Development of a Real-Time High Data Rate Acoustic Link

J. Trubuil*, T. Le Gall*, G. Lapierre**, J. Labat*
* ENST-Bretagne, BP 832, 29285 Brest Cedex, France
** G.E.S.M.A. (Groupe d'Etudes Sous-Marines de l'Atlantique), BP 42, 29240 Brest Naval, France

Abstract- **The GESMA is developing a high data rate acoustic link whose robustness must allow communication between a mobile and a surface vessel. The selected multi-channel blind equalizer is designed to follow strong variations of underwater acoustic channel (UWA). This patented equalizer developed by ENST-Bretagne is the heart of a digital processing whose integration on a DSP (Digital Signal Processor) platform is currently carried out. Acquisition, timing, carrier and phase recovery, equalization, and decision are jointly optimized. Its validation must make it possible the GESMA to transmit real-time informations to data rate higher than 25 kbps in horizontal configuration.**

Fig. 1. Illustration of SEA TWIN.

I. INTRODUCTION

The GESMA has led for several years a project relating to an high data rate acoustic link. The required goal is to equip Redermor (AUV developed by GESMA) with a system allowing wireless communications from the vehicle towards the sea surface. Two acoustic links are integrated (a low data rate for control and a high data rate for significant transfers). After several evaluations of different processing among which beamforming and equalization, the project is currently focussing on the implementation of one of these methods on a DSP, in order to design a real-time high data rate acoustic link. This paper presents the state of this work. In a first part, the objectives are described. Some recalls on the state of the art in coherent receivers will place authors contribution in the field. A description of their solution (detailed in [1]) is then given in section IV. Finally, a description of the real-time processing and some DSP's results are presented.

II. OBJECTIVES OF THE PROJECT

The GESMA will soon carry out a new performance evaluation of an high data rate acoustic transmission in accordance with the last sea trials [2] and [3]. The originality of the approach will consist in testing the whole communication system (acquisition, demodulation, equalization and decision).

The implementation is carried out by the ENST-Bretagne on the basis of a platform including a Texas Instruments DSP (TMS320C6201). Once ready, this platform will be evaluated during the next sea-trials scheduled in 2002. This adaptive receiver will be tested in real time during these sea trials.

The aim of the project is to allow high data rate from 10 to 40 kbps in horizontal configuration (ranges around 1000m).

The transmissions are designed to be carried out in shallow water with mobile displacements.

The type of transmission will not only interest continuous stream of data but also burst transmissions. These last can have a particular interest compared to the low data rate modulations in terms of temporal occupancy. Moreover, the concept of underwater acoustic networks are such as the integration of high data rate acoustic link is imminent in order to save available temporal and spectral bandwidth. A paper in preparation will describe the use of the equalizer presented in [1] as an interesting solution for the small sequences of data.

Sea trials are scheduled for transmission of data coming from either a camera CCD either a database. Signals will be collected on an antenna of 4 hydrophones whose spacing will be adjustable in order to quantify the contribution of spatial diversity on performance. The modem will be carried by a robot of the GESMA - SEA TWIN presented Fig. 1 in order to estimate the influence of movements on the reliability of the transmission.

The objective of the GESMA is to develop a sufficiently robust acoustic link for the purpose of making the mine hunting vehicles autonomous.

The link carried out will be sufficiently modular to test many configurations and to evaluate the robustness of the link on many scenarios.

III. SOME RECALLS ON THE CURRENT STATE OF THE ART ON HIGH DATA RATE ACOMMS

The first coherent treatments appeared as soon as fluctuations of phases brought by underwater acoustic (UWA) channel (reverberation) were on the way to be controlled in particular by means of adaptive algorithms. Phase modulation allows to considerably increase the effective data rate in a given bandwidth. In order to raise the phase ambiguity, differential encoding is used. Nevertheless, the number of states of phase seldom exceed 8.

The first coherent systems were mainly dedicated to transmission in vertical channel as the TIVA system [4] and [5]. Carrier frequencies are rather high and thus limit the range of such systems but offer a broad bandwidth. Then, the STAP developed by the DERA [6] offers bit rates from 20 up to 41 kbps with a carrier frequency of 30 kHz. The university of Birmingham developed a system operating at 50 kHz with a bandwidth of 10 kHz. Lastly, the Woods Hole Oceanographic Institute (WHOI) brought an important contribution with a system using a carrier frequency of 15 kHz which allows transmissions in QPSK with 5 kbps. Very complete papers describe the evolution of coherent treatments [7], [8] and [9].

Coherent treatments must first of all bring back the signal in baseband by a demodulation and a sampling carried out usually by a complex stage of demodulation (recovery of the I and Q component) followed by a sampler using a T-spaced or a fractionally spaced approach. Other systems carry out these operations jointly [10] by sampling the input signal at a rate four times higher than the carrier frequency. Timing recovery is achieved by the Gardner algorithm [11]. Optimal approach in the maximum likelihood sense is not exploitable in UWA context. Generally, an equalizer is used instead and defined on the choice of:

A. *Its structure*

In most systems, linear transversal filters are used. However, when faced to very harsh channels, their global performance turn out to be really poor. That is why decision feedback equalizers are often used in such situations, since they are able to track deep-fading channels.

For that purpose, the multiple-input equalizer introduced and described in [1] will be used and evaluated in this paper. To sum up, this equalizer is mainly characterized by an adaptive self-optimized configuration in terms of both structure and criterion for adaptation. In the first running mode, the equalizer is linear and recursive and its parameters are adapted according to relevant criteria that lead to a solution which is very close to the MMSE solution. Then, at the price of a structural modification, the equalizer is configured as the conventional multiple-input DFE. A relevant signal such as the estimated MMSE or the Godard function allows to select the operating mode. The purpose of this paper is to bring some results about this equalizer in the field of UWA communications.

B. *The criteria for adaptation*

Two approaches are available. Roughly speaking, systems generally use trained approaches. In this case, trained sequences have to be periodically transmitted, decreasing the effective data throughput.

To avoid this drawback, blind approaches turn to be very attractive but before [1], there was not robust solution for adapting the DFE parameters at their right solution. Naturally, some previous works had been involving a linear recursive structure and a global criterion such as the Godard (CMA) one. Unfortunately, the global criterion leads to a solution which can be far from the MSE solution. Otherwise stated, such approaches are not robust at all, at the difference of the solution proposed in [1] which can brings the equalizer parameters very close to the MSE solution.

In order to cope with the demodulation frequency offset as well as Doppler effect, a phase recovery function is generally added. Some references describe dedicated cells of Doppler compensation [12].

Lastly, some improvements focus on the use of DFE in a SIMO (Single Input Multiple Output) context and the use of spatial treatment. In particular one can finds the spatial diversity processing [3], beamforming before equalization [13] even a combination of both [14].

Current work is focusing on the improvement of the reliability of acoustic links [15], the equalization of maximum phase channels (time reversal structure [16]) and to decrease the computational complexity by the reduction of the number of taps (in particular sparse equalization [17]). Let us recall that the delay spread can reach 100 symbol durations (i.e., 10 ms) for a baud of 10 kbauds, as for typical UWA channels met around Brest. [18] confirms this aspect with US channels.

Finally, some efforts in term of acoustic compatibility and integration of high data rate acoustic link in multi-user systems are to be underlined and evaluated in current developments in the field [19] [20].

From a general point of view, LMS or RLS type algorithms can be used. Current developments have brought many results comparing the two approaches [21] [22]. Contradictory comments do not allow to bring safe conclusions.

IV. THE BLIND SOC-MI-DFE

We are now going to briefly recall the adaptive bind equalizer introduced in [1] and called self-optimized configuration multiple input decision feedback equalizer (SOC-MI-DFE).

This blind equalizer can reach a high level of performance, in terms of speed of convergence, mean squared error (MSE) and bit error rate (BER). More over it is characterized by a structural and algorithmic adaptivity which enables it to deal with variations of UWA channel.

In addition only, this equalizer increases data throughput by transmitting only data. Since, it is no more necessary to use preamble or another training sequence.

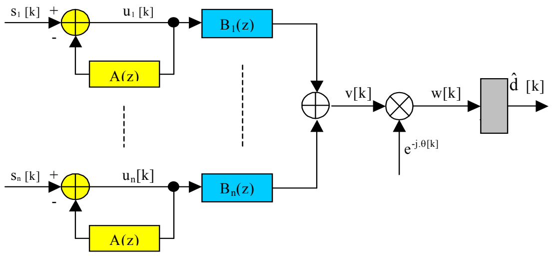

Fig. 2. SOC-MI-DFE (convergence mode).

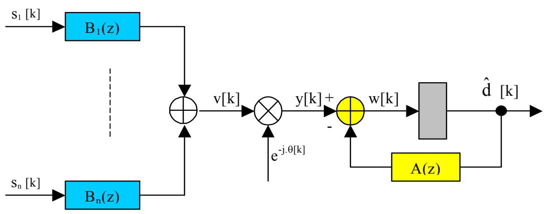

Fig. 3. SOC-MI-DFE (tracking mode).

In its convergence (or starting mode) the SOC-MI-DFE has a linear structure including a recursive filter A common to all branches and a transversal filter B specific to each branch. The recursive filter is adapted according to a criterion J(A) which stands for the cumulated power of outputs of this filter. Transversal filters are adapted from a blind criterion namely the Godard (CMA) algorithm. Only the phase recovery is decision directed in this mode. Fig. 2 presents the equalizer in its convergence mode.

When confidence in decided data is high, this can be evaluated by comparing the estimated MSE to a threshold J_0, one can show [1] that the structure can evolve towards a decision feedback equalizer by simple permutation of filters. Fig. 3 shows the final structure. Once commutation is carried out, filters A and B are adapted by decision directed criteria.

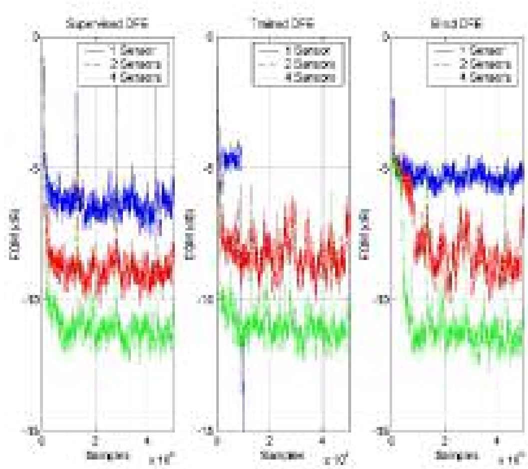

Fig. 4. Evolution of MSE for supervised, trained and blind spatio-temporal DFE (4-PSK).

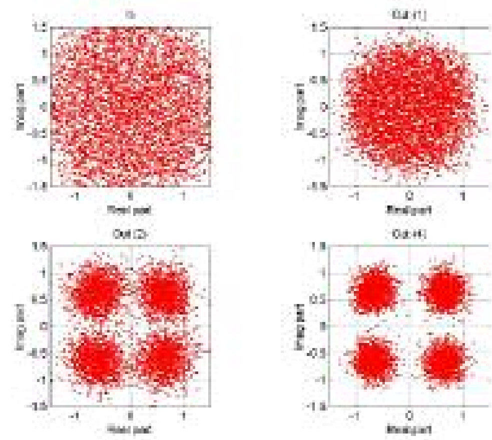

Fig. 5. Input and output constellations for the blind spatio-temporal DFE with 1, 2 & 4 sensors (4-PSK).

The interest of this adaptivity lies in the possibility to switch from one structure to another according to the channel severity. Let us recall that each stage is adapted by a specific criterion which confers an increased robustness to this receiver.

Treatments have confirmed the interest of blind approaches and the need for spatial treatments. Indeed, when the medium offers a too consequent variability, the system can be lost. At this moment, coherent receivers need retraining.

Fig. 4 shows a result synthesizing these two aspects obtained on PSK-4. It concerns a transmission carried out on 50000 symbols with a bit rate of 25 kbps in a range about 1000 meters. The three plot draw the MSE obtained with the same structure of equalization but adapted either in supervised mode, or in trained mode, or in blind mode. The various curves show the MSE reduction obtained with the use of 1, 2 or 4 sensors. Transversal filters contain 15 coefficients (12,1,2) and the recursive filter 60. One can note the very good behavior of the blind approach in term of MSE with respect to the other approaches in particular the supervised approach. One can also note the effect of the depth sounder which disturbed the trained approach at the very beginning of the sequence. One can finally note the influence of spatial diversity.

Fig. 5 shows the corresponding constellations. Moreover extra details are provided in [3].

V. REAL-TIME PROCESSING

The receiver platform is based on an acquisition card which is plugged in a personal computer (PC). The architecture of this card is based on a Texas Instruments digital signal processor (DSP) namely the TMS320C6201 [23]. This DSP has a clock frequency of 200 MHz (5 ns cycle time) which allows a theoretical performance of 1600 MIPS (Million Instructions Per Second) to be reached.

Fixed-point representations in two-complement coding are used and data are quantified in 8, 16 or 32 bits format.

A parallel architecture allows this DSP to execute up to 8 instructions per cycle time.

The real-time processing is implemented with an Integrated Development Environment (IDE) called Code Composer Studio (CCS) [24]. CCS not only provides tools required for development (edition, debug, compilation and linking) but also optimization tools. Furthermore, every signal can be visualized by CCS both in time and frequency representations (Fig. 6).

The program is written in C language and built by splitting the main routine into elementary functional subroutines since the SOC-MI-DFE calls many similar functions. Thus readability as well as portability are increased. Nevertheless, the most expensive functions in term of MIPS are rewritten in assembler language. In spite of CCS optimization tools, these functions were manually optimized. Fig. 7 shows the superiority of this approach in term of processing time to perform a 20 taps FIR filtering.

Considering a hand-coded assembly program using the software pipelining technique [25] , the time to process a 20 taps FIR filtering can be reduced by 90% compared to a C self-optimized program with CCS.

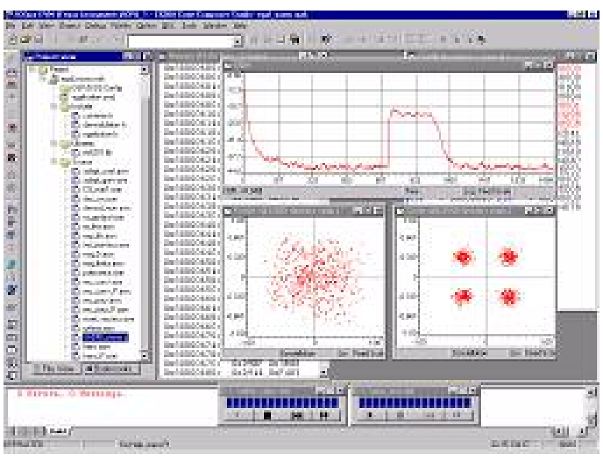

Fig. 6. Code Composer Studio IDE.

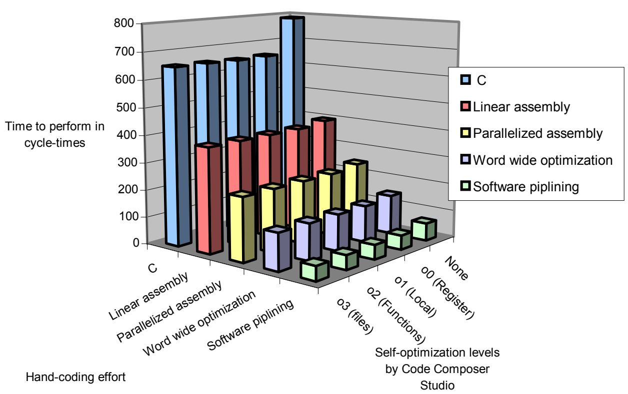

Fig. 7. Hand-coding effort vs. CCS self-optimization . Time to perform a 20 taps FIR filtering.

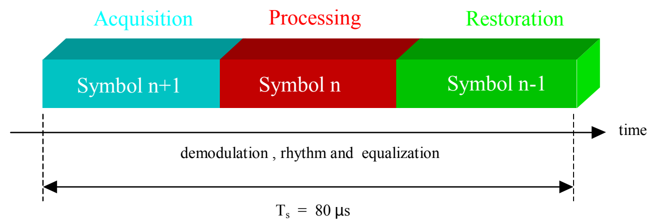

Fig. 8. Real time constraints.

The selected strategy for demodulation consists in synchronously sampling each input signal. As a consequence, demodulation can be performed using digital processing, which allows just one digital analog converter (DAC) to be used for a given input signal. The underlying principle is to oversample every signal at a clock frequency $4f_0$, f_0 denoting the carrier frequency. In addition, symbol duration T as to be chosen in such a way that the product f_0T is integer.

Timing recovery is achieved by the GARDNER algorithm [11]. The error signal u(n) is computed as:

$$u(n) = \sum_{i=1}^{P} \mathrm{Re}\{x_i^*(n-1/2)(x_i(n) - x_i(n-1))\} \cdot \qquad (5.1)$$

The samples respectively labeled $x_i(n-1/2)$ and $x_i(n)$ represent samples at times epochs t = (n-1/2)T and t = nT, for the i-th sensor. Re{.} denotes the real part of the expression between bracket. The error signal is then filtered and used to control a numerically controlled oscillator (NCO).

T-spaced equalization is performed involving the samples $x_i(n)$. However, for timing recovery, samples $x_i(n-1/2)$ and $x_i(n)$ are used and had be previously filtered.

Real-time constraints require to simultaneously process the acquisition of the (n+1)th symbol while processing the n symbol and making the restitution of the (n-1)th one. Fig. 8 shows these different stages to be considered. For a bit rate of 25 kbps, the processing time required by all these treatments must not exceed 80 µs.

VI. DSP'S RESULTS

Presented results correspond to an UWA communication system in shallow water. The distance between transmitter and receiver was about 900m and the depth around 30m.

The symbols were transmitted using QPSK modulation with a carrier frequency of 62 kHz and a bit rate of 25 kbps. DSP processing was carried out involving 1, 2 and 4 sensors.

The demodulated signal was over sampled at 200 kHz and then recorded. For that reason, digital demodulation was not implemented. Only low-pass filtering, timing and carrier recovery, and equalization were implemented.

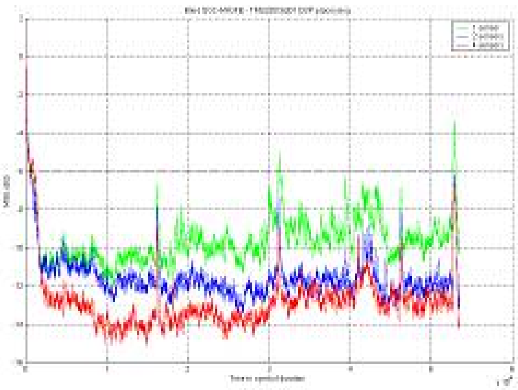

Fig. 9. MSE of the SOC-MI-DFE for 1, 2 and 4 sensors.

The different parameters of the SOC-MI-DFE equalizer are the following : the transversal filter B had 15 coefficients among which 2 were dedicated to the causal part. The recursive filter A had 55 coefficients. Threshold was fixed at J_0 = -6dB.

Fig. 9 shows the evolution of the estimated mean square error (MSE) of the SOC-MI-DFE for 1 , 2 , 4 sensors and 53500 symbols. In this example, the gain brought by use of 4 sensors can reach near 4 dB compared with 1 sensor.

From Fig. 10 depicts constellations taken between iterations 25000 and 25500 of the different signals processed by the DSP. The first graph shows input constellation. The second one shows the output of the single input equalizer. From the third constellation, one can already note a significant improvement with the contribution of a second sensor. The fourth constellation clearly highlight the gain brought by use of 4 sensors.

The first estimations give a whole processing time of about 40 µs, in the 4-sensors case. About 30 µs are required to achieve equalization. While typically 10 µs is required for low-pass filtering. The margin is evaluated to 40 µs, which allows a digital demodulation to be further implemented.

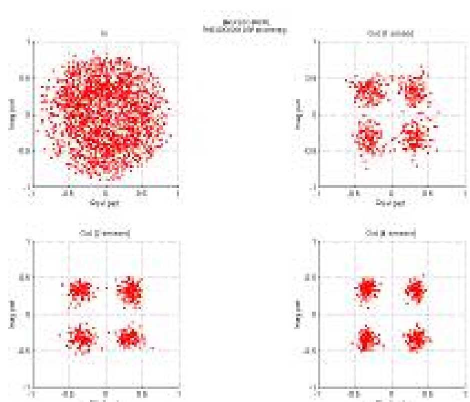

Fig. 10. Input and output constellations of the SOC-MI-DFE for 1, 2 and 4 sensors with DSP processing.

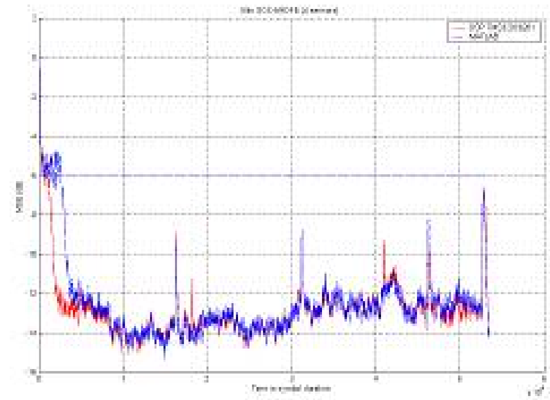

Fig. 11. Comparison between the DSP's results and a 32-bits floating point simulation.

Finally, DSP results are compared with a 32-bits floating point simulation (Matlab) for the purpose of determining the influence of the 16-bits fixed point DSP representation. As depicted on Fig. 11 (Matlab simulation in blue and DSP processing in red), the results in term of estimated MSE are equivalent in practice. In some cases, the lost of precision due to the 16-bits fixed point representation would seem to improve the speed of convergence.

VII. CONCLUSION AND PERSPECTIVES

This paper presents the current state of the GESMA project related to designing an high data rate acoustic link dedicated to the future mines hunting AUVs.

After some brief recalls on multidimensional processing (spatio temporal equalization, beamforming), we focus on a recently introduced blind multiple input equalizer and its implementation on a TMS320C6201 DSP.

First results allows to conclude to the feasibility of transmission systems at bit rates higher than 25 kbps. Integration and first sea trials will constitute the next stage of this project.

REFERENCES

[1] J. Labat and C. Laot, "Blind adaptive multiple input decision feedback equalizer with a self optimized configuration", IEEE trans one Comm, Flight 49, N°4, April 2001.

[2] D. Albonico; F. Fohanno; J. Labat, ' Test of an high date rate acoustic link in shallow water', Proc. Oceans' 98, Nice, France, pp. 1028-1032, 1998.

[3] G. Lapierre, J. Labat, J. Trubuil, ' Evaluation of high data rate acoustic link: contribution of blind spatio-temporal equalization', Proc. ECUA' 2000, Lyon, France, Vol.1, pp. 549-554.

[4] A. Glavieux, J. Labat, 'Transmission d'mages sur canal acoustique sous-marin', GRETSI-NICE, pp. 161-176, 1-5 juin 1987.

[5] G. Ayela, JM. Coudeville, " TIVA : A long range, high baud rate image/data acoustic transmission system for underwater applications ", Proc. Udt, Paris, France 1991.

[6] S. Dunn, J. Davis, S. Pointer, " A real-time high data rate acoustic communications receiver demonstration system ", Proc. In Oceans' 99.

[7] M. Stojanovic, J. Catipovic, J. Proakis, " Phase-coherent digital communications for underwater acoustic channels ", IEEE of Oceanic Engineering, Vol.19, N°1, January 1994.

[8] M. Stojanovic, " Recent advances in high speed underwater acoustic communications ", IEEE of Oceanic Engineering, vol. 21, N°2, April 1996.

[9] D. Kilfoyle, A. Baggeroer, " The state of the art in underwater acoustic telemetry ", ", IEEE of Oceanic Engineering, Vol.25, N°1, January 2000.

[10] T. Le Gall, " Extraction of the complex envelope of a 4-PSK. Influence on the adaptive equalization performances. Application to the UWA channel ", CNAM report, October 2000.

[11] F. Gardner, " A BPSK, QPSK timing-error detector for sampled receivers ", Trans IEEE. one Comm., Flight COM-34, N°5, May 1986.

[12] B. Sharif and Al, " Closed loop Doppler alignment and compensation for non-stationary underwater platforms ", Proc Oceans' 99.

[13] D. Billon, B. Quellec, " Performances of high data rate acoustic underwater adaptive communication systems using beamforming and equalizing ", Proc. OCEANS' 94, Vol3, pp. 507-512, 1994.

[14] P. Beaujean, L Leblanc, " Spatio-temporal processing of coherent acoustic communications data in shallow water ", Proc. Oceans' 99.

[15] M. Stojanovic, J. Proakis, J. Catipovic, " Analysis of the impact of channel errors estimate there the performances of a decision feedback equalizer in fading multipath channels ", IEEE Trans. one Comm. Vol.43, N°2/3/4. Feb, March, April 1995.

[16] S. Ariyavisitakul, " A decision feedback equalizer with time-reversal structure ", IEEE Newspaper one selected areas in Comm. Vol.10, N°3, April 1992.

[17] M. Kovic, D. Brady, M. Stojanovic, " Digital Sparse equalization for real-time underwater acoustic communications ", in Proc. OCEANS' 95, San Diego, CA, Oct. 1995, pp. 1417-1422.

[18] M. Stojanovic, J. Catipovic, J. Proakis, " Adaptive multi-channel combining and equalization for underwater acoustic communications ", J Acoust. Soc.. Amer., Vol. 94, pp. 1621-1631, 1993.

[19] M. Stojanovic, Z. Zonar, " Multichannel processing of broadband multiuser communication signals in shallow water acoustic channels ", IEEE Oceanic Engineering, Flight 21, N°2, April 1996.

[20] B. Welsh, J. Redi, S. Pointer, J. Davies, " Advances in efficient mobile submarine acoustic networks ", International Unmanned Undersea Vehicle Symposium, April 24-28,2000.

[21] J. Labat, " Real time underwater communications ", Proc. in Oceans' 94,Brest, France 1994. vol3,pp. 501-506.

[22] V. Capellano, G. Jourdain, “ Adaptive comparison of algorithms for multi-channel adaptive equalizers. Application to underwater acoustic communications “, Proc. Oceans’98, Nice, France.

[23] TMS320C62x/C67x CPU and Instructions Set Reference Guide Texas Instruments, SPRU189C, March 1998.

[24] Code Composer Studio (CCS) User’s Guide, Texas Instruments, SPRU328B, February 2000.

[25] TMS320C6x Optimizing C compiler User’s Guide Texas Instruments, SPRU187C, February 1998.