EXPERIMENTAL EVALUATION OF THE PERFORMANCES OF A NEW PRESSURE-VELOCITY 3D PROBE BASED ON THE AMBISONICS THEORY

Angelo Farina[a], Enrico Armelloni[b], Lorenzo Chiesi[a]

[a] Industrial Eng. Dept., University of Parma, Via G.P. Usberti 181/ 43124 PARMA, ITALY
[b] A.I.D.A. S.r.l. spin-off company of the University of Parma, Italy

Contact author: Angelo Farina, Industrial Eng. Dept., University of Parma, Via G.P. Usberti 181/ 43124 PARMA, ITALY, tel. +39 0521 905701, fax + 39 0521 905705, E-mail farina@unipr.it, web site HTTP://www.angelofarina.it

Abstract: *A three-dimensional underwater probe, employing 4 standard hydrophones located at the vertexes of a tetrahedron, has been built and tested inside a pool and in the sea. A new 4-channels, self-contained, waterproof recording system was employed ("Soundfish"), capable of storing the signals coming from the 4 hydrophones on an SD card, in standard WAV format.*
The recorded signals are later analyzed by means of the Ambisonics theory, employing a matrix of 4x4 FIR filters. The coefficients of these filters have been computed by inverting a large number of impulse response measurements performed inside a test pool, from many different directions. As result, vector decomposition of the sound field was obtained, containing a set of 4 signals, corresponding the 0th and 1st order spherical harmonics of the sound pressure field (pressure and particle velocity).
Hence, the instantaneous Sound Intensity vector can be computed, from which the three-dimensional direction-of-arrival of the sound can be obtained, and the position of the predominant noise source can be tracked continuously over a marine map of the area.
The paper provides a detailed description of the Soundfish recorder, discloses the mathematical and physical details of the signal processing performed, and presents the analysis of some recordings collected at the Miramare Marine Reserve, Trieste, ITALY.

Keywords: *pressure-velocity probe, vector analysis, location tracking*

1. INTRODUCTION

Boat noise represents a chronic source of harassment for fish species, whose communication for inter- and intra-sexual selection is mainly based on low frequency sound signals [1]. Investigating the impact of boat noise on target fish species is particularly relevant for coastal MPAs, which are biologically rich locations deserving protection from anthropogenic pollutants. Although many fish species are primarily sensitive to the kinematic components of the sound field [2], namely to particle acceleration, both background and boat noises have been characterized so far mostly by means of sound pressure measurement.

In this work, a novel hydrophonic probe ("Soundfish") is presented, developed based on the long-known technology known as Ambisonics [3].

The probe is built employing 4 standard hydrophones, placed at the vertexes of a tetrahedron. The probe is also equipped with a waterproof 4-channels sound recorder, capable of recording continuously for more than 24 hours, or to be self-activated by sounds exceeding a preset threshold for long-term monitoring of noisy events.

The sound pressure signal and the three signals corresponding to the Cartesian components of particle velocity are later obtained by post processing the 4-channels recording, by means of a suitable filter matrix.

This allowed for characterization of the sound field not just in terms of sound pressure, but also of the three Cartesian components of the particle velocity. A further post processing of the pressure-velocity recordings allow for detecting the three-dimensional direction-of-arrival of the sound: this makes it possible to track the trajectory of a boat moving at short distance from the probe.

With suitable playback equipment, it is also possible to employ these pressure-velocity recordings for reconstructing the three-dimensional acoustical soundscape inside a controlled environment (that is, a region of space surrounded by a reasonable number of loudspeakers, say 8). This can be used both for underwater playback experiments (with the goal of assessing the behavioural reactions of marine species to noise stimulation), and for recreating the underwater sound in air (useful for example in marine museums, aquariums, etc, for providing the visitors with a realistic soundscape).

2. DESCRIPTION OF THE "SOUNDFISH" APPARATUS

The probe is simply a small holder for 4 standard hydrophones (Aquarian Audio mod. H2b). These 4 hydrophones are located at the vertexes of a tetrahedron, at a distance form its center (radius) of approximately 150mm. This is the underwater equivalent of a Soundfield™ microphone.

This size of the probe was chosen for optimizing its behaviour in the frequency range of 100 Hz to 10 kHz, but if a lower frequency range is required (for ship noise, for example), the hydrophones can be mounted at a larger radius by means of spacers.

The recording system is based on a modified ZOOM H2 digital sound recorder, re-named Brahma, capable of recording the signals coming from the Soundfilsh probe. The recorder operates at 48 kHz, 24 bits and records standard uncompressed WAV files over a 16Gb or 32 Gb SD card.

These recordings are easily post-processed on a PC. A software tool, named Brahmavolver, was developed for converting the raw signals coming form the 4 hydrophones to the 4 output signals, representing respectively the sound pressure and the three Cartesian components of particle velocity. The processing is based on the use of a matrix of 4x4 FIR filters, currently 2048 points long. In our approach [4] the filter coefficients are computed numerically, inverting a matrix of measured impulse responses, obtained with the sound source placed at a large number D of positions all around the probe, as shown in *Fig. 1*.

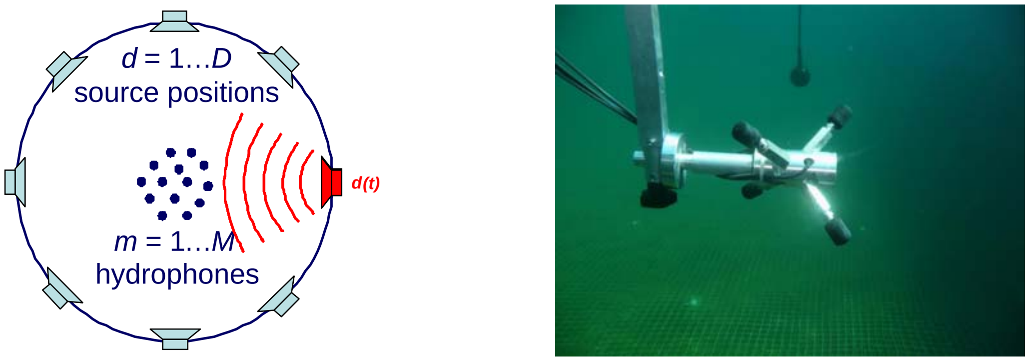

Fig. 1: Impulse response measurements, performed inside a pool.

In the following chapters, the theory for performing the numerical signal processing is first explained, then the results of some preliminary experiments made for testing the Soundfish apparatus in the sea are presented.

3. THEORY

Given an array of transducers, a set of digital filters can be employed for creating the output signals. In our case the M=4 signals coming from the hydrophones need to be converted in V=4 signals yielding the desired virtual directive sensors: so we need a bank of M×V (4x4) filters, as shown in *Fig. 2*. As always, we prefer FIR filters.

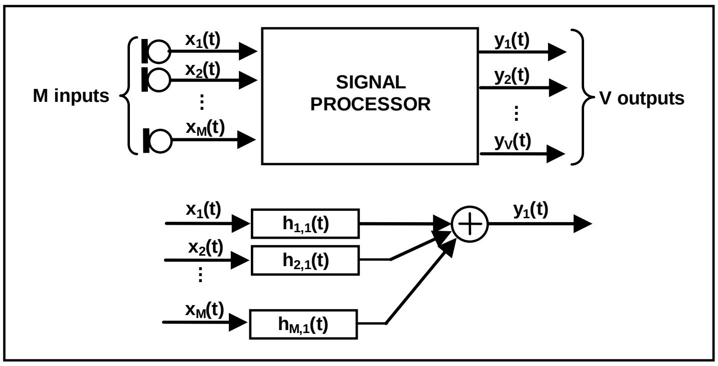

Fig. 2: Scheme of the signal processing.

Assuming x_m (m=1..M) as the input signals of M hydrophones, y_v (v=1..V) as the output signals of V virtual sensors and $h_{m,v}$ the matrix of filters, the processed signals can be expressed as:

$$y_v(t) = \sum_{m=1}^{M} x_m(t) * h_{m,v}(t) \tag{1}$$

Where * denotes convolution, and hence each virtual sensor signal is obtained summing the results of the convolutions of the M inputs with a set of M proper FIR filters.

One of the most used techniques when employing arrays of transducers arranged at the vertexes of a regular polyhedron is the Ambisonics method: in this case the directivity patterns of the virtual sensors correspond to the zero and first order spherical harmonics. The zero-order harmonic Q_0 has a polar pattern shaped as a sphere, that is, an omnidirectional sensor, so in practice it corresponds to an ideally-omnidirectional pressure sensor.

The first-order harmonics are three, $Q_{1,x}$, $Q_{1,y}$ and $Q_{1,z}$ shaped as "figure-of-8", aligned with the three Cartesian axes. The virtual sensors hence are particle velocity transducers, and the signals processed by these filters are the Cartesian components of the particle velocity.

Fig. 3 shows a three-dimensional representation of the polar patterns of the 4 virtual sensors:

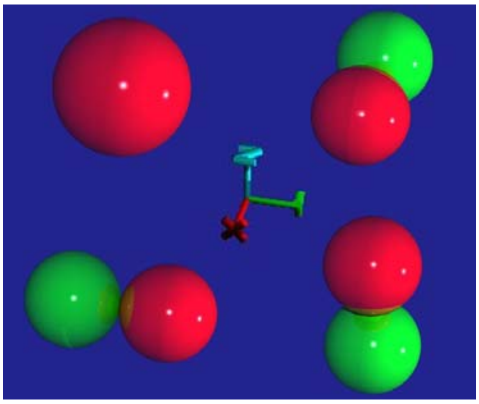

Fig. 3: Zero and First Order spherical harmonics.

The mathematical functions describing these polar patterns are the following:

$$Q_0(\varphi,\vartheta)=\frac{\sqrt{2}}{2} \qquad \begin{cases} Q_{1,x}(\varphi,\vartheta)=\cos(\varphi)\cdot\cos(\vartheta) \\ Q_{1,y}(\varphi,\vartheta)=\sin(\varphi)\cdot\cos(\vartheta) \\ Q_{1,z}(\varphi,\vartheta)=\sin(\vartheta) \end{cases} \tag{2}$$

The processing filters *h* are usually computed following one of several complex mathematical theories, based on the solution of the wave equation, often under certain simplifications, assuming the transducers are ideal and identical. In some implementations the signal of each microphone is processed through a digital filter for compensating its deviation, with a heavier computational load.

In this novel approach no theory is assumed: the set of filters *h* are derived directly from a set of anechoic measurements. A matrix of measured impulse response coefficients *c* is formed and the matrix has to be numerically inverted (usually employing some approximate techniques, such as Least Squares plus regularization); in this way the processed signals are maximally close to the ideal responses prescribed. This method also inherently corrects for transducer deviations and acoustical artefacts (shielding, diffraction, reflection, etc.).

The characterization of the array is based on a matrix of measured anechoic impulse responses *c*, obtained with the sound source placed at a large number D of positions all around the probe, as shown in *Fig. 1*.

The processing filters *h* should transform the measured impulse responses *c* into the prescribed theoretical impulse responses *p*:

$$\sum_{m=1}^{M} c_{m,d} * h_m \Rightarrow p_d \qquad d = 1..D \tag{3}$$

Please notice that in practice the target impulse responses p_d are simply obtained applying a direction-dependent gain *Q*, given by eq. 2, to a delayed unit-amplitude Dirac's delta function: $p_d = Q(\phi,\theta) \bullet \delta$.

Computation is easier in frequency domain (that is, computing the complex spectra, by applying the FFT algorithm to the N-points-long impulse responses *c*, *h* and *p*). Let's call *C*, *H* and *P* the resulting complex spectra. This way, the convolution reduces to simple multiplication between the corresponding spectral lines, performed at every frequency index *k*:

$$\sum_{m=1}^{M} C_{m,d,k} \cdot H_{m,k} \Rightarrow P_d \qquad \begin{cases} d = 1..D \\ k = 0..N/2 \end{cases} \tag{4}$$

Now we pack the values of ***C***, ***H*** and ***P*** in proper matrixes, taking into account all the *M* input microphones, all the measured directions *D* and all the *V* outputs. Hence:

$$[H_k]_{MxV} = \frac{[P]_{DxV}}{[C_k]_{DxM}} \tag{5}$$

This over-determined system doesn't admit an exact solution, but it is possible to find an approximated solution with the Least Squares method, employing a regularization technique for avoiding instabilities and excessive signal boost [5]. The block diagram of the least-squares method is shown in *Fig. 4*:

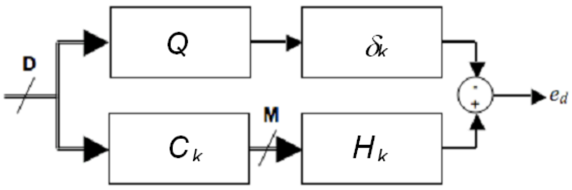

Fig. 4: Scheme of the Least Squared method with a delay in the upper branch.

In this scheme we observe the delay block δ, required for producing causal filters, and the resulting total modelling error *e*, which is being minimized by the least-squares approach. In our case, this delay is embedded in the prescribed "target functions" P_d, as already explained.

In general, the frequency-domain representation of a Dirac's delta delayed by n_0 samples is given by:

$$\delta_k = e^{-j2\pi k\frac{n_0}{N}} \tag{6}$$

In our case, the "optimal delay" n_0 chosen was simply $N/2$, that is, half of the length of the filters.

Furthermore, a regularization parameter is required in the denominator of the matrix computation formula, to avoid excessive emphasis at frequencies where the signal is very low.

So the solution formula, which was first proposed in Kirkeby et al. [5], becomes:

$$[H_k]_{MxV} = \frac{[C_k]^*_{MxD} \cdot [Q]_{DxV} \cdot e^{-j\pi k}}{[C_k]^*_{MxD} \cdot [C_k]_{DxM} + \beta_k \cdot [I]_{MxM}} \tag{7}$$

As shown in the image below (*Fig. 5*), the regularization parameter β is frequency-dependent. A common choice for the spectral shape of the regularization parameter is to specify it as a small, constant value inside the frequency range where the probe is designed to work optimally, and as much larger values at very low and very high frequencies, where conditioning problems are prone to cause numerical instability of the solution.

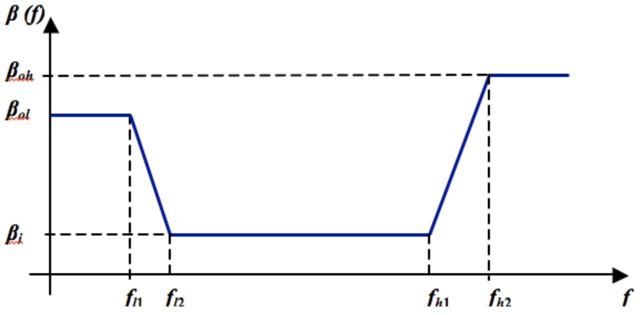

Fig. 5: Regularization parameter in dependence of the frequency.

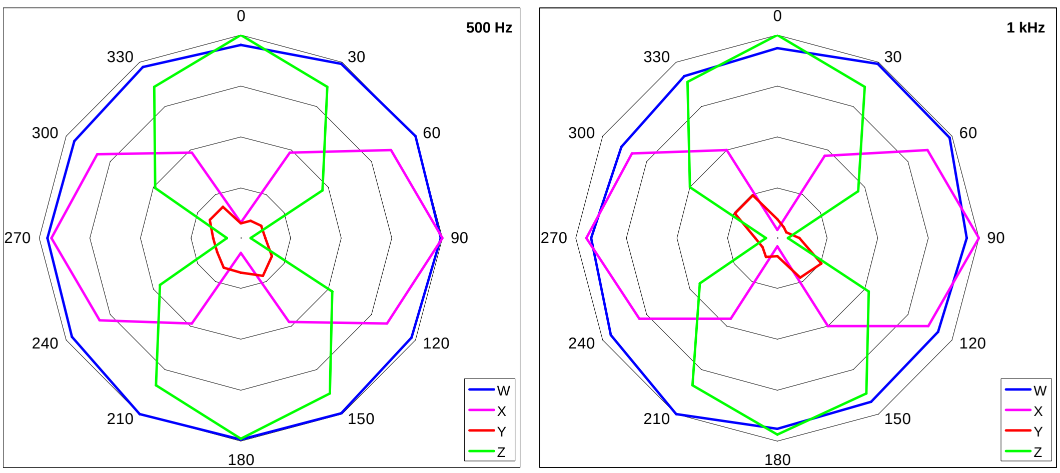

Fig. 6: Polar patterns of the Soundfish probe at 500 and 1000 Hz.

4. FIRST EXAMPLES OF USE OF THE SOUNDFISH PROBE

4.1. Preliminary tests in pool

A suitable number of impulse response measurements were performed on the Soundfish probe inside the test pool, kindly made available by WASS in Livorno, Italy, as shown in *Fig. 1*. A turntable, controlled by our Aurora software, was employed for automatically rotating the probe in steps of 30 degrees both along azimuth and elevation, yielding a set of 72 impulse responses uniformly distributed all around the sphere. Previous *Fig. 6* shows some of the results of these preliminary tests: the polar patterns of the pressure and particle velocity in two octave bands.

4.2. Field recordings

The probe and the Brahma recorder were placed on the sea bottom, at a depth of 8m, inside the protected area of the Miramare Reserve. A 30 minutes long recording of the Sea Ambient Noise (SAN) was performed, followed by recordings of a boat passing near the probe. The analysis of these recordings allowed for the computation of 1/3 octave band spectra of both sound pressure level (SPL) and particle velocity level (PVL), computed with reference to the standard quantities for underwater acoustics (1 μPa and 1nm/s). Furthermore, the particle acceleration levels (PAL) can be easily derived from the values of particle velocity levels, following the procedure described in Picciulin et al. (2010).

Fig. 7 presents the analysis of the recordings, showing the 1/3 octave band spectra in terms of SPL and PVL of the SAN and of the boat passage.

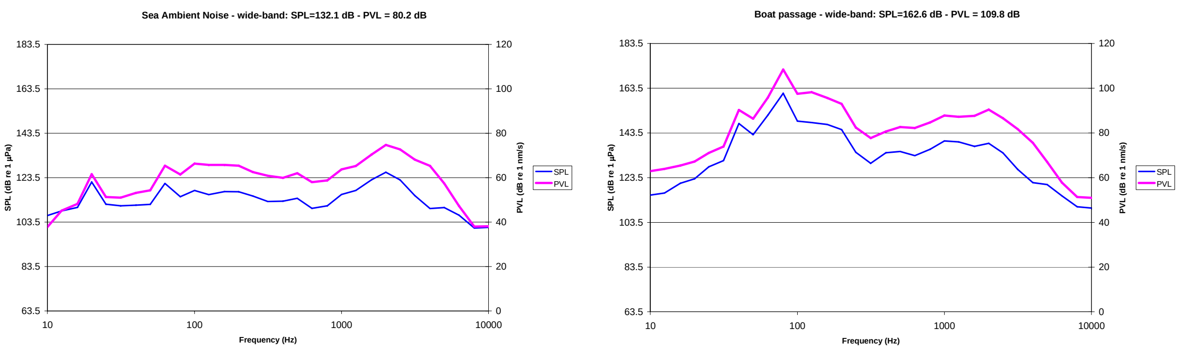

Fig. 7: SPL and PVL spectra: Sea Ambient Noise (left) and boat passage above the probe (right).

It can be seen how the pressure and velocity spectra differ, due to different acoustic impedance occurring for near, coherent and far, diffuse sound fields.

Albeit only the overall magnitude of the particle velocity vector is reported here, the data obtained allow for computation of the direction of the vector, making it possible to know, at any instant, the position of the sound source. *Fig. 8* shows the coordinates of the estimated position of the sound source overplotted on the map of the site. Result is in good agreement with the real trajectory of the boat.

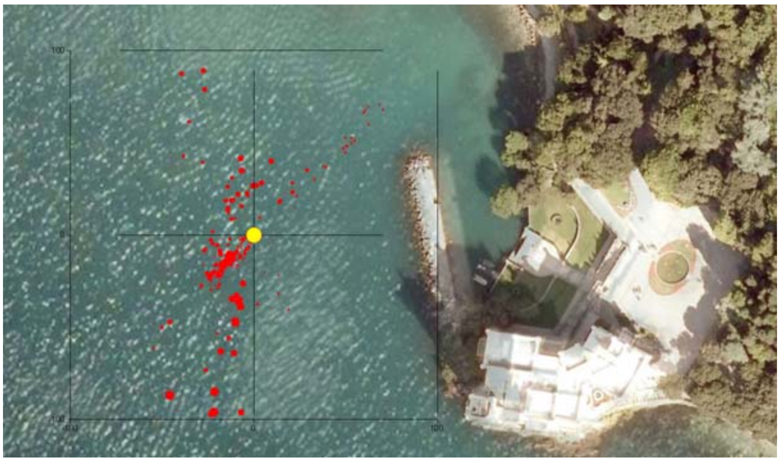

Fig. 8: Coordinates of the estimated position of the sound source (Miramare - TS).

5. CONCLUSIONS

The new *Soundfish* probe can be employed for an analysis of the cause-effect relationship, as at every instant the position of the source, relative to the receiver, is known, alongside with the quantities relevant for assessing the impact of human-produced noise over marine species, either sensitive to sound pressure or to particle motion. The reliability of the new measurement system must now be assessed by employing it in a number of surveys, under different sea conditions, at different depths, and with various kinds of noise sources. It could also be advisable to repeat the calibration in the pool, employing narrower angular steps, for ensuring computation of even better digital filters.

6. ACKNOWLEDGEMENTS

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REFERENCES

[1] **Amorim MCP,** Diversity of sound production in fish, In *Ladich SP et al (eds) Communication in fishes,* Science Publishers, Enfield. pp. 71-104 (2006).

[2] **Popper AN, Fay RR** The auditory periphery in fishes. In: *Comparative Hearing: Fish and Amphibian*. Springer-Verlag, New York, pp. 43–100 (1999).

[3] **M.A. Gerzon**, "Ambisonics in Multichannel Broadcasting and Video", *J. Audio Eng. Soc.,* vol. 33 no. 11, pp. 859-871 (1985 Nov.)

[4] **A.Farina, L.Capra, L.Conti, P.Martignon, F.Fazi,** Measuring spatial impulse responses in concert halls and opera houses employing a spherical microphone array, In *Proc. 19th International Congress on Acoustics*, Madrid (Spain), 2-7 Sept. 2007.

[5] **O. Kirkeby, P. Rubak, A. Farina**, Analysis of ill conditioning of multi-channel deconvolution problems, *106th AES Convention*, Munich, Germany - May 8-11, 1999.