高等学校电子信息类教材

通信与电子信息工程专业英语

◎聂　敏　畅志贤　金　蓉　王雅宁　编著

◎裴昌幸　主审

電子工業出版社
PUBLISHING HOUSE OF ELECTRONICS INDUSTRY
http://www.phei.com.cn

高等学校电子信息类教材

通信与电子信息工程专业英语

聂 敏　畅志贤　金 蓉　王雅宁　编著

裴昌幸　主审

電子工業出版社
Publishing House of Electronics Industry
北京 • BEIJING

内 容 简 介

本书选材广泛，内容新颖，由浅入深，涉及通信系统基础理论、光通信、微波与卫星通信、量子通信、移动通信、多媒体通信、激光技术、纳米技术、有线电视、计算机网络、电话交换、天线与智能天线、计算机断层扫描、人工智能、磁共振成像、传感器网络、电力线通信、数据压缩、电子显微镜、虹膜技术等，旨在通过扩展学生的专业词汇量，提高其阅读、理解专业文献资料的能力。全书共 18 个单元，其中前 17 单元是课堂教学内容，第 18 单元属于高级阅读内容，教师可根据教学计划灵活取舍。

本书可作为通信工程、电子信息、广播电视、信息安全、物联网等专业的本科生或研究生教材，也可作为通信工程技术人员的培训教材和参考书。

本书的电子教学课件可从华信教育资源网（www.hxedu.com.cn）免费下载。

图书在版编目（CIP）数据

通信与电子信息工程专业英语 / 聂敏等编著. —北京：电子工业出版社，2012.1
高等学校电子信息类教材
ISBN 978-7-121-15382-2

Ⅰ. ①通… Ⅱ. ①聂… Ⅲ. ①通信工程－英语－高等学校－教材②电子信息－英语－高等学校－教材
Ⅳ. ①H31

中国版本图书馆 CIP 数据核字（2011）第 252551 号

责任编辑：张来盛（zhangls@phei.com.cn）
印　　刷：
装　　订：北京市李史山胶印厂
出版发行：电子工业出版社
　　　　　北京市海淀区万寿路 173 信箱　邮编 100036
开　　本：787×1 092　1/16　印张：18.75　字数：600 千字
印　　次：2012 年 1 月第 1 次印刷
印　　数：3 000 册　　定价：35.00 元

凡所购买电子工业出版社图书有缺损问题，请向购买书店调换。若书店售缺，请与本社发行部联系，联系及邮购电话：（010）88254888。

质量投诉请发邮件至 zlts@phei.com.cn，盗版侵权举报请发邮件至 dbqq@phei.com.cn。

服务热线：（010）88258888。

前　　言

通信、电子信息类专业英语是通信工程、电子信息、广播电视、信息安全、物联网等专业或相关专业的重要课程之一。通过本课程的学习，学生能够在原有英语词汇的基础上，掌握更多、更广泛的专业词汇，熟悉专业英语的习惯用法和翻译技巧，提高阅读、理解专业文献资料的能力，为以后从事科学研究、工程设计、系统维护和项目研发奠定良好的专业英语基础。

本教材选材广泛，内容新颖。全书包括 18 个单元，内容不仅涉及通信系统基础理论中的调制解调技术、信道容量、多址技术、OFDM 技术、PCM 技术等，还包括光通信、微波与卫星通信、量子通信、第三代移动通信、TCP/IP、IPv6、多媒体通信、激光技术、纳米技术、有线电视、计算机网络、电话交换技术、天线与智能天线、计算机断层扫描、磁共振成像、传感器网络、电力线通信、人工智能、数据压缩、电子显微镜、虹膜技术等，其中前 17 单元属于课堂教学内容，第 18 单元属于高级阅读。考虑到不同专业计划课时的差异，本书在前 17 单元中，每个单元有 1～2 篇课文，还有 1～2 篇扩展阅读的内容，以适应 32 学时、48 学时和 64 学时的教学需要，教师可根据计划学时对教学内容灵活取舍。在第 18 单元中，有 11 篇高级阅读课文，供学有余力或准备考研、出国深造的学生扩展知识视野。本书的电子教学课件可从华信教育资源网（www.hxedu.com.cn）免费下载。

本书由西安邮电学院聂敏、畅志贤、金蓉和华北电力大学王雅宁共同编写，其中第 1～6 单元和第 18 单元中第 1～3 篇由金蓉编写，第 7～12 单元和第 18 单元中第 4～6 篇由畅志贤编写，第 13～15 单元、17 单元和第 18 单元的 7～11 篇由聂敏编写，第 16 单元由王雅宁编写。本书承蒙西安电子科技大学通信工程学院原副院长、博士生导师裴昌幸教授主审。裴教授在百忙之中抽出宝贵时间，全面、仔细地审阅了全书，并提出了十分宝贵的修改意见，在此表示由衷的感谢。

本教材的编写，得到了西安邮电学院副校长、博士生导师范九伦教授，以及西安邮电学院通信与信息工程学院副院长刘毓教授，通信与信息工程学院副院长、博士生导师卢光跃教授的大力支持，编著者对此深表谢忱。此外，还要特别感谢西安邮电学院通信与信息工程学院通信工程系主任杨武军副教授的大力支持和帮助。

在本书的编写过程中，引用了书末所列参考文献的部分内容；这些内容不仅凝聚了原作者的智慧和心血，也使本教材能够全面体现通信、电子信息领域的最新研究成果和技术发展水平。谨向这些成果的原作者表示诚挚的感谢。

由于经验不足，加上编著者水平有限、时间仓促，本教材难免有疏漏之处，敬请读者批评指正。谢谢。

编著者

2011 年 10 月

目　　录

Unit 1 Basics of Communication

1948 年，Shannon 发表了有关信息论的著名研究论文。他首次提出，通信的基本问题是，通信双方如何能够精确或近似地复制对方发出的信息，以及如何对该信息进行定量分析。Shannon 的论文对信源、信道、编码、译码和信宿进行了精确描述，建立了通信系统的数学模型，由此产生了信息论这门新学科。该论文还第一次强调了概率论和随机过程在通信中的重要性，提出了信源编码定理和信道编码定理。现在，信息论与纠错编码、信道容量、信息熵、信源压缩编码等理论已经成为通信与信息学科发展的重要基础理论体系。

Text A

A Mathematical Theory of Communication

The recent development of various methods of modulation such as PCM and PPM which exchange bandwidth for signal-to-noise ratio has intensified the interest in a general theory of communication [1]. A basis for such a theory is contained in the important papers of Nyquist and Hartley on this subject [2]. In this paper we will extend the theory to include a number of new factors, in particular the effect of noise in the channel, and the savings possible due to the statistical structure of the original message and due to the nature of the final destination of the information [3].

The fundamental problem of communication is that of reproducing at one point either exactly or approximately a message selected at another point [4]. Frequently the messages have meaning, that is they refer to or are correlated according to some system with certain physical or conceptual entities [5]. These semantic aspects of communication are irrelevant to the engineering problem. A significant aspect is that the actual message is one selected from a set of possible messages. The system must be designed to operate for each possible selection, not just the one which will actually be chosen since this is unknown at the time of design.

If the number of messages in the set is finite then this number or any monotonic function of the number can be regarded as a measure of the information produced when one message is chosen from the set, all choices being equally like. As was pointed out by Hartley the most natural choice is the logarithmic function. Although this definition must be generalized considerably when we consider the influence of the statistics of the message and when we have a continuous range of messages, we will in all cases use an essentially logarithmic measure.

The logarithmic measure is more convenient for various reasons:

1. It is practically more useful. Parameters of engineering importance such as time, bandwidth, the number of relays, etc., tend to vary linearly with the logarithm of the number of possibilities. For example, adding one relay to a group doubles the number of possible states of the relays. It adds 1 to the base 2 logarithm of this number. Doubling the time roughly squares the number of possible messages, or doubles the logarithm, etc.

2. It is nearer to our intuitive feeling as to the proper measure. This is closely related to (1) since we intuitively measures entities by linear comparison with common standards. One feels, for example, that two punched cards should have twice the capacity of one for information storage, and two identical channels twice the capacity of one for transmitting information.

3. It is mathematically more suitable. Many of the limiting operations are simple in terms of the logarithm but would require clumsy restatement in terms of the number of possibilities.

The choice of a logarithmic base corresponds to the choice of a unit for measuring information. If the base 2 is used, the resulting units may be called binary digits, or more briefly bits, a word suggested by J. W. Tukey. A device with two stable positions, such as a relay or a flip-flop circuit, can store one bit of information. N such devices can store N bits, since the total number of possible states is $2N$ and $\log_2 2^N = N$. If the base 10 is used, the units may be called decimal digits. Since

$$\log_2 M = \log_{10} M / \log_{10} 2 = 3.32 \log_{10} M \tag{1.1}$$

A decimal digit is about $3\frac{1}{3}$ bits. A digit wheel on a desk computing machine has ten stable positions and therefore has a storage capacity of one decimal digit. In analytical work where integration and differentiation are involved the base e is sometimes useful. The resulting units of information will be called natural units. Change from the base a to base b merely requires multiplication by $\log_b a$.

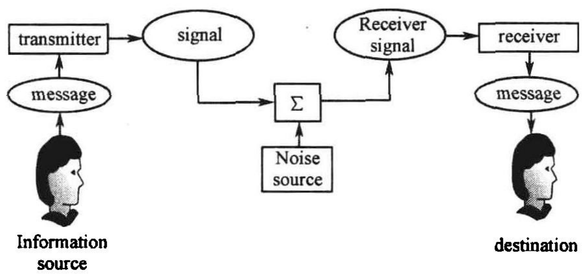

Figure 1.1 Schematic diagram of a general communication system

By a communication system we will mean a system of the type indicated schematically in Figure 1.1. It consists of essentially five parts:

1. **An information source** which produces a message or sequence of messages to be communicated to the receiving terminal. The message may be of various types: (a) A sequence of letters as in a telegraph of teletype system; (b) A single function of time $f(t)$ as in radio or telephony; (c) A function of time and other variables as in black and white television — here the message may be thought of as a function $f(x,y,t)$ of two space coordinates and time, the light intensity at point (x, y) and time t on a pickup tube plate; (d) Two or more functions of time, say $f(t)$, $g(t)$, $h(t)$—this is the case in three dimensional sound transmission or if the system is intended to service several individual channels in multiplex; (e) Several functions of several variables—in color television the message consists of three functions $f(x,y,t)$, $g(x,y,t)$ and $h(x,y,t)$ defined in a three-dimensional continuum—we may also think of these three functions as components of a vector field defined in the region — similarly, several black and white television sources would produce messages consisting of a number of functions of three variables; (f) Various combinations also occur, for

example, in television with an associated audio channel.

2. **A transmitter** which operates on the message in some way to produce a signal suitable for transmission over the channel. In telephony this operation consists merely of changing sound pressure into a proportional electrical current. In telegraphy we have an encoding operation which produces a sequence of dots, dashes and spaces on the channel corresponding to the message. In a multiplex PCM system the different speech functions must be sampled, compressed, quantized and encoded, and finally interleaved properly to construct the signal. Vocoder systems, television and frequency modulation are other examples of complex operations applied to the message to obtain the signal.

3. **The channel** is merely the medium used to transmit the signal from transmitter to receiver. It may be a pair of wires, a coaxial cable, a band of radio frequencies, and a beam of light, etc.

4. **The receiver** ordinarily performs the inverse operation of that done by the transmitter, reconstructing the message from the signal.

5. **The destination** is the person (or thing) for whom the message is intended.

We wish to consider certain general problems involving communication systems. To do this it is first necessary to represent the various elements involved as mathematical entities, suitably idealized from their physical counterparts. We may roughly classify communication systems into three main categories: discrete, continuous and mixed. By a discrete system we will mean one in which both the message and the signal are a sequence of discrete symbols. A typical case is telegraphy where the message is a sequence of letters and the signal is a sequence of dots, dashes and spaces. A continuous system is one in which the message and the signal are both treated as continuous functions, e.g., radio or television. A mixed system is one in which both discrete and continuous variables appear, e.g., PCM transmission of speech.

课文注释：

[1] The recent development of various methods of modulation such as PCM and PPM which exchange bandwidth for signal-to-noise ratio has intensified the interest in a general theory of communication. 各种调制方法，如 PCM 和 PPM，能够完成带宽与信噪比的交换，它们的发展增强了通信理论的重要性。

[2] A basis for such a theory is contained in the important papers of Nyquist and Hartley on this subject. 在本学科，Nyquist 和 Hartley 的重要论文中包含了这种理论的基础。

[3] In this paper we will extend the theory to include a number of new factors, in particular the effect of noise in the channel, and the savings possible due to the statistical structure of the original message and due to the nature of the final destination of the information. 在本文中，我们将这些理论进行扩展，包括许多新的因素，尤其是信道中噪声的影响，以及可能由于原始信息的统计结构和信宿的特性所导致的信息存储。

[4] The fundamental problem of communication is that of reproducing at one point either exactly or approximately a message selected at another point. 通信的基本问题是如何在一点精确或近似地再生来自另一点的信息。

[5] Frequently the messages have meaning, that is they refer to or are correlated according to some system with certain physical or conceptual entities. 信息通常有具体含义，按照一定的物理或概念实体，其含义与这些系统相互关联。

Text B

Information Theory

Claude Shannon laid the foundation of information theory in 1948. His paper "A Mathematical Theory of Communication" published in *Bell System Technical Journal* is the basis for the entire telecommunications developments that have taken place during the last five decades[1]. A good understanding of the concepts proposed by Shannon is a must for every budding telecommunication professional[2].

In any communication system, there will be an information source that produces information in some form, and an information sink absorbs the information[3]. The communication medium connects the source and the sink. The purpose of a communication system is to transmit the information from the source to the sink without errors. However, the communication medium always introduces some errors because of noise[4]. The fundamental requirement of a communication system is to transmit the information without errors in spite of the noise[5].

The requirement of a communication system is to transmit the information from the source to the sink without errors, in spite of the fact that noise is always introduced in the communication medium.

In a generic communication system, the information source produces symbols (such as English letters, speech, video, etc.) that are sent through the transmission medium by the transmitter. The communication medium introduces noise, and so errors are introduced in the transmitted data. At the receiving end, the receiver decodes the data and gives it to the information sink.

As an example, consider an information source that produces two symbols A and B. The transmitter codes the data into a bit stream. For example, A can be coded as 1 and B as 0. The stream of 1's and 0's is transmitted through the medium. Because of noise, 1 may become 0 or 0 may become 1 at random places, as illustrated below:

Symbols produced	A	B	B	A	A	A	B	A	B	A
Bit stream produced	1	0	0	1	1	1	0	1	1	1
Bit stream received	1	0	0	1	1	1	1	1	1	1

At the receiver, one bit is received in error. How to ensure that the received data can be made error free? Shannon provides the answer.

In this block diagram (Figure 1.2), the information source produces the symbols that are coded using two types of coding—source encoding and channel encoding—and then modulated and sent over the medium. At the receiving end, the modulated signal is demodulated, and the inverse operations of channel encoding and source encoding (channel decoding and source decoding) are performed. Then the information is presented to the information sink.

As proposed by Shannon, the communication system consists of source encoder, channel encoder and modulator at the transmitting end, and demodulator, channel decoder and source decoder at the receiving end.

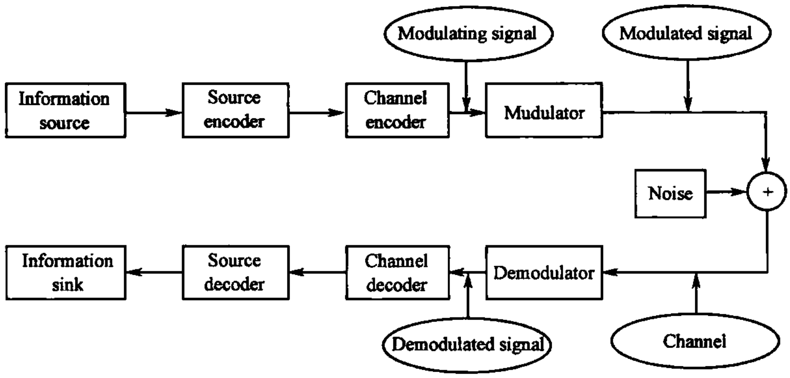

Figure 1.2 The communication system model

Information source: The information source produces the symbols. If the information source is, for example, a microphone, the signal is in analog form. If the source is a computer, the signal is in digital form (a set of symbols).

Source encoder: The source encoder converts the signal produced by the information source into a data stream. If the input signal is analog, it can be converted into digital form using an analog-to-digital converter. If the input to the source encoder is a stream of symbols, it can be converted into a stream of 1 and 0 using some type of coding mechanism. For instance, if the source produces the symbols A and B, A can be coded as 1 and B as 0. Shannon's source coding theorem tells us how to do this coding efficiently.

Source encoding is done to reduce the redundancy in the signal. Source coding techniques can be divided into lossless encoding techniques and lossy encoding techniques. In lossy encoding techniques, some information is lost.

In source coding, there are two types of coding—lossless coding and lossy coding. In lossless coding, no information is lost. When we compress our computer files using a compression technique (for instance, WinZip), there is no loss of information. Such coding techniques are called lossless coding techniques. In lossy coding, some information is lost while doing the source coding. As long as the loss is not significant, we can tolerate it. When an image is converted into JPEG format, the coding is lossy coding because some information is lost. Most of the techniques used for voice, image, and video coding are lossy coding techniques.

The compression utilities we use to compress data files use lossless encoding techniques. JPEG image compression is a lossy technique because some information is lost.

Channel encoder: If we have to decode the information correctly, even if errors are introduced in the medium, we need to put some additional bits in the source-encoded data so that the additional information can be used to detect and correct the errors. This process of adding bits is done by the channel encoder. Shannon's channel coding theorem tells us how to achieve this.

In channel encoding, redundancy is introduced so that at the receiving end, the redundant bits can be used for error detection or error correction.

Modulation: Modulation is a process of transforming the signal so that the signal can be transmitted through the medium.

Demodulator: The demodulator performs the inverse operation of the modulator.

Channel decoder: The channel decoder analyzes the received bit stream and detects and corrects the errors, if any, using the additional data introduced by the channel encoder.

Source decoder: The source decoder converts the bit stream into the actual information. If analog-to-digital conversion is done at the source encoder, digital-to-analog conversion is done at the source decoder. If the symbols are coded into 1 and 0 at the source encoder, the bit stream is converted back to the symbols by the source decoder.

Information sink: The information sink absorbs the information.

What is information? How do we measure information? These are fundamental issues for which Shannon provided the answers. We can say that we received some information if there is decrease in uncertainty. Consider an information source that produces two symbols A and B. The source has sent A, B, B, A, and now we are waiting for the next symbol. Which symbol will it produce? If it produces A, the uncertainty that was there in the waiting period is gone, and we say that information is produced. Note that we are using the term information from a communication theory point of view; it has nothing to do with the usefulness of the information.

Shannon proposed a formula to measure information. The information measure is called the entropy of the source. If a source produces N symbols, and if all the symbols are equally likely to occur, the entropy of the source is given by:

$$H=\log_2 N \quad \text{bits/symbol} \tag{1.2}$$

Shannon introduced the concept of channel capacity, the limit at which data can be transmitted through a medium. The errors in the transmission medium depend on the energy of the signal, the energy of the noise, and the bandwidth of the channel. Conceptually, if the bandwidth is high, we can pump more data in the channel. If the signal energy is high, the effect of noise is reduced. According to Shannon, the bandwidth of the channel and signal energy and noise energy are related by the formula:

$$C = W \log_2(1 + S/N) \tag{1.3}$$

Where C is channel capacity in bits per second (bps), W is bandwidth of the channel in Hz, S/N is the signal-to-noise power ratio (SNR).The value of the channel capacity obtained using this formula is the theoretical maximum. So, we cannot transmit data at a rate faster than this value in a voice-grade line. An important point to be noted is that in the above formula, Shannon assumes only thermal noise. To increase C, can we increase W? No, because increasing W increases noise as well, and SNR will be reduced. To increase C, can we increase the SNR? No, that results in more noise, called intermodulation noise.

The entropy of information source and channel capacity are two important concepts, based on which Shannon proposed his theorems.

In a digital communication system, the aim of the designer is to convert any information into a digital signal, pass it through the transmission medium and, at the receiving end, reproduce the digital signal exactly. To achieve this objective, two important requirements are:

- ▶ To code any type of information into digital format, note that the world is analog—voice signals are analog, images are analog. We need to devise mechanisms to convert analog signals into digital format. If the source produces symbols (such as A, B), we also need to convert these symbols into a bit stream. This coding has to be done efficiently so that the smallest number of bits is required for coding.
- ▶ To ensure that the data sent over the channel is not corrupted. We cannot eliminate the noise introduced on the channels, and hence we need to introduce special coding techniques to overcome the effect of noise.

These two aspects have been addressed by Claude Shannon in his classical paper "A Mathematical Theory of Communication" published in 1948 in *Bell System Technical Journal*, which gave the foundation to information theory. Shannon addressed these two aspects through his source coding theorem and channel coding theorem.

课文注释：

[1] Claude Shannon laid the foundation of information theory in 1948. His paper "A Mathematical Theory of Communication" published in *Bell System Technical Journal* is the basis for the entire telecommunications developments that have taken place during the last five decades. 在 1948 年，Claude Shannon 建立了信息论的基础。他的论文《通信的数学理论》发表在《贝尔系统技术杂志》上，该论文是近 50 年整个通信发展的基础。

[2] A good understanding of the concepts proposed by Shannon is a must for every budding telecommunication professional. 每个通信从业人员，必须对 Shannon 所提出的概念有很好的理解。

[3] In any communication system, there will be an information source that produces information in some form, and an information sink absorbs the information. 在任何通信系统中，都包括以一定形式产生信息的信源，以及获取信息的信宿。

[4] The communication medium connects the source and the sink. The purpose of a communication system is to transmit the information from the source to the sink without errors. However, the communication medium always introduces some errors because of noise. 传输媒体连接信源和信宿。通信系统的目的，是无差错地将信息从信源传送至信宿。但是，由于噪声的存在，传送媒体往往会引入一些差错。

[5] The fundamental requirement of a communication system is to transmit the information without errors in spite of the noise. 尽管有噪声存在，但对通信系统的基本要求是无差错地传送信息。

Expanding reading

Channel Capacity

A variety of impairments can distort or corrupt a signal. A common impairment is noise, which is any unwanted signal that combines with and hence distorts the signal intended for transmission and reception. For the purposes of this section, we simply need to know that noise is something that degrades signal quality. For digital data, the question that then arises is to what extent these impairments limit the data rate that can be achieved. The maximum rate at which data can be transmitted over a given communication path, or channel, under given conditions is referred to as the channel capacity.

There are four concepts here that we are trying to relate to one another:

- Data rate: This is the rate, in bits per second (bps), at which data can be communicated.
- Bandwidth: This is the bandwidth of the transmitted signal as constrained by the transmitter and the nature of the transmission medium, expressed in cycles per second, or Hertz.
- Noise: For this discussion, we are concerned with the average level of noise over the communications path.
- Error rate: This is the rate at which errors occur, where an error is the reception of 1 when 0 was transmitted or the reception of 0 when 1 was transmitted. The problem we are addressing is this: Communications facilities are expensive and, in general, the greater the bandwidth of a facility, the greater the cost. Furthermore, all transmission channels of any practical interest are of limited bandwidth. The limitations arise from the physical properties of the transmission medium or from deliberate limitations at the transmitter on the bandwidth to prevent interference from other sources. Accordingly, we would like to make as efficient use as possible of a given bandwidth. For digital data, this means that we would like to get as high a data rate as possible at a particular limit of error rate for a given bandwidth. The main constraint on achieving this efficiency is noise.

Nyquist Bandwidth

To begin, let us consider the case of a channel that is noise free. In this environment, the limitation on data rate is simply the bandwidth of the signal. A formulation of this limitation, due to Nyquist, states that if the rate of signal transmission is $2B$, then a signal with frequencies no greater than B is sufficient to carry the signal rate. The converse is also true: Given a bandwidth of B, the highest signal rate that can be carried is $2B$. This limitation is due to the effect of inter symbol interference, such as is produced by delay distortion. The result is useful in the development of digital-to-analog encoding schemes.

We referred to signal rate. If the signals to be transmitted are binary, then the data rate that can be supported by B Hz is $2B$ bps. As an example, consider a voice channel being used, via modem, to transmit digital data. Assume a bandwidth of 3 100 Hz. Then the capacity, C, of the channel is $2B$ = 6 200 bps. However, signals with more than two levels can be used; that is, each signal element can represent more than one bit. For example, if four possible voltage levels are used as signals, then each signal element can represent two bits. With multilevel signal, the Nyquist formulation becomes

$$C = 2B\log_2 M \qquad (1.4)$$

Where M is the number of discrete signal elements or voltage level. Thus, for M = 8, a value used with some modems, a bandwidth of $B = 3\ 100$ Hz yields a capacity $C = 18\ 600$ bps.

So, for a given bandwidth, the data rate can be increased by increasing the number of different signal elements. However, this places an increased burden on the receiver. Instead of distinguishing one of two possible signal elements during each signal time, it must distinguish one of M possible signals. Noise and other impairments on the transmission line will limit the practical value of M.

Shannon Capacity Formula

Nyquist's formula indicates that, all other things being equal, doubling the bandwidth doubles the data rate. Now consider the relationship among data rate, noise, and error rate. The presence of noise can corrupt one or more bits. If the data rate is increased, then the bits become shorter in time, so that more bits are affected by a given pattern of noise. Thus, at a given noise level, the higher the data rate, the higher the error rate.

Here is an example of the effect of noise on a digital signal. The noise consists of a relatively modest level of background noise plus occasional larger spikes of noise. The digital data can be recovered from the signal by sampling the received waveform once per bit time. As can be seen, the noise is occasionally sufficient to change 1 to 0 or 0 to 1.

All of these concepts can be tied together neatly in a formula developed by the mathematician Claude Shannon. As we have just mentioned, the higher the data rate, the more damage that unwanted noise can do. For a given level of noise, we would expect that greater signal strength would improve the ability to receive data correctly in the presence of noise. The key parameter involved in this reasoning is the signal-to-noise ratio (or S/N), which is the ratio of the power in a signal to the power contained in the noise that is present at a particular point in the transmission. Typically, this ratio is measured at a receiver, because it is at this point that an attempt is made to process the signal and eliminate the unwanted noise. For convenience, this ratio is often reported in decibels:

$$\mathrm{SNR}_{\mathrm{dB}} = 10\log_{10}\frac{\text{signal power}}{\text{noise power}} \tag{1.5}$$

This expresses the amount, in decibels, that the intended signal exceeds the noise level. A high SNR will mean a high-quality signal.

The signal-to-noise ratio is important in the transmission of digital data because it sets the upper bound on the achievable data rate. Shannon's result is that the maximum channel capacity, in bits per second, obeys the equation

$$C = B\log_2(1+\mathrm{SNR}) \tag{1.6}$$

Where C is the capacity of the channel in bits per second, B is the bandwidth of the channel in Hertz. The Shannon formula represents the theoretical maximum that can be achieved. In practice, however, only much lower rates are achieved. One reason for this is that the formula assumes white noise (thermal noise).

The capacity indicated in the preceding equation is referred to as the error-free capacity. Shannon proved that if the actual information rate on a channel is less than the error-free capacity, then it is theoretically possible to use a suitable signal code to achieve error-free transmission