MA1 Acoustic Labs 1
Introduction to audio signals analysis / synthesis on Matlab

Author : Hervé Lissek

October 19, 2022

Duration :	2 x 4 hours (technical work & report)
Support :	
Prerequisite :	Signal Processing
References :	Matlab Signal Processing Toolbox
	Mario Rossi, *Audio*, Presses Polytechniques Universitaires Romandes, 2007
Hardware/software :	— computer with Matlab ; — headphones ; — microphone ; — loudspeaker.

Objectives of the laboratory

At the end of this laboratory, you should be able to display, generate and analyse simple audio signals in time and frequency with Matlab.

Preliminary work

Hints : prepare the report template (Word, LaTeX, etc.) at the beginning of the lab to spare time at the end.
Also, create a .m code on matlab, that you will name TP01_ yourname.m. Use the % symbol to comment lines in the code, which is useful for explaining the algorithms. You are asked to comment as much as possible the code so that it can be easily corrected. For example, it is expected that you create code sections for each exercise, by writing the following lines :

```
%%
%%%%%%%%%%%%%%%
% Exercise n %
%%%%%%%%%%%%%%%
```

The first %% initiate a section in Matlab. You can then compile the code by section, if needed.

Exercice 1 : Analysis of a sinusoidal ("pure") sound

Throughout this exercise, the sampling frequency F_s is fixed to 44100 Hz. We assume that the different audio signals represent sound pressures in Pa.

1. Justify the choice of Fs for sampling audio signals.
2. Define, with this sampling frequency, a time vector T lasting 1 second in Matlab.
3. Generate a sinusoidal audio signal S at frequency $f_0 = 1034$ Hz, of amplitude 1 Pa, and lasting 1 second.
   What is the theoretical rms value of such a signal ?
4. Remind the general definition of the rms value of a **continuous** signal.
   What is the definition of the rms value of a **discrete** signal ?
   Compute the discrete rms value with the generate signal on Matlab, and compare this value to the theoretical one (preceding question).
5. The *DFT* (Discrete Fourier Transform) of a signal x of length N is given by :

$$X[k] = \sum_{n=0}^{N-1} x[n] e^{-2i\pi \frac{kn}{N}} \text{ with } 0 \leq k < N, k \in \mathbb{Z}, \tag{1}$$

   where n and k account respectively the samples (time domain) and the frequential bins. In opposition to "analog" frequencies f (in continuous domain), expressed in Hz, the digital frequencies k are dimensionless. The two variables are related by :

$$f = \frac{k.F_s}{N} \text{ with } 0 \leq f \leq \frac{F_s}{2} \text{ and } 0 \leq k \leq \frac{N}{2}. \tag{2}$$

   What is the frequential bin corresponding to an analog frequency of $f_0 = 1034$ Hz (sampled at 44100 Hz) for a *DFT* of length $N = 1024$?
6. Compute the *DFT* of signal S on 1024 points. To this end, use the Matlab function *fft*. You can get information on the function on Matlab by quering :

```
doc fft
```

   in the *Matlab Command Window*.
   Create a graph window and draw the module of *DFT* $\|X[k]\|$.
   Explain the shape of this curve (occurrence of peaks, symmetry).
   Justify the amplitude of the first peak.

   In the following, you should truncate the *DFT* vector to avoid information redundancy, by writing :

```
DFT = DFT(1:N/2+1)
```

7. Starting from relation 2, compute the frequency vector F associating each frequential bin to its analog counterpart. Verify that $F[1] = 0$ et $F[513] = 22050$.
8. Now draw the non-redundant *DFT* as a function of analog frequencies, represented on a logarithmic scale.
   Create code lines in order to :
   — Set an amplitude of 1 Pa at frequency $f_0 = 1034$ Hz.
   — Visualize the amplitude in dB (Reminder : $X_{\text{dB}} = 20 \log_{10}(|X|)$.
   Verify that the value at $f_0 = 1034$ Hz is 0 dB.
9. The dB scales are always referred to a reference value A_0 :

$$A_{\text{dB}} = 20 \log_{10} \left(\frac{A}{A_0} \right). \tag{3}$$

In the preceding question, the reference value is 1, which is known as "Full Scale" dB in digital audio, denoted dB FS. The value of 0 dB FS corresponds to the maximal value being coded on a digital system (above which the input saturates). As a consequence, the values expressed in dB FS are usually negative (assuming the audio signals are within the dynamics of the input of the sound card).
In acoustics, the dB reference corresponds to the average auditory threshold at 1 kHz, which is $p_0 = 20\ \mu$Pa, allowing estimating the Sound Pressure Levels (*SPL*) L_p in dB SPL. Then, the SPL corresponding to the audibility threshold is 0 dB SPL, and the usual values of dB SPL are positives.
The sound pressure level of a given *rms* pressure p_{rms} is derived according to :

$$L_p = 20 \log_{10} \left(\frac{p_{rms}}{p_0} \right). \tag{4}$$

Calculate the Sound Pressure Level of a sinusoidal signal of *amplitude* 1 Pa.

10. Assuming the signal S defined in the preceding corresponds to an acoustic signal of amplitude 1 Pa, create the code lines in order to get the *DFT* with the correct value in dB SPL at 1034 Hz (scaling factor).

Exercice 2 : Generation/Recording of an audio signal

1. Write code lines allowing generating and recording simultaneously a sinusoidal signal $s(t)$ with the following functions :

```
info = audiodevinfo;
recorder1 = audiorecorder(Fs,nBits,nChannels,ID);
record(recorder1);
sound(signal,Fs);
pause(time);
stop(recorder1);
```

2. Compare the *waveforms* and the *DFT* of the recording with a microphone and the sound card directly. Comment.

Exercice 3 : Mixtures of sinusoidal signals

1. Synthesize the combination of two sinusoidal signals at $f_1 = 500$ Hz with amplitude 2 Pa and $f_2 = 1500$ Hz with amplitude 1 Pa, over a duration of 1 second at sampling frequency 44100 Hz. We assume the two signals are in phase at time origin.
2. Represent the waveform and the *DFT* of this signal (en dB).
3. What is the difference, in dB, between the two observed peaks. Justify.
4. Synthesize the combination of two sinusoidal signals at $f_3 = 1000$ Hz with amplitude 1 Pa and $f_4 = 1001$ Hz with amplitude 1 Pa, over a duration of 1 second at sampling frequency 44100 Hz. We assume the two signals are in phase at time origin.
5. Represent the waveform and the *DFT* of this signal (en dB). Comment.

Exercice 4 : Periodic signals

1. Generate square and triangle signals as the one represented on figure 1, with period $T = 4$ ms.

2. Generate a sinusoidal signal of same period.
3. Listen to the three signals and comment.
4. Compute the *DFT* of each signal and compare them to each other.

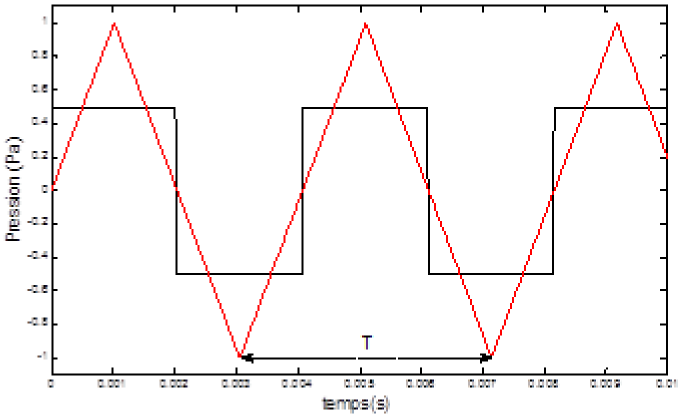

Figure 1 – square and triangle signals of period T