全国高等教育自学考试指定教材

计算机通信工程专业（独立本科）

通信英语

（附通信英语自学考试大纲）

全国高等教育自学考试指导委员会 组编

主编 张筱华

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《自学考试教材》出版前言

高等教育自学考试教材建设是高等教育自学考试工作的一项基本任务。经教育部同意，我们拟有计划、有步骤地组织编写一些高等教育自学考试教材，以满足社会自学和适应考试的需要。《通信英语》是为高等教育自学考试计算机通信工程专业（独立本科）组编的一套教材中的一种。这本教材是根据专业考试计划，从造就和选拔人才的需要出发，按照全国高等教育自学考试指导委员会颁布的《通信英语自学考试大纲》的要求，结合自学考试的特点，组织高等院校一些专家、学者集体编写而成的。

计算机通信工程专业（独立本科）《通信英语》自学考试教材，是供个人自学、社会助学和国家考试使用的。现经组织专家审定同意予以出版发行。我们相信，高等教育自学考试教材的陆续出版，必将对我国高等教育事业的发展，保证自学考试的质量起到积极的促进作用。

编写高等教育自学考试教材是一种新的尝试，我们希望得到社会各方面的关怀和支持，使它在使用中日臻完善。

全国高等教育自学考试指导委员会

1999年2月

编 者 的 话

为了适应我国通信信息产业飞速发展的新形势，提高工程技术人员的专业英语水平，我们编写了这本《通信英语》。

本书共 18 个单元。内容涉及计算机、数字通信、数据通信、光纤通信、数据交换、移动通信、局域网、多媒体技术、SDH、ATM、Internet、电信网和 ISDN 等方面，基本覆盖了当代通信的大部分新技术领域。

本书的课文主要选自美国高等院校的教科书和一些高级别的通信技术刊物。这些课文语言朴实，文字流畅，易于阅读和理解，相信它会受到读者的欢迎。

考虑到读者学习过公共英语，已具有一定的英语基础，所以本书的编写是以扩大通信技术的词汇量，熟悉专业术语，了解科技文章的表达特点和掌握英语翻译技巧为宗旨的。我们认为，只要读者能熟练地阅读和翻译本书的课文，则看一般的通信专业英语文章就不会再感到费力。

本书的前身《通信英语》(北京邮电大学出版社出版)自 1994 年问世以来，已多次再版，并深受工程技术人员的厚爱和好评，许多省、市和院校还将该书指定为工程技术人员和大学生的必读教材，作者在此向他们表示深切的谢意。根据我国信息产业的实际需要和发展，我们再次对原书作了较大的修改和调整，使它更贴近计算机通信工程专业。

本书由张筱华、石方文、王迎春编写，张筱华担任主编。上

海交通大学陈敏逊教授和北京邮电大学倪维桢教授以及汪琛副教授认真审阅了全书，作者向他们表示衷心的感谢。

张筱华　石方文　王迎春

于北京邮电大学函授学院

1999 年 2 月

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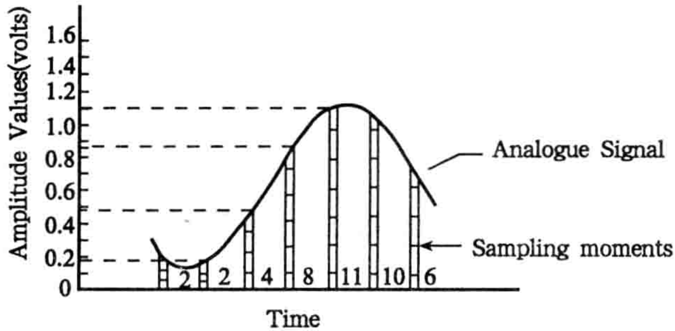
通信英语

UNIT 1

The Principle of PCM[1]

PCM is dependent on[2] three separate operations, sampling, quantizing, and coding. Many different schemes for performing[3] these three functions have evolved during recent years, and we shall describe the main ones. In these descriptions we shall see how a speech channel of telephone quality may be conveyed as a series of amplitude values, each value being represented[4], that is, coded, as a sequence of 8 binary digits[5]. Furthermore, we shall prove that a minimum theoretical sampling frequency of order 6.8 kilohertz (kHz) is required to convey a voice channel occupying[6] the range 300 Hz to 3.4 kHz. Practical equipments, however, normally use a sampling rate of 8 kHz, and if 8-digits per sample value[7] are used, the voice channel becomes represented by a stream of pulses with a repetition rate of 64 kHz. Fig. 1-1 illustrates the sampling, quantizing, and coding processes.

Reexamination of our simple example shows us that the speech signal of maximum frequency 3.4 kHz has been represented by a signal of frequency 64 kHz. However, if only 4-digits per sample value had been used, the quality of transmission would drop, and the repetition rate of the pulses would be reduced to



Amplitude value	Binary coded equivalent	Pulse code modulated signal
1	0000	
2	0001	
3	0010	
4	0011	
5	0100	
6	0101	
7	0110	
8	0111	
9	1000	
10	1001	
11	1010	
12	1011	
13	1100	
14	1101	
15	1110	
16	1111	

If the analogue signal shown above is "sampled ", and then "coded" using the table, the transmitted pulse code modulated signal becomes:

Decimal values: 2 , 2 , 4 , 8 , 11 , 10 , 6

Binary values: 0001, 0001, 0011, 0111, 1010, 1001, 0101

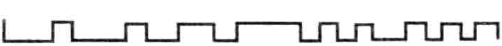
PCM Signal: 

Fig. 1-1 The Sampling and Coding Processes, and the Resultant PCM Signal

32 kHz. Thus the quality of transmission is dependent on the pulse repetition rate, and for digital communication systems these two variables may be interchanged most efficiently[8].

Digital transmission provides a powerful method for overcoming noisy environments. Noise can be introduced into a transmission path in many different ways; perhaps via a nearby lightning strike, the sparking of a car ignition system, or the thermal low-level noise within the communication equipment itself[9]. It is the relationship of the true signal to the noise signal, known as the signal-to-noise ratio[10], which is of most interest to the communication engineer. Basically, if the signal is very large compared to the noise level[11], then a perfect message can take place; however, this is not always the case. For example, the signal received from a satellite, located in far outer space[12], is very weak and is at a level only slightly above that of the noise. Alternative examples may be found within terrestrial systems where, although the message signal is strong, so is the noise power.

If we consider binary transmission, the complete information about a particular message will always be obtained by simply detecting the presence or absence of the pulse. By comparison, most other forms of transmission systems convey the message information using the shape, or level of the transmitted signal; parameters that are most easily affected by the noise and attenuation introduced by the transmission path[13]. Consequently there is an inherent advantage for overcoming noisy environments by choosing digital transmission.

So far in this discussion we have assumed that each voice channel has a separate coder, the unit that converts sampled amplitude values to a set of pulses; and decoder, the unit that performs the reverse operation. This need not be so, and systems are in operation where a single codec (i. e. , coder, and its associated decoder) is shared between 24, 30, or even 120 separate channels. A high-speed electronic switch is used to present the analog information signal of each channel, taken in turn[14], to the codec. The codec is then arranged to sequentially sample the amplitude value, and code this value into the 8-digit sequence. Thus the output to the codec may be seen as a sequence of 8 pulses relating to channel 1, then channel 2, and so on. This unit is called a time division multiplexer (TDM), and is illustrated in Fig. 1-2. The multiplexing principle that is used is known as word interleaving. Since the words, or 8-digit sequences, are interleaved in time.

At the receive terminal a demultiplexer is arranged to separate the 8-digit sequences into the appropriate channels. The reader may ask, how does the demultiplexer know which group of 8-digits relates to channel 1, 2, and so on? Clearly this is important! The problem is easily overcome by specifying a frame format, where at the start of each frame a unique sequence of pulses called the frame code, or synchronization word, is placed so as to[15] identify the start of the frame. A circuit of the demultiplexer is arranged to detect the synchronization word, and thereby it knows that the next group of 8-digits corresponds to channel 1. The synchronization word reoccurs once again after the last channel has been received[16].

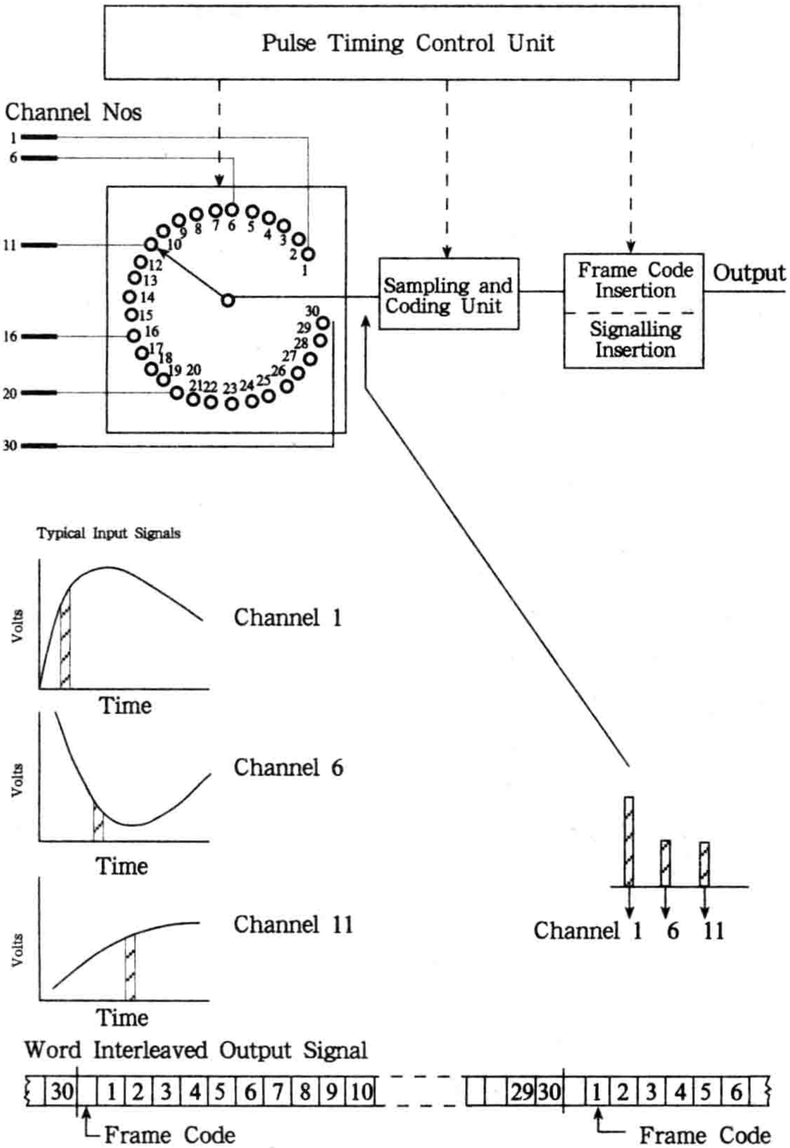


Fig.1–2 The Function of the Time Division Multiplexer (TDM)

NEW WORDS AND PHRASES

principle ['prinsipl] *n.* 原理

be dependent on 依赖，取决于

sample ['sæmpl] *vt.* 采样；*n.* 样值

quantize ['kwɔntaiz] *vt.* 量化，分层

code [koud] *vt.* 编码；*n.* 码

scheme [ski:m] *n.* 方案，设计，安排

describe [dis'kraib] *vt.* 叙述，描述

description [dis'kripʃən] *n.* 叙述，描述

amplitude ['æmplitju:d] *n.* 幅，幅度

binary ['bainəri] *a.* 二进制的

minimum ['minimәm] *n.* 最小值，最小量

theoretical [θiə'retikəl] *a.* 理论上的

repetition [ˌrepi'tiʃən] *n.* 重复，反复

reexamination ['ri:igˌzæmi'neiʃən] *n.* 再审查，重考

maximum ['mæksiməm] *n.* 最大值

reduce [ri:'dju:s] *vt.* *vi.* 减少，缩小

interchange [ˌintə'tʃeindʒ] *v.* 互换，转换

method ['meθəd] *n.* 方式，方法，手段

overcome [ouvə'kʌm] *vt.* 克服，打败，征服

environment [in'vaiərənmənt] *n.* 环境，周围情况

lightning ['laitniŋ] *n.* 电光，闪电，雷电

strike [straik] *v.* 击，敲，打

spark [spɑ:k] *n.* 发火花，打火，闪光

ignition [ig'niʃən] *n.* 点火，点火装置

signal-to-noise ratio 信噪比

satellite ['sætəlait] *n.* 卫星

terrestrial [ti'restriəl] *a.* 地球的，地面的，大地的