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

Document AVC-704
26 October 1994

Source: RAPPORTEUR (Sakae OKUBO)
Title: Correspondence with Rapporteurs of SG13 and SG11
Purpose: Report

This document includes the following correspondence with Rapporteurs of SG13 and SG11, which has been generated according to the discussion at the Grimstad meeting:

	Rapporteurs addressed	Title
1	Q.6/13 (Mr. Katsuyuki Yamazaki)	Priority and error indication primitive parameters at ALL-SAP
2	Q.16/13 (Mr. Kenneth C. Glossbrenner) and Q.6/13 (Mr. Katsuyuki Yamazaki)	Network performance assumptions
3	Q.6/13 (Mr. Katsuyuki Yamazaki)	Audiovisual services between the two terminals accommodated in B-ISDN and N-ISDN
4	Q.6/13 (Mr. Katsuyuki Yamazaki)	AAL1 short interleaver, circuit transport and support of SDT
5	Q.15/11 (Mr. Rajiv Kapoor) and Q.6/13 (Mr. Katsuyuki Yamazaki)	Negotiation and signalling of an operational mode of multimedia multiplex and synchronization

Annex 1 to AVC-704

Source: Rapporteur for Q.2/15 (Sakae OKUBO)
Title: Correspondence to Rapporteur for Q.6/13 (Mr. Katsuyuki Yamazaki)
Subject: Priority and error indication primitive parameters at ALL-SAP
Purpose: Request for action

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At its Norway meeting in July 1994, Experts Group for Video Coding and Systems in ATM and Other Network Environments in Study Group 15 discussed the primitive parameters at the boundary between the AAL layer and the multimedia multiplex (H.222.1) layer for audiovisual communications. After having reviewed input contributions, we concluded that primitive parameters should be defined for priority and error indication as follows:

Table 1 Proposed definition of primitives for use with H.222.1 at AAL-SAP

primitive	direction	parameters
AAL_UNITDATA.request	user to AAL	DATA, PRIORITY
AAL_UNITDATA.indication	AAL to user	DATA, STATUS

where STATUS in AAL_UNITDATA.indication includes error indication that DATA may include errors based on the AAL error correction/detection results. The primitive names and the exact meaning of the parameters in Table 1 require further study. There may be a STRUCTURE parameter associated with the AAL_UNITDATA.request and AAL_UNITDATA.indication primitives.

It is our agreement that the linkage between the higher layer priority request (e.g. transport_priority bit in the H.222.0 Transport Stream) and the primitive parameter setting at the AAL-SAP should not be tight; it should be left to the choice of applications. In the case of an integrated H.32X terminal, the entity which might set the transport_priority bit has access to the AAL-SAP, and can set the PRIORITY parameter as required. In the case of a Transport Stream being accessed remotely, it remains the users option as to how priority should be implemented, and as to whether it should or should not be used. Such choices depend on cost, and network Quality Of Service offerings.

We request SG13 to consider definition of additional primitive parameters at AAL-SAP which are necessary for audiovisual communications as discussed above.

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

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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"

Annex 3 to AVC-704

Source: Rapporteur for Q.2/15 (Sakae OKUBO)
Title: Correspondence to Rapporteur for Q.6/13 (Mr. Katsuyuki Yamazaki)
Subject: Audiovisual services between the two terminals accommodated in B-ISDN and N-ISDN
Purpose: Request for advice

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Experts Group for Video Coding and Systems in ATM and Other Network Environments in Study Group 15 has been studying audiovisual systems and terminals in B-ISDN; H.32X for utilizing new generation audiovisual coding (e.g. H.262|MPEG-2 Video) and H.32Y for adapting existing H.320 to B-ISDN. One of the essential features of such B-ISDN terminals, particularly videoconferencing and videophone terminals, is to interwork with existing H.320 audiovisual terminals accommodated in N-ISDN. According to I.581, the network interconnection between B-ISDN UNI and N-ISDN UNI is supported through Interworking Function at the junction point of the two networks by using the AAL1 circuit emulation service.

Since existing H.320 terminals are not prepared for a long burst of errors caused by the cell loss, we need a cell loss protection mechanism for the B-N interworking in the B-ISDN portion of the connection if the cell loss is excessive. Furthermore, the cell loss protection mechanism should not incur excessive delay in conversational services. The current short interleaver in AAL 1 has been specified to meet these requirements. Hence we understand that AAL 1 with short interleaver in the Convergence Sublayer should be invoked for audiovisual B-N interworking where the cell loss is problematic.

At the July 1994 meeting of the Experts Group in Norway, some questions were raised how this AAL can be invoked at the Interworking Function. When an audiovisual terminal accommodated in B-ISDN places a call, it knows that its N-ISDN destination should also be an audiovisual terminal, thus such AAL should be negotiated. Since AAL should be terminated at the Interworking Function, the Interworking Function may have to support multiple service dependent AALs; such as "Circuit transport", "Video Signal Transport".

When a terminal accommodated in the N-ISDN places a call, it may carry audiovisual attributes in LLC/HLC, thus the Interworking Function may be able to invoke AAL1 with short interleaver for the connection. LLC/HLC, however, is generally not decoded inside the network according to the principle that the network is service independent. If that is the case, there seems to be no way for Interworking Function to identify whether the call is audiovisual or of unrestricted digital information.

Related to the above discussion, we feel that availability of AAL tools (e.g. FEC) should be negotiated at the start of a call among terminals, terminal adaptors and Interworking Functions. If such AAL capabilities are not communicated, it is not clear how terminals and adaptors will decide when to use these capabilities.

We would like to seek advice of the SG13 members on the following:

1) whether our understanding for B-N interworking and operation of the Interworking Function is correct,
2) what provisions are necessary in the audiovisual terminals to achieve B-N interworking, and
3) how the choice of AAL1 tools is negotiated and signalled to the remote end.

END

Annex 4 to AVC-704

Source: Rapporteur for Q.2/15 (Sakae OKUBO)
Title: Correspondence to Rapporteur for Q.6/13 (Mr. Katsuyuki Yamazaki)
Subject: AAL1 short interleaver, circuit transport and support of SDT
Purpose: Request for clarification

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At its July 1994 meeting in Norway, the SG15 Experts Group for Video Coding and Systems in ATM and Other Network Environments gratefully received your input regarding the "circuit emulation" for Broadband and Narrowband interworking. We now understand that the "Circuit transport" service of AAL1 is the right word, that the "Circuit transport" supports the SDT pointer, but the "Video signal transport" with the short interleaver FEC does not support the SDT pointer, and that the support of both SDT and FEC in AAL1 is under study.

It is appreciated if we could receive further clarification on how the choice of an appropriate AAL service be negotiated and signalled for communication between two H.32Y (adaptation of H.320 to B-ISDN) terminals or for interworking between an N-ISDN H.320 terminal and a B-ISDN H.32Y terminal.

END

Annex 5 to AVC-704

Source: Rapporteur for Q.2/15 (Sakae OKUBO)
Title: Correspondence to Rapporteur for Q.15/11 (Mr. Rajiv Kapoor) and Q.6/13 (Mr. Katsuyuki Yamazaki)
Subject: Negotiation and signalling of an operational mode of multimedia multiplex and synchronization
Purpose: Request for advice

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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). Our study is summarized in the network adaptation protocol reference model as attached.

As shown there, we have two options of multimedia multiplex and synchronization methods; PS (Program Stream) and TS (Transport Stream). One of these will be chosen on the basis of required service of the application and the network performance. More generally we will have several H.222.1 operational modes. Though we will have an in-channel negotiation mechanism regarding terminal capabilities, its signal may be multiplexed with audio, video and other signals using PS or TS.

Hence we need to identify what multiplex option should be used in a particular call through a channel other than the one carrying audiovisual signals. One way would be through outband signalling, e.g. by the user data or LLC/HLC fields in Q.2931. We would appreciate any advice of SG11 members with this respect.

END

Attachment: Proposed H.32X network adaptation protocol reference model