电子电信工程专业英语

Fundamentals and New Concepts for Electronics & Telecommunications

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内容简介

本书分为两部分。第一部分介绍了信号分析与处理的基本知识,其中包括数学分析、概率论与随机过程、信号与系统、数字信号处理等有关课程中常用的一些基本概念,对部分词汇作了注释,对文中的部分语句给出了参考译文,适于精读课程使用。

第二部分介绍了无线电通信以及水声工程的一些基本知识,包括无线电通信和现代通信的基本原理、噪声和干扰、水声信号处理以及水下声成像等内容,适于泛读课程使用。

本书适合于电子信息工程、通信工程专业高年级学生使用,对同等水平的科研和工程技术人员也有一定的参考价值。

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前　言

在近年来的专业英语教学实践中，我们发现自己正面临着一种困境：一方面，电子信息技术正在以令世人惊诧的速度发展，大量新技术的出现令人目不暇接，原有教材内容亟待更新；另一方面，多数学生的英语水平普遍较数年前提高很大，原有教材的深度和广度都已不适合现在的教学需要。因此，我们编写了这本专业英语阅读教材。

我们的编写思路是：

一、选择的材料包含电子信息工程专业、通信工程专业的基础内容，突出基本概念、基本公式和定理的说明解释，使学生能够掌握本专业最基本的英文词汇，为将来的研究工作打好基础。

二、参考近年的大量有关技术资料，挑选内容新、范围广、实践性强的材料，提高学生对专业领域新知识的学习兴趣。

本书的大部分内容，在近两年的教学实践中得到了检验，效果良好，现总结编译成书。

本书第一部分和第二部分的第 3 课由姜弢编写，第二部分 1、2 课由田坦编写，第二部分第 4 课由赵春晖编写，第 5 课由徐新盛编写。全书由田坦负责统稿。

在本书的编写过程中，我们得到了电子工程系和水声工程系有关领导的大力支持，在此致谢。此外，朱海峰、程波助教和多名硕士研究生在编写过程中也做出了贡献，在此一并致谢。

由于作者水平有限，书中难免会出现一些错误和不足之处，敬请读者批评指正。

作　者

于哈尔滨工程大学

2001.6

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Part Ⅰ

Part Ⅱ

Part I

Lesson 1 Periodic Signals

1.1 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. 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[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 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 periodic 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

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

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. The general form of the Taylor series for approximating a function in the region of the point is given by

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

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

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 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)[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 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[3]. A continuous sinusoidal function ($\sin \omega t$) is thought of as a 'single frequency' wave of frequency radians/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 signals and vectors. A vector is specified both by its magnitude and direction, familiar examples being force and velocity. Suppose we have two vectors $\boldsymbol{V}_1$ and $\boldsymbol{V}_2$; geometrically, we define the component of vector $\boldsymbol{V}_1$ along vector $\boldsymbol{V}_2$ by constructing the perpendicular from the end of $\boldsymbol{V}_1$ onto $\boldsymbol{V}_2$. We then have

$$V_1 = C_{12} V_2 + V_e \qquad (1-3)$$

where vector 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 V_2. Thus we say that the component of vector V_1 along vector V_2 is given by $C_{12} V_2$, 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

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

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

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

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

$$\varepsilon = \frac{1}{(t_2 - t_1)} \int_{t_1}^{t_2} f_e^2(t) \mathrm{d}t$$

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

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

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

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

$$C_{12} = \int_{t_1}^{t_2} f_1(t)f_2(t)dt \Big/ \int_{t_1}^{t_2} f_2^2(t)dt \qquad (1-6)$$

1.3.2 *Signal description by sets of orthogonal functions*

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 functions $f_2(t), f_3(t), f_4(t)$, etc., which are mutually orthogonal. Suppose the initial approximation is

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

and that the error is further reduced by putting

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

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 C_{12} must be in order that the mean square error is again minimized. We now have

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

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

$$\epsilon = \frac{1}{(t_2 - t_1)}\int_{t_1}^{t_2}[f_1(t) - C_{12}f_2(t) - C_{13}f_3(t)]^2 dt \qquad (1-10)$$

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]

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

In other 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 were 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[7]. 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 functions will, however, be equally valid for all time because of the periodic nature of every member of the orthogonal.

1.4 The Fourier Series

The basis of the Fourier series is that a complex periodic waveform may be analysed 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

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

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 $\int \left| f(t) \right| dt$ 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