Extending Coverage of High Definition TV Services over ADSL2 with Optimized Reception Quality using H.264/AVC Transrating

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Original scientific paper

***Abstract*— In this paper, we present a new Joint Source-Channel Coding (JSCC) architecture to extend the coverage of H.264/AVC High Definition (HD) video delivery over Digital Subscriber Line (DSL). The proposed solution combines low complexity H.264/AVC transrating as well as multi-carrier transmission and takes into account realistic ADSL2 specifications including all OSI layers. Both transrating and bit and power loading transmission parameters are automatically optimized in terms of end-user perceived quality, with respect to the characteristics of the given subscriber's loop. Several originalities have been included: a new optimization algorithm has been developed, as well as a full rate-distortion modelling of the H.264/AVC transrater's performances. Simulation results show that the proposed solution can extend the coverage area of HD video delivery up to more than one kilometre. It should allow the widespread distribution of HD video contents and increase the number of eligible subscribers.**

***Index terms*— ADSL2, H.264/AVC, joint source channel coding, transrating, Quality of Experience (QoE).**

I. Introduction

The performance of the H.264 Advanced Video Coding (AVC) standard [1] has recently allowed the DSL operators to provide new High Definition Television (HDTV) services to their subscribers. However, DSL operators are restricted by the throughput limitation of copper loops (due to attenuation, interferences and network topology), when attempting to extend their HDTV service coverage to subscribers located too far from the central office (also called Digital Subscriber Line Access Multiplexer, or DSLAM). Hence, they need to consider innovative broadcasting solutions in order to overcome this problem and consequently increase the number of subscribers.

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A. *Previous work*

Several solutions for digital television coverage extension have been already proposed in the literature [2-5]. These solutions mainly rely on a joint source channel coding (JSCC) approach, in which the bit rate is lowered by transrating [2] or scalable bit stream extraction [3, 4] and, in a combined way, the transmission parameters (modulation, coding rate) are chosen such that the Quality of Experience (QoE) is optimal for the end user. In particular, Coudoux et al. [5] proposed an original JSCC architecture for coverage extension of standard definition TV services using MPEG-2 over ADSL. This JSCC architecture consists of the optimized combination of on-the-fly low complex MPEG-2 transrating with multi-carrier transmission from an end-to-end video quality point of view.

B. *The proposed solution*

In this paper, we address the problem of efficient coverage extension for emerging H.264/AVC high definition TV services over ADSL2. The H.264/AVC standard offers a gain of compression about two compared to MPEG-2 for the standard definition. Consequently, the DSL operators can now transmit High Definition video services using H.264/AVC at the same bit rate instead of Standard Definition ones using MPEG-2. We propose a JSCC solution as [5], but here based on efficient H.264/AVC video transrating combined with optimized transmission using ADSL2 technology. The aim of the proposed solution is to deliver high definition TV contents with satisfying video quality to subscribers located far from the central office (e.g., in rural areas), for which this type of service was not yet available.

Several novelties have been introduced in the proposed solution compared to the one described in [5], which was based on the MPEG2 standard: first, the new H.264/AVC video coding tools such as intra prediction or more efficient context-adaptive variable length coding algorithms should be carefully taken into account. Consequently it leads us to develop a new H.264/AVC transrating scheme already presented in [9, 10] which is proving to be more complex to implement than in the MPEG-2 case. Moreover, the transrating algorithm has been developed with respects to low complexity constraints as it must be integrated directly into the central office in the context of high-definition video delivery scenarios, and a new rate-distortion modeling of the transrating performances has been proposed.

Secondly, concerning the transmission part, the new ADSL2 specifications are accounted for instead of the ADSL ones including better noise modeling with FEXT and NEXT precise models, and newly compulsory trellis coding. Low complex non iterative algorithm developed by the authors [12] has been also adapted in order to compute bit and power allocation over sub-carriers. Moreover, on the contrary to the approach in [5] we adopt a real cross-layer approach by considering all the OSI layers from the video application layer to the physical one (ADSL2) including the RTP/UDP transport layer.

Finally, the authors present a new JSCC system based on a joint source and channel coding approach in order to extend the coverage area of HDTV services over ADSL2 with best reception video quality. The new contributions of this paper also consist of the efficient combination of both techniques thanks to the optimization algorithm, as well as a thorough analysis of obtained results in terms of extension of the eligibility distance and evolution of minimal end-to-end visual distortion.

The proposed JSCC architecture is depicted on Fig. 1. The system input is digital video content that has been initially compressed using the H.264/AVC standard. Compressed video is transmitted from the central office to the end-user's residential home over the subscriber's phone line. When the end-users are located too far from the DSLAM, the channel quality becomes too poor to deliver high-definition video services with sufficient quality level, introducing a maximum distance beyond which end-user clients are no more eligible to HDTV services.

In order to cope with this problem, we propose a combined source–channel coding approach. First, the input compressed video bitstream is transrated in order to adapt the bit rate to the capacity of the given line; this video bitstream is then transmitted using multi-carrier transmission. Both H.264/AVC transrating as well as multi-carrier bit and power loading transmission parameters are jointly optimized given the feedback information on quality of the corresponding subscriber's loop, in order to offer the best QoE to the end user.

The remainder of this paper is organized as follows. In Section II, we give a detailed description of the different parts of the proposed JSCC architecture. These include a low complexity H.264/AVC transrating algorithm combined with a multi-carrier modulation (MCM) transmission scheme. Then, we demonstrate in Section III the throughput limitation related to ADSL2 technology by evaluating the initial ADSL2 eligibility distance, i.e. the maximum distance threshold beyond which end-user clients cannot access broadband HDTV services. In Section IV, the optimization algorithm is detailed and the simulation results are given and analyzed in Section V for different copper loop configurations. These results clearly demonstrate the effectiveness of the proposed solution, as in some cases, the coverage area can be extended up to more than one kilometer from the initial eligibility distance while introducing minimal visual distortion. Such one-kilometer increase of the coverage area's radius around the DSLAM obviously allows to deliver HDTV contents to a significantly greater number of potential end-user clients, with a progressive but acceptable degradation of received quality as the distance from the DSLAM increases. Finally, in Section VI, we give our conclusions and prospects for future research.

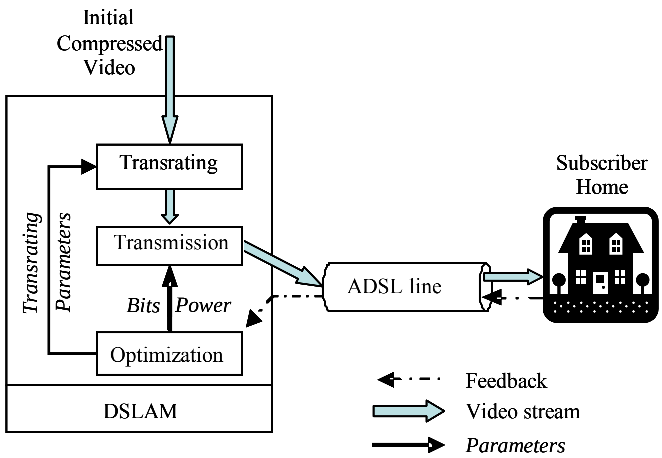


Fig. 1. Block diagram of the proposed JSCC architecture

II. Description of the Proposed JSCC System

In order to extend the coverage area for a given DSL architecture, we propose a joint source-channel coding approach, combining efficient transrating and optimal MCM transmission. The corresponding pseudo-code is given in Table I below.

TABLE I
PSEUDO-CODE OF THE PROPOSED JSCC ALGORITHM

For a given subcribers' line,
BEGIN
- *Compute the maximum available bit rate (D_{max}) under initial power and QoS constraints*
- *IF input video bit rate > D_{max} then*
    - *-Estimate the transrating parameters which lead to best video quality with outpout rate $\leq D_{max}$*
    - *-Transcode the input H.264/AVC compressed bitstream using optimal transrating parameters*
- *Compute bit and power allocation for the final video bit rate and QoS constraints*

END

The video transrating as well as transmission parameters are determined in an optimal way described in Section 4, such that the quality of experience is the best possible for the newly eligible end-user. In what follows, we first describe the different parts of the system architecture. We briefly present the transrating algorithm, and give its performances. Then, we detail the multi-carrier transmission scheme and its parameters.

A. The Proposed Transrating Algorithm

The incoming H.264/AVC compressed bitstream is first transcoded in order to adapt the bit rate to the capacity of the given line, by applying video transrating at the application layer. Several methods have been proposed in the literature for recent years [6-8]. In [8], the authors proposed a low complexity SNR transcoding for H.264/AVC which is limited to inter-coded frames. In the present case, transrating is accomplished using a hybrid H.264/AVC transrating architecture based on selective transmission of transform coefficients [9]:

- Inter-coded frames are transrated using open-loop architecture. Consequently, the subsequent mismatch between the encoded and decoded reference blocks leads to error propagation or drift. However, we verify visually that drift error caused by inter picture transrating remains clearly negligible.
- To avoid the severe visual quality loss due to drift error caused by intra picture transrating, intra-coded reference frames need to be recoded. This is performed by re-using selected modes of the incoming compressed stream. The main advantage of this re-use mode is that it reduces the system complexity compared to a full-decode full-recode transrating algorithm.

Such transrating algorithm has several advantages: it can transcode any kind of picture with no drift error for intra slices, and reduced implementation complexity. It allows real-time high-definition video transrating with great performances compared to existing solutions. In practice, low frequency coefficients that have significant visual influence are left unchanged and kept in a zigzag order until a given frequency position (FP) which varies from 1 (only the DC coefficient remains) to 16 (all coefficients are preserved). To increase the transrater flexibility, the FP parameter used to transrate intra-blocks (FP intra) is chosen independent of the one used to transrate inter-blocks (FP inter). It is shown that, for moderate bit rate reductions, the proposed transrating scheme able to deliver video contents with similar quality compared to the ones originally encoded at the targeted bit rate. Finally, Deknudt et al. [10] demonstrated that this transrating architecture is of low complexity level and allows real time processing required for dynamic bit rate adaptation of H.264/AVC compressed video streams at the DSLAM. The proposed transrating architecture was implemented on a PC using JM reference software, written in the C language, without assembler optimizations. The transrater's performances on 720p YUV sequences is equal to 21 frames per second with an Intel Core 2 Duo E8500 @ 3.16Ghz (one core used) [9]. The transrating architecture was also implemented, for Intra processing only, on a dedicated Field Programmable Gate Array (FPGA) with a Virtex 5 architecture from Xilinx. Only intra picture transrating was implemented, because it is more complex than inter picture transrating. It achieved a real-time processing on High Definition videos (1920x1080@30fps) [10]. Considering the same configuration, the classical transrating architecture based on a full-decode full-recode approach gives a performance equal to one frame by second. So the complexity is reduced by a factor of 21 thanks to the proposed transrater.

In order to illustrate the video quality performances of the transrater, ten well-known 720p50 HD test video sequences (e.g., *Parkrun*, *Mobcal*, *Shields*) have been encoded using JM 12.4 codec software with the Main profile. These video sequences constitute a large database which offers a large variety of high and low spatio-temporal activities. Coding parameters include Exponential Golomb and CAVLC in order to be compatible with all H.264/AVC profiles. The hierarchical (*IBBBPBBBPBBBI*) GOP structure has been also used to limit the inter drift phenomenon. Indeed, the distance between any inter- predicted image and its reference is limited to 3 images and the possible error propagation is consequently restricted. We choose a Constant Bit Rate (CBR) equal to 8 Mb/s. Such bit rate value is in accordance with high video quality levels required by operators for HDTV services over DSL. In what follows, the high-definition video sequence compressed at 8 Mb/s constitutes the reference for further quality evaluation. This is consistent with realistic broadcasting conditions as the end user does never have access to the original uncompressed video sequence.

In order to evaluate the visual quality of transmitted video, we choose to compute the mean square error (MSE) between the reconstructed 8Mb/s compressed video sequence and its transrated-decoded version because MSE, as well as the related peak signal-to-noise ration (PSNR) criterion, remain the most widespread and popular image quality objective metrics in the scientific community, albeit being not always well correlated with human visual assessment. The VQM (Video Quality Metric) criterion was also used [11]. Typical VQM values range from 0 (excellent) to 1 (worse). Hence, both MSE and VQM criteria are monotonically increasing functions of the perceived distortion.

When computing the MSE, we verify that two groups of sequences can be distinguished:

- The first group contains sequences with very little motion and/or few details, for which the MSE values remain low.
- The second group contains sequences with a higher level of activity and/or more texture; the corresponding MSE values are higher.

Because all these sequences are encoded at the same bit rate, the sequences of the second group had to be coded with a higher quantization step (controlled in H.264/AVC by the Quantization Parameter (QP)) compared to the ones belonging to the first group. Hence, we verify that the MSE which depends on the QP parameter varies significantly from one sequence to another. In order to have one unique modeling of MSE distortion whatever the compressed sequence, we propose logically to normalize the MSE by a function of QP. Classically, we suggest considering a polynomial function of the form QP^n. We verify experimentally that the value n=4 gives good results.

Finally, Fig. 2 shows the average normalized MSE (NMSE) curve as a function of the FP intra and FP inter parameters.

We can clearly note that NMSE is not equally influenced by FP intra (bold solid line) and FP inter (bold dotted line), the first one having a greater impact on the reconstructed video quality. These results were confirmed when computing VQM results. These VQM results are presented in Table II for one sequence of each group with FP intra varying and FP inter=16, and vice versa. Again, the video quality is more degraded when intra-coded coefficients are removed, due to the greater efficiency of inter-prediction compared to the intra-prediction as well as to the intra-frame error propagation through the whole GOP because of temporal prediction.

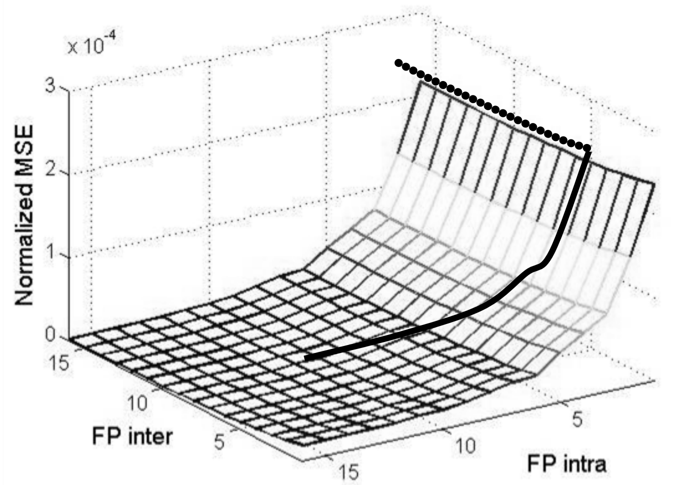


Fig. 2. Evolution of the normalized MSE as a function of the transrating parameters.

TABLE II
VQM NOTES FOR *MOBCAL* AND *PARKRUN* SEQUENCES

	FP intra varying FP inter=16		FP inter varying FP intra=16	
FP value	*Mobcal*	*Parkrun*	*Mobcal*	*Parkrun*
1	**0.53**	**0.48**	**0.03**	**0.09**
4	**0.17**	**0.19**	**0.02**	**0.03**
8	**0.03**	**0.09**	**0.01**	**0.03**
12	**0.02**	**0.03**	**0.01**	**0.03**
16	**0.01**	**0.03**	**0.01**	**0.03**

B. *DSL Transmission Parameters*

As illustrated in Fig. 1, digital compressed video is transmitted from the DSLAM to the end user's residence over the copper loops originally used for telephony services. This is possible thanks to the introduction of advanced digital processing tools including multi-carrier modulation (MCM) which allows to optimize data transmission after channel estimation thanks to the bit and power loading. Among the different optimization algorithms available in the literature, we use the bit and power loading algorithm developed by Goudemand et al. [12]. This algorithm has been selected because of its reduced complexity. Indeed, it has the great advantage to determine the optimal bit and power distribution for MCM using a single iteration compared to other conventional algorithms. Such reduced complexity is necessary in order to fulfill delay constraints of video applications. In this algorithm, the total power constraint is minimized under a fixed Bit Error Rate (BER) and fixed bit rate constraints. Other ADSL2 transmission parameters are defined as follows:

- We consider normalized ETSI line models in our simulations, noted loop1 to loop8, with multiple section sizes from 0.4 to 0.9 mm. The loop3 to loop8 models are defined respectively with minimal lengths of 1500, 2200, 1750, 1750, 4200 and 1100 meters [13], and the loop8 model includes bridge taps.
- We use the normalized ETSI noise model defined in [14]. It includes NEXT and FEXT for alien transmission systems, as well as background noise.
- RS (255,239) Reed Solomon codes combined with a 3 dB gain for Trellis coding [15] were applied to transmitted data.
- The maximum power constraint allowed for ADSL2 transmission (19.9 dBm) [16] as well as the 15% rate overhead due to RTP/UDP/IP and ATM framing were taken into account.

Because we want to ensure the best QoE level to the end user, Quasi Error-Free (QEF) conditions were applied which correspond to less than one erroneous bit per hour of received video [17]. For the given video coding parameters, the resulting bit error rate (BER) is equal to: $BER_{QEF} = 10^{-11}$ under video coding parameters. However, depending on the distance from the DSLAM to the residence, channel quality varies, introducing a maximum distance threshold beyond which end-user clients cannot access full bit rate high definition TV services over DSL under QEF conditions and corresponding to 8Mb/s. Hence, in the next section we estimate this threshold, called hereafter the initial video eligibility distance.

III. INITIAL VIDEO ELIGIBILITY DISTANCE

In order to evaluate the performances of our solution in terms of coverage extension, we first need to compute the initial eligibility distance for which compressed video streams are successfully transmitted from the DSLAM to the subscriber's home without any bit rate reduction or adaptation of the transmission parameters.

For each given loop model (loop1 to loop8), we determine the maximum line length for which the video streams encoded at 8 Mb/s can be transmitted under the total power and QEF BER constraints. Practically, this is achieved by performing a bit and power allocation for each tested distance as the result of the bit and power loading algorithm cannot be predicted analytically in a straightforward way. Fig. 3 gives the results for all loop models.

We can first observe that loop4, loop7 and loop8 cannot offer the 8Mbps throughput necessary to deliver any video service. This is due to the fact that the minimum length of the line is high as well as to the line characteristics. Therefore, it should be of great interest if our solution could make these loops fully operational for HD video delivery. Second, in spite of their non null minimum length, loop3, loop5 and loop6 have an initial eligibility distance of 1680, 1910 and 1790 meters, respectively. Third, loop1 and loop2, which don't have any minimum length, have initial eligibility distances of 1410 and 1720 meters respectively. Note that the difference of initial eligibility distance between the loop1 and loop2 models is due to the variability of the section size.

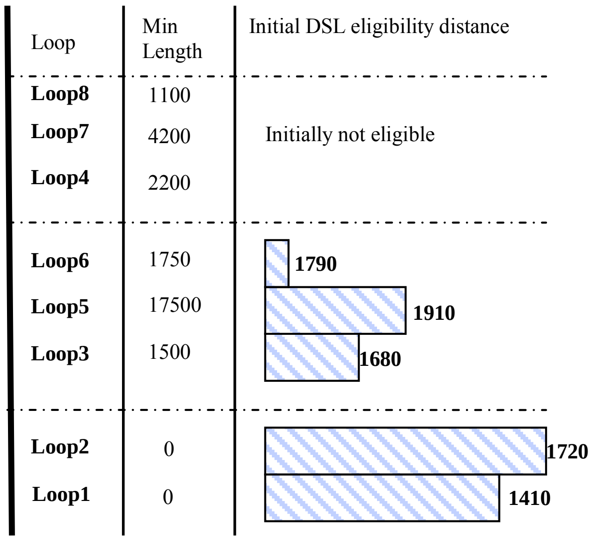


Fig. 3. Initial ADSL2 eligibility distance for standardized ETSI models.

IV. The Optimization Algorithm

In this Section we describe the optimization algorithm which constitutes the core of the proposed JSCC architecture. The aim of the optimization algorithm is to determine both transrating and bit and power distribution parameters, such that the high definition TV services become available to eligible customers, located beyond the initial eligibility distance with the best visual quality.

Let us consider a given subscriber located too far from the DSLAM to be eligible to HDTV services. The first idea could be to evaluate the maximum throughput available on the subscriber's loop depending on its length, then to adapt the incoming video bit stream to this maximum throughput thanks to transrating. However, it should be kept in mind that the objective is to provide the best video quality to the end user, given that this best video quality does not always correspond to the maximum available bit rate but stays close to it. Hence, we will aim at transmitting HDTV contents with maximal visual quality and at a bit rate D_T lower than (or equal to) maximum throughput, under given BER constraints. As explained in the previous section, the BER is chosen to verify QEF conditions, so the overall video distortion is only due to transrating.

The complete flow chart of the optimization algorithm is presented in Fig. 4 and detailed in the following paragraphs. This flow chart corresponds to the pseudo-code given in Table I. The inputs and outputs of this algorithm are the ones previously mentioned in Fig. 1. Clearly, the optimization process allows maximizing the received video quality at the subscriber's home. It is of low computational complexity and can be easily implemented at the central office.

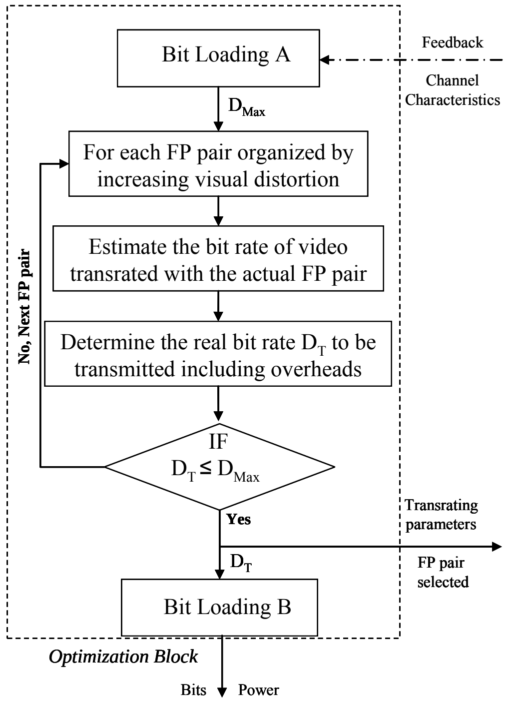


Fig. 4. Flow chart of the proposed optimization algorithm.

The optimization algorithm can be divided into three steps:

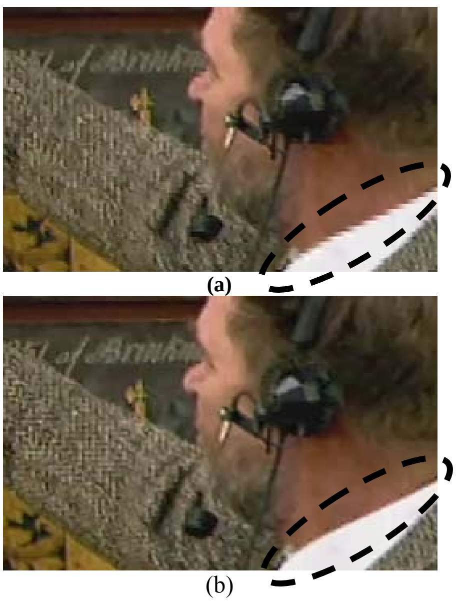
- We use a first bit and power loading algorithm (bit loading A), which is a modified version of the one presented in [12] to determine the maximum bit rate D_{Max} under the following constraints: $P = P_{\mathrm{Max}}$, $BER = BER_{\mathrm{QEF}}$.
- The optimal (FP_{intra}, FP_{inter}) couple is estimated, which leads to the best video quality with respect to the D_{Max} bit rate constraint.
- The second bit and power loading algorithm (bit loading B) minimizes the transmitted power under the following given constraints: BER equals to BER_{QEF}, and the transmitted bit rate is equal to D_{T}.[1]

It should be noted that only two bit and power loading stages are necessary, which limits the complexity of the optimization process. The average running time for most bit loading algorithms is about 1ms for 256 subchannels [18], which is negligible compared to the transrating. Moreover the bit loading is performed once during the initialization stage.

[1] Please note that, here, D_{T} accounts for bit rate overhead due to encapsulation and FEC processes.

Optimal transrating parameters, i.e. transrating parameters obtained at the second step which lead to maximal visual quality for a bit rate lower than $D_{\max}$, are obtained in an efficient way by ranking all (FP_{intra}, FP_{inter}) couples by increasing visual distortion (distortion values are obtained from Fig. 2). For each couple, the estimated bit rate after transrating is obtained straight from a bit rate model (see (1), Section V-B). The real bit rate is deduced from this bit rate estimation by accounting for bit rate overhead due to encapsulation and FEC processes. The optimal couple is the first couple for which the corresponding bit rate is lower than (or equal to) D_{Max}.

Finally, we see in Section II that the proposed transrating algorithm is controlled by the two parameters called FP_{intra} and FP_{inter}, each varying from 1 to 16. This leads theoretically to 256 possible couples. However, the transrating operation results inevitably in a visual distortion. If too high, this distortion might severely affect the QoE level of the end user. Consequently, it is necessary for DSL operators to use an additional criterion to avoid the transrated video signals to be over distorted. As illustrated in Fig. 2, the visual distortion due to transrating remains negligible when FP_{inter} varies. Therefore, only variations of the FP_{intra} parameter are considered to establish this criterion. Moreover, we can note in Fig. 2 that the normalized MSE value dramatically increases when FP_{intra} value is inferior to 3. To confirm this analysis, visual experiments were conducted on a large panel of transrated sequences belonging to the two groups (see Section II.A) to evaluate how the video quality varies as FP_{intra} decreases. Tests results show that in all cases the perceived video quality cannot be considered satisfactory for FP_{intra} values inferior than 3. This is illustrated in Fig. 5 for the *Shields* test sequence.



(a)

(b)

Fig. 5. Illustration of the visual impact of the transrating distortion (enlarged part): (a) $FP_{\text{intra}} = 2$; (b) $FP_{\text{intra}} = 3$.

Fig 5.a) shows that transrating using $FP_{\text{intra}} = 2$ results in an image with very annoying edge artifacts (see inside the area delimited by a dashed line). In Fig 5.b), however, the transrated video signal using $FP_{\text{intra}} = 3$ is visually more pleasant, without noticeable artifacts. Such FP_{intra} limit value is in accordance with human vision frequency sensitivity [19]: when FP_{intra} is equal to 3, at least one transform coefficient is available (if present) in both the horizontal and vertical directions and the frequency content of each transrated block is guaranteed to be well balanced. Based on these statements, we suggest that the transrating process was restricted to 224 possible couples, with FP_{intra} values being higher or equal to 3, and $1 \leq FP_{\text{inter}} \leq 16$.

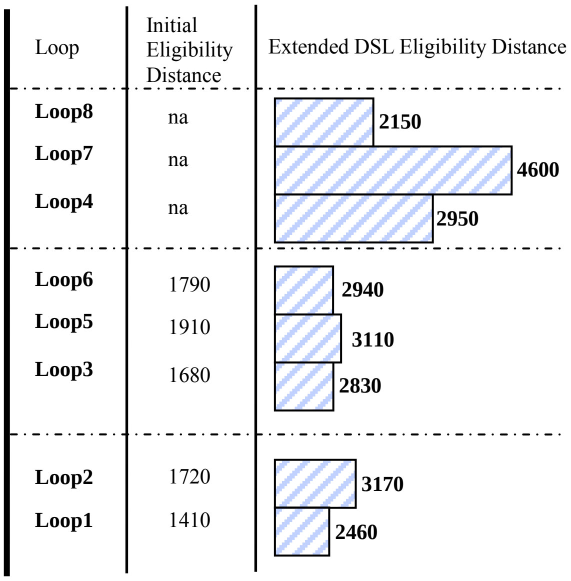
V. Results and Discussion

First, we evaluate the proposed solution in terms of maximum DSL coverage extension for high definition TV services, corresponding to the maximal distance between the DSLAM and a subscriber's home that can be reached after transrating and transmission based on the optimization algorithm. It can also be evaluated in terms of received video quality. Both evaluations are presented in this Section.

A. *Extended Coverage Results*

Fig. 6 presents the new DSL coverage areas for HDTV services after transrating and transmission based on the proposed optimization algorithm. It also shows that our JSCC architecture provides a very significant eligibility extension of more than one kilometer for loop models 1, 2, 3, 5 and 6, ranging from 1050m for Loop1 to 1450m for Loop2. The average coverage extension equals 1200 meters for these loop models.

Let us consider for example a subscriber line having the technical characteristics of the Loop1 model. Such line is characterized by an initial eligibility distance of 1410 meters, i.e., if a subscriber is located farther away than 1410 meters from the DSLAM, he/she cannot receive HDTV services at corresponding bit rates with satisfying QoS level. Consequently, he/she is not eligible to HDTV services over ADSL2. As illustrated in Fig. 6, the proposed algorithm allows extending the eligibility distance from 1410 meters to 2460 meters with an image degradation which remains acceptable. The coverage area can be defined as the area of the circle centered on the DSLAM. So, for the Loop1 model, the increase of the radius from 1410 meters to 2460 meters corresponds to an extension of the coverage area of about 200%, and should allow to deliver HDTV contents to a significantly greater number of end-users clients, albeit with a slightly reduced quality.



***n/a:** non-available because initially not eligible for HDTV services*

Fig. 6. Maximum extended DSL eligibility distance results.

In addition, it is essential to note that the loop models 4, 7 and 8, which don't provide initially sufficient throughput to deliver HDTV services, are now available for high definition video delivery over ADSL2. This constitutes a significant result for DSL operators, since numerous subscribers become newly eligible and can now benefit from HDTV services over DSL with satisfying video quality.

As explained previously, DSL coverage extension for HDTV services is made possible thanks to transrating, but at the expense of visual quality. Nevertheless, the reception video quality inside the new coverage area is known to be maximal in terms of normalized MSE thanks to the optimization algorithm. Fig. 7 presents the evolution of the normalized MSE values as a function of the distance from the DSLAM (line length) for three different loop models: Loop1, Loop3, and Loop6. Similar results were obtained with the other loop models. As we can see on the figure, the general behavior of the Normalized MSE as a function of the distance is similar for all the loops models. Changing the loop model will influence the initial eligibility distance because of the cable section size or the presence of bridge taps. Obviously, we verify that the visual distortion increases as the line length increases: in this case, the available throughput becomes more and more limited and consequently the transrating needs to be stronger. It should be noted also that the evolution of the distortion curves is similar for all loop models: the distortion increases slowly up to point noted A and then evolves more quickly. The slowly varying first part of the curve represents a coverage extension of about 800 meters with very good video quality.

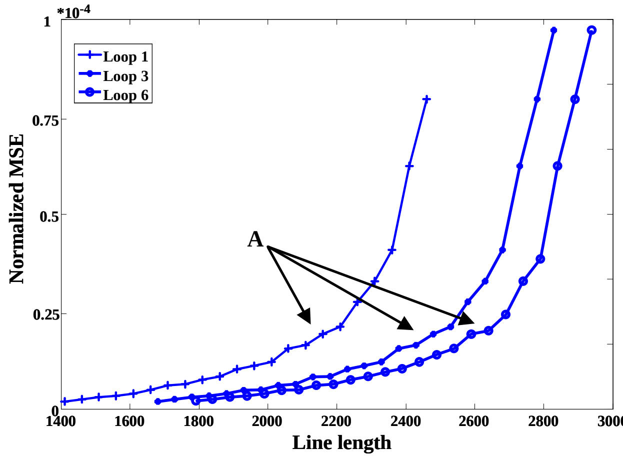


Fig. 7. Evolution of NMSE distortion metric as a function of line length.

B. *Results Analysis*

To analyze this pattern of evolution, we consider the performance of our transrater in terms of bit rate reduction. The evolution of the output bit rate as a function of FP_{intra} and FP_{inter} transrating parameter values has been computed for a wide number of 720p video sequences. The estimated bit rate noted $\hat{B}$ is then modeled in a simple way thanks to the following expression described in [9]:

$$\hat{B}(FP_{\textbf{intra}}, FP_{\textbf{inter}}) = 0{,}5\log_2(FP_{\textbf{intra}}.FP_{\textbf{inter}}) + 4 \qquad (1)$$
$$\text{where: } 1 \leq FP_{\textbf{intra}} \leq 16,\ 1 \leq FP_{\textbf{inter}} \leq 16$$

Fig. 8 presents the evolution of the bit rate of transrated video when both FP_{intra} and FP_{inter} transrating parameters vary from 1 to 16 and for loop1 model. On this figure, we also represent by arrows the series of transrating parameters couples selected by the optimization algorithm in increasing line length order. When increasing the line length of 50 meters, the arrow moves from one couple to a neighbor couple according to the optimization result. To achieve the necessary bit rate reduction, while keeping highest video quality, the FP_{inter} parameter value is first reduced from (16, 16) until point A of coordinates (16,1). Then, it becomes necessary to decrease the FP_{intra} parameter value in order to further reduce the bit rate. This is accomplished by truncating the intra coefficients a lot more, thus causing a faster deterioration of the video quality. In the same way, the couples such that $FP_{\text{inter}} \geq FP_{\text{intra}}$ are never used.

Logically, the evolution of the distortion curve before point A, shown in Fig. 7, is similar to the evolution of the normalized MSE when the FP_{inter} parameter is changing only (bold dotted line in Fig. 2). After point A, this evolution corresponds to FP_{intra} changing from 16 to 3, where FP_{inter}=1 (beginning of bold solid line in Fig. 2).

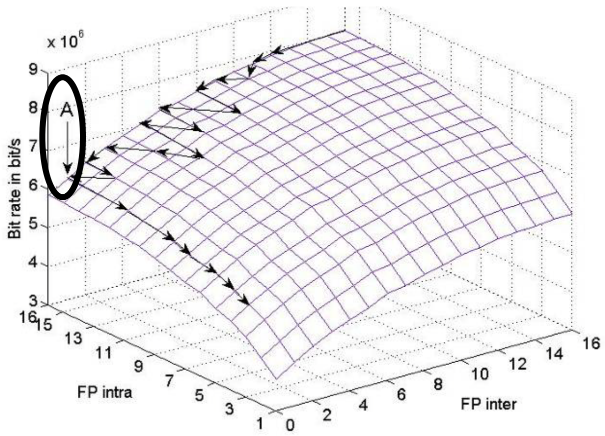


Fig. 8. Evolution of the bit rate as a function of (FPintra, FPinter) transrating parameters. Note that the arrow line represents the selected optimal couples for the loop1 model in increasing line length order.

In order to analyze received video quality in a better way, Table III gives the MSE values for all test sequences at the new maximum eligibility distance. These MSE values differ greatly for each group of sequences. The four sequences in the left column are characterized by a lot of movements/details, and consequently transrating is more severe. This leads to higher MSE values than for the group of sequences in the right column. For these six sequences, fewer frequency coefficients remain after initial encoding at 8 Mb/s, and consequently transrating with the same FP values causes less coefficients to be truncated which leads to lower MSE values.

TABLE III
MSE RESULTS AFTER MAXIMUM COVERAGE EXTENSION

High temporal/spatial activity video sequences		Low temporal/spatial activity video sequences	
Sequence	MSE	Sequence	MSE
Crowed run	178	*Into tree*	42
Ducks	173	*Mobcal part1*	58
Parkrun	181	*Mobcal part2*	42
Park Joy	180	*Old Town*	36
		Shields	59
		Stockholm	50

These MSE values must be compared to the initial MSE value, which are zero except for loop models 4, 7 and 8 which were not eligible initially.

Table IV presents the initial MSE values for the *Shields* sequence transmitted at the minimum distance for these three loop models. Similar results were obtained with other sequences. Again, we would like to put the emphasis on the fact that subscribers associated these loops models had initially no access to the HD video services, but now they do, starting with a reduced but quite acceptable perceptual quality as illustrated by the low MSE values reported in Table IV. We believe that this constitutes a great advantage of our JSCC architecture. Thanks to this solution, DSL operators will have the possibility to significantly increase their customers' base.

TABLE IV
MSE FOR INITIALLY NON ELIGIBLE LOOP4, LOOP7 AND LOOP8

Loop model	Distance	MSE
Loop4	2200	3.9
Loop7	4200	10.5
Loop8	1100	1.3

Finally, it should be noted that the complexity of the proposed solution is limited and compatible with real-time implementation of a complete end-to-end JSCC architecture. The optimization process requires two consecutive bit allocations instead of one in [5]. Nevertheless, we use a low complex bit loading which only introduces very small computational overhead. Indeed, the complexity of the overall architecture is mainly related to the transrating stage which has been accordingly developed not very complex, as explained in Section II.

VI. CONCLUSION

We have proposed an original JSCC architecture for coverage area extension of high definition TV services over ADSL2. This is based on efficient H.264/AVC video transrating and multicarrier modulation and includes an optimization algorithm which minimizes the perceptual distortions of the high definition contents transmitted at the end user's side. The complete solution is original and introduces several novelties in H.264/AVC transrating, ADSL2 transmission as well as optimization processes compared to previous work [5]. Simulation results clearly show that the proposed scheme can increase the coverage area for DSL2 high definition video distribution by about 1200m by jointly adapting both source coding and transmission parameters. Moreover, thanks to the proposed solution, a subscriber whose copper line is of poor quality, such that initially there is no possible access to HDTV services, can become eligible to these services with reduced but satisfying video quality, at last. The proposed video transmission scheme is of low computational complexity and so can be implemented at the central office. Clearly, it constitutes an attractive solution for DSL operators, because they can offer HDTV services to an increased number of subscribers at prices varying according to the available video quality.

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