通信英语

TONGXIN YINGYU

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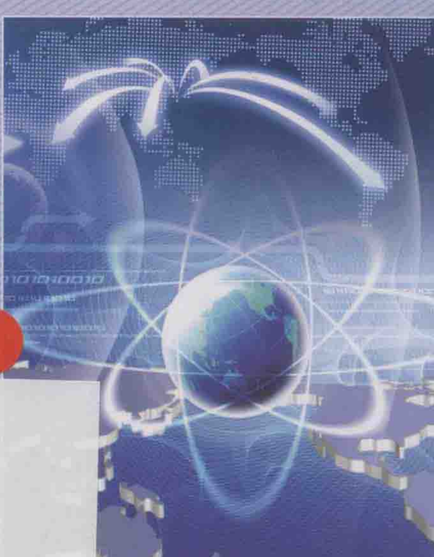

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通 信 英 语

孙　凡　主　编
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内容简介

本书由13篇精选课文、阅读材料和丰富多样的练习题组成。内容涉及物联网、云计算、通信网络等方面，基本覆盖了当代通信业的新技术领域。

本书所选课文语言朴实、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。

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目标，全书以现代通信技术和网络技术为重点。

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前言 (Preface)

为了适应我国通信信息产业飞速发展的需要，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，力求为学生提供未来工作岗位所需的专业英语知识和技能，培养学生的英语翻译能力。

本书由 13 篇精选课文、阅读材料和丰富多样的练习题组成。内容涉及物联网、云计算、通信网络等方面，基本覆盖了当代通信业的新技术领域。

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衷心希望本书的出版能为我国通信业的发展、为通信信息产业员工整体素质的提高、为我国通信专业教育的发展做出贡献。

编　者

目录

(Contents)

Unit 1 PCM ······ 1

Unit 2 Systems Analysis and Design ······ 12

Unit 3 Frame Formats ······ 18

Unit 4 Optical Fibre Communication History ······ 25

Unit 5 SDH Transmission ······ 35

Unit 6 EWSD and the Next Generation Network ······ 45

Unit 7 Data Transmission ······ 51

Unit 8 W-CDMA Mobile Communication Systems ······ 58

Unit 9 4G ······ 65

Unit 10 Internet TV ······ 71

Unit 11 The Internet of Things: In Action ······ 81

Unit 12 How Cloud Computing Works ······ 91

Unit 13 Passive Optical Network ······ 101

参考译文与练习答案 ······ 110

参考文献 ······ 138

Unit 1

PCM

Text

PCM is much more than a technique, it is a system. It includes a coder and, later, a decoder. However, to fully understand the role that these functions play, you need to understand the complete PCM system. To start, look at Fig. 1 – 1.

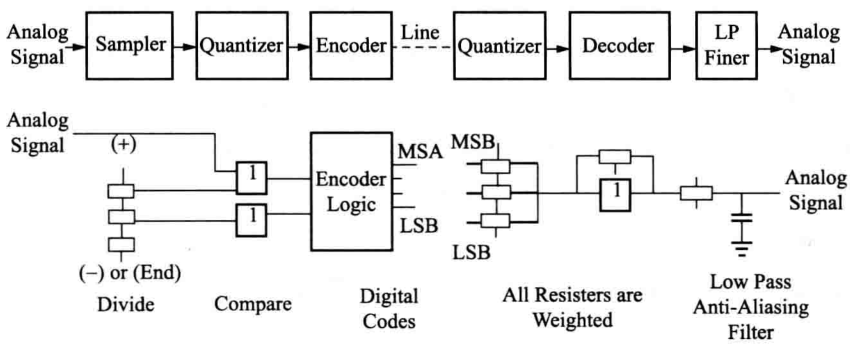

Fig. 1 – 1 PCM

The PCM process is shown in a 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) as being 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 not shown, it is assumed that the appropriate functions in the above figure are controlled by clocking signals.

PCM Sampling, Clocking, and Quantizing

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, present the role of the functions shown in Fig. 1 – 1. Although one of the features of a PCM system is to accommodate almost any input analog waveshape, for clarity, a sawtooth (also termed a ramp) 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, desecrate steps. At each tick of the system clock, the quantizer looks at (samples) 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 (2^1) quantizer would only have two levels, which would be coded as either a 0 or a 1. A 2-bit (2^2) quantizer would have four possible levels; a 3-bit (2^3) quantizer would have eight possible levels; and the common 8-bit (2^8) quantizer would produce 256 levels. All of these level counts must include zero. Fig. 1 – 2(b) shows 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 Fig. 1 – 2 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.

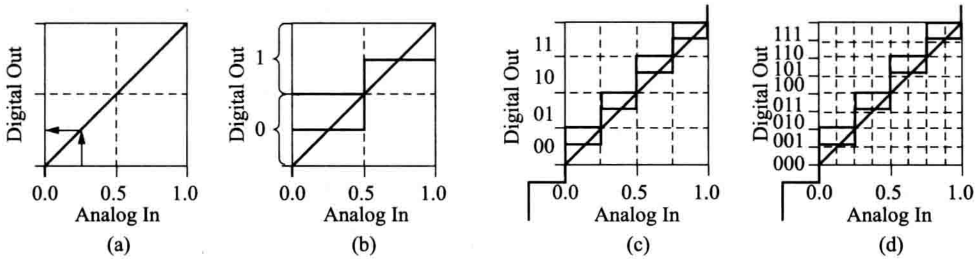

Fig. 1 – 2 Quantize a ramp signal with a 1 – bit quantizer

Fig. 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 (follow the arrows) are shown as the ordinate.

Fig. 1-2(b): It shows the action of a 1-bit (2^1) 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. See the text above for an explanation of the thin, black stair-step line.

Fig. 1-2(c): Here the rust-colored stair step indicates the action of a 2-bit (2^2) quantizer that produces four coded levels from an analog ramp. See the text above for an explanation of the thin, black stair-step line.

Fig. 1-2(d): This shows the action of a 3-bit (2^3) quantizer where the one volt peak analog ramp is converted into (2^3) or 8 digital codes. See the text above for an explanation of the thin, black stair-step line.

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

New Words

access [ˈækses] *n.* & *v.* 接入，通路，存取，访问

ADSL *abbr.* = Asymmetric Digital Subscriber Line 非同步数字用户专线

Alexander Graham Bell 亚历山大·格拉汉姆·贝尔

alpha-numeric *adj.* 字母数字式的

ARPANET *abbr.* = Advanced Research Project Agency Network 阿帕网（美国官方的电脑网络，为 Internet 的前身）

ASCII *abbr.* = American Standard Code for

		Information Interchange 美国信息交换标准码
AT&T		美国电报电话公司
attenuation	[əˌtenjuˈeiʃn]	*n.* 衰减，衰耗，损失
backbone	[ˈbækbəun]	*n.* 骨干；基础
Baltimore	[ˈbɔːltimɔː]	*n.* 巴尔的摩港市（美国马里兰州）
Bell Core		贝尔核心；贝尔中心
benchmark	[ˈbentʃmaːk]	*n.* 水平点；基准，标准检查程序
binary	[ˈbainəri]	*n.* 二进制数
Bluetooth	[ˈbluːtuːθ]	*n.* 蓝牙，一种无线连接技术标准
bps		*abbr.* =bits per second 位/秒，每秒传送位数，=bytes per second 字节/秒
CCITT		*abbr.* =Consultative Committee for International Telegraph and Telephone 国际电报电话咨询委员会
cell	[sel]	*n.* 蜂窝，小巢
cellular	[ˈseljulə]	*adj.* 蜂窝的；多孔的，细胞组成的
channel	[ˈtʃænl]	*n.* 信道，通路
character	[ˈkærəktə]	*n.* 品质；性格；特性；字符
circuit	[ˈsɜːkit]	*n.* 电路，回路
circuit-switched		*adj.* 电路交换
Claude Shannon		克劳德·香农
coax cables		*n.* 同轴电缆
compression	[kəmˈpreʃn]	*n.* 压缩，压挤
connectivity	[ˌkənekˈtiviti]	*n.* （网络）连通性，连接
constellation	[ˌkɑnstəˈleiʃən]	*n.* 星座；星座区域
crossbar	[ˈkrɔsbɑː]	*n.* 闩；横木，纵横制
data network		*n.* 数据网络
Data Center		*abbr.* 资料中心，数据中心
Data Communication		*n.* 数字通信
digitized	[ˈdidʒitaizd]	*adj.* 数字化的
discrete	[disˈkriːt]	*adj.* 分离的；不连接的
Early Bird	[ˈəːlibəːd]	*n.* 早鸟号（卫星名）
Ericsson	[ˈeriksən]	*n.* 爱立信（公司名）
Ethernet	[ˈiːθənet]	*n.* 以太网

exchange	[iks'tʃeindʒ]	*n.* & *v.* 交换，交换机，互换
exploit	[ik'splɔit]	*n.* 功绩，功勋；辉煌的成就
FCC		*abbr.* = Federal Communications Commission（美国）联邦通信委员会
FDM		*abbr.* = Frequency Division Multiplexing 频分复用
fiber	['fɑibə]	*n.* = fibre 光纤，纤维
FSK		*abbr.* = Frequency – Shift Keying 数字信号调频
geostationary	[dʒiːəu'steiʃənəri]	*adj.*（人造卫星）与地球旋转同步的
Guillermo Marconi		吉列尔莫·马可尼
handheld		*adj.* 手持，手持式的
Heinrich Hertz		海因里希·赫兹
IBM		*abbr.* = International Business Machine 国际商务机器公司
IEEE		*abbr.* = Institute of Electrical and Electronic Engineers 电气和电子工程师协会
install	[in'stɔːl]	*vt.* 安装，设置
Intel	['intel]	*n.*（美国）英特尔公司（全球知名的半导体生产厂商）
Internet	['intənet]	*n.* 国际互联网
introduce	[ˌintrə'djuːs]	*v.* 引入，介绍
IPSec		*abbr.* = Internet Protocol Security 网际协议安全
Iridium	[i'ridiəm]	*n.* 铱，铱星计划（大型的低轨道卫星通信计划）
ISDN		*abbr.* = Integrated Services Digital Network 整合服务数字网络
ITU-T		*abbr.* = International Telecommunication Union-Teleco mmunication Sector 电信标准局
L1-carrier		*n.* L1 载波
L2TP		*n.* 第二层隧道协议
LAN		*n.* = Local Area Network 局域网
low-loss		*adj.* 低损耗

Mbps		*abbr.* = million bits per second 兆字节/秒
mobile	[ˈməubail]	*adj.* 移动的
modem	[ˈməudem]	*abbr.* = modulator-demodulator 调制解调器
Motorola		*n.* 摩托罗拉（公司名）
multitone	[mʌltitəun]	*n.* & *adj.* 多音频，多音频的
Nokia		*n.* 诺基亚（公司名）
packet	[ˈpækit]	*n.* 包，数据包，分组
paging	[ˈpeidʒiŋ]	*n.* & *v.* 寻呼
PCM		*abbr.* = Pulse Code Modulation 脉冲编码调制，脉码调制
PCS		*abbr.* = Program Counter Storage 程序计算内存 = punched-card system 打孔卡片系统
POTS		*abbr.* = Plain Old Telephone Service 普通老式的电话服务
protocol	[ˈprəutəkɔl]	*n.* 协议，协约，规约
push-to-talk		*n.* 一按通
QAM		*abbr.* = Queued Access Method 队列存取方法
quadrature	[ˈkwɔdrətʃə]	*n.* 求面积
radio	[ˈreidiəu]	*n.* 无线电；收音机
radio wave	[ˈreidiəu weiv]	*n.* 无线电波
reliable	[rilaiəbl]	*adj.* 可信赖的；可靠的；确实的
ringer	[ˈriŋə]	*n.* 振铃，振铃电路
Robert Metcalf		*n.* 罗伯特·梅特卡夫
Samuel Morse Patent		*n.* 塞缪尔·摩斯专利
satellite	[ˈsætəlait]	*n.* 人造卫星
simultaneously	[ˌsiməlˈteiniəsli]	*adv.* 同时地
sonnet		十四行；十四行诗
spectrum	[ˈspektrəm]	*n.* 频谱；光谱，谱线
Sprint Corp		*n.* 斯普林公司
St. Louis		*n.* 圣路易斯
standard	[ˈstændəd]	*n.* 标准，规范

stationary	[ˈsteiʃənəri]	*adj.* 不动的；定居的
string	[striŋ]	*n.* 字组，字符串
Sweden	[ˈswiːdn]	*n.* 瑞典
system	[ˈsistəm]	*n.* 系统
TCP/IP		*abbr.* = Transmission Control Protocol/Internet Protocol 传输控制/网络通信协议
telecommunication	[ˌtelikəˌmjuːniˈkeiʃn]	*n.* 电信；长途电信，（常复数）电信学
telegraph	[ˈteligraːf]	*n.* 电报，电信 *vt.* 打电报给 *vi.* 打电报
Telstar		*n.* （美国电话电报公司于 1962 年首度发射的）通信卫星
terrestrial	[təˈrestriəl]	*adj.* 地球的；陆地的，地面的 *n.* 地球人
Thomas Watson		*n.* 托马斯 · 沃森
Toshiba		*n.* 东芝（公司名）
transmission	[trænsˈmiʃn]	*n.* 传输
VPN		*abbr.* = Virtual Private Network 虚拟专用网
Washington	[ˈwɔʃiŋtən]	*n.* 美国华盛顿州；美国首都华盛顿
wireless	[ˈwaiələs]	*adj.* 无线的；无线电的

Exercises

Ⅰ. Phrase expressions

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 statements

1. In PCM, an analog signal is first sampled; then it is encoded as a unique series of digital bits, and then that level the 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. Translations

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 begun 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 8000 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 U – 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 users' 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 signals 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.

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 U-Law.

A-Law and U-Law Companding.

A-Law standard is used by European countries and U-Law is used by North America and Japan.

Similarities Between A-Law and U-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). 8 - 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 (2×4 kHz $\times 8$ bits = 64 kbps).

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 U-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 U-Law.

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

A-Law requires 13 bits for a uniform PCM equivalent. U-Law requires 14 bits for a uniform PCM equivalent.

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

(2)

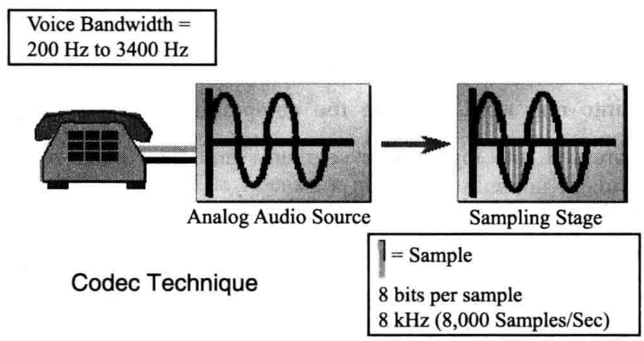

Fig. 1 – 3 Pulse Code Modulation – Nyquist Theorem

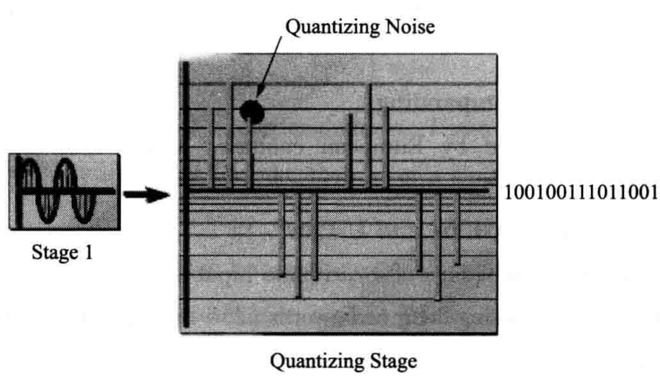

Fig. 1 – 4 Pules Code Modulation-Analog to Digital Conversion

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

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 64, 000 bits per second. Fig. 1 – 2 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. win be Cerrainly, Quantization noise reduces the SNR of a signal. Therefore, an increase in quantization noise degrades the quality of a voice signal. Fig. 1 – 3 shows how quantization noise is generated. For coding purpose, an N bit word will yield 2^N 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 with more code words.

SNR (including quantization noise) is the single most important factor affecting voice quality in uniform quantization. As stated prerionsly, 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 non-uniform quantization process called companding.

Unit 2
Systems Analysis and Design

Text

The hardware can do nothing itself; it requires programs collectively called software. The availability of software justified the purchase of hardware and it is the software that eventually determines how successful a computer system will be in satisfying your needs. Working under control of a stored program, a computer processes data into information. Any given computer application involves at least three components: hardware, software, and data. A system is a group of components that work, people, procedures, input and output, media, files, hardware, and software must be carefully coordinated.

Those people, called users, generally know what is required, but may lack the expertise to obtain it. Technical professionals, such as programmers, have the expertise, but may lack training in the user's field. To complicate matters, users and programmers often seem to speak different languages, leading to communication problems. A systems analyst is a professional who translates user needs into technical terms, thus serving as a bridge between users and technical professionals.

Like an engineer or an architect, a system's analyst solves problems by combining solid technical skills with insight, imagination, and a touch of art. Generally, the analyst follows well defined, methodical process that includes at least the following steps: 1. Problem definition. 2. Analysis. 3. Design. 4. Implementation. 5. Maintenance. At the end of each step, results are documented and shared with both the user and the programmers. The idea is to catch and correct errors and misunderstandings as early as possible.

The first step in the systems analysis and design process is problem definition. The analyst's objective is to determine what the user needs. Note that, as the process begins, the user possesses the critical information, and the analyst must listen and learn. At this stage, the analyst has no business even thinking about programs, files, and hardware, but must communicate with the user on his or her own terms. The idea is to ensure that both the user and the analyst are thinking about the same thing. Thus, a clear, written statement expressing the analyst's understanding of the problem is