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Expert group for ATM video coding

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Title : COMPENDIUM ATM; Coding procedures
Purpose : Discussion, Proposal

This document is a working document, with the following objectives:

- To define unambiguous terminology
- To identify first coding models.
- To study compatibility with the H.261

Futhermore the quality of service and bit rates versus quality are tackled.
In order to compare possible simulation results and to distribute the work efficiently, guidelines for simulations are included.
For the sake of completeness and cross fertilization of simulation and hardware studies the status of the hardware experiments is also included in this document.

COMPENDIUM
ATM; Coding procedures

COST 211-bis Simulation Subgroup
COST 211-bis Hardware Subgroup

November 5, 1990

Contents

Chapter 1

Prose description

Audio visual telecommunication will expand rapidly in the 1990's. Especially the visual part of the telecommunication system is of interest for the COST 211 ter project.

The efficient usage of bandwidth is important to enable the transport of the variety of information. Not only the transport of high speed data but also of video, audio, and text makes bandwidth reduction still necessary but also the growth of the traffic on the network. The asynchronous transfer mode (ATM) is the target transfer mode for the future public broadband ISDN (B-ISDN also IBCN). It is expected to provide an efficient utilization of the network resources by integrating a variety of communication services.

It becomes clear that for the evolution of in COST 211 ter's objectives, various expertise need to be incorporated to achieve a common stance on the coding procedures. Also it was agreed that the project would strive after a compatibility with the already reached coding standards H.261.
Research for telecommunication image coding must therefore be carried out in close cooperation with the groups dealing with the definitions of the new networks.

The objectives of COST is not to define standards COST need to have an appropriate channel to forward the proposals or information. As for the H.261 work ETSI NA3 (Audio visual services) was this standardization body. Also other research and development groups can be identified among which RACE 1018 HIVITS, RACE 1041 FUNCODE and ESPRIT MIAS. Some of the projects are designing systems for constant bit rates (CBR). Variable bitrates (VBR) are considered in HIVITS and FUNCODE. Combining the strength of the projects to make progress in the definition and the design of systems for ATM is desirable.

For ATM at the moment ETSI NA5 can be identified as standardization group which is involved in broadband network. In CCITT, SG XVIII is the global counterpart for ETSI NA5.

Due to various jargon, COST 211 felt it was necessary to compose a document which contains the basic terminology. This compendium is divided in eight chapters and is meant as a working document. The reader is kindly asked to check the date of the document. The information in the document is agreed upon in COST 211 and is also stable information from other groups (for network aspects ETSI NA5 and CCITT SG XVIII).

Chapter 2 entitled "Terminology", contains the basic terminology used in this compendium and utilize as much as possible the terminology used in the ETSI and the CCITT groups. Most important terms following from impacts of ATM on motion video coding and from new coding strategies will be explained. The chapter has been subdivided into two main chapters describing the terminology of the network part and the video coding part.

Coding models; first candidates, chapter 3 deals with different coding models for ATM networks. First some restrictions and characteristics on ATM networks are given after which the basics of one-layered and two-layered coding is given. The section other methods contains at the moment coding techniques which are under development for other than only the conversational services (MPEG).

Chapter 4, the H.261 capabilities for ATM, focusses on the implications of ATM on the H.261 recommendation. The mandate for ATM clarified that the ATM coding procedure should have at least the H.261 capabilities. This means that intercommunication is mandatory. The studies on this topic need to be carried out in the light that modifications to the H.261 are implemented in such a way that there exist an fallback mode to the H.261 standard. The decoder should be able to decode a H.261 bitstream. This chapter related to the previous chapter and gives guidelines which are described in chapter 7 simulations.

Chapter 5, Impact of QOS and Preventive policing on coding performance, intend to give an overview of the parameters defining the quality of service among which parameters like information loss, delay, throughput. Also the influence of policing is tackled.

Chapter 6, Bitrate versus quality, deals with the impact on the picture quality of different coding schemes and the choice whether VBR or CBR should be preferred.

In Chapter 7, Guidelines for simulations, the basic configuration and its parameters are defined in order to stimulate simulations which could be compared. From the previous chapters the basic parameters are taken. Some parameters are not stable but are used for comparison only. Five test sequences are agreed upon and listed in this chapter. For the presentation of the results a table is recommended in order to compare the results more properly.

Chapter 8, guidelines for hardware experiments, will be used in future to start trials on ATM networks.

The last chapter gives the task leaders and its contributors which worked very hard to compose this excellent document.

Chapter 2

Terminology

The asynchronous transfer mode (ATM) is the target transfer mode for the future public broadband ISDN (B-ISDN). It is expected to provide an efficient utilization of the network resources by integrating a variety of communication services. In this chapter of the compendium the most important terms following from the impacts of the ATM on motion video coding and from new coding strategies will be explained.

For an easier overview this chapter has been subdivided into two main chapters describing the terminology of the network part and the video coding part, respectively.

2.1 Terminology, Network Part

Several modes of ATM are known [1], among which the target transfer mode for B-ISDN uses a concept of transmitting information blocks of fixed length, called *cells*. The transmission of several calls over a common link is performed by time-multiplexing the individual cell streams in an asynchronous way, i.e., there is no fixed reservation of time slots for each call. It is important to note that *cell consistency* and *chronological order* of cells for each call are preserved throughout the network. Since in ATM networks no physical but only logical associations exist between the end points of a link the term *virtual channel* is used to denote the realization of a logical connection.

In the following chapters some basic properties and important parameters, which have provisionally been agreed on at CCITT, will be described [(89)56].

2.1.1 Definition of "constant bit rate sources" and "variable bit rate sources" in an ATM environment

According to the asynchronous time-multiplexing of the cell streams, source data may be transmitted with variable bit rate (VBR) to the receiver. The instantaneous bit rate of a VBR source fluctuates, there will be some time intervals during which the rate is higher than the overall average bit rate and other intervals during which the rate is lower. But even if a source sends out cells with constant rate, after passing one or more switch nodes, fluctuations will be obtained to some extend according to the variable delay of the cells caused by the switch buffers (cell delay jitter). Therefore, it has to be defined under which circumstances source data is considered to be constant bit rate (CBR).

Definition:

An ATM source is said to be sending out cells with constant bit rate in a strict sense, if and only if the time interval between the respective start of emission of two consecutive cells is exactly the same for all cells emitted during the connection to the network.

This strong definition can only hold at the very first connection interface to an ATM network. According to the cell delay jitter the cells are not equally spaced any more when they passed the switch nodes in the network.

Definition:

An ATM source is said to be sending out cells with constant bit rate in a wider sense, if and only if a constant rate can be found by which a hypothetical smoothing buffer is read out such that neither buffer overflow nor underflow occurs. The buffer size for this test has to be kept small, only fluctuations due to cell delay jitter shall be smoothed down.

Definition:

An ATM source is said to be sending out cells with variable bit rate, if the rate is not constant in a wider sense.

2.1.2 User Network Interface (UNI)

For the transmission of cells two basic rates will be recommended.

1. bitrate 155.52 Mbit/s, equivalent payload: 149.76 Mbit/s
2. bitrate 622.08 Mbit/s, equivalent payload: 599.04 Mbit/s

Each cell consists of an *information field* and a cell *header* which identifies the virtual circuit the cell belongs to. The following lengths in octets have been proposed (Fig.1):

- total cell size: 53 octets
- information field: 48 octets
- header field: 5 octets

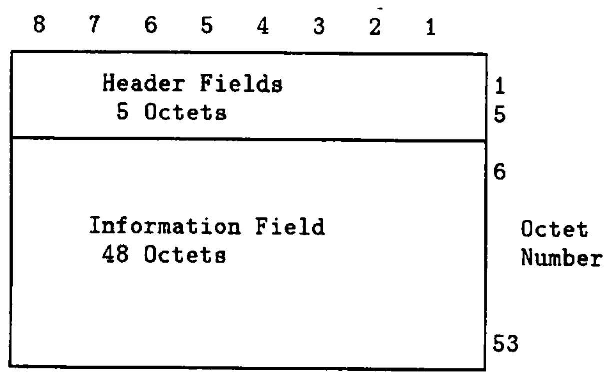

Figure 1
ATM cell structure at the UNI/NNI

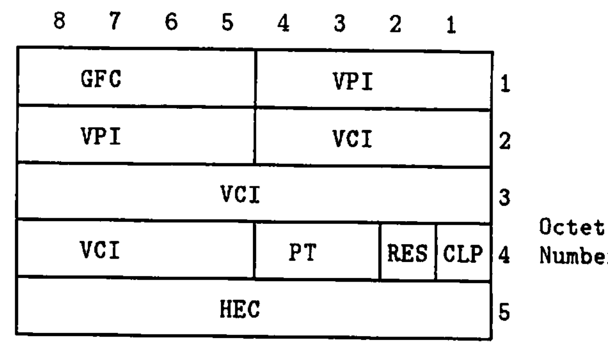

Figure 2
Header structure at the UNI

The cell header contains several fields with control informations for the network, which are:

- Generic Flow Control (GFC), 4 bits
- Virtual Path Identifier (VPI), up to 12 bits
- Virtual Channel Identifier (VCI), 12-16 bits (depending on the customers demand)
- Payload Channel Identifier (PT), 2 bits
- Reserved (RES), 1 bit
- Cell loss priority, 1 bit
- Header Error Control (HEC), 8 bits, generator polynomial X^8+X^2+X+1 (unstable information)

Cell delineation is based on HEC and a selfsynchronizing scrambler applied to the information field.

2.1.3 Network Node Interface (NNI)

For interfacing two network nodes ATM and STM transfer modes are considered. For the STM case a nes synchronous digital hierarchy (SDH), which is based on a rate of 155.52 Mbit/s, has been developed by the CCITT. Cell structure and header funtions are very similar to the UNI.

2.1.4 B-ISDN Protocol Reference Model

A protocol reference model has been established to help the functional definition of the B-ISDN (fig.3). The model is divided in planes and layers, among which the ATM layer and the ATM adaptation layer are of major importance for the work of this group.

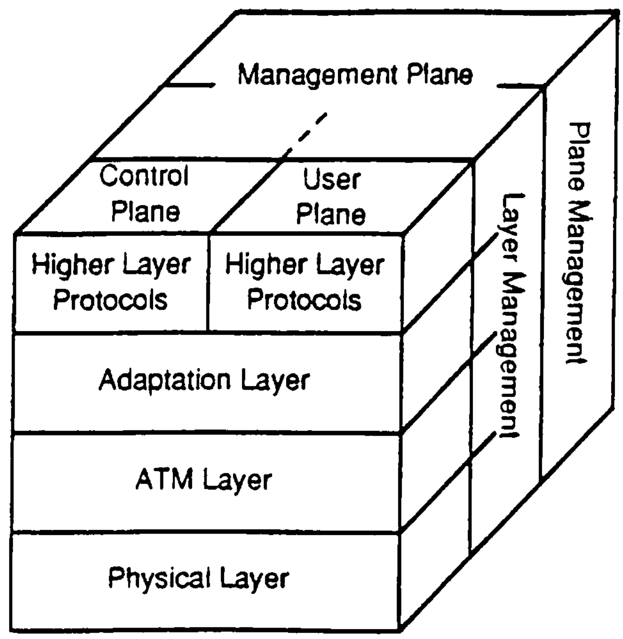

Figure 3. B-ISDN ATM Protocol Reference Model.

2.1.4.1 ATM layer

ATM Layer requirements [7,(89)50]:

- cell delineation
- merging and recovering of cell boundary information
- appending and checking head error control (HEC)
- cell rate decoupling by insertion and suppression of idle cells
- (de-)multiplexing of cells (from) to different virtual channels

The ATM Adaptation Layer (AAL) is located between the ATM layer and the user layer. Functions and structure of the AAL are dependent on the specific service class. Four different service classes have been approved (see Fig.4).

	Class A	Class B	Class C	Class D
Timing relation between source and destination	Required		Not Required	
Bit rate	Constant	Variable		
Conection Mode	Connection oriented			Connectionless

Figure 4. AAL Service Classes

Examples of sevice classes:
class A: Circuit emulation; constant bit rate video
class B: Variable bit rate video and audio
class C: Connection oriented data transfer
class D: Support of connectionless data transfer
Video services can be included either in class A or class B.

2.1.4.2 ATM Adaptation Layer

The AAL is subdivided into the two sublayers:

- Convergence Sublayer (CS), service dependent
- Segmentation And Reassambly sublayer (SAR), depending on the class of service

The lower sublayer (SAR) works on a cell per cell basis, while the higher sublayer (CS) works with information units called Protocol Data Units (PDU). Different structures for the AAL have been proposed for different service classes [(90)90]. The essential functions to be perfomed by the AAL are (for more detailed information see [(90)90]:

- cell assembly and disassembly (production of constant or variable length blocks of user information)
- end to end synchronization by a time stamp (TS) field
- detection of lost or inserted cells by a sequence number (SN) field

- protection of information blocks against bit errors by a cyclic redundancy check (CRC) field
- compensation for the cell delay variation

2.1.4.3 Segmentation and Reassembly Sublayer (SAR) for Class 1 Services

Segmentation and reassembly function is performed on a cell by cell basis. The adopted SAR-PDU (Protocol Data Unit) can be seen in Figure 5.

SN	SNP	SAR-PDU Payload
4 bits	4 bits	47 octets

SN - Sequence Number
SNP- Sequence Number Protection

Figure 5. AAL SAR-PDU for AAL class A

The sequence number (SN) has 4 bits and is used to detect lost or misdelivered cells. A specific value of the SN may indicate a special purpose. The sequence number protection (SNP) has 4 bits and is used to protect the SN against bit errors.

2.1.4.4 Convergence Sublayer (CS)

The convergence sublayer may perform the following functions:

- For high quality audio and video, error correction is performed to protect against bit errors. This may be combined with bit interleaving to give more secure protection against errors.
- For voice services this sublayer provides the clock recovery service for the receiver by controlling the buffer filling. This requires no specific field in the CS-PDU.
- For services requiring explicit time indication this may be provided by means of a time stamp pattern inserted in the payload of the AAL. Other mechanisms may be used to provide this function.
- Sequence number processing is performed at the sublayer to detect the loss and misdelivery of cells. The handling of lost and misdelivered cell condition is also performed in this sublayer.

2.1.5 Quality of service (QOS) specifications

By means of flow control it may be possible to separate traffic of different quality requirements within the ATM network thus enabling the allocation of resources according the needs of the respective services. In the following some items characterizing the quality of service will be described [2-6,(89)56].

2.1.5.1 Information loss

Different kinds of information loss can occur in ATM networks:

- Cell loss due to header errors
- Cell insertion (from other virtual channels) due to header errors
- Cell loss due to buffer overflows within the network
- Bit errors within the information field of a cell

So far no final specifications exist for the cell loss ratio. Buffer queues are usually designed for loss ratios below 10^{-8} while bit error ratios will considerably exceed this value. The effect of cell loss on the picture quality depends on the robustness of the coding algorithm where the relative occurrence of quality degradations will only be annoying for high average data rates.

2.1.5.2 Delay

Basically two performance specifications for the delay have to be considered in ATM networks:

- mean and maximum end-to-end delay (e.g. on a statistical basis)
- mean and maximum delay jitter (e.g. on a statistical basis)

Packetization delay is introduced at the encoder and decoder side since it takes a finite time to fill and empty a cell. For variable bit rates this delay depends on the instantaneous bit rate. Generally, packetization delay takes at most several ms which does not cause serious problems. Sometimes bit stuffing of incomplete cells may be required for cells including synchronization information.

Cell delay jitter is caused by the presence of queues during transmission through the ATM network, expected values are in the range of 200-600 μs. Cell delay jitter for one virtual channel means the deviation of momentary and average transfer time. Delay jitter between virtual channels is the deviation between the momentary transfer times. Delay jitter affects mainly end-to-end synchronization between encoder and decoder.

2.1.5.3 Throughput

For describing the throughput characteristic of an ATM (virtual) channel two main bandwidth allocation methods have to be distinguished, namely:

- Constant bit rate (CBR)
- changing by a call by call basis (CCC)
- variable during the call (VBR)

The latter term denotes one of the most interesting features of ATM for video coding and describes what is usually understood by VBR coding. It is expected (although not yet proven) that VBR coding leads to a stabilized picture quality compared to the fixed bit rate case. For VBR the following parameters are useful to estimate the bandwidth requirements for transmission:

- Peak bit rate, measured over a defined short period
- Average bit rate, measured over a defined long period
- Maximum peak bit rate during a call
- Maximum average bit rate during a call
- Average bit rate during a call
- Burstiness factor

2.1.6 Traffic Control

An ATM nbetwork is designed to transport a wide variety of traffic classes satisfying the required performance. To cope with this requirement an ATM network will provide some levels of traffic control capabilities:

- Connection admission control
- Usage policing
- Priority control
- Congestion control

2.1.6.1 Connection admission control

An ATM network accepts a call requested only when sufficient resources are available to carry it under its required QOS and to maintain the agreed QOS of existing calls.

2.1.6.2 Usage parameter control (Usage monitoring or usage policing)

Usage policing is defined as the set of actions taken by the public network to monitor and control user's traffic in terms of traffic volume and cell routing validity. Its main purpose is to protect network resources from malicious and/or unintentional misbehaviour, which can affect the QOS of other already established connections be detecting violations of negotiated parameters. Usage policing will apply only during the information transfer phase of a connection.

Usage policing is performed on VCs and VPs at the access point wherever they are terminated within the network. A specific policing algorithm has not been standardized. Some suggestions for policing functions are (Fig.6):

- Peak
  The maximum cell rate (peak) can be watched by demanding a minimum time between cells belonging to the same virtual connection.

- Jumping window
  Jumping window is based on counting cells in a fixed time interval. A number of X cells are allowed in an interval containing Y time slots. When time interval ends, the counters are reset and a new interval starts.

- Stepping window
  The same as 'Jumping Window', but after a passive period which is greater than the window size, the counters will restart when the first cell arrives.

- Moving window
  Moving window allow X arriving cells in an arbitrary time interval containing Y time slots. In order to control this, it is necessary to collect information about arrival time for all the cells arriving in the last Y time slots.

- Leaky bucket
  Leaky bucket is based on a counter which is incremented every time a cell arrives and decreases with a constant rate. When the maximum of the counter value is reached, policing action takes place.

- Exponential weighted moving average (EWMA)
  EWMA is an extended version of the jumping window. EWMA takes into

consideration the previous time intervals. A parameter, S, is updated in the end of every time interval:

$$S_n = (1 - \alpha)X_n + \alpha S_{n-1} \tag{2.1}$$

where X_n is the number of cells arriving in interval n, α is a constant less than 1. We demand that S shall be less than a certain value S_{max}. For $\alpha = 0$, EWMA will be identical to 'Jumping Window'.
A basic policing function is to monitor the peak. With only this parameter a natural response would be to have CBR codes using the peak cell rate. A next step could be to monitor the mean cell rate. This can be done with many of the methods mentioned. With the mean cell rate lower than the peak cell rate you allow some kind of VBR.

2.1.6.3 Priority Control

The use of priority control in an ATM network is for further study.

2.1.6.4 Congestion Control

Congestion control techniques will be used in the B-ISDN to assist maintenance of QOS of connections, e.g. connection admission control and usage policing.

2.2 Terminology, Video Coding Part

Two main classes of coding models for ATM are discussed [(89)52], the first one includes:

- codes based on allocation of transmission capacity by statistical parameters of the output rate. Important impacts are:
  - encoder needs statistical monitor (preventive policing function) and rate control and/or:
  - rate control by load information from the network [(89)57]
  - decoder needs error concealment techniques for lost cells

The second class comprises:

- layered (hierarchical) coding models [(89)3,(89)41,(89)44,(89)58]. Important impacts are:

  - one layer for basic (low resolution) picture quality and one (or more) layer(s) for enhanced quality
  - requirement of network with guaranteed (very low cell loss ratio) channel (priority channel)
  - reduced need for policing

Transmission of video data from layered coding models can be done by [(89)58]:

- one virtual channel carrying all layers (identification of layers at the AAL protocol layer)
- several virtual channels carrying one layer each

The latter is the more flexible approach but is also more expensive and has the problem of delay compensation between different layers.

Preventative policing may be required depending upon the coding algorithm. For example 1-layer coding will require this function in order that it does not exceed its negotiated bit requirement. In contrast, 2-layer coding may not require this mechanism as it can cope with a cell loss and can rely on the network.

2.3 Documents from COST 211 bis simulation subgroup

(89)3	BTRL, Video coding for ATM.
(89)41	BTRL, Further studies on 2-layer coding.
(89)42	SIEMENS, Statistical analyses of bit streams from a variable rate H.261-based video codec.
(89)44	CNET, First attempts of picture coding on ATM networks.
(89)56	INESC, B-ISDN: present situation.
(89)50	Telefonica, Starting with cell loss.
(89)52	BTRL, Proposal for ATM compendium: Coding models; first candidates
(89)57	INESC, Is adaptive quality a viable solution?
(89)58	INESC, Coding models examined in relation to compatibility and interworking issues.
(89)72	BTRL, Terminal adaptation for VBR codecs,
(90)90	INESC, Comments on Newcastle AAL conclusions
(90)98	NTA, Video coding and policing functions
(90)114	BTRL, Policing functions for ATM-networks
(90)116	BTRL, Preliminary determination of base layer bit rate in a two-layer coding architecture
(90)119	INESC, Policing in B-ISDN
(90)126	INESC, B-ISDN present situation

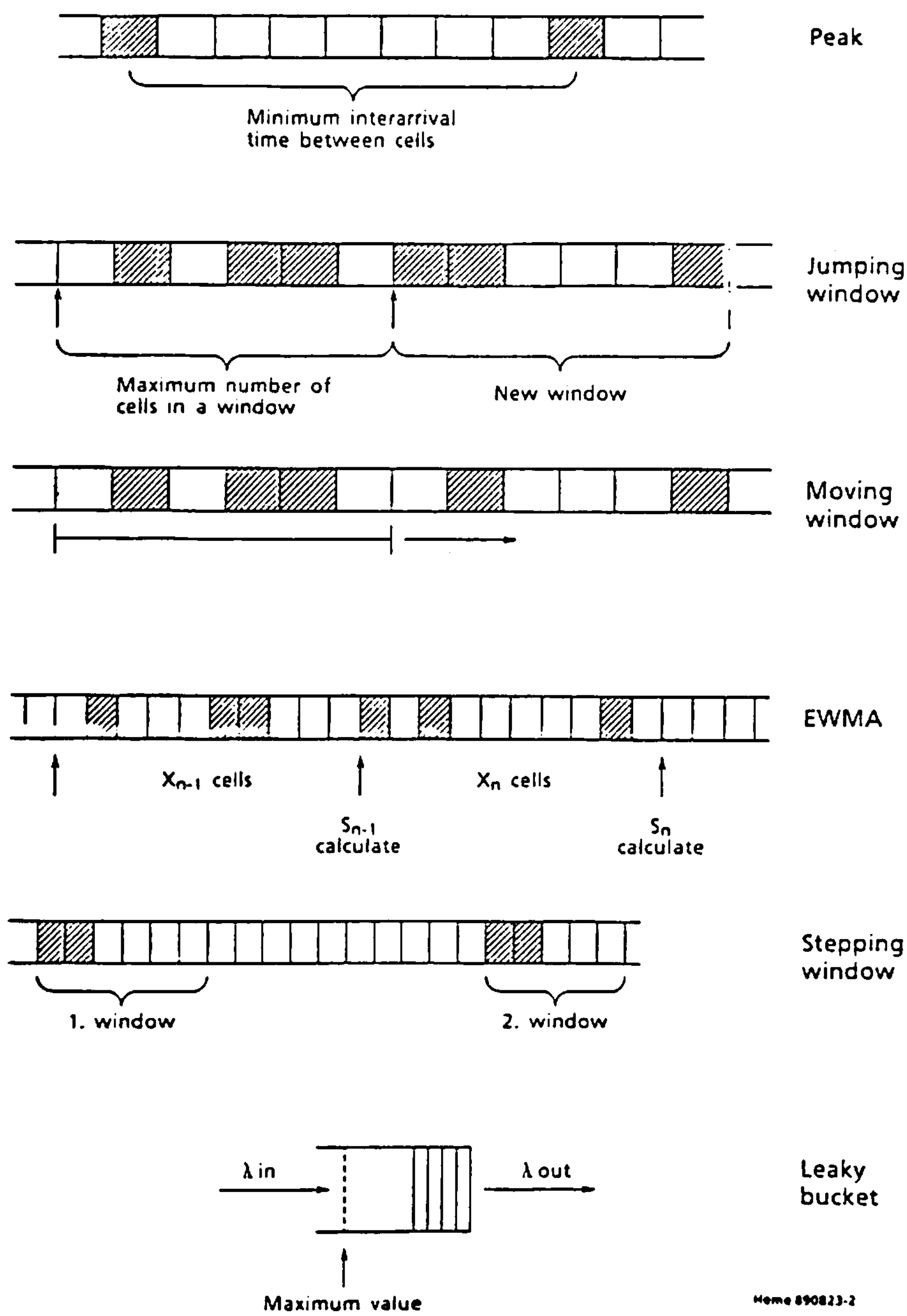

figure 6. Policing functions

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Chapter 3

Coding Models

3.1 Background on ATM networks

The characteristics of ATM networks have not been finalized and different network operators have alternative ideas of how to transport data over an ATM network. One is to load the network at a sufficiently low level so that the total peak demand of the statistically multiplexed sources very rarely exceeds the network's bandwidth. Another method is the use of a switch supporting differing Qualities of Service (QOS) to achieve a higher loading factor.
In order to control the statistical multiplexing of the VBR sources, two methods have been devised [1] :-

a) Bandwidth allocation - Video sources are characterized in terms of their average bit rate requirement and their variance so that at call set up the network finds a path that accommodates those requirements. This method requires 'policing' functions both in the network and the encoder to ensure that the video source does not exceed the statistical description specified at call setup.

b) Differing QOS - Two channels are required for each video connection (unidirectional). The first offering a constant guaranteed bandwidth and the second of variable bandwidth which, due to network loading, may be subject to cell loss.

3.2 Coding Aims for ATM

There are two main aims for the coding of real time video signals for use on ATM networks, they are :-
a) To obtain near constant picture quality.
b) Maintain compatibility with the existing CCITT H.261 specification.
A further aim is
c) To obtain a coding technique that maintains compatability over the various multimedia services.

3.3 Coding Models for ATM

Two basic models have been proposed for coding of video for transmission over an ATM network. These are the one-layer and two-layer coding approaches.

3.3.1 One Layer Coding

3.3.1.1 Existing fixed bit rate model

Existing CBR codecs could be used on an ATM network that provides a guaranteed fixed bit rate channel.

Advantage:

- no modification to existing codecs.

Disadvantage:

- does not exploit VBR for video coding.

3.3.1.2 Variable bit rate model

This codec model relies upon an accurate statistical description of the supplied video source. Cells will be lost if the video source exceeds its declared statistical description which may be incurred by a 'noisy' source, scene change, etc. For mechanisms of cell loss see chapter 2.

At the encoder (Fig.1) the coded data is monitored by a Statistical Monitor which controls the amount of data generated in the Forward Coder; possibly in a similar manner to the existing fixed bit rate models.

If cells are lost, the decoder (Fig.2) must detect the occurrence and instigate remedial action by the Reverse Coder.

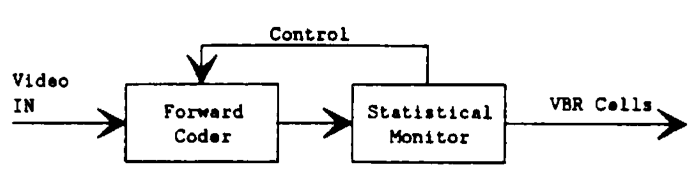

Fig. 1 A block diagram of a VBR encoder

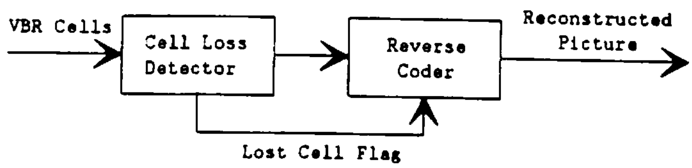

Fig. 2 A block diagram of a VBR decoder

Remarks:

- A concealment technique has to be adopted in the decoder to cope with cell loss; further study needed.
- Relies on statistical analysis of a range of video sources; further study needed.
- Possibly a simpler design than two-layer models.
- Possible compatibility with H.261 - further study needed.
- Detailed models still required.
- Requires statistical monitoring at the encoder and the network.

3.3.2 Two-layer Coding

Two-layer models offer the possibility of coping with cell loss by using a network outlined in 3.1(b).
The first layer contains a base picture which is transmitted over a guaranteed channel. The second layer contains enhancement data which is transmitted over a second channel. This channel could undergo cell loss.
Advantages:

- Guaranteed reception of a base picture, even in the event of cell loss; but requires a network that supports a guaranteed (or very low cell loss ie. 10^{-8}) channel.
- Does not rely on a statistical description of the source and hence a much reduced need for a 'policing' function in the encoder.
- Inherently compatible with H.261 if this specification is used as the first layer coding algorithm.

Disadvantages:

- requires two connections to the network; What is the skew over the network?
- More complex design; need for synchronization of two data channels in the decoder.

3.3.2.1 Quantization Split

In the encoder (Fig.3), the first layer codes a base picture and the second layer codes the difference (the enhancement detail) between the original picture and the base picture. If there is a frame drop in the base picture then the dropped picture could be encoded entirely in the second layer. Hence the picture rate may be kept at a maximum at all times. The consequence is that as the difference data fluctuates the bandwidth requirement varies accordingly.
In the decoder (Fig.4), the two data streams are decoded separately and then combined to form the reconstructed picture.
This scheme has the advantage that when cells are lost that area of the received picture associated with those cells reverts to the received base picture. Furthermore, the first layer could be coded using the CCITT H.261 algorithm ensuring compatibility of the base picture. Several possible methods of implementing a two layer codec have been documented.

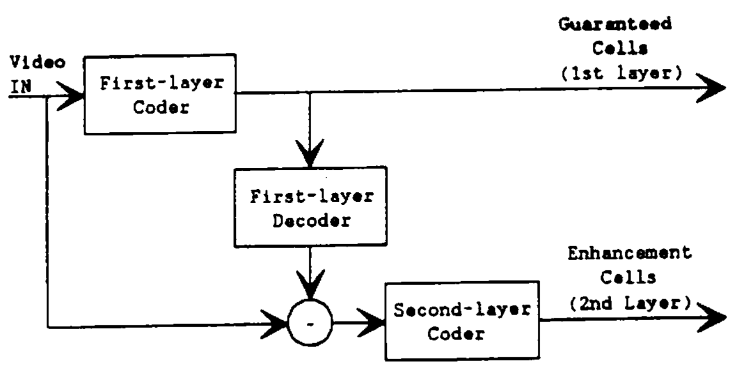

Fig. 3 A block diagram of a two layer encoder

This method has shown resilience to cell loss in the second layer as high as 10^{-3}.

3.2.2.2 Spectral Split

In this approach the high frequency content of the picture is coded in an enhancment layer. This is done by switching the high order coefficients of a block into a second layer, figure 5.
The second layer has a variable data rate. The base channel can be either variable or constant.

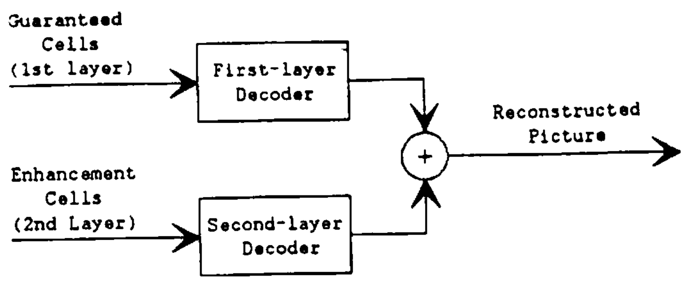

Fig. 4 A block diagram of a two layer decoder

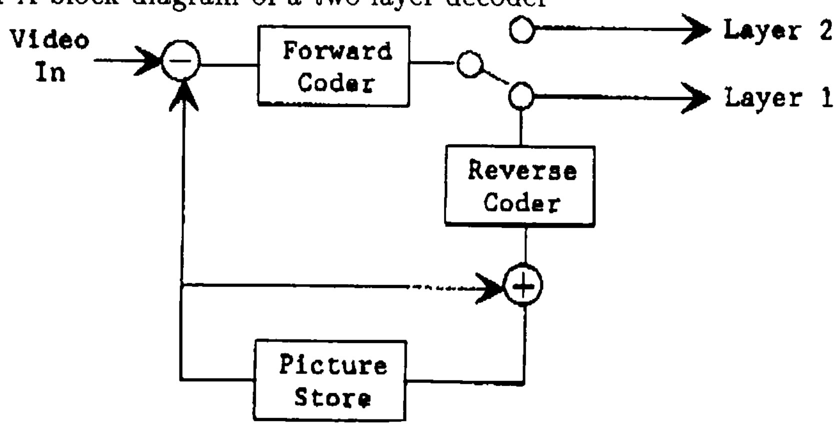

Fig. 5 Block diagram of 2-layer coding by switching coefficients

The overall bit rate for the spectral split approach is lower than the quantization split method. This is due to the fact that coefficients are not coded in both layers in the spectral split method.

3.4 Coding for Storage

An experts group in ISO is working towards standardizing coding for storage (MPEG1). This work is now being extended to cover other applications such as broadcasting (MPEG2). The bit rates under consideration (10Mbits/s) may only be available on B-ISDN networks. Hence the need to consider the effects of ATM on such coding techniques.

3.4.1 MPEG1

Work has almost concluded in ISO on coding for storage. The target bit rate is between 1.1Mbits/s and 1.5Mbits/s. The picture size is CIF.
The coding consists of intra, predicted and interpolated pictures.

A I I P I I P I I............P I I A I I P......

A intra
I interpolated
P predicted

The number of interpolated pictures can be upto six, however it is also valid to have none.
The delay in determining the interpolated frames may be too long for face to face applications. In this case it may be necessary to use intra and predicted pictures only.

3.4.2 MPEG2

Work has begun in ISO on coding of CCIR601 images. The upper bound on the bit rate has been set to 10Mbits/s. No lower bound has been set.
Applications range from digital VTR to broadcast TV. The picture quality will be no less than NTSC/PAL/SECAM and up to CCIR601.
Various picture formats will be covered:

- CCIR601 720x240x2x30; 720x288x2x25
- EDTV (16:9) 960x240x2x30; 960x288x2x25
- Progressive 960x480/576x1x24/25

Compatability with MPEG1 has not been determined yet. Though if compatability is not guaranteed the first phase products may have a short life.

3.5 References

[1] W. Verbiest; 'The Impact of ATM Networks on Video Coding', IEE Colloquium on "Packet Video" , May 1989

3.6 Relevant COST Documents

One-Layer

Daimler Benz; 'Performance of video decoding according to CCITT H.261 in the presence of cell loss', COST211 bis 90/76

Two-Layer Quantization Split
BTRL; 'Video coding for ATM', COST211bis Sim 90/3

CNET; 'Performance of 1-layer VBR verses 2-Layer VBR coding scheme', COST211bis Sim 90/103

BTRL; 'Preliminary studies into cell loss in a 2-layer coding scheme', COST211bis Sim 90/115

BTRL; 'Determination of base layer bit rate in a 2-layer coding architecture', COST211bis Sim 90/116

BTRL; 'Preliminary studies into the effects of 1-layer and 2-layer video coding on loading of ATM networks', COST211bis Sim 90/121

BTRL; 'Application of a threshold to VBR enhancement data in a two layer coding scheme', COST211bis Sim 90/143

IST/CSELT; 'Bit rtae optimization on a 2-Layer constant quality video coding scheme', COST211bis Sim 90/151

Two-Layer Spectral Split

CNET; 'Attempt for a new 2-layer coding scheme', COST211bis Sim 90/125

DBP-T/FI; 'Comparison of two-layer coding scheme', COST211bis Sim 90/136

BTRL; 'An alternative 2-layer coding scheme', COST211bis Sim 90/138

Coding for Storage

RNL; 'Simulation model 2 (SM2)', COST211bis Sim 90/111

RNL; 'Comparison RM8 and MPEG, SM2', COST211bis Sim 90/117

BTRL; 'Coding of CCIR601 images with H.261', COST211bis Sim 90/141

Chapter 4

The H.261 Capabilities for ATM

4.1 Introduction

The mandate for ATM provided by the management committee clarified that the ATM coding procedure should have at least the H.261 capabilities. This means that intercommunication is mandatory.

The studies on this topic need to be carried out in the lignt that modifications to the H.261 are implemented in such way that there exists an fallback mode to the H.261 standard. The decoder should be able to decode a H.261 bitstream.

In this chapter which is related to chapter 7 simulations based on the latest Reference model need to be carried out.

This chapter 4 reports the work already done in this topic within the group. The aim is to try to give stable assumptions for the future work.

4.2 Background

H.261 coding algorithm was studied for STM networks, that means it has been optimised to keep a fixed bitrate independently of the picture content. Due to the regulation process only a small variation of the instantaneous bitrate is permitted (mainly determined by the buffer sizes).

A major target is to increase the global quality of transmitted pictures by using variable bitrate transmission (4).

ATM networks should allow constant video quality with variable bitrate in contrast to the present H.261 in STM applications where we have variable quality with fixed bitrate.

H.261 capabilities for ATM depend strongly on ATM capabilities (cell loss, cell intrusion, delay jitter, allowed burstiness, allowed mean bitrate) , but up to now too many features of the ATM are not specified enough and under discussions.

Even without a well defined ATM network behaviour, we can try to make some assumptions on the advantages of the H.261 coding scheme and the possible modifications or adaptations.

4.2.1 H.261 coding advantages

* H.261 represents the state of the art in picture coding for low -and medium- bitrates.

* H.261 provides reasonable picture quality for various bitrates (Specified for range 64 kbit/s up to 2 Mbit/s, Recommendation p x 64 kbit/s p = 1 to 30).

* Depending on the sequence content, H.261 allows rather good picture quality by adapting the value of p between 1 to 30.

* Independently of specific problems due to ATM networks, in term of coding efficiency H.261 algorithm is in line with the ATM video bitrate range (mean 2 Mbit/s, maximum 10 Mbit/s and burstiness around 4)

* H.261 picture quality is mainly given by the value of the quantizer stepsize. Picture quality can be kept approximatly constant by fixing the quantizer stepsize (leading to variable bitrate).

* H.261 codec is intrinsically a variable bitrate codec which is transformed in fixed bitrate codec by the regulation loop.

* statistical decorrelation of bit streams from different H.261 coded sequences might be exploited.

* components have been already developed to perform some parts of the coding process (DCT,MC...)

* A H.261 based algorithm is a good canditate for storage on CD-ROM

4.2.2 Drawbacks

* Coding efficiency is mainly obtained by using temporal correlation(motion compensation, temporal prediction, conditional replenishment, differential entropy coding), which implies that the decoder must have the same information of the past: H.261 might be more sensitive to errors than other dedicated ATM coding schemes.

* DCT might not allow easy and efficient sub-band coding in multilayer algorithms.

4.2.3 H.261 capabitities for CCIR 601 pictures

The coding method in ATM environment has not only to deal with low bit rate services. The H.261 coding efficiency has been roughly evaluated for CCIR 601 moving pictures at 15 Mbit/s and 20 Mbit/s (18). The quality obtained is largely superior for the specific CCIT 601 algorithms. This can be explained by the following reasons:

- there is half pel accuracy in the motion compensation search in the CMTT algorithm (which seems really efficient in case of zoom and pan).
- there is a weighted quantization in CMTT algorithm.
- the efficiency of the H.261 VLCs for hight bit rates has to be proved.

In order to encode CCIR 601 pictures some modifications of the video multiplex have to be made .

Some preliminary simulations have been made (26) of hierarchical coding of CCIR 601 pictures using a layered H.261 based coding algorithm, the CCIR 601 image being decimated in 4 CIF sub-images using sub-band coding.

4.3 VBR versus CBR

Several results have already shown a better picture quality when H.261 is used in variable bitrate environment (4,3,5,8,15)
The quality depends on the regulation stategy.

The first conclusions are:

* Better quality can be achieved with “open loop” coders

* It is not fair to compare VBR and CBR coding with the same mean bitrate (due to loading network considerations) (17).

*ATM networks seem also to imply to have a minimum regulation loop to reduce the burstiness (where is the optimun between codec and network constraints ?)

* There is no evidence in reduction of H.261 codec complexity with ATM networks

4.4 H.261 adaptations for ATM networks

4.4.1 Influence of cell loss

If some preliminary results show that H.261 can cope with reasonable error rate in existing digital network by using conventional error correction techniques (BCH), the impact of cell loss (burst errors) has not yet been enough examined at different cell loss rates.

The possibility to use H.261 coding scheme, and the way(s) to use it will depend on the results of the work we have to do on the impact of cell loss with H.261 codecs and the cell loss rate in ATM networks. Some simulation works (14) have already demonstrated that the impact of transmission errors (or cell losses) depend on the decoder stategy when an error (or cell loss) occurs.

Several ways have to be explored:

4.4.2 Scenario 1 - Single bearer service

With the assumption that the coding is not too affected by cell loss in ATM networks.

a) Very low level of cell loss

In this case the only problem to adapt CCITT Rec.H.261 is then the burstiness and the mean bitrate control. It is necessary to find a suitable regulation to keep constant picture quality (but this problem is not obviously part of a future recommendation and will depend on specific implementations)(6).

* Quasi-VBR model for packet video (11,22):

With this proposal all the cells are treated in the same way. The coding strategy consists to switch the p value, depending of the activity and the buffer content, adequating it to the minimum value. This modification on the CCITT Rec.H.261 should permit to obtain coded sequences with a constant picture quality.

* Iterative algorithm (9,13)

In order to keep a "constant quality" of pictures in a ATM network which have some constraints on the statistical parameters of the bitstream (e.g peak) it is necessary to control the codec bitrate. An iterative algorithm that allows to maintain constant the Luminance Signal to Noise Ratio (LNSR) by choosing adapted quantization step could be used.

b) Low level of cell loss

Several techniques might be envisaged:

- Break the temporal correlation in the bit-stream (for example by adding more intra coded blocks).

- Use feedback information from the decoder to indicate which cells are missing (influence of the reaction delay, impossible for broadcast services).

- Use decoding strategy in case of cell loss. Determine what the acceptable parameters for BER and CER are to use H.261 codecs in ATM.

4.4.3 Scenario 2 - Layered bearer service

If the ATM networks could provide multiple Quality Of Service (QOS), at least 2, it might be possible to have a guaranteed bandwidth allocated to video signal. This has led to the idea to have two coding levels (1).

The encoded data flow is divided into two bit streams: one transmitted on a guaranteed channel with a very low packet loss probability (first layer), and the other one on a non guaranted channel (second layer). Then a two layers H.261 coding based could be used.

2- layer coding H.261 based

a) First layer:

In the first channel we assume that we don't have to deal with the problem of cell loss ratio. Depending on the bitrate capacity of this guaranteed quality channel it would be possible to use H.261 at low bitrate (p x 64 kbit/s, p = 1 to 5) or to transmit only some important information of the H.261 video bitstream: picture start codes, coding modes, motion vectors...(several kbit/s).

If H.261 is employed for the first layer (fallback mode for STM network), the picture obtained by decoding the first layer provides basic quality independently of the network loading.

The first layer could be CBR or VBR but if the first layer is CBR then the network can more easily guarantee this and thus provide lower cell loss ratios.

b) Second layer: several proposals

The second channel is used to transmit an enhancement of the picture.. This enhanced signal may be used to:

- Improve the picture quality which is not sufficient at low bitrate
- Have a higher frame resolution
- Have a higher spatial resolution
- ....

Several techniques have already been studied and proposed in which the second channel is used to:

- transmit the overflow signal (H.261 bitstream is shared between VBR and CBR channels - in case of buffer overflow, the data stream is sent over VBR)

- transmit an enhanced transform picture signal:the difference between the DCT coefficients of the first layer and their quantized values are quantized again with a fine quantizer (1,7,10,12,16,20)(fig 1).

- transmit only a reduced number of quantized coefficients in the basic layer (fig 2) and the rest in the second channel. In this case a fine quantizer is used for the both channels (stepsize 8) (24).

- transmit an enhanced picture in the pel domain: the difference between the imput picture and the first layer basic decoded picture is encoded using a DPCM method which can be less sensitive to cell loss than hybrid transform coding (1) (fig 3).

- transmit subsampled frames...

- To encode pictures for CD-ROM storage, it is used an interpolative mode which should be invastigated for ATM coding. For the interpolated frames, only motion vectors for interpolation and coefficients for these frames are transmitted. These interpolated frames are not used in the coding loop for prediction then it is possible to send interpolation data in the second channel. The main problem of this coding scheme for videoconferencing or videotelephony applications is the delay introduced by the interpolation (19).

c) Two - layer coding efficiency : If the first simulations showed some problems to keep a constant picture quality the last presentations (simulation and hardware) give promissing results (16).

An evaluation of the two layer scheme is necessary to determine what is the overall bitrate required in such schemes to obtain the same quality as a one layer sheme at a given bitrate. The ratio between the basic layer bitrate over the total mean bitrate is an important parameter for the coding efficiency (21).

4.4.4 Error correction

The BCH error correction conforming to H.261 is usable in ATM networks, if either CBR (class 1) is used, or if the payload of class 2 PDUs is completly used.

As there are two FEC mechanisms for class 2 VBR if a (modified) H.261 codec is used, a decision has to be made, which FEC is better suited. None of them can be switched off (23).

4.4.5 Miscellaneous

Also H.221 is not a part of H.261, it is pointed out that we will need a substitute for videotelephony and videoconferencing services in ATM networks.

Some Relevant COST211Bis Simulation Subgroup documents

(1) COST 211Bis Sim/89/3	Video coding for ATM (BTRL)
(2) COST 211Bis Sim/89/4	First results on videoconferencing over ATM networks (Siemens)
(3) COST 211Bis Sim/89/5	Video services supported by ATM networks (Sweden)
(4) COST 211Bis Sim/89/7	Video codecs for ATM networks: still a hybrid scheme? (CSELT)
(5) COST 211Bis Sim/89/39	VBR video codec output simulator based measurements (Tech.Res. C. Of Finland).
(6) COST 211Bis Sim/89/40	VLC and Buffer Regulation for video transmission in ATM (UCL)
(7) COST 211Bis Sim/89/41	Further studies on 2-Layer Coding (BTRL)
(8) COST 211Bis Sim/89/42	Statistical analyses of bitstreams from a variable rate H.261 based video codec (Siemens)
(9) COST 211Bis Sim/89/43	Extension of CCITT visual communication codec for operatio in ATM networks (CSELT-IST/JNICT)
(10) COST 211Bis Sim/89/44	First attempts of picture coding on ATM networks (CNET)
(11) COST 211Bis Sim/89/49	A quasi-VBR Model for packet video(TELEFONICA)
(12) COST 211Bis Sim/89/53	Quantization of the second layer of an adapted RM8 two-layer codec (BTRL)
(13) COST 211Bis Sim/89/68	"Constant Quality" video coding in ATM networks: an iterative algorithm (IST/JNICT-CSELT)
(14) COST 211Bis Sim/89/84	Performance of video decoding according to CCITT H.261 at the presence of cell loss (DAIMLER).
(15) COST 211Bis Sim/89/86	A comparison of Variable versus Constant Bit Rate and the impact on the picture quality for fixed framerate (PTT RNL)
(16) COST 211Bis Sim/89/91	'Jack in the Box' through a Two-Layer VBR Video Codec (Hardware Demonstrator) (BTRL)
(17) COST 211Bis Sim/89/99	Comparing VBR and CBR coding schemes (STA)
(18) COST 211Bis Sim/89/107	Comparison of CCIR 601 algorithms versus H.261 for 15 Mbit/s and 20 Mbit/s (UCL)
(19) COST 211Bis Sim/89/107	Comparison between the Reference Model RM8 and the MPEG simulation model SM1 (CNET)
(20) COST 211Bis Sim/89/113	Proposal for Two-layer Coding Reference Model (BTRL)
(21) COST 211Bis Sim/89/113	Determination of Base Layer Bit Rate in a Two-Layer Coding Architecture (BTRL)
(22) COST 211Bis Sim/89/122	A quality maintenance based scheme for packet video (TELEFONICA)
(23) COST 211Bis Sim/89/124	Usage of BCH error correction in ATM networks (DAIMLER)
(24) COST 211Bis Sim/89/125	Attempt for a new 2-layer coding scheme (CNET)
(25) COST 2111Bis Sim/90/136	Comparison of 2 layer coding schemes (DBP)
(26) COST 211 Bis SIM/90/138	An alternative 2 layer coding scheme (BTRL)
(27) COST 211 Bis SIM/90/141	Coding of CCIR 601 images with H.261 (BTRL)
(28) COST 211 Bis SIM/90/145	Influence of noise on the code bit rate of an open loop h.261 codec (Siemens)

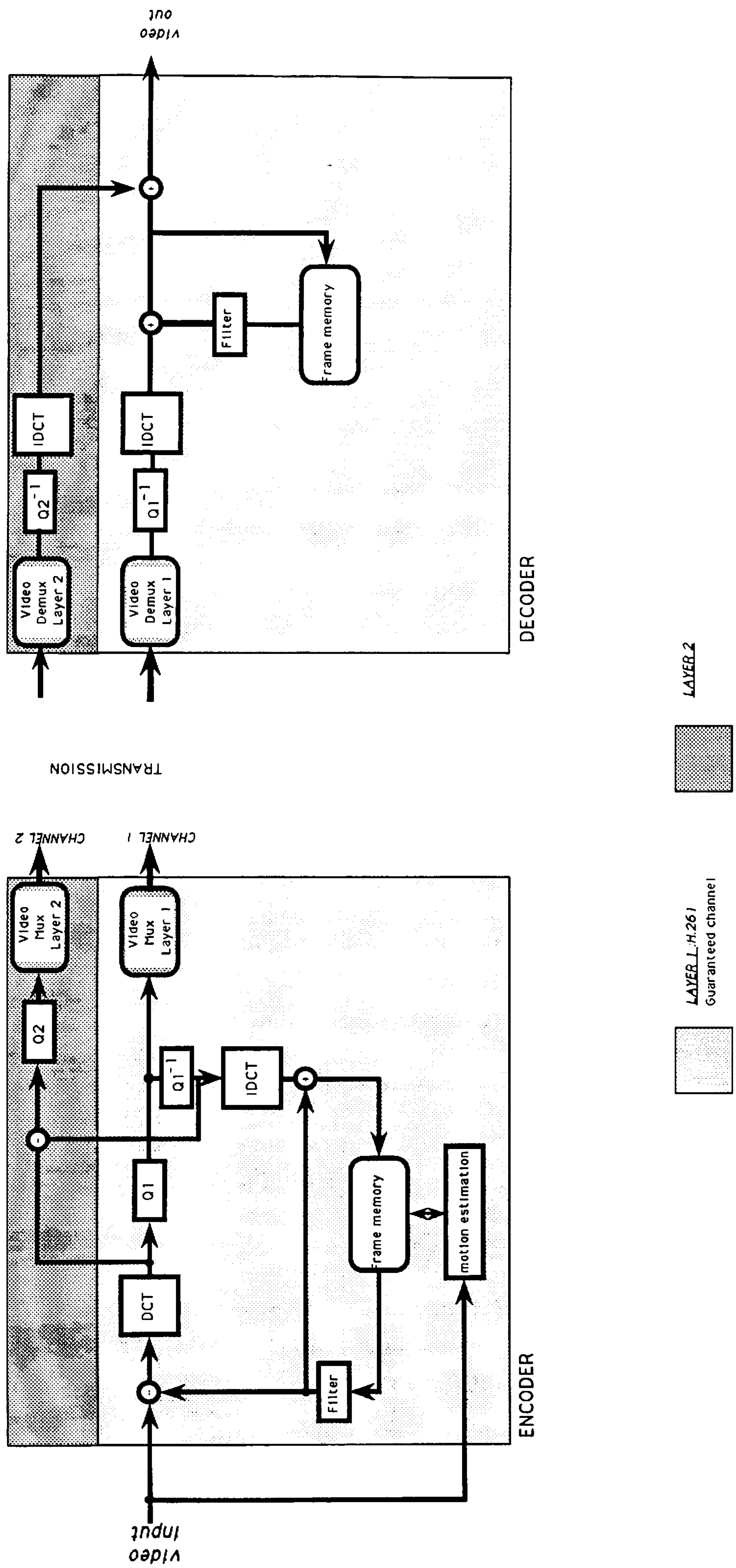

FIG 1: H.261 compatible two layer coding scheme (Transform domain)

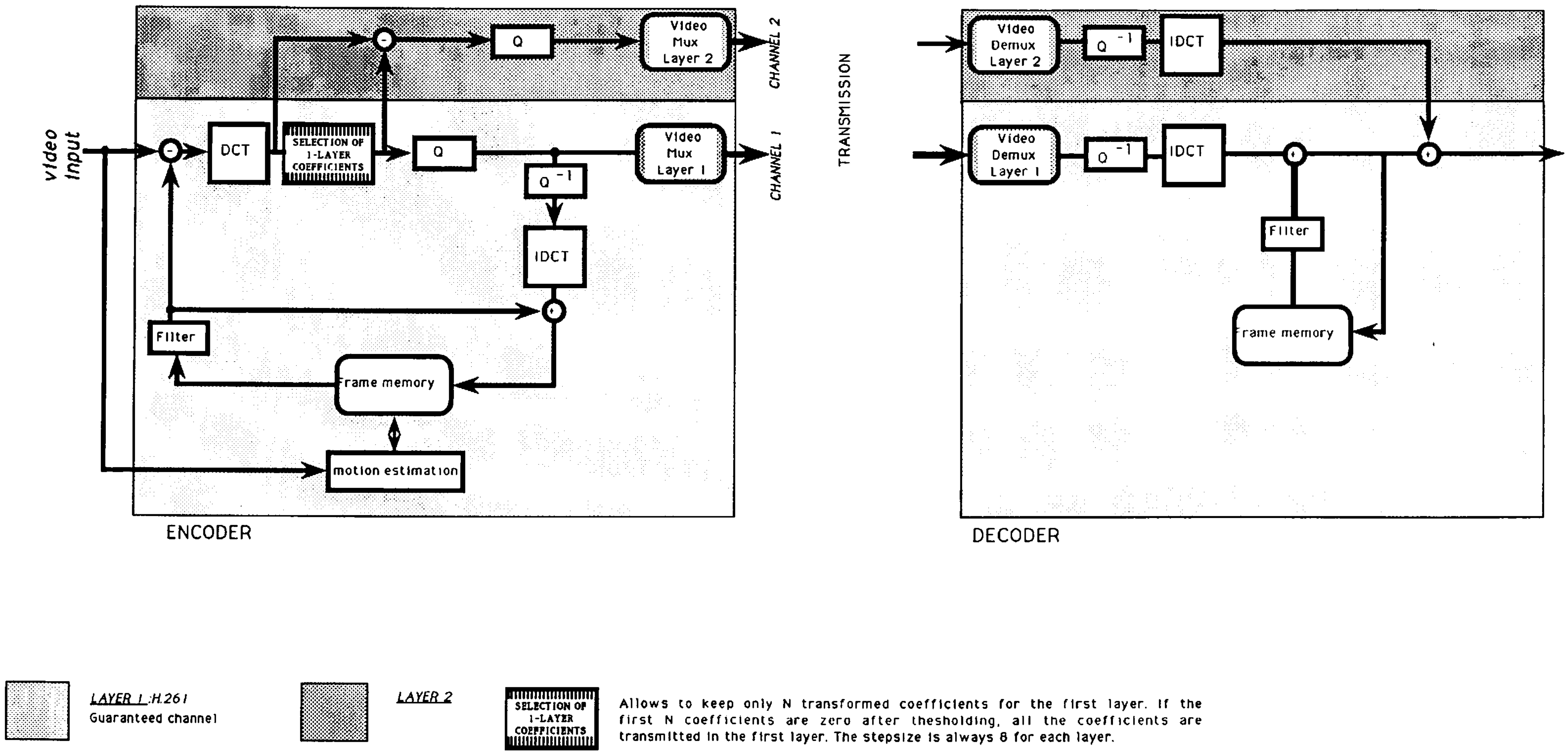

FIG 2: An alternative H.261 compatible two layer coding scheme
(Transform domain)

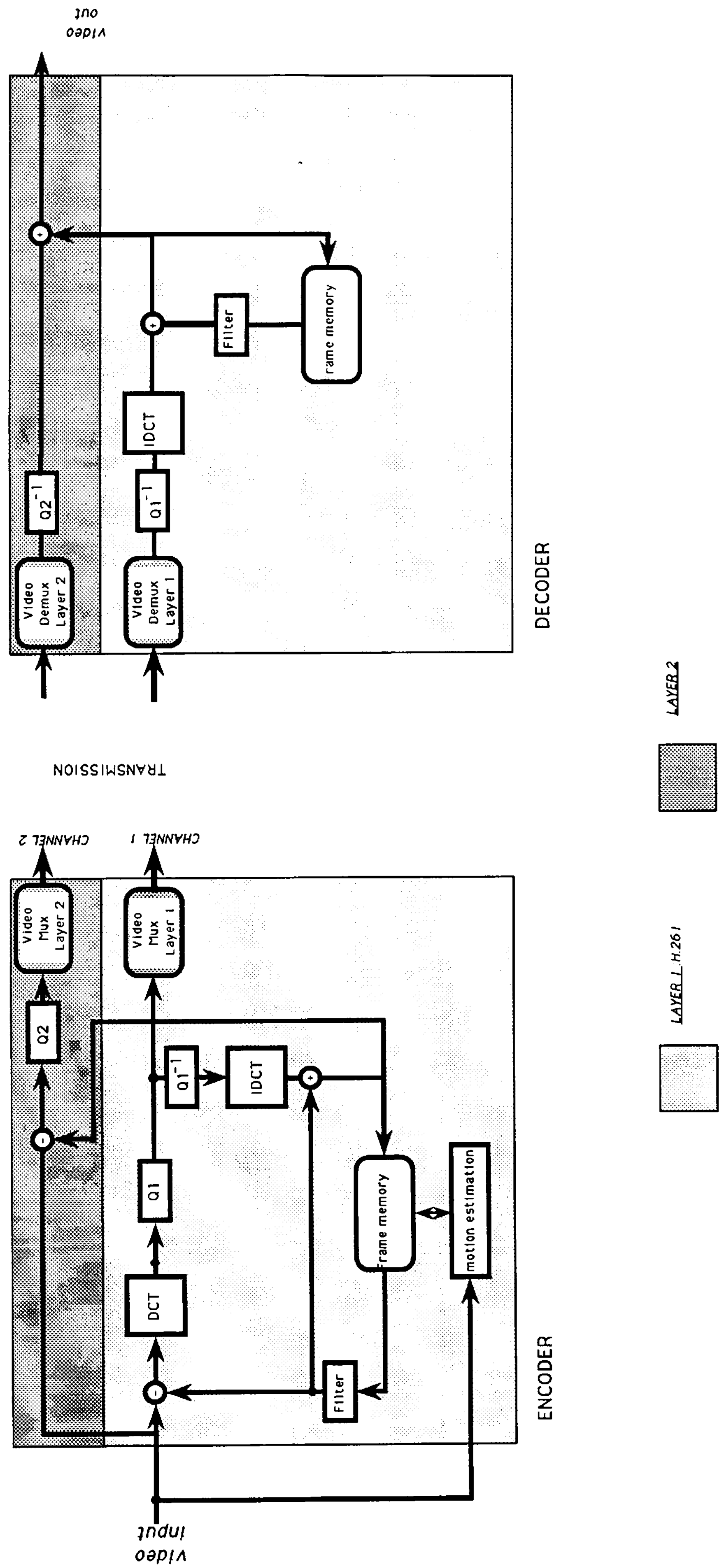

FIG 3: H.261 compatible two layer coding scheme (Spatial domain)

Chapter 5

Impact of QOS and Preventive Policing on Coding Performance

QOS, as defined in chapter 2, is characterized by the parameters information loss, delay and throughput. This chapter intends to give an overview of these parameters and how they and preventive policing influence the coding performance.

5.1 Information loss

Loss of information has to be considered both on cell and bit basis. Loss of information incorporates both cell loss, cell insertion, and bit errors.

5.1.1 Cell loss and insertion

Cell loss may occur due to bit errors in the header or buffer overflow in the network. Cell insertion may occur due to bit errors in the header.

The ATM network is meant to handle all sorts of video services, from low bit rate videophone to high bit rate HDTV. It is a common thought that users will be less tolerant with errors for the high bit rate services (TV, HDTV) than with low bit rate services (videophone).

The probability of cell loss does not change if the bit rate changes. Table 5.1 shows the relations between bit rate, probability of cell loss, and mean time between cell loss for high priority type channel, while table 5.2 shows the same for low priority type channel[1].

When loosing a cell because of buffer overflow in the network, there is a possibility that consecutive cells may be lost. The probability for loss of consecutive cells is small for a high priority type channel using not a very high bitrate. For

[1]Figures are rounded to 1-2 digits. Therefore, the relation between individuals of them are not always exactly correct.

	E-10	E-9	E-8	E-7
140 Mbit/s	8 hours	45 min	5 min	30 sec
34 Mbit/s	1 day	3 hours	20 min	2 min
2 Mbit/s	3 weeks	2 days	5 hours	30 min
384 kbit/s	4 months	2 weeks	1 day	3 hours
64 kbit/s	2 years	2 months	1 week	17 hours

Table 5.1: Mean time between cell loss - high priority channel

	E-4	E-3	E-2	E-1
140 Mbit/s	30 msec	3 msec	300 μsec	30 μsec
34 Mbit/s	125 msec	12 msec	1 msec	125 μsec
2 Mbit/s	2 sec	212 msec	21 msec	2 msec
384 kbit/s	11 sec	1 sec	110 msec	11 msec
64 kbit/s	66 sec	7 sec	663 msec	66 msec

Table 5.2: Mean time between cell loss - low priority channel

a low priority channel consecutive cell loss must be considered as the common case.

5.1.2 Bit errors in the information field

Error correction on the payload has to be considered on basis of coding algorithm and BER (Bit Error Rate). Error correction is easiest to implement if the user information is grouped in constant length blocks.

5.1.3 Effect of information loss

In the litterature there are examples showing the effect of cell loss for different coding algorithms. The effect of cell loss will very much be the same as the effect of bit errors. Simulations show that it is very critical where in the bitstream one or more cells are lost. Loss of cells containing controll information (Picture Start Code, Group of Block Start Code, etc. for H.261 codecs) can give very dramatic results. Two obvious problems are:

- loss of variable length code (VLC) synchronization

- error propagation when using differential coding schemes

5.1.4 Error control coding

In conversational services the realtime aspect does not allow the use of retransmission of information. Therefore, precautions must be taken on the encoder side so that the decoder may discover information loss and do something about it if necessary. Various methods might be used, such as:

- sequence numbering of cells [1, 7]
- FEC (forward error control coding) [10, 11]
- interleaving of data [4]
- duplication of cells with crucial data

Protecting cell headers from bit errors will be very useful. The Header Error Control (HEC) function chosen by the CCITT will be capable of correcting a single bit error in the header. In the case of multiple bit errors which prove to be uncorrectable, the whole cell shall be discarded to prevent misdelivery of the cell, but there can be situations where multiple errors are not detected and this leads to misdelivery (cell insertion).

Sequence numbering of cells can be done in the AAL. The knowledge of which cell is being received by the decoder is of great importance, and can be utilized to have a more efficient error control and recovery.

5.1.5 Error recovery

If cell loss or bit errors are detected by the decoder and it is not capable of correcting it, it has to use some kind of error concealment technique to reduce the visibility of the error, or the coding algorithm can be chosen to be little sensitive to errors.

- use of layered coding schemes [1, 3, 4, 5, 6, 8, 9]
- insertion of zeroes for lost cells [1]
- replacing "lost pixels" (caused by cell loss) with corresponding pixels in previous frame [1, 7]
- forced update (intrafield coding) [2]
- use of synchronization flags [4]

The choice of method/methods will depend on the network parameters and what algorithms are used in the codecs.

5.2 Troughput

Chapter 2 describes 3 different modes of throughput:

1. Fixed bit rate (FBR)

2. Bit rate variable on a call by call basis

3. Bit rate variable during the call (VBR)

While the user this will see this as 3 different alternatives, we can look upon them as 2 different modes when analyzing coding performance:

1. Continuous bit rate (CBR) (1 and 2)

2. Variable bit rate (VBR) (3)

VBR gives possibility to have constant or almost constant quality.

5.3 Delay

5.3.1 End to end delay

The delay in todays codecs are in the order of tens of seconds. With ATM packetization delay is introduced (see 2.1.4.2). Packetization delay is expected to be as most several milliseconds and is not considered as a big problem.

5.3.2 Delay jitter

The delay jitter is how much the time for arrival of cells (or bits) do differ from the ideal arrival time. In an ATM net delay jitter may be caused by that different cells may have been different buffered in the net.

5.4 Preventive Policing

Especially in a network without possibility of prioritized packets the policing function is essential. Violating the policing function leads to that cells may be discarded by the network and give a cell loss bigger than spesified by the QOS. For som coding schemes this may lead to very visible errors, while other schemes cope well with that problem.

As an example we look at two different cases.

5.4.1 One Layer Coding

By one layer coding we are in this case thinking of a coding algorithm where we do not differentiate between the different cells coming from the coder. In this case it is important not to break the contract the terminal has with the network. This could lead to additional cell loss. A way to avoid additional cell loss is to control the bit rate from the codec towards the net - a kind of buffer regulation. This regulation should try to adopt the bit rate to the contracted parameters, but see to that the contract is not broken. Two cases of contracts are considered here:

1. Simplistic case: Constant Bit Rate (CBR) Contract
   - No adaptation of H.261
   - No additional cell loss due to violation of contract
   - Variable picture quality
2. More complex case: Variable Bit Rate (VBR) contract
   - Buffer control in H.261 becomes more complex
   - Possibly more constant quality

   The preventive policing function (buffer control) could monitor parameters like mean, peak and burstiness.

5.4.2 Two Layer Coding

By two layer coding we are in this case thinking of a coding scheme consisting of a CBR base layer and a VBR enhancement layer. The enhancement layer is not used in prediction, and cell loss here will not result in error propagation.

- Guaranteed path (high priority packets) (CBR)
  - regulated CBR
  - no adaption of H.261
  - known cell loss rate
  - minimum quality given by H.261
  - not constant quality
- Enhancement layer (VBR)
  - possible to have open loop coding

- very simple policing function, if any at all
- unknown cell loss rate
- possibly more constant quality

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[10] TELEFONICA I+D, "AAL Versus Videoservices", COST 211bis, Doc. SIM(90)/150.

[11] STA, "Reconstruction of Lost Cells", COST 211bis, Doc. SIM(90)/130.

Chapter 6

Bitrate versus Quality

6.1 Introduction

This chapter studies the impact of different coding schemes on the picture quality, on the resulting bitrate and on the expected network facilities.

The effect of 3 different kinds of coding schemes

- constant bitrate (CBR)
- variable bitrate (VBR), one layered
- variable bitrate (VBR), two layered

is investigated.

Questions to be answered are :

- what is the resulting picture quality?
- what are the parameters of the codec output (average bitrate, burstiness, burst duration, ...)?
- what is the controling possibility of the codec?
- (were the above mentioned parameters calculated after simulation or were they specified when simulation started?)
- what is the cost in terms of bandwidth to be allocated by the ATM network?
- what is expected of the network in terms of quality of service, policing, ...?

6.2 Assessment of quality

In the following the term "quality" is used as a synomym for "picture quality".
The quality assessment problem is subdivided into four tasks:

- "Stand alone" quality assessment
- Quality comparison of different coding schemes
- Impact of coding control stategies on the picture quality
- Impact of network characteristics on the picture quality

These four points will be discussed in the following paragraphs.

Results and conclusions of simulations will be listed in another paragraph. The assumption will be clearly mentioned in order to give a correct weighting to the achieved result.

Some general conclusions are grouped in the paragraph 6.8 Conclusions.

6.3 "Stand alone" quality assessment

6.3.1 General

It is well known, that the usual SNR-measure is insufficient in describing the perceived quality, and sometimes even leads to erroneous results when used for comparison. Nevertheless the SNR is still the most popular quality measure due to its simplicity.

In contrast subjective quality assessment is slow and expensive unless you just want to judge which of two pictures is best, when they are presented to you "side by side" - so to speak - in either time or space.

6.3.2 Applied quality assessment methods

Normally the quality of simulation results are assessed in the following two ways:

1. Objective evaluation using the SNR-ratio.
2. Informal subjective test.

The simulation group simply watches the sequence (sometimes interleaved with some reference e.g. RM8). Afterwards the impression is discussed in rather unprecise terms.

6.3.3 Suggestions for improvement

The subjective evalutation could be improved, if each sequence was evalutated in a more specific way, where specific parameters were assessed. This could include:

1. Spatial resolution
   - Moving part
   - Background
2. Temporal resolution
   - Motion reproduction
   - Jerkiness
3. Coding artifacts
   - Mosquitos
   - Dirty window
   - Background noise
   - Twinkling
4. Colour resolution

Each point could be evaluated using a 5 grade scale.

6.4 Quality comparison of different coding schemes

6.4.1 "Fair" comparison

In order to make a "fair" comparison some parameters must be chosen to establish a common base between the compared schemes. These parameters must be clearly stated.
In addition all factors which influence the complexity of the network must be taken into account (e.g. policing function, cells with different priorities, ...).
It should be pointed out that for a "fair" comparison the "cost" implications of the diffent coding schemes should be considered. For instance, an obvious implication related to CBR and VBR is that their average bitrate are not equivalent, as the VBR scheme costs more in terms of bandwidth than the CBR scheme. The network has to allocate more bandwidth for the VBR source in order to cope with the bursty character of the call. This statement is however not entirely true for the case in which a QoS as low as 10 E-1 in terms of cell loss ratio is accepted for the VBR source.
Some parameter which are relevant for the comparison of the different coding schemes are listed hereafter :

1. Parameters specifying the codec output
   - average bitrate
   - burstiness or peak bitrate
   - variance
   - burst duration and frequency
   - quality
2. Parameters specifying the network
   - link capacity
   - cell loss ratio
   - queing technique
   - buffersize (delay)
   - policing function
   - tariffing
3. Parameters important for or resulting from both network and codec
   - QoS required
   - Priorities required
   - Policing function required at the encoder
   - Measuring period of all statistical parameters

In the future the exact definition of the above parameters and the criteria for "fair" comparison should be established, taking into account the different ways in which the comparison can be done.
Two ways of comparison are used today :

1. The same bandwidth allocation on the network
   In this case it is important to specify the network model which was used to calculate the comparable bandwidths.
2. The same quality achieved
   When two simulation are said to have the same quality it is important to mention on which quality criterias this assumption is based.

6.4.2 Quality of different VBR schemes

6.4.2.1 General

When VBR is used the picture quality can be kept more or less constant and sudden changes in the scene like scene cuts and sudden movements can be overcome without loss of quality. This requires however that the network can cope with the bursty data. Different coding schemes can be designed with a certain trade-off between coding efficiency and error resilience. This will further be discussed under 6.4.2.2 One layered VBR codecs and 6.4.2.3 Two layered VBR codecs.

The codec can also be thought to control its output in order to suite the network in which it will be used. Only few efforts have up till now been made to controle a VBR codec in order to not violate some predefined parameters.

6.4.2.2 One layered VBR codecs

The efficiency of the codec can be compared to the efficiency of a CBR codec, but the bitrate varies with the amout of data to be transmitted. The quality, which can be constant when no cells are lost, can seriously degrade when during network overload cells have to be discarded. One layered codecs are no more error resilient than CBR codecs and relatively low cell loss ratios will probably be required.

6.4.2.3 Two layered VBR codecs

The coding efficiency of an error resilient two layer codec is less than for a one layer codec. However the network can be loaded more heavily due to the error resilience of the codec and no visible degradation of quality is caused by cells lost in the second channel, assuming a cell loss ratio less than or equal to 10-1.

Simulation work has primarily been concentrated on :

- the quality resulting from coding for base channel
- quality gain for the second channel
- the ratio of bitrates for base channel and second channel and the impact on the picture quality

6.5 Coding control strategies

The picture quality is highly dependent on the way in which the bitrate is controlled.

It seems logical that some form of controlling is needed, with a possible liaison with the encoder policing function. The knowledge of the network parameters can be used when controlling the codec.

In most of the coding schemes operating at CBR the output bitrate (or buffer regulation) is controlled by acting on the quantiser stepsize. This is however not the only way to achieve the control of the bitstream. In particular for VBR coding other coding control strategies are possible, influencing the quality of the coded sequence. By means of temporal or spatial subsampling the bitrate regulation can be achieved. Earlier results indicate that there are no advantages in performing the bitrate control by that mean with respect to the variation of the quantiser stepsize.

6.6 Impact of some network parameters on the picture quality

The cell loss ratio has a strong impact on the loading of the network. What is the trade off between efficient coding (probably low cell loss ratio required) and error resilient coding schemes (probably low coding efficiency, but accepting high cell loss ratio) in terms of quality (assuming the bandwidth to be allocated by the network is used as reference)?
Does the effect of overload - where low priority cells (the second channel) are trown - have a strong impact on the overall impression of the image?

6.7 Results and conclusion of simulations

6.7.1 One Layered

Simulations show no significant difference between VBR and CBR when equivalent bitrates are used. Differences can be seen when sequences contain violent motion, scene cuts or zoom.

6.7.2 Two Layered

In most simulations with two layered coding schemes the base channel is CBR. If the second channel is coded with CBR an annoying "Twinkling effect" can be seen. If the second channel is coded with VBR this effect does not occur.

6.8 Conclusion

Up until now the simulations have mainly dealt with improvements of different VBR coding schemes. However, one of the main questions to be answered still remains :

VBR or CBR for video transmission in ATM networks?

So far, rather few quality comparisons of CBR vs VBR coding schemes have been made under the condition of equal network load. It is therefore too early to give a firm answer to this question.

6.9 Related recent COST 211bis/ter papers :

(1)	SIM/89/39	(Tech. Research Center of Finland) VBR video codec output simulator based measurements
(2)	SIM/89/43	(IST/JNICT + CSELT) Extension of CCITT Visual Communication Codec for operation in ATM networks
(3)	SIM/89/44	(CNET) First attempts of picture coding on ATM networks
(4)	SIM/89/49	(TELEFONICA) A quasi - VBR model for packet video
(5)	SIM/89/53	(BTRL) Quantisation of the second layer of an adapted RM8 two-layer codec
(6)	SIM/89/57	(INESC) Is adaptive quality a viable solution
(7)	SIM/89/65	(STA) Comments on picutre quality and variable bitrate
(8)	SIM/89/69	(IST/JNICT + CSELT) "Constant Quality" Coding in ATM networks : Simulation Results
(9)	SIM/90/77	(Siemens AG) Variable Bitrate Image Coding for "Constant quality", Preliminary results
(10)	SIM/90/78	(BTRL) A comparison of coding performance between 1-Layer and 2-Layer codecs
(11)	SIM/90/80	(UCL) Comparison of RM8 in close and open loop; measurements and simulation
(12)	SIM/90/86	(PTT RNL) A comparison of variable versus constant bitrate and the impact on the picture quality for fixed frame rates
(13)	SIM/90/93	(CNET) 2 layered coding scheme simulations
(14)	SIM/90/98	(NT) VBR video coding and policing functions
(15)	SIM/90/99	(STA) Comparing VBR and CBR coding schemes
(16)	SIM/90/104	(FI beim FTZ) Bitrate control in VBR codecs

(17)	SIM/90/105	(PTT RNL) Variable frame rate; Picture quality and Bitrate
(18)	SIM/90/109	(UCL) Picture quality criterions for image data compression (Bitrate versus quality)
(19)	SIM/90/114	(BTRL) Policing functions for ATM networks
(20)	SIM/90/119	(INESC) Policing in B-ISDN
(21)	SIM/90/123	(TELEFONICA) Influence of consecutive cell loss in packet video
(22)	SIM/90/129	(STA) The impact of some parameters on the occupancy of the network
(23)	SIM/90/132	(STA) Some network load calculations based on BTRL simulations
(24)	SIM/90/133	(Siemens AG) Comparison of VBR with CBR coding, evaluation of the statistical multiplexing gain
(25)	SIM/90/137	(BTRL) Further studies into the effects of one layer and two layer video coding on loading of ATM networks
(26)	SIM/90/142	(BTRL) Comparison of CBR and two layer VBR for the same network loading
(27)	SIM/90/145	(Siemens AG) The influence of noise on the code bit rate of an open loop H.261 coder
(28)	SIM/9/149	(CNET) Comparison between two layer and single layer video coding
(29)	SIM/90/151	(IST/CSELT) Bitrate optimization on a two layered constant quality video coding scheme

Chapter 7

Guidelines for simulation

7.1 Introduction

For simulation of video coding for ATM networks many parameters can be used. Some parameter which are relevant for the comparison of the different coding schemes are listed here (see chapter 6):

1. Parameters specifying the codec output
   - average bitrate
   - burstiness or peak bitrate
   - variance
   - burst duration and frequency
   - quality
2. Parameters specifying the network
   - link capacity
   - cell loss ratio
   - queing technique
   - buffersize (delay)
   - policing function
   - tariffing
3. Parameters important for or resulting from both network and codec
   - QoS required
   - Priorities required
   - Policing function required at the encoder

- Measuring period of all statistical parameters

To be alble to compare results in this chapter some guidelines are given for most of these parameters.
The following transfer situations may be assumed in an ATM network:

1. One channel with a certain Quality of Service (QOS)

2. Two channels with different QOS, resulting in a layered coding scheme. This will only be useful if the cell loss rate of the two channels differ significantly.

In the case that one channel is used a simple model can be defined. This model is depicted in figure 7.1 and 7.2.

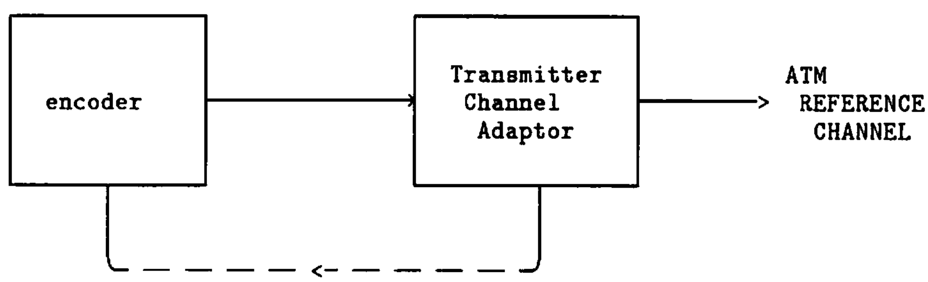

Figure 7.1: ATM Reference Encoder

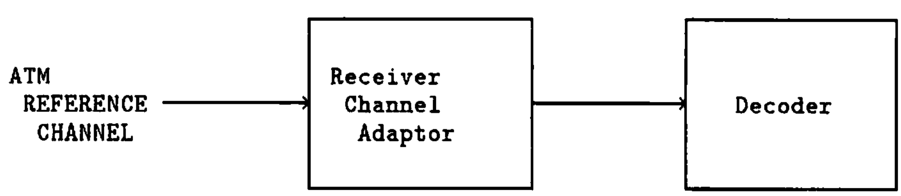

Figure 7.2: ATM Reference Decoder

7.2 Reference Coding algorithm

For simulations H.261 Encoder Reference Model will be used, as described to be discribed as a modification to RM8 (according to modified doc #COST211ter SIM(90)101).[1]

[1]A complete reference model for H.261 encoding (based on RM8) will be available in jan 91

In the case of two layered coding the base layer is encoded according to this reference coding algorithm.

For simulations outside the scope of H.261 other coding algorithms can be used.

7.3 Transmitter channel adaptor

The control process adapts the encoder to the ATM channel. It performs packetization of the bitstream at the encoder and controls parameters of the encoder (e.g. stepsize).

7.4 ATM reference channel

7.4.1 Description

The ATM reference channel is a model based on in a B-ISDN Class B service (see chapter 2). In simulations measuring e.g. burstiness and the impact of cell loss the channel parameters "cell length" and "cell los rate" need to be defined.

7.4.2 Cell length

The cell-size for ATM is 48+5 octets. The ATM -payload consists of 48 octets. Each cell has to spend octets for SAR-header and a SAR-trailer.

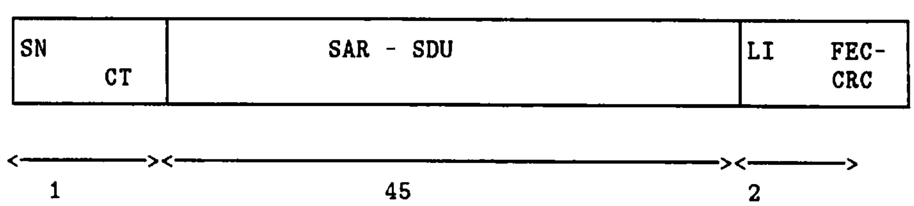

Figure 7.3: ATM cell information field

For simulation purposes it was agreed to use:

- 1 octet SAR Header
- 2 octets SAR Trailer
- 45 octets information (SAR-payload)

As the AAL has not yet been fully defined, for simulations the Convergence sublayer is not take into account. This results in zero bits for CS-header, CS-pads and CS-trailer; the User PDU length is assumed to have no upper limit.

7.4.3 Cell loss rate

Simulations can include a high and a low cell loss rate.

- $P_{loss} = 10^{-3}$ for high cell loss rates.
- $P_{loss} = 10^{-8}$ for low cell loss rates.
  N.B. For simulations the low cell loss rate can be assumed to be: $P_{loss} = 0$

7.5 Receiver channel adaptor

On the receiver side the channel adaptor consists of at least a depacketizer. It can also detect Cell loss and give a "cell loss flag" to the decoder.

7.6 Reference Decoding algorithm

For decoding the H.261 decoder is used. In the case of two layered coding the H.261 is a fall back mode (base channel)
For simulations outside the scope of H.261 other decoding algorithms can be used.

7.7 Measurement of statistics

7.7.1 Burstiness

One of the items that seem useful to be measured or limited on a VBR-codec is the burstiness of the bit stream or cell-stream that is produced, but there is no single definition for the burstiness.

For encoder control the maximum number of cells per GOB is defined to limit burstiness.

- measure the cell rate and bit rate for each frame, GOB and MB.
- Optional: measure momentary cell rates.

  The momentary cell rate at cell N is defined as:

  If a cell arrives at time $= t_i$ and the next cell arrives at time $= t_{i+1}$, the momentary cell rate during the time-interval $\langle t_i, t_{i+1}]$ is defined as $r_{mom} = 1/(t_{i+1} - t_i)$.

  The time base that is used is the average time for a macro block.

7.7.2 Sequences

Three different services can be distinguished:

- Conversational
- Retrieval and messaging
- Distribution

These services all require typical sequences for simulations.
Agreed is to use:

sequence	frame rate [Hz]
SWING	30
SALESMAN	30
TABLE TENNIS	25
DAWN	25
JACK IN THE BOX	30

7.7.3 Simulation parameters

Four different bit rates are considered for simulations

mean Bitrate [kbit/s]	Framerate [frames/s]
64	10
320	15[/30]
1024	30
1856	30

For the 25 Hz sequences the frame rates 10,15 and 30 Hz become resp. 8.33, 12.5 and 25 Hz.
For 2-layered coding the mean bitrate in the two cannels has to be:
CHANNEL 1: 50% of total (mean) bitrate
CHANNEL 2: 50% of total (mean) bitrate

7.7.4 Network Loading

An important parameter for simulations is the total number of calls. For the loading of networks a number of formulas can be used. In apppendix B the network models that are being used are listed.

7.7.5 Presentation of results

Depending on the type of simulations that are being performed, the results can be presented with:

- SNR plots
- Plot of Number of bits per frame
- Statistics see in table 7.1
- burstiness
    1. mean and peak number of cells per macroblock
    2. mean and peak number of cells per group of block
    3. mean and peak number of cells per frame
    4. optional: histograms of
        - the momentary cell rates
        - the number of cells per GOB

  (For items 1,2 and 3 see example table 7.7.5)

Coding results have to be presented on UMATIC TAPE.
It is mandatory to show the results full screen (also when a comparison of two sequences is made).
In the case of two layered coding the fall back mode (base channel) result has to be shown.

Sequence :	Institute :
Modification :	Date :
Number of tracks :	Temporal resolution :

Item		layer 1	[layer 2]
SNR for luminance SNR for chrominance(U) SNR for chrominance(V)			
Mean value of step size			
Mean value of the number of non-zero coefficients			
Mean value of the number of zero-coefficients			
Block type of Macro	Fixed		–
	Inter+coef		–
	Inter+Coef+MC+Fil		–
	Inter+MC+Fil		–
	Intra		–
	Inter+Coef+MC+Fil+Q		–
	Inter+MC+Fil+Q		–
	Intra+Q		–
	Inter+Coef+MC		–
	Inter+MC		–
	Inter+Coef+MC+Q		–
mean number of bits/frame			
Bits for first frame			
Number of forced to fixed mb's			–

Table 7.1: The form for the assesment of the coding results

		Number of Cells		
		max	mean	max/mean
Dawn 25Hz Fixed Quantizer 8	mb	4	0.28	14.53
	gob	31	9.09	3.41
	frame	285	109.05	2.61
Dawn 12.5Hz Fixed Quantizer 12	mb	3	0.18	16.94
	gob	24	5.85	4.11
	frame	213	70.15	3.04

Table 7.2: Mean and peak cell rate example

Chapter 8

Guidelines for Hardware experiments

At the time of writing (17 May 1990) there were no coordinated activites in the COST 211bis Hardware Subgroup towards ATM experiments. At the Hardware Subgroup meeting on 16 May some organisations outlined their individual activities or plans:

In Belgium several establishments are involved in work for an experimental broadband network. Philips MBLE are building a HDTV codec which should be completed in about 2 years. This is a two layer codec, with the first layer coding CCIR 601 and the second layer providing the increased resolution for HDTV. Other companies are working on CCIR 601 and high quality videophone codecs, but these are not H.261 compatible.

Also in Belgium, UCL would like to build an ATM sub-band coder for TV/HDTV as part of COST 211.

BTRL is close to having a functional experimental VBR video codec based on a two layer approach with H.261 in the base layer as described earlier in the compendium. Interfaces have been designed to connect this codec to a demonstrator of BT's ATM switch employing the Orwell protocol.

PKI expressed their wish to adapt H.261 codecs to operate on ATM networks. The timescale is the end of 1991. Preference is for the single layer approach to give simpler hardware and easier interconnection with fixed rate versions of H.261 codecs.

CSELT intend to adapt an existing hardware to variable rate when manpower resources become available on the completion of other projects.

In Spain a project is underway to test services such as videoconference, TV distribution and videotext on an experimental network. This may be ready at the end of 1991.

Chapter 9

Summary

Chapter	Task leader	Company	Contributors	Company
1	Ronald Plompen Dolf Schinkel	PTT Contest PTT RNL		
2	Jurgen Pandel	Siemens	Antonio Gaspar Miguel Roser Ian Parke	INESC-NORTE Telefonica BTRL
3	Ian Parke	BTRL	Antonio Gaspar Jurgen Pandel Gerard Eude	INESC-NORTE Siemens CNET
4	Gerard Eude	CNET	Mauro Quaglia Norbert Diehl Dolf Schinkel Luc Vandendorpe Miguel Roser Ian Parke Jurgen Pandel	CSELT AEG RNL UCL Telefonica BTRL Siemens
5	Helge Sandgrind	NT, Norway	Jan Schneider Jurgen Pandel Miguel Roser Peter List Jurgen Pandel	SEL Siemens Telefonica FI Siemens
6	Christel Verreth	STA	Norbert Diehl Dolf Schinkel Luc Vandendorpe Per Tholin Jan Schneider	Daimler Benz RNL UCL STA SEL
7	Dolf Schinkel	RNL	Ian Parke Bruno Loret	BTRL CNET
8	Geoff Morrison	BTRL		

Glossary

AAL layer ATM Adaptation Layer, layer of the protocol reference model between the ATM layer and the service user layer, *2.1.3*

ATM Asynchronous Transfer Mode. A kind of fast packet switching with cells (=packets) of fixed length and routing after call setup, *2.1*

ATM layer lower layer of the protocol reference model defining the cell processing at the network termination, *2.1.3*

B-ISDN Broadband Integrated Services Digital Network

BER Bit Error Ratio. Mean ratio of erronous received bits and total number of transmitted bits over a communication link.

BSO Bit Stream Oriented, a specific service attribute

CBR Constant Bit Rate, the same meaning as FBR

Cell basic information unit of fixed length of 48+5 octets (bytes), *2.1.1*

Cell Delineation operation performed to detect the cell boundaries, *2.1.3*

Cell, information field comprises 48 octets (bytes) for user information including service dependent control information (see also AAL), *2.1.1*

Cell, header field comprises 5 octets (bytes) for identification and cell function specification, *2.1.1*

Cell, segment number number which is incremented from cell to cell (e.g. modulo 64, [7]) to detect lost cells at the recceiver. The segment number will be processed within the AAL.

CL Connectionless Service; service that provides the means by which network entities can exchange data units without the establishment of a data connection.

CO Connection Oriented Service; service which provides the means for establishing, using, resetting, and terminating data connections

CRC-field cyclic redundancy check field; PDU field at the SAR sublayer for bit error control

CS Convergence Sublayer, sublayer of the AAL for VBR services, *2.1.3*

Distribution Service service characterized by the unidirectional flow of information from a given point in the network to other (multiple) locations. Distribution services are divided into two classes: without user individual presentation control and with user individual presentation control.

Error Concealment technique to conceal the visibility of lost data segments at the decoder by interpolation methods or partly freezing the previous image

FBR Fixed Bit Rate, throughput characteristic in which a constant transmission capacity is provided to the user during a call, *2.1.4.3*

GFC Generic Flow Control, field of 4 bits within the cell header, it 2.1.1

HEC Head Error Control, field of 8 bits within the cell header for bit error correction within the cell header, *2.1.1*

Interactive Service service class which provides the means for bidirectional exchange of information between users and hosts. It is subdivided into three classes: conversational services, messaging services and retrieval services. Layered coding class of hierarchical coding algorithms using different layers of resolution, *2.2.1*

Messaging Services interactive service class which offers user-to-user communication between individual users via storage units with store-and-forward, mailbox and/or message handling (e.g. information editing, processing, and conversion) functions.

NNI Network Node Interface, interface between two network nodes, *2.1.2*

PDU see "Protocoll Data Unit"

PO Packet Oriented, a specific servive attribute

Policing (function) method for supervision of the allocated bandwidth parameters during a call, *2.1.5*

Priority Channel ATM channel with very low rate of cell loss compared to a nonprioritized channel

Protocol Data Unit (PDU) payload of the covergence sublayer plus additional fields corresponding to specific functions

Protocol Reference Model Model of planes and layers for functional definition of the B-ISDN, *2.1.3*

PT Payload Channel Identifier, field of 2 bits within the cell header, *2.1.1*

Quality of service ATM networks will offer different classes of transmission quality, which can be chosen according to the respective service requirements. Characteristic parameters could be cell loss ratio, delay, and diverse throughput parameters, *2.1.4*

QOS see "quality of service"

Retrieval Service interactive service class which provides the capability of accessing information stored in database centres. The information will be sent to the user on demand only. The information can be retrieved on an individual basis, i.e., the time at which an information sequence is to start is under the control of the user.

SAR Segmentation And Reassambly sublayer. Sublayer of the AAL, *2.1.3*

Statistical Monitor Functional block of an encoder with variable bit rate to police the statistical parameters of the rate which have been allocated at call set-up, *2.2.1*

Statistical Multiplexing time-multiplexing of several VBR sources on a common link of transmission capacity less than the sum of the individual peak rates

STM Synchronous Transfer Mode. Transfer mode in which the transmission and switching functions are achieved by permanent allocation of channels/bandwidth between the connections.

SDH Synchronous Digital Hierarchy, a new transfer mode for hybrid schemes of ATM and STM networks, *2.1.2*

SN-field sequence number field; PDU field at the SAR sublayer for detection of lost or inserted cells

TA Terminal Adaptor, required for the interim situation between a video codec which is not a B-ISDN device and the network [(89)72].

TS-field time stamp field; PDU field at the CS sublayer for end-to-end synchronization

UNI User Network Interface, interface that connects the user to the network, *2.1.1*

VBR Variable Bit Rate, throughput characteristic in which an average transmission capacity with limited dynamical changes is provided to the user during a call, *2.1.4.3*

VCI Virtual Channel Identifier, field of 12-16 bits within the cell header, *2.1.1*

Virtual channel logical unidirectional association between the end points of a (physical) link that enables the transfer of cells, *2.1*

Virtual Path a group of virtual channels treated as a logical entity, which can be used, for example, to create virtual private networks or to separate traffic of different types (QOS).

VPI Virtual Path Identifier, field of up to 12 bits within the cell header, *2.1*

Appendix A

Network Models

A.1 On/Off Sources

The source is assumed to be alternatively active, transmitting at a peak rate of P, and silent. The relative durations of the on and off periods is chosen such that the mean rate of the model matches the mean rate of the signal. Further information can be found in COST211bis Sim 90/121. No account is made of the variance of the signal.

A.2 Gaussian Model

A Gaussian model for the superposition of all sources takes into account the variance of the signals. The mean and variance characterizing the Gaussian model are obtained by direct summation of individual source means and variances. Such a model is described in COST211 bis Sim 90/149.

A.3 Other Models Using Variance of the Source

Two other models have been used which are based on the measured variance and the average bit rate of the sources. These are documented in COST211bis 90/132.

Note All these models are for a single service, ie. video, on the ATM network.

Appendix B

Agreements, Questions and Action Points of COST-211ter

B.1 Agreements

It is agreed to continue working on different coding schemes before the start of the CCITT SG XV on ATM, so the following algorithms should be investigated in order to have common views on their performances:

- VBR derived form RM8
- Quasi-VBR scheme (by switching the p-value)
- 2 layers algorithms:
  - RM8 + VBR
  - VBR (or CBR)+ VBR by selecting the transmitted coefficients
  - transmitting subsampled frames in the 2nd layer

In order to have fair comparisons, we should stick to the guidelines for simulations depicted in the chapter 7 of the compendium.
It appears that it is not fair comparing VBR vs CBR using the same overall mean bitrate.

It was agreed that at least a basic policing function is required.

B.2 Questions

- Will policing function be implemented in the network?
- Will policing function be implemented in the users terminal?
- What is the optimum time period for the measurement of the mean bitrate?

- In which fraction of this time is the peak bitrate allowed?
- Do we need error correction in the SAR-SDU?
- How to split the bit rate between 2 layers
  - if the quality is kept constant
  - if we consider the network load
- How to compare schemes (bit rate (1 layer; 2 layers), subjective quality, objective quality, network load, VBR vs. CBR)
- Do we have to use FEC in the payload or (only) use AAL error correction?
- How to control VBR schemes?

B.3 Action points

- Find algorithms to cope with cell loss in 1 layer codecs?
- Buffer regulation in VBR 1 layer codecs.
- Policing functions = possible strategies depending on the network and coder capabilities.
- Define a fair comparison between VBR and CBR schemes: Define the parameters to be specified to the network.
- Loading of the network vs coding algorithm: Confirm the network model.
- *(Video compression using human visual perception)*
- AAL type 2:
  - What has to be specified by NA5?
  - How many bits for serial number?
  - What about serial number protection?
- H.261 adaption for CCIR 601 format
- Cross check the new 2 layer coding scheme
- Write a new specification for the reference model