High performance Architecture of Integrated Protocols for Encoded Video Application

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Abstract —*Despite the evolution of high speed communication network to accommodate an increasingly number of applications with diverse service requirements, there still exist a number of barriers related to the deployment of the encoded video over the ATM network. In fact, additional works have to be devoted to improve protocol architecture and to guarantee the QoS. In this paper, we first analyze the main parameters affecting the visual quality of real video pictures. Then, we define specific services to be implemented at the network interface level. We also discuss the proposed integrated protocols architecture for real time application such as video coding illustrating the function to support the challenges of managing real time services over high speed network. In fact, data cells are exposed to delays and losses, which affect the quality of the video signal. Therefore, 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 an interface between the MPEG-2 standard and the ATM network in order to improve the video visual quality. Our approach 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 compressed video. The performance evaluation shows the effectiveness of the proposed interface architecture with the set of mechanisms in improving the robustness of the video delivery system.*

Key Words — *ATM: Asynchronous Transfer mode, MPEG Motion Picture Expert Group, QoS: Quality of Service, PSNR: Peak Signal over Noise Rate (dB).*

1.INTRODUCTION

Interactive Multimedia applications like video conferencing over high-speed networks require more and more resources in term of bit rate and bandwidth allocation to satisfy the negotiated QoS. In particular, encoded video requires more developments in both video processing and network capacity. The recent developments in the area of video coding and compression technique are enabling the deployment of computer-based video communication systems. However, one of the main challenges to overcome remains the design and deployment of protocol architectures able to cope with the stringent requirements of video communication. In order to prove effectiveness, a video communication system will require the use not only of high-speed networks, but also resource control and error recovery mechanisms 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 regrouping different control mechanisms has to take into account characteristics of the various system elements from the application down to the transmission level.

To investigate integrated protocol architecture, we have to explain characteristics and functions of the different protocol layered in order to understand the design of effective integrated protocol architectures. The Asynchronous Transfer Mode (ATM) is widely used as the transfer technique for such applications [3]. It consists of a connection-oriented network, where data is carried in a short fixed-length packet of 53 octets, called cell. The application data is segmented into fixed size cells of 53 bytes and transferred across the network. It is important to study the cell level performance of these applications and its impact on the application level performance. The cell level performance measures that affect a multimedia application performance include cell errors, cell losses and cell delays. Real time interactive multimedia applications like video conferencing are very sensitive to cell delays. In fact, end-to-end packet delay consists of segmentation and reassembly delay and cell transfer delay that includes the queuing delay within the network. This delay depends on the network traffic and the architecture capacity. The ATM technology was initially designed to permit simultaneous carrying of various services all supported equivalently without any knowledge of its cell content, i.e. without discriminating some cells against others. Among all the services, variable bit rate in multimedia environment and especially digital video communication should contribute as an important fraction of the overall traffic. Due to the nature of the compression technique, the MPEG-2 sequence is inherently variable asynchronous, stochastic and non-stationary. In order to provide an acceptable quality of service, the network interface must be able to ensure minimal congestion conditions and to optimize the bandwidth allocation to meet the real time constraints.

Moreover, the residential audiovisual applications such as TV broadcasting, and Video On Demand (VOD) will work on the basis of high-speed network technique. These video applications will widely use the motion picture expert group (MPEG) compression standards to save network resources. These video compression algorithms are mostly based on efficient DCT (Discrete Cosines Transform) and block-oriented techniques DCT, H.261, ISO JPEG (Joint Picture Expert Group) and ISO MPEG-1 [4]. Unfortunately, they all have been designed to support specific services and applications, and they can't inter-operate without restriction over a based ATM network. For instance, the H.323 is an International Telecommunications Union (ITU) standard that provides specification for computers, equipment, and services for multimedia communication over networks that do not provide a guaranteed quality of service. Moreover, the H.261 scheme was initially designed to operate over specific digital networks like Narrow-band ISDN with a low rate about nx64Kbit/s to carry on teleconferencing applications where motion is naturally more limited. In the same way, JPEG algorithms only emphasizes still images and do not apply any inter-coding algorithms to take into account temporal redundancy of moving picture sequences. For this application, processing delay was not a major concern for H.323, H.261 and JPEG-1 [5].

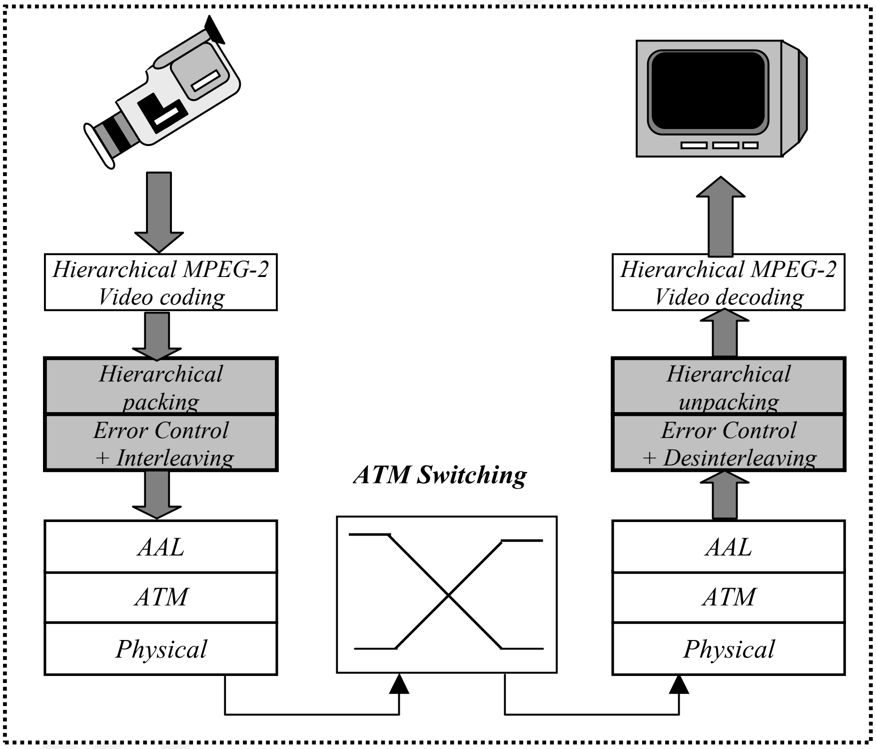

Figure 1. Protocol Architecture

Until now, few works have considered to allow a new higher quality video coding over the B-ISDN (Broad-band Integrated Services Digital Network) network related to the application, which is a delay sensitive [6, 7]. Consequently, an important effort should be devoted to inter-operate a heterogeneous standard like MPEG-2 [8] and ATM network to benefit from this environment for real time applications with agreed quality of service [9]. In fact, the transport of compressed pictures related to video applications over ATM network introduces several issues that should be considered in order to ensure a high end-to-end quality. This includes the choice of adaptation layer packets for the network side, the traffic management mechanism, the cell mapping algorithm, interleaving and recovery schemes. All these operations will be performed to ensure low video quality degradation for end-to-end communication via the ATM network. Therefore, the objective of ATM adaptation layer is not only to map the flow of information generated by the application into the payload of cells, but also to provide above the ATM layer, a service suitable to the need of the specific applications [10].

In order to enhance the robustness of the video transmission process, we propose the use of a hierarchical video encoding scheme system operation. Figure 1 depicts the proposed protocol architecture comprising a set of protocol mechanisms tailored to improve the robustness of the video delivery application.

Our proposed work consists in the development of an adaptation layer between the MPEG-2 and the ATM standards including hierarchical packing/unpacking, error control.

Basic Concepts

The major feature of an integrated protocol for encoded video applications is to allow communication between video coding application across the network with the guarantee of the visual quality of the sequence. Several aspects have to be considered for the interface function: Protocol and format adaptation, traffic regulation, the different service requirements and functions to be supported. Furthermore, the proposed interface should be transparent for the interconnected networks and encoded video applications according to the MPEG-2 standard [11].

These mechanisms have been designed bearing in mind the stringent requirements and characteristics of encoded video according to the specifications of the MPEG-2 standard. The proposed protocol architecture has to provide the appropriate solution to deal with the situation where the video application is aiming to transmit cells faster than the negotiated bandwidth. This situation can easily happen when the application is delivering a compressed video sequence according to the MPEG-2 standard, whereas the receiver is running according to the negotiated bit rate. Two aspects have to be taken into account where we design the integrated protocol architecture for a to real time application such as:

- The aspect of the information representation: it consists of the choice of the format and the compression algorithms, the error control strategy, and the interleaving techniques.
    - The compression technique allows the minimization of data redundancy in order to increase the amount of the exchanged data. This technique has got many advantages. First it increases the storage capacity and reduces the access time to the encoded data and so it reduces the needed bandwidth to be allocated for the data transfer.
    - The error control and the interleaving techniques: because of real time character without retransmission of the data, the standard error recovery technique with retransmission (Backward Error Control Strategy) is not suitable and will be replaced by other techniques based on the error recovery technique (Forward Error Control Strategy) such as the Reed-Solomon code.
- The second aspect concerns the network architecture: in this case, we have to choose the appropriate protocol to satisfy the need of the real applications. These protocols are located in both the connection admission (ATM, MPLS: Multi Protocol Label Switching, with CR-LDP: Constraint-based Routed-Label Distribution Protocol and so on and so forth) and the resources reservation in term of memory and bandwidth allocation in order to satisfy the required quality of service for specific video applications.

    In this respect, a new adaptation encapsulation scheme is proposed to map the video sequence on Transport Stream (TS) level over the Adaptation ATM layer (AAL5). The new AAL5 sub-layer is named advanced AAL5+ and includes real-time extensions to support the VBR (Variable Bit Rate) service related to compressed video sequence according to MPEG-2 standard. In addition, the integration of the protocol architecture includes the error recovery technique and the interleaving/des-interleaving technique to cope with the required visual video quality.

This paper is organized as follows: We briefly analyze the main factors that can affect a picture compressed video quality. The second section is devoted to the network integrated protocol architecture issues including the appropriate techniques. These techniques should allow the maximum flexibility degree for bit rate variation specific services defined at the network interface level. We particularly focus on the new cell mapping algorithm encapsulation process with links the MPEG-2 video sequence and the ATM network the AAL5+ access point. The third section deals with the performance evaluation, forward error control and traffic management. Finally, we conclude.

2. ANALYSIS OF PARAMETERS AFFECTING PICTURE QUALITY

The objective of this section is to understand various parameters affecting the video encoded application performances. From a network standpoint, applications and underlying generated traffic are commonly grouped into service classes according to their traffic characteristics and sensitivity. The first class is related to error and loss-sensitive flows like traditional LAN (Local Area Network) traffics. The second-class concerns delay sensitive applications, which are usually referred to real-time services (voice, non compressed video). With real-time encoded video applications, this artificial border between reliability and stringent temporal requirements are not respected. Indeed, video applications, using compression capabilities are submitted to both error-free and real-time transmission constraints. These applications are very sensitive to cell losses and errors. Using the pipeline technique in the proposed architecture, the segmentation and the reassembly delays as well as the queuing delays don't affect the performance of the encoded video applications.

2.1. Data losses due to cell errors

Random bit errors along the communication path or within the network nodes due to the electrical or physical problems can highly damage the quality of the decoded pictures. At the cell level when such bit errors occur in the header, the cell is either miss-delivered when errors and address modifications are not detected, or discarded by the physical layer or by the receiver in case of uncorrectable detected errors. In both cases the whole cell should be considered as lost and the consequences can be serious for the MPEG decoding process. If the error occurs in the payload field of the cell, the damage is obviously limited to the degradation of the data part of the MPEG packet. If this part belongs to the MPEG Transport Stream (TS) packet header the entire packet may be lost and the impact on the displayed pictures can be also very serious. Fortunately, the probability of such losses is normally extremely low. For instance, in high-speed networks based on optical fibers, it is not exceeding 10^{-13}. Nevertheless, even for these transient error events, new mechanisms related to error detection and correction scheme are required at the MPEG level to ensure low video quality degradation.

2.2. Data losses due to burstiness

During the congestion state, the network performance is being regressed in term of fairness and efficiency. When the traffic is bursty and has variable bit rate requirements with high peak rates such as real time variable bit rate (rt-VBR) and available bit rate traffic classes (ABR), congestion may occur even on high speed network like ATM. These heavy loads related to these traffics are mostly due to inadequate network resources allocation and multiplexing process. Exceeding network capacity leads to the cell discarding by either congested nodes (through UPC: User Protocol Control) or the destination terminal if the delay exceeds a fixed threshold. In the later case, the MPEG packet arrives too late for playback on the terminal. Preventive actions must be applied to minimize the QoS degradation, inside the network. These actions operate through intelligent and tolerant discarding, at terminal nodes, through fast recovery schemes at both ATM and MPEG transport levels.

3. NETWORK INTEGRATED PROTOCOL ARCHITECTURE ISSUES: SPECIFIC SERVICE DEFINITION

In the previous section, we described the services that are possible with the high-speed integrated architecture to support encoded video application and the technical challenges to be solved to make this architecture a success. In this section, we discuss the technical solutions that are implemented in our integrated protocol related to the encoded video applications.

3.1. Proactive network control policies: traffic management algorithm

Traffic management consists of methods to share network resources and balance different traffic loads by satisfying the required QoS parameters and traffic definitions in the worst case since the application does not satisfy the negotiated agreement.

The difficulty is 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 resource management strategy, since congestion leads to the greatest factor in data loss and so in video quality regress. Our purpose is to 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 during 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 percentage for each connection.

We note that the statistical parameter α is fixed according to memory resources and the 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 overlap 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 the delay into account 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 T_{th} by estimating the inter-arrival cell time according to the negotiated bandwidth. This value is then compared to the real time T_r. As depicted in the Figure 2, when the buffer queue length of a connection exceeds an upper threshold $(T_r < T_{th}/1+\alpha)$ where T_{th} is defined by the negotiated bit rate, an early congestion is detected. Therefore, a connection witch arrives in burst with an overflow of rate cannot use other than the space which is reserved to it only a percentage of the available space not used by the other connection. In the case when the new cell arrives after $T_{th}/1+\alpha$, then we will say that the cell is in keeping. Moreover, if the arrival occurs before this moment, the cell will be dropped.

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 [12].

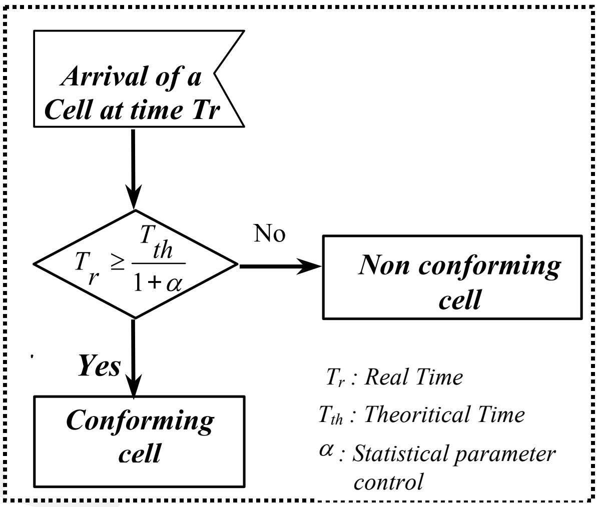

Figure 2. The traffic management algorithm

3.2. Dynamic Memory Management

In order to increase the design performance, and to avoid intermediate packet copy the memory management implements a unique buffer space to process efficiently received and transmitted MPEG-2 packets. 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 either 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.

The connection table (CT) consists of 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, MPEG-2 packets are retrieved from the shared buffer and transmitted across the ATM network.

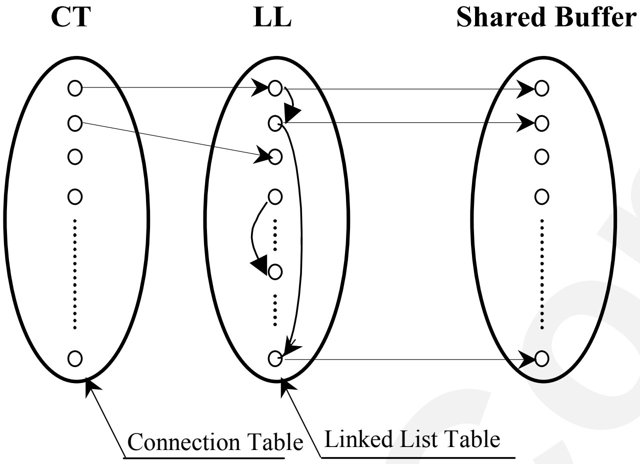

Figure 3. Linked List Principle for Buffer Management

3.3. A new MPEG-2 video stream encapsulation strategy

The MPEG-2 encoded video application is mapped on to the ATM network at the ATM adaptation layer (AAL5) level, which is intended for a point-to-point traffic. We note that the AAL5 with *NULL* Convergence Sub-Layer cannot support any QoS. However, the extended version of AAL5 named AAL5+ with a new SSCS (Service Specific Convergence Sub-layer) could support the QoS according to the defined protocol. Few works have been proposed to define a new sub-layer and have focused their study on the end-to-end delay bound [13]. Some of them have addressed an analysis approach based on a frame level priority data partition [14] and have studied the synchronization problems [6].

In this respect we propose a new encapsulation strategy for an MPEG-2 video over the ATM network because the traditional AAL5 is inadequate for the transmission of variable bit rate video and requires extended features. The aim of the specific CS sub-layer is to allow a first step MPEG data extraction process before the traffic management and ATM 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 the segmentation of each PES into a number of 188-byte fixed length transport stream (TS) packets. At the AAL service access point (SAP), the transport layer passes the TS packets to the SSCS (Service Specific Convergence Sub-layer) using message mode service internal function. As illustrated in the figure 4, SSCS joins up every three TS packets and adds header and trailer information. The header is composed of a 4-bit sequence number (SN), which represent the number of the TS packet of the video sequence. A 4-bit SNP (Sequence Number Protection) represent an error correction code CRC-3 and a bit of parity. The trailer consists of a 3-byte forward error correction (FEC). The FEC scheme uses a Reed-Solomon (RS) code [15], which enables the correction of up to 2-loss bytes at each block of 564 bytes (e.g. 3x188). This encapsulation strategy operates in conjunction with a traffic control according to the proposed algorithm presented in section 3.1.

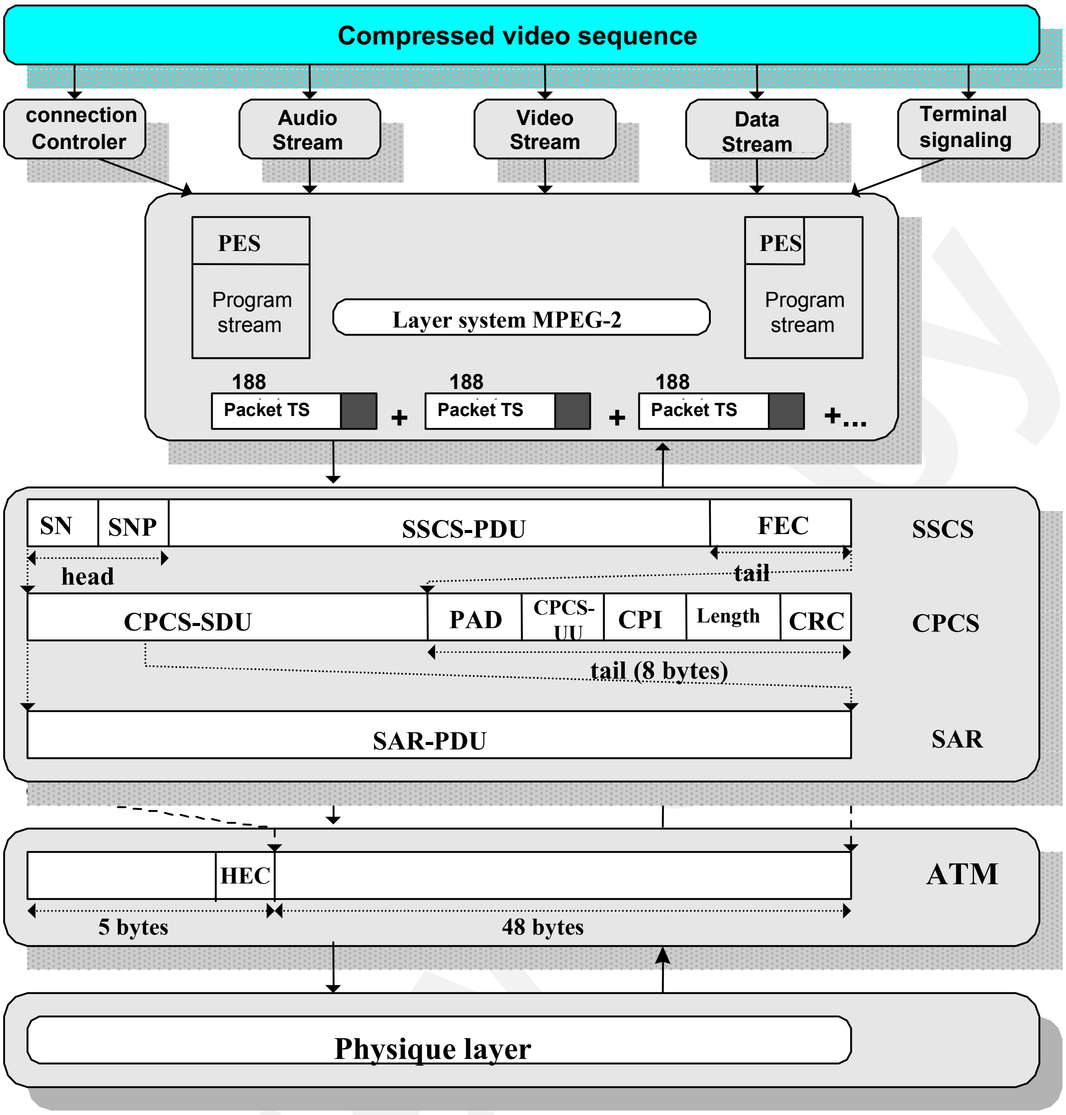

Figure 4. A new mapping of MPEG-2 video stream on AAL5+

3.4. Reactive Policy: The interleaving technique

As we have previously explained, the 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 des-interleaver stores the received data by sequentially increasing 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 chosen 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 the interleaver since the received message block can't be fully decoded until all of the nxm bits arrive at the receiver and are des-interleaved.

4. INTEGRATED PROTOCOL ARCHITECTURE

4.1. Target Model

Many architecture models have been developed according to application domain. Real-time system needs temporal behavior, when data base system considers the optimal organization of data. Consequently, many styles can be adopted in the design process and be classified into three broad categories [16].

- *Data flow model*: The system is defined by a set of activities associated to data. This approach is effective for certain application domains as a digital signal processing (DSP), where temporary variables are used to hold intermediate computation results.
- *Control-dominated model*: In this category, a control part is used to hold commands or control signals. This model is the most appropriate for reactive real-time system in which temporal behavior is very important. This approach has attracted more and more attention because of its wide use in control, communication, and embedded systems.
- *Hybrid model*: The designer can use this model to integrate many characteristics of cited models. This approach allows the designer to represent many aspects related to complex systems.

In our integrated protocol architecture, target architecture is composed of concurrent data flows which are ordered by the system control. This control part can be decomposed in a set of autonomous modules. Architecture is performed with two major parts: a sequencer and a data path. The control part is the most important in the proposed architecture including control signals and intelligent algorithms to satisfy the real-time constraints related to video conferencing for example.

4.2. Architectural configuration

Our proposed architecture of the integrated protocols related to MPEG-2 and ATM standards consists of a set of interface and control components. This architecture is capable of transmitting data and control information in two directions between distant communication entities related to video applications. This architecture includes three different parts as depicted in Figure 5:

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

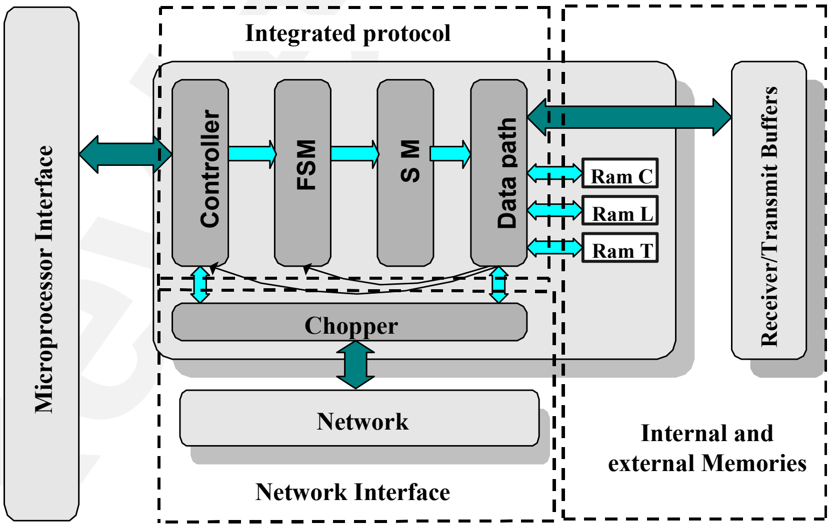

Figure. 5. Synoptic scheme of the proposed architecture

4.2.1 Network Interface unit (Chopper)

This unit is responsible for the reception of data from the network interface to be stored into 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 [17].

4.2.2 Microprocessor interface unit (microprocessor interface)

This interface arbiters dialogue between the Microprocessor and our circuit. It allows the microprocessor to control and to configure the proposed architecture.

4.2.3 The integrated protocol unit

This module consists of the memory controller including:

- The dynamic bandwidth allocation
- Scheduling technique
- Flow control mechanism.

Our proposed architecture is based on the dynamic management of the buffers in both reception and emission. This brought us to define the suitable data structure and to choose the appropriate technique.

The dynamic bandwidth allocation is based on the circular linked list technique. This technique has got many advantages. Besides the optimization of resources in memory, it reduces the access time to data. Furthermore, the use of the linked list technique avoid the multiple copies of ATM cells and consequently optimizes the transfer delays as well as the space used for buffering. Memory resources are based on :

- The linked list memory (RAM L): it is used for the dynamic management of the memory of data. This memory serves as a mapping support of the linked list and performs a buffer management for emission and reception. In addition, the data structure considered for our implementation consists of a descriptor containing two fields of address forming, in fact, the basic element of the linked list. Taking into account the size of the buffer and the nature of the descriptor, we can easily deduct the size of this memory. Indeed, this size is fixed to 32 Kwords of 16 bits.
- The receive/transmit memory (DPRAM: Double Port RAM): the receive and the transmit are used to absorb 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.
- The memory of connection (RAM C): This memory is dedicated for global connection parameters such as Xmin, Xmax. In addition it buffers the address location for every connection in the shared buffer. When the connection changes the negotiated parameters, this memory is updated. In our case, we consider up to 512 connections, so it is a 2 Kwords memory.
- In addition, scheduling technique consists of the mapping of the axis on a special memory called memory of time (RAM T). This memory performs the scheduling of the reception and the emission of cells according to the indication of time axis. It stores in a 2 bytes field the VCI (Virtual Connection Identifier) and the nature of the successful cell. RAM T is of size 256 words

The integrated protocol unit is composed of the following components:

The main state machine (controller)

To administer these four memories, a controller is designed to assure the dynamic memory management of data, as well as the control and the adaptation processing. The controller performs the following operations:

At the reception side, the controller determines the nature of the cell and activates the adequate procedure such as:

- Linked list initialization;
- Data cell buffering according to the conformities with the negotiated parameters;
- Connection parameters update to which belongs this cell.

At the emission side, the controller activates the procedure of emission of the current cell.

The secondary state machine

This machine is formed of two blocks:

- The Finite State Machine (FSM): it receives the primitive of the system (type procedure of emission of a cell, procedure of reception of a cell, etc.) and it defines the primitive execution order.
- The sequencer (SM): it allows the generation of all the necessary commands or control signal for the data path operation.

The data path

This module contains essentially a register interface and dedicated and standard operators (counters, addition, etc…) to perform all the operations required by the entire architecture. It's also responsible for the generation of the memories reading and writing signals.

5. ARCHITECTURE SYNTHESIS

Our interface architecture is designed to operate with an MPEG-2 video sequence in real time manner. However, in the validation process, we have operated with two approaches: Off-line and real-time processing. At the beginning, we need to distinguish between **off-line processing** and **real-time processin**g. In off-line processing, the entire input signals reside in the validation environment at the same time. For example, the medical imaging, such as computed tomography, represents an off-line processing case study. The data set is acquired while the patient is inside the machine, but the image reconstruction may be delayed until a later time. The key point is that all the information is simultaneously available to the processing program. This is common in scientific research and engineering, but not in consumer products. In real-time processing, the output signal is produced at the same time that the input signal is being acquired. For example, this is needed in telephone communication, hearing aids, and radar. These applications must have the information immediately available, although it can be delayed by a short amount of time. For instance, the speaker or listener cannot detect a 10millisecond delay in a telephone call. Likewise, it makes no difference if a radar signal is delayed by a few seconds before being displayed to the operator. Real-time applications input a sample, perform the algorithm, and output a sample, over-and-over. Alternatively, they may input a signal group

Design Results

We summarize in table 1 the design effort and the involved resources in terms of complexity when generating the micro-programmed architecture Fig 6. Through the analysis and verification process, we have used Modelsim simulator. The first validation of the proposed architecture is achieved with RTL description. In the advanced validation of the process, the RTL description is mapped onto the target architecture and synthesized using XILINXS library for the Virtex 2000 family in order to download the description in the test board. The frequency of FPGA processing is 20 Mhz. The primary results of the synthesis are reprinted in table 1, which represents the optimized number of configurable logic block (CLBs) for each component of block diagram presented below. The same table shows the corresponding of Flip_Flop.

Component	Number of CLBs	Number of Flip-Flops
Controller	65	22
FSM	79	40
SM	104	148
Data path	379	662
Chopper	90	130

Table 1: synthesis results with FPGA advantage simulator

6. PERFORMANCE EVALUATION

The experimental setup for the test evaluation consists of a VHDL description of the entire proposed architecture. This description is performed according to the predefined scheme of figure 5. we have used models for all external memory. 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. At the AAL service access point (SAP), the transport layer is the TS packet for the new SSCS sub-layer defined for our purpose. We propose a variant of the selected cell discard for our traffic management algorithm, which provides better performance for carrying video stream over loose 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.

6.1. Qualitative evaluation of the integrated protocol with flow control algorithm

To evaluate our technique, we have considered two encoded video test sequences where three performances evaluations are investigated. The experiment consists of these applications all trying to send a large amount of data from the sender to the receiver. Each application 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.

To perform the validation of our proposed architecture, a real test sequence has been attached to the protocol architecture. We have developped a test layer environment in VHDL in order to interface the real time sequence with our architecture and to analyze the sequence at the output level where the reconstructed sequence has to be compared with the original one. During the estimation, we have simulated various possible cases.

The first case consists in emulating an overtaking of overflow rate. This overflow is perfectly absorbed and resolved by the algorithm that we proposed. In this case there is a good agreement between the sequences of image. The Figure 6 shows a good agreement between the sequence of image. The algorithm guarantees a good visual quality before and after the network cross.

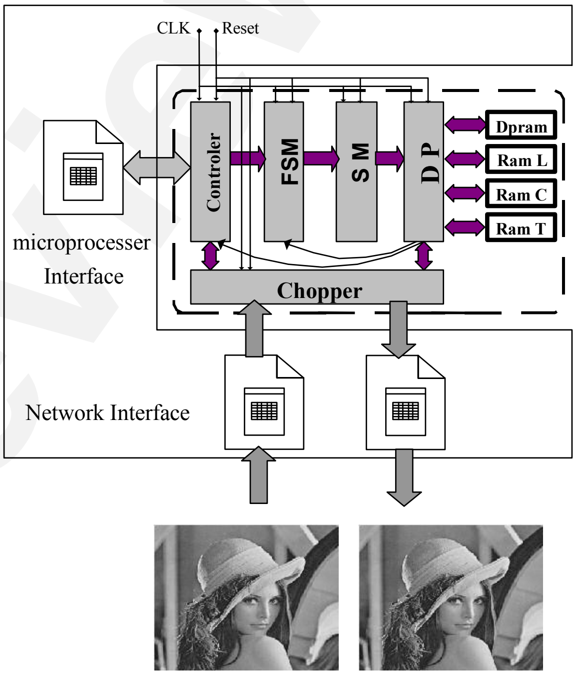

Figure.6. Sequence of picture not sucked before and after passing through the network

The other case consists in pushing the overtaking of the rate to see the limits of the algorithm. In that case, resources in terms of memory cannot tolerate the entire overflow. The control flow algorithm discards all additional cells belonging to this connection. As it is depicted in Figure 7, the sequence of image is partially jerky. It means that a part of the data of the sequence of image is missing person during the passage through the network. The effect of the ATM cell losses is clear in the sequence visual quality. This example shows the effect of ATM cell losses in the visual quality of pictures.

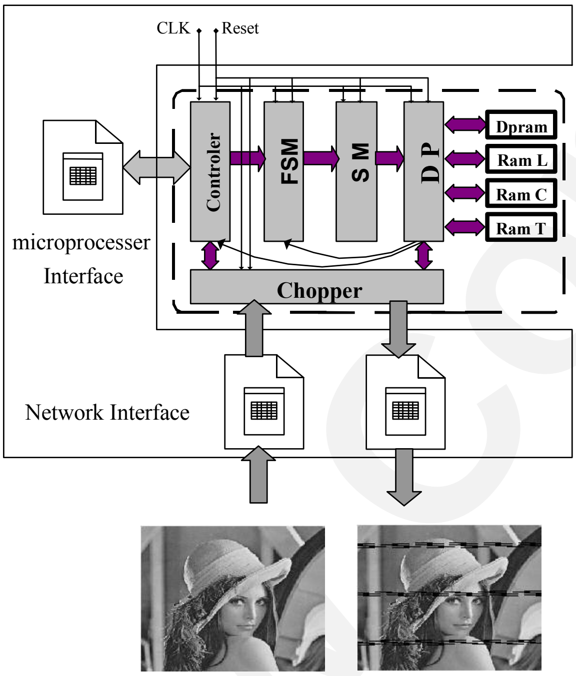

Figure.7. Sequence of picture sucked after passing through the network

6.2. Quantitative evaluation of the integrated protocol with flow control algorithm

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 usually defined as;

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

Where MSE is the mean square error given by:

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

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. Between-sequence comparisons of PSNR are meaningless however. 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 has been carefully analyzed, concerns the situation related to along bursty traffic emulation for the encoded video sequences of test.

Measurements of PSNR obtained for the worst case with a long burst period can prove 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. We have studied the variation of the visual quality which is represented with (1/PSNR) according to the bandwidth overflow. Three values of statistical parameters have been studied (p1 = 10%, p2 = 20% and p3 = 30%, p represents α). Finally, the figure 8 shows different situations related to the encoded video sequences after applying the appropriate cell-dropping scheme. We remark that the loss probability depends on the statistical parameter α. In fact, when 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=20% of the post allocated bit rate, in terms of ATM cells, to each connection supporting the encoded video sequence.

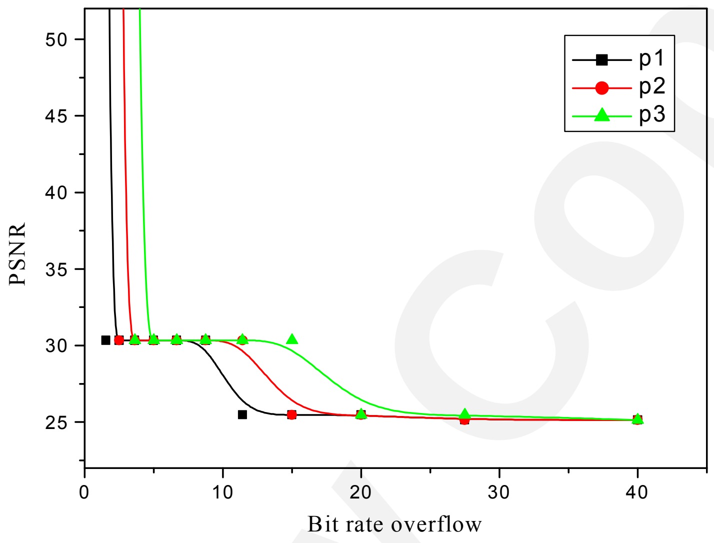

Figure 8. The visual quality over the statistical parameter p of the bandwidth adjustment

7. CONCLUSION

In this paper, we have proposed new integrated protocol architecture to support video application over the ATM network in order to improve the visual video quality. This architecture includes two parts, which are: (i) the encapsulation of the video stream and the associated protocol, (ii) the dynamic bandwidth allocation according to statistical criteria to fit the available resources. 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 is significantly improved. By automatically adjusting 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. The PSNR metric, which is the means to quantify the visual quality, applying the test sequences shows that visual quality degradation is reduced when we use our proposed techniques. Our proposed architecture is decomposed into several modules. Each module is simulated and checked separately. The entire architecture is verified using real sequences of pictures as input data.

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