Pulse-Code-Modulation Recording System

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A pulse-code-modulation tape recorder for eight channels of audio information is described. The sampling rate for each channel is 47.25 kHz, and the channel samples are interleaved in 13-bit code plus parity and phase check bits for each, with three, eight channel blocks in a television-signal horizontal-scanning-line format. A studio-quality video tape recorder is used. Engineering quality considerations are explained and performance data are given. Applications to the mastering of disc records of unprecedented fidelity are indicated.

INTRODUCTION: Studies concerning the problems involved in magnetic tape recording have been conducted in a variety of ways in the past. Nevertheless, completely satisfactory performance has never been attained. The problems involved include distortion, dynamic range, wow, flutter, modulation distortion, etc. Although these problems have their causes in both the tape and the tape mechanism, the extent of improvement by a number of methods to date has been very gradual.

As an entirely different method for solving these problems we have incorporated pulse-code-modulation (PCM) techniques into magnetic tape recording, enabling an unprecedented improvement to be made, and have perfected an operational PCM recording system. It is, therefore, mainly intended for use as a master tape

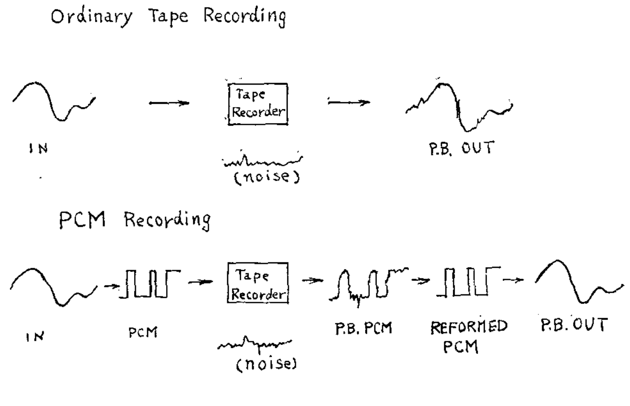


Fig. 1. Principle of PCM recording.

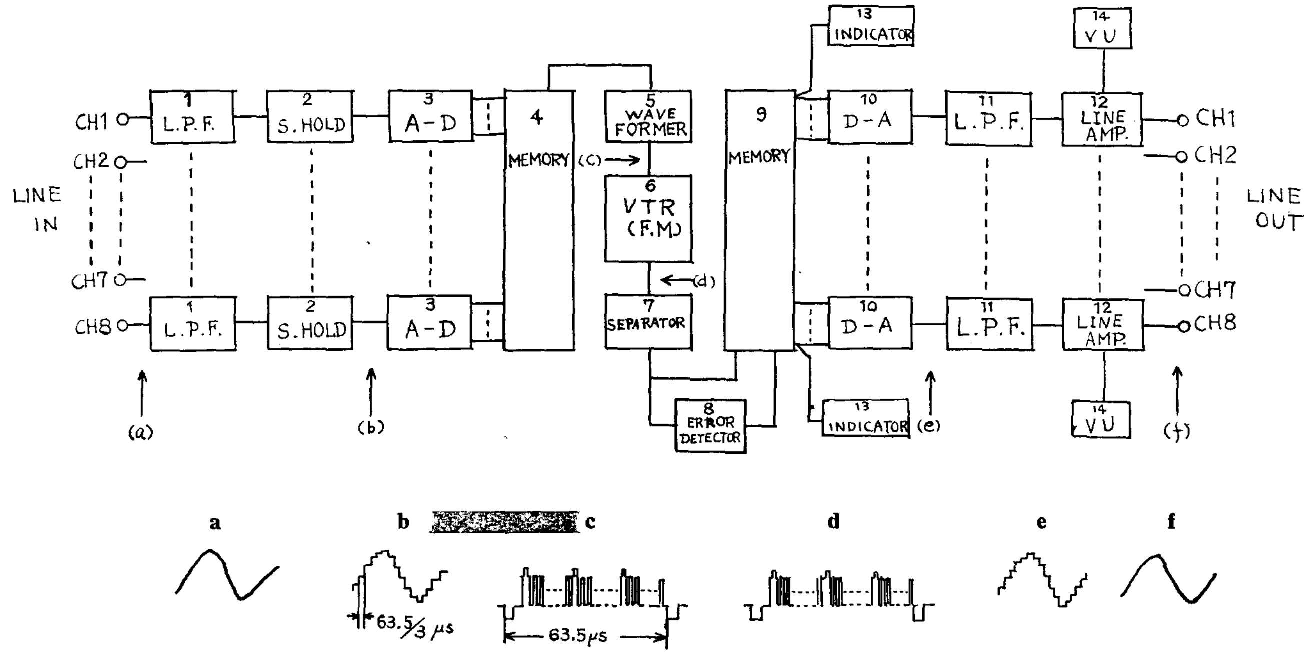


Fig. 2. PCM recording system block diagram. **a.** Line in. **b.** Sampling signal. **c.** PCM recording signal. **d.** PCM playback signal. **e.** D/A out. **f.** Line out.

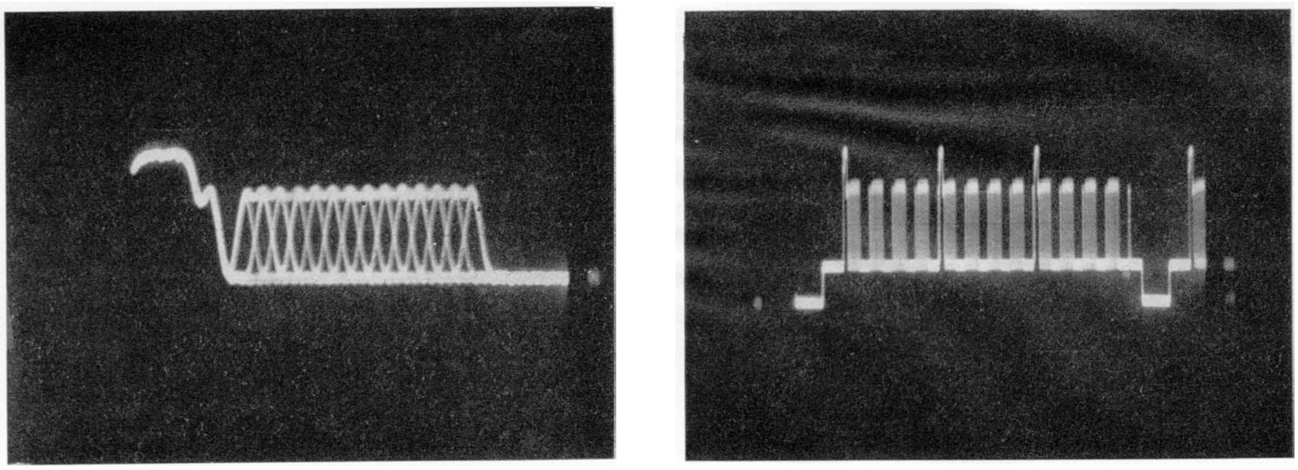
recorder for the cutting of disc records.

The PCM recording system is essentially a system that first converts an audio signal into a binary pulse code for recording on magnetic tape and then reconverts the code into the original audio signal. In ordinary recording, where the recorded representation is the direct analog of the signal, any disturbance causes an exactly corresponding disturbance in the reproduced signal, lowering its quality. The PCM recording system makes it possible, however, to adjust and supplement the coded waveforms to obtain a reproduction that is identical with the recorded signal. Fig. 1 explains the principle of PCM recording.

COMPOSITION OF PCM RECORDING SYSTEM

Fig. 2 is a block diagram of the PCM recording system. Eight-channel transmission is the standard for audio signal input and output. An audio signal is first passed through the low-pass filter to eliminate energy above the maximum frequency $f_{\max}$ of transmission. Then the signal is sampled in the sample hold circuit. The held signal is digitized into a binary code through the analog-to-digital (A/D) converter. The parallel signals of the eight channels are temporarily retained in the recording memory to become a series of PCM signals at required time intervals. A horizontal synchronizing signal is added to the PCM signal. In passing through the wave former, the code is turned into transmission waveforms recordable by the video tape recorder.

The waveforms are recorded and reproduced. The reproduced signal is fed to the signal separator to extract the data, which are put into the playback memory. Errors are detected and corrected here by the error detector. The output is reconverted smoothly into the original audio signal through the digital-to-analog (D/A) converter and the low-pass filter before being sent out through the line amplifier.



a **b**

Fig. 3. PCM recording signal. **a.** In one horizontal scanning-line format. **b.** For one sample of a single channel.

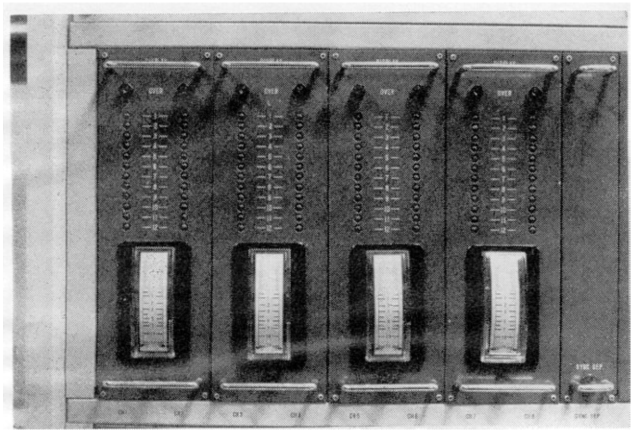


Fig. 4. Level indicator.

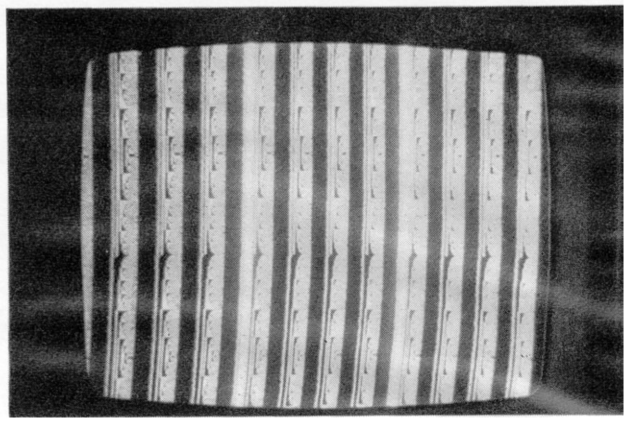


Fig. 5. Image of PCM recording signal, 60-Hz input signal for channels three, five and seven.

CODING

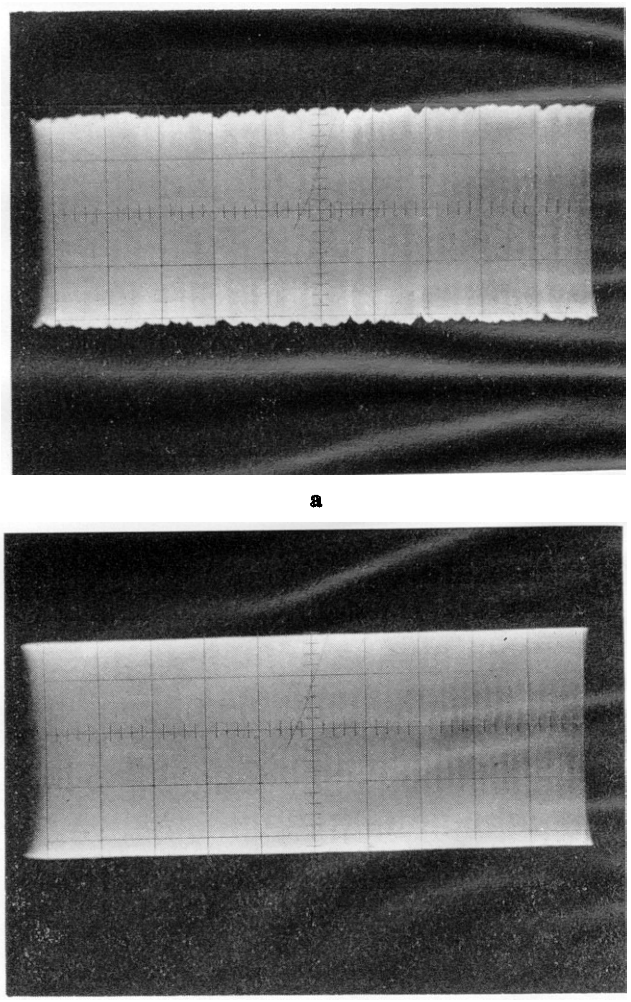
An audio signal is converted into a binary code. In order to do so, a sampling is taken of the instantaneous value of the continuous signal at a fixed period. If the sampling frequency f_s is more than twice the maximum frequency f_{max} of transmission ($f_s > 2f_{max}$), the information contained in the original signal will not be lost by sampling. However, because of limitations resulting from the characteristic of the low-pass filter used to eliminate the sampling frequency and side band, actual practice requires a sampling frequency of about $f_s = 2f_{max} \times 1.2$. If the sampling frequency is three times the frequency of the horizontal synchronizing signal, i.e., $f_s = 3 \times 15.75$ kHz $= 47.25$ kHz, where 15.75 kHz is the horizontal synchronization frequency of the video format, an audio signal of up to 20 kHz can be transmitted. The low-pass filter is needed for the audio signal input so as not to pass more than 20 kHz. The voltage of the sampling is retained in the sample-and-hold circuit until the time of the next sampling and, while being retained, is converted into digital code. If the maximum signal voltage were represented by the nth bit of the binary code, the digital value would be 2^n. Since the minimum voltage can be represented by the value 1, the dynamic range, that is, the ratio of quantizing noise to maximum voltage is $S/N = 20 \log_{10} (2^n)$.

With the use of a 13-bit code, 8192 voltage levels can be represented, and a dynamic range of approximately 78 dB is possible.

FORMATION OF PCM RECORDING SIGNAL

From the sample-and-hold circuit each of 13 bits is taken out in parallel by the eight-channel A/D converter and retained by the recording memory. A parity check bit to detect dropouts during reproduction and a check bit for phase-shift detection are also introduced, bringing the total to 15 bits. The eight-channel part is taken out consecutively as a series of signals. In other words, the data for one sampling consists of $15 \times 8 = 120$ bits. Data synchronization signals are added to help extract data during reproduction. The synchronization signals consist of pulses having a higher voltage level than the data-pulse level to permit their separation in playback. With a sampling frequency of 47.24 kHz, the density of data for 120 bits per sampling becomes 47.25 kHz $\times$ 120 bits $=$ 5670 kbits/s. This high bit rate and consequent bandwidth requirement means that a rotary-head video tape recorder (VTR) must be used.

In order to permit interchangeable use of VTRs and compatibility with other studio equipment, it is desirable to put the PCM signal in a format similar to that of TV signals. Three samplings of data, including the data sync, are placed in each horizontal scan section of the TV signal format. The insertion and extraction of data



a

b

Fig. 6. Reproduced 10-kHz sine wave. **a.** With ordinary two-track recorder at 380 mm/s. **b.** With PCM. Time axis—10 ms/div.

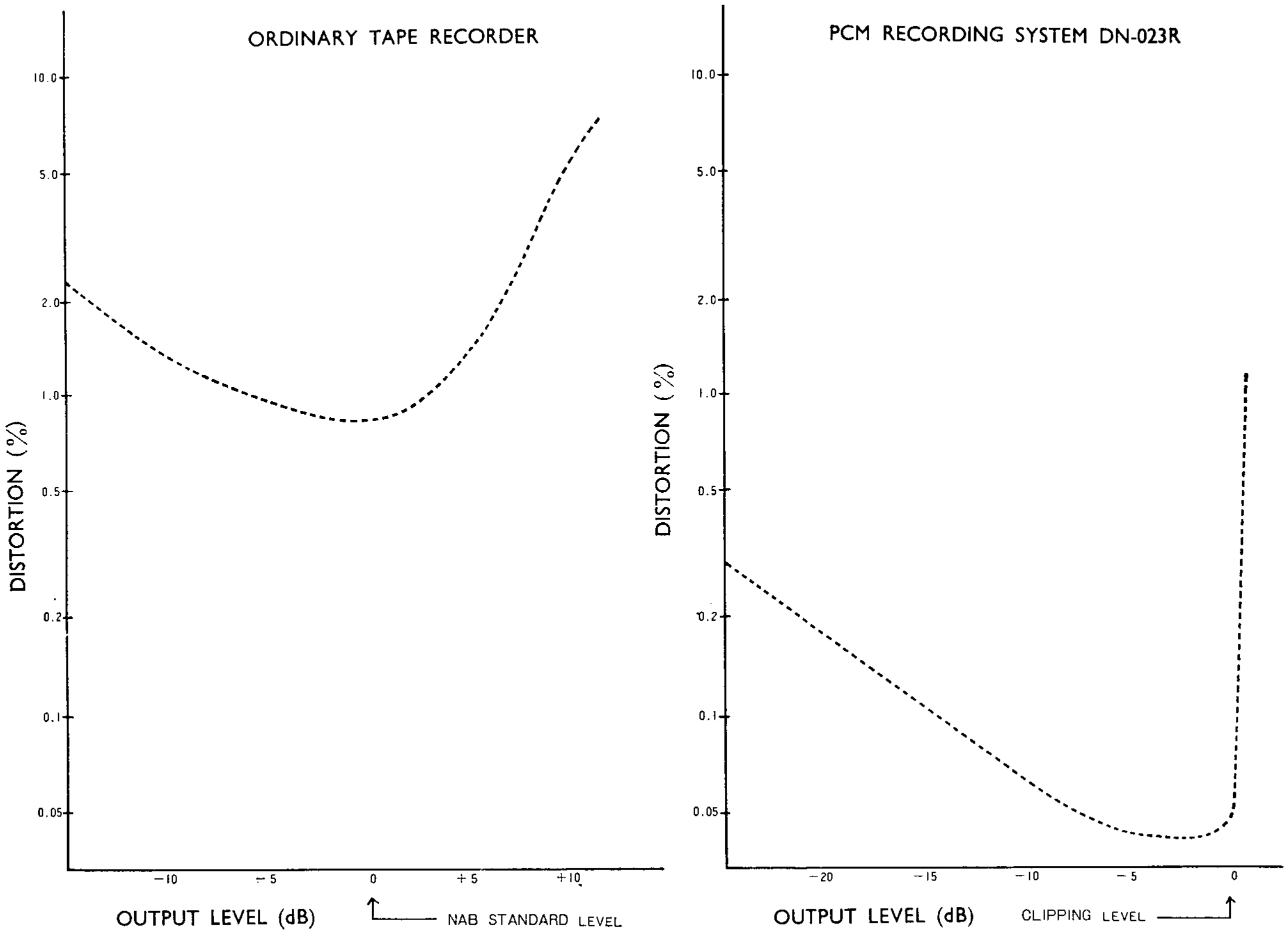


Fig. 7. Distortion characteristics at 1 kHz.

bits is governed by a clock frequency of 7.1825 MHz with the maximum data frequency being 7.1825 MHz × ½ ≈ 3.59 MHz, about the same as the chroma frequency of the NTSC color-image signal. The horizontal synchronizing signal and front and back porch signals are added and passed through a sine-squared filter in order to reduce the risk of waveform distortion in recording and reproduction. This is then sent out as a complete horizontal scanning-line signal. Fig. 3 shows this waveform.

CHANNEL SELECTION

An eight-channel audio-signal capacity is provided. In the case of four- or two-channel transmission, the remaining channels are employed to minimize errors in the recording and reproducing stages. By utilizing the memory, the same data are transmitted twice or four times. These are compared at the time of reproduction to detect errors and select the correct signal. In this way the accuracy of the correction is enhanced.

VIDEO TAPE RECORDER (VTR)

A four-head low-band VTR is used. The functional problems that can be considered are frequency characteristics, waveform distortion, noise, jitter, and dropout. Since there is a need to record the 3.59-MHz signal with the minimum of distortion, the qualities of the FM demodulation filter require that attention be given to such factors as the frequency and phase characteristics. As for jitter, in taking out various data by the clock signal in the phase of the data synchronizing signal, it is satisfactory if the jitter in one horizontal section is less than a minor fraction of 139 nanoseconds per data of the horizontal section, and this requirement will be completely fulfilled by the four-head VTR for broadcasting studio use. Tape with minimum dropouts is used. Since the dropouts become more severe as the FM carrier frequency becomes higher, it is meaningless to resort to a high frequency in an attempt to improve the signal-to-noise ratio.

A master recording system for disc records must be able to provide for variable pitch and depth cutting. In order to make this possible, an advanced head (two-channel direct audio recording) has been installed. To ensure good quality cutting, a stabilized frequency band is used and, in addition, both the number of revolutions of the turntable of the cutting machine and the speed of the master tape are reduced to one half. This is, in other words, half-speed cutting. In order to enable the tape speed of 380 mm/s at the time of recording to be reproduced at half speed of 190 mm/s, both the tape speed and the number of revolutions of the head drum are reduced to one half. Furthermore, the FM demodulation filter and as well as the servo units are also operated at half speed.

Both in recording and reproducing, the VTR's servo synchronization is locked to the standard TV synchronizing signals that control the PCM conversion into the TV format.

REPRODUCTION AND COMPENSATION OF DATA

In order to extract data from the PCM signal reproduced through the VTR, the data synchronizing signal is first separated and removed. A clock signal set for synchronous oscillation in this phase reads the data serially into a shift register memory, and the data bits are removed from the shift register in parallel. Even if jitter is included in the data from the VTR, the data synchronizing signal carried with the data will jitter in the same manner, and it is possible to extract jitter-free data from the reproduced signal. Here the check bits for parity and phase checks confirm whether or not there is a dropout, and any error in the signal is detected. In the case of two- or four-channel transmission, the various bits of the same data in different positions are compared to see if they agree. If an error in data is detected, a correction is made by calculating the average value of the data preceding and following the loss of one sample. When the error is more than one sample, the preceding good data are substituted. In two-channel transmission the information without error can be selected and used.

The signal coming into the memory part includes jitter, but since it is removed by a fixed synchronous clock signal, as at the time of recording, wow and flutter disappear completely.

OUTPUT D/A CONVERTER

The data taken out are converted successively into analog voltages. In this way, a signal of the same form as the sampled signal at the time of recording is obtained. This signal is passed through the low-pass filter to turn it into a smooth continuous signal. During this process the sampling signal is a rectangular pulse rather than an ideal impulse. Therefore, the high-frequency response is altered slightly by the aperture effect. If t_0 is the width of the pulse, the form of alteration is $[\sin(\omega t_0/2)]/(\omega t_0/2)$. This corresponds to linear filtering, and the frequency response can easily be boosted. The phase does not present any problem, since the formation of all the various frequencies is delayed together by $\frac{1}{2}\ t_0$. There are two kinds of low-pass filters, one up to 20 kHz and the other 10 kHz for half-speed reproduction.

LEVEL INDICATOR, WAVEFORM MONITOR

In PCM recording, the level of A/D conversion is set, and the greater the level, the better the signal-to-noise ratio, as in the case of ordinary tape recorders. An ordinary VU meter can also be used, but an even better level monitoring can be obtained by monitoring the various digitalized bits. The significant digits of the binary code gradually increase in number in accordance with the increase of the input level. A light-emitting diode lamp indicator is attached to each digit, and a holding circuit is provided for easy visibility. With a level monitor prepared in this way, the number of monitoring lamps lighted will increase with every 6-dB increase of signal, and the complete instantaneous level can be observed. In addition, an AND circuit for all the bits is set up to

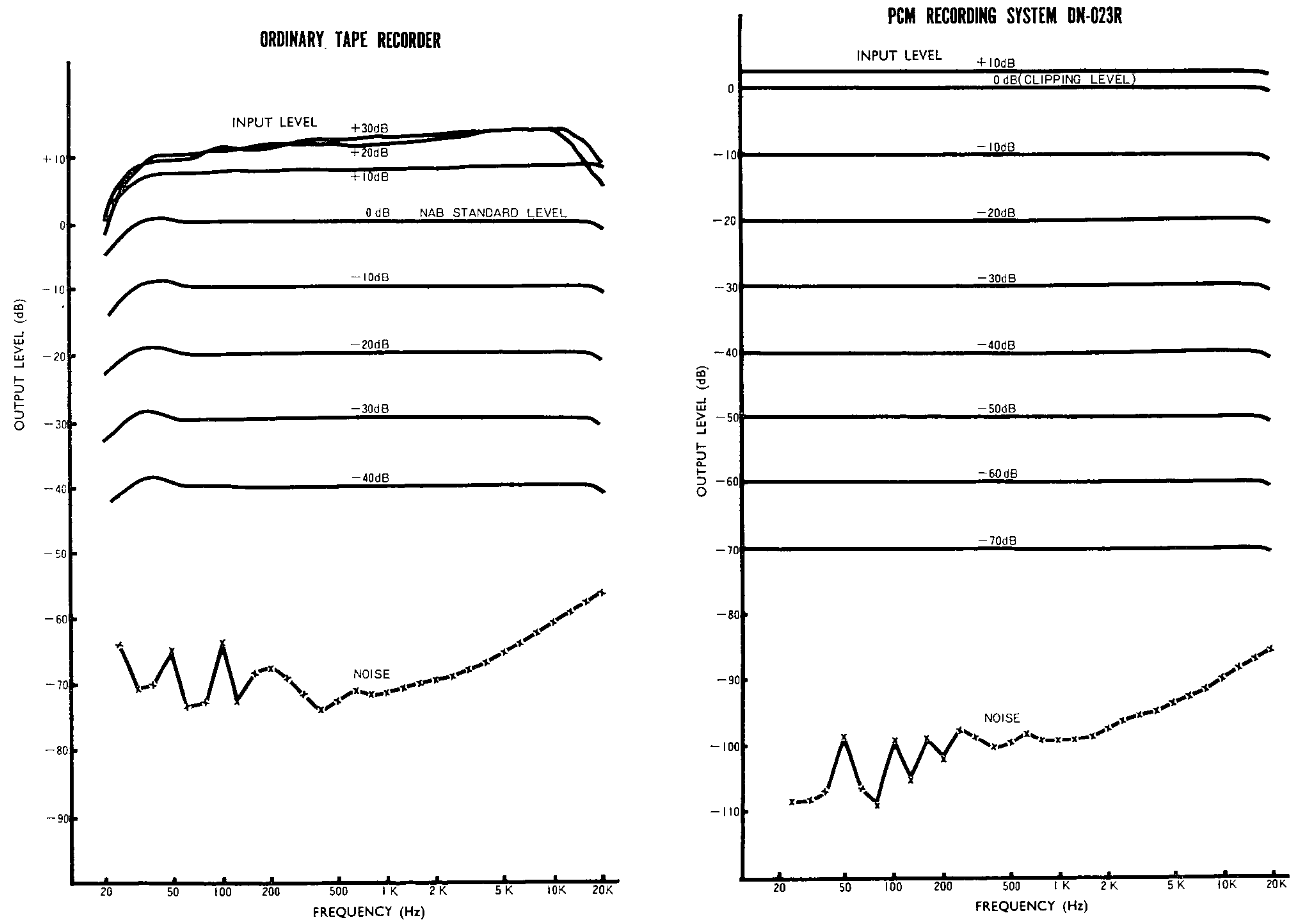


Fig. 8. Frequency response.

indicate the clipping level and give warning. This picture is shown in Fig. 4. In the process of VTR recording of the PCM signal, a monitor device for VTR is also installed. Observation of the waveform can be done with an oscilloscope. It can be observed at the same time as an image through the TV Braun tube. The various bits of the binary code can be seen as vertical stripes. The state and level of the various beams can also be seen. Fig. 5 shows the video picture.

CHARACTERISTICS AND ADVANTAGES OF PCM RECORDING AND REPRODUCING EQUIPMENT

By using the PCM method it is possible to enhance performance in a number of ways. Its features include

1) A wide dynamic range
2) Little distortion
3) No wow and flutter
4) No fluctuations due to tape-speed variations
5) No interchannel talk
6) No fluctuation of level due to record/playback variations
7) No fluctuation of phase
8) No modulation distortion
9) No print-through (can be stored a long time)
10) No signal degradation in duplication
11) A flat frequency characteristic over a wide range.

These are the points of difference with an ordinary tape recorder. The differences are shown in Fig. 6 for the reproduced sine wave, in Fig. 7 for the distortion characteristic, and in Fig. 8 for the frequency response. Another feature is that despite the use of VTR, the same capabilities ordinarily expected of a tape recorder are possessed by the PCM recorder. In other words,

12) Multichannel recording and reproduction possible
13) Half-speed reproduction possible
14) Equipped with an advance head
15) Automatic editing and splicing.

The performance and method of the equipment are shown in Table I.

APPLICATIONS AND RESULTS OF USE

Figure 9 shows the equipment. As explained in the foregoing, the quality of the reproduced signal has been greatly enhanced and, even when heard by the ear, sound practically unchanged from the input can be reproduced. The effect of dropouts, which accompanies magnetic tape as an unavoidable defect, cannot be perceived for all practical purposes because of the accurate operation of the error-correcting circuit.

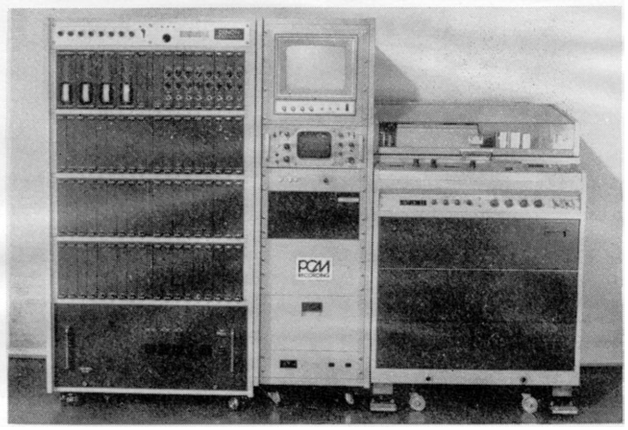


Fig. 9. PCM recording system.

This equipment was used for record cutting. For this, half-speed cutting and nondistortion cutting were carried out together. Half-speed cutting has been explained earlier. In nondistortion cutting, compensatory steps were taken during the cutting to eliminate tracing distortion resulting from the difference in shape between the cutting and the reproducing stylus. From the disc record produced by combining the above described methods, we were successful in obtaining reproduced sound of unprecedented fidelity.

In editing too, a method not too different from that with the ordinary tape recorder can be adopted. For

Table I. Performance and Method of PCM Recording System.

Performance	
Frequency	DC to 20 kHz ± 0.5 dB or less (at 380 mm/s) DC to 10 kHz ± 0.5 dB or less (at 190 mm/s)
Dynamic range	More than 75 dB
Distortion	Less than 0.1 percent at operating level
Channel crosstalk	Not measureable
Wow and flutter	Not measureable
Method	
Pulse-code modulation	13-bit natural code
Audio sampling frequency	47.25 kHz
PCM recording signal	Standard TV signal (except advance-head signals)
PCM data clock frequency	7.1825 MHz
Number of audio channels	Eight, four, two, selectable
Number of advance-head signal channels	Two
Magnetic tape recorder	Four-head low-band VTR
Tape speed	380 mm/s (at recording) 380 mm/s, 190 mm/s (at playback)
Head–tape relative speed	40 m/s
Magnetic tape	2-inch video tape

sound recording, careful planning is necessary to obtain equally high fidelity signals for recording the source. It is believed that this equipment can be used not only as a master recorder for producing records, but also in research on sonority (high-pitched tone) or as a data recorder, as well as in many other applications.

THE AUTHORS

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