Digital Signal Processing

Assignment 3: audio filtering

A short audio sample has been provided with an interference. The sampling rate is 16kHz.

Signal analysis

First step is to analyse the signal, this way we can find out which frequency has to be filtered out.

1. Plot the single-sided frequency spectrum of the audio sample.
2. Determine the frequency of the interference signal.

Useful MATLAB commands: *audioread(), fft()*

Low pass filter design

As we have seen in the lectures the transfer function in the Laplace domain for a simple low pass filter is

$H_l(s) = \frac{1}{1+s\tau}$

Determine the digital equivalent of this filter using the Zero-order Hold (ZoH) approximation technique.

As cut-off frequency we will use ½ the interference frequency

$$f_c = \frac{1}{2} f_{interference}$$

1. Determine $H(z)$
2. Determine the difference equation

Filtering

Using the difference equation write a for loop to filter the distorted audio sample. Plot the single-sided frequency spectrum of the filtered audio sample.

1. How much, in dB, is the interference signal suppressed?

Comparison (NOT GRADED)

It is interesting to compare the digital filter derived by the ZoH technique to its analog counterpart.

1. Calculate the attenuation at the interference frequency of the analog counterpart by hand.
2. Use *freqz()* and *freqs()* to plot both the analog and the digital frequency response of the filters in the same figure.