2 Use of DFT and Windowing

2.1 DFT and windowing of a signal

The exercices in this chapter will illustrate the use of the Discrete Fourier Transform (DFT) to calculate the spectrum of a signal.

2.1.1 "Ideal" Scenario

Read the signal 'oef2id.wav' [1] in Matlab with the *wavread()* function and excise a segment of 256 samples from the signal. Apply a 256 points DFT based on these time domain samples. Can you determine which frequencies are present in the signal, based on its DFT?

2.1.2 "Non-Ideal" Scenario

Now read the signal 'oef2ni.wav' [1] and calculate a 256 point DFT for this signal. Compared to the previous scenario, the sounds in this signal are slightly shifted in frequency. What has happened to the DFT spectrum? Compare the "ideal" and "non-ideal" scenarios?

A possible explanation for this effect can be found by considering the DFT as a Discrete Fourier Series (DFS) decomposition of the signal, obtained by a periodic repetition of the 256 point segment. Illustrate this by reconstructing these periodically repeating signals for both signals (e.g. make a 1024-points long version of both signals by respectively repeating their first 256 samples four times[1]), and compare the reconstructed signals. What goes

[1] Hint: y=[x(1:256); x(1:256); x(1:256); x(1:256)];

"wrong" in the case of the non-ideal scenario?[2] [3] [4]

To reduce the effect of discontinuities at the frame boundary, signal segments are frequently first multiplied with a time varying window that attenuates the initial and final parts of the segment, before a DFT analysis is performed. Use the Matlab 'Window Design and Analysis Tool' (wintool) to generate and analyse these windows: 'Rectangular', 'Hann' and 'Flat Top' (choose a fixed window length of 256 samples). For every window, wintool shows two different graphs: the graph on the left is the time domain representation of the window, while the graph on the right illustrates its frequency domain representation. The most important frequency domain characteristics of a window are the width of its main lobe, and the (maximal) relative height and roll off of its side lobes. Apply the three different windows on the 256 point sound segment and compare the resulting DFTs. What is the influence of the window? It is interesting to point out that the original 256 point segment was actually also windowed, using a rectangular window.

2.1.3 Influence of the window on the DFT spectrum

We can study the influence of the window more accurately by applying a 4096 points DFT on the signal that you obtain by adding zero-valued samples to the original 256-sample window signal (add zeroes until a 4096-sample segment is obtained). This is also known as zero-padding. Calculate now the 4096 point DFT in the ideal and non-ideal case for some different windows. What is the relation with the 256 point DFT? Remember that a DFT is always a sampled version of the spectrum of a signal. You can also compare the 4096 point DFT with a window of 256, 512 and 1024 points from the original signals. What is the influence of the length of the window?

2.2 Composed Signals

Read in the signal 'oef2mul.wav' [1] and apply the different windows[5]. Work with windows of length 256 samples and 4096 DFT points. Which frequency components does the signal have? What is their relative strength? Note: plot the amplitude spectra in a deciBell (dB) scale. The smallest frequency

[2] Hint: You can use *stem()* instead of *plot()* to see the individual samples.

[3] Hint: Use 'Hold on' to plot two signals on the same plot.

[4] Hint: The color of the plot can be chosen by giving an extra argument to the 'plot' function. E.g. plot(x, 'r') for 'red', 'y' for 'yellow', 'b' for 'blue', 'k' for 'black',....

[5] The sampling frequency is 32 kHz. It can be read from the wav-file in the following way: [x,fs]=wavread('file name');

component is very small compared to the other components, and would be (almost) invisible in a linear-scale plot.

Bibliography

[1] prof. Verhelst's website: http://www.etro.vub.ac.be/Personal/wverhelst/

[2] W. Verhelst, Digital signal Processing - Part I: Basic Concepts, VUB uitgaven, 2012-2013.