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Technical Notes

Using Microphone Arrays to Detect Moving Vehicle Velocity

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The noise of motor vehicles is one of the most important problems as regards to pollution on main roads. However, this unpleasant characteristic could be used to determine vehicle speed by external observers. Building on this idea, the present study investigates the capabilities of a microphone array system to identify the position and velocity of a vehicle travelling on a previously established route. Such linear microphone array has been formed by a reduced number of microphones working at medium frequencies as compared to industrial microphone arrays built for location purposes, and operates with a processing algorithm that ultimately identifies the noise source location and reduces the error in velocity estimation.

Keywords: microphone array, sound location, moving sources, vehicle velocity.

1. Motivation and objective

Speed control and processing systems of road vehicles can currently determine traffic speed by using microwave radar systems. These devices can detect objects and determine distance and movement speed by using built-in emitting and receiving antennas. The mechanics of a microwave radar system are in essence simple: the emitting antennas project radio waves continuously onto the road and upon contact with the vehicle in transit, the radio waves are reflected and captured by the receiving antennas, which in turn trigger the start of an internal processing algorithm that produces an estimate of the velocity of the detected vehicle. In a similar fashion, other type of radars can be built based on the same idea, but using laser emitters and receivers. Although these systems have some advantages as they do not need to interrupt traffic for installation and are multilane data collection systems, they also have disadvantages like the possibility of some missed detections if tall vehicles occlude the more distant lanes and mainly that they are easily detected by antiradar systems.

Other systems, like “velocity cameras”, have, in contrast, a slightly different approach to speed vehicle detection. They calculate the average velocity of a vehicle transiting between two points by processing the temporal delay of two or more snapshots of the vehicle taken along the stretch of the road under study. New control systems, such as the DUO and/or Mobile-Vision systems, play with the same principle of capturing images, but use a more advanced technology than the basic “speed camera” system. These systems have main problems with large vehicles, because they can mask trailing vehicles, moreover the presence of shadows, reflections from wet pavement, headlight beams, relative color of vehicles and background, or camera vibration can affect vehicles’ detection.

All existing control devices currently in operation, as well as those that are being developed, aim to reliably detect the speed of road vehicles. Whereas vehicles are traditionally seen as significant noise pollutants, this negative and inherent characteristic could be used to locate and estimate their speed. The use of an acoustic microphone array as a traffic control tool would be an undetectable system which would avoid visual problems and be cheaper than most of those currently used.

As the modeling process and theoretical research of sound sources state, a moving source of real sound placed at a particular distance from the receiver can be considered as a point source, which can be identified by emitting a spherical wave from the physical center of the generating element. Locating the origin of this wave constantly would enable us to establish the position of the vehicle to be analyzed and then its velocity. Microphone array is the most suitable system to identify the angle of arrival of a wave emitted in a particular point of space as their set of sound sensors can determine the origin of a sound wave through gaps in signals received between the elements.

Building on the above idea, the present study investigates the creation of a linear microphone array formed by the fewest possible number of sound receivers to allow an easy installation and a manageable on-site use, but guaranteeing a reasonable degree of accuracy in the results. As one might expect, the reduction of the number of sensors in the array could cause an increase in the error of the results, but such increase can be controlled, as described in this article later on, by the application of a processing algorithm that adjusts the operating procedures and reduces the error to an acceptable limit. Such linear microphone array has been formed by a reduced number of microphones working at medium frequencies as compared to industrial microphone arrays (Kuhn *et al.*, 1998) or directional microphones (Danicki, 2005) built for location purposes.

2. Microphone array background

A microphone array consists of a grid of microphones sampling the sound field at discrete spatial positions. The angle of the array relative to the far field sound source and the strategic position of each microphone within the array cause the plane wave front to hit the receivers with a temporal delay, as Fig. 1 shows. The temporal recording of each microphone needs to be compared to ensure that the measuring signal is the same in every microphone regardless of the specific phase delay of each signal, which depends on the relative position between the transducer and the noise source. The capacity of locating a noise source is known as the Direction Of Arrival, DOA.

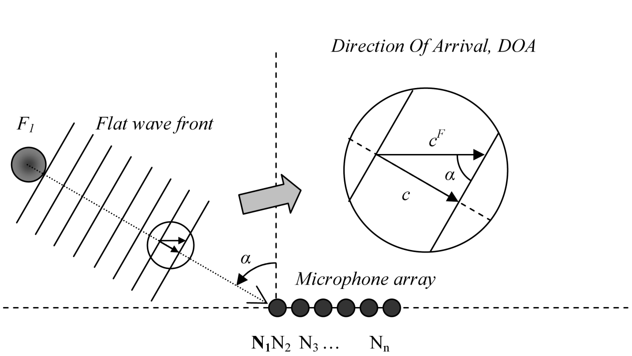

Fig. 1. Noise source wave front consideration for a linear array in the far field.

Developing the optimal microphone array configuration and finding novel applications for microphone arrays are some of the main goals pursued by contemporary scientists specialized in the matter. A clear example of this search is noted in the development of 2-dimensional antennas to detect and identify noise sources in near field, as marketed by Bruel&Kjaer (Christensen, Hald, 2004), and some other technology companies like Bswa Technology from China and Gfai Technology (Döbler *et al.*, 2008) from Germany.

Using simple antennas may give satisfying results. Lopez-Valcarce (2004) and Moran *et al.* (2007) carried out different tests to detect noise sources, localization, and speed through microphone antennas. Other authors (Kodera *et al.*, 2007) suggest a 4-microphone system to locate vehicles in the near field with the purpose of using it in road safety. Using a 2-microphone system (Perez-Gonzalez *et al.*, 2002) there is suggested an algorithmic calculation to detect the position and speed of moving sound sources emitting on a narrow broadband and then optimize results with the Monte Carlo series.

On the other hand, microphone antennas are suggested by Harmonoise work team to locate and identify noise sources of travelling vehicles as well as holographies in the near field by using linear and 2D antennas.

Some applications of microphone array in the far field have been published (Quaranta *et al.*, 2008) and they suggested a model for noise source localization in open spaces using sound sensors placed far away from each other. Other authors (Cigada *et al.*, 2007) developed an experimental test in order to validate a linear microphone array using the delay & sum algorithm.

3. Array design

Characteristics of the environmental test and the noise source determine, to a considerable degree, design restrictions of a microphone antenna as well as the method used to process the incoming data. Important design factors to be taken into account are listed below so the microphone antenna we suggested can achieve all the goals initially established.

3.1. Type of antenna

The spatial distribution of the microphones considered for the antenna designing process will condition its characteristics, its mathematical data processing, and its potential applications. Depending on how sensors are placed in space, systems can be divided into:

- Linear antennas: All sensors are placed on the same spatial line by keeping a distance sequence between each pair of sensors, such as a group of microphones uniformly distributed, Uniform Linear Array, ULA, or an array with a proportional distance between microphones.

- Planar antennas (2 dimensions): In this case, microphones are placed on the same work level. This kind of antennas is commonly used to create holograms in the near field.
- Spatial antennas (3 dimensions): Finally, in these antennas, the relative position between sensors varies depending on a particular volume, being able to detect noise sources moving in a specific volume.

As one of main objectives of this application is to work with the simplest solution and due to the first simulations shown good results, it was established that a linear antenna with uniformly placed microphones will be used, ULA.

3.2. Processing method

Bibliography is composed of an important variety of processing methods: narrowband methods such as "delay and sum"; Capon or MUSIC method; and broadband methods such as the Double Fourier Transform or Spatial Cross Spectrum method.

Results of other research works (Genescà *et al.*, 2009) and previous experiences (Peral, 2009) determine that the most suitable method is the Spatial Cross Spectrum, SCS (Boone, 1987) according to the frequency range of the sound source as well as to the needs of the angular resolution and the number of the antenna sensors.

3.3. Distance between microphones

Distance between microphones would be conditioned by testing frequency and it should fulfill Eq. (1) to avoid spatial aliasing:

$$d \leq \lambda/2 = c/(2f), \tag{1}$$

where d is the distance between microphones, λ is the wavelength of the expected signal, c is the speed of sound, and f is the frequency of the expected signal.

After analyzing spectral characteristics of an average group of passing vehicles (see Fig. 2), it has been detected that 1000 Hz was the frequency band that kept more sound power, which allowed us to determine that the distance between microphones for the narrow band system should be not more than 0.17 meters to avoid spatial aliasing.

3.4. Distance between source and receiver

To assure a correct identification, the proposed test should be carried out under the hypothesis based on a point source emitting in the far field. To do so, the minimum distance between source and receiver has to be established so these assumptions can be guaranteed. Different authors (Maekawa, 1970) state that a noise source can be considered a point-like one when the distance from the receiver shows a minimum value that depends on source size and the frequency of an emitted wave. Given all the size characteristics of the tested source and the location frequency, the reference distance between the microphone array and the closest

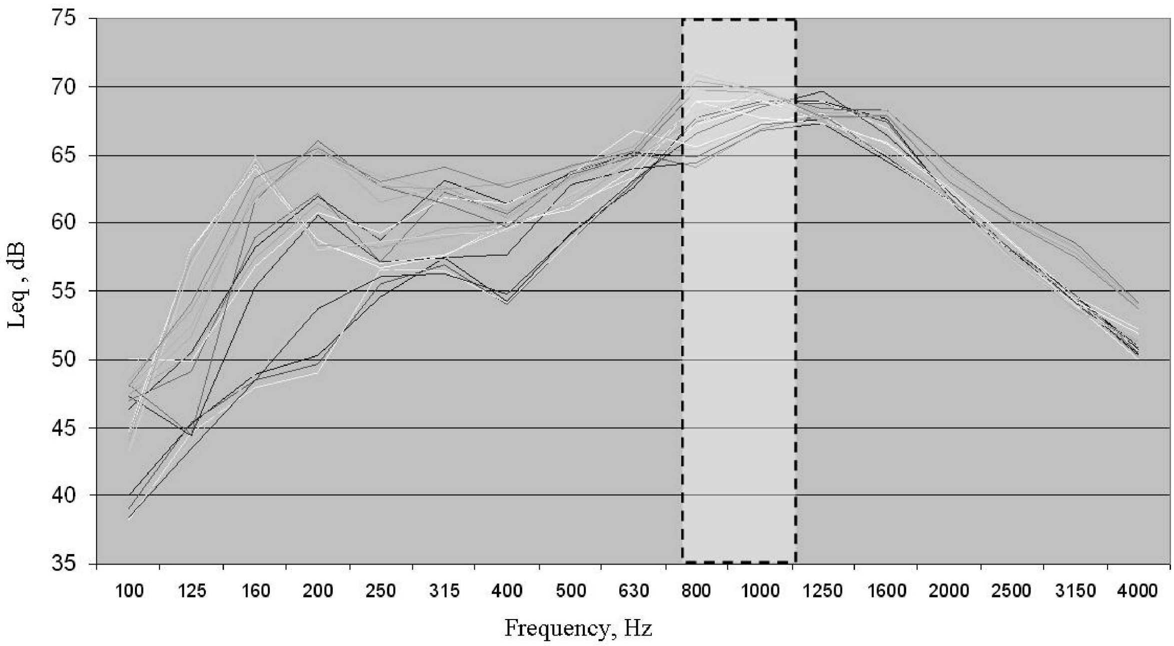

Fig. 2. 1/3 octave band noise spectrums of a measured group of passing vehicles with different characteristics used to detect main working frequency.

point to the travelling trajectory is 30 meters and the length of the test track in which the source will be detected is 300 meters (see Fig. 3).

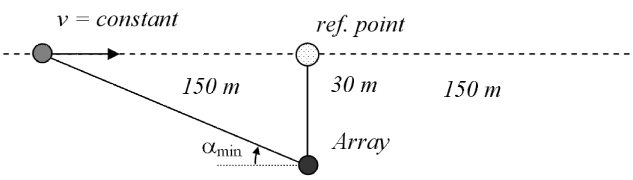

Fig. 3. Schematic sketch of the test and the minimum location angle.

3.5. Number of microphones and angular resolution

The angular resolution of a linear array will be determined by its total aperture or length capacity as well as the frequency of the captured signal. After establishing the distance between the sensors, the total aperture capacity will be directly determined by the total number of array microphones. When there are more sensors, the angular resolution of the array is less, which makes localization results even more reliable.

Angular resolution is determined by the antenna steering vector and it varies depending on the focal angle as well as on the processing method used. The angular resolution for SCS processing method will be as follows (Cigada *et al.*, 2007):

$$\Delta\alpha = \frac{\lambda}{Ap\cos(\alpha)} = \frac{c}{2L\,f\cos(\alpha)}, \tag{2}$$

where $\Delta\alpha$ is the angular resolution of the microphone array, Ap is the aperture of the array, L is the total length of the microphone array, f is the frequency of the signal, and α is the focal angle.

Considering that during the test, no other significant noise source would be in the stage and knowing that the extreme angle of the noise source will be, correspond with Fig. 3, $\alpha_{\min} = \arctan(30/150) = 0.197$ rad with respect to the antenna, the minimum length of the antenna should fulfil the condition

$$\frac{c}{2L_{\min} f\cos(\alpha_{\min})} \leq \alpha_{\min}, \tag{3}$$

giving, as a result,

$$L_{\min} \geq 0.89 \text{ m}. \tag{4}$$

Knowing that the distance between microphones is 0.17 m, the antenna should need at least 7 microphones to reach the minimum distance, $L_{7\text{-micros}} = 1.02$ m. The resulting beam pattern from a 7-microphone antenna at working frequency of 1000 Hz is shown in Fig. 4.

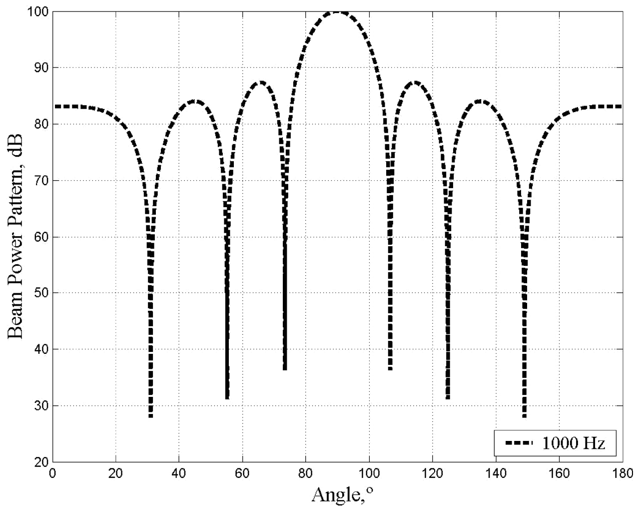

Fig. 4. Beam Pattern of the ULA of 7 microphones at 1 KHz, depending on the source location angles.

3.6. Linear antenna direction

To determine antenna direction that guarantees the best results on the angle of source arrival, angular variation of linear microphone distribution will be taken into consideration. Resolution of a linear antenna will decrease as the source focal point moves away from the zero angle (seen itself as the center of the trajectory of the travelling vehicle to the perpendicular direction). Two alternatives were then suggested for the linear layout: sensor system in a perpendicular line to the vehicle trajectory, *Perp*, and antenna parallel to the travelling trajectory, *Parl*. Both layouts have been tried out to determine the most suitable one for the proposed test, as shown in Fig. 5.

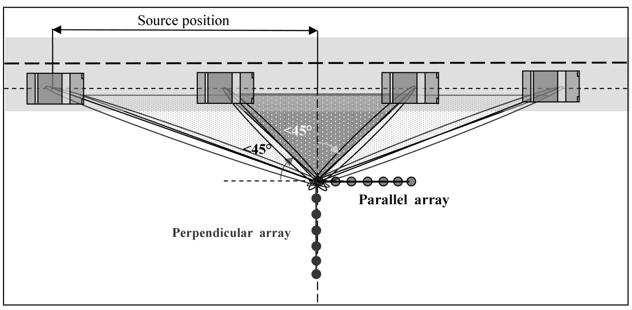

Fig. 5. Trial zone for the two different microphone positions.

3.7. Sampling frequency

As the main location frequency was considered to be 1000 Hz, a sampling frequency, f_s, of 10000 Hz would be enough for testing purposes. This way we would avoid problems such as temporal aliasing and using vectors with a too large data content which would make it difficult to obtain data as well as its subsequent processing.

4. Data processing

Figure 6 shows the process diagram of data processing.

Each sensor of the operating antenna will provide a data vector, $S\prime_n(t)$. These will then be filtered and processed to obtain the instant velocity of the sound source. Signals will initially be filtered, $S_n(t)$. Taking into account the range in which the vehicle will provide us with the highest amount of sound energy, like the Doppler Effect, a FIR filtering called "band-pass" is carried out between 940 and 1060 Hz frequencies for each signal captured by each transducer.

All signals obtained are divided into K snapshot blocks (see Eq. (5)), which will be analyzed separately assuming a fixed position of the source when capturing. Each block will have an enough number of snapshots to carry out the correct analysis and will be as small as possible to minimize source movement when capturing. To stop the source from varying its position abruptly, and knowing that a Hanning window will subsequently be used to reduce values on extreme sides, samples of $0.1 f_s$ snapshots were taken (this amount of data guaranteed that the vehicle movement was less than 2 meters for every measuring instant):

$$K = \frac{\text{Time}}{0.1\, f_s}, \tag{5}$$

where Time is the testing total time, f_s is the sample frequency, and K is the number of snapshot blocks.

Each average time, t_a, will be the new reference time for data blocks. Namely, data block j, delimited by an initial time $t_{i,j}$ and a final time $t_{f,j}$, is associated with time $t_{a,j}$ defined as

$$t_{a,j} = \frac{(t_{i,j} + t_{f,j})}{2}, \tag{6}$$

where j takes values from 1 to K.

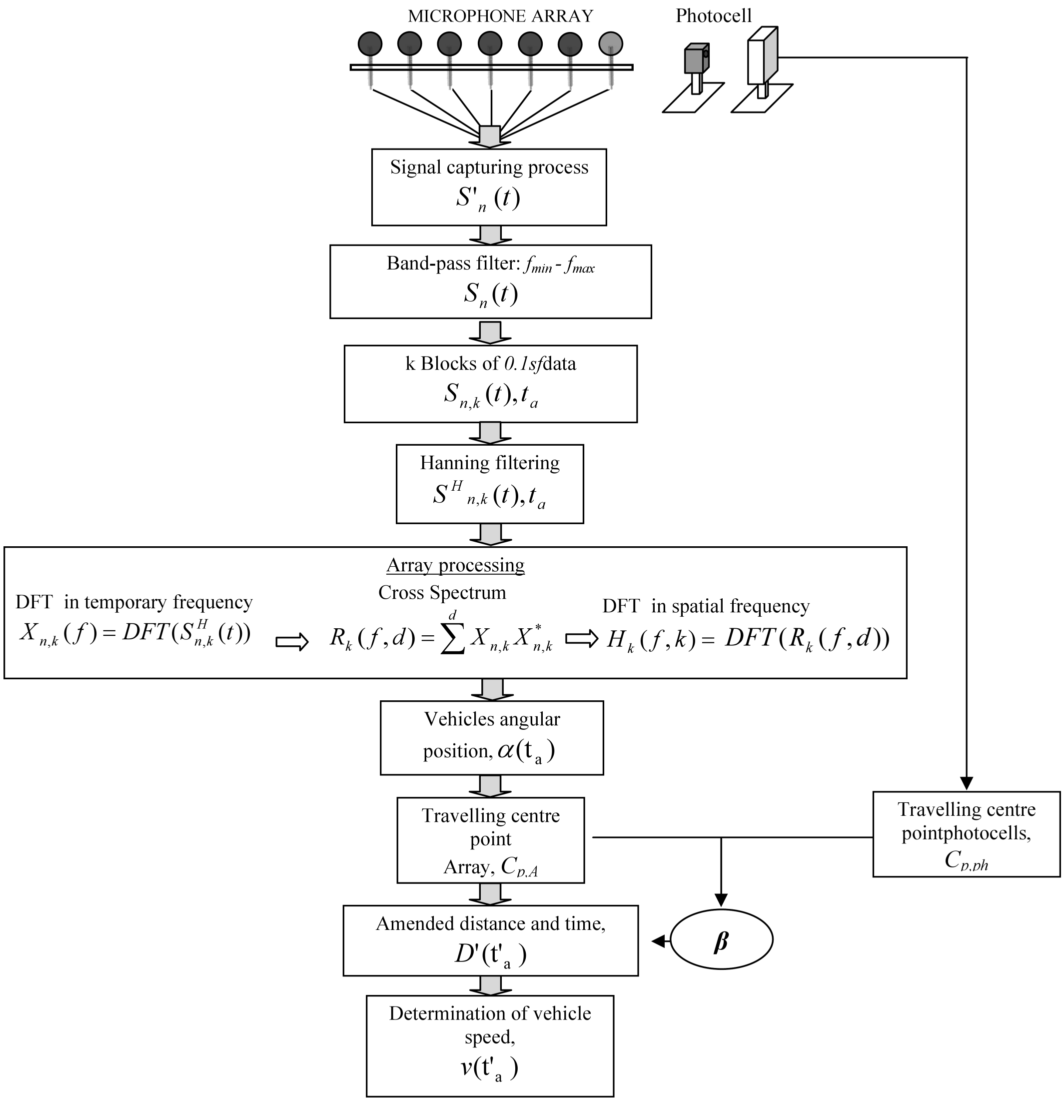

Fig. 6. Diagram of signal processing flow to obtain the vehicle travelling speed through a microphone linear antenna.

Each block then goes through an IIR filter, of the Hanning window type, to avoid leakage problems before going through Spatial Cross Spectrum algorithm (Jonhson, Dudgeon, 1993). As a response, the algorithm gives a localization function from which the direction of arrival of the main source can be inferred. Each block will therefore have the value of the direction of arrival of the signal, $\alpha(t_a)$.

The direction of arrival of the sound wave will lead to the travelling velocity of the vehicle during the test. Consequently, deviations in the source position are amended due to possible angular variation between the array and the travelling line of the sound source (to do so, a given time interval should be found by using a couple of photocells which will lead us to the exact vehicle position), wave displacement time from the emitting source to the array, and point sound sources in the area being able to move the antenna focalization away. To adjust these results, an algorithm has been implemented to locate the vehicle travelling line and get rid of all samples that have not detected it as main noise source, working as a 'clean function' (see Fig. 7).

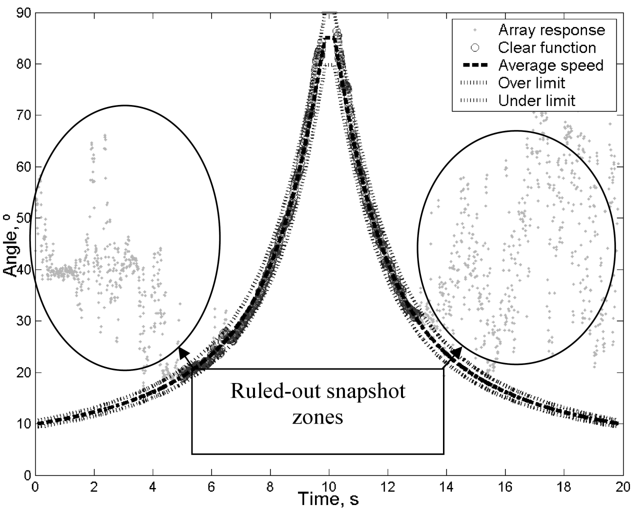

Fig. 7. The zones bounded by the vehicles average speed rule out points giving no information on sound source localization, *'clean function'*.

Given the vehicle trajectory and the angular position for any moment in time $\alpha(t_a)$, displacement will be obtained as

$$D(t_a) = \frac{dist}{\tan(\alpha(t_a))}, \tag{7}$$

where $D(t_a)$ is the theoretical displacement of the noise source, *dist* is the constant perpendicular distance between the microphone array and the source track, and $\alpha(t_a)$ is the focusing angle of the antenna of each data block.

After being adjusted as previously mentioned, the corrected value will be

$$D'(t_a) = \frac{D(t_a)\,\sin(\alpha(t_a))}{\sin(\pi - (\alpha(t_a) + \beta))} = \frac{\cos(\alpha(t_a))\,dist}{\sin(\pi - (\alpha(t_a) + \beta))}, \tag{8}$$

where $D'(t_a)$ is the real displacement of the noise source and β is the correction angle (angular difference between real direction and theoretical direction of the antenna, see Fig. 8).

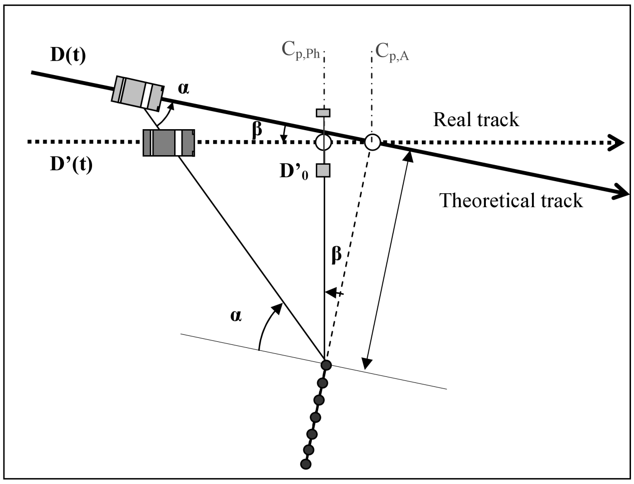

Fig. 8. Sketch of the angular correction to obtain the real displacement of the noise source.

Likewise, the theoretical distance R that the sound wave has to travel to arrive at the microphone antenna is

$$R(t_a) = \frac{dist}{\sin(\alpha(t_a))}. \tag{9}$$

But considering the correction angle, the real distance R' between the sound source and the receptor is

$$R'(t_a) = \frac{dist}{\sin(\alpha(t_a))} \pm \frac{D'(t_a)\sin(\beta)}{\sin(\alpha(t_a))} = \frac{dist \pm (D'(t_a)\sin(\beta))}{\sin(\alpha(t_a))}. \tag{10}$$

So actually, the angular function will provide the position of the vehicle at the instant

$$t'_a = t_a - R'(t_a)/c = t_a - \frac{dist \pm (D'(t_a)\sin(\beta))}{c\,\sin(\alpha(t_a))}, \tag{11}$$

where c is again the speed of sound.

Linking this position with the moment in time in which the vehicle is traversing the center of the real travelling line reference D'_0, an average variation of the position will be obtained as

$$\Delta D'(t_a) = |D'(t_a) - D'_0|\,. \tag{12}$$

Taking the velocity definition into account,

$$v(t_a) = \frac{\Delta D'(t_a)}{\Delta t'_a} = \frac{|D'(t_a) - D'_0|}{|t'_a - t'_0|}. \qquad (13)$$

Due to the antenna features and the test itself, results will fall into a margin of error which will be minimized by using an approximation of least squares. In order to achieve this, a speed value range between 10 and 30 m/s should be used and the obtained speed vector would be $v_k = [10, ..., 30]$.

Taking the center of the vehicle travelling line into account, the velocity value is established and minimizes the difference between squares of the snapshot block captured in 0.5 seconds.

Data processing was implemented in MATLAB and tested in a simulation as it is explained in the next section.

5. Simulation

To guarantee that this microphone system operates accurately, tests under different assumptions have been carried out. To achieve this, the simulation was based on a point sound source on the move emitting pink noise, proceeding all the way along a straight line at a constant velocity. The distance covered was 300 m and the microphone system placed at a distance of 30 m. During the simulation, parameters such as displacement velocity and sound intensity of background noise were varied. Figure 9 shows signals of the reference microphone (first microphone of the linear array) for the 3 assumptions.

Figures 10 and 11 show results of simulations obtained through the antenna placed perpendicularly and parallel to the vehicle travelling line. The background noise changes for each assumption as Table 1 shows, and affects source localization when it is placed far away from the capture system. In both cases, it is possible to find a limit angle from which the antenna results are not accepted because of high deviation between results and real source position. However, because of the area of low angular resolution, parallel antenna determines sources position during a lower time period.

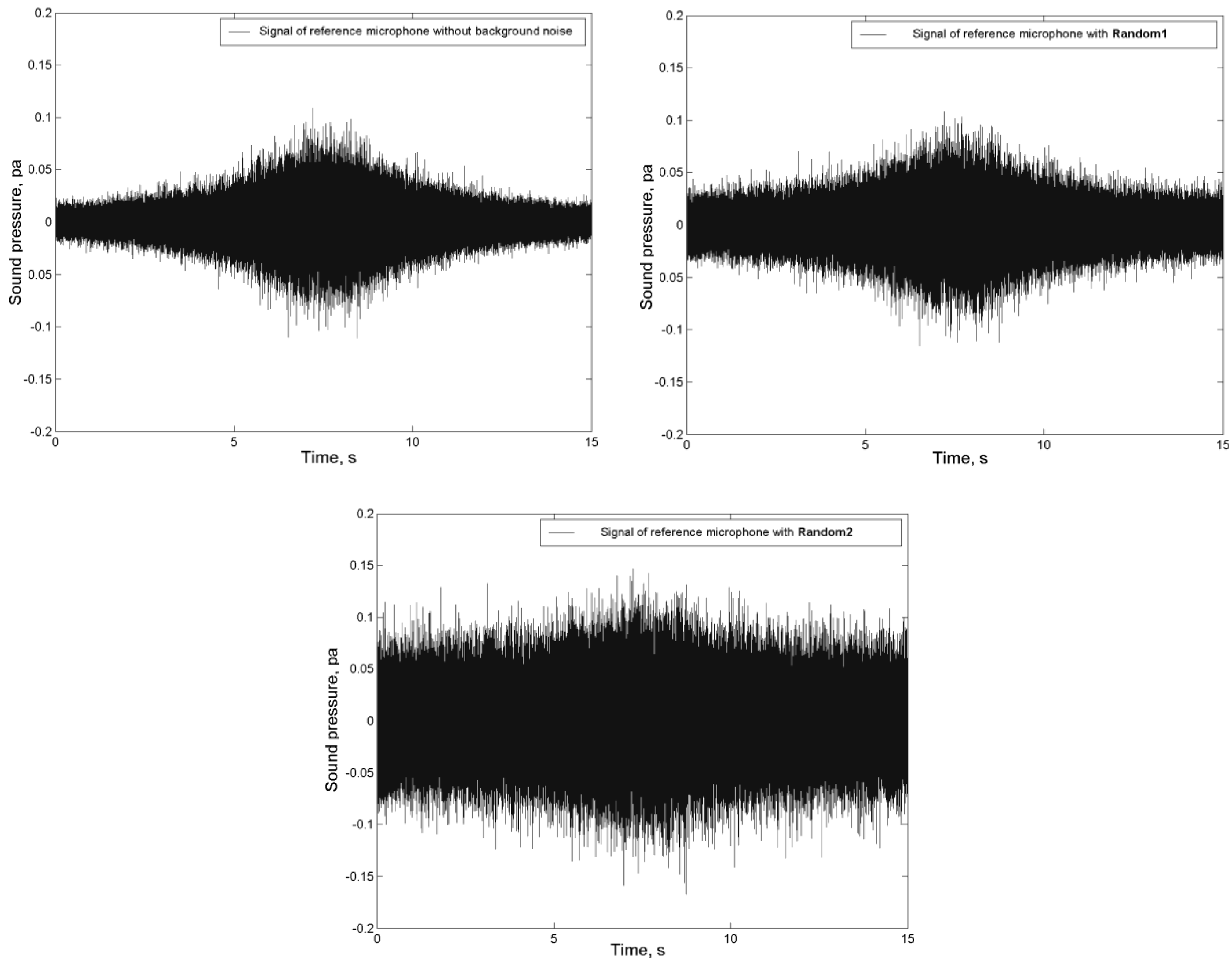

Fig. 9. Signals simulated.

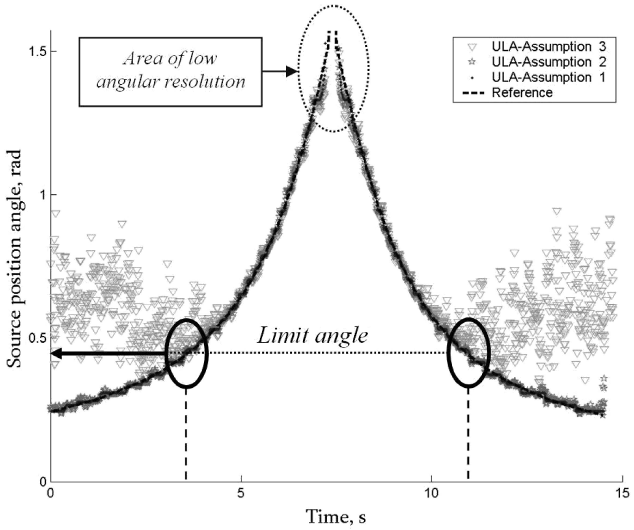

Fig. 10. Results of simulation of ULA-7 placed perpendicularly to the vehicle travelling line.

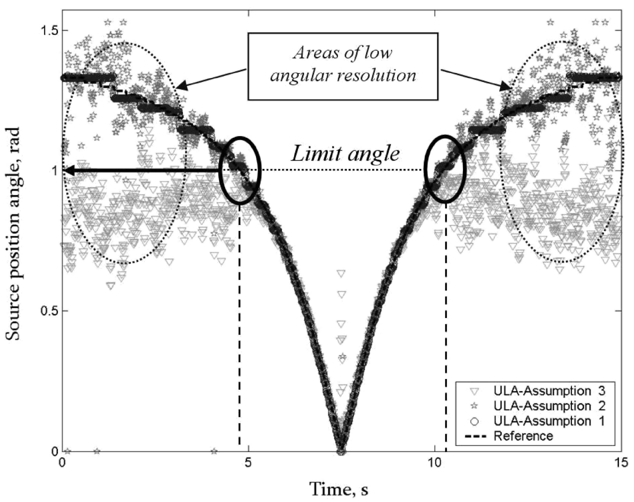

Fig. 11. Results of simulation of antenna parallel to the travelling trajectory.

Table 1. Assumptions simulated.

	Velocity [m/s]	Sampling Frequency [Hz]	Background noise
Assumption 1	20/30	10,000	*Null*
Assumption 2			*Random1* (9 dB lower than emitted in ref point)
Assumption 3			*Random2* (same level like emitted in ref point)

Table 2 contains the mean squared errors (10) of the results obtained from every simulation carried out. For each assumption, calculations have been performed by using the position values given by the algorithm which have been compared with the real source position:

Table 2. Mean Squared Error obtained through both antennas and for each of the assumptions.

	ULA-L [rad]	ULA-II [rad]
Assumption 1	0.0149	0.016
Assumption 2	0.018	0.0942
Assumption 3	0.223	0.3041

$$MSE = \sqrt{\sum_{i=1}^{N} \frac{(\alpha_{\mathrm{exp},i} - \alpha_{\mathrm{real},i})^2}{N}}, \tag{14}$$

where MSE is the Mean Square Error, N is the number of results obtained, $\alpha_{\mathrm{exp},\,i}$ is the experimental angle of each instant of time, and $\alpha_{\mathrm{real},\,i}$ is the real angle of each instant of time.

Uniform linear array works better in orientation perpendicular to the vehicle travelling line, obtaining a narrow deviation in vehicle position for different background noise conditions.

Following the data processing procedure suggested, vehicle speed can be established if position data given by the system are considered. Finally, Fig. 12 presents speed values for assumption 2 as well as deviation observed at a constant speed of 20 m/s.

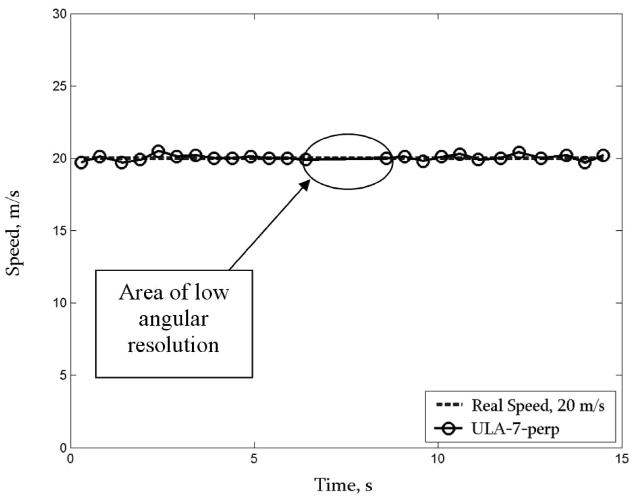

Fig. 12. Source speed obtained by processing results of ULA-7 with antenna perpendicular to the vehicle travelling line for assumption 2.

6. Conclusions

This article analyzes main aspects of the designing process of a linear microphone antenna for vehicle localization in higher speed traffic conditions. Design tasks included the study of different types of microphone arrays, distance and number of sensors, direction and distance of the antenna with respect to the travelling line of the sound source. The antenna eventually suggested has 7 omnidirectional microphones, strategically placed 0.17 meters away from each other, and set up perpendicularly to the travelling line 30 meters away from the closest point of it. The algorithm that has been designed for data processing is based on filtering and conditioning of signals captured by different microphones, implementation of signals through the calculation method *Spatial Cross Spectrum*, proposal for a system that reduces deviation of results, and obtaining instantaneous function with the velocity source based on mean squares.

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