A FORMAT OF STATIONARY-HEAD DIGITAL AUDIO RECORDER COVERING WIDE RANGE OF APPLICATION

1677(H-6)

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Presented at the 67th Convention 1980 Oct. 31/Nov. 3 New York

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ABSTRACT

A format is developed for stationary head digital audio recorder. The channel number covers from two (1/4" tape) to 48(1" tape). The nominal speeds are settled as 30, 15, and 7.5 IPS, and the tracks assigned for one channel are different as one, two, and four accordingly. While the basic electronic boards are kept as same in any speed. A strong error correction is adopted for protection from dusts, scratches and finger prints, as well as smooth punch-in/out and tape-splice editing.

1. INTRODUCTION

In conventional analogue tape recording for professional application, the speeds of 30ips, 15ips and 7.5 ips are commonly used. The reason why three of tape speeds are necessary is to solve verious demand of users among the trade-off between quality and type consumption.

The same thing is expected also in digital audio. It will not be very far away that the digital audio means not only a very high grade technology which is required by limited people but the most popular recording method covering wide range of application. It should be noted that one side of digital compact disc will have a playing time of more than one hour, which requires very low tape speed for master recorder such as 15ips or 7.5ips otherwise a huge size of tape reel is necessary.

On the other hand, for multi-channel recorder, one track per one digital sound channel is desirable for economical reasons. In this case higher tape speed is unavoidable and tolerable as far as it is sufficient to record one piece of music.

A very flexible format of stationary head tape recording is developed according to the user's requirement [1].

As to the tape speed, the above mensioned three speeds are adopted because any objectionable reason can not be found. According to the modern technology, number of tracks required to record one channel of digital audio sound quantized 16 bit and sampled at 50.4 KHz, are one, two, and four corresponding to the tape speed of 30ips, 15ips and 7.5ips. Ofcause for expermental purposes, much more density is possible by very sophisticated devices and technology, but cost and reliability should be considered preferentially.

The sampling rates are selected as 50.4 KHz, 44.1 KHz and 32 KHz. The tape speed is reduced at the lower sampling rates and in total nine speeds are prepared.

One of the remarkable points of this format is that the circuit boards for digital signal processing can be common to all of these versions. Therefore one machine covering several versions will be easily designed.

Table 1 shows the rough description of the format. The channel number is prepared from two to 48.

A new channel coding (modulation scheme) with density ratio of 1.5 is developed in order to improve reliability. Very strong error correcting scheme named cross-interleave code [2] is adopted with some modifications which is essential for punching in/out and tape-cut editing.

Two of analogue tracks for editing purposes, one external track for time code or mixing information, and one digital track for internal control are prepared separately from digital sound tracks.

The details are described in the followings.

2. SOURCE ENCODING

A quantization is adopted according to the recommendation of AES Digital Audio Technical Committee [3].

The dynamic range achieved by 16 bit linear is considered as more than enough as far as the converter is really accurate (this is very difficult today). But if someone would like to use wider dynamic range in the future, DPCM can be used without changing the main hardware. In this case, a dynamic range of 110dB at 1 KHz can be obtained. In order to indicate this kind of information, sufficient number of control bit is prepared in each channel, which will be useful to design a machine with automatic switch for quantization.

3. SAMPLING RATE

Three sampling rates of 50.4 KHz, 44.1 KHz, and 32 KHz are adopted. The 32 KHz is a transmission standard in Europe and in Japan, and the most of the broadcaster's applications are considered as sufficient in this frequency.

The 44.1 KHz is adopted as a sampling rate for the digital compact disc system [4]. Therefore, if master tapes are prepared in this frequency, they can be easily used for cutting of digital compact disc without going through analogue signal, and without using an expensive rate converter.

There are two frequencies for domestic digital audio recorders using video format, which are 44.056 KHz for NTSC and 44.1 KHz for PAL or SECAM [5][6]. The difference is only 0.1% and if the recorded tapes are delivered at 44.1 KHz even for NTSC area, there will not be any problem (i.e. the speed tolerance of analogue tape recorder is no better than ± 0.1%). As to the professional application, 44.1 KHz is quite recommendable rather than 44.056 KHz, not only by the reasons mentioned above but also the integer relationship to 32 KHz [7].

The 50.4 KHz is proposed because it has the interger relationship of 8 to 7 to 44.1 KHz [7],[8],[9].

4. PACKING DENSITY

Fig. 1 shows an example of the relationship between block error rate and recorded wave length. While heads and tapes of normal video tape recorders are used and a simple waveform equalizer with three ports is adjusted at every point. In other words, this characteristics can be obtained by economical devices which are being produced in large quantity today.

It is shown that the wave length below 1.5μm is a little bit dangerous and that over 2.0μm will not largily help to improve error rate. Therefore the wave length of λ = 1.8848 m is chosen with a new channel coding named HDM-1 at the density ratio of 1.5. (see Chapter 5)

Table 2 shows bit rate, maximum frequency to be recorded, minimum wave length to be recorded, recorded bit length, and the packing density for each verion of the format. The format is designed so as to keep the packing density equal to each version of tape speed and sampling rate.

5. CHANNEL CODING. (MODULATION SCHEME)

A new channel coding named HDM-1 (HDM=High Density modulation) is developed in order to improve reliability at high packing density. The main characteristics of HDM-1 are shown below.

TABLE 3 HDM-1

1.	Window Margin;	$T_W = 0.5T$
2.	Minimum Distance Between Transition;	$T_{min} = 1.5T$
3.	Density Ratio;	D.R. = 1.5
4.	Maximum Distance Between Transition;	$T_{max} = 4.5T$
5.	Constraint Length;	$L_c = 5.5T$
6.	Clock Rate;	CLK = 2/T

T = The length of one bit of input data

The density ratio is equal as 3PM [10], but T_{max} is improved from 6T to 4.5T and L_c, from 9T to 5.5T. T_{max} is important to keep good clock recovery at splice-edited point, at punched in/out point and at long burst errors. The hardware required for modulator and demodulator is simpler than 3PM, and is not very much expensive than that of MFM.

6. BLOCK STRUCTURE

One block for error correction consists of 288 bits by the following structure.

TABLE 4 Block Structure

Item	Word Number	Bit Number
cynchronization	-	11
Block Address	-	3
Flag Bits	-	2
Data Words	12	192
Parity Words for Error Correction	4	64
Parity Word for Error Detection	1	16
Total	18	288
Redundancy	33.3%	
Efficienty	66.7%	

Flag bits are forming one frame by eight blocks which is indicated by block address, and in total, sixteen bits are prepared to indicate some kind of informations necessary for individual tracks, such as emphasis or quantization.

7. ERROR PROTECTION

7.1 EVALUATION OF ERROR PROTECTION

Since early stage of digital audio recording, a method of single erasure correction has been adopted as an error protection against dropouts [11],[12],[13]. This method is capable of correcting one word out of one block with the aid of error detection code like CRCC (Cyclic Redundancy Check Code).

Fig. 2 shows block error rate after and before correction while one block is 288 bits. The curve of above mentioned single erasure code is proportional to the square of error rate before correction.

Recently, codes with stronger error correctability are introduced to the field of digital audio, such as "b-Adjacent Code" or "Read-Solomon Code" [6],[14]. Those codes are classified as double erasure correction code, and the curve of error correction is proportional to the third power of error rate (Fig. 2).

A very strong error correcting scheme named "Cross Interleave Code [2] is developed by authors especially for the application to digital audio disc systems [15], the curve of which is shown in Fig. 2 to be proportional to the fourth power of error rate with the same redundancy as double erasure correction codes.

It has been discussed in many literatures about correctability of codes in the manner such as "the error rate is measured as 10^{-5} so that the probability of un-corrected block is once in every ten hours for single erasure code, once in every century for double erasure code, and once in every ten million years for cross interleave code." This discussion is completely wrong, and un-corrected block will be observed quite more often than once in ten hours for single erasure code with using very clean tapes. It is not necessary to wait ten million years to hear clicks from a system with cross interleave code. This is because the real errors are combination of random errors, short burst errors, and long burst errors, but Fig. 2 is a results of the calculation only for random errors.

The authors have been proposing an evaluation method using two parameters, error rate and error correlation [16][17]. Fig. 3 shows a plane with two axis of error rate and error correlation. Good clean tapes of broadcasting quality is located on this plane around error rate of 10^{-5} and correlation coefficient of 0.3. The area for dirty tape is very difficult to define, because the parameters are so much distributed by each tape, and by each portion on the tape. Fig. 3 shows a concept that in dirty tapes not only error rate but also correlation are larger than clean tapes. If synchronization network or servo system of the machine are not properly working, a short error is expanded into a longer one, and the correlation coefficient is observed as larger than original one.

Therefore the whole area of Fig. 3 is important for the evaluation of error correction codes, tapes and machines.

7.2 FINGER PRINTS

Finger print is one of the weak points of digital tape recorders comparing to analogue recorders. Fig. 4 shows an example of dropout caused by a heavy finger print and its estimated spacing between tape and head.

In analogue recording, this spacing loss will reduce higher frequency compornent, but still music can be reproduced because the major part of power spectrum stays in lower frequency.

In digital recording, however, the lack of high frequency component will easily make peak shift and code errors, and almost nothing can be reproduced at that point. After investigating finger print problems, the following conclusions are obtained.

(1) Heavy burst error is unavoidable at the point of finger print.

(2) It is not very easy to clean up finger prints once they have adhered.

(3) This problem can not be solved by reducing packing density as far as it is denser than one tenth of that shown in Table 2. Significant difference can not be observed and heavy burst error will still exist at lower packing density.

(4) The treatment of the machine should be recommended as not to touch tapes.

(5) Error correction should solve a solitary finger print which might be accidentally put on the tape.

(6) There is not any good solution if a lot of finger prints are put on narrow area of the tape.

7.3 OUTLINE OF THE CODING

The word sequence is devided into odd and even numbers, and the cross interleave code is applied independently to odd and even sequence. One block of error correction consists of six words of each sequence, and two parities P and Q are formed before and after interleaving by sub-delay "d = two blocks", A unit delay of "D = seventeen blocks" makes another interleave for burst error correction [2].

The odd and even sequences are separated by large delay "$\mathcal{D}$ = 204 blocks", which makes possible to carry out cross-fading at splice edit points, without changing time scale on the tape.

The words are distributed into two or four tracks for lower speed version as is shown in Fig. 5. It should be noted that the burst correction length D is equal to each version, but the concealment is different because it depends on the distance between odd and even sequence.

The values of d, D, and $\mathcal{D}$ are optimized so as to obtain the best correctability in the condition that the total delay time for encoding and decoding to be a certain value which is defined by the requirement from editing.

7.4 THE PERFORMANCE OF THE CODE

7.4.1 RANDOM ERROR CORRECTION

Random error correctability of cross interleave code is shown in Fig. 2, which is not full ability but practical values by triple decoder [2] [15]. Further more correctability is expected by more-multiple decoder with keeping full compatibility of the code [2] [15], but it does not seem to be necessary for practical use.

High rate of random error is seldomly observed in the field, but it can happen when the adjustment of wave form equalizer is not suitable or S/N ratio deteriorates by some reasons. In this case, error rate of 10^{-2} is still tolerable by this code.

7.4.2 BURST ERROR CORRECTION

Table 2 shows burst error correctability of each version. The format is designed according to the following policies.

(1) The most of scratches and dusts should be perfectly corrected. For this reason the burst correctability of distance of 3 mm on the tape is kept for each version of the format. In other words, if 3 mm of tape is completely destroyed, exact the same data as encoded one can be reproduced by virture of the error correcting code.

(2) Finger prints or large scratches which might be scarcely adhered should also be corrected in sufficient quality.

In order to correct certain erroneous area, some other area is required to be error free. This area is called as guard space. Fig. 6 shows area of perfect correction and its required guard space. For medium speed version, the area of perfect correction is either of (3.08mm) x (half tape width) or (1.54mm) x (full tape width). For low speed version, both the area and the guard space are reduced into approximately half size.

Fig. 7 shows the area of finger print protection. High speed version has the biggest area, and it is reduced according to the speed. The area for low speed version is not considered as sufficient for finger print, and careful treatment is necessary.

Fig. 8 shows strength against an accident by which the most of the data are destroyed. If 89% of the data on the tape are lost, still a poor concealment method is possible, which should interpolate three words out of every four words. For instance, words W_1, W_5, and W_9 are alive, but W_2, W_3, W_4, W_6, W_7, and W_8 are lost. In this case surprisingly good result is obtained if the values of interpolated words are calculated as follows, as far as it does not happen frequently.

$$W_2' = \frac{3}{4} W_1 + \frac{1}{4} W_5$$
$$W_3' = \frac{1}{2} W_1 + \frac{1}{2} W_5$$
$$W_4' = \frac{1}{4} W_1 + \frac{3}{4} W_5$$
$$W_6' = \frac{3}{4} W_5 + \frac{1}{4} W_9$$
$$W_7' = \frac{1}{2} W_5 + \frac{1}{2} W_9$$
$$W_8' = \frac{1}{4} W_5 + \frac{3}{4} W_9$$

Where upper script ' means interpolated value. The multiplication by coefficient 1/2 or 1/4 can easily be executed by shifting the word one or two bits, and that of 3/4 is obviously their combination.

7.4.3 EVALUATION ON THE PLANE OF ERROR RATE AND CORRELATION

Fig. 9 shows a result of computer simulation, where triple decoder is assumed. Even for the nominal value for clean tape; error rate of 10^{-5} and correlation of 0.3, correctability is approximately four orders worse than the results of random error shown in Fig. 2.

It is considered that the word error rate of 10^{-8} after correction is sufficient for practical use, if machine has some means of concealment. And if the value is worse than 10^{-4}, user might hear some clicks.

Fig. 10 shows an evaluation on the plane of error rate and error correlation. The plane is devided into three regions. The region (i) is recommended for practical use, and the block error rate after correction is better than 10^{-8}. The region (ii) needs some warning because there are some troubles on the machine or on the tape; the error rate is between 10^{-4} and 10^{-8}. The region (iii) should not be used, because the error rate is worse than 10^{-4}.

8. EDITING

8.1 PUNCHING IN/OUT

In analogue recorders, the procedures of punching in and out is just playback, record, and playback, but in digtal, it is devided into seven modes shown in Fig. 11.

(i) Playback Mode; The cross fader does not work in this mode and the played back signal goes to monitor speakers directly.

(ii) Playback-Record Mode; The played back signal is re-recorded at the same place on the tape. The distance between playback head and record head is adjusted by digital delay memory. Again, the signal just pass through the cross fader. This mode works very short period just before punching in point.

(iii) Cross Fade Mode; The signals from microphone and from playback head are mixed and are automatically cross-faded. The mixed signal is recorded on the tape.

(iv) Record Mode; The signal from microphone is fed to record head directly.

(v) Cross Fade Mode; The same process as mode (iii), but in this case the signal from microphone is gradually decreased and that from playback head, increased.

(vi) Playback-Record Mode = (ii)

(vii) Playback Mode = (i)

Thus, digital cross fading is easily executed and far smoother punching in and out is possible comparing to analogue machines. The real discontinuity exist between mode (i) and (ii), and (vi) and (vii), because the tolerances of the distance between two heads and that of tape stretch are far larger than one bit of recorded signal. Therefore, always one block is destroyed by this discontinuity.

This problem can be solved by the following method, without using inter-block-gap.

(1) The switching point for entry and out going of recording is not arbitrary, but specified, for instance, every other D-blocks.

(2) By virture of cross interleave code, the two sequences

for error correction (P-and Q-series) do never meet [2]. Above mensioned switching points are always on P-series, and related blocks belonging to Q-series are therefore never destroyed after how many times punching in and out is proceeded at the small area on the tape.

(3) If one of the error correction series is destroyed, the cross interleave code is equivalent to single erasure code. Namely, this code keeps the error correctability equivalent to single erasure code after loosing a lot of blocks by punching in and out.

(4) The time necessary to wait for the next entry point is 4 m.sec. for 50.4 KHz and 6.4 m.sec. for 32 KHz version, and will not cause any noticable delay of punching in.

8.2 TAPE SPLICE EDITING

At the spliced point of the tape, it is unavoidable to loose a lot of data because compliance of tape is suddenly changed at the point. The tape-splice editing is performed by the following strategy.

(1) Time scale on the tape should not be changed because it is very difficult to make an electronic editing of splice-edited tapes if time scale is changed at the point and it is adjusted by tape speed afterwards.

(2) Cross fading should be done for smooth editing.

(3) Therefore, the information beyond the spliced point should be reproduced by some ways, and overlap should be prepared.

(4) An interleave between odd and even words is prepared for that purpose. As is shown in Fig. 12, overlapping is possible after de-interleave the odd-even delay. The lost area at the spliced point is dispersed and back-uped by good data.

(5) All parts are fully covered by error correction because the code is independent in odd and even requence of words.

(6) Lost area should be concealed by interporlation, and the overlapped area should be cross-faded for smooth editing.

9. CONCLUSION

The authors have been developping several stationary-head formats shown below in these several years.

(i) X-22DTC = 2 channel, quater inch tape machine, with tape speed of 15 ips. Using b-adjacent code for error correction. March 1978.

(ii) PCM-3224 Prototype = 24 channel, one inch tape machine with tape speed of 22.5 ips. A combination of single erasure correction code and very strong concealment ability is used. November 1978.

(iii) PCM-3204 Prototype = 4 channel, quater inch tape machine with tape speed of 15 ips. Cross interleave code is used. November 1979.

After field testing these machines, with the aid of experience from various rotary head machines [5], the following points are concluded.

(1) Packing density is increased significantly within several years (Fig. 13), but reliability has improved greatly.

(2) The ability of random error correction will contribute to improve reliability against miss adjustment or various defects of machines.

(3) The condition where machines are used practically is far worse than that of laboratory, and various error protection methods should be prepared in order to save the precious source even if the most of the data are destroyed.

(4) The ability for editing should be treated the most carefully. Cross fading is recommended for smooth editing.

(5) There are so many fields of application of digital audio, and single speed with single sampling rate is almost impossible.

The format described above chapters is developed according to these conclusions. The points of the format are shown below.

(a) The versions of three speeds and three nominal sampling rates are prepared in order to cover wide range of application. The digital circuitry can be common to all of these versions.

(b) A new modulation scheme named HDM-1 is developed in order to improve reliability at high packing density.

(c) A new error correction code named cross interleave code is developed to get strong error correctability with low redundancy.

(d) The structure of cross interleave code is a combination of two set of error correction code with different interleave, and this is utilized in punching in/out. The blocks destroyed by switching between playback and record mode are designed always to belong to one error correction series, and the correctability of single erasure correction is always kept despite of loosing a lot of blocks.

(e) Cross fading can be applied to tape on splice edited point without changing time scale. This is made possible by applying the code independently to odd and even series and by making a large delay between two series. Thus the half of the data from both sides of the spliced point are overlapped with full ability of error correction.

(f) Error correction methods are prepared as the following four steps.

    (i) perfect error correction

    (ii) good error concealment

    (iii) normal error concealment

    (iv) poor error concealment.

    The correctable area, and the guard space are different in each of the method, and even if 89% of the data is destroyed in some small constraint area, the method (iv) is possible to apply.

(g) An evaluation method using a plane of error rate and error correlation is developed, which is considered to describe the ability in the field far better than just calculating random and burst error correctability. But it should be noted that for medium and low speed version,correlation between tracks should also be taken account. This point can not be treated in this paper.

ACKNOWLEDGEMENT

The authors would like to express sicere gratitude to Dr.Heitaro Nakajima, Managing Director of Sony Corporation for his consistent support, and Messieurs T.Sonoda, M.Kubo, T.Watanabe, C.Kanai, Y.Kishi, H.Fujikura, N.Nakamura, I.Iwamoto, and Y.Sako of the project team, for their contribution in this study. The authors also gratefully acknowledge the helpful suggestions of Dr. R.Lagadec of Willi Studer.

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Table 1 The Outline of the Format

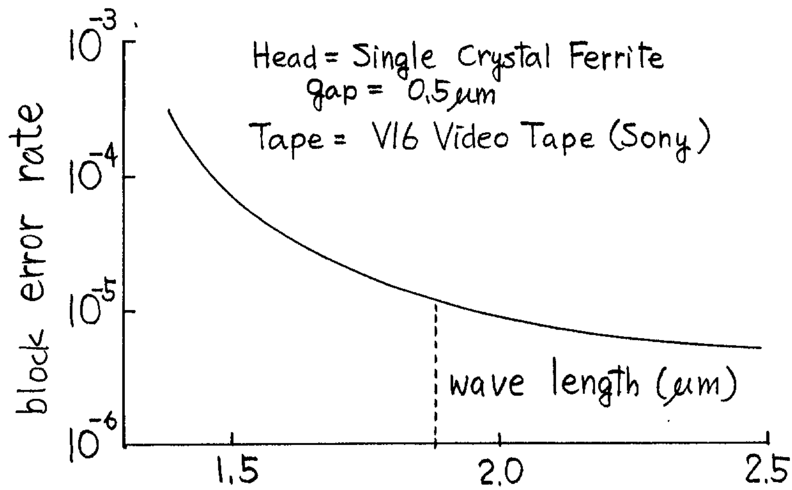
Version		High Speed			Medium Speed			Low Speed
Tape Width (inch)		1/4	1/2	1	1/4	1/2	1	1/4
Number of Digital Channels		8	24	48	4	12	24	2
Number of Tracks per Digital Channel		1			2			4
Analogue Track		2						
External Track		1						
Samplimg Rate f_s(KHz)		50.4 or 44.1 or 32.0						
Tape Speed	f_s = 50.4 KHz	76.00			38.00			19.00
(cm/s)	f_s = 44.1 KHz	66.50			33.25			16.63
at	f_s = 32.0 KHz	48.25			24.13			12.06
Quantization		16 Bit Linear / Word						
Channel Coding		HDM - 1						
Error Correction		CRCC , Cross - Interleave						
Redundancy		33% (including parities and sync. word)						

Table 2 Packing Desity and Burst Error Correctability of the Format

Version		High Speed			Medium Speed			Low Speed		
Sampling Rate (KHz)		50.4	44.1	32.0	50.4	44.1	32.0	50.4	44.1	32.0
Tape Speed (cm/s)		76.00	66.50	48.25	38.00	33.25	24.13	19.00	16.63	12.06
Bit Rate/TRK (kb/s)		1,209.6	1,058.4	768.0	604.8	529.2	384.0	302.4	264.6	192.0
Density Ratio (Channel Coding)		1.5 (HDM-1)								
Max. Frequency to be Recorded (KHz)		403.2	352.8	256.0	201.6	176.4	128.0	100.8	88.2	64.0
Min. Wave Length to be Recorded (um)		1.8844								
Bit Length (um)		0.6281								
Packing Density (bqi)		40,426								
Burst Error Correction	Length (mm)	5.427*			3.075					
	Time (msec)	7.141*	8.161*	11.25*	8.095	9.252	12.75	16.19	18.50	25.50
	Bits	8,640*			4,896					
Burst Error Protection (Good Concealment)	Length (mm)	21.35			15.20 x 1/2 tape width			12.12 x 1/4 tape width		
	Time (msec)	28.08	32.10	44.24	40.00	45.71	63.00	63.78	72.29	100.5
	Bits	33,984			24,192			19,296		
Burst Error Protection (Normal Concealment)	Length (mm)	36.90			24.60 x 1/2 tape width					
	Time (msec)	48.56	55.49	76.48	32.37	37.00	50.99			
	Bits	58,752			39,168					
Burst Error Protection (Poor Concealment)	Length (mm)	52.28			26.14			13.07		
	Time (msec)	68.81	78.64	108.38	68.81	78.64	108.38	68.81	78.64	108.38
	Bits	83,232			41,616			20,808		

* These values are theoritical limit by multiple decoder, while other values can be obtained by single decoder.

Fig 1 Block Error Rate vs Packing Density



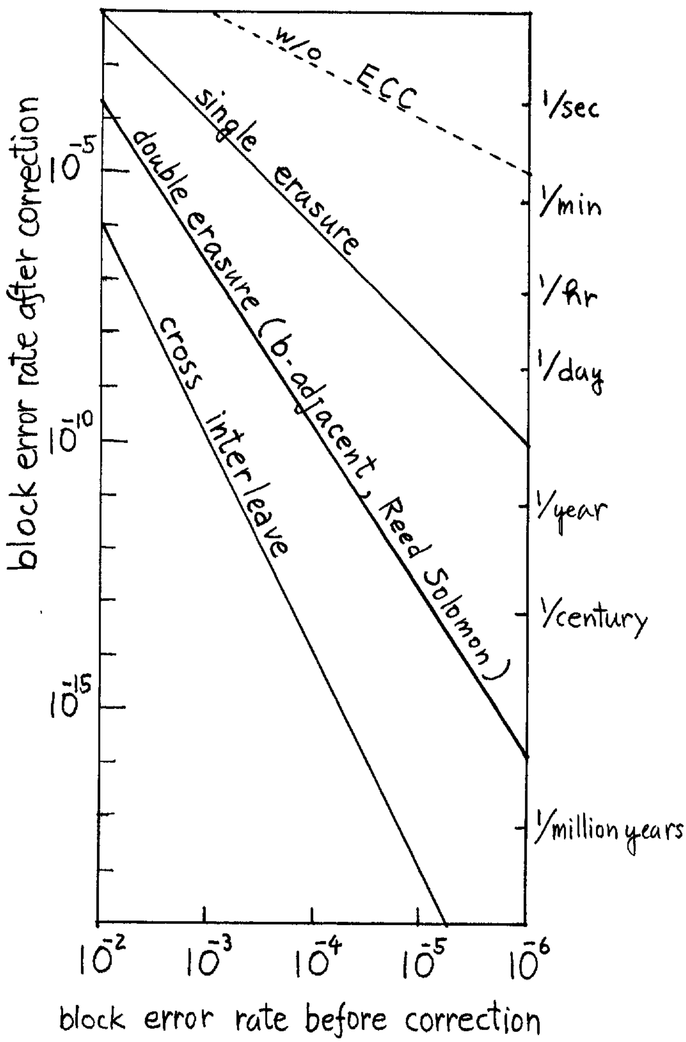
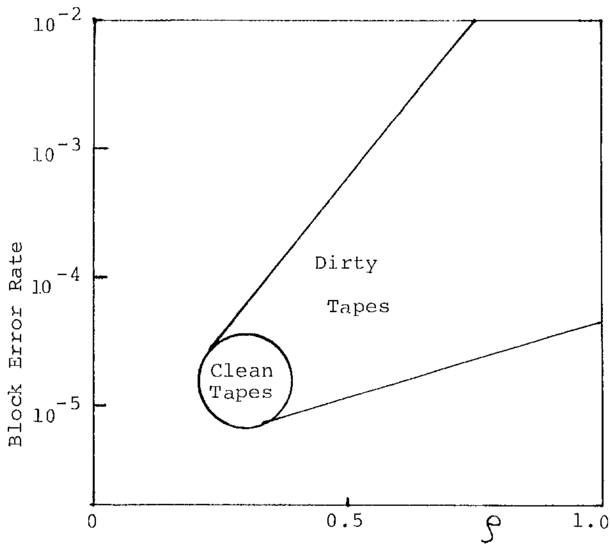


Fig.2 Random Error Correctability of Various Codes

Fig 3 Error Characteristics for Clean & Dirty Tapes



Block Error Correlation Coefficient

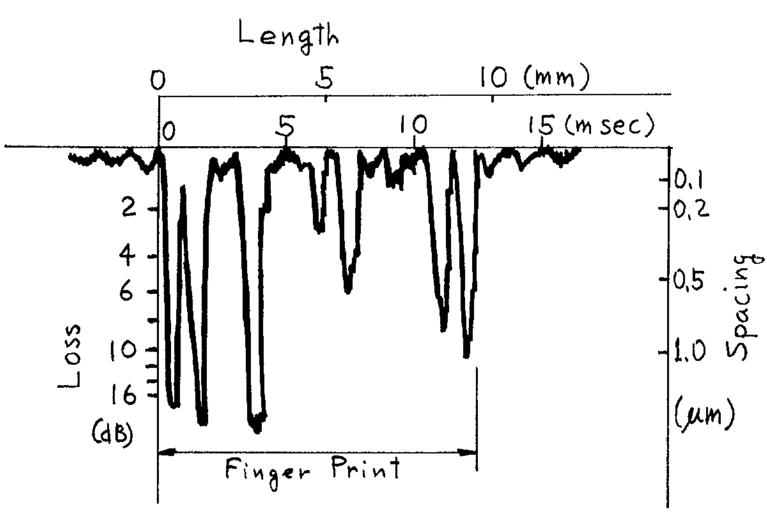


Fig.4. An Example of Heavy Finger Print

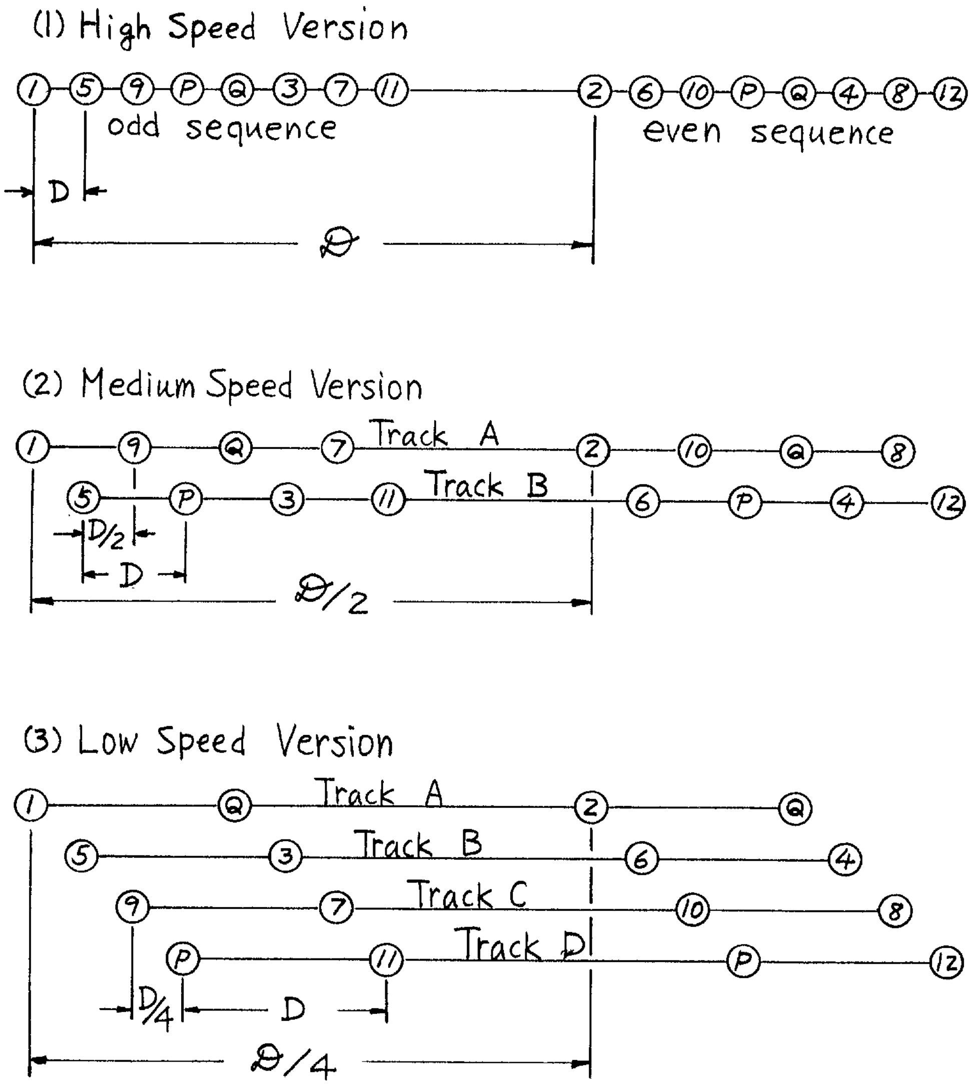
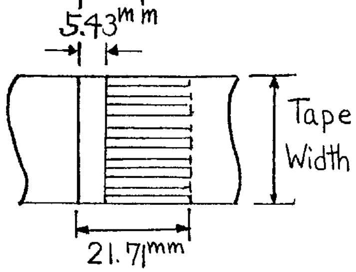
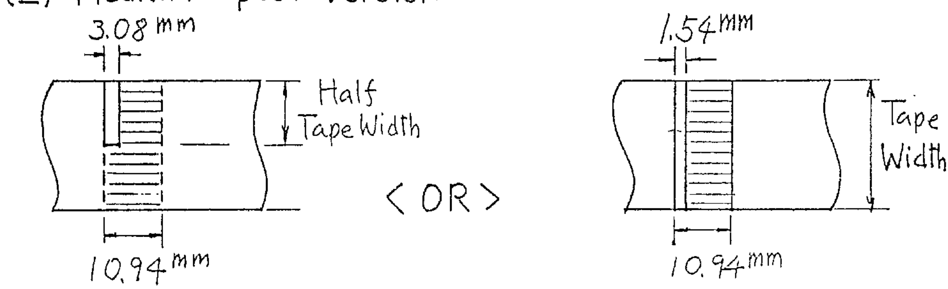


Fig. 5 Word Sequence on the Tape

(1) High Speed Version



(2) Medium Speed Version



(3) Low Speed Version

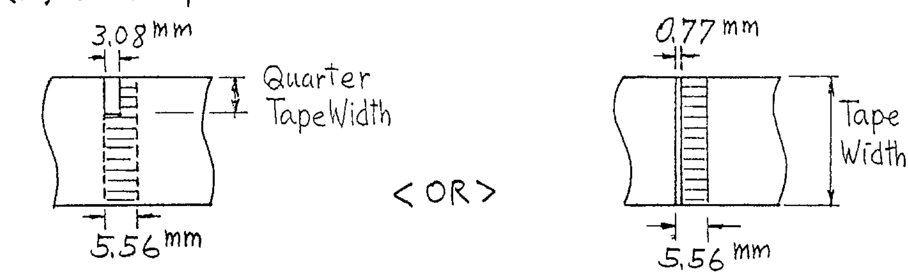


Fig. 6 Correctable Area and Guard Space

The Area of Perfect Correction

Guard Space Necessary

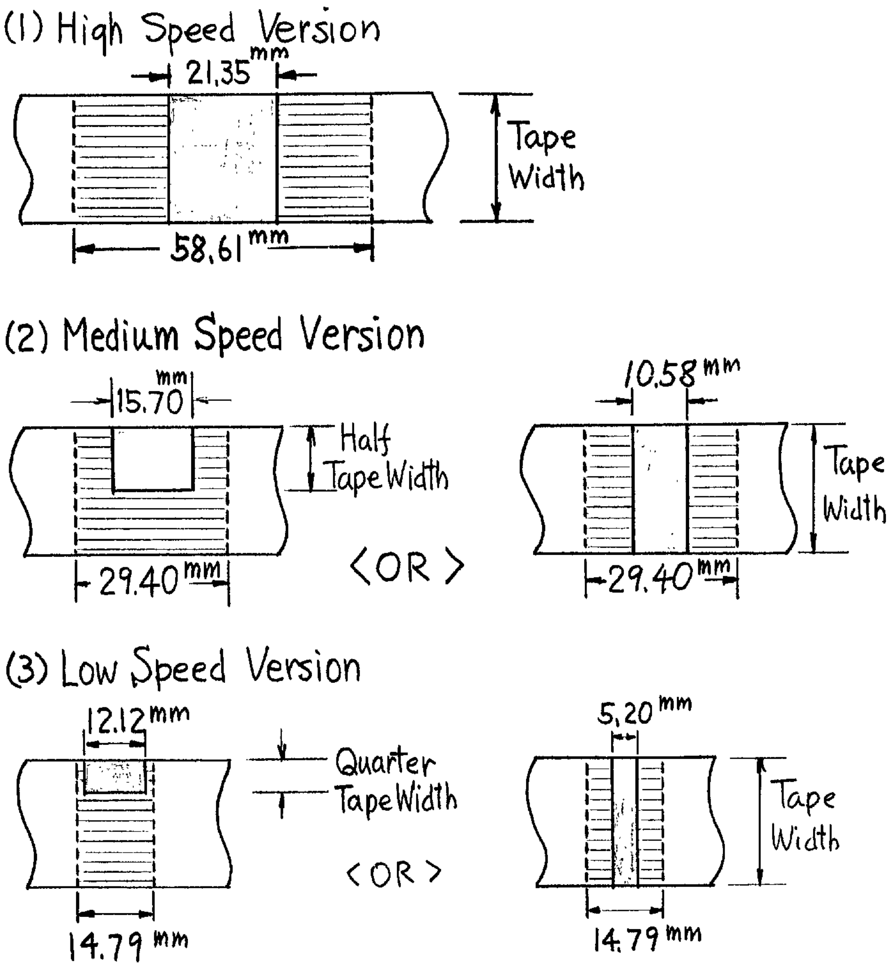
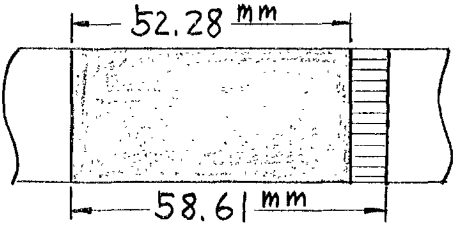


Fig 7 Finger Print Protection

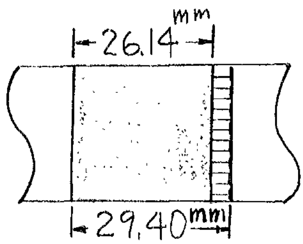
Correctable Area by Good Concealment

Guard Space Necessary

(1) High Speed Version



(2) Medium Speed Version



(3) Low Speed Version

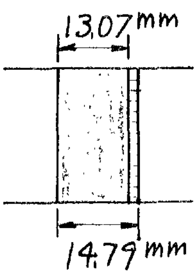


Fig 8 Accident Protection

Correctable Area by Poor Concealment

Guard Space Necessary

Fig 9. Error Correction v.s. Error Correlation

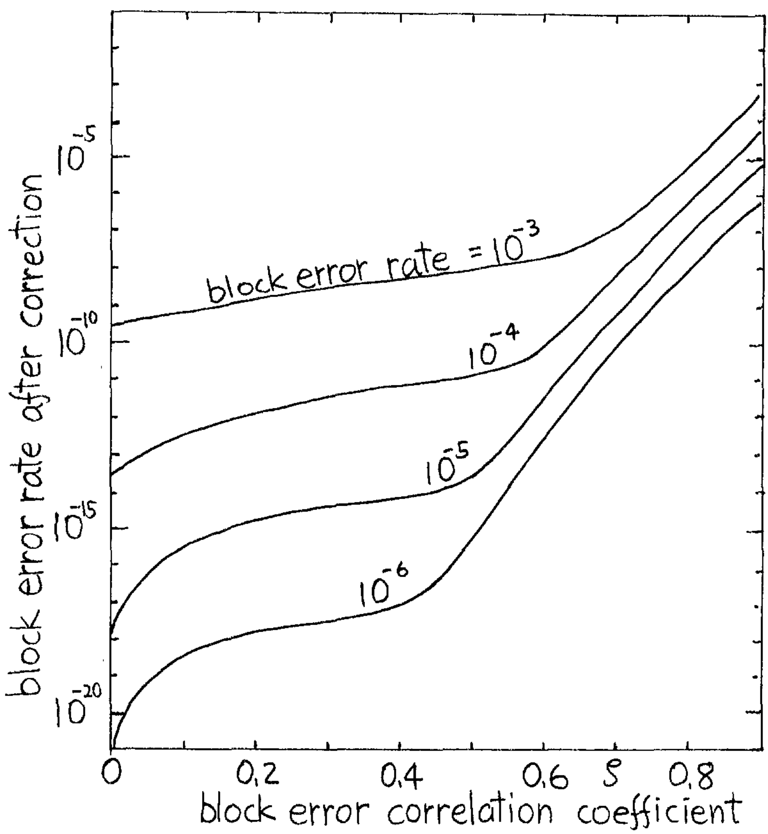


Fig. 10 Evaluation on the Plane of Error Rate & Correlation

REGION (i) Good to Use

REGION (ii) Warning

REGION (iii) Inhibited

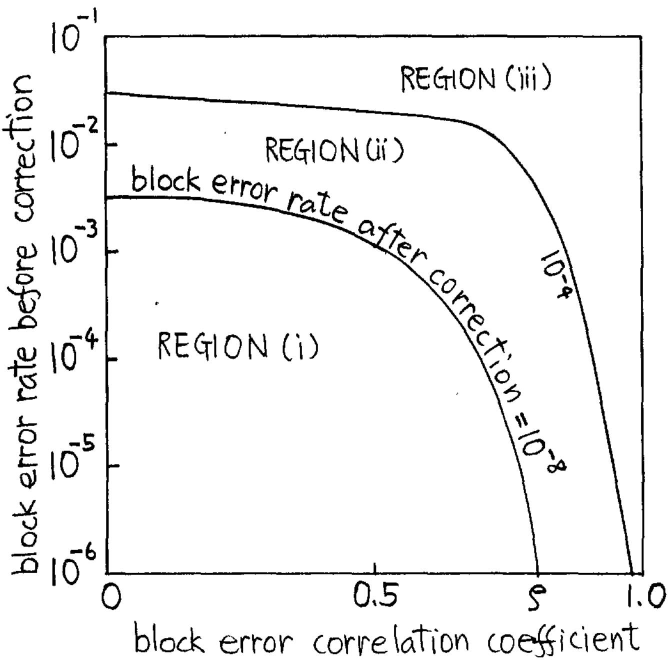
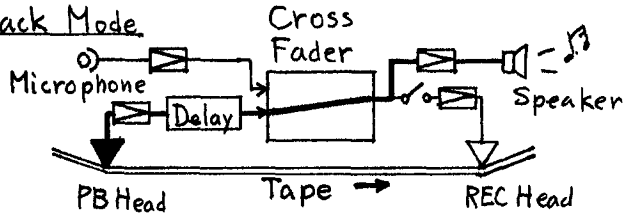
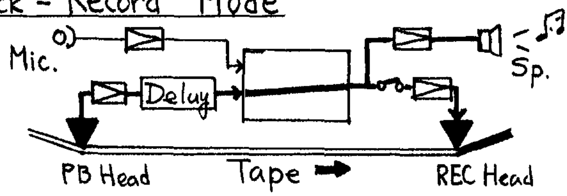


Fig. 11 The Procedure for Punching In/Out

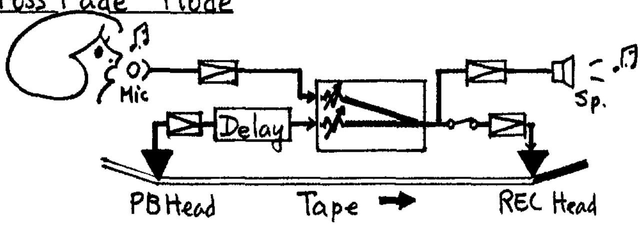
(i) Playback Mode



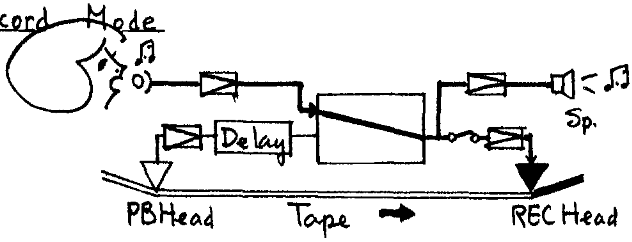
(ii) Playback - Record Mode



(iii) Cross Fade Mode



(iv) Record Mode



(v) Cross Fade Mode

(vi) Playback - Record Mode

(vii) Playback Mode

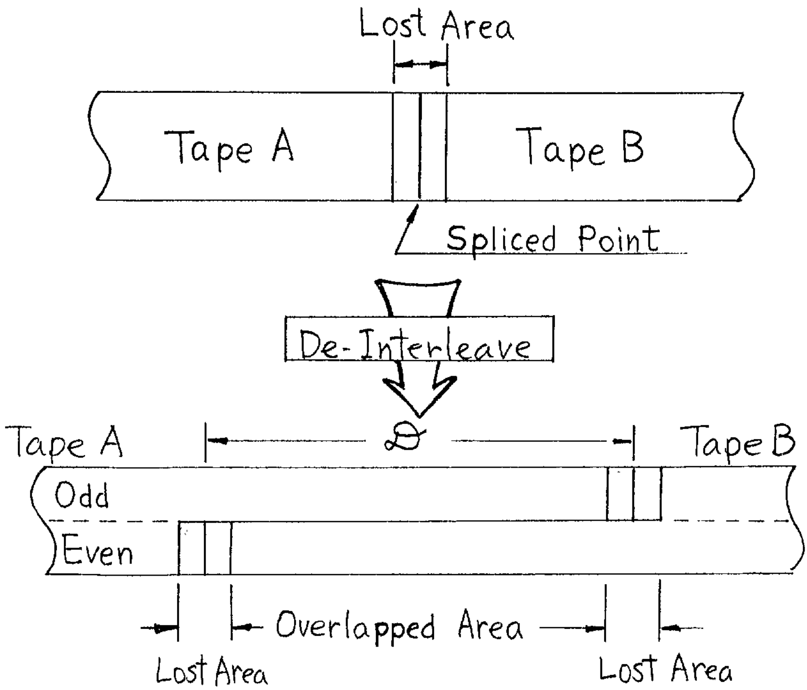


Fig. 12 Tape Splice Editing

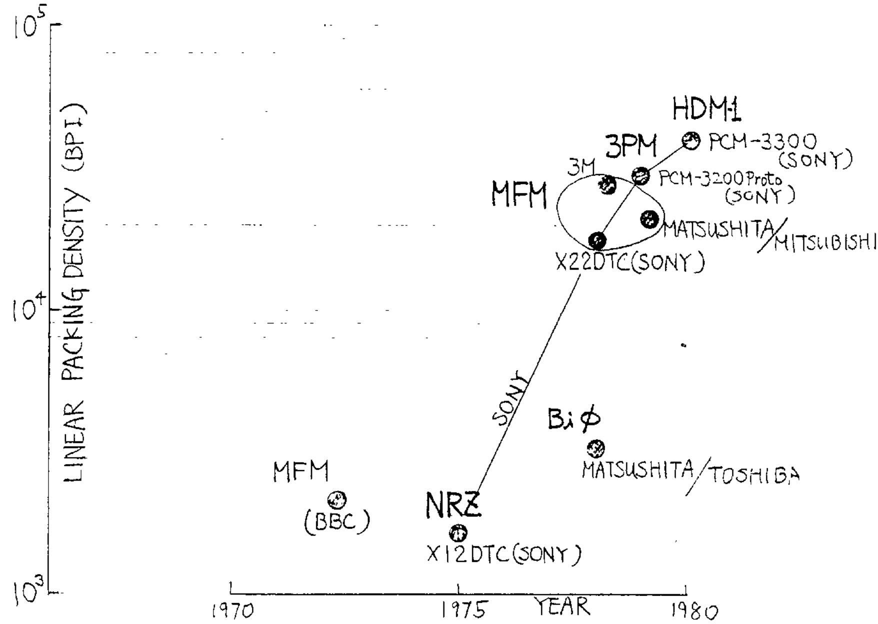


Fig. 13 Packing Density of Stationary Head Digital Audio Recorders