A DIGITAL AUDIO SYSTEM BASED ON A PCM STANDARD FORMAT

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Abstract

By use of Pulse Code Modulation (PCM) technique, we can successfully overcome the limitations of conventional analog audio systems.

Especially PCM recording will be the most useful field in the digital audio systems.

Fortunately, a Standard Format for consumer use PCM recording adopted for the standard TV signal format has been established recently. Consequently, we are now able to design PCM tape recorder having full compatibility with the PCM processor and magnetic tape as well. We developed a new error correcting code suitable for rotary head PCM recording, "interleaved Matrix Code" which has an extremely strong error correcting ability and which can be designed at a comparatively low hardware cost.

1. Specifications for digital audio systems

Table 1-1 shows the basic specifications required for developing consumer use digital audio systems which should be realized in a good balance between the quality of reproduced audio signals and its hardware complexity. These specifications are based not only on our requirements but also on PCM Standard Format. Fig. 1-1 shows the frequency relationship between the sampling frequency f_s and NTSC TV signals. The sampling frequency based on NTSC TV format has been already discussed precisely, and determined to be f_s = 44.056 KHz.[1)]

In the case of PAL/SECAM TV format, we calculated the frequency relationship according to various combinations of D and B numbers as shown in Fig. 1-4. Here, D is the number of unused H line periods for data storage in one frame. B is the number of unused bits for data storage in 1H. And we found the combination shown in Fig. 1-5 will be the best choise for the PAL/SECAM TV Format. As is clear from Fig. 1-1 and Fig. 1-5, the PCM frequency relationship between NTSC and PAL/SECAM TV format is very similar to each other. So, in this paper, we'd like to discuss mainly the PCM signal format based on the NTSC TV format. Signal allocation in 1H and 1V are shown in Fig. 1-2 and Fig. 1-3.

2. Error correction scheme

2-1. Data alignment during 1H and interleaved recording

Dropout characteristics of VTR cassette tapes are already discussed precisely at the 61st AES Convention.[2] According to its results, the basic characteristics of dropouts are as follows;

a. Once dropout occurs, it continues to certain length. (burst errors)
   ---- interleaved arrangement of data recording is required.

b. The length of burst error may become more than 10 horizontal line periods
   ---- an interleave distance of more than 10H line periods is required, and 16H line periods is selected for interleave distance in our PCM Standard Format.

c. By converting the PCM data into blocks, the burst error probability becomes the simple block error rate.
   ---- we can easily estimate the error correcting ability of any error correcting code.

Based on the observed above dropout characteristics, the PCM data alignment in 1 H period and its interleaved data recording on magnetic tape is carried out as shown in Fig. 2-1 and Fig. 2-2, respectively. In Fig. 2-1 and Fig. 2-2, D is the interleave distance, and is selected as 16 H line periods.

2-2. "interleaved Matrix Code"

"interleaved Matrix Code" is an error correcting code which can correct 2 error words in 6 words. It has 2 redundant words P.Q. and is recorded on a magnetic tape as shown in Fig. 2-1 and Fig. 2-2. Several characteristics and advantages of the "interleaved Matrix Code" over conventional error correcting code, are as follows.

a. Strong error correcting ability, and consequently very effective to the rotary head PCM recording.

b. Every data is treated word by word, and is also very effective to the rotary head PCM recording.

c. This code is effective only in combination with the interleaved data recording.

d. The error correcting ability can be varied widely by its hardware construction while maintaining compatibility.

e. The "interleaved Matrix Code" can be realized by comparatively simple, which is suitable to consumer use, hardware.

2-3 The processing of "interleaved Matrix Code"

The relationship between 6 data words and 2 redundant words P.Q. is shown in Fig. 2-3, and its interleavedarrangement is shown in Fig. 2-4. In these figures, the data words W1, W2, ---, W6 correspond to the data words Ln, Rn, ----- Rn + 2, each respectively as shown in Fig. 2-1 and Fig. 2-2.

The redundant words P.Q are generated according to the equations (2-1) and (2-2) as follows.

$$P = W_1 \oplus W_2 \oplus W_3 \oplus W_4 \oplus W_5 \oplus W_6 = \sum_{i=1}^{6}\!{}^{\circ}\; W_i \quad ----- \qquad (2\text{-}1)$$

$$Q = T^6 W_1 \oplus T^5 W_2 \oplus T^4 W_3 \oplus T^3 W_4 \oplus T^2 W_5 \oplus T W_6 = \sum_{i=1}^{6}\!{}^{\circ}\; T^{7-i} \cdot W_i \quad -- \qquad (2\text{-}2)$$

In these equations, the redundant word "P" is the result of exclusive sum of 6 data words, namely the result of Mod. 2 operation on these 6 words. And the redundant word "Q" is the result of exclusive sum of $T^{7-i} \cdot W_i$, where i varies from 1 to 6.

Further "T" is the Q generating matrix as shown in Fig.2-5. Data stream incoming from VTR is continuously checked by CRCC to detect whether there exists any error or not. Suppose one data word, for example "Ln" in Fig.2-2, among 6 data words is false, and the other interleaved data words in the another blocks are correct. The incorrect data "Ln" can be easily recovered by calculation of equation (2-1) or (2-2). Suppose now there exists 2 error words in the same group which consistsof 6 data words and 2 redundant words. Since there are 2 equations for 2 unknown words (namely 2 error words), we can reconstruct unknown words by using these two equations (2-1) and (2-2).

By writing received data as W_i, W_j and error values as e_i, e_j, we can get following equations in the case of double errors.

$$\hat{W}_i = W_i \oplus e_i \qquad \text{----- (2-3)}$$

$$\hat{W}_j = W_j \oplus e_j \qquad \text{----- (2-4)}$$

$$P \oplus \sum_{k=1}^{6} \hat{W}_k = P \oplus \sum_{k=1}^{6} W_k \oplus e_i \oplus e_j = e_i \oplus e_j = S_1 \qquad \text{----- (2-5)}$$

$$Q \oplus \sum_{k=1}^{6} T^{7-k} \cdot \hat{W}_k = Q \oplus \sum_{k=1}^{6} T^{7-k} \cdot W_k \oplus T^{7-i} \cdot e_i \oplus T^{7-j} \cdot e_j$$

$$= T^{7-i} \cdot e_i \oplus T^{7-j} \cdot e_j = S_2 \qquad \text{----- (2-6)}$$

The Syndromes S_1, S_2 are calculated from the received words by equations (2-5), (2-6) respectively. From equations (2-5), and (2-6), we can calculate the error value e_i and e_j as follows.

$$e_j = (I \oplus T^{i-j})^{-1} \cdot (S_1 \oplus T^{i-7} \cdot S_2) \qquad \text{----- (2-7)}$$

$$e_i = S_1 \oplus e_j \qquad \text{----- (2-8)}$$

In equations (2-7) and (2-8), I is the unit matrix, i, j is the location of error words detected by CRCC. The correct data words W_i, W_j are calculated as follows.

$$W_j = \hat{W}_j \oplus e_j \qquad \text{----- (2-9)}$$

$$W_i = \hat{W}_i \oplus e_i \qquad \text{----- (2-10)}$$

Q generating matrix "T" having 14 x 14 construction is shown in Fig. 2-5. This matrix "T" is determined based on the following two main reasons.

a. Under various combination of i, j ($1 \leqq i < j \leqq 6$), equation (2-5) and (2-6) must be linearly independent to each other.

b. A hardware to calculate above equations must be as simple as possible.

There exist many other matrices "T" which satisfy the above condition (a), but judging from the condition (b), the matrix "T" shown in Fig. 2-5 has been considered to be the best choise. A kind of operation of matrix "T" has been reported (3). In order to generate redundant words P, Q and achieve interleaved recording on magnetic tapes, we developed an "interleaved Matrix Code" generator shown in Fig. 2-6.

The "interleaved Matrix Code" can correct two errors simultaneously occurring as mentioned before. Therefore the probability of no correction is of the order of $P_H{}^3$ and is written as follows.

$$\text{Puncorrectable} = P_H \cdot ({}_7C_2 \cdot P_H{}^2) = 21 \cdot P_H{}^3 \qquad \text{-----} \quad (2\text{-}11)$$

Where P_H is the error rate observed in H units.

Table 2-1 shows the main characteristics of "interleaved Matrix Code" and indicates that the number of uncorrected error is less than one per hour even if we use a fairly bad VTR tape. We examined the error correcting ability of this code using standard home use VTR and a hardware simulator. The construction and operation of our hardware simulator and its results have been already reported at the 61st AES convention.[2)]

Fig. 2-7 (a) and (b) show the results obtained from the hardware simulator. As a whole, there were only slight difference between the calculated values shown in Table 2-1 and the simulated values shown in Fig. 2-7. As shown in Fig. 2-7 (a), there exists no uncorrectable error in more than one hour.

2-4. Encoding of "interleaved Matrix Code"

Encoding of "interleaved Matrix Code" is basically executed by performing exclusive sum, multiplication of $T \cdot W_i$ and delay as shown in Fig. 2-6.

The matrix "T" is the associated matrix of polynomial $x^{14} + x^8 + 1$. And the multiplication of matrix "T" is executed by the linear shift register shown in Fig. 2-8. This multiplication of matrix "T" with word "W_i" will be clear from equation (2-12).

$$[T] \cdot \underset{(W_i)}{\begin{pmatrix} x_1 \\ x_2 \\ x_3 \\ \\ \\ x_{14} \end{pmatrix}} = \underset{(T \cdot W_i)}{\begin{pmatrix} x_{14} \\ x_1 \\ x_2 \\ \\ x_8 \oplus x_{14} \\ \\ x_{13} \end{pmatrix}} \qquad \text{-----} \quad (2\text{-}12)$$

Fig. 2-9 shows the shift signal and gate control signal "G" for the multiplication circuit shown in Fig. 2-8. The operation of this circuit is as follows. First of all one word W_1 consisting of 14bits is shifted into 14bit shift register while AND gate is opened. Then the gate is closed, and shift register shifts by one bit, now the result of $T \cdot W_1$ is stored in the shift register.

As a next step, we have a series of same operation with a data word W_2, and we get a result as follows.

1st operation: $\quad W_2 \oplus T \cdot W_1$

2nd operation: $\quad T \cdot (W_2 \oplus T \cdot W_1) = T \cdot W_2 \oplus T^2 \cdot W_1$

By continueing 6 times of this operation, we can get the redundant word Q.

$$T^6 \cdot W_1 \oplus T^5 \cdot W_2 \oplus T^4 \cdot W_3 \oplus T^3 \cdot W_4 \oplus T^2 \cdot W_5 \oplus T \cdot W_6 = Q$$

The delay means for obtaining interleaved data words is the most essential part of "interleaved Matrix Code" generator. As shown in Fig. 2-6, comparatively large memories are required for constructing delay means. One of the most effective way of realizing above delay is to use the random access memories (RAM). Fig. 2-10 is an example of this intelligent memory structure with using most popular 4K bit RAM. The memory structure is composed of 4-parallel and 2-serial construction. 14 bits of one word is achieved by parallel combination of 4-RAMs. Addressing of 128 x 8 (= 1024 words) is done as shown in the figure. When recording PCM signal on magnetic tape, data words and redundant words are written into RAMs with a fixed interleave distance D. If the first data word "Ln" is addressed (x_1, y_0), the next data word "R_n" is addressed (x_2, y_{16}), the third data word "L_{n+1}" is addressed (x_3, y_{32}) and so on. The redundant word "Qn" is addressed (x_8, y_{112}). Here 16, 32, 112 are the delayed positions expressed in the unit of H line period. In this memory construction, there remains 144H space for storing data. This space is purposely reserved for absorbing the jitter of VTR output. Usually we don't need such a large jitter margin. But accidentally some VTRs show very large jitters, therefore it is recommendable to design a memory structure of having large jitter margin.

The second step of recording PCM signal on magnetic tape is to read out data words and redundant words from RAMs and to write into magnetic tape. In this step, RAM is addressed in order of (x_1, y_0), (x_2, y_0), (x_3, y_0) ---- ----- (x_8, y_0), (x_1, y_1), (x_2, y_1) ----- . Therefore data words and redundant words are recorded on magnetic tape with a fixed interleave distance.

When reproducing PCM signal from VTR, RAM is addressed in the opposite way from that mentioned above. The data stream and block diagram of PCM taperecorder are shown in Fig. 2-11. Memory addressing to achieve "interleaved Matrix Code" encoding and decording is controlled by RAM address control circuit.

2-5. Decoding of "interleaved Matrix Code"

The basic operation of decoding "interleaved Matrix Code" is as follows.

a. Finding out the error words by using CRCC code.

b. De-interleaving the reproduced words by RAMs with RAM address control circuit.

c. Deriving error values e_i, e_j by performing equations (2-7) and (2-8).

Operation of a. and b. is already explained in 2-4. The operation of c. is a little complicated, and we need some knowledge to accomplish the operation with a simple circuit. In equation (2-7), we must operate the two main equations (2-13) and (2-14).

$$(S_1 \oplus T^{i-7} \cdot S_2) \qquad \text{-----} \quad (2\text{-}13)$$

$$(I \oplus T^{i-j})^{-1} \qquad \text{-----} \quad (2\text{-}14)$$

In both equations, the operation of matrix "T" is not very simple because the values of "i" and "j" vary with errors. The equation (2-13) is performed with (T^{-7}) circuit, and Q-Synd generating circuit. The equation (2-14) is performed by $(I \oplus T^{i-j})^{-1}$ ROM circuit. The "interleaved Matrix Code" decoding circuit diagram is shown in Fig. 2-12. And one example of detailed circuit executing equation (2-13) is shown in Fig. 2-13. The operation of $T^{i-7} \cdot S_2$ will be explained with Fig. 2-13. By putting W_1 through T^{-7} circuit composed of seven exclusive OR gates, $T^{-7} \cdot W_1$ is executed. When one shift pulse generates, $T \cdot (T^{-7} \cdot W_1)$ is produced and stored in the shift register. Next step, word W_2 comes from RAM, and $T \cdot (T^{-7} \cdot W_2 \oplus T^{-7} \cdot W_1 \cdot T) = T^{-7} \cdot (T \cdot W_2 \oplus T^2 \cdot W_1)$ is generated and stored in the shift register. By repeating this operation, only excepting redundant word P, we can get the following result, $T^{-7} \cdot S_2$, when word Q appears.

$$T^{-7} \cdot (Q_n \oplus \sum_{k=1}^{6} \widehat{W}_k) = T^{-7} \cdot S_2 \qquad \text{-----} \quad (2\text{-}15)$$

Just after the operation of $T^{-7} \cdot S_2$, i times of shift pulse is applied to the shift register. And we can get the result of $T^i \cdot (T^{-7} \cdot S_2) = T^{i-7} \cdot S_2$. Finally the operation of $(S_1 \oplus T^{i-7} \cdot S_2)$ is carried out quite easily by exclusive OR circuit.

Here, the operation of equation (2-14) will be explained below.

In the equation, assume $i < j$ and re-write, $k = j - i$, we can get the equation (2-15).

$$(I \oplus T^{i-j})^{-1} = (I \oplus T^{-(j-i)})^{-1}$$
$$= (I \oplus T^{-k})^{-1} = M_k \qquad \text{-----} \quad (2\text{-}15)$$

There are five values of K according to the combination of i and j, where $1 \leq i < j \leq 6$. In order to operate the equation (2-7), it is very clever to use the matrix ROM storing the value of M_k. In that case, the equation (2-7) becomes the simple equation (2-16).

$$e_j = M_k \cdot (S_1 \oplus T^{i-7} \cdot S_2) \qquad \text{-----} \quad (2\text{-}16)$$

Five values of matrix M_k, $M_1 - M_5$, are shown in Fig. 2-14. A more datailed circuit of M_k matrix addressing and multiplication of M_k with $(S_1 \cdot \oplus T^{i-7} \cdot S_2)$ is shown in Fig. 2-15.

3. Consumer use digital audio systems

In the near future, every audio system will be processed by digital technique, not only for professional use but for consumer use as well. For realization of consumer use digital audio systems, its hardware cost must be kept sufficiently low. In order to realize this object, we have developed an example of consumer use digital audio systems.

3-1. PCM recording processor with digital dubbing function

One of the special features of the PCM recording processor shown in Fig. 2-11 is its RAM construction. In this construction. RAM is used commonly for recording and reproducing. This is one effective way of realizing a less expensive and also higher quality PCM taperecorder.

For the purpose of realizing digital audio systems, the output signal of a PCM taperecorder must be digitally transferred to other audio equipments in order not to cause signal quality deterioration. Especially as a consumer use system, it is more convenient to get the digital dubbing signal in the form of TV signal in order to record a PCM signal onto VTR. This type of PCM recording processor is achieved with additional P.Q encoding circuit and RAM as shown in Fig. 3-1.

The PCM signal from VTR 1 is de-interleaved in RAM 1 and its errors are corrected in P,Q decoding circuit. Then that digital audio signal is again P.Q encoded, interleaved in RAM2 in Digital Dubbing section and is sent to VTR2 for recording. Of course an analog output and digital output can be obtained directly from the P.Q decoding circuit.

3-2. Consumer use digital audio systems

So long as the sampling frequency of PCM Recording processor is based on PCM Standard Format, its digital input and output can be easily connected to other digital audio equipments as shown in Fig. 3-2. Of course other related digital audio equipments can be constructed based on the format. Such as an editing machine. As consumer use digital audio systems, PCM taperecorder or recording processor, A/D and D/A unit and digital mixer will be considered to be the most essential equipments. In connecting these digital audio equipments to each other, one equipment must be selected as a master clock generator.

In Fig.3-2, analog signal line is marked with "A", digital signal line is marked with "D" and its timing signal line is marked with "T". Fig. 3-2 (a) shows the digital audio system where the PCM processor operates as a master clock generator, and its frequency is synchronized to the reproduced signal from VTR 1 by means of PLL.

In the system of Fig. 3-2 (b), A/D unit 1 is selected as a master clock generator. Usually consumer use VTR cannot be synchronized to the external sync. signal, but on the other hand, however, most of the professional VTR can be synchronized to the external sync. signal. Digital audio system, being used as a studio recording or other professional purposes, can be achieved using professional type VTR as shown in Fig. 3-2 (c). In this case, every equipment is synchronized to the master clock.

A series of digital audio equipments shown in Photo 3-1 have been developed. Clock signal of every equipment can be selected as a master or a slave.

3-3. Digital Interface

Fig. 3-3 shows an example of digital interface, connecting respective digital equipments. This kind of digital interface is suitable for consumer use digital audio systems because of its simplicity in hardware.

Only three twisted-pair lines are required. One is for transmitting data signals serially, the second for shift clock signal, and the third for R-L recognizing signal. Interval time for one sample is set as

$\frac{1}{f_s}$ = 22.7 μS. This one sample time is divided into 40 slots, namely 20 slots for each of L channel and R channel. And 14 positions of each 20 slots are used for transmitting one word. Its transfer ratio of data transmission is 1.76 MB/S and is easily transmitted to the other equipment with the interface condition of TTL and with twisted-pair lines.

Acknowledgements

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(1) M. Kosaka: "Sampling-Frequency Considerations", Journal of the AES, PP. 234-240, April 1978

(2) T. Seno, R. Wada: "A Consideration of the Error Correcting Codes for PCM Recording System", 61st AES Convention, preprint No. 1397 (H-4), Nov. 1978

(3) D.C. Bossen: "b-Adjacent Error Correction", IBM J. RES. DEVELOP, PP. 402-408, July 1970

No.	Items	Characteristics
1	Channels	2 (L, R)
2	Freq. response	DC ~ 20 KHz
3	Sampling freq.	f_s = 44.056 KHz (NTSC) Fig.1-1
4	Quantization	14 bit linear
5	Composite signal	Standard TV format
6	Signal allocation in 1H	Standard TV format Fig.1-2
7	Signal allocation in 1V	Standard TV format Fig.1-3
8	Drop out compensation	Error correction by "interleaved Matrix Code"
9	VTR	Any home use VTR Fig.1-2, 3

Table 1-1 Basic specifications for digital audio systems

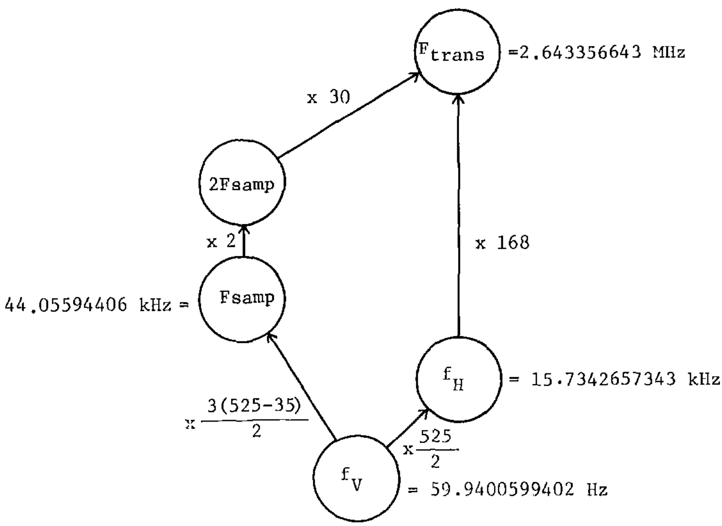


Fig.1-1 Frequency relationship between f_s and fv of NTSC

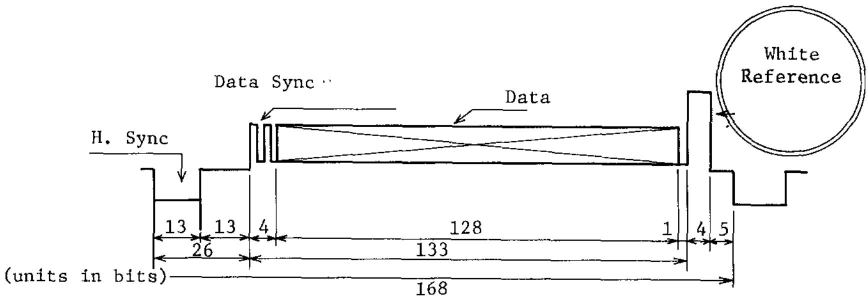


Fig.1-2 Signal allocation in 1H

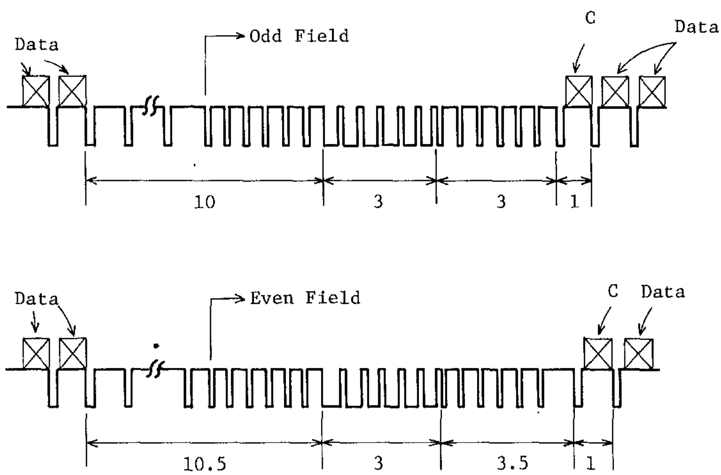


Fig.1-3 Signal allocation in 1V

D \ B	38	40	42
35	D= 35 B= 38 F orig= 918,187,500 F trans= 2,593,750 F samp= 44,250 P= 10375 Q= 354 N= 29.309	D= 35 B= 40 F orig= 154,875,000 F trans= 2,625,000 F samp= 44,250 P= 1750 Q= 59 N= 29.661	D= 35 B= 42 F orig= 940,312,500 F trans= 2,656,250 F samp= 44,250 P= 10625 Q= 354 N= 30.014
37		D= 37 B= 40 F orig= 55,125,000 F trans= 2,625,000 F samp= 44,100 P= 625 Q= 21 N= 29.762	
39		D= 39 B= 40 F orig= 769,125,000 F trans= 2,625,000 F samp= 43,950 P= 8750 Q= 293 N= 29.863	
40	D= 40 B= 38 F orig= 910,406,250 F trans= 2,593,750 F samp= 43,875 P= 10375 Q= 351 N= 29.558	D= 40 B= 40 F orig= 307,125,000 F trans= 2,625,000 F samp= 43,875 P= 3500 Q= 117 N= 29.915	D= 40 B= 42 F orig= 932,343,750 F trans= 2,656,250 F samp= 43,875 P= 10625 Q= 351 N= 30.271

Fig.1-4 Frequency relationship of PAL TV format

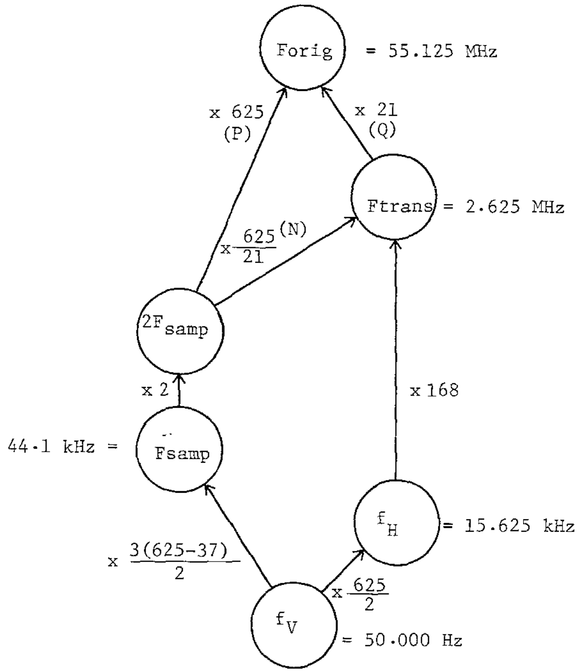


Fig.1-5 Frequency relationship between f_s and f_V of PAL format

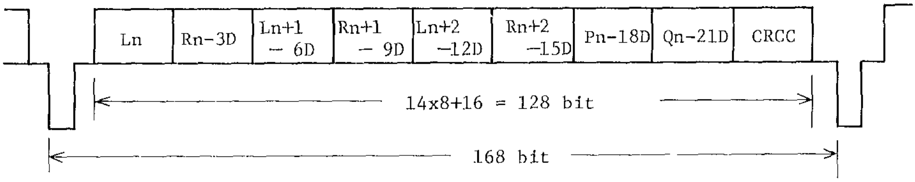
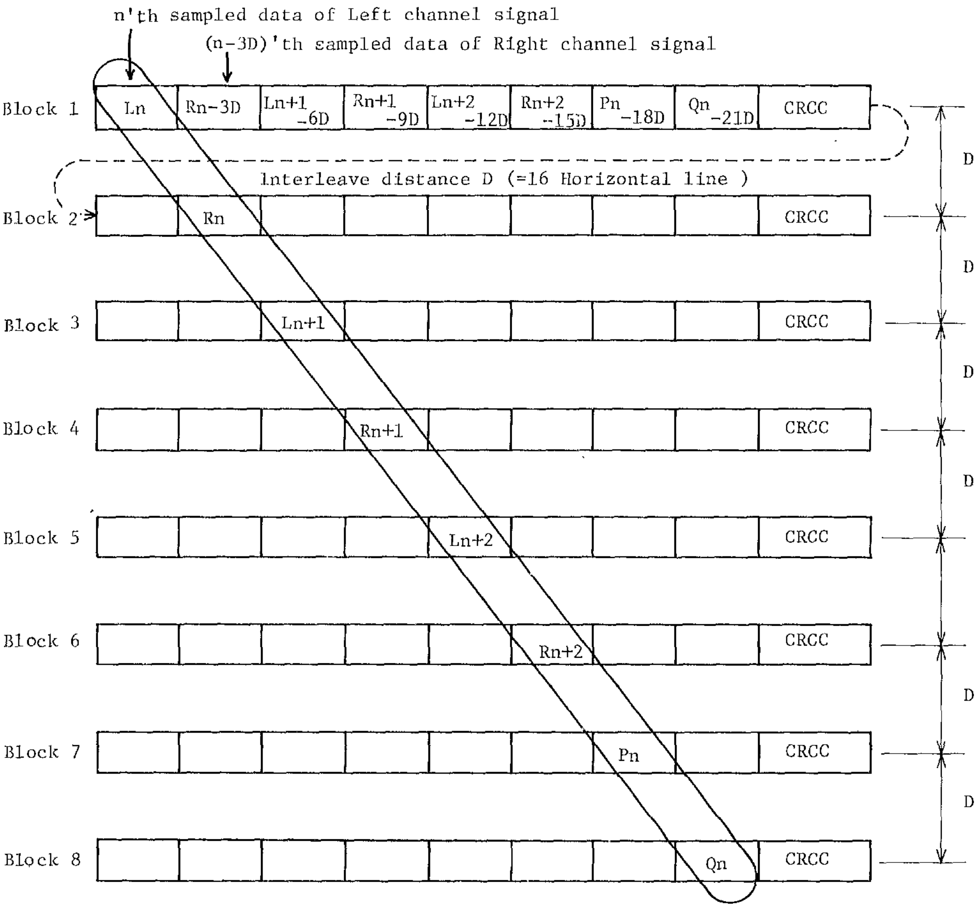


Fig.2-1 PCM data allocation in 1H



D: Interleave Distance (= 16 Horizontal Line)

Fig.2-2 Interleaved data on magnetic tape

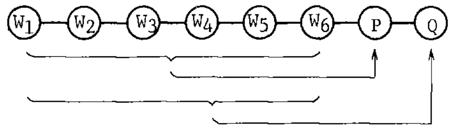


Fig.2-3 Data words and redundant words P,Q.

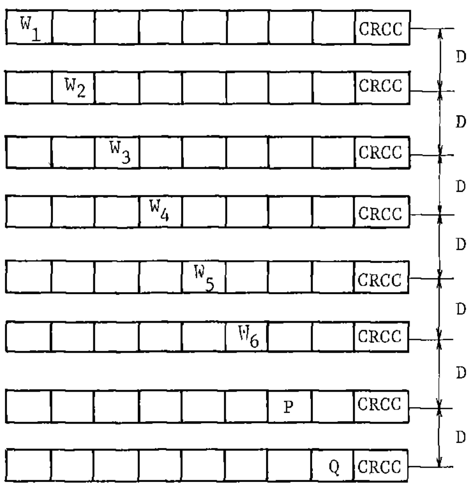


Fig.2-4 Interleaved data on magnetic tape

$$
T = \begin{pmatrix}
0&0&0&0&0&0&0&0&0&0&0&0&0&1\\
1&0&0&0&0&0&0&0&0&0&0&0&0&0\\
0&1&0&0&0&0&0&0&0&0&0&0&0&0\\
0&0&1&0&0&0&0&0&0&0&0&0&0&0\\
0&0&0&1&0&0&0&0&0&0&0&0&0&0\\
0&0&0&0&1&0&0&0&0&0&0&0&0&0\\
0&0&0&0&0&1&0&0&0&0&0&0&0&0\\
0&0&0&0&0&0&1&0&0&0&0&0&0&0\\
0&0&0&0&0&0&0&1&0&0&0&0&0&1\\
0&0&0&0&0&0&0&0&1&0&0&0&0&0\\
0&0&0&0&0&0&0&0&0&1&0&0&0&0\\
0&0&0&0&0&0&0&0&0&0&1&0&0&0\\
0&0&0&0&0&0&0&0&0&0&0&1&0&0\\
0&0&0&0&0&0&0&0&0&0&0&0&1&0
\end{pmatrix}
$$

Fig.2-5 Q generating matrix "T"

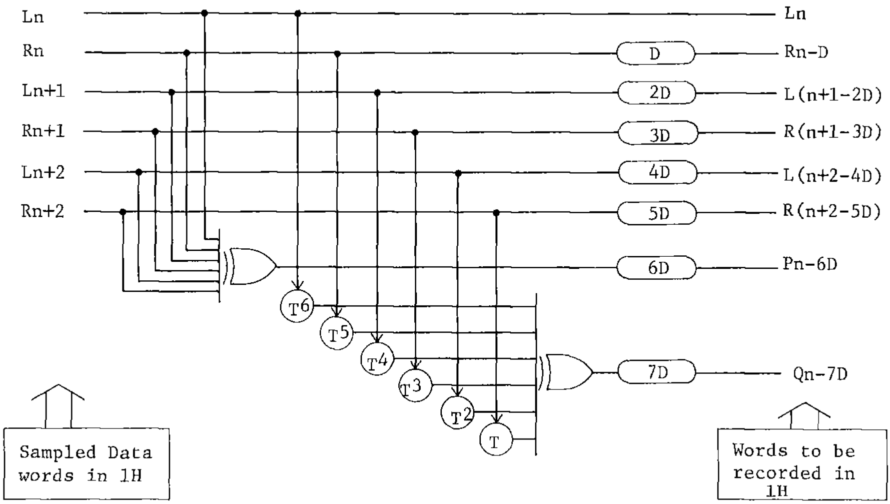


Fig.2-6 "interleaved Matrix Code" generator

Items		Characteristics
Uncorrectable probability		$21 \cdot P_H{}^3$
uncorrectable error number per hour	$P_H{}^* = 10^{-3}$	1.11/hour
	$P_H{}^* = 5x10^{-3}$	0.139/hour
	$P_H{}^* = 10^{-4}$	$1.1x10^{-3}$/hour
error correcting ability VS burst error length		0 ~ 2D: Corrected
		2D ~ 4D: 1 wd or 2 wds continuous error
		4D ~ : more than 3 wds continuous error

* Error rate in H

Table 2-1 Main characteristics of "interleaved Matrix Code"

```
file No.
15
total time (min)
         77.21
total Hs
      68032747
total H errors
          5712
H error rate
     8.396E-05
total errors
             0
error rate
     0.000E 00
**distribution**
   L  channel
  single     0
   R  channel
  single     0
calculating time
        +36.00
         (min)
```

(a)

```
file No.
0
total time (min)
         51.39
total Hs
      45278167
total H errors
         22070
H error rate
     4.874E-04
total errors
             8
error rate
     2.945E-08
**distribution**
   L  channel
  single     1
  2words     2
   R  channel
  single     1
  2words     1
```

(b)

Fig.2-7 Error correcting ability of "interleaved Matrix Code" tested by hardware simulator

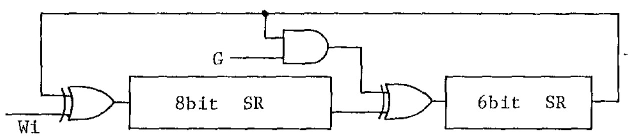


Fig.2-8 T·Wi processing circuit

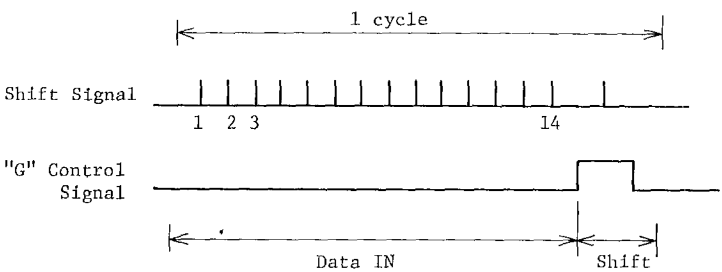


Fig.2-9 Shift signal and gate control signal for T·Wi process

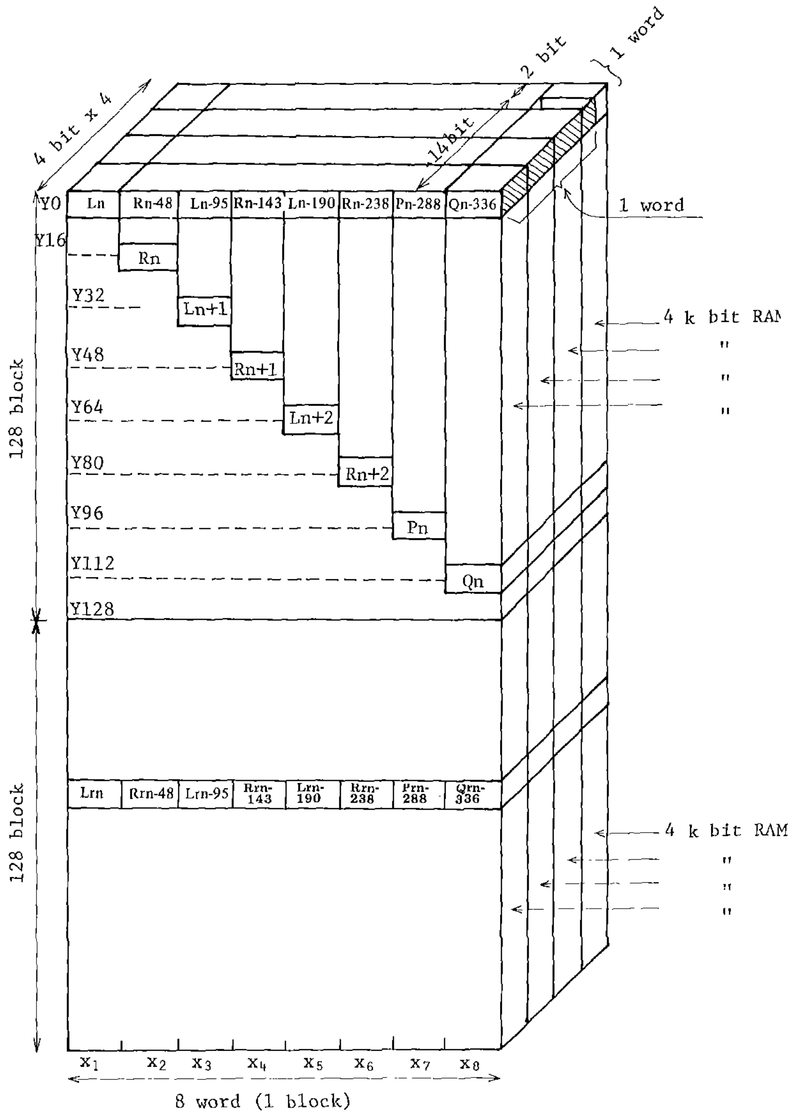


Fig.2-10 Memory adressing for interleaved recording

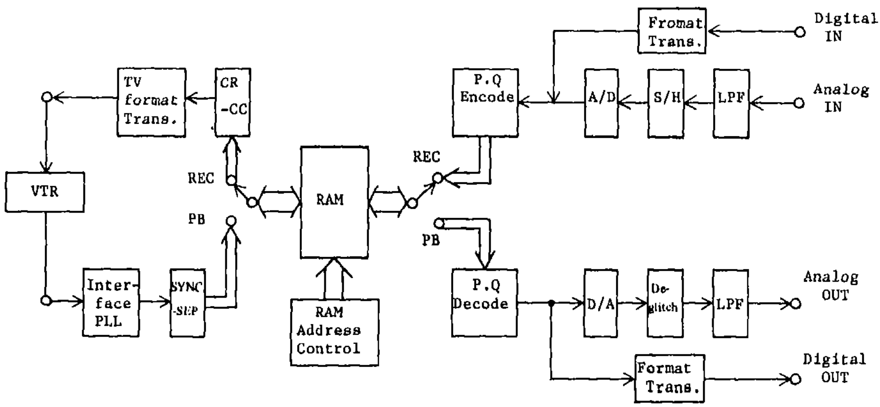


Fig.2-11 Construction of PCM taperecorder

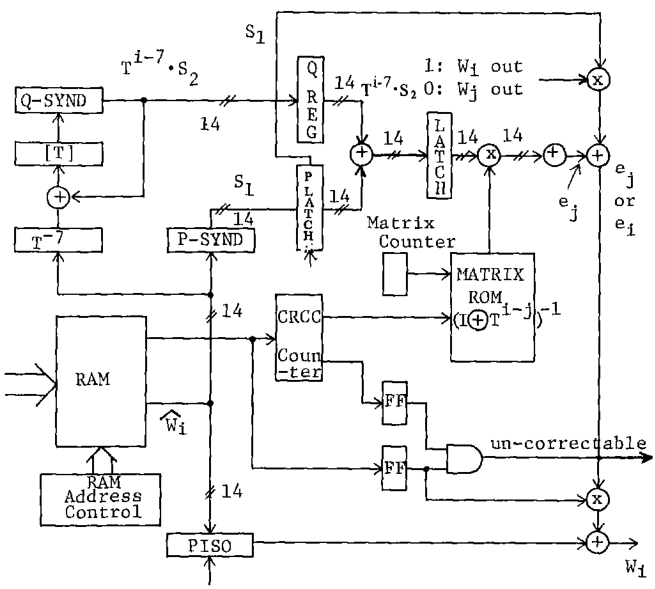


Fig.2-12 "interleaved Matrix Code" decoding circuit

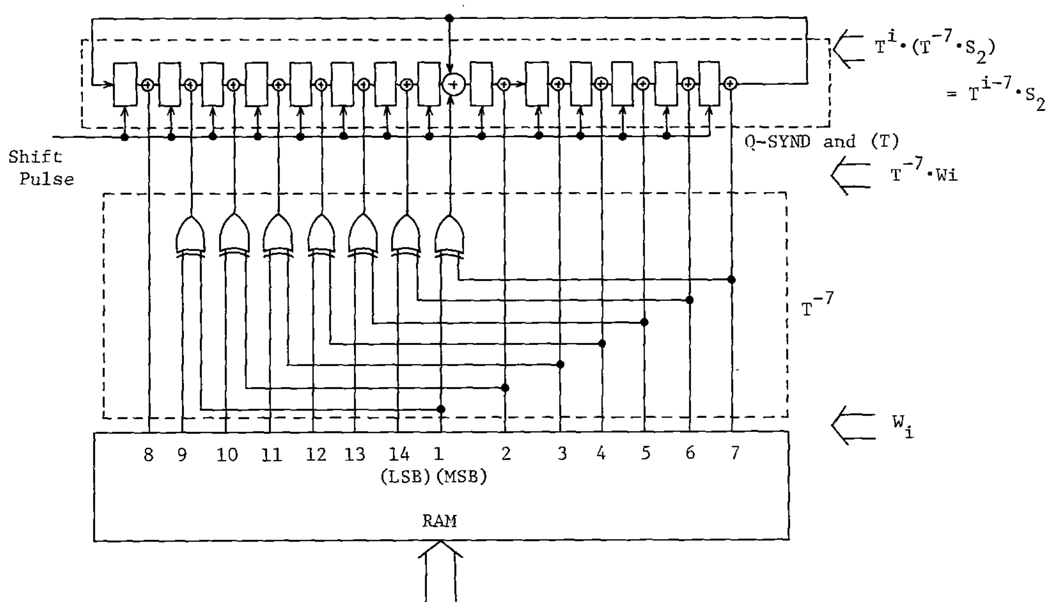


Fig.2-13 $T^{i-7} \cdot S_2$ processing circuit

$$
M1 = \begin{pmatrix}
1&1&1&1&1&1&1&1&1&1&1&1&1&1\\
0&1&1&1&1&1&1&1&1&1&1&1&1&1\\
0&0&1&1&1&1&1&1&1&1&1&1&1&1\\
0&0&0&1&1&1&1&1&1&1&1&1&1&1\\
0&0&0&0&1&1&1&1&1&1&1&1&1&1\\
0&0&0&0&0&1&1&1&1&1&1&1&1&1\\
0&0&0&0&0&0&1&1&1&1&1&1&1&1\\
0&0&0&0&0&0&0&1&1&1&1&1&1&1\\
1&1&1&1&1&1&1&1&0&0&0&0&0&0\\
1&1&1&1&1&1&1&1&1&0&0&0&0&0\\
1&1&1&1&1&1&1&1&1&1&0&0&0&0\\
1&1&1&1&1&1&1&1&1&1&1&0&0&0\\
1&1&1&1&1&1&1&1&1&1&1&1&0&0\\
1&1&1&1&1&1&1&1&1&1&1&1&1&0
\end{pmatrix}
\qquad
M2 = \begin{pmatrix}
1&0&1&0&1&0&1&0&1&0&1&0&1&0\\
0&1&0&1&0&1&0&1&0&1&0&1&0&1\\
0&0&1&0&1&0&1&0&1&0&1&0&1&0\\
0&0&0&1&0&1&0&1&0&1&0&1&0&1\\
0&0&0&0&1&0&1&0&1&0&1&0&1&0\\
0&0&0&0&0&1&0&1&0&1&0&1&0&1\\
0&0&0&0&0&0&1&0&1&0&1&0&1&0\\
0&0&0&0&0&0&0&1&0&1&0&1&0&1\\
1&0&1&0&1&0&1&0&0&0&0&0&0&0\\
0&1&0&1&0&1&0&1&0&0&0&0&0&0\\
1&0&1&0&1&0&1&0&1&0&0&0&0&0\\
0&1&0&1&0&1&0&1&0&1&0&0&0&0\\
1&0&1&0&1&0&1&0&1&0&1&0&0&0\\
0&1&0&1&0&1&0&1&0&1&0&1&0&0
\end{pmatrix}
\qquad
M3 = \begin{pmatrix}
1&0&0&1&0&0&1&0&0&1&0&0&1&0\\
0&1&0&0&1&0&0&1&0&0&1&0&0&1\\
0&0&1&0&0&1&0&0&1&0&0&1&0&0\\
0&0&0&1&0&0&1&0&0&1&0&0&1&0\\
0&0&0&0&1&0&0&1&0&0&1&0&0&1\\
0&0&0&0&0&1&0&0&1&0&0&1&0&0\\
0&0&0&0&0&0&1&0&0&1&0&0&1&0\\
0&0&0&0&0&0&0&1&0&0&1&0&0&1\\
1&0&0&1&0&0&1&0&1&1&0&1&1&0\\
0&1&0&0&1&0&0&1&0&1&1&0&1&1\\
0&0&1&0&0&1&0&0&1&0&1&1&0&1\\
1&0&0&1&0&0&1&0&0&1&0&1&1&0\\
0&1&0&0&1&0&0&1&0&0&1&0&1&1\\
0&0&1&0&0&1&0&0&1&0&0&1&0&1
\end{pmatrix}
$$

$$
M4 = \begin{pmatrix}
0&0&1&0&0&0&1&0&0&0&1&0&0&0\\
0&0&0&1&0&0&0&1&0&0&0&1&0&0\\
1&0&0&0&1&0&0&0&1&0&0&0&1&0\\
0&1&0&0&0&1&0&0&0&0&0&0&0&1\\
1&0&1&0&0&0&1&0&0&0&1&0&0&0\\
0&1&0&1&0&0&0&1&0&0&0&1&0&0\\
1&0&1&0&1&0&0&0&1&0&0&0&1&0\\
0&1&0&1&0&1&0&0&0&1&0&0&0&1\\
1&0&0&0&1&0&0&0&0&0&0&0&0&0\\
0&1&0&0&0&1&0&0&0&0&0&0&0&0\\
0&0&1&0&0&0&1&0&0&0&0&0&0&0\\
0&0&0&1&0&0&0&1&0&0&0&0&0&0\\
1&0&0&0&1&0&0&0&1&0&0&0&0&0\\
0&1&0&0&0&1&0&0&0&1&0&0&0&0
\end{pmatrix}
\qquad
M5 = \begin{pmatrix}
0&1&1&0&1&0&1&1&0&1&0&1&1&0\\
1&0&1&1&0&1&0&1&1&0&1&0&1&1\\
0&1&0&1&1&0&1&0&1&1&0&1&0&1\\
1&0&1&0&1&1&0&1&0&1&1&0&1&0\\
1&1&0&1&0&1&1&0&1&0&1&1&0&1\\
1&1&1&0&1&0&1&1&0&1&0&1&1&0\\
1&1&1&1&0&1&0&1&1&0&1&0&1&1\\
0&1&1&1&1&0&1&0&1&1&0&1&0&1\\
1&1&0&1&0&1&1&0&0&0&1&1&0&0\\
0&1&1&0&1&0&1&1&0&0&0&1&1&0\\
1&0&1&1&0&1&0&1&1&0&0&0&1&1\\
0&1&0&1&1&0&1&0&1&1&0&0&0&1\\
1&0&1&0&1&1&0&1&0&1&1&0&0&0\\
1&1&0&1&0&1&1&0&1&0&1&1&0&0
\end{pmatrix}
$$

Fig. 2-14 Mk matrix

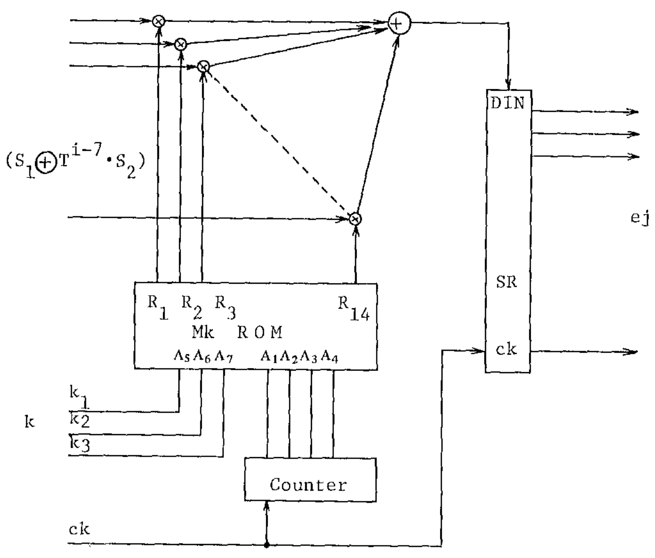


Fig.2-15 $Mk \cdot (S_1 \oplus T^{i-7} \cdot S_2)$ processing circuit

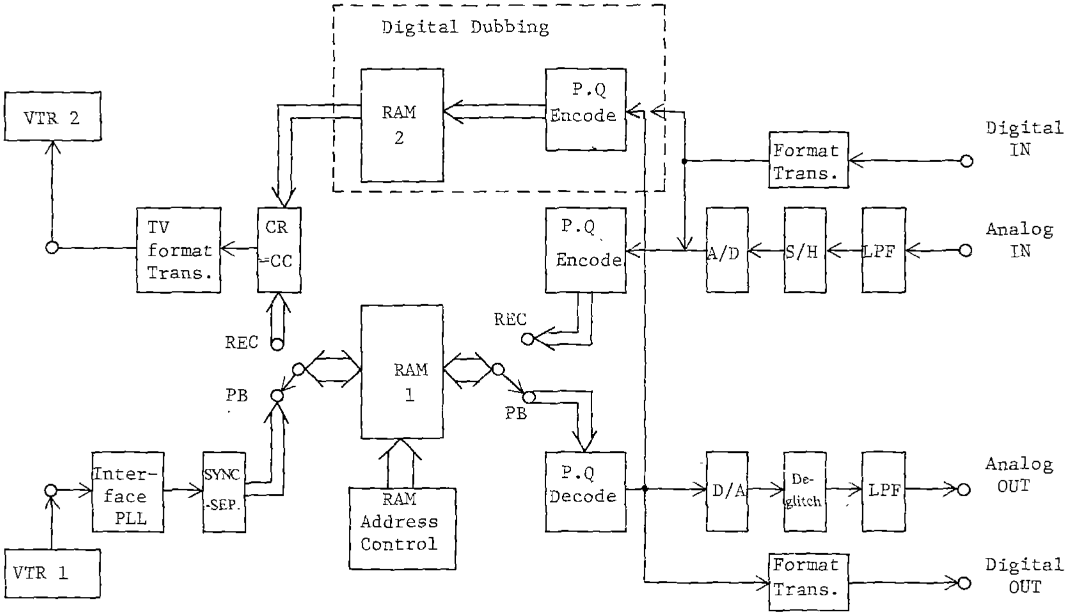


Fig.3-1 Construction of PCM processor with digital dubbing function

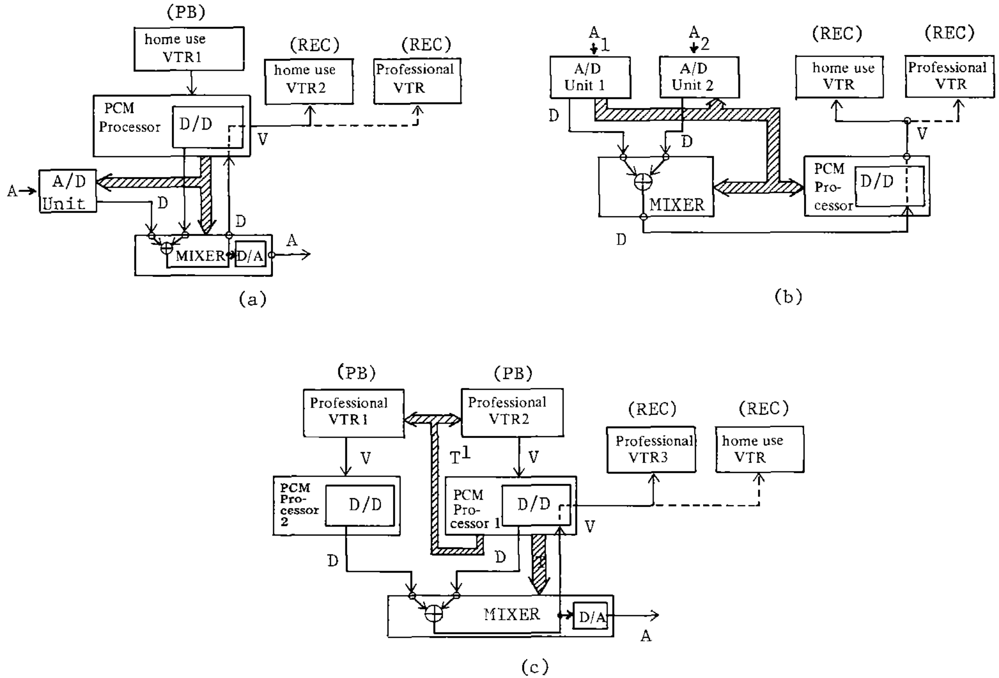


Fig.3-2 Interconnection of digital audio systems

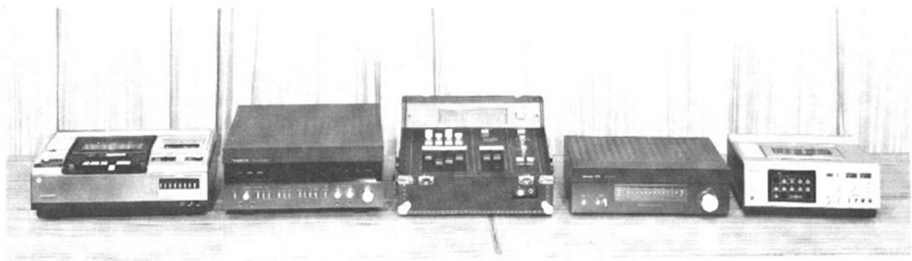


Photo 3-1 A series of digital audio equipments

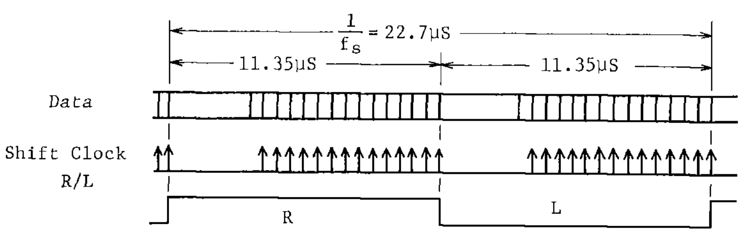


Fig.3-3 Digital interface