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Cross-Layer Adaptation of MPEG4 Video Streaming over Wireless Networks using Unequal Error Protection and MC-CDMA

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Abstract

A novel methodology for the efficient multiplexing and streaming of MPEG4 video over wireless networks is presented and discussed. The proposed cross-layer adaptation jointly exploits variable-bitrate (VBR) multi-carrier code-division multiplexing (MC-CDM) and MPEG4 Fine-Grain-Scalability (FGS) in order to provide unequal error protection to the transmitted video stream. A shared bandwidth is partitioned into orthogonal sub-channels in order to multiplex different layers of MPEG4-coded signals. Lower layers are assigned a higher number of sub-channels (and hence an increased frequency diversity) as compared to FGS enhancement layers, in order to provide a differentiated protection against channel degradations. A 2-GHz LEO multicast satellite transmission system has been considered as an application testbed of the proposed methodology. Results achieved in terms of PSNR show that the VBR MC-CDM technique can provide better results than conventional MPEG4 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 a convenient solution for reliable multimedia transmissions in mobile environments.

Keywords: Video coding, Unequal Error Protection, Multimedia Communication, Multicarrier modulation, Multiaccess Communications, Wireless Networks

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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 present, 3G mobile standards (i.e. UMTS and UMTS-like systems) are supporting services like i-mode, videophone, high-speed data access (up to 384Kb/s), and video-on-demand over PDAs, etc. [1]. Next generation mobile networks (4G and beyond) will focus on 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 [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 efficiently exploit 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/video signals are generally quite vulnerable 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 media streaming 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. Several companies and research institutes recently 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].

In this work, the focus is primarily put on real-time multicast video streaming for mobile users. The MPEG4 standard made available a number of useful features for the development of flexible and robust video streamers over unreliable channels [7]. In particular, temporal scalability and *Fine-Grain-Scalability* (FGS) [34], which are core features of MPEG-4 standard, allow one implementing multi-layered video coders able to address in an 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, focusing on different aspects of video transmission.

As in [6], it is possible to define three main approaches to resilience: i) joint source/channel coding, ii) post-processing at the decoder upon detection of errors, iii) interaction between 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]. Such approaches involve the introduction of redundancy in the bitstream as well as temporal delays, which may significantly decrease the efficiency and throughput. For this reason, dynamic and adaptive mechanisms are being studied to protect the information in a selective way. According to this principle, the parts of the transmitted bitstream that carry more significant information should be protected more effectively, thus leading to the concept of Unequal Error Protection (UEP). Several solutions have been proposed for both wired and wireless networks. In particular, UEP is applied in [11] for video broadcasting over wired channels such as DSL lines. In this case, different layers of scalable image/video contents are multiplexed over a variable number of DSL sub-channels. According to the basic DSL concept, a higher bitrate is associated to video layers transmitted over sub-channels characterised by a higher channel-gain-to-noise-ratio (CGNR) by using spectrally efficient multi-level modulations. In [12], Jil et al. proposed the application of an improved MIMO-OFDM modulation scheme for reliable transmissions of FGS MPEG4 video streams over wireless channels. Such a scheme is characterized by an adaptive power allocation performed at sub-channel level in order to minimize the total distortion while satisfying the transmission power constrains. Another UEP technique exploiting the potentialities of MIMO-OFDM systems with application to MPEG4 streaming has been presented by G. Ren *et. al.* in

[13]. Such a scheme uses a space-time block coding with variable transmission gain for a MIMO-OFDM system. The information bits related to the encoded video stream are grouped into three different data partitions, each one characterized by a different sensitivity to channel errors. The different data partitions are then space-time encoded with a different diversity gains depending on their importance for the decoding phase.

As far as UEP techniques based on channel coding are considered, several approaches have been recently proposed [14-18]. In [14], a convolutional code is employed to achieve UEP by separately encoding the I- and P-frames of an H.263-coded video sequence. Interleaving/de-interleaving is assumed to be ideal, so that decoding errors should be regarded as independent by the source decoder. The solution proposed by Kim and Mersereau in [15] consists of a FEC-based bitplane-wise UEP algorithm aimed at providing unequal amount of FEC protection to each bit-plane of a JPEG2000-coded still picture or to a 2D/3D SPIHT-coded video sequence. The combination of advanced error resilience tools such as Reversible Variable Length Coding, Rate Compatible Punctured Convolutional Codes (RCPC), and Reed Solomon encoding was also deeply investigated [16-18]. In [19] Gharavi proposed the use of a UEP approach based on joint source/channel coding performed at physical layer. Such methodology provides error resilience in a transmission system based on a 16-QAM modulation with pilot symbols for channel estimation. UEP is obtained by subdividing the code in two partitions according to the partition mechanism is implemented in the H.263 standard, and assigning VLC data to the partitions on the basis of the relevant content. A dual-priority transmission mechanism is then implemented by exploiting the different resilience of the bits making up a symbol in a Gray-coded 16-QAM. Wavelet diversity is employed in [20] to transmit compressed still pictures over wireless channels. Multiple copies of the wavelet coefficients are transmitted over separated channels, in order to exploit diversity against frequency-selective multipath fading. UEP is jointly achieved by using different Reed-Solomon encoders for different coefficients, as to increase redundancy in lowest-resolution subbands (which are characterized by the most critical information content). Finally, we can mention the methodology presented in [21], targeted at implementing an efficient and robust TDM-based multiplexing of MPEG4 streams provided with UEP. In this approach, two multiplexing layers may be selected according to the expected QoS. The former combines in a single stream several

MPEG4 sources belonging to the same class (audio, video, or control data) and requiring the same QoS. The latter is employed for the MPEG4 streams characterized by specific QoS constraints. Error protection is performed differently for the two layers: in the first case it is directly applied to the multiplexed stream (thus acting in a homogeneous way), while in the second case it is separately applied to every single stream according to the requested QoS.

Recently, some interesting works focused the attention on cross-layer approaches, targeted at providing robust video streaming over unreliable access networks. In wireless environments, channel conditions change rapidly over time due to noise, interference and multi-path propagation. For this reason, transmission schemes have to dynamically adapt to both application requirements and channel conditions [22]. As far as multimedia is concerned, cross-layer techniques are proposed in [22] and [23] for video streaming applications over wireless networks in order to dynamically adapt FEC coding, maximum MAC retransmission limit, and packet sizes. In [24] a formal energy-based approach is described, targeted to the optimal use of transmission power in the presence of distortion and power constraints. It is demonstrated that the adaptive scheme outperforms fixed power schemes as well as previous approaches based on heuristics. Another cross-layer approach is proposed in [25], where a layered 3-D SPIHT-coded video stream is mapped onto several DS/CDMA channels, each one characterized by different FEC protection and adaptive power allocation, whereas the processing gain of each channel is fixed. Here, rate and power allocation for each DS/CDMA channel is optimised with respect to the minimization of the average distortion by using a formal approach. A Minimum-Mean-Squared-Error (MMSE) multi-user detection algorithm is employed at the receiver side in order to reduce the impact of multi-channel interference on the system performance.

In this context, the paper proposes a novel methodology for the efficient multiplexing and transmission of MPEG4 video signals over wireless networks. The proposed approach is based on a cross-layer adaptation, which makes physical and transport layers to interact to achieve higher error resilience through unequal error protection (UEP) mechanisms. The flexibility of a variable-bit-rate multi-carrier code-division multiplexing (MC-CDMA) at the physical layer [26-29] allowed to design a physical layer whose robustness against frequency-selective channel distortions is adaptively matched to the significance of different video signal contents. In order to split the stream in layers of different importance, MPEG4 Fine-Grain-Scalability (FGS) was considered at the video transport layer. The shared bandwidth is partitioned into orthogonal sub-channels in order to multiplex different layers of MPEG4-coded signals. A higher number of sub-channels (and hence an increased frequency diversity) is assigned to base layers while fewer sub-channels are proportionally assigned to FGS enhancement layers, in order to provide a differentiated protection against channel degradations. The proposed scheme was tested through extensive simulations related to a satellite multicasting scenario. Results achieved in terms of PSNR point out that the proposed technique outperforms conventional MPEG4 single-layer MC-SS transmission techniques.

The paper is structured as follows: Section 2 introduces the proposed approach and provides some insight, in three sub-sections, of the application scenario, the video codec structure, and the VBR MC-CDMA technique. Aspects related to UEP through joint multi-carrier modulation and multi-layered MPEG4 coding are properly evidenced. Section 3 presents selected experimental results. Finally, Section 4 draws paper conclusions.

2. Multi-layer MPEG4 video over VBR MC-CDM

The aim of the present work is to develop a robust video streaming approach for mobile multimedia applications, by exploiting a cross-layer optimization involving the physical and transport layers of a wireless network.

The core of the proposed methodology is the combination of a multi-layer MPEG4 video coder with a variable-bitrate Multi-Carrier Code-Division-Multiple-Access (VBR MC-CDMA) transmission technique, aimed at achieving a UEP based on frequency-diversity for each coded layer. The implemented MPEG4 scalable video coder can be regarded as a compliant extension of MPEG4 FGS standard, able to provide a hierarchical structure consisting of a base layer (carrying the most critical information content at low bitrate), and several FGS enhancement layers, characterized by increasing bitrate and decreasing importance of the information content. VBR-MC-CDMA techniques (basically derived from the OFDM-CDMA concept [26-27], i.e., 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 [28-29]. VBR transmission is performed over MC-CDMA channels consisting of sets of orthogonal sub-channels and different spreading codes (belonging to the family of the Orthogonal Variable Spreading Factor (OVSF) codes [33]).

The proposed transmission methodology enables both the multiplexing of different MPEG4 coding layers of a single transmitting user (Variable-Bit-Rate Multicarrier-Code-Division-Multiplexing) and the multi-user transmission of different MPEG4 video streams. The latter can be achieved by overlapping in MC-CDMA modality the coding layers of the multiple users over the shared bandwidth. The frequency diversity provided by the VBR MC-SS transmission, and hence the 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 importance) are transmitted over the MC-CDMA channels provided with the highest number of orthogonal sub-channels. On the other hand, the highest-bit-rate coding layers (useful for refining the perceptual quality of the decoded video sequences) are transmitted over the MC-CDMA channels provided with the lowest number of orthogonal sub-channels. In such a sense, we can speak of cross-layer adaptation of MPEG4 video streaming, because the physical layer and the radio resource management of the network are adaptively designed with respect to the importance of the information carried on by the different FGS coding layers.

The proposed approach can be regarded as a substantial innovation over state-of-the-art methodologies for error resilience using unequal error protection summarized in Section 1. In fact, error resilience is implemented here by means of different degrees of frequency diversity, selectively applied to the different MPEG4 coding layers. As compared to other UEP techniques exploiting multi-carrier modulations [11-13], the proposed approach differs substantially from state-of-art techniques in the fact that it is strongly based on spread-spectrum and frequency-diversity, whereas other work rely on narrowband OFDM and DSL modulations, space/time diversity, and optimised power resource allocation managed at OFDM sub-carrier level. Moreover, Spread-Spectrum-based MC-CDMA employed for multi-layered video transmission is based on a completely different concept with respect to the one proposed in [25], because signal spreading is performed in the time domain and does not adapt to the layer importance. There, adaptation is achieved by FEC coding and power allocation.

As far as the technical feasibility of the proposed method is concerned, it is to be pointed out that multi-carrier spread-spectrum modulations and flexible radio resource management are going to characterize “4G and beyond” wireless standards [2], profitably exploiting concepts like network reconfigurability and software-defined-radios [39]. No particular technical drawback can hinder a prototypical implementation of the video streaming system proposed in this paper. Recent works evidenced how MPEG4 coding can be ported over DSP infrastructures [30], as well as full-digital implementations of MC-SS transceivers by means of Fast-Fourier-Transform (FFT) software tools [26-27]. The following sub-sections describe a possible application scenario for the envisaged system, and provide and in-depth description of the developed technologies.

2.1 THE APPLICATION SCENARIO

The application scenario for the proposed video streaming technique is depicted by Figure 1. The underlying idea is to provide a multicast video content distribution to mobile users in a vehicular environment. The central node of the mobile network is a LEO satellite. The bi-directional communication between mobile terminals and earth station is allowed in asymmetric mode. The video contents are provided “on-demand” to the mobile users by a downlink channel. Such a multicast transmission is broadband and synchronous. On the other hand, the reverse uplink channel aims at collecting users’ requests of video contents. The uplink transmission channel can be regarded as narrowband and asynchronous. The collected users’ requests are then forwarded to the earth station through the downlink. 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 lost information.

<INSERT FIGURE 1>

The employment of LEO satellites instead of GEO ones is motivated by the necessity of reducing the latency time in delay-sensitive multimedia applications such as 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 [31] and for S-UMTS standard [32]. Such a choice can be motivated by a future integration perspective concerning mobile cellular networks and satellite networks as pointed out in [2] and [32].

As far as the current development is concerned, the attention was mainly focused on the downlink channel, which is the most critical issue due to the necessity of satisfying precise QoS requirements in the presence of frequency-selective signal distortions involved by multipath fading. In the architecture of Figure 1, LEO satellites might be or not regenerative without losing generality and without modifying substantially the core of the proposed approach for scalable video transmission. The adoption of regenerative satellites could be suggestible in order to improve the flexibility of the system and reduce latencies.

2.2 MULTI-LAYER MPEG4 CODING

MPEG4 standard for audio-visual coding is currently regarded as one of the most efficient techniques, because of its flexibility and of the possibility of implementing different compliant codec architectures depending on the application [7]. One of the main advantages of the standard is the capability of obtaining scalable streams, i.e., the encoded video signal can be partitioned into different bitstreams, whose main property is to be hierarchically decodable in a progressive way. Different types of scalability have been proposed. It is possible to scale the bitrate on the basis of the frame rate (temporal scalability), of the coefficient quantization (quality scalability), of the frame size (spatial scalability), or even of the scene contents (region-of-interest scalability). From the point of view of the code structure, several possibilities have considered, including layered coding, multiple descriptions and fine granularity. Fine Granularity Scalability (FGS) is very useful in many streaming applications [7] [34]. In analogy with layered coding, FGS allows splitting the stream in two different bitstreams: a base-layer (BL), which allows a video reconstruction at base quality, and an enhancement layer, which contains the additional data necessary to progressively increase the video quality. The peculiarity of the FGS enhancement layer is that the DCT coefficients are coded in a bitplane wise modality, and ordered starting from the most significant bit. Thanks to this structure, the video reception can be stopped anywhere without compromising the decoding process, and exploiting all the received information at the decoder. This property allows one to easily handle a transmission process by taking into account several different parameters such as channel/traffic conditions, computational capabilities of the decoder, power consumption, and so on.

The proposed multi-layered coding methodology is based on the above-mentioned criteria, with some modifications. In particular, the proposed encoder exploits the capability of MPEG4 of effectively combining FGS and temporal scalability [34] in order to partition the coded bitstream into multiple layers, in order to make the video coding process more versatile with respect to user's needs and bandwidth constraints. In the specific application context described in Section 2, we decided to use a 4-layer encoding process that generates four streams characterised by different properties. To achieve this goal the codec first subdivides a full frame-rate video into three layers whose structure is shown in Figure 2a [35-36]:

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

In case of successful decoding of the all three layers, a user would perceive a full frame-rate high-quality video. The video-coding scheme described above can be applied with success in many situations, such as for instance a multicast transmission, in which different users may receive a scaled quality of the video depending on connection/terminal characteristics. A possible drawback lies in the fact that, in the presence of hard channel conditions, FGS and FGST layers have a not negligible probability of being almost entirely lost due to the fact that UEP schemes would mainly protect the base layer. ARQ techniques may partially solve the problem but cannot be used in applications where real-time, multicast or high interactivity are requested. According to these considerations, the stream architecture has been modified in this work by further specifying the layer structure. To this purpose, a new layer is introduced by suitably truncating the FGS and FGST layers and by mixing them in order to achieve a more adaptable combination (see Figure 2b).

<INSERT FIGURE 2>

The first layer of Figure 2b provides a low quality, low frame-rate video, as in the MPEG standard [7]. The enhancement layers aim at improving the quality of the video in the most flexible and gradual way. In particular, the FGS layer is truncated in two different streams, whereas the FGST layer is divided in three layers. The obtained sub-layers are hierarchically combined so as to achieve a progressive quality. In detail, the second layer (namely: Base T-layer or BTL) provides a low-quality temporal-enhanced video, obtained by truncating at a very low bit-rate the FGST layer. Actually, the successful decoding of BL and BTL layers would provide a low quality full-rate video, that is the minimal result that should be guaranteed also in the presence of harsh noisy channels. The upper layer provides a selective quality enhancement. The successful decoding of FGS1 and FGS2 enhancement layers would provide a full-rate video with increased quality with respect to BTL. The layer stratification shown in Figure 2b can be read vertically and horizontally. Vertically, the different grey tones of boxes show the encoded layers obtained from the same original layer, whereas horizontally we can find the layers that are delivered at the same rate. Although the proposed structure requires the definition of a non-compliant layering, this problem can be easily solved by using a glue-procedure at the decoder site that re-combines the received information in order to make the stream decidable by a standard MPEG4 decoder.

Figure 3 draws the PSNR vs. frame number achieved by the incremental error-free decoding of the four layers previously described. Note that the PSNR increase achieved by BTL decoding is negligible, BTL not being aimed at improving the visual quality of single frames. A significant visual quality improvement is instead achieved by successful decoding of FGS1 and FGS2 enhancement layers.

<INSERT FIGURE 3>

Figure 4 shows three sample pairs of frames extracted from the decoded multi-layered video stream in case of error-free transmission, and provides a visual confirmation of the above considerations on the progressive video quality improvement. The first pair (Figure 4a) concerns the separate decoding of BL and BTL layer. One can easily note the low quality of the video, but no significant artifacts are present in the frames. The second pair (Figure 4b) is related to the joint decoding of BL, BTL and FGS1 layers. A global increase of the visual quality is clearly perceivable. The last pair (Figure 4c) depicts the situation after the joint decoding of all layers: the quality of the decoded sequence is the best possible.

<INSERT FIGURE 4>

In Table 1, some numerical results on frame rate, compression and PSNR are provided. Data are related to 5 seconds of the “carphone” sequence, grabbed at 25fps for a total of 125 frames. The full frame-rate for the encoded sequences is therefore equal to 25fps, whereas BL frame-rate is 1/3.

<INSERT TABLE 1>

2.3 VARIABLE-BITRATE MULTICARRIER CODE-DIVISION MULTIPLEXING

In this section it is described how the multi-layer MPEG4-based video coder described in Section 2.2 can be supported by a robust multiplexing and transmission layer able to provide unequal error protection over the different layers on the basis of the information contents. An efficient and flexible methodology for achieving such kind of resilience at the 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 MPEG4 video transmission is shown in Figure 5.

The shared bandwidth is partitioned into sets of orthogonal sub-channels. Each coding layer is associated to a set of sub-channels, 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. A similar approach has been already demonstrated in the field of MC-CDMA transmission of multi-user variable-bit-rate signals over LEO satellite networks and indoor WLAN networks [28-29]. 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 of MC-CDMA of easily employing all the received signal energy scattered in the frequency domain to estimate the transmitted symbol. This is mainly due to the fact that in MC-CDMA schemes the received signal is, in a sense, combined in the frequency domain [26]. In contrast, it is much more difficult for a DS/CDMA rake receiver to make full use of the received signal energy scattered in the time domain, due to the heavy effects of multi-user interference that should be removed by appropriate multi-user detection algorithms [26-27]. A clear demonstration of this can be found in

[25], where MMSE multi-user detection is explicitly pointed out by authors as a key element required to achieve BER performances compatible with a correct decoding of the multi-layered video.

Other wideband multi-carrier 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), while providing negligible performance improvements in case of frequency-selective multipath fading [26]. As far as the considered video streaming system is concerned, the VBR characterization of UEP provided to different MPEG4 coding layers could not be straightforwardly implemented by MC-DS-CDMA and MT-CDMA modulation schemes (a feasibility study about these aspects might be subject of further investigations). On the contrary, VBR MC-CDM can be used for both single- and multi-user transmission of multi-layered MPEG4-coded video signals. In the first case, the different MPEG4 coding layers of the single user are multiplexed and transmitted over different MUX channels. In the second case, different MPEG4 coding layers belonging to the same class but related to different users' streams share the same MUX channel in MC-CDMA modality.

<INSERT FIGURE 5>

It is to be pointed out that the scheme in Figure 5 is customized on the basis of the 4-layer MPEG4 video coder described in Section 2.2. Nevertheless, the number of Multi-carrier Spread-Spectrum (MC-SS) MUX channels can be either arbitrarily extended according to the number of coding layers generated by the source encoder, or reduced whether it is decided to transmit only a subset of the coding layers produced. The detail on the actual implementation of each MUX channel is shown in Figure 6.

<INSERT FIGURE 6>

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 bitrates of each MPEG4 sub-stream $L_k(t)$ corresponding to the *k*-th layer can be expressed as integer multiples of the base layer bitrate 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 state that $Q = 6$, *k*=1 (FGS2 layer), *k*=3 (FGS1 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 [28-29], 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 [26] (usual values for such a parameter are *F*=1 or *F*=2). Note that *k* and *Q* should change in case of different layered coding and different rate assignment, but the orthogonal sub-channel allocation rule of equation (2) would remained the same.

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. In fact, the highest number of sub-channels (512) is attributed to the most important layer, i.e., the base-layer, while the lowest number of sub-channels (16) is attributed to FGS2, which incidentally is the highest bitrate channel.

<INSERT TABLE 2>

It should be also noted that the multicarrier spreading of the transmitted sub-streams is performed by a "full-digital" I-FFT block that may work either at IF or in baseband without loss of generality (in baseband, f_0 will be null). Nevertheless, the current trend of full-digital SDR-based devices is to work up to IF, in order to allow efficient software implementation of the complete demodulation chain (including carrier recovery algorithms) [39].

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

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

where *P* is the carrier power, assumed to be equal for all layers and all users (no power management is performed at present, although it can be considered in future investigations), $c_k[u,n]$ 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 [28-29], Orthogonal Variable Spreading Factor codes [33] have been employed), and $\Pi(t)$ is the digital waveform assumed for simplicity as rectangular NRZ pulse of unit amplitude. Since a fixed power has been adopted, the transmission signal-to-noise ratios $E_b(k)/\eta$ (where η is the one-sided power spectral density of AWGN noise) are different from a transmitted layer to another, 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, FGS2 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 to IF can be expressed as follows:

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

$g_n(t) = \alpha_n(t)e^{j\phi_n(t)}$ is the complex channel coefficient related to the *n*-th orthogonal sub-channel, and *z(t)* is an additive complex Gaussian noise. 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 [26]. Such a 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 by the channel by means of a coherent detection [26]. 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 as 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 with in [41]. Afterwards, the received baseband signal is low-pass filtered and sampled before symbol decision (timing recovery is assumed to be ideal). The advantages of the EGC scheme are mainly related to the reduced algorithmic complexity. The EGC scheme is theoretically optimal in the case of synchronous multi-user transmission over AWGN channel [26]. In such a case, ideal user orthogonalization is retained and multi-user interference (MUI) does not arise. On the other hand, 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 limit receiver performance.

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 mode 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 treated in future works.

<INSERT FIGURE 7>

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 distance between satellite and earth station. According to the exhaustive analysis of non-linear distortions in MC-CDMA transmission systems presented in [42], we considered the presence in the transmission chain of a Solid-State-Power-Amplifier (SSPA) installed on-board LEO satellite, whose normalized AM/AM characteristic is given as follows (a SSPA does not usually introduce phase distortion [42]):

$$\gamma_{SSPA}(t) = \frac{\gamma(t)}{\left(1+\gamma(t)^{10}\right)^{1/10}} \tag{6}$$

where $\gamma(t)$ is the envelope of the multi-user VBR MC-CDMA signal transmitted onto the channel.

3. Experimental results

3.1 SIMULATION PROCEDURES

A realistic simulator of the video transmission system described in Section 2 has been implemented in order to validate the proposed methodology. The simulator has been developed in two steps. The first step dealt with the implementation of the multi-layered MPEG4 video codec. The second step was related to simulation of a variable-bitrate multi-carrier CDMA of MPEG4-coded video streams, together with their transmission over LEO satellite downlink channel.

The MPEG4 standard reference software was adopted as codec kernel [40]. Two open-source versions of such software have been released, a first version known as *MoMuSys* developed in ANSI C, and a second one, developed by Microsoft® in C++ language. Both codecs provide the most important features mechanisms of the standard, while none of them implements tools to manage stream errors. In our simulations the latter was preferred because of the possibility of easily monitoring the effects of channel errors on the decoded stream.

As far as the multiplexing and transmission aspects are concerned, a frequency-selective LEO satellite channel was simulated by means of a Ricean tapped-delay-line model [5], parameterised by keeping into account the experimental data presented in [37]. In Table 3, the numerical parameterisation of the tapped delay line channel model is shown:

<INSERT TABLE 3>

From Table 3, one can derive that the delay spread of the channel is equal to 500nsec, corresponding to a coherence bandwidth of 2MHz. The parameterisation in terms of Doppler spread of the LEO satellite channel was extrapolated by results shown in [38]. In particular, a Doppler spread equal to 32 KHz has been chosen for our simulations. Both satellite channel and whole modem chain have been simulated in MATLAB SIMULINK V6.5 environment.

3.2 SIMULATION RESULTS

In this section, a selection of simulation results achieved in an extensive testing phase are presented. Due to the large computation required by the simulations, it was not feasible to iterate several times the simulation of long test sequences. For this reason, we articulated the simulation into two types of trial: a single-trial simulation of a 5 seconds video clip (consisting of 125 QCIF frames), and an iterated simulation (10 trials) of a shorter clip (1.24 second of video consisting of 31 QCIF frames). Simulations of single user and multi-user transmission have been performed on the following widely used test video sequences:

- *Single-user transmission*: the user transmits a clip of "*Carphone*";
- *Multi-user transmission*: K=3 users. User 1,2, and 3 transmits clips of "*Carphone*", "*Foreman*", and "*Mother and daughter*", respectively.

It is worth noting that also the above-mentioned "single-user" transmission practically is a multi-user one, 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 affected by inter-layer interference, which is actually a form of multi-user interference.

Results achieved by the proposed transmission technique are compared with a more conventional single-layer MC-SS transmission using the same bandwidth. 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 to 6 referring to single-user, multi-user single-layer, and multi-user multi-layer transmission, respectively. The BER values achieved by simulations are in agreement with theoretical values typical of VBR MC-CDMA transmissions over multipath fading channels.

<INSERT TABLES 4, 5, AND 6>

In Figure 8, a set of sample frames is shown related to the single-user transmission of the long sequence for a *SNR* value of the FGS2 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 the latter case, the *SNR* of the FGS2 layer is equal to 12dB. These results make evident that multi-layer VBR MC-CDM transmission outperforms single-layer MC-SS transmission.

<INSERT FIGURES 8, 9, 10 AND 11>

The PSNR provides an objective confirmation of the perceived quality shown in the above figures. We considered three different measures:

- Average PSNR values vs. FSG2 *SNR* values (namely: *SNR(FGS2)*) achieved for the long "carphone" sequence after decoding all received layers (i.e., BL, BTL, FGS1, FGS2). Such results are shown in Figure 12 (single-user transmission) and Figure 13 (multi-user transmission);
- Incremental PSNR results vs. frame number achieved for the short "carphone" sequence by layer-after-layer decoding. Such results have been shown for the single-user case @*SNR(FGS2)*=11dB (Figure 14), and for the multi-user case @*SNR(FGS2)*=12dB (Figure 15) being able to point out the incremental quality achievable by the multi-layered coding;
- Incremental average PSNR results vs. *SNR(FGS2)* achieved for the long "carphone" sequence by layer-after-layer decoding. Such results are shown in Figure 16 in order to validate in further way results shown in Figures 14 and 15.

<INSERT FIGURE 12, 13, 14, 15, 16>

In Figure 12, one can notice that the two curves related to the single layer and multi-layer transmission tend to become coincident for *SNR(FGS2)* > 14dB (when both cases are "error-free"). On the other side, a dramatic quality increase for the multi-layer transmission can be also observed when *SNR(FGS2)* is in the range [10-13]dB. This fact is consequential to the UEP provided by the VBR MC-CDM mechanism to the different coding layers. A quite similar PSNR pattern can be observed in Figure 13 for the multi-user transmission case. Of course, the increasing multi-user interference amount (due to the simultaneous presence of inter-layer and multi-layer interference) involves a performance degradation with respect to the single-user case, measured in about 2dB. PSNR curves vs. frame number of Figures 14 and 15 clearly shows the effectiveness of the proposed video transmission methodology, as compared to the corresponding curves of Figure 3 (error-free transmission case). We can observe that in both cases, the first three layers (BL, BTL and FGS1) are successfully decoded, because they are received "error-free". This means that a mobile user can play at least a fairly high-quality full-rate continuous video. But also FGS2 layer provides a considerable quality improvement, even though channel errors affecting such a layer prevent users to play a top-quality video. Figure 16 summarizes the overall simulation results. It is noticeable that BL and BTL layers can be always successfully decoded also in case of low *SNR(FGS2)* values. This means that also in this case a mobile user can play at least a full frame-rate low-quality video. Moreover, we can observe that for *SNR(FGS2)* > 8dB, FGS1 layer can be decoded without errors, thus providing a good quality continuous video. Finally, we can point out that for *SNR(FGS2)* > 10dB, FGS2 layer begins to yield a positive contribution in the decoding of the video sequence, progressively enhancing the perceived visual quality.

4. 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. Extensive tests provided the experimental proof of the robustness of VBR MC-CDM approach, achieved through a UEP mechanism implicit in the multi-carrier multiplexing. Future developments will include the investigation of the use of multi-user detection algorithms, the adaptive management of power resources attributed to each MUX channel, the integration of error resilience tools in the standard MPEG-4 encoding. Furthermore, aspects related to the actual implementation of the proposed video transmission system over HW/SW architectures (DSP) will be considered.

Finally, it should be mentioned that the proposed analysis, limited to a test-case related to a mobile satellite video streaming application, can be easily extended to other application scenarios working in different wireless environments (e.g., cellular MAN, indoor WLAN, etc.).

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CAPTIONS OF THE FIGURES

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

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

FIGURE 3: Incremental PSNR values vs. frame number related to the progressive decoding of the four MPEG-4 layers without transmission.

FIGURE 4: Sample couples of frames for: BLT layer (a), FGS1 layer (b), FGS2 layer (d).

FIGURE 5: Block diagram of the single-user VRB MC-SS video multiplexer;

FIGURE 6: Detail of the implementation of the generic VBR MC-SS MUX channel.

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

FIGURE 8: Triplets of sample frames of the received decoded sequence (single-user transmission, SNR of the FGS2 layer = 10dB): (a), (b), (c): single-layer MC-SS transmission, (d), (e), (f): VBR MC-SS transmission with multi-layered coding.

FIGURE 9: Triplets of sample frames of the received decoded sequence (multi-user transmission, SNR of the FGS2 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.

FIGURE 10: Triplets of sample frames of the received decoded sequence (multi-user transmission, SNR of the FGS2 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.

FIGURE 11: Triplets of sample frames of the received decoded sequence (multi-user transmission, SNR of the FGS2 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.

FIGURE 12: Average PSNR values vs. SNR of FGS2 layer for the single-user transmission case (long “carphone” sequence).

FIGURE 13: Average PSNR values vs. SNR of FGS2 layer for the multi-user transmission case (User 1 – long “carphone” sequence).

FIGURE 14: Incremental PSNR values vs. frame number related to the progressive decoding of the four MPEG-4 layers in the single user transmission case (short “carphone” sequence, *SNR (FGS2)*=11dB).

FIGURE 15: Incremental PSNR values vs. frame number related to the progressive decoding of the four MPEG-4 layers in the multi-user transmission case (User 1 – short "carphone" sequence, *SNR(FGS2)*=12dB).

FIGURE 16: Average incremental PSNR values vs. SNR(FGS2) for the multi-user transmission case (User 1 – long "carphone" sequence).

CAPTIONS OF THE TABLES

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

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

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

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

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

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

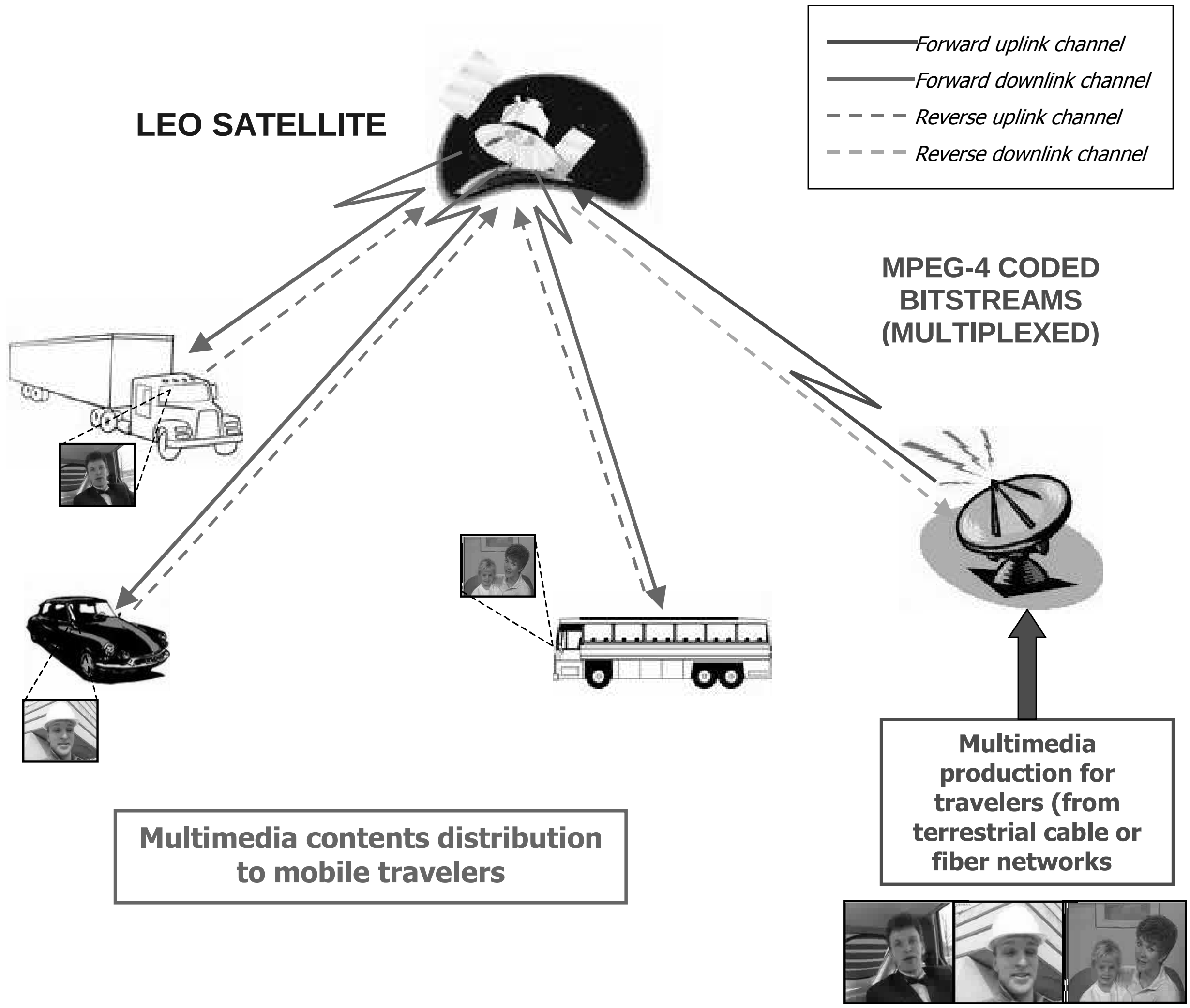
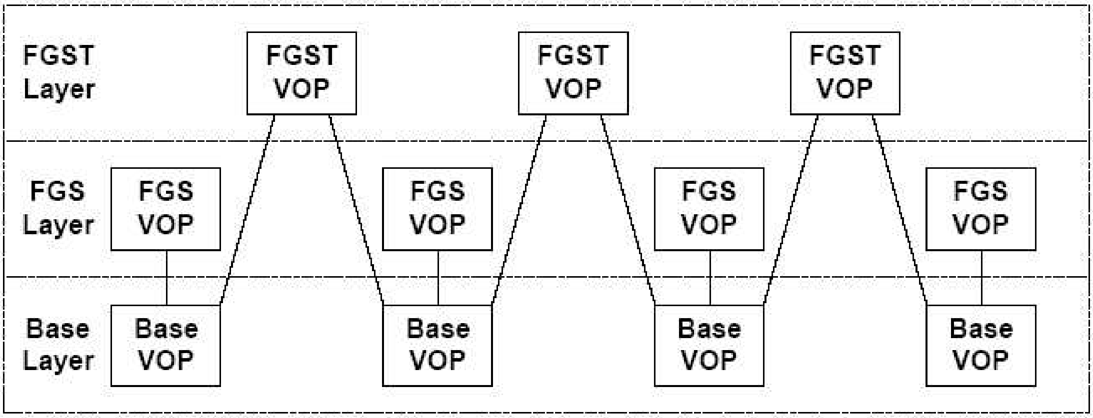
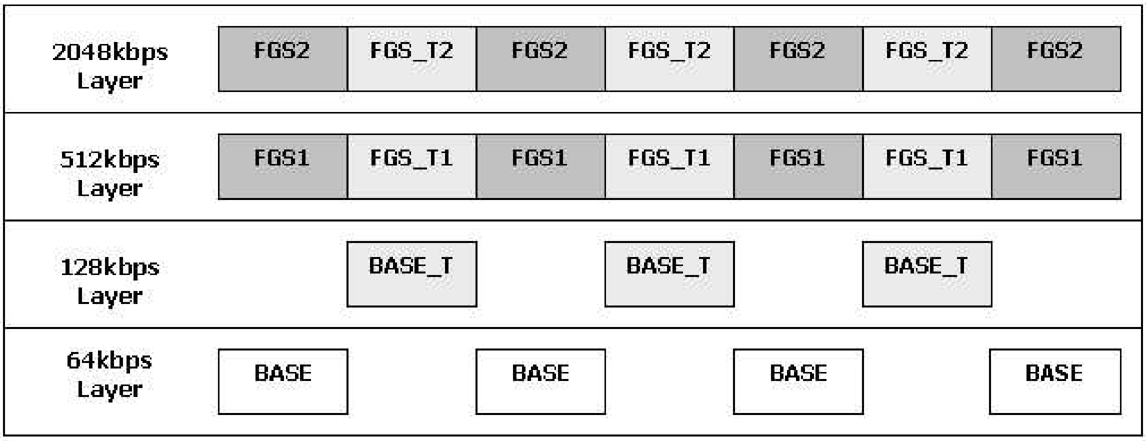

LEO SATELLITE
Forward uplink channel
Forward downlink channel
Reverse uplink channel
Reverse downlink channel
MPEG-4 CODED BITSTREAMS (MULTIPLEXED)
Multimedia production for travelers (from terrestrial cable or fiber networks
Multimedia contents distribution to mobile travelers

FIGURE 1


FGST Layer
FGST VOP
FGST VOP
FGST VOP
FGS Layer
FGS VOP
FGS VOP
FGS VOP
FGS VOP
Base Layer
Base VOP
Base VOP
Base VOP
Base VOP

(a)


2048kbps Layer
FGS2 FGS_T2 FGS2 FGS_T2 FGS2 FGS_T2 FGS2
512kbps Layer
FGS1 FGS_T1 FGS1 FGS_T1 FGS1 FGS_T1 FGS1
128kbps Layer
BASE_T BASE_T BASE_T
64kbps Layer
BASE BASE BASE BASE

(b)

FIGURE 2

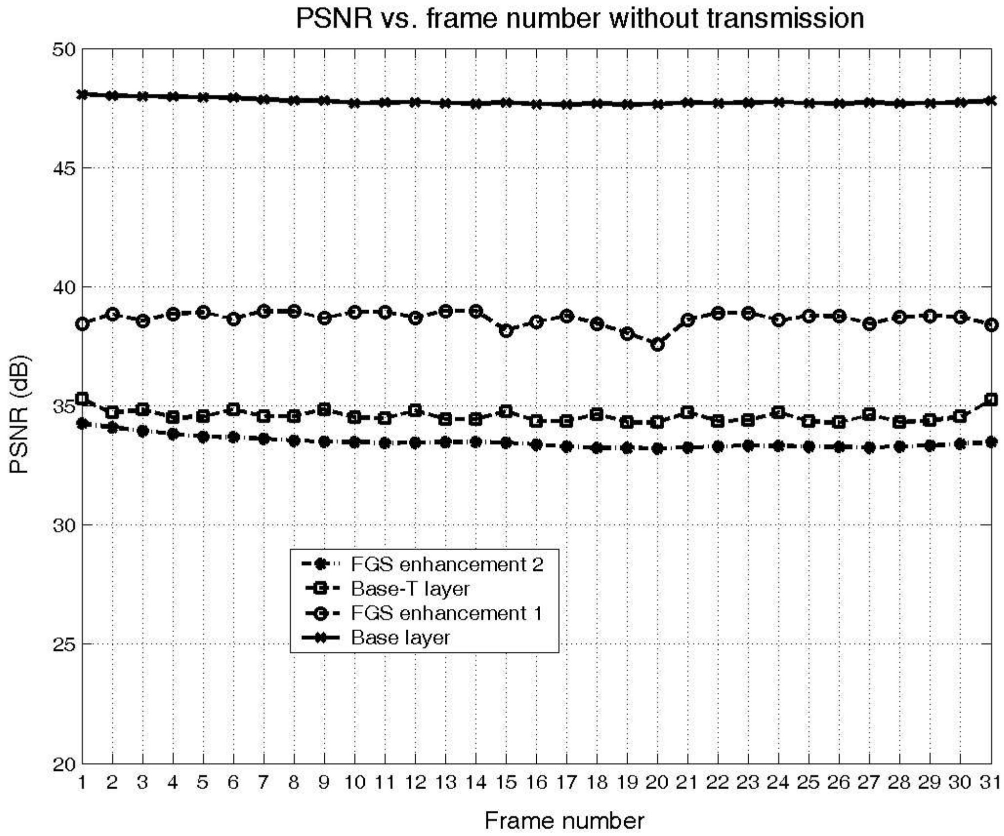

PSNR vs. frame number without transmission
PSNR (dB)
Frame number
FGS enhancement 2
Base-T layer
FGS enhancement 1
Base layer
50
45
40
35
30
25
20
1 2 3 4 5 6 7 8 9 10 11 12 13 14 15 16 17 18 19 20 21 22 23 24 25 26 27 28 29 30 31

FIGURE 3

(a)

(b)

(c)

FIGURE 4

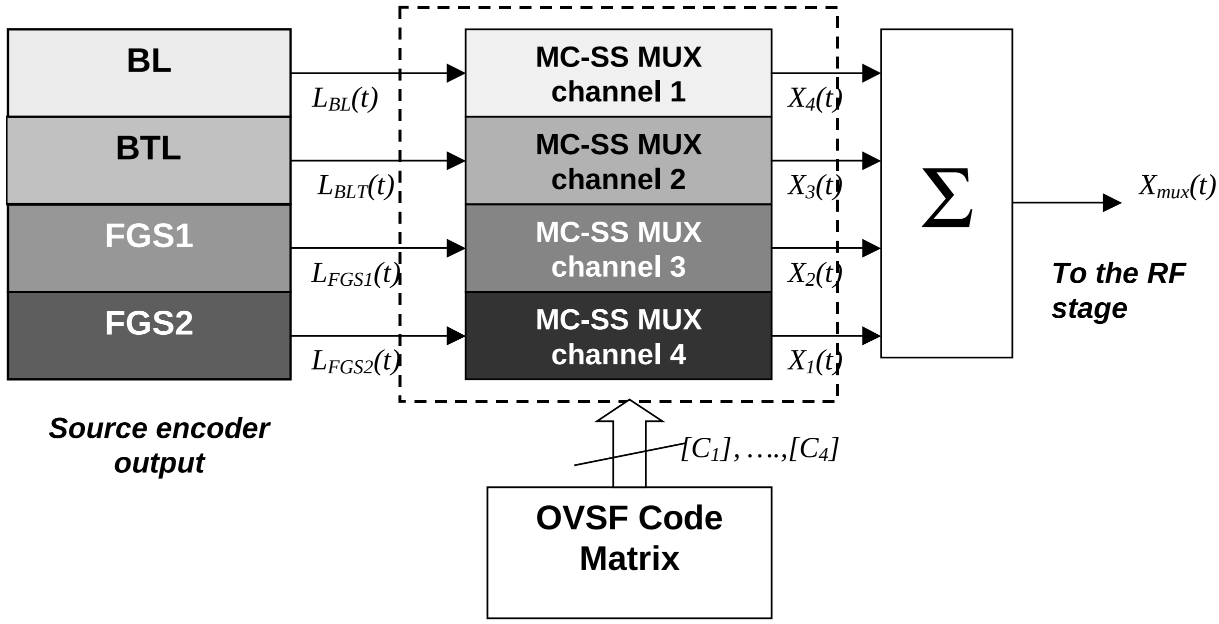


FIGURE 5

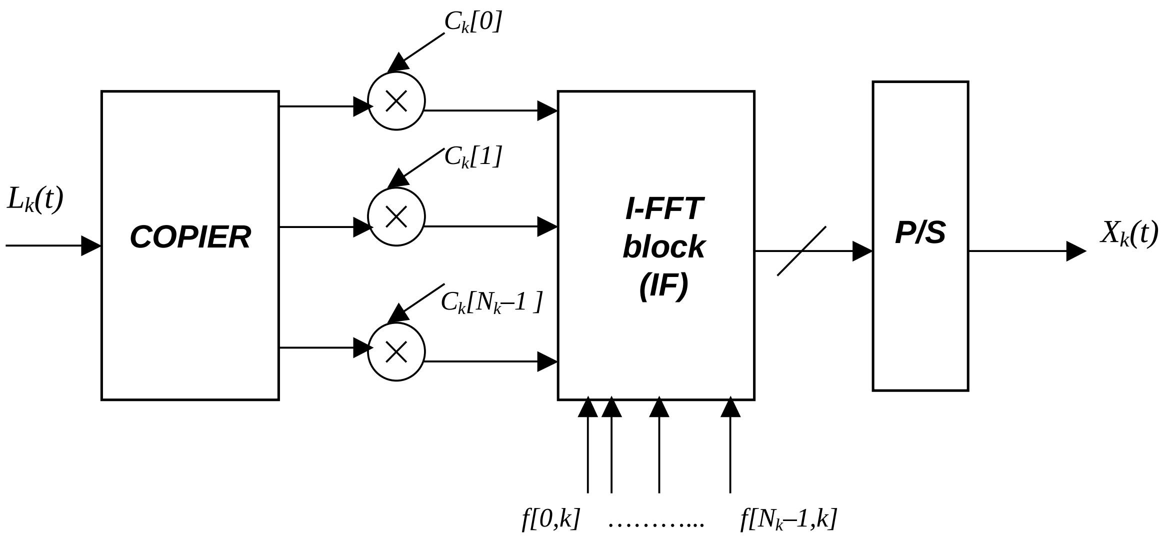


FIGURE 6

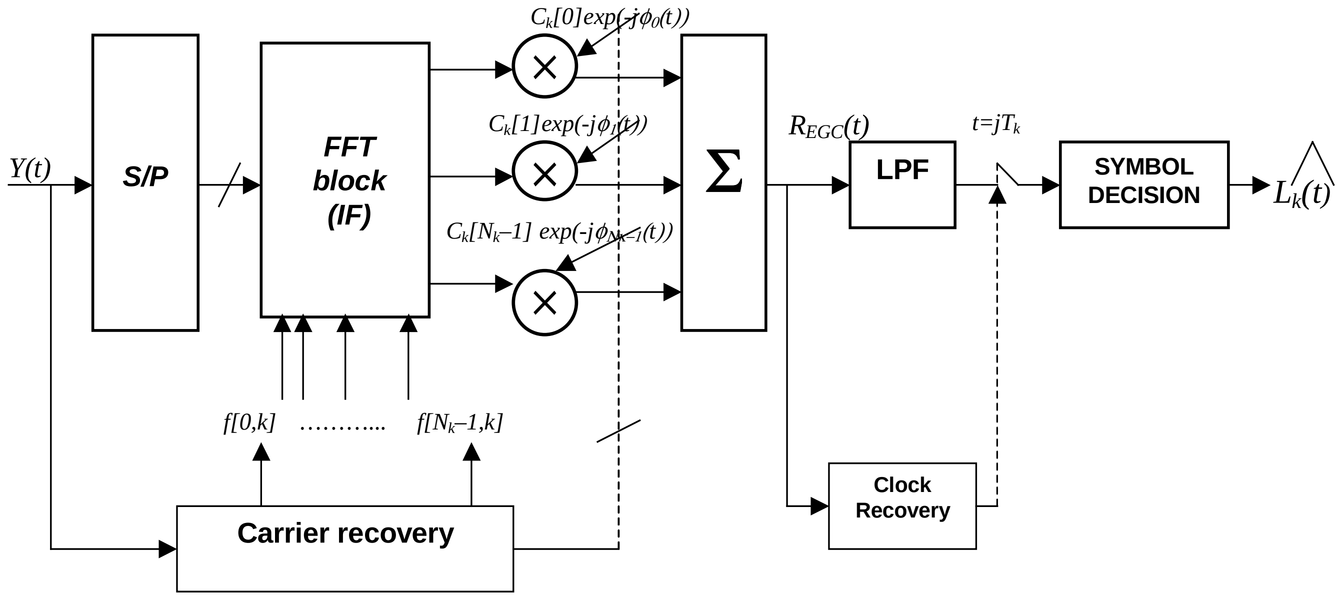
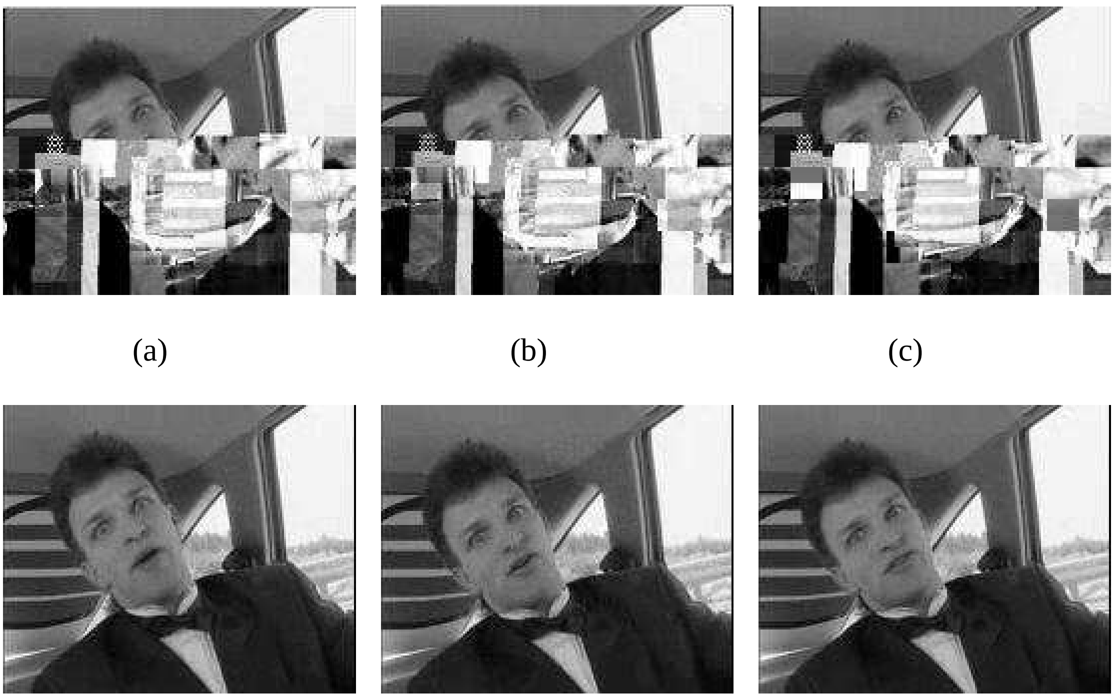


FIGURE 7



(a) (b) (c)

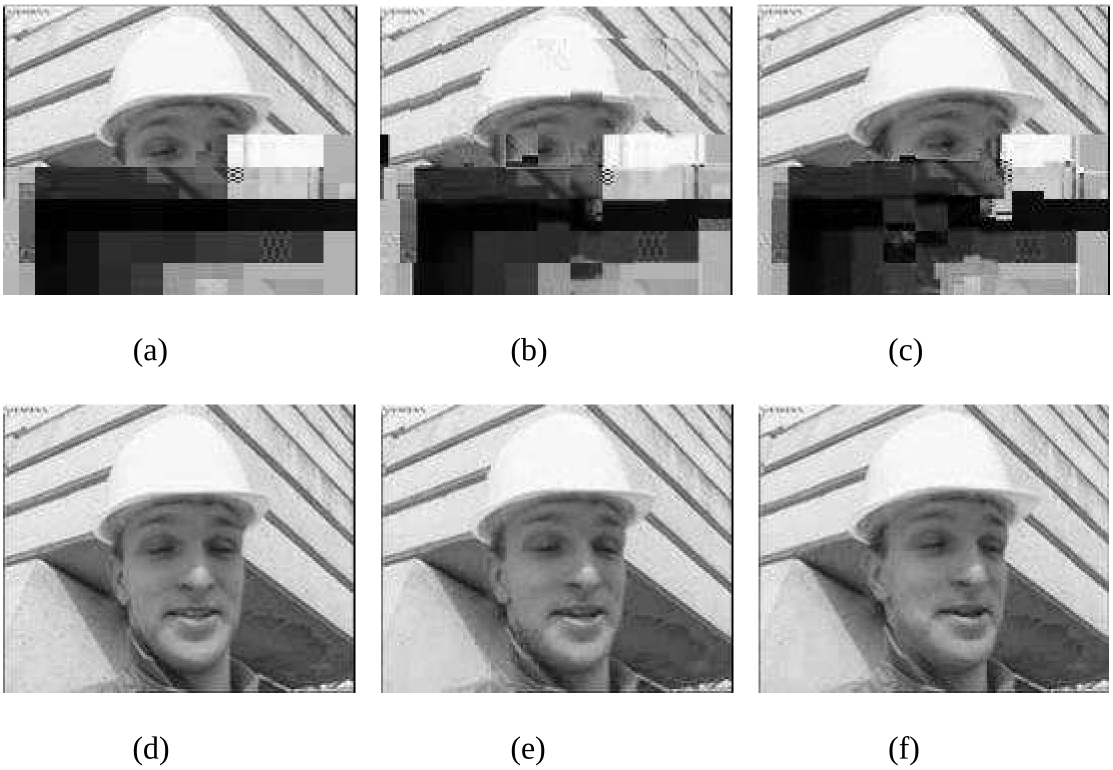
(d) (e) (f)

FIGURE 8

(a) (b) (c)

(d) (e) (f)

FIGURE 9



(a) (b) (c)

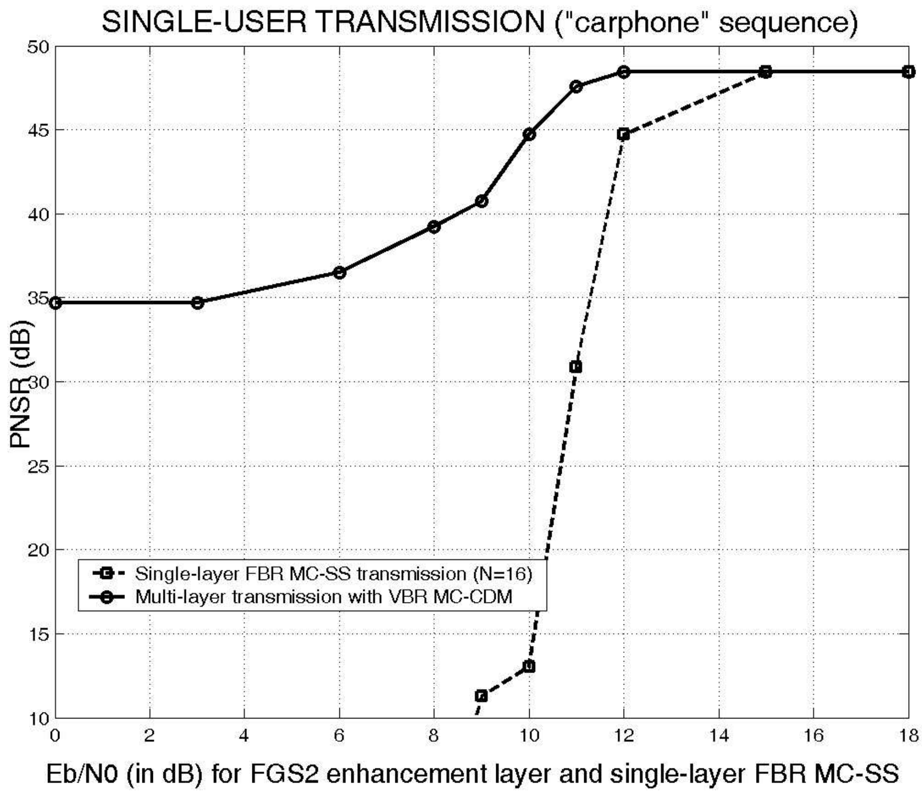
(d) (e) (f)

FIGURE 10

(a) (b) (c)

(d) (e) (f)

<u>FIGURE 11</u>



<u>FIGURE 12</u>

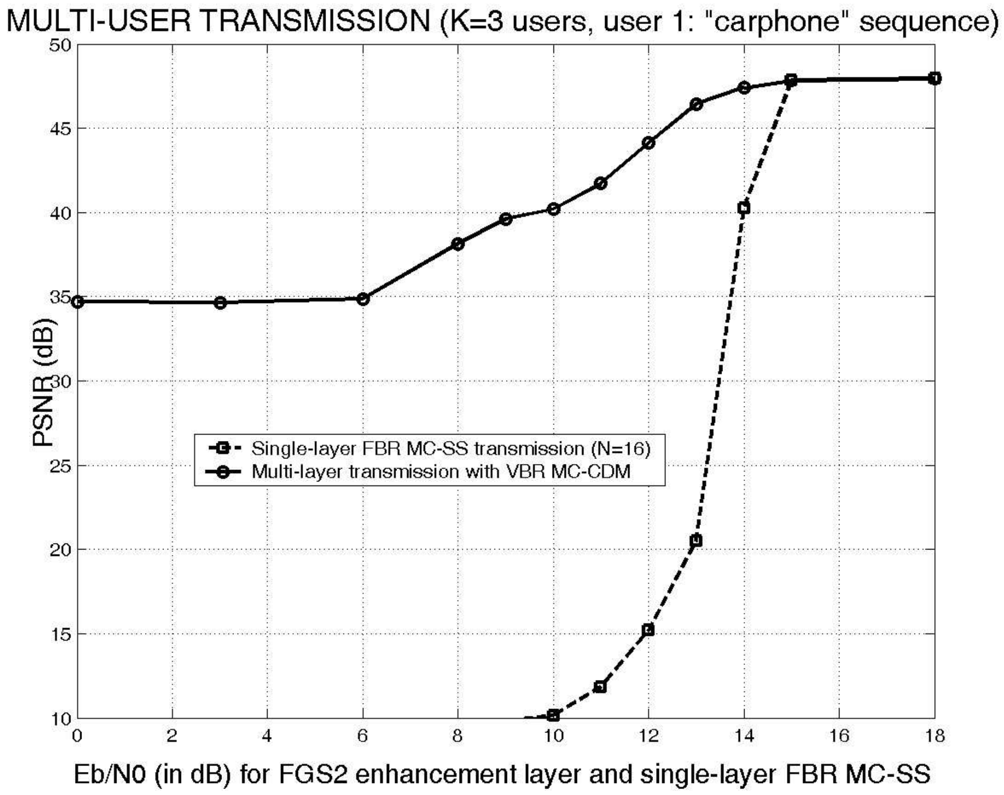


FIGURE 13

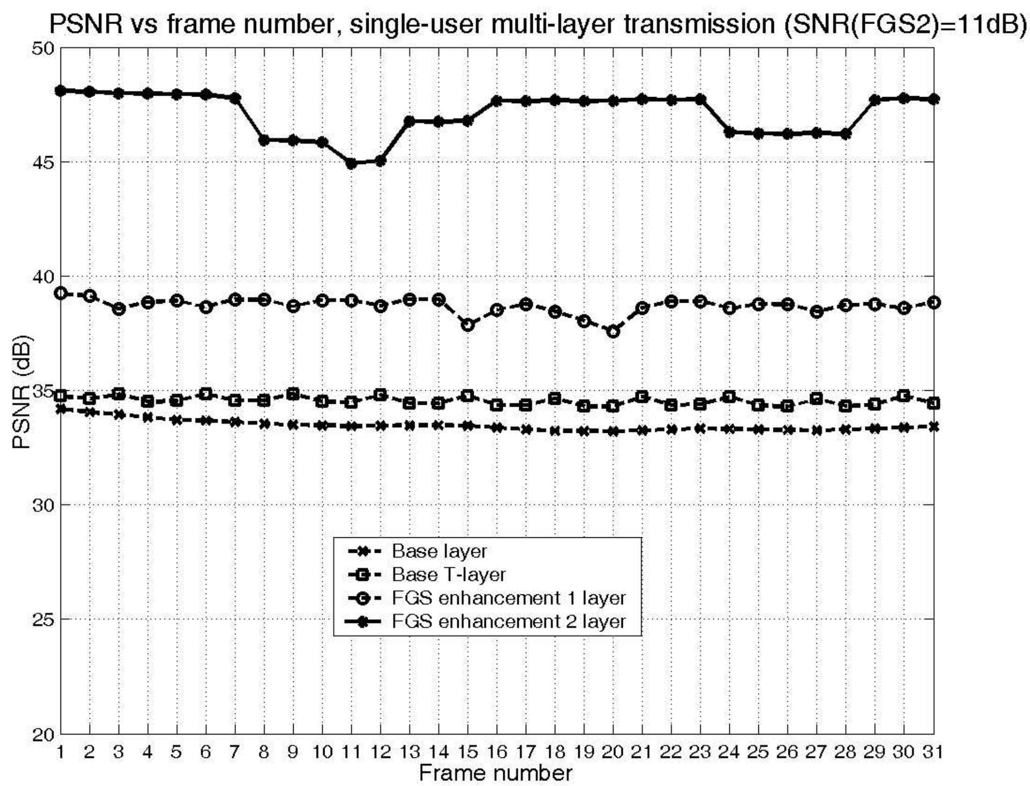


FIGURE 14

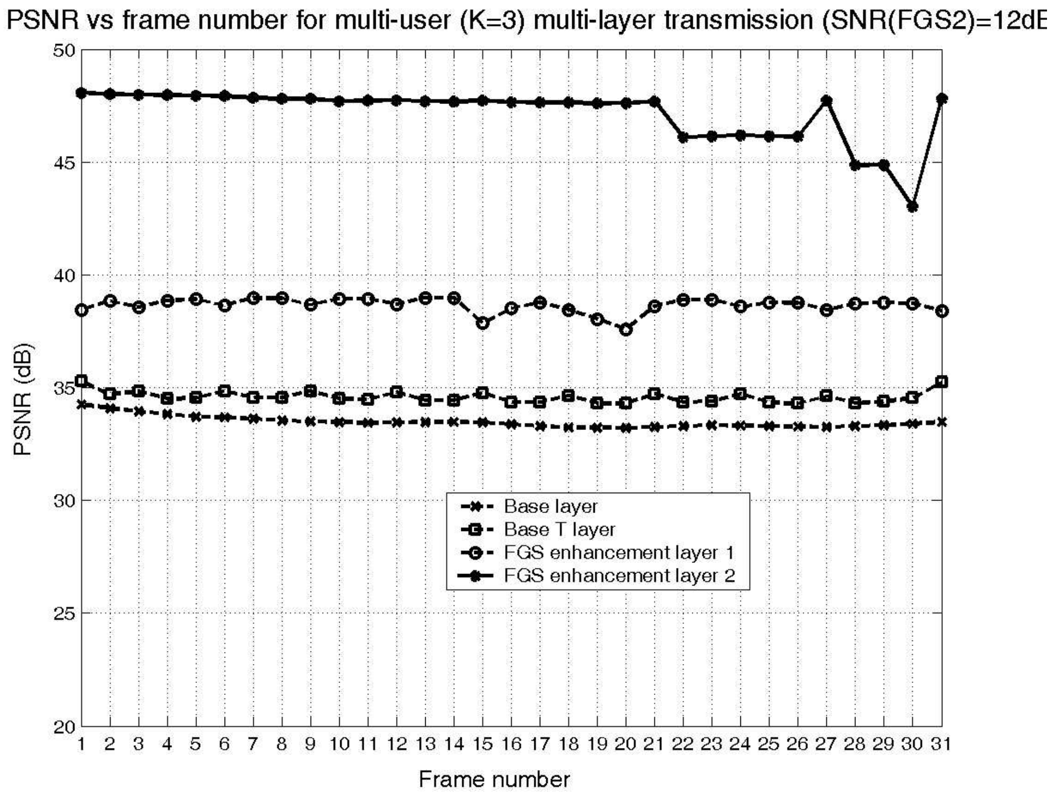


FIGURE 15

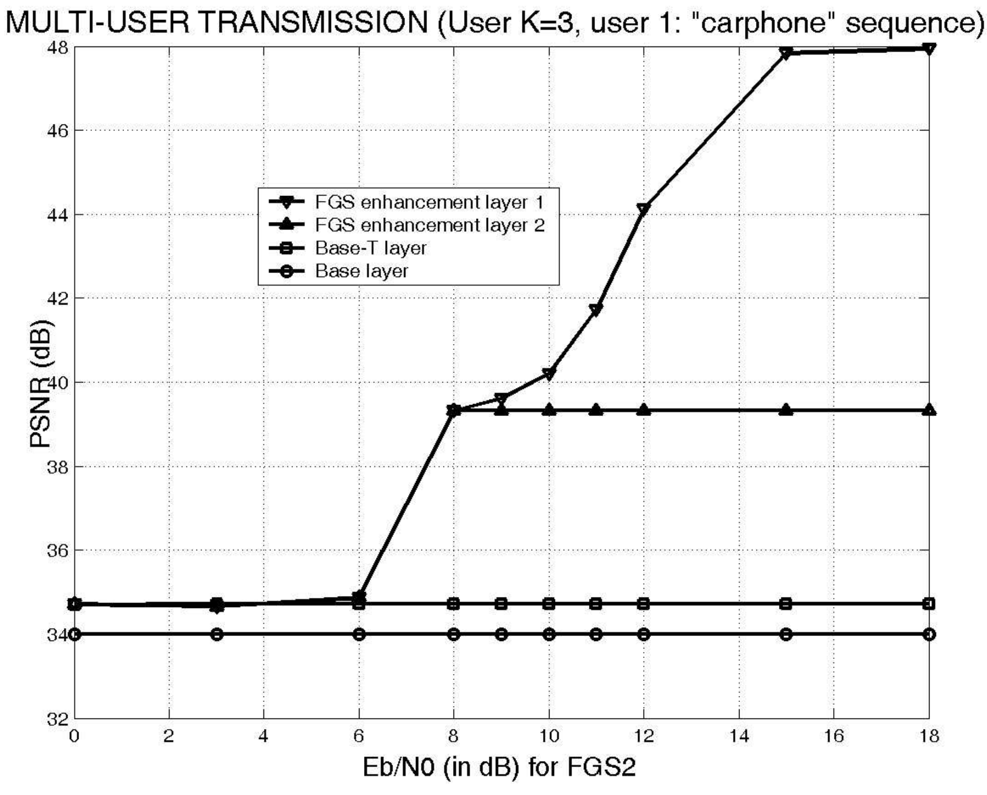


FIGURE 16

TABLE 1

MPEG-4 CODING LAYER	FRAME-RATE	SIZE (KILOBYTES)	INCREMENTAL AVERAGE PSNR	CHANNEL BIT-RATE	TIME REQUIRED FOR TRANSMISSION (SEC.)
BL	8.33fps	20.1	34.1dB	64Kb/s	2.51
BTL	25fps	52.2	35.3dB	128Kb/s	3.27
FGS1	25fps	189.2	39.8dB	512Kb/s	2.96
FGS2	25fps	472.2	48.2dB	2048Kb/s	1.84

TABLE 2

	MPEG-4 LAYER	NUMBER OF ORTHOGONAL SUB-CHANNELS (N_l)	TRANSMISSION SNR (E_{bk}/η)
MUX CHANNEL 1	BL	512	SNR(FGS2)+15dB
MUX CHANNEL 2	BTL	256	SNR(FGS2)+12dB
MUX CHANNEL 3	FGS1	64	SNR(FGS2)+6dB
MUX CHANNEL 4	FGS2	16	SNR(FGS2) (assigned)

TABLE 3

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

TABLE 4

	SINGLE-USER TRANSMISSION (“CARPHONE SEQUENCE)				
	BASE LAYER	BASE-T LAYER	FGS ENHANCEMENT LAYER 1	FGS ENHANCEMENT LAYER 2	SINGLE LAYER MC-SS TRANSMISSION
SNR(FGS2)=0dB	Error free	Error free	$5.6*10^{-2}$	$9.42*10^{-2}$	$9.42*10^{-2}$
SNR(FGS2)=3dB	Error free	Error free	$2.70*10^{-4}$	$3.29*10^{-2}$	$3.28*10^{-2}$
SNR(FGS2)=6dB	Error free	Error free	$1*10^{-6}$	$5.4*10^{-3}$	$5.4*10^{-3}$
SNR(FGS2)=8dB	Error free	Error free	Error free	$8.4*10^{-4}$	$8.23*10^{-4}$
SNR(FGS2)=9dB	Error free	Error free	Error free	$2.33*10^{-4}$	$2.30*10^{-4}$
SNR(FGS2)=10dB	Error free	Error free	Error free	$5.79*10^{-5}$	$5.79*10^{-5}$
SNR(FGS2)=11dB	Error free	Error free	Error free	$9.93*10^{-6}$	$1,02*10^{-5}$
SNR(FGS2)=12dB	Error free	Error free	Error free	$1.66*10^{-6}$	$1,47*10^{-6}$
SNR(FGS2)=15dB	Error free	Error free	Error free	Error free	Error Free
SNR(FGS2)=18dB	Error free	Error free	Error free	Error free	Error Free

TABLE 5

	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

TABLE 6

	MULTI-LAYER MULTI-USER MC-CDMA TRANSMISSION USER 1 (“CARPHONE”)			
	BASE LAYER	BASE-T LAYER	FGS ENHANCEMENT LAYER 1	FGS ENHANCEMENT LAYER 2
SNR(FGS2)=0dB	Error free	Error free	$5.5*10^{-2}$	$9.89*10^{-2}$
SNR(FGS2)=3dB	Error free	Error free	$2.57*10^{-4}$	$3.86*10^{-2}$
SNR(FGS2)=6dB	Error free	Error free	$5.03*10^{-7}$	$9*10^{-3}$
SNR(FGS2)=8dB	Error free	Error free	Error free	$2.3*10^{-3}$
SNR(FGS2)=9dB	Error free	Error free	Error free	10^{-3}
SNR(FGS2)=10dB	Error free	Error free	Error free	$4.53*10^{-4}$
SNR(FGS2)=11dB	Error free	Error free	Error free	$1.7*10^{-4}$
SNR(FGS2)=12dB	Error free	Error free	Error free	$6.45*10^{-5}$
SNR(FGS2)=13dB	Error free	Error free	Error free	$2.29*10^{-5}$
SNR(FGS2)=14dB	Error free	Error free	Error free	$7.18*10^{-6}$
SNR(FGS2)=15dB	Error free	Error free	Error free	$2.70*10^{-6}$
SNR(FGS2)=18dB	Error free	Error free	Error free	Error free