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in ATM and Other Network Environments

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SOURCE : JAPAN
TITLE : Cell loss correction method for MPEG 2 System Streams transmission
PURPOSE : Discussion
Relevant sub-group : System

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1. Introduction

For the error correction method in AAL type 1, the Reed-Solomon (94,88) code with diagonal interleaving was reviewed in AVC-569 at Brussels meeting. For constant bit-rate services, this method can be realized using ring buffer, however, it is not appropriate for variable bit-rate transmission. Therefore, the double buffer method which has the closed form of error correction matrix, was proposed for variable bit-rate services.

These correction method can be applied for the transmission of MPEG 2 system streams (i.e. Program Stream, Transport Stream and Packetized Elementary Stream) over the ATM networks without ATM cell alignment.

In this document, we discuss the error correction methods in AALs for Transport Stream packet (188 bytes) transmission with ATM cell alignment.

2. Error correction method for the alignment between TS packet and ATM cell

Two error correction methods which can maintain the alignment between TS packets and ATM cells are described.

2.1 Using the AAL type 1

Figure 1 shows the Transport Stream packet and the AAL type 1. Sequence numbers are available in AAL type 1 headers. To achieve the alignment and cell loss protection, the combination method of byte interleaving and RS code can be applied. The byte interleaving is organized as a matrix of 47 rows and 192 columns. In order to detect the bit error within a TS packet, 16-bit CRC(Cyclic Redundancy Code) is calculated using the same polynomial of the HDLC system i.e. $x^{16}+x^{12}+x^{5}+1$. The TS packets with CRC and 2-byte RS(192,190) code are horizontally written in the transmitting interleaver, and 47-byte cell payloads are vertically read out. This method can correct either 2 cell losses within 192 cells or 1 byte random error in a row of 192 bytes. The overhead of this method is 2.1%. For synchronization of the interleave matrix, PTI(user to user indication) or CSI(CS indicator) bit can be used.

2.2 Using a new AAL type

Figure 2 shows the TS packets and a new AAL type. A TS packet is stored into four ATM cell payloads, each cell has one byte header. The header consists of 4-bit Sequence Number field and 4-bit CRC field. The 4-bit CRC field is combined over the four ATM cells in order to transmit a 16-bit CRC which is calculated for the TS packet. In order to correct the cell loss and/or random byte error, four Reed-Solomon code cells are added for every *m* TS packets transmission. By using RS($4m$+4, $4m$) code, this method can correct either 4 cell losses within $4m$+4 cells or 2 bytes random error in a row of $4m$+4 bytes. The overhead is $(m+1)^{-1}$.

In case of m=12, 24, 48, the correction performance for cell loss and random byte error are shown in figure 1 and 2 of Annex 1, respectively.

3. Conclusion

For MPEG 2 Transport Stream packet transmission, the cell loss correction methods using AAL type 1 and new AAL type are discussed to achieve the alignment between TS packets and ATM cells.

END.

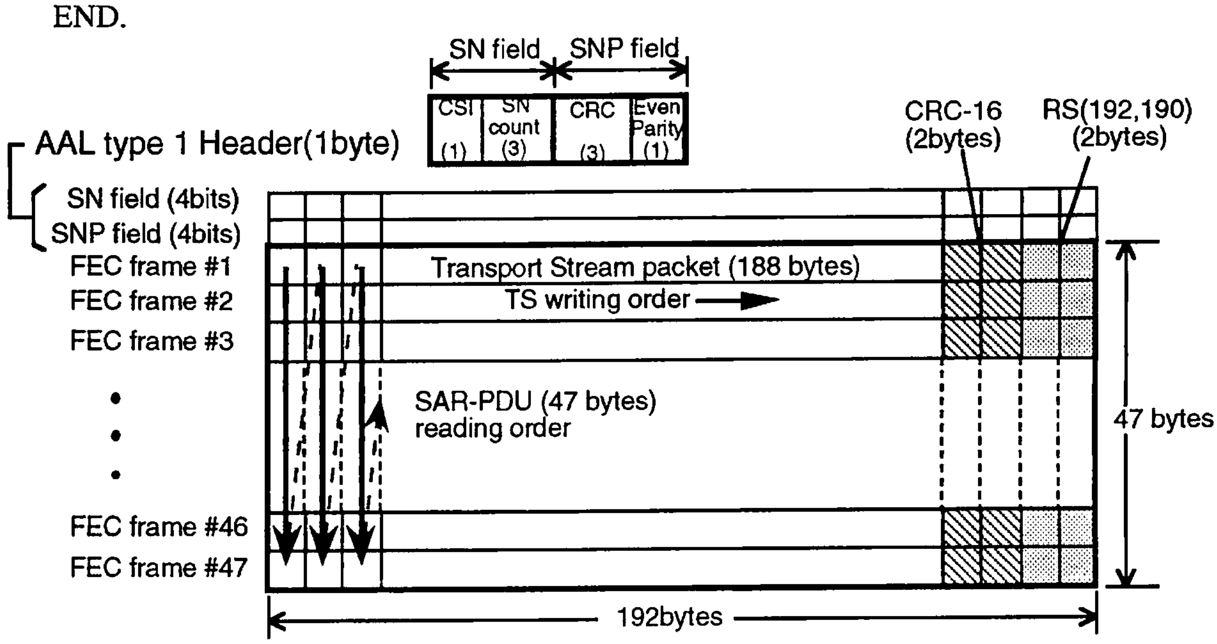

Figure 1. Transport Stream packet and CRC-16 + RS(192,190) in AAL type 1.

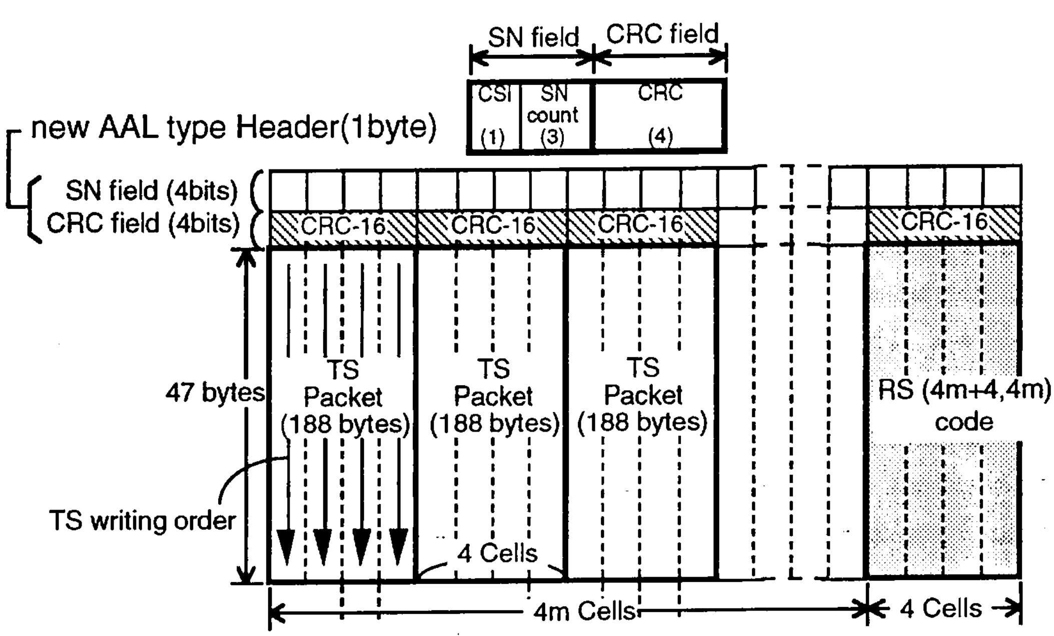

Figure 2. Transport Stream packet and new AAL type.

Annex 1 to AVC-594

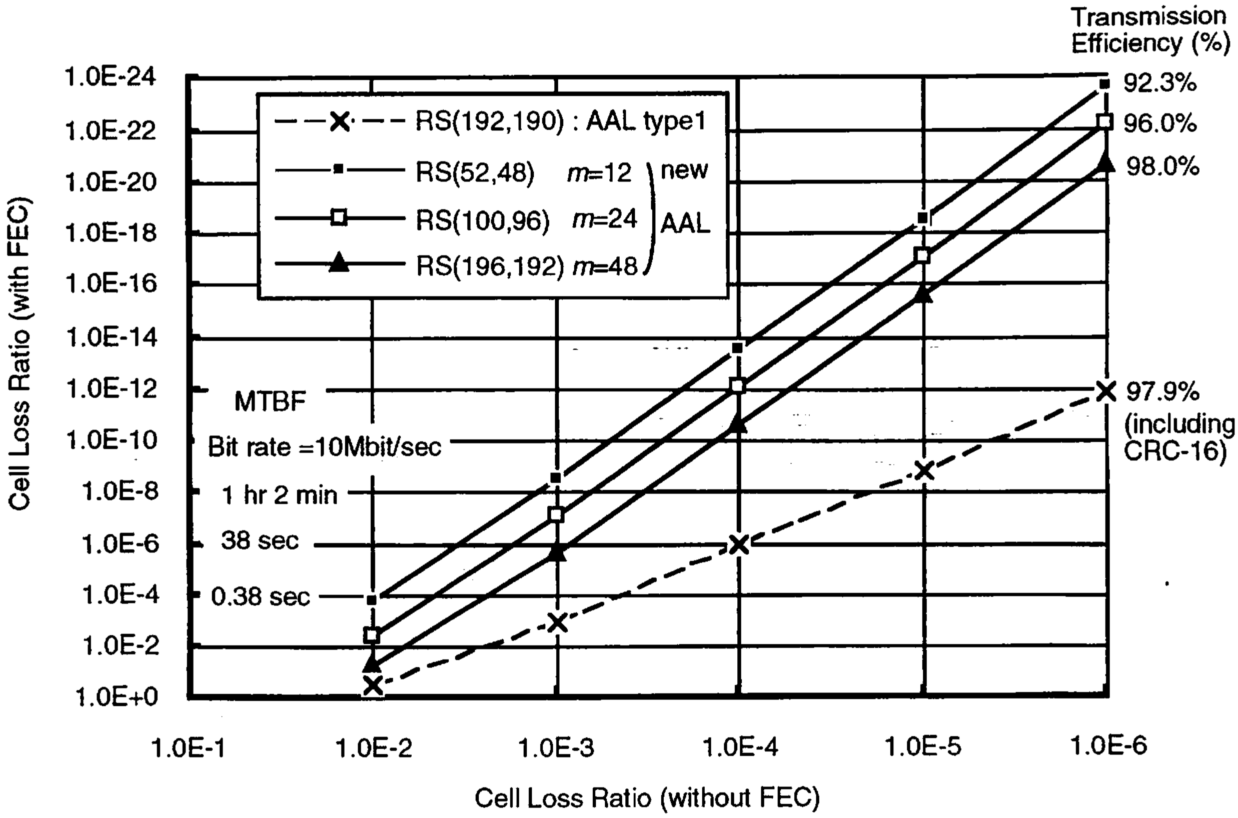

Figure 1 / Annex 1. Correction performance for cell loss.

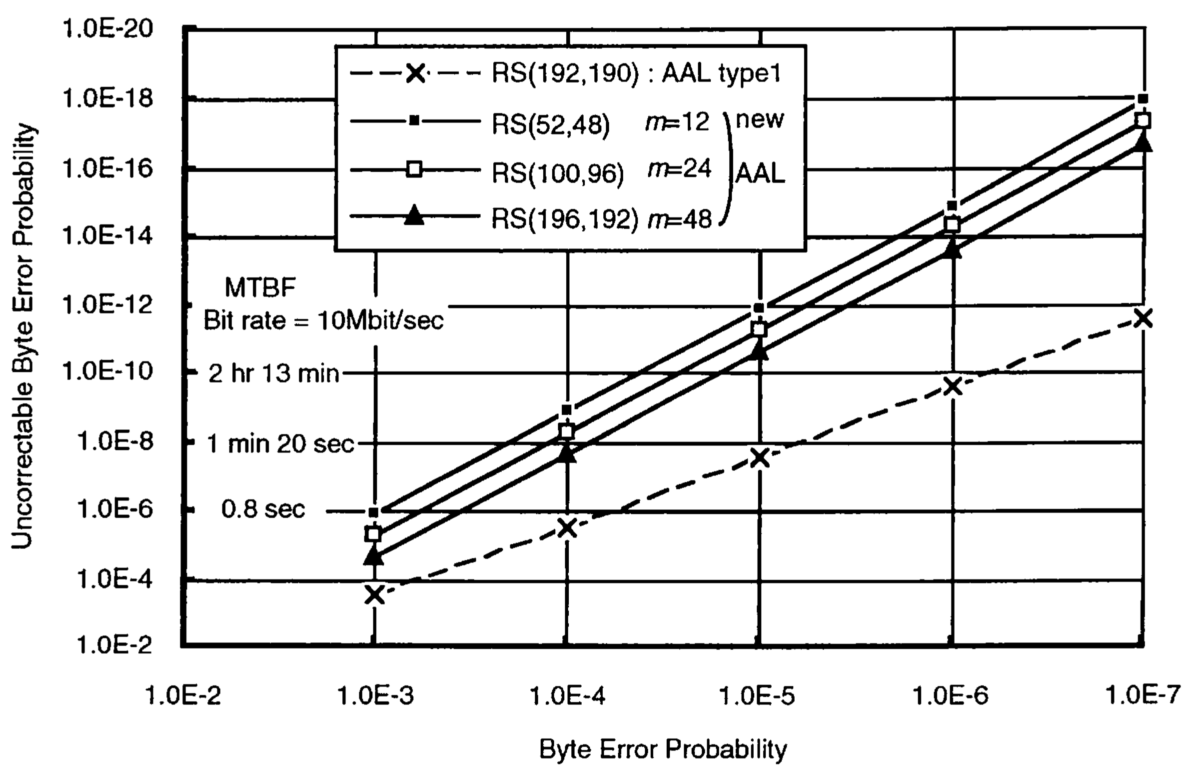

Figure 2 / Annex 1. Correction performance for random byte error.

gateway converts communication protocol and video-audio data frame format only, because these both equipments implement ITU-T Rec. H.261 video coding scheme and G.728 audio coding scheme. However when an ISDN visual telephone has only G.711 audio coding scheme, the gateway transcodes these coding schemes mutually.

3. Communication protocols and congestion control

3.1 Communication protocols

Figure 3 shows a call setup and release sequence over LAN. TCP protocol is used for these connection establishment sequence, and UDP datagram transmission is used for realtime transmission of coded audio and video data.

(1) Call setup

The adapter initiates a call when the adapter gets 'call' command with IP address parameter from PC/WS. The adapter transmits call setup commands to a destination adapter in TCP. The destination adapter responds with call setup accept command when the adapter establishes the call. These commands, call setup and call setup accept, have some parameters which indicate audio data length, video encoder transmission capability, video decoder mode request, etc.

(2) Coded audio and video data transmission

The adapters send and receive audio/video coded data with UDP. The audio data and video data are combined in a packet to decrease the number of packets, and transmitted at regular intervals. This is to achieve better transmission efficiency. The packet structure is shown in Figure 4. A packet consists of a control command, a sequence number, an audio data length, a video data length, coded audio data and coded video data. The control command indicates fast update request, loop back request and, transmission and reception bit rates. The sequence number shows the packet number and is used for detecting packet loss by checking it continuity. The audio data is fixed at 0, 70, 110 or 150 bytes in a packet. The video data is filled with variable length coded data at regular intervals in a packet.

(3) Call release

The adapter releases a call when the adapter gets 'release' command from PC/WS. The adapter transmits release command to the destination adapter by TCP, then stops audio-video data transmission through UDP data packets.

3.2 Congestion control

The UDP is very simple and easy to process, so it can suit realtime data transmission. However, when LAN traffic becomes heavy, UDP packets are discarded. The adapter can detect a packet discard by checking the sequence number in the data packet, then lowers video throughput by one step automatically to relieve congestion and sets the encoder in fast update mode to clear decoded video. When congestion persists, it adjusts rate one step further. Transmission rates return