A Novel Method for Inserting an MPEG-2 TS into Ensemble in a DMB Transmission System

Gwang Soon Lee, Byungjun Bae, Young Kwon Hahm, and Soo In Lee

ABSTRACT—This paper presents an effective algorithm for inserting an MPEG-2 transport stream (TS) into a Digital Audio Broadcasting (DAB) ensemble without any bandwidth waste in a Digital Multimedia Broadcasting (DMB) transmission system. The key technologies of this algorithm include packet rate control and program clock reference correction, which are important for TS processing. The proposed algorithms are applied to the various DMB transmission systems based on Eureka-147, and the performance of the proposed algorithm is confirmed through the experimental DMB broadcasting.

Keywords—DAB, DMB, MPEG-2 TS, packet rate control, Ensemble.

I. Introduction

Since the Eureka-147 Digital Audio Broadcasting (DAB) system [1]-[3] was announced in the middle of the 1990s, many kinds of applications have been introduced in many countries in the world including Europe. Digital Multimedia Broadcasting (DMB) is one of the applications which have emerged from the Eureka-147 DAB system. Particularly in Korea, DMB focuses on the broadcasting of moving pictures and their reception in harsh conditions such as in places surrounded by high buildings and on highways where vehicles are moving at a very high speed. Although the DMB system is an improved version of the DAB system, the existing devices from the DAB system should still be used in many parts of the DMB system. In this paper, we propose an algorithm for effectively inserting an MPEG-2 transport stream (TS) into a DAB ensemble which constitutes the transmission frame in DMB, and we confirm the proposed algorithms with implementation in various DMB transmission systems, such as the DMB stream caster and Ensemble remultiplexer, to which the above algorithm is applied.

II. Layer Structure of the DMB System

The layer structure for a DMB system is illustrated in Fig. 1. The DAB ensemble is logically composed of a fast information channel (FIC) and main service channel (MSC), where the MSC is used to carry audio and data service components and the FIC is used for rapid access of information by a receiver. The MSC is made up of a sequence of common interleaved frames, transmitted every 24 ms. The smallest addressable unit of a common interleaved frame is the capacity unit (CU), the size of which is 8 bytes. An integral number of CUs are grouped together to constitute the basic transport unit of the MSC, called a sub-channel. In DMB, one sub-channel is used for one audio/video (AV) service, and the number of CUs for the transmission of an AV media stream is allocated and fixed by the multiplex configuration. To insert the AV media stream encoded by the DMB standard [4]-[5] into the DAB ensemble, it is capsulated into an MPEG-2 TS and Reed Solomon (RS) packet (204 bytes) generated by RS encoding[6]-[7]. The key point for the above process is that the size of a sub-channel in a DAB ensemble is exactly fixed by a multiple of 8 bytes; on the other hand, an AV media stream is not fixed but is at a constant bit rate. Moreover, the RS coding is processed by an RS packet unit. In this case, if the packet rate is controlled by an RS packet unit during a frame period (24 ms), the excess null stream may be filled at a part of the sub-channel so that it causes bandwidth waste. In this paper, to satisfy the two constraints—the processing of the RS coder by an RS packet unit and the ensemble composition by an 8-byte unit without any bandwidth waste—

Manuscript received July 21, 2004; revised Oct. 5, 2004.

Gwang Soon Lee (phone: +82 42 860 1676, email: gslee@etri.re.kr), Byungjun Bae (email: 1080i@etri.re.kr), Young Kwon Hahm (email: ykhahm@etri.re.kr), and Soo In Lee (email: silee@etri.re.kr) are with the Digital Broadcasting Research Division, ETRI, Daejeon, Korea.

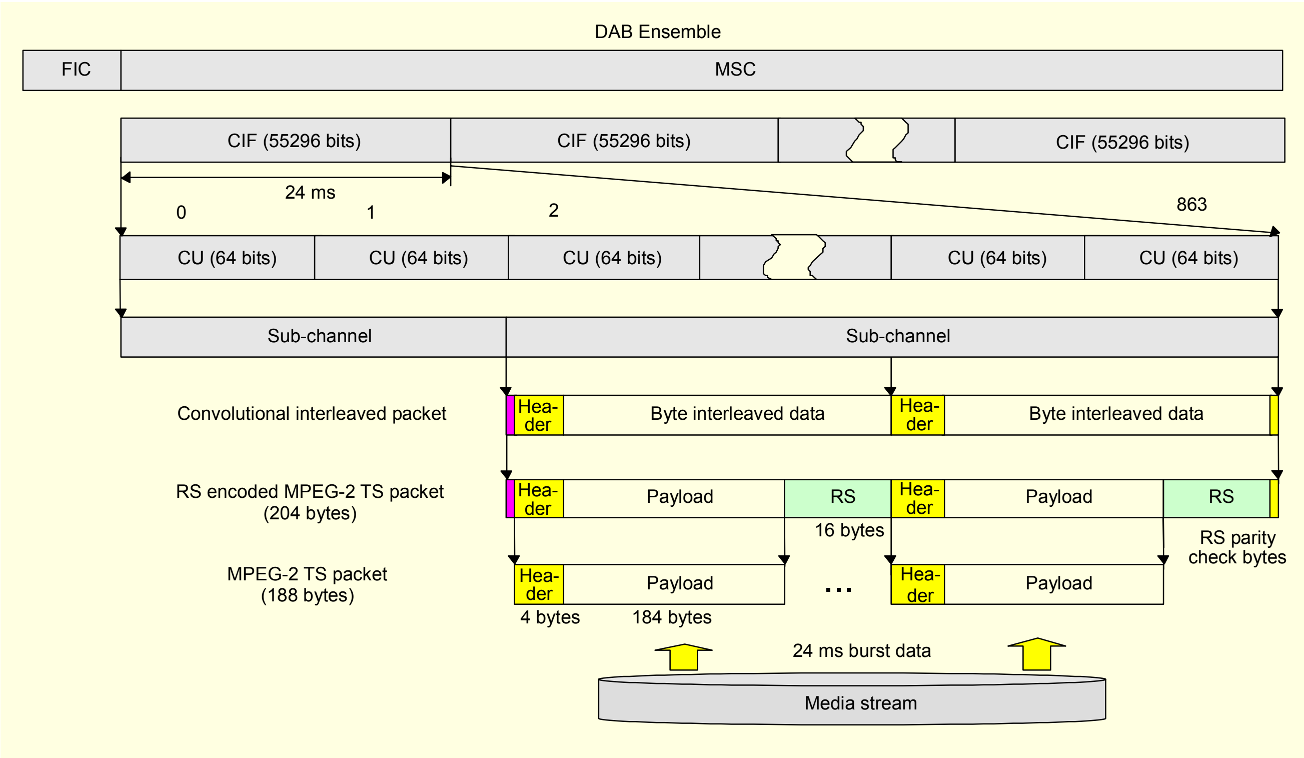

Fig.1. Layer structure of the DMB system.

we present a novel method to control the packet rate through a virtual buffer.

III. The Proposed Algorithm for Inserting an MPEG-2 TS

For inserting an MPEG-2 TS into the DAB ensemble effectively, we present the block diagrams as shown in Fig. 2. After a synchronizing process, the received MPEG-2 TSs are analyzed so that the bit rate of a payload MPEG-2 TS packet is calculated, excepting null packets, and the information is input into the management block. By using this information and sub-channel size configured by the FIC, the management block calculates the exact amount of data to be inserted into the DAB ensemble per frame period (24 ms) and sends this information to the packet rate control and stream data buffering block.

1. Packet Rate Control Algorithm

The major function of the packet rate control block is to supply the bytes by an RS packet unit (204 byte) to the outer coder as well as maintaining the exact number of bytes allocated at a sub-channel of the DAB ensemble. For this processing, we present a new method to control the packet rate through a virtual buffer that is conceptually connected to the packet rate control block. The status of the virtual buffer is calculated using division processing of both the sub-channel size and packet length per frame period.

In the proposed algorithm, the number of packets N_i written to the virtual buffer during the i-th frame period is a quotient of sub-channel size divided by RS packet size (204 bytes) P_{RS}, in the following way:

$$N_i = S \ DIV \ P_{RS} \,, \tag{1}$$

where DIV indicates integer division and S is the number of bytes allocated at the sub-channel during a frame period T_{fr} (24 ms). When B_{sc} is the bit rate of a sub-channel in the ensemble, S is computed as

$$S = (B_{sc} \times T_{fr}) / 8 \,. \tag{2}$$

The remainder R_m is then accumulated at R_T per each frame period as follows:

$$R_T = R_T + R_m \,, \tag{3}$$

where $R_m = S \,\%\, P_{RS}$, and $\%$ indicates modulus operation. Finally, if $R_T > P_{RS}$, the number of packets N_i written to the virtual buffer and the remainder R_T during the i-th frame period are updated as

$$N_i = N_i + 1, \tag{4}$$

$$R_T = R_T - P_{RS.}$$

In the above procedure, if the accumulated amount R_T exceeds

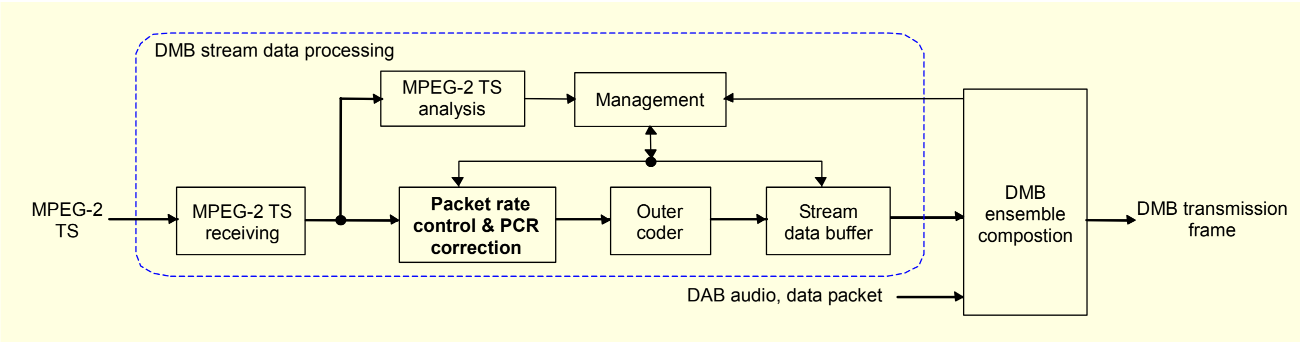

Fig. 2. Block diagram for effective inserting MPEG-TS into the DAB ensemble.

RS packet size P_{RS}, the number of packets written to the virtual buffer during one frame period is increased by 1. The output of a virtual buffer is always maintained as the same size as sub-channel size S per frame period. If there are no MPEG-2 TSs coming to the buffer, the null TS packets are generated and written to the buffer.

With the above processing, the status of the virtual buffer can be maintained without any overflow or underflow as shown in Fig. 3, satisfying the two constraints illustrated in section II.

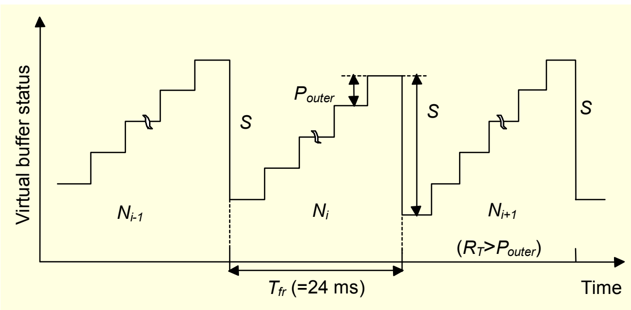

Fig. 3. Virtual buffer status after proposed packet rate control.

2. Program Clock Reference Correction Algorithm

Due to the removing and adding of null TS packets for the packet rate control, jitter may occur, which makes synchronization of the receiver unstable. Therefore, program clock reference (PCR) correction is essential for a stable MPEG-2 system. In a conventional method, due to the multiplication and division at the PCR correction process, the PCR correction [8] implementation becomes complicated.

The PCR in an MPEG-2 system is composed of the base field and extension field, driven at 90 kHz counter and 27 MHz counter, and is defined in [9] as

$$PCR(i) = PCR_{base}(i) \times 300 + PCR_{ext}(i)\,, \tag{5}$$

where $PCR_{base}(i)$ is the PCR base field value and $PCR_{ext}(i)$ is the PCR extension field value. In this paper, we present a simple

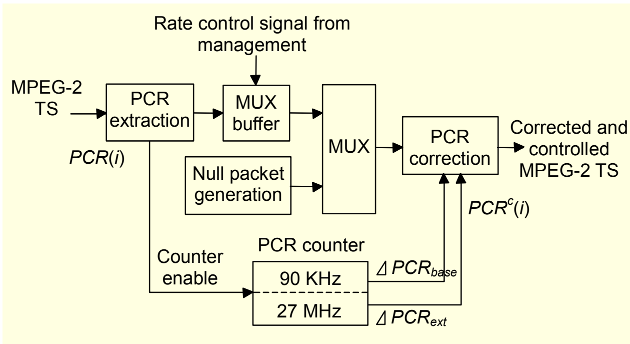

Fig. 4. The proposed block diagram for the PCR correction and packet rate control.

method to correct the PCR of an MPEG-2 TS, which is proper for the packet rate control introduced in this paper. The block diagram for the two proposed functions, packet rate control and PCR correction, is shown in Fig. 4. In the proposed method, the PCR is corrected at each of two counters—a 90 kHz counter for the PCR base and a 27 MHz counter for the PCR extension—so that its implementation becomes easier than in the conventional method, which needs multiple computations.

In Fig. 4, the PCR fields are extracted from the TS packets, and the PCR base and extension counter are increased respectively by the synchronized 90 kHz and 27 MHz clock pulse as soon as the PCR is extracted from the TS. Also, if there are PCR fields at the output of MUX, the PCR fields are corrected as

$$PCR^c_{base}(i) = PCR_{base}(i) + \Delta PCR_{base} + carry, \tag{6}$$

$$PCR^c_{ext}(i) = PCR_{ext}(i) + \Delta PCR_{ext}\,,$$

where ΔPCR_{base} and ΔPCR_{ext} are increased values of each counter, and *carry* is the carry value from the summation of the 27 MHz extension field. Finally, the corrected PCR is illustrated as

$$PCR^c(i) = PCR^c_{base}(i) \times 300 + PCR^c_{ext}(i)\,. \tag{7}$$

IV. Experimental Results

We verified the proposed algorithms through implementation of the DMB transmission system, such as the DMB stream caster and Ensemble remultiplexer [10]. The implemented DMB steam caster can be connected to the commercial DAB ensemble multiplexer using a TCP/IP connection, as shown in Fig. 5. In the DMB steam caster, an MPEG-2 stream file is able to be selected for transmission, and its bit rate is automatically calculated and controlled to be proper for a sub-channel of the ensemble, which is informed by the ensemble multiplexer. The results of experimental DMB broadcasting are shown in Table I. After video and audio sources are encoded by an MPEG-4 AVC of 512 kbps and an MPEG-4 BSAC of 96 kbps, they are encapsulated into an MPEG-2 TS of 796 kbps. In this Table, the fixed size of the sub-channel allocated for transmission of the MPEG-TS is 864 kbps. After the RS coding of the MPEG-TS, the bit rate of RS packets were maintained at about 863.74 (=796 × 204/188) kbps. From this result, we can see that the bit rate of the MPEG-2 TS is maintained almost as high as the bit rate requested by the sub-channel of the ensemble without bandwidth waste. Further, through the proposed PCR correction, the jitter of the PCR is kept at an acceptable range (±500 ns), which is specified by MPEG-2. The proposed algorithms are also applied to the Ensemble remultiplexer, which was implemented by full hardware.

V. Conclusion

We have newly developed the DMB system including its transmission system and receiver for a mobile TV service. Also, we have confirmed the effect of DMB broadcasting through an experimental broadcast, at which the digital TV contents were displayed without any problem under an80 Km/h bus speed in the coverage area. For this project in Korea, we have introduced an effective algorithm for inserting an MPEG-TS into a DAB ensemble without any bandwidth waste in the DMB transmission system, having implemented the DMB stream caster and Ensemble remultiplexer using the proposed algorithm. The proposed algorithm can be applied to any DMB transmission system for inserting an MPEG-2 TS into the DAB ensemble.

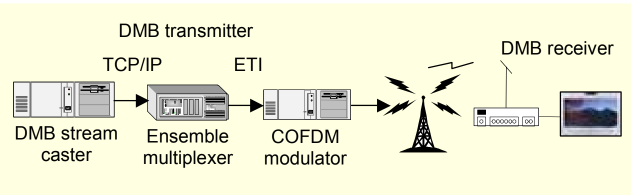

Fig. 5. Application of the implemented DMB stream caster for DMB broadcasting.

Table 1. The experimental result.

Sub-channel capacity		864 kbps, 648 CU
RS packet rate		863.74 kbps
MPEG-2 TS rate		796 kbps
Multimedia stream	Video	MPEG-4 AVC, 512 kbps, 15 fps
	Audio	MPEG-4 BSAC, 96 kbps
	System	MPEG-4 & MPEG-2
PCR jitter		<± 30 ns

Acknowledgment

We would like to thank the anonymous reviewers whose comments improved this paper.

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