ENGLISH

通信英语

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孙 凡 李鄂强 / 主编

湖南科学技术出版社

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内容简介

本书是高职高专院校通信类专业英语教材。全书包括15篇精读课文、阅读材料和丰富多样的练习题。内容涉及交换、计算机、网络等相关通信技术领域。选材于各类现代国内外通信出版物和互联网。全书以现代通信技术和网络技术为重点，根据在校高职高专学生学习的特点编写而成。

本书适合各类高职高专学生、通信工程技术人员和各类英语爱好者使用。

本书在编著过程中承蒙长沙通信职业技术学院周训斌老师、胡远萍老师提出很多宝贵建议，梅勇老师提供了部分语法材料、并对全书的最终内容审定提出了中肯的修改意见，在此谨表示衷心感谢。

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前 言

为了适应我国通信信息产业飞速发展的需要，提高通信行业高级技术人员的专业英语水平，我们编写了这本《通信英语》。本教材从高级应用型人才培养的总体目标出发，结合学生毕业后的工作实际，力求向学生提供未来工作岗位所需的专业英语知识和技能，培养学生英语翻译能力。

《通信英语》系专门用途英语教材中的一本，旨在提高通信工程专业和计算机通信专业的学生和通信行业人员在通信领域的资料阅读和翻译能力。

本书包括15篇精选课文、阅读材料和丰富多样的练习题。内容涉及交换、计算机、网络等相关方面，基本覆盖了当代IT业的所有新技术领域。例如，第一单元：脉冲表编码调制(Pulse Code Modulation)就是有关交换技术的内容。课文共分五个部分：课文、生词、练习、语法和补充阅读。补充阅读部分的内容是对主课文的拓展和延伸。语法部分的重点从以往的传统语法转移到了阐述电信英语的特点上来。并对电信英语中的名词化结构、被动语态、长句等进行了阐述并举例说明。第二单元：介绍MS-6117主板(Introduction to MS-6117 Mainboard)涉及的是计算机的内容。书中介绍了主板的结构及功能。本书课文主要选自当代国外通信出版物和互联网络。这些课文语言朴实，文字流畅，易于阅读和理解，有助于扩大读者的视野，提高阅读能力。因此，我们相信它会受到读者的欢迎。

考虑到读者学习过基础英语，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英语词汇量、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。全书以现代通信技术和网络技术为重点，根据在校学生的特点而编写。本教材实用性强，尤其突出了通信技术的实际需要，选材新颖，练习的设计兼具实用性和针对性。为了便于教学，各单元每一部分均注有生词和短语，书末还配有总词汇表。

衷心希望本书的出版能为我国IT业的发展，为通信信息产业职工整体素质的提高，为我国高等教育的发展作出贡献。

长沙通信职业技术学院的领导和教务处以及通信管理系对本教材的编写给予了大力支持，还有周训斌、梅勇等老师对本书提出了一些宝贵意见，在此一并表示感谢。

编者

2007 年 7 月

目　录

Unit One

Text

Pulse Code Modulation

PCM is much more than a technique—it is a system. Yes, it includes a coder and, later, a decoder. However, to truly understand the role that these functions play, you need to understand the complete PCM system. To start, look at Figure 1.1

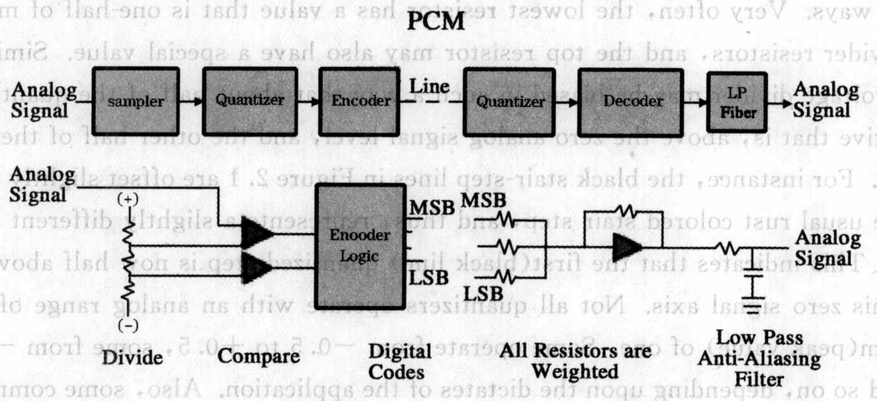

Figure 1. 1 The Process of PCM

Figure 1. 1 The PCM process is shown in block-diagram form and also in a simplistic schematic form. By a somewhat convoluted process, an analog signal, such as audio or video, is first sampled; then that sample is tagged(compared) at a certain level; and then that level is encoded as a unique series of digital bits. These successive digital bit-codes can then be transmitted down a medium, such as a transmission line or, more often, processed in the digital domain. The transmitted or processed digital signals can then again be quantized, decoded, and sent through an(analog) low-pass, anti-aliasing filter to again emerge as an analog signal. The circuits shown are for illustration and are not necessarily the actual configurations that would be used. Although it is not shown, it is assumed that the appropriate functions in the above figure are controlled by clocking signals.

When the words sampling and clocking are used in the description of a process, they are glowing clues that this process is, indeed, in the digital domain. To illustrate the PCM process, we will start with an analog signal and, step by step, show the role of the functions shown in Figure 1. 1. Although one of the features of a PCM system is to accommodate almost any input analog waveshape, for clarity, a sawtooth waveshape will be used as the test analog input. The role of the quantizer is usually to divide the amplitude of the input signal into a series of evenly-spaced, discrete steps. At each tick of the system clock, the quantizer looks at a small portion of the input signal and decides which one of several predetermined levels that portion of the signal represents. These predetermined levels are established by the design of the quantizer. A 1-bit quantizer would only have two levels, which would be coded as either a 0 or a 1. A 2-bit quantizer would have four possible levels; a 3-bit quantizer would have eight possible levels; and the common 8-bit quantizer would produce 256 levels. All of these level counts must include zero. Figure 1. 2 [b] show one way to quantize a ramp signal with a 1-bit quantizer.

Note: Giving examples of PCM systems is difficult since there are so many different approaches. For instance, there are several different approaches to the overall A/D process including the design of the quantizer. Various texts will show graphics representing the quantizing process with slightly different details. This is because a quantizer, which is often simply a series of resistors connected as a tapped DC voltage divider, can be configured in many ways. Very often, the lowest resistor has a value that is one-half of most of the other divider resistors, and the top resistor may also have a special value. Similarly, the tapped voltage divider may be biased in such a way that about half of the quantized levels are positive that is, above the zero analog signal level, and the other half of the levels are negative. For instance, the black stair-step lines in Figure 2. 1 are offset slightly downward from the usual rust colored stair step, and thus, represents a slightly different quantizing process. This indicates that the first(black line) quantized step is now half above and half below this zero signal axis. Not all quantizers operate with an analog range of zero to a maximum(peak value) of one. Some operate from -0.5 to $+0.5$, some from -1.0 to $+1.0$, and so on, depending upon the dictates of the application. Also, some communication industries, such as telephone companies, use resistor dividers with an uneven(logarithmic) value distribution for their resistor networks. This special log distribution is used for signal compression.

Figure 1. 2 (a) This is the template that will be used to show how an analog ramp waveshape can be quantized. In each illustration, the amplitude of the analog wave is given as the abscissa and the corresponding digital codes are shown as the ordinate.

Figure 1. 2 (b) This shows the action of a 1-bit (21) quantizer. The rust-colored stair step and the "{" indicate that any analog value from 0 to 0. 5 is quantized as a digital 0. Likewise, all analog values from 0. 5 to 1. 0 are quantized as a digital 1.

Figure 1. 2 (c) Here the rust-colored stair step indicates the action of a 2-bit quantizer that produces four coded levels from an analog ramp.

Figure 1. 2 (d) This shows the action of a 3-bit quantizer where the one volt peak analog ramp is converted into 8 digital codes.

Information Contributed By: Bob Libbey, Retired RCA Engineer and Adjunct Professor, New Jersey Institute of Technology

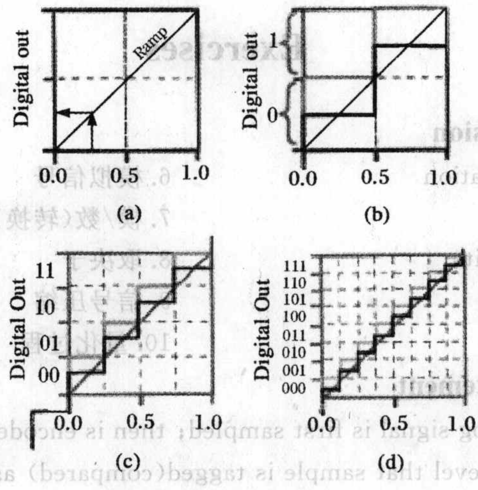

Figure 1.2 Analog in

New Words

access /ækses/ *n.* & *v.* 接入，访问，通路，存取
circuit /'sə:kɪt/ *n.* 电路
compression /kəmpreʃən/ *n.* 压缩
DC=Direct Current *n.* 直流电
system /sɪstəm/ *n.* 系统
PCM=Pulse Code Modulation *n.* 脉冲编码调制
transmission /trænzmɪʃən/ *n.* 传输
wireless /waɪələs/ *n.* 无线的
low-loss *n.* 低损的
sample /sɑ:mpl/ *v.* 采样
clock/klɔk/ *n.* 时钟
simplistic /sɪm'plɪstɪk/ *adj.* 简单的
schematic /skɪmætɪk/ *adj.* 图表的
convoluted /kɒnvəlu:tɪd/ *adj.* 回旋的
low-pass *adj.* 低通的
anti-aliasing filter /'æntɪ-'eɪlɪəsɪŋ'fɪltə/ *n.* 消侧音电话
configuration /kənˌfɪgjureɪʃən/ *n.* 配置
abscissa /æb'sɪsə/ *n.* 横座标
ordinate /'ɔ:dɪneɪt/ *n.* 纵座标
distribution /ˌdɪstrɪ'bju:ʃən/ *n.* 分布
domain /də'meɪn/ *n.* 域
digital /'dɪdʒɪtl/ *adj.* 数字的
accommodate /ə'kɒməˌdeɪt/ *v.* 调节

Exercises

Ⅰ. Phrase expression

1. pulse code modulation
2. block-diagram
3. series of digital bits
4. anti-aliasing filter
5. sawtooth
6. 模拟信号
7. 模/数(转换)过程
8. 取决于
9. 信号压缩
10. 量化过程

Ⅱ. True/False statement

1. In PCM, an analog signal is first sampled; then is encoded as a unique series of digital bits, and then that level that sample is tagged(compared) as being at a certain level. ( )

2. When the words sampling and clocking are used in the description of a process, this process is in the digital domain. ( )

3. There are several different approaches to the overall A/D process including the design of the quantizer. ( )

4. Some communication industries, such as telephone companies, use resistor dividers with an uneven(logarithmic) value distribution for their resistor networks. This special log distribution is used for signal coding. ( )

5. Very often, the lowest resistor has a value that is one-half of most of the other divider resistors, and the top resistor may also have a special value. ( )

Ⅲ. Cloze

Each input sample is (1) a quantization (2) that is closest to its amplitude height. If an input sample is not assigned a (3) interval that matches its actual height, then an error is (4) into the PCM process. This error is (5) quantization noise. Quantization noise is equivalent to the random noise that impacts the signal-to-noise ratio (SNR) of a voice signal. SNR is (6) in decibels (dB). The (7) the SNR, the better the voice quality. Of course, quantization noise (8) the SNR of a signal. Therefore, an increase in quantization noise degrades the (9) of a voice signal. Figure 3 shows how quantization noise is (10). For coding purpose, an N bit word will yield 2N quantization labels.

( )1. A. assigned	B. assigns	C. assigning	D. to be assigned
( )2. A. level	B. invert	C. interva	D. involve
( )3. A. quantizating	B. quantization	C. quantity	D. quality
( )4. A. to be introduced	B. introduced	C. being introduced	D. introducing
( )5. A. calls	B. calling	C. being called	D. called
( )6. A. measured	B. measure	C. counted	D. count
( )7. A. high	B. higher	C. highest	D. most highest
( )8. A. reduction	B. reduces	C. increase	D. increased
( )9. A. quality	B. quantity	C. number	D. amount

( )10. A. regenerated B. generated C. generator D. regenerator

Ⅳ. Translation

The word TELECOMMUNICATION is a combination of two words TELE+COMMUNICATION.

The word TELE in Latin means distance. Hence telecommunication is distance communication. The necessity of communication began as early as the existence of mankind on this earth. Communication has become the vital tool for mankind to strife prosperously in this world.

Short for pulse code modulation, a sampling technique for digitizing analog signals, especially audio signals. PCM samples the signal 8 000 times a second; each sample is represented by 8 bits for a total of 64 kbps. There are two standards for coding the sample level. The Mu-Law standard is used in North America and Japan while the A-Law standard is use in most other countries.

PCM is used with T-1 and T-3 carrier systems. These carrier systems combine the PCM signals from many lines and transmit them over a single cable or other medium.

Communication networks employ a variety of transmission media ranging from copper wires to satellite channels to transport user's information. The transmission media is the physical path for the communication signal. Transmission media can be classified into two major categories: guided media, which may constrain and guide the communication signal, and unguided media, which permits signal to be transmitted but not guide them. Examples of guided transmission media are metallic cable and optical fibers. Examples of unguided transmission media are the radio signals and satellite signals. An important characteristic of these different media is the bandwidth or simply the range of frequencies each can transmit. In general, the greater the bandwidth of a given media, the more it can carry.

After filtering and sampling(using PAM) an input analog voice signal, the next step is to digitize these samples in preparation for transmission over a telephony network. The process of digitizing analog voice signals is called PCM. The only difference between PAM and PCM is that PCM takes the process one step further by encoding each analog sample using binary code words. Basically, PCM has an analog-to-digital converter on the source side and a digital-to-analog converter on the destination side. How does PCM encode these samples? PCM uses a technique called quantization.

Grammar

电信英语特点概述

叙述电信技术的英语是科技英语的一个分支,其语言特性基本符合科技英语的普遍特征,本节将介绍科技英语的一些特点,以便读者在以后的章节学习过程中逐步体会,加深理解,从而更好把握科技英语的语言规律,为阅读专业英语打好基础。

可以说，科技英语只是一个科学技术的一个载体，它是为科技内容服务的，因此，科技文体的特点便决定了科技英语的特色。科技文章通常要求简洁明了、客观严密、逻辑性强、信息量大，相应地，科技英语便具有这样一些特点：

* 大量使用名词化结构
* 大量使用被动语态
* 常采用长句
* 频繁使用缩略词和复合词
* 使用表示逻辑关系的词

1. 名词化结构

科技英语中，经常使用名词化结构代替动词性结构、从句等，使得句法更简单、文章更简练。试比较下列几组句子：

例 1 The first step in converting the signal from analog to digital is to filter out the higher frequency component of the signal, mainly because it's going to make things easier downstream for converting this signal.

例 2 The second step in converting an analog voice signal to a digital voice signal is to sample the filtered input signal at a constant sampling frequency.

例 3 The analog voice signal could be sampled at a million times per second or at two to three times per second.

2. 被动语态

科技文章中主要描述客观事实，强调事物的客观性，因此，在科技英语中，通常避免使用"我们"、"厂家"等主观感情色彩较浓的字眼，而采用被动语态，突出施动对象。

例 1 PCM is a type of coding that is called "waveform" coding because it creates a coded form of the original voice waveform.

例 2 This band-limiting filter is used to prevent aliasing(anti-aliasing).

3. 长句

科技文章信息量大，要表达的内容往往比较复杂，因此常常出现含多个从句或并列句的长句，用来表达复杂事体的逻辑关系。

例 1 Over time, it has become obvious that digital coding is more immune to noise corruption on long-distance connections, and the world's communications systems have converted to a digital transmission format called pulse code modulation(PCM).

例 2 The low-pass output filter, used to reconstruct the original input signal, is not smart enough to detect this overlap, so it creates a new signal that did not originate from the source.

例 3 Accomplished by using a process called pulse amplitude modulation (PAM), this step uses the original analog signal to modulate the amplitude of a pulse train that has a constant amplitude and frequency.

4. 缩略词和复合词

从词法上看，科技文章使用缩略词和复合词的频率很高，有时是为了方便，如将"Stored Program Control"缩写为"SPC"，将"Telecommunication"缩略为"Telecom"等；有时则是适应新技术、新产品的出现，在原有词汇的基础上较形象地构造新词，如 read-me(说明程序)，

terminal-to-terminal(端对端),bandwidth(带宽),open-loop(开环)等。

5. 表示逻辑关系的词

科技英语力求结构严谨、层次分明,因此常使用表示逻辑关系的词来叙述一个复杂的过程。常见的词有:consequently, then therefore, hence, as a result, thus, in addition, furthermore, moreover, besides, however, nevertheless, on the other hand, meanwhile, in the meantime, at the same time, finally, at first, at last, in conclusion, in fact, in other words, in order to, generally speaking, in general, as a whole, 等等。

以上列举了科技英语较具普遍性的五个特征。当然,我们无法十分准确地描述科技英语,而只能勾画其轮廓,抓住其要点。况且,对于不同的专业、不同的领域,科技英语亦是千姿百态、大相径诞的。但我们相信,只要掌握了科技英语的共性,并在专业英语的过程中进一步深化、具体化,是不难学好各种专业英语的。

Supplementary Reading

(1)

Companding refers to the process of first compressing an analog signal at the source, and then expanding this signal back to its original size when it reaches its destination. The term companding was created by combining the two terms, compressing and expanding, into one word. During the companding process, input analog signal samples are compressed into logarithmic segments and then each segment is quantized and coded using uniform quantization. The compression process is logarithmic, where the compression increases as the sample signals increase. In other words, the larger sample signals are compressed more than the smaller sample signals, causing the quantization noise to increase as the sample signal increases. A logarithmic increase in quantization noise throughout the dynamic range of an input sample signal will keep the SNR constant throughout this dynamic range. The ITU-T standards for companding are called A-law and μ-law.

A-law and μ-law Companding

A-law standard is used by European countries and μ-law is used by North America and Japan.

Similarities Between A-law and μ-law

Both are linear approximations of logarithmic input/output relationship.

Both are implemented using 8-bit code words (256 levels, one for each quantization interval). Eight-bit code words allow for a bit rate of 64 kilobits per second(kbps), calculated by multiplying the sampling rate(twice the input frequency) by the size of the code word (2×4kHz×8bits=64kbps).

Both break a dynamic range into a total of 16 segments:

8 positive and 8 negative segments.

Each segment is twice the length of the preceding one.

Uniform quantization is used within each segment.

Both use a similar approach to coding the 8-bit word:

First(MSB) identifies polarity.

Bits 2, 3, and 4 identify segment.

Final 4 bits quantize the segment are the lower signal levels than A-law.

Differences Between A-law and μ-law.

Different linear approximations lead to different lengths and slopes.

The numerical assignment of the bit positions in the 8-bit code word to segments and the quantization levels within segments are different.

A-law provides a greater dynamic range than μ-law.

μ-law provides better signal/distortion performance for low level signals than A-law.

A-law requires 13 bits for a uniform PCM equivalent. μ-law requires 14 bits for a uniform PCM equivalent.

An international connection should use A-law, μ to A conversion is the responsibility of the μ-law country.

(2)

Quantization is the process of converting each analog sample value into a discrete value that can be assigned a unique digital code word.

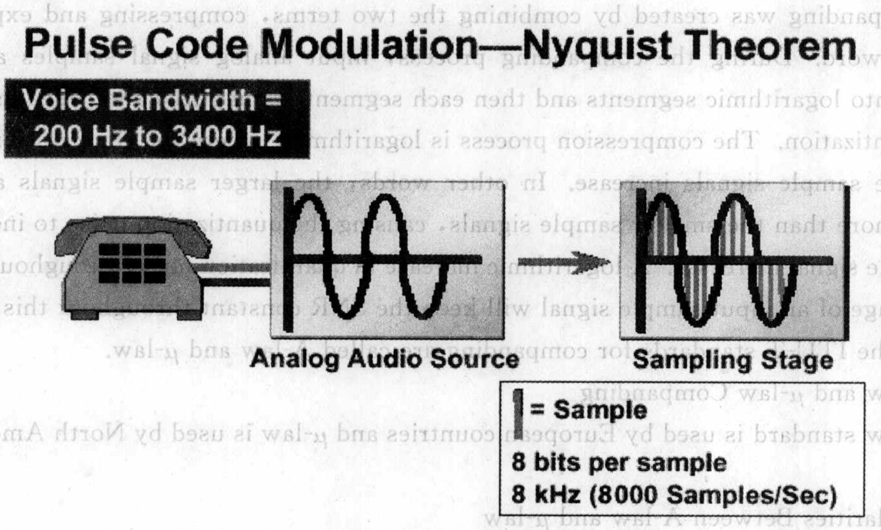

Figure 1. 3 Code Technique

As the input signal samples enter the quantization phase, they are assigned to a quantization interval. All quantization intervals are equally spaced (uniform quantization) throughout the dynamic range of the input analog signal. Each quantization interval is assigned a discrete value in the form of a binary code word. The standard word size used is 8 bits. If an input analog signal is sampled 8000 times per second and each sample is given a code word that is 8 bits long, then the maximum transmission bit rate for telephony systems using PCM will be 64000 bits per second. Figure 1. 3 illustrates how bit rate is derived for a PCM system.

Each input sample is assigned a quantization interval that is closest to its amplitude height. If an input sample is not assigned a quantization interval that matches its actual height, then an error is introduced into the PCM process. This error is called quantization noise. Quantization noise is equivalent to the random noise that impacts the signal-to-noise ratio (SNR) of a voice signal. SNR is measured in decibels(dB). The higher the SNR, the better the voice quality. Of course, quantization noise reduces the SNR of a signal. Therefore, an increase in quantization noise degrades the quality of a voice signal. Figure 1.4 shows how quantization noise is generated. For coding purpose, an N bit word will yield 2N quantization labels.

One way to reduce quantization noise is to increase the amount of quantization intervals. The difference between the input signal amplitude height and the quantization interval decreases as the quantization intervals are increased(increases in the intervals decrease the quantization noise). However, the amount of code words would also have to be increased in proportion to the increase in quantization intervals. This process would introduce additional problems dealing with the capacity of a PCM system to handle more code words.

SNR (including quantization noise) is the single most important factor affecting voice quality in uniform quantization. As stated earlier, uniform quantization uses equal quantization levels throughout the entire dynamic range of an input analog signal. Thus low signals will have a small SNR (low-signal-level voice quality) and high signals will have a large SNR (high-signal-level voice quality). Considering that most voice signals generated are of the low kind, having better voice quality at higher signal levels is a very inefficient way of digitizing voice signals. To improve voice quality at lower signal levels, uniform quantization(uniform PCM) was replaced by a nonuniform quantization process called companding.

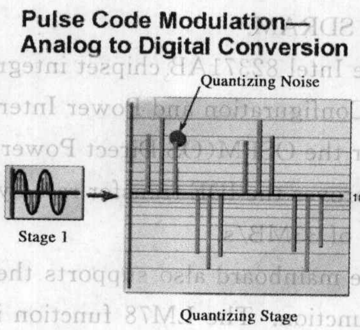

Figure 1.4 Quantizing Stage