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Expert Group for ATM Video Coding

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Title: Features of MUPCOS
Purpose: Discussion, Information

Introduction

The MUPCOS is a highly flexible system which can suit the needs for many different applications. Different applications have different requirements. The requirement claims met by MUPCOS are given. MUPCOS is developed by PTT Research and LEP-Philips (Paris). Part of this work has been done in the VADIS/COST 211-ter collaboration, and part of this work has been done in the RACE HIVITS WPA1.5 project.

1 Requirements of different applications

In document AVC-109 a provisional list of requirements for the H.32x terminal are given. In the MPEG Packet Proposal Description a number of requirements for different applications are listed, the requirements with the most important implications on the algorithms are: compatibility, low codec delay, cell loss resilience and scalability. Constraints concerning for instance bit error resilience can mostly be handled with a FEC.

1.1 Compatibility

A very obvious requirement is the **compatibility requirement**, this is a must for for instance:

The conversational services, **H.261 and H.320 compatibility**:
- Future systems should be able to interwork with existing systems
- Stable investment in equipment
- A stable platform for the market

Interworking between H.32x and H.320 terminals is outlined in figure 1.

The so called 'compact disk interactive' systems, need MPEG1 compatibility to have a:
- Stable investment in equipment
- Stable investment in software (compact disks)
- A stable platform for the consumer market
- A compatible product range with increasing performance which can be offered to the customer

For those applications not needing this compatibility, it might be too a heavy burden in price or maybe in performance. Therefore the MUPCOS allows to switch off the compatible

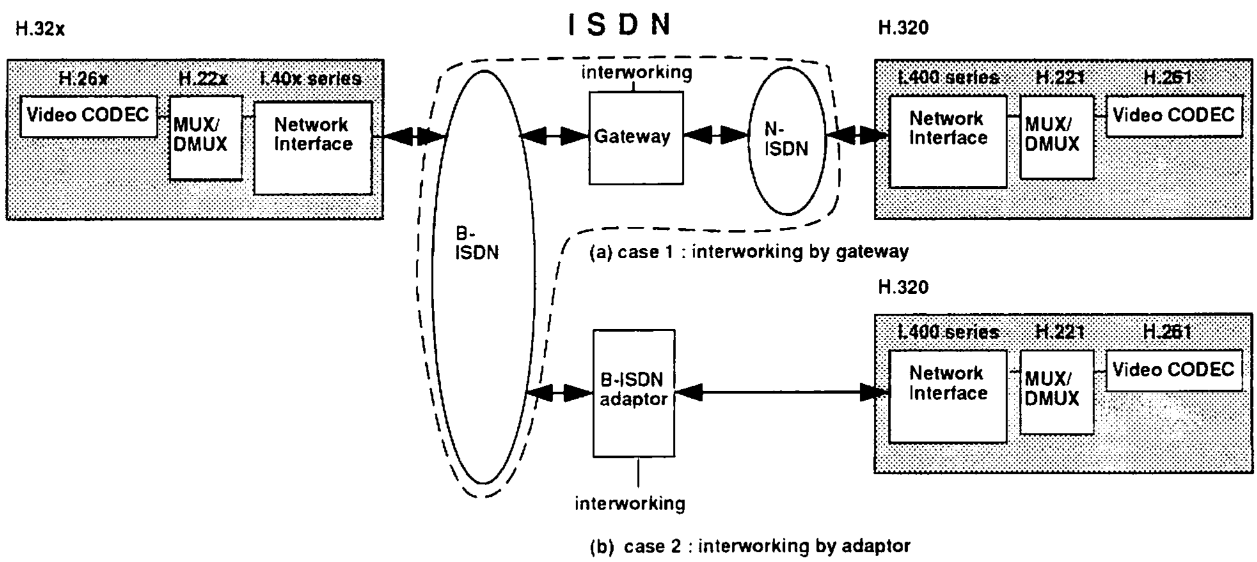

Figure 1 : Two cases of interworking between H.32x and H.320

mode in the encoder and the decoder, so for those applications it is not necessary to implement this.

1.2 Low end to end delay

A second important requirement is the **low delay requirement**, this requirement is a must for for instance:

The conversational services:
- For proper videophone use or video conferencing low delay is required.
- The total end to end delay should be low, desirably less than about 150 ms. This means that the maximum delay should not too often exceed this value, and that is very desirable to offer the customer a delay which is much less than the above mentioned 150 ms.

For these applications there is always the trade off between picture quality and delay. The less delay, the less the image quality is. The reason for this as follows due to less delay the future and past time windows will be small, with a small time window low correlation in picture sequences will be found. The advantage of low delay systems is that when the time window gets smaller the memory to store the information also will reduce. The MUPCOS can be switched, in a similar way as MPEG1, to a mode which only uses P (forward predict only) fields. This mode is similar to H.261. By using field coding it can be demonstrated that it can result in lower delays than by using frame coding.

A model to visualise the different components of the total end to end delay is given below:

Examples for the items given in table 6.1 are for a 50Hz interlaced source and a 50Hz processing format:
- Format conversion: no format conversion is required; $d_{fc} = 0$ ms.

— Line to Block conversion: d_{lb} = 1 ms.
— Initial processing delay, this time is required to retrieve for instance some motion vectors or perform an image analysis: d_{pe} = about 5 ms to 25 ms.
— Frame/Field reordering delay:
  • For frame coding: d_{re} = 20 ms.
  • For field coding: d_{re} = 0 ms, if no interpolation is used.
— Buffer delay: this delay plus the buffer delay from the decoder are almost constant.
— Network delay: depending on the infrastructure in the range of about 1 ms to several 100s ms. If satellite links are used, network adaptation is included. In this example we will assume d_n = 0 ms.
— Buffer delay: the total encoder and decoder delay equals to the total number of bits in the encoder and decoder buffer over the average bitrate when a constant bitrate is assumed, so $d_b = d_{bc} + d_{bd}$. The H.261 hypothetical reference decoder specifies a B value which is a measure for the buffer delay: $B = 4R_{max}/29.97$, where R_{max} is the maximum video bit rate to be used in the connection.
  For this example the formula can be modified to: $B = F*R_{max}/50$, where F is a peak factor.
  The buffer delay bd is then, combining the two formulas, in the worst case about $(F*R_{max}/50)/R_{max}$ seconds, for F=4, d_b = 80 ms.
— Frame/Field reordering delay:
  • For frame coding: 20 ms.
  • For field coding: 0 ms, if no interpolation is used.
— Initial processing delay: about 1 ms.
— Block to Line conversion: 1 ms.
— Format conversion: no format conversion is required; 0 ms.

The biggest delay in this example is the buffers delay, it is suggested to take this item into account in further studies.

The typical maximum total delay is, for a frame coding algorithm, in this example about 128 ms to 148 ms. The typical maximum total delay for a field coding algorithm is in this example about 88 ms to 108 ms.

Conclusion: for the end to end delay it is important to select the processing format close to the source input format.

1.3 Cell loss resilience

A third important requirement is the **cell loss resilience** requirement, this requirement is a must for applications in the ATM network.

The future communications network will be the Broadband Integrated Services Digital Network (B-ISDN). This network will be of the Asynchronous Transfer Mode (ATM) type. Some of the characteristics of such networks are: low delay and packet switched. These packets are called cells in the B-ISDN network, these cells have a netto size of 386 bits, and one of the consequences of the low delay is that cells can be lost in the network.

Video services which use the network should be able to cope with these cell losses and therefore should be able to handle the loss of non recoverable data chunks of at least 386 bits.

The MUPCOS uses a hierarchical coding structure. Utilising this in a layered coding system the visibility of errors can be effectively reduced and recovery from errors can be achieved.

1.4 Scalability

An other requirement is the **scalability** requirement, this requirement is of interest for window oriented applications.

MUPCOS is by nature a scalable system, which comes implicit from the forward and backward compatibility, which makes the MUPCOS also upward and downward compatible.

2 The H.261 forward and backward compatibilty claim

The H.261 compatibility claim is demonstrated for practical reasons with a MPEG1 codec in the base layer in the only forward prediction mode, this mode is similar to H.261. See also figure 2, the low delay claim and the cell loss resilience claim.

3 Forward, backward compatibility and scalability claim

The 14 sequences coded for the TEST are all coded compatible to the MPEG1 core. For the 4 Mbit/s coded sequences, 1 Mbit/s are used for the MPEG1 and 3 Mbit/s for the enhancement data. For the 9 Mbit/s coded sequences, 1 Mbit/s is used for the MPEG1 and 8 Mbit/s is used for the enhancement data.

These claims are demonstrated with the following processed sequences:
- Susie coded at 1.5Mbit/s in the MPEG1 channel, and with a total bit rate of 5Mbit/s.
- Table Tennis coded at 1.5Mbit/s in the MPEG1 channel, and with a total bitrate of 5Mbit/s.
- Flower Garden coded at 1.5Mbit/s in the MPEG1 channel, and with a total bitrate of 5Mbit/s.

In the demonstration the sequences are shown decoded with the MUPCOS decoder and with a MPEG1 decoder. The MPEG1 quality is of about the quality demonstrated of the so called 'SM3+' software simulations in previous MPEG meetings.

4 Low delay claim

This claim is demonstrated with the following processed sequences:
- Susie coded at 1Mbit/s in the MPEG1 channel, and with a total bitrate of 4Mbit/s.

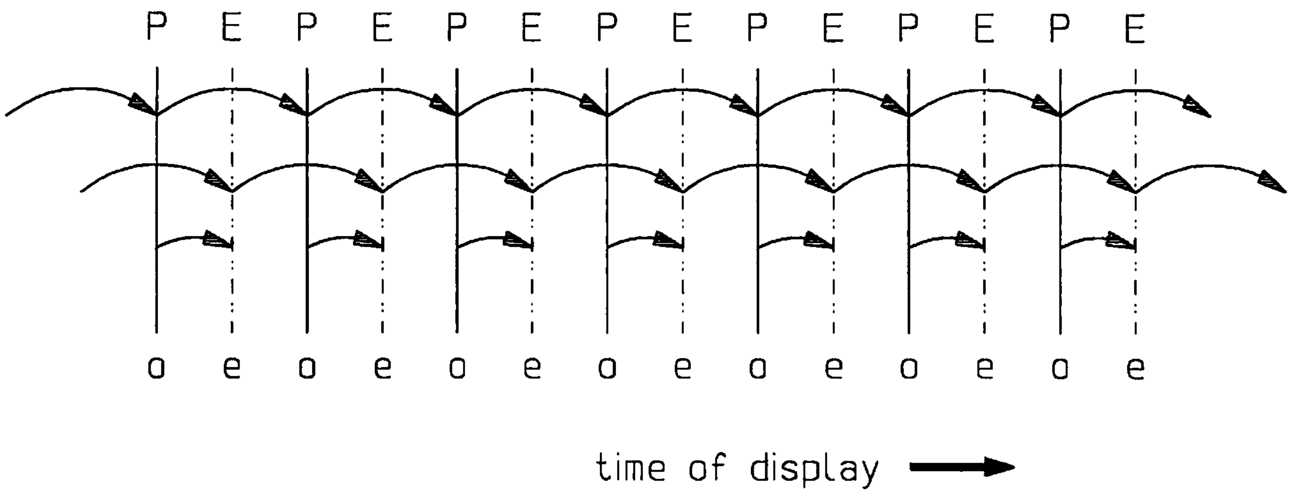

Figure 2 : Low end to end delay prediction information flows

— Dawn coded at 1Mbit/s in the MPEG1 channel, and with a total bitrate of 4Mbit/s.
— Table Tennis coded at 1Mbit/s in the MPEG1 channel, and with a total bitrate of 4Mbit/s.

Note: Dawn is a sequence originated by PTT Research that has been used in COST 211-ter for ATM studies.

The forward prediction information flow compatible to MPEG1 is shown in figure 2.

The delay for the MUPCOS in the compatible low delay mode is given in table 1.

Format conversion	0 ms
Line to Block conversion	1 ms
Initial processing delay	5 ms
Frame/Field reordering delay	0 ms
Buffer delay	20 ms
Network delay	0 ms
Buffer delay	20 ms
Initial processing delay	1 ms
Frame/Field reordering delay	0 ms
Block - Line conversion	1 ms
Format conversion	0 ms
Total	48 ms

Table 1 : MUPCOS end to end delay

One of the advantages low delay codecs can have is that less memory is required for the implementation of these systems and also the computational performance requirements are less. The memory requirements for the MUPCOS in this low delay mode is 3 CCIR 601 field memories for the encoder and 3 CCIR 601 field memories for the decoder.

5 Cell loss resilience claim

To overcome the effect of cell loss in an ATM network the MUPCOS can be used in a two layer coding configuration.

The first layer contains a base picture which is transmitted over a guaranteed channel. The second layer contains enhancement data which are transmitted over a second channel. This channel could undergo cell loss.

This claim is demonstrated with tho following processed sequences in the low end to end delay mode:

- Susie coded at 1Mbit/s in the MPEG1 channel and with a total bit rate of 4Mbit/s.
- Dawn coded at 1 Mbit/s in the MPEG1 channel and with a total bit rate of 4Mbit/s.
- Table Tennis coded at 1Mbit/s in the MPEG1 channel, and with a total bit rate of 4Mbit/s.

The target value for cell loss for B-ISDN as being standardised by CCITT SG XVIII is not yet clear. Cell loss probabilities could be in the order of 10^{-3} for low priority and 10^{-8} for high priority (utilising the priority bit, CLP, in the cell header). Recent developments

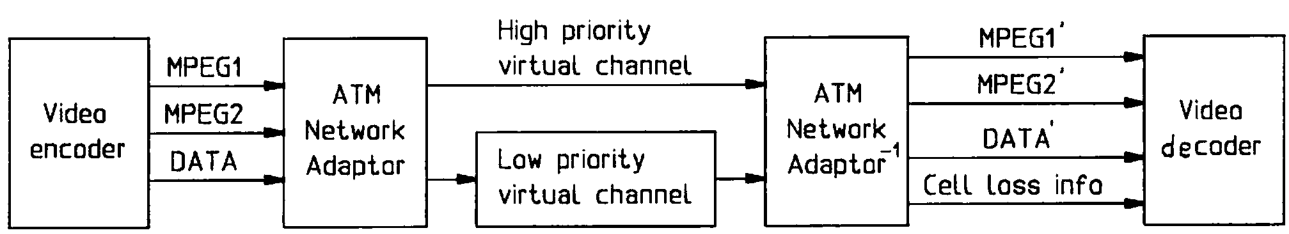

Figure 3 : Example of encoder, network, decoder diagram

In figure 3 an example of an encoder, network and decoder block diagram is given. The ATM Network Adaptor handle the packetisation of the MPEG1 and MPEG2 video bit streams. Note: the DATA channel is not transmitted over the ATM, and thus is DATA' not used in the simulations. The Low priority virtual channel is simulates the cell losses as specified in the PPD.

indicate a move towards better quality with even lower cell loss probabilities.

In the simulations the video data is transported via two constant bit rate virtual channel, the first one carring all high priority cells, the second one carring low priority cells with once in a while a high priority cell.

The simulations used a cell loss rate of 10^{-3} for the total cell stream, and the cell loss only occurs in the enhancement channel. A model of the encoder, network and decoder is shown in figure 3, a standard random generator is used to determine the lost cells. The high/low priority and cell loss ratios for the sequences coded, 1 Mbit/s in the base layer and totally with 4Mbit/s is shown in table 2.

High and low priority channels	
Cells high priority channel/total cells	0.25
Cell loss probability	10^{-3}
Cells high priority/total cells	0.34
Low priority channel	
Total bit rate	3 Mbit/s
Num cell CLP = High/Total cells	0.12
Cell loss probability	$1.5*10^{-3}$
High priority channel	
Total bit rate	1 Mbit/s
Cell loss probability	10^{-8}

Table 2 : Cell loss statistical information used in the simulations

Demonstrated is the effect on the full resolution pictures. Cell loss gives disturbances in blocks, macro blocks or slices. The decoder however recovers from these losses within a few frames using the MPEG1 prediction. This simple method does however not fully utilize the potential in cell loss resilience. More advanced methods which will not increase the complexity are under study.