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PARAMETER OPTIMIZATION AND SIMULATED PERFORMANCE OF A DVB-T DIGITAL TELEVISION BROADCASTING SYSTEM

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Abstract

A prototype of a Digital Television Terrestrial Broadcasting System has been implemented according to the European DVB-T standard. The first step in the construction of this prototype involved parameter optimization via simulation.

This paper presents the optimization process of the system parameters and the final achieved BER performance. Important design aspects such as interleaving, convolutional puncturing codes, Viterbi decoding, pilot-based channel correction and optimal pilot power are considered.

The DVB-T standard offers some suggested values of the C/N ratio needed to achieve the desired BER requirements in the receiver. However, they have been obtained assuming ideal channel correction and synchronization.

Simulation results are provided here in non-ideal conditions.

1. INTRODUCTION

In the European DVB Project (Digital Video Broadcasting), founded by initiative of more than 190 organizations, the standards for digital television broadcasting have been specified for the different transmission media: satellite (DVB-S), cable (DVB-C) and terrestrial (DVB-T) [1]. The later is a result of the RACE Project dTTb (Digital Television for Terrestrial Broadcasting), which has covered all the aspects of digital television terrestrial broadcasting in Europe, with the work of 30 partners from 10 different countries [2].

The system specified within the dTTb Project is very complex [1,2]. Taking the input signal coming from an MPEG-2 encoder, all the elements for its correct transmission and reception are defined, adapting the signal to the terrestrial channel. With this aim, a combined encoding and modulation process [3] is applied to the data source, with different coding rates [4] (which imply different bit rates), according to the channel characteristics and the expected quality at the receiver. Besides, different services, depending on the quality of the received TV signal (standard, enhanced and high definition) and several configurations (by combination of these services) are defined.

In order to build a prototype for such a system, we have developed a methodology to allow the use of software design tools to implement in a rapid and efficient way communications systems prototypes using digital signal processors (DSPs) [5]. The first step in the implementation process is the optimization of every parameter to achieve the system BER requirements.

In particular, the influence on the BER performance of the following elements has been studied:

- ♦ interleaving
- ♦ convolutional punctured codes and Viterbi decoding
- ♦ pilot-based channel correction
- ♦ optimum power of pilot sub-carriers

In the next section, the transmitter and receiver are described. Following, the optimized system parameters and the final achieved BER performance are presented. These results are compared with the DVB-T standard suggested performance.

2. DESCRIPTION OF TRANSMITTER AND RECEIVER ELEMENTS

In this section, the transmitter and receiver of the prototype, which implement the main functions defined in the DVB-T standard [1], are described.

The elements of the transmitter, whose block diagram is shown in Figure 1, are the following:

Figure 1 Transmitter block diagram

2.1 Inner coding

Inner coding is based on $\frac{1}{2}$ convolutional code, with constraint length 7 and generator polynomials 171_{Oct} , 133_{Oct} . This mother code is used to produce several punctured codes with different coding rates [4].

Table 1 indicates the bits that must be punctured in order to construct the different codes with x1, x2, x3, ... the output bits of mother code.

Table 1 Puncturing patterns

rate	pattern	transmitted sequence
1/2	11	x1 x2
2/3	1101	x1 x2 x4
3/4	110110	x1 x2 x4 x5
5/6	1101100110	x1 x2 x4 x5 x8 x9
7/8	11010101100110	x1 x2 x4 x6 x8 x9 x12 x13

2.2 Inner interleaving

The DVB-T standard specifies a combination of bit and symbol inner interleaving: a first block interleaving is applied to the bits with 126-bit blocks and a second block interleaving is applied to the digital modulation symbols. The block length of this second interleaving is equal to the number of information sub-carriers in the OFDM frame (1512 in 2K-OFDM and 6048 in 8K-OFDM).

2.3 Digital modulation

Digital modulation is based on QPSK, 16-QAM or 64-QAM. It is a multi-resolution modulation which allows one to design a hierarchical error protection scheme.

The constellation is divided in clusters that are further divided in sub-clusters. The higher priority bits (HP) are mapped to the clusters and the lower priority ones to the sub-clusters (MP) or constellation points (LP).

The euclidean distances between clusters d_{E1} , sub-clusters d_{E2} and points d_{E3} can be changed in order to obtain different degrees of protection. Figure 2 shows an example for 16-QAM with $\alpha_1 = d_{E1}/d_{E2}$.

Figure 2 Multi-resolution 16-QAM

Besides modulation and coding are combined following a multi-level coding scheme [3].

2.4 Orthogonal Frequency Division Multiplexing (OFDM)

OFDM is one of the best alternatives to alleviate the multipath effects in mobile communications [6]. For reception in built-up areas with delay spreads of some microseconds, the channel is highly frequency selective and sophisticated equalization techniques are required to achieve high bit rate transmissions. However, a combination of OFDM and coding associated with interleaving in the frequency domain (COFDM) can take advantage from the diversity associated to multipath [7]. Besides, multi-carrier systems have been proved to be more computationally efficient than single-carrier systems with equalization under some multipath channel conditions [8].

In an OFDM system the spectrum associated to each elemental data is a small portion of the total bandwidth, which is divided in N sub-channels. Each of them is modulated with one symbol and they are all multiplexed in frequency.

If T represents the OFDM symbol duration, that is:

$$T = \frac{N}{B}, \text{ the } N \text{ carriers are placed in the frequencies:}$$

$$fk = f_o + k / T, \quad k = 1, 2, \dots, N - 1$$

A higher spectral efficiency is achieved because the different sub-channels are allowed to overlap.

Orthogonality between them simplifies the separation process in the receiver, allowing the use of the Discrete Fourier Transform [9].

The complex envelope of the transmitted OFDM signal, sampled with sampling frequency $f_s = \frac{1}{B}$, is:

$$s(n) = \sum_{m=-\infty}^{\infty} \sum_{n=N_g}^{N-1} \sum_{k=0}^{N-1} s_{m,k} \cdot e^{j \frac{2\pi}{N} kn} \prod (n - m(N + N_g))$$

This means that every OFDM symbol has $N+N_g$ values: the N obtained by an Inverse Discrete Fourier Transform (IDFT) plus an added Time Guard of N_g . This Time Guard consists of the last N_g samples of the IDFT which are added at the beginning of each symbol in order to cope with the ISI caused by multipath delay spread.

$\Pi(n)$ is a rectangular pulse of duration $N+N_g$ samples.

Two frequency guards are also inserted, leaving some sub-carriers without modulation at the lower and upper sides of the spectrum, as a protection against adjacent channel interference (ACI).

Besides some sub-carriers (called pilots) are transmitted with a known amplitude in order to provide a means of channel estimation and correction.

2.5 Transmitted signal

The COFDM signal is transmitted in QAM (quadrature amplitude modulation) with a first intermediate frequency $IF_1=4.57$ MHz, that is, half the sampling frequency of the complex lowpass signal. The complex signal is interpolated by a factor of two before translation to the IF frequency. Its spectrum is shown in Figure 3.

Figure 3 Spectrum of QAM transmitted signal

The transmitter block diagram is shown in Figure 4.

Figure 4 Receiver block diagram

Its main elements are the following:

2.6 I-Q generation

Three means of generating the I-Q components from the real received signal have been investigated:

- Traditional I-Q generation
- Frequency conversion using a sampling frequency equal to $4 IF_1$ as suggested in dTTb
- Use of the Hilbert transform

The traditional method exhibits some disadvantages, such as sensitivity to frequency or phase mismatch between transmitter and receiver local oscillators, that are overcome by the two proposed digital methods.

While both digital schemes offer the same performance in AWGN, the method based on the Hilbert transform performs better in situations of frequency instability or timing jitter.

2.7 OFDM frame synchronization and Orthogonal Frequency Division Demultiplexing

Three synchronization operations are performed in this system:

- Identification of the number of sub-carriers
- Time Guard estimation
- FFT window adjustment

They are all accomplished by using the following correlation function [10]:

$$corr(n) = \sum_{k=n-N_I+1}^n x(k) \cdot x^*(k-N)$$

The Orthogonal Frequency Division Demultiplexing (OFDD) process is implemented by means of a Fast Fourier Transform [9].

2.8 Channel estimation and correction

The channel effects in OFDM can be corrected by interpolating in time and frequency the DFT of the channel impulse response with the aid of the pilot sub-carriers inserted in the OFDM frame. This process is shown in Figure 5. More details can be found in [11].

Figure 5 Channel correction by pilot-based interpolation

2.9 Digital demodulation and inner decoding

The use of multi-level coding allows one to implement the demodulation-decoding process in a sub-optimal way by means of multistage decoding [12], reducing the complexity of the optimal maximum likelihood decoder. This technique consists in decoding in a sequential fashion the different levels of priority, using the information of the already decoded HP bits for the decoding of the MP bits, and both the HP and MP information for the decoding of the LP bits.

The decoding of each bit stream is accomplished by using the Viterbi algorithm with a previous insertion of dummy bits for the codes generated by puncturing.

The inclusion of channel state information (CSI) in the Viterbi metrics has been considered.

3. PARAMETER OPTIMIZATION

The main parameters related to the described elements have been optimized so as to achieve the system BER requirements. Some of them are discussed here.

3.1 Interleaving

As mentioned in the preceding section, the DVB-T standard specifies a combination of bit and symbol inner interleaving. Such an interleaving requires memory and causes delay in the transmitter and receiver.

Figure 6 BER in a 2K-OFDM system with QPSK: no interleaving (-), symbol interleaving (--), bit and symbol interleaving (*)

It can be questioned if such a complex interleaving is needed or only the symbol interleaving, which corresponds to the COFDM concept, would be enough. Simulations have been run to answer this question when the OFDM sub-carriers are modulated in QPSK and 16-QAM. Results are shown in Figures 6 and 7.

Figure 7 BER in a 2K-OFDM system with 16-QAM: no interleaving (-), symbol interleaving (--), bit and symbol interleaving (*)

In these simulations the OFDM signal has been transmitted through a multi-path channel named F1 whose definition can be found in [11].

It can be noticed that a combination of bit and symbol interleaving offers a negligible gain when the digital modulation of the sub-carriers is QPSK. However, when using a higher level modulation such as 16-QAM, a combined interleaving can be necessary to achieve the BER requirements.

3.2 Inner decoding

Two main parameters have to be adjusted in relation to the inner decoder: the Viterbi decoding path length and the number of quantization levels. Figure 8 shows the bit error rate for different values of the decoding path length when using a $\frac{3}{4}$ rate punctured code. As it can be seen, a value of 30 corresponds to an optimum trade-off between memory and performance.

Figure 8 BER of $\frac{3}{4}$ rate punctured code vs. decoding path length for $C/N = 3$ dB

Figure 9 shows the performance of the same code when two different numbers of quantization levels are used: the upper curve corresponds to an 8-level quantization while the lower curve has been obtained with 32 levels.

Figure 9 BER of $\frac{3}{4}$ rate punctured code vs. C/N

Although the use of 32 levels of quantization instead of 8 for the mother $\frac{1}{2}$ convolutional code leads to a poor gain (0.25 dB), we can see that this gain is much bigger when using punctured codes.

3.3 Pilot-based channel correction

Figure 10 shows the symbol error rate (SER) obtained in an OFDM system after the F1 channel [11] when pilot-based channel correction is used [11]. For 2K and 8K systems, channel correction leads to a better performance, but this only occurs when the C/N ratio exceeds some value (around 10 dB).

Figure 10 SER versus C/N with and without channel correction

Since the system working point can be below this value, some simulations have been run on order to check in what situation a correction scheme is needed.

Figure 11 BER of QPSK-OFDM (2K) after F1 channel

The bit error rate (BER) for a 2K system using QPSK can be seen in Figure 11. Three cases have been considered: channel correction without interleaving (o),

no channel correction nor interleaving (*), no channel correction and with symbol interleaving (x).

It can be noticed that for the considered system and C/N values, the correction of the channel effects leads to a worse performance, since the channel estimation is corrupted by noise.

However, when using 16-QAM instead of QPSK, the working point corresponds to higher C/N values and the channel correction scheme provides better performances. This fact can be seen in Figure 12 in which BER after channel correction (-) and without channel correction (.-) are shown.

Figure 12 BER of 16 QAM-OFDM (2K) after F1 channel

On the other hand, when the channel is more frequency-selective as P1 channel defined in [11], channel correction is always needed. Figure 13 shows the BER performance of a QPSK-OFDM (8K) system after this P1 channel: (1) and (2) have been obtained without correction and (3) and (4) with channel correction (without and with interleaving respectively).

Figure 13 BER of QPSK-OFDM (8K) after P1 channel

3.4 Optimum power of pilot sub-carriers

As mentioned before, some pilot sub-carriers are inserted in the OFDM frame for channel estimation purposes. The power of the pilots can be varied with respect to that of the information sub-carriers in order to reduce estimation errors. However a trade-off between C/N ratio of information sub-carriers and channel estimation is required.

The parameter $P = \frac{pot_{pilots}}{pot_{resto}}$ has been defined and

its optimum value has been searched by simulation. Results for a QPSK-OFDM 2K system are shown in Figure 14 leading to an optimum value of 2.

The same value is obtained when the digital modulation and/or the number of sub-carriers are changed.

Figure 14 SER versus parameter P in a QPSK-OFDM 2K system

4. ERROR PERFORMANCES

DVB-T standard [1] defines many different transmission modes by selecting the inner code, digital modulation, Guard Time duration and number of sub-carriers. For some of these combinations the C/N ratios required to obtain a $BER=2 \cdot 10^{-4}$ (this means Quasi Error Free performance after Reed-Solomon decoding) are provided in the standard specifications as indicative values that must be confirmed by testing. These values assume ideal channel estimation and synchronization in the presence of F1 and P1 channels [1].

With the aim of reducing the number of cases for study, the following modes have been considered in the dTTb project [2]:

- Mode 1: 64-QAM, 2/3 code.
- Mode 2: 16-QAM, 1/2 code.
- Mode 3: 4-QAM (QPSK), 1/2 code.
- Mode 4: 64-QAM, 1/2 code.

- Mode 5: 16-QAM, 3/4 code.

- Mode 6: 64-QAM, $\alpha_1=2$, $\alpha_2=1$, two levels of hierarchy and 1/2 code for HP, 3/4 code for MP-LP.

All of them consider 2K and 8K sub-carriers.

The C/N values required to obtain this BER in non-ideal conditions are presented here for Modes 2-5 and compared to those provided in [1]. Table 2 presents these values for a 2K system.

Table 2 C/N (dB) to achieve $BER=2 \cdot 10^{-4}$ in 2K-OFDM

Mode	AWGN: DVB-T	AWGN: results	F1: DVB-T	F1: results	P1: DVB-T	P1: results
Mode 2	8.8	9.8	9.6	12	11.2	NO
Mode 3	3.1	3.3	3.6	4.1 (*)	5.4	NO
Mode 4	14.4	14.5	14.7	(**)	16.0	NO
Mode 5	12.5	13.5	13.0	20.6	16.7	NO

(*) without channel correction

(**) saturation in $BER=4 \cdot 10^{-4}$

The values for an 8K system are shown in Table 3.

It should be noted that the C/N values that are given in [1] are identical for 2K and 8K sub-carriers both in AWGN and multi-path channel conditions. This is due to the fact that ideal channel estimation has been considered in the standard. Obviously the situation is different when the channel is really estimated as shown in Tables 3 and 4.

Table 3 C/N (dB) to achieve $BER=2 \cdot 10^{-4}$ in 8K-OFDM

Mode	AWGN: DVB-T	AWGN: results	F1: DVB-T	F1: results	P1: DVB-T	P1: results
Mode 2	8.8	9.8	9.6	11.4	11.2	34
Mode 3	3.1	3.3	3.6	4.2 (*)	5.4	14.4
Mode 4	14.4	14.5	14.7	17.5	16.0	(**)
Mode 5	12.5	13.5	13.0	16.5	16.7	(***)

(*)without channel correction

(**) saturation in $BER=5 \cdot 10^{-3}$

(***) saturation in $BER=4.5 \cdot 10^{-3}$

On the other hand, it is not always convenient to correct the channel effects, as discussed in section 3.3. Values marked with (*) have been obtained without this correction.

The C/N values required in our prototype under AWGN conditions are very similar to those given in [1].

When the channel is F1 the difference between the C/N values given in the standard and the obtained ones increases due to the non-ideal channel estimation. On the other hand, it can be seen that channel effects are best corrected with 8K sub-carriers.

Finally, if the channel is P1 the real obtained performance is much worse than expected in [1] for 8K and it is not possible to achieve the desired BER values with 2K because of the smaller number of pilot sub-carriers.

These performance results have been obtained without using channel state information in the inner decoder. The required C/N values could be lowered if this information was used. This fact is subject to further work.

5. CONCLUSIONS

The European Digital Television system for Terrestrial Broadcasting specified within the dTTb Project and the DVB-T standard is very complex, allowing a variety of configurations and operation modes. A parameter optimization process has been carried out in order to build a prototype containing the main elements considered this standard.

In order to achieve the best bit error rates, the following points have been studied:

- interleaving
- convolutional punctured codes and Viterbi decoding
- pilot-based channel correction
- optimum power of pilot sub-carriers

Finally, a realistic BER performance has been provided and compared to the suggested ideal values given in the standard.

Although the results in the AWGN channel agree, the performance achieved in non-ideal conditions is poorer than expected in multi-path environments.

It should be noted that the use of channel state information in the inner decoder could lead to a better performance. However, this fact is subject to further work.

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