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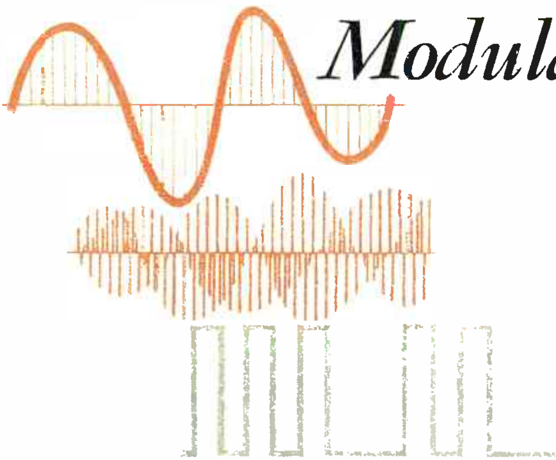
the Lenkurt® Demodulator

Pulse Code Modulation



The constant search for better communications at lower cost has led the telephone industry to a radically different method of transmitting speech information. This method, using binary digital pulses rather than conventional analog signals, provides high quality transmission and has proven to be very economical in short-haul carrier systems.

Up to the present time, telephone communications has been accomplished almost entirely with an analog electrical carrier wave which is varied continuously in proportion to the speech signals. Such systems have always been haunted by noise and crosstalk.

In recent years, the telephone industry has shown great interest in a method of coding speech information into digital electrical pulses. Unlike analog signals, these pulses, after becoming distorted by noise during transmission, can be completely regenerated at repeaters along the transmission path and at the receive station.

In the nineteenth century there were many attempts to code speech and music into digital electrical signals for transmission, using the techniques employed in telegraphy. Unfortunately, early experimenters did not have the mathematical tools provided by what is now termed information theory and were denied success because their coding schemes were too simple and did not convey enough information. Before they were able to advance their coding techniques, Alexander Graham Bell successfully transmitted speech using an analog electrical signal. The success of Bell's experiment was so immediately overwhelming that an immense telephone communications industry revolutionized around analog speech transmission.

Because of the outstanding success of analog transmission techniques, such as frequency division multiplexing (FDM), many years passed before serious attention was given to other methods of transmitting speech signals. However, with the ever-present problems of noise and crosstalk and the rising complexity and cost of electrical filters and other devices found in frequency division systems, it was certainly natural for engineers to search for more practical and efficient transmission methods. One of the most significant methods under investigation has been *time division multiplexing.*

It was demonstrated experimentally even before the development of FDM that time division techniques could be used to transmit many speech messages simultaneously over the same circuit. But such techniques could not be put into practical use at the time because of the limitations of mechanical devices for high-speed switching. The invention of the vacuum tube and the electric wave filter made frequency division multiplexing much more attractive for use in telephone transmission systems. However, researchers continued to investigate time division methods.

The first useful time division multiplex systems were developed in the early 1930's. In these systems a number of circuits share a common transmission path but at separate time intervals. Time division systems employ some type of pulse modulation, in contrast to the more familiar amplitude and frequency (AM and FM) techniques used in FDM.

The most popular type of pulse modulation has been pulse amplitude. In pulse amplitude modulation (PAM), a continuous signal, such as speech, is represented by a series of pulses called samples. The amplitude of each sample is directly proportional to the instantaneous amplitude of the continuous signal at the time of sampling. Since the amplitudes of the samples are continuously variable, the problems of cumulative noise and dis-

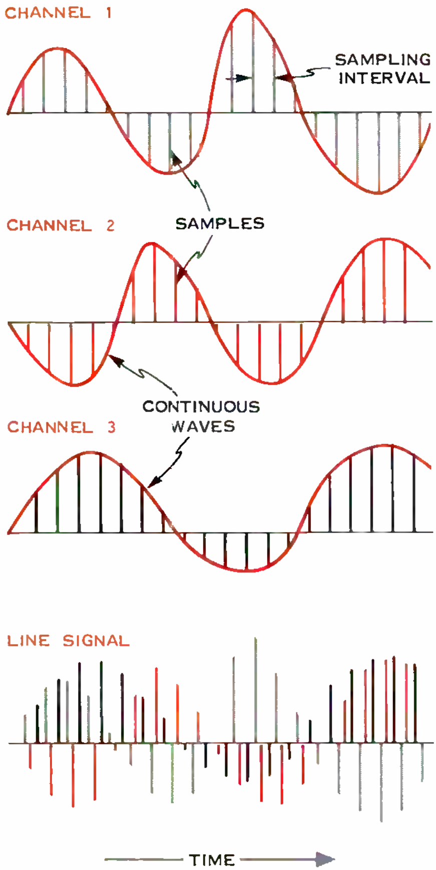


Figure 1. Example of time division multiplexing using pulse amplitude modulation. A pulse amplitude sample is placed on the line as each channel is sampled in turn.

tortion associated with analog signals are present in pulse amplitude modulation systems.

In 1937, Alec H. Reeves, then a member of the International Telephone and Telegraph Corporation Laboratory in Paris, resolved that the problems of cumulative noise and distortion could not be overcome in pulse modulation systems using pulses of varying amplitude. This prompted him to review the early idea of transmitting speech using coded pulses of constant amplitude, similar to those used in telegraphy. His investigation resulted in the invention of a radically different approach to transmitting speech signals. In 1938, Reeves patented his invention which became known as *pulse code modulation*. Unfortunately, the development of practical pulse code modulation systems had to await the arrival of high-speed solid-state switching devices, which occurred after World War II.

Pulse code modulation involves transforming continuously variable speech signals into a series of digitally coded pulses and then reversing the process to recover the original analog signals. This procedure can be carried out in three successive operations.

The first operation is to *sample* the speech signals at a suitable rate and to measure the amplitude of the signal at the time of sampling. This operation is equivalent to pulse amplitude modulation (PAM). Next, the voltage amplitude of each sample, which may assume *any* value within the speech range, is assigned to the nearest value of a set of discrete voltages. This process is known as *quantizing* and is equivalent in mathematics to rounding off to the nearest whole number or integer. The final step is to *code* each discrete amplitude value into binary digital form, similar to coding the letters of the alphabet for telegraphy. Now a series of binary coded digital pulses can be used to carry the message over a transmission line. These binary pulses are in fixed and predetermined time positions and only the presence or absence of a pulse determines the information content of the signal. Since the precise magnitude of the pulses is no longer critical, the problems of cumu-

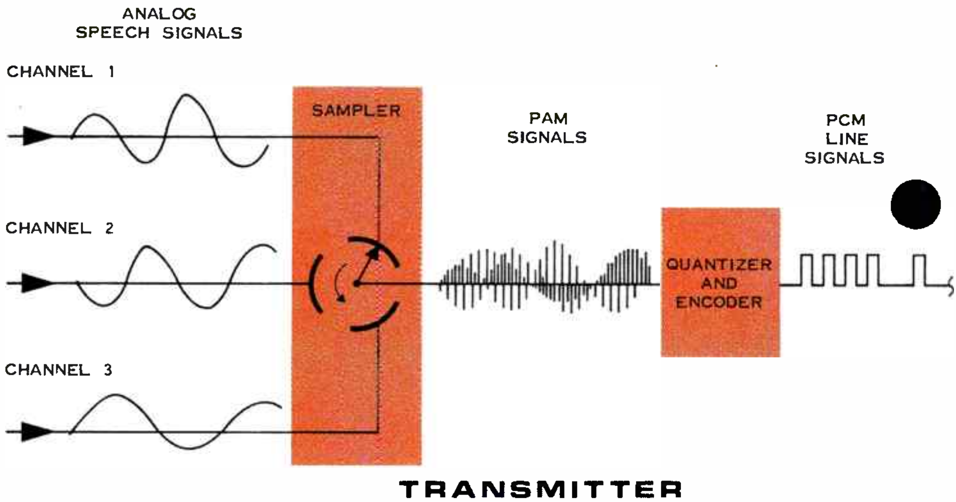


Figure 2. Simplified time division multiplex PCM system.

lative distortion and noise associated with pulses of varying amplitude are greatly reduced.

Sampling

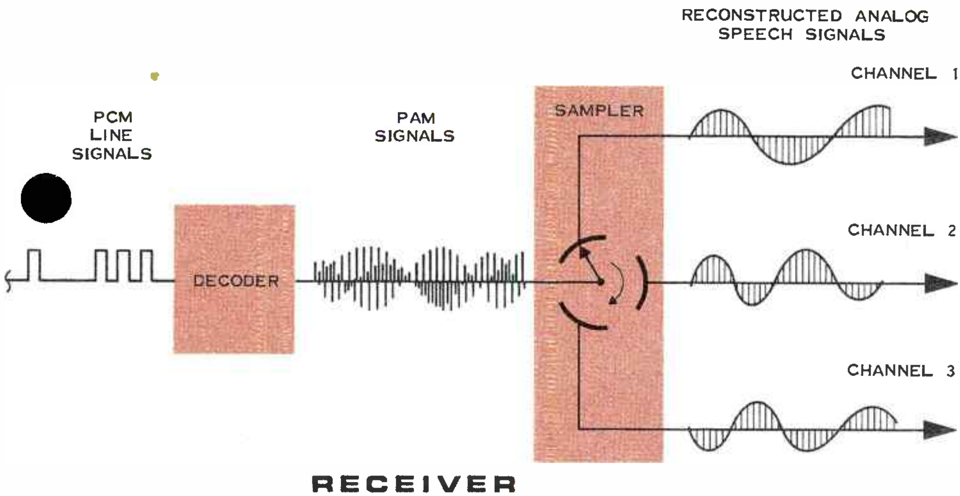
It has been proven mathematically that if a continuous electrical signal is sampled at regular intervals at a rate of at least twice the highest significant signal frequency, then the samples contain all of the information of the original signal. This principle is known as the *sampling theorem.* A continuous signal waveform, therefore, can be represented completely if at least two amplitude samples are transmitted for every cycle of the highest significant signal frequency.

In PCM systems designed for speech signals, a sampling rate of 8000 Hz, or one sample every 125 microseconds (1/8000 second), is ordinarily used. This sampling rate is sufficient since the bandpass of ordinary speech or telephone channels has an upper cutoff frequency below 4000 Hz. The 125 microsecond interval between samples of one voice channel can be allocated to other voice channels by means of time division multiplexing.

The number of channels that can be time division multiplexed using an 8000-Hz sampling rate depends, of course, on the duration of the time slot assigned to each sample — the shorter the duration, the greater the number of channels. In practice, the duration of the samples depends upon the operation and characteristics of a physical circuit. Thus, the number of time division channels is limited by the performance requirements and capabilities of a particular transmission system.

Quantizing

As previously stated, sampling a continuous speech signal at regular intervals results in a series of pulses whose voltage amplitudes are proportional to the level of the signal at the



time of sampling. The amplitudes might be any of an infinite number of values within the intensity range of speech. The usual intensity range encountered in telephone systems is about 60 dB, or a voltage ratio of 1000 to 1.

After sampling, the next step in the PCM process is to divide or quantize the 60-dB intensity range into increments or amplitude levels to permit binary digital coding. These discrete levels, known as quantum steps, are used to represent any level within the speech range. This is accomplished by using the quantum step nearest to the actual amplitude value of the pulse sample. For example, an actual amplitude sample with a value of say 8.24, would be represented by quantum step 8. A sample value of 8.61 would be represented by quantum step 9, and so on.

Since the quantum step only approximates the actual value, there is always some error. The maximum error is equal to one-half the size of the quantum step. In speech signals, such errors are random and cause what is usually referred to as *quantizing error or noise.* Quantizing noise is the major source of signal distortion in PCM systems. The degree of quantizing noise is mainly a function of the number of quantum steps used—the more quantum steps, the less the quantizing noise. However, increasing the number of quantum steps increases the bandwidth required to transmit the coded signals.

It is, of course, necessary that the quantizing process detect all of the positive and negative amplitude levels within the dynamic speech range. Experiments have shown that approximately 2048 *uniform-size* quantum steps are required to cover the speech range and to provide sufficient signal fidelity. An excessively large bandwidth is required to transmit the coded line signals representing such a large number of uniform quantum steps.

One way of reducing the number of quantum steps without sacrificing quality is to make the size of the quantum steps *non-uniform,* thereby taking advantage of the statistical distribution of speech amplitudes. Most of the information in speech signals is concentrated at low amplitude levels. If the quantum steps are all equal in size, then low level or weak signals suffer the greatest amount of quantizing error. Therefore, small quantum steps are needed more at the low amplitude levels than at the higher levels. If very small quantum steps are assigned where most of the speech information is concentrated, that is at low amplitude levels, and larger steps assigned to the rest of the amplitude range, then the total number of steps required can be greatly reduced. Varying the size of the quantum steps requires sophisticated coding techniques which are presently under development.

Another method is to *compress* the amplitude range of the pulse samples before uniform quantization and then to expand the range back to normal at the receiving end of the circuit. This technique, called instantaneous compression and expansion, or *companding*, achieves the same results as varying the size of the quantum steps. Instantaneous companding, which must be very fast-acting to respond to the short pulse samples, should not be confused with the slower-acting syllabic companding technique used in certain analog telephone circuits — although the principles are the same. The syllabic compandor responds to the envelope of analog speech signals directly while the instantaneous compandor responds to PAM samples of the analog signals.

Signal compression modifies the normal distribution of speech amplitudes by imparting more gain to weak signals than to strong signals. In typical applications, the technique reduces the amplitude ratio from 1000 to 1 to 63 to 1. Using a certain compression characteristic that reduces the speech range from about 60 dB to about 36 dB, and one that varies logarithmically with signal amplitude, the number of quantum steps can be reduced from

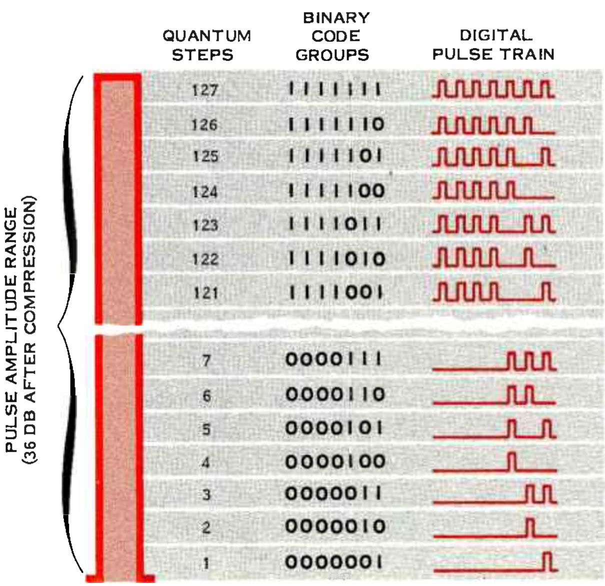


Figure 3. In PCM, amplitude samples of speech signals are compressed, quantized, coded into binary form, and placed on the line as digital pulses.

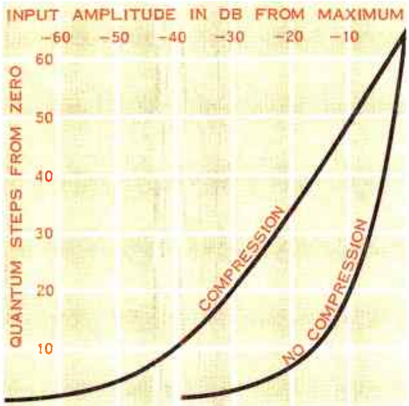


Figure 4. Signal compression using typical logarithmic compression characteristic.

2048 to 128 while maintaining the same quantizing noise performance. Signal compression using a typical logarithmic compression characteristic is shown in Figure 4.

Coding

The final step in the PCM process is to code the quantum steps into digital form. If each quantum step is numbered in decimal form, then some type of digital code can be developed to represent each of the numbers. Ordinarily, a binary code is used that consists of a combination or *code group* of binary 1's and O's, each group representing a decimal number. Once the code is established, a series of on-off binary pulses, representing the code groups, can be used for transmission.

The number of quantum steps that can be represented with a binary code is 2^n, where n is the number of binary digits, or bits, required in each code group. Thus, a 5-bit code is required for 32 (or 2^5) quantum steps, while a 7-bit code is needed for 128 (2^7) steps. In systems using a 7-bit code, the speech amplitude range is normally divided into 127 quantum steps; step 64 is zero reference, with 63 steps positive and 63 steps negative.

The bandwidth required to transmit digital pulses is directly proportional to the number of bits in the code group. A code representing 2048 uniform quantum steps would have required 11 bits per code group ($2^{11} = 2048$). Compressing the amplitude range of the sample pulses before quantization, therefore, reduces the number of bits per code group required for quality speech transmission from 11 to 7.

With a 7-bit code, the first bit position has a value of $2^6 = 64$, the second has a value of $2^5 = 32$, and so on. The value of all seven positions is shown in the following table.

Bit Position	1	2	3	4	5	6	7
Value	2^6	2^5	2^4	2^3	2^2	2^1	2^0
	64	32	16	8	4	2	1

Typically, the coded line signal consists of a train of pulses in which binary 1's are represented by positive or negative pulses and binary O's are represented by spaces (or no-pulses). A binary 1 in any of the bit positions means that the value of the position is to be summed. A binary O in any of the positions means that the value of the position is *not* to be summed. As an example, the pulse train and code group representing quantum decimal step number 100 would be:

Bit Position	1	2	3	4	5	6	7
Pulse Train							
Binary Code	1	1	O	O	1	O	O
Value 100	64 +	32 +	0 +	0 +	4 +	0 +	0

Present Applications

One of the most outstanding features of PCM systems is that the coded line pulses can be regenerated at repeater stations. Since only the presence or absence of a pulse determines the message, the line signal can be completely renewed each time it passes through a repeater. This allows a high signal-to-noise ratio to be maintained through a long string of repeaters, thus overcoming most of the problems of cumulative noise which characterize analog transmission systems.

Unfortunately, the advantages of PCM are obtained at the expense of increased bandwidth. For example, the bandwidth of a voice channel in a PCM system using an 8000-hertz sample rate and a 7-bit code would be approximately 56 kHz compared to 4 kHz required for a single-sideband suppressed-carrier FDM system.

In typical long-haul high density transmission systems, especially microwave radio systems, the availability of bandwidth is usually very critical. Presently, PCM does not provide sufficient economical or technical improvements over analog techniques to justify its use in these long-haul systems. But the same is *not* true in short-haul cable transmission systems. There have been continuing efforts by the telephone industry to shorten the economical *prove-in* distance of multichannel carrier systems since they were introduced into the short-haul cable plant. The tremendous population growth around urban areas has greatly increased the need for low cost carrier systems in short-haul interoffice trunks. This need has stimulated

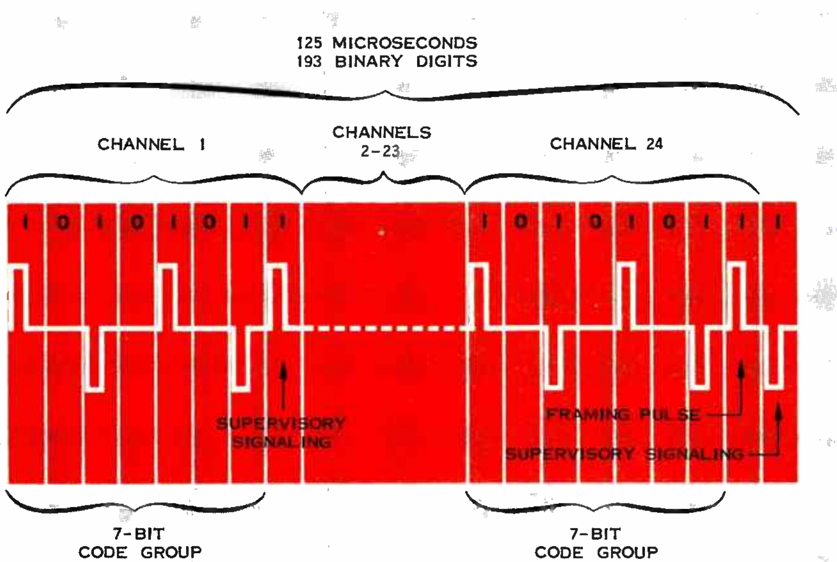


Figure 5. In the 24-channel T1 carrier system, 125 microsecond sampling interval or frame is divided into 193 time slots — 168 slots for speech, 24 slots for supervisory signaling, and 1 slot for synchronization.

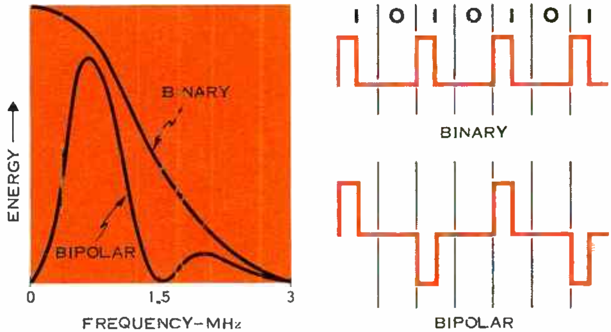


Figure 6. Energy distribution for binary and bipolar pulses with 50-percent duty cycle. In bipolar pulses most of the energy is concentrated at half the pulse repetition frequency and there is no dc component.

interest in PCM systems for use in telephone exchange cable trunks ranging in lengths from about 6 to 50 miles.

In 1962, the Bell System began production of a 24-channel PCM carrier system called the T1. Installation of T1 carrier systems in Bell's exchange plant marked the first large-scale use of time division multiplexing in commercial telephony.

The T1 carrier system was designed primarily for use with two non-loaded 22-gauge cable pairs in exchange area trunks. One cable pair is required for each direction of transmission. Regenerative repeaters, used with the T1 system, are spaced at intervals of about 6000 feet. This interval corresponds to the spacing of Western Electric's H-88 load coils on 22-gauge cable pairs. Since the load coils must be removed when the line is to be used for PCM operation, it is convenient to replace them with a regenerative repeater.

Each voice-frequency input channel in the T1 is sampled once every 125 microseconds or 8000 times per second. The variable amplitude pulses resulting from the sampling process are then compressed and quantized into one of 127 quantum steps coded into 7-digit binary code groups. An eighth digit or bit is added to the code group for each channel sample and is used to carry supervisory signaling information.

The time slots which make up one 125-microsecond period constitute what is called a *frame*. An additional bit time slot is added to each frame for use in synchronizing the two system terminals. This makes a total of 193 time slots per frame (24 channels x 8 bits per code group + 1 synchronizing slot). Multiplying the 193 time slots times the 8000 hertz sampling rate provides an output pulse train with a maximum bit rate of 1,544,000 bits per second.

The binary coded pulses transmitted to the cable pair have a fifty percent duty cycle, which means the width of the pulses is one-half the time slot allocated to each pulse. Bipolar transmission is used with successive pulses, representing binary 1's, alternating in polarity. Figure 5 illustrates a pulse train representing one frame.

There are several advantages of the bipolar pulse pattern over straight binary or unipolar transmission. As shown in Figure 6, most of the energy of bipolar signals is concentrated at frequencies of about half the pulse repetition frequency. Accordingly, there is much less energy coupled into other systems in the same transmission cable because of increased crosstalk coupling

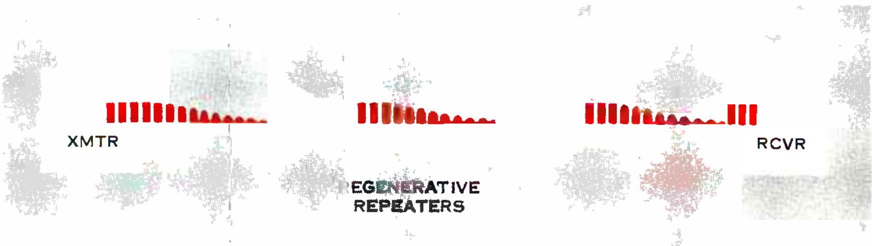


Figure 7. One of the most outstanding features of PCM systems is that the binary coded digital line pulses can be regenerated at repeaters and at the receiver station.

loss. Also, bipolar pulses do not have a dc component, thus permitting simple transformer coupling at repeaters. The unique alternating pulse pattern can also be used for error detection since errors tend to violate the pattern.

As the line signals travel along the cable pairs the pulses become distorted by the usual signal impairments such as noise and attenuation. When the pulses reach a repeater, they are retimed and reshaped so that a new undistorted pulse is produced for each pulse received. At the receive terminal the line pulses are again reconstructed before they are fed into the receiver detection and decoding equipment. The PCM coding process is reversed in the receiver in order to recover the original continuous speech signal. The continuous signal at the output of the PCM receiver should be a replica of the original signal, except for some distortion resulting from quantization.

However, noise troubles, like energy, seem to be conserved and only changed from one form to another. So it is with pulse code modulation. Although noise in PCM systems does not accumulate, it does prevent the perfect timing of regenerated pulses and shows up as jitter on the retransmitted pulse train. Successful practical solutions to this timing problem are the key to successful PCM cable carrier transmission.

Conclusions

New digital technology promises much more than just carrying out the tasks of transmission systems developed in the past. PCM systems will eventually handle all of the transmission functions of today's frequency division multiplex systems more efficiently and more economically. The development of digital techniques will enable different types of services such as voice and facsimile to be treated alike in transmission systems. Once the various types of analog signals are formed into digital signals they are all similar.

The new PCM cable carrier systems must carry on the tradition of the telephone industry. They must be used with the telephone plant that exists today, complete with its inheritance of old cables and switching systems produced by many different manufacturers.

In addition to operating over existing cable systems, there are already means to interconnect FDM systems with PCM networks using a device called an en*co*der-*dec*oder or codec. Network television in color will also be handled. Such high density systems will require line transmission rates close to 300 megabits per second. Also, the use of a technique called *pulse stuffing synchronization* will permit adding and dropping systems by digital means, and provide an easy method of interconnecting PCM systems.

High density PCM systems will use thousands of transistors in circuit configurations where switching times may be as fast as a fraction of a nanosecond. In the future, the use of integrated circuits instead of discrete components promises great economies as well as excellent reliability.

PCM systems are not without problems. Long chains of repeaters in tandem challenge the ingenuity of engineers to produce reliable repeaters at economical prices. The T1 system, for example, may use 50 repeaters in tandem for a 50 mile link. Further problems arise because PCM requires so much bandwidth. Bandwidth is readily available on cable but is not easily obtained from the available microwave spectrum. This fact ensures that single sideband frequency division multiplex will be around for a long time. PCM systems are also vulnerable to impulse noise which may prohibit their use in situations where cable plant and switching machines are not up to modern standards. In such situations present day FDM cable carrier systems, which do not exhibit the noise threshold characteristics of PCM, may do a better job. Also, cables already carrying FDM carrier systems will have to be filled out with the same type of systems since it is presently not possible to mix T1 and FDM systems in the same cable.

Although the T1 carrier system was developed primarily for the transmission of analog information in the form of processed voice signals, its repeatered line is a very fine high-speed digital transmission facility. Techniques have been developed to use these digital transmission systems to handle up to eight 50-kilobit data channels or two 250-kilobit data channels.

Pulse code modulation systems provide better handling of telephone supervisory signaling than the usual inband methods used with FDM systems. The systems employ time division signaling methods which are very economical and avoid the problems of speech simulation or *talkdown* inherent in inband signaling systems.

Some small switching machines employ time separation instead of the familiar space separation techniques used for so many years by electromechanical machines. Digital transmission is used with these time separation switching machines, and it is only a short technical step to join digital transmission and switching into an integrated communications system. It seems likely that in the 1970's integrated electronic switching and PCM transmission systems will be operating both in the United States and Europe.

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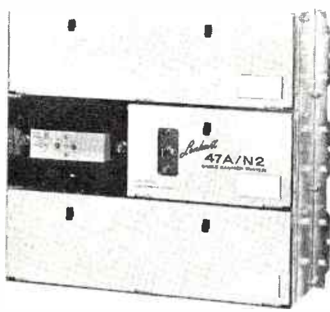

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