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Winter Term 2018/2019

Foundations of Audio Signal Processing
Exercise sheet 10

To be uploaded in eCampus till: 11-01-2019 22:00 (strict deadline)

Exercise 10.1

[4+4+2 = 10 points]

(a) You are given the following sequences:

$$x = (x(0), x(1), x(2)) := (1, 2, 3),$$

$$y = (y(0), y(1)) := (4, 5).$$

Calculate (by hand) the convolution of x and y. Show all the steps in between.

(b) Write a Matlab function that produces an animation of the convolution. The function should have as input two general signals x and y to be convolved and it should show the signal x, the result of the convolution and all the steps in between.
Do not use the built-in Matlab command to calculate the convolution.

(c) Write a Matlab script which calculates and shows (using the function you created in (b)) the convolution of the following signals:
$x = (x(0), x(1), x(2), x(3)) := (1, 1, 1, 1)$ and $y = (y(0), y(1), y(2), y(3)) := (1, 1, 1, 1)$.

Exercise 10.2

[2+4 = 6 points]

Let $x, y \in \ell^2(\mathbb{Z})$ with $(x * y) \in \ell^2(\mathbb{Z})$. Prove that:

(a) the convolution operator is commutative,

(b) the Fourier transform transfers convolution to pointwise multiplication, i.e.,

$$\widehat{(x * y)}(\omega) = \hat{x}(\omega) \cdot \hat{y}(\omega).$$

Exercise 10.3

[$2 + 2 + 3 + 2 + 2 = 11$ points]

This exercise is based on the demo about filters, which you have seen in the lecture and has to be solved using Matlab.

(a) Create a sine wave with sampling frequency 1000 Hz, length 1 second and frequency 1 Hz. Add random noise to the sine wave and plot the result.

(b) Create an averaging filter of length 50.

(c) Apply the filter created in (b) to the signal created in (a) and plot the result.

(d) Use a Hann filter of length 50 and apply it to the signal you created in (a). Plot the result and compare it with the one in (c). What do you notice?

(e) Plot the frequency response of the two filters you created in (b) and (d). Explain the differences.