CMPE 362 - Introduction to Signals for Computer Engineers
Homework 2

April 24, 2020

1 Question 1 - Fourier Series

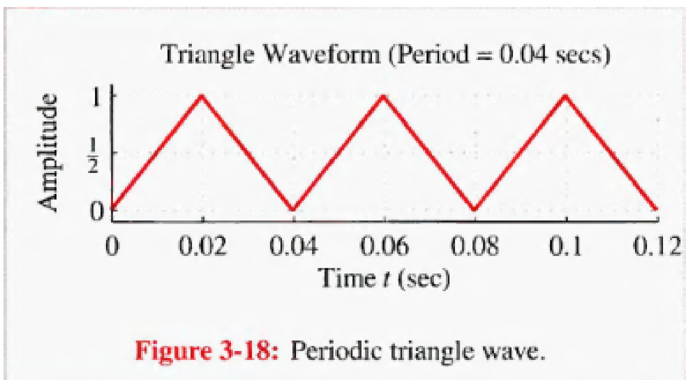

Figure 1: Wave

1.1 Part A

First, you have to find Fourier series coefficients for the given signal in figure 1. Please write on a piece of paper clearly without skipping any steps. You can include the photo of the paper in your report.(You can also do these calculations using LaTeX or any equation program.)

Please write the values for a_k when k is 0,even and odd and draw the frequency spectrum of the signal.

1.2 Part B

Construct this triangular wave in figure 1 using MATLAB . You can use built-in sawtooth function. Take DFT(Discrete Fourier Transform) of your signal using FFT and FFTShift and compare the frequency spectrum with the one you found in the first part.(Do not forget normalization after FFT.)

Reproduce the Fig. 2 in MATLAB and add it to your report.

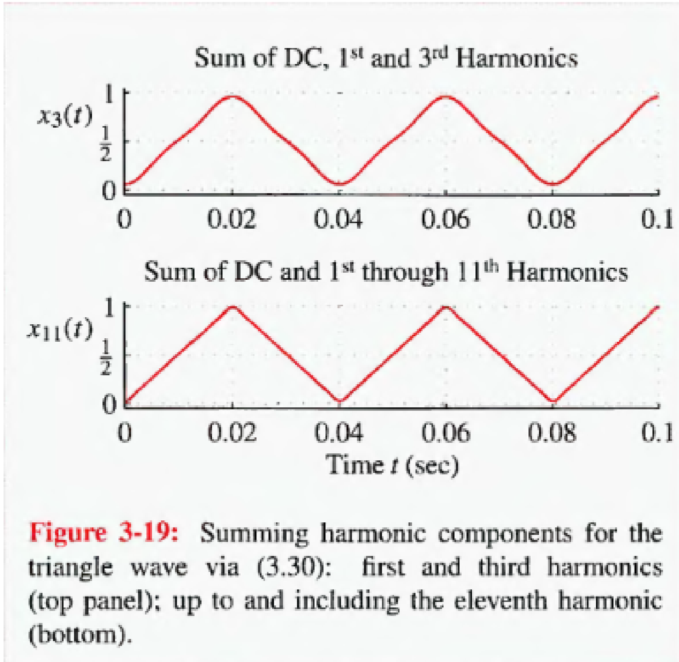

Figure 2: Harmonic Summation

Please solve Part B in a single script (question1.m) and include it in your submission.

2 Question 2 - Sampling and reconstruction

In this part, you should construct a digital signal which is a summation of 2 signals with different frequencies(blue plot). Then you will take samples from this digital signal with different sampling rate(orange plot). Afterwards, you will reconstruct the original signal with interpolation using sinc pulses(yellow plot). Please find the Nyquist sampling rate for your signal empirically.

You should produce 3 figures for 3 cases like the figures 3 and 4: Undersampling, sampling at Nyquist rate and oversampling.

Write your formula of your signals, sampling rates in your report and add these 3 figures to the report. Send me your script as well. (question2.m)

3 Question 3- Noise cancellation

In this question, you will use a open-source dataset: https://datashare.is.ed.ac.uk/handle/10283/2791

Download clean testset wav.zip and noisy testset wav.zip. There are people speaking in the clean test set without any backgroud noise and in noisy test set with background noise. Your task is to design a filter as good as you can so that you can obtain a voice file similar to the one

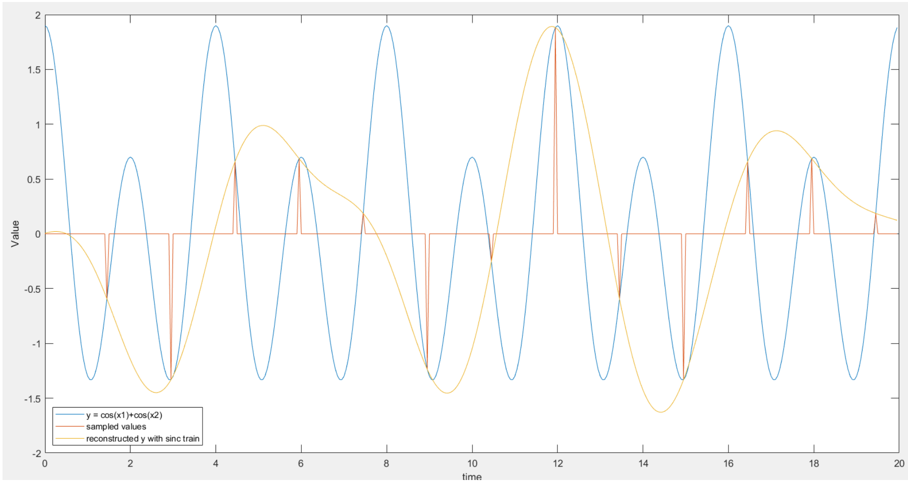

Figure 3: Undersampling

in clean test set by processing the noisy voice file. In this question, you can use your creativity. You can design averaging filters, band-pass filters or search for useful filters on web.However, it is forbidden to use any built-in function or copy paste of the code of any built-in function except circular convolution(cconv). You should be able to apply your filter both in frequency domain and time domain by using convolution.

I will illustrate your task with an ideal filter for a specific signal and implementing this example will grant you one-third of the question.

I select "p232 090 clean.wav" and "p232 090 noisy.wav". Frequency spectrum for 2 signals can be seen in figure 5 :

Then, I find the frequency response of my ideal filter by dividing FFT(clean signal)/FFT(noisy signal). The frequency response of my filter in figure 6:

Now, let's look how we could use this filter in time domain:

I can find the time-response of my filter by using the functions ifftshift and ifft.

There is a duality between time-domain and frequency domain. The multiplication operation in frequency domain is equivalent to the convolution operation in time domain. Therefore, you can obtain a new clean recording by taking convolution of noisy recording and time-response of your filter. Here, you should use circular convolution(cconv) because fft operation in MATLAB is Discrete Fourier Transform. You can read more about this on web.

You should tell in your report which voice sample(s) you choose, which method you choose to implement for your filter and how you implemented it. You should include time and frequency spectrum of your clean and noisy recordings and of your filter in your report as well. Please send me your script(s) too.

As I said above, you will get one-third of the points if you find the ideal filter for a single recording in one domain and you can apply it in the other domain by using convolution. However, you have to design your own filter working for multiple recordings to get more credit and you have to design your own filter working for random multiple recordings to get the full credit.At the end, your code should be able to produce a new clean voice recording after processing the noisy one. I

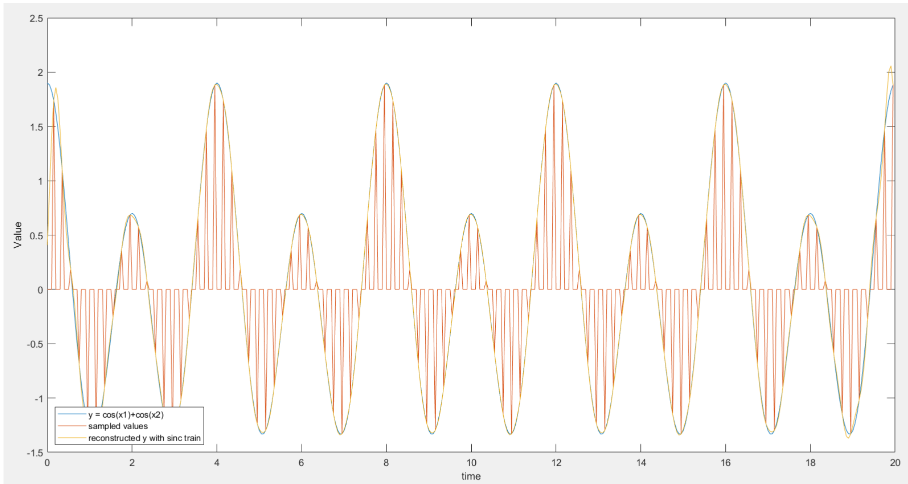

Figure 4: Oversampling

am aware of the fact that this is a machine learning data-set and I will try to be optimistic while grading your homeworks. You will get points according to performance of your filter and you should show the equivalence relation by using convolution in either frequency domain or time domain.
Deadline: 5 May, 11:59 P.M. (You will be graded out of 100)
Late deadline: 7 May, 11:59 P.M. (You will be graded out of 70)

Prepare a report (pdf file) includes your code, explanations and comments of your code for each question. You will compress everything into a zip file. Name it as YourNumber-CmpE362-HW2.zip and submit it via Moodle.

The Moodle upload limit is 2MB, do not forget to shrink your report pdf before submission. I will not accept any e-mail submissions. You will compress everything into a zip file and submit it via Moodle. When copying is detected, both parties will get zero.

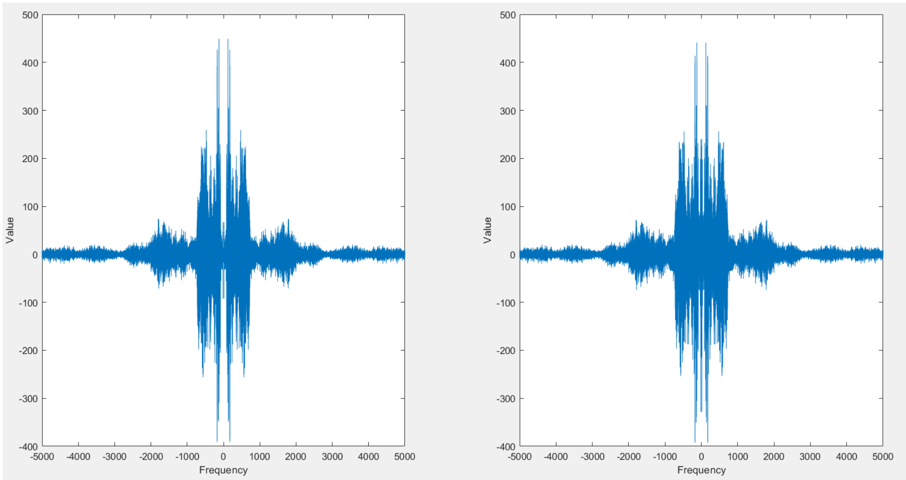

Figure 5: Frequency spectrum of my voice recordings

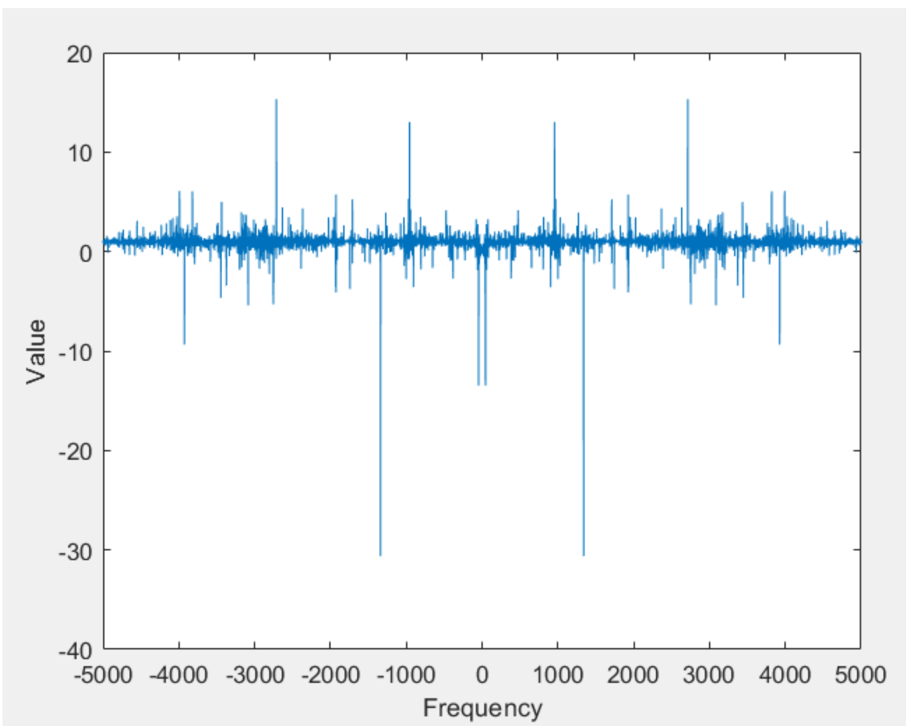

Figure 6: Frequency spectrum of my filter