

A LONG PLAY DIGITAL AUDIO DISC SYSTEM

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AN AUDIO ENGINEERING SOCIETY PREPRINT

A LONG PLAY DIGITAL AUDIO DISC SYSTEM

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ABSTRACT

The development of video disc systems has turned out the possibility of a long play digital (PCM) audio disc systems. The band width required for two channel of digital audio signals is less than that of video signals and reduction of the revolution makes the longer playing time be possible. For the further improvement of packing density, a kind of run length limited code is adopted. The code is called as 3PM (three-position modulation), and the packing density can be 150 percent of MFM coding for the same minimum wave length to be recorded. This is achieved at the expence of decreasing jitter margine, which is relatively easy to solve in optical disc systems.

The playing time of two and a half hours is achieved on one side of optical disc with diameter of 30cm. The sampling rate is 44.056kHz and each of two channels is coded by 16 bit linear quantization. The revolution is 450 rpm. Code errors are analyzed for each revolution of 1800, 900 and 450 rpm on the plane of bit error rate and bit error correlation coefficient. An effective error correcting scheme named "Cross Interleave" is developed which is possible to decode by various levels of decoder from simple erasure type to the complex "Corss Word" type with keeping full compatibility.

1. INTRODUCTION

A long play digital audio disc system has developed after three year studies of increasing packing density. Table 1 shows our steps of development^{(1),(2),(3)}, which indicates the playing time on the same size of a disc has increased by five times without decreasing the minimum wave length to be recorded.

The advance is mainly achieved by changing coding method from NRZ-FM (video format) into MFM and then into 3PM⁽⁴⁾ accordingly revolution could be reduced as 1800 r.p.m., 900 r.p.m. and 450 r.p.m..

Table 1

Steps for Development of A Long Play Digital Audio Disc System in Sony

Step	1	2	3
Completed Date	1976 Sep.	1977 Sep.	1978 Sep.
Coding	NRZ-FM (Analogue Video Format)	MFM	3PM
Playing Time (minutes)	30 minutes	60 minutes	150 minutes
Revolution (rpm)	1800	900	450
Track Pitch (μ m)	2.5	1.7	1.3
Minimum Wave Length (μ m)	1.1	3.4	2.4
Diameter of a Disc (mm)	303 ϕ	303 ϕ	303 ϕ
Sampling Rate (kHz)	44.1	44.1	44.056
No of Channels	2	2	2
Quantization (bits)	13 Non Linear	13 Non Linear	16 Linear
Error Correcting Schemes	Combination of Random and Burst Error Correction		Cross Interleave System
References		(1), (2)	(3)

The quantization is changed from 13 bit nonlinear into 16 bit linear, so that the quality of the master recording system will be exactly obtained until the final pressed disc to be played at homes.

Another great advance in these developments is that of error correcting schemes. The characteristics of code errors are different by the revolution of the discs, which are analyzed and an effective correcting method is described in this paper.

2. DIGITAL AUDIO DISC SYSTEMS

Fig 1 shows the total system of digital audio discs.

The mastering recorder is PCM-1600, VTR based, two channel, 16 bit system, and all the signal processing is carried out in digital until the final D/A converters in the player. For that reason, the sampling rate of the system is chosen as 44.056 kHz in line with PCM-1600.

The process from cutting to the production of discs is not different from that of video disc systems. The player consists of signal pick-up part (disc drive, tracking and focus servo, and He-Ne laser tube and its optics) and signal processing part (PLL, frame and bit synchronization, demodulation of 3PM⁽⁴⁾, time base correction, error correction, and D/A). The former is almost the same as video disc player, but as the revolution is 450 r.p.m. the higher stability is required. The latter is described in the following sections.

3. ANALYSIS OF CODE ERRORS

The causes of code errors of optical digital audio disc systems are expected as follows.

- (1) defects of pits on discs
- (2) defects of metallic coating on discs
- (3) bubbles, irregular reflection or other defects of discs
- (4) dusts, scratches, finger-prints on discs
- (5) de-focusing
- (6) miss-tracking
- (7) noise
- (8) jitter
- (9) fluctuation of RF levels
- (10) inter-symbol interference

If the revolution of discs is reduced and the packing density is increased, the length of the code errors caused by (1)~(4) will be increased, but the bit error rate will remain in the same value. But the reduction of the revolution might make the flutter of the disc worse and the code errors by (5), (6) might increase. Besides that, the more the packing density increase, the severer be the code errors caused by (7)~(8), generally.

Design of error correcting schemes deeply depends on the characteristics of code errors and thus the selection of the revolution should be very careful. In this paper, the analysis of code errors is carried out based on a simple statistical model called "Gilbert Model^{(7),(8)}" which is shown in Fig 3. Where B and G express the states of error and no error, respectively, and the letters T and t are the transition probabilities. The characteristics of error are expressed by two parameters, the bit error rate ρ , and the bit error correlation coefficient ξ .

$$\rho = \frac{T}{T + t} \quad (1)$$

$$\xi = 1 - T - t \quad (2)$$

Fig 4 shows an example of the difference of the error characteristics due to the revolution of the discs on the " $\rho - \xi$ plane". The calculation is carried out by comparing the block error rate before and after error correction. Error correction is chosen as (1) 1 bit error, (2) 2 bit random error, and (3) 15 bit burst error, while the length of the block is between 60 and 75 bits. The error rate of n-bit-block is expressed as follows.

$$E_{BLK} = 1 - \frac{T}{T + t} (1 - T)^{n-1} \quad (3)$$

The probability of 1 bit error E_1 , 2 bit random error E_2 and that of burst error within 15 bits E_{15B} in n-bit-block are expressed as follows.

$$E_1 = \frac{Tt}{T + t} (1 - T)^{n-2} \{2(1 - T) + (n - 2)t\} \quad (4)$$

$$\begin{aligned} E_2 &= \frac{Tt}{T + t} (1 - T)^{n-3} \{ (1 - T)^2 \{2(1 - t) + T\} \\ &\quad + (n - 3)(1 - T)t \{2T + (1 - t)\} \\ &\quad + \frac{1}{2}t^2(n - 3)(n - 4) \} \end{aligned} \quad (5)$$

$$E_{15B} = \sum_{i=1}^{15} \frac{Tt}{T + t} (1 - t)^{i-1} (1 - T)^{n-i-1} \{2 + (n - i - 1)t\} \quad (6)$$

The data shown in Fig 4 is not meaningful enough, because the discs are produced in various times and in the normal room condition. Usually the quality of discs much depends on production lots rather than the revolution or the error correcting schemes. If precise data are necessary, a disc including various revolution in various part should be prepared.

Nevertheless, a trend of errors is obviously found in Fig 4, that the value of bit error correlation coefficient decreases as the revolution reduced, and the bit error rate does not significantly depend on the revolution.

4. ERROR CORRECTING SCHEMES

In this system, an effective error correcting scheme named "Cross Interleave Correction" is adopted, which consists of delayed interleave and at least two sets of delayed parity word⁽⁴⁾. Decoding is based on pointer-erasure method, where the pointer of the erroneous word is obtained by CRCC. For the further correcting ability, "Cross Word Decoding" is possible with keeping full compatibility to the pointer-erasure decoding.

4-1. ENCODER

Fig 5 shows the format and the encoder. One frame consists of 18 bit synchronization word and three sub-blocks. Each sub-block includes two information words (L_i , R_i) and two parity words (P_i , Q_i) and CRCC. The encoder consists of simple delay memories, exclusive-or gate, and CRCC encoder. The parities P_i , Q_i are composed by the following equation.

$$P_i = L_i \oplus R_i \quad (7)$$

$$Q_i = L_i \oplus R_{i-1} \oplus P_{i-1} \quad (8)$$

Where, \oplus means modulo-two summation (exclusive-or).

4-2. THE SIMPLEST BASIC DECODER

Fig 6 shows the simplest decoder, which only utilize the parity words P_i (the parity words Q_i are ignored). Each sub-block is checked by CRCC decoder and the error information of 1 bit is fed to the parity P-decoder after giving the same delay as the main information words. In P-decoder, a syndrome word

$$S_{pi} = \acute{L}_i \oplus \acute{R}_i \oplus \acute{P}_i \quad (9)$$

is calculated. Where the prime mark indicates the received word which might include errors. If $S_{pi} = 0$, or all the error information bits indicate no error, the words \acute{L}_i and \acute{R}_i are considered not to include any code errors. If $S_{pi} \neq 0$, and the error information bit of \acute{L}_i indicates error (\acute{R}_i and \acute{P}_i are no error), for instance, \acute{L}_i is supposed to be erroneous, and since the error pattern appears in S_{pi} , the error correction is carried out by the following formula.

$$L_i = \overset{\cdot}{L}_i + S_{pi} \quad (10)$$

Error Correction is possible, as far as two of the related sub-blocks for one syndrome are not erroneous. Each related sub-block is separated 16 sub-blocks each other, therefore a burst error within 16 sub-blocks (1290~1308 bits) can be corrected by this simplest decoder. While a guard space of 32 sub-blocks is necessary for that correction.

4-3. MULTIPLE DECODERS

Fig 7 shows a double decoder, which computes two syndromes S_{pi} , and

$$S_{qi} = L_i \oplus R_{i-3} \oplus P_{i-8} \oplus Q_i \quad (11)$$

Using each syndromes S_{pi} and S_{qi} , one error word can be corrected by the method described above. Supposing $\overset{\cdot}{L}_i$ and $\overset{\cdot}{R}_i$ are error, it can not be corrected by the basic decoder shown in Fig 6. But using double decoder (Fig 7), both words can be corrected by using S_{qi} .

$$S_{q_i} = L_i^x \oplus R_{i-2}^x \oplus P_{i-7}^x \oplus Q_i^x \quad (12)$$

$$S_{q_4} = L_4^x \oplus R_1^x \oplus P_{-4}^x \oplus Q_4^x \quad (13)$$

Where the upper script x indicates erroneous word. Supposing L_i , R_i , and R_{i-2} are erroneous, two word errors are included both in S_{pi} and S_{qi} , nevertheless, error correction is possible by the double decoder as the procedure shown below.

<Step 1: Q-Decoder>

$$S_{q_i} = L_i^x \oplus R_{i-2}^x \oplus P_{i-7}^x \oplus Q_i^x \quad (14)$$

$$S_{q_4} = L_4^x \oplus R_1^x \oplus P_{-4}^x \oplus Q_4^x \quad (15)$$

R_1^x is corrected by S_{q_4} , but L_i^x and R_{i-2}^x are still erroneous.

<Step 2: P-Decoder>

$$S_{p_i} = L_i^x \oplus R_i^x \oplus P_i^x \quad (16)$$

$$S_{p_{i-2}} = L_{i-2}^x \oplus R_{i-2}^x \oplus P_{i-2}^x$$

Since R_i^x is corrected by the preceding step, L_i^x is corrected by S_{p_i} , and R_{i-2}^x by $S_{p_{i-2}}$.

Fig 8 shows the four times decoder. Supposing L_1^x , R_1^x , L_{-2}^x , R_{-2}^x , and Q_{-2}^x are erroneous, four times correction is necessary.

<Step 1: Q-Decoder>

$$S_{q_{-2}} = L_{-2}^x \oplus R_{-3}^x \oplus P_{-10}^x \oplus Q_{-2}^x \quad (18)$$

$$S_{q_i} = L_i^x \oplus R_{i-2}^x \oplus P_{i-7}^x \oplus Q_i^x \quad (19)$$

$$S_{q_4} = L_4^x \oplus R_1^x \oplus P_{-4}^x \oplus Q_4^x \quad (20)$$

R_1^x is corrected by Sq_4 , but others can not be corrected.

<Step 2: P-Decoder>

$$Sp_1 = L_1^x \oplus R_1^x \oplus P_1^x \quad (21)$$

$$Sp_2 = L_2^x \oplus R_2^x \oplus P_2^x \quad (22)$$

Since R_1^x is corrected at the preceding step, L_1^x can be corrected by Sp_1 .

<Step 3: Q-Decoder>

$$Sq_1 = L_1^x \oplus R_2^x \oplus P_7^x \oplus Q_1^x \quad (23)$$

$$Sq_2 = L_2^x \oplus R_3^x \oplus P_6^x \oplus Q_2^x \quad (24)$$

R_2^x is corrected by Sq_1 .

<Step 4: P-Decoder>

$$Sp_2 = L_2^x + R_2^x + P_2^x \quad (25)$$

L_2^x is corrected by Sp_2 , and error correction is completed.

While if the double decoder (Fig 7) is used for the above example, L_2 and R_2 are remained as uncorrectable. This is one of the features of the Cross Interleave System that the more the steps of decoding increase, the better correctability is obtained.

Ofcause there is some probability of uncorrectable errors, however steps of decoding are prepared. One of the examples is shown below.

$$Sq_1 = L_1^x \oplus R_2^x \oplus P_7^x \oplus Q_1^x \quad (26)$$

$$Sq_4 = L_4^x \oplus R_1^x \oplus P_4^x \oplus Q_4^x \quad (27)$$

$$Sp_1 = L_1^x \oplus R_1^x \oplus P_1^x \quad (28)$$

In this case, L_1^x , R_1^x , Q_1^x , and Q_4^x are erroneous. All related syndromes, eqs. (26)~(28) include two erroneous words.

4-4. CROSS WORD DECODING ^{(10), (11), (12), (13)}

If an information word relates to plural syndromes, cross word decoding is possible by comparing those syndrome patterns. In above format, three types of syndrome Sp_i , Sq_i and residual of CRCC can be utilized.

In the case shown above by eqs. (26)~(28), related syndromes by CRCC, Sci are as follows.

$$Sc_1 = \text{Res}(L_1^x, R_1^x, P_1^x, Q_3^x, C_1^x) \quad (29)$$

$$Sc_{17} = \text{Res}(L_{17}^x, R_1^x, P_{15}^x, Q_{13}^x, C_{17}^x) \quad (30)$$

$$Sc_{36} = \text{Res}(L'_{36}, R'_{20}, P'_4, Q'_1, C'_{36}) \quad (31)$$

$$Sc_{39} = \text{Res}(L'_{39}, R'_{23}, P'_1, Q'_4, C'_{39}) \quad (32)$$

Where $\text{Res}(\quad)$ means residual of CRCC, and C_i is received CRCC word.

In the case described above as uncorrectable by erasure method, all the equations (26)~(32) are not zero. If the words L_i^x , R_i^x , Q_1^x , and Q_4^x are really erroneous, they can not be corrected by all means. But the error position in eqs. (29)~(32) can not be indicated and the following error patterns are also considered as uncorrectable.

$$(i) \quad L_i^x, R_i^x, R_{20}^x, C_{39}^x$$

$$(ii) \quad L_i^x, L_{17}^x, P_4^x, Q_4^x$$

If the word error is considered as random, real uncorrectable errors are 1% of uncorrectable errors by above decoders, and 99% will be corrected by the Cross Word Decoder. The principle of Cross Word decoding is to find the error location by comparing syndromes; for instance, in case (i) the equation

$$Sp_1 = Sq_1 + Sq_4 \quad (33)$$

will be satisfied, and in case (ii),

$$Sq_1 = Sp_1. \quad (34)$$

There is another method of Cross Word decoding, that is to put the parity words into CRCC decoder, and after appropriate shifting, compare with eqs. (29)~(32).⁽¹³⁾ If the error is within 11 bit burst, the error location in eqs. (29)~(32) can be found by this method.

5. 3PM CODE

Among the various class of the run length limited code, 3PM (three-position modulation)⁽¹⁴⁾ is selected because of its relatively simple hardware and fairly good efficiencies. The principle of 3PM, shown in Table 2, is to convert three bit of original data word into six-bit word, in which "1" means transition, and any 1's are always separated by two 0's. In other words, the minimum duration between transitions is 1.5 L, that is 50% larger than that of MFM, where L is the length of original data bit cell.

Table 2 3PM

No	Original Data Word			Transition Positions					
1	0	0	0	0	0	0	0	1	0
2	0	0	1	0	0	0	1	0	0
3	0	1	0	0	1	0	0	0	0
4	0	1	1	0	1	0	0	1	0
5	1	0	0	0	0	1	0	0	0
6	1	0	1	1	0	0	0	0	0
7	1	1	0	1	0	0	0	1	0
8	1	1	1	1	0	0	1	0	0
	← L →			← L →					

At the junction of the words, another consideration is necessary. If the word No.8 comes after No.1, the transition pattern is

0 0 0 0 1 0 1 0 0 1 0 0
 No.1 No.8

, and there is a "1 0 1" pattern which violates above law. In this case, "1 0 1" pattern is changed into "0 1 0" pattern as follows.

0 0 0 0 0 1 0 0 0 1 0 0
 No.1 No.8

In order to allow this nonlinear junction, the last transition in Table 2 is always "0", and the coding is always proceeded watching the preceding and the following words.

Table 3 shows a comparison between 3PM and MFM. The packing density of 3PM can be 50% higher than that of MFM, but jitter margin is 50% worse and the maximum transition is 100% longer. Therefore, bit synchronization and the mechanisms of the player should be designed carefully.

Table 3 Comparison between 3PM and MFM

		MFM	3PM
Duration Between Transition	Minimum	$L(1/2\lambda_{min})$	$1.5L(1/2\lambda_{min})$
	Maximum	$2L(\lambda_{min})$	$6L(2\lambda_{min})$
Jitter Margin		$0.5L(1/4\lambda_{min})$	$0.5L(1/6\lambda_{min})$

L: length of data bit cell

λ_{min} : minimum wave length to be recorded

6. CONCLUSION

A digital audio disc system realizing the playing time of two and a half hours on one side of a 30cm-disc is described. The redundancy of the error correcting schemes is 64.2% including synchronizing bits, and transmitting bit rate is 3.568536 Mb/s for two channels of 16 bit signals. But on account of 3PM coding, the maximum recording frequency is 1.189512 MHz, and the minimum wave length at the inside of discs(120mm) is 2.4 μ m for 450 r.p.m. revolution.

These values are decided tentatively for experimental systems. For the systems to be commercialized, the redundancy and the minimum wave length might be able to be reduced on account of highly controlled processes, and the more playing time or smaller size of the disc will be possible.

Another great problem for commercialization is the compatibility of the players between digital audio and video disc systems. The system described in this paper has that compatibility, but consequently extremely long playing time is achieved, which might be unnecessary from software point of view. The final standard should be established by mutual studies of various software and hardware makers.

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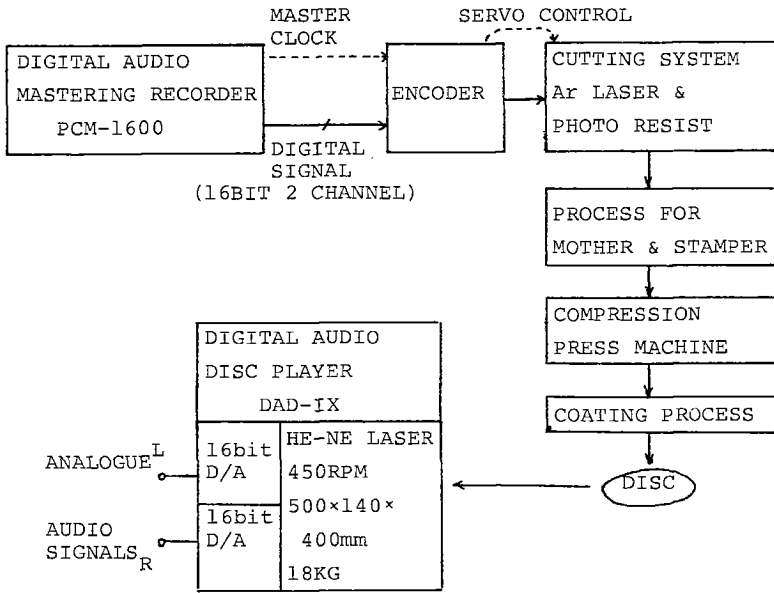


Fig 1. Digital Audio Disc Systems



Fig 2 Digital Audio Disc and the Player DAD-1X

Specifications

General

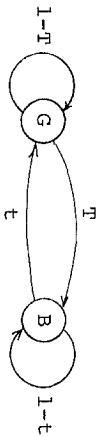
Playing time:	Two and a half hours (maximum)
Revolution:	450 rpm
Number of channels:	2 channels, digital
Sampling frequency:	44.056kHz
Quantization:	16-bit linear/channel
Dropout compensation:	Newly developed error correcting code
Dynamic range:	Better than 95dB
Total harmonic distortion:	Better than 0.03%
Frequency response:	2Hz ~ 20kHz±0.25dB

Player system

Signal detection:	Optical reflection type with a helium-neon laser
Wow and flutter:	Undetectable
Dimensions:	520(W) × 140(H) × 400mm(D)
Weight:	18 kg
Power consumption:	75W

Disc

Dimensions:	303mm (dia.)
Thickness:	1.1mm
Material:	Polyvinyl chloride, coated with reflection material



G: State of No Error

B: State of Error

$\left. \begin{matrix} T \\ t \end{matrix} \right\}$: Transition Probability

$r = \frac{t}{T+t}$: Bit Error Rate

$\rho = 1 - T - t$: Bit Error Correlation Coefficient

Fig 3 Gilbert Model

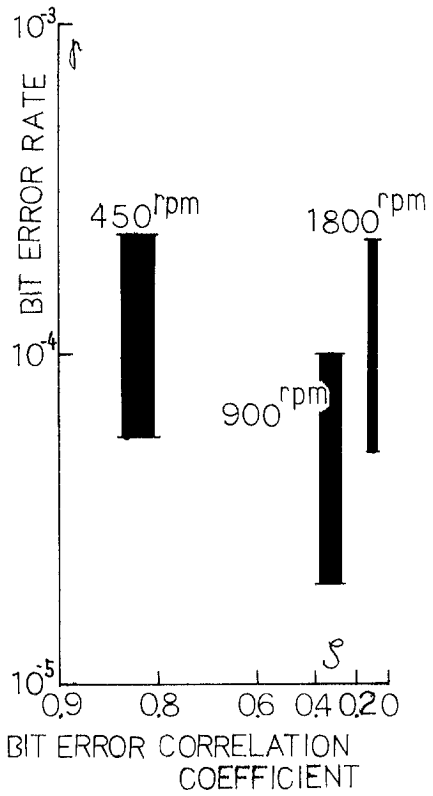
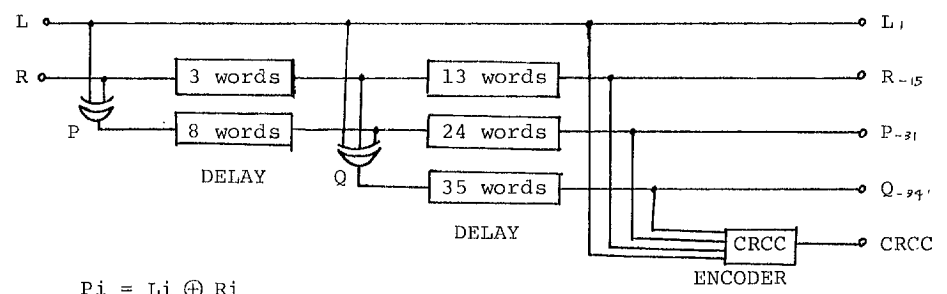
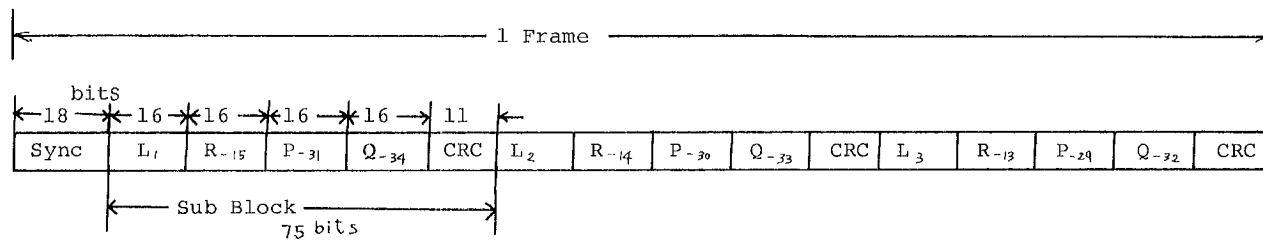


Fig 4 Code Error Analysis on δ - ρ plane



$$P_i = L_i \oplus R_i$$

$$Q_i = L_i \oplus R_{i-3} \oplus P_{i-8}$$

Fig 5 Format and Encoder

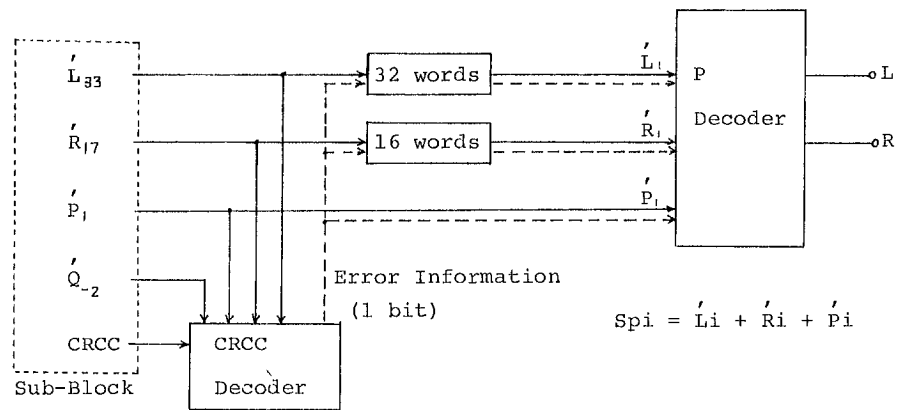


Fig 6 Basic Decoder (Ignoring Parity Q)

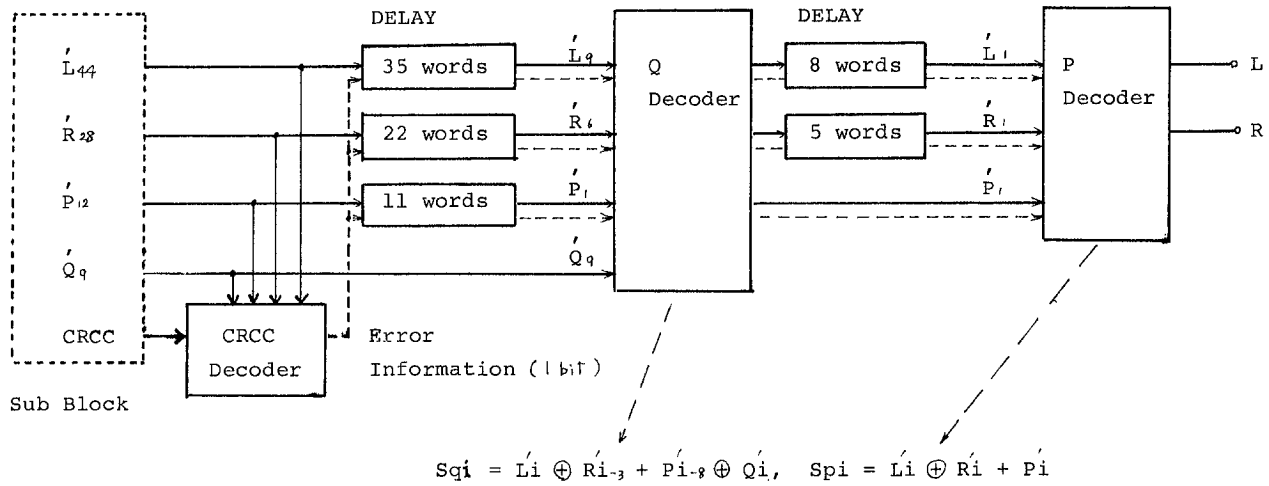


Fig 7 Double Decoder

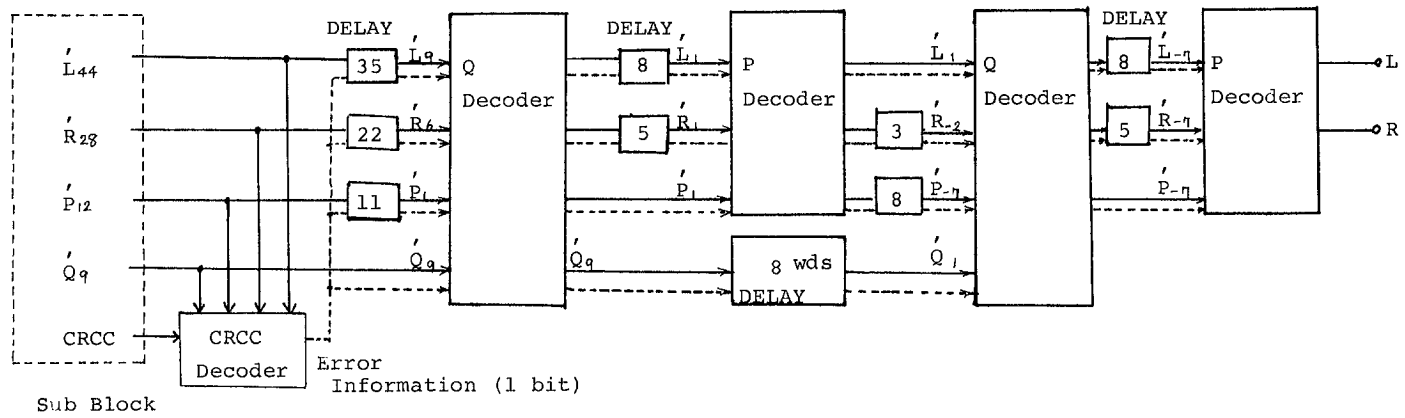


Fig 8 4-Times Decoder