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# Scalable Video Coding in Fading Hybrid Satellite-Terrestrial Networks

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## 1. Introduction

Broadband satellite multimedia (BSM) systems will be an integral part of the global information infrastructure as one of the major technologies providing both broadband access and broadcast services (Skinnemoen & Tork, 2002). Recent commercial deployments show that users not only would like to have access to value-added services (e.g., mobile internet, multimedia streaming, etc.) but are also willing to pay more for them and in particular for video services (Sattler). The introduction of video coding technology in the satellite application space opens up new and challenging topics; digital video applications have to face potentially harsher transmission environments than ones they were originally designed to work with (e.g., HDTV, Mobile TV), especially as regards traversing packet networks with the presence of satellite links. Towards approaching the satellite multimedia application delivery needs, H.264/MPEG4 Advanced Video Coding (AVC) (Ostermann et al, 2004), as the latest entry of international video coding standards, has demonstrated significantly improved coding efficiency, substantially enhanced error robustness, and increased flexibility and scope of applicability relative to its predecessors (Marpe et al, 2002). In the last decade, there is a growing research interest for the transmission and study of multimedia content over IP networks (Chou & van der Schaar, 2007) and wireless networks (Rupp, 2009). In an increasing number of applications, video is transmitted to and from satellite networks or portable wireless devices such as cellular phones, laptop computers connected to wireless local area networks (WLANs), and cameras in surveillance and environmental tracking systems. Wireless networks are heterogeneous in bandwidth, reliability, and receiver device characteristics. In (satellite) wireless channels, packets can be delayed (due to queuing, propagation, transmission, and processing delays), lost, or even discarded due to complexity/power limitations or display capabilities of the receiver (Katsaggelos et al, 2005). Hence, the experienced packet losses can be up to 10% or more, and the time allocated to the various users and the resulting goodput1 for multimedia bit stream transmission can also vary significantly in time (Zhai et al, 2005). This variability of wireless resources has considerable consequences for multimedia applications and often leads to unsatisfactory user experience due to the high bandwidths and to very stringent delay constraints. Fortunately, *multimedia applications* can cope with a certain amount of packet losses depending on the used sequence characteristics, compression schemes, and error concealment strategies available at the receiver (e.g., packet losses up to 5% or more can be tolerated at times). Consequently, unlike file transfers, real time multimedia applications do not require a complete insulation from packet losses, but rather require the application layer *to cooperate* with the lower layers to select the optimal wireless transmission strategy that maximizes the multimedia performance. Thus, to achieve a high level of acceptability and proliferation of wireless multimedia, in particular wireless video (Winkler, 2005), several key requirements need to be satisfied by multimedia streaming solutions (Wenger, 2003) over such channels: (i) easy adaptability to wireless bandwidth fluctuations due to cochannel interference, multipath fading (Pätzold, 2002), mobility, handoff, competing traffic, and so on; (ii) robustness to partial data losses caused by the packetization of video frames and high packet error rates. This chapter tackles in a unified framework both the (satellite) wireless channel modeling and scalable video coding components in the context of satellite-terrestrial broadcasting/multicasting systems (Kiang et al, 2008). It should be mentioned that the literature is poor in the analysis of the effects produced by corrupted bits in compressed video streams (Celandroni et al, 2004), and an attempt is done here to contribute some results to this open field of research. Some technical aspects both in terms of the video coding system and the satellite channel are provided in Section II. Section III deals with the joint source and channel simulation, and Section IV presents the simulation results. The last Section V contains the conclusions and future improvements on the proposed work.

## 2. Technical background

### 2.1 Video coding scheme (AVC, SVC)

H.264, or MPEG-4 AVC (advanced video coding) (ITU-T, 2003) is the state-of-the-art video coding standard (Richardson, 2005). It provides improved compression efficiency, a comprehensive set of tools and profile/level specifications catering for different applications. H.264/AVC (Ostermann et al, 2004) has attracted a lot of attention from industry and has been adopted by various application standards and is increasingly used in a broad variety of applications. It is expected that in the near-term future H.264/AVC will be commonly used in most video applications. Given this high degree of adoption and deployment of the new standard and taking into account the large investments that have already been taken place for preparing and developing H.264/AVC-based products, it is quite natural to now build a SVC scheme as an extension of H.264/AVC and to reuse its key features. Furthermore, its specification of network abstraction layer (NAL) separate from the video coding layer (VCL) makes the standard much more network-friendly as compared with all its predecessors. The standard is first established in 2003 jointly by ITU-T VCEG (video Coding Experts Group) and ISO/IEC MPEG (Moving Picture Experts Group). The partnership, known as JVT (Joint Video Team), has been constantly revising and extending the standards ever since. SVC Considering the needs of today's and future video applications as well as the experiences with scalable profiles in the past (Cycon et al, 2010), the success of any future SVC standard critically depends on the following essential requirements. Similar coding efficiency compared to single-layer coding—for each subset of the scalable bit stream.

- Little increase in decoding complexity compared to single layer decoding that scales with the decoded spatio–temporal resolution and bit rate.
- Support of temporal, spatial, and quality scalability.

- Support of a backward compatible base layer (H.264/AVC in this case).
- Support of simple bit stream adaptations after encoding.

SVC (Scalable Video Coding) (Schwarz et al, 2003) is the newest extension established in late 2007. Formally known as Annex G extension to H.264, SVC allows video contents to be split into a base layer and several enhancement layers, which allows users with different devices and traffic bearers with different capacities to share the video without provided multiple copies of different qualities.

## 2.2 SVC in our approach

Although scalability in video is not a new concept, the recent standardization acts as a catalyst to its acceptance into different market segments. In our approach, a layered approach to video coding similar to SVC is used to split video payload into 2 streams (Kiang et al, 2008). The base layer provides near-guaranteed, low resolution video whereas the enhancement layer provides the additional information required to improve the base-layer to a the low- and high-fidelity videos, it should be transmitted with a higher protection against corruption due to channel errors. Video coding in this work involves both AVC and a layered approached similar to SVC (based on AVC). For completeness, some crucial factors that make AVC a superior video coding standard are listed below:

- INTRA pictures and INTRA- regions within INTER pictures are coded with prediction from neighboring blocks. The prediction can be done in different directions, depending on the way the regions are textured (e.g. horizontal or vertical striped, checked boxed patterns etc.)
- Variable block-sizes are allowed in both INTRA- (16x16 and 4x4) and INTER-modes (16x16, 16x8, 8x16, 8x8 and other sub-8x8 blocks in multiple of 4).
- Motion estimation with possible resolution down to ¼- pixels.
- New integer-based 4x4 transform and options 8x8 transform.
- 6-tap filters for ½-pixel and bilinear filter for ¼-pixel luma-sample resolutions.
- Quantization based on logarithmic-scale.
- In-loop loop filter for removing blocking effects.

The SVC extension enables the AVC encoder to produce and base layer and incrementally improve the quality by providing differential information. Three types of scalability can be

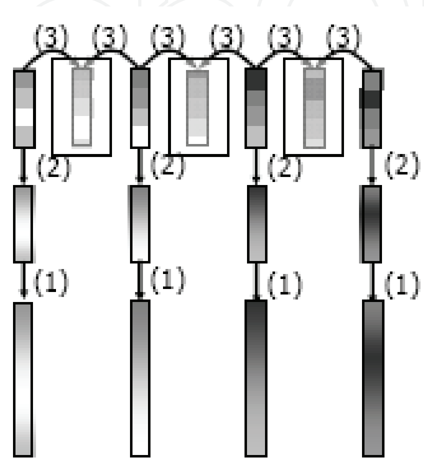


Fig. 1. Different scalabilities: (1) Spatial; (2) SNR (quality); (3) temporal (Kiang et al, 2008).

identified based on how the incremental information is used to improve quality. They are (1) spatial scalability (variation of picture resolution), (2) SNR scalability (variation of quality) and (3) temporal scalability (variation of frame rate). The 3 forms of scalability are illustrated in the figure 1. Different combinations of scalability can be used to adapt to the channel conditions.
In this approach, spatial scalability is issued to produce the enhanced video layer.

### 2.3 Fading hybrid satellite terrestrial networks

In mobile radio communications, the emitted electromagnetic waves often do not reach the receiving antenna directly due to obstacles blocking the line-of-sight path. In fact, the received waves are a superposition of waves coming from all directions due to reflection, diffraction, and scattering caused by buildings, trees, and other obstacles. This effect is known as *multipath propagation (*Pätzold, 2002). A typical scenario for the terrestrial mobile radio channel is shown in Figure 2. Due to the multipath propagation, the received signal consists of an infinite sum of attenuated, delayed, and phase-shifted replicas of the transmitted signal, each influencing each other. Depending on the phase of each partial wave, the superposition can be constructive or destructive. Apart from that, when transmitting digital signals, the form of the transmitted impulse can be distorted during transmission and often several individually distinguishable impulses occur at the receiver due to multipath propagation. This effect is called the *impulse dispersion*. The value of the impulse dispersion depends on the propagation delay differences and the amplitude relations of the partial waves. Multipath propagation in a frequency domain expresses itself in the non-ideal frequency response of the transfer function of the mobile radio channel. As a consequence, the channel distorts the frequency response characteristic of the transmitted signal. The distortions caused by multipath propagation are linear and have to be compensated for on the receiver side, for example, by an equalizer.

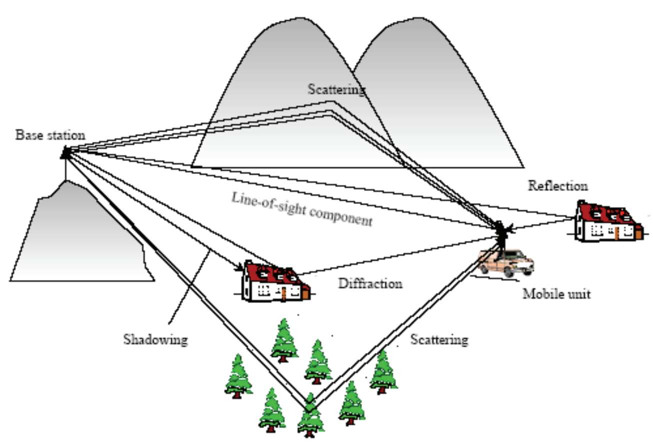


Fig. 2. Fading phenomena in a multipath wireless network (Pätzold, 2002).

Besides the multipath propagation, also the *Doppler effect* has a negative influence on the transmission characteristics of the mobile radio channel. Due to the movement of the mobile unit, the Doppler effect causes a frequency shift of each of the partial waves. Our analysis in this work considers the propagation environment in which a mobile-satellite system operates. The space between the transmitter and receiver is termed the channel. In a mobile satellite network, there are two types of channel to be considered: the mobile channel, between the mobile terminal and the satellite; and the fixed channel, between the fixed Earth station or gateway and the satellite. These two channels have very different characteristics, which need to be taken into account during the system design phase. The more critical of the two links is the mobile channel, since transmitter power, receiver gain and satellite visibility are restricted in comparison to the fixed-link.

By definition, the mobile terminal operates in a dynamic, often hostile environment in which propagation conditions are constantly changing. In a mobile's case, the local operational environment has a significant impact on the achievable quality of service (QoS). The different categories of mobile terminal, be it land, aeronautical or maritime, also each have their own distinctive channel characteristics that need to be considered. On the contrary, the fixed Earth station or gateway can be optimally located to guarantee visibility to the satellite at all times, reducing the effect of the local environment to a minimum. In this case, for frequencies above 10 GHz, natural phenomena, in particular rain, govern propagation impairments. Here, it is the local climatic variations that need to be taken into account. These very different environments translate into how the respective target link availabilities are specified for each channel. In the mobile-link, a service availability of 80–99% is usually targeted, whereas for the fixed-link, availabilities of 99.9–99.99% for the worst-month case can be specified.

*Mobile satellite systems* (Ibnkahla, 2005) are an essential part of the global communication infrastructure, providing a variety of services to several market segments, such as aeronautical, maritime, vehicular, and pedestrian. In particular, the two last cases are jointly referred to as the *land mobile satellite (LMS)* segment and constitute a very important field of application, development, and research, which has attracted the interest of numerous scientists in the last few decades. One fundamental characteristic of an LMS system is the necessity to be designed for integration with a terrestrial mobile network counterpart, in order to optimize the overall benefits from the point of view of the users and network operators. In essence, satellite and terrestrial mobile systems share the market segment along with many technical challenges and solutions, although they also have their own peculiar characteristics. A classic and central problem in any mobile communication system is that of modeling electromagnetic propagation characteristics. In LMS communications, as for terrestrial networks, multipath fading and shadowing are extremely important in determining the distribution of the received power level. In addition, it is common to also have a strong direct or specular component from the satellite to the user terminal, which is essential to close the link budget, and which modifies significantly the statistics with respect to terrestrial outdoor propagation In terms of *modeling the LMS propagation channel (*Lehner & Steingass, 2005), there are three basic alternatives: geometric analytic, statistical, and empirical. Generally speaking, the statistical modeling approach is less computationally intensive than a geometric analytic characterization, and is more phenomenological than an empirical regression model. The most remarkable advantage of statistical models is that they allow flexible and efficient performance predictions and system comparisons under different modulation, coding, and access schemes. For these reasons, in the first part of this chapter we focus our attention on a thorough review of statistical LMS propagation models, considering large- and small-scale fading, single-state and multistate models, first- and second-order characterization, and narrowband and wideband propagation.

### 2.4 Land mobile satellite channel

Both vehicular and pedestrian satellite radio communications are more commonly referred to as the Land Mobile Satellite (LMS) channel. LMS constitutes a very important field of application, development, and research, which has attracted the interest of numerous scientists in the last few decades (Ibnkahla, 2005). In the LMS channel, received signals are characterized by both coherent and incoherent components including direct signals, ground reflections, and other multipath components. The relative quality and intensity of each component varies dynamically in time (Mineweaver et al, 2001), based on various parameters. Shadowing of the satellite signal is caused by obstacles in the propagation path, such as buildings, bridges, and trees. Shadowed signals will suffer deep fading with substantial signal attenuation. The percentage of shadowed areas on the ground, as well as their geometric structure, strongly depend on the type of environment. For low satellite elevation the shadowed areas are larger than for high elevation. Especially for streets in urban and suburban areas, the percentage of signal shadowing also depends on the azimuth angle of the satellite (Lutz et al, 2000). Due to the movement of non-geostationary satellites, the geometric pattern of shadowed areas is changing with time. Similarly, the movement a mobile user translates the geometric pattern of shadowed areas into a time series of good and bad states. The mean duration of the good and bad state, respectively, depends on the type of environment, satellite elevation, and mobile user speed (Lutz et al, 2000). A popular and relatively robust two-state model for the description of the land-mobile satellite channel was introduced by (Lutz et al, 1991). The fading process is switched between Rician fading, representing unshadowed areas with high received signal power (good channel state) and Rayleigh/lognormal fading, representing areas with low received signal power (bad channel state) (Lutz, 1998). An important parameter of the model is the time-share of shadowing, $A$, representing the percentage of time when the channel is in the bad state, ranging from less that 1% on certain highways to 89% in some urban environments.

## 3. Joint source and channel estimation

The basic simulation involves video application encoding, channel simulation, video decoding and finally video quality analysis. Figure 3. below depicts the overall simulation system. For the SVC simulation, the base layer and enhancement layer are passed through separate channel simulators and are corrupted independently. For the AVC case, only channel is used as there is no enhancement layer. This model is used to simulate different channel conditions and a fixed set of iterations are used to collect statistical data. The following sections provide a detailed description of each functional block.

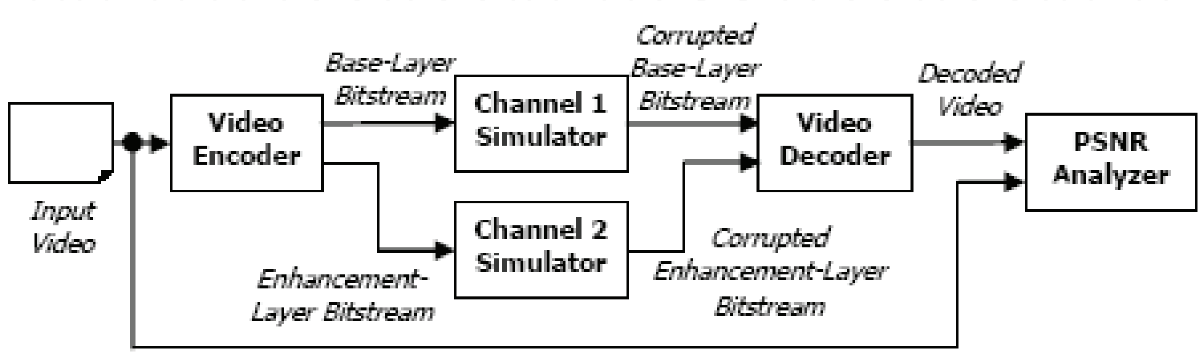


Fig. 3. Overall simulation system architecture (Kiang et al, 2008).

### 3.1 Video encoder

The 2-layer encoder system is illustrated in Figure 4. below:

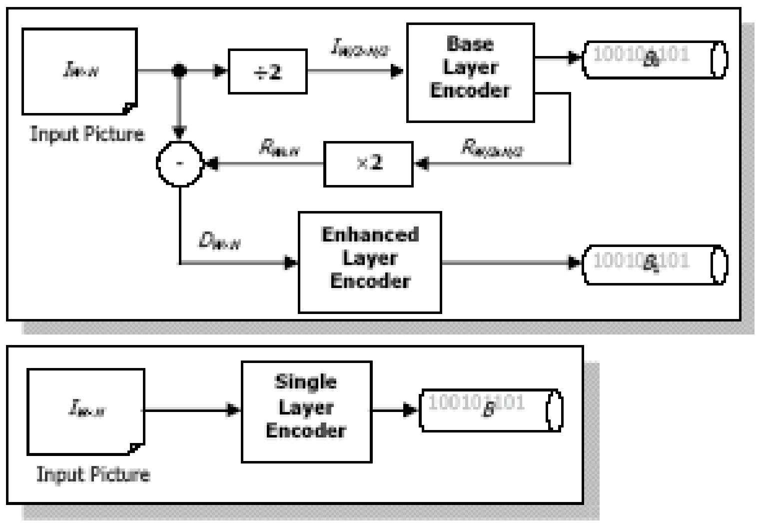


Fig. 4. Encoder architectures (top: 2-layer, bottom: single-layer) (Kiang et al, 2008).

Every input picture $IW×H$ , is decimated by 2 via a simple decimation filter. The resulting decimated picture $IW/2×H/2$ serves as an input to the Base Layer AVC encoder to obtain the base layer bit-stream, $B0$. The reconstructed picture from the base layer encoder ($RW/2×H/2$) is up-sampled by 2, and the resulting picture $RW×H$. is subtracted pixel-by-pixel from the input picture $IW×H$. The 'difference' picture ($DW×H$)is the input to the enhancement layer encoder which produces enhancement layer bit-stream, $B1$. $B0$ and $B1$ is output to their respective channel simulators. For the case of a single layer encoder, only $B0$ is output. However, it should be noted that as a reference, we ensure that the bit-rate of the single layer encoder, $R$, is similar in value of the total bit rates of the base-layer $R0$ and enhancement-layer $R1$. That is:

$$R \approx R_0 + R_i \tag{1}$$

### 3.2 Channel simulator

The error model in the channel simulator is based on a Gilbert-Elliot 2-state Markov model. Based on field measurements, the Gilbert-Elliot model has been shown to approximate the land mobile satellite channel quite well (Schodorf, 2003).

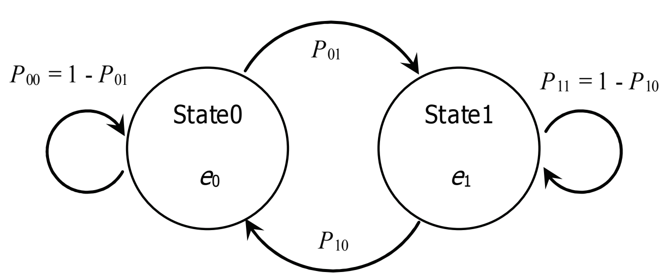


Fig. 5. Gilbert-Elliot channel model.

It assumes that a channel has a good and bad state, *S*0 and *S*1. Each state has a bit-error rate (BER), *e*0 and *e*1. The BERs in general depend on the frequency and coding scheme and on environmental conditions (e.g., number of paths between source and destination). The good state has a lower BER. The state transition probability *P*01 is the probability of the channel changing from *S*0 to *S*1. The four transition probabilities form the transition probability matrix:

$$\begin{bmatrix} P_{00} & P_{10} \\ P_{01} & P_{11} \end{bmatrix} = \begin{bmatrix} 1-P_{01} & P_{10} \\ P_{01} & 1-P_{10} \end{bmatrix} \tag{2}$$

Continually multiplying (2) will achieve a steady condition in which any 2-valued column vector, when premultiplied with the resulting matrix will achieve an invariant column vector; the value of this column vector denotes the long-term probability the *S*0 and *S*1 will occur respectively. This is the probability at which the states are likely to occur and is given by:

$$P_0 = \frac{P_{10}}{P_{01}+P_{10}} \quad ; \quad P_1 = \frac{P_{01}}{P_{01}+P_{10}} \tag{3}$$

This probability distribution { *P*0, *P*1 } is used to initialize the state at the beginning of each transmission packet. The transition probabilities (only two of which are independent) determine the mean duration and frequency of the error bursts. Thus the mean duration of periods of time spent in the bad state (i.e., the mean burst length) is given by (Carey, 1992):

$$D_b = \sum_{j=0}^{\infty} P_{11}^{j} = \frac{1}{1-P_{11}} = \frac{1}{P_{10}} \tag{4}$$

Similarly the mean duration of periods between bursts is (Carey, 1992):

$$D_g = \frac{1}{P_{01}} \tag{5}$$

Simulation of each packet is carried out independently. If an encoded video frame is smaller than the fixed packet size, the whole frame is transmitted within one packet. Else the frame is fragmented into fixed size packets (with the exception of the last packet) and transmitted independent of each other. Every bit within a packet is checked via a random number generated between 0 and 1.0. If the number is less than the BER value of the current state, the bit is deemed to be corrupted and the whole packet is discarded. A frame with one or more discarded fragment is also deemed to be lost and will not be decoded by the decoder. At every bit, the state is checked for transition based on the current transition probability. This description assumes transmission of data one bit at a time, so that the model's decision, in terms of state transition, occurs for each bit. In systems (e.g., using QAM) where a single transmitted symbol carries more than one bit, the decision occurs once per symbol (Carey, 1992). The following figure contains the flow chart of the process:

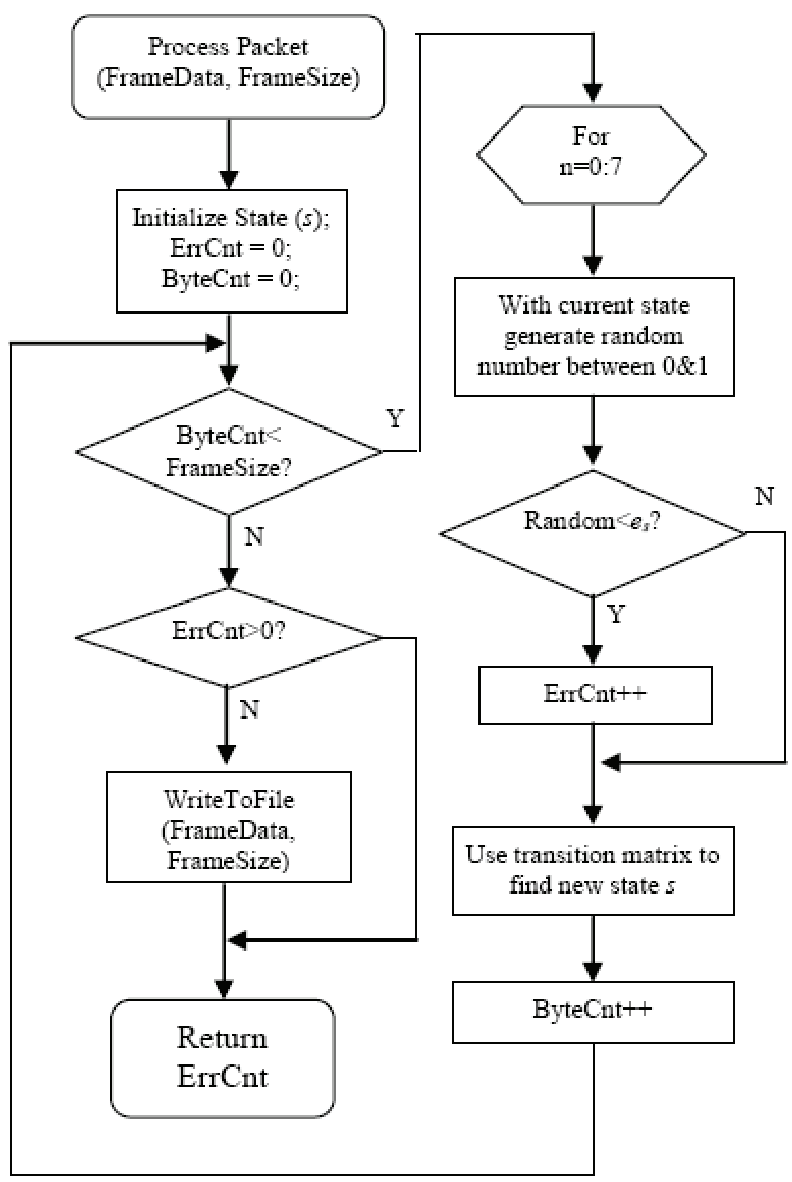


Fig. 6. Bit-based corruption and state-transition flow chart in channel simulator (Kiang et al, 2008).

### 3.3 Video decoder

The single-layer and 2-layer decoder architectures are shown in Figure 7. below:

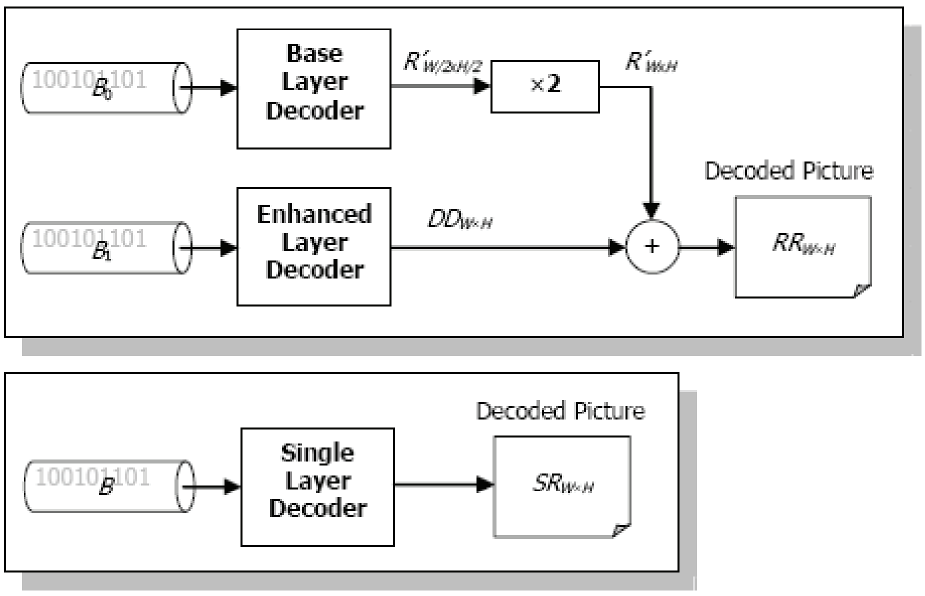


Fig. 7. Decoder architectures (top: 2-layer, bottom: single-layer) (Kiang et al, 2008).

It should be noted that in absence of errors in channel 1, $R'W\times H$ in Figure 4. and $R'W\times H$ in Figure 7. are identical. When channel 2 is corrupted, $RRW\times H = R'W\times H$. Hence, in the case of corruption in Channel 2, the 2-layer decoder can still output a relatively good quality video from channel 1. Similar case cannot be said of the single-layer system. Of course, the above claim is only true provided $B0$ in channel 1 is not corrupted. When the latter happens, the overall quality of the 2-layer system may be lower than that in the single layer system. However, chances of that happening are relatively small when $B0$ is more protected from errors than $B$. Furthermore, due to the fact that $R0 > R$, packet errors are much lower in $B0$ even if both are protected equally and the channel conditions are identical.

In the presence of packet errors, some decoded pictures may have been corrupted. Intra pictures (I) encoded separately whilst Predictive pictures (P) and bidirectional-predictive pictures (B) are predicted from previously decoded pictures and hence they bear dependencies with other pictures. Same can be said of the relationship between pictures from the base- and enhancement-layers. In spite of the numerous error concealment techniques available, this paper applies the simple method of repeating previously decoded picture.

In Figure 8., I$n$ pictures are Intra pictures and P$nm$ is the $m$th Predictive picture since $n$th Intra picture, which is reference from P$n,m$-1. E frames are enhancement picture based on the corresponding base-layer picture. The figure depicts which picture is displayed in the presence of packet losses.

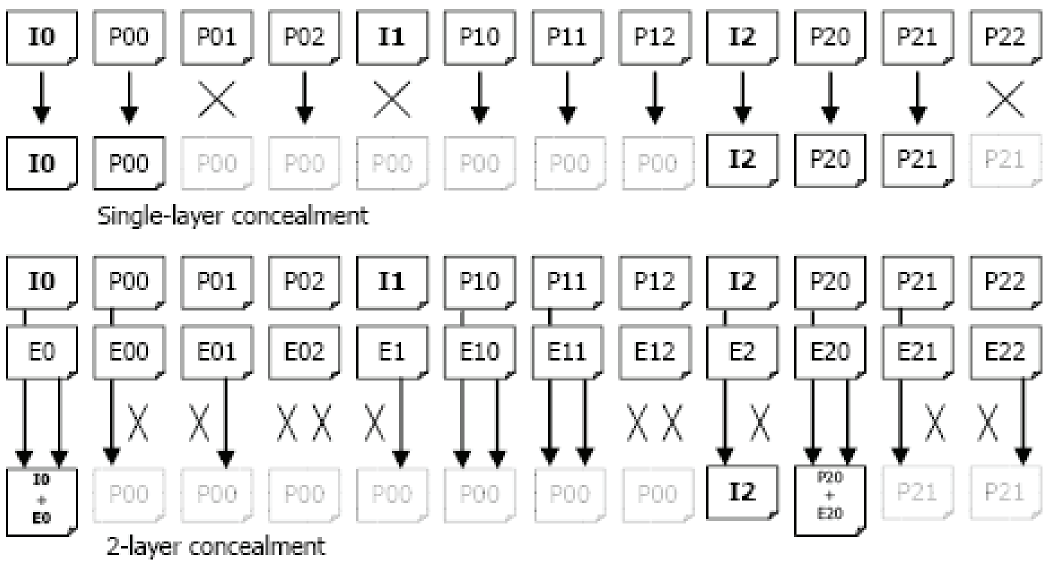


Fig. 8. Simple concealment based on "previous good picture" method (Kiang et al, 2008).

### 3.4 PSNR quality analyzer

We have decided to use the Peak Signal-to-Noise Ratio (PSNR) as a measurement of received video quality in the presence of losses. PSNR is traditionally used by video coding community to measure the fidelity of the compressed video with respect to its original input. Assuming a picture $I(x,y)$ of dimension $W \times H$ is compressed and the reconstructed picture after decompression is $R(x,y)$ . The fidelity measure of PSNR between $I$ and $R$ is given as:

$$PSNR(I|R) = 10 \cdot \log_{10}\left( \frac{255^2 \times W \times H}{\sum_{x=0}^{W-1}\sum_{y=0}^{H-1} \left(I(x,y) - R(x,y)\right)^2} \right) \tag{6}$$

In our case, the PSNR value used in (6) of the current reconstructed picture is reference with the original input picture used by the encoder. Since the encoded bit-streams are corrupted by error channel some frames may have been lost. This results in the loss of synchronization between the input and the reconstructed pictures. This problem is circumvented by tagging each encoded picture with a sequence number. This number is traced within the decoder and the PSNR analyzer. Lost pictures will be substituted with a previous good picture. This is a typical scenario found in practical systems.

### 3.5 Simulation conditions - assumptions

The primary objective of the simulation phase is to evaluate the performance of the video coding techniques (i.e., single layered and two-layered), in different satellite channel quality conditions. Two experiments have been carried out in connection to the aforementioned objective:

1) A 2 Mbps single-layered video coded stream is transmitted over a satellite link. Different channel quality conditions are simulated by configuring the input parameters of the Gilbert-Elliot model, namely *P01*, *P10*, *e0*, and *e1*. The coded video sequence is then injected with errors and the performance of the coding scheme is recorded in terms of the PSNR objective quality metric.
2) In this experiment, the two-state Markov channel model characterizing the shadowing process (i.e., switching between good and bad channel states) was extended to two separate channels (Lutz et al, 1998). More specifically, two layered video coded streams, namely the base and enhancement layers (each one at 1Mbps), are transmitted over two separate satellite links. Enhancement layers encode additional information that, using the base layer as a starting point, can be used to reconstruct higher quality, resolution, or temporal versions of the video during the decode process. Without the base layer, no video reconstruction is possible. Simulated channel combinations are divided into two categories: the first one considers both channels under the same conditions in terms of BER, and the second one applies more protection to the base layer (lower BER) and less protection to the enhancement layer (higher BER). In our simulations, the following conditions and assumptions are made:

- Simulations have been carried out using the channel simulator fed with the well known Foreman video sequence.
- Packets affected by at least one (1) bit in error are discarded, and the decoder loses the entire payload they contain.
- Each video packet is small enough to fit into one communication frame. Each video coded stream has 250 4CIF (704x576) picture frames at 25 fps. Hence duration of 10 seconds is assumed.

The first frame of the coded video sequence always passes uncorrupted through the channel simulator – this is a valid assumption as in the long run, channels unable to transmit any frames at all are practically useless.

- For each channel, the video stream is broadcasted to 500 stationary users. This means that all users of a specific broadcasting session experience the same channel conditions.

Channels are emulated by configuring the input parameters of the Gilbert-Elliot channel simulator. In the simulated scenarios the value of P10 is set to a specific value, and the value of P01 varies so as to represent a range of channel conditions with different values of the average error probability (Masala et al, 2004) (BER in the "bad" state). The idea is to distinguish between various classes of users that lie within the satellite spot-beam coverage area.

## 4. Simulation results

To illustrate the multilayer video, we down-sample the 4CIF (704x576) sequence by 1/2 and encode the base layer with AVC. We then employ the spatial scalability to produce the enhancement layer. The simulation results concerning the objective quality vs. channel BER of both single- and two-layered coding schemes have been plotted in Figure 9. The three lines correspond to different channel quality conditions for: 1) single-layered video coding, 2) two-layered video coding with equal BER conditions for base and enhancement layers, and 3) two layered video coding with more protection applied to the base and less to the enhancement layers (BERbase< BERenhancement).

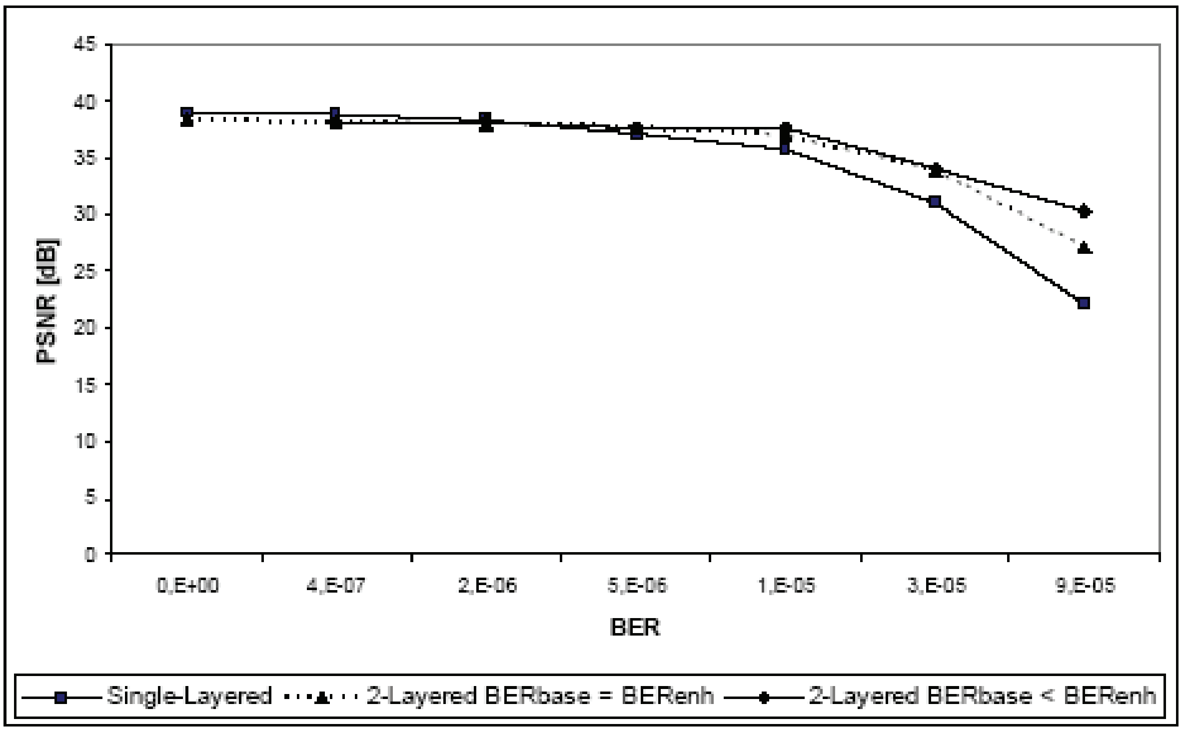


Fig. 9. PSNR (left axis) versus BER for transmission of a H.264 coded sequence. Three cases are reported: the first one refers to singe layered H.264 AVC encoded video, the second refers to a two-layered scalable video based on H.264 assuming that both channels fall under the same conditions in terms of BER. The third refers to a two-layered scalable where more protection is applied to the base layer and less protection to the enhancement layer, resulting in lower BER in the base layer.

The quality measure PSNR reflected in the plot is the 'aggregate' value, or the average value of all the simulations performed with the same channel conditions. The first point in the graph is derived from ideal (i.e., lossless) channel conditions (i.e., *eb* = 0), where the PSNR indicates maximum quality of the decoded video stream. As we move towards the right side of the graph, channel conditions tend to deteriorate, and the same happens to the quality of the received video stream. For low BERs and up to a specific transition point (situated around a value of 5x10-6), the single-layered scheme shows better performance than the two-layered case. This is mainly attributed to the fact that more overhead is introduced to the scalable scheme for error resilience purposes. The cross-point in general depends on the channel coding scheme, packetization, and also error concealment techniques. The superiority of the two-layered case is evident as harsher channel conditions start to dominate. The maximum observed difference in objective quality is about 5 dB (22.18 dB for single-layer and 27.15 dB for two-layered at 9x10-5). In the case of the unequal error protection, where the base layer has higher protection (e.g., channel block coding) than the enhancement layer, the quality of the decoded sequence is further enhanced as we move towards more unfavorable channel conditions. This shows that under severly bad channel conditions, higher protection should be given to the base layer in order for the 2-layer decoder to consistently produce a relatively good-quality video.

## 5. Conclusion and future directions

In this chapter, we tackled the multimedia (video) application over a fading satellite-terrestrial network by applying the scalable video coding over a land mobile system assuming a 2-state Markov model. By splitting video into 2-layers (Base/Enhanced) and transmitting them in 2 separate satellite channels, the overall quality measured in terms of aggregate PSNR value is improved over the single-layered scenario. Results from our 2-state Markov channel model support this claim. Moreover, applying higher protection to base layer at the expense of the enhancement layer further improves the aggregate viewer experience. However, the current system has the following room for improvements and will be covered in future work:

- More layers of SVC (including temporal scalability should be used.
- Research into optimal packetization and layering strategies should be looked into.
- Error concealment strategies in connection with the video coding scheme shall be investigated.
- Current channel model should be augmented by a good physical and MAC layer simulator.
- Interleaving and channel error protection schemes (e.g., parity bits, FEC codes) should be examined towards and end-to-end simulation.
- Simulations should include mobile users.

## 6. Acknowledgments

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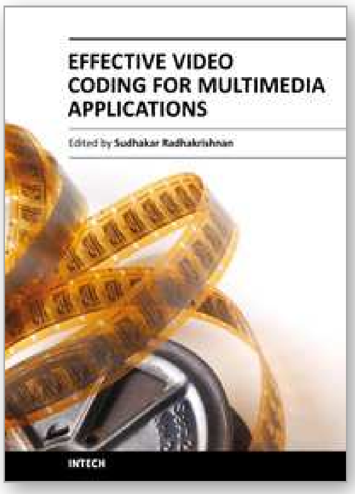


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Information has become one of the most valuable assets in the modern era. Within the last 5-10 years, the demand for multimedia applications has increased enormously. Like many other recent developments, the materialization of image and video encoding is due to the contribution from major areas like good network access, good amount of fast processors e.t.c. Many standardization procedures were carrried out for the development of image and video coding. The advancement of computer storage technology continues at a rapid pace as a means of reducing storage requirements of an image and video as most situation warrants. Thus, the science of digital video compression/coding has emerged. This storage capacity seems to be more impressive when it is realized that the intent is to deliver very high quality video to the end user with as few visible artifacts as possible. Current methods of video compression such as Moving Pictures Experts Group (MPEG) standard provide good performance in terms of retaining video quality while reducing the storage requirements. Many books are available for video coding fundamentals.This book is the research outcome of various Researchers and Professors who have contributed a might in this field. This book suits researchers doing their research in the area of video coding.The understanding of fundamentals of video coding is essential for the reader before reading this book. The book revolves around three different challenges namely (i) Coding strategies (coding efficiency and computational complexity), (ii) Video compression and (iii) Error resilience. The complete efficient video system depends upon source coding, proper inter and intra frame coding, emerging newer transform, quantization techniques and proper error concealment.The book gives the solution of all the challenges and is available in different sections.