Protocol Architecture for MPEG-2 Video on High Speed Network

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ABSTRACT. *Real time communication services related to the encoded video transmission over the high-speed network like ATM need a new integrated protocol architecture to meet the required quality. In fact, data cells are exposed to delays and losses, which affect the video signal quality. So, we have to perform the adequate processing in order to keep the quality of service on an acceptable level. In this article we propose the design of a new adaptation of MPEG-2 video in ATM Networks in order to improve the video visual quality. Our approach, based on a new MPEG-2 protocol architecture, tries to overcome the difficulty imposed by traditional random cell discarding due to the bursty aspect of the traffic and the variable bit rate transmission, nature of encoded video. We demonstrate the importance of including a dynamic bandwidth reallocation and appropriate interleaving technique for a bursty encoded video traffic being carried in ATM Network by comparing the degradation quality of MPEG-2 sequence with and without our proposed techniques. The results show that the proposed interface is effective in solving the cell loss problem and thus enhances the QoS for MPEG-2 video transmission in ATM Networks.*

Keywords: MPEG-2 (Moving Picture Expert Group) ATM (Asynchronous Transfer Mode), AAL5+ (Advanced ATM Adaptation Layer), QoS (Quality of Service), VBR (Variable Bit Rate).

1. Introduction

Recently, the traffic conveyed over high-speed network has been growing dramatically requiring more developments in both coding techniques and network transmission capacity. In addition, developments in the area of video coding and compression techniques are enabling the deployment of computer-based video communication systems. One of the main challenges remains in the design and the deployment of protocol architectures able to cope with stringent video communication requirements. In order to guarantee its effectiveness, the video communication system requires high-speed networks with intelligent control resources and error recovery mechanisms. These requirements should be able to properly manage the system resources and cope with system errors [1, 2]. The design of an integrated interface between the video application and the ATM based network, including different control mechanisms, has to take into account characteristics of various system elements from the application down to the transmission mechanism.

Carrying the encoded video applications over ATM networks introduces several issues that must be considered in order to ensure a high end-to-end quality [3]. This includes the choice of adaptation layer packets for the network side, the traffic management mechanism and the cell mapping algorithm definition, and the interleaving and recovery schemes implementation. All these operations will be performed with the aim of ensuring a low video quality regress for an end-to-end communication via the ATM network. Therefore, the objective of ATM adaptation layer is not only to map the information flow generated by the application into the payload of cells, but also to provide the suitable service required by the specific application above the ATM layer [4, 5].

In order to enhance the robustness of the video transmission process, we propose the use of a hierarchical video encoding scheme system operation.

[Insert figure 1 about here]

Figure 1 depicts the proposed protocol architecture comprising a set of protocol mechanisms tailored to enhance the robustness of the video delivery application. These mechanisms have been designed bearing in mind the stringent requirements and characteristics of video encoded according to MPEG-2 Standard specifications. The proposed protocol architecture has to provide the appropriate solution to deal with the situation where the video application would transmit cells faster than the negotiated bandwidth. This situation can happen when the application is delivering a compressed MPEG-2 video sequence [6, 7], whereas the receiver is running according to the negotiated bit rate.

In this respect, a new adaptation encapsulation scheme is proposed to map the video sequence on Transport Stream (TS) level over the ATM Adaptation layer (AAL5). The new AAL5 sub-layer is named advanced AAL5+ and includes real-time extensions to the Variable Bit Rate (VBR) service related to the MPEG-2 standard [8, 9,10].

This paper is organized as follows. In section 2, we briefly review technical challenges for video quality enhancement. Section 3 is dedicated to our proposed techniques related to the network integrated protocol architecture. A particular focus is made on the new cell mapping algorithm encapsulation process which links the MPEG-2 video sequence to the ATM network at the AAL5+ access point. Section 4 is devoted to the description of the integrated protocol architecture configuration described in VHDL language. In section 5, we present the performance evaluation of our implemented techniques and we discuss the results. Finally, we conclude and present directions for future works in section 6.

2. Technical challenges for video quality enhancement

To address the transmission problem of compressed VBR video over lossy networks, several techniques and solutions have to be proposed to minimize the video quality regress. This section looks at the technical challenges for the design of integrated protocols supporting an encoded MPEG-2 video applications.

2.1. Quality in carrying encoded video over packet network

Supporting encoded video services over the ATM network is considered as a promising trend in the telecommunication activity. Particularly, the design of the integrated protocol supporting video services is an important issue, which should be studied. Today, it becomes important to understand how packet networks can deliver QoS related to encoded MPEG-2 video traffic. To implement the needed bandwidth with regard to the timing guarantees constraints, packet based network must first identify the video streams and then make the appropriate traffic control to the video packet. In addition, the encoded video packet has to be encapsulated onto the network format in order to reduce the transfer delay and packet loss.

2.2. Protection by packet interleaving

Interleaving MPEG-2 video frames consists in structuring a set of AAL5+ PDU as a square block embedding cells performed an each column. This procedure involves a rearrangement of the original frames to ensure that previously consecutive frames are separated at transmission and rearranged back in their original sequence at the receiver. At the destination system, this can be seen as a virtual N*M cell matrix. Inside the interleaving technique, a single lost packet will only result in multiple short gaps in different streams of the received data. In no interleaved data stream, a packet lost allows a long gap into consecutive frames compared to precedent one. By spreading the source bits over time, it becomes possible to make use of error control coding which protects the source data from corruption by the channel or by a bursty period of traffic [11, 12]. Unfortunately, frames interleaving has the disadvantage of increasing the processing delay.

3. Network integrated protocol architecture issues: specific service definition

In this section, we discuss technical solutions that are implemented in our integrated protocol related to encoded video applications.

3.1. Proactive network control policies: traffic management algorithm

The problem is actually to deliver the grade of service that has been promised for all connections, even when one of them does not respect the negotiated flow. This operation requires some kind of resources management strategy, since congestion will be by far the greatest factor in data loss and so in video quality regress. For our purpose, we focus on a shared buffer approach for all encoded MPEG-2 frames. There are a number of parameters and functions that need to be considered. In our case, the traffic parameters that have been proposed for resource management are:

- mean bit rate = 1/ *Xave* where *Xave* is the average packet inter-arrival time of the current connection. The transmission mean rate is measured in number of cells per second along a fixed period of time T.
- peak bit rate = 1/ *Xmin* where *Xmin* represents the minimum packet inter-arrival time of the current connection,
- statistical control parameter α.: it represents the tolerated exceed parameter expressed in percent for each connection.

We note that the statistical parameter α is fixed according to memory resources and traffic nature. This parameter gives the rate of exceeded bandwidth tolerated by the shared buffer without discarding any cell according to the occupancy on the buffer space. For our implementation, we have fixed α to 0.2 in order to tolerate an overlapping between different ATM connections related to MPEG-2 frames equal to 20% for each connection.

The control is performed at a cell level and includes cell loss and cell errors to better take into account the delay and loss sensitive characteristics of MPEG-2 encoded applications. Given an accepted ATM connection carrying an encoded video application, according to the allocated bandwidth, we can easily compute the theoretical arrival time by estimating the inter-arrival cell time according to the negotiated bandwidth. This value is then compared to the real time. As depicted in the Figure 2, when the buffer queue length exceeds an upper threshold $(T_r < T_{th}/1+\alpha)$ where T_{th} is defined with the negotiated bit rate, an early congestion is detected.

[Insert figure 2 about here]

Consequently, the traffic management algorithm tries to reallocate an additional buffer space according to the statistical parameter (α) and to the occupancy on the buffer space.

3.2. Dynamic memory management

In order to increase the design performance, the memory management implements unique buffer space to process efficiently received and transmitted MPEG-2 packets and to avoid intermediate packet copy. Data from each connection is stored in per-connection data queues, so that each connection can be served separately. On the reception of MPEG-2 encoded sequence, the corresponding AAL5+ segment is stored in the shared buffer location in order to be analyzed. The received sequence could be whether queued in the appropriate connection queue or discarded depending on the error control message. The Figure 3 presents the relation between the connection table (CT), the linked list table (LL) and the shared buffer area.

[Insert figure 3 about here]

The connection table (CT) consists on a list of descriptors containing the traffic parameters related to MPEG-2 connection such as a peak bit rate and a mean bit rate. On the transmit side, an MPEG-2 packets are retrieved from the shared buffer and transmitted across the ATM network.

3.3. A new MPEG-2 video stream encapsulation strategy

Our entry point for the ATM network is located at the ATM Adaptation Layer-5 which is intended for the point-to-point transport of variable bit rate. In order to map the MPEG-2 encoded video application onto the ATM network, an appropriate adaptation protocol is necessary. The ATM Forum has recommended to carry the MPEG-2 Transport Stream packets per AAL5-PDU with *NULL* Convergence Sub-Layer. In this case, the QoS of bursty interactive VOD applications are not guaranteed and this point of view is still an open issue. Few works have proposed to define a new sub-layer and have focused their study on the end-to-end delay bound [13, 14, 15]. Some of them have addressed an analysis approach based on a frame level priority data partition [16] and have studied the synchronization problems [17].

[Insert figure 4 about here]

Our proposed strategy consist in a new encapsulating the MPEG-2 standard into the AAL sub layer because the traditional AAL5 is inadequate for the transmission of variable bit rate video and requires extended futures. The aim of the specific CS sub-layer is to allow a first step MPEG data extraction process before the traffic management and the network processing. Uncompressed video frames are individually encoded according to the MPEG-2 standard in a packet elementary stream (PES). This means that an access unit may start at any point within a PES packet. Instead of encapsulating MPEG-2 video data at the macro-block level, we propose that each PES is segmented into a number of 188-byte fixed length transport stream (TS) packet. At the AAL service access point (SAP), the transport layer passes the TS packets to the SSCS using message mode service internal function. As illustrated in Figure 4, SSCS groups every three TS packets and adds header and trailer information. The header is composed of a 4-bit sequence number (SN), and a 4-bit SNP (Sequence Number Protection). The trailer consists of a 3-byte forward error correction (FEC).

3.*4. The interleaving technique*

As previously explained, the MPEG-2 video source is placed into the interleaver by sequentially increasing the column number for each successive bit, and filling the rows. The interleaved source data is then read out column-wise and transmitted to the network. At the receiver of the interface component, the de-interleaver stores the received data by increasing sequentially the column number of each successive bit, and then clocks out the data column-wise, one word (column) at time. In our implementation, we have choosen ***N*** and ***M*** both equal to 48 which represents the size of the ATM cell. We note that there is an inherent delay associated with an interleaver since the received message block can't be fully decoded until all of the ***NxM*** bits arrive at the receiver and are de-interleaved.

4. Integrated protocol architecture configuration

Our proposed architecture consists in a set of interface and control components. This architecture is capable to transmit data and control information in two directions between distant communication entities related to encoded video applications. This architecture includes three different parts as depicted in Figure5 :

- The network interface unit,
- The microprocessor interface unit,
- The main part of the integrated protocol architecture.

[Insert figure 5 about here]

4.1. The network interface unit

This unit is responsible for the reception of data from the network interface to be stored into the receive buffer. It's also responsible for data transmission from the interface to the network. The implemented version of this unit represents a simple version of the UTOPIA interface [18].

4.2. The microprocessor interface unit

This interface arbiters dialogue between the Microprocessor and our circuit. It allows the microprocessor to control and to configure the proposed architecture. It can also handle the video sequence at the reception side and display it in order to analyze the video visual quality.

4.3. The integrated protocol architecture

Our proposed architecture is based on the dynamic management of the buffers in both reception and emission. This configuration brought us to define the suitable data structure and to choose the appropriate technique. Based on the circular linked list, this technique presents many advantages. Besides the optimization of resources in memory, it reduces the data access time. Memory resources are based on:

- The linked list memory: it is used for the dynamic management of the data memory which serves as a support of mapping of the linked list and performs a buffer management for emission and reception. In addition, the data structure considered for our implementation consists in a descriptor containing two fields of address forming consequently the basic element of the linked list. By taking into account the size of the buffer and the nature of the descriptor, one can easily deduce the size of this memory. Indeed this size is fixed to 32 Kwords of 16 bits.
- The receive/transmit memory (DPRAM): the receive and the transmit is used for absorbing the latencies incurred by different data rates between cell interface. It is an external component with a size of 256 Kwords of 16 bits divided into slots of 24 words receiving the 48 bytes of data of every cell.

Each of the connection table and the scheduling technique is mapped onto two SRAM memories operating in parallel way. We remark that the flow control technique is implemented according to the algorithm described in section 3.1.

5. Performance evaluation

The experimental setup for the test evaluation consists of a network simulation model composed of a local ATM network with an ATM switch and an advanced AAL5 sub-layer. At the AAL service access point (SAP), the transport layer is fixed as the TS packet for the new SSCS sub-layer defined for our purpose. We consider a new cell-discarding scheme based on a statistical parameter for our traffic management algorithm, which provides better performance for carrying video stream over loossy environment. The proposed scheme is associated with dynamic and statistic buffer allocations with an extended AAL5 to form MPEG-2 to ATM interface. During the bursty traffic, we propose to drop a cell after making a first correction in the allocated buffer space using the statistical parameter α. This approach is applied to minimize discarding cell within the capacity of the network architecture. Our proactive control approach is performed gradually to avoid congestion situation. In addition, we use the interleaving technique with the Forward Error Correction mechanism in order to make better the visual video quality.

We have also described the entire architecture with the VHDL language. All memories are considered as external components. Our VHDL is described at the RTL level with V-SYSTEM tools. The integrated protocol architecture is situated at the interface level between the MPEG-2 standard at the TS level and the ATM network at the AAL5 layer.

5.1. Interleaving effects on the video quality

We consider three sequences of image, the original one (a), the sequence with degradation because of data losses (b) and finally the third sequence (c) that represents the second sequence but followed by the interleaving technique. This evaluation shows the interest of the interleaving technique to improve the video quality in the worst case where some cells are discarded because of the limitation in the bandwidth of the network.

[Insert figure 6 about here]

5.2. Evaluation of the integrated protocol with flow control algorithm

To evaluate our technique, we have considered two encoded video test sequences as an MPEG-2 video applications where three performances evaluations are investigated. These applications are trying to send a large amount of data from the sender to the receiver. Each one of them has different bandwidth requirements. The first one consists of an ATM connection that doesn't exceed its negotiated bit rate. In the second situation, this connection presents a bit rate exceeding the negotiated value but this variation is limited in time (limited period of burst). The third case consists of a long bursty traffic emulation where bandwidth can’t be shared according to the negotiated parameters and ATM cells must be dropped after a first correction in order to avoid the network congestion.

For the first approach of comparison, we have used the PSNR metric, more appropriate for the encoded video applications than the SNR one. The PSNR is defined in the usual way as:

$$PSNR = 10\log_{10}\frac{255^2}{MSE}$$

Where MSE is the mean square error given by :

$$MSE = \sum_{m}\sum_{n}\left[x(m,n) - x^*(m,n)\right]^2$$

The PSNR is measured in decibels (dB) where *x(m,n)* represents the original encoded video sequence pixels and *x*(m,n)* the rebuilt one. The PSNR measure is also not ideal, but is in common use. However, it remains a good measure for comparing restoration results for the same sequence, but between-sequence comparisons of PSNR are meaningless. One sequence with 30 dB PSNR may look much better than another sequence with 20 dB PSNR. Where the burst is limited in time, it is perfectly absorbed and resolved by our proposed traffic control algorithm. So we obtain a good quality of the encoded video sequence through the ATM network. The critical case, which is analyzed carefully, concerns the situation related to a long bursty traffic emulation for the encoded video sequences of test (Vanseg and Engine).

Measurements of PSNR obtained for the worst case with a long burst period are shown in the following table. It can be seen that there are significant improvements of the encoded video quality. We notice that the PSNR decreases when the loss probability increases which means that there is an increasing of the video quality degradation.

Test Encoded video sequence	***Loss Probability***	***PSNR(dB) without Interleaving***	***PSNR (db) with Interleaving***
Engine benchmark	10^{-4}	28.2	19,4
	10^{-3}	19.8	16.8
	10^{-2}	10	9.5
Vanseg benchmark	10^{-4}	30	27.4
	10^{-3}	20	19.7
	10^{-2}	12.1	10

Table1: Comparison of PSNR values (in dB)

In order to enhance the video quality of our sequence, we have implemented a 48x48 bytes interleaver in the integrated protocol architecture. The table 1 shows that the video quality can be improved using the interleaver. For example, we note that the Engine Benchmark, using the interleaver, with a PSNR equal to 28,8dB might look much better than the same one without the interleaver where the PSNR is equal to 19,4 dB. Finally, the above table shows different situations related to the two encoded video sequences after applying the appropriate cell-dropping scheme. We remark that the loss probability depends on the statistical parameter α. In fact, where α increases the cell loss probability decreases. However, the α

value is limited with the resource management capacity. In the case of our implementation, the best results are obtained with α=15% of the post allocated bit rate, in terms of ATM cells, to each connection supporting the encoded video sequence.

6. Conclusion

In this paper, we have proposed new integrated protocol architecture to support video application over the ATM network in order to improve the video visual quality. This architecture includes three parts which are: (i) the encapsulation of the video stream and the associated protocol, (ii) the dynamic bandwidth allocation according to statistical criteria to more (iii) and the interleaving technique combined with correction code. According to the presented results, we conclude that the quality of service in terms of visual quality and cell loss probability provided by encoded video application has significantly improved. By adjusting automatically its allocated bandwidth to the network capacity, the proposed traffic control mechanism reduces discarding cells in the worst case when a bursty flow occurs for a long period. In addition, the interleaving techniques improve the visual video quality. The PSNR metric, which is the means to quantify the visual quality applied to two encoded video test sequences shows that the visual quality degradation is reduced when we use our proposed techniques.

Our future work will involve studying improved version of the traffic control algorithm taking into account the nature of the ATM cell (data cell of MPEG-2 and control cell) with a dynamic priority assignment to better suit with MPEG video transmission over ATM network. Further version should be based on a content-based priority scheme.

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Figure Captions

Figure 1. Protocol Architecture

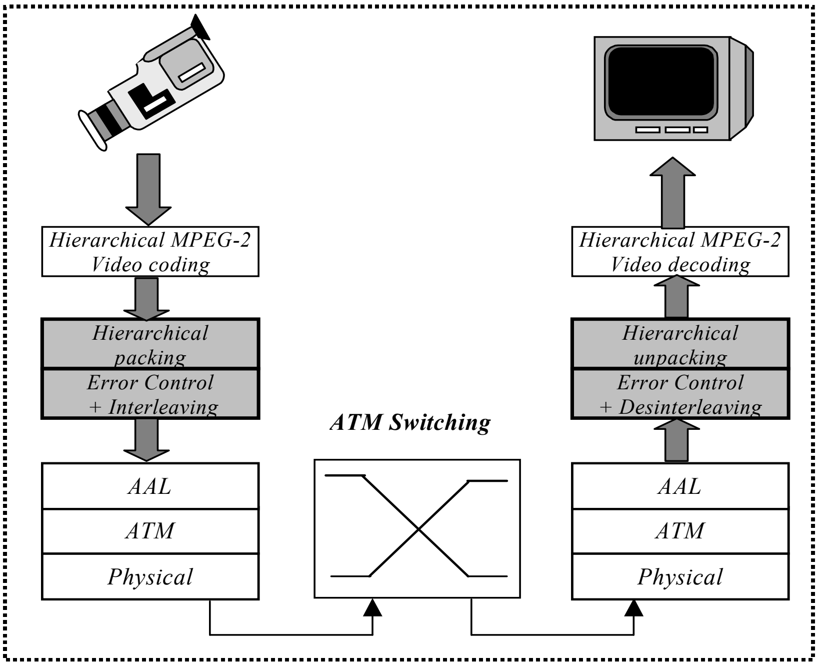
Figure 2. The traffic management algorithm

Figure 3. Linked List Principle for Buffer Management

Figure 4. A New mapping of MPEG-2 video stream on AAL5

Figure 5. Synoptic scheme of the proposed architecture

Figure 6. Video quality improvement with the interleaving technique.



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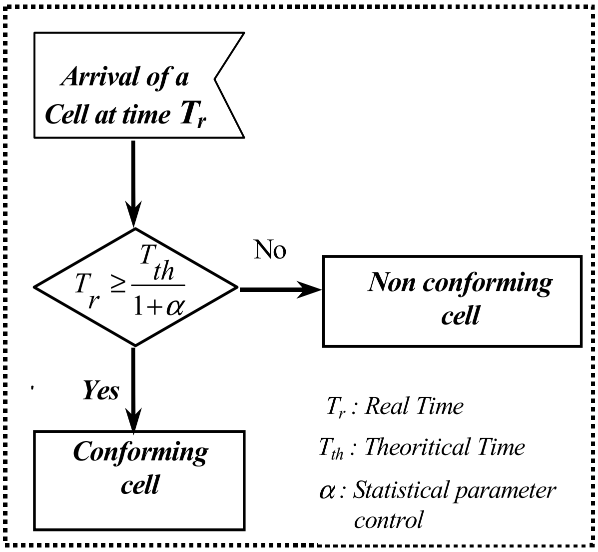
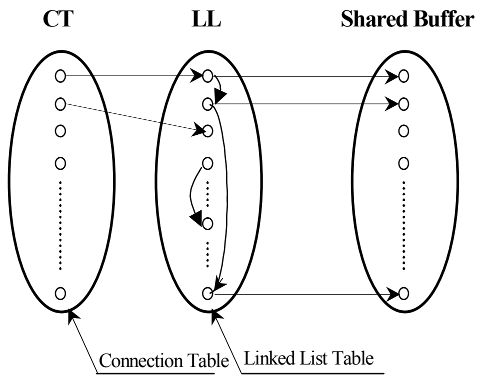


Figure 2. The traffic management algorithm



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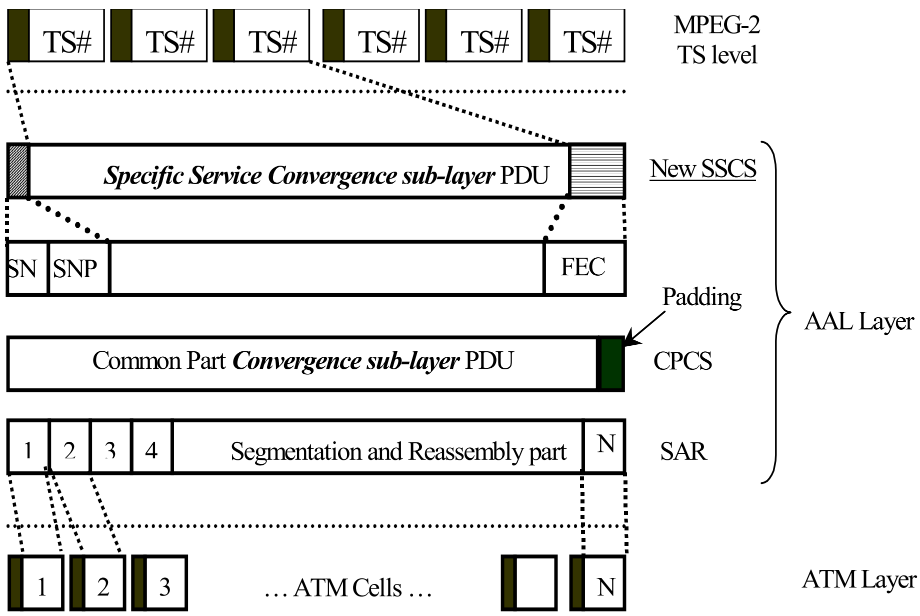

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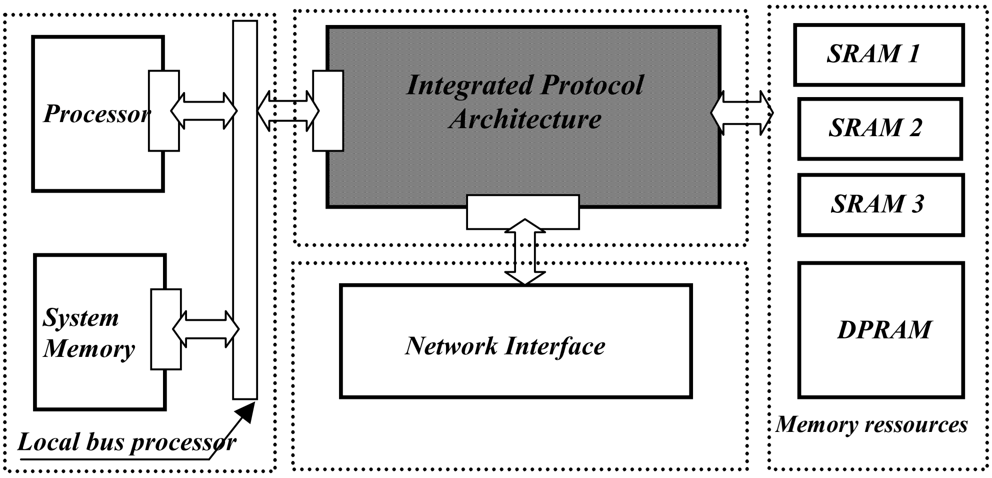

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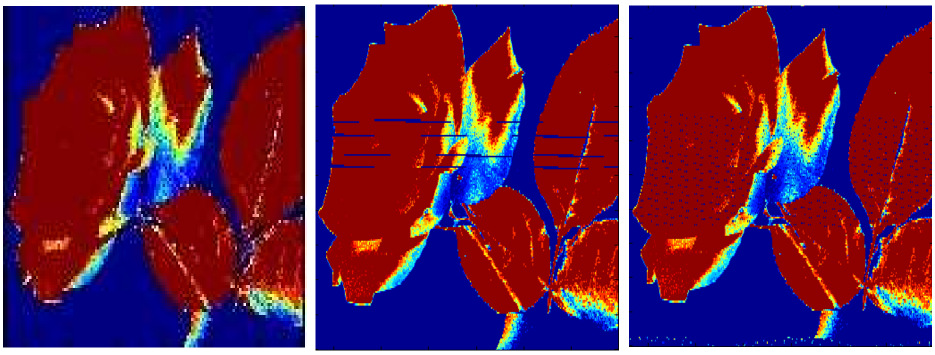

(a) (b) (c)

Figure 6. Video quality improvement with the interleaving technique