AN EFFECTIVE RAM ACCESSING METHOD
FOR PCM PROCESSING SYSTEMS

1726 (D-4)

Shin-ichi Ohashi, Takao Arai
Hitachi Ltd., Japan

Masahira Fujimoto, Harukuni Kobari
Victor Co. of Japan Ltd., Japan

presented at the
68th Convention
March 17. – March 20. 1981
Hamburg

AES

AN AUDIO ENGINEERING SOCIETY PREPRINT

This preprint has been reproduced from the author's advance manuscript, without editing, corrections or consideration by the Review Board. For this reason there may be changes should this paper be published in the Audio Engineering Society Journal. Additional preprints may be obtained by sending request and remittance to the Audio Engineering Society, Room 449, 60 East 42nd Street, New York, N.Y. 10017.

An Effective RAM Accessing Method for PCM Processing Systems

Shin-ichi Ohashi*, Takao Arai*,
Masahira Fujimoto**, Harukuni Kobari**

* Consumers Product Research Center,
Hitachi Ltd.

** Audio Engineering Research Center,
Victor Co. of Japan, Ltd.

ABSTRACT

An effective method to control RAM in PCM digital audio processer using VCR has been studied. The main functions of the RAM is to interleave/de-interleave datas during recording and play back respectively, and to absorb time base error of data caused by jitter of VCR. Newly developed RAM control method (called corn type) can reduce the necessary RAM size to a half of that of conventional method.

1. INTRODUCTION

Instead of conventional analog type audio systems, the digital audio techniques are remarkably progressing. In 1979, a technical file for PCM encoder/decoder used with VCR was adapted by EIAJ (Electronic Industries Association of Japan) and it's commertial products are apearing in the market.

By adapting PCM system, the fundamental performance of audio product, such as S/N, distortion, flatness of frequency response, won and flutter etc, are improved, but on the contrast, the complexity of circuit systems will remarkably increase.

In this paper, a new concept to minimize RAM capacity in PCM systems is described.

The major function of PCM digital audio processing are to add correcting words to digital audio signals, then to interleave them.

Finally they are recorded to VCR after adding error detecting and synchronizing signals. For play back mode the above functions work exactly in reverse.

RAM is used for interleaving/de-interleaving and also for absorbing time base error of VCR.

In this paper, we analyze the condition of minimum RAM size and then propose a new RAM accessing method which enable the RAM size to reduce down to ½ of conventional method.

The new method can be effectively applied not only to PCM digital audio processor using VCR, but also to PCM disc player's and general digital data interleaving/de-interleaving processors.

2. PCM DIGITAL AUDIO PROCESSING SYSTEM USING VCR.

The block diagram and the major specification of PCM digital audio processor are shown in Fig. 1, and Table 1. Left and Right audio input signals are converted to digital data by S/H (Sampled and Hold) circuits and A/D (Analog to Digital) converter. The data with error correcting words P and Q are stored in RAM. The stored data in the RAM are read out following the interleaving seqences defined by Technical File of EIAJ. Finally they are recorded on VCR with error detecting words (CRCC), horizontal synchronizing signals and vertical synchronizing signals.

For playback mode the above functions work exactly in reverse.

3. CONVENTIONAL RAM CONTROL METHOD FOR INTERLEAVING/DE-INTERLEAVING.

The conversion processes from analog to PCM signal is illustrated in Fig. 2.

Left and right audio signals are sampled alternatively at 44.056 KHz rate. .... (a)

Digitally converted datas are stored in RAM. .... (b)

The stored datas are read out sequentially at 3D intervals to make interleaved data.

At the sametime, the two error correcting words P and Q are made from the data. .... (c)

The purpose of interleaving is to record data dispersedly on cassette tape to inprove error correcting capability against burst drop out data.

Finally, synchronizing signals are added to data block to form EIAJ signal format which is reffered to video signal format.

Fig. 3 illustrates the concept of interleaving methode.

Fig. 4 shows RAM map of interleaved data.

Input data of three pair of L and R channel and corresponding error correcting words P and Q are stored from lowest row to upwards successively. The interleaved data are made, as shown in Fig. 4 , from the data located on the dotted line. Similarly, next interleaving is done from upper adjacent data from L4, and so on. The data in the lowest row is erased after read out since they are no more necessary for interleaving. The minimum RAM size for interleaving is calculated as Eq. (1).

$$\text{(Data in a row)} \times \frac{21}{3} D = 12.6 \text{ K bits} \qquad (1)$$

; Datas in a row = 14 bit X 8 datas = 112 bit

; D = 16

4. NEW RAM ACCESSING METHOD FOR INTERLEAVING/DE-INTERLEAVING

Since the shaded area in Fig. 4 is used already for interleaving and unnecessary to keep data. We developed a new RAM accessing method to save the shaded area in Fig. 4 and achieved to reduce the RAM size down to ½ of that of conventional method.

Fig. 5 illustrate the principle of new method. Each circles carrespond to the data area of L, R and P, Q in RAM. For instance, the circle at right side correspond to the data area of Q and on which 112 data are stored as Qn, Qn-3, Qn-6 .... Qn-21D successively.

When Qn is stored in position 1, Qn-21D in position 2 is read out.

On the next step, a new data Qn + 3 is in position 2 and adjacent data Qn - 21D + 3 is read out from position 3.

Thus, the process of writing new data and reading out of delayed data are done by shifting data positions in counter-clockwise successively.

The accessing process of other 7 group of data (from Pn - 18D to Ln), are similar.

Since the length of interleaving become shorter by 3D interval, the corresponding RAM areas become smaller and finally the RAM area of Ln is only 1 word length.

Fig. 4 and 5 showed the process of recording,the de-interleaving process for play back is simillar but just reverse.

Fig. 6 illustrate comparison of our new method and the conventional method.

Each contours correspond to data areas of each L, R and P, Q.

The side surface areas are proportional to total RAM size. Obviously, the new method forms a corn type and its side surface area is ½ of that of conventional cylinder type method.

5. THE EFFECTIVE METHOD OF TIME BASE ERROR ABSORPTION

In playback mode, the additional RAM area to absorb time base errors such as by jitter of VCR is necessary.

Conventionally, the RAM area to absorb jitter is equally allocated to all the data, but the area for error correcting words, P and Q, are not necessary because the time base error correction is required to L and R channel data only.

Fig. 7 illustrate the necessary RAM area for jitter absorption (jitter margin) and for de-interleaving.

The jitter margin for P and Q can be eliminated by the method described below.

The de-interleaved data L and R are corrected by error correcting words P and Q if some of data are erroneous.

This correcting process is done immediately after reading out the de-interleaved data from RAM, and corrected L, R data are written in to RAM again. Then data P and Q become unnecessary to be stored.

As a result, the RAM area for jitter margin is reduced to 3/4 of that of conventional method as shown in Fig. 7 (b).

6. CONCLUSION

We developed an effective RAM accessing method used in PCM digital audio processor which comply the signal format defined by Technical File of EIAJ.

This method can reduces the RAM size for interleaving/de-interleaving of data to ½ of that of conventional method.

The method can be applied to general interleaving/de-interleaving signal processing systems also.

A new data accessing method can also reduce the additional RAM area, necessary to absorb time base errors caused by jitter of VCR, to 3/4 of that of conventional method.

Table 1. Specification of PCM digital audio used with VCR

No. of Audio Channels	2
Sampling Frequency	44.056 kHz
No. of Coding Bits	14-bit linear coding
Sync. Signal	Standard NTSC
Data Error Recovery	16-bit CRCC/H detection and correction
Reproduction Frequency Range	DC to 20,000 Hz
Dynamic Range	More than 85 dB
Harmonic Distortion	Less than 0.03% (at any recording level)
Wow & Flutter	Below measurable limit

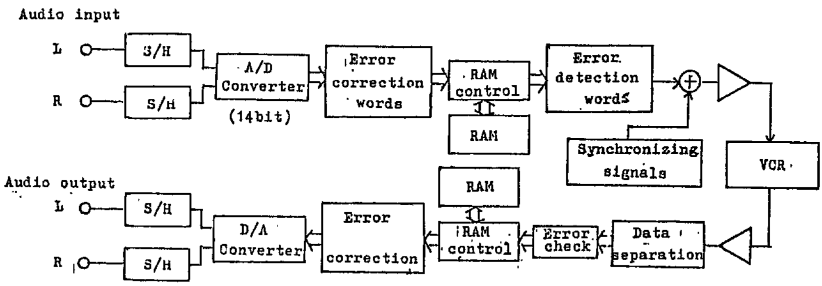
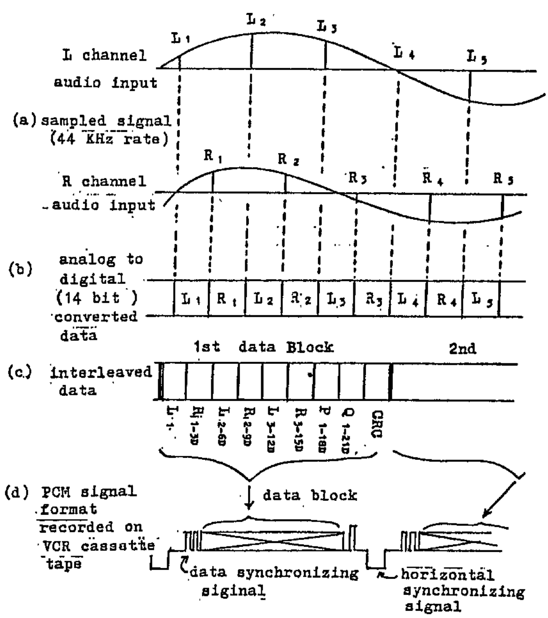


Fig.1. Block diagram of PCM digital audio processor using VCR.



Fig,2 The conversion process from analog to digital signal

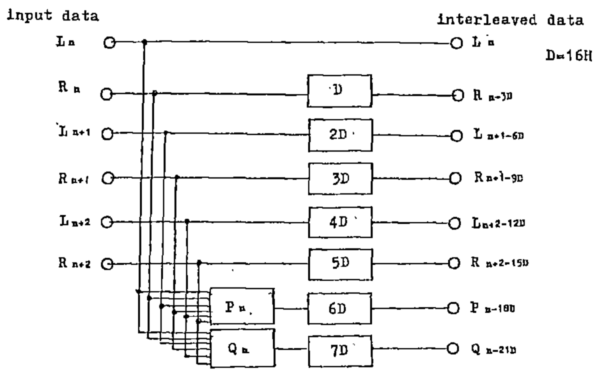


Fig.3. Concept of interleaving data

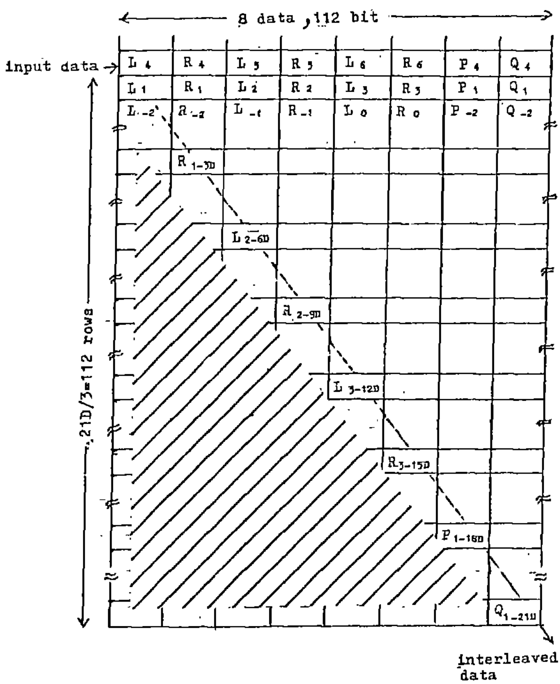
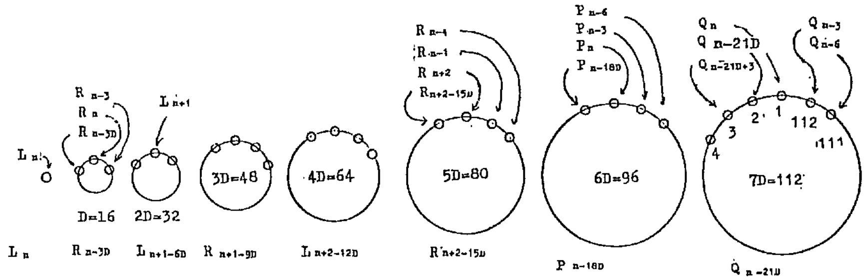
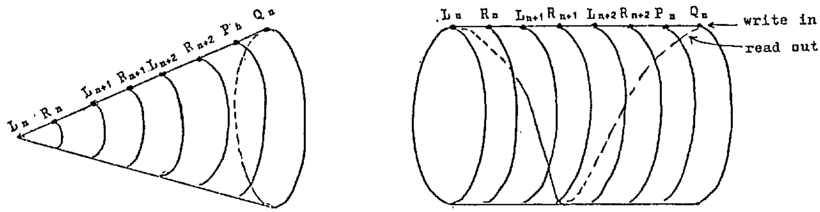


Fig.4. RAM map for interleaving



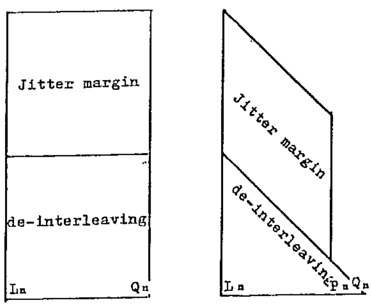
Fig,5. New RAM accessing method to interleave datas (Recording process)



(a) New corn type

(b) Conventional cylinder type

Fig,6. Comparison of RAM size (Recoring process)



(a) conventional (Cylinder type)

(b) new method (Corn type)

Fig,7. Comparison of RAM size required to de-interleave. data and absorb time base error of VCR at play back.