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Processamento Digital de Sinais

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Introduction

This Lab project addresses *sampling* and *aliasing* concepts with synthetic and real signals. Digital computers are powerful tools which allow us to manipulate and store signals. However, digital computers can only manipulate discrete signals. This means that continuous signals (e.g., speech, audio, biomedical signals) have to be converted into discrete signals by a sampling operation before being processed in a computer. It can be shown that this conversion is a lossless operation if the sampling rate is high enough: the sampling theorem states that no information is lost if the sampling frequency is higher than twice the highest frequency of the signal. If this condition is not met, there is loss of information caused by spectral folding. This is known as aliasing and it creates an annoying effect since high frequency components of the signal are shifted in the spectral domain by a multiple of 2π and distort the signal.

This work studies these effects using synthetic and real (audio) data. In the case of audio data, we simulate the sampling process by using a *down sampling* operation. The artifacts created by aliasing can be easily identified. The work also shows the effectiveness of the anti-aliasing filter.

Experimental work

1. Consider a continuous chirp signal (t in sec.)

$$x(t) = \cos\left[2\pi\left(\frac{1}{2}kt^2 + f_0 t + \phi_0\right)\right] \tag{1}$$

   with instantaneous frequency $\omega(t) = 2\pi(kt + f_0)$. In this work, we will assume that $k = 1000$, $f_0 =$ 1000Hz.

   Build a Matlab vector $x(n)$ obtained by sampling the chirp $x(t)$ at a sampling rate of 8000 samples per second, in the interval $[0, 5]$ seconds. Listen to the obtained signal by using the following function:

   \>>*soundsc(x,8000)*. Please comment what you heard.

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.