V. CHETVERIKOV

DATA PROCESSING FOR CONTROL AND MANAGEMENT

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USSR, 129820, Moscow, I-110, GSP, Pervy Rizhsky Pereulok, 2, Mir Publishers

Printed in the Union of Soviet Socialist Republics

ИЗДАТЕЛЬСТВО · МИР ·

В. Н. ЧЕТВЕРИКОВ

ПРЕОБРАЗОВАНИЕ И ПЕРЕДАЧА ИНФОРМАЦИИ В АСУ

МОСКВА «ВЫСШАЯ ШКОЛА»

V. CHETVERIKOV

DATA PROCESSING FOR CONTROL AND MANAGEMENT

Translated from the Russian
by
B. Kuznetsov

MIR PUBLISHERS · MOSCOW

First published 1977

Revised from the 1974 Russian edition

The Greek Alphabet

Αα	Alpha	Ιι	Iota	Ρρ	Rho
Ββ	Beta	Κϰ	Kappa	Σσ	Sigma
Γγ	Gumma	Λλ	Lambda	Ττ	Tau
Δδ	Delta	Μμ	Mu	Υυ	Upsilon
Εε	Epsilon	Νν	Nu	Φφ	Phi
Ζζ	Zeta	Ξξ	Xi	Χχ	Chi
Ηη	Eta	Οο	Omicron	Ψψ	Psi
Θθϑ	Theta	Ππ	Pi	Ωω	Omega

The Russian Alphabet and Transliteration

Аа	a	Кк	k	Хх	kh
Бб	b	Лл	l	Цц	ts
Вв	v	Мм	m	Чч	ch
Гг	g	Нн	n	Шш	sh
Дд	d	Оо	o	Щщ	shch
Ее	e	Пп	p	Ъъ	"
Ёё	e	Рр	r	Ыы	y
Жж	zh	Сс	s	Ьь	'
Зз	z	Тт	t	Ээ	e
Ии	i	Уу	u	Юю	yu
Йй	y	Фф	f	Яя	ya

На английском языке

CONTENTS

PREFACE

In the design of management information and control systems, a good deal of attention is devoted to what are called peripherals — facilities external to the central processor (or computer), whose purpose is to gather, reduce and convey data from source to sink. Peripherals go a long way towards making a success two-way communication between man and machine in systems like MIS's. This is why so much emphasis is placed in this text on these devices that link together man and computer. Through these devices, man can collect and enter any source data and programs, retrieve results, and address the central processor.

In the general case, a MIS may utilize data on the progress of a production process from automatic transducers. If the transducers generate analog signals, that is, signals continuously varying with time, the need arises to convert these signals to numbers before data can be entered in the central processor. In turn, in order that the numbers generated by the computer can be utilized to run the process, they should first be converted back to appropriate analog signals. These two aspects are dealt with in the last two chapters.

Where users are remote from the central processor of a MIS, resort is made to data communication over various communication channels. The physical channels are not discussed in the book, but ample attention is given to the information aspects of data communication.

The book is largely based on the lectures the author read at the Baumann Higher Technical School in Moscow.

The author wishes to thank the faculty members of the Computer Engineering Chair at the Moscow Institute of Electronic Engineering and Professor K. A. Sannikov, D. Sc. (Tech), for their review of the manuscript and for their suggestions, and also V. N. Kononykhin, V. A. Galkin, A. E. Osminin and G. I. Revunkov for their assistance in the preparation of some sections.

CHAPTER 1

INFORMATION AND ITS MEASURE

1.1. PRESENTATION OF INFORMATION

Information may be defined as any data about an event or an object. The concept of information, as it is understood in cybernetics, is akin to that of reflection dealt with in dialectic materialism. The property of reflection is to be found not only in objects but also in processes; in the latter case it manifests itself as a definite relation, or correspondence, between the states of interacting objects. While philosophers are mainly concerned with qualitative differences between forms of reflection, cyberneticists concentrate on a quantitative treatment.

Whether one deals with correspondence between sensation and reality or between the indication of a voltmeter and the voltage across its terminals, the situations are similar because the former object is reflected, or mapped, into the latter, that is, the latter contains some information about the former. On this basis, we may say that *information* is a mapping of a state of one object into a state of another, provided there is a correspondence between their states. States of an object may be mapped into those of several objects (with a varying degree of accuracy).

It is convenient to use the term *message* for the form in which the information is presented to a communication system. Whatever the contents of a message, it is always conveyed through the system in a form (electrical, aural, light, etc.) called a *signal*.

A signal is always formed when a message is to be conveyed from a sender (source) to a recipient (sink) which are, in the general case, separated in space and time. Therefore, a signal may be defined as a means of conveying information in space and time. However this definition of the signal is purely functional and does not describe it as an object of investigation.

From an analysis of any situation involving the use of signals, we may readily conclude that, although signals are always related to a material object, most of the specific qualities of that object are of minor significance. For example, in learning the contents of a printed text it is unimportant what kind of ink or paper has been used; differences between texts (signals) are above all recognized from differences in letters, that is, states of the objects. Therefore, we may say that signals are not the objects themselves, but their states. A signal is formed by causing the object to change state. To preserve correspondence between a message and a signal, that is, to preserve the possibility of extracting the original message from the signal received, the latter should be formed according to definite rules. Such a transformation is called *coding*, and the related rules are called a *code*.

Leaving out the physical aspects of a signal, we may note that in the transmission of information from source to sink, a number of tasks has to be tackled in order to make information more convenient to process and convey. This point can best be understood from the following example.

Let a source generate messages as code combinations (words) made up of characters (letters and/or numerals) out of an alphabet X having a total of l characters. Each message may be thought of as being a selection a out of an alphabet A. It may be represented by a selection of characters, x, out of the alphabet X. This decomposition of messages into elementary units greatly simplifies the transformation of a message into a signal. Instead of using a long dictionary (an infinite one in the general case) in order to establish correspondence between all selections out of the alphabet A and signals, it will suffice to set up unique correspondence between signals and a limited number x of elementary units out of the alphabet X. Obviously, the number of signals needed to represent such elementary units will be determined by the number l of characters in the alphabet.

Thus, each selection a out of the alphabet A may be represented as a sequence, $a = (x_1, x_2, \ldots, x_n)$, where n is the word length.

To transmit a message a represented by a string of elementary units x_i, signals $z_i(t)$ should be sent into the communication channel consecutively.

In practice, the number of characters, l, in the alphabet X may be fairly great (decimal notation uses ten numerals, Russian alphabet has 32 characters, etc.). That is why resort is made to a further transformation, called coding. By coding, the alphabet X having l characters is replaced by another alphabet, Y, having m characters (where $m < l$). This coding assigns to each elementary unit, or character, x, a certain series of symbols, y.

In the wider sense, coding has to do with the transformation of a message into a signal. In the narrower sense, this is a process by which discrete messages are mapped into a series of agreed symbols.

The reduction in the number of characters in the alphabet entails a reduction in the number of distinct signals, $z_i(t)$, necessary for their transmission, and also eases the limitation imposed on the duration of each signal. As a result, transmission is simpler to organize.

For information to be reliably conveyed in space and time, the signals used must be immune to spatial and temporal variations. Qualitatively, this immunity is specified in relation to specific conditions under which the signal is used. In terms of stability, all signals may be divided into static and dynamic.

Static signals are those which utilize stable states of physical objects (printed texts, states of flip-flops, states of a register, the position of a mechanical element, etc.).

Dynamic signals are those which utilize the time-varying states of physical objects (variations in the electromagnetic field, variations in electric parameters, etc.).

Dynamic signals are mainly used to convey information, and static signals, to store it. This division of functions is not mandatory, however, and dynamic signals may well be used to store information and static signals, to convey it.

As to the structure, signals may be divided into continuous and discrete in terms of function and argument. A signal is *continuous* if all the values it can take form a continuum (examples are continuously varying electric current, voltage, or mechanical displacement). A signal is *discrete* if it is limited to a finite (countable) set of values. Signals may further be classed according to the behaviour of the function and its argument.

A continuous function of a continuous argument. At arbitrary instants of time, the function describing such a signal may take on any value out of an infinite set of values within a finite interval

$$F(t)_{\min} \leqslant F(t) \leqslant F(t)_{\max}$$

No limitations are imposed on the behaviour of the function.

Examples are signals generated by pressure, voltage or position transducers as continually varying voltages or currents.

A continuous function of a discrete argument. The function describing such a signal may take on any value out of a continuous set at predetermined instants of time, $t_k = k\Delta t$, where $k = 0, 1, \ldots, n$.

A discrete function of a continuous argument. At any instant of time, the function describing such a signal may take on any value, $F(t)_j$, out of a finite set. Examples are all scale mechanisms from which data at any instant are read off as numerical values accurate to within the least scale division, ΔF.

A discrete function of a discrete argument. The function describing such a signal may take on any value out of a finite set at predetermined discrete instant of time, $F(\Delta t_k)_j$. Such signals are generated by digital computers or clock-controlled digital devices.

Data processing systems usually employ both continuous and discrete signals and have, therefore, to resort to conversions from continuous to discrete form and/or back.

1.2. SAMPLING, QUANTIZING AND CODING

In most cases, information about a physical process is generated by appropriate transducers as signals (variables) which are continuous functions of time, $F(t)$. Before such a continuous (or analog) signal can be presented to a discrete (or digital) communication system or a digital computer, the continuous waveform should be converted to a discrete or digital waveform. This can be done in analog-to-digital converters by techniques known as *sampling* (or time quantization) and *quantizing* (or amplitude quantization). It is possible to apply sampling and quantizing separately and together.

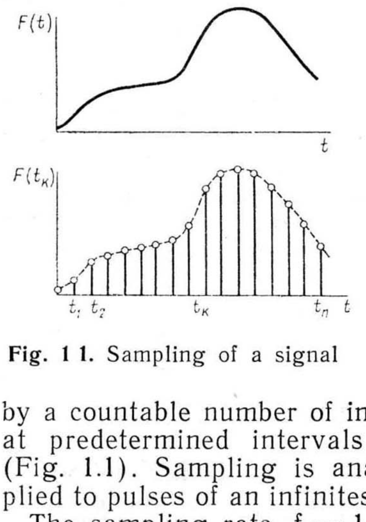

Fig. 1.1. Sampling of a signal

Sampling. In this case, a continuous function, $F(t)$, having an infinite number of values, is replaced by a countable number of instantaneous values, or samples, taken at predetermined intervals (equally or unequally spaced), Δt (Fig. 1.1). Sampling is analogous to amplitude modulation applied to pulses of an infinitesimal duration.

The sampling rate, $f = 1/\Delta t$, is selected so as to facilitate the subsequent reconstruction of the original continuous signal from its samples taken at instants t_k with sufficient accuracy. Generally, an arbitrary continuous function, $F(t)$, can be faithfully reproduced within a finite time interval T, only if samples of this function are taken at all points of this interval, that is, if there is an infinite number of samples spaced an infinitesimal interval apart.

Reconstruction of a continuous function from a finite number of samples within a finite time interval T produces an error determined by both the number of samples taken within that interval (or the sampling rate) and the method of interpolation adopted. There is, however, a class of functions which can faithfully be reconstructed from a finite number of samples. This is true, for example, of bandlimited functions.

The sampling rate for bandlimited functions can be found on the basis of V. A. Kotelnikov's sampling theorem * which states: *If a continuous time function $F(t)$ contains no frequencies higher than f_s, it is completely determined by giving its instantaneous values, $F(k\Delta t)$, at a series of instants spaced $\Delta t = 1/(2f_s)$ apart.* Here, $k = 0, 1, 2, \ldots, n$ and f_s is the highest frequency present.

By this theorem, any function $F(t)$ extending from 0 to f_s may be expanded into a series such that

$$F(t) = \sum_{-\infty}^{+\infty} F(k\Delta t) \frac{\sin \omega_s (t - k\Delta t)}{\omega_s (t - k\Delta t)} \tag{1.1}$$

where $\omega_s = 2\pi f_s$.

It follows from Eq. (1.1) that any bandlimited function $F(t)$ may be represented by an infinite sum in which each term is expressed by a function of the form

$$z = y(\sin x)/x$$

where

$$y = F(k\Delta t)$$

and

$$x = \omega_s (t - k\Delta t)$$

(Fig. 1.2), and differs from the other terms in amplitude and time shift (phase). The functions defining $\sin x/x$ at sampling instants, that is, at $t = k\Delta t$, take on maximum values equal to unity. On the other hand, the sum, Eq. (1.1), at each kth instant is defined by only one kth term because all other terms vanish at that instant (Fig. 1.3). Over the interval Δt the reconstructed function is defined by all terms.

* Known as Shannon's sampling theorem in the American literature. — Tr.

Thus, a bandlimited function $F(t)$ can be completely specified by giving a finite number of instantaneous values (samples) taken at equally spaced intervals Δt. On the other hand, if we have the numerical values of a function $F(t_k)$ at all sampling points (spaced Δt apart), we can completely reconstruct the original function by adding together functions of the form $z = y\ (\sin x)/x$.

Kotelnikov's theorem applies to a bandlimited function, $F(t)$, that is, a function unlimited in time. In practice, however, one has to deal with physical processes which are time-limited, that is, they have both a beginning and an end. Therefore, functions applicable to them are likewise time-limited. Time-limited functions cannot be bandlimited (that is, their spectral density is non-zero outside the finite interval), which conflicts with Kotelnikov's theorem. The conflict can be resolved, however, by a reasonable assumption, that is, by defining an *effective bandwidth*, that is a frequency range outside which the spectral density falls below some predetermined value. Then Kotelnikov's theorem stating that a function of duration T can be specified by giving a set of $n = 2f_sT$ samples, will hold. However, this does not imply that

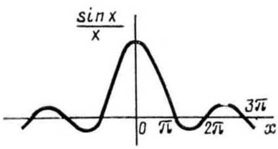

Fig. 1.2. Plot of the function sin x/x

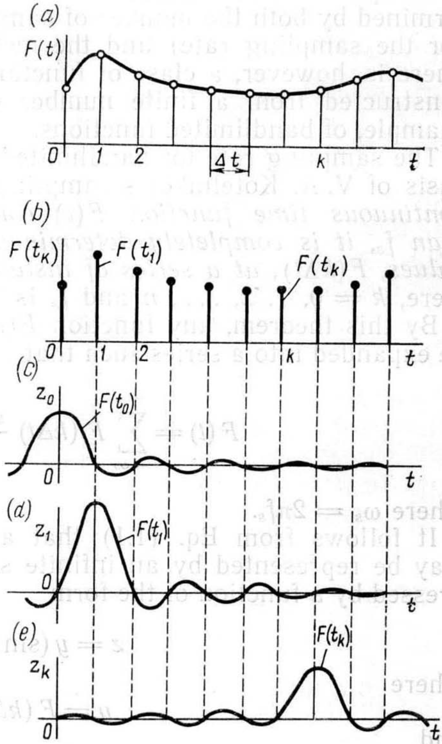

Fig. 1.3. Plots of (*a*) the function $F(t)$; (*b*) of its instantaneous values, $F(t_k)$; and (*c*), (*d*), (*e*) its terms