Physical layer DVB-SH performance prediction based on mutual information

Wilfried Chauvet, Jérôme Lacan, Caroline Amiot-Bazile, Frederic Lacoste, Benjamin Ros

To link to this article: DOI: 10.1002/sat.1011
URL: http://dx.doi.org/10.1002/sat.1011

Physical layer DVB-SH Performance Prediction Based on Mutual Information

W. Chauvet[1], J. Lacan[1,2]*, C. Amiot-Bazile[3], F. Lacoste[3] and B. Ros[3]

[1] *TéSA,14-16, Port Saint-Etienne 31000 Toulouse (France) ,*
[2] *ISAE, University of Toulouse, 10, avenue E. Belin, BP 54032 31055 Toulouse cedex 4*
[3] *CNES, 18 avenue E. Belin, 31401 Toulouse Cedex 9*

SUMMARY

DVB-SH (Digital Video Broadcasting- Satellite Handled) is a hybrid satellite terrestrial broadcasting standard dedicated to provide video or audio services for handheld terminals. On the satellite part, this standard can make use of interleaving mechanisms to mitigate the effects of the Land Mobile Satellite (LMS) channel. As result, these mechanisms enables the in-time distribution of a codeword over a duration ranging from 100 ms to about 30 s, depending of their parameters. This mechanism significantly improves the error recovery performance of the code however, in the literature, a theoretical evaluation at system level of this improvement is missing. Moreover, carrying out Monte-Carlo simulations implementing real decoding processes on significant traveled distances is time prohibitive.
We propose hereafter a prediction method compatible with fast simulations to quantitatively evaluate the system performance in function the Packet Error Rate (PER), Erroneous Second Ratio (ESR) and zapping time. This method is based on the computation of the mutual information between emitted and received symbols for QPSK modulation and turbo coding.
We demonstrate that our method reaches a prediction precision of the order of 0.1 dB, which is significantly better than two classical prediction methods. Moreover, our solution reduces the simulation time by a factor of 500 compared to Monte-Carlo. Beyond DVB-SH application, the presented approach can be applied in a large panel of satellite mobile systems and is completely new for the satellite community.

Received . . .

KEY WORDS: DVB-SH, mutual information, prediction

1. INTRODUCTION

DVB-SH [1][2] (Digital Video Broadcasting Satellite Handled) is a standard dedicated to hybrid broadcasting systems combining satellite and terrestrial parts. This system operates in S band at 2 GHz and provides high quality video or audio services for light terminal with limited battery capacities. We focus in this paper on the satellite component of this hybrid system, and on its performance evaluation.

The propagation channel LMS (Land Mobile Satellite) has been widely studied ([3], [4], [5], [6]). This channel is time varying which induces very dispersive values of received power, with several distance scales (fading effect at short scale, shadowing effect at medium scale). In order to mitigate the effects of such propagation channel, DVB-SH provides a time slicing process that relies on convolutional interleavers that can distribute the symbols of a single codeword (of the turbo code) over up to 30 seconds. This mechanism can be seen as a fade averaging and results in a significant improvement in terms of error recovery.

A major difficulty associated to this mechanism resides in its performance evaluation on sufficiently long distances in order to be able to tune its parameters. Indeed, the use of time slicing leads to a non stationary distribution of the Signal to Noise Ratio (SNR) inside a codeword of the turbo code. Nevertheless, there is no theoretical expression for the performance of a turbo code for such a complex non stationary distribution, and therefore, the performance evaluations can only be accomplished through simulations. Nevertheless, in order to gain access to statistical values like the variations of the Packet Error Rate (PER) in time, Erroneous Second Ratio (ESR) and zapping time, simulations based on Monte-Carlo methods become time-prohibitive. Thus, a modeling step turns out to be necessary.

In this context, the modeling task can be divided into two parts: modeling the channel and modeling the performance of the DVB-SH physical layer, more precisely the coding/decoding process on interleaved channel samples. Concerning the LMS channel, models [5] and [3] respectively based on a Markov Chain with three states and a semi-Markov Chain with two states have been proved to be accurate. With regard to the coding/interleaving performance on this channel, the issue to be addressed is as follows: given a perfect Channel State Information (CSI), which metric allows to obtain the best prediction of performance? Several methods can be found to answer this question, which differ on the measure retained, denoted as the Link Quality Metric (LQM).

The most intuitive approach consists in choosing the mean Signal to Noise Ratio (SNR) inside a codeword [7] [8] p.70. However, the larger the SNR dispersion inside a codeword is the larger the prediction error will be. In particular, the PER prediction error of this kind of approach is demonstrated in [9]. Reference [10] points out the necessity of using a convex function of the SNR to take this result into account. Among the propositions of convex functions, we can cite the Q function (gaussian complementary cumulative distribution function), the exponential function, or more complex functions [11]. More recently, a new LQM which consists in considering the Mutual Information (MI) between the coded bits and the Log Likelihood Ratio (LLR) at the input of the decoder, was studied for terrestrial MIMO transmissions [12], [13], [14], [10] [15]. The main objective of this paper is to explore the potentialities of this latter method for our specific context.

After having evaluated several LQM metrics and demonstrated the very good precision of the method retained, we carried out an evaluation of ESR performances on representative traveled channel distances as a function of the system margin conjointly with an evaluation of the statistical zapping time, which represents a key and a new entry for future DVB-SH systems dimensioning.

Beyond DVB-SH application, the presented approach can be applied to a large panel of satellite mobile systems and is completely new for the satellite community, as demonstrated by the long monte-carlo simulation driven for DVB-SH performance characterization in DVB-SSP standardization group [8].

In the following section, we present the prediction issue by highlighting the LMS channel and the time slicing mechanism for the DVB-SH system. Then, in Section 3, we present three different LQMs for PER prediction. We comparatively evaluate the three selected prediction methods in terms of BER and PER in Section 4. Section 5 presents a validation of the PER and ESR prediction of the DVB-SH physical layer on a LMS channel. In Section 6, we present results in terms of ESR and statistical zapping time for a wide range of interleaver parameters considered in the standard. Finally, we present in Section 7 a comparison of complexity and execution time with the traditional Monte-Carlo approach.

2. PERFORMANCE EVALUATION ISSUE IN DVB-SH SYSTEMS

2.1. *LMS channel modeling*

In DVB-SH systems, coverage is mostly ensured by satellite in rural and suburban areas. At S-band, the Land Mobile Satellite (LMS) propagation channel can generally be considered as non frequency selective. Several propagation channel models have been developed in the past. All these models rely on a representation of the LMS channel by a first order Markov chain corresponding to the large scale (several meters) level changes affecting the transmitted signal. Some of these Markov chains take into account two states (Line Of Sight (LOS) and Shadow) [4], three states (LOS, Shadow, Heavy Shadow/Blockage) [5] or even recently a two states semi Markov model [3]. In the following, the three-state model [5] is retained according to ITU-R recommendation [6] for performance evaluations.

The model makes the assumption that the received signal is composed of the sum of two components : the direct signal and the multipath component.The direct signal is assumed to be Log-Normally distributed with a mean (relative to LOS) and a standard deviation, while the multipath component is considered as random and characterized by its average power (dB relative to LOS). The model is based on the assumption of the existence of three distance scales (very slow, slow and fast) in the received signal corresponding to different phenomena.

Statistical parameters of this model depend on the environment and the elevation angle. Based on measurements, reference [5] has defined matrices for four environments : Open, Intermediate tree shadowed, Heavy tree shadowed, Suburban and Urban and for elevation angles above 40°.

2.2. *Key issues in DVB-SH systems*

DVB-SH systems aim at providing TV and audio broadcasting services for handheld terminals. In the context of a constrained satellite budget link, two main performance issues have to be handled: the first one is the availability and QoS of the system often characterized by means of the ESR5 criterion and the second one is to ensure an acceptable zapping time for the end user. The use of long traditional interleavers enhances the availability and QoS of the system but highly degrades the zapping time. For this reason, DVB-SH introduces advanced interleaving solutions designed to enhance statistical zapping times, allowing fast zapping times in good reception conditions and longer zapping times when reception conditions are bad.

2.3. *Description of DVB-SH interleavers*

DVB-SH standardizes convolutional interleavers acting on Interleaver Unit (IU) made of 126 bits. Each interleaver is characterized by five parameters given in [8]: (Nof_late_taps, Common_multiplier, Nof_slices, Slice_distance, Non_late_increment). Three classes of interleaving parameters setting proposed in [8] are considered

- short uniform interleaver (5,48,1,0,0): spreading between 100 and 240 ms,
- long uniform interleaver (40,0,12,4,2): spreading between on 6.4 s and 15.6 s,
- long uniform late interleaver (10,24,9,5,12): spreading between 12.8 s and 31.2 s.

2.4. *Propagation channel variation in a codeword*

Depending on the physical layer parameters and the interleaver choice, a given codeword will be spread from 100 ms to 31.2 s. Because of the LMS channel, several output IU bits of a same codeword will be affected by different attenuations as illustrated in Figure 1. This Figure illustrates how an input codeword is spread on the LMS channel. The realizations of this channel were obtained with the Perez Fontan Model for elevation of $40°$ in a suburban environment and for a mobile speed of 60 km/h.

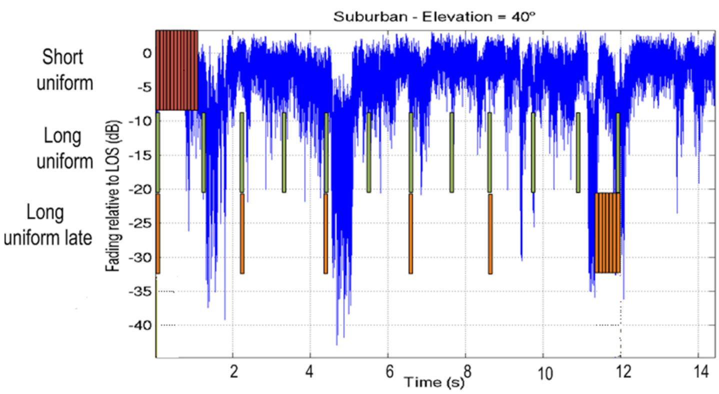


Figure 1. Illustration of the interleaving/slice effect on the LMS channel

The consequence of the use of the previous interleavers with the LMS channel is that predicting performance of DVB-SH system can equivalently be modeled by predicting the performance of codewords with time varying channel SNR according to Figure 2.

2.5. *Performance criteria retained: PER, ESR5 and zapping time*

2.5.1. PER estimation Time varying values inside a codeword are caused by varying attenuations. The main issue is to assess whether we are able to predict PER performance of such a codeword. The answer depends on the distribution of the fading. When this distribution is stationary according to a simple law (constant distribution, Rayleigh distribution..), it is possible to derive a theoretical

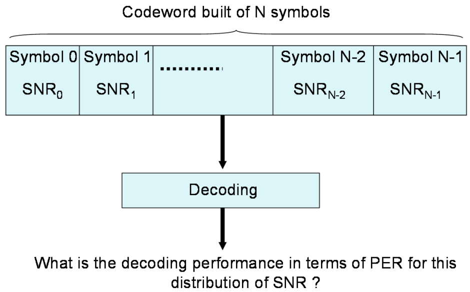


Figure 2. Formulation of PER estimation problem

expression of the PER. However, in the context considered here, interleaving and time slicing leads to a non stationary distribution because of the strong variations of the channel. A theoretical expression of the PER can not be derived. Monte-Carlo simulations are thus the only solution to obtain these values. However, as explained in Section 7, such simulations are prohibitively time consuming.

2.5.2. ESR estimation ESR5 (Erroneous Seconds Ratio 5%) is another fundamental performance indicator in DVB-SH systems, which is more closely related to the final Quality Of Service than PER. More generally, ESRX is defined as the ratio of 20 seconds windows in which the number of erroneous seconds is smaller than X percent. Actually, for DVB-SH services, ESR5 has been shown to be the most suitable criterion for quality characterization at system level [8]. As a consequence, we need for system dimensioning to compute the mean ESR5 value to characterize DVB-SH performances depending on the system margin, the environment, and the physical layer characteristics (interleaver, modulation, . . .). Similarly to PER, an ESR5 theoretical expression can not be derived and the other alternative is to perform simulations. In order to plan such simulations, it is important to note that performance convergence for the LMS channel is rather low and requires to represent long traveled distances (see section 6.2). Such physical layer simulations implementing the coding and decoding processes are extremely time consuming and can not realistically be performed for sufficient distances.

2.5.3. Zapping time estimation Since the main application of DVB-SH is video streaming, the delay observed by a user when he joins a channel or when he switches between two channels is a key parameter. This delay, called *zapping time*, is generally lower than 2 s for terrestrial transmissions [8], however, the use of long interleavers for the satellite component can drastically extend it. There is a trade-off between the robustness, which requires a long interleaver depth and the zapping time, which requires a short depth. This analysis of the zapping time delay must be then carefully done because it directly impacts the QoS perceived by the user.

3. PREDICTION METHODS PRESENTATION

This Section aims to address the performance of the system when a codeword is affected by different levels of SNR. In this section, after proposing the general framework of the prediction methods, we expose two standard methods for performance prediction. We then present the innovative mutual information method, previously confined to the context of terrestrial MIMO system, which appears to be very relevant in our mobile satellite context. Finally, a theoretical comparison of the three methods in the case of a QPSK modulation is provided.

3.1. Generic formulation of the prediction methods

All the bibliographic methods for PER performance analysis share a common approach based on the notion of equivalent constant SNR denoted in the following as SNR_{gauss}. They assume that the performance of a codeword subjected to different levels of SNR_i (for each symbol $0 \leq i \leq N-1$) is given by the performance for a Gaussian Channel with equivalent constant SNR denoted as SNR_{gauss}. SNR_{gauss} can be expressed in a generic way as a function of the successive levels of SNR as follows

$$SNR_{gauss} = F^{-1}\left(\frac{1}{N}\sum_{i=0}^{N-1} F(SNR_i)\right) \tag{1}$$

where F is the Link Quality metric (LQM) function. This approach has the advantage to be able to compute, for example, the PER on a varying channel using the performance curves $PER = G(SNR)$ for a Gaussian channel.

3.2. Standard link quality metrics

We consider in the following that received complex symbol y_k (k-th symbol in the considered codeword) is obtained at the output of the channel according to

$$y_k = \rho_k x_k + n_k \tag{2}$$

where x_k is the complex emitted symbol (k-th symbol of the codeword), ρ_k is the real and positive attenuation of the channel (we make the assumption that phase rotation is perfectly corrected at reception) that affects x_k and n_k is the additive Gaussian Noise with Power Spectral Density (PSD) $\frac{N_0}{2}$ on the independent imaginary and real parts that affects x_k.

3.2.1. Mean SNR as LQM The most intuitive method involves considering the mean SNR [7] over the codeword. In the definition, Eq. (1), this method leads to consider F as the identity function and the equivalent SNR_{gauss} is given by

$$SNR_{gauss} = \left(\frac{1}{N}\sum_{k=0}^{N-1} SNR_k\right) \tag{3}$$

where SNR_k is the SNR of the k-th symbol of the considered codeword.

3.2.2. BER as LQM A second method involves considering the BER on each coded symbol before decoding. For a given modulation and coding scheme, the BER is a function of the SNR through the use of the function Q. If we denote $BER(SNR)$ as the general BER function of the channel, the equivalent SNR_{gauss} is given by

$$SNR_{gauss} = BER^{-1}\left(\frac{1}{N}\sum_{k=0}^{N-1} BER(SNR_k)\right) \tag{4}$$

where SNR_k is the SNR of the k-th symbol of the codeword.

3.3. Mutual Information approach

The mutual information is used to predict physical layer performance in some wireless terrestrial networks [15]. These scenarii are very different from those in the satellite mobile DVB-SH context where the codewords are distributed over a large time period on a highly varying channel. Our proposal in this paper is to evaluate and adapt this method to the LMS context.

If we consider the mutual information between emitted and received symbols, the equivalent SNR_{gauss} of the codeword is given by:

$$SNR_{gauss} = I^{-1}(\frac{1}{N}\sum_{k=0}^{N-1} I(SNR_k)) \tag{5}$$

where I is the mutual information between emitted symbol X and received symbol Y computed according to the general formula

$$I(X, Y) = H(Y) - H(Y/X) \tag{6}$$

where $H(U)$ is the entropy of random variable U defined by:

$$H(U) = -\int f(u)\log_2(f(u))du \tag{7}$$

where $f(u)$ is the probability density of the random continuous variable U. The derivation method of the mutual information is detailed in the appendix for QPSK modulation.

3.4. Theoretical comparison between the three methods for QPSK

Here we propose to theoretically derive the previous expressions in case of QPSK modulation. As SNR indicator, we choose $\frac{E_b^c}{N_0}$ the ratio between the energy per coded bit and the noise PSD. We assume that simulations provide us with performance curve $PER = G(\frac{E_b^c}{N_0})$ for a AWGN channel, for QPSK modulation and a code of rate R.

3.4.1. Expression of the theoretical $\left.\frac{E_b^c}{N_0}\right]_{Gauss}$ **mean SNR method:**

In this method, function F is explicit since it is the identity function and the PER of the considered codeword is given by

$$PER_{mean} = G\left(\left.\frac{E_b^c}{N_0}\right]_{Gauss}^{mean}\right) \tag{8}$$

with

$$\left.\frac{E_b^c}{N_0}\right]_{Gauss}^{mean} = F_{mean}^{-1}\left(\frac{1}{N}F_{mean}\left(\sum_{k=0}^{N-1}(\rho_k)^2\ \frac{E_b}{N_0}\right]\right)\right) \tag{9}$$

and $F_{mean}(x)$ is the identity function, *i. e.* $F_{mean}(x) = x$.

BER channel method:

In case of QPSK, BER is given by $Q\left(\sqrt{\frac{2E_b^c}{N_0}}\right)$ with $Q(x) = \frac{1}{\sqrt{2\pi}}\int_x^{\infty} e^{-\frac{t^2}{2}}\,dt$. Therefore, the PER_{BER} of the codeword is given by

$$PER_{BER} = G\left(\left.\frac{E_b^c}{N_0}\right]_{Gauss}^{BER}\right) \tag{10}$$

where

$$\left.\frac{E_b^c}{N_0}\right]_{Gauss}^{BER} = F_{BER}^{-1}\left(\frac{1}{N}F_{BER}\left(\sum_{k=0}^{N-1}(\rho_k)^2\ \frac{E_b}{N_0}\right]\right)\right) \tag{11}$$

and

$$F_{BER}(x) = Q(\sqrt{2x}). \tag{12}$$

Mutual Information method:

A derivation of the theoretical expression of the Mutual Information $I(\frac{E_b^c}{N_0})$ between emitted and received symbols in case of a QPSK modulation is found in the appendix. The PER_{MI} of the codeword is therefore given by

$$PER_{MI} = G\left(\left.\frac{E_b^c}{N_0}\right]_{Gauss}^{MI}\right) \tag{13}$$

where

$$\left.\frac{E_b^c}{N_0}\right]_{Gauss}^{MI} = F_{MI}^{-1}\left(\frac{1}{N}F_{MI}\left(\sum_{k=0}^{N-1}(\rho_k)^2\ \frac{E_b}{N_0}\right]\right)\right) \tag{14}$$

and $F_{MI}(x)$ is the mutual information curve, *i. e.* $F_{MI}(x) = I(x)$.

Figure 3 illustrates functions $F_{mean}(\frac{E_b}{N_0})$, $F_{BER}(\frac{E_b}{N_0})$ and $F_{MI}(\frac{E_b}{N_0})$ for $\frac{E_b}{N_0}$ in a logarithmic scale.

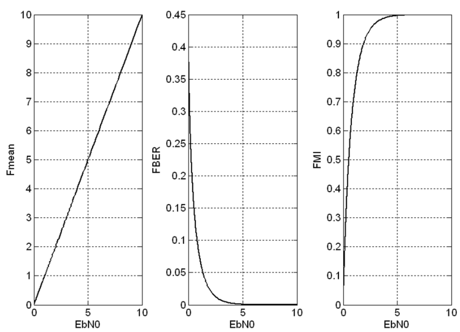


Figure 3. Functions F for mean SNR method, BER method and Mutual Information method

3.4.2. Convexity consideration of the inverse functions A function f is convex if it fulfills, for any set of P positive real α_k and real x_k for $0 \leq k \leq P-1$ such that $\sum_{k=0}^{P-1} \alpha_k = 1$,

$$f\left(\sum_{k=0}^{N-1} \alpha_k x_k\right) \leq \sum_{k=0}^{N-1} \alpha_k f(x_k) \tag{15}$$

Functions F_{MI}^{-1} and F_{BER}^{-1} are shown to be convex. This convexity property is graphically illustrated by Figure 4, even if not demonstrated through a theoretical approach. Using convexity inequality

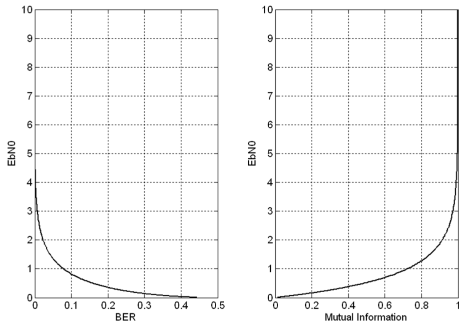


Figure 4. Illustration of the convexity of F_{MI}^{-1} and F_{BER}^{-1}

(15) for function F_{MI}^{-1} in equation (14) leads to

$$\left.\frac{E_b^c}{N_0}\right]_{Gauss}^{MI} \leq \frac{1}{N}\left(\sum_{k=0}^{N-1} (\rho_k)^2 \left.\frac{E_b}{N_0}\right]\right). \tag{16}$$

But $\frac{1}{N}\left(\sum_{k=0}^{N-1} (\rho_k)^2 \left.\frac{E_b^c}{N_0}\right]\right)$ is exactly $\left.\frac{E_b^c}{N_0}\right]_{Gauss}^{mean}$. For the same reasons, we get

$$\left.\frac{E_b^c}{N_0}\right]_{Gauss}^{BER} \leq \frac{1}{N}\left(\sum_{k=0}^{N-1} (\rho_k)^2 \left.\frac{E_b}{N_0}\right]\right). \tag{17}$$

A theoretical expression of F_{MI}^{-1} is not available, but graphical considerations seem to show that convexity is uniformly stronger for F_{BER}^{-1} than for F_{MI}^{-1}. This remark leads to the following set of inequalities for the three methods:

$$\left.\frac{E_b^c}{N_0}\right]_{Gauss}^{BER} \leq \left.\frac{E_b^c}{N_0}\right]_{Gaus}^{MI} \leq \left.\frac{E_b^c}{N_0}\right]_{Gauss}^{mean}. \tag{18}$$

4. PERFORMANCE COMPARISON OF THE DIFFERENT PREDICTION METHODS

We propose now to evaluate prediction performance for the three previous methods on arbitrary fading sequences of one codeword length in the case of a QPSK 1/3 turbocoded DVB-SH transmission [2].

The mean BER and mean PER are simulated for the different channels considered through Monte-Carlo simulations performed by repeating the attenuation sequence on the successive codewords and calculating the mean BER or the mean PER. These reference BER or PER performances are then compared to prediction results obtained relying on the three previous methods and using as reference the BER and PER turbocode Gaussian performance curves.

4.1. *Fading sequences applied*

For a varying channel, the receiver observes a varying SNR (respectively $\frac{E_b}{N_0}$ and $\frac{E_s}{N_0}$) that can be decomposed into an attenuation (*i.e.* a fading) term, and a fixed contribution that we call in the following nominal SNR (respectively, nominal $\frac{E_b}{N_0}$ and nominal $\frac{E_s}{N_0}$) and which we denote SNR_n (respectively, $\left.\frac{E_b}{N_0}\right]_n$ and $\left.\frac{E_s}{N_0}\right]_n$).

When the attenuation is equal to zero, $\frac{E_b}{N_0}$ is equal to $\left.\frac{E_b}{N_0}\right]_n$. The value of $\left.\frac{E_b}{N_0}\right]_n$ is represented on the abscissa axis on the performance curves whereas the successive attenuation values are considered fixed for a given simulation channel. The three different sequences of attenuation are presented in Figure 5.

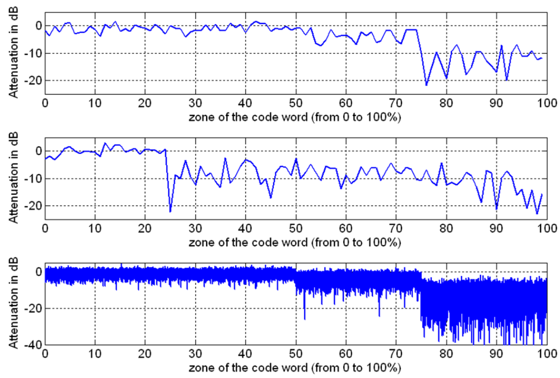


Figure 5. The 3 channel realizations for the coding rate 1/3

4.2. *Evaluation of the BER prediction methods*

Figure 6 represents the BER predictions using the different methods in terms of $\left.\frac{E_b}{N_0}\right]_n$. It shows how mutual information method provides an excellent BER estimation for both coding rates for the different channels. In addition, the relative position of the BER curves is coherent with inequality (18).

4.3. *Evaluation of the PER prediction methods*

Figure 7 represents the PER predictions using the different methods in terms of $\left.\frac{E_b}{N_0}\right]_n$ for the three channels.

It shows how mutual information method provides an excellent estimation of the PER estimations for the three channels for the coding rate 1/3. A slight shift of about 0.1 dB appears in the PER estimation with the mutual information method. Complementary simulations (not presented here) tend to show that this offset only depends on the modulation and coding scheme, and could then be

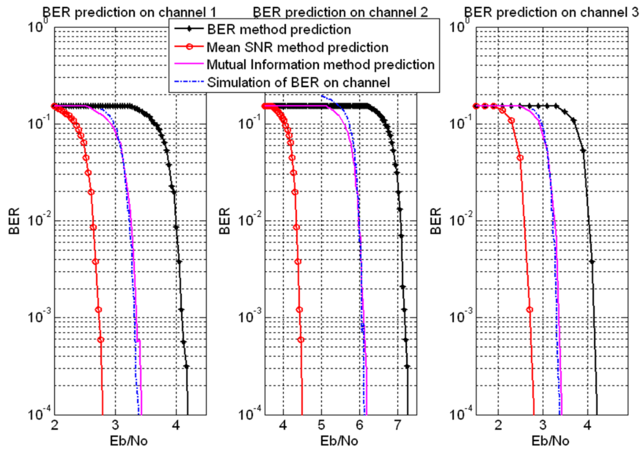


Figure 6. BER predictions for coding rate 1/3

corrected independently of the propagation channel. We have compensated the residual offset in the following simulations using an offset equal to 0.09 dB for QPSK 1/3.

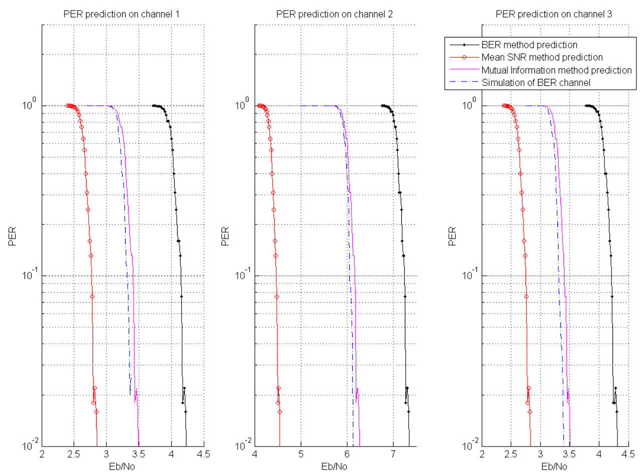


Figure 7. PER predictions for coding rate 1/3

4.4. *Conclusion on the comparison of prediction methods*

Based on these results it appears clearly for QPSK turbo-coded transmission following the DVB-SH standard that the prediction method proposed in this paper based on the mutual information parameter provides predictions better than 0.1 dB for both the mean BER and mean PER performance calculation. It also appears that more traditional methods like those based on mean SNR or the mean channel BER are less accurate and represent respectively lower and upper performance bounds. All these results were confirmed by complementary simulations carried out for 1/2 coding rate (not presented here).

We propose then to focus in the following on the mutual information prediction method and to go deeper into its validation.

5. MUTUAL INFORMATION METHOD VALIDATION FOR A LMS CHANNEL

We propose in this section to complete the evaluation and validation of the mutual information method for a more realistic DVB-SH transmission through the LMS channel.

5.1. *Physical layer parameters and propagation channel*

We consider the parameters in Table I (from [8]), corresponding to a QPSK modulation with a $1/3$ rate turbo code. A realization of this LMS Intermediate Tree Shadowing (ITS) [5] channel will be

Waveform	QPSK1o3_U
Channel configuration	LMS-ITS
Speed (km/h)	50
Common_multiplier	40
Nof_late_taps	0
Nof_slice	12
Slice_distance	4
Non_late_increment	2
Coding rate	1/3
nominal C_N	5

Table I. Value of physical parameters

used in the following simulations.

5.2. *PER simulations*

We propose here to compare the estimated PER using Monte-Carlo method, and the predicted PER obtained using the mutual information prediction method.

The estimated PER corresponds to an averaging of 10 Monte-Carlo simulations performed on the same channel. This channel corresponds to 16283 codewords. The codewords are subjected to a fixed attenuation sequence whereas each of the ten simulations is done with a different random seed noise. The $PER_{est}(i)$ of the $i-th$ codeword is estimated using

$$PER_{est}(i) = \frac{1}{10}\sum_{j=1}^{10} X_i^j \tag{19}$$

where X_i^j is the random variable that equals to 1 if the $j-th$ realization of the $i-th$ codeword is wrong (the packet is not perfectly decoded) and equals to 0 if the decoded packet is correct.

The predicted PER $PER_{pred}(i)$ on the $i-th$ codeword relies on mutual information and is computed for the same channel.

Figure 8 presents the evolution of the estimated PER, the predicted PER and the error between them, considering a sliding window of 3 successive PER values. Zooming in an interesting section of the plot (Figure 8), we see on these two figures that the mutual information method leads to a very efficient prediction for packet error rate evolution. Figure .

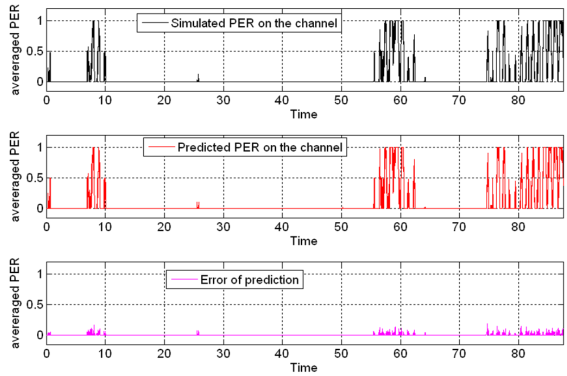


Figure 8. Estimated and predicted averaged PER

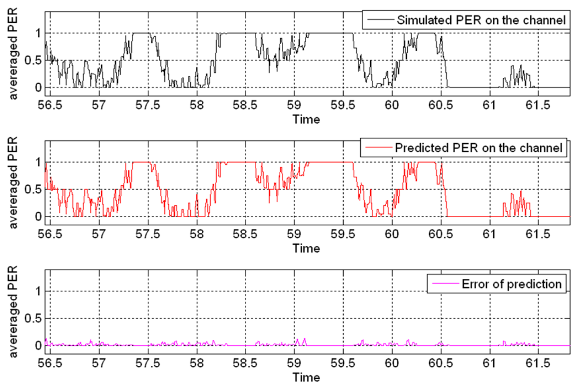


Figure 9. Zoom for estimated and predicted averaged PER

5.3. *ESR5 simulations*

To perform an ESR5 estimation, an additional step is necessary using PER predictions. Traditionally, ESR5 was computed from traces provided by physical layer simulations. We describe hereafter how to compute ESR5 directly from PER predictions.

We denote by $PER_{i,j}$ the predicted PER of the $j-th$ codeword of the $i-th$ second. Making the assumption that the PERs of two consecutive seconds are independent and that L is the number of codewords per second, we compute first the probability that the $i-th$ second is errorless , P_i, by

$$P_i = \prod_{j=1}^{L}(1 - PER_{i,j}). \tag{20}$$

Based on this error-free second probability, we calculate the number of error-free seconds on the $k-th$ window with duration 20 s (time between k and $k+19$). This value is represented by the random variable Y_k given by

$$Y_k = \sum_{m=k}^{k+19} A_m \tag{21}$$

where A_m is a Bernoulli random variable that equals 1 if the $m-th$ second is error-free (with probability P_m defined in (20)) and A_m equals 0 if the the $m-th$ second is false (with probability $1-P_m$). Making the assumption that the A_m variables are independent, we can consider that their sum is distributed according to the convolution of Bernoulli distributions. If we denote F_k as the cumulative distribution function of the random variable Y_k, we obtain the theoretical expression of ESRX as

$$(ESRX)_k = P(Y_k \geqslant 20X) = 1 - F_k(20X). \tag{22}$$

Figure 10 presents the evolution of ESR5 computed from estimated PER and predicted PER. We

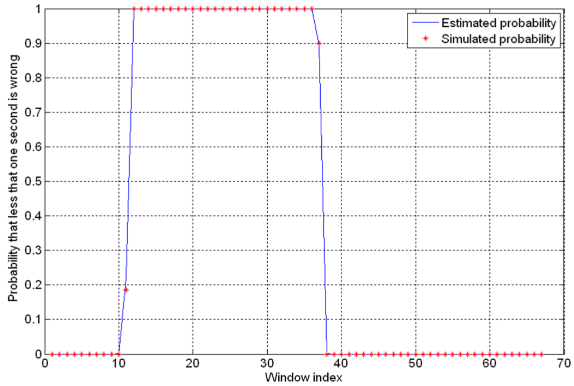


Figure 10. ESR prediction and estimation

can see on this figure that for this channel, ESR5 computed from estimated PER and predicted PER are identical.

5.4. Conclusion

This completes the validation results obtained by analyzing the evolution of PER and ESR5 for a realistic LMS propagation channel. The results show an excellent convergence between Monte-Carlo simulations and the mutual information prediction method. The proposed method has been validated for the purpose exposed in the beginning of the paper, that is, a large scale performance evaluation in terms of QoS for DVB-SH transmissions using long interleavers for realistic LMS propagation channels.

6. EXPLOITATION OF THE METHOD FOR DVB-SH AIR INTERFACE TRADE-OFF

We propose now to exploit the above validated prediction method to compute ESR and zapping time performance in terms of the system margin for three kinds of interleavers.

6.1. Description of the parameters

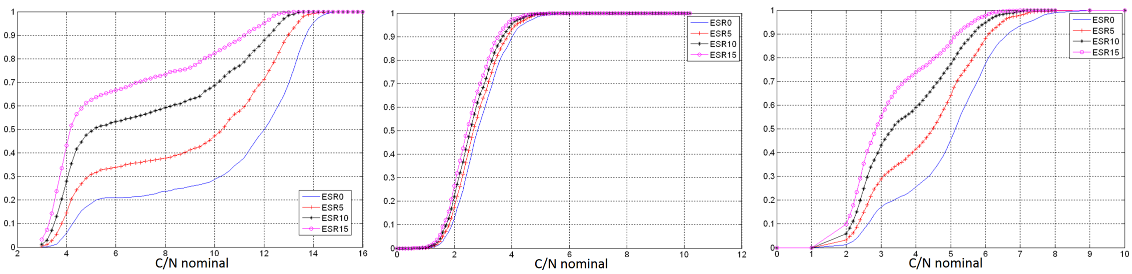
We use the following parameters for the simulation. We propose to use the short uniform, long uniform and long uniform late interleavers whose parameters are given in Section 2.3.

Channel		Physical layer	
Mode	Suburban	**Mode**	OFDM/2K
Elevation	40°	**Modulation**	QPSK
Speed	50 km/h	**Band**	5MHz
		Coding rate	1/3

Table II. Parameters of the scenario

6.2. *Simulation results for ESR performance*

6.2.1. ESR5 computation according to system margin For each value of $\frac{C}{N}\big]_n$, we simulated a 100 km channel run and performed the PER prediction. Using this prediction of PER, we computed the ESR5 for the three considered interleavers, as illustrated in Figure 11. These curves are of



(a) short uniform interleaver (b) long uniform interleaver (c) long uniform late interleaver

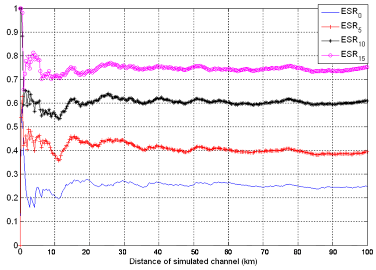
Figure 11. ESRX according to the $\frac{C}{N}]_n$ for the three interleavers

paramount interest for DVB-SH system dimensioning to establish the necessary system margin for a given service availability. The simulation durations with traditional Monte-Carlo simulations (see Section 7) made them very difficult to obtain in the past and explain the lack of similar extensive results to the knowledge of the authors.

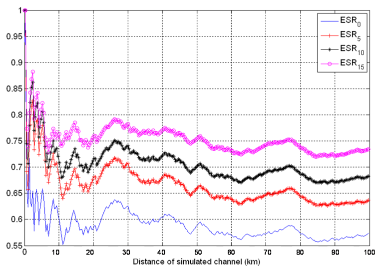
Besides, the curves give interesting elements to position the different interleavers in terms of performance. In Figure 11a, we see how the short interleaver is able to overcome very fast fades (strong slope of the curve between 3 and 5 dB) but does not overcome the fast and slow fades (weak slope of the curves between 5 and 10 dB) because it is too short. Only a strong value of $\frac{C}{N}\big]_n$ enables to get a satisfying value of ESR5. In Figure 11b, we see how the long uniform interleaver mitigates all fades (uniform slope between 3 and 5 dB). Finally, Figure 11c shows that the behavior of uniform late interleaver is a mix of the two previous behaviors.

Finally we can see that for an ESR5 target of 90% (coherent with the expected system availability in ITS environments), the required $\frac{C}{N}\big]_n$ in LOS conditions is 13 dB for the short uniform interleaver, 3.8 dB for the long uniform interleaver, and 6.1 dB for the long uniform late interleaver showing a very good efficiency in the case of long interleavers at 50 km/h. These results show typical outputs that can be obtained with the proposed prediction approach, and represent new and important entries for system performance evaluation and system design. Clearly, these results, which concern a particular case of environment and speed, can be obtained for complementary scenarios.

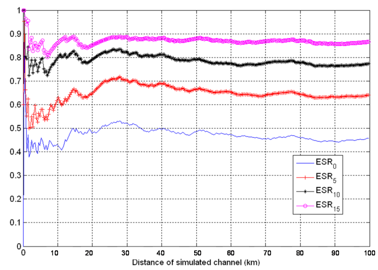
6.2.2. ESR5 convergence results Figure 12 presents convergence results obtained for the three different interleavers with $\frac{C}{N}\big]_n = 8.6$ dB. We can underline that these convergence figures confirm



(a) short uniform interleaver



(b) long uniform interleaver



(c) long uniform late interleaver

Figure 12. Convergence of ESRX for the three interleavers

that a minimum distance simulation of about 15 km is required in order to ensure the convergence of the ESR as it was assumed in Section 2.5.2.

6.3. Results of simulation for zapping time performance

As explained in Section 2.5.3, zapping time is one of the key issues of the DVB-SH standard [2]. Practically, when a user joins a channel at a given time, if a long interleaver is being used, he will receive the final part of codewords whose first parts were sent before his arrival time. The receiver then tries to decode codewords with only this final part. For each new data burst, it receives an increasing part of the codewords and, after a time equal to the interleaving depth, the considered codewords are complete. To have a short zapping time, the user must be able to decode incomplete codewords. It follows that the global performance in terms of zapping time depends directly on the decoding probabilities of incomplete codewords. The correspondence between the zapping time and the decoding of incomplete codewords is illustrated in Figure 13. In this figure, we assume that a user joins the channel t seconds before the end of the transmission of a codeword (sent with a uniform late interleaver). We consider that the zapping time is equal to the minimum t allowing a successful decoding. To evaluate the decoding probabilities of incomplete codewords,

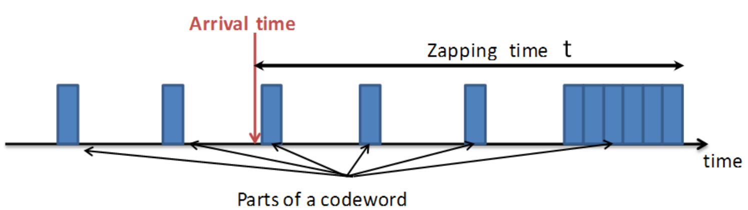


Figure 13. Zapping time and decoding of incomplete codeword

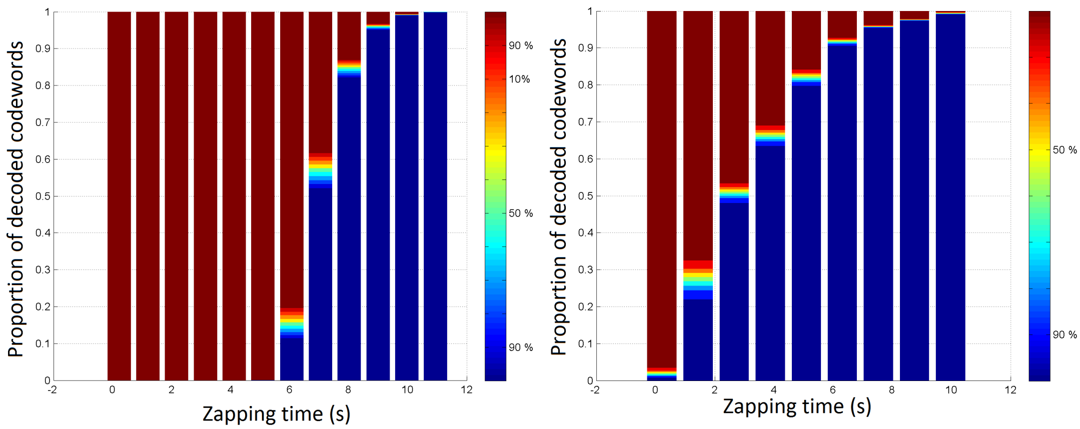
we considered a transmission on a suburban channel with long interleavers described in section 6.1 with $\frac{C}{N}]_n = 4$ dB, and for each codeword, we estimate the decoding probability. Note that the scenario using short interleaver was not studied here because zapping time is not an issue in this case.

This estimation can also be done with the mutual information method. Indeed, the non-received parts of a codeword brings a mutual information equal to zero, the decoding is possible if the received parts of the codeword bring sufficient mutual information for the whole codeword. An illustration is shown in Table III. The first two rows extracted from Table 7.8 of [8] give the required $\frac{C}{N}]_n$ for received data to be decoded successfully when a part of the data is erased. The third row indicates the corresponding required mutual information per received bit and the fourth row gives the corresponding required mutual information per transmitted bit (*i. e.* the global received mutual information averaged over all bits -received or erased- of a codeword). It is noteworthy that this global mutual information is quasi-constant whatever the proportion of erased data. The results are

IU received	288	264	240	216	192	168	144
req. C/N	-0.9	0.0	0.6	1.4	1.8	3.0	4.4
req. MI per rec. bit	0.4	0.46	0.50	0.57	0.60	0.69	0.8
req. MI per transm. bit	0.39	0.41	0.41	0.42	0.39	0.40	0.39

Table III. Required mutual information per codeword bit

presented in Figures 14a and 14b for, respectively, long uniform and long uniform late interleavers. In these figures, the interleaver depth is split into time ranges corresponding to 1 second. For each second, we give the proportion of codewords that can be decoded with a given probability. Figure 14a shows that the long uniform interleaver can only decode the first codewords after 6



(a) long uniform interleaver

(b) long uniform late interleaver

Figure 14. Decoding probability for incomplete codeword with long interleavers on suburban channel with a nominal $\frac{C}{N}]_n = 4$dB

seconds. Ideally, on a noiseless channel, since the code rate is $1/3$, a codeword could be decoded as soon $1/3$ of its coded bits are received. Since the long uniform interleaver uniformly spreads the codeword over 12 seconds, a codeword can be decoded after 4 seconds. The result obtained here is not surprising because the codewords transmitted under low noise conditions are decoded when half of the codeword is received, *i. e.* after 6 seconds. The very good property of the long uniform late interleaver is that it provides full reliability after 10 seconds. The uniform distribution of the codeword over time is indeed optimal in terms of correction capability. The behavior of the long uniform late interleaver (Figure 14b) is very different. Since half of the codeword is sent on the same second, some codewords can be decoded at the first second. After 5 seconds, more than 80% of the codewords are decoded. The counterpart of this good zapping time is that, in the studied case ($\frac{C}{N}]_n = 4$ dB), this interleaver does not reach the full reliability (100% of decoded codeword). However, if we consider the trade-off between reliability and zapping time in the global case, the long uniform late interleaver appears to be the best solution between the two long interleavers.

7. COMPLEXITY ANALYSIS

7.1. Complexities of parameter estimation

7.1.1. PER estimation An alternative solution to obtain a PER estimation of the DVB-SH coding and interleaving performance for a given channel is to carry out simulations of the physical layer. Given a realization of the channel of duration d, let us assume that we want to get an estimation of the PER of each codeword of the DVB-SH system for this channel. Since PER is a statistical value, Monte-Carlo methods are suitable for obtaining this value. This method consists in generating a large number of realizations of the same codeword and after decoding these realizations, PER estimation is obtained by computing the ratio of wrong decoded codewords. Generation of the codeword is continued until the estimation of the PER converges. Classically, the number of generations of the same codeword is chosen to be greater than $100/PER_{theoretical}$ where $PER_{theoretical}$ is the theoretical value of PER.

As an example, for the particular case of a QPSK with 1/3 coding rate and OFDM modulation in 8k mode with 1/4 guard interval, the duration of a SH frame is about 150 ms and the duration of a codeword is about 3.3 ms. As a consequence, for a channel with duration d seconds, the number of codewords is about $d/3.3 * 10^3$. Assuming that we aim at estimating PER performance on a channel with duration 60 s, this leads to about 18000 codewords. With the Monte-Carlo method for every codeword, it leads to a minimal number of 1800000 codewords to be simulated.

7.1.2. ESR estimation ESR5 can be computed from the PER estimation as explained in Section 5.3. The necessary length of simulated channel to ensure convergence of the ESR5 value depends on the parameters of margin, environment, and physical layer as illustrated in Section 6.2.2, but it appears that 15 km is a good order of magnitude. As a consequence, with the same numerical values as in the previous example, if we denote by v the mobile speed (in km/h), the number of codewords during this time equals $15/v * 3600 * 10^3/3.3$. Numerical examples for $v = 50km/h$ and $v = 3km/h$ lead to a number of transmitted codewords of 330 000 and 5 450 000. With the Monte-Carlo method, it leads to a minimum number of $3.3 * 10^7$ and $5.45 * 10^8$ codewords to be simulated. It is obvious that these simulations require prohibitive computation time. In addition, all these operations only provide a single value of ESR5. But, the output we are interested in is the value of ESR5 in terms of the system margin. As a consequence this process has to be reiterated a large number of times for every value of system margin of interest.

7.1.3. Zapping time estimation As explained in Section 2.5.3, zapping time is evaluated through the decoding probabilities of incomplete codewords. The number of codewords that must be evaluated is thus similar than that for PER estimation, but for each codeword, several decoding attempts are done. For example, in the results presented previously, a decoding is attempted each time a new part of the codeword is received, *i. e.* each second. Like for PER, performing Monte-Carlo-based estimations is not realistic.

7.2. Comparison of simulation times

Two physical layer DVB-SH receiver simulators were developed. The input of the two simulators is a sequence of fading values generated with a propagation channel simulator developed at CNES (and adopted by the DVB-SSP group for the definition of the DVB-SH standard [2]).

The first simulator, called "full simulator", implements deinterleaving, demodulation and turbo decoding operations. The second one, called "simplified simulator", implements de-interleaving operations and replace the demodulation/turbo decoding operations by the decoding prediction method based on mutual information. As shown in Sections 3 and 5, simulation results provided by the two simulators in terms of PER are extremely similar.

On the contrary, if we consider the execution times, the results are extremely different. For example, for the scenario corresponding to Table I with a short interleaver, the full simulator needs 239 minutes to simulate a distance of 3000 meters on a Penthium D 2.8 GHz with 1 GB of RAM. For the same scenario, the simplified simulator only needs 16 seconds on the same computer. The gain factor is about 800. Even if the full simulator can be optimized, we can considered that a factor of about 500 is a good approximation of the gain. In light of this comparison, it is clear that the results presented in Figures 11 or 14 could not be obtained without this gain. Note that, despite the interest of these results, to the knowledge of the authors, this kind of results have not yet been published or available in any public documents.

8. CONCLUSION

The purpose of this paper is to evaluate a new method for performance assessment of the DVB-SH standard. We point out that a theoretical evaluation of performance is not achievable because of the non stationary distributions of the SNR within a codeword. Moreover, Monte-Carlo simulations proved to be time prohibitive, especially for ESR evaluation. The proposed prediction method relies on the computation of mutual information between emitted and received symbols. Reliability of this method is first evaluated on arbitrary channels with non stationary distribution of noise variance and compared to two other methods, namely the mean SNR method and the mean BER method. These simulations show that the mutual information method exhibits accuracy better than 0.1 dB. Then, a more complete validation is proposed by simulating the DVB-SH physical layer for the LMS channel. Results confirmed the very good precision of this prediction method. Finally, we present possible outputs of a simulator based on this method that was shown to be a key tool for future mobile satellite system dimensioning by providing ESR5 and zapping time precise statistic estimations for representative mobile route lengths.

ACKNOWLEDGMENTS

The authors would like to thank Emmanuel Lochin, Guillaume Jourjon and the reviewers for their careful reading and their helpful comments.

9. APPENDIX

9.1. *Derivation of the mutual information for QPSK modulation*

We first consider a BPSK modulation. Given a realization of the emitted BPSK symbol random binary variable x and a realization of the noise n, assuming perfect knowledge of the channel attenuation ρ, the continuous random variable y of output symbol is given by

$$y = \rho\sqrt{E_b}x + n. \tag{23}$$

The values that the random variable x can take are -1 and 1 and the noise variable n has an independent Gaussian distribution with variance $\frac{N_0}{2}$.

For convenience, we consider y' given by

$$y' = \frac{\rho\sqrt{E_b}}{N_0}y = \rho^2\frac{E_b}{N_0}x + n\frac{\rho\sqrt{E_b}}{N_0}. \tag{24}$$

Variable y' has a bimodal Gaussian distribution whose continuous probability density is given by

$$\begin{aligned} f(y') &= f(y'/x=-1)P(x=-1) + f(y'/x=1)P(x=1) && (25)\\ &\rightarrow \frac{1}{2}N(-\rho^2\frac{E_b}{N_0}, \rho^2\frac{E_b}{2N_0}) + \frac{1}{2}N(\rho^2\frac{E_b}{N_0}, \rho^2\frac{E_b}{2N_0}) && (26)\end{aligned}$$

where $N(m, \sigma^2)$ is a Gaussian distribution with mean m and variance σ^2. In the following we denote $\rho^2\frac{E_b}{N_0}$ as α and $\rho^2\frac{E_b}{2N_0}$ as β^2.

Since y' has the same information as y (they only differ by a multiplication by a constant), the mutual information between random variable x and y equals the mutual information between random variables x and y' and it is given by

$$I(x, y') = H(y') - H(y'/x) \tag{27}$$

where $H(U)$ is the entropy of the continuous random variable U defined by

$$H(U) = -\int_{-\infty}^{\infty} f(u)\log(f(u)) \tag{28}$$

where $f(u)$ is the probability density of U.

In (27), y'/x is a Gaussian random variable with variance β^2. It is well known that the entropy of such a random variable only depends on the variance and is given by:

$$H(y'/x) = \frac{1}{2}\log_2\left(2e\pi\beta^2\right). \tag{29}$$

In addition, using (26) and (28), we have for $H(y')$

$$H(y') = -\int_{-\infty}^{\infty} G(u)\log(g(u))du \tag{30}$$

where $G(u)$ is defined by

$$G(u) = \frac{1}{2\sqrt{2\pi\beta^2}}\left(\exp\left(-\frac{(u-\alpha)}{2\beta^2}\right) + \exp\left(-\frac{(u+\alpha)}{2\beta^2}\right)\right). \tag{31}$$

Expression (30) can not be expressed into a closer form and requires using numerical integration to calculate it.

As a consequence, $I(x, y')$ only depends on $\alpha = \rho^2\frac{E_b}{N_0}$ and $\beta^2 = \rho^2\frac{E_b}{2N_0}$ and it is therefore only a function of $\rho^2\frac{E_b}{N_0}$. Figure 15 plots this function in terms of $\rho^2\frac{E_b}{N_0}$ for a BPSK. In case of QPSK modulation, because of the independence of the imaginary and real parts, the function is the same.

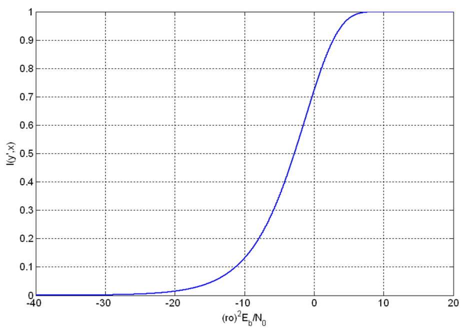


Figure 15. Mutual information for QPSK/BPSK modulation

REFERENCES

1. ETSI. System Specifications for Satellite services to Handheld devices (SH) below 3 GHz, TS 102 585.
2. ETSI. Framing Structure, channel coding and modulation for Satellite Services to Handheld devices (SH) below 3 GHz, EN 302 583.
3. Burzigotti P, Prieto-Cerdeira R, Bolea-Alamanac A, Perez-Fontan F, Sanchez-Lago I. DVB-SH analysis using a multi-state land mobile satellite channel model. *Advanced Satellite Mobile Systems, 2008. ASMS 2008. 4th*, 2008; 149–155.
4. Lutz E, Cygan D, Dippold M, Dolainsky F, Papke W. The land mobile satellite communication channel-recording, statistics, and channel model. *IEEE Transactions on Vehicular Technology* 1991; **40**(2):375–386, doi: 10.1109/25.289418.
5. Perez-Fontan F, Vazquez-Castro MA, Buonomo S, Poiares-Baptista JP, Arbesser-Rastburg B. S-band LMS propagation channel behaviour for different environments, degrees of shadowing and elevation angles. *IEEE Transactions on Broadcasting* 1998; **44**(1):40–76.

6. ITU. Rec. ITU-R P.681-6, Propagation data required for the design of Earth-space land mobile telecommunication systems 2003.
7. Simoens S, Bartolome D. Optimum performance of link adaptation in HIPERLAN/2 networks. *IEEE Vehicular Technology Conference, 2001. VTC 2001 Spring*, vol. 2, 2001; 1129 –1133 vol.2.
8. ETSI. Digital Video Broadcasting (DVB) DVB-SH Implementation Guidelines, TS 102 584.
9. Lampe M, Rohling H, Zirwas W. Misunderstandings about link adaptation for frequency selective fading channels. *IEEE International Symposium on Personal, Indoor and Mobile Radio Communications, PIMRC*, vol. 2, 2002; 710 – 714.
10. Kim J, Ashikhmin A, de Lind van Wijngaarden A, Soljanin E, Gopalakrishbab N. Reverse Link Hybrid ARQ: Link Error Prediction Methodology Based on Convex Metric", 3 GPP2 TSG-C WG3. *Technical Report* April 2003.
11. Strinati E, Simoens S, Boutros J. New error prediction techniques for turbo-coded OFDM systems and impact on adaptive modulation and coding. *IEEE International Symposium on Personal, Indoor and Mobile Radio Communications, PIMRC*, vol. 2, 2005; 1116 –1119.
12. Brueninghaus K, Astely D, Salzer T, Visuri S, Alexiou A, Karger S, Seraji GA. Link performance models for system level simulations of broadband radio access systems. *IEEE International Symposium on Personal, Indoor and Mobile Radio Communications, PIMRC*, vol. 4, 2005; 2306 –2311.
13. Jensen TL, Kant S, Wehinger J, Fleury BH. Mutual information metrics for fast link adaptation in IEEE 802.11n. *IEEE International Conference on Communications, 2008. ICC '08*, 2008; 4910 –4915.
14. Sayana K, Zhuang J, Stewart K. Short term link performance modeling for ML receivers with mutual information per bit metrics. *Global Telecommunications Conference, 2008. IEEE GLOBECOM 2008.*, 2008; 1 –6.
15. IEEE 80216 Broadband Wireless Access Working Group. IEEE 802.16m Evaluation Methodology Document (EMD), IEEE 802.16m-08/004r2 July 2008.