Contributions to Passive Acoustic Oceanic Tomography – Part II : Extraction of propagation features from single hydrophone measurement

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I. INTRODUCTION

Acoustic tomography is a way to produce a fast, accurate and cheap monitoring of water mass. This monitoring requires an inversion procedure made of two steps. The first one is to estimate acoustic properties (such as the sound speed profile of the water column) from the measurement of a propagated known acoustic waveform between fixed sources and receivers. Then a second step consists in inferring some physical ocean parameters (temperature, bottom nature) from these previous estimated acoustic characteristics. Large scales deep water and small scales shallow water configurations have been successfully studied and associated to matched delay, matched field and matched impulse response inversion processing.

Accurate estimates of acoustic properties demand the emission of powerful and recurrent signals in the adapted bandwidth and in agreement with the scale of the monitoring. But we would rather not send these hard active sounds through the water column in a potential military underwater warfare context, or if mammal species health is considered. A recent solution has emerged in the community to tackle this problem with the passive tomography processing. Passive tomography processing consists in estimating acoustic properties by using opportunity sources present in the channel at the time of interest. Some experiments have recently been carried out using ships, marine mammals and surface noises.

Different levels of complexity can be formulated to insure the discreetness of tomography processing. The first one is the Active Discreet Tomography (ADT) where active emission is allowed but with a waveform chosen to insure a low probability of interception using for instance a copy of a noise component or a spread spectrum signal. In that case Signal to Noise Ratio (SNR) is reduced compared to classic active tomography. The second one is the Aided Passive Tomography (AiPT) where active emission is forbidden but where a cooperate entity of known position can produce an acoustic emission linked to its natural activity. Blind estimation of the impulse response of the channel is performed with the losses of absolute time and magnitude references, and SNR is also reduced. The last one is the Autonomous Passive Tomography (AuPT) where active emission is forbidden but where an entity of unknown position can produce an acoustic emission linked to its natural activity. As in the case of the AiPT, blind estimation of the impulse response of the channel is performed with the losses of absolute time and magnitude references, SNR is reduced, and on top of that, position of the source considered as a nuisance parameter has to be estimated jointly with the parameters of interest.

Passive Acoustic Tomography must be tackled as an inverse problem and successful completion is achieved by treating the following steps:

- step 1 : hypothesis making,
- step 2: direct modelling,
- step 3: performances prediction and optimal design of experiments,
- step 4: inversion stage
    - step 4.1 : digital signal processing for propagation features extractions from measurements
    - step 4.2: acoustical properties estimation by model fitting between measured features and simulated ones
- step 5: real world data applications.

While a first companion paper deals with steps 2 and 3 and a second ones deals with extraction of space and temporal features of the multi-paths arrivals, the present paper is aimed at extracting the temporal features of the acoustics arrival (step 4.1) from a single hydrophone measurements (real world data - step 5). The tools produced in this paper are suitable with AiPT and AuPT. Two bandwidths of natural sources are under study here, a lower ones (bandwidth B_1) from 0 to 500 Hz for which group propagation times versus frequency and mode order (under a modal dispersive propagation hypothesis) are the features extracted from the data and a higher ones (bandwidth B_2) from 2 kHz to 4 kHz for which time and magnitude of arrivals (under a multi paths acoustical rays propagation hypothesis) are the features extracted from the data. Theses tools use interception of sonar activities or explosive sound signals in AiPT or marine mammals vocalises (Beluga calls for the high bandwidth, Right Whales Gunshot calls) in AuPT. Applications are carried out on marine mammals calls in AuPT where whales positioning is performed by hyperboling fixing via a network of hydrophones and propagation features extraction is performed from a single hydrophone measurement given the estimated position of the source.

This paper is organized in three parts. Part I presents the material for real world data application, Part II presents extraction of time and magnitude of arrival from sources of bandwidth B_1 whereas

Part III presents extraction of group propagation times from sources of bandwidth B_2.

II. MATERIAL FOR REAL WORLD DATA APPLICATION

To evaluate our algorithms, 2 data sets are used.

II-1 Laurentian Channel Data Set

For Part II, we use the data provided by the “Institut des Sciences de la Mer” (ISMER) from the University of Quebec at Rimouski (UQAR). Saint-Lawrence channel presents two critical habitat areas of marine mammals and a very dense ship traffic (see [4]). To understand interaction between marine mammals behavior and ship radiated noise, passive recordings of a few marine mammals sounds were performed thanks to a network of hydrophones (5 multi-electronics Aural M1 moorings and a 6 ocean bottom hydrophones coastal array) in order to identify and to locate these animals during summers 2003 and 2004. Regular and calibrated acoustic signals were also sent from a seismic sparker, installed on the research boat Coriolis, to help refining more precisely the positions of the coastal array. Several CTD (conductivity - temperature – depth) records were taken during this experiment and the bathymetry for the environment was available from previous studies. More details on the data can be found in [5] .

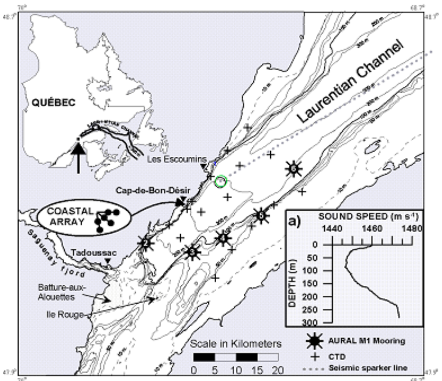

Fig. 1. Study area with the location of both the coastal array and the 5 buoys a) A typical celerity profile measured in this area.

II-2 Bay of Fundy Data Set

For part III, we use the data provided for the workshop on detection and localization of marine mammals using passive acoustics [6] . In this data set, marine mammals calls were recorded by a network of 5 Ocean Bottom Hydrophones (Bandwidth : 0-800 Hz) in the bay of Fundy whereas celerity profile were estimated thanks to XBT and CTD records. Bottom is made of a thin layer of LaHave clay over a thick layer of Scotia Shelf drift. Thickness and acoustic properties of these sediment layers can be found in [7] . The next figure presents location of Bay of Fundy, Bathymetry, OBH Locations and typical temperature profile.

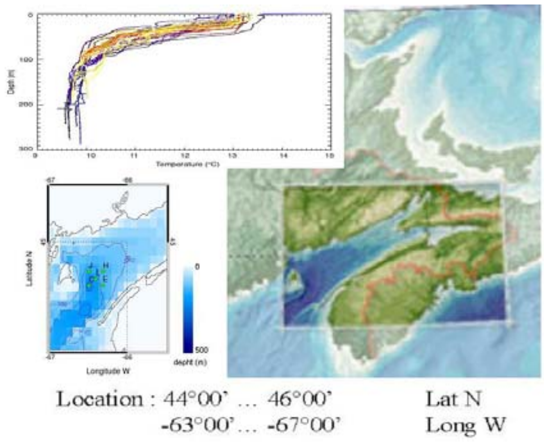

Fig 2 : Bay of Fundy Data Set

III. FEATURES EXTRACTION FROM OPPORTUNITY SOURCES OF BANDWIDTH B_2

III-1 Algorithm for blind estimation of time and magnitude of arrivals

When the central frequency of the opportunity source is high enough, acoustic ray paths propagation takes place and the signal at the receiver can be seen as a sum of attenuated and delayed versions of the emission. Then, if the emission has a clear time frequency content such as marine mammal vocalizes, time frequency processing can be used advantageously. A theoretical Time Frequency mapping of the received signal *m(t)* concentrates the tempo-spectral power density around N time translated versions of the instantaneous frequency curve of the source *s(t)*. The proposed processor is based on the main characteristics of this Time Frequency mapping. A first stage is dedicated to the instantaneous frequency estimation of the opportunity source followed by a second stage which estimates the time and magnitude of arrivals (main features to extract) of each resolved replica of the emission.

To estimate the source’s instantaneous frequency function, a local maximum is sought thanks to an optimal time frequency mapping which deletes the interference terms without significantly increasing the spread of the auto-terms (Reduced Interferences Distributions, RID). These signal-dependent time frequency representations are based on the optimal weighting of the ambiguity function by a radially signal-dependent Gaussian kernel (Radial Gaussian Kernel, RGK), or based on the optimal weighting of local ambiguity function (Adaptative Optimal Kernel, AOK) developed by Baraniuk and Jones [2] . Our approach may be biased but is stable over noise and interferences. The algorithm used to estimate the source’s instantaneous frequency function is the following:
- compute the signal-adapted Time Frequency mapping of the received signal, ($RGK_m(t,f)$),
- for each frequency bin f_i, estimate the time of the first local maximum of the function of time $RGK_m(t,f_i)$.

The source’s instantaneous frequency function $\widetilde{f}_i(t)$ obtained at this stage is used to estimate the channel impulse response.

The reference [1] details a blind low-resolution estimation of the features of interest thanks to a blind time-frequency matched filter.

As soon as celerity profile estimation is concerned, first acoustics paths carry a lot of information about it, but are usually not resolved by the matched filter and require high-resolution processing. We now focus on the development of **a blind high-resolution time frequency processor**.

This tool is dedicated to signals *s(t)* having a curvilinear distribution of time spectral power density. Each signal of this family can be locally approximated by a Chirp signal, and then if the area of validity of this assumption and the chirp parameters (central frequency and bandwidth) are known, the MUSIC algorithm can be applied to *m(t)* in order to estimate each delay τ_i.

The algorithm is described by the steps below :

- step 1: necessary assumption making: the sound of opportunity is frequency modulated and a ray propagation hold, so the received measurement is given by: $$m(t)=\sum_{i=1}^{N}a(i)e(t-\tau_i);\ e(t)=a(t)\exp(2\pi j\int_0^u f_i(u)du+j\varphi(0))$$ $\forall t\in[0,T]$,
- step 2: time windowing of the signal around any time t_0 ($t_0\in[0,T]$) such as on the observation area, $f_i(t)$ can be approximated by its first order Taylor series development and then m(t) can be approximated by a sum of delayed linear frequency modulation $$m(t)\approx\sum_{i=1}^{N}a(i)\exp(2\pi j(f_i(t_0)(t-\tau_i)+\alpha(t_0)(t-\tau_i)^2))$$ $$\alpha(t_0)=\frac{\partial f_i(t_0)}{\partial t},$$ $$\forall t\in[t_0-\frac{L}{2};t_0+\frac{L}{2}]$$
- step 3: Dechirping by the multiplication of the windowed signal by the conjugate of the linear frequency modulation $$d(t)\approx m(t)\times\exp(-2\pi j(f_i(t_0)t+\alpha(t_0)t^2))$$ reference $$\approx\sum_{i=1}^{N}K_i\exp(-4\pi j\alpha(t_0)\tau_i t),$$ $$\forall t\in[t_0-\frac{L}{2};t_0+\frac{L}{2}]$$
- step 4: for each t_0, estimation of τ_i by spectrum analysis of d(t) thanks to MUSIC algorithm for example and building of a binary time frequency (BTFM) representation by setting $BTFM(t_0-\tau_i,f_i(t_0))=1\ \forall i\in\{0,\dots,N\}$,
- step 5 : estimation of the impulse response of channel by time frequency correlation $$\hat{h}(t)=\int_{-\infty}^{\infty}BTFM(u,f)\delta(f-f_i(u-t))dudf.$$

The critical point of the algorithm is to determine automatically for each time t_0, the optimal neighborhood where the chirp-like assumption is valid. This is achieved by looking for the length L of a rectangular time window ($w_L(t-t_0)$) to apply to *m(t)* which minimizes the spread of the Fractional Fourier Transform of the windowed signal. More about this algorithm can be found in [4].

III-2 Real world data application

Among the large set of collected data from laurentian Channel data set, impulse sound with bandwidth from 0 to 1 kHz, frequency modulation from 500 Hz to 8 kHz, and narrow band sound produce the three major families of marine mammals sounds where multi-paths structure of the measurement can be observed using the first two ones. A Beluga vocalize with decreasing frequency modulation of 2 kHz bandwidth (see fig3) is selected as a test bench for our algorithms.

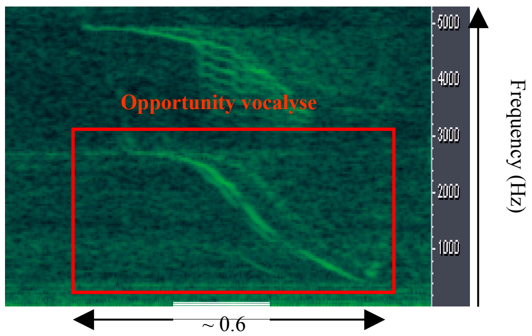

Figure 3. Spectrogram of the opportunity vocalize (Window used: Hamming 25ms)

The time frequency in figure 3 is the spectrogram of the test sound recorded at the first hydrophone of the coastal array.
Resolved multi-paths can be observed on the spectrogram. It seemed to confirm our hypothesis of acoustic ray paths propagation, then it is proposed to estimate the impulse response of the channel between beluga and the receiver with low resolution algorithm developed in [1] and high resolution algorithm presented in this section and to compare them with a simulated one given by Bellhop code using the position of Beluga obtained by passive triangulation between the six ocean bottom hydrophones of coastal array and true bathymetry and sound speed profile.

Low-resolution algorithm provides estimates of instantaneous frequency law:

$$e(t)=\exp(2\pi j\int_0^t f_i(u)du),\ t\in[0,0.6s]$$

with $f_i(t)=10^4(-1.78t^3+1.43t^2-0.4t+0.18)\,(Hz)$

High-resolution algorithm applied on the same test sound provides the BTFM of the arrivals (step 4 of the algorithm, see fig.5) and the estimated impulse response (step 5 of the algorithm, see fig.7).

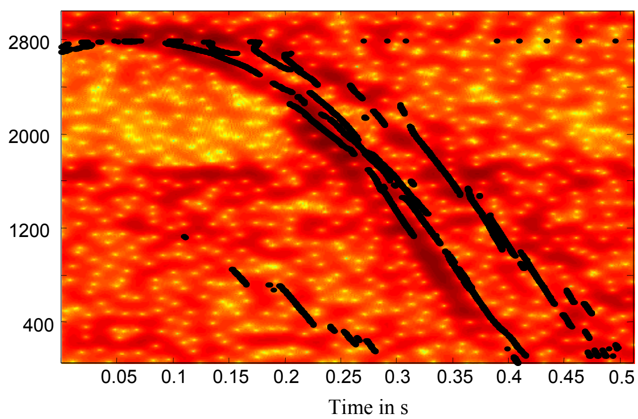

Figure 5. Spectrogram of the opportunity vocalize (in red colormap) and the 4th order MUSIC BTFM (black point)

This algorithm seemed to give good results and succeeded in finding 4 coherent paths. However, to definitely validate our approach, we had to locate the source to be able to simulate the impulse response between beluga and receiver. For this, location is estimated by blind triangulation with relative time of arrival between all the hydrophones of the coastal array thanks to hyperboling fixing method described in [5] .
Because of the small aperture in deep, we were only able to locate the whale in Latitude and Longitude (48,2657 °N, -69.4640°W). To handle this problem, we simulated the propagation in the channel thanks to Bellhop (parameterized with true bathymetry and sound speed profile) for a source positioned at the Latitude and Longitude found by hyperboling fixing and for all depths between 0 to 240m.
On figure 6, an example of plots of the rays generated by BELLHOP is presented.

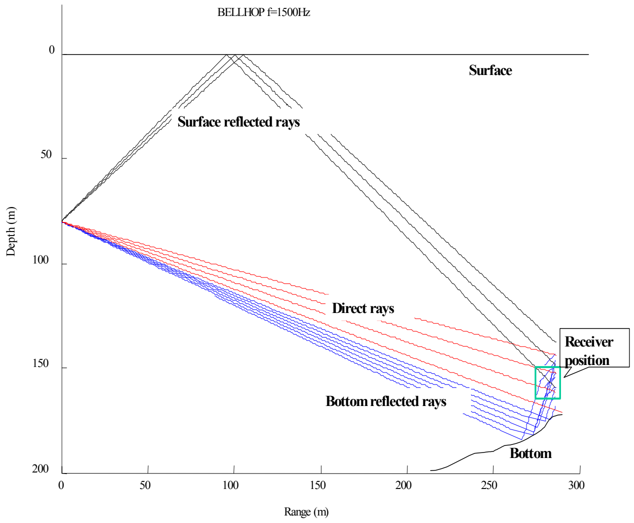

Figure 6. Plots of the rays generated by Bellhop obtained with a 80m-depth source.

Thus, several channel impulse responses are obtained as a function of depth, and they are compared to the impulse response estimated with the high-resolution algorithm described in the second paragraph. Finally, the more similar channel impulse response was found for an almost 80 meters depth.
On figure 7, a comparison between 5 channel impulse responses estimates is given:
- One simulated with Bellhop with a 80m depth source (fig.7 Curve 5)
- One obtained by an Active-Adapted Filter (we extracted the instantaneous frequency law of the real vocalize, then we constructed the corresponding signal and we made it forward in the simulated channel (fig.7 Curve 4)
- One given by high-resolution passive algorithm (fig.7 Curve 2)
- One given by low-resolution passive algorithm (fig.7 Curve 1)
- One obtained by an Active-Adapted Filter but with a smaller band. This band is equal to the larger chirp one, obtained during step 2 of the algorithm, actually 272 Hz. (fig.7 Curve 3)

This comparison seemed to prove the efficiency of our high-resolution algorithm because of the good fit observed between theoretical time of arrival given by Bellhop and passive estimation of the impulse response given by the algorithms. The level of resolution offered by the high-resolution passive algorithm is still far from the active matched filter one but is very close to the one obtained by active matched filter with local time windowing. This level of resolution is satisfying enough in our case to explain the temporal structure of the arrival.

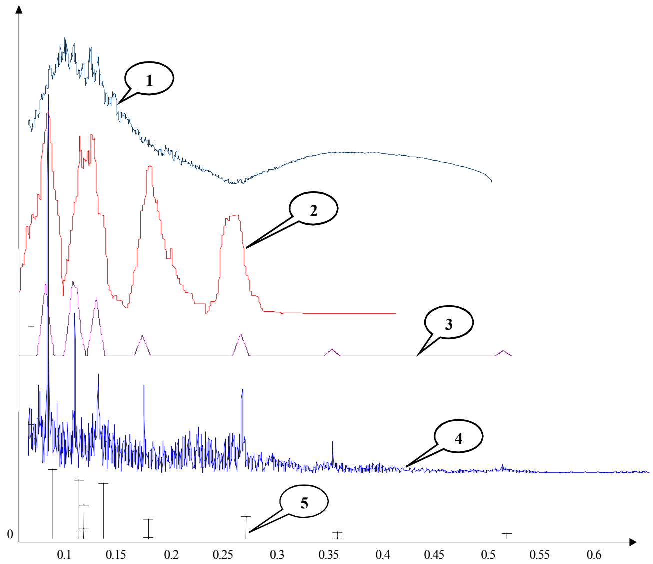

Figure 7. Comparison between Channel IR obtained with different methods

IV. FEATURES EXTRACTION FROM OPPORTUNITY SOURCES OF BANDWIDTH B_1

Acoustic sources belonging to bandwidth B_1 have great interest because low frequency waves penetrate far into the bottom and then allow to estimate its geo-acoustic properties.

Whereas time and magnitude of arrivals are the features of interest for bandwidth B_2, for low frequency, dispersion phenomena (a different group celerity between frequencies in a same acoustic mode and between different mode at a same frequency) carry lots of information and a feature to extract from measurements is the distribution of group time delay versus frequency and mode index of an acoustic wave propagation from an emitter to a receiver. In the Bay of Fundy data set, a lot of right whales calls stand for and they can be approximate to an impulse in B_1 bandwidth.

So the signal measured by an hydrophone is directly an estimation of the channel impulse response between whale and hydrophone. Then after location of emitted whale (by hyperbolic fixing) a fit (least square minimization for example) between theoretical and measured impulse responses may be performed to asses the geo-acoustic properties of the channel. In this chapter , we are presenting primilary results to this fit . Simply, we compare the time frequency structure of a right whale call (see Figure 8) after emission and propagation through the channel (periodogram, 1200 Hz sampling frequency, Kaiser 180 dB window, 256 samples length) and a simulated dispersion curve obtained with ORCA sets in line with chapter II-2 (see Figure 9) and 500 m range between whale and receiver. The time frequency transform of the estimated impulse response channel presents a good agreement with the simulated dispersion curves. These first statements of fact raise good perspectives to estimate geo-acoustic properties using right whale calls interceptions.

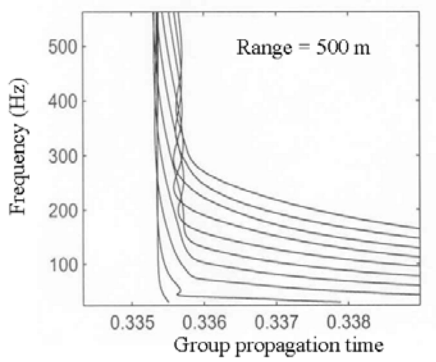

Figure 8 : Simulated dispersion curves

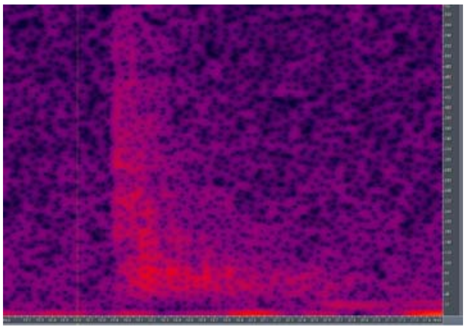
Figure 9 :Periodogram of measured Impulse Response Channel

V. CONCLUSION

In this paper, we address passive acoustic tomography using marine mammals vocalises and more precisely extraction of propagation features from single hydrophone measurement. Extraction procedures are tested on real world data.

For high frequency modulated calls, we propose a blind channel impulse response estimation algorithm. Its capabilities to perform the identification of the impulse response channel without using the knowledge of the emitted source signal has been demonstrated in the case of a single hydrophone. We have succeeded in applying it to real data obtained from a Laurentian Channel experiment performed during the summer 2003. Performances obtained with the high-resolution algorithm are close to classical active matched filtering methods. We will definitely validate our algorithms with the remaining Laurentian Channel data (data from the 5 ocean bottom hydrophones deployed in the centred square configuration, Fig. 1), which will allow us to improve the quality of the location of the emitted whales.

For low frequency impulse calls, we compare the time frequency structure of the impulse response estimate with simulated dispersion curves. Good agreements between simulations and measurements raise promising perspective to passive geo-acoustic inversion thanks to right whale calls.

VI. ACKNOWLEDGMENTS

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