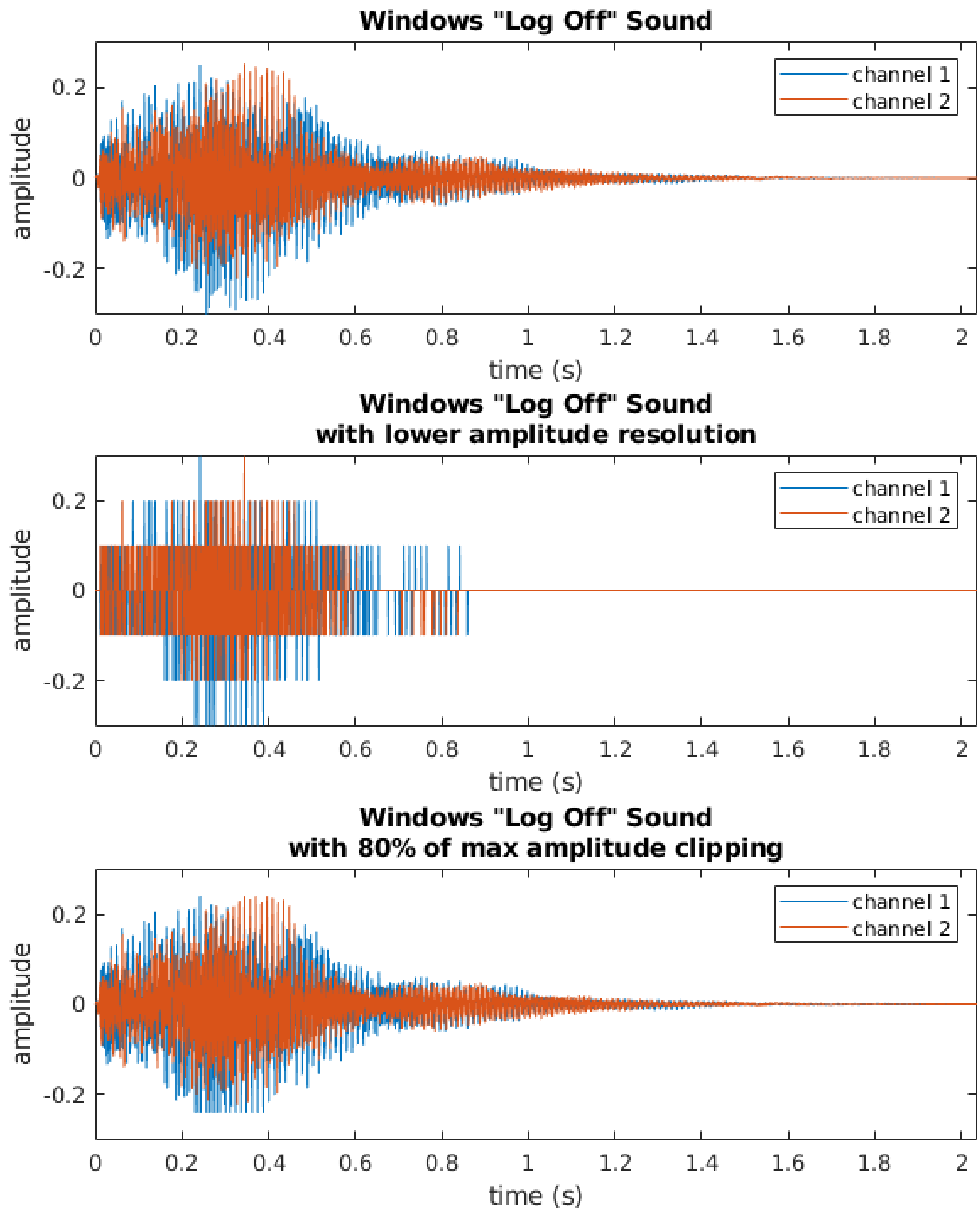
Assignment 3 - Dimitar Dimitrov - s1018291

3.2 Clipping

```
[logoff_sound,sampleRate] = audioread('logoff.wav');
high = max(abs(logoff_sound), [], 'all');
%sound(logoff_sound, sampleRate);
duration = (length(logoff_sound) - 1)*1/sampleRate;
t = 0:1/sampleRate: duration;

subplot(3,1,1);
plot(t, logoff_sound);
xlim([0 duration]);
ylim([-1*high high]);
title('Windows "Log Off" Sound');
legend('channel 1', 'channel 2');
xlabel('time (s)');
ylabel('amplitude');

lowampres = round(logoff_sound * 10)/10;

subplot(3,1,2);
plot(t, lowampres);
title([{'Windows "Log Off" Sound'},{'with lower amplitude resolution'}]);
xlim([0 duration]);
ylim([-1*high high]);
legend('channel 1', 'channel 2');
xlabel('time (s)');
ylabel('amplitude');
% the amplitude seems to be using only 7 values (6 steps)
%sound(lowampres, sampleRate);
lowerampres = round(lowampres * 10)/10;
%sound(lowerampres, sampleRate);
% during the first downscaling of the resolution the amplitudes were
% assigned to their closest value (and it had only 7 values) which means a
% lot of sounds were assigned to a value higher than their original one
% during the rounding. Now those already higher values are amplified again
% which produces an even louder soundwave (larger amplitudes).

A_max = max(abs(logoff_sound), [], 'all');
clipped_sound = logoff_sound;
clip_factor = 0.8;
clipped_sound(clipped_sound> clip_factor*A_max) = clip_factor*A_max;
clipped_sound(clipped_sound< -1*clip_factor*A_max) = -1*clip_factor*A_max;

subplot(3,1,3);
plot(t, clipped_sound);
title([{'Windows "Log Off" Sound'},{'with 80% of max amplitude clipping'}]);
xlim([0 duration]);
ylim([-1*high high]);
legend('channel 1', 'channel 2');
xlabel('time (s)');
ylabel('amplitude');
%saveas(gcf,'windows_logoff.png');
%sound(clipped_sound, sampleRate);

% at 80% I personally can't hear the difference but plotting the sound
% makes it obvious that the sound never goes beyond a particular amplitude

% as we lower the amplitude the sound gets quiter, which makes sense since
% we've decreased the amplitude which is effectively the loudness
```

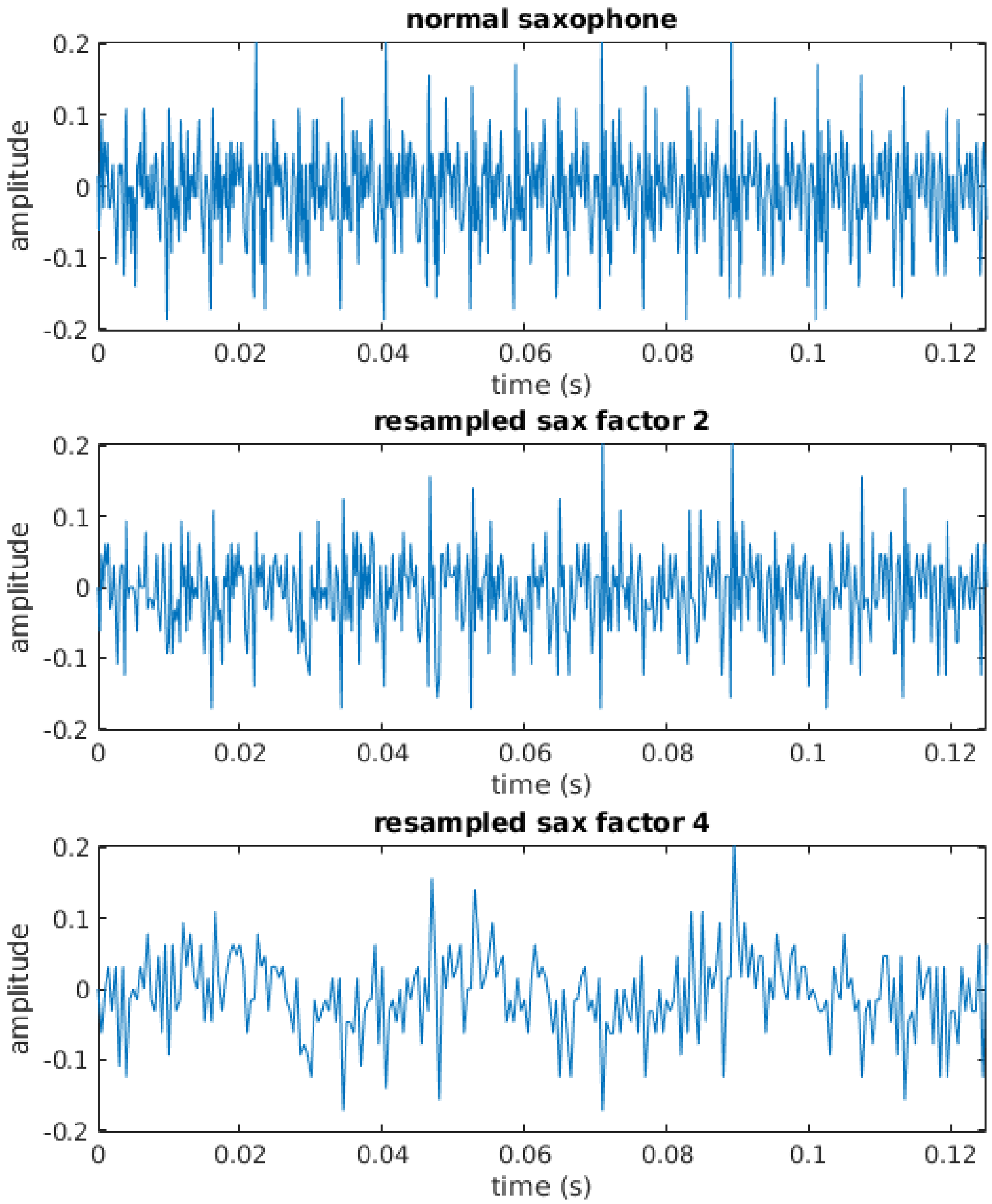


3.3 Sound

3.3.1 Saxophone

```
[sax, Fs] = audioread('Saxophone.wav');
% the shape of sax is 70000;1 so it is a mono recording
% the sampling frequency is 8000hz (8khz)
% nyquist frequency of this clip = 8000/2 = 4000hz (4khz)

% sound(sax, Fs)
nyq_sax = sax(1:2:length(sax));
nyq_Fs = Fs/2;
% sampling frequency is now 4000hz

%sound(nyq_sax,nyq_Fs);
% the sound is muffled

q_sax = sax(1:4:length(sax));
q_Fs = Fs/4;
% sampling frequency is now 2000hz
```

```
% sound(q_sax, q_Fs);
% the sound is even more muffled and particular segments sound like a
% vibration/rattling rather than a tone

section = sax(24000:25000);
section2 = nyq_sax(24000/2:25000/2);
section4 = q_sax(24000/4:25000/4);

ts = 0:1/Fs:(length(section) - 1) *1/Fs;
ts2 = 0:1/(Fs/2):(length(section2) - 1)*1/(Fs/2) ;
ts4 = 0:1/(Fs/4):(length(section4) - 1)*1/(Fs/4) ;

high = max(abs(section));
high2 = max(abs(section2));
high4 = max(abs(section4));

subplot(3,1,1);
plot(ts, section);
title('normal saxophone');
xlim([0 ts(end)]);
xlabel('time (s)');
ylabel('amplitude');
ylim([-1*high high]);

subplot(3,1,2);
plot(ts2, section2);
title('resampled sax factor 2');
xlim([0 ts2(end)]);
xlabel('time (s)');
ylabel('amplitude');
ylim([-1*high2 high2]);

subplot(3,1,3);
plot(ts4, section4);
title('resampled sax factor 4');
xlim([0 ts4(end)]);
xlabel('time (s)');
ylabel('amplitude');
ylim([-1*high4 high4]);

saveas(gcf, 'sax_comparison.png');

% we can see that with quarter the original sampling rate a sinusoid is
% starting to form where there wasn't one before, which is a sign of
% aliasing taking place

[newS,newFs] = downsampleSound(sax,Fs,7);
% sound(newS, newFs);
% with factor 7 it's already hard to tell, with factor 8 it could be
% anything
```



3.3.2 Downsample Sound

```
function [newSound, newFs] = downsampleSound(soundMatrix, sampleRate, downsampleFactor)
% function [new_sound, newFs] = downsampleSound(soundMatrix, sampleRate, downsampleFactor)
% downsamples a sound according to a given factor
% takes a:
% soundMatrix - matrix of samples comprising the sound
% sampleRate - the sample rate of that matrix (integer, samples per second)
% resampleFactor - factor by which the sampling will be divided/degraded
% and returns:
% newSound - the resampled versions of the sound matrix
% newFS - the new sampling rate (integer, samples per second)
% downSampe(A,B,2) returns newSound, a sound matrix that has half the samples
% of A and newFs, the new sampling rate, equal to B/2.

newSound = soundMatrix(1:downsampleFactor:length(soundMatrix));
newFs = round(sampleRate/downsampleFactor);
```

3.5 EEG data

```
load('EEGdata.mat');
plot(t, EEGdata);
xlim([t(1) t(end)]);
% there should be an amplitude spike shortly after a stimulus
ind_event = find(event==1);
% ind_event has length 200 so the stimulus was presented 200 times
between = diff(ind_event);
% there's different amount of samples between events
min_samples = min(between);
% the minimum amount of samples between 2 consecutive events is 292
EEGsorted= [];
for i=1:200
        v = EEGdata(ind_event(i):ind_event(i)+min_samples);
        EEGsorted = [EEGsorted v];
end
EEGmean = mean(EEGsorted, 2);
sampleLength = mean(diff(t)); % get the average temporal difference between samples
time = 0:sampleLength:sampleLength*(min_samples);
plot(time,EEGmean)
title('mean EEG signal accross 200 trials')
xlabel('time (s)');
xlim([time(1) time(end)]);
ylim([min(EEGmean) max(EEGmean)]);
ylabel('amplitude (v)');
%saveas(gcf, 'mean_eeg_time.png');

% we can see a positive ERP right before the 50th sample, followed by a
% negative potential at around the 80th sample, and another positive
% potential at around 130th sample.
% after looking at the t vector it seems it's sampling once every 0.0042
% seconds, so after converting to that timeline it seems that the potentials
% occur at respectively ~200ms, ~350-400ms and ~550ms which are likely
% components P200 (aka P2), N400 and P600
% the latter two are associated with grammatical error/unexpected words
% if I recall correctly
```