



通信与信息工程专业英语

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**English in Communication
and Information Engineering**

哈尔滨工业大学出版社

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内容提要

本书共 11 单元。第 1~5 单元分别为连续信号、离散和随机信号、信号处理基础、调制和解调、滤波器,介绍了信号分析与处理的基本知识;第 6~8 单元分别为通信网络、光纤通信、移动通信,介绍了现代通信的新技术;第 9 单元多媒体信号处理系统给出了信息处理系统实例;第 10 单元可编程 DSP 和第 11 单元 ARM 体系结构介绍了通信与信息工程中常用的核心设备 DSP 与 ARM。

本书给出了大量的通信与信息工程专业词汇、短语和专有缩写词,并对较难的长句给出了译文。本书可供大学本科或专科通信工程、电子信息工程及电子信息科学与技术等相关专业高年级学生使用,也可供相关的科研和工程技术人员参考。

图书在版编目(CIP)数据

通信与信息工程专业英语/刘露主编. —哈尔滨:哈尔滨工业大学出版社,2010.6
ISBN 978-7-5603-3045-7

I. ①通… II. ①刘… III. ①通信工程-英语 ②信息技术-英语 IV. ①H31

中国版本图书馆 CIP 数据核字(2010)第 127055 号

责任编辑 王桂芝
封面设计 卞秉利
出版发行 哈尔滨工业大学出版社
社 址 哈尔滨市南岗区复华四道街 10 号 邮编 150006
传 真 0451-86414749
网 址 <http://hitpress.hit.edu.cn>
印 刷 哈尔滨市石桥印务有限公司
开 本 880mm×1230mm 1/32 印张 10.5 字数 310 千字
版 次 2010 年 6 月第 1 版 2010 年 6 月第 1 次印刷
书 号 ISBN 978-7-5603-3045-7
定 价 22.00 元

(如因印装质量问题影响阅读,我社负责调换)

前 言

本书是为通信与信息工程相关专业学生学习专业英语课程而编写的,其目的是加强通信与信息工程相关专业学生科技英语的学习及训练,为学生打下良好的科技英语基础,使其能够熟练掌握通信与信息工程相关专业英语词汇,提高对通信与信息工程相关专业英文文献的阅读能力和学术交流能力。

本书共 11 单元,分为基础原理和专业技术两部分。内容包括:连续信号,离散和随机信号,信号处理基础,调制和解调,滤波器,通信网络,光纤通信,移动通信,多媒体信号处理系统,可编程 DSP,ARM 体系结构等。

本书内容丰富,在教学安排上各学校(任课教师)可根据学生的英语水平和学校对该课程的课时要求灵活安排,其中有些内容可作为学生课外阅读材料。

本书的特色是:① 针对性强。本教材针对通信与信息工程专业学生介绍与该专业基础课程和专业课程有关的英语基础知识和专业知识。② 方便学生阅读。本书给出了大量的通信与信息工程专业词汇、短语和专有缩写词,并对较难的长句给出了译文。

本书第 1、2、3、4、5 单元由哈尔滨理工大学刘露编写,第 6、7、8 单元由张斌编写,第 9、10、11 单元由马建为编写,全书由刘露统稿。

在本书的编写过程中,朱非甲、闫俊丽、冯少华、马思明、白祥云等研究生在词汇及音标的录入和校对方面做了大量工作,在此表示衷心感谢!由于地域和时间的原因,难以与引用的参考文献原作者取得联系,在此一并表示感谢!

本书可供大学本科或专科通信工程、电子信息工程及电子信息科学与技术等相关专业高年级学生使用,也可供相关的科研和工程技术人员参考。

由于本书作者水平所限,书中难免出现疏漏或不当之处,敬请广大读者和同仁提出宝贵意见,以便今后再版时加以改进。

编 者
2010 年 4 月

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Continuous Signals

1.1 Signals and Descriptions

1.1.1 Signals

Signals describe quantities that change. Fig. 1.1 depicts the electrical voltage that a microphone produces in response to the spoken word “car”. This voltage corresponds largely to the acoustic pressure on our ear, which reacts to the changes in this pressure over time. The curve in Fig. 1.1 shows the value of microphone voltage in relation to time. Since there is a voltage value for every point in time, we term this a *continuous-time* signal. We call time the *independent* variable and the voltage changing over time the dependent variable or signal **amplitude**. We usually represent the independent variable horizontally (*x*-axis) and the dependent variable vertically (*y*-axis).

Fig. 1.2 depicts another continuous signal. The diagram shows the temperature curves for a house wall, not over time, but in relation to the location. The curves show the temperature profile

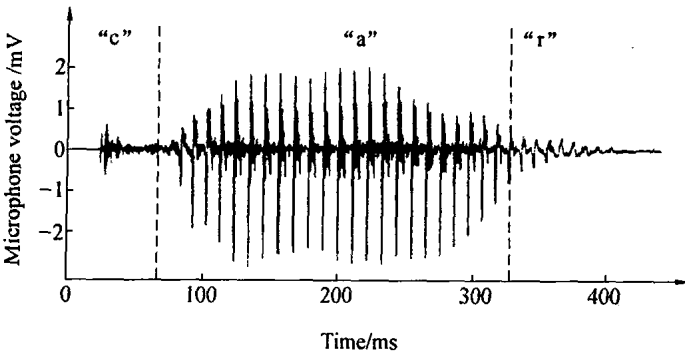


Fig. 1.1 Example of a continuous-time signal: a voice signal for the syllable “car”

inside a 15cm thick brick wall where the air temperature at the right side suddenly rose by 10 K. One hour later the local temperature follows the curve represented by the thick time. At another time we would have a different temperature curve. In contrast to Fig. 1.1, time here is a parameter of a family of curves; the independent continuous variable is the location in the wall.

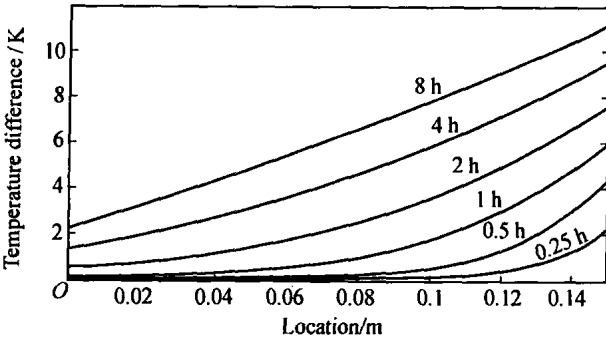


Fig. 1.2 Temperature curve for a house wall

Fig. 1.3 shows another kind of variable quantity, the stock market index over time. Although this index changes all the while the stock market is open, the diagram shows only the weekly average. Thus the depicted value does not change continuously,

but only once a week. When the signal amplitude occurs only at certain fixed points in time (discrete times), but not for points in between, we call the signal *discrete* or, more precisely, *discrete-time*. In our example, however, the signal amplitude itself is not discrete but continuous.

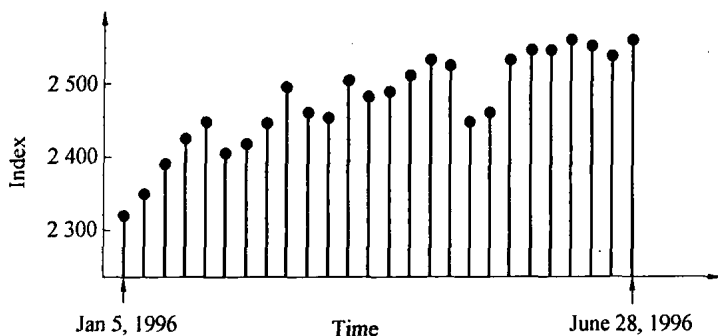


Fig. 1.3 The weekly German stock market index between January 5, 1996, and June 28, 1996

In Fig. 1.4 we have entered the frequency of earned marks for a test in system theory at the University of Erlangen-Nurnberg in April, 1996. The individual marks assume only discrete values (1.0 ~ 5.0); the frequencies (in contrast to the average stock index) are whole numbers and so likewise discrete. In this case both the independent and the dependent variable are discrete.

The signals we have considered thus far have been quantities that depend on a *single* independent variable. However, there are quantities with dependencies on two or more variables. The grayscale images of Fig. 1.5 depend on both x and the y co-ordinates. Here both axes represent independent variables. The dependent variable $s(x, y)$ is entered along one axis, but is a grayscale between the extreme values black and white.

When we add motion to pictures, we have a dependency on three independent variables (Fig. 1.6): two co-ordinates and

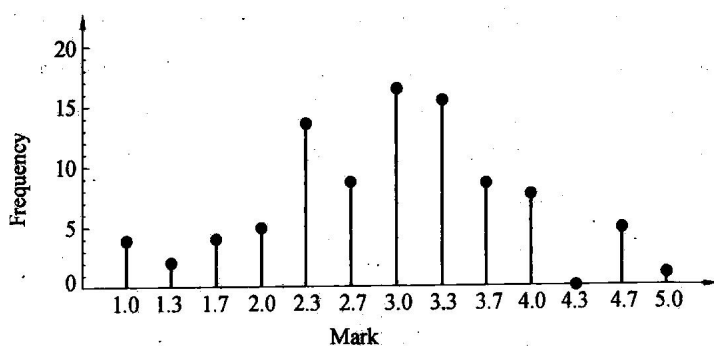


Fig. 1.4 Frequency of earned marked for a test in systems theory



Fig. 1.5 A picture as a continuous two-dimensional signal

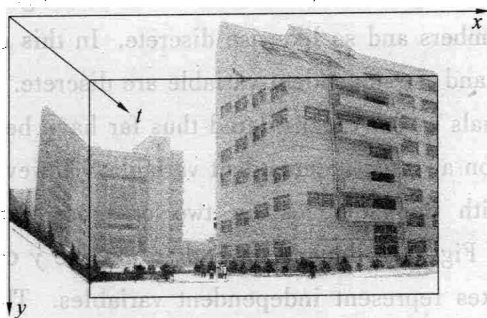


Fig. 1.6 Moving pictures as an example of a continuous three-dimensional signal

time. We call these two-or three-dimensional (or generally

multidimensional) signals. When grayscale values change continuously over space or over space and time, these are continuous signals.

All our example have shown parameters (voltage, temperature, stock index, frequencies, grayscale) that change in relation to values of then independent variables. Thereby they transmit certain information. In this book we define a signal as follow:

A signal is a function or sequence of values that represents information.

The preceding examples have shown that signals can assume different forms. Signals can be classified according to various **criteria**, the most important of which are summarized in Tab. 1.1.

Tab. 1.1 Criteria for classifying signals

| continuous(-time) | discrete(-time) |
|----------------------|--------------------|
| amplitude-continuous | amplitude-discrete |
| analogue | digital |
| real-valued | complex-valued |
| unidimensional | multidimensional |
| finite domain | infinite domain |
| deterministic | stochastic |

We have already discussed the difference between continuous and discrete signals on the basis of Fig. 1.1 and Fig. 1.5. Discrete signals are also termed discontinuous. Most of the preceding signals have been amplitude-continuous, because their dependent variable can take on any value. However, the signal in Fig. 1.4 is amplitude-discrete, for the dependent variable (number of examinees) can assume only integer values. Taken precisely, the stock index in Fig. 1.3 is likewise amplitude-discrete, since the stock

index is specified to only a certain number of decimal places. Signals whose dependent and independent variables are continuous are called *analogue* signals. If both variables are discrete, we call the signal *digital*. The output voltage of microphone is an analogue signal, for at any given time amplitude values can be read with any desired precision. Sequences of values stored in a computer are always digital, since the amplitude values can be stored only with finite word length in distinct (discrete) storage cells.

All of signals we have considered so far had real amplitudes and so are classified as *real-valued*. Signals whose dependent variable assumes complex values are called *complex-valued*.

The signals in Fig. 1.1 to 1.4 are unidimensional, while those in Fig. 1.5 and 1.6 are multidimensional. For reasons of graphic representation, all the signals in the previous examples had finite domains of their independent variables and so are classified as *finite-domain* signals. However, if we consider the signal in Fig. 1.6 as the picture of a television camera, then the domain of the location variable becomes finite again due to the restricted picture excerpt, but the domain of the time variable is infinite (neglecting the finite lifetime of the camera).

Signals are termed *deterministic* if their behavior is known and can be represented, e. g. , by a formula. The deflection voltage of an oscilloscope is a deterministic signal, for its behavior is known and can be represented as a **sawtooth** wave. By contrast, we cannot define the amplitude values of a voice signal (see Fig. 1.1) by means of formulae or graphical elements; furthermore, their continued behavior is not known. Such signals are termed *stochastic*. Since it is impossible to specify their behavior in terms of functions, such signals are described by expected values (mean, variance and many others).

1.1.2 Time-domain Descriptions

The fact that the great majority of functions which may usefully be considered as signals are functions of time lends justification to the treatment of signal theory in terms of time and of frequency^[1].

A periodic signal will therefore be considered to be one which repeats itself exactly every T seconds, where T is called the period of the signal waveform; the theoretical treatment of periodic waveforms assumes that this exact repetition is extended throughout all time, both past and future. In practice, of course, signals do not repeat themselves indefinitely. Nevertheless, a waveform such as the output voltage of a **mains rectifier** prior to smoothing does repeat itself very many times, and its analysis as a strictly periodic signal yields valuable results. In other cases, such as the **electrocardiogram (ECG)**, the waveform is **quasi-periodic** and may usefully be treated as truly periodic for some purpose. It is worth noting that a truly repetitive signal is of very little interest in a communication **channel**, since no further information is conveyed after the first cycle of the waveform has been received. One of the main reasons for discussing periodic signals is that a clear understanding of their analysis is a great help when dealing with **aperiodic** and random ones.

A complete time-domain description of such a signal involves specifying its value precisely at every **instant** of time. In some cases this may be done very simply using mathematical notation. Fortunately, it is in many cases useful to describe only certain aspects of a signal waveform, or to represent it by a mathematical formula which is only approximate. The following aspects might be relevant in particular cases:

- (1) the average value of the signal;
- (2) the **peak** value reached by the signal;
- (3) the proportion of the total time spent between value a and b ;
- (4) the period of the signal.

If it is desired to **approximate** the waveform by a mathematical expression, such techniques as a **polynomial expansion**, a **Taylor series**, or a **Fourier series** may be used. A polynomial of **order** n having the form

$$f(t) = a_0 + a_1t + a_2t^2 + a_3t^3 + \dots + a_nt^n$$

may be used to fit the actual curve at $(n + 1)$ **arbitrary** points. The accuracy of fit will generally improve as the number of polynomial **terms** increases. It should also be noted that the error between the true signal waveform and the polynomial will normally become very large away from the region of the fitted points, and that the polynomial itself cannot be periodic. Whereas a polynomial approximation fits the actual waveform at a number of arbitrary points, the alternative Taylor series approximation provides a good fit to a smooth continuous waveform in the vicinity of one selected point. The coefficients of the Taylor series are chosen to make the series and its derivatives agree with the actual waveform at this point. The number of terms in the series determines to what order of derivative this agreement will extend, and hence the accuracy with which series and actual waveform agree in the region of the point chosen^[2]. The general form of the Taylor series for approximating a function $f(t)$ in the region of the point $t = a$ is given by

$$f(t) = f(a) + (t - a) \times \frac{df(a)}{dt} + \frac{(t - a)^2}{2!} \times \frac{d^2f(a)}{dt^2} + \dots +$$

$$\frac{(t-a)^n}{n!} \times \frac{d^n f(a)}{dt^n}$$

Generally speaking, the fit to the actual waveform is good in the region of the point chosen, but rapidly **deteriorates** to either side. The polynomial and Taylor series descriptions of a signal waveform are therefore only to be recommended when one is concerned to achieve accuracy over a limited region of the waveform. The accuracy usually decreases rapidly away from this region, although it may be improved by including additional terms (so long as t lies within the region of convergence of the series)^[3]. The approximations provided by such methods are never periodic in form and cannot therefore be considered ideal for the description of repetitive signals.

By contrast the Fourier series approximation is well suited to the representation of a signal waveform over an extended interval. When the signal is periodic, the accuracy of the Fourier series description is maintained for all time, since the signal is represented as the sum of a number of sinusoidal functions which are themselves periodic^[4]. Before examining in detail the Fourier series method of representing a signal, the background to what is known as the 'frequency-domain' **approach** will be introduced.

1.1.3 Frequency-domain Descriptions

The basic conception of frequency-domain analysis is that a waveform of any complexity may be considered as the sum of a number of sinusoidal waveforms of suitable amplitude, periodicity, and relative phase^[5]. A continuous sinusoidal function ($\sin \omega t$) is thought of as a 'single frequency' wave of frequency ω **radians/second**, and the frequency-domain description of a signal involves its breakdown into a number of such