A NEW TRANSMISSION STRATEGY FOR SCALABLE MULTIMEDIA DATA ON OFDM SYSTEMS

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ABSTRACT

We consider the real time transmission of scalable multimedia data with a constrained end-to-end Quality of Service. In the Rayleigh channel case, the resulting optimization problem is solved by the Flexible Transmission (FT) algorithm. This resources allocation strategy allows to transmit the sole layers with guaranteed QoS at the receiver. Here, we propose to extend the FT method to the OFDM case. The resulting algorithm is incorporated into a multimedia application using MPEG-4 speech frames, in order to compare the performance of our approach with those of other unequal error protection schemes. Simulation results show the efficiency of our solution and its adaptivity to frequency channel selectivity.

1. INTRODUCTION

The growing demand on multimedia content delivery has reinforced the necessity of efficient transmission systems dedicated to multimedia data. These systems have to comply with the diversification of the transmission networks, and more particulary the emergence of new wireless and xDSL communication technologies, and the development of new multimedia scalable coding tools. Their design is related to the major constraint of *end-to-end Quality-of-Service* (QoS). This QoS reflects the distortion introduced on the transmitted data by the source coding and the transmission operations. It is defined by the application (streaming, ...) with respect to the source coding algorithm and the reception device (mobile phone, PC...). Thus, our research study context[1] is focused on the development of an efficient multimedia transmission scheme that ensures the QoS for any source coding tools and application scenarios.

Regarding State-of-The-Art strategies, including joint source-channel coding [2], the design of such a system requires the adaptation of each aspect of the transmission scheme (from source coding algorithm to resources allocation policy) with respect to the channel condition and the reception device. First, the source coding algorithm choice can be simplified when the source encoding technique is scalable. Indeed, the bitstream frames are split into *layers*, corresponding to the compressed version of the same multimedia data with different rates and different source coding qualities. Therefore, the source distortion can be directly related to the number of transmitted layers, without changing the source coding algorithm. Furthermore, this scalable structure establishes a hierarchical organization of the emitted data into sensitivity classes to channel transmission errors[2]. This error sensitivity, known to be unequal [3], requires to adapt (even optimize) the transmission resources with respect to the emitted data sensitivity, channel conditions and system constraints, yielding Unequal Error Protection (UEP) schemes. State-of-The-Art transmission strategies have two major drawbacks in our application scenario. First, most strategies [4] do not take advantage of the degree of freedom offered by the scalable encoding process on the source rate adaptation. Second, most resources allocation procedures [3, 5] are based on the minimization of an empirical distortion metric over possible resources. This metric is often tabulated with respect to a considered scalable encoder for various transmission conditions. Nevertheless, it is a complex task, which can become costfull when the number of coding tools supported by the transmission scheme increases. Moreover, the considered optimization criterion is not relevant in our research study context, where the end-to-end QoS is specified by the application and belongs to the system constraints.

In [6], we propose a resources allocation policy, called Flexible Transmission (FT), suitable for this application context in the Rayleigh channel case. It is based on: (1) an original optimization criterion, that maximizes the source rate (i.e. the number of transmitted layers) under QoS and load constraints and (2) a simple expression of QoS as Bit Error Rates (BER) required by each layer [4]. In this paper, we propose an extension of this algorithm to the Orthogonal Frequency Division Multiplexing (OFDM) context, taking into account the scalable property of the transmitted data and the specificities of the OFDM channel. The adapted transmission parameters are chosen with respect to the current channel state information and the system constraints, provided by a cross-layer strategy.

The paper is organized as follows. Section 2 describes the transmission system. Section 3 sums up the FT resources allocation policy designed for Rayleigh channels and details its expansion to the OFDM case, yielding the FT-OFDM procedure. Section 4 draws up the algorithm efficiency through an application example (MPEG-4 audio streaming for telephony over ADSL). Simulation results are presented and compared to standard protection schemes. Section 5 concludes by summing up the major contributions of the proposed method.

2. REFERENCE TRANSMISSION SYSTEM DESCRIPTION

The reference transmission system is presented in figure 1 and is detailed in the following subsections.

2.1 Source coding parameters

Thanks to the scalable encoding process, multimedia data are represented by a bitstream structured in frames or Transmission Units (TU). These frames are split into I_{max} layers: one base layer and $I_{max}-1$ enhancement layers, denoted by $\{\mathscr{L}_i\}_{i\in[\![1,I_{max}]\!]}$ with decreasing importance degrees. The length of the i-th layer (in bits) is equal to a proportion p_i of the total frame length, denoted by N, so that: $\sum_{i=1}^{I_{max}} p_i = 1$. The importance degree affected to each layer characterizes [2]:

- the weight of each layer on the distortion D_s introduced by the source encoding process on the reconstructed multimedia data: the more enhancement layers are used during the source decoding, the smaller the source distortion is.
- the unequal sensitivity of each layer to channel transmission errors and therefore their weight on the distortion D_c introduced by the channel on the decoded multimedia data.

As proposed by [4], we suppose that the error sensitivity of each layer i can be featured by a bit error probability value, denoted by B_i. This value is defined according to some perceptual quality criterion so that, when the BER affecting the transmission of the i-th

[1]This study is part of the FP6/IST project M-Pipe [1] and is co-funded by the European Commission.

[2]We assume that the source decoder is not robust (as it is the case for most existing standards).

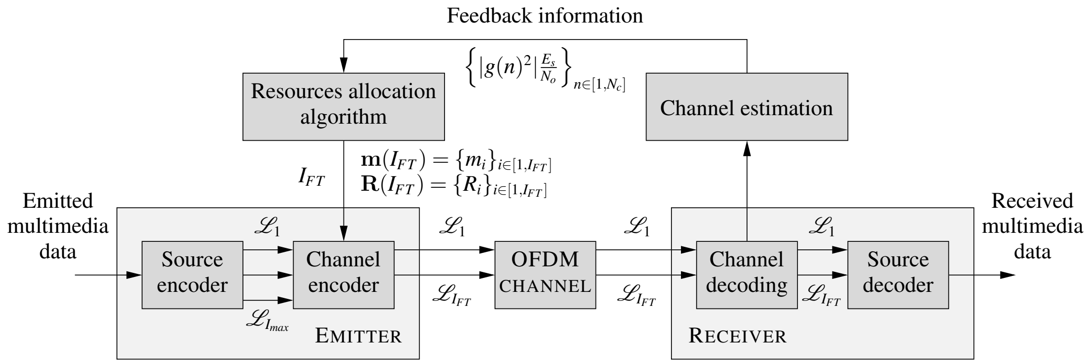

Figure 1: Transmission scheme piloted by the FT algorithm.

layer is lower than B_i, the source decoding of this layer has no (or few) influence on the channel distortion D_c. The set of $\{B_i\}_{i\in[1,I_{max}]}$ is then directly related with the end-to-end QoS of transmission systems: ensuring the QoS consists in ensuring the BER affecting each layer i, denoted by BER_i, to be lower than B_i.

2.2 Transmission model

2.2.1 Available resources at the emitter

At the emitter, source data are first encoded with a channel coding, chosen among rate-compatible channel encoder [7]. In this paper, we use *Rate Compatible Punctured Convolutional* (RCPC) codes, but it could easily be extended to any other rate-compatible encoding process (such as turbo codes). Thanks to the puncturing process, using a mother convolutional coding rate $R_1 = \frac{1}{N_1}$, different coding rates R_l are available: $R_l = \frac{P}{P+l}$, with $1 \leq l \leq (P-1)N_1$, where P is the puncturing period.

The coded data are then mapped with one of the available signalling constellations. These constellations are BPSK or 2^{2m}-QAM with constant symbol energy E_s. The order m denotes the number of bits per symbol and will refer to a specific modulation choice ($m = 1$ for BPSK, $m = 2$ for QPSK, etc...).

2.2.2 OFDM channel

Thanks to a long enough cyclic prefix, the frequency selective channel is turned into scalar channel gains $\{g_n\}_{n\in[1,N_c]}$ on the N_c different OFDM carriers: the Signal-to-Noise-Ratio (SNR) for the n-th sub-channel is $|g_n|^2\frac{E_s}{N_o}$. Assuming that the channel varies slowly, we define T as the number of periods while channel time response is constant. The maximum symbol load per coded TU, S_{max}, is chosen equal to TN_c and is defined at the physical layer. It refers to the physical payload. Moreover, we assume that our OFDM system is synchronization error free.

2.2.3 Receiver

At the receiver, after Zero-Forcing equalization, a soft output maximum *a posteriori* demapper is used to compute soft decision values, applied as input of a classical Viterbi decoder. A channel estimation module is linked with the emitter through a feed-back channel so to inform him of the estimated channel conditions (i.e. $\left\{|g_n|^2\frac{E_s}{N_o}\right\}_{n\in[1,N_c]}$). The feedback channel and the channel estimation at the receiver are supposed perfect and the bitstream header transmission is assumed to be error-free.

3. THE PROPOSED RESOURCES ALLOCATION PROCEDURE

Regarding the system characteristics, the resources allocation procedure aims at choosing the adapted system parameters that ensures the QoS constraints. The system parameters are the number of transmitted layers, denoted by I_{FT}, the modulation order m_i and the coding rate R_i for each transmitted layer $\mathscr{L}_i$ with $i \in [1, I_{FT}]$. Furthermore, the QoS constraints are stated by the $\{B_i\}_{i\in[1,I_{max}]}$ bounds.

The proposed strategy is based on the key idea that the enhancement layers should not be transmitted while the previous transmitted layers are not sufficiently protected to respect the QoS bounds. Thus, the proposed allocation policy yields to the adaptation of the number of transmitted layers I_{FT} at the channel coding stage. With scalable data, the I_{FT} parameter is directly related to the source rate $\mathbb{R}_s(I_{FT})$:

$$\mathbb{R}_s(I_{FT}) = \sum_{i=1}^{I_{FT}} p_i N. \quad (1)$$

As a result, the allocation strategy is related to an optimization problem:

- maximizing the source rate (or I_{FT}) and finding the associated modulation orders $\mathbf{m}(I_{FT}) = \{m_i\}_{i\in[1,I_{FT}]}$ and channel coding rates $\mathbf{R}(I_{FT}) = \{R_i\}_{i\in[1,I_{FT}]}$ over the possible transmission parameters combinations, denoted by $\mathscr{P}$,
- with respect to QoS bounds, data throughput constraints (given by the symbol rate S_{max}) and channel transmission conditions.

In this section, we briefly remind the FT resources allocation procedure designed to solve this optimization problem in the Rayleigh channel case and expound its adaptation to the OFDM context.

3.1 FT algorithm: Rayleigh channel case

In the Rayleigh channel case, the channel conditions are given through an unique $\frac{E_s}{N_o}$ value.

Since the bit error rate of the i-th layer is a function of the SNR, the modulation order m_i and the coding rate R_i, we will denote it: $\text{BER}_i\left(\frac{E_s}{N_o}, m_i, R_i\right)$. Moreover, we define $S_i(m_i, R_i) = \frac{p_i N}{m_i R_i}$, the spectral efficiency of $\mathscr{L}_i$ (that is the number of symbols transmitted within the coded $\mathscr{L}_i$). The optimization problem related to the allocation policy can now be set by the following equation, given the SNR value $\frac{E_s}{N_o}$:

$$\begin{cases} (I_{FT}, \mathbf{m}(I_{FT}), \mathbf{R}(I_{FT})) = \arg \max\limits_{(I,\mathbf{m}(I),\mathbf{R}(I))\in\mathscr{P}} \mathbb{R}_s(I), & \text{(2a)} \\ \forall i \in [1, I], BER(\frac{E_s}{N_o}, m_i, R_i) \leq B_i & \text{(2b)} \\ S_{TU} = \sum_{i=1}^{I} S_i(m_i, R_i) \leq S_{max} & \text{(2c)} \end{cases}$$

where S_{TU} represents the transmission load of the whole transmitted data (i.e. the number of symbols used to transmit the coded layers $\mathscr{L}_1$ to $\mathscr{L}_{I_{FT}}$).

The FT solution, we bring to this problem in [6], is an iterative algorithm. It processes from the first layer $\mathscr{L}_1$ to the last one by progressively filling the coded TU until the maximum symbol load S_{max} is reached. During the process of the I-th layer $\mathscr{L}_I$, the algorithm selects the modulation and coding rate pair (m_I, R_I) that minimizes the spectral efficiency S_I of $\mathscr{L}_I$ over the possible pairs which respect the BER constraint (2b). The matched number of transmitted classes is obtained as soon as $\sum_{i=1}^{I} S_i > S_{max}$: indeed $I_{flex} = I - 1$.

3.2 Adaptation to the OFDM case

We now propose to extend the FT algorithm to the OFDM context. The resources allocation procedure has to define, besides the matched number of transmitted layers and the matched transmission parameters, an adapted carrier allocation policy. The allocation policy has two purposes:

- Take the channel state into account: this state is now given through multiple SNR values on the different sub-carriers, that is $\left\{|g_n|^2 \frac{E_s}{N_o}\right\}_{n\in[1,N_c]}$. This SNR variability prevents us from directly applying the FT algorithm, since it has been designed for an unique SNR input value. Using the FT algorithm with the average SNR doesn't satisfy the QoS on the carriers with lower SNR than the average and using the algorithm with the minimal SNR value results in a severe overprotection (yielding a waste of resources).
- State the assignment between OFDM sub-channels, layers and resources as presented in figure 2: additional parameters have therefore to be considered by the allocation policy, such as the set of OFDM sub-channels, denoted by $\mathscr{N}_i$, used to transmit each layer $\mathscr{L}_i$ with modulation m_i and coding rate R_i.

3.2.1 *Adapted SNR values choice to process FT*

The OFDM SNR variability modifies the QoS constraints formulation (2b). Indeed, the BER_i affecting the i-th layer transmission now depends on the BER values characterizing each OFDM sub-channels assigned to $\mathscr{L}_i$. We will denote these BER values by $\left\{\mathrm{BER}_n^{(i)}\right\}_{n\in\mathscr{N}_i}$. They can be expressed with respect to the SNR values $\left\{|g_n|^2 \frac{E_s}{N_o}\right\}_{n\in[1,N_c]}$, the modulation order m_i and the coding rate R_i. For the i-th layer, the QoS constraint given by equation (2b) can be simply ensured by considering the worst BER configuration: this configuration is achieved by the OFDM sub-channel assigned to $\mathscr{L}_i$ with the lowest SNR or channel gain (i.e. $\min_{n\in\mathscr{N}_i} |g_n|^2$). The QoS constraints will then be stated by the following overestimation:

$$\forall i \in [1,I], \mathrm{BER}_i \approx \mathrm{BER}^{(i)}_{n_{min}(i)} \leq B_i \text{ with } n_{min}(i) = \arg\min_{n\in\mathscr{N}_i} |g_n|^2 \tag{3}$$

This overestimation, sometimes too severe, can be relaxed considering that only a percentage α (for instance 80%) of the layer $\mathscr{L}_i$ comply with the QoS requirement. Thus, denoting $\mathscr{N}_i^{(\alpha)}$ the set of OFDM sub-channels used to transmit the $\alpha p_i N$ first bits of $\mathscr{L}_i$, the QoS constraints will finally be formulated considering the worst BER configuration on the $\mathscr{N}_i^{(\alpha)}$ OFDM sub-channels distribution, that is:

$$\forall i \in [1,I], \mathrm{BER}_i \approx \mathrm{BER}^{(i)}_{n^{(\alpha)}_{min}(i)} \leq B_i \text{ with } n^{(\alpha)}_{min}(i) = \arg\min_{n\in\mathscr{N}_i^{(\alpha)}} |g_n|^2 \tag{4}$$

3.2.2 *Sub-carriers allocation policy*

The OFDM SNR diversity can also be exploited to enforce the unequal protection of the data. State-Of-The-Art resources allocation algorithms [4] propose to sort the sub-channels in a decreasing SNR order: thus, the best sub-channels (in terms of SNR)

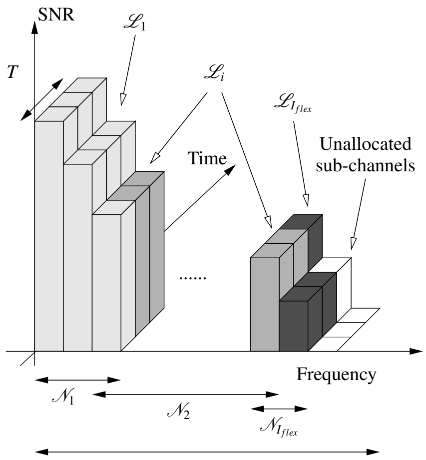

Figure 2: Sub-channels allocation policy.

can be allocated to the most important layers, while the least one are used to transmit the least important layers. We propose to follow the same allocation policy in the extended FT algorithm. Supposing the OFDM sub-channels sorted with decreasing SNR value, the layer $\mathscr{L}_1$ coded with its associated protection scheme $(\mathbf{R}_1(I_{FT}), \mathbf{m}_1(I_{FT}))$ will be transmitted over the best SNR sub-channels, whose index belongs to $\mathscr{N}_1 = \left[1, \left\lceil \frac{p_1 N}{m_1 R_1 T} \right\rceil\right]$. On the other side, $\mathscr{L}_{I_{FT}}$ coded with $(R_{I_{FT}}, m_{I_{FT}})$ will be transmitted with the least SNR ones, whose index only depend on the previous layers mapping.

3.2.3 *Problem formulation and solution*

The FT procedure in the OFDM context deals with finding the adapted number of transmitted layers I_{FT}, the adapted modulation order $\mathbf{m}(I_{FT})$, coding rates $\mathbf{R}(I_{FT})$ and sub-carriers distribution $\mathscr{N}(I_{FT}) = \{\mathscr{N}_i\}_{i\in[1,I_{FT}]}$ (starting from the best SNR sub-channels to the least ones) over the possible configurations set $\mathscr{P}'$ that satisfy QoS and system constraints. It can now be stated by the following optimization problem:

$$\begin{cases} (I_{FT}, \mathbf{m}(I_{FT}), \mathbf{R}(I_{FT}), \mathscr{N}(I_{FT})) = \arg\max_{(I,\mathbf{m}(I),\mathbf{R}(I),\mathbf{N}(I))\in\mathscr{P}'} \mathbb{R}_s(I) & \text{(5a)} \\ \forall i \in [1,I], \mathrm{BER}^{(i)}_{n^{(\alpha)}_{min}(i)} \leq B_i, \text{ with } n^{(\alpha)}_{min}(i) = \arg\min_{n\in\mathscr{N}_i^{(\alpha)}} |g(n)|^2 & \text{(5b)} \\ S_{TU} = \sum_{i=1}^{I} S_i(m_i, R_i) \leq S_{max} & \text{(5c)} \end{cases}$$

The solution to this problem, called FT-OFDM, follows the same principles as the iterative FT algorithm design for the Rayleigh channel case. It processes from the first layer $\mathscr{L}_1$ to the last one by progressively filling the coded TU until the maximum symbol load S_{max} is reached. During the process of the I-th layer $\mathscr{L}_I$, the algorithm computes for each possible configuration (m_i, R_i) the required sub-channels distribution $\mathscr{N}_I$ and the associated subset $\mathscr{N}_I^{(\alpha)}$ (with respect to the sub-channels distributions of the previous layers). Then, the targeted SNR value ($\min_{n\in\mathscr{N}_i^{(\alpha)}} |g(n)|^2 E_s/N_o$) over the considered $\mathscr{N}_I$ sub-channels is computed to evaluate the respect of the BER constraint defined by equation (4). The $(m_I, R_I, \mathscr{N}_I)$ satisfying the QoS constraint and minimizing the spectral efficiency S_I of $\mathscr{L}_I$ over the possible triplets is finally selected. The matched number of transmitted classes is obtained as soon as $\sum_{i=1}^{I} S_i > S_{max}$, with $I_{FT} = I - 1$.

4. EXPERIMENTAL RESULTS

4.1 Application to MPEG-4 speech scalable data

An application of the proposed algorithm to the transmission of scalable speech data is at stake to evaluate the algorithm performance. Among the several source coding tools of the MPEG-4 standard [8], we focus on the CELP encoder with the MultiPulse Excitation and the Bit-Rate Scalability (BRS) tools. This coder can compress a speech signal sampled at 8 kHz with a scalable bitstream characterized by a 12 kbps bitrate and a 4-layers structure. The base layer (with 120 bits per frame) is generated with a core CELP encoder operating at 6 kbps using a speech production model based on an excitation signal passed through an auto-regressive filter. The BRS tool provides the 3 remaining layers and adds a 2 kbps per layer information, that refines the excitation signal description (with 40 bits per frame and per layer). The whole bitstream is finally constructed as the succession of frames, each containing $N = 240$ bits.

4.2 Test plan

4.2.1 Audio scalable data parameters

According to the chosen scalable coder structure, the layer proportions are the following: $p_1 = \frac{1}{2}$, $p_2 = p_3 = p_4 = \frac{1}{6}$. We assume that the QoS to ensure for our application can be described by the following BER upper bound of each class: $B_1 = 3.10^{-3}$, $B_2 = 4,6.10^{-3}$, $B_3 = 8.10^{-3}$, $B_4 = 9.10^{-3}$.

4.2.2 Standard link parameters

RCPC codes are generated from a mother convolutional with rate $\frac{1}{3}$ and enumerator polynoms $G_1 = [133]_8, G_2 = [145]_8$ and $G_3 = [175]_8$. The puncturing period is chosen equal to 8. The list of available code rates $\mathscr{R}$ is finally: $\mathscr{R} = \left\{\frac{8}{9}, \frac{8}{10}, \frac{8}{12}, \frac{8}{14}, \frac{8}{16}, \frac{8}{18}, \frac{8}{20}, \frac{8}{22}, \frac{8}{24}\right\}$.

The modulation schemes choice are limited to BPSK and 2^{2m}-QAM with $m \in \{2,3,4\}$ (QPSK to 64-QAM).

4.2.3 OFDM parameters

The OFDM parameters are the following: the number of sub-channels in an OFDM symbol is fixed to $N_c = 120$, the OFDM coherence time is $T = 3$, yielding a maximum symbol load per coded TU $S_{max} = TN_c = 360$.

4.2.4 Performance evaluation

The FT-OFDM algorithm efficiency for the transmission of speech scalable data is evaluated by the perceived quality measurement of the decoded speech signal. This measure is computed using the Perceptual Evaluation of Speech Quality (PESQ) software, described by the IUT recommendation P.862 [9]. It compares the uncoded original speech signal to the received speech signal and results in a Mean Opinion Score (MOS), in the range $[0;4]$, that reflects the distortion introduced by the source coding and transmission operations on the decoded data. This measure is computed using a 10s-duration signal, sampled at 8 kHz, and is averaged over 50 transmissions.

The performance of the FT-OFDM algorithm will be compared to three State-of-the-Art resources allocation strategies, summed up in table 1: these strategies achieve data protection against error by applying different coding rates and modulation orders on each layer. Following the ordered subcarrier selection algorithm proposed by [4] in the OFDM channel context to minimize the average error probability affecting each layer transmission, the modulation order and the associated OFDM sub-channel distribution are chosen as follows: the layers are distributed on the sub-channel regarding their transmission error sensitivities (the higher sensitive the layer is to errors transmission, the better sub-channel SNR is used). Modulation orders are chosen so that the whole coded frame data only fill the half OFDM sub-channels (OFDM sub-channels with the lowest SNR are therefore discarded).

Scheme	Layer $\mathscr{L}_1$	Layer $\mathscr{L}_2$	Layer $\mathscr{L}_3$	Layer $\mathscr{L}_4$
Strategy 1	$R_1 = 8/24$ 16-QAM	$R_2 = 8/24$ 16-QAM	$R_3 = 8/24$ 16-QAM	$R_4 = 8/24$ 16-QAM
Strategy 2	$R_1 = 8/18$ QPSK	$R_2 = 8/12$ 16-QAM	$R_3 = 8/12$ 16-QAM	$R_4 = 8/12$ 16-QAM
Strategy 3	$R_1 = 8/16$ QPSK	$R_2 = 8/22$ 16-QAM	$R_3 = 8/16$ 16-QAM	$R_4 = 8/9$ 16-QAM

Table 1: Considered standard resources allocation strategies

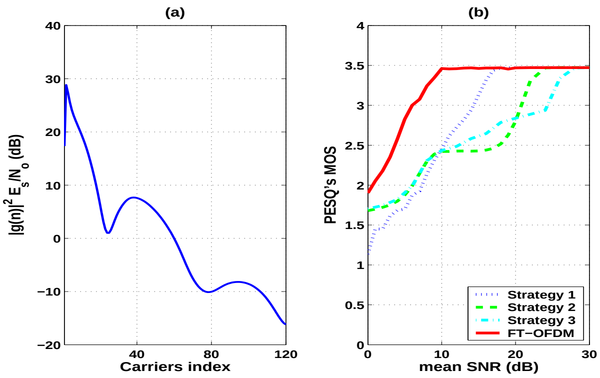

Figure 3: (a) Frequency response of the considered ADSL channel, (b) Performances of the different resource allocation schemes with respect to the ADSL channel mean SNR.

4.3 Obtained results

4.3.1 Performance in terms of perceived quality

With aim at evaluating the performance of the FT-OFDM algorithm on the perceived quality of the decoded data, we use an ADSL channel model, measure through experimental simulations on real ADSL channel. The frequency response of the considered ADSL channel model is presented in figure 3 (a).

The mean MOS obtained with the PESQ algorithm after 50 transmissions of a speech signal are drawn in figure 3 (b) for the three reference resources allocation strategies and the FT-OFDM algorithm with respect to the ADSL channel mean SNR value. These curves prove the efficiency of our resource allocation procedure: the FT-OFDM algorithm points out that the QoS can be guarantee until a 7 dB mean SNR. In this operating range, the perceived quality of the decoded speech data obtained with FT-OFDM slowly varies in comparison to the reference schemes. Moreover, the ADSL channel mean SNR value, from which the maximum MOS value (obtained when the speech data are only distorted by the source coding process) is reached, is 7 dB lower than those of the State-Of-The-Art strategies. This improvement is the result of the FT-OFDM adaptivity on channel conditions contrary to the reference schemes.

For mean SNR values lower than 7 dB, the FT-OFDM algorithm points out that the QoS can not be guarantee and selects a resources configuration by default. Since the QoS constraints are no more satisfied, the FT-OFDM MOS values roughly decrease and reach similar performance to those of the reference schemes (due to the sub-carriers distribution).

4.3.2 Resources allocation results with respect to channel dynamic

With aim at evaluating the influence of OFDM channel (and more particulary its frequency selectivity) on the FT algorithm, we design an arbitrary simulation channel with the following principles: given a mean SNR value denoted by $\overline{\frac{E_s}{N_o}}$, the SNRs $\left\{|g(n)|^2 \frac{E_s}{N_o}\right\}_{n\in[1,N_c]}$ of each OFDM sub-channel are computed in order to follow a linear

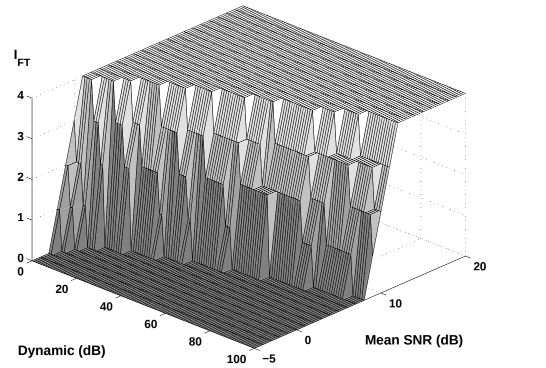

Figure 4: Variation of the number of transmitted layer I_{FT} with respect to the mean SNR and the dynamic values of the OFDM channel (with $\alpha = 100\%$).

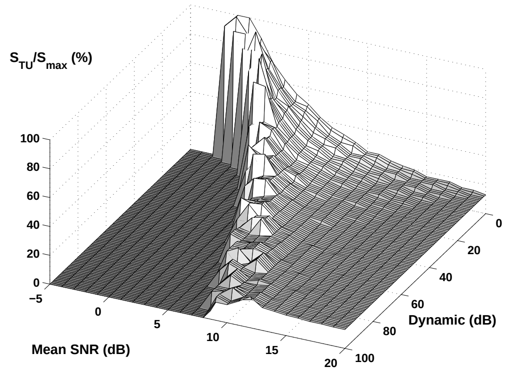

Figure 5: Allocation ratio of an OFDM symbol with respect to the mean SNR and the dynamic values of the OFDM channel (with $\alpha = 100\%$).

decrease (in dB) around the mean SNR value. In other words:

$$|g(n)|^2 \frac{E_s}{N_o}\bigg|_{(\mathrm{dB})} = \overline{\frac{E_s}{N_o}}\bigg|_{(\mathrm{dB})} + \left(\frac{\Delta}{2N_c} - n + 1\right) \text{ with } n \in [1, N_c],$$

where Δ represents the difference between the best sub-channel SNR and the least one. In this case, Δ reflects the frequency selectivity of the OFDM channel and will be referred to as channel dynamic.

Figure 4 and 5 clarify the FT-OFDM algorithm process with respect to the OFDM channel. Figure 4 presents the coding source rate (in terms of number of transmitted layer I_{FT}), computed by the FT-OFDM algorithm, with respect to the mean SNR and the dynamic values. Figure 5 draws the variation of the allocation ratio of an OFDM symbol (i.e. the ratio between the number of symbols S_{TU} used to transmit the coded data and the symbol load S_{max}) with the OFDM channel (mean SNR and dynamic).

Figure 4 exhibits the strong adaptation of the coding source rate for medium mean SNR value (between -3 and 10 dB) imposed by the QoS requirements. Moreover, the algorithm shows a strong adaptivity of the source rate to the channel dynamic: in this range, the source rate decreases when the channel dynamic increases. Indeed, the proportion of low SNR sub-carriers (compared to mean SNR value) increases with the dynamic and the QoS constraints become more restricting, since they are related to the lowest SNR value of the allocated OFDM sub-channel.

With high mean SNR values (greater than 10 dB), source coding rate is no more adapted ($I_{FT} = 4$), since the channel conditions are good enough to transmit all layers. The FT algorithm adaptivity to the OFDM channel is now deferred to the sub-channel allocation policy. Indeed, figure 5 shows that the allocation ratio of an OFDM symbol decreases with the mean SNR value increase. For high mean SNR configurations, layers transmission is concentrated on OFDM sub-channels with highest SNR. These sub-channels, thanks their high SNR, are able to support the layers transmission with high modulation orders (generally 64-QAM) and low coding rates and still respect the QoS constraints.

Finally these figures exhibit the system operating limits under the end-to-end QoS constraints: when the mean value SNR is lower than -3 dB, any QoS bounds can be ensured. Therefore, the FT algorithm decides to transmit any scalable data (by indicating $I_{FT} = 0$). These limits could be improved by relaxing the constraints (QoS or symbol load).

5. CONCLUSIONS

In this paper, we proposed a resources allocation procedure for scalable multimedia data transmission over a frequency selective channel using an OFDM system. Our solution allows not to transmit all layers in order to protect enough the transmitted ones, using a channel-adapted carriers allocation. The choice of the transmitted layers, yielding the source coding rate adaptation, is no more performed by the source encoding process but by the channel encoder.

Simulation results show the efficiency of the proposed source coding rate adaptation. It also exhibit good performance of the proposed algorithm in terms of QoS delivery and robustness to frequency channel selectivity. Thus, the proposed resources allocation strategy seems to be suitable for designing wireless or xDSL communication systems that may : (1) be strongly constrained by channel variations (outdoor, indoor channels ...), (2) require flexibility and low complexity (especially for multicast applications).

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