CMPE362 Homework 2

İrem Uğuz
2015400165

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1 Advanced Peak Finder

In this question, I designed a moving average filter to filter the signal. With different sample numbers to take average from, I compared how the number of peaks that are found changes.

I wrote a Matlab script for the filter. In the script, I am using some of the built in functions of Matlab: filter and findpeaks. I chose the use those functions because I believe that the built-in functions usually work more efficient than the functions that are written by hand.

General trend of the number of peaks that are detected is that the number of peaks decrease as the number of the samples that are averaged increases. The reason for this trend is quite understandable: If there are more than one local maxima in the averaged numbers, this local maximas are eliminated to one possible local maxima, as we are taking the average of some numbers that are neighbouring.

The number of local maximums that are found in the last homework's ExampleSignal.csv file versus the number of samples that the average is taking of(varying between 2 and 30) is shown in the plot below. The n=1 value is the un-filtered original verison of the Signal file.

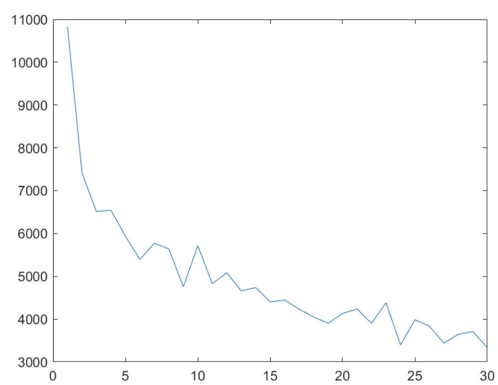

Figure 1: Graph of N versus number of peaks that are detected in filtered signal

2 Frequency (Pitch) of the Sound

2.1 Exercise 1

In this exercise, I was asked to quadruple the frequency of the data and play the data. I make the frequency quadruple by playing with the pitch by taking only one of the four samples. When I played the .wav file, I heard that the frequency increased greatly.

2.2 Exercise 2

In this exercise, I was asked to half the frequency of the data and play the data. I make the frequency halve by playing with the pitch by taking adding extra sample to the frequency. When I played the .wav file, I heard that the frequency decreased.

2.3 Exercise 3

In this exercise, I was asked to double the sampling frequency of the data and play the data. When I played the .wav file, I heard that the frequency increased as the sampling frequency increased.

2.4 Exercise 4

In this exercise, I was asked to halve the sampling frequency of the data and play the data. When I played the .wav file, I heard that the frequency decreased as the sampling frequency decreased.

2.5 Explanations

In the example part that was given in the question, the pitch of the wave is changed by taking the only even numbered samples. The resulting sound of that example was the same as the result of the doubled sampling frequency. This was an expected result as the both applications double the frequency of the sound that is heard.

Also, the result of the halving the sampling frequency and the halving the pitch was the same. The reason is the same as I explained above. Halving the sampling frequency means taking doubled number of samples which is the same as increasing the pitch of the data.

3 N-Tap Filter

In this question, I designed an N-Tap filter compare the values of SNR of original signal and the noisy signal. The original signal file is named mike.wav. The filter takes the signal in which the original signal plus the delayed version of the original signal exists.

I wrote a function named NTapFilter which takes a signal, k value, n value, alfa and the sampling frequency of the signal as parameter. The function basically takes the input signal and adds the delayed versions times exp(-alfa,n) to the original signal.

3.1 Changing Alfa Values

In this part of the question, the alfa values are varies between 0 and 1. The N and K are constant. The N values that is given to the filter is 10. And the K value is 100 millisecond, which is the default K value.

The effect of the alfa to the SNR value looks like a polynomial. At first the increase in the alfa decreases the SNR value, making it closer to the more positive. But after a point, the SNR value starts to decrease.

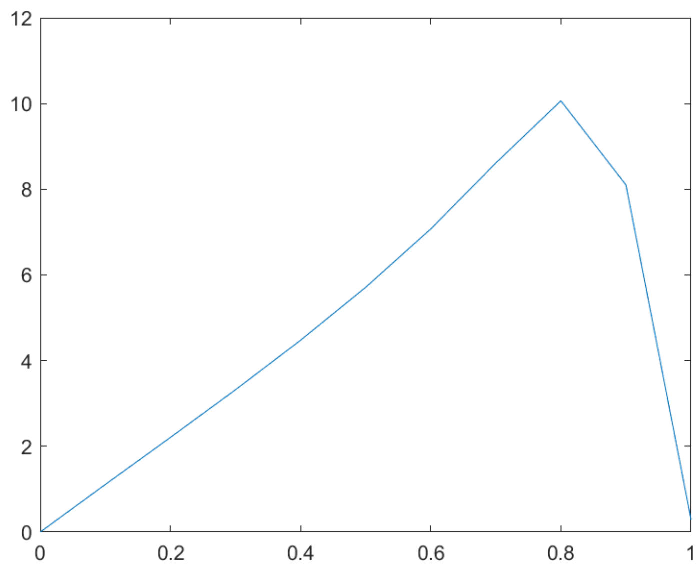

Figure 2: Alfa Values Versus SNR values

3.2 Changing N Values

In this part of the question, the N values are varies between 1 and 50. The alfa and K are constant. The alfa that is given to the filter is 0.7. And the K value is 100 millisecond, which is the default K value.

The effect of the N to the SNR value looks like a zigzag. At the first N values, the SNR value, changes drastically increasing. But after some point of the N, the N value stops its effect on the SNR and the found SNR value starts to be stable.

3.3 Changing K Values

In this part of the question, the K values are varies between 0.1 seconds and 0.4 seconds. The alfa and N are constant. The N that is given to the filter is 10. And the alfa value is 0.7.

Increasing the K, at first makes the SNR value lower. But at some point of K, the K value nearly stops affecting the SNR values.

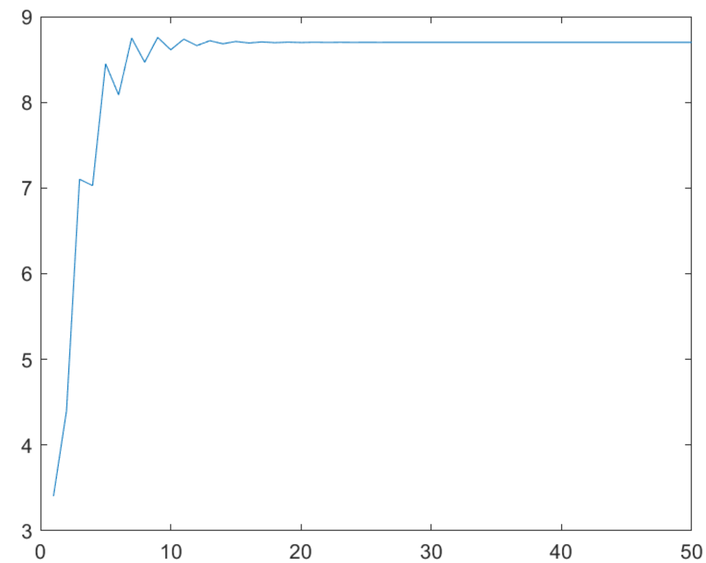

Figure 3: N Values Versus SNR values

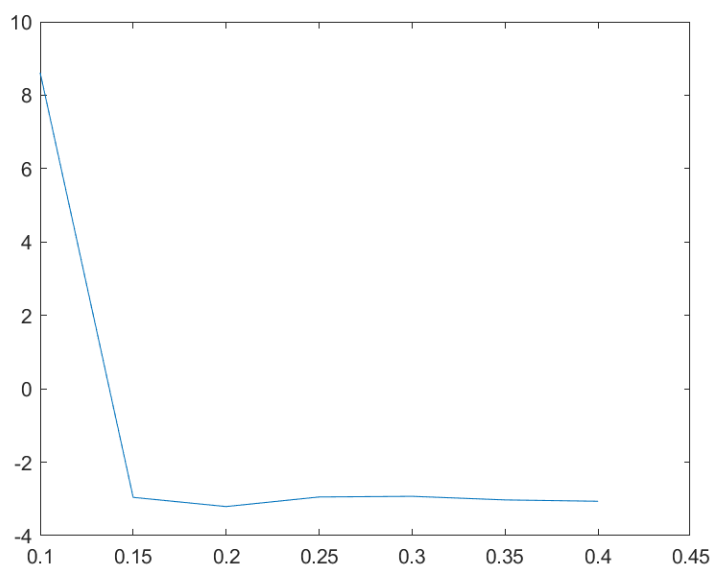

Figure 4: K Values Versus SNR values