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1. Introduction

In the description of the AAL 1 in the Rec. I.363, a cell loss correction method has been specified in the CS for video. The method relies on a combination of FEC and octet interleaving, enabling the correction of 4 lost cells in a group of 128 cells (i.e. in an interleaving matrix). However, because of the processing delay it implies, this method is more suitable to high bit rates encountered in the transport of video programs.

Besides, during the interregnum meeting of the SWP XVIII/8-3 the question of the definition of cell loss correction mechanism for interactive low bit-rates video services (e.g. H.320 terminals) arose, based on a liaison statement of SG XV responding to SG XVIII on this issue. Views were expressed then on the interest to define an adapted mechanism for such services.

On the other hand, the TG CMTT/3 (where the transport of High Quality sound in the B-ISDN is currently addressed) has discussed about a cell loss correction method adapted to HQ sound. As a result, a liaison statement has been issued to the SG 13 of the TSS.

The present contribution is in line with both the living list of the SWP XVIII/8-3 and the liaison statement of the CMTT/3, and it proposes that a cell loss correction method **common** to all real-time low bit rate services should be specified.

2. Cell loss correction for real-time services

When correction of cell losses is mandatory (i.e. when concealment mechanisms in the layers above the AAL do not apply although a high quality of service is required) the method based of a combination of a FEC and data interleaving appears as a classical method, because it enables large amounts of errored data to be recovered. It is a generic method as well, in the sense that its parameters are to be adapted to particular service requirements. For example, when a short processing delay is aimed at, the size of the interleaving matrix may be adapted.

The parameters that are to be specified in relation to service requirements are discussed in the liaison statement of the CMTT/3.

These are :

- correction capabilities, i.e. number of cell losses to be corrected in a block of data
- type of FEC, (e.g. RS codes), which influences the overhead, i.e. the transmission overcost
- interleaver size, (i.e. the block of data) which influences the processing delay

For low bit rates, the processing delay is the key issue. The lower the bit-rate, the higher the processing delay for a given interleaver size. Therefore the interleaver size should be as low as possible, which reduces the amount of data which can be corrected for a given FEC. On the other hand, for low bit-rates, it seems acceptable to correct only one cell in a block of data for the reasons explained in the liaison statement from the CMTT/3.

3. Discussion

3.1. Processing delay

The liaison statement of the CMTT/3 proposes a list of parameters for the specification of the method. Table 1 gives the value of the processing delay depending on the bit-rate, and the size of the interleaver.

Delays are given when using the classical interleaving method consisting in writing user data horizontally in the transmitting interleaver and reading them out of it vertically to fill in SAR-PDUs, with a symmetrical process in the receiving part (see Fig.1).

The delay (including transmission and reception) is d = 2.K.R/D where R is the depth of the interleaving matrix and D the bit rate of the signal before being processed by the RS (N,K) coder.

The 128 cell interleaver as specified in Rec. I.363 is given as a comparison basis.

	128 kbit/s	256 kbit/s	384 kbit/s	1 Mbit/s
16 cells	88 ms	44 ms	29.3 ms	11.2 ms
20 cells	111 ms	55.7 ms	37.1 ms	14.2 ms
32 cells	176 ms	88 ms	58.6 ms	22.4 ms
128 cells	728 ms	364 ms	242 ms	91 ms

Table 1 : Delays vs bit-rates and interleaver size (classical interleaver)

3.2. Improved interleaving mechanism

A technique is proposed to reduce the delay due to the interleaving process of data. This technique is based on a diagonal reading of the transmitting interleaver combined with a symmetrical diagonal writing of the receiving desinterleaver (see Fig.2)

In the transmitting AAL, incoming RS words of length N are stored horizontally in the interleaver. Then, data are read out of the interleaver **diagonally**. In the receiving AAL, SAR-PDU payloads are stored diagonally in the interleaver and then RS words are read out horizontally.

Fig. 2 illustrates this method and in order to explain it as simple as possible, small values have been chosen for N, R and L (size of the SAR-PDU payloads) : N = 10, R = 4, L = 5.

With this mechanism, the processing of incoming RS words is done continuously (fig. 3). It is not necessary to wait until a whole interleaver has been stored before reading out data to SAR-PDU.

The major advantage of this mechanism is **to divide by 2** the processing delay, both in the transmitting and the receiving AALs.

As a result, when using the diagonal interleaving mechanism, the delays of table 1 are divided by 2 as mentioned in table 2 :

	128 kbit/s	256 kbit/s	384 kbit/s	1 Mbit/s
16 cells	44 ms	22 ms	14.6 ms	5.6 ms
20 cells	55.7 ms	27.8 ms	18.5 ms	7.1 ms
32 cells	88 ms	44 ms	29.3 ms	11.2 ms

Table 2 : Delays vs bit-rates and interleaver size (diagonal interleaver)

4. Proposals

1. For the correction of cell losses it is proposed to define a method **common** to all real-time low bit-rate services, including High Quality sound and interactive audiovisual services.

2. The method should be based on a combination of FEC using RS codes and octet interleaving as proposed in the liaison statement from the CMTT/3.

3. The dimension of the interleaving matrix should be 16 cells.

4. A delay reduction mechanism (e.g. diagonal interleaving mechanism) enabling to divide by two the processing delay of the method should be used.

5. Conclusion

This contribution supports a common cell loss correction method for all real-time low bit-rates services using the AAL type 1. Due to an improved interleaving mechanism, processing delays are very low.

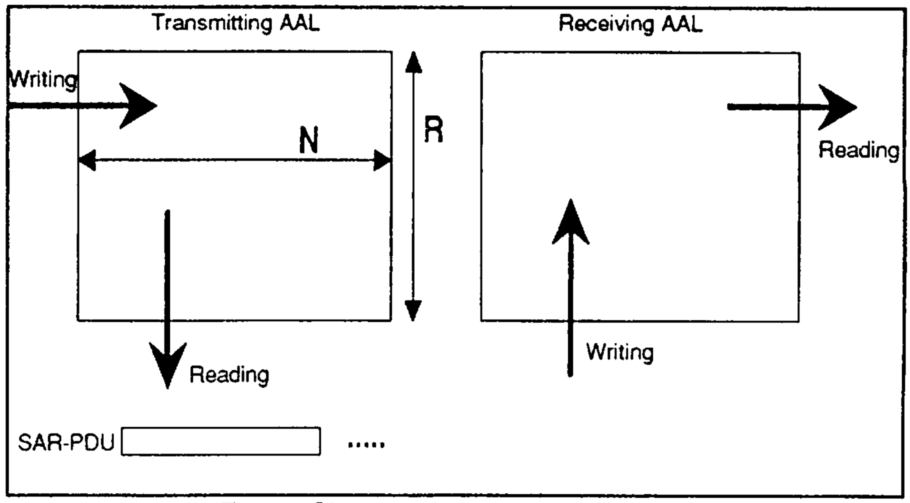


Fig.1 : Classical interleaving method

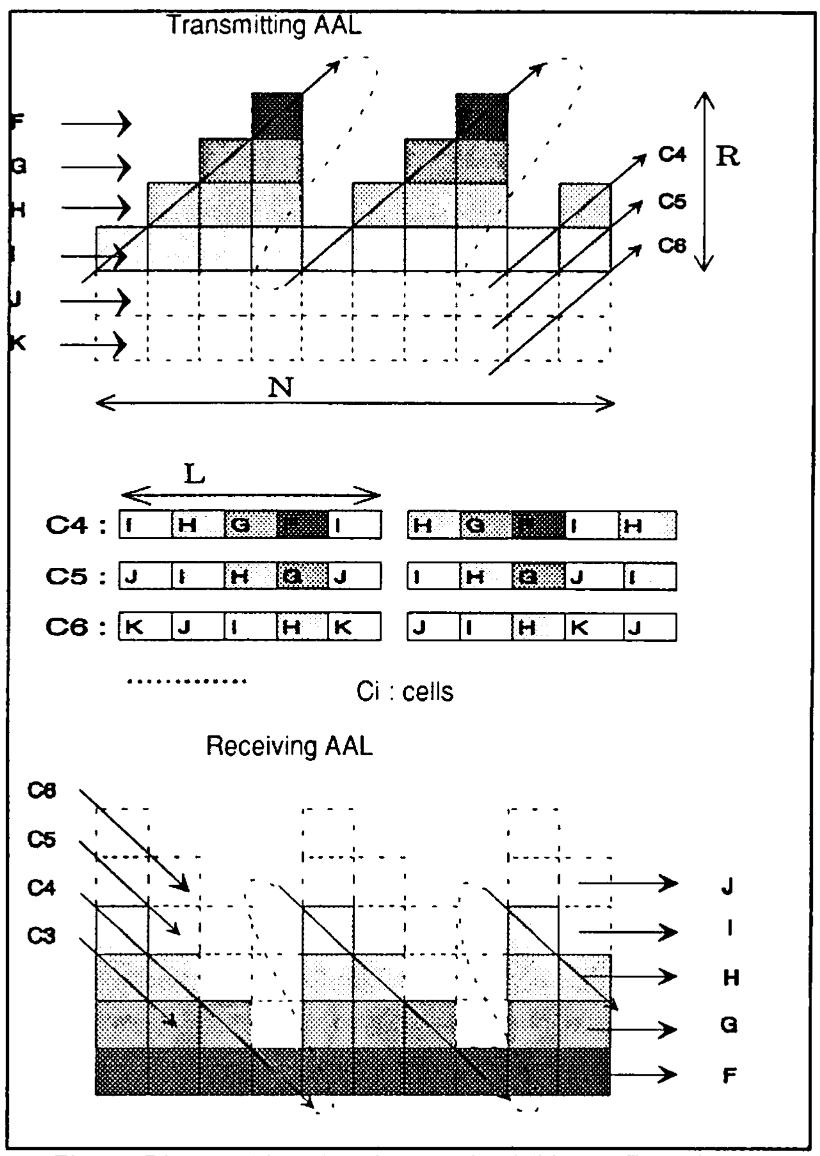
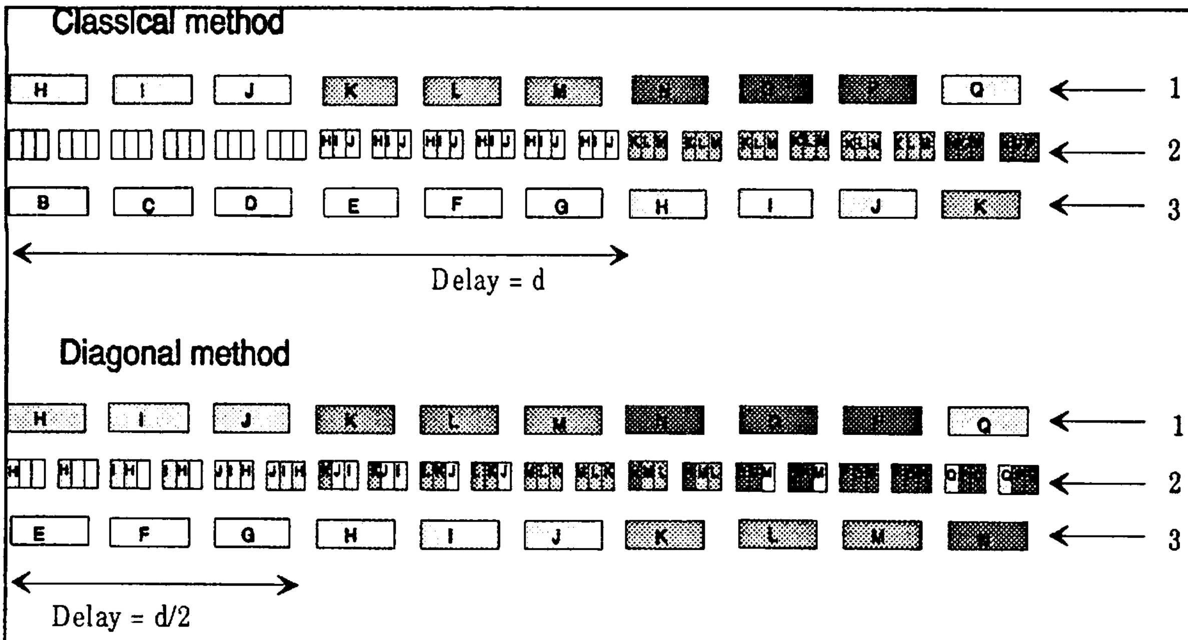


Fig.2 : Diagonal interleaving method (N=10, R=4, L=5).



1 : Writing into the interleaver

2 : Reading out of the interleaver / writing into the desinterleaver

3 : Reading out of the desinterleaver

Fig.3 : Processing delay (N=6, R=3, L=3)

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1. Introduction

On the proposal of the CMTT/TG3, the WP XVIII/8 of the CCITT decided during its January 1993 meeting to define a CS for the transport of high quality sound. In the list of functions proposed by the CMTT/TG3, most of the functions can be performed by using techniques already defined for the other CS. However, the correction of cell losses cannot be covered by the technique already defined for video because of the high delay it implies for the low bit rates encountered in audio. Therefore a technique for the correction of cell losses adapted to high quality audio signals is still to be defined.
A method is described which, on the assumption that the correction of the loss of one cell is sufficient, would provide full protection against all errors occurring in the unidirectional transmission of high quality audio signals in ATM networks by combining a FEC code with an interleaving technique.

2. Correction of cell losses : basics

For the correction of cell losses, a method has already been defined in Rec. I.363. This method, which appears to be classical, is based on a combination of a Forward Error Correction (FEC) redundancy and an interleaving process aiming at dispersing errors in the data flow, in order to keep them within the correction capabilities of the redundancy. This method is powerful enough to correct cell losses, with as a counterpart a processing delay proportional to the dimension of the interleaver. In practice, this delay becomes critical as the bit rate decreases : for example, the method described in Rec. I.363 introduces a total delay of 91 ms at 1 Mbit/s, and up to 364 ms at 256 kbit/s (taking account of both transmitting and receiving ends, i.e. (124 + 124) x 47 octets at the AAL-SAP). For that reason, its use is limited to unidirectional video, and therefore another method has to be defined for high quality audio where bit rates are typically in the range of 128 kbits/s to about 1 Mbit/s.

3. Rationale for the definition of the parameters

3.1. Delay :

The major requirement is the processing delay, which should be as low as possible. As it is a function of the interleaver length, this length must be as small as possible.

3.2. Correction capabilities :

Due to the low bit rate of the service, burst of cell losses are not to be expected. The major reason why cell loss will be experienced in the network is a congestion in the waiting queues leading to a burst of cell losses. However, these congestions affect high bit rate multiplexes, so the probability that a low bit rate service will undergo two consecutive cell losses during a congestion is very small. As a result correction capabilities are only required to correct one cell loss in the group corresponding to the octet interleaver.

3.3. Redundancy and overhead :

The overhead due to redundancy must be kept as low as possible. The minimum value for the redundancy to correct one lost cell is one redundancy cell in the interleaver.

3.4. Correction code :

Due to their correction capabilities in the case of cell loss, Reed-Solomon (RS) codes are preferred. The erasure mode of RS codes is quite adapted to the recovery of the dummy information inserted after a cell loss has been detected.

4. Proposal

4.1. RS codes

The definition of the RS code must take into account the following points :
- in order to get a small dimension of the interleaver, the correction block must contain an integer number of SAR-PDU payloads, i.e. 47 octets.
- for practical reasons, RS codes with a maximum correction capability of 255 octets are the most easily implemented.

As a result, the possible figures for the correction block are : 47, 94, 141, 188, 235.

As the redundancy must enable the correction of one cell in the interleaver, the definition of the relevant codes is linked to the interleaver structure. Possible values are :
- RS (235, 223) : 12 octet redundancy, interleaver dimension : 20 cells (235 x 4 octets)

- RS (188, 176) : 12 octet redundancy, interleaver dimension : 16 cells (188 x 4 octets)
- RS (141, 133) : 8 octet redundancy, interleaver dimension : 18 cells (141 x 6 octets)
- RS (94, 88) : 6 octet redundancy, interleaver dimension : 16 cells (94 x 16 octets)
- RS (47, 44) : 3 octet redundancy, interleaver dimension : 16 cells (47 x 16 octets)

4.2. Correction capabilities

For all solutions, one cell loss is corrected in the interleaver.

Correction capabilities for errored bits are :
- RS (235, 223) : 6 errored octets in the block of 235 octets
- RS (188, 176) : 6 errored octets in the block of 188 octets
- RS (141, 133) : 4 errored octets in the block of 141 octets
- RS (94, 88) : 3 errored octets in the block of 94 octets
- RS (47, 44) : 1 errored octet in the block of 47 octets

As the latter solution shows poorer capabilities for bit errors, it should be discarded.

4.3. Interleaver structure

In the transmitting AAL, the interleaving matrix is written row by row, and it is read column by column. Therefore, the structure of the matrix for each RS code is as follows :
- 235 columns and 4 rows, i.e. 20 cells
- 188 columns and 4 rows, i.e. 16 cells
- 141 columns and 6 rows, i.e. 18 cells
- 94 columns and 8 rows, i.e. 16 cells

In any case, in the receiving AAL, each cell loss corresponds to 47 dummy octets inserted in consecutive slipping columns of the interleaver, thus enabling the correction row by row.

4.4. Delay

The delay is due to the processing both in the transmitting and the receiving AAL. For a given bit rate, this delay is proportional to the number of octets provided at the AAL-SAP which is contained in the interleaver (i.e. the redundancy must not be counted, because it is not seen by the user).
Values are given hereafter for the proposed solutions :
223 octets x 4 x 2, i.e. 1784 octets (approx. 38 cells)
176 octets x 4 x 2, i.e. 1408 octets (approx. 30 cells)
133 octets x 6 x 2, i.e. 1596 octets (approx. 34 cells)
88 octets x 8 x 2, i.e. 1408 octets (approx. 30 cells)

4.5. Overhead

The overheads of the four codes are :
- for RS (235, 223) : 5.4%
- for RS (188, 176) : 6.8%
- for RS (141, 133) : 6.0%
- for RS (94, 88) : 6.8%

5. Conclusion

A comparison should be made between the above four proposals, and relevant other proposals, if any, to choose the best solution for the correction of cell losses in the CS for High Quality audio. The comparison should take into account the following criteria :

- acceptable delay for the service
- acceptable overhead for the service
- implementation aspects
- suitability for other services (e.g. audiovisual interactive low bitrate services).

It is currently intended that there willl be further study of this subject within the CMTT.

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