Digital Signal Processing - DSP
(PDS - Processamento Digital de Sinais)
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In this lab the **aliasing** phenomenon will be illustrated. The sampling process will be simulated by using the **down sampling** operation applied to discrete signals. The main goal of this work is to observe (listen to) the artifacts due the spectral overlapping and, simultaneously, to show the effectiveness of the **anti-aliasing** filter to eliminate them.

Experimental work

1. Build a vector $x(n)$ that represents the following *chirp* signal, $x(n) = \sin(2\pi f[t(n)]t(n))$ with $f(t) = f_a + t(n)(f_b - f_a)$, $f_a = 1000$ Hz and $f_b = 2000$ Hz. The vector $t(n)$ contains the sampling moments of a continuous signal in the interval $[0, 5]$ seconds at 8000 samples per second. Listen to the obtained signal by using the following function:

   >>*soundsc(x,8000)*. Comment.

2. Build the spectrogram of x by using the function *spectrogram*. What is the relation of this spectrogram with the listened sound?

3. Sample the signal $x(n)$ by obtaining the following signal: $y(n) = x(2n)$. Listen to $y(n)$ and observe the respective spectrogram. Explain the observation.

4. Load the sound file using the following commands:
   >>*[romanza,FS,NBits]=wavread('romanzasmall');*
   >>*x=romanza(:,1);*

   Listen to its contents and register the respective sampling frequency, Fs.

5. Sample x at a sampling rate ten times (/10) less than Fs and listen to the result. Describe the main differences and explain them.

6. Filter the original signal x by using the following command:
   >>*x=filter(fir1(100,0.1),1,x);*.

   This is a 100 order low pass FIR filter with a cut-off frequency of $\pi/10$. Repeat the previous item using the filtered signal and explain the differences obtained.