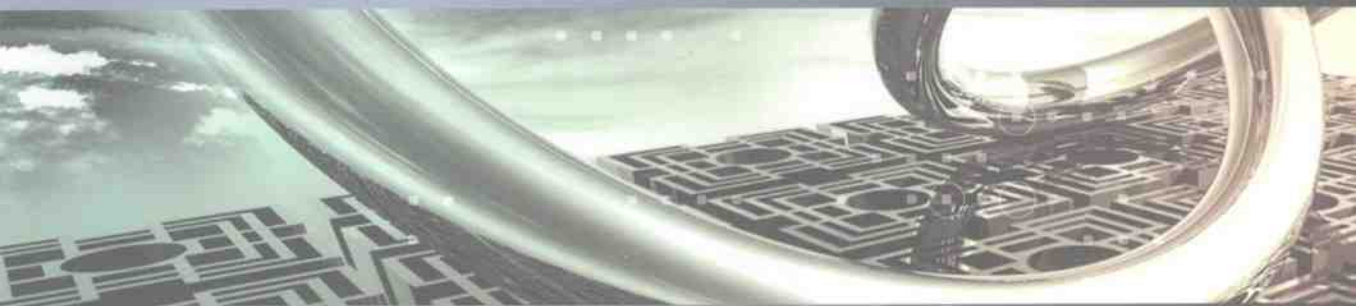
高等学校专业英语教材

测控技术与仪器专业英语教程

（第2版）

刘曙光　曹军义　主编



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

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北京 · BEIJING

内容简介

本书旨在使读者掌握测控技术与仪器专业英语术语及用法，培养和提高读者阅读和翻译专业英语文献资料的能力；主要内容包括电子技术、数字系统、信号处理、测试技术、仪器仪表、遥感通信等。本书由16篇课文和16篇阅读材料组成，并附有所有课文的参考译文。为了方便教学，本书另配有电子教案，向采纳本书作为教材的教师免费提供。

本书可以作为测控技术与仪器专业的专业英语教材，也可供从事相关专业的工程技术人员学习参考。

图书在版编目(CIP)数据

测控技术与仪器专业英语教程/刘曙光主编. 2版. 北京：电子工业出版社，2009.8
高等学校专业英语教材
ISBN 978-7-121-09329-6

Ⅰ. 测… Ⅱ. 刘… Ⅲ. ①测量系统：控制系统—英语—高等学校—教材 ②电子测量设备—英语—高等学校—教材 Ⅳ. H31

中国版本图书馆CIP数据核字(2009)第128790号

策划编辑：杨丽娟
责任编辑：杨丽娟
印　　刷：北京市天竺颖华印刷厂
装　　订：三河市鑫金马印装有限公司
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：787×980　1/16　印张：22.5　字数：505千字
印　　次：2009年8月第1次印刷
印　　数：4 000册　　定价：32.00元

前　　言

随着信息化社会进程的不断加快，越来越多的人们认识到语言已成为获取信息的重要手段。为了更快、更准确地了解本专业的发展动向，应学习和借鉴国外的先进技术和有效管理经验。阅读和翻译专业英语文献资料的能力已经成为高等院校学生及科研工作者所必备的素质之一。

目前，学生们在完成大学公共英语课程的学习之后，要想顺利阅读本专业的英语书刊、文献尚存在着不少困难。教育部颁布的"大学英语教学大纲"把专业英语阅读列为必修课而纳入英语教学计划，强调通过四年的教学使学生达到顺利阅读专业刊物的目的。根据这一精神，我们编写了本书，以满足测控技术与仪器专业学生学习的需要，也可供从事上述专业的工程技术人员学习参考。

本书由16篇课文和16篇阅读材料组成，涉及的内容包括：电子技术、数字系统、信号处理、测试技术、仪器仪表、遥感通信等。课文内容新颖，文体规范，难度适中。为了适应专业英语教学的要求，书中所涉及的内容既对学生所学过的课程进行了必要的覆盖，又有所拓宽和延伸，力求反映测控技术与仪器的现状和发展趋势，既可提高读者英语阅读水平，又能使读者了解学科前沿。为了方便教学，本书另配有电子教案，向采纳本书作为教材的教师免费提供（获取方式：登录电子工业出版社华信教育资源网 www.hxedu.com.cn 或电话联系 010-88254537 获得）。

本书由刘曙光、曹军义主编，参加编写的有费佩燕、屈萍鸽、侯志敏、郭亚青、张莉。由于水平有限，书中难免有不足和欠妥之处，恳请广大读者批评指正。

编　　者

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Lesson 1 Periodic Signals

1.1 Time-Domain Description

The fact that 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. 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 main rectifier prior to smoothing does repeat itself very many times, and it analysis as a strictly periodic signal yields valuable results.[1] In other cases, such as the electrocardiogram, the waveform is quasi-periodic and may usefully be treated as truly periodic for some purpose. It is worth nothing 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 periodic and random ones.

A complete time-domain description of such a signal involves specifying its value precise 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 as a polynomial expansion, a Taylor series, or a Fourier series may be used. A polynomial of order n having the form

$$f(t) = a_0 + a_1 t + a_2 t^2 + a_3 t^3 + \cdots + a_n t^n \tag{1-1}$$

may be used to fit the actual curve at $(n+1)$ arbitrary points. The accuracy of fit will generally improve as number of polynomial terms increases. It should also be noted that the error Figure 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 point chosen. The general form of the Taylor series for approximating a function in the region of the point is given by

$$f(t) = f(a) + (t-a) \times \frac{\mathrm{d}f(a)}{\mathrm{d}t} + \frac{(t-a)^2}{2!} \times \frac{\mathrm{d}^2 f(a)}{\mathrm{d}t^2} + \cdots + \frac{(t-a)^n}{n!} \times \frac{\mathrm{d}^n f(a)}{\mathrm{d}t^n} \tag{1-2}$$

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 outside this region, although it may be improved by including additional terms (so long as t lies within the region of convergence of the series).[2] 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. 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.

1.2 Frequency-Domain Description

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.[3] A continuous sinusoidal function ($\sin \omega t$) is thought of as a 'single frequency' wave of frequency radians per second, and the frequency-domain description of a signal involves its breakdown into a number of such basic functions. This is the method of Fourier analysis.

There are a number of reasons why signal representation in terms of a set of component sinusoidal waves occupies such a central role in signal analysis. The suitability of a set of periodic Functions for approximating a signal waveform over an extended interval has already been mentioned, and it will be shown later that the use of such techniques causes the error between the actual signal and its approximation to be minimized in a certain important sense. A further reason why sinusoidal functions are so important in signal analysis is that they occur widely in the physical world and are very susceptible to mathematical treatment; a large and extremely important class of electrical and mechanical systems, known as 'linear systems', responds sinusoidally when driven by a sinusoidal disturbing function of any frequency. All these manifestations of sinusoidal function in the physical world suggest that signal analysis in sinusoidal terms will simplify the problem of relating a signal to underlying physical causes, or to the physical properties of a system or device through which it has passed. Finally, sinusoidal functions form a set of what are called 'orthogonal function', the rather special properties and advantage of which will now be discussed.

1.3 Orthogonal Functions

1.3.1 Vectors and Signals

A discussion of orthogonal functions and of their value for the description of signals may be conveniently introduced by considering the analogy between vectors and signals. A vector is specified both by its magnitude and direction, familiar examples being force and velocity. Suppose we have two $\boldsymbol{V}_1$ and $\boldsymbol{V}_2$; geometrically, we define the component of vector $\boldsymbol{V}_1$ along vector $\boldsymbol{V}_2$ by constructing the perpendicular form the end of $\boldsymbol{V}_1$ onto $\boldsymbol{V}_2$. We then have

$$\boldsymbol{V}_1 = C_{12}\boldsymbol{V}_2 + \boldsymbol{V}_e \tag{1-3}$$

where vector $\boldsymbol{V}_e$ is the error in the approximation. Clearly, this error vector is of minimum length when it is drawn perpendicular to the direction of $\boldsymbol{V}_2$. Thus we say that the component of vector $\boldsymbol{V}_1$ along vector $\boldsymbol{V}_2$ is given by $C_{12}\boldsymbol{V}_2$, where C_{12} is chosen such as to make the error vector as small as possible. A familiar case of an orthogonal vector system is the use of three mutually perpendicular axes in co-ordinate geometry.

There basic ideas about the comparison of vectors may be extended to signals. Suppose we wish to approximate a signal $f_1(t)$ by another signal or function $f_2(t)$ over a certain interval $t_1 < t < t_2$; in other words,

$$f_1(t) \approx C_{12} f_2(t) \qquad \text{for} \quad t_1 < t < t_2$$

We wish to choose C_{12} to achieve the best approximation. If we define the error function

$$f_e(t) = f_1(t) - C_{12} f_2(t) \tag{1-4}$$

it might appear at first sight that we should choose C_{12} so as to minimize the average value of $f_e(t)$ over the chosen interval. The disadvantage of such an error criterion is that large positive and negative errors occurring at different instants would tend to cancel each other out. This difficulty is avoided if we choose to minimize the average squared-error, rather than the error itself (this is equivalent to minimizing the square root of the mean-squared error ,or 'r. m. s' error). Denoting the average of $f_e^2(t)$ by ε, we have

$$\varepsilon = \frac{1}{(t_2 - t_1)} \int_{t_1}^{t_2} f_e^2(t)\mathrm{d}t = \frac{1}{(t_2 - t_1)} \int_{t_1}^{t_2} [f_1(t) - C_{12} f_2(t)]^2 \mathrm{d}t \tag{1-5}$$

Differentiating with respect to C_{12} and putting the resulting expression equal to zero gives the value of C_{12} for which is a minimum.[4] Thus

$$\frac{d}{\mathrm{d}C_{12}} \left\{ \frac{1}{(t_2 - t_1)} \int_{t_1}^{t_2} [f_1(t) - C_{12} f_2(t)]^2 \mathrm{d}t \right\} = 0$$

Expanding the bracket and changing the order of integration and differentiating gives

$$C_{12} = \int_{t_1}^{t_2} f_1(t) f_2(t)\mathrm{d}t \bigg/ \int_{t_1}^{t_2} f_2^2(t)\mathrm{d}t \tag{1-6}$$

1.3.2 Signal description by sets of orthogonal function

Suppose that we have approximated a signal $f_1(t)$ over a certain interval by the function $f_2(t)$ so that the mean square error is minimized, but that we now wish to improve the approximation. It will be demonstrated that a very attractive approach is to express the signal in terms of a set of function $f_2(t)$, $f_3(t)$, $f_4(t)$, etc. , which are mutually orthogonal. Suppose the initial approximation is

$$f_1(t) \approx C_{12} f_2(t) \tag{1-7}$$

and that the error is further reduced by putting

$$f_1(t) \approx C_{12} f_2(t) + C_{13} f_3(t) \tag{1-8}$$

where $f_2(t)$ and $f_3(t)$ are orthogonal over the interval of interest. Now that we have incorporated the additional term $C_{13} f_3(t)$, it is interesting to find what the new value of must be in order that the mean square error is again minimized. We now have

$$f_e(t) = f_1(t) - C_{12} f_2(t) - C_{13} f_3(t) \tag{1-9}$$

and the mean square error in the interval $t_1 < t < t_2$ is therefore

$$\varepsilon = \frac{1}{t_2 - t_1} \cdot \int_{t_1}^{t_2} [f_1(t) - C_{12}(t) f_2(t) - C_{13}(t) f_3(t)]^2 \mathrm{d}t \tag{1-10}$$

Differentiating partially with respect to C_{12} to find the value of C_{12} for which the mean square error is again minimized, and changing the order of differentiation and integration, we have again [5]

$$C_{12} = \int_{t_1}^{t_2} f_1(t) f_2(t) \mathrm{d}t \Big/ \int_{t_1}^{t_2} f_2^2(t) \mathrm{d}t \tag{1-11}$$

In order words, the decision to improve the approximation by incorporating an additional term in does not require us to modify the coefficient, provided that $f_3(t)$ is orthogonal to $f_2(t)$ in the chosen time interval. [6] By precisely similar arguments we could show that the value of C_{13} would be unchanged if the signal was to be approximated by $f_3(t)$ alone.

This important result may be extended to cover the representation of a signal in terms of a whole set of orthogonal functions. The value of any coefficient does not depend upon how many functions from the complete set are used in the approximation, and is thus unaltered when further terms are included. The use of a set of orthogonal functions for signal description is analogous to the use of three mutually perpendicular (that is , orthogonal) axes for the description of a vector in three-dimensional space, and gives rise to the notion of a 'signal space'. [8] Accurate signal representation will often require the use of many more than three orthogonal functions, so that we must think of a signal within some interval $t_1 < t < t_2$ as being represented by a point in a multidimensional space.

To summarize, there are a number of sets of orthogonal functions available such as the so-called Legendre polynomials and Walsh functions for the approximate description of signal waveform, of which the sinusoidal set is the most widely used. [9] Sets involving polynomials in t are not by their very nature periodic, but may sensibly be used to describe one cycle (or less) of a periodic waveform; outside the chosen interval, errors between the true signal and its approximation will normally increase rapidly. A description of one cycle of a periodic signal in terms of sinusoidal function will, however, be equally valid for all time because of the every member of the orthogonal.

1.4 The Fourier Series

The basis of the Fourier series is that complex periodic waveform may be analyzed into a number of harmonically related sinusoidal waves which constitute an orthogonal set. If we have a periodic signal $f(t)$ with a period equal to T, then $f(t)$ may be represented by the series

$$f(t) = A_0 + \sum_{n=1}^{\infty} A_n \cos n\omega_1 t + \sum_{n=1}^{\infty} B_n \sin n\omega_1 t \tag{1-12}$$

where $\omega_1 = 2\pi/T$. Thus $f(t)$ is considered to be made up by the addition of a steady level A_0 to a number of sinusoidal and cosinusoidal waves of different frequencies. The lowest of these frequencies is ω_1 (radians per second) and is called the 'fundamental'; waves of this frequency have a period equal to that of the signal. Frequency $2\omega_1$ is called the 'second harmonic', $3\omega_1$ is the 'third harmonic', and so on. Certain restrictions, known as the Dirichlet conditions, must be placed upon $f(t)$ for the above series to be valid. The integral $\int |f(t)| \mathrm{d}t$ over a complete period must be finite, and may not have more than a finite number of discontinuities in any finite interval. Fortunately, these conditions do not exclude any signal waveform of practical interest.

1.4.1 Evaluation of the coefficients

We now turn to the question of evaluating the coefficients A_0, A_n and B_n. Using the minimum square error criterion described in foregoing text, and writing for the sake of convenience, we have

$$\left.\begin{aligned} A_0 &= \frac{1}{2\pi}\int_{-\pi}^{\pi} f(x)\mathrm{d}x \\ A_n &= \frac{1}{\pi}\int_{-\pi}^{\pi} f(x)\cos nx\,\mathrm{d}x \\ B_n &= \frac{1}{\pi}\int_{-\pi}^{\pi} f(x)\sin nx\,\mathrm{d}x \end{aligned}\right\} \tag{1-13}$$

Although in the majority of cases it is convenient for the interval of integration to be symmetrical about the origin, any interval equal in length to one period of the signal waveform may be chosen.

Many waveforms of practical interest are either even or odd functions of time. If $f(t)$ is even then by definition $f(t) = f(-t)$, whereas if it is odd $f(t) = -f(-t)$. If $f(t)$ is even and we multiply it by the odd function $\sin n\omega_1 t$ the result is also odd. Thus the integrand for every B_n is odd. Now when an odd function is integrated over an interval symmetrical about $t=0$, the result is always zero. Hence all the B coefficients are zero and we are left with a series containing only cosines. By similar arguments, if $f(t)$ is odd the A coefficients must be zero and we are left with a sine series. It is indeed intuitively clear that an even function can only be built up from a number of other functions which are themselves even , and vice versa.

We have already seen how the Fourier series is simplified in the case of an even or odd function, by losing either its sine or its cosine terms. A different type of simplification occurs in the case of a waveform possessing what is know as 'half-wave symmetry'. In mathematical terms, half-wave symmetry exists when

$$f(t) = -f(t+T/2) \qquad (1\text{-}14)$$

In other words any two values of the waveform separated by $T/2$ will be equal in magnitude and opposite in sign. Generalizing, only odd harmonics exhibit half-wave symmetry, and therefore a waveform of any complexity which has such symmetry cannot contain even harmonic components. Conversely, a waveform know to contain any second, fourth, or other harmonic components cannot display half-wave symmetry.

Usually, we have always integrated over a complete cycle to derive the coefficients. However in the case of an odd or even function it is sufficient, and often simpler, to integrate over only one half of the cycle and multiply the result by 2. Furthermore if the wave is not only even or odd but also display half-wave symmetry, it is enough to integrate over one quarter of a cycle and multiply by 4. These closer limits are adequate in such cases the function that is being integrated is repetitive, repeating twice within one period when it also exhibit half-wave symmetry.

1.4.2 Choice of time origin, and waveform power

The amount of work involved in calculating the Fourier series coefficients for a particular waveform shape is reduced if the waveform is either even or odd, and this may often be arranged by a judicious choice of time origin (that is, shift of time origin).[10] This shift has therefore merely had the effect of converting a Fourier series containing only sine terms into one containing only cosine terms; the amplitude of a component at any one frequency is, as we would expect, unaltered. For a complicated waveform which is neither even nor odd, it must be expected to include both sine and cosine terms in its Fourier series.

As the time origin of a waveform is shifted, the various sine and cosine coefficients of its Fourier series will change, but the sum of the squares of any two coefficients A_n and B_n will remain constant, which means that the average power of the waveform, a concept familiar to electrical engineers, is unaltered.

The above ideas lead naturally to an alternative trigonometric of the Fourier series. If the two fundamental components of a waveform are

$$A_1 \cos \omega_1 t \quad \text{and} \quad B_1 \sin \omega_1 t$$

their sum may be expressed in an alternative form using trigonometric identities

$$A_1 \cos \omega_1 t + B_1 \sin \omega_1 t = \sqrt{(A_1^2 + B_1^2)} \cos\left(\omega_1 t - \arctan \frac{B_1}{A_1}\right)$$

$$= \sqrt{(A_1^2 + B_1^2)} \sin\left(\omega_1 t + \arctan \frac{B_1}{A_1}\right) \qquad (1\text{-}15)$$

Thus the sine and cosine components at a particular frequency are expressed as a single cosine or sine wave together with a phase shift. If this procedure is applied to all harmonic components of the Fourier series, we get the alternative forms

$$f(t) = A_0 + \sum_{N=1}^{\infty} C_n \cos(n\omega_1 t - \phi_n) \quad \text{or} \quad f(t) = A_0 + \sum_{N=1}^{\infty} C_n \sin(n\omega_1 t + \theta_n) \tag{1-16}$$

where

$$C_n = \sqrt{A_n^2 + B_n^2}, \phi_n = \arctan(B_n/A_n), \theta_n = \arctan(A_n/B_n) \tag{1-17}$$

Finally, we note that sine the mean power represented by any component wave is

$$(A_n^2 + B_n^2)/2 = C_n^2/2 \tag{1-18}$$

and the power represented by the term A_0 is simply A_0^2, the total average waveform power is equal to

$$P = A_0^2 + \frac{1}{2}\sum_{n=1}^{\infty} C_n^2 \tag{1-19}$$

But P may be expressed as the average value over one period of $[f(t)]^2$, using again the convention that is considered to represent a voltage waveform applied across a ohm resistor. Hence

$$P = A_0^2 + \frac{1}{2}\sum_{n=1}^{\infty} C_n^2 = \frac{1}{T}\int_{-T/2}^{T/2} [f(t)]^2 \mathrm{d}t \tag{1-20}$$

This result is a version of a more general one known as Parseval's theorem, and shows that the total waveform power is equal to the sum of the powers represented by its individual Fourier components. It is, however, important to note that this is only true because the various component waves are drawn from an orthogonal set.

Words and Expressions

accuracy [ˈækjurəsi] *n.* 精确性,准确度,精度

amplitude [ˈæmplitjuːd] *n.* 振幅,幅度

aperiodic [ˈeipiəriˈɔdik] *adj.* 非周期的

approach [əˈprəutʃ] *n.* 接[逼]近;近似法[值];途径,方法

approximation [əˌprɔksiˈmeiʃən] *n.* 近似值

arbitrary [ˈɑːbitrəri] *adj.* 任意的

channel [ˈtʃænl] *n.* 信道,频道

coefficient [ˌkəuiˈfiʃənt] *n.* 系数

convergence [kɔnˈvəːdʒəns] *n.* 收敛

conversely [kɔnˈvəːsli] *adj.* 相反的,逆的