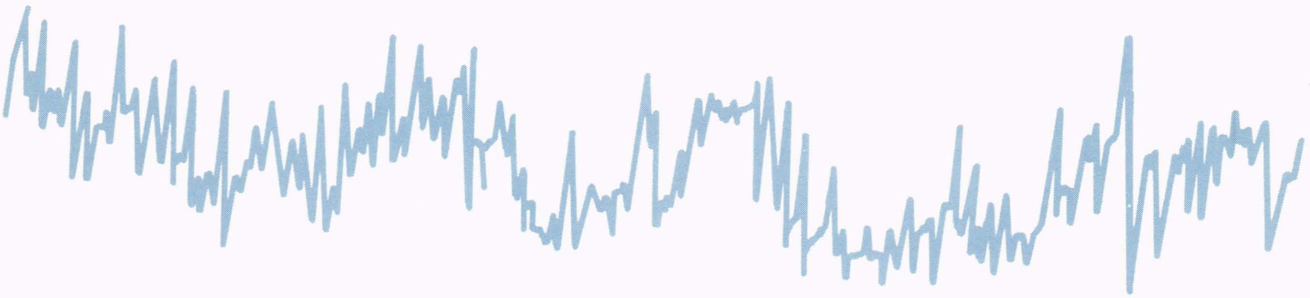
The problem of reliably demodulating or decoding noisy signals has long been of prime concern in the field of communications engineering. Filtering, frequency modulation, pulse code modulation, and the development of information theory have all contributed to solving the problem. Recently the digital computer has been brought into use as a signal-decoding device. This article describes a method for using a digital computer to decode a digitized or sampled PCM *waveform. A comparative analysis is made of the computer method and the more conventional real-time methods and their respective advantages and disadvantages are noted.*



IMPROVED DETECTION of PCM WAVEFORMS

R. B. McDowell

As early as the turn of the century Marconi and others were concerned about the degradation of spark radio transmission caused by various types of interference. Marconi's first approach was to increase the signal power. Later, following the work of Oliver Lodge, filtering began to be used.

By the early 1920's, E. H. Armstrong had begun experiments with frequency modulation (FM) to demonstrate that this method provided a substantial improvement in signal quality in the presence of noise. He showed that the improved signal quality did not derive primarily from increased signal power or filtering.

In 1948, C. E. Shannon published his general theorem on information transmission in the presence of noise. A binary digit transmission system is better known as a pulse code modulation (PCM) system and Shannon's theorem is the fundamental statement of the theoretical information capacity of such a system.* With this breakthrough, PCM became the strongest contender for minimum error in signal transmission through a noisy system.

Figure 1 shows four representative waveforms. Each waveform contains exactly the same information and each type has certain advantages relative to the other types. For purposes of our discussion here, however, these advantages will not be treated, since any facility that processes PCM data rarely has any control over the PCM that must be handled. As with FM, decreased susceptibility to noise does not depend only on signal power, but also on the form of the signal itself. The nonreturn

* Shannon's theorem: "Let P be the average transmitter power, and suppose the noise is white thermal noise of power N in the band W. By sufficiently complicated encoding systems it is possible to transmit binary digits at a rate $C = W \log_2 \frac{P+N}{N}$ with as small a frequency of errors as desired. It is not possible by any encoding method to send at a higher rate and have an arbitrarily low frequency of errors."

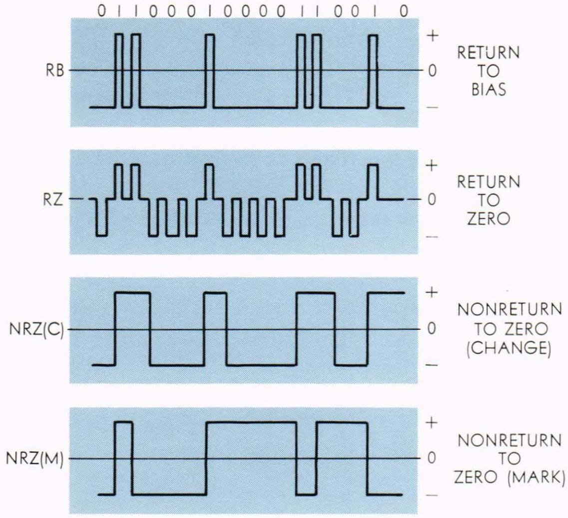


Fig. 1—Types of waveforms.

to zero (change) waveform, identified as NRZ(C), was used in the APL experiments because it is the most commonly used PCM waveform of the several that are in vogue.

By the late 1950's, it had become practical to combine the PCM and FM systems. Figure 2 shows the essentials of a PCM/FM system. The FM is used to provide a carrier for radio transmission, FM being relatively noise-free compared to other carrier systems. Theoretically, if the PCM could be transmitted long distance by itself, less error would be present at the receiving end. The PCM/FM method has proved to be more reliable than any other practical long distance transmission system. This fact notwithstanding, a discrepancy still exists between the performance of the actual systems and the performance predicted by theory. Using Shannon's formula and assuming that the system bandwidth equals the bit rate, as is the case for most practical systems, error-free PCM transmission should theoretically be possible for signal-to-noise ratios greater than 0 db.

Current Methods

The incorporation of the phase-lock loop, a feedback method of filtering and detecting signals, enabled the development of better PCM/FM systems. The phase-lock loop is applicable both to the demodulation of the FM and also to the detection of the PCM waveform itself.

The second stage of detection, the "cleaning up" and decoding of the PCM waveform after recovery from the FM signal, is performed by a bit synchronizer. The bit synchronizer is a device, normally made up of logic elements and a phase-lock loop filter, that accepts the relatively noisy PCM waveform from the output of the FM receiving and demodulating equipment, further filters the waveform, detects the waveform transitions, and decodes these transitions into a standard binary waveform.

From this point the standard waveform is converted into any digital computer format desired to facilitate the analysis of the information carried by the PCM. Or, in many cases, the numerical values represented by the waveform are simply printed out for immediate use.

In 1962, a report was published describing the theoretical optimum for the PCM/FM systems currently in use.[1] Accepting the fact that bit synchronizers are often, if not generally, used in such systems, the report treated these systems accordingly in its analysis. Figure 3 shows the optimum performance in terms of the binary bit error probability as a function of the signal-to-noise RMS voltage ratio in decibels. This optimum assumes that bit synchronization is continuously maintained. It also assumes that the FM carrier has a

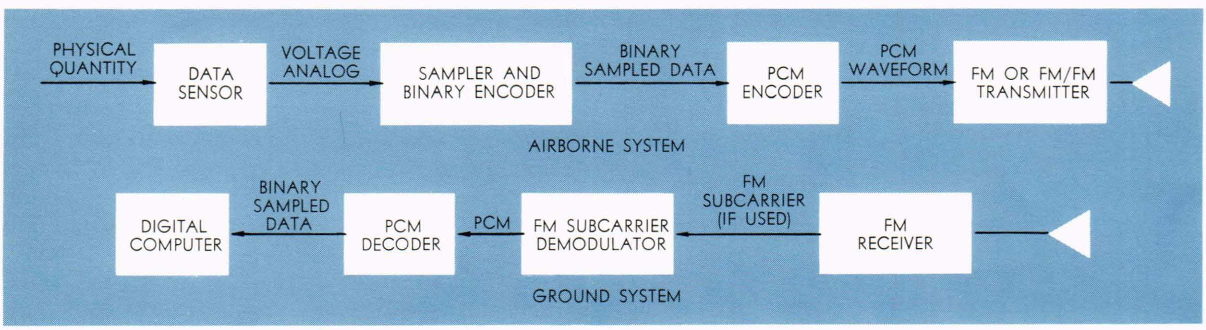


Fig. 2—Simplified PCM/FM or PCM/FM/FM telemetry system.

[1] E. F. Smith, "Attainable Error Probabilities in Demodulation of Random Binary PCM/FM Waveforms," *IRE Transactions on space Electronics and Telemetry,* **8-9,** 1962-1963, 290-297.

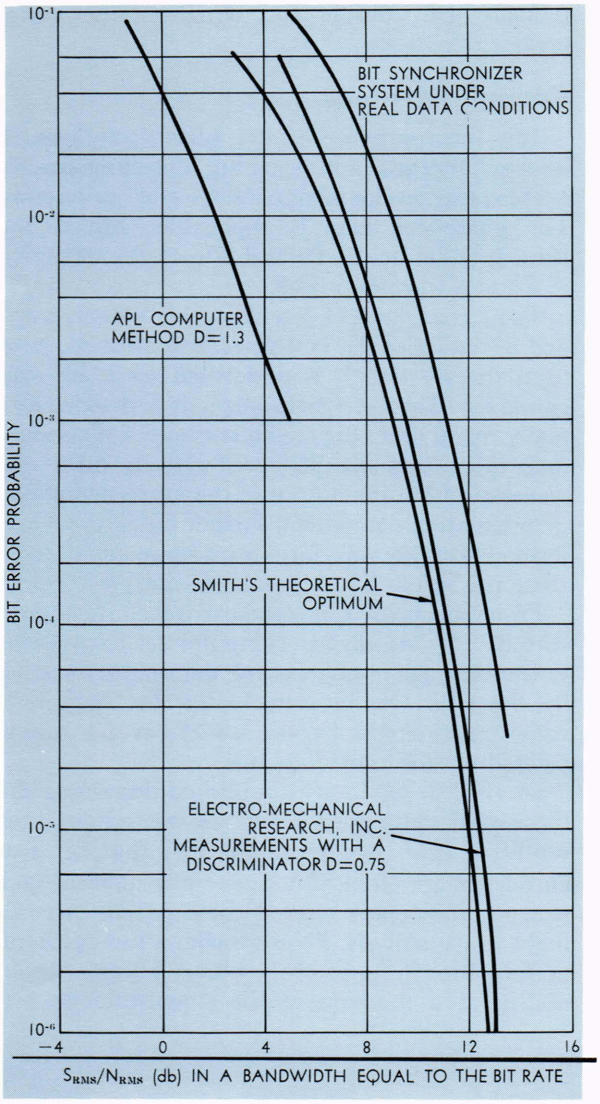


Fig. 3—Optimum performance in terms of binary bit error probability as a function of signal-to-noise ratio.

deviation ratio, D,† of approximately 0.7 and that the demodulation equipment has appropriate bandpass characteristics.

Unfortunately, this optimum performance in terms of the binary bit error probability falls far short of what Shannon's theorem indicates is possible, since the error rate is not zero when the signal-to-noise ratio is greater than 0 db, using the previous assumptions on bandwidth and bit rate.

At this point, there are three possible approaches to gaining further improvement. The first approach is to find a signal transmission system that is inherently more efficient (less subject to noise-induced error) than present systems. This is a problem of coding, the production of a new signal form to carry the information. The inventions of FM and PCM are examples of more efficient coding.

The second approach is to find a more efficient means of decoding for present systems. The bit synchronizer is an example of a more efficient decoding device, as compared to an FM discriminator alone.

The third approach is a combination of the first two. Special codes containing some pattern of redundancy are used along with a particular scheme for detecting this pattern, such as correlation of some sort. Examples of this range from parity bits to so-called error-correcting codes.

A Computer Program for PCM Recovery

At APL the decision was made to pursue the second approach and search for a better PCM decoding method. The stimulus for these efforts was supplied by the nearly complete loss of useful telemetry data from the TRAAC and 5E satellites after the high-altitude atomic test carried out on July 9, 1962.

The telemetry signals, which were PCM/FM/FM, could not be recovered despite the use of phase-lock loop FM detection and bit-synchronizer PCM detection. However, it was noticed from oscillographic records made of the output of the FM subcarrier discriminator, that some semblance of the PCM wave could be detected visually over short intervals. This led to the hope that a method could be devised that would perform as well automatically and, thereby, recover at least a small portion of the data.

This in turn led to the general approach of trying to evolve a purely graphical or geometrical method of extracting the signal waveform. This method was soon abandoned, however, in favor of a more fundamental approach involving the statistics of the signal and noise and a digital rather than an analog instrumentation technique.

The result of this work has been to produce a digital computer program that accepts and operates on a digitized PCM waveform. The output is the decoded binary information. The program was used on the telemetry data received from the above mentioned satellites with satisfactory results. This particular telemetry when processed by the computer method yielded approximately 50% usable data. These data had previously been unrecoverable with standard instrumentation. The program has also been tested quantitatively under laboratory conditions and exceeds by a substantial margin the

† The deviation ratio, in this case, is defined as the ratio of total frequency deviation of the carrier to the bit rate of the PCM waveform.

best that could be expected from standard methods, as shown in Fig. 3.

Perhaps the primary advantage that the computer method has over the standard methods is its ability to scan the data for some finite time interval and make tentative decoding decisions. These are then used and, based on discrepancies that develop, are re-evaluated and better, final decisions are made. In this way, the statistical characteristics of the input data can be accounted for on a trial-and-error or adaptive basis. The computer method does not require any assumptions to be made as to the noise amplitude probability distribution. This minimizes the a priori knowledge required. In this situation, the computer is used to try to produce an optimum result by successively adjusting certain parameters, such as Δt, the expected bit cell period. In short, it is possible to use a sort of regression method to separate the signal from the noise. The usual statistical problems are dealt with only in a gross way as they directly affect the result and not in an explicit analytical fashion.

As part of the regression method used, successive tests are made for the number of points, K, that lie above or below L, the transition level. In fact, the basis of the program is founded on the fact that the computer can store data, and then successively adjust it to fit certain known, fixed characteristics.

A second, practical advantage of the computer method is that the drifts, noise, and phase shifts, characteristic of analog or hybrid-type equipment such as bit synchronizers, can be minimized by going directly from the FM detector output to the digital format by sampling (digitizing) the PCM waveform.

Comparison of Decoding Methods

For a clearer understanding of the differences between the computer method and the bit synchronization method for decoding PCM, a brief comparison of the character or qualities of each method will now be made.

1. Acquisition time for the computer is always one frame or less, if synchronization can be achieved at all. A PCM frame is a sequence of numbers, or pulses representing the numbers, assigned to a set of measurements which are cyclically repeated. The acquisition time is zero except when a frame-to-frame change in bit rate exceeds 12%. At a $\pm 12\%$ change in bit rate it is necessary to change the expected bit cell period parameter in the program or the program will not operate. Why 12% is the point of failure could not be ascertained. This was an empirical result. A bit synchronizer will require from one to many frames for synchronization.
2. Up to 12% changes in bit rate can occur in the PCM waveform without producing any effect on data recovery and without requiring additional parameters to be supplied to the program. Bit synchronizers generally can cope only with changes of about $\pm 5\%$ before losing data.
3. The transition density of the PCM waveform is unimportant for the synchronization of the computer method. (Transition density is the average number of waveform transitions relative to either time or the number of bits.) Bit synchronizers will either fail to recover the data

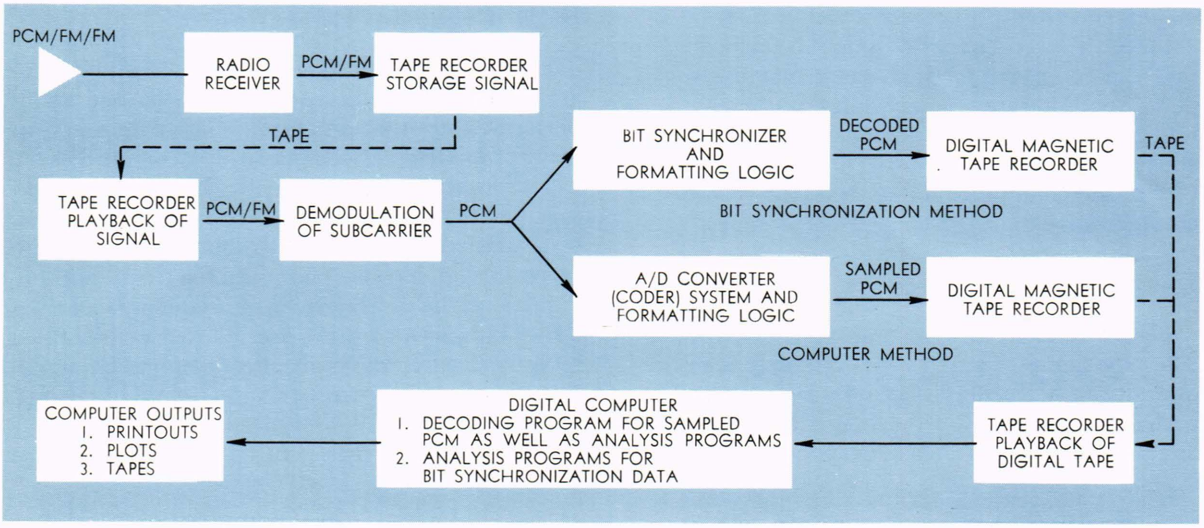


Fig. 4—General purpose PCM data processing system, showing bit synchronization and computer methods for decoding PCM.

if the transition density falls below some minimum level, or, at least, the bit error will increase sharply, making the recovered data highly questionable.

4. No critical adjustments of parameters are required for the computer program. In their present state of development, bit synchronizers require rather critical adjustment of loop bandwidth and other internal circuitry.
5. The current state-of-the-art in analog-to-digital conversion and in transfer of the resulting digital information to the computer is such that the PCM data rate which can be handled by the computer system is less that that which can be decoded by the bit synchronizer system. (Bit synchronizers can handle on the order of 1,000,000 bits per second. Digitizing equipment can, at six samples per bit, handle on the order of 80,000 bits per second.)
6. Digitization of the PCM over a six-samples-per-bit basis requires the computer to handle several times as much data as there is information in the PCM data. The economics of the situation depends on computer availability in any particular case, but as an indication it may be noted that the data processing costs have been approximately doubled when using the computer method.

From the preceding discussion it is fairly obvious that the computer method would not likely supplant the bit synchronizer for data recovery of routine interest nor would it necessarily be more efficient or more economical. The prime application of the computer method is in the recovery of essential data from extremely noisy signals.

General PCM System

Figure 4 illustrates a general type of PCM system. The PCM signal originates as a remotely measured quantity in a missile, a satellite, a biomedical experiment, etc. Usually, the PCM is directly modulated on a carrier, or on a subcarrier which is then remodulated on an RF carrier, if several signals are to be multiplexed and transmitted by radio link.

This composite signal is then received at some convenient location, demodulated, and recorded on magnetic tape. Normally, the complete processing of the remaining signal cannot be adequately performed at the receiving site. The tape-recorded information is then sent to a central data-processing facility.

At this central facility, the information is reproduced from the tape, any necessary further demodulation, filtering, and other signal conditioning is performed, and the remaining PCM plus noise is then either fed through a bit synchronizer or is sampled, depending on whether the old or new method is to be used. If the former, the output from the bit synchronizer is stored on magnetic tape in suitable format for computer entry. The digital computer is necessary only for analysis of the data in terms of its physical meaning. The communications problem ends with the final decoding of the PCM waveform carried out by the bit synchronizer.

In the other case where the PCM waveform is sampled, but not decoded in any way, the decoding is carried out by the digital computer and the bit synchronizer is not required.

PCM Program Operation

Figure 5 is a block diagram of the computer program. Figure 6 illustrates the basic operations performed by this program, relative to the PCM waveform itself.

The frame word is found by successively masking the bit pattern, moving the mask one bit at a time until the frame word is found containing no more than a specified number of errors. The beginning of the pulse or number sequences forming the PCM cycles or frames are indexed or identified by a fixed set of values or pulse patterns known as the frame word. When two successive frame words are found, the Δt is adjusted incrementally by small amounts in order to determine exactly the

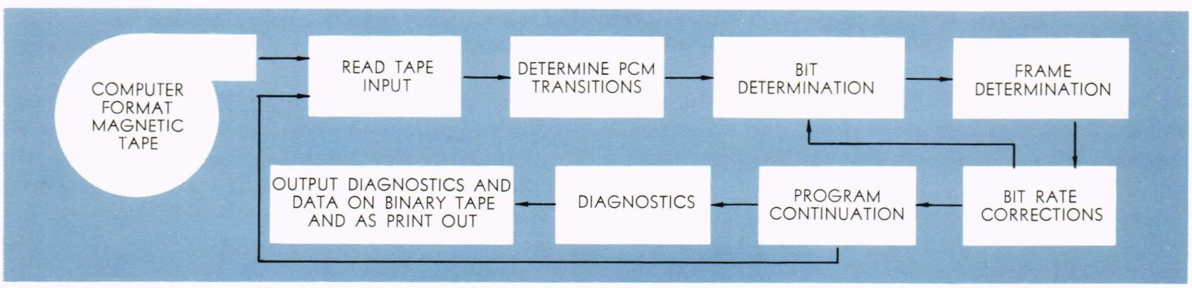


Fig. 5—Block diagram of PCM detection and decoding program.

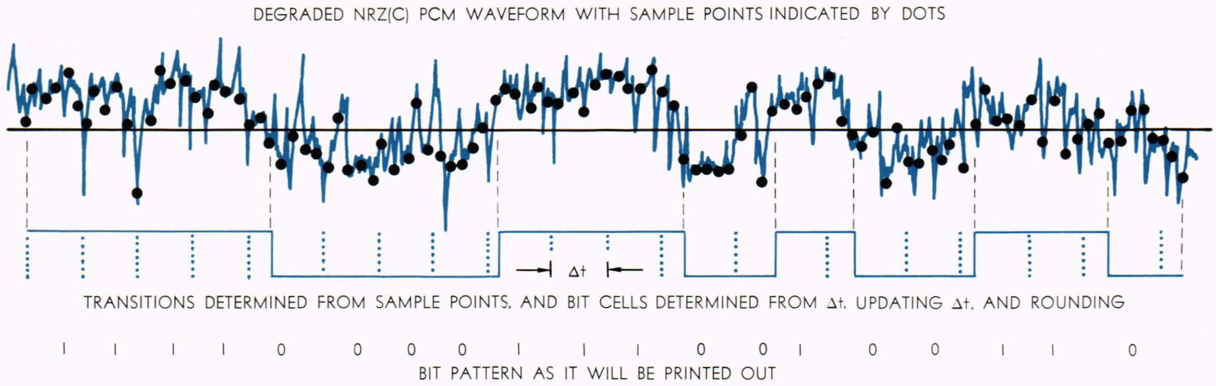


Fig. 6—General procedure for detecting degraded PCM.

proper number of bits within the frame. This process largely compensates for the imprecision, due to noise, by which the PCM transitions are detected.

The following outline summarizes the operation of the program:

1. *Transition Determination.*
   a. Take a moving sequence of sample points. The number of these sample points, identified as N, is determined by the requirements for describing a single bit cell. A bit cell is that portion of a PCM waveform which, when decoded, yields one binary digit. However, N may be any number easily stored in the computer in excess of this minimum.
   b. Compare each value of the sequence with a parameter L, the transition level, until K values in succession are found above or below L. The value of K is predetermined; it represents the number of sample points required to be reasonably certain that a true transition is actually represented by the cross-over of the waveform from one side of the transition level to the other. The intent is to rule out momentary transitions caused by noise spikes. The value of K cannot be larger than, or consist of more than, the number of points contained within, a single bit-cell period, Δt. By trial-and-error, the optimum value was found to be the number of points making up two-thirds of a bit-cell period, which for a six-samples-per-bit system involves four points.
   c. A sequence of K points above the level L establishes the "ones" state. If the sequence is below L, the "zeroes" state is established.
   d. Continue examining points until a sequence of K points is found which lies on the opposite side of L from the first set.
   e. Store the time associated with the first point of the second sequence. This is a transition.
   f. Continue this process until P points have been compared. The P points represent the number of points required from along the waveform to include a little more than one PCM frame. It is important to include an interval long enough to contain two frame words. Once the transitions have been determined from this set of points, the bit pattern may be determined and, then, from this the frame words can be found.
2. *Bit Determination.*
   a. Divide the time interval between transition times by the parameter Δt, the expected bit-cell period. Do this also for the initial period up to the first transition.
   b. Store these numbers (rounding any fractional values).
   c. Depending on the polarity of the leading transition, store these numbers sequentially as the corresponding number of 1's or 0's.
   d. Continue the above process until an accounting has been made of all transition periods.
3. *Frame Determination.*
   a. Take a moving sequence of 1's and 0's in the order stored above and compare it with a mask of M bits (the mask being a standard pattern of 1's and 0's, in this case the frame word) until a match is obtained for all but E bits of the mask. The value M represents the number of bits making up a frame word. The E bits are the acceptable errors that will be allowed in the determination of frame words. Since the frame must be crudely detected before other refinements of the detection process can be carried out, the initial detection must allow for some error in the frame word.

b. Continue this frame determination until all bits have been checked.
c. Store sets of bits in sequence beginning with the first bit of the frame word as determined by the masking operation above.
d. If no frame word is found for bits not accounted for by other frames, list the bits detected in the interval in blocks of F bits or less. The parameter F is the number of bits in a proper frame.

4. *Bit Rate Corrections.*
   a. Determine the number of bits between first bits of successive frame words.
   b. If the number is less than F, decrease Δt by T seconds and recompute the bit patterns of each frame as done in paragraph 2 above. The T seconds increments are essentially arbitrary and depend partly on how long the program should try to correct Δt and partly on judgment as to the amount by which Δt should be incremented for each try at correction.
   c. Continue to increment Δt until F bits are determined or Υ trials have been made. The Υ trials, like the T increments, are essentially arbitrary, and are determined by judgment for each individual occasion.
   d. If the number of bits in a frame is greater than F, increase Δt by T seconds and recompute as done in paragraph 2 above.
   e. Continue to increment Δt until F bits are determined or Υ trials have been made.
   f. If F bits are determined or Υ trials have been made, go to the next frame and compute the necessary corrections.
   g. Use the corrected Δt for the first trial on the next frame. Instructions in paragraphs 2, 3, and 4 are carried out one frame at a time so that a continually corrected Δt is available for both frame detection and the first trial for bit determination in successive frames.

5. *Program Continuation*
   a. Take a second sequence of points beginning where the first left off in paragraph 1 above.
   b. Load the computer output tape with framed data in a format suitable for input to analysis programs or printout.

6. *Diagnostics* (*Printout*)
   a. Record time of first bit of each frame word.
   b. Record number of bits short or long for a frame after corrections.
   c. Estimate bit rate after correction for each frame.
   d. Note frame number or other fixed or predictably changing data word contained within all or some frames.
   e. If no frame word is found, print statement to that effect.
   f. List all program parameters.
   g. Record total of good frames and total of all frame intervals.
   h. Tabulate framed 1's and 0's.

It might be supposed that the higher the sampling rate along the original waveform the better. This did not prove to be the case in the experiments conducted. Instead, six sample points per bit cell was optimum. Higher sample rates caused a gradual decrease in detection reliability and lesser rates caused dramatic decreases. Deviation ratio for the FM signal, filtering, signal-to-noise ratio, and other factors had no appreciable effect on this optimum. So far, there is no adequate explanation for this situation.

Summary

A continuing effort of growing magnitude in the field of communications is producing a multitude of new codes and coding methods, all aimed at transmitting information more reliably. A complementary effort is also going on to produce better methods of detection and decoding of the transmitted information. Great strides have been made in recent years, among which FM, PCM, and Shannon's theorem on information capacity are foremost.

The use of computers as opposed to conventional instrumentation promises certain special advantages in the decoding process. Specifically, many operational difficulties with drift, equipment noise, operator judgment and error, etc., are avoided or minimized. But most importantly, considerable improvement appears possible when the data and noise statistics can be taken into account, not just on a fixed or a priori basis, but also in an adaptive way. The computer also enables performance of numerical operations representing highly nonlinear processes, which cannot in general be implemented by conventional electronic instrumentation.

The method presented in this article is an attempt to capitalize on some of these advantages. The work has been of an empirical nature rather than being based on an analytical foundation, except as previous analytical works cited within this article contributed to guiding the work discussed. Work is going on which it is hoped will provide a sound analytical approach to the use of adaptive or regressive techniques for decoding signals. In the meantime, the computer program presented here is a practical means for recovering PCM information that would otherwise be lost.