EEE391 - Matlab Assignment 1 Report

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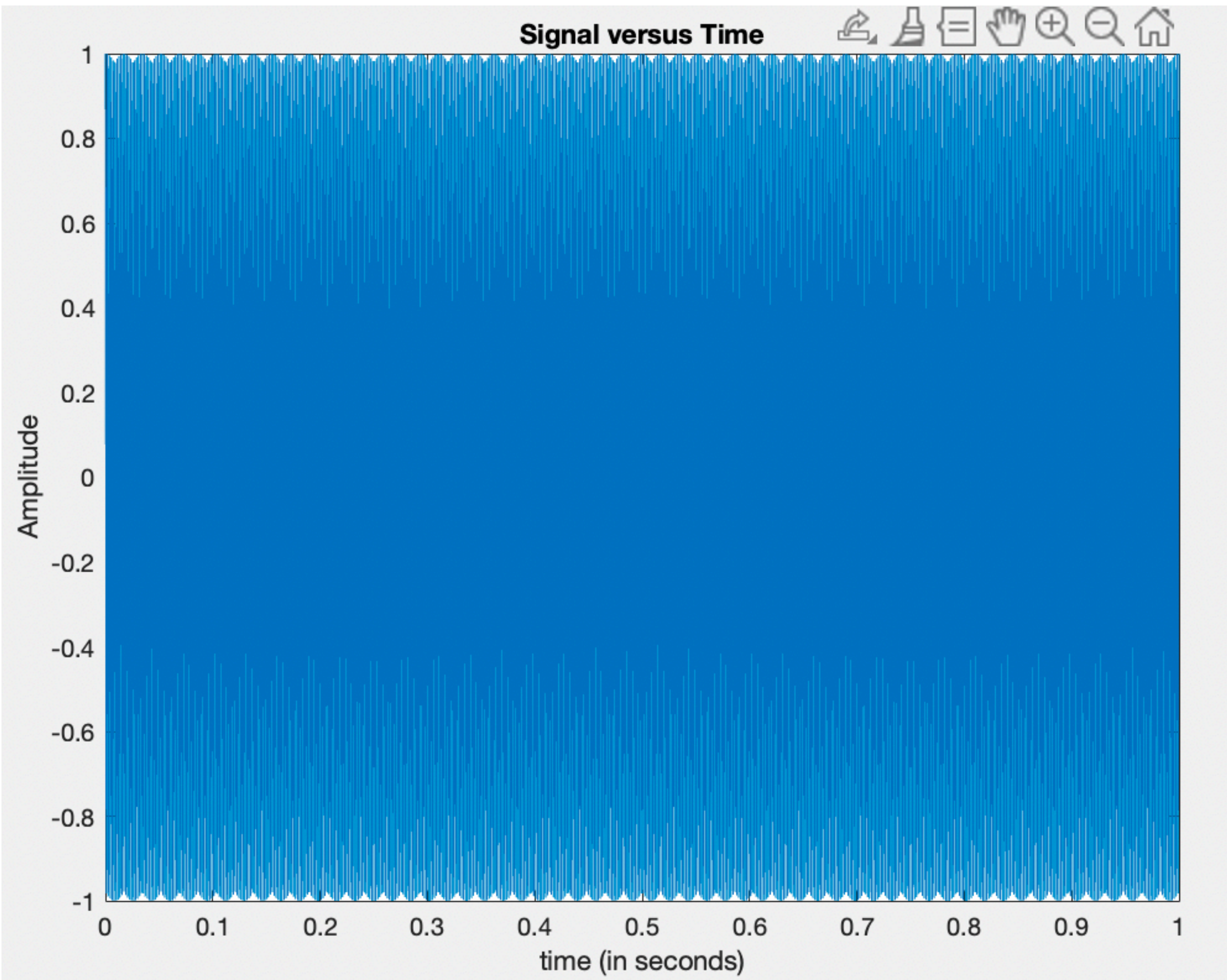
EEE391-1

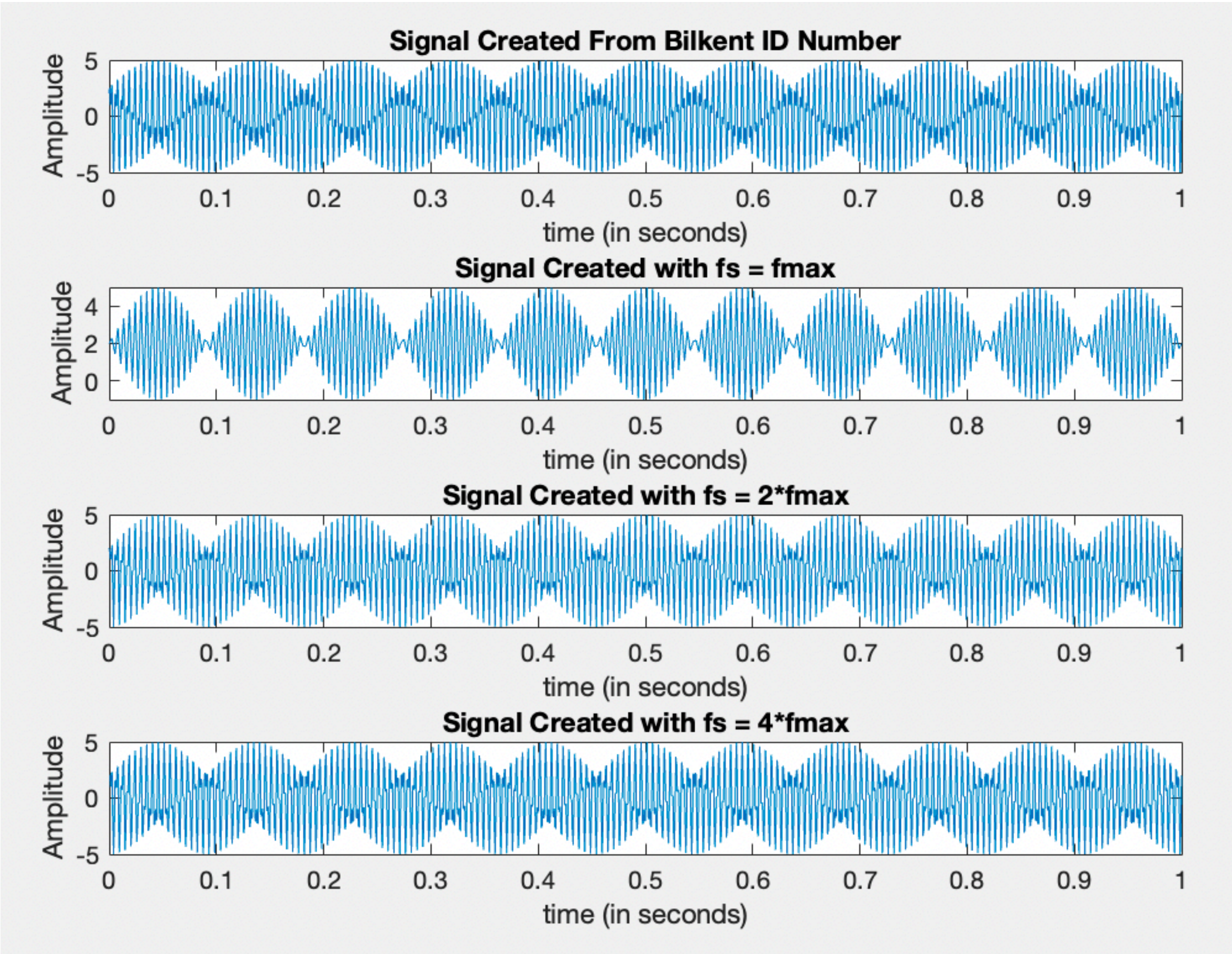
Part A:

Part 1-2-3:



When going from part 1 to part 2, the sound got thicker as the frequency of the sound decreased. Then, as the frequency of the sound increased while going from part2 to part3, the sound became thinner.

Part 4:



Appendix:

```
%1
f0 = 550;
t = 0 : 1/8192 : 1;
x1 = cos(2*pi*f0*t);
figure;
plot(t,x1);
xlabel('time (in seconds)');
ylabel('Amplitude')
title('Signal versus Time');
%soundsc(x1);

%2
f0 = 430;
t = 0 : 1/8192 : 1;
x1 = cos(2*pi*f0*t);
```

```
%soundsc(x1);

%3
f0 = 750;
t = 0 : 1/8192 : 1;
x1 = cos(2*pi*f0*t);
%soundsc(x1);
%}
%4
f1 = 196;
f2 = 190;
t = 0 : 1/8192 : 1;
y = 2*cos(2*pi*f1*t) + 3*sin(2*pi*f2*t);
%figure;
%subplot(4, 1, 1)
%plot(t , y);
%xlabel('time (in seconds)');
%ylabel('Amplitude')
%title('Signal Created From Bilkent ID Number');

fmax = max([f1, f2]);
sr1 = fmax;
dt1 = 1 / sr1;
t1 = 0 : dt1 : 1;
y1 = 2*cos(2*pi*f1*t1) + 3*sin(2*pi*f2*t1);
%subplot(4, 1, 2)
%plot(t1 , y1);
%xlabel('time (in seconds)');
%ylabel('Amplitude')
%title('Signal Created with fs = fmax');

sr2 = 2 * fmax;
dt2 = 1 / sr2;
t2 = 0 : dt2 : 1;
y2 = 2*cos(2*pi*f1*t2) + 3*sin(2*pi*f2*t2);
%subplot(4, 1, 3)
%plot(t2 , y2);
%xlabel('time (in seconds)');
%ylabel('Amplitude')
%title('Signal Created with fs = 2*fmax');

sr3 = 4 * fmax;
dt3 = 1 / sr3;
t3 = 0 : dt3 : 1;
y3 = 2*cos(2*pi*f1*t3) + 3*sin(2*pi*f2*t3);
%subplot(4, 1, 4)
%plot(t3 , y3);
%xlabel('time (in seconds)');
%ylabel('Amplitude')
%title('Signal Created with fs = 4*fmax');
```

Part B:

Part 2:

The sound of piano note E4 is saved and it is imported to Matlab. The fundamental frequency of note E4 is 329.63. Therefore, its fundamental period is 0.0030 second.

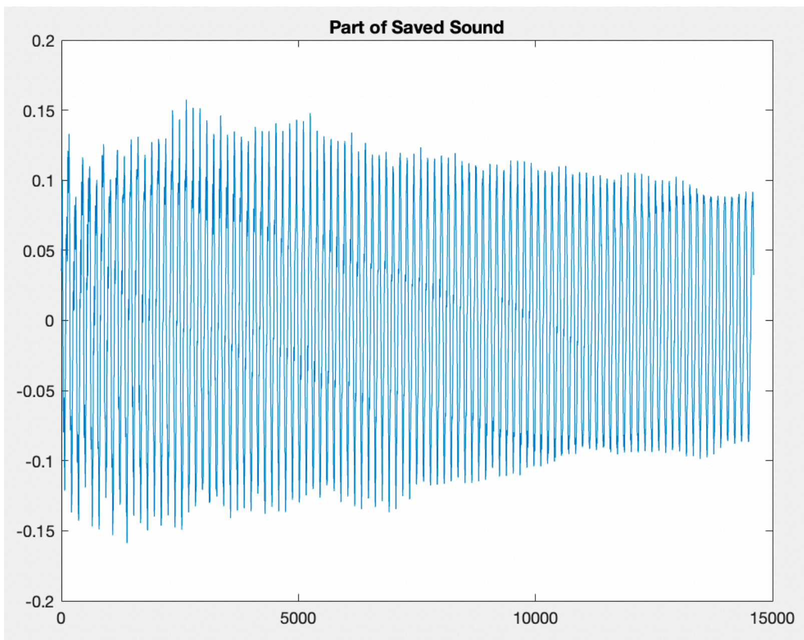


Figure 1: Part of saved note

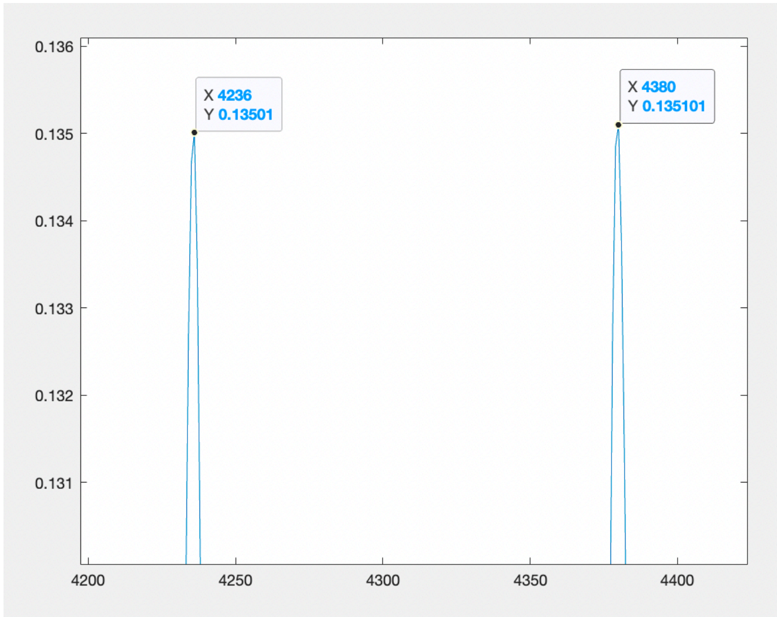


Figure 2: One period of note

By looking the figure 2, it is concluded that one period of note contains 144 points. Since the sampling frequency is 48000, 144/48000=0.003 as it is calculated theoretically.

Part 3:

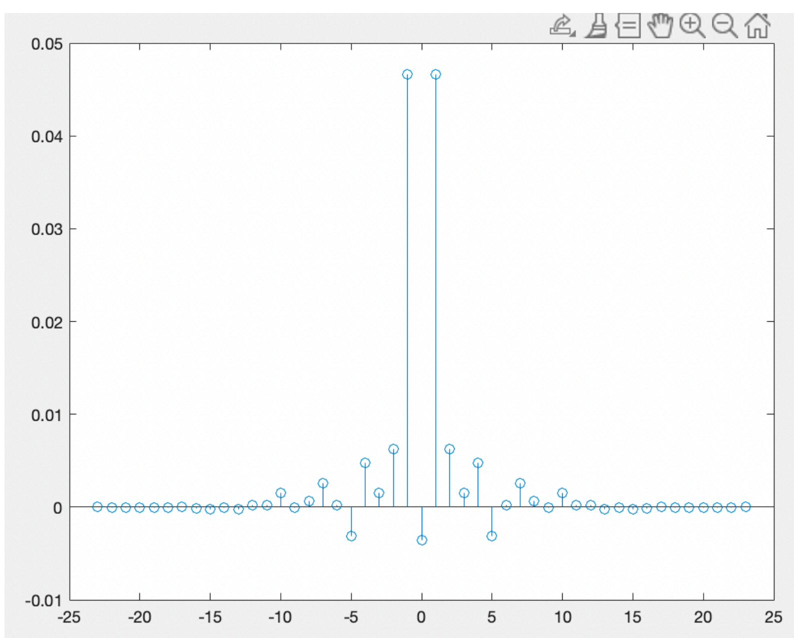


Figure 3: Fourier Series Coefficients of note

Part 4:

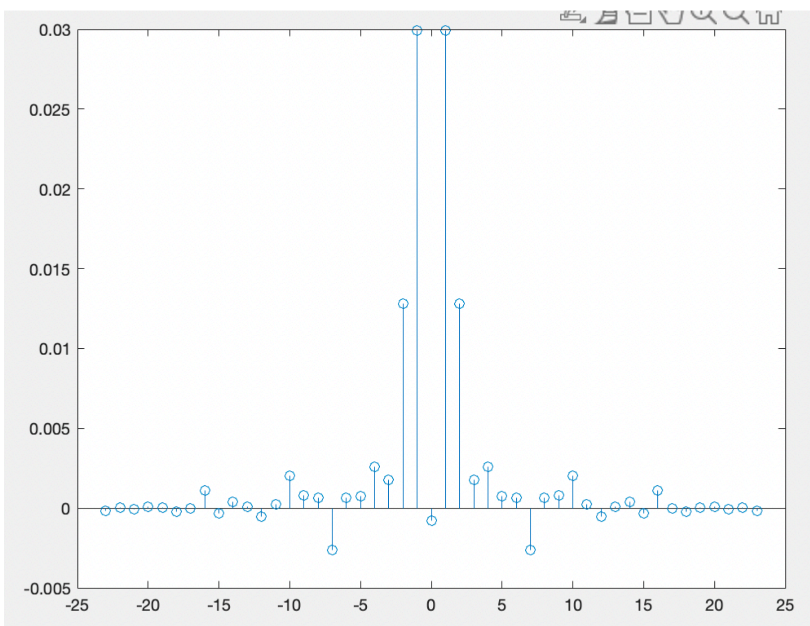


Figure 4: Fourier Series Coefficients of E4 in guitar

Although, they are the same note, there are differences in the amplitudes of the Fourier coefficients. The reason of the difference may be the tone of the instruments.

Part 5:

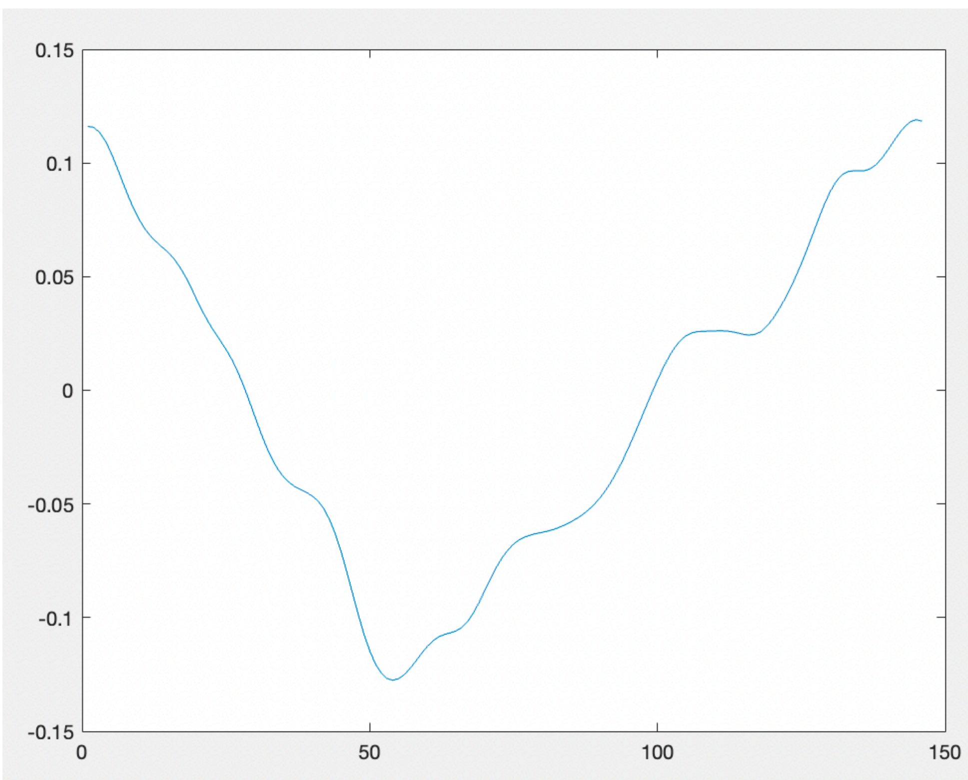


Figure 5: Re-synthesized signal

After reproducing signal, same signal is obtained. By adding replicates of this signal, sound is generated again.

Appendix:

```
[soundArray,Fs] = audioread('sound.wav');
[gitar,Fs] = audioread('E4Piano.wav');
T_real = 1/329.63;
size=round(T_real*Fs);
partOfSoundArray = soundArray(48000:48000+size*100);
partOfSoundArray_gitar = gitar(48000:48000+size*100);
figure(1)
plot(partOfSoundArray_gitar)
title("Part of Saved Sound");

t = linspace(0,T_real,size);
firstPeriodOfSoundArray = partOfSoundArray(7293:7293+size-1);
firstPeriodOfSoundArray_gitar = partOfSoundArray_gitar(11523:11523+size-1);

figure(2)
plot(firstPeriodOfSoundArray);
FSC = zeros(47,1);
FSC_gitar = zeros(47,1);

for n=1:47

    FSC(n)= trapz(1/48000,firstPeriodOfSoundArray.*exp(-1i*2*pi*(n-24)*t/
T_real))/T_real;

end
for n=1:47

    FSC_gitar(n)=
trapz(1/48000,firstPeriodOfSoundArray_gitar.*exp(-1i*2*pi*(n-24)*t/T_real))/
T_real;

end
figure(3)
stem(linspace(-23,23,47),FSC)
figure(4)
stem(linspace(-23,23,47),FSC_gitar)

ReSound = zeros(size,1);

for k = 1:145
    for n=1:length(FSC)
        ReSound(k) =  ReSound(k) + FSC(n).*exp(1i*(n-24)*2*pi*t(k)/T_real);
    end
end
ReSound = real(ReSound);
re_sound = repmat(ReSound,100,1);
sound(partOfSoundArray)
figure(5)
plot(firstPeriodOfSoundArray)
```