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ADAPTIVE MMSE TURBO EQUALIZATION USING HIGH ORDER MODULATION: EXPERIMENTAL RESULTS ON UNDERWATER ACOUSTIC CHANNEL

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Abstract: *This paper presents some results on adaptive minimum mean-square error (MMSE) turbo equalization obtained from underwater experiments in the Atlantic ocean. Single-carrier transmission with high-order modulations is considered (8-PSK, 16-QAM, 32-QAM and 64-QAM), at a coded bit rate as high as 28800 bps on the underwater acoustic channel. The all-digital receiver performs timing recovery, equalization, interleaving and channel decoding. The results are evaluated in term of bit error rate (BER) and mean square error (MSE). Turbo equalization improves the performance of the receiver and allows reliable high order transmissions when the signal to noise ratio is sufficient.*

Keywords: *underwater acoustic communications, high order modulation, turbo equalization*

1. INTRODUCTION

This paper presents a high data rate acoustic link between two boats in motion. The proposed receiver is based on the TRIDENT receiver [1], developed by GESMA (Groupe d'Etudes Sous-Marines de l'Atlantique, Brest, France), in collaboration with Telecom Bretagne and SERCEL. This receiver was designed for text, images and speech data transmission in a shallow water environment. Initially designed for QPSK, the TRIDENT receiver has been extended in this paper to high-order modulations (8-PSK, 16-QAM, 32-QAM and 64-QAM). This results in the transmission of coded bit rates as high as 28800 bps over distances greater than 1 km.

To maximize the spectral efficiency of the link, a single-carrier modulation is used. Data transmission is organized into long bursts. In contrast to OFDM systems, this scheme avoids the spectral efficiency loss due to the insertion of a guard interval or a cyclic prefix. The proposed receiver must be able to deal with time- and frequency-selective channels. Therefore we use efficient synchronization schemes and an adaptive multiple-input equalizer. In addition, channel coding is used to increase the robustness of the transmission.

In this paper, we evaluate the performance of an adaptive MMSE turbo equalizer. Minimum mean square error (MMSE) turbo equalization [2-3] has proven to be effective for removing intersymbol interference. The equalizer and the channel decoder exchange soft information in an iterative process. It is shown that, as expected, the performance improves when the number of iteration increases especially when the channel is very frequency selective. Experimental results are given on the performance of the turbo equalizer using high-order modulations.

2. TRANSMISSION MODEL

The transmission scheme is depicted in Fig. 1. A convolutional code is fed by binary data. An interleaver shuffles the coded bits. Each set of $m = \log_2(M)$ interleaved coded bits is mapped onto an M -ary complex symbol d_k taken from an M -PSK or M -QAM signal set, using Gray or quasi-Gray labelling. Data transmission is organized into long bursts of several seconds. Each burst is made of an initial preamble used for frame detection and synchronization, followed by several fixed-size data blocks separated by pilot sequences. A transducer transmits the modulated signal on the time- and frequency-selective underwater channel. The receiver is equipped with a chain of $N_R = 4$ hydrophones spaced 25cm apart. An all-digital single-carrier receiver is used, relying on a multiple-input adaptive MMSE turbo equalization scheme.

Let $s(t)$ be the transmitted waveform, $s(t) = \Re e \left\{ \sum_{n=-\infty}^{+\infty} d_k g(t - kT) e^{j(2\pi f_c t + \psi)} \right\}$, where f_c is the carrier frequency, ψ is the carrier phase uncertainty, $1/T$ is the symbol rate with T the symbol duration, d_k are the transmitted symbols with variance σ_d^2 and $g(t)$ is a square-root raised-cosine filter with roll-off factor 0.25.

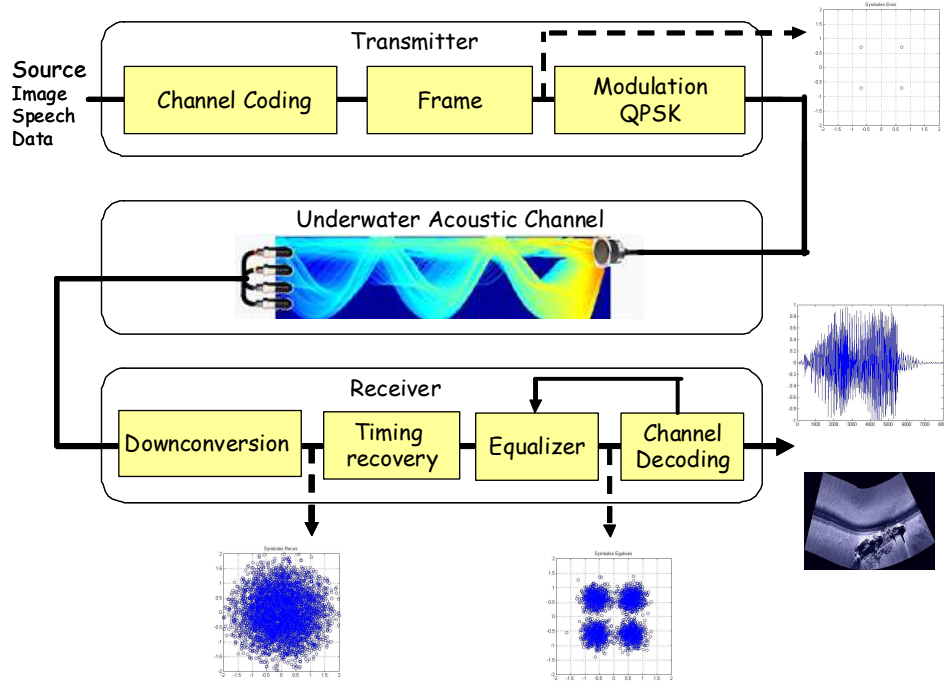


Fig. 1: Single-carrier transmission scheme overview

When the signal is centred on a relatively low carrier frequency ($f_c = 17.5$ kHz in our experiments - see Section 4), an all-digital receiver is feasible [4]. Oversampling is performed at the rate $1/T_s$ where T_s is chosen so as to satisfy the sampling theorem. In this paper, we have chosen $T = 20T_s$. The down-conversion is performed digitally, and a timing synchronization scheme based on a sample rate converter is used to determine to optimum sampling epochs. The resulting all-digital receiver is depicted in Fig. 2.

In wide-band transmission, as it is the case in underwater acoustic communications, the Doppler effect introduces a scaling of the symbol period which must be taken into account in the design of the timing recovery scheme [5]. The optimum sample time not only depends on the propagation delay at hydrophone j ; $j = 1, \dots, N_R$ but also on a common Doppler shift depending on the relative speed of the boats and the propagation wave velocity [5]. Because the receiver is all-digital, the optimum sampling time $kT + \tau_k^{(j)}$ is not necessarily a multiple of T_s , a sample rate conversion based on interpolation, filtering and decimation is then required [4]. The optimum sampling time is unknown and must be estimated. Initial compensation of the common Doppler shift is performed by using the short preamble inserted at the beginning of the transmission to estimate the relative velocity. Note that this preamble is also used to perform frame detection and synchronization. Then, a non-data-aided (NDA) timing recovery scheme is designed which takes into account the residual Doppler shift due to the moving platforms and the different channel delays at each antenna.

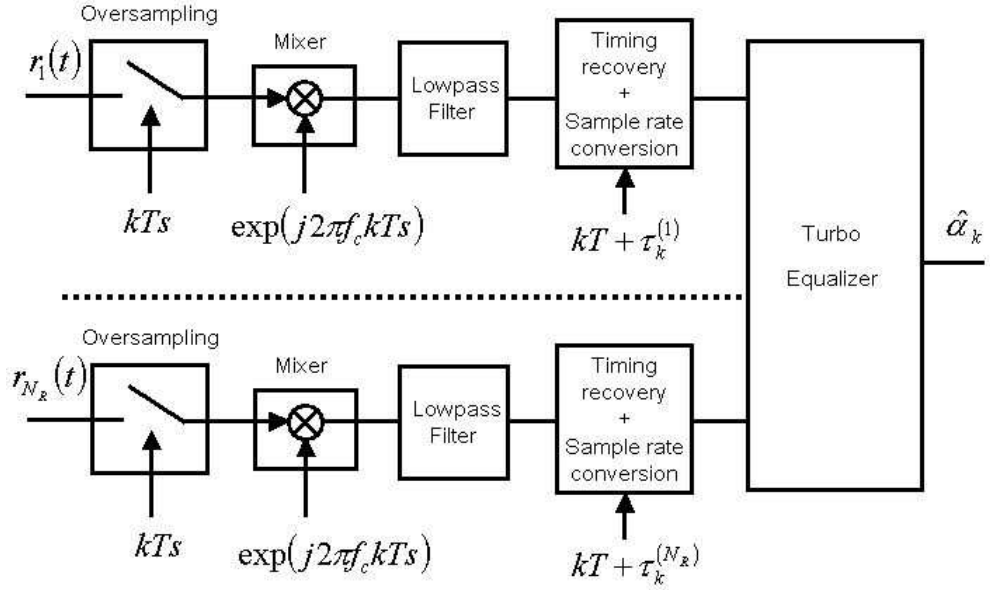


Fig. 2: Structure of the all-digital receiver

3. TURBO EQUALIZATION PRINCIPLE

The adaptive turbo equalizer is depicted in Fig. 3. Equalization and channel decoding exchange soft information in an iterative manner. Each iteration consists of a multiple-input equalizer, a soft-input soft-output (SISO) demapper, a deinterleaver Π^{-1} , a binary SISO channel decoder, an interleaver Π and a SISO mapper. The equalizer is fed in by the received signal samples $r_k^{(j)}$ and also by the estimated data \bar{d}_k obtained from the previous iteration. The channel decoder produces soft decisions on the coded bits, which are used in turn by the SISO mapper to compute the soft symbol estimates \bar{d}_k to be used by the equalizer in the next iteration.

The multiple-input equalizer combines the outputs of the feedforward transversal filters fed by the signals received from the hydrophones. Second-order phase-lock loops (PLLs) are optimized jointly with the equalizer filters in order to compensate for the residual frequency offsets. When *a priori* information is available from the channel decoder at the previous iteration, a feedback filter fed in by the estimated symbols \bar{d}_k is used to suppress the interference at the combiner output.

An adaptive least mean square (LMS) algorithm is used to obtain the filter coefficients allowing tracking of the channel time variations. The adaptive algorithm is composed of two distinct phases: the training phase and the tracking phase. The training phase makes use of pilot sequences known to the receiver (data-aided (DA)) to initialize the equalizer coefficients. Next, during the tracking period, the coefficients are continuously updated in a decision-directed (DD) manner, based on the receiver decisions on the transmitted symbols.

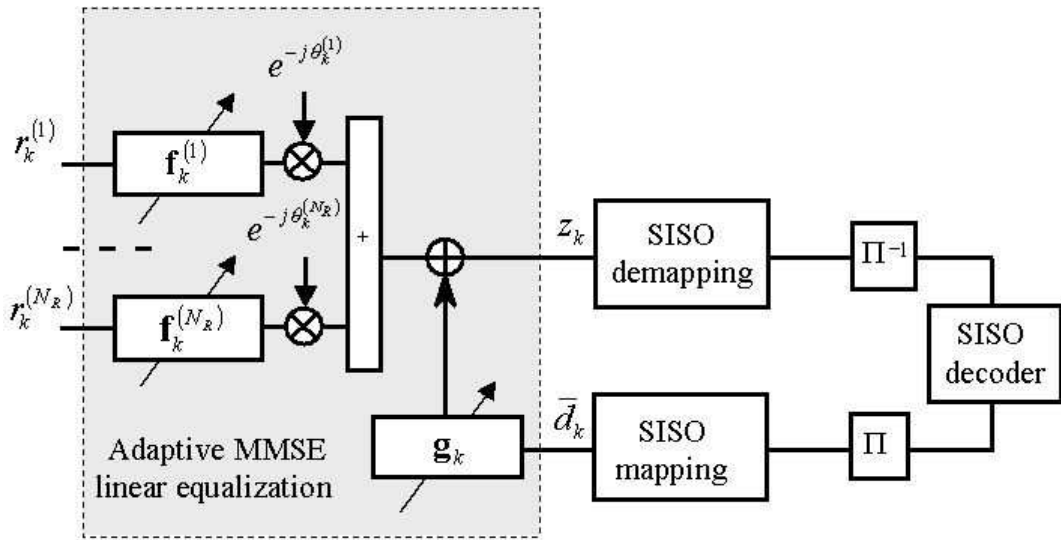


Fig. 3: Turbo-equalization scheme

This paper focusing on experimental results, additional information on the SISO mapping, the SISO demapping and the adaptive LMS equalizer used in this paper are available in previous references [2][6].

4. EXPERIMENTALS RESULTS

Our simulations emphasize the performance of the MMSE turbo equalizer over real sea trials. Experimental sea trials were carried out on March 2010 in the site "bay of Brest", France, by DGA/GESMA. The aim of these sea trials was to transmit data from a ship to another boat in a shallow water environment with a water depth of 10 to 30 meters. These trials were carried out in a context of rough sea. As depicted in the Fig. 4, the transmitter was placed on the board of the "Aventurière II" and the receiver on board of the boat "Idaco". At the receiver side, the antenna array was a vertical chain of $N_R = 4$ hydrophones spaced with 25 cm.

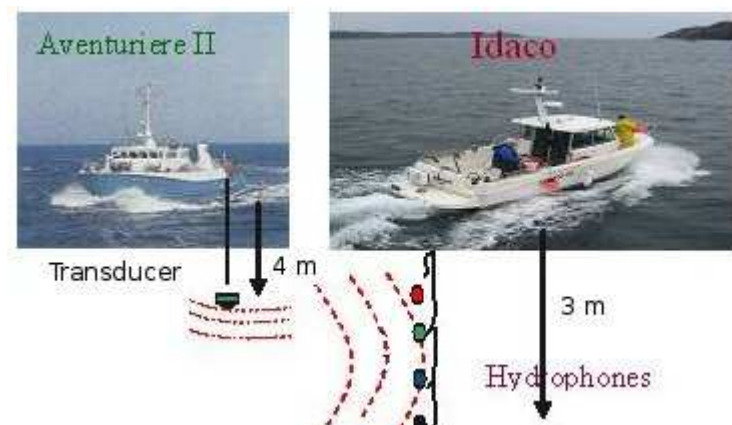


Fig. 4: Sea trial configuration

During these trials, two particular sequences were recorded (AIT44 and AIT63) with the objective to test the turbo-equalization with high order modulations. Each recorded sequence included consecutive bursts of symbols (8PSK, 16QAM, 32QAM and 64-QAM) separated by a guard interval of 5 seconds. Single carrier transmissions with a carrier frequency of 17.5 kHz were used. The symbol rate was 4800 symbols per seconds. The features of the two recorded sequences are given in Table I.

To build a burst of symbols, a rate $R_c = 1/2$ convolutional code with constraint length 5 was fed by a block of binary data. An interleaver shuffled 22500 coded data. Each set of $m = \log_2(M)$ interleaved coded data was associated with M -ary complex symbol. A transmitted burst resulted on the concatenation of one preamble of 1 second followed by a repetition of m blocks of $22500/m$ symbols separated by a pilot sequence of 256 symbols. The burst duration was 25 seconds. The ratio between the user bit rate and the channel bit rate was approximately 0.45.

In order to emphasize on the improvements with the proposed turbo equalizer, we show the behavior of the receiver in terms of decision-directed minimum mean square error (DD-MSE) at the equalizer output versus the iteration number over duration of 25 s. The

DD-MSE is estimate by $DDMSE_{k+1} = \lambda DDMSE_k + (1 - \lambda) |z_k - \hat{d}_{k-\Delta}|^2$ where $\lambda = 0.99$.

Modulation type	8-PSK	16-QAM	32-QAM	64-QAM
Channel Bit Rate (bits/s)	14400	19200	24000	28800
User Bit Rate (bits/s)	6840	8640	10800	12960

Table I : Features of the real sea trials

In Table II, we give the bit error rate (BER) at the equalizer and channel decoder outputs versus the iteration. The BER was computed on duration of 25 seconds. The improvement of the BER is rather weak and only visible for 32-QAM and 64-QAM.

Sequence AIT63	8-PSK	16-QAM	32-QAM	64-QAM
BER equalizer output (1 st iter)	3.7e-4	1.1e-3	1.3e-2	1.8e-1
BER decoder output (1 st iter)	0	0	1.77e-4	2.3e-1
BER equalizer output (3 rd iter)	3.7e-4	1.1e-3	1.0e-2	1.2e-1
BER decoder output (3 rd iter)	0	0	8.53e-5	1.5e-1

Table II : Features of the real sea trials

In Fig. 5, we have plotted the DD-MSE at the output of the equalizer versus iteration and the constellations for 1 second duration (4800 symbols) from the 5th to the 6th second. Constellations shown that receiver detects the 64-QAM modulation. Nevertheless, the SNR is rather weak and the channel decoder runs in a SNR range where the BER increases instead of decrease. The performance gain between the first and the third iteration of the turbo equalizer is weak because the intersymbol interference on this transmission is not significant. In Fig. 6 is depicted the channel state information (CSI) after Doppler shift compensation. Performing the characterization of the frequency response, it can be shown that frequency selectivity is weak with fading lower than 5 dB. Because it is well-known

that the turbo equalization is essentially efficient when the frequency selectivity is high, the performance results obtained for these real sea experiments are in accordance.

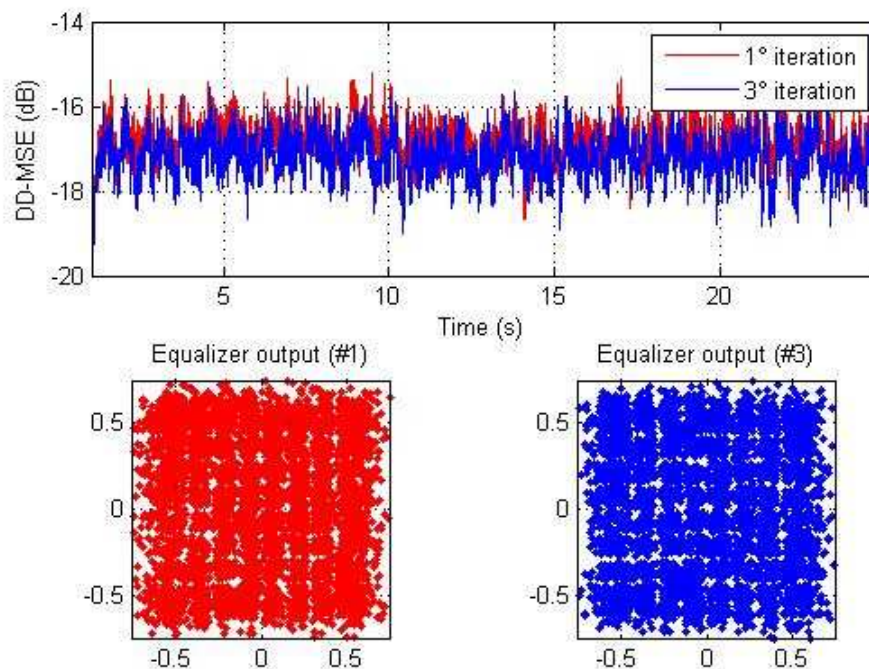


Fig. Sequence AIT63: 64-QAM, 4800 symbols/s, $f_c = 17.5$ kHz, distance=640m, $v=1.7$ m/s

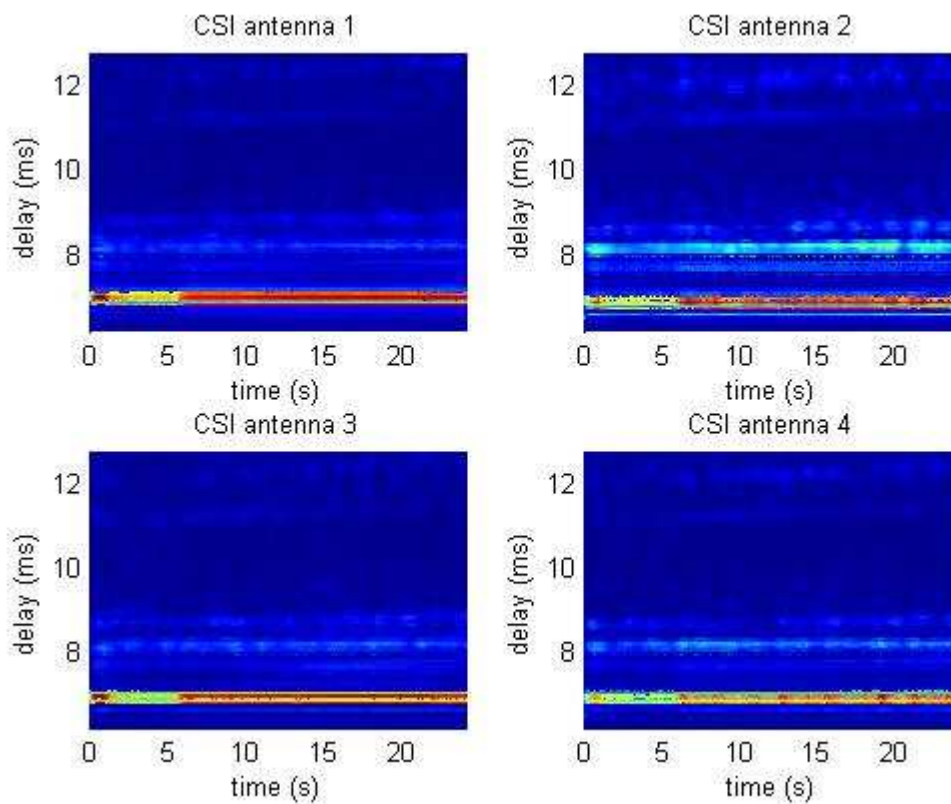


Fig. Sequence AIT63: channel state information, 4800 symbols/s, $f_c = 17.5$ kHz, distance=640m

5. CONCLUSIONS

Single-carrier underwater acoustic transmissions based on high-order modulations and adaptive MMSE turbo equalization have been successfully demonstrated in real conditions, with user data rates greater than 10 kbps. Unfortunately, from the database of real recorded signals, the performance improvement obtained by the turbo equalization process have not been emphasized because of a low frequency selectivity of the channel. We hope that the recorded signals in a future campaign of measure will be more corrupted with strong intersymbol interference in order to prove the efficiency of the turbo equalization.

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