PCM Recorder
A New Type of Audio Magnetic Tape Recorder

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The PCM recorder is a new type of audio magnetic tape recorder that records audio signals in the form of pulse codes. The composition and performance of the PCM recorder, are explained in comparison with conventional magnetic tape recorders. The PCM recorder owes its superiority in performance compared to conventional audio recorders to the adoption of a digital recording system. Quality of reproduction satisfies the most rigid standards for recording of music signals.

INTRODUCTION: Rapid progress has been made in signal-recording technology since the invention of the magnetic tape recorder. The performance of such recorders has satisfied many requirements in various fields of application. However, in certain fields, such as music signal recording requiring very high fidelity and acoustical analysis requiring precise measurements, there has been a need for increased improvement in the quality of reproduced signals. These requirements include flat amplitude and phase characteristics in a wide frequency band, an excellent signal-to-noise ratio (SNR), a very wide dynamic range, a very low distortion factor and inter- or crossmodulation distortion factor, and excellent transient response characteristics. At present it is difficult to obtain satisfactory results on these points, even with the highest grade professional audio tape recorder available. Of course, efforts are being made to improve the performance, but the present performance is close to the theoretical limits of the magnetic recording system. Hence the development of a new recording system, which is different from the conventional one, is believed to be necessary in order to break through the theoretical barrier. One of the conceivable methods of improvement would be to employ a recording medium basically different from the magnetic medium, and another method would be to use the conventional magnetic recording medium in combination with some signal modulation method which is not affected by the limitations of the conventional magnetic recording medium. The PCM recorder is of the latter type. In recent years pulse-code modulation (PCM) has attracted attention for its excellent transmission characteristics and has been used in many communication systems. In PCM systems, information is generally transmitted in the form of binary pulses whose presence or absence in each time slot indicate each bit of information. Hence, sensing the presence or absence of a pulse in each time slot is required at the receiving end of the path through which the PCM signals are transmitted. The main characteristics required are as follows:

1) The transmission path must not introduce a timing jitter large enough to cause time slot misjudgment.

2) The system must have a frequency band wide enough to avoid mutual pulse interference.

3) Noise interfering with pulse sensing must be avoided.

If the above conditions are met, the PCM system is capable of perfect reception of the transmitted information. Desirable characteristics for the binary PCM transmission of two channels of audio signals in the 0–16-kHz band are a timing jitter below 80 ns, a frequency band between 0 and 1.28 MHz, and an SNR above 6 dB. No conventional audio recorder can meet all of these requirements. Conventional audio magnetic recorders have a more than adequate SNR, but the frequency band is too narrow and the timing jitter too large. However, the above problems can be overcome if a helical-scan two-head type video tape recorder (VTR) is used as a recorder with wide frequency band, and if a special synchronization system is used in combination with a buffer memory to remove the timing jitter. A recording system using digital signals is available in the market as a data recorder, and in the field of acoustic analysis, a high-tape-speed type is used for PCM recording of audio signals in narrow band. But even such a data recorder is unable to record music signals, which have a frequency band as broad as 20 kHz, for prolonged periods. The reason is that such a data recorder has time limits in recording because of the high-speed tape drive, and the frequency band is inadequate in spite of the high-speed properties.

COMPOSITION OF THE RECORDER

The PCM recorder, as shown in Fig. 1, is composed of an analog-to-digital (A/D) converter, a magnetic recorder, a buffer memory, and a digital-to-analog (D/A) converter. The following is a description of the operation of the recording system. The audio signals of two channels which have come into the A/D converter pass through a low-pass filter (LPF) to restrict the band to 0–16 kHz. Then time-division multiplexing is conducted at the multiplexing gate to form pulse-amplitude modulation (PAM) signals. The PAM signals are converted into natural binary codes of 13 bits by an encoder to form PCM signals. The PCM signals obtained are rearranged in a pulse train in a form suitable for magnetic recording and fed to the magnetic recorder. In the magnetic recorder, the pulse train is modulated into an FM signal with a small modulation index by the frequency modulator. The FM carrier is then recorded on magnetic tape using the two helical-scan magnetic heads alternately. The FM signals reproduced from the magnetic tape by alternating use of two magnetic heads are formed into a continuous FM signal by a changeover switch for the heads. This continuous FM signal is demodulated into a pulse train with an FM demodulator and fed to the buffer memory. When the pulse train passes through the buffer memory, the timing jitter produced in the magnetic recording system is removed, and the jitter-free pulse train is fed to the D/A converter as a regular PCM signal. The PCM signal that reaches the D/A converter is decoded into a PAM signal by the decoder. The PAM signal passes through a demultiplexing gate and is demultiplexed into two channels. Each of the demultiplexed signals passes through a low-pass interpolating filter, is interpolated, and is reconstituted as the original audio signal. In the processes of recording and playback, the magnetic-tape drive system is controlled by a servo mechanism using a crystal oscillator as a standard.

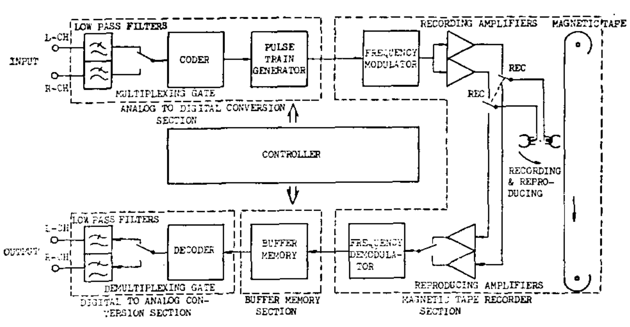

Fig. 1. Construction of PCM recorder.

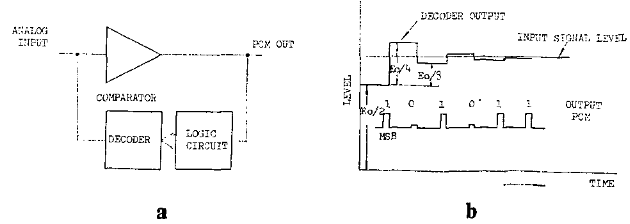

Fig. 2. Construction and encoding process of sequential comparison feedback encoder. **a**. Construction. **b**. Encoding process: E_0—maximum encoding level.

ENCODER

The system using PCM features sampling frequency, code length, and type of code. The present PCM recorder is designed to record and reproduce stereophonic music signals with very high fidelity. Music signals are said to have frequency bands ranging from 20 to 20 000 Hz and a dynamic range of 80 dB. In order to pass a music signal through the PCM system and then reproduce it with very high fidelity, a code length of about 13.5 bits and a sampling frequency of at least 40 kHz are required. The PCM recorder uses a sampling frequency of 40 kHz and a code length of 13 bits to handle music signals prescribed as 0–16 kHz in frequency band and 75 dB in dynamic range. In encoding the signal, no processing (such as special emphasis on the frequency characteristics, or compression and expansion of levels) is employed, so that the naturalness of the recorded sound will be retained. Music signals are scarcely affected by a slow change of level, but they are greatly affected by a rapid change of level or by pulse noise. Therefore, the encoder used for music signals must be of an error-free type. The present PCM recorder utilizes a sequential-comparison feedback encoder. As shown in Fig. 2**a**, the encoder is composed of a comparator, a logic circuit, and a partial decoder.

The system of encoding is described below. First the output of the decoder is set at half the maximum level E_0 and is compared with the input signal. The code 1 is given if the input signal level is higher, or if lower, code 0 is given. Thus the most significant bit (MSB) is decided upon. Then the MSB code is sensed by the logic circuit; if the MSB is 1, the output in the decoder

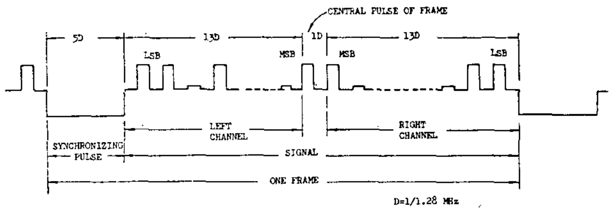

Fig. 3. Composition of PCM pulse train.

is set at $(\frac{1}{2} + \frac{1}{4})E_0$; if the MSB is 0, the output is set at $\frac{1}{4}\ E_0$. By making a comparison between the output level in the decoder and the input signal, the second bit is decided upon. The decision as to the nth bit is made by setting the output in the decoder at

$$\left(\sum_{j=1}^{n-1} \delta_j 2^{-j} + 2^{-n}\right) E_0, \qquad \delta_j = \begin{cases} 1 & j\text{th bit is } 1 \\ 0 & j\text{th bit is } 0 \end{cases}$$

and comparing the output level with the input signal. In this way, the decision must be made bit by bit until the least significant bit (LSB) is obtained. Fig. 2**b** shows the states of change in the decoder output in the process of encoding. Gradually coming closer to the input signal, the decoder output finally accords with the input signal with an accuracy within half the LSB level, as shown in Fig. 2**b**. This means that if the same decoder employed in the process of encoding is used to decode PCM codes, a signal agreeing with the input signal reappears with an accuracy within half the LSB level, whatever the form of code may be. Therefore, such a decoder is free from any errors which would produce a pulse noise harmful to music signals.

COMPOSITION OF PULSE TRAIN

The PCM signal from the encoder is arranged in a pulse train, as shown in Fig. 3, before it is recorded. In the pulse train, narrow pulses in the plus direction are code pulses, and wide pulses in the minus direction are synchronizing pulses. The pulse train between the leading edge of each synchronizing pulse and that of the succeeding synchronizing pulse is called one frame. In the frame are serially arranged a synchronizing pulse, a check pulse, and 13 bits of code pulses, each, for the samples of the right and left channels of the stereophonic signal. The 13 bits of code pulses are called one word. One bit of check pulse is in the center of the code pulse train, with the right and left channel words preceding and following the check pulse. In each word the MSB comes nearest to the center, as shown in Fig. 3. During reproduction the synchronizing pulse is used as the time standard for synchronization of the clock signal and, at the same time, is used as the junction time slot

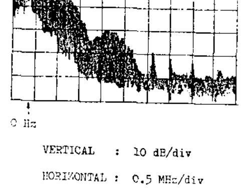

Fig. 4. Power spectrum of PCM pulse train.

when the output signals from the two magnetic heads are connected to obtain a consecutive signal. The pulse repetition frequency in the pulse train thus composed is 1.28 MHz.

RECORDER

The PCM system has superior transmission characteristics to other modulation systems; however, it requires a very broad transmission frequency band. In the present PCM recorder, stereophonic signals of two 16-kHz channels are transmitted as a train of pulses having a pulse repetition frequency of 1.28 MHz. The power spectrum of the pulse train is shown in Fig. 4. Therefore, instead of a conventional audio tape recorder, a video tape recorder (VTR) capable of recording a broadband signal must be used to record the PCM signal. The present PCM recorder uses a helical-scan type two-head VTR. This VTR employs a magnetic tape, 1 inch in width. The tape speed is 19 cm/s. The relative speed of heads to the tape is 16 m/s.

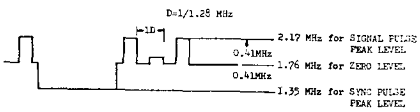

Fig. 5. Frequency allocation of FM signal.

The FM system used for recording video signals is also suitable for the PCM recorder. The use of frequency modulation removes most of the amplitude fluctuations of the signal in playback. But because of the small modulation index used, the SNR is only about 35 dB. However, this is more than adequate for PCM transmission. The frequency allocation of FM signals used in this recorder is shown in Fig. 5. PCM signals have information in the time slot. Therefore it is very important to reduce the timing jitter of the reproduced signals. For this purpose, the tape driving mechanism must be servo controlled. In the present PCM recorder the timing jitter is controlled within $\pm 50\ \mu s$. A two-head type VTR causes discontinuity of timing when the output is changed over from two heads, but this discontinuity is controlled within $\pm 2.5\ \mu s$.

SYNCHRONISM

In decoding PCM signals it is necessary to synchronize the clock of the decoder with the time slot of PCM signals. A synchronizing method utilizing the statistical nature of signals is used in many PCM systems. With the present recorder system, however, the reproduced signals have a considerable degree of timing jitter. Therefore an established synchronizing method was adopted. The frequency spectrum of timing jitter contained in the reproduced signals shows a tendency for the jitter to become smaller with higher frequencies. Accordingly, if the reproduced signals are divided into short sections, the amount of timing jitter in each section will be insignificantly small. By paying attention to this point, a method of synchronizing at each small section of signals was adopted in the present PCM recorder. More particularly, this method, as shown in Fig. 3, consists of inserting synchronous signal pulses of the same minus direction as used in TV signals into the code train at

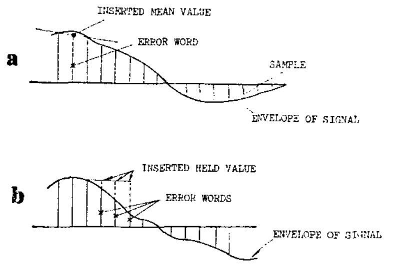

Fig. 6. Error correction. **a**. One-word error. **b**. Sequential error.

the required intervals in advance, detecting these pulses in reproduction, and using the detected pulses as the time standard to achieve synchronization. A timing generator used in the decoder consists of a pulsed oscillator which starts oscillation at the trailing edge of synchronizing pulses and stops at the leading edge of the following synchronizing pulses. This generator produces jitter-free clock signals with the same frequency as the pulse repetition frequency of PCM signals, and in a period as short as one frame it is capable of remaining in phase with the time slots of the PCM signals. Hence by using such an oscillator, synchronization is achieved within each frame of PCM signals.

BUFFER MEMORY

The above method achieves acceptable synchronization within each frame, but the frame intervals may be accompanied by jitter. This is due to low-frequency jitter components contained in the reproduced signal. In order to decode PCM signals satisfactorily, this jitter must be removed. A buffer memory is used for removal. The reproduced PCM signal is stored in the memory in a unit of one frame, using the output of the pulsed oscillator as a clock, and the stored signal is read out by the jitter-free clock. By so doing, the jitter contained in the reproduced signal can be removed. If the jitter were not removed, overlapping in the signal would result. Since the jitter, which is contained in the signal reproduced by a servo-system-controlled VTR, is objectionable, a buffer memory with a capacity twice as high as the maximum value of the amount of jitter can remove all the jitter completely without deterioration or overlapping of the signal. In the present PCM recorder, a memory with a capacity of about 300 μs is used for correcting jitters of ± 50 μs. This capacity corresponds to ten frames or 320 bits of PCM signal. This has been accomplished by ten shift registers of 32-bit length.

DECODER

The PCM signals from the buffer memory are decoded into PAM signals by a decoder. The decoder used is the same as the one used in the encoder. A decoder combining a ladder resistance network, a constant-current source, and a switching circuit was used in the present PCM recorder.

ERROR CORRECTION

Usually many dropout phenomena are seen in signals reproduced from magnetic tapes. The dropout is caused by fine dust on the magnetic tape, by peeling of the magnetic coating, or by unevenness in coating. Conventional audio recorders are not significantly affected by the dropout pheomenon, but a PCM recorder will be seriously affected because the recording area for each bit is very small, and a very small dropout causes a code error. This results in a clicking noise like the sound of a scratched phonograph disc. In order to reduce such noises, the present PCM recorder uses error detecting and correcting techniques.

For the detection of code errors, two methods are used concurrently. One method comprises judging the presence or absence of the check pulse in each frame to detect the disorder of signal in the frame. The other method involves watching the level of the reproduced FM signal to detect whether or not the dropout causes lowering of level. Although these methods are indirect, most errors can be successfully detected. In case the detected errors are confined to one frame only, they are corrected by replacing the signal in the erroneous frame by the algebraic mean value of signals in the frames preceding and following that frame. In case of continuous errors over several frames, a signal in the last frame of the signal before the errors occur is substituted for these erroneous frames.

Fig. 6 shows the methods for correcting errors. The error correction actually is made at a stage where the signal is still in the PCM form, even though Fig. 6 shows the error correction in an equivalent PAM signal because the PAM signal is easier to understand. Moreover, the detection and correction of errors are carried out in the unit of a single frame. In other words, they are carried out simultaneously in the signals of right and left channels. This is done because errors are made in both channels in many instances. By correcting the errors, nearly all clicking noises can be eliminated.

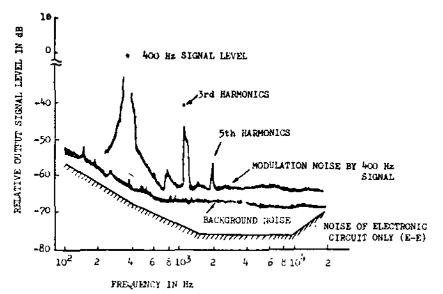

Fig. 7. Example of power spectrum of reproduced signal obtained by a conventional audio tape recorder. Tape speed 38.1 cm/s; spectrum analyzer bandwidth 100 Hz.

SIGNAL-TO-NOISE RATIO

Conventional magnetic recorders for audio signals often produce background noise, modulation noise, crosstalk, or harmonic distortion, so that the SNR in the reproduced signal is not always satisfactory. Fig. 7 shows the spectrum of a signal reproduced from a conventional audio magnetic recorder. There are three spectra, the noise spectrum of the electronic system (*E-E* system) alone (apart from magnetic heads or magnetic tape), the noise spectrum produced by the running of blank magnetic tape, and the reproduction spectrum of a 400-Hz sine wave recorded at a level causing 1 percent harmonic distortion. In the case shown in Fig. 7, the SNR is less than 35 dB, relative to the modulation noise. Moreover, the dynamic range relative to the background level is 50 dB at most.

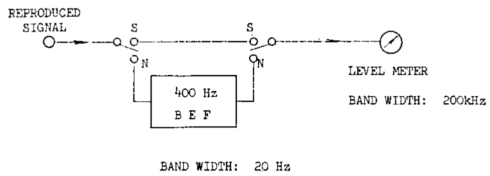

Fig. 8. SNR measurement system. Level of reproduced signal is 1-percent harmonic level.

The SNR is measured using a level meter and a band elimination filter, as shown in Fig. 8. At first the level of the reproduced signal, which contains a 400-Hz sinusoidal wave with 1 percent harmonics and modulation noise, is measured using a level meter with 200-kHz bandwidth. This level indicates the signal S. Then the 400-Hz component in the reproduced signal is eliminated using a 400-Hz band elimination filter with 20-Hz bandwidth, and the remaining level is measured using the same level meter. This indicates the noise level N. The SNR is the ratio between the signal level and the noise level. The dynamic range is defined as the ratio between above signal level and the background noise level. The background noise level is also measured using the same level meter.

On the other hand, PCM recording is never accompanied by noise related to magnetic recording characteristics, such as modulation noise, as shown in Fig. 7. This is because the PCM signal is composed of a binary pulse train, and decoding the signals involves only the presence or absence of the pulses. Most of the noise in the PCM recorder is generated when analog signals are converted into digital codes. This noise is called quantization noise and is given by the following equation:

$$N_Q = \frac{1}{12}\left(\frac{2}{K}\right)^2 \int_{-1}^{+1} \frac{1}{g'(y)} P(y)\, dy$$

where

N_Q quantizing noise power
K number of quantization steps
g compressing–expanding function
$P(y)$ probability density function of signal level.

Thus the magnitude of the quantization noise N_Q is related to the number of quantization steps, that is, the code length.

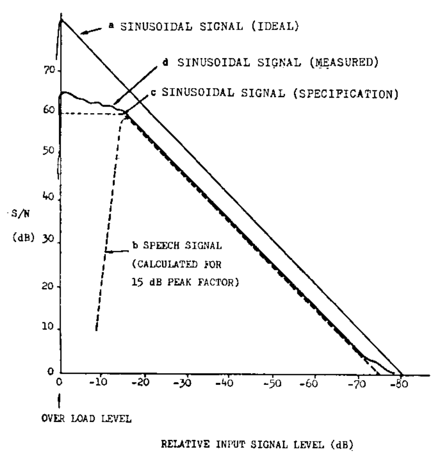

Fig. 9. SNR of 13-bit encoding.

Fig. 9, curve a, shows the ratio of signal to quantization noise in the case of a 13-bit linear code. As can be seen, a recorder with a dynamic range of about 80 dB can be realized theoretically in PCM recording using a 13-bit code.

The maximum input signal level in a conventional audio magnetic recorder is set with reference to the magnitude of the harmonic distortion in the reproduced signal. This harmonic distortion is derived from the characteristics of the magnetic tape. When the input signal is at a higher level than the stipulated level, greater harmonic distortion is caused, but the rate of increase is comparatively slow. On the other hand, the signal level in the case of PCM is expressed in a pulse code, and the number of transmissible signal levels is decided upon by the number of bits. For this reason the maximum input signal level is determined by the limit of code length. For example, when a 13-bit straight code is used, the number of transmissible levels is $2^{13} = 8192$. If the size of one step is 1 mV, the maximum transmissible level is 8.192 V. In the case of usual PCM systems, all the input signals exceeding the maximum level are encoded the same as the maximum level. Therefore, levels higher than the maximum level are all clipped at the maximum level, whereby great harmonic distortion is produced. This distortion is called overload distortion noise. Fig. 9, curve a, shows that the SNR curve at levels higher than the overload level falls rapidly. Fig. 9, curve b, shows the SN_Q characteristics against a mean input level in the case of audio signals with a peak factor of 15 dB encoded into 13-bit codes.

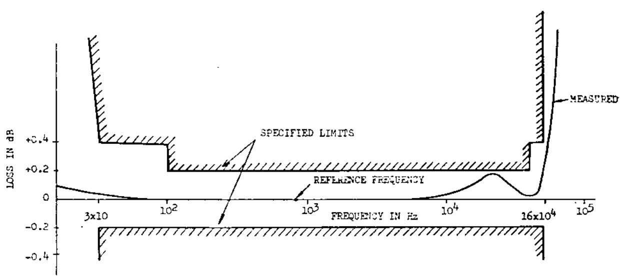

Fig. 10. Frequency response of PCM recorder.

The deterioration of the SN_Q at high levels is caused by the clipping of overload levels. In PCM recording, it is important to keep a proper recording level in order to prevent overload distortion noise, taking into account the peak factor of the recording signal.

Fig. 9, curve c, shows the SNR standard for the present PCM recorder. The reason for using a margin about 5 dB higher than the theoretical value is to take into account the deterioration of the SNR caused by imperfection of the encoder as well as other factors. The measured SNR is shown in Fig. 9, curve d. Even in the case of PCM recording, crosstalk exists. The crosstalk is produced when the signals of two channels are subjected to time-division multiplexing while in the PAM stage. The amount of crosstalk is determined by the bandwidth of the electronic circuit. The crosstalk between right and left channels in the present PCM recorder is −55 dB. If further reduction of crosstalk is desired, instead of time-division multiplexing at the PAM stage, the signals should be encoded with two encoders and

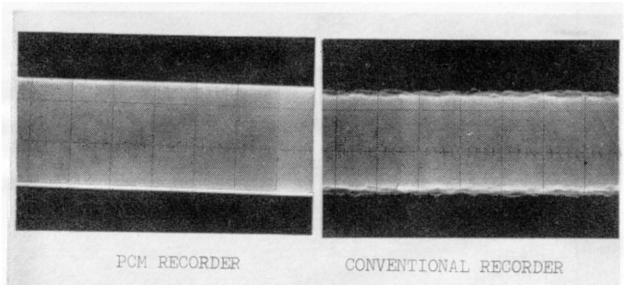

Fig. 11. Level fluctuations of reproduced signal. 10-kHz sine wave; horizontal scale 5 ms/div; vertical scale 0.5 V/div.

then subjected to time-division multiplexing. This method eliminates crosstalk completely.

FREQUENCY CHARACTERISTICS

The frequency characteristics of conventional audio magnetic recorders depend chiefly upon the magnetic tape system and the equalizing circuit. The characteristics of the PCM system have no relation to those of the magnetic tape system. They are determined mainly by the characteristics of the two LPFs. One LPF is used for restricting the bandwidth of input signals at the A/D converter, and the other is used for interpolating PAM signals at the D/A converter. Therefore, in PCM recording very flat characteristics can be obtained as shown in Fig. 10. With conventional audio recorders, the reproduced signals are accompanied by considerable level fluctuation. This is due to the lack of uniformity in the magnetic material and the uneven contact between the magnetic head and the magnetic material. In contrast, PCM recording does not cause such level fluctuations. As has been previously explained, the level fluctuations normally occurring do not in any way affect the sensing of binary pulses, because recording is done in the form of PCM-FM. Fig. 11 shows an example of the reproduced waveform of a single sine wave. With conventional audio recorders the SNR of the reproduced signal varies with recorded frequency. This variation results in a comparatively high SNR value for midband frequencies, but for higher or lower frequencies the modulation noise or harmonic distortion increases. On the other hand, PCM recording has a uniform noise and harmonic distortion for all recorded frequencies, and the recorded waves are reproduced almost perfectly. This is why the PCM recorder can be considered an excellent waveform recorder. Fig. 12 clearly shows the difference between waveforms reproduced from two types of recorders.

PHASE DIFFERENCE AMONG CHANNELS IN MULTICHANNEL RECORDING

When multichannel signals recorded on magnetic tape are reproduced, the phase difference between channels can be minimized in the case of PCM recording, and the residual amount is always uniform. This is because the phase difference is determined only by the two LPFs. In contrast with conventional audio recorders which record multichannel signals in the form of space division, the phase difference among channels is rather large and the difference fluctuates. Therefore, when stereophonic signals are reproduced, the acoustic source is heard as if it were moving over a certain range, and the system is deficient in that perfect localization is not accomplished. However, with PCM recording, the acoustic source is heard fixed on almost one point, and very good localization is achieved.

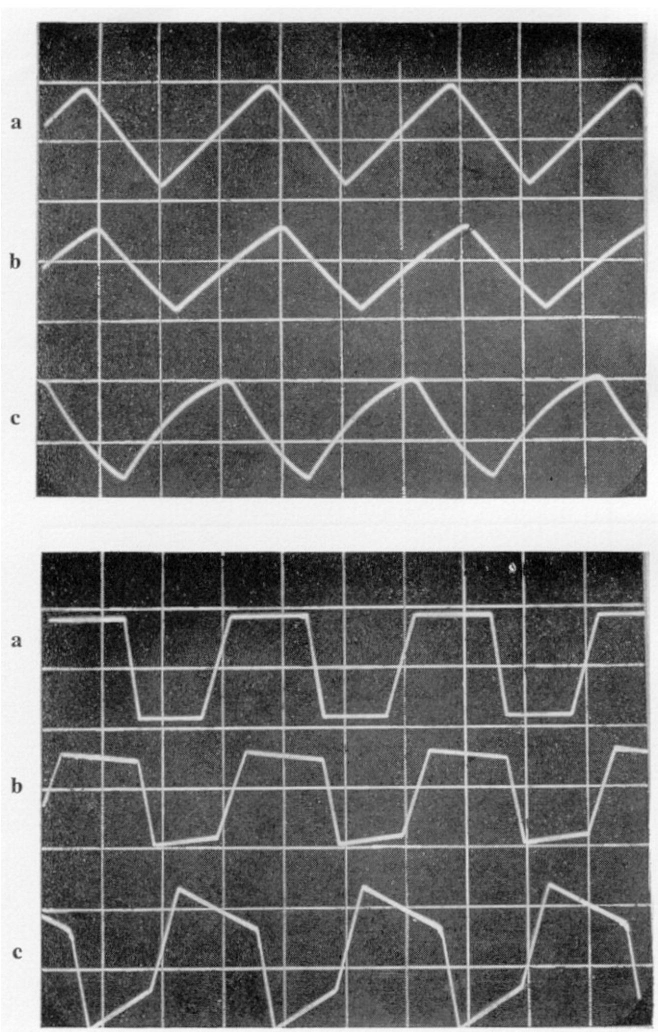

Fig. 12. Waveform responses. Horizontal scales 1 ms/div. *a*—input signal; *b*—output signal obtained by PCM recorder; *c*—output signal obtained by conventional recorder.

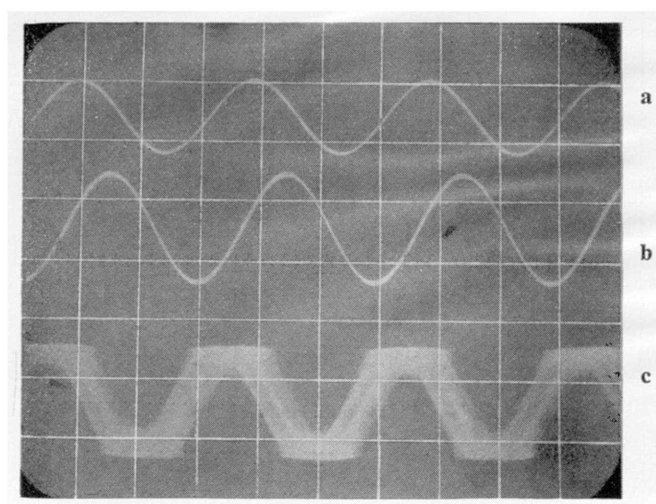

Fig. 13. Wow and flutter of reproduced signal. *a*—input signal; *b*—output signal obtained by PCM recorder; *c*—output signal obtained by conventional recorder.

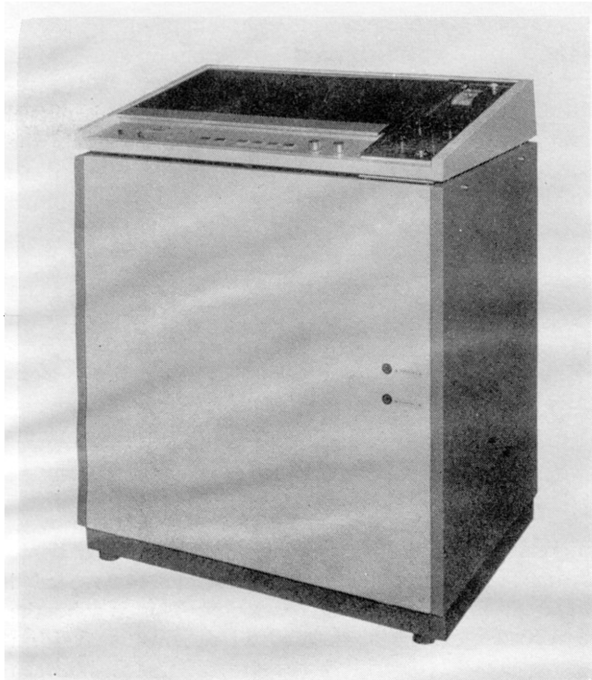

Fig. 14. PCM recorder.

WOW AND FLUTTER

Conventional magnetic tape recorders inevitably produce wow and flutter in reproduced signals. Wow and flutter caused by eccentricities in rotating mechanical parts or by stretching and contraction of the magnetic tape create FM noise in the reproduced signals. On the other hand, the PCM recording using a pulsed-oscillator synchronous system and a buffer memory can completely absorb any timing variations arising in the tape transport, and wow and flutter are, therefore, completely and thoroughly eliminated.

Fig. 13 shows an example of two types of wow and flutter. With PCM recording the frequency of reproduced signals is determined on the basis of the oscillation frequency of a very stable crystal oscillator used as a standard. Therefore, the recorded frequency is reproduced without undergoing any change whatsoever.

CONCLUSION

Many of the defects in conventional recorders which have posed problems can be remedied by recording audio signals in the form of pulse-code modulation. Using PCM techniques, a very high fidelity audio recorder has been developed. The PCM recorder owes its excellent performance to the transmission characteristics of the PCM signals and easy-to-handle digital signals. Because the PCM signals occupy a very wide frequency band, a wide-band recorder like a VTR must be used to record audio signals. For this reason, an easily operable helical-scan two-head type VTR was used in the present PCM recorder. A problem found in this type of recorder is the difficulty in editing tapes. An electronic editing system is required, and the development of such a suitable system is one of the most important and immediate projects for development. Another important project is to establish standards of PCM systems for recording signals with high fidelity.

The PCM recorder has a very wide range of applications. Recording of music signals with very wide dynamic range and recoding of signals which require preservation with high fidelity over long periods of time are easily achieved with the PCM recorder. Moreover, it can be used as a precision measuring instrument where high accuracy is required, and as an acoustic analysis recorder for processing large amounts of data in digital form. Fig. 14 shows the external appearance of the PCM recorder, model LB1007.

ACKNOWLEDGMENT

The author wishes to express his hearty thanks to K. Hayashi of NHK Technical Research Laboratories who first suggested to him the idea of the PCM recorder and has furnished guidance in the development of the recorder. Many other people have helped in the development, and the author is grateful for their assistance.

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