THE SIGNAL PROCESSING OF A PROFESSIONAL USE DIGITAL AUDIO RECORDER

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Kazuhiro Sugiyama
Ken Onishi
Yoshinobu Ishida
Yasuo Kondou
Yoshitsugu Kusunoki
Tetsuya Yamaguchi
Mitsubishi Electric Corporation

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Kazuhiro Sugiyama*, Ken Onishi*, Yoshinobu Ishida*,
Yasuo Kondou**, Yoshitsugu Kusunoki**, and Tetsuya Yamaguchi**

*Consumer Electronics Development Laboratory
**Communication Equipment Works
Mitsubishi Electric Corporation

ABSTRACT

The PD(PRODIGI) format is the widely accepted format of the professional use stationary head digital audio tape recorders.

The X-86 two channel digital audio tape recorder based on this PD format has lots of features. The number of quantization bits is 20 bit linear slot and the error correcting code is the robust RSC code. Moreover, both insert recording which is called punch in/punch out and tape-cut editing, are capable.

This paper describes the digital signal processing of the X-86.

1. INTRODUCTION

In the two channel PD digital audio recorder with a 20 bit linear slot for audio quantization and a robust error correction code, high reliability is achieved and high quality sound recording and reproducing are available.[1][2] The authors report on the professionally used two channel digital audio recorder, X-86, based on this PD format.

2. SPECIFICATIONS AND SYSTEM STRUCTURE OF THE RECORDER

This recorder adopts the two channel PD format mode I shown in Table-1 and its specifications are given in Table-2.

Then the system block diagram is illustrated in Fig. 1. The signal processing circuits are replaced by 7 LSIs for achieving enhanced system reliability, low cost, and substantial scale down. Table-3 shows the LSIs list. Four identical pieces of the MODEM LSI and one piece of each LSI except for the MODEM LSI are used in one recorder.

The input/output terminal has not only line analog terminals but also digital I/O terminals. For the head configuration, There are 4 heads consisting of a preceding playback head, erase head, record head, and playback head. When the simultaneous monitor reproducing is operated, two channel digital audio signals are rearranged in the recording format via the A/D converter and each LSI, such as EDIT, INTERLEAVE, RSP, MODEM. Then the formated data is recorded on the magnetic tape via the recording amplifier.

On the playback side, the playback signal reproduced by the playback head is converted to the two channel audio signals and the audio signals are derived via the head amplifier, wave shaping circuit, and each LSI such as MODEM, ID.CONT, DEINTERLEAVE, RSP, EDIT, and D/A converter. In the punch in/out operation, the recording audio signal which is transferred to the INTERLEAVE LSI, is given by switching the output signal from the A/D converter or DIGITAL-IN terminal to the playback signal from the DEINTERLEAVE LSI.

The error correcting code is a sub-class of the generalized products code constructed from C2 composed of (16,12,5) Reed-Solomon code, C3 composed of (8,4,5) Reed-Solomon code and C1 composed of (344,328) CRC .[3] C2 is made from the digital audio data and C3 is made from the ID data. In this apparatus, an RSP type LSI (Reed-Solomon code Processing) can execute all the encoding and decoding of the Reed-Solomon codes C2, C3. The processing numbers of RSP during one block interval (1 msec) are 20 times for each encoding or decoding of the C2 code and 1 time for each encoding or decoding of the C3 code. Fig. 2 shows the sequence of the time sharing process. We don't use the algorithm that assigns a fixed processing time beforehand in each encoding or decoding, but rather a time sequence algorithm in which the next encoding (decoding) is started after one encoding (decoding) is completed. Even if some of the encoding (decoding) periods are temporarily different from the others, the RSP LSI is used in time sharing process unsynchronously. The RSP LSI is connected to the INTERLEAVE, ID CONT., DEINTERLEAVE LSIs in the common data-bus, and controlled by the time sharing processing controller.

3. SIGNAL PROCESSING OF THE EDITING

The digital audio recorder stationary head type can be edited both by tape-cutting and electronically. The advantages of tape-cutting and electronic editing include the following:

(1) Tape-cut editing
- It can be handled in the same manner as in conventional analog tape recorders.
- Editing is easy by using one recorder and editing time is short.

(2) Electronic editing
- Rehearsal editing is possible and it is possible to relocate repeatedly.
- Tape damage free editing is possible.

This digital audio tape recorder can use both editing schemes.

3.1 SIGNAL PROCESSING BY TAPE-CUT EDITING

The tape-cut editing has already been reported.[4] In this system, the following technologies have been newly developed.

(1) A two-mode decoding system is adopted, which can vary the error correcting ability depending on the tape-cut editing segment and other segments.
(2) The cross-fading time can be varied by optimizing the interleaving.

A conceptual diagram of signal processing of tape-cut editing is shown in Fig. 3. When the edited tape of (A) is played back, the reproduced signal of tape-1 and tape-2 can be taken out by overlapping in time by deinterleaving as shown in (B). In the bordering portion of tape-1 and tape-2, that is, in the edited portion shown in (B), the C2 code cannot be composed correctly. On the other hand, by C2's decoding method, in ordinary segments, it is necessary to correct up to 4 erasures by sufficiently utilizing the correction ability of the codes, but in the tape-cut edited segments, the risk of miscorrection increases. In this system, therefore, by employing the two-mode decoding method, the decoding algorithm is changed over to the editing mode greatly enhancing the error detection ability in the tape-cut editing region as compared with other ordinary regions. A decoding block diagram is shown in Fig. 4. At both ends (1),(2) of this editing segment, errors are corrected by setting up flags, and the sound quality is improved.

Moreover, this format is interleaved so that the cross-fading time can be varied in tape-cut editing, and 2.5, 5 and 10 msec can be selected in this model.

3.2 SIGNAL PROCESSING BY ELECTRONIC EDITING

In the case of electronic editing, there are two types of connecting methods which are used in case of switching from playback to recording or from recording to playback.

(1) Assembling or insert recording used by the punch in/out function
(2) After recording

In this model, a 4 head configuration is employed allowing both types of editing.

In the punch in(out) process, writing is started(ended) by using the time difference of the magnetic tape passing through the preceding(1st) reproducing head and the recording head so that the error correction code (C2) is connected to the same position of the re-coded tape as the reproduced music signals. It is difficult, however, to connect exactly because of the error in distance between the reproducing head and the recording head or the wow-flutter of the tape transport, and it is necessary, by some method, to fit the data array at the punch in/out's switching point which is generated by the recording-reproducing. Accordingly, for a 2-channel PD format, the ID area of each block is provided with a block address.

Fig. 5 shows the block format. ID1, ID0 means ID (identification data) which is needed for the system control data, such as sampling frequency, tape speed, emphasis ON/OFF and so on, and BA1, BA0 is a continuous block address having a 16 bit length. And the Reed-Solomon code is generated from this data for achieving high reliability. In this format, ID, BA data can be perfectly corrected if 4 track errors occur due to head clogging.

Now follows a description of the reproducing signal processing at the punch in/out. Fig.6 is a schematic drawing of the deinterleave memory for the reproducing signals including those before and after switching between recording and reproducing. (A) shows the case where the reproducing signal is given as it is into the de-interleave memory, and indicates that each symbol of the error correction code can not be composed again near the switching point, and that error correction isn't possible. (B) is the method used by this system (block address system). Reproduction of the block address is made in advance, the written address of the deinterleave memory is calculated from this block address, and continuous blocks are composed again on the memory. It is , therefore, possible to adjust the array of error correction codes, and the data of the dropped or duplicated areas are reproduced correctly.

4. CONCLUSION

The authors mentioned the signal processing of the two-channel digital audio recorder by the PD format, especially the two-mode decoding method and editing scheme. The development of the custom designed LSIs has made this recorder compact in its hardware size, inexpensive in production cost, high reliable, and a machine with extremely low power consumption. In the future, these LSIs will be used in other two-channel PD recorders. In conclusion, the authors would like to thank the following personnel, Messrs. M.Itoga, K.Shouji of Mitsubishi Electric Corporation.

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TABLE 1 TWO CHANNEL PD FORMAT VERSION

Mode	I	II
Sampling Frequency (KHz)	48/44.1	96/48/44.1
Quantization (bits/sample)	20	16
Number of Channels	2	2
Number of Data Tracks	8	8
Tape Speed (cm/s)	38.1/35.0	38.1/19.05/17.5
Transmission Rate (MBPS)	2.88/2.65	4.60/2.30/2.12
Linear Recording Density(KBPI)	24.0	38.4
Error Correction Code	Generalized Product Code C1 (344,328)CRC $\in$ GF(2) C2 (16,12)RS $\in$ GF(2^8)	
Redundancy (%)	31.1	31.1

TABLE 2 SPECIFICATIONS OF THE X-86

Number of channels	PCM 2ch Analog (cue) 2ch Time code 1ch Aux. digital 1ch
Tape speed	38.1 cm/s (15ips) ±10 %
Recording time	2 hours max. (14 inch reel size)
Tape	6.3 mm width digital audio tape
Sampling frequency	48 / 44.1 KHz
No. of quantization bits	16 bit linear (20 bit capable)
Error Correcting Code	RSC (Generalized Product Code)
Modulation code	2/4 M (Run Length Limited Code)
Editing	Tape cut & Punch in/out
Frequency response	20 Hz to 20 KHz +0.5/-1.0 dB
Dynamic range	Over 90 dB (Unweighted rms)
Distortion	Less than 0.05 %, 50Hz to 20 KHz

TABLE 3 THE SIGNAL-PROCESSING LSIS

No.	MODEL	FUNCTIONS	PACKAGE
1	EDIT	Concealment and crossfade, Level control	100 pin-Flat
2	INTERLEAVE	Interleaving, ID and CI code generation.	100 pin-Flat
3	RSP	Encoding and Decoding of Reed-Solomon Code	100 pin-Flat
4	MODEM	2/4M modulation and demodulation, Sync. Pattern Gen.& Det.	80 pin-Flat
5	ID.CONT	Punch in/out detection, CI check	100 pin-Flat
6	DEINTERLEAVE	Deinterleaving, Splice detection	100 pin-Flat
7	CLOCK & SERVO	System clock generation, Servo control	100 pin-Flat

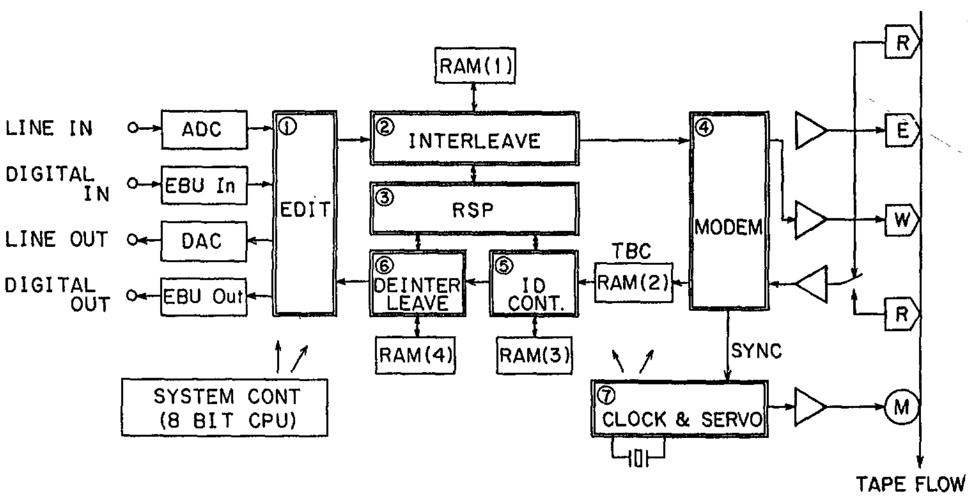

FIG. 1 SYSTEM BLOCK DIAGRAM

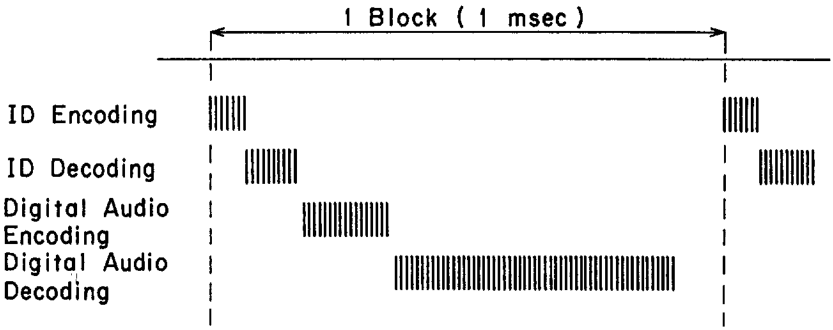

FIG.2 TIME SHARING CONTROL OF THE RSP LSI

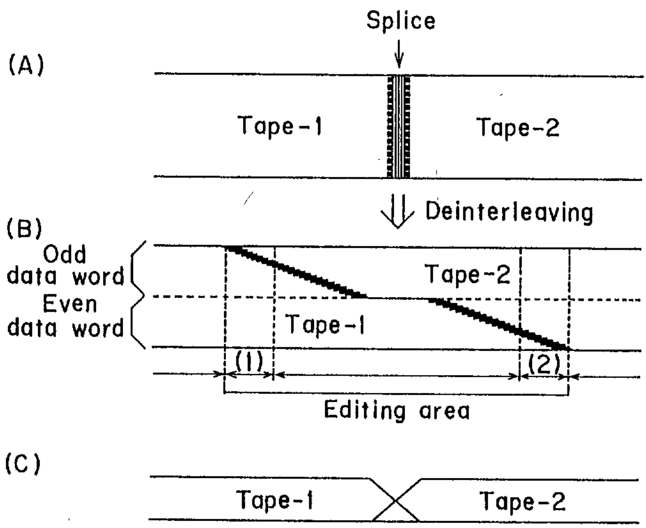

FIG.3 TAPE-CUT EDITING

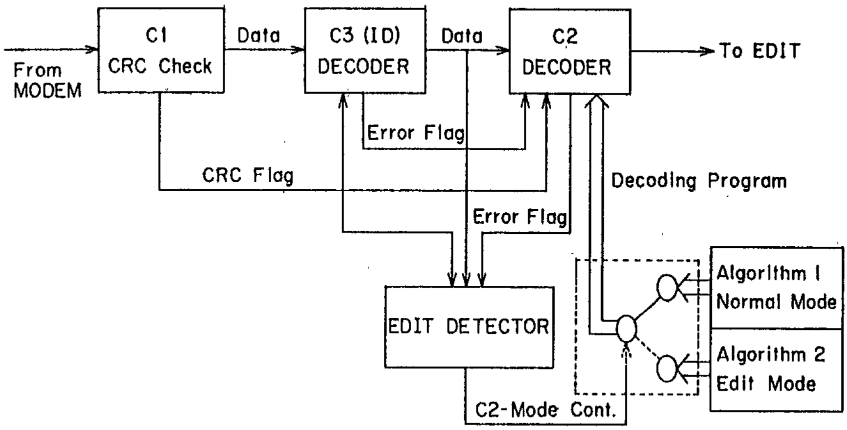

FIG.4 TWO-MODE C2-DECODING

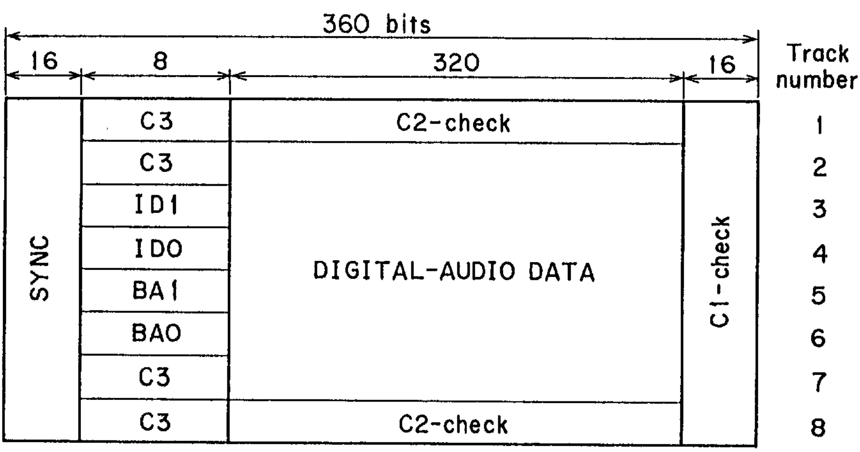

FIG. 5 BLOCK FORMAT

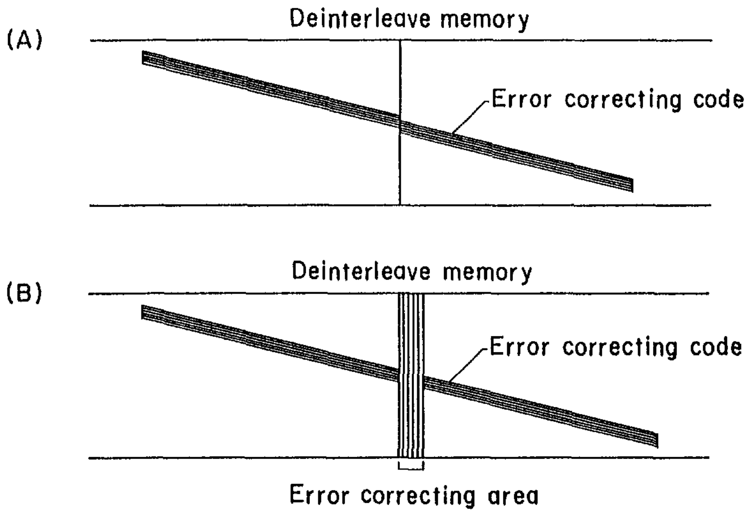

FIG. 6 ERROR CORRECTING PROCESS OF THE PUNCH IN/OUT ERROR