UE Traitement avancé du son
Compte-rendu du TP 1 :Acoustics and Sound Localization

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1.1 In preparation

In this first tutorial, space is represented as a plane with an orthonormal basis [x,z] and a reference point O. A microphone array is placed on the x-axis and centred at O. It is equipped with N microphones (N even), placed at the coordinates $(x = 0, z = z_n)$, $n \in [1, N]$, and equally spaced by the distance d.

1. Express z_n as a function of the spacing d between the microphones :

$$z_n = (n - \frac{N+1}{2})d$$

with
n : current microphone.
N : number of microphones.
d : the interspace between two successive microphones.

1.1.1. Wave Fronts

Assumption 1 : the wave front is assumed to be a plane

2. Express dn as a function of z_n and the angle θ_s, which is the angle of incidence defined with respect to the axis [O, z] of the microphone array :

$$d_n = z_n.cos(\theta_s)$$

with
θ_s : the angle of incidance
z_n : spacing between microphones

3. Deduce the angle of incidence θ_s, as a function of the delay δ_{τ_n} .

$$\delta_{\tau_n} = \frac{z_n}{c} cos(\theta_s) \implies \theta_s = arccos(\delta_{\tau_n} \frac{c}{z_n})$$

Assumption 2 : the wave front is assumed to be spherical

4. Express the (Cartesian) coordinates of the source (x_s, z_s) as a function of the distance r_s between the source and the centre of the microphone array, the position of the n^{th} microphone z_n, the distance d_n and the angles α_s and $d\alpha_s$, the angle between the wave arriving at the n^{th} microphone and that arriving at O

$$\alpha_s + d\alpha_n + 90 + \omega = 180 \implies \omega = 90 - (\alpha_s + d\alpha_n)$$

$$\begin{cases} x_s = (r_s + d_n).sin(90 - (\alpha_s + d\alpha_n)) \\ z_s = (r_s + d_n).cos(90 - (\alpha_s + d\alpha_n)) \end{cases} \implies \begin{cases} x_s = (r_s + d_n).cos(\alpha_s + d\alpha_n) \\ z_s = (r_s + d_n).sin(\alpha_s + d\alpha_n) \end{cases}$$

5. Deduce the equation verified by d_n :

$$d_n^2 + 2d_n r_s = z_n^2 - 2r_s z_n cos\theta_s \quad (1)$$

According to the generalized Pythagorean theorem, we have

$$(d_n + r_s)^2 = r_s^2 + z_n^2 - 2r_s z_n cos(\theta_s) \implies d_n^2 + r_s^2 + 2r_s d_n = r_s^2 + z_n^2 - 2r_s z_n cos(\theta_s)$$

$$\implies d_n^2 + 2r_s d_n = z_n^2 - 2r_s z_n cos(\theta_s)$$

6. Propose a method to derive the position of the source, given by the angle of incidence θ_s with respect to the axis [O, z) of the microphone array, and its distance from the origin r_s.

$$\begin{cases} x_s = r_s sin(\theta_s) \\ z_s = r_s cos(\theta_s) \end{cases}$$

   We need to find the expression of the angle as a function of the positions and then differentiate it as usual

7. What happens to equation (1) when $r_s \to +\infty$ Conclude.

   We rewrite (1) as follows :

$$\frac{d_n^2}{r_s} + 2d_n = \frac{z_n^2}{r_s} - 2z_n cos\theta_s$$

   si $r_s \to +\infty$

$$\implies d_n = -z_n cos\theta_s$$

   That means that we are in the case of a planar wave front. unlike to the case in question (2) we have an angle of incidence $\theta = -\theta_s$ which means the source is in the opposite side .

1.2 In session

For all the experimental part of this tutorial, define an orthonormal reference frame $[0, x, z)$ on your workbench and place the microphone array on the axis $[0, z)$ centred at O.

1.2.1 Getting started

To begin, start the acquisition of the audio system, and capture one audio buffer. Plot the resulting signals as a function of time.

1.2.2 Speed of Sound

1. Place the source on the [0, z) axis towards the origin O, at a small distance to the edge of the microphone array.
2. Generate a pure tone at a fixed frequency chosen between 100 Hz and 1 kHz using the app previously installed on your smart phone.
3. Record the sound wave with the microphones of the array, over a period of about 1 s, and then display the waveforms recorded by the first and the last microphones in the array on the same graph, with different colours.

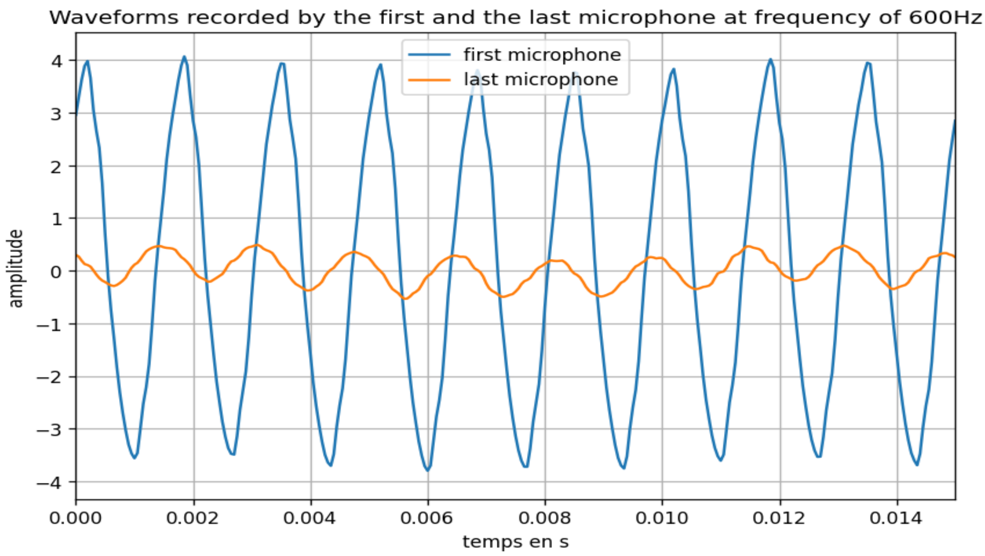

FIGURE 1 – Display of the signals recorded by the first and last microphone.

We observe that both microphones have recorded sinusoidal audio signals. We notice an attenuation for the sound recorded on microphone 8 compared to microphone 1, due to the distance between the two microphones (the source is closer to microphone 1).

4. Calculate their Discrete Fourier Transforms on a judiciously chosen number of points. For each Fourier transform, plot the modulus and the phase.

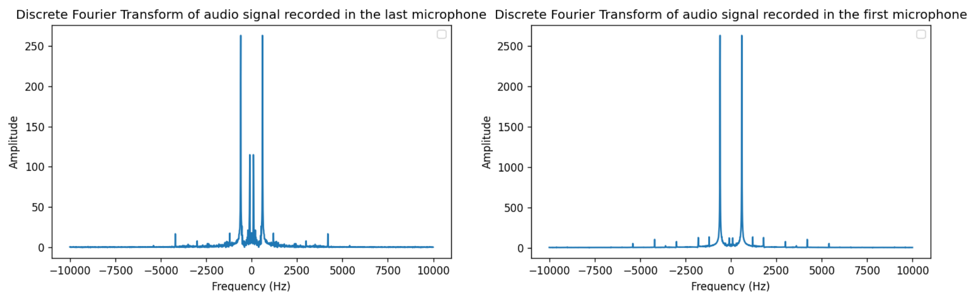

FIGURE 2 – Discrete Fourier Transform of the signals recorded by the first and the last microphones

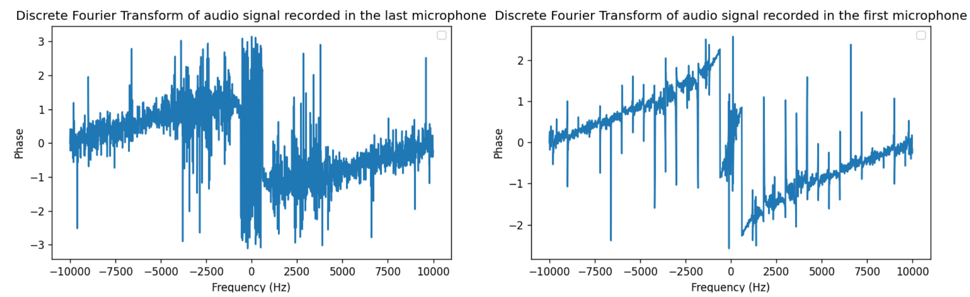

FIGURE 3 – Discrete Fourier Transform of the signals recorded by the first and the last microphones

5. Measure the time needed by the wave to travel the distance between the microphones
   a the observation of the waveform

   By measuring between two peaks of the signals obtained by acquisition, we observe that the time delay $= 0.0012$s b the phase of spectra

By using the phase difference between the two signals, we can estimate the time delay by applying the following formula :

delay time = $(\Phi_1 - \Phi_8)/(2\pi f)$, with $\Phi_1 = 3.9$ $\Phi_8 = 0.1$ et f = 600Hz

we find a delay time = 0.001s approximately.

6. Deduce the speed of sound propagation, also known as the celerity. Compare with the theoretical value at room temperature

$$vitesse = distance/timeDelay$$

by using a time delay = 0.0012s we find through the observation of the waveforms we obtain : $vitesse = 7d/0.0012 = 350m/s$

by using the time delay = 0.001s found with the phases difference of the signals, we obtain $vitesse = 7d/0.001 = 420m/s$

We notice that the calculation made using the waveform observation is closer to reality in the absence of noise observed on the phase.

The signals observed at frequency f = 600Hz appear to be in phase, which allows us to estimate correctly the time delay between them and then getting results close to reality.

7. Repeat the previous steps with a white noise sound source. Calculate the Discrete Fourier Transforms (DFT) (modulus + phase) of the waveforms recorded by the first and the last microphones in the array. Propose a method using the phase difference of the two DFTs to estimate the speed of sound.

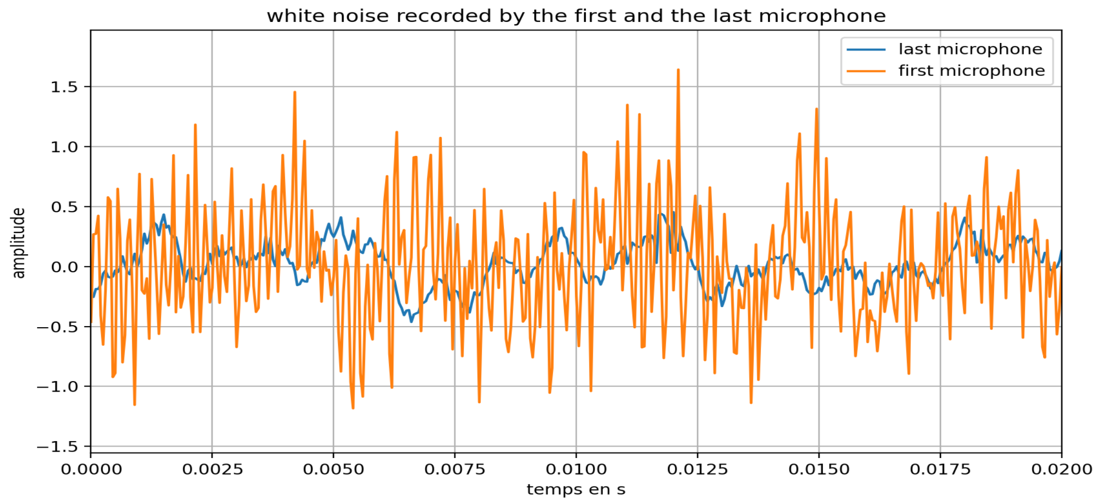

FIGURE 4 – Display of white noise recorded by the first and last microphone

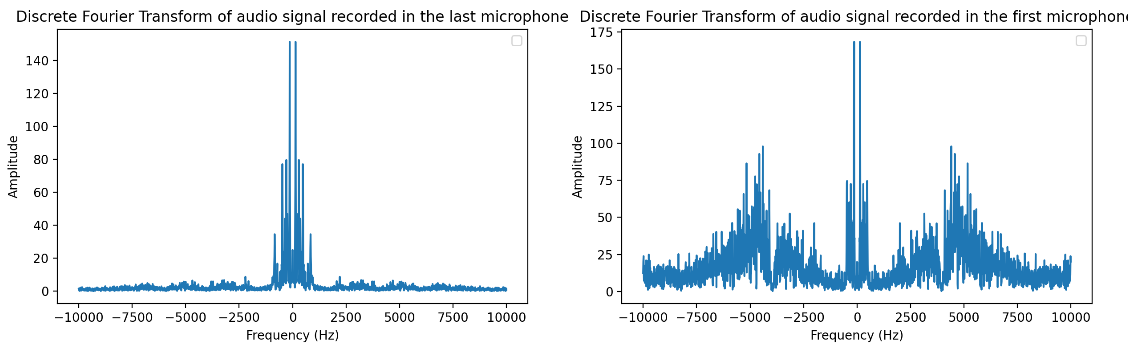

FIGURE 5 – Modulus Discrete Fourier Transform of white noise recorded by the first and the last microphones

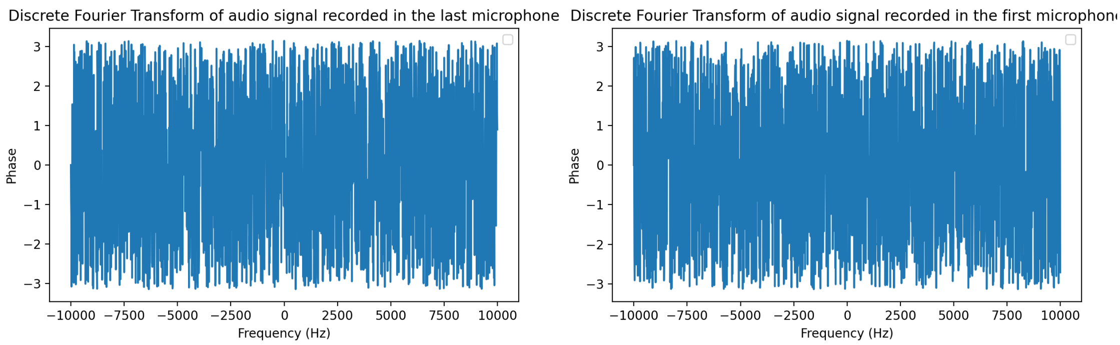

FIGURE 6 – Phase Discrete Fourier Transform of white noise recorded by the first and the last microphones

8. Discuss and compare the accuracy of the three methods tested above.

   The accuracy obtained using the different methods with frequency of 600Hz, (for real sound speed c = 340 m/s) :

   - By using waveform observation,we achieve an accuracy of $(340/350) * 100 = 97.14\%$ which is the most satisfactory method.

   - By using the phase difference of the spectra of the two signals, we obtained an accuracy of $(340/420) * 100 = 80.95\%$, we observe that this is lower than the accuracy obtained with waveform observation. That is likely due to noise in the phase.

   - In the case where white noise is recorded, and using the phase difference between the spectra, we obtain a delayTime of 0.0017s, which gives us a speed of sound c = 247 m/s. The accuracy is therefore 72.64%, making it the least interesting method for estimating the speed of sound.

9. Repeat previous steps at a frequency of 3Khz

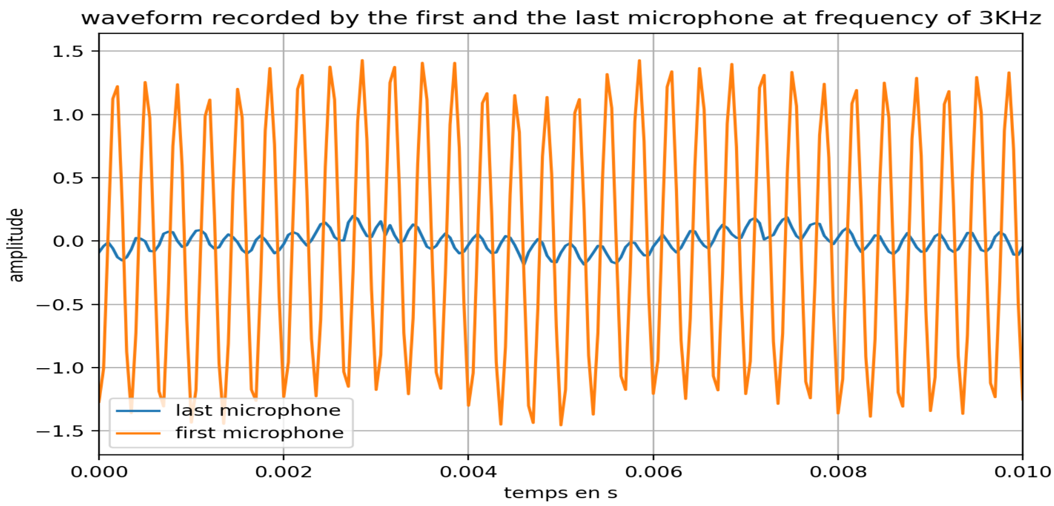

FIGURE 7 – Display of waveform recorded by the first and last microphone at frequency of 3Khz

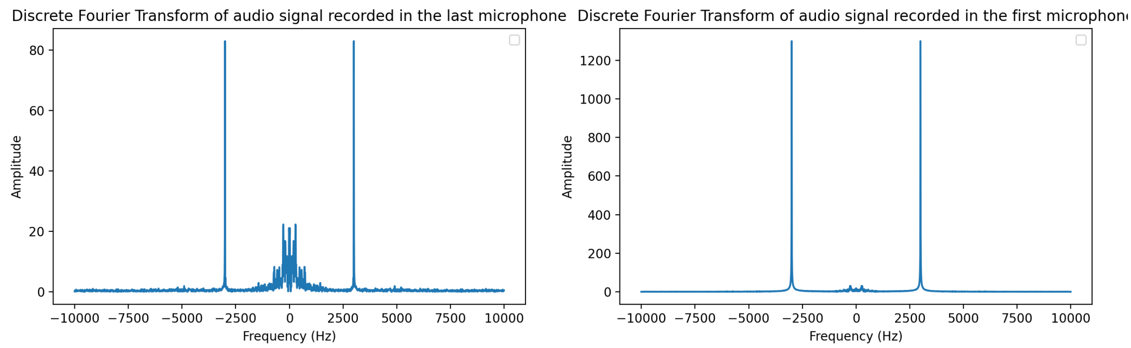

FIGURE 8 – Modulus Discrete Fourier Transform of white noise recorded by the first and the last microphones at frequency of 3Khz

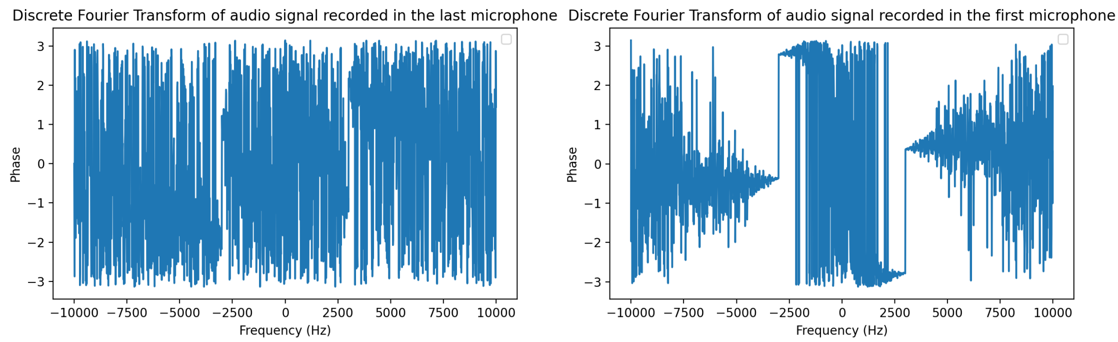

FIGURE 9 – Phase Discrete Fourier Transform of white noise recorded by the first and the last microphones at frequency of 3khz

10. Use the first two methods described above - using a pure tone to estimate the speed of sound :

    (a) in the time domain : By calculating the time delay between two peaks, we find a delayTime of 0.0001s, which gives a speed of sound c = 4200 m/s

    (b) in the frequency domain : on faisant le calcul de la difference de phase entre les deux signaux on trouve un de delayTime = 0.00018s, ce qui donne une célérité de c = 2223.33 m/s.

    It can be observed that the estimates are completely distorted. This is due to the fact that the frequency is too high to be estimated correctly. Beyond a certain frequency value, the shorter wavelength leads to a greater difference in propagation time compared to the wavelength, which can result in incorrect estimations of the speed of sound.

11. Deduce the limit frequency above which it is no longer possible to determine the delay between the acoustic waves measured. In the following, this limit frequency is noted f_{lim}. If the frequency is too high, it results in a smaller wavelength (inversely proportional relationship). However, in order to accurately estimate the time delay between two microphones separated by a distance d, the distance d must be smaller than half of the wavelength. To summarize $D < \frac{\lambda}{2} = \frac{c}{2f} \implies f_{lim} = \frac{c}{2D} = 404Hz$, in this case we take D as a distance between the microphone 1 and 8 (D = 7*d = 0.42m)

1.2.3 Radiation from the source

12. Place the source in the axis of one microphone of the array, as close as possible (1 cm) to the microphone capsule and directed towards the microphone under consideration. Generate a pure tone below the limit frequency. Display the recorded waveform and calculate its RMS value.

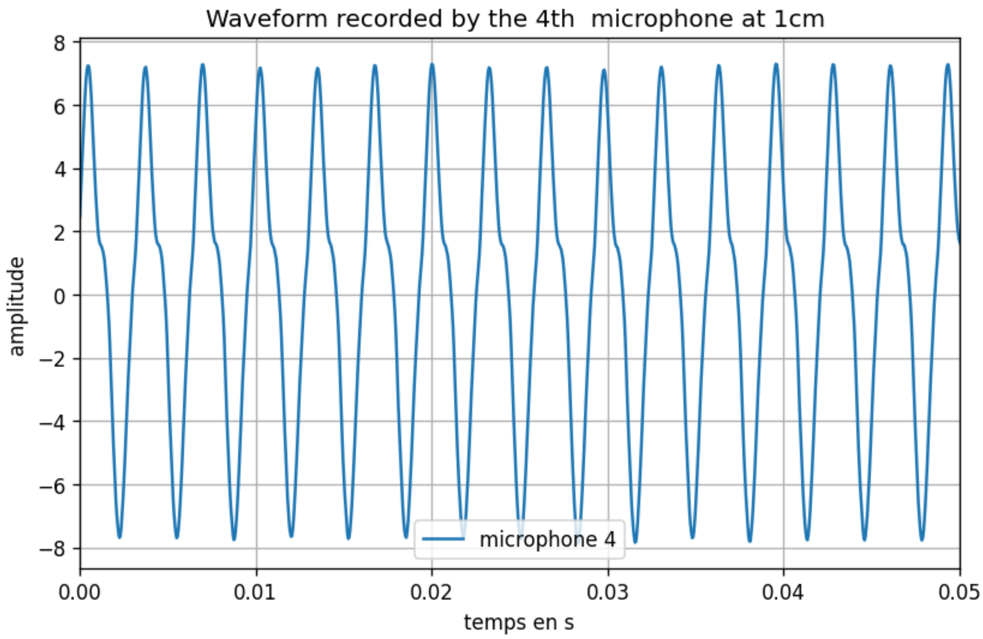

FIGURE 10 – Display of the waveform recorded with frequency of 307Hz, with source toward the microphone

To calculate the RMS (Root Mean Square) which calculates the effective value of the signal, we use the formula : $RMS = \sqrt{\frac{1}{N}\sum_{i=1}^{N} x_i^2}$
where
N : is the number of samples contained in the recording,
x_i : is the value of sample i
RMS = 4.54

13. Increase the distance between the source and the microphone under consideration. Repeat the above steps for different values of this distance.

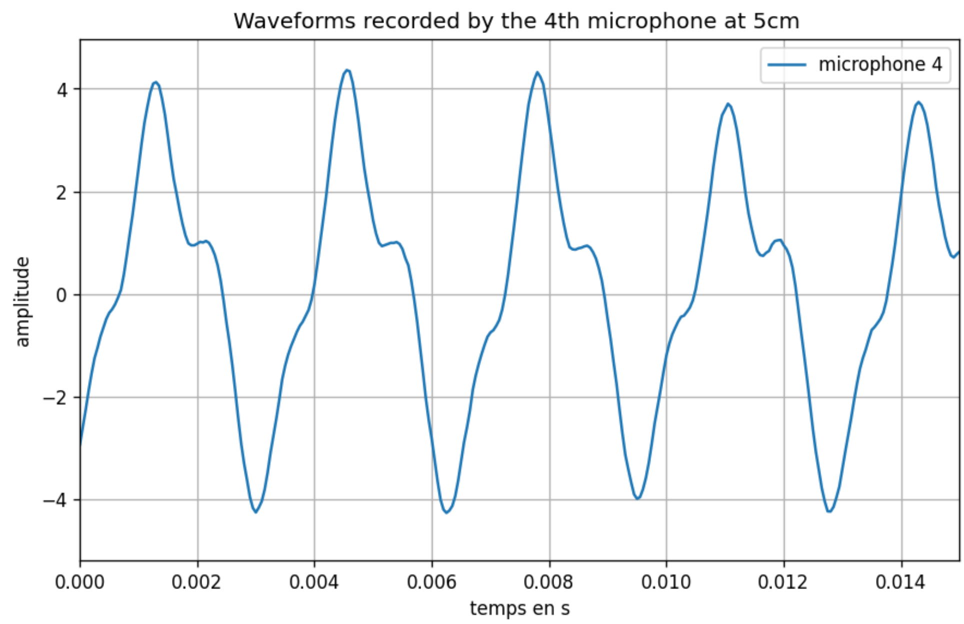

FIGURE 11 – Display of the waveform recorded with frequency of 307Hz, with source toward the microphone at 5cm

RMS = 2.4

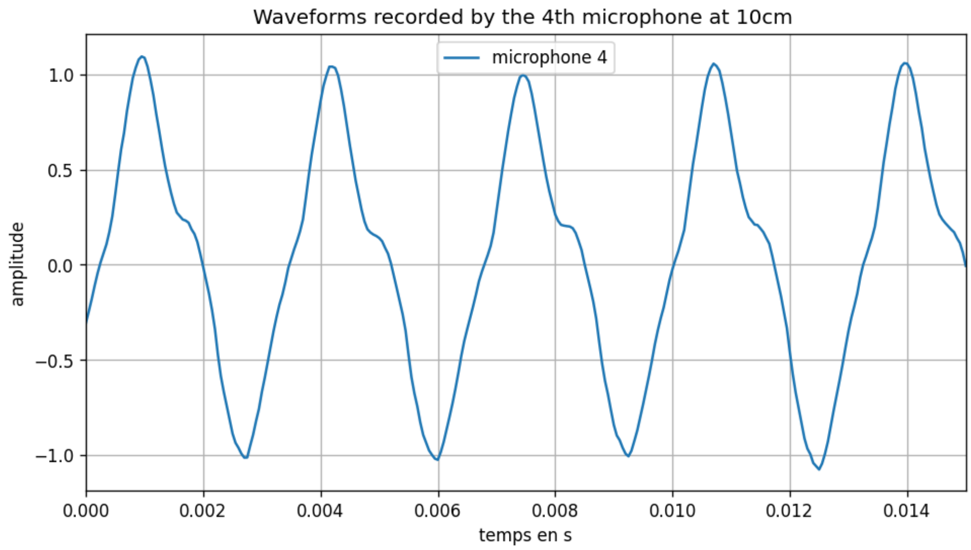

FIGURE 12 – Display of the waveform recorded with frequency of 307Hz, with source toward the microphone at 10cm

RMS = 0.623

14. Plot the RMS values as a function of the distance between the sound source and the microphone under consideration.

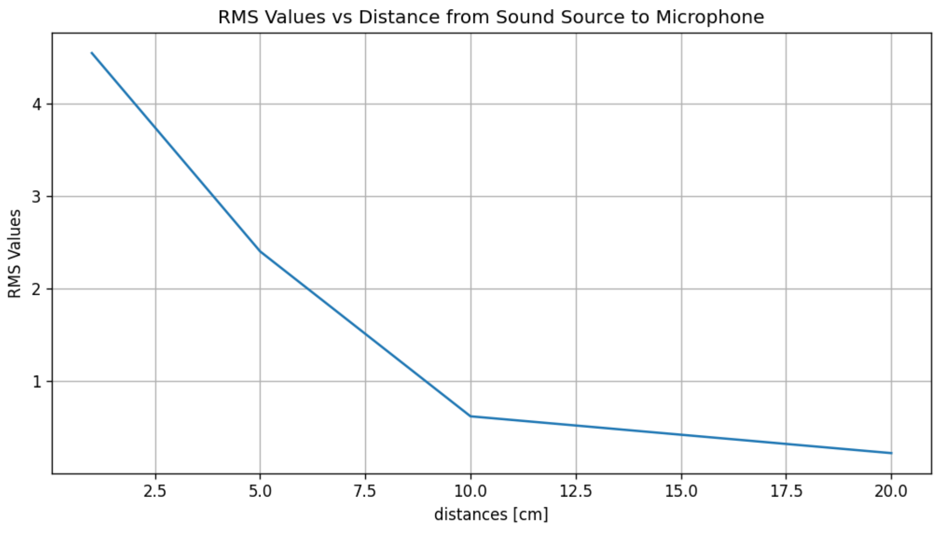

FIGURE 13 – RMS values curve estimated through 4 distances

15. **Interpretation** : Sound propagates in waves that disperse in space. As these waves move away from the source, their intensity decreases. This justifies the fact that the RMS decreases.

1.2.4 Wave fronts

16. Place the source in the direction of the centre of the microphone array, at a distance greater than the length of the array.
17. Read the coordinates (x_s, y_s) of the source and deduce the angle of incidence θ_s of the source, relative to the $[O, z)$ axis of the microphone array.

$(x_s, y_s) = (0, 47\text{cm})$
$\theta_s = 80$

18. Generate a pure tone at a frequency below the limit frequency previously measured. Record and display the waveforms measured by each microphone of the array.

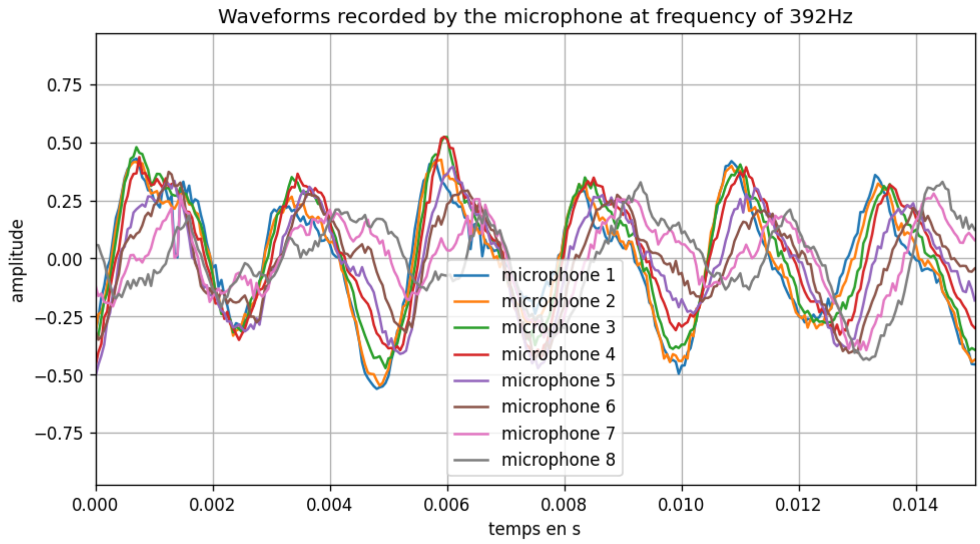

FIGURE 14 – waveforms recorded by the micrphones with source placed in the center of the array

19. Estimate the time delay between each of these waveforms. Deduce for each microphone, the delay δ_n needed by the wave to travel the distance d_n defined in preparation (see Fig.1).
$\delta_{\tau_1} = -0.000107$
$\delta_{\tau_2} = -7.66\ 10^{-5}$
$\delta_{\tau_3} = -4.59\ 10^{-5}$
$\delta_{\tau_4} = -1.532\ 10^{-5}$
$\delta_{\tau_5} = 1.532\ 10^{-5}$
$\delta_{\tau_6} = 4.59\ 10^{-5}$
$\delta_{\tau_7} = 7.66\ 10^{-5}$
$\delta_{\tau_8} = 0.000107$

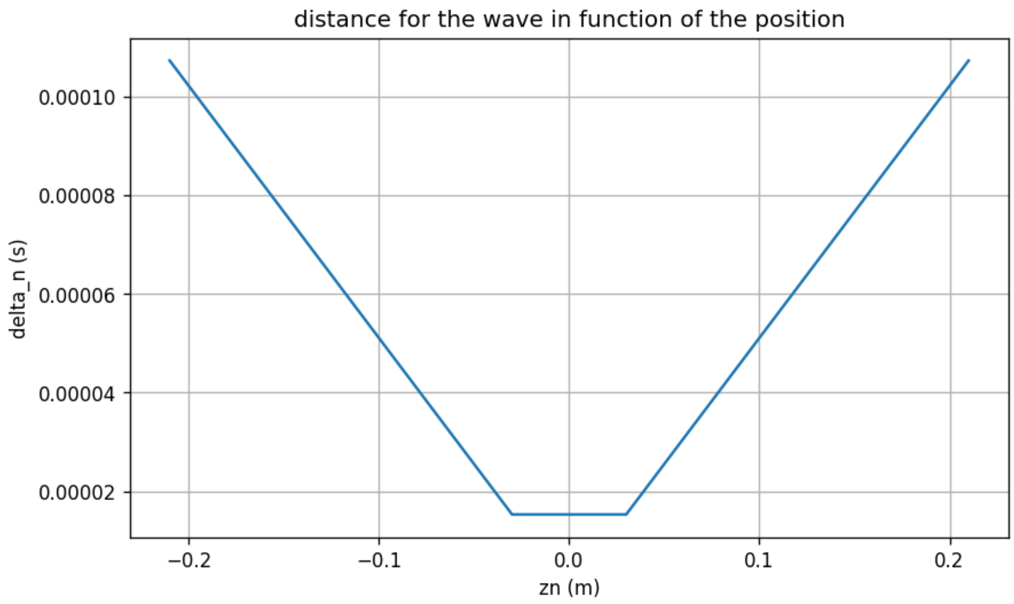

FIGURE 15 – the delay needed by the wave to travel distance

20. Locate the source using these measurements and the equations obtained in preparation.Compare with the real position of the source.
$\theta_s = arccos(\frac{\tau_n.c}{z_n})$
We obtain an approximation of the angle at which the source was, which is about 71°. In reality, we measured about 69° but these measurements are not entirely accurate since we made them with a ruler. Additionally, the consideration of the wave is not totally plane as it is supposed.

21. Place the source towards the origin O, at a distance of the same order of magnitude as the length of the microphone array. Then, repeat all the previous steps

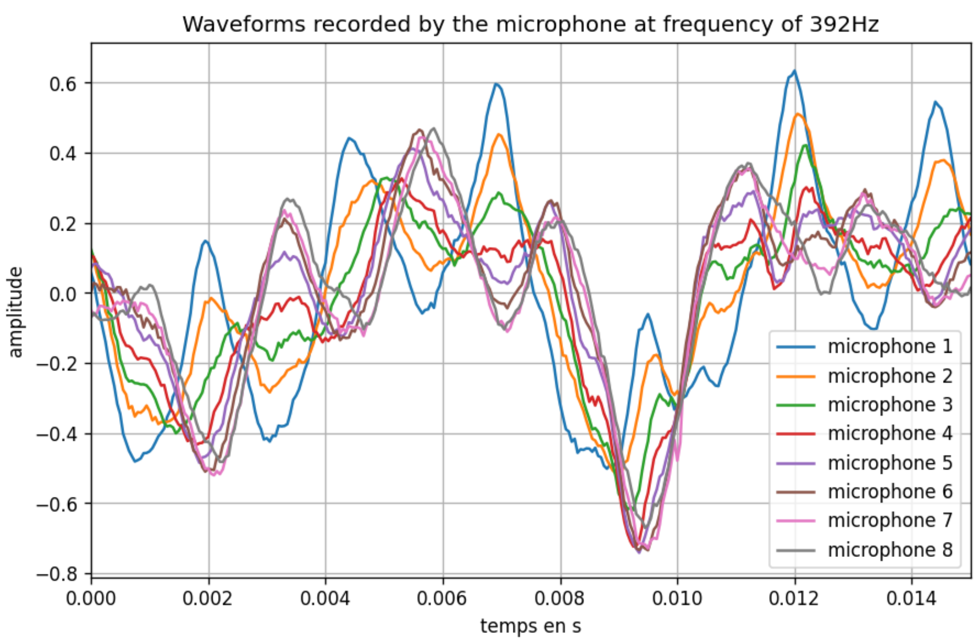

FIGURE 16 – Source towards the origin at the same order of magnitude as the length of the microphone array

22. At what distance from the source to the microphone array can the sound wave be considered as a plane wave ?

    we can consider that the sound wave is a plane if the source is placed at a distance greater than the length of the microphone array.