

Noise localization using a low-cost microphone array

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ABSTRACT

Identifying where a sound is coming from can be a complex task when not using the correct tool. Array signal processing techniques can be a solution for sound localization but are computationally intense and require extensive hardware and advanced software. However, many noise localization applications do not call for the accuracy the before mentioned techniques offer. If the goal is to simply give the direction of noise sources, simplified methods and hardware are needed. Thus, an array of 16 microphones arranged in a circle to establish sound pressure level directivity is presented in this paper. The hardware is controlled via an App. The App computes the location of a sound source in the horizontal plane via triangulation. A color map on a 22-degree grid shows the sound pressure level distribution in 360 degrees. Applications in monitoring stations show reliable results to determine the direction of noise sources. The principal advantage of the system is the simplified implementation of typical beamforming applications, making it easy to use and cost-efficient.

Keywords: Noise localization, Microphone array, Signal processing

I-INCE Classification of Subject Number: 71

1. INTRODUCTION

Sound source localization is a wide topic covered for years and many signal processing techniques were developed for that purposes. Microphone arrays allow to do a spatial sampling of a front wave giving more information about the nature of a sound than using only one microphone. Acoustic cameras and antennas use this principle of operation applying signal processing techniques such as beamforming to localize sound sources in 2D and 3D environments. Therefore, the implementation of this devices can involve complex systems able to process up to 120 channels of microphones increasing the computational requirements of the hardware and advanced software. Due this, the systems can be cost-elevated and difficult to carry because of its dimensions.

However, many noise localization problems do not call for the accuracy that an acoustic camera can offer for beamforming applications. If the objective is to simply know where a noise is coming from simplified methods and hardware can be applied.

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Situations like a road near a railroad or a harbour can demand just only identify whether the major contribution of sound pressure level is coming from the cars traffic, the passing of the train or the loading and unloading of ships.

Thereby a simplified and innovated system is proposed in this paper to solve many typical beamforming applications. The system is composed of a sensor controlled by an App to perform measurements and show results. It is able to identify the direction of sound sources in a horizontal plane. The location of them can be estimated by triangulation of several sensor positions within a measurement.

The new system presented in this paper brings straightforward and reliable solutions to typical beamforming applications with the advantage of the simplified implementation making it easy to use and cost-efficient.

2. PRINCIPLE OF THE SYSTEM

Delay-and-sum beamforming is the simplest technique applied for sound source localization [1]. When a sound wave reaches an array the same signal is received by each microphone but at different times depending on the position of the radiating sound source. Knowing the distances between microphones and sound source, and the velocity of sound is possible to calculate the individual time delay to align in phase all the signals received by all the microphones. Then, the summation of the channels is performed to arise the desire signal and attenuate other sources coming from other directions. In devices such as acoustic cameras the image plane of interest is divided into a variety of small individual surfaces (pixels). The time delays are calculated for each pixel and the sum is performed. The sound pressure level (SPL) is sorted in a color scale and assigned to the current pixel. Thus, the virtual plane is reconstructed and only for the pixel where the sound source is located, the signals are perfectly aligned. The result gives a color map representing the sound pressure level distribution in the image plane.

Despite the simplicity of the delay-and-sum beamforming principle, in practice it can demand high computation time and hardware requirements. For instance, the time delays are calculated using the digitalized signals from the microphones. This imply that the running steps are defined by the sample rate F_s of the AD converter but usually the time delay is not an integer of $1/F_s$. Due that, commercial systems can use sample rates up to 192 kHz to have the enough accuracy, making them complex. Another way of doing that is by interpolation. The advantage of this technique is more accuracy in time, lower sample rates can be used but the time consuming make it slow for real time applications [2].

Considering the limitations above mentioned among others, a simpler and innovative technique is proposed. Often, many typical noise localization problems do not require the accuracy of beamforming to give efficient solutions. In this context, the principle applied in this paper for sound source localization is inspired in beamforming. Instead of using the arriving time delays, the amplitude differences between the channels of the array are evaluated. An acoustic sensor called PUCK is presented to apply this principle. It consists in a 16 microphones array disposed in a circumference with a slightly different design than conventional arrays. The microphones are pointing outward the circumference in the same plane covering 360 degrees.

2.1. MICROPHONE ARRAY DESIGN

The finalized version of the PUCK sensor is shown in Figure 1. Condenser capsules of ¼” are used for each channel attached to an exponential aperture, covering 22 degrees of a circumference with 20 cm diameter. A pre-amplifier drives the signal received by each microphone and all the channels are digitalized by a 24 bits AD converter. An intern algorithm evaluates the amplitude differences between channels to determine the direction where the highest sound pressure level is coming from in a horizontal plane. Each microphone measures the pressure in 22 degrees to represent all together the complete circumference. Then an angle value is obtained in degrees showing the direction where the sound source is located. The biggest arrow impressed on the top of the sensor chassis indicates the zero-degree reference as it can be seen in Figure 1.



Figure 1 – Array of 16 microphones disposed in a circumference of 20 cm diameter (PUCK)

The PUCK sensor measures 6 angle values per second, giving a dynamic description of an acoustic event, and they are stored in an intern SD card. The device is controlled by an App via Wi-Fi. A battery gives the sensor up to one entire day of autonomy.

2.2. CALIBRATION

The calibration process is an important stage to provide reliable results. An acoustic calibrator is used to adjust all the channels with the same sensitivity. Each microphone was excited with a 1 kHz sinusoidal tone of 94 dB and using an oscilloscope the same output level was achieved in all the channels. The output waveforms were observed in the display to ensure that no distortion was introduced. This is an important step considering that the intern algorithm of the sensor evaluates the amplitude level differences between channels to set the direction of a sound source.

2.3. SOFTWARE

The software for the PUCK sensor is an App developed for mobile devices. It controls the sensor to perform measurements and store the results. The wireless connection is via Wi-Fi through a hot spot emitted by the PUCK. The App has two functionalities: the first is to operate with a single sensor and the second to triangulate several sensor positions within a measurement.

The first configuration controls only one sensor to perform individual measurements. Two displays modes are available for real time operation and to show the results. These are the direction mode and segment mode, and they can be seen in Figure 2.

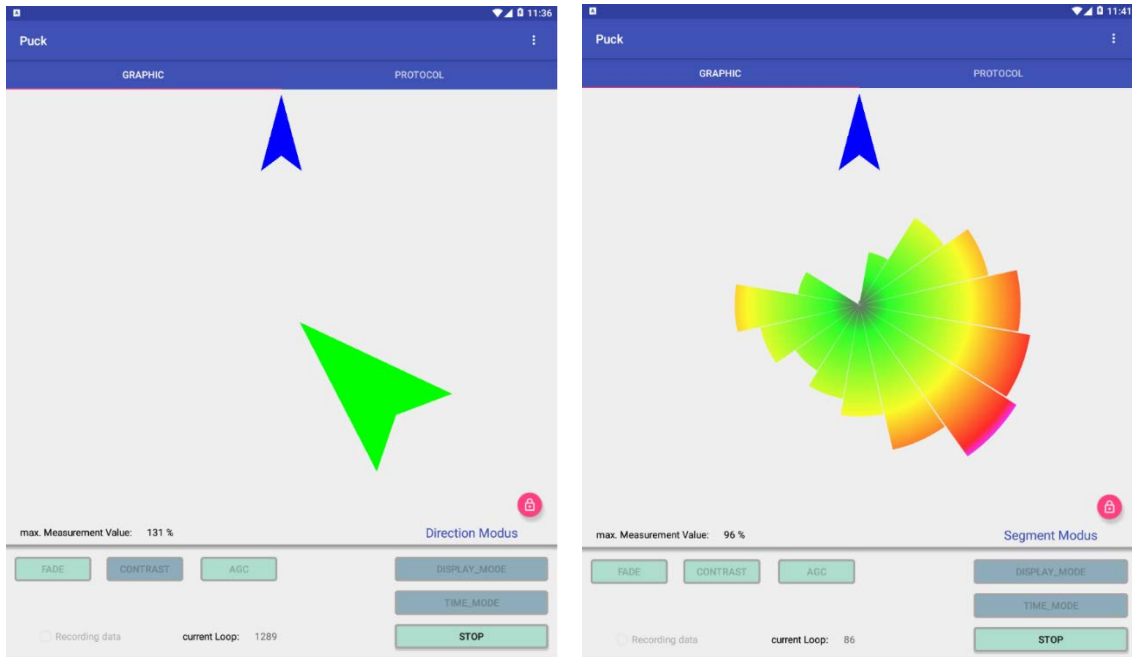


Figure 2 – Display screens of the PUCK’s software. On the left: Direction mode. On the right: Segment mode.

The direction mode an arrow to clearly identify the direction where the loudest sound pressure level is coming from (left). For both cases the small arrow on the top of the screen indicates the zero-degree reference plotted in the sensor chassis. In the segment mode not only the primary direction is shown but also other ones where the sound pressure level contribution is lower. This can be seen in Figure 2 (right) where the primary direction is indicated (same as the direction mode) but also is possible to detect a smaller contribution coming from the left side. This mode uses 16 segments to represent the 360°. The amplitudes are associated to a color scale where the red represents the maximum level and a background image of the area can be introduced. A results file (.txt) is generated by the App and stored in the SD card with the instantaneous angle values calculated during the all measurement time. The amplitude values of the 16 channels are also included.

However, both display modes only show the directions of sound sources but not the distances to them. For this purpose, another feature of the software computes the location of a sound source via triangulation of several sensor positions. The angle values measured at each position are combined to determine the source location at every instant. If measurements are taken for some minutes or several hours, an integration time can be adjusted to have the representative values. For dynamic events, either moving sources or non-stationary noises several sensors must be used at the same time. On the other hand, for stationary noises only one sensor can be placed at the desired measurement points. The results are shown as a set of images representing the acoustic event. Figure 3 shows an example of this, with the triangulation of a sound source location by using four sensor positions. A background image of the measurements area is also included. The color scale represents the SPL radiation in decibels full scale (dBFS) assuming free field conditions.

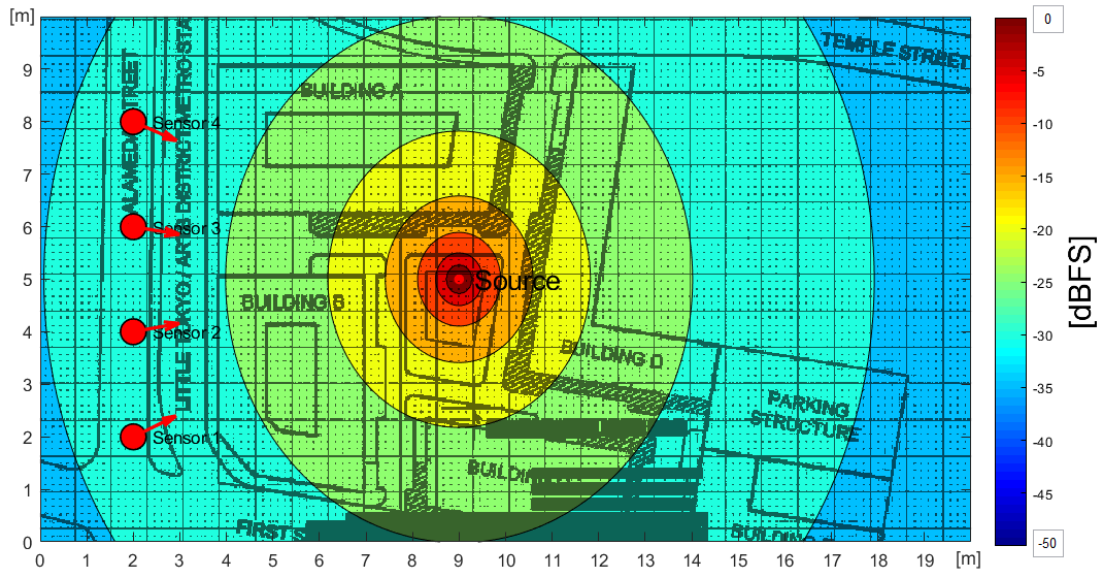


Figure 3 – Source triangulation by using 4 measurement points. The red circles are the sensor positions with the direction arrows. The background image represents a typical situation of urban noise. The SPL is indicated by the color scale in dBFS.

The measurement area is composed by streets and buildings, showing an example of a typical situation for urban noise localization.

3. TEST MEASUREMENTS

3.1. Measurement I

The first one was using four sensors to triangulate the location of a moving sound source. The sensors were disposed in line every 2 m of separation and a broad band source was displaced from a point A to a point B. The experiment was carried out in a conference room. Figure 4 shows the measurement setup and the room dimensions.

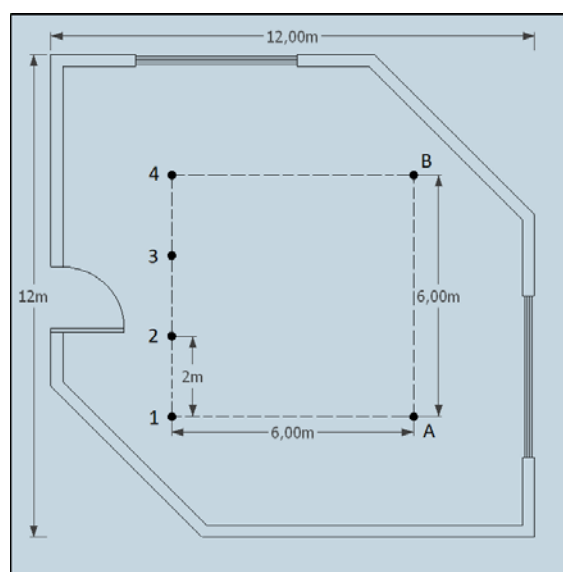


Figure 4 – Measurement setup with the sensor positions (1, 2, 3, 4), the source path (A to B) and room dimensions.

Since only one sensor prototype was available, the measurement was carried out placing it at each position at a time. The broad band source was displaced from point A to point B and the PUCK saved the angle values describing the path. Then, the sensor was placed at position 2 and the process repeated until complete the four measurement points. The time of the source displacement from A to B was 20 s at constant speed in order to have repeatability of the acoustic event.

3.2. Measurement II

The second measurement was carried out using an individual sensor to identify the sound pressure level directivity. The PUCK was attached to a sound level meter (SLM) class 1 to measure noise inside a classroom (4 x 6 x 2.5 m). A broad band source was moved around the room and the SPL was recorded. Figure 5 shows the scheme of the setup measurement and the sound source path. The zero-degree reference coincide with the microphone position of the SLM. A measurement of 1 minute was performed with a slow integration time and A-weighting. The background noise was also measured.

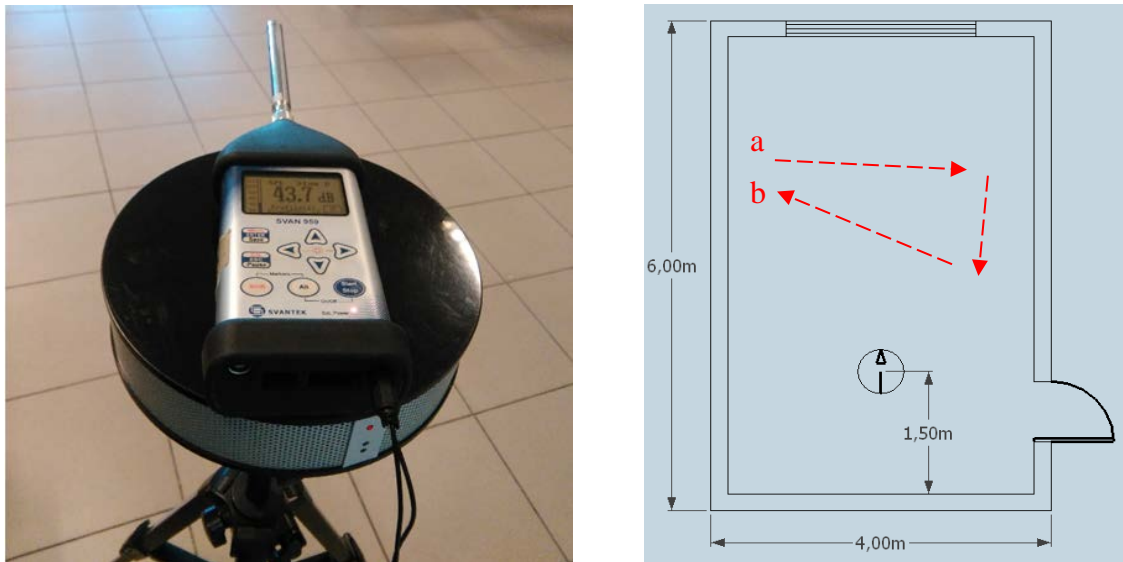


Figure 5 – Left: PUCK sensor attached to a sound level meter class 1. Right: scheme of the classroom with the measurement point and source path (from a to b).

4. ANALYSIS OF THE RESULTS

The result of measurement I is shown in Figure 6. The sequence of images represents the dynamic behaviour of the sound source moving from point A to point B. The instantaneous angle values were averaged introducing an integration time of 4 s. The system was able to reconstruct the original path of the sound source. It can be seen that some frames have the source located in the expected path (from A to B) but some of them are out of it. This is due to instabilities introduced in the angle values provided by the sensor. The sound reflected on the walls affects the calculation of the direct sound directivity. In consequence, the triangulation of the sound source can result out of the expected path.

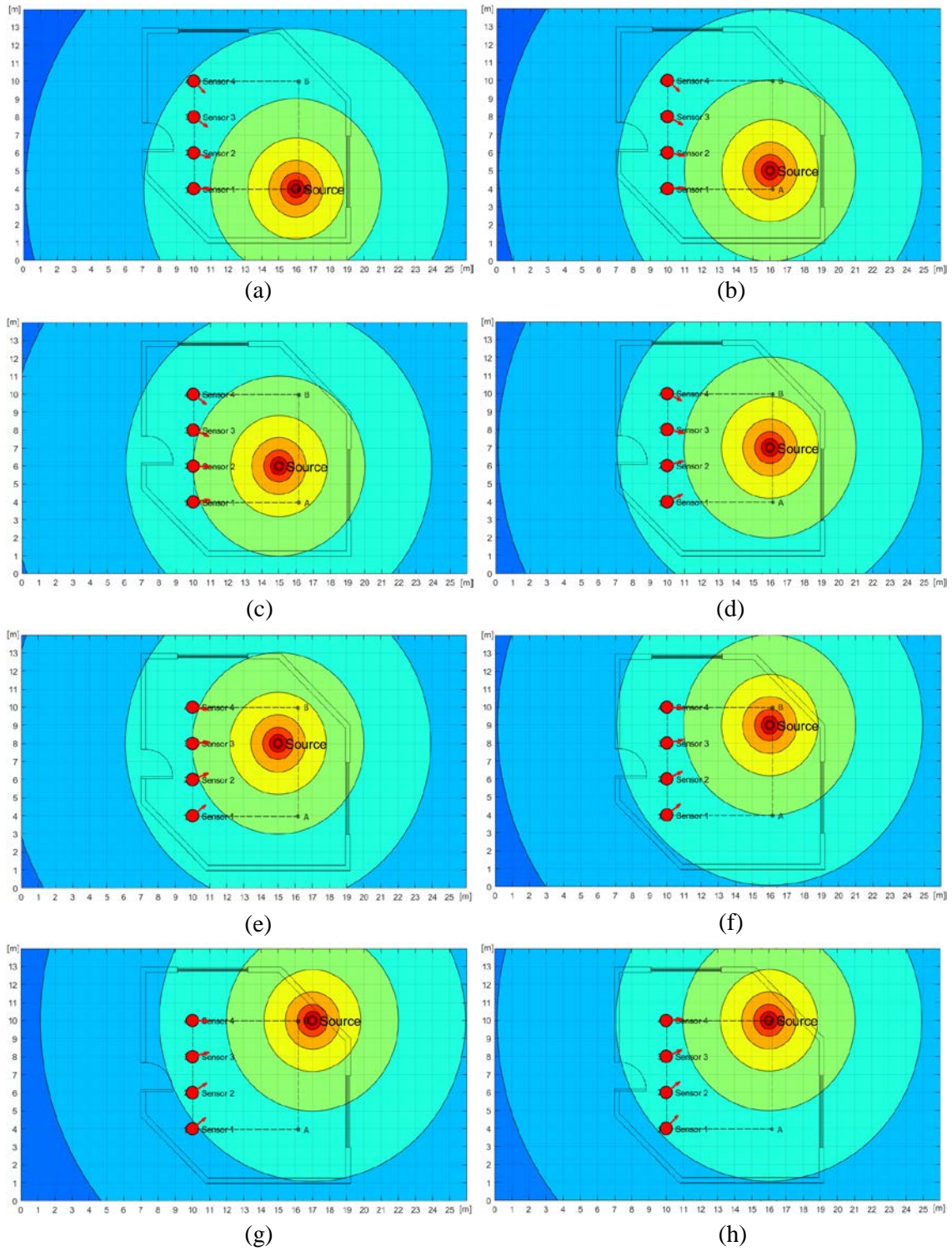


Figure 6 – Sequence of images generated by the PUCK's software (ordered from (a) to (h)) describing the measured acoustic event. The sound source was triangulated at every instant, defined by the integration time.

Another feature to take into account is the integration time. Values from 1 to 8 s were used to perform the triangulation. If this value is too short the angles presents wide variation of about 20°, making the triangulation unstable. If the time is too large the averaged values do not describe the complete path of the source accurately. A balanced integration time of 4 s was found to represent the acoustic event clearly and have less instabilities as possible.

The SPL is a simple representation of the sound source radiation considering free field conditions. Despite this do not describe the behaviour inside the room, is a first step for future development.

The results of measurement II are shown in Figure 7. The SPL in dB(A) is exposed over the entire measurement together with the PUCK localizations. The graphs in the bottom show the direction of the sound source for six instants of time. It starts on the left where the source was originally located in the measurement, then goes to the right and finish again on the left. The peak level of 70 dB shows the instant when the source was closest to the sound level meter. The background noise measured was 43.2 dBA.

As the sound level meter can only measure the magnitude of the SPL, the PUCK sensor complements it providing information about the direction of the loudest SPL. The identification is accurate when the sound is nearly 15-20 dB higher than the background noise, otherwise the direction tends to fluctuate.

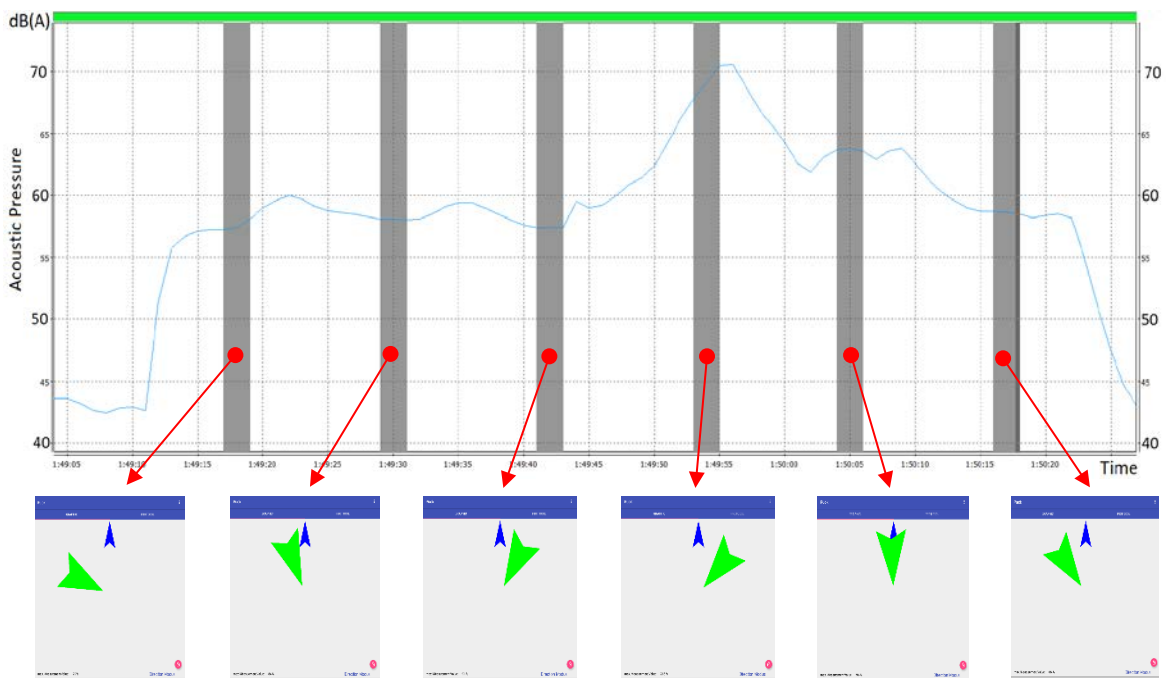


Figure 7 – SPL in dB(A) vs. time for measurement II. The big arrows of the graphs in the bottom show the source direction measured by the PUCK for every time selection.

5. CONCLUSIONS AND FUTURE WORK

A low-cost system for sound source localization was presented in this paper. The system is composed of a microphone array (PUCK sensor) controlled by an App. The sensor is able to establish the direction of the loudest SPL through an angle value represented in a 360° polar plot.

Two measurements were conducted to study the capabilities of the system. First, the localization of a moving sound source was conducted via triangulation of several sensor

positions. The results are shown as a sequence of images describing the acoustic event. Despite the acoustic environment used was not the ideal, since reflections tends to affect the localization of the direct sound, the PUCK sensor showed reliable results in determining the location of a sound source. The integration time of the angle values seems to be a factor to get stable results. Typical beamforming applications that do not call for the accuracy of system such as the acoustic cameras can be solve in a simpler and less expensive way. Identifying the loudest zone of a road, a railway or a harbour and in what moment of the day, can be the first step to find a solution to a noise problem.

The second measurement expose the capabilities of the system to be implemented in monitoring station for urban noise. Since a sound level meter can only provide the magnitude of the SPL generated during a measurement, the PUCK can bring the direction from where the noise is coming from. Therefore, the nature of a noise can be study with more accuracy.

The advantages of the system are the simplified implementation for typical beamforming applications, the ease of use and cost-efficient.

Future works are focused on improving the dynamic range of the sensor and the capability of identifying two or more sources at the same time.

6. REFERENCES

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