A DESIGN OF PROFESSIONNAL
DIGITAL AUDIO RECORDER

1885 (G-2)

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**Presented at
the 71st Convention
1982 March 2-5
Montreux**

AES

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AN AUDIO ENGINEERING SOCIETY PREPRINT

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ABSTRACT

It is several years since digital audio recorders were introduced in professional field, and another revolution is expected in the technology of multi-channel recorder design.

By virtue of the advance of semi-conductor technology, head technology, error correction technology, and system technology, now it is possible to design a high reliable, smaller size, less hardware, and less tape-consumption digital audio recorders.

Newly developed A/D and D/A converters, single chip and 16-bit accuracy without any trimming, reduced the cost, increased reliability and improved the sound quality of multi-channel recorders greatly.

A newly developed single-crystal ferrite head made it possible to obtain full compatibility of tapes carrying 24 digital channels and 4 auxiliary tracks on half inch tape. It is also expected that 48 channel digital recorder is possible on half inch tape by thin film head technology and tape compatibility will be sufficiently kept. And also the tapes will be compatible between conventional 24 channel machines and the initial 24 channels of coming 48-channel machines.

The advance of error correcting schemes and its signal processing technology improved the machine reliablility in real studio environment.

Various signal processing technologies for punching in/out, splice editing and electronic editing are also discussed.

1. INTRODUCTION

It is observed that the progress of professional digital audio recorders comes to the second step after several years of introduction period. This second step is supported by the following technologies.

(1) Monolithic A/D and D/A converters obtaining 16-bit accuracy without any trimming [1], [2].

(2) Head technology for high density recordings. The linear density improved greatly by applying video head technology, and the track density is going to improve by thin film head technology [3].

(3) Tape transport mechanisms developed especially for digital recordings, which is capable of handling very thin tape and of obtaining good tape compatibility in spite of narrow track width.

(4) Error correction technologies [4], which is one of the most advanced area of technologies since the introduction of digital audio into professional market.
This achievement is effective not only to improve the reliability of the machines in studio environment, but also to expand sophisticated editing capability.
It should be also noted that the size of circuits has remarkably reduced in spite of great improvement in correctability, which made the size of the machines even smaller than analogue ones.

(5) Modulation schemes [5] especially developed for digital audio application. Both packing density and reliability

can be improved by using a modulation scheme the density ratio of which is 1.5.

(6) Editing technologies. In early days editing capability of digital recorders is limited, but the technology of signal processing has improved enough to introduce a wide range of editing from sophisticated electric editing into conventional tape-splice editing.

In this paper, details of these basic technology is described, and an example of a design of a professional recorder is shown, with the possibility of future expansion of the format.

2. BASIC TECHNOLOGIES

2.1 MONOLITHIC A/D, D/A CONVERTER [1], [2]

Designers of digital audio recorders have long been suffered from the price as well as the quality of A/D and D/A converters. It is not a great mistake to say a converter with real 16-bit accuracy has never been supplied. And even if installing acceptable converter in the machine, sometimes it is found the quality deteriorates after a certain period. The most severe point in sound quality is "the monotonicity" of converters, which can easily fail by a very slight shift of resistor values representing MSB, 2SB, or 3SB.

Most of the converters , up to now, are designed to obtain the accuracy by resistor network, which easily drifts by temperature or by shock.

On the other hand, it is well known that integration-type converters will never fail in "monotonicity", and also the accuracy is easier to obtain because it mainly depends on the stability of clock signal. The only reason which prevented the adoption of integration-type converter into digital audio is the speed of solid state devices.

Fig. 1 shows monolithic A/D and D/A converters which is an example of integration-type monolithic converters completed by a high speed I^2L/ECL technology.

Fig. 2 shows the block diagram of A/D converter CX-899. The conversion principle consists of the following four-phase operation, as shown in Fig. 3.

(1) t_1-t_2 : instantaneous sampling

t_1 : the switch associated with the integrator S_1 is turn on, and the analogue input level is sampled and held.

(2) t_2-t_3 : coarse integration

t_2 : both switches associated with the current sources S_2, S_3 are turned on, so that the coarse reference current (I_0+i_0) discharges the sampled charge of the integrator.

(3) t_3-t_4 : fine integration

t_3 : when the output voltage of the integrator crosses the threshold voltage V_T, the comparator 1 operates and turns off S_2, and the fine reference current (i_0) discharges the capacitor.

(4) t_4-t_1 : idling

t_4 : when the output voltage of the integrator crosses the voltage V_0, the clock for counting is inhibited.

In this procedures, the coarse integration corresponds to the values from MSB to 9SB, and the fine one, that from 10SB to LSB. Therefore, the ratio of the coarse and the fine reference current $(I_0+i_0)/i_0$ is selected as 128/1, the accuracy of which is obtained within 1LSB to keep full monotonicity. This was possible by untrimmed poly-Si resistor.

The clock required for integration is approximately 50 MHz, which was enabled by a new I^2L technology with deep UV lithography to realize minimum feature size of 1.5μm X 2.0μm. The structure is shown in Fig. 4.

The same principle of dual slope integration is also adopted to the monolithic D/A converter CX-890, with the same I^2L/ECL process. In this case, coarse and fine integration start simultaneously (t_1 in Fig. 6), and stop individually according to the bit pattern (t_2, t_3). t_4 is the timing for output after glitch.

Table 1 shows basic specifications of both converters.

Table 1 Basic specifications of monolithic converters

1. Name	A/D converter CX-899 [1]	D/A converter CX-890 [2]
2. Quantization	16 bit linear	16 bit linear
3. Sampling Rate	up to 50 KHz	up to 50 KHz
4. Clock Rate	50 MHz	33 MHz
5. Distortion	0.0028%	0.0022%
6. Power Dissipation	450 mW	430 mW
7. Supply Voltage	5V	5V
8. Process	I^2L/ECl	I^2L/ECL
9. Number of Elements	990	1020
10. Chip Size	2.1X 2.3 (mm)	1.9X 2.4 (mm)
11. Number of Pins	28	28

2.2 SINGLE-CRYSTAL FERRITE HEAD

Fig. 7 shows a comparison of structure between conventional head (A) and newly developed single-crystal ferrite head (B).

The structure (A) is made of two parts, front core and back core, and after inserting coils they are put together. Consequently, there are side-gaps between coils and front gaps, which decrease the efficiency and increase cross-talk. In this structure, shielding between cores is essential, thus the structure is complex and cost is high.

The structure (B) stemmed from the technology of video head, but the shape and permeability are optimized for digital audio application. In this structure, there is not any extra gap between front gap and coil, and the efficiency can be higher as well as the cross-talk is low enough even without any shielding. The structure is simple and that makes possible to reduce the total size of head, which is, again, better for the efficiency and the cross-talk.

Fig. 8 shows the cross-talk versus guard band at several different values of core width "t". If "t" can be reduced less than 2 mm, a head with a guard band of 0,1 mm can be used for digital audio.

Fig. 9 shows a completed head for half inch tape. Digital tracks of 26 are realized in line, with track pitch of 0.44 mm.

2.3 THIN FILM HEAD [3]

Fig. 10 shows thin film heads developed for digital audio deck using conventional compact cassette for consumer [3]. The one (above) is multi-turn recording head and the other (below) is MR-type play back head.

Thin film heads in general have the following features.

(1) Cross-talk is extremely low and guard band can be reduced significantly.

(2) The output level of MR-type head is independent from tape speed but responding to flux density. It is expected to obtain higher level than conventional heads at the tape speed lower than 2 m/s.
(3) The cost of head does not depend on the number of tracks, and the process is suitable for mass-production.
(4) The precision of the head is decided by photo-lithography. This high precision combined with low cross-talk enable to realize very narrow track pitch.

This technology can be applied to the design of professional recorders without any doubt, and it is expected that the track density will be doubled compared with the conventional heads with keeping the same tape compatibility and the same ability of tape splice editing.

2.4 TAPE TRANSPORT MECHANISM

Several years ago, it was considered that the tape transport mechanisms for digital audio can be less sophisticated than analogue machines, because time base correction is built in and jitter does not affect to the sound quality. This was proved as wrong after introducing various editing methods in digital recorders. The following are the conditions for tape transport mechanisms for digital audio.

(1) Tape tension should be controlled precisely. Transient tension at any mode should be kept within $\pm 20\%$ of normal tension. This is very important because digital audio recorders are handling very thin tapes the thickness of which is between $25\mu m \sim 35\mu m$.
It is also to be considered that the spliced point of tape is easily stretched if transient tension is not well controlled.
For these reasons, a tape drive mechanism of friction-type (pinch roller less) is recommended.
Fig. 11 shows the tape transport mechanism developed especially for digital audio application.

(2) Track width and pitch are far narrower than analogue machine, and it is recommended to regulate the tape support to one edge, just like video tape recorders. An example of tape-edge regulation method is shown in Fig. 11 (below).

(3) At punching in and out points, smoother editing is possible by adopting cross-fading between old and new data. In order to achieve this, the information played back from a head should be re-recorded by a different head at the same position on the tape. The distance between two heads corresponds to time delay of signal processing.

The accuracy of the position of re-recording is determined by the following points.

(i) Precision of the distance between two heads.

(ii) Jitter.

(iii) Azimuth tolerance of both heads.

In the format called "DASH" [4] (described below), the tolerance of the position is within one block of signal, and all the parameters should be guaranteed in high precision.

Fig. 12 shows fluctuation of re-recorded signal, which is shown to be less than 0.02 ms or 0.08 block length.

2.5 ERROR CORRECTION TECHNOLOGIES

Error correction is one of the most improved areas of technology after the introduction of digital audio.

Initially, correction code was designed by a limited experiments in laboratories, but soon it was found that the condition of code error of open reel recorders in real studio environment is far severer than expected. This is caused by finger prints, dusts, scratches, and smoke of cigarettes.

Fig. 13 shows correctability of various codes against random block error, where one block consists of 288 bits.

At good condition, block error rate of 10^{-5} can be easily obtained and single erasure code is considered as sufficient because un-correctable block will apear at the rate of once in several hours, but most of them can be concealed.

Accordingly, there was a discussion that Cross Interleave Code [6] is too much for digital audio because the probability of un-correctable block is less than once every million years.

After some investigation in studio environment, these arguments are found wrong and an evaluation method is developed for the mixture of random error and various length of burst error. Fig. 14 shows an evaluation of "DASH Format [4]" which will be described in this paper.

A parameter named block error correlation coefficient is introduced and various grade of tapes as well as error correctability are evaluated on the plane of block error rate and said correlation coefficient [4].

Editing capability should also be considered in error correction code. The behavior of the code at spliced point as well as punching in/out point are very important. A code named Cross-Interleave code is adopted [4], and its improved version with superior correctability (Fig. 13) should be given up, by this reason.

It is also noted that burst error correctability is also increased after a lot of experience in studio environment. The final version of the disign is capable of protecting a burst error longer than 80,000 bits.

In spite of these significant achievement, electronic circuitly necessary for signal processing was greatly reduced, and consequently the size of the machines can be even smaller than analogue ones.

2.6 PACKING DENSITY AND MODULATION SCHEMES

A channel code named HDM-1 is developed for digital audio applicaiton [5], which achieved 50% higher packing density than conventional channel code such as MFM with keeping the same wave length to be recorded.

The coding rules are shown in Fig. 15, and the basic parameters are described in Table 2.

Table 2 Basic parameters of HDM-1

Window margin	0.5T
Minimum distance between transition	1.5T
Maximum distance between transition	4.5T
Constraint length	5.5T
Density ratio	1.5T

T = The length of one data bit

2.7 EDITING TECHNOLOGIES

2.7.1 PUNCHING IN/OUT

The following strategies are developed for the execution of punching in/out in digital recorders [4].

(1) Every block begins with a synchronization word which is out of modulation rule and is strong enough to make valid of a new block punched in at the different phase from the original data.

(2) The distance between playback head and synchronized recording head is equivalent as system delay of digital signal processing.

(3) The real punching point is shifted from that of signal punching. Namely, just before the signal punching in point, the synchronized recording head starts to record the signal read from playback head.
Similarly, the real punching out point comes just after the signal punching out.
(4) Cross fading is executed at both punching in and out point between the input signal and played back signal.
(5) In order not to destroy a lot of blocks by punching, mechanical tolerance between two heads, azimuth and jitter are kept in a certain precision.
(6) In order not to destroy multiple blocks by multiple punching, the entry and exit points are set to be a certain interval which corresponds to one of the sequence of Cross Interleave code.
Consequently, the correctability of single erasure is always kept by another sequence of the code after multiple punching.
The time rag caused by this settled entry and exit points is 4.2 ms in worst case at 48 KHz sampling rate, which can be ignored in editing.

2.7.2 TAPE-SPLICE EDITING

The following strategies are developed for tape-splice editing.

(1) In addition to the interleave delay for burst error correction, a large delay is set between even and odd number of data words. And the encoding of Cross Interleave Code is executed separately to even and odd number of words (Fig. 16).
(2) The address of blocks on the tape is recorded on the control track with using 28 bits.
The spliced point can easily be detected by reading this absolute address. And even if it is miss-detected by address, the error of interleave sequence will show the rough area of splice.

(3) At playback stage, above mentioned delay between even and odd words will form a certain period of overlap between the signals on both tapes beyond the spliced point.
This overlapped period is utilized for crossfading and consequently smooth editing can be achieved.
Fig. 17 shows a block diagram of playing back system.

2.7.3 ELECTRONIC EDITING

Electronic editing is now commonly used in professional digital audio field [7]. In the design of multi-channel recorders, several other features can be added, which are described below.

(1) SEQUENTIAL PUNCHING:

Supposing several takes are recorded on different channels, and their best parts are going to be assembled into one channel (Fig. 18), this can be done fully in digital domain without using digital mixing console.

This is named as "sequential punching", and the built-in cross fader can be used at each editing point.

(2) CROSSFADE TIME:

Supposing several channels are edited at a certain point, it is sometimes required that the crossfade time is defferent to each channels. This is easily carried out if cross fader is designed independently to each channels.

(3) EDITING POINT OFFSET:

In the editing of multi channel recorders, it is more flexible that the editing point can be offsetted channel by channel. This can be easily done, by appropriate timing operation to cross fader.

Fig. 19 shows the concept of multi-channel electronic editing, which is impossible in conventional tape-splice editing.

Table 3 The outline of the DASH Format

Version		Fast			Medium			Slow
Tape width (inch)		1/4	1/2	1	1/4	1/2	1	1/4
Digital channels		8	24	48	4	12	24	2
Digital tracks		8	24	48	8	24	48	8
Analogue tracks		2	2	2	2	2	2	2
Time code track		1	1	1	1	1	1	1
Control track		1	1	1	1	1	1	1
Total track		12	28	52	12	28	25	12
Sampling rate fs (KHz)		50.4 , 48.0 , 44.1 , 32.0						
Tape Speed	fs=50.4KHz	76.00			38.00			19.00
	fs=48.0KHz	72.38			36.19			18.10
	fs=44.1KHz	66.50			33.25			16.63
	fs=32.0KHz	48.25			24.13			12.06
Quantization		16 bit linear / channel						
Channel coding		HDM-1						
Error correction		Cross Interleave , CRCC						
Redundancy		33% (Error correction , detection , and synchronization)						
Bit length (μm)		0.6281						
Minimum wave length to be recorded (μm)		1.8844						
Packing density (bpi)		40,426						

3. THE FORMAT "DASH" AND ITS EXPANSION

The format for professional digital audio recorder has already reported [4], but several additional details are described below. The format is named as "DASH" for convenience, which is a shortened form of Digital Audio Stationary Head.

3.1 SAMPLING RATE

A sampling rate of 48 KHz is added to the format, and the related parameters are shown in Table 3. The format itself is sampling rate free, and the tape speed is adjusted so as to obtain the same bit length to be recorded.

3.2 BLOCK FORMAT

The detailed block configuration on the data track is shown below.

Table 4 Block configuration

Item	Bit number	Word number
1. Synchronization	11	1
2. Block Address	2	
3. Overwrite Number	2	
4. Control (Emphasis)	1	
5. Data words	192	12
6. Error Correction	64	4
7. Error Detection	16	1
Total	288	18

Redundancy 33.3%

Efficiency 66.6%

The control bit is set as one bit per each block, but with using block address, four different meanings can be assigned. The control bit at the block address of "00" is assigned as emphasis information (0 = no, 1 = emphasis), while the other three bits are kept open (zeros should be recorded until it is determined).

Thus the emphasis information can be controlled channel by channel, for instance automatic switching is possible at playing back mode.

The even and odd numbers of words are coded separately with using Cross Interleave code as is shown in Fig. 16 and 17, and each delay for interleaving is as follows.

Table 5 Interleave delay

Even-odd delay D =	204 blocks
Cross interleave delay D =	17 blocks
d =	2 blocks

3.3 CONTROL TRACK FORMAT

The block on the control track is called as "SECTOR" in order to discriminate from that on the data track. The length of one sector is equivalent as four blocks on data tracks, and the configuration is shown below.

Table 6 Control Track Configuration

Item	Bit Number
1. Synchronization	4
2. Control	16
(Sampling Rate)*	(4)
(Version)	(3)
(To be defined)	(9)
3. Address	28
4. Error detection	16

* Assign for sampling rates

0000	50.4 KHz
0001	48.0 KHz
0010	44.1 KHz
0011	32.0 KHz

3.4 FORMAT EXPANSION

After the development of thin film head, there arised a possibility of further expansion of the format of higher track density. This is because of the following reasons.

(1) The most critical point in tape interchangability is the margin in over-writing, which is decided by the minimum difference of track width between the recording and the playing back head.
This difference is a summation of the original difference and the tolerance of head dimension, strech of tape by temperature and humidity, the static tolerance and temperature characteristics of tape guiding, and the dynamic movement of running tape.

(2) The guard band of thin film head can be reduced because the cross talk is far better and the space for coil wire is not necessary.

(3) The track width of playing back head can be narrower because the sensitivity of MR-type head is better.

(4) The mechanical tolerance of thin film head is decided by photo-rithography, which is ten times better than conventional heads.

(5) Consequently, even if the track density of thin film head is doubled, it is possible to improve the mimimum difference of track width between recording and playing back head compared with the conventional heads.

In addition, the delicacy of tape splice editing is mostly decided by the width of recording head. Again, thin film head with double track pitch has the same ability as the conventional head in tape-splice editing.

In the future, it might be possible to introduce a digital recorder which carries 48 channels on half inch tape, for instance. And the initial 24 channel can be full compatible with conventional recorders.

4. AN EXAMPLE OF DESIGN, PCM-3324

Fig. 20 shows the completed digital audio recorder PCM-3324, and its dimension. Fig. 21 shows tape support configuration and the arrangement of heads.

Most of electronic circuit board can be accessed from front panel, as is shown in Fig. 22.

The main specifications are shown in Table 7.

This machine is designed to be suitable for the mastering of Compact Disc systems [8], and the sampling rate (44.1 KHz) as well as the time constant of emphasis are selected in line with it. In other words, if music is recorded by this machine, it is not necessary to go into analogue domain, nor digital sampling converters.

The machine also operates at the sampling rate of 48 KHz (switchable) for the future application for the audio of digital video systems.

The synchronization (gen-lock) is possible by three means.

(1) sampling rate

(2) sector rate

(3) composite video signal (option)

The error correction is based on the strong Cross Interleave code which is considered as triple erasure correction for random block error, with additional interleaving for burst error protection of four levels shown in Table 7 (No. 18).

It should be noted that for the perfect correction, a long area of guard space is required, but for the lower level of protection, guard space is far less. This strategy is proved as very effective in real studio environment.

The editing capability is fully prepared from tape-splice editing into various electronic editing as is described in the previous section. A special editor is required for the electronic editing, and the high speed remote interface (SRIF-3) is prepared for the arbitrary control of the machine and for the future expantion of editing strategy.

It should be noted that the editing of multi-channel digital recorders is completely different from that of two channel ones or video recorders. Because, at every editing point, all the information of cross fade time and editing point offset of each channel, the status of the function keys, and some special control like sequential punching are necessary to be transmitted.

If the interface is designed in slow speed, the full features of digital editing is only possible, although in very limited manner, by preparing all the memories at the main frame. In this case, if somebody would like to introduce a new concept of editing, it might be difficult because the main frame has frozen the strategy of editing at initial design.

Today, the electronic editing has just started, and the concept is very flexible with the introduction of digital consoles, therefore the author strongly recommend a high speed interface with the flexibility for the future.

The machine, PCM-3324, is equipped with other two remote interfaces (Table 7). The one (SRIF-1) is an interface with using relay contact which is effective to connect with present automated mixing consoles. The other (SRIF-2) is a pararell format with capability of the control of all the function keys, auto-locator, automatic punching in / out, and the pitch control, and the information for remote level meter is also included.

The size of the machine is even smaller than some of the analogue ones. This was achieved without using any of custom made LSI's nor master slices, which means that a further reduction in size is expected with the aid of the introduction of these semiconductor technologies.

This shows a hope for a machine with larger number of channels if it is required from the industry.

The tape consumption is also better than analogue machines. The half inch tape will help the handling as well as the archives in studios.

5. CONCLUSION

Basic technologies for the design of professional recorders, the format, and an example of the design are described in this paper. It is noted that some of these technologies are developed for consumer applications.

In the past, the most modern technologies in recorders were developed for professional applicaitons and afterward they flew down to the consumer field. But in digital, this is somewhat reversed because of the investment required for the development of LSI's, A/D and D/A, thin film heads and so on, which are remarkably large compared with the size of the market of professional audio.

It is, therefore, adviced to settle the professional standard as close as possible to that of consumer, otherwise the industry can not enjoy the state of arts and is left behind in expensive and inconvenient tools.

Fortunately enough, a format "DASH" and a recorder "PCM-3324" have been developed with the full support from technologies developed for consumer applications.

It is also lucky that many people related in digital audio agreed with the dual sampling rates of 44.1 KHz and 48 KHz. The former is suitable for all the mastering works for Compact Disk systems, and the latter is reserved for the future application with digital video systems. If all the professional recorders are designed to be switchable between these frequencies, people need not to worry about sampling rates.

The format "DASH" is mutually supported by Sony, Willi Studer and MCI, and therfore the tapes are compatible among numerous machines.

ACKNOWLEDGEMENT

The author would like to thank to the following personnel of Sony Corporation; Messieurs M. Morizono, H. Nakajima, T. Miamoto and K. Suetsugu for their support to the project, Messieurs M. Takeda, A. Iga and K. Asano for the development of A/D and D/A, Messieurs N. Okamoto and T. Watanabe for the development of the single crystal ferrite head, Messieurs T. Sekiya and H. Miyairi for the development of thin film head technology, Messieurs M. Tanaka, and N. Watanabe for the development of the format "DASH" and Messieurs Y. Tsuchiya, Y. Kishi, S. Hoshimi and other members of the team for the development of PCM-3324.

The author also thanks to Dr. R. Lagadec of Willi Studer for his useful advices.

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Table 7 PCM-3324 Specifications

1. Number of tracks :

Digital channel	24
(Digital track	24)
Analogue track	2
Time code track	1
Control track	1
Total track	28

2. Tape speed : 66.50 cm/s (at f_s = 44.1 KHz)
72.38 cm/s (at f_s = 48.0 KHz)
with $\pm$12.5% vernier

3. Tape = digital audio master tape
$D\text{-}1/_2$ -2920 (65 minutes at fs = 44.1 KHz, 14 inch reel)
$D\text{-}1/_2$ -1460 (33 minutes at fs = 44.1 KHz, 10,5 inch reel)

4. Quantization : 16 bit linear / channel

5. Dynamic range : more than 90 dB

6. Frequency responce : 20 Hz to 20 KHz $\left\{\begin{matrix}+0.5\\-1.0\end{matrix}\right\}$ dB

7. Total harmonic distortion : less than 0.05%

8. Wow and flutter : undetectable

9. Emphasis : ON/OFF switchable independently to each channel at recording, and automatic switching at playing back.

Time cpnstants 50μs / 15μs (compatible with Compact Disc Digital Audio, EIAJ format of rotary head recorders (STC-007, 008), and PCM-1610)

10. Print through : Not measurable

11. Erasure : No measurable residual signal

12. Play time :

65 minutes (44.1 KHz)	60 minutes (48 KHz)	14 inch reel
33 minutes (44.1 KHz)	30 minutes (48 KHz)	10.5 inch reel

13. Fastforward, rewind time : 270 second (14 inch reel)
    150 second (10.5 inch reel)

14. Format : DASH-F [4]
    (Fast version of DASH format)

15. Block rate : Signal track block rate
    3,675 block/s (44.1 KHz)
    4,000 block/s (48.0 KHz)
    Control track sector rate
    918.75 sector/s (44.1 KHz)
    1,000.00 sector/s (48.0 KHz)

16. Channel Coding : HDM-1 [5]

17. Error Control : Cross Interleave Code [6]

18. Burst error protection :

	Length on the tape (mm)	Time (ms)		Bits
		44.1 KHz	48 KHz	
Correction	3.78	5.71	5.25	6,048
Guard space	17.91	26.94	24.75	28,512
Good concealment	22.25	33.47	30.75	35,424
Guard space	36.36	54.69	50.25	57,888
Normal concealment	36.90	55.49	50.98	58,752
Guard space	21.71	32.65	30.00	34,560
Marginal concealment	52.28	78.64	72.25	83,232
Guard space	6.33	9.52	8.75	10,080

19. Heads :

Digital recording (advance)	1
Digital recording (shynchronization)	1
Digital playing back	1
Analogue recording / playing back	1
Analogue erase	1

20. Tape transport : Friction drive mechanism especially developed for digital audio.

21. Digital audio I/O level :

maximum	24 dBM	(600Ω load)
reference	4 dBM	
adjustable range	±10 dB	

22. I/O terminals :

1	Digital input	x24	
2	Digital output	x24	
3	Analogue input	x 2	
4	Analogue output	x 2	
5	Digital interface input	x24	
6	Digital interface output	x24	
7	Time code input	x 1	
8	Time code output	x 1	
9	Word synchronization input	x 2	
10	Word synchronization output	x 2	
11	Sector synchronization input	x 2	
12	Sector synchronization output	x 2	
13	Sector address input	x 1	
14	Sector address output	x 1	
15	Composite synchronization input	x 1	(option)
16	Remote 1 input / output	(format SRIF-1)	
17	Remote 2 input / output	(format SRIF-2)	
18	Remote 3 input / output	(format SRIF-3)	
19	Speed control input	x 1	
20	Phase control input	x 1	
21	Tach pulse output	x 2	(ϕ1 and ϕ2)
22	Ground terminal	x 1	
23	AC input	x 1	

23. Special control functions :

1. Group keys x4
2. Set up memories x4
   (backed-up by battery)
3. Edit stop
4. Automatic punching in / out
5. Rehearse

24. Editing functions :

① Punching in / out

* Accuracy within 1 ms
  (by interval absolute address
  = sector address)
* Crossfade time
  variable between
  1.45 ms -- 372 ms (44.1 KHz)
  1.33 ms -- 341 ms (48 KHz)

② Electronic editing

* Editing point can be settled independently in each channels.
* Crossfade time can be settled independently in each channels.
* Sequential punching is possible on digital domain.

③ Tape splice editing

* Crossfade time (not variable)
  5.66 ms (44.1 KHz)
  5.20 ms (48 KHz)
* Tape cut at any point.
* Two spliced points should be apart at least 21mm.

25. Dimensions : 830 (W) x 1003 (H) x 740 (D) mm

26. Weight : 220 kg

27. Power requirement : AC 100V / 120V / 220V / 240V ±10%
50 Hz or 60 Hz

28. Power consumption :
approximately 3 KVA, 2 KW (peak)
2 KVA, 1.5 KW (recording)

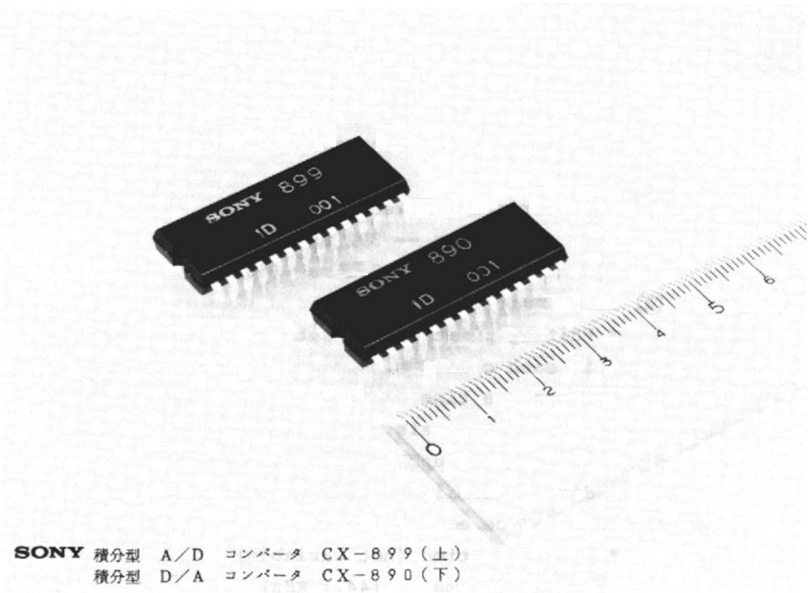


Fig. 1 A/D converter CX-899
and
D/A converter CX-890

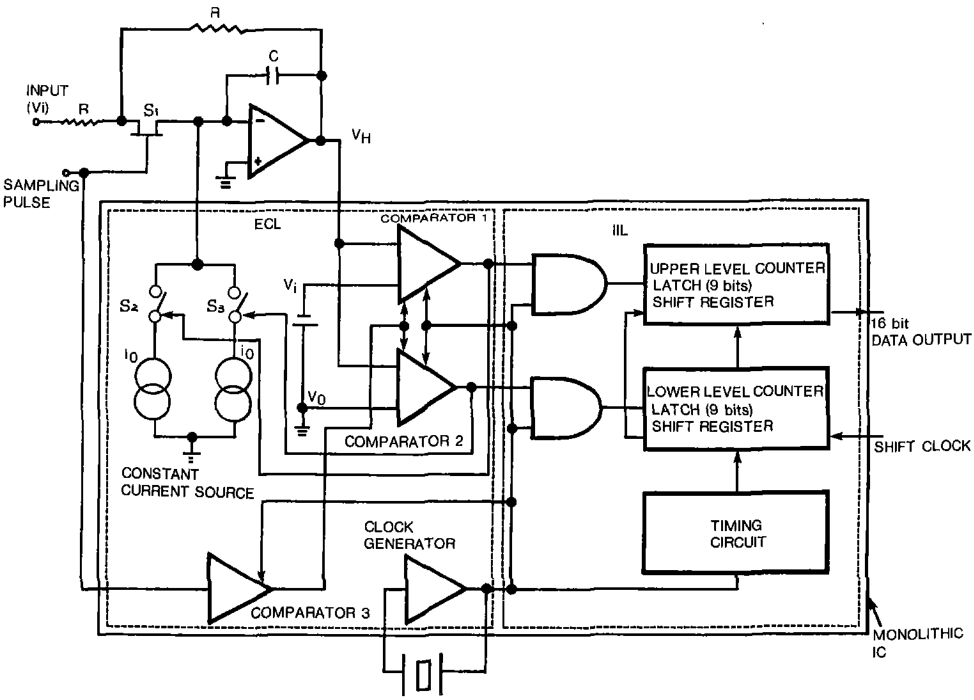


Fig. 2 Block diagram of A/D converter CX-899

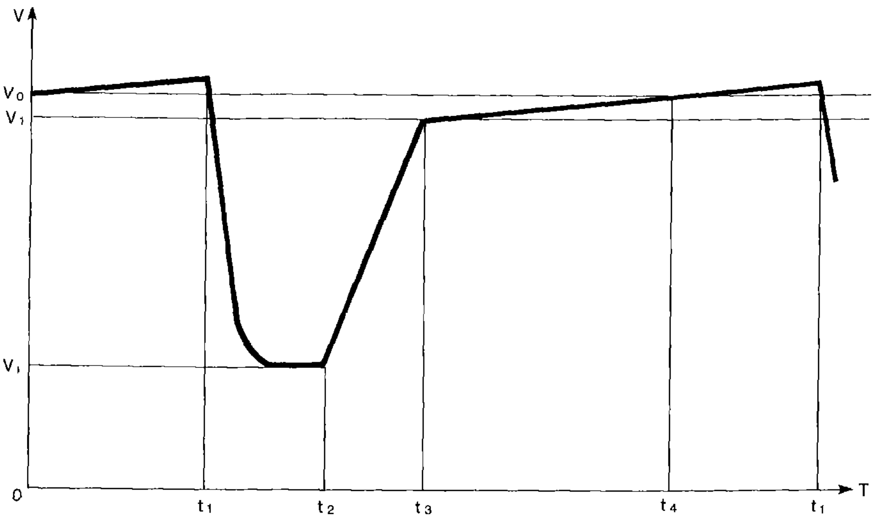


Fig. 3 Basic principle of double integration-type A/D converter (CX-899)

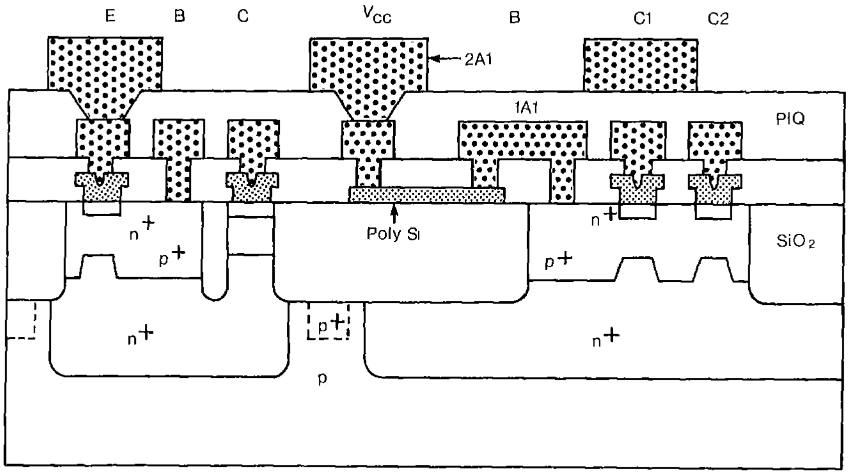


Fig. 4 The structure of high speed I^2L

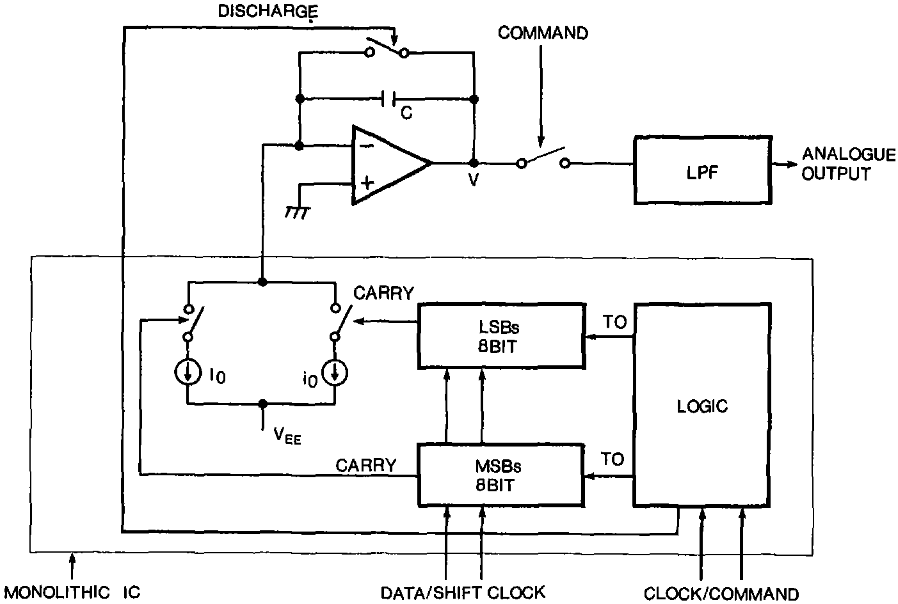


Fig. 5 Block diagram of D/A converter CX-890

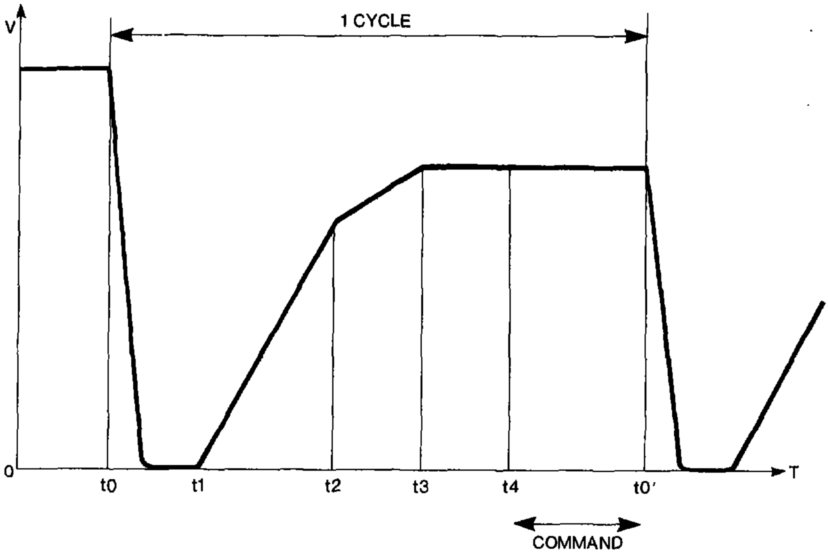
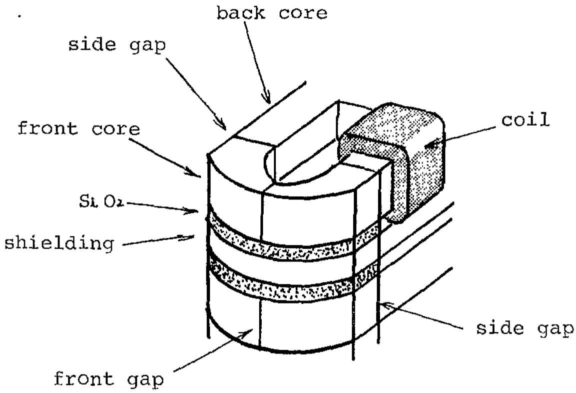
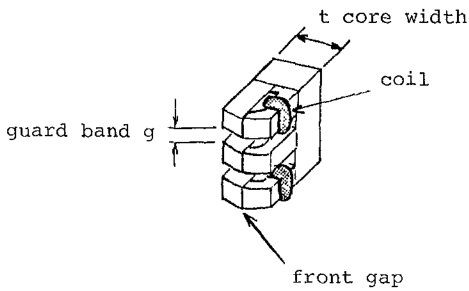


Fig. 6 Basic principle of D/A converter CX-890



(A) conventional head structure



(B) newly developed single crystal ferrite head

Fig. 7 Head structure

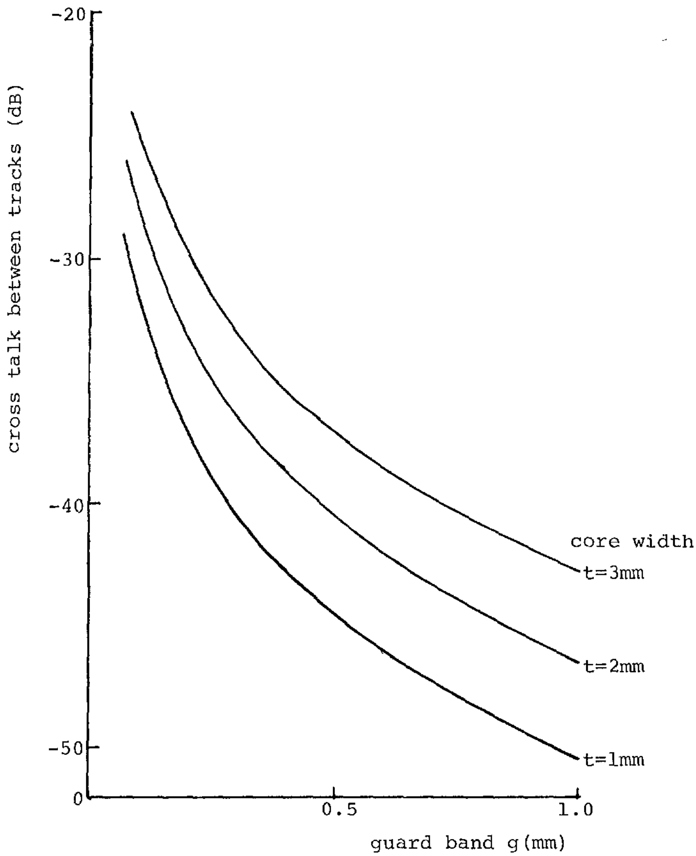


Fig. 8 Cross talk

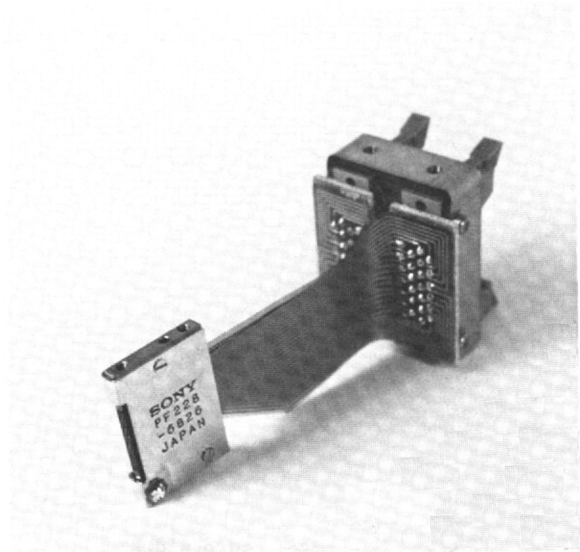


Fig. 9   Single crystal ferrite head for
          half inch tape (26 tracks)

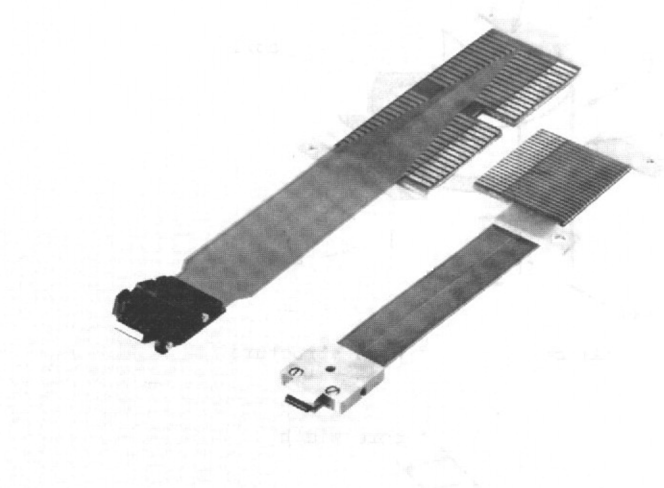


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Fig. 10 Thin film heads developed for compact cassette digital audio deck

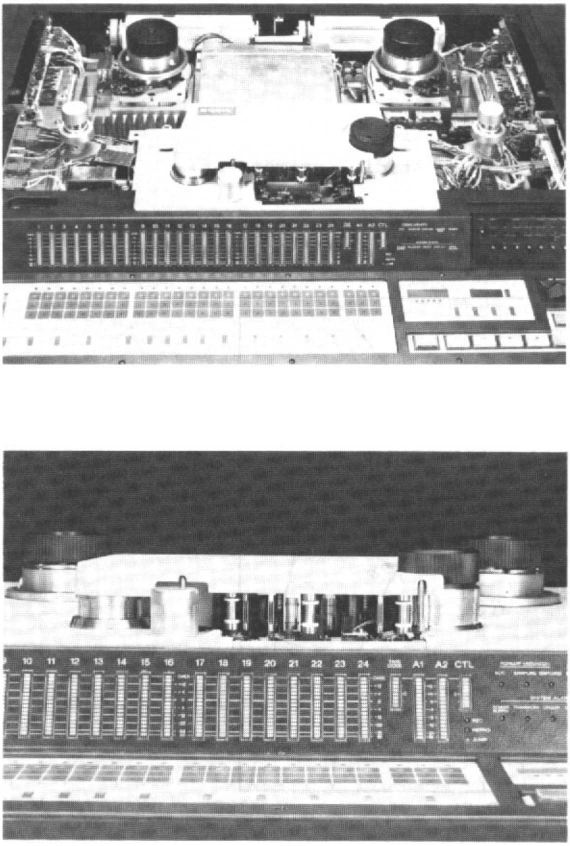


Fig. 11 Tape transport mechanism

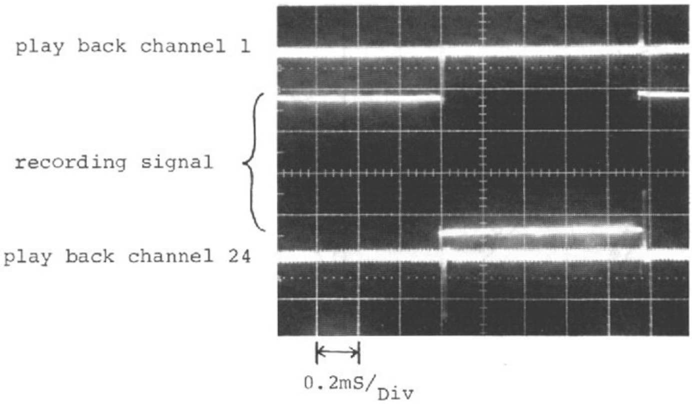


Fig. 12 Fluctuation of re-recorded position

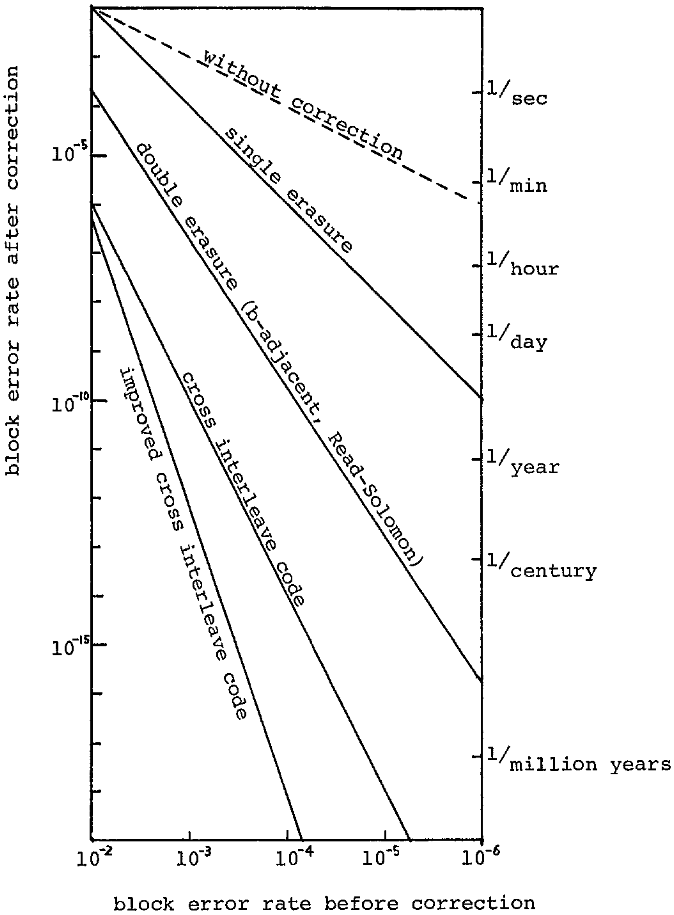


Fig. 13 Random error correctability of various codes

region (i)	: good to use, block error rate after correction $< 10^{-8}$	
region (ii)	: warning , $10^{-8} <$ 〃	$< 10^{-4}$
region (iii)	: inhibited to use, $10^{-4} <$ 〃	

area A : best tuned condition

area B : deteriorated in studio environment

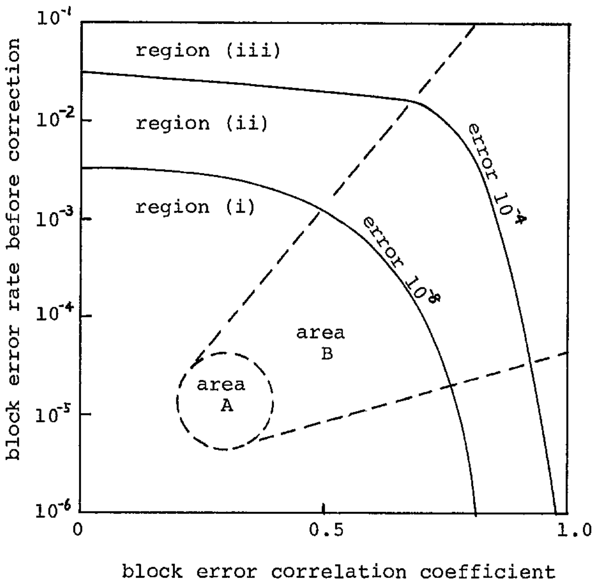
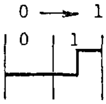


Fig. 14 Evaluation of "DASH Format"

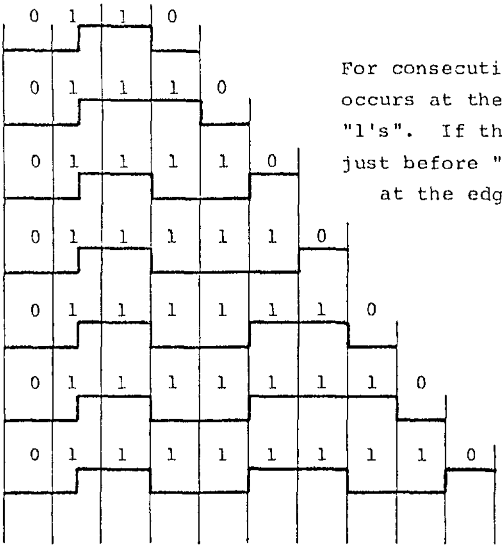
Fig 15 Coding Rules of HDM-1

(1)



Transition at the center of "1" bit cell.

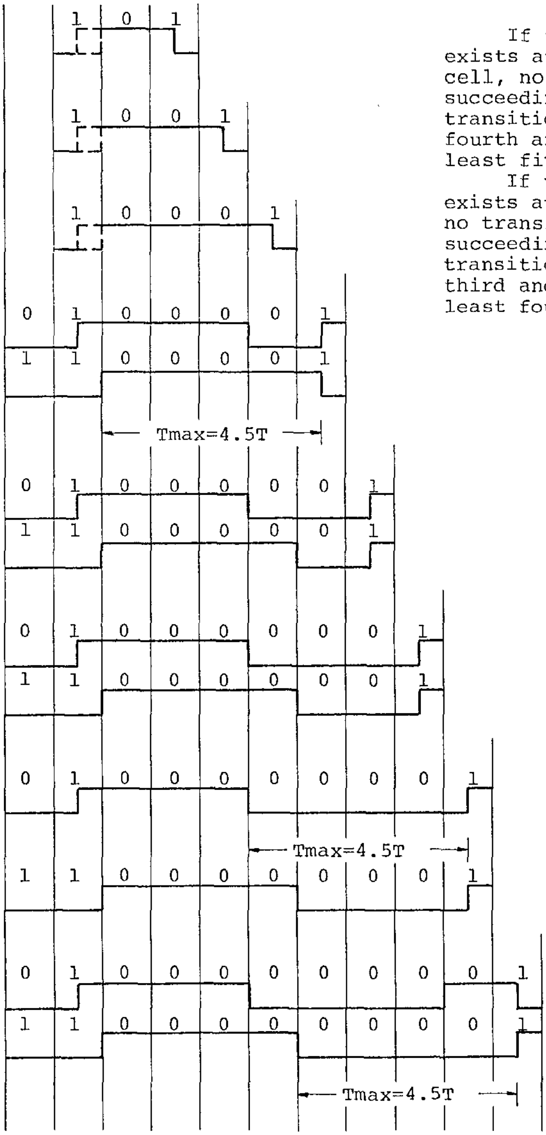
(2) Consecutive "1"



For consecutive "1", transition occurs at the edge of every two "1's". If three "1's" are left just before "0", transition occurs at the edge between "1" and "0".

Fig 15 (Continued)

(3) Consecutive "0s"



If the preceeding transition exists at the boundary of a bit cell, no transition occurs for the succeeding zeros unless the transition at the boundary of the fourth and the fifth zero for at least five consecutive zeros.

If the preceeding transition exists at the center of a bit cell, no transition occurs for the succeeding zeros unless the transition at the boundary of the third and the fourth zero for at least four consecutive zeros.

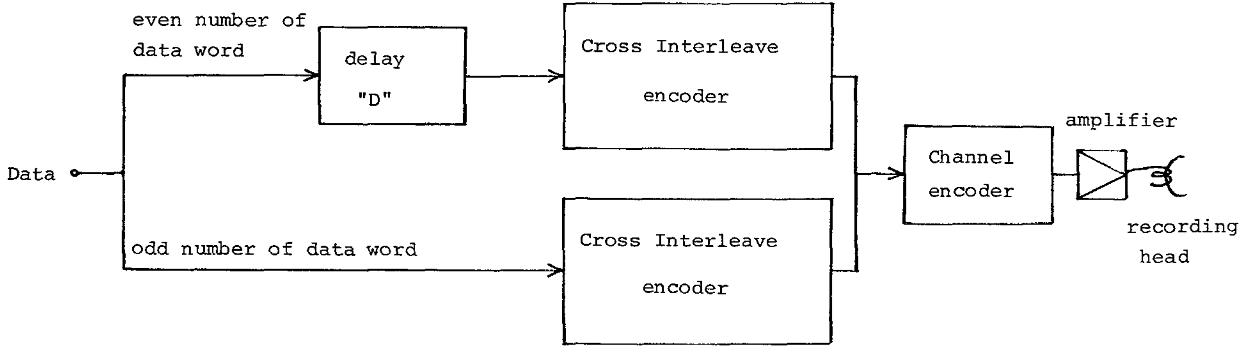


Fig 16   Recording system

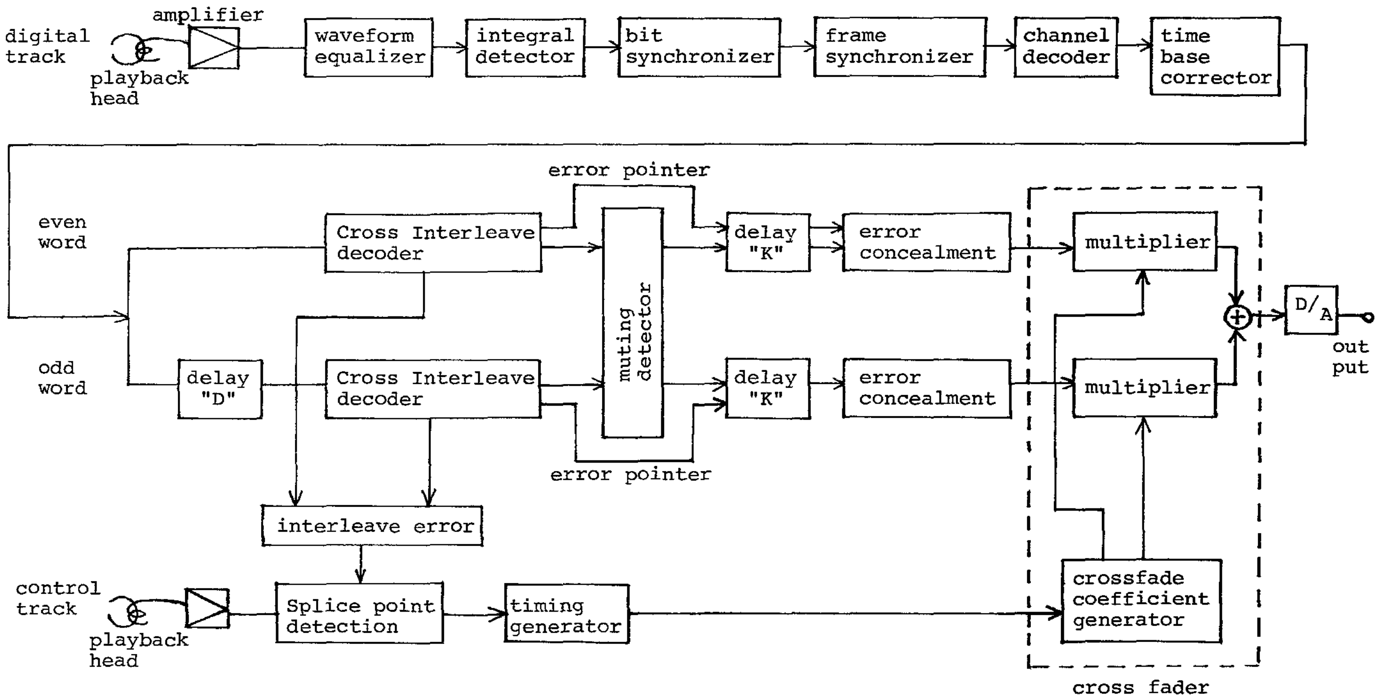


Fig. 17 Playing back system

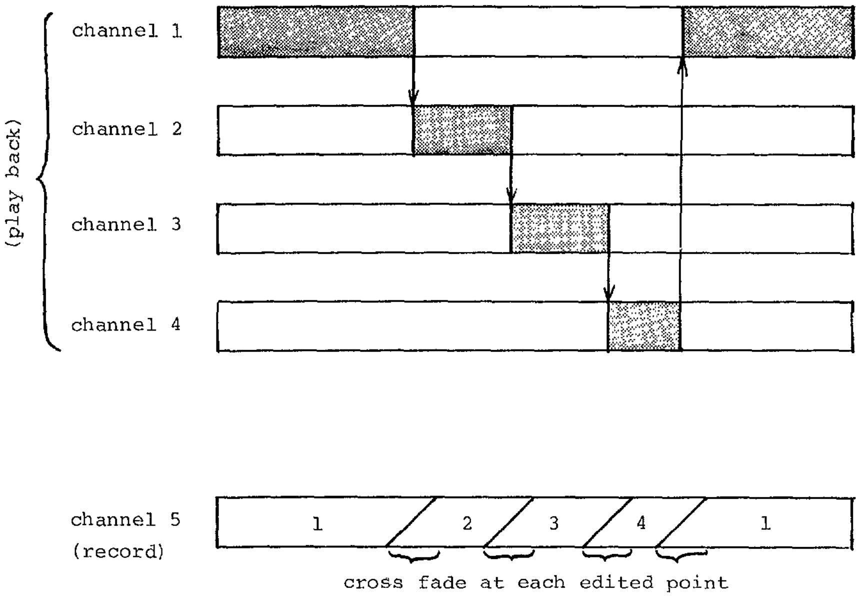


Fig 18   Sequencial punching

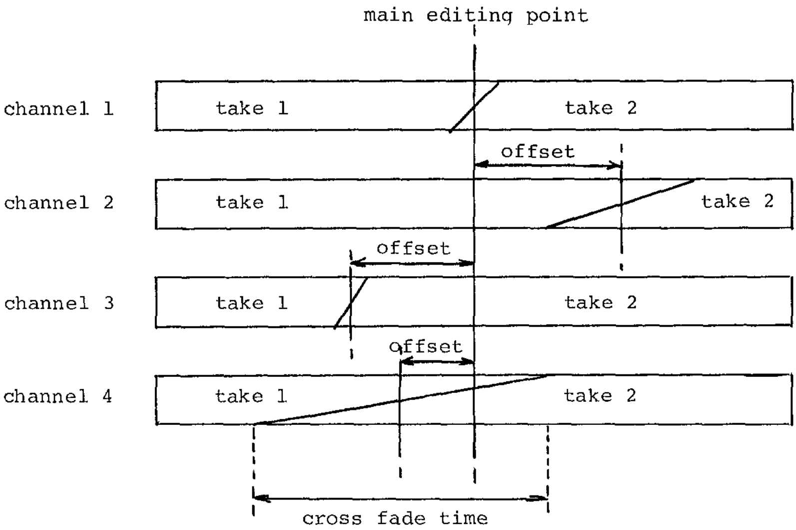


Fig 19 Electronic editing with different corss fade time and offset in each channels

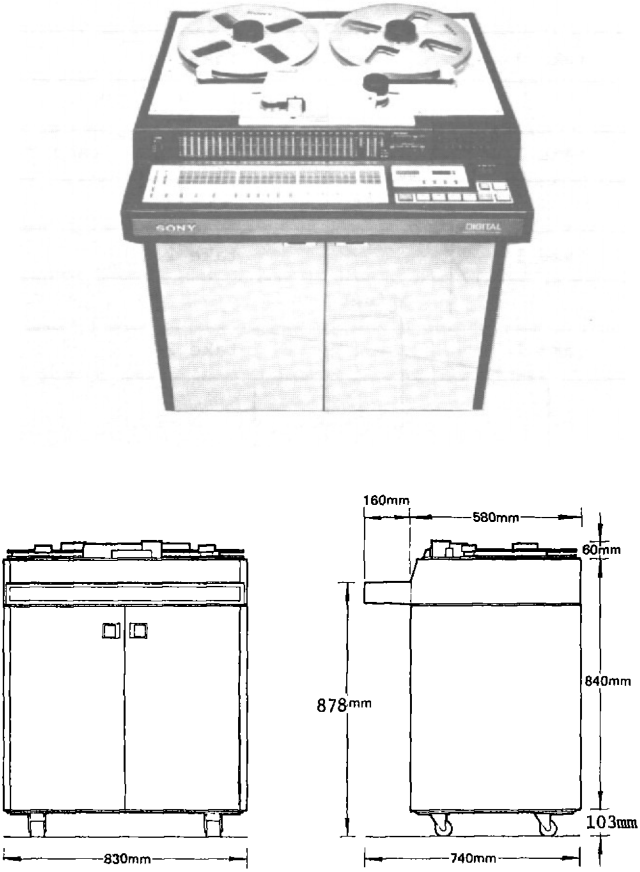
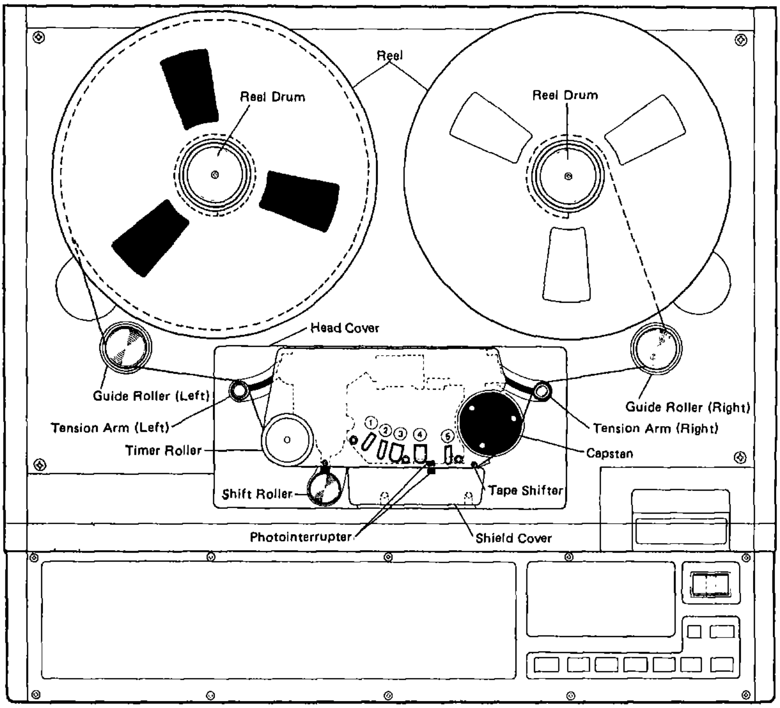


Fig. 20   PCM-3324 Outside view



1. Digital REC Head (advance REC head)
2. Digital P·B Head
3. Analog Erase Head
4. Analog REC/P·B Head
5. Digital REC Head (sync REC head)

Fig. 21 PCM-3324 Tape support configuation

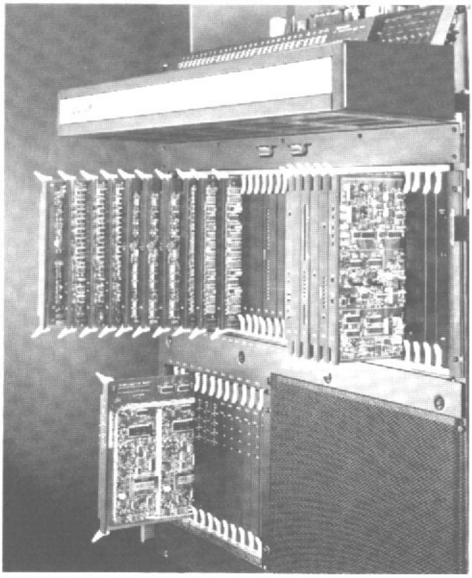


Fig. 22 PCM-3324 Electronics configuration