Prediction of performance of the DVB-SH system relying on Mutual Information

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***Abstract*—DVB-SH (Digital Video Broadcasting- Satellite Handled) is a broadcasting standard dedicated to hybrid broadcasting systems combining a satellite and a terrestrial part.**

On the satellite part, dedicated interleaving and time slicing mechanisms are proposed to mitigate the effects of Land Mobile Satellite (LMS) channel, based on a convolutional interleaver. Depending on the parameters of this interleaver, this mechanism enables to split in time a codeword on duration from 100 ms to about 30s. This mechanism significantly improves the error recovery performance of the code but in literature, exact evaluation at system level of this improvement is missing.

The objective of this paper is to propose a prediction method compatible with fast simulations, to quantitatively evaluate the system performance in terms of Packet Error Rate (PER). The main difficulty is to evaluate the decoding probability of a codeword submitted to several levels of attenuation. The method we propose consists in using as metric the Mutual Information (MI) between coded bit at the emitter side and the received symbol. It is shown that, by averaging the MI over the codeword and by using the decoding performance function g such that PER=g(MI) determined on the Gaussian channel, we can significantly improve the precision of the prediction compared to the two other methods based on SNR and Bit Error Rate (BER). We evaluated these methods on three artificial channels where each codeword is transmitted with three or four different levels of attenuations. The prediction error of the SNR-based (resp. the input BER-based) method varies from 0.5 to 1.7 dB (resp. from 0.7 to 1.2 dB) instead of the MI-based method achieves a precision in the order of 0.1 dB in the three cases. We then evaluate this method on real LMS channels with various DVB-SH interleavers and show that the instantaneous PER can also be predicted with high accuracy.

I. Introduction

DVB-SH [1] [2] (Digital Video Broadcasting Satellite Handled) is a broadcasting standard dedicated to hybrid broadcasting systems combining a satellite and a terrestrial part. This type of system operating in S band at 2 GHZ aims at providing a high quality video or audio services for light terminal using battery with low autonomy. We focus in this paper on the satellite component of this hybrid system, and on the evaluation of its performances.

Channel of this system can be described as a LMS (Land Mobile Satellite) whose modeling was largely studied ([3], [4], [5], [6]). This channel is a time varying channel inducing a very dispersive value 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 relying on convolutional interleavers that can split the symbols of a single codeword (of the turbo code) over up to 30 seconds of channel. This mechanism can be seen as a fading averaging and results in a significant improvement of performance in terms of error recovery.

A major difficulty associated to this mechanism is to evaluate its performances on sufficient channel distances in order to be able to tune their parameters. Actually, the use of time slicing leads to a non stationary distribution of Signal to Noise Ratio (SNR) inside a code word of the turbo code and there is no theoretical expression for the performance of a turbo code for such a complex non stationary distribution. As a matter of fact, simulations are required. If we are concerned about statistical values like mean Packet Error Rate (PER), simulations relying on Monte Carlo methods become time prohibitive. A modeling step turns out to be necessary. This modeling task can be divided into two parts : modeling the channel and modeling the performance of the DVB-SH physical layer on this channel, more precisely the coding/decoding process on interleaved channel samples. Concerning the LMS channel, models [5] and [3] respectively based on Markov Chain with three states and semi-Markov Chain with two states reveal to be quite efficient. With regard to the coding/interleaving performance on this channel, the issue to be addressed is the following : given a perfect Channel State Information (CSI), what is a good measure to be considered for the prediction of the performance? Several methods can be found to answer this question, which differ on the measure retained, denoted as the Link Quality Metric (LQM) performance.

The most intuitive approach consists in choosing the mean Signal to Noise Ratio (SNR) inside a code word [2] p70, [7]). However, the more the SNR dispersion inside a codeword is high around the mean value the more the prediction error is high. Bad reliability in PER prediction of this method is pointed out in [10]. Reference [15] 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 (complementary cumulative distribution function), the exponential function, or more complex functions [9].

More recently, a new LQM which consists in considering the Mutual Information (MI) between coded bits and Log

Likelihood Ratio (LLR) at the input of the decoder, was studied for terrestrial MIMO transmissions [11], [12], [13], [15] [16] [17]. We propose in this paper to explore the potentialities of this method for our specific context. Originality of this paper are multiple : first, we present a modified version of this method by considering Mutual Information between coded symbols and received symbols (and not between coded bits and LLR). In addition, we propose to transpose this method from the terrestrial framework to the satellite one with LMS channel. Finally, in an effective goal, thanks to our simulator, we provide PER results for many parameters of time slicer for DVB-SH system.

In part 2, we present the prediction issue by highlighting the LMS channel and the time slicing/interleaving mechanism for the DVB-SH system. In part 3, we present three different LQM for PER prediction. In part 4, we propose a compared evaluation of the three methods. Part 5 is devoted to a validation of the PER prediction on a LMS channel at the DVB-SH physical layer.

II. Performance evaluation issue in DVB-SH systems

A. LMS channel Modeling

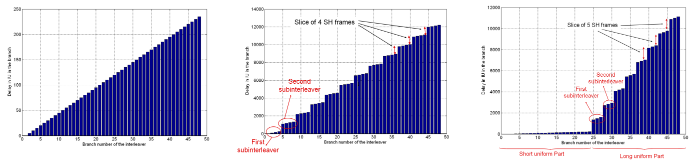
In the DVB-SH system, coverage is mostly ensured by satellite in rural and suburban areas. At S-band, the Land Mobile Satellite (LMS) propagation channel can generally be considered as non frequency selective.Several propagation channel models have been developed in the past. All these models rely on a representation of the LMS channel through a first order Markov chain to represent the large scale (several meters) changes of propagation fading levels affecting the transmitted signal. Some of these Markov chains take into account two states (LOS, 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.

B. Overview of the DVB-SH system

Because of the strong level of shadowing of the LMS channel when mobile is crossing bad states of the Markov Chain, a very large margin is required in order to achieve target PER. Then, time interleaving/time slicing is proposed to counteract the effects of signal fading [2], p 107. This time slicing is achieved thanks to a convolutionnal interleaver of 48 branches acting on Interleaver Units (IU) made of 126 bits. Value of the 48 delays are entirely defined by the choice of five parameters given in [2]: (Nof_late_taps, $Common\_multiplier$, Nof_slices, $Slice\_distance$, $Non\_late\_increment$).

Without detailing the exact meaning of the interleaver parameters, it is essential to emphasize that this interleaver enables a time slicing operation, as well as interleaving. This slice option is set off by a parameter value Nof_slices greater than one and means that a SH frame is interleaved and also split into several SH frames. Three classes of interleaving parameters setting are proposed in [2] that are reviewed here.

This class of Short Uniform Interleaver (Nof_late_taps=48, Nof_slices=1) does not carry out the slice operation and consists of a regular interleaving where the i-th branch is delayed by $i * Common\_multiplier$ meaning that two successive UI in the same branch are $48 * Common\_multiplier$ apart at the interleaver output (number 48 is because interleaver is periodic with period 48) . Figure 1(a) illustrates delay scheme for such an interleaver and for parameters (5,48,1, 0,0). Depending on the physical layers parameters, this interleaver spreads one SH-frame on 100 up to 240 ms.



(a) (b) (c)

Fig. 1. Interlavers : (a) short uniform interleaver, (b) long uniform interleaver, (c) long uniform late interleaver

Long Uniform Interleaver carries out slice operation and it is an hybrid interleaver since since it both performs time slicing and uniform sub interleaving on each slice . Effect of these sub interleavers is to interleave input IU within a SH frame and effect of slice is to split output IU of each subinterleaver into Nof_slices SH frames that are $Slice\_distance$ apart. Depending on the physical layers parameters, this interleaver spreads one SH-frame on 6.4s up to 15.6 s. Figure 1(b) illustrates the delays for such an interleaver with parameters (40,0,12,4,2). We can see that in a subinterleaver, IU in two successive branches are separated from 3840 at the output of interleaver as for the uniform interleaver. In addition, two IU in two successive sub interleavers are separated from 4 SH frame at the output interleaver.

The third family of interleaver is the long uniform late interleaver. This interleaver is composed of two parts. The first part is a short uniform interleaver that processes the first Nof_late_taps branches. The second part is a long uniform interleaver acting on the $48 - Nof\_late\_taps$ remaining branches. Depending on the physical layers parameters, this interleaver spreads one SH-frame on 12.8s up to 31.2 s. Figure 1(c) illustrates the delays for such an interleaver with parameters (10,24,9,5,12). We can see that in the short uniform part, IU in two successive branches are separated from 480 at the output of interleaver. In addition, in the long uniform part, IU in two successive branches in the same subinterleaver are separated from 5760 at the output of interleaver. In addition, two IU in two successive sub interleavers from the long uniform part, are separated from 5 SH frames at the output interleaver.

C. Difficulties encountered for performance prediction

We aim at predicting performance of DVB-SH coding/interleaving mechanisms. The retained coding for DVB-SH is turbo code with input block size of 12282 bits. Whatever the code rate is (2/3 up to 1/5), output code word size will be more greater than 3 periods of the interleaver. As a consequence, depending on the physical layer parameters and the interleaver choice, any codeword will be split from 100 ms up to 31.2s. Because of the LMS channel, several output IU of a same SH frame will be affected by different attenuation as illustrated in figure 2. This figure highlights how an input SH frame is split on the LMS channel. Realization of this channel is obtained with Perez Fontan Model for elevation of 40° in a suburban environment and a speed of the mobile of 60km/h.

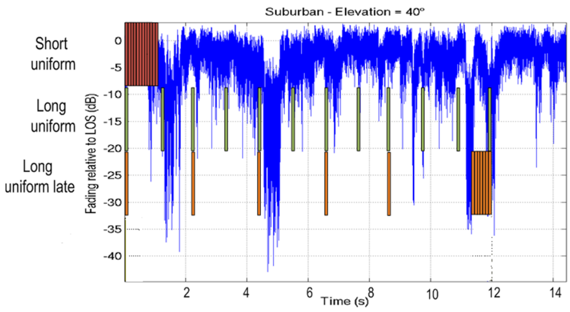


Fig. 2. Illustration of the interleaving/slice effect on the LMS channel

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 code words with time varying SNR according to Figure 3.

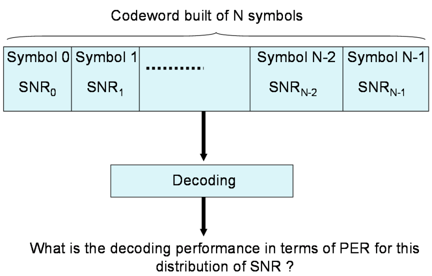


Fig. 3. Formulation of PER estimation problematic

Time varying value inside the code word is caused by the different value of attenuations. The issue is therefore : are we able to predict PER performance of such a code word? 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 expression of the PER. However, in the application we are interested in, interleaving and time slicing leads to a non stationary distribution because of the different Markov states. A theoretical expression of the PER can not be derived.

An alternative solution for obtaining 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 on a duration d, let assume that we want to get an estimation of the PER of each codeword of the DVB-SH system on this channel with duration d. PER is a statistical value. As a consequence, Monte Carlo methods are suitable to get this value. This methods consists in generating a large number of realizations of the same code word and after decoding this codeword, PER estimation is obtained by computing the ratio of wrong decoded code word. Generation of the codeword is done until the estimation of the PER has converged. An empirical rule says that the number of generation of the same codeword must be at least $100/PER_{theoretical}$ 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 duration of codeword is about 3.3 ms. As a consequence, for a channel with duration d in second, the number of codewords is about $d/3.3*10^3$. Assuming that we aim at estimating PER performance on a channel with duration 60s, this leads to a number of about 18 000 codewords. With Monte Carlo method for every codeword, it leads to a minimal number of 1 800 000 code words to be simulated.

III. Prediction methods presentation

In the light of the previous part, it is obvious that a modelling step is required if we are looking for PER estimation for the DVB-SH system. In our context, the issue to address is: what is the performance of the system when a codeword is affected by different levels of SNR? The most intuitive approach [2], [7] is to consider the average SNR on the codeword. However, although simple, this method results in a prediction error [10] that is all the more high that SNR dispersion is high around the mean value. [15] highlights that the use of a convex function can reduce this prediction error. Many functions have been proposed like the Q function [8] or more complex functions [9]. In this part, we propose the general framework of this prediction methods and we put in light a recent method that was confined to the terrestrial context with MIMO channels. This method relies on the use of Mutual Information.

A. Generic formulation of the prediction methods

All the bibliographic methods of PER performance share a common approach based on a notion of equivalent constant SNR denoted in the following as SNR_{gauss}. They assume that the performance of a codeword submitted to different levels of different levels of SNR_i (for each symbols i $0 \leq i \leq N-1$, cf figure 7)) is given by the performance of a Gaussian Channel with an equivalent constant SNR_{gauss}.

SNR_{gauss} can be expressed in a generic way as a function of the successive levels of SNR by the following formula :

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

where F is the Link Quality Metric (LQM) function.

The advantage of this approach is to be able to compute for example PER on the varying channel using the performance curves $PER = H(SNR)$ on a Gaussian channel at this value of SNR_{gauss}:

$$PER = H(SNR_{gauss}) \tag{2}$$

B. Presentation of the different link quality metrics

We consider in the following that received complex symbol y_k^m (k th symbol the m th code word) is obtained at the output of the channel according to:

$$y_k^m = \rho_k^m x_k^m + n_k^m \tag{3}$$

where x_k^m is the complex emitted symbol (k th symbol of the m th code word), ρ_k^m is the real and positive attenuation of the channel (we make the assumption that phase rotation is perfectly corrected at the reception) that affects x_k^m and n_k^m is the additive Gaussian Noise with DSP $\frac{N_0}{2}$ on the independent imaginary and real parts that affects x_k^m. Interleaving and time slicing are considered by considering the interleaved equivalent channel (taking into account the interleaving/slice process) according to Figure 4.

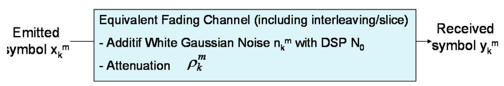


Fig. 4. Description of the equivalent "interleaved" channel

1) Mean SNR as LQM: The most intuitive method consists in considering the mean SNR [7] on the code word. In the description (1), this method leads to consider F as the identity function and the equivalent SNR_{gauss}^m of the m-th code word is given by:

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

where SNR_k^m is the SNR of the k-th symbol of the m-th code word.

2) BER on the channel as LQM: A second method consists in considering the equivalent BER on the channel on the coded symbols before decoding. For a given modulation and coding scheme, 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}^m of the m-th code word is given by:

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

here SNR_k^m is the SNR of the k-th symbol of the m-th code word.

3) Mutual Information as LQM: If we consider the Mutual information between emitted symbol and received symbol, the equivalent SNR_{gauss}^m of the m-th code word is given by:

$$SNR_{gauss}^m = I^{-1}\left(\frac{1}{N}\sum_{k=0}^{N-1} I(SNR_k^m)\right) \tag{6}$$

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

$$I(X,Y) = H(Y) - H(Y/X) \tag{7}$$

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

$$H(U) = -\int f(u)\log_2(f(u))du \tag{8}$$

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

C. Methods derivation for QPSK

Here we propose to theoretically derive the previous expressions in case of QPSK modulation. As SNR indicator, we choose the $\frac{E_b^c}{N_0}$ as the ratio between the energy per coded bit and the noise DSP. We assume that simulations provide us performance curve $PER = G(\frac{E_b^c}{N_0})$ on a AWGN channel and for a QPSK modulation and a code choice with coding rate R. We denote as $\left.\frac{E_b}{N_0}\right]$ the System margin, i.e the ratio between the energy per coded bit and the noise DSP in case of no attenuation.

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 PER of the $m-th$ code word is given by:

$$PER_{mean}^m = G\left(\left.\frac{E_b^c}{N_0}\right]_{Gauss}^{m,mean}\right) \tag{9}$$

with :

$$\left.\frac{E_b^c}{N_0}\right]_{Gauss}^{m,mean} = F_{mean}^{-1}\left(\frac{1}{N}F_{mean}\left(\sum_{k=0}^{N-1}(\rho_k^m)^2\left.\frac{E_b}{N_0}\right]\right)\right) \tag{10}$$

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

BER channel method:

In case of a QPSK, BER channel 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, PER_{BER}^m of the $m-th$ code word is given by:

$$PER_{BER}^m = G\left(\left.\frac{E_b^c}{N_0}\right]_{Gauss}^{m,BER}\right) \tag{11}$$

where,

$$\left.\frac{E_b^c}{N_0}\right]_{Gauss}^{m,BER} = F_{BER}^{-1}\left(\frac{1}{N}F_{BER}\left(\sum_{k=0}^{N-1}(\rho_k^m)^2\,\frac{E_b}{N_0}\right]\right)\right) \tag{12}$$

and

$$F_{BER}(x) = Q(\sqrt{2x}) \tag{13}$$

Mutual Information method:

Derivation of the theoretical expression of the Mutual Information $I(\frac{E_b^c}{N_0})$ between emitted and received symbol in case of a QPSK modulation is found in Appendix. PER_{MI}^m of the $m-th$ code word is therefore given by:

$$PER_{MI}^m = G\left(\left.\frac{E_b^c}{N_0}\right]_{Gauss}^{m,MI}\right) \tag{14}$$

where,

$$\left.\frac{E_b^c}{N_0}\right]_{Gauss}^{m,MI} = F_{MI}^{-1}\left(\frac{1}{N}F_{MI}\left(\sum_{k=0}^{N-1}(\rho_k^m)^2\,\frac{E_b}{N_0}\right]\right)\right) \tag{15}$$

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

IV. Comparison of the different prediction methods

In this part, we propose to evaluate performance of prediction for the three previous methods on particular channel realisations for a QPSK turbocoded DVB-SH transmission [1].

Two different coding rates are considered : 1/3 and 1/2. For each coding rate, mean BER and PER are simulated on the different channels considered through monte-carlo simulations. The simulation channels are defined on one code word length. The simulations are then performed repeating the channel attenuations on each successive code word and calculating the mean BER or the mean PER.

These "real" BER or PER performances are then compared to prediction results performed relying on the 3 previous methods and using as reference the BER and PER performance of these codes on a Gaussian channel.

A. Presentation of the simulation channels

The following simulation channels are defined as a succession of attenuations. When attenuation is equal to zero, E_b/N_0 is equal to (E_b/N_0)n (nominal). The nominal E_b/N_0 is variable and represented on abscissa axis on the performance curves whereas the successive attenuation values are considered fixed for a given simulation channel.

For 1/3 (respectively 1/2) coding rate simulations, we propose the three different schemes of attenuation presented on Figure 5 (respectively on Figure 6).

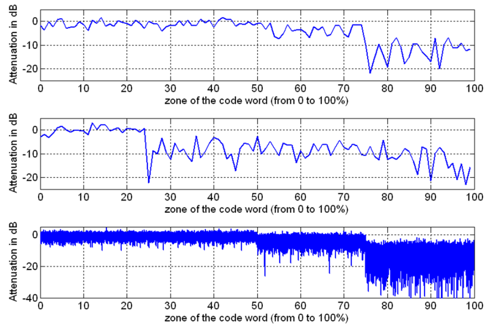


Fig. 5. The 3 channel realizations for the coding rate 1/3

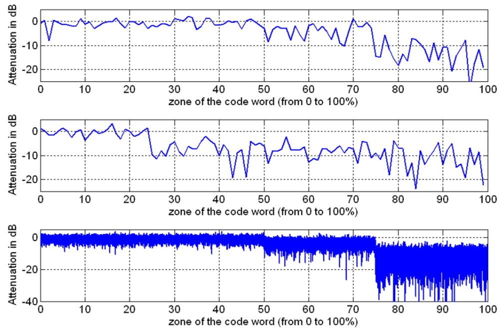


Fig. 6. The 3 channel realizations for the coding rate 1/2

B. Evaluation of the methods for BER prediction

1) 1/3 coding rate: Figure 7 represents the BER prediction of the different methods in terms of nominal E_b/N_0.

This figure shows that Mutual Information methods provides a confident estimation of the output BER of the rate 1/3 turbo code on the three channels.

2) 1/2 coding rate: Figure 8 represents the BER prediction of the different methods in terms of nominal E_b/N_0.

This figure shows that Mutual Information methods provides a confident estimation of the output BER of the rate 1/2 turbo code on the three channels.

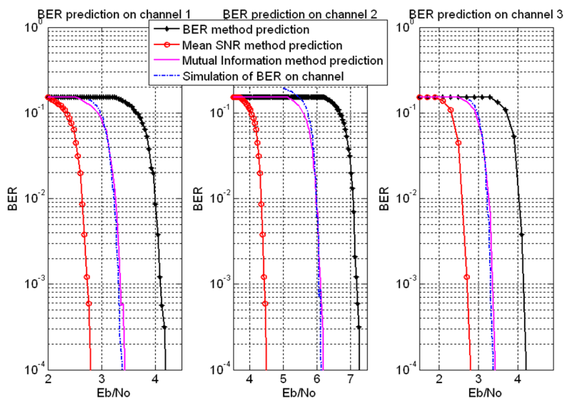


Fig. 7. BER predictions for coding rate 1/3

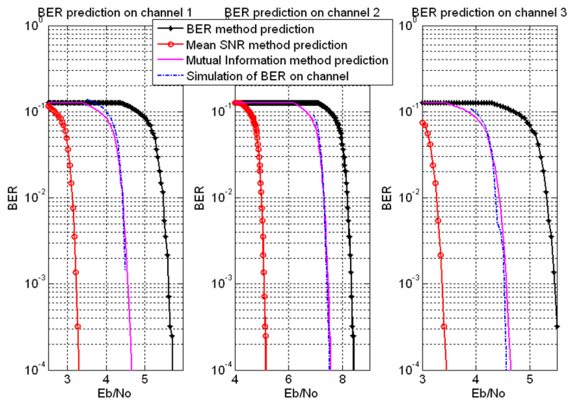


Fig. 8. BER predictions for coding rate 1/2

C. Evaluation of the methods for PER prediction

This subsection is identical to the previous one, but PER is considered and not BER and only coding rate 1/3 is considered.

1) 1/3 coding rate simulations: Figures 9, 10 and 11 represent the BER prediction of the different methods in terms of system margin $\left.\frac{E_b}{N_0}\right]$ (E_b is the energy per information bit) for the three channels.

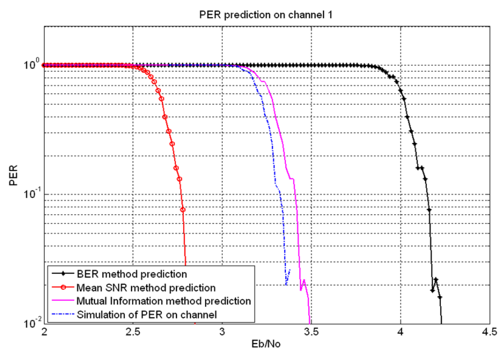


Fig. 9. PER predictions for coding rate 1/3 and channel 1

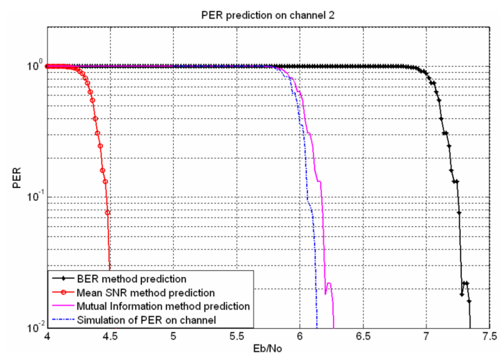


Fig. 10. PER predictions for coding rate 1/3 and channel 2

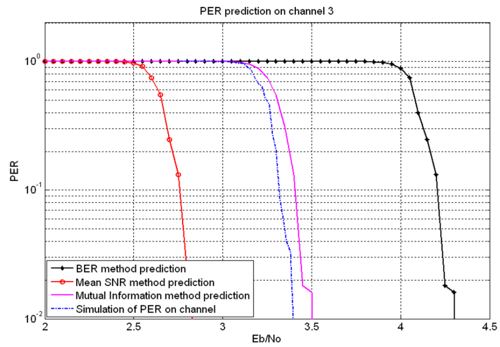


Fig. 11. PER predictions for coding rate 1/3 and channel 3

2) PER results analysis: These figures show that Mutual Information methods provides a confident estimation of the PER estimation on the three channels for the coding rate 1/3. A light shift of about 0.1 dB appears in the PER estimation with the Mutual Information method. Complementary simulations tend to show that this offset only depends on the modulation and coding scheme and could then be corrected independently of the propagation channel. We propose to compensate the residual offset in the following simulations. This offset is set to 0.09 dB for QPSK 1/3.

D. Conclusion on prediction methods comparison

Based on these results it appears clearly for QPSK turbo-coded transmission following DVB-SH standard that the proposed prediction method based on mutual information performs very precise predictions better than 0.1 dB for both output BER and PER performance calculation. We show that more traditional methods like mean SNR or mean channel BER give more approximate predictions. We propose then to focus in the following on mutual information prediction method and to go more in depth into its validation.

V. Validation for a LMS channel

We propose in this part to complete the evaluation of the Mutual Information method on a more realistic DVB-SH transmission on our propagation channel of interest that is LMS channel.

A. Presentation of physical layer parameters

We consider the following parameters extracted from the DVB-SH guidelines [2].

These parameters mean that a QPSK modulation with a 1/3 coding rate turbo coding is used. The interleaver belongs to the long uniform family of interleavers (figure 5). Value of $\left.\frac{C}{N}\right]$ target is related to the the $\left.\frac{E_b}{N_0}\right]$ margin system previously defined by the relation for OFDM :

$$\left.\frac{E_b}{N_0}\right] = \left.\frac{C}{N}\right] - 10 * log10(coding\_rate * modulation\_order) - \alpha \tag{16}$$

Waveform	QPSK1o3_U
Configuration	LMS-ITS
Channel (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
C_N target	5

TABLE I
VALUE OF PHYSICAL PARAMETERS

α value is because of pilot symbols and depends on the OFDM mode (2k, 4k,8k). For example, for 2k mode, it equals 0.35.

B. Presentation of the channel

Channel is a LMS channel defined by the model of section 3. Environment retained is ITS. A realization of this channel on one minute duration will be used in the following simulations and is given by Figure 12.

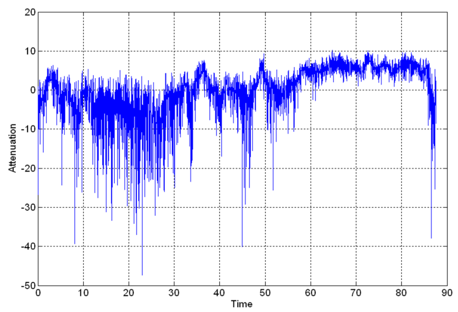


Fig. 12. Realisation of the channel on about one minute

C. Presentation of the simulations

In this part we propose to compare the two following values:
- Simulated PER: PER is simulated with previous parameters and channel. The one minute channel corresponds to 16283 code words. Monte Carlo methods are used and these 16283 code words are simulated 10 times. $PER_{est}(i)$ of the $i-th$ code word is estimated by:

$$PER_{est}(i) = \frac{1}{10} \sum_{j=1}^{10} X_i^j \tag{17}$$

where X_i^j is the random variable that equals 1 if the $j-th$ realization of $i-th$ code word is is wrong (Paquet is not perfectly decoded) and equals 0 if decoded paquet is good.

- Predicted PER: Using method relying on Mutual Information, and the realization of channel (figure 22), prediction of PER is performed and we denote $PER_{pred}(i)$ as the predicted PER for the $i-th$ code word. PER prediction is done using a shift of 0.09 dB according to figure 19,20 and 21.

1) Simulations results: In a first step, we consider the instantaneous PER estimation and simulation for the LMS channel on figure as well as the PER error. Figure 13 illustrates results.

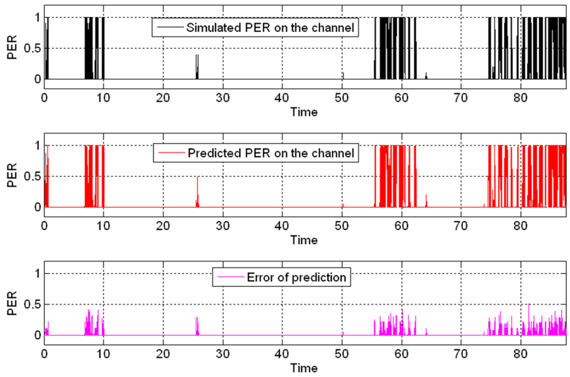


Fig. 13. Estimated and predicted instantaneous PER

Making a zoom on an interesting zone, we get the Figure 14:

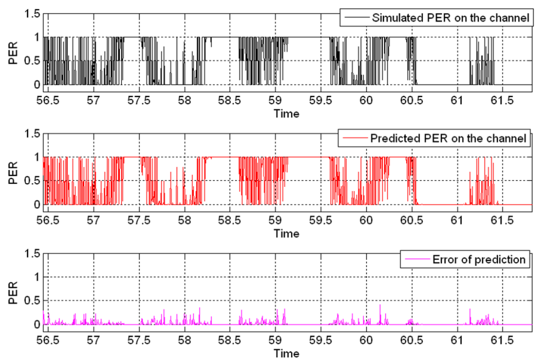


Fig. 14. Zoom for Estimated and predicted instantaneous PER

We can conclude from these results that instantaneous PER is quite well estimated by the Mutual Information method.The general behavior is indeed respected, and instantaneous error is less than 0.3.

In a second step, for better visibility, we propose to consider the averaged PER (averaging on the 3 last values). Figure 15 presents these results.

And making a zoom on the same zone than figure 14, we get the results of Figure 16.

When we consider the averaged PER (on only 3 code words), we see that the mutual information method leads to a very efficient prediction. It is interesting to remark than $PER_{est}(i)$ (9) has a large variance since only 10 realizations are available. As a consequence, error prediction for instantaneous PER in Figures 13 and 14 may be caused by this variance and not the efficiency of the prediction. The fact that averaging PER on only 3 words leads to a strong decrease of error prediction seems to corroborate this assumption.

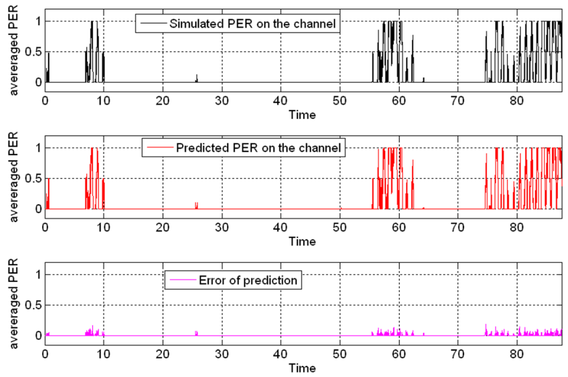


Fig. 15. Estimated and predicted averaged PER

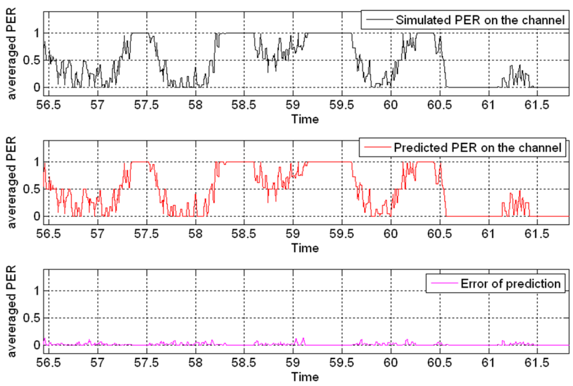


Fig. 16. Zoom for estimated and predicted averaged PER

D. Conclusion of the Validation for a LMS channel

This part completes the validation results with the evolution of PER on a realistic LMS propagation channel. The results show an excellent convergence between monte-carlo simulation and mutual information prediction method once the averaging is sufficient on monte-carlo simulations. The proposed method is then validated for the initial purpose exposed in the paper that is large scale performance evaluation in terms of QoS for DVB-SH transmissions using long interleavers on realistic LMS propagation channels.

VI. Conclusion

In this paper, we were interested in the performance evaluation of DVB-SH system. DVB-SH interleavers spread over a long period several parts of a FEC codeword. The main difficultly consists thus to evaluate the decoding performance of a codeword submitted to various noise levels. A theoretical analysis of this performance seems extremely hard and a monte-carlo-based evaluations reveal to be time prohibitive.

We propose a method to quickly evaluate PER performance of the system. This method relies on the computation of Mutual Information between coded bits and received symbols. Reliability of this method was first demonstrated using simulated channels with non stationary distribution of noise variance and compared to two other methods, namely Mean SNR method and BER method. These simulations show that the mutual information method exhibits a very good reliability. Then, a more detailed validation was proposed in simulating the DVB-SH physical layer and the LMS channel on one minute duration. PER estimation was based on these simulations and compared to prediction based on the Mutual Information method. Results show that the prediction method is very accurate.

In this paper, the method was proposed for performance evaluation of QPSK modulation. Work is under progress to generalize this methods for higher order modulation.

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