IST, Digital Signal Processing, Lab. #2 – Filtering Pedro M. Q. Aguiar¹ January, 2012

In this lab., we will consider two types of systems, namely, linear time-invariant (LTI) ones and the median filter. The goal is to compare these systems in what respects to their capability to remove or attenuate the noise that typically corrupts old vinyl records. In the audio signal that will be used in the experiments, this kind of noise is simulated by saturating random samples, process that is conveniently termed by salt-and-pepper. Between brackets ({command}), you will find suggestions of relevant MatLab[©] commands for the proposed tasks. Naturally, you should use the system help to obtain the description of how to use the command {help command}.

- 1. Observation of the corrupted signal.
 - a) Load the sound file *fugee*. {wavread}
 - b) Plot (segments of) the signal. Comment. {plot}
 - c) Visualize its magnitude spectrum. {fft}
 - d) Listen the signal. Comment. {soundsc}

2. Consider the LTI filter with a generic system function,

$$H(z) = \frac{b_0 + b_1 z^{-1} + \dots + b_M z^{-M}}{1 + a_1 z^{-1} + \dots + a_N z^{-N}},$$

where, naturally, the parameters $\{b_0, b_1, \ldots, b_M, a_1, \ldots, a_N\}$ determine its frequency selectivity.

- a) Compute the parameters of a Butterworth low-pass filter of order 10, with cut-off frequency of $\pi/2$. {butter}
- b) Visualize the frequency response of the filter and its pole-zero map. Comment. {freqz}, {roots}
- c) Filter the audio signal. {filter}
- d) Plot (segments of) the filtered signal. Comment.
- e) Visualize its magnitude spectrum. Comment.
- f) Listen the filtered signal. Comment.
- g) Repeat a)-f) for other values of cut-off frequency, attempting to improve the results. Comment.

3. Consider the median filter whose input-output relation is

 $y[n] = \text{median}\{x[n-M], x[n-M+1], \dots, x[n], \dots, x[n+M]\},\$

where the parameter M controls the order.

- a) Characterize this system in what respects to causality, stability, linearity, and time-invariance.
- b) Filter the audio signal with a median filter with parameter M = 1. {medfilt1}
- c) Plot (segments of) the filtered signal. Comment.
- d) Visualize its magnitude spectrum. Comment.
- e) Listen the filtered signal. Comment.

¹Parts of this document are based on a lab. assignment authored by João Sanches.