PULSE CODE MODULATION OF DIGITAL COMMUNICATION SYSTEM OVER TWO-WIRE LINES

by 500

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I. INTRODUCTION

One of the major problems of telephone transmission is the reduction of noise and distortion that is occurred along the path or introduced by associated equipment. The development of Pulse Code Modulation, commonly referred to as PCM, tends to eliminate the problems of noise and crosstalk.

The analog speech signal is sampled thousands of times per second. These samples are converted into digitally coded pulses. The pulses from many conversations (or channels) are then transmitted sequentially over the same cable pair. In this way, speech signals are stacked in time. At the receive end, the pulses are converted back into their original analog form. The PCM system is the newest cable carrier system. It requires its own special equipment and can play a distinctive role in the communications network.

In this paper, some basic concepts of pulse code modulation and an experiment in transmitting speech by PCM system will be discussed. Also, some basic ideas of TONE-DIAL telephones and its applications will be discussed later. With touch-tone calling (pushbutton signaling) using voice frequency signaling, a call can be placed in less than half the time it takes with a conventional dial.

II. PCM AND ITS FEATURES

2.1 Sampling[13]

The function of sampling is to replace a continuous bandlimited signal by a discrete sequence of its samples without the loss of any information. Such discrete information can be transmitted by a group of pulses whose amplitudes may be varied according to sample values.

If the signal is sampled instantaneously at regular intervals and at a rate slightly higher than twice the highest signal frequency, then the samples will contain all of the information of the original signal.[11] This is called Shannon's sampling theorem. In other words, a bandlimited signal which has no spectral components above a frequency f_m Hz is uniquely determined by its values at uniform intervals less than $1/2f_m$ seconds apart.[13]

Consider a bandlimited signal f(t) which has no spectral components above f_m. This means that F(w), the Fourier transform of f(t), is zero for $|w| > w_m$ ($w_m = 2\pi f_m$). Suppose that the signal function f(t) is multiplied by a periodic impulse function $\delta_T(t)$ with regular intervals of T seconds. Then the sampled function is $f_s(t)$.

$$f_s(t) = f(t)\delta_T(t) \tag{1}$$

And the Fourier transform of $\delta_T(t)$ and f(t) is (see Appendix)

$$\mathcal{F}[\delta_T(t)] = w_o \delta_{w_o}(w) \tag{2}$$

$$\mathcal{F}[f(t)] = F(w) \tag{3}$$

where $$\delta_{w_o}(w) = \sum_{n=-\infty}^{\infty} \delta(w - nw_o) \tag{4}$$

$$w_o = 2\pi/T \tag{5}$$

According to the frequency convolution theorem, the Fourier transform of sampled function will then be

$$F_s(w) = \mathcal{F}[f_s(t)] \tag{6}$$

$$= \mathcal{F}[f(t)\delta_T(t)] \tag{7}$$

$$= \frac{1}{T} \sum_{n=-\infty}^{\infty} F(w - nw_o) \tag{8}$$

Equation (8) represents F(w) repeating itself every w radian per second. Note that F(w) will repeat periodically without overlap as long as $w_o \geq 2w_m$, or

$$\frac{2\pi}{T} \geq 2(2\pi f_m)$$

That is

$$T \leq \frac{1}{2f_m} \tag{9}$$

2.2 Reconstruction [8, 13]

It is easy to recover F(w) from $F_s(w)$ by allowing the sampled signal to pass through a low-pass filter which will only allow frequency components below f_m and attenuate all the higher frequency components. The output of this filter will then be identical to the input signal.

If the sampling interval T becomes larger than $1/2f_m$, then there is an overlap between successive cycles, and F(w) can not be recovered from $F_s(w)$. Therefore, if the sampling interval T is made too large, the information is partly lost, and the signal f(t) can not be recovered from the sampled signal $f_s(t)$.

2.3 Quantization

The fundamental operation of pulse code modulation is the conversion of a signal sample into a code combination of on-off pulses (or binary digits). In any practical system, a continuous range of signal values can not be reproduced since only a finite number of combinations can be made available.[12] So that in PCM system, it is necessary to represent a continuous range of signal amplitudes by a finite number of discrete steps, and each sample value of the signal is represented by its nearest equivalent in a discrete set of amplitudes to permit digital encoding. This process is spoken of as quantization.

In Fig. 1(a), a straight line representing the relation between input and output samples in a linear continuous system is replaced by a flight of steps, and the height of the step is the quantum.

For n-digit codes, there are 2^n levels. The problem then is to determine the smallest number of steps which voice signals may be quantized without serious distortion. Presently the 7-digit code is used in PCM system. The distortion from this process is referred to as quantizing noise.

In order to reduce the quantizing noise, tapered steps could be used rather than uniform ones, as shown in Fig. 1(b). In this way a given number of steps can be assigned in greater proportion

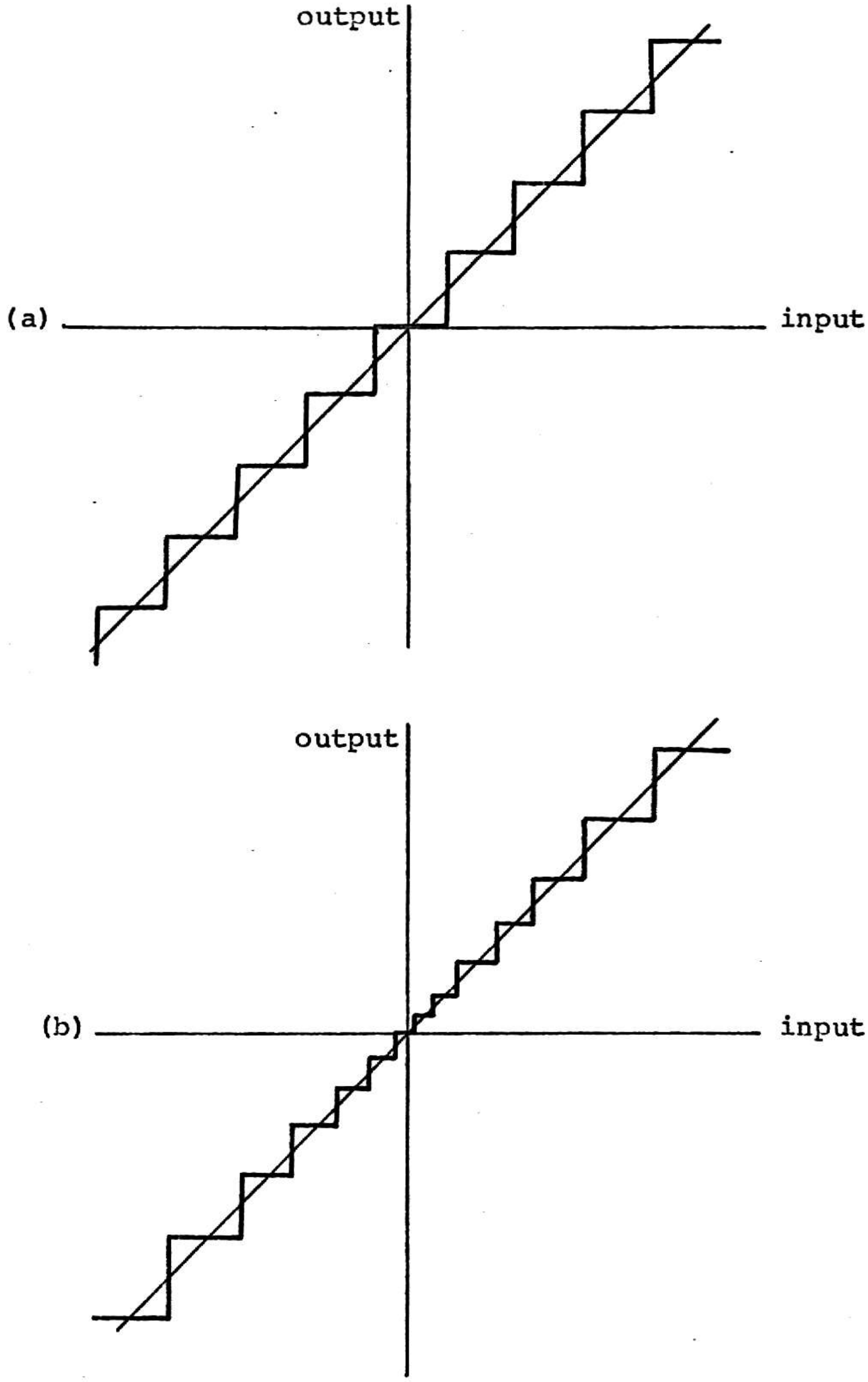


Fig. 1. Relation between input and quantized output, with quantization uniform in (a) and tapered in (b).

to the low amplitudes than to the highs. There results a degree of step subdivision sufficient to care adequately for the low-amplitude sounds including noise.[12] This is used in nonlinear coding process, but it still poses a serious problem in coding accuracy for the smallest steps.[7] In general, it is more practical to use logarithmic companding to reduce the noise which will be described later.

2.4 Time-Division Multiplexing [13]

The sampling theorem described previously makes it possible to transmit the complete information in a continuous bandlimited signal by transmitting samples of f(t) at regular intervals. The transmission of these samples engages the channel only part of the time, and it is possible to transmit several signals simultaneously on the time-sharing basis. This is done by samping all of the signals to be transmitted and interlacing these samples as shown in Fig. 2. [13]

Fig. 3 shows a block diagram representation of a transmitter and a receiver of a time division multiplexed system. At the transmitter, the commutator is switched from channel to channel in a sequence by the timing circuit, which also generates the sampling pulses. Thus the commutator connects different channels in a sequence to the sampling circuit, which samples all of the signals in a sequence by pulses generated by the timing circuit. The commutator switching and the sampling pulses are in synchronism. The output of the sampling circuit is thus a signal which consists of samples of all of the signals interlaced. At the

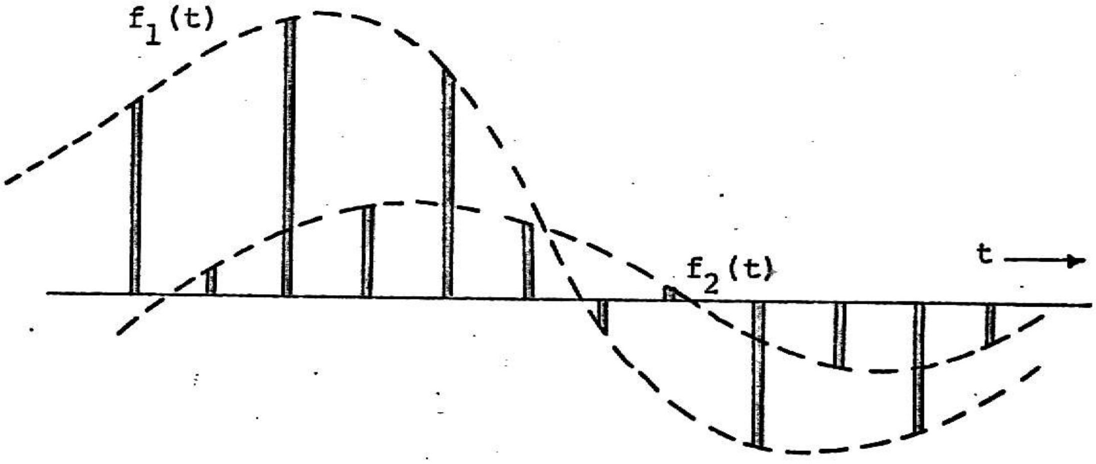


Fig. 2. Time multiplexing of two signals.

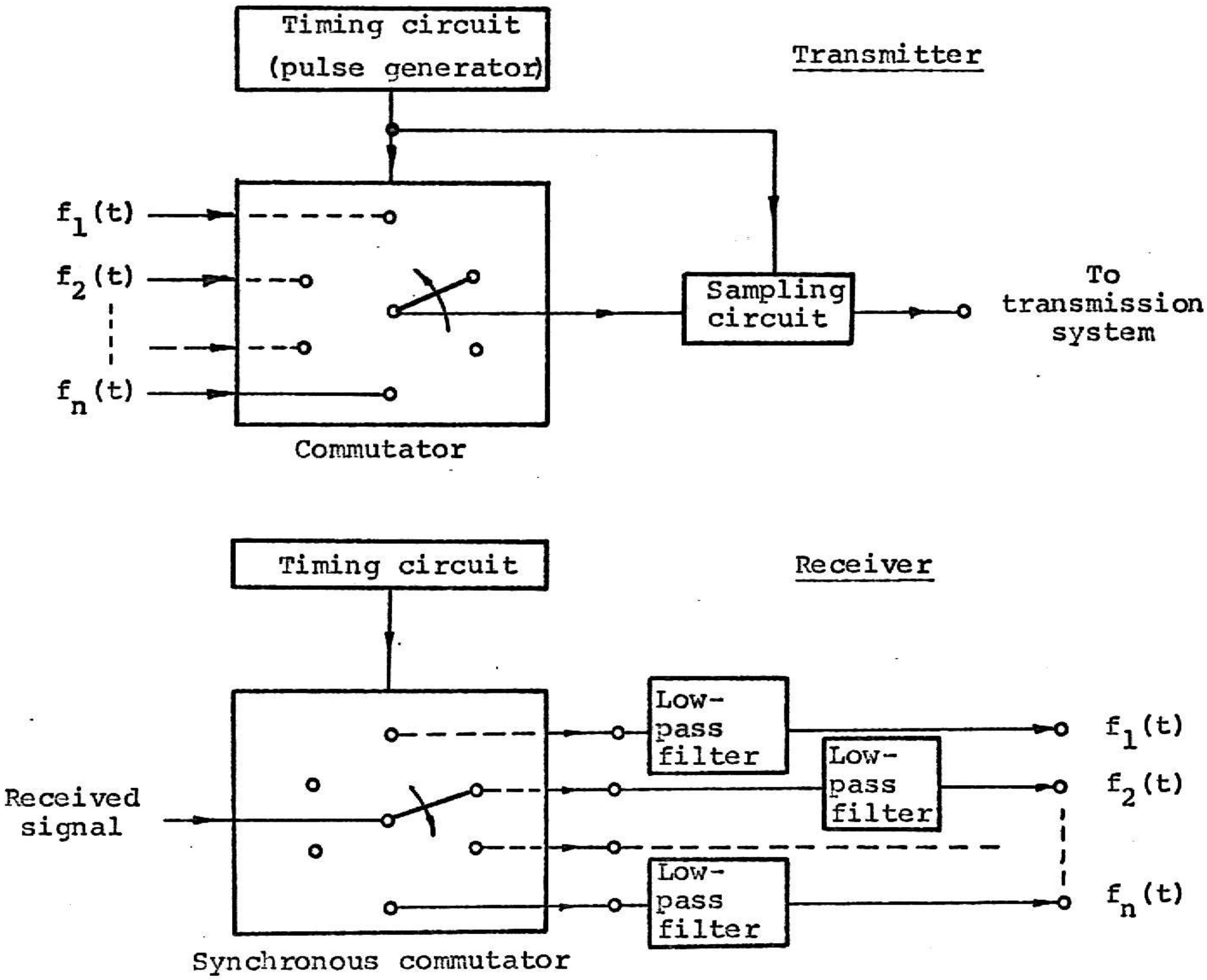


Fig. 3. Time multiplexing of n channels.

receiver, another timing circuit which is in synchronism with that at the transmitter is used to switch the commutator to different channels. The samples of various signals are now properly separated. The desired signal is recovered from each channel by a low-pass filter.[13]

2.5 Coding

The coder is required to set up a pulse code combination for each quantized signal value. A great many codes are available, but in practice, the binary code was found to be the simplest one to express the quantized signal.[12]

Coding is the process of conversion of signals to a group of binary code, and vice versa. At the transmitter, a quantized signal is expressed as a seven-digit binary code, and then transmitted over the line. This process is called encoding. Decoding, at the receiver, is the inverse operation of encoding, restores the digit binary code received to original signal which is an input to the coder.

Suppose that the three pulses used for expressing the digits can have the combinations representing the amplitudes of signals, as shown in Table 1.

In Fig. 4, the continuous signal was sampled and quantized at the instants indicated, and coded into the binary number at the right, which represented the amplitude of the signal to the nearest half quantum.[10]

TABLE 1

Amplitude Represented	Code
0	000
1	001
2	010
3	011
4	100
5	101
6	110
7	111

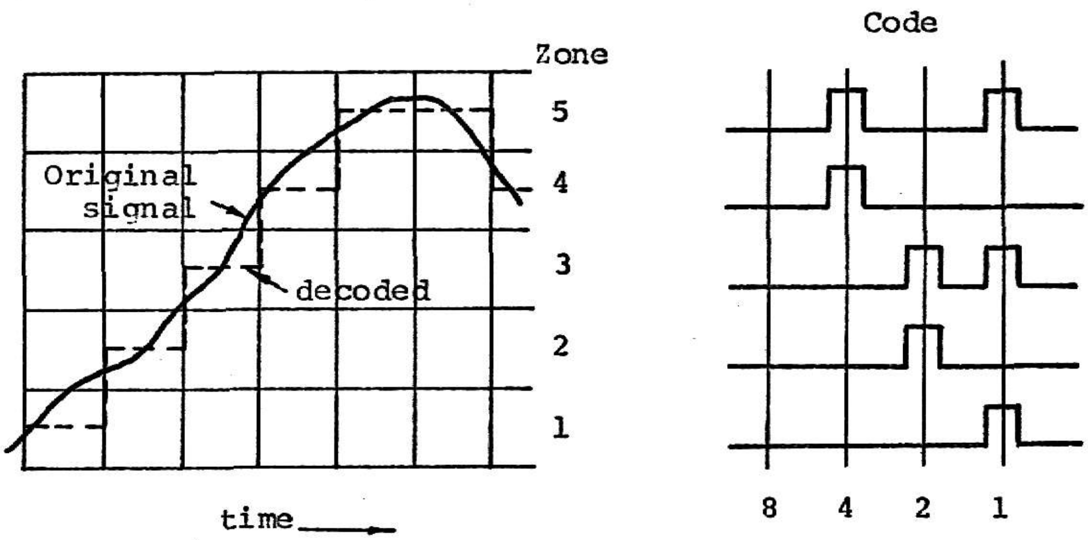


Fig. 4. Pulse-code modulation

2.6 Decoding

In Fig. 4, the binary codes to the right are transmitted to the destination, where the code groups are decoded into a voltage proportional to the code, producing the received signal represented by the dotted "staircase" signal. If the staircase signal is passed through a low-pass filter to smooth the steps, an improved approximation of the original signal is produced.[10] The decoded signal is made a still better approximation of the original by decreasing the interval between samples (i.e., increasing sampling frequency) and increasing the number of amplitude zones (digits per code). However, this causes a limit to the number of channels that can be used by time-division multiplex and increases the complexity of equipment.

The decoding and encoding are inverse operations. That is, the "units" pulse is decoded first, and the pulse with the highest place value last.[8]

III. SYSTEM ORGANIZATION [5]

The pulse code modulation (PCM) system is the newest cable system. It is economical for short distances of less than ten to more than 25 miles,[5] working on exchange cable pairs. The function of repeaters between the terminals is to reconstruct the transmitted pulse train after it has traveled through a dispersive, noisy medium.

In this paper, a PCM system with terminals of 24 speech channels will be discussed. The amplitude of speech samples were expressed in seven-digit binary code. The repeatered line was designed for use with 19- or 22-gauge cable pairs.[5] Nominal repeater spacing was 6000 feet.

A block diagram of the speech portion of the system is shown in Fig. 5.[5] Incoming speech to a channel unit, after passing through the hybrid, is band-limited to reject all frequencies above 4 KHz. This band-limited signal is sampled 8000 times per second by the sampling gate associated with this channel. The resultant sample, whose amplitude is proportional to the signal level at the instant of sampling, is passed through the compressor, which gives preferential gain to low-level signals, and presented to the coder. The coder expresses the sample amplitude as a seven-digit binary number or one of 128 different possible levels. The first digit has the weight of 64, the last digit has weight of 1. The signal arrives at the coder on a pedestal 64 units high so that code 64 corresponds to zero signal amplitude. The seven-digit code goes onto the trnasmission line, followed by

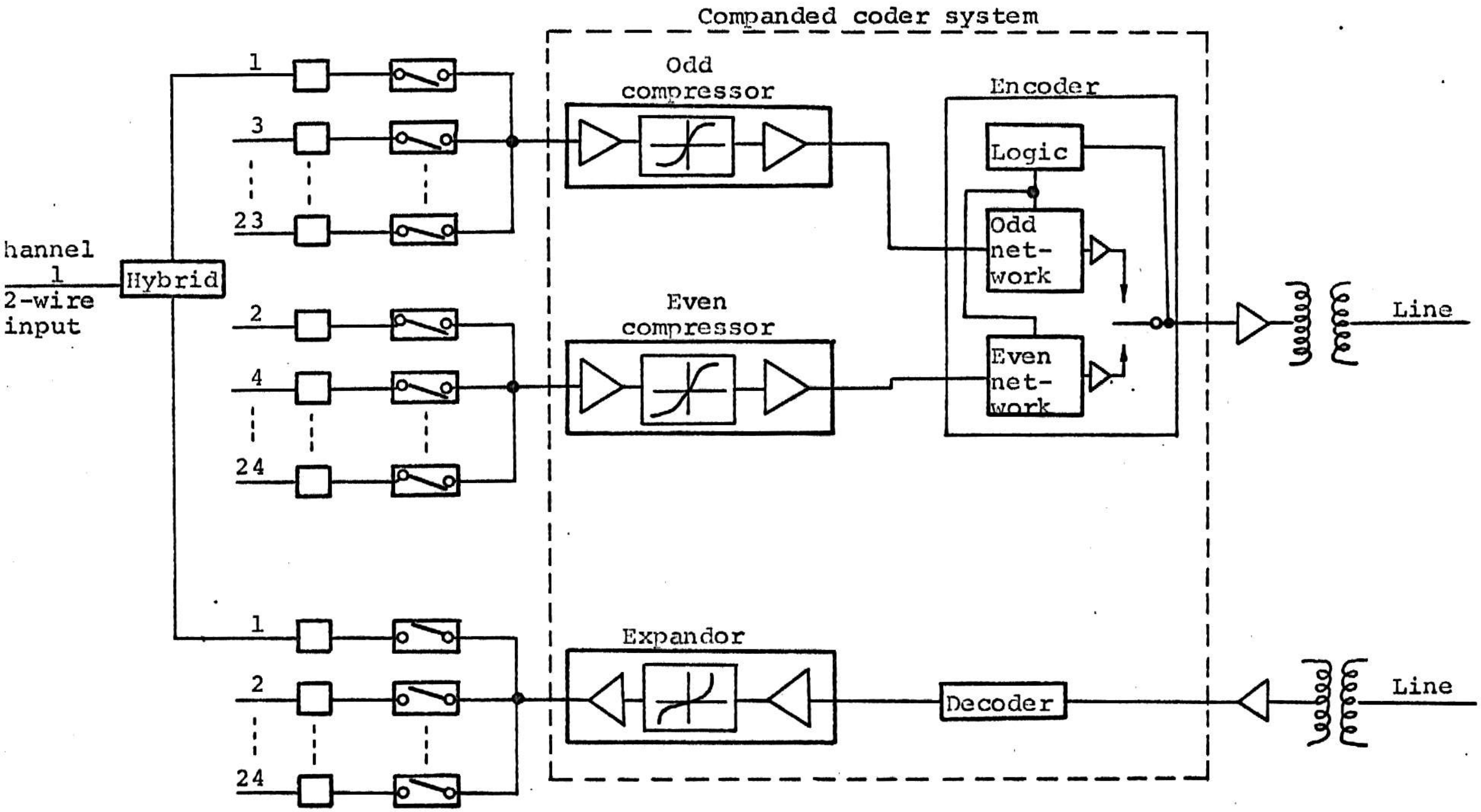


Fig. 5. PCM System

an eighth time slot which carries the supervisory signaling for that channel.

The channels are sampled in a recurring sequence, one sample from each channel or 24 samples being encoded and transmitted every 125 μsec. (=1/8000 sec.). Since each sample requires eight time slots including the signaling, these 24 samples require a total of 192 time slots on the line. An additional or 193rd time slot is added to permit synchronizing or framing the two ends of the system. These 193 time slots comprise a framing period. This is illustrated in Fig. 6.[5] There being 8000 such periods each second, the repetition rate of pulses on the lines is 1.544 million pulses per second (193 x 8000 pps). The time assigned to one bit is about 0.65 (=125/193) μsec.. The pulses have 50 percent duty cycle and are therefore 0.325 μsec. wide.

These pulses, containing speech, signaling and framing information, arrive at the receiving terminal after being reconstituted several times by the regenerative repeaters which are spaced at 6000-foot intervals on the line. At the receiver, the pulses are sorted out, the signaling pulses being directed to the individual channel signaling units and the speech-coded signals going into the decoder. The decoder output is a pulse amplitude modulated (PAM) signal whose amplitude is equal to the amplitude of the input to the coder. The decoded PAM pulse passes through the expandor, which has the inverse characteristic of the compressor, providing more gain for higher-level signals. The expandor is followed by a wide-band power amplifier which raises the signals to a level sufficient to require no further amplification after being switched to the channel units. A low-pass filter in the receiving section of the channel unit integrates the samples to yield the original signal.[5]

With this general description, some more detailed consideration of the circuits used to process the signal will be discussed in the following sequence.

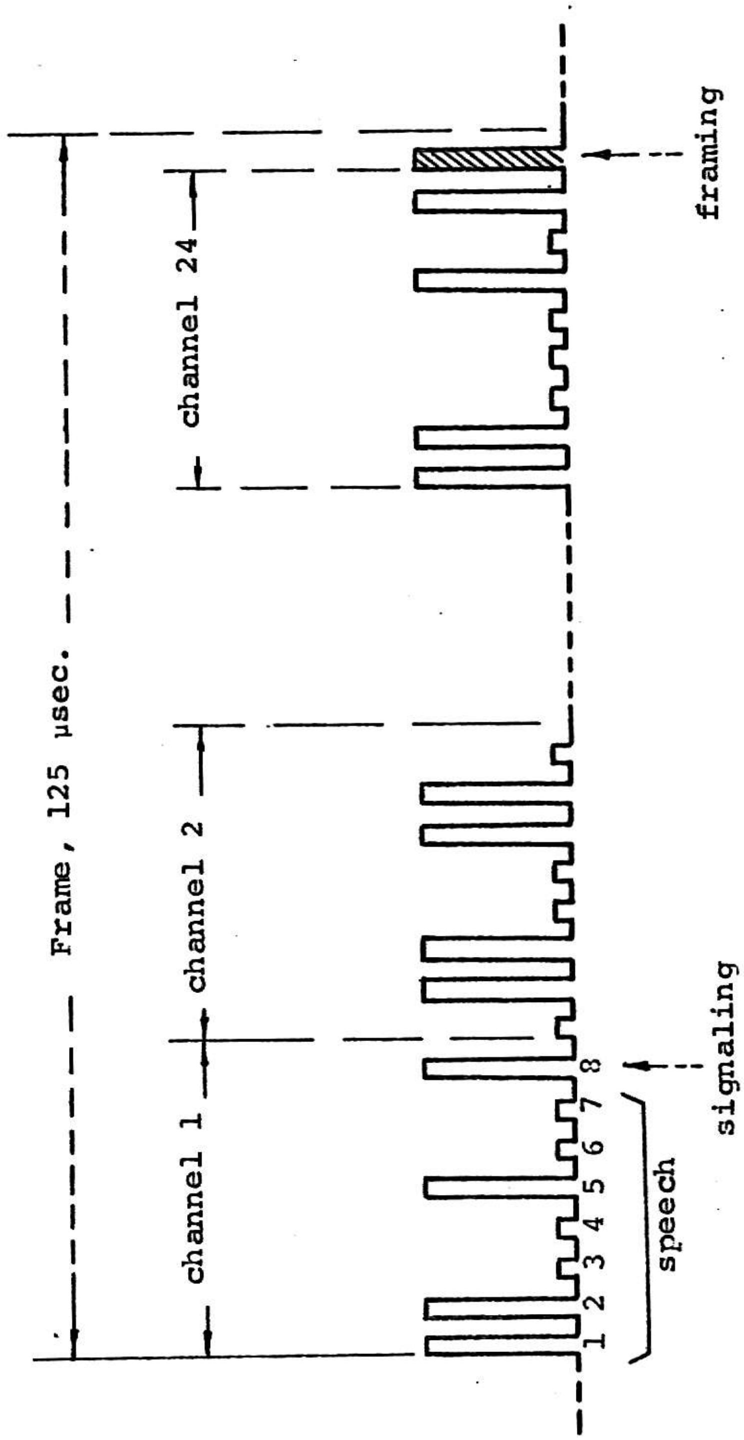


Fig. 6. System Time Assignment

IV. SAMPLE AND HOLD CIRCUITS

Since the effects of crosstalk and noise are a function of the signal level, it is advisable to keep the signal level as high as possible before encoding so that interference will have the smallest effect.[5] The energy sampling was considered to approach this result in no attenuation theoretically.[5]

It is important that the signal should be held until coding is completed. So at the common equipment level, energy sampling requires an erase interval between samples to avoid interchannel crosstalk. The time necessary for adequate erasure and new sample buildup leads to a requirement for two compressors and two encoders (as shown in Fig. 5) plus transfer circuits, so that one set can erase and build up while the other is coding, and vice versa.[2]

In the new PCM carrier system (VICOM terminal), a two-step sampling process is used, utilizing unbalanced channel gates without transformer drive, and a single compressor and encoder.

A typical channel gate is shown in Fig. 7.[2] Sampling by channel gates is the first step in a two-step sampling procedure. The second step of the sampling process is accomplished by the circuit shown as Fig. 8.[2] The sample-and-hold operation provides an input to the coder which is stationary through the coding interval.

During coding, the switch is open and A2 holds its output because the holding capacitor supplies its input. The high

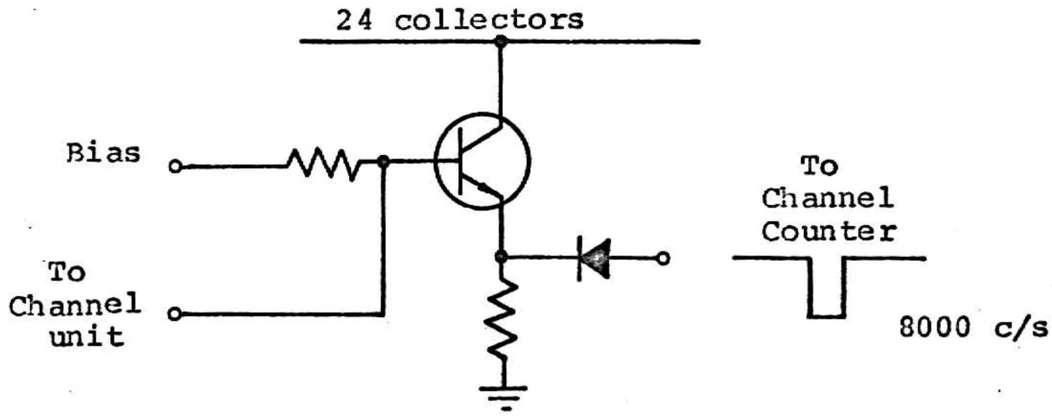


Fig. 7. Channel sampling gate

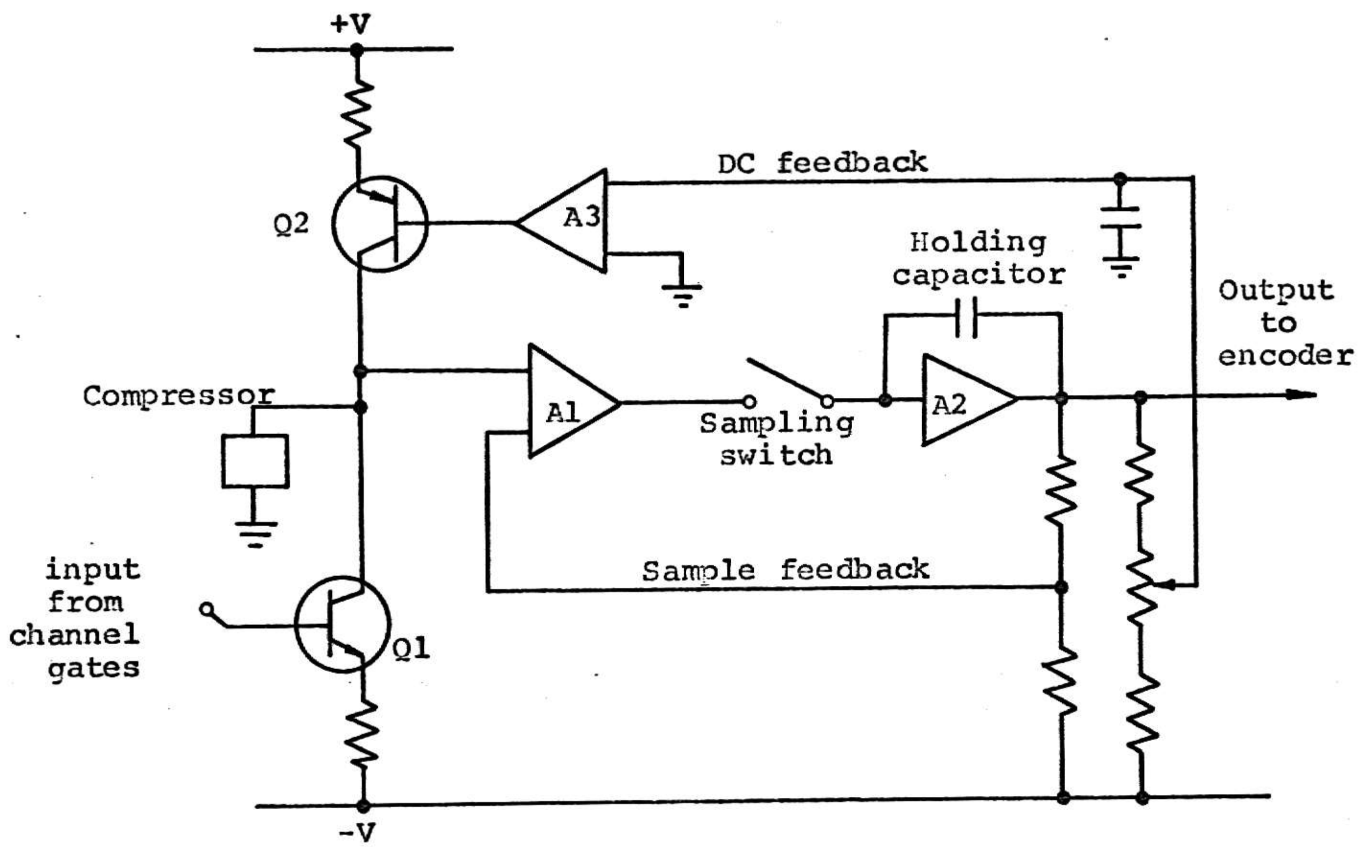


Fig. 8. Sample and hold circuit

input impedance of A2 prevents discharge of the holding capacitor during the coding interval. After coding, the switch closes again and a new sample value is established at the output of A2 before the next coding interval. It is not necessary to erase the holding capacitor between samples because, during sampling, A2 drives to the present sample value regardless of the initial condition at the start of sampling.[2]

The timing diagram of Fig. 9 illustrates the time relationship between the two steps of the sampling process.

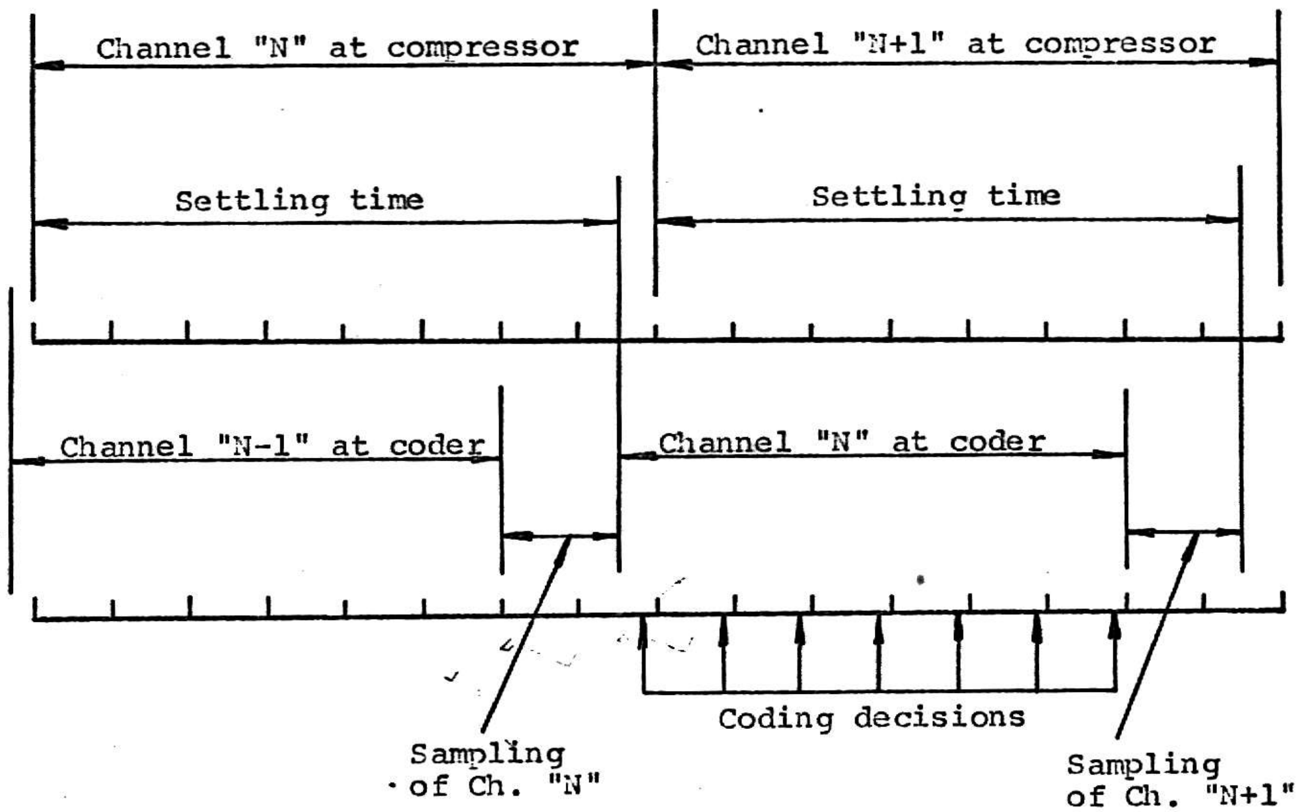


Fig. 9. Timing of sample and hold circuit

V. COMPANDED CODER SYSTEM

The companded coder system is the major part of the terminal for a PCM system. It consists of a logarithmic instantaneous compandor plus a linear (equal-step) coder. It performs the analog-to-digital and digital-to-analog conversions for the PCM terminal.

5.1 Companding Characteristic[7]

As shown in Fig. 10(a), if a moderate number of equal steps are used to quantize a given range of signals, the weakest signals have the most serious quantizing distortion. For instance, a single quantum step has a 50 percent error.

It is more practical to taper the signal in the manner shown in Fig. 10(b) to spread weak signals over a considerable number of quantum steps and reduce the quantizing noise. It is referred to as logarithmic compression.

A modified logarithmic characteristic of the type defined in Fig. 11[7] has been found to be desirable when the message is speech. Through its use, quantizing distortion may be reduced to an acceptable value for weak signals, with an acceptable impairment for strong signals, where quantizing noise is not so noticeable.

B. Smith has proposed a logarithmic compression characteristic of the form [6]

$$y = \frac{\ln (1 + \mu x)}{\ln (1 + \mu)} \qquad (10)$$

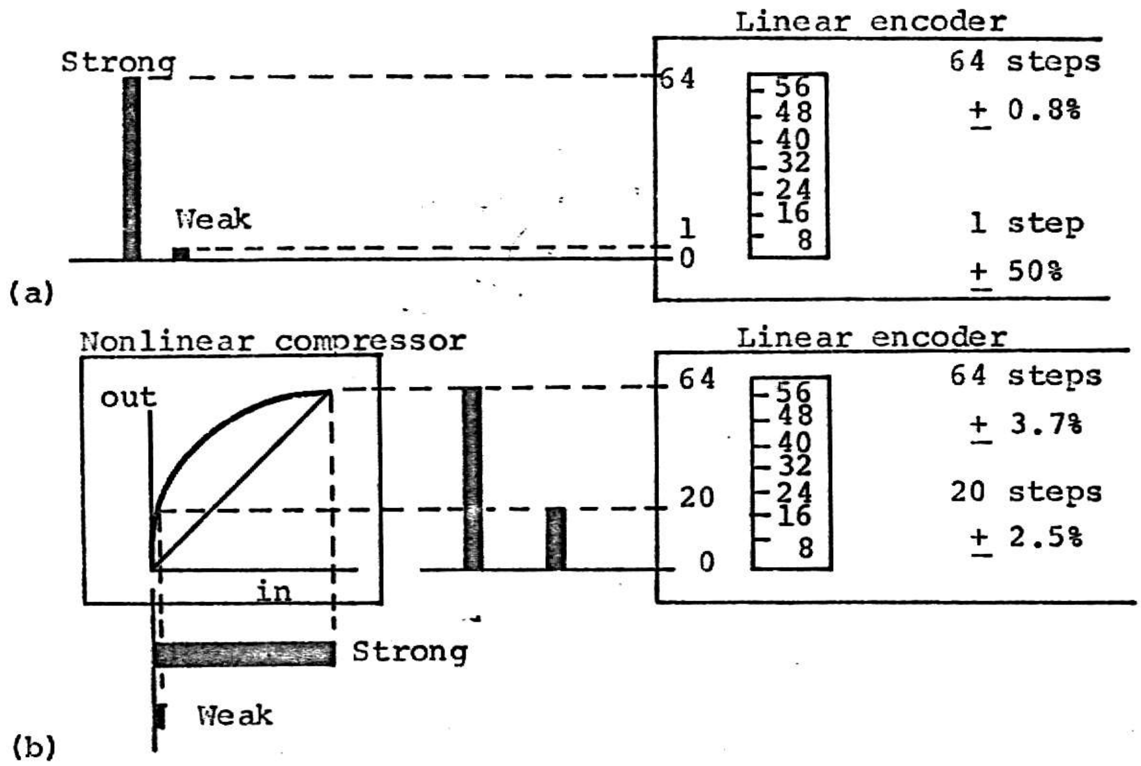


Fig. 10. Reduction of quantizing distortion

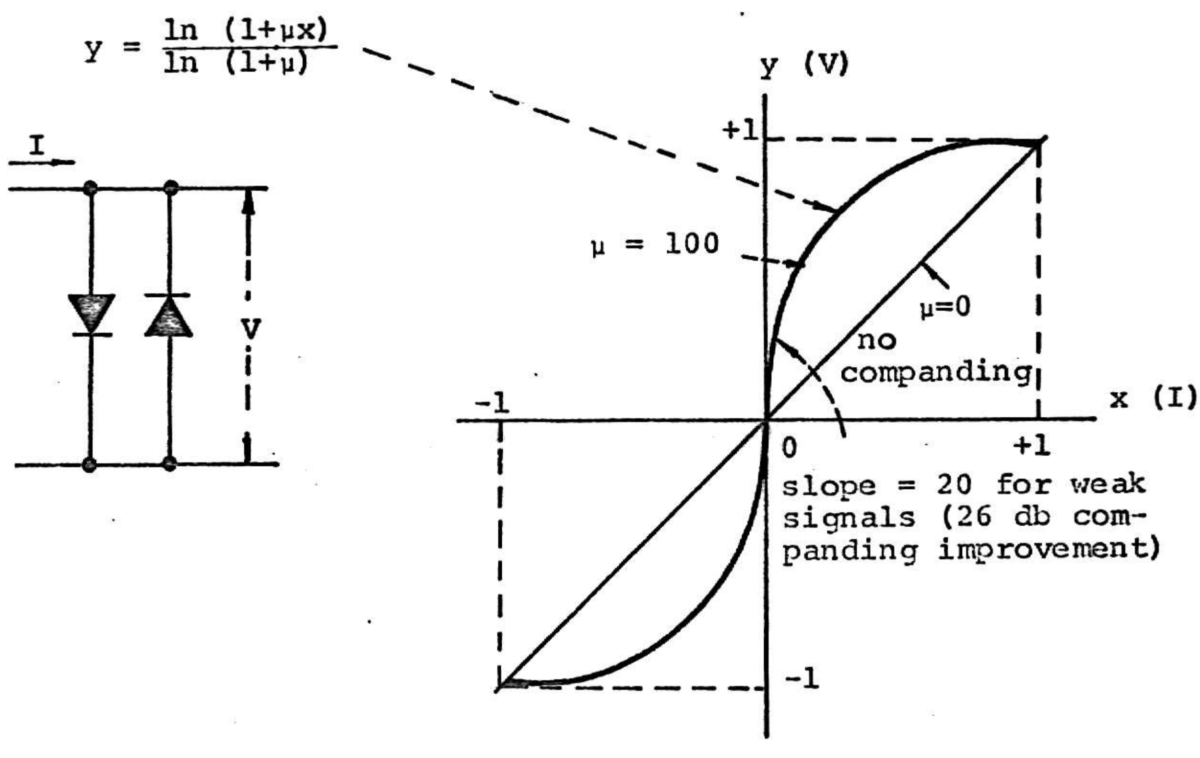


Fig. 11. Companding characteristic

where μ is the compression ratio.

The advantages of high values of μ are:[7]

(a) Obtaining a large companding improvement for weak signals.

(b) Reduction of idle circuit noise and interchannel crosstalk due to the irregular excitation of weak signals.

(c) Minimizing the clipping of large signals. For this, a high system overload value relative to the weak signal level has to be maintained.

The disadvantages of high values of μ are:[7]

(a) The difficulty of achieving sufficient stability in system net loss for high level signals.

(b) The difficulty of achieving and maintaining satisfactory "tracking" (true inversity) between compressor and expandor.

(c) The difficulty of achieving sufficient bandwidths in the compandor networks.

(d) The difficulty of holding the dc component of the multiplexed signals to a value low enough for full exploitation of high μ.

The practical value of μ selected is a compromise between the two. A choice of $\mu = 100$ was made for the experimental compandor.

In Fig. 11, x represents the input and y the output in the case of a compressor; while for the expandor, y represents the input and x the output.

5.2 Instantaneous Compandor[1,7]

The practical companding network is shown as Fig. 12.[1] For the compressor, Fig. 12(a), a current proportional to the signal is injected into the network, and the coresponding voltage is read out and transmitted through the system to the expandor.

For the expandor, Fig. 12(b), the voltage is impressed on the network, and the corresponding current is read out as the received signal.[1]

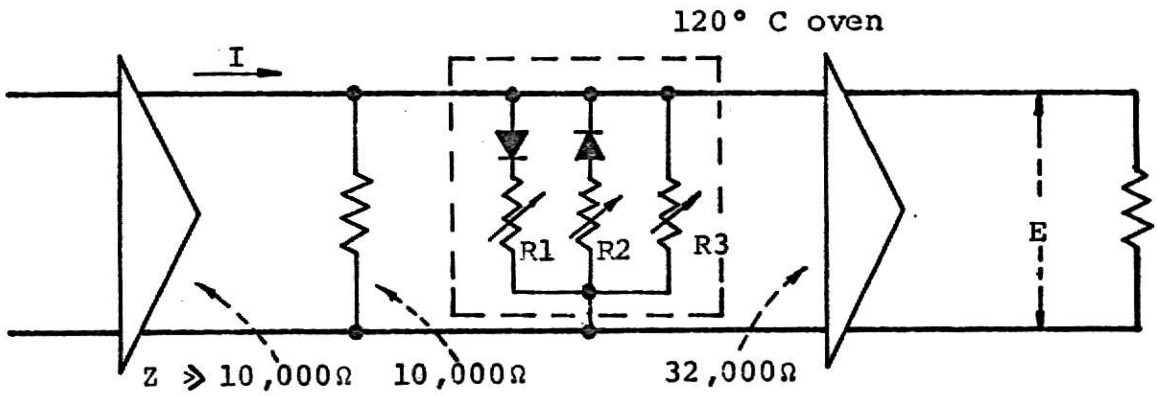
The compandor achieves its non-linear characteristic by the use of diodes whose forward voltage-current characteristic is logarithmic. As shown in Fig. 12, the diodes shunt the signal path in the compressor and are in series with it in the expandor. Initial pairing of diodes at medium currents plus appropriate adjustments of R1, R2, and R3 (or R1', R2', and R3'), will tend to linearize the characteristic and made the "standard" curve as shown in Fig. 12(c).

This addition of resistances yields the following advantages in practice:[7]

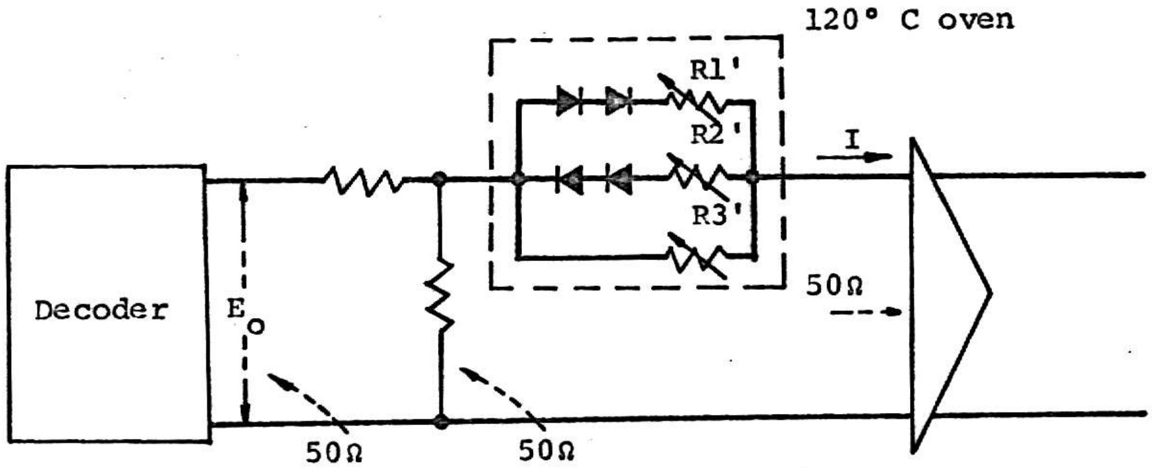
(a) Partially compensating the initial differences between diodes by appropriate choice of the resistors.

(b) Shunt resistors avoid the need of an infinite-impedance generator and load for the compression network. Series resistors avoid the need to zero impedance generator and load for the expansion.

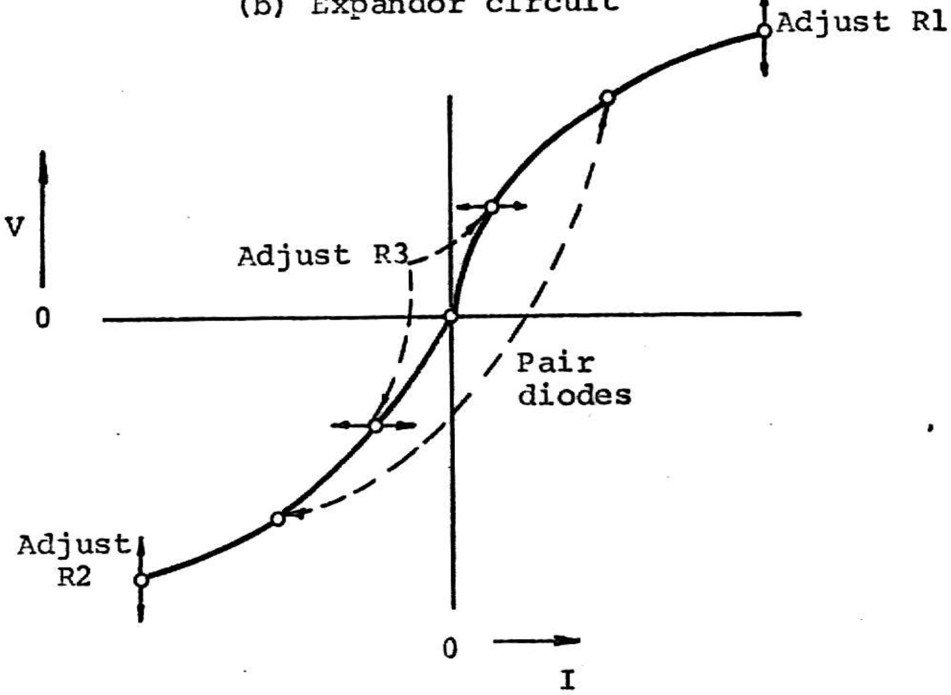
(c) It allows the direct and stray capacitances of the diodes and their interconnecting configuration to be reasonably high without sacrifice of the bandwidth required.



(a) Compressor circuit



E_o = Open circuit voltage of decoder
(b) Expandor circuit



(c) Characteristic adjustment

Fig. 12. Practical companding network

(d) It provides masking of those changes that will inevitably occur in the diodes (due to temperature variation and aging).

(e) Reducing the network sensitivity to those d-c components that will inevitably occur in the multiplexed signals.

A good network stability is achieved through the use of stable diodes and resistors housed in an oven at a temperature of 120°C.

5.3 Equal-Step Seven-Digit Encoder[5,7]

A simplified block diagram of digit-at-a-time or sequential comparison encoder is shown in Fig. 13. This coder successively compares the input pulse amplitude modulation (PAM) signal to binary weighted currents and generates the PCM signals as the comparison is taking place.[5]

The switches 1 through 7 are swung from ground to battery in sequence under the control of leads from the digit generator (D1-D7).

Each switch closure substracts an amount of current from the current-summing point proportional to the conductance in series with the switch. Thus, closing switch No. 1 subtracts 64 units of current; switch No. 2 subtracts 32 units; etc.[5]

For example, connect the first switch to the $-E_{REF}$, while all other resistors are connected to ground. If the resultant signal at the summing amplifier output is positive, indicating that the reference current exceeds the signal current (i.e., signal current is less than 64 units), then a most significant PCM bit is transmitted. A feedback signal to the logic causes the memory element to reset; i.e, a pulse is generated and the first switch is released, and the signal is then equivalent to 32.

If the resultant output of the summing amplifier is negative, thereby indicating that the signal exceeds the reference current (i.e., signal is greater than 64 units), the decision circuit transmits a zero for the most significant PCM bit. In this case,

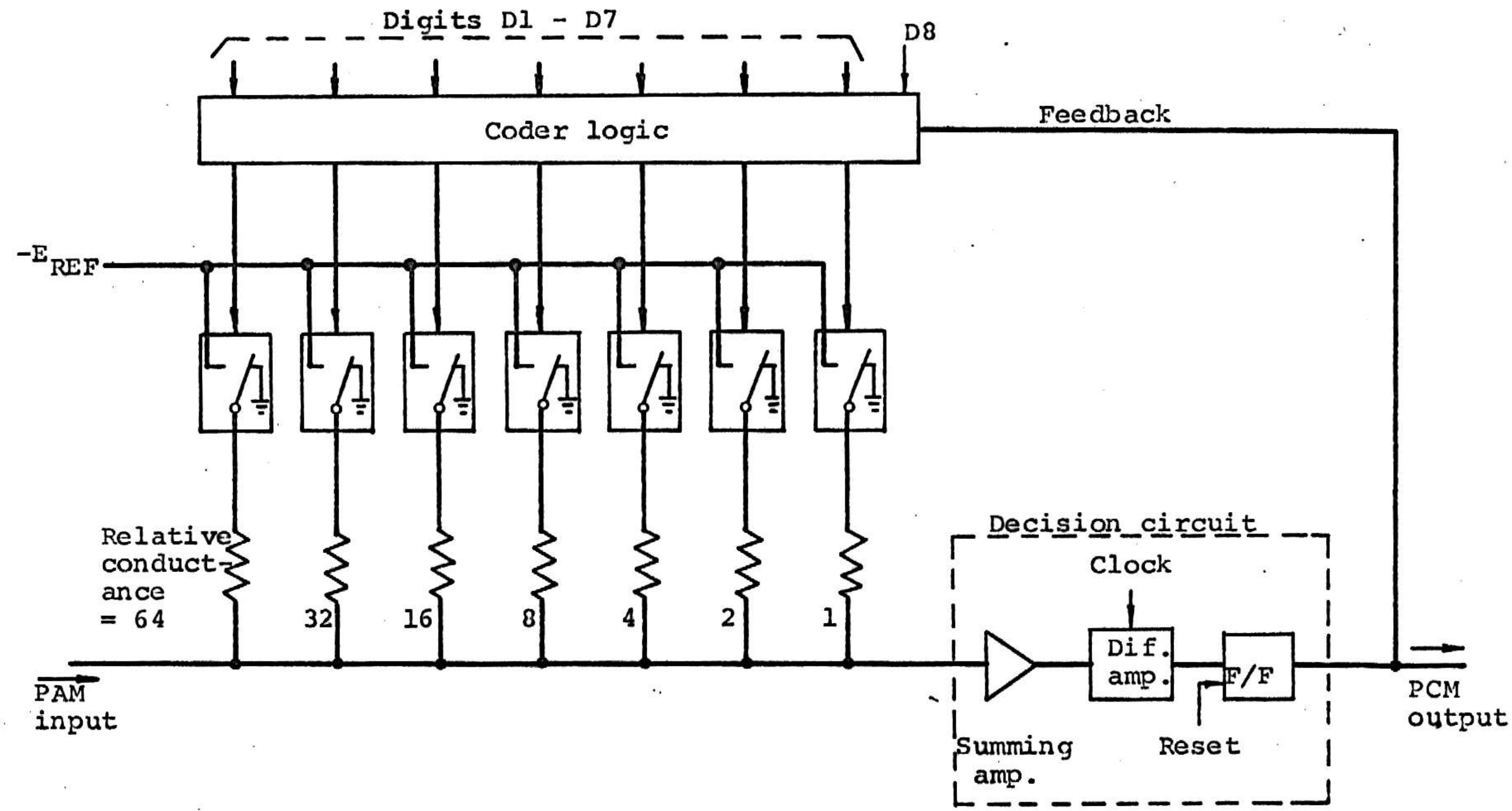


Fig. 13. Sequential comparison encoder.

no feedback signal is generated, and thus the first switch is connected to the reference voltage for the remainder of the encoding cycle. That is, no pulse is generated, and the next comparison will determine whether the signal exceeds 96.

This process is contined through the seven digits, a pulse being generated each time the PAM signal is less than the sum of the reference currents. Thus the encoder produces the prime of the code corresponding to the signal amplitude. For example,

$$11 = 0' \times 8 + 1' \times 4 + 0' \times 2 + 0' \times 1$$
$$= 1 \times 8 + 0 \times 4 + 1 \times 2 + 1 \times 1$$

where 11 is the magnitude of the signal and 8, 4, 2, and 1 are the magnitudes of the references. The transmitted code corresponding to the signal amplitude, for this example, is 0100.

The logic portion of the coder uses flip-flops to keep selected switches operated. All the flip-flops are reset by the eighth digit from the digit generator. The entire encoding process is shown as Fig. 14.

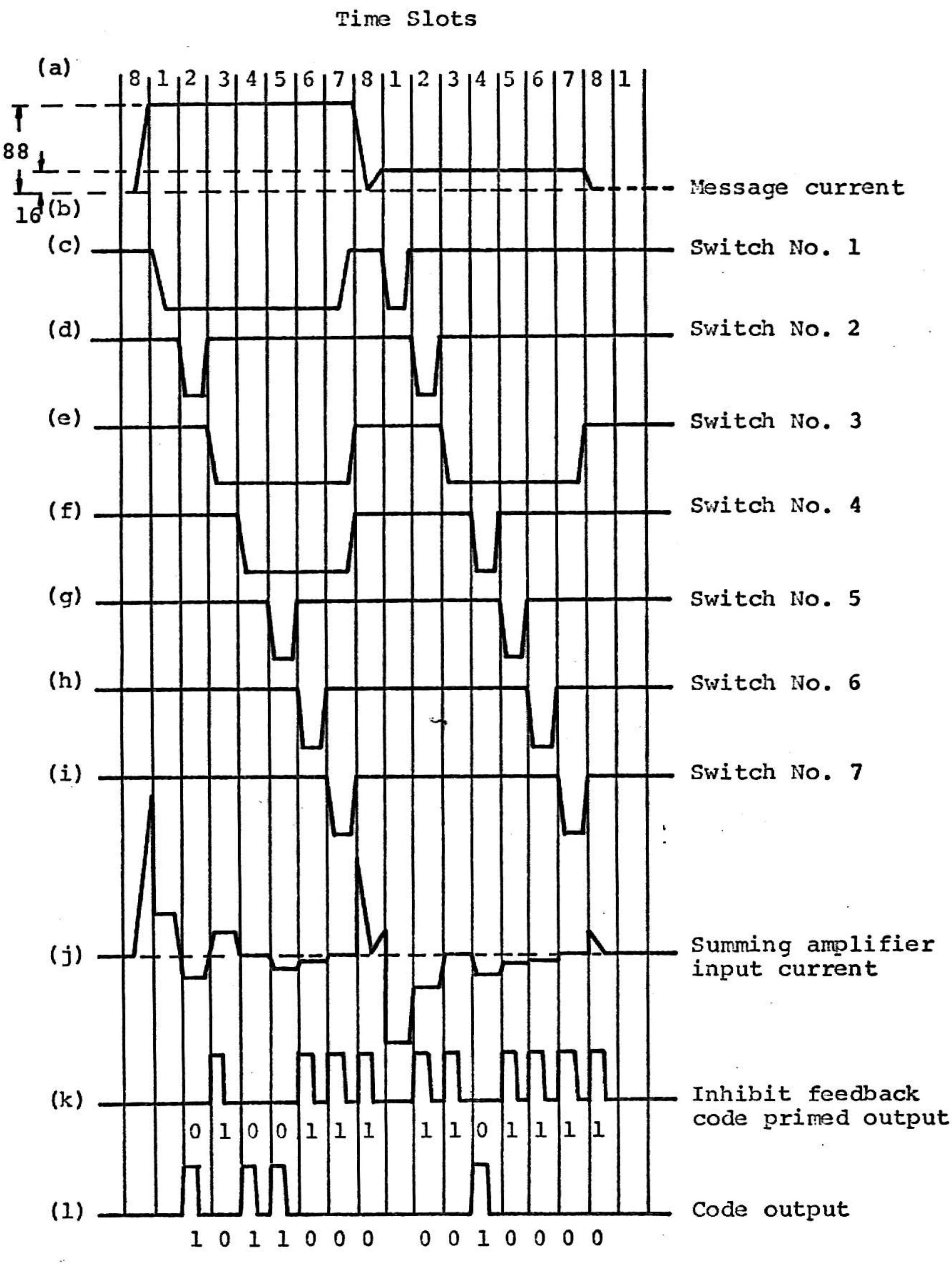


Fig. 14. The encoding process

5.4 Decoder[5,7]

The decoder is similar to the encoder in its operation. Its simplified block diagram is shown as Fig. 15.[5] A decoder network consists of a weighing network (including the relative conductances), switches, and storage devices that permit serial-to-parallel conversion of the received code.

In response to a pulse in a given time slot on the PCM line, a reference voltage is applied to a binary-weighted resistor. This supplies into a summing point a current proportional to the weight of the pulse. Pulses from the PCM line are steered to the correct switches by logic gates using pulses from the receiving digit pulse generator. The resultant decoded signal is then expanded and amplified.

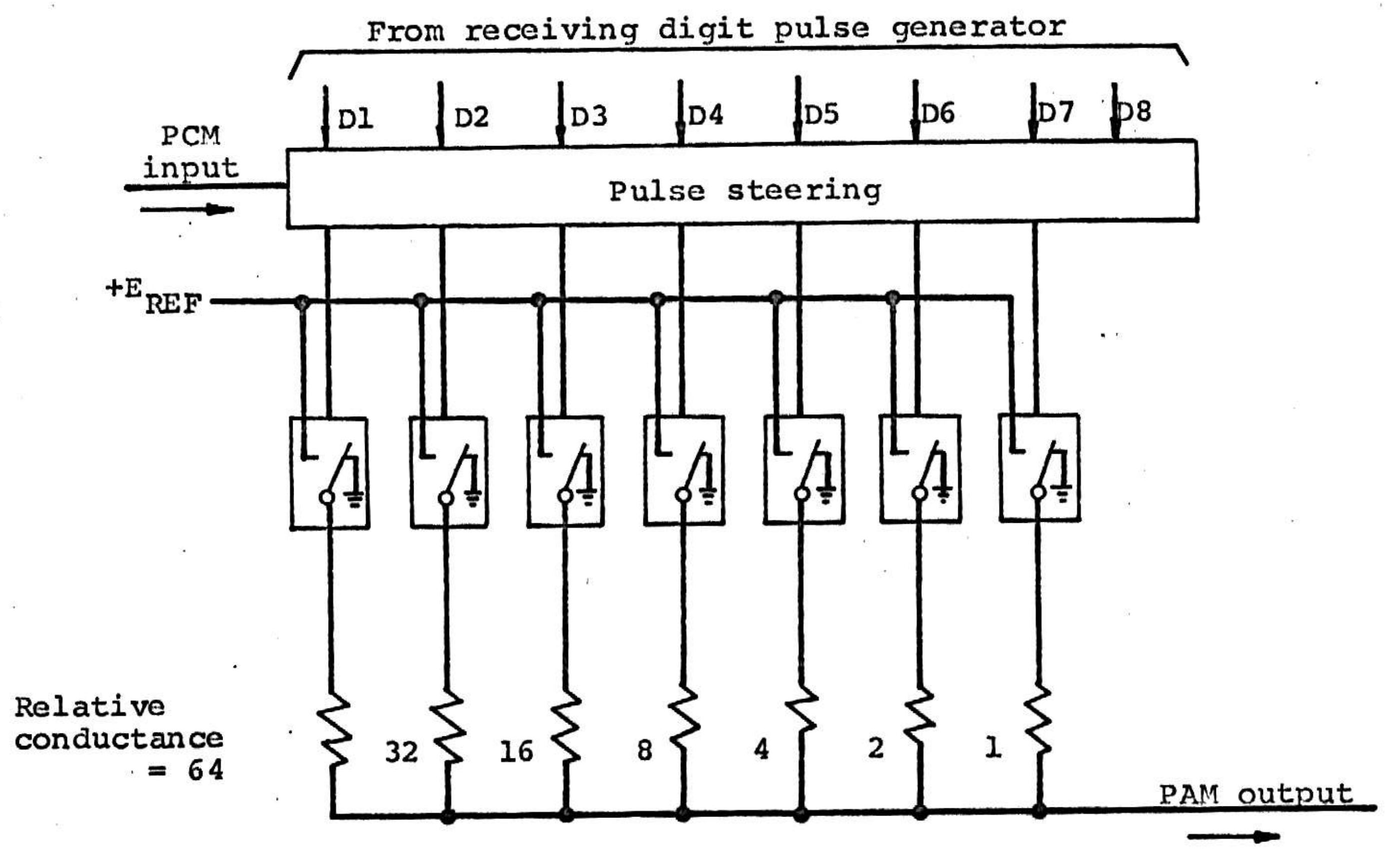


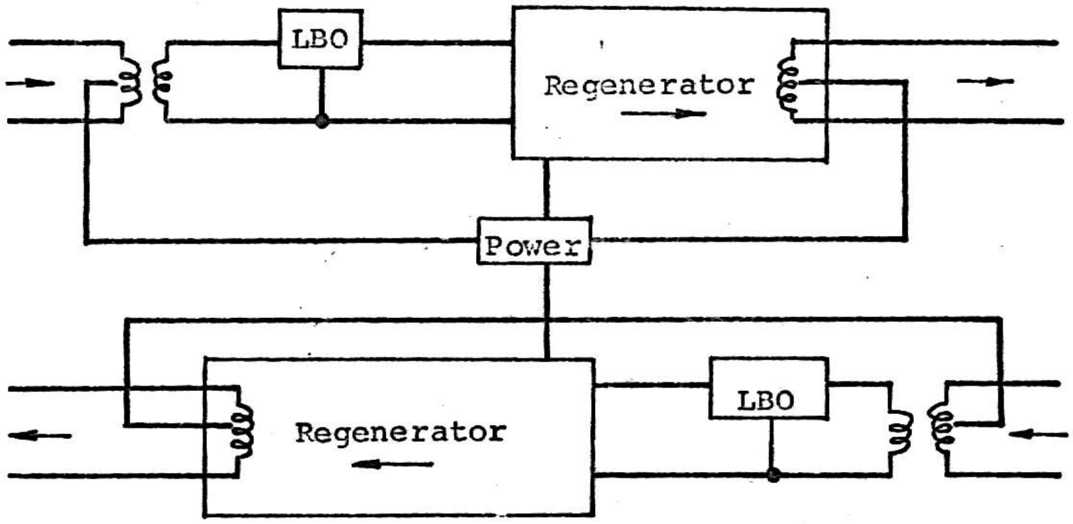
Fig. 15. Simplified block diagram of decoder

VI. REPEATERS

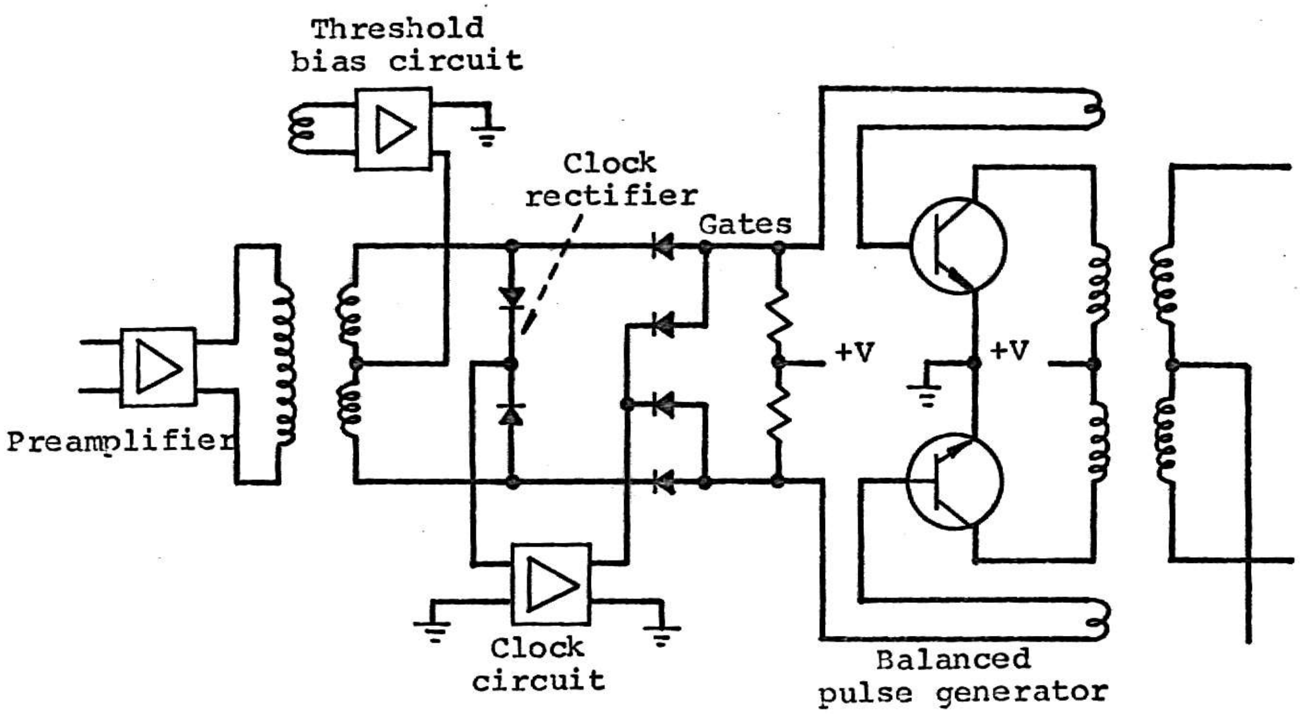
The digital transmission line, or repeated line, extends from terminal to terminal of a system, and consists of two cable pairs equipped with repeaters for two directions of transmission. The repeaters are fully regenerative and receive their timing information for the regenerative process from a resonant circuit driven by the incoming pulse train. Very simply, the repeater function is to "look" at a received pulse train and emit a "new" pulse for each received pulse.

If the noise or interference of PCM is sufficiently strong, then the network can not distinguish signal pulse from interference. In practice these parameters are controlled by repeater spacing. The nominal spacing was found to be 6000 feet,[4] since such spacing produces a very difficult interference situation. Also, a bipolar pulse pattern was used in the transmission line to reduce crosstalk between systems.[3,4]

The typical line repeater[1] is shown in Fig. 16(a), and Fig. 16(b) is a regenerator. Each repeater contains two regenerators, mountings for two line-build-out (LBO) networks, which make any cable length appear as approximately 6000 feet to the preamplifier. The preamplifier amplifies and equalizes the incoming signal, reshaping each pulse to reduce its dispersion into adjacent time slots. Its output drives not only the regenerating circuit but also the clock circuit and the threshold bias circuit. The threshold bias circuit sets the decision level which determines for each time slot whether or not a pulse is to be regenerated. The clock rectifier converts the



(a)



(b)

Fig. 16. (a) Repeater configuration;
(b) Repeater regenerator

incoming bipolar signal into a unipolar pulse train which contains a strong component of energy at the original repetition rate of 1.544 megacycles. This 1.544-megacycle component is selected by a tuned circuit, amplified, and shaped in the clock circuit to provide a turn-on and turn-off timing pulse for the beginning and end, respectively, of each pulse position. The generator puts out a "new" pulse when the preamplifier output exceeds the threshold level during the sampling instant. For bipolar transmission the regenerator is a balanced circuit of identical halves in push-pull.

In addition to the normal line repeaters, special repeaters are required at the transmitting and receiving terminals.[4] The transmitting repeater is peculiar only in that it must convert the unipolar signals to a bipolar pattern. The receiving repeater must deliver a unipolar signal and clock to the receiving terminal.

VII. TONE-DIAL TELEPHONES

7.1 Introduction

Prior to 1941 all automatic telephone switching systems were controlled by d-c pulse generated by contacts that make and break line current. A new system called "Tough-Tone Calling" was designed and carried out in 1959.[15] The rotary dial was replaced by ten buttons. When a customer pushed any one of the ten buttons, a pair of voice-frequency tones was activated which were used for transmitting digital information from telephone stations to central offices.

The code chosen for touch-tone calling is a "four-by-four" code. It consists of two frequency ranges, each containing four signal frequencies, and in which each character comprises one frequency from each group. The code chosen for practical use is 3 x 4 code. The arrangement of pushbuttons of telephone set is shown in Fig. 17. The remaining two empty spaces are for special service.

7.2 Choice of Signals

Past experience with voice-frequency signaling indicated that it is difficult to protect single-frequency tones against imitation by speech and noise.[16] In a four-by-four code, each signal is composed of two frequencies, in which each frequency is from each frequency range. This two-frequency code was found to be one which is not easily imitated by speech.[14]

The choice of frequenices of the signals is based on the

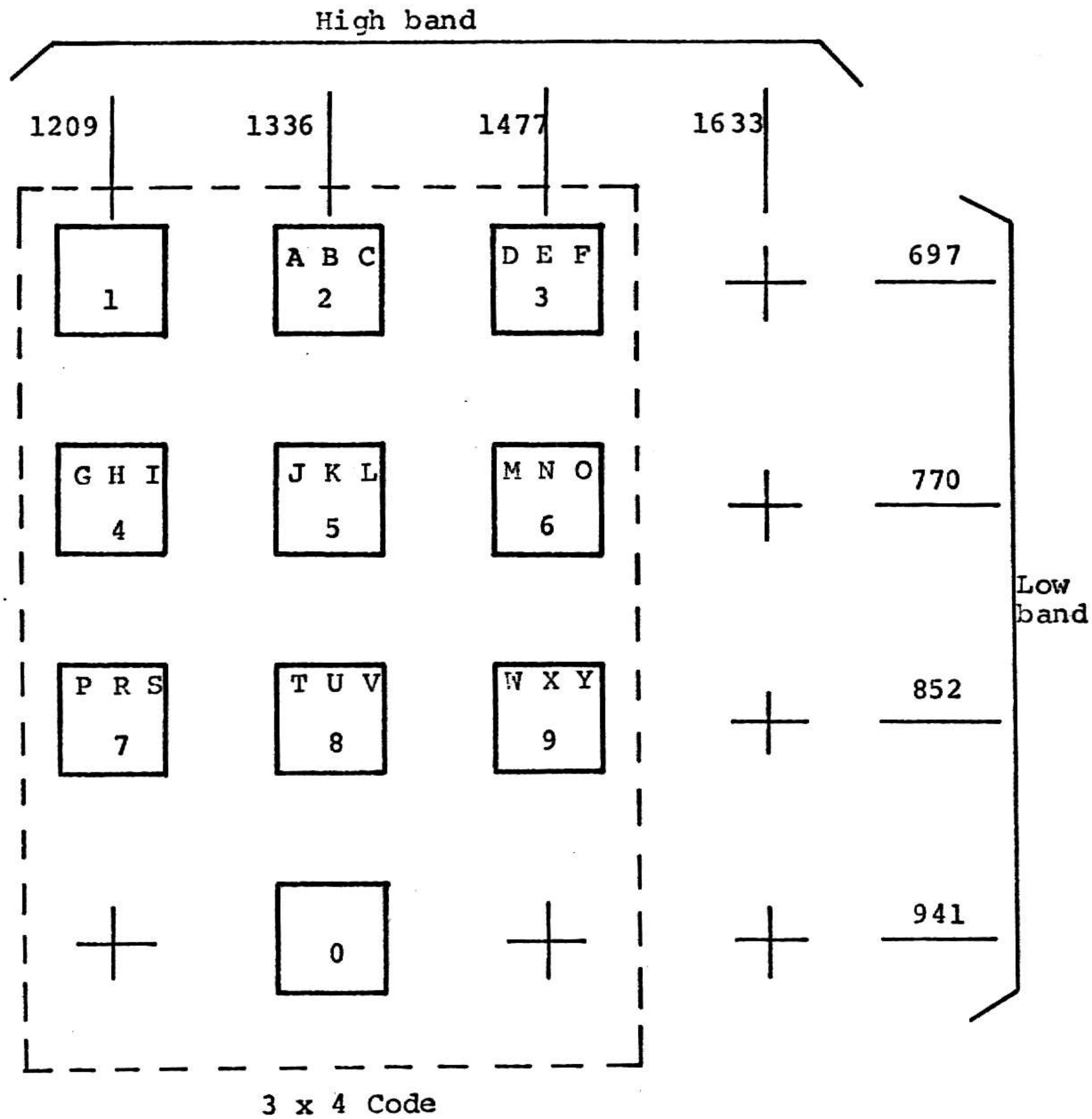


Fig. 17. Digit frequency assignments

following considerations.

(1) Frequencies used in the code must be confined to the transmission band of the telephone plant and therefore must lie wholly within the voice-frequency range.[16]

(2) The signals must be distinguishable from speech.

(3) Attenuation and delay distortion with frequency are least troublesome in the frequency range from approximately 700 to 1,700 Hz.[20]

(4) The frequencies in one code group should not be harmonically related to those in the other group.[16]

The following set of frequencies was chosen for a 4 x 4 code with previous considerations.

Low Group Hz	High Group Hz
697	1209
770	1336
852	1477
941	1633

Although the signal duration is almost entirely under control of customers, two time intervals need to be specified.[15] First, the signal must be presented and must endure long enough to be recognized; second, a recovery interval must be allowed before a second signal is presented. In general, 40 ms duration was taken as the minimum pulse duration to which the receiver must respond. The maximum pulsing rate designed for the receiver is 12 pulses per second.[16]

7.3 Signaling System[16]

The basic signaling system is shown in Fig. 18.[16] A two-frequency signal is generated at the telephone set, which contains a dual-frequency oscillator controlled by pushbuttons. This signal passes over the line to the receiver at the central office.

The separation filter at the receiver separates the two frequencies of a valid signal into their respective groups. A limiter for each group is for standardizing signal amplitudes. It can distinguish between valid signals and speech or noise, that is, guard action, the purpose of which is to reduce the probability of false response to speech or other unwanted signals. The response of each limiter to the single touch-tone frequency is a symmetrical square wave of fixed amplitude, which contains the fundamental and odd harmonics of the signal frequency. In each group, the tuned circuit selects the corresponding incoming signaling frequency and responds to the fundamental of the square wave with an amplitude sufficient to operate a detector. The detector makes a check to determine the presence of one and only one tone in each of the two frequency groups and converts it to dc marks, which are then forwarded to the converter. The converter, in turn, translates the dc marks into electrical signals suitable for the control of the switching elements.

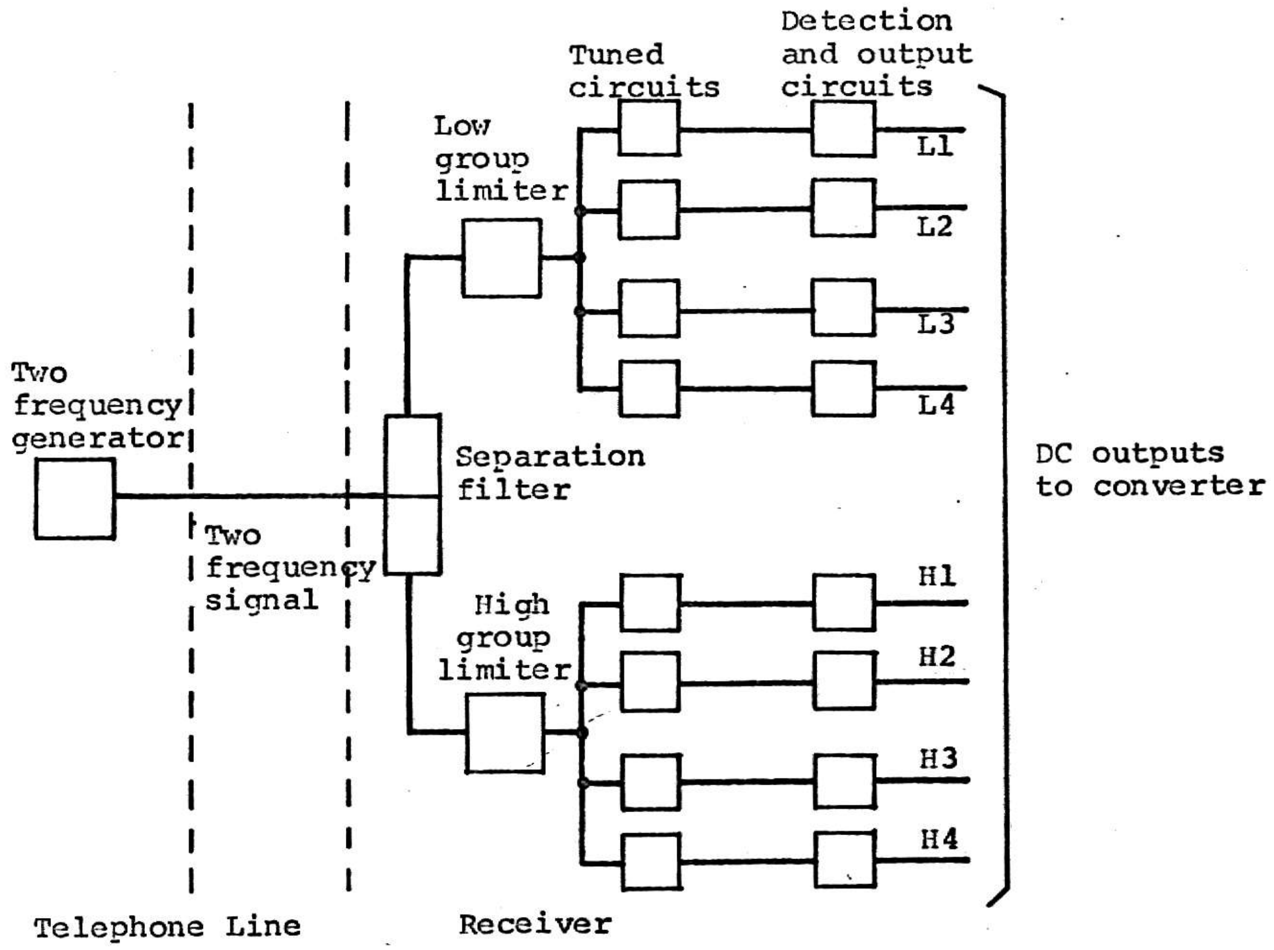


Fig. 18. Basic signaling system

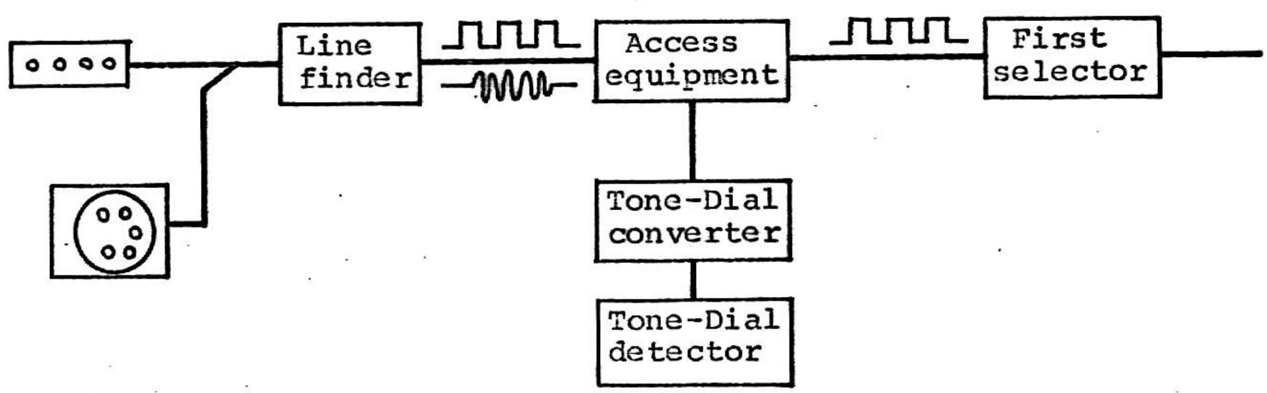


Fig. 19. Tone-Dial switching system

7.4 Tone-Dial Switching System[19]

The simple block diagram of tone-dial switching system is shown as Fig. 19. It consists of three main functional units: access equipment, tone detectors, and converters. The access equipment is used to connect the subscriber to a tone detector and converter combination during the dialing period, and then to switch through to the forward dial train and dismiss the detector and converter after signaling has been completed. The tone detector is used to receive the tone signals from the telephone set, check signals for validity, and convert them to dc marks. The converter is used to control the switch train during the signaling period, receive and store the marks from the detector, and convert the stored information into standard dial pulses that are transmitted to the forward switch train.

If the subscriber uses a rotary dial instrument, the pulses of the first digit are repeated to the first selector and, at the end of that digit, the access equipment is given a release signal to cause connection between the linefinder and first selector. Thus the converter is dismissed from the connection, and subsequent digits are pulsed through the switch train in normal fashion.

VIII. CONCLUSION

The pulse-code modulation technique tends to eliminate the problems of noise and crosstalk. Through the use of integrated circuits and solid-state devices, PCM system could be much more economical to manufacture than the current system.

The principal sources of noise in PCM system are quantizing, distortion and crosstalk interference. Quantization is the process of converting the sample of the signal to a finite number of discrete steps to permit digital encoding. The error or noise produced by this procedure is the major source of signal impairment. For time-division speech channels, non-uniform quantization was found to yield the best over-all signal-to-noise performance for a given bandwidth. With uniform quantization, in order to obtain comparable performance, it would require an increase in sampling frequency and/or an increase in the number of quantizing levels for the same signal range. As discussed previously, non-uniform quantization could be achieved by using a logarithmic compandor and a separate uniform encoder-decoder combination.

Crosstalk is generally due to the carryover from one pulse to the adjacent pulse. This problem arises when a single 24-channel system is considered to transmit in both directions, and is further compounded when many 24-channel systems are transmitted on separate pairs in the same cable bundle.[3] This problem could be reduced by using bipolar code and spacing repeaters by 6000 feet each.

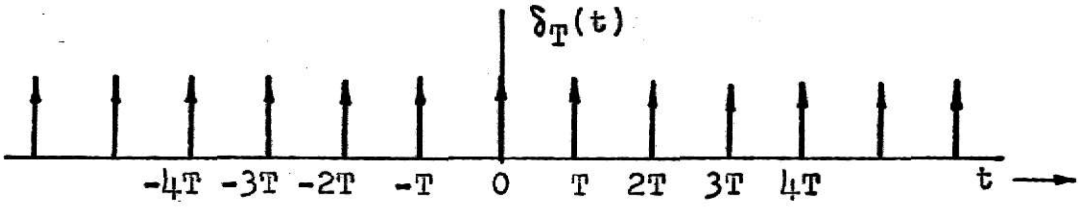
Since there is the possibility of regenerating signals in an ideal manner and thus avoid, theoretically, any loss of information, it is possible to obtain a quality of transmission independent of distance. But it is a little difficult to produce a reliable repeater at economical prices. Also PCM system are vulnerable to impulse noise, whereas FDM (Frequency Division Multiplexing) systems are much less troubled with this problem. In general, PCM is used more economically in many short-haul applications, and FDM works much better for medium and long-haul applications.

The advantages of pushbutton dialing are: greater each of dialing, increased speed by which a call can be made, and greater accuracy in dialing. These advantages will lead to the increasing demands of business, industry, and government for the faster transfer of information.

IX. APPENDIX

Shannon Sampling Theorem

Suppose a sequence of equidistant impulses of unit strength is given and separated by T seconds, as shown in the figure below.



It can be represented as the function $\delta_T(t)$.

$$\delta_T(t) = \delta(t) + \delta(t-T) + \delta(t-2T) + \ldots\ldots + \delta(t-nT) + \ldots$$
$$+ \delta(t+T) + \delta(t+2T) + \ldots\ldots + \delta(t+nT) + \ldots\ldots$$
$$= \sum_{n=-\infty}^{\infty} \delta(t-nT) \tag{A-1}$$

It is a periodic function with period T and its Fourier Series is

$$\delta_T(t) = \sum_{n=-\infty}^{\infty} F_n \, e^{jnw_o t} \quad (-\infty < t < \infty) \tag{A-2}$$

where

$$F_n = \frac{1}{T} \int_{t_o}^{t_o+T} \delta_T(t) \, e^{-jnw_o t} \, dt \tag{A-3}$$

$$w_o = 2\pi/T \tag{A-4}$$

Note that the choice of t_o is immaterial. Then the function $\delta_T(t)$ in the interval $(-T/2, T/2)$ is simply unit impulse $\delta(t)$.

Hence,

$$F_n = \frac{1}{T} \int_{-T/2}^{T/2} \delta(t) \, e^{-jnw_o t} dt \tag{A-5}$$

Since $\delta(t) = 0$ everywhere except $t = 0$. So Equation (A-5) can be written as

$$F_n = \frac{1}{T} e^{0} \int_{0^-}^{0+} \delta(t)\,dt = \frac{1}{T} \tag{A-6}$$

then

$$\delta_T(t) = \frac{1}{T} \sum_{n=-\infty}^{\infty} e^{jnw_o t} \tag{A-7}$$

Take the Fourier transform of Equation (A-7)

$$F[\delta_T(t)] = \frac{1}{T} F \sum_{n=-\infty}^{\infty} e^{jnw_o t} \tag{A-8}$$

$$= \frac{1}{T} \sum_{n=-\infty}^{\infty} F[\cos nw_o t + j \sin nw_o t] \tag{A-9}$$

$$= \frac{1}{T} \sum_{n=-\infty}^{\infty} \{\pi[\delta(w-nw_o)+\delta(w+nw_o)]$$

$$-\pi[\delta(w+nw_o)-\delta(w-nw_o)]\} \tag{A-10}$$

$$= \frac{1}{T} \sum_{n=-\infty}^{\infty} [2\pi\delta(w-nw_o)] \tag{A-11}$$

$$= \frac{2\pi}{T} \sum_{n=-\infty}^{\infty} \delta(w-nw_o) \tag{A-12}$$

$$= w_o\, \delta_{w_o}(w) \tag{A-13}$$

where $F(\cos w_o t) = \pi[\delta(w-w_o) + \delta(w+w_o)]$ (Ref. 13) (A-14)

$F(\sin w_o t) = j\pi[\delta(w+w_o) - \delta(w-w_o)]$ (Ref. 13) (A-15)

$$\sum_{n=-\infty}^{\infty} \delta(w-nw_o) = \delta_{w_o}(w) \tag{A-16}$$

If a signal function f(t), which has no spectral components above f_m , is multiplied by a periodic impulse function $\delta_T(t)$ with regular intervals of T seconds. Then the sampled function is $f_s(t)$.

$$f_s(t) = f(t)\delta_T(t) \tag{A-17}$$

According to the frequency convolution theorem, Eq. (A-17) can be represented as Fourier transform $F_s(w)$.

$$F_s(w) = \mathcal{F}[f_s(t)]$$

$$= \mathcal{F}[f(t)\delta_T(t)]$$

$$= \frac{1}{2\pi}\{\mathcal{F}[f(t)] * \mathcal{F}[\delta_T(t)]\} \tag{A-18}$$

$$= \frac{1}{2\pi}[F(W) * w_o\delta_{w_o}(w)] \tag{A-19}$$

$$= \frac{1}{T}[F(w) * \delta_{w_o}(w)] \tag{A-20}$$

$$= \frac{1}{T}[F(w) * \sum_{n=-\infty}^{\infty} \delta(w-nw_o)] \tag{A-21}$$

$$= \frac{1}{T}\sum_{n=-\infty}^{\infty}[F(w) * \delta(w-nw_o)] \tag{A-22}$$

Use the convolution theorem with a unit impulse function which says

$$g(t) * \delta(t - T) = f(t - T) \tag{A-23}$$

Then the Equation (A-22) will be

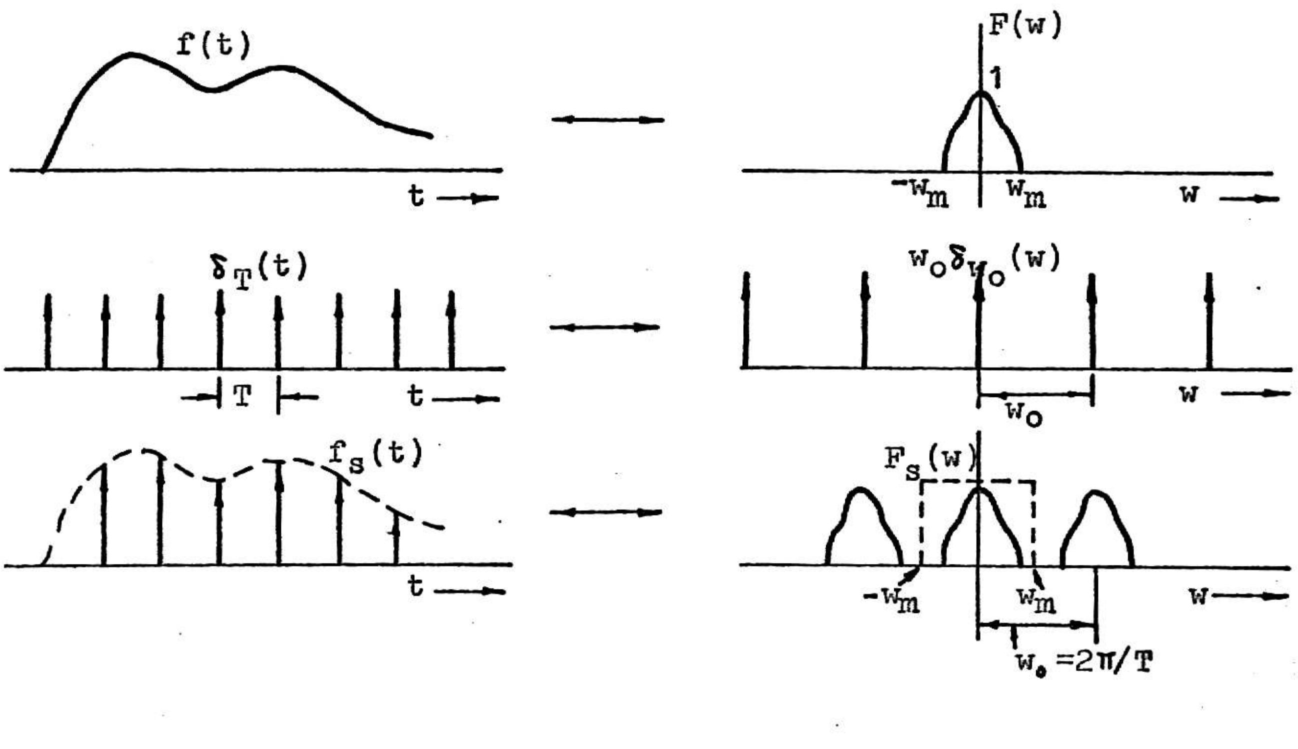
$$F_s(w) = \frac{1}{T} \sum_{n=-\infty}^{\infty} F(w-nw_o) \qquad \text{(A-24)}$$

The right-hand side of Equation (A-24) represents function F(w) repeating itself every w_o radian per second. From the figures[13] as shown below, it is evident that F(w) will repeat periodically without overlap as long as $w_o \geq 2w_m$, or

$$\frac{2\pi}{T} \geq 2(2\pi f_m)$$

That is $T \leq \frac{1}{2f_m}$. (A-25)

Therefore, as long as f(t) is sampled at regular intervals less than $1/2f_m$ seconds apart, $F_s(w)$ will be a periodic replica of F(w) and therefore contains all the information of f(t). The maximum interval of sampling $T = 1/2f_m$ is also called the <u>Nyquist interval</u>.



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PULSE CODE MODULATION OF DIGITAL COMMUNICATION SYSTEM OVER TWO-WIRE LINES

by

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B. S., Taiwan Provincial Cheng Kung University, 1965

AN ABSTRACT OF A MASTER'S REPORT

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Manhattan, Kansas

1969

This report presents a brief discussion of Pulse Code Modulation (PCM) and Touch-Tone calling. Also their application -- PCM systems and Tone-Dial Telephones -- will be discussed in the paper.

Section II gives a brief description of some fundamental concepts of PCM, and section III through section VI discusses PCM transmission systems. The analog speech signal is sampled thousands of times per second. An encoder at the transmission terminal converts these speech amplitude samples into code digits and a decoder at the receive terminal converts these digits back into speech samples. It requires time division multiplexing of speech signals to increase the amount of traffic carried on telephone cables. The function of repeaters between the terminals is to reconstruct the transmitted pulse train after it has traveled through a dispersive, noisy medium.

Section VII describes some basic ideas of Tone-Dial telephones and its applications. With touch-tone calling using voice frequency signaling, a call can be placed in less than half the time it takes with a conventional rotary dial.