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IMPROVED PROFESSIONAL USE
2 CHANNEL DIGITAL AUDIO RECORDER

2270 (A-4)

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**Presented at
the 79th Convention
1985 October 12-16
New York**

AUDIO ES

AES

AN AUDIO ENGINEERING SOCIETY PREPRINT

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ABSTRACT

The stationary head 2-ch digital audio recorder with tape-cut editing has been well received by professional users ever since it was introduced in 1980, mainly because of its extremely high reliability and high performance.
To cope with the increasing demands for two channel analog audio, auxiliary functions and so on, the authors have developed the improved signal format. The punch in / punch out, the optimum splice, and the electronic editing, etc, have been installed.

1. INTRODUCTION

There are two kinds of systems of the professional use 2-ch digital audio recorders. One is the stationary head open reel type using a multi-track recording method [1] and the other is the PCM conversion processor using a VCR [2]. Both of these have been put in practical use since around 1980.

As to consumer use digital audio system breakthroughs, there was the standardization of PCM processor (EIAJ STC-007, 1979), the sale of the CD player (1982), the debut of digital satellite broadcasting in Japan (1984), and the release of technical specifications of DAT (Digital Audio Tape recorder, 1985) and so on. Therefore the 2-ch digital master recorder is more and more necessary. Recently the progress of LSI technology, magnetic head, and magnetic tape have been remarkable, and by these technologies the professional use recorder has gained higher reliability and easier handling.

This paper describes the required specifications, error protection methods, tape-cut and electronic editing, and so on, for the signal format of the stationary head digital audio recorder.

2. OUTLINE OF THE TAPE FORMAT

2.1 REQUIREMENTS FOR THE 2-CH DIGITAL AUDIO RECORDER

There are two kinds of professional use 2-ch digital audio recorders.

1. For master recording use; This is utilized for the mixdown from the multi-channel recorder, or for the transmission for disk cutting, and it is important that the system has high efficiency, high reliability and easy handling for tape-cut editing.

2. For program exchange use; This is utilized for FM or DBS broadcasting, or program exchange between recording studios or broadcasting stations. It is important that the system has high reliability and that the recording time is as long as possible.

Moreover, the signal format requires the following:

3. Considering the LSI technology and the device progress, in the near future the signal format should have a simple extension of the sampling frequency, the quantization bit, and those compatible functions.

4. It should have parameters which enable digital audio interface, CD sub-code interface, and synchronization with a VCR.

2.2 BASIC PARAMETERS

The authors have adopted the following basic parameters considering the above requirements.

Basic Parameters	Contents
Number of tracks	12 Total tracks
	8 Digital audio
	2 Auxiliary analog
	1 Time code
	1 Auxiliary Digital
Tape speed	7.5 / 15 ips
Error correction	RSC-IV code
Modulation scheme	2 / 4 M [3]
Sampling frequency	48 / 44.1 kHz
Quantization	16 bits linear

2.3 TRACK FORMAT

Fig.1 shows the track format on the tape. The main track shall be used for recording digital audio signals. Sub-track No.1 shall be recorded in SMPTE time code and shall be used for auto locating, synchronization with a VCR, etc. Sub-track No.2 shall record auxiliary digital data for a users requirement and the recording of the CD sub-code signal. Sub-track No.3 and No.4 shall record the audio signal for cueing.

2.4 TAPE FORMAT

Fig.2 shows the frame and block configurations. The same block configuration is used at the tape speeds of 7.5 and 15 ips. The total length of one frame is 360 bits, where 16 bits are used for the synchronizing signal, 8 bits are used for the control signal, 320 bits are used for the digital audio signal or the C2 check code, and 16 bits are used for the C1 check code. Fig.3 shows interleaving in the case of tape speed at 15 ips. Continuous wording (even and odd sample) such as W0, W1, is recorded on the tape at intervals of 36 frames with the track position separated. The error correcting code consists of black words which are separated 4 frames from each other.

2.5 RSC-IV CODE

The degradation of the error rate is anticipated in the case of the tape speed at 7.5 ips because the linear recording density is about 40 kbpi. On the other hand, in the case of 20 kbpi, the frame error rate is good from about 10^{-6} to 10^{-7}.[4] The authors have adopted the RSC-IV code that has efficient error correcting capabilities even if the frame error rate is 10^{-3}, the error is located at the spliced point, or at the punch in / punch out point.

The RSC-IV code is a sub-class of the generalized products code.[5][6] This code is constructed by the (16,12,5) Reed-Solomon code over $GF(2^8)$, as code C2, and by the (344,328,4) CRC over GF(2), as code C1. Symbol interleaving has been performed between C2 encoding and C1 encoding. Generator polynomials for code C2, and code C1 are given by the following equations.

$$G_2(X) = \prod_{i=0}^{3} (X + \alpha^i)$$

$$G_1(X) = X^{16} + X^{12} + X^5 + 1$$

, where α is a root of the primitive polynomial, $X^8+X^4+X^3+X^2+1$.

3. ERROR CORRECTING CAPABILITY

3.1 PERFORMANCE

Burst errors which occurs in the tape-running direction, are detected by CRC's. One frame length consists of 344 bits. A RS codeword consists of 16 symbols which are taken two symbols within them. The RS code corrects up to 4 erasures which are detected by CRC's. As it is rather difficult to make a proper model fitting for burst channel, the authors evaluate the performance analysis on random error channels. As is well known, numerical results of random error channels will give us more severe conditions than those of burst channels.

(1) Performance of the CRC

The CRC based upon CCITT V41 is used. The generator polynomial is already given by $G_1(X)$. Undetection probability P_M per one frame is approximately,

$$P_M = 2^{-16} \times \binom{43}{2} Ps^2 (1-Ps)^{41}$$
$$= 1.37787 \times 10^{-2} Ps^2$$

where, Ps gives the symbol error rate. Therefore, detection probability P_D per frame is,

$$P_D = \sum_{i=1}^{43} \binom{43}{i} Ps^i (1-Ps)^{43-i} - P_M$$
$$\cong 43Ps$$

where, P_M gives the frame undetection probability. The symbol erasure rate, which means the ratio of errors with error detection flags versus whole errors, is given by,

$$Pst \cong (1/43) \times \binom{43}{1} Ps$$

$$= Ps$$

The false erasure rate, which means the ratio of the correct symbol with error detection flags versus whole errors, is given by,

$$Psf \cong (42/43) \times \binom{43}{1} Ps$$

$$= 42Ps$$

and the symbol error rate, which means ratio of errors without flags versus whole errors, is given by

$$Pse \cong P_M$$
$$= 1.37787 \times 10^{-2} Ps^2$$

(2) Decoding for (16,12,5)RS code

The (16,12,5)RS decoder corrects up to 4 erasures detected by CRC. The decoding results are shown in Table 1. Two kinds of compensation methods were adopted. One method is to compensate only erasure symbols with flags when more than 3 erasures come, because of the beyond error correcting capability in that case.

This is called compensation type A. On the other hand, if uncorrectable errors are detected, and the number of flags is less than 5, there entire RS codeword is compensated. This is called compensation type B.

3.2 NUMERICAL RESULTS

Probability of the occurrence for mis-corrected sound is given approximately by,

$$Pm \cong \frac{5}{16}\Big[A_5(16)\binom{5}{2}\left(\frac{Pse}{255}\right)^2\binom{3}{1}\left(\frac{Pst}{255}\right)$$

$$+A_5(16)\binom{5}{1}\left(\frac{Pse}{255}\right)\binom{4}{2}\left(\frac{Pst}{255}\right)^2\binom{2}{1}\left(\frac{254}{255}Pst+Psf\right)$$

$$+A_5(16)\binom{5}{1}\left(\frac{Pse}{255}\right)\binom{4}{2}\left(\frac{Pst}{255}\right)^2\binom{2}{2}\left(\frac{254}{255}Pst+Psf\right)^2$$

$$+A_5(16)\binom{5}{3}\left(\frac{Pst}{255}\right)^3\binom{2}{2}\left(\frac{254}{255}Pst+Psf\right)^2\Big]$$

$$+\frac{1}{16}\binom{16}{1}Pse\binom{15}{1}Pst\binom{14}{4}(Pst+Psf)^4+\ldots$$

Probabilty of occurrences for compensated sound is also given approximately by,

$$Px \cong \binom{16}{2}Pse^2\binom{14}{1}Pst - A_5(16)\binom{5}{2}\left(\frac{Pse}{255}\right)^2\binom{3}{1}\left(\frac{Pst}{255}\right)$$

$$+\binom{16}{1}Pse\binom{15}{2}Pst^2\binom{13}{1}(Pst+Psf)$$

$$-A_5(16)\binom{5}{1}\left(\frac{Pse}{255}\right)\binom{4}{2}\left(\frac{Pst}{255}\right)^2\binom{2}{1}\left(\frac{254}{255}Pst+Psf\right)$$

$$+\binom{16}{1}Pse\binom{15}{2}Pst^2\binom{13}{2}(Pst+Psf)^2$$

$$-A_5(16)\binom{5}{1}\left(\frac{Pse}{255}\right)\binom{4}{2}\left(\frac{Pst}{255}\right)^2\binom{2}{2}\left(\frac{254}{255}Pst+Psf\right)^2$$

$$+\binom{16}{3}Pst^3\binom{13}{2}(Pst+Psf)^2$$

$$-A_5(16)\binom{5}{3}\left(\frac{Pst}{255}\right)^3\binom{2}{2}\left(\frac{254}{255}Pst+Psf\right)^2$$

$$+\frac{5}{16}\binom{16}{1}Pse\binom{15}{1}Pst\binom{14}{4}(Pst+Psf)^4+\ldots$$

Fig.4 shows the RSC-IV performance curve.

4. EDITING

The editing function is important for the professional recorder, and it is desirable that the signal format enable both tape-cut editing and electronic editing.

4.1 TAPE-CUT EDITING

The authors have already developed reports about the PCM recorder which has achieved tape-cut editing.[7][8] There are a lot of problems to be solved when it is applied to the PCM recorder. Those main problems and solutions are as follows:

1. Loss of the phase lock of the capstan servo at the spliced point. This is solved by the selection of a sub-survo pulse series or adoption of a lower phase lock frequency.

2. Large drop-out at the spliced point. This is solved by adopting interleaving, or removing of error data using memories.

3. Signal level discontinuity at the editing point. This is smoothed by fading-in and fading-out.

After the above basic problems are solved, a reproduced signal should appear as in fig.5-1. The signal before the editing point changes from a normal reproduced area to an error correction area, then to an error interpolation area. The signal after the editing point changes exactly as the reverse process. Fade-in / fade-out is achieved where both the before and after signals were reproduced at the same time. Interleaving at the signal format is decided by an interval of tape-cut editing and a fade-in / fade-out time.

An interval of tape-cut editing is selected at a few centimeters, and this value is about the same as with an analog recorder and therefore the exchange of a musical note of a piano, for example, is capable. At this time, the fade-in / fade-out time becomes about 20 msec. maximum. From fig.5-2, the fade-in / fade-out time decreases when a gap at a spliced point occurs. If memories are used, the time frame will increase. Fig.5-3 shows this. For example, if a gap at a spliced point is set at -13mm, that is, if the signal after a spliced point is read ahead of normal reproducing, the fade-in / fade-out time can be set up perfectly free.

As to the re-recording of a spliced tape, the reproduced signal is completely corrected by the powerful error correcting code and no degradation of audio signal generates.

4.2 ELECTRONIC EDITING

For the signal format, which enables the electronic editing without the signal degradation, tape-cut editing is required. It is necessary that the punch-in / punch-out be performed at an editing point, and that the overwrite be performed at a front and rear portion of the punch-in / punch-out so as not to degrade the audio signal. Accordingly, there are two kinds of head arrangements as shown in fig.6. In the case of both head arrangements, the process of decoding, signal mixing and encoding must be made while the tape goes from the PB-2 head to the REC-2 head. Therefore, the interleaving length is restricted by a practical head interval. Fig.7 shows a general idea of the punch-in / punch-out function. The signal data discontinuity and error occurs at a boundary between the original portion and the overwrite portion. If the control signal at every block is used, a re-arrangement of signal data is capable, and error data can be completely corrected by the error correcting code.

5. CONCLUSION

The signal format for the professional use 2-ch digital audio recorder has been developed, which has high reliability and flexibility for various requirements. This format is considered not only for the use in recording studios and broadcasting stations, but also for the use in program exchanges and for the extension of basic parameters in the future. The major improved points of this format are as follows:

1. There are 4 auxiliary tracks for cueing (L ch and R ch), time code, and users digital data.
2. Tape-cut editing and electronic editing (punch-in / punch-out) is available.
3. Long play mode (tape speed 7.5 ips) is available.
4. A powerful error correcting code (the RSC-IV code) is adopted.

This basic model professional use 2-ch digital audio recorder based upon this improved format is inexpensive and easy to handle.

6. ACKNOWLEDGMENT

The authors would like to thank the following personnel of Mitsubishi Electric Corporation; Messrs. M. Itoga, K. Ishii and S. Kunii for their support to the project, Messrs. Y. Kawakami, Y. Kusunoki, M.Ozaki and other members for the development of this signal format, Mr. B. Bearman for his assistance of this paper.

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NUMBER OF ERRORS

NUMBER OF ERASURES		0	1	2	3
0		C	C	C	M/B
1	(○)	C	C	C	M/B
	(●)	C	C	M/B	M/B
2	(○○)	C	C	B	M/B
	(●○)	C	C	M/B	M/B
	(●●)	C	C	M/B	M/B
3	(○○○)	C	C	B	M/B
	(●○○)	C	C	M/B	M/B
	(●●○)	C	M/B	M/B	M/B
	(●●●)	C	M/B	M/B	M/B
4	(○○○○)	C	C	A	M/A
	(●○○○)	C	C	M/A	M/A
	(●●○○)	C	M/A	M/A	M/A
	(●●●○)	C	M/A	M/A	M/A
	(●●●●)	C	M/A	M/A	M/A
5	(○○○○○)	C	C	A	M/A
	(●○○○○)	C	A	M/A	M/A
	(●●○○○)	C	M/A	M/A	M/A
	(●●●○○)	M/A	M/A	M/A	M/A
	(●●●●○)	M/A	M/A	M/A	M/A
	(●●●●●)	M/A	M/A	M/A	M/A

C --- CORRECT DECORDING
A --- COMPENSATION TYPE-A
B --- COMPENSATION TYPE-B
M --- MISCORRECTION

TABLE.1 C2-DECORDING MAP

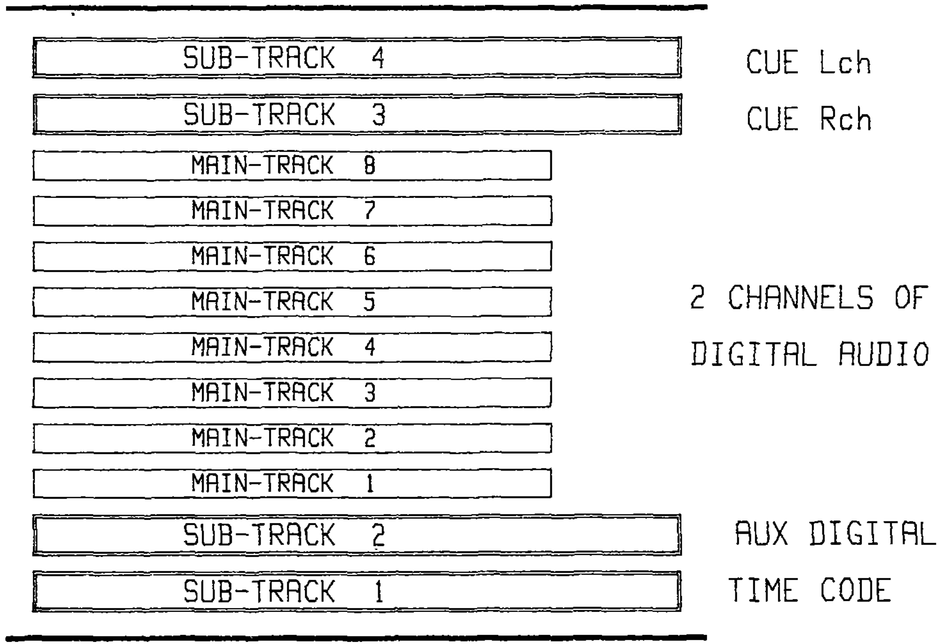


FIG.1 TRACK FORMAT

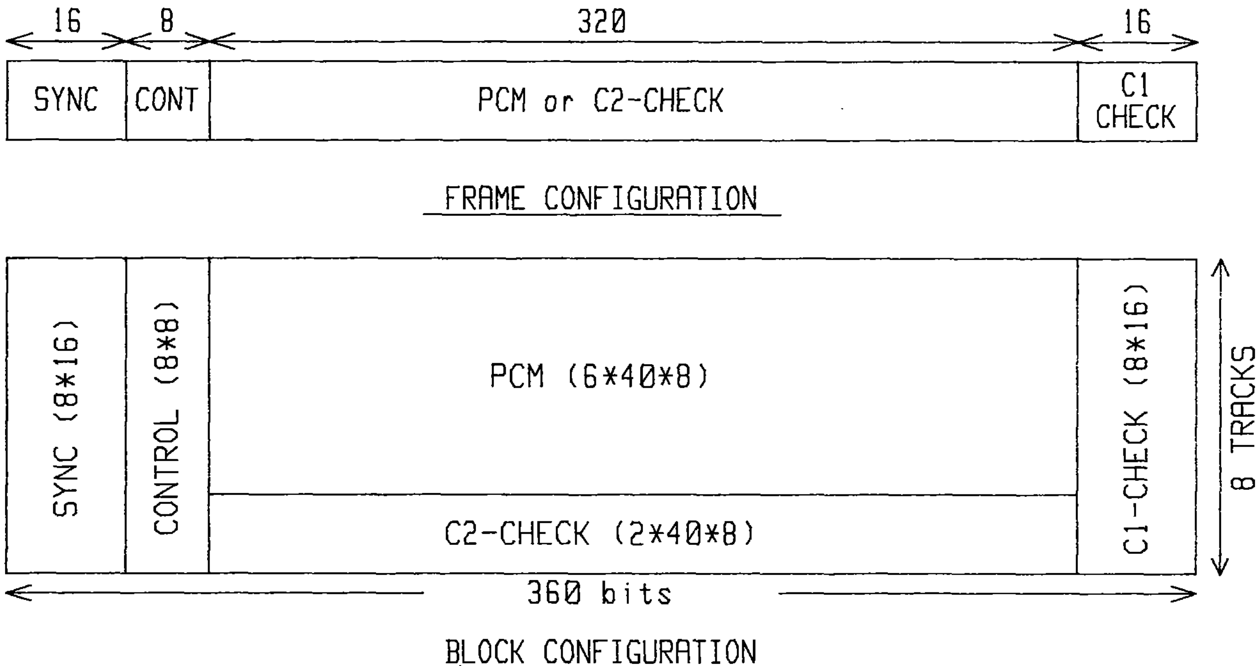


FIG.2 FRAME AND BLOCK CONFIGURATION

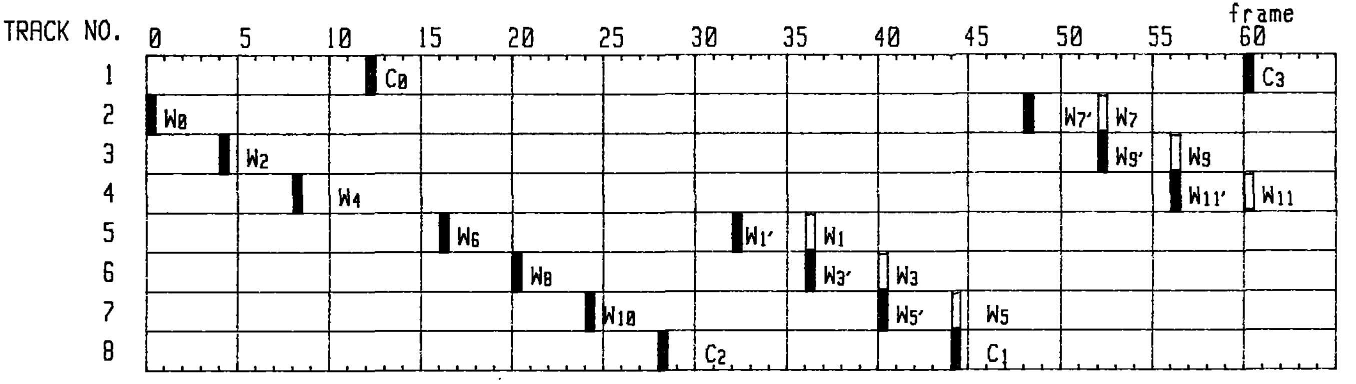


FIG.3 INTERLEAVING ON THE TAPE

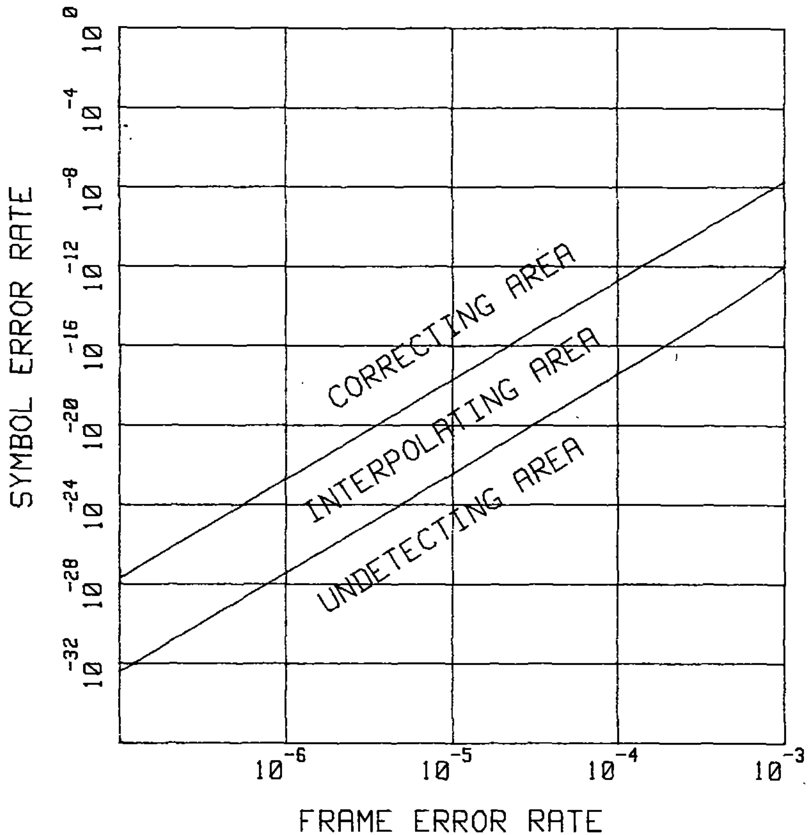
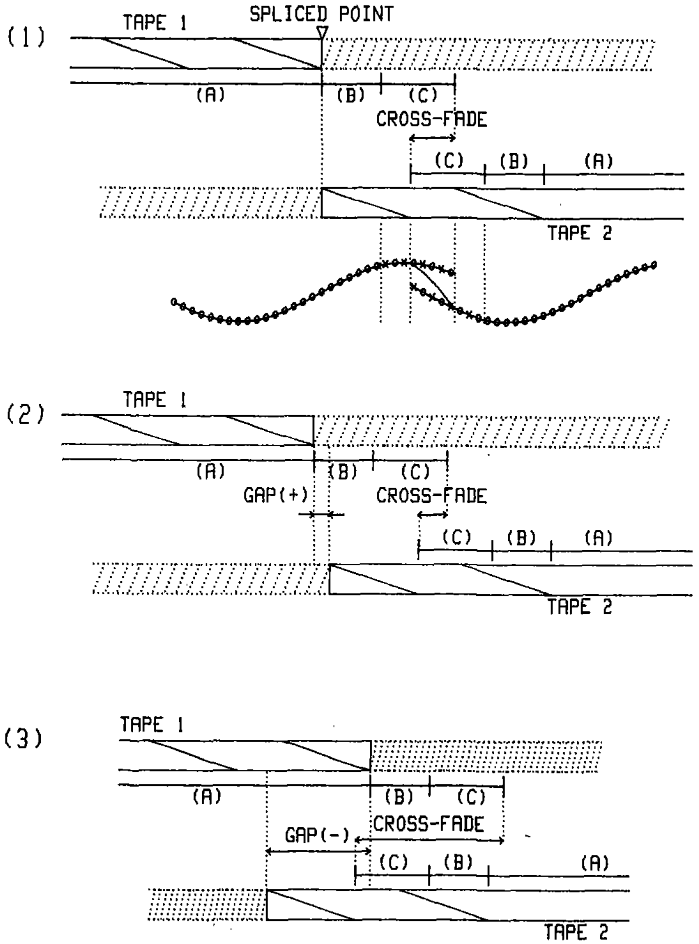


FIG.4 ECC PERFORMANCE CURVE



(A) NORMAL REPRODUCTION AREA
(B) ERROR CORRECTION AREA
(C) ERROR INTERPOLATION AREA

FIG.5 SIGNAL PROCESSING OF TAPE-CUT EDITING

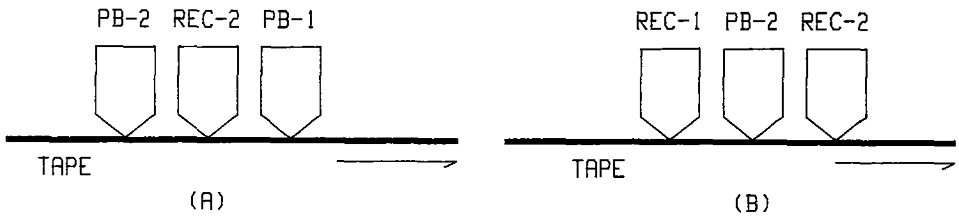


FIG.6 HEAD ARRANGEMENT

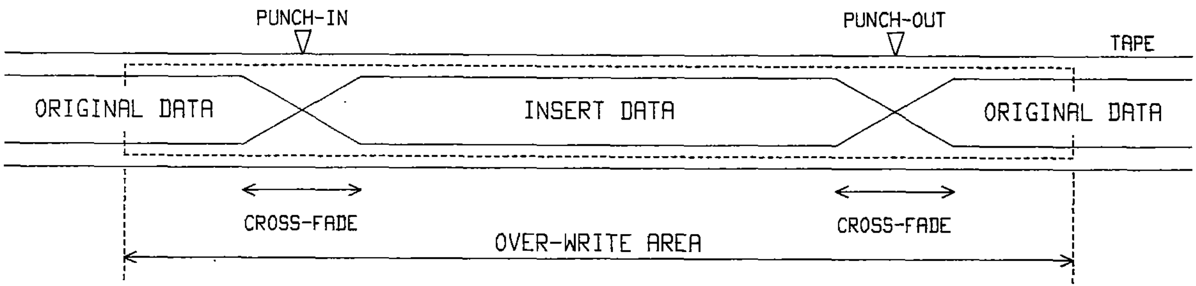


FIG.7 A GENERAL IDEA OF THE PUNCH-IN AND PUNCH-OUT FUNCTION