A 24-CHANNEL STATIONARY-HEAD DIGITAL AUDIO RECORDER

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presented at the 61st Convention November 3-6, 1978 New York



AN AUDIO ENGINEERING SOCIETY PREPRINT

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A 24-CHANNEL STATIONARY HEAD DIGITAL AUDIO RECORDER

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Abstract

A 24-channel longitudinal digital audio recorder is reported here. The code format is especially developed considering the usability (punching in/out, synchronized recording, tape-cut editing, and so on) and the reliability (finger prints, dusts and scratches on the tapes). Results of computer simulation for the correctability of errors are also shown.

1. Introduction

The history of the practical digital audio recorders has started as an adapter of video tape recorders [1]. The digital technology has been proved to obtain a great improvement in performances of audio recorders. Nevertheless, the advent of true digital audio age is not expected until multi-channel digital audio recorders are realized.

Recently, several longitudinal digital audio recorders has been developed [2],[3]. The purpose of this paper is to show an example of the next generation of multi-channel digital audio recorders in full consideration of the following points.

- (1) All features and usability of analogue recorders, such as tape-cut editing, punching in/out, and so on, should be possible in digital recorders.
- (2) Strong protection from finger prints, dusts, and scratches on tapes should be devised.

In other words, the usability and the reliability of digital audio recorders should be equal to or better than those of analogue recorders. The problem (1) mainly lies in (A) the complete block of information words and (B) re-synchronization of bits and frames when a new stream of information with different phase comes in. The problem (2) depends on error correction and interpolation schemes, and it is necessary to make a special consideration on the fact that any noise must not be detected even when all of the tracks are destroyed for fairly long period.

Another very important point for digital recorders is that they will be used as a master for "a digital audio disc", which is supposed to have a palying time of more than one hour on one side. That means digital audio mastering systems should have playing time of at least 70 minutes.

The PCM-3200 digital audio mastering series are one of the examples solving above problems. Fig.l shows a sketch of PCM-3224, a 24 channel digital audio recorder.

2. Fundamental Specifications

(1) PCM-3200 series (Number of Channels)

| No. | Number of Channels | Tape Width |
|----------|--------------------|------------|
| PCM-3248 | 48 | 2 inches |
| PCM-3232 | 32 | 2 |
| PCM-3224 | 24 | 1 |
| PCM-3216 | 16 | 1 |
| PCM-3208 | 8 | 1/2 |
| PCM-3204 | 4 | 1/4 |
| PCM-3202 | 2 | 1/4 |
| | † | 1 |

In addition to the above digital audio channels, two analogue audio channels (to find editing point) and one SMPTE time code track are prepared for each.

- (2) Number of Tracks per Channel: 2 TRK/CH
- (3) Tape Speed: 22.5 IPS (Trying to reduce into 15 IPS)
- (4) Sampling Frequency: $f_1=44.056$ kHz $f_2=50.350$ kHz

 $\rm f_1$ coincides with the sampling frequency of rotary head systems (PCM-1, PCM-1600) [1], and digital frequency converter between $\rm f_1$ and $\rm f_2$ will be prepared.

- (5) Quantization: 16 bit linear/CH
- (6) Packing Density: 30,720 BPI

20,480 Flux Reverse/Inch

for 22.5 IPS

- (7) Modulation: 3PM
- (8) Error Correcting Schemes: Modified Cross Word Code
- (9) Redundancy: 41.7% (including synchronization bits)
- (10) Tape: V-16 (Sony)
- (11) Head: Single Crystal Ferrite Head.
- (12) Dynamic Range: more than 90dB (20Hz 20kHz)
- (13) Harmonic Distortion: better than 0.05% (20Hz 20kHz)
- (14) Crosstalk between Channels: better than 90dB
- (15) Frequency Range: 20Hz 20kHz (+0.5, -1.0 dB)

- (16) Wow/Flutter: Quatz Precision
- (17) Playing Time: 80 minutes (14" Reel) 40 minutes (10" Reel)

for 22.5 IPS

- (18) FF/REW Time: within 2 minutes (10" Reel)
- (19) Pitch Vernier: ± 12.5%
- (20) Editing, and Punch In/Out, Accuracy: 1.67m sec.

3. Code Format

Fig. 2 shows the format for one channel. Two tracks are separated by a certain distance to avoid error correlation between tracks.

The letter S shows synchronization signal of 25 bits. Each number shows 16 bit data word and its sequence. The letter P shows 17 bit check word which is not a mere parity, but full summation of two information words (odd and even) and its subscript expresses sequence. In other words, if the word #1 expresses a value of 1,234V and the word #2, 2,345V, then the check word becomes $P_1 = 1,234V + 2,345V = 3,579V$. The letter Q shows a 16 bit check word which works as an error detecting code for its previous line of information. For instance, Q_1 detects errors in the sequence of words 1, 3, P_{-279} , 5, ... 27, P_{-255} .

Each block starts with S and terminates by Q. The check words P's are delayed by ten blocks in each track.

4. Error Correcting/Interpolation Ability Table 1 shows error correcting and interpolation ability of this format for every pattern of block errors. In this table, 4 blocks $\sharp 1$, $\sharp 2$, $\sharp 21$ and $\sharp 22$ are shown, where the information words 1, 2, \cdots 28 are in the blocks $\sharp 1$ and $\sharp 2$, and their check words P_1 , P_3 , \cdots P_{27} are in the blocks of $\sharp 21$ and $\sharp 22$.

If there is not any errors in the blocks #1 and #2 (pattern No. 3, 4 and 5), all the information words in the blocks can be played back without any trouble. Of course, if we note the information words in the blocks of #21 and #22, the pattern No. 3, 4 and 5 are equivalent to No. 1, 2 and 10, respectively.

If one of two blocks #1 and #2 is bad, and #21 and #22 are without any errors (pattern No. 1, 2), error correction is carried out perfectly by the well-known erasure method. If one of the blocks #1 and #2 is bad and also some errors exist in #21 or #22 (No. 6, 7, 8, 9, 11 and 12), one word interpolation can be carried out. In that case, if both #21 and #22 are bad (No. 11, 12), the interpolation appears once in every two words, and if either #21 or #22 is good (No. 6 $^{\sim}$ 9), once in every four words. The latter case is because half of missed words are correctable by using check words P.

If both #1 and #2 are bad and both #21 and #22 are good (No. 10), one word interpolation can be done by using $P_{1/2}$, $P_{3/2}$, $\cdots P_{27/2}$. This is the merit of using the full added check word in stead of pure parity. If both #1 and #2, as well as either #21 or #22, are bad (No. 13, 14), one word in every four words is good and three words must be interpolated.

If all the blocks are bad, naturally, we have to give up any kind of error compensation method.

<Burst Error> Up to 3,840 bit (or 5.5 m sec.) burst error can be corrected. The burst error correctability depends simply on the amount of delay memory, and we adopted this value considering a trade-off between cost and ability.

<Random Error> This format is very strong for random errors,
because one word interpolation is possible, if 2/4 of blocks are
bad with using conventional erasure method. In addition to that,
cross word demodulation [4] is also possible with keeping full
compatibility to the erasure decoding. In this case, by adding
a certain amount of decoding circuits, the strength for random
error compensation can be increased $10^3 \sim 10^5$ times higher than
erasure decoding.

<Finger Prints, Dusts, and Scratches> This format is strong for
defects on tapes, because error compensation is possible even if
3/4 of blocks are bad, and even if both of the tracks are bad for
a long period.

Fig.3 shows results of computer simulation [4] of this format. The vertical axis represents occuring rate of errors exceeding correctability or the ability of interpolation of this format. While the horizontal axis is "bit error correlation coefficient ρ ", and hatched area is measured value of ρ in steady state. When finger prints or scratches stain the tape, the value of ρ increases.

5. Punch In/Out and Tape Cut Editing
In this format, all the information words are complete at the vertical line of the tape, for instance, at the end of blocks of #1 and #2. If the tape is cut, or new information is punched in, the sequence of old information is kept until the final living blocks. The only problem is the delayed check words P's, and thus error correcting is impossible at the final 5.5 m second of the old information. Nevertheless, with the aid of the strong error detecting words Q of 16 bits, one word interpolation can be carried out for one track error at that period, and practically there is not any trouble.

Synchronizing word consists of 25 bits for each block in order to achieve re-synchronization of the new information at the middle of old blocks. Consequently, punching in/out and tape cut editing are possible without adding inter-block gap, and this enables to save redundancy several percent.

Conclusion

o. Conclusion
So far, we have described our newly developed stationary head
digital audio recorder systems. One of the big problems of formatting is the number of tracks to be used to record one channel
of audio signals. The smaller number is the better for economical
reasons. But if one track per channel is adopted, a long interleaving must be taken in order to obtain sufficient correctability
for burst errors [2]. In that case, if tape-cut editing or punching-in is carried out, a huge number of words will be crippled by
losing even or odd half, thus the usability might be limited comlosing even or odd half, thus the usability might be limitted comparing to analogue recorders. Therefore, the minimum number of tracks per channel for general-use digital audio recorders is rather considered to be two.

There is another problem for a complete block of error correcting schemes of two tracks. The size of the block can not be too large because of the requirement of editing accuracy, and the burst error correctability is limited by the size of the block. In other words, a two track format, with both sufficient editing accuracy and sufficient correctability, is almost impossible by a complete block.

The code format presented by this paper, a complete information block and delayed parity words, is one of the solutions for above two problems.

Another distinguished point of this format is that the parity is not a mere exclusive-or but a full add of the information values, thus the interpolation is possible even when all the information words are destroyed.

Tape speed is temporarily decided as 22.5 IPS by using 3PM coding, but engineering effort will be tried to reduce it into 15 IPS, in the future. A series of digital audio recorders, from 48 to 2 channels with the same code format, is being planning.

In closing this article, the authors would like to thank Dr. H. Nakajima, managing director of SONY Co. Ltd. for his supervision and all the stuffs of the PCM project team for their cooperation in development.

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Fig 1 PCM-3224

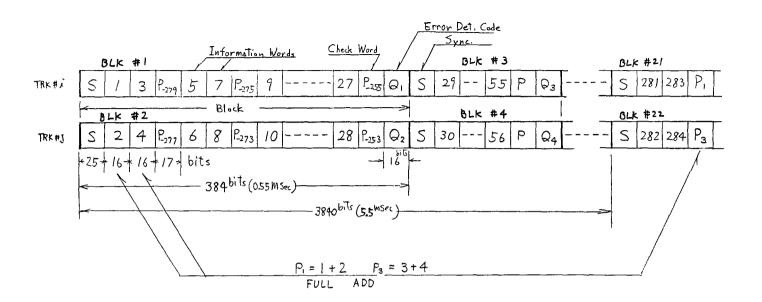


Fig 2 Code Format for 1 Channel (Redundancy 41.7%)

| • | | | | | | |
|----------------|--------------------------|----|----------------|-----|----------------|-------------------|
| pattern No. | No. of Error Block | #1 | #2 | #21 | #22 | |
| | | | | | | |
| 1 | 1. | × | 0 | О | 0 | |
| 2 | | 0 | × | o | 0 | |
| 3 | | 0 | o | × | 0 | Correction |
| 4 | | 0 | О | o | × | |
| 5 | 2 | | _o - | x | _x - | |
| 6 | | × | 0 | × | 0 |) |
| 7 | | × | o | o | × | 1/4 |
| 8 | | 0 | × | × | 0 | l-word |
| 9 | | 0 | × | o | × | Interpolation |
| 10 | | × | × | 0 | 0 |) |
| 11 | 3 | x- | o- | - × | x_ | 1/2 |
| 12 | | 0 | × | × | × | J |
| 13 · | | × | × | 0 | × | 3-Word |
| 14 | | × | × | × | 0 | 3/4 Interpolation |
| 15 | | × | × | × | × | Give Up |
| 4 | 4 | | | | | 40 9710 |

Table 1. Error Correcting and Interpolation Ability for Each Track Error Pattern

x: error

o: no error

Fig 3. Results of Computer Simulation
Bit Error Rate = 10-4

