Performance Comparison and Optimization of Channel Coding for Acoustic Communication in Shallow Waters

[Extended Abstract]

Ufuk Uensal, Steffen Moser and Frank Slomka
Institute of Embedded Systems/Real-Time Systems
Faculty of Engineering and Computer Science, Ulm University
Albert-Einstein-Allee 11, 89081 Ulm, Germany
ufuk.uensal@alumni.uni-ulm.de, {steffen.moser | frank.slomka}@uni-ulm.de

ABSTRACT

The coded communications using non-coherent orthogonal modulation and capacity-approaching binary channel codes namely low-density parity check code (LDPC) and turbo code are investigated in this paper with the focus on the main three characteristic effects of an underwater channel, namely, multi-path propagation, Doppler spread and ambient noise. Additionally, a new method was implemented and tested successfully to identify and eliminate the high-amplitude noise from the received dataset.

Categories and Subject Descriptors

E.4 [**Coding and information theory**]: Error control codes; C.4 [**Performance of systems**]: Performance attributes

General Terms

Algorithms, Design, Performance

Keywords

Underwater Acoustic Communication, Channel Coding, Turbo Code, LDPC Code, BER (Bit Error Rate), Multi-Carrier Communication, High-Amplitude Noise

1. INTRODUCTION

The low-power underwater modem developed by Woods Hole Oceanographic Institution (WHOI) offers a robust modulation scheme, namely, non-coherent frequency-shift keying with frequency hopping (FH-FSK) to allow operation under harsh conditions of the underwater acoustics channel[1].

The most recent, powerful, *near-Shannon-limit* channel coding techniques like *turbo codes* or *LDPC codes* can be used to achieve a significant performance improvement compared to classical convolution codes. In this paper we show how these methods increase the performance of the currently available WHOI micro modem implementation. We will also present a method to identify and eliminate the high-amplitude noise from the received dataset.

WUWNET '14 Nov 12-14 2014, Rome, Italy
ACM 978-1-4503-3277-4/14/11
http://dx.doi.org/10.1145/2671490.2674566 .

2. LLR COMPUTATION OF NON- COHERENT FSK DEMODULATOR

With a vector $u \in \{0,1\}^K$, a codeword $b' \in \{0,1\}^N$ is produced by a binary encoder, where K is the number of message bits, while N is the number of codeword bits. The M-ary orthogonal modulator produces the $\mathrm{M} \times \mathrm{L}$ matrix of symbols $S = [s_0^T, ..., s_{L-1}^T]$ where $L = [N/\log_2 M]$ is the number of symbols to be transmitted sequentially, while M is the number of orthogonal channels. Each column of S represents one M-ary symbol and is represented as an elementary vector $\mathrm{e_m}$ with only a single one in the $\mathrm{m^{th}}$ position. If the symbol s is transmitted, the first $\mu = \log_2 M$ bits in b are gathered to form the symbol $s \Leftrightarrow \{b_0, ..., b_{\mu-1}\}$. The energy per coded symbol ε_s is related to the energy per message bit ε_b by $\varepsilon_s = K\varepsilon_b/L$. The following symbols which are also the input to the non-coherent MFSK demodulator are passed to the receiver:

$$Y = [(s_0^T + n_0^T), ..., (s_{L-1}^T + n_{L-1}^T)]$$

For an arbitrary symbol s, y is the received signal vector. With received symbols Y and an estimate of the average symbol signal-to-noise ratio $\varepsilon_\mathrm{s}/\mathrm{N_o}$, MFSK demodulator computes soft information as log-likelihood ratio for each code bit b_k, as in Equ. 1 which is explained in detail in [3] where the set $S_k^{(1)}$ contains the indices of all symbols labeled with $b_k = 1$, and $S_k^{(0)}$ contains the other symbols.

$$z_k = \log \frac{\sum\limits_{i \in S_k^{(1)}} I_0\left(\frac{2\varepsilon_s |y_i|}{N_0}\right)}{\sum\limits_{i \in S_k^{(0)}} I_0\left(\frac{2\varepsilon_s |y_i|}{N_0}\right)} \tag{1}$$

3. DATA PRE-PROCESSING

The presence of high-amplitude noise emerging due to ambient noise and multi-path propagation leads to a degradation of the detection and correction abilities of the channel coding algorithms. The bit constellation namely $0-1-0$ is prone to error which mostly cannot be corrected by soft-in soft-out (SISO) decoder. A method was tested successfully to identify and eliminate the high-amplitude noise from the received dataset. This arithmetic mean of amplitudes $\bar{b}_k$ encountered over the last time windows is being calculated and updated continuously only during time slots with no data transfer. A more accurate mean is obtained by removing the outliers with a difference is bigger than standard deviation s_{b_k}. Two values are calculated as the noise tendency about a corresponding position only with its immediate neighbors. The value with the bigger deviation is used.

$$s_{nt} = \begin{cases} 1, & \bar{b}_k - b_k[i-1] \geq \bar{b}_k - b_k[i+1] \\ 0, & \text{otherwise.} \end{cases}$$

$$b_k[i] = \begin{cases} b_k[i] + \left(\bar{b}_k - b_k[i-1]\right), & \text{for } s_{nt} = 1 \\ b_k[i] + \left(\bar{b}_k - b_k[i+1]\right), & \text{otherwise.} \end{cases} \quad (2)$$

The bit samples at these marked positions will be modified by removing the deviation of the high-amplitude noise from the average noise from the value of the corresponding input sample as shown in Equation 2.

4. CHANNEL CODING STRATEGIES

To find the compromise between satisfying the need for safe communication and holding the complexity and latency at a low level, the codeword length is set to 288 and the dimensions of LDPC parity check matrix are defined as 288×576 with a code rate 1/2. The time-varying binary convolution codes is used in this work. The code symbols of a convolution code have a time index associated with them. The convolution code memory(M_s) is set to 289. The dimensions of parity check matrix are 288×290. The length of the code memory is 2.

The Turbo encoder and decoder applied in long term evolution (LTE) standard which is used in our analysis contains two 8-state constituent encoders which is appropriate for a swallow water channel. Our observations revealed that the increase the number of registers more than 4 does not lead to an improvement. The BCJR algorithm is used for MAP decoding and operates as in the LDPC case with LLRs.

5. COMPUTER SIMULATION

The acoustic channel simulator Bellhop [2] which is based on an ray tracing approach is used in combination with recorded several temperature profiles to compute the impulse responses for 3 waters, namely, Lake Constance, Pfuhler See, and a quarry pond in the village Schabringen. AWGN Noise and other factors of an swallow water channel have been added additionally. Because of handling of fresh waters, the salinity is accepted as 0 %. The underground is modeled as a continuous sinusoidal ground elevation with a maximum height of 0.5 m and a period of 10 m. To determine the BER, 288 bit random data was transmitted in one frame with 80 symbols per second. The frequency band employed for the frequency hopping is between 7 and 12 kHz and the used sampling frequency was 48 kHz. The attached results are obtained while the transmitter and receiver are placed at 9 meter and 2 meter depth respectively. The bandwidth of WHOI System is equal to 4 kHz. The noise level for the ambient noise is 75 dB re 1 μPa per Hertz. The power level is set to 8 mW.

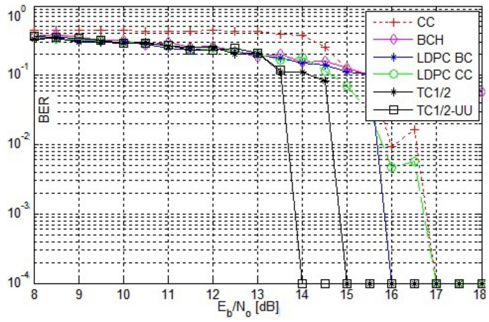

Figure 1: Performance Measurements

6. RESULTS & CONCLUSIONS

The number of iterations selected for the both decoding algorithms are different and adapted to lead to the best results. The turbo code algorithm needs 8 iterations, while the LDPC decoders need 40 iterations for the optimum results.

There is 1 dB difference between the best two algorithms, namely, turbo codes and LDPC block codes as shown on the Figure 1. The poor performance of LDPC codes under severe multi-path influence bases on very large selected channel memory and the selected decoding structure. If the channel quality varies rapidly, the large memory leads to a long-term effect of the distorted parts.

The E_b/N_0 values varies between 10.5 and 17 dB. During the tests in Pfuhler See, it has been observed that the further the destination, the lower the BER values are measured. For the other waters, exactly the opposite is ascertained. The same test for the distance 100 meters is executed for a channel with different Doppler spread values. With 16 Hz Doppler spread, a significant difference up to 1 dB can be ascertained in comparison to the case without a Doppler spread. The results of every algorithm worsened at almost the same rate.

The simulations of the implementation of high-amplitude noise removal have revealed a significant gain up to 1 dB as shown in the Figure 1. The curve marked with "TC 1/2-UU" indicates the BER values, if the high-amplitude noise elimination is handled before decoding.

The three introduced algorithms are practical for implementation with a minimum increase in transmitter and receiver complexity. It was shown that all those three techniques ensured additional coding gains up to 3 dB in comparison to the already algorithms available in the WHOI micro-modem. This coding gain is a very important resource to increase the data rate or decrease the used power level.

7. REFERENCES

[1] E. Gallimore, J. Partan, I. Vaughn, S. Singh, J. Shusta, and L. Freitag. The WHOI Micromodem-2: A Scalable System for Acoustic Communications and Networking. In *OCEANS 2010*, pages 1–7, Sept 2010.

[2] M. Porter. Ocean Acoustics Library, July 2014.

[3] M. C. Valenti, E. Hueffmeier, B. Bogusch, and J. Fryer. Towards the capacity of noncoherent orthogonal modulation: BICM-ID for turbo coded NFSK. In *in Proc. IEEE Military Commun. Conf. (MILCOM*, pages 1549–1555, 2004.