

L.EEC025 - FUNDAMENTALS OF SIGNAL PROCESSING

*Academic year 2025-2026, week 10
TP (Recitation) problems*

Topics: Exercises on the DFT

Exercise 1

Find the DFT, of length N, of the discrete-time signal $x[n] = \frac{1}{2} \left(1 - \cos \frac{2\pi}{N} n \right)$. Validate your analytical result in Matlab by making, for example, N=8.

Exercise 2

Consider the discrete-time signal $x[n] = 2^{-n} u[n]$.

- a) Find its Fourier transform $X(e^{j\omega})$.
- b) Show that the discrete-time signal $y[n]$ of length N and whose DFT is obtained as
$$Y[k] = X\left(e^{jk\frac{2\pi}{N}}\right), \quad k = 0, 1, \dots, N-1$$
, is given by $y[n] = \frac{2^{-n}}{1 - 2^{-N}}, \quad n = 0, 1, \dots, N-1$.
- c) Verify numerically in Matlab the previous result using N=64.

Exercise 3

Consider the following set of Matlab commands:

```
x=[3 2 1 0 0 0];
X=fft(x);
Y=real(X);
y=x-iff(Y);
Z=X.*X;
z=ifft(Z)
```

- a) Say what the purpose of this Matlab code is, and identify the main DFT properties that are implied.
- b) Without computing the DFT or the IDFT, find the contents of vector y. Explain.
- c) Replace the two lines of the above code involving Y and y by different Matlab commands delivering the same result. Explain.
- d) Without computing the DFT or the IDFT, find the contents of vector z. Explain.
- e) The result in vector z is equivalent to the result of a linear convolution. Would this hold true if $x=[3 2 1 1 0 0]$?

Exercise 4

Consider the following discrete-time signal $x_0[n] = [0 \ 1 \ 1 \ 0]$.

- a) Find its DFT.
- b) Express the DFT of the signal $x[n] = [0 \ 0 \ 0 \ 0 \ 0 \ 1 \ 1 \ 0]$ as a function of the DFT of $x_0[n]$.
- c) Confirm your results using Matlab.

Exercise 5

Use a microphone correctly plugged to your computer and the following Matlab code (or an alternative audio recorder) in order to record on a WAV file the sound of your voice during 10 seconds. During this recording time you should utter different vowels, for example, a-e-i-o-u, as well as your name. This Matlab code sets the sampling frequency to 22050 Hertz and sets the sample resolution to 16 bits.

```
FS=22050; duration=10; NBITS=16;
r = audiorecorder(FS, NBITS, 1);
fprintf('Start speaking...\n');
record(r); % speak into the microphone...
pause(duration); stop(r);
% p = play(r); % listen to the complete recording
x = getaudiodata(r, 'single'); % get data normalized to +-1.0
x=x(FS:end); % avoid first second, may contain noise
fprintf('Stop speaking. Now playing...\n');
sound(x,FS);
fprintf('Stop playing.\n');
audiowrite('soundfile.wav', x, FS, 'BitsPerSample', NBITS);
```

After you create the WAV file (you only need to do this once), use the following Matlab code to read and play the recorded sound:

```
[x,FS,NBITS]=wavread('soundfile.wav'); % or
[x,FS]=audioread('soundfile.wav');
sound(x,FS,NBITS); %NOTE: x values are in the range [-1, 1]
N=length(x);
samples=[0:N-1];
figure(1)
plot(samples/FS, x);
xlabel('Time (s)');
ylabel('Amplitude');
title('soundfile.wav');
```

Our objective now is to filter the recorded sound using different filters and to listen to the result. We will use the (incomplete) Matlab code `aeiou_name.m` as a baseline. In order to complete this Matlab command file, you should replace 'COMPLETEHERE' by the appropriate Matlab commands according to the following indications:

- a) design an FIR “equiripple” filter (comand `firpm`, or `remez`), or order 126 (i.e., length 127) having passband between 300 Hz and 3200 Hz, and stopbands between 0 Hz e 200 Hz, and between 4000 Hz and the Nyquist frequency,
- b) design a low-pass FIR filter using the window method (comand `firl1`) whose cut-off frequency is 2000 Hz,
- c) using the filter designed in b) modify its impulse response vector in order to obtain a high-pass filter (i.e., without using again command `firl1`).

For each of the designed filters (one at a time!) first check, using the instructions already available in the supplied `.m` file, whether the resulting frequency response corresponds to the desired frequency response. Listen to each filtered signal and compare it to the original sound. Discuss what changes you notice as a result of each filtering.

Optional: You may use the Matlab filer design environment `fdatool` (or `filterDesigner` in the most recent Matlab version) in order to repeat each one of the above designs and check their characteristics.