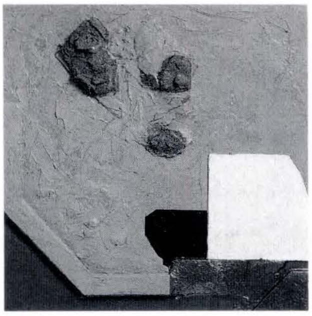
ISSN 0016-2523

# FUJITSU SCIENTIFIC & TECHNICAL JOURNAL

## Summer 1992 VOL. 28, NO. 2

Special Issue on Telecommunications

FUJITSU Sci. Tech. J., **28**, 2, pp. 123-297, Kawasaki, Summer, 1992



**The Issue's Cover** :
ZONA CONFINARIA (Boundary Zone)
by Junroh MANABE

FUJITSU Scientific & Technical Journal is published quarterly by FUJITSU LIMITED of Japan to report the results of research conducted by FUJITSU LIMITED, FUJITSU LABORATORIES LTD., and their associated companies in communications, electronics, and related fields. It is the publisher's intent that FSTJ will promote the international exchange of such information, and we encourage the distribution of FSTJ on an exchange basis. All correspondence concerning the exchange of periodicals should be addressed to the editor.

FSTJ can be purchased through KINOKUNIYA COMPANY LTD., 38-1, Sakuragaoka 5-Chome, Setagaya-ku, Tokyo 156, Japan, ( Telephone : +81-3-3439-0162, Facsimile : +81-3-3706-7479 ).

The price is US$7.00 per copy, excluding postage.



FUJITSU LIMITED 1015 Kamikodanaka, Nakahara-ku,
Kawasaki 211, Japan
Cable Address : FUJITSULIMITED KAWASAKI
Telephone : +81-44-777-1111
Facsimile : +81-44-754-3562

Printed by MIZUNO PRITECH Co., Ltd. in Japan

FUJITSU

SCIENTIFIC & TECHNICAL JOURNAL

Summer 1992 VOL. 28, NO. 2

Special Issue on Telecommunications

CONTENTS

**Featuring Papers**

## Papers

Fujitsu corporate pavilion

# A World Beyond Borders! AT TELECOM 91

There are numerous "borders" in our daily lives — time, distance, language, experience and the interface between man and machine. These borders hinder both business and our personal activities. At Fujitsu, we believe that we can help build a new world goes beyond these borders with telecommunications and information processing technologies.

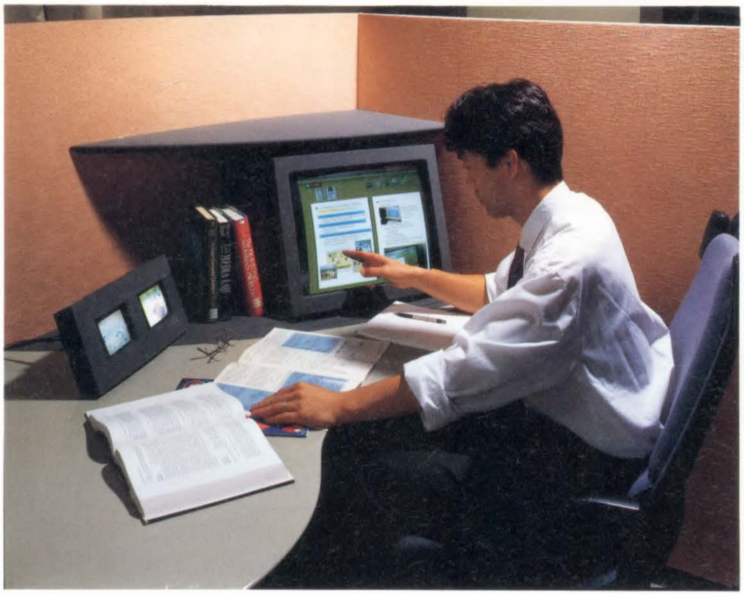

Fujitsu booth in Japan pavilion

FETEX-150
B-ISDN Switcing
System



Ultra-long Distance,
Huge Capacity
Lightwave
Transmission

Monster
(Multimedia Oriented Super
Terminal)



UDC 621.395.345
FUJITSU Sci. Tech. J., **28**, 2, pp. 132-140(1992)

## ATM Switching Technologies

• Kazuo Hajikano • Tetsuhiro Nomura • Koso Murakami

This paper describes the ATM switching architectures and service-specific functions enabling the ATM network to provide multimedia communication. After reviewing various switching architecture, ATM switching architecture based on the self-routing principle, named Multi-Stage Self-Routing (MSSR) was selected. This paper describes the principle, the configuration, and evaluations of MSSR, and the Bi-CMOS switch LSI for MSSR. Service-specific functions for connectionless data transfer and B-ISDN service trial switching systems using MSSR are also described.

UDC 621.395.74
FUJITSU Sci. Tech. J., **28**, 2, pp. 141-149(1992)

## ATM Transmission Systems Technologies

• Kazuo Iguchi • Hirohisa Gambe • Kazuo Yamaguchi

ATM is the key to creating the next generation of telecommunication networks, or B-ISDN. This paper discusses the architecture of B-ISDN and the required network elements from the viewpoint of the transmission network. Flexibility conforming to B-ISDN standards is most important for the introductory phase of B-ISDN. Also, the economical aspect should be considered in the spread phase. The architectures of the trunk network, subscriber access network, and customer premises network, and their required performance were studied. This paper discusses the feasibility of the studies and shows the prototype systems configurations and results. These experiments confirm that the ATM transmission technologies are now applicable to B-ISDN.

UDC 621.395.345.037.3
FUJITSU Sci. Tech. J., **28**, 2, pp. 150-160(1992)

## Enhancement of Digital Switching System FETEX-150 towards Broadband ISDN and Intelligent Network

• Akihiro Sera • Toru Masuda • Yoichi Fujiyama

This paper reports recent enhancements of the digital switching system FETEX-150. First, the architecture enhancement based on the subsystem concept and multiprocessor ring bus is described. Then recent developments in Broadband ISDN (B-ISDN) based on ATM switching technology are described focusing on the service trial system, which has a maximum throughput of 80 Gb/s. Finally, developments in the Intelligent Network (IN) are described, including Intelligent Network 1 (IN/1) based on telecommunication-processor and Advanced Intelligent Network (AIN) utilizing general-purpose computers.

UDC 621.395.74.037.3.072.9
FUJITSU Sci. Tech. J., **28**, 2, pp. 161-171(1992)

## Synchronous Digital Network Systems

• Takeshi Sakai • Yoshikuni Toko • Akira Tokimasa

The Synchronous Digital Hierarchy (SDH) is the world standard recommended by CCITT in 1988. The Synchronous digital network system based on the recommendations introduces flexibility or plane expansion to the existing point-to-point network and has enhanced the versatility to cope with changes in demand and improved network operation and maintenance. Furthermore, it will become a platform for the implementation of broadband ISDN. Fujitsu has developed SDH-based fiber optic and microwave radio transmission systems conforming to the Japan and North America specifications. This paper outlines the characteristics and aims of the SDH, the concept for the new SDH-based synchronous digital network, and the developed SDH systems.

UDC 621.397.2
FUJITSU Sci. Tech. J., **28**, 2, pp. 172-180(1992)

## Multimedia Communication Technology

• Ryusaku Imai • Teruo Tobe • Tsuneo Katsuyama

One of the major developments in the communications and computer industry is the multimedia information environment. Integration of these multimedia technologies with daily activities has become a critical issue. In particular, office communications is one of the most important areas for application of multimedia services. The fundamental aims are natural conversation through telecommunications, information-sharing between geographically separated users, and a user-friendly interface to facilitate use of the new services. This paper introduces and discusses three major research activities on teleconferencing and multimedia user interfaces.

UDC 334.722:621.395.74
FUJITSU Sci. Tech. J., **28**, 2, pp. 181-191(1992)

## Corporate Information Network System: COINS

• Tatsuki Hayashi • Kozo Nakamura • Toshitaka Tsuda

Fujitsu has developed COINS, a system of products that integrates operation and building techniques for use in corporate information network systems. Demand for private ISDN has continued to grow since its release in 1984. As a result, the overall demand for COINS products has also increased.
This paper describes the philosophy and methods used in developing COINS, with particular focus on features for private ISDN, and also overviews the private B-ISDN of the future.

UDC 621.395.345:621.395.74
FUJITSU Sci. Tech. J., **28**, 2, pp. 192-205(1992)

## ISDN PBX for the International Market

• Jun Shimada • Toshiyuki Sato • Toshihito Yamashita

Both public and private ISDN facilities have been incorporated in the FETEX-9600 and FETEX-600 series, and were released in the U.S. and Australian markets in April, 1990. The success of that introduction has proven the FETEX-9600/600 to be a fully ISDN compatible PBX.
This paper discusses the key technologies used in the introduction of ISDN into the FETEX-9600/600 and looks at the future directions of FETEX-9600/600 development as the technologies of computing and telecommunications merge.

UDC 334.722:621.395.74
FUJITSU Sci. Tech. J., **28**, 2, pp. 206-215(1992)

## Local Area Network Systems

• Satoshi Nojima • Takashi Matsuda • Takashi Nakamura

This paper describes the development of Fujitsu's LAN systems. Based on the premises of connecting the products of multiple vendors and multimedia communications, Fujitsu has developed a variety of LAN systems, some based on the standard system and some based on Fujitsu's own systems. The main feature of Fujitsu's activities is the development of a total system, covering transmission equipment, LAN interconnection systems, and maintenance and administration systems. This paper also describes some research and development activities directed towards the next generation of LAN systems.

UDC 621.391.6
FUJITSU Sci. Tech. J., **28**, 2, pp. 216-227(1992)

## Fiber Optic Transmission

• Hideo Kuwahara • Akira Miyauchi • Akira Mitsuhashi

The development of fiber optic transmission systems by Fujitsu is reviewed. The equipment developed for NTT's F-1.6G system, FTM-2.4G system and 1.8 Gb/s 1.55 $\mu$m long span system is first described with the main system parameters. Then research and development activities are introduced, including evolutionally developed optical amplification, ultra-high speed optical transmission towards 10 Gb/s, and coherent lightwave transmission, with a description of their related components. Optical device technologies such as erbium firber doping, wavelength division multiplexing combining the optical signal and pumping power, and $LiNbO_3$ external modulation are also described.

UDC 621.391.8.037.3
FUJITSU Sci. Tech. J., **28**, 2, pp. 228-240(1992)

## Digital Signal Processing Technology for Communications

• Kiichi Matsuda • Yoshitsugu Nishizawa • Fumio Amano

This paper describes the history, major techniques, and developing trends in video and audio signal processing technology in the field of communications. Communications via transmission lines and wireless systems are discussed. Communications via digital storage media, which has recently attracted much interest, is also dealt with in this paper. Topics related to ATM, expected to become the next-generation transmission method, are also discussed for both video and audio.

UDC 621.395.74:681.3.06
FUJITSU Sci. Tech. J., **28**, 2, pp. 241-246(1992)

## Globalization of Software Development for Reliable Telecommunications Systems

• Tadamichi Suzuki • Katsuyoshi Konishi • Kiyoh Nakamura

The users of a globalized communication network service have diverse needs. Meeting these needs is the single most important requirement for the success of globalized software development.

This paper discusses methods of developing reliable software on an international scale for telecommunications systems such as switching systems.

Software development support functions such as Computer Aided Software Engineering (CASE) technologies have been used to assure high reliability of telecommunications systems. These functions need to be further developed and made more available to users and programmers around the world.

UDC 621.382:621.395.4
FUJITSU Sci. Tech. J., **28**, 2, pp. 247-259(1992)

## Integrated Circuits for Digital Transmission Systems

• Norio Ueno • Takashi Touge • Kazuo Yamaguchi

Integrated circuits will be a key element in the next generation of digital transmission systems. These systems must be fast, flexible, compact, cost effective, and highly reliable. Also, the power efficiency of these systems must be maximized to compensate for the increased power consumption and consequent increase in heat output that are associated with an increase in transmission speed. One of the most effective ways to achieve such a system is to use advanced integrated circuit technology.

This paper looks at Fujitsu's latest integrated circuits for digital transmission systems, focusing on N-ISDN/B-ISDN transmission LSIs and gigabit optical regenerator ICs.

UDC 621.395.74.072.9
FUJITSU Sci. Tech. J., **28**, 2, pp. 260-271(1992)

## Flexible Management and Control of Transmission Network Based on SDH

• Takafumi Chujo • Hiroaki Komine • Keiji Miyazaki

The installment of Synchronous Digital Hierarchy (SDH) systems has already begun in various parts of the world. Now that the technologies for initial implementation have been established, path management and control are the key to fully exploit the advantages of SDH by improving the reliability, economical efficiency, and serviceability of networks. After describing the state-of-the-art technologies for SDH implementation, this paper describes a flexible path management and control system. Then, network restoration is discussed as an example of flexible path management and control, and new distributed restoration techniques are proposed. The applicability of the proposed technique is discussed, based on the results of computer simulation.

UDC 334.722:621.395.74
FUJITSU Sci. Tech. J., **28**, 2, pp. 272-278(1992)

## Intelligent Network Architecture and Service for Private Network

• Moo Wan Kim • Akira Hakata • Norihiro Aritaka

This paper proposes a new network architecture and services for the private network. First the requirements for the private network are described, then a new network architecture which has three planes (physical, logical, and service) is proposed. The prototype system, and software configuration and services based on the proposed architecture are then described. Also, the effectiveness of AI technology for realizing the advanced communication services is demonstrated with the AI secretary service, which can handle incoming calls like a human secretary.

UDC 621.391.5:621.395.721.1
FUJITSU Sci. Tech. J., **28**, 2, pp. 279-288(1992)

## The Development of Optical Fibers for Subscriber Loops

• Shigeyuki Unagami • Hiroshi Ogasawara

This paper describes an optical fiber system that has been tailored to applications in the subscriber loop network. The paper discusses the basic concepts of using the system, based on the authors' recent studies of the Fiber In The Loop (FITL). The paper identifies the key points of the system technology, the device technology, and power feeding. The paper compares the network architecture and several transmission schemes, and outlines the progress being made in developing devices used in the subscriber systems. The paper also analyzes one approach designed for the U.S. market, in which the double active star network topology is applied. In addition, the paper discusses the cost parity with copper and the use of a short wavelength laser diode.

UDC 535.14:535.417
FUJITSU Sci. Tech. J., **28**, 2, pp. 289-297(1992)

## Quantum Noise Suppression in an Optical Interferometric System Using Optical Squeezing

• Masataka Shirasaki • Hermann A. Haus • Keren Bergman
• Christopher R. Doerr

A nonlinear interferometric optical system that operates with quantum noise below the shot noise limit is proposed, analyzed, and confirmed experimentally. Sub-shot noise performance cannot be attained with conventional linear optical systems. The proposed system uses optical squeezing produced through a new scheme, that is simple and stable. This noise suppression scheme can be applied directly to a variety of optical interferometric measurements which limit enables, detection of a signal with less noise than the shot noise.

Preface

# Special Issue on Telecommunications

● **Ryoichi Sugioka**
**Executive Director**

Half decade has passed since Communications was featured in the September 1986 issue of FUJITSU Scientific & Technical Journal. During this period, communications technology has undergone dramatic change. This has motivated us to organize another Special Issue on Telecommunications at this time.

Rapidly diminishing political and economic barriers are speeding up international coordination and cooperation on an increased scale, which brings to us bright hopes for a peaceful and prosperous world towards the 21st century. No doubt that the globalization of telecommunications and broadcast networks are playing a major role in accelerating this trend.

At the same time, we are confronted with the challenging task of preserving our planet's resources and environment to ensure quality life for future generations. With such tasks of a global scale to be overcome, **"exchange of information"** is expected to bear increasingly significant roles, whereas **"exchange of goods"** used to be the central activities in social and economic life of the past. Continuing efforts in advancing telecommunications technologies are needed to lay a sound infrastracture for the future.

Until recently, telecommunications network was optimized in terms of voice communications, mainly telephone. As the range of day-to-day communications activities has broadened, however, this approach is no longer sufficient. With technical advance now enabling individuals to use powerful new information processing capabilities with ease, a new approach is needed–one that takes advantage of the continuing trend toward "downsizing" and techniques that enable distributed proccessing functions to be combined for increased functionality and widespread use. The success of multimedia processing made possible by the foregoing developments depends, however, on how much networking capabilities can be enhanced and on how network bottlenecks can be resolved. This will require a higher-speed, widerband infrastructure flexible enough to handle multimedia information at the users' option, and at a lower per-bit cost.

Ever aware of the above trends and the growing need for such new technology as pulse-code modulation, digital exchanges, fiber optics communication, and ISDN, Fujitsu has invested resources toward accelerating these trends. Among the more recent developments are high-speed fiber optics communication based on the Synchronous Digital Hierarchy (SDH) and broadband ISDNs using ATM.

To this end, corporate reorganization in late 1991 involved restructuring our major framework divisions–switching and transmission–into 2 divisions, a division devoted to infrastructure network technologies and a division devoted to business communication network technologies, which covers end-user oriented technologies. This restructuring is expected to facilitate switching and transmission technology synergy. The roots of this reform go back to the reorganization of the base technology and engineering division in 1985, and now have been extended to cover a scheme of overall system implementation addressing the needs of the times for fully integrated systems.

As stated earlier, telecommunications network architecture is at a major crossroads making it imperative that we also merge information processing and telecommunications technologies. Fujitsu proposes to help direct future trends using its major strengths–telecommunications, information processing, and device technologies–unified under the slogan **"What mankind can dream, technology can deliver,"** which has guided our approach to the 21st century.

The article in this issue will be introducing our goals, how we propose to approach them, and what we have gained thus far. With gratitude and deep appreciation, we here acknowledge our debt to our customers, without whose cooperation and support none of the progress we have made could have been achieved.

UDC 621.391.1.001.1

# Towards the Age of New Telecommunications

• Michio Fujisaki • Hirobumi Takanashi • Akio Moridera

*(Manuscript received April 28, 1992)*

This issue will be introducing you to cutting-edge developments in telecommunications at Fujitsu. This keynote address briefly outlines the guiding principles and corporate policy underlying Fujitsu's R&D work, but leaves it to the articles that follow to provide the technical details.

## 1. Introduction

The telecommunications network has expanded on a global scale, now supporting telephone based communications in almost all corners of the world. The achievements so far has been enabled by a worldwide effort in technological development, consistent investment and reliable operation in the area of switching, transmission, and terminal systems.

The technologies to construct the very basic infrastructure of telecommunications have essentially been established. Telecommunications and broadcast networks can speedily deliver up-to-date information throughout the globe. It would not be an overstatement to say that these technologies have had profound effects on the recent dramatic developments in the international political and economical arena. Given the continuously changing global situation, however, the role of telecommunications is becoming even more important.

Looking to the future, the current capabilities of telecommunications are inadequate in terms of network performance and cost constraints. In order to meet the growing demands for higher performances and expanding applications, further advances in network infrastructure and service functionalities need to be pursued.

Fujitsu has contributed to the construction of today's telecommunications by providing state-of-the-art technologies and highly reliable products. Based on its five decades of experience in telecommunications, Fujitsu will make all out efforts in delivering future telecommunications that would serve as a basic engine for bringing prosperous and comfortable human life in the 21st century.

This issue provides a glimpse into the latest activities and achievements at Fujitsu, focusing on the development of the state-of-the-art telecommunications technologies.

## 2. Goals for the 21st century

As the 21st century approaches, expectation is mounting towards a universal telecommunication services where communication by anyone, anywhere, at any time, and by any media can be achieved. These new capabilities coupled with the widespreading advanced information processing technologies promise dramatic changes in our daily lives.

Nurturing a new technology and finding practical applications for it requires a finely tuned intuition and a clear understanding of future perspectives and needs. This requires some careful consideration of some of the potential possibilities information communication will offer in the 21st century. The following several paragraphs depict our imagination of how work and life will be conducted in the next century.

The 21st century should see a private telephone for everyone, most likely a portable set providing a flexible and easy-to-use communication means. At the same time, integrated linkage between telecommunications equipment and computers will become an essential element. This will require powerful networking capabilities suppotred by developments in easy-to-use customer equipments, distributed processing, and open systems architecture implementation. The progress of multimedia information processing technologies will make efficient transfer of video and audio information imperative. As information processing becomes increasingly pervasive, the present telecommunications networks could be a bottleneck to further advances in multimedia information processing technologies.

At least one personal computer, workstation, or similar data terminal equipment will be found in every home, controlling household appliances as well as providing information processing and data retrieval necessary for daily life. Routine work will be done, in whole or at least in part, via a broadband multimedia workstation either at home, from the nearest satellite office, or while traveling, diminishing the traditional role of commuting.

In the office, large amounts of information will be moved via a broadband fiber optics network. Access to management support information or data for conducting advanced scientific computations will make corporate activities more creative and efficient than ever. Cooperative work among people separated geographically will become popular, and telecommunications will be required to provide a near perfect transfer of not only raw information but also the atmospheric conditions so that a natural cooperative environment can be created over the network.

Advances in telecommunications have the potential to ease congestion and concentration in the urban areas, to direct traveling more for leisure, enhance environmental protection, and promote energy conservation. The 21st century is also expected to bring about practical applications in machine translation, which combined with advanced communications intelligence will accelerate international cultural exchanges and business interactions.

We are fully commited to provide technologies and products that would make these expectations for the 21st century a reality, and would enrich the lives of all mankind through partnership of people and telecommunication. Building a global networking partnership without significant cost penalty, however, will involve the following major prerequisites:

1) Design concepts to materialize revolutionary ideas through graceful migration of technologies.
2) Integration of technological innovations in telecommunication, information processing, and software.
3) Technological development centered on the needs of the individual.
4) International cooperation to create standards applicable worldwide.
5) User oriented regulatory and legislative environment.

Fujitsu intends to approach these tasks using its state-of-the-art telecommunications, information processing and semiconductor technologies, and perhaps more important, through exploitation of fully integrated synergistic capabilities.

## 3. Telecommunication network innovations

### 3.1 Information-oriented network

We believe that a network tailored to the 21st century should be "information-oriented" or "information-driven."

Previous networks have mainly concentrated on simply communicating anytime, anywhere, and with anyone on a person-to-person or point-to-point basis. As personal terminals such as personal computers and workstations come into common use and how information is used becomes critical to larger numbers of people, telecommunications networks must also offer ready access to required information in the optimum mode at the required time. For example, a typical function of telecommunications networks of the future would be to give the user access to information via personal computers or workstations, where the obtained information is processed and stored then re-transferred wherever needed.

Networks tailored to such a mode of information transfer have traditionally been optimized and made available through different approaches from conventional telephone network. The future network, however, should be integrated for various types of multimedia information to effectively support variety of personal activities. As the need to access information without considering its physical location increases, networks should also support directories and associated resources. Given this background, such multimedia information processing networks, should be called "information-oriented" or "information-driven."

### 3.2 New requirements for networks

Information-oriented networks must meet some of typical technical requirements:

1) Low cost and volume-sensitive billing resulting from the integration of backbone networks using Asynchronous Transfer Mode (ATM), and cuts in physical distances and in the per-bit cost derived from advances in fiber optics transmission technologies (extended repeater spans by optical amplification and coherent transmission).
2) Enhanced compatibility with multimedia information.
3) Support of diversified user interfaces enabled by integrated backbone networks for services such as Frame Relay, Switched Multi-megabit Data Service (SMDS), ATM, and ISDN.
4) Intelligent Network (IN) facilities to provide a ready solution to enhanced services and added personal communication capabilities.
5) Networking technologies to meet wireless access by portable telephones or terminals.
6) Private networks that have been built around leased circuits will grow into hybrids that optimize use of the ATM virtual path and other facilities of a public switched network. In this situation, the linkage between the private and public networks, including IN facilities and network management functions, will assume added importance.
7) New services, such as point-to-multipoint communication, in which multiple information sources are accessed, will also add to the existing repertoire of communication services, such as point-to-point communication. Broadcast applications will also be highlighted, making the merger of broadcasting and telecommunications a likely scenario.

### 3.3 Enhancement of telecommunication technologies

Enhancement of telecommunication technologies will be essential for achieving the goals above. To this end, Fujitsu has been working on the following key technologies:

1) Broadband ISDN (B-ISDN)

Genuine multimedia and economic data communication.

2) Intelligent network

Enhanced networking facilities resulting from efficient implementation of new services and computing capabilities.

3) Personal communication

Creation of a comfortable communication environment, realization of "anytime, anywhere" communication, and support of transmission of "information of any kind".

4) Global communication

Economization on global communication to cut relatively expencive international communication costs and structuring of R&D environments which promote international interaction.

## 4. Fujitsu's conceptual approaches to technological development

Summarized below are Fujitsu's conceptual approaches to realizing the above technological development goals:

First, Fujitsu stresses the importance of maintaining a clear vision of its role in pursuing its R&D activities. Fujitsu's slogan, "What mankind can dream, technology can deliver." illustrates Fujitsu's policy of exploring new technologies which will provide the foundation and stimulus, for further development and spin-off technologies. For example, Fujitsu's present emphasis on high-tech areas such as high-speed fiber optics communication and B-ISDNs was motivated by recognition of the immense effects past achievements have had on technological evolution. Fujitsu aims to establish a framework for recycling advanced technologies to enable economical development of next-generation networks. Detailed descriptions of such technological accomplishments are given in the articles that follow.

Second, since its beginnings as a communications equipment manufacturer, Fujitsu has expanded its line of products to cover information processing equipment, devices and semiconductors, from components to systems. In developing and marketing information processing equipment, it is imperative that systems engineering concentrate on building and presenting systems tailored to the needs of the user. Operating under this dictum, Fujitsu has developed products with the needs of end users in mind. As stated earlier, the growing complexity of the modern network is making the support of new media, as well as telephone traffic, increasingly important, and precise determination of the needs of a new evolving class of users is essential. The development of telecommunications infrastructures must be directed at handling a wide assortment of applications and terminals. This need has already been identified in ISDN evolution. Fujitsu operates as a total systems supplier by making the best of its management resources and by linking telecommunications and information processing, from infrastructures to applications. Synergy of such parallel technologies will be the watchword governing Fujitsu's future technological development.

Third, Fujitsu will be stressing the early implementation of key technologies, in particular, those mentioned in Section 3, by taking full advantage of its engineering and personnel assets and its post experience. Emphasis will also be placed on achieving advanced technologies through a synergy of information processing and device technology.

Fourth, and most important, Fujitsu will continue to maintain a customer-first attitude. Customers range from telecommunication network carriers to network users.

A thorough understanding of the needs of users is important. Thus, Fujitsu has concentrated on early implementation of the latest technological concepts and cooperated in field testing with users. For example, in 1985, immediately after the CCITT standards for the ISDN were established, Fujitsu took part in an ISDN trial in Singapore that involved building ISDN capability into existing exchanges, without stopping the service, to test how they would work in the field. This was the world's first standards-based trial. More recently, Fujitsu has been participating in B-ISDN field trials in cooperation with the Bell Operating Companies of North America.

Thus, Fujitsu has been committed to implementing evolving technological concepts by taking full advantage of its management resources. A more detailed insight into some of Fujitsu's approaches is given below.

## 5. Fujitsu's approaches and activities in key technological areas

Fujitsu makes the best of its strengths—full round technologies and systems integration capabilities—to take a lead in both R&D activities and product delivery in the area of information and communication. Some strategic approaches of Fujitsu are illustrated below with reference to actual technological accomplishments to date.

### 5.1 Creation of system solutions through rapid introduction of cutting-edge technologies

In addition to technologies for telecommunication, information processing, and electronic devices, Fujitsu has demonstrated high capability in integrating these technologies through application of advanced software and system design. With this background, Fujitsu holds a good position to offer solutions for future large-scale systems, an important example of which is the B-ISDN.

Implementation of B-ISDN depends on the integration of extensive and diverse technological achievements. These include a total solution to the basic ATM transport scheme, systems development, including network configurations, switching and transmission systems, super-large capacity trunk systems, and fiber optics communication, together with the latest in LSI and software technology.

Before the CCITT discussions started, Fujitsu had already embarked on a B-ISDN R&D program. Completed in 1988, full-scale prototype ATM switching system was the first demonstration of the feasibility of B-ISDN. The system incorporated an SDH-compatible ring access system and a unique switching architecture called Multi-Stage Self-Routing (MSSR). Since then, Fujitsu has continued its efforts toward delivering a commercial model, and delivered a field trial model in end of 1992 (see Fig. 1). At the same time, Fujitsu is now preparing technologies for full scale deployment of B-ISDN, e.g. HEMT LSI that implements a switching highway speed of 9.6 Gb/s and a maximum throughput of 640 Gb/s, high capacity optical communication at 10 Gb/s and above, ATM multiplexers and crossconnects that facilitate efficient ATM trunking.

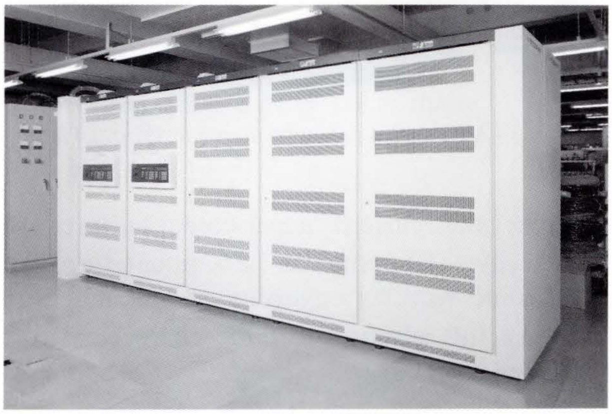


Fig. 1–FETEX-150 B-ISDN switching system.

The development of new applications attractive to users is a prerequisite if B-ISDN is to be universally accepted. In this respect, Fujitsu is jointly exploring new telecommunication services with network carriers as well as developing communication workstations with advanced human interfaces.

### 5.2 Application of "cross-culture" of telecommunication and information processing

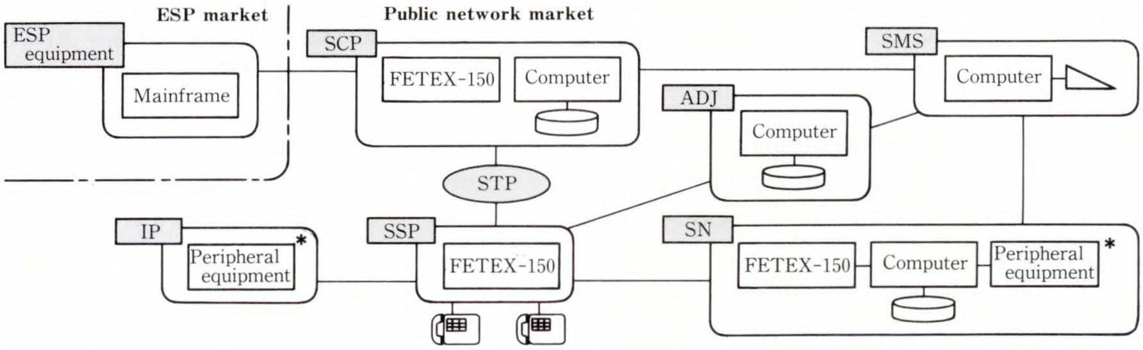
Linkage with information processing will be integral to future telecommunications systems. Fujitsu is thus promoting cultural mix of telecommunications and information processing to produce synergistic effects in the following areas.

1) Intelligent Networks (IN)

For the public network, Fujitsu is developing a system architecture based on the FETEX-150 switching system. Figure 2 shows our vision of IN which encompasses B-ISDN capabilities. For corporate networks, needs for diversified IN services are rapidly emerging, and Fujitsu is developing a system that facilitates tight coupling of PBXs and computers. The Telecommunication Computer System Interface (TCSI) is adopted as a standard interface, which provides new networking capabilities applicable to multipoint node systems.

2) Computer networking

Fujitsu is also working to enhance and expedite interaction among host computers,




ESP : Enhanced Service Provider
IP : Intelligent Peripheral
*: Voice storage and response unit, media convertor, and others
SCP : Service Control Point
STP : Signal Transfer Point
SSP : Service Switching Point
SMS : Service Management System
ADJ : Adjunct
SN : Service Node

Fig. 2–Intelligent network concept.

databases, and workstations, and upgrading the distributed processing environment. For example, Fujitsu has been developing distributed OSs and object-oriented software, while maintaining real-time processing requirements in telecommunications. Switched Multi-megabit Data Service (SMDS) equipment has been developed using ATM switch as the basic platform. This system is expected to provide high speed (up to 45 Mb/s) data communication between LANs, and a trial system has already been delivered to the customers.

### 5.3 Creation of user-friendly communications environments

In the 21st centruy, the availability of sophisticated but easy to use telecommunications services will be the key to a network's success. Based on the past experiences in software and system technologies, Fujitsu is working on developments in the following areas:

1) Universal Personal Telecommunications (UPT) networking

UPT is a new concept in telecommunication service providing full personal mobile communications with personal telephone number for everyone. Fujitsu is developing a vast range of technologies necessary to implement this service. Fujitsu has already developed a 150-cc portable telephone for the current cellular access system and is planning to provide pocket-size phones without delay. For future UPT, flexible wireless access via cellular, satellite or LAN will be coupled with sophisticated networking technologies based on distributed database and call processing. At the same time, careful considerations of privacy must be embedded in the service to protect users from unwanted disturbances. Fujitsu has developed a prototype intelligent electronic secretary system based on advanced artificial intelligence as a case study of this problem.

2) Human interfaces for workstations

To enhance the human interface to the workstation, Fujitsu is working on a prototype Multimedia-oriented superterminal called MONSTER, that uses concepts such as browsing, filing, and indexing to approach and even surpass the current paper-based culture electronically. Hypermedia documentation and electronic working environment are under trial to take full advantage of the characteristics of electronic displays.

a) General view

b) Automated warehouse and carrier

Fig. 3—Automated production line at Oyama Plant.

### 5.4 Global marketing systems

Fujitsu will furnish user-customized systems using its global marketing, R&D, manufacturing, and marketing networks that will involve implementations of the ISDN, SDH, ATM, and other evolving standards. For example, Fujitsu has already developed techniques enabling joint software development between remotely separate design groups.

### 5.5 Fully automated production technologies for high density packaging and customized manufacturing

Fujitsu makes efforts to move ahead not only in technologies but also in manufacturing.

Anticipating growing importance of high density packaging for future personalized terminals and customized low-volume manufacturing, Fujitsu has installed the latest in automated production line at its Oyama Plant, which includes automated warehouses (see Fig. 3).

### 5.6 Efficient linkage with users and outside organization

Fujitsu will strengthen ties with users and outside organizations recognizing that the development of large scale and complex systems/services of the future can be achieved only through cooperations with many capable partners.

1) Expanding scope of telecommunication applications

Close cooperation among end-users, network operators, and manufacturers is becoming essential in this area. To strengthen global ties with users, Fujitsu has joined a number of projects involving specific B-ISDN applications, including hospital systems, electronic publishing systems, and distributed processing computer networks. We are also promoting R&D to make network administration and resources management more efficient and to develop new business opportunities jointly with network operators around the world.

2) Standardization and open systems architecture implementation

In the area of global open system and network architectures, Fujitsu is participating in activities to globally establish standardization through CCITT, TTC, T1, ISO, and other standards organizations. In addition to directly contributing to the drafting of standards, Fujitsu is also contributing to demonstrate the feasibility of the recommendations by early prototyping.

3) Research efforts

Fujitsu has sponsored many exchange and joint research programs with educational and research organizations at home and abroad, to advance basic, interdisciplinary, and evolving technologies.

## 6. Conclusions

As has been described, Fujitsu is working to strengthen its R&D efforts to contribute to building the technological foundation on which the information-oriented community will stand. However, widespread acceptance of new telecommunication networks can be hindered by inertia of old infrastructure and the massive initial investment required. Discovering a framework that significantly diminishes initial cost and promotes smooth migration is of utmost importance for expediting the needed breakthrough. Needless to say, ingenuity and advanced expertise will be essential, and this is where Fujitsu will be concentrating its efforts. At the same time, we will cooperate with the users in preparing social and economical environment for the future telecommunications.

This issue features the articles illustrating some of Fujitsu's R&D accomplishments in this context.

**Michio Fujisaki**
Director
FUJITSU LIMITED

**Hirobumi Takanashi**
Director
FUJITSU LABORATORIES LTD.

**Akio Moridera**
Deputy General Manager
Telecommunication Network Systems Group
FUJITSU LIMITED

UDC 621.395.345

# ATM Switching Technologies

● Kazuo Hajikano ● Tetsuhiro Nomura ● Koso Murakami

*(Manuscript received November 7, 1991)*

This paper describes the ATM switching architectures and service-specific functions enabling the ATM network to provide multimedia communication. After reviewing various switching architecture, ATM switching architecture based on the self-routing principle, named Multi-Stage Self-Routing (MSSR) was selected. This paper describes the principle, the configuration, and evaluations of MSSR, and the Bi-CMOS switch LSI for MSSR. Service-specific functions for connectionless data transfer and B-ISDN service trial switching systems using MSSR are also described.

## 1. Introduction

Broadband ISDNs (B-ISDNs) handle a wide range of advanced multimedia services including high-speed data, high-definition image, and full-motion video. Bit rates range from a few kb/s for telemeter data to several hundred Mb/s for high-quality video. Some services, such as voice, impose severe delay limitations on the network, while other services do not. Some services such as high-quality video are sensitive to errors as well as delay, but a few errors are acceptable for other services. Asynchronous Transfer Mode (ATM) is considered a key technology for B-ISDNs[1]. In an ATM network, all information is segmented into a series of "cells". The number of cells assigned to each block of information is proportional to its bandwidth. Since ATM switching is hardware based, its bandwidth flexibility is great.

At an ATM switching node, a series of cells pertaining to one call arrive in a non-cyclic, or random, manner. The asynchronous arrival of cells causes cell loss due to outlet contention (multiple cells destined for the same outlet arriving simultaneously at different inlets). At the ATM switching nodes, the traffic flow between the input highway and the output highway changes dynamically, and in cases of such instantaneous congestion, the switching architecture must maintain a cell-loss rate below the value determined adequate to maintain the particular service quality.

Various switching architectures have been proposed. They are classified into four categories and reviewed here. After that, the principle, configuration, and evaluation of our original switch architecture, named Multi-Stage Self-Routing (MSSR) are described. Bi-CMOS LSI for this architecture is also described.

ATM network provides the basic transport service common to all services. To support various services, service-specific functions are to be performed at customer equipment and/or equipment in the switching system. For example, to provide a constant bit-rate transfer service for speech, customer equipment requires functions which perform not only cell assembly and disassembly, but also dejittering, timing recovery and compensation for lost cells.

LAN is expected to be used as a network extending over several offices and businesses over a wide area. Inter-LAN connection is a promising service of B-ISDN. Therefore, service-specific functions for offering connectionless data transfer used for inter-LAN connection are discussed in the latter half of this paper.

Finally, our B-ISDN service trial system using MSSR is described. To verify the practical use of B-ISDN, this system is used in trial service with a telephone company and potential end user.



FUJITSU Sci. Tech. J., **28**, 2, pp. 132-140 (June 1992)

## 2. Switching architecture

Before reviewing the various architecture, let us look at the ATM switching requirements. The ATM switch should be designed to satisfy, for example, the following requirements.

1) Service quality

A switching delay of less than 0.4 ms and cell-loss rate of less than $10^{-10}$. Interactive speech traffic requires the shortest delay, possibly up to 30 ms, end-to-end. Data and compressed video traffic require the lowest cell-loss rates ($10^{-9}$, end-to-end). The required service quality is decided on the assumption that the maximum number of relaying nodes is 10. For cell delay, four delay components contribute to the total end-to-end delay: the transmission delay for the end-to-end distance, the time to assemble a cell, the sum of node switching delays, and the time to absorb switching delay fluctuations at the receiver terminal. The maximum transmission delay for 3000 km is about 15 ms. Cell assembly time is 6 ms for 64-kb/s PCM speech in the case of a 53-octet cell according to the CCITT agreement. Assuming that the absorption switching delay fluctuation time is the same as the maximum switching delay, the acceptable switching delay per switching node is about 0.4 ms.

2) Switch capacity range and highway throughput

The highway throughput objective complies with the latest standard trends. 622 Mb/s user-network interface speed is recommended for high-definition video and inter-supercomputer applications. Capacity range should cover up to several thousands or more as well as existing Narrowband ISDN (N-ISDN) switching system. To cover the wide range of capacity, modularity is essential. Modular growth also helps to eliminated dispersions when the system is expanded.

### 2.1 Typical switching architectures

Essentially, the operating speed of an ATM switch depends on the buffer structure because the memory access speed of buffers is dramatically slower than the operating speed of the logic gates. According to the buffer structure, ATM switches are classified into the following categories[2)] (see Fig. 1).

| Category | | Shared buffer | Input buffer | Output buffer | Crosspoint buffer |
|---|---|---|---|---|---|
| | | MUX, Buf., DMUX; Address control | Buf., Buf.; Contention or scheduling | MUX, Buf., Buf. | Buf., Buf., Buf., Buf. |
| Buffer speed | Write | NV | V | NV | V |
| | Read | NV | V | V | V |
| Buffer control | | Multiple FIFO | RIRO | FIFO | FIFO |
| Buffer size | | O (N) | O (N) | O (N) | O ($N^2$) |
| | | Small → | | | large |
| Feature | | Small buffer<br>Complicated buffer control | Low throughput without additional control<br>Low buffer speed | Simple control<br>Bus bottleneck | Simple control<br>Low buffer speed<br>Large buffer |

N : Number of lines  FIFO : First-In First-Out
V : Line speed  RIRO : Random-In Random-Out

Fig. 1—ATM switch categories.

1) Shared buffer switch[3)]

The shared buffer switch uses centrilized control to handle contentions. The size of the shared buffer can be obtained by calculating the convolution of M/D/1 queue length distributions. Sharing memory among all queues (dedicated queue for each outgoing line) significantly reduced the overall buffer size, but it is inevitable to use complicated, high-speed memory control logic. One problem is that one heavily loaded output can affect the other outputs. Another problem is the difficulty of achieving requirements for high fault tolerance.

2) Input buffer switch[4)]

The input buffer switch has a dedicated buffer for each incoming line. If a buffer is simply formed of First-In First-Out (FIFO), the so-called Head-of Line (HOL) effect causes throughput degradations. To prevent this phenomenon, some additional control functions, e.g. scheduling, are required. The size of the input buffer depends on the additional functions, but it is smaller than the buffer for an output buffer switch.

3) Output buffer switch[2)]

The output buffer switch has a dedicated buffer for each outgoing line. In case the input ports are multiplexed and broadcast to output ports, a high-speed bus must be included in the switch. A broadcast functions is easily realized in an output buffer switch and less buffer memories are required than those in the cross-point buffer switch. Higher speed buffer memories are required than those in the crosspoint buffer switch and input buffer switch.

4) Crosspoint buffer switch

The crosspoint buffer switch has the most distributed control scheme and lowest operation speed of the four categories. Each crosspoint buffer is a FIFO. Contention control involves determining from which buffer a cell will be selected. Though the control mechanism is relatively simple, the buffer size is large.

## 3. Multi-Stage Self-Routing (MSSR) switching[5)]

For the switch capacity, the multi-stage architecture excels over all others proposed previously in flexibility. For example, to expand the switch size, a typical self-routing network such as a Banyan network[6)] always requires cutting or rearranging the existing wires. Since multi-stage configurations are building-block architectures, such problems do not happen. Not only to avoid exceeding the limitation of the switch size, but also to obtain the most efficient highway utilization, we take the approach of raising the highway throughput by speeding up the switching operation. It is known that to increase the highway throughput is one of the best ways from the viewpoint of highway efficiency because efficient multiplexing of many burst data is statistically determined[7)]. Crosspoint buffer switching is suitable for high-speed operation, because the required speed of the buffer is the lowest and complicated external control functions are not required.

### 3.1 Principle and configuration

ATM switching requires two major functions: a Virtual Channel Identifier (VCI) conversion function which converts an incoming VCI to an outgoing VCI, and a cell-by-cell switching function. As shown in Fig. 2, the MSSR consists of VCI Converters (VCC) and Self-Routing Modules (SRMs). The MSSR network is constructed by connecting the SRMs in a multi-stage link configuration. This configuration has multiple routes between a first-stage SRM and a third-stage SRM. This allows the traffic flow to be balanced between each link. However, since dynamic path selection on a cell-by-cell basis requires complicated cell sequence integrity mechanisms, a series of cells pertaining to one call are routed along only one path, in principle.

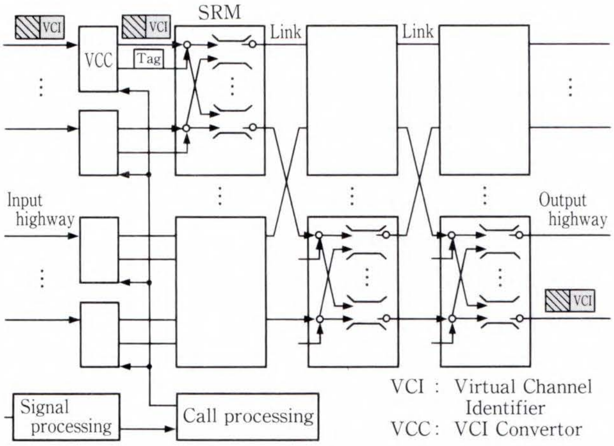



Fig. 2—Multi-stage self-routing switching.

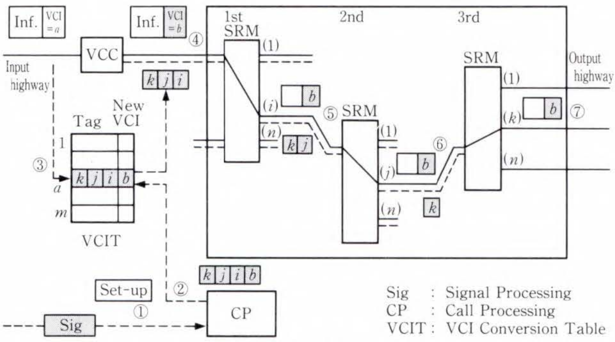



Fig. 3—Switching operation.

Figure 3 shows the operation of the MSSR. In the call set-up phase, call processing determines the output highway, the value of VCI for that output highway, and the path through the switch. It also creates a path information tag that indicates the outlet port number at each SRM that the call should be switched to. Then it writes the tag and the new VCI into the VCI Conversion Table (VCITT) in the VCC. In the information transfer phase, when a cell arrives, the VCC searches the VCIT using the incoming VCI, replaces this VCI with the value in the table, and transfers the cell with the tag to the switching network. In the SRM, the cell is routed to the outlet port pointed to by the tag without external software control. For example, when a cell with tag = $i$ arrives at an SRM, the cell is routed to the ith output port in the SRM.

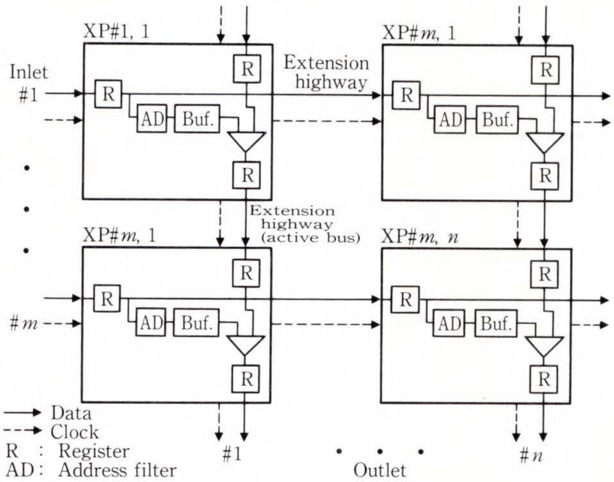



Fig. 4—Self-routing module structure.

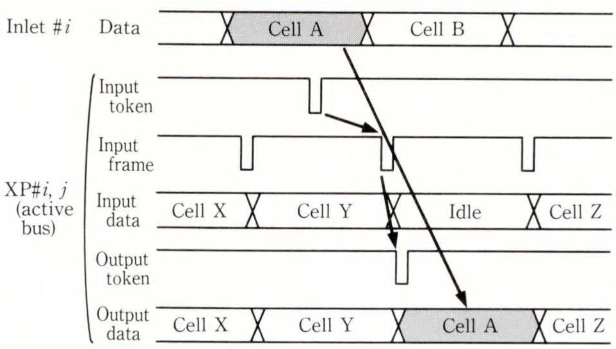



Fig. 5—Multiplexing.

### 3.2 Self routing module structure

As shown in Fig. 4, every function required by the SRM is distributed to each crosspoint to enable the SRM to be constructed with only one IC. The buffer outputs at an outlet are connected to an active bus. Each crosspoint has three functions.

1) Address filtering of the incoming cells with the tag of its outlet number
2) Temporary buffering for absorbing link contentions
3) Cell multiplexing onto an outlet out of the distributed buffers

A cell arriving at an inlet is loaded into the buffer at the crosspoint between the inlet and the outlet pointed to by the tag. The cells stored in the buffers of an outlet are taken out on a FIFO basis and multiplexed onto the active bus. On the active bus, the signal lines of the cells are transmitted with their clock line and retimed by the clock at each crosspoint. The active bus is effective for high-speed multiplexing because the timing deviations between the signal lines and the clock are minimized.

Moreover, the disadvantageous increase in buffer size posed by dividing the buffer at each inlet can be resolved by a buffer read mechanism. If multiple buffers corresponding to an outlet operate as one FIFO buffer, it is possible to reduce the buffer size to as much as in the output buffer switch. For example, every cell has the stamp of the time when the cell was written into the crosspoint buffer and is read from buffers in order of the stamp. The SRM is physically a kind of crosspoint buffer switch and logically a kind of output buffer switch.

A token line for multiplexing cells onto an outlet is prepared and connects the crosspoint in a daisy chain in the vertical direction in Fig. 4 As shown in Fig. 5, if the token finds the cell to be multiplexed at the crosspoint switch (XP#$i$, $j$), it gives the right to access the active bus at the next cell-frame period. The token is then sent to the next crosspoint switch (XP#$i$ + 1, $j$) with the cell. If there is no cell to be multiplexed, the token is immediately sent to the next crosspoint switch (XP#$i$ + 2, $j$).

### 3.3 Performance evaluation

To confirm the feasibility of MSSR switching, we evaluated the required buffer size to satisfy cell switching delay and cell-loss rate. The traffic model is approximated with the three-stage serial queuing network model of M/D/1(m) considering the cell arrival process as the superposition of the renewal process[8]. Each stage queue with finite waiting rooms represents a FIFO buffer at an outlet of an SRM. It is assumed that the load of each link is balanced and the cell-loss rate at each queue is adequately small. Therefore, it can be considered that the cell loss at each queue has no effect on the cell-arrival rate at the next stage.

Figure 6 shows the buffer length at each outlet of an SRM spotted against the cell-loss rate for several link traffic loads. When a cell-loss rate is less than $10^{-10}$ under 90 percent link utilization, the buffer length is 110 cells. For the

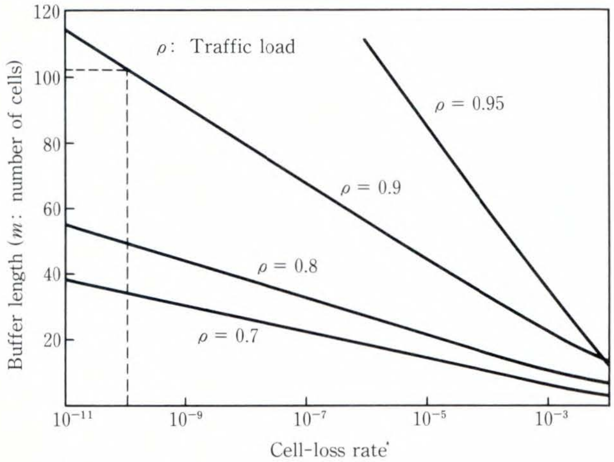



Fig. 6—Required buffer capacity.

Table 1. IC specifications

| Item | Specifications |
|---|---|
| Function | Cell switch (1:1, 1:$N$)<br>VCI conversion<br>Traffic monitoring |
| Highway throughput | 1.2 Gb/s<br>80 Mb/s x 16-bit parallel |
| Cell length | Up to 72 bytes, any length |
| Buffer capacity | 16 cells |
| Device | Bi-CMOS gate array<br>7 kgate + 10 Kbit RAM |
| Interface | TTL/ECL100K |
| Power dissipation | 4.7 W/chip |
| Package | 256-pin PGA |

cell switching delay characteristics, the maximum delay is composed of the queuing delay and holding time. The queuing delay is bounded by the queuing buffer length. The maximum queuing delay is the time taken for a cell to pass all the way through m buffers in each SRM. With a buffer length equal to 110 cells, the maximum switching delay is about 0.23 ms at a highway throughput of 622 Mb/s. Thus, the target objectives are satisfied.

### 3.4 Bi-CMOS ATM switch LSI[9)]

The key to obtaining a large-scale ATM switch is how to achieve both high-speed and low power dissipation IC switches. High-speed inter-package signal transmission requires a strong drive capability. We selected Bi-CMOS with an ECL interface as the most appropriate technology. By Bi-CMOS technology, an operation speed of 75 Mb/s is obtained for 500 of 156-Mb/s channels.

The estimated hardware requirements for each outlet are 35 kgates and 47 Kbits of RAM. Though it is preferable to implement all functions on one chip for high-speed operation and switch size reduction, in practice, the integration scale of the Bi-CMOS process at present is 8 kgates and 40 Kbits. In the early stages of ATM switch development, it was unavoidable that the SRM had to be constructed of several ICs. So, the ATM switch IC was developed as a single crosspoint switch and implemented in a Bi-CMOS gate array having a 0.8-$\mu$m rule logic and RAM. The maximum access time of the RAM is 10 ns. The chip measures 13 m x 13 mm and is packaged in a 256-pin PGA. The 7 kgates and 2 blocks of 5-Kbit RAM cells were used.

Table 1 shows the specifications. The IC has the capability of a 1.2 Gb/s throughput (80 Mb/s x 16 bit-parallel operation). The IC has VCI conversion and tag insertion logic, dual-port RAM for storage of up to 16 cells, RAM control logic for FIFO operation, cell multiplexing logic, a traffic monitor, and a processor interface. In additional to the input and the output highway, extension highways are provided for easy expansion, and the signals are retimed by a register for high-speed operation. A token is cycled in a multiplex control line that connects the ICs in a daisy chain.

## 4. Service-specific functions enabling ATM network to provide varied services

The ATM network provides basic transport service common to all services. To support varied services, service-specific functions are to be performed at customer equipment and/or equipment in the switching system. For example, to provide constant bit-rate transfer service for speech, customer equipment requires functions which perform not only cell assembly and disassembly, but also dejittering, timing recovery, and compensation for lost cells.

LAN is expected to be used as a network extending over several offices and businesses over a wide area. Inter-LAN connection is essential to expand the area for LAN, and the inter-LAN connection service will be demanded by B-ISDN users even at an early stage. LAN generally offers connectionless communication using variable-length messages. On the other hand, ATM network offers connection-oriented communication. Therefore, service-specific functions for connectionless data transfer are required and are a very important issue.

### 4.1 Offering connectionless communication on ATM network[10)]

In Connectionless (CL) communication, there is no call setup phase and CL messages are routed using E.164 address in the message in network, message by message. Figure 7 shows connectionless communication in ATM/B-ISDN. Gateways in customer premises translate LAN address to E.164 address, assemble CL messages, assemble cells and vice versa (see Fig. 8). In order to provide the CL communication on ATM, the switching system has to be equipped with so-called Connectionless Service Function (CLSF). CLSF handles variable-length messages and performs message routing based on the results of address translation using E.164 address in the message. CLSF also performs functions such as termination of interface protocols between the subscriber and network and between exchanges, and switching (see Fig. 9). ATM switching provides a path between the subscriber and CLSF and a path between CLSFs, and decouples connectionless communication from connection-oriented communication.

There are two methods of message processing in the CLSF. One is message-based processing. In this method, the message is recovered from cells and address translation, routing, switching, etc. are performed in the form of the message.

The other is cell-based processing, so-called pipelining, using the fact that the address to be used for routing is always included in the BOM cell. In this method, message routing is performed in the form of the cell, at the expense of introducing slightly complicated mechanisms. That is, CLSF gets the routing information using the destination address in the BOM cell, routes the BOM cell and memorizes this information temporally. Then, the CLSF routes subsequent cells belonging to same message using memorized information, and erased routing information in the memory when the EOM cell arrives.

All other functions are also performed in the form of the cell. In pipelining method, switching can be performed by the switching fabric of connection-oriented communication and the delay for cell assembling and disassembling is eliminated. Therefore, the pipelining method is preferable.

We propose the CLSF architecture which

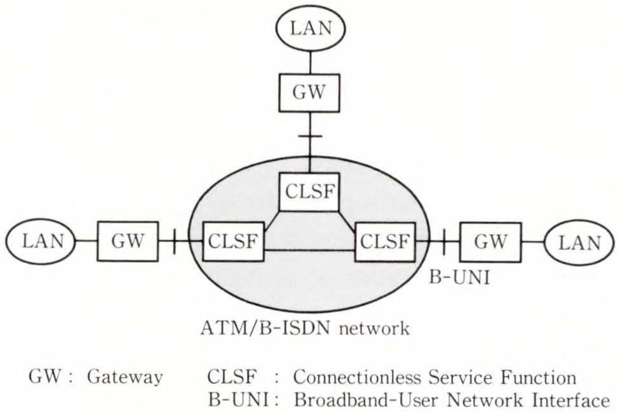



Fig. 7—Connectionless communication in ATM/B-ISDN network.

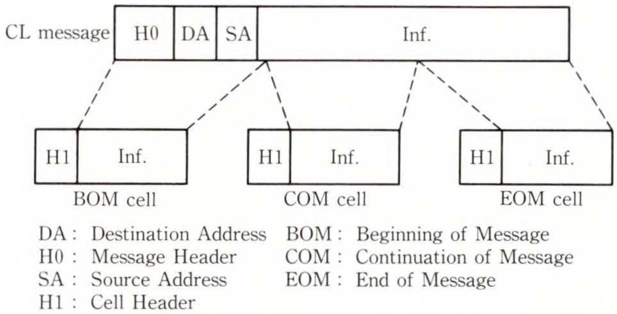



Fig. 8—Message format.

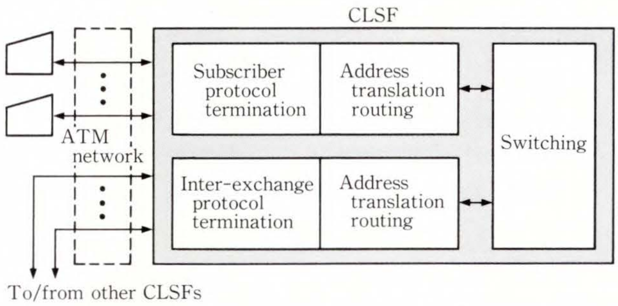



Fig. 9—Function element of CLSF.

consists of Subscriber Message Handler (SBMH) and Gateway Message Handler (GWMH), and ATM switching is shared by both connection-oriented communication and connectionless communication. SBMH performs subscriber protocol termination, address translation and routing. GWMH performs inter exchange protocol termination, address translation and routing. SBMHs and GWMHs are connected to each other by a Permanent Virtual Channel (PVC) having a mesh topology.

Next, the arrangement of Message Handler (MH) in the B-ISDN switching system is discussed. Assume the B-ISDN switching system consists of Distribution Switch (DSSW) and Concentration Switch (CNSW). There are two alternatives (see Fig. 10). One is a centralized arrangement, in which all MHs are placed at the DSSW. The other is distributed arrangement, in which each SBMH is placed at the CNSW and GWMH is placed at the DSSW.

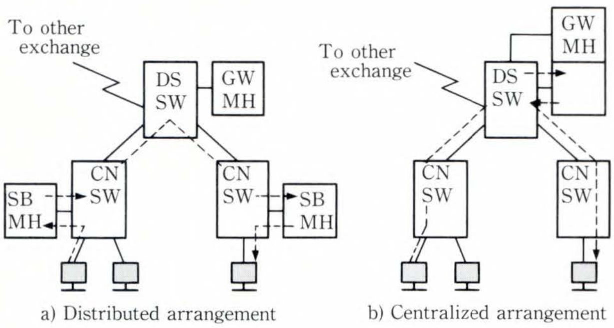



Fig. 10–Arrangement of MHs.

In the distributed arrangement, each MH accommodates subscribers connected to the CNSW at which this MH is placed. Addresses of these subscribers have no relationship to each other in general. The originating MH must translate all the digits of the destination address to identify the terminating MH.

In the centralized arrangement, each MH can accommodate any subscriber connected to any CNSW. Therefore, it is possible to arrange for each MH to accommodate subscribers the first few digits of whose addresses are the same. Then the originating MH has to translate only the first few digits of the destination address to identify the terminating MH. This helps reduce the required size of the routing table (address translation table) of MH.

PVCs are established between the subscriber and MH and between MHs. The PVC between the subscriber and MH will be used at low efficiency. In the centralized arrangement, this PVC is established across the CNSW and DSSW. On the other hand, this PVC is established across only the CNSW in the distributed arrangement. Therefore, the bandwidth utilization of the switch in the centralized arrangement is lower than that in distributed arrangement. When

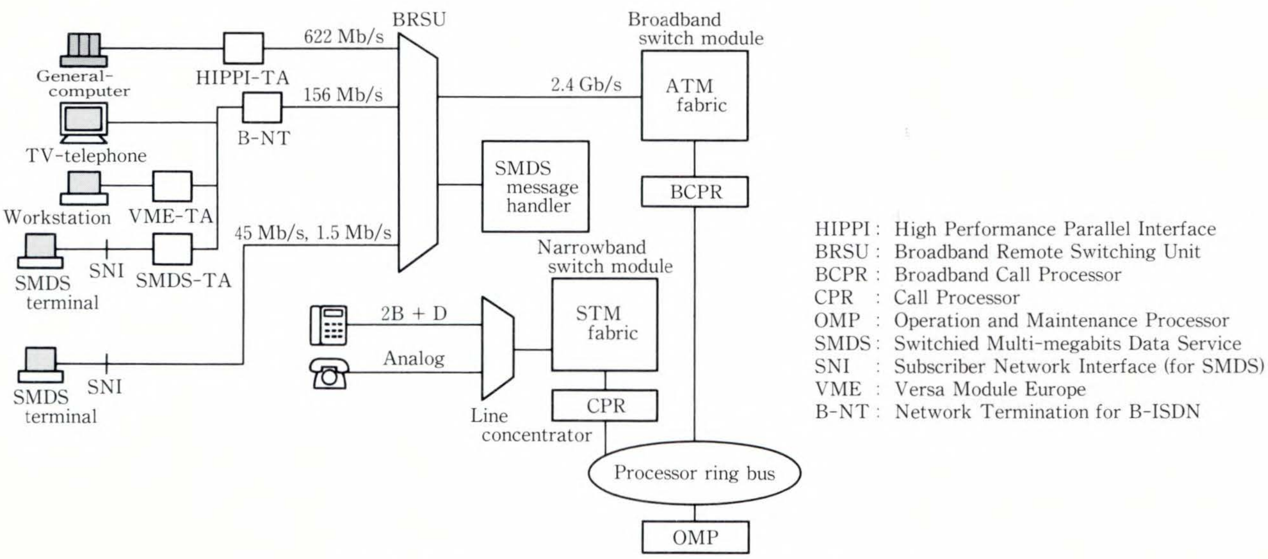



Fig. 11–Configuration of B-ISDN service trial system.

Table 2. General specifications of B-ISDN service trial system

| Items | Specifications |
|---|---|
| Host switch | ATM: MSSR switching method<br>Internal link speed: 1.2 Gb/s<br>Throughput: 80 Gb/s<br>Max 64 BRSUs per host |
| BRSU | ATM concentration (1:1 to 32:1)<br>Max 256 OC-3c lines per BRSU<br>Max 64 OC-12c |
| UNI | OC-3c (156 Mb/s, ATM)<br>OC-12c (622 Mb/s, ATM)<br>DS1 or DS3 (for SMDS SNI)<br>Customer premises bus: optical active bus based on DQDB |
| Terminal and services | Video telephone terminal adapters for:<br>VME bus<br>SMDS (1.5 Mb/s, 45 Mb/s)<br>HIPPI (622 Mb/s) |

DQDB: Distributed Queue Dual Bus
VME: Versa Module Europe
HIPPI: High Performance Parallel Interface

connectionless traffic is not such a small part of the total traffic, the distributed arrangement is preferable.

## 5. B-ISDN service trial system

Figure 11 shows the configuration of B-ISDN service trial system and Table 2 shows its general specifications. In the figure, Switched Multimegabit Data Service (SMDS) is a connectionless data service specified by Bellcore. The broadband switch module is added to the existing narrowband switching system. An Operation and Maintenance Processor (OMP) provides unified operation, and administration and maintenance functions for both N-ISDN and B-ISDN.

The broadband switch module is composed of the broadband host switch and Broadband Remote Switching Unit (BRSU). The BRSU provides an interface with subscribers and concentrates the traffic from subscribers. BRSU is equipped with an SMDS Message Handler. The broadband host and the BRSU adopt MSSR switching using the Bi-CMOS LSI. A 64 x 64 switch fabric, where each input highway has 1.2 Gb/s of throughput, constructed by connecting 8 x 8 SRMs in the MSSR configuration is housed in 4 cabinets. The maximum capacity of each switch is 512 of 156-Mb/s channels. In the switch, information in the highway is handled in a sixteen-bit parallel manner and 80-MHz switching operation is established.

For subscribers with the User-Network Interface (UNI) standardized in CCITT, a 156 Mb/s or 622 Mb/s single fiber optic interface with Wave Length Division Multiplexing (WDM) is provided. The policing function is provided for each subscriber interface. For subscribers requiring a direct interface with SMDS Subscriber Network Interface (SNI), 1.5 Mb/s and 45 Mb/s 4-wire metallic interfaces are provided.

## 6. Conclusion

This paper reviewed various switching architectures, which are classified into four categories, input buffer, output buffer, shared buffer, and crosspoint buffer. The characteristics of each category are discussed. The paper also introduced the author's original architecture, MSSR, for ATM switching. The MSSR has the following features to deal with the various types of traffic required for B-ISDNs.

1) Variable bit-rate switching based on a self-routing principle.
2) Suppression of cell losses due to the minimization of the probability of outlet contention
3) Modular configuration consisting of small self-routing modules

A crosspoint buffer structure for the SRM was also introduced. This structure has the advantages of suitability for high-speed switching. For connectionless data transfer, the architecture of a message handler performing connectionless service functions are discussed. We established a highway throughput of 1.2 Gb/s for our B- ISDN service trial system with a Bi-CMOS technology. The service trial system proved the feasibility of a commercial public switching network accommodating up to 4 000 subscribers at a concentration ratio of 1/8.

## References

1) CCITT: Draft Recommendation. SGX VIII I.121, Dec., 1990

2) Suzuki, H. et al.: Output-buffer Switch Architecture for Asynchronous Transfer Mode. Proc. ICC'89, 4.1, **Vol. 1**, 1989, pp. 99-103.
3) Kuwahara, H. et al.: A Shared Buffer Memory Switch for an ATM Exchange. Proc. ICC'89, 4.4, **Vol. 1**, 1989, pp. 188-122.
4) Obara, H. et al.: High Speed Transport Processor for Broadband Burst Transport System. Proc. ICC'88, 29.5, **Vol. 2**, 1998, pp. 922-927.
5) Hajikano, K. et al.: Asynchronous Transfer Mode Switching Architecture for Broadband ISDN. Proc. ICC'88, 29.3, **Vol. 2**, 1988, pp. 911-915.
6) Muise, R.W. et al.: Experiments in Wideband Packet Technology. Proc. Zurich seminar '86, 1986, pp. 135-139.
7) Wernik, M. R.: Architecture and Technology Consideration for Multimedia Broadband Communications. Proc. GLOBECOM'88, 21.2, **Vol. 2**, pp. 663-667.
8) Abe, S et al.: Traffic Design for an Asynchronous Transfer Mode Switching Network. Proc. ITC-12, 1988, pp. 1.3A.4.1.
9) Kato, Y. et al.: A VLSIC for the ATM Switching System. Proc. ISS'90, P5, **Vol. III** pp. 27-32.
10) Kawasaki, T. et al.: A Study on High Speed Data Communication Systems Using ATM. Proc. GLOBECOM'91, 59.5., pp. 2105-2109.



**Kazuo Hajikano**

Overseas Switching Hardware Development Dept.
Switching Systems Dev.
FUJITSU LIMITED
Bachelor of Electronics Eng.
Waseda University 1978
Master of Electronics Eng.
Waseda University 1980
Specializing in Broadband Switching System



**Koso Murakami**

Integrated Communication Network Laboratory
FUJITSU LABORATORIES, KAWASAKI
Bachelor of Electronics Eng.
Osaka University 1971
Dr. of Electronics Eng.
Osaka University 1991
Specializing in Digital Communication Network



**Tetsuhiro Nomura**

Integrated Communication Network Laboratory
FUJITSU LABORATORIES, KAWASAKI
Bachelor of Engineering in Electronics Eng.
Meiji University 1972
Specializing in Electronic Switching System

Featuring Paper

UDC 621.395.74

# ATM Transmission Systems Technologies

● Kazuo Iguchi ● Hirohisa Gambe ● Kazuo Yamaguchi

*(Manuscript received February 25, 1992)*

ATM is the key to creating the next generation of telecommunication networks, or B-ISDN. This paper discusses the architecture of B-ISDN and the required network elements from the viewpoint of the transmission network. Flexibility conforming to B-ISDN standards is most important for the introductory phase of B-ISDN. Also, the economical aspect should be considered in the spread phase. The architectures of the trunk network, subscriber access network, and customer premises network, and their required performance were studied. This paper discusses the feasibility of the studies and shows the prototype systems configurations and results. These experiments confirm that the ATM transmission technologies are now applicable to B-ISDN.

## 1. Introduction

Broadband ISDN (B-ISDN), which can integrate many services having different bit rates, like voice, high-speed data and video, is expected to become the next generation network and is being studied at CCITT. The key feature of B-ISDN is the unique transfer mode, called the Asynchronous Transfer Mode (ATM), which can be applied effectively to both Continuous Bit-rate Services (CBR) and Variable Bit-rate Services (VBR).

The basic parameters of B-ISDN, like ATM cell and interface structures, were agreed upon at the CCITT Geneva meeting and approved in May, 1991[1]. B-ISDN then entered the stage of practical study[2)-4)]. The items agreed upon, however, are not sufficient for B-ISDN implementation, but they are effective for studying prototype B-ISDN systems. These problems will be resolved by CCITT recommendations in 1992.

This paper discusses the architecture of B-ISDN, based on the current agreement of CCITT, which includes the ATM functions, ATM structure, ATM network and interfaces. This paper also studies the Trunk Network (TN), Subscriber Access Network (SAN) and Customer Premises Network (CPN) configurations, which will form the infrastructure of B-ISDN. The authors examined the requirements for these networks and propose a two-phase approach concerning to the evolution of B-ISDN.

The first phase of B-ISDN is the introductory phase. In this phase, the feasibility of each network and putting into effect the practical implications of the current agreement of the B-ISDN recommendations are the most important issues. The second phase is the spread phase of B-ISDN, to be implemented around the year 2000. The economical aspect will be the most important, and some modifications of B-ISDN recommendations are required, especially in B-SAN and B-CPN.

## 2. Architecture of B-ISDN

Broadband ISDN was originally intended to provide multimedia services. Therefore, the architecture of B-ISDN should be flexible enough to meet changes in service, media, and communication methods.

The ATM concept is very attractive for integrating both services and networks. The key features which enable such integrations are discussed below.

1) Cell-based multiplexing scheme

There are no hierarchies in the ATM net-

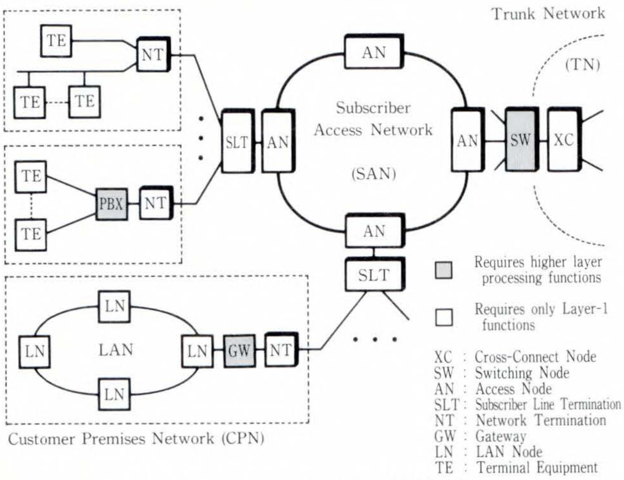



Fig. 1—Architecture of B-ISDN.

works and any kind of service can be accommodated. Each user can send ATM cells, according to its requirements, through a unified user-network interface. CCITT supports 155.52-Mb/s and 622.08-Mb/s user-network interfaces. Within the network, ATM cells from each user are multiplexed and conveyed by optical transmission systems. This cell-based multiplexing enables effective use of network resources and provides more flexibility than conventional time-division multiplexing systems, which were optimized based on a 64-kb/s hierarchy. Network element configurations are simplified by non-hierarchical multiplexing schemes. Also, network configurations are simplified, improving the reliability of the network.

2) Virtual path scheme

The quality required of each service is different and it is difficult to manage at the same time. The Virtual Path (VP) concept is effective for this application. That is to say, the services requiring the same quality are arranged in the same VP group and each VP group is integrated within the transmission media. Each terminal negotiates the required service quality, like cell loss rate and delay, in the Switching Node (SW) of a public network in the call setup phase. The SW registers the information from each Terminal Equipment (TE), like peak and mean traffic parameters of the cell generation statistics, and assigns an appropriate Virtual Path Identifier (VPI) to the TE. After the call setup phase, ATM cells are supervised at the entrance of the network, or Subscriber Line Termination (SLT), and normal cells are multiplexed by their VPI values and sent to SW using SAN.

Figure 1 shows the architecture of B-ISDN. The authors applied the cross-connect systems for TN to handle high-capacity traffic and to provide flexible virtual path networks. For SAN, the authors applied a ring-based network for the feeder loop considering the required fiber length, reliability, and effectiveness of the transmission capacity. This approach is effective in the introduction phase of B-ISDN. SAN consists of SLT and AN. SLT has the function of line concentration and policing of ATM cells. AN has the add/drop functions and provides flexible VPs between each SLT and between SLT and SW, utilizing the add/drop function of VPI. Customer Premises Network (CPN) provides access to public networks and enables the customer premises equipment to become part of the network.

The configuration of CPN will vary in the evolution stage of B-ISDN according to the network size required by users. It is important to categorize the architecture of CPN according to the requirements. Also, the change in service should be considered because bi-directional communication services and data services will be dominant in the introductory phase but distribution services like CATV will increase in the spread phase of B-ISDN.

### 2.1 Trunk network (TN)

High efficiency, high reliability and flexibility are important for TN. A cell-based multiplexing scheme enables non-hierarchical Cross-Connect (XC) capabilities and provides simple XC mechanisms for network elements. Also, Virtual Path (VP) schemes enable simple and flexible path configurations. These features cannot be provided by conventional systems. Path connection and path capacity allocation are designed independently. Therefore, virtual mesh networks can be established in the introductory phase, regardless of their capacity. XC systems are passive devices and no changes are required in XC control mechanisms, even when the con-

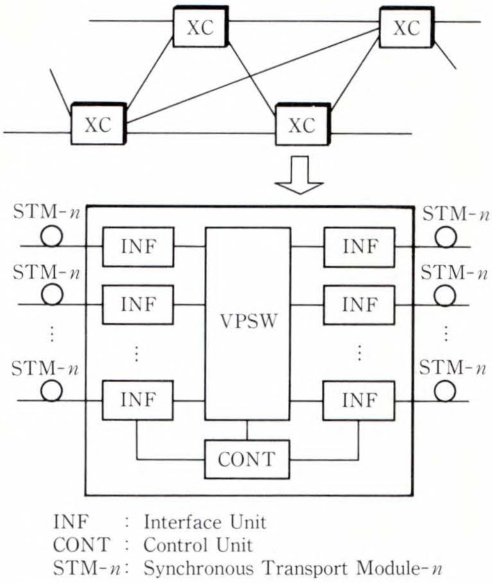



Fig. 2–Configuration of basic XC system.

nection is changed, thereby enabling a highly reliable network configuration.

Figure 2 shows the basic XC system configuration. It consists of optical interface units, VPI converter units, and VP cross-connect units. As mentioned before, VP cross-connect functions are implemented on a cell-by-cell basis, and are independent of the interface speed. For the TN, a high throughput of over 10 Gb/s, and a low cell loss of less than $10^{-10}$ are required. VP cross-connect mechanisms to obtain these specifications are important.

### 2.2 Subscriber access network (SAN)

A ring-based network configuration is effective for constructing a cost-effective SAN. This is because high-speed optical transmission links are shared by ANs and flexible bandwidths can be allocated between each AN. It is very important in the introductory phase of B-ISDN.

Figure 3 shows the SAN and AN configurations. The authors used Media Access Control (MAC) protocol based on Distributed Queue Dual Bus (DQDB) for the ring access[5], which is effective for both large-scale networks and obtaining high efficiency. DQDB-based MAC processing is performed at the transmission speed, for example 2.4 Gb/s, so as to make effective use of the transmission capacity and to effectively accommodate the broadband serv-

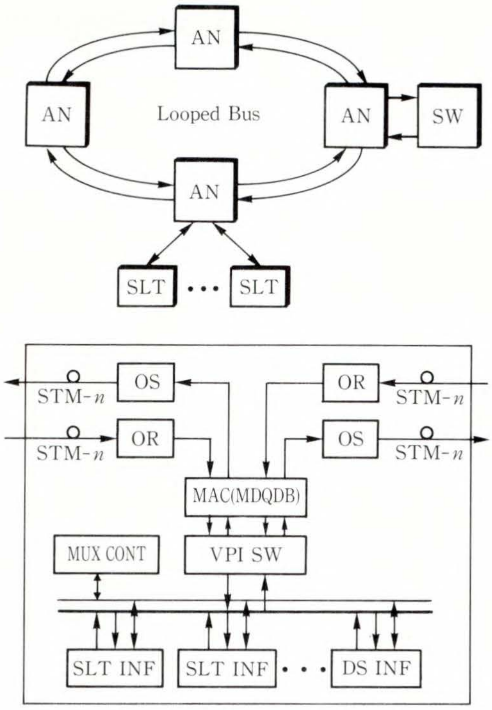



Fig. 3–Configurations of SAN and AN.

ices. The weak point of DQDB is its access capability, which causes variations in the delay characteristics at each AN location. However, the variations are negligible in high-speed processing as with this application.

VP path connection and bandwidth allocation are changed when a failure occurs to ensure high reliability. The authors suppose dual ring architecture and loopback functions are necessary for SAN. SAN network configuration may not be unique and several alternative approaches will exist. Therefore, AN should have the ability to change the add/drop functions to simple SLT for a double-star network.

### 2.3 Customer premises network (CPN)

The CPN is smaller and has fewer terminals than SAN. The requirements for CPN are different from SAN[6]. The functions regarded as necessary for CPN are as follows:

1) Arbitrary communication speed support
2) Simultaneous multi-terminal connectability
3) Multimedia communication support
4) Multipoint communication support
5) Terminal portability
6) Wiring expandability
7) Reliability

Figure 4 shows the configurations of CPN which can provide these requirements. Star topology applying PBX as NT2 is the most popular approach and provides a simple CPN, but requires higher layer processing. Therefore, the complex NT2 function, like the distribution of signaling information and the compatibility checking mechanism, is required in NT.

The ring topology is the same as in SAN, and enables the transmission bandwidth to be used effectively. However, in an active configuration, one node failure disables the entire network. Also, it requires medium access control overhead.

A bus topology, which is applied to Narrowband ISDN (N-ISDN), is attractive for B-CPN, but in an active configuration, the economy and reliability is the same as with a ring topology. N-ISDN incorporates a passive bus scheme, using the D-channel access protocol, which is very simple and only requires one-bit overhead. If a simple MAC protocol can be achieved, which enables TB/SB interface equal to Z, passive bus technologies would be very attractive in terms of making the CPN reliable, expansible, and economical.

Based on the above, the authors studied the CPN configuration, dividing B-ISDN into two separate phases, the introductory phase and the spread phase.

**2.3.1 Introductory phase of B-ISDN**

In this phase, it is necessary to establish a simple CPN conforming to the CCITT agreement. Also, flexible NT and TE configurations are desired for future CPN. The CCITT agreement restricts point-to-point connection at the PMD layer to hasten the establishment of CPN.

The authors propose a physical star/logical bus CPN architecture. Physical star provides a point-to-point configuration conforming to CCITT standardization. The logical bus provides the inherent features of a bus network. The logical bus is realized by the internal bus in NT. Therefore, NT requires no higher layer processing functions and makes the hardware simple and reliable.

Figure 5 shows the NT configuration. ATM cells from each TE are buffered once in NT and multiplexed on a cell mux/demux bus under the control of Control Unit (CONT), thereby eliminating cell congestion. No MAC overhead is

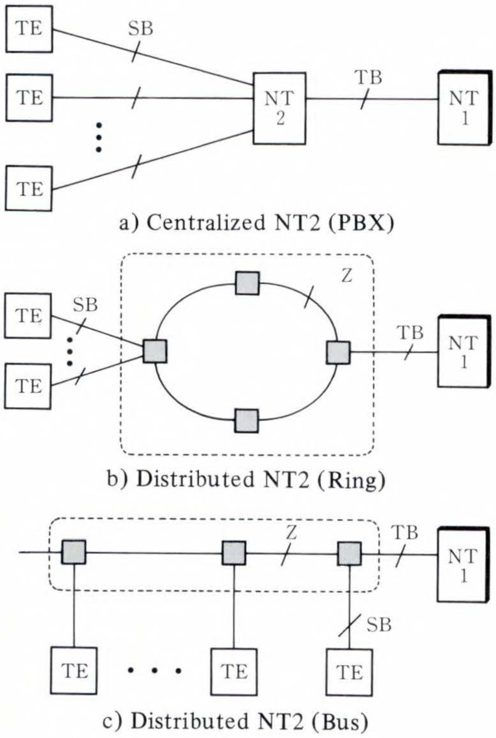



Fig. 4—Configuration of B-CPN.

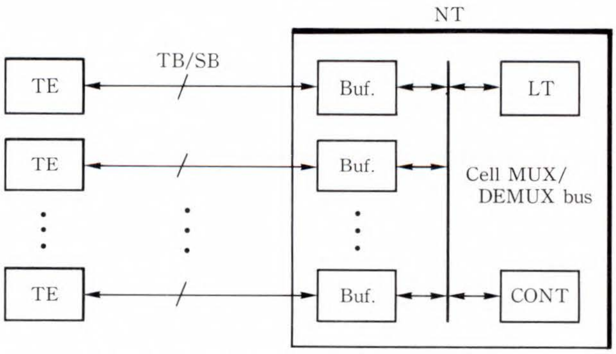



Fig. 5—Configuration of bus-based NT.

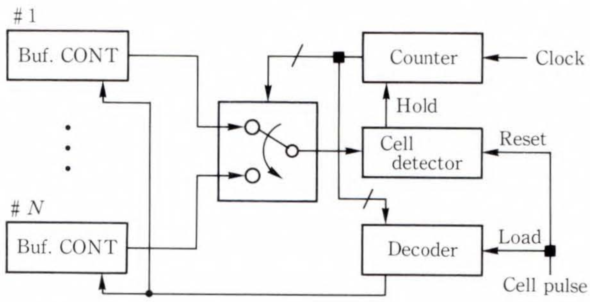



Fig. 6—Collision control scheme.

required on the transmission line. This architecture has the merits of both the bus and star architecture, and provides good expansibility and reliability.

Figure 6 shows the configuration of the collision control circuit. A cell detector checks each buffer status sequentially using a counter and, if an available cell is found in the buffer, stops the counter and enables the buffer to transmit the ATM cell via the bus. This circuit operates at about 5 Mb/s and can be created using one CMOS programmable logic device having an interface bit rate of 155.52 Mb/s with 16 terminals.

### 2.3.2 Spread phase of B-ISDN

For the spread phase of B-ISDN, it is very important to provide a distribution service and cost-effective configurations in CPN. The introductory phase requires one pair consisting of an optical transmitter and receiver per TE in NT. If the number of optical transmitters and receivers can be decreased, the cost of the CPN will be substantially reduced. An optical passive bus satisfies this requirement and it will be the most promising approach to realize a cost-effective CPN in the spread phase of B-ISDN. However, there are several problems, caused by differences in the round trip delay and suchlike, to be overcome before the optical passive bus becomes usable. Problems are as follows:

1) Limitation of bus length
2) Transmission efficiency
3) Number of TEs that can be connected
4) ATM cell access contention
5) Intra-CPN multipoint communication

Figure 7 shows the configuration of the optical passive bus. The authors have already proposed new techniques to overcome some of these problems[7),8)]. One of the most important features is intra-CPN multipoint communication and the techniques should be developed in this phase.

### 2.3.3 Multipoint communication

This mechanism has many advantages for small and medium-sized businesses. It has the networking capabilities of distributed equipment, without the need to introduce expensive LANs. In intra-CPN communication, internal communication traffic should not affect the outgoing traffic or public network. The following requirements must be taken into account:

1) Minimum influence on public networks
2) Simple functions required in NT and TE
3) No special protocols needed in NT and TE
4) Independent of CPN configuration

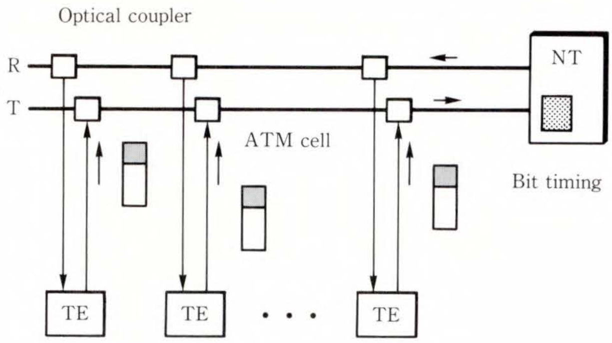



Fig. 7—Optical passive bus.

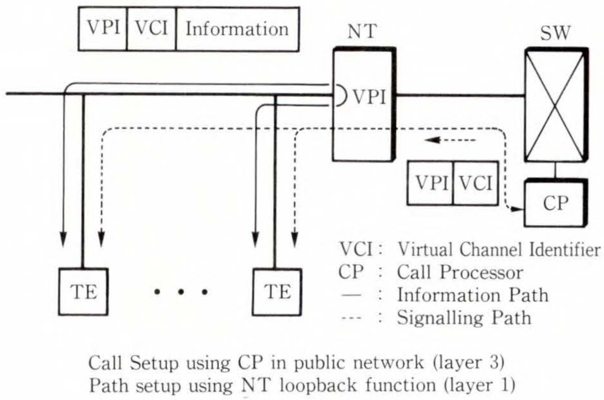



Fig. 8—Intra-CPN communication.

Figure 8 shows an example of the intra-CPN communication function. Higher layer processing for call setup is executed between the TEs and SW to set the communication path. The intra-CPN communication path is set by using the loopback function in NT. In this system, TEs can communicate irrespective of the CPN configuration or communication type, thereby enabling the full benefits of an ATM-based logical network to be used.

An actual communication path can be provided by the CPN function, or loopback, to suppress the increase in outgoing traffic. Only the signaling traffic path through the NT is necessary and it will be very small. This intra-CPN communication scheme satisfies all the above requirements.

## 3. Prototype systems

The authors studied the feasibility of B-ISDN system, having the above mentioned requirements, and examined the cross-connect systems, add/drop systems, and optical passive bus systems.

### 3.1 Cross-connect systems (XC)

To develop high capacity, high throughput cross-connect systems, many switching configurations were considered. Of those, the VP switching architecture was found to be the most important.

The authors focused on input buffer-type VP switching, which requires a smaller buffer and less internal processing speed[9]. In this type of VP switching, the buffer control mechanism and contention control schemes are the keys to high throughput. If a simple and effective algorithm can be developed, this approach should be one of the candidates for the high-capacity, high-throughput XC configuration.

Figure 9 shows the configuration of XC. Table 1 shows the specifications. RAMs were used for input buffers instead of FIFO memories to eliminate the headline effect[10] which impedes the throughput. Random read control mechanisms were used to solve the headline effect, and a random write control with empty address management mechanisms was used to reduce the buffer memory. For the S-SW, a newly developed high-speed CMOS matrix LSI operating at 116 Mb/s was used to provide non-block operation[11]. These mechanisms are quite effective for input buffer XC systems. For the congestion control, Polling and Reservation (P-RSV) schemes were used to obtain high throughput. P-RSV controls one reservation for each input and output port for one cell period, which is required for the input buffer XC. Queuing memory management techniques were used. Figure 10 shows the congestion control configuration and basic principle of P-RSV. This enables a high throughput of over 90 percent, as shown in Fig. 11.

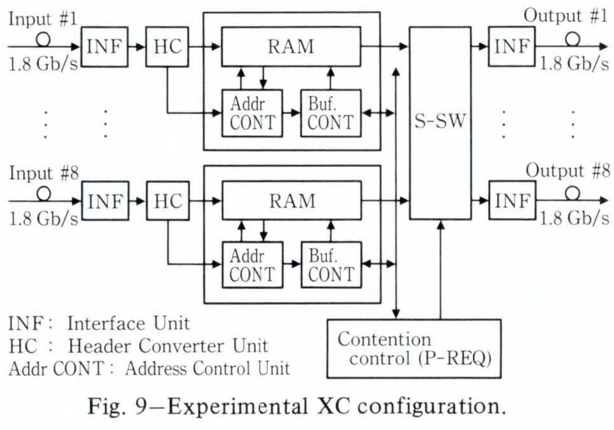


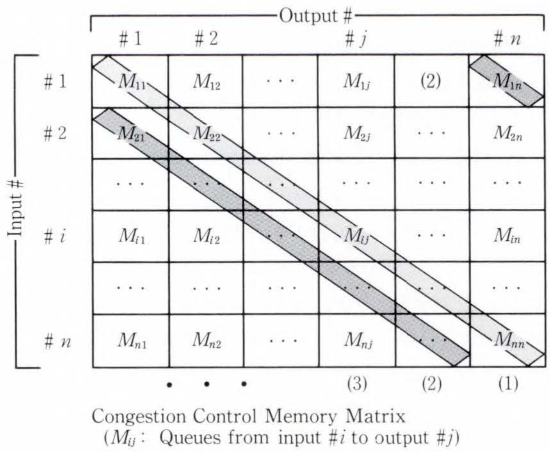

Fig. 9–Experimental XC configuration.

### 3.2 Add/drop multiplexer systems (ADM)

DQDB-based MAC protocol was used for the dual ring ADM system. In the ring network,

Table 1. Specifications of XC system

| Items | | Specifications |
|---|---|---|
| Interface rate | | 1.866 Gb/s |
| Number of links | | 8 (input and output) |
| Total throughput | | 11.2 Gb/s |
| Swithing scheme | | Input buffer + space SW |
| | Input buffer | Random-in, Random-out (RIRO) (RAM + address management) |
| | Contention control | Scheduling P-RSV (Polling and Reservation) |
| | Space SW element | 16 x 8 crosspoint SW (CMOS LSI: 116 Mb/s) |




Congestion Control Memory Matrix
($M_{ij}$: Queues from input $\#i$ to output $\#j$)

**Basic principle of P-RSV**

1) Poll $M_{ij}$ $(i = j)$ for $j = 1$ to $n$.
Reserve $O_j = i$ if $M_{ij} = 1$.
($O_j$: input # for output $\#j$)
Note: see (1) in Fig.
2) Repeat 1) for $i = j + k$ except $j$ at $O_j \neq 0$, $k = 1$.
Note: see (2) in Fig.
3) $k = k + 1$: repeat 2) to $k = n - 1$.

(Total $n$ times/one cell period)

Fig. 10–P-RSV scheduling scheme.

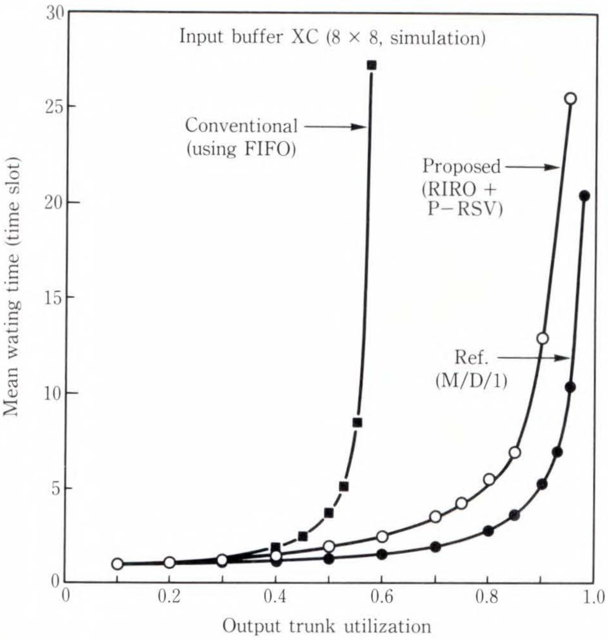


Fig. 11–Average delay characteristics.


Fig. 12–Experimental ADM system.

Table 2. Specifications of ADM systems

| Items | Specifications |
|---|---|
| Interface rate and topology | 1.866 Gb/s (dual-ring) |
| | 156 Mb/s or 622 Mb/s (Star: max 8 line cards) |
| Total throughput | 2.8 Gb/s |
| Access protocol | M-DQDB for dual-ring Skip polling for star |
| Connection mode | Point to point (1:1) Distribution (1:$n$) |

Table 3. Specifications of optical passive bus

| Items | Specifications |
|---|---|
| Interface rate | 156 Mb/s (STM-1) |
| Access scheme | Cell-based burst access |
| Contention control | Polling Request (P-REQ) |
| Phase adjustment | Phase Aligned Bus (PAB) |
| Timing recovery | Bit shifted composition using CMI preamble code |
| Number of TEs | Max 16 |
| Line length | Max 1 km |
| Connection mode | 1:1, 1:$n$ (distribution) |

ATM cells for the distribution mode circulate multiple times, a phenomenon which does not occur in bus system. Therefore some mechanisms should be taken to prevent this phenomenon. One indicator bit was added to the DQDB protocol. If ATM cells pass the master node, this bit is turned on. The master node monitors the indicator bit and ATM cells whose indicator bit is turned on are discarded. This technique is effective for a ring system.

Bus-based architecture was used for the AN to enable the service to be accommodated flexibly. ATM cells from each interface unit are multiplexed on the bus under the control of MUX-CONT to eliminate cell congestion. Skip polling multiplexing is used in MUX-CONT, which monitors the buffer status sequentially and polls only valid buffers. This system was evaluated using computer simulation. The resulting throughput characteristics were the same as those of the M/D/1 model.

Table 2 shows the specifications and the prototype ADM systems, and Fig. 12 shows an external view of the system.

### 3.3 Optical passive bus systems

The authors have already resolved some of the difficulties related to optical passive bus as mentioned before, focusing on the medium access control scheme. The authors also analyzed several MAC protocols that can be applied for B-CPN. The centralized MAC protocol, such as Polling (POLL) and Request-Assign (RAMA), and the distributed control, such as Carrier Sense (CSMA) and Distributed Queue (DQDB), are candidates for B-CPN MAC protocol. The authors propose Polling-Request (P-REQ) scheme[12], which is not the best solution but it

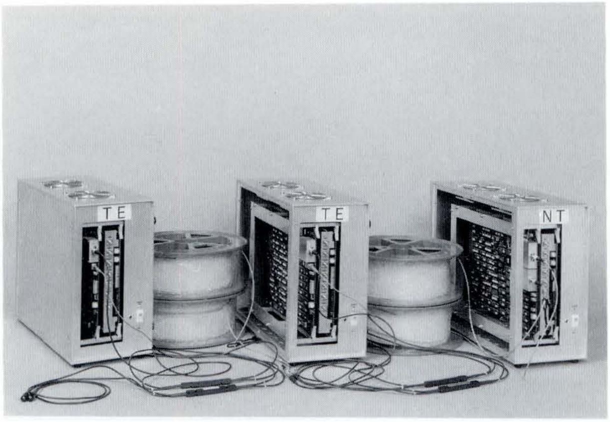



Fig. 13–Experimental optical passive bus.

is applicable to B-CPN. P-REQ has throughput that is slightly inferior to the distributed control methods, but the number of TEs that can be connected is about double that for CSMA and DQDB, where two optical couplers are needed in the T-line for each TE. The structures of NT and TE are simple, requiring little hardware. Consequently, it is a system suited to B-CPN, where the emphasis is on economy.

Table 3 shows the specifications of B-CPN systems based on the prototype optical passive bus. Figure 13 is a photograph of the system. The MAC part is integrated into one IC, using programmable logic device. The optical interface and the MAC part, including the cell buffer, are integrated into one printed circuit board.

## 4. Conclusion

The paper discusses the architecture of B-ISDN and prototype systems. The paper proposes cross-connect systems for a trunk network, add/drop multiplexer systems for a subscriber access network, and physical star/logical bus systems for the introductory phase and optical passive bus systems for the spread phase of B-ISDN. The authors confirm Fujitsu's ATM transmission technologies can now be applied to for B-ISDN systems.

## References

1) CCITT Working Party XVIII/8: Draft Recommendation. CCITT COM XVIII-R34-E, 1990.
2) Tokizawa, I., Kanada, T., and Sato, K.: A New Transport Network Architecture Based on Asynchronous Transfer Mode Techniques. Proc. ISSLS 1988, pp. 217-221.
3) Murano, K., Murakami, K., Iwabuchi, E., Katsuki, T., and Ogasawara, H.: Technologies Towards Broadband ISDN. *IEEE Commun. Mag.*, **28**, 4, pp. 66-70 (1990).
4) Sato, Y., Yamanaka, N., Sato, K., and Tokura, N.: Experimental ATM Transport Systems and Virtual Path Management Techniques. Proc. GLOBECOM '91, pp. 2110-2116.
5) IEEE document 802.6: "IEEE Draft 802.6 MAN", Ver. d. 8, June 1989.
6) Iguchi, K., Takeo, H., Amemiya, S., and Tezuka, K.: Subscriber Access Scheme for Broadband ISDN. Proc. ICC '90, pp. 663-669.
7) Iguchi, K., Amemiya, S., Soejima, T., and Murano, K.: Optical Passive Bus for Broadband User-Network Interface. Proc. ISSLS 1988, pp. 235-239.
8) Iguchi, K., Amemiya, S., Takeo, H., and Tezuka, K.: A Study on Architecture of Optical Passive Bus for Broadband Customer Premises Network. *Electronics and Communication,* in Japan, Part 1, **74**, 1, pp. 1-12 (1991).
9) Obara, H., and Yasushi, T.: High Speed Transport Processor for Broadband Burst Transport System. Proc. ICC '88, pp. 922-927.
10) Karol, M. J., Hluchgj, M. G., and Morgan, S. P.: Input vs. Output Queuing on a Space-Division Packet Switch. Proc. GLOBECOM '86, pp. 659-665.
11) Soejima, T., Ezaki, Y., Iguchi, K., Yamamura, T., Komaki, M., and Ikezawa, T.: CMOS LSI for Flexible Telecommunication Node in Broadband ISDN. Proc. ICC '89, pp. 1403-1407.
12) Amemiya, S., Takeo, H., Tezuka, K., and Iguchi, K.: An Architecture of Optical Passive Bus for Broadband CPN. Proc. GLOBECOM '89, pp. 1647-1654.

**Kazuo Iguchi**

Broadband Communication Systems Laboratory
FUJITSU LABORATORIES, KAWASAKI
Bachelor of Electrical Eng.
Yokohama National University 1973
Specializing in Broadband ISDN Transmission Systems

**Kazuo Yamaguchi**

Broadband Communication Systems Laboratory
FUJITSU LABORATORIES, KAWASAKI
Bachelor of Metallurgical Eng.
Yokohama National University 1971
Master of Metallurgical Eng.
Yokohama National University 1973
Specializing in Broadband ISDN Transmission Systems

**Hirohisa Gambe**

Signal Processing Dept.
FUJITSU LIMITED
Bachelor of Electronic Eng.
Tokyo Institute of Technology 1973
Master of Science in Electrical Eng.
Polytechnic Institute of New York 1978
Specializing in Communication VLSI and Digital Signal Processing

Featuring Paper

UDC 621.395.345.037.3

# Enhancement of Digital Switching System FETEX-150 towards Broadband ISDN and Intelligent Network

• Akihiro Sera • Toru Masuda • Yoichi Fujiyama

*(Manuscript received November 29, 1991)*

This paper reports recent enhancements of the digital switching system FETEX-150. First, the architecture enhancement based on the subsystem concept and multiprocessor ring bus is described. Then recent developments in Broadband ISDN (B-ISDN) based on ATM switching technology are described focusing on the service trial system, which has a maximum throughput of 80 Gb/s. Finally, developments in the Intelligent Network (IN) are described, including Intelligent Network 1 (IN/1) based on telecommunication-processor and Advanced Intelligent Network (AIN) utilizing general-purpose computers.

## 1. Introduction

The FETEX-150 is a digital switching system for central offices and is aimed at overseas markets. The development of the FETEX-150 started in the late 1970s and the first system was put into service in 1982. More than 11 200 000 lines of the FETEX-150 have been put into service or are on order in 18 countries.

The FETEX-150 was originally designed for telephone and narrowband ISDN switching. Recently telecommunication has become more and more important in business activities and daily life. As customers expect more powerful and flexible telecommunication services, it is necessary for telecommunication networks to evolve towards broadband networks, intelligent networks, and personal communication networks. The broadband network enables transfer of a large amount of information in a short time. The intelligent network enables flexible and sophisticated services. The personal communication network eliminates the need for each person to be present at terminal equipment located at a fixed place.

This paper describes the enhancements of FETEX-150 in these directions. First, system architecture enhancement of the FETEX-150 is described and then functional enhancement towards broadband networks and intelligent networks are described. The enhancement towards personal communication network is also progressing but will be reported on another occasion.

## 2. Architecture enhancement

### 2.1 Subsystem concept

The functions required for switching systems are increasing each year, and the switching system is becoming very large and complicated. Apparently it is not feasible to put all the functions into one system. One method to cope with this problem is to divide the functions into separate subsystems.

The switching system of the next generation will consist of a group of subsystems which are independent of each other. Under this concept, even if one subsystem becomes faulty, the other subsystems are not affected. On the other hand, if one subsystem is closely related with other subsystems, the communication between these two subsystems becomes large, which may cause a bottleneck.

### 2.2 Multiprocessor ring bus FESNET-II

To realize this subsystem architecture, a powerful communication between processors is necessary. The mechanism for interconnecting processors must be able to provide direct communication between any pair of processors for a sufficient number of processors. The restrictions on the distance between processors must be minimized and a sufficiently redundant configuration must be provided to ensure high reliability. It is also important for the processors to be able to be expanded easily.

Based on the abovementioned requirements, a multiprocessor ring bus called FESNET-II has been developed[1]. Optic fiber cable is used for high data transmission capability, and flexibility in the location of the processors (see Table 1). The number of nodes (processor pairs) is 64, which is 8 times larger than the multiprocessor system of the current FETEX-150. The Fiber Distributed Data Interface (FDDI) protocol is used for layers 1 and 2 because the FDDI is well standardized and the standard LSI is available.

Table 1. Specification of multiprocessor ring bus FESNET II

| Item | Specification |
|---|---|
| Transmission speed | 100 Mb/s |
| Access method | Token-ring access |
| Configuration | Duplicated FDDI ring bus |
| Number of nodes | 64 |
| Inter-node distance (max) | 100 m |
| Ring length (max) | 6 400 m (= 100 x 64) |
| Protocol layer 1 and 2 | FDDI protocol (PMD, PHY, MAD, SMT), LCC |
| Protocol layer 3 to 7 | Based on OSI |

The FESNET-II provides a good solution to the problem of bottlenecks in communication. Its high data transmission capability eliminates the delay in communication between two subsystems.

### 2.3 FETEX-150 with multiprocessor ring bus

Separation of functions to subsystems must be as optimized considering the advnatages and disadvantages of function separation. For example, analog telephone switching and narrowband ISDN switching should be in the same subsystem to avoid the duplication of software, considering their similarity and close relationship. An example of planned subsystems is as follows (see Fig. 1):

1) Plain Old Telephone Service (POTS) and narrowband ISDN switching
2) Packet switching
3) Broadband ISDN switching
4) Local Service Control Point (L-SCP)
5) Application processor
6) Mobile telephone switching

From the viewpoint of Operation and Maintenance (O&M), it is not preferable to maintain a number of subsystems with separate O&M functions. The solution adopted for the FETEX-150 is to organize the system with a

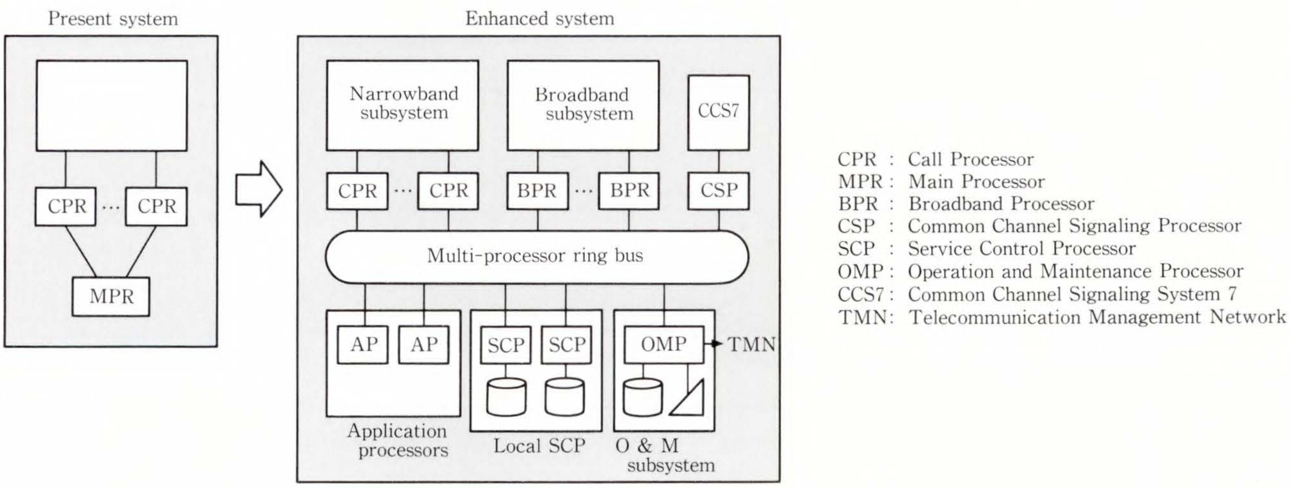



Fig. 1—Architecture enhancement of FETEX-150

O&M subsystem common to many subsystems. The O&M subsystem provides unified operation and maintenance for all the other subsystems within the system. The O&M functions provided by the O&M subsystem is non-volatile, so that even if a certain subsystem goes completely down, the O&M functions survive and can take some recovery action for the faulty subsystem. The O&M subsystem also provides a database common to some subsystems and an interface to Operation Systems (OS) via the data network.

## 3. Broadband switching capability

### 3.1 Standardization and market trends

CCITT is aggressively promoting the standardization of Broadband ISDN (B-ISDN)[2]. In 1988 it was agreed to adopt Asynchronous Transfer Mode (ATM) for B-ISDN, and in 1989, the basic items for ATM (e.g. cell size) were agreed upon[3]. It is expected that initial recommendations for B-ISDN will be issued in 1992, which will encourage B-ISDN trials in many countries. In 1994, further details of B-ISDN recommendation will be issued, which will enable the development of a commercial B-ISDN switching system.

The need for high-speed data transmission and LAN interconnection in the business area is increasing rapidly. Businesses currently use dedicated leased lines of 1.5 Mb/s, 2 Mb/s, etc. Image transmission in the medical area, and in publishing and advertising, etc., also needs high speed data transmission. Domestic applications, such as entertainment, education and medical fields are also a good application of B-ISDN but mass introduction will not take place until the 21st century. So the target of B-ISDN at the initial stage is the business market.

The study of B-ISDN in CCITT is now focusing on ATM technology and optical subscriber lines with speeds of 156 Mb/s and 622 Mb/s. However, before the introduction of such pure B-ISDN, some pre-broadband services will be introduced.

One such pre-broadband service is the frame relay, which is a connection oriented, switched or non-switched packet data service with speed of 64 kb/s to 2 Mb/s. Another candidate is Switched Multimegabit Data Service (SMDS) standardized by Bellcore of U.S., which is a connectionless, packet data service with speed of 1.5 Mb/s to 156 Mb/s. These services will be

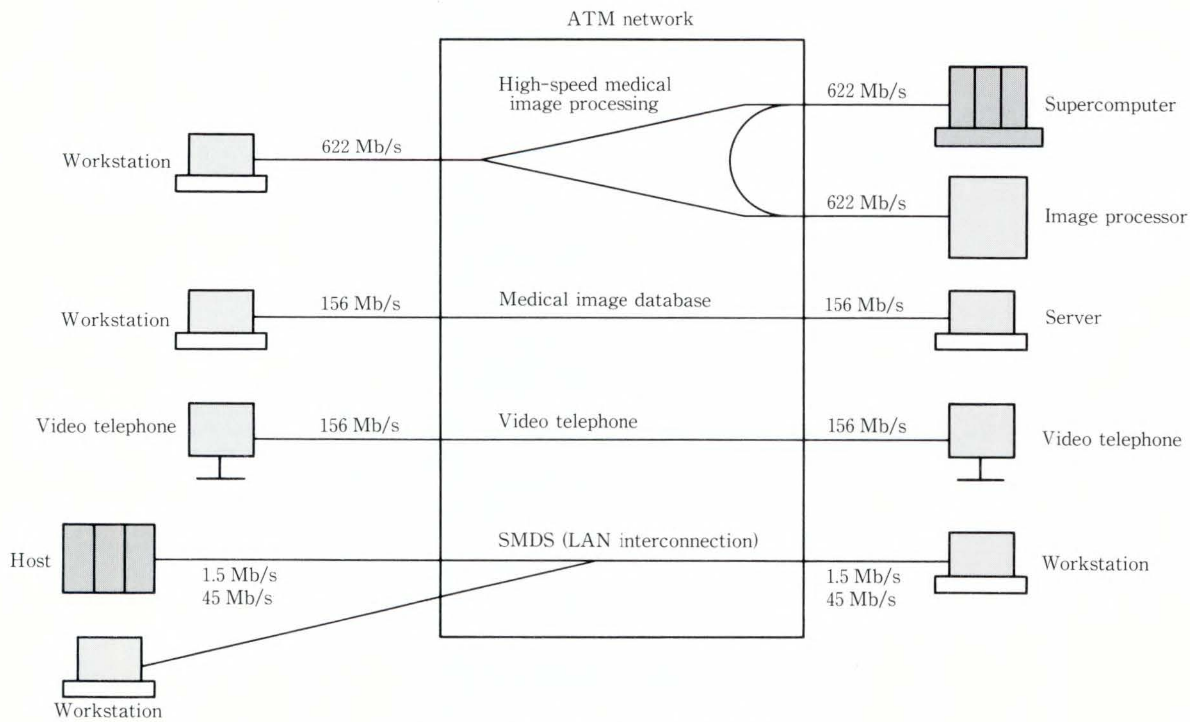



Fig. 2–B-ISDN service trial in U.S.

introduced using current transmission facilities but will be gradually integrated into ATM-based B-ISDN in future.

### 3.2 B-ISDN development history

The following is the history of development and the plans for Fujitsu B-ISDN systems:

1988: Laboratory system
1989: Prototype system
1991: Service trial system
1993: Commercial system

The 1988 Laboratory system was developed by Fujitsu Laboratories for confirmation of the ATM switching architecture called Multi-Stage Self-Routing (MSSR) switching[4]. The 1989 prototype system is an actual implementation of ATM switching capability using MSSR on the FETEX-150 platform[5].

It is said that Narrowband ISDN (N-ISDN) was developed in a rather technology-oriented way. So far the user demand for N-ISDN is less than expected because of lack of good applications which can utilize N-ISDN capabilities. In

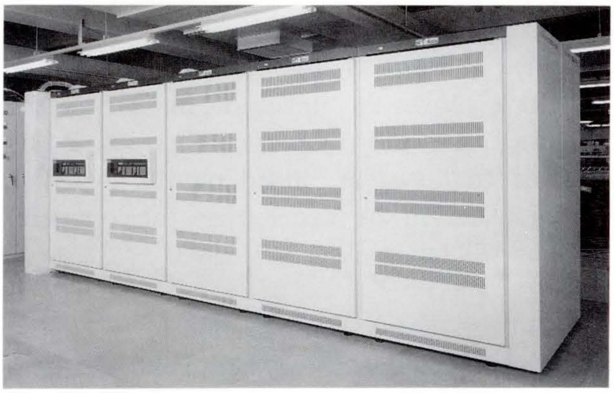
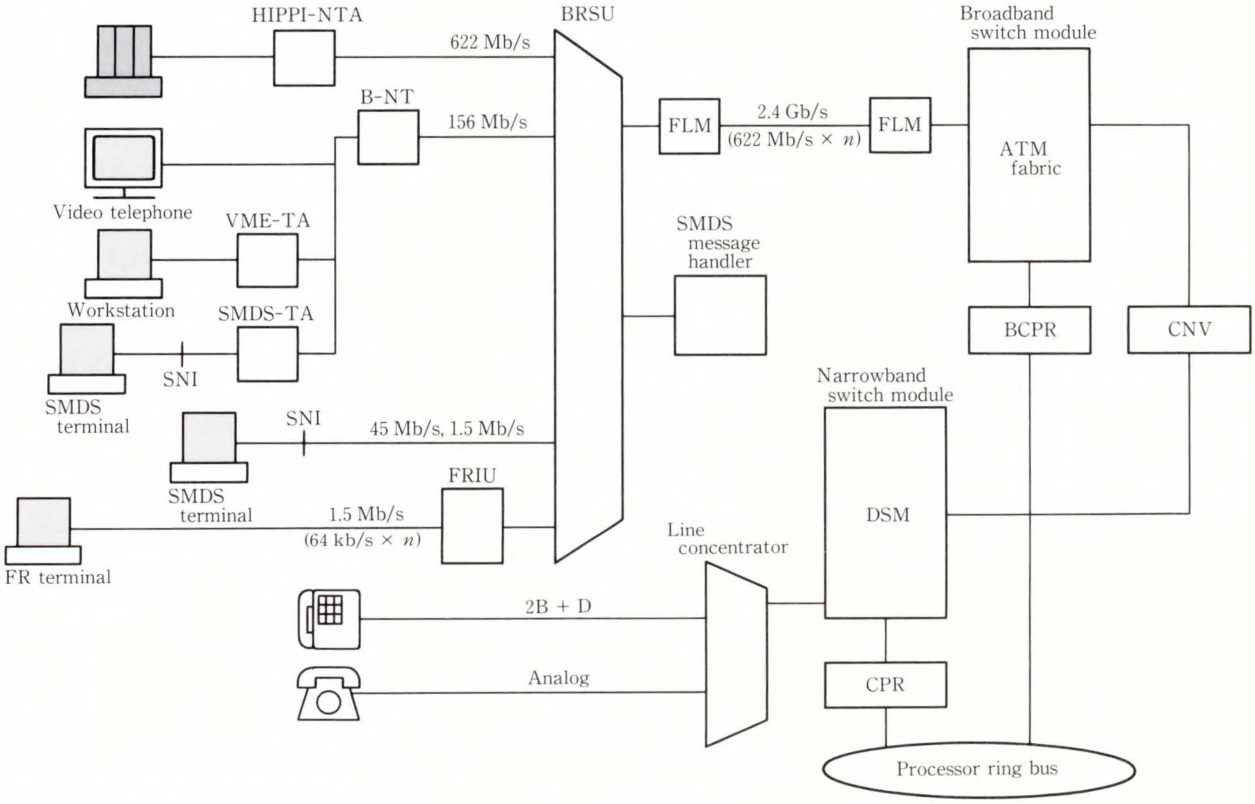


Fig. 3–External view of FETEX-150 B-ISDN service trial system.




HIPPI-NTA : High Performance Parallel Interface Network Termination and Terminal Adapter
B-NT : Broadband Network Termination
SNI : SMDS Subscriber Network Interface
VME-TA : Versa Module Europe Terminal Adapter
SMDS-TA : SMDS Terminal Adapter
FRIU : Frame Relay Interworking Unit
BRSU : Broadband Remote Switching Unit
FLM : Fiber Loop Multiplexer
DSM : Digital Switch Module
BCPR : Broadband Call Processor
CNV : ATM/STM Converter
CPR : Call Processor
OMP : Operation and Maintenance Processor

Fig. 4–System configuration of FETEX-150 with B-ISDN capability.

this context, the most important point in the development of B-ISDN is to find good applications and to encourage end customers to use B-ISDN.

The B-ISDN service trial system is now under development with the purpose of a trial operation with actual customers in a real network. B-ISDN service trial using this system is planned in the U.S. and Asia from early 1992. Figure 2 shows the image of the service trial. Figure 3 shows the external view of B-ISDN service trial system. Based on the experience accumulated in the service trials, the commercial system will be developed by 1993.

### 3.3 B-ISDN system configuration

Figure 4 shows the system configuration of the FETEX-150 with B-ISDN capability. Table 2 shows the general specifications of the B-ISDN service trial system. The broadband switch module is added to the existing narrowband switching system. An Operation and Maintenance Processor (OMP) provides unified operation, administration and maintenance functions for both N-ISDN and B-ISDN switch modules.

The broadband switch module consists of the broadband host switch and the Broadband Remote Switching Unit (BRSU). The BRSU provides an interface with subscribers and concentrates the traffic from subscribers. Various topologies can be adopted for the link between the BRSUs and the broadband host switch, e.g. star, ring, etc. using Fiber Loop Multiplexer (FLM), depending on the geographical conditions. For subscriber lines between the BRSU and customer premises, a simple star topology is adopted.

### 3.4 Implementation of pre-broadband service

As mentioned in section 3.1, the pre-broadband services such as frame relay and SMDS will be implemented before B-ISDN. The following two alternatives are possible to implement these services:

1) A switching system dedicated to frame relay and SMDS
2) An B-ISDN system based on ATM also provides frame relay and SMDS.

Considering future expandability and flexibility, the B-ISDN system based on ATM is adopted in the FETEX 150 (see Fig. 5). The

Table 2. General specification of B-ISDN service trial system

| Item | Specification |
|---|---|
| Host switch | ATM: MSSR switching method<br>Internal link speed: 1.2 Gb/s<br>Throughput: 80 Gb/s<br>(156 Mb/s x 512 ATM highways)<br>Max 64 BRSUs per host |
| BRSU | ATM concentration (1 : 1 to 32 : 1)<br>Max 256 OC-3c lines per BRSU<br>Max 64 OC-12c lines per BRSU |
| UNI | OC-3c (156 Mb/s, ATM)<br>OC-12c (622 Mb/s, ATM)<br>DS1 (1.5 Mb/s) or DS3 (45 Mb/s) for SMDS SNI<br>Customer premises bus:<br>Optical active bus based on DQDB |
| Terminal equipment and services | Video telephone<br>Terminal adapters for:<br>VME bus<br>SMDS (1.5 Mb/s, 45 Mb/s)<br>HPPI (622 Mb/s) |

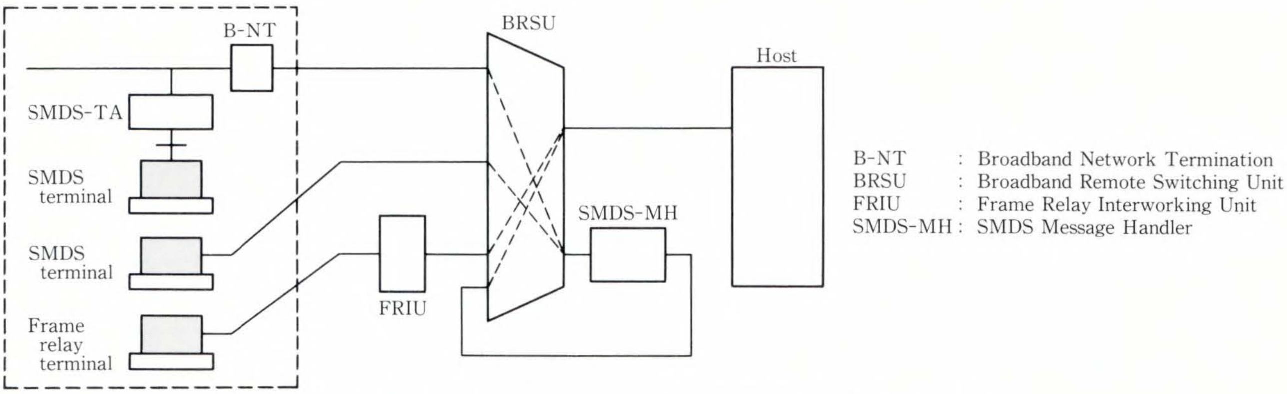



Fig. 5—SMDS and frame relay implementation in FETEX-150

frame relay service is provided with the Frame Relay Interworking Unit (FRIU), which provides conversion between the frame relay and ATM. The connection between FRIUs is provided by Virtual Channel (VC) on ATM, which is set up on a call-by-call basis.

In the case of SMDS, special eqiupment called the SMDS Message Handler (SMDS-MH) is used to provide various SMDS-oriented services such as address screening, routing of messages, group addressing (point-to-multipoint connection), and checking of illegal messages. Since the SMDS is a connectionless service, the subscriber is always connected to the SMDS-MH with a semi-permanent connection of the VC on ATM. By this configuration, the following two types of subscriber interface are uniformly provided:

1) Metallic interface (direct accommodation of subscriber interface)
2) Optical interface (common use of optical subscriber line with various ATM based terminal equipment).

**3.5 Broadband host switch**

The role of broadband host switch is to switch traffic between BRSUs and to switch traffic to and from other B-ISDN switching systems. Broadband host switch supports variable bit rate connection-oriented communication services of up to 622 Mb/s.

The ATM fabric is called Multi-Stage Self-Routing (MSSR) switching, which consists of three stages of Self-Routing Modules (SRMs). The throughput of internal highway is 1.2 Gb/s. The SRM consists of specially designed BiCMOS VLSI[5]. Each MSSR switching module can accommodate eight 1.2-Gb/s input/output highways. By grouping up to eight MSSR switching modules, a maximum of sixty four 1.2-Gb/s input/output highways can be provided, giving approximately 80 Gb/s throughput at maximum configuration. This is equivalent to 512 of 156-Mb/s ATM channels. Figure 6 shows the ATM switch card of the service trial system.

The broadband host switch is connected to BRSUs via SONET STS-12c fiber optic interface. If the BRSU is placed next to the host switch, SONET STS-12c direct connection is possible. The number of fiber optic interfaces depends on the traffic between the host switch and each BRSU.

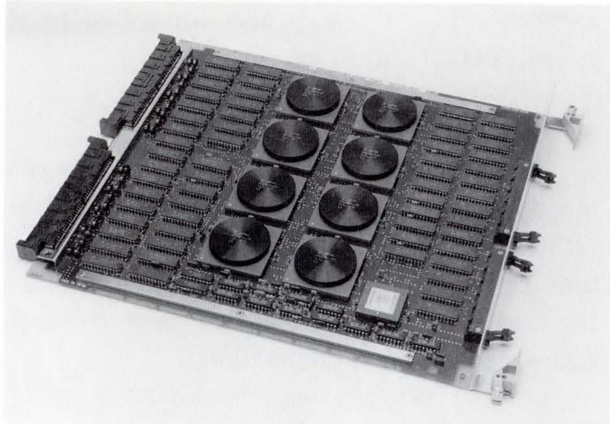


Fig. 6—ATM switch card.

**3.6 Broadband Remote Switching Unit (BRSU)**

The BRSU consists of the ATM concentration switching unit, the subscriber interface, and the host switch interface. Concentration switching unit consists of a single stage or two-stage SRM which is the same as the SRM used in host switch. Concentration switching unit has a total of thirty-two 1.2Gb/s input/output highways at maximum configuration where subscriber interfaces are accommodated. The concentration ratio is determined by the ratio of the total throughput of subscriber interfaces vs. the total throughput of the host switch interface.

The following four types of subscriber interface are provided:

1) 156-Mb/s single fiber optic interface,
2) 662-Mb/s single fiber optic interface,
3) 1.5-Mb/s DS1 metallic interface and
4) 45-Mb/s DS3 metallic interface.

For subscribers with User-Network Interface (UNI) standardized in CCITT, a 156-Mb/s or 622-Mb/s single fiber optic interface with Wave Length Division Multiplexing (WDM) is provided. A policing function is provided for each subscriber interface. For subscribers requiring a direct interface with the SMDS Subscriber Network Interface (SNI), 1.5-Mb/s (DS1) and 45-Mb/s (DS3) 4-wire metallic interfaces are provided. The subscriber requiring a frame relay service is also connected to the 1.5-Mb/s (DS1) interface.

### 3.7 Customer premises equipment

There are two types of optical subscriber interfaces, 156 Mb/s and 622 Mb/s. The interface of 622 Mb/s can connect only one terminal unit per subscriber line for simplicity. This is because this interface is used only for super high-speed data communication.

On the other hand, the interface of 156 Mb/s can connect multiple terminal units and Customer Premises Network (CPN) is provided at the customer premises. The CPN configuration and the media access protocol are currently being studied by various standards organizations. For a service trial system, the bus configuration is adopted. The technique of distributed control using request bits and busy bits, which is similar to Distributed Queue Dual Bus (DQDB) protocol, is adopted.

The following four kinds of terminal equipment are developed.

1) Terminal Adapter with Versa Module Europe bus interface (VME-TA)

The VME-TA enables the high-speed data communication of ATM for current workstations/computers with VME bus interfaces.

2) Terminal Adapter with SMDS interface (SMDS-TA)

The SMDS-TA can connect the SMDS terminal equipment and provide SMDS on 156-Mb/s B-ISDN suscriber lines.

3) Terminal Adapter with High Performance Parallel Interface (HIPPI-TA)

The super high-speed data communication is provided by the HIPPI-TA which has the HIPPI interface. Since the maximum speed of HIPPI is 800 Mb/s, the HIPPI-TA has a large buffer memory to connect the subscriber line of 622 Mb/s without the performance deteriorating.

4) Video telephone

The video telephone provides voice and moving color picture communication.

## 4. Intelligent network capability

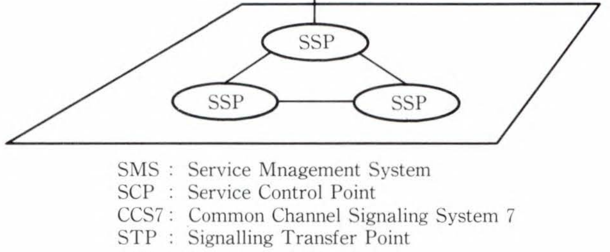
### 4.1 Standardization and market trends

The concept of the Intelligent Network (IN) was originally developed by the U.S. Bell Operating Companies and its standardization has been promoted by Bellcore[7]. The IN aims at the following points:

1) Introduction of database common to the whole network which enable network-wide services
2) Separation and centralization of service control functions which enable rapid introduction of new services and easy modification of services
3) Introduction of a service creation environment which enables development of services by telephone companies

Bellcore has completed the standardization of the initial component of IN called IN/1, which mainly covers network-wide databases. Now Bellcore is promoting the standardization of Advanced Intelligent Network (AIN) which will cover the separation of service control functions and the service creation environment. Recently CCITT also started the standardization of AIN, and recommendations are expected to be issued in 1992, 1993 and 1996[8].

Many countries are now eagerly introducing the IN capability to increase revenue by introducing advanced services using IN/1, and are waiting for the standardization of AIN, expecting a telecommunication network with a new structure which enables various new services.




SMS : Service Mnagement System
SCP : Service Control Point
CCS7 : Common Channel Signaling System 7
STP : Signalling Transfer Point
SSP : Service Switching Point

Fig. 7–IN/1 network configuration in FETEX-150.

**4.2 IN/1 implementation**

The implementation of IN capability on the FETEX-150 started with IN/1, which was requested by many countries using FETEX-150. Figure 7 shows the FETEX-150 IN/1 network configuration.

The Service Switching Point (SSP) is the FETEX-150 switching system itself with software modification to interwork with the Service Control Point (SCP). The SCP is equipped with a network-wide database and provides information necessary for the services in response to inquiries from the SSP. The SCP is configured based on the FETEX-150 architecture using its telecommunication-oriented processor.

**4.3 Evolution towards AIN**

Generally speaking, AIN consists of four elements: Service Switching Point (SSP), Service Control Point (SCP), Service Management System (SMS) and Service Creation Environment (SCE). Unlike IN/1, general-purpose computers are used in the SCP, SMS and SCE.

Figure 8 shows the plan for migrating from IN/1 to AIN. AIN will be realized in the following steps:

Step 1: Introduction of new AIN services

The AIN platform is introduced over existing IN/1. Some new AIN services are introduced on the AIN platform. Current IN/1 services are still provided on the existing IN/1 platform to prevent conflicts with current services. A limited number of SSPs are upgraded for AIN functions.

Step 2: Nationwide AIN

The conventional IN/1 SCP is upgraded to AIN SCP by introducing general-purpose computers. The local SCP (L-SCP) will be introduced in addition to the centralized SCP (C-SCP). The SMS is introduced to manage a number of SCPs and the SCE is introduced to enable service creation. Existing IN/1 services are all transported to the AIN platform and all SSPs are upgraded for AIN capability.

**4.4 AIN Service Control Point (SCP) architecture**

The AIN SCP architecture consists of two subsystems, the Signaling Subsystem (SGS) and the Service Logic Control Subsystem (SLCS) as illustrated in Fig. 9.

The SGS is configured with the standard FETEX-150 system and terminates Common Channel Signaling System 7 (CCS7) and selects SLCSs. The SGS communicates with SLCSs by

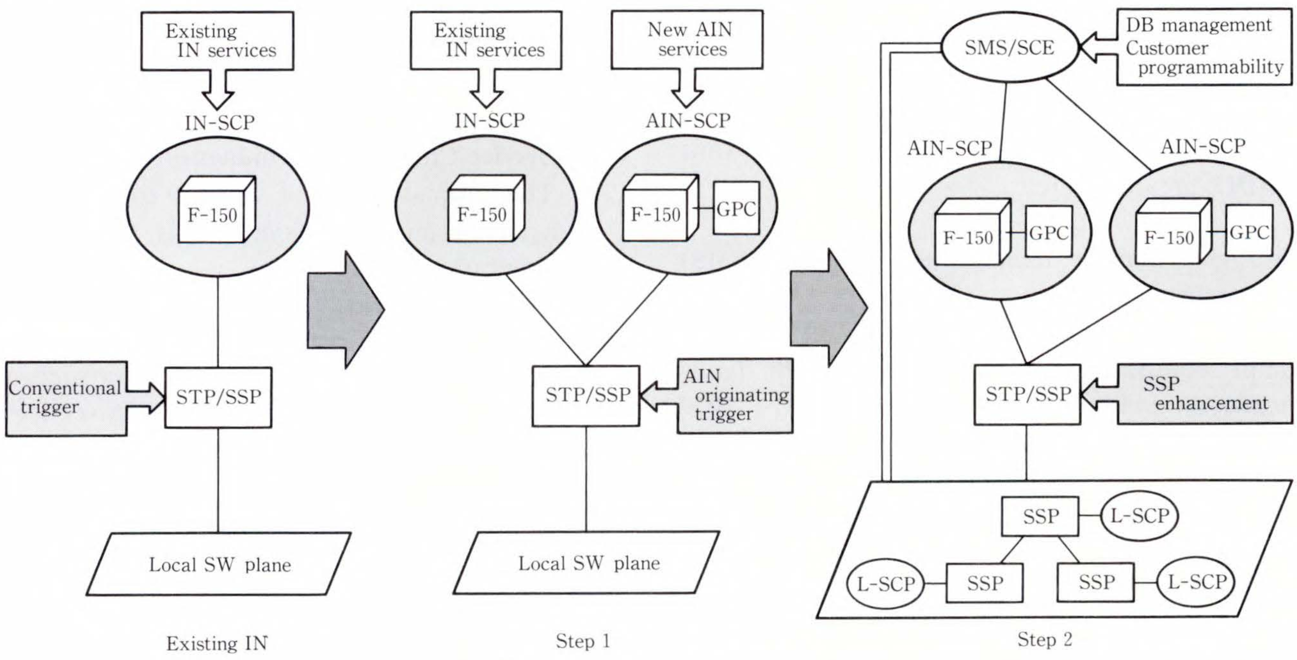



Fig. 8–Migration from IN/1 to AIN.

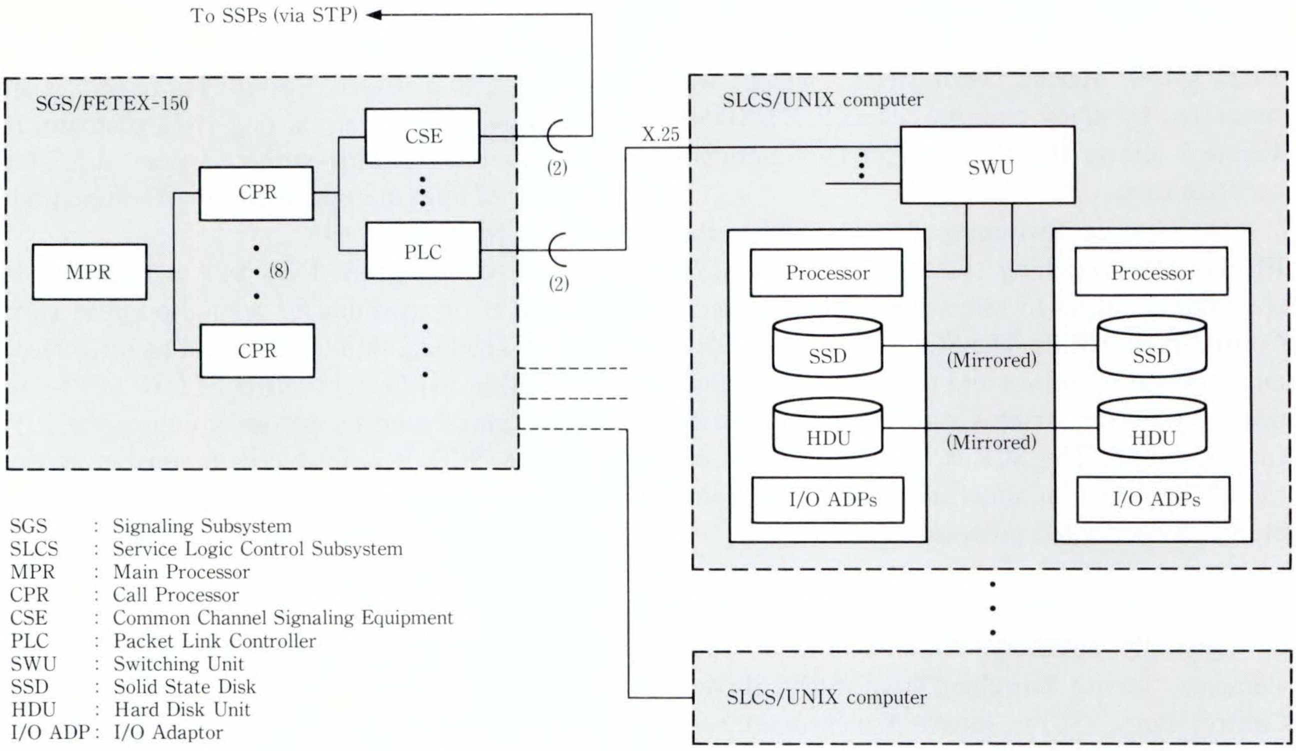



Fig. 9—AIN SCP hardware configuration.

using IN application protocol defined on Transaction Capability Application Part (TCAP) over X.25 links, Ethernet or FDDI according to the service traffic growth.

The SLCS is configured with duplicated UNIX based super-mini computers, which operate in the hot-standby mode. An external interface such as X.25 links are switched in the Switching Unit (SWU) and mass-storage such as Solid State Disk (SDD) and Hard Disk Unit (HDU) are duplicated.

### 4.5 AIN Service Management System (SMS) architecture

The SMS architecture consists of a super-mini computer and workstation with fault-tolerant architecture, the same as the SCP architecture. The SMS provides the following functions:

1) Updating and backing up the database in the SCP
2) Controlling the activation/deactivation of services
3) Customizing the database by service users and downloading to each SCP
4) Supervising the remote operation of the SCPs
5) Collecting and statistically processing service traffic handled by the SCP

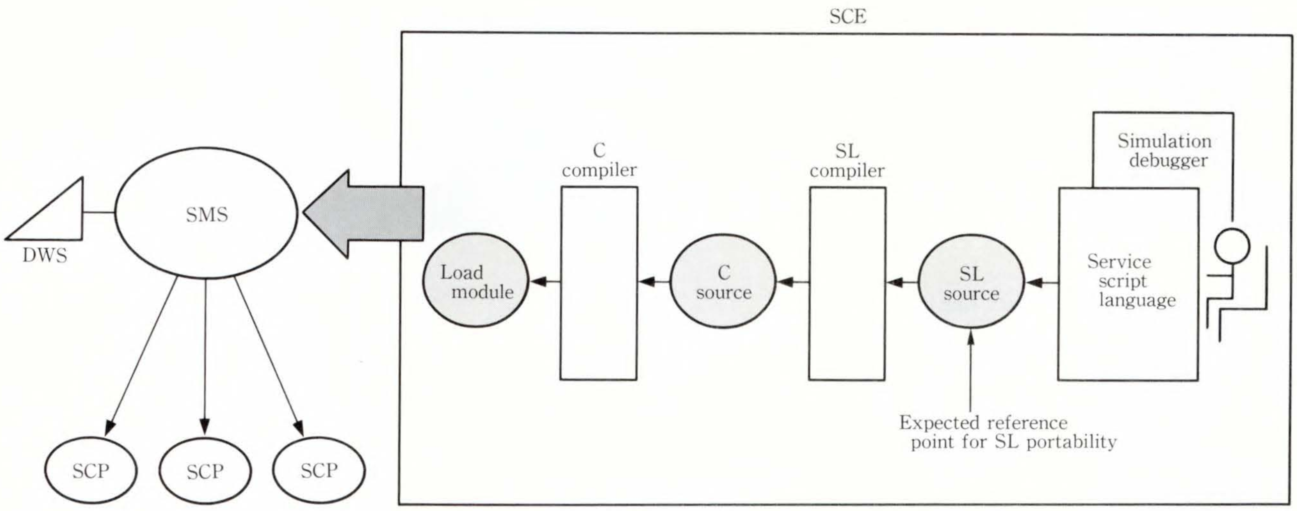
The workstation called Database Management Workstation (DWS) provides a user-friendly human-machine interface for customization databases.

### 4.6 Service Creation Environment (SCE)

The ultimate goal of AIN is to realize an open programming environment with full customer programmability.

To utilize the full capabilities of AIN, an enhanced Service Creation Environment (SCE) is required which includes capabilities such as planning, managing projects, designing services, and programming and debugging. The SCE is intended to be able to be programmed and debugged by people who are not programming experts. The SCE hardware consists of a super-mini computer and workstation. It is also possible to integrate both SMS and SCE in the same hardware platform.

The service creation process is shown in




DWS : Database Management Workstation    SCE : Service Creation Environment

Fig. 10–Service creation process.

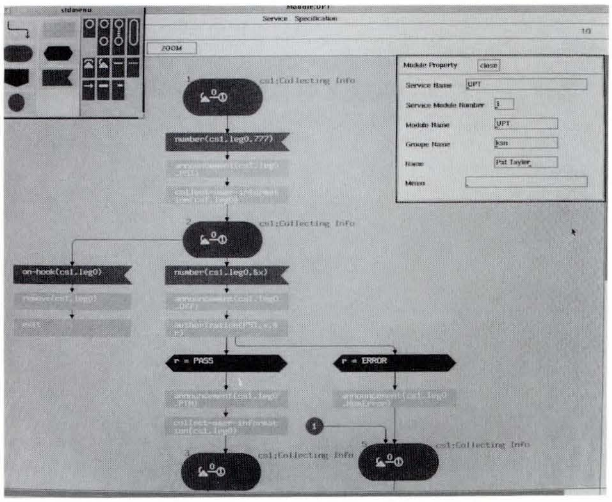


Fig. 11–Example of service script.

Fig. 10. A service programmer programs the service script using the Service Script Language which is specially developed for telecommunication service programming. An example of service script is shown in Fig. 11. The produced service script is called the Service Logic (SL) source, which is debugged by a simulation debugger on the workstation called the Service Programming Workstation (SPWS). Then the SL source is compiled into C or C++ source code by the SL compiler, then compiled by the C compiler into a load module. The load module is called the Service Logic Program (SLP), and is transferred to each SCP through the SMS.

## 5. Conclusion

This paper describes recent enhancements of the FETEX-150 digital switching system. The FETEX-150 will evolve to a multi-service platform which can provide services for various media and interfaces. Through this enhancement, FETEX-150 will evolve into a next generation switching system with extended flexibility and capabilities.

## References

1) Fujiyama, Y. et al.: Architecture Enhancement of Digital Switching System. Proc. ISS '90, **Vol. V**, 1990, pp. 217-220.

2) Walters, S.M.: A New Direction for Broadband ISDN. *IEEE Commun. Mag.* **29**, 9, pp. 39-42 (1991).

3) CCITT: "Broadband Aspects of ISDN". Draft Recommendation I.121, Study Group XVIII, Geneva, May 1990.

4) Hajikano, K. et al.: Asynchronous Transfer Mode Switching Architecture for Broadband ISDN. Proc. ICC '88, **Vol. 2**, 1988, pp. 911-915.

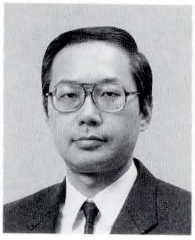
5) Takahashi, A. et al.: A Broadband Switching System for Public Network. Proc. ISS '90, **Vol. V**, 1990, pp. 103-109.

6) Kato, Y. et al.: A VLSIC for the ATM Switching System. Proc. ISS '90, **Vol. III**, 1990, pp. 27-32.

7) "Advanced Intelligent Network Release 1 Network and Operation Plan". Bellcore Special Report, SR-NPL-001623, June 1990.

8) CCITT: "Baseline Document for Intelligent Network Studies at CCITT". SG XI-R24, April 1990.

**Akihiro Sera**
Overseas Switching Hardware Development Dept.
Switching Systems Div.
FUJITSU LIMITED
Bachelor of Electrical Eng.
Hiroshima University 1968
Specializing in Switching Hardware



**Yoichi Fujiyama**
Overseas Systems Engineering Dept.
Switching Systems Div.
FUJITSU LIMITED
Bachelor of Applied Physics
Tokyo Institute of Technology 1973
Specializing in Switching Systems Engineering

**Toru Masuda**
Overseas Software Development Dept.
Switching Systems Div.
FUJITSU LIMITED
Bachelor of Electrical Eng.
The University of Tokyo 1968
Specializing in Switching Software

UDC 621.395.74.037.3.072.9

# Synchronous Digital Network Systems

● Takeshi Sakai ● Yoshikuni Toko ● Akira Tokimasa

*(Manuscript received Oct. 31, 1991)*

The Synchronous Digital Hierarchy (SDH) is the world standard recommended by CCITT in 1988. The Synchronous digital network system based on the recommendations introduces flexibility or plane expansion to the existing point-to-point network and has enhanced the versatility to cope with changes in demand and improved network operation and maintenance. Furthermore, it will become a platform for the implementation of broadband ISDN. Fujitsu has developed SDH-based fiber optic and microwave radio transmission systems conforming to the Japan and North America specifications. This paper outlines the characteristics and aims of the SDH, the concept for the new SDH-based synchronous digital network, and the developed SDH systems.

## 1. Introduction

Technological advances such as fiber optic line transmission and multilevel modulation for microwave radio transmission have enabled the analog transmission systems supporting the telecommunications infrastructure to become digital. Such systems now provide transmission capacities of gigabits ($10^9$ bits) per second, as opposed to megabits ($10^6$ bits) per second. This has contributed to a significant cut communication costs. The "big wire" concept of the conventional point-to-point digital transmission system aimed at an efficient telephone signal transmission. However, it was unable to fully address networking needs, such as flexibility in meeting changing network demand, high reliability and enhanced operation, and compatibility with new high-speed broadband services.

A key technology to resolve these problems is synchronization of the entire network, which introduces network flexibility or network plane expansion and moves away from the "big wire" concept. In 1988, CCITT established the synchronous digital interface standard supporting synchronous multiplexing in the form of the Synchronous Digital Hierarchy (SDH)[1]. In North America, this took form as the Synchronous Optical Network (SONET)[2]. Such standardization has promoted the worldwide development of synchronous digital network systems.

In 1988, Nippon Telegraph and Telephone Corporation (NTT) started development of a new network based on the SDH[3),4]. Fujitsu developed fiber optic and microwave radio transmission systems for the new network[5]. Similarly, in the U.S., the Bell Operating Companies (BOCs) were working on transmission systems built to the SONET standards. Fujitsu also developed the SONET transmission system to address this need[6].

## 2. Synchronous multiplexing scheme

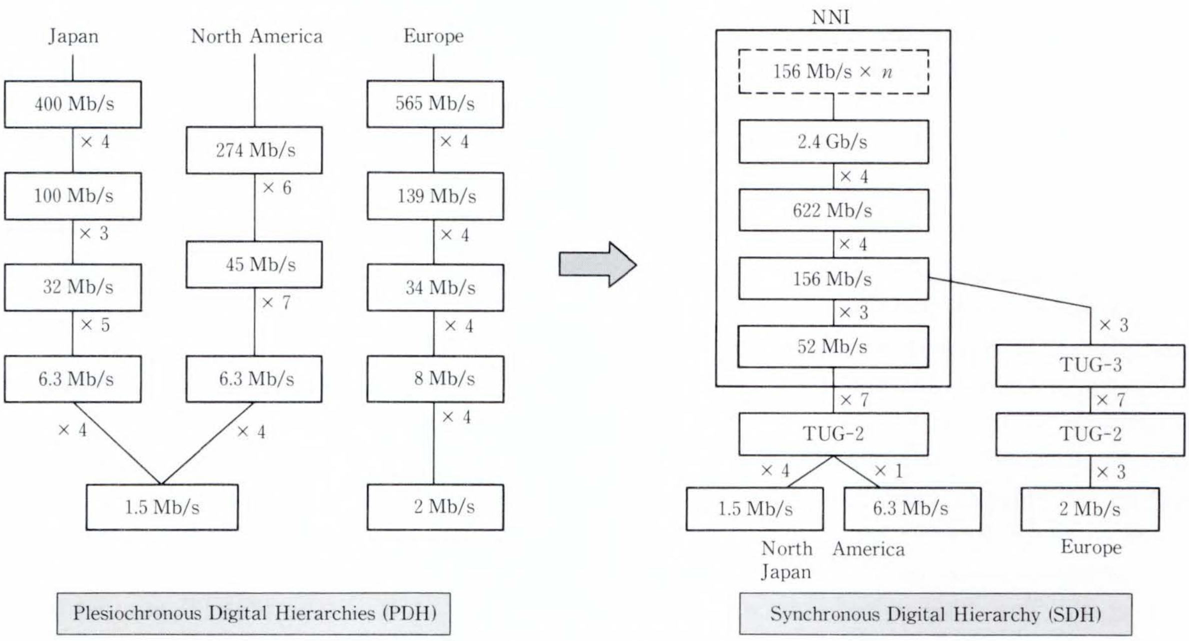
Conventional transmission technology was mainly concerned with communicating information as cheaply as possible between two remote points. This involved a single transmission line packed with pulse streams (digital multiplexing) to enable higher line use and cut the cost per line. The two types of digital multiplexing now used are justification multiplexing and synchronous multiplexing.

Justification multiplexing involves insertion or extraction of extra pulses, called justification pulses, to make slightly different bit rates of pulse streams uniform. Synchronous multiplexing involves distributing highly stable clock pulses across the network to equalize the bit rate at all points.

Justification multiplexing is limited to point-to-point communication because I/O signal pulse streams (digital multiplexing signals) are nonsynchronous among multiplexers. Synchronous multiplexing enables skip-level multiplexing to an optional rate or free routing of specific pulse streams among the multiplexing signals with the electronic switch (digital cross-connect). This is possible because all signal pulse streams are synchronous. Synchronous multiplexing enables an entire network to be comprehensively managed.

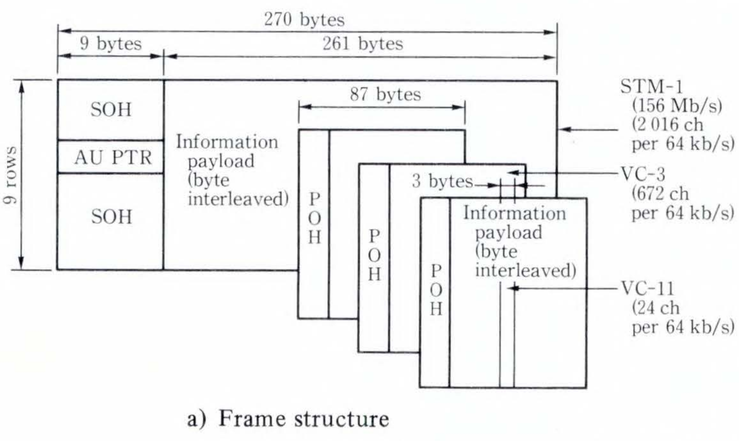
## 3. Synchronous Digital Hierarchy (SDH)

Multiplexing bit rates are arranged in what is called a digital hierarchy. Three main hierarchies have been implemented involving bit rates not synchronized between the levels of the hierarchies, because they were based on justification multiplexing. Such hierarchies are called the Plesiochronous Digital Hierarchies (PDH). The internationally standardized SDH operates using a rate which is a multiple of the




TUG-$n$: Tributary Unit Group level-$n$

Fig. 1—Digital hierarchy.




a) Frame structure

| | |
|---|---|
| SOH | Reframing for STM-1 frame<br>Line error performance monitoring<br>Section maintenance signals<br>OAM data communication channels<br>Protection switching control |
| AU PTR | Indication of VC-3 head<br>Frequency justification |
| POH | Path error performance monitoring<br>Path maintenance signals |

SOH : Section Overhead
POH : Path Overhead
AU PTR : Administrative Unit Pointer
OAM : Operation, Administration and Maitenance

b) Overhead and function

STM-1 : Synchronous Transport Module level-1
VC-$n$ : Virtual Container level-$n$

Fig. 2—Frame structure of 156 Mb/s synchronous interface.

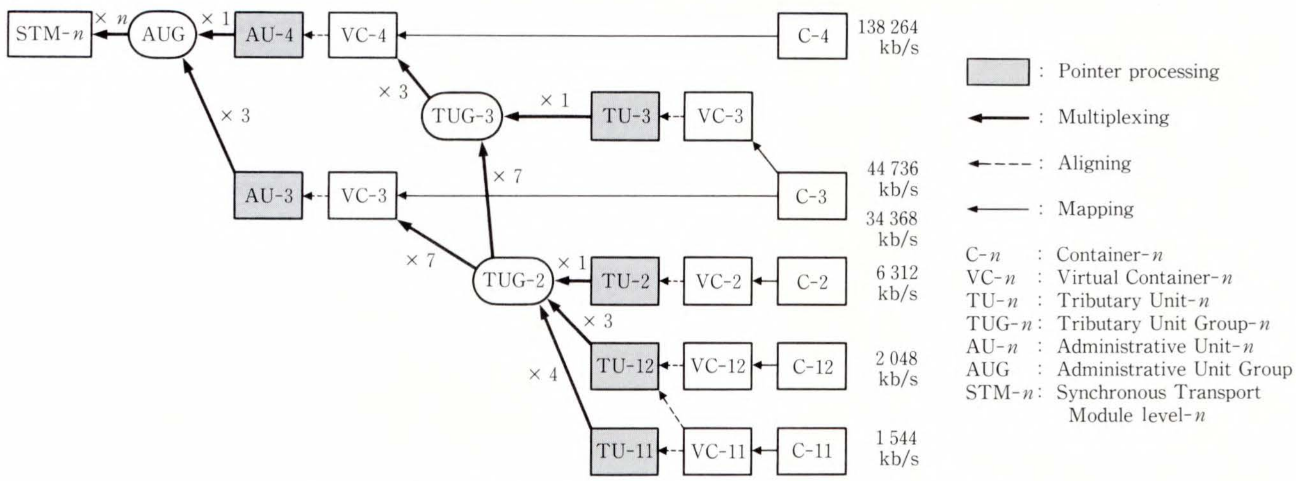



Fig. 3—Multiplexing structure of NNI. Source: reference 9).

basic 156 Mb/s rate (see Fig. 1). CCITT has standardized interfacing in the form of the Network Node Interface (NNI). This has also been standardized as SONET in North America by the T1 Committee and in Japan by the Telecommunication Technology Committee (TTC)[7]. In North America and Japan, a low speed of 52 Mb/s has been added to the SDH in addition to 156 Mb/s.

Figure 2 shows the NNI frame structure. Figure 3 shows its multiplexing structure. The NNI implements three new concepts:

1) Phase synchronization using pointers,
2) Virtual containers used as multiplexing units, and
3) Overhead bytes enhancing Operation, Administration, and Maintenance (OAM).

In phase synchronization, the frame phase of input signals is adjusted to the specific time location by accommodating delay variations or phase changes in the input signal in the transmission line, such as jitter and wander, within the end office equipment. Phase synchronization using pointers transfers the address indicating the starting position of the multiplexing signal. Synchronization is maintained by updating this pointer. Because the timing relationship between frame synchronizing pulses and multiplexing signals is not fixed, unlike in current phase synchronization, less buffer memory is needed to absorb phase changes in the transmission line, thus reducing the processing time and downsizing circuits.

Unlike in the conventional scheme, a structure in which low-speed signals are multiplexed into a high-speed signal does not directly depend on a specific rate or interface, i.e. low-speed signals are multiplexed in standardized multiplexing units called "virtual containers". This makes it easy to multiplex low-speed signals for the conventional system and diverse signals for new services. Virtual containers support both plesiochronous interfaces using justification multiplexing and lower-level synchronous interfaces connected to a digital switch. Plesiochronous signals are multiplexed by positive and negative frequency justification, which can be fine-tuned to accommodate phase changes in the transmission line.

The NNI is divided into a hierarchy of sections and paths managed by the transmission network. The NNI maintains an area for transferring OAM information on each of these sections and paths, called an overhead. Overheads are used to help supervise the network's operating status, detect problems such as transmission quality degradation, localize failures, and back up failed transmission lines automatically. Supervisory and control information is transferred via data communication channels.

## 4. New synchronous digital network

The purpose of the new SDH-based synchronous digital network is to make "big wire"

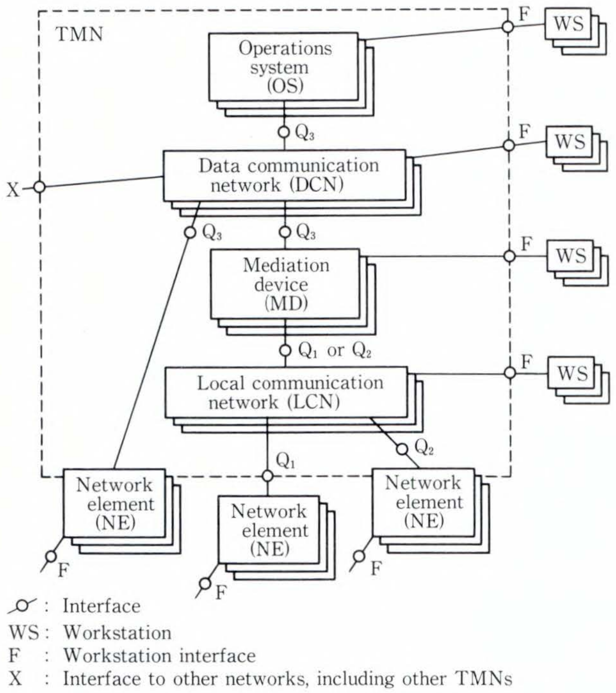



Fig. 4—Physical architecture for TMN. Source: reference 10).

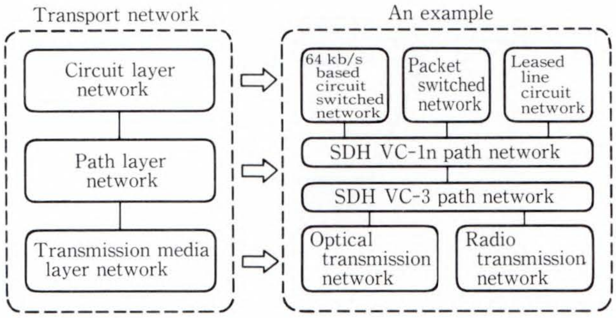



Fig. 5—Layered model of the transport network. Source: reference 11).

networks more efficient and easier to operate and manage. To develop such networks, the architecture is introduced to organize network elements into a layered hierarchy for functions and managed objects.

To upgrade network management and make systems easier to expand, network elements are divided into transport and network management layers. The network management layer includes the Telecommunication Management Network (TMN), for example, whose standardization the CCITT is studying. The TMN is a hierarchical and open information network using enhanced information processing technologies such as object-oriented and distributed processing. Its interface with a transmission system, called the Q interface, consists of seven OSI protocol layers (see Fig. 4).

The three transport layers are the circuit layer defined between service nodes such as switches, the transmission medium layer which does the actual transmission, and the path layer, a unit of network operation shared by transparent services and transmission media (see Fig. 5). These layers are designed, operated, and managed independently.

Transmission media include optical fiber and microwave radio. Transmission line interface rates are standardized at 156 Mb/s × n. Paths are in SDH multiplexing units (1.5 Mb/s or 52 Mb/s, for example).

The transmission network elements are standardized and tailored for the target network topology and network size to flexibly configure the equipment: Transmission line termination, multiplexing of low-speed interfaces into the SDHs, and digital cross-connection and add-dropping to optimize the use of resources such as paths. The Add/Drop Multiplexer (ADM) is especially useful in ring network operations. These functions are used in actual devices either individually or in combination.

## 5. SDH system features

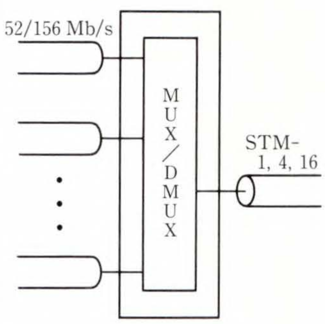
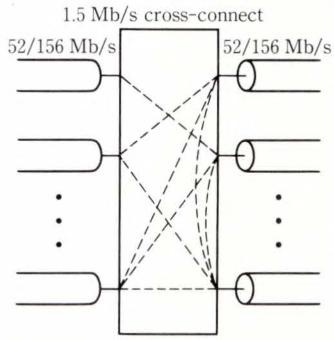
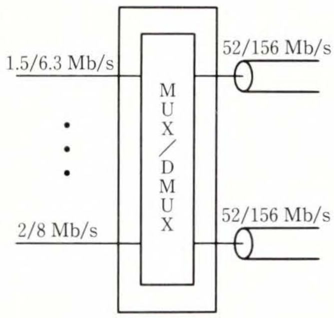
This section describes the SDH transmission system and microwave radio system for Japan, and the SONET-standard transmission system for North America developed by Fujitsu.

### 5.1 SDH transmission system

Japanese networks were previously synchronized with the digital hierarchy's 1.5 Mb/s and 6.3 Mb/s rates. Rates of 32 Mb/s and above used a plesiochronous network and justification multiplexing. These have been replaced with an SDH network.

New network elements include module A, long-haul fiber optic transmission equipment, module B, a digital cross-connection operating on paths at 1.5 Mb/s, and Module C, an SDH skip-level multiplex equipment with 1.5 Mb/s ADM function (see Table 1 and Fig. 6). Figure 7

Table 1. Features of modules A, B, and C

| Features \ Module | A | B | C |
|---|---|---|---|
| Function | **Long-haul fiber optic transmission**<br>Direct multiplexing from intra-office 52/156 Mb/s signals to STM-1 (156 Mb/s), 4 (622 Mb/s), 16 (2.4 Gb/s) | **Digital cross-connect**<br>1.5 Mb/s circuit cross-connect between 52/156 Mb/s signals | **Skip-level multiplexing**<br>Skip-level multiplexing from existing lower-speed signals to 52/156 Mb/s signals |
| Configurations |  |  |  |
| Number of circuits accommodated (per 64 kb/s voice channel) | 156 Mb/s: 16 128 ch/bay<br>622 Mb/s: 24 192 ch/bay<br>2.4 Gb/s: 32 256 ch/bay | 24 192 ch/M-bay (one-way)<br>Max 193 536 ch<br>(M-bay × 8 & J-bay × 2) | 10 080 ch/bay |
| Bay configuration | Self-support INS cabinet: 800(*l*) × 600(*b*) × 1 800 (*h*) (mm) | | |

STM-*n*: Synchronous Transport Module level-*n*
MUX/DMUX: Multiplexer/Demultiplexer
M-bay: Multiplexer-bay
J-bay: Junctor-bay

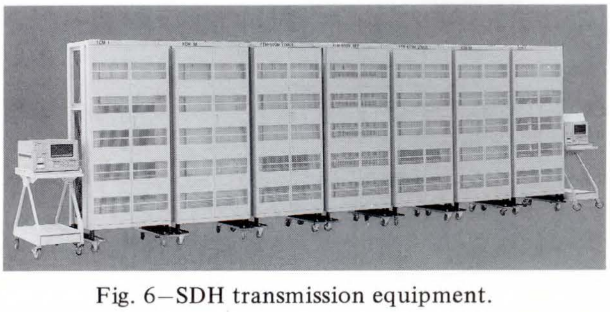


Fig. 6—SDH transmission equipment.

shows the new synchronous network configuration.

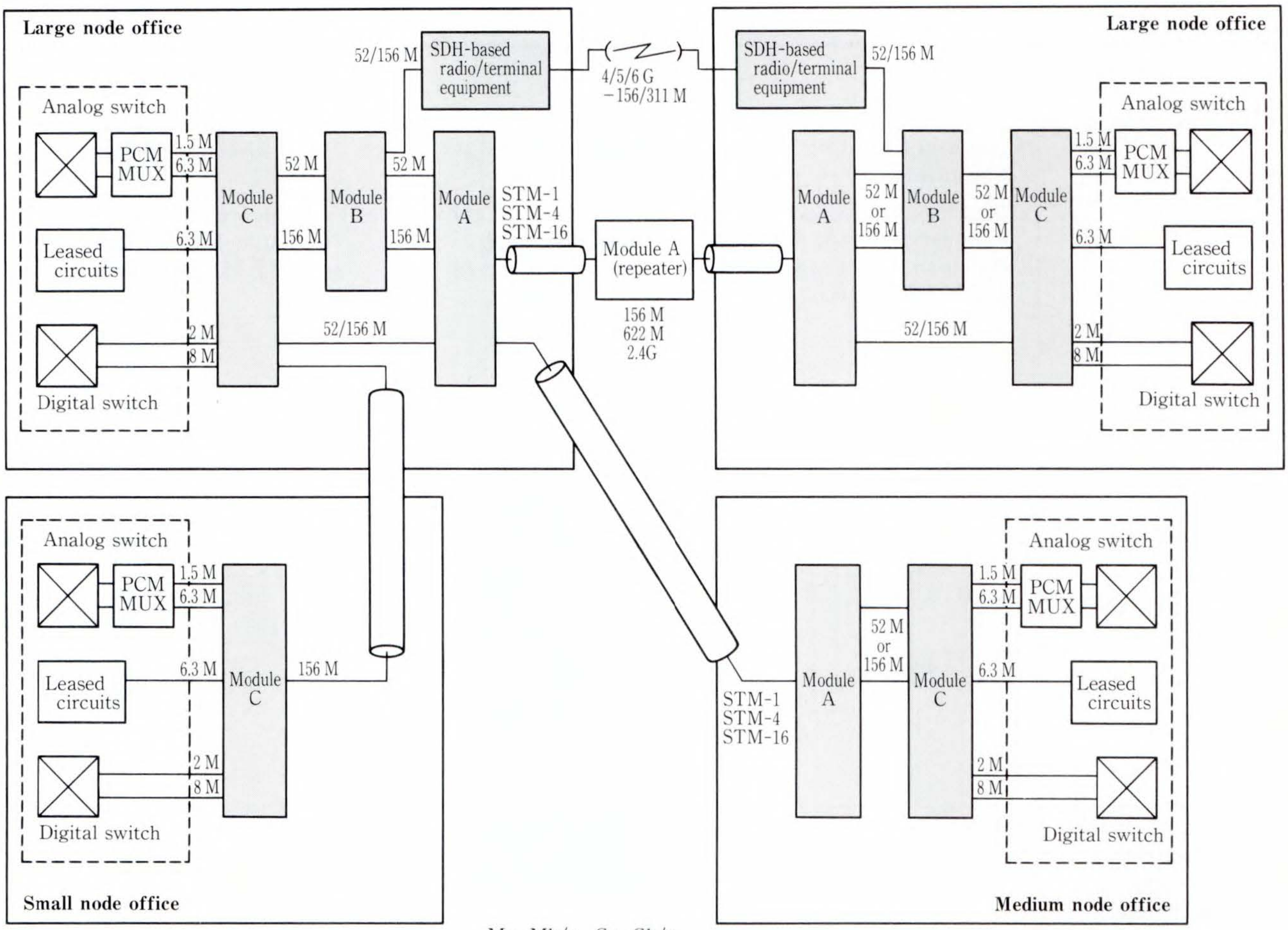
Intra-office interfaces that connect modules are standardized in the 52 Mb/s or 156 Mb/s optical interface. The fiber optic transmission line interface for module A supports rates of 156 Mb/s, 622 Mb/s, and 2.4 Gb/s and wavelengths of 1.31 $\mu$m and 1.55 $\mu$m to enable transmission as far as 40 km and 80 km respectively.

Module B's cross-connect capability is implemented by a combination of time and space switches (T + TST). It offers a maximum non-blocking switch capacity of 1.5 Mb/s by 8 064.

Module C multiplexes four lower-level synchronized interfaces from 1.5 Mb/s to 8 Mb/s into optional Virtual Containers (VC-11 and VC-2) on the SDH interface.

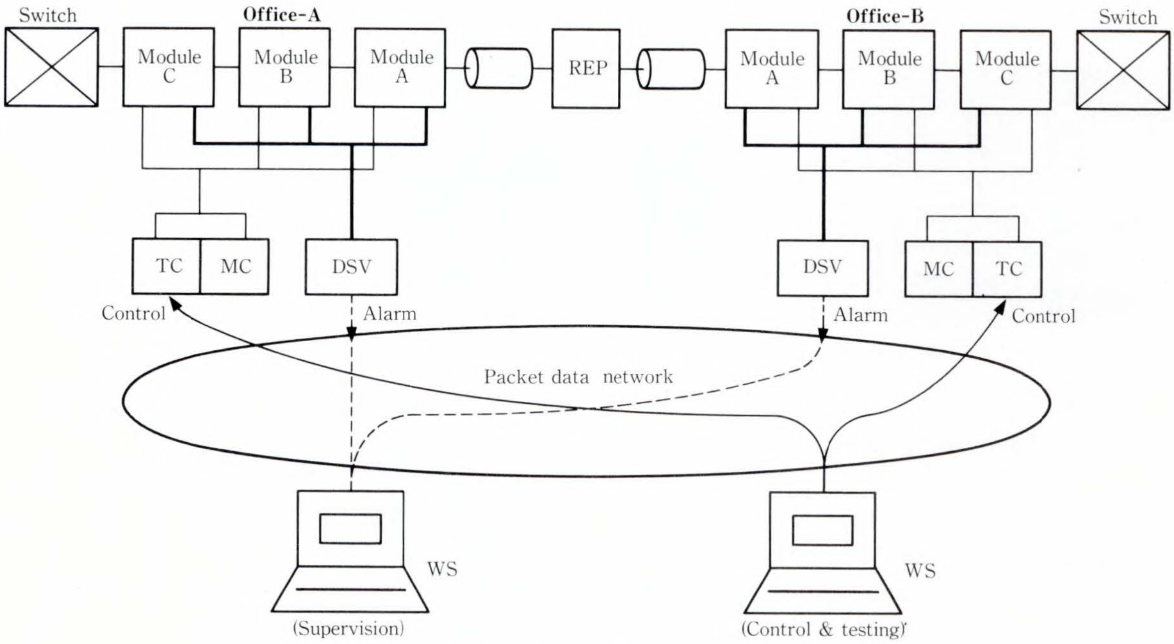
The new transmission system simplifies network maintenance and operation tasks using the following features;

1) automatic recovery within failed equipment and automatic line protection using SDH overheads,
2) remote monitoring of network performance and status logs,
3) remote provisioning of equipment operation modes and parameters, such as cross-connect setting, and




M: Mb/s G: Gb/s

Fig. 7—An example of new synchronous network configuration.




DSV : Digital Supervisary Equipment
TC : Terminal Control Equipment
MC : Measurement Control Equipment
WS : Workstation
REP : Repeater

Fig. 8—An example of operation system for SDH network.

4) remote testing of optional transmission paths.

These functions are placed under the centralized transmission network operation system over a broad network (see Fig. 8).

### 5.2 SDH microwave radio system

In Japan there are principally used 4-, 5-, and 6-GHz bands for long-haul trunk lines, while 11- and 15-GHz bands are used for short-haul applications in spur lines and local areas. The newly developed synchronous digital radio equipment series with SDH interface operating in the above frequency bands, which features 256 and 16 QAM for superior spectral efficiency in high-capacity trunk line applications, is applied for various links. Table 2 summarizes the features of SDH microwave radio equipment, and Fig. 9 shows the 256 QAM digital radio equipment.

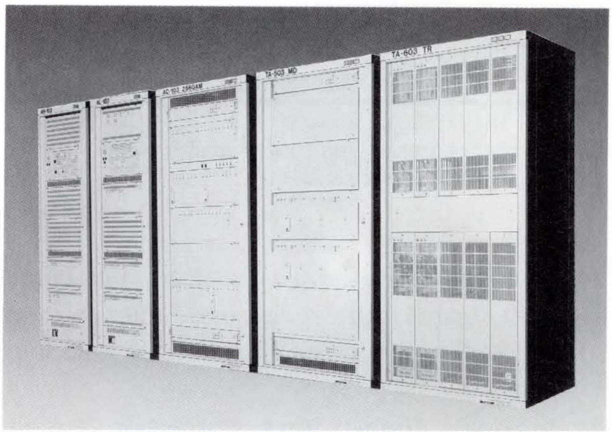


Fig. 9—SDH 256 QAM digital radio equipment.

Because radio uses limited-frequency resources, its compatibility with conventional systems is important. It supports bit rates such as 156 Mb/s x 1 or 2 for the new synchronous interface in the same frequency allocations, such as 4, 5, and 6 GHz, as existing systems.

In the radio system, a line switching section consists of two terminal stations and several relay stations. Figure 10 gives an example of the system configuration of the trunk transmission (156 Mb/s) method using the 4-, 5-, and 6-GHz

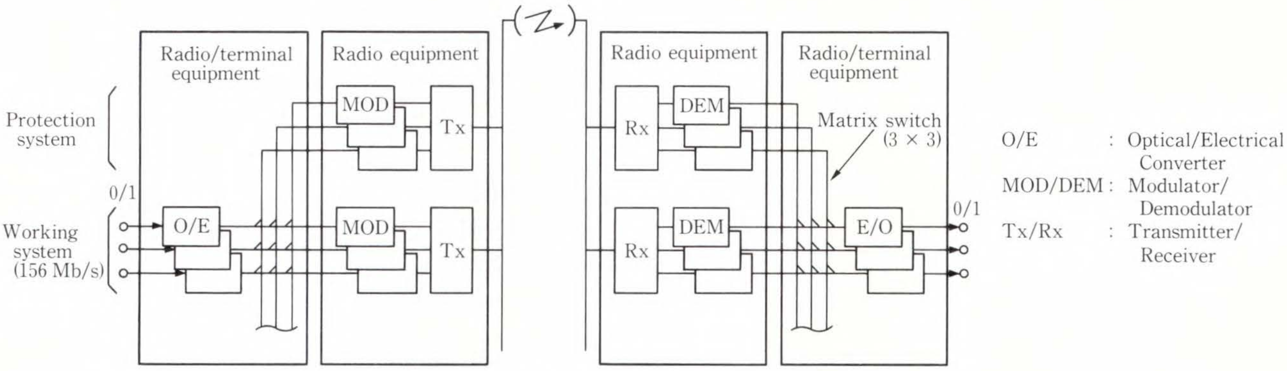



Fig. 10—Block diagram of SDH microwave radio system.

Table 2. Features of SDH microwave radio equipment

| Item | 4, 5, 6 GHz/300 Mb/s | 4, 5, 6 GHz/150 Mb/s | 11 GHz/150 Mb/s | 11 GHz/50 Mb/s |
|---|---|---|---|---|
| Frequency bands | 4, 5, 6 GHz bands | | 11,15 GHz bands | 11 GHz bands |
| Transmission capacity | 312 Mb/s | 156 Mb/s | 156 Mb/s | 52 Mb/s |
| IF frequency | 110/130/150 MHz | 110/130/150 MHz | 140 MHz | 70 MHz |
| Modulation | 256QAM (3 multi-carrieres) | 16QAM (3 multi-carrieres) | 8PSK | 4PSK |
| Transmitting power | 27 dBm/carrier | 19 dBm/carrier | 33/37 dBm | 28 dBm |
| Space diversity | Phase-detecting in-phase combiner | | – | – |
| Fading equalizer | 7 taps, digital | 7 taps, digital | – | – |
| Error correction | Double forward error correction by BCH code | | | |

QAM: Quadrature Amplitude Modulation PSK: Phase Shift Keying BCH code: Bose-Chaudhuri-Hocquenghem code

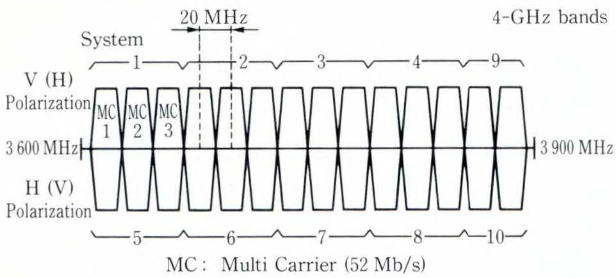



Fig. 11—Radio channel frequency allocation.

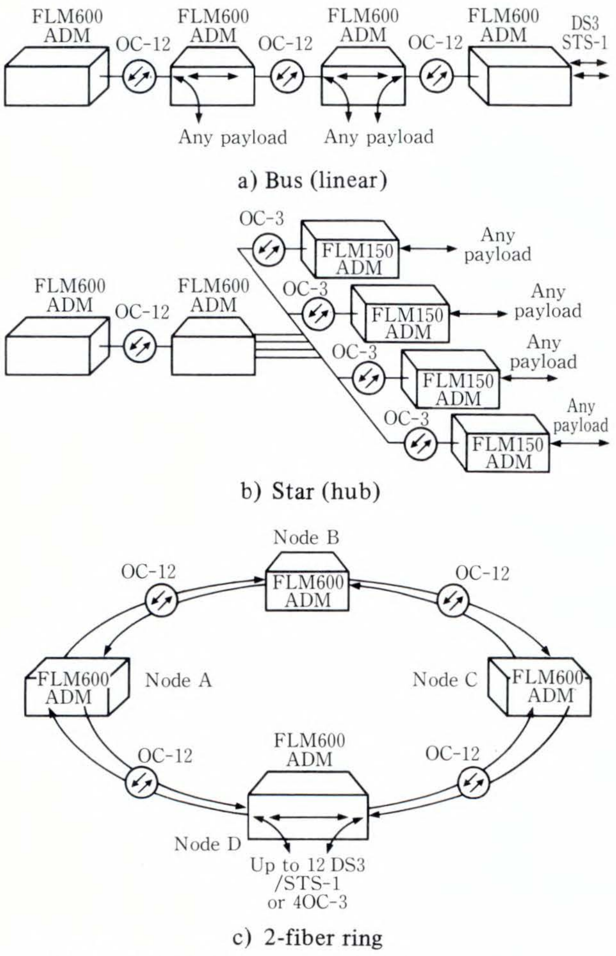



Fig. 12— Network topology for subscriber network.

bands. A radio transmission path contains up to 27 operating systems and one standby system. To protect against fading, a phenomenon specific to radio links, and to improve line performance, the SDH microwave radio system;

1) splits each system into three carriers, each forming a 52-Mb/s link (see Fig. 11),
2) enables hitless switching with the three standby carries for any faulty carrier of any operating system via matrix switches,

Table 3. Features of the FLM series

| System | Function |
|---|---|
| FLM 6 | Optically extends DS1s directly from FLM 150 ADM<br>4 x DS1 ⇄ optical 6.9 Mb/s |
| FLM 150 ADM | DS1, DS3 ⇄ OC-3<br>Point-to-point, hubbing (OC-3)<br>ADM (bus/ring) |
| FLM 600 ADM | DS3, STS-1, OC-3/OC-3c ⇄ OC-12<br>Point-to-point, hubbing (OC-12)<br>ADM (bus/ring) |
| FLM 2400 ADM | DS3, STS-1, OC-3/OC-3c, OC-12/OC-12c ⇄ OC-48<br>Point-to-point, regenerator<br>ADM (bus/ring) |

DS*n* : Digital Signal level *n*
STS-*n* : Synchronous Transport Signal level-*n*
OC-*n* : Optical Carrier level-*n*

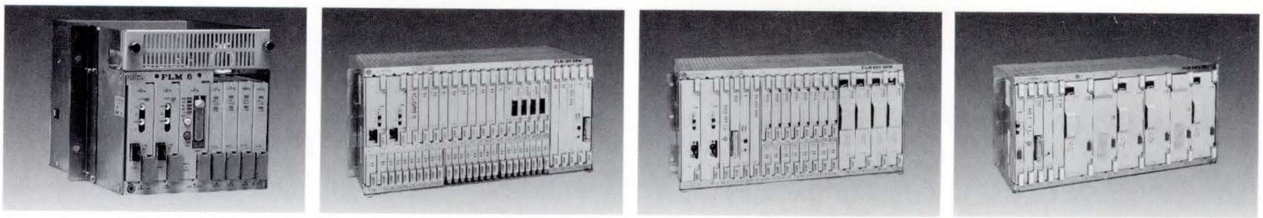
3) equalizes waveforms distorted by fading with transversal equalizers, and provides space diversity using two or three antennas, and
4) provides double error correction circuits based on BCH coding.

The system also monitors the received power, error rates, and other characteristics in radio sections, and is capable of isolating faulty sections.

### 5.3 SONET system

Networks operating in North America are characterized by a broad subscriber service area covered by a single switching office, and the separation of the network into BOCs. Geographical gaps are filled by Inter-exchange Carriers (IXCs). These make it necessary for the SONET system to be able to form networks efficiently in subscriber areas and to provide system compatibility among carriers.

For subscriber areas in North America, Fujitsu has developed the transmission system featuring optical transmission, SDH multiplexing, and ADM facilities, standardized to line speeds from 6.9 Mb/s to 2.4 Gb/s, to meet the needs of the diversified network topology (see Fig. 12). Table 3 summarizes the equipment, generally called the Fiber Loop Multiplexer (FLM) series that make up the system, shown in Fig. 13.



a) FLM6 b) FLM150 ADM c) FLM600 ADM d) FLM2400

Fig. 13–FLM series.

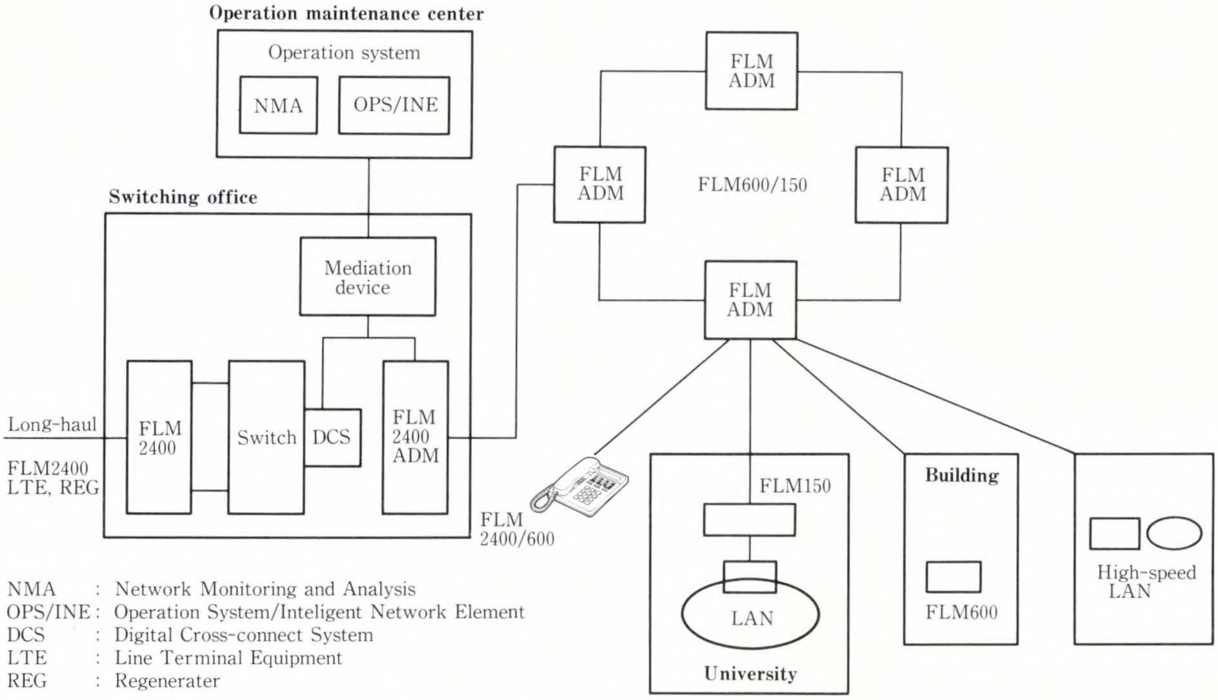
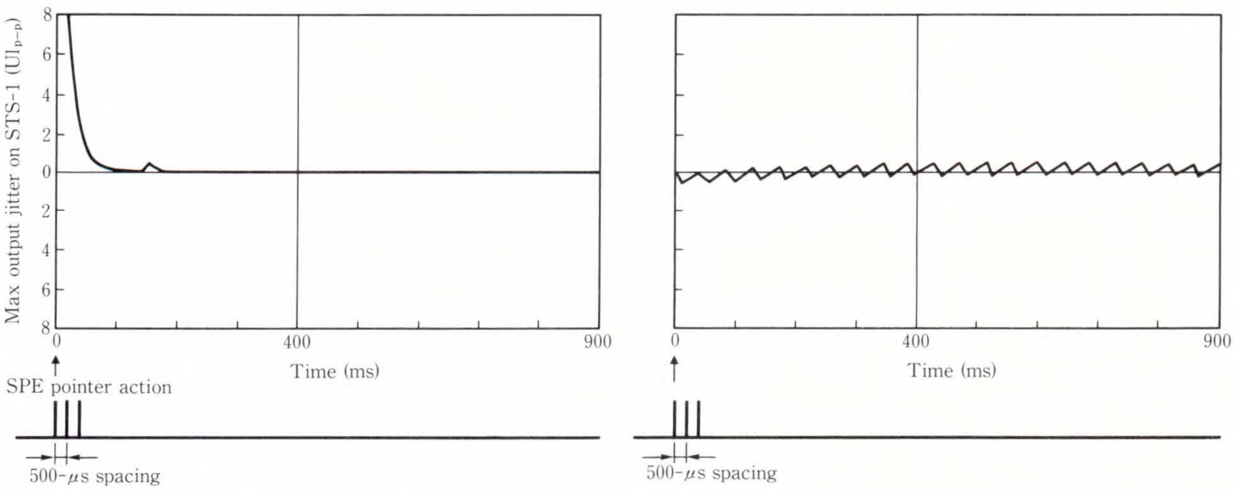



Fig. 14–System application for SONET.




a) Without jitter suppression function at tributary b) With jitter suppression function at tributary

STS-1 : Synchronous Transport Signal level-1 (52 Mb/s) SPE : Synchronous Payload Envelope

Fig. 15–Jitter suppression using adaptive bit leaking scheme.
Three consecutive same-direction SPE pointer action with the minimum spacing (3 frames).

Figure 14 shows a typical SONET configuration. Subscriber service information collected by a subscriber loop system is transmitted to a switching office and routed to switched and leased service circuits by the Digital Cross-connect System (DCS) installed in the subscriber loop system and in the switching office. The operation system manages all DCS operation. Bell Communications Research Inc. (Bellcore) is developing the Network Monitoring and Analysis (NMA) system and Operation System/ Intelligent Network Elements (OPS/INE) system for network routing management[8].

Differences in synchronization clocks of different carriers are accommodated by justification (pointer action). Due to the nonperiodic justification in the SDH by byte (8-bit) units, as opposed to cyclic justification by bits in conventional justification multiplexing, jitter may not be fully suppressed by brief batch justification. To remove this concern, "bit leak" is introduced to manipulate each 8-bit sequence with a wider bit spacing, rather than manipulating all eight bits at once. Figure 15 shows jitter suppression using a bit leak with the previous scheme.

## 6. Conclusion

New synchronous digital network systems have been developed to make good use of the SDH characteristics for addressing evolving networking needs.

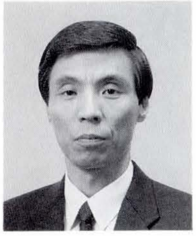
Since 1989 these systems have been used widely in Japan. In North America, the BOCs are actively implementing the SONET system. The standardized basic specifications of the system are expected to make their application possible all over the world.

The new synchronous interface basically operates at 156 Mb/s, the standardized user channel speed for future Broadband ISDN (B-ISDN). The scope of the new interface is expected to be extended to cover offices and general households to implement services such as home shopping and home medical care. These will be supported by high-quality visual communication such as High-Definition Television (HDTV) and supercomputers. The new interface is thus expected to help build the B-ISDN infrastructure.

Standardization organizations including the CCITT are studying possible enhancements to achieve a higher level of functionality and enhanced operation and maintenance to help prepare for the time when B-ISDN arrives. Fujitsu intends to promote global contributions to develop the next generation network responding to society's needs.

## References


1) CCITT: Recommendations G. 707, G. 708, and G. 709. CCITT Blue Book, 1989.
2) Bellcore: Synchronous Optical Network (SONET) Transport Systems: Common Generic Criteria. Bell Commun. Res., Morristown, N.J., TA-NWT-000253, Issue 6, September 1990.
3) Ueda, H., Tsuji, H., and Tsuboi, T.: New Synchronous Digital Transmission System with Network Node Interface. Proc. GLOBECOM' 89, 42. 4, **Vol. 3**, pp. 1506-1510.
4) Ichikawa, H., Sango, J., Tanaka, K., and Sindo, Y.: Digital Radio System Design with the Network Node Interface. Proc. ICC, 42. 4, 1989, pp. 1297-1302.
5) Special Isssue: New Synchronous Digital Network System. (in Japanese), *FUJITSU*, **42**, 4, pp. 291-380 (1991).
6) Shinbashi, M., Ishikawa, A., and Sato, K.: SONET System for North America. *FUJITSU Sci. Tech. J.*, **27**, 1, pp. 92-106 (1991).
7) TTC: JT-G 707, JT-G 708, and JT-G 709. Network Node Interface III. 1 (in Japanese), TTC STANDARDS, 1990.
8) ANSI: "Operations, Administrations, Maintenance, and Provisioning (OAM&P) Lower-layer Protocol for Interfaces between Operations Systems and Network Elements." ANSI, T1. 204-1990.
9) CCITT: "Revised draft Recommendation G. 709-Synchronous multiplexing structure." STUDY GROUP XVIII-REPORT R33, 1990.
10) CCITT: "Principles for a Telecommunication Management Network." Recommendation M. 30, 1989.
11) CCITT: "Annex 1: Text of draft Recommendation G. snal." Annexes to the report of WP XVIII/7, Q4/XVIII, 1990.



**Takeshi Sakai**
Transmission Architectures Dept.
FUJITSU LIMITED
Bachelor of Electronic Eng.
Tokyo Institute of Technology 1969
Specializing in Digital Transmission Systems

**Akira Tokimasa**
Transmission Div.
FUJITSU LIMITED
Bachelor of Communication Eng.
Kyushu University 1966
Specializing in Digital Transmission Systems

**Yoshikuni Toko**
Radio & Satellite Communications Div.
FUJITSU LIMITED
Bachelor of Electronic Eng.
Yamanashi University 1968
Specializing in Radio Systems

Featuring Paper

UDC 621.397.2

# Multimedia Communication Technology

● Ryusaku Imai ● Teruo Tobe ● Tsuneo Katsuyama

*(Manuscript received January 6, 1992)*

One of the major developments in the communications and computer industry is the multimedia information environment. Integration of these multimedia technologies with daily activities has become a critical issue. In particular, office communications is one of the most important areas for application of multimedia services. The fundamental aims are natural conversation through telecommunications, information-sharing between geographically separated users, and a user-friendly interface to facilitate use of the new services. This paper introduces and discusses three major research activities on teleconferencing and multimedia user interfaces.

## 1. Introduction

Recent progress in multimedia technology is making multimedia communications a reality. As the basis for the communications network, ISDN technology has already been introduced worldwide; transmission speeds of 64 kb/s and 1.5 Mb/s make it possible to transmit not only text and data, but also visual information which appears natural. Furthermore, Broadband ISDN (B-ISDN) is currently being developed, which at a speed of 156 Mb/s, is about one thousand times faster than ISDN. This will make it possible to transfer high-resolution graphics almost instantaneously, and to transmit multiple videos without loss of quality. New image compression techniques and input-output devices have further contributed to the progress of multimedia communications.

This paper discusses how to make the best use of such multimedia technologies in the workplace. One of the most important applications is the communication of information in the office. With the rising popularity of decentralization, or 'satellite' offices, the communication of information between offices is becoming more important. Although telephone and facsimile machines currently play a very important role, they are not always sufficient for all office environments or circumstances. For example, it is very difficult to explain the shape of a complex object or to describe a map over the telephone. Multimedia communications services will help to alleviate such problems. Video teleconferencing has already made a significant contribution to effective, efficient communications between geographically separated offices.

There are several key issues common to multimedia communications. Firstly, it is essential that conversation and live images be natural for effective conferences. The users should not be aware of technological limitations. Secondly, information must be shared between the users. If each user can point to and mark the same document, for example, then a common base for good communications can be established. Finally, it is equally important to have a friendly user interface. The technology must be easy to use.

This paper describes some of the current technological issues behind multimedia communications, and the approaches being taken to resolve them: natural teleconferencing services for realistic live conferences (Chapter 2), image data transfer between users using ISDN for desktop conferencing (Chapter 3), and services and user interfaces in future B-ISDN multimedia environments (Chapter 4).

## 2. Teleconferencing

### 2.1 Video teleconferencing systems

A large part of office time is taken up in meetings and conferences. If these meetings are to be implemented electronically, then a more natural communications system which simulates face-to-face communications is necessary. This is the goal of teleconferencing systems.

The use of teleconferencing systems has been rising to counteract the spiralling costs of travel and time out of the office. However, video conferencing offers a better human interface, and is thus making practical conferencing systems a reality–this has been made possible by the development of an international standard codec and installation of ISDN line services. The following section outlines Fujitsu's VS-700 video teleconferencing system which complies with the recommendations, and is scheduled for widespread use.

Picture quality and usability are key issues. For many years, manufacturers have concentrated their R&D efforts on the efficiency of image coding technology. As a result, high quality images can be produced even with a high compression ratio. However, since the prime object has been the video codec, usability and convenience have tended to be ignored. This is clearly not satisfactory, since even beginners should be able to use the equipment to carry on a normal conversation as if in a real meeting, with a minimum of training.

Based on extensive in-house use, Fujitsu has been placing great emphasis on the man-machine interface while working on video conferencing systems. The goals for Fujitsu's development work are therefore:

1) Transmission of virtual reality

Due to its physical restrictions, telephone conferencing is not suited to conveying the fine nuances of real communication. Our object is therefore to transmit both voice and images in a system that is as close to natural face-to-face communications as possible.

i) Voice transmission

Voice transmission must enable the voice of the speaker to be clearly distinguishable from the voices of other participants, regardless of the room acoustics.

ii) Image transmission

Participants in the conference should be able to appreciate the facial expression of the speaker and to feel the atmosphere of the conference. To simulate real face-to-face communication, the entire picture, and a close-up of the speaker must be displayed simultaneously rather than in turn. The expression of the speaker must be clearly visible.

2) Usability

Conference participants may have little or no knowledge of how to use video conferencing systems. Therefore, the basic functions of the conferencing system should be usable without specialist knowledge. However, those who already have some experience should be able to fine-tune the system easily.

3) Expanding an installed system

A system that does not require specific lighting conditions can be implemented simply by choosing the appropriate equipment. This sort of system can easily be installed even in a normal office, simply by connecting the various pieces of equipment. The system can also be moved easily.

### 2.2 The Fujitsu video teleconferencing system

This conference system consists of a

Fig. 1–FACOM video terminal.

Table 1. Specifications of teleconferencing system: FACOM 2285 video terminal

| Items | | Specifications |
|---|---|---|
| Image | Coding system | Conforms to CCITT H.261 TTC JT-H261. |
| | Signal | NTSC color composite signal |
| | Multiplex mode | Two-channel multi-rate transmission and composite display |
| Voice | Coding system | Conforms to CCITT G.772 TTC JT-G722. |
| | Voice band | 50Hz-70kHz |
| | Processing system | Automatic gain or microphone control |
| Control | Camera control | Various types of control such as automatic photographing of speaker and remote camera presetting |
| | Console | Infrared wireless keypad |
| Mainframe | Multiplex mode | Conforms to CCITT H.221 TTC JT-H221 |
| | Line speed | 56 to 1 920 kb/s |
| | External dimensions | 1 680($l$) x 850($b$) x 700($h$) (mm) |

FACOM 2285 video terminal and various subsystems. Table 1 lists the specifications, and Fig. 1 shows the FACOM 2285 video terminal. The functions and characteristics of the system are described below.

1) Voice

An automatic voice gain control system was developed to counteract feedback howl. The voice frequency bandwidth is 7 kHz, which is twice that of a telephone, and the coding system conforms to CCITT G.722. The voice quality is natural and well articulated.

2) Two-channel multi-rate transmission and composite image display

This system is equipped with two cameras: a wide-angle camera for transmitting pictures of the conference room, and a close-up camera for capturing the face of the speaker. Switching between cameras is not necessary since a composite view of both images is displayed at the receiver's side (see Fig 2). Multi-rate transmission, in which the higher transmission rate is assigned to the close-up image, realistically conveys the expression of the speaker and the atmosphere of the conference room.

Fig. 2–Example of a composite screen.

3) Operation

i) Voice activation

Microphone gate control recognizes the speaker and automatically triggers a camera close-up on the speaker. This allows the remote participants to see immediately the expression of the speaker, and facilitates a smooth and natural conference.

ii) Camera control of remote conference room

This function allows participants in the remote conference room to manually set the camera position, for example, to see a close-up of a participant who is not currently speaking, or attendees. This camera is controlled by a dedicated wireless keypad.

iii) Document transmission control

Many documents and materials are often used in a conference. This system can display materials on the remote screen along with a close-up of the person explaining them, which increases the efficiency of the conference. The close-up image of the speaker can also be moved around on the monitor screen.

### 2.3 Future expansion

The usefulness of video teleconferencing has been recognized for many years. However, systems developed so far have both advantages and drawbacks. Since the functions to be provided by a system depend on the purpose of the conference and the needs of the participants, it is difficult to select and balance the necessary functions. This is partly due to the fact that neither manufacturers nor users have had sufficient experience using these systems yet. To ensure the growth of teleconferencing, it is vital that multimedia systems be made as accessible as possible, perhaps by linking them to computer networks.

## 3. Visual desktop conferencing

### 3.1 Requirements of visual desktop conferencing

Decentralization of businesses is likely to increase in the near future as satellite offices become more common. Thus, the need for preliminary meetings between remote sites and 'formal' teleconferencing as previously described will increase. The spread of ISDN and improvements in color image processing technology have now created the environment for visual desktop conferencing. However, it is not sufficient to simply transfer images to achieve a natural conversation; all parties must simultaneously share the same materials–this is known as WYSIWIS, or 'What-You-See-Is-What-I-See'. This gives the participants the impression that they share the same information environment. This section describes the Visual Network Station (VNS), which was developed by Fujitsu for desktop applications. The system has the following requirements.

1) Image communication

During preliminary office meetings, verbal explanations usually have to be backed up with real documents and models which are to be discussed during the teleconference. Movable, easy-to-use cameras are thus needed to photograph documents and other items. Furthermore, to produce natural images of better quality than normal television, high-speed transmission is required, which must follow international standard codec.

2) Presentation

In most offices, G3 facsimile machines are becoming commonplace. Preliminary meetings are usually conducted over the telephone based on documents sent by fax. However, conventional telephone and facsimile technology does not allow discussion points to be raised easily, and items cannot be 'pointed out' directly by marking them. It is important in a visual desktop conference that complete presentation is possible, whereby things can be pointed out and marked. Facial images must also be transmitted to complete the sense of reality.

3) Communication using only one B-channel

To keep line costs down and to allow existing terminals and facsimile machines to be used in a visual desktop conference, it is important that communication is possible using only one B-channel.

4) Compactness

Although the cost must be kept low, it is also important that the equipment can still fit on a regular desk, preferably occupying no more space than a telephone.

### 3.2 Visual network station technology

1) Progressive display for a fast response

The image section of the equipment uses the Joint Photographic Coding Experts Group (JPEG) algorithm hardware codec that is the international standard coding system for still color pictures. In the JPEG algorithm, two modes of operation are defined, baseline sequential coding and progressive build-up coding. Figure 3 shows a block diagram of baseline coding. In the encoder, $8 \times 8$ input samples are transformed using the Forward Discrete Cosine Transform (FDCT) into an $8 \times 8$ array of quantized values, and the quantized output is fed to a procedure which converts the coefficient value to a set of symbols. In the spectral selection mode, the DCT is segmented into 'frequency' bands and each band is sent as a separate scan. The codec implements both JPEG baseline sequential coding and progressive build-up coding by spectrum selection.

We use progressive coding as the standard since it takes about ten seconds to transform a $640 \times 480$ pixel image through a 64 kb/s communication line. The low frequency is divided

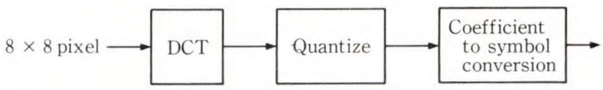



Fig. 3–Block diagram of baseline sequential coding.

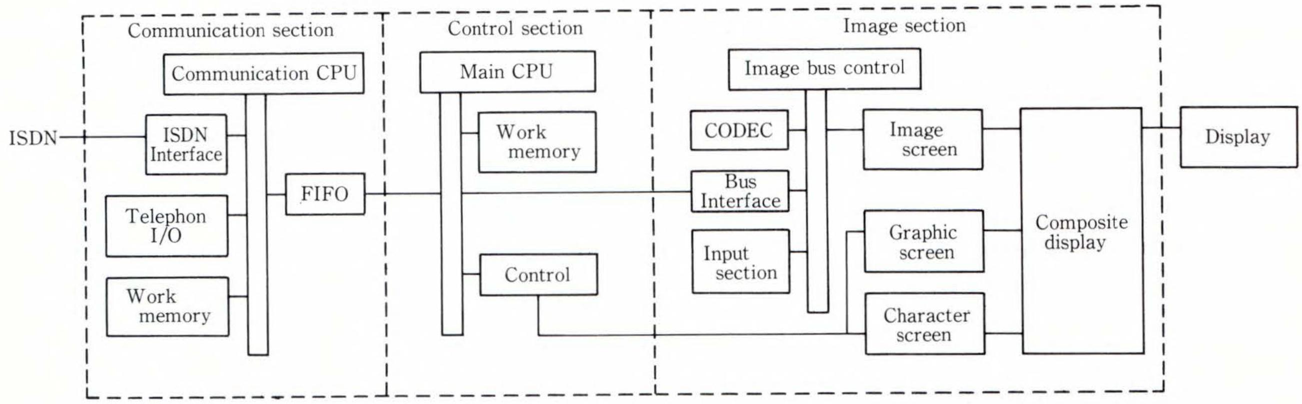



Fig. 4—VNS block diagram.

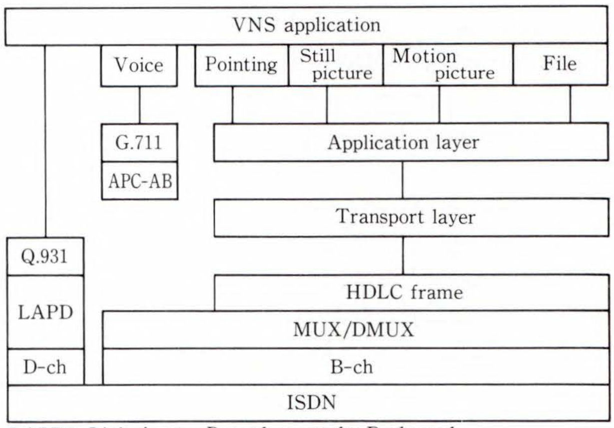
into three stages (DC, first AC coefficient, second AC coefficient) to give a fast response. However, the high frequency (from third to 14th AC coefficient) is not divided, because a change of display cannot be noticed. An outline display takes three seconds through a 64 kb/s line, and the complete display takes about ten seconds. Also we display a small motion picture by continuously processing a 160 x 100 pixel screen using the sequential function. A motion picture of about three screens per second can be achieved over a 64 kb/s line. Simple video communication can be attained without expensive motion video codec.

2) Compact desktop equipment

The above functions can be implemented in a desktop-size unit by using four A5-size printed circuit boards. Figure 4 shows a block diagram of the VNS. The image section contains two boards, and there is one board in each of the communication and control sections.

i) Image section

The image memory is used for both the display frame memory and the coding memory, and it uses an efficient Y, Cr, Cb format for compactness. There is also an image bus, and the image memory and all I/O sections are connected to this bus. Access to the image memory from the I/O sections has been simplified by using a dedicated image bus control section. In addition to the image memory, there is also a graphic screen for the pointing and marking functions, and a character screen for displaying electronic directories. This configuration enables rapid pointing and marking.




LAPD: Link Access Procedure on the D-channel

Fig. 5—Communication protocol configuration.

ii) Communication section

Figure 5 shows the entire configuration of the communications protocol which supports the desktop conferencing system. To enable communication through one B-channel, in layer 2 the B-channel is divided into a 16 kb/s voice channel and a 48 kb/s data channel. The Adaptive Predictive Coding-Adaptive Bit allocation (APC-AB) system is used for voice coding, because it gives excellent voice quality and good coding results.

Processing of the 48 kb/s data channel is simplified as follows. Framing is processed in the HDLC procedures, error and flow control are processed in the transport layer, and other control is processed in the application layer. It is thus possible to achieve simultaneous communication of images and voice through only one B-channel. The communication section is also used for pointing and marking, which helps to give a concise communications protocol, rapid re-

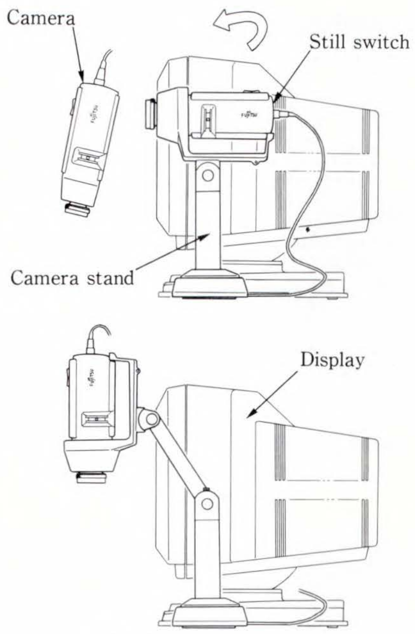



Fig. 6–Camera section.

sponse and compact hardware.

3) Freedom of movement of camera

The arm mechanism of the camera can be moved smoothly when photographing people or documents. The position of the camera can be adjusted by tilting it forward so that one A4-size document fills the screen. The camera can also easily be detached to photograph other documents or objects, as shown in Fig. 6, and the camera has a pause switch to allow images to be taken manually.

## 4. Multimedia information environments and the human interface

### 4.1 Integrated information environments

In multimedia communication systems such as teleconferencing and desktop conferencing, the human interface is critical. In order to get the general acceptance, the human interface should be consistent with the current environments. From this view point, this section describes a new approach to the human interface for multimedia information services.

Visual information, which is the significant feature of a multimedia environment, is currently often expressed on paper, and most information handled by humans is still communicated by paper. Despite the rising popularity of voice mail and computer networks which can operate even while one is out, most information is nevertheless produced, stored, copied and transferred on paper. In meetings and conferences, large volumes of paper documents are required. The office workers of today are still swamped in paperwork.

The medium of paper is an intuitive and direct medium. Browsing allows fast retrieval of information, and page and chapter numbers give the reader the structure of the document. Space is also part of the structure, and should itself be considered as information.

Since paper is a physical medium, it inevitably has its defects. Information can really only be presented as text, figures, or photographs. The process of physically binding pages together, necessary for management, imposes a centralized structure on the information. Retrieval and processing of the information is time-consuming, and much storage space is required. Furthermore, the use of paper is seriously affecting world natural resources and the environment.

Mankind has been using paper for about 5 000 years. In around 3100 B.C., the ancient Egyptians started recording important events on papyrus paper made from reeds instead of the traditional rock, wood and clay tablets. Paper was later reinvented in Han China in around 105 A.D. by Ts'ai Lun. Paper is one of mankind's most important cultural assets, and it holds most of the accumulated knowledge of generations of the human race.

To allow the multimedia environment to become the cornerstone of information in the future, it must have the benefits of paper, without the associated problems.

### 4.2 Characteristics and issues of the multimedia environment

Before considering the human interface of a multimedia system, we should summarize the characteristics of multimedia information, especially visual information. Multimedia, especially images, has characteristics that voice and text lack. Images are more real representations which therefore have greater impact. However, they can be interpreted in different ways depending on the viewer and the circumstances. One good analogy is a typical mental test, in which images can be interpreted in many ways. Even identical images can have different meanings depending on the context. The image should therefore be presented in its original form without titling or other secondary information management.

The medium of images is also not limited to a one-dimensional arrangement that is true of strings of characters. An image only takes on a meaning when viewed as a whole; the normal method of viewing text by scrolling is not suitable for viewing images.

To allow broadband multimedia information systems to become part of our daily life, the services offered must make the best use of all the above characteristics. Furthermore, a good human interface that makes it possible to use these services easily and efficiently must be established.

### 4.3 Human interface for the multimedia environment

Since most information that is handled by people is either written or printed on paper, the advantages of paper are well known, while the disadvantages have been largely ignored. To create a system which makes the most of the characteristics of multimedia information, the following guidelines have been established.

1) A better interface than paper

When multimedia information includes moving images, the use of post-it notes, name cards, and bookmarks that can be used with paper should be maintained.

2) A frame for effective management of space

Paper is wellsuited to the human ability to manage space; information can be easily acquired and consolidated by physically moving it or creating interrelationships. The optimum frame is around the space in which the reader's eyes move naturally. To make life easier for the reader, a special frame must be provided.

3) Real, intuitive representation and operation

The information contained in images represents an actual object in an intuitive manner.

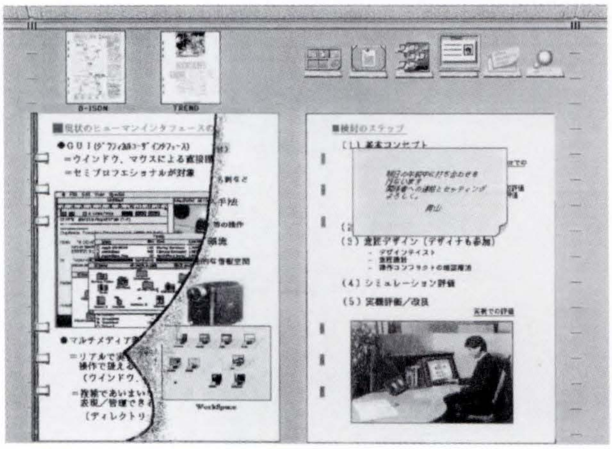


Fig. 7—Example of a desktop scene.

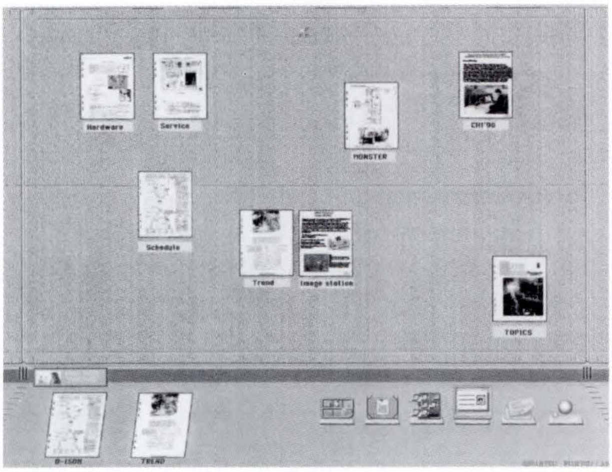


Fig. 8—Example of a room scene.

These images should therefore be used in a similarly intuitive manner by touching them.

4) Elimination of the disadvantages of paper

The power of expression that is available in multimedia information is a significant improvement over paper. Hypermedia technology may be useful for managing many documents that are related in a complex way, and for providing flexible access to documents. The way that the user interacts with the system should provide a new approach to understanding the contents of the documents.

Figure 7 shows an example of a human interface that implements all of these guidelines. There is a 'binder' that binds more than one sheet of paper to the desk. If the page is opened, a video or animated sequence starts playing. If the user touches the video the sequence stops.

The user controls all document operations in the same direct manner. For example, if the user strokes or 'flips' the page from right to left, the page turns over. If the screen is stroked more quickly, more than one page is flipped at a time. Stroking in the opposite direction goes back pages. If the upper right of the page is tapped twice, the screen shrinks and displays at the top of the screen. This miniature image can still be recognized as the original document, however. If a keyword on a page is touched, then the hypermedia system retrieves reference information about that item. This information appears as a page which slowly slides out from under the current page onto the right side of the screen.

Figure 8 shows a partition at the back of the desk. This simulates actual desks which often have a partition at the back and one to the side. This partition has more than one surface, and these surfaces can be seen by turning the partition to the right or left.

The miniature document described above can be put in a particular place on the partition, and when the miniature is touched, it moves onto the desktop. The document can then be read, as shown in Fig. 7.

To implement this human interface, many broadband communication interface functions, such as those available in B-ISDN, are required. It is also necessary to be able to control the presentation of broadband media such as video in realtime. A prototype system has been developed in which a multimedia accelerator is added to a general purpose workstation. The accelerator offers a way to control the presentation of broadband media in a flexible, realtime way. The prototype human interface is being evaluated experimentally for its recognition and perception characteristics (see frontispiece: Monster).

## 5. Conclusion

This paper has introduced multimedia communication services and human interface technology behind the terminals used for such services and interfaces, based on both current and future office communication networks.

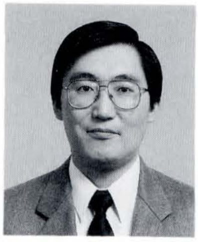
For cost-effective, user-friendly communications, office systems in large, complex, widely distributed working environments are likely to provide practical solutions. This will therefore require the integration of various user documents, which are held on different media and have complex relationships. Multimedia technology provides one key way to integrate these information needs. It also leads naturally to practical applications, since it offers a realistic simulation of the way we handle information in the real world.

Multimedia services actually assist our management of information. Future research and development must work from this base to establish more user-friendly communications technology, which will then become the multimedia basis of much of human activity.

## References

1) Nishimura, N. et al.: An Architecture of High-speed Picture-Codec. (in Japanese), Proc. 1992 IEICE Spring Conf., D-374, pp. 7.118.
2) Umezaki, Y. et al.: An Architecture of Simple-Motion-Picture by Still-Picture-Codec. (in Japanese), Proc. 1992 IEICE Spring Conf., D-373, pp. 7.115.
3) H. Kamata, et al.: Communication Workstation for B-ISDN: MONSTER (Multimedia Oriented Super Terminal). Proc. GLOBECOM '89, pp. 959-964.

**Ryusaku Imai**
Video Communication Dept.
FUJITSU LIMITED
Bachelor of Electronics Eng.
Waseda University 1974
Specializing in Developing Teleconference System



**Tsuneo Katsuyama**
Distributed Systems Laboratory
FUJITSU LABORATORIES, KAWASAKI
Bachelor of Instrumentation Eng.
Keio University 1974
Master of Instrumentation Eng.
Keio University 1976
Specializing in Communications and Personal Information Services

**Teruo Tobe**
Network Software Dept.
FUJITSU LIMITED
Bachelor of Electronics Eng.
Saitama University 1972
Specializing in Multimedia Communication Terminals

UDC 334.722:621.395.74

# Corporate Information Network System: COINS

● Tatsuki Hayashi ● Kozo Nakamura ● Toshitaka Tsuda

*(Manuscript received December 16, 1991)*

Fujitsu has developed COINS, a system of products that integrates operation and building techniques for use in corporate information network systems. Demand for private ISDN has continued to grow since its release in 1984. As a result, the overall demand for COINS products has also increased.

This paper describes the philosophy and methods used in developing COINS, with particular focus on features for private ISDN, and also overviews the private B-ISDN of the future.

## 1. Introduction

Changes in society and the economy have brought about a demand for effective corporate information networks. In May 1984, Fujitsu took the lead by announcing COINS (Corporate Information Network System). In August 1985, we launched COINS II, and in February 1987, COINS II V2. In April 1988, to take advantage of NTT's INS Net 64 service, we announced COINS III. Moreover, Fujitsu has continued to increase the range of COINS products. COINS users have been able to take full advantage of NTT's INS Net 1500 service, launched in June 1989, and the INS packet switching service that debuted in June 1990.

Demand for corporate information networks systems from the need for information exchange between bodies such as related firms and overseas branches, and of course within corporations themselves. There is also a growing demand for domestic information services. To answer these demands, COINS must be able to provide economical broadband network building, high-reliability, high-speed transmission and connection to public networks, and be able to integrate with yet-to-be-launched network services. ISDN enables the construction of broadband, integrated digital networks that offer high-speed, economical communication. By exploiting ISDN's capabilities to the full, COINS enables the construction of effective private ISDN networks.

This article focuses on how COINS products exploit ISDN's features, and looks at possible future developments. The probable effects of the introduction of B-ISDN are also covered[1].

## 2. COINS

### 2.1 Goals of COINS and COINS products

COINS enables the construction of an in-company information communications network system (private ISDN) using high-speed digital leased lines and public ISDN. This private ISDN can handle greatly increased quantities of data in a wide variety of formats. Also, by supporting connection to public ISDN and public networks, it enables the most effective construction of a system for communication with related firms and private residences[2]. Figure 1 shows how private ISDN is positioned within the system.

Private ISDN can be realized using COINS products, such as the FETEX-3000A series ISS (Information Switching System) digital switching system, the F2650 series DMIX-E multimedia multiplexer, and the MHLINK series LAN system. These COINS products are monitored and managed by the COMS-C network monitor system.

COINS products solve many of the problems

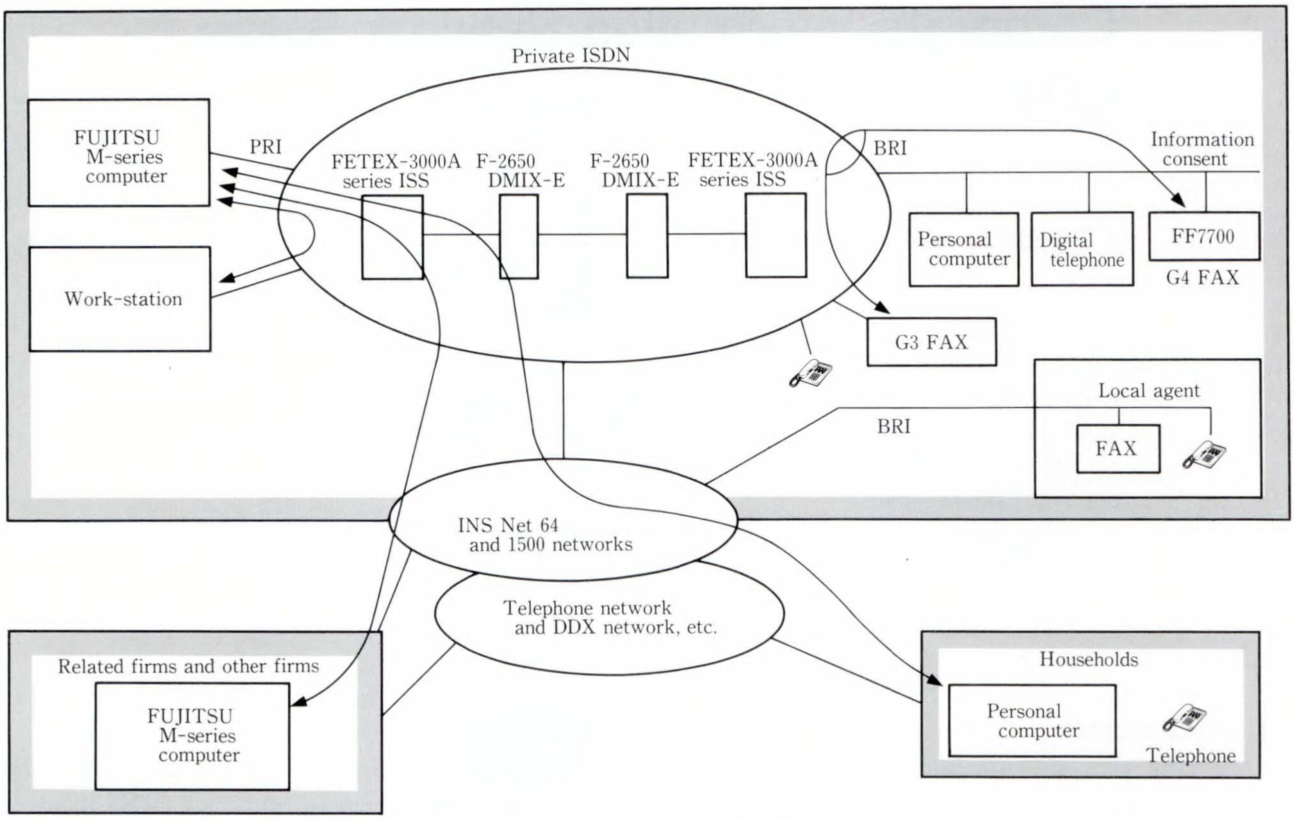



Fig. 1–Private ISDN positioning.

of conventional communications networks. They also offer the following advantages[3]:

1) Reduced costs

Conventional networks that handle voice, data, and image data separately can be replaced with a single integrated system using high-efficiency signal compression. This drastically reduces communications costs.

2) Increased reliability

The centralized management of private ISDN, together with overall COINS product monitoring, guarantees high reliability.

3) Flexible configuration

The network configuration can be tailored to the purpose of the communications system. For example, point-to-point for 1 : 1 communication, star (or multipoint) for communications from a center station, and drop/insertion for branch/bypass at an intermediate station.

4) Switched connection to public ISDN

If a high-speed digital leased line should fail, the system can automatically switch to a public ISDN line to maintain communications. A digital l-link network, offering low line cost, can be configured by connecting to the public ISDN only when necessary.

### 2.2 Basic policy driving private ISDN implementation

The basic points to consider when implementing a private ISDN system with COINS products are shown below.

1) Full compatibility

A corporate communications network must provide an ISDN environment for the services offered by public ISDN. Such services include high-speed data transfer and access to video databases. Each COINS product is fully compatible with ISDN.

2) Complete support

Networks need not be restricted to providing communication between a company's head office and branches. There is also a demand for the exchange of information with local offices, related firms and private residences, through the public network. COINS products can be connected to, and fully support the auxiliary functions of, public ISDN.

3) Minimum disruption

COINS products and functions make the best use of existing resources throughout the extended period needed for private ISDN conversion.

4) Standards conformity

COINS products conform to all relevant standards, such as those laid down by the CCITT.

**2.3 Development policy**

Our development goals for the COINS product group, as related to ISDN features, are listed below.

1) ISDN-type interface

To provide standard information exchange and enable easy access to public and private ISDN from computers and terminals, we set out to develop a user network interface similar to the public ISDN interface.

The digital FETEX-3000A series ISS (Information Switching System) incorporates a push-button telephone feature and the standard ISDN interface for trunk lines, as well as extension and station lines. The FETEX-3000A series ISS links with the FETEX-5000A series packet switching device to provide access to the INS packet service from its incorporated ISDN terminals, and also from existing terminals. The ISDN standard terminal interface can be installed in the F2883/2890 series optical LAN.

2) Connection between corporate and public ISDN

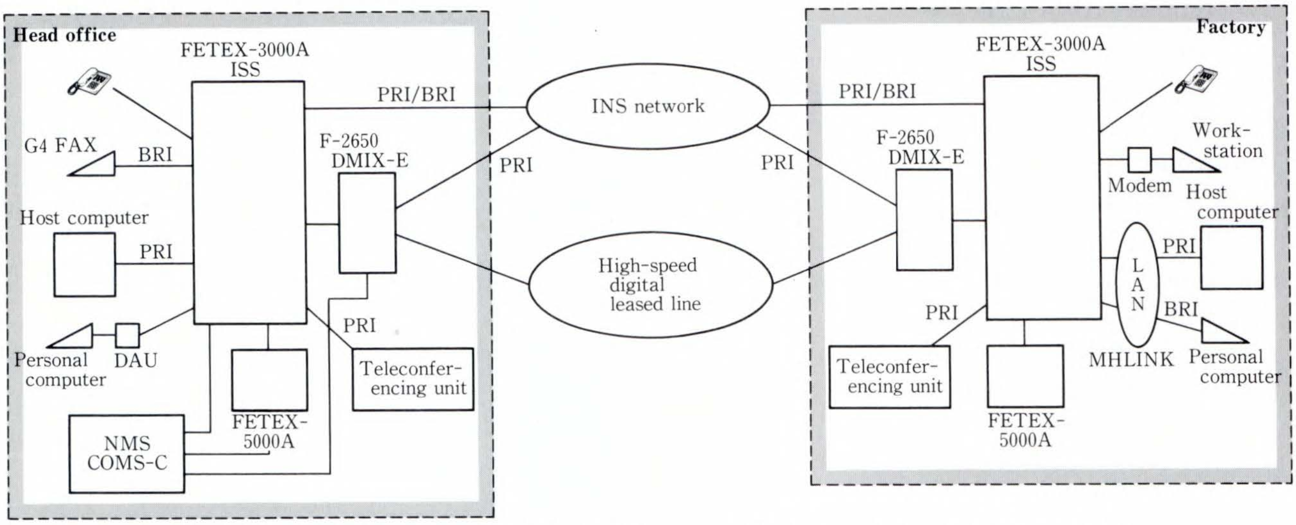
Compensation and connection features that combine both private ISDN and public ISDN are provided to ensure highly reliable and economical broadband digital networks. The FETEX-3000A series ISS and F2650 series DMIX-E can be connected to public leased lines for communication between public ISDN and the leased line. Both units also allow the user to connect telephones on a toll number via public ISDN. The DMIX-E has a rerouting feature to automatically switch to INS Net 64 or 1500 if the high-speed digital leased line fails.

3) Efficient installation

A feature enabling communication between ISDN products and existing networks (i.e. non-

Table 1. ISDN support of COINS equipment

<table>
<tr><th colspan="2">Equipment model</th><th>Outline of support</th></tr>
<tr><td rowspan="2">Digital switching system</td><td>FETEX-3000A</td><td>Interconnection between existing data terminal and ISDN terminal by data format conversion function<br>BRI and PRI interface on both extension line and station line sides<br>Inter-ISDN terminal communication via digital leased line between FETEX-3000A provided by ISDN interface support in trunk line<br>Data format conversion function, interconnection between existing data terminal and ISDN terminal by modem to ISDN digital signal mutual conversion function<br>B/D channel packet switching with FETEX-5000A linkage<br>High-speed channel switching in extension/station/trunk line<br>Toll dial number connection between US extension lines via public ISDN<br>Interconnection between data terminal included in the leased line and public ISDN</td></tr>
<tr><td>FETEX-5000A</td><td>Bypass function to public ISDN<br>Packet communication between terminals connected to FETEX-5000A end terminals and public ISDN<br>In-plant B/D channel packet switching with FETEX-3000A linkage<br>INS station line B/D channel packet switching with FETEX-3000A linkage</td></tr>
<tr><td>Multimedia multiplexer</td><td>F2650DMIX-E</td><td>BRI and PRI interface with public ISDN<br>Automatic backup<br>Digital network enlargement to branch system possible by forming specified time operation network using public ISDN</td></tr>
<tr><td>LAN</td><td>F2880/2890</td><td>BRI and PRI interface</td></tr>
</table>




DAU: Data Access Unit NMS: Network Management System DMIX: Multimedia Multiplexer

Fig. 2—Configuration of private ISDN.

ISDN) allows users to build a private ISDN while minimizing investment of both time and capital. Perhaps the best approach is to gradually add ISDN features to an existing communications system until, ultimately, the entire network is replaced. Given the length of time needed to implement a private ISDN, it is important to use existing systems to their greatest effect.

The FETEX-3000A series ISS provides a data format conversion function that makes it possible to connect an existing data terminal and an ISDN terminal. It also supports a digital signal conversion function between the modem and ISDN. The FETEX-5000A series, when linked with the FETEX-3000A series ISS, supports a Packet Assembly/Disassembly (PAD) function that allows an existing ISS terminal to use the INS NET packet service.

Table 1 shows ISDN support provided by the COINS product group, resulting from the above development policies. Figure 2 shows a private ISDN configuration consisting of COINS products. The most important roles in a private ISDN are played by the digital switching systems, usually a FETEX-3000A series that supports telephones and data terminals, and the multimedia multiplexer that transmits multimedia information through high-speed digital leased lines.

The following sections explain the ISDN features of digital switching systems and multimedia multiplexers, and how they might be further developed.

## 3. Digital switching system

### 3.1 Purpose of development

In response to the advent of the ISDN era, the FETEX-3000A series ISS was developed to access a wide range of services[4]. It offers the following features:

1) Ability to handle future ISDN traffic load

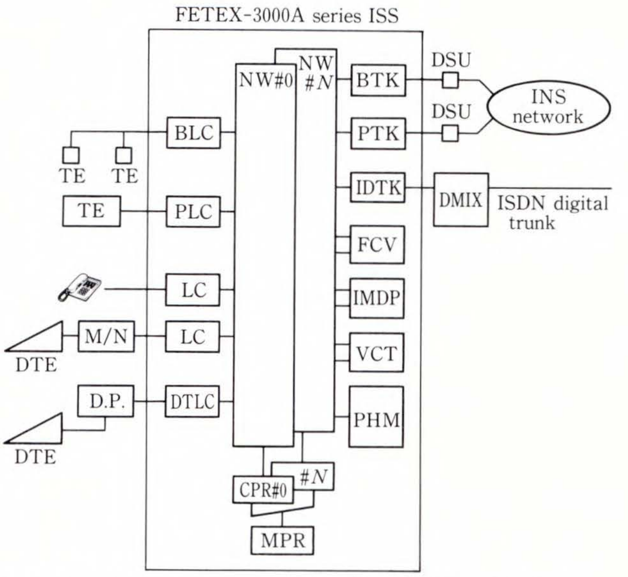
As ISDN is applied more and more to the public network, the widespread introduction of ISDN terminals into offices is expected. The ISS can handle the resulting increase in ISDN traffic.

2) Ability to cope with ISDN feature enhancements

Because of the addition of features to public ISDN, the appearance of various ISDN terminal types, and changes in the working environment within corporations, there is more and more diversity in applications using PBX. The ISS has a system configuration that can adjust to future enhancements in both the hardware and software.

3) Ability to link to existing systems

Over 3 500 FETEX-3000/3000A series PBXs have been delivered since the model was launched in April 1986. ISDN features need to be added to this large number of installed systems. This has been made possible by Field Upgrades




NW : Network
TE : ISDN Terminal Equipment
BLC : Line Card for BRI Extension Line
PLC : Line Card for PRI Extension Line
LC : Line Card
DTE : Data Terminal
M/N : Modem/NCU (Network Control Unit)
D.P. : FETEX-3000/3000A Series Digital Telephone
DTLC : Line Card for Digital Telephone
BTK : BRI Station Line Trunk
PTK : PRI Station Line Trunk
DSU : Line Terminator
IDTK : ISDN Digital Trunk
FCV : Data Format Converter Trunk
IMDP : ISDN Modem Pool Trunk
VCT : Voice Compression Trunk
PHM : Packet Handler Module
CPR : Call Processor
MPR : Management Processor

Fig. 3—System configuration of FETEX-3000A.

(FUG).

### 3.2 System configuration

Figure 3 shows the system configuration of the FETEX-3000A series ISS.

1) Full support of ISDN interface

Both the extension line and network sides offer a Basic Rate Interface (BRI) and a Primary Rate Interface (PRI). When linked with the FETEX-5000A series, the system can support an X. 31 terminal.

2) Networking functions

An ISDN D-channel signaling system, based on the TTC standard, connects ISSs. This system provides:

i) Access to private ISDN services such as caller number notification, subaddress forwarding, clamp-on, and call forwarding.
ii) Interoperable data transmission functions between public ISDN and the leased line.
iii) The option of adding a voice compression function which, with the FETEX-3000A series ISS media recognition function, allows voice and data traffic to be dynamically transmitted over the same channel.

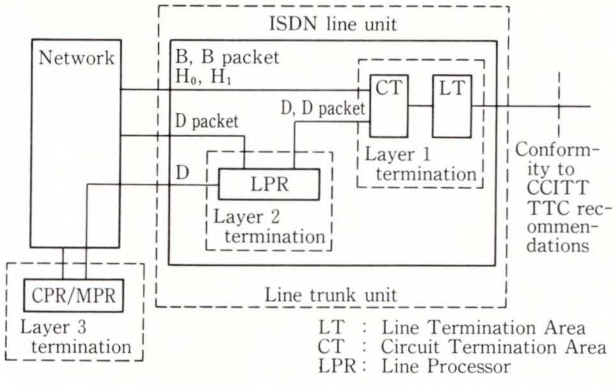



Fig. 4—ISDN Line/Trunk system configuration.

For example, 64 kb/s voice data is compressed to 32 kb/s prior to transmission. This maximizes the efficient use of a channel.

iv) A maximum exchange rate of 1.5 Mb/s.

3) Use of ISDN with existing interfaces

To provide interconnection between existing terminals, such as modems and ISDN terminals, a data format conversion function and modem signal conversion function are built into the ISS.

4) Automatic protocol selection

Some areas of the U.S. have a 56 kb/s standard, so naturally the protocol is different. The FETEX-3000A series ISS terminal automatically selects the correct protocol according to the data transmission speed of the sending terminal. Thus it is possible to communicate with networks that have different network speeds.

### 3.3 Hardware configuration

1) Voice channel

In high-speed channel switching of PRI $H_0$ (384 kb/s) and $H_1$ (1.536 Mb/s), frame time assurance is important. The voice channel has a buffer on each side.

2) ISDN line interface equipment

Figure 4 illustrates the ISDN line/trunk system configuration. The LT/CT area that terminates electrical/physical interface layer 1 contains five types of specially-developed LSI, including specially-developed master-slice LSIs with a maximum of 8 000 gates. The LPR area uses a communication LSI that enables simultaneous D-channel LAPD protocol processing for 16 lines. Network layer 3 terminates at the CPR/MPR area via the LPR area. The LPR separates the D-channel signal into D-channel packet information and call control signals. Packet information is sent to the FETEX-5000A through the voice channel. Several specially developed interface circuits provide maximum miniaturization and reliability for the line interface equipment. Printed wiring boards (PWB) with up to 8 layers cope with the multi-layered patterns resulting from the increased number of LSIs employed.

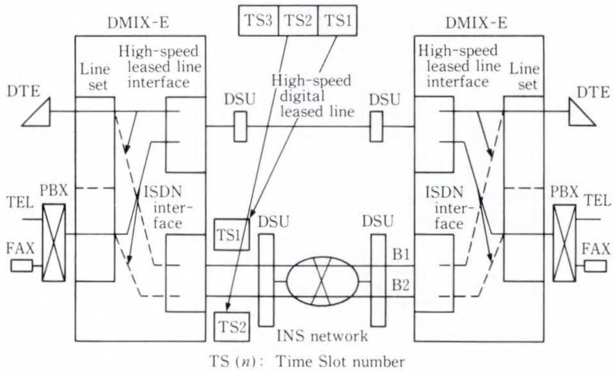


Fig. 5—Circuit rerouting for ISDN.

### 3.4 Software configuration

1) Support of several ISDN protocols

An ISDN user interface, extension-line network interface, and the ISDN leased line interface between ISSs are provided. These interfaces are realised by making the protocol into a data structure that is referenced during protocol processing. This makes it easy to add and modify functions, and to add or replace protocols in the future.

2) Practical use of existing software resources

A wide variety of exchange services have already been developed for the FETEX-3000A series. The ISDN protocol process transforms ISDN I/O information into existing I/O information, enabling these services to be used through most existing software modules.

3) Common software

Based on the one-machine concept of the FETEX-3000A series, all models, from small to large capacity, use the same software. This concept applies not only to existing switching parts, but also to ISDN control parts, thus improving software productivity.

## 4. Multimedia multiplexer

### 4.1 Purpose of development

The F2650 DMIX-E multimedia multiplexer can transmit voice, data, and video conference signals over several high-speed digital leased lines to provide a corporate INS.

NTT has already introduced its INS Net 64 and 1500 services. In response to this, the DMIX has an interface compatible with public ISDN, in addition to the interface necessary for high-speed digital leased lines. As a result, conventional 24-hour fixed-operation networks have gained the following features:

1) High reliability

If a high-speed digital leased line should fail, the DMIX automatically detects the failure and bypasses that line by switching to the public ISDN circuit. In this way, a high-reliability network is achieved at minimum cost.

2) Economy

A cheap digital 1-link network can be built by connecting a DMIX to an ISDN circuit only during chosen times (for example, daily between 8.00 a.m. and 5:00 p.m.).

It is also possible to economically digitize conventional networks based on analog leased lines, such as branch systems in large-scale networks, by applying DMIX as a gateway.

3) Interoperable linkage between COINS equipment

To make the best use of ISDN, COINS equipment provides advanced functions for linking to existing equipment, such as the FETEX-3000A series PBX.

### 4.2 System outline

#### 4.2.1 System configuration

Figure 5 shows the DMIX-E ISDN support configuration. ISDN functions are installed in the area where the conventional high-speed digital leased line is connected. This enables the system to be configured for multimedia multiplexing over an ISDN line.

#### 4.2.2 System features

The DMIX-E can be connected to both INS

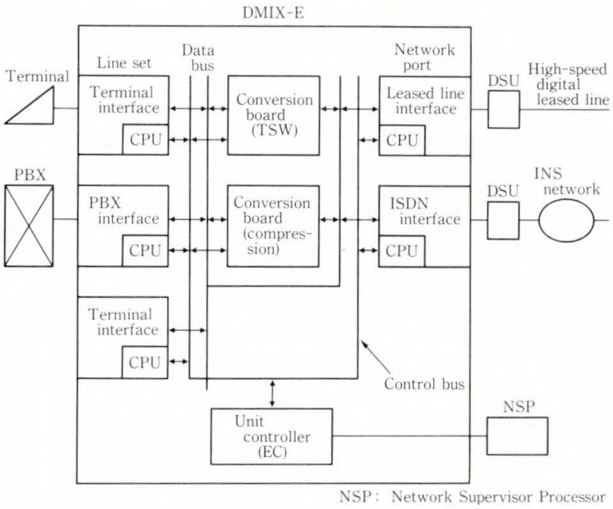


Fig. 6–Hardware configuration of DMIX-E.

Net 64 and 1 500. Features include:

1) Wide range of transmission rates

The system can support all the B (64 kb/s), $H_0$ (384 kb/s) and $H_1$ (1.536 Mb/s) rates used by ISDN. Therefore, the system can handle a wide range of transmission rates.

2) Preset network operation

ISDN branch lines with little traffic can be preset to use a particular band only during certain times. With metered rate systems, this enables an economical network to be configured.

3) Conformity to standards

The ISDN interface conforms to CCITT and TTC recommendations.

### 4.3 System unit configuration

Figure 6 shows the part of the DMIX-E configuration that enables ISDN support. A DMIX-E consists of a line set, network port, and a conversion board. Each unit connects to a duplex data bus and control bus. A local CPU, mounted in each unit, sets each unit according to commands received from the Equipment Controller (EC) that controls the entire DMIX. With the CPU connected to the Network Supervisor Processor (NSP) for DMIX, an ordinary console can monitor and manage the DMIX network. The function of each unit is briefly explained below.

1) Line set

The line set is the interface unit for any type of data terminal or PBX that supports multimedia. Several types are available.

2) Network port

The network port outputs multiplexed multimedia signals to a high-speed digital leased line. There are two types of line network port, one for a high-speed digital leased line, the other for an ISDN line. There are also two types of ISDN line network port, one for ISDN Net 64 and the other for 1 500.

3) Conversion board

This provides the voice compression, multiplexed format conversion, and line edit functions in units of 8 kb/s, 3.2 kb/s, and 0.4 kb/s.

4) Unit controller

This CPU board controls the entire unit via the control bus to enable initial start, failure management, and inter-DMIX communication. The CPU board can also interface with the NSP for DMIX.

### 4.4 Software configuration and outline of processing

The software has a modular structure. The functions of the main modules are outlined below.

1) EC control module

This module is the system's main control unit. It manages the internal data bus, makes setting changes, and monitors the alarm notification status for each unit.

2) Network port control module

This module monitors and manages the status of the high-speed digital leased line and controls inter-DMIX communication.

3) ISDN control module

This module handles the ISDN protocol and can interface with the INS Net. It receives execution commands for call out to a network, call in, and data selection for transmission over a network from the EC control module.

### 4.5 Rerouting

If the high-speed digital leased line or network port hardware fails, communication can be switched from the failed line or port to the public ISDN. This makes the network highly reliable. The following outlines how rerouting is

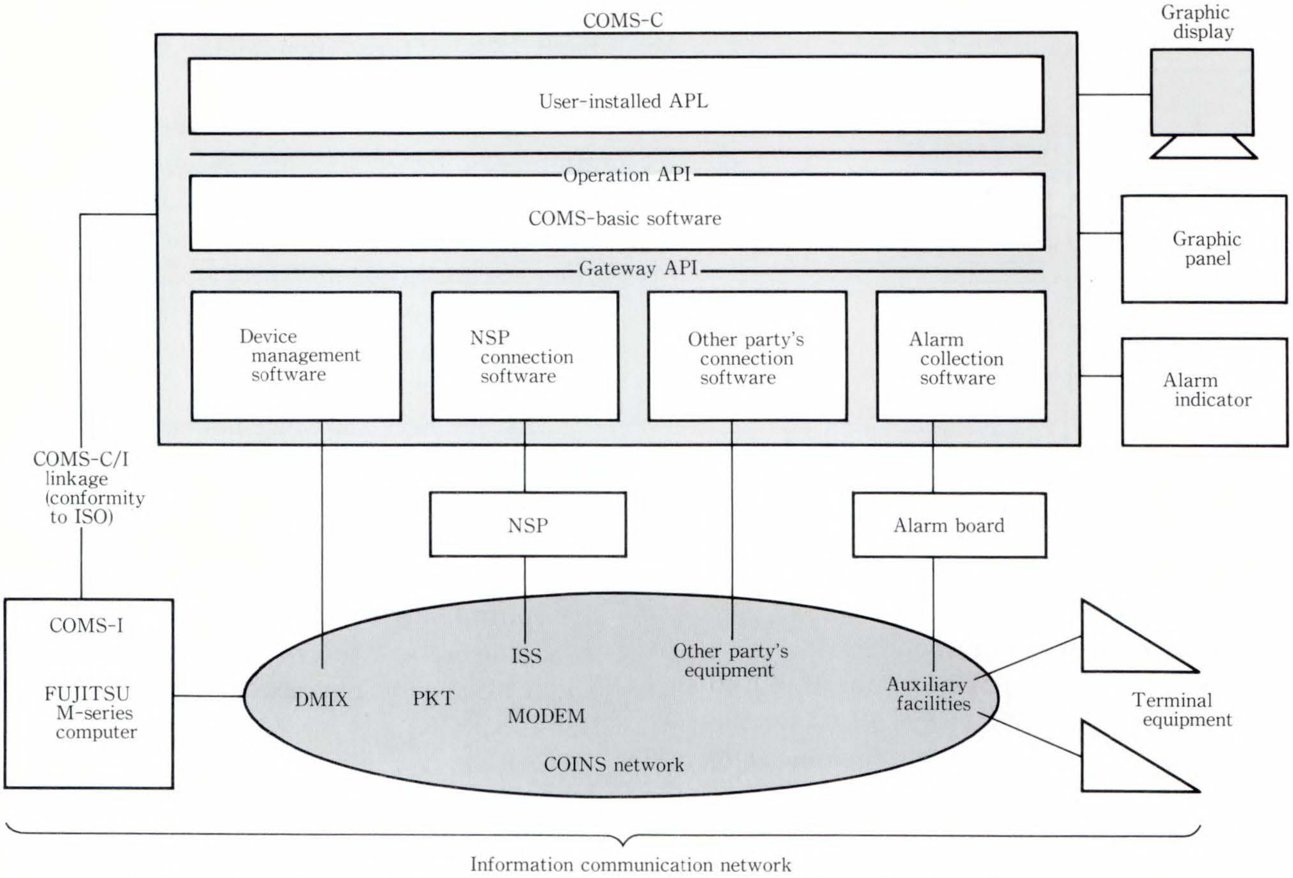



Fig. 7—System configuration and monitor mode of COMS-C.

performed.

1) Instigating and switching back

Rerouting can be instigated in two ways: either automatically by the DMIX equipment or manually by the system operator. The former is triggered when the high-speed digital leased line or hardware port fails provided that a predefined reroute condition is satisfied. The latter case may arise during system operation testing and maintenance. The system operator can force the NSP to perform rerouting by inputting a command. Switching back is left up to the system operator's judgment. Only manual control is available. It is performed with the ISDN control module, via the EC control module, by inputting a command from the NSP console.

2) Connecting to the public ISDN

If a high-speed digital leased line fails, the network port module sends an alarm signal to the EC control module. If the EC control module decides rerouting is necessary it issues a network call-in instruction to the ISDN control module. The ISDN control module connects the ISDN line, then connects the defined data path. At the same time, the monitor control communication path is reported to the DMIX.

When the ISDN control module notifies the EC control module that ISDN connection is complete, the EC control module switches the data path from the high-speed digital leased line to the ISDN line.

## 5. LAN system[5)]

### 5.1 Purpose of development

MHLINK was developed as a multimedia backbone LAN to handle voice, data, and video data, and enables the building of a corporate ISDN based on COINS products. The following points were considered during development:

1) Standard support

In future LAN systems, there will be an ever-increasing demand for multi-vendor environments. Therefore, a standard interface must be adopted.

2) Inter-LAN communication

For LANs, standardized branch lines such as CSMA/CD (ISO8802-3) and FDDI (ISO9314) are already established. To communicate between branch LANs LANs can be connected by means of a MAC bridge system with a self-learning function. Also, by linking with a COINS system, a remote LAN can be connected, enabling larger and wider-range inter-LAN connections.

3) Enhancement of line interface support

By supporting high-speed line interfaces such as V.35, X.21, RS449, and the ISDN interface, expected to prevail in the future, higher-speed terminals can be connected.

## 6. Network Management System (NMS)

### 6.1 Purpose of development

CMOS-C is a centralized network monitoring system for COINS networks. It has been developed with the following aims:

1) To monitor and manage each component within a network, including auxiliary equipment.
2) By linking the NSP of each COINS component, to manage all devices as a unit.
3) To supervise networks including non-COINS components provided by other vendors, by supporting the OSI-standard NMS interoperable interface.

### 6.2 System outline

#### 6.2.1 System configuration

Figure 7 shows the CMOS-C configuration. Network components such as DMIX, ISS, packet switching equipment, and modems are connected directly, or via each NSP, to CMOS-C. COMS-C makes it possible to manage a network incorporating various devices by modeling the network using the three elements of node, port, and line.

It is also important to manage the auxiliary equipment in the machine room. This might include monitoring the air conditioning status and whether the room door is open or closed to ensure network security. COMS-C enables this kind of management with standard data collecting devices.

Other companies' equipment in a network can be macro-managed by the above-mentioned standard data collecting devices. It can also be managed in the same way as the COINS equipment by installing connection software. COMS-C, provided with the standard gateway Application Program Interface (API), enables other vendors' software to be installed easily. Moreover, the desired management system can be built by installing application software with the operation API.

#### 6.2.2 Outline of functions

COMS-C includes the following standard OSI system functions. These functions support the entire cycle of network design, installation, operation, and evaluation. They also keep the service level stable.

1) Configuration management

For defining configuration information using simple language, managing data expansion, collecting data on network operation status and operation control

2) Failure management

For detecting failures, isolating them, reporting, indicating the affected range and called party, recovery action guidance, and history information management

3) Performance management

For managing traffic information for toll dial networks and packet networks

4) Accounting management

For collecting, editing, and output of toll dial network accounting information

5) Security control

For checking access permission by means of a user ID and password, controlling output information in accordance with user attributes, and for control command control.

## 7. B-ISDN overview

Currently, public networks are being prepared for the introduction of Broadband ISDN

(B-ISDN) and flexible networks[6]. The next COINS generation will support B-ISDN as shown below.

### 7.1 Wide area connection (inter-LAN communication)[7]

Inter-LAN communication is becoming increasingly common. For Asynchronous Transfer-Mode (ATM) we are targeting a LAN-specific connectionless communication environment where line capacity is used most effectively and where there is no loss of communication in the event of a failure. Future development will focus on[8]:

1) B-ISDN Switched Multi-megabit Data Service (SMDS).
2) Addition, to PBX, of a function equivalent to SMDS.
3) Dynamic routing processing within the PBX trunk.

### 7.2 Effective line accommodation algorithm

Since B-ISDN is based on cell multiplexing, it is difficult to insert an effective line accommodation algorithm developed for conventional circuit switch-based fixed-rate transmission lines. Consequently, we intend to develop a line accommodation algorithm based on statistical multiplication. The main technologies involved are:

1) For voice/graphics coded communication, high quality variable rate coding that takes advantage of ATM dynamic capacity allocation.
2) Effective accommodation of burst traffic such as data.

### 7.3 Simplified local network architecture

We aim to use ATM to simplify networks by unifying various service access interfaces into "cells". To do this, we will position a LAN nucleus within the corporate network configuration and build up a system which features expandability, improved robustness to failure and improved personal services. The main development technologies are:

1) An expandable LAN that can easily accommodate the services provided by B-ISDN.
2) In addition to a simplified wiring system, a distributed processing technology using several, small-scale machines linked by LAN.

### 7.4 Improved network configuration flexibility

We foresee that various public network services, such as services that allow users to freely define network configurations, are likely to appear in the future. Accordingly, we intend to improve the service quality and economy of corporate networks.

We aim to achieve this by providing network design tools for planning the optimum network configuration based on traffic status decisions.

## 8. Conclusion

We have outlined ISDN and COINS, focusing on the FETEX-3000A series ISS, DMIX-E, LAN systems, and COMS-C. In future network environments that feature COINS, high-speed, broadband, and wide-area communication will become ever more prevalent and, indeed, will become central to global networks. The direction in which COINS must move to meet the challenge of B-ISDN has also been discussed. ISDN will be a long time in coming to most households. However, high-definition TV (HDTV) will provide a means of visual communication, supporting animation and the like, and will show the real merit of broadband (156 Mb/s, 622 Mb/s) communication.

COINS will continue to offer more attractive products and services, for example, greater availability and reduced failure rates and data transmission delays. As a failure example, if traffic is low, a higher-quality voice/image service should be supplied. Today, we are developing variable-speed coding technologies and arranging the technologies that can support such improvements.

Progress toward B-ISDN will be made in stages, and some of these features can be installed without waiting for B-ISDN.

## References

1) Fujitsu: Fujitsu ISDN System. (in Japanese), *FUJITSU Journal*, **14**, 9 pp. 40-43 (1988).
2) Fujisaki, Y., Nakamura K., and Hara, T.: COINS Products. (in Japanese), *FUJITSU*, **41**, 4 pp. 300-312

(1990).
3) Fujisaki, Y., and Hayashi, T.: Integrated Services Private Digital Communication System. Proc. PTC' 85, pp. 226-230.
4) Muramoto, T., Nomura, Y., and Sato, Y: FETEX-3000 Series Digital PBX. (in Japanese), *FUJITSU*, **37**, 5 pp. 368-373 (1986).
5) Nojima, S. et al.: Local Area Network. *FUJITSU Sci. Tech. J.*, **28**, 2 (Special Issue on Telecommunications), pp. 216-227 (1992).
6) Asatani, K., and Ikeda, Y.: Present Status and Future Trends of the Study on the Broadband ISDN. (in Japanese), *Trans. IECE*, **J72-B-I**, 11, pp. 886-895 (1989).
7) Takahashi, A., Nishino, T., Murakami, K., and Dunning, S.C.: A Broadband Switching System for Public Network Proc. ISS' 90, **Vol. V**, 1990, pp. 103-109.
8) Iguchi K. et al.: A study on Architecture of Optical Passive Bus for Broadband Customer Premises Network. (in Japanese) *Trans. IECE*, **J72-B-I**, 11, pp. 961-969 (1989).

**Tatsuki Hayashi**
Transmission Network Dept.
FUJITSU LIMITED
Bachelor of Electronics Eng.
Yamanashi University 1967
Master of Electronics Eng.
Yamanashi University 1969
Specializing in Transmission Equipment

**Kozo Nakamura**
Switching Network Engineering Dept.
FUJITSU LIMITED
Bachelor of Control Eng.
Tokyo Institute of Technology 1970
Specializing in Switching Equipment

**Toshitaka Tsuda**
Optical Interconnection Dept.
FUJITSU LABORATORIES, KAWASAKI
Bachelor of Electrical Eng.
The University of Tokyo 1970
Dr. of Electrical Eng.
The University of Tokyo 1975
Specializing in Digital Signal Processing and Communication LSIs

UDC 621.395.345:621.395.74

# ISDN PBX for the International Market

● Jun Shimada ● Toshiyuki Sato ● Toshihito Yamashita

*(Manuscript received November 1, 1991)*

Both public and private ISDN facilities have been incorporated in the FETEX-9600 and FETEX-600 series, and were released in the U.S. and Australian markets in April, 1990. The success of that introduction has proven the FETEX-9600/600 to be a fully ISDN compatible PBX.

This paper discusses the key technologies used in the introduction of ISDN into the FETEX-9600/600 and looks at the future directions of FETEX-9600/600 development as the technologies of computing and telecommunications merge.

## 1. Introduction

Integrated Services Digital Network (ISDN) is probably the most significant telecommunications technology yet developed. The ISDN network offers a level of service and performance not possible in the Public Switched Telephone Network (PSTN). ISDN features include faster call set up and a higher grade of telecommunications such as Calling Line Identification (CLI), indication of call progress, and greater data transmission capacity through the use of 64 kb/s digital channels.

The development program of the FETEX-9600 and FETEX-600 series has been designed to keep in step with developments in public ISDN networks in Fujitsu's international markets. Fujitsu implemented private ISDN networking on the FETEX-9600/600 after a detailed analysis of the private ISDN networking requirements of the international markets. The FETEX-9600/600 now provides full interconnection to public and private ISDN networks, providing users with a complete digital link, new applications and advanced networking features.

This paper describes the architecture and technology used to implement market requirements in the FETEX-9600/600. Special emphasis is given to the key technologies used for the introduction of ISDN. This paper also discusses the future directions of the FETEX-9600/600 as the technologies of computing and telecommunications merge.

## 2. ISDN development and ISDN PBX

The history of ISDN development is one of ISDN international standardization. The CCITT have been working to standardize the Primary Rate Interface (PRI) and Basic Rate Interface (BRI) for the public ISDN network, and have published various CCITT recommendations: the Red book in 1984, the Blue book in 1988, and a Yellow book is planned for 1992.

The private ISDN network, or inter-PBX signaling standard in the ISDN environment, has been discussed by various standards bodies. In the CCITT, work has started within the ISO working group.

Prior to the completion of an international ISDN PBX networking standard, Fujitsu adopted an ISDN D-channel based inter-PBX signaling system. This system was implemented in the FETEX-9600/600 in 1989. Since 1990, a number of FETEX-9600/600 systems have been commercially delivered and installed, making them one of the first PBXs in the world to incorporate ISDN D-channel based signaling for both public and private ISDN networks.

Although ISDN PBX is still called a PBX, it should be classified as another category of switch separate from non-ISDN PBX, due to its tremendous potential and impact on the telecommunications market. The requirements for ISDN PBX are described below.

1) Co-existence with existing networks

Migration from non-ISDN to full ISDN will not be carried out in one step. Both kinds of network need to exist simultaneously until migration is complete. In addition to the ability to function together with non-ISDN networks, the provision of a smooth migration path to the world of full ISDN is mandatory.

2) Flexible protocol signaling system

Even when an international standard has been established, there will still be differences in implementation. A flexible ISDN signaling system is important.

3) Interconnection to the computer world

Although ISDN is based on a telecommunications platform, it must provide a bridge to the


Fig. 1–FETEX-9600.

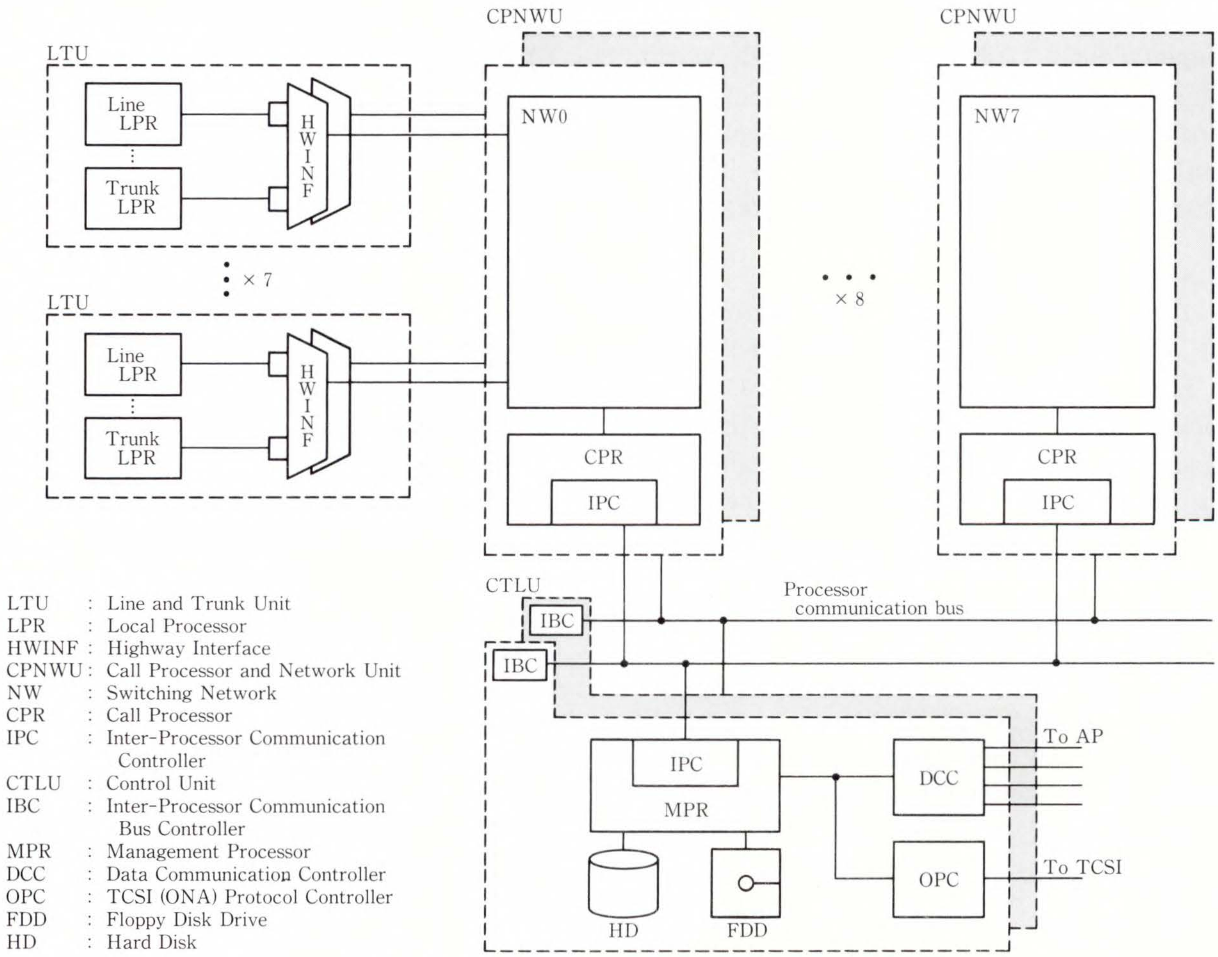



Fig. 2–FETEX-9600XL system architecture.

computer world as the technologies of telecommunications and computers merge.

4) Open architecture

As the requirement for common elements of telecommunications and communications systems becomes increasingly important, an open architecture interface, either physical or logical, is essential.

5) Future potential

ISDN technology is still being developed. The PBX architecture must be designed so that its functions and performance can be enhanced in future.

## 3. FETEX-9600/600 system architecture

In line with the above requirements, the FETEX-9600/600 has been developed to provide PBX customers with advanced applications and network services for the ISDN telecommunications environment. (The domestic versions of FETEX-9600 and FETEX-600 are FETEX-3000 and FETEX-2000 respectively.) To ensure complete compatibility with the ISDN world, Fujitsu has incorporated international ISDN standards throughout the system design, while simultaneously creating a sufficiently flexible design to include future ISDN enhancements.

### 3.1 System structure

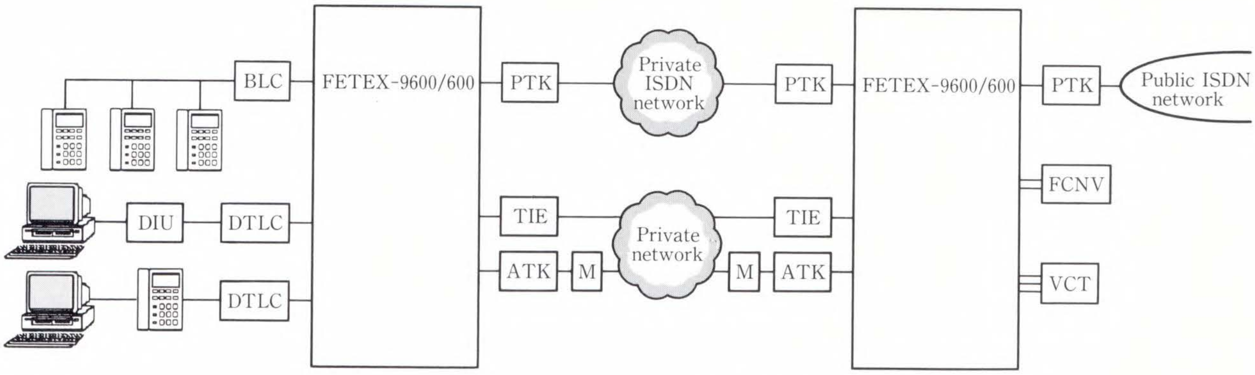
The FETEX-9600 is a high performance PBX with a modular architecture based on 16- or 32-bit multi-processor control and a non-blocking single stage digital switching network designed to accommodate ISDN traffic[1),2)]. This design covers a range of line numbers from 100 to 9 600 lines in four hardware packages, VS, M, MS and XL. Commonality is maintained throughout these systems in all terminals, line and trunk cards, as well as software and other ISDN facilities. The system can be field-upgraded from the smallest to the largest number of lines while still maintaining complete ISDN compatibility. Figure 1 is a photograph of the outside of the FETEX-9600. The architecture of the largest configuration, the FETEX-9600XL, is shown in Fig. 2. System parameters are summarized in Table 1.

The FETEX-600, on the other hand, is designed for a smaller number of lines (50 to 400 lines). The FETEX-600 is controlled by a 16-bit main processor, and the network configuration is also non-blocking single stage. The system has been optimized, depending on the number of lines, in three products, the FETEX-610, FETEX-620 and FETEX-640. The system parameters of the FETEX-640 are summarized in Table 2.

Table 1. FETEX-9600 system summary

| Item | Specification |
|---|---|
| Control scheme | Stored program distributed control<br>Hierarchical control architecture<br>(LPR-CPR-MPR)<br>Management Processor<br>(MPR) : 1 (duplicated)<br>Call Processors<br>(CPR) : 1 to 8 (duplicated) |
| Network | Single stage time division PCM network<br>(16 384 internal time slots) |
| Processors | MPR: 32-bit micro processor (i80386)<br>CPR: 16/32bit micro processors (i80286/386)<br>LPR: 4/8/16-bit micro processors<br>DCC: 16-bit micro processor (i80186)<br>OPC: 16-bit micro processor (i80186) |
| Main memory | MPR: 16 Mbytes<br>CPR: 8 Mbytes |
| File memory | Hard disk (40 Mbytes) and floppy disk |
| Line capacity | 9 600 lines |
| Trunk capacity | 3 000 trunks |
| Applications processor ports | 16 |
| Voice coding | PCM $\mu$-law or PCM A-law |
| Proprietary terminal interface | 64 + 64 + 16 kb/s (2B + D)<br>Time compression multiplex, 2 wires |
| Primary rate interface | 1 544 kb/s (23B + D) for T1<br>2 048 kb/s (30B + D) for CEPT |
| Basic rate interface | S-interface |
| Cabinet dimensions | $850(l) \times 500(b) \times 2\,000(h)$ (mm) |

The table describes the top model, the FETEX-9600XL.
LPR: Line Processor
DCC: Data Communication Controller
OPC: TCSI (ONA) Protocol Controller




BLC* : BRI Line Circuit
DIU : Data Interface Unit
DTLC : Digital Telephone Line Circuit
PTK : PRI Trunk
TIE : TIE Trunk
ATK : Analog PRI Trunk
FCNV : Format Converter
VCT* : Voice Compression Trunk
M : Modem
(* : not available for FETEX-600)

Fig. 3–FETEX-9600/600 ISDN facilities.

Table 2. FETEX-600 system summary

| Item | Specification |
|---|---|
| Control scheme | Stored program distributed control |
| Network | Single stage time division PCM network (1 024 internal time slots) |
| Processors | 16-bit micro processor (i80186) |
| Main memory | EPROM for generic program: 2 Mbytes<br>RAM for customer data: 1.25 Mbytes |
| File memory | Floppy disk (for customer data backup) |
| Call handling capacity | 4 000 BHCA |
| Line capacity | 424 lines |
| Trunk capacity | 64 trunks |
| Applications processor ports | 2 |
| Voice coding | PCM $\mu$-law or PCM A-law |
| Proprietary terminal interface | 64 + 64 + 16kb/s (2B + D)<br>Time compression multiplex, 2 wires |
| Primary rate interface | 1 544 kb/s (23B + D) for T1<br>2 048 kb/s (30B + D) for CEPT |
| Basic rate interface | (planned for future release) |
| Cabinet dimensions | $900(l) \times 530(b) \times 510(h)$ (mm) |

The table describes the top model, the FETEX-640.

### 3.2 ISDN facilities

The ISDN facilities of the FETEX-9600/600 are shown in Fig. 3. The design of the FETEX-9600/600 allows all these facilities to be incorporated into the current operating system without system modifications such as add-on switch modules and so forth.

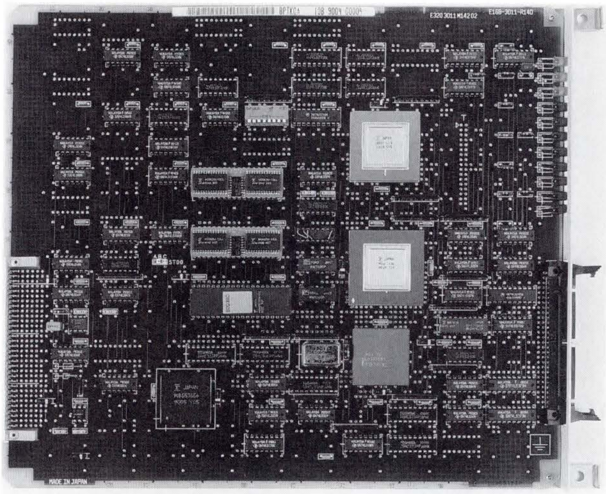


Fig. 4–FETEX-9600 PTK card.

1) Primary Rate Interface (PRI)

The PRI Trunk (PTK) provides a primary rate digital link to the public ISDN network or to other PBXs in private ISDN networks. Both 1.544 Mb/s (23B + D) for T1 and 2.048 Mb/s (30B + D) for CEPT have been developed. All functions and components are mounted on a single set of boards. This has been achieved through high density circuit design. Figure 4 is a photograph of the FETEX-9600 PTK card.

In addition to the standard primary rate network, the PTK can be used to connect to internal PBX facilities such as multiplexers to create greater capacity transmission paths ($N \times 64$ kb/s).

2) Basic Rate Interface

The BRI Line Circuit (BLC) provides a Basic Rate Interface (2B + D) to allow ISDN telephones and terminal adapters to be installed. The BLC provides a four-wire interface conforming to CCITT S-interface recommendations. The BLC board can accommodate up to eight circuits. Both point-to-point and point-to-multipoint (eight terminals per circuit) connections are provided.

3) Analog D-channel interface

Although the digital telecommunications era has now arrived, private analog network lines are still used extensively, and they offer an economic advantage in certain existing, voice-oriented, PBX interconnections. To improve these networks, a common channel, inter-PBX signaling scheme is more effective than simple, digit based addressing.

The Analog D-channel trunk (ATK) has been developed to meet this requirement. The ATK is designed to terminate ISDN D-channel signaling. The D-channel signal extracted from the ATK is converted into an analog signal and transmitted via a synchronous modem (up to 19.2 kb/s) to the other end of the private network connection.

4) Rate adaptation

The Format Converter (FCNV) can convert digital streams from pre-ISDN data terminals into an ISDN compatible format (V110). By using an FCNV, any data terminal device that can be connected to the FETEX-9600/600 PBX (V.24/V.28, V.35, X.21) can communicate with ISDN equipment located either within the PBX or in the public ISDN network.

When receiving incoming ISDN data calls, the FETEX-9600/600 identifies the type of ISDN terminal from the Low Layer Compatibility (LLC) element of the ISDN setup message. It then automatically converts to the format appropriate to the type of terminal equipment (ISDN or non-ISDN) attached to the FETEX-9600/600.

5) Voice compression

Voice compression technology can double the efficiency of a private network ISDN channel. The Voice Compression Trunk (VCT) converts either $\mu$-law or A-law PCM signal into CCITT compatible ADPCM (32 kb/s) signals, then bundles them into a 64 kb/s channel or vice-versa.

The VCT can also detect Group 3 facsimile signals and convert them appropriately, thus smoothing the migration from non-ISDN to ISDN private networking.

6) ISDN terminals

The FETEX-9600 can accommodate existing Single Line Terminals (SLTs), Proprietary Digital Terminals (MLDTs) and Data Interface Units


Fig. 5–SRS-400.

Table 3. Public ISDN protocol implementation

| Interface | Protocol Specification | Vendor | Exchange |
|---|---|---|---|
| BRI | TA-TSY-000286 | Bellcore | – |
| | 5D5-900-311 | AT & T | – |
| PRI | PUB41449, PUB41459 | AT & T | 4ESS |
| | 5D5-900-312 | AT & T | 5ESS |
| | NIS A211-4 | Northern Telecom | DMS250 |
| | NIS A211-1 | Northern Telecom | DMS100 |
| | TPH1856 | L.M. Ericsson | AXE10 |
| | TIF-218 | Fujitsu | FETEX-150 |

(DIUs). It can also accommodate ISDN terminal equipment such as BRI Terminal Adapters (TAs), BRI stations and personal computers with internal BRI cards. Table 3 lists the BRI protocols currently supported by the FETEX-9600.

The SRS series of Fujitsu ISDN terminals can be connected with the FETEX-9600 (see Fig. 5). Other vendor equipment supporting the protocols listed in Table 3 can also be connected. Such terminals allow a variety of interfaces and applications to be used with the FETEX-9600.

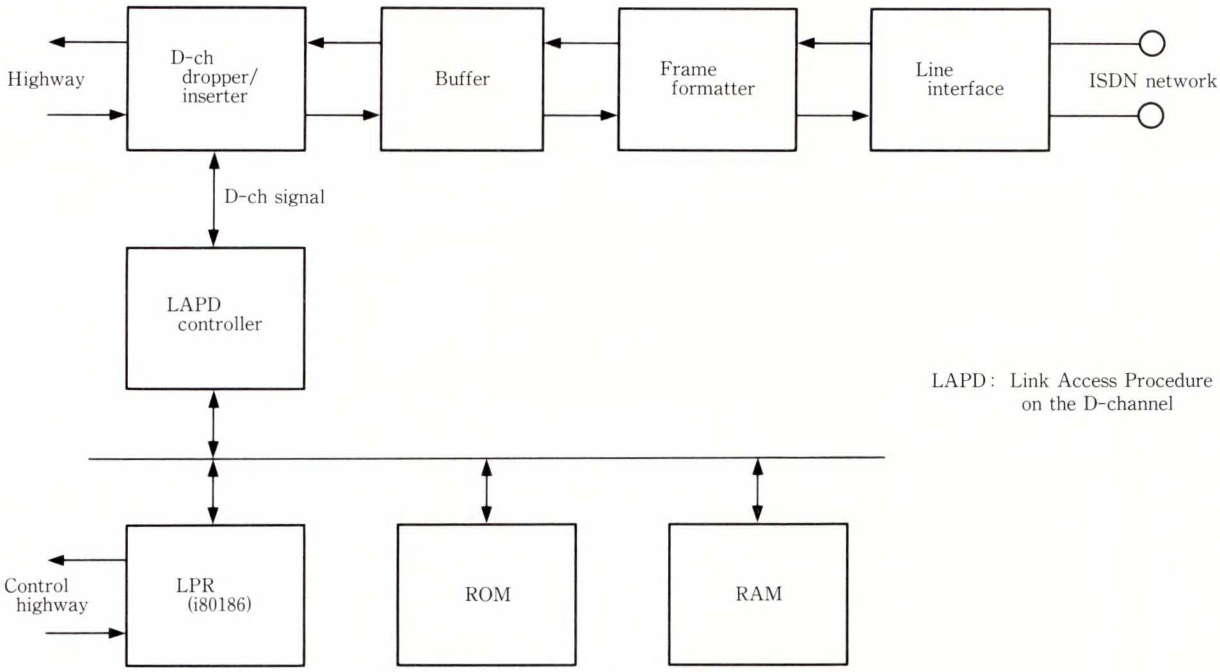



Fig. 6–Hardware structure of PTK.

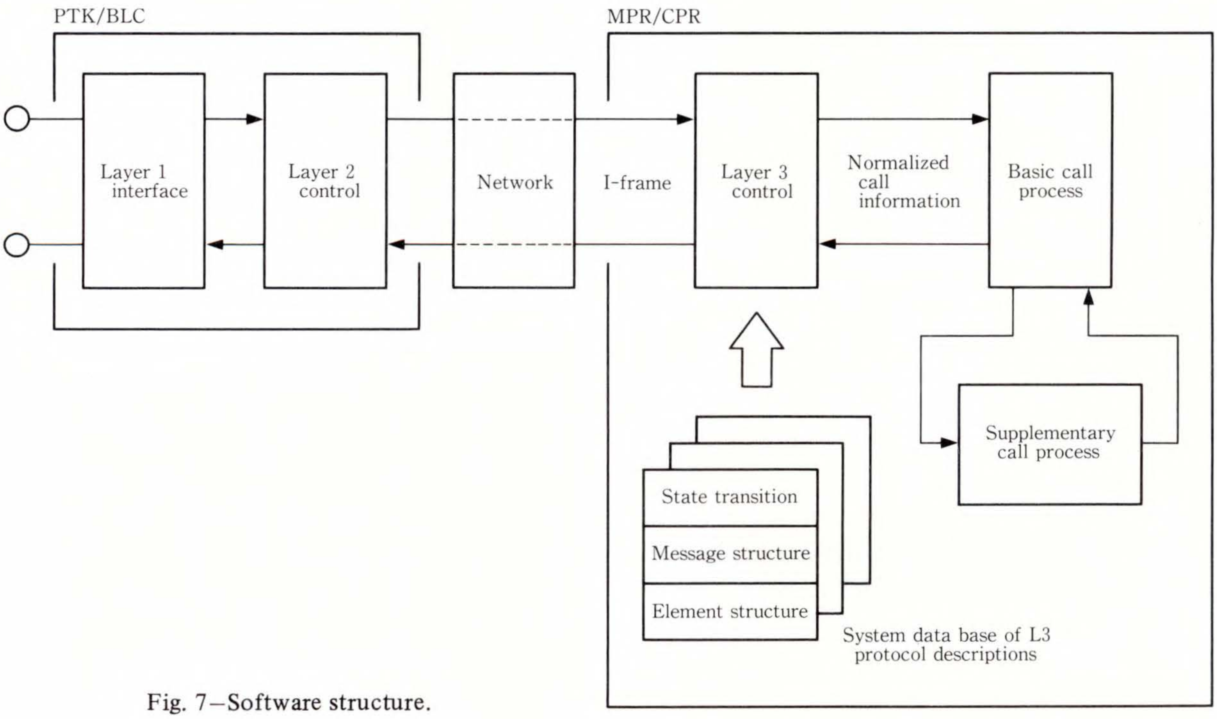



Fig. 7–Software structure.

### 3.3 Facility controls

The characteristics of the structure of the hardware and software used to control the ISDN facilities and protocols of the FETEX-9600/600 are described below.

1) Hardware structure

The ISDN D-channel facilities, such as the PRI Trunk (PTK) and the BRI Line Circuit (BLC), are designed to include both D-channel signaling termination and bearer channel transmission in the board as shown in Fig 6. The 16-bit high performance Line Processor (LPR) controls ISDN layer 2, while the PBX Management and Call Processor (MPR/CPR) controls layer 3 and the call processing under the distributed control architecture (see Fig. 7). The communication path between the LPR and the MPR/CPR is through the PBX non-block digital network.

As with non-ISDN line and trunk cards, these facilities are designed so that they can be universally installed in any card slot in the line and trunk units. This allows the system installer to allocate the cards flexibly, regardless of the type of card, and hence achieve the most cost effective arrangement.

2) Software structure

Although based on the CCITT international standards, there are minor deviations in the protocol specifications for each country and/or ISDN exchange. This has been caused by different interpretations of the standards and vendors' implementation strategy.

The table driven software structure in the layer 3 protocol control section and the layered control architecture (corresponding to protocol and call processing) allow multiple protocol control with a standardized interface (see Fig. 7). This software structure also features independent protocol control and the flexibility to incorporate new protocols and supplementary services in future.

The distributed ISDN control scheme supported by this hardware and software produces a linear relationship between PRI accommodation and traffic load as shown in Fig. 8 for the FETEX-9600XL.

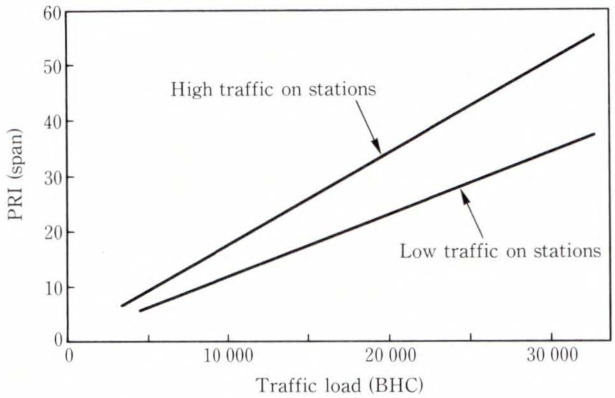



Fig. 8—PRI traffic characteristics.

### 3.4 Protocol implementation

#### 3.4.1 User-Network interface

The protocols which allow connection to the public ISDN network and ISDN terminals including Primary Rate Interface and Basic Rate Interface are listed in Table 3. These interconnectabilities have already been verified and authorized by the ISDN exchange authorities or its equivalent in each country.

#### 3.4.2 User-User interface

1) Inter-PBX signaling history

Inter-PBX signaling has been studied and developed by various PBX vendors to provide enhanced PBX-to-PBX networking using common channel signaling. The X.25 based signaling system was developed in the early 1980s[3]. Then, in the mid 1980s, an HDLC based signaling system called Digital Private Networking Signaling System (DPNSS) was developed in Europe for inter-PBX signaling[4]. The CCITT Signaling System 7 (SS7) was also used for inter-PBX signaling.

Since ISDN D-channel signaling was still under discussion during the 1980s, the early inter-PBX signaling schemes were non-ISDN and tended to develop proprietary protocols. However, as open network systems took hold toward the end of the 1980s, the ISDN D-channel signaling system became a serious contender for inter-PBX signaling system development.

2) International standards

Work towards an international standard was started by the Joint Technical Committee 1 within the ISO working group[5]. Initially, private protocols such as SS7, DPNSS and ISDN

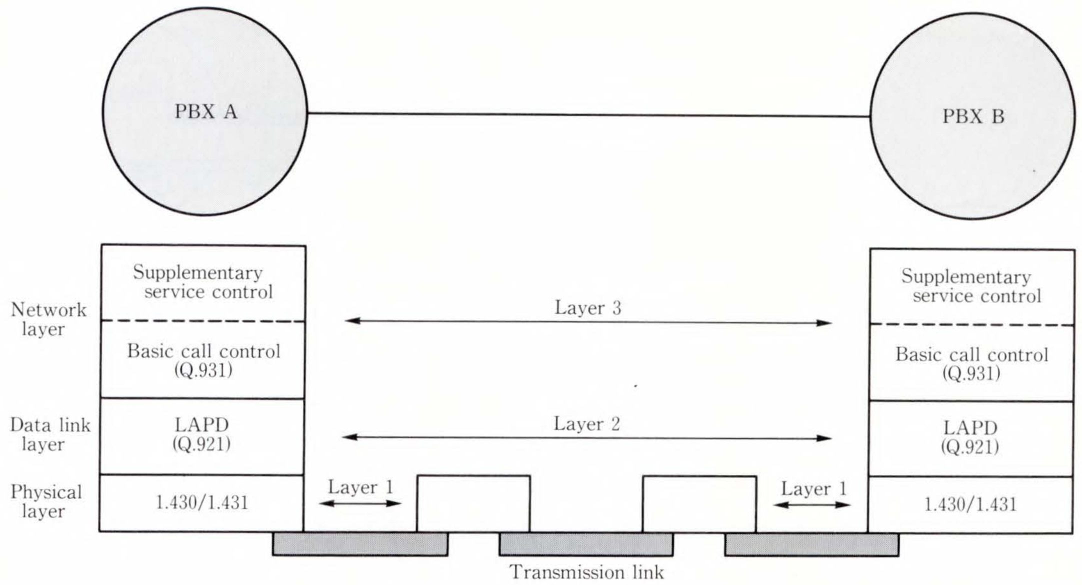



Fig. 9–Protocol structure.

PRI D-channel based signaling had been tabled for consideration for private networking systems. The outcome of these discussions was that SS7 should be applied to public network signaling, and that ISDN PRI D-channel based signaling would be used for private network signaling and access to customer premises.

The international standard for basic call control should be available in early 1993, the generic supplementary service control procedure is expected in the middle of 1994, and the first supplementary service standard is due to be released in 1995.

3) Fujitsu's approach

Due to the slow progress toward mutual agreement on an international protocol standard and Fujitsu's desire to be the leader in this technology, we have adopted an ISDN D-channel based inter-PBX signaling protocol called TeLinc[Note]. This was originally developed by Telecom Australia[6]. As illustrated in Fig. 9, the structure of the protocol layer, including the Physical Layer (L1), the Data Link Layer (L2) and the Network Layer (L3), complies with the ISDN User-Network interface standard. Control for supplementary services is also built into layer 3 to provide feature transparency across private networks. TeLinc is believed to provide the smoothest migration path toward the developing international standard.

Note: TeLinc is a registered trade mark of the Australian Telecommunications Commission.

## 4. ISDN applications

The applications of ISDN are of direct interest to an account manager who is responsible for the decision to introduce an ISDN PBX. As described below, the FETEX-9600/600 provides many benefits to businesses which are not available from non-ISDN PBXs.

### 4.1 High speed data transfer

The 64 kb/s clear channel enables high speed, reliable data communications on an ISDN network wide basis. Applications for this kind of bearer service include Group 4 facsimile, picture videotex and file transfer.

By using Data Interface Units (DIUs), the Data Terminal Adapter (DTA) in the Proprietary Digital Telephone (MLDT) and BRI Terminal Adapters (TAs), the FETEX-9600/600 provides various standard interfaces for high speed data transmission such as X21 and V35. As a direct result of the FETEX-9600/600's development strategy, all these terminals can communicate with each other, be they ISDN or non-ISDN.

For applications such as video conferencing, LAN bridging, and CAD/CAM interconnection which require higher data speeds, $N \times 64$ kb/s channel aggregation can be achieved by combin-

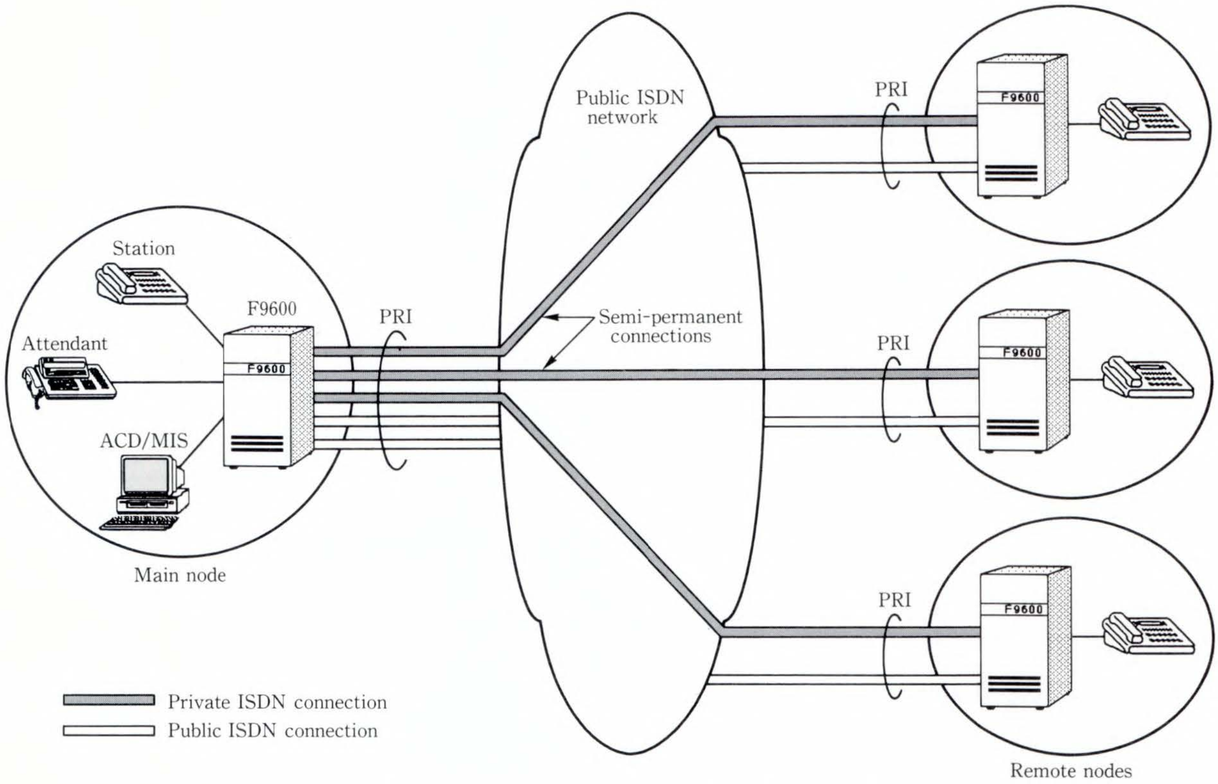



Fig. 10–Example of FETEX-9600 ISDN private network.

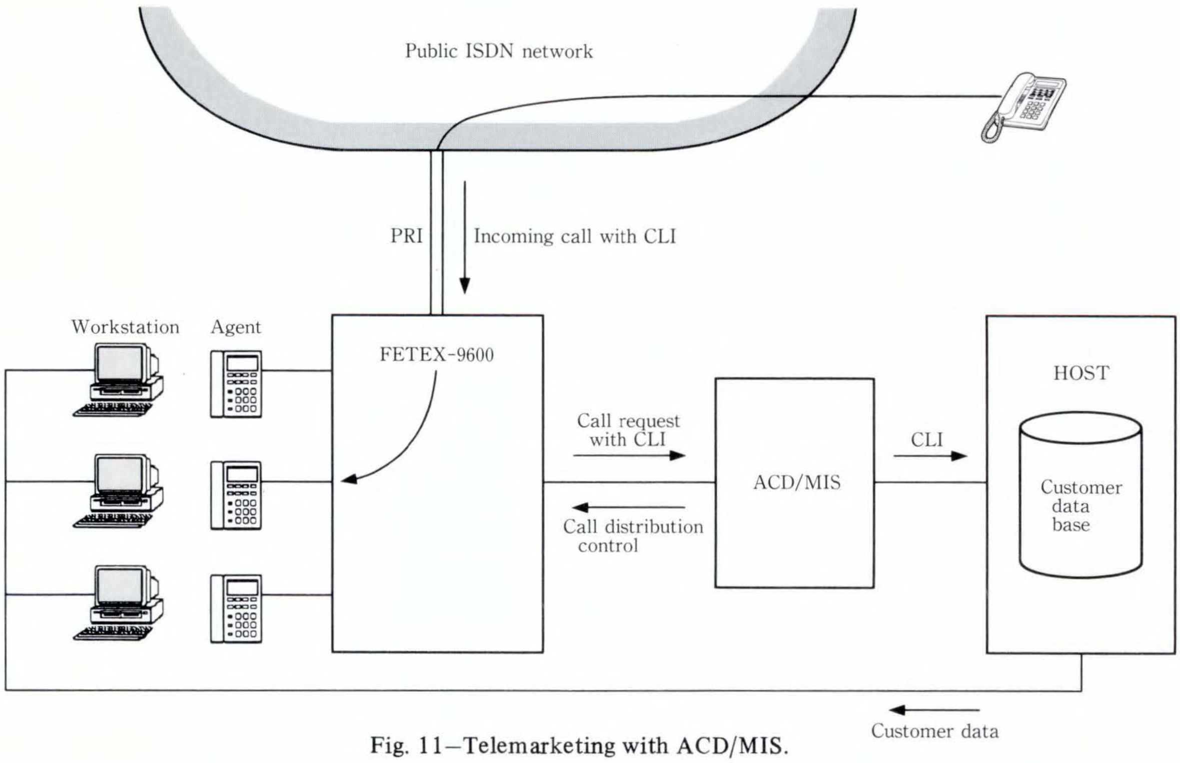



Fig. 11–Telemarketing with ACD/MIS.

ing the basic 64 kb/s B-channel speed of the PRI interface with an ISDN multiplexer. This provides switched high speed data communications.

### 4.2 ISDN networking

The Fujitsu ISDN Private Network (FIPN), based on TeLinc, provides facilities such as

Table 4. ISDN private networking supplementary features

| Service type | Feature name |
|---|---|
| Unsuccessful call | Station camp-on<br>Executive busy override<br>Executive camp-on<br>Data camp-on |
| Call forwarding | Call forwarding all calls<br>Call forwarding on busy<br>Call forwarding on no answer |
| Night service | Night mode |
| Three party call | Consultation hold/transfer<br>Three party conference |
| Multi-party conference | Eight way conference<br>Meet me conference |
| Extension features | Distinctive ringing<br>Do not disturb<br>Malicious call trace |
| Call information display | Calling/called number display<br>Connected number display<br>Station/trunk name display<br>Call status display<br>Reason for call fail |
| Private network services | Uniform numbering plan<br>Automatic network routing<br>ACD overflow<br>Look ahead routing<br>Multiple ACD groups<br>Centralized voice mail system<br>Route optimization |
| Operator services | Incoming trunk type display<br>Transfer<br>Alternating<br>Three way conference<br>Operator call<br>Camp-on<br>Verify<br>Centralized attendant service |
| Maintenance service | Annoyance call trace |

uniform numbering, a variety of transparent network PBX features and network routing. FIPN allows users to make and receive calls in a "feature transparent" manner, as if the PBX network was unified into one PBX.

Another advantage of FIPN is its ability to centralize network facilities. For example, operator location, voice mail systems, and PBX management facilities which serve the entire private network can be centralized. The centralized attendant facility allows operators to be located in a single network node, yet make and receive calls on behalf of all locations in the network. An example of the existing FIPN network is shown in Fig. 10, and the networking features available are listed in Table 4.

Interworked with the public ISDN network, FIPN could enlarge existing private communications into a national or even international private network. ISDN networking features can thus be extended internationally.

### 4.3 Telemarketing

One of the major features that ISDN offers over PSTN is the provision of Calling Line Identification (CLI). CLI conveys the caller's number through the ISDN network to the call destination. Not only can the caller's number be displayed on the receiver's telephone, but it can be passed through the PBX to a host computer interface, opening up the possibility for a variety of inbound telemarketing applications.

The FETEX-9600 presently interworks this facility with its Automatic Call Distribution & Management Information System (ACD/MIS), which is one of the Application Processors (APs) supported by the FETEX-9600.

By passing CLI information through the PBX to a host computer interface, the host computer can use the CLI to make data base inquiries, identify and present a customer profile prior to answering the call, as shown in Fig. 11.

### 4.4 Telecommunications and computer integration

The trend over the past ten years to downsize computer equipment has led to more personalized data processing facilities, typically connected together to form Local Area Networks (LANs) or Wide Area Networks (WANs). However, most telecommunications and computer networks are still confined to customer premises.

This limitation can be removed by the PBX-to-computer bridge facility which provides a gateway from the PBX to a computer or computer network. A PBX network with voice facilities and a LAN with work stations can be merged through this bridge, thus enabling the computer to switch voice calls or the PBX in order to control data processing.

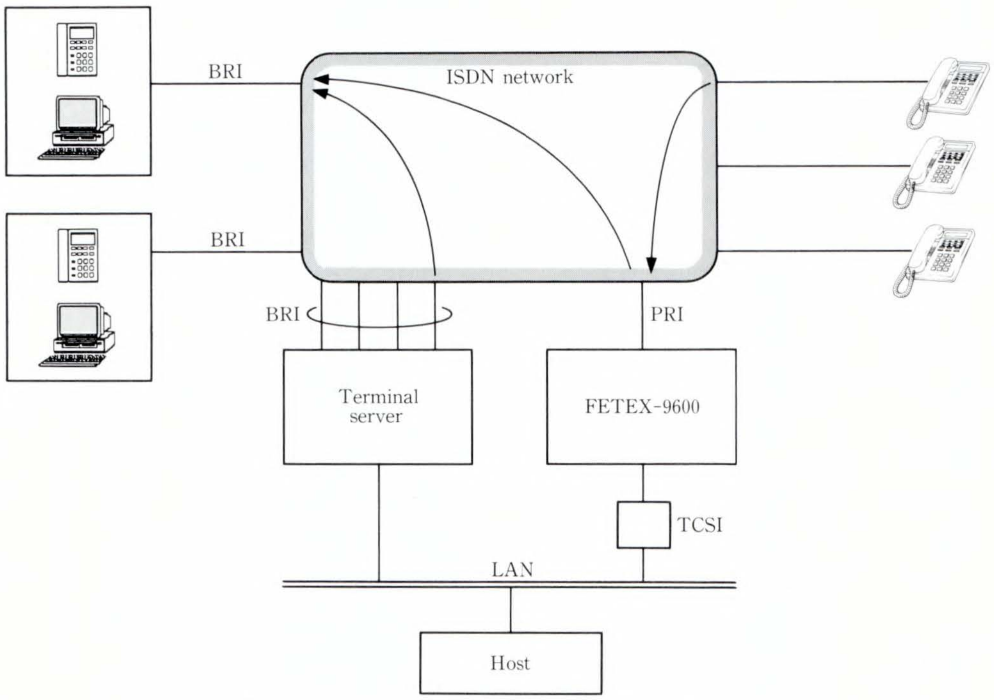



Fig. 12–Telecommunications and computers.

**4.4.1 Telecommunications and Computer Service Interface (TCSI)**

TCSI is a PBX-to-computer interface developed by Fujitsu. It is based on a simplified functional model, and designed so that the bridge between telecommunications and computers can be implemented easily. The major features of the model include:

1) Call control functions such as call setup and call release
2) Call data management functions for the data base used by PBX call processing
3) Input/output control functions including PBX terminal control
4) Fault and system management functions such as data base synchronization

Call control is the major PBX switching function controlled by computer, and it includes an optimized call status model defined in the call transition diagram, by referring to the ISDN call status. As a result, the number of basic call statuses is reduced to only five, which simplifies the design work for computer service applications[7].

**4.4.2 TCSI applications**

One application of telecommunications and computer integration is network wide voice and data base switching as shown in Fig. 12. Under control of the host computer, an inquiry call is switched and transferred by PBX through a voice network to the most appropriate location with the relevant data base on a LAN network.

## 5. Future directions

Fujitsu maintains a long term commitment to developing the internal and local switching capabilities of the FETEX-9600/600. Fujitsu will continue to adapt the FETEX-9600/600 to the changing environment to ensure that it remains a key player on the general communications platform, and is not limited to telecommunications alone.

The major issues still outstanding in the network, transmission and data processing environments are described below.

1) Public network: coordination with future network enhancements to provide more intelligence, by Intelligent Networks (IN) and Advanced Intelligent Networks (AIN), including interworking with Centrex.
2) Private network: provision of interconnectability to other private networks, both multi-vendor and multi-national, as well as integration with newer technologies such

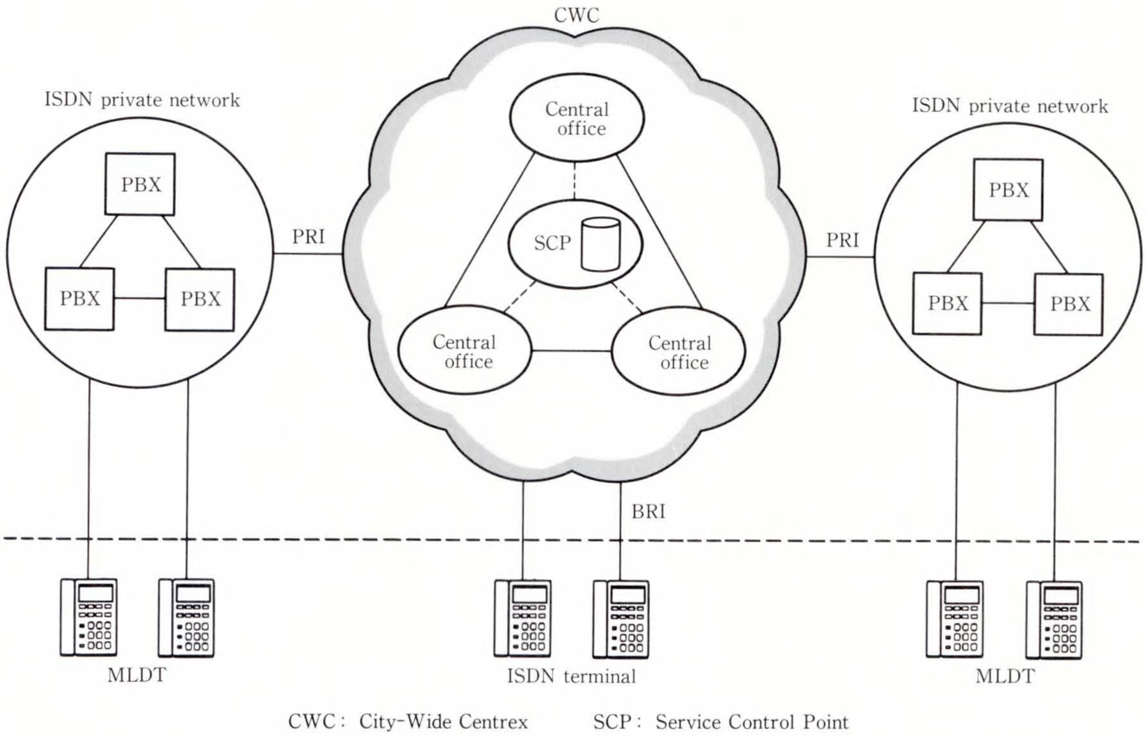



Fig. 13—Interworking with City-Wide Centrex (CWC).

as cordless PBX.

3) Computers: better integration with computer equipment.
4) Transmission: increase line transmission capacity for higher speed transmission schemes such as broadband ISDN.

Our approach to solving these outstanding issues is described below.

### 5.1 Interworking with City-Wide Centrex (CWC)

Like the development of D-channel based inter-PBX protocols, work is being conducted on inter-exchange signaling systems. This work is concentrating on the implementation and improvement of the feature control part of SS7 in order to overcome the public network's current lack of transparent feature control.

If Centrex services within a single node can be relaxed, network-wide and City-Wide Centrex (CWC) services could be made available. For economic reasons, CWC can serve a relatively small number of subscribers in a single location, but a very large number of subscribers over a wide area. However, PBX would be more economical if a large number of users are located in a single area.

By combining both CWC and PBX private networks through a ISDN D-channel signaling system similar to the existing inter-PBX signaling system, a huge integrated network could be established. Figure 13 shows the concept of CWC-PBX interworking where PBX station users and exchange subscribers participate as equals in the same network.

### 5.2 Broadband ISDN (B-ISDN)

For bulk data transfer for media and broadcasting, bearer speeds of around 100 Mb/s are required. Since Narrowband ISDN (N-ISDN) has a relatively low prime bearer speed of 64 kb/s, a new switch technology, B-ISDN is being developed.

Even though B-ISDN is introduced, N-ISDN will still carry a large percentage of telecommunications traffic. It is therefore important that, since the FETEX-9600 is developed as a B-ISDN switch, the system still supports N-ISDN. Figure 14 shows the approach for introducing B-ISDN to the FETEX-9600.

The system under development contains an

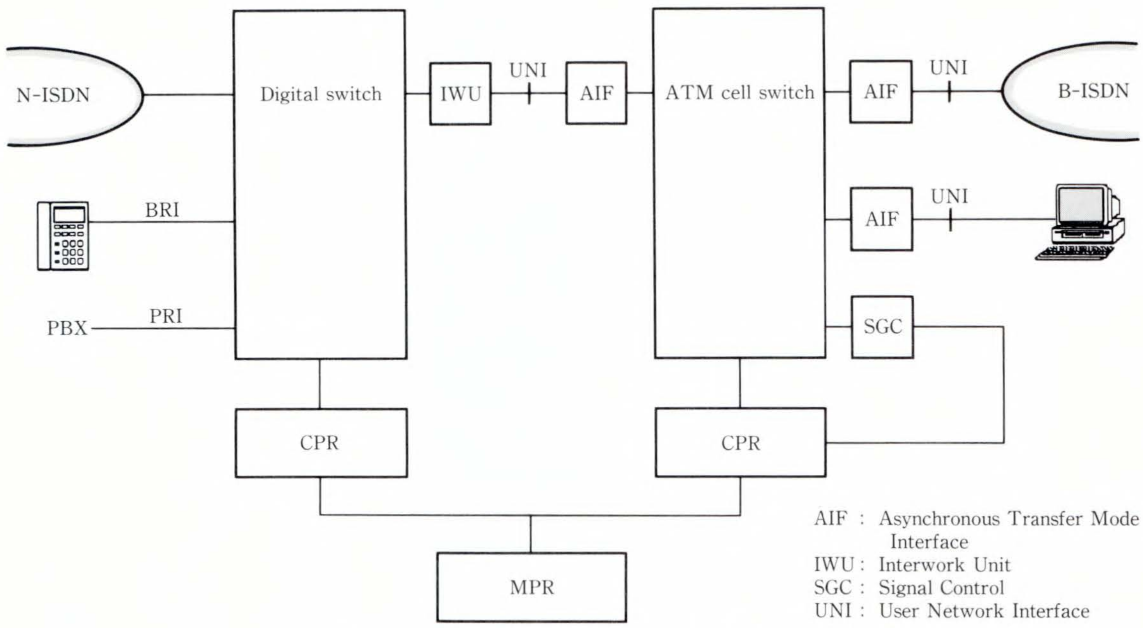



Fig. 14—Broadband ISDN configuration.

Asynchronous Transfer Mode (ATM) switch to support terminals for B-ISDN switching. All B-ISDN traffic is carried through this switch. Both transmission schemes are coupled through an interconnection facility. Although B-ISDN and N-ISDN traffic is carried through separate switch blocks, both are controlled by the current FETEX-9600 call processing architecture.

### 5.3 New node concept

Currently the PBX is a single service node or, in the case of a private network, a group of single service nodes. As network interconnection matures, it will become possible to provide the services residing in one PBX node to other PBXs or even to intelligent exchange nodes.

Rather than confining the rich features and services developed over many years on the FETEX-9600 to local users only, Fujitsu proposes to release them in such a way that they will be available to other nodes from a "service provider" node.

The service provider node contains features such as the property management system, network management system and ACD/management & information system. The service provider node could provide these services to other nodes through ISDN links carrying feature control signals and data transaction messages.

## 6. Conclusion

This paper began by reviewing the history of the development of ISDN through international bodies, then discussed the requirements for ISDN PBX, the development of the FETEX-9600/600 as an ISDN switch, and focused on future directions. To date, a large number of FETEX-9600/600 switches equipped with ISDN services have been installed and many customers are enjoying the advantages offered by the ISDN network.

Telecommunications technology is constantly evolving and growing. Fujitsu plans to develop the FETEX-9600/600 continually, and offer the latest advances and improvements in telecommunications technology.

## References


1) Ogawa, T., Koyano, T., and Fukasawa, H.: A New Digital PBX System for Integrated Office Services. *FUJITSU Sci. Tech. J.*, **23**, 2, pp. 87-100 (1987).
2) Ogawa, T., Kamioka, S., Ogawa, Y., and Koyano, T.: A New Generation PBX for Integrated Office Services. Proc. 12th Int. Switching Symp., 1987, pp. 550-556.
3) Foard, C.F., Newell, J.A., and Ryva, G.J.: Private Networks: Evolution to Value-Added Applications. Proc. 13th Int. Switching Symp, 1990, pp. 55-60.
4) British Telecommunications PLC: British Telecom-

munications Network Requirements. N-188, issue 5, December 1989.
5) ISO/IEC JTC1: Information Technology - Private Integrated Services Network - Circuit mode 64 kbit/s bearer service - Service description, functional capabilities and information flows. SC6N6869, July 1991.
6) Telecom Australia: Australian ISDN Private Network Protocol TPH2256. 1st ed., Melbourne, 1988, 796p.
7) Aritaka, N., Katsuyama, T., Osawa, J., and Tachieda, H.: Intelligent Networking and Services in the Business Communications Environment. Proc. 13th Int. Switching Symp., 1990, pp. 147-152.

**Jun Shimada**

Systems Development Dept.
Business Switching Systems Div.
FUJITSU LIMITED
Bachelor of Electrical Eng.
Kyoto University 1974
Master of Electrical Eng.
Kyoto University 1976
Specializing in Switching System

**Toshihito Yamashita**

Software Development Dept.
Business Switching Systems Div.
FUJITSU LIMITED
FUJITSU Technical College 1981
Specializing in Switching Systems Software Design

**Toshiyuki Sato**

Hardware Development Dept.
Business Switching Systems Div.
FUJITSU LIMITED
Bachelor of Electrical Eng.
Science University of Tokyo 1984
Master of Electrical Eng.
Science University of Tokyo 1986
Specializing in Switching Systems Hardware Design

UDC 334.722:621.395.74

# Local Area Network Systems

● Satoshi Nojima ● Takashi Matsuda ● Takashi Nakamura

*(Manuscript received December 5, 1991)*

This paper describes the development of Fujitsu's LAN systems. Based on the premises of connecting the products of multiple vendors and multimedia communication, Fujitsu has developed a variety of LAN systems, some based on the standard system and some based on Fujitsu's own systems. The main feature of Fujitsu's activities is the development of a total system, covering transmission equipment, LAN interconnection systems, and maintenance and administration systems. This paper also describes some research and development activities directed towards the next generation of LAN systems.

## 1. Introduction

A network which provides communication between devices such as data processing terminals and computers in a geographically limited area is called a Local Area Network (LAN). In the early days, the major objective of the LAN system was to create a resource-sharing system which provided easy wiring between the terminals. This feature is based on two basic ideas, a broadband and broadcast transmission medium and a distributed communication control scheme. When the LAN was introduced, it became popular very quickly and became used in laboratories, factories, and offices. The rapid expansion and growth in number of LANs has been such that nowadays LANs have a basic role in virtually all communication systems.

This paper describes the approach to the development of Fujitsu's LAN and the elementary technology of LANs. This paper also discusses the state-of-the-art research and development for the next generation of LAN systems.

## 2. Fujitsu's LAN

### 2.1 Fujitsu's approach

As the LAN is widely used, the requirements for LANs are diverse. According to the demand, Fujitsu has developed various types of LAN systems over the past decade[1]. This section discusses Fujitsu's approach to development and surveys its product line-up.

Among the various requirements of LANs, the following are especially important.

1) Various types of LANs

In today's computer network environment, end systems consist of various terminals ranging from a host computer to personal terminals. This means that the required communication speed varies from 1 Mb/s to over 100 Mb/s and that the required types of communication, such as interactive and/or bulky data transmission, are also diverse. In response to these requirements, various types of LANs have been introduced and connected to each other, creating a hierarchical LAN network incorporating a variety of transmission speeds and functions.

2) Multi-vendor configuration

As the usual installation process is conducted independently and in a step-by-step fashion, the system may include various types of terminals according to the applications or the environment. The LAN system must therefore support a configuration consisting of products of multiple vendors, which means standardization of the LAN interface is an important issue. When developing LAN system products, the system should follow the standard in every aspect, such as layer protocols and management functions.

3) Multimedia wiring

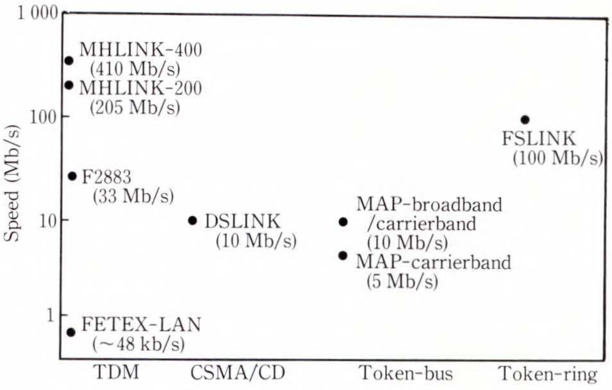



Fig. 1—Fujitsu's LAN products.

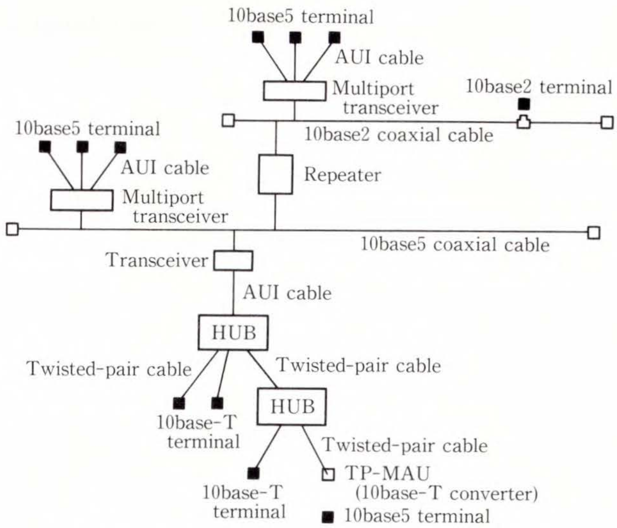



Fig. 2—DSLINK physical configuration.

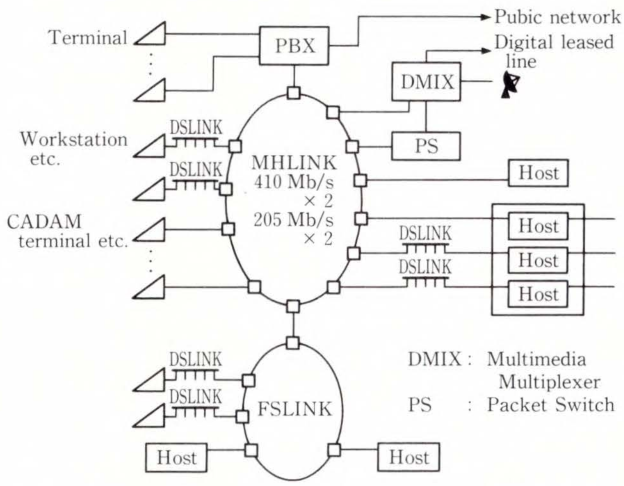



Fig. 3—Fujitsu's main factory (Kawasaki) LAN system.

Optimized dedicated systems, such as various LANs for computers, including personal computers and workstations, PBX networks and ITV systems are now in widespread use. Their scale and functions are growing and becoming more complicated. The multimedia LAN, which integrates various kinds of media and provides unified operations, is popular because of its cost-effectiveness and minimal cable requirements.

In response to the above trends, Fujitsu's approach is to develop and offer systems that:

1) Satisfy the various user requirements,
2) Conform to international standards,
3) Have a hierarchical configuration and the ability to be interconnected, and
4) Develop and offer Fujitsu's own systems according to the user requirements, such as multimedia LAN.

## 2.2 Fujitsu's products

Figure 1 shows the line-up of the Fujitsu LAN products. In the basic system, there are two products: DSLINK, which conforms to the Carrier Sense Multiple Access/Collision Detection standard (CMSA/CD) and FSLINK, which conforms to the Fiber Distributed Data Interface standard (FDDI). Fujitsu's own system is MHLINK, a multimedia LAN. Fujitsu also has a Media Access Control (MAC) bridge system (LLU), and a brouter system (LLU-E). There is also a LAN network management system, LAN Monitoring Processor (LAMP), which provides network management, and administration and maintenance functions.

DSLINK is a basic LAN system used for a branch LAN or small-site networking, and conforms to the standards IEEE802.3 and IS08002-3 for medium-speed LANs (10 Mb/s). It has a bus or tree topology, and three types of physical layer configurations, 10base5, 10base2 and 10base-T. Figure 2 shows a typical configuration. FSLINK is a high-speed token-ring LAN (100 Mb/s), which conforms to the standards ANSI X3T9.5 and IS09314 FDDI. Last of all is MHLINK, Fujitsu's own multimedia integrated LAN which can accommodate voice, image, and data communication.

## 2.3 System configuration

Using these elementary systems, the user can construct a hierarchical communication environment. Figure 3 shows a typical system configuration, that of the LAN system in Fujitsu's Kawasaki factory. There are about 3 500 system designers and hardware/software engineers in this building. It was necessary to install section LAN systems on each floor for networking workstations and personal computers of various applications, such as designing tools. To establish a floor communication environment, DSLINK and FSLINK systems were installed on each floor. To provide connection between the LANs on each floor, MHLINK systems are used as a high-speed information highway in the building. In a hierarchical configuration like this, there are various technologies. In the following section, some of the major technologies are discussed in more detail.

## 3. Technologies

### 3.1 Interconnecting devices

In designing its LAN-LAN connection devices LLU (MAC bridge) and LLU-E (brouter), Fujitsu took into account the following requirements:

1) Accommodation of various kinds of LANs,
2) The ability to handle various kinds of protocols, and
3) High filtering and forwarding throughput.

For the requirements of various LAN interfaces, each LAN interface is designed as an individual unit and the system is composed of these units. For multiple protocols and high throughput, the system is designed in a multiprocessor configuration. In the architecture, each LAN interface is controlled by a dedicated high-speed processor, the MAC bridge function is controlled by another high-speed processor, and the network level routing function is controlled by a general-purpose microprocessor.

The hardware configuration is shown in Fig. 4. The Line Control Processors (LCPs), which control the LAN interface, are connected to a system bus, and controlled by a Main Control Processor (MCP) through an IPU bus and a common memory. In a router module, a 32-bit microprocessor handles the routine functions.

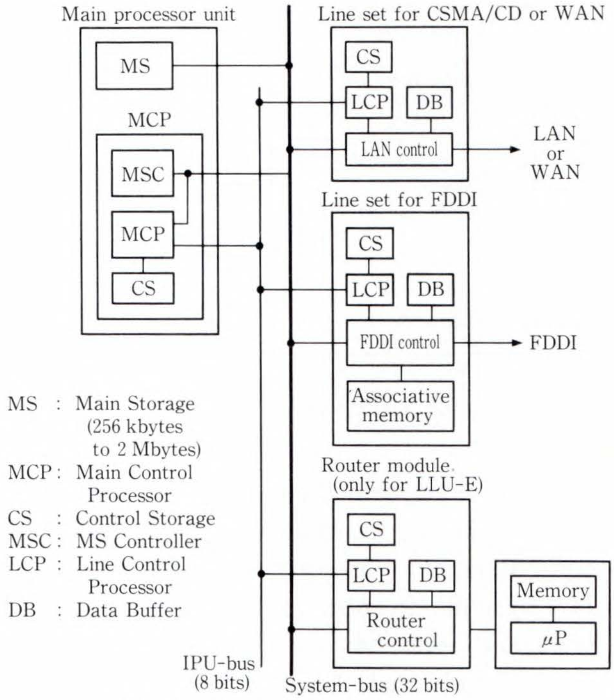



Fig. 4–LLU/LLU-E architecture.

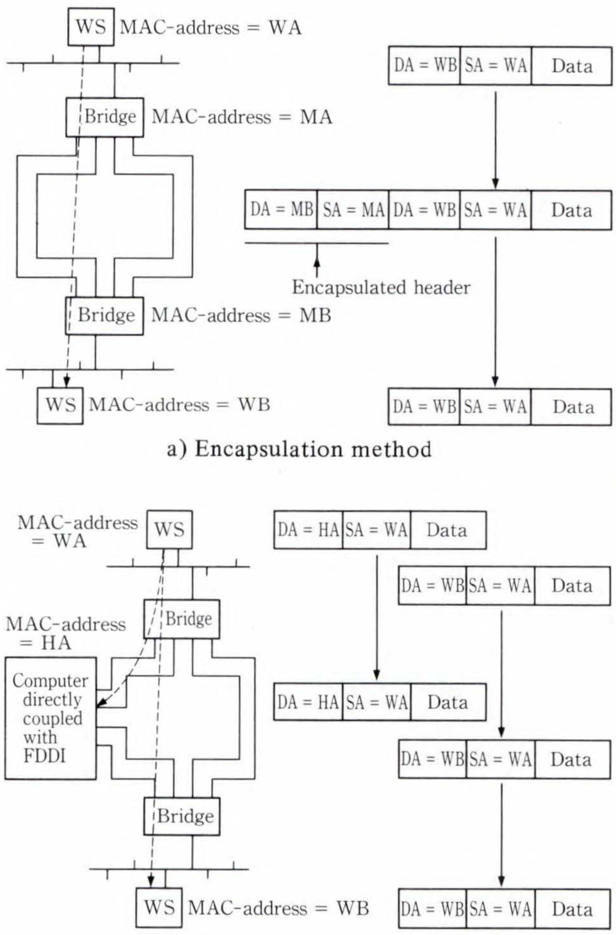



Fig. 5–FDDI bridging methods.

### 3.2 The basic functions of FDDI

Fujitsu's FDDI product, FSLINK, is designed to accommodate both branch LANs, such as DSLINK, and high-speed computing devices. The basic function of FSLINK is to provide transparent communication for branch LANs, while the station provides the MAC bridge function between FSLINK and DSLINK. However, DSLINK and FSLINK have different frame formats and different types of media access control. If only the basic function is required, the encapsulation method shown in Fig. 5a) simplifies designing the MAC bridge function. This is because the MAC bridge only has to filter and strip frames according to a single MAC address.

However, FSLINK has to connect high-speed computing devices directly. In this case, the MAC header of LAN frames cannot be changed or added. To satisfy these contradictory requirements, Fujitsu adopted transparent frame format bridging as shown in Fig. 5b). The architecture of the station is same as LLU and LLU-E mentioned above, shown in Fig. 4. Under the control of a Main Control Processor (MCP), the FDDI line set accommodates an FDDI ring and provides the FDDI ring access and management functions.

Under this method, the FDDI station has to filter every frame transmitted on the FDDI ring and to strip the frames it originates itself. To improve the capacity of the bridge function between FSLINK and DSLINK, it is important to reduce the number of frames to be processed. For these purposes, Fujitsu developed the distributed MAC bridge function. MCP provides the bridge function between FDDI and the branch LANs, and a local filtering function is provided in an FDDI line-set using an associative memory as a MAC address learning table.

The functions of local filtering are as follows:

1) When transmitting frames to the FDDI ring, the source address is learned and stored in a learning table, table A.
2) When receiving a frame from the FDDI ring, the source address part of the frame is stored in another learning table, table B, and is compared with those registered in learning table A. If they match, the frame is stripped from the ring because the frame was originated by the station itself.
3) If they don't match, the frame is from another station, and the destination address part is compared to table B. If they match, the frame is discarded without MCP bridging, because the frame is communicated between another two stations on the FDDI ring.
4) If they don't match, the frame needs to be transmitted to the branch LANs, and the LCP reports the fact to the MCP. The MCP then performs MAC bridging. That is, MCP learns the source MAC address of the received frame, and forwards it to another line set according to its destination MAC address.

This process reduces the number of frames processed by the MCP and enables the processing capacity of the MCP to be used efficiently.

### 3.3 Multimedia LAN[2)]

In designing a multimedia LAN, to create an information highway according different types of communication in the FA and OA environment, there are various contradictory requirements. First, a multimedia LAN must have sufficient capacity for various media, such as LAN interconnection, voice communication, and video communication. Second, it must accommodate continuous and instantaneously large volumes of traffic. Finally, since the multimedia LAN is a key part of the infrastructure of the total communication system, high RAS capabilities (reliability, availability, and serviceability) should be provided enabling non-stop operation.

#### 3.3.1 Network capacity

The requirement network capacity greatly depends on the size of the area covered by the LAN. Fujitsu estimates the required capacity of each service to be supported by the LAN as follows:

1) Interconnection of branch LANs:
   100-200 Mb/s (considering FDDI)
2) Voice services: 50-100 Mb/s
   (assuming 1 000-2 000 telephone lines)
3) Video services: 50-100 Mb/s

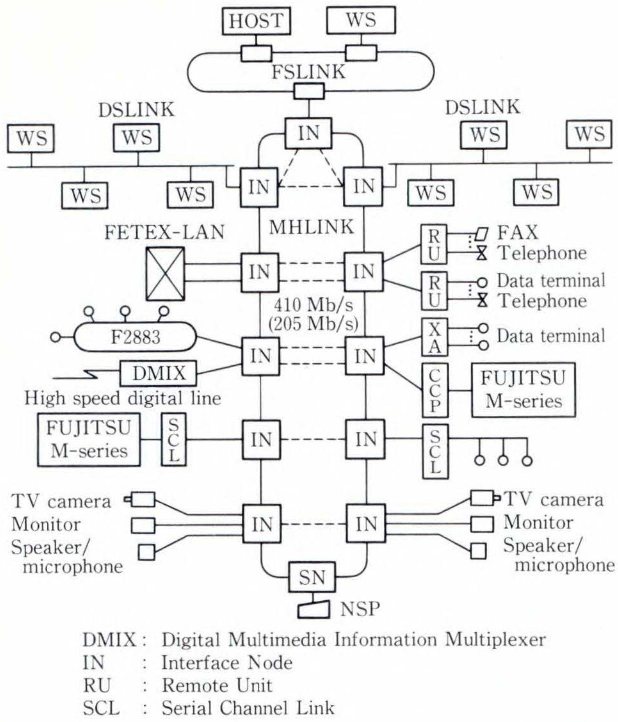



Fig. 6—MHLINK system configuration.

Table 1. MHLINK system specification

| Item | | MHLINK-200 | MHLINK-400 |
|---|---|---|---|
| Network topology | | Duplicated ring | |
| Transmission speed | | 205 Mb/s | 410 Mb/s |
| Components | | SN, IN, XA, and NSP | |
| Media access | | Slotted ring | |
| Communication | | 1:1, $n$:$n$, broadcast | |
| Number of INs | | Up to 64 | |
| Node span | | Up to 10 km | |
| Interface menu | IN | ISO8802-3 (CSMA/CD)<br>ISO9314 (FDDI-I)<br>Digital leased line (up to 6.3 Mb/s)<br>Multiplexed interfaces (2 Mb/s, 8 Mb/s)<br>NTSC (video) | |
| (up to 8XA/IN) | XA | V.24/28, V.35, RS449, X.21, I.430 | |
| RAS | | Centralized network management by NSP<br>Automatic ring reconfiguration (ring switching, loopback, node-bypass)<br>Duplexed SV | |

(assuming a few lines of compressed video) Fujitsu has concluded the LAN has to incorporate 205-Mb/s and 410-Mb/s series machines.

### 3.3.2 System configuration

Figure 6 shows the MHLINK system configuration. The system specifications are listed in Table 1. The system consists of a Supervisory Node (SN), Interface Nodes (IN), Extension Adaptors (XA) which multiplex low-speed lines, a Network Service Processor (NSP), and optical fiber cables. The SN has the central supervisory and control functions, including slot generation for data transmission. The IN accommodates XAs, terminals, communication equipment, and branch LANs. The NSP is connected to the SN and provides a human/machine interface for centralized system monitoring and management.

### 3.3.3 Key technologies

1) Access method

MHLINK is designed to accommodate communication speeds from DSO (64 kb/s) to DS3 (45 Mb/s) and to support the communication topologies of point to point, point to multipoint and multi-point to multi-point. A major technological problem is how to handle these variations efficiently. Fujitsu adopted a slotted-ring access method and a fixed-length mini-packet scheme.

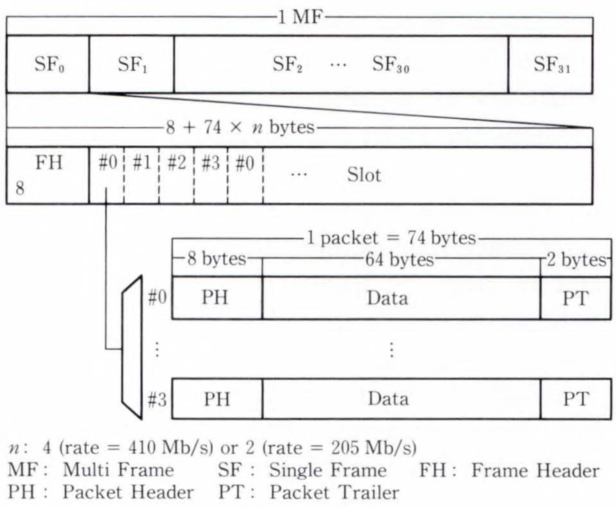



Fig. 7—MHLINK frame format.

The frame (data stream) shown in Fig. 7 circulates on a ring. A frame is divided into 4 or 2 slots according to the transmission speed. The slot is for carrying a mini-packet, which has a 64-byte payload area. Every information flow is formatted as a mini-packet and communication is made between terminals by obtaining the slot and sending the mini-packet. This scheme

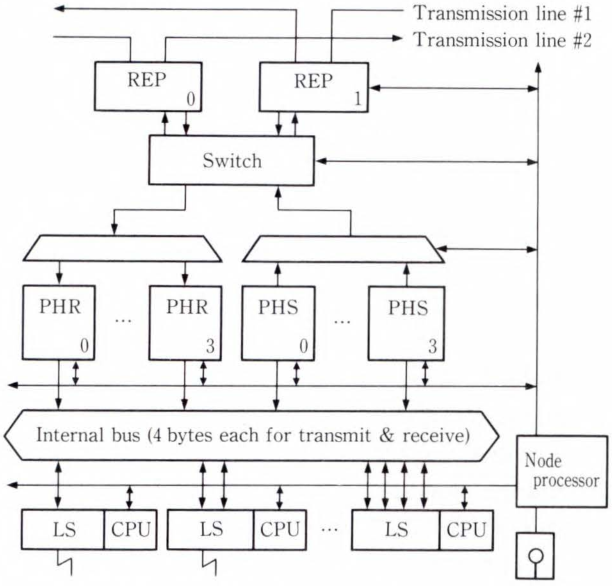


Fig. 8–MHLINK node configuration.

flexibly accommodates various traffic loads, however it requires both the slotted ring access function and the mini-packet assembly and disassembly function. In particular, when the ring transmission speed is 400 Mb/s, the node requires a packet processing capability of as high as 0.7 million packets per second. To satisfy these requirements, it is necessary to create a high-performance, low power-consuming, and economical system.

To meet these requirements, Fujitsu uses parallel processing. The 400-Mb/s bit stream is divided into 100-Mb/s streams and each is processed in parallel. This enables using high-density, low-power C-MOS gate array LSI technology. The system is shown in Fig. 8. The Transmit Packet Handler (PHS) and the Receive Packet Handler (PHR) can handle datastreams of 100 Mb/s (170 kilo-packets per second). The handlers use a C-MOS custom-designed LSI. Using four sets of PHS/PHR, it is possible to process 400 Mbits of data per second. Although this system needs four times the number of LSI gates compared to ECL technology, CMOS technology has marked advantages in power consumption and integration. In addition, 400-Mb/s systems and 200-Mb/s systems are provided with a unified architecture, and the power consumption is reduced.

2) RAS function

Redundant configurations and centralized network management are the key points of the RAS function. The main components of the system such as fiber optic transmission lines and their repeaters in the nodes are duplicated. Total control of the redundant configuration is managed by a Supervisory Block (SV) in the SN. If a fault occurs, the faulty components are automatically isolated and the standby system is activated. Fujitsu has also developed a dedicated Network Management Processor (NSP) which offers a Human Machine Interface (HMI) for monitoring the network operation status, logging failures, and providing traffic statistics for centralized network management. The details of network management are discussed in the next section.

**3.3.4 Communication feature**

This subsection describes how to integrate continuous and instantaneously large volumes of traffic. As MHLINK is based on a slotted ring, it can easily handle a continuous communication bandwidth by reserving a multiple slot sequence. However, to handle instantaneously large volumes of traffic, as with a LAN interconnection, MHLINK has to provide connectionless communication between a lot of nodes. The destination of the transmitted information is changed dynamically, and the required communication speed has to be the same as the end system LAN such as CSMA/CD or FDDI. Therefore, the major problem is how to provide a connectionless communication capability and efficient use of the basic LAN.

In MHLINK, Fujitsu adopted the token-ring access method for interconnection of end system LANs. The token frame runs around the ring using a multi-frame header. The MHLINK node (IN) which has LAN frames to be transmitted, captures the token and sends the frame using a sequence of empty slots. When the frame has been sent, the IN releases a token. In this way, MHLINK establishes a basic LAN like FDDI LAN. ISO 8802-3 (CSMA/CD) LANs and ISO 9314 (FDDI) LANs can be connected to this LAN. Each IN has a LAN bridge function, and the MAC bridge with an automatic learning process is used for efficient interconnection. This reduces the total traffic through the MHLINK system.

### 3.4 Network management

#### 3.4.1 Requirements

The LAN environment consists of multiple protocols and products made by multiple vendors. When making available a LAN system, the top consideration should be the network management system. A characteristic feature of LANs is that the transmission medium and inter-LAN connecting devices are shared by many computers. This means that a failure in a transmission medium or an inter-LAN connecting device may disable communication to all computers, and cause major damage to the whole computer system. Therefore, it is very important to offer the facilities such as localization of the failure, quick recovery using a duplicated configuration or some reconfiguration function, and preventive maintenance to minimize the chance of failure.

As the computer can be very easily connected to a LAN, and is highly portable, the computer can also be a cause of problems. Broadcast storming or a meltdown can occur if a user fails to set the addresses, which causes the traffic to become enormous, and disables communication between the other computers. Therefore, it is also important to offer facilities such as monitoring the addition and movement of computers, especially address management, and constant monitoring of the traffic load. To summarize the above, network management in the LAN environment must provide the following:

1) Accommodation of multiple protocols and the products of multiple vendors
2) Fault management of the transmission medium and inter-LAN connecting devices (localization and quick recovery)
3) Preventive maintenance (logging of statistics, and so on)
4) Monitoring of the addition and movement of computers (address management)
5) Monitoring of the traffic load.

#### 3.4.2 Fujitsu's action

Up until now, two types of network management equipment are available, the LAN Monitoring Processor (LAMP) for standard LANs, and the Network Service Processor (NSP) for multimedia LANs. This equipment offers various management functions independent of the higher communication protocols. In addition, LAMP supports Station Management (SMT) commands specified in the FDDI international standard, which enable LAMP to manage the equipment of multiple vendors.

The three major functional categories of this network management equipment, and its main functions are described below.

1) Configuration Management

Graphically displays the network configuration and operational status.

2) Fault Management

Displays the error messages, location of the error, logging, and diagnostics.

3) Performance Management

Displays and statistically processes the traffic load in the communications lines.

In the near future, enhanced LAN management functions such as computer management using address information are planned to be introduced. The integration of LAMP and NSP based on the Simple Network Management Protocol (SNMP) is also planned to be introduced, which will provide users with a unique user interface to manage the whole LAN system.

## 4. Preparing for the future

Based on the technologies accumulated through developing various types on LAN systems, Fujitsu is hoping to provide better LAN systems in the future. In developing future LAN systems, the emphasis will be on improved capacity and flexibility, wide area interconnection, and personal communication. This chapter introduces some research activities concerning these points of emphasis.

### 4.1 Improved capacity and flexibility[3)]

This is a constant theme for LAN. Especially now, when bulky information such as image data is becoming popular, technology able to

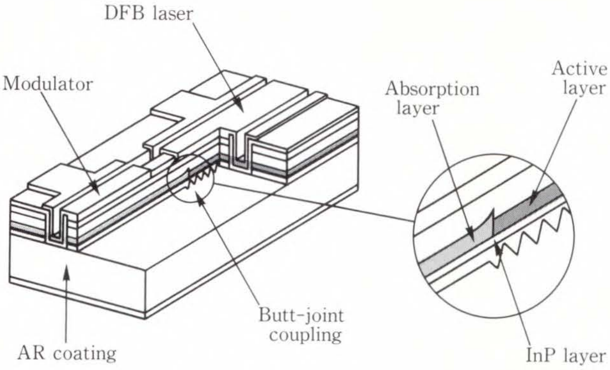


Fig. 9–Monolithic electro-absorption modulator.

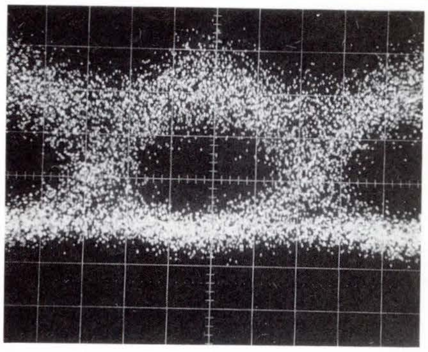


Fig. 10–Eye opening at 10 Gb/s.

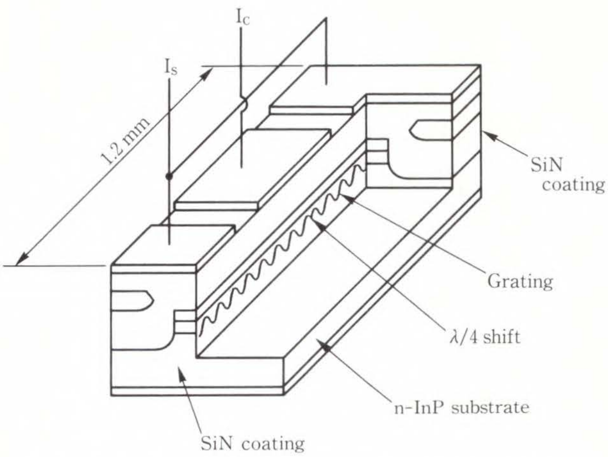


Fig. 11–Wavelength tunable laser diode structure.

handle the data volume is in heavy demand. The following two examples are technologies recently developed in the Fujitsu Laboratories.

The first example is a straightforward way of increasing the capacity, the multi-gigabit optical signal modulation technology[4]. Figure 9 shows the device structure of the monolithic electro-absorption modulator. The continuous light output by the Distributed Feedback (DFB) laser diode is either absorbed or transparently fed through the modulator part according to the signal voltage input to the modulator electrode. This device can transmit 10 Gbits of data per second over 65 km. Figure 10 shows the eye opening.

The second example shows use of the optical wavelength[5]. The transmission capacity can be increased by wavelength multiplex technology. In addition, this can enable a switching function using the wavelength as address information, providing more flexibility in the construction of a LAN system. Figure 11 shows the wavelength can be changed by a bias voltage.

### 4.2 Wide area interconnection

LAN interconnection using public networks is becoming more and more important. As Broadband ISDN (B-ISDN) will be used in future, LAN interconnection will become easier because a high transmission capacity equivalent to a high-speed LAN will be provided for each customer. It is expected that the first useful application of B-ISDN will be LAN interconnection, and the technology to support this, such as Switched Multi-megabit Data Service (SMDS), is being studied by the standardization authorities. This is an attempt to find a solution by connecting currently existing LANs to B-ISDN.

For an efficient interconnection, however, there is another approach, by designing a new LAN to have a certain degree of compatibility with B-ISDN. In line with this approach, Fujitsu has developed an experimental LAN named Shuttle Bus[6]. Figure 12 shows the system configuration. It employs a 1.8-Gb/s optical loop and the same frame format as the B-ISDN Synchronous Transport Module (STM) 12. Eight STM-1s are assigned to carry cells whose format is the same as the Asynchronous Transfer Mode (ATM), and the remaining four STM-1s are reserved for the synchronous transfer mode. An ATM switching system is connected and handles ATM cells. Using the ATM switching system as a gate, the Shuttle Bus is easily connected to an ATM-based B-ISDN subscriber loop.

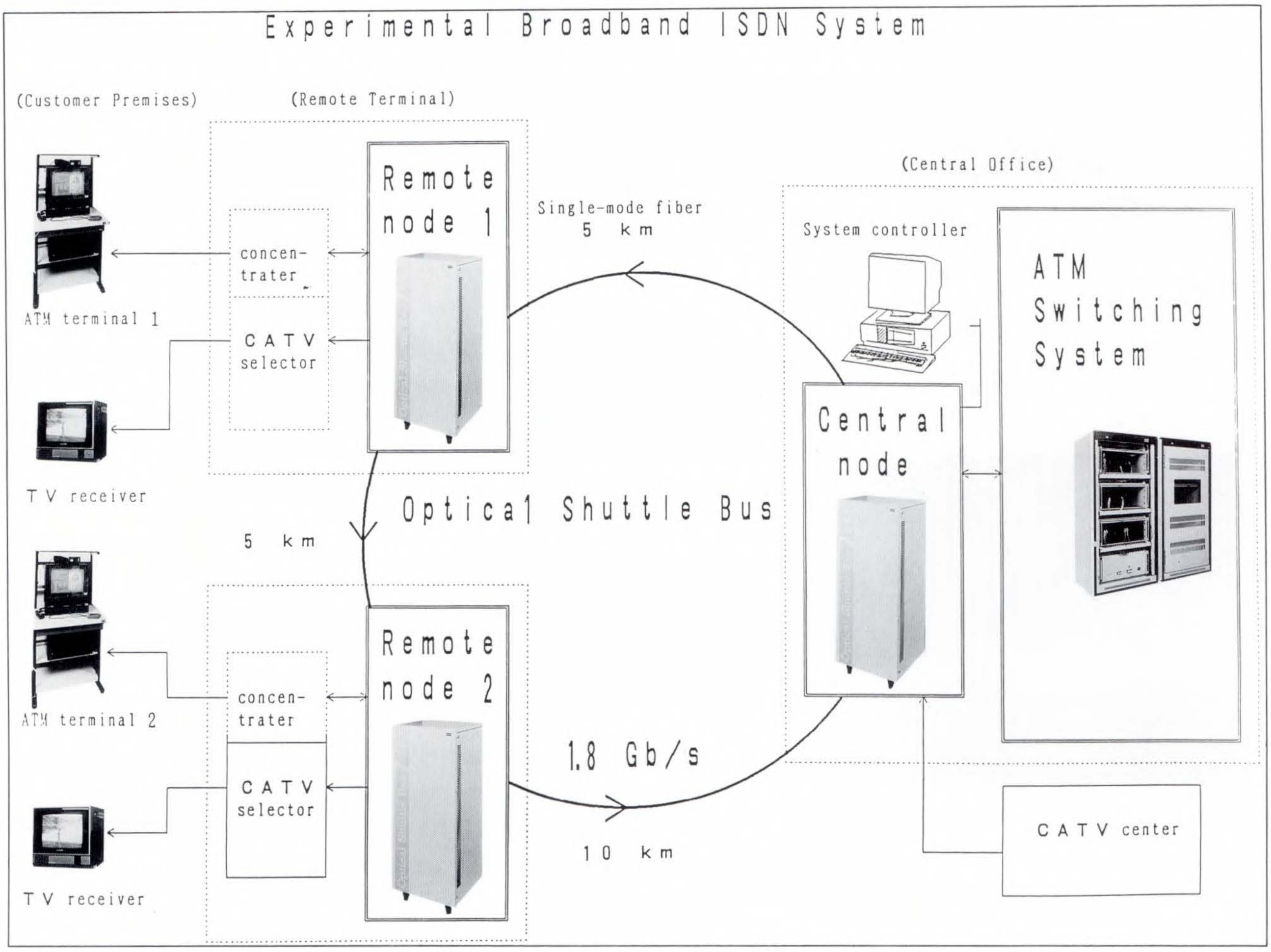



Fig. 12–Shuttle bus configuration.

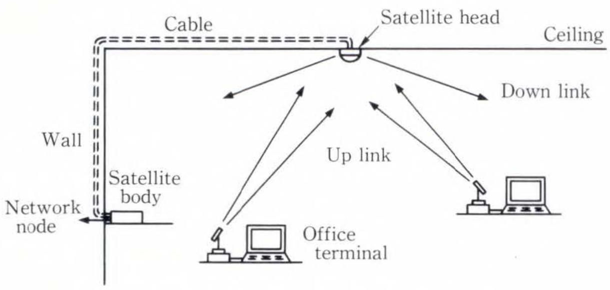



Fig. 13–Optical wireless LAN configuration.

### 4.3 Personal communication

Personal communication is what provides users with mobility. A wireless LAN is a typical example. As a related technology, the optical wireless LAN Fujitsu developed in the past[7] is described below. This system provides a wireless link, which is one of the key elements of the personal communication. Figure 13 shows the system configuration of Fujitsu's optical wireless LAN for intra office application. Data is transmitted using space propagation of an optical signal. Each terminal can establish a communication link from anywhere in the office via a satellite head on the ceiling. Because of its low transmission capacity, this systems is not the basis of Fujitsu's current wireless LAN study, but the experience is useful for the overall study of the wireless LAN.

Fujitsu is conducting an intensive study of the wireless LAN with the Japanese standardization authorities, and will report the results in the near future.

## 5. Conclusion

This paper describes Fujitsu's approach to the LAN environment, the product line-up and the elementary technologies. State-of-the-art approaches to future technologies are also described. In the future, there will be various broadband communication technologies, and data communication will grow significantly. This means that data communication traffic will become an important part of distributed computing and processing. To ensure that the LAN system is a key component, Fujitsu is developing a full range of products which support interconnection.

## References


1) Special Issue: Communications: *FUJITSU Sci. Tech. J.*, **22**, 4, pp. 211-388 (1986).
2) Matsuda, T. et al.: A 400-MBPS MULTIMEDIA LAN. Proc. Int. Telecommun. Symp., 1990, pp. 11.3.1-11.3.5.
3) Horimatsu, T. and Clendening S. J.: Device Technologies for the Next Generation of Multigigabit Optical Transmission Systems, Proc. MCI Technol. Symp., 1990.
4) Furutsu, M. et al.: Monolithic Electro-absorption Modulator/DFB Laser Lightsource with Semi-Insulating InP Burying Layers. Ext. Abs. 21st Conf. Solid-State Devices and Materials, 1989, pp. 445-448.
5) Fujimoto, N. et al.: Photonic Cross-connection Node Architecture Utilizing Novel Multiwavelength Selective Filters. Proc. 16th ECOC, **Vol. 1**, 1990, pp. 199-202.
6) Yamashita, H. et al.: Flexible Synchronous Broadband Subscriber Loop System: Optical Shuttle Bus. *J. Lightwave Technol.*, **7**, 9, pp. 1788-1987 (1989).
7) Minami, T. et al.: Optical Wireless Modem for Office Communication. Proc. NCC'83, pp. 721-728.



**Satoshi Nojima**

Integrated Communication Network Laboratory
Communication Space Div.
FUJITSU LABORATORIES, KAWASAKI
Bachelor of Electronics Eng.
Waseda University 1976
Master of Electronics Eng.
Waseda University 1978
Specializing in Research and Development of Local Communication Networks



**Takashi Matsuda**

Business Communication Dept.
Business Communication Div.
FUJITSU LIMITED
Bachelor of Electrical Eng.
Tohoku University 1973
Specializing in Development of Communication Equipment



**Takashi Nakamura**

Computer Systems Architecture Dept.
Main Frame Div.
FUJITSU LIMITED
Bachelor of Electrical Eng.
Tohoku University 1971
Specializing in Development of LAN Systems

UDC: 621.391.6

# Fiber Optic Transmission

● Hideo Kuwahara ● Akira Miyauchi ● Akira Mitsuhashi

*(Manuscript received April 2, 1992)*

The development of fiber optic transmission systems by Fujitsu is reviewed. The equipment developed for NTT's F-1.6G system, FTM-2.4G system and 1.8 Gb/s 1.55 $\mu$m long span system is first described with the main system parameters. Then research and development activities are introduced, including evolutionally developed optical amplification, ultra-high speed optical transmission towards 10 Gb/s, and coherent lightwave transmission, with a description of their related components. Optical device technologies such as erbium firber doping, wavelength division multiplexing combining the optical signal and pumping power, and $LiNbO_3$ external modulation are also described.

## 1. Introduction

Fiber optic transmission systems have been widely introduced all over the world because of their superior characteristics such as wide bandwidth, low transmission loss, and light weight, compared to conventional coaxial cable systems.

In order to reduce transmission costs, we have been developing technologies to increase transmission speed and to lengthen repeater spacing. We have developed increased transmission speed equipment for NTT's F-32M, F-100M, F-400M, and F-1.6G optical transmission systems[1),2)].

The 1.55 $\mu$m range is very attractive, because the fiber has the minimum transmission loss in this range of wavelength. Another limiting factor is dispersion in the fiber, that is, the different propagation velocities of different wavelengths which degrades long distance transmission characteristics. In Japan, NTT adopted Dispersion Shifted Fiber (DSF) for this reason. Dispersion in the DSF fiber is negligible, making 1.55 $\mu$m equipment easy to develop. However, conventional 1.3 $\mu$m zero-dispersion fiber is widely installed all over the world. It is desired to transmit 1.55 $\mu$m optical signals over this conventional 1.3 $\mu$m zero-dispersion fiber, but dispersion again becomes the main factor which limits repeater spacing. In spite of this problem we successfully developed 405M, 810M, and 1.8G equipment, whose repeater spacing is 90 km with conventional 1.3 $\mu$m zero-dispersion fiber. To increase the repeater spacing even more, we are developing new technologies such as external modulators, optical amplifiers, and coherent transmission systems.

From the view point of systems there is a dramatic change from asynchronous systems like NTT's F-1.6G optical transmission system to synchronous systems. This synchronous system has a CCITT-recommended standard interface all over the world. We successfully developed equipment for NTT's FTM-600M and FTM-2.4G system as a synchronous digital hierarchy system[3),4)].

Chapter 2 of this paper summarizes the successful development of equipment for NTT's F-1.6G and FTM-2.4G systems, and for the 1.8 Gb/s 1.55 $\mu$m system. Chapter 3 reviews recent research and development activities. Chapter 4 describes related optical components.

## 2. Developed technologies

This chapter describes F-1.6G optical transmission equipment as high-speed transmission equipment, 1.8 Gb/s 1.55 $\mu$m optical transmission equipment as long repeater spacing equipment with a conventional fiber, and FTM-2.4G optical transmission equipment as equipment for a synchronous system.

### 2.1 F-1.6G optical transmission equipment

Table 1 shows the main parameters of the equipment. The F-1.6G was the first equipment whose transmission speed exceeded 1 Gb/s. In order to obtain stable high-speed operation and to realize high productivity, we developed high-speed optical devices and ICs, and packaging technologies for them[5].

Table 1. Main parameters of F-1.6G equipment

| Item | | Parameters |
|---|---|---|
| Channel capacity | | Phone: 23 040 ch/system<br>32 MTV: 48 ch/system |
| Bit rate | Intra-office | 397.200 Mb/s × 4 |
| | Line | 1 820.900 Mb/s |
| Transmission codes | | 10B1C |
| Cable | | Single-mode fiber cable (1.3 μm zero-dispersion) |
| Wavelength | | 1.31 μm |
| Optical source | | FBH DFB LD |
| Optical detector | | GaInAs APD |
| Output power | | −4 dBm (standard)/ 0 dBm (option) |
| Minimum detectable power ($P_e = 10^{-11}$) | | −30 dBm (standard)/ −30.5 dBm (option) |
| Repeater spacing | | 30 km (standard)/40 km (option) |
| Error rate | | $<10^{-8}$/2 500 km, $<10^{-11}$/ repeater |
| Line supervision | | Line error rate monitoring (Parity check, and C-bit rule violation) |

Table 2. ICs for F-1.6G regenerator

| IC | Function | Characteristics |
|---|---|---|
| EQL 1 | Pre amplifier | Input equivalent noise current density: 13 pA/$\sqrt{\text{Hz}}$<br>Maximum gain: 88 dBΩ (V/A)<br>Dynamic range: 30 dB<br>Bandwidth: 1.0 GHz |
| EQL 2 | AGC amplifier, post amplifier, 2 outputs | |
| TIM 1 | Timing extraction | Maximum gain: 55 dB<br>Dynamic range: 30 dB |
| TIM 2 | Limiter amplifier | |
| TIM 3 | Limiter amplifier, R-IN down det, 2 outputs | |
| DEC | Slice amplifier, decision, CLK out | Discrimination sensitivity: 4 mVpp |
| REG | NRZ/RZ converter, LD driver | Maximum drive current: 45 mA |

#### 2.1.1 Optical devices and ICs

The higher the transmission speed becomes, the shorter the repeater spacing when using a Fabry-Perot Laser Diode (LD). We adopted a Distributed Feedback (DFB) LD to avoid degradation by mode partition noise, which limits the repeater spacing.

For high sensitivity we used a 50 μm diameter InGaAs Avalanche Photodiode (APD) instead of a Ge APD. The receiver sensitivity of InGaAs APDs is about 2 dB superior to that of Ge APDs.

We developed seven high-speed repeater ICs and two low-speed control ICs. Table 2 shows the main parameters of the ICs.

#### 2.1.2 Packaging technologies

In order to guarantee the reliability of our equipment we designed special packaging for the IC chips. Stray capacitance and lead inductance of the IC package cause degradations of the signal waveform and signal to noise ratio ($S/N$). The feature of our IC packages are:

1) Several IC chips, chip capacitors and chip resistors are included in a package in order to avoid undesirable parasitics.
2) A multi-layered structure is adopted to obtain effective isolation.
3) High-speed signals are sent over strip lines.
4) Via-holes are used to obtain stable ground and isolation.

Figure 1 shows the structure of the IC package. It has seven layers and 36 pins.

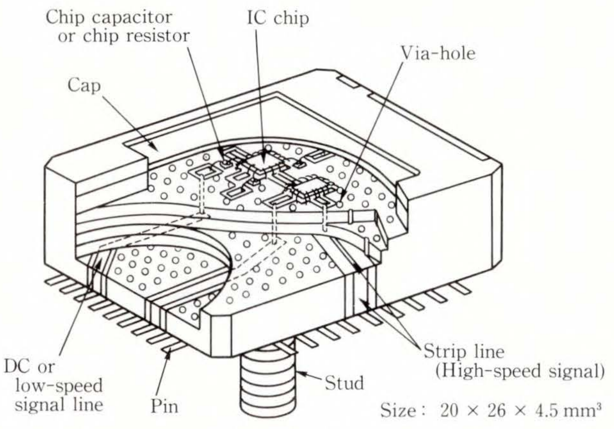



Fig. 1 – IC package for F-1.6G regenerator.

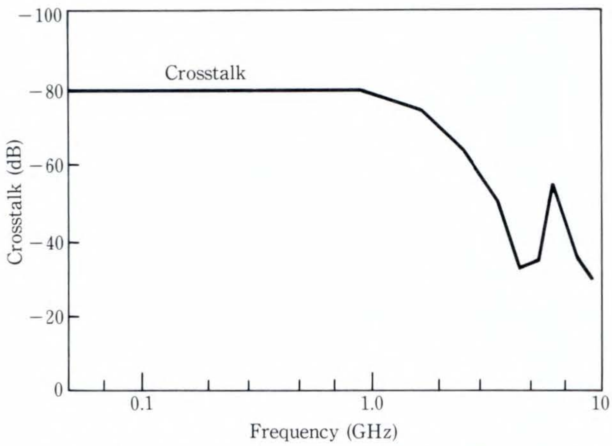


Fig. 2—Crosstalk characteristic of F-1.6G IC package.

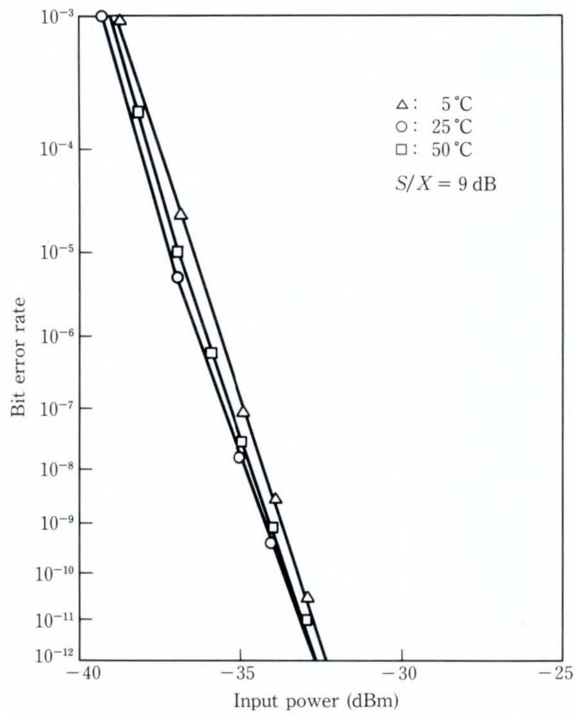


Fig. 3—Bit error rate characteristics of F-1.6G equipment.

Figure 2 shows its crosstalk characteristic. This is acceptable because the maximum gain of our IC chips is 55 dB at 1.8 GHz[6].

### 2.1.3 Performance

Figure 3 shows our measurements of the Bit Error Rate (BER) characteristics of the equipment with $S/X = 9$ dB from 0 °C to 50 °C (Ta: equipment ambient temperature). In order to guarantee performance despite aging degradation we adopted the S/X method[7]. Using the technologies mentioned above we obtained good characteristics against temperature change and aging degradation.

Table 3. Main parameters of 1.8 Gb/s 1.55 μm equipment

| Item | Parameters |
|---|---|
| Bit rate | 1 866.24 Mb/s |
| Cable | Single-mode fiber cable (1.3 μm zero-dispersion) |
| Dispersion | Max 1 630 ps/nm |
| Wavelength | 1.55 μm |
| Optical source | FBH DFB LD |
| Optical detector | InGaAs APD |
| Output power | 0 dBm |
| Minimum detectable power ($10^{-11}$) | −25.8 dBm (including transmission power penalty) |
| Repeater spacing | 90 km (fiber loss 0.25 dB/km) |

Table 4. Output power of 1.8 Gb/s 1.55 μm equipment

| Temperature | 0 °C | 25 °C | 50 °C |
|---|---|---|---|
| Output power | +2.2 dBm | +2.0 dBm | +2.0 dBm |

## 2.2 1.8 Gb/s 1.55 μm optical transmission equipment

Table 3 shows the main parameters of the equipment. This 1.8 Gb/s 1.55 μm optical transmission equipment is the first equipment whose transmission speed exceeds 1 Gb/s using 1.55 μm light through conventional 1.3 μm zero-dispersion fiber[8].

### 2.2.1 Optical devices

The chirping characteristic of the DFB LD degrades transmissions in high-dispersion fibers. In order to obtain good transmissions it is very important to reduce chirping. Optimizing the structure of the DFB LD, e.g. corrugation depth, successfully reduces chirping. In order to obtain stable single mode operation of the DFB LDs we adopted a thermo-electric cooler to stabilize the temperature of the laser.

### 2.2.2 Performance

We obtained good temperature characteristics for output power, as shown in Table 4. Deviation of the output power is only 0.2 dB between 0 °C and 50 °C (Ta).

We measured the BER characteristics of the 1.8 Gb/s 1.55 μm optical transmission equipment against temperature in the range 0 °C to 50 °C (Ta), with and without 100 km of conventional 1.3 μm zero-dispersion fiber (total dispersion = 1 630 ps/nm). Figure 4 shows the

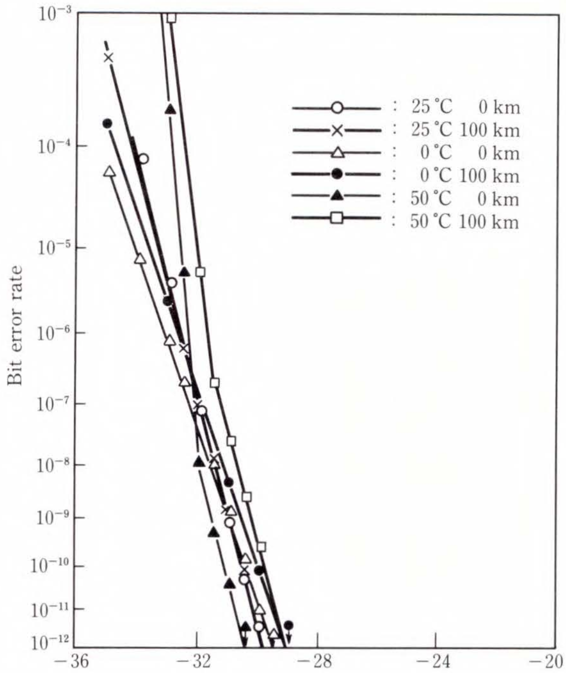



Fig. 4—Bit error rate characteristics of 1.8 Gb/s 1.55 μm equipment.

Table 5. Main parameters of 2.4 Gb/s equipment

| Item | Parameters | |
|---|---|---|
| Bit rate | 2 488.36 Mb/s | |
| Wavelength | 1.3 μm | 1.55 μm |
| Optical source | FBH DFB LD | |
| Optical detector | InGaAs APD | |
| Output power (dBm) | 0 to +6/+3 to +6 | −1 to +5/+2 to +6 |
| Minimum detectable power ($10^{-11}$) | −29 dBm | −30 dBm |
| Repeater spacing | 40 km | 80 km |

results. We obtained good minimum detectable power of −29.4 dBm.

### 2.3 FTM-2.4G optical transmission equipment

Table 5 shows the main parameters of the equipment. We developed not only 1.3 μm equipment for conventional 1.3 μm zero-dispersion fiber but also 1.55 μm equipment for dispersion shifted fiber[9].

#### 2.3.1 Optical devices and ICs

The output power of this equipment is much higher than that of conventional equipment. We used an asymmetric facet reflection type FBH-DFB LD to maintain output power at

Table 6. ICs for 2.4 Gb/s regenerator

| IC | Process | Function | Characteristics |
|---|---|---|---|
| PRE | GaAs | Pre-amplifier | Input equivalent noise current density = 5 pA/$\sqrt{\text{Hz}}$ Trans-impedance = 59 dBΩ |
| EQL | Si-ESPER | AGC amplifier Post amplifier | Gain = max 37 dB |
| TIM 1 | Si-ESPER | Timing extraction | |
| TIM 2 | Si-ESPER | Limiter amplifier R-IN DOWN detect | Gain = max 40 dB |
| TIM 3 | Si-ESPER | Limiter amplifier Phase shift | Gain = max 37 dB |
| DEC | Si-ESPER | Slice amplifier Decision CLK out | Discrimination sensitivity = max 10 mVpp |
| REG | GaAs | LD driver | Pulse current = min 60 mA |

ESPER: Emitter-base Self aligned with Polysilicon Electrodes and Resistors

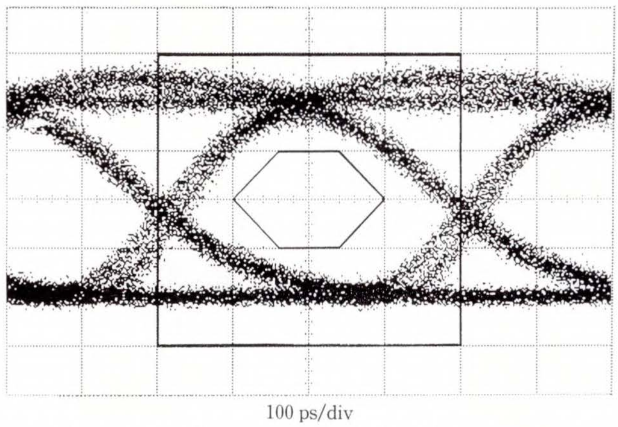



Fig. 5—Optical waveform of 2.4 Gb/s equipment. (100 ps/div)

a stable, high level. For high sensitivity we used a 30 μm-diameter InGaAs APD, which has a high M·B-product and low capacitance.

We developed seven high-speed repeater ICs[10] and two low-speed control ICs. Table 6 shows the main parameters of the ICs. To obtain superior low noise characteristics we used a GaAs preamplifier IC.

#### 2.3.2 Performance

Figure 5 shows the output waveform with the filter recommended by CCITT. Figure 6 shows the spectrum characteristics of the equipment. We measured the BER characteristics of

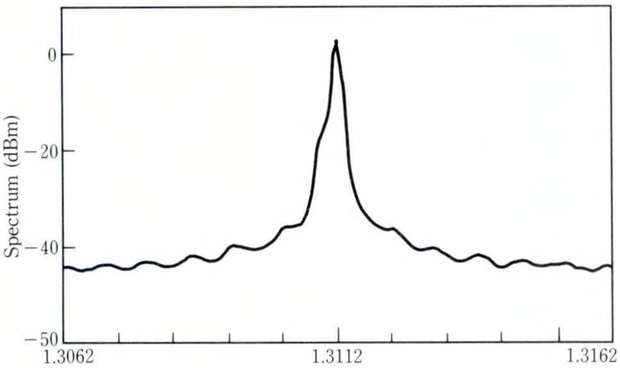


Fig. 6—Optical spectrum of 2.4 Gb/s equipment.

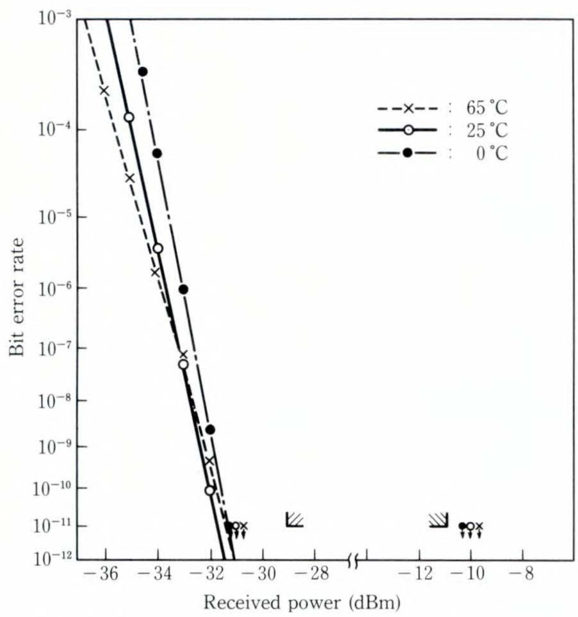


Fig. 7—Bit error rate characteristics of 2.4 Gb/s equipment.

the equipment in the temperature range 0 °C to 65 °C (equipment case temperature). Figure 7 shows our results[11),12)].

## 3. Recent developments in optical transmission

This chapter outlines three of the technologies which have been part of recent research and development towards longer spacing and larger capacity. They are, first, optical amplification which boosts the optical signal directly. Second, ultra-high speed optical transmission which increases the modulation speed to higher bitrates. And third, coherent lightwave transmission which exploits a wide optical frequency spectrum.

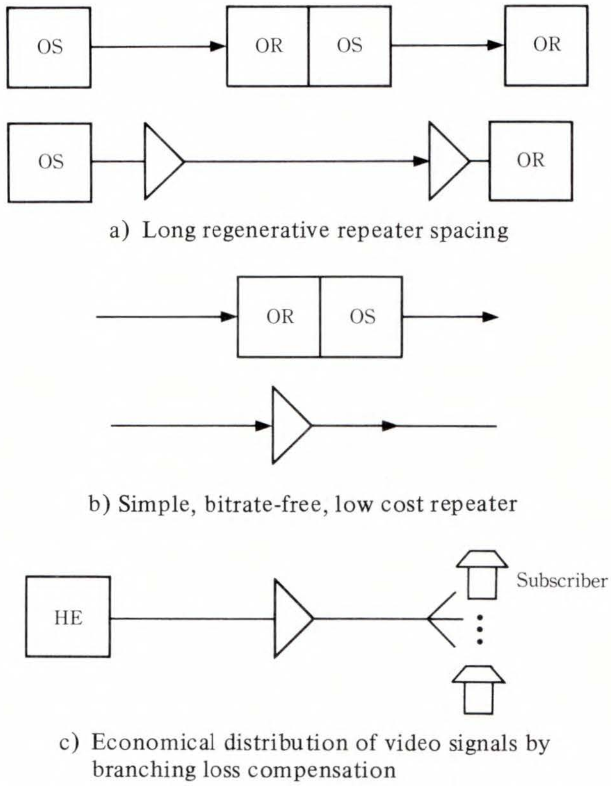


Fig. 8—Impact and application range of optical amplifiers.

### 3.1 Optical amplification

#### 3.1.1 Features and application range

Conventional optical repeaters change the received light signal into an electric signal, amplify or regenerate this using high speed electronics, then reconvert it into an optical signal. Optical amplifiers can boost the optical signal directly, and have these advantages:

1) independence of bitrate, modulation format;
2) common amplification of frequency or wavelength multiplexed signals; and
3) simplified configuration of the optical repeater.

Optical amplifiers have a wide application range as shown in Fig. 8. An optical power amplifier used after the optical transmitter increases the optical power. An optical preamplifier with low noise characteristics is used before the optical receiver. Using these amplifiers can drastically increase repeaterless transmission distance[13)]. In-line optical amplifiers, inserted for longer transmission line, however, generate optical noise which degrades signal to noise ratio. Figure 9 shows how $S/N$ degradation due

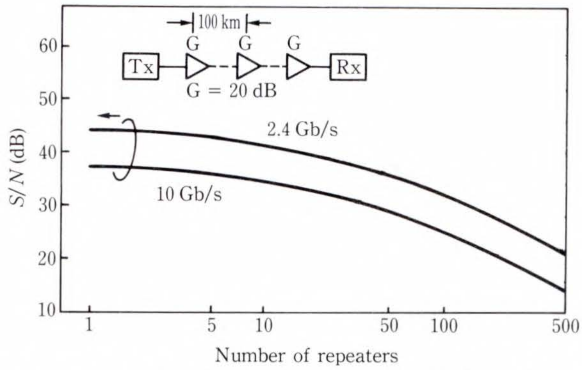



Fig. 9—$S/N$ degradation due to optical amplifier chain.

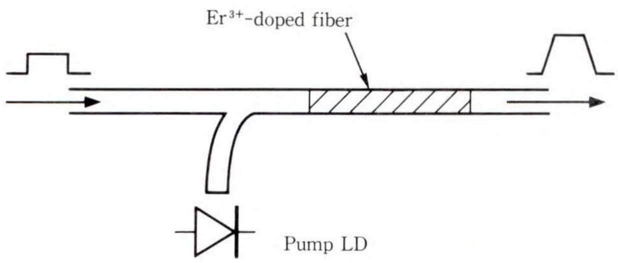



Fig. 10—Basic configuration of EDFA.
Advantages: plarization-insentive gain and low coupling loss.

to the amplifier chain can be serious when more than 100 amplifiers are connected in a chain.

Optical amplifiers are also important in the subscriber loop system or the CATV distribution system. Optical amplification will be an inevitable technology to realize broadband ISDN.

### 3.1.2 Amplifier technology

There are two types of optical amplifier; the optical fiber amplifier and the semiconductor laser amplifier. In the first, optical fibers having a core doped with the rare earth element erbium provide an efficient amplifier medium in the 1.55 $\mu$m range. This type has been developed extensively in recent years and is becoming practical. The principle of the Erbium Doped Fiber Amplifier (EDFA) is shown in Fig. 10. The energy level of the erbium ion is elevated by light with a shorter wavelength than the 1.55 $\mu$m signal light, i.e. 1.48 $\mu$m or 0.98 $\mu$m, supplied from a pumping laser. The other type, the Semiconductor Laser Amplifier (SLA), which has a low reflectivity facet to prevent oscillation of the laser, still needs some study to reduce the dependence of gain on polarization and to reduce the coupling loss to the fiber.

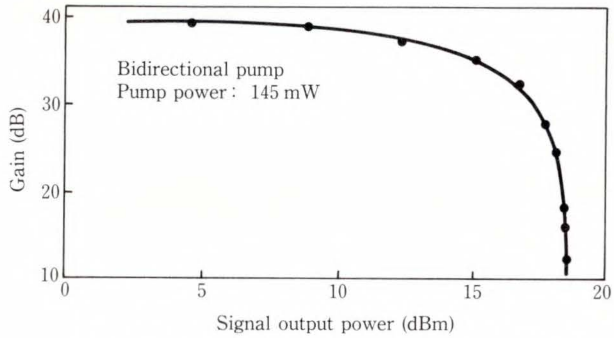



Fig. 11—Gain versus output characteristics.

### 3.1.3 Amplifier characteristics

This subsection summarizes typical characteristics of the EDFA. Figure 11 shows gain versus output characteristics for a pumping power of 145 mW. A small signal gain of 40 dB and saturated power in the high power region of +18 dBm (64 mW) were obtained. Although wave distortion in the saturated region prevents the normal use of electrical linear amplifiers, the signal waveform in the EDFA does not suffer distortion due to the long relaxation time (milliseconds) of the erbium ion. The amplifier bandwidth of the EDFA is usually a few nanometers, corresponding to several hundred GHz, which is wide enough to permit a single stage amplifier for a single-wavelength signal. Wavelength multiplexed signals or multiple stage operation of amplifiers, however, require more bandwidth, and aluminum co-doping into the fiber core achieves a wide bandwidth of 30-40 nm[14].

## 3.2 Ultra-high speed optical transmission

Current research into higher modulation bitrates is striving for 10 Gb/s and higher[15]. In such time-division multiplexing transmissions, the main obstacle in the transmission characteristics is the limitation due to fiber dispersion. Figure 12 shows typical tolerable fiber dispersions using an external intensity modulator, monolithically integrated electro-absorption modulator, and conventional direct modulation of a DFB laser[16]. Key technologies of such ultra-high speed optical transmission systems are a low-chirp optical modulator, wideband optical receiver using flip-chip type photodetector, optical amplifier, and ultra-high speed ICs.

### 3.2.1 Low-chirp optical modulator

A $LiNbO_3$ Mach-Zehnder external modulator is very attractive because a tolerable fiber dispersion of more than 450 ps/nm is possible at 10 Gb/s. The modulator is described in detail in Chapter 4.

### 3.2.2 Wideband optical receiver using a flip-chip type photodetector

A flip-chip type photodetector is very effective for optical receivers at speeds higher than 10 Gb/s. The flip-chip photodetector eliminates bonding wires and avoids the resonance caused by photodetector capacitance from the bandwidth of the optical receiver. A 10 GHz-bandwidth optical front end using a Si-bipolar preamplifier IC and a flip-chip APD has been reported. A slant-end fiber is used for fiber-to-photodetector coupling[17].

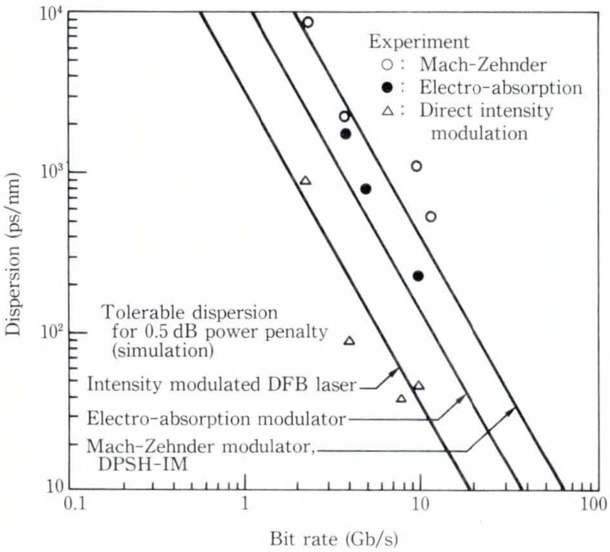



Fig. 12–Tolerable fiber dispersion corresponding to each modulation scheme.

### 3.2.3 Optical amplification

Optical amplification will play an important role, especially in ultra-high speed optical transmission systems. An optical booster amplifier at the transmitter output as well as an optical preamplifier at receiver input will be required to realize a 10 Gb/s-90 km transmission. A 10 Gb/s transmission using in-line optical amplifiers like the one shown in Fig. 13, is also attractive. A regenerative span of more than 400 km will be possible by using the $LiNbO_3$ Mach-Zehnder modulator.

### 3.2.4 Ultra-high speed ICs[10]

Si bipolar, GaAs MESFET, and III-V hetero-junction bipolar processes are under study for this application. Each process has advantages and disadvantages, and the best choice will greatly depend on future research developments. It is worth noting that Si bipolar is still a viable process.

## 3.3 Coherent lightwave transmission

This technology is expected to be the next

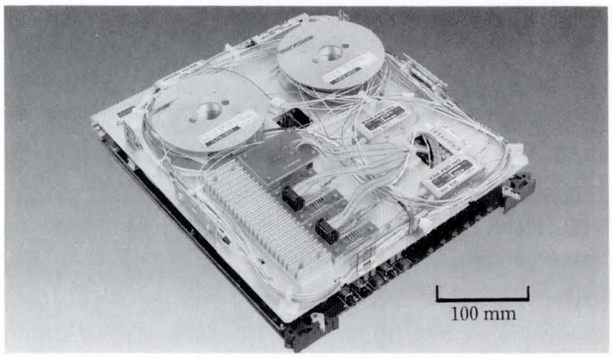



Fig. 13–Assembled optical amplifier for 10 Gb/s system.

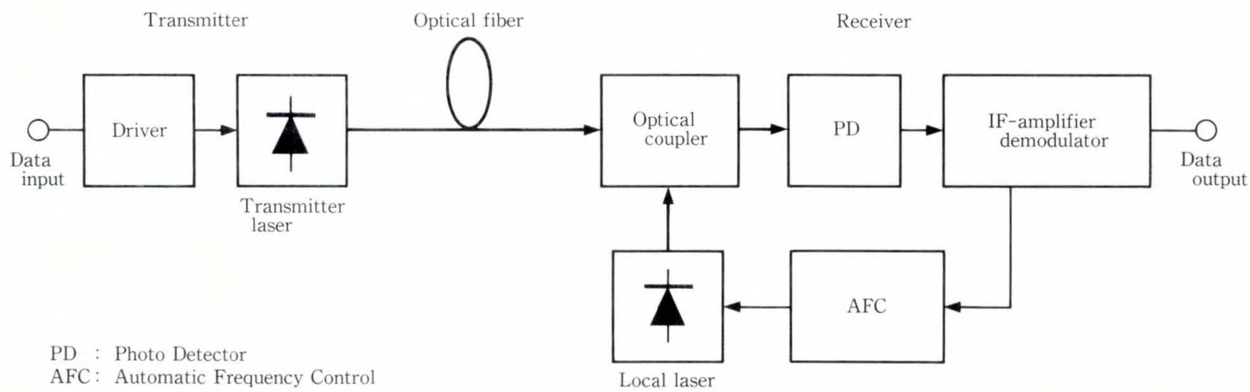



Fig. 14–Basic system configuration of the coherent transmission.

generation of optical communication because its high receiver sensitivity and high frequency selectivity can utilize the full bandwidth of a single-mode fiber. Figure 14 shows the basic system configuration. The phase, frequency or amplitude of the lightwave is modulated, it is transmitted through a single-mode fiber and is detected by heterodyne or homodyne reception using a local laser.

Near-term application of this coherent lightwave technology is for repeaterless undersea transmissions over 300 km; transmissions over more than 2 000 km are possible using optical amplifier repeaters and coherent terminal equipment[18,19]. The long-term target is broadband ISDN and optical switching systems using enhanced frequency selectivity in optical frequency division multiplexing.

We are now developing a 2.5-4 Gb/s Continuous Phase FSK (CPFSK) system to enable repeaterless transmission over more than 250 km. To realize the CPFSK system, we have developed the following technologies.

### 3.3.1 Three-electrode DFB-LD module

We have developed the 1.55 $\mu$m long-cavity three-electrode DFB-LD and its module shown in Fig. 15[20]. This module has a very narrow spectral linewidth of less than 1 MHz, a flat FM modulation bandwidth of 8 GHz, and high output power exceeding 10 dBm. The optical frequency is well stabilized by controlling its temperature and a 60 dB built-in optical isolator.

### 3.3.2 Balanced receiver

We have also developed a Dual-detector Balanced Optical Receiver (DBOR) consisting of a twin-PIN photodiode (see Fig. 16) and High Electron Mobility Transistors (HEMT) as a front-end preamplifier. The DBOR has a 14 GHz bandwidth and very low noise characteristics of 14 pA/$\sqrt{\text{Hz}}$. Using two DBORs, we have developed a polarization diversity receiver to overcome polarization fading in the transmitting fiber. This technology enables coherent transmission via installed conventional single-mode fibers.

### 3.3.3 2.4 Gb/s CPFSK system and its characteristics

Our prototype system of a 2.4 Gb/s CPFSK transmitter and a heterodyne polarization diversity receiver[21] is shown in Fig. 17. Figure 18 shows its bit error rate characteristics. The receiver sensitivity is better than 44.1 dBm at a BER of $10^{-9}$, which is 10 dB better than that of conventional IM/DD systems. We have also demonstrated 254 km single-mode fiber

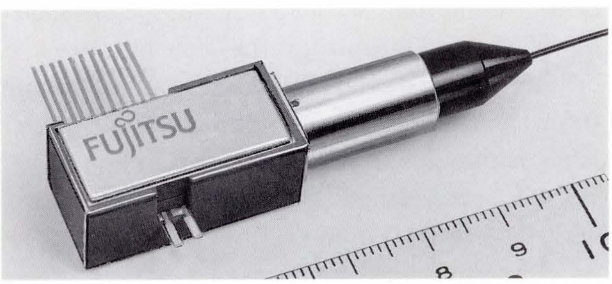
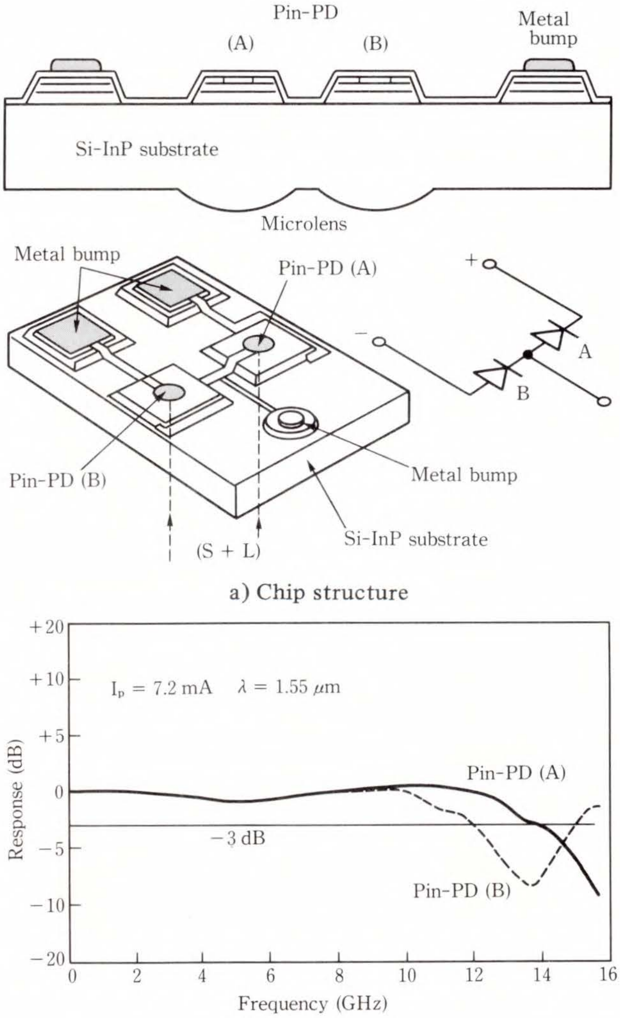



Fig. 15—Laser module for three-electrode DFB-LD.




a) Chip structure

b) Characteristics

Fig. 16—Twin-PIN photodiode.

(total dispersion: 4 854 ps/nm) transmission with a small dispersion penalty of 1.4 dB, and confirmed the stable operation of this equipment over about 2 months.

Advances in coherent lightwave technology continue to be made and we should see a practical system in the near future.

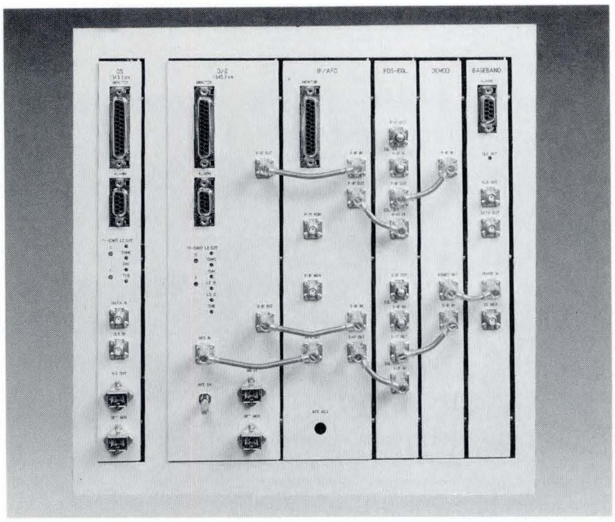


Fig. 17–Prototype system of a 2.4 Gb/s CPFSK transmitter and receiver.

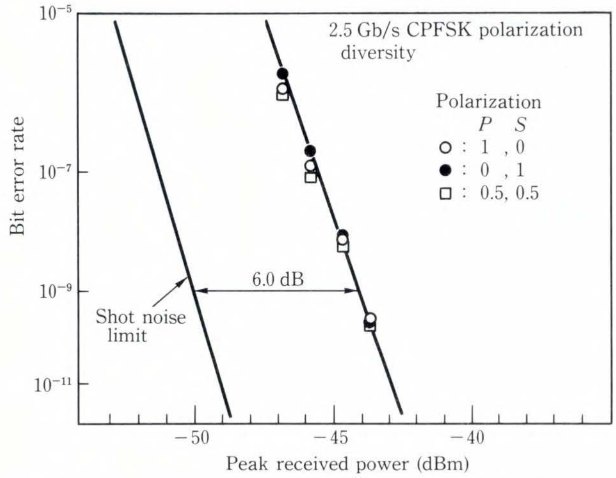
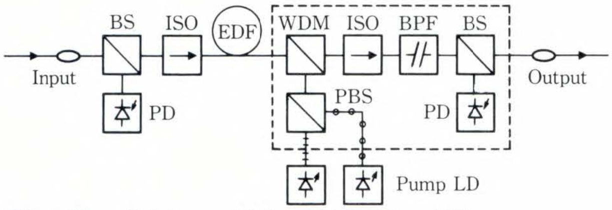



Fig. 18–Bit error rate characteristics of 2.4 Gb/s CPFSK system.




BS : Beam Splitter EDF : Erbium-Doped Fiber
ISO : Optical Isolator WDM : Wavelength Division Multiplexer
BPF : Band Pass Filter PBS : Polarization Beam Splitter

Fig. 19–Configuration of the erbium-doped fiber amplifier (Backward pumping configuration for power amplifier).

## 4. Optical devices

In this chapter, optical devices for the optical amplifier such as erbium-doped fiber and a WDM module are described. A $LiNbO_3$ Mach-Zehnder external modulator for multi-gigabit optical transmission is also described.

### 4.1 Optical devices for the optical amplifier

The erbium-doped fiber amplifier has been extensively developed in recent years, and is coming into practical use in 1.55 μm transmission systems. Its optical circuit consists of the erbium-doped fiber, a pumping LD module and other optical devices such as a WDM coupler, isolator, coupler for monitoring and bandpass filter. The configuration and characteristics of the optical devices for an erbium-doped fiber amplifier pumped by 1.48 μm LDs are described in this section.

Figure 19 shows the configuration of the erbium-doped fiber amplifier. Figure 20 shows the external appearance of the optical devices.

#### 4.1.1 Erbium-Doped Fiber (EDF)

EDF has good amplification characteristics in the 1.55 μm wavelength band. To achieve flat wavelength characteristics, we developed the erbium-aluminum co-doped fiber. Using an MCVD method, co-doping erbium and aluminum in the center area of the fiber core, we obtained broadband (30-40nm) fiber[14]. Gain-wavelength characteristics are shown in Fig. 21. High power conversion efficiency is also important, especially in power amplifiers. We realized high efficiency by reducing the EDF background loss. The background loss is closely related to the concentration of erbium and co-doped aluminum. We achieved a slope efficiency of 90 % by optimizing the concentration of erbium and co-doped aluminum to 400-500ppm, 0.5 wt% respectively. Amplification characteristics vs pumping power are shown in Fig. 22.

#### 4.1.2 Wavelength Division Multiplexer (WDM) module

To reduce insertion loss and size, we developed a WDM module in which six components

(WDM coupler, polarization beam splitter, isolator, band pass filter, coupler and photodiode) are integrated by micro-optics technology[22]. Configuration of this module is shown in Fig. 23. The module is 40 x 35 x 8 mm and the insertion loss is 1.8 dB for the signal path, 0.8 dB for the pump path. A mode field diameter conversion lens system has been developed to reduce coupling loss between fibers with different mode field diameters (6 $\mu$m, 8 $\mu$m, 10 $\mu$m).

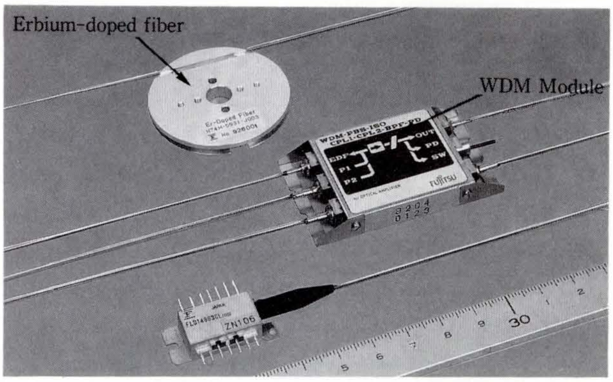



Fig. 20—External appearance of optical devices for the erbium-doped fiber amplifier. (Erbium-doped fiber coil, pumping laser, WDM module).

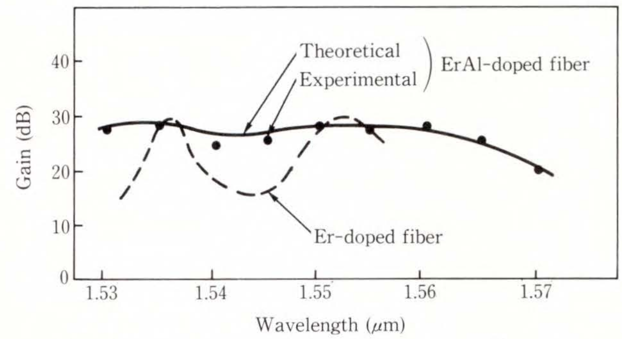



Fig. 21—Wavelength dependence of amplification characteristics.

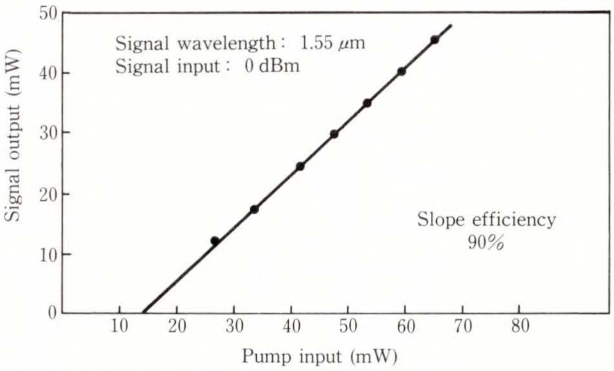



Fig. 22—Amplification characteristics vs pumping power.

## 4.2 $LiNbO_3$ Mach-Zehnder external modulator

The Ti-diffused $LiNbO_3$ Mach-Zehnder external modulator is a key device for high-speed, long-haul transmission systems because of its low chirp characteristics. We obtained $LiNbO_3$ modulators reliable enough for use in a commercial system. By contriving their buffer layers, we have reduced the DC drift which has been the largest problem in bringing $LiNbO_3$ devices into practical use. Our modulators are superior also in the stability of their modulation characteristics and insertion loss against temperature change[23),24)].

A schematic configuration of the $LiNbO_3$

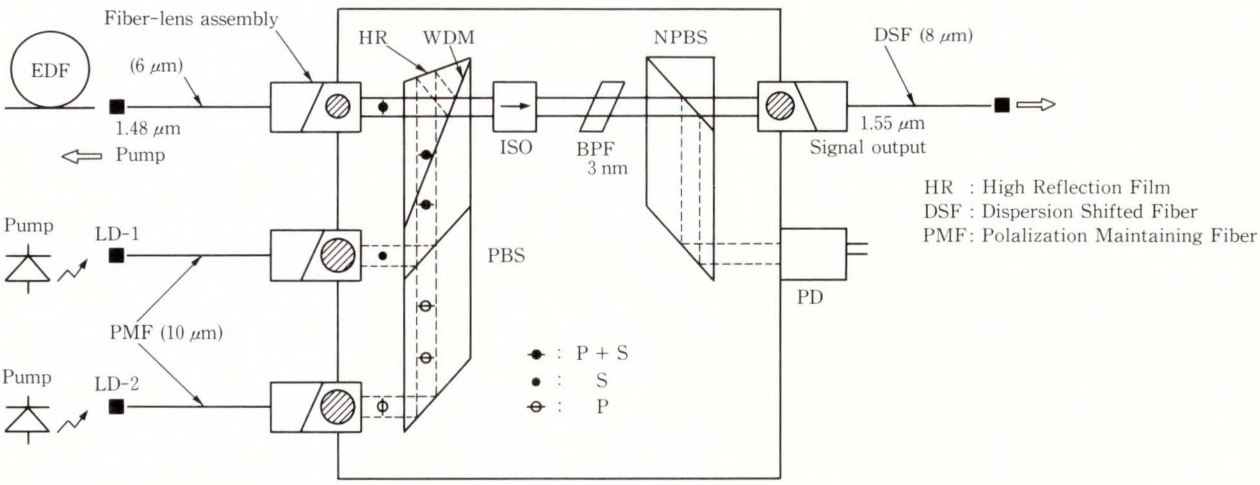



Fig. 23—Configuration of the WDM module.

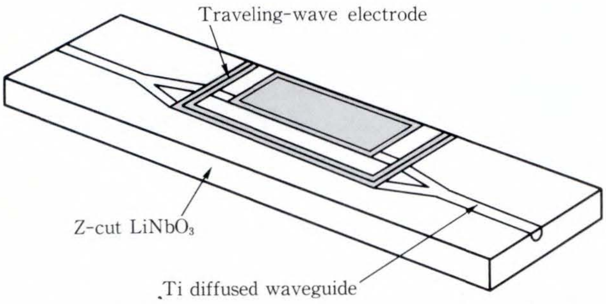



Fig. 24—$LiNbO_3$ Mach-Zehnder external modulator.

Table 7. Characteristics of $LiNbO_3$ Mach-Zehnder external modulator

| Items | Characteristics |
|---|---|
| Low insertion loss | 2.8 dB |
| High-speed operation | 3 GHz |
| Wide wavelength band | 1.53-1.57 $\mu$m |
| Low drive voltage | 3.0 V |
| High temperature stability | 0-85 °C |

A shematic configuration of the $LiNbO_3$ Mach-Zehnder external modulator is shown in Fig. 24. Table 7 shows its characteristics. Figure 25 shows its external appearance.

## 5. Conclusion

This paper reviews the development of fiber optic transmission systems in Fujitsu, and introduces our research and development activities and optical component technologies. Systems described in Chapter 2 have been effective in reducing the transmission cost per channel and are widely installed in many countries.

The fiber optic transmission systems developed so far, however, are mostly trunk line systems. As broadband and multimedia information network traffic including optical CATV increases, broader bandwidth in subscriber lines and optical cable installations will be demanded. Mass production will reduce costs and encourage further penetration of optical technology synergetically. With the evolution of subscriber loop systems the capacity in trunk line systems will increase further and require "Tera-bit" transmission systems.

Developments in ultra-high speed TDM technology will enable the installation of commercial 10 Gb/s transmission systems in the near future.

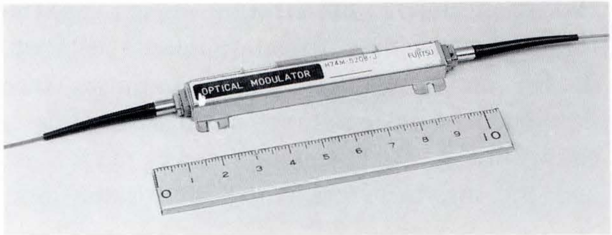



Fig. 25—External appearance of $LiNbO_3$ Mach-Zehnder external modulator.

Also, soliton transmission is a candidate for ultra-long submarine systems. Beyond the limits of the TDM approach, WDM or optical FDM technologies can be powerful solutions. Ultimate transmission capacity will be realized by coherent transmission technology using the ultra-wide spectrum of optical frequency resources. Network flexibility will also be enhanced with optical switching or optical cross connect functions. Optical transmission technology will also be used within computers. We will continue to develop the next generation of fiber optic transmission by effectively exploiting the advantages of our own technologies.

### References


1) Shirakawa, H., Watanabe, R., and Miura, H.: Trends in Optical Fiber Systems Development at NTT. Proc. IEE Int. Conf. 1985, pp. 171-174.
2) Kimura, H., and Nakagawa, K.: F-1.6G System Overview. *Electr. Commun. Lab. Tech. J.*, **36**, 2, pp. 153-160 (1987).
3) CCITT: Recommendation G. 707, G.708, G. 709. Blue Book, 1989.
4) Nakajima, A., and Tsuji, H.: A Study on Optical Interface of 2.4 Gb/s Fiber Transport Module for SDH System. (in Japanese), 1990 Autumn IEICE Natl. Conv. Rec., B-606, pp. 3.268.
5) Miyauchi, A., and Onishi, M.: 1.6-Gbit/s Optical Fiber Transmission Equipment. *FUJITSU Sci. Tech. J.*, **25**, 3, pp. 228-247 (1989).
6) Miyauchi, A., Nishimoto, H., Okiyama, T., Kitasagami, H., Sugimoto, M., Tamada, H., and Emori, S.: Fujitsu Limited, Integrated Circuit Device Having Stacked Conductive Layers Connecting Circuit Elements Therethrough, US Patent 4 827 327, 1989-05-02.
7) CCITT: Recommendation G957. 1990.
8) Miyauchi, A., Suzuki, K., and Nagakubo, Y.: 1.8 Gb/s 1.55 $\mu$m Optical Transmitter and Reciver.

(in Japanese), 1990 Spring IEICE Natl. Conv. Rec., B-924, pp. 4.103.

9) Miyauchi, A., Onishi, M., and Okamoto, A.: Optical Modules for SDH Optical Transmission Equipment. (in Japanese), *FUJITSU*, **42**, 4, pp. 372-380 (1991).

10) Ueno, N. et al.: Integrated Circuits for Digital Transmission Systems. *FUJITSU Sci. Tech. J.*, **28**, 2, (Special Issue on Telecommunications) pp. 243-259 (1992).

11) Tsuda, T., Onishi, M., Yamashita, M., Nagakubo, Y., Kawai, M., Yako, H., and Terayama, Y.: 2.4 Gb/s Optical Receiver. (in Japanese), 1991 Spring IEICE Natl. Conv. Rec., B-946, pp. 4.108.

12) Suzuki, K., Miyauchi, A., Masuko, T., Shima, M., Komatsu, M., Tajima, K., and Imai, K.: 2.4 Gb/s Optical Transmitter. (in Japanese), 1991 Spring IEICE Natl. Conv. Rec., B-945, pp. 4.107.

13) Hagimoto, K., Iwatsuki, K., Takada, A., Nakazawa, M., Saruwatari, M., Aida, K., Nakagawa, K., and Horiguchi, M.,: 250 km non-repeatered transmission experiment at 1.8Gb/s using LD pumped $Er^{3+}$-doped fiber amplifiers in IM/Direct detection system. *Electron. Lett.*, **25**, pp. 662-664 (1989).

14) Tagawa, K., Okamura, K., Inagaki, S., Takeda K., and Yoshizawa, S.: Broadening of gain-wavelength characteristics by erbium-aluminum-doped fiber. Proc. OEC' 90, PD-6, pp. 16-17.

15) Nakagawa, K., Hagimoto, K., Nishi, S., and Aoyama, K.,: A bit-rate flexible transmission field trial over 300 km installed cables employing optical fiber amplifier. Proc. Optical Amplifiers and their Applications (OAA '91), PDP11, 1991.

16) Minami, T., Nishimoto, H., Suyama, M., Kuwahara, H., Horimatsu, T., and Touge, T.: Optical amplification and modulation techniques for multigigabit IM/DD long-haul transmission systems. Proc. ICC' 90, 219.3 pp. 183-188.

17) Hamano, H., Yamamoto, T., Nishizawa, Y., Oikawa, Y., Kuwatsuka, H., Tahara, A., Suzuki. K., and Nishimura, A.: 10Gbit/s optical front end using Si-Bipolar preamplifier IC, flipchip APD, and slant-end fibre. *Electron. Lett.*, **27**, 18, p. 1602 (1991).

18) Hayashi, Y., Ohkawa, N., Fushimi, H., and Yanai, D. : A fully-engineered coherent optical trunk transmission system. Proc. ECOC '91, pp. 393-396.

19) Saito, S., Imai, T., Ito, T. : An over 2 200 km coherent transmission experiment at 2.5 Gb/s using erbium-doped fiber in-line amplifiers. *IEEE J. Lightwave Technol.* **9**, 2, pp. 161-169 (1991).

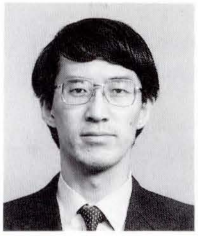
20) Onaka, H., Miyata, H., Kotaki, Y., Kuwahara, H., Takada, T., and Takeuchi, Y.: Distributed laser diode module with 8-GHz flat FM response and +10-dBm output for coherent lightwave transmission. Proc. OFC' 90, FD4, pp. 208.

21) Chikama, T., Naito, T., Kiyonaga, T., Ishikawa, G., and Kuwahara, H.: Optical heterodyne continuous phase FSK transmission experiment up to 4 Gb/s. Proc. 16th ECOC, 1990, pp. 65-66.

22) Naganuma, N., Kubo, T., Murai, T., and Abe K.: Compact low-loss integrated microoptics circuit for er-doped fiber amplifiers. Proc. MOC' 91, PD3, pp. 10-13.

23) Seino, M., Nakazawa, T., Kubota, Y., Doi, M., Yamane, T., and Hakogi, H.: A low DC-drift Ti: $LiNbO_3$ modulator assured over 15 years. Proc. OFC' 92, PD3, pp. 325-328.

24) Suzuki, K., Onishi, M., Nishimoto, H., Amemiya, I., Kuwata, N., Okushima, H., and Miyauchi, A.: A compact 2.5-Gbit/s Optical Transmitter Module with a $LiNbO_3$ Mach-Zehnder modulator. Proc. OFC' 92, WM3, pp. 169.



**Hideo Kuwahara**

Optelectronic Systems Laboratory
FUJITSU LABORATORIES, KAWASAKI
Bachelor of Electronics Eng.
The University of Tokyo 1972
Dr. of Electronics Eng.
The University of Tokyo 1984
Specializing in Optical Transmission Systems

**Akira Miyauchi**

Optical Transmission Engineering Dept.
Transmission Division
FUJITSU LIMITED
Bachelor of Electronics Eng.
Kyushu University 1974
Specializing in Development of High-speed Optical Transmission Systems

**Akira Mitsuhashi**

Optical Devices Development Dept.
FUJITSU LIMITED
Bachelor of Precision Mechanical
The University of Tokyo 1972
Specializing in Optical Passive Devices

UDC 621.391.8.037.3

# Digital Signal Processing Technology for Communications

● Kiichi Matsuda ● Yoshitsugu Nishizawa ● Fumio Amano

*(Manuscript received January 9, 1992)*

This paper describes the history, major techniques, and developing trends in video and audio signal processing technology in the field of communications. Communications via transmission lines and wireless systems are discussed. Communications via digital storage media, which has recently attracted much interest, is also dealt with in this paper. Topics related to ATM, expected to become the next-generation transmission method, are also discussed for both video and audio.

## 1. Introduction

This paper describes processing of audio and video signals for telecommunications. Telecommunications invariably requires transmission lines which have limited capacities. A key point in telecommunications systems is how efficiently this capacity is utilized. Moreover, in bidirectional communications such as TV conferencing where each terminal uses loudspeakers and microphones, echoes and howling also have to be considered.

Asynchronous Transfer Mode (ATM) provides variable bit rate transmission as one of its main features. Studies on the ATM network are being conducted both for video transmission and audio transmission. Coding algorithms for new media such as digital storage (i.e. CD-ROM) are also under development.

Chapter 2 of this paper reviews Fujitsu's activities in the development of digital video and audio signal processing technology, and Chapter 3 explains some of these areas in more detail.

## 2. The development of signal processing technology: background

This chapter describes Fujitsu's main activities in both video and speech processing from about 1985 to the present.

### 2.1 Video processing technology

#### 2.1.1 Universal standard TV coder

Fujitsu started its development of video coding technologies and codecs in the early 1970's[1]. Activities were divided into two categories: One was high bit rate coding for Industrial Television (ITV) which adopted intraframe prediction, and the other was low bit rate coding for videophones and TV conferencing that started several years later and adopted an interframe prediction scheme. Hardware development was on a customer-by-customer basis since:

1) high speed transmission lines could not be easily used,
2) the quantity of hardware required was great since the processing capacity of devices was not very high,
3) the quality of picture reproduction was not satisfactory, and
4) there was no recommended standard.

In the 1980s, high-speed digital transmission lines, such as leased digital lines and ISDNs, became available and hardware was made smaller owing to progress in the processing capaicty of ICs. These new high-speed lines brought new visual services such as TV conferencing and CATV. Also at this time, CCITT started work on standardization of a video coding algorithm

Fig. 1–TV conferencing system "Station E".


Fig. 2–TV conferencing system "VS-700".

and recommended two algorithms in 1984; one was the Phase Alternation Line (PAL) system at 2 Mb/s, and the other was the National Television System Committee (NTSC) system at 1.5 Mb/s. Adaptive interframe/intraframe prediction with Motion Compensation (MC) was employed in both.

In the latter half of the 1980s, new video coding algorithms were developed with lower transmission rates than before. These new methods adopted block coding, vector quantization, and orthogonal transform coding, in addition to conventional interframe prediction coding. Vector quantization carries out quantization on a block-by-block basis, while the transform coding method quantizes prediction errors after converting them into the frequency domain. Fujitsu started development of a video codec for TV conferencing which used vector quantization in 1984, and started delivery of TV conferencing system "Station E" in 1985 (see Fig. 1).

The CCITT also began investigating a video coding algorithm with transmission bit rates of less than 1.5 Mb/s. Various kinds of coding algorithms such as Vector Quantization (VQ) and Discrete Cosine Transform (DCT) were proposed by Japan and countries in both Europe and North America. After evaluation and discussion, CCITT finally recommended a DCT based method for $n \times 384$-kb/s codecs.

Fujitsu developed a transform coding technology and completed an experimental codec in 1988. Based on experience obtained in the development of this codec, Fujitsu has contributed to the CCITT expert group for video coding with regards to:

1) the specification of a loop filter that enhances coding efficiency and improves reproduced picture quality, and
2) the precision of DCT calculation necessary to avoid accumulated errors that become noticeable.

However, a new study aimed at a coding algorithm for much lower transmission bit rates has begun. The recommendation for $n \times 384$ kb/s has widened to $p \times 64$ kb/s, where $p$ ranges from 1 to 30. This new standard was recommended in December 1990 as CCITT H.261.

Under the circumstances mentioned above, Fujitsu has mainly been developing video codecs for use in TV conferencing. Fujitsu developed a video codec based on a 1984 CCITT recommendation and delivered it to NTT in 1987. Fujitsu then developed a new video codec based on vector quantization and delivered it to NTT in 1988. In the same year, Fujitsu developed and started delivery of a new TV conferencing system named "Station E" that included a video codec based on hybrid quantization, which is a combination of scalar and vector quantization.

Another new TV conferencing system, called "VS-700", was developed in 1991 (see Fig. 2). It operates in a mode compatible with CCITT H.261, and when both terminals in communication are VS-700, it provides enhanced features such as:

1) superimposed display of the main speaker's face on a wide angle view or document, and
2) remote camera control by a wireless keypad.

#### 2.1.2 HDTV coder

In addition to the development of NTSC signal transmission systems, Fujitsu has also been developing transmission technologies for HDTV signals. Fujitsu developed a compact transmission system with a transmission bit rate of 100 Mb/s using intraframe prediction[2]. In 1989, HDTV codec, FV-600M with distribution-level picture quality, was developed and later demonstrated by NTT at the Flower Exposition. Another new experimental HDTV codec for use in ATM systems was developed by modifying the previously developed 100-Mb/s codec. This codec transmits HDTV signals with transmission bit rates between 60 Mb/s to 130 Mb/s and has cell loss resilience, the significance of which is described in Subsection 3.1.2 (see Fig. 3).

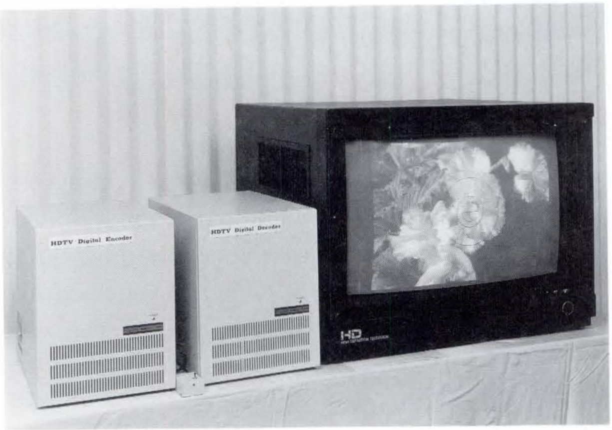


Fig. 3 – HDTV codec for ATM system.

#### 2.1.3 Digital image storage coder

The video codecs explained above were developed mainly for transmission services. However, with the recent proliferation of the CD-ROM, the study of coding algorithms for storage and recording of moving pictures is becoming increasingly more active. ISO recommended a coding algorithm for transmission bit rates of about 1 Mb/s. The algorithm adopted

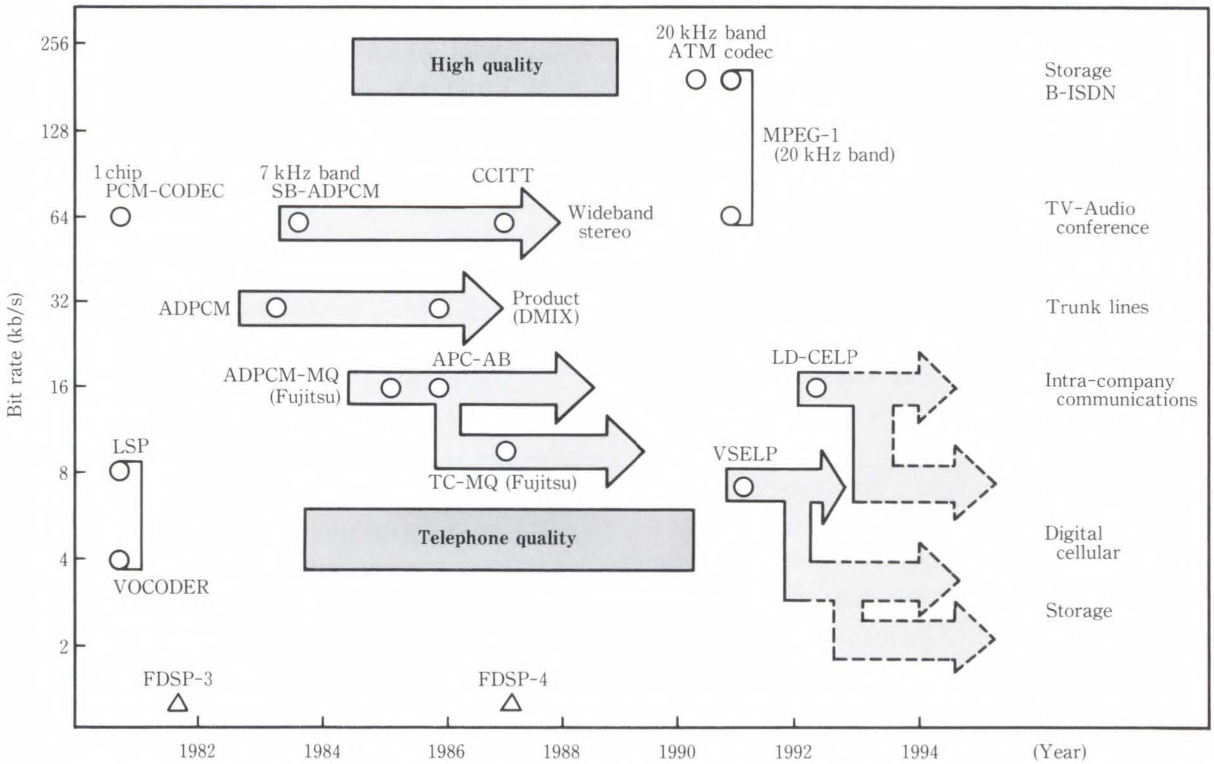



Fig. 4 – Speech coder development at Fujitsu.

the same MC + DCT scheme as CCITT H.261. In addition, other features such as random access, fast forward playback, and reverse playback are implemented using a coding control mechanism on a frame-by-frame basis.

Fujitsu has been investigating the possibility of standardization from the standpoint that this algorithm will become a key technology for multimedia coding in both communication and information services[3]. Fujitsu has also been contributing to this technology by presenting reproduced pictures created using original algorithms. A study group called MPEG is now engaged in the next step, developing technology for transmission rates below 10 Mb/s.

## 2.2 Speech processing technology

### 2.2.1 Speech coder

Speech coder development can be divided into two major paths: One for high-quality coders and the other for telephone band coders (see Fig. 4).

Fujitsu began development of a 7-kHz 64-kb/s high-quality speech coder for TV conferencing in 1983. In November of the same year, CCITT began work on speech coding standardization. Fujitsu made a large contribution by providing hardware for evaluation. When the subband ADPCM (SB-ADPCM) system was recommended as CCITT G.722, Fujitsu developed dedicated LSI and completed a reduced-scale system in March 1987.

Toward 1989, the demand grew for coders with the reproduction quality of a compact disk (CD) on a level sufficient for Broadband ISDNs (B-ISDNs) and storage media systems. To satisfy this demand, Fujitsu developed a 20-kHz, 384-kb/s, embedded stereo coder for ATM networks in 1990. The same year, ISO began to standardize the coding system for storage of moving images and high-quality audio on CDs and other digital storage media. ISO set the coding speed of audio at 128 kb/s (256 kb/s for stereo signals). Four systems were proposed. Fujitsu proposed Adaptive Transform Audio Coding (ATAC) and put together a trial system in a joint effort with the Victor Company of Japan, Ltd., the Sony Corporation, and NEC Corporation.

Fujitsu started full-scale development of the COINS telephone-band speech coder in 1984. In 1986, Fujitsu developed a 32-kb/s ADPCM coder conforming to CCITT G.721. The coder could also transmit 9.6-kb/s modem signals. Fujitsu carried out a trial manufacturing of a 16-kb/s coder for high-level band compression using Adaptive Predictive Coding with Adaptive Bit allocation (APC-AB) in 1986. After this trial, Fujitsu succeeded in creating a 16-kb/s coder using two Fujitsu MB86232 floating-point Digital Signal Processor (DSP) chips at the end of 1987, and completed a single-chip ASIC device at the end of 1988. Through continuous research and development since the end of 1984, Fujitsu has developed a unique 16-kb/s ADPCM with Multiquantizer (ADPCM-MQ) system. A 16-kb/s coder based on this system was created using two Fujitsu MB8764 16-bit fixed-point DSP chips. More efficient lines were also in great demand for overseas, intra-company communications networks. To satisfy this demand, Fujitsu started development of an 8-kb/s coder in 1986. A Time-domain Compression ADPCM-MQ (TC-MQ) system was developed by combining band splitting and time axis compression with ADPCM-MQ. Based on this concept, a product using six MB8764 chips was completed by the end of 1987.

Currently, many manufacturers are developing digital cellular mobile systems which rely greatly on speech coder technology. In Japan, 11.2-kb/s Vector Sum Excited Linear Predictive coding (VSELP) was selected as the system standard in June 1990. Fujitsu is promoting development using a dedicated DSP.

### 2.2.2 Echo canceller

Echo canceller development began in 1982. In 1983, a trial 360-tap unit for telephone conference systems and incoming call transfers was created using three MB8764 chips. After several manufacturing trials, an IC that suppresses echoes of up to 64 ms in duration was developed in 1986. This IC is used for incoming call transfers, three-party conferencing, and telephone conferencing. After the development of this IC, in the latter half of 1986, Fujitsu transferred

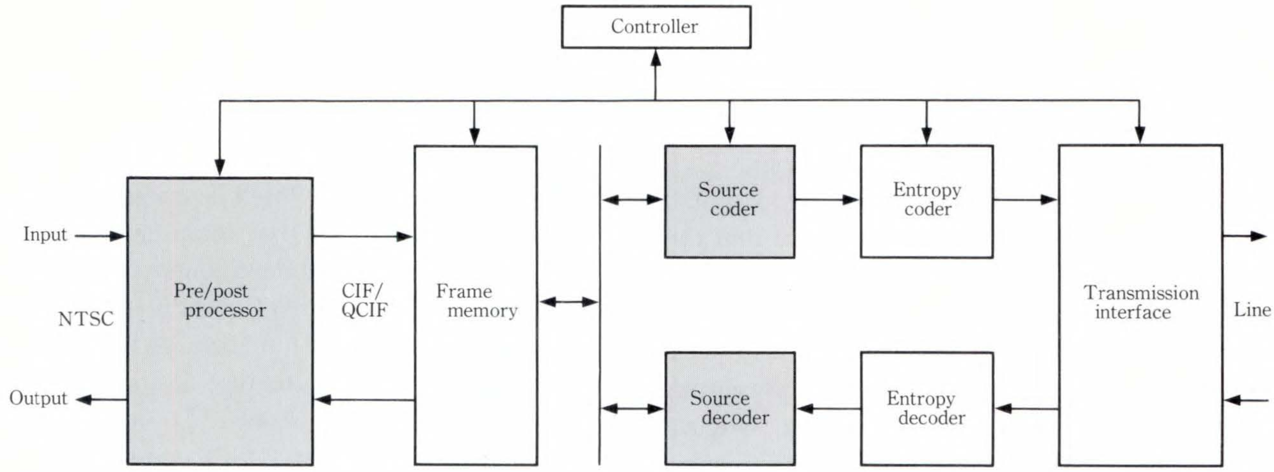



Fig. 5–Block diagram of a video codec based on CCITT H.261.

the emphasis of their research to the development of an acoustic echo canceller for TV conferencing systems. At the end of 1987, a trial unit for suppressing broadband (7 kHz) echoes of up to 256 ms in duration was realized with twenty-two MB86232 chips. To further save on processing and enhance convergence speed, a frequency domain processing system and a band splitting system are now under development.

## 3. Signal processing technology

Coding algorithms and processors, which have been developed to attain compact hardware, are explained in this chapter.

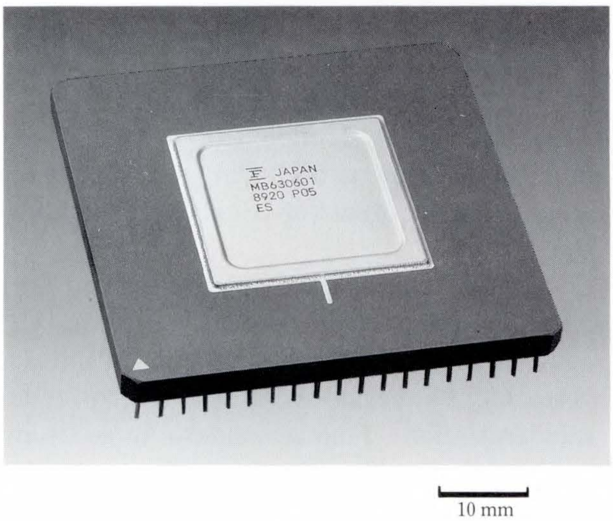



Fig. 6–VSP-LSI.

### 3.1 Video signal coding technology

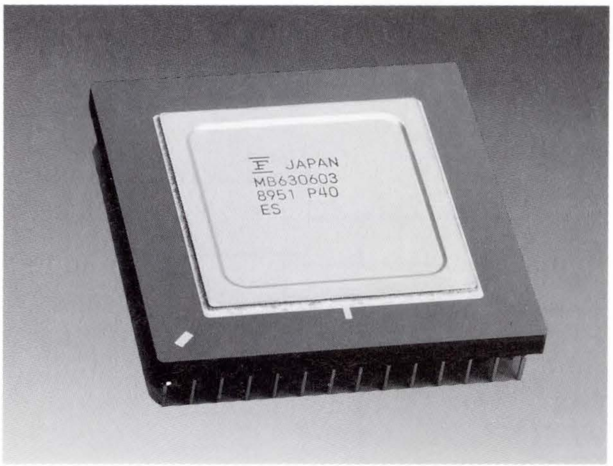
#### 3.1.1 LSIs for video signal processing

In accordance with the CCITT H.261 video coding algorithm for TV conferencing and videophone services recommended in December 1990, many manufacturers have started developing video codecs. Fujitsu developed TV conferencing terminal VS-700 in 1991. The main objective of development was to make the codec as small as possible and operational at transmission bit rates between 64 kb/s and 2 Mb/s.

In accordance with this development policy, three kinds of new LSIs were developed to make the codec smaller. The new codec adopted a hybrid architecture of fast dedicated circuits and processor based programmable circuits[4]. The block diagram of a video codec based on CCITT H.261 is shown in Fig. 5. In the figure, blocks that use these LSIs are shadowed.

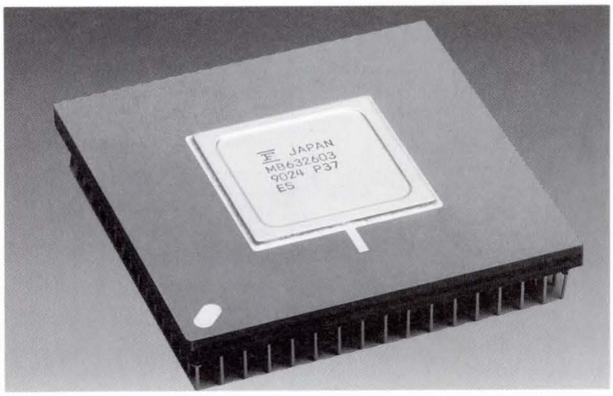
1) Video Signal Processor LSI (VSP-LSI)[5]

This LSI is a programmable video signal processor (see Fig. 6). Operations are carried out with a precision of 16 bits and with a cycle time of 70 ns. A pipelined architecture was adopted to execute block-matching operations typical in MC at high speeds. Although parallel schemes are widely used for high-speed operation, they tend to require a parallel data input/output mechanism for synchronous operation of multiple processors. In cases in which parallel input/output cannot be attained because of a limited memory capacity and/or limited signal lines in the case of a single-chip LSI with multiple processors, the processors have to operate




10 mm

Fig. 7–DCT-LSI.



10 mm

Fig. 8–VFC-LSI.

asynchronously. While some processors are reading new data, others are carrying out arithmetic operations and writing results back into memory.

This mode of operation describes the pipelined architecture used in this LSI. It has three parallel input/output ports: Frame memory (F) port, Control (C) port, and Extensional (E) port. The F port is especially designed for video data input/output and has a data transfer capability of 54 Mbytes per second.

2) Discrete Cosine Transform LSI (DCT-LSI)[6]

This LSI was developed for DCT operations of several block sizes including 8 by 8. Its data input/output port has a zigzag scanning converter (see Fig. 7). As for the precision of inverse DCT operation, it satisfies the requirements of H.261. This LSI can also be used for loop filtering of 8 by 8 blocks, an operation required by codecs conforming to H.261, as described in Subsection 2.1.1.

3) Video Format Converter LSI (VFC-LSI)[7]

H.261-based codecs require format conversion between conventional NTSC/PAL and the new Common Intermediate Format (CIF)/ Quarter CIF (QCIF) to make communications possible between regions with different TV signal formats. This LSI was especially developed for this conversion (see Fig. 8), and is applicable to both forward conversion, from NTSC/PAL to CIF/QCIF, and backward conversion, from CIF/QCIF to NTSC/PAL.

To reduce the number of gates in the LSI, multipliers that require many gates are shared by both conversions by controlling the signal flow and switching coefficients of the digital filter. Furthermore, the three component signals, Y, R-Y, and B-Y, are separated from the composite input signal and multiplexed or demultiplexed in the LSI to reduce peripheral circuits.

**3.1.2 Coding for ATM**

Studies of ATM networks are currently under progress. ATM, which provides variable bit rate transmission capability, is attractive for video signal coding since it makes efficient use of network resources and can improve picture quality. Fujitsu is investigating and developing video signal coding technology for ATM networks with the belief that his coding technology will become a key factor in future communications[8].

In ATM networks, cell loss occurs due to buffer overflows and transmission errors. Conventional video codecs with interframe prediction cannot be used under such circumstances since they have inconsistencies in data between the transmitting side and receiving side. This causes severe degradation in picture quality after a cell loss. New coding algorithms being studied to cope with cell loss can be divided into two groups: Concealment by source coding and concealment by transmission coding.

Some of the algorithms Fujitsu has been investigating are as follows. One example of source coding is separation of the source signal into a lower frequency part and higher fre-

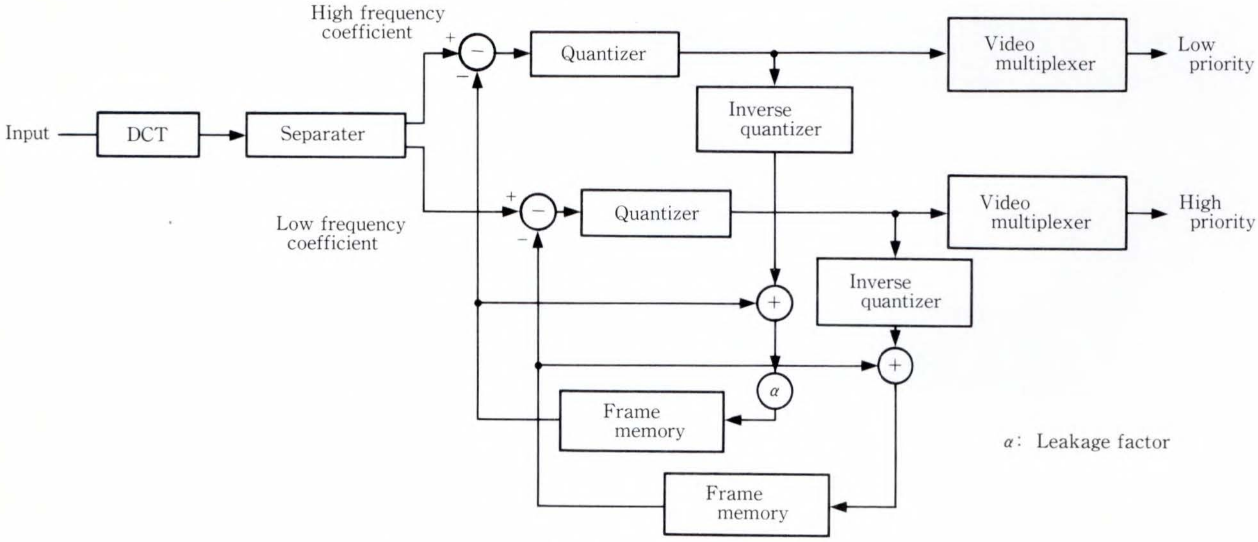



Fig. 9–Block diagram of ATM coder.

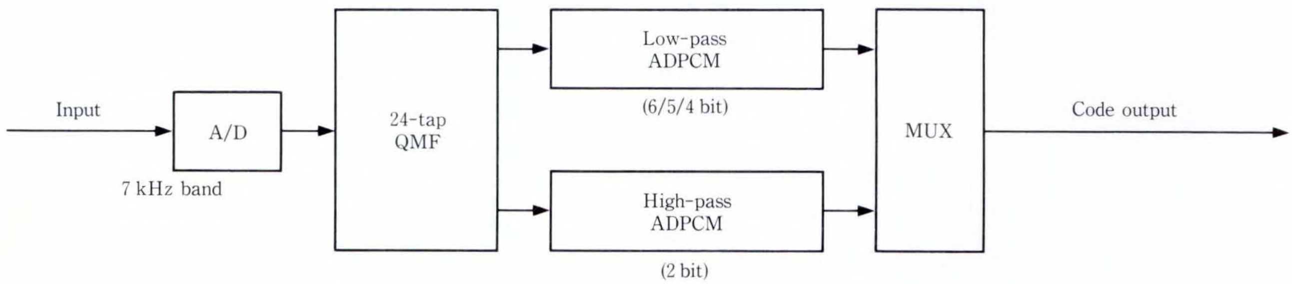



Fig. 10–SB-ADPCM coder block.

quency part, with the lower part being transmitted on a high priority channel with interframe prediction and the higher part encoded by interframe prediction with a leakage to absorb the effects of errors (see Fig. 9). In another method for fast recovery from transmission errors, the degree of picture quality degradation on the receiving side caused by transmission errors is estimated on the transmitting side. Intraframe prediction is selected at the next frame if the estimated error is large. In concealment by transmission coding, a line-block buffer is prepared to store entropy coded data which is reused in place of discarded data.

The next target of video coding is the development of a universal coding algorithm that increases picture quality and is suitable for a wide range of transmission bit rates while maintaining compatibility with H.261.

### 3.2 Speech processing technology

#### 3.2.1 Broadband speech coder

1) 7-kHz speech coder

Figure 10 shows the block configuration of the 7-kHz 64-kb/s speech coding algorithm recommended by CCITT G.722 for use in TV and speech conferencing systems [9),10)]. Signals converted from analog to digital at a sampling frequency of 16 kHz are split at 4 kHz into high-frequency band and low-frequency band signals by a Quadrature Mirror Filter (QMF). The low-frequency band signals are coded by ADPCM in six bits, and the high-frequency band signals in two bits. This algorithm has a data transmission mode. If there is a data communications request, the low-frequency band signals are quantized in five or four bits and transmitted at coding speeds of 56 or 48 kb/s.

To implement this algorithm in small-scale hardware, Fujitsu developed a dedicated IC based on its MB8764 DSP. Since the I/O analog filter for band splitting is subject to strict characteristics, a hybrid IC using an active filter was developed. Figure 11 is a photograph of the CODEC board. The coder and decoder consist of a single chip each.

2) 20-kHz audio coder

In the field of B-ISDNs, ATM networks are under active development. One of the main features of ATM networks is that they can transmit multiple media signals such as speech, audio, image, and data, all together at variable bit rates with a common cell format. However, cells may be lost depending on the network status or congestion. To solve the problem of cell loss, Fujitsu developed an experimental 20-kHz, 384-kb/s stereo coder using its general-purpose MB86232 DSP[11].

A band-splitting ADPCM is used for coding. Input signals are divided into four bands by a QMF, and embedded coding resilient to cell loss is performed for the lower two bands. The priority control system discards acoustically insensitive high-frequency band components first. If more cells must be lost, the low-order bits of the low-frequency band component are discarded. This enables communication with a cell loss rate of up to 62.5%. Table 1 lists the specifications of this coder, and Fig. 12 shows the trial unit.

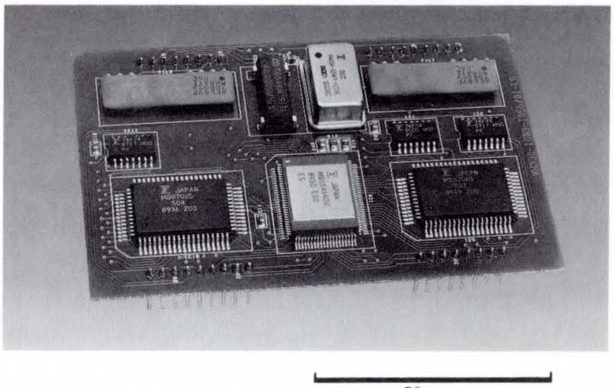


Fig. 11–SB-ADPCM coder board.

Table 1. Main specifications of ATM high-quality coder

| Items | Specifications |
|---|---|
| Coding algorithm | Band-splitting ADPCM |
| Bandwidth | 0 - 20 kHz |
| Sampling frequency | 48 kHz |
| Number of split bands | 4 (equal band) |
| Number of quantized bits | 8 : 4 : 4 : 2 |
| Number of predictable inverse-quantized bits | 4 : 2 : 2 : 2 |
| Coding bit rate | 192 kb/s |
| DSP type | MB86232 (14 chips) |

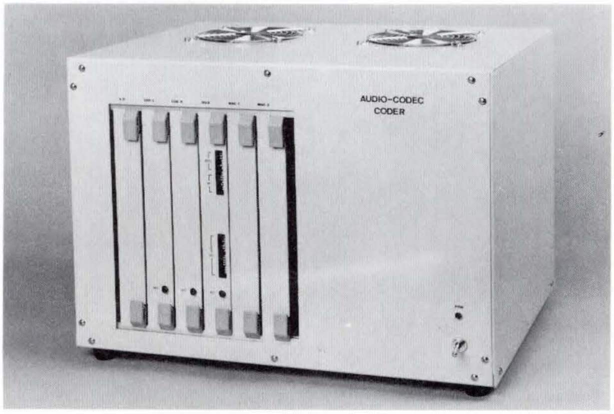


Fig. 12–ATM high-quality coder trial.

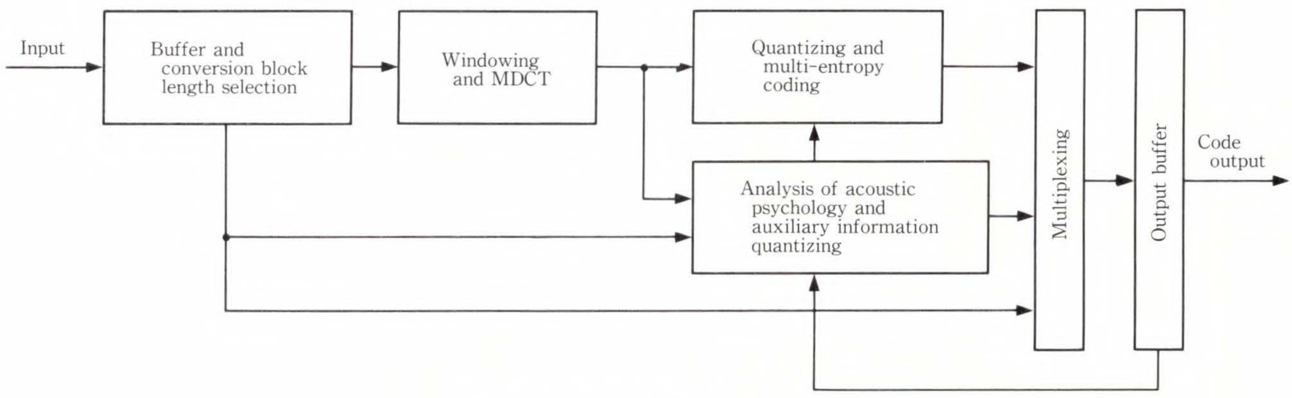


Fig. 13–ATAC coding algorithm.

The coding system that ISO is now standardizing for multimedia storage systems codes signals at 256 kb/s (128 kb/s for monaural signals). Acceptable quality cannot be achieved without a complicated algorithm. In addition, fast play, return, and other functions specific to the storage media must be considered. Four groups proposed different solutions for such a system. ISO made a subjective quality evaluation of each of these proposals in July 1990. Fujitsu proposed the ATAC system, together with the Victor Company of Japan, Ltd., the Sony Corporation, and NEC Corporation.

Figure 13 shows the configuration of the coding section. According to the results of the evaluation, a system combined with Audio Spectral Perceptual Entropy Coding (ASPEC), proposed by AT&T and CNET, and Masking pattern adapted Universal Subband Integrated Coding And Multiplexing (MUSICAM) proposed by IRT and Phillips, was submitted as a tentative recommendation[12].

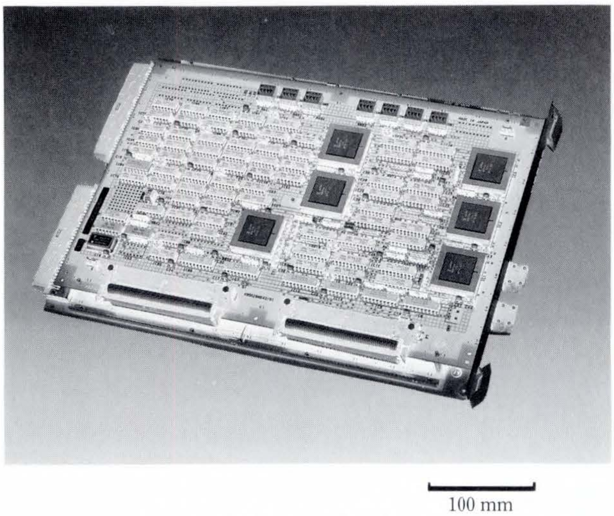



Fig. 14–8-kb/s TC-MQ coder.

**3.2.2 Telephone band speech coder**

For Multimedia Time Division Multiplexing (MTDM) systems, an APC-AB coder module was developed mainly from a dedicated IC (MB87528) which is based on a floating-point DSP (MB86232)[13]. The module can be switched between 16 kb/s and 8 kb/s. The module has also been applied in COINS. To enhance the speech quality regenerated at 8 kb/s, an 8-kb/s TC-MQ coder consisting of six MB8764 DSP chips was also developed[14] (see Fig. 14).

In the field of digital cellular mobile systems, digital systems are being developed to cope with the quick increase in subscribers. Speech encoders are also very important for these systems. For domestic systems, research in standardizing speech coding algorithms began under the leadership of the Research and Development Center for Radio Systems in September 1989. Motorola's VSELP system was selected as the standard in June 1990.

The system was improved with Code-Excited Linear Predictive coding (CELP) to enhance the sound quality, to reduce the amount of opera-

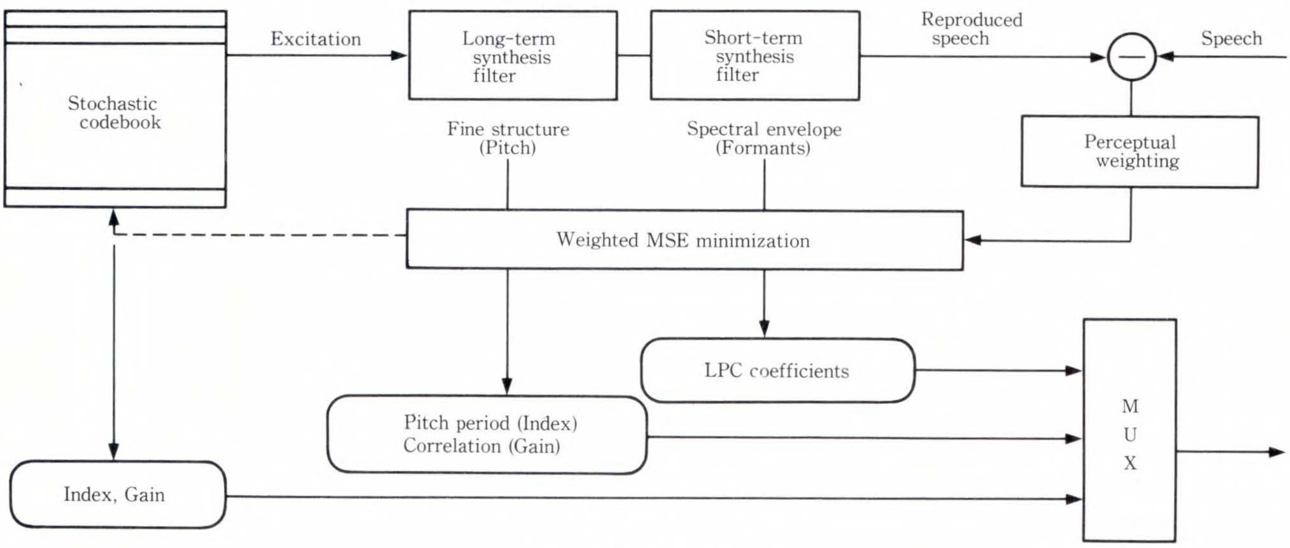



Fig. 15–Principle of CELP.

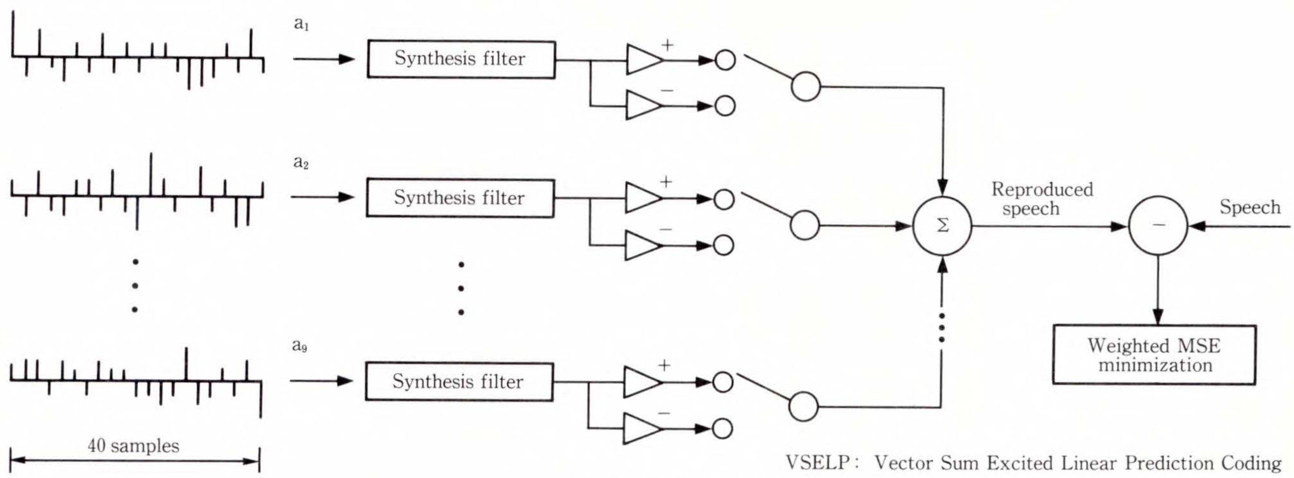



Fig. 16—VSELP decoder.

tions, and to increase resilience against transmission errors. As Fig. 15 shows, the CELP system has a code-book which stores several sound source waveforms as vectors. Speech is regenerated for each vector in the code-book through a synthesis filter, then a vector with the least difference from the input speech is selected and its corresponding address in the code-book is transmitted.

This system has the disadvantage in that an enormous volume of operations is necessary since all the vectors are filtered[15]. The VSELP system for Japanese represents code vectors by the linear combination of nine basic vectors to greatly reduce the required operations and to enhance resilience against transmission errors. Regenerated voice quality is enhanced by adding a pitch prefilter and a spectrum postfilter[16)-18)] (see Fig. 16). Fujitsu created the VSELP system using four MB86232 DSP chips. At this time, a single-chip dedicated DSP is under development to reduce power consumption.

### 3.2.3 Echo canceller

1) Echo canceller for telephone networks

An echo canceller has many applications, such as for satellite communications, incoming call transfer, and three-party conferencing. An effective means for developing ICs applicable to as wide a range of applications as possible involves:

i) creating a dedicated IC for the convolution section for pseudo-echo generation and the tap factor updating section, which are common in terms of function, and
ii) using a general-purpose DSP to control the individually required functions, such as speech detection and gain control amplification.

Table 2. Main specifications of echo canceller LSI

| Items | Specifications |
|---|---|
| Process | CMOS |
| Gate length | 1.5 $\mu$m |
| Gate delay (typ) | 1.0 ns |
| Number of gates | 9 300 |
| RAM capacity | 12 kbit |
| Machine cycle | 100 ns |

developed a dedicated IC that can be controlled by a DSP. This chip handles a delay of 64 ms to cover the maximum domestic (Japan) delay. The configuration is easily adaptable to tandem connection for application in acoustic echo cancellers. Table 2 lists its major specifications[19)].

The MB8764 control DSP performs several functions: bi-directional simultaneous speech control, normalization of residual echoes, impulse signal detection, protection from incorrect coefficient adaptation, detection of echo path changes, and input DC offset compensation.

2) Acoustic echo canceller

A speech system for TV conferencing uses microphones and speakers. To ensure high-quality voice and sound echoes and howling must be suppressed. An echo canceller is used for this purpose. Unlike a telephone network, the following conditions must be considered in an acoustic system:

i) The bandwidth is 7 kHz, much wider than that of a telephone channel.
ii) An echo impulse response continues for several hundred ms or sometimes as long as several seconds.
iii) The echo suppression requirement is 40 dB or greater.

A transversal echo canceller that uses the learning identification method popular in telephone networks must process several thousand taps. As a result, this type of canceller requires a large number of operations, and response to system fluctuation may be delayed.

Table 3. Main specifications of trial system

| Items | Specifications |
|---|---|
| Frequency band | 0.05-7 kHz |
| Sampling frequency | 16 kHz |
| Echo suppression time | 250 ms |
| Number of taps | 4 000 |
| Convergence algorithm | Correlation eliminating type |
| Configuration | Band splitting (in half) |
| Suppressed volume of echo | >40 dB |

Considering these problems, Fujitsu put together a trial system using a correlation eliminating algorithm to enhance the convergence speed and a band split configuration to reduce the size of hardware[20),21)]. Table 3 lists the main specifications and Fig. 17 shows the trial system. MB86232 DSP chips are used to obtain echo suppression of 40 dB or greater. The total number of chips is 22.

Frequency domain processing is effective in greatly reducing the number of operations[22)]. To further reduce the volume of operations, Fujitsu proposed a combined system consisting of a band split configuration using a QMF, and down-sampling technology[23),24)] (see Fig. 18).

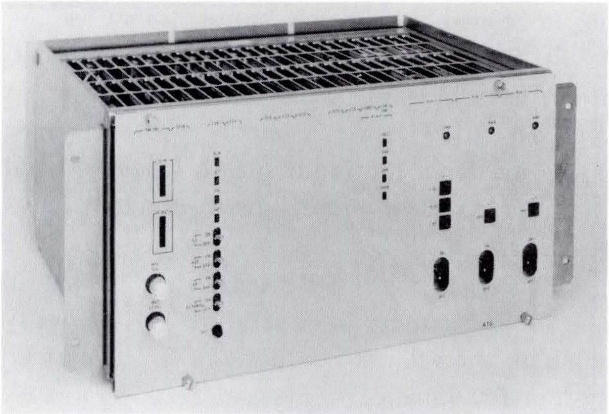


Fig. 17—Trial acoustic echo canceller.

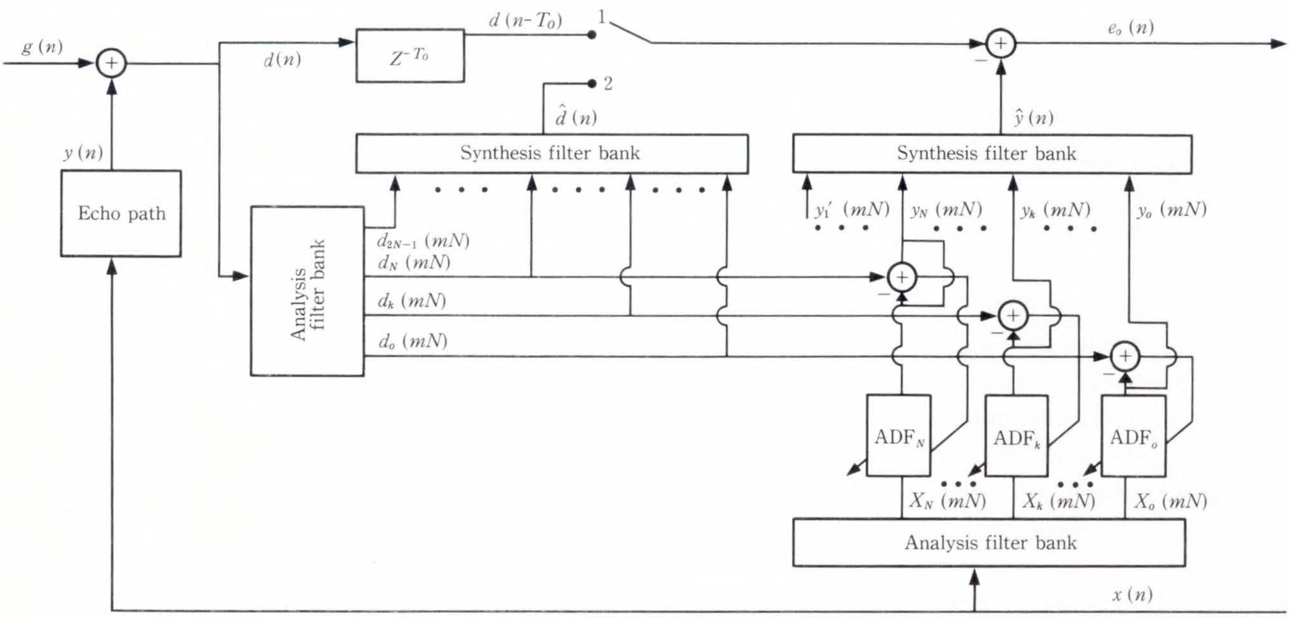



Fig. 18—Band-splitting echo canceller.

## 4. Conclusion

The major digital signal processing technologies of video and audio signals used in communications systems and digital storage media have been covered. As described, new coding technologies and dedicated ICs have been developed for use in new transmission systems such as ISDN and B- ISDN, and to provide new services such as TV conferencing and digital cellular mobile systems.

Several applications of up to 1.5 or 2 Mb/s are becoming available on ISDN, but ISDN itself is not yet widespread. Many other useful services may appear with the proliferation of ISDN, and it will be important to develop new coding algorithms, processing devices, and related technology in order to provide such services to customers at a reasonable price.

### References


1) Tsuda, T., and Hiraoka, M.: Efficient Video Bandwidth Compression. *FUJITSU Sci. Tech. J.*, **22**, 4, pp. 355-366 (1986).
2) Kawai, O., Okazaki, T., Matsuda, K., Hosokawa, T., and Yamanaka, T.: A Compact 100-160 Mb/s HDTV DPCM-AQ Codec. Proc. GLOBECOM'89, pp. 236-241.
3) Konoshima, M., Kawai, O., and Matsuda, K.: A Study of Coding Scheme for Digital Storage Media. (in Japanese), Proc. PCSJ'89, pp. 73-74.
4) Morimatsu, E., Itoh, T., Matsuda, K., Nishizawa, Y., Takenaka, Y., and Narahira, S.: An Experimental Video Codec Based on CCITT H.261. Proc. PCS'91, pp. 117-120.
5) Kawai, O., Matsuda, K., Itoh, A., and Narahira, S.: A Development of a Video Signal Processor LSI VSP-1. (in Japanese), Proc. SITA'89, pp. 257-262.
6) Kosugi, Y., Matsuda, K., Takenaka, Y., Itoh, A., Ishihara, T., and Nagasawa, T.: A Development of Discrete Cosine Transform VLSI. (in Japanese), 1990 Spring IEICE Natl. Conv. Rec., D-400 pp. 7.152.
7) Itoh, T., Oono, M., Matsuda, K., Miyasaka, H., Terui, Y., Fukuda, K. and Tanaka, S.: Development of CIF Conversion and Frame Memory Control LSIs. (in Japanese), 1990 Autumn IEICE Natl. Conv. Rec., D-277, pp. 6.279.
8) Hamano, T., Sakai, K., and Matsuda, K.: Cell-Loss Compensation in Variable Bit-Rate Video Coding. Proc. PCS '91, pp. 35-38.
9) Taka, M., Suzuki, J., Shimoyama, H., and Tomita, Y.: 64 kbps High Quality Speech CODEC for Video Conference. (in Japanese), *IEICE Tech Rep.*, CS83-196, pp. 77-84 (1984).
10) Tomita, Y., Unagami, S., Kajiwara, M., and Matsumura, T.: An Implementation of the 64 kbps Audio Codec Based on CCITT Recommendation G.722. Proc. GLOBECOM'87, pp. 463-467.
11) Kurihara, H., Mase, H., Yamada, H., Yamashita, H., and Amano, F.: An Implementation of Audio Codec for ATM Networks. (in Japanese), 1990 Autumn IECIE Natl. Conv. Rec., B-628, pp. 3.290.
12) Unagami, S., and Iseda, K.: DSP Application for Audio Signal Compression/Decompression. *JAS Journal*, **30**, 10, pp. 18-25 (1990).
13) Satoh, K., Tomita, Y., Unagami, S., Kurihara, H., and Kajihara, M.: An Implementation on a Single Chip APC-AB Voice Codec Using an Application Specificed LSI. (in Japanese), 1989 Spring IEICE Natl. Conv. Rec., A-10, pp. 1.10.
14) Amano, F., Iseda, K., Okazaki, K., and Unagami, S.: An 8 kbps TC-MQ (Time domain Compression ADPCM-MQ) Speech Codec. Proc. ICASSP'88, pp. 259-262.
15) Schroeder, M.R., and Atal, B.S.: Code-Excited Linear Prediction (CELP): High Quality Speech at Very Low Bit Rates. Proc. ICASSP'85, pp. 937-940.
16) Wakao, M., and Oguchi, M.: Activity of the Codec Algorithm Selection for the Japanese Digital Cellular System. (in Japanese), *IEICE Tech. Rep.*, RCS90-25, pp. 27-34 (1990).
17) Gerson, I.: Vector Sum Excited Linear Prediction (VSELP) Speech Coding for Japan Digital Cellular. *IEICE Tech. Rep.*, RCS90-26, pp. 35-40 (1990).
18) Mclaughlin, M.: Channel Coding for Digital Speech Transmission in the Japanese Digital Cellular System. *IEICE Tech. Rep.*, RCS90-27, pp. 41-45 (1990).
19) Kobayashi, N., Gambe, H., and Aoki, K.: An Adaptive Transversal Filter VLSI. Proc. ICASSP'86, pp. 1529-1532.
20) Yasukawa, H., Furukawa, I., and Ishiyama, Y.: Characteristics of Acoustic Echo Canceler Using Sub-band Sampling and Decorrelation Methods. *Electr. Lett.*, **24**, 16, pp. 1039-1040 (1988).
21) Sakai, Y., Amano, F., Unagami, S., Ito, A., Chujo, K., and Abe, S.: Hardware Implementation of Echo Canceler by Using MB86232. (in Japanese), Proc. IEICE Symp. DSP, 1988, pp. A4-4.
22) Asharif, M. R., Amano, F., Unagami, S., and Murano, K.: Acoustic Echo Canceler Based on

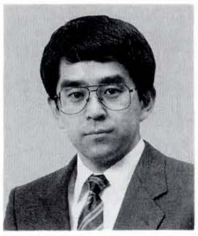
Frequency Bin Adaptive Filtering (FBAF). Proc. GLOBECOM'87, pp. 1940-1944.

23) Perez, H., and Amano, F.: Acoustic Echo Cancelation Using Multirate Techniques. *IEICE Trans.*, **E74**, 11, pp. 3559-3568 (1991).

24) Amano, F., and Perez, H.: A New Subband Echo Canceler Structure. Proc. ICASSP'91, pp. 3585-3588.

**Kiichi Matsuda**

Broadband Communication Systems Laboratory
FUJITSU LABORATORIES, KAWASAKI
Bachelor of Electrical Communication
Waseda University 1972
Master of Electrical Eng.
Waseda University 1974
Specializing in Video and Audio Signal Processing



**Yoshitsugu Nishizawa**

Video Communication Dept.
Business Communication Div.
FUJITSU LIMITED
Bachelor of Electrical Eng.
Nagoya University 1978
Master of Electrical and Electronic Eng.
Nagoya University 1980
Specializing in Video Coding System

**Fumio Amano**

Radio and Satellite Communication Systems Laboratory
FUJITSU LABORATORIES, KAWASAKI
Bachelor of Electronic and Electrical Eng.
Tokyo Institute of Technology 1975
Master of Electronic and Electrical Eng.
Tokyo Institute of Technology 1977
Specializing in Digital Signal Processing of Speech Signal

UDC 621.395.74:681.3.06

# Globalization of Software Development for Reliable Telecommunications Systems

● Tadamichi Suzuki ● Katsuyoshi Konishi ● Kiyoh Nakamura

*(Manuscript received January 14, 1992)*

The users of a globalized communication network service have diverse needs. Meeting these needs is the single most important requirement for the success of globalized software development.

This paper discusses methods of developing reliable software on an international scale for telecommunications systems such as switching systems.

Software development support functions such as Computer Aided Software Engineering (CASE) technologies have been used to assure high reliability of telecommunications systems. These functions need to be further developed and made more available to users and programmers around the world.

## 1. Introduction

The 1990s will see increasingly diverse demands being made on international communication network services[1]. Also, the embedded software for advanced communication-information systems will become far more complex, and the success of these systems will greatly depend on the globalization of software development.

CASE was developed to increase software quality and productivity. However, the current CASE cannot assure that integrated technology is suitable for globally distributed and concurrent development.

## 2. Key aspects for globally distributed software development

High-level communication network services of intelligent networks link not only computers and communication networks, but also individuals and organizations around the world. Therefore, new methods must be developed to enable the production of high-quality software that takes cultural and linguistic differences into account.

The basic requirements for the dispersion and concurrent development of software on a global scale will be an understanding of client needs; mutual exchange of expertise and high-reliability software; and a high efficiency in areas such as design, test, maintenance, and control. How can these requirements be fulfilled?

1) Distributed process management

For certain aspects of a process it may be advantageous to introduce flexibility and ambiguity into the definition of a job. To compensate for differences in the software development styles of different cultures, the reliability of software must be assured. In particular, great care is needed in multi-stepwise refinement at the design stage to ensure that the design specifications are appropriate and that information is correctly inherited from the previous process.

2) Multi-lingual management

The many design languages and human languages in use around the world make globalization a difficult task. One way to overcome this difficulty is to unify views using graphical representations. Test and design must be consistent regardless of the language in which it is done.

3) Organizational management

For good organizational management, it is important to accumulate and transfer expertise to overcome the international differences in relationships between groups and individuals. To achieve this, it is essential to reuse informal information.

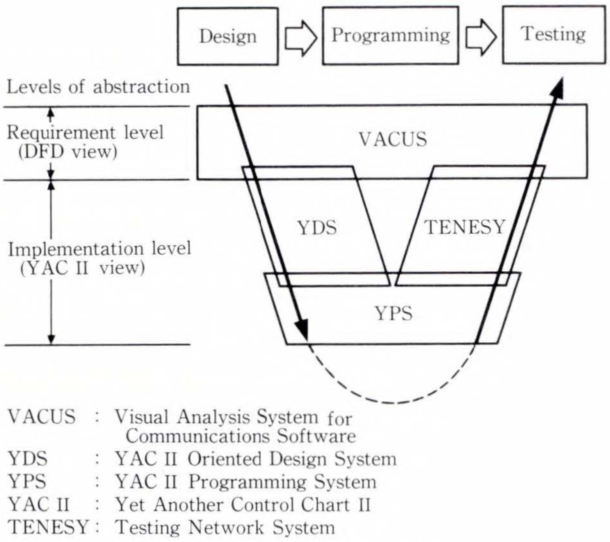



Fig. 1–CASE approach.

## 3. Practical approach for reliability and productivity

This chapter looks at the following key aspects of the globalization of software development: the development environment, management, and the infrastructure.

### 3.1 Software development environment

The development environment must be unified, and a consistent development process must be supported. Figure 1 shows the CASE approach to software development support systems using wide area networks consisting of workstations and a host computer.

1) Integration of software tools

Software tools must support consistency in

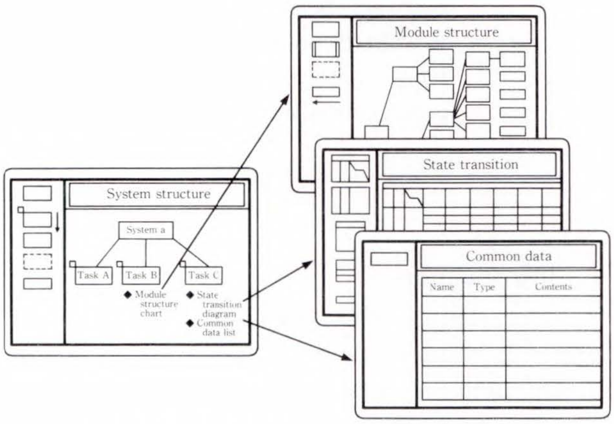



Fig. 2–Automatic guidance by YDS.

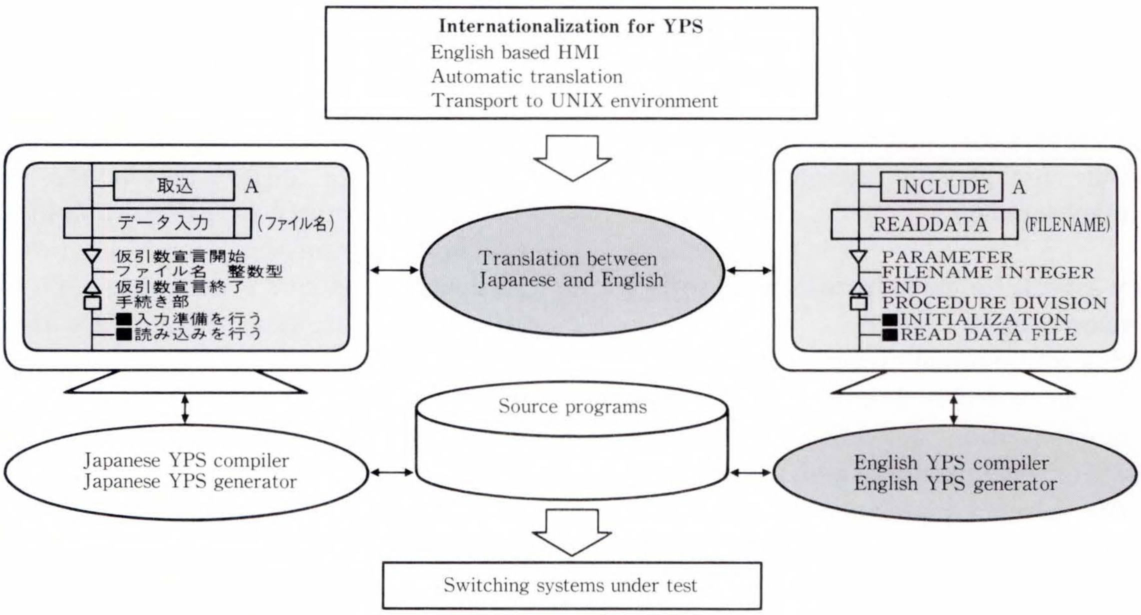



Fig. 3–Globalization in design language.

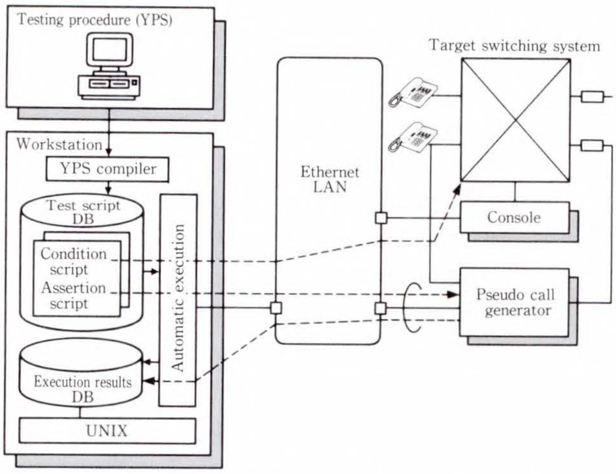



Fig. 4—Automation of system testing.

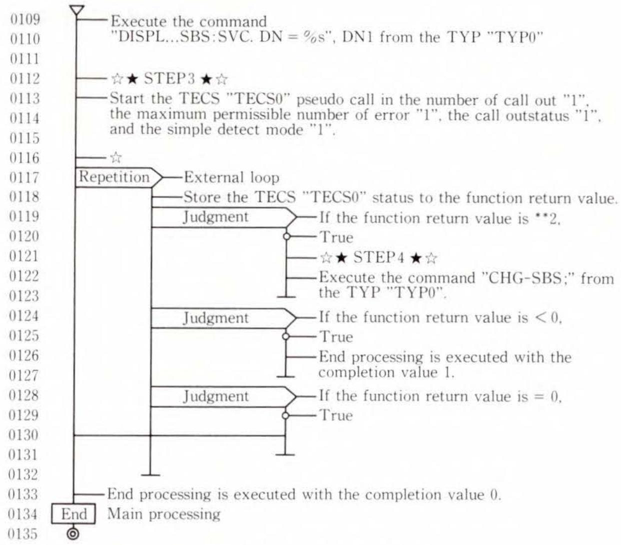



Fig. 5—Test script.

both design and testing according to the requirements definition.

To minimize the effect of language differences, text is converted to graphics. Multi-lingual management can be enhanced by the use of the visualization function[2] for test and design. The use of two-dimensional charts in all processes assures that the software is always viewed in the same way independent of development skills. This function significantly improves distributed process management in globalized software development.

2) Design support

The YAC II Oriented Design System (YDS, YAC II: Yet Another Control Chart II)[3] has a guidance function that helps prevent description errors in design documents. Also, YDS incorporates logic verification into each process, for example, design forms to be described are displayed automatically and the attributes are then checked. This function is useful for distributed process management. Moreover, when readability is improved, designers are better able to reuse software resources (see Fig. 2).

The automatic programming function of the already developed YAC II Programming System (YPS)[4] enables the coding process to be omitted without the introduction of errors. Input using this function is done using a pseudo-natural language and two-dimensional charts. The first steps (e.g. multi-lingual management) in the globalization approach can be made using a graphic specification language and automatic translation between Japanese and English as shown in Fig. 3.

3) Testing support

Switching system functions must be tested in an operational environment before shipment. Globalization requires the following:

i) Improvement of software quality before the systems are tested at installation.

ii) Increased efficiency of system testing.

To meet the first requirement, existing testing rules are used for logical verification. Our Switching Software Verification Expert System (SVEX)[5] performs logical verification automatically by simulating the original switching program according to these testing rules.

For the second requirement, test scripts are transferred to machines via the network for remote and automatic testing[6] (see Fig. 4). The test scripts can be used repeatedly for regression testing (see Fig. 5). This testing method is especially valuable for distributed process management. The language of these test scripts is based on YAC II. Because YAC II is also the language used for design, this choice of language makes multi-lingual management easier.

### 3.2 Software quality management

It is particularly important to establish management methods for high-quality software development. The improvement of software quality by using historical data should be promoted for total development control and standardization.

1) Software engineering norm

Fujitsu took more than ten years to establish its Software Engineering Norm for Telecommunications Software Development (SEN). This norm is a documented software development methodology that will ensure a high quality of software in the international age. However, great care must be taken when applying the norm in different cultures because it can be applied in various ways. For good distributed process management and organizational management, the norm should be compiled with as much flexibility and cooperation as possible. Therefore, local rules must be established before the norm is applied.

2) Software management system

For many years Fujitsu's Software Management System for Telecommunications Software Development (SMS)[2] has been controlling the progress of development and resources at distributed development points.

Databases enable input to be equally accessible from any development point for distributed process management, and enable data to be visualized for multi-lingual management.

To realize practical reuse of software[7], other technological breakthroughs are needed. Software reuse means the common ownership of "assets"[8]. Software reuse deals with expertise and informal information, which are not reused as much as cataloged program modules. We believe that software is an asset only if it can be profitably reused. Software asset management is a key issue in global software reuse. Therefore, better methods of organizational management have been developed to improve software asset management.

3) Software QC program

To achieve globalization, the quality of feedback must be improved, and the quality of the overall system must be distinguished from the quality of individual software components. In the area of distributed process management, we have built and are operating a software quality control system called the Quality Assurance of Communications Software (QACE). This system controls the entire company activity and provides a highly reliable development control system and a quality assurance staff organization.

A program called AYUMI[9] has been developed to enhance quality assurance. AYUMI is a program similar to the AYUMI reports used in Japanese elementary schools. AYUMI is a high-performance program with three feedback loops: process evaluation loops within a division of software development, incoming inspection loops prepared by quality assurance staff, and periodic quality control meeting loops. AYUMI evaluates in five stages and 55 items for each feedback loop. AYUMI is small group activity, so developers can use it to develop high quality software (see Fig. 6).

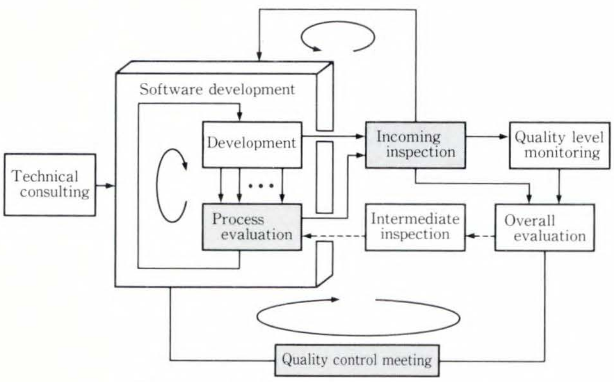



Fig. 6—Quality assurance system.

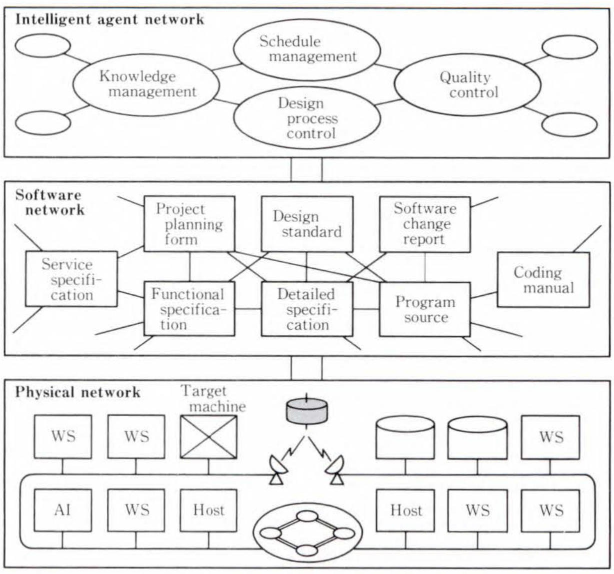



Fig. 7—Intelligent software network.

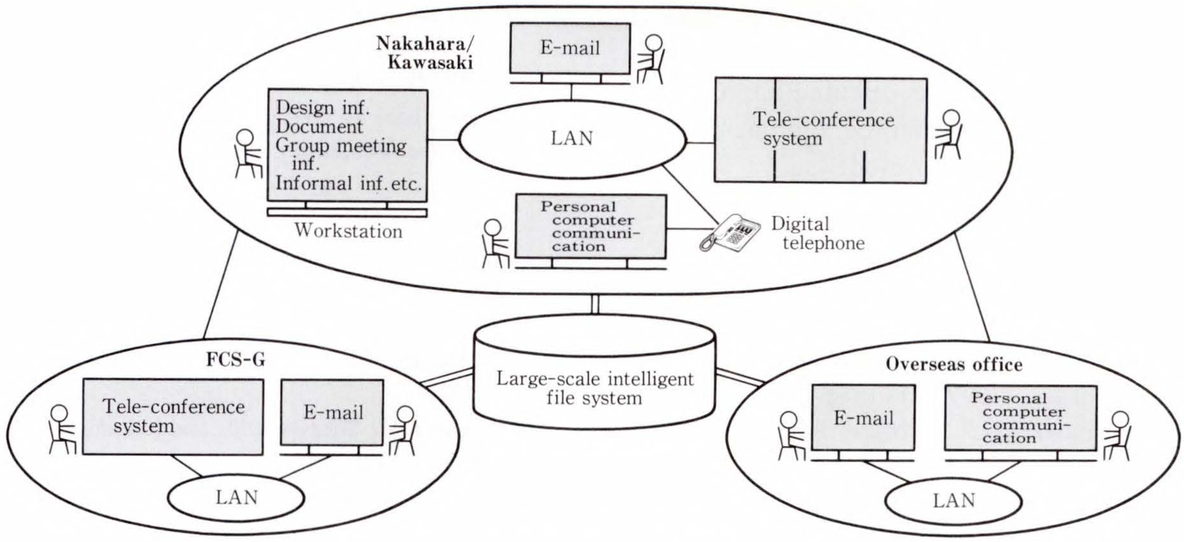



Fig. 8–Expansion of communication channels.

### 3.3 Infrastructure for globally distributed software development

Globalization can overcome the problems in advanced communication-information systems that are due to differences in local time and location. We are striving to overcome the problems due to local time by connecting development points using multiple communication channels. Some typical examples of infrastructure, environmental framework, and work style are described below.

1) Intelligent Software Network[10)]

Figure 7 shows the concept of "Intelligent Software Network" (ISN). ISN forms the framework of a cooperative distributed development environment for the most important tasks of organizational management. ISN is composed of the following three layers:

i) The physical network,
ii) the software network containing the design information required for cooperative activities, and
iii) an intelligent agent network that gives clear instructions to project managers, quality controllers, software design consultants, etc.

The physical network supports information exchange among international development points via high-speed local area networks or global broadband networks based on ATM. The software network provides software development teams with information on the relational networks of various software assets (e.g. software specifications, design manuals, and program sources) for a specific software development team. Expert information will also be managed in this network for easy circulation. The intelligent agent network supports collaborative work and the administration of distributed process management. Intelligent agents such as project managers make suggestions to software engineers according to guidelines.

2) Teleworking

Problems caused by the differences between collective work styles and individual work styles can be resolved by introducing flexible working conditions, for example, teleworking[11)]. In teleworking, people work at home instead of at the office. When properly managed, teleworking can encourage personnel to work at their best. To make teleworking a feasible proposition, a common graphic language, verification of each process, and standardization are required. Based on the concept of ISN described above, communication channels have just been expanded to form the infrastructure for globalization among head offices, subsidiaries, and overseas offices (see Fig. 8).

## 4. Conclusion

Software development is ultimately dependent on people. The globalization of high-reliability software development will require a virtual work space that mimicks the "face to face" environment. Such a work space should be developed using a human-oriented approach (as opposed to a product-oriented approach) that is based on human behavior modeling.

### References


1) Hass, R.J., and Robrock, R.B.: Intelligent Network of the Future. Proc. GLOBECOM '86, pp. 1311-1315.
2) Nakamura, K., Fujimoto, H., Suzuki, T., Tarui, Y., and Kiyokane, Y.: Visual Programming Environment in Communications Software. Proc. GLOBECOM '86, pp. 435-439.
3) Fujimoto, H., Toyoshima, Y., Nishiyama, Y., and Fukao, I.: Design Support Environment for Communications Software. (in Japanese), *Trans. IPSJ*, **31**, 7, pp.1113-1122 (1990).
4) Mikami, J., and Higashimura, A.: YPS : YACII Programming System. (in Japanese), *FUJITSU*, **39**, 1, pp. 29-35 (1988).
5) Yamazaki, J., Miyazaki, K., and Suzuki, T.: SVEX: Switching Program Verification Expert System. *FUJITSU Sci. Tech. J.*, **25**, 4, pp. 306-312 (1989).
6) Kodama, H., Fujii, Y., and Yamane, S.: Automated Test System for Switching Software. (in Japanese), *IEICE Tech. Rep.*, SSE90-75, pp. 19-24 (1990).
7) Prieto-Diaz, R. and Freeman, P.: Classifying Software for Reusability. *IEEE Software*, **4**, pp. 6-16 (1987).
8) Suzuki, T., Kiyokane, Y., Yamane, S., Domon, T., and Yoshio, S.: Software Assets Management and Reuse in Distributed and Concurrent Development - An Approach to Software Assets Inventory. (in Japanese), *IEICE Tech. Rep.*, SSE91-135, pp. 73-78 (1992).
9) Toyama, M., Sugawara, M., and Nakamura, K.: High-Quality Software Development System-AYUMI. *IEEE. J. Select. Areas Commun.*, **8**, 2, pp. 201-209 (1990).
10) Sugioka, R.: Telecommunications Software Development: Today and Tomorrow. Proc. GLOBECOM '88, pp. 730-734.
11) Nakamura, K., Tsuboi, J., Ito, H., and Kiyokane, Y.: A Trial of Teleworking and Telelearning-Practical Approach in the Communication Software Field. Proc. PTC'92, pp. 730-736.



**Tadamichi Suzuki**
Engineering Dept.
Communications Software Development Div.
FUJITSU LIMITED
Bachelor of Electronic Eng.
Tohoku University 1973
Specializing in Telecommunications Software Development Technology



**Katsuyoshi Konishi**
Communications Software Development Div.
FUJITSU LIMITED
Bachelor of Applied Mathematics and Physics
Kyoto University 1967
Specializing in Telecommunications Software Development Technology



**Kiyoh Nakamura**
Communications Software Development Div.
FUJITSU LIMITED
Bachelor of Electrical Communications Eng.
Waseda University 1964
Specializing in Telecommunications Software Development Technology

UDC 621.382:621.395.4

# Integrated Circuits for Digital Transmission Systems

• Norio Ueno • Takashi Touge • Kazuo Yamaguchi

*(Manuscript received January 28, 1992)*

Integrated circuits will be a key element in the next generation of digital transmission systems. These systems must be fast, flexible, compact, cost effective, and highly reliable. Also, the power efficiency of these systems must be maximized to compensate for the increased power consumption and consequent increase in heat output that are associated with an increase in transmission speed. One of the most effective ways to achieve such a system is to use advanced integrated circuit technology.

This paper looks at Fujitsu's latest integrated circuits for digital transmission systems, focusing on N-ISDN/B-ISDN transmission LSIs and gigabit optical regenerator ICs.

## 1. Introduction

After a period of vigorous research and development, the 64 kb/s Narrowband ISDN (N-ISDN) is now being put to practical use. The number of ISDN users is rapidly increasing throughout the world, and N-ISDN is expected to continue to develop and expand. Furthermore, there is now a great increase in research and development of network and equipment architectures and device technologies for Broadband ISDN (B-ISDN). These efforts will enable high-quality motion video and ultra high-speed data services.

For a world-wide extension of N-ISDN to suceed, it must be economical. Furthermore, the digital transmission systems for B-ISDN having 156 Mb/s User Network Interfaces (UNIs) must be not only fast and flexible but also compact, cost effective, and highly reliable. High-speed, very large scale, and low-power integrated circuit technologies are essential for the achievement of such systems. In response to this situation, Fujitsu have developed LSIs for N-ISDN/B-ISDN digital transmission systems and gigabit optical regenerator ICs for inter-and intra-office optical interfaces of transmission systems.

Section 2.1 of this paper looks at some of the latest transmission LSIs including Synchronous Digital Hierarchy (SDH)/Asynchronous Transfer Mode (ATM) transmission LSIs for B-ISDN and interface LSIs for N-ISDN. Section 2.2 looks at signal processing LSIs for image and voice signals. Sections 3.1 and 3.2 look at optical regenerator ICs, focusing on ICs for gigabit optical transmission.

## 2. LSIs for digital transmission systems

This chapter looks at LSIs for ISDN transmission and signal processing.

### 2.1 ISDN transmission LSIs

This section looks at high-speed digital LSIs for SDH/ATM transmission on B-ISDN and interface LSIs for digital subscriber loop transmission on N-ISDN.

#### 2.1.1 High-speed digital LSIs for SDH/ATM transmission on broadband ISDN

The development of evolution strategies for B-ISDN can now be accelerated in CCITT on the Network Node Interface (NNI) and User Network Interface (UNI). Rapid progress in the area of fiber optics, electronics, and information processing now give a sound basis for economically feasible implementations of

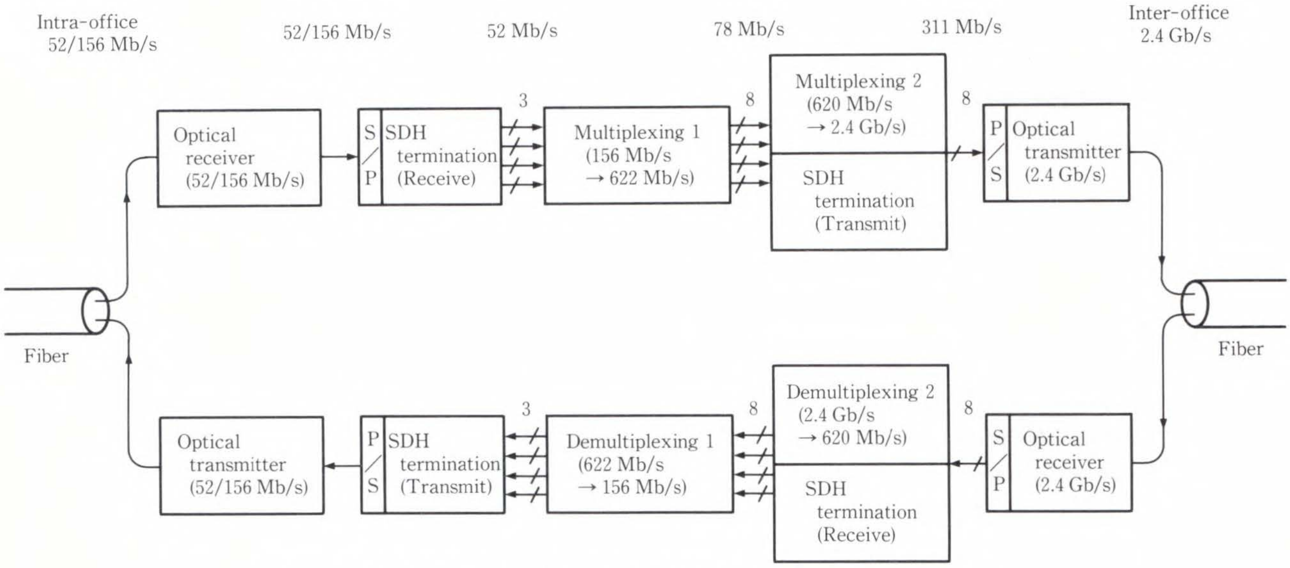



Fig. 1–Block diagram of 2.4 Gb/s SDH multiplexer/demultiplexer.

B-ISDN. In particular, new transport technologies such as SDH for NNI and ATM for UNI provide efficient and flexible means of transferring diverse information.

The introduction of SDH transmission systems are expected to form the backbone of a universal digital fiber network and to facilitate a smooth transition from existing networks to future broadband networks. Furthermore, the introduction of ATM must integrate both services and network components. A block diagram of a 2.4 Gb/s SDH multiplexer/demultiplexer is shown in Fig. 1. The SDH multiplexer/demultiplexer basically consists of SDH terminations, a synchronous multiplexer/demultiplexer, and an optical transmitter/receiver. The SDH terminations also perform pointer processing, overhead data insertion/drop, and error monitoring. The synchronous multiplexer/demultiplexer is divided into two parts, multiplexing 1/demultiplexing 1 and multiplexing 2/demultiplexing 2 with SDH termination of the 2.4 Gb/s stage.

The 2.4 Gb/s SDH multipexer/demultiplexer uses 11 newly developed LSIs. Current CMOS gate arrays are operated at 52 Mb/s or less, GaAs gate arrays at 78/311 Mb/s, and GaAs full-custom LSI chips[1] at Gb/s. The power consumption of GaAs gate arrays is 1/3 that of ECL gate arrays, which makes them a good choice for low power use. The SDH termination CMOS-LSI (transmitter) is shown in Fig. 2. The 2.4 Gb/s multiplexing/demultiplexing board with two GaAs gate arrays plus a cooling fin (multiplexing 2/demultiplexing 2 with SDH termination) is shown in Fig. 3.

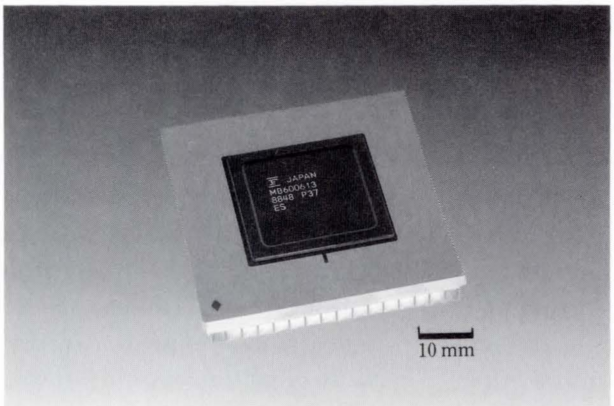



Fig. 2–SDH termination CMOS-LSI.

The maximum operating speeds of LSIs will continue to increase. For example, CMOS gate arrays made using Bi-CMOS or high-speed pure CMOS process technology will be used at 100-300 Mb/s. The maximum number of gates per LSI is also increasing rapidly, and many functions that are currently performed by several LSIs will be performed by a single LSI within a few years. For example, the current SDH multiplexer/demultiplexer uses 20 k-gate CMOS gate arrays that operate at 19 Mb/s and 5 k-gate

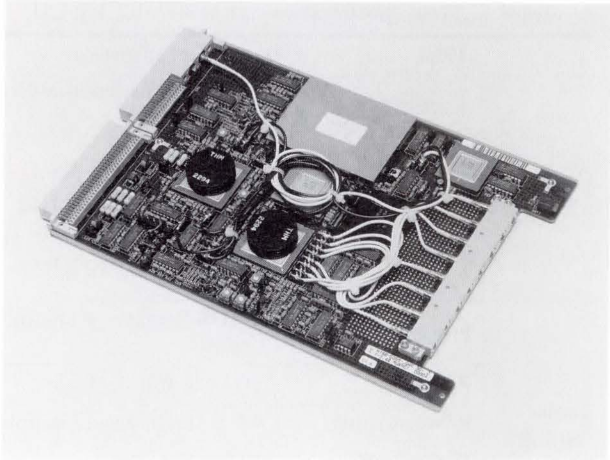


Fig. 3—2.4 Gb/s multiplexing/demultiplexing board with two GaAs gate arrays plus cooling fin. (size: 280 x 200 mm)

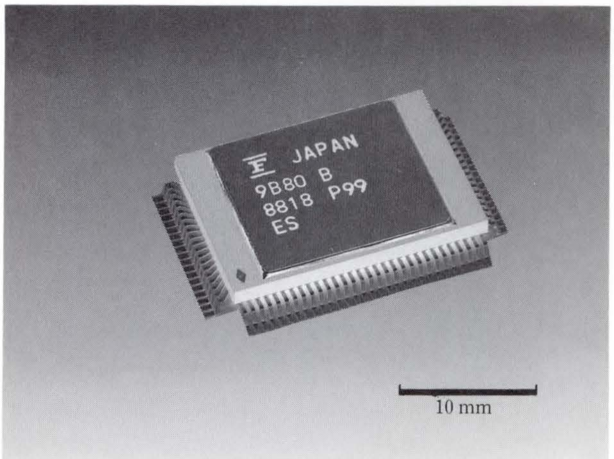



Fig. 4—16 x 8 space switch LSI.

GaAs gate arrays that operate at 311 Mb/s. However, the next generation of the ATM multiplexer/demultiplexer will use 200k-gate Bi-CMOS gate arrays that operate at 156 Mb/s and 30k-gate GaAs gate arrays that operate at 620 Mb/s.

ATM transmission systems require a broadband switch for the cross-connect and ATM multiplexer/demultiplexer. A $16 \times 8$ space switch LSI[2)] for a broadband switch and its input/output waveforms are shown in Figs. 4 and 5 respectively. The maximum operating speed of this switch is 210 Mb/s. The low power consumption of this switch (500 mW at 150 Mb/s) was realized using high-speed CMOS process technology.

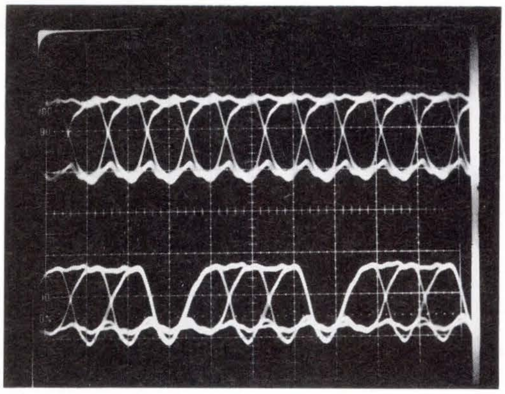



Fig. 5—Waveforms of 16 x 8 space switch LSI.

An ATM multiplexer/demultiplexer generally consists of an SDH/ATM termination, a cell multiplexer/demultiplexer, and an optical transmitter/receiver. The optical transmitter/receiver used in ATM transmission systems is the same as the one used in SDH transmission systems.

Many FIFOs (two port RAMs) are used for delay justification in the ATM multiplexer/demultiplexer. A sophisticated high-speed and low-power FIFO is therefore essential for these LSIs. Two types of FIFO have been developed: a Bi-CMOS FIFO that can operate at 809 Mb/s, and a GaAs FIFO that can operate at 200 Mb/s. For ATM transmission systems, Bi-CMOS gate arrays that operate at 156 Mb/s and GaAs gate arrays that operate at 620 Mb/s will be used. These gate arrays will include the new FIFOs. The gate scale per chip will be 5-10 times greater than that of the previously described SDH transmission LSIs[3)].

**2.1.2 Interface LSIs for ISDN**

In April 1988, NTT started its INS Net services, which in turn initiated the spread of N-ISDN. The B-ISDN described in the previous section are expected to be in wide use in the future. Meanwhile, many subscribers are already using N-ISDN. Fujitsu is promoting the development of equipment for N-ISDN, and the domestic specifications for N-ISDN are being established. To improve miniaturization and economy, we have developed CMOS standard-cell LSIs containing random logic, memory, and analog circuits on a single chip[4)]. These were developed to promote the implementation of N-ISDN.

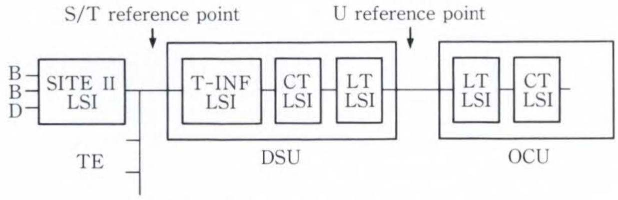



Fig. 6–U and S/T interfaces.

Table 1. Specifications for ping-pong subscriber line transmission (NTT)

| Item | Specification |
|---|---|
| Transmission | Time Compression Multiplex (TCM) |
| Capacity | 144 kb/s + 3.2 kb/s + 3.2 kb/s |
| Burst cycle | 2.5 ms |
| Line bit rate | 320 kb/s |
| Transmission Line code | AMI (6 $V_{op}$) |
| Line equalization | $\sqrt{f}$ equalization, bridged tap equalization |
| Applicable cable | In-place metal subscriber line |
| Maximum transmission distance | 7 km (0.65 $\phi$) |
| Bridged tap condition | Optional length, 2 taps |

Figure 6 shows the U and S/T interfaces for N-ISDN.

1) U interface LSIs

The U interface connects a switching system to a subscriber using in-place metal wire-pair transmission lines. The U interface is standardized by CCITT G.960/961. Table 1 gives the specifications for the ping-pong transmission used by NTT's INS Net64.

The LSI used for subscriber line transmission is called the Line Termination (LT) LSI.

i) LT LSI

For the early INS Net64, we developed an LT LSI based on CMOS digital and analog standard-cell LSI technology. This LSI was first applied to an NTT INS model in Mitaka, a Tokyo suburb[5]. We have since improved the performance and cost effectiveness of this LSI. Out latest version of this LSI has a die size of 9.0 x 7.6 mm and a power consumption of 110 mW. Table 2 gives the main specifications of the MB87592 LT LSI. This LSI provides the following functions

Table 2. Main specifications of MB87592 LT LSI

| Item | | Specification |
|---|---|---|
| Function | Transmission | Time Compression Multiplex (TCM) |
| | Line bit rate | 320 kb/s |
| | $\sqrt{f}$ equalization *gain range | 0~50 dB (at 160 kHz) |
| | suppresible echo length | 4.5 $T$ ($T$ = 1/320 KHz) |
| Technology | Process | CMOS 8 kgates + analog cells |
| | Package | 100 pin flat package |
| | Power-supply | +5 V single power supply |
| | Power consumption | 110 mW 50 mW (in power down mode) |

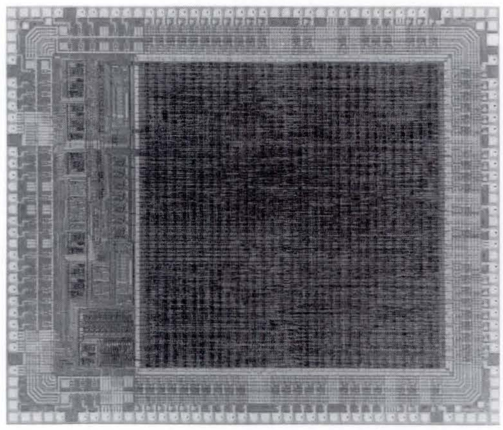


Fig. 7–LT LSI. 1 mm

for 320 kb/s subscriber line transmission:

a) Unipolar/bipolar transmission code conversion
b) Metal subscriber line equalization
c) Digital control timing extraction

The die of the LT LSI is shown in Fig. 7.

ii) CT LSI

The Circuit Termination (CT) LSI multiplexes and separates the signals received by the LT. In the early stage of development, we used gate array and processor-type LSI technology to meet a variety of specifications. Now that the specifications have been established, we are using CMOS standard-cell LSI technology to reduce the price.

2) ISDN S/T interface LSI

To reduce the price and power consumption of the ISDN (2B + D) basic interface, we have developed LSI sets which provide S/T reference points. The functions of these LSIs are explained below.

i) SITE II

S Interface Terminal Equipment (SITE) supports ISDN basic user-network interfaces that conform to CCITT I.430, Q.920, and Q.921. The MB87588 SITE II LSI has an interface and driver/receiver for the S interface that conforms to I.430. The die of SITE II is shown in Fig. 8. SITE II has the following functions:

a) Layer-1 status management
b) Primitive interface
c) Automatic check of Terminal Equipment Identifier (TEI)
d) Inclusion of R Interface (RI) random generator for TEI automatic assignment
e) Multiframe data transmission by CPU interface
f) Driver/receiver for S interface

To reduce its price, SITE II is made using Fujitsu's current CMOS digital and analog standard-cell technology. The maximum operating frequency of SITE II is 6.114 MHz, and the power consumption is only 80 mW. only 80 mW.

ii) T-INF

T-INF is a full-duplex four-wire driver/receiver LSI used in the ISDN basic user-network interface that conforms to CCITT I.430 (NT1). T-INF is used in the Data Service Unit (DSU), and interfaces the 2B + D (64 + 64 + 16 kb/s) signal to the CT LSI.

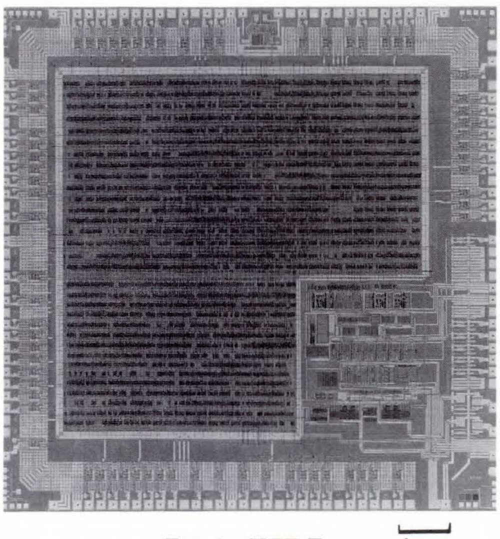



Fig. 8—SITE II.

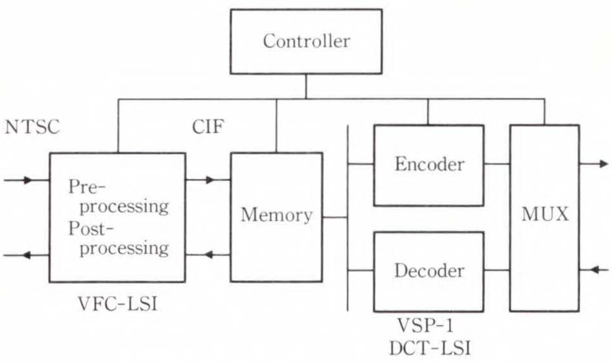



Fig. 9—Image encoder/decoder.

## 2.2 Signal processing LSIs

This section describes the signal processing LSIs. These LSIs enable effective transfer of image and voice signals.

### 2.2.1 Image signal processing

Because image signals require a larger bandwidth than voice signals, image signal communication has not come into wide use. However, high-compression image signal processing and wide-band digital signal transmission have now made high-quality and economical image communication possible. The LSIs described here enable moving image signal processing conforming to the CCITT standards for video-conferencing telephone image coding.

Our image encoder/decoder is used for moving image signal transmission conforming to CCITT H.261[6),7)] (see Fig. 9). The key to achieving variable transfer rates from 64 kb/s to 2 Mb/s in a single CODEC is flexibility. Custom LSIs are used to achieve this flexibility using a minimum hardware configuration.

Fujitsu's latest Sea-of-Gates (SOG) CMOS gate array technology is used to fabricate large-scale logic LSIs with built-in ROM and RAM. We have developed the following chip set:

1) Video Signal Processor-1 (VSP-1)[8)]

The VSP-1 is a programmable image encoder/decoder LSI (see Fig. 10). It has an operation word length of 16bits and an instruction cycle of 70 ns. The absolute value operation

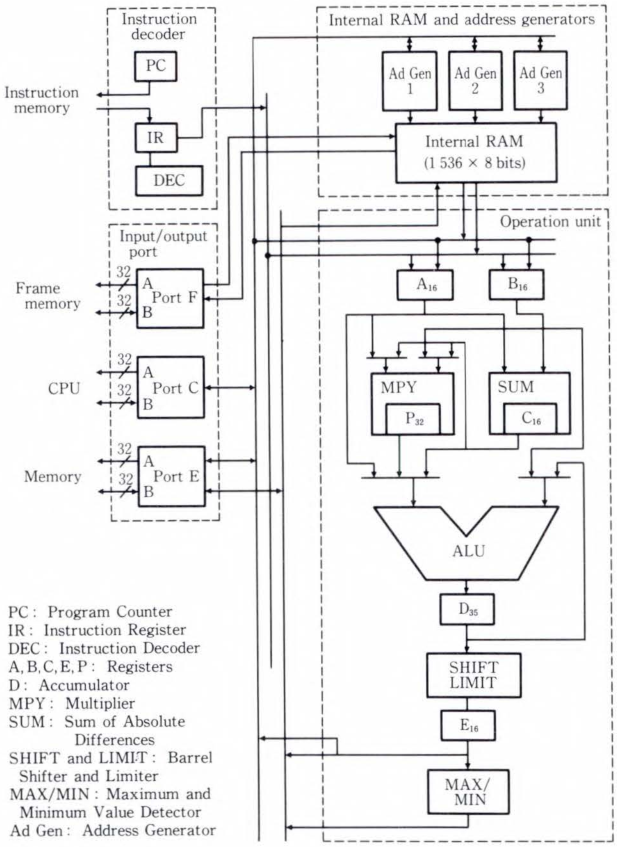



Fig. 10—Block diagram of VSP-1.

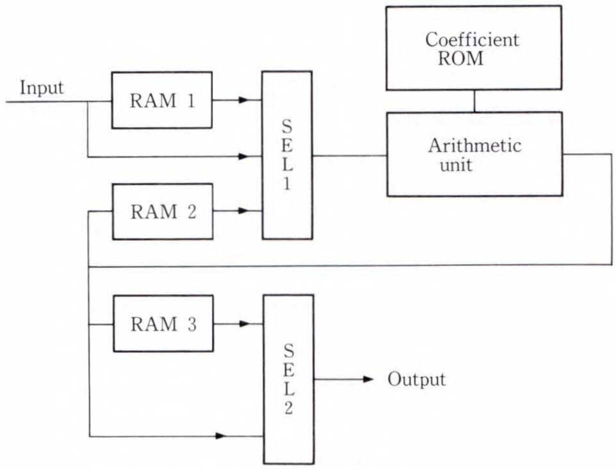



Fig. 11—DCT-LSI.

unit, Arithmetic Logic Unit (ALU), and maximum/minimum value detector have a four-stage pipeline structure. Three system input/output ports are used to improve throughout. The image data input/output port has Direct Memory Access (DMA) functions to transfer data blocks between the port and memory at a maximum rate of 54 Mbyte/s.

2) Discrete Cosine Transform-LSI (DCT-LSI)[9]

The DCT-LSI performs parallel high-speed discrete cosine transforms. The size of DCT operation can be 2 x 2, 4 x 4, 8 x 8, or 16 x 16 pixels. Because of the variable data size, this LSI can be used for both still and moving image processing. The matrix operation unit performs high-speed processing using eight parallel processing circuits.

After two-dimensional cosine transform, a zigzag scanning is applied to transfer coefficient data of moving image signal. Each RAM block can be used for forward/backward conversion. Figure 11 shows the block diagram of the DCT-LSI.

3) Video Format Converter LSI (VFC-LSI)[10]

The VFC-LSI converts between conventional National Television System Committee (NTSC) image signals and Common Intermediate Format (CIF) or Quarter CIF (QCIF) signals. The VFC-LSI performs this conversion by preprocessing or postprocessing the VSP-1 input or output (see Fig. 9).

The VFC-LSI's main function is to convert the number of scanning lines while maintaining image resolution using a transverse filter. The VFC-LSI also has a multiplex/demultiplex function for time division processing of the Y, R-Y, and B-Y components of a composite signal. This LSI can be used with NTSC and the European Phase Alternation Line (PAL). This is achieved by changing the timing at which data and coefficients are input to the filter.

Using a frame memory control LSI, and a CCITT H.221 based data multiplexing LSI in addition to the LSIs above reduces the equipment size. More and more conventional circuits will be replaced with LSIs to further downsize the video signal processing equipment.

### 2.2.2 Voice signal processing

The technology for digital signal processing LSI has grown with the requirement for better transmission efficiency. In the early 1980s, Fujitsu developed the MB8764 FDSP-3 Digital Signal Processor (DSP) for voice and audio signals. This LSI uses 16-bit fixed-point

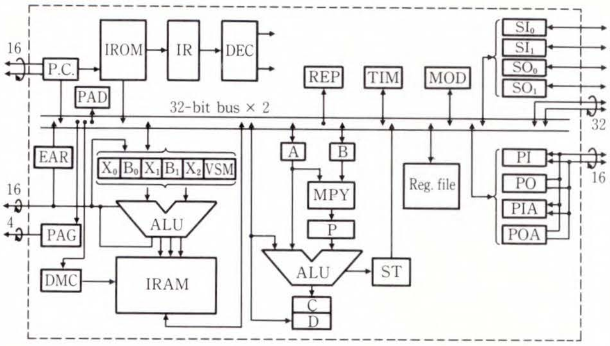



Fig. 12—Block diagram of FDSP-4.

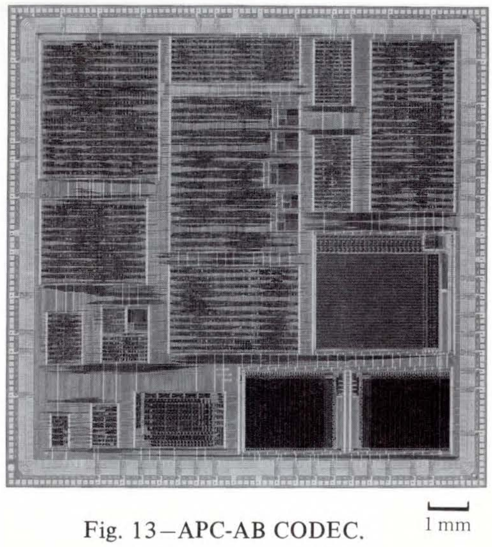



Fig. 13—APC-AB CODEC.

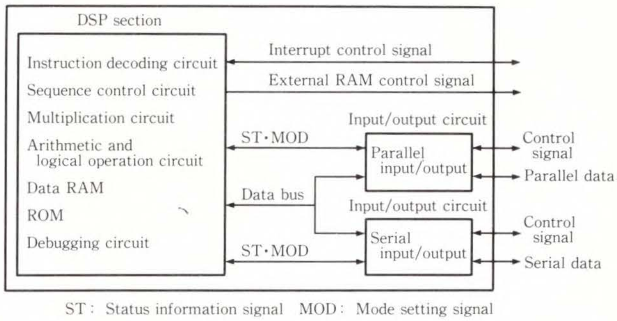



Fig. 14—Concept of MB86220.

arithmetic. In the latter half of the 1980s, we developed a high-performance general-purpose DSP, the MB86232 FDSP-4, which uses 32-bit floating-point arithmetic. Figure 12 shows the block diagram of the FDSP-4[11].

The following problems may arise when a general-purpose DSP is used:

1) The DSP may have unnecessary functions or a performance that exceeds the basic requirements.
2) The DSP may not be able to handle the processing.
3) Peripheral circuits may be required.

When constructing a system using a general-purpose DSP, there will be only a few problems in small-scale applications. but the price will become a major problem in large-scale applications. Therefore, Fujitsu developed the FDSP-4, which is a basic DSP having the standard architecture used by the customized ASIC DSPs that were developed later.

ASIC DSPs are low cost and low power devices. For digital standard-cell LSI, cell libraries have been enhanced and automatic wiring techniques greatly improved. These advances have enabled significant reductions in the design period and development costs of large-scale ASIC DSPs, which contain hundreds of thousands of gates. Typical ASIC DSPs are described briefly in the following section.

1) MB87528 Adaptive Predictive Coding with Adaptive Bit allocation (APC-AB) CODEC

The APC-AB method assigns adaptive bits to parameters such as residual power and pitch frequency for a 64 kb/s PCM signal, and compresses the bandwidth to 16 kb/s or 8 kb/s by predictive quantizing. The data word length is reduced to 24 bits, which is 8 bits less than the word length of the basic DSP. For optimization, the ROM is expanded and the RAM is minimized. Figure 13 shows the die of this CODEC.

2) MB86220 series ASIC DSP core

The MB86220 is a 24-bit floating-point DSP that includes a user-defined specifications area. Figure 14 shows the concept of the MB86220. The MB86220 consists of the following:

i) An instruction decoding circuit, sequence control circuit, multiplication circuit, arithmetic and logical operation circuit, data RAM and ROM, and
ii) a customized input-output circuit.

This IC can be applied to audio signal processing such as low bit rate encoding, and image processing on personal computers. Figure 15 shows the die of this circuit.

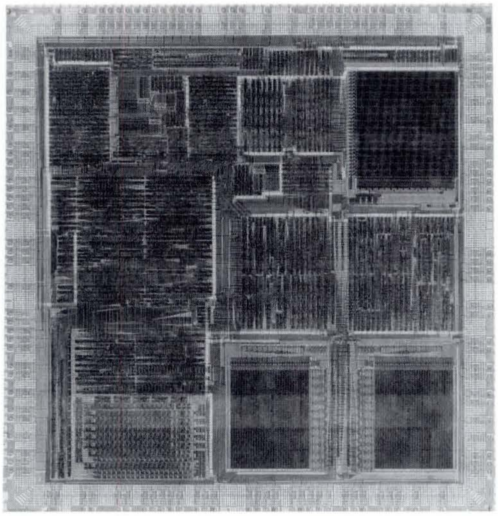


Fig. 15–MB86220.

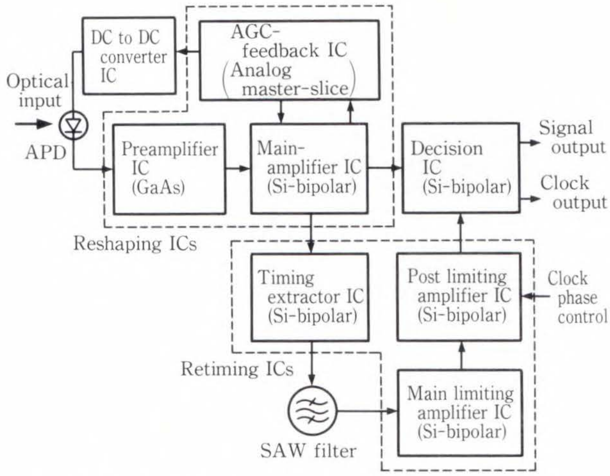



Fig. 16–Block diagram of 2.4 Gb/s optical receiver.

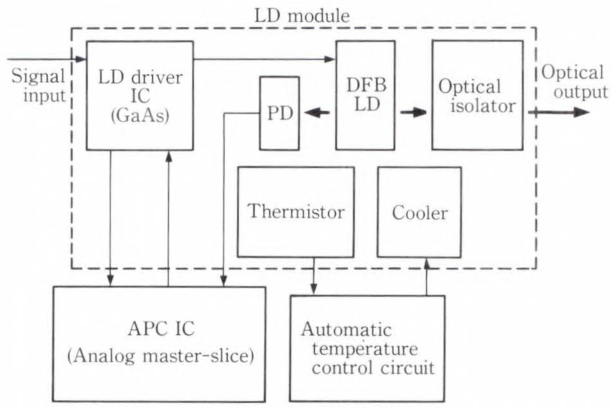



Fig. 17–Block diagram of 2.4 Gb/s optical transmitter.

## 3. Optical regenerator ICs

This chapter describes some optical regenerator ICs for gigabit digital transmission systems.

### 3.1 2.4 Gb/s regnerator ICs

Regenerator ICs for 1.544 Mb/s to 1.6 Gb/s communication systems based on Si-bipolar technology have already been developed[12]. Also, in 1990, regenerator ICs for 2.4 Gb/s communication systems were developed[13]. The repeater spans of a long-haul optical communication system cannot easily be reduced to compensate for an increase in the transmission rate. Therefore, 2.4 Gb/s optical communication systems require a high-sensitivity receiver, a high-power output transmitter, and high performance ICs with higher speed. The block diagrams of a 2.4 Gb/s optical receiver and transmitter that use regenerator ICs are shown in Figs. 16 and 17. The post limiting amplifier IC in the retiming section has a clock phase shifter that can adjust the retiming clock phase of received data in the decision circuit without adjusting the cable or strip line length. For the receiver and transmitter, nine regenerator ICs were developed. Five of these ICs are made using an ultra high-speed Si-bipolar process with transistors having a unity gain bandwidth of 13-16 GHz. Two are made using a high-transconductance GaAs MESFET process with a 0.8 $\mu$m gate length. The last two are made using a Si-bipolar analog master-slice for AGC/APC control ICs.

Table 3. Main specifications of 2.4 Gb/s regenerator ICs

| Item | | Specification |
|---|---|---|
| Receiver | | Receiver sensitivity $< -29$ dBm. |
| | Reshaping ICs | Bandwidth $>$ 2 GHz |
| | | Transimpedance gain $>$ 95 dB$\Omega$ |
| | | Equivalent input noise current $< 5$ pA/$\sqrt{\text{Hz}}$ |
| | Retiming ICs | Bandwidth $>$ 3 GHz |
| | | Gain $>$ 60 dB (at 2.4 GHz) |
| Transmitter | | Optical output power $>$ +3 dBm |
| | LD driver IC | Operation speed $>$ 2.4 Gb/s |
| | | Maximum driving current $>$ 80 mAp-p. |

The main specifications of these ICs are shown in Table 3. For high-sensitivity, the preamplifier chip is made using the GaAs process. The equivalent input noise current of 5 pA/$\sqrt{\text{Hz}}$ is 1/3 of the figure achieved using the bipolar process. The LD driver is also made using the GaAs process in order to achieve a large driving pulse current and bias current. The maximum driving pulse current of 80 mA is twice as large as that achieved using the bipolar process.

The design of the new multi-chip modules for the transmitter and reciever pay special attention to high frequency impedance matching, reduction of the internal resistance of the grounds and power lines, and the crosstalk characteristics. One or two ICs and associated capacitors, resistors, and optical components (i.e. Avalanche Photodiode (APD), LD, and optical isolator) are housed in a single module.

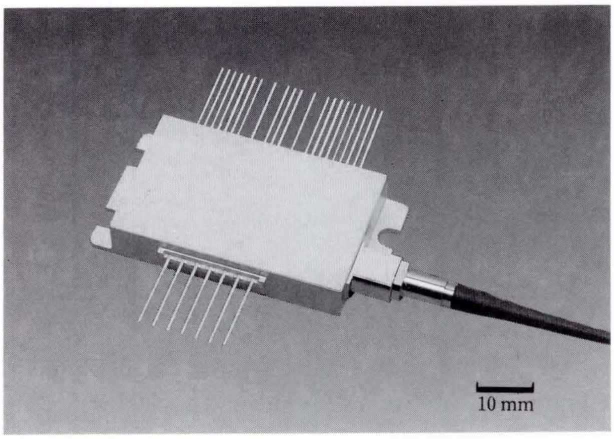



Fig. 18–LD module.

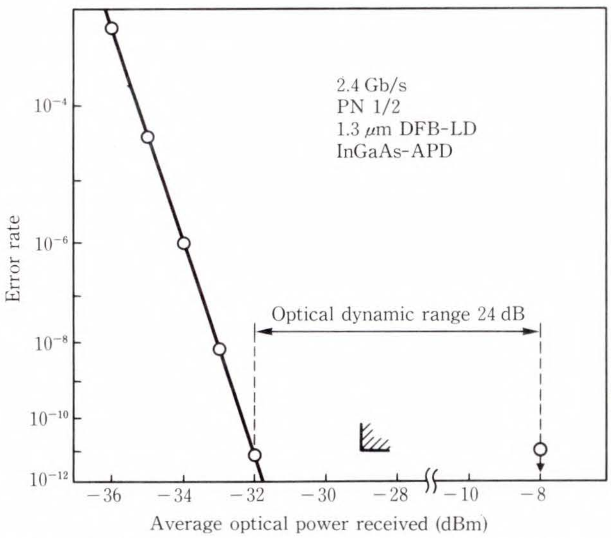



Fig. 19–Error rate characteristics of optical receiver.

Figure 18 shows the LD module, which contains the LD driver IC, DFB-LD, optical isolator, and thermal cooler. The module is connected via a single-mode fiber.

The average optical power output of the transmitter module is +3 dBm. The error rate characteristics of the optical receiver are shown in Fig. 19. The average optical power reception required to maintain a $10^{-11}$ error rate is −32.0 dBm, and the optical dynamic range is more than 24 dB. The performance of these modules is sufficient for 40 km transmission using 1.3 μm single mode fiber or 80 km transmission using 1.55 μm single-mode fiber.

### 3.2 ICs for beyond 2.4 Gb/s

#### 3.2.1 Design concept

The performance of ICs in ultra high-speed systems (i.e. systems operating at about 10 Gb/s) is seriously degraded by various parasitic elements such as the bonding wires between ICs and other components. Therefore, new design concepts that consider not only circuit parameters but also the effect of parasitic reactances will be needed.

#### 3.2.2 Transistor choice

In the last several years, we have studied which kinds of transistors are suitable for use

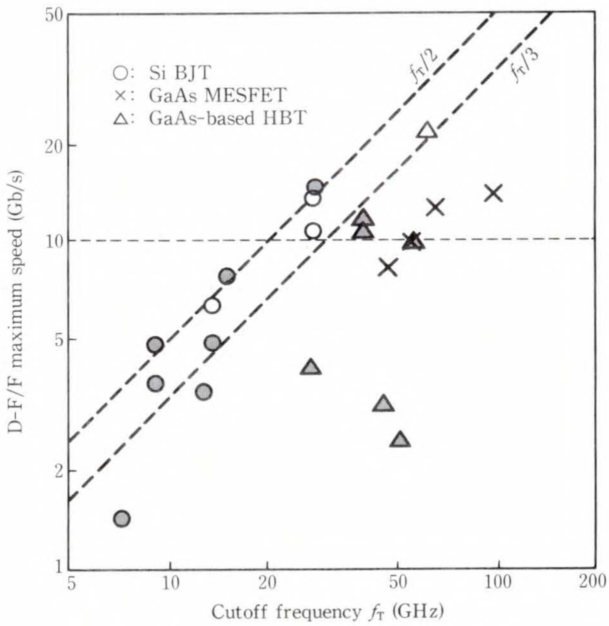



Fig. 20–Performance of high-speed ICs.

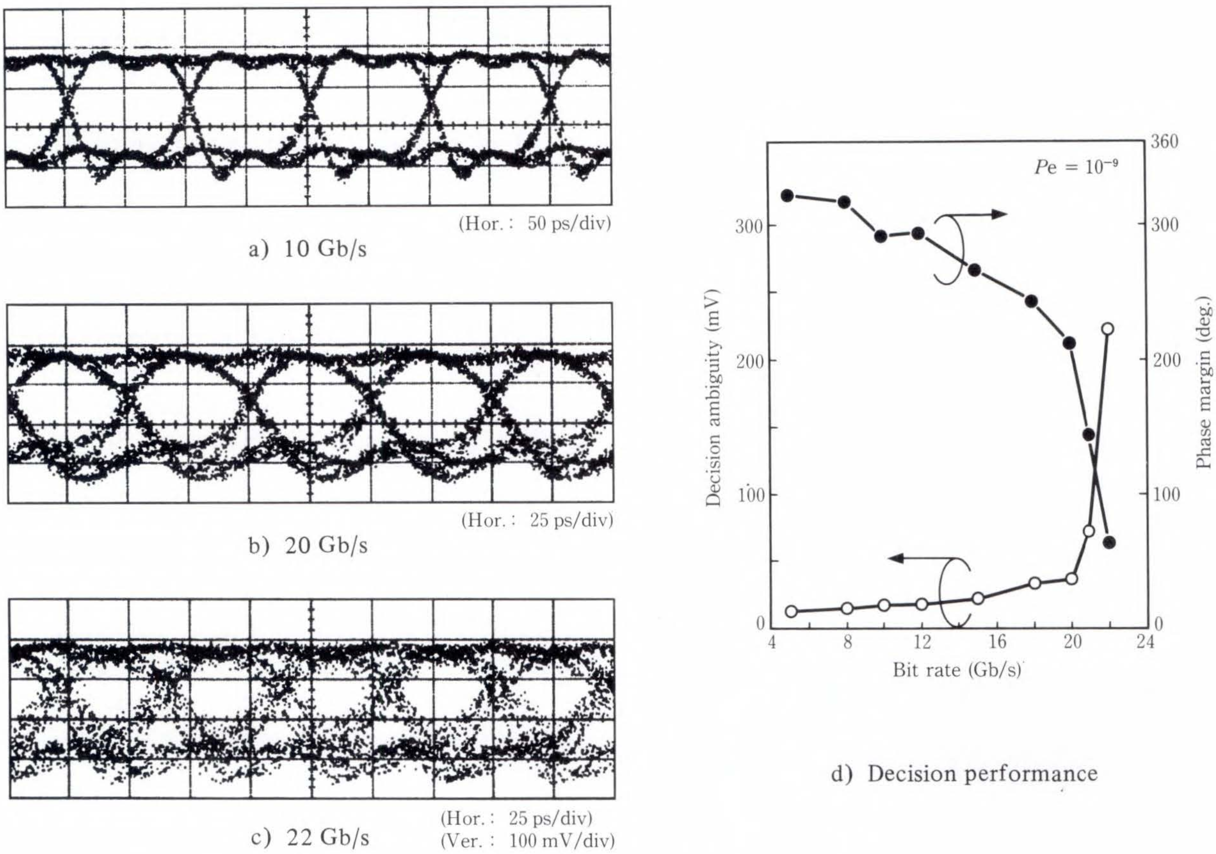
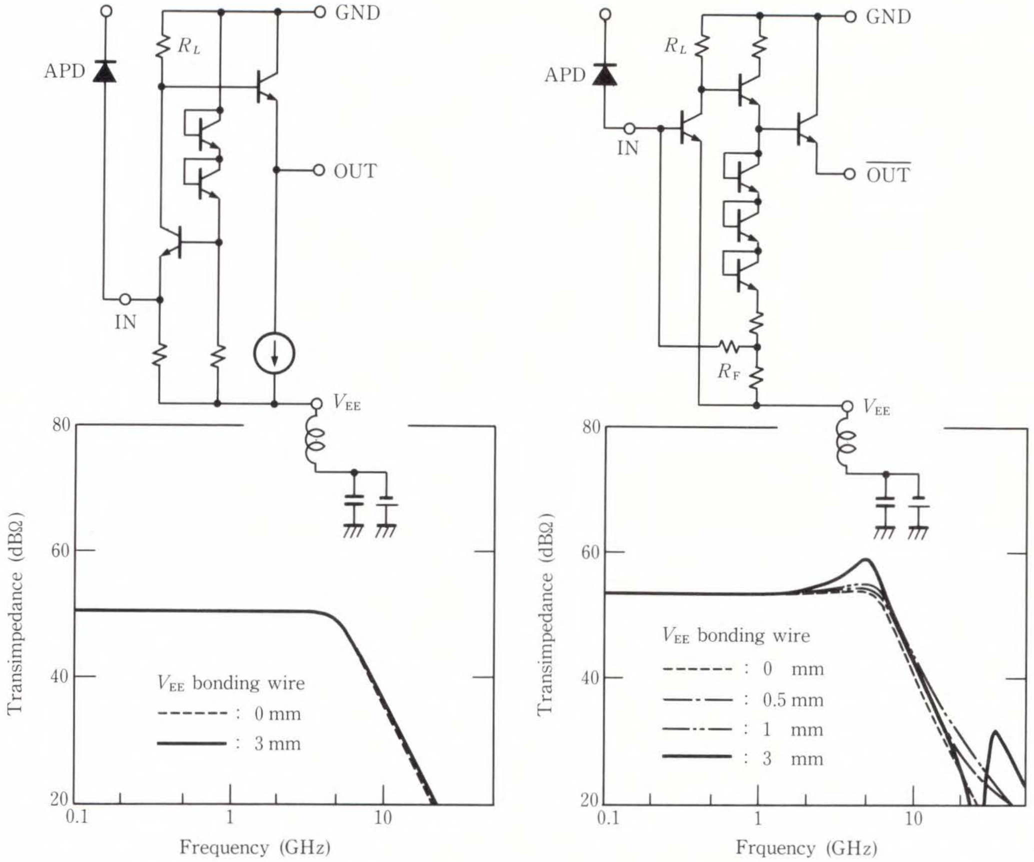



Fig. 21—Waveforms and decision performance of AlGaAs/GaAs-HBT decision IC.

in 10 Gb/s systems. In this study, we fabricated small scale ICs such as decision circuits and preamplifiers using various transistors to assess the transistors' speed and bandwidth capabilities. Figure 20 shows the maximum operating speeds of three decision ICs made using various transistors[14]. In this figure, the shaded symbols indicate the performance of individual transistors. The maximum operating speed of 22.7 Gb/s was achieved by a $2 \times 5$ $\mu$m emitter AlGaAs/GaAs Hetero-junction Bipolar Transistor (HBT) IC[15]. An Si-Bipolar IC using the ESPER configuration operated up to 13.5 Gb/s[16]. A GaAs MESFET IC with a 0.6 $\mu$m gate-length operated up to 8 Gb/s[17]. Based on these results, we are now investigating the performance of 10 Gb/s regenerator ICs that use a combination of Si-Bipolar transistors and AlGaAs/GaAs HBTs. The output waveforms and decision characteristics of the AlGaAs/GaAs HBT IC[15] are shown in Fig. 21.

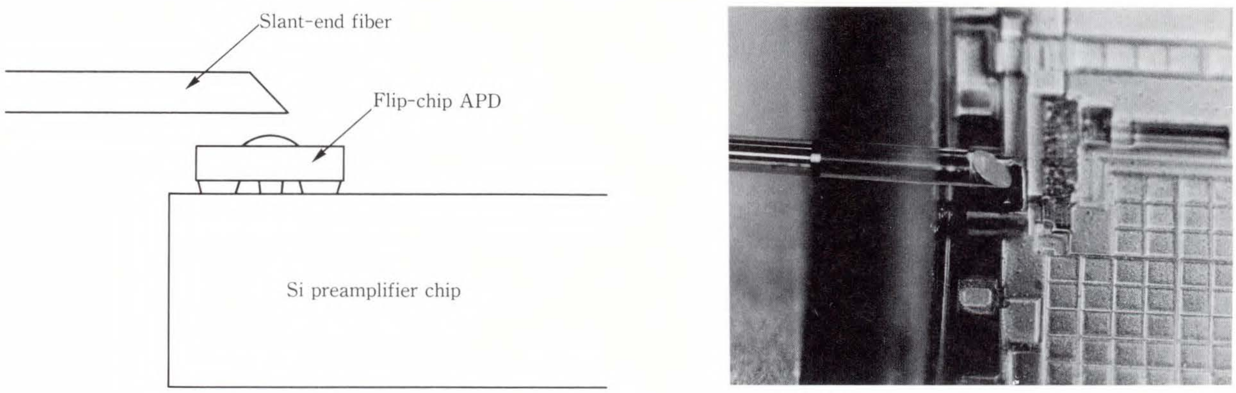
### 3.2.3 Circuit configuration[18]

Figure 22 shows two types of preamplifier. Figure 22a) shows a common-base circuit, which is more suitable for 10 Gb/s applications. Figure 22b) shows the conventional transimpedance circuit. This circuit has a wide bandwidth, low noise, and wide dynamic range. Figure 22 also shows simulated frequency responses for these two circuits for various lengths of bonding wire between the IC and the voltage source pads. As can be seen in Fig. 22b), the conventional transimpedance circuit has gain peaks that increase with the bonding wire length, indicating that this circuit is easily affected by parasitics. On the other hand, the common-base circuit is not affected by the bonding wire length. The noise and dynamic range of the common base circuit were estimated to be comparable with those of the transimpedance circuit. These results show that new circuit configurations, such as the




a) Common-base circuit b) Transimpedance circuit

Fig. 22–Preamplifier circuits.




a) Configuration b) Fabricated optical receiver front end

Fig. 23–Flip-chip bonding method.

common-base circuit will be needed for 10 Gb/s regenerator ICs. We have already proposed several new circuit configurations for ultra-high speed systems.

### 3.2.4 New mounting technique

Noise and bandwidth are greatly dependent on the quality of the connection between the APD and the preamplifier. We are now developing a new mounting technique using a flip-chip APD.

Figure 23 shows the shematic diagram and a close-up of the fabricated optical receiver front end[16]. In the conventional method, the connection between the APD and preamplifier is made

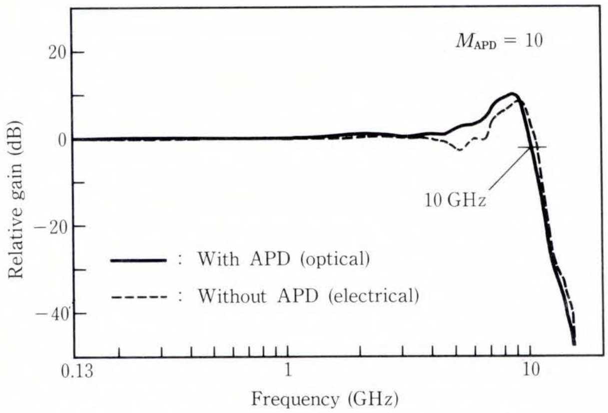



Fig. 24—Frequency response of preamplifier.

using bonding wires. In this method, resonances due to the APD's capacitance and the inductance of the bonding wire seriously degrade the output response of the preamplifier. Directly bonding the flip-chip APD to the preamplifier IC shifts these resonances far beyond the limits of the 10 Gb/s signal. Figure 24 shows the frequency response of the Si-Bipolar preamplifier with and without the APD. As the figure shows, there is only a slight difference in the responses of the preamplifier and the flip-chip APD/preamplifier combination, which proves the effectiveness of the flip-chip APD technique at 10 Gb/s. The receiver sensitivity of the flip-chip APD/Si-Bipolar preamplifier combination was −23 dBm at 10 Gb/s[16].

## 4. Conclusion

This paper looked at Fujitsu's latest integrated circuits for digital transmission systems, focusing on N-ISDN/B-ISDN transmission LSIs and gigabit optical regenerator ICs. Very large-scale and economical CMOS gate arrays are very suitable for use in the N-ISDN interface and for image/voice signal processing. Bi-CMOS gate arrays operating at 156 Mb/s and GaAs gate arrays operating at 620 Mb/s have been successfully used for SDH/ATM transmission for B-ISDN.

2.4 Gb/s optical regenerator ICs have been developed using the following: an ultra high-speed Si-bipolar process that produces transistors having a unity gain bandwidth over the range of 13-16 GHz, and a high-transconductance GaAs MESFET process with a 0.8 $\mu$m gate length. We are now studying the performance of 10 Gb/s optical regenerator ICs made using various processes, for example, Si-bipolar and AlGaAs/GaAs HBT, in order to find the most suitable process for the ICs of 10 Gb/s systems.

The LSIs/ICs described in this paper make the production of compact, low-power, and economical equipment for N-ISDN/B-ISDN transmission systems a very realistic goal.

## References


1) Naito, H., Kawai, M., Ohtsuka, T., Ishihara, T., Yamaguchi, K., Taniguchi, A., Onodera, H., and Ando, T.: 3 Gbit, 16-Channel GaAs Multiplexer and Demultiplexer LSIs. Tech. Dig. GaAs IC Symp. 1989, pp. 321-324.
2) Soejima, T., Ezaki, Y., Iguchi, K., Yamamura, T., Komaki, M., and Ikezawa, T.: CMOS LSI for Flexible Telecommunication Node in Brodband ISDN. Conf. Rec. ICC'89, pp. 1403-1407.
3) Iguchi, K. et al.: ATM Transmission Systems Technonogies. *FUJITSU Sci. Tech. J.* **28**, 2 (Special Issue on Telecommunications) pp. 142-149 (1992).
4) Moridera, A., Motida, Y., and Goto, S.: ISDN Systems in Public Networks. (in Japanese), *FUJITSU*, **41**, 4 pp. 283-299 (1990).
5) Chujo, T., Ueno, N., Takada, A., Hino, Y., and Fukuda, M.: A Line Termination Circuit for Burst-mode Digital Subscriber Loop Transmission. *IEEE J. Select. Areas Commun.*, **4**, 1, pp. 176-183 (1986).
6) Morimatsu, E., Itoh, T., Matsuda, K. Nishizawa, Y., and Takenaka, Y., Itoh, A.: An Experimental Video Based on CCITT H. 261. Proc. PCS'91, 3·15, pp. 117-120.
7) Matsuda, K. et al.: Digital Signal Processing Technology for Communication. *FUJITSU Sci. Tech. J.* **28**, 2 (Special Issue on Telecommunications), pp. 228-240 (1992).
8) Kawai, O., Matsuda, K., Narahira, S., Itoh, A., Ishihara, T., and Ikezawa, T.: An Architecture of a Video Signal Processor LSI VSP-1. (in Japanese), 1989 Autumn IECE Natl. Conv. Rec., D-105, pp. 6. 105.
9) Kosugi, Y., Matsuda, K., Takenaka, Y., Itoh, A. Ishihara, T., and Nagasawa, T.: A Development of Discrete Cosine Transform VLSI. (in Japanese), 1989 Spring IECE Natl. Conv. Rec., D-400, pp. 7. 152.
10) Itoh, T., Oono, M., Matsuda, K., Miyasaka, H., Terui, Y., Fukuda, K., and Tanaka, S.: Development

of CIF conversion and frame memory control LSI. (in Japanese), 1989 Spring IECE Natl. Conv. Rec., D-277, pp. 6. 279.

11) Gambe, H., Ikezawa, T., Kobayashi, N., Tanabe, T., and Unagami, S.: A 32bit floating point digital signal processor FDSP-4 and its application to the communication systems. Proc. GLOBECOM '87, pp. 442-446.
12) Yamaguchi, K., Kitasagami, H., Kawai, M., Amemiya, I., Touge, T., Tamada, H., Kokado, M., and Sugimoto, M.: Gigabit Receiver ICs for Optical Communications. *IEEE J. Select. Areas Commun.*, **6**, 3, pp. 460-467 (1988).
13) Kawai, M., Yamaguchi, K., Miyauchi, A., Onodera, H., Nakamura, K., and Yako, H.: 2.4 Gbit/s Regenerator ICs for Optical Communications. Regular Papers 100C-E00C'91, pp. 509-512.
14) Hamano, H.: ICs for 10-Gb/s and Faster Optical Transmission Systems. Digest LEOS'91 Meeting, pp. 20.
15) Hamano, H., Ihara, T., Amemiya, I., Fukatsugi, T., Ishi, K., and Endoh, H.: 20 Gbit/s AlGaAs/GaAs-HBT 2:1 Selector and Decision ICs. *Electron. Lett.*, **27**, 8, pp. 663-664 (1991).
16) Hamano, H., Yamamoto, T., Nishizawa, Y., Oikawa, Y., Kuwatsuka, Y., Tahara, A., Suzuki, K., and Nishimura, A.: 10 Gbit/s Optical Front End Using Si-bipolar Preamplifier IC, Flipchip APD and Slant-end Fiber. *Electron. Lett.*, **27**, 18, pp. 1602-1604 (1991).
17) Hamano, H., Amemiya, I., Ihara, T., Onodera, H., Onodera, T., and Shimizu, S.: 8 Gbit/s GaAs Logic ICs for Optical Fiber Communication Systems. *Electron. Lett.*, **24**, 24, pp. 1522-1523 (1988).
18) Hamano, H., Yamamoto, T., Nishizawa, Y., Tahara, A., Miyoshi, N., Suzuki, K., and Nishimura, A.: High-speed Si-Bipolar IC Design for Multi-Gb/s Optical Receiver. *IEEE J. Select. Areas Commun.*, **9**, 5, pp. 645-651 (1991).

**Norio Ueno**

Communication Device Technology, Dept.
FUJITSU LIMITED
Bachelor of Electronics Eng.
Kogakuin University 1973
Master of Electical Eng.
Kogakuin University 1975
Specializing in Telecommunication Devices

**Kazuo Yamaguchi**

Broadband Communication Systems Laboratory
FUJITSU Laboratories, KAWASAKI
Bachelor of Metallurgical Eng.
Yokohama National University 1971
Master of Metallurgical Eng.
Yokohama National University 1973
Specializing in Broadband ISDN Transmission Systems

**Takashi Touge**

Optical Transmission Engineering Dept.
FUJITSU LIMITED
Bachelor of Electronics and Communication Eng.
Waseda University 1967
Master of Communication Eng.
Waseda University 1969
Specializing in Optical Communication

UDC 621.395.74.072.9

# Flexible Management and Control of Transmission Network Based on SDH

● Takafumi Chujo ● Hiroaki Komine ● Keiji Miyazaki

*(Manuscript received December 4, 1991)*

The installment of Synchronous Digital Hierarchy (SDH) systems has already begun in various parts of the world. Now that the technologies for initial implementation have been established, path management and control are the key to fully exploit the advantages of SDH by improving the reliability, economical efficiency, and serviceability of networks. After describing the state-of-the-art technologies for SDH implementation, this paper describes a flexible path management and control system. Then, network restoration is discussed as an example of flexible path management and control, and new distributed restoration techniques are proposed. The applicability of the proposed technique is discussed, based on the results of computer simulation.

## 1. Introduction

It is now three years since the concepts and detailed structure of Synchronous Digital Hierarchy (SDH) were established as standards by CCITT[1)]. SDH has gained wide acceptance as a vehicle for removing bottlenecks in existing networks and for accommodating new services, eventually providing a sound basis for the infrastructure of the future Broadband ISDN (B-ISDN)[2)].

Synchronous networks based on the SDH recommendations bring a wide range of benefits:

1) Worldwide unique network node interface
2) Reduction of transmission network investment
3) Enhanced Operation, Administration and Maintainance (OA&M) capabilities
4) Increased reliability
5) Efficient and flexible path manipulations (add/drop, crossconnect)
6) Low-delay digital network
7) Compatibility with existing digital transmission systems
8) Future proof (bearer for B-ISDN)

Since SDH is uniquely defined, it will enable the construction of a global-scale digital network, overcoming the differences in transmission hierarchies adopted previously in various parts of the world. Through its efficient path manipulation capabilities, combined with the ample built-in Operation, Administration, Maintainance and Provisioning (OAM&P) capacity, it will facilitate flexible, reliable, and economical network construction. The transmission frame is designed to be compatible with existing transmission systems and, at the same time, future proof for broadband channels transport. Thus it will provide the vehicle for smooth migration from the Narrowband ISDN (N-ISDN) to B-ISDN. The introduction of SDH systems is the first step towards construction of the B-ISDN.

Implementation of equipment conforming to the SDH standards and deployment in the network are already well under way in various parts of the world[3),4)]. This paper first overviews the state-of-the art technologies for SDH implementation and then focuses on the subject of constructing a flexible and reliable network, fully exploiting the capabilities inherent in the SDH concepts.

## 2. Initial SDH implementation

SDH systems were first deployed in Japan in July 1989. The SDH system configuration adopted by NTT consists of three major components[3]. The fiber optic line system integrates the line termination with the function of skip multiplexing to STM-*n* level. The line termination is designed either for a fiber link or a radio link, the highest level termination for the fiber case being 2.4 Gb/s[5]. The crossconnect system currently provides crossconnection at the VC11 (Virtual Container with 24 channels) level. The adaptation system provides interfaces for existing digital systems.

In North America, systems based on similar technologies are now being introduced. For example, Fujitsu is providing SDH systems targeted mainly for loop multiplexing, called the Fiber Loop Multiplexer (FLM) series, which includes STM-1 (155.52 Mb/s: FLM150), STM-4 (622.08 Mb/s: FLM600), and STM-16 (2488.32 Mb/s: FLM2400) systems. (FLM2400 is to be introduced in 1992.)

For these initial implementations, many leading-edge technologies were developed. For example:

1) High-speed, low-power LSI (CMOS, GaAs)
2) High-speed optical transmitter/receiver (2.4 Gb/s)
3) High-stability clock generation and synchronization ($10^{-11}$)
4) Large-scale, real-time processing firmware
5) High-density packaging

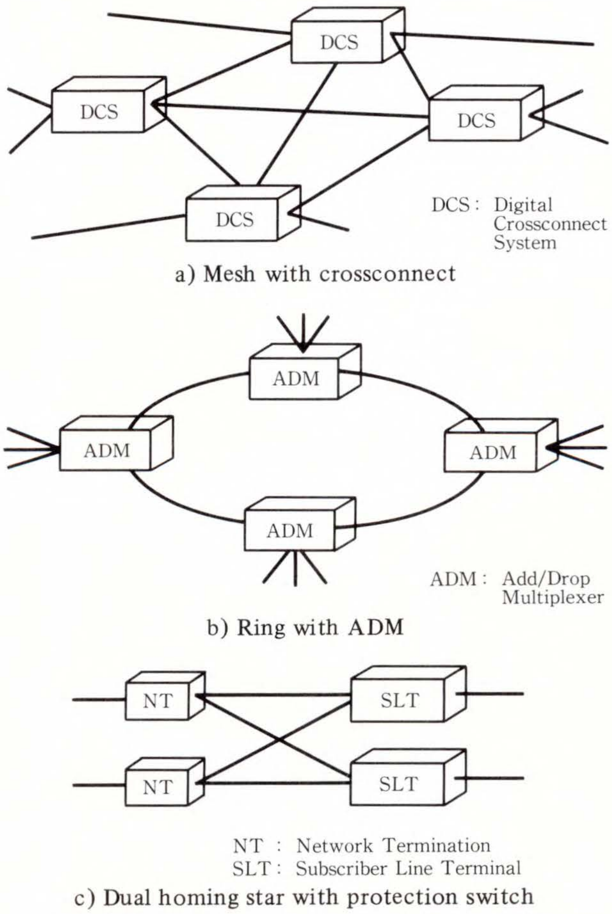
This SDH equipment already has much more extensive OAM&P capabilities than the previous equipment. For example, the FLM series is equipped with the following OAM&P functions:

1) Physical inventory data stored on all plug-in units
2) Software downloading (local and remote)
3) Automatic internal turn-up diagnostics
4) Full access to performance monitoring data (physical, section, line, path level)
5) Provisioning performed locally or remotely via the local terminal or from the OS
6) Automatic provisioning for protection units with the parameters of the failed unit

However, these functions are still limited to the OAM&P level of equipment itself or to the use of point-to-point communications. To fully exploit the capabilities of the SDH systems, further work in both the standards and the implementation is required. In particular, network architecture principles and OAM&P features and capabilities are very important for constructing a flexible and highly reliable network. At the same time, it is necessary to reduce the size and cost of the equipment to promote full SDH penetration, where user-to-user connection by SDH is to be accomplished.

## 3. Flexible and reliable network configuration

Through the efficient path manipulation capabilities of SDH, such as skip multiplexing, add/drop and crossconnection, combined with the ample built-in OAM&P capacity, it has become possible to facilitate various types of network configuration, as shown in Fig. 1. This chapter compares these network configurations,




a) Mesh with crossconnect

b) Ring with ADM

c) Dual homing star with protection switch

Fig. 1—Network configurations based on SDH.

underscoring their flexibility and application ranges. This chapter also outlines a prospective total network architecture combining these configurations to meet various flexibility and reliability requirements.

1) Mesh with crossconnect

Using the time slot exchange function of crossconnect, this configuration facilitates path routing by software control in near-real time. This enables the route to be altered when a link or node failure occurs. Also the route alteration capability can be used to avoid traffic congestion at some portion of the link. For example, the route can be altered season by season, month by month, day by day, or hour by hour according to the traffic pattern. Thus, mesh network with crossconnect has the highest degree of reliability and flexibility, and is most effective when there is a large volume of traffic with high physical network connectivity. This means that this configuration is most suitable for long-haul networks such as trunk or inter-city networks.

2) Ring with ADM

Limiting the number of interfaces or line connections and consequently confining the time slot manipulation to add/drop, Add/Drop Multiplexer (ADM) enables the construction of a flexible and reliable network on a limited scale, but with a much simpler structure. For example, there has recently been a lot of study of the self-healing ring, and the possibility of a high-reliability network has been reported[6]. The ADM is suitable for medium-haul transport networks such as intra-city networks or feeder loops in subscriber networks, where the average traffic volume is comparatively small but the traffic variance may be quite large.

3) Dual homing star with protection switch

Besides simplicity, the advantage of the star configuration is that it is superior in security. Since security is particularly important in subscriber loop networks, the star configuration is indispensable. However, the simple star configuration is lacking in flexibility and reliability. One alternative solution is the dual homing star. In this configuration, a Network Termination (NT) is connected to two different Subscriber Loop Terminals (SLTs) in parallel, one is the working line and the other is the spare line. When the working line fails, the line is switched to the spare line using the protection switch function in the NT. The information exchange for switching between NT and SLT is done using the overhead bytes reserved in the SDH transmission format. Alternatively, the spare line can be used in the normal state as an extra channel. For example, one can use it as a lower-priority channel which can tolerate outage when the working line carrying higher priority traffic fails and is

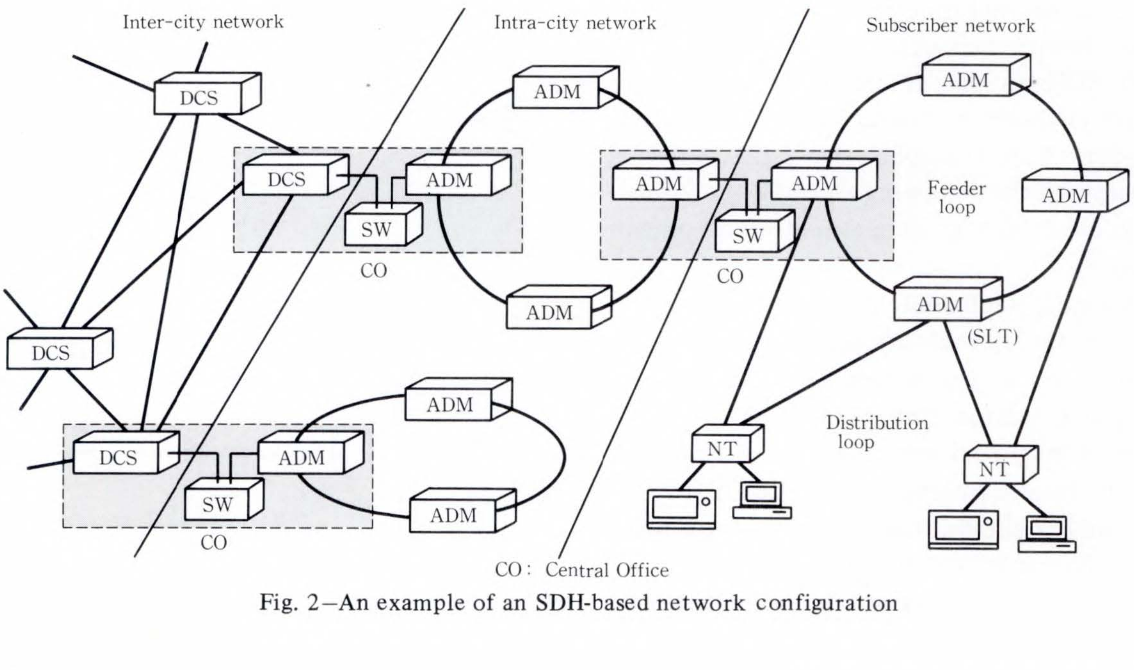



Fig. 2—An example of an SDH-based network configuration

switched over.

Figure 2 illustrates one example of total network configuration which combines the various advantages of the SDH network described above.

All network configurations described here must be linked with appropriate control and management mechanisms in order to make optimum use of their flexibility and reliability. The following chapters discuss this issue focusing on the meshed network with crossconnect, since this requires the highest degree of sophistication.

## 4. Flexible path management and control

The SDH network will provide operating companies with increased network flexibility and survivability through software control of the network elements described in Chapter 2. This chapter discusses management functionalities in crossconnection-based SDH networks. In particular, the focus is on configuration management of the Virtual Container-path (VC-path) layer network, a key to enhancing network flexibility and survivability through management and control.

### 4.1 VC-path management functionalities

Before discussing configuration management, a brief overview of the VC-path management functions is presented. As defined in the CCITT Recommendations[7], the transport network consists of three sub-network layers: a circuit layer network, a path layer network, and a transmission media layer network. The path layer is commonly used by different circuit layer networks such as public switched telephone networks, packet switched networks, and leased-line networks. The function of the managed path layer network is to create standard bandwidth pipes between two path termination points in the upper layer network, then to request the lower layer network to accommodate those logical pipes into physical transmission media. SDH crossconnect equipment plays an important role in rapidly providing these capabilities via software control and creates new opportunities for network management and control.

The management goal which we think important is operational enhancement of network flexibility and survivability through software control. Network flexibility is defined as the ability to reconfigure the VC-path network as the customer requires and to accommodate unexpected growth and changes in traffic. Network survivability is defined as the ability to restore the VC-path network from facility failures such as fiber cuts, equipment faults, fires, and earthquakes.

According to CCITT Recommentation M. 30[8], these management functions are categorized into five areas: performance management, configuration management, security management, and accounting management. To achieve the management goals, it is first necessary to detect, localize, and report failures as quickly as possible through in-service performance monitoring and the operation support interface. After a failure is reported, configuration management then attempts to reconfigure and restore the network using the available facilities.

As described in the previous section, the embedded OH bytes in the SDH frame allow fault and performance management to greatly enhance these capabilities through the continuous error checking and embedded data communication channels. CCITT Recommendation G. 784[7] proposes how the data communication channel should be used for fault management and performance management. However, configuration management, a key to enhance both flexibility and survivability, has yet to be discussed in detail.

The following sections give the authors' view on configuration management in the VC-path network, then propose new management and control techniques to enhance network survivability.

### 4.2 VC-path configuration management

Configuration management consists of sub-management functions, i.e. path setup, path restoration, spare capacity management, and resource management, as shown in Fig. 3. Path

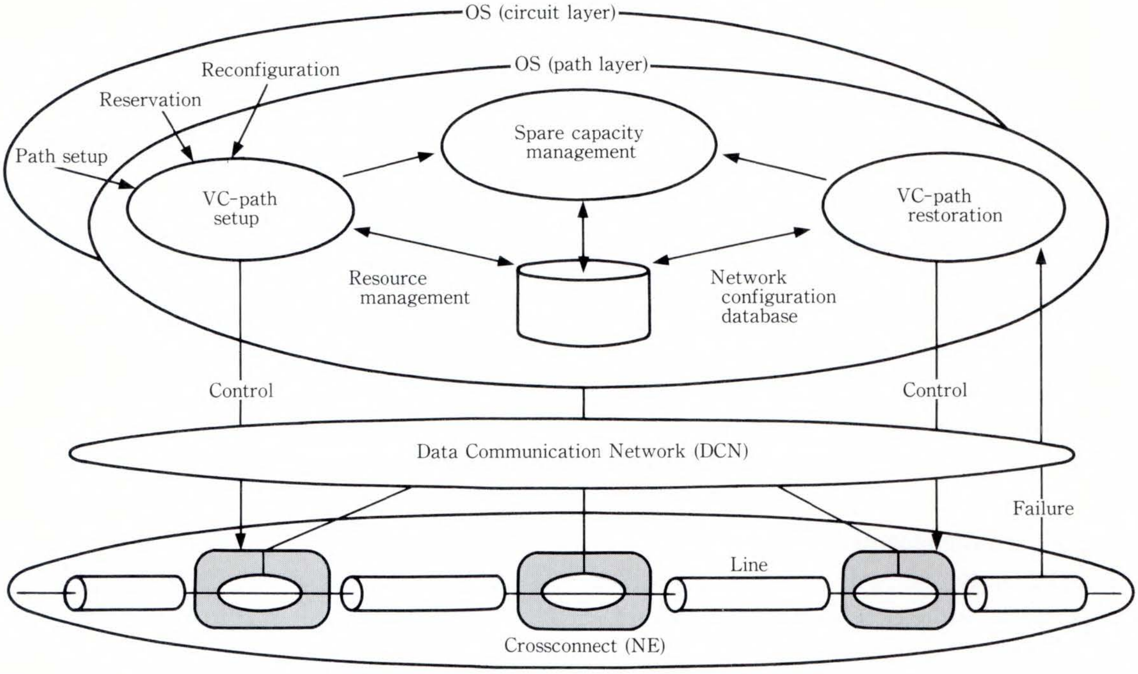



Fig. 3 – VC-path management and control

setup is the process of configuring the network to provide a connection between path termination points. Upon receiving a request, the path setup function finds a route between two nodes using currently available capacity in the network. A failure is detected by the performance management function, which then requests the restoration function to restore the failed path. The restoration function also finds an alternate route using the currently available capacity in the network.

The ability to reconfigure and restore the network is heavily dependent not only on the physical network connectivity and traffic, but also on the amount and location of spare capacity in the network. The spare capacity function manages the amount and location of spare capacity, according to performance and network cost constraints. Figure 4 is a schematic diagram of the managed spare capacity of a transmission link. The spare capacity can be partitioned conceptually into three groups:

1) Spare capacity for restoring current working paths
2) Spare capacity for restoring newly accommodated working paths, and
3) Spare capacity for accommodating new path

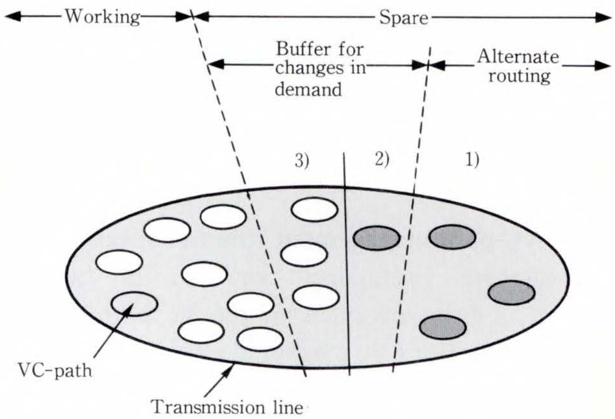



Fig. 4 – Spare capacity management.

setup requests.

These reserved capacities constitute the spare capacity management process. After receiving a new request, the path setup function requests the spare capacity manager to allocate the spare capacity needed to set up the requested path. The spare capacity manager then determines whether there is enough capacity to restore the requested path. If the required capacity is available, the spare capacity request is acknowledged. The spare capacity manager interacts with the path setup and path restoration functions without operator intervention, and continuously monitors the network element states.

## 5. VC-path restoration

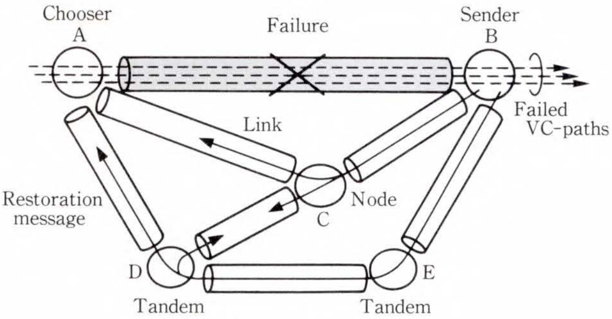
The service restoration methods are increasingly important as transmission capacity grows with the deployment of high-speed optical fiber systems. This chapter proposes a distributed restoration algorithm and a spare capacity allocation algorithm based on the authors' management concept described above.

### 5.1 Management architecture

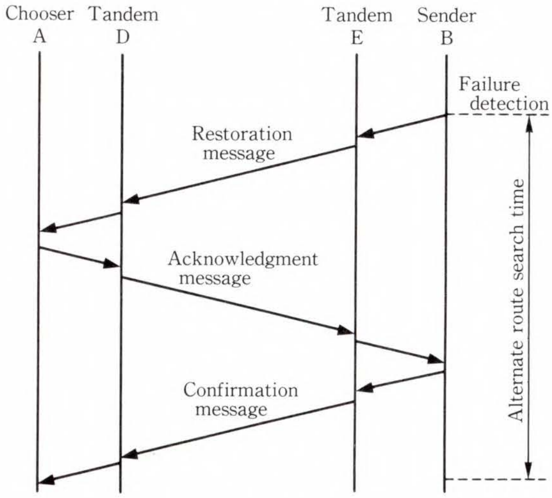
The many algorithms developed to restore networks can be classified as either centralized[9] or distributed algorithms[10)-12)]. In centralized control, the network is controlled and managed from a central operation center based on the state of a global network, but must keep data synchronized with the true state of the available capacity in the network. In distributed control, the processing load is distributed among the nodes and restoration is thus quite faster. To communicate between the nodes, the SDH OH bytes for Embedded Operation Channels (EOCs) can be used. Therefore, the time-critical path restoration function is distributed over all cross-connect nodes, while the spare capacity management function is centralized and makes optimum decisions based on the overall global network state.

### 5.2 Restoration algorithm: multi-destination flooding algorithm

The distributed algorithms proposed so far[10)-12)] are based on simple flooding. When a link fails, one of the two nodes connected to the failed link becomes the sender and the other becomes the chooser (see Fig. 5). The sender broadcasts restoration messages to all links with spare capacity. Every node except the sender and the chooser responds by rebroadcasting the message. Broadcasting messages far away from the sender may cause message congestion. However, this can be avoided by limiting the number of links the message traverses, called the hop limit. When the restoration message reaches the chooser, the chooser returns an acknowl-




a) Link failure




b) Message passing

Fig. 5—Simple flooding.

edgment to the sender. The acknowledgment message is relayed back to the sender to confirm the existence of an alternate route. The sender then sends a confirmation message to the node in the alternate path to assign spare capacity and thus creates a new route for failed paths.

These algorithms usually assume a single-link failure, but in reality, some links which go to different nodes may be in the same conduit. Therefore, if the conduit is cut, several links may fail at the same time[11)]. Fires and earthquakes can also damage a large number of nodes, so the restoration algorithm must also be able to handle these multiple-link and node failures.

Simple flooding methods assume just one chooser. This is extended here to allow multiple choosers to be message destinations (see Fig. 6)[13)]. When a node detects the loss of a signal from a link, the node cannot tell whether the link or the node at the other end has failed. The node sends a restoration message directed to

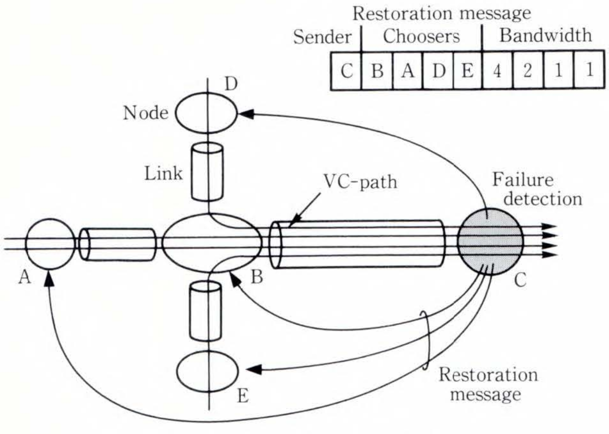



Fig. 6—Multi-destination flooding.

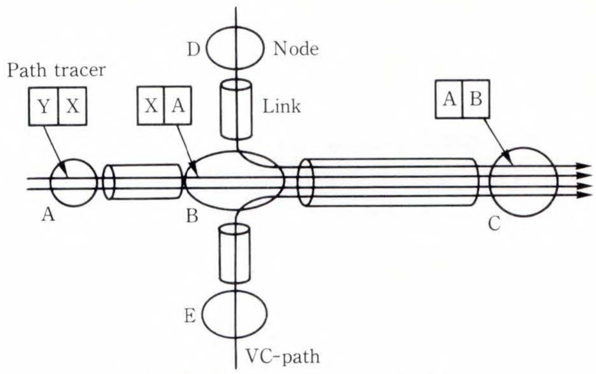



Fig. 7—Path route monitoring.

the node which is the chooser in a link failure, as well to those that are choosers in a node failure. The restoration message contains all choosers and the required capacity for each sender-chooser pair. The node which received the restoration message checks the destination field of the message, and if it is a chooser candidate, it returns an acknowledgment to the sender. Thus, by extending simple flooding into multi-destination flooding, link or node failures do not have to be distinguished because there is always at least one chooser. The sender does not send different restoration messages to each chooser candidate, but instead broadcasts only one restoration message listing all candidates. Therefore, the number of restoration messages decreases and congestion is reduced.

### 5.3 VC-path route monitoring

For multi-destination flooding, each node must have the route information of the paths passing through the node. One approach is to have the central office distribute such route information to all nodes. However, routes are changing dynamically under customer control, and nodes might receive inconsistent route information because updating route data takes time. The authors propose a path route monitoring method in which each node collects the route time[13].

Assuming a single-link, double-link, or single-node failure, the chooser candidates could be the last two consecutive nodes in every path before the sender. Therefore, the route information items required at every node are the IDs of the last two consecutive nodes in every path before the node. This information is collected as follows. Node IDs are sent through assigned space in the path overhead (see Fig. 7). For every path going through a node, the data in the ID area is shifted and the ID of the node it is going through is written in. In this way, every node receives continuous and real-time route information.

### 5.4 Spare capacity assignment algorithm

The spare capacity assignment algorithm is the heart of the spare capacity manager and assigns spare capacity to each physical link of a network to minimize the total spare path length. The multi-destination flooding algorithm chooses $k$ link-disjoint paths which are successively the shortest, to multiple chooser candidates. The time taken to execute the algorithm at each node is assumed to be negligible, and $k \leq$ connectivity of sender $-$ 1. Based on these assumptions, the problem of spare capacity design can be formulated as follows.

1) Given: The physical network topology and the working path capacity of each physical link
2) Objectives: To assign spare capacity of each physical link to minimize the total spare path length.
3) Subject to: $k$ link-disjoint paths which are successively the shortest to multiple chooser candidates, and the hop limit

To solve this problem, the authors developed a heuristic add/delete algorithm[14] (see Fig. 8). In the initial design step, the software accumu-

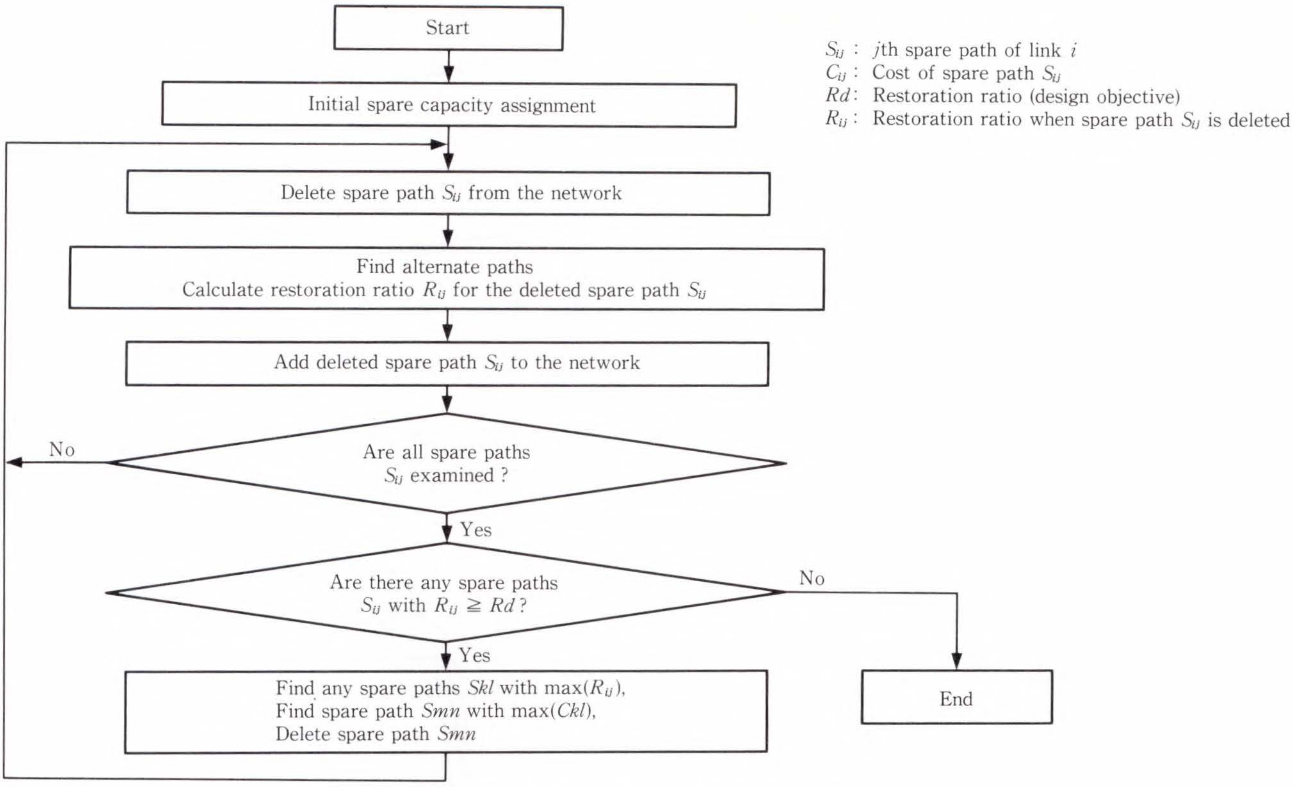



Fig. 8—Spare capacity assignment algorithm.

lates all alternate paths for each possible failure. These include single-link, multiple-link, and node failures. Since alternate paths for one failure can share a spare path with another failure, this may result in redundancy. The algorithm finds such redundancies and deletes as many spare paths as possible.

### 5.5 Results of simulation

The authors studied restoration performance using mesh network models. The length of each link and the number of the working paths of each link were randomly generated. Based on the network topology and the number of working paths, the spare capacity design software assigns spare capacity to each link to recover from single-link, double-link, and single-node failures. Then the authors examined all these failures and evaluated their alternate route search times and restoration ratios using a network simulator (see Fig. 9).

#### 5.5.1 Spare capacity ratio, network connectivity and hop limit

Figure 10 shows the spare capacity ratio,

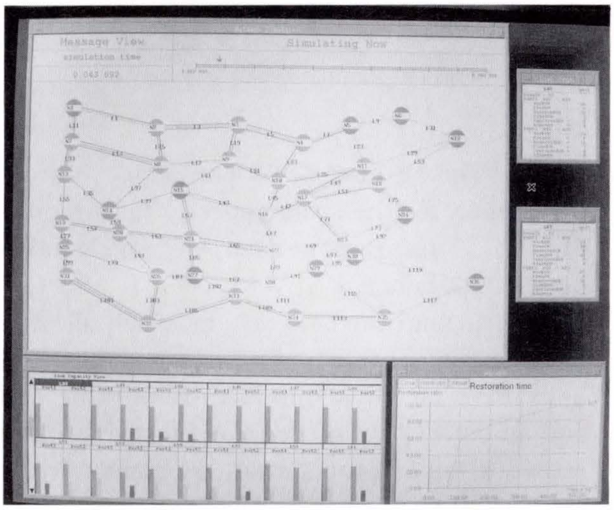


Fig. 9—Network simulator.

which is the ratio of the total spare path length to the total working path length, for various mesh networks. In the 100-node mesh network, the spare capacity ratio for 100 % restoration is 53 % for single-link failures, and 165 % for double-link failures.

The spare capacity ratio is heavily dependent on physical network topology and working path

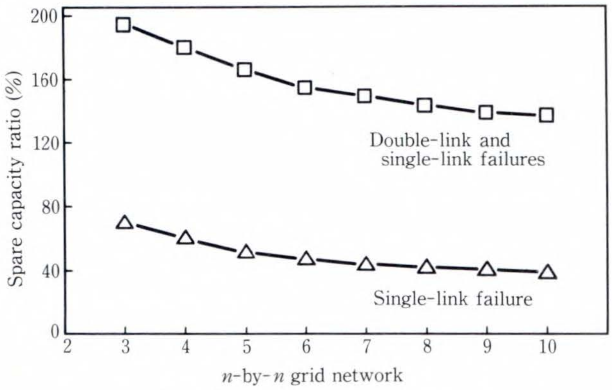



Fig. 10—Spare capacity ratio.

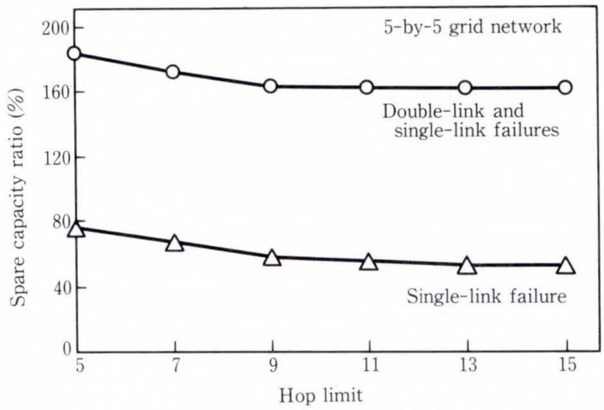



Fig. 12—Spare capacity ratio with hop limit.

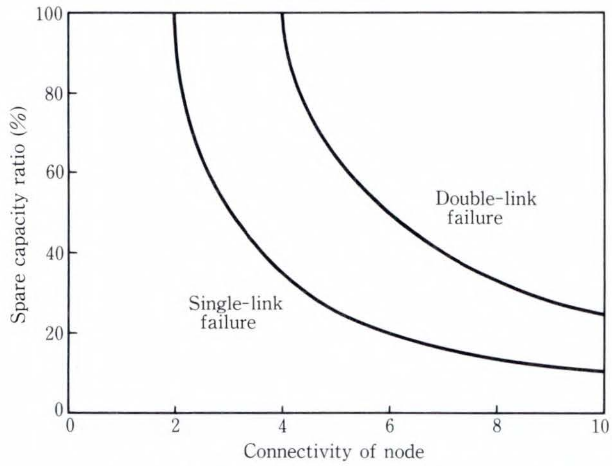



Fig. 11—Required spare capacity ratio.

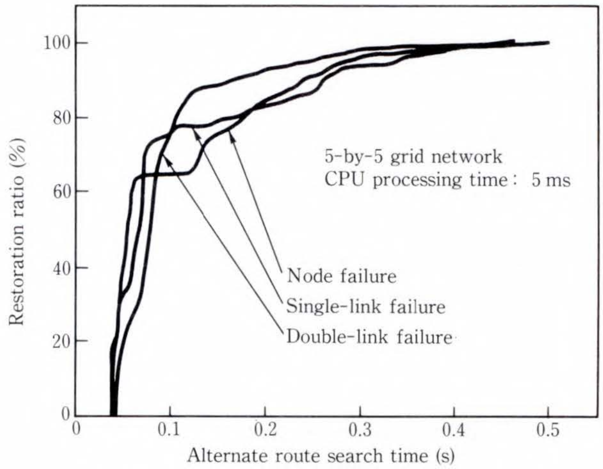



Fig. 13—Alternate route search time.

layout. The authors developed a rule based on experience for the spare capacity ratio required for 100 % restoration from any single-link or double-link failure.

For a uniform network, in which all of the nodes have equal connectivity, and all of the links have an equal number of working paths and an equal length, the required spare capacity ratio is given as a function of the connectivity of the node ($c$):

Required spare capacity ratio

$$= \frac{1}{c-1} \quad \text{(for single-link failure)}$$

$$= \frac{2}{c-2}. \quad \text{(for double-link failure)}$$

Figure 11 shows the required spare capacity ratio for 100 % recovery from single-link and double-link failures.

Another factor affecting the spare capacity ratio is the hop limit for alternate routes. Figure 12 shows the results for spare capacity ratio vs. hop limit for single and double link failures in a 5-by-5 grid network. These results show that the spare capacity ratio increases as the hop number decreases.

### 5.5.2 Alternate route search time and hop limit

Figure 13 shows the cumulative restoration ratios of single-link, double-link, and single-node failures. The restoration ratio of the network is the ratio of restored to lost paths. For node failure, the paths terminating at the failed node are not counted as lost paths because it is impossible to restore them. The alternate route search time is reduced in proportion to the decrease in hop limit. The results indicate that the proposed algorithm can handle multiple-link and node failures as well as single-link failures. All alternate route searches are completed

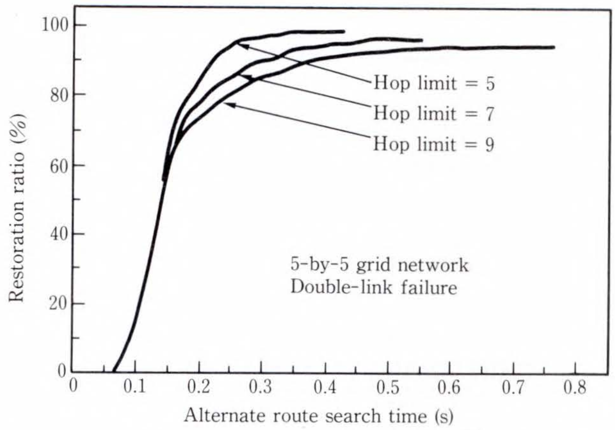



Fig. 14—Alternate route search time vs. hop limit.

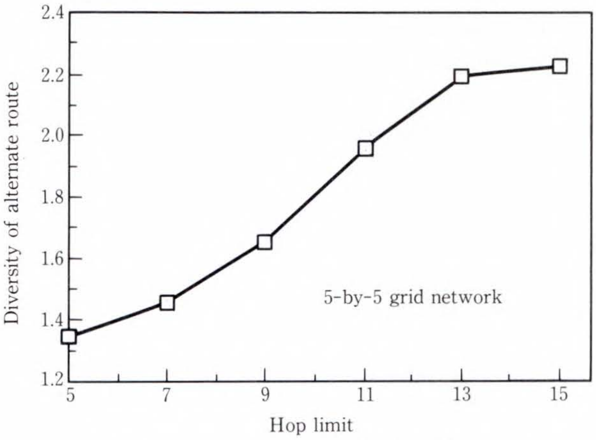



Fig. 15—Diversity of alternate routes.

within 0.5 s, with the message processing delay at the nodes assumed to be 5 ms.

Figure 14 shows the results for alternate route search time vs. hop limit. Figures 13 and 14 show that there is a clear tradeoff between the alternate route search time and the spare capacity ratio. As the hop limit increases, the spare capacity ratio decreases but the alternate route search time increases. Therefore, there is an optimum hop limit that gives the best cost-performance.

### 5.5.3 Diversity of alternate routes

This section describes how many alternate link-disjoint routes can restore one failed route. Figure 15 shows the diversity of the alternate route which is defined as the number of link-disjoint alternate routes per failed route. This figure shows that the route diversity rises as the hop limit of alternate routes increases. One failed route is restored by two alternate routes

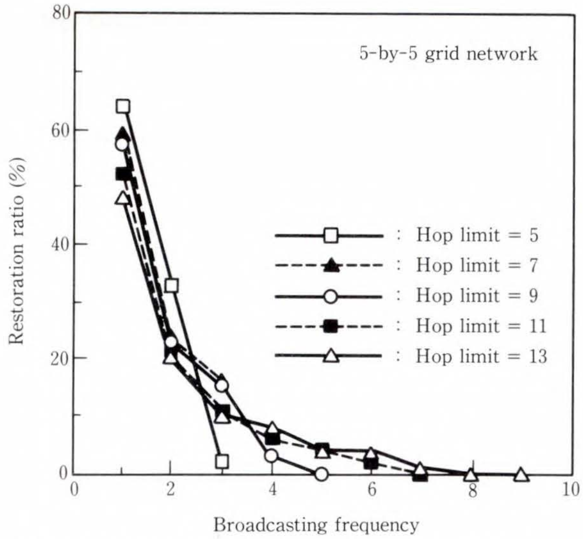



Fig. 16—Broadcasting frequency in spare capacity assignment.

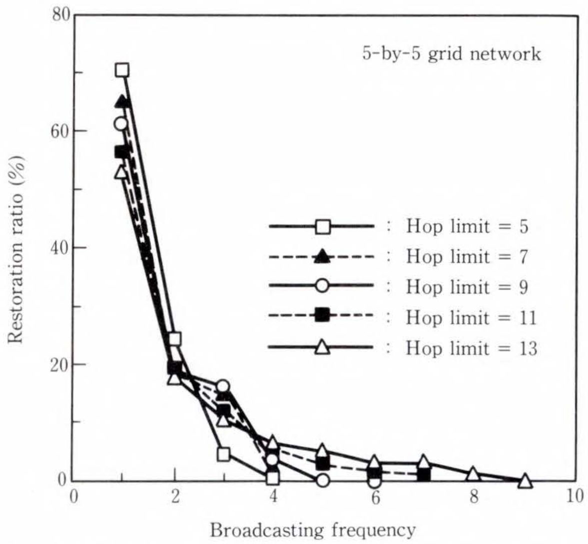



Fig. 17—Broadcasting frequency in simulation.

in a 5-by-5 grid network when the hop limit is 11. When the hop limit is 15, the increment of the route diversity is small compared with one from 11 to 13. This is because it is very difficult to route alternate routes with hop numbers higher than 13 in a 5-by-5 grid network. Therefore, the hop limit of the alternate routes is restricted by the network size.

### 5.5.4 Restoration ratio and broadcasting frequency

Multi-destination flooding is recursively executed to broadcast the restoration messages until all failed paths are restored or additional alternate routes cannot be found. Therefore, this section describes the restoration ratio for each iteration of message broadcasting. Figures 16 and 17 show the spare capacity assignment and simulation results. In these figures, the broadcasting frequency increases as the hop limit of alternate routes increases, and the results in assignment and simulation are very similar. But for spare capacity assignment, the restoration ratio acquired by the first broadcast is less than the simulation. This is because the spare capacity assignment algorithm takes account of the worst case.

## 6. Conclusion

State-of-the-art technologies in LSI devices, optical transmission, and high-density packaging have made possible the initial implementations of SDH systems. The deployment of the systems in the networks has begun, commencing a major step towards constructing a globally unique digital network. For the network to form a sound infrastructure for the coming Broadband ISDN, it is vital that it be constructed with flexibility and reliability on a much greater scale, then built into the existing networks. Exploiting the operation and management capabilities embedded in the SDH specifications and utilizing the current advanced distributed processing technologies, it seems that this goal can be achieved. It is important to study the various network architectures and the corresponding control-and-management shcemes, and moreover establish firm standards in this area so that the global network operates as a single coherent network in today's multi-carrier and multi-vendor environment.

This paper has described the current state of technologies in terms of system implementation and proposed a means for achieving the objectives described above.

## References


1) CCITT: Recommendations G.707-G.709. Blue Book, 1988.
2) Murano, K. et al.: Technologies Towards Broadband ISDN. *IEEE Commun. Mag.*, **28**, 4, pp. 66-70 (1990).
3) Kasai, H. et al.: Synchronous digital transmission systems based on CCITT SDH standards. *IEEE Commun. Mag.*, **28**, 8, pp. 50-59 (1990).
4) Sandesara, N.B. et al.: Plans and considerations for SONET deployment. *IEEE Commun. Mag.*, **28**, 8, pp. 26-33 (1990).
5) Arai, H. et al.: Field Investigation Results of 2.4 Gb/s Fiber Transport Module for SDH System. (in Japanese), 1991 IEICE Spring Natl. Conv. Rec., B-783, pp. 3.335.
6) Wu, T., and Lau R.C.: A class of Self-Healing Ring architectures for SONET network applications. Proc. GLOBECOM '90, 403.2.1, pp. 444-451.
7) CCITT: "Synchronous Digital Hierarchy (SDH) Management". Draft Recommendation G.784, 1990.
8) CCITT: "Principles for a Telecommunications Management Network (TMN)". Recommendation M.30, 1988.
9) Falconer, W.: Services Assurance in Modern Telecommunications Networks. *IEEE Commun. Mag.*, **28**, 6, pp. 32-39 (1990).
10) Grover, W. D.: The Selfhealing Network: A Fast Distributed Restoration Technique for Networks Using Digital Crossconnect Machines. Proc. IEEE GLOBECOM '87, 28.2.1, pp. 1 090-1 095.
11) Yang, C. H., and Hasegawa, S.: Failure Immunization Technology for Network Service Survivability. Proc. GLOBECOM '88, 47.3.1, pp. 1 549-1 554.
12) Chujo, T., et al.: The Design and Simulation of an Intelligent Transport Network with Distributed Control. Proc. IEEE NOMS, 1990, pp. 11.4.1-12.
13) Komine, H. et al.: A Distributed Restoration Algorithm for Multiple-link and Node Failures of Transport Networks. Proc. GLOBECOM '90, 403.4.1, pp. 459-463.
14) Miyazaki, K. et al.: Spare Capacity Assignment for Multiple-Link Failures. Proc. IWACA, 1992, pp. 191-197.

**Takafumi Chujo**

Broadband Communication Systems Laboratory
FUJITSU LABORATORIES, KAWASAKI
Bachelor of Electronics Eng.
Kanazawa University 1976
Master of Electronics Eng.
Nagoya University 1978
Specializing in Network Operations and Management

**Keiji Miyazaki**

Broadband Communication Systems Laboratory
FUJITSU LABORATORIES, KAWASAKI
Bachelor of Mathematics Sci.
Kumamoto University 1986
Master of Mathematics Sci.
Kumamoto University 1988
Specializing in Network Operations and Management

**Hiroaki Komine**

Broadband Communication Systems Laboratory
FUJITSU LABORATORIES, KAWASAKI
Bachelor of Electrical Eng.
Science University of Tokyo 1985
Specializing in Network Operations and Management

UDC 334.722:621.395.74

# Intelligent Network Architecture and Service for Private Network

● Moo Wan Kim ● Akira Hakata ● Norihiro Aritaka

*(Manuscript received November 7, 1991)*

This paper proposes a new network architecture and services for the private network. First the requirements for the private network are described, then a new network architecture which has three planes (physical, logical, and service) is proposed. The prototype system, and software configuration and services based on the proposed architecture are then described. Also, the effectiveness of AI technology for realizing the advanced communication services is demonstrated with the AI secretary service, which can handle incoming calls like a human secretary.

## 1. Introduction

The R&D for realizing an intelligent network is being undertaken in various organizations around the world. The basic concept of an intelligent network is to provide advanced communication services more quickly by combining the information transport capabilities of a communication network with the information processing capabilities of a computer. To provide an intelligent network for the private networks, vendors of switching systems and computers have developed the switch-computer interworking systems[1]. At present, efforts are being made[2] to establish a standard interface between the switching system and the computer and, at the same time, many application services have been developed based on the each vendor's own interface.

This paper proposes a new network architecture for the private network which enables the vendors and service providers to develop the network and services systematically. First, the requirements for the private network are described and the new network architecture based on those requirements is proposed. Finally, the prototype system and services based on the proposed architecture are described.

## 2. Requirements

The following are the main conditions to be considered for developing a private network.

1) Flexibility and service expansibility

The office environment is continuously being enhanced with the advancement of Office Automation (OA). Various network resources (e.g. terminals, common service equipment, and databases) are replaced more frequently than those in the public network. Therefore, first of all, a private network must be capable of step-wise expansion to keep pace with these changes.

Second, with the enhancement of new office environments, a private network must allow for continuous and rapid development of new services and modification of existing services. It must therefore provide a flexible service creation and control mechanism, which enables service providers and users to modify and develop each service independently of other services.

Third, when a new service is introduced, new corresponding service traffic would increase steadily with the expansion of user's scope and needs. Therefore a private network also should be capable of step-wise expansion of service traffic.

Finally, vendors must supply private networks with physical network configurations appropriate for each kind of office.

2) Intelligent user support

There are already many switching services provided by Private Branch Exchange (PBX) systems and Application Processors (APs), but an ordinary user cannot make full use of these services. This is partly because of the poor human interface offered by the terminal. However, the main reason is the insufficient user support for selection of services. The users must determine their own requirements, the relationship between services, and the effect of various network states before selecting a service. This problem will become more serious as new services continue to be developed.

A powerful intelligent user support function is indispensable to help users select adequate services.

## 3. Network architecture

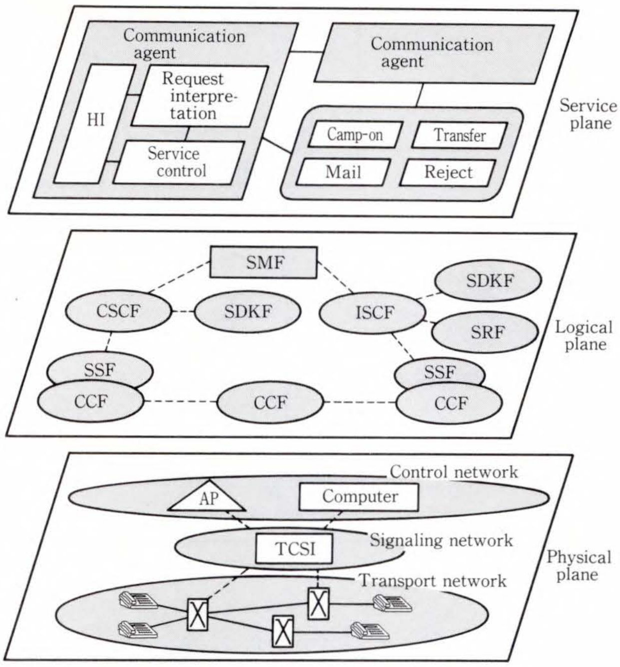
Figure 1 shows the proposed network artchitecture based on the conditions described in chapter 2. Each plane corresponds to different viewpoint of the same network: physical equipment viewpoint, logical function viewpoint and service viewpoint. The outline and features of each plane are as follows.

1) Physical plane

The physical plane specifies the physical network configuration according to various office conditions. It consists of three subnetworks: transport network, control network, and signaling network. The transport network contains transport and switching systems such as subscriber line controller, transmission equipment, and PBX. The control network contains various service control systems such as an adjunct application processor and general-purpose computer with database. The signaling network contains signaling systems to connect transport networks and control networks such as a PBX-computer interface processor and common signaling equipment.

2) Logical plane

The logical plane specifies the logical Function Entities (FEs) and the interfaces between FEs. As shown in Fig. 1, main FEs basically




TCSI : Telecommunication and Computer Service Interface
—— : Information line ---- : Signaling line

Fig. 1—Network architecture.

correspond to those in the architecture model discussed at CCITT[3]. Each FE is defined as follows.

i) Call Control Function (CCF)

Basic call processing function to control point-to-point connection.

ii) Service Switching Function (SSF)

Interface function to combine CCF with CSCF or ISCF.

iii) Call Service Control Function (CSCF)

Advanced call processing function to control service with call state transition such as additional switching service (e.g. conference service).

iv) Information Service Control Function (ISCF)

Service control function to combine a service with information processing (e.g. telemarketing service).

v) Service Database/Knowledge base Function (SDKF)

Management and access function of database/knowledge base.

vi) Service Resource Function (SRF)

Control function for specific service resources such as mailing system.

vii) Service Management Function (SMF)

Management function to install service

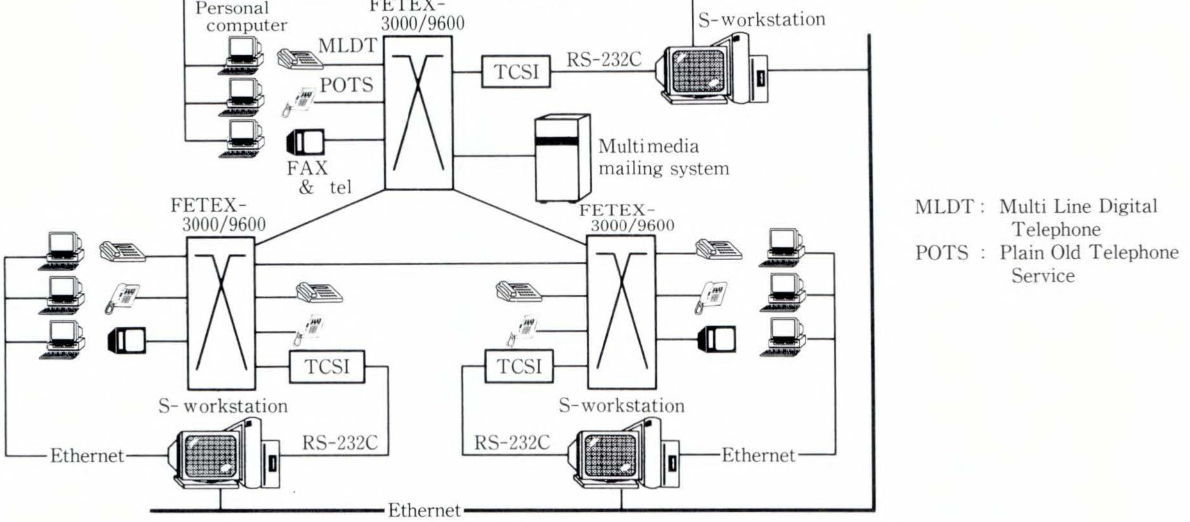



Fig. 2–Hardware configuration.

software in CSCF and ISCF. Using this FE, service providers and users can develop customized service.

The separation of the physical plane and the logical plane allows vendors to provide various types of private networks easily. This is because a private network fitting a specific physical network configuration can be obtained by simply mapping the necessary logical FEs onto the physical plane.

3) Service plane

The service plane consists of a set of services provided by a private network and the user support function (say, a communication agent) for each user. Services are classified into basic services realized only by CCF and advanced service by interworking multiple FEs. As shown in Fig. 1, a communication agent has a Human Interface (HI), a function for interpreting user requests, and a service control function.

These functions are enhanced by applying AI technology, enabling the communication agent to realize a more flexible service management function like a human secretary. This feature makes all the communication services implicitly provided in the network available to the ordinary user.

## 4. Prototype system

### 4.1 System configuration

Figure 2 shows the system configuration based on the network architecture proposed in chapter 3. The Fujitsu PBXs (FETEX-9600 series for overseas and FETEX-3000 series for domestic) are connected by transmission lines, and the Fujitsu S-series workstations are connected as application processors to the PBXs by Telecommunication and Computer Service Interface (TCSI). TCSI is a Fujitsu standard interface between a PBX and computer system. It provides 47 primitives, including "Connect Request", "Disconnect Request", and "Hold Request"[1]. An application processor controls the basic call processing in PBX by executing these primitives, and advanced sophisticated services can be provided. A multi-media mailing system which can store voice and FAX mail is connected to PBX as an intelligent peripheral device. Plain Old Telephone Service (POTS), FAX, and Multi Line Digital Telephone (MLDT) terminals with message-indication functions are also connected to the PBXs. Personal computers to input service control data are connected to the application processor through Ethernet.

On the physical plane, a transport network consists of PBXs and various terminals, a control network consists of application processors, and a signaling network consists of TCSI processors. On the logical plane, CCF and SSF are implemented in the PBXs, CSCF, ISCF, and SDKF are in the application processor, SRF is in the multi-media mailing system, and SMF is in

Fig. 3—Apearance of the prototype system.

the personal computers.

Figure 3 shows the appearance of the PBX and the application processor. This prototype system was exhibited at Telecom 91.

### 4.2 Software configuration

Figure 4 shows the software configuration of the application processor[4]. All the software components are designed based on the object-oriented paradigm and are classified into two groups, the application layer and virtual layer, to simplify changing the program modules or adding the new ones.

1) Application layer

This layer includes the program which executes the service logic.

i) Communication agent manager

Controls the assignment of the communication agent to each user.

ii) Communication agent

Controls the service unit which best satisfies the user's request in place of each user.

iii) Service units

Service units are controlled by the communications agent. They are classified into two groups: telecommunication service and deskwork service. The telecommunication service group includes the originating service, the accept service, the call transfer service, the mail service and the reject service. The deskwork service group includes the call-processing instruction registration service and the schedule management service. The call-processing instruction registration service helps users to register the call instruction data which indicates how to process incoming calls. The schedule management service manages the user's personal schedule database.

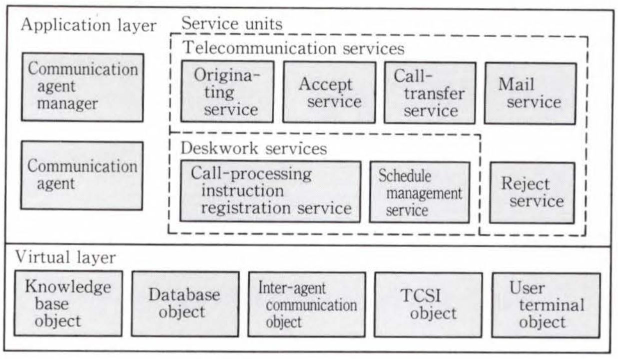



Fig. 4—Software configuration.

2) Virtual layer

All the programs in this layer hide the location and the method for physically handling the resources in the network.

i) Knowledge base object

The logical name of the knowledge base and its logical handling instructions are analyzed to find its location and to generate actual operations for the knowledge base.

ii) Database object

The logical name of the database and its logical handling instructions are analyzed to find its location and to generate actual queries to the database.

iii) Inter-agent communication object

Controls the communication between communication agents, and the logical name of the communication agent is analyzed to find its location.

iv) TCSI object

A logical PBX handling instruction is analyzed to generate the primitives supported by TCSI.

v) User terminal object

The logical name of the terminal and its logical handling instruction (input/output) is

| From | To | User status | Calling party | Service |
|---|---|---|---|---|
| 10 : 00 | 12 : 00 | absent | Jones | call-transfer |
| 10 : 00 | 12 : 00 | absent | Schmidt | mail |
| 16 : 00 | 18 : 00 | absent | Jones | call-transfer |

The condition (time scope, user status, and calling party's name) and a service which satisfies the conditions are specified.

Fig. 5–Call-processing instruction database.

| From | To | Place | Business |
|---|---|---|---|
| 10 : 00 | 12 : 00 | meeting room | meeting |
| 16 : 00 | 18 : 00 | guest room | meeting |

The time scope, user destination, and activity are entered.

Fig. 6–Schedule database.

analyzed to find its location and to generate actual operations for the terminal.

### 4.3 Prototype services

#### 4.3.1 AI secretary service[5)]

1) Service feature

This service processes incoming calls according to the call-processing instruction data which instructs how to process incoming calls, and schedule data registered by the user. But the user cannot register complete instructions for all possible calling parties. The feature of this service is that it can process incoming calls by using the knowledge base even if no call instruction data is registered. In this service, the knowledge of call handing which the human secretary has is used to determine the most adequate service. The following are some examples. The call-processing instruction data and schedule data are listed in Figs. 5 and 6.

*A call from Mr. Jones at 11 am.* (The call-instruction data is registered.)

In this case, the call is transferred to the meeting room. PBX sends the telephone number of the calling party and called party with the call arrival signal to the application processor. The communication agent manager invokes the communication agent of the called party. The communication agent retrieves the call-processing instruction data and the schedule data from the database through the database object. In this case, the communication agent finds that the call-transfer service is registered, and the user is in the meeting room. The communication agent selects the call-transfer service in the tion agent selects the call-transfer service in the service controls PBX through the TCSI object, and forwards the call to the meeting room.

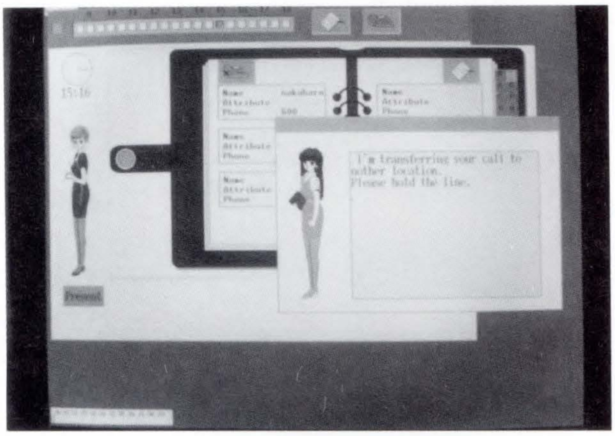


Fig. 7–Typical display of the prototype system.

*A call from Mr. Schmidt at 5 pm.* (No call-instruction data is registered.)

In this case, the call from Mr. Schmidt is transferred to the mailing system. The communication agent cannot find an applicable call-processing instruction, and determines the appropriate service by inferring the status of the calling party and called party. In this case, the status of the calling party is judged not to be important because the calling party is a colleague, and the status of the called party is judged not to be disturbed because he is in the guest room with a visitor. The communications agent concludes that the mail service is most appropriate. The communication agent transfers the call to the mailing system and invokes the mail service.

Figure 7 shows the typical display on the personal computer to input service control data. The communication agent is referred to as an AI secretary because this service can process incoming calls like a human secretary.

2) Knowledge base

Human secretaries process incoming calls

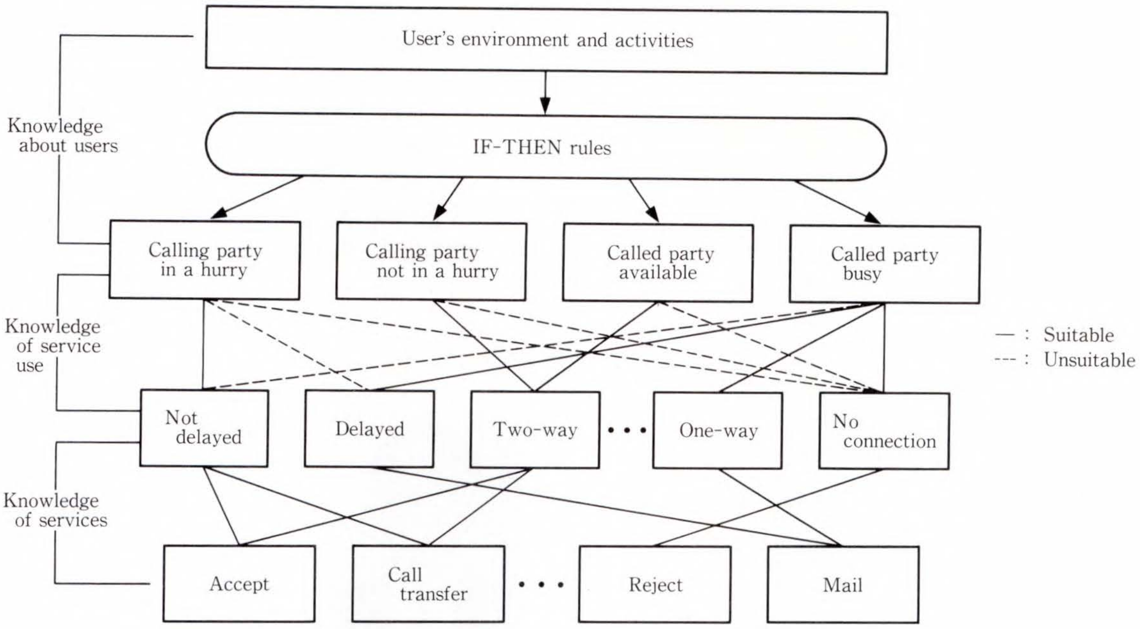



Fig. 8—Knowledge base.

appropriately without detailed instructions. Using their knowledge of the organization, location, and communication services, they select the proper service based on relatively simple information, such as calling party's name and the intended called party's destination. Therefore to realize the intelligent service feature described in the previous section, it is necessary to formalize the secretary's knowledge about call handling.

Figure 8 shows the structure of the knowledge base for this service which consists of three parts. The knowledge about users is used to infer the user status from the user's environment and activities (organization, business, etc.). This knowledge is represented by IF-THEN rules. The knowledge of services describes the properties of services available in the network. The knowledge of service use describes the relationships between abstract user statuses and service properties. It is used to evaluate the suitability of the service for the user status. The knowledge base also has inference mechanisms, which mainly use prolog-like backward chaining and are written in C for good performance and ease of future extension.

### 4.3.2 Cooperative schedule management service with mail service

This service automatically enter the data in the personal schedule database from the received mail. For example, if the received mail reports a meeting, the data about the date, place, and subject of the meeting is extracted and is entered in the schedule database. Because identifying the data to be extracted is very difficult, we fix the mail format for simplicity.

When the communication agent selects the mail service, it checks the subject in the fixed mail and judges whether the received mail reports the meeting. If it reports the meeting, the communication agent extracts the date, place, and subject of the meeting using information about the mail format, and gives them to the schedule management service. The schedule management service uses the user interface object to notify the user of a meeting or inform the user of a schedule conflict.

## 5. Conclusion

This paper proposes a new network architecture for the private network and describes the prototype system and services based on the architecture. Also, the AI secretary service, an example of the prototype services, shows how AI technology can help realize the advanced communication services. Hereafter we are going to evaluate the prototype system and services and to incorporate the results to advance the proposed architecture and services.

## References


1) Aritaka, N. et al.: Intelligent Networking and Services in the Business Communications Environment. Proc. ISS'90, B8.3, **Vol. V**, 1990, pp. 147-152.
2) ECMA: Final Draft TR/CSTA. March 1990.
3) CCITT: Baseline Document for Intelligent Network Studies at CCITT. SG XI-R24, April 1990.
4) Hakata, A. et al.: An Intelligent Network Service Prototype for Private Network. Proc. GLOBECOM '90, 408.1.1, 1990, pp. 629-634.
5) Kim, M.W. et al.: An Advanced Service Application for Private IN –Intelligent Secretary Service–. Proc. ISS'90, P6, **Vol. III**, 1990, pp. 33-36.



**Moo Wan Kim**

Integrated Communication Network Laboratory
FUJITSU LABORATORIES, KAWASAKI
Bachelor of Electronics Eng.
Osaka University 1974
Dr. of Electronics Eng.
Osaka University 1980
Specializing in Communication Software and Intelligent Network



**Norihiro Aritaka**

Systems Development Dept.
Business Switching Systems Div.
FUJITSU LIMITED
Bachelor of Electrical Eng.
Yokohama National University 1976
Master of Electrical Eng.
Yokohama National University 1978
Specializing in Business Switching System



**Akira Hakata**

Integrated Communication Network Laboratory
FUJITSU LABORATORIES, KAWASAKI
Bachelor of Instrumentation Eng.
Keio University 1977
Master of Instrumentation Eng.
Keio University 1979
Specializing in Personal Communications Network

UDC621.391.5:621.395.721.1

# The Development of Optical Fibers for Subscriber Loops

• Shigeyuki Unagami • Hiroshi Ogasawara

*(Manuscript received February 17, 1992)*

This paper describes an optical fiber system that has been tailored to applications in the subscriber loop network. The paper discusses the basic concepts of using the system, based on the authors' recent studies of the Fiber In The Loop (FITL). The paper identifies the key points of the system technology, the device technology, and power feeding. The paper compares the network architecture and several transmission schemes, and outlines the progress being made in developing devices used in the subscriber systems. The paper also analyzes one approach designed for the U.S. market, in which the double active star network topology is applied. In addition, the paper discusses the cost parity with copper and the use of a short wavelength laser diode.

## 1. Introduction

As the telecommunication infrastructure is becoming increasingly integrated, many major new technologies are being developed. In particular, many studies and field trials for Fiber In The Loop (FITL) are being conducted around the world. The development and standardization of the Broadband ISDN (B-ISDN) has been making steady progress on an international scale. At the same time, studies of optical subscriber loops, which introduce the fiber to the subscriber areas, have started independently and are tailored to the conditions prevailing in each country and how the loops will be used.

To cope with the anticipated requirements of the next centruy, it is necessary to develop a new generation of optical subscriber loop systems. The new system will be based on fiber optics which are as cost effective as the copper system for the Plain Old Telephone Service (POTS). At the same time, it is necessary to move slowly from the present system to the new digital services supported by the future B-ISDN.

This paper discusses subscriber loop topologies, transmission schemes, and device technologies, and gives an example of the authors' approach to FITL in the United States. The paper concludes with a technical discussion of the future of optical subscriber systems.

## 2. Approach to the optical subscriber loop system

To establish an economical subscriber network architecture, it is necessary to examine the transmission technology, the fields of application, the geographic distribution of the subscriber premises, and the service grade which mainly relates to the transmission bit rate. Recently, the requirements for the high-speed transfer of computer files, video signals, and many telephone channels in clustered subscriber areas have increased in Japan[1]. NTT provides a subscriber loop system, the so-called CT/RT system, in which high-speed digital data is transmitted over optical digital transmission lines. In this system, the information is multiplexed at the Remote Terminal (RT), which is set up in the local building, and transmitted to the Central Terminal (CT) located at the switching office. This is designed for the Fiber To The Building (FTTB) system. The same system can be seen in the United States. Digital Loop Carrier

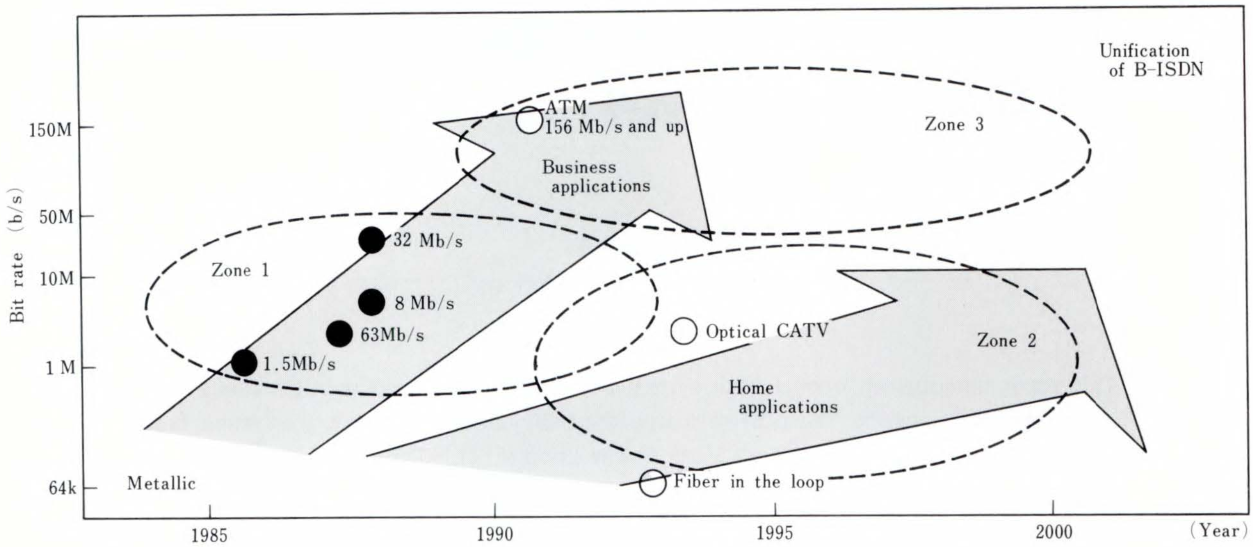



Fig. 1—Progress of the optical subscriber system.

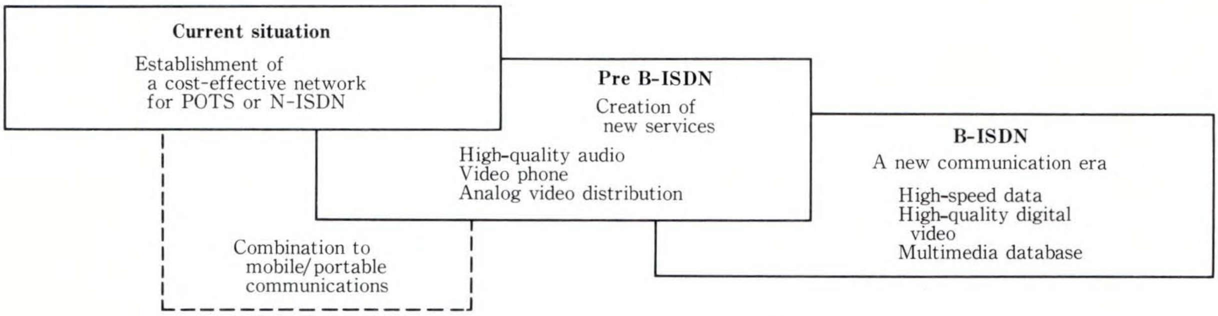



Fig. 2—Procedure for implementing FITL.

(DLC) is a feeder transmission system in which a remote terminal is set up in a plant near the curb-side[2]. DLC also transmits analog voice signals in PCM format. PCM signals are converted back to analog signals at the remote terminal and delivered to the subscriber premises.

The subscriber loop systems can be categorized into the three regions shown in Fig. 1. Current business applications using a channel capacity of less than 50 Mb/s (zone 1) is successful in providing an economical solution for businesses. The Fiber To The Curb (FTTC) systems with optionally attached video services (zone 2) is a new issue discussed here. These systems very much require new technologies and need to be used in large volumes to bring the cost down to the same level as existing copper systems. The higher bitrate transmission services (zone 3), such as computer file transfer, bidirectional video transmission, and multi-media processing, are the exclusive domain of B-ISDN and provide versatile services over a completely unified network.

B-ISDN will probably be extended to business applications before home applications due to the increasing demand for large-volume data transfer. However, the ultimate network must involve domestic customers by providing new broadband services. It is therefore necessary to make a thorough study of the FITL system for preparing the first step of the fiber infrastructure. This paper mainly focuses on the implementation of zone 2 in relation to FITL.

The procedure for implementing FITL is illustrated in Fig. 2. The main services provided by the current applications are POTS and a narrowband ISDN to replace the copper-based communication facilities with equivalent cost performance. As it is easier to install the network in the newly built areas than to replace the

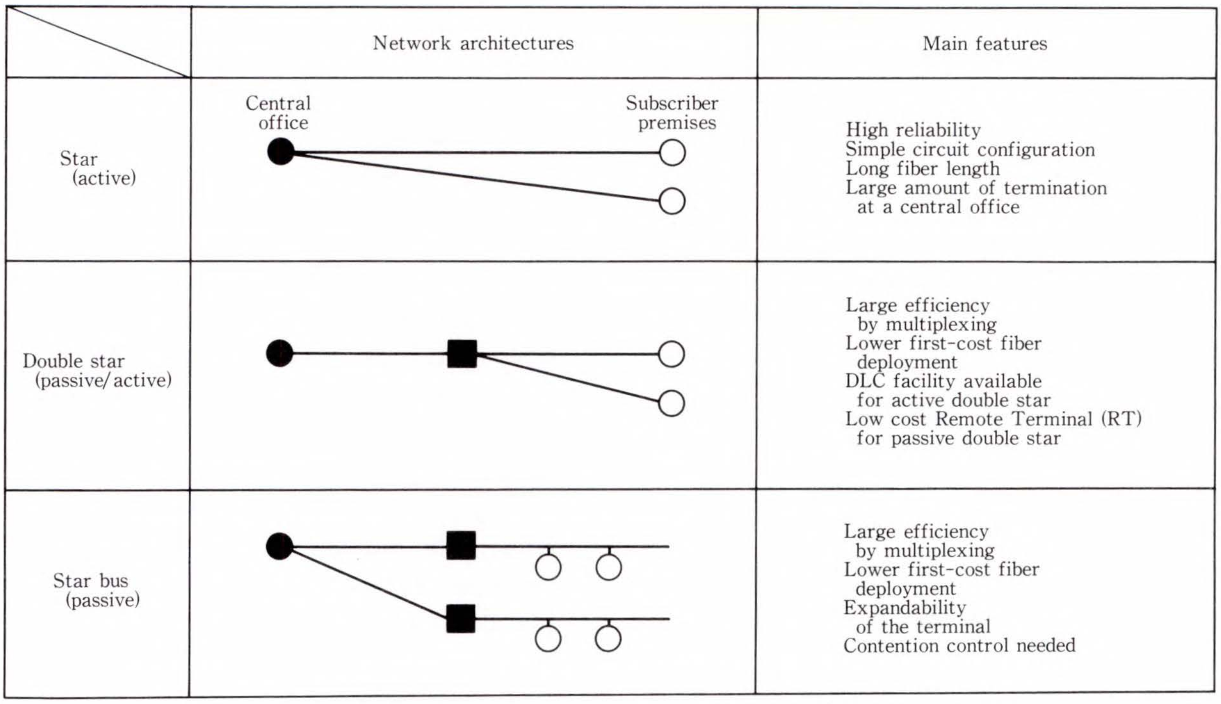



Fig. 3—Typical network architectures and main features.

copper system, it is important to consider the strategy, particularly in the first stage. Video distribution and services having higher bit rates, such as Hi-fi audio, high-speed facsimile, and video conference services, are the attractions offered by B-ISDN.

## 3. Technologies for FITL

Major issues in the development of the optical subscribed loop system are as follows[3].

### 3.1 System technology

This section mainly focuses on the network topology and the transmission schemes, which are the main issues of the system technology.

1) Network topology

Figure 3 shows the typical network architectures and their main features. The early networks for POTS were implemented with the following objectives.

i) To eliminate as many electronic components as possible
ii) To enable the facilities to be shared by many customers
iii) To use existing facilities and established technologies

The passive split star and active star networks are typical candidates which satisfy some of the above objectives. The passive split star topology extends the network like a tree structure. An optical fiber is branched at nodes located midway along the optical transmission line. Data transmitted from the center which contains all the subscriber information is broadcast to all subscribers (down link). Conversely, data from each subscriber can be sent to the center by the Time Division Multiple Access (TDMA) scheme (up link). The means of implementing the passive split star network, the fast convergence of Automatic Gain Control (AGC) and Phase Locked Loop (PLL) timing recovery functions are key technologies to be developed. The reports of several trials have been reported[4)-6)].

The other approach is an active star network utilizing the existing digital loop carrier system. Using this approach makes it is easy to displace the current remote terminal located in the curb. Although an optical fiber is dedicated to each unit at the curb side, the initial cost of the transmission equipment is reduced. The active star approach seems to be feasible in short-haul applications between the feeder cabinet and the curb terminal. Datails of the active star network are described later.

2) Transmission scheme

Figure 4 shows some typical optical transmission schemes. The system of Space Division Multiplexing (SDM) uses separate fibers for both the up and down links. Although the implementation is very simple, the applied distance is limited to about two kilometers because the cable used in about twice as expensive.

Time Domain Compression Multiplexing (TCM), or so-called optical ping-pong transmission, transmits data alternately between the transmitter and the receiver in a short frame. In this scheme, the bilateral operation of an edge emitting LED or a laser diode for both the optical transmitter and detector is proposed to reduce the number of optical components[7]. The basic advantage of TCM is that it eliminates the interference between the transmitter and

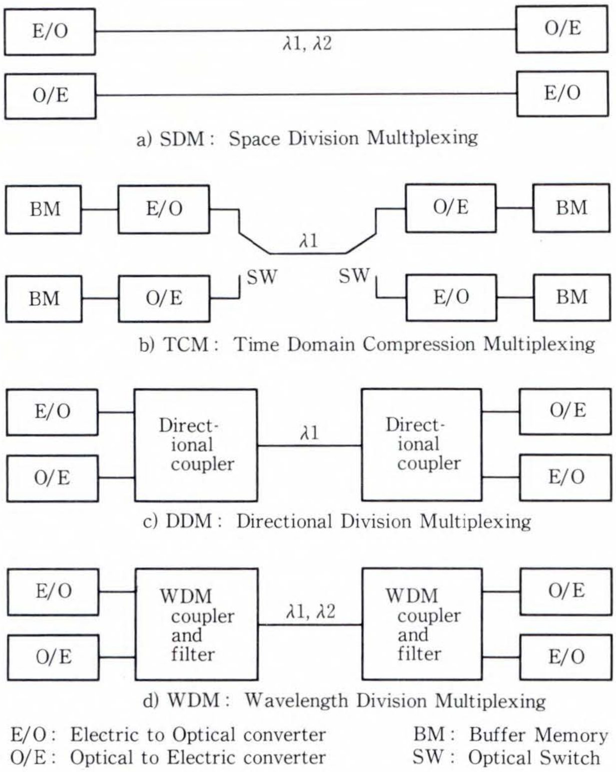



Fig. 4—Typical optical transmission schemes.

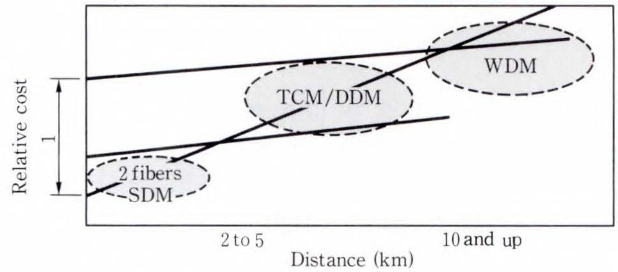



Fig. 5—Relative cost comparison.

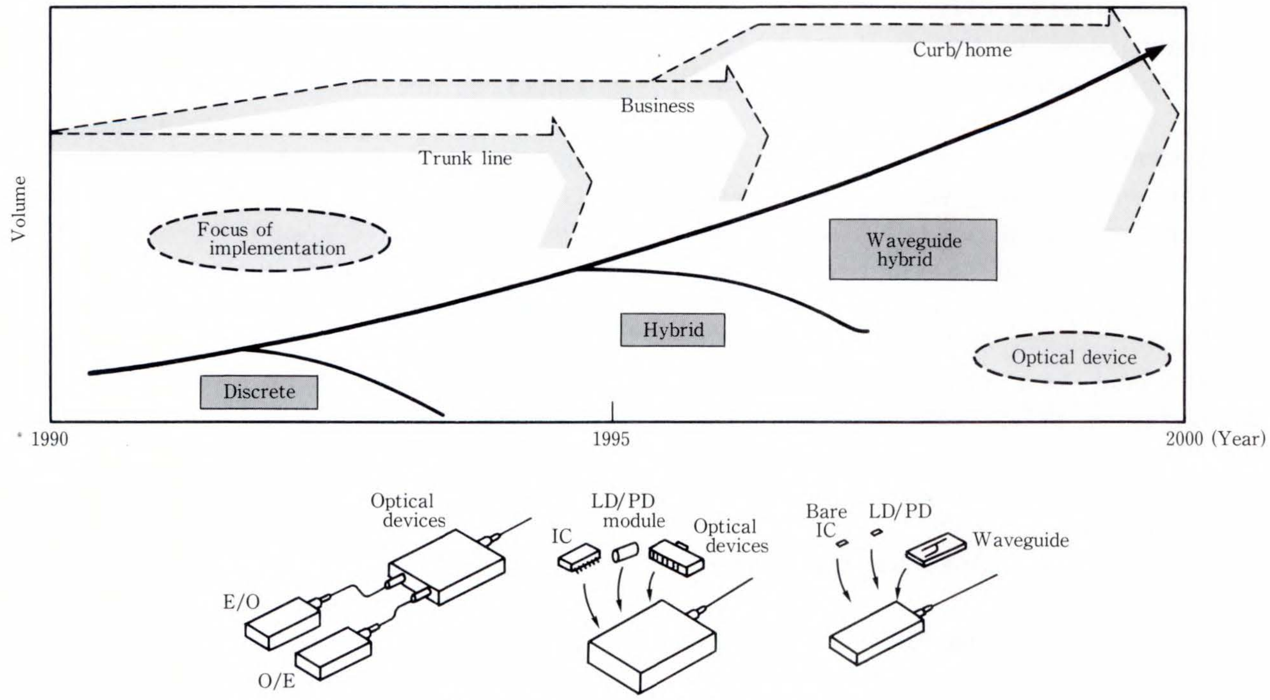



Fig. 6—Progress of optical modules.

receiver, while the transmitted bit rate increases to more than twice the terminal data rate. Consequently, there is a trade-off between the maximum amount of data to be transmitted and the hardware implementation.

Directional Division Multiplexing (DDM) is also effective for single-fiber single-wavelength facilities. The poor reflectivity of the laser diode limits the transmission distance.

Wavelength Division Multiplexing (WDM) can be used in bidirectional transmission on one fiber with two wavelengths or in video transmission with a subsidiary wavelength.

Figure 5 shows a relative cost comparison with applied fields.

### 3.2 Device technology for subscriber loop systems

Figure 6 shows the progress of optical modules[8]. The optical waveguide integrated circuit is one of the key devices enabling optical modules to be small and cost-effective in high-volume production. In the near future, alignment-free mounting technology is expected to reduce the manufacturing costs and production time.

Figure 7 shows the configuration of the waveguide WDM module. The transmitter portion, the receiver, and the WDM function are integrated on a silicon substratum. It is necessary to make the connection between the lens and the waveguide IC carefully, taking account of the sensitivity of the mode-field diameter of the waveguide IC, which is about 10 times larger than that of the laser diode.

Figure 8 shows the characteristics of the WDM with the waveguide IC. It is also important to implement an LSI which is dedicated to the electric functions.

Figure 9 shows the application field, indicating the relationship between the integrated numbers and the required bit rate. CMOS technology will be applied in a wide range of fields which use advanced technologies such as larger-scale integration by Sea of Gate (SOC)

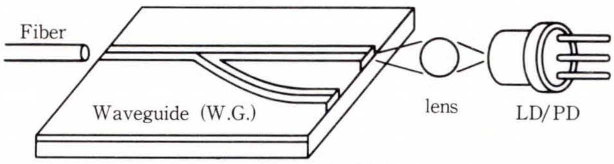



Fig. 7—Waveguide WDM module.

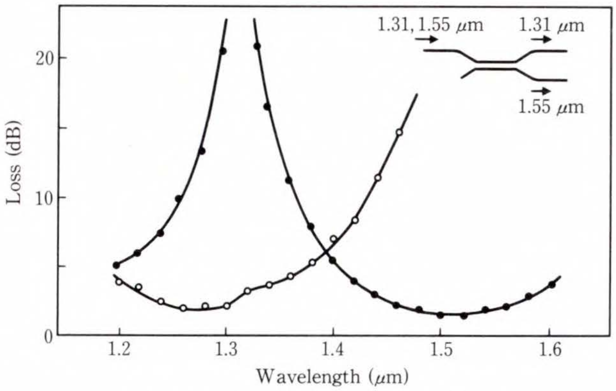



Fig. 8—Characteristics of WDM.

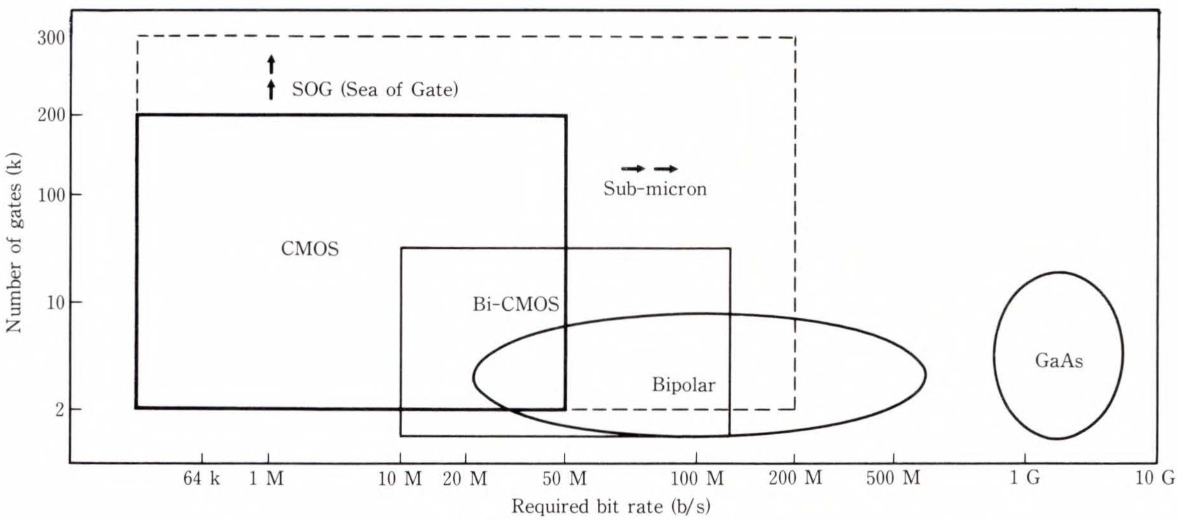



Fig. 9—Application field of LSIs.

and high-speed processing using the submicron pattern rule.

### 3.3 Power feeding

Power feeding is another important issue especially in the subscriber loop system. This section discusses remote feeding and local feeding. Table 1 compares the remote power feeding and local power feeding. Table 2 lists the main features of the battery. The design of a power feeding system is now being studied.

Table 1. Comparison of remote power feeding and local power feeding

| | Remote power feeding | Local power feeding |
|---|---|---|
| Fiber expand-ability | Poor (Power line is needed) | Good |
| Reliability | Large damage | Limited damage |
| Maintenance | Remotely controllable | Periodic check for battery |
| Economic penalty | Power line | Battery |
| Technology | Conventional technology | Small battery is required |

Table 2. Comparison of various kinds of batteries

| | Lead | Ni-Cd | Lithium |
|---|---|---|---|
| Capacity | Large-medium | Medium | Small |
| Energy density | 50 WH/l | 115 WH/l | 250 WH/l |
| Voltage | 2 V/cell | 1.2 V/cell | 3 V/cell |
| Discharge cycle | >200 times | >500 times | >200 times |
| Tolerance of over charge or discharge | Good | Excellent | Fair |
| Self discharge | Good | Fair | Excellent |
| Life cycle | 4 to 5 years | 4 to 5 years | 10 years |
| Cost | Excellent | Good | Fair |

## 4. Example of an approach for FITL

Figure 10 shows the proposed FITL system that will soon be introduced in the United States[9]. Taking account of the current subscriber transmission facilities, such as the digital loop carrier system, the double active star approach seems to be the candidate that will reduce the costs of first-stage deployment. SONET/DLC technologies are applied in the feeder portion. The 0.78-$\mu$m optical transmission rate is used in the distribution portion. The POTS interface uses DLC/PCM electronics both to exploit the economies of the large-scale implementation of current technologies and because of their proven record in the field. The sharing scale of $N$ living units at each pedestal is determined by considering the balance of costs between the curb electronics and the drop facilities.

Figure 11 shows the application environments for introducing FITL. The cost parity with copper is the single most important requirement. To meet this requirement, it is necessary to use a direct digital interface to the switch. Preparations for standardizing the transmission and next-generation switching systems are now underway.

Figure 11 shows also the impact of introducing environment systems that meet these new standards and the related requirements. Waiting three to five years before introducing FITL products appears to be too conservative when it is economically feasible to do so now. Satisfying the necessary financial conditions via the use of available new technologies seems to be the reasonable appproach.

Figure 12 shows typical trends in reducing the size of important components as the product evolves. The size of functional devices, such as

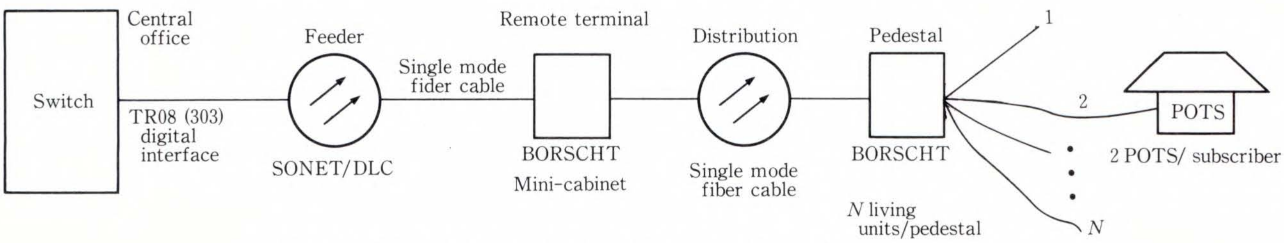



Fig. 10—FITL configuration. (Fujitsu's FITL model for active star)

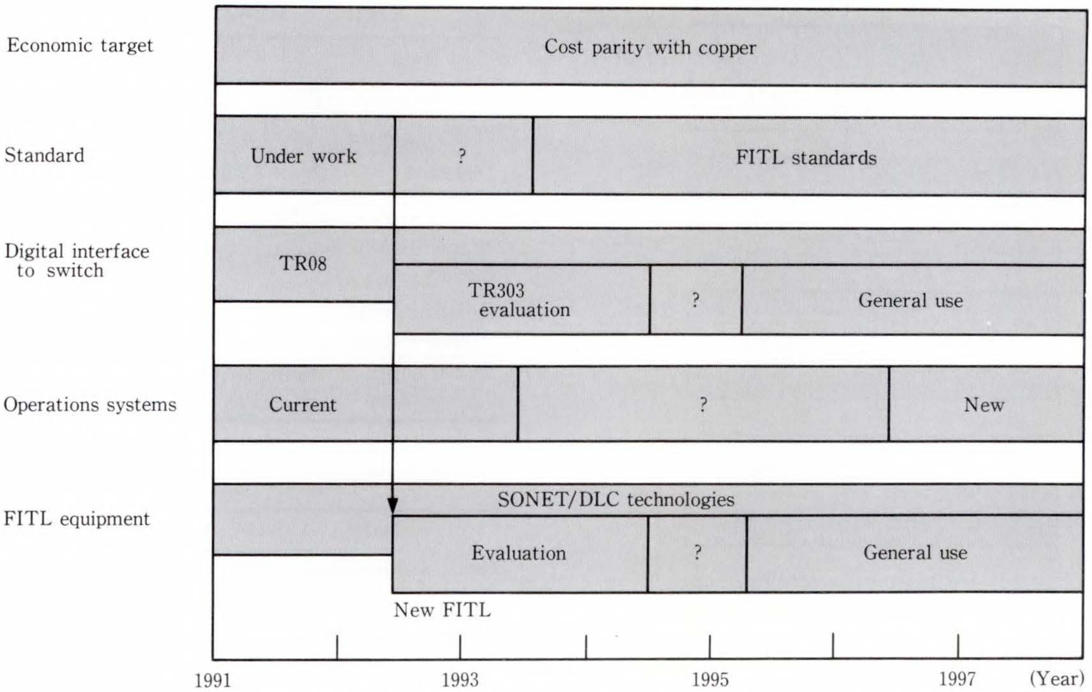



Fig. 11–FITL application environments.

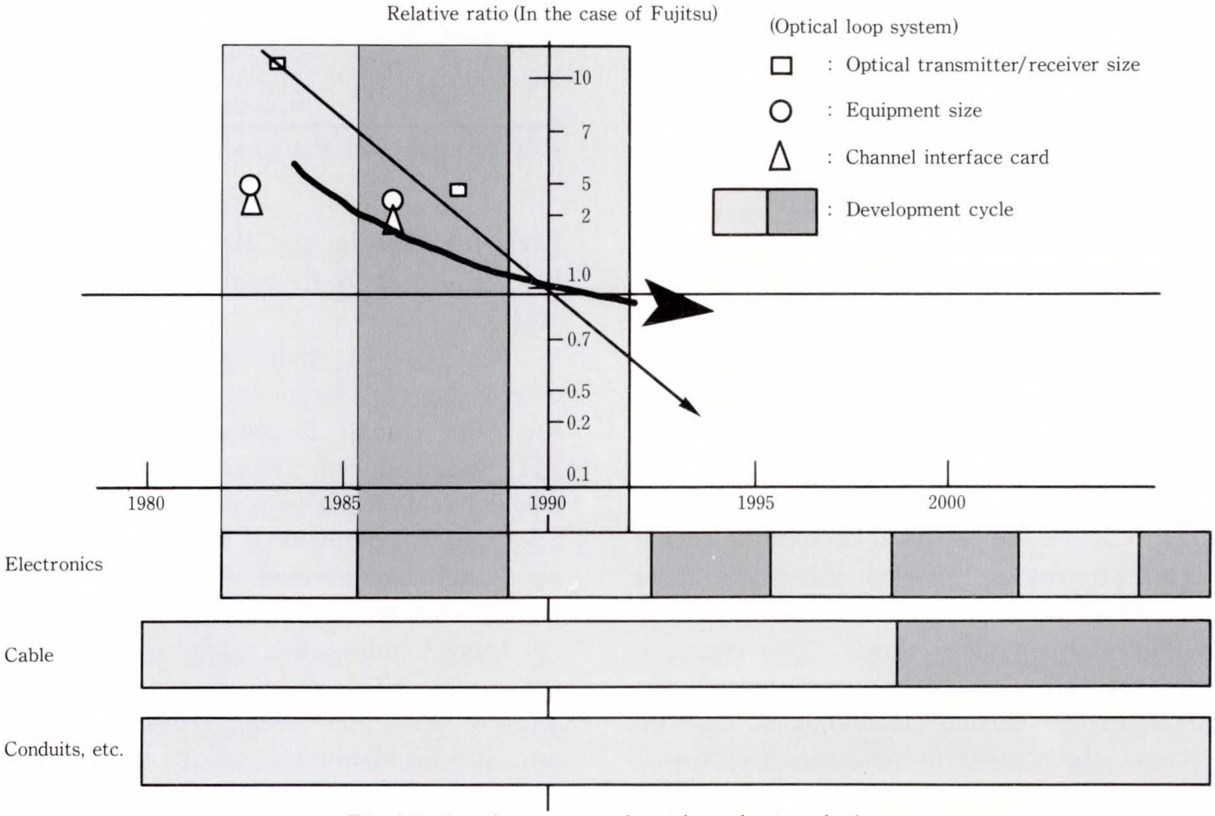



Fig. 12–Development cycle and product evolution.

optical transmitter and receiver modules, is still rapidly decreasing, while items relating to external interfaces, such as drop cable interfaces, have reached saturation, and substantial size reductions are no longer expected. Assuming that the rapid evolution of devices will continue,

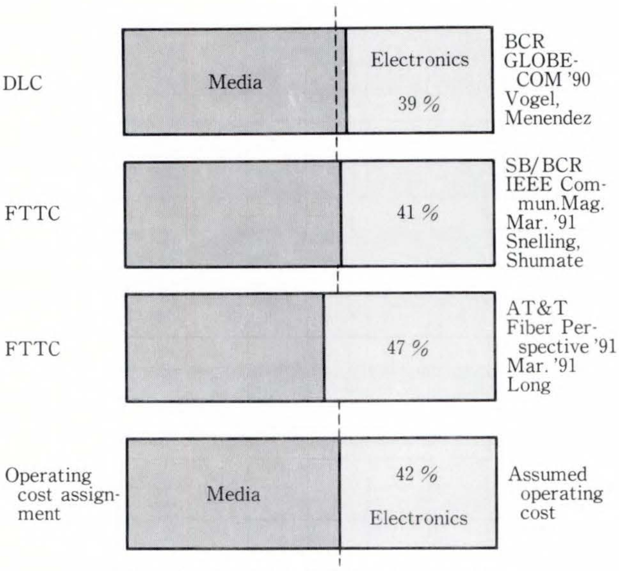



Fig. 13–Estimate of operating costs.

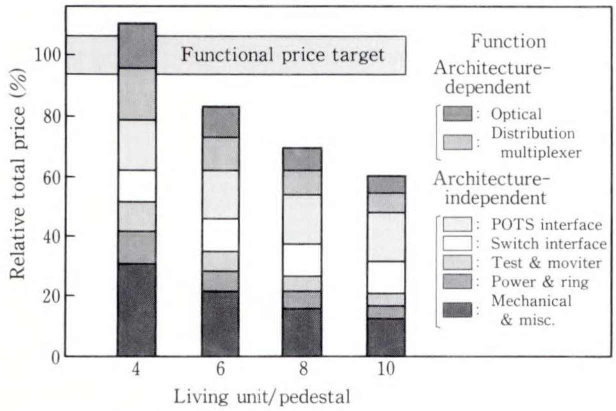



Fig. 14–Normalized total price.
(Examination of cost parity with copper)

a new product will come onto the market every three or four years. These new products will be able to accommodate new devices without much charge in size of the electrical equipment used in the curb electronics. However, things like cables and conduit are more infrastructure intensive and evolve much more slowly. This electrical equipment should be introduced through the application of current technologies that incorporate the ability to accommodate future requirements.

For this study of cost parity with copper, Fig. 13 gives the functional cost of copper distribution in the active star approach. The cost of the electrical equipment normalized to the cost parity with copper is shown in Fig. 14.

Table 3. Main specifications of FTTC

| Item | Description |
|---|---|
| Service | POTS |
| POTS/pedestal | 2 to 12 POTS/pedestal |
| Distance | Max 4 km |
| Fiber | Single-mode fiber |
| Transmission quality | BER $< 10^{-9}$ |
| Environmental temperature | $-40$ °C to $+65$ °C |
| Power supply to pedestal | Remote powering<br>DC 130V constant voltage $<$ 100 VA |

Table 4. Optical transmission

| Item | Description |
|---|---|
| Optical transmission | SDM (2 fibers/$T_x$ + $R_x$) |
| Line rate | 1.544 Mb/s |
| Line code | CMI |
| Wavelength | 0.78 $\mu$m |
| Optical source | 0.78 $\mu$m LD, $-10$ dBm |
| Optical detector | Si-PIN PD, $-46$ dBm |
| Dynamic range | 36 dB |
| Supervision | CMI code monitor<br>LD power monitor<br>Receive monitor |

SDM: Space Division Multiplexing

Each function in the electrical equipment is itemized to show its contribution to the total cost.

The cost of each function was assessed considering the current lean volume availability, since this climate is assumed to prevail when FITL is introduced. The cost parity with copper is given at six living units per pedestal. Note that functions independent of the architecture make up a significant portion of the price. POTS and the switch interfaces occupy the larger portions.

Table 3 shows the major specifications of the typical FITL system in the United States. Table 4 gives the design requirements of its optical transmission system. To ensure the most economical implementation for optical transmission within four kilometers, the 0.78 $\mu$m LDs and the silicon PIN photo diodes are used. The 0.78 $\mu$m LD is suitable in high-temperature environments of 65 °C to 85 °C and also has a good high-temperature linearity. The PIN

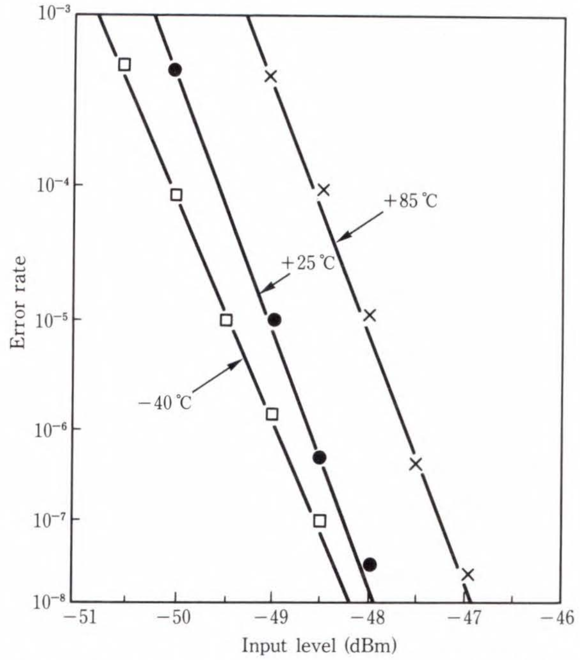



Fig. 15—Transmission performance of 1 300 SM fiber.

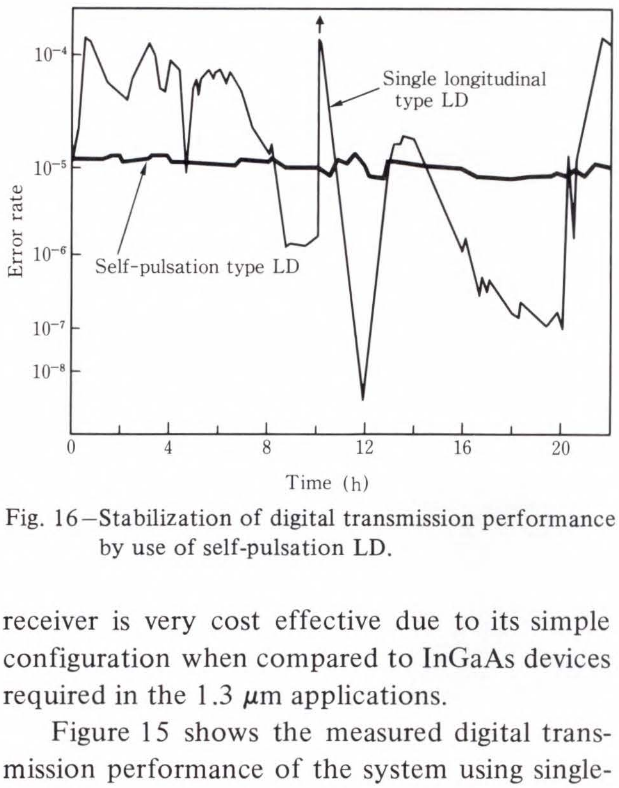



Fig. 16—Stabilization of digital transmission performance by use of self-pulsation LD.

receiver is very cost effective due to its simple configuration when compared to InGaAs devices required in the 1.3 $\mu$m applications.

Figure 15 shows the measured digital transmission performance of the system using single-mode fiber optimized for 1.3 $\mu$m. The results indicate that 1.3 $\mu$m optimized single-mode fiber has good performance at an optical transmission of 0.78 $\mu$m. The modal noise in a fiber cable section causes a variation in the transmission performance.

Figure 16 shows the authors' solution to

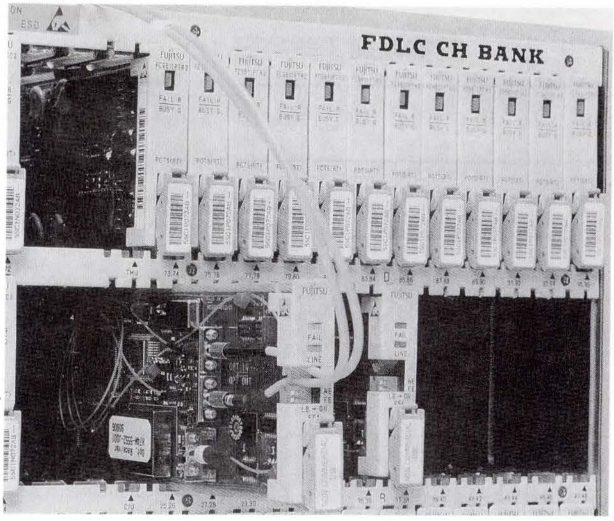



Fig. 17—Channel bank unit of FTTC.

stabilize this variation. Using a self-pulsation 0.78 $\mu$m LD reduces the deterioration in performance due to modal noise. Based on the experimental results, 2 dB of the maximum power penalty was assigned due to the modal noise when using 0.78 $\mu$m transmission on single-mode fiber optimized for 1.3 $\mu$m transmission.

The next step in introducing FITL internationally is considered to be developing a more economical 1.31 $\mu$m LD for FITL applications to replace the 0.78 $\mu$m LD.

Figure 17 shows the DLC channel bank unit.

## 5. Technical trends for future systems

Several approaches are being studied as candiates for upgrading systems in the future. The systems of the future must meet the following three requirements: To be able to handle high-speed data transmission, to serve a large number of customers, and to provide an extended range of channel numbers.

For high-speed transmission, the star configuration appears to have a number of advantages due to the inherent topological feature of point-to-point transmission. Replacing the electrical equipment with the high-speed devices is a simple means of implementing broadband services. An optical amplifier is a sophisticated device to extend the transmission distance. Use of an optical amplifier puts a margin in the budget for the transmission system. The optical amplifier is especially attractive for delivering video signals to many customers simultaneously. An erbiun-doped fiber amplifier has now reached the practical stage of development and is available for distributing video signal[10].

A high-density frequency multiplexing transmission system is a candidate for increasing the channel capacity. The Sub-Carrier Multiplexing technology (SCM) and systematic transmission is required for this purpose[11].

## 6. Conclusion

Investigation of the fibers in loop systems is now underway. This paper describes several key technologies, including network and transmission technologies, device technology and power feeding. The paper outlines the active star approach used in the United States. To reduce costs, the paper proposes the use of a short wavelength laser diode. The paper concludes with an outline of some future systems.

The investigation of the passive double star approach has been started at Fujitsu. The details will be reported later.

## References


1) Special issue: ISDN system. (in Japanese), *NTT R & D*, **38**, 7, (1989).
2) Carroll, R.L.: Optical Architecture and Interface Lightguide Unit for Fiber-to-the Home Feature of the AT & T SLC Series Carrier System. *J. Lightwave Technol.*, **7**, 11, pp. 1727-1732 (1989).
3) Mochida, Y. and Unagami, S.: Technical Subjects in the Development of the Optical Subscriber System. (in Japanese), IEICE Natl. Conv. Rec., P1-5, Panel discussion, Mar. 1990, pp. 4-185-188.
4) Proc. IEEE Workshop on Passive Optical Networks for the Local Loop, May 1990.
5) Proc. the Second IEEE Workshop on Passive Optical Networks for the Local Loop, Dec. 1990.
6) Proc. the Third IEEE Workshop on Local Optical Networks, Sep. 1991.
7) Soejima, T., Yamashita, H., and Kuwahara, H.: Experimental Optical Ping-pong Transmission System. Proc. Int. Zurich Seminar, Digital Communications, 1986.
8) Hashi, T., Isono, H., Okamoto, A., and Kunikane, T.: Optical Modules for Local Optical Netowrks. Proc. the Third IEEE Workshop on Local Optical Networks, 7.3, Sep. 1991.
9) Ogasawara, H., Kutsuwada, N., and Sentlingar, J.: An Approach for FTTS, Fiber to the Curb System, Proc. the Third IEEE Workshop on Local Optical Networks, G1.5, Sep. 1991.
10) Kinoshita, S., Okiyama, T., Sasaki, M., Horimatsu, T., and Inagaki, S.: Noise Characteristics of Erbium-Doped Amplifier by 1.48 $\mu$m Laser Diode. (in Japanese), IEICE Natl. Conv. Rec., B-784, Sep. 1991.
11) Kuwahara, H., and Unagami, S.: Optical Coherent Distribution Network using Subcarrier Multiplexing. Proc. the Third IEEE Workshop on Local Optical Networks, G3.5, Sep. 1991.



**Shigeyuki Unagami**
Subscriber Systems Develpment Dept.
FUJITSU LIMITED
Bachelor of Electrical Eng.
Nagoya University 1970
Specializing in Digital Signal Processing and Subscriber Systems



**Hitoshi Ogasawara**
Transmission Division
FUJITSU LIMITED
Bachelor of Electronics Eng.
Tokyo Institute of Technology 1966
Specializing in Digital Transmission Systems

UDC 535.14:535.417

# Quantum Noise Suppression in an Optical Interferometric System Using Optical Squeezing

● Masataka Shirasaki ● Hermann A. Haus ● Keren Bergman
● Christopher R. Doerr

*(Manuscript received February 3, 1992)*

A nonlinear interferometric optical system that operates with quantum noise below the shot noise limit is proposed, analyzed, and confirmed experimentally. Sub-shot noise performance cannot be attained with conventional linear optical systems. The proposed system uses optical squeezing produced through a new scheme, that is simple and stable. This noise suppression scheme can be applied directly to a variety of optical interferometric measurements which enables detection of a signal with less noise than the shot noise limit.

## 1. Introduction

Coherent optical light has an intrinsic quantum noise called shot noise, which is due to photon statistics. Any coherent light sent through a linear optical system remains coherent. Therefore, shot noise, which remains after all the excess noise is completely suppressed, determines the intrinsic lower noise limit in classical optical systems, i.e. systems using linear optical elements. Future optical systems will require improved performance with a high signal to noise ratio. Thus the shot noise suppression will become an important issue.

Optical squeezing was proposed as the only way of quantum noise suppression below the shot noise limit in an optical interferometer[1), 2)]. Squeezed light is produced through a nonlinear optical interaction and has quite different photon statistics from conventional coherent light. Squeezed light has also been produced experimentally using optical parametric oscillation and four-wave mixing[3)-6)]. These previous optical squeezing methods and their applications to an optical interferometer had, however, some problems, which are not only that the optical system is complicated and unstable, but also that the high pumping power which is used for squeezing is wasted after squeezing. This means that a high signal to noise ratio obtained using a squeezed light is not an advantage over a scheme that uses the high pump power itself with a correspondingly high signal to noise ratio. For these reasons, the advantage of using squeezed light has not been clear.

A new squeezing scheme using a nonlinear symmetric Mach-Zehnder interferometer[7)] was proposed to solve the above problems. This scheme is a traveling wave system which enhances the nonlinear effect. Many of the conventional schemes employ resonators for enhancement of the nonlinearity. The traveling wave system allows the proposed scheme to work with high frequency and short optical pulses. This scheme reuses the pump power employed for the squeezing as the probe light in the interferometer. The optical power is not wasted in this scheme, and thus the advantage of using squeezed light is unequivocal.

This scheme can be realized practically with a nonlinear ring reflector when the input light consists of pulses[8)]. It was theoretically confirmed that this scheme can reach a quantum noise which is lower than the noise produced by conventional schemes using a squeezed vacuum and coherent probe[9),10)]. Noise suppression using this scheme was confirmed experimentally[11),12)], and approximately 5 dB suppression below the shot noise level was observed.

In this paper, the proposed squeezing method is reviewed and some recent experimental results are shown.

## 2. Proposed scheme for squeezing

The nonlinear symmetric Mach-Zehnder interferometer for squeezing is shown in Fig. 1. The interferometer consists of two 50/50 beam splitters and an identical Kerr medium in both arms. The pump light in a coherent state is fed into one of the two input ports. The small-signal light, which is supposed to be vacuum in the discussion later, is fed into the other input port. After the pump and the signal are equally mixed at the first 50/50 beam splitter, two coherent lights, whose average amplitudes are nearly equal, enter the Kerr medium in each arm. These coherent lights, which are self-phase modulated in the Kerr media, interfere at the second 50/50 beam splitter. Then one output will be a squeezed signal and the other the pump light, which is slightly different from the input pump. When the input signal is vacuum, squeezed vacuum is generated. For the theoretical treatment of this squeezer, Appendix 1 should be consulted.

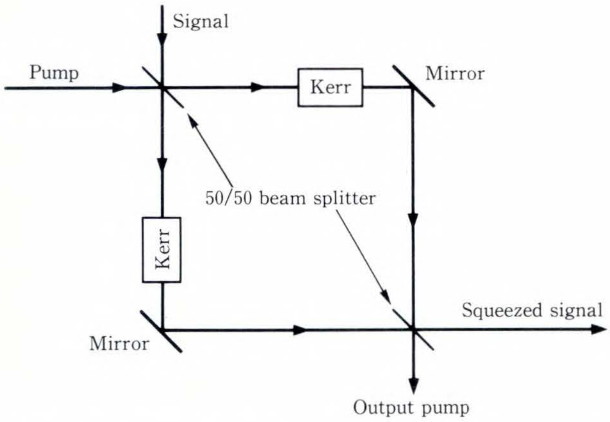



Fig. 1—A scheme proposed for squeezing. It is a nonlinear symmetric Mach-Zehnder interferometer consisting of two 50/50 beam splitters and identical Kerr media in both arms. A small signal and a large pump with a photon number of $|\beta|^2$ are the inputs to the squeezer. A slightly modulated pump and a squeezed signal are the outputs.

The results show that the angle of the squeezing axis is approximately

$$\frac{\phi}{2} = \frac{\pi}{4} + \frac{1}{2}\tan^{-1}\left(\frac{\kappa}{2}|\beta|^2\right). \qquad \ldots\ldots\ldots (1)$$

Note that this expression does not include the phase due to the classical cross-phase modulation from the pump to the signal. Here, $\kappa$ is the quantity of the nonlinearity in the arm and $|\beta|^2$ corresponds to the pump intensity. The squeezed amplitude fluctuation $U_1$ is a function of the pump intensity, and is shown with an approximation as

$$U_1 = \frac{1}{4}\left(\sqrt{1 + \frac{\kappa^2}{4}|\beta|^4} - \frac{\kappa}{2}|\beta|^2\right)^2. \qquad (2)$$

The fluctuation $U_1$ calculated without approximation is shown in Fig. 2 as a function of the nonlinearity when the input signal is vacuum. The amplified fluctuation $U_2$, which means the fluctuation in the direction perpendicular to the squeezed fluctuation direction, and the product of $U_1$ and $U_2$ are also shown in Fig. 2. This figure shows that highly squeezed vacuum is obtained without an enhancement of the uncertainty product for a properly chosen nonlinearity.

It is known that the output noise of a

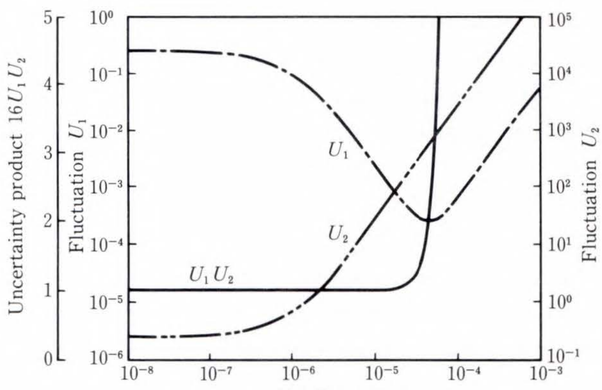



Fig. 2—Theoretical curves for the minimum amplitude fluctuation $U_1$, maximum amplitude fluctuation $U_2$, and their product $U_1U_2$ as a function of the nonlinearity when the input pump photon number is $|\beta|^2 = 10^6$, and the input signal is vacuum.

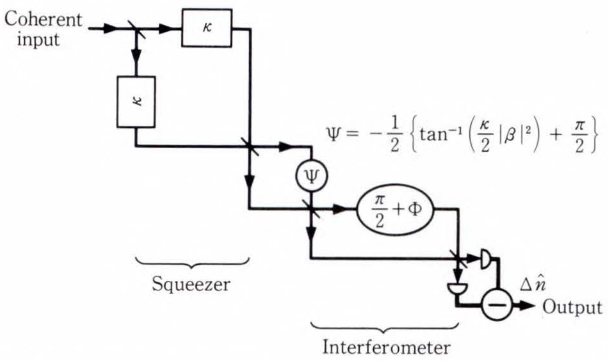



Fig. 3—Proposed phase sensitive interferometer. The output pump is reused as the probe light for the interferometer. $\Psi$ adjusts the phase so that the squeezing axis is perpendicular to the probe phase. The output is the photon number difference detected by the balanced detector. The small value $\Phi$ is to be measured.

balanced detector is less than the shot noise in a phase sensitive interferometer when squeezed vacuum is fed into the unused port with a proper phase. The squeezed light generated with the squeezer described above is applied to a phase sensitive interferometer. Here, the output pump of the squeezer is reused as probe light for the interferometer as shown in Fig. 3. Because the additional phase in the signal due to the cross-phase modulation is canceled via the output-pump phase due to the self-phase modulation, these phase factors do not affect the interference. The phase shifter $\Psi$ adjusts the axis of the squeezed vacuum to be parallel to the amplitude of the probe light. The outputs from the interferometer are received by a balanced detector, which counts the photon number difference between the two detectors. The formalism for the output of the balanced detector is derived without approximation in Appendix 2. This interferometer system theoretically shows much lower quantum noise than the shot noise limit as shown in Fig. 4, where the input photon number is assumed to be $10^6$. The quantum noise is the shot noise which is equal to the input photon number, when the nonlinearity is negligibly small. This figure also shows that the noise can be suppressed by 4 orders of magnitude. Of course, the noise suppression will be limited by the quantum efficiency of the optical detectors, optical

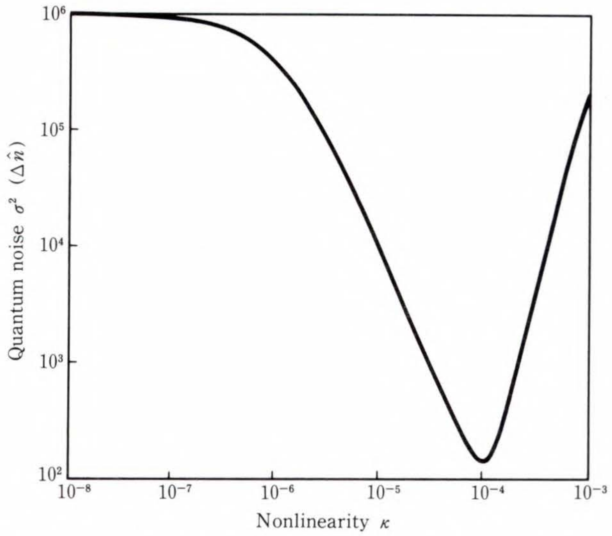



Fig. 4—Theoretical curve for the output noise of the system shown in Fig. 3 as a function of the nonlinearity $\kappa$ when the input photon number is $|\beta|^2 = 10^6$ and $\Phi = 0$. The shot noise $|\beta|^2$ appears when the nonlinearity is negligibly small.

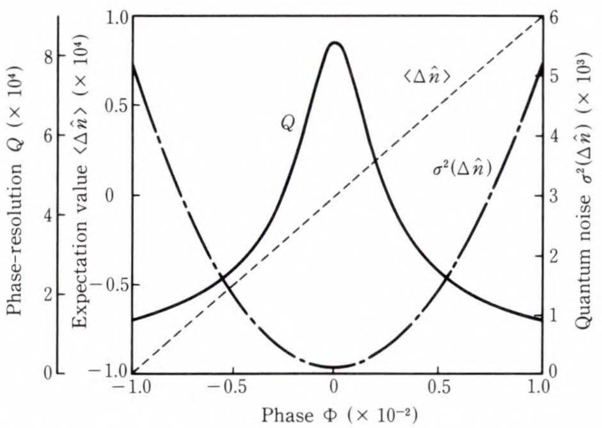



Fig. 5—Theoretical curve for the average and the noise in the output of the system shown in Fig. 3 as a function of the phase $\Phi$. $Q$ value which corresponds to the phase resolution is also shown. The input photon number $|\beta|^2 = 10^6$ and nonlinearity $\kappa = 10^{-4}$

losses, misalignment of the system, etc. For example, 1/10 of optical loss or 90 % quantum efficiency of the detector will limit the noise suppression to 10 dB.

Using the noise described earlier, the phase resolution in an interferometer is discussed. $Q$ is defined as the inverse of the phase fluctuation corresponding to the quantum noise in the output,

$$Q = \frac{d < \Delta\hat{n} >}{d\Phi} / \sqrt{\sigma^2(\Delta\hat{n})}. \qquad \ldots\ldots (3)$$

A high $Q$ value will provide a high phase resolution. Figure 5 shows $<\Delta \hat{n}>$, $\sigma^2(\Delta \hat{n})$, and $Q$, derived from Equations (A 9), (A 10), and (3), as functions of the interferometer phase $\Phi$, when $|\beta| = 10^3$. Here $\kappa$ is $10^{-4}$, which makes $\sigma^2(\Delta\hat{n})$ minimum as described in Appendix 2. For classical interferometers, which use only coherent light, the maximum $Q$ value is $|\beta|$ (with the shot noise limit). Figure 5 shows that the $Q$ value with the proposed system increases up to about 80 times of $|\beta|$, $Q$ value of classical interferometers. The best $Q$ value for an input photon number of $10^6$, $0.84\times|\beta|^{5/3}$, is higher than the maximum $Q$ value, $|\beta|^{3/2}$, obtained with an interferometer using squeezed vacuum and a coherent probe.

When the input pump light is in the form of optical pulses, the squeezer can be realized with a nonlinear ring reflector. Figure 6 shows such a squeezer using a fiber ring, where the fiber has Kerr nonlinearity. The scheme of the nonlinear ring reflector is equivalent to the nonlinear symmetric Mach-Zehnder interferometer as shown in Fig. 1. Every formula in Appendix 1 can be applied to the ring reflector. The incoming pulses (corresponding to the pump in Fig. 1) are divided into two equal pulses with the 3 dB coupler. These pulses cotravel in the nonlinear fiber ring. The fiber maintains the polarization of the light. These pulses have no mutual interaction except when the pulses meet in the fiber. If the pulse duration is much shorter than the pulse separation, this interaction can be neglected. They return to the 3 dB coupler, where they interfere to become squeezed vacuum and the output pump. The output pump is propagating backward along the pump input fiber, and the squeezed vacuum is emitted from the other output port of the 3 dB coupler as shown in Fig. 6.

The backward-travelling pump light can be completely extracted using an optical circulator, or may be partially extracted with a partially reflecting mirror. This latter situation results in a reduction of interferometer sensitivity compared to the case when the pump is saved in its entirety. However, for convenience, a partially reflecting mirror is used in the experimetns.

## 3. Experiments for the noise suppression

The noise suppression was experimentally confirmed using homodyne detection of the squeezed vacuum with the output pump as the local oscillator, as shown in Fig. 7. This situation is equivalent to the phase sensing interferometer shown in Fig. 3, and the noise in the system in Fig. 7 should be the same as that in Fig. 3. In the experiment, a 1.3 $\mu$m wavelength mode-locked Nd: YAG laser was used as the pump power. The pulse duration is 100 ps, the pulse

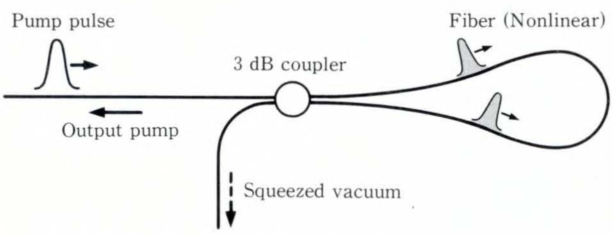



Fig. 6–Practical realization of the proposed squeezer with pump pulses. The input pulses are divided by a 3 dB coupler into two identical versions counter-propagating in a polarization maintaining fiber with Kerr nonlinearity. The output pump returns in the input fiber and the squeezed vacuum is emitted from the other port of the coupler. This scheme enables the self-stabilization of the lengths of the two arms in the squeezer.

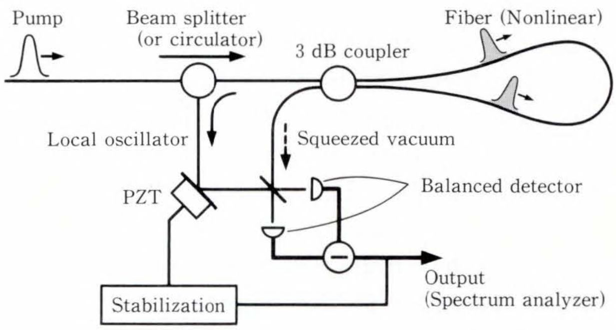
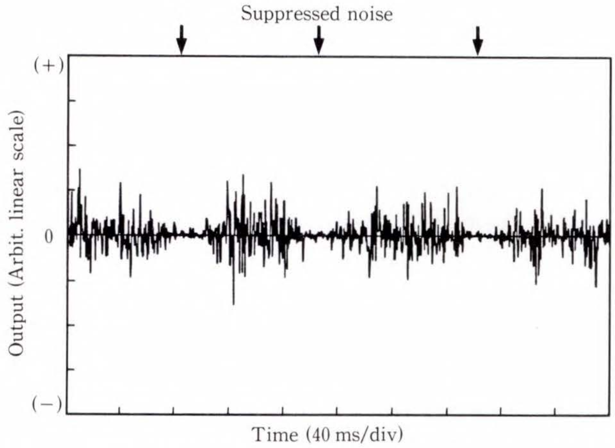
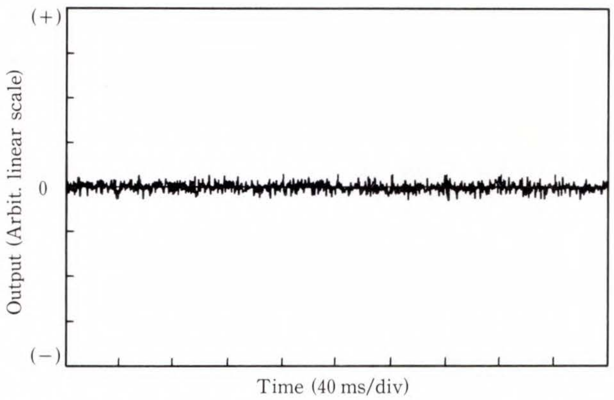



Fig. 7–Experimental setup using the squeezer shown in Fig. 6. The output pump is taken out with a beam splitter and phase-adjusted with a PZT phase modulator. The generated squeezed vacuum is observed with the homodyne detector with the output pump as a local oscillator. For the noise spectrum observation, the PZT phase is stabilized by an electronic circuit.




a) The output of the balanced detector with the PZT phase swept without operation of the stabilization circuit. At certain phases, the noise is suppressed by the squeezed vacuum.




b) The output corresponding to the shot noise is observed by blocking the squeezed vacuum entering the the balanced detector.

Fig. 8—Observation of the output of the balanced detector in the time domain. The output is filtered with a 49-51 kHz pass band. The vertical axes in a) and b) have the same scale. The suppressed noise with the squeezer in a) is lower than the shot noise in b).

repetition rate is 100 MHz, and the fiber length is 50 m. Thus, the pulse separation is 100 times the pulse duration. The average pump power in the fiber is 100 mW, which gives a nonlinear phase shift of 1.4 rad through the fiber.

The noise level obtained for squeezed vacuum with a balanced homodyne detector depends on the relative phase between the local oscillator and the squeezed vacuum. That is, the noise level varies above and below the shot noise level when the phase $\Psi$ is swept as in Fig. 3. In the experiment, this phase change was provided by a mirror mounted on a piezo-electric crystal (PZT) inserted in the probe light path as shown in Fig. 7. The noise variation was observed in the time domain as a function of the local oscillator phase with the system free-running {see Fig. 8a)}. The horizontal axis indicates the time or increasing $\Psi$ swept by the PZT modulator. The vertical axis indicates the actual photocurrent of the balanced detector after filtering with a 2 kHz bandwidth at 50 kHz. It is clear that the noise changes, depending on the local oscillator phase. For comparison, Fig. 8b) shows the shot noise observed by blocking the squeezed vacuum input. This noise level was also confirmed by the calibration with a flash light with the same photocurrent. When the phase of the local oscillator is correct, the noise is clearly less than the shot noise.

The noise observation described above was carried out for a frequency with a narrow band width. It is important for applications of squeezed light that the noise is suppressed in a certain frequency band. Therefore, it is desirable to observe the noise spectrum is the frequency domain. In order to do this, the system must be stable, because it takes some time for a spectrum analyzer to sweep the frequency in the band width. An electronic feedback circuit to the PZT modulator was developed for the stabilization. This circuit controls the PZT to minimize the output noise of the balanced homodyne detector. The PZT is modulated at 250 Hz with a small amplitude. This makes a modulation of the noise intensity in the output. This modulation of the noise can be extracted by the following procedure. First, a part of the output is taken out and filtered with a high-pass filter with a 100 kHz cutoff frequency. This eliminates the large noise caused by the laser classical noise. This also removes the DC component which may occur from some imbalance. Then this signal has only the quantum noise with zero average. Next, the magnitude of the noise in this signal is obtained by a half-wave rectifier. Next, this is filtered with a band-pass filter at 250 Hz. Then this is multiplied by the PZT modulation signal to get the feedback signal. If the phase $\Psi$ is

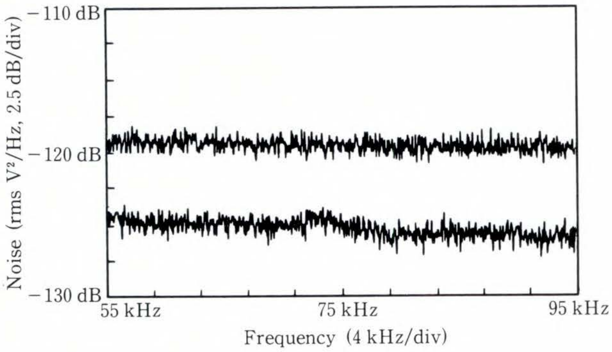



Fig. 9—Noise spectra for the output of the balanced detector with an observation range of 55-95 kHz. The upper trace shows the shot noise and the lower trace shows the suppressed noise with the squeezer. 5-6 dB noise suppression below the shot noise level is apparent in the entire range using the system.

correctly adjusted, this feedback signal will be zero. The feedback signal is superposed upon the PZT driver.

The stabilization circuit developed enabled the observation of the noise spectrum. The result is shown in Fig. 9. The observation window is 55-95 kHz. The upper trace indicates the shot noise level obtained by the way explained for Fig. 8b). The lower trace indicates the suppressed noise using the squeezed vaccum. Each trace consists of 100 averages. It is clear that the noise levels are almost flat within this window, and that 5-6 dB noise suppression compared with the shot noise level was attained by applying the squeezed vacuum.

## 4. Conclusion

Optical interferometry with quantum noise below the shot noise limit was achieved using a new method for generation of squeezed light. This system is simple and can be stabilized with an electronic feedback circuit.

The optical power source in this experiment is a mode-locked Nd: YAG laser which produces short pulses. Pulsed light is effective for nonlinear optical systems because of its high peak power with low average power. In the future, this pulsed light will be generated by mode-locked semiconductor lasers or fiber lasers for squeezing applications, allowing for more simple and compact systems.

## Appendices

### Appendix 1

The input photon operators for the signal and the pump of the squeezer are defined as $\hat{a}$ and $\hat{b}$, respectively. The lightwaves in the two arms of the squeezer are linear superpositions of the signal and the pump in the input ports. The lightwave in each arm undergoes a phase shift caused by the Kerr nonlinearity, which is proportional to the photon number. The lightwaves then interfere at the output. The exact expression for the output signal using the input operators is

$$\begin{aligned}\hat{f}=&\frac{1}{\sqrt{2}}[\exp\{\frac{i\kappa}{2}(\hat{a}^\dagger+\hat{b}^\dagger)(\hat{a}+\hat{b})\}\frac{\hat{a}+\hat{b}}{\sqrt{2}}\\&+\exp\{\frac{i\kappa}{2}(\hat{a}^\dagger-\hat{b}^\dagger)(\hat{a}-\hat{b})\}\frac{\hat{a}-\hat{b}}{\sqrt{2}}]\\=&\frac{1}{2}[\exp\{\frac{i\kappa}{2}(\hat{a}^\dagger\hat{a}+\hat{a}^\dagger\hat{b}+\hat{a}\hat{b}^\dagger+\hat{b}^\dagger\hat{b})\}\\&+\exp\{\frac{i\kappa}{2}(\hat{a}^\dagger\hat{a}-\hat{a}^\dagger\hat{b}-\hat{a}\hat{b}^\dagger+\hat{b}^\dagger\hat{b})\}]\hat{a}\\&+\frac{1}{2}[\exp\{\frac{i\kappa}{2}(\hat{a}^\dagger\hat{a}+\hat{a}^\dagger\hat{b}+\hat{a}\hat{b}^\dagger+\hat{b}^\dagger\hat{b})\}\\&-\exp\{\frac{i\kappa}{2}(\hat{a}^\dagger\hat{a}-\hat{a}^\dagger\hat{b}-\hat{a}\hat{b}^\dagger+\hat{b}^\dagger\hat{b})\}]\hat{b}. \quad \ldots \text{(A 1)}\end{aligned}$$

where $\kappa$ is a measure of the Kerr nonlinearity, proportional to the length of the Kerr medium. In the following calculations, $\hat{b}^\dagger\hat{b}$ is a large pump and $\kappa$ is small, yet the nonlinear phase shift $\kappa\hat{b}^\dagger\hat{b}$ is assumed to be larger than unity. Also, $\kappa^2\hat{a}^\dagger\hat{a}\hat{b}^\dagger\hat{b}$ is assumed to be much less than unity. First, the term $\hat{a}^\dagger\hat{a}$ in the exponent is neglected because of the small signal assumption. Next $\kappa\hat{a}^\dagger\hat{b}$ and $\kappa\hat{a}\hat{b}^\dagger$ are regarded to commute with $\kappa\hat{b}^\dagger\hat{b}$ because the commutators of these terms are extremely small under the above assumption. Then the term $\kappa\hat{b}^\dagger\hat{b}$ in the exponent of Equation (A1) can be factored out. After these steps, the output formula is

$$\begin{aligned}\hat{f}=&\exp(\frac{i\kappa}{2}\hat{b}^\dagger\hat{b})\,(\frac{1}{2}[\exp\{\frac{i\kappa}{2}(\hat{a}^\dagger\hat{b}+\hat{a}\hat{b}^\dagger)\}\\&+\exp\{-\frac{i\kappa}{2}(\hat{a}^\dagger\hat{b}+\hat{a}\hat{b}^\dagger)\}]\hat{a}\\&+\frac{1}{2}[\exp\{\frac{i\kappa}{2}(\hat{a}^\dagger\hat{b}+\hat{a}\hat{b}^\dagger)\}\\&-\exp\{-\frac{i\kappa}{2}(\hat{a}^\dagger\hat{b}+\hat{a}\hat{b}^\dagger)\}]\hat{b}). \quad \ldots\ldots \text{(A2)}\end{aligned}$$

Because the product terms of $\hat{a}$ and $\hat{b}$ in the exponent are small, one approximately obtains

$$\hat{f} = \exp(\frac{i\kappa}{2}\hat{b}^\dagger\hat{b})\{(1+\frac{i\kappa}{2}|\beta|^2)\hat{a} + \frac{i\kappa}{2}\beta^2\hat{a}^\dagger\}$$
$$= \exp(\frac{i\kappa}{2}\hat{b}^\dagger\hat{b})(\mu\hat{a} + \nu\hat{a}^\dagger), \qquad \text{(A3)}$$

where

$$\mu = 1+\frac{i\kappa}{2}|\beta|^2, \nu = \frac{i\kappa}{2}\beta^2. \qquad \text{(A4)}$$

Here, a coherent state is used for the pump, and, therefore, the square root of the fluctuation of $\hat{b}$ is much smaller than the expectation value of $\hat{b}$. This allows the treatment of $\hat{b}$ as a classical quantity $\beta$ in the linearized terms. But the exponential factor must remain as an operator, because the fluctuation of this factor may affect the squeezing. In addition, the values of $\mu$ and $\nu$ satisfy

$$|\mu|^2 - |\nu|^2 = 1. \qquad \text{(A 5)}$$

Equation (A 3) with (A 5) is the well-known formula for a squeezing. Thus optical squeezing is obtained with this scheme. In particular, squeezed vacuum is obtained from no excitation in the signal input port.

On the other hand, most of the pump power emerges from another output port of the interferometer. By the same procedure, one finds the output pump.

$$\hat{g} = \exp(\frac{i\kappa}{2}\hat{b}^\dagger\hat{b})\,\hat{b}. \qquad \text{(A 6)}$$

It should be noted that the output pump and the output signal have the same exponential factor. This suggests that this exponential factor is canceled when the output pump and the output signal interfere.

**Appendix 2**

A phase sensitive interferometer whose inputs are provided by the squeezer is analyzed. It is known that the minimum output noise at the balanced detector is obtained when the long axis of the squeezed vacuum is parallel to the complex amplitude of the probe light. The axis of the squeezed vacuum is obtained from the phase of $\sqrt{\mu\nu}$ in Equation (A4), and so

$$\Psi = -\frac{1}{2}\{\tan^{-1}(\frac{\kappa|\beta|^2}{2}) + \frac{\pi}{2}\}. \qquad \text{(A7)}$$

The system in Fig. 3 has two inputs at the first beam splitter, whose photon operators are $\hat{a}$ and $\hat{b}$. But the input $\hat{a}$ is in a vacuum state. The other input is in a coherent state with an amplitude $\beta$. In the following calculations, $\hat{c}$ and $\hat{d}$ defined as the photon operators for the two arms in the squeezer are used, where they are in coherent states $|\beta/\sqrt{2}>$ and $|-\beta/\sqrt{2}>$, respectively. These operators produce the output photon operators $\hat{c}_o$ and $\hat{d}_o$. Using this notation, the noise in the output of the balanced detector is analyzed.

The lights $\hat{c}$ and $\hat{d}$ are phase-modulated in Kerr media, then converted into $\exp(i\kappa\hat{c}^\dagger\hat{c})\hat{c}$ and $\exp(i\kappa\hat{d}^\dagger\hat{d})\hat{d}$, respectively. The output of the balanced detector, $\Delta\hat{n}$, is

$$\begin{aligned}\Delta\hat{n} &= \hat{c}_o{}^\dagger\hat{c}_o - \hat{d}_o{}^\dagger\hat{d}_o\\ &= \sin\Psi\cos\Phi(\hat{c}^\dagger\hat{c} - \hat{d}^\dagger\hat{d})\\ &\quad - (\sin\Phi + i\cos\Psi\cos\Phi)\,\hat{c}^\dagger\exp\{-i\kappa\hat{c}^\dagger\hat{c}\}\exp\{i\kappa\hat{d}^\dagger\hat{d}\}\,\hat{d}\\ &\quad - (\sin\Phi - i\cos\Psi\cos\Phi)\,\hat{d}^\dagger\exp\{-i\kappa\hat{d}^\dagger\hat{d}\}\exp\{i\kappa\hat{c}^\dagger\hat{c}\}\hat{c}. \qquad \text{(A8)}\end{aligned}$$

After some calculations, the first and the second order moments of the output are

$$<\Delta\hat{n}> = |\beta|^2\exp\{|\beta|^2(\cos\kappa - 1)\}\sin\Phi. \qquad \text{(A9)}$$

$$<(\Delta \hat{n})^2> = |\beta|^2 + \frac{|\beta|^4}{2}(1 - \sin^2\Psi \cos^2\Phi)$$
$$+|\beta|^4 \exp\{|\beta|^2(\cos\kappa - 1)\} \sin\kappa \sin 2\Psi \cos^2\Phi$$
$$+\frac{|\beta|^4}{2}\exp\{|\beta|^2(\cos 2\kappa - 1)\}(\sin^2\Phi - \cos^2\Psi\cos^2\Phi). \quad \cdots\cdots\cdots\cdots\cdots \text{(A10)}$$

In particular, when $\Phi = 0$, the noise is

$$\sigma^2(\Delta\hat{n}) = <(\Delta\hat{n})^2> - <\Delta\hat{n}>^2$$
$$= |\beta|^2 + |\beta|^4 \exp\{|\beta|^2(\cos\kappa - 1)\}\sin\kappa \sin 2\Psi$$
$$+\frac{|\beta|^4}{2}[1 - \exp\{|\beta|^2(\cos 2\kappa - 1)\}]\cos^2\Psi. \quad \cdots\cdots\cdots\cdots\cdots\cdots\cdots\cdots \text{(A11)}$$

Next, the behavior of Equation (A11) is analyzed around the noise minimum point. Here, $\kappa|\beta|^2 >> 1$ and $\kappa^2|\beta|^2 << 1$ are assumed. Then Equation (A11) is approximately represented

$$\sigma^2(\Delta\hat{n}) \cong \frac{1}{\kappa^2|\beta|^2} + \frac{5}{12}\kappa^4|\beta|^6. \quad \ldots \text{(A12)}$$

For a given $\beta$, this noise is minimized to be

$$\left(\frac{45}{16}\right)^{1/3} \times |\beta|^{2/3},$$

when $\kappa = \left(\frac{6}{5}\right)^{1/6} \times |\beta|^{-4/3}$.

## References


1) Caves, C. M.: Quantum-mechanical noise in an interferometer. *Phys. Rev.*, **D23**, pp. 1693-1708 (1981).
2) Bondurant, R. S., and Shapiro, J. H.: Squeezed states in phase-sensing interferometers. *Phys. Rev.*, **D30**, pp. 2548-2556 (1984).
3) Wu, L.-A., Kimble, H. J., Hall, J. L., and Wu, H.: Generation of Squeezed States by Parametric Down Conversion. *Phys. Rev. Lett.*, **57**, 20, pp. 2520-2523 (1986).
4) Yuen, H. P., and Shapiro, J. H.: Generation and detection of two-photon coherent states in degenerate four-wave mixing. *Opt. Lett.*, **4**, 10, pp. 334-336 (1979).
5) Levenson, M. D., Shelby, R. M., Aspect, A., Reid, M., and Walls, D. F.: Generation and detection of squeezed states of light by nondegenerate four-wave mixing in an optical fiber. *Phys. Rev.*, **A32**, 3, pp. 1550-1562 (1985).
6) Kumer, P., and Shapiro, J.H.: Squeezed-state generation via forward degenerate four-wave mixing. *Phys. Rev.*, **A30**, 3, pp. 1568-1571 (1984).
7) Shirasaki, M., and Haus, H.A.: Broadband squeezing with cotraveling waves. Nonlinear Guided-Wave Phenomena: Physics and Applications. paper SA2, 1989, and 1989 Opt. Soc. Am. Tech. Digest Series, **Vol. 2** pp. 232-234.
8) Shirasaki, M., and Haus, H. A.: Squeezing of pulses in a nonlinear interferometer. *J. Opt. Soc. Am. B*, **7**, 1, pp. 30-34 (1990).
9) Shirasaki, M.: Squeezing performance of a nonlinear symmetric Mach-Zehnder interferometer using forward degenerate four-wave mixing. *J. Opt. Soc. Am. B*, **8**, 3, pp. 672-680 (1991).
10) Shirasaki, M.: Quantum-noise reduction in a phase-sensitive interferometer using nonclassical light produced through Kerr media. *Opt. Lett.*, **16**, 3, pp. 171-173 (1991).
11) Bergman, K., and Haus, H. A.: Squeezing in fibers with optical pulses. *Opt. Lett.*, **16**, 9, pp. 663-665 (1991).
12) Rosenbluh, M., and Shelby, R.M.: Phys. *Rev. Lett.*, **66**, pp. 153-156 (1991).

**Masataka Shirasaki**

Optoelectronic Systems Laboratory
FUJITSU LABORATORIES, KAWASAKI
Bachelor of Applied Physics
Waseda University 1977
Dr. of Applied Physics
Specializing in Squeezing and Nonlinear Optics, and Optical Components for Fibers

**Hermann A. Haus**

Dept. of Electrical Engineering and Computer Science
MASSACHUSETTS INSTITUTE OF TECHNOLOGY
Bachelor of Science
Union College 1949
Dr. of Science
M. I. T. 1954
Specializing in Electrical Engineering and Teaching

**Keren Bergman**

Dept. of Electrical Engineering and Computer Science
MASSACHUSETTS INSTITUTE OF TECHNOLOGY
Bachelor of Science in Electrical Eng.
Bucknell University 1988
Master of Science in Electrical Eng.
M. I. T. 1991
Specializing in Fiber Optics

**Christopher R. Doerr**

Dept. of Electrical Engineering and Computer Science
MASSACHUSETTS INSTITUTE OF TECHNOLOGY
Bachelor of Science in Aeronautical Eng.
M. I. T. 1989
Bachelor and Master of Science in Electrical Eng.
M. I. T. 1990
Specializing in Optical and Electrical Engineering

# International Network

## Offices

### Abu Dhabi Office
P.O. Box 47047 Suite 802.
A1 Masadood Tower,
Sheikh Hamdan Street,
Abu Dhabi. U.A.E.
Telephone : (971-2)-333440
FAX : (971-2)-333436

### Algiers Office
9 Rue Louis Rougie Chateau Neuf.
El Biar, Alger 16030. Algeria
Telephone : (213)-2-78-5542
Telex : 408-67522

### Amman Project Office
P.O. Box 5420, Ammân. Jordan
Telephone : (962)-6-662417
FAX : (962)-6-673275

### Bangkok Office
3rd Floor, Dusit
Thani Bldg., 1-3, Rama IV.
Bangkok10500, Thailand
Telephone : (66-2)-236-7930
FAX : (66-2)-238-3666

### Beijing Office
Room 2101, Fortune Building,
5 Dong San Huan Bei-lu,
Chao Yang District, Beijing,
People's Republic of China
Telephone : (86-1)-501-3261
FAX : (86-1)-501-3260

### Brussels Office
Avenue Louise 176, Bte 2
1050 Brussels, Belgium
Telephone : (32-2)-648-7622
FAX : (32-2)-648-6876

### Harare Office
CABS House, Corner Central Avenue,
4th Street, Harare, Republic of Zimbabwe
Telephone : (263-4)732-627
FAX : (263-4)732-628

### Hawaii Office
6660 Hawaii Kai Drive, Honolulu,
Hawaii 96825, U.S.A.
Telephone : (1-808)-395-2314
FAX : (1-808)-396-7111

### Jakarta Representative Office
16th Floor, Skyline Bldg.,
Jalan M.H. Thamrin No.9,
Jakarta, Indonesia
Telephone : (62-21)-333245
FAX : (62-21)-327904

### Kuala Lumpur Office
Letter Box No.47, 22nd Floor,
UBN Tower No.10, Jalan P.
Ramlee, 50250, Kuala Lumpur, Malaysia
Telephone:(60-3)-238-4870
FAX : (60-3)-238-4869

### Munich Office
c/o Siemens Nixdorf Informationsysteme
AG, D8 SC,
Otto-Hahn Ring 6, D-8000,
München 83, F.R. Germany
Telephone:(49-89)-636-3244
FAX : (49-89)-636-45345

### Nairobi Office
Flat No.10, Fairview Flats, Bishop Road,
P.O. Box 21290, Nairobi, Kenya
Telephone : (254-2)727934
FAX : (254-2)723201

### New Delhi Liaison Office
Mercantile Home
1st Floor, 15 Katsurba, Gandhi Marg
New Delhi-110001, India
Telephone : (91-11)-331-1311
FAX : (91-11)-332-1321

### New York Office
680 Fifth Avenue, New York,
N.Y. 10019, U.S.A.
Telephone : (1-212)-265-5360
FAX : (1-212)-541-9071

### Shanghai Office
Room 1504, Ruijin Bldg.,
205 Maoming Road South,
Shanghai, People's Republic of China
Telephone : (86-21)-433-6462
FAX : (86-21)-433-6480

### Sucursal de Colombia
Cra. 9a. A No. 99-02 ofc. 811
Edificio Seguros Del Comercio
Santa Fe De Bogotá, Colombia
Telephone : (57-1)-618-3147
FAX : (57-1)-618-3134

### Taipei Office
Sunglow Bldg., 66, Sung Chiang
Road, Taipei, Taiwan
Telephone : (886-2)-551-0233
FAX : (886-2)-536-7454

### Washington, D.C. Office
1776 Eye Street, N.W.,
Suite 880, Washington, D.C.,
20006, U.S.A.
Telephone : (1-202)-331-8750
FAX : (1-202)-331-8797

## Subsidiaries

### ASIA AND OCEANIA
Fujitsu Australia Ltd.
Fujitsu Australia Software Technology. Pty. Ltd.
Fujitsu Component(Malaysia)Sdn. Bhd.
Fujitsu Electronics(Singapore)Pte. Ltd.
Fujitsu Hong Kong Ltd.
Fujitsu Korea Ltd.
Fujitsu Microelectronics Asia Pte. Ltd.
Fujitsu Microelectronics(Malaysia)Sdn. Bhd.
Fujitsu Microelectronics Pacific Asia Ltd.
Fujitsu New Zealand Ltd.
Fujitsu(Singapore)Pte. Ltd.
Fujitsu(Thailand)Co., Ltd.
Fujitsu Trading Ltd.

### NORTH AMERICA
Fujitsu America, Inc.
Fujitsu Business Communication Systems, Inc.
Fujitsu Canada, Inc.
Fujitsu Computer Packaging Technologies, Inc.
Fujitsu Computer Products of America, Inc.
Fujitsu Imaging Systems of America, Inc.
Fujitsu Microelectronics, Inc.
Fujitsu Network Switching of America, Inc.
Fujitsu Network Transmission Systems, Inc.
Fujitsu Systems Business of America, Inc.
Fujitsu Systems Business of Canada, Inc.
Open Systems Solutions, Inc.
Fujitsu Personal Systems, Inc.

### EUROPE
Fujitsu Deutschland GmbH
Fujitsu España, S.A.
Fujitsu Europe Ltd.
Fujitsu Europe Telecom R&D Centre Ltd.
Fujitsu Finance(U.K.)PLC
Fujitsu France S.A.
Fujitsu International Finance(Netherlands)B.V.
Fujitsu Italia S.p.A.
Fujitsu Microelectronics Ireland Ltd.
Fujitsu Microelectronics Italia S.r.l.
Fujitsu Microelectronics Ltd.
Fujitsu Mikroelektronik GmbH
Fujitsu Nordic AB
Fujitsu Systems Business of Europe Ltd.
Fulcrum Communications Ltd.

### LATIN AMERICA
Fujitsu do Brasil Ltda.
Fujitsu Vitória Computadores e Serviços Ltda.

**FUJITSU LIMITED**

6-1, Marunouchi 1-chome, Chiyoda-ku, Tokyo 100, Japan
Phone: National (03) 3216-3211 International (Int'l Prefix) 81-3-3216-3211 Telex: J22833 Cable: "FUJITSULIMITED TOKYO"