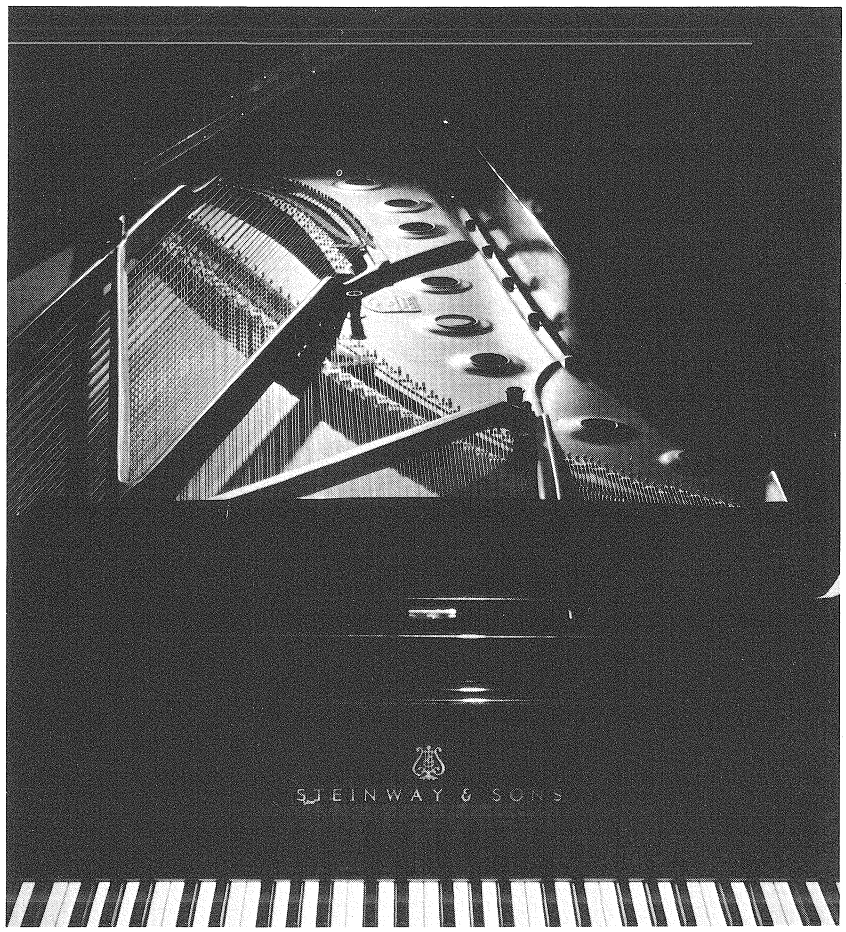
PCM-A Breakthrough in Audio Technology

SONY®



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Introduction

Unless we play a direct cutting record, the music we listen to and enjoy comes to us via some form of tape recording process. In other words, the quality of the music we hear always depends largely on the quality of the original tape recording.

In our pursuit of the ultimate in high-fidelity music reproduction, we have found that even the best of conventional recording systems, and their high state-of-the-art, are still limited by a number of drawbacks in the form of noise and dynamic range limitations. These limits are inherent in the tape, heads, and other mechanical factors, and although they can be minimized by conventional means, it is virtually impossible to eliminate them completely.

How would it be then, if, instead of further refining and perfecting the presently known recording technology, we were to make a system whose performance is no longer limited by these mechanical parameters? If there were only a way to leave the limitations of present recording technique behind and set out in a new direction, the quality of the music we hear could take a giant step.

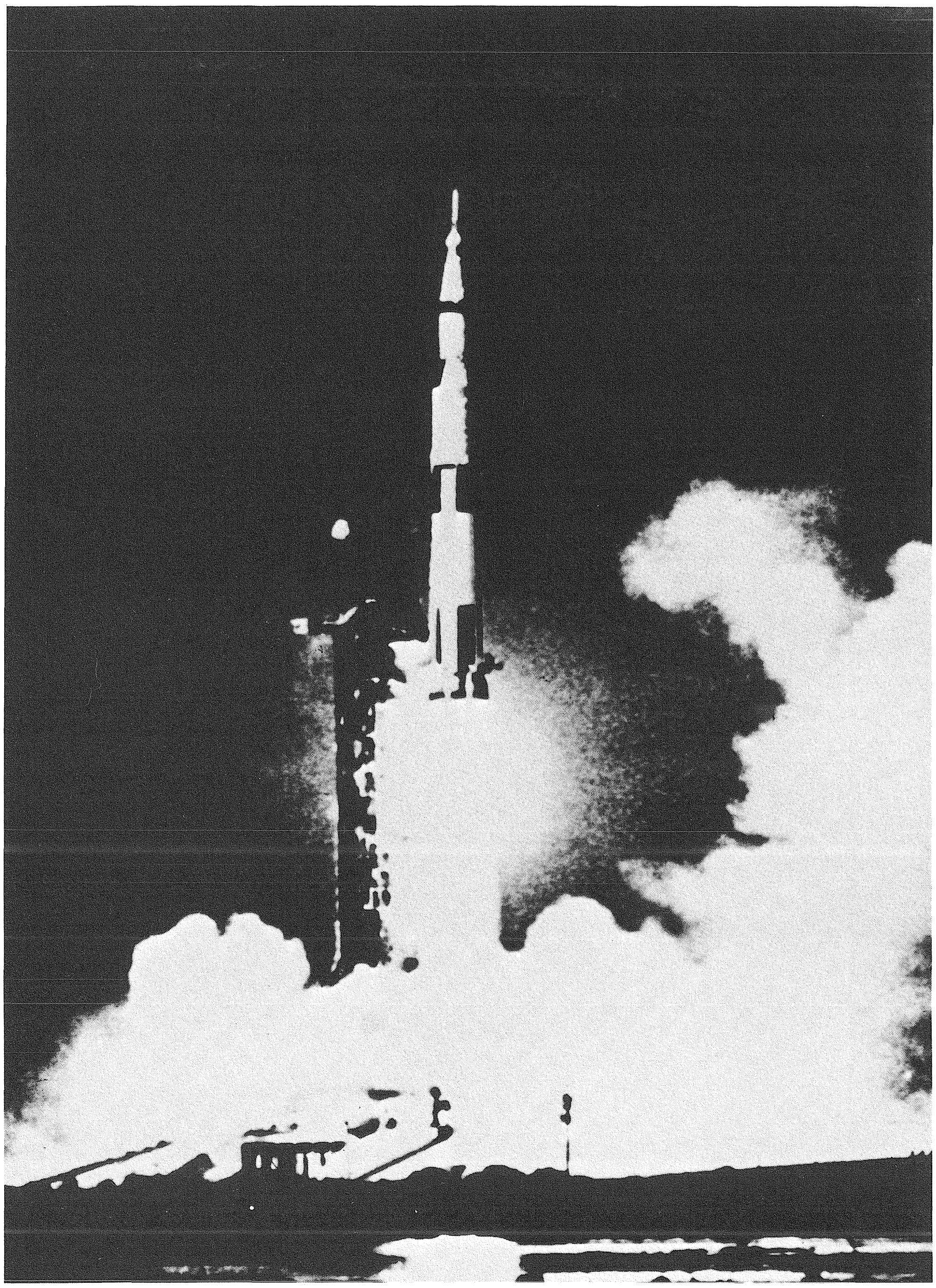
That new direction is PCM (Pulse Code Modulation) and it represents a breakthrough in the audio field that will affect the entire industry today.

With the PCM, the sound level and frequency are represented by a binary numbering system in which only "1" and "0" are used, and then recorded as equal-amplitude digital pulses. In playback, all that has to be done is to discriminate between this presence and absence of signals. The quality of recording and playback is thus not governed by the performance of the tape used or the heads. Because of this advantage, PCM is often referred to as the ideal recording system.

By introducing PCM, Sony's technology has left the old boundaries of audio fidelity and crossed over into a new field of investigation that has just begun to reveal its true potentials.

Chapter 1 The Attraction of PCM

The distinctly audible achievements ahead
as the conventional methods
approach their limits



1. What is PCM?

PCM is an acronym for Pulse Code Modulation, originally devised by A. H. Reeves in 1937. In short, this is a system which modulates signals just like AM (Amplitude Modulation) and FM (Frequency Modulation). The sounds or pictures are given "1" (ON) or "0" (OFF) signals; these are encoded and then reproduced by computer processing. There are no effects of distortion or noise interference along the transmission paths or relay points. The transmission characteristics are fundamentally superior.

This is why NASA adopted PCM for space repeater system using satellites. PCM has already been widely applied also for TV relay circuits and telephone multiplexing based on frequency division. Adopting PCM, more than 10,000 different circuits can be transmitted through a single undersea cable simultaneously and with virtually no signal deterioration.

What would happen then if PCM were used for audio?

In a sense, it is possible to use the potential of PCM even more effectively than in communication systems. Sony was one of the first audio companies to turn the spotlight on the superiority of PCM. We have been researching PCM now for many years, and as a result of our endeavors, succeeded in introducing PCM in audio equipment in 1974. Since that announcement, Sony has done more research and revealed more prototypes. One of the outstanding developments is an adaptor-type PCM audio processor directly connectable to a video tape deck. Sony has brought PCM much closer to our everyday lives.

The application of PCM to audio and the establishment of audio technology using PCM were actually started by Sony. The resultant remarkable improvements of the specifications include a substantially improved signal-to-noise ratio, a greatly expanded dynamic range, a wow & flutter below measurable limits and the reduction of distortion. The progress marked by the introduction of PCM can now be compared to the advances and refinements in audio over the last twenty years.

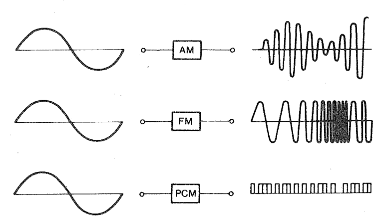


Fig. 1: Modulation Systems and Waveforms

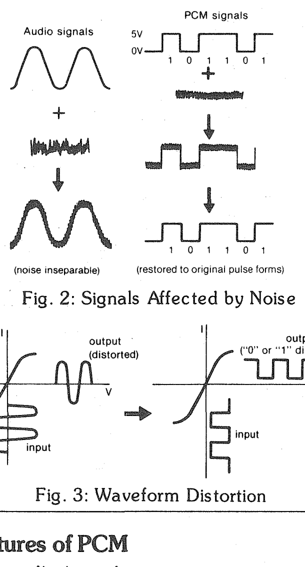


Fig. 2: Signals Affected by Noise

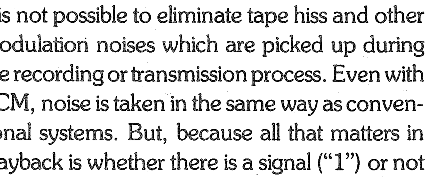


Fig. 3: Waveform Distortion

2. Features of PCM

(1) Immunity to noise

It is not possible to eliminate tape hiss and other modulation noises which are picked up during the recording or transmission process. Even with PCM, noise is taken in the same way as conventional systems. But, because all that matters in playback is whether there is a signal ("1") or not ("0"), noise can be virtually eliminated. As an example, if pulses corresponding to 5V, as in Fig. 2, are set for "1" and 0V for "0", it is possible to compute the original pulse code by correction to 5V (if over 2.5V) and to 0V (if under 2.5V) even if noise is applied to the signal.

(2) Minimal distortion

In magnetic recording, problems are posed by the non-linearity of the tape, heads, and transmission elements, and these are the major factors behind signal distortion. This distortion is the same as noise in that it cannot be removed once generated. Fig. 3 compares analog and digital signals applied to a circuit whose input/output characteristics are non-linear.

As is clear, linearity does not cause any problems with the PCM. The distortion figure, therefore, can be dramatically improved. It is easy to yield a total harmonic distortion of less than 0.03%.

(3) Time compensation possible

In ordinary tape recorders, the tape and the head come into mechanical contact, and variations in their relative speed are detected as wow & flutter. Even with PCM, the signals which are read out during playback are affected by this kind of variation, and the intervals between the signals may be thrown out of sync. However, the signals are not reproduced as they are but are stored once in the memory circuit, and transmitted again in the same time intervals as when they were recorded. Because of this, there is no wow & flutter, speed deviation, or level fluctuations—or, at least, they are theoretically zero.

In the past, problems associated with mechanical factors have thwarted improvements to the sound quality. In this area, however, PCM has achieved nothing less than a breakthrough.

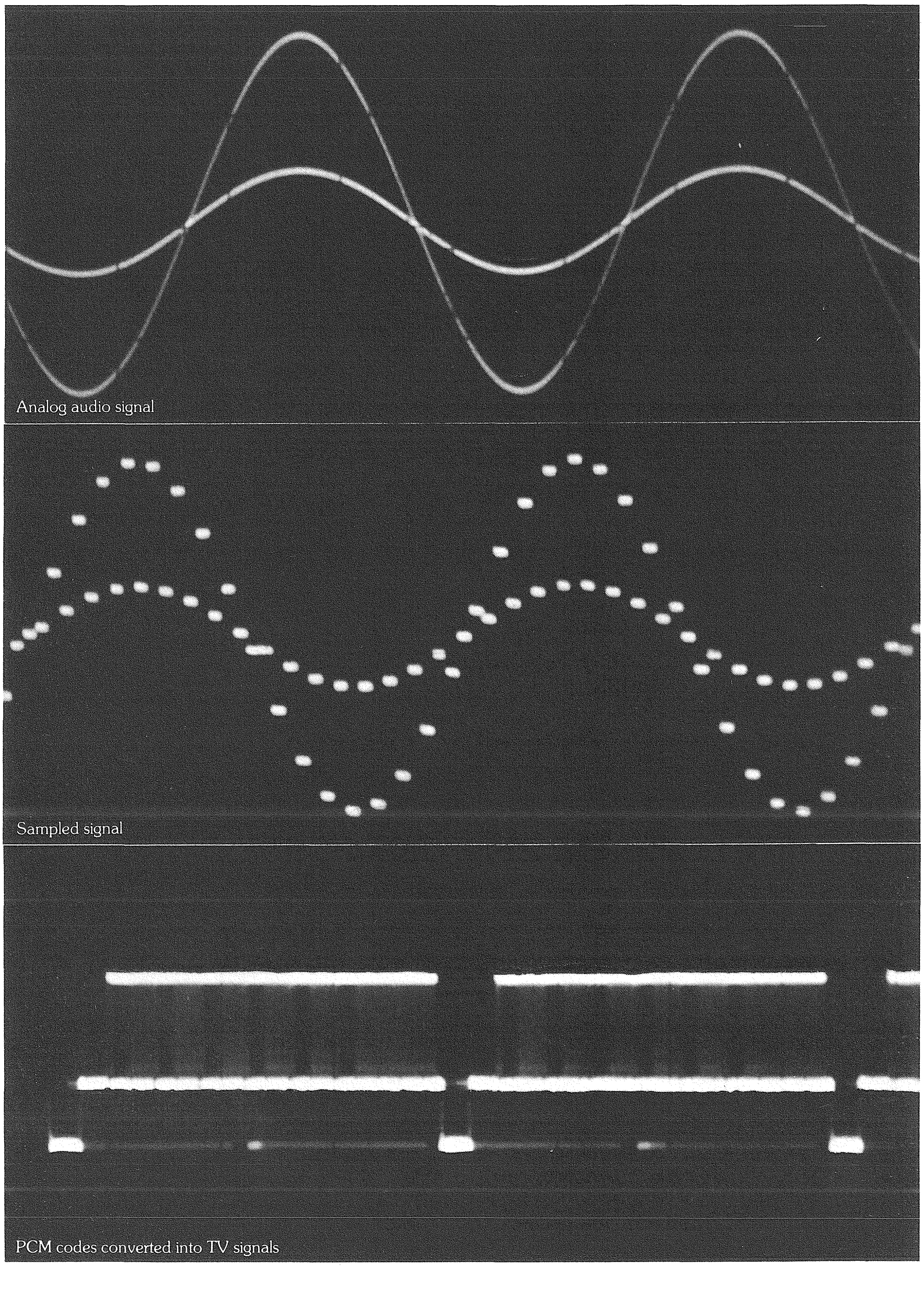
(4) Wide dynamic range

PCM characteristics are determined by the number of pulse codes per unit time. With the Sony PCM-1 digital audio processor, the number of pulse codes per second is approximately 1.4 million, and a wide dynamic range of 85dB has been achieved. Moreover, it is theoretically easy to improve on this since the dynamic range becomes wider in proportion to the increase in the code number.

In conventional recording systems, it was hard to hope for ultra-low-range reproduction. With PCM, great progress has been achieved in this respect, too. DC playback is possible. In addition, PCM recorded tapes show no signal deterioration even when repeatedly copied. There are also numerous other advantages inherent only in PCM. Detailed explanation of why this almost ideal performance can be obtained follows in the next chapter.

Chapter 2 PCM Operation

The sound quality
of direct-to-disc technique
and full adaptability to replace analog systems



Analog audio signal

Sampled signal

PCM codes converted into TV signals

1. Analog and digital

Before we discuss the principles of PCM, we must understand the difference between "analog" and "digital".

"Analog" as an electrical term has the meaning of "continuous". Voltage is a good example. When increasing from 1V to 2V, it goes up continuously from 1.1V, 1.2V,..... to 1.9V, and then 2V.

In contrast, the minimum value of the digital system of counting is fixed and quantity is expressed in integers. Say you are counting pencils. You are counting 1, 2, 3, ...and the minimum unit is 1. There is no such quantity as 1.1 pencils. In other words, the values which express quantity within a given range are finite. PCM is a system of converting analog quantities into digital and then into binary codes using combinations of "1" (ON) and "0" (OFF). "1" or "0" in this case is known as bit (binary digit).

2. All about bits

Bit is an abbreviation for binary digit. It is the smallest unit that can have meaning in binary notation and is usually marked as "1" or "0".

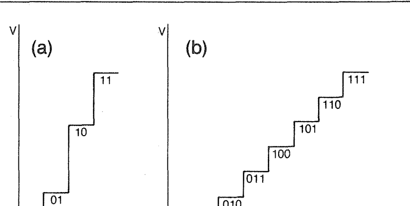


Fig. 4: Bit Number and Code Combinations

In Fig. 4-(a), voltage increase along with the time is expressed by 2 bits. Four voltage levels can be indicated when 2 bits are used. If codes corresponding to the interim voltages are required, the number of bits which configure the codes can be increased. Using 3 bits instead of 2, the four possible code combinations change to eight. Provided that the range to be expressed is set, if the number of bits is increased, the corresponding voltage can be expressed more accurately and the analysis capacity can be increased. Listed below are examples of relations between the bit number and the corresponding expressions.

1 bit: $2^1 = 2$ expressions
0, 1
2 bits: $2^2 = 4$ expressions
00, 01, 10, 11
3 bits: $2^3 = 8$ expressions
000, 001, 010, 011, 100, 101, 110, 111

As you see now, the bit number, in other words, refers to the number of digit. The greater the bit number, the greater the content which can be expressed. Simply saying, an additional bit doubles the number of expressions and, in case of digital audio, offers extra 6dB of dynamic range. Sony's PCM-1 digital audio processor, for example, employs 14-bit system, and incoming audio signals are thereby analyzed and expressed in 16,380 different ways, consequently assuring a dynamic range of 85dB.

3. PCM circuitry

(1) A/D conversion

a) Sampling

Audio signals provide continuous values with regard to their amplitude. At the same time, they are continuous values in time as shown in Fig. 6-(a). These 2 elements are broken down into discrete values in PCM, and this concept of handling discontinuous form of signals is the very core of PCM which makes it possible to realize unprecedentedly high performance.

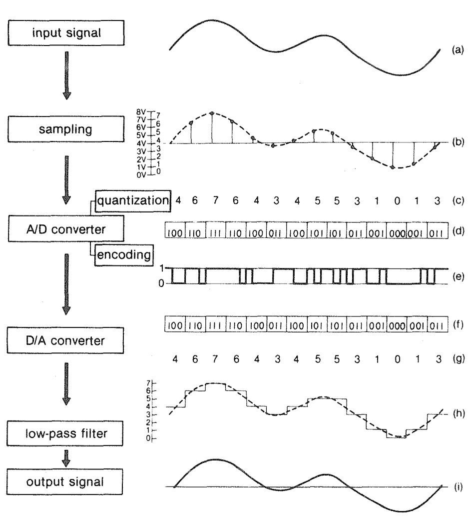


Fig. 5: PCM Encode/Decode Process

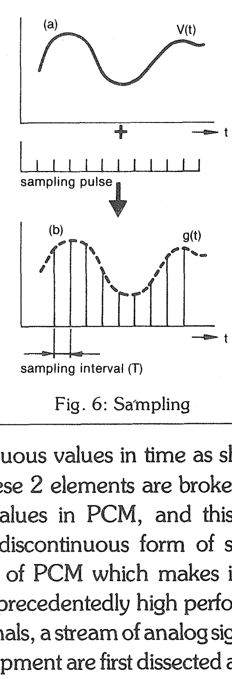


Fig. 6: Sampling

Audio signals, a stream of analog signals, fed to a PCM equipment are first dissected and made into discrete values at fixed time intervals as in Fig. 6-(b). This process is called sampling. As shown in Fig. 7-(a), it takes a voltage of V1 (=4V) at time T1 to transmit the sampled signal, a voltage of V2 (=5V) at time T2, and a voltage of Vn at time Tn.

However, it is likely to happen that signals including high frequencies are transmitted in the sampling time width as in Fig. 7-(b). In other words, sampling interval must be shortended in order to be accurate. It is generally accepted that the sampling speed (frequency) must be twice that of the fastest changing component (the highest frequency) of the input signals. Shannon's sampling theorem states that if the highest frequency of signals to be transmitted is less than half the sampling frequency, the sampled signals will be perfectly reproduced. Frequency response up to 20kHz is generally sufficient for music reproduction. The sampling frequency, therefore, is acceptable if it is double at 40kHz. A PCM standard of 44.056kHz has sufficient leeway.

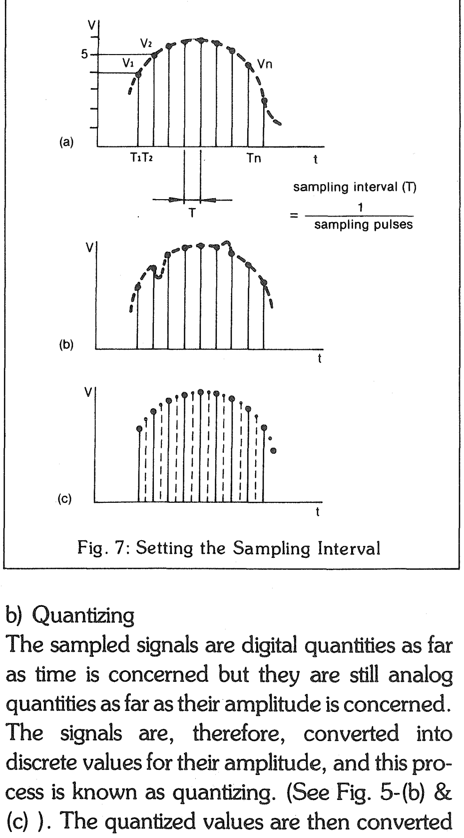


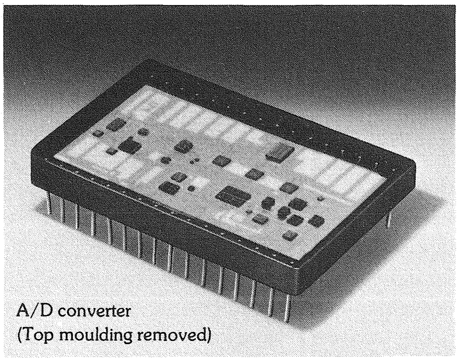
Fig. 7: Setting the Sampling Interval

b) Quantizing

The sampled signals are digital quantities as far as time is concerned but they are still analog quantities as far as their amplitude is concerned. The signals are, therefore, converted into discrete values for their amplitude, and this process is known as quantizing. (See Fig. 5-(b) & (c)). The quantized values are then converted into codes determined by "1" or "0", as explained in Chapter II-2.

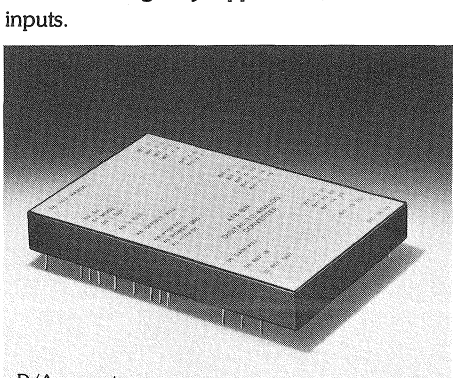
Eight-bit quantization is acceptable when sound such as that used with the telephone is involved since the dynamic range does not have to be that wide. However, in general, about 60dB or wider dynamic range is said to be required for hi-fi applications. This corresponds to 10 bits in PCM. The PCM-1 adopts 14-bit quantization for a wide dynamic range of 85dB.

The sampling and quantizing are quite complex electronic operation. Nowadays, this task of digitalizing audio signals is performed usually by an IC known as an A/D (analog-to-digital) converter. The digital signals are then encoded serially in ON/OFF patterns as in Fig. 5-(e).


A/D converter
(Top moulding removed)

(2) D/A conversion

For reproduction, the digital signals must first be converted back into analog signals. This reverse operation is performed by a D/A (digital-to-analog) converter. (See Fig. 5-(f) to (i).) The signals passing through the D/A converter and the following low-pass filter are restored to the same continuous analog waveforms as those which were originally supplied to the recorder as inputs.


D/A converter

(3) Dropout compensation

Dropouts on magnetic tapes are mainly caused by scratches and dusts. In playback, these dropouts become code errors, and if reproduced without being restored, reliability will be critically downgraded in spite of the great advance in performance specifications. Dropout compensation, indeed, is one of the most important aspects in designing digital audio equipment.

The PCM-1 employs CRCC (Cyclic Redundancy Check Code) to provide for such defects. By applying appropriate bits of CRCC, it is possible to detect the erroneous data at almost 100% probability. Then, with the aid of a special configuration of code format called interleaving, the code error is substituted by the average value of those data before and after the error. This method is known as a liner interpolation.

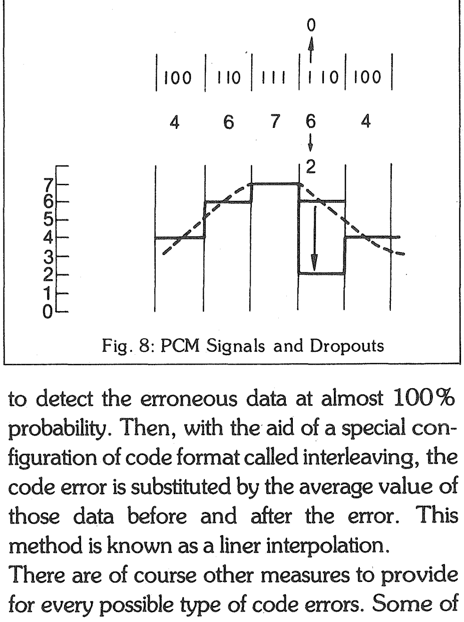
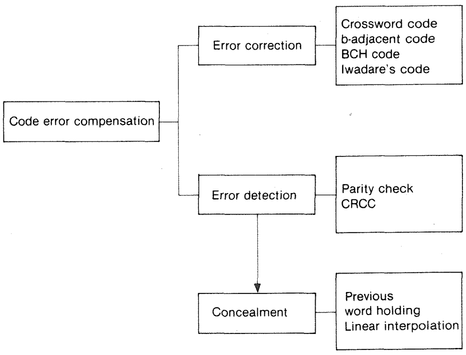


Fig. 8: PCM Signals and Dropouts

There are of course other measures to provide for every possible type of code errors. Some of them are more effective and sophisticated than the method mentioned above. Using the CROSSWORD code specially developed for the Sony professional PCM equipment, for example, errors can perfectly be corrected, instead of being interpolated, in order to achieve the ultimate in high-fidelity sound reproduction. But in ordinary commercial or home applications, the interpolation method is quite sufficient.



* Listed here are only the typical methods. According to whether the errors are of the burst or randam type, an appropriate code or a combination of codes and interleaving are selected for utmost effectiveness.

Fig. 10: Code Error Compensation Scheme

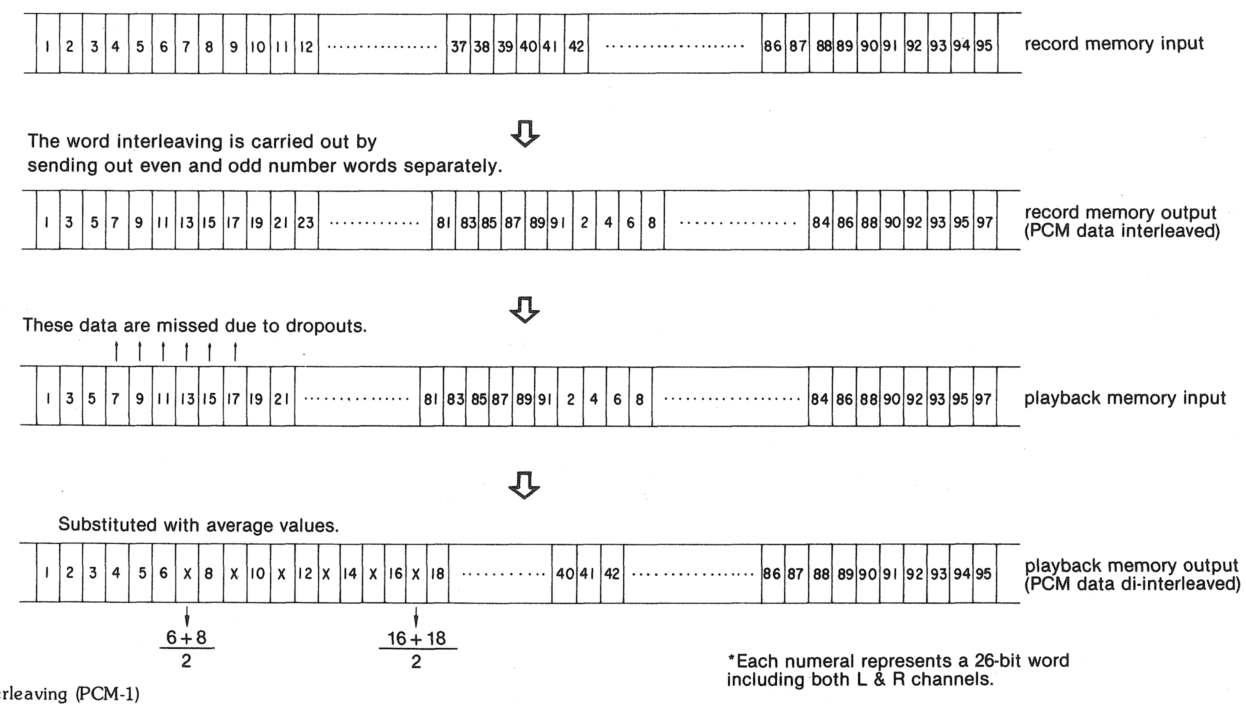


Fig. 9: Interleaving (PCM-1)

(4) PCM recording

With the PCM-1, sampling is performed 44,056 times a second and the quantized code each time is 32 bits. As you now see, the sampling and quantization result in an enormous quantity of binary codes of 44,056 x 32=1,409,792 bits/second. In other word, more than 1.4 million data are fed out every second. In terms of frequencies, this is equivalent to 1.4MHz.

It is, of course, not possible for ordinary tape decks to record such a high frequency, and so a video deck, Sony's Betamax or U-matic videocassette recorder, is required.

TV signals have a frequency bandwidth extending over 4MHz and, therefore, the video deck can certainly record the PCM codes, which require less than a half the frequency bandwidth of that for TV signals. Naturally, the PCM codes must be superimposed on TV signals in order to be recorded on video decks.

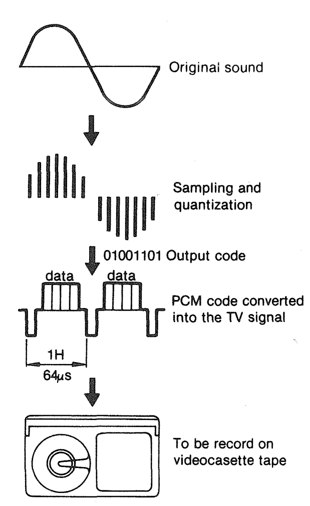
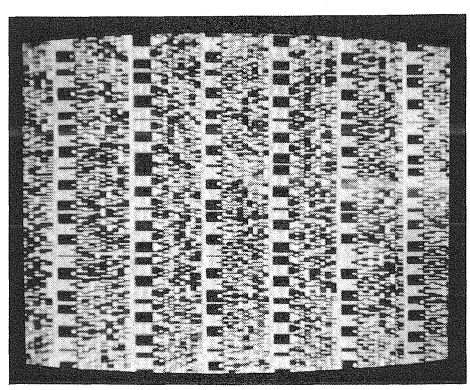
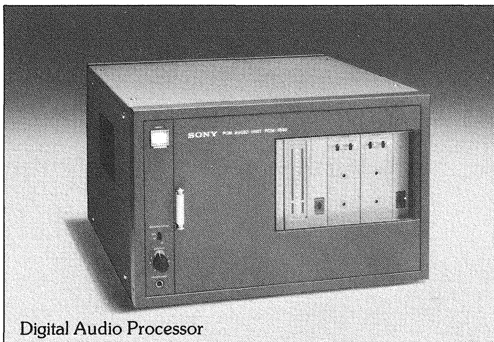


Fig. 11: Conversion into TV Signals

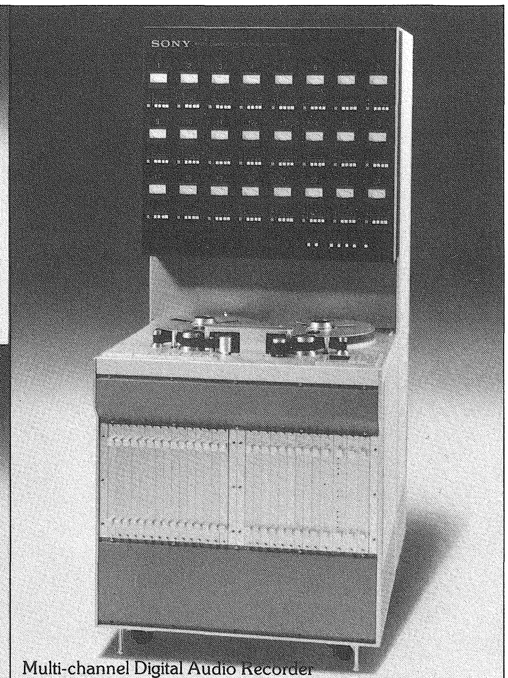

PCM encoded codes on TV signal

Chapter 3 Potentials of PCM

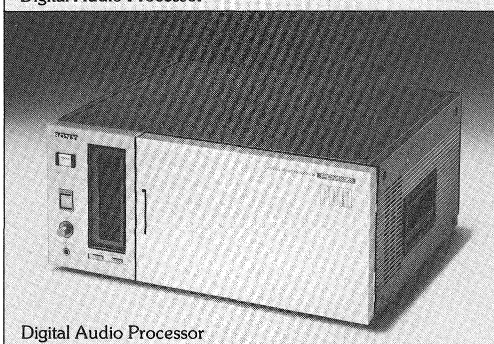
Enormous potential of digital systems
—the very attractive prospects
for audio systems in the future



Digital Audio Processor

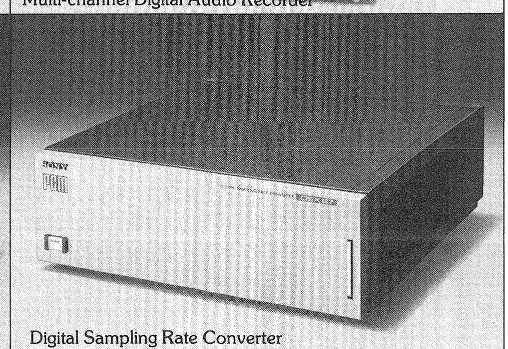


Multi-channel Digital Audio Recorder

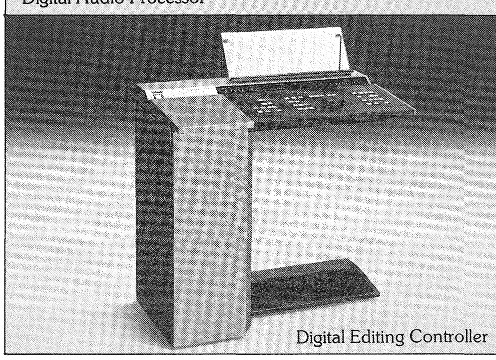


Digital Audio Processor

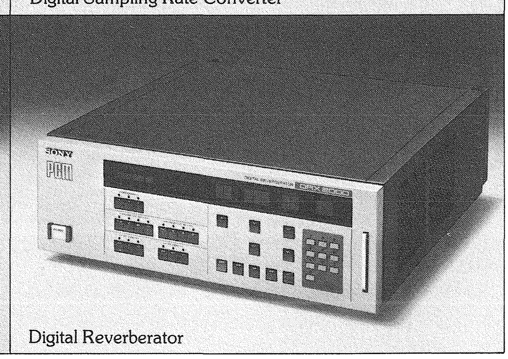
Digital Audio Processor



Digital Sampling Rate Converter



Digital Editing Controller



Digital Reverberator

1. Sony and PCM

Though digital technology has proved especially effective for improving the performance of audio recorders, most of the digital audio recorders previously developed are restricted as exclusively experimental or professional. Why has the practical application of PCM been so slow?

One reason is the problem with the development of circuit elements. Another is the price—even if developed, PCM systems have been considered too expensive to replace analog systems. Sony succeeded in developing a professional-use PCM equipment some years ago. That was the first step toward the introduction of PCM into audio. Nevertheless, the history of the truly practical digital audio recorder started a couple of years later. To meet the demands for PCM systems which can be used in the home, attention was focused by Sony on video decks which began to penetrate the market rapidly at that time. The company succeeded—the first in the world, in fact—in developing the PCM-1 digital audio processor, an adapter which could be hooked up to the video deck to provide PCM recording and playback.

Following on the heels of the PCM-1 came the PCM-1600 for professional application, and this spread widely among some of the world's leading record companies and recording studios. Sony's reputation as a digital audio manufacturer was even further enhanced by the announcement at the 1978 AES Convention of the 24-channel stationary head PCM recorder. Sony won high praise from the industry.

Digital techniques are also making inroads into records. Sony has also developed an optical digital audio disc system which uses a laser beam for signal detection. Furthermore, by developing digital mixers, reverberators, and various other peripheral equipment, Sony aims to develop a total digital audio system.

Alongside its efforts to develop new products, Sony is also contributing a great deal to the standardization of specifications in the industry. PCM standards for consumer applications will be decided any day now. Sony is also urging other companies in the industry to consider its optical readout disc system in view of its technical excellence.

2. Future of PCM

The possibilities of PCM are unlimited for music enjoyment. It follows that we can always lend our ears to PCM sound, which is undoubtedly closest to the live sound. Consider, for instance, that PCM music tapes have become available on the market. Of course, what we actually buy will be a tape which has been copied many times over from a master tape. But with PCM tapes, the quality does not deteriorate in principle however many copies are made and at whatever speed. When we listen to such a tape, it is as though the only device between us and the performers is a single microphone to pick up sounds. Once PCM broadcasts and PCM tuners are realized, concert hall sound will be transmitted accurately via PCM and thus the sound heard in the home will be exactly the same as the original performance. Since the summer of 1978, PCM-recorded FM broadcasts have been aired several times experimentally in Japan. Even though ordinary FM tuners and transmitters were used for reception and transmission, the high sound quality of the PCM-recorded sources made a deep and lasting impression on listeners.

From a different perspective, PCM will make it easy to create one's own sound. The output of the PCM recorder remains exact replica of the original sound waveform, and how it is changed or characterized can solely depend on individual preference. The preferred tone, reverberation and other effects can be freely added to make an individual sound and increase the enjoyment of sound creativity. A new dimension to Hi-Fi may well be added.

In the field of semiconductor engineering, advanced ICs and LSIs have been successively developed recently. And, before long, these will surely bring the PCM systems down into a price range within everyone's reach. When we consider this fact and the numerous merits of digital techniques, the day when audio systems go totally digital cannot be very far away.

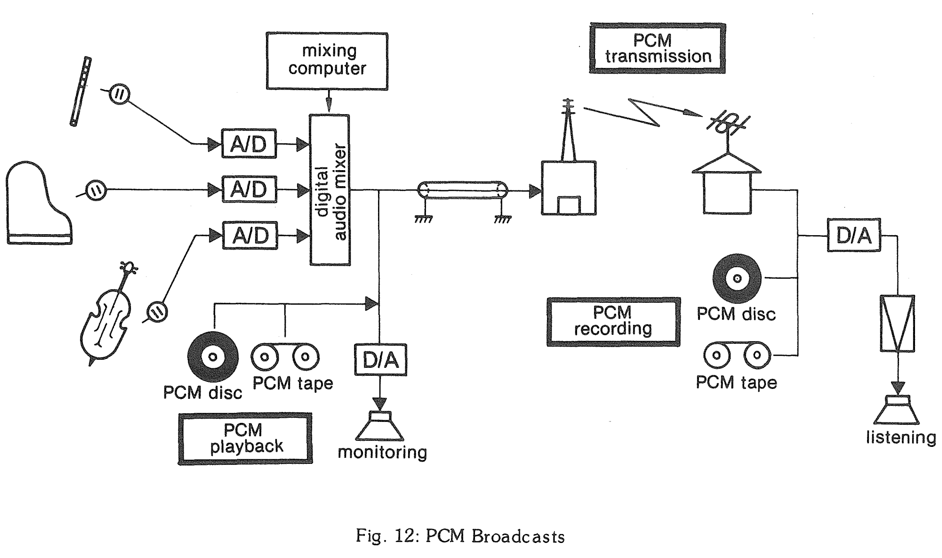


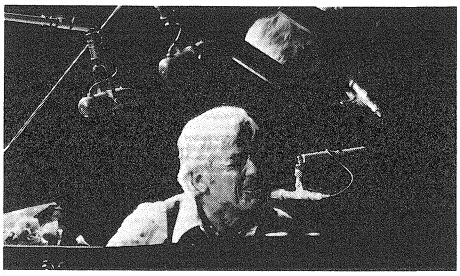
Fig. 12: PCM Broadcasts

Clock pulse
In sync system digital equipment, fixed period pulses are used to gate information and to ensure the correct operation of all the circuits. These pulses are also called clock pulses. The circuits are actuated when these pulse commands are given. The operation speed of circuit, therefore, is determined by these pulses.

Encoding
Encoding denotes the conversion of quantized amplitudes into pulse codes. A binary code is most commonly used. In actual operation, the quantizing and encoding are performed simultaneously by the analog-to-digital converter.

Quantizing and quantizing noise
Quantizing refer to the process whereby the range of values of the sampled amplitude of an analog signal is divided into a finite number of subranges, each represented by an assigned (quantized) value, and substituted for the digital signals. There is a slight error between the original signal and the quantized value. This is heard as noise and is known as quantizing noise, or quantization distortion. The signal-to-noise ratio of the PCM and dynamic range are determined by this noise.

Dynamic range
In PCM, dynamic range is expressed as the ratio between the maximum acceptable signal input and the quantizing noise. With linear quantizing, the range is virtually proportional to the number of bits, and with non-linear quantizing, the obtainable dynamic range is much wider with the same number of bits.



Dropout
This refers to distinct but temporary gaps in the signal level during the playback of recorded data caused by marks or dust on the surface of the tape. In PCM systems, it results in a direct code error. When burst-formation type of dropouts occur, the errors are scattered and corrected by bit interleaving.

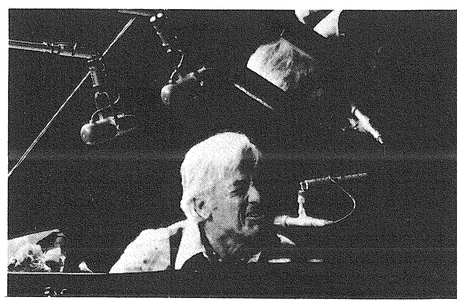
Jitter
This term denotes the instability of a signal in either its amplitude, its phase, or both. It is generated when signals are played back on a tape recorder with wow and flutter, whereby noise is added to the signals. In PCM systems, it is the cause of code errors along with dropouts in the tape medium.

Code error
This refers to an erroneous 1 or 0 in the encoded signals. it is caused by dropouts, jitter, noise, etc. If the recording is played back with these code errors, it will come through as a clicking sound. In order to compensate for these errors, a number of methods are used in the circuitry: previous word holding, linear interpolation and error correction code.

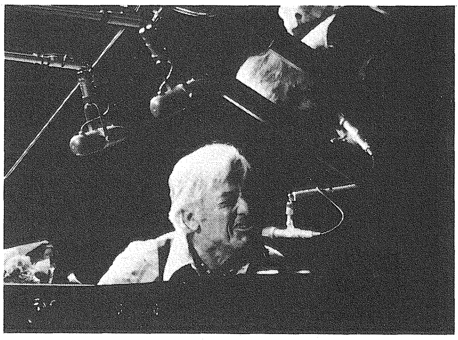
Parity check
This is a method employed to detect word (binary) errors. An extra bit is added to a number, and if even parity is used, the sum of all the 1's in the word and its corresponding parity is always even; if odd parity is used, the sum of the 1's and the parity bit is always odd. With a parity check, the total number of binary 1's (or 0's) is always even or odd. Either an even-parity or odd-parity check can be conducted.

Previous word holding
When a word error is detected, the erroneous word is replaced by the preceding word so that there is no audible difference.

Linear interpolation
When a word error has been detected, the average value of the preceding and succeeding word is substituted for the erroneous word. The error is thus compensated for so that there is no audible difference. The compensation capability of this interpolation is superior to that of the previous word holding.



Interleaving
Interleaving is a method to disperse dropout errors by changing the sequence of information words in recording. Restored to the original order in playback, the erroneous word is invariably placed between the correct words, and thus average-value interpolation, etc. can be easily performed. With the PCM-1, the word interleaving is carried out by sending out even and odd number words separately.



Analog
This word is used in contrast with digital. Analog quantities denote quantities that change continuously like the temperature or voltages. Ordinary audio signals are called analog signals, and VU meters have an "analogy" with the variation in these analog signals which is indicated by the deflection of the pointer. This type of meter can be called an analog display meter.

Digital
This word originates from "digit" meaning finger and it is used in contrast with analog. A digital quantity denotes a quantity by which a variable amount is discontinuously encoded (numerical values). In other words, a digital quantity is an encoded analog quantity. The word also denotes numbers expressed in digits and in a certain scale of notation to represent all the variables that occur in a problem.

Bit
This is an abbreviation for binary digit. It is a unit of information equal to one binary decision, so that 1 digit is referred to as 1 bit. Three bits refer to a 3-digit code. With n bits, it is possible to indicate and subdivide 2^n types of information.

Sampling theorem
Sampling refer to the extraction of the amplitude of an analog signal at regular time intervals.
The sampling theorem (developed by Shannon) states that the sampling rate must be twice the highest frequency component.

Sampling and holding
This is a circuit which is used in an analog-to-digital converter to make a measurement of a analog signal (sampling) and to increase the duration of that signal (holding). Holding is necessary as a fixed period of time is required to convert the analog signal into a digital signal.

Word
A group of bits that express a single quantizing value is called a word. It usually comprises at least 9 bits. In some cases, the sampled values of the left and right channels are lumped together and called a word. The redundancy bit may also be included and the resulting combination is called a word.

PCM
This acronym means pulse code modulation. It refers to a system of modulation whereby the ordinary analog signals like audio signals are replaced by pulses, their amplitudes are turned into digital codes, and the resulting signals are transmitted or recorded. The signals are all expressed in binary digits, 1 for every pulse and 0 for every non-pulse, and so the signals are resistant to noise, and distortion can be kept down to a very low level right up to high frequencies without being dependent on the frequency.

Recording density
Recording density denotes the number of bits which can be recorded per given unit length on the data tracks of a tape or disc. The unit BPI (bit per inch) is often used to denote how many bits can be written along an inch of magnetic tape.

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PCM