Processamento Digital de Sinais
Instituto Superior Técnico

Sampling and Aliasing

This lab assignment addresses *sampling* and *aliasing* with synthetic and real-life signals. Digital computers are powerful tools which allow us to manipulate and store signals. However, digital computers can only manipulate discrete-time signals. This means that continuous-time signals (e.g., speech, audio, biomedical signals) have to be converted into discrete-time signals, by a sampling operation, before being processed in a computer. This conversion is lossless 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 *aliasing* (also called *spectral folding*). This creates an annoying effect, since high-frequency components of the signal are moved into the low frequencies, and distort the signal. This work studies these effects using synthetic and real-life audio data. The work also illustrates the usefulness of *anti-aliasing* filters.

Notes

This assignment is to be performed in class. At the end of the class you must submit, through fenix, a zip file with your results and your MatLab code. The items that you should address are marked, below, in a format such as **R1.a)**.

Experimental work

1. Consider the following continuous-time signal (with t measured in seconds),

$$x_c(t) = \cos(2\pi F_0 t) \tag{1}$$

   with frequency $F_0 = 300Hz$.

   R1.a) Build a MATLAB vector `x` by sampling $x(t)$ at a rate of 3000 samples per second in the interval $[0, 0.5]$ seconds. Plot signal `x` as a function of time.

   R1.b) Compute and display the spectrogram of `x` using the command

   ```
   N=512; spectrogram(x, hann(N), 3*N/4, 4*N, 3000, 'yaxis').
   ```

   The first argument of the `spectrogram` function, `x`, is the signal of which you wish to compute the spectrogram. The second argument, `hann(N)`, indicates the window that should be used to compute the spectrogram, in this case the Hanning window. By varying the value of `N`, you can control the length of the window (`N` should be a multiple of 4). The fifth parameter, `3000`, indicates the sampling frequency of the signal. For more information about the `spectrogram` function, use MATLAB's help.

   R1.c) Comment on what you observe in the time and spectrogram plots.

2. Repeat the previous item using frequency $F_0 = 2700Hz$.

   R2.a) Comment and explain what you observe.

3. Load the sound file `Happier.mp3` using the command

   ```
   [x, Fs] = audioread('Happier.mp3');
   ```

   Write down the signal's sampling frequency, contained in variable `Fs`. Listen to the signal's contents (don't forget to use the appropriate sampling frequency in the `soundsc` command).

   Note: The signal is rather long. You can stop playing it with the command `clear sound`.

Visualize the spectrogram of the first 15 seconds of the signal. In the computation of the spectrogram, choose a window length that allows you to clearly see the spectral lines of the sound.

R3.a) Choose the window length that yields, in your opinion, the best representation of the signal's time-frequency structure. Justify your choice.

4. Sample `x` so that the new sampling rate is `Fs/5`, and listen to the result (again, don't forget to use the appropriate sampling frequency in the `soundsc` command).

   Visualize the spectrogram of the first 15 seconds of the sampled signal (don't forget to properly set the window length and the sampling frequency).

   R4.a) Describe, and try to explain, what you heard and observed.

5. Filter the original signal `x` by means of the following command:

```
xf = filter(fir1(100, 0.2), 1, x);
```

   This command performs filtering by a order-100 low-pass FIR filter with a cut-off frequency of $0.2\,\pi$ (expressed in the angular-frequency scale of the discrete-time Fourier transform).

   Sample the filtered signal `xf` so that the new sampling rate is `Fs/5`. Listen to the result and visualize its spectrogram, using the appropriate sampling frequency.

   R5.a) Explain the differences in what you heard and observed, relative to what you heard and observed in item 4.