An Introduction to
Digital Integrated Communications Systems

Hiroshi Inose

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FOREWORD

Professor Hiroshi Inose was a pioneer in digital switching and transmission in days before these arts had become part of the world's communication systems. Today the digital art provides a substantial and growing amount of both transmission and switching, and it is clear that in the long run digital communication will supplant both analog transmission and analog space-division switching.

Inose's own invention of time slot interchange has been essential to this triumph of the digital art. Now, through support derived from the Marconi International Award, which he received for 1976, he has been persuaded to write a book that clearly presents the philosophy and techniques of digital communication and illustrates these with examples of actual and projected standards and systems.

The story he tells is a marvelous one. It is the story of concepts for encoding analog signals into a digital bit stream, for combining several bit streams in time-division multiplex, and for processing, switching and transmitting such signals by using various time-division techniques. All of this has been made practical by a solid-state revolution, including large scale integration (LSI), that has made verv complex digital circuits inexpensive and reliable.

Inose's book explains clearly the overall impact of a single digital art that is replacing the formerly different and divergent arts of transmission and switching. His book is a philosophical work in that it tells why as well as how all of communication is becoming one unified, interconnected digital world. But the book is far more than philosophy. It explains in detail all of the key features of digital communication, including switching and transmission, and tells how these features have a place or are embodied both in standards and in particular systems in many countries.

I don't think that anyone else could have written this book. To do so required someone who has worked closely in the field of communication for many years (as Inose has), who has had experience with actual systems (Inose directed the construction of a digital time-division switching system that worked in the Electrical Engineering Department of the University of

Tokyo for a year and a half starting in 1962), who is thoroughly familiar with computers as well as communication (Inose is Director of the Computer Center at the University of Tokyo), and who is familiar with communications in all countries through attending international congresses and serving on international committees (as Inose is). He has also gained practical experience with American communication through spending various periods at Bell Telephone Laboratories.

In all, there is only one book like this because there is only one Hiroshi Inose to write such a book. I know that the book will find a wide use among communicators, and I hope that it will be used as a text in American universities, where the realities of communication have been sorely neglected.

December, 1978

John R. Pierce
California Institute of Technology

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PREFACE

The motivation and the opportunity for writing this book were provided by the Second Marconi International Fellowship which was awarded to the author by the Marconi Fellowship Council at the Royal Society of Arts, London, on May 6, 1976, H.R.H. the Duke of Edinburgh presiding. As the commissioned work for the Fellowship, the author has chosen to write a book on digital integrated communications. This seemed appropriate because the author's work on this topic was one of the major citations for the award. Thus, the book was originally intended to summarize the studies on digital integrated communications which have been conducted by the author and his associates over a period of twenty-five years.

The contents of the proposed book were later changed to cover more broadly the state of the art of digital integrated communications, with less emphasis on some of the author's own works so as to provide readers with a general perspective on this important emerging technology. The author felt that he could thus produce a more useful book, for to his knowledge no comprehensive book covering the state of the art has yet been written.

Thus, the present book is intended to serve as a textbook or a reference book for those engineers and scientists who have specific interests in digital integrated communications, and in particular, in the switching aspect of this technology. In view of the enormous potential impact of digital integrated communications on various branches of technology and on society as a whole, it is hoped that the book may provide, in chapters 1, 2 and 6 and in the introductory sections of chapters 3, 4 and 5, a general understanding on the state of the art for non-technological readers and for engineers and scientists in broader disciplines of science.

The author's studies on integrated digital communications, carried on over a period of a quarter of a century, have not been accomplished without the support of many others. With utmost gratitude, the author recollects the encouragement and support provided by numerous leaders, including Professor Toshifusa Sakamoto at the University of Tokyo, and Drs. John R. Pierce, W. Deming Lewis and Messrs. W. H. C. Higgins and

H. Earle Vaughan of Bell Telephone Laboratories. Furthermore, the author wishes to mention with thanks that his work could not have been carried through without the dedicated efforts of the author's colleagues and associates including Professors Hiroya Fujisaki, Tadao Saito, Yuichi Yoshida, Yasuhiko Yasuda, Mikio Takagi and Masami Kato and Drs. Zenya Koono, Toshiharu Aoki, Takehisa Tokunaga and Shoichiro Asano.

The author is particularly indebted to Professor John R. Pierce, presently at the California Institute of Technology, for writing a foreword to the book and for perusing the entire manuscript and making detailed corrections. The completion of the book has been made possible by Professor Pierce's enthusiastic encouragement and dedicated assistance. Thanks are also due to Drs. Masami Kato, Sadahiko Kano and Shoichiro Asano for critically reviewing the book chapters 2, 3 and 4 respectively, to Ms. Ruriko Moriya for typing and proof reading, and to Ms. Megumi Shimizu of the University of Tokyo Press for editorial assistance.

The writing of the manuscript took more than two years because of the author's involvement in many different activities in and out of the university. The manuscript was mainly written in three successive summers in Aspen, Colorado, hosted by the Aspen Institute for Humanistic Studies that administers the Marconi International Fellowship. The magnanimous hospitality of Dr. Walter O. Roberts, the trustee and the author's host, remains in the author's memory together with magnificent views of the Colorado mountains. The patience and encouragement of my wife Mariko will also never be forgotten.

Last but not least, the author wishes to acknowledge with thanks the Marconi Fellowship Council; Mrs. Gioia Marconi Braga, Chairperson; and Dr. Walter O. Roberts, Secretary; for their continuing support and encouragement.

January 5, 1979

Hiroshi Inose
University of Tokyo

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Chapter 1 Introduction

1.1 Digital Communications in Retrospect

As is well known, electrical communication in its earliest stage was digital, in the form of telegraphy. Messages were coded into a combination of marks and spaces or the presence or absence of pulses to be transmitted over very noisy and highly attenuating channels. At that time, quite a few people even tried to convert speech signals into interruptions of electric current using vibrating membranes. In terms of modern terminology, this is a crude form of zero crossing modulation, the quality of which is now known to be unsatisfactory for practical use.

At the height of interest in digital communications in the last quarter of the nineteenth century, there were, however, a few people who worked on non-digital techniques. The most outstanding of them was Alexander Graham Bell, who invented the telephone which generates an electric current that changes instantaneously in accordance with speech intensity. "Leave the beaten track occasionally and dive into the woods. You will be certain to find something that you have never seen before." These are the words we can read on a monument dedicated to Bell, signifying his insight into the future of non-digital technology at a time when digital technology was prevalent.

In contrast to the telegraph signal which takes either one of two discrete amplitudes, namely on and off, the telephone signal takes continuous amplitude values analogous to speech and is therefore called an analog signal. The communications systems carrying analog signals emerged rapidly since then owing to a large extent to the invention of the electron tubes that made possible such important functions as the amplification, oscillation, modulation and detection. As the result, analog communication surpassed digital communication in the first quarter of the present century, and has dominated up to the present in telecommunications services. The technology that characterizes analog communication is carrier telephony employing frequency-division-multiplex (FDM) technique. By modulating carriers, a plurality of speech signals are aligned along the

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frequency axis, to be transmitted over a single transmission medium as shown in Fig. 1.1. Carrier telephony by microwaves and coaxial cables now serves some 400 million telephone sets all over the world. Likewise, radio and television broadcasting that disseminates audio and visual signals by modulating radio waves provide programs to the people of the world through more than one billion radio and television receivers. In contrast to the extensive use of electronic means in the transmission aspect of communications technology, however, communication switching technology continued to use electro-mechanical components to connect individual speech signals. This is called space-division-multiplex (SDM) switching and has been predominantly used by the world's major switching systems, although electronic computer technology has been utilized in recent systems for control functions.

In the heyday of analog communications in the 1930's, there were a few people who were interested in digital techniques. Among them was A. H. Reeves who invented a method of speech transmission using a combination of presence and absence of pulses.[1] The method, known as the

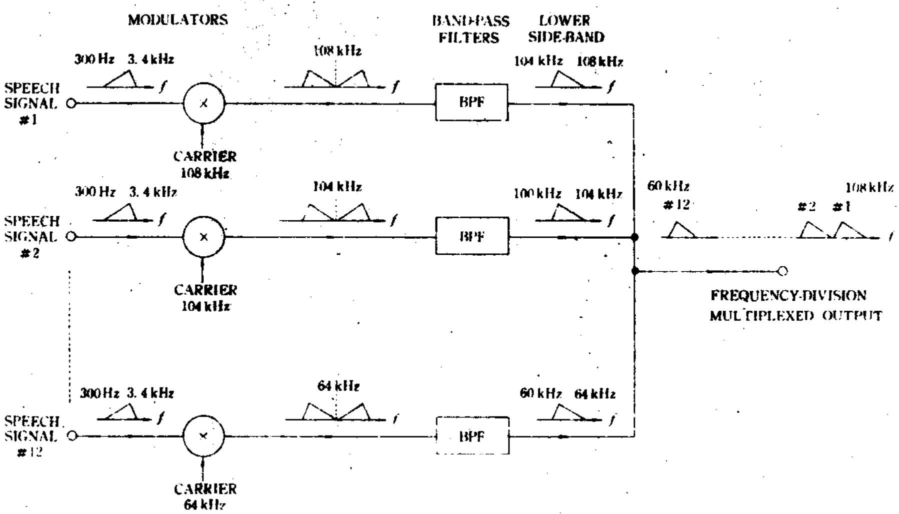

Fig. 1.1 Principle of Frequency-Division-Multiplexing (FDM)

Twelve speech signals, each occupying a bandwidth of slightly less than 4 kHz, are applied to the modulators. The output of a modulator, which is the multiplication of the speech signal and the carrier, consists of an upper side-band and a lower side-band which are centered around the carrier. The outputs are then fed to the band-pass filters that select only the lower side-bands. Twelve lower side-bands when combined occupy the band from 60 kHz to 108 kHz. The combined signal, which is called a group, is further combined with four other groups to form a super group of 60 speech signals. And five super groups form a master group of 300 speech signals. (In the Bell System, ten super groups form a master group of 600 speech signals.) As much as 10,800 speech signals can now be frequency-division-multiplexed and transmitted over a coaxial cable.

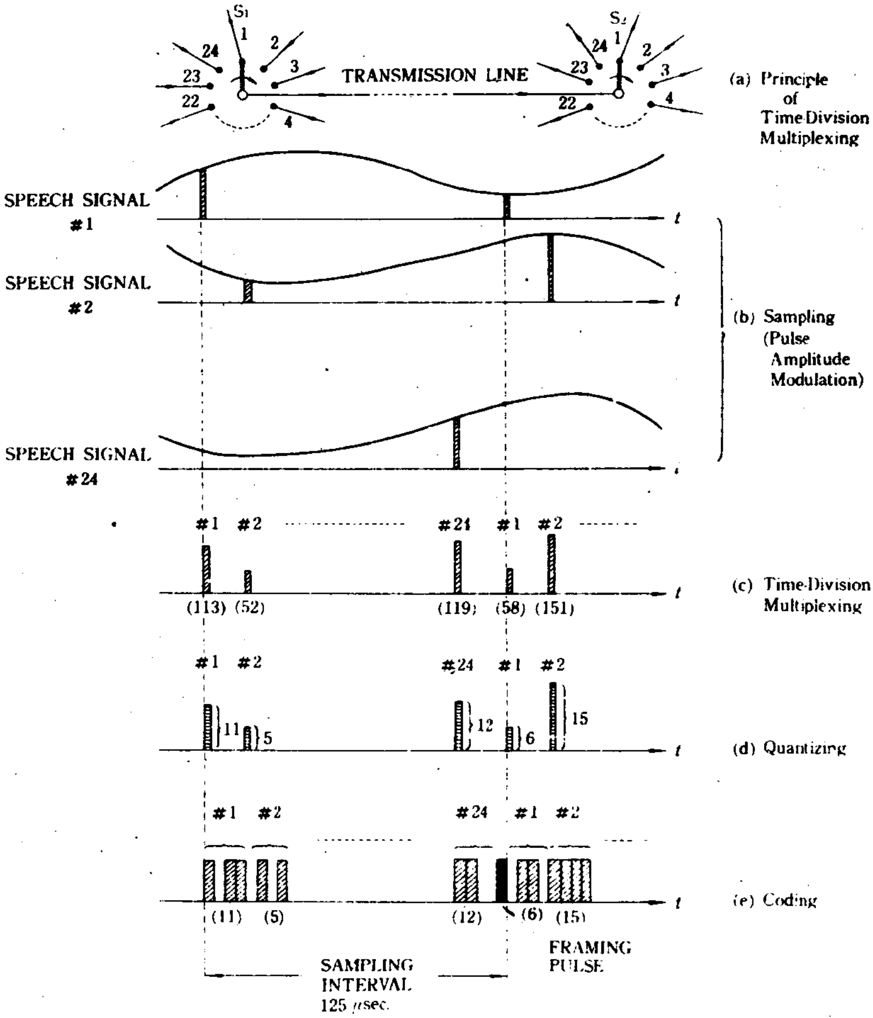

Fig. 1.2 The Principle of Time-Division-Multiplexing (TDM) and Pulse Code Modulation (PCM)

A pair of switches at the transmitting and receiving ends rotate in synchronism across 24 contacts, to each of which 24 individual speech signals are fed (a). The instantaneous amplitudes of the speech signals are repeatedly sampled as the switch rotates (b). The sampled speech signals in the form of the pulse amplitude modulation (PAM) are combined to form time-division-multiplexed (TDM) signals (c). The TDM-PAM signals are then quantized (d) and coded (e). In this illustrative example coding into four bits is shown. The PAM sequences, i.e., 113, 52, 119, 58 and 151 millivolts, are divided by the quantizing step of 10 millivolts into 11, 5, 12, 6 and 15 steps, respectively (d), and are converted into four-bit binary codes, 1011, 0101, 1100, 0110 and 1111 which represent decimal numbers 11, 5, 12, 6 and 15 (e). The truncation error in the quantizing process is called the quantizing noise. In practice, seven-bit coding is generally employed to reduce quantizing noise to less than 1%. The framing pulse shown in (e) is inserted to tell the receiving end when the next sampling interval begins. The number of time slots in a sampling interval ranges from 24 to 5,760.

pulse code modulation or PCM, was characterized by three processes, namely, sampling, quantizing and coding. Sampling is to scan the analog speech signal periodically and obtain instantaneous amplitude. Quantizing is to devide the sampled amplitude into small pieces and thereby convert the continuous property into a discrete property. Coding is to count the number of the small pieces and generate a combination of pulses representing the number. This may be considered as a sophistication of the earlier attempt, the zero crossing technique. Pulse code modulation, however, is superior in that it retains the instantaneous amplitude information which is lost in zero crossing. As is shown in Fig. 1.2, another technique, known as time-division-multiplexing (TDM), is generally combined with pulse code modulation to transmit a plurality of speech signals over a common transmission medium. The period of sampling, which is typically 125 microseconds, is divided into a number of time slots, and each of the encoded speech signals is carried within each of the time slots.

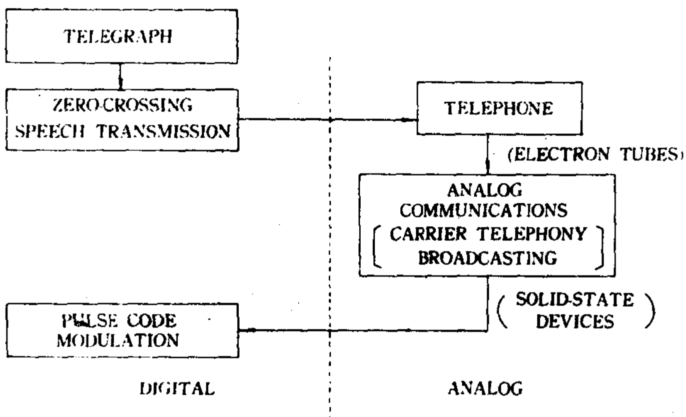

Fig. 1.3 Digital Communications in Retrospect

One of the benefits of pulse code modulation is its immunity to noise. By using a number of simple devices called regenerative repeaters, each of which identifies the presence or absence of received pulses and then reshapes the pulses to be transmitted, the original speech quality can be retained even when transmitted through noisy channels over long distances. The superiority of pulse code modulation over other existing techniques was later proven theoretically to be the most efficient yet physically realizable method of transmitting signals that can attain a transmission rate close to the theoretical limit.(2) The inscription on Bell's monument was again valid.

Because of the complexity of the circuitry that performs sampling, quantizing and coding, practical use of pulse code modulation was delayed until the late 1950's. The invention of transistors and the remarkable progress in solid state devices and circuitry that followed played a decisive role in bringing pulse code modulation into practical existence. Now pulse code modulation transmission systems are rapidly implemented the world over. The number of time slots ranges from 24 to 5,760 depending upon the types of transmission media being employed. As is illustrated in Fig. 1.3, transmission technology is now turning from analog to digital world.

1.2 PCM Integrated Communications

Integration of various aspects of human activities has been a major motive force in the formation of our society. Transportation networks, electric power networks and communications networks, among other things, have been developed to help integrate almost all social and economic activities world wide. Strangely enough if we look into the communications technology that plays a vital role in society's integration, the integration of its major functions, transmission and switching, was not accomplished until recently. Frequency-division-multiplex (FDM) analog systems have been extensively used for transmission by microwaves and cables. Space-division-multiplex (SDM) switching systems that employ mechanical switches to carry individual speech have been dominant. As

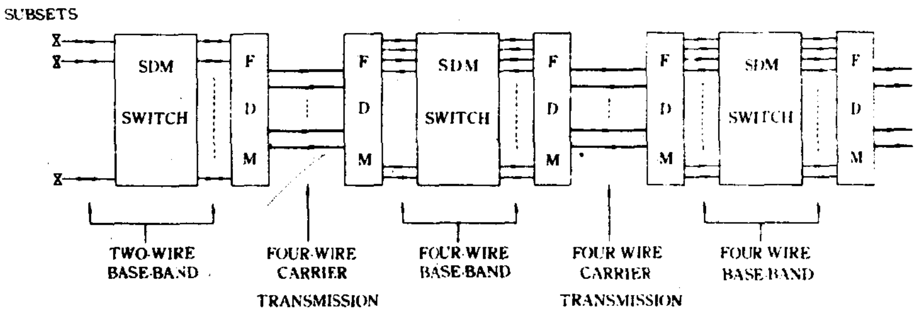

Fig. 1.4 Network with FDM Transmission and SDM Switching

When frequency-division-multiplex (FDM) is used for transmission, modulated and multiplexed speech signals have to be demultiplexed and demodulated before being switched by a space-division-multiplex (SDM) switch, and, after switching, they have to be modulated and multiplexed to be transmitted. The repetition of the procedure results in an accumulation of noise caused by the modulation and demodulation, in addition to increasing the cost. A pair of wires from a subset transmits the speech signal two ways, and hence are called the two-wire base-band lines. After being switched the speech signals are modulated and frequency-division-multiplexed at the transmitting end. The carrier transmission path is one-way so that two pairs of wires are required to transmit and receive a signal. Such a path is called a four-wire path. At the receiving end the speech signals are demultiplexed and demodulated and enter the SDM switch over four-wire base-band trunks.

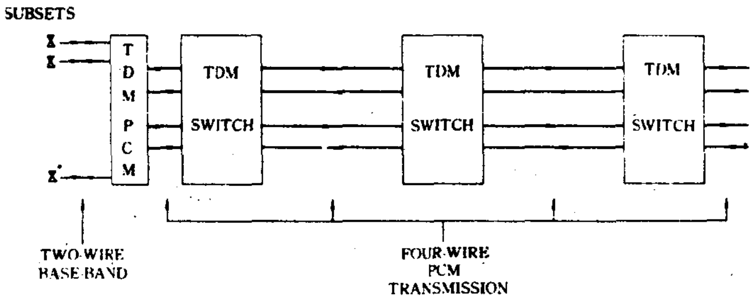

Fig. 1.5 Network with TDM-PCM Transmission and TDM Switching
When time-division-multiplex (TDM) by means of pulse code modulation (PCM) is used for transmission and switching, the speech signals are encoded and multiplexed at the entrance of the network, and are then transmitted and switched, without any decoding or demultiplexing, all through the network until they reach the exit of the network, where they are demultiplexed and decoded. Except for the two-wire base-band lines that connect the subsets to the TDM-PCM terminal, all the transmission paths are four-wire PCM buses.

the result, at the input of each of the switching nodes of the network, SDM signals have had to be demultiplexed and demodulated to provide individual speech signals to SDM switches, and at the output, the individual speech signals have had to be modulated and multiplexed in order to be transmitted by an FDM link, as shown in Fig. 1.4. This is not only costly but also results in the accumulation of noise introduced in each of the modulation and demodulation procedures.

It was the late 1950's when the concept now known as PCM integrated communications emerged. When proposed by Earle Vaughan in his experimental system called ESSEX,[3] the concept was acknowledged world-wide to be revolutionary in the sense that the two separately developed functions of telecommunications, transmission and switching, were integrated by employing PCM. In this system, as shown in Fig. 1.5, speech signals are coded and time-division-multiplexed at the points of origin, and demultiplexed and decoded only at the points of destination. In other words, the speech signals in coded and time-division-multiplexed form are not only transmitted over the links but are also switched at the nodes without any change in their form. This prevents the quality of the speech signal from being degraded, no matter how many links and nodes are involved in its path. This arrangement also drastically cuts down the cost of the network by avoiding modulation and demodulation at each link and by replacing the inefficient space-division-switching at each node.

Since then, a number of significant contributions have been made in the application of this concept, including the time slot interchange technique that makes it possible to switch the speech signal from one time slot to another by means of temporary memories. The fact that PCM switching

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