CCITT SGXV
Working Party XV/1
Experts Group for ATM Video Coding

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SOURCE : Japan
TITLE : Revision to the Annex 3 to Doc. AVC-22R
(Annex 3 : Liaison statement to SGXVIII on requirements for B-ISDN network model as it impacts on video coding)
PURPOSE : Proposal

1. Background

The annex 3 to Doc. AVC-22R had been sent to SGXVIII Matsuyama meeting. However, the answer is not at sufficient level. Besides, next SGXVIII meeting will be held from June 11 to 25. We think that further information is effective to accelerate the discussion in SGXVIII.

2. Proposal

It is proposed that some information, such as the following underlined sentences, should be added to the annex 3 to Doc. AVC-22R.

Addenda

Ad.1 Cell loss ratio

Various cell-based video coding systems have been developed or simulated, collectively capable of satisfactory performance in the face of a variety of cell loss ratios. However, the actual figure to be expected from the network for a particular video service application and bit rate will determine both the need for cell loss protection or recovery and the method to be used. The Experts Group expects to identify appropriate cell loss ratios for video services on the B-ISDN, and will input this information to SGXVIII when available.

The cell loss ratio has fundamental implications for the video coding strategy and its efficiency. If, for example, layered video coding systems are to be used, exploiting the availability of the cell loss priority indicator, an indication of the cell loss rate for each priority level is necessary. Figures for expected cell insertion rates are also required.

a. High quality class (low cell loss ratio)
There exist several techniques including layered coding to cope with cell loss. However, these techniques cause increase of the hardware. Besides, for the purpose of videoconference the picture degradation once in 10 hours is considered to be permissible. Therefore, if the cell loss ratio is smaller than 10^{-10}, the special techniques for cell loss tolerance are not necessary even for 100Mbit/s TV codec. Is the cell loss ratio for high quality class smaller than 10^{-10} ?

b. Low quality class (high cell loss ratio)
Liaison statement from SGXVIII (Traffic control and resource Management Aspect) requires the input about the cell loss ratio for CLP lower priority. The three cases are considered for low quality class.

- case 1 : cell loss ratio is assured
  In this case, if network charge is considerably cheaper than that of high quality class, video codec may use it.
- case 2 : cell loss ratio is not assured
  In this case, users cannot estimate the picture degradation.
  Therefore, video codecs cannot use it.
- case 3 : time percentage with assured cell loss ratio is specified.
  If the percentage of 'not assured' time is sufficiently small (for example, corresponding to one short burst in ten hours) and in 'assured' time cell loss ratio is small enough (for example 10^{-10}),
  user will be glad to use this class.

Whether codec use this class or not depends on the network charge (resource). What is the difference between case 1 and case 3 from the standpoint of the network resource?

Ad.2 Cell loss burst characteristics

The question of whether cells are lost in isolation or in bursts is fundamental for the video coding approach. Some coding schemes are proposed which provide a means of protecting against bursts of cell loss, but they may not be necessary if cells are lost in isolation (i.e. if cell losses are uncorrelated).

-Will cell loss be dominated by network congestion?
-Will bursts of cell loss result from network congestion?
-Will the cell loss burst length be service rate dependent?
-Will high priority cells be affected by network congestion?

Clarification of these points is sought from SGXVIII, and additional guidance to an appropriate statistical model to characterise bursts of cell loss would be welcome.

The Liaison statement from SGXVIII (Information on ATM network performance) says that the two simulation results show the followings:
(1)Cells tend to be lost consecutively.
(2)The cell loss process is not random process and may be described by the Gilbert model.
In that situation, how can we estimate the average interval time T in which no cell loss occurs? If T>>1/((bit rate)x(cell loss ratio)), our requirement for cell loss ratio will be relaxed.

Ad.3 Use of CLP indicator

The use of CLP is useful for some coding schemes to provide tolerance to cell loss.

-Under what circumstances would the service provider set the CLP indicator?
-Could the CLP be changed by the service provider after a user has set it?
If CLP bit is not changed in the network, layered video coding can use it as layer indicator.
-Will the usage monitoring structure encourage the use of both high and low priority cells?
-Will the rate of high and low priority cells be negotiated independently with the network?
-Will any relation exist between VBR/CBR services and high/low quality classes?
-What kind of negotiation should be done to use CLP bit?
Some kind of layered coding techniques can control the total rate but cannot control the rate of both priority classes accurately. In such case, for example, the following negotiation can be considered :
*maximum 8Mbit/s
*average 5Mbit/s
*The rate of low priority cells can fluctuate above a certain bound such that the percentage be greater than 10%.

Other information concerning call admission control and usage monitoring that would impact on the user's choice of a combination of high and low priority cells would be welcome.

Ad.4 Usage parameters

In our group, the term "window" means the policing time for average bit. The following methods are considered for the policing in the network.
- Jumping window : there is no time interval between succeeding two windows.
- Moving window : (sliding window) window is sliding at a time step smaller than window
- Stepping window : there is a time interval between succeeding two windows, which always start with a valid cell.
- Leaky bucket : cells are put into the buffer and take out from the buffer at an average bit rate. If buffer overflow occurs, cells are discarded.

If a codec cannot know when the window starts in the network, it should control the bit rate by sliding window (most severe one). Is there any technique to know when the window starts in the network?

The information generated by the video codec varies in nature. Therefore, VBR service is expected to be suitable for video transmission. However, if the window size is small, the merit of VBR service for video codec will vanish. What is the maximum window size in each of the following case?
case 1 : average bit rate=1.5Mbps maximum/average=3
case 2 : average bit rate= 5Mbps maximum/average=3
Furthermore, what is the relation between window size and delay?

VBR service is not always suitable for video transmission. The merit is related to the network charge. How is the network resource management done for VBR? We adopted the first network model as included in our previous liaison statement. A study results based on this model is shown in Fig. 1. (such as contained in AVC-47)

Fig.1 Statistical multiplex

Is there any other equations or models suitable for estimating the VBR resource management?

When network becomes grown, user rate becomes comparatively smaller. In that situation, it seems that there is only a little difference of required network resource between low cell loss class and high cell loss class. Will high cell loss class continues to exist in the future?

Ad.5 Multimedia connections

The ability of the B-ISDN to perform the multiplexing task provided by the terminal on circuit-switched networks makes it attractive to consider cell-by-cell multiplexing (by use of different Virtual Channels, or possibly on a single VPI/multiple VCI's) for the provision of multimedia connections. However two issues arise here, as discussed below:

-Will the network be capable of providing connection admission and monitoring based on the group of VC's constituting a multimedia connection? If not, would users see a penalty in the use of multiple VC's, and be encouraged to perform multimedia service multiplexing at a higher layer? SGXVIII should be aware of this possibility and consider whether this capability can be accommodated. Previous experience with multimedia services suggest that a group of at least seven VC's may be necessary, but we would like to know if there is an upper limit.

-Differential VC delay. If multimedia connections (video and associated audio in particular) are supported over multiple VCs, there exists the possibility of differential delay. If excessive, this may require end-to-end signalling overheads to add time-stamps and permit resynchronization. What is the expected limit on differential delay between VCs? Is there any technique to reduce the difference of delay for two VCs? For example, if two VCs for video and audio can select the same virtual path, it will be a merit for multi-media communications.

What kind of network service is available for multi-media? Is the outband negotiation available?

What kind of multi-media multiplex method is preferable from the standpoint of network resource management.

Ad. 6 Bit error rates

We assume that cell payloads are subject to a small probability of transmission bit errors. The statistics of such errors will determine the need for, and type of, error correction mechanism and the overhead necessary to achieve this. It could also influence approaches to, and efficiency of, video coding and choice of code word assignment schemes.

What is the expected probability of transmission bit errors, and are these errors likely to be uncorrelated of bursty? Draft Rec.

I.363 notes (section 2.3 and 3.3) that one of the functions of type 1 and 2 AALs is the '...monitoring of user information field or bit errors and possible corrective action...'. The Experts Group wishes to work with SGXVIII to further clarify the functionality of the AAL in this respect.

From the standpoint of interworking between N and B-ISDN video codecs, the bit error ratio must be the same level as that of N-ISDN.

Ad.7 Cell delay and jitter

The fixed component of end-to-end delay is an important factor for conversational video services. It will impact on the choice of coding method and allowable buffering within the encoder and decoder.

What is the expected maximum B-ISDN delay, including processing and queuing within the B-ISDN switching equipment?

The variation in delay, or jitter, determines the size of receive buffers necessary for its removal, and therefore again influences the total end-to-end delay. What are the expected statistics of cell delay jitter? Is a hypothetical reference connection available or planned, that would assist in these matters?

Ad.8 Network model for hardware experiments

It is the initial intention of this Experts Group to target hardware trial of ATM video codecs for the second half of '93. Success of such trials will depend on the availability of network equipment or simulators. Would these be expected in this timeframe?

Ad.9 AAL

Draft Rec. I.363 describes AAL type 1 & 2 structures which could be used for real-time video services. To make progress, the experts group intends to distinguish between:

-an AAL suitable for existing video services (e.g. H.261), that could be standardized in the relatively short term,
-for future ATM video coding standards, an AAL matched to the specific coding algorithms will be necessary. It is premature to define an AAL for these applications at this time.

What other considerations should be taken into account, when we investigate the AAL type 2? Is the function of AAL type 2 decided only from a standpoint of video coding?