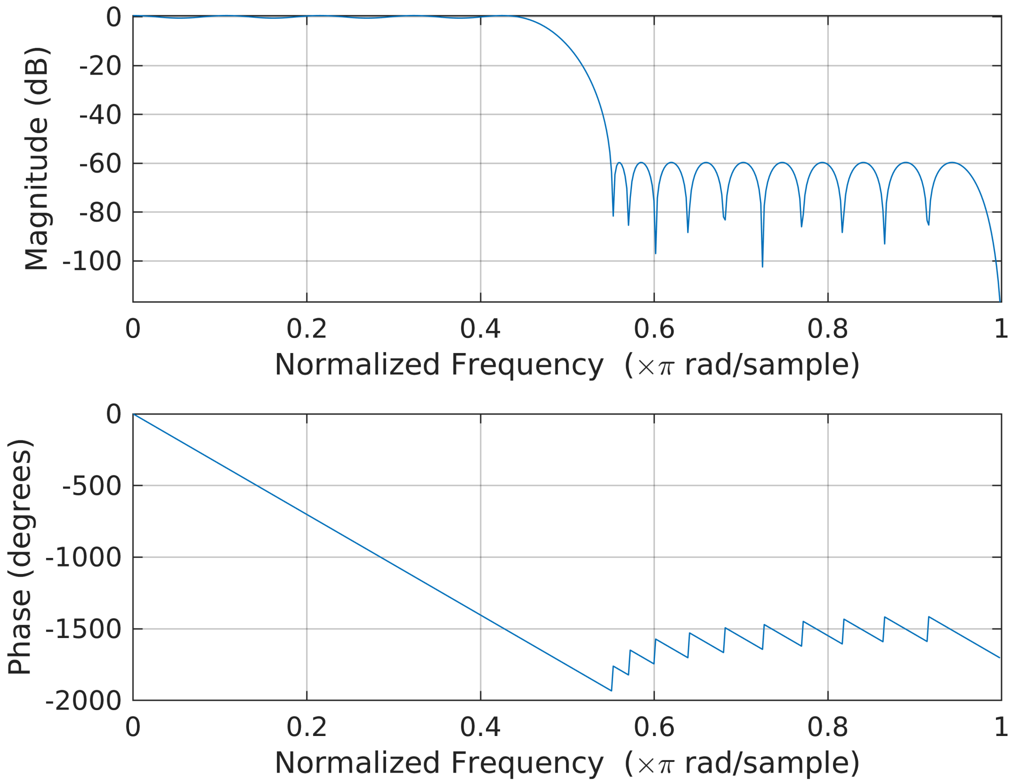


Part 2: Equiripple Filter Design and Downsampling with AAF

2.1 Equiripple Filter Design

Using Parks-McClellan optimal FIR filter design (firpm API) to design an AAF (low-pass filter) with cutoff at $\frac{\pi}{2}$

```
cutoff = [0.45*pi, 0.55*pi];
% Passband and Stopband Ripples in dB
rpass = 1;
%rstop = -30;
rstop = -60;    % Vary rstop to see if "stronger" filters have significant impact
filtFreqAudio = cutoff*Fs_audio/(2*pi);
filtFreqSpeech = cutoff*Fs_speech/(2*pi);
a = [1, 0];     % Desired amplitudes at the passband and stopband frequencies
dev = [(10^(rpass/20) - 1)/(10^(rpass/20) + 1), 10^(rstop/20)];
[n, fo, ao, w] = firpmord(filtFreqAudio, a, dev, Fs_audio);
bAudio = firpm(n, fo, ao, w);
[n, fo, ao, w] = firpmord(filtFreqSpeech, a, dev, Fs_speech);
bSpeech = firpm(n, fo, ao, w);
freqz(bAudio);
```



2.2 Downsampling after Anti-Alias Filtering

```
audioAA = filter(bAudio, 1, audio16k);
speechAA = filter(bSpeech, 1, speech8k);
M  = 2;
%M  = 4;
audioAAD = audioAA(1:M:end);
speechAAD = speechAA(1:M:end);
Fs_audioD = Fs_audio/M;
Fs_speechD = Fs_speech/M;
player1 = audioplayer(audioAAD, Fs_audioD);
player2 = audioplayer(speechAAD, Fs_speechD);
playblocking(player1);
playblocking(player2);
audiowrite('out/music16k-AAD2.wav', audioDown, Fs_audioD);
audiowrite('out/speech8k-AAD2.wav', speechDown, Fs_speechD);
```

Inference

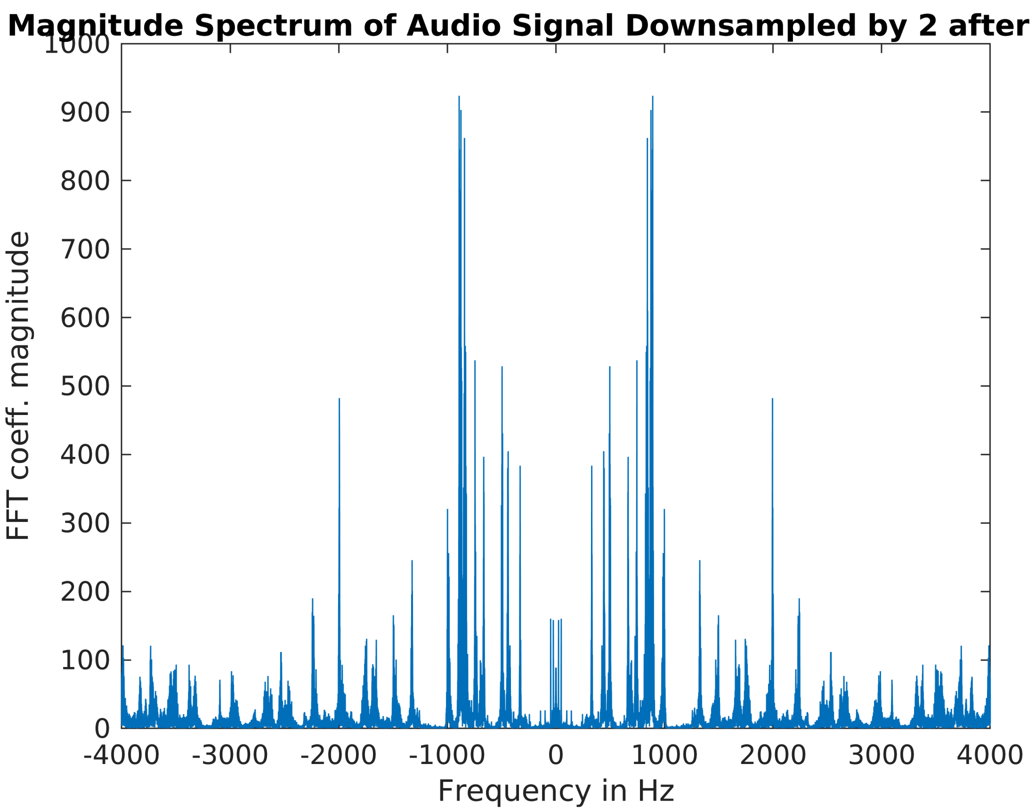
The anti-aliasing filter helps reduce aliasing in the downsampled signals. As a result, the audio quality of these signals is better than that of vanilla Downsampling (without AAF). However, for M = 4, the resultant audio signals have significant distortion.

It is also noticed upon trying various values for the stopband ripple (-10 dB, -30 dB, -60 dB) that the stopband ripple did not have a significant impact on the quality of the audio signal. Since M = 2 is not satisfactory, the next option is to downsample by a fraction => upsampling followed by downsampling (or vice versa).

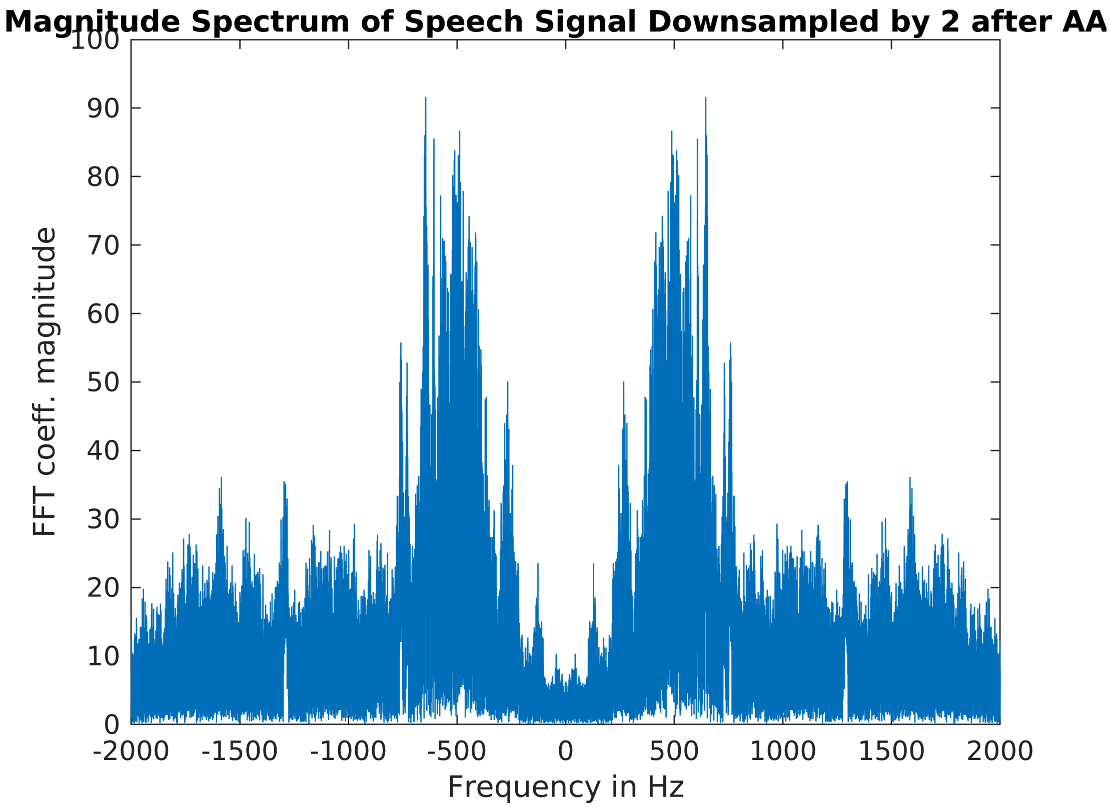
The following cells plots the magnitude response of the audio and speech signals for the above case (AAF followed by Downsampling by 2).

2.3 Magnitude Spectrum of Signals Downsampled after AAF

```
%Magnitude Spectrum of AAD Signals
fft_audioAAD = fftshift(fft(audioAAD, NaudioD));
fft_speechAAD = fftshift(fft(speechAAD, NspeechD));
wAudio = linspace(-Fs_audioD/2, Fs_audioD/2, NaudioD);
wSpeech = linspace(-Fs_speechD/2, Fs_speechD/2, NspeechD);
figure();
plot(wAudio, abs(fft_audioD));
xlabel("Frequency in Hz");
ylabel("FFT coeff. magnitude");
title("Magnitude Spectrum of Audio Signal Downsampled by 2 after AAF");
```



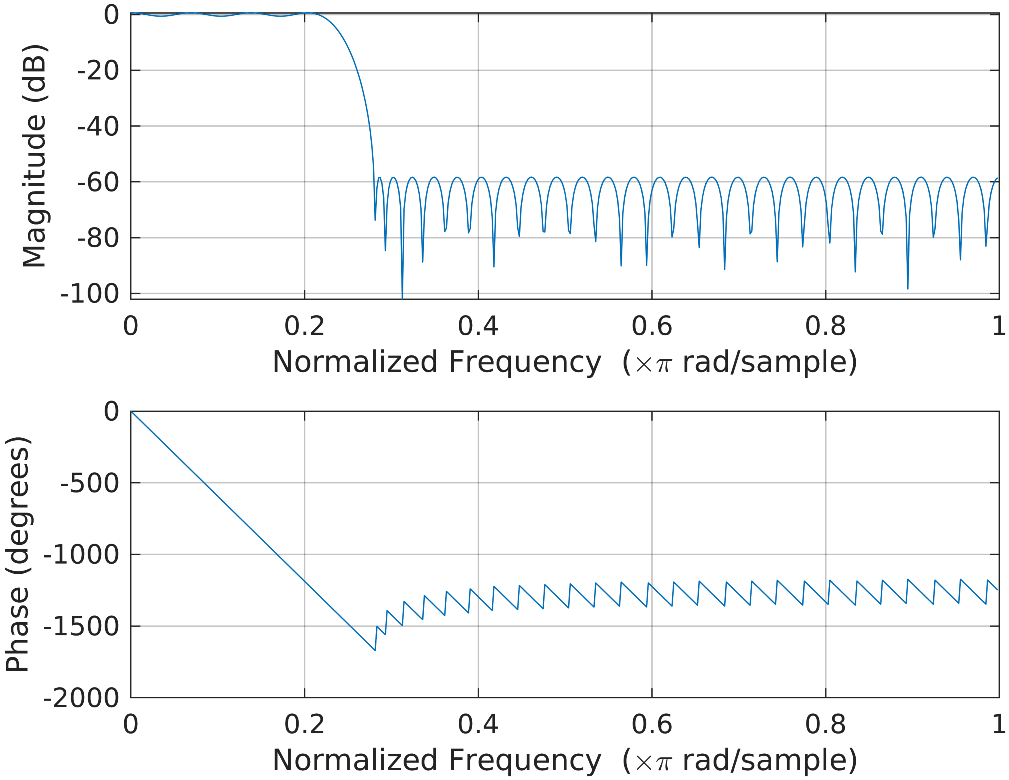

```
figure();
plot(wSpeech, abs(fft_speechD));
xlabel("Frequency in Hz");
ylabel("FFT coeff. magnitude");
title("Magnitude Spectrum of Speech Signal Downsampled by 2 after AAF");
```



Part 3: Equiripple LPF Design

Lowpass Filter with cutoff at $\frac{\pi}{4}$

```
cutoff = [0.22*pi, 0.28*pi];
% Passband and Stopband Ripples in dB
rpass = 1;
%rstop = -30;
rstop = -60;    % Vary rstop to see if "stronger" filters have significant impact
filtFreqAudio = cutoff*Fs_audio/(2*pi);
filtFreqSpeech = cutoff*Fs_speech/(2*pi);
a = [1, 0];     % Desired amplitudes at the passband and stopband frequencies
dev = [(10^(rpass/20) - 1)/(10^(rpass/20) + 1), 10^(rstop/20)];
[n, fo, ao, w] = firpmord(filtFreqAudio, a, dev, Fs_audio);
lpfAudio = firpm(n, fo, ao, w);
[n, fo, ao, w] = firpmord(filtFreqSpeech, a, dev, Fs_speech);
lpfSpeech = firpm(n, fo, ao, w);
freqz(lpfAudio);
```



Part 4: Fractional Downsampling by Upsamling before Downsampling

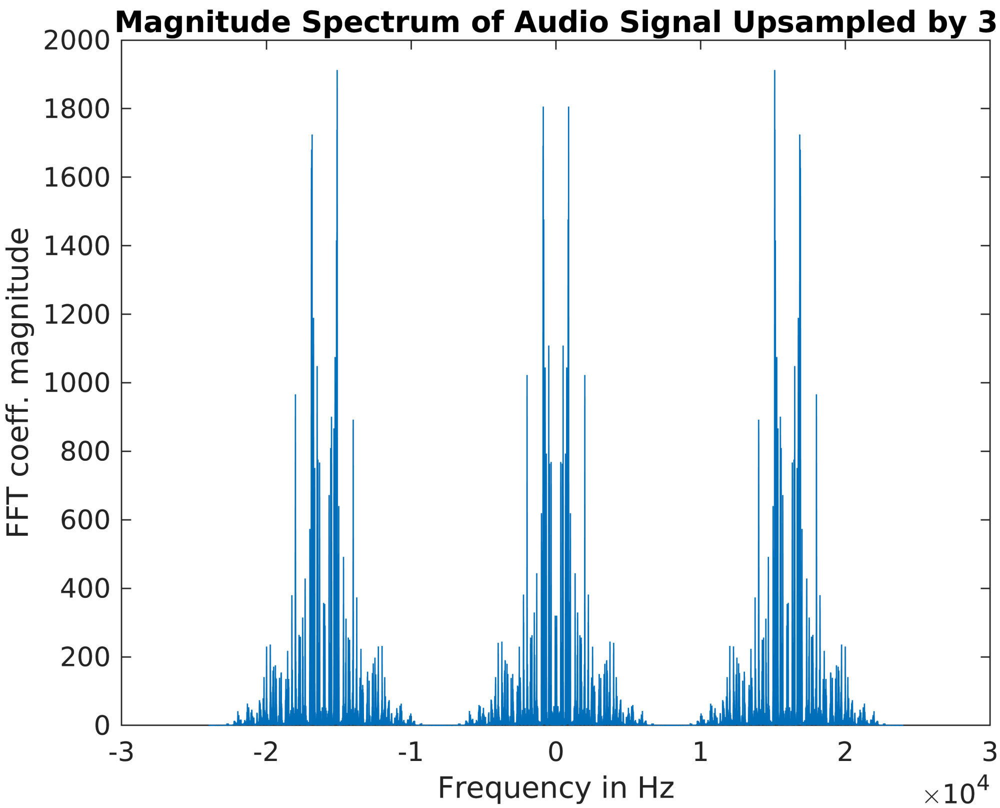
In this section, the audio signals are first upsampled by a factor of 3, then multiplied by the LPF and then downsampled by a factor of 4. Here, the LPF behaves like an interpolation filter for the upsampling process and as an Anti-Aliasing Filter for the downsampling process.

4.1 Play Back and Magnitude Spectrum of Upsampled Signal

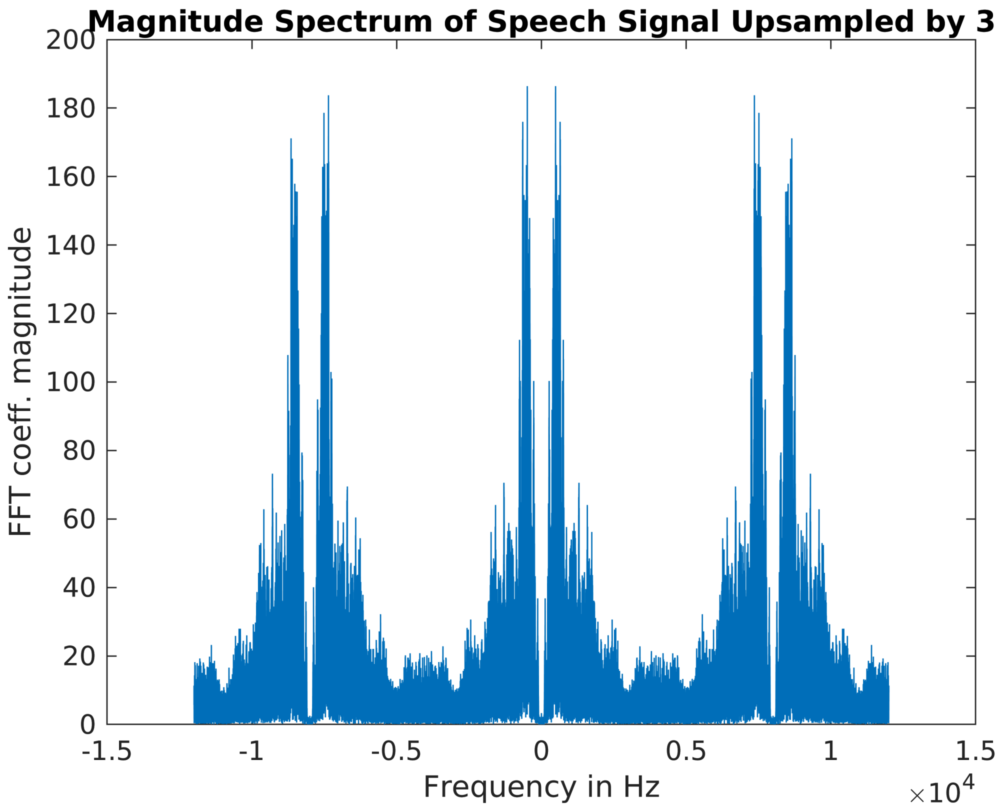
```
L = 3;
audioUp = upsample(audio16k, L);
speechUp = upsample(speech8k, L);
%{
playback of upsampled signal: Notice high freq 'ringing' as the signal
is not interpolated.
%}
player1 = audioplayer(audioUp, Fs_audio*L);
player2 = audioplayer(speechUp, Fs_speech*L);
playblocking(player1);
playblocking(player2);
```

```
% Magnitude Spectrum of Upsampled Signal
NaudioUp = 2^(ceil(log2(length(audioUp))));
NspeechUp = 2^(ceil(log2(length(speechUp))));
Fs_audioUp = Fs_audio*L;
Fs_speechUp = Fs_speech*L;
fft_audioUp = fftshift(fft(audioUp, NaudioUp));
```

```
fft_speechUp = fftshift(fft(speechUp, NspeechUp));
wAudio = linspace(-Fs_audioUp/2, Fs_audioUp/2, NaudioUp);
wSpeech = linspace(-Fs_speechUp/2, Fs_speechUp/2, NspeechUp);
figure();
plot(wAudio, abs(fft_audioUp));
xlabel("Frequency in Hz");
ylabel("FFT coeff. magnitude");
title("Magnitude Spectrum of Audio Signal Upsampled by 3");
```

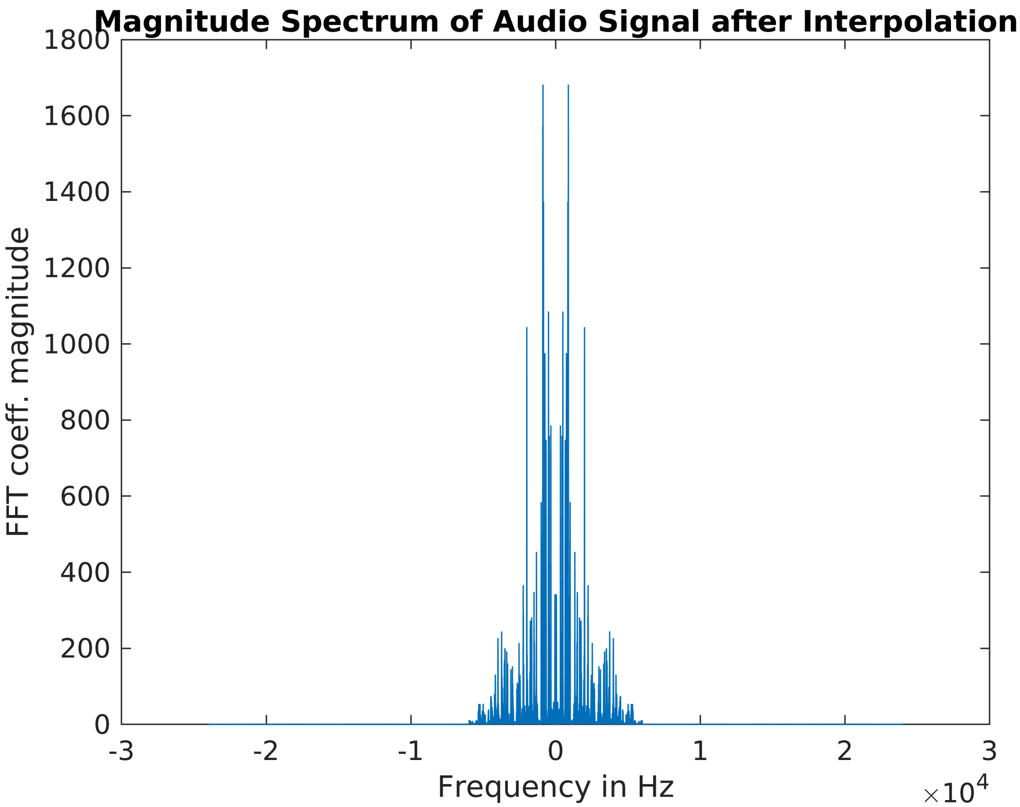



```
figure();
plot(wSpeech, abs(fft_speechUp));
xlabel("Frequency in Hz");
ylabel("FFT coeff. magnitude");
title("Magnitude Spectrum of Speech Signal Upsampled by 3");
```

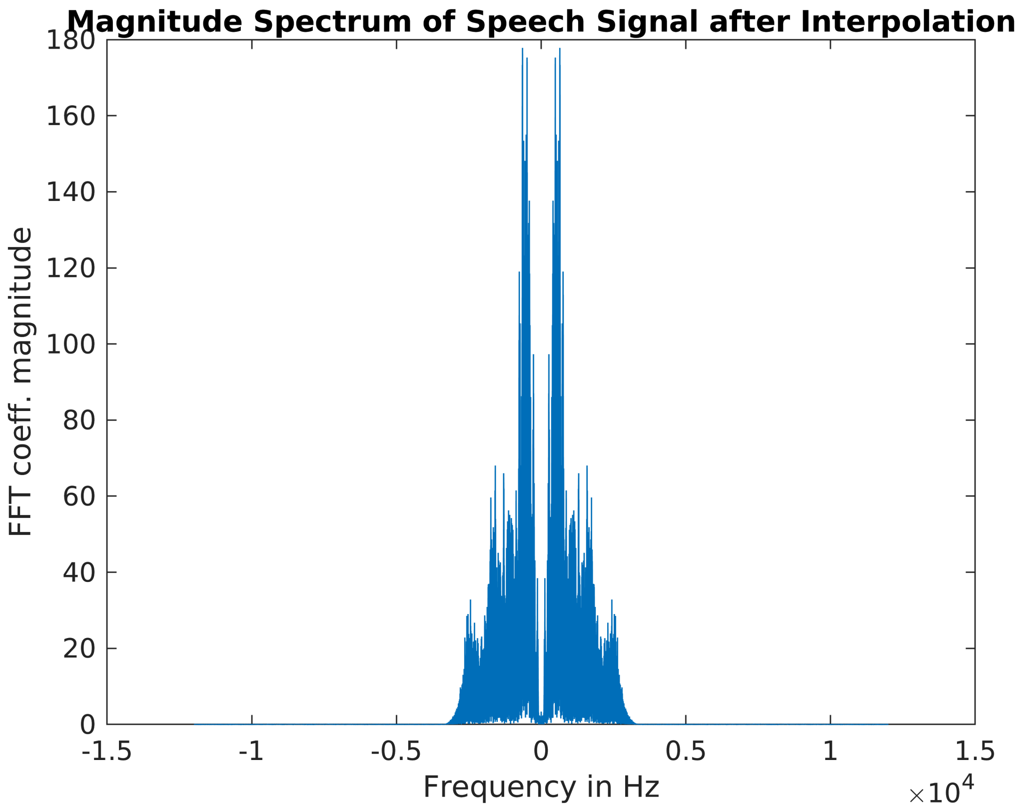


4.2 Play Back and Magnitude Spectrum of Interpolated Signal

```
audioUAA = filter(lpfAudio, 1, audioUp);
speechUAA = filter(lpfSpeech, 1, speechUp);
%{
playback of interpolated signal: notice the better sound quality
%}
player1 = audioplayer(audioUAA, Fs_audio*L);
player2 = audioplayer(speechUAA, Fs_speech*L);
playblocking(player1);
playblocking(player2);
% Mag Spectrum of Interpolated Signals
fft_audioUAA = fftshift(fft(audioUAA, NaudioUp));
fft_speechUAA = fftshift(fft(speechUAA, NspeechUp));
figure();
plot(wAudio, abs(fft_audioUAA));
xlabel("Frequency in Hz");
ylabel("FFT coeff. magnitude");
title("Magnitude Spectrum of Audio Signal after Interpolation");
```




```
figure();
plot(wSpeech, abs(fft_speechUAA));
xlabel("Frequency in Hz");
ylabel("FFT coeff. magnitude");
title("Magnitude Spectrum of Speech Signal after Interpolation");
```

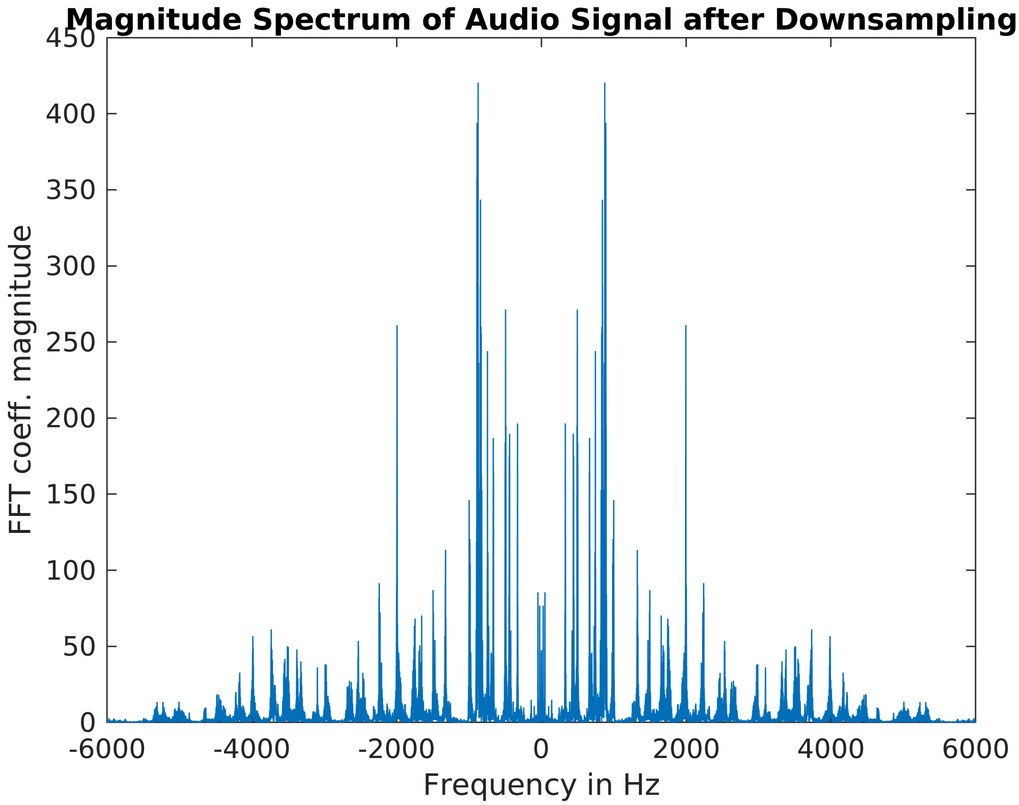


4.3 Downsampling Filtered Signal

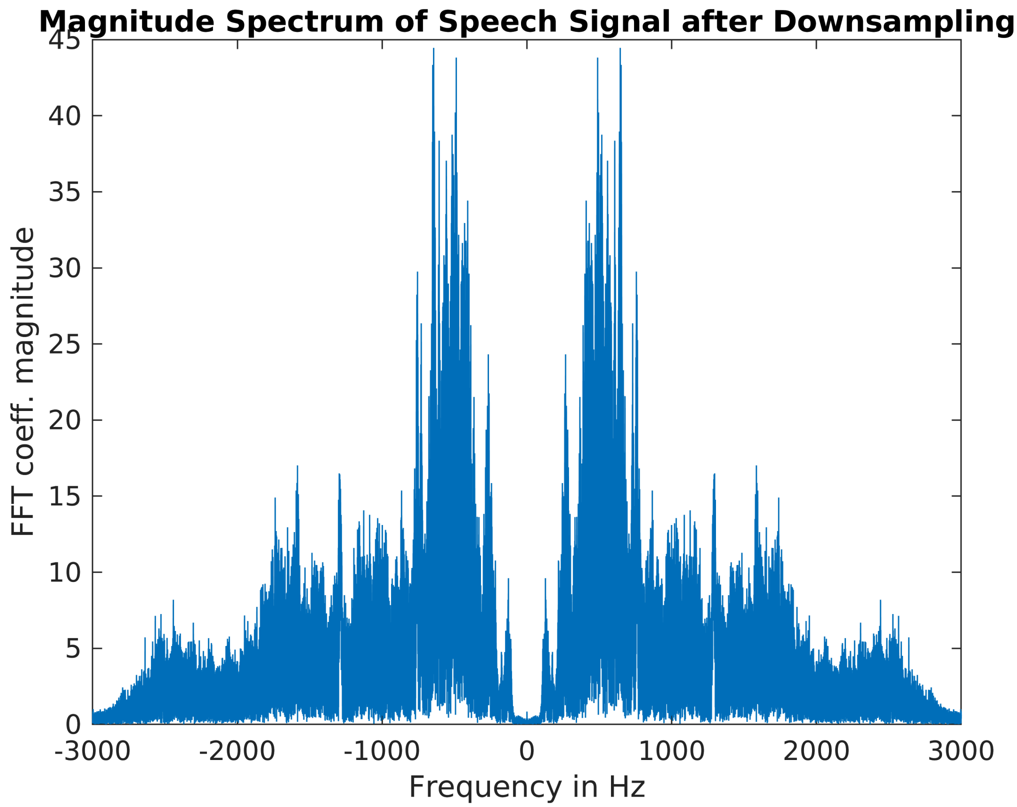
```
M = 4;
audioFrac = audioUAA(1:M:end);
speechFrac = speechUAA(1:M:end);
Fs_audioFrac = Fs_audio*(L/M);
Fs_speechFrac = Fs_speech*(L/M);
%{
playback of downsampled signal: fractional downsampling (effective) of 4/3.
Naturally the audio quality for this is the best among Parts 1, 2 and 4.
%}
player1 = audioplayer(audioFrac, Fs_audioFrac);
player2 = audioplayer(speechFrac, Fs_speechFrac);
playblocking(player1);
playblocking(player2);
audiowrite('out/music16k-UAAD.wav', audioFrac, Fs_audioFrac);
audiowrite('out/speech8k-UAAD.wav', speechFrac, Fs_speechFrac);
```

```
% Mag Spectrum of Fractionally Downsampled Signals
NaudioFrac = 2^(ceil(log2(length(audioFrac))));
NspeechFrac = 2^(ceil(log2(length(speechFrac))));
fft_audioFrac = fftshift(fft(audioFrac, NaudioFrac));
fft_speechFrac = fftshift(fft(speechFrac, NspeechFrac));
wAudio = linspace(-Fs_audioFrac/2, Fs_audioFrac/2, NaudioFrac);
wSpeech = linspace(-Fs_speechFrac/2, Fs_speechFrac/2, NspeechFrac);
figure();
```

```
plot(wAudio, abs(fft_audioFrac));
xlabel("Frequency in Hz");
ylabel("FFT coeff. magnitude");
title("Magnitude Spectrum of Audio Signal after Downsampling");
```




```
figure();
plot(wSpeech, abs(fft_speechFrac));
xlabel("Frequency in Hz");
ylabel("FFT coeff. magnitude");
title("Magnitude Spectrum of Speech Signal after Downsampling");
```



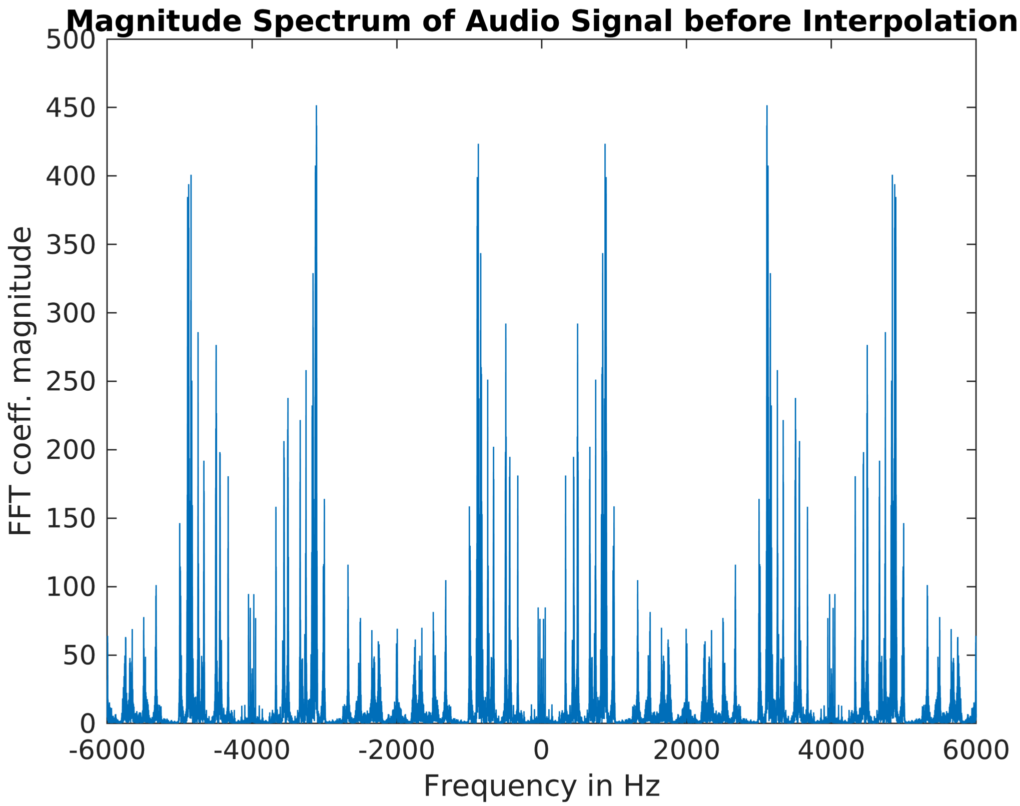
Part 5: Fractional Downsampling by Downsampling before Upsampling

5.1 Play Back and Magnitude Response for Output before Interpolating

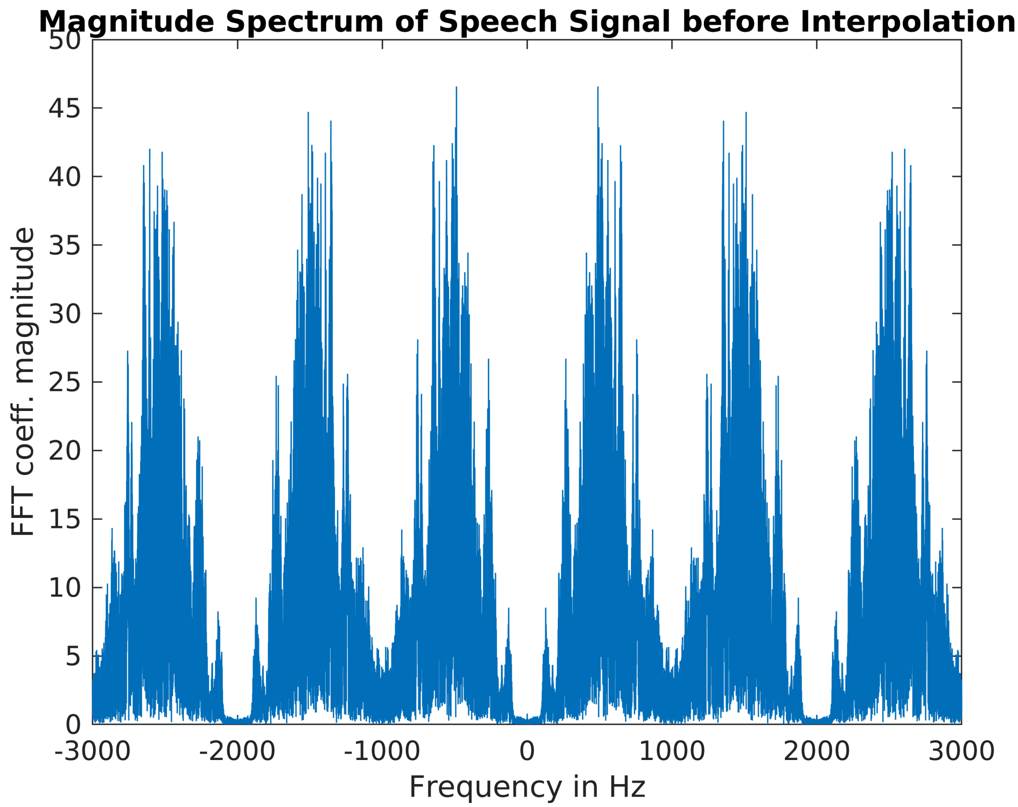
```
M = 4;
L = 3;
audioAA = filter(lpfAudio, 1, audio16k);
speechAA = filter(lpfSpeech, 1, speech8k);
audioD4 = audioAA(1:M:end);
speechD4 = speechAA(1:M:end);
audioAADU = upsample(audioD4, L);
speechAADU = upsample(speechD4, L);
%{
playback of upsampled signal: fractional downsampling (effective) of 4/3.
Naturally the audio quality is poor because performing downsampling first leads to alia
%}
player1 = audioplayer(audioAADU, Fs_audioFrac);
player2 = audioplayer(speechAADU, Fs_speechFrac);
playblocking(player1);
playblocking(player2);
```

```
% Mag Spectrum of Fractionally Downsampled Signals
NaudioFrac = 2^(ceil(log2(length(audioAADU))));
NspeechFrac = 2^(ceil(log2(length(speechAADU))));
fft_audioFrac = fftshift(fft(audioAADU, NaudioFrac));
fft_speechFrac = fftshift(fft(speechAADU, NspeechFrac));
```

```
wAudio = linspace(-Fs_audioFrac/2, Fs_audioFrac/2, NaudioFrac);
wSpeech = linspace(-Fs_speechFrac/2, Fs_speechFrac/2, NspeechFrac);
figure();
plot(wAudio, abs(fft_audioFrac));
xlabel("Frequency in Hz");
ylabel("FFT coeff. magnitude");
title("Magnitude Spectrum of Audio Signal before Interpolation");
```




```
figure();
plot(wSpeech, abs(fft_speechFrac));
xlabel("Frequency in Hz");
ylabel("FFT coeff. magnitude");
title("Magnitude Spectrum of Speech Signal before Interpolation");
```



5.2 Interpolation Filter Design

```
bInt = intfilt(L, 3, 1);    % interpolation filter of support 17
audioOut = filter(bInt, 1, audioAADU);
speechOut = filter(bInt, 1, speechAADU);
%{
playback of interpolated signal: fractional downsampling (effective) of 4/3.
%}
player1 = audioplayer(audioOut, Fs_audioFrac);
player2 = audioplayer(speechOut, Fs_speechFrac);
playblocking(player1);
playblocking(player2);
audiowrite('out/music16k-AADUI.wav', audioOut, Fs_audioFrac);
audiowrite('out/speech8k-AADUI.wav', speechOut, Fs_speechFrac);
```

Inference

The information loss caused by downsampling first is irretrievable, hence the audio quality does not improve significantly upon interpolation of the upsampled signal. Of the downsampled signals from Parts 1 to 5, the following is the order of the observed audio quality in ascending order:

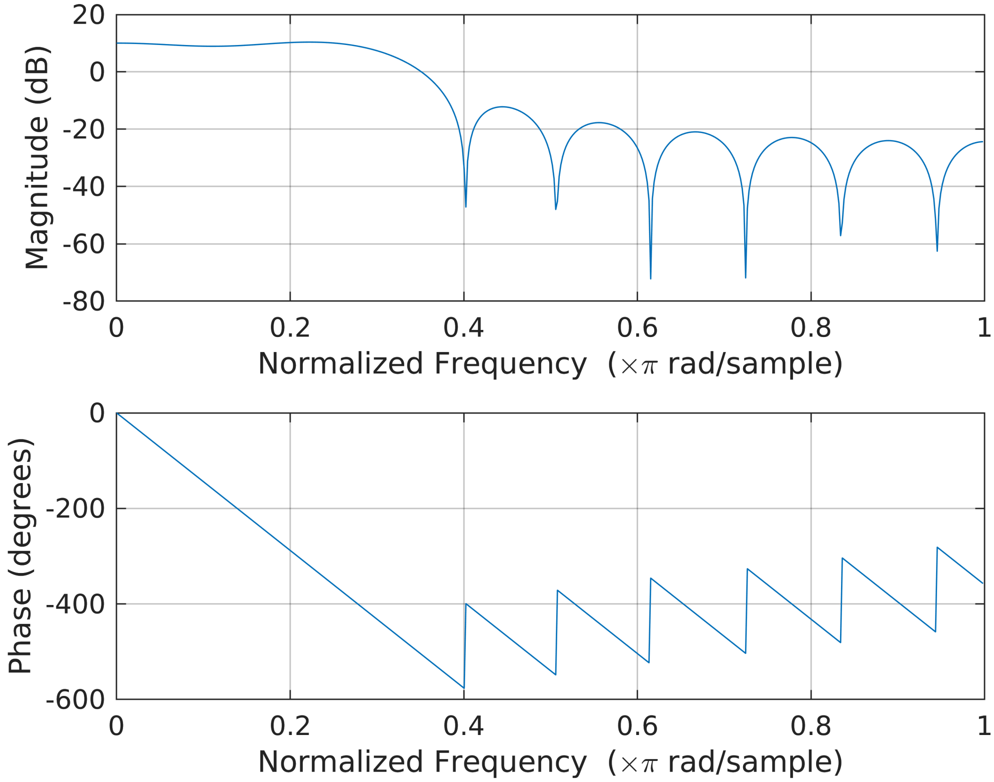
Note: Only output signals (that have been saved to the out/ directory) of each part are being compared here. Remarks on intermediate signals such as upsampled but not interpolated signals are made as comments in the appropriate code sections.

1. Part 5: Anti-Aliasing -> Downsample by 4 -> Upsample by 3 -> Interpolation

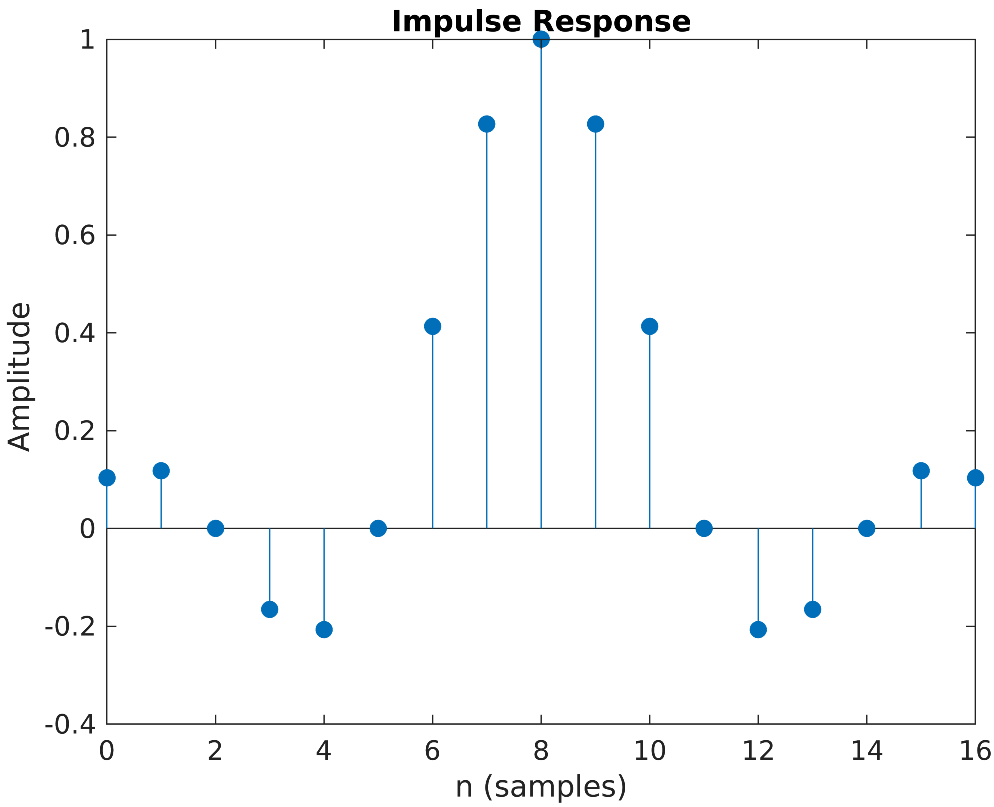
2. Part 1: Downsample by 2
3. Part 2: Anti-Aliasing -> Downsample by 2
4. Part 4: Upsample by 3 -> Interpolation-cum-Anti-Aliasing Filtering -> Downsample by 4
5. Original Signal

5.3 Magnitude Response of Interpolating Filter and Output

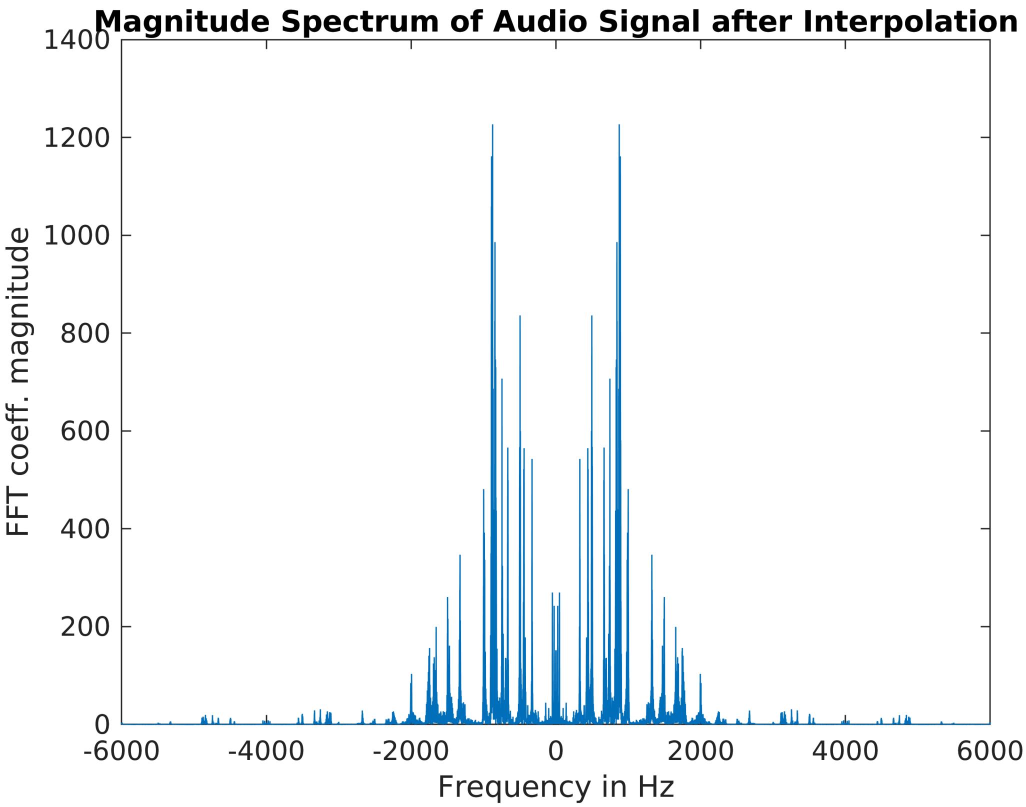
```
freqz(bInt);    % Frequency Response Spectrum
```



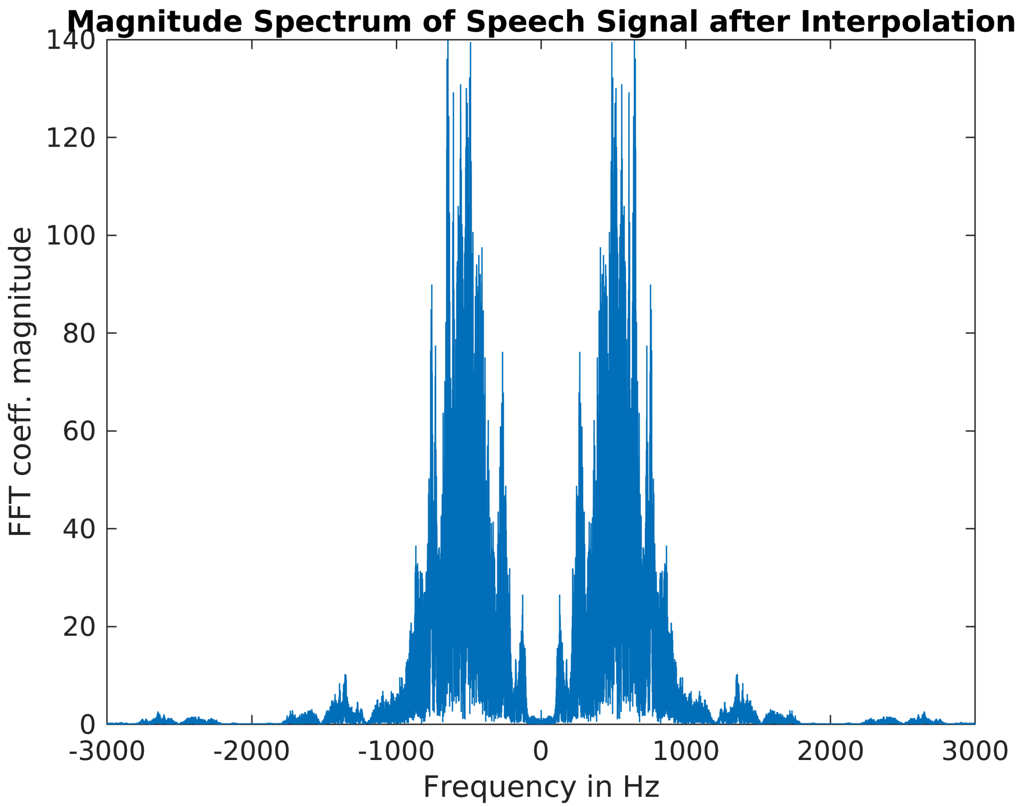

```
impz(bInt); % Impulse Response of the Interpolation Filter
```




```
% Mag Spectrum of Interpolated Signals
fft_audioOut = fftshift(fft(audioOut, NaudioFrac));
fft_speechOut = fftshift(fft(speechOut, NspeechFrac));
figure();
plot(wAudio, abs(fft_audioOut));
xlabel("Frequency in Hz");
ylabel("FFT coeff. magnitude");
title("Magnitude Spectrum of Audio Signal after Interpolation");
```




```
figure();
plot(wSpeech, abs(fft_speechOut));
xlabel("Frequency in Hz");
ylabel("FFT coeff. magnitude");
title("Magnitude Spectrum of Speech Signal after Interpolation");
```


Magnitude Spectrum of Speech Signal after Interpolation
FFT coeff. magnitude
Frequency in Hz
0
20
40
60
80
100
120
140
-3000
-2000
-1000
0
1000
2000
3000