# CONCEPTS OF INTEGRATED ADAPTIVE DATA TRANSMISSION SYSTEMS

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**SUMMARY** Telemetry systems require the transmission of information in analog, sampled, and digital forms. There also are requirements for spectrum conservation, for minimum spurious transmission outside the assigned band, and for highest flexibility in multiplexing and in accepting a wide range of sensors.

The paper describes the conceptual design of an advanced information transmission system that operates basically in the sampled mode, but that can accept analog inputs through an information compressing sampler or digital inputs by taking one sample of each input bit and making binary decisions in the receiver.

The system uses three different encoding processes in an integrated manner.

The first encoding process is performed by a redundancy reducing computer-like subsystem called the *contractive encoder*. Its purpose is to eliminate unnecessary samples while keeping the essential samples at their correct place on the time scale. Repetition of already transmitted samples. or the insertion of samples from other channels will fill the space of samples that have been removed.

The second encoding process is performed by a *distributive encoder* The purpose of this part of the information processor is the protection of the information against sudden pulsive disturbances. The distributive encoder spreads the information content of each sample over tens to hundreds of other samples in an ordered manner. If, during transmission, one sample or a group of samples is heavily mutilated by pulsive disturbances, the corresponding decoder in the receiver will recover most of the information of these mutilated samples from the other undisturbed samples while spreading the energy of the pulsive disturbances over all samples so that it contributes only a negligibly small error to any one of them.

The third encoding process is performed by the *modulative encoder*. This device has the task of shaping the samples that are produced by the distributive encoder into

bandlimited waveforms so that a noise-like, bandlimited composite transmission signal with a uniform spectral distribution is finally fed into the communication channel.

Many adaptive features and the option to use a return channel for repetitions of blocks of samples give the system high stability, high efficiency, and low error ratio.

**INTRODUCTION.** The 1966 Telemetry Standards call for utmost care in designing telemetry systems as far as spectrum conservation is concerned.[1] The problem of assigning channels so that there is a minimum of mutual interference is becoming increasingly difficult to solve. The calamity of the overcrowded spectrum affects all services to approximately the same degree.

There is no single answer to the problem of spectrum conservation. It can be attacked only by a combination of many efficient design measures in concert with a well-planned integrated system optimization. This idea of integrated data transmission systems is not new. The individual design measures that have to be combined into an efficient integrated data transmission system evolved in the 1950-60 period. They were summarized into ten cardinal principles in 1959:[2]

- Redundancy reduction
- Statistical evaluation of the transmission medium
- Matched energy distribution in time and frequency domain
- Large signal alphabet in an economical signal base (WT product)
- Bandlimited waveforms (preferably orthogonal sets)
- Protective encoding
- Variable transmission rate or power matched to the variations of the channel characteristics
- Optimum signal extractors in the receiver (matched filters or cross-correlators)
- Optimum deciders in digital systems (maximum likelihood deciders)
- Application of reliability checks and mode control over a return channel.

The engineering means for the implementation of all these ten cardinal principles underwent extensive exploration since 1960. For the first time, it now seems possible to attempt at least a conceptual design of an integrated information transmission system. It is exactly twenty years since C. E. Shannon published his mathematical theory of communications. [3] In that theory Shannon separately defines a discrete channel and a continuous channel. It is the continuous channel for which he derived the famous formula that relates the maximum error-free information transmission rate (capacity) of a channel to the bandwidth and the signal-to-noise power ratio. Shannon states that this capacity can be reached with a “sufficiently involved encoding system.” Error correcting systems, also called protective encoding schemes (cardinal principle number six), represent only one of the methods that are available to design a “sufficiently involved encoding system." All the other nine cardinal principles have to be applied as well and many of them are indirectly discussed in Shannon's 1948 publication. Because of this concentration on the sixth principle, however, most approaches preferred binary digital encoding schemes.

It is only in the last few years that communication theory has reconsidered the older art of sampled information transmission. Encoding processes have now been suggested that involve the transmission of linear real number codes.[4] This fact, together with the development of advanced circuitry for implementing efficiently all the ten cardinal principles, leads this author to believe that sampled information transmission rather than binary data transmission will be the preferred subject of advanced development efforts in the next two decades.

This paper intends to show, by way of a conceptual design example, how all the ten cardinal principles could be implemented in an information transmission system operating on a sampled basis. Evidently a system that can handle continuous samples can handle discrete samples as well; i.e., two level samples (binary) or multilevel samples (nonbinary). A decision device in the receiver will make full use of any *a priori* information that should indicate a digital input. The implementation of the processing circuits may well be on a binary digital basis, though the binary words will represent continuous samples to any desired degree of precision.

**THE OVERALL SYSTEM DESIGN** A triple encoded sampled information transmission system (TESIT) consists basically of three different encoders at the transmission end and three corresponding decoders at the receiving end. Around these subsystems a designer may arrange a number of additional circuits and devices to implement virtually all the ten cardinal principles listed in the introduction.

The first encoder may be called the *contractive encoder*. It accepts the samples (or digits) at a constant rate, either directly from the input source or from a synchronous sampler, if an analog input source is connected to the equipment (block 2 in figure 1). If large scale integrated (LSI) hardware is used throughout the TESIT equipment, the designer may prefer to perform all the processing operations in binary logical circuits. In this case the designer may provide a quantizer at the input to encoder 2 and all samples will be processed in the form of binary words.

A sufficiently large number of samples is submitted to the contractive encoder 2 before any output is released to the next encoder. An analyzer/programmer (block 3 in figure 1) investigates the statistical characteristics of the samples that will be in the contractive encoder at any given time. It programs the contractive encoder in accordance with the result of the analysis or in line with a prepared information compression program. Some examples of this action will be given later. The result of the contractive encoding process is two outputs: one output transfers over line 4 a smaller number of samples, on the average, than had been received by the encoder. The other output transfers over line 5 a few digits of service information for each group of samples that goes over line 4. These digits, which are called “flags, 11 indicate the kind of encoding that had been applied to the group of samples. This group of samples itself is called a “contractive word. “

Notice that this contractive encoding process performs a redundancy reduction process according to cardinal principle number one. Many such redundancy reduction processes have become known in the last decade. Some that are particularly useful in a TESIT system will be explained later in the paper.

The second encoder may be called the *distributive encoder* (block 6, figure 1). It first accepts several groups of “contracted samples” along with the corresponding flags and forms them into a longer “frame” of such contracted samples and digits. Notice that any binary digits, such as the flag digits, will be processed just as any other samples. They simply represent samples of fixed positive or negative height. Then these longer frames (comprising, for example, 32 samples) are submitted to the distributive encoding process. The result of this process is a new frame, the output. frame; usually, but not necessarily, it has the same number of samples as the input frame. The output frame contains in each sample contributions from all samples of the input frame. Such distributive encoding processes can be performed with the help of orthogonal binary sequences as commutating means. [5] However, it may be expected that nonbinary orthogonal sequences will lead to more efficient distributive encoders.

Notice that this encoding into long frames is in agreement with cardinal principle number four. To meet also the requirement of spectrum conservation the distributive encoder may be programmed in the multiorthogonal mode. [5] In this mode many spread-out samples will be linearly superimposed so that they all occupy the same frequency band in the same time interval. In this limiting case each output frame of the distributive encoder will have the same number of samples as the corresponding input frame. In this mode of operation the distributive encoding process offers protection against pulsive disturbances. Alternately the distributive encoder can be programmed to produce output frames that are much longer than the corresponding input frames. Redundancy is thereby introduced and the process is, in sampled form, the equivalent of the digital error correction encoding process (cardinal principle number six). In this mode the information is protected against Gaussian noise as well as against pulsive noise. The transmission can then take place over channels with low signal-to-noise ratio, but the larger number of samples will lead to an increased bandwidth occupancy or to a lower information transmission rate.

The third encoder is designated as modutative encode,% (block 8, figure 1). It receives the output samples, called “distributed samples, from the distributive encoder through an amplifier. The modulative encoder modulates special bandlimited waveforms with the individual sample values. , Usually the designer will use orthogonal sets of bandlimited waveforms. [6] These sets may be operated in one of several modes:

> The modulative encoder may be programmed to use only one waveform for each sample interval. In this single signal mode the encoder operates in a bandspread mode with the result that transmission may take place with a slightly smaller energy per sample than without bandspread action, provided the noise power density is constant over the whole band over which the energy can be spread.
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> Alternately the modulative encoder may be programmed to operate in the multisignal mode or in the multiorthogonal mode. In these modes several or all of the orthogonal bandlimited waveforms will be transmitted simultaneously. Each waveform interval will now carry a larger number of samples. These modes will be used if sufficient transmitter power is available and spectrum conservation is of highest importance.

Notice that this modulative encoding process meets the requirements of cardinal principle number five and, in connection with the distributive encoder, it also meets the requirements of cardinal principle number three, If the mode selection is coupled to a measurement of the channel characteristics, the system will adapt itself to variable channel conditions. This operation follows the seventh cardinal principle.

The output of the modulative encoder, whether in the single signal mode, the multisignal mode, or the multiorthogonal mode, is called the composite transmission signal. This signal is mutilated in the transmission channel (15 in figure 1) and noise is added to it. In most channels there is a background of random noise on top of which individual noise spikes (pulsive noise) appear occassionally.

The receiving system (lower part of figure 1) extracts the information from the disturbances of the channel and restores the original input samples as precisely as possible and in the correct order and position on the time scale. If digital input is used, digital output will be restored at point 23 in figure 1. If sampled input is received, the corresponding output samples will appear at the same place. If analog input is received, a smoothing filter will be applied to restore a continuous wave from the sampled values.

The essential subsystems of the receiving system perform the converse functions of the corresponding subsystems in the transmitting system.

The *modulative decoder* (block 16, figure 1) is the principal signal extractor in the receiver. It cross-correlates individually in separate correlators all the participating orthogonal waveforms with the incoming composite signal. At the outputs of some or of all of these correlators will appear the distributed samples evidently still contaminated by noise and distortions.

Notice that the modulative decoder performs the function demanded by cardinal principle number 8. If the transmitter encoder operates in the single signal mode, the receiver cross-correlator will apply decision rules or a ptioti information to determine the one output that corresponds to the waveform that has been selected by the transmitter modulative encoder. If the transmitter operates in the multisignal mode, the receiver crosscorrelator will attempt to determine or will select those outputs that correspond to the waveforms that the transmitter modulative encoder has modulated. If the transmitter operates in the multiorthogonal mode, the receiver cross-correlator will use all the available outputs. Evidently, since the total energy that can be applied at the transmitter to any waveform interval is limited, there is the possibility of giving the whole energy to (1) one waveform in the single signal mode, (2) a limited numb, er of waveforms in the multisignal mode, or (3) all waveforms in the multiorthogonal mode. Correspondingly, the system will operate with low sampled transmission rate in low SNR, with medium rate in medium SNR, or with high rate in high SNR.

It is also important to recognize that no final decision device can be used for the information samples immediately after the modulative decoder. Remember that the output of the modulative decoder is the contaminated distributed samples and these samples have to be reconverted to the contractive samples in the next receiver subsystem. Decision devices may be used at this point in the system for any service signals (synchronization etc. ) that must be branched off (25 in figure 1), or when a hybrid modulative encoding process should be used with some quantization involved.

The second essential subsystem in the receiver is the *distributive decoder* (17 in figure 1). It performs the converse function of the distributive encoder in the transmitter. The distributive decoder reassembles each of the original contracted samples by cross-correlating each word of Contaminated distributed samples in a separate correlator with replicas of all the orthogonal binary sequences that had been used in the distributive encoder. The outputs of some or of all of these cross-correlators deliver the original word of contracted samples.

Notice that this second decoding process completes the application of cardinal principle number four. If the distributive encoder has been programmed into a redundant mode, the resulting large WT can be used as a means of improving the transmission efficiency. That efficiency is also called the “utility”, [7-9] and it is defined by the ratio of the noise power density in the channel to the energy per bit of information as referred to the receiver input. The improvement in utility is particularly significant if the original input is digital and if the distributive decoder is equipped with a maximum likelihood decider. Such a design approach will meet the requirements of cardinal principle number nine.

The third subsystem, the *contractive decoder* (block 18, figure 1), will receive two outputs from the distributive decoder. Output 20 in figure I will carry the words of samples that originally had left the contractive encoder over connection 4. Output 21 will carry the flag digits that are necessary to tell the contractive decoder (group by group) what to do with the contracted samples for restoring the original information to an optimum degree.

Notice that at this point in the system the information samples do not correspond exactly to the original contracted samples. The sample error evidently is due to the distortion and due to the noise in the channel. If the samples are taken from digital input data, the decider within the distributive decoder may take care of much of these sampling errors and will submit to the contractive decoder clearly defined quantized samples (two levels for binary input). Naturally, some of these quantized samples will be wrong; i.e., digital errors will have resulted from sample errors that are too large. Yet, notice that the preceding two decoding processes will have “smeared out” most of the pulsive disturbances. If any band-spreading or time-spreading modes were chosen for the distributive and/or modulative encoders in place of the fastest mode, the multiorthogonal mode, the preceding decoding steps will have resulted in eliminating most of the damaging influence of Gaussian noise as well as the pulsive noise. If it is known in advance that digital input will, be applied, the distributive decoder will contain a decider that will be programmed with the optimum decision rule for the case at hand. Under such circumstances the distributive decoder becomes an error-correcting decoding system with the redundancy having been selected by the distributive encoder. Thus, we see that any such action completes the fulfillment of the requirements of cardinal principle number six.

**OPTIONAL SYSTEMS FEATURES** One of the most impressive characteristics of a TESIT system is its extreme flexibility. When applying the latest progress (1968) of LSI solid state hardware, it is possible to assemble a tailor-made TESIT system for almost all particular systems requirements. Hence, a designer does not need to feel limited in his choice of word length, coding and decoding complexity, buffer capacity, etc. A basically flexible system design will be executed in the future in hundreds of special variations, provided the basic building blocks will be designed in perfectly compatible modules.

Another feature of TESIT is the programmable character of all the three principal encoders or decoders. They all are basically small special purpose computers and they can be designed in a way that they may be programmed to operate in hundreds of different modes. An adaptive design of TESIT will permit an automatic operation whereby the system will continuously monitor its own performance, and it will adapt its mode of operation to the slowly changing, characteristics of the input information and of the transmission channel.[10] Some of the optional systems features that will support such an adaptive system design will now be mentioned.

**The Incorporation of a Return Channel** The most important optional feature in TESIT is the use of a return channel to make the system an adaptive system. Such a return channel may be a separate channel of much smaller capacity than the forward channel or it may be another forward channel in a full duplex system where each forward channel reserves a small percentage of its total capacity for return messages of the other forward channel.

The application of a return channel is shown as connection 14 in figure 1. The primary purpose of a return channel is the transmission of requests for the repetition of completely destroyed parts of forward messages. For this purpose it is necessary to provide in the transmitter a repeat buffer (9 in figure 1) where duplicates of all distributed samples are kept until they are obsolete; i.e., until the time has lapsed during which a repetition could take place. These duplicate samples are preferably stored in smaller groups, called blocks, and not in complete frames of the distributive encoder. In the present design example they are stored in block lengths suitable for the modulative encoder. In the receiver each of these blocks of samples is checked in the modulative decoder by methods to be explained later. If such a check shows heavy noise or distortions during any one of these blocks of samples, a request for repetition of the mutilated block goes at the next opportunity over the return channel back to the repeat buffer. There the stored copy of that block of samples will be selected and the block will be transmitted a second time.

The receiving system needs another repeat buffer (19 in figure 1) so that all other groups of samples can be held back until the repetition is completed. All distributed samples must finally be submitted in their correct sequence to the distributive decoder.

A secondary purpose of the application of a return channel is the transmission of short status reports from the receiver to the transmitter. Such status reports may cause the transmitter to change the mode of operation in many ways. The transmitter may elect to use more redundancy by going from the multiorthogonal mode to the multisignal mode or to the single-signal mode. Conversely, the transmitter may go to a higher rate by switching modes in reversed order. Such mode changes may take place in the modulative encoder or in the distributive encoder. The transmitter programmer alternately may elect to change the mode in the contractive encoder; for example, it may decide to tolerate more redundancy in the messages so that errors may be more clearly identifiable in the final output.

Notice that the application of a return channel meets the requirements of the second, seventh, and tenth cardinal principles.

**Average Power Control** The use of orthogonal binary sequences and of Orthogonal bandlimited waveforms offers an excellent opportunity to apply a forward-acting automatic power control system. This is shown by line 13 and line 12 in figure 1. Line 13 receives a control signal from the distributive encoder for regulating the gain of amplifier 13a. This control signal will remain constant during each frame of distributed samples, but it will change from frame to frame. The control signal will be of such a magnitude that the total energy used for each frame will remain constant, regardless of the value (height) the individual samples will assume in any particular frame. Because each frame will contain certain samples in digital form (i.e., with known amplitudes), it will be easy at the receiving side to evaluate (on an average basis over all digital samples) the value of the control signal that has been applied to any particular frame of distributed samples. The reciprocal value of this control signal can be applied to an amplifier in the distributive decoder with the result that the contracted samples leaving the distributive decoder will have been restored to their correct values.

Evidently this method will contribute to an improvement in signal-to-noise ratio (SNR) during the transmission of all those frames where a majority of samples has rather low values. The contractive encoder can be programmed to make all non-essential samples zero, thus automatically causing the essential samples to absorb a larger energy per sample.

The same principle of power control can also be applied to the modulative encoder in connection with the final amplifier (11 in figure 1). In this case the correction at the receiving side must be applied in the modulative decoder. In FM transmission channels this same process may be used for deviation control.

**Optional Multiplex Capability** The high flexibility of TESIT permits its application for telemetry and also for voice or picture transmission; it may operate with a single input or with multiple inputs. Any input may be a time-division multiplex tr i ain of hybrid character; i.e., the pulses may be quantized digits,or sampled pulses.

One particularly interesting possibility is indicated in figure 2. That figure shows a more detailed block diagram of a TESIT transmitter, which will be explained completely in one of the following sections. Here attention shall be drawn to the two inputs that should symbolize a so-called delay sensitive source (at point 30) and a delay proof source (at point 42). The first source is an analog source, such as the output of a microphone during voice conversation or the output of a mechanical sensor for vibration measurements in telemetry. In both cases there may be the requirement to process the signals with a maximum delay, not exceeding, for example, 500 ms. The second source may be a digital message source; for example, outputs of some counters or some telegraphic messages or some digitized temperature measurements of routine nature--in any case, a kind of information that may be delayed up to a minute or more.

The contractive encoder will multiplex these two sources in a way that the first source will always have priority access to the channel while information from the second source will only be accepted when there are gaps in the information flow from the first source. Such asynchronous, yet information-dependent, multiplexing methods have been suggested in the past;[11] and a prototype model of a svstern that places digital data into the gaps of voice transmission has been built. [12] Another distantly related system is Time Assignment Speech Interpolation (TASI) where individual voice channels are switched off from the transmission medium during any measurable gap in the speech waveform, while they get instantaneously a new transmission channel assigned when the speech waveform continues. [13]

The TESIT system has an advantage over these systems of prior art: the asynchronous multiplexing function is part of the overall adaptive TESIT system. Automatic trade-offs can be programmed into the system so that the multiplexing function can contract the information from several sources to a stronger and stronger degree at the cost of a gradually decreasing quality when other elements in the system (for example, pulsive disturbances in the channel) demand such an action. This may be the case when frequently required retransmissions have cut out some of the useful transmission intervals. It is this feature of "graceful degradation" that makes the optional multiplexing capability particularly attractive in connection with TESIT.

**Continuous Channel Analysis** Cardinal principle number two calls for a statistical evaluation of the transmission medium. In adaptive systems this evaluation has to be performed as frequently as possible to enable the analyzer/programmer (3 in figure 1) to update the optimum mode selection in real time.

The selection of trigonometric product waveforms [6, 16] as the bandlimited signals in the modulative encoder offers a very interesting option for channel analysis. The method is called a "noise window." It has been discovered that one waveform out, of an orthogonal set of product waveforms has a very flat energy density spectrum (white noise character). Consequently, it has been proposed to use this waveform as a noise window. This means that this waveform will never be generated by the modulative encoder. It is not used to carry any information sample; nevertheless, the noisy composite transmission signal is cross-correlated with this waveform in the modulative decoder in the receiver. The result of this cross-correlation process will be a good measure for the total noise in the channel. A new measurement can be taken during each waveform interval or after integration over several waveform intervals. Remember that all the waveforms constitute an orthogonal set of bandlimited functions. Thus they all cross-correlate to zero output. The noise window waveform will not receive any contribution from other waveforms in its correlator output, no matter what kind of information is carried in an interval. Noise and distortions, however, do not cross-correlate to zero with the noise window waveform, except for very special noise patterns that should coincide exactly with one of the other waveforms of the orthogonal set. Following this idea, a designer has only to provide a monitor for the output of that particular correlator and to let it report any essential changes to the transmitting side. Even in systems without return channel, one can use the variations at this output to give an indication of the changes in the transmission reliability (error probability).

**Adaptive Power Division for Synchronization** The use of trigonometric product waveforms as bandlimited transmission signals offers one other significant advantage--a very easy method to maintain and update synchronism.

It was pointed out in the first publication about trigonometric product waveforms [6] that in each set of orthogonal waveforms there will be one waveform with discontinuities at the ends of the interval. It is suggested that this waveform be used for synchronization. If, at the transmitter modulative encoder, the input corresponding to this "pulsive waveform" is kept constant, the waveform will smoothly continue from one interval to the other. It will alternately have positive and negative peak values exactly at the separation line between the waveform intervals. Since all other waveforms of an orthogonal set go to zero at the ends of an interval, it can be seen that only the synchronization waveform will supply energy to the receiver during a very small time interval exactly at the transition between two waveform intervals. Moreover, this pulsive waveform is orthogonal to all other waveforms of a set. These desirable characteristics make this waveform ideally suitable as a synchronization signal. Notice that no separate time slot nor any separate pilot channel is needed for this kind of synchronization. Yet, because of the orthogonal character of the whole set of waveforms, one can arbitrarily decrease and increase the amplitude of this synchronization waveform, thus dividing the total available power to a lesser degree or to a higher degree in favor of the synchronization signal. This process can be made adaptive via a return channel so that more synchronization power is demanded when synchronization difficulties are threatening or when updating of the synchronization epoch is required (in mobile systems). More power can be assigned to the information-carrying waveforms as soon as there is a firm locked-in condition.

**AN OVERSIMPLIFIED DESCRIPTION OF THE BASIC OPERATION OF TESIT** An adaptive system of the complexity of TESIT is difficult to describe, due to the many different modes of operation that such a system may assume. For the benefit of readers who are not familiar with the many facets of an adaptive communication system, we shall now try to give an oversimplified description of a typical operation by using an example with reasonable numbers.

The first encoding process, the contractive encoding, is explained in figure 3. We assume that a typical analog telemetry signal of less than 1 kHz information bandwidth is entering in point I in figure I and is being sampled 2000 times per second. All these samples go into the contractive encoder (31 in figure 2). The encoder must process these samples and send them in the form of “contracted words” to a “sample-and-hold” buffer B, which is shown as 34 in figure 2. Figure 3 shows a 56-ms-long time interval of the analog input. The first 12 ms is an active period that will best be transmitted in a continuously sampled mode. although any of the predictive compression modes could be used likewise. [14] For the purpose of this example we assume that the words at the output of the contractive encoder contain 8 samples each, plus 2 ternary digits called a “flag. 11 In an actual system these words will have 12 to 24 samples to keep the percentage of the redundancy for the flag smaller. For simplicity of the following simulation in numbers, the samples are represented by integers +9, +8, ...0. ..-8, -9. In an actual system they may assume any real number between the plus and minus peak values.

A typical flag code is shown in figure 3. The simplest contractive encoding procedure calls for the omission of words when the channel is inactive. In our example that is the case from 13 to 32 ms on the time scale. Five words can be omitted; this is indicated by the flag, +-, at the third word. When reading this flag, the contractive decoder in the receiver will wait 20 ms after having decoded and delivered the third word before delivering the continuation of the output signal. The ternary digits are transmitted as bipolar samples of maximum height (+9, 0, -9). It will be shown that this procedure gives them a larger share of the transmission energy and thereby offers a higher protection to the important information conveyed by the flag. At the designer’s discretion a binary code or a code of higher order may be used for the flags. As will be seen, the system handles flags or any other digits (quantized samples represented by integer numbers) exactly in the same manner as it handles analog samples (real numbers). The only difference for handling digits is that the final detector (reading device) will have one or more thresholds, preferably of the variable kind, to be adjusted manually or by program.

The flag code in figure 3 shows examples of many other contractive encoding modes. One mode is used for the fifth and sixth words. The flag of the fourth word predicts by the code, -+, that a service word from another channel (in this case, a data channel) will follow. This again is an oversimplification. The data channel (42 in figure 2) symbolizes any number of other channels. The flag could be used as an address and words could be selected from low capacity data channels with sporadic activity. In this manner TESIT would work as an asynchronous multiplex system. [11] The example in figure 3 shows that such side information from data channels is inserted into the main analog channel whenever the latter displays low activity. Figure 2 shows that the acceptance of such side information is adoptively dependent on the level of contracted words in the message buffer (34). If there are less than 10 words in the buffer, a signal goes over line 45 to ask for such side information, here symbolized by the term “service messages.” If, on the other hand, the message buffer is being filled much faster than the system can transmit the words, a signal goes over line 38 to the compression programmer, which will adapt by selecting an encoding mode with higher compression ratio. The switch in the encoding mode will be communicated to the receiver by an “inner service word” to enable the contractive decoder to switch the mode at the correct instant. These inner service words will be particularly protected by redundant digital codes. They are called inner service words because they will be encoded in the distributive encoder in the same way as all the contracted information words. In addition, the system requires “outer service words” that are added only after the distributive encoding process; they are branched out in the receiver before the distributive decoding process (63 in figure 2 and 137 in figure 7).

Two further adaptive procedures may be mentioned in connection with message buffer 34 in figure 2. The one procedure is a kind of overflow protection. When a dangerously high word level (50 in our example) is reached in the buffer, line 46 orders the repeat controller to reject repetitions. The other procedure is a kind of protection against the other extreme; i.e., against an interruption of continuous operation by an empty message buffer. This is a natural situation at the beginning of an operation or during any interruption of the input from the main channel. Connection 47 signals this event when less than three contracted words are in the message buffer. A special null frame generator inserts dummy words into the system until at least three contracted words are assembled in the message buffer. The dummy words are flagged as inner service words, recognized by message reader 162 in the receiver (figure 7), and, consequently, are dumped.

Returning to figure 3, one can see that the fifth word has the flag, indicating that another word follows from the data channel. This sixth word has the flag, -- , indicating that an outer service word will follow with an announcement of a change in the encoding procedure. The code for such change announcements may be designed specifically with the special application in mind for which the system will be programmed. Indeed it may be foreseen that at least compression programmer 36 and service generator 41 in figure 2 and their counterparts 162 and 109 in the receiver (figure 7) will be programmed computer-like devices. It will then be easy to program any TESIT equipment for any special application. Assume, for example, that the analog input to the TESIT transmitter is actually a timedivision multiplex train, where pulse position modulation (PPM) channels are intermixed with pulse amplitude modulation (PAM), with analog, and with digital channels. The designer may then program a digital transmission of the position and/or the’ height of a pulse into his contractive encoder. The encoder would identify the position within a given interval and the height of the pulse, and would encode this information into a code word such as the seventh word in figure 3. In this particular example the code has three groups of ternary digits:

- A group of two digits indicates the length of the interval to be encoded (5 words in this case, represented by the combination, +-).

- A group of three digits indicates the position of the pulse within the interval (here, in the second fifth of the third word, represented by the code, +0+).

- A group of three digits to indicate the polarity and the amplitude of the pulse (here, amplitude minus seven in a scale, -9 to +9; this is expressed by the code, -- 0).

Finally, we see that the flag of the seventh word in figure 3 indicates that a service word will follow with data indicating a gap of more than 5 words. Figure 4 continues this exercise for an interval of 250 ms, incorporating at the beginning the first 56 ms of figure 3. It is the purpose of figure 4 to go through one complete cycle of operation including a retransmission. Any details of the word contents are therefore omitted. However, the figure shows the timing and the buffer levels with fair details. Note that at the end of the 56-ms interval described in figure 3 there follows a longer period of a constant high positive level at the input to the main channel (lowest line in figure 4). The contractive encoder operates with a processing delay, $\Delta_1$, which is assumed to be 28 ms (7 standard word intervals) in our case. In a large scale system it may be much longer in terms of the number of samples involved. (Here the contractive encoder will take only a maximum of 56 samples into one observation and processing interval). This will be sufficient to conclude that the long positive constant level from 60 to 82 ms will be better encoded by a differential process (first order prediction). The eighth word will therefore be a standard word with 12 samples transmitting the slope (56 to 60 ms) in full detail. The ninth word will be a service word announcing the code change to a first order prediction code and no further transmission will be required until the constant value starts changing again. Before continuing with the discussion of figure 4, we must understand the distributive encoder and the adaptive interaction between all TESIT functions.

Figure 2 shows that the distributive encoder in our example uses a 32 by 32 element matrix. Three contractive words, W1, W2, W3, and their flags, F1, F2, F3, are transferred from the message buffer to special buffer 49. Two “distributed flags” 55 are added. The total of 32 samples forms the input to one distributive frame. Writing the distributive flags first, one arrives in our example at the column of samples shown at the left side of figure 5. The three words are the first three words formed in figure 3.

Using this input the distributive encoder forms a new sequence of 32 samples simply by multiplying each input sample with every element of an orthogonal binary sequence and then adding all the products column by column. The sums of each of the 32 columns are the new 32 samples. This process has been called a multiorthogonal encoding process. It can be easily simulated on paper as it is shown in figure 5. A set of orthogonal binary sequences required for this process can be generated in many different ways. In this example an orthogonalized binary pseudo-noise (PN) sequence is used. [15] All binary, elements in such a sequence are either +1 or -1. The first sequence of the set is a sequence of 32 positive elements. It has to be multiplied with the first sample--in this case, of zero value. All products in the first row are zero in figure 5. The second sequence and all the following sequences start with a positive element that is followed by one phase of the PN sequence. This first positive element in all sequences is the reason that the first column is always the input sequence itself. The algebraic sum of all numbers in this column is therefore the sum of the input sequence itself (cumulative dc value). It is +36 in the case of our example. The sums of all the columns are marked as $s_1$.... $s_{32}$ and their numerical values are marked on the right side of figure 5.

The second orthogonal binary sequence is written out on the top of figure 5. It can be formed by putting 5 minus symbols into the second to sixth element and then forming always the n-th element by the modulo two addition (the so-called half adder operation) of the (n-3)th element and the (n-5)th element. Thus the seventh element is $- \oplus - = +$. The eighth and ninth elements are also +. The tenth element results from the modulo two addition of the seventh element (+) and the fifth element (-). By applying the truth table on the right, the reader can easily continue the operation until the whole sequence is formed.

| $\oplus$ | - | + |
|---|---|---|
| - | + | - |
| + | - | + |

The second row is the multiplication of the second input sample (+2) with the second binary sequence as shown at the top. The third binary sequence is formed by shifting the PN sequence for one element to the right and taking the last element of the original sequence to the second place. The fourth binary sequence is again started by a positive element and is followed by the second phase shift of the PN sequence. Accordingly, one arrives at the last sequence being a positive element followed by the n-2 shift of the PN sequence of n-1 elements, or, if one prefers, by the PN sequence shifted for one element to the left. Notice, however, that the last input element has a negative sign. Therefore, all products in the last row have the opposite signs of the last orthogonal binary sequence.

The sums of all columns (new sample values) are marked at the right side of figure 5. The largest sum is negative (-60). The smallest absolute sum is 2. It is always useful also to form the squares of all new samples (sample power). This is done at the utmost right. The sum of all these squares represents the total energy of one distributed frame, 22976 in this case. Dividing this number by 32 gives the average power of a sample. It is interesting to note that this average power must be 32 times the average power of the input samples, no matter what message is being processed. The root of the average power is the root mean square (RMS) level of the samples. It must be $\sqrt{32}$ times the RMS level of the input samples.

The two distributed flags at 55 in figure 2 are for the purpose of transmitting a measure of the RMS value (or peak value or average value, depending on the application) and for permitting a noise check. The first objective is desirable whenever the structure of the frames may be significantly different from one to the next. In this case it may be desirable to adjust the gain of an amplifier anew for each frame so that the transmitter power may be continuously used in the most efficient way. The second objective is achieved by keeping one input to the distributive encoder permanently at zero level. Any output signal at the corresponding output terminal indicates the presence of disturbances (or cross-talk). Again the selection of only two such distributed flags is imposed by the small total number of elements in all the encoding processes. If very large frames were used, more such flags could reasonably be used to communicate with higher precision between transmitter and receiver. Also it is conceivable to use the absolute value of those digits that are known to be of constant absolute level (binary or ternary digits) as an indication of any automatically operating amplifier adjustment. As there will be a number of such digits in most frames of greater length, one can make a statistical evaluation at the receiving side and use the RMS value of such digits as scaling factor for restoring the correct levels of all samples at the receiving side while using the mean square deviation from this value as a measure of the residual noise in the received frame.

Returning to figure 2 one can recognize that the new distributed samples leave the distributive encoder via connection 53 and amplifier 54 until they are stored in serial-to-parallel converter 56. There they are split into words of smaller length, suitable for the modulative encoder. These words are now called ‘modulative blocks’ to distinguish them from the contracted words and the distributed frames. In particular they may be called modulative blocks at the input side of modulative encoder 33 and modulated blocks at the output side of the encoder. Notice that modulative blocks consist of discrete samples while modulated blocks consist of analog bandlimited waveforms that are continuous within one block, each waveform having been continuously modulated in at least one of its parameters with one of the samples of a modulative block.

In the present example the 32 samples of a distributed frame are loaded into seven modulative blocks in the following manner: the first two distributed samples of a frame are carried in a start block and again in an end block. The other 30 samples are carried in 5 modulative blocks that are interleaved between the start and the end blocks. The start and end blocks carry, in addition to the two distributive samples, four service digits indicating in a special code all the information that is necessary to operate an automatic repeat request (A RQ) system over a return channel.

Although any kind of bandlimited waveform can be used in the modulative encoder, sets of orthogonal bandlimited waveforms are recommended. In the present conceptual TESIT design, trigonometric product waveforms are preferred. [6, 16] Figure 6 shows the waveform pictures of a set of eight orthogonal product waveforms with a nominal

WT-product of 4. A frequency synthesizer (58 in figure 2) provides the fundamental frequency, $f_o$, and the second and fourth harmonic, all in sine and in cosine phase with respect to the central clocking phase supplied by timer 71 in figure 2.

Again it must be stressed that the particular numbers are an oversimplification. A system will operate more efficiently with sets of 32 or 64 orthogonal waveforms. Yet, all the fundamentals can be better explained with an example of only eight waveforms. Notice that only six of the eight waveforms are used to carry samples. The seventh will be used for synchronization and the eighth is never transmitted. It is, however, used in the receiver correlator to measure the noise content of each block.

For synchronization it is advantageous to use the waveform with a sine factor of the fundamental frequency and cosine factors in all other harmonics. This is the only waveform in any set with a discontinuity at the ends of the interval. If the modulative encoder input S at 67 in figure 2 receives a constant value, this waveform will not be interrupted at the ends but will produce alternately positive and negative peaks at the separation line between blocks. At these instances all other waveforms have zero amplitude so that the modulative blocks can be changed without causing transients.

If an orthogonal set of waveforms is used in the modulative encoder, the designer has the choice of transmitting one waveform at a time or of superimposing several or all of them in any waveform interval. If only one waveform is used in each interval, the full average power (or peak power or peak deviation in FM) can be assigned to that single waveform. If more waveforms are used simultaneously, the available power or deviation must be shared. The extreme case, when all waveforms are used all the time, is called the multiorthogonal case and it is the one used in the present example. It will be preferred when the SNR is relatively good and the bandwidth is costly. The other cases require that the waveform intervals be made shorter than the block intervals, since several waveforms have to follow each other in time division during any block interval. Accordingly, the bandwidth must increase in proportion to the number of waveform intervals that are required for each block interval. The system operates in a bandspreading mode.

Evidently there are many different encoding procedures, depending on the number of waveforms used in each interval. The simplest case is the one applied in this example where each sample in buffer 57 modulates the amplitude of one of the six waveforms. All waveforms are then superimposed in summation network 59. The waveform interval is identical with the block interval. Since all the waveforms are orthogonal to each other, they form a composite signal similar to that in a frequency division multiplex system where each component can be raised and lowered in amplitude, independently of the other components. Thus, one can indiscriminately use continuous samples or fixed amplitudes for binary, ternary, or higher order digits at any one of the input terminals (57 and 67 in figure 2). This characteristic can be used to adjust the share of the synchronization power in an adaptive mode. A special sync amplifier will control the sync power according to the information to be received over the return channel from the other side. High: sync power will be supplied during the initial acquisition phase or when operating with low SNR in the receiver (slant range). Relatively small sync power will be required under constant operating conditions with good SNR.

Modulative blocks that have been transmitted are not discarded; they are ‘ kept in repeat buffer 35 whence they can be selected for retransmission when so requested by the other station.

The receiver block diagram of figure 7 shows the complementary subsystems of the transmitter. Modulative decoder 102, distributive decoder 103, and contractive decoder 104 are self-explanatory. Two devices, however, have no counterpart in the transmitter. These are noise sampler 122 and comparator 133.

The noise sampler receives the output from the correlator that samples the “noise window.” As explained before, one should never transmit one of the orthogonal waveforms with the uniform spectrum; for example, the eighth waveform of figure 6. That particular waveform has a nearly uniform spectrum over the transmission band. It will therefore be best suited to sample the white background noise. This is done by cross-correlating in an integrate-and-dump correlator a locally generated waveform of this kind with the incoming noisy composite signal. Because all information-carrying components of the composite signal are orthogonal to this noise sampling waveform, the special noise correlator output, N (in 113), will contain contributions only from the noise- -not from the composite signal. The noise window enables the system to ‘look through’ the useful signal and to watch continuously the noise only. Distortions of the composite signal will likewise cause some indication in the noise window. An excessive output from the noise window during a single block will cause a repeat request (via 124) for that one block only. A repeatedly high output from the noise window will cause a change of the encoding modes or a readjustment of the repeat threshold (via 123).

The comparator (133) has the task of comparing the start block with the end block. It was said before that start block and end block carry the service information for operating the ARQ system. It is particularly important that this information be well protected. Start block and end block are therefore identical and the comparator assures that the maximum advantage is taken from operating in this time diversity mode. Simultaneously, it assures that the beginning and end of each frame are duly recognized. The comparator is also used to compare any original block with its retransmission, again to make maximum use of this kind of adaptive time diversity operation in the ARQ system.

It is impossible to explain in this paper in full detail all the other features of figure 7, but it is hoped that, with the help of the extensive explanation of the transmitter, the reader will be able to identify the corresponding blocks in the receiver.

We may now return to figure 4 to go completely through the example of an operational phase of 250 ms. The symbols used in figure 4 are explained in the legend on that figure. The numbers in the squares and circles indicate the continuing sequence of contracted words entering buffer 34 in figure 2. On the bottom of figure 4 one can see in the first 56 ms the same input signal as has been used in figure 3. The contractive encoder takes eight samples during any 4-ms period. The total processing delay of the contractive encoder is 28 ms. This means that as many as seven such 4-ms periods (i.e., up to 56 samples) are simultaneously in the contractive encoder. At the end of this 28-ms period a decision is made, and, if necessary, a contracted word will leave the contractive encoder. Thus, we see that the first, second, and third words go in 4-ms intervals into buffer 34. If the buffer does not contain any other words from previous encoding periods, null frames will be taken from generator 48. Such null frames will contain special inner service words to indicate. to message reader 162 (figure 7) that those words have to be dumped.

Notice that figure 4 indicates the number of contracted words that are at any time in buffer 34. They can be counted by drawing a vertical line at the desired time instant and counting the number of horizontal lines that are crossed. At instant 74 ms, one may count 6 words in the buffer. The crossings are marked by small circles. At instant 218 ms, there are 13 words in the buffer. Thus, one can see that a diagram such as the one in figure 4 clearly displays the adaptive operation of the system.

We assume that the system in our example will be programmed to form a distributed frame as soon as there are at least three contracted words in buffer 34. It can be seen that this situation is first achieved at 44 ms. Naturally, a distributed frame will be encoded only when all words of the previous frame and all repeated blocks have left buffer 56. The distributive encoder may then use a time interval equivalent to one modulative block length to encode a new frame and to deliver all distributed samples to buffer 56. It is assumed that the first such opportunity will happen at 76 ms. The large triangle A symbolizes that the first frame containing words number 1, 2, 3 will be formed and delivered in a maximum processing time of 4 ms. From 80 ms to 108 ms the small triangles indicate that a start block, five message blocks, and an end block will be encoded by the modulative encoder. These blocks will leave the transmitter in time sequence. While the modulative encoder will process the end block from 104 to 108 ms, the distributive encoder will process the next frame, marked B.

The reader will now be in a position to follow the rest of the operation. Three interesting events may be pointed out.

(1) The word nine is an inner service word indicating the switch from standard contractive encoding to first order prediction. Therefore, no contracted words will leave the contractive encoder from 96 ms to 112 ms. During all this time there are still at least three words in buffer 34. No null frame will be inserted. At 136 ms the contractive encoder sensed that the main input was inactive for about 50 ms. It therefore calls for words from the side channel (13 to 18).

(2) A particularly interesting operation is the ARQ procedure; i.e., the repetition of mutilated blocks. It has been mentioned previously that the noise window (or error detection codes in digital operation) will indicate to the receiver the desirability to call for the repetition of a block. To show one such operation we assumed a one-way transmission delay of 20 ms, marked by small crosses at the top of figure 4. The small triangles with falling slope indicate the periods when the received blocks will be processed in the modulative decoder in the receiver. Assume now that noise contaminated the second block of frame A. This is marked by the letter N in figure 4. At 116 ms a repeat request goes via 124, 119, 128, 129, and 130 back to the transmitter of figure 2. There it arrives over 61 and 62 at the repeat controller. The request is read and the word that is to be repeated will be selected by 63 and transferred to buffer 56. It will be transmitted after the end block of the frame currently in transmission. Figure 4 shows that the retransmission takes place from 220 to 224 ms. The corresponding diagram of the receiver operations (not enclosed) would show that selector 140 in figure 7 would have transferred the originally received block number 2 of frame A to buffer 127 at the time when the repetition of this word would arrive from 114 via 126 at the same buffer. Comparator 127 would then make the best weighed decision, using the information from the original transmission and from the repeat transmission along with any additional information gained from noise analysis (not shown in detail).

(3) The contracted word number 19, marked T, is a special inner service word, transferring a time reference. This may be necessary in complicated encoding modes, where it could happen that certain errors could cause the receiver decoders, particularly the contractive decoder, to lose track of the correct time line when counting omitted words or when switching decoding modes. The transmission of a time reference word will be necessary to correct the receiver timing. The reference word will carry a code indicating where, placed on a time line, the word originated (in our case, 176 to 180 ms). The receiver can correct its own timing when the reference word does not appear at the correct place. If an absolute clock is available at the transmitter, such time reference words may be initiated; for example, every

second at 000 ms. This procedure is particularly important in mobile systems with variable transmission delay.

**A Numerical Decoding Example** It is very instructive to demonstrate the noise reducing characteristics of the distributive decoding process by means of a numerical example. It may be recalled that the purpose of the distributive encoder is the "smearing out" of the information contained in any one sample over all the other samples of a frame. It has been claimed in this paper that this distributive encoding process offers protection of the information against pulsive noise when applied in the fastest mode, the multiorthogonal mode. It has also been claimed but not discussed in detail, that a certain amount of protection against Gaussian noise may be achieved when applying the distributive encoding process in a slower mode; for example, one of the multisequence modes.

It shall now be demonstrated in figures 8, 9, and 10 that there seems to be a good basis for such claims. At the present time no rigid analytical investigation of the multisequence modes and of the multiorthogonal mode is known to this author. It is hoped that such an analytical treatment of the problem will be performed in the near future.

The multiorthogonal encoding process has been demonstrated numerically in figure 5. The output of the distributive encoder after performing this process is the 32 samples listed on the right side of figure 5 from $s_1$ = +36 to $s_{32}$ = +2. The RMS value of this sequence of samples is 26.796. The reader is invited to modify any one of the input samples and to observe that surely all 32 output samples will change. Raise, for example, the third input sample from +1 to +2 and one can see that the third row will change to +2, +2, -2, ... -2, +2. Correspondingly, the sum of the first column (i.e., the first output sample) will change to +37; $s_2$ will change to +17; $s_3$ will change to +19; and so on to $s_{31}$, which will change to -25; and $s_{32}$ will change to +3. If the output sequence is reduced to the same RMS value as the input sequence (i.e. , to 4.737), one can see that each output sample will only be changed by plus or minus 0.177 when one of the input samples is changed by ±1.0. The process is actually smearing out the information of any input sample over all the output in such a way that each output sample contains a small contribution from each input sample.

This sequence of distributed samples, $s_1$, $s_{2,}$ ... $s_{32}$, is shown in the second column in figure 8. Gaussian noise is now added in a simulated paper experiment. The third column in figure 8 is a random selection of noise samples from a Gaussian distribution with zero means. This particular random selection (noise record) has a mean value of +0.172, an RMS value of 4.85, and a peak value of +10.8. As the signal sequence has an RMS value of 26. 8, one can easily calculate that the SNR is +14.8 db.

The fourth column shows the signal-plus-noise of the received samples. This sequence is now submitted to the distributive decoding process and the output samples of the decoder are reduced in size (32:1) so that the output RMS value in the noise-free case would be the same as the original input RMS value, (4. 737). Comparing each decoded sample in the fifth column with the corresponding input sample in the first column shows the error of each sample. All the errors are written out in the sixth column. As to be expected, the RMS error is exactly the same as the RMS value of the noise when accounting for the power reduction after the decoder (32:1). It is, however, interesting to note that the peak-to-average ratio of the error is much smaller than the peak-to-average ratio of the noise record.

Though the encoding process is basically operating in the multiorthogonal mode, one can see in the same example the advantage of the multisequence mode. The latter may better be designated in this case as “sample diversity.” Assume that the 9th, 110th, 13th, and 14th input samples all carried the same information. One sample had been submitted simultaneously to four encoder inputs. Under this assumption one would transmit the same information four times and one could call this process sample diversity. If the receiver had full *a priori* information about this mode of operation, it would take the mean value of the corresponding four received samples as the best possible estimate . . Evidently, such a sample diversity mode is fundamentally identical with any wellknown time diversity or frequency diversity mode. The advantage is primarily in the flexibility that the system offers. Diversity operation can be introduced at any time without any change of any fixed parameters of the system. It can be programmed into compression programmer 36 (figure 2) and into expansion programmer 109 (figure 7) without requiring any change in the structure or the hardware of the system. The switching of the diversity modes can be done adoptively during the operation of the system. This is particularly desirable when the system has to operate in the presence of intentional disturbances. The extreme case of such diversity operation is reached when the same sample value would go to all 32 (or at least 30) inputs of the distributive encoder. The received decoder can then calculate the expected value of all the noisy outputs and it would eliminate from the calculation those output samples with the largest deviation from the expected value. Notice, however, that when using orthogonalized PN sequences as encoding sequences, the sample must be put in positive polarity to half the inputs and in negative polarity to the other half of the inputs. This measure will avoid an excessively large sum in the first column, while all the other columns will be zero.

Figure 9 shows the same numerical example for the case of Gaussian noise plus pulsive noise in the channel. The first three columns are identical to those of figure 8 except that the noise sample marked A is taken as a high impulse of 6.3 times the RMS value of the Gaussian component. The example shows clearly how much,this pulsive disturbance is flattened out in the decoding process. The largest error is now -2.72 (i.e., only 0.58 times the RMS value of the noise-free sequence), while the noise pulse was 30.6 or 1.14 times the RMS value of the undisturbed sequence in the channel.

Figure 10 shows that, even under the influence of a noise burst that would catastrophically contaminate five of the transmission samples, the decoding process can flatten out the influence of the noise. In this last simulation example, the SNR is only 6.54 db. The peak-power to averagepower ratio of the noise is 10.75 db, while the peak-error to averageerror ratio has been reduced to 7.2 db. The largest noise spike in the transmission channel is 41.1 or 1.5 3 times the R MS value of the signal, while the largest output error is 6.12 or only 1.08 times the RMS value of the output samples. Ternary digits (the last two input samples) would be detected correctly despite the low SNR and the pulsive noise character in the channel.

Despite these good results on numerical examples, the reader should be warned that more fundamental analytical investigations are required before generally valid results can be made available. For example, a devil’s advocate could always invent a noise record (sequence of noise samples) that, at least in the polarity sequence, would completely match one of the 32 orthogonal sequences. In such a case a very large output w6uld occur in only one of the correlators--due to the noise and not due to the signal. An error would result. There are special cases where the receiver decoder would actually increase the peak-to-average ratio of the noise instead of decreasing it. Evidently, the longer the encoding sequences are, the less will be the probability that any such coincidence could ever happen. Moreover, in any TESIT system the modulative encoding process has a further randomizing influence and the other adaptive procedures, such as the noise window together with the ARQ procedure, will eliminate any periods with excessive noise. Exactly how well the combination of all these measures will really work will have to be studied by simulation. The precise analytical evaluation would be too complex.

**CONCLUSIONS** it has been demonstrated that the realization of an integrated information transmission system operating on a sampled basis is possible with methods known today. Ways have been shown how to implement in a single system all the ten cardinal principles discussed in the introduction. So far the system that has been discussed in this paper is only a conceptual design. Many more investigations of an analytical and experimental nature are required before it will be possible to estimate the cost and performance of such systems. Simulation studies will be needed to arrive at trade-off curves for the many design parameters. Yet, this author is convinced that adaptive systems along the lines of TESIT will be essential for the conservation of one of our most precious national resources, the frequency spectrum.

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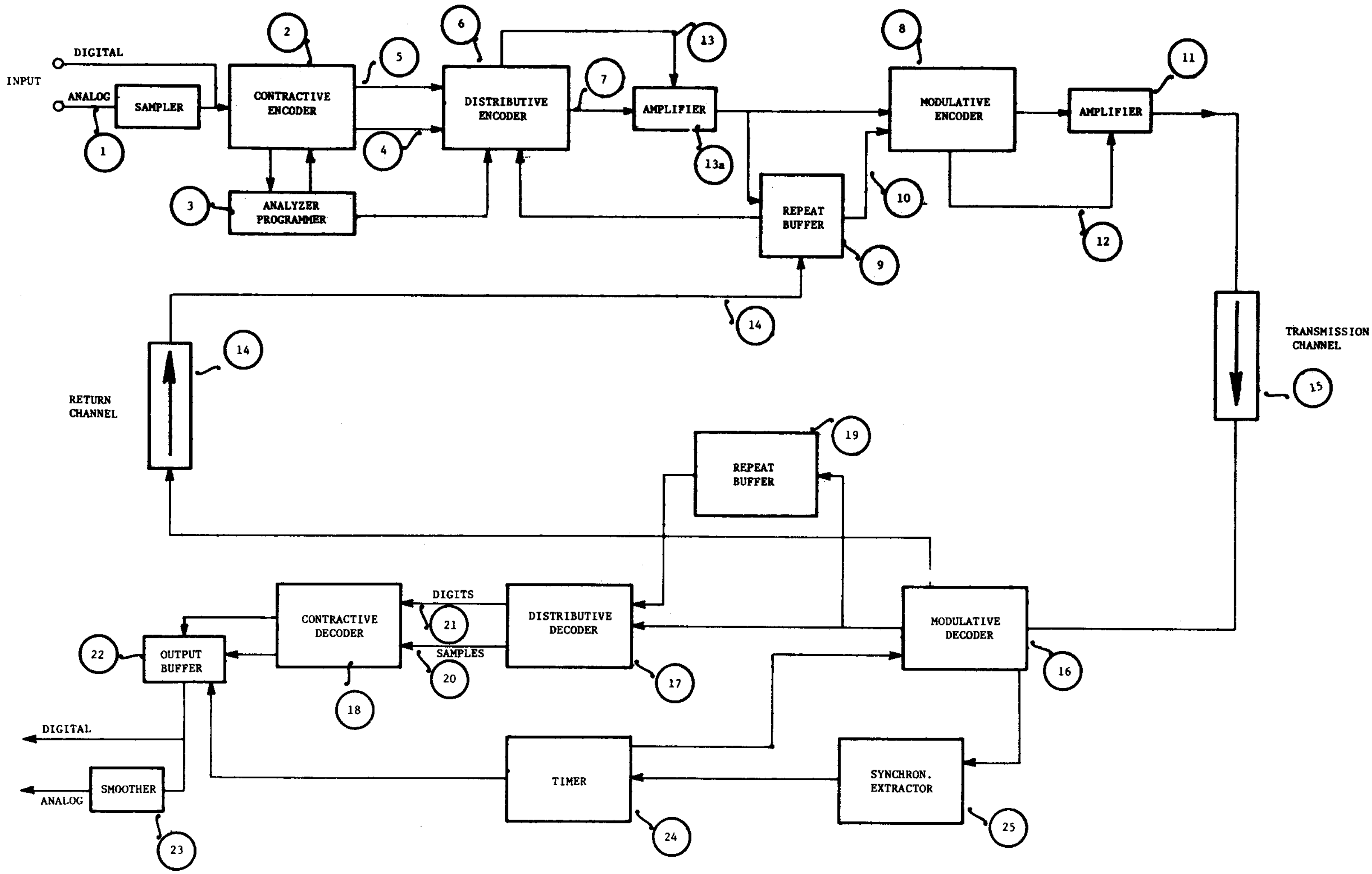



**Figure 1. Simplified Block Diagram of TESIT**

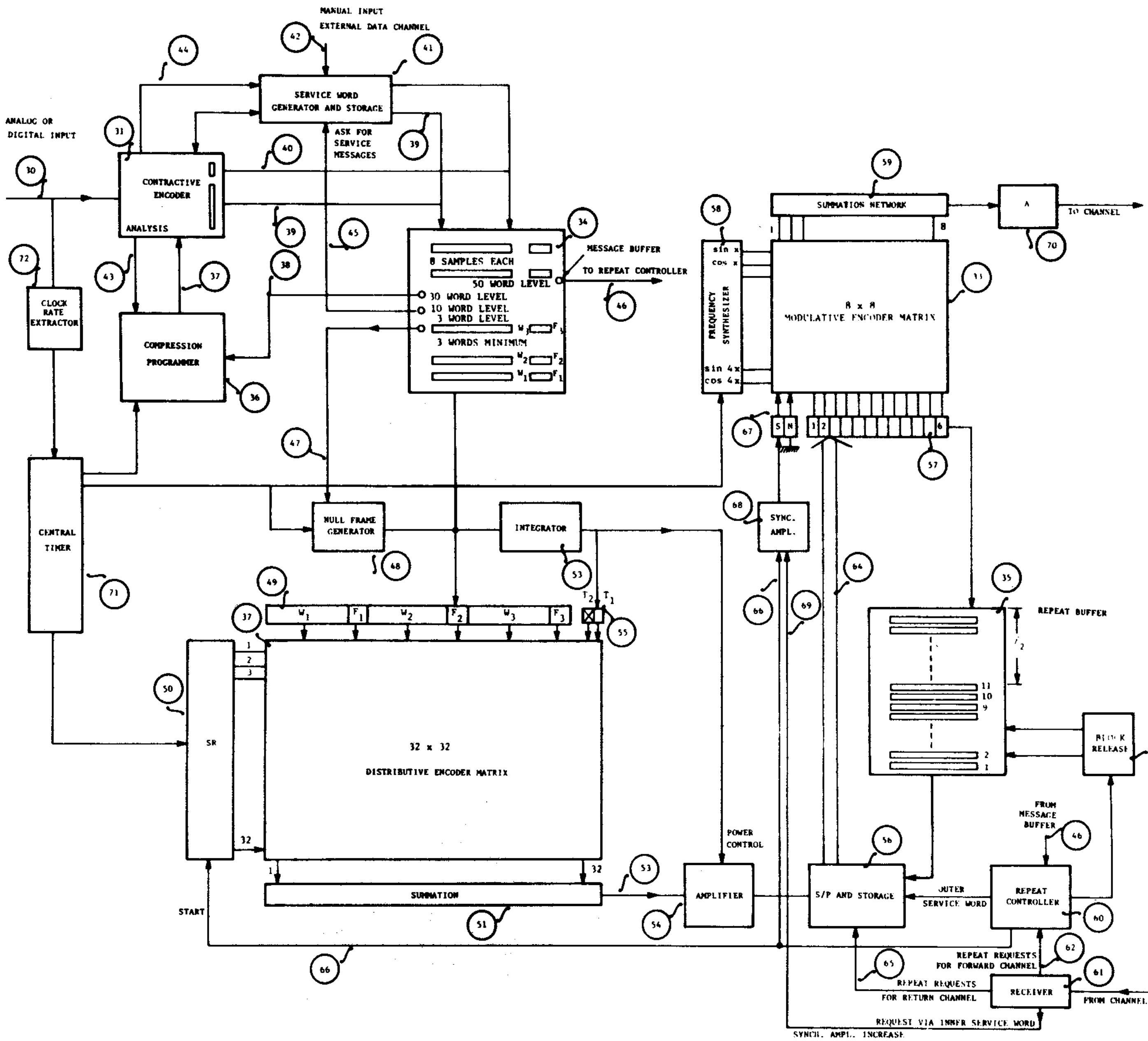



**Figure 2. Complete Block Diagram of a TESIT Transmitter (Simplified Example)**

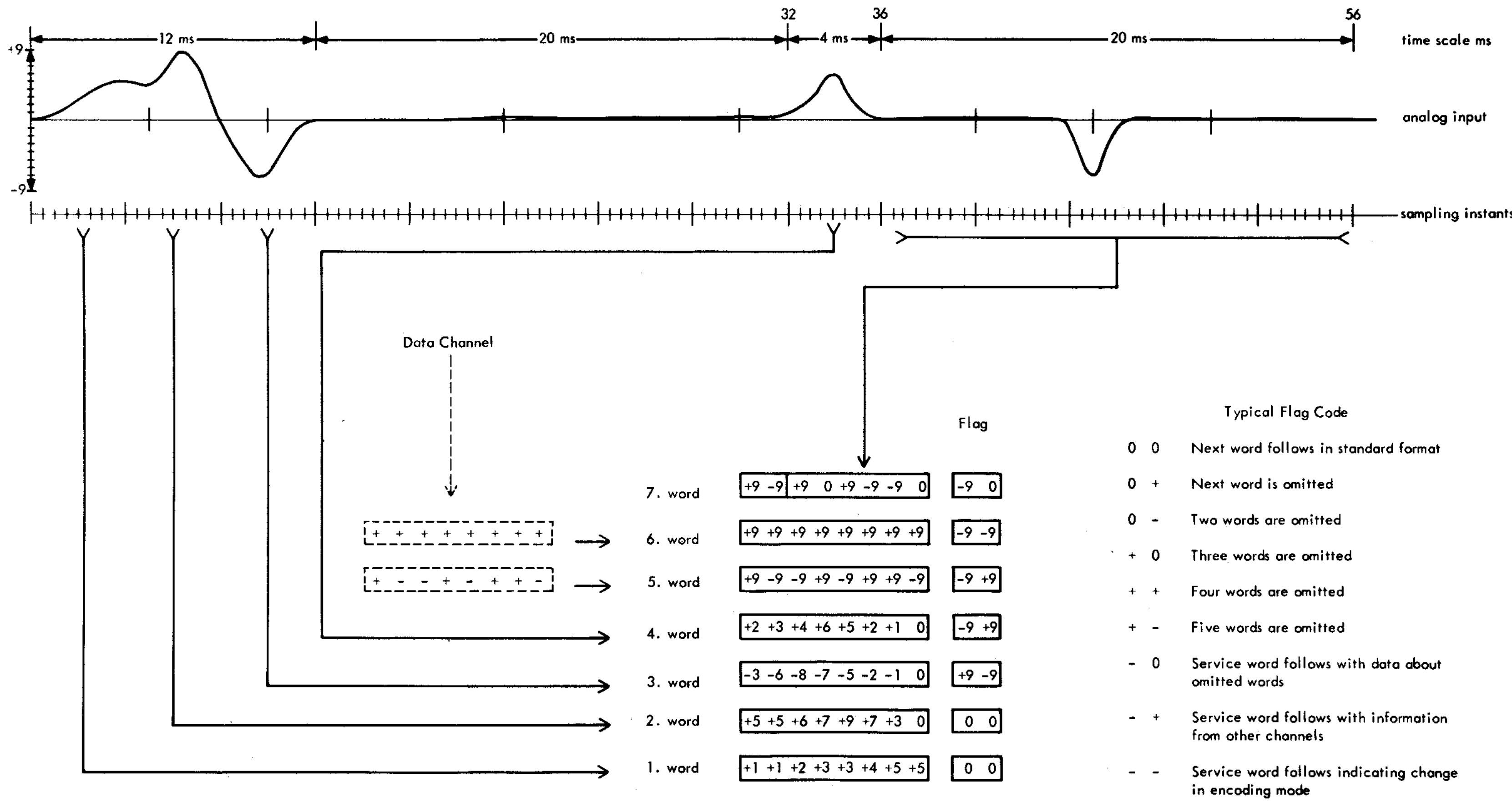



**Figure 3. Encoding of 56 ms of a Test Message**

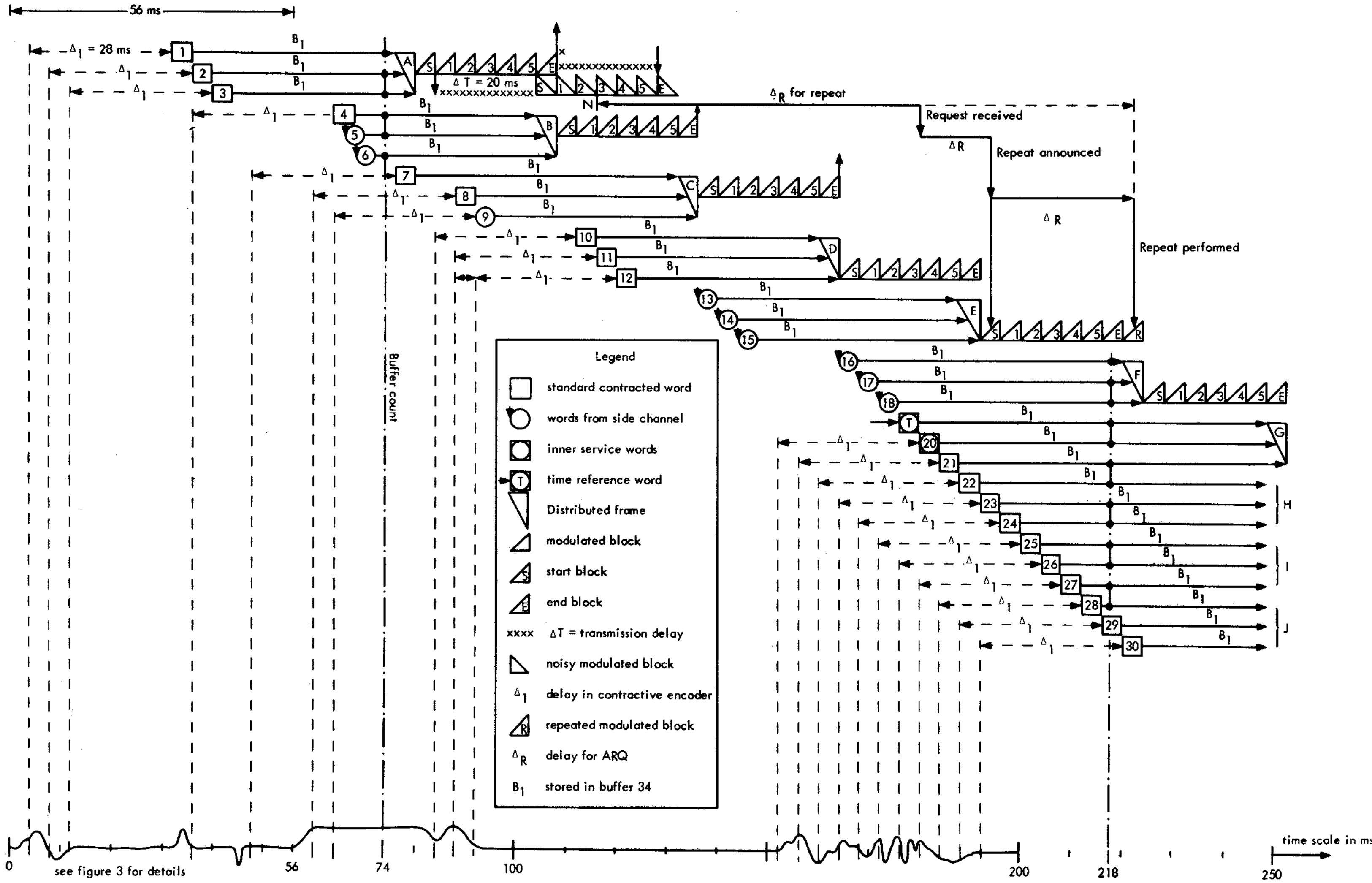



**Figure 4. Encoding Operation Lasting 200 ms**

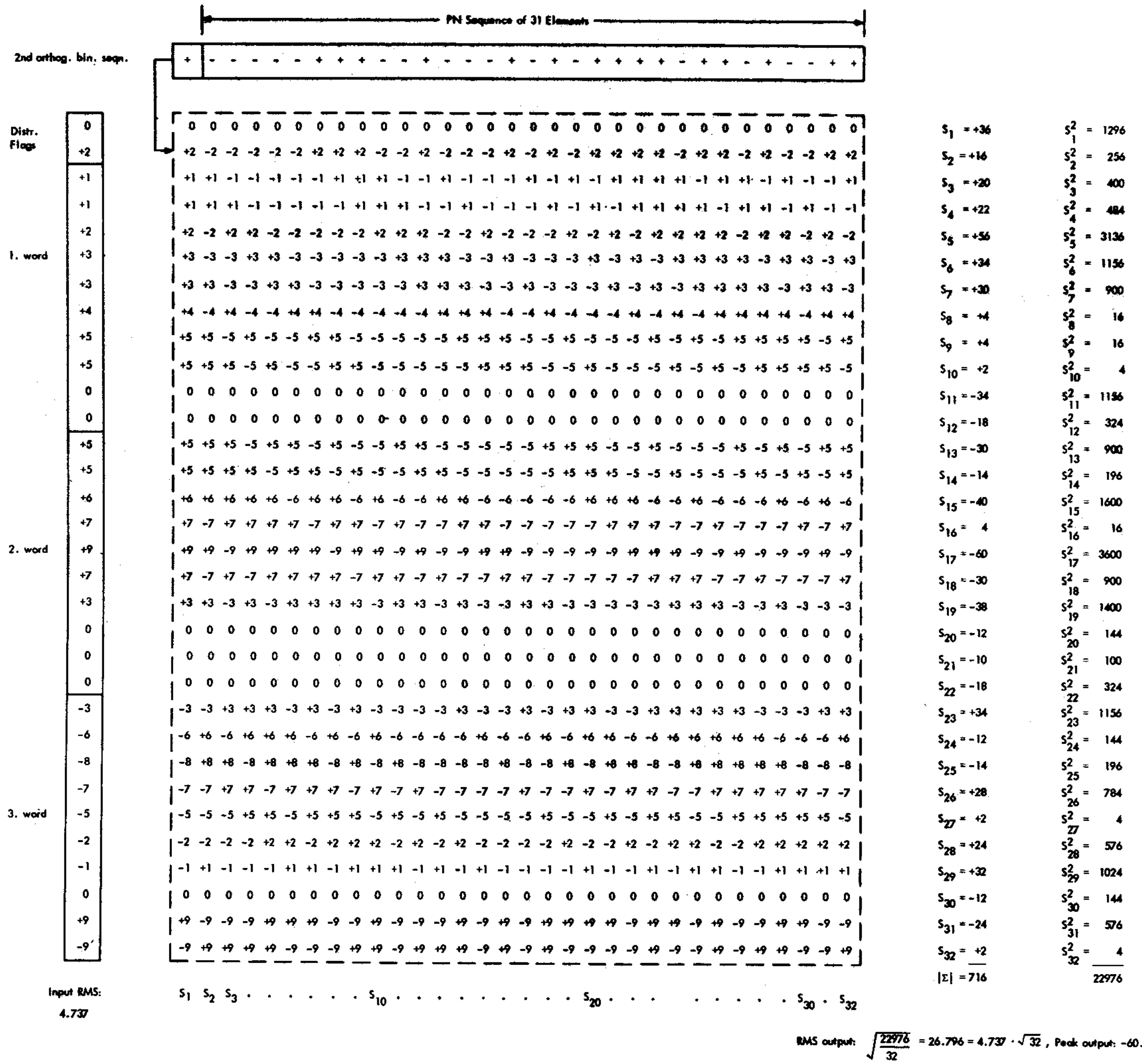
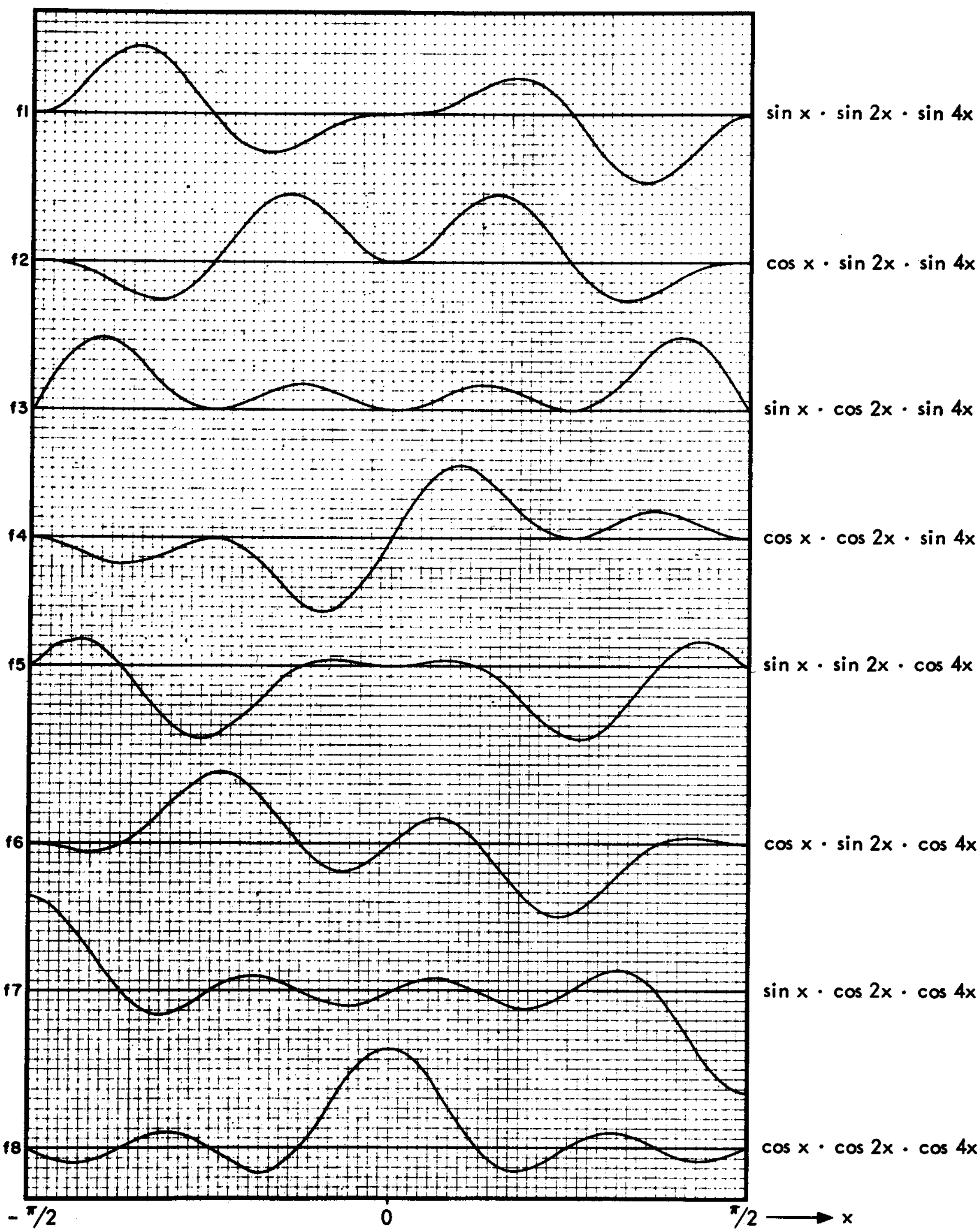



**Figure 5. Example of a Distributive Encoding Process**




$x = \omega_o t$; $\omega_o = 2\pi f_o = \frac{2\pi}{T}$; T --- Waveform Interval

Essential Bandwidth $W = \frac{4}{T}$; Number of Waveforms $n = 2WT$

**Figure 6. Set of 8 Orthogonal Trigonometric Product Waveforms**

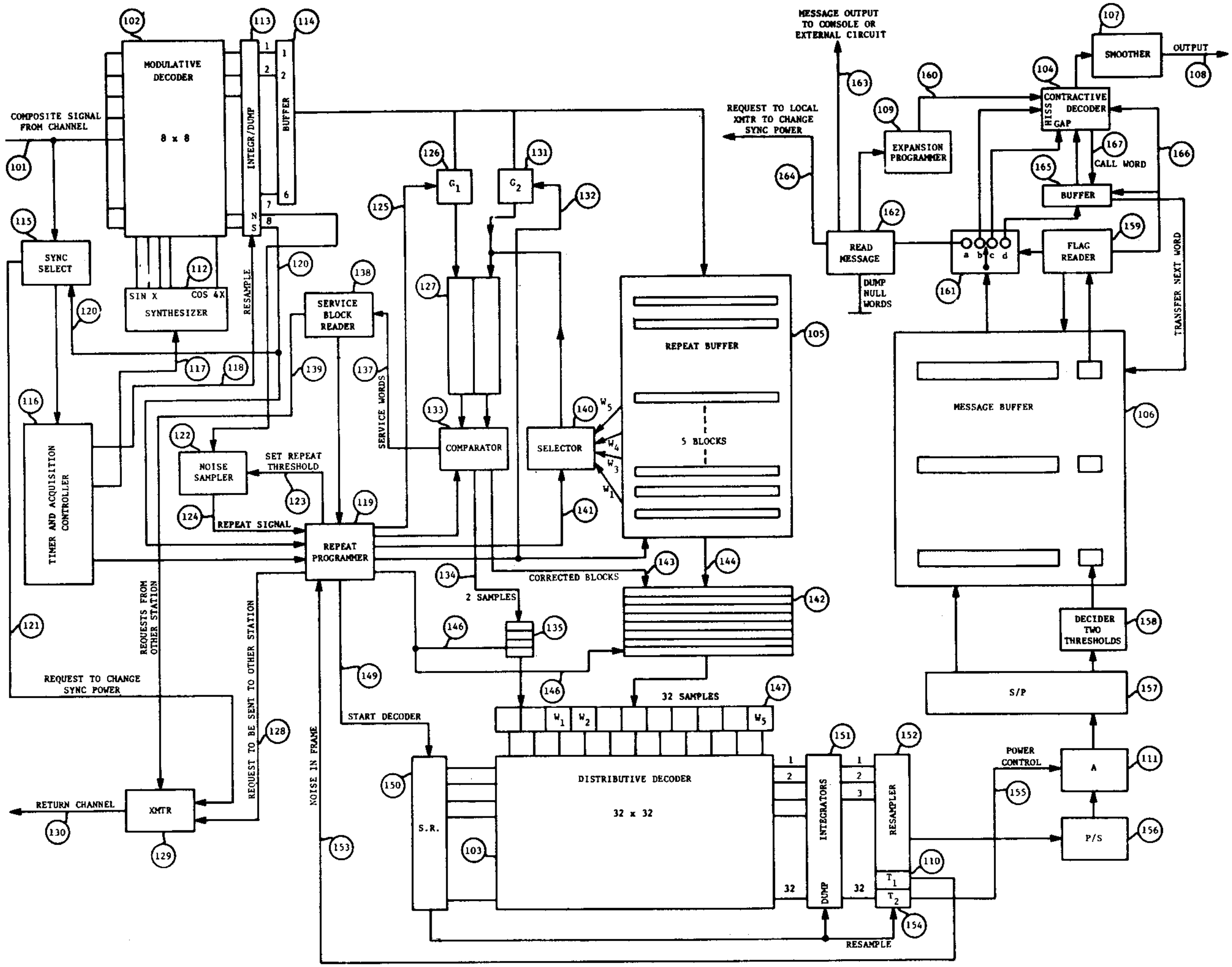



**Figure 7. Complete Block Diagram of a TESIT Receiver (Simplified Example)**

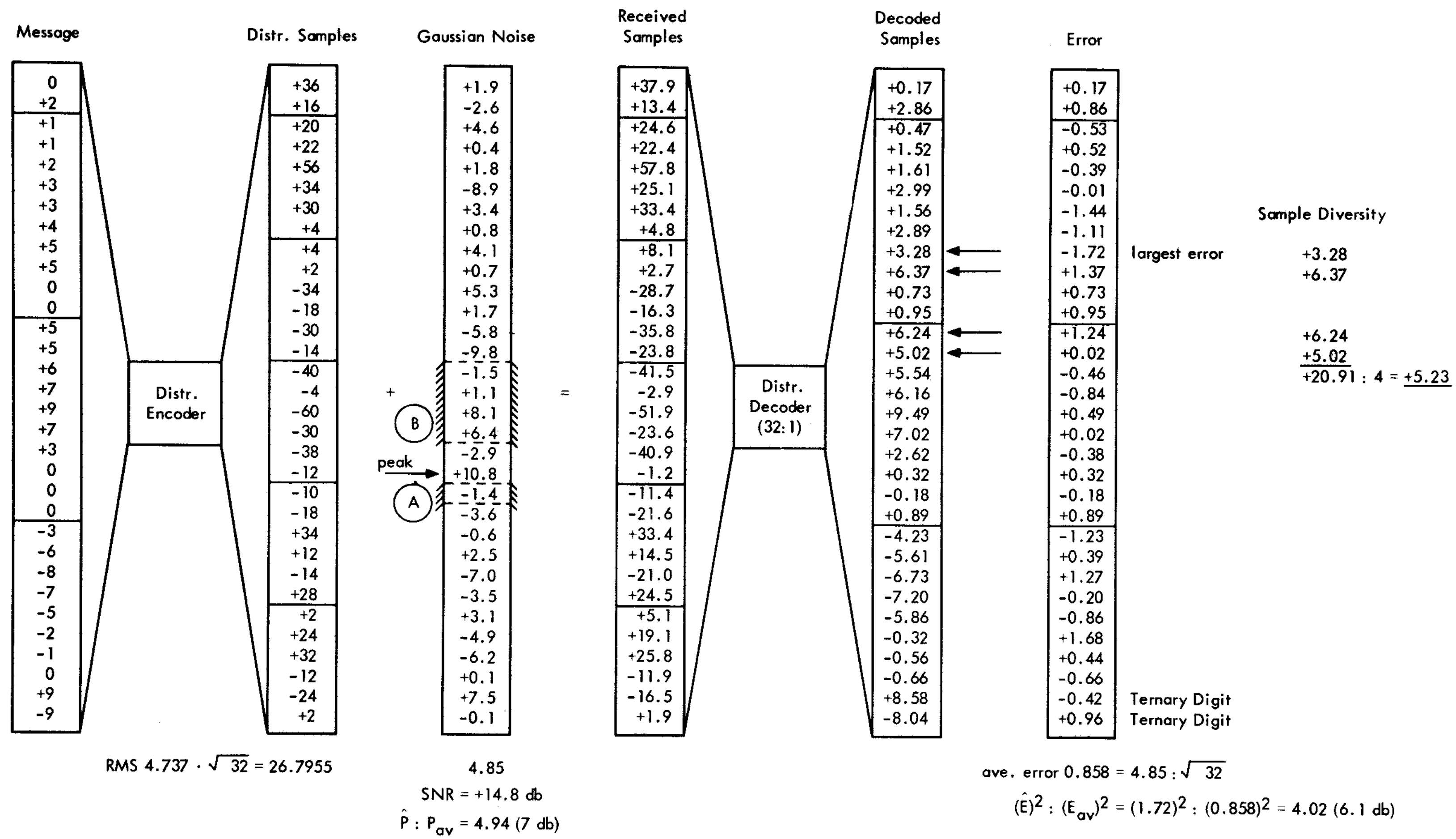



**Figure 8. Transmission over a Channel with Gaussian Noise**

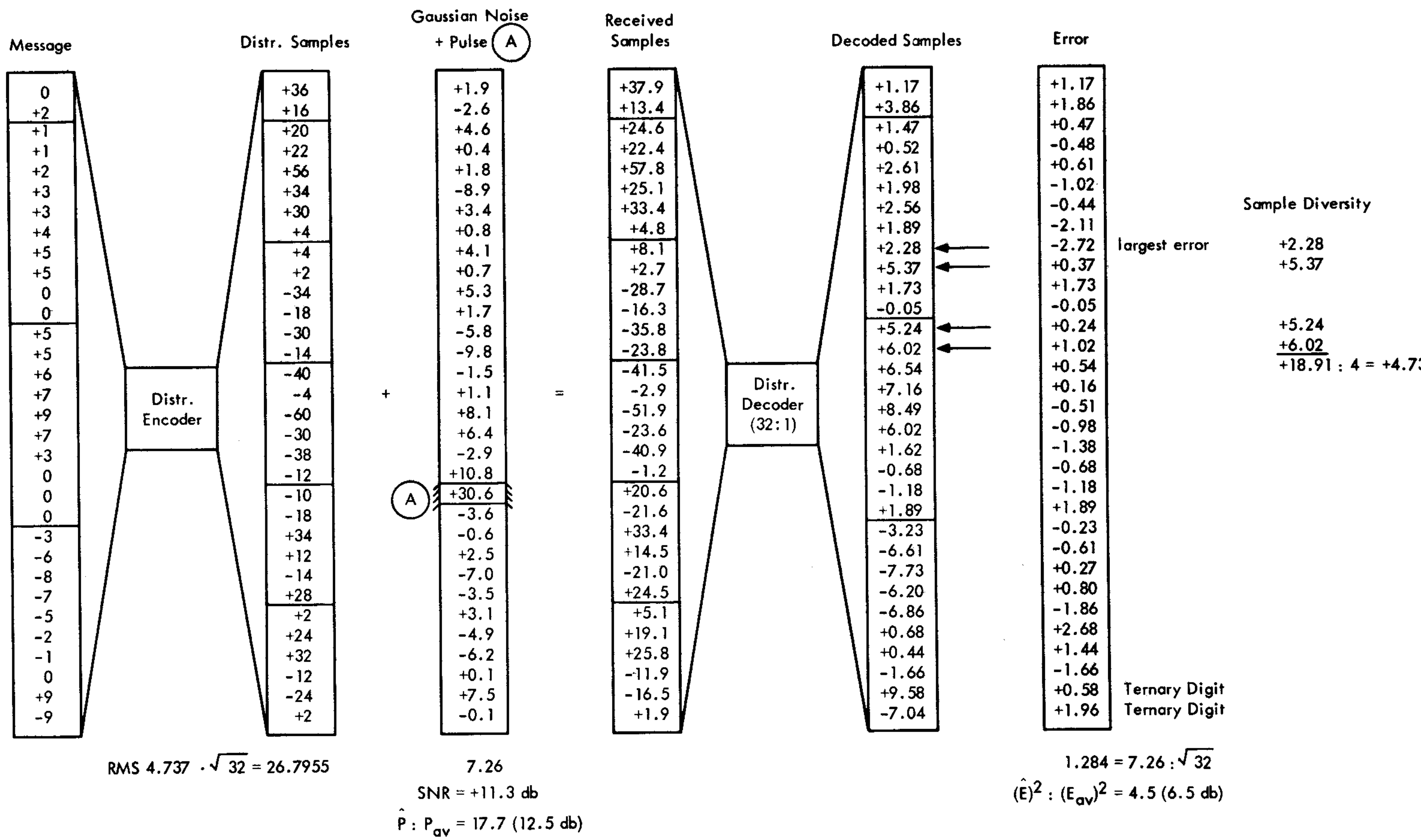



**Figure 9. Transmission over a Channel with Gaussian Noise Plus one Single Pulse A**

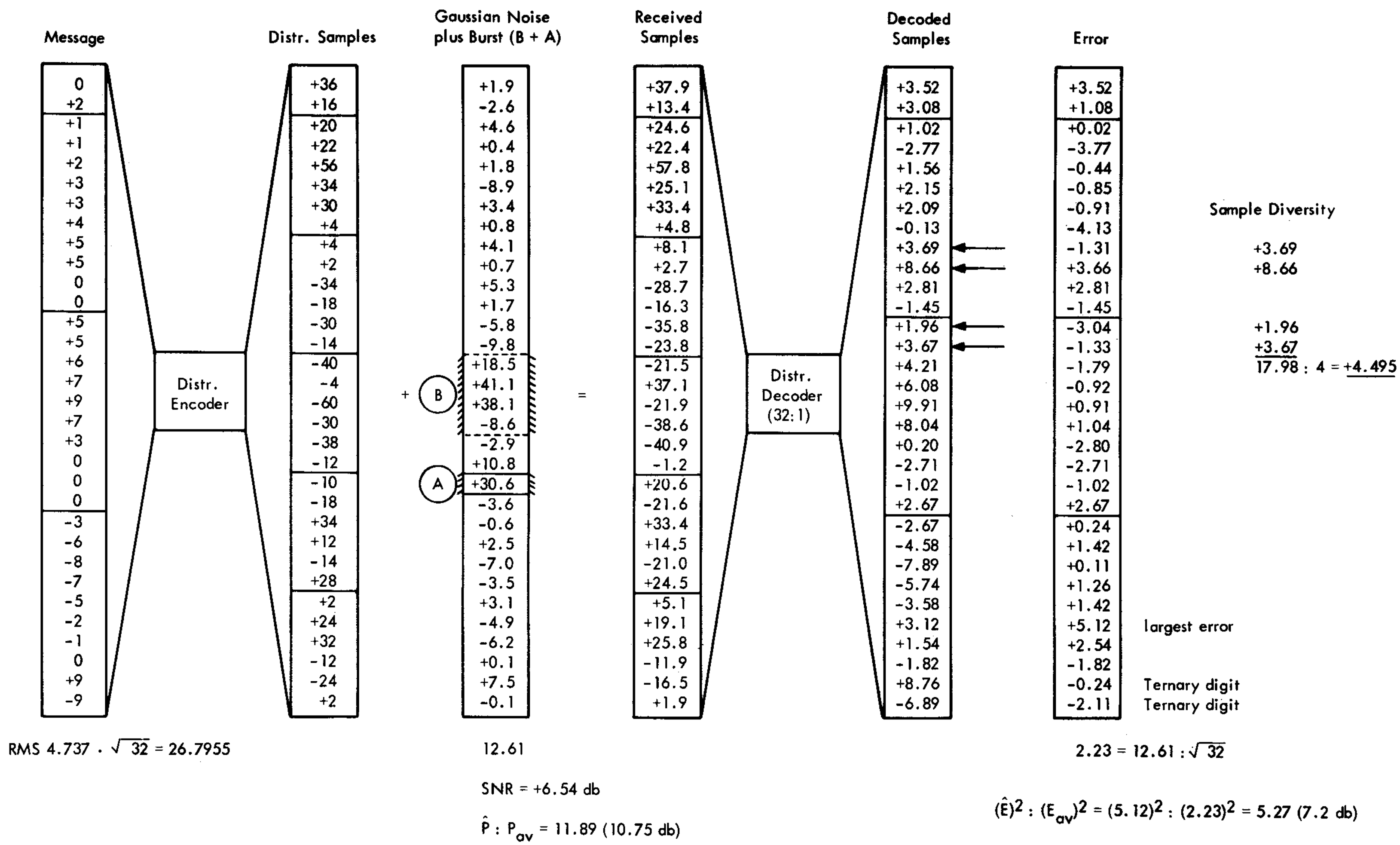



**Figure 10. Transmission over a Channel with Gaussian Noise Plus a Noise Burst A + B**