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ELECTRIC AND ELECTRONIC ENGINEERING

ON IDENTIFICATION OF TRANSMISSION CHANNELS FOR SONAR BASED APPLICATIONS

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Impulse response function and time delay estimation from the impulse response signal are considered. The work is looking from the sonar based applications point of view, where the time difference between transmitted pulse and received echo is important for target and object detection and recognition. Two steps are considered: identification of the sonar channel (air), by impulse response signal and, second, estimation of the time delay from the shape of the identified impulse response. The results are at the level of other previous works and create the motivation to implement these algorithms in practice.

Key words: systems, time-domain models, signal processing, estimators, sonar.

1. INTRODUCTION

The basic structure of an in air ultrasonic transmission system contains a chain of blocks as: source (S), which is the ultrasonic transmitter; direct path (DP); Scatter (SR); inverse path (RP), and the receiver (R). At the first raw modeling level it is supposed that each block can be modeled by a linear function and time invariant.

For this work we consider that direct and inverse path as being composed of the same medium (air) and the scatter block is making a total reflection. No directivity pattern is considered for ultrasonic transducers. For all modeling block of the transmission system impulse response function is considered and estimated. Such an approach allows estimating the shapes of the received echoes for target recognition and classification purposes. Such an approach is used also in some works as [2], [8], [16], [17], or [23]. Time delay estimation is considered after impulse response estimation. The research is part of the adbiosonar project, [24].

Section 2 describes the models of the ultrasonic transmitters and receivers, the air channel models, and introduces the signals for source (S) modeling. Section 3 presents the used methods for system identification, in this case the transmission channel. Section 4 investigates four basic methods for time-delay estimation by having the estimated impulse response. Section 5 presents the simulation setup and evaluates the obtained results.

2. SYSTEM DESCRIPTION

A simple but general model is considered, which is composed of signal source *s(t)* and a transmission path characterized by impulse response function, *h(t)*. Signals at the site of measurements are affected by noise, $n_1(t)$ and $n_2(t)$. Thus, the signal acquired by each sensor is a delayed and attenuated version of the original source signal plus additive noise, as it is presented in Figure 1. For simplicity reasons, noise signals are considered independent.

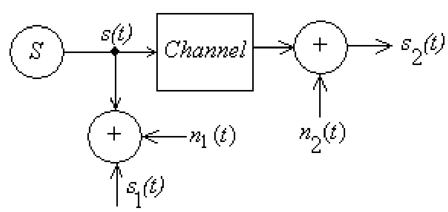

Fig. 1 - Signal modeling under single path propagation hypothesis

The model of the involved signals is described by the set of equations:

$$\begin{aligned} s_1(t) &= s(t) + n_1(t) \\ s_2(t) &= h(t-D) \otimes s(t) + n_2(t) \end{aligned} \tag{1}$$

The measured signal $s_1(t)$ is considered different of the transmitted one, *s(t)*. This does not affect other used algorithms and methods, but reflects more realistic the experiment. In the case of the transmission only, without measurements, the signal *s(t)* at the input of the channel is also affected by noise coming form connecting wires and physical connectors.

The first considered problem is to find an estimate of the impulse response function, *h(t)*, and - next problem - to estimate time delay *D* which characterize the air transmission channel.

Generation of the echo is based on absorption and delay features of the air as propagation medium. The transducer model is considered as pass band linear filter. Starting from a model of a narrow range frequency type transducer and considering a pair, transmitter-receiver, the following model could be used, verified also by experimental measurements, [2], and [15]:

$$h_{tr/rec}(t) = \exp\left(-(t-\tau)^2 / 2\sigma^2\right) \cdot \sin\left(2\pi \cdot f_0 t\right) \tag{2}$$

where f_0 is the resonance frequency and σ is inversely proportional to the centralized bandwidth of the filter, and τ models the inertia of the transducers.

An elementary transmission channel of length L, including the time return of the echo, working on frequency f_i, is modeled by an impulse response function as

$$h_{air,i}(t) = \begin{cases} \exp\left[-a_i \cdot c_i\left(\cdot t - D\right)\right] / a_i, & t \geq D \\ 0, & t < D \end{cases} \tag{3}$$

where a_i is attenuation at the distance L and c_i is a constant correction factor for frequency f_i and D is the delay. In the case of wide frequency range, Eq. (3) must consider a summation for all frequency components. By using the convolution operator, see e.g. [14], the received signal is described by

$$s_2(t) = s(t) \otimes h(t) + n_2(t) = s(t) \otimes h_{tr/rec}(t) \otimes h_{air}(t) + n_2(t) \tag{4}$$

3. IDENTIFICATION OF THE CHANNEL

The transmission channel means the direct path, an ideal reflector and the return path. The medium of propagation is air. Taking into account two basic phenomena during the propagation of sound in air, absorption and delay, the transmission channel is considered modeled as a linear passive system. A model of such system is the impulse response function, described by Eq. (3), which can be estimated in various ways. Two methods are important: (*i*) correlation method, e.g. [5], [7]; (*ii*) estimation of parameters model (filter), which are the samples of the impulse response signal, e.g. [7], [13].

In this work an adaptive estimation method is used for impulse response estimation. The mean-square-error (MSE) error criterion is used, which is based on ensemble averaging of the squared estimation error *e(n)* between desired signal, *d(n)*, and the estimated one, $\hat{d}(n)$,:

$$J(n) = E\left[|e(n)|^2\right] = E\left[\left|d(n) - \hat{d}(n)\right|^2\right] \tag{5}$$

Desired signal *d(n)* is equal with the input, *d(n)= u(n)*, for the case of one step ahead prediction. The used input vector at time *n* into transversal filter is

$$\mathbf{u}(n) = [u(n) \quad u(n-1) \quad ... \quad u(n-M+1)]^T \tag{6}$$

and the estimated weights vector at time *n*:

$$\hat{\mathbf{w}}(n) = [\hat{w}_0(n) \quad \hat{w}_1(n) \quad ... \quad \hat{w}_{M-1}(n)]^T \tag{7}$$

By considering real values only for all involved signals and parameters, the Normalized Least-Mean-Square Adaptive Filters (nLMS) formula for computing the *M*-by-1 tap-weight vector is used as

$$\hat{\mathbf{w}}(n+1) = \hat{\mathbf{w}}(n) + \tilde{\mu}\frac{\mathbf{u}(n)}{\|\mathbf{u}(n)\|^2} \cdot e(n) = \hat{\mathbf{w}}(n) + \tilde{\mu}\frac{\mathbf{u}(n)}{\|\mathbf{u}(n)\|^2} \cdot \left(d(n) - \mathbf{u}^T(n) \cdot \hat{\mathbf{w}}(n)\right) \tag{8}$$

where $\tilde{\mu}$ is a real positive scaling factor and

$$\|\mathbf{u}(n)\|^2 = \sum_{k=0}^{M-1} u^2(n-k) \tag{9}$$

is the squared Euclidean norm of the tap-input vector **u**(*n*).

The computed value for the vector **w** represents an estimate whose expected value may come close to the Wiener solution, $\mathbf{w}_0$, as the number of iterations, *n*, approaches infinity. The tap-input vector **u***(n)* and the desired response *d(n)* are drawn from a jointly wide-sense stationary environment. More details and particular cases are discussed in literature of adaptive filtering, e.g. [10].

Once the impulse response function is acceptable, the time delay can be computed by various methods, e.g. thresholding, as presented in the next section, and might be used to predict the shape of the future coming signal (echo).

4. TIME DELAY ESTIMATION (TDE) FROM IMPULSE RESPONSE FUNCTION

Ref [5], for example, shows how to compute the delay from an impulse response function. Two basic ideas are presented now, from the same reference. Let $\hat{h}(t)$ the estimation of the impulse response. The time-delay can be estimated by either of the following two approaches:

1). Separation of time-delay and dynamics and then estimation:

(a) Separate the pure time-delay $\delta(t - D)$ from the dynamics $h_0(t)$ (system with no delay) of the system $h(t)$.

(b) Estimate the time when the maximum of the estimate $h_d(t)$ of the time-delay $\delta(t - D)$ occurs.

2). Direct detection of the start of the impulse response:

(a) Detection of the main lobe; (b) Estimation of the start time.

Based on the particular shape of the impulse response for this application, in this work we have used four basic methods, which are close to the second approach. All methods are described from the second step, i.e. estimation, because detection process of the impulse response it is supposed already done.

The structure of the first two methods is presented in Figure 2. A Direct Method (DM) uses computation of the coordinate of the maximum value of the impulse response. Various case studies and simulations scenarios might be found in [1], [3], [6], [9], and [11]. The estimated delay is

$$\hat{D}_{DM} = \left(\arg\max_{k} \hat{h}(k) \right) \cdot T_s \tag{10}$$

The second one, called Gradient Method (GM), computes firstly the gradient of the impulse response and then compute de coordinate of the maximum gradient. The reasons of the second method is coming from the fact that the impulse response is starting to rise close to the moment when the impulse response function has the greatest change, from low values to highest value. The estimated delay is

$$\hat{D}_{GM} = \left(\arg\max_{k} \frac{d\hat{h}(k)}{dk} \right) \cdot T_s \tag{11}$$

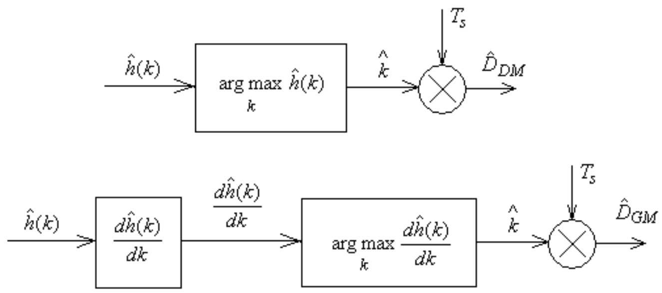

Fig. 2 - Structure of the direct (DM) and gradient (GM) time delay estimation methods, based on estimated impulse response function

The principle of the third method, the method of line (LM), is presented in Figure 3. The time delay is given by the intersection of the line (*d*) with horizontal axis *Ox*, i.e. the value of the x_3. The necessary two points to compute the line's equation are given by the point where the gradient of *h(t)* is maxim (the point P_2), and – the second point - by the maxim value of the impulse response function (the point P_1). This method of computation of x_3 will be referred as the line method (LM). The value of x_0 is the real value of the delay. From geometrical representation at least, it seems that the line method estimates by missing and the first two methods (direct and gradient based) estimate by truncation. In the most general case, an efficient estimation must take the mean of the three basic estimations, i.e. x_1, x_2 and x_3. The structure of the method is presented in Figure 4.

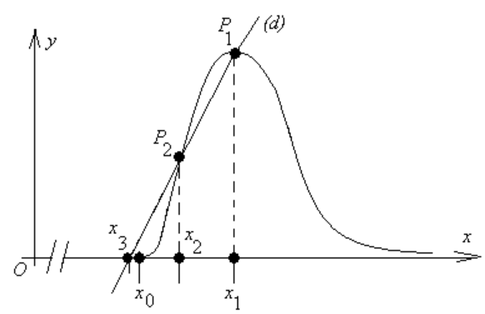

Fig. 3 - Estimation of the time delay by method of line (LM)

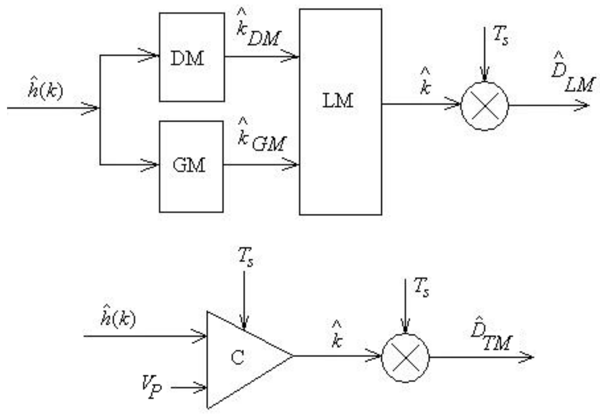

Fig. 4 - Structure of line (LM) and triggering (TM) time delay estimation methods, based on estimated impulse response function

The triggering method (TM), described in Figure 4, is the simplest from implementation point of view. A circuits counts the samples of the impulse response function, until a greater value then a threshold is found. The time delay is equal with the locked decimal value of samples multiplied by sampling interval, T_s. There are some helpful guidelines concerning the choice of the threshold, e.g. [5]. The threshold must have a higher value then any spike coming from the transmission channel, i.e. from noise. Otherwise, wrong results could be obtained.

In the next we suggest a formula based on the probability of the peak noises over transmission channels. For practical reasons, a zero mean white Gaussian noise*)* *N(t)*, is considered with *w(x)* being the probability density function. There is a relation between the peak value $V_p > 0$ of the noise and the probability to take values outside the interval $(-V_p, V_p)$ as

$$P(|N(t)| > V_p) = 1 - \int_{-V_p}^{V_p} w(x)dx \qquad (12)$$

and for Gaussian signals with variance σ^2 we have

$$P_V = 1 - \frac{1}{\sqrt{2\pi\sigma^2}} \int_{-V_p}^{V_p} \exp\left(-\frac{x^2}{2\sigma^2}\right)dx \qquad (13)$$

If the probability of false alarm (or non-detection) P_v is imposed, numerically the value of peak value V_p can be computed, which will be equal – in this context – with the necessary threshold to use in time delay estimation from impulse response function. Thus, by considering the error function as

$$erf(x) = \frac{2}{\sqrt{\pi}} \int_0^x \exp\left(-t^2\right) dt = 1 - erfc(x) \tag{14}$$

the next relation is valid:

$$P_V = 1 - erf\left(V_p / \sqrt{2\sigma^2}\right) = erfc\left(V_p / \sqrt{2\sigma^2}\right) \tag{15}$$

with

$$erfc(x) = \frac{2}{\sqrt{\pi}} \int_x^{\infty} \exp\left(-t^2\right) dt \tag{15.a}$$

For a given power noise and false alarm probability (P_v) Eq. (15) gives the value of the necessary trigger value (or threshold). Figure 5 presents the evolution of the V_p for various values of the probability, P_v.

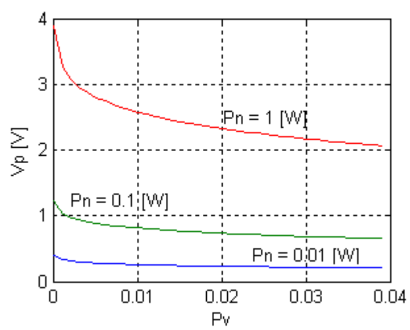

Fig. 5 - Threshold – probability dependency; one Ohm resistor load.

Other criteria for threshold computation are based on the power of the noise over the channel, estimated or measured, as it presented in [18], or [20].

5. SIMULATION SETUP AND RESULTS

A. Signal generation

Roughly, the general guidelines of the generation signals from system identification point of view are considered, i.e. the input signal should be close to the random behavior, white and with Gaussian distribution.

The input signal is binary (pseudo)-random and flat distribution of the amplitude spectrum inside of the useful frequency range. The signal might be generated by digital devices available on any signal processing board.

The ultrasonic transmission system could have two kind of transducers, which work on fixed frequency (called here, narrow range) or on wide range of frequency. The last case is covered by band pass ultrasonic transducers, e.g. EMFi based transducers.

For the narrow frequency range a central frequency of 40 kHz, a 3dB bandwidth B = 5 kHz, and an internal delay $\tau = 10\mu s$, were considered For the wide band case, the frequency range is from 40 kHz to 200 kHz, and B = 300 kHz.

Figure 6 shows the basic signal involved in simulation, for the case of a narrow range and Figure 7 for the case of wide frequency range. In both cases, the input signal in the transmission system, the output of the digital device (microcontroller), is a binary random signal with flat amplitude spectrum over the bandwidth of the channel, as it is presented in Figure 7. In the narrow case, the amplitude spectrum of the signal after ultrasonic transducer corresponds to a band pass filter with a very small value of the 3dB bandwidth (~ kHz).

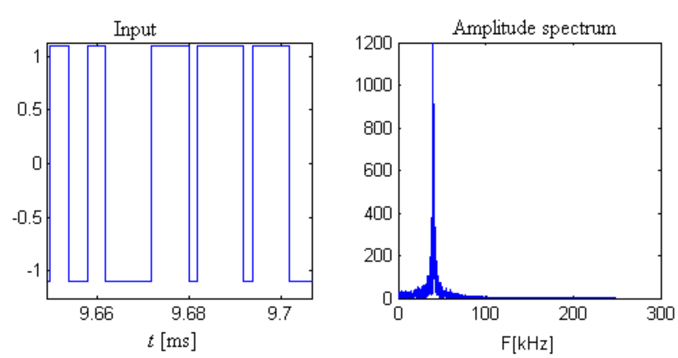

Fig. 6 - Input signal for the narrow frequency range case

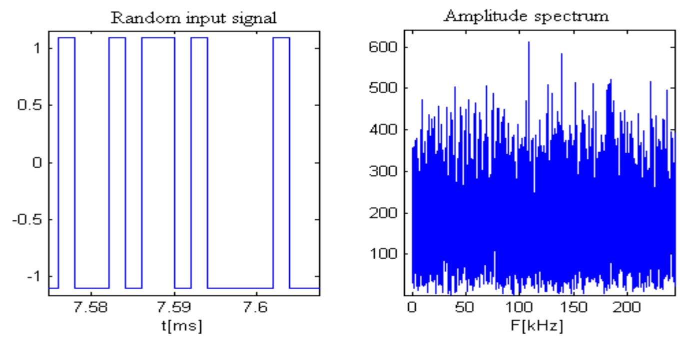

Fig. 7 - Input signal for the wide frequency range case

B. Identification results

The first round of simulation was with a transversal filter of length M = 2000. Taking into account a sampling interval of $T_s = 2\mu s$, a range resolution of 0.68 mm is obtained, which is more than enough for many applications. Figure 8 presents the final estimated impulse responses for various signal-to-noise ratios (SNR) and narrow frequency range (NR). Even the number of considered processed samples is quite large, there still are some small errors. The second round with M = 1500 gives similar results.

Figure 9 presents the final estimated impulse responses for various signal-to-noise ratios (SNR) and wide frequency range (WR), i.e. from 40 kHz up to 200 kHz. As expected, because wide range frequency decrease the variance of the estimation error (see e.g. [12], [19], [21], [22]), the estimation errors after Q = 10000 updates are much smaller comparing with the previous case.

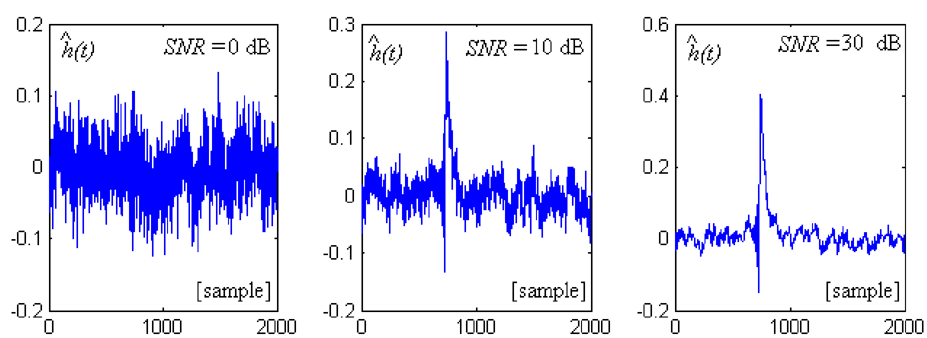

Fig. 8 - Identification results: narrow frequency range (NR)

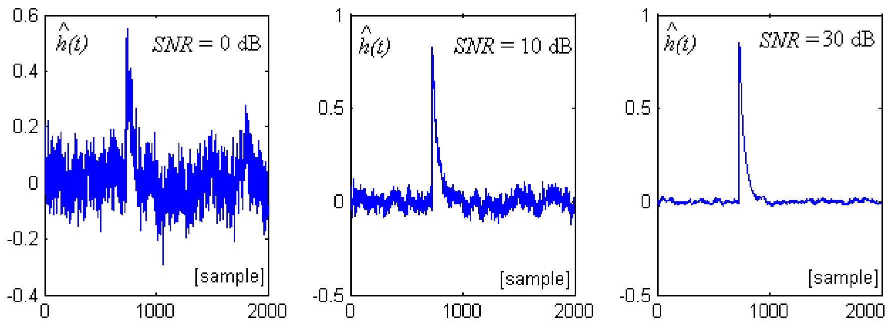

Fig. 9 - Identification results: wide frequency range (WR)

C. Time Delay Estimation (TDE) results

Figure 10 presents some waveform (impulse responses) from the time-delay estimation by using direct thresholding method. The threshold level, represented with dashed line, is imposed as ten times higher the effective amplitude of the noise computed from the measured power over the transmission channel. It is the simplest method and widely used in practice.

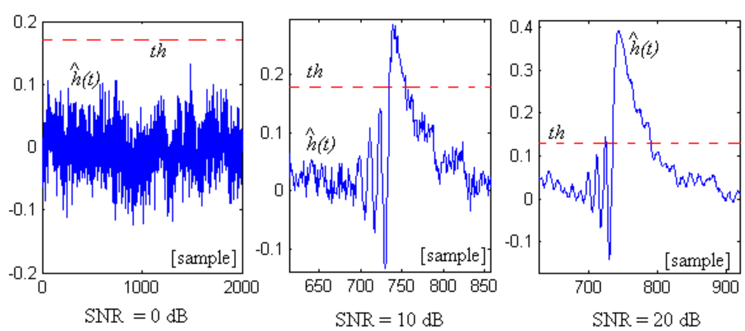

Fig. 10 - Results of the triggering method (TM) , NR case

Figure 11 presents the estimation error for three used methods: DM, GM, LM, and TM for narrow range case (NR). Figure 12 shows the same criterion but for wide frequency range (WR). The numerical values associated to these figures are presented below in Table 1, with bold characters for the best obtained estimations. The right hand colon shows the mean value computed for signal-to-noise ratio from 10 to 30 dB. The case of 0 dB was not considered because it generated bad results, far of values for high SNRs.

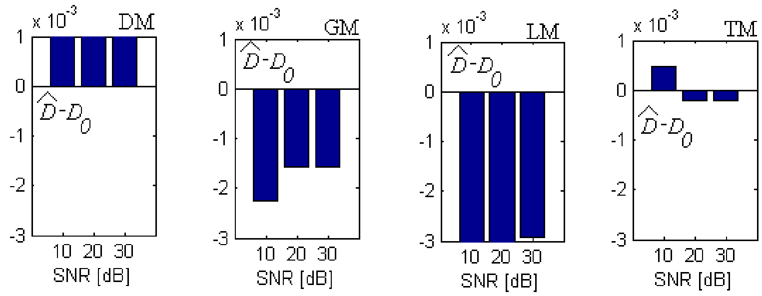

Fig. 11 - TDE bias results, narrow range, under various SNRs; L = 0.5 m

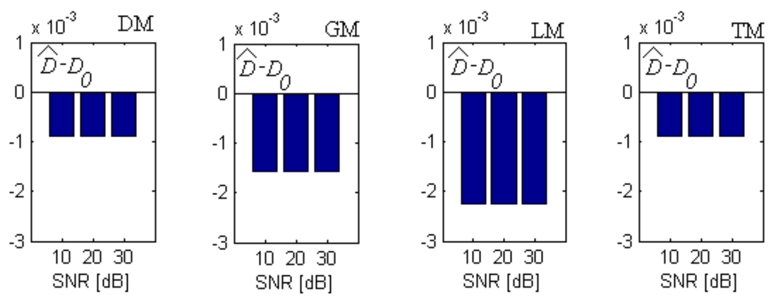

Fig. 12 - TDE bias results, wide range, under various SNR; L = 0.5 m

Table 1

Numerical values of distance estimations [m]

		SNR [dB]				
		0	10	20	30	Mean*
WR	DM	0.5032	**0.4991**	**0.4991**	**0.4991**	**0.4991**
	GM	**0.4984**	0.4984	0.4984	0.4984	0.4984
	LM	0.4950	0.4978	0.4978	0.4978	0.4978
	TM	1.3600	**0.4991**	**0.4991**	**0.4991**	**0.4991**
NR	DM	1.0132	0.5025	0.5059	0.5052	0.5045
	GM	**0.6678**	0.4978	0.4984	0.4984	0.4982
	LM	0.1742	0.4964	0.4964	0.4971	0.4966
	TM	1.3600	**0.5005**	**0.4998**	**0.4998**	**0.5000**

* without values of 0 dB.

6. CONCLUSIONS

The objective of the work was to evaluate some methods and algorithms from the general field of linear systems identification, to obtain valid models for the air transmission channels used in sonar applications and, based on this, to estimate the time-delay of the channel which is an important parameter in the model of the channel and sonar applications.

Two case studies were considered, as transmission systems: narrow and wide frequency ranges. The first one corresponds to ultrasonic transducers working on fixed frequency, and the second one corresponds to wide range ultrasonic frequency. Both cases are considered and explored in the adbiosonar project, [24].

Impulse response function was estimated by using an adaptive filter with transversal structure, for simplicity reasons.

Concerning time delay estimation, for wide range identification signals two equivalent methods are suitable (from the estimation error criterion point of view): impulse thresholding and direct method, based on the index estimation of the highest value. For narrow range, the best results are obtained by thresholding. These are valid for high values of the SNRs, between 0 and 30 dB.

For low values of the SNR the best estimations, even far from real ones, are given by the gradient method. The line method gives bad results, at least for the considered signals and simulation scenarios.

More attention must be paid in the future works to the evaluation of the performance of the new introduced methods (GM and LM) in order to evaluate the variance of the estimation error and to compare with some theoretical limits, as the Cramer-Rao bound.

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