ITU Telecommunication Standardization Sector
Study Group 15
Experts Group for Video Coding and Systems
in ATM and Other Network Environments

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Source: SG13 Rapporteur for Q.16/13
Title: Liasion Statement on ATM Network Performance
Purpose: Report

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Question(s): 16/13, 2/15

SOURCE: SG 13, Question 16/13, November 14-25, 1994

TITLE: Liaison to SG 15 for action

Subject: Assumptions Concerning Possible ATM Network Performance Levels

Deadline for reply: June, 1995

Contact point: Kenneth C. Glossbrenner
Rapporteur for Q16/13
AT&T Bell Laboratories
Rm 3L-531
Crawfords Corner Rd.
Holmdel, NJ, U.S.A. 07733

Tel: +1 908 949 1857
Fax: +1 908 949 1652
email: kcg@honet6.att.com

Thank you for the opportunity to review the assumptions you are making concerning possible ATM network performance levels. The following discussion presents the best common understanding of Q16/13 participants concerning ATM network performance at this point in time. Unfortunately, for the most part this understanding is not yet based on substantial test results. As in your ATM Performance Assumptions document, the numbers presented here do not represent official numbers from any national or international standards organization.

As we pursue revisions to Recommendation I.356 (B-ISDN ATM Layer Cell Transport Performance) we are maintaining a living document of the performance needs coming from the various communications applications that will make use of the BISDN. Input from the AVC group of SG 15 on video system requirements would be most appreciated. When the document reflects the needs of each of the major applications, we will endeavor to establish ATM network performance objectives sufficient for those applications. The enhanced version of I.356 is scheduled to be approved in April, 1996.

- In addition to the factors you named, CDV performance will be strongly influenced by the bit rate of the facility carrying the ATM cells. Lower bit rate systems such as primary rate systems will yield substantially more CDV.
- The range you presented for CLR is reasonable, assuming you are not including cells lost during bursts. In our definition of CLR, burst events are eliminated by counting them as contributors to severely errored cell blocks (SECB). A SECB will probably result in a major video disruption, such as a freeze frame. SECBs will likely occur with a frequency equivalent to today's SES events. (<0.2% end-to-end network commitment, per G.826. The actual SES performance is often much better.)
- I.356 specifies bit error performance using cell error ratio (CER), the fraction of cells that deliver one or more bit errors to the AAL. The worst case long-term BER delivered from this process is reasonably assumed to be $<10^{-7}$. Typical BER performance may be much better.
- Burst bit error events may occur when an FEC is used on the physical link. When they occur they are likely to be one cell payload in length or less. An SES at the physical layer will likely be converted into a burst of cell loss and, possibly, some small number of errored cells.

- The end-to-end delay figures you assume may be too large. After subtracting the propagation delay, the Q16/13 experts believe the mean end-to-end delay may be between 1 and 10 milliseconds.
- The CDV numbers you quote may be optimistic. Most CDV studies to date have focused on networks carrying CBR traffic only. Furthermore, the queuing delay variation for a single low speed link could more than 1 millisecond by itself.
- In note c. to Table 1, it would be more appropriate to reference Recommendation G.826 (1993).

We have a few questions for you:

- We would appreciate your opinion on the appropriate values of N and M for our definition of SECB. What lengths and intensities of bursts of cell loss will be critical thresholds for your video equipment?
- Because propagation delay will dominate the end-to-end delay and because codecs themselves contribute substantially to the customer perceived delay, why would you be concerned about the relatively small processing and queuing delays in the network? What value of end-to-end network delay is important you? We do not understand why you feel that 10 milliseconds is large for ordinary telephony.
- How will your equipment compensate for CDV? Will you use a phase lock loop to average out the variations or will there be a timestamp in your payload that will be used to subtract out the variations? How would you prefer to see network CDV expressed? Is your concern about long term (low frequency) CDV greater than your concern for high frequency CDV?
- Finally, what are your thoughts about the expected values of each of the performance parameters? Are any of them believed to be a potential source of problems for your video equipment?

*Appendix (original question from the SG15 Experts Group)*

Annex 2 to AVC-704

Source: Rapporteur for Q.2/15 (Sakae OKUBO)
Title: Correspondence to Rapporteur for Q.16/13 (Mr. Kenneth C. Glossbrenner) and Q.6/13 (Mr. Katsuyuki Yamazaki)
Subject: Network performance assumptions
Purpose: Request for comments

CONTACT: Sakae OKUBO
Rapporteur for Q.2/15
Graphics Communication Laboratories
6F ANNEX TOSHIN BLDG, 4-36-19 Yoyogi,
Shibuya-ku, Tokyo 151 Japan
Phone: +81 3 5351 0181
Fax: +81 3 5351 0185
e-mail: okubo@gctech.co.jp

Experts Group for Video Coding and Systems in ATM and Other Network Environments in Study Group 15 has been studying the network adaptation which connects the elementary stream (such as audio, video) coding layer and the ATM layer in audiovisual communication terminal (projected Recommendation H.32X). Typical functionalities of the network adaptation includes multimedia multiplexing and synchronization, bit error/cell loss handling, reduction of CDV.

Though one of the decisive factors for the choice of network adaptation solution(s) is network performance, we understand, after having jointly met with the AAL1&2 group of SG13 in March 1994, that sufficient advice will not be obtained in the near future One way for proceeding is to make a couple of network scenarios and evaluate the network adaptation solutions based on them. These network scenarios are intended for internal use, but it is also expected that its exposure to the outside may induce comments from the network people.

At the July 1994 meeting in Norway, we formulated such scenarios by educated guess. The outcome is documented as attached. It should be noted that these assumptions are our own yardstick in the lack of definitive performance figures and do not represent official views from any national or international standardization bodies.

We see the following implications from the network performance assumption:

1. If the mean error free time is calculated for a 6 Mbit/s connection, it becomes 2.5 minutes for bit errors even in the best case scenario. Hence it is concluded that bit error correction is indispensable for higher bit rate communications, while cell loss becomes a problem only in the worst case scenario (mean error free time = 64 seconds).

2. End-to-end delay is of the order of 10 ms excluding the propagation delay. This looks rather large if we consider the B-ISDN support of ordinary telephone as part of audiovisual services.

3. It is noted that the short interleaver FEC method, which protects against one cell loss in a 16-cell block, may not be required under several transmission conditions, particularly at low bit rates such as 64 kbit/s. However, a simpler FEC approach which only protects the data against BER conditions (i.e., no cell loss protection) might be desired. Currently, AAL-1 does not support such a mechanism (i.e., error correction capability only).

We would welcome comments of the SG13 members on our network scenarios as attached and their implications as listed above.

END

Attachment: SG15 Experts Group Document AVC-635 "ATM performance assumptions"