Performance Analysis of Substituting DVB-S2 LDPC Code for DVB-T Error Control Coding System

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Abstract

As HDTV (High Definition Television) services are the next big thing in the television broadcasting, the work for the next generation terrestrial Digital Video Broadcasting (DVB-T2) standard has started. Since HDTV channels require more capacity, new technologies as compared to current terrestrial DVB-T standard are necessary. One improvement is to select a more efficient Forward Error Control (FEC) code. A potential candidate for this is the Low Density Parity Check (LDPC) code specified in the in the DVB-S2 (second generation satellite Digital Video Broadcasting) standard. It is stated in the call for technologies issued by the DVB project, that this code is the working assumption. It is further stated that if the DVB-S2 LDPC code is not suitable for terrestrial channels, other channel coding schemes could be considered. In this paper the performance of a system where DVB-T concatenated Reed-Solomon-convolutional coding scheme is replaced by the DVB-S2 LDPC code is studied by simulations. The aim is to investigate the performance of the LDPC code in a terrestrial channel model.

Keywords

DTV, Standards, Channel Coding

INTRODUCTION

The current European standard for Terrestrial Digital Video Broadcasting (DVB-T) [1] was approved by European Telecommunications Standards Institute (ETSI) in December 1995. The DVB-T system is based on multicarrier OFDM (Orthogonal Frequency Division Multiplex) scheme to combat effects of the terrestrial transmission channels. The forward error control coding (FEC) in the DVB-T system is based on concatenation of a Reed-Solomon (RS) and a convolutional code. Since the introduction of DVB-T standard in 1995, coding theory has advanced. Recently, more efficient FEC methods have been discovered and most importantly computational resources for decoding them are available even in portable handheld devices. LDPC (Low Density Parity Check) codes together with turbo codes are reported to perform very close to the theoretical upper limit for the spectrum efficiency introduced by Claude Shannon already in 1948.

As digital television is moving towards HDTV (High Definition Television) based services, more capacity from the transmission network is required. Therefore, new terrestrial digital broadcasting standard DVB-T2 is being developed by the DVB project. To allow for increasing capacity, a more efficient FEC code is among the necessary improvements. One probable option for this code is the concatenation of LDPC and BCH (Bose-Chaudhuri-Hocquenghem) code specified in the second generation satellite Digital Video Broadcasting (DVB-S2) standard [2]. The error correction capability of the BCH code is rather modest, being from 8 to 12 bits in each codeword of several tens of thousands of bits. In this paper the performance of the LDPC code is compared to the performance of DVB-T FEC based on bit error rates (BER) and error distributions. The performance is evaluated based on only the performance of the LDPC code, since the main task of the concatenated BCH code is to remove the error floor after the LDPC decoding. Error floor is a phenomenon where the error rate of a code doesn't tend to zero as quickly with high signal strengths as with lower signal strengths. These error floors for LDPC codes tend to happen at such low BER levels that even the weak BCH code is enough to make error free transmission possible.

This paper is organized as follows. First, the simulation system is described. Then the performance comparison of the different coding schemes is presented. Also the effect of an additional time interleaver is investigated. Further, the error distributions after the decoding are studied. Finally, concluding remarks are given.

SIMULATION SYSTEM

In this chapter the simulation system is described. First, it is necessary to go through the most important characteristics of the used LDPC code and how the code is incorporated into the DVB-T system simulator.

DVB-S2 LDPC Code

LDPC codes are block codes, i.e. a block of data is encoded into a codeword on the contrary to convolutional codes where a continuous data steam is encoded. As their name indicates, LDPC codes are specified by sparse (Low Density) parity check matrixes. DVB-S2 LDPC codes are binary systematic codes, meaning that the information bits are visible in the codeword. DVB-S2 standard specifies two possible code lengths, namely 16200 and 64800 bits. The longer codes are specified for broadcast applications and the shorter ones are used for non-broadcast services. As broadcasting is of main interest in this paper, only the longer codes are considered. Code rates specified in the standard are: 1/4, 1/3, 2/5, 1/2, 3/5, 2/3, 3/4, 4/5, 5/6, 8/9 and 9/10, while the highest one is specified only for the

long code length. Individual parity check matrixes are defined for each code rate in [2], instead of utilizing shortening or puncturing. This way the coded data block coming from the encoder is always of the same length while the amount useful information in the data block varies as a function of code rate.

To decode LDPC codes, several algorithms exist. The sum-product algorithm (SPA) is an efficient soft decision decoding method for LDPC codes based on belief propagation. It is an iterative process where the reliability (soft) information of the received bits is refined iteration by iteration. Iteration consists of computing parity-check sums and updating reliability information based on the results of the parity-checks. The output of iteration is used as an input for the next one. This process of updating the reliability continues until all parity checks are fulfilled or maximum iteration count is reached. After this, hard decisions on the symbols are made to come to an estimation of what was sent. In our simulations soft information in the form of LLR (log-likelihood-ratio) from the demodulator is used. For more detailed information on the decoding algorithms as well as the LDPC codes themselves, interested reader can turn to [3].

Simulation Model

To obtain the results presented in this paper, a DVB-T simulator was modified by substituting the interleaved concatenated RS-convolutional coding by DVB-S2 LDPC code. The block schema of the simulator is shown in Figure 1.
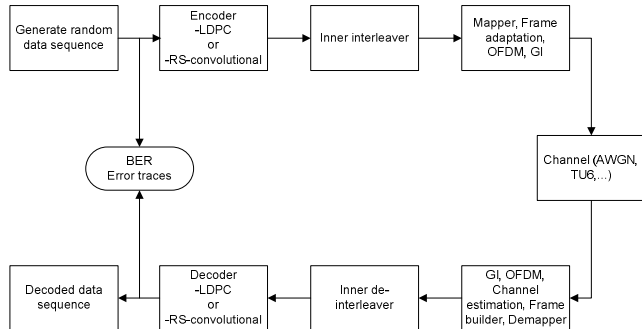

Figure 1: Block schema of the simulator

First, random data sequence is generated. The generated data is encoded either by the described LDPC code or by the conventional DVB-T coding scheme. Inner interleaver used in the simulations is the one defined in the DVB-T standard [1]. The function of the inner interleaver is to shuffle the bits so that bits that are close to one another in the coded stream are not transmitted in the same modulation symbol. Further, the interleaver permutes the data to available OFDM carriers to achieve frequency diversity. Mapper maps the information to modulation symbols. Frame adaptation creates the OFDM frame structure, consisting of 68 OFDM symbols. After this OFDM symbols are created and guard interval inserted. In

the receiver, the reverse chain is performed. In the simulations knowledge on the channel is available, and ideal channel correction is performed. The demapper is modified from the DVB-T one to output bitwise LLRs to enable LDPC decoding based on this reliability information. LLR is defined as follows:

$$LLR = \log \frac{P(b = 0 \mid r = (x, y))}{P(b = 1 \mid r = (x, y))}, \qquad (1)$$

where $P(b{=}0|r{=}(x,y))$ is the probability of bit being zero based on the observation r, and $P(b{=}1|r{=}(x,y))$ is the probability of bit being one based on the observation r. Calculating exact LLR values involves calculating distances of the received signal to all the constellation points. For higher order modulations this can be a tedious job. In our simulations we have used approximate LLRs. In the calculation of approximate LLRs only the nearest constellation point with 1 and 0 in corresponding bit location is considered rather than all constellation points. For Gray coded QAM (Quadrature Amplitude Modulation) as the one in DVB-T, the LLR approximation can be based on the decision regions due to intelligent bit ordering of Gray coding. Mechanism for this kind of LLR calculation is presented in [4].

Finally, LDPC decoding using belief propagation or RS and convolutional decoding is performed. Original and the received data are compared to obtain the bit error rate and byte error trace indicating which bytes (8 bit segments) are correct and which are not. Byte level error information is gathered to enable comparison to concatenated RS-convolutional code, where the RS decoder operates on bytes rather than bits.

DVB-T Coding Scheme

The outer code in the DVB-T concatenated coding scheme is RS(204,188) code shortened from the RS(255,239) code. This RS code is capable of correcting 8 byte errors. The code operates on Transport Stream (TS) packets of length 188 bytes, i.e adds 16 bytes of redundancy to each TS packet. The output of the RS code is interleaved using a convolutional interleaver. The output of the interleaver is fed to the inner code. The inner code is 64-state rate ½ convolutional code defined by generator polynomials $G_1{=}171_{OCT}$ and $G_2{=}133_{OCT}$. Other code rates in the system are obtained by puncturing this mother convolutional code with the puncturing patterns given in the standard [1]. Code rates defined in the standard are 1/2, 2/3, 3/4, 5/6 and 7/8. In the receiver of our simulation chain, soft information is used in the viterbi decoder to perform soft decision decoding of the convolutional code. Hard decision RS decoding is used.

PERFORMANCE ANALYSIS

In this chapter the performance of the LDPC code is analyzed and compared to the performance of the RS-convolutional code used in DVB-T systems. In the simulations, approximately 10 Mbits of data was simulated.

First, the effect of the maximum iteration count on the performance of the LDPC decoding is studied. This is done to justify a single value for the maximum iteration count to be used in the following simulations. Then the error performance is presented in Additive White Gaussian Noise (AWGN) channel and six tap typical urban (TU6) [5] mobile channel. The transmission parameters used in the following simulations are given in Table 1.

Table 1: Simulated system parameters

Parameter	Value
Guard interval	¼
FFT size	8k
Modulation	16-QAM
LDPC code length	64800 bits
LDPC max. Iterations	50
Effective code rates	1/2, 2/3, 3/4, 5/6

Maximum Iteration Count

The maximum number of iterations for each codeword in LDPC decoding must be set to some value, to stop the process in case there are too many errors for the code to correct. This way the worst case decoding latency can be set, which is important in streaming applications such as Digital Television. On the other hand if we set the iteration number too small, we can loose a lot in the error correction capability. The effect of the amount of iterations as well as the performance of the LDPC code rate 1/2 in the TU6 mobile multipath channel is presented in Figure 2. The Doppler frequency in the simulation is f_D=10Hz while the system parameters are as given in Table 1.

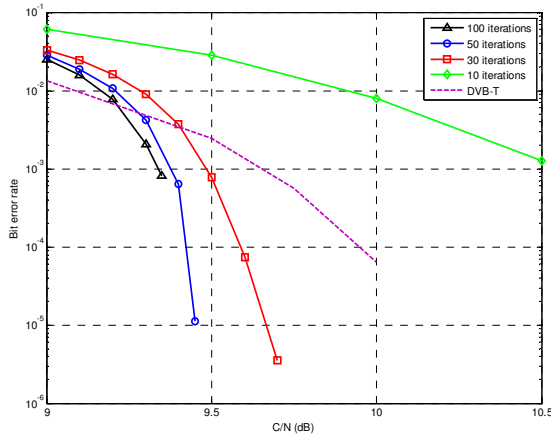**Figure 2. The effect of LDPC iteration count and comparison to DVB-T in TU6 channel f_D=10Hz**

It is visible that increasing iterations from 50 to 100 doesn't give as much gain as when increasing from 10 to 30 or 30 to 50. The amount of iterations directly affects the decoding complexity and latency. Therefore, from the performance point of view 50 iterations would seem a nice compromise between error performance and decoding complexity and is thus used in the following simulations in this paper.

Error Performance

The error performance of the LDPC coded system is compared to the standard DVB-T system based on bit error rates after the decoding. It should be noted that the true code rate of the concatenated scheme is lower than that of LDPC because of the concatenation of the RS code (true code rate (1/2)*(188/204)=0.46 instead of code rate 1/2 etc.). Let us first compare the performance of the codes in widely used AWGN channel to have a reference comparison of different coding schemes.
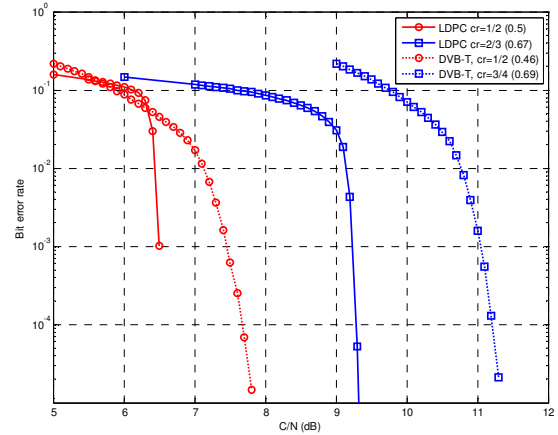

Figure 3: Comparison of LDPC and DVB-T coding in the AWGN channel

Gain is visible with all of the considered code rates in the AWGN channel. Figure 3 shows the performance curves for true code rates 1/2 and 2/3. LDPC code rate 2/3 is compared to the DVB-T RS-convolutional coding scheme with convolutional code rate 3/4. This because the true code rate of the DVB-T coding with convolutional code rate 3/4 (shown in parentheses) is closer to that of LDPC 2/3 than for the DVB-T coding with convolutional code rate 2/3. In the same way, LDPC code rates 3/4 and 5/6 are compared to DVB-T coding with convolutional code rates 5/6 and 7/8 respectively. It is also observed that the slopes of the BER curves for LDPC codes are steeper than those of the RS-convolutional coding. The simulated coding gains at bit error rate 10^{-4} for the studied LDPC code rates 1/2, 2/3, 3/4 and 5/6 are 1.2 dB, 2 dB, 1.8 dB and 0.6 dB respectively.

Performance comparison of the coding schemes in TU6 channel with f_D=10Hz is shown in Figure 4. This Doppler frequency corresponds to relatively slow receiver movement assuming that the center frequency of the transmitted signal is around 500 MHz. The true code rates are given in parentheses in the figure. The coding schemes with rather similar overall code rates are again compared. Coding gain for the LDPC code over the DVB-T coding is observed. At bit error rate level 10^{-5}, for true code rates 1/2, 2/3, 3/4 and 5/6 gains of 0.6 dB, 2.5 dB, 3 dB and 2.7 dB respectively are observed. On the other hand, it can be seen, that for example for LDPC code rate 3/4 and DVB-T coding with convolutional code rate 3/4 almost similar

BER is obtained with similar C/N values. Thus, with LDPC code higher data throughput is obtained with similar C/N value since true code rate for LDPC code is 0.75 and for DVB-T coding it is 0.69. In this exemplary case, the gain of LDPC code in data throughput is nearly 9%.
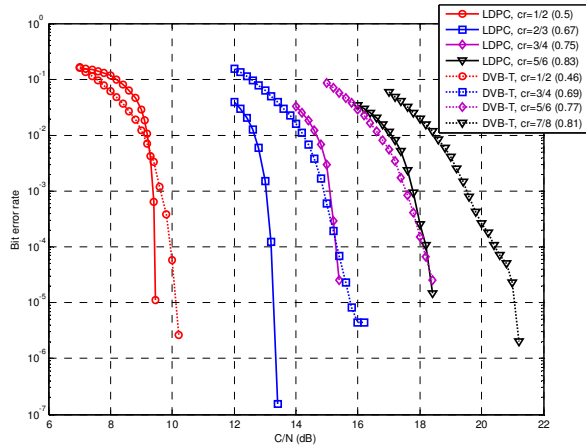

Figure 4: Performance comparison of LDPC and DVB-T coding in TU6 channel with f_D=10Hz

Additional Time Interleaver

It is known that the DVB-T system doesn't necessarily have enough time diversity to combat the challenges of fading and impulsive channels. These kinds of channels are present at least in mobile use cases. To make mobile reception of the digital broadcast services based on DVB-T possible, DVB-H [6] was introduced. DVB-H is an amendment to DVB-T. The most important new element in DVB-H was the MPE-FEC (MultiProtocol Encapsulation – Forward Error Correction) that introduces additional time diversity to the system. To have also the robustness against fading and impulsive channels in the second generation DVB-T2, longer time interleaving than what is in the present DVB-T system is necessary. Therefore, let us further study the effect of adding a time interleaver over 34 and 68 OFDM symbols (that is over half and one OFDM frame) to our simulation system. A matrix interleaver of total size equal to *num_symbols* x *num_carriers* x *bits_per_carrier* is used. For interleaver over 34 OFDM symbols and the other system parameters as given in the Table 1 this corresponds to 34x6048x4= 822528 bits and for interleaver over 68 OFDM symbols to 68x6048x4=1645056 bits. Thus interleaving depths of about 13 and 25 LDPC codewords are obtained. The results with code rate 1/2 LDPC code in TU6 channel with f_D=10Hz using the additional time interleaver are shown in Figure 5. It is seen that some gain (0.1dB – 0.2dB) can be obtained by already adding this rather un-optimized interleaver to the system. Most likely, a time interleaver providing better result in this particular system could be found by careful system design. The time diversity doesn't come without any cost though. For our example, assuming that 5 bits are used for presenting the soft information, the deinterleaver in the receiver would require 5x822528 ~ 4

Mbits of memory for the interleaver over 34 OFDM symbols. Also, the delay of the system increases, since the data cannot be decoded before the interleaver is completely filled.
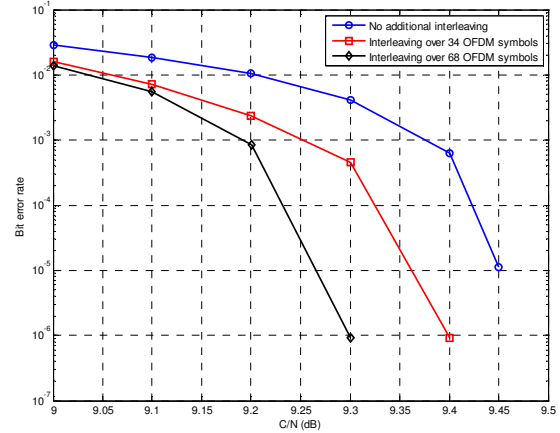

Figure 5: The effect of the additional time interleaver

Error Distribution

We further continue our analysis by comparing the byte error distributions after the decoding with similar propagation scenarios. Figures for only code rate 1/2 in TU6 with f_D=10Hz are presented, but conclusions are similar for other code rates also. Byte error rates within codewords (4050 bytes for LDPC 1/2 and 188 bytes for RS-convolutional) for both systems are shown in Figure 6 and Figure 7.
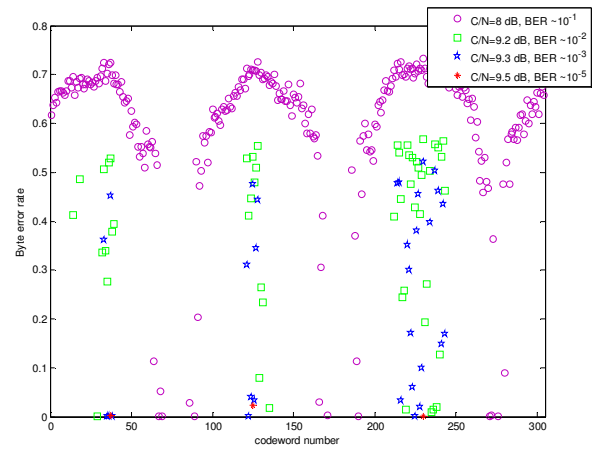

Figure 6: Byte error rates in codewords after the LDPC decoding

The same amount of data is simulated in both figures, thus the number of shorter codewords in DVB-T is greater than the number of longer LDPC codewords. The error distributions are compared at C/N values giving similar overall BER of the simulation. It is clearly visible that errors are more densely packed after the LDPC decoding as compared to decoding concatenated RS-convolutional code. When the byte error rate is approximately 10^{-3} for

both systems, i.e. C/N=9.3 dB for LDPC code and C/N=9.7 dB for RS-convolutional code, it is seen that byte error rate in LDPC codewords is up to 50% and for RS-convolutional the rate it is below 10%. On the other hand, error free durations between the error bursts are longer with the LDPC coding. Which kind of error behavior is desirable depends on the application. Some applications may work better with bursty errors while others may prefer more uniformly distributed errors.
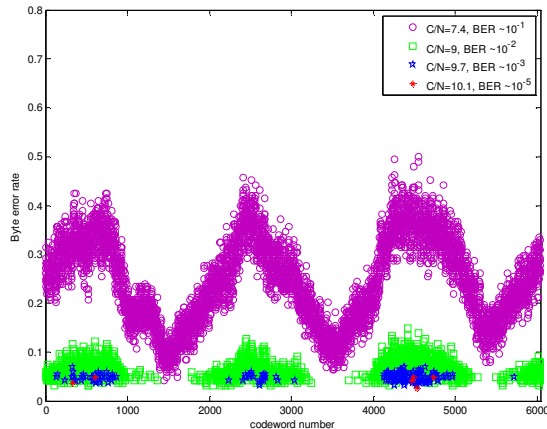

Figure 7: Byte error rates in RS codewords after the RS decoding

CONCLUSIONS

In this paper we compared the performance of DVB-T RS-convolutional coding to the performance of the same system where the RS-convolutional code is replaced with DVB-S2 LDPC code. It is seen that the LDPC code outperforms the concatenated code in AWGN and TU6 f_D=10Hz channels with all studied code rates. Also, the maximum amount of iterations necessary in LDPC decoding was considered and it was seen, that 50 iterations could be used as a reasonable compromise between the performance and the complexity. It was also observed, that by adding a time interleaver, the performance of the system

in mobile channels, such as TU6, can be enhanced. As for the error distributions after the decoding, it was observed that LDPC decoding provides more dense error bursts than the concatenated decoding scheme. Which kind of error distribution is desirable depends of the application that is being transmitted over the system. As a future topic, the interleaver could be further optimized to match the system and the possible gain obtained by the longer time interleaver could be studied.

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