PCM Tutorial

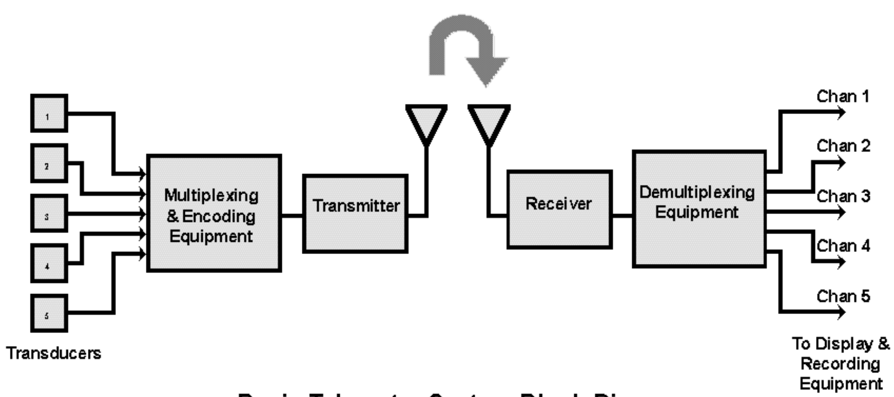
Introduction

The process of telemetry starts with data obtained from various sensors called transducers. These are the devices which translate physical data such as temperature and pressure into electrical signals.

In the simplest telemetry system, which monitors only one such transducer, the signal from the transducer may be applied directly to a radio transmitter (or sent directly over a cable). At the other end of the radio link, the signal goes directly from the receiver to a suitable display device or recorder. For example, if the transducer is monitoring temperature, the information may be displayed on a meter calibrated directly in degrees.

A telemetry system is rarely called upon to monitor only one transducer. In general, the number of data sources is great and a separate transmission link for each one is not practical. Therefore, most telemetry systems combine the signals from a number of transducers into a single composite signal for transmission over one radio link. The process of combining these signals is called multiplexing, and each individual signal occupies a channel in the multiplex system.

At the receiving end, the channels are separated by a process called demultiplexing and routed to the various display or recording devices. The accompanying figure shows a basic telemetry system block diagram.



Basic Telemetry System Block Diagram

Multiplexing

Many individual signals are combined in the multiplexing process. However, they cannot occupy the same portion of the transmission link's frequency spectrum at the same time. Otherwise, there would be no means of separating them at the receiver. Channel separation within a single transmission link is accomplished by either of two basic methods: by separating the channels in frequency or by separating the channels in time.

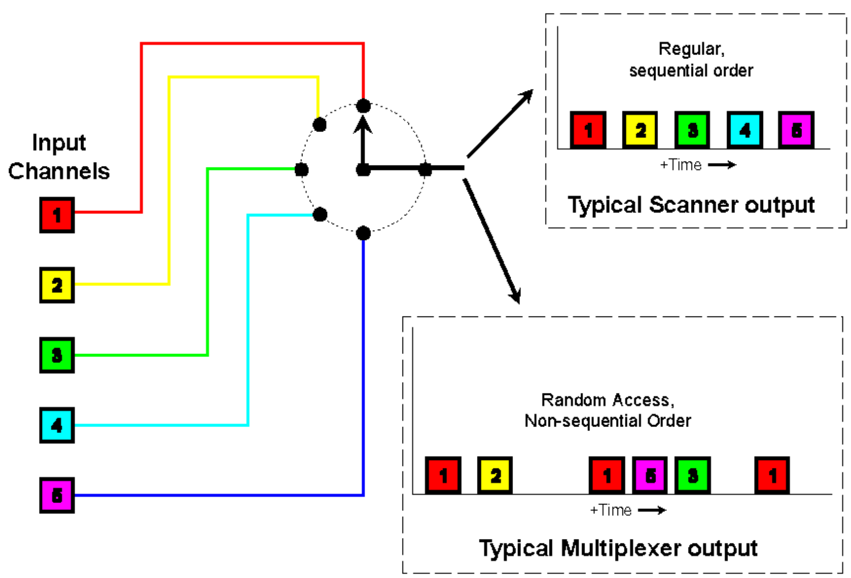
All channels may share the transmission link all the time, with each channel occupying a different portion of the frequency spectrum. This type of multiplexing is called frequency division multiplexing or FDM. Frequency-sensitive filters can then separate the bands at any desired point.

The other type of multiplexing allows all channels to share the transmission link, with each channel connected to the link for only a short time. This type of multiplexing is called time division multiplexing or TDM.

While each type of multiplexing has its particular advantages and disadvantages, the method of time division multiplexing is most commonly used in telemetry.

Time Division Multiplexing

All channels of a time-division multiplex system use the same portion of the transmission links' frequency spectrum - but not at the same time. Each channel is sampled in a regular sequence by a multiplexer (or less commonly by a scanner) as shown in this figure. When all channels have been sampled, the sequence starts over with the first channel. Thus samples from a particular channel are interleaved in time between samples from all other channels.



Time Division Multiplexing Techniques

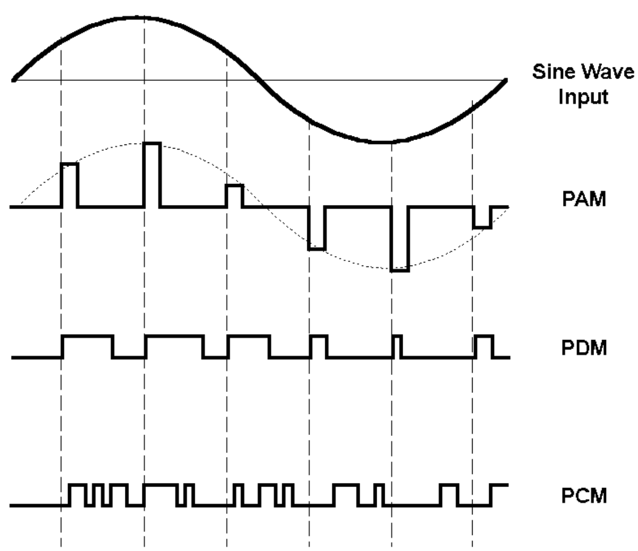
Since no channel is monitored continuously in a time-division multiplex system, the sampling must be rapid enough so the signal amplitude in a particular channel does not change too much between samples. Theoretical studies based on idealized conditions have shown that no information is lost if the sampling rate is at least twice the highest frequency component in the sampled signal. This sampling rate, however, is based on "perfect" electrical characteristics, which cannot actually be attained in electronic equipment. Practical telemetry systems use sampling rates much higher in order to preserve all the information in the original signal without the need for unnecessarily complex circuitry.

The per-channel sample rate in a typical telemetry system is set about 5 times the highest frequency component in the sampled signal. For example, if the highest frequency component in a particular channel is 40 Hz, the channel is sampled about 200 samples per second or SPS. If there are 8 such channels in a system, the multiplexer must operate at 1,600 SPS.

At the receiving end of the system, a decommutator operating in the same sequence as the multiplexer, distributes the parts of the multiplexed signal to the proper output channels. Since a time-division multiplex system is based on precise timing, it is vitally important the decommutator be synchronized exactly with the information coming from the multiplexer. Otherwise, information on fluid flow, for example, might be misinterpreted as temperature information.

Pulse Modulation Techniques

The output from a multiplexer is pulse-amplitude modulated or PAM. That is, the height of each pulse represents the instantaneous amplitude of the sampled waveform, as shown in this figure. A smooth curve drawn through the tops of the pulses would reconstruct the original waveform as shown in the figure. PAM is the simplest modulation technique for use with time-division multiplexing. It is often used where little interference is encountered, or where the accuracy requirements are not too stringent.



Pulse Modulation Techniques

The disadvantage of PAM is that any noise "riding" on the signal changes the pulse height, thereby introducing distortion. One solution to this problem is to use pulse-duration modulation or PDM, in which the PAM signal goes to a keyer which produces new pulses of uniform height but varying length. The information is then carried by the pulse length, or duration, rather than by the pulse height as in PAM. PDM is somewhat less susceptible to noise than PAM. However, any distortion of the pulse shape may change the apparent pulse duration, thereby producing a distorted output signal.

Pulse Code Modulation or PCM offers a method of overcoming some of the disadvantages of other types of pulse modulation. In PCM, the instantaneous amplitude of the sampled signal is represented by a coded arrangement of binary digits or bits resulting in a series of pulses and spaces. All pulses are the same height and the same shape. Therefore, it is only necessary for the receiving equipment to detect the presence or absence of a pulse. A distorted pulse does not degrade the signal as long as the pulse can still be recognized. Thus, PCM is less sensitive to noise than either PAM or PDM and is easily implemented using modern electronic technology.

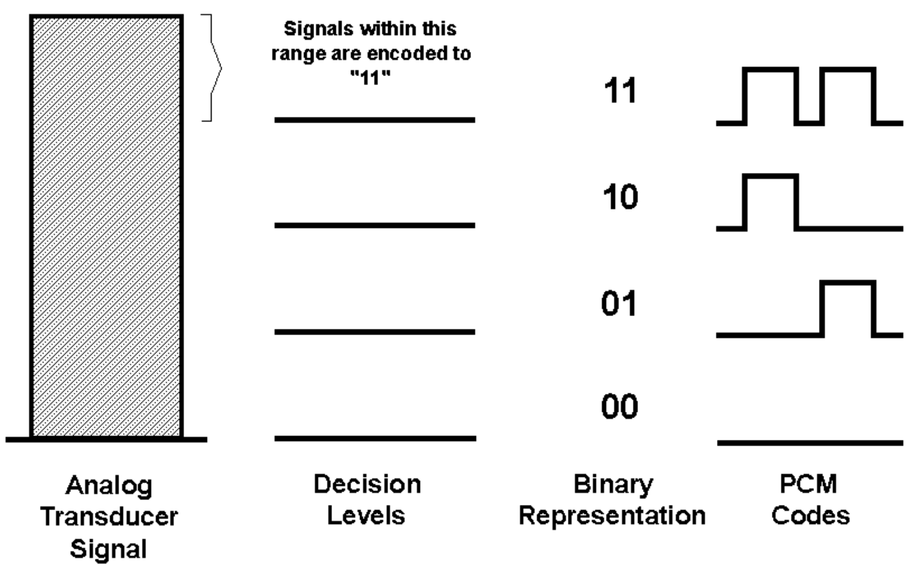
Telemetry Standards

Telemetry Standards are contained in the latest version of IRIG-106, Telemetry Standards. These standards provide the necessary criteria on which to base equipment design and modification. The ultimate purpose is to ensure efficient spectrum utilization, interference-free operation, interoperability between ranges, and compatibility of range user equipment with the ranges.

The Basics of PCM

As previously mentioned, considerable immunity to noise and other transmission difficulties can be achieved if Pulse Code Modulation or PCM techniques are used in telemetry systems. The multiplexed signal is coded as a series of identical pulses and spaces. The receiving equipment need only make a simple "yes or no" decision as to the presence or absence of a pulse at a particular time.

Before it is coded for transmission, the analog signal is sampled just as in other forms of pulse modulation. The range of possible pulse heights, from zero to full scale, is then divided into discrete steps so that each step can be represented by a particular arrangement of binary pulses and spaces, as shown in the figure "Converting an Analog Signal to a Digital Signal". This coded arrangement of binary pulses is the PCM signal.



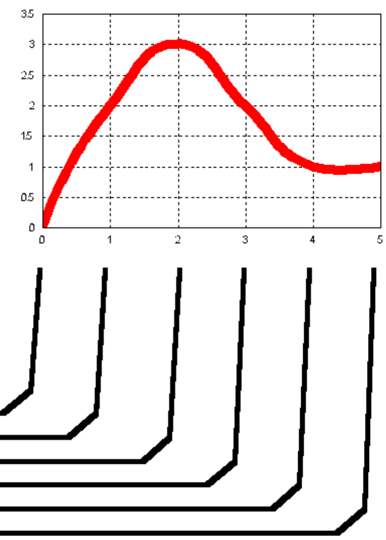
Converting an Analog Signal to a Digital Code

Since the PCM signal is an approximation of the original signal to the nearest discrete level at each sampling instant, the reconstructed waveform at the receiving end of the system is distorted. It is a quantized approximation of the original signal. The disparity between the original waveform and the quantized digital representation is termed quantizing noise. It results from the difference between the signal amplitude and the nearest quantizing level represented by the code at each sampling instant.

In binary systems, the number of quantizing levels possible in any given code is 2^n where n is the number of digits in the code. Thus in a two-bit code, there are 2^2, or 4, discrete codes. This type of coding allows the system to assume four amplitude levels above zero.

In an eight-bit code, which is frequently used in telemetry, 256 discrete codes are possible. In a ten-bit system, 1,024 numbers are possible, allowing the decimal numbers from 0 through 1,023 to be represented. A twelve-bit system provides numbers from 0 through 4,095 to be represented. The figure "ADC Conversion and Resolution" illustrates the relationship between the number of bits used in the code known as the bits per word, and the resolution of the data provided by the system.

A Typical Analog Input Signal is Sampled at 1 second intervals. The signal is then converted to digital format by 8-bit, 10-bit, and 12-bit ADCs. The resulting PCM output codes are shown below in decimal format.



The resolution available from the various ADCs is listed below each column.

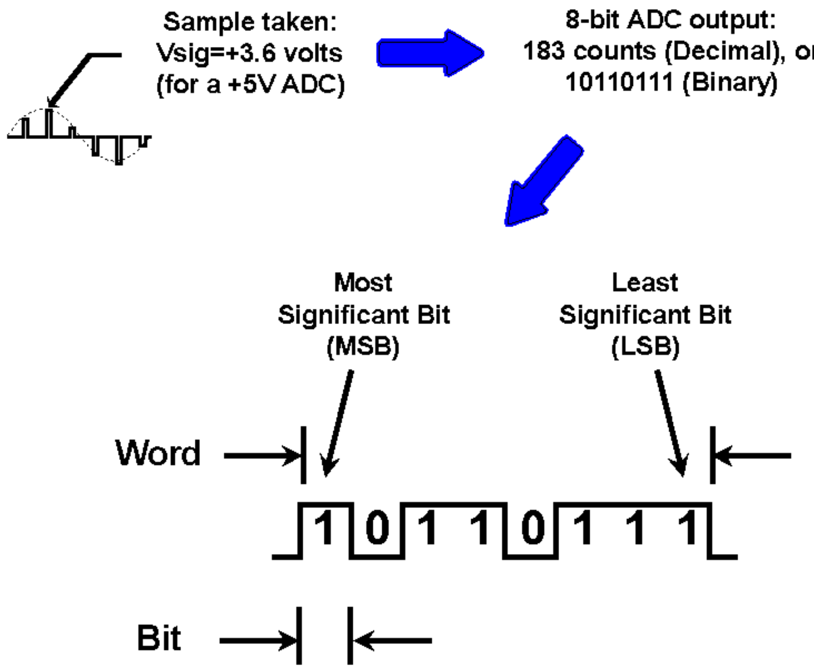
Time	Voltage	8-bit Code (Decimal)	10-bit Code (Decimal)	12-bit Code (Decimal)
0	0 Volt	3	12	48
1	2 Volts	103	412	1648
2	3 Volts	153	612	2448
3	2 Volts	103	412	1648
4	1 Volt	53	212	848
5	1 Volt	53	212	848
	Resolution:	.4%	.1%	.025%

ADC Conversion and Resolution

Thus, a PCM signal is always an approximation of the original signal. How good an approximation is primarily a matter of economics. When the quantization is made as fine as that possible with a 12-bit code, the resolution of the PCM system exceeds that of many laboratory instruments, and it is considerably above that attained in most types of FM data transmission.

PCM Word Format

The series of pulses which represents a single sample from a single channel is called a word as shown in this figure. One complete sampling cycle, including a word from each channel, is called a minor frame. Each pulse or space in the word is called a bit (from "binary digit"). The total number of bits in each word determines the resolution which is a measure of the number of different discrete signal levels which can be identified and coded.



PCM Word Format example

The actual encoding occurs in an analog-to-digital converter or ADC. One such ADC encoding process is known as successive approximation. It is done by comparing the amplitude of the analog sample to a series of precision voltages, one for each digit in the code. The first voltage is 50% of maximum amplitude, the second is 25%, the third is 12.5%, etc. Thus, each comparison voltage is half the preceding one.

When the sample is compared to the first voltage, the first digit of the code is determined. If the sample is greater than 50% of maximum, logic circuits generate a binary ONE in the Most Significant Bit or MSB position. If it is less than 50%, the first digit is a ZERO.

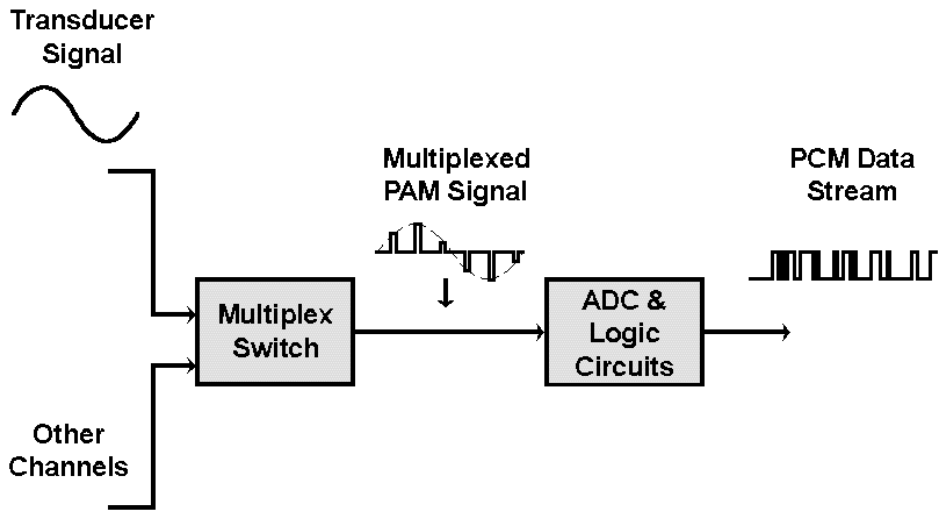
For example, suppose an 8-digit code is being used, and the sample of interest is +3.6 volts which represents 72% of maximum amplitude. Since this is greater than the first comparison voltage, the first digit is ONE. Now the second comparison voltage (25%) is added to the first and the sample is compared again. Since the sample is less than 75%, the second digit is ZERO. The second comparison voltage is then switched out, and the third (12.5%) is switched in. The sample is then compared against 62.5% of full value. Since the sample exceeds this, the third digit is ONE. The fourth comparison voltage is then switched in, and the sample is compared against 68.75%. The fourth digit is therefore ONE.

This process is continued through the eight comparison voltages, and a binary number is obtained which represents 183 (72% of the full-scale value of 256 levels). The binary number is 10110111.

There are a number of modulation techniques to represent the logic levels ONE and ZERO, such as the presence or absence of a pulse with respect to time, a switch from voltage level to another, etc. Information on the se other modulation techniques can be found in the IRIG standards.

A Basic PCM Encoder

This figure shows a simplified block diagram of a PCM encoder. A number of transducer signals are applied to the input of a multiplexer switch. Signals are sampled in any order and at any rate as defined by the user of the system. Many systems provide programmability in order to select channel sampling conditions.



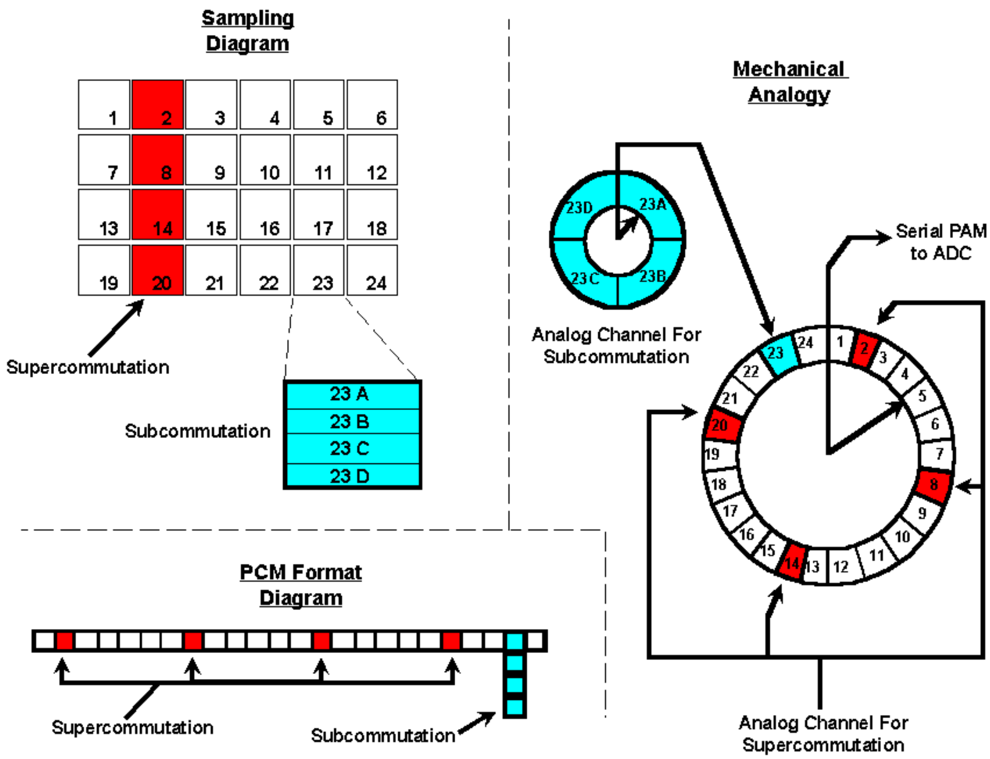
PCM Encoder Simplified Block Diagram

The output of the multiplexer switch is a PAM signal which carries time-division-multiplexed samples of each input channel. The ADC samples this PAM signal and executes an analog-to-digital conversion on each sample. The output from the ADC is serialized and formatted into a PCM wavetrain in accordance with the applicable telemetry standards (IRIG-106).

PCM Format Diagram

The PCM Format Diagram is the "road map" which describes the order and sequence of data contained within a PCM data stream. Channels appear in the data stream at different rates and different orders, all depending on the requirements of the mission. Over the years, common "shorthand" has been developed to present this type of information in what is called the PCM Format Diagram.

The accompanying figure illustrates the concept of the PCM Format Diagram. A PCM Encoder can be described in terms of a mechanical analogy -- the rotary switch or commutator. 24 input channels are sampled as the commutator shaft rotates. Channel 2, 8, 14, and 20 are all tied to the same input channel thereby increasing the sample rate of that channel. Channel 23 is coupled to a second-tier commutator operating at a sub-multiple rate of the primary commutator. Thus the data appearing in location 23 actually represents four other input channels, all sampled at a lower sampling rate.



PCM Format Diagram

The Serial PAM output from the commutator can be expressed by a Sampling Diagram. Each commutator position is represented in the diagram to show the order in which the channels are sampled. Again, channels 2, 8, 14, and 20 all represent the same physical channel measurement. Channel 23 is "exploded" to illustrate the types of data sampled at an overall lower rate.

Instead of viewing the sampling sequence with the Sampling Diagram, a better "shorthand" presentation has developed through the years. The PCM Format Diagram is the universally recognized method for presentation of PCM formats. Now a few terms will be discussed in more detail relating to the PCM format.

A typical PCM Encoder must handle many channels of data, each with its own particular characteristics. Some signals change very rapidly and require a high sampling rate, while other signals change quite slowly and would "waste" a high sampling rate. The bulk of the channels, however, require only a moderate sample rate. Therefore, the PCM format is usually designed to accommodate most of the data in a single sampling cycle called a minor frame. The minor frame is divided into segments, or time "slots" called words. A single channel of high speed data may be connected to several words in each minor frame. This is called supercommutation. At the other extreme, subcommutation permits a channel of slowly varying data to appear only once in several minor frames. The time in which all data samples are sampled at least once is called a major frame. Thus each channel is sampled at nearly an optimum rate with no wasted data capacity.

PCM versus FM

Whenever data is to be transmitted from one point to another, the question of whether to use an analog (FM) or a digital (PCM) transmission system arises. Older equipment installations may become the target for modernization, in which case the conversion from FM to PCM may result in significant performance improvements and reduction of cost. In general, the choice of FM vs. PCM is clear, while in other cases, some study is necessary. No simple formula has been developed for comparing the two techniques because there are so many aspects to such a comparison. However, it is possible to contrast them in general terms.

One of the most important points of comparison is the required accuracy of the system. If the data must be accurate to better than about 1%, PCM is usually the choice. If the requirements are very stringent, this is almost certain to be the deciding factor.

Where large numbers of channels are involved, PCM also has advantages, chiefly in size and weight. An airborne FM system of several hundred channels would be quite bulky. However, an offsetting factor may be the frequency of the data to be handled. High frequency data channels require even higher sampling rates. In the extreme case, a few such channels could absorb the entire capacity of a PCM system which, in another arrangement, could handle many more low-speed or medium-speed channels. To handle many channels of vibration data, for example, a constant bandwidth FM system is likely to be the answer.

PCM has an advantage over FM when low power or a noisy transmission link results in a low signal-to-noise ratio. This comes about because the receiving equipment needs only to detect the presence or absence - not the height or shape - of a pulse. Once the signal power is high enough so the decision level is clear of the noise, additional signal power merely increases the "safety margin," with little effect on signal quality. If the signal is at all intelligible, only a small increase in signal-to-noise ratio may make the transmission nearly perfect.

Because PCM is a method of representing an analog signal in digital form, it is particularly well adapted to work directly with digital data-processing equipment.

Even though a signal may be developed by time-division multiplexing, it is normally transmitted by frequency modulation. For example, consider a "PAM/FM" system. This means that the PAM output from a commutator is used to frequency modulate the transmitter directly. The terminology can even be carried a step farther. An example is a "PAM/FM/FM" system. This means that a PAM signal modulates an FM subcarrier. Typically, the FM channel is then combined with other similar channels by frequency-division multiplexing, and the composite signal is used to frequency modulate the transmitter. The resulting system is a "hybrid" which combines time-division and frequency-division multiplexing.