The *Picturephone*® System:

Digital Encoding of the Video Signal

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We describe in this paper the theory and implementation of a differential pulse code modulation system that is employed in the Picturephone® *network. Established and original coding techniques have been combined to allow transmission of a coded one-megahertz analog signal over a 6.312-megabit-per-second digital channel.*

I. INTRODUCTION

Pulse code modulation (PCM) is used for processing of analog signals for digital transmission. Quantizing noise and increased bandwidth are the penalties for the noise immunity inherent in digital transmission systems.

Differential systems[1] such as delta modulation[2–4] and the more general differential pulse code modulation (DPCM) have been investigated because of the ability to shape the spectrum of the noise and to take advantage of the sample-to-sample correlation in the signal. In a series of papers in 1952, B. M. Oliver,[5] E. R. Kretzmer[6] and C. W. Harrison[7] considered the use of linear prediction for "decorrelation" of video signals. In 1956 W. F. Schreiber[8] performed entropy measurements on television signals to determine the predominant redundancies present. In 1958 R. E. Graham[9] demonstrated by computer simulation the feasibility of using 3-bit DPCM for the television transmission of still black and white pictures. More recently J. B. O'Neal, Jr., has analyzed delta modulation[10] and DPCM[11] for the transmission of video signals. The noise structure of DPCM systems in slope overload has been investigated by E. N. Protonotarios.[12] Subjective evaluations have been recorded by R. C. Brainard.[13] These studies have led the way for the application of DPCM to *Picturephone* transmission.

II. DIGITAL PROCESSING OF THE VIDEO SIGNAL

2.1 *Pulse Code Modulation*

The conventional method of digital processing of analog signals is PCM. The signal, $s(t)$, is sampled, $s(nT)$, quantized to discrete levels,

$s_q(nT)$, and encoded, $Z(nT/N)$, in a binary form for transmission at the digital line rate (see Fig. 1).

The main source of impairment in the decoded and filtered signal is noise introduced at the quantizer.

$$n_q(nT) = s(nT) - s_q(nT). \tag{1}$$

This quantizing noise takes the form of background noise of a generally random nature. In addition, as the number of bits per sample is reduced, the background noise level rises, and contouring patterns appear in the video picture.

2.2 *Predictive Feedback Systems*

Independent processing of each sample is assumed for PCM systems. For signal sources having statistical properties like those of white noise, zero sample-to-sample correlation, this is appropriate. However, most video signals are highly correlated not only from sample-to-sample but from line-to-line and from frame-to-frame as well.

In predictive quantizing systems the value of the signal is predicted (predictor P) at each sampling instant from its recorded past history (Store) and only the difference, $e(nT)$, between the signal sample and its predicted value, $s_p(nT)$, is quantized, $e_q(nT)$, and transmitted (see Fig. 2).

The *Picturephone* signal has a 1-MHz bandwidth with an 8-kHz line rate. There are 271 lines per field with a 2:1 interlace. Table I shows the minimum amount of storage required (transmitter and receiver) for various feedback systems assuming that information for use in the predictor, P, is stored with an accuracy of eight bits per sample.

The costs and complexities of systems containing large stores weighed heavily against the use of frame-to-frame correlation[14] at the time the design of a codec for the *Picturephone* network was begun. The work of O'Neal on television signals, whose line-to-line statistics are much the same as those of *Picturephone* signals, demon-

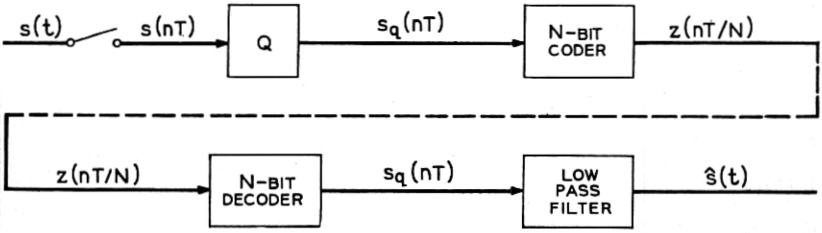

Fig. 1—Pulse code modulation (PCM) system.

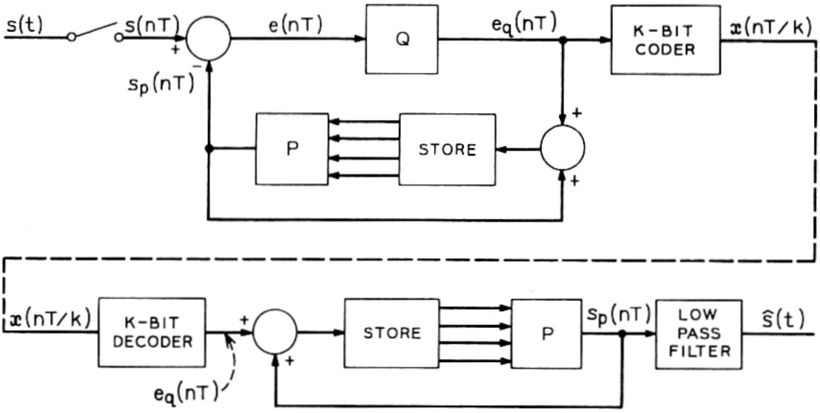

Fig. 2—Predictive quantizing system.

strated that only about a 2-dB gain in signal-to-noise ratio (S/N) could be achieved using linear prediction if horizontal and vertical correlations were approximately equal. Somewhat less could be achieved with a 2:1 interlace format. The design effort was therefore focused on sample-to-sample decorrelation procedures.

For sample-to-sample predictive systems, the store takes the form of a weighted tapped delay line (T is the sampling interval) with the weights, a_i, to be determined by the statistics of the signal (see Fig. 3).*

$$s_p(nT) = \sum_{i=1}^{\ell} a_i \{e_q[(n - i)T] + s_p[(n - i)T]\} \tag{2}$$

where the a_i ; $i = 1, 2, \cdots, \ell$ satisfy the equations

$$R_{0i} = a_1 R_{1i} + a_2 R_{2i} + \cdots + a_\ell R_{\ell i} \tag{3}$$

and R_{ij} is the correlation function,

$$R_{ij} = E\{s[(n - i)T]s[(n - j)T]\}. \tag{4}$$

Measurements[9] have shown that, for samples taken on the same line, television signals have an autocorrelation function which is very nearly Laplacean.†

* The assumption here is that the quantizer makes very little difference in the statistical properties of the error signal, an assumption which becomes more nearly correct as the number of quantizing levels is increased. Note that an exact analysis is impossible using linear prediction techniques with the non-linear quantizer in the network.

† As with all statistical properties, care should be exercised in interpreting this result. Although a typical subject on the average will exhibit this type of correlation, there are likely to be regions of the picture where much less sample-to-sample correlation is present.

TABLE I—FRAME STORAGE REQUIRED FOR VARIOUS PREDICTIVE SYSTEMS

Frame Correlation Predictor	1,084,000 bits
Line Correlation Predictor	4,000 bits
Sample Correlation Predictor	16 bits

$$R_s(\tau) \approx \exp\left(-\alpha \mid \tau \mid\right). \tag{5}$$

For this type of source, the optimum solution to minimize the mean square error between input and output (predicted signal) is one where all the $a_i = 0$ except a_1.[11] This means that only the previous sample is effective in decorrelating the signal. The predictive feedback coder then takes the form shown in Fig. 4. A measure of the validity of the assumptions and the effectiveness of the decorrelation process is the amount of correlation removed from the video signal and the amount remaining in the error signal. The adjacent sample correlation coefficient

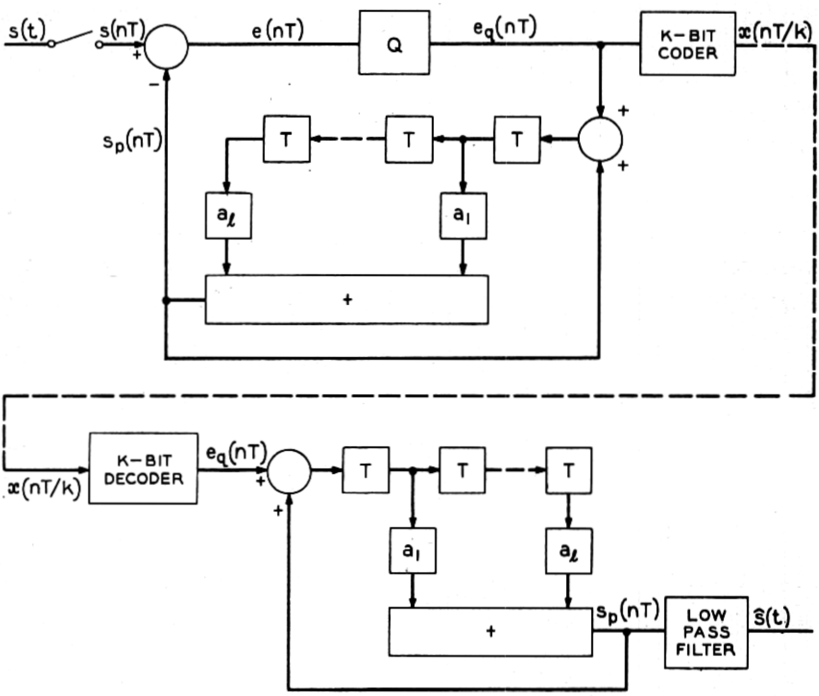

Fig. 3—Sample-to-sample predictive quantizing system using a tapped delay line.

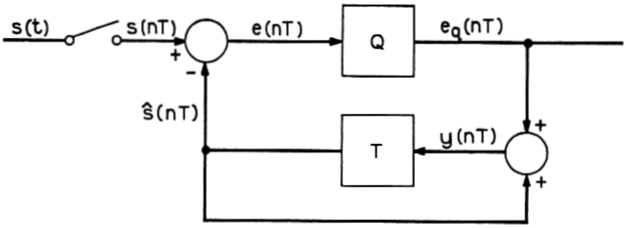

Fig. 4—One-tap predictive quantizing coder.

$$\rho_s = \frac{E\{s(t)s(t + T)\}}{E\{s^2(t)\}} \tag{6}$$

was calculated from sample statistics and found to be 0.93 for the still picture "Rosalie" as referred to in Section 2.6. The adjacent sample correlation coefficient

$$\rho_e = \frac{E\{e_q[nT]e_q[(n - 1)T]\}}{E\{e_q^2(nT)\}} \tag{7}$$

for the same picture is 0.4407 as calculated from laboratory data for the three-bit DPCM coder using the quantizer of Fig. 5. This shows that considerable redundancy has been removed from the signal but that some sample-to-sample correlation remains.

2.3 *Quantizing Noise*

To demonstrate that the noise introduced in predictive feedback systems can be determined by analyzing the error at the quantizer itself, consider the one-tap system of Fig. 4. The quantizing error QE is given by

$$QE = e_q(nT) - e(nT) \tag{8}$$

where

$$e(nT) = s(nT) - \hat{s}(nT), \tag{9}$$

$$e_q(nT) = y(nT) - \hat{s}(nT), \tag{10}$$

$$y(nT) = \hat{s}[(n + 1)T]. \tag{11}$$

Substituting equations (9), (10) and (11) into (8) gives

$$QE = s(nT) - \hat{s}[(n + 1)T] \tag{12}$$

which is the system error in transmitting the signal in the absence of any channel errors. The system noise can therefore be determined directly from the characteristic of the error signal and a knowledge of the quantizer. The manner in which the quantizing noise manifests itself as picture impairments will be discussed in later sections.

2.4 *Differential Pulse Code Modulation*

Consider the single tap predictive feedback system shown in Fig. 4. The feedback signal, $\hat{s}(nT)$, is given by

$$\hat{s}(nT) = \sum_{i=1}^{\infty} e_q[(n - i)T]. \tag{13}$$

If a loss factor $\beta < 1$ is introduced into the delay section then equation (13) is modified to give

$$\hat{s}(nT) = \sum_{i=1}^{\infty} \beta^i e_q[(n - i)T]. \tag{14}$$

Past history is weighted giving greater importance to the more recent samples. In practice, the feedback network in the *Picturephone* codec is replaced by an analog network which consists of an integrator, F, with leak (see Fig. 6);

$$F(s) = \frac{K}{s + \alpha}, \tag{15}$$

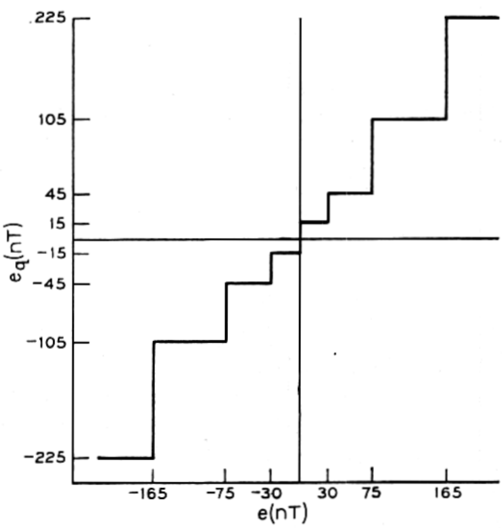

Fig. 5—Three-bit DPCM quantizer characteristic.

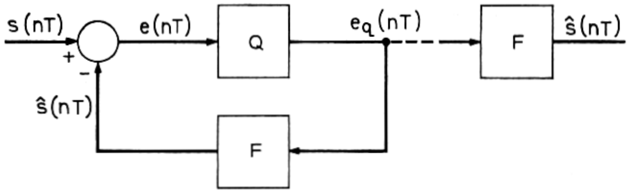

Fig. 6—DPCM system with feedback integrator.

where the β [equation (14)] and α [equation (15)] are related in the following manner:

$$\beta \approx \exp(-\alpha T). \tag{16}$$

A system organized in this fashion is a DPCM system. A differential system very nearly transmits the differences between successive samples rather than the samples themselves. The receiver accumulates the received difference and reconstructs an approximation to the original waveform. The presence of a quantizer provides the potential for an accumulation of systematic quantizing error. The feedback loop around the quantizer prevents this accumulation.

The feedback network used in the *Picturephone* coder has a 27μs time constant. For subjective effect the time constant was made as large as possible without unduly magnifying the effects of transmission errors on the viewed picture. This is discussed further in Section 2.8.

2.5 *Subjective Considerations and Noise Shaping*

In DPCM, the subjective effect of a video observer can be taken into account and the spectrum of the noise shaped accordingly. As Graham points out,[9] the "concept of predictive quantizing as a truly perceptual coding technique" is one "in which the primary object is not necessarily accurate prediction all or almost all of the time, which frequently is impossible, but only accurate prediction in those signal regions where the observer perception is high." It is known[9] that the eye is more tolerant of noise located at black-white interfaces than in flat regions of the picture. Interfaces are characterized by large values of the difference signal and flat regions by small values. It is therefore possible to use companding on the difference signal to locate the noise in region of the picture where that noise is least objectionable.

Companding is accomplished through the use of a nonlinear quantizing characteristic providing a fine structure and a high S/N for low level difference signals (flat regions) and a coarse structure with its lower S/N for large difference signals (interfaces). The nonlinear characteristic used in the *Picturephone* codec is shown in Fig. 5. The choice of this particular set of step sizes will be discussed in a later section.

Because of the subjective aspects of rating picture quality, S/N is not a sufficient measure of system performance. However, it is useful to compare the theoretical S/N performances of PCM and DPCM. Based on some simplifying assumptions* O'Neal[11] has shown that for the optimum one tap predictive filter the S/N improvement for DPCM over PCM is related to the sample-to-sample correlation of the signal, $R_s(T)$, in the following manner.

$$\text{S/N IMPROVEMENT} = 10 \log \left[\frac{1}{1 - R_s^2(T)} \right]. \tag{17}$$

For video signals with only a moderate amount of detail this results in an improvement of approximately 12 dB which is equivalent to two bits of quantizing information per sample.

2.6 *Signal Impairments*

DPCM systems are subject to a unique set of subjective impairments. Among these are overload noise, granular noise, contouring and edge busyness, all resulting from the quantizing process (see Section 2.3) but different in appearance to the observer.

Unlike PCM systems which are amplitude limited, DPCM systems overload on slope. If the sample-to-sample difference is greater than the largest quantizer step, then the system is said to be in slope overload. A system exhibiting slope overload effectively reduces the horizontal resolution of the scene being processed.

In low detail regions of the picture, the noise power is determined by the rather fine quantizing structure designed for small differences. This is termed the granular noise region. Figure 7 shows a typical video waveform having an overload region and a granular noise region.

In the *Picturephone* codec the small step is 15 mV and the large step is 225 mV on a full scale video signal of ±750 mV (cf. Fig. 5). The resolution error for any brightness change is plus or minus one-half of a small step. The slope handling capability of the codec is 450 mV per μs.

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* The assumptions on which the theory is based are that (*i*) the distribution of the error signal can be approximated by the Laplace density function:

$$P(e) = \frac{1}{\sqrt{2}\sigma_e} \exp\left(-\frac{\sqrt{2}}{\sigma_e} \mid e \mid\right),$$

(*ii*) the quantizer is optimized to the statistics of the error signal and (*iii*) the probability of overload is small.

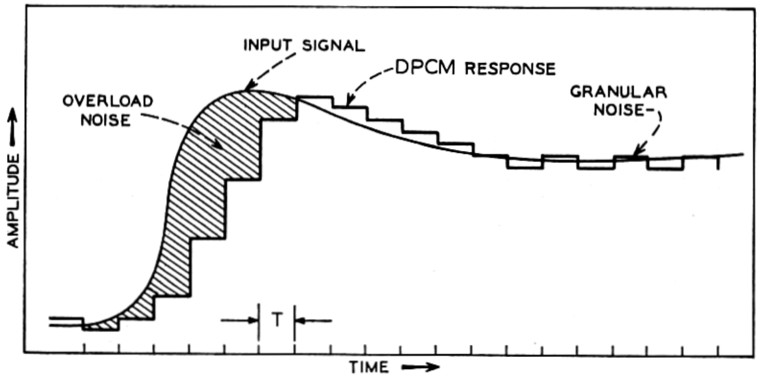

Fig. 7—Illustrated video waveform and DPCM response showing an overload region and a granular noise region.

Figure 8 shows a typical picture ("Rosalie") uncoded, (see Fig. 8a), and exhibiting slope overload after three bit encoding (see Fig. 8b). This is most noticeable as smearing on the ear ring (white-to-black interface).

If the small steps are too large, the strong correlation between the quantizing noise and the signal can lead to visible patterns that are subjectively objectionable. In flat regions of the picture (zero sample-to-sample difference), the DPCM feedback loop oscillates using alternately positive and negative small steps. The integrator leak breaks up this pattern and if the leak time constant is too small, leak contouring can occur. This is illustrated for two different values of video level in Fig. 9. The dashed component of the signal is the leak contour.

Even in the absence of leak, contouring patterns can occur in sloping regions of the video signal. A slope which slightly exceeds (falls short of) the ability of a particular step size to follow it will require periodic correction, through the use of the next larger (smaller) step. If this waveform is repeated through several lines (strong vertical correlation in the picture) and if the differences between the adjacent steps in question are too large, a visible contour pattern will result. Figure 10 demonstrates this effect.

Figure 11 is a photograph of an oscilloscope trace showing a leak contouring pattern. A well-designed three bit codec does not produce visible contours. This picture resulted from changing the leak time constant from 27 μs to 5 μs.

The requirements for minimizing the preceding impairments are in conflict, given a fixed number of steps in the quantizer. Granular noise requirements specify a fine quantizing structure. Slope overload and

Fig. 8—(a) Analog *Picturephone* signal "Rosalie." (b) Photograph of 3-bit DPCM encoded *Picturephone* signal "Rosalie." Exhibit of slope overload.

hence horizontal resolution requirements specify a large outer step. Contouring and, as will be shown below, edge busyness, can both be reduced through the use of a uniform spacing of the steps. Quantizer design is therefore a matter of finding the proper compromise among the subjective effects of the various impairments.

2.7 *Synchronization and Edge Busyness*

Since the system was designed to concentrate the noise in the high frequencies corresponding to the interfaces or edges in the picture, the resulting degradation becomes a limiting factor. Quantizing noise at slightly nonvertical edges takes the form of "stairsteps." This is due to the discrete nature of the sampling process. If in going from frame-to-frame in a still picture there are changes in the response of the

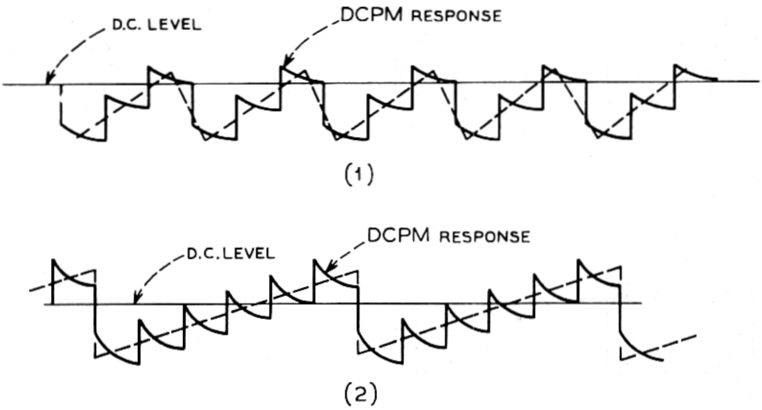

Fig. 9—Illustration of leak contouring patterns.

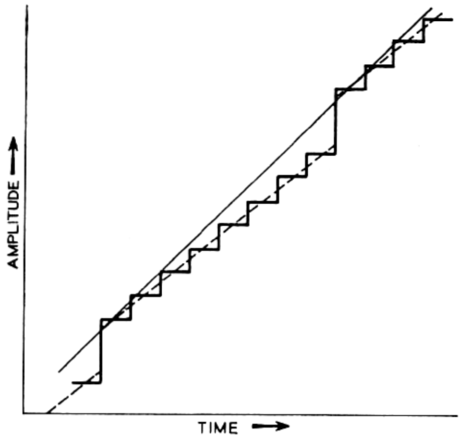

Fig. 10—Illustration of contouring pattern.

DPCM loop, it follows that the edge noise will shift. This gives rise to the degradation known as edge busyness.

Considerable improvement is realized by phase locking the sampling clock in the coder to the line rate of the video source. This ensures that each sample is taken in the same horizontal position from line-to-line and from frame-to-frame. Edge busyness is also a function of noise, quantizer characteristic, threshold tolerances and past history due to the memory in the feedback loop. It follows that synchronization will substantially improve picture quality but will not eliminate edge busyness completely. Figure 12 illustrates a busy edge pattern

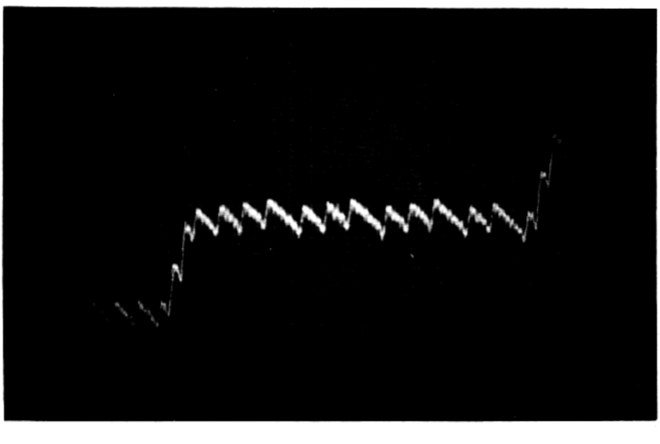

Fig. 11—Photograph of oscilloscope trace of leak contouring in the 3-bit DPCM encoded *Picturephone* signal "Gray Scale" ($5\mu s$ leak time constant).

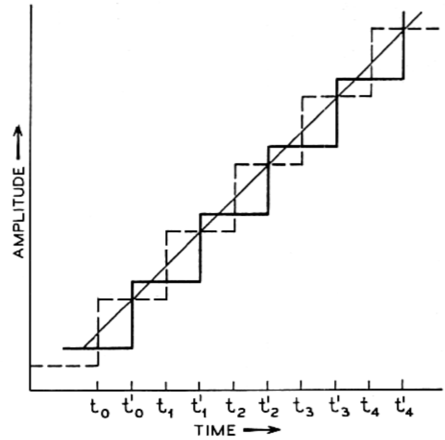

Fig. 12—Illustration of edge busyness.

caused by differences in past history and in clock phase. Figure 13 is a photograph of an oscilloscope trace showing a "busy edge." The different paths taken in making the black-to-white transition are clearly visible in this multiple trace picture.

Of the above factors, the quantizing structure and the leak time constant are under the control of the designer. Since edge busyness results from the variability in the selection of the set of steps to approximate the input ramp voltage it follows that minimizing the horizontal difference between the two responses will minimize the edge busyness in a mean square sense. The horizontal differences are minimized by making the differences between adjacent steps as small as possible. This would be accomplished through the use of a uniform quantizer. However, the existence of the other impairments with their conflicting requirements makes the use of a uniform quantizer impractical as stated above.

2.8 *Effects of Channel Noise*

Since differential encoding removes redundancy in the signal source it provides for more efficient transmission of information. However, the removal of redundancy from a signal will in general, make the signal more susceptible to channel noise.*[15,16]

For DPCM systems the degradation due to digit errors takes the form of horizontal streaks in the picture. Each streak originates at a decoded sample difference which is incorrect due to a transmission

* Since the transmission is in digital form, the susceptibility is to bit errors in the channel.

error. Decoder memory causes the effect of a channel error to propagate through successive reconstructed samples for a period of time determined by the leak time constant. The visible effect is therefore lessened by using an integrator with a small leak time constant. The creation of leak contour patterns is a lower bound for reduction in the size of the leak time constant.

III. CIRCUIT DESIGN

Figure 14 is a block diagram of the overall circuit. The transmitting and receiving circuits are each made up of three major parts, the analog interface, the digital interface and the DPCM coder or decoder circuits.

The input signal may be a pre-emphasized 1-MHz video signal such as that sent from one *Picturephone* station set to another or it may be a 460.8 kb/s binary data signal sharing the same facilities. Level variations as large as ±6 dB about the nominal level are acceptable and will be corrected to within ±0.1 dB before coding.

The digital signals are B6ZS signals of the proper format for transmission on a T2 facility.[17] The circuits are matched to a 100-ohm balanced line and sufficient equalization is provided in the decoder to compensate for a 22-gauge cable pair of up to 2000 feet.

3.1 *The Transmitting Portion Analog Interface*

All input signals, video or data, pass through common input circuits (Fig. 15), which provide 40 dB of longitudinal suppression and which

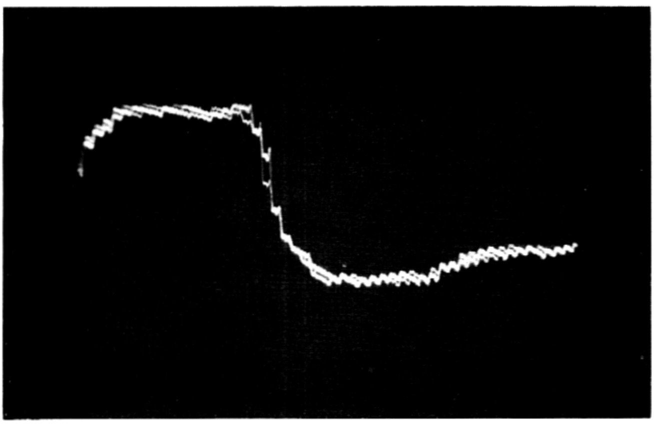

Fig. 13—Picture of oscilloscope trace of edge busyness in 3-bit DPCM encoded *Picturephone* signal "Rosalie."

drive a phase-linear 6-pole low-pass filter with a gaussian roll-off 6 dB down at frequency $f = 1$ MHz,

$$\text{Filter loss} = 20 \log_{10} \exp (0.69f) \text{ dB}. \tag{18}$$

After filtering, the signal is processed in one of three ways. Picture signals, which are recognized by the 8-kHz sync format, must be de-emphasized[18] in order to obtain a normal video waveform, similar to that at the output of a *Picturephone* camera tube and suitable for coding with a DPCM coder; the de-emphasis circuits have the inverse characteristics of the pre-emphasis circuits used in the station set.

$$\text{De-emphasis loss} = 20 \log_{10} \left| \frac{10(jf + 0.1)}{jf + 1} \right| \text{dB}, \tag{19}$$

where f is the frequency in MHz.

The signal is then clamped and passed through automatic gain control (AGC) circuits which maintain a constant 0.375-volt sync-tip to back-porch voltage; thus at the input to the DPCM coder the video signal has a nominal level of 1.5 volts peak-to-peak. Binary data are controlled by a separate suitable AGC circuit with 7.2 dB less gain which adjusts the nominal level to the coder to be 0.9 volts peak-to-peak. Low-level signals, specifically the 12-kHz tone used for testing the channel continuity, pass through fixed gain circuitry. Information as to the presence and nature of the received signal is multiplexed onto the output bit stream to control functions at the decoder that are complementary to those in the coder.

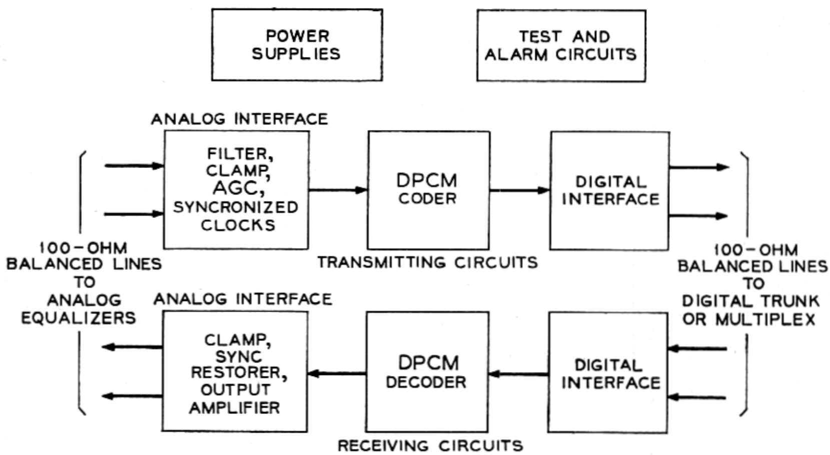

Fig. 14—Codec terminals.

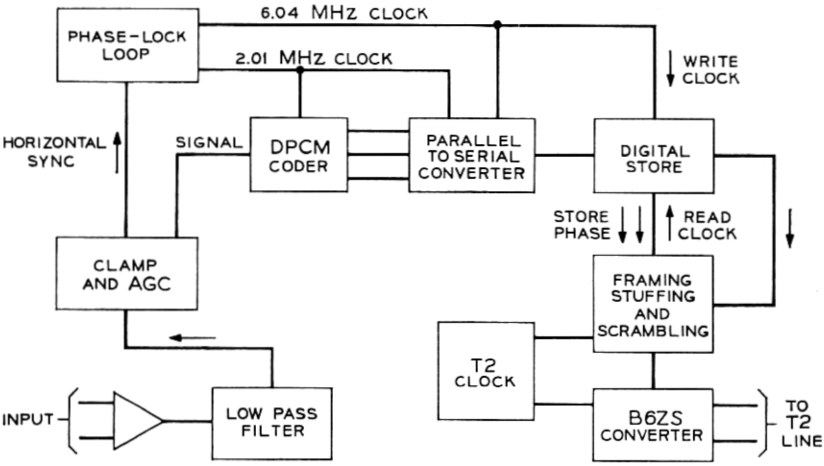

Fig. 15—Block diagram of the transmitting circuit.

The basic clock frequency for the coder circuits, 6.048 MHz, is an exact multiple of three times the *Picturephone* line frequency, 8 kHz, (it is the 252nd harmonic). This allows the sampling clock to be synchronized to the *Picturephone* line rate so that the rectangular video picture is sampled with a uniform grid. This 6.048-MHz frequency is generated by the voltage controlled oscillator (VCO) in a phase-locked loop and is divided by three to obtain the 3-phase 2.016-MHz sampling clock that drives the actual coder. The 6.048-MHz frequency is counted down by a factor of 12 in the digital store addressing circuits and the resulting frequency is counted down by an additional factor of 63 to obtain an 8-kHz square wave. This signal is one input to a phase comparator whose other input comes from the sync separator within the video clamp circuits; the output of the phase comparator is used to control the frequency of the VCO. In the absence of external noise, any individual sample of the picture is made at the same temporal point along the same line with an accuracy of less than ten nanoseconds, a figure that corresponds to less than 0.001 inches displacement across a *Picturephone* display tube. This synchronization process minimizes the busy edge effects. The clock is not synchronized to the data signals.

3.2 *The DPCM Coder*

The differential pulse code modulator circuits that are used in the codec are shown in Fig. 16. At each sampling instant of time the video input is compared directly against the decoded video output signal

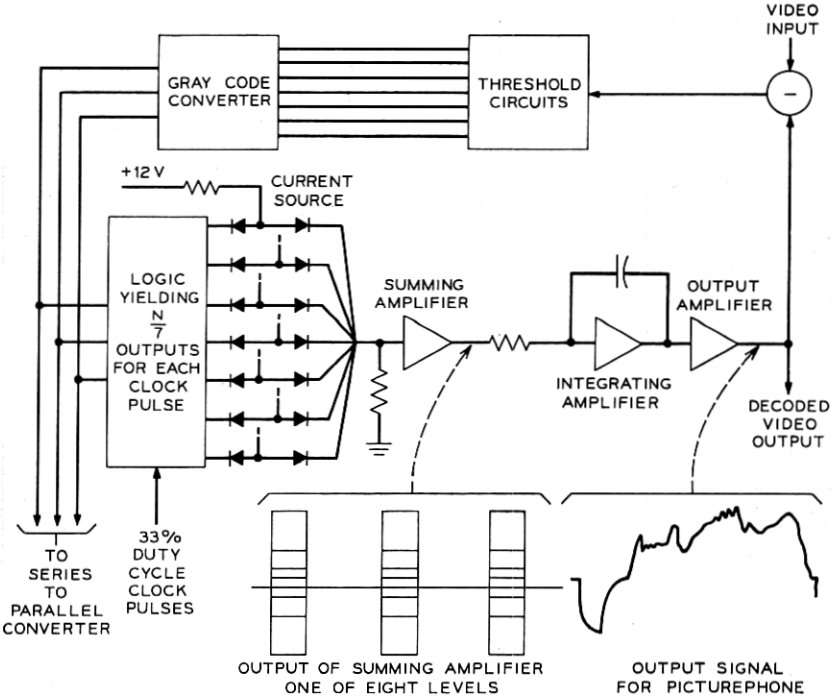

Fig. 16—DPCM coder.

last transmitted. The output of the difference amplifier is connected in parallel to the inputs of seven threshold detectors set to 0, ±2 percent, ±5 percent and ±11 percent of the nominal 1.5 volt peak-to-peak video input signal (cf. Fig. 5). The cyclic coded outputs of these detectors are the output of the coder and are taken to the digital-to-analog circuit shown in the lower half of the circuit of Fig. 16 whose logic, gating, summation, integration and amplification circuits produce a decoded video signal that is a close replica of the original input signal. The various circuit functions of the digital-to-analog circuitry are kept separate to avoid undesirable voltage interactions and the duty cycle is set at 33 percent to avoid similar trouble in the time domain. The required (nonlinear) companding characteristic is achieved by making the current sources appropriate for output voltage changes, $e_q(nT)$ of magnitude ±1 percent, ±3 percent, ±7 percent, or ±15 percent of the peak-to-peak quantizer input video signal (cf. Fig. 5).

Low-speed data signals[19] are coded in exactly the same manner as video signals. However the magnitude of the data signals is such that the largest step size is exactly 25 percent of the total peak-to-peak excursions. This ensures that each binary transition can be coded within four sampling intervals, a time interval that is slightly less than the period of 460.8 kb/s data. Between binary transitions extensive use is made of the smallest steps whose magnitude is only ± 1.7 percent of the binary peak-to-peak signal.

3.3 *The Transmitting Portion Digital Interface*

After conversion to a serial bit stream, the coder output bit stream passes through a digital store (see Fig. 15), with a capacity of four 3-bit words each of which occupies a fixed position within the memory. The information is not taken out of the store at the steady 6.048 MHz rate but at a 6.312 MHz rate that is sometimes interrupted so that the bit stream can be framed. The framing process must allow the inclusion of three signaling elements and must make adjustments for small variations in the frequency of the sampling clock which is synchronized to the video horizontal scanning frequency. The three signaling elements are transmitted as zeros except when

$BZ = 1$, if a signal is detected at the coder input, or
$PD = 1$, if a video signal is being coded, or
$RF = 1$, where RF is a store framing signal.

There are two types of 108-bit words R and S where

$$\{(\text{Bit } 1 = 1), (\text{Bit } 33 = 1), (\text{Bit } 65 = 1), (\text{Bit } 92 = BZ)\} \in R,$$

and

$$\{(\text{Bit } 1 = 0), (\text{Bit } 33 = 0), (\text{Bit } 65 = 0), (\text{Bit } 76 = PD), (\text{Bit } 92 = RF)\} \in S,$$

and in each case all other bits or elements of the word carry DPCM information. The number of information bits is different for the two words and which word is sent is determined by monitoring the relative phases of the elastic store write and read addressing.

The output bit stream passes through a one-cell scrambler to remove any preponderance of 1's or 0's and is converted to B6ZS format for transmission on the T2 line. The scrambling is performed by delaying the output through a one-cell shift register, the input of which is the modulo-2 sum (exclusive-OR function) of the input and the output of the shift register.

3.4 *The Receiving Portion Digital Interface*

After clock recovery in a T2 receiver the unipolar signal is recovered from the B6ZS input signal and is descrambled to yield the

108-bit words clocked at a 6.312-MHz rate (see Fig. 17). The descrambler contains a one-cell shift register the input and output of which are applied as inputs to an exclusive-OR logic circuit to derive the output data stream. The bit stream is passed through a 12-cell store using a write clock that is obtained by analysis of the 108-bit words and which clock only DPCM information bits into the store. The signaling bits are recovered and the digital store framing information is used to ensure that the four 3-bit words are placed in the decoder digital store in the same position that they were taken out of the coder digital store.

A jitter-free regenerated clock is needed for reading data from the store and for use in the series-to-parallel converter and the decoder; the 6.048-MHz VCO clock is locked to the incoming signal by means of a phase-lock loop which includes the divide-by-12 circuits used for addressing the digital store. After passing through the digital store, the bit stream is processed in a serial-to-parallel converter which provides the required inputs for the DPCM decoder.

3.5 *The DPCM Decoder and Output Interface*

The decoder consists of a digital-to-analog circuit exactly the same as that used in the coder and shown in the lower portion of the circuit in Fig. 16. Since the circuits at the two ends of the digital transmission system are of identical design, an accurate representation of the input signal is assured. Although not shown in the figure, a resistive leak is provided across the integrator capacitor in each circuit; the leak is necessary to ensure decay of the effects of digital errors and mistracking and a time constant value of 27 microseconds is used.

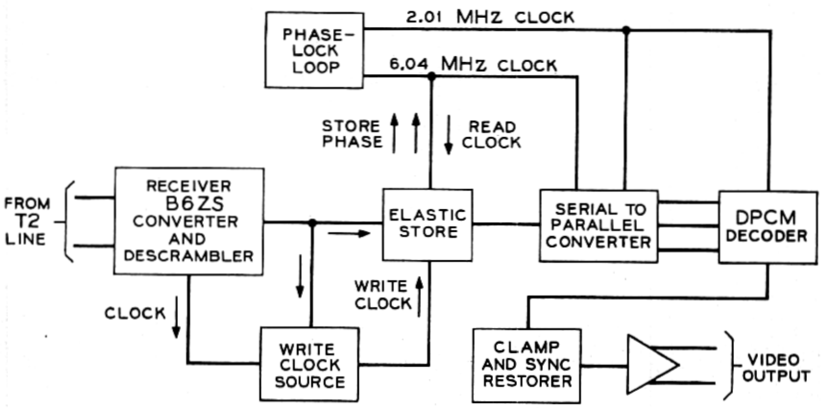

Fig. 17—Block diagram of the receiving circuit.

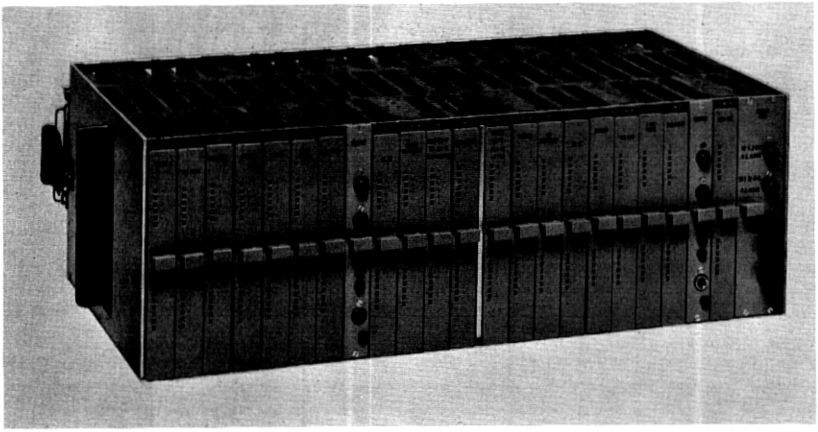

Fig. 18—The codec shelf.

At the output of the decoder, data mode signals are adjusted to the correct level and then passed to the balanced amplifier that drives the 100-ohm line. For video signals some additional processing of the video signal is necessary after pre-emphasis because the station set dc restorer would be adversely affected by the quantizing noise at the tips of the sync pulses which might cause low-frequency beats in the *Picturephone* clamp circuits.[20] Accordingly a circuit is included that attenuates the high frequency quantizing noise during the occurrence of sync pulses.

3.6 *Equipment Design*

The circuits for the two-way codec terminal are placed on 21 printed circuit packs located in a 7-inch-high shelf as shown in Fig. 18. Four of the cards carry the circuits that do the actual coding and decoding of the video signal, two of eight test cards provide power and alarm circuitry and the remaining cards provide the interface and synchronization circuits.

IV. EVALUATION

Many codecs have been built and tested; six were in service on the TD-2 radio relay route between New York and Pittsburgh for a 6-month period in 1969. Even with the additional noise introduced by the route, a subjective evaluation of the quantizing noise showed less than 1 dB of degradation in the field.

After the trial some changes were made in the design used in the first service model codecs to provide more flexible usage and to conform to new *Picturephone* standards. Circuits were added to the input and output interfaces to allow 460.8 kb/s data to be processed, to activate the appropriate signal paths and to distinguish between video and data signals. The new *Picturephone* set employs crystal controlled frequency standards allowing the use of the small buffer stores and the simple phase-lock loops that have been described above. The sync format has also been changed. However, no fundamental change was made to the coding algorithm or to the digital interface. Work is continuing to seek improved coding techniques for use in the *Picturephone* network.

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