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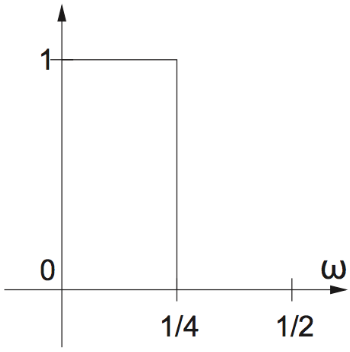
Foundations of Audio Signal Processing
Exercise sheet 11

To be uploaded in eCampus till: 19-01-2019 22:00 (strict deadline)

Exercise 11.1

[4+2+2 = 8 points]

In this task, an ideal low pass filter along with its properties will be be investigated. Consider an ideal low pass filter with cut-off frequency $\omega_0 = 1/4$, as illustrated below.



Prove that:

(a) The filter coefficients of an ideal low pass filter with real coefficients and a cut-off frequency of $\omega_0 \in (0, \frac{1}{2})$ are given by

$$h(n) = 2\omega_0 \mathrm{sinc}(2\omega_0 n), \quad n \in \mathbb{Z}.$$

(b) C_h is not causal.

(c) C_h has an infinite number of filter coefficients different from 0, i.e., C_h is IIR.

Exercise 11.2

[2 + 3 = 5 points]

Let us consider the Haar filters h and g defined as:

$$h(n) = \begin{cases} 0.5 & \text{if } n \in \{0, 1\}, \\ 0 & \text{otherwise.} \end{cases}$$

and

$$h(n) = \begin{cases} 0.5 & \text{if } n = 0, \\ -0.5 & \text{if } n = 1, \\ 0 & \text{otherwise.} \end{cases}$$

(a) Find an explicit formula for the frequency responses of both h and g. (Hint: Calculate the Fourier transform by hand.)

(b) The filters h and g have the properties that they divide the signals in low-frequency and high-frequency parts. Prove that by adding the two filtered signals, one gets back the original signal.

Exercise 11.3

[4 + 4 + 6 = 14 points]

(a) Write a Matlab function which creates a lowpass filter (use the function *fircls*) that, if applied to a signal of sampling frequency 44100 Hz, lets the frequencies below 10000 Hz pass. Plot the frequency response of the above created filter.

(b) Plot the frequency response of the filter (created in (a)) of order 10, 50 and 200. What happens if you change the order of the filter?

(c) Write a Matlab function that performs the following:

- Creates a sum of two sine waves of frequencies 5000 Hz and 15000 Hz. The sampling frequency has to be of 44100 Hz.
- Applies the filter created in (a) to the sum of sine waves.
- Plays back the original signal and the filtered version and plots their Fourier transforms.