

Umut Mücahit Köksaldı  
21402234  
EEE391 – 01  
13.11.2017

## Assignment 1

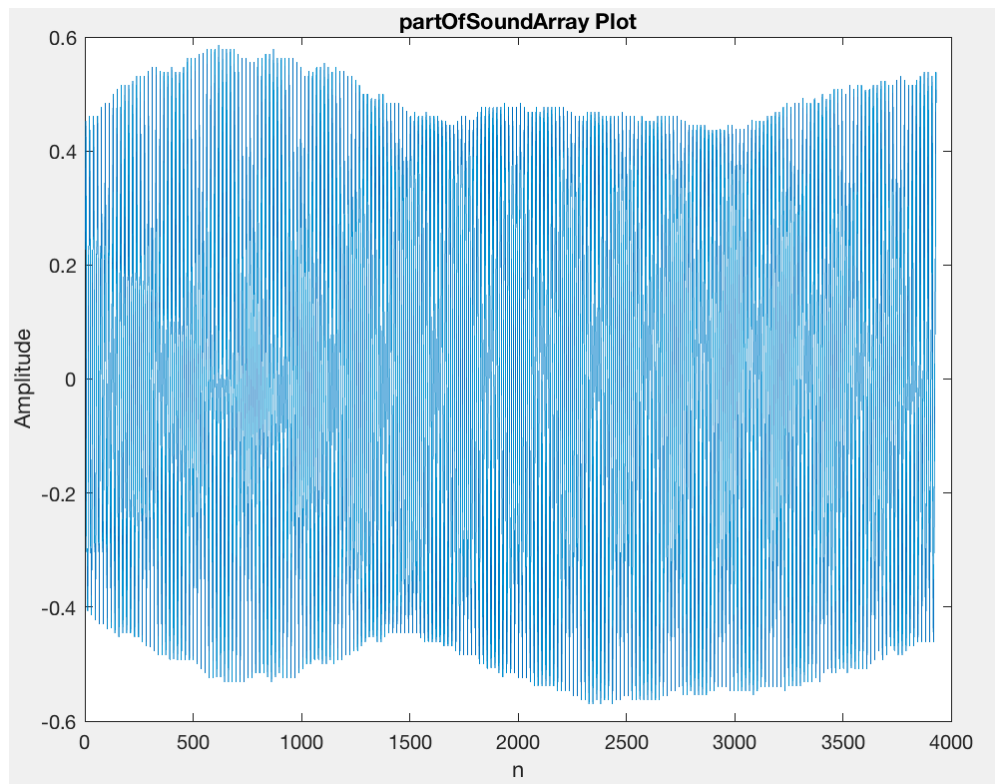
### 1. Voice Recording

I have recorded the A4 note on a real acoustic piano. I have used the following code segment to record the sound as an audiorecorder in MATLAB, and then extract the contents of the recorded signal to an array.

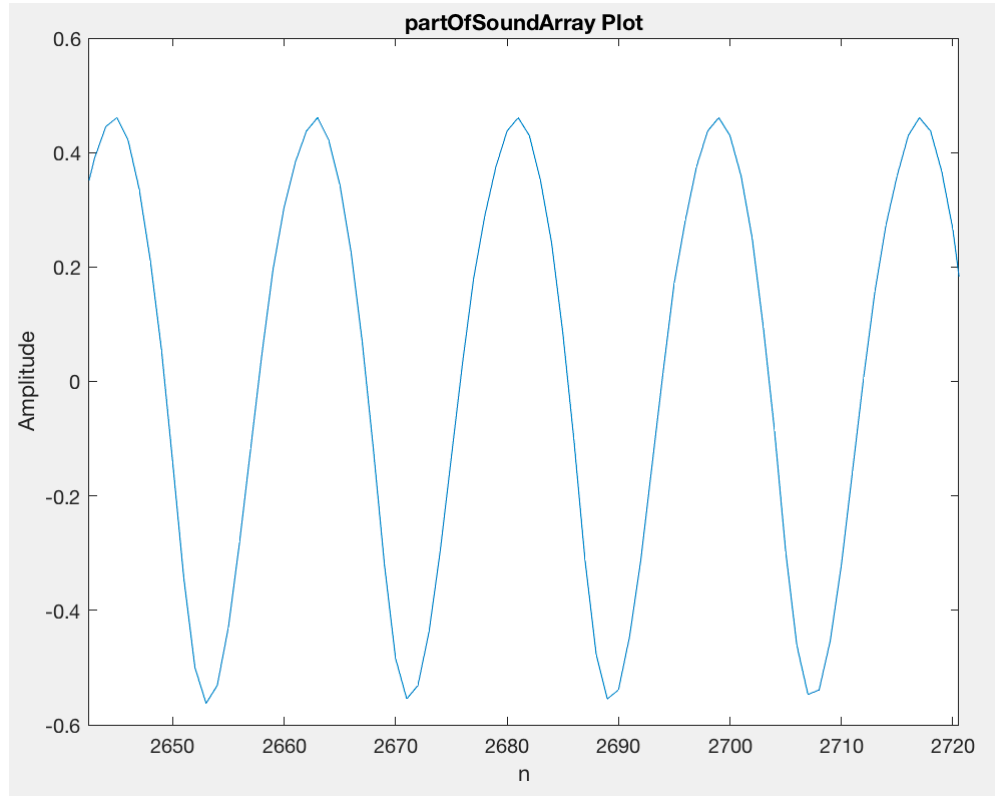
```
recObj = audiorecorder;  
disp("Start recording.");  
recordblocking(recObj, 6);  
disp("End of Recording.");  
soundArray = getaudiodata(recObj);
```

### 2. Fundamental Period Calculation

The note I have recorded is A4, which has a period of 440 Hz as evidenced by the website provided in the assignment description. The plot of partOfSoundArray is provided below.



Upon zooming in we can calculate the frequency and the period of the sampled sound by looking at the distance between two peak points. A plot of the zoomed in portion of partOfSoundArray is below.



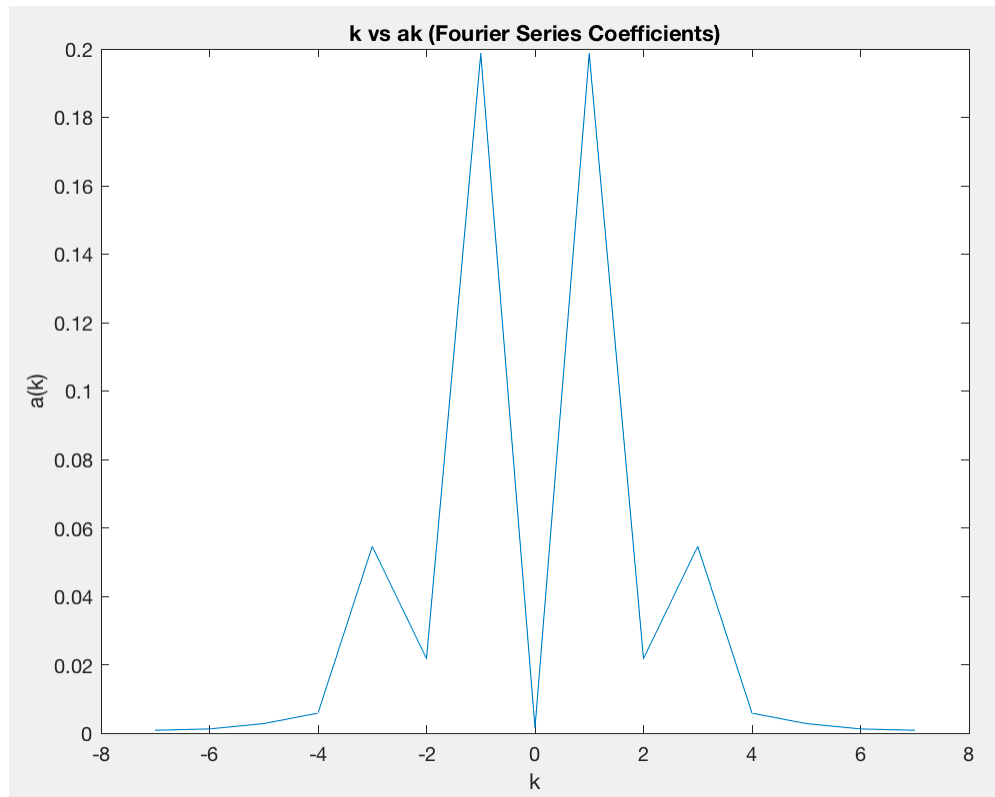
From this plot, we find that the distance between two peak points is 18 samples. Since the signal is sampled at a rate of 8Khz, we can find the fundamental frequency:

$$f_0 = \frac{8000}{18} = 444Hz$$

The result seems satisfactory considering that the listed frequency of A4 is 440 Hz.

### 3. Fourier Series Analysis

I have calculated the Fourier Series Coefficients of the sampled signal by utilizing the formula inside a for loop and using the trapz command of MATLAB to evaluate the integrals. N was chosen to be 7 upon observing that the Fourier Series Coefficients approached zero after the  $k = \pm 6$ , so it was determined that  $N=7$  was sufficient to calculate the non-zero coefficients. The plot of the fourier series coefficients and the code to generate it is as follows:



```

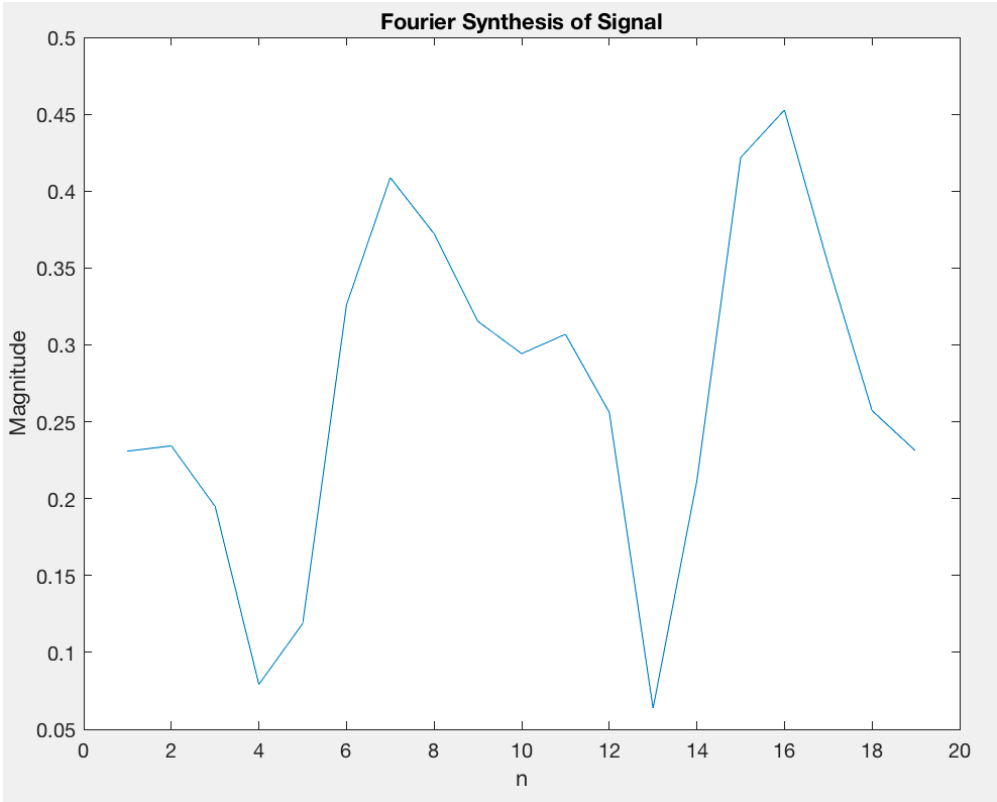
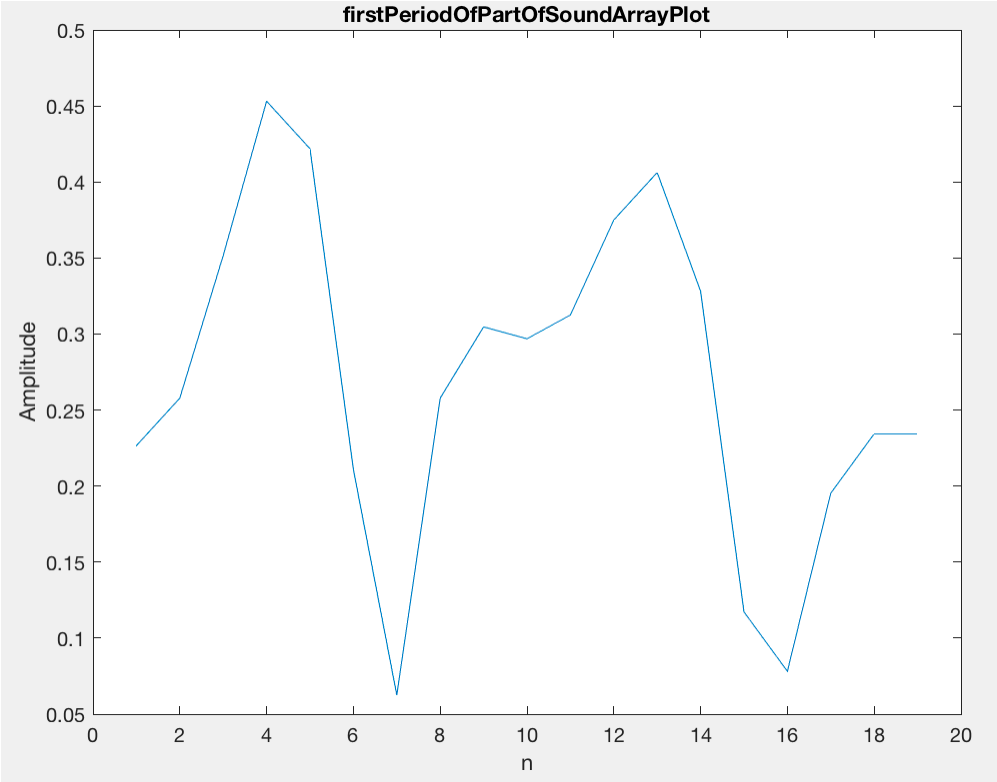
%% part 3: calculate fourier series coefficients
firstPeriodUpperBound = cast(period * 8000, 'int8');
firstPeriodOfPartOfSoundArray = partOfSoundArray(1:firstPeriodUpperBound+1);
plot(firstPeriodOfPartOfSoundArray);
t = 0:1/8000:period;
t_t = t.';
k = 7;
fcd = zeros(2*k+1, 1);
for m = -k:k
    fcd(m+k+1) = (trapz(t_t, exp(-
1i*2*pi/period*m.*t_t)*1/period.*firstPeriodOfPartOfSoundArray));
end

x = linspace(-k, k, 2*k+1);
y = abs(fcd);
plot(x, y);

```

#### 4. Fourier Series Synthesis

I have utilized the formula provided in the textbook inside a for loop in MATLAB to synthesize one period of the signal from the Fourier Series Coefficients calculated in part 3. The plots of both the original and the synthesized signal (first periods) are as follows.



As can be observed, the two plots look extremely similar, they look like reflections of one another along the y-axis. The result is satisfactory. The code used to synthesize and generate the plots are as follows.

```
%%% part 4: fourier series synthesis
x = zeros(19,1);
for m = 1:2*k+1
    y = fcd(m)*exp((-1i*2*pi/period)*(m-k-1)*t_t);
    x = x+y;
end
plot(abs(x));
```

## 5. Voice File Generation

The voice files for both the synthesized and the original partOfSoundArray were generated. Upon playback, both files sounded quite similar to one another, as expected. The code used to generate the audio files are below.

```
%%% part 5: voice file generation
w_signal = repmat(x, 200, 1);
audiowrite('fourierSynthesis.wav', abs(w_signal), 8000);
audiowrite('original.wav', abs(partOfSoundArray), 8000);
```

## 6. Synthesis of the Voice from Partial Fourier Series Coefficients

For the purposes of seeing the difference between each sound file, we will add another Fourier Series Coefficient each time starting from  $a(0)$ . Every time we include another coefficient in our calculation, the signal will change and exhibit different behavior. Since we have 13 coefficients that are non-zero (as can be observed from part 3), we will need to include 6 harmonics. The code to synthesize and generate the sound files is below.

```
%%% part 6: synthesis from partial fourier coefficients
for j = 0:6
    x = zeros(19,1);
    for m = (8-j):(8+j)
        y = fcd(m)*exp((-1i*2*pi/period)*(m-k-1)*t_t);
        x = x+y;
    end
    audiowrite(strcat('synthesis', j, '.wav'), abs(x), 8000);
end
```

In the initial iteration, the generated sound is silent since  $a(0) = 0$ . After adding the other Fourier Series Coefficients to the calculation, the signal gets gradually closer to the original value. However, the coefficients that do the most change are the initial ones  $a(1)$  and  $a(-1)$ .

## 7. Synthesis of the Voice from Fourier Series Coefficients whose Magnitudes are equated to one

In this section, we go through the array that contains the Fourier Series Coefficients and normalize them by dividing them to their absolute values. Then, as we have done in part 4, we synthesize the signal again and generate the audio file.

```
%%% part 7: Synthesis of theVoice fromFourier SeriesCoefficients
%%% whoseMagnitudes are equated to one
equated_fcd = zeros(15, 1);
for j = 1:length(equated_fcd)
    equated_fcd(j) = fcd(j) / abd(fcd(j));
end

x = zeros(19,1);
for m = 1:2*k+1
    y = equated_fcd(m)*exp((-1i*2*pi/period)*(m-k-1)*t_t);
    x = x+y;
end
audiowrite('equated.wav', repmat(abs(x), 200, 1), 8000);
```

## 8. Synthesis of the Voice from Fourier Series Coefficients whose Phases are equated to zero

For this section, we will calculate a new set of Fourier Series Coefficients which is generated from taking the absolute value of our initial coefficients (their magnitudes). Then we synthesize and generate the sound file as in part 4. Since the frequency of the file stayed the same and the only change was made to the phase, the sound is similar to the original signal, as expected.

```
%%% part 8: Synthesis of the Voice from Fourier Series Coefficients whose
Phases
%%% are equated to zero
absolute_fcd = abs(fcd);
x = zeros(19,1);
for m = 1:2*k+1
    y = absolute_fcd(m)*exp((-1i*2*pi/period)*(m-k-1)*t_t);
    x = x+y;
end
audiowrite('zeroed.wav', repmat(abs(x), 200, 1), 8000);
```

**All sound files discussed in this report are included in the submission mail.**