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IMPROVED PCM (PULSE CODE MODULATION) RECORDING SYSTEM

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SUMMARY Introduced below is a new pulse-code modulation (PCM) tape recorder for professional applications. It is capable of eight-channel recording and reproduction. This system is an improved version of the system which was described in the Journal of AES published in September 1973 (volume 21, number 7). Both systems are compatible, and a four-head low-band video tape recorder is used as the mechanical part of the recording system. The new system features an emphasis circuit and 14-bit A/D and D/A conversions to improve the dynamic range of the system, as well as a selection circuit for the digital data for high quality duplicating and synchronized recording with two video tape recorders and a PCM converter. Engineering quality considerations and performance data are detailed below.

Introduction

The PCM (Pulse-Code Modulation) recording method has been spotlighted in recent years as a method by which daterioration in sound quality, which occurs when recording in studios and halls, can be minimized. It works in a different way from the conventional analog recording system.

Nippon Columbia Co. came out with the first PCM recording system(DN-023R) for disc master applications in 1972, and it was discussed in the Journal of AES. Two years later, the company brought out the second model which was even more compact and lightweight than its predecessor. Both systems were installed as part of the record production process and they have been used in the manufacture of 150 or so LP records to date.

In the meantime, however, the company continued to conduct research on the basic principles of PCM recording and sought ways and methods to eliminate the problems besetting these systems. Based on the findings of these activities, the company successfully completed the third model, the DN-034R, which is not only compatible with the two previous varsions, but which also displays an improved performance and is more compact and lightweight.

This report aims to outline the principle of PCM recording, the results of error measurements and the workings of the recently developed system itself.

New requirements

Listed below are the specifications and electrical performance of the first two PCM recording systems developed by Nippon Columbia.

Specifications of the DN-023R
Modulation : PCM
Coding : 13-bit sign and magnitude binary code
Transmission clock frequency : 7.1825 MHz
Audio sampling frequency : 47.25 kHz
Transmission waveform : standard TV signal (except vertical sync signal)
Number of audio channels : 8/4/2, selectable
Usable tape : 2-inch video tape

Electrical performance data of the DN-023R
Frequency response : DC - 20kHz ±0.5 dB
Dynamic range : more than 75 dB
Distortion : less than 0.1 % (at operating level)

These two recording systems were used to record various programs in studios and halls, and despite the fact that in almost every case, the programs were of a satisfactory quality, the company decided that something was lacking in the dynamic range when recording in a studio with microphones close to the percussion instruments. Furthermore, the mixing engineers gave high marks to the overall quality when a circuit displaying the same saturation characteristics as the tape was inserted into the systems.

A number of methods were tried out with duplicating and synchronized recording, and the company soon realized the desirability of developing a recording system which could turn the merits of the PCM system to advantage for high-quality duplicating and synchronized recording.

Dynamic range

With PCM recording, quantizing noise in theory is generated. It is a well-known fact that the relationship of quantizing noise power N_Q, caused by uniform quantization, to the quantizing steps ΔV can be expressed by the following equation:

$$N_Q = \frac{(\Delta V)^2}{12} \quad (W)$$

Furthermore, the relationship between the S/N ratio with the maximum amplitude sine waves and the quantizing noise, and the number of bits M can be expressed as:

$$S/N_Q = \frac{3}{2} \cdot 2^M = 6M + 1.8 \text{ (dB)}$$

Therefore, to expand the dynamic range requires

(i) An increase in the number of bits.
However, to expand the dynamic range while keeping the number of bits at the same level offers the following possibilities:

(ii) Emphasis

(iii) Companding in the analog signal stage

(iv) Non-linear quantization

(v) Differential PCM

(vi) Forecasting quantization

(vii) Conversion of orthogonal conversion, etc.

The third possibility can be brought to reality by connecting a Dolby or dbx system, and the recording system's frequency response and other characteristics can be made superior to those of an ordinary tape recorder. Therefore, this method produces better results than those obtained in normal circmstances.

It is necessary to analyze the signals if the second, fourth, fifth, sixth and seventh methods are introduced. With the latter three methods, it is necessary to take the utmost care when reducing the signals' redundancy since the effect of the error is great.

Nippon Columbia decided to study the first and second possibilities as part of its plan to expand the dynamic range, and it opted for 14-bit conversions and an emphasis circuit. With the emphasis circuit introduced in the system the recorded programs were analyzed. Based on the findings and on the limitations of the disc cutting system's dynamic range, the emphasis characteristics shown in Fig. 1 were adopted.

The addition of the emphasis circuit resulted in an audible improvement in the S/N_Q of about 8-9 dB. and with 14-bit quantization, an audible S/N_Q of the order of over 90 dB was obtained.

Duplicating and synchronized recording

When using the PCM recording system to duplicate, three methods can be considered (refer to Fig. 2):

(i) Audio signal duplication

(ii) Transmission signal duplication

(iii) Data duplication

With method (i), the quantizing noise increases every time the duplication process is repeated, and this causes the S/N_Q to deteriorate. With method (ii), the noise, distortion and jitter of the transmission signals increase and this results in an increase in errors. With method (iii), the only increase comes from the errors caused by tape dropout, and this is therefore a method which makes the most of the PCM's merits.

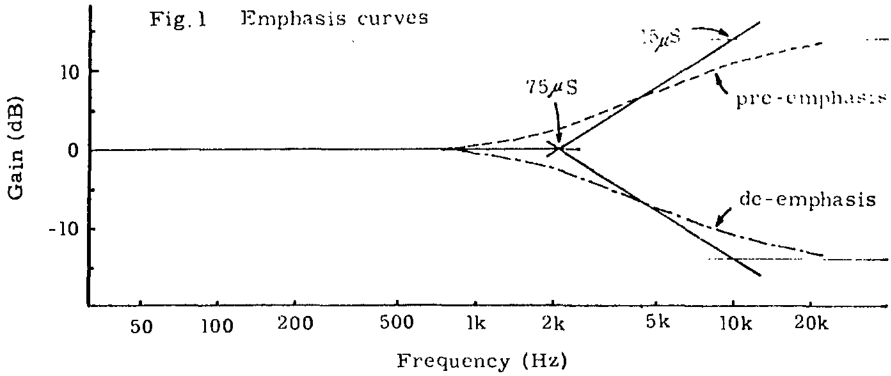
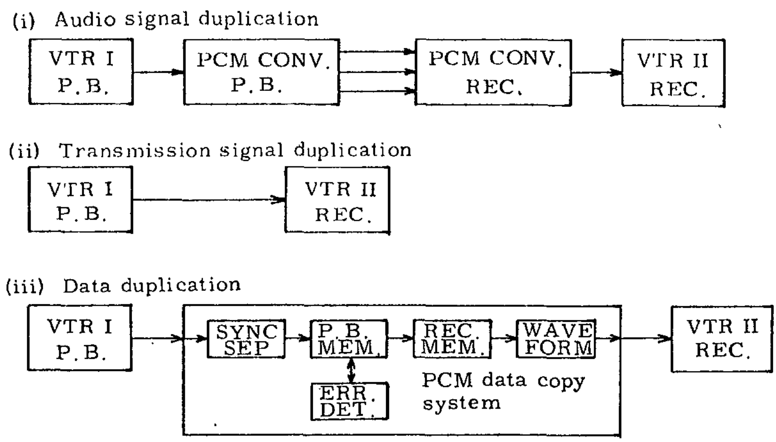


Fig. 1 Emphasis curves

Fig. 2 Duplication



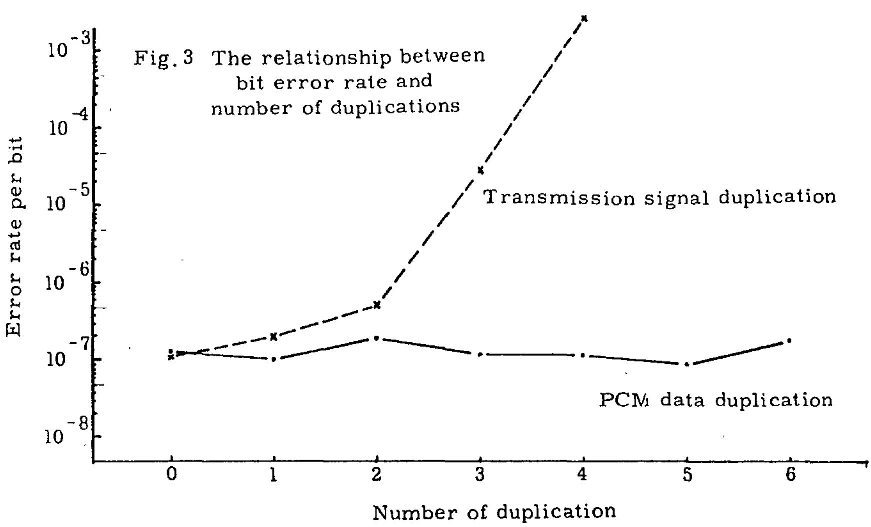


Fig. 3 The relationship between bit error rate and number of duplications

Fig. 3 shows the results of measuring the relationship between the bit error rate and the number of duplications when the transmission waveform and data duplication methods are applied. It is apparent that data duplication gives the superior quality.

In a system which records serial signals which are time division multiplexed and which employs a video tape recorder as the recorde, like the PCM recording system does, it is extremely difficult to record synchronously with one recorder - unlike using an ordinary tape recorder. Nevertheless, if two recorders are employed and a recording data selection circuit provided (refer to Fig. 4), then the above data duplication method becomes the most appropriate since it allows high-quality synchronized recording.

Data errors and interpolation

It is very important to give consideration to methods of detecting data errors in PCM recording systems. It is also necessary to interpolater in such a way as cannot be heard at the same time as detecting the errors accurately. The errors in PCM recording systems can be traced to tape dropouts. With the Scotch 420 video tapes which are currently being used, the error rate per bit ranges from 10^{-8} to 10^{-7}, and the detection error when conducting a parity check is 10^{-2} or so. This value is allowable for all practical purposes.

Three interpolation methods can be considered: preceding signal retention, preceding/following signal averege interpolation and interpolation using a function generator. The PCM recording system mentioned here employs preceding/following signal average interpolation when there is an error in an isolated sample, and preceding signal retention when there is an error in two or more consecutive samples. To check the effect of this interpolation, interpolation was conducted with pure tone, and the results of the findings concerning the threshold frequency of abnormal noise perceptibility are shown in Fig. 5 when measured both with and without an emphasis circuit (Fig. 1). The results prove that the preceding/following signal average interpolation is far more effective than the preceding signal retention method. They also indicate that the emphasis circuit not only improves the S/N_Q mentioned above, but it is also effective in reducing the abnormal noise when errors are generated.

Ordinary music signals, not pure tone, differ according to the musical sounds. With the emphasis circuit is used with preceding signal retention, abnormal noise can be perceived in cases where interpolation (21 x 4 μsec.) is performed with four or more consecutive samples.

Improved PCM Recording System (DN-034R)

All the above findings were incorporated into the development of the Improved PCM Recording System (DN-034R) which is compatible with the conventional models.

The only difference in the specifications of the improved PCM recording system from those of its predecessors is the 14-bit coding - nothing else has been changed. Refer to Fig. 6 for the improved version's block diagram.

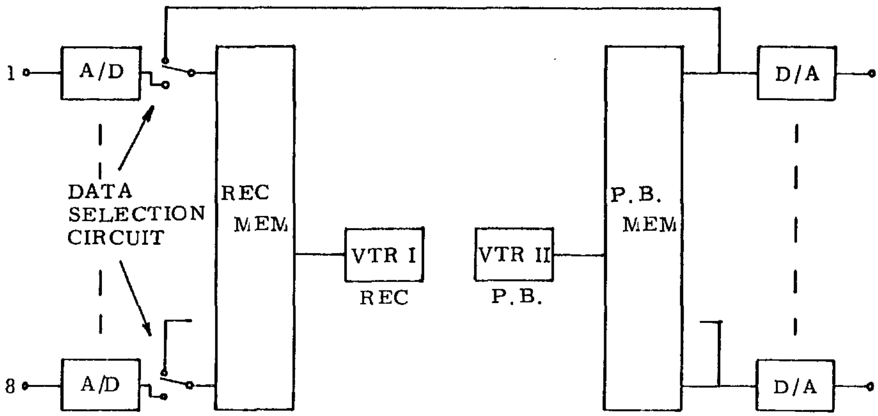


Fig. 4 Block diagram of synchronized recording using two VTRs

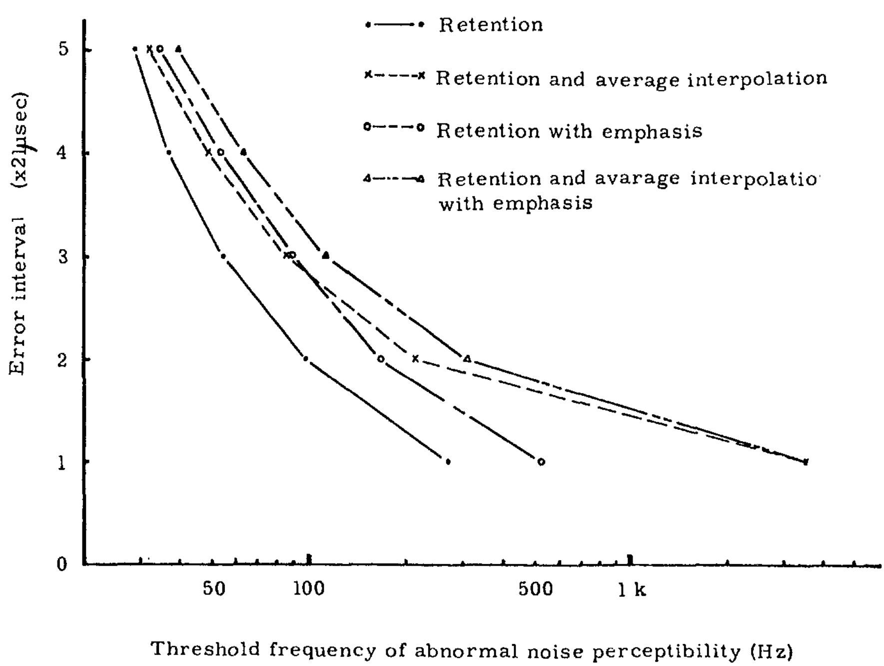


Fig. 5 The relationship between error interval and threshold frequency of abnormal noise perceptibility

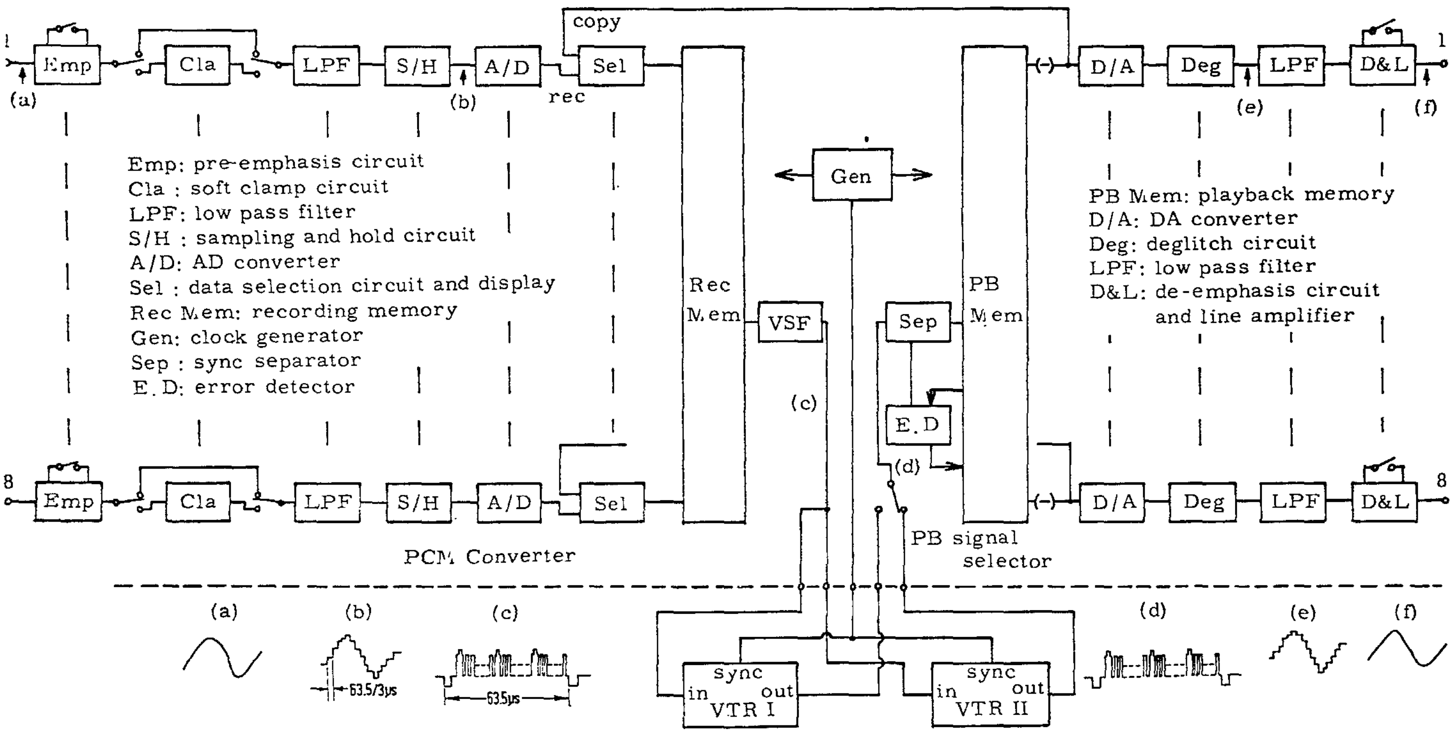


Fig. 6 Block diagram of DN-034R

As shown in Fig. 6, this system digitalizes eight channel audio signals and records PCM transmission waveforms, which are recorded as FM on the tape by the four-head low-band VTR.

With reproduction, the recorded PCM signal is first demodulated, and then data errors are detected and corrected.

After correction, the data are converted into analog signals, and eight channel audio signals are fed out.

The main features of the PCM recording system are outlined below.

(1) Input stage

The PCM recording system features an eight-channel audio signal input. The signals which are applied to each of the inputs pass through the pre-emphasis circuit whose characteristics are shown in Fig. 1, and through the soft clamp circuit which has saturation characteristics resembling those of an ordinary tape. They are then fed to the input filters.

The signals can also bypass the emphasis circuit and the soft clamp circuit. They normally employ the emphasis circuit for ordinary recordings but bypass the soft clamp circuit except when special effects are required.

The input filters display flat characteristics from DC up to 20 kHz and they feature an attenuation of over 72 dB with a frequency of over 23.625 kHz, which is half the sampling frequency.

(2) Sampling and coding

In these processes, each of the eight channels comprises an independent sample and hold circuit and A/D converter for maintenance.

The sampling frequency is triple the frequency of the TV horizontal sync signal: 15.75 x 3 = 47.25 kHz (approx. 21 μsec).

The audio signal is linearly converted into a 14-bit natural binary code by the A/D converter, and its LSB becomes 61 ppm.

Therefore, the accuracy of the components before A/D converter must be significantly less than 61 ppm.

There are the reasons why companding by nonlinear coding has not been adopted here: it is necessary to know in advance the level distribution of the program sources for the optinum companding characteristics, it is difficult to determine the optimum level for individual music, and the system is apt to be affected by distortion.

(3) Transmission waveforms

The A/D converter's 14-bit output data enter into the data selection circuit which is provided for data duplication and synchronized recording. Their level is displayed and after the data to be recorded are selected between the A/D converter's output data and the reproduction output data, they are converted into a sign and magnitude binary code.

They are then temporarily stored in the recording memory and turned into series signals. Parity check bits and sync signals are added and the signals are then formed into transmission waveforms in the shape of TV signals (refer to Fig. 7).

As shown in Fig. 7, three data samples are placed in the horizontal scanning section, and horizontal sync, front and back porch signals are added. A horizontal interval is equivalent to 456 bits, so the clock frequency becomes 7.1825 MHz.

As the signal is a NRZ (non return to zero) type, the maximum data frequency becomes 3.59 MHz, which is half the clock frequency.

Therefore, an ordinary VTR is suitable for the PCM system. This waveform is similar to the TV signal (except the vertical sync signal), and so it is easy to apply broadcasting VTR techniques to this system for editing and duplicating.

(4) Audio transmission channels
Standard audio transmission uses eight-channels.
In the case of transmission with fewer than eight channels, the remaining channels are employed to minimize the transmission errors.
In 4-channel transmission, the coded signals for channels 1 to 4 are each staggered by 1 sample and fed to channels 5 to 8, so that at the time of reproduction the bit of the same signal in different positions can be compared to see if it tallies or not. The reason for putting signals in different sample positions is that they must be prevented from being taken out by the same synchronizing signal and also prevented from simultaneous omission due to a large dropout.

In 2-channel transmission, the same signal is also sent to channels 3 to 4 so that the right one is chosen at the time of reproduction. This enhances the accuracy of the correction.

(5) Magnetic tape recorder
An ordinary 4-head low-band VTR is employed.
In the VTR for the PCM recording system, the consideration must be given to jitter, frequency response, SN ratio, waveform distortion (including phase distortion), and other factors.

Jitter should be less than 139 nano-seconds (1 bit) in intervals of eight channel serial data (120 bits), because data are extracted by the clock signal which is locked to the reproduced data synchronizing signal.
The 3.59 MHz signal must be reproduced without any distortion in the waveforms and phase characteristics.
Tapes with minimum dropouts should be selected.
When the FM carrier frequency falls, the dropouts decrease and although the carrier leakage and the moire increase, they have a negligible effect on the PCM recording system as compared with common use, and so the low band VTR is employed.

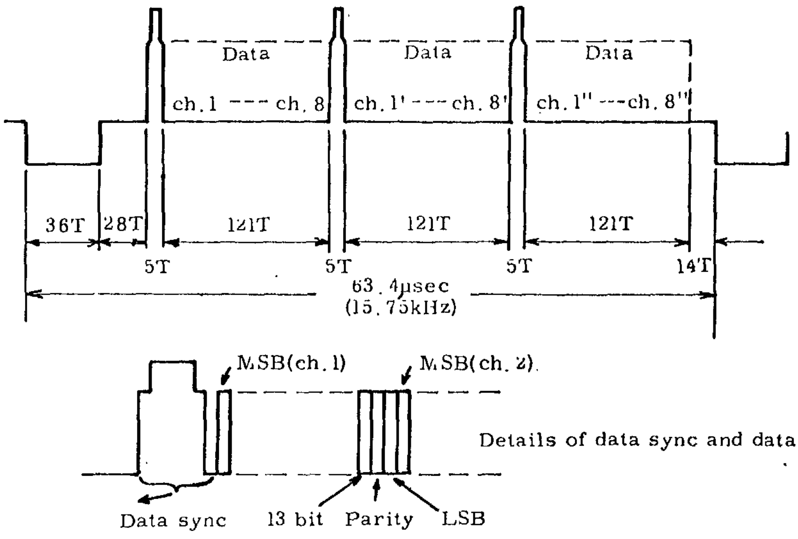


Fig. 7 Transmission waveform

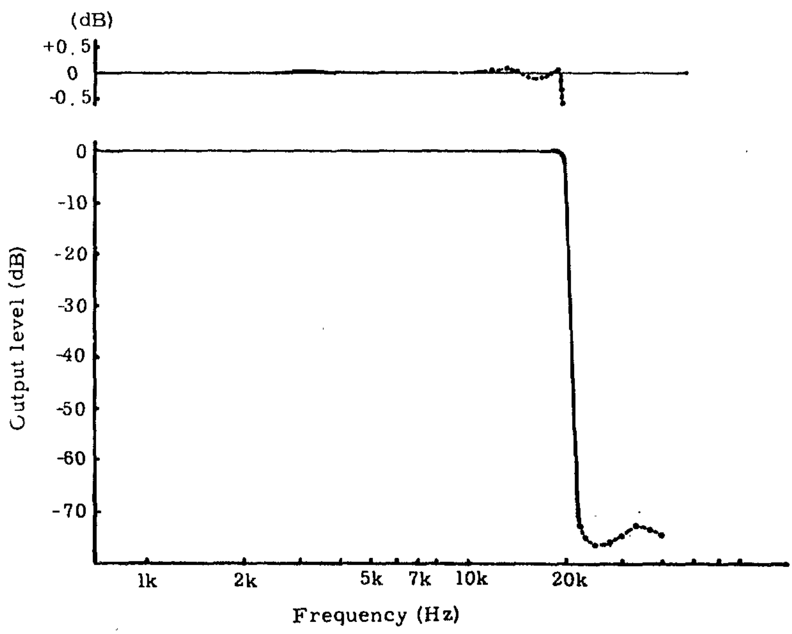


Fig. 8 Frequency response

(6) Data extraction, error detection and interpolation.
The data are extracted by the clock signal locked to the data sync signal and stored in a shift register.
The data are checked by the parity bit, and checked for dropouts by the level detector. With 2- or 4-channel transmission, all the data of the same sample are compared to correct errors.
Interpolation consists of adding the preceding/following average signals for each omission of one sample.
When more than two samples are omitted, the preceding signal is retained.
In 2-channel transmission the right data can be chosen. These processes permit the data error noise to be minimized in the analog output signal.
The extracted and interpolated data are converted into a nutural binary code and sent out as eight channel parallel data.

These output data pass through the data output connectors and then through the data input connectors (these are usually connected to the output connectors). They are supplied to the D/A converter and the recording data selection circuit.

The data input and output connections are provided to facilitate connection to a computer or digital mixing console.

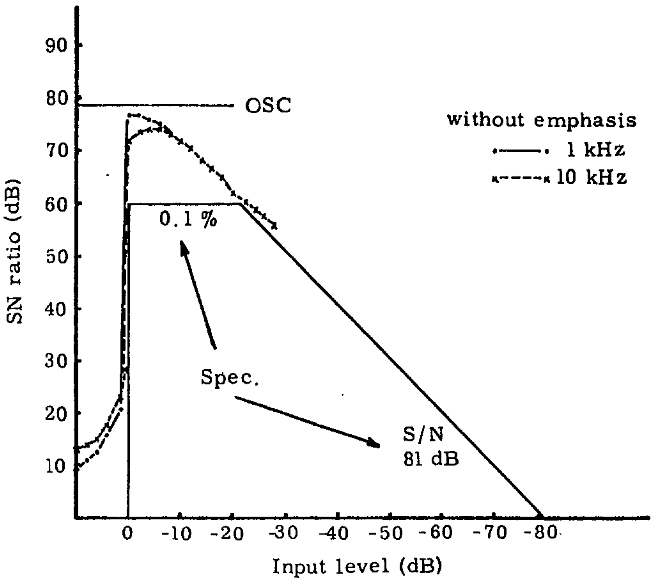
(7) Decoding and output stage
The eight channel parallel data are converted from digital signals to analog signals by the 14-bit D/A converter. They pass through a deglitch circuit and a low-pass filter. After being fed to the emphasis circuit, the signals are then given de-emphasis characteristics (Fig.1) and are fed out from the line amplifier.

On the other hand, when playing back tapes on the improved PCM recording system which were recorded on the conventional versions, 13-bit D/A conversion is performed. The same applies for tapes which were recorded on the improved system and are played back on the conventional models.

Performance data

Fig. 8 and Fig. 9 show the measured frequency response and the S N ratio, respectively. The performance of the PCM recording system based on these results is as follows:

(1)	Frequency response	DC - 19 kHz $\pm$ 0.2 dB DC - 20 kHz $^{+0.2}_{-1.0}$ dB
(2)	Dynamic range	More than 81 dB (without emphasis) 89 dB (with emphasis)
(3)	Distortion	Less than 0.1% (at operating level, DC - 20 kHz)



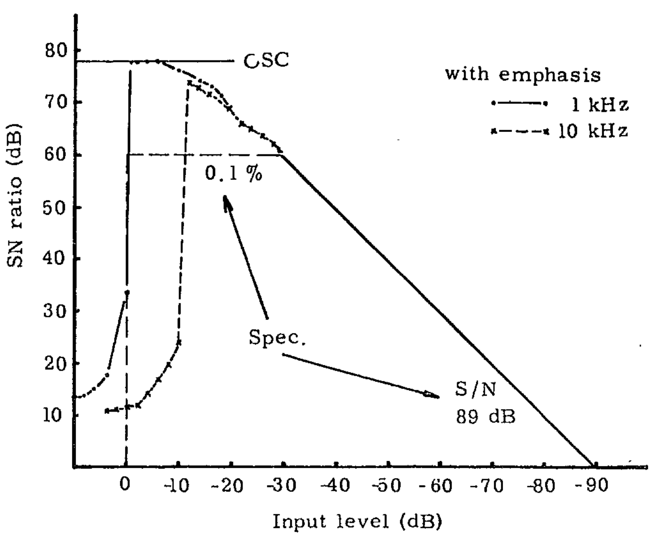


Fig. 9 SN ratio characteristics

When Scotch 420 is used as the recording tape and the FM carrier is low-band, the error rate per bit is 3×10^{-8} and that is 4.2×10^{-7} per sample of one channel. This means that the error occurs once every 50s. But in the case of high band, the error rate per bit is 1.2×10^{-7} on the same tape. Thus the low band is shown to be better than the high band. The detection error is about 10^{-2} in the case of eight-channel reproduction.

Conclusion

As a result of including all the findings of the research on PCM recording in the development of the new PCM recording system, Nippon Columbia claims that its new invention provides the characteristics which it originally forecast and that it turned in a very impressive performance with input/output audition comparison tests.

The new system also brings to reality high-quality duplicating and synchronized recording. There is absolutely no difference in the reproduction sound when duplicating has been repeated about 10 times.

The new system is provided not only with analog input and output connectors but also with digital data input and output connectors. If it is connected to a digital mixing console with delay, addition/ subtraction, multiplication/ division functions, then panpot, level control, equalization, delay and reverberation adding operations can be undertaken, and it is also possible to mix down with digital signals.