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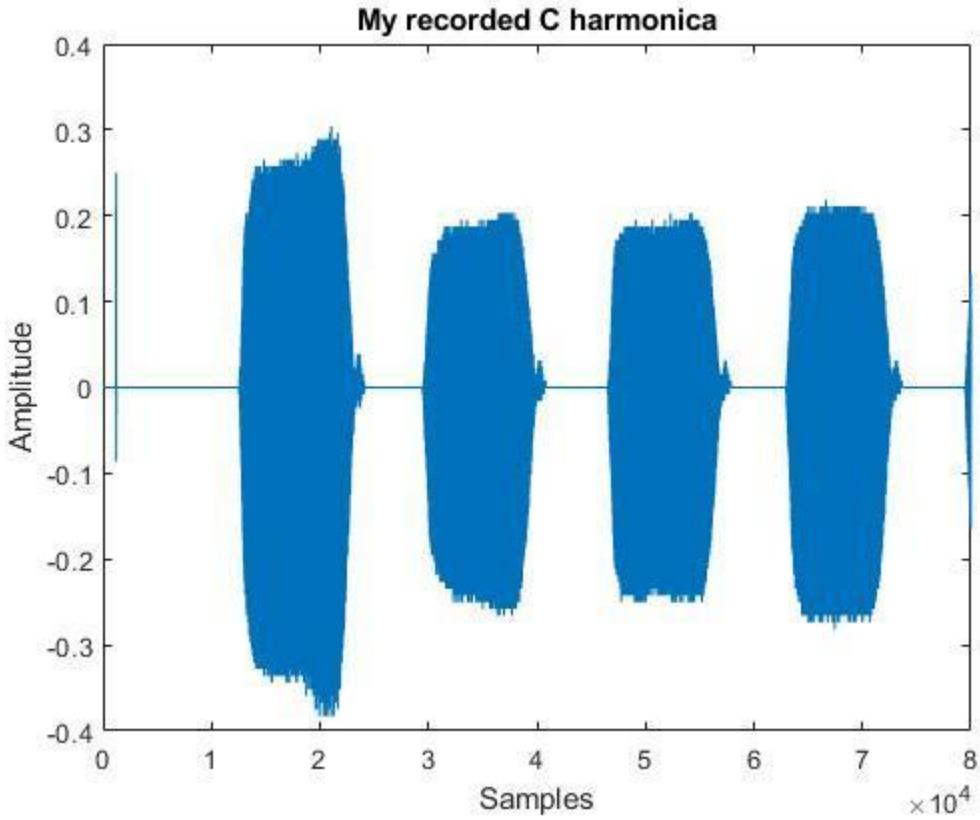
Section 1

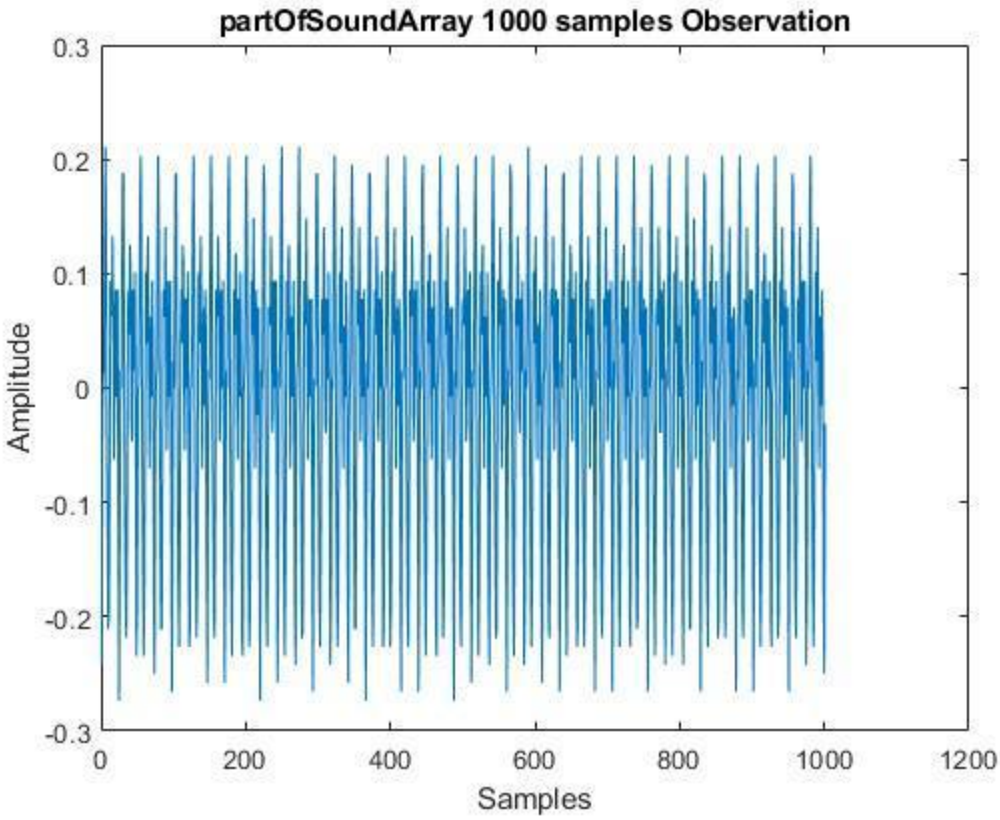
EEE 391 MATLAB ASSIGNMENT 1



This is the plot that was regenerated from the recording scripts in Part1. I recorded these soundwaves with my C harmonica. Below was my code in comments now:

```
%{
recObj = audiorecorder(16000, 8, 1);
disp("Start recording.");
recordblocking(recObj, 5);
disp("End of Recording.");


play(recObj);
%}
```

Extracting partOfSound



This plot shows a part of the sound I've created beforehand with my C harmonica. I decided to take 1000 samples because the fundamental frequency of the note I played was 659.25, therefore there are at least 100 periods here in this graph.

Fourier Series Analysis:

The fundamental frequency of my recorded E note is theoretically 659.29 Hz. Multiplying fundamental period with sampling period gives us the sample count in one period. In my case, it's approximately 24.37. However, due to the imperfect recording, I can observe a period in 10 samples. Therefore I miscalculated, because of environmental errors I believe, the fundamental period as 0.00063, nearly as twice as lower. When I extend the signal to the larger one by using the graphical fundamental frequency, I obtain a G sound whereas when I use the theoretical frequency, I obtain a sound which is just a little bit more flat than the original one. I can distinguish that because I think I have a high-pitched musical ear. However, while using the theoretical fundamental frequency, I see 2.5 crests % troughs (periods) which is not so strange because recording a perfect signal is quite hard. I plotted the FSC by using the absolute values of coefficients. The following were my code and plots for this part and extracting partOfSound:

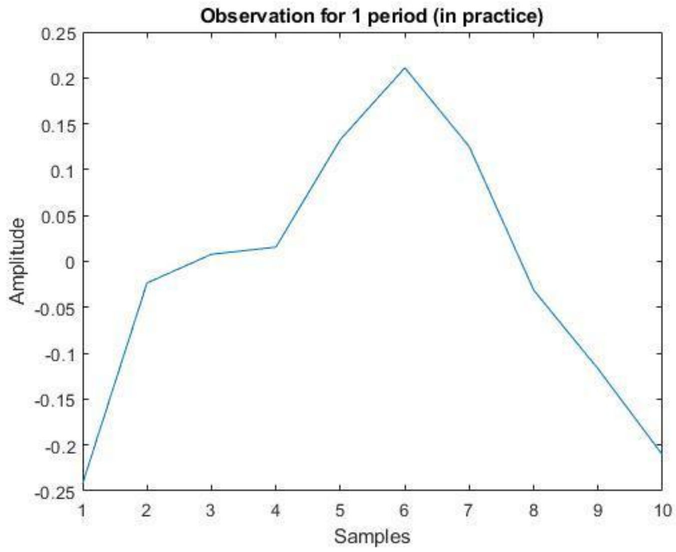


figure 1

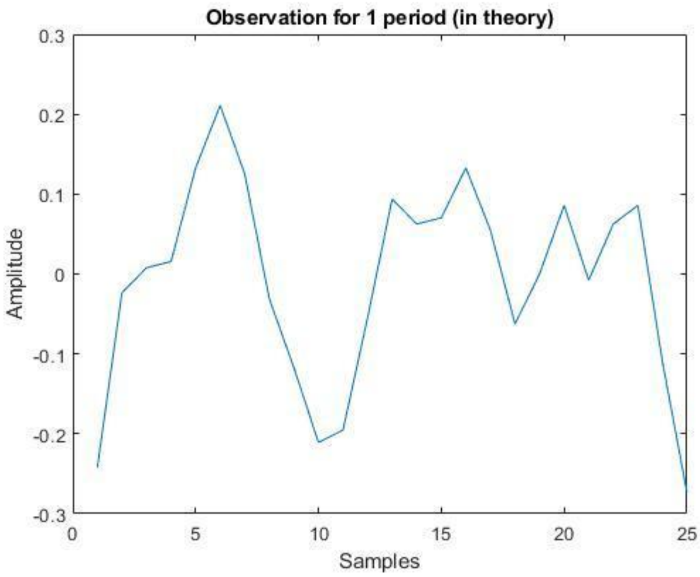


figure 2

```
fundamentalFreq = 659.25;
fundamentalPeriod = 1 / 659.25;
sampleRate = 16000;
samplingPeriod = 1 / 16000;
wavelength = 52.33;
noOfSamples = fundamentalPeriod * sampleRate;
sample_count_oop = fundamentalPeriod / samplingPeriod;
sampleCountInOnePeriod = ceil(sample_count_oop);
graphicalFundamentalPeriod = 25;

firstPeriodOfPartOfSoundArray =
partOfSoundArray(1:1:graphicalFundamentalPeriod);
```

```
figure;
plot(firstPeriodOfPartOfSoundArray);
title('Observation for 1 period (in theory)');
xlabel('Samples');
ylabel('Amplitude');

figure;
plot(partOfSoundArray(1:1:10));
title('Observation for 1 period (in practice)');
xlabel('Samples');
ylabel('Amplitude');



x = firstPeriodOfPartOfSoundArray.';

%quantizing time.
t = 0:1:graphicalFundamentalPeriod - 1;
TsOne = graphicalFundamentalPeriod;

ak = zeros(1,TsOne);


%evaluating the integral.

for k = -(TsOne-1):TsOne-1
    expTerm = exp(-j*(2*pi)*k*(t/TsOne));
    ak(k+TsOne) = trapz( (x.*expTerm) / TsOne);
    abs_ak = abs(ak);
end

% the interval -N,N
N = -(TsOne - 1):1: TsOne - 1;


figure;
stem(N, real(abs_ak))
title('Magnitude of Fourier Series Coefficients in One Period');
xlabel('|ak| coefficients')
ylabel('complex amplitude')
```

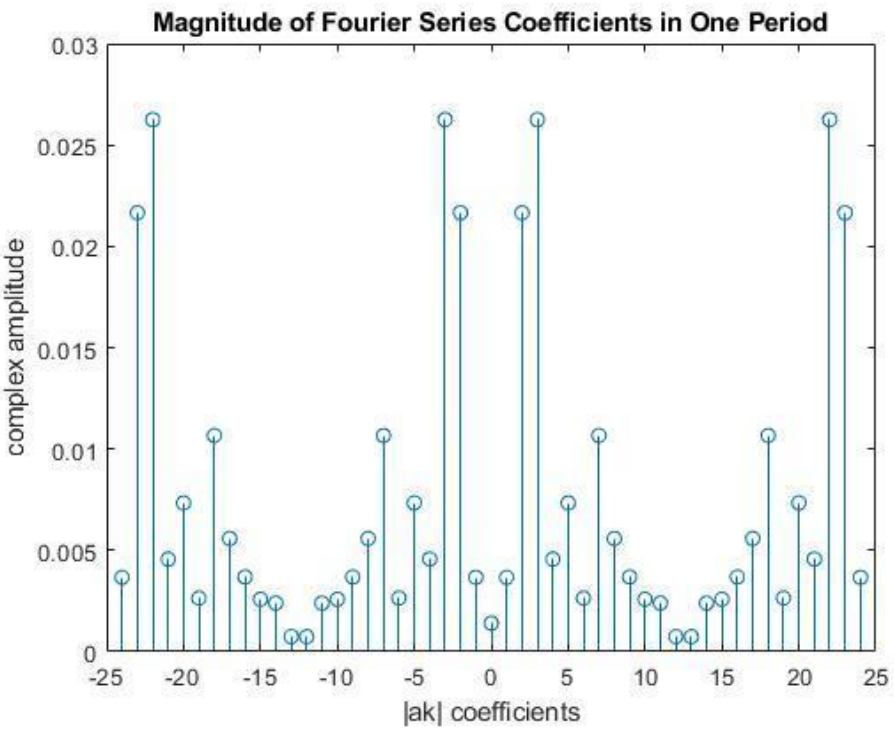
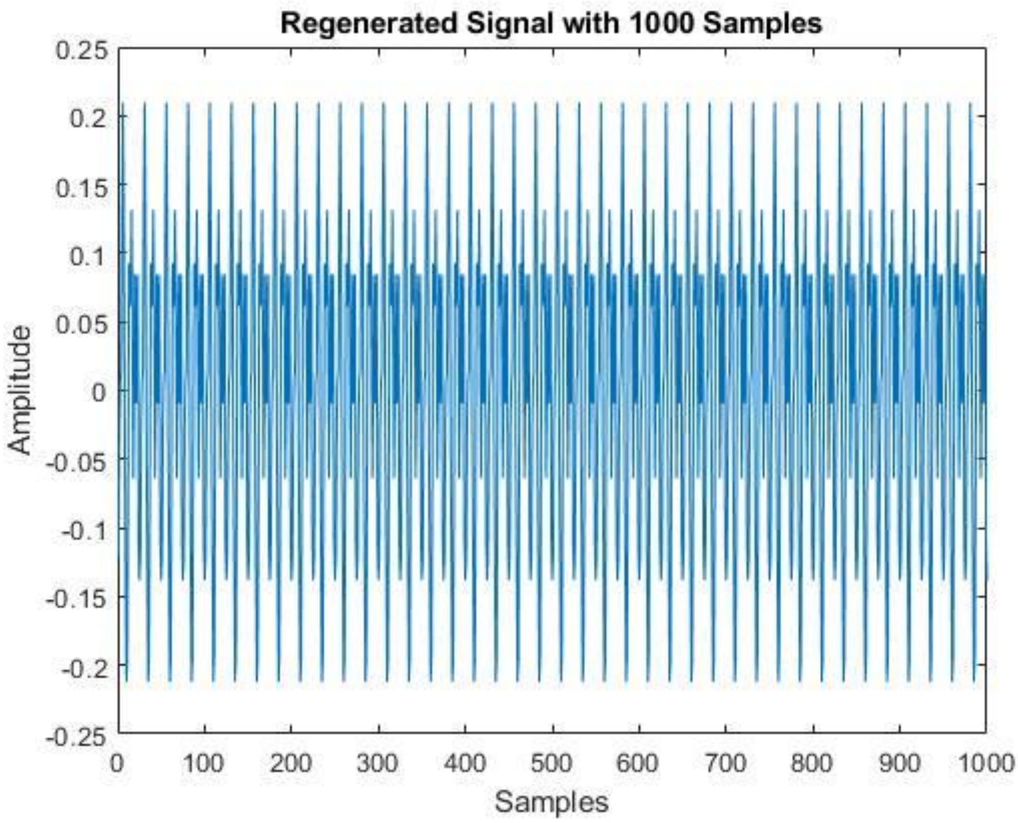


figure 3

Fourier Series Synthesis:

I have synthesized the signal back with the same amount of samples, with the theoretical fundamental frequency I obtained. The experimental theoretical frequency gave me a very low G sound. This one gave me a slightly lower sound than E5. Not even an Eb. Below is my plot my codes, also writing the audio file I generated:




```
t_second = 0:graphicalFundamentalPeriod/TsOne: graphicalFundamentalPeriod -
graphicalFundamentalPeriod/TsOne;
synthesizedSignal = zeros(size(t_second));
```

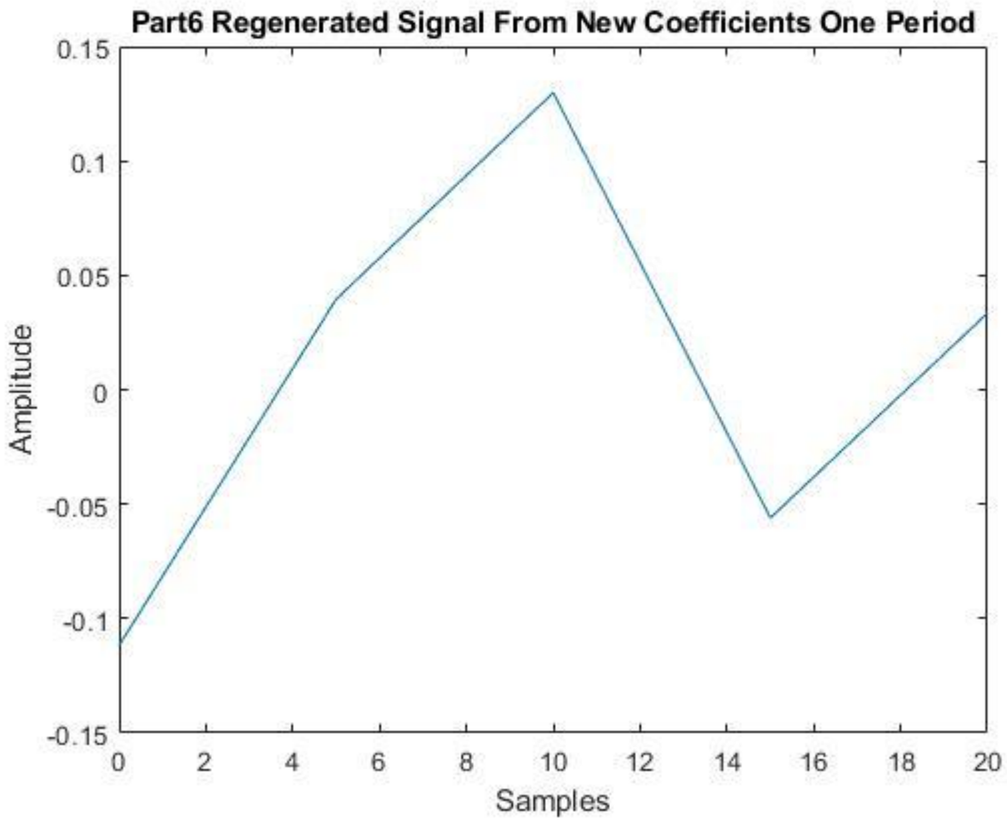
```
for k = -(TsOne-1):TsOne-1
   synthesizedSignal = synthesizedSignal + (ak(k+graphicalFundamentalPeriod)*
exp(k*1j*2*pi*(t_second/graphicalFundamentalPeriod)));
end
figure;
plot(t_second,real(synthesizedSignal)); % The one period signal

n = 40;
newArray = repmat(synthesizedSignal,1,n);
regeneratedGraph = real(newArray);
figure;
plot(regeneratedGraph);
title('Regenerated Signal with 1000 Samples');
xlabel('Samples')
ylabel('Amplitude')

% Now write the audiofile...
%sound(regeneratedGraph);
audiowrite('harmonica_reproduced.wav', regeneratedGraph, 16000);
%}
```

Fourier Series Synthesis With Known Coefficients:

In this part, if I did not get wrong, I created new bk's from ak's that are not zero. I took a small interval and extended it. I used to hear an E5 which was the usual note that I played from my harmonica, however after I did this part I heard a high pitched beep sound. My code and plot for the new signal for one period is below:

```
%PART 6

bk(6) = ak(25);        %This is ak(0).
bk(7) = ak(26);
bk(5) = ak(24);
bk(4) = ak(23);
bk(3) = ak(22);
bk(2) = ak(21);
bk(1) = ak(20);
bk(8) = ak(27);
bk(9) = ak(28);
bk(10) = ak(29);

TsTwo = 5;
t_second_2 = 0:graphicalFundamentalPeriod/TsTwo: graphicalFundamentalPeriod -
graphicalFundamentalPeriod/TsTwo;
reSynthesizedSignal = zeros(size(t_second_2));

for k = 1:10
   reSynthesizedSignal = reSynthesizedSignal + (bk(k)*
exp(k*1j*2*pi*(t_second_2/graphicalFundamentalPeriod)));
end
figure;
plot(t_second_2,real(reSynthesizedSignal)); % The one period signal
title('Part6 Regenerated Signal From New Coefficients One Period');
xlabel('Samples')
ylabel('Amplitude')


n = 40;
newArray2 = repmat(reSynthesizedSignal,1,n);
```

```
reSynthesizedSignal = real(newArray2);
%figure;
%plot(reSynthesizedSignal);
%title('Part6 Regenerated Signal From New Coefficients');
%xlabel('Samples')
%ylabel('Amplitude')

sound(reSynthesizedSignal);
```



Fourier Series Synthesis With Uniform Coefficients:

I used to hear an E5 which was the usual note that I played from my harmonica, however after I did this part I again heard a high pitched beep sound. My code and plot for the new signal for one period is below:

```
%Part 7

ck(6) = 1;          %This is ak(0).
ck(7) = 1;
ck(5) = 1;
ck(4) = 1;
ck(3) = 1;
ck(2) = 1;
ck(1) = 1;
ck(8) = 1;
ck(9) = 1;
ck(10) = 1;

TsTwo = 5;
```

```
      t_second_2 = 0:graphicalFundamentalPeriod/TsTwo:
graphicalFundamentalPeriod - graphicalFundamentalPeriod/TsTwo;
      reSynthesizedSignal = zeros(size(t_second_2));

      for k = 1:10
         reSynthesizedSignal = reSynthesizedSignal + (ck(k)*
exp(k*1j*2*pi*(t_second_2/graphicalFundamentalPeriod)));
      end
      figure;
      plot(t_second_2,real(reSynthesizedSignal)); % The one period signal
      title('Part7 Regenerated Signal From New Coefficients One Period');
      xlabel('Samples')
      ylabel('Amplitude')


      n = 40;
      newArray2 = repmat(reSynthesizedSignal,1,n);
      reSynthesizedSignal = real(newArray2);
      %figure;
      %plot(reSynthesizedSignal);
      %title('Part6 Regenerated Signal From New Coefficients');
      %xlabel('Samples')
      %ylabel('Amplitude')

      % Now write the audiofile...
      sound(reSynthesizedSignal);
```