1pAB2. Design and field test of a low-cost-portable linear array for marine mammal localization

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Marine mammals are reliable biindicators of aquatic systems health. Within this group, cetaceans are well known by their high dependence on sound for many of their vital activities such as: socialization and mating, prey catching and navigation. Due to its high dependence on sound, bioacoustic methods become very important in the study of these species. Acoustic monitoring in the field is usually performed using omnidirectional hydrophones to assess the presence of mammals, but for some behavioral studies it is also important to locate the animals, something which is not possible with that arrangement. Although there are very well known techniques to detect the Direction of Arrival of the sound, the equipment required is highly specialized and expensive. In this paper the design and field test of digital and analog versions of a portable linear array of hydrophones capable of locating animal sounds by beamforming, using low cost and easily available equipment is presented. The array was tested in La Paz bay, Mexico by experts of the Marine Mammals Research Program of the University of Baja California Sur which were able to locate dolphins (Tursiops truncatus) only by their sound, despite strong sources of noise in the area.
INTRODUCTION

Marine mammals are adapted to the aquatic environment where light attenuates rapidly, but sound propagates well over long distances; hence, active and passive use of sound plays an important role for many of their vital activities such as: socialization and mating, prey catching and navigation. Due to its high dependence on sound, bioacoustic methods become very important in the study of these species.

Acoustic monitoring in the field is usually performed using omnidirectional hydrophones to assess the presence of mammals, but for some behavioral studies, individual and group identification, and acoustic communication studies, it is also important to have a method by which the azimuth of a calling animal can be determined acoustically [1, 2], something which is not possible with this arrangement. Although there are well known techniques to detect the Direction of Arrival of an underwater source of sound, unfortunately, the equipment required is highly specialized and expensive. For this reason, there is a need to create a portable system to locate marine mammals, a system that is affordable and easy to assemble.

Here, we designed and field-tested digital and analog versions of a portable linear array of four hydrophones placed in a lightweight structure that is capable of locating animal sounds by beam forming. Additionally, we used low cost and easily available equipment. The array was tested in Bahía de La Paz on the western side of the Gulf of California, under controlled conditions and in the field by experts of the Marine Mammals Research Program of the University of Baja California Sur. The device was able to locate dolphins (*Tursiops truncatus*) by their sound, despite strong sources of noise in the area.

BASIC TERMINOLOGY AND CONCEPTS

The hydrophones normally used in bioacoustics are transducers designed to receive sound in any direction so that a vocalizing animal might be detected no matter where it is in relation to the receiver. This is because the beam pattern (pick-up pattern) of a single hydrophone is relatively wide and provides low values of directivity. Higher directivity characteristics can be accomplished by forming a set of hydrophones in a geometrical configuration referred to as an array. The total field of the array is determined by the vector addition of the fields covered by the individual elements. To provide very directive patterns, it is necessary that the fields from the elements of the array interfere constructively (add) in the desired directions and interfere destructively (cancel each other) in the remaining space. Ideally this can be accomplished, but practically it is only approached (Figure 1).

In an array of identical hydrophones, there are key points that can be used to shape the overall pattern of the arrangement and the most important are: the geometrical configuration of the overall array (linear, circular, rectangular, spherical, etc.), the number of elements and the distance between them [3].

![FIGURE 1. Polar response of the beam pattern of (a) one hydrophone, (b) two hydrophones array, and (c) four hydrophones array. Note: The beam pattern is the field covered by a single hydrophone or an array of hydrophones.](image)

The main pickup pattern of an array (main lobe) becomes narrower by increasing the number of elements, but it also starts to cover other directions (side lobes) besides the main one (Figure 1c). Nonetheless, the array works for a broad range, is designed for a specific frequency, and the spacing between elements corresponds to half of its length wave for which nulls appear at 90° and 270° [3, 4].

There are well-known signal processing techniques to locate underwater sound sources using arrays, like “multi-hydrophone ranging”, that uses the differences in sound arrival times to the array with widely spaced hydrophones, or the “triangulation” technique, detecting the angle of arrival, which one measures from different positions, the directions to the sound source, and estimates the location where these directions cross. In this work, a method called “beam-forming”, where multiple closely-spaced hydrophones are correlated to obtain the direction of arrival of the sound (AOD) is described [5].
Beam-forming is a standard procedure when multiple hydrophones form a compact array. The primary objective is that, if hydrophones are close enough, the received signal from a given direction should be only time-delayed versions of the same basic signal, each hydrophone will be contaminated by different ambient noise fluctuations, instead of estimating the time delays between hydrophone pairs, one uses all hydrophones and sums the time-delayed amplitudes to form what is called a beam pattern (Figure 1c), which emphasizes coherent signals and minimizes the overall noise contribution to the array output [5, 6].

In a line array, the direction from which a wave front originates has an effect on the time at which the signal meets each element in the array. When arriving from $-45^\circ$, the signal reaches the left hand hydrophone first, when arriving from perpendicular to the array, the signal reaches each hydrophone at the same time, and from $+45^\circ$, the right hand hydrophone receives the signal first (Figure 2a) [7].

![Figure 2](image2.png)

**FIGURE 2.** Analysis of three different sound waves to an array with three hydrophones.

The figure above illustrates that the maximum output amplitude is achieved when the sound source is located perpendicular to the array; the signals arrive at the same time to each hydrophone, they are highly correlated in time and reinforce each other. Alternatively, if a sound source is located in a non-perpendicular direction, the signals will arrive at different times, so they will be less correlated and will result in a lesser output amplitude [7].

Normally, the beam pattern of a line array is fixed to a single direction perpendicular to the array (Figure 1c), so there are two ways to locate a source: the array could be rotated, physically looking for the angle where the maximum output gain is or electronically steer the beam pattern by adding a delay time to each of the hydrophones, such that the signals from a particular direction are aligned before they are summed. By controlling this feature, the main lobe direction of an array can be steered, searching for the angle of maximum gain output (Figure 3a) [7].

![Figure 3](image3.png)

**FIGURE 3.** (a) Electric steering of the array; (b) Geometry to estimate the delay time.

From Figure 3(b), the wave front time delay can be calculated using the difference in distance a wave front must travel between the reference point and the element on interest. The time is then calculated by dividing this distance by the speed of sound. According with Figure 3(b)

\[
\text{Distance} = x \sin \theta \\
\text{Delay} = \frac{\text{Distance}}{\text{Speed of sound}} = \frac{(x \sin \theta)}{c}
\]

Based on theoretical knowledge, the design of the array will be presented.
LINEAR ARRAY

The design of the array was proposed for a 2 KHz signal because many marine mammals produce sound with signals around this frequency; the resulting distance between hydrophones will not be too long, and because for real-time acoustic monitoring with headphones, it is a good frequency in a noisy environment. The optimal distance between elements is computed as half the wave length \( \lambda \), and thus, for a 2 kHz frequency, the wave length \( \lambda \) is:

\[
\lambda = \frac{c}{f} = \frac{(1500 \text{ m/s})}{(2000 \text{ Hz})} = 0.75 \text{ m}
\]  

Thus, the distance between elements (L) will be 0.375 m. With this spacing, a four-element array was selected, since an array larger than 1.5 m would make the device bulky and difficult to handle in a small boat. Accordingly, the array will be placed in an horizontal lightweight and folding structure (“horizontal” to avoid any variation with the surface and floor sound reflections), capable of lowering to an estimated depth of 3 m, and with a steering handle at the top to hold the arrangement.

Linear Array Supporting Structure

Copper pipes were selected as the material to use since they are low cost, easily available, rigid and lightweight, and suitable for use in water. The proposed linear array follows the shape of an inverted “T”, as shown in Figure 4a. The parts required are listed below.

- 4 hydrophones, with 6 m of cable [8].
- 4-jack 3.5 mm mono mini female audio connectors (for chassis mount).
- 2 copper water pipes, each 3 m long each (¾ inch dia).
- 4 couplers for copper pipe (¾ inch dia), two single type and two “T” type.
- Accessories: screws, plastic cable ties.

Follow the scheme shown in Figure 4(a). Notice that one of the copper pipes is first cut into three 1 m segments and then joined again using single couplers (secured by a screw); in this way the length can be reduced, as required. From the dimensions suggested, only the spacing between hydrophones becomes compulsory to keep the design at the selected frequency of 2 kHz. A steering handle is added for the analogue operation of the array.

![FIGURE 4. (a) Scheme and size of the array; (b) Low cost portable line array.](image)
BEAMFORMING

There are two options proposed to process the signals. The first uses an analogue system to add the signals from the four hydrophones to maximize the signals coming from the front; in this case, the array has to be manually steered to locate the sound source, as described before. The second uses a digital system to add the signals and steer the array electronically by adding delays to the signals of the hydrophones according to the angle of interest. Both options will be described

Analogue Beamforming

According to Figure 2, adding the signals coming from the front of the array will increase the total gain, while signals from other directions will interfere. This can be easily done using an analogue four-channel audio mixer. The gain of each channel can be controlled to weigh the array using standard windows to minimize side lobes. A low cost soundtrack mixer was used for this propose [9].

Digital Beamforming

To process the signals digitally, an analogue-to-digital converter is required. The price of this component depends mainly on the number of channels and the sampling frequency. For the design frequency of 2 KHz, a sampling frequency of 4 kHz would be enough, since it meets the Nyquist criteria. An alternative to the very high cost of professional DAQs, a Sony playstation eye camera (PS3) was selected. It is a digital webcam camera for Playstation 3 that comes with a small microphone array (4 microphones), used for voice location tracking, echo cancellation, and background noise suppression with a sampling rate of 16 kHz and 16 bits resolution. It can be connected to the computer by USB cable. After some minor changes in the circuit, it can be used as a four audio channel interface for a very low price [10]. A standard laptop computer could be used for the processing the signals.

Audio interface

To use the PS3 camera as a four channel analog-to-digital converter, some modifications must be made. There are some sites that show how to open the camera, such as [11]. Once opened, find a ground point in the circuit and wire it to the ground of the jacks. Identify the microphones soldered to the circuit. Cut the leg nearest the edge of the circuit board and connect it to the audio jack. Connect the borne at the circuit board to the jack as shown in Figure 5d. This arrangement permits switching between the microphone array, when no plug is inserted, and the hydrophones.

FIGURE 5. T PS3 Eye camera, (a) Intern circuit; (b) Audio jacks; and (c) Interface ready to use it, (d) Schematic of the audio jack switch function.

Signal Processing

The processing of the signals was implemented in Matlab (R2010b). Figure 6 shows the processing flowchart.
The signal is processed in batches of 2 s acquired for all channels. In a first preprocessing stage, a band pass filter (First-order Butterworth 1–7 kHz) is applied to discard unwanted low frequency signals generated by the waves hitting the boat. According to formula (2) and Figure 3, the beamforming performs the electrical steering by computing the signals at different angles using the delays shown in Table 1. Results from each angle are normalized and only the maximum value is displayed. Therefore, this algorithm shows the location of the signal with the highest amplitude within this band.

### Table 1. Delays required to electronically steering the array.

<table>
<thead>
<tr>
<th>Angle</th>
<th>–65°</th>
<th>–45°</th>
<th>–30°</th>
<th>–15°</th>
<th>0°</th>
<th>15°</th>
<th>30°</th>
<th>45°</th>
<th>65°</th>
</tr>
</thead>
<tbody>
<tr>
<td>Delay</td>
<td>–0.226 ms</td>
<td>–0.176 ms</td>
<td>–0.125 ms</td>
<td>–0.0647 ms</td>
<td>0 s</td>
<td>0.0647 ms</td>
<td>0.125 ms</td>
<td>0.176 ms</td>
<td>0.226 ms</td>
</tr>
</tbody>
</table>

### TESTING

Two different scenarios were devised to assess the performance of the linear array. In the first case, tests were conducted at a dock to provide controlled conditions. Then the array was tested in the field.

#### Test under controlled conditions

The first test was conducted at Playa Pichilingue near La Paz, Baja California Sur. The array was immersed holding it away from the dock (Figure 7c) and a fixed source was placed at –90°, –45°, 0°, +45°, and +90° with respect to the orientation of the array at ~25 m, as shown in Figure 7a. The sound source was generated by an alarm device inside a plastic bag and immersed 1 m from the surface. The sound generated oscillates between 3.5 and 4.5 KHz, as shown in the spectrogram of Figure 7b. Figure 7d shows the plot obtained, using the digital implementation of the beamforming when the source was located at –45°. Encouraged by these results, the array was field tested.
FIGURE 7. Laboratory test under controlled conditions. (a) Scheme of the source and array position. (b) Spectrogram of the record when the source was at −45°. Note: Only one channel is represented; settings: 512 point FFT, Blackman-Harris window. (c) Dock and array. (d) Matlab AOD graphic

Field Test

Bahía de La Paz is located at the southwest side of the Gulf of California between 24.1° and 24.8° N and 110.2° and 110.8° W, with a surface area of ~1970 km² (Figure 8). The bay has a shallow marine depression, which gets deeper progressively from south to north [12]. The region is semi-arid and warm with annual average temperature of 24.7°C, the water’s average salinity is 36‰, and average annual rainfall is 210 mm. The bay lacks a source of freshwater other than rainfall [13, 14].

FIGURE 8. The study area is the Bahía de La Paz in the Gulf of California. (a) The bay is part of the State of Baja California Sur, Mexico; (b) The bay is delimited by Isla Espiritu Santo and the Baja California Peninsula.

The field test was conducted by a reconnaissance in the bay with the array aboard a small vessel (~5 m long), by members of the Marine Mammals Research Program (PRIMA) of the University of Baja California Sur, moving to strategic locations in search of the cetacean.

A family of dolphins (*Tursiops truncatus*) appeared on the surface at approximately −20° with respect to the array, as shown in Figure 9a. The boat was stopped. There was also a strong sound coming from a dredge at a
nearby petrol storage facility. The signal from the array was first recorded for analysis using the digital version. Figure 9c shows the spectrogram from one channel. The whistles from the dolphins are highlighted by the blue circles. In this graph, the strong source of noise from the dredge can be seen at lower frequencies. Figure 9b shows the results from the digital beamforming. The black squares correctly located the dolphins at $-20^\circ$, but between whistles the algorithm locates another sound source at $+7^\circ$. This sound source wasn’t identified in the field, but it might be one high frequency component from the dredge, such as the ones shown in figure 9c.

To test the analog beamforming, members from the PRIMA steered the array manually and the output of the array was connected to the mixer and monitored through headphones. They were able to locate the dolphins, substantially suppressing the noise from the drill. The movement of the dolphins was followed by the array until no visual contact was possible, but they were acoustically detected. Navigation in the direction indicated by the array confirmed their bearings.

**FIGURE 9.** Field test. (a) Dolphin, dredging activities noise and array position. (b) Matlab AOD graphic (c); Spectrogram of the dolphin whistle (1,2) noise (3), Note: Only one channel is represented. Settings: 512 point FFT, Blackman-Harris window. (d) Dolphins at the back before lowering the array; (e) Array lowered into the water.

**CONCLUSIONS**

Both solutions (analog and digital) are useful at the location time, but each one has different characteristics that can be adjusted better to the conditions of the offshore research. Even when the array is designed for a given frequency, it works for many other frequencies around the main one, but in a suboptimal way. Although sometimes it seems that the technology tools designed for scientific research is commonly expensive and inaccessible, there are reliable alternatives at affordable prices. By taking care of each step, this open source Low-cost Portable Linear Array is ready to help those involved in marine mammal research.
FUTURE WORK

To locate a specific cetacean, the implementation of matched filters and correlation algorithms regarding to the whistles, clicks or burst-pulse emitted by the mammal is required. Also, to avoid carrying a portable computer aboard the vessel, it will be good to design an embedded system that can do the digital processing and, with a small display, show the results.

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