Digital Signal Processing - DSP
(PDS - Processamento digital de Sinais)
Instituto Superior Técnico - 2º Semester 2008/2009
João Sanches

Filtering

In this work two types of linear and non linear filters will be used to attenuate or, if possible, to remove the noise corrupting an audio file. The noise corrupting the music is **salt and pepper** to simulate the traditional noise present in the old vinyl long play records. This type of noise, non additive nor Gaussian, is characterized by maximum and minimum saturations at random instants.

The elimination of this type of noise using linear filtering is usually not efficient. Conversely, non linear filtering is usually quite efficient when dealing with this type of non-linear degradation process, e.g., the **median** filter.

The output of a basic 1D median filter is the median value of the set of values of the samples in the neighborhood of the current sample plus the current sample, that is, the median of the sample set contained in the $2M+1$ dimension neighboring window. In this strategy the noisy samples tend to accumulate at the extrema of the window which leads to its elimination.

1. Read the audio file, listen to it and represent it graphically. Can you identify, in the graph, the noise corrupting the music?

2. Develop a routine to implement the difference equation corresponding to the following filter

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

   The routine is called by using the following command: $y = filtro(x, A, B)$; where $A = [1, a_1, ..., a_N]^T$, $B = [b_0, b_1, ..., b_M]^T$, y is the output signal and x is the input signal.

3. Design a 10 order Butterworth filter with a cut-off frequency equal to $\pi/2$ by using the function $butter()$. Visualize its frequency response using the function $freqz()$ and the pole localization using the function $roots()$.

4. Use the routine implemented in question 2) to filter the signal $x(n)$ with the filter designed in question 3). Comment the results, namely concerning the quality of the audio output signal.

5. Filter now the audio signal with the median filter, using the following command: $y = medfilt1(x, N)$; where $N = 3$ is the window dimension. Listen the result and compare it with the one obtained in 4).

6. Try other dimensions for the median filter in order to improve the results. Comment the results as a function of the window dimension of the median filter.