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

by 500

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

A MASTER'S REPORT

submitted in partial fulfillment of the

requirements for the degree

MASTER OF SCIENCE

Department of Electrical Engineering

KANSAS STATE UNIVERSITY
Manhattan, Kansas

1969

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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