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MULTI-CARRIER CODE DIVISION MULTIPLEXING OF MULTI-LAYERED MPEG4 VIDEO SIGNALS FOR REAL-TIME MOBILE STREAMING APPLICATIONS

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0. Summary

In this report, a novel methodology for the efficient multiplexing and transmission of MPEG4-coded video signals over wireless networks will be presented and discussed. The proposed approach relies on the joint exploitation of variable-bit-rate (VBR) multicarrier code-division multiplexing (MC-CDM), together with MPEG4 coding with Fine-Grain-Scalability (FGS) in order to provide unequal error protection to the transmitted video stream. The innovative scheme proposed employs a shared bandwidth partitioned into orthogonal sub-channels in order to multiplex different layers of MPEG-4-coded signals. The highest number of sub-channels (and hence an increased frequency diversity) is assigned to the lowest-bit-rate base layer and the lowest number of sub-channels is assigned to the highest bit-rate enhancement layer. In such a way, base layer information contents are more protected against channel degradations than information contained in FGS enhancement layers, which can only yield a refinement of the quality of the decoded streams. A 2GHz LEO multicast satellite transmission to mobile users has been regarded as the application testbed for the proposed method. Results achieved in terms of PSNR point out that the VBR MC-CDM technique can provide better results than a conventional MPEG-4 single-layer MC-SS transmission. In the framework of a full-digital implementation of reconfigurable multimedia transceivers, the proposed VBR MC-CDM technique may be regarded as an interesting solution for reliable multimedia transmissions in mobile environments.

1. Introduction

Multimedia communications will become in the next future more and more mobile and ubiquitous. Tachikawa reported in [1] some impressive statistics about the penetration of the wireless Internet service in the Far East (more than 60 million of people access the Internet using mobile phones). Such a trend is also noticeable in USA and Europe, where the penetration rate of the mobile phones is close to saturation and users' expectations are going more and more towards new mobile multimedia services. At the present time, 3G mobile standards (i.e. UMTS and UMTS-like systems) are currently supporting services like i-mode, videophone, high-speed data access (up to 384Kb/s), video-on-demand over PDAs, etc. [1]. The evolution of next generation mobile networks (4G and beyond) will forecast as key issues higher data rates, increasing heterogeneous and asynchronous multimedia traffic fluxes, flexible network architectures, seamless services characterized by global coverage [1][2]. In such a perspective, the tight integration of terrestrial and satellite wireless network infrastructures might allow in the next future the provisioning of high-speed data transmissions (rate up to 1Gb/s) and interactive multimedia services to users spread everywhere, as clearly stated in [2]. Open problems, still hindering the efficient provision of broadband multimedia services over wireless networks, are mainly related to the tradeoff between the adoption of efficient source coding techniques, which are necessary in order to exploit efficiently the available bandwidth [3], and the severe Quality-of-Service (Q.o.S.) requirements generally imposed by commercial customers. In fact, it is known that compressed audio and video signals are generally quite vulnerable with respect to channel errors because of the use of predictive coding and Variable Length Coding (VLC) by the source encoder [3]. On the other hand, requirements of error-free transmission cannot often be achieved over wireless channels. In fact, signal propagation impairments, in particular rain fading [4] and multipath fading [5], can lower very much physical layer performances, consequently producing clusterization of channel errors, heavy packet losses, huge latency times and other drawbacks whose impact on the perceived Q.o.S. is generally dramatic. In such a framework, the provision of reliable multimedia services with controlled Quality-of-Service is surely a challenging research task. Lot of companies and research institutes have focused their attention on these aspects. The common target is to provide the transmission of multimedia contents with an *improved resilience* against channel degradations [6]. Our interest is devoted, in particular, to real-time multicast video streaming applications for mobile users. Recently, the standardization of scalable MPEG-4 coding techniques [7] brought to considerable advancements in the development of flexible and robust video streaming over unreliable channels. In particular, temporal scalability and *Fine-Grain-Scalability* (FGS) [29], which are core features of MPEG-4 standard, allow one to implement a multi-layered video coding able at addressing in effective way several technical problems inherent to video service delivery. Scalable MPEG-4 coding with adaptive rate control can actually support real-time video streaming over 3G cellular systems based on CDMA2000 standard, as shown in [8]. Various methodologies for error resilient video streaming have been proposed in literature, each one concentrates on specific aspects of the video transmission. Devised solutions can be listed into three main categories, as mentioned in [6]: i) those exploiting joint source and channel coding, ii) those invoked at the decoder upon detection of errors, iii) those which require interactions between the source encoder and decoder. Referring to source and channel coding techniques, possible solutions employ Forward Error Correction (FEC) and Automatic Repeat & Request (ARQ) mechanisms [9-10]. Solutions like these ones may involve introduction of redundancy in the bitstream, which, in some cases, could significantly lower system efficiency and throughput. For this reason it would be necessary to have dynamic mechanisms able at applying the correct amount of protection in a selective way. In fact, some parts of the transmitted bitstream carry information contents more relevant than other parts. For this reason, it is necessary to provide *Unequal Error Protection (UEP)* to the different parts of the coded video stream. Possible solutions have been proposed both for cable communication and wireless networks. In particular, the UEP concept is applied in [11] over spectrally shaped wired channels, such as DSL lines for video broadcasting and HDTV. The resulting method consists of a procedure aimed at multiplexing different layers of scalable image/video contents over a different number of DSL sub-channels. Following the basic DSL concept, the method shown in [11] enables higher bit-rate transmission (by using spectrally-efficient multi-level modulations) for those coding layers transmitted over sub-channels characterised by a higher channel-gain-to-noise-ratio (CGNR). In [12], Jil, Zhang *et al.* proposed the application of an improved MIMO-OFDM modulation scheme for reliable transmissions of FGS MPEG-4 video streams over wireless channels. Such a scheme is characterized by an adaptive power allocation made at sub-channel level in order to minimize the total distortion while satisfying the transmission power constrains.

As far as UEP techniques based on channel coding are considered, we can mention some approaches shown in recent works [13-18]. In [13], convolutional codes are employed for UEP by separately encoding single I-frames and P-frames of an H.263-encoded video sequence. Ideal interleaving/de-interleaving is supposed in [13], so that decoding errors should be regarded as independent by the source decoder. The solution proposed by Kim and Merserau in [14] consists of a FEC-based bit-plane-wise UEP algorithm aimed at providing unequal amount of FEC protection to each bit-plane of JPEG2000-coded still images and 2D/3D SPIHT-coded video sequences. Other UEP approaches have been investigated in order to allow good-quality wireless video communications and involve in most cases error resilient tools, such as: Reversible Variable Length Coding, Rate Compatible Punctured Convolutional Codes (RCPC), Reed Solomon encoding, etc. as shown in [15-17]. A different UEP approach that is based on joint source and channel coding performed at physical layer level has been proposed by Gharavi [18]. Such a methodology is aimed at providing error resilience by handling a transmission system based on a 16-QAM modulation with pilot symbols for channel estimation. The UEP is obtained in [18] by distributing VLC information amongst two partitions on the basis of the importance of the information contents carried by VLC coefficients (such a partition mechanism is implemented in the H.263 standard). Afterwards, a dual-priority transmission mechanism is implemented by exploiting the differing error resilience of the bits making up a symbol in a Gray-coded 16-QAM. Wavelet diversity is employed in [19] for transmission of compressed still images over wireless channels with UEP. Multiple copies of the wavelet coefficients are transmitted over separated channels, in order to exploit diversity against frequency-selective multipath fading. Different Reed-Solomon encoders are employed for the different subband coefficients. In such a way, UEP is exploited in addiction to diversity by adding an increasing channel coding redundancy to the lowest-resolution subband coefficients (which are characterized by the most relevant information content). Finally, we can mention the methodology presented in [20], targeted at implementing an efficient and robust TDM-based multiplexing of MPEG-4 streams provided with UEP. Two multiplexing layers have been considered for classes of MPEG-4-coded data differentiated on the basis of the expected Q.o.S. A first multiplexing layer is employed for combining in a unique stream some MPEG-4 data streams belonging to the same class of information (audio, video, and control data) and requiring the same Q.o.S. After combination, a unique error protection strategy is applied to the multiplexed data stream. The second multiplexing layer is employed for those MPEG-4 data streams characterized by specific Q.o.S. constraints. In this latter case, error protection is separately applied to each single coded stream in a differentiated way. The methodology shown in [20] allows one to reduce overhead needed for error control and is specifically targeted to transmission of multimedia streams over mobile channels.

In this work, we proposed a novel method for real-time MPEG-4-based video streaming over mobile channels affected by frequency-selective multipath fading. The core of the proposed methodology is the combination of multi-layered MPEG-4 video coding and variable-bit-rate Multi-Carrier-Code-Division-Multiple-Access (VBR MC-CDMA) transmission techniques in order to obtain a frequency-diversity-based unequal error protection for each different transmitted coding layer. The implemented MPEG-4 scalable video coding can be regarded as an extension of the MPEG-4 FGS standard. In this specific context, we developed a multi-layered encoder, which allowed us to obtain a hierarchical structure consisting of a base layer, characterized by low bit-rate and very relevant information content, and some FGS enhancement layers, each one characterized by increasing bit-rate and decreasing relevance of the information contents. VBR-MC-CDMA techniques (basically derived from the OFDM-CDMA concept [21] [22] that is the Spread Spectrum extension of the OFDM modulation) rely on the transmission of digital streams characterized by different symbol rates over a shared bandwidth partitioned into orthogonal sub-channels [22]. The lowest number of sub-channels is attributed to the highest bit-rate users, and consequently the highest number of sub-channels to the lowest bit-rate users. Orthogonal Variable Spreading Factor (OVSF) codes [28] are employed for pseudo-random access of VBR users to the common channel. The proposed transmission methodology enables both the multiplexing of different MPEG-4 coding layers of a single transmitting user (Variable-Bit-Rate Multicarrier-Code-Division-Multiplexing) and the multi-user transmission of MPEG-4-coded video streams. This latter one can be achieved by overlapping in MC-CDMA modality the coding layers of the different users over the shared bandwidth. The frequency diversity provided by the VBR MC-SS transmission, and hence the degree of resilience against channel distortions, is incremental with respect to the importance of the coding layer, as the lowest-bit-rate coding layers (characterized by the highest information relevance) are transmitted over the highest number of orthogonal sub-channels.

The proposed approach can be regarded as a substantial step-ahead in the state-of-the-art of error resilient video streaming applications and Unequal Error Protection. In fact, error resilience is implemented here by means of different degrees of frequency diversity selectively applied to different MPEG-4 coding layers. Such kind of approaches based on multicarrier modulations and flexible radio resource management will characterize "4G and beyond" wireless standards [2], profitably exploiting concepts like network reconfigurability and software-defined-radios [34]. Nevertheless being quite futuristic, no particular technical drawback can hinder a prototypical implementation of the video streaming system proposed in this paper. Recent works evidenced how MPEG-4 coding can be ported over DSP infrastructures [24], as well as full-digital implementations of MC-SS transceivers by means of Fast-Fourier-Transform (FFT) software tools [21][22].

The paper is structured as follows: Section 2 will deal about the wireless video streaming application chosen as testbed. Section 3 will be devoted to the description of the proposed Variable-Bit-Rate Multicarrier Code Division Multiplexing for the transmission of MPEG-4 coded video streams. Aspects related both to multicarrier modulation and multi-layered MPEG-4 coding will be properly evidenced. Section 4 will present selected experimental results. Finally, Section 5 will draw paper conclusions.

2. Description of the multicast satellite video streaming application

The research work presented in this paper is inserted in the framework of the activities related to the SHINES (*Satellite and HAPS integration*) project. This project, funded by the Italian Ministry of University and Scientific Research during the biennium 2002-2004 [25], is aimed at studying the integration of LEO and GEO satellite infrastructures with stratospheric High-Altitude-Platforms (HAPS) in order to provide interactive multimedia services characterised by global coverage. Various application scenarios have been forecast for SHINES [25]: one of these can be depicted by the scheme reported in Figure 1. In particular, an application of multicast video content distribution has been designed for mobile users in a vehicular scenario. This application will be the testbed for the proposed video transmission methodology. The central node of the mobile network is a LEO satellite. The bi-directional communication between mobile terminals and earth station is allowed in asymmetrical modality. The video contents are provided "on-demand" to the mobile users by the forward downlink channel. Such a multicast transmission is broadband and synchronous. On the other hand, the reverse uplink channel is devoted at collecting users' requests of video contents. Reverse uplink transmission can be regarded as narrowband and asynchronous. The collected users' requests are then forwarded to the earth station by the reverse downlink channel. Finally, requested multimedia contents are dispatched to the LEO satellite by the point-to-point forward uplink channel. The transmission of source-coded video contents is performed in real-time and "raw" modality, without any kind of packetization and retransmission of noise-altered information.

The employment of LEO satellites instead of GEO ones can be motivated by the necessity of reducing the latency time in delay-sensitive multimedia applications like multicast video streaming. We considered satellite transmission over S-band (2GHz), which is the radio spectrum portion commonly used for land mobile satellite transmission, for satellite telephony service [26] and for S-UMTS standard [27]. In the present dealing, we shall focus our attention on the forward downlink channel, which is almost critical due to the necessity of satisfying precise Q.o.S. requirements in presence of frequency-selective and signal distortions involved by multipath fading. In the architecture of Figure 1, LEO satellites might be regenerative or not without losing generality and without modifying substantially the core of the proposed approach for scalable video transmission. The adoption of regenerative satellite could be suggestible in order to improve system flexibility and reduce latencies.

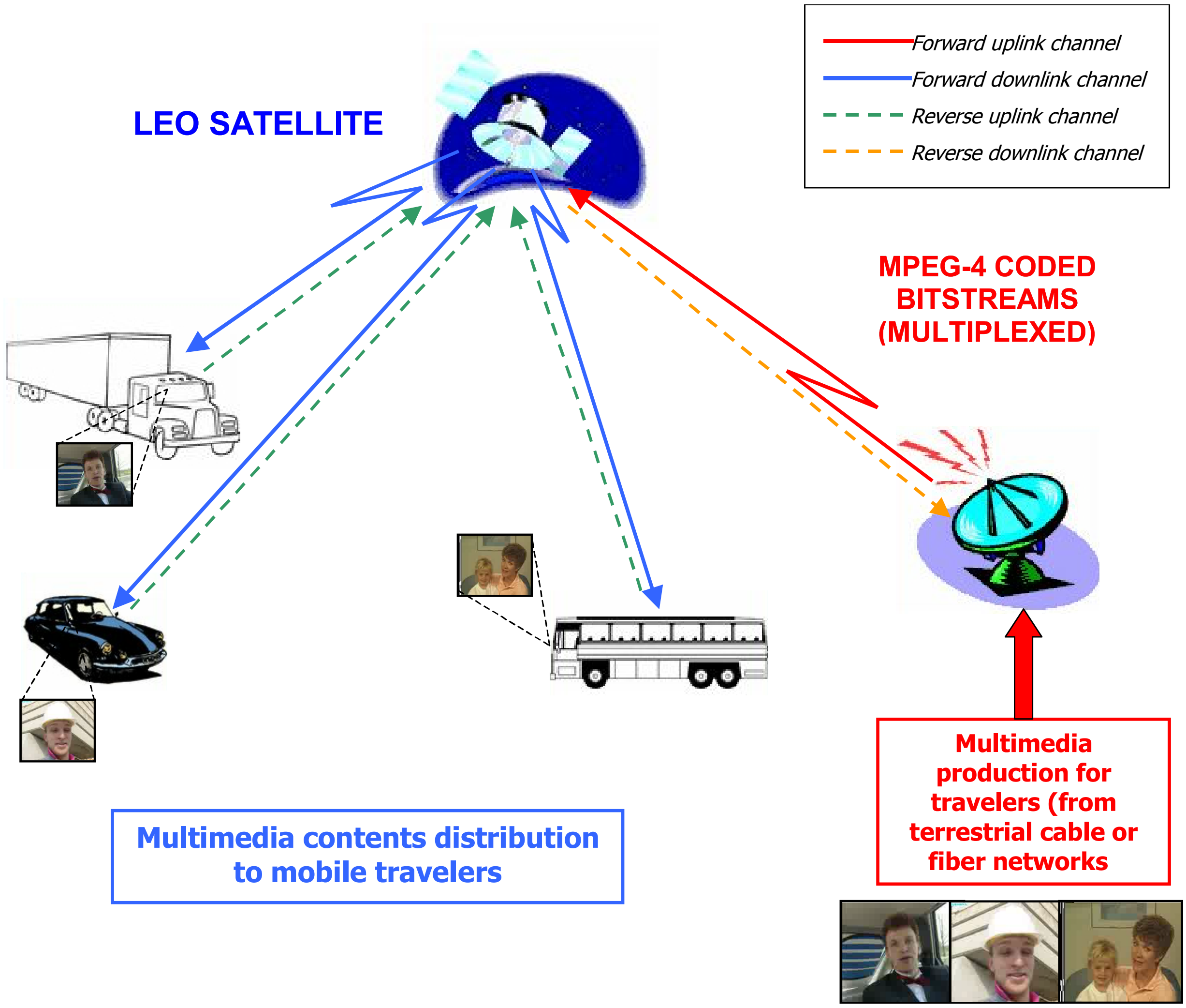


FIGURE 1: *The application scenario for the proposed video transmission methodology*

3. The proposed VBR MC-CDM methodology for transmission of multi-layered MPEG-4 video streams

3.1 MULTI-LAYERED MPEG-4 CODING

ISO-IEC MPEG-4 video standard is regarded as one of the most reliable video coding techniques, because of its flexibility and the possibility of implementing different structures of the codec with the only constraint of a standard-compliant bitstream. More in general one of the main advantages of the new generation coding techniques is the capability of obtaining scalable videos, i.e. the encoded video signal can be partitioned into different bitstreams, which can be decoded independently. Different typologies of scalability techniques have been proposed, but the most effective in the video-streaming application framework is the Fine Granularity Scalability (FGS) [7] [29]. This kind of encoding methodology allows one to generate two different bitsteams: the first one is the *base-layer* (BL), which contains the basic information contents of the video. The other bitstream consists of the *enhancement layer*, which contains the additional data necessary to reconstruct an almost perfect video. The peculiarity of the enhancement layer (namely: FGS layer [7] [29]) is that the DCT coefficients are coded in a bit-plane wise modality, and they are ordered starting from the most significant bit. Thanks to this procedure one can handle video transmission by considering several aspects like e.g.: channel conditions, available computational power at the decoder side and other relevant impact factors.

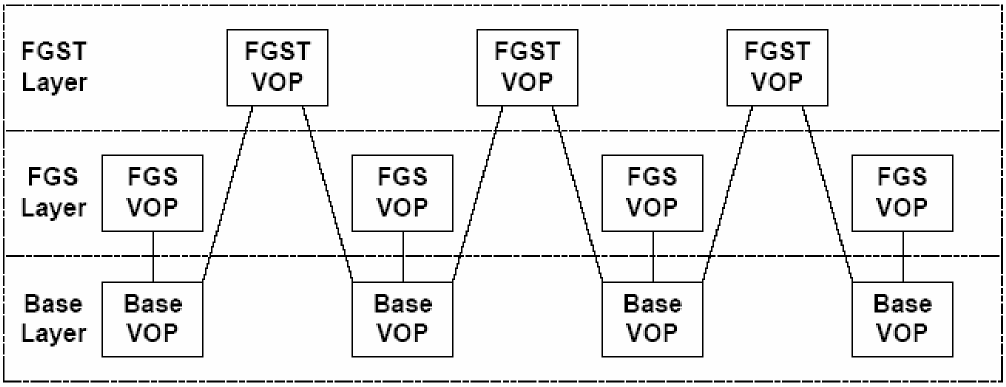
The proposed multi-layered coding methodology is based on the principles mentioned above, but with some modifications. In particular, we exploited the capability of MPEG-4 standard to combine effectively FGS scalability with temporal scalability [29] in order to partition the coded bitstream into multiple layers, so to make the video coding process very versatile with respect to user's need and bandwidth constraints.

In the specific application context described in Section 2, we decided to use a 4-layer encoding process able at creating four different streams, each one characterised by different properties. The starting point of the codec implementation is the partition of a full frame-rate video stream into three layers whose structure is described in details in [30] and [31] and shown in Figure 2a:

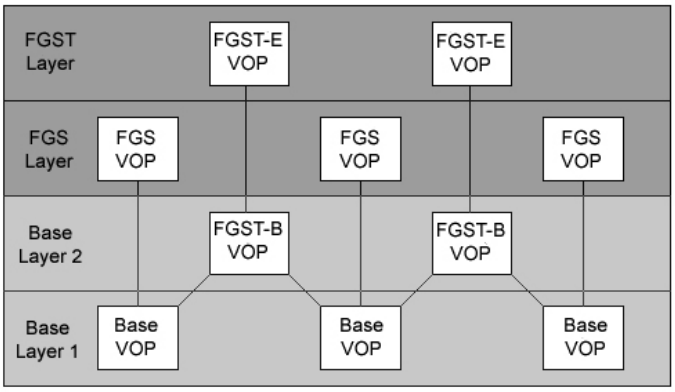
- The *base layer* (BL), characterised by reduced frame-rate and low PSNR. This layer can be received and decoded even when the bandwidth resources are almost scarce;
- the *FGS enhancement layer* containing the residual information for the base layer;
- The *FGST enhancement layer*, containing the coded information related to the frames discarded by the two lower layers listed above.

As perceptual result, in case of successful decoding all three layers, the user would play the full frame-rate high-quality video.

The video-coding scheme described above is quite flexible and efficient, but not very suitable to real-time video streaming applications like those one considered in this work. In fact, the choice of avoiding retransmission in case of channel errors can involve the failure of FGS and FGST layers decoding with a not negligible probability. This is because in a UEP scheme the BL would result the most protected one against channel errors, whereas the degree of protection of FGS and FGST layers would be lower and decreasing. So, only a low-quality slow-motion video could be played by the user's device. This fact would be quite annoying in the context of the fulfilment of a video-on-demand service. According to these assumptions, the proposed layered structure has been improved by adding a fourth layer, aimed at providing a temporal continuous video at low quality. Such an improvement can be easily obtained by truncating the FGST layer conveniently in order to have a second base layer, which introduces temporal continuity (namely: *temporal base layer* – BLT). The enhancement layers have been treated almost in the same way. In particular, two enhancement layers are available: the former one is related to the BL and is exactly the FGS layer of the original encoder, whereas the latter one is related to the BLT and is part of the original FGST layer, as mentioned above. The structure of the improved layered encoder proposed is shown in Figure 2b.



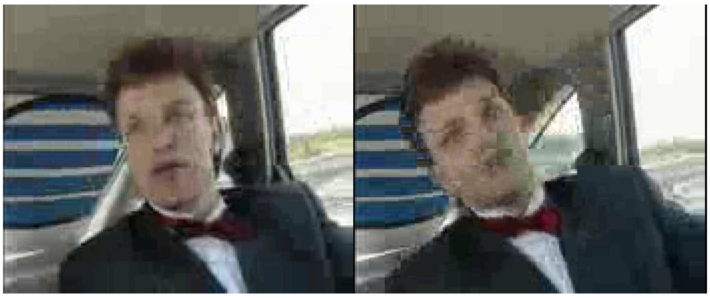
(a)



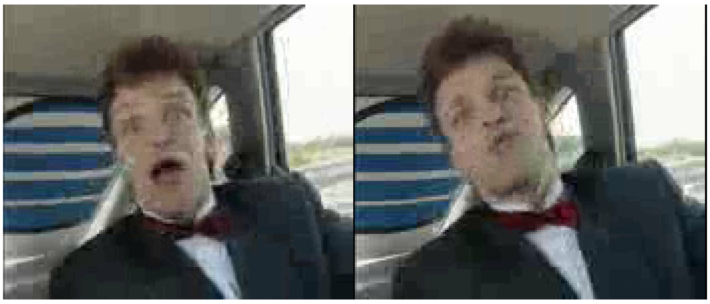
(b)

FIGURE 2: *(a) Standard three-layered FGS MPEG-4 coding architecture, (b) Improved four-layered FGS coding architecture*

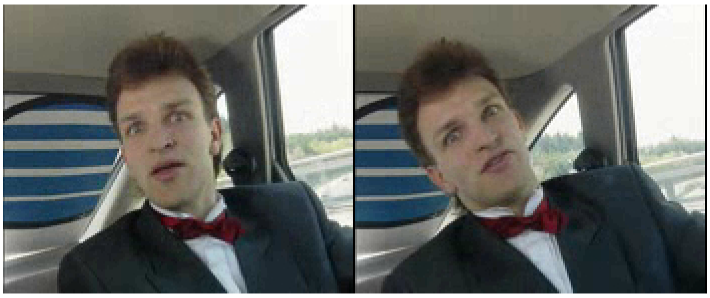
Figure 3 contains four couples of sample frames taken after decoding of a multi-layered video stream in case of error-free transmission. The first couple of frames (Figure 3a) are related to the separate decoding of the BL layer. One can easily note the low quality of the video, but no significant artifacts are present in the frames. The second couple of frames (Figure 3b) are related to the joint decoding of BL and BLT layers. As previously mentioned, the BLT is designed for providing a low quality video (the increment of average PSNR is negligible with respect to the decoded sequence of Figure 3a), but the sequence is played at full-rate. This latter one would be a minimally satisfactory result that should be guaranteed also in worst cases of transmission over harsh noisy channels. The third couple of frames (Figure 3c) are related to the joint decoding of BL, BLT and FGS layers. We have a global increment of the quality of the decoded sequence, but users would perceive some discontinuities among enhanced BL frames (like ones shown in Figure 3c) and low-quality BLT frames (see Figure 3b). In any case, the human eye can tolerate quite well such kind of video distortions. The last couple of frames (Figure 3d) depict the situation corresponding to the joint decoding of all layers: the quality of the decoded sequence is the best possible.



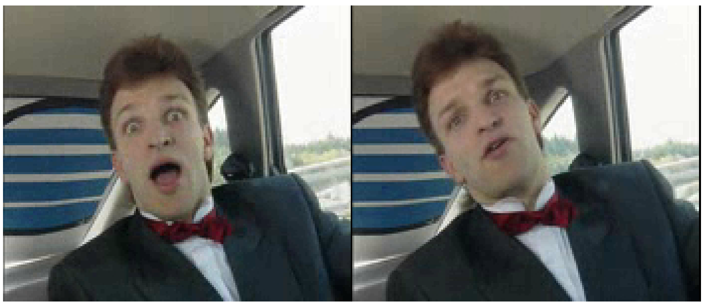
(a)



(b)



(c)



(d)

FIGURE 3. *Sample couples of frames for: base layer (a), BLT layer (b), FGS layer (c), FGST layer (d)*

In Table 1, some numerical data about frame rates, bit-rate, size, time of transmission, and PSNR achievable related to the proposed multi-layered MPEG-4 video coding are shown. Data are related to 5 seconds of the "carphone" sequence, grabbed at 25fps and consisting of 125 frames. The full frame-rate for the encoded sequences is equal to 25fps, whereas BL and FGS frame-rates equal to 1/3 times the full-rate. One can see that the real-time constraints can be satisfied. Indeed, the code-division-multiplexing strategy adopted (and described in the following) allowed us to receive simultaneously all the coded layers. So, the total transmission time is actually the time required for transmitting the biggest layer. In our case the transmission time is 3.88 sec and is the transmission time required by the FGS layer. A proper buffering mechanism and fast processing architectures working at the decoder side should guarantee the respect of the real-time requirements imposed by the considered application.

TABLE 1. *Numerical parameterization of the proposed multi-layered MPEG-4 coding strategy*

	MPEG-4 LAYER	NUMBER OF ORTHOGONAL SUB-CHANNELS (N_l)	TRANSMISSION SNR (E_{bk}/η)
MUX CHANNEL 1	FGST	16	SNR(FGST) (assigned)
MUX CHANNEL 2	FGS	64	SNR(FGST)+6dB
MUX CHANNEL 3	BLT	256	SNR(FGST)+12dB
MUX CHANNEL 4	BL	512	SNR(FGST)+15dB

3.2 VARIABLE-BIT-RATE MULTICARRIER CODE-DIVISION MULTIPLEXING

The multi-layered MPEG-4-based video coding detailed in Section 3.1 should be supported by a robust multiplexing and transmission layer able at providing unequal error protection to the different layers on the basis of the relevance of the information contents carried by them. An efficient and flexible methodology for achieving such kind of resilience at physical layer level is provided by the variabile-bit-rate multicarrier code division multiplexing (VBR MC-CDM). The global scheme of the VBR MC-CDM system for multi-layered MPEG-4-based video transmission is shown in Figure 4.

The shared bandwidth is partitioned into sets of orthogonal sub-channels. Each coding layer is provided with its own sub-channels set, whose cardinality depends on the degree of frequency diversity (and therefore of protection against frequency-selective channel distortions) attributed to it. A set of sub-channels forms a physical MUX channel. This kind of approach has been already shown for MC-CDMA transmission of multi-user variable-bit-rate signals over LEO satellite networks [23]. The choice of MC-CDMA techniques as baseline instead of single-carrier DS/CDMA and other wideband multi-carrier modulation schemes has been motivated by the intrinsic capability exhibited by MC-CDMA of easily employing all the received signal energy scattered in the frequency domain to estimate the transmitted symbol [21]. This is true because, in MC-CDMA schemes, the received signal is, in a sense, combined in the frequency domain [21]. As contrast, it is more difficult for the DS/CDMA rake receiver to make full use of the received signal energy scattered in the time domain. This is due to the heavy effects of multi-user interference that should be removed by appropriate multi-user detection algorithms [21]. Other wideband multicarrier modulation schemes, like MC-DS-CDMA and Multi-Tone-CDMA (MT-CDMA), which are based on the temporal spreading of the signal over each sub-carrier, present an increased complexity with respect to MC-CDMA (MT-CDMA requires a bank or rake receivers [21]), but performance improvements in case of frequency-selective multipath fading distortions are not so greatly appreciable [21]. Moreover, the VBR characterization of UEP provided to the different MPEG-4 coding layers could not be straightforwardly implemented by MC-DS-CDMA and MT-CDMA modulation schemes (a feasibility study about these aspects might be a further research step). In our application, the VBR MC-CDM approach can be used both for single-user and multi-user transmission of multi-layered MPEG-4-coded video signals. In the former case, the different MPEG-4 coding layers of the unique user will be multiplexed and transmitted over different MUX channels. In the latter case, different MPEG-4 coding layers belonging to the same class but related to different users' streams will share the same MUX channel in MC-CDMA modality.

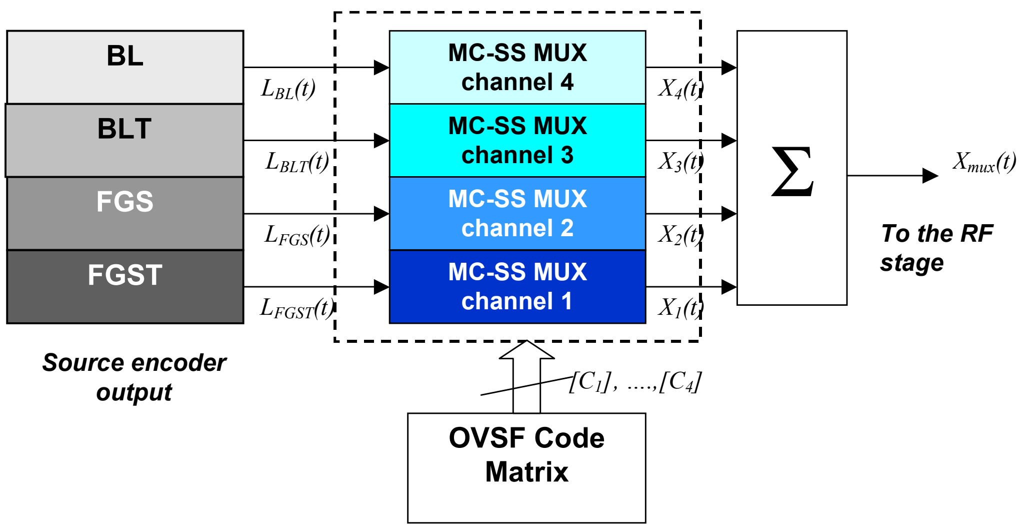


FIGURE 4. *Block diagram of the single-user VRB MC-SS video multiplexer*

Note that the scheme of Figure 4 is customized on the basis of the 4-layer MPEG-4-based video coding described in Section 3.1. Nevertheless, the number of Multicarrier-Spread Spectrum (MC-SS) MUX channels could be both arbitrarily extended as the number of coding layers provided as output by the source encoder increases and reduced whether it is decided to transmit only a subset of the coding layers produced. The detail of how each MUX channel is actually implemented is shown in Figure 5.

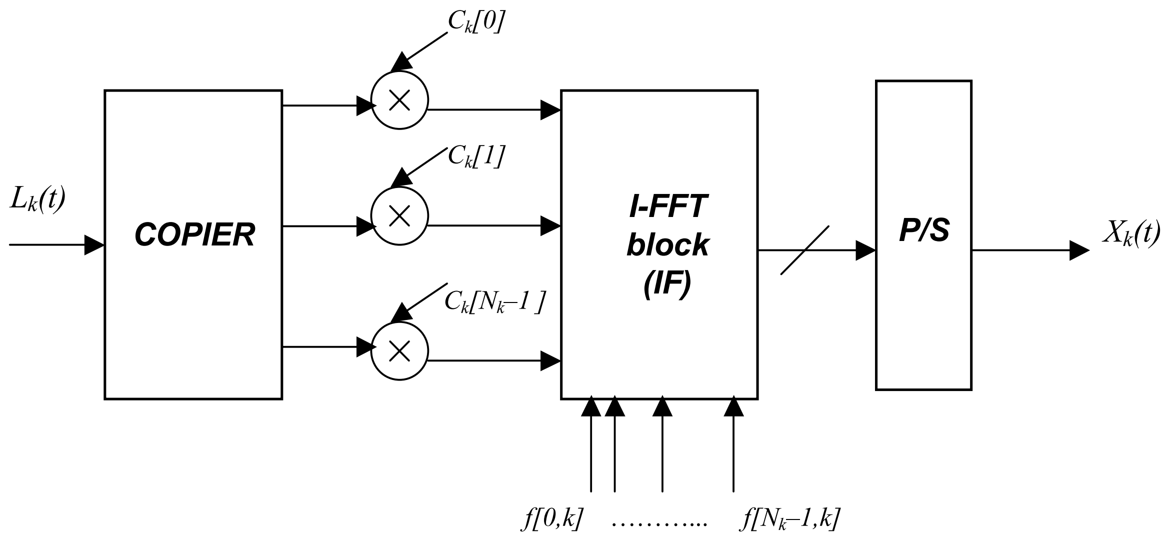


FIGURE 5. *Detail of the generic VBR MC-SS MUX channel*

Each MUX channel (index $i = 1,..,M_{MUX}$, where $M_{MUX} \leq 4$) occupies a given bandwidth equal to B_{MUX}. We can note from Table 1 that the source bit-rates of the MPEG-4 layers binary sub-streams $L_k(t)$ can be expressed integer multiples of the base-layer bit-rate r_{BL} =64Kb/s:

$$r_k = 2^{Q-k} r_{BL} \quad k = 1,..,Q \tag{1}$$

Referring to the rate allocation shown in Table 1, we can say that $Q = 6$, *k*=1 (FGST layer), *k*=3 (FGS layer), *k*=5 (BLT layer), and finally *k*=6 (BL).

The bandwidth B_{MUX} is then partitioned into N_k orthogonal sub-channels upon the MC-SS VBR strategy shown in [23], i.e.:

$$f[n,k] = f_0 + \frac{F}{2}\left(2^{k-1} - 1\right) r_{BL} + nFr_k \quad n = 0,..,N_k - 1 \tag{2}$$

where the values of *k* have been defined above and f_0 is the IF value (IF values commonly chosen for wideband satellite applications are 70MHz and 140MHz). *F* is the sub-carrier spacing factor [21] [22] (usual values for such a parameter are *F*=1 or *F*=2). Note that *k* and *Q* should have changed in case of different layered coding and different rate assignment, but the orthogonal sub-channel allocation rule of equation (2) would have remained unchanged. The pictorial scheme of Figure 6 can easily describe with some approximations and simplifications the sub-channel allocation strategy chosen. The Power Spectral Densities (PSDs) of four multi-carrier MUX channels are drawn in Figure 6, i.e.: MUX channel #1 partitioned into 8 orthogonal sub-channels, MUX channel #2 partitioned into 4 orthogonal sub-channels, MUX channel #3 partitioned into 2 orthogonal sub-channels, and finally MUX channel #4 partitioned into a single sub-channel (single-carrier case). The highest number of sub-channels (8 in the example) is attributed to the lowest bit-rate signals that can exploit the highest degree of frequency diversity against frequency-selective channel impairments. The lowest number of sub-channels (1 in the example) is attributed to the highest bit-rate signals that can exploit the lowest degree of frequency diversity. Such a tradeoff between transmission rate and robustness against channel effects is reasonable in VBR Spread Spectrum systems (it is forecast also by single-carrier W-CDMA standard [27] [28]), as the bandwidth resources are generally constrained.

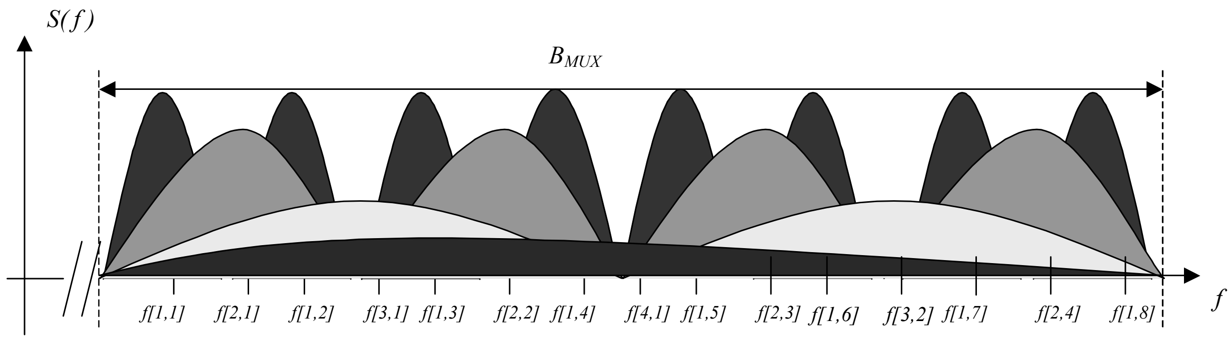


FIGURE 6. *Example of orthogonal sub-channel allocation strategy in the context of a VBR MC-SS transmission*

Of course, the sub-channel assignment strategy chosen for the actual application described in Section 2 is different with respect to the simplified scheme shown in Figure 6. In Table 2, the detail of sub-channel allocation for the considered application is shown. Using a subcarrier spacing factor F=1, the amount of shared bandwidth employed for video transmission is 33MHz. It is clear from Table 2 how UEP is applied to the different coding layer by the proposed VBR MC-CDM scheme.

TABLE 2. *Sub-channel allocation strategy for the multi-layered VBR MC-SS video transmission*

	MPEG-4 LAYER	NUMBER OF ORTHOGONAL SUB-CHANNELS (N_l)	TRANSMISSION SNR (E_{bk}/η)
MUX CHANNEL 1	FGST	16	SNR(FGST) (assigned)
MUX CHANNEL 2	FGS	64	SNR(FGST)+6dB
MUX CHANNEL 3	BLT	256	SNR(FGST)+12dB
MUX CHANNEL 4	BL	512	SNR(FGST)+15dB

It should be also noted that the multicarrier spreading of the transmitted sub-stream is performed by a "full-digital" I-FFT block working at IF. It would be also correct to suppose to make such block working in baseband without losing any generality. Nevertheless, the current trend of full-digital SDR-based devices is to work up to IF, in order to effectively implement by software the complete demodulation chain (including carrier recovery algorithms) [34].

The VBR MC-CDM IF signal generated by each user u of the system has the following expression:

$$x_{MUX,u}(t) \triangleq \sqrt{2P}\,\mathrm{Re}\left\{ \sum_{k=1}^{M_{MX}(u)} \sum_{j=-\infty}^{+\infty} \sum_{n=0}^{N_k-1} c_k[u,n] S_k[u,j] e^{2\pi j f[n,k]t} \Pi_{Tk}(t - jT_k) \right\} \quad (3)$$

where P is the carrier power, assumed to be equal for all layers of all users (we doesn't perform any kind of management of the power resources. This might be matter for future works), $c_k[u,n]$ is the n-th chip of the PN signature code assigned to the k-th multiplexed layer of the u-th user (according to considerations made in [23], Orthogonal Variable Spreading Factor codes [28] has been employed), and $\Pi(t)$ is the digital waveform assumed for simplicity as rectangular NRZ pulse of unit amplitude. Having used a fixed power resource, it must be evidenced that the transmission signal-to-noise ratios $E_b(k)/\eta$ (η is the one-sided power spectral density of AWGN noise) are different from a transmitted layer to another one, due to the different symbol duration $T_k \propto 1/r_k$ (see Table 2). This is another aspect of the UEP provided by the proposed video transmission methodology. In fact, the FGST layer is transmitted at a signal-to-noise ratio 15dB lower than BL one. $S_k[u,j]$ is the j-th symbol transmitted by the u-th user, and related to the k-th multiplexed layer. We assume a BPSK modulation, so that $S_k[u,j] \in \{-1,1\}$.

The received multi-user signal coming from the forward downlink channel (synchronous transmission) downconverted at IF can be expressed as follows:

$$Y(t) = \sqrt{2P}\,\mathrm{Re}\left\{\sum_{u=1}^{U}\sum_{k=1}^{M_{MX}(u)}\sum_{j=-\infty}^{+\infty}\sum_{n=0}^{N_k-1} g_n(t)c_k[u,n]S_k[u,j]e^{2\pi jf[n,k]t}\Pi_{Tk}(t-jT_k)+z(t)\right\} \quad (4)$$

$g_n(t) = \alpha_n(t)e^{j\phi_n(t)}$ is the complex channel coefficient related to the n-th orthogonal sub-channel, and *z(t)* is the additive complex Gaussian noise. Note that the structure of the single-user de-multiplexing receiver is shown in Figure 7. Such a receiver is the basic coherent multi-carrier matched filter receiver, known also as *Equal Gain Combining (EGC)* receiver [21]. Such a receiver structure can be implemented by means of a FFT algorithm and does not require any kind of channel estimation. Actually, the EGC receiver can compensate the time-varying phase shifts involved the frequency-selective channel distortion by means of a coherent detection [21]. The signal provided as output by the EGC receiver of Figure 7 can be expressed as follows:

$$R_{EGC}(t) = \sqrt{2P}\,\mathrm{Re}\left\{\sum_{u=1}^{U}\sum_{k=1}^{M_{MX}(u)}\sum_{j=-\infty}^{+\infty}\sum_{n=0}^{N_k-1} \alpha_n(t)c_k[u,n]S_k[u,j]\Pi_{Tk}(t-jT_k)+z(t)\right\} \quad (5)$$

We supposed here to obtain ideal carrier synchronization, so to achieve the perfect knowledge of $\phi_n(t)$ in order to compensate it. The problem of carrier synchronization in VBR MC-CDMA transmission systems working over multipath fading channels has been dealt in [36]. Afterwards, the received baseband signal is low-pass filtered and sampled before symbol decision (ideal timing recovery has been also supposed in our dealing). The advantages of the EGC scheme are mainly related to the reduced algorithmic complexity. The EGC scheme is theoretically optimal in case of synchronous multi-user transmission over AWGN channel [21]. In such a case, ideal user orthogonalization is retained and multi-user interference (MUI) doesn't arise. On the other hand, the EGC scheme becomes sub-optimal when user orthogonalization is lost in case of frequency-selective channel distortions. In this latter case, multi-user interference will bound receiver performances. In the proposed video transmission scheme, two levels of MUI may degrade the perceived quality of service: the *inter-layer interference*, occurring also in the single-user transmission case as the different coding layers are overlapped in CDM modality over the shared bandwidth, and the *multi-layer interference*, occurring when different layers belonging to different users are overlapped in CDMA modality over the shared bandwidth. The adoption of more sophisticated multi-user detection strategies in order to reduce the impact of MUI on system performances will be matter for future works.

Another problem to be mentioned is the sensibility of MC-CDMA to non-linear distortions involved by high-power amplifiers (HPA) working in saturation mode. Satellite applications generally require such kind of hardware devices in order to compensate the heavy pathloss originated by the long kilometric distance existing between Earth and satellite. An exhaustive analysis of non-linear distortions in MC-CDMA transmission systems has been presented in [37]. The total BER performance degradation involved by non-linear distortions is measured in about 5dB for the downlink and 1.2dB for the uplink. An AWGN channel with non-linear distortion of the transmitted signal has been supposed in [37], and the fixed-bit-rate case is dealt. The performances of the variable-bit-rate MC-CDMA transmission over AWGN channel in presence of non-linear distortions have been evaluated in [38]. Different classes of VBR users have been considered in [38]. The total degradation involved by the presence of non-linearity ranged from 1.5dB for the slowest users up to 7dB for the fastest users. In this paper, the problem of non-linear distortions has been neglected in evaluating performances, focusing our attention on the robustness provided by the proposed video transmission method against frequency-selective linear distortions. In literature, methodologies are proposed in order to compensate the effects of non-linear distortions on multicarrier modulations. As example, pre-distortion techniques are proposed in [37] for equalizing amplifier characteristics in MC-CDMA applications. The impact of performance degradations on the proposed video streaming methodology, together with possible countermeasures could be open issues for future research works.

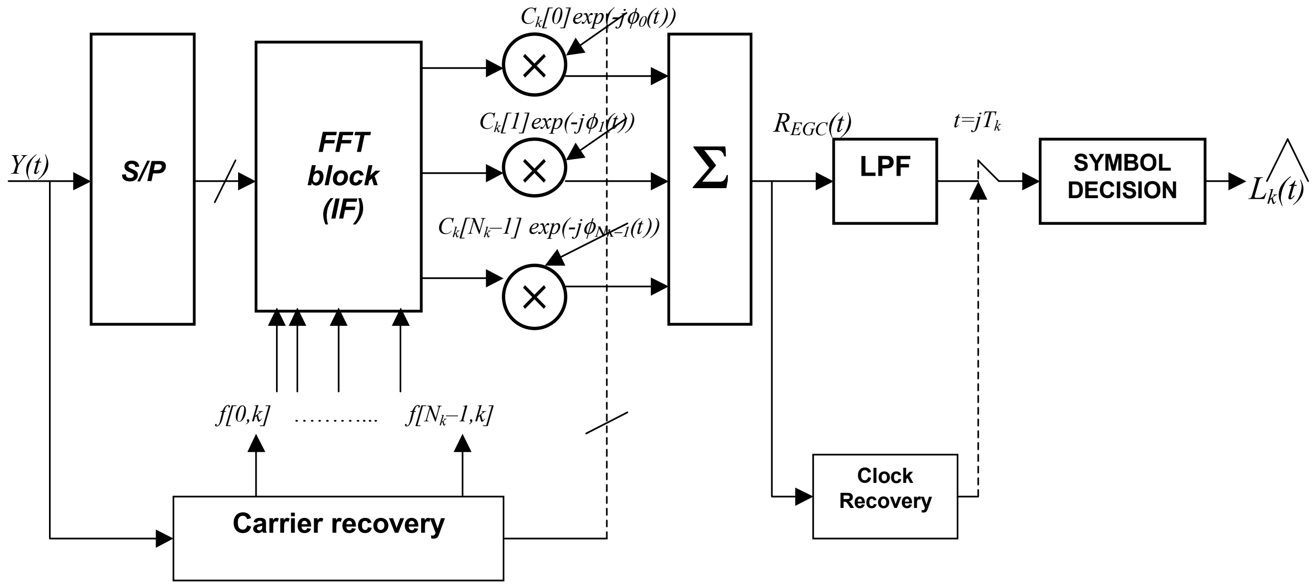


FIGURE 7. *Detail of the generic VBR MC-SS demultiplexer*

4. Experimental results

4.1 SIMULATION PROCEDURES

A realistic simulator of the video transmission system described in Section 3 has been implemented in order to validate the proposed methodology. The simulator has been developed in two steps. The first step was devoted at the implementation of the multi-layered MPEG-4 video coding. The second step was related to the variable-bit-rate multi-carrier code division multiplexing of MPEG-4 coded video streams, together with their transmission over LEO satellite downlink channel.

The core for the implementation of the multi-layered MPEG-4 coding has consisted in the reference software of the MPEG-4 standard [35]. Two versions of such software have been proposed, a first version known as *MoMuSys* developed in ANSI C, and another one, developed by Microsoft® in C++ language. This latter version has been chosen for our work, because it allows an effective monitoring of the effects of channel errors on the decoded streams, together with an easier debug. It should be mentioned that neither MoMuSys nor Microsoft® codec implementations have embedded error resilience tools. Such kind of tools should be implemented by the codec users. In this paper, we did not consider the implementation of any resilience mechanism. The integration of the error resilience in the MPEG-4 multi-layered coding will be matter for future works.

Considering aspects related to multiplexing and transmission, the frequency-selective LEO satellite channel has been simulated by means of a Ricean tapped-delay-line model [5], parameterised by keeping into account the experimental data presented in [32]. In Table 3, the numerical parameterisation of the tapped delay line channel model is shown:

TABLE 3. *Numerical parameterization of the tapped-delay-line channel model*

PATH	DELAY (NSEC.)	AMPLITUDE DISTRIBUTION	RICE FACTOR	AVERAGE ATTENUATION WITH RESPECT TO LOS (DB)
Line-on-Sight (LOS)	0	Rice	30	0
Delayed 1	100	Rice	3	-15
Delayed 2	200	Rayleigh	-	-20
Delayed 3	300	Rayleigh	-	-26
Delayed 4	400	Rayleigh	-	-28
Delayed 5	500	Rayleigh	-	-30

From Table 3, one can derive that the delay spread of the channel equal to 500nsec, corresponding to a coherence bandwidth of 2MHz. The parameterisation in terms of Doppler spread of the LEO satellite channel has been derived by results shown in [33]. In particular a Doppler spread equal to 32 KHz has been chosen for our simulations. Both the satellite channel and the entire modem chain have been simulated in MATLAB SIMULINK V6.5 environment.

4.2 SIMULATION RESULTS

The achievement of selected simulation results has been articulated into two steps: a first step related to a single-trial simulation of a long sequence transmission (5 seconds of video consisting of 125 QCIF frames), and a second step related to multiple re-iterated simulation trials (i.e. 10 trials) of a short sequence transmission (1 second of video consisting of 25 QCIF frames). Such a choice has been motivated by the huge computational time required by the MATLAB-SIMULINK part of the simulator that hindered us to re-iterate the simulation of the long sequence transmission. In any case, the average results achieved by multiple simulation trials (not presented here for sake of brevity) have fully confirmed results achieved by the single-trial simulation step. Simulations of single user and multi-user transmission have been performed by using the following well-known sample sequences:

- *Single user transmission*: the single user transmits the "carphone" sequence;
- *Multi-user transmission*: K=3 users. User 1 transmits the "carphone" sequence, user 2 transmits the "foreman" sequence, user 3 transmits the "mother and daughter" sequence.

It is worth noting that also the above claimed "single-user" transmission is truly multi-user, as the four different layers of the transmitting user are multiplexed in CDM modality over a shared bandwidth. Therefore, the multi-layer synchronous transmission over frequency-selective multipath channel is subjected to inter-layer interference, which is actually a form of multi-user interference.

Results achieved by the proposed VBR MC-CDM system for video transmission, characterized by UEP, are compared with a more conventional single-layer MC-SS transmission using the same bandwidth occupation. This latter transmission is characterized by higher bit-rate (2048Kb/s), lower number of sub-channels (N=16) and absence of UEP mechanisms.

The performances in terms of measured BER for the simulated video transmission are shown in Table 4 for single-user transmission, Table 5 and 6 for multi-user single layer and multi-user multi-layer transmission respectively. The BER values achieved by simulations are quite in line with theoretical values typical of VBR MC-CDMA transmissions over multipath fading channels.

TABLE 4. *BER performances achieved by simulations for the single-user VBR MC-SS video transmission*

	SINGLE-USER TRANSMISSION ("CARPHONE SEQUENCE)				
	BASE LAYER	TEMPORAL BASE LAYER	FGS ENHANCEMENT LAYER	FGST ENHANCEMENT LAYER	SINGLE LAYER MC-SS TRANSMISSION
SNR(FGST)=0dB	Error free	Error free	$5.6*10^{-2}$	$9.42*10^{-2}$	$9.42*10^{-2}$
SNR(FGST)=3dB	Error free	Error free	$2.70*10^{-4}$	$3.29*10^{-2}$	$3.28*10^{-2}$
SNR(FGST)=6dB	Error free	Error free	$1*10^{-6}$	$5.4*10^{-3}$	$5.4*10^{-3}$
SNR(FGST)=8dB	Error free	Error free	Error free	$8.4*10^{-4}$	$8.23*10^{-4}$
SNR(FGST)=9dB	Error free	Error free	Error free	$2.33*10^{-4}$	$2.30*10^{-4}$
SNR(FGST)=10dB	Error free	Error free	Error free	$5.79*10^{-5}$	$5.79*10^{-5}$
SNR(FGST)=11dB	Error free	Error free	Error free	$9.93*10^{-6}$	$1,02*10^{-5}$
SNR(FGST)=12dB	Error free	Error free	Error free	$1.66*10^{-6}$	$1,47*10^{-6}$
SNR(FGST)=15dB	Error free	Error free	Error free	Error free	Error Free
SNR(FGST)=18dB	Error free	Error free	Error free	Error free	Error Free

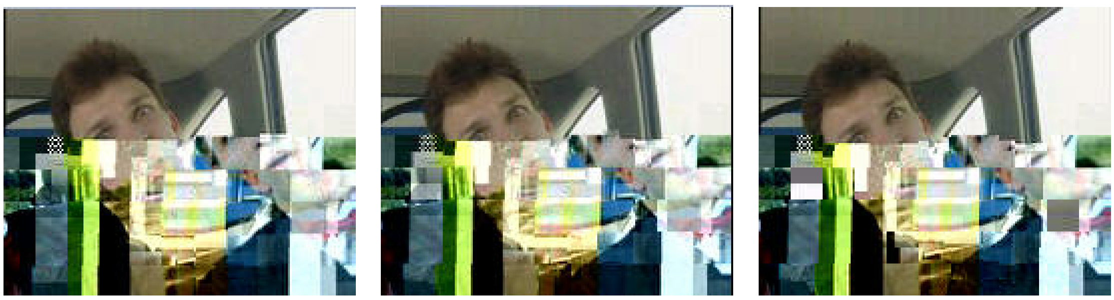
TABLE 5. *BER performances achieved by simulations for the multi-user single-layer MC-SS video transmission*

	SINGLE-LAYER MULTI-USER MC-CDMA TRANSMISSION		
	USER 1 ("CARPHONE")	USER 2 ("FOREMAN")	USER 3 ("MOTHER & DAUGHTER)
SNR=0dB	$8.18*10^{-2}$	$8.21*10^{-2}$	$8.17*10^{-2}$
SNR=3dB	$5.18*10^{-2}$	$5.16*10^{-2}$	$5.19*10^{-2}$
SNR=6dB	$8.53*10^{-3}$	$8.49*10^{-3}$	$8.56*10^{-3}$
SNR=8dB	$2.4*10^{-3}$	$2.3*10^{-3}$	$2.4*10^{-3}$
SNR=9dB	$1.1*10^{-3}$	$1.0*10^{-3}$	$1.1*10^{-3}$
SNR=10dB	$4.4*10^{-4}$	$4.1*10^{-4}$	$4.8*10^{-4}$
SNR=11dB	$1.7*10^{-4}$	$1.6*10^{-4}$	$1.8*10^{-4}$
SNR=12dB	$5.8*10^{-5}$	$5.6*10^{-5}$	$6.2*10^{-5}$
SNR=13dB	$2.2*10^{-5}$	$1.4*10^{-5}$	$2.4*10^{-5}$
SNR=14dB	$4.9*10^{-6}$	$5.1*10^{-6}$	$5.0*10^{-6}$
SNR=15dB	$1.6*10^{-6}$	$6.4*10^{-7}$	$1.1*10^{-6}$
SNR=18dB	Error free	Error free	Error free

In Figure 8, we have shown some sample frames related to the single-user transmission of the long sequence for a *SNR* value of the FGST layer equal to 10dB (this is also the *SNR* of the MC-SS single-layer transmission). Sample frames 8a, 8b, and 8c are related to the single-layer transmission, sample frames 8d, 8e and 8f are related to the VBR MC-CDM transmission of multi-layered MPEG-4 coded stream. Similar results are shown in Figures 9, 10, and 11 for the multi-user MC-CDMA case. In this latter case, the *SNR* of the FGST layer is equal to 12dB. It is evident from such kind of results that multi-layer VBR MC-CDM transmission provides much better results than the single-layer MC-SS transmission.

TABLE 6. *BER performances achieved by simulations for the multi-user multi-layer VBR MC-SS video transmission*

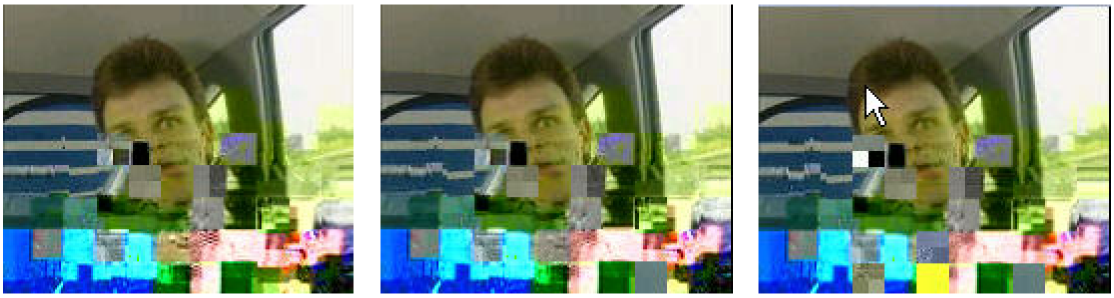
	MULTI-LAYER MULTI-USER MC-CDMA TRANSMISSION USER 1 ("CARPHONE")			
	BASE LAYER	TEMPORAL BASE LAYER	FGS ENHANCEMENT LAYER	FGST ENHANCEMENT LAYER
SNR(FGST)=0dB	Error free	Error free	$5.5*10^{-2}$	$9.89*10^{-2}$
SNR(FGST)=3dB	Error free	Error free	$2.57*10^{-4}$	$3.86*10^{-2}$
SNR(FGST)=6dB	Error free	Error free	$5.03*10^{-7}$	$9*10^{-3}$
SNR(FGST)=8dB	Error free	Error free	Error free	$2.3*10^{-3}$
SNR(FGST)=9dB	Error free	Error free	Error free	10^{-3}
SNR(FGST)=10dB	Error free	Error free	Error free	$4.53*10^{-4}$
SNR(FGST)=11dB	Error free	Error free	Error free	$1.7*10^{-4}$
SNR(FGST)=12dB	Error free	Error free	Error free	$6.45*10^{-5}$
SNR(FGST)=13dB	Error free	Error free	Error free	$2.29*10^{-5}$
SNR(FGST)=14dB	Error free	Error free	Error free	$7.18*10^{-6}$
SNR(FGST)=15dB	Error free	Error free	Error free	$2.70*10^{-6}$
SNR(FGST)=18dB	Error free	Error free	Error free	Error free



(a) (b) (c)

(d) (e) (f)

FIGURE 8. *Triplets of sample frames of the received decoded sequence (single-user transmission, SNR of the FGST layer = 10dB): (a), (b), (c): single-layer MC-SS transmission, (d), (e), (f): VBR MC-SS transmission with multi-layered coding*



(a) (b) (c)

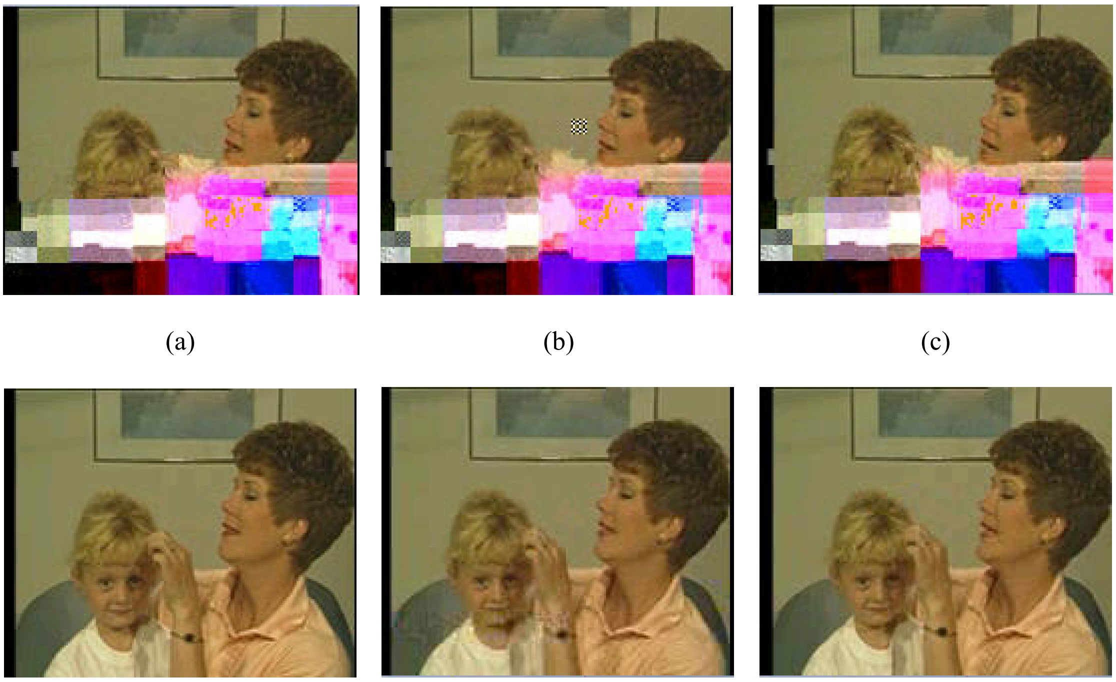
(d) (e) (f)

FIGURE 9. *Triplets of sample frames of the received decoded sequence (multi-user transmission, SNR of the FGST layer = 12dB – User 1: "carphone" sequence): (a), (b), (c): single-layer MC-SS transmission, (d), (e), (f): VBR MC-SS transmission with multi-layered coding*

(a) (b) (c)

(d) (e) (f)

FIGURE 10. *Triplets of sample frames of the received decoded sequence (multi-user transmission, SNR of the FGST layer = 12dB – User 2: "foreman" sequence): (a), (b), (c): single-layer MC-SS transmission, (d), (e), (f): VBR MC-SS transmission with multi-layered coding*



(a) (b) (c)

(d) (e) (f)

FIGURE 11. *Triplets of sample frames of the received decoded sequence (multi-user transmission, SNR of the FGST layer = 12dB – User 2: "mother and daughter" sequence): (a), (b), (c): single-layer MC-SS transmission, (d), (e), (f): VBR MC-SS transmission with multi-layered coding*

The visual results shown above can be confirmed by *PSNR* results shown for the "carphone" sequence in Figure 12 (single-user transmission) and Figure 13 (multi-user transmission). All *PSNR* curves are plotted versus the FSGT *SNR* value (namely: *SNR(FGST)*). The two curves related to the single layer and multi-layer transmission trend to become coincident for *SNR(FGST)*>14dB (when both cases are "error-free"), but one can note an impressive quality gain provided by the multi-layer transmission for 10dB<*SNR(FGST)*<13dB. This fact is consequential to the UEP provided by the VBR MC-CDM mechanism to the different coding layers. In Figure 14, the PSNR achieved by the decoding of the four layers of the "carphone" sequence vs. *SNR(FGST)* in the multi-user case is shown. It is noticeable that BL and BLT layers can be always successfully decoded. This tells us that at least a full frame-rate low-quality video can be played by mobile users. Moreover, we can note that for *SNR(FGST)*>8dB, the FGS layer can be decoded without errors, thus providing a fair quality video with some discontinuities among BL enhanced frames and BLT frames not yet enhanced. Finally, we can point out that for *SNR(FGST)*>10dB, the FGST layer begins to yield a positive contribution in the decoding of the video sequence, progressively enhancing the perceived visual quality.

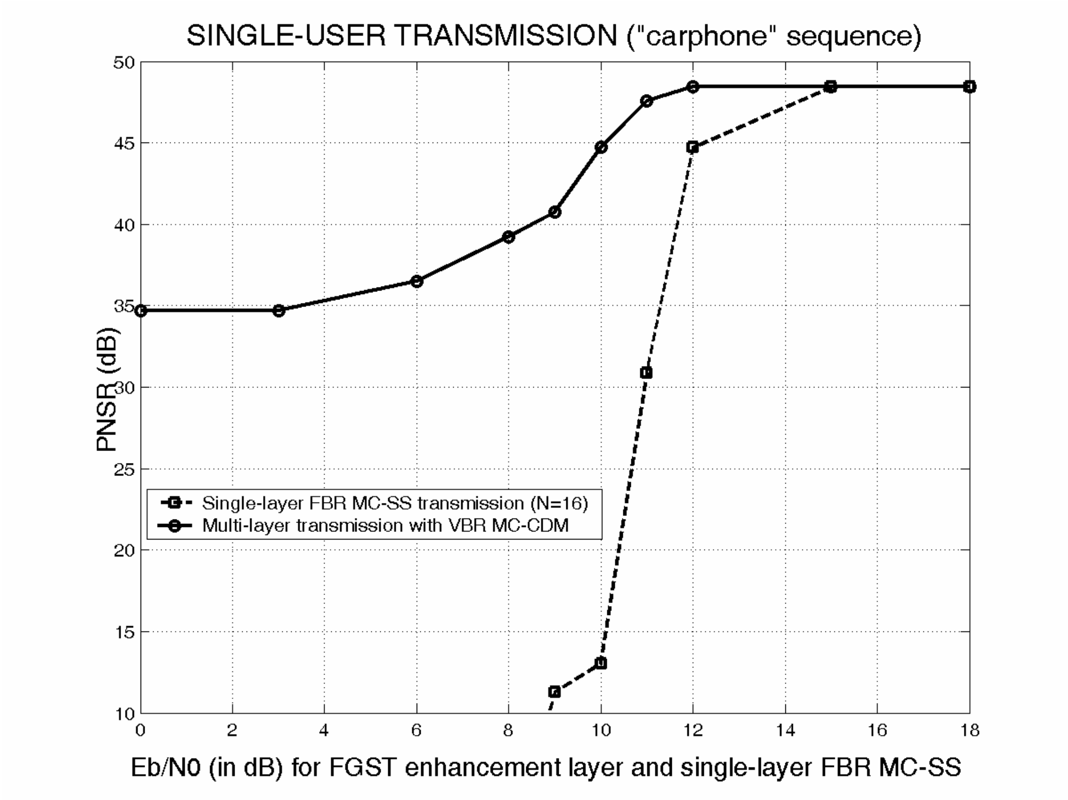


FIGURE 12. *PSNR values vs. SNR of FGST layer for the single-user transmission case*

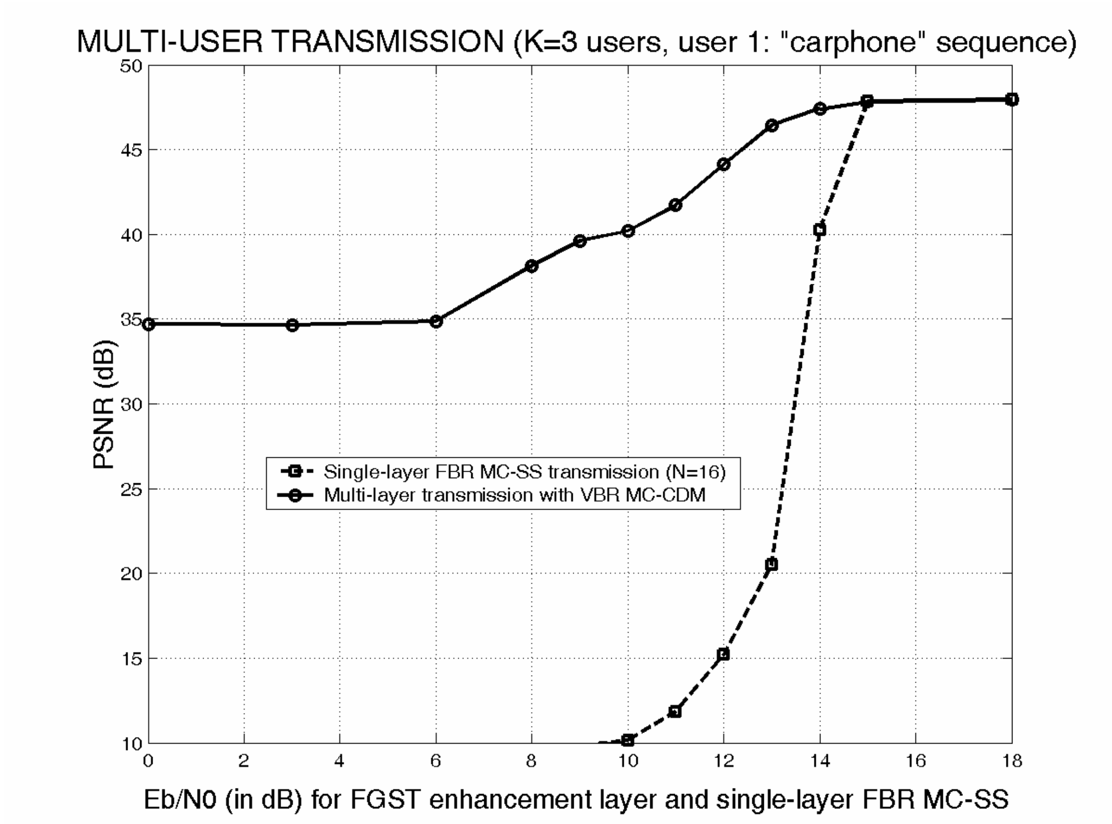


FIGURE 13. *PSNR values vs. SNR of FGST layer for the multi-user transmission case*

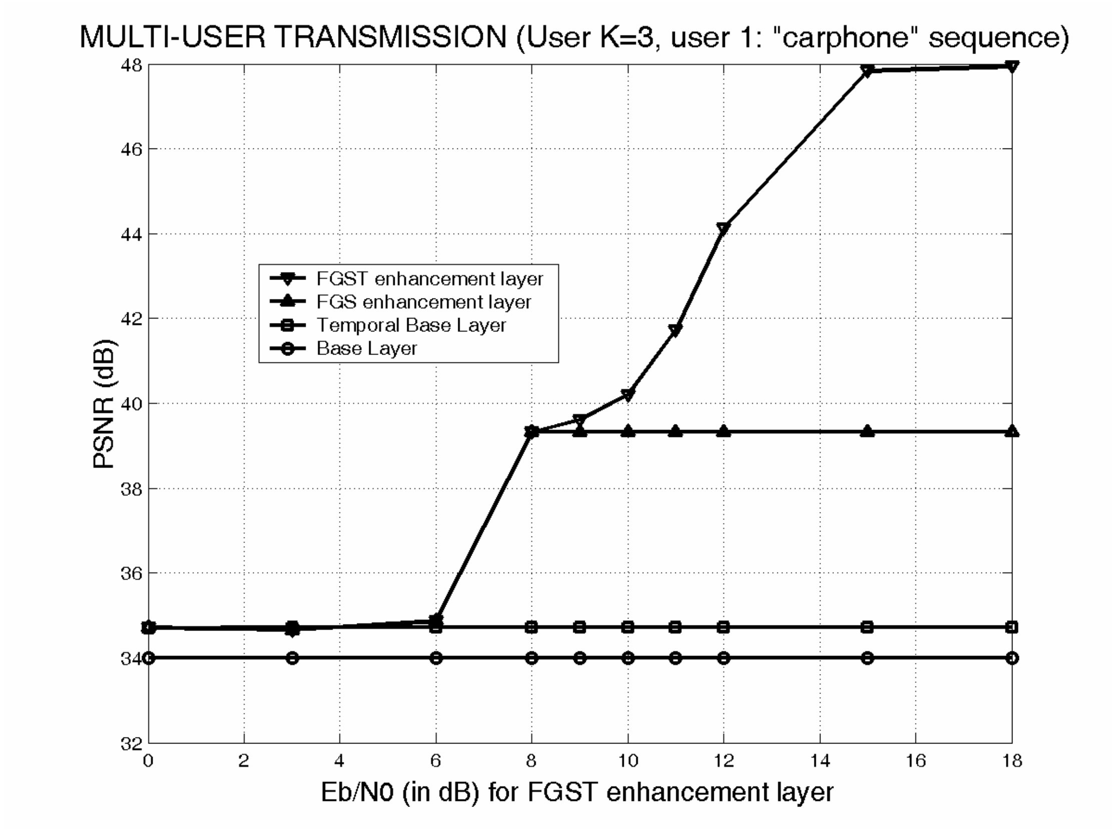


FIGURE 14. *Incremental PSNR values vs. SNR of FGST layer for the multi-user transmission case*

5. Conclusion

In this paper, a novel methodology for reliable video streaming over wireless networks has been discussed. The technical basis of the proposed approach relies on advanced concepts like multicarrier modulations for variable-bit-rate transmissions and multi-layered scalable MPEG-4 coding that will characterize future generations of multimedia wireless networks. The application testbed considered for assessing the proposed video transmission method is related to a satellite multicast video streaming service targeted to vehicular users. Experimental results shown can prove the robustness of the VBR MC-CDM approach. This is due to the unequal error protection inherent to the multicarrier multiplexing technique adopted, able at providing an incremental degree of frequency diversity against multipath channel distortions to the coding layers characterized by the most relevant information content. Future works might deal, among the others, with aspects like employment of multi-user detection algorithms, adaptive management of the power resources attributed to each MUX channel, effects of non-linear distortions (and possible countermeasures) on perceived video quality, integration of error resilience tools in the standard MPEG-4 encoding, etc.

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