Experiment 03

Part 1: Upsampling and Downsampling audio signal

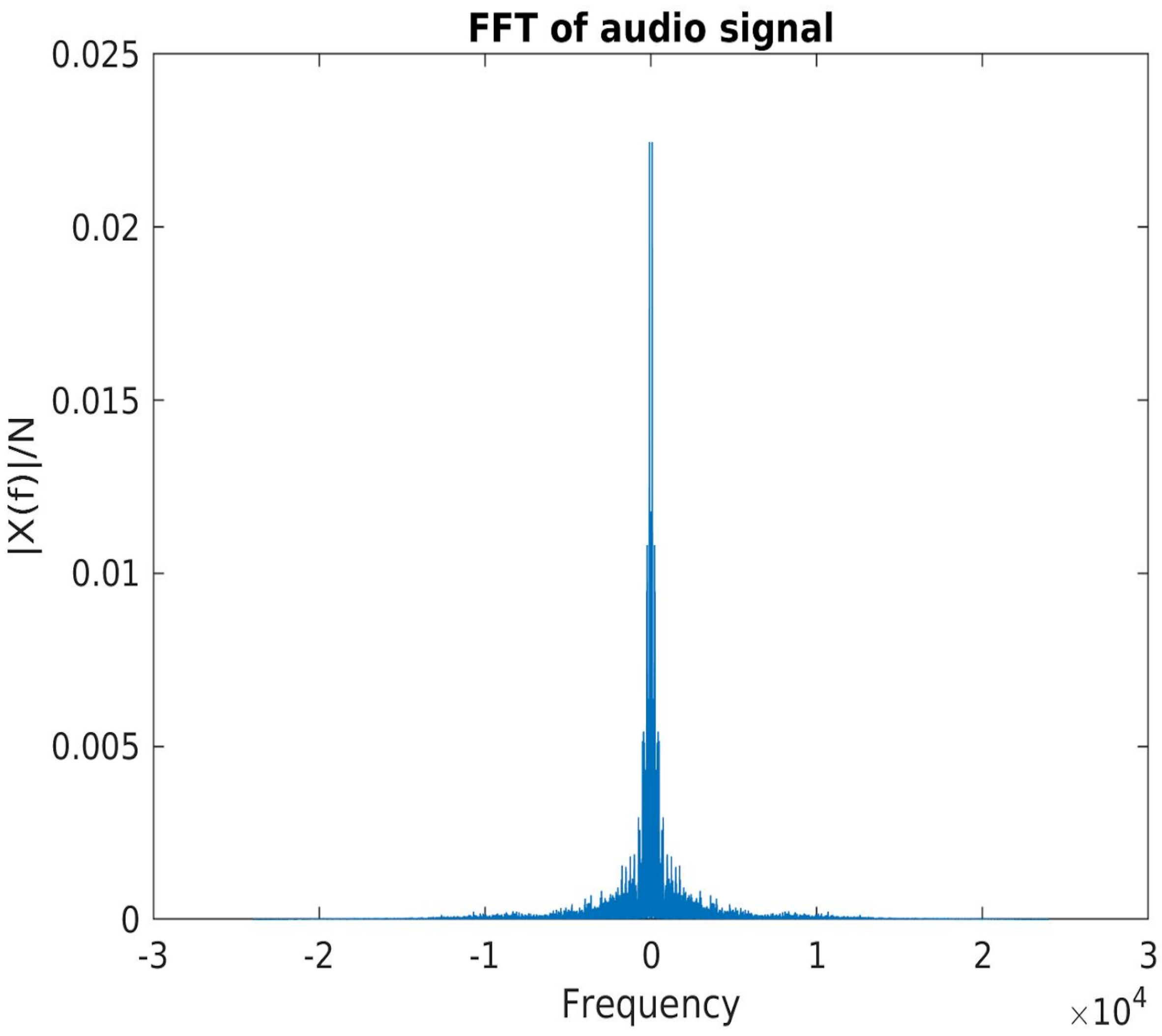
Code:

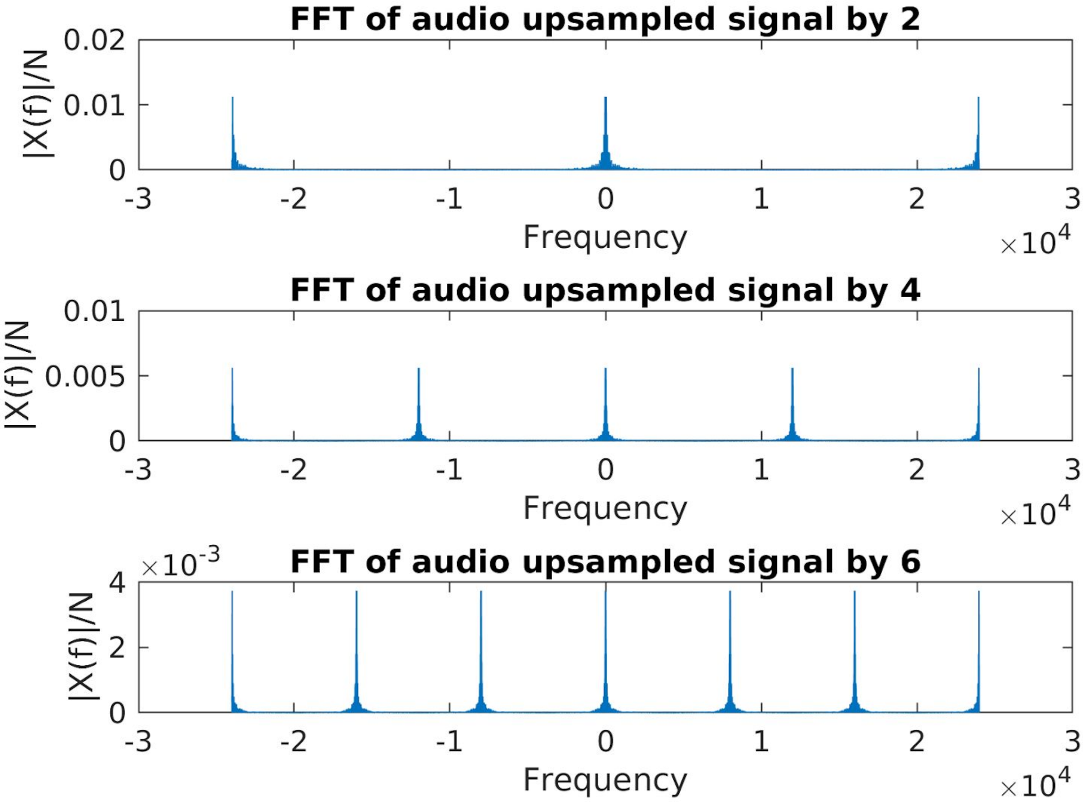
```
[signal , fs]=audioread('audio48kHz.wav');
N=length(signal);
df=fs/N;
freq=-fs/2:df:fs/2-df;
%FFT of the audio_file
X=fftshift(abs(fft(signal))/N);
plot(freq,X);
title('FFT of audio signal');
xlabel('Frequency');
ylabel('|X(f)|/N');
upsig2=zeros(1,2*N);
upsig4=zeros(1,4*N);
upsig6=zeros(1,6*N);
%Upsampling the signal by 2,4,6
for i=1:N
upsig2(1,2*i)=signal(i);
upsig4(1,4*i)=signal(i);
upsig6(1,6*i)=signal(i);
end
%FFT of upsampled signals
Xup2=fftshift(abs(fft(upsig2(1,:)))/(2*N));
Xup4=fftshift(abs(fft(upsig4(1,:)))/(4*N));
Xup6=fftshift(abs(fft(upsig6(1,:)))/(6*N));
figure
subplot(3,1,1);
frequp2=-fs/2:df/2:fs/2-df/2;
plot(frequp2,Xup2);
title('FFT of audio upsampled signal by 2');
xlabel('Frequency');ylabel('|X(f)|/N');
subplot(3,1,2);
frequp4=-fs/2:df/4:fs/2-df/4;
plot(frequp4,Xup4);
title('FFT of audio upsampled signal by 4');
```

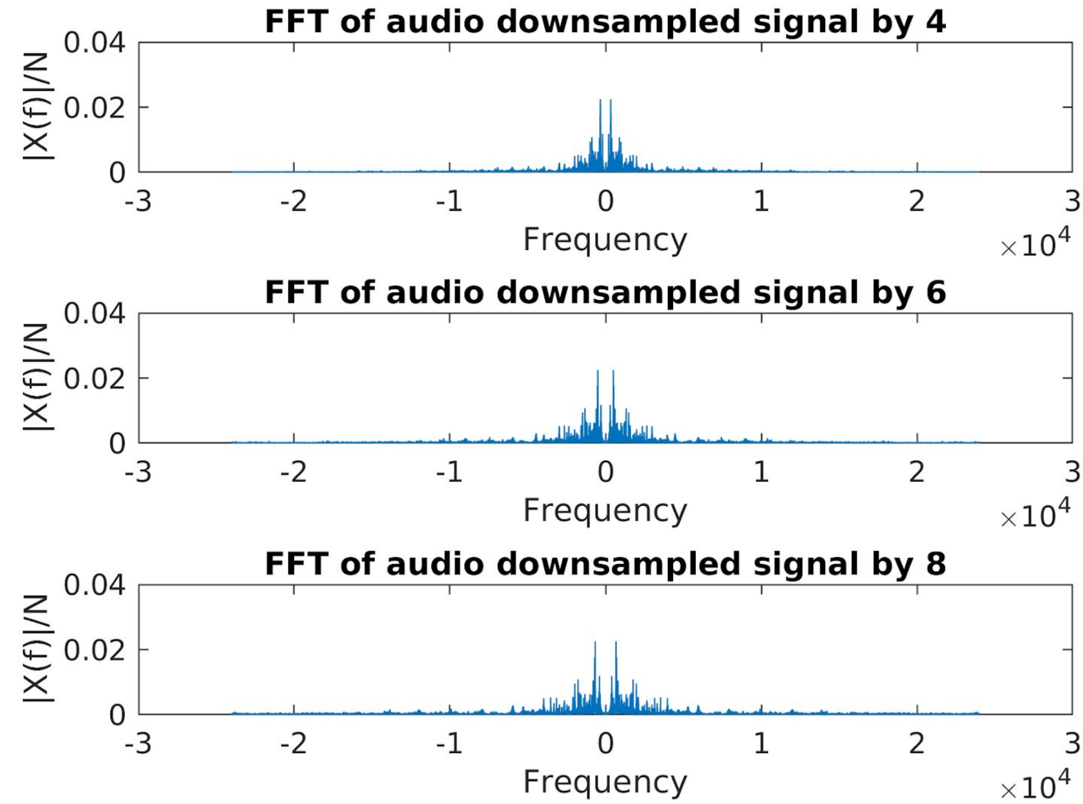
```
xlabel('Frequency');ylabel('|X(f)|/N');
subplot(3,1,3);
frequp6=-fs/2:df/6:fs/2-df/6;
plot(frequp6,Xup6);
title('FFT of audio upsampled signal by 6');
xlabel('Frequency');ylabel('|X(f)|/N');

%Downsampling by 4
downsig4=zeros(1,floor(N/4));
for i=1:floor(N/4)
downsig4(1,i)=signal(4*i);
end
%Downsampling by 6
downsig6=zeros(1,floor(N/6));
for i=1:floor(N/6)
downsig6(1,i)=signal(6*i);
end
%Downsampling by 8
downsig8=zeros(1,floor(N/8));
for i=1:floor(N/8)
downsig8(1,i)=signal(8*i);
end
%FFT of downsampled signals
Xdown4=fftshift(abs(fft(downsig4(1,:)))/(floor(N/4)));
Xdown6=fftshift(abs(fft(downsig6(1,:)))/(floor(N/6)));
Xdown8=fftshift(abs(fft(downsig8(1,:)))/(floor(N/8)));
figure
subplot(3,1,1);
freqd4=-fs/2:4*df:fs/2-4*df;
plot(freqd4,Xdown4);
title('FFT of audio downsampled signal by 4');
xlabel('Frequency');ylabel('|X(f)|/N');
subplot(3,1,2);
freqd6=-fs/2:6*df:fs/2-6*df;
plot(freqd6,Xdown6);
title('FFT of audio downsampled signal by 6');
xlabel('Frequency');ylabel('|X(f)|/N');
subplot(3,1,3);
```

```
freqd8=-fs/2:8*df:fs/2-8*df;
plot(freqd8,Xdown8);
title('FFT of audio downsampled signal by 8');
xlabel('Frequency');ylabel('|X(f)|/N');
%Creating audio files of corresponding upsampled signals
audiowrite('audioup2.wav',upsig2,2*fs);
audiowrite('audioup4.wav',upsig4,4*fs);
audiowrite('audioup6.wav',upsig6,6*fs);
%Creating audio files of corresponding downsampled signals
audiowrite('audiodown4.wav',downsig4,fs/4);
audiowrite('audiodown6.wav',downsig6,fs/6);
audiowrite('audiodown8.wav',downsig8,fs/8);
```




FFT of audio upsampled signal by 2
|X(f)|/N
0.02
0.01
0
-3
-2
-1
0
1
2
3
Frequency
$\times 10^4$
FFT of audio upsampled signal by 4
|X(f)|/N
0.01
0.005
0
Frequency
$\times 10^4$
FFT of audio upsampled signal by 6
$\times 10^{-3}$
|X(f)|/N
4
2
0
Frequency
$\times 10^4$


FFT of audio downsampled signal by 4
|X(f)|/N
0.04
0.02
0
-3
-2
-1
0
1
2
3
Frequency
$\times 10^4$
FFT of audio downsampled signal by 6
|X(f)|/N
0.04
0.02
0
Frequency
$\times 10^4$
FFT of audio downsampled signal by 8
|X(f)|/N
0.04
0.02
0
Frequency
$\times 10^4$

- The clarity of the upsampled signal is more clear as the sampling frequency is increased.
- This is because upsampling involves elongation in time domain which consequently results in compression in frequency domain.
- The clarity of the downsampled signal is not that good when compared to that of the upsampled version of the signal as the sampling frequency is reduced.
- Reduction in sampling frequency results in essentially results in loss of information to some extent.
- This is because downsampling essentially results in compression in time domain and elongation in frequency domain.
- We can conclude that the upsampling is better than the downsampling as there is no loss of information regarding the signal.

  Link to audios :
  https://drive.google.com/open?id=124b0IhX7Lx_-6G4__yXq_BiIaL4AI6Lr

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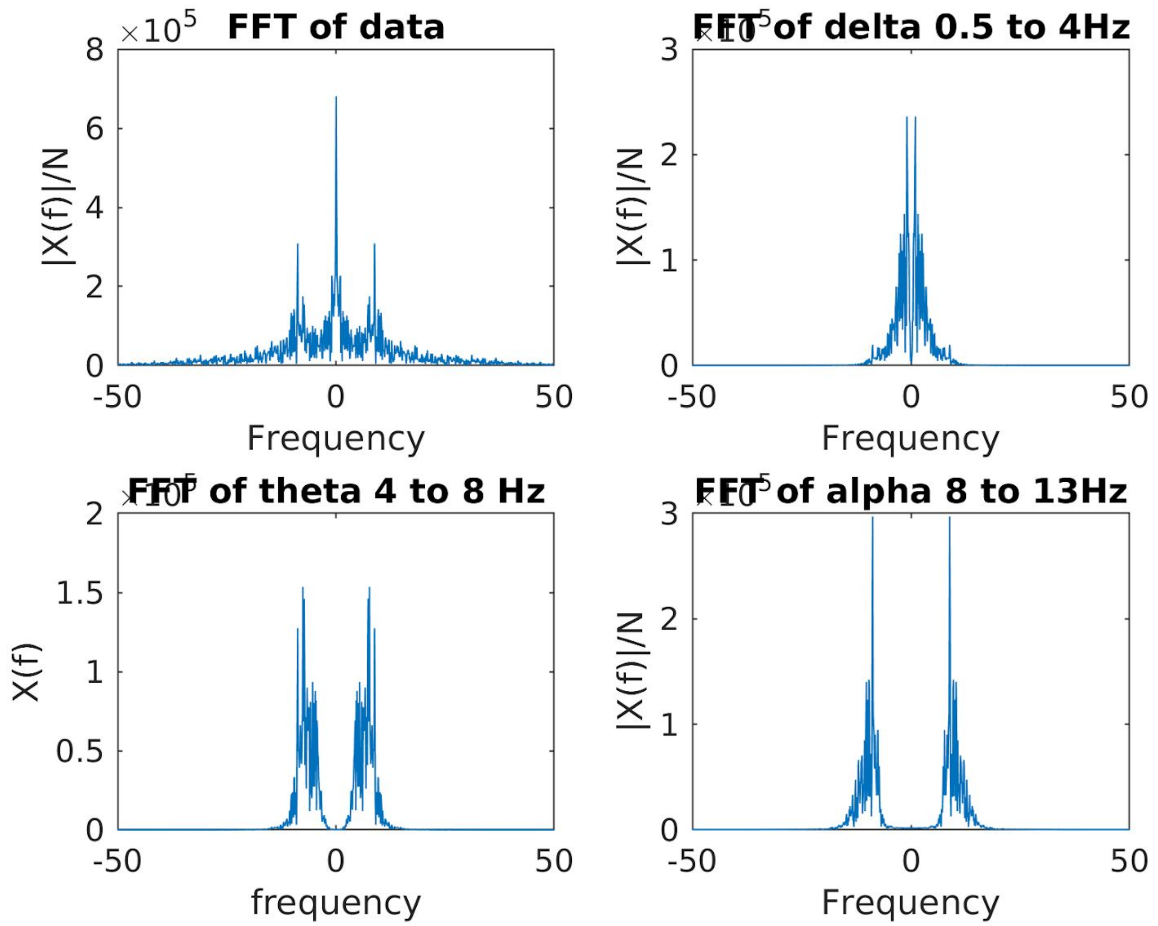
Part 2 : Filtering EEG Signals

Code:

```
%%%%%%%%Exp03b%%%%%%%%%
clc
clear all
close all

data = load('eeg1-f3.dat') ;
Fs = 100; %%(0 to 1 ----> 1 corresponds to 50Hz ----> 0.01 corresponds to 0.5Hz)
t= 1:1:length(data);
T = 1/Fs;
% subplot(4,2,1);
% stem(t,data);

nf=1024; %number of point in DTFT
f =-Fs/2:(Fs)/(nf-1):Fs/2;
```

```
Y = fftshift(abs(fft(data,nf)));
subplot(2,2,1);
plot(f,Y);
title('FFT of data');
xlabel('Frequency');ylabel('|X(f)|/N');


[b1,a1] = butter(3,[0.01 0.08],'bandpass');
y1 = filter(b1,a1,data);
% subplot(4,2,3);
% stem(t,y1);
% title('Delta frequency 0.5 to 4 Hz');
Y1 = fftshift(abs(fft(y1,nf)));
subplot(2,2,2);
plot(f,Y1);
xlabel('frequency');
ylabel('X(f)');
title('FFT of delta 0.5 to 4Hz');
xlabel('Frequency');
ylabel('|X(f)|/N');

[b2,a2] = butter(3,[0.08 0.16],'bandpass');
y2 = filter(b2,a2,data);
% subplot(4,2,5);
% stem(t,y2);
% title('Theta frequency 4 to 8 Hz');
Y2 = fftshift(abs(fft(y2,nf)));
subplot(2,2,3);
plot(f,Y2);
xlabel('frequency');
ylabel('X(f)');
title('FFT of theta 4 to 8 Hz');

[b3,a3] = butter(3,[0.16 0.26],'bandpass');
y3 = filter(b3,a3,data);
% subplot(4,2,7);
% stem(t,y3);
% title('Alpha frequency 8 to 13Hz');
Y3 = fftshift(abs(fft(y3,nf)));
subplot(2,2,4);
plot(f,Y3);
xlabel('frequency');
```

```
ylabel('X(f)');
title('FFT of alpha 8 to 13Hz');
xlabel('Frequency');
ylabel('|X(f)|/N');

print(gcf,'03b.png','-dpng','-r300');
```



- The electroencephalogram (EEG) is a recording of the electrical activity of the brain from the scalp. The recorded waveforms reflect the cortical electrical activity.
- Delta band tends to be highest in amplitude and slowest wave
- Presence of all the three bands is confirmed from the DFT of the signals

Part 3

Code:

```
[signal,fs]=audioread('fivewo.wav');
N=length(signal);
nb=[1,2,3,4,8,16];
f1=90;
f2=5760;
%Random noise
noise=rand(1,N);
sig=zeros(6,N);
for p=1:6
n=nb(p);
k=nthroot(f2/f1,n);
band=zeros(1,n+1);
%Creating the edge frequencies for given number of bands
for i=1:n+1
band(i)=f1*(k^(i-1));
end
bandsig=zeros(n,N);
noisebp=zeros(n,N);
%bandsig:Filetered component of the signal in respective bands
%noisebp:Filetered component of the noise in respective bands
for i=1:n
[b, a]=butter(2,[band(i) , band(i+1)].*(2/fs),'bandpass');
bandsig(i,:)=filtfilt(b,a,signal);
noisebp(i,:)=filtfilt(b,a,noise);
end
env=zeros(n,N);
%Envelop of bandpass filtered components in respective bands
for i=1:n
y=hilbert(bandsig(i,:));
env(i,:)=abs(y);
end
%Designing lowpass filter with fcutoff=240Hz: 2pi*240/fs
fcut=240*2/fs;
```

```
[blow,alow]=butter(2,fcut,'low');
envlow=zeros(n,N);
envnoise=zeros(n,N);
%envlow:Lowpass filtered componentof envelop in respective
bands
%envnoise:Envlow multiplied elementwise with respective
bandpass
%component of noise
for i=1:n
envlow(i,:)=filtfilt(blow,alow,env(i,:));
envnoise(i,:)=envlow(i,:).*noisebp(i,:);
end
%Final ouput signal by elementwise addition of envnoise
for j=1:N
for i=1:n
sig(p,j)=sig(p,j)+ envnoise(i,j);
end
end
end

%Scaling by multiplying with (number of bands)^0.5
for p=1:6
sig(p,:)=(sig(p,:))*sqrt(nb(p));
end
%Converting the signals into audio files
audiowrite('audio_nb=1.wav',sig(1,:),fs);
audiowrite('audio_nb=2.wav',sig(2,:),fs);
audiowrite('audio_nb=3.wav',sig(3,:),fs);
audiowrite('audio_nb=4.wav',sig(4,:),fs);
audiowrite('audio_nb=8.wav',sig(5,:),fs);
audiowrite('audio_nb=16.wav',sig(6,:),fs);
```

- Irrespective of number of bands the audio signals was observed to be attenuated in all the cases.
- The envelop signal extracted in each band is used to modulate the white noise that is spectrally limited by the same bandpass filter.

- The audibility of the signal obtained by this process is somewhat low when compared to that of the original signal.
- The intelligibility of the signal is mainly affected by the quality of the signal.
- 8 bands are sufficient to clearly understand the sound.
- There wasn't much improvement in the signal quality when the number of bands is increased from 8 to 16.
- The signal quality was not that good when number of bands are 1,2,3,4 and the words of the speech were not clearly understandable.
- Thus we can conclude that the low frequency temporal structure of speech and frequency content of speech is sufficient for speech recognition to a greater extent.

Link to audios :
https://drive.google.com/open?id=18zpNH6ejpRXOOh_uv2K3Keh98VoF3ZD4