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Improvement of Underwater Acoustic Communication using Reed Solomon Block Turbo Codes

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Abstract— TRIDENT project was launched a few years ago by GESMA (Groupe d'Etudes Sous-Marines de l'Atlantique) in collaboration with TELECOM Bretagne. First objectives were to develop a multiple rate underwater acoustic link for images, speech signal, text and data transmission. This link was designed to provide a wireless communication to AUVs. Recently the platform was extended to low bit rate speech transmission using a MELP coder working at 2400 bps. Such a bit rate allows the use a lower carrier frequency able to reach longer ranges for speech transmissions. More recently two channel coding options were added to the TRIDENT platform. Convolutional codes (CC) and Reed Solomon (RS) block codes were only checked. These simple codes were not able to improve significantly the BER at the channel decoding output. That is why, last year GESMA decided to check the turbo codes options and more precisely the Reed Solomon Block Turbo Codes (RS BTC) to enhance the channel decoding efficiency at the receiver side. This paper presents the different possibilities offered by the platform: images and speech transmissions and the different channel coding options. The data transmitted are interleaved and frame recovery is performed in reception. After system validation in Brest River (November 2009), sea trials have been recently conducted in Brest bay (March 22 to 26, 2010).

Keywords—Underwater acoustic communication, BER (Bit Error Rate), speech coding, synchronization, Reed Solomon Block Turbo Codes (RS BTC)

I. INTRODUCTION

The first objectives of TRIDENT project (TRansmission d'Images et de Données EN Temps réel) were to equip Autonomous Underwater Vehicles (AUVs) with acoustic communications. This system had to be able to transmit continuous information such images, speech signal and text. The TRIDENT system [1] can use four carrier frequencies (11.2, 17.5, 20.0 and 34 kHz) with a QPSK modulation (Quadrature Phase Shift Keying) and different bit rates, ranging from 2.8 to 23.3 kbps. Multipath propagation, noise Doppler effect and also impulse noise bring a lot of perturbation on underwater acoustic communication, such as time dispersion and a variability to the received signal. It is to note that carrier frequencies and available bandwidths are much lower than communication channels ones. A blind

spatio-temporal equalizer [2] is used to reduce these various perturbations. GESMA also aimed to increase the link's reliability with the objective to decrease the BER from 10^{-2} to 10^{-4} . In order to do so, a channel coding was integrated to the system. Two kinds of error correcting schemes had been tested including Convolutional Codes (CC) and Reed Solomon (RS) block code [3][4]. These two channel coding schemes have low code rates (CC: 0.5 and RS: 0.7) and do not bring us enough correction. To improve the symbol correction and obtain a higher code rate, Reed Solomon Block Turbo Codes (RS BTC) were recently tested, in static and in moving conditions. The iterative decoding process based on the Chase-Pyndiah [5][6] algorithm uses a soft version with four to eight iterations. Berlekamp [7] and Chien algebraic algorithms are also used to correct received symbols. A differential coding/decoding soft option is used to solve the phase ambiguity. Three different synchronization words are wisely included in the coded frame to synchronize the interleaver and recover the information frame at the receiver.

This paper provides an overview of the high data rate acoustic link. Firstly, we present the TRIDENT platform and its RS Block Turbo Codes extension. Strategies for synchronizing interleaver, channel decoder and solving the phase ambiguities are also described. Then, we present the sea trials and show the first acoustic communication results in the 2.8 to 14 kbps range, applied both to images and low bit rate speech (2400 bps) transmissions.

II. ITERATIVE DECODING OF REED SOLOMON BLOCK TURBO CODES

The underwater acoustic channel is characterized mainly by strong fading, multipath noise, Doppler effect and also impulse noise. As part of the wireless communication system TRIDENT, the Quadrature Phase-Shift Keying (QPSK) modulation is usually the preferred technique to transmit information. It is used, in reception, with a blind spatio-temporal equalizer. This equalizer uses input signals sampled on several sensors (4 in this version) only coming from a single emission source. This space diversity improves the Signal to Noise Ratio (SNR) compared to the mono-sensor

version. This equalizer runs according to a convergence or starting mode and a tracking mode, which are differentiated both on the structure and the algorithm. The interest of this adaptivity lies in the possibility of switching from one structure to another according to the channel severity. Moreover preamble or other training sequence become unnecessary, because only user data are transmitted. This approach seems particularly appropriate for implementation simplicity, interoperability and better spectral efficiency. By using a higher code-rate, the bit rate user becomes higher in the case of fixed bit rate transmission. To improve the underwater acoustic link performance and obtain a higher code rate, Reed Solomon Block Turbo Codes (RS BTC) were tested.

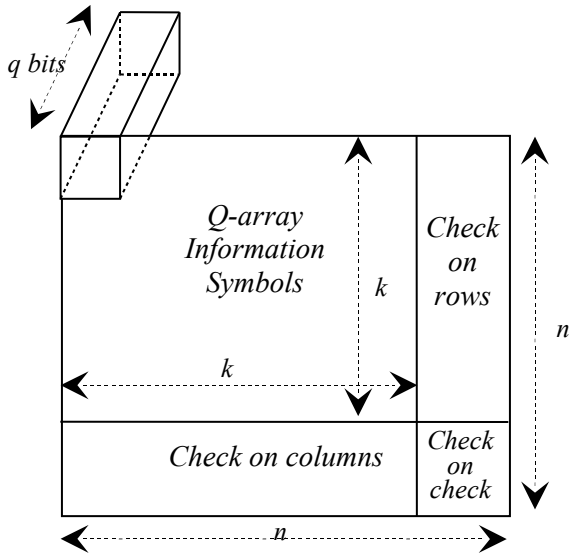


Figure 1. RS product codes with Q-ary symbol concatenation

The concept of product code introduced by P. Elias in 1954 is a single and efficient method to construct powerful codes. Product codes are obtained by means of serial concatenation of two (or more) linear block codes, e_1 having parameters (n_1, k_1, δ_1) and e_2 having parameters (n_2, k_2, δ_2) . Parameters n_i, k_i and δ_i ($i=1,2$) stand respectively for code length, code dimension and Minimum Hamming Distance (MHD) of each code. The parameters of the constructed product code P are given by $n = n_1 \times n_2, k = k_1 \times k_2, \delta = \delta_1 \times \delta_2$. The code rate of P is $R = R_1 \times R_2$ where R_i is the code rate of its component codes. It is shown that all n_1 rows of the matrix are code words of e_1 just as all n_2 columns are code words of e_2 by construction. RS codes are a subclass of non-binary BCH codes. $RS(n, k, \delta)$ is a Maximum Distance Separable (MDS) code which has the minimal code length for a given code rate and error correcting capability. RS product codes can be constructed based on either Q-ary symbol concatenation or bit concatenation. In order to get smaller block size, Q-ary symbol based RS product code is used

here. It is constructed from $(k_1 \times k_2)$ Q-ary ($Q=2^q$) information symbols, which means $(k_1 \times k_2 \times q)$ data bits. The q binary elements corresponding to a Q-ary symbol are at the intersection of a row and a column. Serial concatenation is realized by coding Q-ary symbols along the rows and columns successively. Our RS product code P is constructed by using two identical component codes $e_1 = e_2 = RS(n, k, \delta)$ and is shown in figure 1.

By using a Galois Field (GF) generator polynomial, the above RS product code can be represented by a matrix of n rows and $(n \times q)$ columns after the Q-ary to binary conversion. The transmission with QPSK modulation over Additive White Gaussian Noise (AWGN) channel can be regarded as the superposition of two independent Binary Phase-Shift Keying (BPSK) modulations applied respectively to the in-phase and orthogonal carrier. Each bit of the RS product code is associated with one binary value according to the mapping rule ($0 \rightarrow -1, 1 \rightarrow +1$). At the channel output, the bit level Log-Likelihood Ratio (LLR) is computed and fed to the following turbo decoder.

The turbo decoder is made up of several cascaded Soft-Input-Soft-Output (SISO) decoder. Each SISO decoder is used for decoding rows and columns. A SISO decoder can be divided into Soft-Input-Hard-Output (SIHO) decoder and a reliability estimation of each decoded bit at the output of the SIHO decoder. The Chase's algorithm [7] is extended to construct the SIHO decoder for $RS(n, k, \delta)$ over $GF(Q)$. Let the vector $R = (r_1, r_2, \dots, r_i, \dots, r_{qn})$ be the normalized bit level LLR sequence after soft demodulation for one received word and $Y = (y_1, y_2, \dots, y_i, \dots, y_{qn})$ be the hard decision estimation of the transmitted sequence following the equation 1.

$$y_i = 0.5 \times [\text{sign}(r_i) + 1] \text{ for } 1 \leq i \leq (q \times n) \quad (1)$$

Firstly the s least reliable positions corresponding to the smallest absolute values in vector R are determined. Test patterns of length qn of weight "0" to s with the "1" restricted to the least reliable positions are added to the sequence Y to obtain new binary sequences. A Q-ary sequence $H = (h_1, h_2, \dots, h_i, \dots, h_n)$ is then obtained based on new binary sequence. The algebraic RS decoder processes these Q-ary sequences and produces the Maximum-Likelihood (ML) RS codewords. These ML codewords are then represented by binary sequence of length qn after the Q-ary to binary decomposition and noted C^i . The Chase algorithm computes the Euclidian Distance (ED) to the input R and selects codeword D at minimum ED from R . For each bit d_j ($1 \leq j \leq qn$) of D , we search for the concurrent code word C at minimum ED from R such that $c_j \neq d_j$ at positions j . If such codeword is available, the soft output for bit d_j is given by (2).

$$r_j = \left(\frac{M^C - M^D}{4} \right) \times d_j \quad (2)$$

MD is ED between D and R, and MC is ED between R and C. Else we use the predefined value β and process the vector R elements (equ. 3).

$$r'_j = r_j + \beta d_j \quad (3)$$

The extrinsic information W which is the difference between soft output and soft input for each component is computed and sent to the next SISO decoder. Input to the next SISO decoder is given by $R + \alpha W$ where α is used to reduce the influence of extrinsic information in the first iteration.

The complexity of this SISO decoder depends on parameter s . Given that the soft decoding criterion is to find the RS binary codeword of length qn at minimum ED from observation vector R , then s is given by $\delta_b/2$ where δ_b is the minimum Hamming distance of RS binary codewords. In the case of single-error correcting codes, $\delta_b = \delta = 3$ and thus $s = 1$ is sufficient for SIHO decoding.

III. SPEECH MELP CODER/DECODER

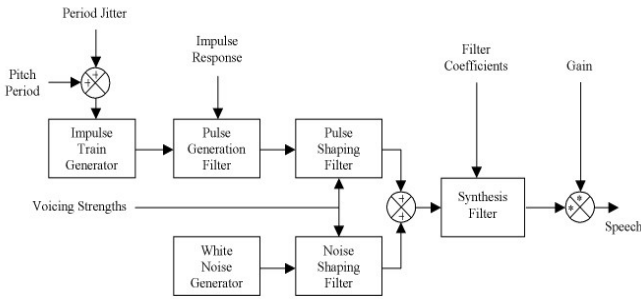


Figure 2. The MELP model of speech production

The Mixed Excitation Linear Prediction (MELP) (Figure 2) voice coder model is based on the traditional Linear Prediction Coefficient (LPC) vocoder. However, the synthesizer has additional abilities that allow MELP to mimic more of the characteristics of natural human speech.

The coder employs Fourier magnitude coding of the prediction residual to improve the speech quality, as well as vector quantization techniques to efficiently encode the LPC and Fourier information. The speech is segmented in frame having a length of 22.5 ms and the sampling rate is set to 8 kHz. Each speech sample is 16 bits-quantized giving a

number of 2560 bits per frame at the MELP coder input. After a compression factor close to 53, this bit rate is down reduced to 54 bits per frame, that is to say 2.4 kbps. Synchronization bits are added to the bit stream to find back the speech frame at the decoder.

IV. FRAME SYNCHRONIZATION AND DIFFERENTIAL CODING

Information transmitted is generally framed. In reception, synchronization words are conventionally used to synchronize both the source and channel decoders. Interleaver needs another synchronization. Several level of synchronization word lowers the whole correcting rate but makes it more flexible design of different transmission system stage.

In the TRIDENT system, with the RS BTC channel coding, three different synchronization words S_i ($i \in \{1, 2, 3\}$) are included to retrieve the frame, at the receiver. S_1 (respectively S_2 or S_3) is used to detect the frame start (respectively middle or end frame). The emitted coded frame, with S_i in (respectively out) the frame, is 4805 bits in length (respectively 5084), thus the whole correcting rate is $R = 0.82$. As we intrinsically have a synchronization words, it is likely that the differential coding, used to correct the phase ambiguity, becomes useless. Let us recall that differential coding leads to a 3 dB loss in terms of coding gain. Having a look to the frame length, the differential coding is required. Indeed, a phase jump at the frame start can generate too many errors and the expectation of a new detection is too long. Soft decoding use the Log-Likelihood Ration (DC LLR). To improve the performance of the TRIDENT acoustic link, Reed Solomon Block Turbo Codes are under consideration. Channel coding goal is to lower the bit error rate from 10^{-2} to 10^{-4} and improve images or speech synthesis quality. After equalization and synchronization efficiently realized, channel coding may add extra performance.

V. RESULTS

A. Simulations

The first results presented are obtained by simulations in presence of Additive White Gaussian Noise. Figure 3 presents performance obtained with Reed Solomon Block Turbo Codes. It clearly appears that differential coding (DC) decreases performance by doubling the bit error rate. We take the BER at the output of the channel as a reference for the 3 options, without DC and with DC Hard or LLR. When using DC hard or soft decoding with 4 iterations, we can not lower the BER from 10^{-2} to 10^{-4} . This performance is obtained with DC LLR after 8 iterations. Without DC, we obtain the performance wished. Let us recall that in the case of phase ambiguity, the decoding process may not work correctly. The use of synchronization words S_i is able to

solve the phase ambiguities. Several phase jumps can appear during the frame transmission. Thus, it is better to use differential encoding with LLR decoding.

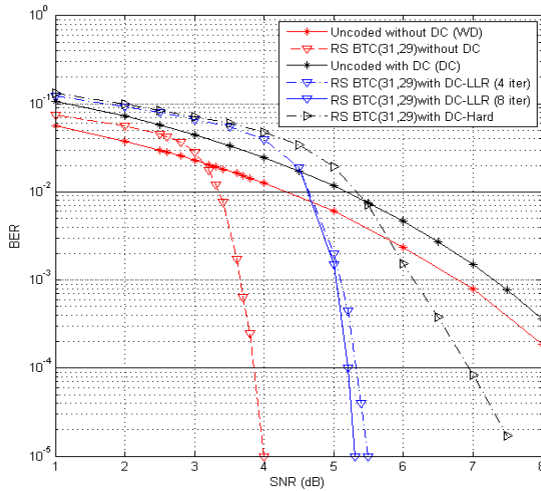


Figure 3. RS BTC with and without differential decoding on Gaussian Channel

Figure 4 presents an image transmission on AWGN channel with a SNR at 5.5 dB. The BER at the channel decoder input is $0.8 \cdot 10^{-2}$ and $1.7 \cdot 10^{-5}$ at the output. Performance are obtained with a differential coding using LLR. We can see that with one iteration of decoding channel, residual errors can not reconstruct the image correctly. After 4 iterations, the few remaining errors do not degrade the image quality. Soft decoding options DC-LLR answer the specification after 8 iterations, with of course an extra computational load. At the BER of 10^{-4} , a loss of 1.3 dB appears when using DC-LLR.

B. River trials

Trials were carried out in November 2009 in Penfeld Brest river to validate and evaluate the TRIDENT system performance over horizontal shallow transmissions. Two carrier frequencies (17.5 or 34.0 kHz), were checked during this trials, with a data bit rate range from 5.8 to 17 kbps.

Transmitter-receiver distance was about 400 m and around 7 m depth. The emitter was positioned in a laboratory van on the quay side. The transducer was deployed at 5 m depth below a buoy. The receiver was on board the "Aventurière II". Acoustic signals were received on 4 sensors, interspaced close to 25 cm (around 5 wavelengths). First hydrophone was 3 meters depth (about 1 meter below the ship's draught).

During these trials, 100 sequences of 3 minutes were emitted and recorded, representing 5 hours. All the 100

experiments carried out during these trials concluded successfully, giving also very good results.

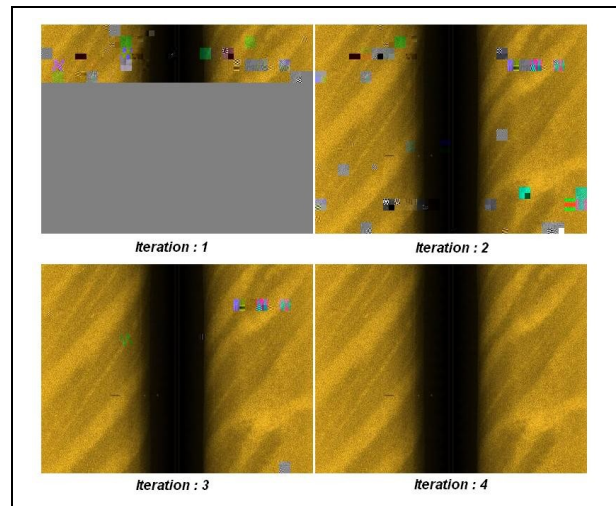


Figure 4. Images after channel decoding (1, 2, 3, 4 iterations)

C. Brest bay trials

Brest Bay trials (Fig. 5) took place from March 22 to 26, 2010. Brest bay is a shallow underwater acoustic channel (10-50 m depth). Two boats were used. The transmitter is on board the "Aventurière II". The transducer is at a 4 m depth. The receiver is on board the boat "Idaco". The 4 hydrophones are at 3 m depth. These trials were carried out in a context of rough sea.

During these trails, five hours and half of signal transmitted, were recorded representing 162 sequences of 2 minutes length. Both emitter and receiver were moving (less than 4 knots) with a ranges from 300 m up to 3000 m. Next results show the behavior of the receiver for a speech transmission with a 17.5 kHz carrier frequency at a bit rate fixed at 11.6 kbps. Transmitter and receiver speeds are around 1 knot.

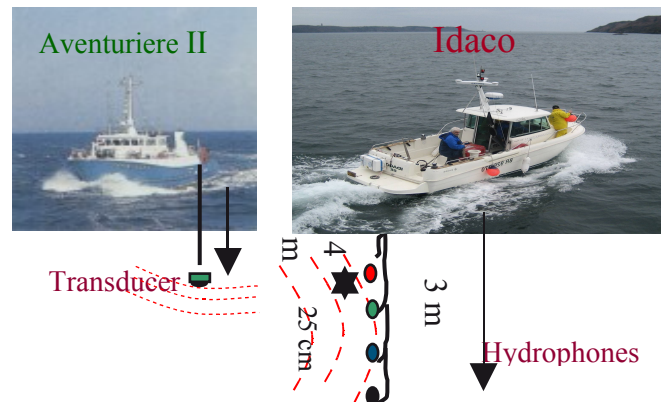


Figure 5. Sea trial configuration

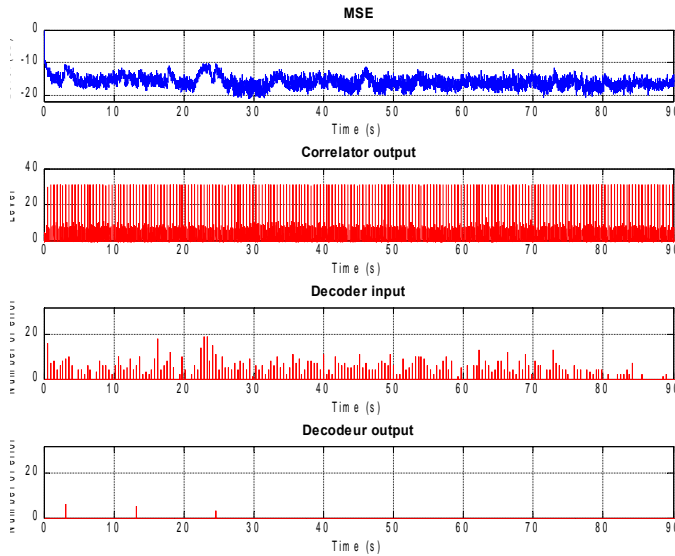


Figure 6. MSE, correlator output, decoder input and decoder output

In figure 6, the Mean Square Error (MSE) show the success of the equalization process. A very good overall behavior is to be noted during this sequence. It clearly appears that the transmission was processed without any kind of difficulties. In this trial, phase ambiguity was solved using a DC-LLR. After both phase ambiguity correction and decoder synchronization, the BER at the decoder input (BER_{in}) was $1.0 \cdot 10^{-3}$ and, $1.5 \cdot 10^{-5}$ at the decoder output. For a number of 227 frames transmitted, no frame was lost, 201 (88.5%) were detected false (one or more errors) at the channel decoder input and 3 (1.3%) at the decoder output. 8 iterations were processed for the channel decoding. Equalizer time was only 0.13 s. During the convergence stage, frames may be lost (only a very small lot of them during this trial). The speech signal was correctly decoded. The transmission bit rate was 11.6 kbps with 2.4 kbps used for the speech signal.

VI. CONCLUSION AND PERSPECTIVES

This paper presents the extension of the TRIDENT system, developed by GESMA and TELECOM Bretagne. This underwater acoustic link is able to transmit different kinds of data as text, images and speech signal. A blind spatio-temporal equalizer is used to reduce different shallow underwater acoustic effects. To improve the underwater acoustic link performance and obtain a higher code rate, Reed Solomon Block Turbo Codes (RS BTC) was tested. The goal was to decrease the BER from 10^{-2} to 10^{-4} . A differential coding has been used to solve the phase ambiguities. First sea trials of real time transmission of speech signal and images, using channel turbo coding, were carried out in the underwater acoustic channel, in stationary state, in Penfeld Brest river, (November, 2009). One hundred experiments were then carried out on this site with success.

In March 2010, sea trials were carried out in the Brest bay, in moving conditions and rough sea. More than 5 hours of transmitted signal were recorded. Only 20% of these experiments have been analyzed. The first results show a good behavior of the channel coding/decoding when synchronization and equalization work correctly.

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