Peter Bocker

ISDN
The Integrated Services Digital Network

Concepts, Methods, Systems

Springer-Verlag
Berlin Heidelberg GmbH

Peter Bocker

ISDN
The Integrated Services Digital Network

Concepts, Methods, Systems

In collaboration with
G. Arndt, V. Frantzen, O. Fundneider, L. Hagenhaus,
H. J. Rothamel, L. Schweizer

With 115 Figures and 32 Tables

Springer-Verlag Berlin Heidelberg GmbH

Dr. rer. nat. Peter Bocker
Gerhard Arndt
Viktor Frantzen
Oswald Fundneider
Lutz Hagenhaus
Dr.-Ing. Hans Jörg Rothamel
Lutz Schweitzer

Siemens AG
Telecommunication Networks and Security Systems Group,
Central Laboratories
Hofmannstrasse 51, D-8000 München 70,
Federal Republic of Germany

This is a completely revised and international edition on the basis of
ISDN – Das diensteintegrierende digitale Nachrichtennetz,
2nd edition, Springer-Verlag 1987

Library of Congress Cataloging-in-Publication Data.
Bocker, P., 1933–
ISDN, the integrated services digital network.
International edition of: ISDN, das diensteintegrierende digitale Nachrichtennetz. Bibliography: p.
Includes index. 1. Integrated services digital networks. I. Title.
TK5103.7.B6313 1988 384 88-6545

ISBN 978-3-662-08038-2 ISBN 978-3-662-08036-8 (eBook)
DOI 10.1007/978-3-662-08036-8

Originally published by Springer-Verlag Berlin Heidelberg New York in 1988.
Softcover reprint of the hardcover 1st edition 1988

Typesetting: With a system of the Springer Produktions-Gesellschaft, Berlin.
Dataconversion: Brühlsche Universitätsdruckerei, Giessen.

2161/3020-54321

Preface

The present state of development of communication technology is characterized by two features, namely the digital representation of all signals transmitted and processed, irrespective of information type – voice, text, data or images – and the integration of systems and services, this integration only being completely possible using digital technology. The boundaries between switching and transmission are shifting, and functions are being redefined and redistributed between terminals and communication networks. Multiservice terminals – unlike telephones, teleprinters, video data terminals – are designed to handle more than one information type. Lastly, the communication network allows voice, text, data and video information to be transmitted on the same circuit; the user obtains access to this network via a non-dedicated "communication socket".

The essential features of this Integrated Services Digital Network (ISDN) have been standardized over the last eight years by experts from all over the world under the aegis of the CCITT [1]), the international standardizing body of the carriers of public communication networks. All the leading network carriers are working towards ISDN implementation because of the substantial benefits it will offer to users, network carriers and manufacturers alike:

- Users will obtain additional and advanced services, most of them designated to cater for the growth in non-voice traffic. The ISDN subscriber access will also enable users to operate existing systems more cost-effectively than via various dedicated networks.
- Network carriers will benefit from the universal network concept which allows them to introduce new services and supplementary services without large specific capital investment. Economic advantages will also flow from service integration, especially at subscriber line level, and from standardization of operation and maintenance for the range of services provided.
- Manufacturers will welcome above all the standard network access, enabling one terminal to access several communication services, i.e. opening up a wider market.

The driving forces behind the ISDN are thus cost-effective communication in various information types via the same subscriber line, and sophisticated new services and supplementary services, especially in the increasingly important non-voice area.

This book describes the ISDN with reference to its services, its technology and the principles on which ISDN terminals operate. The main focus of attention concerns those features of the ISDN which can be implemented using the cables of the existing telephone network; this type of ISDN is occasionally termed "narrowband ISDN". However, full-motion video communication in color and with a quality comparable to

1 Comité Consultatif International Télégraphique et Téléphonique

that of television pictures will only be possible when subscribers are connected via optical fibers to the "broadband ISDN". There is no basic difference between narrowband and broadband ISDN: the configuration of the station, the principles of subscriber access and of signaling are the same for both.

After analyzing the aims and objectives of telecommunication, the services and supplementary services possible in the ISDN are discussed. This is followed by a description of the structure of the network and its basic dimensions. Much space is devoted to the subscriber access and associated user-network interfaces and signaling. The next section deals with the basic features of ISDN terminals. Switching in the ISDN is examined in detail and the interworking of network nodes and user terminals is illustrated with the aid of examples, with particular reference to the special features of ISDN private branch exchanges. The transmission methods and transmission paths used in the ISDN are then described, and the final section of the book contains a critical evaluation of the ISDN for users in the office, at home and en route. Finally, the current standards and recommendations relating to the ISDN are listed in an Annex.

The book is intended for engineers who need to be familiar with the requirements and features of the ISDN in order to design, construct or operate communication systems. It is also aimed at those engaged in teaching and research, as well as a broader public with an interest in technical matters. It thus provides an overview of the background of this new development in communication technology and places its various features in context. In view of the expected international readership, the technical terms used in this book have been chosen such as to comply with international standards, in particular, the International Electrotechnical Vocabulary IEV and relevant CCITT recommendations such as G.701, I.112 and Q.9. When a term is used first, some national terms are given in brackets where appropriate.

This general description of the ISDN concept and its implementation was prepared in collaboration with Dipl.-Ing. Gerhard Arndt, Dipl.-Ing. Viktor Frantzen, Dipl.-Ing. Oswald Fundneider, Dipl.-Ing. (FH) Lutz Hagenhaus, Dr.-Ing. Hans Jörg Rothamel and Dipl.-Ing. Lutz Schweizer. Chapter 1 was written by P. Bocker; Sections 2.1 to 2.3.3 by H. J. Rothamel; Sections 2.3.4 to 2.5, Chapter 5 and Chapter 8 by G. Arndt; Chapter 3 by L. Hagenhaus; Sections 4.1 to 4.3 by O. Fundneider; Sections 4.4, 4.5 and Chapter 6 by V. Frantzen; Chapter 7 and the Annex by L. Schweizer. The book is also the product of much advice and information supplied by colleagues in the various development departments of Siemens AG.

Thanks are due to my professional collegues for their support in producing this book. However, I would like to express my special appreciation to L. Schweizer for his effective and most valuable help in preparing the manuscript of the English edition and of course to my co-authors. Without their understanding and cooperation it would not have been possible to present this work as a unified whole.

Munich, April 1988 P. Bocker

Contents

1 The Function of Telecommunications

1.1 Types of Communication

Communication means the exchange of information. The function of telecommunications is to facilitate communication for people.

Communication is based on the exchange of information between two parties in an *interactive dialog*. Since the invention of telegraphy, electric current on wires and later high-frequency electromagnetic waves have been used to enable individuals to communicate over distance in a simple, convenient manner without physical transport of information media. People can thus send each other texts (telegraphy), talk to each other (telephony) and exchange drawings and illustrations (facsimile transmission, videotelephony) by electrical means. With the development of computers, the need also arose for information transfer between man and machine and from machine to machine (data transmission).

The advent of radio broadcasting also opened the way to *distributive communication*, i.e. communication between one transmitter and a number of receivers which can be reached without recourse to physical transport. Both freely propagating and guided electromagnetic waves are used for transmitting voice, sound and picture (sound broadcasting and television). Anyone who can be reached by these waves has immediate access to distributive systems of this kind.

Recently another type of communication has evolved, namely *retrieval communication*. In retrieval communication, which is being considered as a form of interactive communication (cf. Sect. 2.1), the user ascertains what is available from an information center and requests the selected information at a time chosen by him. He is not able to alter the contents of this information. Examples of this are the information services provided by videotex, broadcast teletext and full-channel broadcast videography [1.1]. The dialog is restricted to initial selection of the information to be retrieved.

1.2 Information Types and Signals

The information handled in communication systems is available in the form of the spoken or written word, as sound, picture or as information to be processed, i.e. "data". Equally varied are the electrical signals physically representing these information types, and their demands on the communication system.

One of the characteristics of these signals is their form of representation. Their parameters may occur continuously or only in the form of discrete values, and they

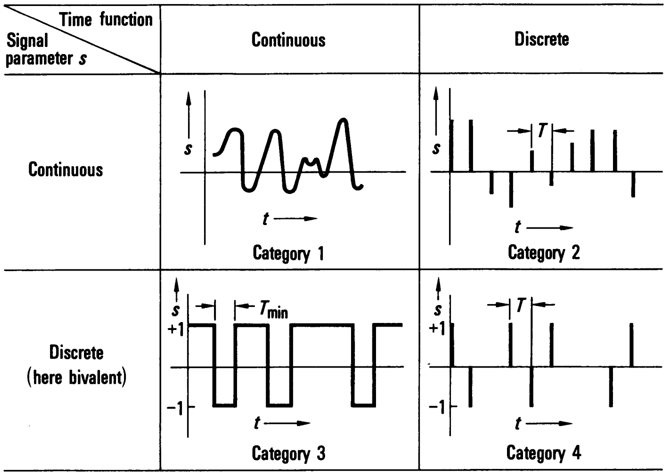


Fig. 1.1. Classification of Primary Signals.
t Time

may have a relevant value at all times ("time-continuous") or only at certain discrete moments (Fig. 1.1). Table 1.1 shows the relationship between these signal parameters and the information types. Speech and sound basically have continuous signal parameters and a continuous time function (Category 1). Although frame sequences in moving picture communication (full-motion video) and still (fixed) image sequences also have continuous signal parameters, the pictures follow each other discretely in time (Category 2). Text and data signals always have a discrete signal parameter; with anisochronous letter or character sequences (i.e. sequences transmitted arrhythmically) the time function is continuous (Category 3); with the blocks of isochronous text and data mostly used nowadays, it is discrete (Category 4).

Technological progress has made it increasingly desirable for communication systems to handle signals that are discrete both in time and value; for instance, discretely timed signal sequences can be interleaved for transmission on a time-division

Table 1.1. Parameters of Primary Signals and Information Types

Signal parameter	Time function	
	Continuous	Discrete
Continuous	Speech Sound	Full-motion video Fixed-image sequences
Discrete	Texts (anisochronous letter sequences) Data (anisochronous character sequences)	Blocks of text Blocks of data

multiplex basis (cf. Sect. 7.2). The primary signals mentioned are therefore converted more and more frequently into *binary, isochronous* signals.

Aside from the mode of representation, the other important signal characteristic is the quantity of information to be transported in a unit of time. This determines either the upper limit frequency at which signal components must be transmitted, or – for binary isochronous signals – the required bit rate. The upper limit frequencies agreed by the CCITT[1] and CCIR[2], the international standardizing bodies of the telecommunications carriers, are 3 400 Hz for telephony, 15 kHz for sound broadcasting and 5 MHz for television; the two channels thus defined for telephony and television correspond to bit rates of 64 kbit/s and 140 Mbit/s respectively. The bit rate for text, data and still-image signals can be anywhere between 50 bit/s, 64 kbit/s and above.

1.3 Systems for Interactive Communication

A wide range of communication systems exists for the above mentioned information types: voice, text, data, image. Here we are primarily concerned with systems for interactive and communication (including retrieval). Some examples are listed in Table 1.2. In each of these systems, communication can take place in different forms, e.g. between two or between several partners, without or with storage of information.

Table 1.2. Systems for Interactive Communication (Examples)

Information type	Communication system
Speech	Telephony
Text	Telex Teletex Videotex (message service)
Data	Data transmission
Picture	Facsimile Telescript (telewriting) Videotex (information service)

Each of the hithero existing systems for interactive communication is designed to handle a single information type. This is in marked contrast to distributive communication, such as today's television service in which picture and sound can be transmitted together.

However, the ability to handle several information types in one system is also required for interactive communication: voice + moving picture, but also voice + fixed image (facsimile), voice + videotex, voice + data, text + data etc. The conventional communication networks and services can only satisfy this requirement to a very limited extent.

1 Comité Consultatif International Télégraphique et Téléphonique;
2 Comité Consultatif International des Radiocommunications

1.4 Conventional Communication Networks and Services

With over 650 million stations worldwide, the *telephone network* is the most extensive network for interactive communication. However, with suitable equipment every telephone subscriber cannot only hold conversations but also transmit facsimile, data, or teleaction information, as well as access videotex databases (Table 1.3). However, the telephone network is only "shared" by these other services; in other words it is still designed for voice communication as far as signaling, call set-up time and other characteristics are concerned.

Another public switched network, the *telex network*, was introduced about 50 years ago for text communication. This has approximately 1.7 million subscribers worldwide and permits text traffic between all connected telex machines using a standardized procedure. To achieve this, it was necessary to agree not only on the basic characteristics of the transmission and switching equipment – as in the telephone network – but also on terminal equipment characteristics such as source code, character structure and transmission rate [1.2].

In addition to the switched telephone network and the telex network, many countries have other independent *networks for text and data communication.* These employ various switching methods (circuit switching, packet switching; see Sect. 3.2.2) and have a number of subscriber classes (Table 1.4), for which different data transmission rates and different codes and speeds in the call set-up and clear-down phase have been specified [1.3 to 1.5]. However, there are no detailed specifications covering codes, user protocols, etc. in the data transmission phase. In data communication, it is normally the users themselves who specify the functions and the method of interoperation of data sources and sinks.

More recently, the teletex service was set up to provide a second, more advanced general text communication service in which every subscriber can exchange texts with every other subscriber. With teletex, the transmission rate is considerably higher than for telex and the character set is the same as that of office typewriters. In accordance with international agreements [1.6, 1.7], this service is handled either in a subscriber class of a special text and data network, or in the switched telephone network (as in the case of e.g. the facsimile service).

As well as these switched networks, *dedicated circuits* can be provided for voice, text and data transmission. By taking appropriate special measures, the transmission characteristics of these circuits can be readily adapted to suit their particular function [1.8].

Table 1.3. Services in the Switched Telephone Network (Examples)

Telephony
Facsimile group 2 (3 minutes per A4 page)
group 3 (1 minute per A4 page)
Data transmission with modems
– Modems for serial transmission
300, 1200, 2400, 4800 bit/s
– Modems for parallel transmission
10–40 characters/s
Videotex access

Table 1.4. The Classes of Service in Public Data Networks According to CCITT Recommendation X.1 [1.3]

Class	Data transmission phase	Call set-up and clear-down phase
Start/stop data terminal equipment		
1	300 bit/s, 11 signal elements per start/stop character	300 bit/s, 11 signal elements per start/stop character CCITT alphabet no. 5
2	50 to 200 bit/s, 7.5 to 11 signal elements per start/stop character	200 bit/s, 11 signal elements per start/stop character, CCITT alphabet no. 5
Synchronous data terminal equipment		
3	600 bit/s	600 bit/s, CCITT alphabet no. 5
4	2.4 kbit/s	2.4 kbit/s, CCITT alphabet no. 5
5	4.8 kbit/s	4.8 kbit/s, CCITT alphabet no. 5
6	9.6 kbit/s	9.6 kbit/s, CCITT alphabet no. 5
7	48 kbit/s	48 kbit/s, CCITT alphabet no. 5
Packet mode terminal equipment		
8	2.4 kbit/s	As per CCITT Rec. X.25
9	4.8 kbit/s	As per CCITT Rec. X.25
10	9.6 kbit/s	As per CCITT Rec. X.25
11	48 kbit/s	As per CCITT Rec. X.25
12	1.2 kbit/s	As per CCITT Rec. X.25
Start/stop data terminal equipment connected to packet switching networks		
20	50 to 300 bit/s, 10 to 11 signal elements per start/stop character	As per CCITT Rec. X.28
21	75 to 1200 bit/s, 10 signal elements per start/stop character	As per CCITT Rec. X.28
22	1200 bit/s, 10 signal elements per start/stop character	As per CCITT Rec. X.28

1.5 Integration of Services

Examination of the existing state of telecommunications has revealed that compatible communication networks and services on which the numerous different communication systems are operated are capable of providing communication between individual partners on a worldwide basis. However, interworking between the separate systems for voice, text, data and image communication is only possible in exceptional circumstances; both the terminals and the procedures for network access and communications handling are different and specifically geared to their particular task.

The further development of communication networks is therefore not merely a question of continually improving, extending and modernizing these systems, but also one of giving telecommunications a new direction; of moving away from the existing separate systems to a universal approach which will overcome the existing barriers between the different information types and technologies, the better to meet the primary function of telecommunications: to facilitate communication for people in all its aspects. In doing so, however, the possibility of interworking with existing networks must always be an attendant consideration.

Since for economic reasons telecommunications will be generally based on digital signals, a uniform digital communication network for all information types is the obvious solution.

Thus, this uniform network must therefore achieve two objectives. The first is the *integration of technologies*. On the one hand, the technologies used in the various services must converge so that e.g. voice and text signals can be transmitted by the same means on a subscriber line. On the other, the switching and transmission technologies within the network have to be integrated. For example, in a digitized network it is not necessary to fan out the time-division multiplexed signals before they reach the exchanges; these signals are fed directly to the switching equipment which then performs channel assignment in both time and space (see Sect. 6.2.3 and [1.9]).

The second objective for a uniform digital communication network is to introduce more advanced *new services* and to provide *service integration*. This means that the voice, text, data and image communication services are handled using a standard procedure for call set-up and clear-down and via a standard interface between terminal equipment and network. It also means that several different information types can be simultaneously supported at a user station, and that interworking facilities can be established between individual information types, e.g. between text and data (cf. Sect. 1.8).

These two objectives are achieved in a consistent way by the concept of the Integrated Services Digital Network [1.10].

1.6 The Integrated Services Digital Network ISDN

A concept as fundamental as that of a universal Integrated Services Digital Network requires international agreements from an early stage. On the basis of discussions held worldwide by the CCITT, agreements have been reached (see Annex) whereby the Integrated Services Digital Network ISDN is characterized by the following basic features (Fig. 1.2):

- It is based on the digitized telephone network, in other words a network based on the digital 64 kbit/s telephone channel. The ISDN is therefore essentially a circuit switching network; however, packet-switched data traffic can also be handled in the ISDN (see Sects. 4.4.4 and 4.5).
- In the ISDN, connections from subscriber to subscriber are entirely digital.
- The basic access for a user (see Sect. 4.2.1.2) provides two 64 kbit/s basic channels (*B channels*) and one 16 kbit/s signaling channel (*D channel*) in each direction; the connections established over the two 64 kbit/s channels can be to different destinations. A primary rate access (see Sect. 4.2.3), mainly for connecting larger ISDN private branch exchanges, is also defined; this may comprise, depending on the multiplexing system used (see Sect. 7.2.3), up to 24 channels or up to thirty 64 kbit/s basic channels and one 64 kbit/s signaling channel. Basic and primary rate access can be provided on the copper wire pairs of existing subscriber lines (see

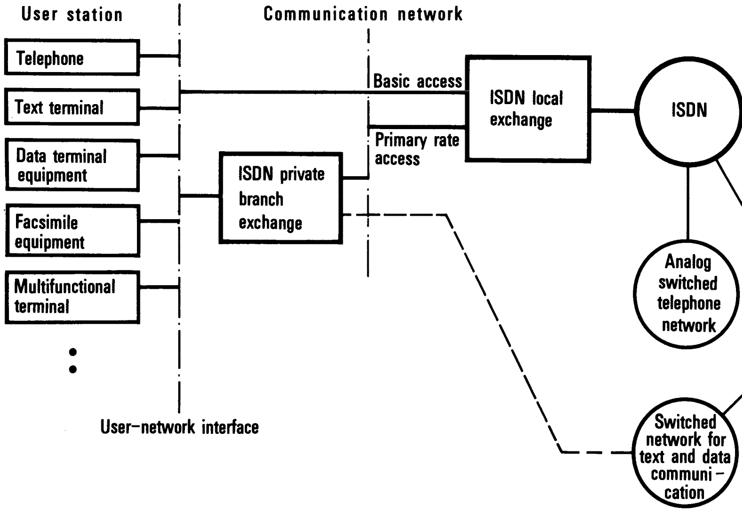


Fig. 1.2. The Integrated Services Digital Network ISDN

Sect. 7.4.3). If optical fibers are used, it is also possible to specify ISDN access lines containing broadband channels, e.g. for video transmission.

- To each subscriber a single directory number is allocated, irrespective of the number and type of voice, text, data and image communication services he requires.
- The universal user-network interface defined for the ISDN allows different terminals to be connected by a standard "communication socket" (Sect. 4.2). Accordingly, standard user procedures for call set-up and clear-down (Sect. 4.3) have been defined.
- The various terminals of a user station can be connected in a bus or star configuration (Sect. 4.1.3). The network not only establishes connections between the user stations (cf. Fig. 1.2) but additionally between those terminals within the user stations which are appropriate for the particular service required and are compatible (see Sect. 4.3.3).

Subscribers in existing networks, e.g. in the analog switched telephone network, are reached via the ISDN with the aid of interworking units (see Sect. 3.5). If no specific interworking unit is available, the subscriber terminal must be provided with a separate connection to the network in question (the dashed line in Fig. 1.2 shows how an IDSN private branch exchange is directly connected to the switched network for text and data communication).

The ISDN thus provides effective integration of technologies and of services for interactive and communication.

As far as the *user* is concerned, the wide range of useful services and features available through the ISDN offers the following advantages:

- New and more versatile applications of communication due to the possibility of exchanging information of different types such as voice and text or text and data either simultaneously or alternately with one or more stations.

- Information exchange with the network through the high-capacity signaling channel even during an existing call without disturbing transmission of user information, e.g. for supplementary services such as *call waiting* or *advice of charge* (cf. Sect. 2.3.3).
- Improved accessibility of the user through the two basic channels and the facilities offered by the signaling channel and by change of service.
- Connection of terminals which can provide simple, uniform access not just to one but to many communication services supporting a variety of information types; even during an established connection, it is possible to effect a change of service including activation or deactivation of appropriate supplementary services (see Sect. 2.3.3).
- More widespread use of communication systems with high bit rates, mainly important for non-voice communication, e.g. facsimile (see Sect. 5.3) and data transmission.

The main advantages for the ISDN *carrier* are as follows:

- ISDN services and capabilities stimulate new applications and increased use of communication networks.
- A single, multi-purpose communication network with uniform technology for all services leads to uniform operation and maintenance.
- The flexible digital principle on which the network is based enables new communication services to be introduced at comparatively low cost, even if only for trial purposes.

1.7 Integration of Services with Higher Bit Rates

In addition to voice, text, data and fixed-image communication, the ISDN concept also allows moving picture (video) communication, because essential features of the ISDN such as the user signaling (see Sect. 4.3) or the control units within an exchange (see Sect. 6.2.5) are also suitable for switching higher rate channels [1.11]. Optical fibers (see Sect. 7.3.1) are then required instead of copper wire pairs for transmission on the subscriber line, and the switching networks in the exchange equipment (see Sect. 6.2.3) must provide sufficient capacity [1.12].

A digital network of this type in which all the services for communication between individual partners are integrated is also able to handle distributive communication services.

1.8 The Communication Network of the Future

The basic function of communication networks – the telephone, text and data networks as well as the ISDN – is to *transport information* between individual users. In the case of the telephony, telex, teletex, and telefax services, for instance, this information consists of the spoken word, written text or pictures; conversion of this primary information to the electrical signals to be transported takes place in the service-specific terminals in accordance with the appropriate standards which are usually universal. On the other hand, with data transmission the information to be transported exists in the form of a quantity of decision units, of "bits", the meaning of which is agreed individually between the operators of the data sources and sinks; in this case, the service only encompasses the *transportation of bits*, which the network receives in the form of electrical signals in accordance with certain specifications and which it forwards to the remote destination.

The increasing universality of terminals and networks and the proliferation of terminals and systems for text and data communication will make it increasingly desirable to overcome such qualitative differences between individual services. Comprehensive, application-neutral protocols and services must be defined and service interworking facilities must be created in order to dismantle the barriers to compatibility and to satisfy communication requirements in their broadest sense.

The following tasks can be mentioned in this context:

1. Ensuring compatibility between terminals and different information centers.
   For example, in the *videotex* service it is already possible to retrieve information from different manufacturers' computer centers with one and the same terminal. Compatibility can be achieved not only by agreement but also by protocol matching in the network, as is done e.g. in the PADs (Packet Assembly/Disassembly Devices) for connecting different terminals to the packet switching network [1.13].
2. Providing interworking between services.
   To enable e.g. the teletex subscriber to reach telex subscribers throughout the world with his equipment, the telex-teletex converter has been specified [1.14]. Further conversions between services within a network, such as the ISDN, and also between services in different networks are required (see Sect. 2.4).
3. Message storing and processing, and message switching by the communication network.
   The purpose of this is to organize communication on a personal basis (and not merely with regard to stations) and thus independently of time and place; this means that sending and receiving a message virtually no longer needs to take place "at the same time", and the personal "electronic mailbox" can be accessed from every communication station (telephone, telex, teletex, ISDN station). In this way mobile users too will be better accommodated. The CCITT recommendations for message handling systems [1.15] already provide an excellent internationally agreed basis for messaging; systems of this type also widen the scope for traffic between different types of terminal by means of protocol matching. The message service provided by videotex as well as services incorporating text and voice storage are examples of functions of this type which have already been introduced.
4. Providing flexible, user-controlled functions in the network.
   The technical components of the ISDN enable special functions to be provided through access to suitable databases in the network on the basis of very efficient signaling. These features allow the subscriber to establish private networks, either switched or with permanent connections, within the public network [1.16].

The communication network of the future is thus a universal digital integrated services network that will offer the user all possible forms of communication in the different information types, i.e. voice, text, data, fixed image and video. This network will be able to support services for interactive and distributive communication. The technical basis is the ISDN. Generally accessible "open" communication systems will be possible in this network through protocol agreements and equipment for information processing and storage; within the network, the user will also be able to create private networks. The communication network of the future based on the ISDN thus meets all the requirements for facilitating information exchange for the user – either with other users or with information centers – even more effectively than at present.

2 Telecommunication Services

The communication facilities in a network are made available to the user in the form of defined telecommunication services. In the ISDN, the 64-kbit/s B-channel (basic channel), the H-channels with bit rates of 384 kbit/s or more and the 16 kbit/s or 64 kbit/s D-channel (signaling channel) (see Sect. 4.2.1.1) enable powerful new services and supplementary services to be provided. For reasons of continuity the ISDN also includes the existing services of the present-day telephone network. Furthermore, existing terminals from the telephone network and public dedicated data networks for circuit-switched or packet-switched data transmission (CSPDN, PSPDN) can also be operated on an ISDN line. This means that the user can enjoy the advantages of integrated services without having to replace all his existing terminal equipment.

To clarify the terms used to describe the individual services, the method used for classifying and describing telecommunication services will first be explained in Sections 2.1 and 2.2.

2.1 Definition of Services

The term *services* is used here to describe all the telecommunication means and facilities provided to users by the telecommunication carriers (administrations and recognized private operation agencies) for communication over public and private networks. Examples are the telephony, teletex, telefax, videotex and data transmission services. The services are characterized by their technical, operational and administrative service attributes. These describe all the communication functions and protocols required for handling the communication facilities associated with the particular service. In this chapter only the technical and operational service attributes for the user are described in detail.

The communication protocols comprise all the rules and specifications for the communication processes between user and network or from user to user over the network.

These functions and protocols can be classified according to the hierarchical structure of the seven layers of the OSI reference model (seven-layer reference model for Open Systems Interconnection [2.1]) (Fig. 2.1; also see Sect. 4.3.1).

The protocols of layers 1 to 4 are transportation oriented, i.e. they control both the access to the network (layers 1 to 3) and the transport of information through the network from user to user (end-to-end). To layer 1 are assigned the physical connection conditions such as the user-network interface (ISDN interface S, see Sect. 4.1, or existing interfaces X.21 (CCITT), RS–232–C, RS–449, etc.), transmission bit rate and electrical characteristics.

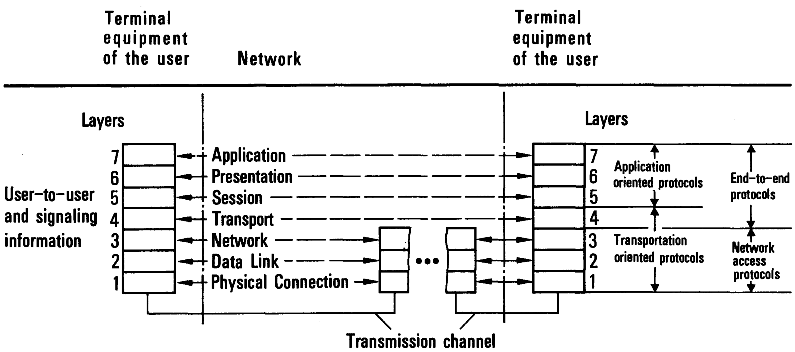


Fig. 2.1. Structure of Protocols for Communication Between Two Terminals via a Switched Network

Layer 2 contains the procedures for reliable data transfer (with error detection and recovery) both for signaling between user and network and from user to user, and for transmission of user information from user to user. In the ISDN, both user-network and user-user signaling take place in the D-channel (see Sect. 4.3); the user information can be transmitted in circuit-switched or packet-switched mode via the B-channel or in packet mode (data and teleaction) via the D-channel (see Sect. 4.2).

The protocols of the call-control or network layer (layer 3) are used for set-up, clear-down and supervision of the physical connection (routing). In the case of packet-switched communication, layer 3 protocols are additionally used for controlling and monitoring packet transport (flow control, supervision of the packet sequence, correction if necessary) for signaling and user information. Layer 4, the transport layer, supervises and controls the transport and the logical linking of message blocks from terminal to terminal (end-to-end).

The application oriented protocols are assigned to layers 5 to 7. Layer 5 contains the session control with set-up and clear-down and monitoring of the logical connection, e.g. between a processing program in a data terminal and a database in an information center (host) for the retrieval of information. Switching between different operation modes of a service (e.g. from text to facsimile in the mixed mode service, Sect. 2.3.1), and control of the correction of transmission errors which could not be rectified in the transport oriented layers also take place in this layer. (The last-mentioned function would involve retransmission e.g. of an information entity such as a page of text).

Layer 6 comprises the functions for presentation of information such as text and graphics character sets, format and structure of a document.

To layer 7 are assigned the functions and protocols for controlling the applications (e.g. text transmission, text and data retrieval) and for editing and processing message contents for communication. The layer 7 functions also include the necessary evaluation of message-related information for controlling the communication process, such as the type of information, quality requirements, name or address of the communication partner or of the application process of a computer, authorizations and data protection (encryption).

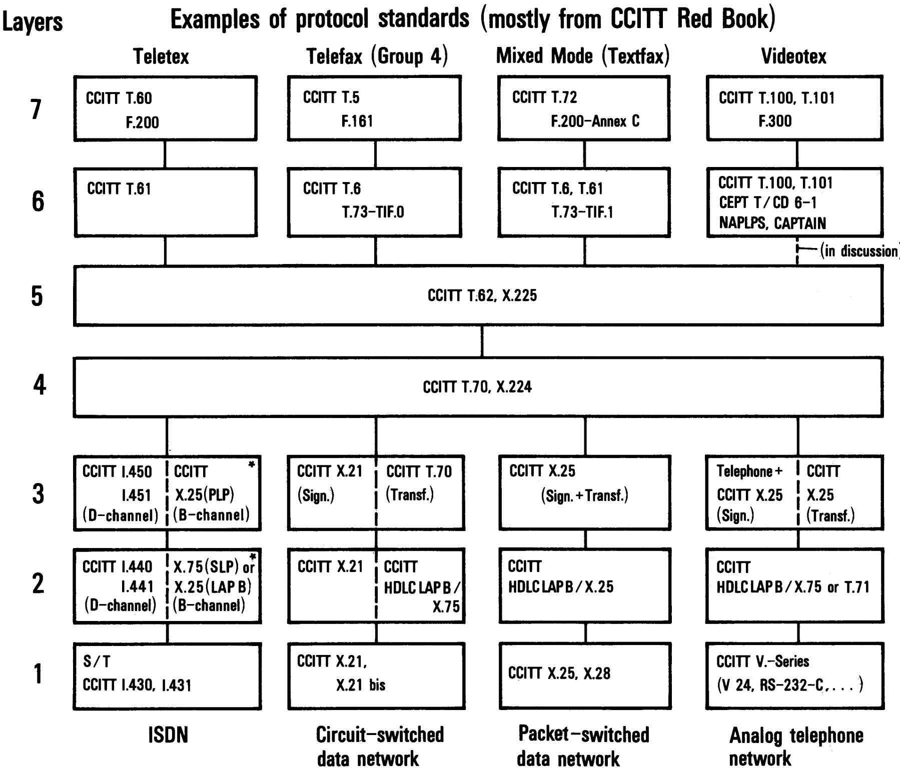
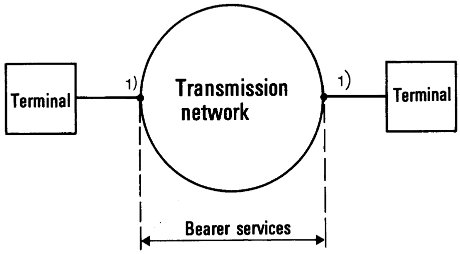


Fig. 2.2. Incorporation of ISDN Protocols into the Structure of Telematic and Data Transmission Protocols.
Sign Signaling, Transf. Transfer of user information: text, data, graphics etc., TIF Text Image Format, LAP Link Access Procedure, NAPLPS North American Presentation Level Protocol Syntax, CAPTAIN Character And Pattern Telephone Access Information Network, CEPT Conférence Européenne des Administrations des Postes et des Télécommunications
* Refer to CCITT Blue Book (protocols not yet specified in the Red Book).

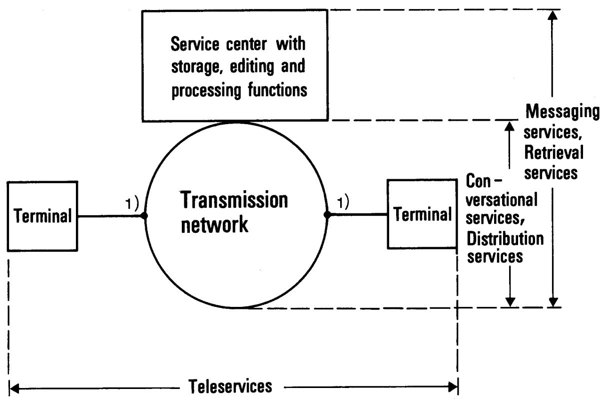
Figure 2.2 shows by way of example the ISDN protocols [2.2 to 2.7], some data transmission protocols and telematic protocols [2.8 to 2.30] for text and still image transmission classified in accordance with the seven-layer structure.

Depending on the extent of standardization of the communication functions and protocols, the services are subdivided into two groups by the CCITT: *bearer services* and *teleservices* [2.31 to 2.33].

Bearer services are used for unrestricted data and text transmission (clear channel) as implemented hitherto in circuit-switched and packet-switched data networks. The technical specifications of these services cover the transmission functions of OSI reference model layers 1 to 3 required for the transport of information. In circuit-switched bearer services, these are the functions assigned to layers 1 to 3 for signaling between user and network (ISDN: D-channel protocols) and the layer 1 functions (physical interface, transmission bit rate) for the transmission of user information (ISDN: layer 1 functions of the B-channel). In packet-switched bearer services the functions of layers 2 and 3 (transmission protocol, packet format) are also specified for transporting the user information. A bearer service only ensures the transport of



a



b

Fig. 2.3. **a** Sphere of Definition of the Bearer Services. **b** Spheres of Definition of the Teleservices
[1] User-network interface

information between the respective user-network interfaces. In other words the compatibility of the terminal equipment high layer protocols, in contrast to the teleservices, is the responsibility of the users of these terminals (Fig. 2.3a). The users may apply proprietary protocols or – in the future – they may use protocols standardized by organizations other than CCITT (in particular ISO).

The *teleservices* are services for user-to-user and user-to-host communication, including specification of the communication functions of the terminals (Fig. 2.3b); these services include telephony, teletex, telefax and videotex. The communication functions comprise on the one hand all the transmission functions and communication protocols of layers 1 to 3, as mentioned for the bearer services. On the other, they include the functions and protocols for controlling the communication processes, if necessary for different information types (e.g. for transmission of alphanumeric characters or of picture elements (pixels) of a facsimile image), for communication oriented editing and processing of information at the transmit end, and for presenting the transmitted information on reproduction (output) at the receive end (layers 4 to 7). By virtue of their specifications, teleservices ensure compatibility between the terminals dedicated to the particular service, i.e. with respect to coding (character sets) and structuring (format) of the user information to be transmitted.

In addition to this OSI-oriented structuring a further classification of the services has been defined in CCITT covering ISDN services up to 64 kbit/s and the future

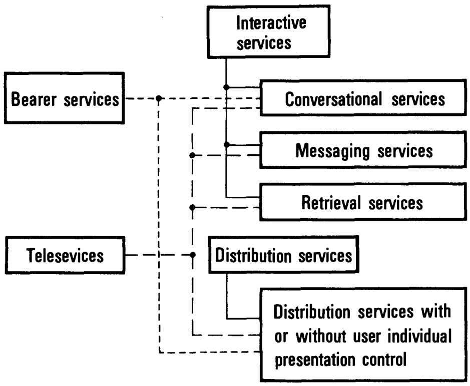


Fig. 2.4. Classification of Services

broadband services. Depending on the different applications, communication forms and network resources required by the services, two main service categories have been identified: *Interactive services* and *distribution services* (Fig. 2.4). The interactive services are subdivided into three service classes, viz. the *conversational services*, the *messaging services* and the *retrieval services*. The class of distribution services contains distribution services with and without user individual presentation control.

The interactive services and the distribution services may be standardized by the CCITT and offered by the public network carriers as teleservices and in some cases as bearer services.

Conversational services in general provide the means for bidirectional dialog communication with bidirectional, real-time (no store-and-forward) end-to-end information transfer from user to user or between user and host (e.g. for data processing). The flow of user information may be bidirectional symmetric, bidirectional asymmetric and in some specific cases (e.g. such as surveillance) the flow of information may be undirectional. The information is generated by the sending user or users, and is dedicated to one or more individual communication partners at the receiving site. Examples of conversational services are telephony, audio conference, teletex, telefax, and data transmission. For those services, ISDN has to provide bidirectional connections with an unrestricted (transparent) bearer capability or a bearer capability dedicated to the type of information to be transported. Conference services require special conference facilities.

Messaging services offer uset-to-user communication between individual users via storage units with store-and-forward, mailbox and/or message handling (e.g. information editing, processing and conversion) functions. Examples of messaging services are message handling services and mail services for audio information, text, data, graphics and high resolution images, such as voice mail, text mail, fax mail. These services require storage (messaging) units which may be provided centrally in the ISDN or in private networks.

The user of *retrieval services* is able to access information stored in information centers and generally provided for public use. This information will be sent to the user on demand only. The information can be retrieved on an individual basis, i.e. the time at which an information sequence is to start is under the control of the user. Examples are retrieval services for text, data, graphics, high resolution images and audio information, such as videotex. Retrieval services need connections with bearer capabilities dedicated to the types of information to be transported from the retrieval center to the user and to the user's request and selection of information in the backward direction. Retrieval centers may be installed in the ISDN (e.g. videotex center) or on the information provider's premises (e.g. as "external computer").

High layer functions of messaging and retrieval services, such as the input, storage and retrieval of information from retrieval centers, the mailbox functions and possible conversion functions for interface, protocol or bit rate adaptation, are also called "value added services" or "enhanced services" realized in special network nodes or external service centers as additions to the transport capabilities of a transmission network. These value added services are only communication functions and will be offered to the users as parts of telecommunication services, not as stand-alone services.

Distribution services with or without user-individual presentation control distribute a stream of information from a central source to an unlimited number of authorized receivers connected to the network. Distribution services *without* user-individual presentation control are the so-called broadcast services. They provide a continuous flow of information. The user can access this flow of information but cannot control the start and order of presentation of the broadcast information. As the user accesses the information at any point in time, it may not be presented from the beginning. Examples are broadcast services possibly for data and voice, and in the future broadband ISDN also for television and sound programs. To support those services, the network has to provide distribution nodes and/or unidirectional (broadcast) connections from the distribution node to the user, and, in most cases, connections from the user to the distribution node for program selection by the user.

Distribution services *with* user-individual presentation control provide an information stream structured as a sequence of information entities (e.g. frames) with cyclic repetition. The user can individually select the distributed information and can control its start and order of presentation. Due to the cyclic repetition, the information entity or entities selected by the user will always be presented from the beginning. One example of such services is broadcast videography or teletext. In general, the broadcast information is selected at the user's terminal, i.e. the selection is not the task of the network.

2.2 Definition of Service Attributes and Supplementary Services

As indicated in Fig. 2.5 and [2.34], the service attributes describing a communication service can be divided into:

- user related, technical service attributes
- carrier related, operational and commercial service attributes.

The service attributes for the user can be further subdivided into *basic service attributes* and *supplementary services*. The basic service attributes describe a service with its basic and fundamental communication functions (basic, minimum configuration) such as the *subscriber line equipment*, the *type of connection* (switched connection or permanent connection) or the *type of representation* of information (e.g. character sets).

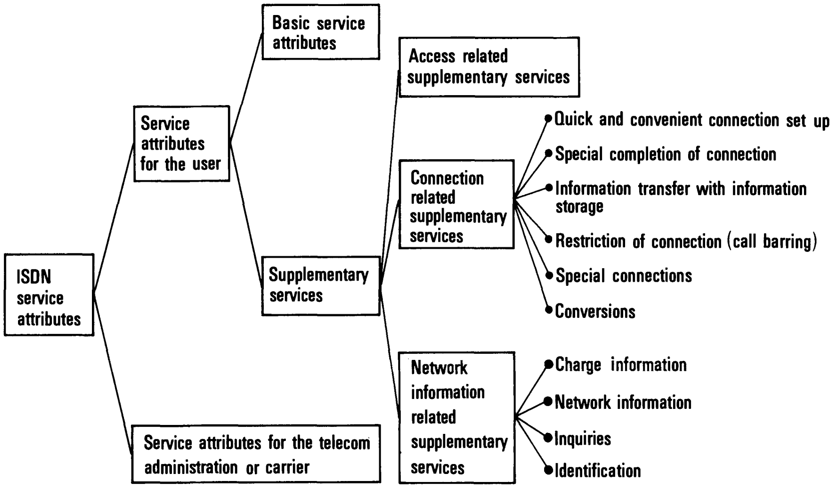


Fig. 2.5. Service Attributes and Supplementary Services in the ISDN (Ref. [2.34])

The *supplementary services* modify or augment the basic attributes and characteristics of a service, e.g. in terms of higher communication quality and enhanced communication convenience. They are not offered to a user separately as stand-alone services but only in conjunction with the basic service attributes of a communication service. The same supplementary service can be used with different services. Examples of this are *abbreviated dialing*, *call waiting*, *closed user group*. The main supplementary services in the ISDN are described more fully in Sect. 2.3.3

Corresponding to their assignment to the lower layers 1 to 3 or the higher layers 4 to 7 of the OSI reference model, the service attributes are designated low layer attributes describing access to the network, or high layer attributes concerned with communication between terminals (end-to-end) and oriented towards applications [2.31]. The basic service attributes and supplementary services are implemented in the communication networks and/or terminals with the aid of appropriate functional elements. Here, too, the same functional element, e.g. *transmission of the subscriber line identification*, *terminal compatibility check* or *access authorization check*, can be used for several basic service attributes and supplementary services.

2.3 Services in the ISDN with Bit Rates up to 64 kbit/s

Table 2.1 gives a list of possible ISDN services. In the table, each group of services, namely the services via B-channels or via the D-channel, the existing services from the telephone network and some data services of digital data networks as well as the services for connecting existing terminal equipment, are arranged according to the classification given in Sect. 2.1. The services are described in detail below. Concepts and plans for introducing ISDN services in several countries are given in [2.35 to 2.37].

Table 2.1. Possible Services in the ISDN with Bit Rates up to 64 kbit/s

Service classes	ISDN services via B-channels (64 kbit/s)	ISDN services via D-channels	Existing services taken over from the telephone network	Services for terminals of dedicated text and data networks
Conversational services	*Bearer services:* ISDN Data Transmission *Teleservices:* ISDN Telephony, Telephone Conference ISDN Teletex ISDN Telefax (Group 4) ISDN Mixed Mode (ISDN Textfax) Telewriting Still Image Transfer Service Videophone Service[2)] ISDN Alarm Services ISDN Teleaction	ISDN Data Transmission (packet-switched) ISDN Alarm Services ISDN Teleaction	Telephony Telefax (Group 2/3) Data Transmission with V-interfaces (parallel, serial) Alarm Services Teleaction	For Teletex (2400 bit/s) Possibly for Facsimile Group 4 (9600 bit/s) For Data Transmission with X.21 and X.25 interfaces
Messaging Services	Voice Mail Text Mail Fax Mail			
Retrieval Services	ISDN Videotex		Videotex	Videotex
Distribution Services	Data Distribution Voice Distribution Still Image Distribution			

2 With highly reduced spatial and temporal resolution

2.3.1 ISDN Services up to 64 kbit/s via B-Channels

The ISDN 64 kbit/s channel connections between any users allow ISDN services to be introduced which are optimized for this transmission rate [2.38, 2.39]. (For an interim period some ISDNs or parts thereof may only be capable of supporting 56 kbit/s unrestricted bearer services, cf. Sect. 7.2.3, page 183). The standardized signaling (D-channel protocol) and the uniform transmission bit rate on the traffic channel (B-channel) will also facilitate rapid and cost-effective implementation of new services and terminals.

2.3.1.1 Conversational Bearer Services

The powerful 64 kbit/s ISDN transmission channels provide sufficient transmission capacity for most data transmission applications in the office and the home. Data transmission services established on these communication paths using both switched and permanent connections will open up new applications in the business and private sectors, far exceeding the capabilities of today's bearer services with their different transmission speeds. Example of this range of services are the 64 kbit/s unrestricted bearer services standardized by CCITT for the transmission of text, data and still images [2.32].

2.3.1.2 Conversational Teleservices

The *ISDN telephony service* represents a considerable advance in voice communication for both the business subscriber and the private user. Due to the technical characteristics of the ISDN, this new telephony service provides enhanced speech quality and greater convenience: improved signal-to-noise ratio, attenuation unaffected by distance, improved handsfree talking facilities. At a further stage in its development, ISDN telephony could offer a greater voice bandwidth, e.g. 7 kHz, and stereo sound – an important service attribute for audio conferencing. The user is also offered a large number of new supplementary services as part of this service (Sect. 2.3.3).

The *ISDN teletex service* is a possible further development of today's teletex service standardized by CCITT [2.8] and offered via circuit-switched and packet-switched digital data transmission networks [2.40]. Due to the higher transmission rate of 64 kbit/s, the time taken to transmit a page of ISO A4 or North American legal paper size is reduced to less than 1 s. The protocols and functions for text transmission and presentation (functions of OSI layer 4 to 7) are based on those of the existing teletex service [2.8 to 2.12], so that interworking between these two services via service conversion facilities is possible. Rapid text transmission with ISDN teletex or a combined text/facsimile service (ISDN mixed mode) will become increasingly important for future office communication.

The facsimile service *ISDN telefax* is used for the transmission and high-quality reproduction of pictures, drawings and handwritten text. As far as the protocols (layers 4 to 7), coding and resolution are concerned, this service will be based on the CCITT Recommendations for facsimile group 4 [2.11, 2.12, 2.15 to 2.18]. These Recommendations provide for resolutions of up to 300 ppi, optionally 400 ppi up to 1 200 ppi; a resolution of 400 ppi corresponds to the quality of present-day office copiers. At this resolution and with black-and-white reproduction, the average amount of pixel-encoded information contained in a page of ISO A4 or North American paper size can be transmitted in about 15 s on ISDN telefax (64 kbit/s). ISDN telefax can thus provide for the first time a practical high-quality facsimile transmission system which can accompany voice communication.

The combined text and facsimile service, *ISDN mixed mode* (also called ISDN textfax), is particularly suitable for cost-effective transmission of black-and-white documents consisting of text and graphics or having hand-drawn illustrations, and of letters with letter heading and signature. This service provides two modes of operation, one for the character-coded transmission of texts, and the other for the pixel-coded transmission of graphics and pictorial information (facsimile). The service attributes such as character sets, resolution, communication protocols, transmission time, etc. correspond to those of the ISDN teletex and ISDN telefax services [2.10 to 2.12, 2.17 to 2.20]. The extension of the mixed mode service to other information types besides text and facsimile, such as graphics (geometric mode), still images with high resolution and audio (voice annotations), may be possible in the future.

A further communication service intended primarily for private use is *telewriting*. Short messages are written with an electronic pen on a note pad and sent to the receiver directly or stored in his electronic mailbox. The messages may be reproduced on the domestic television screen by vector graphics using an accessory unit or on other terminals with video screens in the home or office of the addressee. This service could be introduced to supplement telephone communication (using integrated voice and telewriting terminals).

The *ISDN still image transfer service* is used to transmit individual TV freeze frames. It is also possible to transmit still image sequences using special compression methods so that, depending on content, a new image is reproduced every 1 to 10 s. In this way the impression of moving pictures can be produced to a limited extent ("slow motion").

The *videophone service* with 64 kbit/s or 2 × 64 kbit/s offers users a moving-picture transfer capability for person-to-person, person-to-group and conference communication. However, due to the bit rate of 64 kbit/s (or 2 × 64 kbit/s), this service provides only a low video quality with highly reduced spatial and temporal resolution compared with the video quality offered by existing television broadcast services and planned to be offered by the future broadband video services (cf. Sect. 2.5). Applications are likely to be found in the commercial sector in cases where high video quality is not required. Moreover, the integration of this video service with broadband video telephony, television distribution services, and other visual services such as videotex in multifunctional terminals and combined service centers is not easily achievable because of the complex coding techniques and different display characteristics giving rise to excessive implementation costs.

The *alarm services* include facilities for emergency calls. *Teleaction services* are used for telemetering (e.g. meter reading), monitoring and controlling of commercial and domestic installations (heating, electricity, gas, water) and for controlling road traffic systems.

These services generally require transmission paths with high availability, a low bit error ratio, special provision for reliability and data protection and short transmission times.

Teleaction and alarm signals with higher data volumes and higher priority can be transmitted in the ISDN via circuit-switched and via permanent B-Channels. For applications involving very small data streams, transmission in the D-channel is preferable (see Sect. 2.3.2.2).

2.3.1.3 Messaging Services

Voice mail, text mail, fax mail and the message handling service within the videotex service are messaging services incorporating mailbox functions for voice, text, facsimile or data. Using teleservices of this type, information can be sent to communication partners even from unattended terminals by depositing voice, text or graphic information in their personal electronic mailbox. As a supplementary service, the user will be automatically informed by the mailbox system that he has received a message in his mailbox. This notification may be sent to the user over the signaling (D) channel independently of the operational status of the user's terminal and network access or via a free B-channel by a special control message. The user can retrieve individually the messages placed in his mailbox. The electronic mailboxes assigned to individual users or user groups can be implemented in storage facilities connected to the ISDN at central points (cf. Sect. 6.1). Access by the subscriber to the storage centers using the ISDN voice and text/facsimile/data services will be considerably facilitated by the service integration features of the ISDN mentioned in Sect. 1.6.

With suitable equipment, the messaging services can also be used for data transmission during low-charge periods, for multiaddress (broadcast) transmission and for communication between noncompatible terminals (via service interworking unit with conversion and processing functions). Messaging systems of this type are termed "message handling" systems (MHS) by the CCITT and recommendations for these have already been agreed [2.41].

2.3.1.4 Retrieval Services (Teleservices)

ISDN videotex is a further refinement of the present-day videotex service [2.42]. It is designed for combined application of the alphamosaic, geometric and photographic modes of operation [2.23 to 2.24] for the efficient transfer of mixed text and graphic information. The processing speed matched to the transmission rate of 64 kbit/s, and the optimized coding methods (reduced redundancy) permit rapid image buildup (within 1 to 20 s depending on the image content, operating mode and coding method) for information retrieval and rapid image input by the information providers at the videotex centers. It is only at a rate of 64 kbit/s that the transmission time of (pixel-coded) images and graphics in the photographic mode is acceptable for the user.

Due to the picture reproduction method of existing domestic television sets, the resolution of the information that can be represented with videotex is today limited to 480 pixels horizontally and 240 pixels vertically. When digital television technology is generally established and picture tubes with higher resolution are used, e.g. for future high-definition television (HDTV), it will be possible to formulate the ISDN videotex standard accordingly for a higher resolution.

2.3.1.5 Distribution Services

64 kbit/s distribution services will not have the same importance as future broadband distribution services (e.g. for television programs).

They are only suitable for special applications of voice, data and still image distribution. Due to the unidirectional transmission paths, transmission error correction techniques with error indication in the backward direction (ARQ) cannot be used. The use of such distribution services is only practicable in conjunction with forward error correction techniques (FEC), or if the messages are transmitted with cyclic repetition and are overwritten in the terminal of the receiver at short intervals. The application of such services is still unresolved.

2.3.2 ISDN Services via the D-Channel

The D-channel of the ISDN is primarily used for signaling between user and network. The D-channel is also used – always taking the signaling priority into account – for transmitting data in packet mode from data transmission services and e.g. alarm or teleaction signals from security and teleaction services.

The available transfer rates of additional services of this kind via the D-channel vary as a function of the signaling load on the D-channel. Average throughputs of up to 10 kbit/s are conceivable on a D-channel with a transmission capacity of 16 kbit/s.

2.3.2.1 Conversational Bearer Services with Low Throughput

To supplement the packet-switched bearer services via B-channels (see Sect. 2.3.1), packet-oriented bearer services with virtual call and permanent virtual circuit facility via the D-channel are subject to CCITT standards [2.32]. These can be used, in compliance with the respective national specifications, for transparent transmission of text and data with low throughput. The abovementioned restrictions in terms of throughput and priority apply.

2.3.2.2 Conversational Teleservices

Possible teleservices using the D-channel will be *alarm services* for emergency calls and *teleaction services* for remote meter reading etc. and remote control as already described in Sect. 2.3.1.2

For applications with very small data streams, it is generally preferable in the ISDN to use the signaling channel with its packet-oriented transmission structure. Again, the above-mentioned restrictions in respect of throughput and priority apply.

2.3.3 Supplementary Services

Supplementary services supplement the basic service attributes (see Fig. 2.5). Whereas the basic service attributes describe the minimum configuration of a service and are thus a fixed component of it, a user has the option of changing or improving this basic service by means of supplementary services.

Table 2.2. Important Supplementary Services in the ISDN and Their Assignment to Some ISDN Services[1)]

ISDN supplementary services	ISDN services				
	ISDN data transmission	ISDN telephony	ISDN teletex	ISDN telefax (Group 4)	ISDN videotex
Access related supplementary service					
Change of service during a connection	×	×	×	×	
Multiple subscriber number	×	×	×	×	×
Connection related supplementary services					
Call waiting		×			
Registration of incoming calls		×			
Call forwarding unconditional		×			
Call forwarding no reply		×			
Call forwarding busy		×			
Completion of calls to busy subscribers		×			
Direct dialing in	×	×	×	×	×
Reverse charging/freephone service	×	×	×	×	×
Closed user group	×	×	×	×	×
Conference calling		×			
Abbreviated dialing	o	o	o	o	o
Call hold service		×			
Outgoing call barring		o			
Incoming call barring	×	×	×	×	
Redialing		o	o	o	o
Multi-address calling	o		o	o	
ISDN networking services (City-wide centrex)	×	×	×	×	×
Network information related supplementary services					
Calling line identification presentation		×	× o	× o	
Date and time			×	×	o
Malicious call identification		×			
Advice of charge	×	×	×	×	× o
Announcements		× o			o

× Implementation in switching equipment;
o Implementation in terminals or special service centers.

[1] These are only examples of possible ISDN services and supplementary services. A particular ISDN may not offer all these ISDN services and supplementary services. Some telecommunication carriers may offer ISDN-based services which have not been defined or identified by CCITT.

The supplementary services are subdivided into the following groups in accordance with [2.34] (Fig. 2.5):

- Access related supplementary services,
- Connection related supplementary services,
- Network information related supplementary services.

The supplementary services possible in the ISDN are illustrated below using appropriate examples. Table 2.2 provides an overview of the main supplementary services, assigned to five new ISDN services. This table also indicates whether the individual supplementary services should preferably be implemented in switching equipment, in terminals or in special service centers (e.g. videotex centers).

Possible access related supplementary services

- *Change of service during the connection*, e.g. change from voice communication to facsimile transmission.
- *Status interrogation* of the entered (activated) service attributes, e.g. whether one of the connection related *call forwarding supplementary services* (see below) is switched on.
- *Multiple subscriber number* for selecting specific terminals on the passive bus of the called user, if necessary with distinctive ringing.

Possible connection related supplementary services

These are ISDN supplementary services for *quick and convenient connection set-up*, *special completion of connection*, *information transfer with information storage*, *restriction of connection (call barring)*, and *special connections* and *conversions (service, network and terminal adaptations)*.

- Quick and convenient connection set-up and convenient communication
  - *Abbreviated dialing*: for frequently used directory numbers the user can employ two-digit abbreviated numbers for connection set-up.
  - *Direct dialing in* to extensions in private branch exchanges.
  - *Redialing*: the directory number last dialed is stored. Dialing is repeated by pressing a specific key or automatically after a preset time.
  - *Fixed destination call*: after actuating any key or merely by lifting the handset, a connection is set up automatically to a line with a specific directory number.
  - *Handsfree dialing.*
  - *Handsfree speaking* via a separate microphone.
- Special completion of connection
  - *Completion of calls to busy subscribers*: the calling user can activate this feature if a called line is busy. When the called line is released the network will inform the calling user that the called party is no longer busy. On request of the calling user a recall to the called party will be set up by the network.
  - *Call waiting*: during an existing connection the called user is audibly and/or visually informed of the presence of a further request for connection, possibly with indication of the calling number. He can accept the second connection within a distinct time interval.
  - *Call forwarding unconditional*: the user can forward all incoming calls to any other line by entering its directory number.
  - *Call forwarding no reply*: the user can enter any other directory number, to which an incoming call is forwarded if he does not accept this call within a certain time (e.g. three rings).

  - *Call forwarding busy*: incoming calls are forwarded to the entered directory number if the called subscriber is busy.
  - *Selective call forwarding*: calls from certain directory numbers specified to the switching system by the subscriber are not forwarded.
- Information transfer with information storage
  - *Absent subscriber service* with or without storage of the replies: the functions are the same as those of automatic answering equipment.
  - *Registration of incoming calls*: if calls are made while the called subscriber is absent, the date, time, and directory number of the calling subscriber are recorded for the information of the called subscriber.
- Restriction of connection (call barring)
  - *Total call barring*: barring of the line for all outgoing and incoming connections on request by the subscriber.
  - *Outgoing call barring*: on request by the subscriber for e.g. intercontinental connections, international connections or long-distance connections.
  - *Incoming call barring*: on request by the subscriber, barring at certain times (e.g. do-not-disturb service) or permanent barring (line for outgoing calls only) of all connections or services specific connections.
- Special connections
  - *Freephone service*: the called subscriber accepts all charges for incoming calls. Certain directory numbers are made available for this, such as *Service 800* in the USA.
  - *Reverse charging* (case by case): the called subscriber can accept the connection charges on a case-by-case basis during the call or before accepting the call.
  - *Wakening call service*: the user arranges for a wakening call on his line by specifying the date and time.
  - *Multi-address calling*: for transferring one-way messages, several main station lines are called simultaneously or consecutively (information distribution).
  - *Call hold service*: the user can switch the existing connection with a second user to a wait status, in order to set up an additional connection to a third user for inquiry purposes or to accept a waiting connection.
  - *Conference calling*: connections with three users or more at the same time, each being able to communicate with all the others.
  - *Closed user group*: the users form groups with specific, agreed restrictions for access to and from users of the public network (private branch exchange functions).
  - *ISDN networking services* (*city-wide centrex*): PABX-related supplementary services to support easy set-up and control of the call (e.g. *private numbering plan*), to increase the accessibility of the communication partner, for access authorization, special accounting, directory inquiries and other facilities provided to subscribers by means of specially equipped public exchanges.
- Conversions: If no conversion function is provided, interworking is only possible between compatible services, networks and terminals. To support unrestricted (open) communication, conversion functions have to be provided for service, network and terminal adaptation: conversion of bit rates, protocols, interfaces, etc.

Possible network information related supplementary services

These include possible ISDN supplementary services such as *charge information, network information, directory inquiries*, and *identification.*

- Charge information
  - *Advice of* the current or accumulated *charge* of a call as charge units or amount of money during a connection or after disconnection.
  - *Printed record of call charge* (in text communication services) after disconnection.
  - *Toll ticketing*: itemized billing for each connection and service (date, time, directory number of the called subscriber).
- Network information
  - *Announcements*: indication or announcement e.g. of a changed directory number.
  - *Call progress signals* for user guidance during connection set-up.
  - Indication of *date and time* on connection set-up.
  - *Incoming message waiting indication*: the user is automatically informed by the mailbox system that a message has been deposited in his mailbox.
- Directory inquiries
  - *Public recorded information services* such as weather, sports results, news.
  - *Telephone directory inquiry services.*
  - *Directory inquiry services* for telex, teletex, other services.
- Identification
  - *Malicious call identification*: recording the directory number of malicious callers.
  - *Calling line identification presentation* at the called subscriber's terminal.
  - *Connected line identification presentation* (for the calling user).

2.3.4 Existing Services from the Telephone Network in the ISDN

In the ISDN, the services of the telephone network can be retained alongside newly introduced services at least for a transitional period, so that existing terminals of present-day services of the analog telephone network can continue to be used.

Consequently, the existing telephone service with its analog telephones can also be initially accommodated in the ISDN. The subscriber of this service is assigned an analog line at his ISDN exchange (Fig. 2.6, see also Sect. 6.1). All text and data services of the existing telephone network can be handled as before on this line. This applies to data transmission services with modems (in accordance with CCITT V.-series Recommendations, some of which correspond to EIA standards, e.g. RS−232−C) at the transmission rates currently defined for the telephone network, and also to the telefax and videotex services.

However, these services can also be provided at the ISDN access via a special terminal adaptor (see Sect. 4.1.4) with t/r ("tip and ring") interface (Figure 2.7). In these cases the analog modem signals are digitized in the adaptor unit and are switched and transmitted as digital signals. The service and terminals remain the same, but using the service via the ISDN access has the advantage that ISDN capabilities can also be used for the existing service, e.g. the integrated multichannel access or the improved call

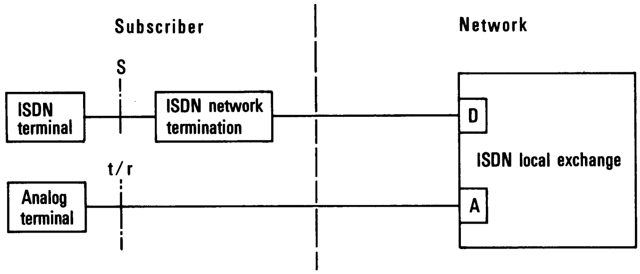


Fig. 2.6. Digital and Analog Subscriber Lines Connected to an ISDN Local Exchange.
A analog line module, D digital line module, t/r analog user-network interface, S ISDN user-network interface

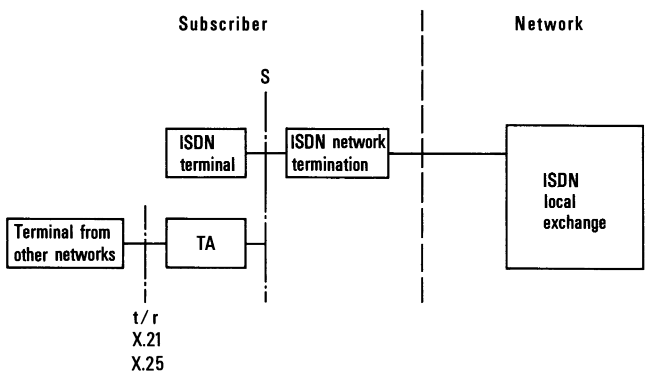


Fig. 2.7. Examples of Adaptation of Terminals from Other Networks to the ISDN Subscriber Line.
S ISDN user-network interface, t/r analog user-network interface, X user-network interface as specified for data networks, TA terminal adaptor

establishment by means of an ISDN terminal within the user station (cf. Sect. 1.6). For example, due to the multichannel capability of the ISDN line, the telephone circuit is not blocked while videotex is being used.

2.3.5 Services in Conjunction with Terminals from Dedicated Text and Data Networks

As far as required, the ISDN will also provide access possibilities, via terminal adaptors, for existing terminals intended for operation in dedicated switched text and data networks (Fig. 2.7). Here, too, the user who needs ISDN access to make use of new ISDN services can enjoy the advantages of the integrated ISDN subscriber line even if he retains some conventional terminal equipment. The main options for access to the ISDN are as follows:

- Data terminals with interface in accordance with CCITT Recommendation X.21 for circuit-switched traffic (cf. Table 2.3).

Table 2.3. User Classes in Circuit and Packet Switched Data Networks

Terminals with interface as per CCITT Rec. X.21		Terminals with interface as per CCITT Rec. X.25	
User class[a]	Speed in bit/s	User class[a]	Speed in bit/s
4	2400	8	2400
5	4800	9	4800
6	9600	10	9600
		11	48000

[a] Numbering as per CCITT Rec. X.1.

- Data terminals with interface per CCITT Recommendation X.25 for packet-switched operation (cf. Table 2.3).
- Teletex terminal equipment (2.4 kbit/s)
- Group 4 facsimile equipment.

Although no changes will be made to these terminals, the subscriber who connects them to the ISDN will receive a service that differs from that provided in the text and data network. This is due to the different characteristics of the network; for example, the connection setup times may be different.

2.4 Service Conversion and Network Interworking Facilities

In specifying services, the aim is to facilitate communication between as many users of a network as possible. The same applies to interworking between terminals of different

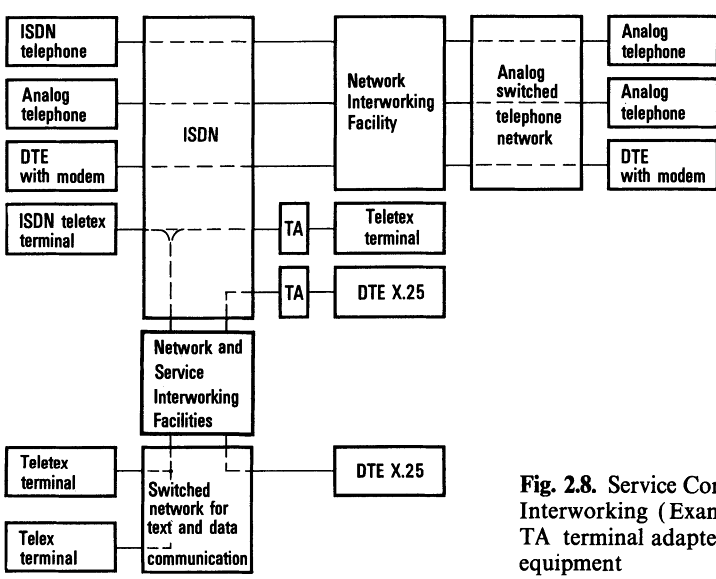


Fig. 2.8. Service Conversion and Network Interworking (Examples).
TA terminal adapter, DTE data terminal equipment

services and of different networks. This is achieved by means of service conversion and network interworking facilities as required (Fig. 2.8).

2.4.1 Service Conversion within the ISDN

It goes without saying that voice communication between existing analog telephones and ISDN telephones should be possible for as long as analog telephones are in use. When connected to subscribers of the existing public telephone network, the ISDN subscriber naturally cannot fully utilize the supplementary services of the ISDN telephony service. For example, *calling line identification presentation* (cf. Sect. 2.3.3) cannot be realized.

In connection with services for non-voice communication, the following service conversions are important within the ISDN:

- ISDN teletex service (64 kbit/s) – teletex terminals (2.4 kbit/s) with terminal adaptor.
- ISDN mixed mode service – ISDN separate text and facsimile service.
- Conversions between services for different groups of facsimile equipment, especially groups 3 and 4 facsimile machines.

2.4.2 Service Conversions in Conjunction with Network Interworking Facilities

Just as within the ISDN voice communication between telephones connected to an analog line and to an ISDN line must be possible, so telephones connected to the analog telephone network or to not yet converted analog parts of the network must be able to communicate with telephones connected to the ISDN. For the ISDN subscriber, essentially the same restrictions as mentioned in Sect. 2.4.1 apply.

In countries with dedicated text and data networks, interworking facilities between the ISDN and these networks are of considerable importance. For example, they enable a computer connected to the data network to be accessed from within the ISDN; all the teletex terminals of both networks can communicate with each other, and there exists (via teletex) a connection from the ISDN to the telex service.

2.5 Services with Higher Bit Rates

Besides the services based on the B-channels (64 kbit/s) or on the D-channel of the ISDN subscriber line, as the ISDN develops further services with higher bit rates can be expected.

The CCITT has already defined so-called H-channels for the ISDN (see Sect. 4.2) with bit rates of 384 kbit/s, 1920 kbit/s for the "European" digital systems (cf. Sect. 7.2.3) and 1536 kbit/s for the "American" digital systems. The 384 kbit/s channel is intended to be used among other things for broadcast sound transmission. The 1920 kbit/s and 1536 kbit/s channels are intended for video conferencing (connection of conference rooms) and for data transmission (e.g. file transfer, newspaper transmission).

As the ISDN described here grows into a broadband network, full-motion video communication will become possible. Channels with bit rates from about 30 Mbit/s to about 140 Mbit/s (depending on the use of redundancy reducing measures) will be required in order to offer a picture transmission quality equaling that of color television [2.43]. For this purpose, it will be necessary to use optical fibers instead of copper wire pairs for the subscriber line (see Sect. 7.3).

Channels operating at these high bit rates will in turn provide a host of new services. The videotelephony service has a key role here, for in principle every telephone subscriber is a potential videotelephone subscriber. A special type of videotelephony is videotelephone conferencing, involving both the connection of several videotelephones in conference mode and the connection of conference studios. Further services are based on the retrieval of television quality still pictures and of film scenes. Closely related to this is a further refined form of the videotex service. Television with a larger number of programs is possible. It is conceivable that messaging services will be established for storage and retrieval of still pictures and film scenes in electronic mailboxes.

In addition to this, high-capacity transmission services for data traffic will be provided. These could promote a useful decentralization of data processing system architectures, in the form, for example, of distributed databases. Based on these high-capacity transmission channels, a standard "document transfer" service would be a desirable possibility; it would realize the concepts of a document containing information in the form of character-coded text, facsimiles and speech (for annotation), and of being able to access the document rapidly on a "page-turning" basis.

3 ISDN Structure

3.1 Network Organization

Communication networks essentially comprise three components (Fig. 3.1): the subscriber line network, the exchanges and the trunk network.

- The *subscriber line network* connects the subscribers' terminals to the local exchanges to which they are assigned. The subscriber lines may be provided on a per subscriber basis, as in most of the telephone network, or shared by several subscribers by means of multiplexing. The latter approach is frequently adopted in the subscriber line area of text and data networks, as the lower density of these networks results in greater distances between subscriber and exchange. Copper wire pairs 0.4 to 0.8 mm in diameter are used for the subscriber lines (see Sect. 7.4.3). In the subscriber line network of the telephone system, each subscriber is served by a two-wire circuit. In text and data networks, due to the greater distances involved, data transmission systems using modulated carriers or baseband transmission are employed. These systems exist in both two-wire and four-wire versions. Four-wire circuits are also provided in some private networks.
- The *exchanges* establish the connection, depending on the directory number dialed, between the subscriber line and an outgoing trunk circuit (local exchange, local office) or interconnect the trunk circuits of different trunk groups (transit or tandem exchange). In many cases the same exchange carries out both functions (combined local/transit exchange). As it is not economic in national networks for all the local exchanges to be interconnected in a mesh network, local networks are

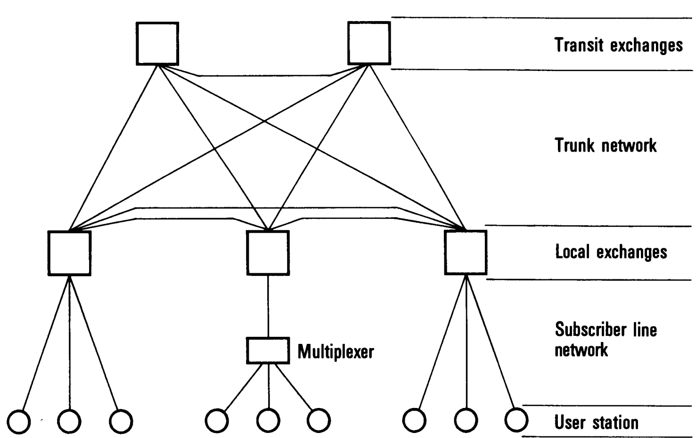


Fig. 3.1. Basic Components of a Communication Network

formed within defined geographical areas. The long-distance traffic of these local networks is fed in concentrated form to the long-distance (toll) network which interconnects the regional centers. This network arangement creates a distinction between exchanges for local and long-distance traffic.

- The *trunk network* interconnects the exchanges. *Local trunks* connect the exchanges within the local network, *long-distance (intertoll) trunks* the exchanges of the toll network. The local and toll networks are linked via *trunk junctions (called toll connecting trunks* in the USA).

In addition, functions and equipment for control and maintenance are required for operating a network. These include call charge registration, call data administration, quality and grade of service measurement, and maintenance of the network equipment.

3.2 Initial Situation for the ISDN

The ISDN cannot be regarded independently of the existing networks. These are the *telephone network*, designed primarily for handling telephone traffic, and the special networks for text and data traffic.

3.2.1 Telephone Network

Telephone networks are mostly configured as combined star/mesh networks and are subdivided into several hierarchical levels. The number of hierarchical levels depends on a number of factors, important variables being the size of a country and the subscriber

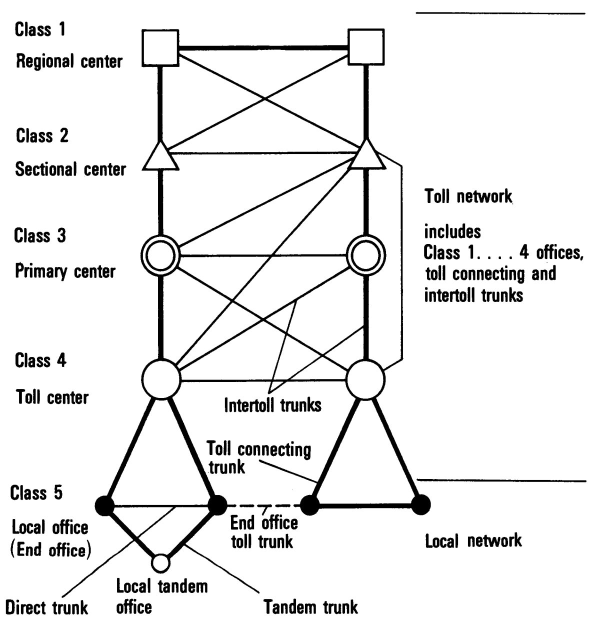


Fig. 3.2. The Hierarchical Network Switching Plan of the USA

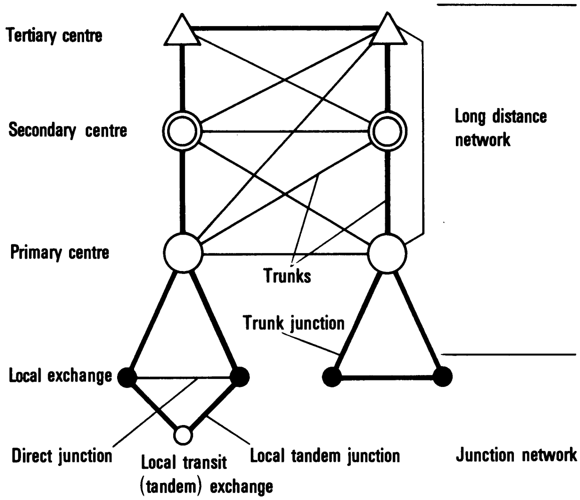


Fig. 3.3. A Four-Level Hierarchical Network Example (Mainly CCITT Terms)

density. For example, the telephone network in the United States of America is divided into five hierarchical levels (Fig. 3.2), whereas in other countries a maximum of four suffices (Fig. 3.3). Various terms are commonly used to designate the exchanges and the trunk groups connecting them (cf. Figs. 3.2 and 3.3).

Due to the hierarchical structure of the network, the number of exchanges is greatest at the lowest level and decreases towards the higher levels. To give an example of very different network dimensions, Table 3.1 compares the telephone network of the USA with that of the Federal Republic of Germany.

Each exchange is connected to the next exchange above it in the hierarchy via the trunk groups of the star network (heavier lines in Figs. 3.2 and 3.3). In addition to these circuits forming the so-called final trunk group, there are high usage trunk groups which connect exchange at the same and different hierarchical levels via the mesh network. These high usage trunk groups are provided between two exchanges if there is sufficient traffic volume to justify them on economic grounds. Whether such high usage trunks are provided also depends on the technical capabilities of the switching systems. While modern SPC-controlled exchanges generally impose no limitations, the older electromechanical systems will not allow every economically viable high usage trunk group to be implemented. This may be because the maximum number of connectable trunk groups is limited or there are restrictions as regards call number interpretation.

Telephone networks are further subdivided into local and long-distance (toll) networks. The local network is used for handling traffic within a defined geographical area, e.g. a city, and may itself be subdivided into several hierarchical levels. Long-distance and international traffic is handled via the toll network. This division into local and toll network is attributable to the different technical requirements of the two parts of the network. Whereas with analog transmission and switching a two-wire arrangement is usually adequate in the local network due to the shorter distances to be spanned, in the toll network a four-wire arrangement is necessary. Furthermore, lower

Table 3.1. Comparison of the Public Telephone Networks in the United States of America and the Federal Republic of Germany (1982) [3.1 and 3.2]

Hierarchical level	Number of exchanges	
	USA	Federal Republic of Germany
Class 1/regional center (Quaternary centre[a])	10	–
Class 2/sectional center (Tertiary centre[a])	52	8
Class 3/primary center (Secondary centre[a])	168	56
Class 4/toll center (Primary centre[a])	933	417
Class 5/local office (Local exchange[a])	18 803	6 200
Number of main stations (millions)	92.3	23
Area (million square km)	9.36	0.25

[a] CCITT term

requirements are placed on the local exchanges than on the long-distance exchanges in terms of traffic routing and call charge registration. However, the progressive digitization of the networks, with four-wire transmission and switching throughout, and stored-program control of the exchanges make it possible to transfer toll network functions to the local exchanges. For example, in the United States there are combined local/long-distance exchanges which are directly interconnected at local exchange level via long-distance trunks (end office toll trunk, see Fig. 3.2).

By connecting voice-band data modems to the subscriber line, the telephone network can also be used for data transmission. For this purpose the digital data signals must be converted by the modem to voice frequency signals [3.3].

However, there are a number of limitations affecting data transmission over the telephone network, which for certain applications have no satisfactory solution. These include:

- the limited data transmission rate of 4 800 bit/s, or 9 600 bit/s at the most,
- the relatively high bit error ratio due to transmitting data as analog signals,
- connection setup times (including entry of the call number) in excess of 10 s,
- possible network blocking during peak traffic periods.

Nevertheless, in many countries data transmission over the telephone network is one of the most frequently used forms of data communication. This is due among other things to the fact that the equipment already installed for the telephone service can be used instead of setting up a separate network.

The *videotex service* [3.4] and the *telefax service* [3.5] for facsimile transmission are two other services mainly handled via the telephone network. For videotex, the telephone network establishes the connection between the videotex users and the information centers; for the telefax service, the connections are dialed-up as for a telephone call and then used for facsimile transmission.

In all these services the telephone network is used solely for information transport and does not provide any supplementary services such as *closed user group* (see Sect. 2.3.3) in addition to the service attributes for the telephone system.

3.2.2 Text and Data Networks

The special requirements of text and data communication, such as higher data transmission rates, shorter connection set-up times and lower error ratios for data transmission as compared with the telephone network, have led many countries to construct separate networks for text and data communication.

These networks employ different through-connection techniques (circuit switching, packet switching) and, due to the smaller number of subscribers, have fewer hierarchical levels than the telephone network. Owing to the lower subscriber density, it is usually only viable to construct exchanges in large population centers. More remote subscribers therefore have to be connected via comparatively long subscriber lines. To save costs in this part of the network, concentrators and multiplexers are generally employed; these concentrate the traffic on the subscriber lines and forward it to the exchanges over a reduced number of lines.

For the text and data networks the CCITT has produced separate Recommendations [3.6] which take into account the special requirements of these networks. For example, these networks provide a number of supplementary services, such as *closed user group*, *reverse charging*, and *multi-address dialing*, and permit synchronous and asynchronous operation of the connected terminal equipment as well as half-duplex and duplex operation.

The main advantages of networks operating on the *packet-switching* principle include matching of the different transmission procedures and rates of the connected terminals, short connection set-up times, use of the subscriber lines for several connections at the same time, and call charge advantages in certain applications, e.g. in the interactive mode.

In many data transmission applications, the connection set-up times and the risk of the desired connection not being immediately available due to lack of free transmission paths or due to the subscriber line being busy, are unacceptable. Moreover in many cases the advantage provided by a switched network of being able to reach a large number of lines is irrelevant because the actual traffic relations are limited to a few specific communication partners. For these applications, dedicated circuits become a viable option. These circuits are used to interconnect two data terminal equipments on a permanent basis. Connection set-up times are virtually negligible, the transmission path is always available and the line cannot be occupied by other calls. Some links of this kind are routed via analog transmission paths; as with data transmission over the telephone network, these require modems at the terminals to convert the digital data signals to voice frequency signals.

3.3 Functions of the ISDN Network Components

The ISDN is based on the digital network components of the telephone network (see Sect. 3.8). These facilities, designed to meet the requirements of the telephone network, have to be supplemented in order to realize the potential of the ISDN.

3.3.1 Subscriber Lines in the ISDN

In the ISDN all information is transmitted digitally, even on the subscriber line (Fig. 3.4a). (This represents a considerable departure from the digital telephone network, in which information is transmitted in analog form as far as the local exchange; see Fig. 3.4b). The existing copper wires will be used as transmission paths.

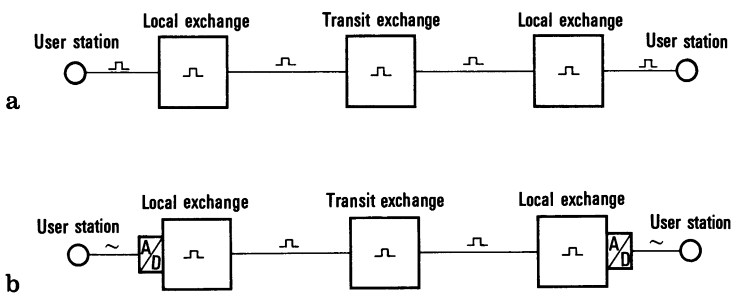


Fig. 3.4a, b. Network Concepts. **a** ISDN; **b** Integrated digital network (IDN). ~ Analog signal, ⎍ Digital signal, A/D Analog/digital converter

With suitable transmission methods (see Sect. 7.4.3), the copper wire pair of the subscriber line can be used to transmit the net bit rate of 144 kbit/s required for the ISDN basic access. In the subscriber line network, no additional expenditure, e.g. for regenerative repeaters, is normally required.

3.3.2 Local Exchanges in the ISDN

In the local exchange, the enhanced communication facilities provided by the ISDN result in a more complex subscriber access than that of the digital telephone network and place additional requirements on call control in the establishment and clearing phases (see Sect. 4.3.5). The essential functions of the subscriber line circuit in the exchange include splitting up the 144 kbit/s information stream into the two 64 kbit/s channels and the 16 kbit/s channel and separate processing of the two 64 kbit/s channels. These are processed separately because they can be switched through to different destinations and can be used for different services. It must be possible, during an established connection, to add-on a second 64 kbit/s channel and to change the service within a channel. This means that subscriber signaling must also be possible during an established connection. The ISDN differs in this respect from the telephone network, in which only call charge information and the criterion for clearing a connection are transmitted in the course of a call. The ISDN also enables several terminals to be connected to a subscriber line under a single directory number. For incoming calls, this necessitates an exchange of information between local exchange and terminal prior to through-connection, and sometimes between the terminals at either end in order to ascertain the terminal to which the incoming call can be assigned.

3.3.3 Signaling Between ISDN Exchanges

Signaling between ISDN exchanges requires a more extensive range of features compared with conventional telephone signaling. The method used is CCITT Signaling System No. 7 with central signaling channels between the exchanges, supplemented by additional functions for ISDN (see Sect. 6.3). The formats for identifying the services have been expanded, and changes and additions have been made to the signaling protocols for controlling new service attributes. Some of the signaling information to be passed between exchanges only relates to the orginating and destination exchanges. To handle this information, end-to-end signaling has been introduced as a new function. End-to-end messages are identified by a special code and are merely passed on by the intervening transit exchanges.

3.3.4 Assignment of ISDN Supplementary Services

For the assignment of supplementary services (see Sect. 2.3.3) to the network components of the ISDN, the following two groups can be distinguished:

- Supplementary services implemented in the terminal,
- Supplementary services provided by the network.

The supplementary services implemented in the terminals include those which assist the user during dialing and which do not require access to data in other network equipment. Examples of this are abbreviated dialing assigned individually to a terminal (using abbreviated directory numbers, fixed destination call, name keys) or redialing (see Sect. 2.3.3).

The network, on the other hand, primarily provides supplementary services requiring access to information stored in the exchanges or at other centralized locations in the network. These typically include call charge related services, such as indication of the charge accumulated for a call, or the closed user group facility restricting traffic to a specific circle of subscribers, and call forwarding.

Services which are only used by a comparatively small number of subscribers, such as retrieval services and mailbox services (see Sect. 2.1) are implemented in ancillary equipment – *service modules* (see Sect. 6.2.9). They are associated with ISDN exchanges located at central points in the network and serve a large "catchment area", i.e. all subscribers connected to their respective subordinate exchanges.

3.3.5 Operation and Maintenance in the ISDN

The operation and maintenance functions for the ISDN network components are largely subject to the same requirements as those affecting the digital telephone network, particularly where the exchanges and the trunk network are concerned. The main functions are detailed in Sect. 6.2.7. Additional requirements basically concern the subscriber line network, as the analog subscriber access is now replaced by a digital one. The subscriber access consists of several functional groups (see Sect. 4.1), such as the line terminal equipment at the exchange, the network termination unit on the subscriber side and the subscriber line. In order to enable a defective network component to be identified in the event of a fault, the subscriber line circuit is divided into several test sections within which test loops can be inserted [3.7]. These test loops can be remotely controlled from the exchange. By comparing the transmitted and received data in the test loops it is possible to determine whether a fault is present in the test section in question.

3.4 Network Dimensioning

3.4.1 Basic Considerations

The aim of network dimensioning is to design the network equipment used jointly by the subscribers, i.e. the exchanges and the trunk network, in such a way that the desired connections can be established without significant blocking even at peak traffic periods [3.8]. In the case of the exchanges, it is necessary to specify the number, position, size and traffic processing capacity; for the interexchange trunks, it is necessary to determine the number of circuits combined to form a trunk group.

In dimensioning switching systems a distinction is drawn between systems operating in the delay mode and those operating in the loss mode [3.9], depending on how switching tasks are handled when blocking occurs.

In the *delay mode*, if blocking occurs the subscriber can wait for a connection path or other switching facilities to become free. Information concerning the mean length of the delay, the probabilities of delays occurring and of exceeding specific delay times is used to describe the grade of service. Exchange control equipment normally operates in the delay mode.

In the *loss mode*, seizure requests which encounter blocking are rejected. The subscriber receives the busy signal. In this case the percentage loss – the ratio of the number of rejected calls to the total number of calls expressed as a percentage – indicates the grade of service. Switching networks employing circuit switching normally operate in the loss mode. If no free connection path can be found within the switching system ("internal blocking") or if no free serving trunk is available ("external blocking"), the seizure request is rejected (Fig. 3.5).

For network dimensioning it is important to specify a suitable grade of service, i.e. the acceptable loss in the case of circuit switching. In electromechanical systems with *limited availability*, in which not every outlet is accessible from every inlet of the switching system, in general a loss of 1 % per line section was the norm. As several line sections are cascaded within a connection path, and losses also occur in the exchanges, the total loss caused by the network for connections via electromechanical systems is in the order of 4 to 8 %. To this must be added losses attributable to the subscribers, e.g. due to "subscriber busy" and "subscriber does not answer" (Fig. 3.5).

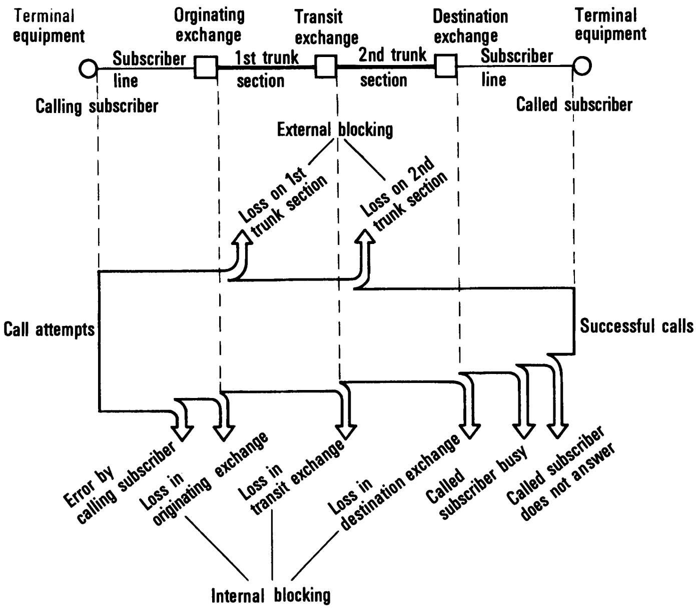


Fig. 3.5. Losses When Setting up a Call

In the digital switching systems with *full availability* used in the ISDN, with large trunk groups a loss of 1 % per line section leads, even under normal load conditions, to a very high level of trunk utilization. For the overload conditions which are inevitable even with careful network dimensioning, such as the higher traffic volumes on certain weekdays ("Mothers day"), insufficient load reserves would then be available. As subscriber busy conditions occasion frequent redialing, the number of seizure attempts snowballs. This results in an additional burden on the control equipment in the exchanges and generates additional dummy traffic in the network [3.10]. For these reasons trunk groups are today frequently designed on the basis of losses under particular overload conditions, e.g. for 3 % loss with 20 % overload. Under normal load conditions, this results in a loss which diminishes as the number of circuits per trunk group increases. Using this approach, an adequate grade of service is ensured even under peak load conditions.

3.4.2 Effects of Service Integration

3.4.2.1 Holding Time, Traffic Intensity, Busy Hour Call Attempts

In the ISDN voice calls will predominate. The mean holding time of such calls[1] (Fig. 3.6) is substantially determined by subscriber behavior; in contrast to data or text traffic, different bit rates for speech coding (see Sect. 7.2.1) do not alter the mean

1 Call is any attempt to seize a switching equipment or trunk, irrespective of the success of this seizure attempt.

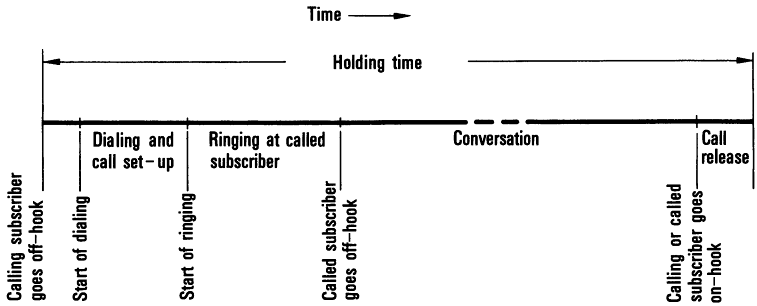


Fig. 3.6. Composition of the Holding Time for a Successful Voice Call

holding time. Only the shorter call establishment times in the ISDN (1 to 2 s compared with up to about 15 s in the analog telephone network) bring about a reduction in the holding time. The typical figure of approximately 100 s for the mean holding time of voice calls will thus remain substantially unchanged in the ISDN. The same considerations apply to calls with retrieval services such as videotex, and certain types of interactive calls using computers. Here it is mostly the subscriber response times which determine the mean holding time; the duration of information input and output, though affected by the transmission speed (bit rate), represents only an insignificant proportion of the overall holding time of such calls.

The effect of transmission rate on the holding time is much greater for calls which do not depend on subscriber response times but in which it is only the quantity of information and the transmission rate which determine the mean holding time. This applies, for example, to calls for text, data and facsimile transmission. The mean holding times of these calls, for instance typically 10 s for 2.4 kbit/s teletex, – already short compared with voice traffic – will be reduced still further with the transition to the uniform transmission rate of 64 kbit/s (Table 3.2).

Less pronounced are the effects of service integration on the total traffic in the network. Due to their mainly short holding times, text and data calls generate little traffic compared with voice traffic for the same frequency of calls. For example, for the

Table 3.2. Estimated Traffic Volumes in the Busy Hour for Different Communication Services with a Transmission Bit Rate of 64 kbit/s

Service	Mean holding time s	Mean BHCA per line, outgoing and incoming calls per hour	Mean traffic intensity per line, outgoing and incoming Erlangs
Videotex	300	0.36	0.03
Telephony	100	3.6	0.1
Data transmission	15	24	0.1
Facsimile	10	7.2	0.02
Teletex	2.5	7.2	0.005

BHCA Busy hour call attempts

same number of connections the traffic intensity of 64 kbit/s teletex connections is only about one fortieth of the traffic intensity of speech connections.

Of greater significance, on the other hand, are the effects of service integration on the number of busy hour call attempts (BHCA). Especially for interactive calls in data traffic, call frequencies per subscriber line many times greater than the values hitherto encountered in the telephone service can be expected (Table 3.2). This is due to the fact that these applications are mainly prevalent in the business sector where intensive utilization of the existing terminals is an economic necessity. Furthermore, a single call only lasts for a few seconds, and so considerably more calls can be set up within a given time than in telephony.

Generally speaking, the traffic requirements will change in the following way compared with the telephone network:

- The traffic volume to be processed per line will increase due to the text and data services.
- The mean holding time will be reduced.
- In the network the number of BHCA will rise considerably more than the traffic carried. The switching equipment in the network will have to provide enhanced control performance.

Further effects will result from the following:

- Implementation of *supplementary services* (see Sect. 2.3.3), such as completion of calls to busy subscriber, call forwarding/diversion and call waiting.
- Implementation of *mailbox services* with storage of speech and text information which is selectively retrieved on request (see Sect. 2.3.1.3).
- The possibility of change of service during a call, or of the choice of an alternative service in the case of subscriber busy (see Sect. 2.3.1).

Although these facilities initiate additional control procedures in the network, they also increase the availability of the desired communication partner and thus reduce the number of unsuccessful call attempts. They will have the effect of changing the existing pattern of telephone calls [3.11, 3.12] (Table 3.3) to provide a higher proportion of successful calls.

3.4.2.2 Shared Use of the Network Equipment

A further traffic-related advantage of service integration arises from the sharing of the network's switching and transmission equipment by the different services. This applies

Table 3.3. Call Mix in the Public Telephone Network

Distribution of calls	in %
Error by the calling subscriber, e.g. off-hook without dialing, incomplete dialing	13
Congestion in the trunk network	6
Called subscriber busy	18
Called subscriber does not answer	8
Successful calls	55

in particular to subscriber lines which represent a significant proportion of the capital costs of a network. In the ISDN the copper wire pair of the subscriber line can handle two services simultaneously (e.g. telephony and videotex). Although it is even now possible for more than one service to be handled in the same transmission system, separate channels are provided for these services within that system. Full sharing is only possible if each channel of the system is accessible to each service. This confers the following advantages:

- Text and data service traffic, low in volume compared with the telephone service, can also use the economical high usage interexchange trunk groups installed for bulk telephone traffic. A single service does not in itself always generate a sufficient volume of traffic to justify a high usage trunk group. This concentrating of traffic becomes increasingly important when digital transmission systems are used, because they provide a high channel capacity (30, 120, 480, 1 920 or 24, 96, 672 channels or multiples thereof, each operating at 64 kbit/s, see Sect. 7.2, Fig. 7.2).
- Different busy hour times and other traffic fluctuations in the individual services largely balance each other out if common trunk groups are used for all services. The network which only carries light telephone traffic during the night, can also be used for transmitting text and data.

3.4.3 Traffic Routing

There are usually several possible routes within a network between the origin and destination of a call. If the traffic volume between exchanges is sufficiently great, high usage trunk groups are set up which handle 80 to 90 % of the offered traffic. For the remaining 10 to 20% of the traffic, next choice routes and last choice routes via one or more transit exchanges are available (Fig. 3.7). The aim of traffic routing is to find the most suitable path for a call from these possible alternative paths and seize it, provided at least one circuit is still free in the desired trunk group. When setting up calls between ISDN subscribers who require a wholly digital link, care must be taken with traffic routing in mixed analog/digital networks to ensure that only digital paths are selected.

In both local and long-distance networks, digitalization has the following effects on traffic routing:

- The channel capacity of the basic digital system (30 traffic channels in the 2 Mbit/s system or 24 in the 1.544 Mbit/s system; see Sect. 7.2.3) is high compared with both

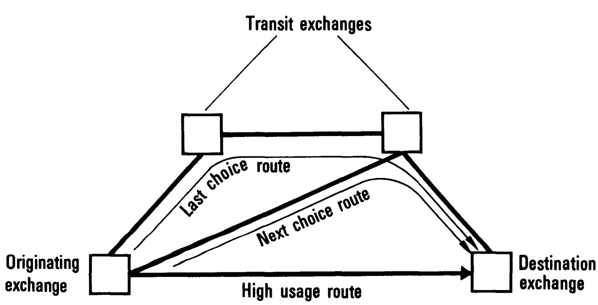


Fig. 3.7. Traffic Routing Schematic

analog local trunks and an FDM system with e.g. 12 voice channels, and raises the economic threshold for installing high usage trunk groups. The traffic previously handled on small high usage trunk groups (less than 10 circuits) becomes transit traffic.

- For transmission engineering reasons, four-wire switching is anyway necessary for long-haul traffic, even with analog switching. As digital switching systems basically operate on a four-wire basis, the additional costs for implementing the transit function are reduced compared with analog technology in which four-wire switching itself means increased expenditure. Instead of separate transit exchanges, transit points can be installed throughout the network on the site of digital exchanges, thereby considerably increasing the options for alternative routing.

Overall, transit traffic in the network will increase somewhat; this is offset by the fact that more traffic can be handled at lower levels of the network [3.13].

An additional refinement of traffic routing results from the increased efficiency of Signaling System No. 7 used in the ISDN. It provides among other things information concerning the originating address of a call and the service indicator and thus enables more effective traffic routing, particularly under overload conditions (see Sect. 6.3). For example, traffic to overloaded destination exchanges can be limited in the originated exchange, or specific services can be prioritized in the event of overload.

However, even with a high degree of integration of services, not all parts of the network will provide the same range of capabilities, because not all the network nodes will provide the facilities required for the less frequently used services (see Sect. 6.2.9). Traffic routing must take this into account. It may come about that for some services fewer high usage trunk group options are available for traffic routing than for telephony.

3.5 Interworking with Other Public Networks

In addition the telephone network, many countries have separate networks for text and data communication (see Sect. 3.2.2).

Special services will still be handled via these networks for some time to come. Even when the ISDN is widely established, several of these networks and large parts of the analog telephone network will continue to exist.

In order to provide general availability of communication partners, it is therefore desirable for ISDN subscribers to be able to interwork with subscribers using corresponding services in existing networks. This presupposes conversion facilities (gateways) between the existing networks and the ISDN (see Sect. 6.2.9).

3.5.1 ISDN and the Analog Telephone Network

Due to the number of subscribers involved and hence also the frequency of traffic relations, interworking between the ISDN and the telephone network is of paramount importance. The ISDN is based on the digital telephone network; it is embedded in this network and uses its network components. The facilities for conversion between the

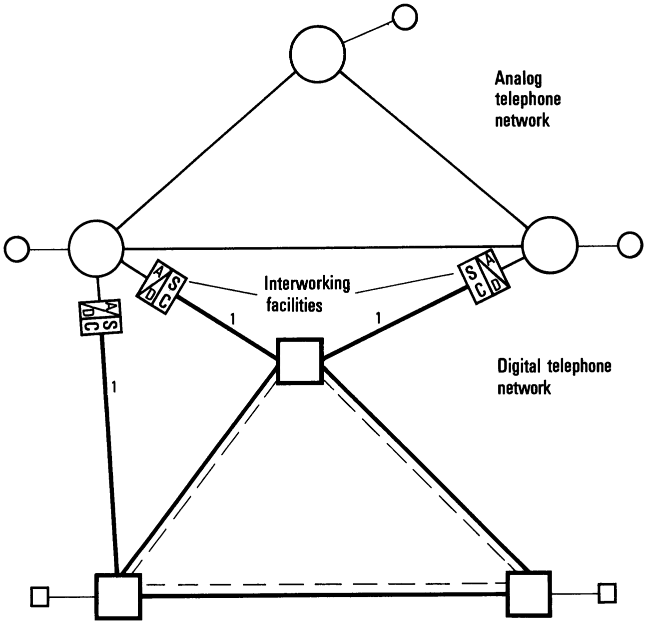


Fig. 3.8. Interworking Between the Analog and Digital Telephone Networks.
[1] Voice circuit signaling, adapted to suit the analog exchanges

Analog	Digital	
○	□	User station
○	□	Exchange
——	——	Trunk circuit

A/D Analog/digital converter, SC Signaling converter, – – – Signaling channel for CCITT Signaling System No. 7

analog and digital telephone networks are also available between the ISDN and the analog telephone network (Fig. 3.8). As the numbering plan of the telephone network is also used for the ISDN there are no fundamental numbering problems, and no access codes are required for crossing the interface between the ISDN and the analog telephone network. It is merely necessary that no analog network equipment be used between two ISDN subscribers who require a transparent connection. The calling subscriber must indicate this prior to call establishment (see Sect. 4.3.3.2).

3.5.2 ISDN and Public Data Networks

In contrast, conversion to the public circuit-switched or packet-switched data networks involves interworking with networks having their own numbering systems, their own signaling methods and protocols, and transmission rates which differ from those of the ISDN.

Interworking between the ISDN and these networks therefore necessitates matching of the signaling protocols and the transmission speeds (e.g. conversion from 64 kbit/s to 2.4 kbit/s text transmission). The type of internetwork gateway is affected among other things by the extent to which existing public data networks and their services are integrated into the ISDN.

The possibilities for interworking with these networks are described in Sect. 4.4.1.

3.6 Interworking with Private Networks

3.6.1 ISDN Private Branch Exchanges

In the business sector, it is common that there are telephones as well as terminals for non-voice communication such as teleprinters, facsimile equipment and data terminal equipment. There is therefore a considerable impetus in this sector to connect these terminals to a common switching system, namely the ISDN private branch exchange (Fig. 3.9). The ISDN private branch exchange has only one line per subscriber which

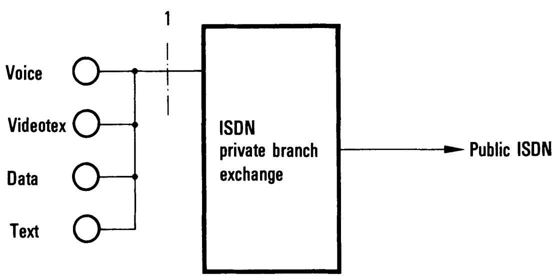


Fig. 3.9. Integration of Services in the ISDN Private Branch Exchange.
[1] Universal Interface (see Sect. 4.2)

is used jointly for all types of information [3.14]. In the forefront of this development are the large private branch exchanges, as use of the ISDN capabilities offers economic advantages even within one PBX network. The private branch exchanges are connected to the ISDN local exchanges via the basic access or primary rate access (see Sect. 4.2.1.2). During a transitional phase, some of these systems will also have direct access to the other public data networks and to the analog telephone network (see Sect. 6.4.1).

3.6.2 Local Area Networks

A particular form of private network has arisen in the shape of *local area networks* (LANs). These serve as a communication medium for rapid data exchange between workstation systems, computers and centrally located high-performance equipment situated within a defined "local" area, e.g. a building or geographical site. Common to all LAN systems is a type of bus, to which all components included in the system are connected and via which all information is transported in packet mode (Fig. 3.10).

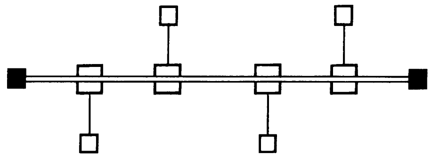


Fig. 3.10. Local Area Network with Bus Structure

Local area networks are characterized by the following features [3.15]:

- The transmission rates are considerably higher than those currently available in private branch exchanges (10 Mbit/s and above).
- The distance between the individual components is restricted to a few kilometers.
- Twisted pair, coaxial and optical fiber cables are used as the transmission medium.
- Local area networks have a distinctive network topology: bus or ring systems without central control.
- Speech transmission creates difficulties because the high bit rates of PCM and the high traffic intensity of bulk telephone traffic place heavy loads on the network, and also because with speech transmission, unlike data transmission, only very short delay times are allowed (< 200 ms).

For the special applications described, local area networks may be used in conjunction with ISDN private branch exchanges.

Connecting a LAN to the ISDN requires an interworking facility ("gateway") in the LAN to perform the necessary signaling adaptation functions and the rate conversion. The ISDN therefore perceives the LAN as a whole as a data terminal equipment with an ISDN S or T interface (see Sect. 4.1.5). This means that the functions of a terminal adapter (see Sect. 4.1.4) are implemented in the gateway (Fig. 3.11).

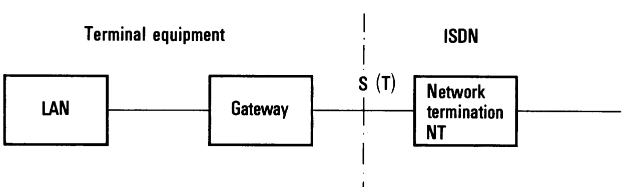


Fig. 3.11. Connection Between Local Area Network and ISDN

3.7 Numbering

The ISDN will adopt the telephone network numbering system. The ISDN subscriber numbers can consist of a maximum of 15 digits including the country code and the national destination code (see Sect. 4.3.3.2). The digits required solely for traffic discrimination (trunk prefix in CCITT parlance), such as 0 for national toll traffic or 00 for international traffic, are not included in the 15 digits. In the case of subscribers with several terminals (see Sect. 4.1.3), the ISDN numbers can be used to address the terminals selectively, as with direct dialing-in to private branch exchanges. In addition to the ISDN number, the ISDN provides the ISDN subaddress. According to recent agreements, this consists of a maximum of 40 digits and is transmitted transparently from the calling to the called subscriber within the call set-up procedure. The ISDN subaddress and ISDN number are separated by a coded character (delimiter). The subaddress enables the subcomponents of the subscriber called under the ISDN number to be addressed more precisely than the scope of the basic ISDN number allows (see Sect. 4.3.3.2). As the ISDN provides several communication services, on call establishment a criterion is required as to which service the subscriber wishes to use. For this purpose the service identifier is used. This is automatically transmitted by the terminals and can be used by the network to ensure that the connection is established over suitable paths, e.g. over a wholly 64 kbit/s connection, and at the called subscriber's end to check whether suitable terminals are available for the desired service (compatibility check) (see Sect. 4.3.3.2).

In the case of the teleservices (see Chap. 2), this check ensures complete compatibility; in the case of the bearer services, it only ensures the presence of a terminal of the appropriate speed category, i.e. not necessarily compatibility of the transmission protocols also.

For connections between ISDN subscribers and subscribers in dedicated networks, the following cases may arise depending on whether interworking units are provided between these networks and the ISDN:

- *No interworking unit between ISDN and dedicated networks*
  If the ISDN subscriber wishes to use the services of other networks, e.g. telex, teletex or data transmission services of text and data networks, he requires, in addition to his ISDN access, a subscriber line to the dedicated network with its associated directory number.
- *Interworking unit provided between ISDN and dedicated networks*
  The ISDN subscriber is connected only to the ISDN and as a rule only has the ISDN number. He can dial subscribers in dedicated networks via the directory numbers of those networks. The ISDN requires a discrimination criterion – e.g. a special prefix – in order to detect that the dialed digits are those of a number in the dedicated network, making it necessary to access a gateway to that network.
  A special case exists if, in interworking with dedicated networks, the ISDN only acts as the feeder port to these networks, the actual service being offered by the dedicated network (port method, see Sect. 4.4.1). In this method the ISDN subscriber is a quasi-subscriber of the specific service network and therefore incorporated into the numbering plan of this network. The ISDN subscriber then has one subscriber line but two directory numbers, namely the ISDN number and a number belonging to the dedicated network. This concept is important for packet-switched services which, due to their fundamentally different switching systems (see Sect. 4.4.4), may not be readily assimilable within the ISDN at the initial stage.

3.8 Implementation Strategies

Telephone networks worldwide are currently being converted from analog to digital transmission and switching – *Integrated Digital Network* – because this confers economic advantages. Since it is desirable for a number of reasons (transmission quality, cost-effectiveness) to avoid several analog/digital conversions within a connection, network digitization is mostly aimed at providing wholly 64 kbit/s connections between originating and destination exchanges at an early date. As a further conversion measure within the trunk network, signaling associated with the traffic channel will be replaced by the more powerful Signaling System No. 7 which employs central signaling channels (see Sect. 6.3). Once the subscriber lines are also converted to digital transmission, all the conditions will be created for extending the digital telephone network into the ISDN.

3.8.1 Overlay Network and Cell Approach

Two basic implementation strategies can be identified for the transition from the existing network to the ISDN: the overlay network and the cell approach [3.16].

The aim of implementation by the *overlay network strategy* is to make the new technology available in a large area even in the initial phase, and with as little new equipment as possible. At suitable points in the network new exchanges will be built which are connected entirely via digital transmission systems and which exchange their switching information via Signaling System No. 7. Interworking with the existing network will be implemented with as few gateways as possible in order to minimize expenditure on conversion equipment, e.g. for analog/digital and signaling conversion. Calls from ISDN subscribers to subscribers in the existing network will be routed as far as possible within the ISDN, while calls in the opposite direction will be transferred to the ISDN at the nearest gateway (Fig. 3.12). As well as replacing complete exchanges, the introduction of new switching systems can also take the form of system expansions. In this case they will primarily be used to cover the demand for ISDN subscriber lines.

The disadvantages of the overlay network are the initially low usage of the overlay links, the limited traffic routing possibilities and problems of operation and maintenance, as a network with a large extent but comparatively low density must be operated and maintained by the operating centers of the existing network.

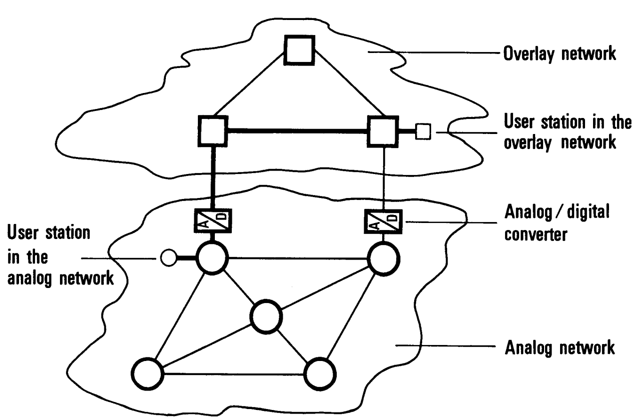


Fig. 3.12. Network Implementation by the Overlay Method.
—— Connection between user stations in the overlay network and in the analog network

The main feature of the *cell approach* is that it concentrates the introduction of the new network components in a defined area, perhaps even providing complete network conversion within that area before new switching and transmission systems are introduced in other areas. This approach is more theoretical than practical in nature, as even within a limited area it is generally unrealistic to replace the entire existing network at a stroke. Furthermore, with this implementation strategy the subscribers of one region would be given preferential treatment over the others.

3.8.2 Pragmatic Implementation Strategy

The overlay network and the cell approach represent two basic implementation strategies which, however, do not take sufficiently into account, or even ignore, a number of practical factors some of which may be specific to particular countries. These include:

- the scale of conversion from analog to digital transmission and switching and the geographical distribution of the existing digital network equipment,
- the extent and location of the demand for new services,
- the condition and age structure of the existing network equipment,
- the space available for installing new network equipment in the buildings,
- the existing infrastructure for operation and maintenance of the network and the qualification level of the personnel involved,
- the financial resources of the network carrier.

As the two implementation concepts are not incompatible, network implementation will mostly be a combination of the overlay network and cell approach, taking into account the factors listed above. To give an example of this approach to network conversion, known as the *pragmatic implementation strategy*, the steps by which the Federal Republic of Germany intends to bring about the transition from the analog to the digital telephone network and to the ISDN are described below [3.2, 3.17].

- Since the mid-seventies, digital transmission systems have been used mainly in the short-haul network, i.e. the trunks connected to primary centers (which correspond to US class 4 centers). By the mid-eighties, an initial phase involving 2 Mbit/s and 34 Mbit/s transmission systems (30 or 480 64-kbit/s channels) was operational at this level. In the long-haul network, transmissions systems operating at 140 Mbit/s and 565 Mbit/s (1920 or 7680 64-kbit/s channels) are also being introduced.
- Since 1985, digital local and long-distance exchanges have been installed, and it is intended that from the end of the eighties only digital switching systems will be acquired. For the *long-distance exchanges*, network introduction from "top to bottom" is planned, i.e. digital switching equipment will first be installed in the largest transit exchanges in the major cities. These exchanges will be given "digital expansion units". Introduction of digital *local switching* is generally intended to be coordinated with long-distance switching, i.e. each time there is a digital expansion in the associated long-distance switching system. It is expected that by 1990 approximately 20 % of the long-distance exchanges will be equipped with digital systems, and that in 1995 approximately 20 % of the lines in the Federal Republic, i.e. approximately 6 million lines, will be connected to digital local exchanges.

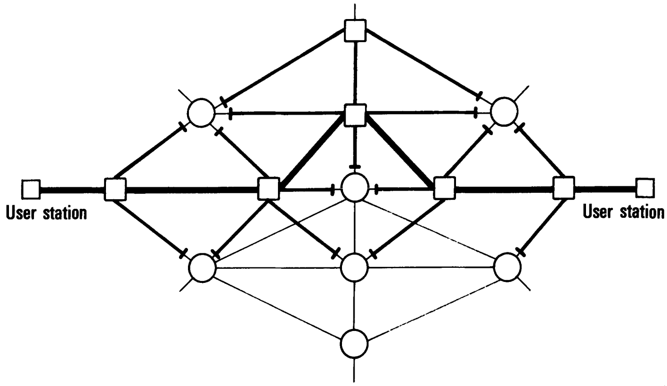


Fig. 3.13. ISDN Connection in a Hybrid Analog/Digital Network.

Analog	Digital	
○	□	Exchange
——	**——**	Trunk circuit
——	ISDN connection	
I	Analog/digital converter	

- Approximately coinciding with the introduction of digital switching systems, i.e. from about the mid-eighties, a start is made on implementing the common-channel Signaling System No. 7.
- After an ISDN pilot project (from 1986) the ISDN, which requires mainly modification to the local exchanges, is to be introduced from 1988 onwards [3.17].

From the above description the following conclusions can be drawn for the ISDN network configuration in the Federal Republic of Germany:

- Even in the introductory phase the ISDN will not form a separate overlay network, i.e. a separate network superimposed on the telephone network which has access to the analog parts of the telephone network via *central* gateways. It will rather be a network embedded in the digital telephone network, in which only digital network equipment will be used for connection set-up between two ISDN subscribers (Fig. 3.13). Unlike an implementation strategy based on the overlay principle, this less rigid approach is expected to provide greater economic advantages, especially during the transitional phase [3.13, 3.18].
- As the ISDN is developing out of the telephone network, its network structure will largely be based on that of the present-day telephone network (Figs. 3.2 and 3.3). The stored program controlled exchanges enable more flexible traffic routing; for example, high usage trunk groups can be installed between local exchanges in adjacent local networks (Fig. 3.14), which was not possible using electromechanical exchange equipment. The greater flexibility of traffic routing also enables the number of hierarchical levels in the long-distance network to be reduced.
- The ISDN will radiate out from the large local networks which have within their area transit exchanges of the upper hierarchical levels. These are largely interconnected via digital transmission paths; in this way a wide-area network reaching

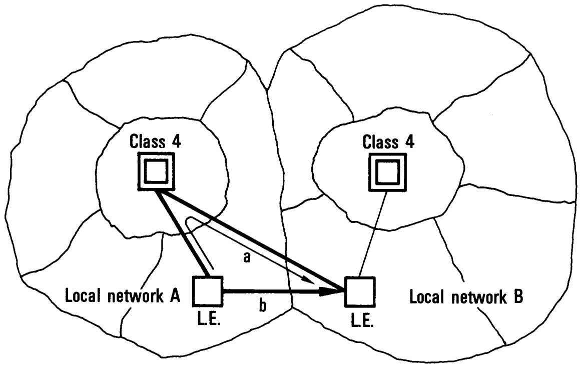


Fig. 3.14. Extended Traffic Routing Options With Stored-Program Controlled Exchanges.
a Shortest possible connection in the analog telephone network,
b Possible connection with stored-program controlled exchanges,
L.E. Local Exchange

the major centers of population can be provided at an early stage. As it is there that the majority of medium and large private branch exchanges are located, the potential users of the ISDN will be able to have an ISDN line without delay.

3.8.3 Satellite Links in the ISDN

One of the basic conditions of the ISDN is an all-digital connection from subscriber to subscriber. For ISDN exchanges located at great distances from each other there may be no all-digital transmission paths available in the terrestrial network in the introductory phase. In these cases, an alternative is connection via satellite. Besides the INTELSAT satellites for intercontinental traffic which have been in operation since the sixties, national satellite projects are operating or being planned in several countries. These provide channels with bit rates of 64 to 2048 kbit/s for transmitting speech, text, data and pictures and are being considered for connecting ISDN exchanges or ISDN private branch exchanges separated by great distances.

The disadvantage of satellite links is the longer signal delay compared with terrestrial connections (approximately 260 ms for the link from earth station to satellite to earth station) [3.19, 3.20]. For text and data connections, in which control instructions and acknowledgements must be transmitted in both directions in the transfer phase, this may considerably prolong the holding time. In telephone connections, the long signal delay makes conversation more difficult (cf. Sect. 7.7.3); as in the analog telephone network, echo suppressors or echo cancelers will be necessary.

Due to residual orbit ellipticity and to the gravitational pull of the moon, signal delay variations occur in satellite connections which cannot be fully compensated by correcting the satellite orbit. These signal delay variations, of the order of a millisecond, must be equalized in buffer memories.

Thus, although satellite links form a flexible supplement to the terrestrial trunk network, their use in digital communication networks requires a number of additional measures to ensure a transmission quality comparable with that of terrestrial links.

3.9 ISDN Implementation Schedules

Virtually all the telecommunication administrations and operating companies in the western industrialized world have defined objectives as to the methods and schedules by which the transition from the analog to the digital telephone network and then to the ISDN is to be completed [3.21]. For the transition to the ISDN, several evolutionary phases are generally discernible:

- Temporary solutions and field trials are as a rule intended to demonstrate the technical and practical feasibility of individual functions, such as the digital subscriber line or of 64 kbit/s switched connections.
- Pilot projects constitute the next phase. The purpose of these is to test the ISDN products operationally and technically before final commercial introduction. Pilot operation is restricted to a few locations, and the number of users is limited.

The countries listed in Table 3.4 are aiming for commercial ISDN operation before the beginning of the nineties.

As far as the channel structure on the subscriber line is concerned, most operating companies have opted for a system using two 64 kbit/s traffic channels and one 16 kbit/s signaling channel ($2 \times B + D$). Different solutions will initially be adopted by the United Kingdom and Japan, as these countries had already taken their decision with regard to ISDN implementation at an early date. The United Kingdom specified a net bit rate of 80 kbit/s subdivided into one 64 kbit/s channel for speech and data transmission, one 8 kbit/s channel for data only and one 8 kbit/s signaling channel [3.22]. In Japan the net bit rate is currently 88 kbit/s, divided into two traffic channels of 64 kbit/s and 16 kbit/s respectively and one 8 kbit/s signaling channel [3.23]. However, by 1988 both countries intend to adopt the channel arrangement of two 64 kbit/s traffic channels and one 16 kbit/s signaling channel which has become the world standard.

Table 3.4. Planned ISDN Implementation Dates

Country	Tempory solution, field trial	Pilot operation	Commercial operation
Australia	1984/85	No details	1988
Belgium	1984/85	1988	1989
Denmark	–	1987/88	1989/90
Federal Republic of Germany	1984	1986/87	1988
Finland	–	1987	1989
France	1986	–	1988
Italy	1984	1987/88	1989/90
Japan	1983/84	–	1984/85
Norway	–	–	1987
Spain	1985	1987	1988
Sweden	1985	1987	1988
Switzerland	–	1987	1989/90
United Kingdom	1983	1984/85	[a]
USA	1985	1986	1987

[a] The pilot project is operated as a normal commercial service.

In the USA, the AT&T has developed a three stage plan [3.24]. The first stage provides for optimization of the existing analog and digital networks, the second stage covers introduction of the ISDN. The third stage is designed to bring about a fully integrated network incorporating circuit switching and packet switching. The whole system is designed for services with virtually any bandwidth ("bandwidth on demand"). In 1986 some of the Bell Operating Companies began to introduce an ISDN based on 64 kbit/s channels. From about 1989 all Bell Operating Companies will provide ISDN services.

The user potential for the ISDN will initially come mainly by business users, most of which have ISDN private branch exchanges or use Centrex services. For this reason, in many countries, when the ISDN is implemented the primay rate access will be provided as well as the ISDN basic access. Different strategies will be pursued by the carriers or operating companies in terms of interworking with existing dedicated networks and of provision of packet-switched services in the ISDN. Common to these strategies is the desire to provide a range of services at least equivalent to those offered by the dedicated networks.

4 Subscriber Access

Compared with the digital telephone network in which analog signals are transmitted to the user equipment, the essential technical innovation of ISDN is the digitization of the subscriber line (see Fig. 3.4). The majority of the internationally agreed ISDN specifications (see list of CCITT Recommendations of the I Series in the Annex to this book) on which the following remarks are based therefore deal with subscriber access.

The main aspects of user access are the configuration of the user station (Sect. 4.1), the user-network interfaces within the user station (Sect. 4.2) and user signaling (Sect. 4.3).

4.1 Configuration of the User Station

4.1.1 Functional Groups of the User Station

Figure 4.1a shows the configuration of a user station as specified in CCITT Recs. I.410 [4.1] and I.411 [4.2]. One or more terminal equipments (TE) are connected to the network termination (NT). The terminal equipments can be of the same type, e.g. a number of telephones, or there may be a combination of different types such as voice and non-voice terminals.

The network termination NT physically connects the terminal equipment to the subscriber line and enables the latter to be used jointly by several terminals. For these two functions the NT is divided into two functional groups, NT1 and NT2, as shown in Fig. 4.1b: NT1 provides the physical connection to the subscriber line, NT2 enables it to be used by several terminals.

The functional group TE can either be a terminal equipment type 1 (TE1) specially designed for the ISDN (Fig. 4.1b) and connected directly to the interface at reference point S, or a terminal equipment type 2 (TE2) with a conventional interface connected via a terminal adaptor (TA).

Reference points between the functional groups have been defined: reference point T between network terminations NT1 and NT2, reference point S between network termination NT2 and the terminal equipments TE, or more precisely TE1 or TA. A standardized physical interface can, but does not always have to be provided at these reference points (cf. Sect. 4.1.3).

Depending on national or network-specific regulations, the responsibility of the network operator ends at reference point S, T or U (cf. Fig. 4.1). If it ends at reference point S, the network operator is responsible for network terminations NT2 and NT1. If it ends at T, he is responsible for network termination NT1 only and if it ends at U, he is responsible for neither NT2 nor NT1. The particular reference point (S, T or U) at

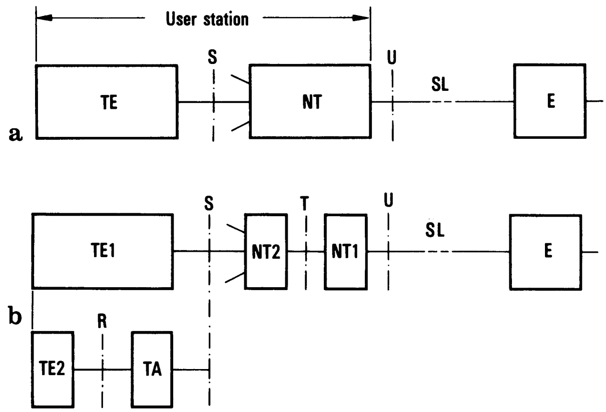


Fig. 4.1a, b. Configuration of the User Station.
SL subscriber line, NT, NT1, NT2 network termination, R, S, T, U reference points, TA terminal adaptor, TE terminal equipment, TE1 terminal equipment with ISDN interface, TE2 terminal equipment with conventional interface, E exchange

which the responsibility of the network operator ends, is both the point at which the network operator supplies a defined service – access to the communication services (cf. Sect. 2) – and the point up to which he accepts responsibility for maintenance. In those cases in which the network operator is responsible for network terminations NT2 or NT1, national or network-specific rules define whether this point is located directly at the NT output or whether the installation as far as the sockets for user terminals (cf. Fig. 4.8) also falls within the network operator's sphere of responsibility.

To provide the user with universal access to the ISDN communication services, the interface at reference points S and T is internationally standardized. For reference point U, on the other hand, only different national specifications have so far been agreed; further national variants and perhaps international standards may emerge (cf. Sect. 4.1.2). The standard for reference points S and T covers not only the mechanical and electrical specifications (Sect. 4.2), but also the specifications for the operating procedure (Sects. 4.2 and 4.3) with the aim of allowing problem-free interconnection of terminal equipments and networks of different origin.

For this purpose the same set of specifications is used for both reference points S and T. If no special NT2 functions (e.g. internal traffic) are required, functional group NT2 can be reduced to a "zero NT2" (cf. Sect. 4.1.3); a terminal equipment designed for reference point S can therefore also operate at reference point T.

Between TE2 and TA lies reference point R at which a conventional interface is normally implemented (cf. Sect. 4.1.4), e.g. in accordance with the CCITT V.-Series Recommendations (see Annex), CCITT Recs. X.21 [4.3] or X.25 [4.4] or the RS232 interface defined by the EIA (Electronic Industries Association, Washington D.C., USA). In order that an ISDN terminal equipment of type TE1 and a combination of TE2 and TA (see Fig. 4.1b) can be handled uniformly by the network (e.g. in terms of addressing), there is provided a separate terminal adaptor TA for each TE2, at least from the functional viewpoint.

4.1.2 Network Termination NT1

The network termination NT1 converts the signals at reference point T to signals suitable for transmission on the subscriber line and vice versa (see Sect. 7.4.3).

As subscriber lines can differ greatly from country to country and even within a country (e.g. in terms of length, cable characteristics, branches etc.), the transmission method for the subscriber line has not yet been internationally standardized (cf. Sect. 4.1.1); network operators define their own specifications for their areas. Due to the nature of functional group NT1, the other functional groups of the user station (i.e. NT2, TE1, TA, TE2) operate independently of the transmission method on the subscriber line.

The network termination NT1 has another important role in fault location. If malfunctions occur, it is necessary to establish, from the switching equipment, the location of the fault in order to enable maintenance personnel to be efficiently deployed. The following test loops (Fig. 4.2, cf. Sect. 3.3), some of which may be standardized by the CCITT and which can be controlled if required from the switching equipment [4.5], may be used for this purpose:

- With test loop c, a switching equipment self-test is performed before each connection set-up
- Test loop b indicates whether the subscriber line is properly terminated (by an NT1) at the user end so that the switching equipment can detect any fault present on the subscriber line.
- With test loop a, the switching equipment can ascertain whether the signals are being correctly transmitted in both directions on the subscriber line; under control of the switching equipment, the NT1 loops some or all of the information sent by the switching equipment back to the switching equipment which then determines whether the bit error rate is within the permitted tolerance range. Depending on how much information is looped back (e.g. all channels or only one channel, see Sect. 4.2.1.1) with the test loop activated, the line is either not available at all or only available to a limited extent to the user.

In addition to this test method which, though effective, impairs operation, the network or the user can if necessary use in-service test methods; for example, suitable data protection information can be added to the transmitted information, enabling the user to ascertain the bit error rate.

Also under discussion are tests which can be initiated by the terminal equipments, e.g. a facility for requesting the network to test the line, or test loops in which information sent by a terminal equipment is looped back to it by the local NT or by the distant NT.

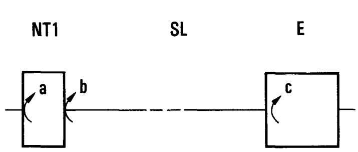


Fig. 4.2. Test Loops for Fault Location.
a, b, c Test loops
SL Subscriber line
NT1 Network termination
E Exchange

4.1.3 Network Termination NT2

The main function of network termination NT2 is to allow joint use of a network access by more than one terminal equipment (cf. Fig. 4.1).

The number of terminal equipments, their spatial arrangement and also, therefore, the implementation of the NT2 may vary considerably (Fig. 4.3).

Figure 4.3a shows the user station referred to in Sect. 4.1.1 with one terminal equipment and a "zero NT2". The "passive bus" (see Sect. 4.2.2.1) shown in Fig. 4.3b is another example of a "zero NT2"; up to eight terminal equipments can be connected in this way. This configuration is only possible with the basic access (see Sect. 4.2.1.2).

However, the NT2 can also be very powerful equipment, such as a private branch exchange which concentrates the traffic of many terminal equipments (Fig. 4.3c) and provides the terminal equipments with additional facilities (e.g. internal traffic). An NT2 of this type can itself provide the abovementioned "passive bus" (Fig. 4.3d).

The connection between NT2 and NT1 may be implemented in various ways (Fig. 4.4a to d).

The normal configuration used for many applications (especially where the user traffic requirement is low) consists of an NT2 connected to a single NT1 (Fig. 4.4a).

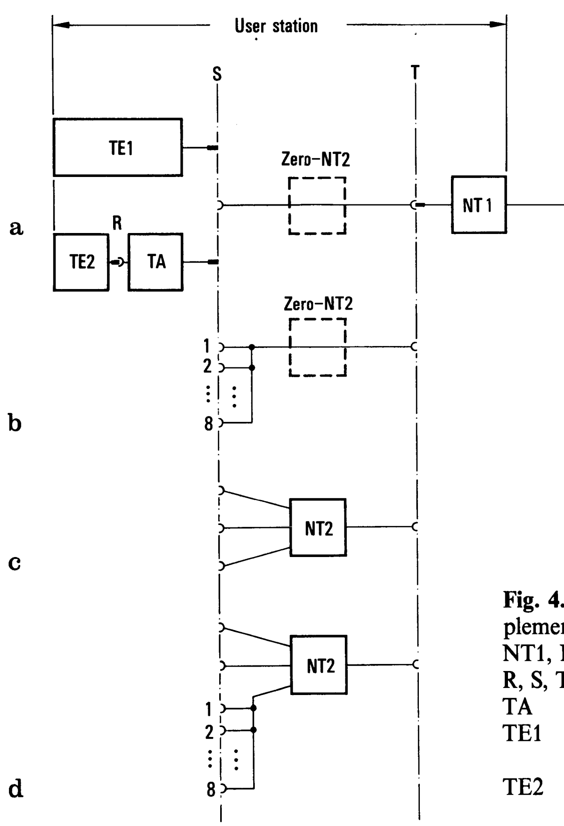


Fig. 4.3a–d. Examples of User Station Implementation.

NT1, NT2	Network terminations
R, S, T	Reference points
TA	Terminal adaptor
TE1	Terminal equipment with ISDN interface
TE2	Terminal equipment with conventional interface

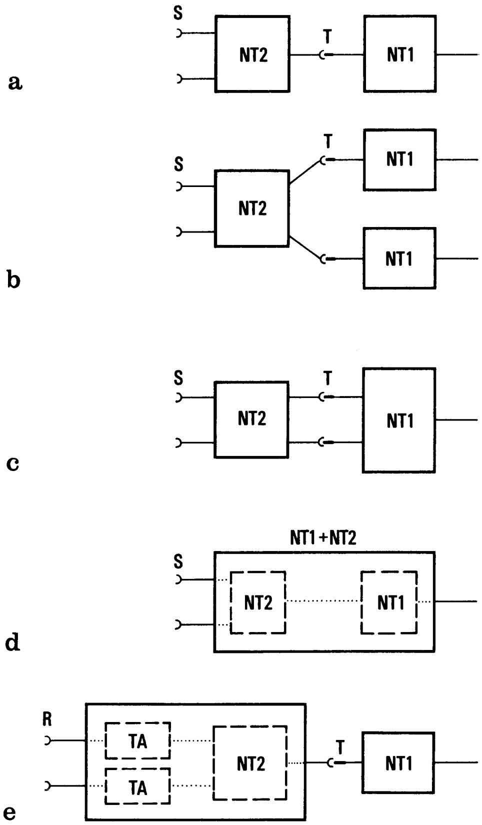


Fig. 4.4a–e. Examples of NT1, NT2, and TA Implementation.
NT1, NT2 Network terminations
R, S, T Reference points
TA Terminal adaptor

To meet more exacting requirements in terms of traffic and availability, an NT2 can have multiple connections to NT1 units (Fig. 4.4b and c).

For economic reasons it is also possible to combine various functional groups to form a single physical entity (Figs. 4.4d and e).

4.1.4 Terminal Adaptor TA

From the economic and organizational standpoint, the possibility of connecting conventional terminals to the ISDN via a terminal adaptor TA facilitates access to the ISDN for the user. The latter should be able to reach his communication partners with both conventional and new terminal equipments, even if his partners are still connected to dedicated service networks (cf. Chap. 2).

The terminal adaptors are normally connected to the interface at reference point S (Fig. 4.3); one or more terminal adaptors can also be combined with the NT2 to form a single entity (Fig. 4.4e) which is then connected to the interface at reference point T.

In drafting the specifications for reference point S, care was taken to ensure that the main CCITT interfaces, e.g. the interfaces defined in CCITT Recommendations X.21 [4.3], X.25 [4.4] and in the V.-Series (see Annex), could be adapted without difficulty. The specifications for the terminal adaptors are given in CCITT Rec. I.461

[4.6] for X.21, in I.462 [4.7] for X.25 and in I.463 [4.8] for the V.-Series interfaces. The interface corresponding to the access point for an analog subscriber line in the conventional telephone network ("t/r i.e. tip and ring interface") was also taken into account. The extent to which other interfaces, too, can be adapted must be checked for each individual case.

4.1.5 Connecting Private Networks

The possibilities for connecting private networks to the ISDN can be illustrated by examining the connection options for private branch exchanges (Fig. 4.5).

There are three alternative methods of connecting private branch exchanges (Fig. 4.5): connection at reference point S, connection at reference point T and direct connection to the subscriber line at reference point U. With connection at reference point S (Fig. 4.5a) the private branch exchange is perceived by the network to be connected like a terminal equipment; in this case NT1 + NT2 does not need to provide any NT2 functions (cf. zero NT2 in Sect. 4.1.3). With connection at reference point T (Fig. 4.5b) the private branch exchange acts as an NT2; with direct connection to the subscriber line (Fig. 4.5c) it acts like an NT1 and NT2.

As the same interface specifications apply to reference points S and T (cf. Sect. 4.1.1), the first two alternatives are technically identical. They may, however, differ as regards the user facilities provided by the network. An example of this is direct dialing-in using the ISDN number (see Sect. 4.3.3.2): as CCITT Rec. I.330 [4.9] currently stands, the network address range corresponding to the ISDN number only extends as far as reference point S and is not designed for sub-addressing within terminal equipments (for this purpose there is the *ISDN subaddress*; see Sect. 4.3.3.2). A private branch exchange featuring direct dialing-in (using the ISDN number) must therefore be connected as an NT2 unit at reference point T.

Direct connection to the subscriber line can give cost benefits as compared with the first two alternatives, especially if a standard interface between NT1 and NT2 is

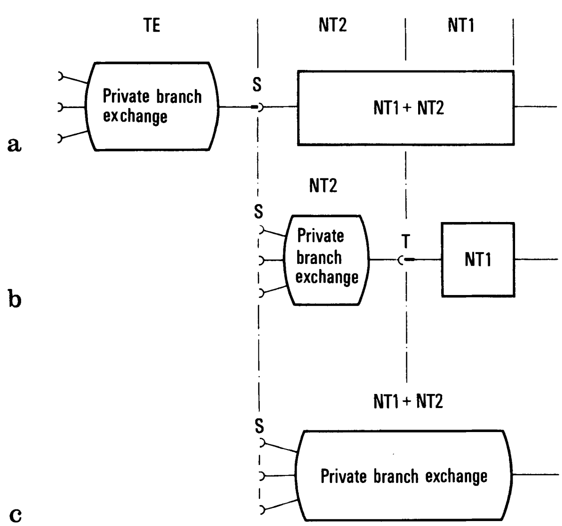


Fig. 4.5a–c. Connection of Private Branch Exchanges.
NT1, NT2 Network terminations
S, T Reference points
TE Terminal equipment

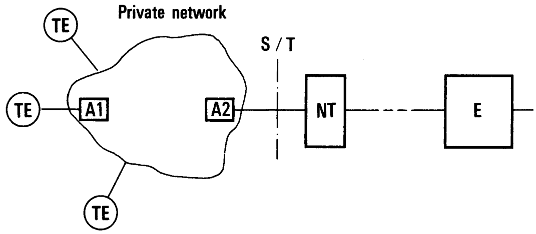


Fig. 4.6. Connection of Private Networks.

A1 Adaptor for connection of ISDN terminal equipments
A2 Adaptor for connecting the private network to the ISDN
TE Terminal equipment
NT Network termination
S, T Reference points
E Exchange

not used; however, the private branch exchange is then subject to the interface conditions (e.g. transmission methods) of the subscriber line, which have yet to be internationally standardized.

In the user domain it is possible to install a complex private network, e.g. a network which can consist of a combination of interlinked private branch exchanges, of local area networks (LANs) meeting ISO Draft International Standard 8802 [4.10] and communication components of a private computer network (Fig. 4.6). A network of this type can provide ISDN and other communication services (cf. Sect. 3.6.2), and terminal equipments with ISDN interfaces can be attached as well as terminal equipments with other interfaces. Such complex networks can be connected to the public ISDN in the same way as private branch exchanges (cf. Fig. 4.5).

A private network not implemented in accordance with the ISDN concept can in some circumstances be endowed with ISDN capability by means of suitable terminal adapters within the private network: terminal adaptor A1 (Fig. 4.6) can enable ISDN terminal equipments to be connected, terminal adaptor A2 (Fig. 4.6) the connection of the private network to the (public) ISDN.

The ISDN addressing mechanisms (see Sect. 4.3.3.2) are sufficient to permit individual addressing of the terminal equipments connected to the private network, and also to interlink several local private networks with he aid of the (public) ISDN to form one large private network.

4.2 User-Network Interfaces

4.2.1 Preliminary Remarks

The specifications relating to the user-network interface have a special significance in communication networks: nationally and internationally, they define the demarcations between terminal equipments and network components.

The user-network interface specified for the ISDN is based on the concept of the ISDN as a universal communication network intended to enable any person worldwide to communicate by any method – using voice, text, data and picture simultaneously –, a network to which it must be possible to connect both specialized and multifunction terminal equipments.

All communication over the ISDN is based on digital links, each with a certain transmission capacity. A user's transmission capacity requirement depends on the type and number of communication services used at the same time, and may vary considerably from user to user; for example, one user may wish to connect an equipment serving as a combined high-quality video telephone set and videotex terminal, another may require a conference facility, a third may need several terminal equipments such as telephones and data terminals and a fourth a large private branch exchange with a high traffic handling capacity.

In order to cover such a variety of requirements without an excessively large number of interface variants, the number of access types defined has been kept to a minimum (see Sect. 4.2.1.2) with transmission capacity levels spread as widely as possible. For the two access types currently specified, the basic access and the primary rate access, the net transmission capacity differs by at least a factor of 10.

In order to adapt the transmission capacity provided by the network to suit actual user requirements, access arrangements of the same or different types can be "connected in parallel"; in addition, the network may not make the full transmission capacity available at the interface (see Sect. 4.2.1.3).

The characteristics of the user-network interface are defined in detail for the basic access and primary rate access. These include specifications for the functional, electrical and mechanical characteristics (see Sects. 4.2.2 and 4.2.3) and for the operational procedures (see Sect. 4.3). The basic specifications described below for the user-network interfaces – channel types, access codes and interface structures – are based on CCITT Rec. I.412 [4.11].

4.2.1.1 Channel Types

The net transmission capacity available at the user-network interface is subdivided in a specific manner – depending on the type of access (see Sect. 4.2.1.2) – into one or more *traffic channels* (e.g. B channels) and normally an additional *signaling channel* (D channel). In special cases (see Table 4.3) the interface structure does not contain a signaling channel, or the active signaling channel can be routed via a different access.

The traffic channels are used by the network to provide circuit-switched and normally also packet-switched communication modes; the D channel is used for communication between user terminal and network, in other words for signaling (cf. Sect. 4.3).

Insofar as the signaling channel has spare capacity in addition to that required for signaling, it can be used for transmitting packet-mode data (also for teleaction purposes) in accordance with the relevant national or network specifications. Signaling and packet-mode data are interleaved for transmission on the same channel, signaling having priority.

Types of Traffic Channel

The traffic channels currently defined are the three types with different transmission capacities listed in Table 4.1.

Table 4.1. Types of Traffic Channel

Channel designation	Bit rate kbit/s
B	64
H0	384
H11	1536
H12	1920

The most important channel type is the 64 kbit/s channel: the B channel, also known as the basic channel. Its bit rate is based on the 8-bit PCM encoding (octet structure) of the telephony signal (see Sect. 7.2.1). The 384 kbit/s H0 channel and two variants of the H1 channel operating at 1920 kbit/s (H12) and 1536 kbit/s (H11) are defined as higher-capacity types of the traffic channel.

In order to be able to provide the basic octet structure (see Sect. 7.2.1) for the PCM-encoded telephony signal in all these channel types without using complicated ancillary equipment, the sender may structure the information in 8 kHz units for all channel types. An information structure of this kind is retained all the way through the network to the receiver. One 8 kHz unit consists of as many bits as can be transmitted within 125 μs (corresponding to 8 kHz): i.e. 8 bits (octet) for 64 kbit/s channels, 48 bits for 384 kbit/s channels, etc.

Broadband channel types with bit rates up to about 135 Mbit/s allowing integration of communication services with even higher transmission capacity requirements (e.g. high-quality full-motion video communication) are under discussion.

Also under consideration are channels with a lower capacity than the B channel, so-called "sub-rate channels" of 8, 16 and 32 kbit/s. Using such channels, the network could make more connections to different destinations simultaneously on one access than on a B-channel basis; however, these connections have less information-carrying capacity than 64 kbit/s connections, and the switching equipment becomes more costly. Another problem with sub-rate channels is that they may lead to a proliferation of classes.

Types of Signaling Channel

Two types of signaling channel have been defined: the D channel normally employed and the E channel (Table 4.2). Depending on the type of access (see Sect. 4.2.1.2), the D channel has a bit rate of 16 or 64 kbit/s; the E channel only operates at 64 kbit/s.

The basic difference between the D channel and the E channel is the link layer protocol employed (layer 2 protocol, see Sect. 4.3.4): in the case of the D channel, the ISDN standard "D-channel protocol" (see Sect. 4.3.4) is used; for the E-channel, a special protocol derived from the Message Transfer Part of Signaling System No. 7 (CCITT Rec. Q.710 [4.12]) would be employed.

Some network operators had announced that they would like to use the E channel in the ISDN introduction phase, so as to allow private branch exchanges primary rate access to the ISDN at an early stage, it being assumed that on the network side the protocols required for the E channel will be easily derivable from the Signaling System

Table 4.2. Types of Signaling Channels

Channel designation	Link layer protocol	Bit rate kbit/s
D	D-channel protocol (LAPD) as specified in CCITT Recs. I.440 [4.27] and I. 441 [4.28]	16 or 64
E[a]	Derived from Signaling System No. 7 in accordance with CCITT Rec. Q.710 [4.12]	64

[a] This option, provided for in the CCITT Red Book, will very likely be deleted in the CCITT Blue Book (1988).

No. 7 used within the network. On the user side, the special protocol required for the E channel generally would have to be implemented separately, in some circumstances even in addition to the D channel protocol normally employed. The option of the E channel will probably be deleted in the new versions of the pertinent recommendations in the CCITT Blue Block.

4.2.1.2 Access Types and Interface Structures

Access Types

Two access types are currently defined for reference points S and T: the *basic access* and the *primary rate access.*

With the basic access, a signal with a total bit rate of 192 kbit/s is used in both directions; the net bit rate then available for the traffic channels and the signaling channel is 144 kbit/s (cf. Sect. 4.2.2.4 and Fig. 4.10).

With the primary rate access, a signal with a total bit rate of 2048 or 1544 kbit/s is used in both directions; the net bit rate is 1984 kbit/s or 1536 kbit/s (see Sect. 4.2.3 and Fig. 4.14).

Subdivision of the net bit rate into channels produces defined *interface structures* at the interface.

Interface Structures

Table 4.3 shows the interface structures specified for the two above-mentioned access types. They are identical for both transmission directions; consequently all channels can be used simultaneously in both directions.

In the case of the basic access, there is only one interface structure with two B channels (64 kbit/s each) and one D channel (16 kbit/s) for signaling.

Several interface structures are defined for the primary rate access, the B-channel –, H0-channel and H1-channel interface structures and the "mixed" interface structure in which the net bit rate can be subdivided into any combination of B and H0 channels (cf. Sect. 4.2.3). The bit rate of the signaling channel (normally the D channel is used) is always 64 kbit/s. It is only with the B-channel interface structures of the primary rate access that the E channel was intended to be used as the signaling channel (cf. Sect. 4.2.1.1).

Table 4.3. Interface Structures at Reference Points S and T

	Basic access	Primary rate access	
Net bit rate	144 kbit/s	1984 kbit/s	1536 kbit/s
B-channel structures	$B+B+D_{16}$	$30 \cdot B+D_{64}$[a]	$23 \cdot B+D_{64}$[a]
	–	$30 \cdot B$[b]	$24 \cdot B$[b]
H0-channel structures	–	$5 \cdot H0+D_{64}$	$3 \cdot H0+D_{64}$
	–	$5 \cdot H0$[b]	$4 \cdot H0$[b]
H1-channel structures	–	$H12+D_{64}$	–
	–	$H12$[b]	$H11$[b]
Mixed structures	–	$n \cdot B+m \cdot H0+D_{64}$	$n \cdot B+m \cdot H0+D_{64}$
	–	$n \cdot B+m \cdot H0$[b]	$n \cdot B+m \cdot H0$[b]

D_{16} 16 kbit/s D-channel
D_{64} 64 kbit/s D-channel
[a] The E-channel has been stipulated as an option for these cases (cf. Sect. 4.2.1.1 and Note to Table 4.2).
[b] The associated signaling channel, if required, is routed via a different access.

If appropriate specification exists nationally or in respect of individual networks, the signaling for a primary rate access may be routed via the 64 kbit/s signaling channel of another primary rate access. (With the H1-channel interface structure the 16 kbit/s signaling channel of a basic access can also be used in exceptional cases.) Assignments are made when the access arrangements are set up and can be changed if required. The signaling channel capacity which is released is available for traffic channels in the 1544 kbit/s variant, normally remaining reserved in the 2048 kbit/s variant.

Figure 4.7 illustrates a possible configuration for a variety of hybrid access arrangements. A hybrid access arrangement consists of a digital interface structure used in conjunction with an analog channel. At reference points S and T, the basic interface structure may be used. In addition to the analog channel, the hybrid access

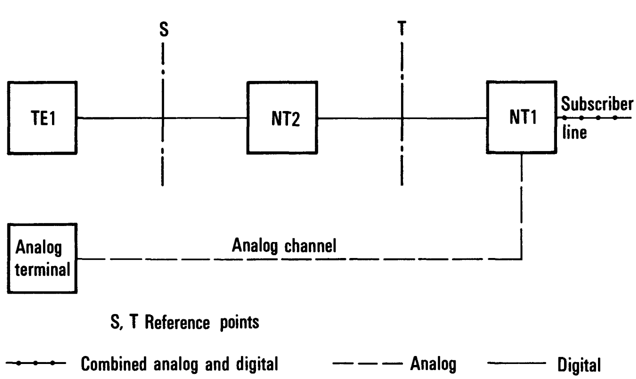


Fig. 4.7. Reference Configuration for Hybrid Access Arrangements for ISDN User-Network Interface

arrangement includes one of the following digital access capabilities (cf. CCITT Rec. I.412 [4.11]):

(i) D
(ii) B + D or
(iii) 2B + D

4.2.1.3 Operation of Traffic Channels

With certain access arrangements or generally with a specific access type, the network may operate some traffic channels either not at all or only with restricted transmission capacity. These channels are nevertheless incorporated with their full bit rate in the interface structure (cf. Sect. 4.2.1.2) at the user-network interface.

If it is economically desirable, the network operator may thus offer the user the available transmission capacity in a more graduated manner (cf. Sect. 4.2.1), and may tailor the equipment associated with the access to suit the user's actual requirements. For example, in the case of the primary rate access with B-channel interface structure, the network may only operate some of the many B channels contained in the interface structure.

During the introduction phase, in some countries one of the two B channels of the basis access will only transmit over the subscriber line at 8 or 16 kbit/s and not at the full bit rate (64 kbit/s) (see Sect. 3.9). Naturally, the network can only assign correspondingly restricted network connections to such a channel. In the case of interworking with a 64 kbit/s channel suitable rate adaption methods have to be used (standardized rate adaption methods are described in CCITT Recs. I.460 [4.13], I.461 [4.6], I.463 [4.8] and I.464 [4.14] (cf. Sect. 4.4)). After an introduction phase all network operators (as far as is currently known) intend to operate the basic access at full capacity.

4.2.2 User-Network Interface for the Basic Access

The user-network interface is the means by which the user terminals gain access to the channels. The timing for information exchange is supplied by the network (see Sect. 7.6.2). The electrical characteristics (see Sects. 4.2.2.2 and 4.2.2.6) and the procedural characteristics (see Sects. 4.2.2.3 to 4.2.2.5) are designed for specific, defined model configurations (see Sect. 4.2.2.1). This does not preclude other configurations yet to be standardized. The standardized specifications for layer 1 (cf. Sect. 4.3.1) of the basic user-network interface are detailed in CCITT Rec. I.430 [4.15].

4.2.2.1 Reference Configurations

The definition of the electrical and procedural characteristics is based on three reference configurations (Fig. 4.8): the point-to-point configuration, the passive bus and the extended passive bus. None of these configurations require NT2 functions (zero NT2, cf. Fig. 4.3a and b). However, the two passive bus configurations require the network to be capable of controlling several (e.g. up to eight) terminal equipments on one access.

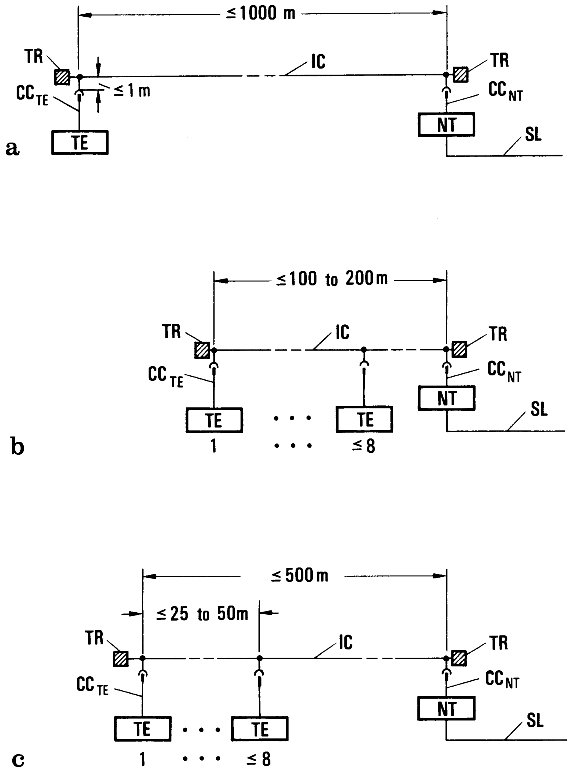


Fig. 4.8a–c. Reference Configuration for Specifying the Characteristics for the Basic Access. **a** Point-to-point configuration; **b** (short) passive bus; **c** extended passive bus.

SL	Subscriber line
TR	Terminating resistor
CC_{NT}	Connecting cord for network termination (≦3 m)
CC_{TE}	Connecting cord for terminal equipments (≦10 m)
NT	Network termination
TE	Terminal equipment
IC	Interchange circuit

Each terminal equipment TE (i.e. TE1 or TA+TE2, cf. Fig. 4.1b and Sect. 4.1.1) is attached via a connecting cord CC_{TE} up to 10 m long and an internationally standardized eight-pin plug (ISO 8877) to a socket at the interchange cable IC (Fig. 4.8). Any stub connecting the socket to the interchange cable must not exceed one meter in length. The network termination NT is connected to the cable either permanently or via the same type of plug as the terminal equipments. As the electrical characteristics of the connecting cord may have a considerable effect on transmission performance, especially in the passive bus arrangement, these characteristics have recently been standardized by CCITT. An extension cord up to 25 m long is allowed for cord CC_{TE} in the point-to-point configuration, provided this does not increase the overall attenuation to over 6 dB.

Normally two wire pairs are used at the interface; these can be used for transmission (one wire pair for each direction) and optionally for power-feeding of terminal equipments from the NT (see Sect. 4.2.2.6 and Fig. 4.13). The advantages of two wire pairs include cost, operational reliability, the range of the passive bus and ease of implementation.

In special cases (depending on the network operator's specifications) the NT can power the terminal equipments via an additional third wire pair (see Sect. 4.2.2.6 and Fig. 4.13). The eight-pin plug arrangement allows yet another wire pair to be used; the latter is not required by ISDN standards, but could be used e.g. for mutual feeding of two directly interconnected terminal equipments.

The interchange cable IC is subject to no special requirements; it will normally consist of two unshielded, balanced, twisted pairs. For the transition from an analog line to the ISDN, normally the subscriber line (cf. Sect. 7.4.3) and frequently also the wiring on the user premises can continue to be used. The two wire pairs can be combined with other wire pairs in one cable. The complete cable run including sockets is purely "passive" in all three configurations, i.e. it contains no amplifying, storing or processing functions. To prevent reflections, a terminating resistor is required at each end of the cable (Fig. 4.8); the terminating resistor at the NT end can be incorporated in the NT.

In all the configurations, the range is limited by the signal delay and the attenuation. Both these variables depend on the connecting cable used: Fig. 4.8 shows possible ranges using typical cabling by way of illustration.

In the point-to-point configuration (Fig. 4.8) the attenuation – measured at 96 kHz – must be less than 6 dB. Specific rules apply to the other two configurations (see CCITT Rec I.430 [4.15]).

In order to enable the NT to loop back the D-channel information transmitted by the terminal equipments with correct timing in the D-echo channel (cf. Fig. 4.10 and Sect. 4.2.2.3), the round trip delay NT-TE-NT must be less than 42 μs (approximately eight times the bit period). This assumes that the NT needs a maximum of 10.4 μs (twice a bit period) for this loopback process. This condition is normally uncritical for all configurations.

In the passive bus arrangement (Fig. 4.8b), terminal equipments can be connected at any point – e.g. one nearby the NT, another at the very end of the passive bus – and can transmit simultaneously. The signal delays of the individual terminal equipments vary with distance. In order to ensure that the signals transmitted by several terminal equipments in the same frame can still be interpreted by the NT, it is necessary to limit among other things (cf. Sects. 4.2.2.2 to 4.4.4.4) the permitted round trip delay and hence also the range of the passive bus. The round trip delay NT-TE-NT – disregarding the frame offset between TE input and TE output (10.4 μs) – must therefore be less than 3.6 μs (approximately 70 % of a bit period) for all terminal equipments. If in the passive bus arrangement the phase readjustment required for the other two configurations is not deactivated in the NT, this figure is reduced to 2.1 μs (approximately 40 % of a bit period); the possibility of permitting up to 2.6 μs (approximately 50 % of a bit period) in future is under discussion by CCITT.

With the extended passive bus (Fig. 4.8c), the reception conditions for the NT are less favorable than with the passive bus (Fig. 4.8b). Consequently, the NT-TE-NT round trip delays for all the terminal equipments must vary by no more than 1.4 μs (approximately 25 % of the bit period); in some circumstances the number of connectable terminal equipments must also be limited (to less than eight). The possibility of limiting the number of connectable terminal equipments to four and allowing up to 2.0 μs (approximately 38 % of a bit period) for the NT-TE-NT round trip delay is the subject of CCITT discussions.

Other configurations can be implemented with the aid of appropriate NT functions, e.g. a star configuration or an "active bus" (in which case the sockets for terminal equipments contain storing or processing components).

4.2.2.2 Electrical Characteristics for Information Transmission

All the component signals, i.e. the D-channel signals, B-channel signals, control signals, etc. (cf. Sect. 4.2.2.4) are combined by time-division multiplexing; a multiplexed signal is thus transmitted in each direction on a wire pair (cf. Sect. 4.2.2.1). Coupling is effected via transformers or their equivalents.

In the case of the passive bus, it is possible that some or all of the terminal equipments (on the same wire pair) may be transmitting at the same time. Normally this only applies to framing signals (see Sect. 4.2.2.4) and D-channel signals (see Sect. 4.2.2.3). The procedure for access of terminal equipments to the D channel (see Sect. 4.2.2.3) requires that the NT always receives a "zero" bit when at least one terminal equipment is transmitting a "zero" in the D channel, and that the NT only receives a "one" when all the terminal equipments are transmitting a "one" or nothing.

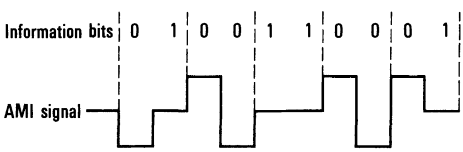


Fig. 4.9. Transmission Code for the Basic Access: AMI Code with 100% Pulse Width.
One-bit: no pulse (space)
Zero-bit: pulse (successive pulses have opposite polarity)

This requirement is met by the following specifications:

- The terminal equipments of a bus configuration transmit bit-synchronously, as all the terminal equipments derive their transmit timing from the same signal received from the NT; the NT for its part derives its timing from the network (see Sect. 4.2.2.1 for delay limitations).
- The transmission code used is AMI [4.16] with 100 % pulse width. Unlike the conventional AMI code, a zero is transmitted as a mark (pulse) and a one as a space (no pulse) (Fig. 4.9). This variation was specified to take account of D-channel access by the terminal equipments (see Sect. 4.2.2.3) and of the link access procedure (see Sect. 4.3.4.2) (fill characters between information frames are sequences of ones).
- Suitable specifications for the frame structure (see Sect. 4.2.2.4) ensure that all terminal equipments send zero-bits with the same polarity in the D channel, so that the pulses corresponding to zeros cannot cancel each other out.
- The terminal equipments can transmit by injecting either current or voltage. Even if some or all of the terminal equipments transmit a pulse simultaneously, the voltage at each transmitter output must stay within certain tolerance limits; this is achieved by each transmitter controlling the current or voltage it injects as a function of its output voltage (voltage-limited current or voltage injection). All the transmitter outputs and receiver inputs always present high impedance, even when deactivated; certain limitations apply only during transmission of a pulse.

Table 4.4 shows an overview of the main CCITT-approved electrical characteristics for the basic access (for further details see CCITT Rec. I.430 [4.15]).

Table 4.4. Electrical Characteristics for Information Transmission via the Basic Access (in accordance with CCITT Rec. I.430 [4.15])

Characteristics	Values
Characteristics of interchange circuit	
– Terminating resistor	$100\,\Omega \pm 5\%$
– Maximum attenuation (at 96 kHz) in the point-to-point configuration	6 dB
– Minimum longitudinal conversion loss (at 96 kHz)	43 dB
– Upper limit value for NT-TE-NT round trip delay[a]	42.0 µs
– Maximum NT-TE-NT round trip delay for all terminal equipments in the passive bus arrangement:	
NT without phase control[b]	3.6 µs
NT with phase control[b]	2.6 µs[e]
– Maximum differential round trip delay NT-TE-NT for all terminal equipments in the extended passive bus	2.0 µs[e]
Transmitter characteristics	
– Voltage-limited current or voltage feeding	
– Pulse amplitude at 50 Ω[c]	$750\,\text{mV} \pm 10\% = V_{nom}$
400 Ω	90[e] to 160% V_{nom}
5.6 Ω	$\leqq 20\%$ V_{nom}
– Output impedance	
when transmitting a pulse	$\geqq 20\,\Omega$[e]
during a space	Specified values depend on frequency[d]
– Minimum longitudinal conversion loss	54 dB
– Jitter in NT output signal (peak-to-peak)	$\leqq 0.26$ µs
– Jitter in TE output signal	$\leqq \pm 0.36$ µs
– Phase displacement in the TE between input and output[b]	-0.36 µs to $+0.78$ µs
– Frame offset between TE input and TE output	10.4 µs
Receiver characteristics	
– Input impedance	Specified values depend on frequency[d]
– Minimum longitudinal conversion loss	54 dB
Radiated emissions	Not yet defined
Insulation requirements	Not yet defined

[a] Due to the D channel access procedure (see Sect. 4.2.2.3).
[b] Not taking into account the frame offset of 10.4 µs between TE input and output.
[c] See tolerance mask (in CCITT Rec. I.430 [4.15]).
[d] See CCITT Rec. I.430 [4.15].
[e] These values, not yet contained in the Red Book version, are specified in the revised version of Rec. I.430 (1986).

4.2.2.3 D-Channel Access by Terminal Equipments

To prevent mutual interference between the terminal equipments in bus configurations during simultaneous transmission in the D channel, these equipments must observe a defined access procedure.

In the case of B channels, normally only one terminal equipment transmits at a time. The exchange ensures this by assigning each of the two B channels to a single terminal equipment at any one time using the signaling.

Actions prior to Transmission

A terminal equipment only begins transmitting in the D channel if it has first ascertained by monitoring that the D channel is free in the direction toward the NT. The relevant criterion is at least eight consecutive ones. The link access procedure (see Sect. 4.3.4.2) ensures that this "pause signal" never occurs within a transmitted frame but only between information frames, and that each information frame begins with a zero.

Actions During Transmission

During transmission the terminal equipments checks, by monitoring in the D channel and by comparison, whether the transmitted information is being corrupted by other terminal equipments transmitting at the same time. The electrical specifications (cf. Sect. 4.2.2.2) are such that terminal equipments transmitting a zero win over terminal equipments transmitting a one bit, at the same time. The winning terminal equipment may continue to transmit, while the other terminal equipments must cease transmitting before the next bit. The link access protocol (see Sect. 4.3.4) ensures that the frames from different terminal equipments all have a different content. Thus, after a short time only one terminal equipment will still be sending zero bits. Only this terminal equipment can successfully complete sending its frame while all the other terminal equipments must stand down. As soon as the D channel is free once more (criterion: at least eight consecutive one bits; see above), the terminal equipments that had to yield retry to transmit their information frames.

Priority

The priority for signaling, which is higher than for packet-mode data (cf. Sect. 4.2.1.1), is governed by the number of consecutive "ones" which a terminal equipment must allow to elapse before it can commence transmission; the more ones, the lower the priority class. Terminal equipments which have successfully completed transmission use the same principle to temporarily downgrade their own priority, so as to give way to waiting terminal equipments of the same priority class.

D-echo Channel

In order to save the expense of providing the terminal equipments with a separate receiver for monitoring in the transmit direction, the NT loops the D-channel information originating from the terminal equipments back to these equipments in a separate *D-echo channel* (cf. Fig. 4.10). The terminal equipments therefore only need to evaluate the D-echo channel for monitoring purposes.

4.2.2.4 Frame Structure

All the control signals and user signals in both transmission directions (Fig. 4.10) are time-division-multiplexed to form a 48-bit frame transmitted 4000 times per second. This corresponds to a total bit rate of 192 kbit/s.

Rapid, unambiguous frame alignment is provided by using a double AMI code violation (see Sect. 4.2.2.2 and example in Fig. 4.10); this is detected no later than the

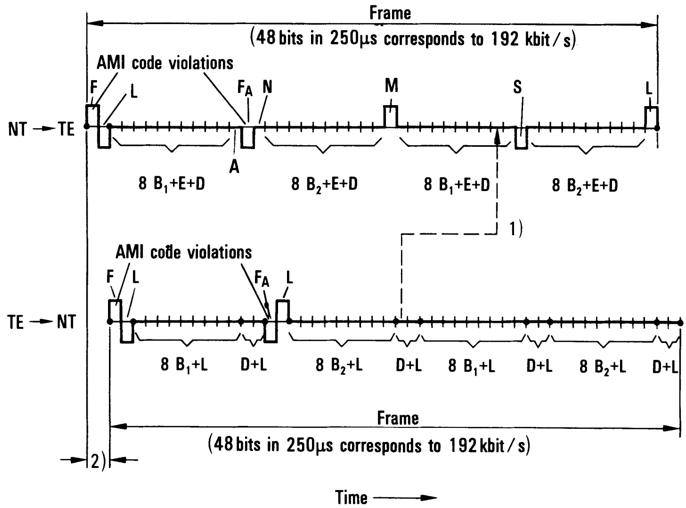


Fig. 4.10. Frame Structure for the Basic Access. To demonstrate essential features, this example has binary ones for all B- and D-channel bits.

●—● Frame section with no DC component
A Bit used for activation protocol
B_1 Bit within B-channel 1
B_2 Bit within B-channel 2
D D-channel bit
E D-echo-channel bit
F Framing bit
F_A Auxiliary framing bit (usually $F_A=0$; for Q bit option see Rec. I.430)
L DC balancing bit

N Inverted value of F_A (therefore usually $N=1$)
NT Network termination
M Multiframing bit (usually $M=0$)
S Bit reserved for future expansions (currently = 0)
TE Terminal equipment
1) Loopback of a D-channel bit in the D-echo-channel
2) The terminal equipments transmit with 2 bits frame offset

14th bit in the frame. In the NT direction, all the terminal equipments simultaneously transmit the signals required for frame alignment (F, F_A and the associated L signals). Frame alignment is ensured even if the wires of a wire pair are reversed. An exception applies to bus configurations where the wires for transmission from TE to NT must not be reversed as otherwise the pulses of simultaneously transmitting terminal equipments would cancel each other out.

With the transformer coupling method used (cf. Sect. 4.2.2.2) dc components would cause interference; therefore, the transmitter balances the dc component of each "frame section" delimited by dots in Fig. 4.10 independently by an L pulse of appropriate polarity.

The first pulse of each frame section is transmitted by the terminal equipments with opposite polarity to that of the framing bit (Fig. 4.10). This ensures both that the pulses for the D-channel signal are transmitted with the same polarity by all the terminal equipments and therefore cannot cancel each other out, and that the AMI violations required for frame alignment (see Sect. 4.2.2.2) occur.

In order to enable the octet structure essential for PCM-encoded voice transmission to be retained, the signals for the two B channels (B1 and B2) are each assigned as octets to eight successive bit positions in the frame, whereas the signals for the D channel and the D-echo channel occupy non-contiguous bits in the frame because of the D-channel access protocol (see Sect. 4.2.2.3 and CCITT Rec. I.430 [4.15]).

The revised version of CCITT Rec. I.430 has the option of using every fifth F_A bit (TE to NT) for an auxiliary "Q" channel – possibly subdivided by means of the "M" bit into four subchannels.

4.2.2.5 Activation and Deactivation

The activation and deactivation protocols enable the network to transfer the network termination and the network-fed terminal equipments to an energy-saving status in periods of inactivity. However, it is necessary for these units at all times to be capable of returning to active status, e.g. in the event of an incoming call.

Activation

The trigger for activation (Fig. 4.11) can originate from the exchange (*Activate* signal) or from any terminal equipment (*Initiate activation* signal). All equipments, such as telephones, which can expect incoming calls must therefore understand the *Activate* signal from the exchange even in the idle phase; consequently, during this phase only those functional units that are not required for detection of this signal may be deactivated in these equipments.

The terminal equipments inform the network that their activation is complete using the *Activation completed* signal; this signal must be transmitted by the terminal equipments not later than 100 ms after receipt of the *activate* signal. The network then acknowledges with an *Activation completed* signal.

Deactivation

Deactivation (Fig. 4.12) is only initiated by the exchange (*Deactivate* signal). All the terminal equipments then assume the deactivated status and confirm this with the

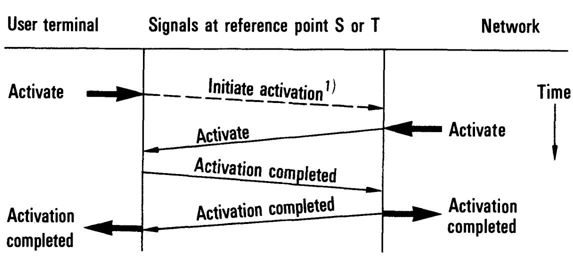


Fig. 4.11. Activation Protocol.
1) Only if activation is initiated by the user terminal

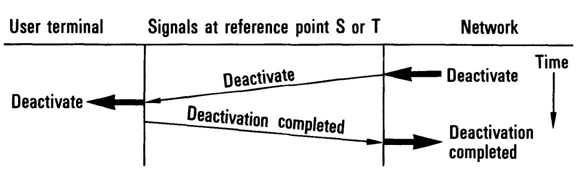


Fig. 4.12.
Deactivation Protocol

signal *Deactivation completed.* The *Deactivate* and *Deactivation completed* signals are coded in such a way that no pulses – not even the frame alignment pulses – are transmitted. Hence in bus configurations the NT does not receive the *Deactivation completed* signal until all the terminal eqipments have been deactivated.

It is left to the network to decide whether and when to deactivate. For example, a network can keep all or specific customer accesses permanently active, or deactive them as soon as all the connections on the access have been released and all signaling activities have ceased.

The NT receives no electrical signal also in cases when the terminal equipments connected to a passive bus are out of service (e.g. unplugged or switched off). It is then up to the network to decide when to deactivate.

4.2.2.6 Electrical Characteristics for Power Feeding

Terminal equipments can be powered from the NT. There are two methods available for this purpose (Fig. 4.13): the method normally used is to supply power in a phantom circuit via the same four wires on which information is transmitted (Fig. 4.13: source 1/sink 1); alternatively two special additional wires can be used (Fig. 4.13: source 2/sink 2). It depends on the type of NT used whether the NT provides power feeding at all, and if so whether method 1 or 2 is used.

In the case of phantom power feeding, the NT informs the terminal equipments via the polarity of the feeding voltage (Fig. 4.13) whether it is providing only the specified emergency power (emergency operation) or a higher normal power (normal operation). The purpose of emergency operation is to enable users, for instance, to telephone even if the public AC mains supply fails. An NT which is powered during normal operation from the AC mains can be remotely fed by the exchange during emergency operation, and then supplies only selected terminal equipments authorized to receive emergency power.

Table 4.5 gives an overview of the electrical characteristics specified for power feeding.

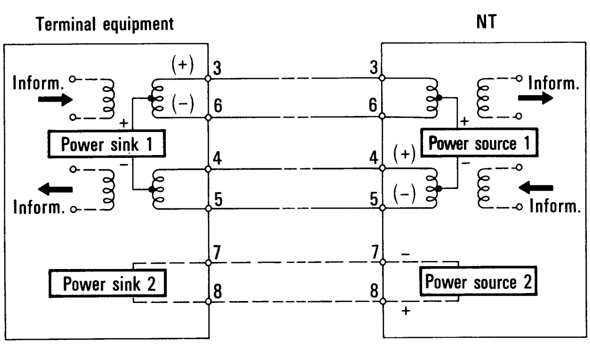


Fig. 4.13. Power Feeding of Terminal Equipments by the NT.
+ – Feeding voltage during normal operation (reversed in emergency operation)
(+) (–) Voltage when transmitting a framing bit pulse (Fig. 4.10)
3...8 Access leads (numbering corresponds to revised version of Rec. I.430 (1986), cf. note "e" to Table 4.4)

Table 4.5. Electrical Characteristics for Power Feeding of Terminal Equipments at the Basic Access

Characteristics	Power feeding methods (cf. Fig. 4.13)		
	Source 1/sink 1		Source 2/ sink 2
	Emergency condition	Normal condition	
Feeding power available at			
NT output	≥ 420 mW	[a]	[a]
TE input	≥ 380 mW[c]	[a]	
Feeding voltage:			
NT output	$-(40\,V+5\%\ldots-15\%)$	$+(40\,V+5\%\ldots-15\%)$[c]	[a]
TE input	$-(40\,V+5\%\ldots-20\%)$	$+(40\,V+5\%\ldots-40\%)$[c]	
Maximum power consumption per TE when deactivated	25mW[b]	100 mW[c]	[a]
Current transient	≤ 5 mA/µs		[a]

[a] Depends on network and NT design.
[b] Up to 100 mW may be specified by the network operator for a transitional period.
[c] Values correspond to revised Rec. I.430 (1986), cf. Note "e" to Table 4.4.

4.2.3 User-Network Interface for the Primary Rate Access

With this interface the user equipment obtains access to the ISDN at reference points S and T on the basis of the interface structures defined for the primary rate access (see Table 4.3). The associated specifications are given in CCITT Rec. I.431 [4.17].

Two interface variants are defined for the primary rate access, one with a total bit rate of 2048 kbit/s, the other with a rate of 1544 kbit/s. Many countries including the European countries use the 2048-kbit/s variant; the 1544-kbit/s variant is used by the USA, Japan and Canada among others. The two different gross bit rates correspond to the digital transmission systems already in use in the trunk networks of these regions (see Sect. 7.2).

In order to provide easier connection of user equipment to both variants in future, the mechanical characteristics (e.g. connectors) still to be finalized and any new features will require standardization in a uniform manner.

The primary rate access is only available in the point-to-point configuration; there is no passive bus (cf. Sect. 4.2.2.1), which would be difficult to implement due to the bit period being reduced to less than one tenth that of the basic access.

The maximum length of the wiring between the network termination NT and the terminal equipment is limited by the maximum permitted attenuation (6 dB at center frequency, see CCITT Rec. G.703 [4.18]); the resultant range for typical circuits is approximately 150 m.

The user equipment is not fed by the NT, so that there is no facility for deactivating the user station and no relevant specification.

As with the basic access, the traffic and control signals are combined in a frame by time-division multiplexing; hence only a single electrical signal is transmitted in each direction.

Electrical Characteristics

The specifications for the electrical characteristics (bit rate, pulse shape, impedance, transmission code) are different for the two interface variants; they have been adopted as they stand from CCITT Rec. G.703 [4.18]. For the 1544-kbit/s variant, CCITT Rec. G.703 [4.18] specifies both AMI and B8ZS as transmission code. The AMI code is simpler but is dependent on the bit sequence: a fairly long sequence of zeros such as may occur in non-voice services can result in synchronization being lost at the receive end, as pulses are only transmitted in response to ones.

In order to be able to handle communication services requiring bit-sequence-independent transmission via the primary rate access in the same way as via the basic access, the standards only allow the B8ZS code to be used on the subscriber line (CCITT Rec. I.431 [4.17]). (Connections routed over trunks via conventional 1.5-Mbit/s transmission systems using AMI code can only be operated on a restricted basis due to the lack of bit-sequence independence within the network, e.g. at 56 kbit/s instead of 64 kbit/s (see Sect. 7.2.3)).

Frame Characteristics

In both variants the specified frame (Fig. 4.14, cf. Sect. 7.2.3) is transmitted 8000 times a second; the frame length is different (256 bits for the 2048-kbit/s variant, 193 bits for the 1544-kbit/s variant). The frame is divided into 31 or 24 8-bit time slots (time channels) preceded by an 8- or 1-bit time slot for control purposes. This gives a net

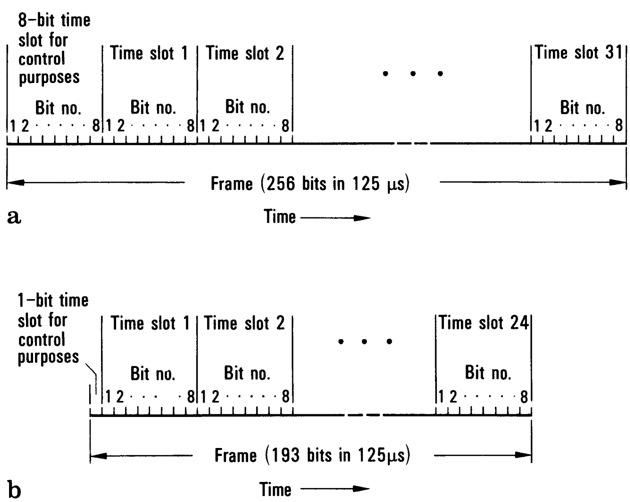


Fig. 4.14a, b. Frame Structure for the Primary Rate Access. **a** Frame structure in both transmission directions for variant with 2048-kbit/s total bit rate; **b** frame structure in both transmission directions for variant with 1544-kbit/s total bit rate

capacity of 1984 or 1536 kbit/s available for the traffic channels and the signaling D-channel.

Part of the transmission capacity of the above-mentioned time slot for control purposes is occupied by the frame alignment signal. This signal is repeated periodically after 2 resp. 24 frames. As any kind of user information can be transmitted in the traffic channels, care must be taken to ensure that the framing does not lock onto an erroneous position; for this purpose, cyclic test data can be derived from the user information and transmitted in the above-mentioned time slots for control purposes (cf. CCITT Rec. G.704 [4.19] and Sect. 7.2.3). A useful side effect of test data of this kind is that it gives the receiver an indication of the bit error ratio on the subscriber line.

Allocation of 8-Bit Time Slots and Channels

In the case of interface structures incorporating a signaling channel (cf. Table 4.3), in both variants this channel always occupies the same, defined 8-bit time slot: in the 2048-kbit/s variant time slot 16, in the 1544-kbit/s variant time slot 24. If an interface structure without a signaling channel is used, the "signaling channel time slot" is normally reserved in the 2048-kbit/s variant while it is used for a traffic channel in the 1544-kbit/s variant.

In B-channel interface structures (see Table 4.3) each B channel occupies an 8-bit time slot; in H1-interface structures, one H1 channel occupies all 8-bit time slots available for traffic channels.

In order to minimize congestion, in the case of mixed interface structures the standards allow completely flexible allocation of 8-bit time slots to traffic channels (see Table 4.3). The individual 8-bit time slots of an H0 channel need not occur in immediate succession.

Nor do the standards specify any particular time-slot assignment for the H0-interface structures; a possible assignment is described in the Annex to CCITT Rec. I.431 [4.17] (cf. also CCITT Rec. G.735 [4.20] and G.737 [4.21] for sound program applications).

4.3 User Signaling

The main purpose of user signaling is to provide mutual understanding between user terminal and network if services and supplementary services are used (cf. Sect. 2.3.3) (*user-to-network signaling*). Unlike in conventional networks, the user signaling also allows a degree of transparent information exchange between two user terminals via the D channel (*user-to-user signaling*; see Sect. 4.3.5.5).

Section 4.3.5 describes user signaling for circuit-switched connections. User signaling for packet-switched connections and also for terminal adaptors in circuit-switched and packet-switched connections is described in Sect. 4.4, together with the adaptation of existing interfaces to the ISDN.

User signaling is standardized in a uniform way for all services, for all access types and for all interface structures (cf. Sect. 4.2.1.2) as well as for reference points S and T (cf. Fig. 4.1b). Where a more precise distinction is not required, the following description therefore refers simply to *user equipment* or *user* (TE1, TA, and also NT2 seen from the network) and *network* (NT1, exchange and also NT2 from the perspective of the terminal equipments).

4.3.1 Protocol Architecture

The protocols for user signaling are structured in accordance with the OSI reference model (OSI = Open Systems Interconnection) (see CCITT Recs. I.320 [4.22], I.420 [4.23], I.421 [4.24] and X.200 [4.25], as well as ISO 7498 [4.26]; cf. Sect. 2). In the OSI reference model, functions are abstracted from the communication processes and assigned to seven hierarchical "layers".

User-network signaling takes place in the lower three protocol layers which, broadly speaking, correspond to the following functions (see Fig. 4.15):

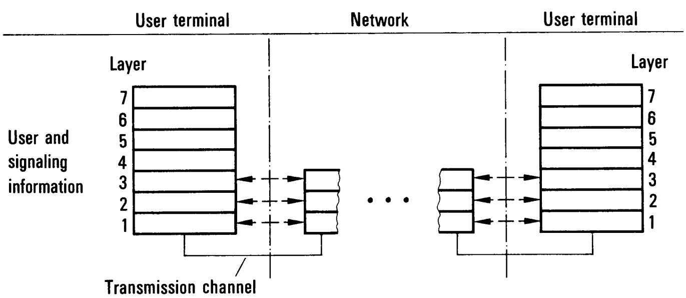


Fig. 4.15. OSI Reference Model for User-Network Signaling and User Information Transfer. Layers 1 to 3: user-network signaling; layer 1: physical layer; layer 2: data link layer; layer 3: network layer

The physical layer (layer 1) provides simultaneous, bidirectional transmission of the information signals in synchronism with the network. In the case of the basic access, the physical layer also enables orderly activation and deactivation (cf. Sect. 4.2.2.5) and regulates simultaneous access by several terminal equipments to the common D-channel (cf. Sect. 4.2.2.3).

The data link layer (layer 2) of the D channel serves the network layer by ensuring protected transmission of signaling information and of any packet-mode data transmitted in the D channel in both directions between network and user. The data link layer also allows selective addressing of individual terminals and of groups of terminals in the broadcast case (see Sects. 4.3.3.3 and 4.3.4).

The network layer (layer 3) of the D channel handles user-network signaling in its narrower sense (see Sect. 4.3.5). With packet-switched connections, the network layer, together with the lower layers, is also involved in the transfer of the user information packets (see Sect. 4.4.4).

The protocols of the "higher" layers (layers 4 to 7), also known as application-oriented protocols, are always interpreted by the network users; they may be interpreted, e.g. in the case of messaging services (see Sect. 2.3.1.3), by the network too.

The structuring (not the protocols) of layers one to three, which mainly concern us here, roughly corresponds to the Systems Network Architecture (SNA) protocol structure on which IBM bases its computer networks. However, as regards the actual protocols, the OSI reference model is preferred because it has been further refined – on the basis of the SNA – into an international standard which is neutral in respect of manufacturer [4.25 and 4.26] and is being updated in line with the ISDN (cf. CCITT Rec. I.320 [4.22]).

4.3.2 Types of Connection

The main function of user-network signaling is call control, i.e. control of call establishment and clearing. Three types of connections are distinguished: circuit-switched connections on traffic channels (B, H0, H1), packet-switched connections on traffic channels, and packet-switched connections on the D-channel. In some circumstances the ISDN may also offer a connectionless packet bearer service e.g. for telemetry purposes; this will not be further discussed here.

Circuit-Switched Connections

In the ISDN, user-network signaling for circuit-switched calls (cf. Sect. 4.3.5) only takes place via the separate signaling channel (outslot). This is in contrast to conventional networks such as the analog telephone network, where signaling takes place in the same channel in which the circuit-switched connection is established (inslot). The appropriately modified OSI reference model is shown in Fig. 4.16 (cf. CCITT Rec. I.320 [4.22]). The specifications for the physical layer are given in CCITT Recs. I.430 [4.15] (see Sect. 4.2.2) and I.431 [4.17] (see Sect. 4.2.3); the link access protocol is described in CCITT Recs. I.440 [4.27] and I.441 [4.28] (see Sect. 4.3.4),

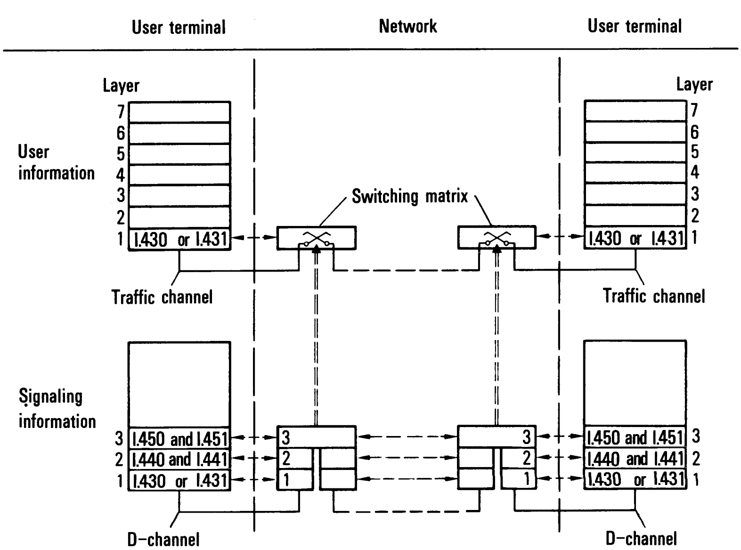


Fig. 4.16. Protocol Architecture for Circuit-Switched Connections

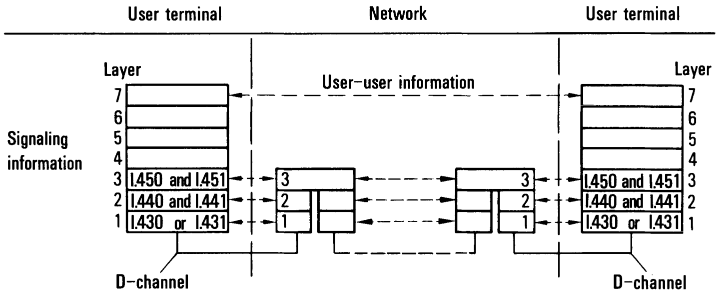


Fig. 4.17. Protocol Architecture for User-User Signaling on the D-channel

the connection-control procedures in CCITT Recs. I.450 [4.29] and I.451 [4.30] (see Sects. 4.3.5 and 4.4).

A circuit-switched connection is only through-connected and handled in the network in the physical layer; the protocols of the other layers are not normally interpreted by the network (Fig. 4.16, upper part).

Outslot signaling has the advantage that signaling (e.g. to indicate a waiting call request) can be accomplished without restriction even if connections already exist, and that the network can easily assign channels and lines (cf. Sect. 4.2.1.2) to connections and if necessary also assign transmission capacity to channels (cf. Sect. 4.2.3).

However, in case of outslot signaling through-connection of a circuit-switched call is not so closely correlated with signaling as in inslot signaling (cf. CCITT Rec. X.21 [4.3] and Sect. 4.4.2); moreover, here the user equipment has to handle the signaling channel as well as the traffic channels.

User-user signaling can be used to a limited extent by user terminals to exchange packet-mode information via the signaling channel, as on a virtual connection

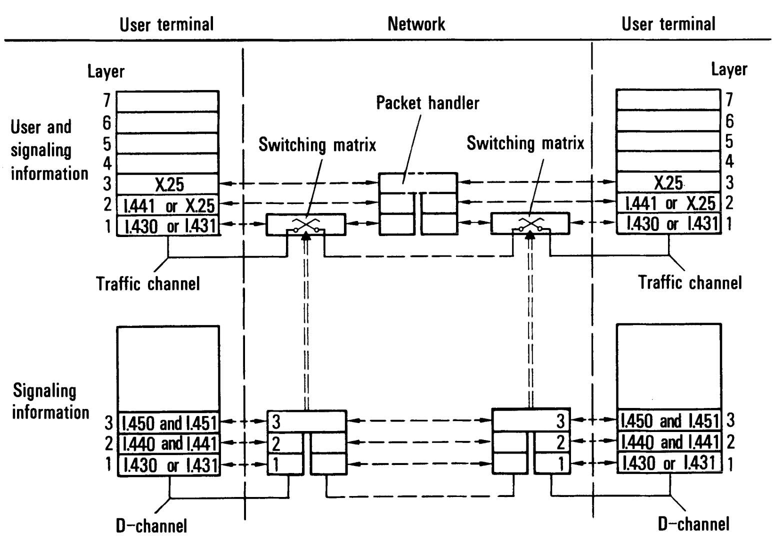


Fig. 4.18. Protocol Architecture for Packet-Switched Connections on a Traffic Channel (e.g. B-channel)

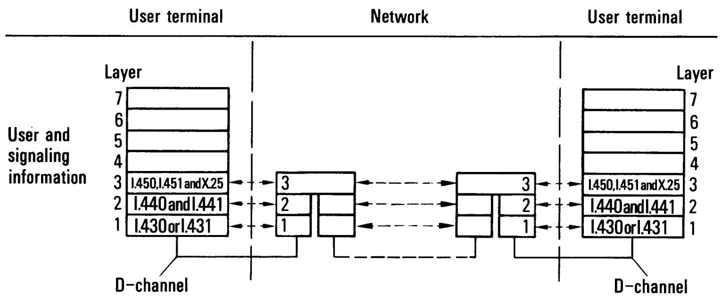


Fig. 4.19. Protocol Architecture for Packet-Switched Connections on the D-channel

(Fig. 4.17). This information is not interpreted by the network and is transmitted with the original packet sequence retained (cf. Sect. 4.3.5.5).

Packet-Switched Connections

A packet-switched connection on a *traffic channel* (currently only B-channels) is set up in two stages: first circuit-switched connection is between the user equipment and the packet handler (Fig. 4.18) is set up (using outslot signaling); on this circuit-switched connection the conventional packet switching protocols (conforming to CCITT Rec. X.25 [4.4]) are handled (inslot) in the data link and network layers, enabling the virtual connections to be set up and cleared down and the packets to be transmitted (cf. Sect. 4.4.4).

In packet-switched connections on the *D channel* on the other hand, the entire call control process takes place in the same (D)channel as the user information transfer (inslot). As soon as the virtual connection has been set up, layers 1 to 3 handle the transfer of the user information packets (Fig. 4.19 and Sect. 4.4.4).

4.3.3 Special Features of ISDN Signaling

4.3.3.1 Functions of the Network Terminations NT1 and NT2

Whereas the NT1 (cf. Sect. 4.1.2) only performs layer 1 functions, the NT2 (cf. Sect. 4.1.3) may perform a wide variety of functions (Fig. 4.20). It may have no function at all ("zero NT2", Fig. 4.20a and Sect. 4.1.3); in an "active bus" configuration it may handle layer 1 functions only (Fig. 4.20b; the sockets for terminal equipments contain amplification, storage or processing components); as a "statistical multiplexer" it may process layers 1 and 2 (Fig. 4.20c; the information frames of several D-channels are multiplexed in the data link layer); or as a private branch exchange it may process layers 1 to 3 (Fig. 4.20d).

In all four cases (Figs. 4.20a to d) the same set of specifications applies to reference points S and T (cf. Sect. 4.1.1).

4.3.3.2 Call Establishment

In the ISDN, different bearer services and teleservices can be accessed on one line (see Sect. 2.1). At the called end, the ISDN selects not only the correct subscriber line but

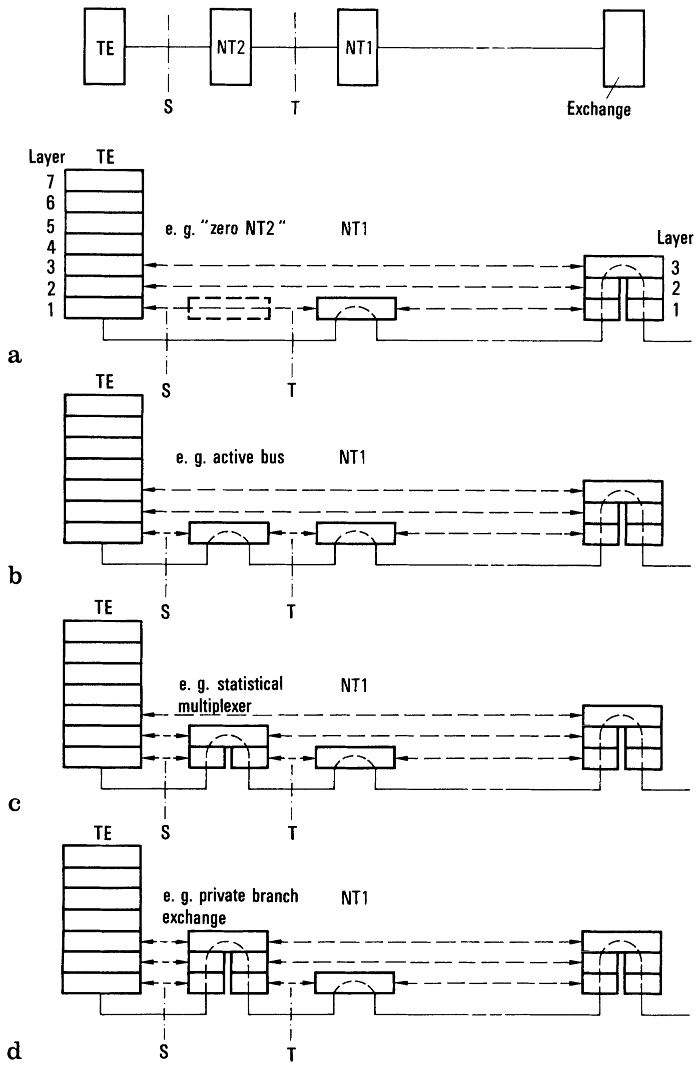


Fig. 4.20a–d. Roles of Network Terminations NT1 and NT2.
NT1, NT2 network terminations, S, T reference points, TE terminal equipment

also a terminal equipment compatible with the required service. To do this, during call establishment the calling terminal gives the network not only address information (ISDN address, see below) but also information on the desired transmission characteristics of the network connection and the compatibility requirements for the called terminal.

Suitable information fields of variable length are provided for this purpose in the connection request message (Table 4.6); these contain the bearer capability, ISDN address, compatibility information and user-user information.

Table 4.6. Information Fields in the Connection Request Message

Bearer capability	ISDN address		Compatibility information		User-user information
	ISDN number	ISDN sub-address	for layers 1 to 3	for layers 4 to 7	

Bearer Capability

The bearer capability information element (cf. CCITT Rec. I.451 [4.30]) contains the transmission characteristics required for the network connection. The main parameters are shown in Table 4.7. Some describe the transmission requirements, e.g. the transmission capacity, the symmetry (unidirectional, bidirectional), the suitability for specific communication types (e.g. voice or transparent transmission) and the

Table 4.7. Bearer Capability Parameters (Summary)

Parameter	Possible alternative values
Information transfer capability	Speech, audio (3.1 or 7 to 15 kHz bandwidth), video or unrestricted digital information
Information transfer mode	– Circuit mode – Packet mode
Information transfer rate	64, 384, 1536 or 1920 kbit/s
Configuration	– Point-to-point – Multipoint
Establishment of connection	– Demand (switched) – Reserved, permanent (for further study)
Transmission symmetry	– Bidirectional symmetric (i.e. same bit rate) – Unidirectional – Bidirectional asymmetric (i.e. different bit rates in the two directions)
Protocol options used for transmitting user information	
in layer 1	– Rate adaption (if required) as per standard (cf. Sect. 4.2.1.3) – Standardized speech encoding method – Undefined
in layer 2	– Per CCITT Recs. I.440 [4.27] and I.441 [4.28] – Per CCITT Rec. X. 25 [4.4], data link layer – Per CCITT Rec. Q.710 [4.12][a] – Undefined
in layer 3	– Per CCITT Recs. I.450 [4.29] and I.451 [4.30] – Per CCITT Rec. X.25 [4.4], packet layer – Undefined

[a] This option will probably be deleted in the Blue Book (1988).

protocol used in layer 1 (e.g. bit rate adaption, cf. Sect. 4.2.1.3 and 4.4). Other parameters describe the mode of establishment of a connection the network connection configuration (e.g. conference) and the switching method (circuit-switched, packet-switched). For packet-switched calls, the protocol to be used in layers 2 and 3 is specified.

On the basis of the given bearer capability, the network checks whether it can provide the desired service (compatibility check in the network). If the test result is negative, the connection request is rejected; if it is positive, connection set-up to the called user equipment proceeds; the latter in turn checks whether it is compatible with the connection request.

Addressing

The ISDN addressing concept goes beyond the "directory number" of the analog telephone network. The ISDN address consists of the ISDN number and optionally the ISDN subaddress (Table 4.8, cf. CCITT Recs. I.330 [4.9] and I.331 [4.31]).

The ISDN number is used by the network to identify a particular country and the desired access or access group. If appropriate specifications exist between user and network, the ISDN number can be additionally used at the called end for direct dialing-in (e.g. to private branch exchanges), or in some countries for selecting specific terminal equipments in bus configurations.

The ISDN number contains up to 15 digits including the country code and national destination code, but excluding any discriminating digit sequences (e.g. 00 for international calls). The country codes are the same as for the analog telephone network (see CCITT Recs. E.163 [4.32] and I.331 [4.31]).

The national destination code (Table 4.8) can be used to address gateways to dedicated networks or in some cases (e.g. in the USA) to different ISDNs.

The ISDN subaddress is used for more precise addressing of subcomponents within the user premises equipment selected using the ISDN number. It is transferred transparently from the calling to the called user equipment in a specific information field during call establishment. Its length is currently limited to a maximum of 32 decimal digits; extension to 20 octets (40 digits) is under discussion by the CCITT.

The ISDN subaddress can also be used by a private network on call establishment in the ISDN to transfer a complete private address to the next (private) node for further network and route selection.

Table 4.8. Structure of the ISDN Address

<table>
<tr><td colspan="4">ISDN address</td></tr>
<tr><td colspan="3">ISDN number
(max. 15 digits)</td><td rowspan="2">ISDN subaddress
(max. 32 digits)[a]</td></tr>
<tr><td>Country code</td><td>National destination code</td><td>Subscriber number</td></tr>
</table>

[a] To be extended to 20 octets (or 40 digits) in the Blue Book.

Compatibility Information

While the bearer capability describes the desired transmission characteristics of the connection, the *compatibility information* (see Table 4.6) contains more detailed information on the intended use of such a network connection by the user's equipment. For instance, a terminal adaptor (cf. Sects. 4.1.4 and 4.4) can inform the terminal adaptor on the other side of the network that a transparent 64-kbit/s network connection is to be used with an effective bit rate of 2.4 kbit/s only.

The addressed user equipment only accepts the connection request if it can accept the parameters in the bearer capability and in the compatibility information.

The compatibility information, if any, for layers 1 to 3 (cf. Fig. 4.15) and for layers 4 to 7 is specified in separate information fields; the information field for layers 1 to 3 may contain the same parameters as those defined for the bearer capability (cf. Table 4.7).

The compatibility information relates only to standardized service attributes. The user terminals can also optionally exchange non-standardized information (see *user-user information* field in Table 4.6 and Sect. 4.3.5.5). This information is transmitted transparently by the network and can be used by the terminal equipments e.g. for additional addressing or for compatibility checks.

4.3.3.3 Bus Configurations

The basic access can support bus configurations as well as point-to-point configurations. Bus configurations (see Figs. 4.8b and c) make specific new demands on the signaling. The point-to-point configuration (Fig. 4.8a) is normally handled by the network as a special type of bus configuration.

Unique Terminal Endpoint Identifier

Every terminal equipment operated in a bus configuration must have a different terminal endpoint identifier so that the network can identify the sending terminal equipment when a frame is received and can send an information frame to a specific terminal equipment. The terminal endpoint identifier is incorporated in the transmitted information frames as part of the data link layer address (cf. Sect. 4.3.4).

Knowledge of the Configuration not Required in the Network

As long as there is no connection request outstanding, the network does not need to know the current status of the terminal equipments in a user station, neither their characteristics nor their terminal endpoint identifiers (TEI). When an outgoing call is made, the network knows the TEI of the calling terminal equipment as soon as the latter transmits the connection requests. With incoming calls, the network first transmits the connection request to all the terminal equipments (cf. Sect. 4.3.5). The latter evaluate the connection request by checking the bearer capability and any address and compatibility information contained therein (see Sect. 4.3.3.2). All the terminal equipments for which this check is positive inform the network accordingly. From their response, the network knows the TEIs of the terminal equipments suitable for the connection, and can now selectively address these terminal equipments at least for the duration of this signaling activity.

Depending on the type of additional address information, several, one or even no terminal equipment may respond to an incoming connection request for the service desired. If the connection request contains no additional address information, all the terminal equipments for which the compatibility check is positive respond. If more than one terminal equipment responds, the network assigns the connection to the terminal equipment that responded first and rejects the other terminal equipments (cf. Sect. 4.3.5).

Configuration Flexibility

As the network does not normally recognize the current status of the connected terminal equipments during inactive periods, the user can change the configuration, e.g. unplug, transfer, plug in, switch off or switch on terminals without informing the network. If, however, a connection is to be maintained when a terminal equipment is transferred, the user must first explicitly "park" the connection in the network (see Sect. 4.3.5.4).

4.3.3.4 Simultaneous Signaling Activities

Several circuit-switched connections – one per traffic channel – can exist simultaneously on one subscriber line (cf. Table 4.3). Consequently it has to be possible for several signaling activities – one for each connection – to be handled simultaneously via the signaling channel. Other signaling activities are needed for controlling packet-switched connections (cf. Sect. 4.4) and for registering, controlling or cancelling supplementary services.

In order to enable the transmitted signaling messages to be assigned to the correct signaling activity, every signaling message contains a unique call reference which, in combination with the data link layer address, is unambiguous. The call reference is issued by the entity that initiates the signaling activity, i.e. by the user equipment for an outgoing call and by the network for an incoming call.

4.3.4 Link Access Procedure on the D channel

The LAPD (Link Access Procedure on the D channel) is responsible for protecting all the information (signaling information and packet-mode data, cf. Sect. 4.2.1.1) transmitted on the D-channel in both directions against transmission and sequence errors and for allocating unique terminal endpoint identifiers (TEI, see Sect. 4.3.4.3).

The detailed specifications for LAPD are given in CCITT Recs. I.440 [4.27] and I.441 [4.28] (see Sect. 4.2.1.1 for the E-channel, not dealt with further here and probably not realized).

4.3.4.1 Features of the Data Link Layer

The features of the data link layer are oriented towards bus configurations (Fig. 4.8b and c); the point-to-point configuration (Fig. 4.8a) is normally operated by the network as a special type of bus configuration.

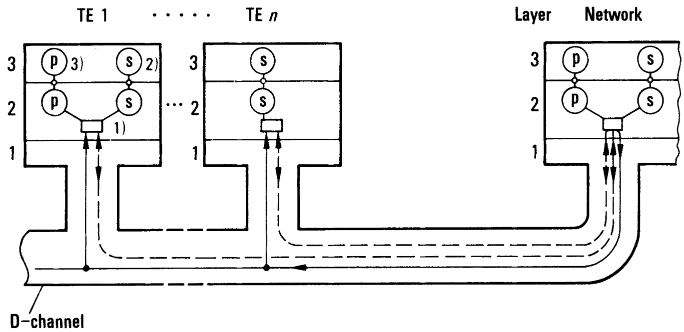


Fig. 4.21. Features of the Data Link Layer.
—— Broadcasting of s or p information, – – – – transfer of s or p information from the network to selectively addressed terminal equipments and vice versa.
p packet-mode data, s signaling information, TE terminal equipment.
1) Multiplexing/demultiplexing equipment for s and p information frames, 2) Processing of signaling information, 3) Processing packet-mode data

The main features offered by the data link layer to the network layer (cf. Fig. 4.15) for information transfer are as follows (Fig. 4.21):

- Unacknowledged transfer of signaling information (s) and packet-mode data (p) from the network to all terminal equipments (broadcast mode) which process s and p information (continuous line in Fig. 4.21)
- Acknowledged and unacknowledged transfer of s and p information from the network to selectively addressed terminal equipments and vice versa (dashed lines in Fig. 4.21).

In the case of acknowledged transfer (see Sect. 4.3.4.2), the data link layer is responsible for detection of transmission errors, error correction and frame sequence monitoring; with unacknowledged transfer, it only handles error detection (incorrectly received information frames are ignored by the receiver).

4.3.4.2 Data Transfer Protocol

The structure of all the protocol elements fully meets HDLC standards developed prior to the ISDN (HDLC: High Level Data Link Control procedure, e.g. ISO 3309 [4.33]).

To operate several terminal equipments simultaneously, including terminal equipments handling both signaling information (s) and packet-mode data (p) (see Fig. 4.21), an address field two octets long is normally used. In the octet transmitted first, a distinction is drawn between s and p information and management information (see Sect. 4.3.4.3); the second octet contains the unique terminal endpoint identifier of the terminal involved or a broadcast identifier (cf. Sect. 4.3.3.3). With regard to access to the D channel in the basic access (see Sect. 4.2.2.3), contiguous "ones" are used as interframe time fill between information frames in both transmission directions; in the case of the primary rate access, contiguous flags [4.17] are used (Table 4.9).

Table 4.9. Main Parameters of the Data Transfer Protocol

Parameters	Parameter values	
	Basic access	Primary rate access
Interframe time fill	contiguous ones	contiguous flags
Modulo	128	128
Window size	for s:1; for p:3	7
Maximum information field length	260 octets	260 octets
Acknowledgement supervision time	1 s	1 s
Frame repetition in the event of fault	up to 3 times	up to 3 times

The protocol element coding also complies with the HDLC standards (e.g. ISO 4335 [4.34]; cf. also ISO 7809 [4.35] and CCITT Recs. X.25 [4.4], X.75 [4.36], T.70 [4.37]).

Unnumbered information frames (UI frames) are used for unacknowledged transmission (cf. Sect. 4.3.4.1), in compliance with ISO 4335 [4.34].

For acknowledged data transfer, multiple frame modulo 128 operation is now specified. (The earlier version of CCITT Rec. I.441 had three options – single frame, multiple frame modulo 8, and modulo 128).

Basic features of the data transfer protocol are that the network and the user equipment have equal access (balanced procedure) and that several layer 2 connections e.g. from different terminal equipments can be operated simultaneously – as far as necessary with different parameters. The data transfer procedure is based on the following principle: the receiver acknowledges error-free reception of information frames to the sender and thus implicitly allows further frames to be sent. If an expected acknowledgement is not sent within a specific time, the transmission error is corrected by repeating the frame. Sequence errors such as loss or duplication of a frame are detected from the consecutive numbering of the information frames (sequence number) and corrected; the highest sequence number is followed once more by the lowest in each case.

The window size (Table 4.9) determines how many frames the sender may transmit without receiving appropriate acknowledgements: for example, with window size 1 the sender must delay transmission of the next information frame until the acknowledgement for the previous frame has arrived. In order to ensure that the sequence number is unambiguous, the window size must always be smaller than the modulo. The window size, in case of the basic access, is 1 for signaling information and 3 for packet-mode data; for the primary rate access, the window size is generally 7 (special regulations apply to satellite routes with long delay times). The maximum information field length is 260 octets in all cases.

The transfer procedure specification is similar to LAPB (Link Access Procedure Balanced, see CCITT Rec. X.25 [4.4]), but somewhat simplified compared with LAPB (see Sect. 4.4).

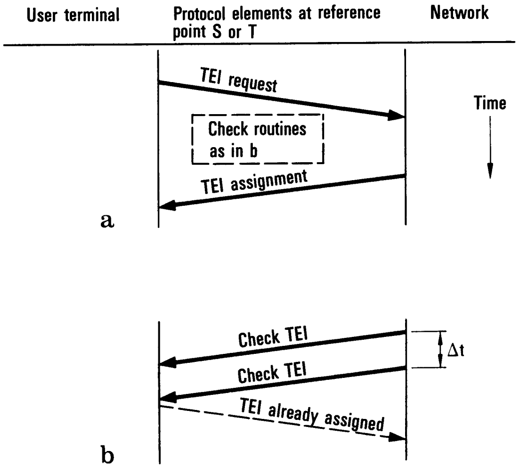


Fig. 4.22a, b. TEI Assignment Protocol. TEI Terminal endpoint identifier

4.3.4.3 Assignment of Unique Terminal Endpoint Identifiers

As described in Sect. 4.3.3.3, every terminal equipment in a bus configuration must have a different terminal endpoint identifier (TEI). A management entity assigned to the data link layer automatically handles TEI assignment on the network side: as soon as a terminal equipment requires a new TEI value (e.g. after the voltage supply has been restored), it sends a request to the network (Fig. 4.22a) and is then assigned a free TEI value by the network. The network ascertains which TEI values are available by interrogating the terminal equipments (Fig. 4.22b) to check whether a certain TEI value has already been used; if no terminal answers after two interrogations, the TEI value in question is considered to be free.

Normally the network will hold a store of free TEI values so that it can assign a TEI immediately in response to a TEI request, thereby eliminating the time-consuming test indicated in Fig. 4.22a. As well as testing the TEI values individually, the network is also able to test all TEI values simultaneously by sending "test TEI" to all terminal equipments on a broadcast basis, as in Fig. 4.22b ([4.28]; cf. Sect. 4.3.4.1).

The network can also declare on a broadcast basis one or all of the TEI values in the terminal equipments invalid; the network will use this option if it concludes from certain protocol deviations that the same TEI value is erroneously being used by more than one terminal equipment.

The TEI value can also be preset in terminal equipments. In this case the terminal equipment must have the TEI value it requires tested for uniqueness by the network according to the same rules (Figs. 4.22a and b). The TEI value to be tested is transferred in the TEI request (Fig. 4.22a).

The TEI assignment procedures (Fig. 4.22) are differentiated from the signaling information and packet-mode data by a special management entity identifier in the first address octet (cf. Sect. 4.3.4.2). During TEI assignment, no TEI value is yet available for addressing the terminal equipment concerned; a terminal-generated random number 16 bits long is therefore used for addressing the terminal equipment in this phase.

4.3.5 Signaling for Circuit-Switched Connections

The signaling handled by the call-control procedure (Fig. 4.15) enables the calling user to select the desired user and communication service. The signaling protocol specified for reference points S and T is uniform for all services. The associated standard is contained in CCITT Recommendations I.450 [4.29] and I.451 [4.30].

4.3.5.1 Simple Call Establishment

Figure 4.23 shows a simple example of circuit-switched call establishment for telephony. Let two telephones (x and y in Fig. 4.23) be arranged in a bus configuration at the called end (cf. Sect. 4.2.2.1). Let the ISDN address of the called user be transferred by the calling user equipment as a complete block (block dialing); block dialing is a typical feature of user terminals with automatic dialing (e.g. text terminals), but is also frequently used on advanced telephones (e.g. with name key dialing). The user-terminal sequences illustrated in Fig. 4.23 are only intended as examples.

After the name key has been pressed, the calling telephone transfers to the network, in the SETUP message, the bearer capability (see Sect. 4.3.3.2), the ISDN address of

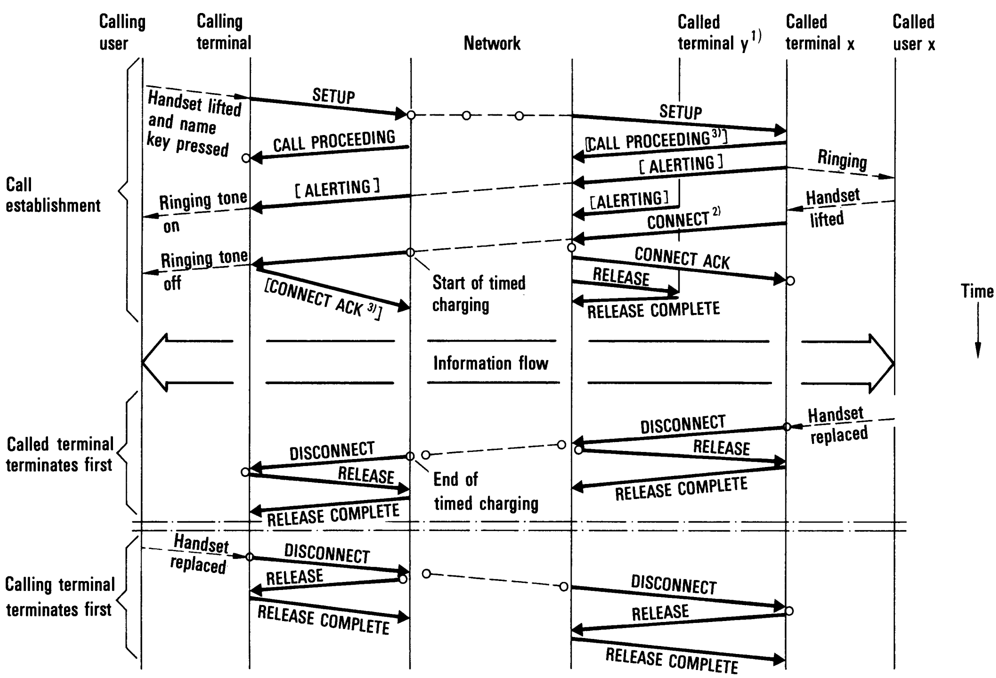


Fig. 4.23. Establishment and Clearing of a Circuit-Switched Call (Set-up with Block Dialing). [...] Transfer of this message may be omitted, o typical through-connection sequence (for call establishment) and disconnection sequence (for call clearing) for the connection, ACK acknowledge

1) the procedures for the terminal user are not shown; 2) the called terminal x answers before terminal y; 3) permitted for reasons of symmetry (normally ignored by the receiver)

the called terminal and – if defined – the compatibility information for the "telephony" service. The network confirms with CALL PROCEEDING that call establishment is in progress and at the same time assigns a vacant B-channel to the calling user terminal, to which the latter normally switches immediately. This mainly applies to telephones in networks providing audible tones (e.g. ringing tone) or announcements in the assigned B-channel at the calling end.

As soon as the destination exchange has detected the connection request, it selects the addressed line, if necessary tests the compatibility and transmits the connection request (SETUP) to all the terminal equipments on a broadcast basis (see Sect. 4.3.4.1). The SETUP message contains the bearer capability and if applicable additional address information and compatibility information (cf. Sects. 4.3.3.2 and 4.3.3.3). This message is also used to assign a free B-channel: only the specific terminal equipment to which the connection is subsequently assigned will then switch to this B-channel (see below). All the terminal equipments which process signaling information and are compatible with the desired service (terminals x and y in Fig. 4.23) inform the network accordingly with ALERTING (the telephones "ring"), and only send CONNECT when they are able to accept the call (in the case of telephones: "handset lifted"). The connection is now switched through by the network and assigned, on CONNECT ACK, to the terminal equipment which had first sent CONNECT. The other terminals are cleared with RELEASE and confirm this with RELEASE COMPLETE.

The calling terminal is informed of the progress of call establishment by the ALERTING and CONNECT messages: ALERTING corresponds to the ringing tone, CONNECT means that the connection has been switched through within the network. The calling and called user terminals are informed by the signaling in the D-channel when the connection is switched through in the network, but not when it is switched through from user terminal to user terminal (end-to-end) (cf. Sect. 4.4.2).

To facilitate direct connection of private branch exchanges (without the public network), the layer 3 protocol has been defined symmetrically as far as possible; for this reason, sending of the CALL PROCEEDING and CONNECT ACK messages, for instance, (Fig. 4.23) is also permissible for user terminals equipments.

If the connection cannot be set up as desired, the user terminal is notified and informed of the cause.

The signaling information is transferred with acknowledgement by the data link layer as far as possible (see Sect. 4.3.4). Only broadcast messages (e.g. normally only SETUP on the called side; cf. Fig. 4.23) have to be transferred unacknowledged, as the data link layer does not provide acknowledged transfer for broadcast messages.

4.3.5.2 Simple Call Clearing

The calling user terminal, the called user terminal and the network can initiate connection clear-down at any time and independently of each other by sending the DISCONNECT message (lower half of Fig. 4.23).

As discussions stand within CCITT, call clearing is handled as follows:

If the initiative for call clearing comes from a user terminal, the latter disconnects itself from the traffic channel and releases it before sending the DISCONNECT message. After receiving the DISCONNECT message, the network disconnects the traffic channel from the connection within the network, initiates connection cleardown within the network and acknowledges with RELEASE. Finally the user terminal terminates signaling activity (cf. Sect. 4.3.3.4) by sending RELEASE COMPLETE, whereupon the network in turn terminates signaling activity.

On the other, "passive" side of the network, the network likewise disconnects the traffic channel from the internal network connection and, with the DISCONNECT message, requests the user terminal to disconnect itself immediately from the traffic channel.

The user terminal confirms this with RELEASE; the network finally terminates signaling activity by sending the RELEASE COMPLETE message, on receipt of which signaling activity is also terminated for the user terminal.

If the initiative for disconnection comes from the network, both user terminals are informed by the network with the DISCONNECT message. For both user terminals, the subsequent course of events corresponds to the procedure shown at the bottom right in Fig. 4.23.

4.3.5.3 Refined Call Establishment and Clearing

Call Establishment with Digit-by-Digit Dialing at the Calling End

Manually operated terminals do not necessarily have to collect the digits of the ISDN number into one block (block dialing), but can transfer the digits to the network singly or in groups.

Direct Dialing in with Point-to-Point Configurations

Part of the ISDN number can be used for direct dialing in to private branch exchanges or comparable installations (cf. Sects. 4.1.5 and 4.3.3.2). The network then transfers to the called user equipment (e.g. PABX) that part of the ISDN number not required for selecting the line either as a block or digit by digit; however, digit-by-digit transfer is only possible in the point-to-point configuration as otherwise unaddressed terminals connected to the bus could compete for the connection prematurely.

Terminal Selection in Bus Configurations

Also under discussion is the possibility of allowing part of the ISDN number to be used for selecting specific terminals in bus configurations in some countries or networks; for this purpose, the called terminals must know their individual selection address and compare it with the remainder of the ISDN number transferred in the SETUP message (Fig. 4.23).

Channel Negotiation by the Calling User Terminal

For private branch exchanges (PBX) it can be useful in the case of an outgoing call to through-connect the switching matrix before sending SETUP ("forward through-connection"), and for the PBX itself to negotiate a B-channel for the connection. The ISDN permits the calling user terminal to propose the channel but reserves the right to reject the proposed channel.

Channel Negotiation by the Called User Terminal

In packet-switched connections (see Sect 4.4.4) the called user terminal can inform the network of the channel on which it wishes to receive the connection: on a new traffic channel or on one already being used by it (currently only a B-channel) or on the D-channel. This channel negotiation facility can also be useful in circuit-switched connections: a called private branch exchange can use it to avoid possible blocking of its switching matrix. For circuit-switched connections, however, this negotiation facility is provided by the network only as an option and only for primary rate accesses.

4.3.5.4 Control of Supplementary Services

Many supplementary services (cf. Sect. 2.3.3) must be registered in advance in the network before they can be called up or activated; standardized operational procedures are therefore required for registering and handling supplementary services. International standards only exist for a few supplementary services, e.g. for "parking" a connection. This supplementary service has special significance for reconfiguration (e.g. transferring or exchanging a terminal), especially in the bus configuration. A user can temporarily park an existing connection in the network under a password and retrieve it again within a specified time under the same password (cf. Sect. 4.3.3.3).

In the absence of international specifications, national or network-specific specifications are likely to provide an interim solution.

4.3.5.5 User-User Signaling

The ISDN can optionally provide user-user signaling on the D-channel as a supplementary service. With this facility, user terminals have a limited capability, as with a virtual connection, of sending each other information not interpreted by the network during the establishment (cf. Sect. 4.3.3.2), clear-down and holding time of a circuit-switched connection. Possible applications include exchange of keywords during call establishment, non-standardized communication between two user terminals about the use of an established connection, and above all signaling for tie lines between private branch exchanges.

User-user signaling is a network option and can be organized in a great variety of ways by the individual networks in terms of the maximum permitted information length per packet (32 or 132 octets), throughput and call charges.

There are two possible modes of transfer: piggyback transfer as a part of a signaling message forwarded from user terminal to user terminal, (e.g. SETUP, CONNECT, etc; cf. Fig. 4.23), and transfer with a special message defined for that purpose [4.30].

4.3.5.6 Stimulus Protocol

Review of the Functional Protocol

The handling of terminals (input and output units) and the use of procedures by the terminal user already varies considerably for the terminals employed in conventional networks. (The term "user-terminal interface" refers collectively to the man-machine interface, i.e. to the handling of terminals and the related procedures). Present-day telephones, for example, use different numbers of keys assigned in different ways as input units as well as different visual displays (e.g. a variety of lamp indications and displays in a variety of arrangements) and a wide range of acoustic signal generators as output units. The user-terminal interface with non-voice terminals, e.g. text and data terminals, is usually quite different again. In the use of voice (announcements) and text, the user-terminal interface is also dependent on the particular national or regional language.

Further developments will continue to occur in this area: integration of services in the ISDN will favor the trend towards multifunction terminals in which the same input and output units are used for different communication functions and local functions. Further impetus can be expected from progress in sensor technology, voice input and output systems, etc.

In order to avoid burdening the public network with this growing number of possible interfacing solutions, but without letting the public network stand in the way of new solutions, the signaling procedures at the ISDN user-network interface are described as functionally as possible (see Sects. 4.3.5.1 to 4.3.5.5) i.e. as independently as possible from the concrete physical input/output units of a specific terminal.

A software module in the terminal undertakes the conversion between the user-terminal interface procedures (actions of the user and indications for the user) and the functional signaling events at the user-network interface. Hence, the optimum user interface solution can be implemented for each terminal without affecting the network.

Concept and Applicability of the Stimulus Protocol

Although as a general approach the *functional protocol* concept described above is the answer to the requirements discussed, in some circumstances it can also be useful to operate certain terminal types with special functions realized essentially in the network. This applies especially to few but widely distributed terminal types, such as terminals designed exclusively for telephony. The *stimulus protocol*, standardized in outline (see CCITT Rec. I.451 [4.30]) is intended for a user-terminal interface optimized in this way.

The stimulus protocol is based on a hypothetical "stimulus mode terminal", the functional units of which, especially the input and output units, are controlled as directly as possible by the network (Sect. 5.1). It is ensured that stimulus mode terminals can be operated in conjunction with "functional terminals" in the same bus configuration and can also be connected (via the network) to functional terminals.

A stimulus mode terminal reports user actions such as keyboard inputs directly to the network as "stimuli", without itself processing them further. Whereas in the functional protocol it is the terminal that interprets the user actions and converts them to the functional protocol, here it is the network that interprets these stimuli in order to assign them their functional signaling significance depending on previous stimuli or previous states. The stimuli transferred from the network in the other direction are control instructions aimed specifically at functional units in the terminal, e.g. write commands for the screen.

A stimulus mode terminal has the following advantages compared with a functional terminal:

- A stimulus terminal is more flexible in use. By means of suitable control programs in the network, the same terminal can be adapted to suit different applications and different users; new and conventional service attributes can be offered with different user interface procedures and even added subsequently, without preparatory operations or changes (e.g. in the software) being necessary in the terminal. As the logic in the terminal only transmits the stimuli and does not interpret them, a stimulus mode terminal can more easily have a new input or output unit added (e.g. an additional keypad).

- A stimulus mode terminal is – at the network's expense – simpler and thus somewhat cheaper, i.e. more comparable in cost with a single telephone connectable to an analog telephone line.

The stimulus concept has the following disadvantages:

- Greater storage requirements and processing load in the network:
  - The number of transmitted signaling messages is greater.
  - The network must convert between the functional protocol used exclusively within the network and the stimuli transmitted at the user-network interface.
  - The more different terminal types are to be operated, the more program storage must be provided by the network.
  - The network may have to store a large number of variables: terminal-type-specific variables (e.g. tables for the interpretation of key codes) are not critical; line-specific variables (e.g. directory number assignment to name keys) are more costly; terminal- or even terminal-userspecific variables are particularly costly.
- Limited general connectability of the terminals: the terminals can only be usefully operated on another access if the network provides the necessary control software and variables for the new access too (especially problematic for connecting a portable stimulus mode terminal to different networks).

These disadvantages are less serious for connecting terminals to private branch exchanges than for connecting them to the public network, because a private branch exchange can be more readily adapted to suit individual user requirements, and because a terminal normally remains within the area of the private branch exchange.

Current standards define only a general framework for the stimulus concept. Realistic support of a stimulus mode terminal by the network requires additional specifications, including a more precise definition of the characteristics (e.g. the screen capacity) of each stimulus mode terminal, as perceived by the exchange, of the operating procedures.

A combination of functional protocol and stimulus protocol is also possible. The following approach seems particularly promising: basic functions such as call establishment and clearing and the main supplementary services, for which operating procedures are or will be internationally standardized, are handled in accordance with the functional protocol; additional, nonstandardized supplementary services are handled in accordance with the stimulus protocol concept with the aid of the information elements KEYPAD and DISPLAY [4.30]. KEYPAD is here used to transfer the terminal user's keyboard inputs to the network, DISPLAY to transfer information from the network to the terminal user.

Table 4.10. Adaption of Existing Interfaces for Text and Data Communication to ISDN

Switching principle	Network	Interface at reference point R	CCITT Recommendation for adaption and network interworking	Function of terminal adaptor (TA)
Circuit switching	Telephone network (PSTN)	t/r	I.530	– *Signaling conversion* t/r – S by *TA-t/r* (Fig. 4.24) with digitized modem signals for conventional videotex, facsimile gr. 2/3 and data transmission (*indirect* V.-S adaption)
		CCITT Rec. V.24, V.25, V.25bis	I.463 (V.110); I.530, I.515	– *Signaling conversion* V.25(bis) – S by *TA-V* (Fig. 4.24) for *direct*, purely digital V.-S adaption for data transmission *without* modem at ISDN subscriber access – *Bit rate adaption* (two-stage per I.461) Bit rates from CCITT Rec. V.5 to 64 kbit/s – *Alignment procedure in the user information channel* Status information transfer between the TAs
	Text/data network (CSPDN)	CCITT Rec. X.21, X.21bis, X.20bis	I.461 (X.30); I.540 (X.321), X.81	– *Signaling conversion* X.21, X.21bis, X.20bis – S in TA-X.21 (X.21bis, X.20bis) (see Fig. 4.26) – *Bit rate adaption* (two-stage, see Fig. 4.27) X.1 – 64 kbit/s[b] – *Ready for data alignment procedure in the basic channel* Status information transfer between the TAs
Packet switching	Text/data network (PSPDN)	CCITT Rec. X.25	I.462 (X.31); I.550	– *Bit rate adaption* X.1 – 64 kbit/s Alternative 1: flag stuffing[a] Alternative 2: per I.461

[a] Cf. HDLC data transfer procedure [4.32].

[b] Three-stage method for X.20bis with additional stage performing asynchronous-to-synchronous conversion using the same technique as defined in V.22 for support of X.1 user rates.

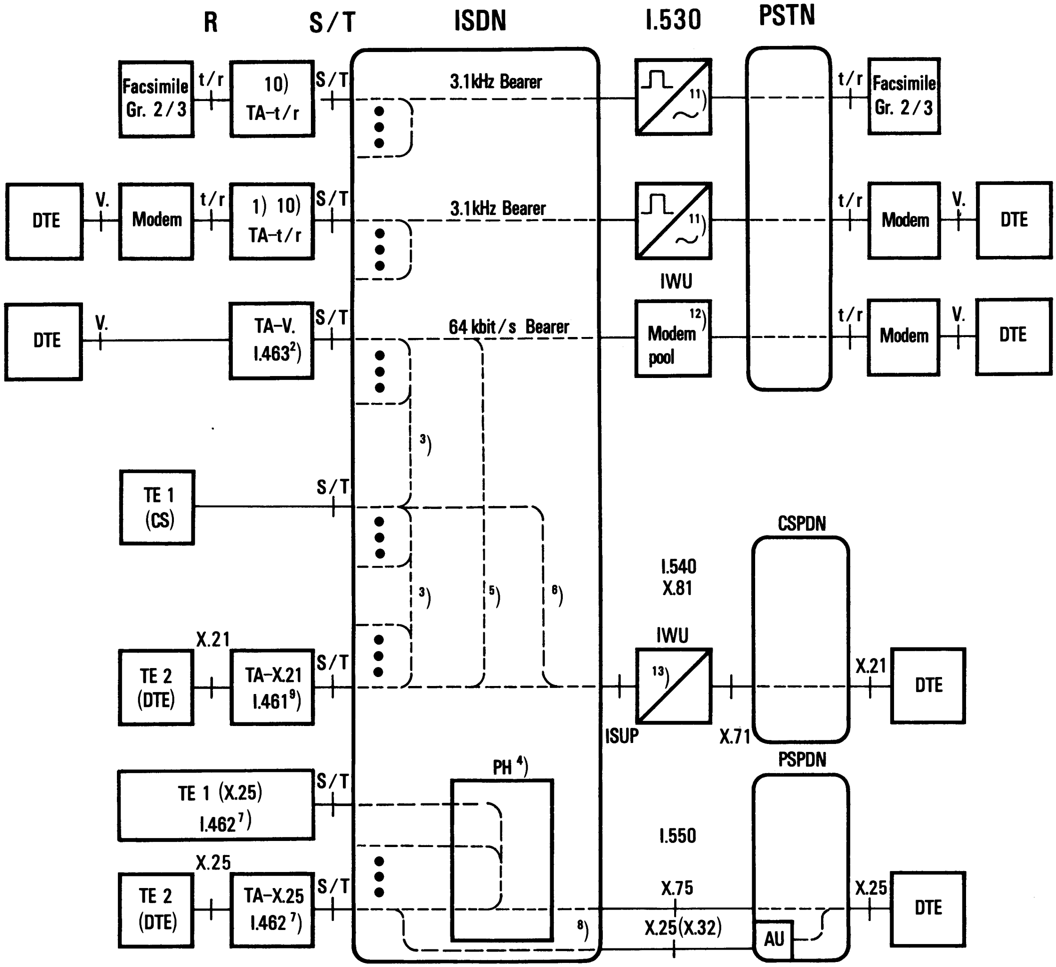


Fig. 4.24. Configuration for the Support of Terminal Equipments with Non-ISDN Interfaces by an ISDN.

1) Adaption of the analog tip/ring interface: digitized modem signals (indirect adaption of V to S interface, decentralized conversion);
2) Method in accordance with CCITT Rec. I.463: digital signals (direct adaptation of V to S interface, central conversion in IWU: modem pool);
3) Only possible for 64 kbit/s (user class of service 30 as defined in CCITT Rec. X.1);
4) Packet-mode bearer service supported within ISDN by a remote X.25 packet handler (PH, cf. Sect. 4.4.4.1);
5) Possible for identical user bit rates as a result of commonality in inslot frame structure and bit rate adaption for TA-V. and TA-X.21 (cf. Sect. 4.4.2.2);
6) Bit rate conversion between non-identical data transmission services provided in the IWU by flag insertion/extraction (use of LAP B in the user information channel) and buffering in the IWU (e.g. between user classes of service 4 and 30 in the case of Teletex terminals);
7) Identical procedures apply to the combination of an X.25 DTE and its TA as well as to a packet-mode TE 1;
8) ISDN provides transparent access channel to PSPDN;
9) Adaption of X.21bis and X.20bis is also covered by Rec. I.461;
10) A/D conversion conforming to Rec. G.711 (A-law, μ-law);
11) D/A conversion conforming to Rec. G.711 (A-law, μ-law);

4.4 Connection of Terminals with Conventional Interfaces to the ISDN

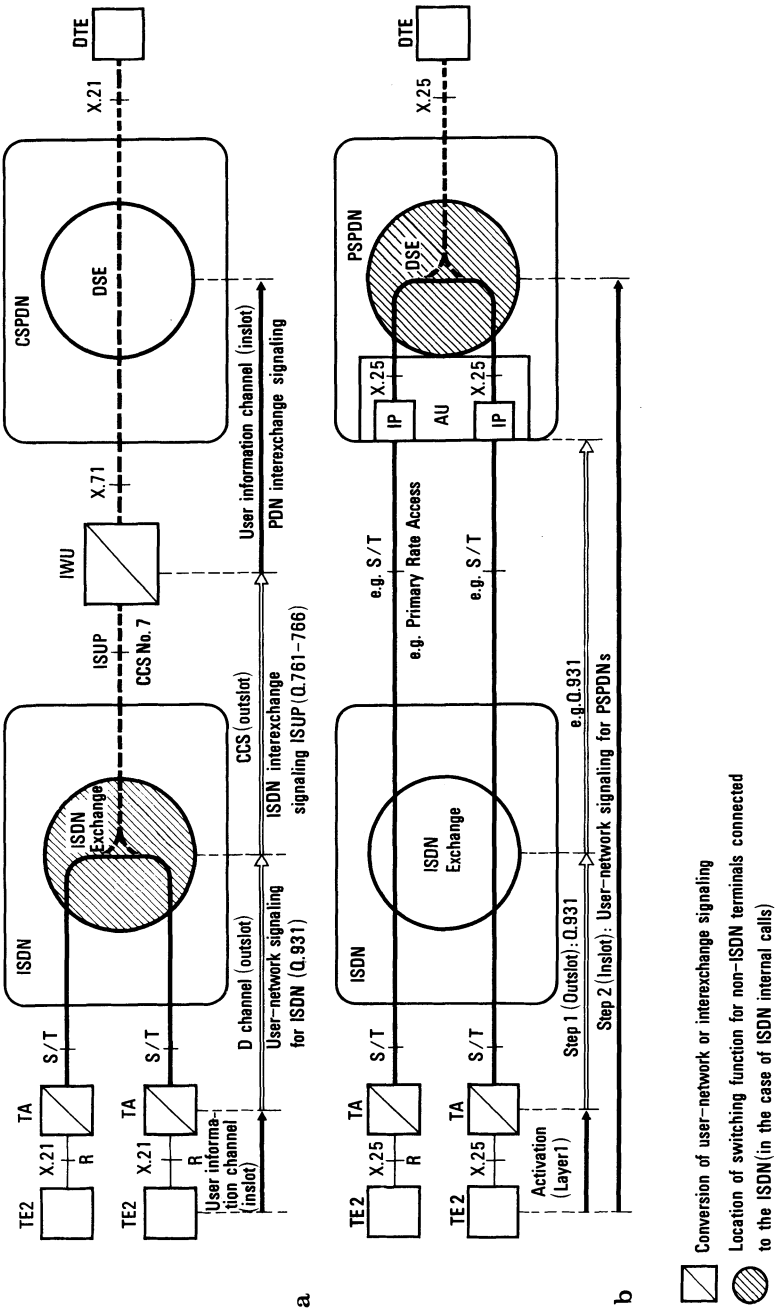
A flexible ISDN implementation strategy must ensure that, for a transitional period at least, not only new ISDN terminals (TE1) connected via the ISDN-specific S interface (see Sects. 4.2 and 4.3), but also existing terminals (TE2) from other networks can be operated at the ISDN user access (see Sects. 2.3.4 and 4.1.4). The main consideration here is to be able to continue using existing terminals and to take advantage of the higher ISDN transmission rate of 64 kbit/s without having to implement a new interface. The options relevant to the connection and interworking of various terminal equipments are illustrated in Fig. 4.24.

ISDN functions to support non-ISDN interfaces and networks can be subdivided into three areas:

- Adaption of the interfaces per CCITT V. and X. Series Recommendations to the requirements of the ISDN S/T interface by means of special *terminal adaptors* TA (see Sect. 4.1.4). In Fig. 4.24 four main categories of terminal adaptors (TA) are shown. These correspond to the basic adaption possibilities that are summarized in Table 4.10 in conjunction with the relevant CCITT Recommendations: *TA-t/r* supporting the analog tip/ring interface (see Sect. 4.4.3.1), *TA-V.* supporting V.24/RS232C or V.35 (see Sect. 4.4.3.2), *TA-X.21* (see Sect. 4.4.2) and *TA-X.25* (see Sect. 4.4.4). Practical TA implementations are likely to also offer combinations of the above-mentioned basic adaption possibilities. Options include combination of modem and TA-t/r (Codec) for accessing, via the ISDN, resources still connected to the telephone network PSTN, integration of the X.3 PAD capability for X.25 conversion into the TA-X.25, as well as combined voice/data adaptors offering an analog voice port for connecting a conventional handset (substituting the conventional simultaneous voice/data technology of pre-ISDN in conjunction with CENTREX – see Sect. 6 – in the U.S.). As an economical alternative to external TAs, plug-in TAs e.g. connected to a PC bus will turn existing non-ISDN terminals (TE2) into ISDN Terminals (TE1).

Fig. 4.24 (contd.)
12) Bit rate adaption in accordance with Rec. I.463;
13) Bit rate adaption in accordance with Rec. I.461 for identical transmission services.

TE1	ISDN terminal equipment with S interface
TE2	Terminal equipment with other interfaces (adaption to the S interface with terminal adaptor TA)
DTE	Data terminal equipment
IWU	Interworking unit
CS	Circuit mode
PSTN	Public switched telephone network
CSPDN	Circuit-switched public data network
PSPDN	Packet-switched public data network
– – – – –	Cases of interworking
PH	Packet handler
AU	Access unit
t/r	Analog tip/ring interface


a
TE2
X.21
R
TA
S/T
ISDN
ISDN Exchange
ISUP
CCS No. 7
IWU
X.71
CSPDN
DSE
X.21
DTE
User information channel (inslot)
D channel (outslot)
User-network signaling for ISDN (Q.931)
CCS (outslot)
ISDN interexchange signaling ISUP (Q.761–766)
User information channel (inslot)
PDN interexchange signaling
b
TE 2
X.25
R
TA
S/T
ISDN
ISDN Exchange
e.g. S/T
e.g. Primary Rate Access
IP
AU
X.25
PSPDN
DSE
DTE
Activation (Layer1)
Step 1 (Outslot): Q.931
e.g. Q.931
Step 2 (Inslot): User-network signaling for PSPDNs
Conversion of user-network or interexchange signaling
Location of switching function for non-ISDN terminals connected to the ISDN (in the case of ISDN internal calls)

- *Terminal interworking between ISDN terminals (TE1) and other terminals (TE2) via the ISDN* – at least if the net transmission rate and the switching principle are compatible (see cases of interworking in Fig. 4.24).
- *Network interworking functions* ensure that all the terminals connected to the ISDN can reach communication partners both in the telephone network and in public data networks (see Fig. 4.24). CCITT specifications concerning interworking between ISDN and other networks are still under discussion. A provisional list of Draft Recommendations scheduled for adoption in 1988 can be found in Annex 2.10.

4.4.1 ISDN Bearer Service and Public Data Network Access Solutions

The basic possibilities for connecting existing text and data terminals to the ISDN are briefly explained below (Fig. 4.25); these possibilities are the use of *ISDN bearer services*, discussed with reference to adapting the X.21 interface [4.3], and the *access to data transmission services that are provided by public data networks (PDN)*. This latter case is restricted to the adaption of the X.25 interface [4.4].

These two approaches, the salient features of which are summarized in Table 4.11, differ mainly in terms of where the switching function for the TE2 terminals connected to the ISDN is handled: either in the ISDN (ISDN bearer service method) or, as before, in switching nodes of an independent public data network (PDN). The latter case is known as the *PDN access solution*; here, in contrast to the ISDN bearer service method, the gateway to the PDN is not only used to communicate with data network users but is also required for ISDN internal traffic between the TE2 terminals. The choice between the ISDN bearer service or the PDN access approach basically depends

Fig. 4.25a, b. Switching Function for Non-ISDN Terminal Equipments connected to the ISDN, provided either entirely within the ISDN or by accessing a data network through ISDN. **a** Use of ISDN bearer services; **b** Access to data transmission services provided by PSPDNs.

TE2	Terminal equipment with R interface (complying with interface Recommendation other than I.-Series)
TA	Terminal adaptor
DTE	Data terminal equipment
IWU	Interworking unit
AU	ISDN access unit of the data network
IP	ISDN port
PDN	Public data network
DSE	Data switching exchange
CSPDN	Circuit switched public data network
PSPDN	Packet switched public data network
CCS	Common channel signaling
S	ISDN user-network interface
X.71	Interexchange signaling for synchronous circuit switched data networks [4.45]
ISUP	ISDN interexchange signaling (ISDN User Part, cf. Sect. 6.3): Q.761 – Q.766

1) Handled transparently in the ISDN. 2) In accordance with Rec. X.32. ——— Connection between two data terminal equipments both connected to the ISDN; ----- connection to a terminal equipment connected to the PDN; ⟹ *outslot* signaling: in the D-channel (subscriber side) or in the common signaling channel (ISDN interexchange side); ⟶ *inslot* signaling: in the user information channel

Table 4.11. Support of Terminal Equipments with Non-ISDN Interfaces by an ISDN

Concept	Use of ISDN bearer services[d]	Access to PDN services
• *Switching function* in: for TE2 terminal equipments connected to ISDN via R interface X.21, X.25	ISDN exchange	Data switching exchange
• *Numbering* for TE2	E.164 (ISDN numbering plan)	X.121 (PDN numbering plan)
• *Services and supplementary services*	I.211	X.1, X.2, X.25, X.32
• *Logical subscriber access* to: in respect of – Handling of supplementary services – charging – O & M	ISDN	Packet switched public data network (PSPDN)
• *Call establishment/clearing*	*Single-step approach*[a] – by TE2 with conventional user-network signaling for non-ISDN interface X.21, X.21bis, V.24/V.25/V.25bis – beyond TA with ISDN signaling I.451/ISUP	*Two-step approach* (port access) 1. ISDN signaling I.451/ISUP to set up a transparent B-channel connection to the PSPDN 2. X.25 virtual call control procedures between TE2 and PSPDN exchange in established ISDN B-channel connection to set up the virtual circuit
• *User-network signaling conversion* in the terminal adaptor	X.21 – I.451 for TA-X.21[a] t/r – I.451 for TA-t/r[f] V.25 (bis) – I.451 for TA-V	Not applicable
• *ISDN/PDN interworking* – for ISDN internal traffic between TE2 terminals	Not required	Needed
– Procedure	Interexchange signaling	User-network signaling as part of two-step approach[b]
in circuit switching	– CCITT Rec. X.71	– not applicable
in packed switching	– CCITT Rec. X.75	– CCITT Rec. X.25
– Implementation	*Interworking unit (IWU)* carries out interexchange signaling conversion ISDN-UP-X.71 in the case of circuit switching[c]	*ISDN access unit (AU) for the PSPDN* makes ISDN ports available

on national conditions, such as the presence of special data networks, the ISDN introduction strategy, etc.

In the case of the *ISDN bearer service solution* (Fig. 4.25a), ISDN exchanges which provide circuit switching can handle signals from TE2 terminals designed for access to circuit-switched networks; this requires no extension of the functions of the exchanges.

By contrast, connection of terminals for packet-switched networks with X.25 interface [4.4] as part of the ISDN bearer service method, i.e. the provision of ISDN virtual call and permanent virtual circuit bearer services, requires more far-reaching measures in the ISDN (cf. Sect. 4.4.4).

With the ISDN bearer service approach, the functional range of a terminal adaptor (TA) for circuit switching includes the following:

- *Mapping of call control procedures* between the interface at reference point R (e.g. conforming to CCITT Rec. X.21) and the S/T interface (CCITT Rec. I.451 [4.30]); the user only needs to carry out *one* call establishment and clearing procedure in accordance with the call control protocol in the data network, e.g. per CCITT Rec. X.21 [4.3]: *single-step call establishment* (see Sect. 4.4.2.1).
- *Bit rate adaption* between the usable data rate of the terminal (cf. CCITT Recs. X.1 [4.38] and V.5 [4.39]) and the ISDN information transfer rate of 64 kbit/s (Sect. 4.4.2.2).
- *Ready for data alignment*, i.e. the task of synchronizing the entry to and exit from the data transfer phase between the two interfaces at reference point R (e.g. per CCITT Rec. X.21), as in data networks. This is achieved with a synchronizing procedure handled in the B-channel (see Sect. 4.4.2.3).

With the *PDN access approach* (Fig. 4.25b) which is confined to packet switching, the only support that an ISDN gives to packet calls is a physical 64 kbit/s circuit-mode transparent connection between an X.25 TA or packet-mode TE1 and the appropriate ISDN port of a packet switched public data network (PSPDN); this method is therefore also known as the *port method.* In this case, although the TE2 terminals are connected physically to the ISDN, logically they remain data network users. This mainly affects subscriber numbering which conforms to CCITT Rec. X.121 [4.40] instead of I.331 [4.31] which applies to ISDN users (I.331 is identical to E.164). In addition, other network functions such as the available service (and possibly supplementary services), call charge accounting, and operation and maintenance functions, are also affected.

With the PDN access solution, data network signaling is not converted to ISDN signaling or vice versa. Instead, call establishment and clearing is always performed in two steps: ISDN user-network signaling (I.451) and ISDN interexchange signaling

Table 4.11 (contd.)
PDN Public data network
[a] Two-step approach of call establishment in the case of packet switching.
[b] Two-step approach following establishment of the circuit-switched connection to the ISDN port by means of the ISUP.
[c] Two-step approach in case of packet switching, i.e. no signaling conversion in the IWU.
[d] Only method for adaption of circuit-mode terminal equipment.
[e] For packet mode only.
[f] Analog tip/ring interface.

(ISUP) are in this case only used to establish the 64-kbit/s circuit-switched access connection between TA and data network port IP (cf. Fig. 4.25b). On completion of *physical* connection establishment (outslot signaling), another signaling procedure (conforming to the packet layer protocol of CCITT Rec. X.25) is required for setting up the *virtual* circuit in the PSPDN; as in data networks, this procedure is handled directly between TE2 and data exchanges: *two-step call establishment.* In this case X.25 call control procedures (cf. Sect. 4.4.4) are conveyed – transparently for the ISDN – via the ISDN B-channel connection already established (inslot signaling).

4.4.2 Connection of X.21 Terminal Equipment with Single-Step Call Establishment

The functions (mentioned in Sect. 4.4.1) of a terminal adaptor (TA) designed on the single-step call establishment principle for X.21 adaption in compliance with CCITT Rec. I.461 (identical to X.30) [4.6] are described in greater detail below:

- Mapping of X.21 call control procedure to the D-channel protocol (see Sect. 4.4.2.1)
- Bit rate adaption from the X.1 bit rate to 64 kbit/s (see Sect. 4.4.2.2)
- Ready for data alignment in the 64 kbit/s channel (see Sect. 4.4.2.3).

4.4.2.1 Mapping of the Call Establishment and Clearing Procedures Between the X.21 and S Interfaces

Technical details for mapping the X.21 signaling events to the corresponding D-channel signaling messages (cf. CCITT Rec. I.451 in Sect. 4.3.5) are given in Fig. 4.26. Only the following remarks need therefore be made here:

- On connection setup the individual dialed digits from the X.21 side are first collected in the TA before a call establishment message (SETUP) is sent with en-bloc dialing to the network.
- On cleardown, the DTE Clear Request per CCITT Rec. X.21 causes the TA to initiate the release procedure on the network side both in the D-channel and in the B-channel (see Sect. 4.4.2.3).

4.4.2.2 Adaption Between the X.1 User Rates of X.21 Terminal Equipment and the ISDN Information Transfer Rate of 64 kbit/s

The purpose of the bit rate adaption functions carried out by the X.21 terminal adaptor is to match the X.1 user rate of the connected X.21 terminal equipment (Table 4.12) to the B-channel rate of 64 kbit/s by adding additional and filler information.

With the exception of 48 kbit/s, a *two-stage method* (Fig. 4.27) is specified for this purpose in CCITT Rec. I.461 [4.6]:

- In *stage 1 (RA1)* an intermediate bit rate of 8 or 16 kbit/s is initially produced (Table 4.12) by creating a 40-bit frame in the terminal adaptor (TA). These 40-bit frames (Table 4.13) are exchanged between the communicating terminal adaptors in the B-channel.

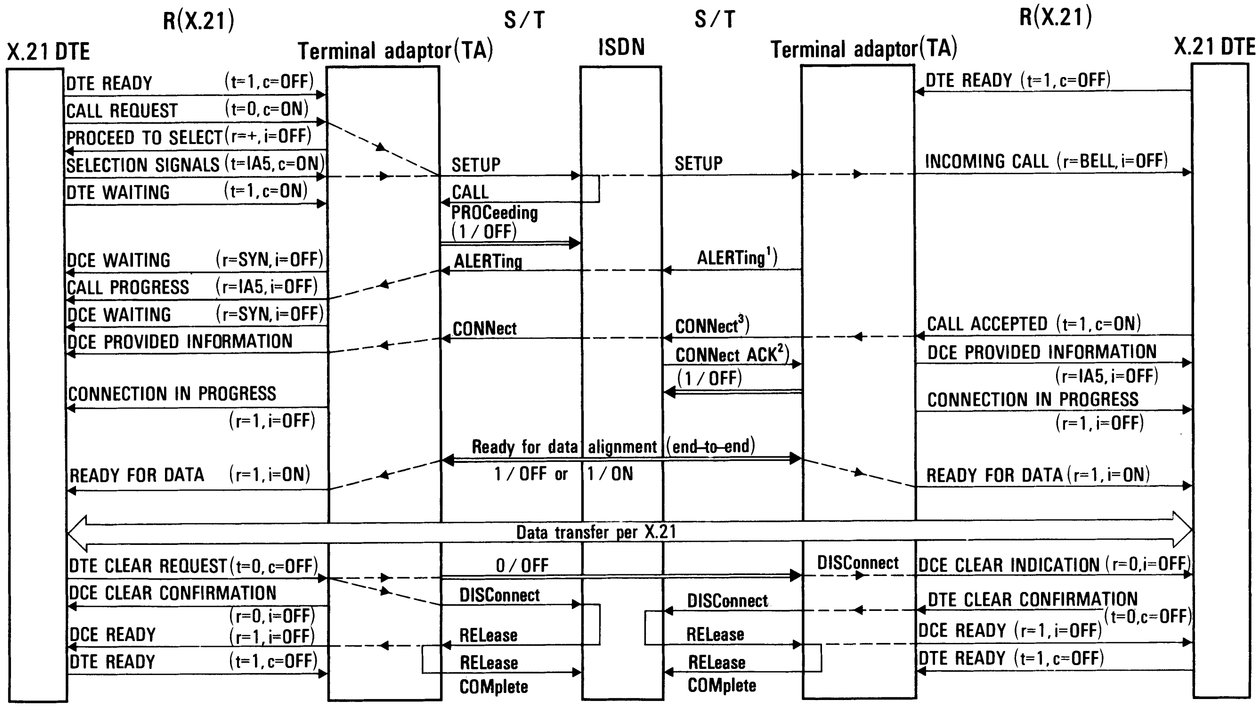


Fig. 4.26. Mapping of X.21 Signaling Events to the Signaling Messages of the D-channel Protocol in the X.21 Terminal Adaptor (TA-X.21).

1) only during manual operation; 2) if several call-compatible TA-X.21s accept an incoming call with ALERTing and/or CONNect, the network sends RELease to the TA-X.21s not selected for the call; 3) if the connected X.21-DTE is in status *controlled not ready*, *uncontrolled not ready* or *busy*, the called TA-X.21 answers the call with RELease COMPlete instead of ALERTing and/or CONNect. The idle status is reported to the calling side with a DISConnect message.

—— D-channel signaling per CCITT Rec. I.451 (cf. Sect. 4.3.5), ═══ signaling in user information channel (cf. Sect. 4.4.2.3)

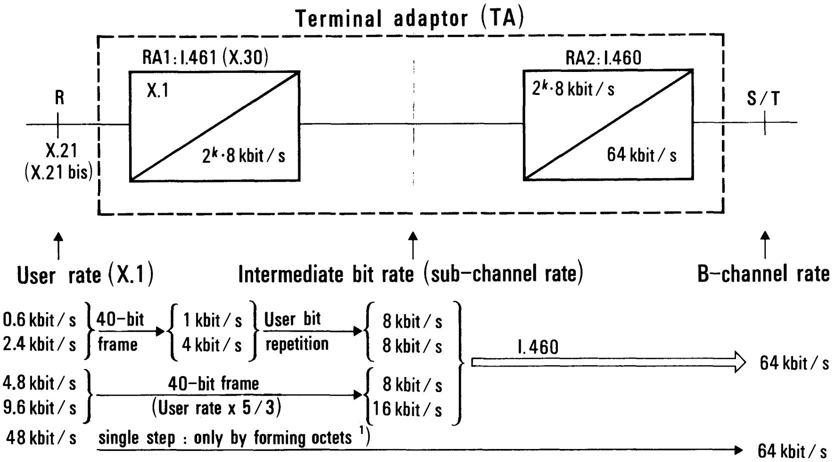


Fig. 4.27. Two-stage Bit Rate Adaptation.
1) Six user information bits + status bit + frame alignment bit

Table 4.12. X.1 User Rates and Intermediate Bit Rates as Specified in CCITT Rec. X.30

X.1 user class of service	X.1 user rate in kbit/s	Intermediate bit rate in kbit/s
3	0.6	8
4	2.4	8
5	4.8	8
6	9.6	16
7	48	64

- In *stage 2 (RA2)* the intermediate bit rate is increased to 64 kbit/s in accordance with CCITT Rec. I.460 [4.13] by setting all unused bit positions within a B-channel octet to binary "1"; in the 8 kbit/s case, this produces a 1111 111X bit sequence in the B-channel and at 16 kbit/s a 1111 11XX bit sequence.

This relatively complex two-stage method of padding out a *single* bit stream to 64 kbit/s was introduced for reasons of compatibility with *synchronous time-division multiplexing of several independent 8, 16 or 32 kbit/s bit streams*, likewise defined in CCITT Rec. I.460; this allows any combination of these subrate streams (sub-channels) in *one* B-channel (cf. the example in Fig. 4.28). Subdivision of the ISDN user access into sub-channels is initially only planned as part of an end-to-end 64 kbit/s connection between two users.

4.4.2.3 Ready for Data Alignment in the 64 kbit/s Channel Between the Terminal Adaptors and Between the X.21 Terminal Equipments

Using the 40-bit frame illustrated in Table 4.13, the terminal adaptors (TA) exchange not only user information (cf. the 8-bit groups for user information P, Q and R) but also the status information provided at the X.21 interface for call control. With the aid

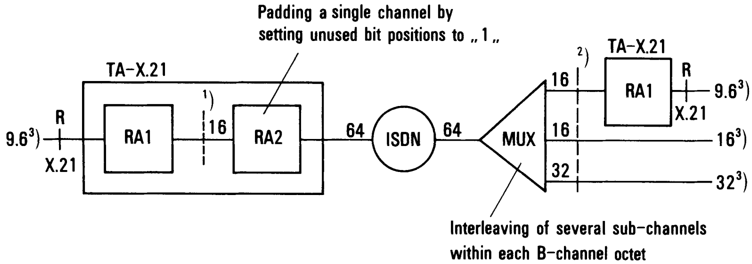


Fig. 4.28. Rate adaption of a Single Bit Stream and Multiplexing of Several Independent Substreams into a 64 kbit/s-channel. All bit rates in kbit/s.
1) Intermediate bit rate; 2) sub-channel bit rate; 3) user bit rate

Table 4.13. Structure of the 40-bit Frame for Bit Rate Adaption in Stage 1 (RA1)

	Bit no.								
	0	1	2	3	4	5	6	7	
Octet no.									
0	0	0	0	0	0	0	0	0	odd frames
	1	E1	E2	E3	E4	E5	E6	E7	even frames
1	1	P1	P2	P3	P4	P5	P6	SQ	
2	1	P7	P8	Q1	Q2	Q3	Q4	×	
3	1	Q5	Q6	Q7	Q8	R1	R2	SR	
4	1	R3	R4	R5	R6	R7	R8	SP	

P, Q, R — 8-bit groups for user information
SP, SQ, SR — Status bits
E1...E7 — These bits provide additional inslot signaling capacity between TAs, e.g. for inslot coding of the X.1 user rate
× — Used for end-to-end flow control between TAs supporting asynchronous DTEs operating at different user rates

of the status bits SQ, SR and SP, the X.21 signaling states on interface circuits t, c, r, i can be transmitted via the ISDN 64 kbit/s connection already set up between the terminal adaptors.

The X.21 interface can also be used at the ISDN with a user bit rate of 64 kbit/s corresponding to user class of service 30 (CCITT Rec. X.1). In this case, because the B-channel rate of 64 kbit/s is fully utilized for data transmission, no status information can be transferred over the 64 kbit/s channel as soon as an X.21 connection has been established. Consequently, it is not possible to use an inslot alignment method at layer 1 when leaving the data transfer phase. For interworking with directly connected ISDN terminals with S interface (TE1), the alignment procedure is also omitted when entering the data transfer phase, as ISDN terminals (TE1) perform alignment above layer 1.

4.4.3 Connection of Data Terminal Equipment with V.-Series Type Interfaces to the ISDN

Non-voice terminals with interfaces conforming to the CCITT V.-Series Recommendations for use in analog telephone networks can be adapted to the S/T interface of the ISDN user access via two fundamentally different terminal adaptors (TA) (see Table 4.10 and Fig. 4.24). Using the ISDN bearer service solution, the S/T interface can either be converted

- to the analog tip/ring interface of the telephone network (TA-t/r), or
- to a V.-series interface for data transmission in the telephone network (TA-V).

4.4.3.1 Support of the Analog Tip/Ring Interface in the ISDN

The terminal adaptor TA-t/r performs analog/digital conversion of the analog (modem) signals of the t/r interface, representing the inverse of the digital/analog conversion at the boundary between the ISDN and the analog telephone network. Besides, decadic pulsing or touch tone signaling is converted to the D-channel protocol.

The main advantage of this method is that the network interworking capabilities for the conventional telephone network required for voice communication can also be used without modification for data traffic.

4.4.3.2 Support of V.-Series Interfaces in the ISDN

The terminal adaptor TA-V specified in CCITT Rec. I.463 (identical to V.110) [4.8] performs a direct conversion of the V.24 interface signals [4.41] to the S interface using purely digital signals. Therefore, no modem is required for internal ISDN traffic. TA-V functions comprise alternate voice/data circuit switched services in conjunction with manual call control performed by a separate digital telephone set.

In addition to that, mapping functions necessary to convert automatic calling and/or automatic answering procedures of CCITT Recs. V.25 [4.42] and V.25bis [4.43] to the ISDN D-channel protocol (CCITT Rec. I.451) are also planned.

As in the case of the TA-X.21, TA-V performs

- Conversion of the electrical, mechanical, functional and procedural characteristics of the V.-series type interface to those of the ISDN S/T interface.
- Bit rate adaption of V.-series user data to 64 kbit/s. By adopting for the TA-V the 40-bit frame and the two-stage rate adaption method defined for TA-X.21 (see Sect. 4.4.2.2), terminal interworking between X.21/X.21bis/X.20bis DTEs and V.-series DTEs via the ISDN is possible (see Fig. 4.24). In contrast to synchronous V.5 bit rates, an additional asynchronous-to-synchronous conversion stage RA0 is added in the case of asynchronous V.5 bit rates (three-stage method). In RA0, incoming asynchronous data is padded by the addition of stop elements to fit the nearest channel rate defined by 2^n times 600 bit/s (see Table 4.14).
- End-to-end alignment of entry to and exit from the data transfer phase. For compatibility with X.30 (I.461), channel status information associated with V.24 interface circuits 108, 107, 105 and 109 is conveyed in-slot by making use of the status bits in the 40-bit frame (see Table 4.13).

Table 4.14. Adaption of Synchronous and Asynchronous Rates as Specified in CCITT Rec. V.110

V.5 user rate in bit/s	RAØ output rate in kbit/s	RA1 output rate in kbit/s
Synchronous:		
600	–	8[1]
1 200	–	8[1]
2 400	–	8[1]
4 800	–	8
7 200	–	16[1]
9 600	–	16
12 000	–	32
14 400	–	32[1]
19 200	–	32
48 000	–	–[2]
56 000	–	–[2]
Asynchronous:		
50	0.6	8
75	0.6	8
110	0.6	8
150	0.6	8
200	0.6	8
300	0.6	8
600	0.6	8
1 200	1.2	8
2 400	2.4	8
3 600	4.8	8
4 800	4.8	8
7 200	9.6	16
9 600	9.6	16
12 000	19.2	32
14 400	19.2	32
19 200	19.2	32

RAØ Stop bit manipulation for asynchronous to synchronous conversion
RA1 See section 4.4.2.2
1 Including user bit repetition
2 Single step adaption is used, i.e. there is no RA1 output

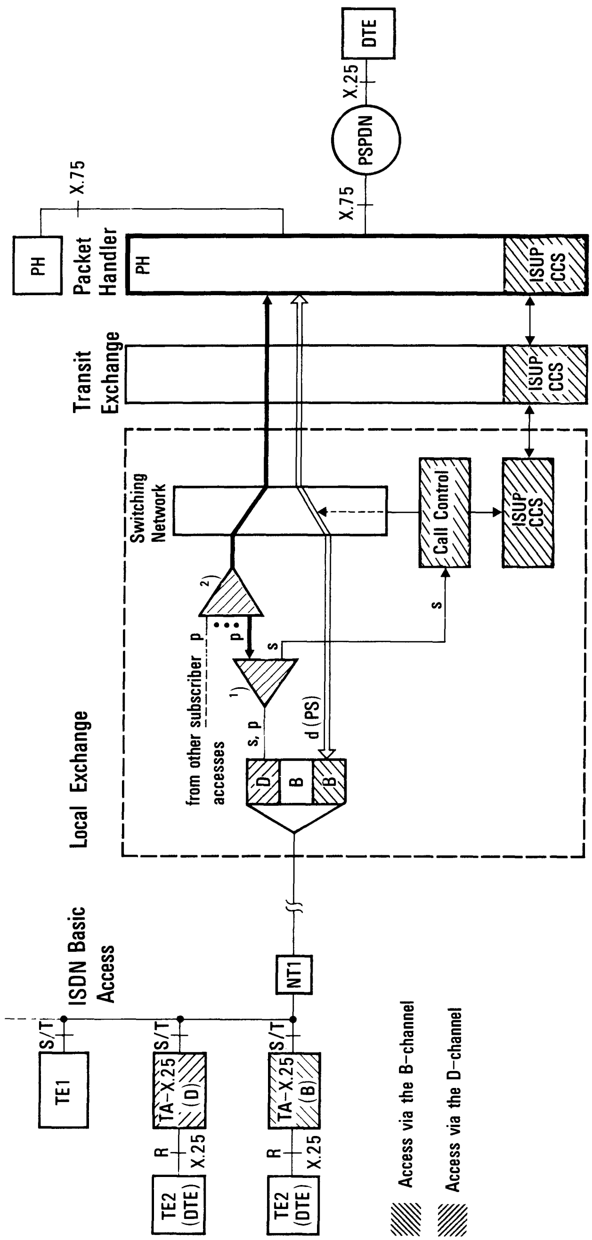
A flow control option allows the connection of asynchronous DTEs operating at different user data rates by reducing the output rate of the faster to that of the slower. Between two communicating TAs the x-bit is used to carry flow control information (see Table 4.13).

4.4.4 Connection of Terminals with X.25 Interface to the ISDN

4.4.4.1 Basic Characteristics

The integration of packet switching into the ISDN [4.44, 4.45] basically follows the two approaches presented in Sect. 4.4.1:

- With the packet-switched access to an *ISDN virtual circuit bearer service*, the packet-switching function is part of the ISDN. Virtual connections between packet



ISDN Basic Access
Local Exchange
Transit Exchange
Packet Handler
PH
X.75
X.25
PSPDN
DTE
ISUP CCS
Switching Network
Call Control
from other subscriber accesses
s, p
d (PS)
D
B
NT1
S/T
R
TE1
TA-X.25 (D)
TA-X.25 (B)
TE2 (DTE)
Access via the B-channel
Access via the D-channel

terminals connected to the ISDN are therefore handled via ISDN-internal X.25 switching devices known as *packet handlers* (PH) or packet switch service modules (PSSM). The PH functions can either be incorporated in an ISDN exchange or located remotely from the ISDN circuit-switching exchanges and assigned as centralized packet switching devices to the higher ISDN network level (Figs. 4.29 and 6.4a).

The packet handlers can be fully interconnected via permanent 64-kbit/s ISDN connections, by means of packet-switching interexchange call control and data transfer procedures in accordance with CCITT Rec. X.75 [4.36], thereby forming an overlay network for packet switching within the ISDN. A packet handler can also assume, if required, the function of a gateway to an independent packet network (PSPDN) on the basis of CCITT Rec. X.75.

- With the circuit-switched *access to PSPDN services* the packet-switching function of an independent packet network (PSPDN) is used. Virtual connections between ISDN users are thus always routed via a separate PSPDN in which the processing of the X.25 call is carried out (Fig. 4.25b).

  In this case, packet-switched services and supplementary services in accordance with specifications for PSPDNs are available to the user (see CCITT Recs. X.1 [4.38], X.2 [4.46] and X.25 [4.4]). In addition, the general procedures and principles of CCITT Rec. X.32 [4.47] apply which define the access to a PSPDN through a circuit-switched public network.

The ISDN procedures for the packet-switching service are given in CCITT Rec. I.462 (identical to X.31) [4.7]. These specifications apply not only to existing X.25 terminal

Fig. 4.29. Provision of ISDN Virtual Circuit Bearer Services by Access to Packet Handling Function within ISDN.

TE1	ISDN terminal equipment with ISDN S interface
TE2	Terminal equipment with non-ISDN R interface
DTE	Data terminal equipment
TA-X.25(B)	X.25 terminal adaptor for B-channel access to PH, i.e. X.25 link and packet layer procedures are conveyed through the B-channel
TA-X.25(D)	X.25 terminal adaptor for D-channel access to PH, i.e. X.25 packet layer procedures are conveyed through the D-channel
NT1	Network termination
PH	Packed handler within ISDN
PSPDN	Packed switched public data network
ISUP	ISDN interexchange signaling (cf. Sect. 6.3)
CCS	Common channel signaling
X.25	User interface for public packet networks [4.4]
X.75	Interexchange interface between public packet networks [4.36]
s	Signaling information (SAPI="s")
p	Packet data in the D-channel (SAPI="p")
d(PS)	Packet data in the B-channel
SAPI	Service access point identifier (cf. Sect. 4.3.4)

1) Separation of signaling information from packet data; 2) layer 2 message interleaving of packet data of different ISDN subscriber accesses.

═══ Access to the PH via the B-channel: in the network, an individual connection (i.e. one per TE2) is used (demand or permanent), ─── Access to the PH via the D-channel: in the network a permanent connection is used jointly with other TE2 terminals by message interleaving

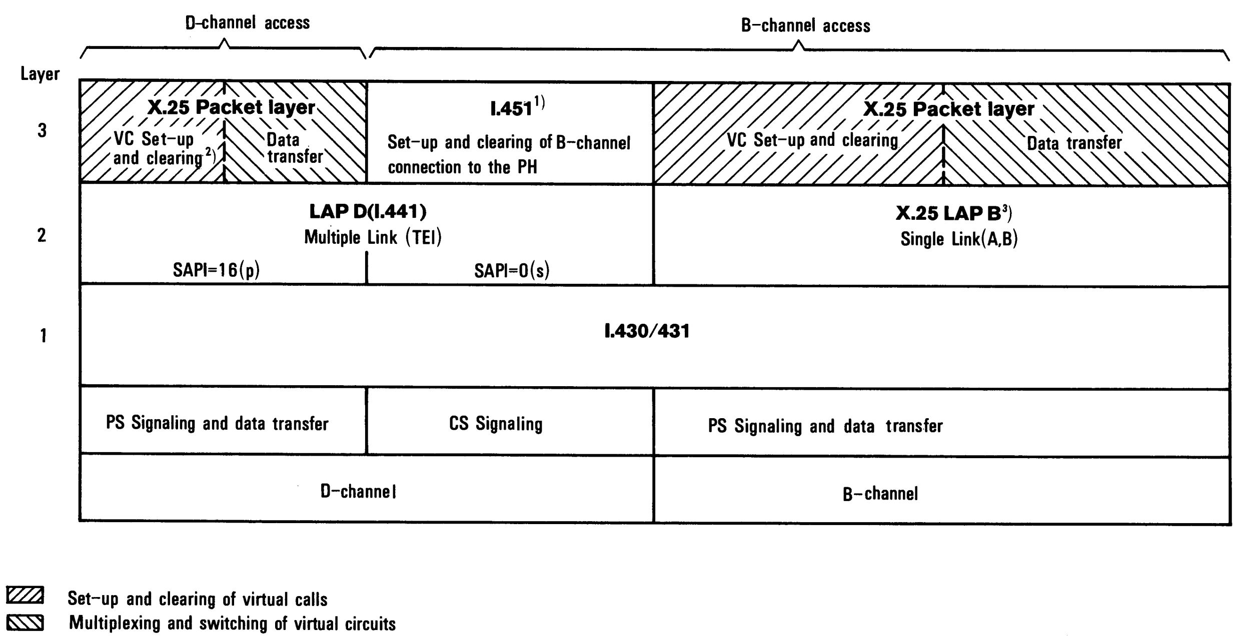


Fig. 4.30. Protocol Architecture for X.25 Based Packet Mode Services in ISDN According to CCITT Rec. I.462 (X.31).
1) Two-step call establishment: I.451 and X.25
2) At the called side: additional call offering procedure based on I.451 (cf. Sect. 4.4.4.2)
3) I.441 (LAP D) in the case of logical link multiplexing in the B-channel (cf. Fig. 4.36)

SAPI	Service Access Point Identifier	CS	Circuit switched
TEI	Terminal Endpoint Identifier	PS	Packet switched
PH	Packet Handler	VC	Virtual call

equipments TE2 [4.4], matched to the ISDN user access via a terminal adaptor (TA), but also to future directly-connected ISDN packet-mode terminals TE1 with S/T interface.

Unlike the ISDN bearer service method of connecting terminal equipments designed for circuit-switched networks (Fig. 4.25a), the ISDN virtual circuit bearer service approach is based on the principle of *two-step call establishment* (see Sect. 4.41). The two-step procedure for access to PSPDN services reproduced in Fig. 4.25b therefore also applies in principle to access via the B-channel to a packet handler (PH) in the ISDN; cf. the protocol architecture for X.25 based packet mode services in ISDN, reproduced in Fig. 4.30. However, the other characteristics such as subscriber numbering in accordance with the ISDN numbering plan (CCITT Rec. I.331 [4.31]), ISDN virtual circuit bearer services in accordance with CCCITT Rec. I.211 [4.48], subscriber administration, etc. correspond to the ISDN bearer service solution described in Sect. 4.4.1.

Even farther-reaching integration of packet switching using the approach of connection set-up in a single step, as described in Sect. 4.4.1, and utilizing a *common* out-slot signaling protocol for circuit-mode and packet-mode ISDN bearer services can be expected in a later phase of ISDN development. Such an approach is under discussion within CCITT and provides for virtual connections to be set up and cleared down by means of the extended D-channel protocol (CCITT Rec. I.451) instead of the X.25 call control procedure (cf. Sect. 4.5 and Fig. 6.4b).

With the ISDN virtual circuit bearer service method (Fig. 4.29), a packet terminal at the ISDN basic access ($B_{64}+B_{64}+D_{16}$) can either use a *B-channel* (see Sect. 4.4.4.3) or the *D-channel* (see Sect. 4.4.4.4) for end-to-end exchange of X.25 control and data packets with a packet handler. However, the network operator decides which of the access types, described in more detail in the following sections, are actually provided:

- B-channel access only (B)
- D-channel access only (D)
- both access types (B/D).

In networks which have both access types available, the type of channel actually used for the delivery of a new X.25 incoming call packet at the user access can be preset by prior agreement, e.g. in the form of customer profile information stored in the network.

However, if both access types are permitted on one ISDN access, the network must perform channel selection individually for each incoming virtual call as part of the call establishment procedure, because the ISDN does not have permanent knowledge of the current user configuration (cf. Sect. 4.3.3.3); therefore, in the case under consideration the network does not know whether terminal adaptors (TA) for B- and/or D-channel access are connected to the ISDN bus at the called side; such an adaptor may even be a combined device designed for both access types. Therefore, in the most general case, the network offers every incoming virtual call, i.e. every new X.25 incoming call packet, globally to all packet-mode TAs/TE1s at the called ISDN access by means of a special point-to-multipoint signaling procedure for packet switching known as the *call offering procedure* (cf. Sect. 4.3.5.1).

4.4.4.2 Point-to-Multipoint Signaling for Incoming Virtual Calls

With the ISDN virtual circuit bearer service method a special point-to-multipoint procedure based on the incoming call procedure for circuit-switched calls is used to offer every new incoming virtual call to all packet mode terminals present at the called ISDN access, and to determine the channel type to be used for the delivery of the X.25 incoming call packet: the *call offering procedure* already mentioned in Sect. 4.4.4.1 (Fig. 4.31). As in the case of circuit-switched call control the network selects the first user equipment which responds to the call offering message SETUP with a CONNect message. By virtue of the access possibilities for packet switching offered by the network in the SETUP message (B or D or B/D), the responding call-compatible packet-mode terminals can request, on an individual-call basis by means of the CONNect message, the channel type to be used for each incoming virtual call:

- a new (idle) B-channel to be set up for the virtual connection (*new B*) or
- a B-channel already used by the same packet-mode terminal for other virtual connections to the PH (*est*ablished *B*) or
- the D-channel (*D*).

In layer 2 of the D-channel (cf. Sect. 4.3.4) the layer 3 messages of the call offering procedure are carried in frames indicating either signaling information (service access point identifier $SAPI=s$, value 0) or packet data ($SAPI=p$, value 16). As s and p represent network-specific signaling variants, terminal adaptors intended for world-wide use must be able to handle both variants. At the time of this writing, discussions in CCITT take place with a view towards eliminating the SAPI = p option.

With *s signaling*, in the event of an *established B* or *D* channel request, the network terminates by means of RELease the signaling activity (signaling transaction) towards the packet-mode terminal selected for the virtual call, as this activity is no longer needed for the subsequent course of the virtual X.25 connection. If, however, the selected packet-mode terminal requests channel type *new B*, the s-type call offering procedure is continued as a normal incoming call procedure (as in circuit switching, cf. Sect. 4.3.5.1) with CONNect ACKnowledge (see B-channel access in Fig. 4.32). If channel type *D* is requested, the network uses the terminal endpoint identifier (TEI) specified in the CONNect message to establish an HDLC-LAPD data link with $SAPI=p$ (p link) to the packet terminal selected in the preceding call offering phase, provided such a link does not already exist for handling other virtual X.25 connections to the same terminal.

With *p signaling* the call offering procedure is generally terminated by the network with RELease. If the packet terminal selected in the call offering phase has requested channel type *new B*, the network then sets up a B-channel connection to the terminal in question in a second step by means of the normal incoming call procedure for circuit switching. If the selected terminal responds with *D* in the call offering phase, a p link is then established by the packet handler as in s signaling (cf. D-channel access in Fig. 4.33).

4.4.4.3 Access to Packet Switching via the B-Channel

For access to a packet handler in the ISDN (ISDN virtual circuit bearer service) or to an independent packet network PSPDN (circuit-switched access to PSPDN

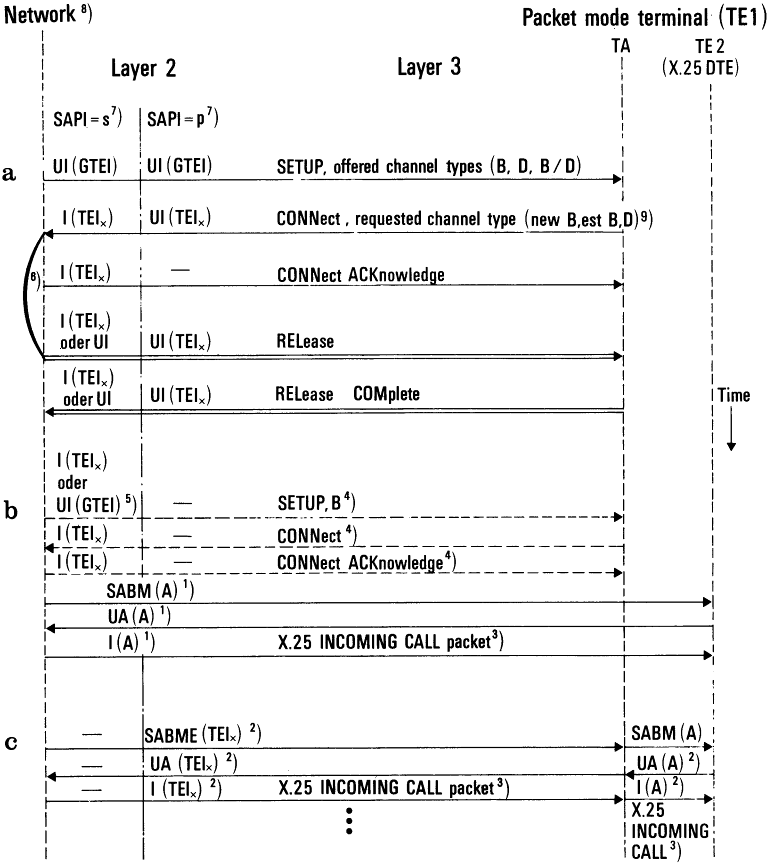
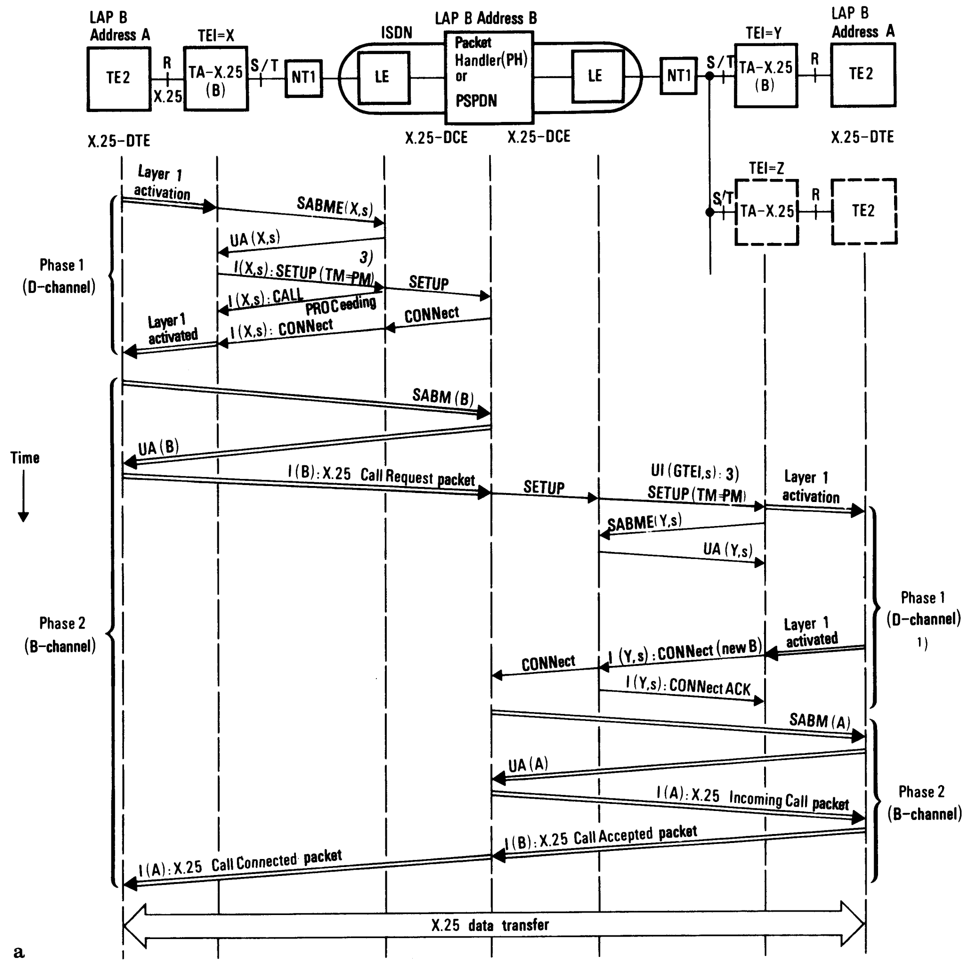


Fig. 4.31a–c. Point-to-Multipoint Signaling Procedure for Incoming Virtual Calls.
a Call offering procedure; **b** continuation *new B* (cf. Fig. 4.32); **c** continuation *D* (cf. Fig. 4.33).

TE1 ISDN terminal equipment with S/T interface
TE2 Terminal equipment with non-ISDN interface
TA Terminal adaptor
SAPI Service access point identifier (cf. Sect. 4.3.4)
GTEI Global TEI (unacknowledged information transfer to all terminal equipments, cf. Sect. 4.3.4)
TEI Terminal endpoint identifier (cf. Sect. 4.3.4)
SETUP CONNect ...: I.451 call control messages (cf. Sect. 4.3.5)
SABM, UA, UI, I: HDLC commands or responses
est ≙ established

1) The X.25 terminal adaptor for B-channel access TA(B) is transparent to X.25 LAPB;
2) the X.25 terminal adaptor for D-channel access TA(D) is transparent in layer 2 except for address conversion LAPD(TEI) – LAPB(A, B) and conversion between flag stuffing (LAPB) and all-ones (LAPD);
3) TA(B) and TA(D) are transparent to the X.25 packet level;
4) not applicable after call offering with SAPI = s;
5) specific terminal selection at the bus in layer 3;
6) SAPI = s: only if CONN contains "estB" or "D"; SAPI = p: always applicable;
7) network-specific alternatives;
8) ISDN local exchange or packet handler
9) CALL PROCeeding and ALERTing may proceed as in the case of circuit switching; a packet call is accepted by CONNect only.



a

Fig. 4.32a, b. Procedure for Access to Packet Switching Services via the B-channel: ISDN provided Packet-Mode Bearer Service and Access to PSPDN provided Data Transmission Service.

a Set-up of the first virtual call; **b** clearing of the last virtual call.

TE2	Terminal equipment with non-ISDN interface
TA	Terminal adaptor
GTEI	Global TEI (unacknowledged broadcasting to all terminal equipments; cf. Sect. 4.3.4)
TEI	Terminal endpoint identifier (cf. Sect. 4.3.4)
TM	Transfer mode
PM	Packet mode
LE	ISDN local exchange
NT1	Network termination
PSPDN	Packet switched public data network
SAPI	Service access point identifier (cf. Sect. 4.3.4)
——	D-channel signaling (outslot)
═══	Signaling in the B-channel (inslot)

SETUP, ALERTing...: ISDN I.451 call control messages (cf. Sect. 4.3.5)

SABM, DISC, UA, UI, I: HDLC commands or reponses

1) Call offering procedure using SAPI = "s" (see Sect. 4.4.4.2);

2) after clearing of the last virtual call

3) PM = "packet mode" indicates an ISDN packet-mode bearer service provided by a PH within the ISDN (case A) while TM = "circuit mode" specifies transparent access to a PSPDN service (case B).

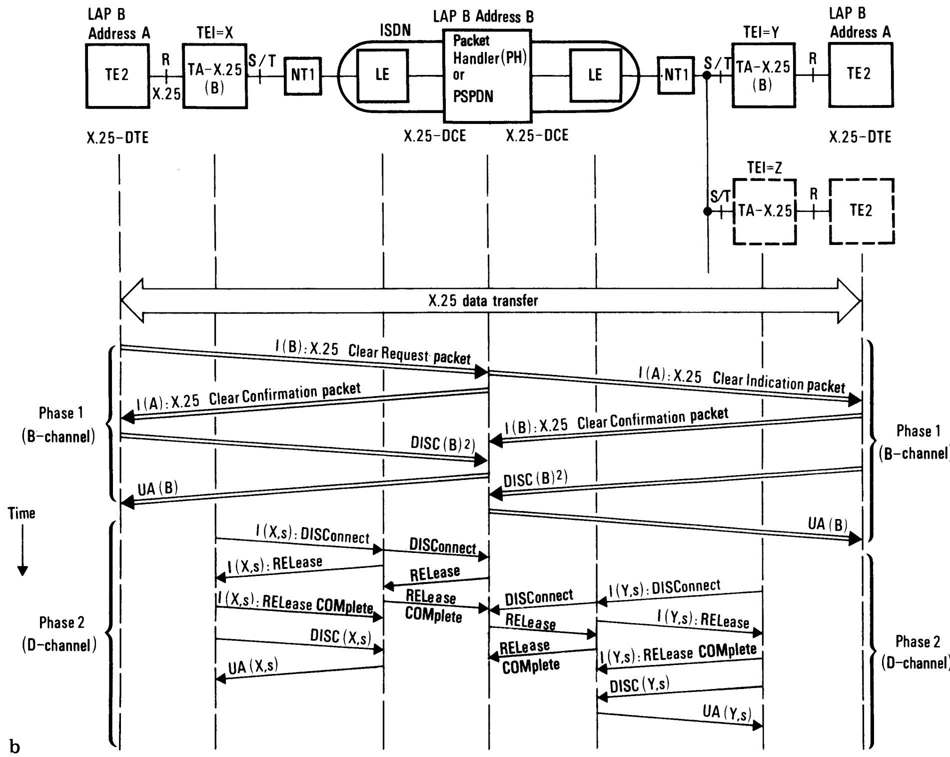


Fig. 4.32b. *Notation for D-channel frames:*
I(Y, s): ALERTing (new B)
- I.451 message (layer 3)
- SAPI
- TEI
- HDLC command (layer 2)

services), a permanent or "on demand" ISDN B-channel connection is used as the transparent feeder between TA-X.25 on the one hand and the packet handler or packet network on the other. As mentioned in Sect. 4.4.4.1, connections are set up in accordance with the two-step call establishment principle (Fig. 4.32); cf. the protocol architecture for X.25 based packet mode services in Fig. 4.30.

Except where a permanent B-channel connection is used or a virtual connection to the same packet terminal already exists, the TA-X.25 for an outgoing virtual connection must first employ the D-channel signaling procedures defined for circuit-switched calls (see Sect. 4.3) to create a 64-kbit/s connection to an input port of the packet handler or packet network: *step 1*; the X.25 terminal can cause the terminal adaptor to do this by means of appropriate activation measures in layer 1 of the X.25 interface, such as the X.21/X.21bis hotline procedure. When the B-channel connection to the packet handler or packet network is set up, the TA-X.25 puts layer 1 of the X.25 interface into the data transfer state. The X.25 protocols of layer 2 (establishment of the HDLC-LAPB data link) and 3 (virtual call) can then be handled directly between the X.25 terminal equipment and the packet handler or packet network: *step 2.*

At the *called end*, two-step call establishment basically operates in the same way, except that the initiative comes from the packet handler or packet network. The first step covers the multipoint signaling for incoming virtual calls using the SAPI = s variant (see Sect. 4.4.4.2). Finally – as previously at the calling end – the X.25 protocols in layers 2 and 3 are performed directly between the packet handler and X.25 terminal: *step 2.*

CCITT Rec. I.462 does not specify the procedure between the ISDN local exchange and the PSPDN or the packet handler. In Fig. 4.32 it is assumed that the D-channel protocol is employed for switched B-channel connections.

The *call clearing procedure* (Fig. 4.32b) also takes place in two steps in reverse order: release of the virtual connection per CCITT Rec. X.25 is followed – if it is the last virtual connection to the terminal in question – by cleardown of the B-channel connection on the basis of D-channel signaling procedures defined for circuit-switched calls (CCITT Rec. I.451).

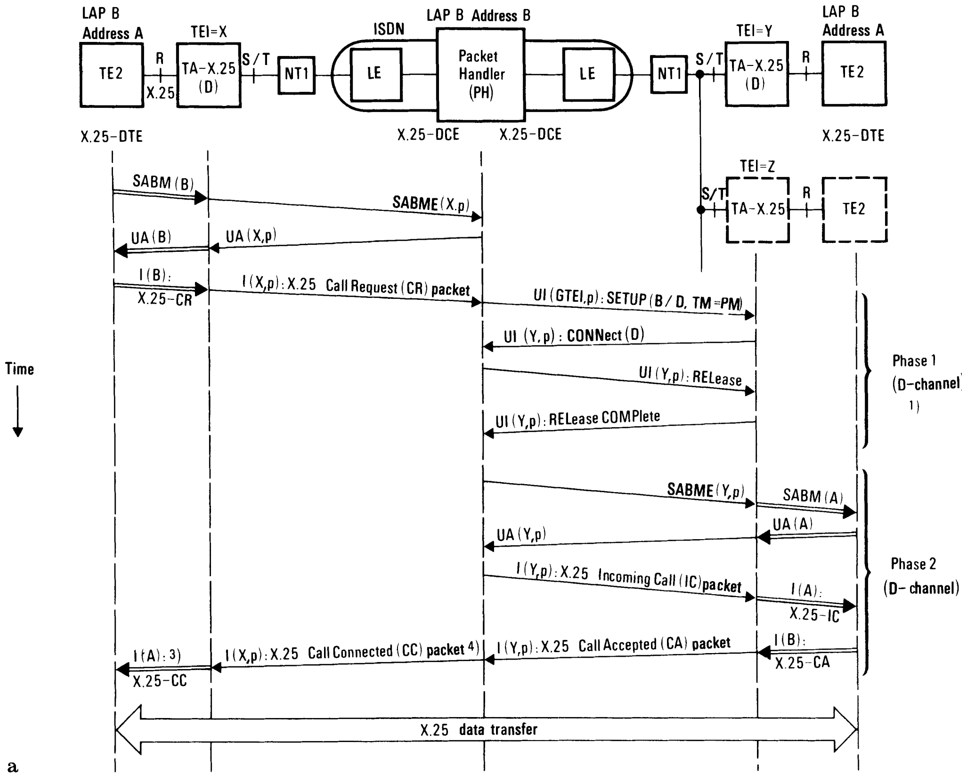
4.4.4.4 Access to Packet Switching via the Signaling Channel (D-channel)

For access to an ISDN packet handler via the D-channel (Fig. 4.33), packet data p and signaling information s share the physical transmission capacity of the D-channel in a multiplex arrangement. This is effected in layer 2 on a *message interleaving* basis; distinction is made in the HDLC address field by means of the service access point identifier $SAPI=p$ or $SAPI=s$ (cf. Sect. 4.3.4).

Unlike access via the B-channel, in this case the X.25 packets in layer 2 are only transported across the X.25 interface using the LAPB as defined in CCITT Rec. X.25; between TA-X.25 and the local exchange or packet handler, the D-channel LAPD is used. In comparison with B-channel access, utilization of the D-channel gives rise to certain restrictions in the use of the X.25 protocol, because exchange of signaling information must not be hindered: these restrictions include curtailing the length of the data field in the X.25 data packets to a maximum of 256 octets and limiting the possible packet throughput.

As the common channel signaling network (cf. Sect. 6.3) was designed for the exchange of signaling information between exchanges, in most networks it is not suitable for packet-switched data traffic. Consequently even with D-channel access on the subscriber line, permanent 64 kbit/s channels are used for the further transport of X.25 packets to the packet handler. For this purpose packet data ($SAPI=p$) in layer 2 of the D-channel must be separated from the signaling information ($SAPI=s$) in the ISDN local exchange (see Fig. 4.29). The p information is then forwarded on to the packet handler – transparently, i.e. without any processing in the ISDN exchanges through which it passes – e.g. via permanent 64 kbit/s channel connections on a message interleaving basis.

In Fig. 4.33 it is assumed that the SAPI = p variant of the call offering procedure (see Sect. 4.4.4.2) is employed at the called side. In the case of the SAPI = p variant the signaling messages of the call offering procedure (step 1) are conveyed transparently between the packet handler and the TA-X.25 over the same B-channel connection which is used for carrying the X.25 packets to and from the packet handler.

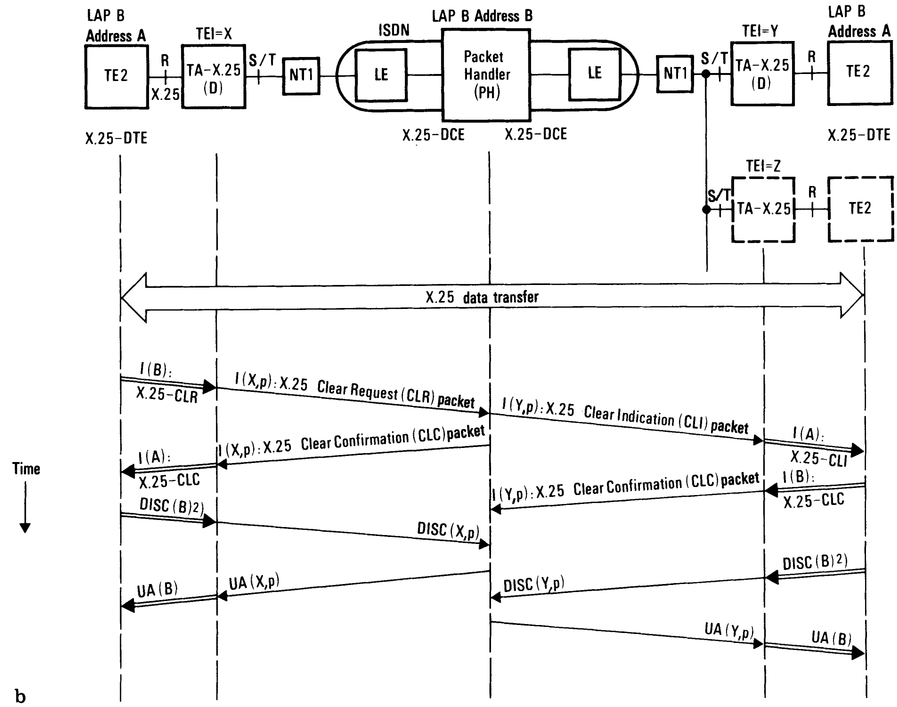


a

Fig. 4.33a, b. Procedure for Access to Packet Switching Services via the D-channel: ISDN provided Packet-Mode Bearer Service.
a Set-up of the first virtual call; **b** clearing down of the last virtual call

TE2 Terminal equipment with non-ISDN interface
TA Terminal adaptor
GTEI Global TEI (unacknowledged broadcasting to all terminal equipments)
TEI Terminal endpoint identifier (cf. Sect. 4.3.4)
SAPI Service access point identifier (cf. Sect. 4.3.4)
TM Transfer mode
PM Packet mode
LE ISDN local exchange
NT1 Network termination
SETUP, CONNect...: ISDN I.451 call control messages (cf. Sect. 4.3.5)
SABM, DISC, UA, UI, I: HDLC commands or responses
—— D-channel signaling (outslot)
═══ inslot signaling

1) Call offering procedure using SAPI = "p" (see Sect. 4.4.4.2);
2) after clearing of the last virtual connection;
3) transport of X.25 packets in layer 2 of the X.25 interface: X.25 LAPB;
4) transport of X.25 packets in layer 2 of the D-channel: LAPD with SAPI = "p" (LAPB and LAPD data links coupled in the TA)



b

Fig. 4.33b. *Notation for D-channel frames:*

UI(Y, p): CONNect (D)
- D — I.451 message (layer 3)
- p — SAPI
- Y — TEI
- UI — HDLC command (layer 2)

I(Y, p): X.25 call accepted packet
- X.25 call accepted packet — X.25 packet (layer 3)
- p — SAPI
- Y — TEI
- I — HDLC command (layer 2)

4.5 Additional Packet Switching Techniques in ISDN

ISDN packet switching based on the conventional X.25 protocols in accordance with CCITT Recommendation I.462 (see Sect. 4.4.4) will be the standard method adopted for packet mode services in the first phase of the ISDN. It seems reasonable, however, that the consistent separation between signaling (control plane) on the one hand and the transfer of user information (user plane) on the other which is practised for circuit-mode services in the ISDN should be extended to packet-mode services in ISDN. By applying the principle of outslot signaling to packet switching it is possible to dispense with the special signaling protocols (see CCITT Recommendation X.25 [4.4] and X.75 [4.36]) and special signaling equipment that have hitherto been necessary. More important than this, however, is the possibility of establishing *common* call control for all types of bearer services, including uniform handling of supplementary services, irrespective of the special equipment which will continue to be required for

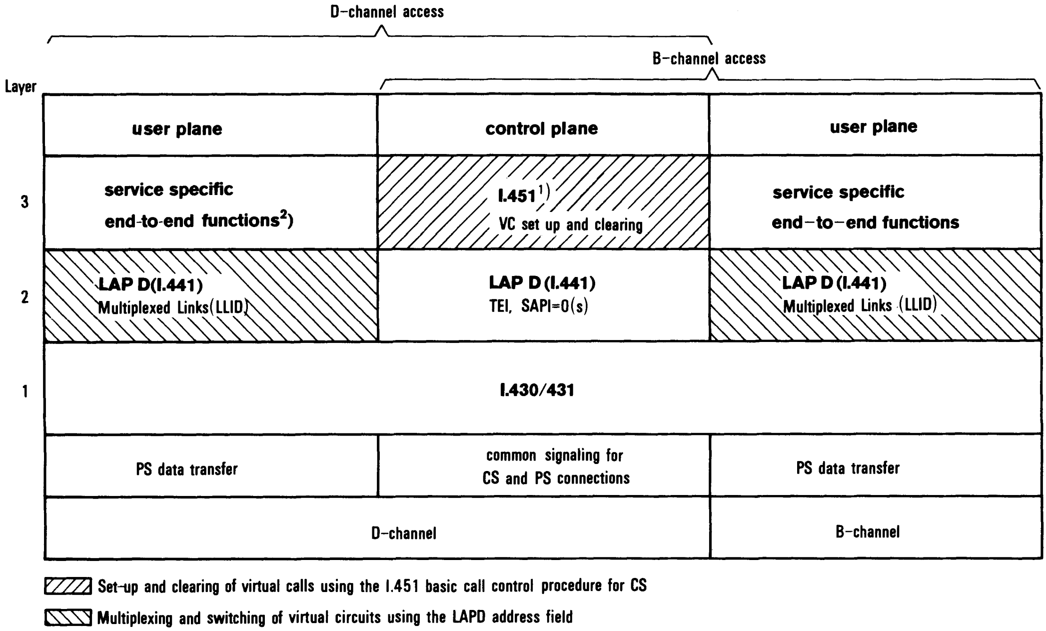


Fig. 4.34. Protocol Architecture at the User-Network Interface for Additional Packet Mode Bearer Services.
1) Single-step call establishment and clearing using I.451 messages with suitable extensions, e.g. LLID information element in the SETUP, CALL PROCeeding and CONNect messages.
2) E.g. data transfer functions of the X.25 packet level protocol.
SAPI Service access point identifier
TEI Terminal endpoint identifier
LLID Logical link identifier (equivalent to the X.25 logical channel number)
CS Circuit switched
PS Packet switched
VC Virtual call

connection control (see Sect. 6.1). In the CCITT, therefore, discussions have started on *additional packet mode bearer services in ISDN.* These new services would be used for data communications at "narrowband" rates up to 2 Mbit/s over D, B, H0 and H1 channels. The objective is to complete a framework recommendation on the overall protocol architecture and the potential bearer services by 1988 [4.55] and to finalize the work during the 1989–92 study period of CCITT.

A comparison of the protocol architectures in ISDN for additional ISDN packet mode bearer services (Fig. 4.34) and conventional X.25 packet switching (Fig. 4.30) shows that the two most important differences are as follows [4.49]:

- Signaling for *setting up and clearing down virtual connections* is based on the *D channel protocol* for circuit-switched connections, i.e. use of *outslot signaling with single-step call establishment* in accordance with I.451 (see Sect. 4.3.5) instead of the X.25 packet layer call control procedure. In this case, the SETUP message is used even if an additional virtual connection is to be set up via an already active physical channel (such as e.g. a B-channel).

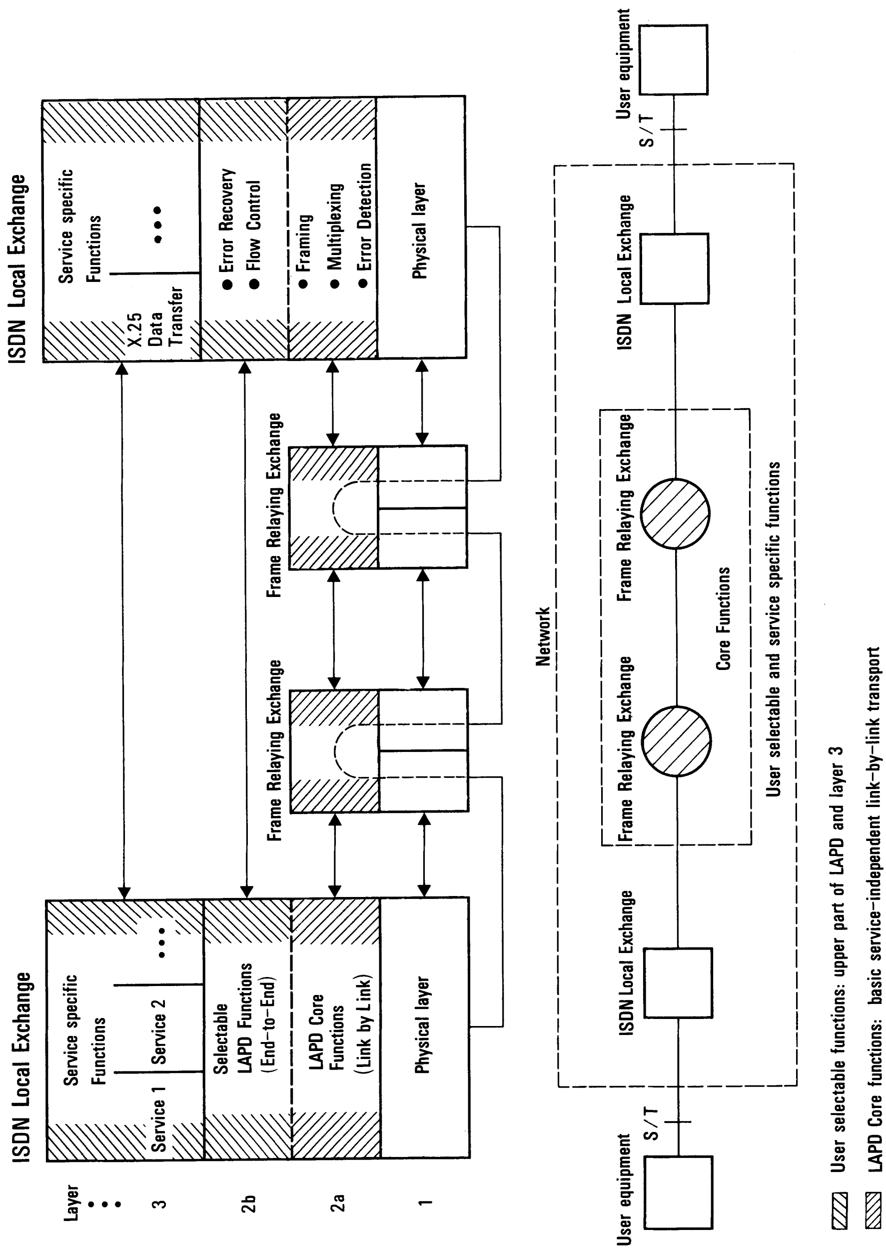


Fig. 4.35. User Plane Functions and Network Architektukture for Additional Packet Mode Bearer Services

- In contrast to X.25, *multiplexing and switching of virtual connections* take place in *layer 2*, with LAP D (I.441, see Sect. 4.3.4) being used even in the B and H channels instead of the X.25 LAP B. In *frame multiplexing* according to the multiple link principle, the *Logical Link Identifier (LLID)* in the LAP D address field (instead of TEI and SAPI) assumes the function of the logical channel number of X.25; like that number, it has local significance only.

Regarding the *control plane*, it is likely that the existing I.451 basic call control procedures for circuit-switched connections can be retained to a very large extent for logical connections as well. On the other hand, information elements specific to packet switching have to be added to the existing I.451 signaling messages, e.g. inclusion of the logical link identifier LLID in the SETUP, CALL PROCeeding and CONNect messages.

As for the *user plane*, the LAP D protocol may not be fully exploited in the network in contrast to X.25. Instead, the LAP D functionality which is terminated in every exchange may be restricted to a basic LAP D subset – see *core functions* in sublayer 2a of Fig. 4.35. These core functions result in a low-cost service-independent frame relaying mechanism on a link-by-link basis which is characterized by a comparatively low overhead and can be used by various applications. The core functions are

- frame delimiting, alignment, and transparency
- frame multiplexing/demultiplexing using the address field
- detection of transmission errors

By supplementing the common core transport functions with *user selectable and service specific protocol functions* above the frame relaying mechanism it will be possible to support different applications. Typically, these additional protocol functions will be performed on an end-to-end basis – either between users or between local exchanges or network gateways.

The network may terminate additional protocol functions as requested by the user. Depending on the degree of protocol termination in the network, the following additional packet mode bearer services can be derived:

- The *frame relaying* bearer service provides only the *LAP D core functions* in at least one point in the network. A frame received is discarded if e.g. the frame checking sequence is in error or if it cannot be routed further due to internal network conditions.
- The *frame switching* bearer service provides *full LAP D functions* in at least one network node.
- The *X.25 based* additional packet mode service provides in addition to full LAP D functions the X.25 data transfer protocol in at least one network node.

CCITT considered the use of frame relaying as an access arrangement to an X.25 packet handler (see Sect. 4.4.4.3) via the B channel as an enhancement to Recommendation I.462: LAP D would be used in the B channel while retaining the X.25 protocol in Layer 3. However, it was decided not to include frame relaying access in the 1988 version of I.462 and to further study this subject in the next study period (1989–1992). As shown in Fig. 4.36, the function of a frame handler in the network would be to multiplex the incoming LAP D connections from various subscriber lines into a common physical connection to the packet handler; this

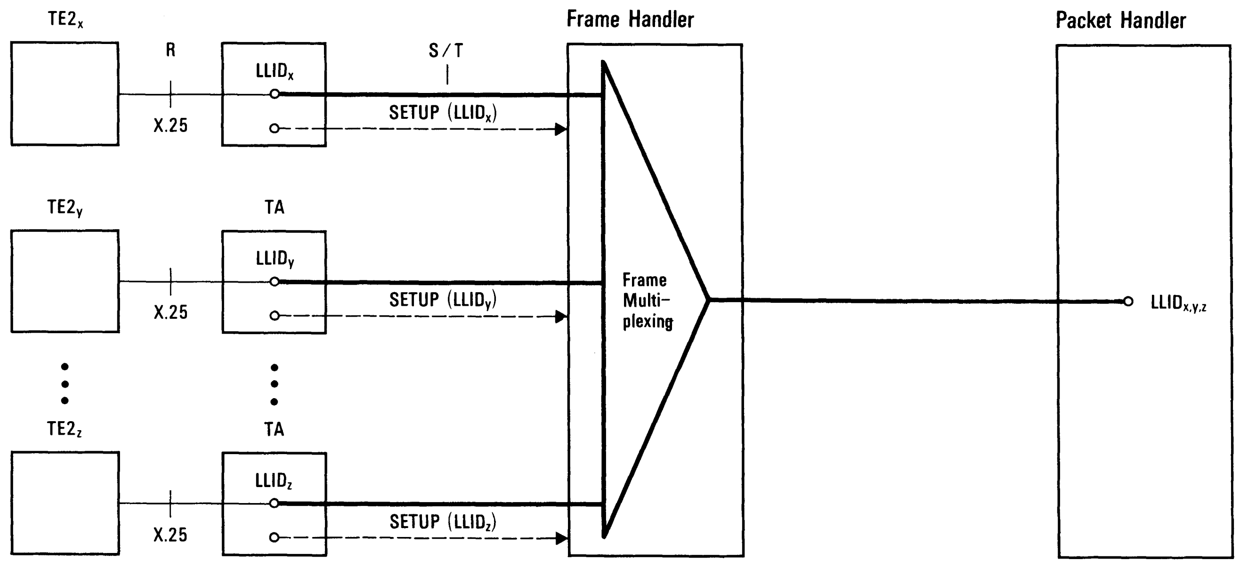
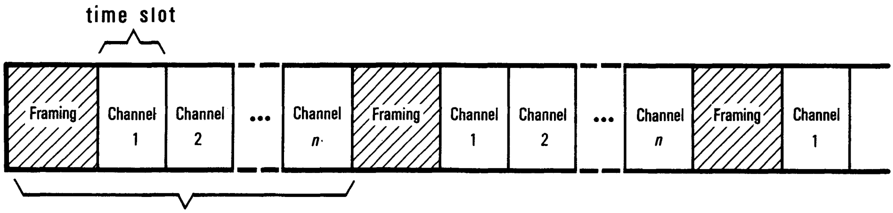
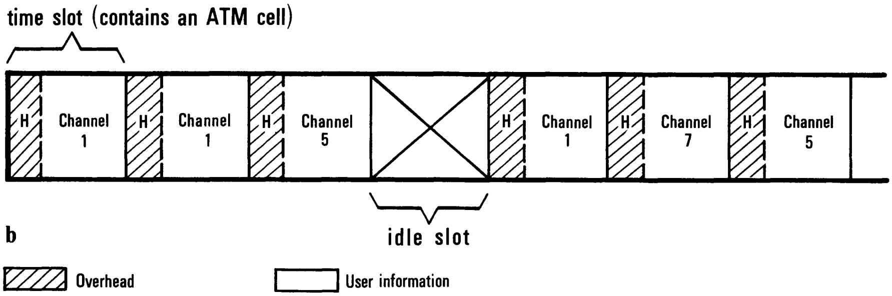


Fig. 4.36. Extension of CCITT Rec. I.462 for Logical Link Multiplexing on the B-Channel.

LLID	Logical link identifier
TA	Terminal adaptor
TE2	Terminal equipment with non-ISDN interface
———	LAP D logical link (I.441)
– – – –	Signaling (I.451)



a



b

Fig. 4.37a, b. Synchronous versus Asynchronous Time Division.
a Synchronous time division multiplexing (STM); **b** asynchronous time division multiplexing (ATM) with fixed packet length (the channel numbering contains an arbitrary example).
H Header (logical channel number)

would result in a more efficient use of the connection to the packet handler by a number of X.25 terminals.

The primary field of applications for the additional packet mode bearer services described above is data communication up to 2 Mbit/s, e.g. interconnection of local area networks (see Sect. 3.6.2) via ISDN. However, the application of packet switched techniques is being discussed in the CCITT for bit rates around 135 Mbit/s in conjunction with Broadband ISDN as well (see Sect. 6.1) under the term asynchronous transfer mode (ATM) – also referred to as fast packet switching (FPS) or dynamic time division multiplexing (DTDM) [4.50]. Even though ATM uses a similar control plane architecture for setting up and clearing down virtual connections as the additional packet mode bearer services of the narrowband ISDN, the transfer of user information can be expected to be different. This is due to the fact that ATM is regarded as a technique intermediate between circuit and packet switching by combining their advantages, i.e. ability to support real time services as a result of improved time transparency and flexible bandwidth resulting from statistical multiplexing.

An important ATM feature is flexible bit rate allocation by decoupling the bit rate requirements of user services from the bit rate of the physical access. In this way a large variety of narrowband and broadband services and combinations thereof can be supported over a single broadband user-network interface without requiring a corresponding proliferation of physical channel types and interface structures.

If the requested bandwidth is allocated to a call on a guaranteed reservation basis, ATM can also support continuous bitstream oriented traffic besides bursty traffic:

deterministic instead of probabilistic ATM. Therefore, at least the entire payload capacity of the broadband user-network interface may be structured into "packets" called ATM cells. In addition to providing a flexible multiplexing technique for the user-network interface, ATM may in the longer term substitute conventional packet and circuit-switching as a universal network concept. ATM is the target solution for implementing Broadband ISDN [4.56]. In the foreseeable future, however, hybrid solutions will dominate that can support coexisting STM (synchronous time division) and ATM based services [4.51, 4.52, 4.53].

While frame relaying uses variable size HDLC frames in layer 2, CCITT has adopted a fixed length approach for ATM called asynchronous time division multiplexing ATD (see Fig. 4.37 and [4.54]). Reduced complexity and less variation in delay may be regarded as the fundamental advantages of fixed cell size enabling universal applicability to a wide range of different services.

In contrast to frame relaying ATD cells start and end at the boundaries of fixed length synchronous time slots in layer 1. The essential difference to STM (circuit switching) is that the time slots are used in an asynchronous manner by the various logical channels that are identified by a logical channel number within the header of the time slot.

5 ISDN Terminals

5.1 Preliminary Remarks

The services of a communication network are accessed via the terminal equipments connected to the user-network interfaces (see Sect. 4.2). The term *terminal equipment* covers terminals for interpersonal communication (e.g. the telephone), terminals for communication between user and data processing systems (e.g. the VDU), and also data processing systems themselves. As the integration of services does not give rise to any special considerations for data processing systems, this chapter therefore deals solely with terminals enabling the human user to gain access to ISDN services.

Terminal equipments are not normally regarded as component parts of a communication network, at least from an administrative or legal standpoint. The network usually ends at the user-network interface, as the term *network termination* (see Sect. 4.1.1) implies. Nor are terminals always component parts of a communication service (cf. Sect. 2.1). Nevertheless, the potential for service integration in the ISDN provided by the service-independent, universal user-network interface (see Sect. 4.2.2) will have considerable effects on the evolution of terminals.

The ISDN-generated impetus for terminal innovation is mainly directed at the terminal functions required for exploiting the new ISDN capabilities. However, ISDN terminals will also exhibit other advanced features.

An important type of terminal made possible by the ISDN is the *multiservice terminal* (often called a "multifunction terminal"). The multichannel ISDN access using a common directory number for all channels and with automatic, service-specific terminal selection (see Sect. 4.3.3.2) enables more than one information type to be used alternately or simultaneously for communication both in the office and at home. In many cases this type of multiple communication is only practicable with multiservice terminals (cf. Sect. 5.4).

Due to the expanded range of functions of the new ISDN terminals, every effort must be made to ensure that the terminals are easy to use with all their functional complexity.

Even the terminals available today have many different user interfaces for information input, information output and operation.

Telephones, for example, use different numbers of keys assigned in different ways for the input of signaling information as well as different tone generators and displays for output of that information. With the development of new terminals, especially multiservice terminals, this multiplicity of user interfaces may increase still further.

In order to avoid burdening the public network with this increasing diversity of user interfaces, and also to ensure that the development of new user interfaces is not held back by the public network, the signaling events at the ISDN user-network

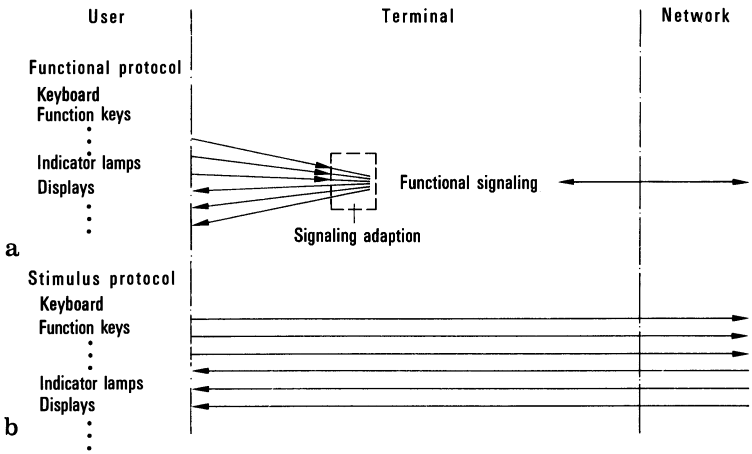


Fig. 5.1a, b. Man-machine Interface at a Terminal and User-Network Interface in the ISDN. **a** With the functional protocol, **b** with the stimulus protocol

interface are described as functionally as possible, in other words as independently as possible from the concrete physical input/output units of a specific terminal.

According to the *functional protocol* concept (see Sects. 4.3.5.1 to 4.3.5.5), it is the terminal itself which makes the conversions between the events at the man-machine interface and the functional signaling events at the user-network interface (signaling adaption, Fig. 5.1a). This allows each terminal to provide the interface best suited to its user without individual requirements on the network.

In some national networks it is intended to use (at least in an interim phase) a "stimulus protocol" (cf. Sect. 4.3.5.6) in which the man-machine interface is controlled directly by the network. This may be applied in connection with terminals that are used in large numbers (especially telephones). The terminal reports events at the man-machine interface, such as keyboard inputs, directly to the network as "stimuli" without processing in the terminal (Fig. 5.1b). The stimulus mode terminal has greater flexibility because features can be introduced or changed centrally by the network without modification to the terminal. However, for this purpose program-specific and equipment-specific data must be maintained in the network for each terminal type. CCITT Recommendations [5.1] exist for the functional protocol and the basic concepts of the stimulus protocol.

5.2 Basic Features of an ISDN Terminal

An described in Sect. 4.2.2, with the basic access the ISDN offers the user an interface with bus capability permitting the operation of several terminals or of multiservice terminals at a *single* user access via parallel-connected sockets (Fig. 5.2).

The functions which the terminals must incorporate to enable them to be operated at this user-network interface are described below. Fig. 5.3 shows the connection unit required in the terminal for the ISDN basic access.

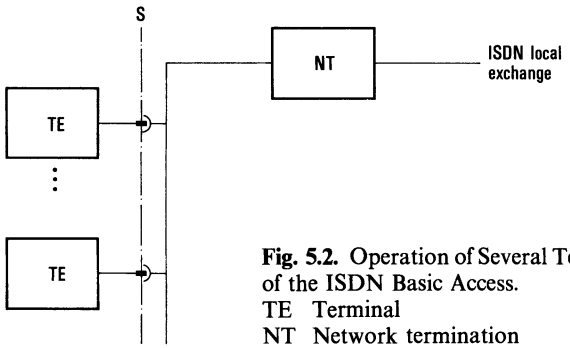


Fig. 5.2. Operation of Several Terminals at User-Network Interface S of the ISDN Basic Access.
TE Terminal
NT Network termination

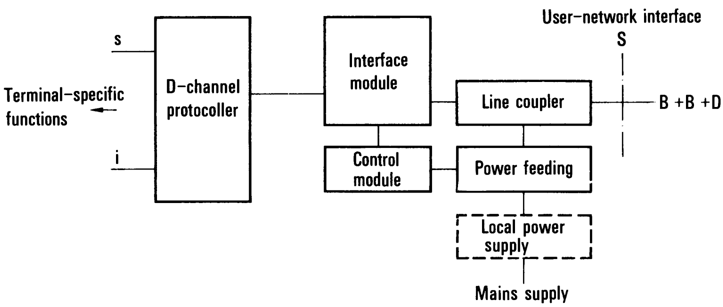


Fig. 5.3. Connection Unit in the Terminal for the ISDN Basic Access.
s Signaling data
i User information

The *line coupler* module provides the electrical connection to the four wires of the interface, e.g. using transformers. It is also responsible for extracting the feeding current supplied across the interface. The *power feeding* module conditions the current from the network termination NT or – if the local AC mains power voltage fails – from the exchange (this implies e.g. voltage stabilization). Where comparatively high power is required (e.g. for operating VDUs), this module enables the terminal to be connected to the AC mains supply.

The *control* module contains the functions necessary for controlling the user-network interface e.g. for initializing its operating status and for access to the common signaling channel if the terminal is operated on a passive bus (cf. Sect. 4.3.3.3). The *interface module* is primarily responsible for generating the multiplex structure (B+B+D) at the interface. The *D-channel protocoller* handles signaling with the exchange using the protocol provided for the D-channel, layers 2 and 3 (see Sect. 4.3).

The above summarizes the functions generally required for connecting a terminal to the ISDN. Additional functions are necessary depending on the specific task of each terminal; these are explained in the following sections.

5.3 Single-Service Terminals Connected to the ISDN

This section describes terminals which provide communication only in one information type, e.g. telephony only or text/data communication only.

5.3.1 ISDN Telephone

The single-service terminal which will probably see the largest number of changes due to the influence of the ISDN, in terms of both user functions and appearance, is the telephone [5.2, 5.3]. ISDN telephones can have different levels of sophistication.

Figure 5.4 shows the basic structure of the ISDN telephone. An important feature is the *alphanumeric display* (e.g. one row with 16 positions) which can be used to indicate directory numbers and other information (cf. Sect. 2.3.3). The *keypad* contains the pushbuttons for connection set-up and function keys for activating supplementary services of the ISDN telephone service.

A basic telephone will only have a small number of function keys for selected supplementary services. The substantially larger number of function keys on advanced telephones can be arranged in blocks according to function: one block of name keys, a second containing the keys for specific call establishment functions such as call forwarding/call diversion and completion of calls to busy subscribers. To a third block can be assigned supplementary services providing the subscriber with information, such as registration of incoming calls and *advice of charge* (cf. Sect. 2.3.3, Table 2.2). A further facility on advanced telephones might be a card reader for identifying the user for different purposes such as assigning charges to individual users of the line, or call diversion initiated at a distant telephone.

An incoming call is indicated acoustically with *VF ringing*.

The *voice unit* converts the speech signals from analog to digital form and vice versa (PCM encoding/decoding; in some cases using other codes such as ADPCM

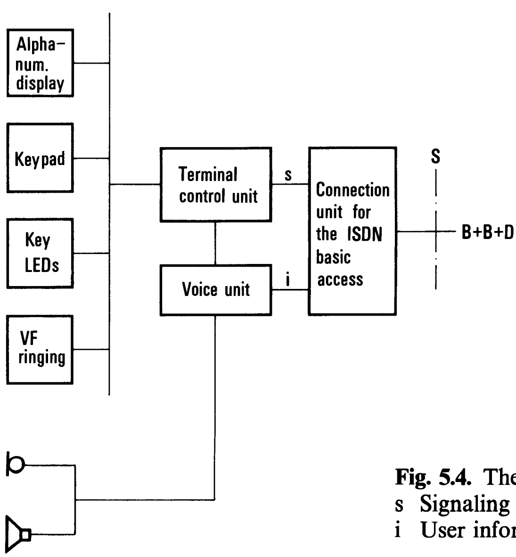


Fig. 5.4. The ISDN Telephone.
s Signaling data
i User information

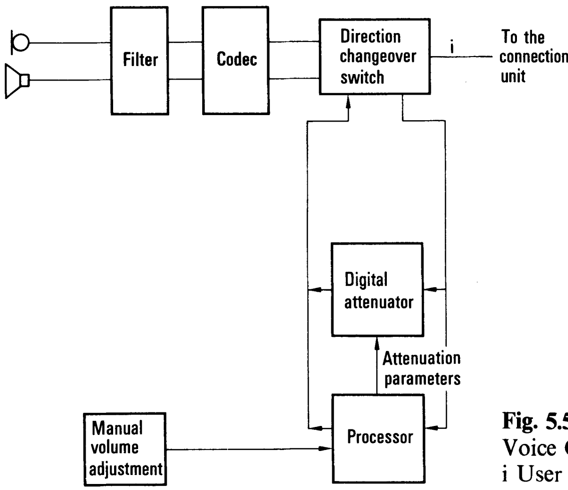


Fig. 5.5. Handsfree Equipment with Digital Voice Control.
i User information

(Sect. 7.2.1) in order to obtain enhanced transmission quality). The *terminal control* contains the scanning function for the keys and the controller for the telephone's alphanumeric display. B-channel selection also takes place here as initiated by the exchange (see Sect. 4.3.5).

Telephoning without a handset – *handsfree talking* [5.4] – leaves the speaker's hands free during a call and also allows active participation by other persons. In the analog telephone network, the imperfect attenuation of the hybrid circuit responsible for separating the transmission directions of the speech signals causes the outgoing signal to be fed back via the telephone loudspeaker. This can cause "singing"; to prevent that effect, the transmission direction inactive at any one time must be artificially attenuated and, at the same time, in order to obtain an adequate receive volume, the other direction must be amplified depending on the line attenuation. With analog handsfree speaking, the effect of this voice-controlled operation may be so great that the impression of "half-duplex" communication is produced. With the all-digital connections in the ISDN with separate go and return paths, there is no electrical coupling at a hybrid. However, unwanted feedback may be produced by acoustic coupling at the distant end; for this reason voice control is also necessary for digital connections, though in reduced form only, due to the lack of attenuation inherent in digital transmission. Consequently, normal duplex operation is largely maintained.

With digital handsfree talking, all control functions are performed digitally: processing and attenuation of the digitized voice signal, speech direction recognition, automatic volume control to the level set by the user, and selection of the different operational states: handsfree speaking/open listening/handset.

Figure 5.5 contains a block diagram showing the handsfree equipment with digital voice control. The digital attenuator provides the digital code words of attenuated PCM-encoded speech for both directions. The code words are derived from the original code words in accordance with the intended attenuation. The processor determines the necessary attenuation values by comparing the relative volumes in both transmission directions.

The use of digital control technology has advantages for the manufacture of handsfree talking equipment (compactness by use of integrated circuits) and further improves the quality of handsfree operation through rapid control response and through precise counterbalancing of the transmit and receive channel attenuation.

5.3.2 Terminals for Non-Voice Communication in the ISDN

For the user, the most important new features of terminals for text and data communication as well as facsimile transmission [5.5] in the ISDN is rapid information transfer on the B-channel.

As with the ISDN telephone, the ISDN supports communication activities (call establishment, etc.) for the user of terminals for text and data services. For example, these terminals can also have function keys and an alphanumeric display like the ISDN telephone, in addition to their screen. However, not all the ISDN supplementary services for call establishment, etc. are provided on non-voice terminals, as not all the supplementary services are applicable (cf. Table 2.2).

Figures 5.6 and 5.7 show the basic structure of ISDN terminals for text and facsimile communication.

A typical ISDN terminal for text and image communication is the "mixed mode" terminal for the ISDN textfax service (see Sect. 2.3.1.2), as it is only the high transmission rate of 64 kbit/s and the correspondingly short transmission time for the facsimile parts of documents that make effective communication with text and image possible.

Admittedly, the cost of the high-resolution screen with large refresh memory required to display an A4 facsimile with sufficient sharpness, together with the scanning

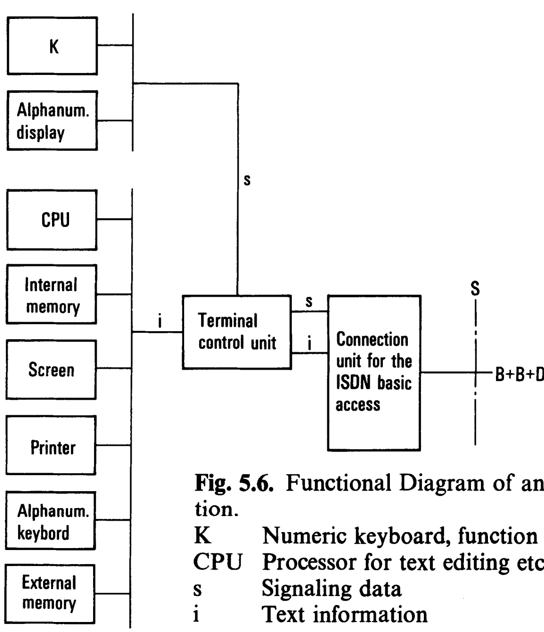


Fig. 5.6. Functional Diagram of an ISDN Terminal for Text Communication.
K Numeric keyboard, function keys for connection setup
CPU Processor for text editing etc.
s Signaling data
i Text information

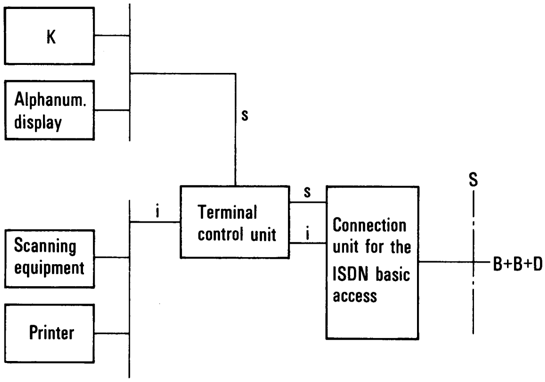


Fig. 5.7. Functional Diagram of an ISDN Terminal for Facsimile Communication.
K Numeric keyboard, function keys for connection setup
s Signaling data
i Facsimile information

and printing equipment, is not inconsiderable. Figure 5.8 shows the basic structure of an ISDN mixed mode terminal. At its highest level of sophistication, a terminal of this type can read existing documents in such a way that the textual material is represented in alphanumeric code (for which automatic character recognition is required) and only the pictorial material appears as facsimile, if appropriate in vector graphics. Thus the volume of information to be stored and transmitted can be reduced to a minimum by using the most economical encoding method in each case.

On-screen text editing is carried out using normal text editing software, as the text is available in character representation. The printer is designed to output both text and

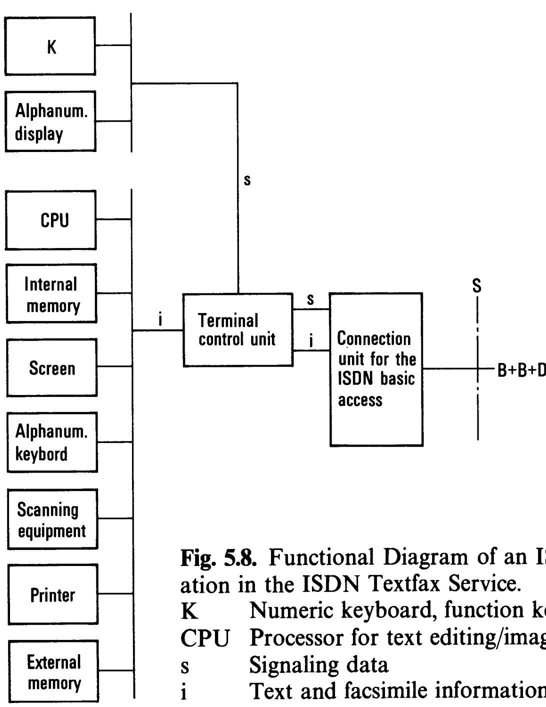


Fig. 5.8. Functional Diagram of an ISDN Terminal for Mixed Mode Operation in the ISDN Textfax Service.
K Numeric keyboard, function keys for connection setup
CPU Processor for text editing/image processing
s Signaling data
i Text and facsimile information

facsimiles. Like all text editing equipment, the terminal is equipped with an external memory (e.g. floppy disk).

In order to allow unrestricted data communication worldwide, the difference between the digital systems (for an interim period, 56 kbit/s basic channel only usable for non-voice e.g. in the USA, as opposed to 64 kbit/s e.g. in Europe) must be overcome. This may have implications for terminal equipment.

5.4 Multiservice Terminals

Multiservice terminals are equipments which have communication capability in more than one information type, either alternately or simultaneously [5.3]. In principle this is also possible with several dedicated terminals; however, the integration of several information types in one unit has decisive advantages:

- A multiservice terminal occupies much less space than the corresponding number of single-service terminals.
- Change of service during a call and simultaneous communication in two services are easier (this includes voice annotations to text and image).
- Only one set of call-establishment equipment is required.
- With a multiservice terminal for voice and data, telephone communication can be improved by the screen being used to display not only data but also a locally stored telephone directory.

An initial example of a multiservice terminal is an integrated voice and data terminal (IVDT) which can be used for ISDN telephony and videotex alternately, or even simultaneously, thanks to the two B-channels in the ISDN basic access. It is equally suitable for use in the office and in the home. Figure 5.9 shows the basic design of this IVDT. Telephoning is enhanced by useful features such as registration of incoming calls, personal telephone directory in the internal memory and user prompting [5.6, 5.7], the screen being used for information output in connection with these features. A particularly effective support facility for telephone traffic is automatic initiation of call set-up where the cursor is used to mark the desired directory number in the directory

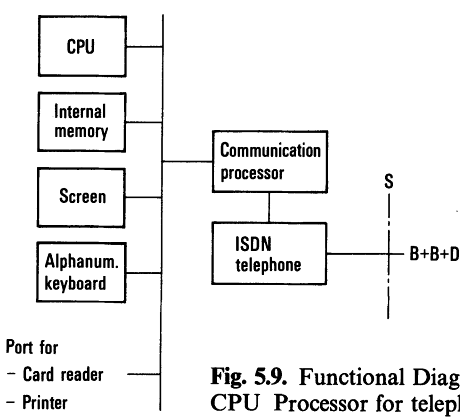


Fig. 5.9. Functional Diagram of an Integrated Voice and Data Terminal.
CPU Processor for telephony support functions

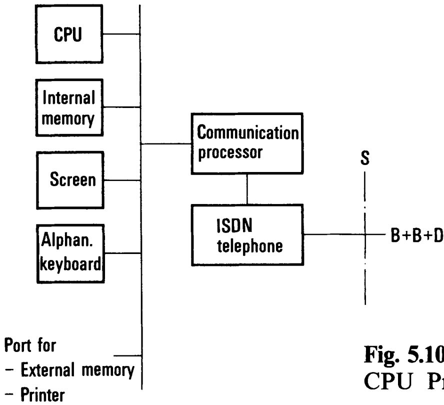


Fig. 5.10. Functional Diagram of an ISDN Workstation.
CPU Processor for text and data functions

displayed on the screen: dialling is then started by pressing just one additional key. This is a typical example of avoidance of "media discontinuity": manual entry of the number after consulting the directory is no longer necessary.

The IVDT has an alphanumeric keyboard for using the videotex messaging service and for local data entry. Other local functions are conceivable, such as keeping an appointment diary and operating a card reader. The information content of one's own IVDT can be stored on the card and loaded to other IVDTs of the same type so that, for instance, one's personal telephone directory can be used there, too.

In most cases, the IVDT will support a connected printer.

Another advanced multiservice terminal is an IVDT in the form of an office workstation. As well as optional utilization of several telecommunication services, workstations provide a range of local functions similar to that found in personal computers. Figure 5.10 shows the basic layout of a workstation.

In addition to telephony, it is possible to communicate with public and private videotex centers, with public and private message handling systems and with special application programs in data processing systems. Workstations can also operate with teletex protocols and use the teletex service.

Local functions can include

- text editing
- pocket calculator functions
- loading and running application programs
- office service functions such as appointment diary, reminders, work schedule
- memory functions such as storage with automatic retrieval, or storing personal notes
- support of telephony.

The ISDN workstation also allows simultaneous voice and non-voice communication (e.g. for an enquiry to a database during a telephone conversation).

Like personal computers, workstations provide connection facilities for various peripherals such as printers and external memories.

6 Switching in the ISDN

6.1 Introduction

The cost-effective realization in LSI chips of digital time-division multiplexing and speech digitization (64 kbit/s pulse code modulation – PCM – in accordance with CCITT Rec. G.711 [6.1]) has resulted in a new concept for switching equipment.

Typical of the new generation of equipment is that used in modern data networks [6.2, 6.3] and for the digitized telephone network (Integrated Digital Network IDN) [6.4]. The digital telephone network develops from the existing analog network as a consequence of the cost benefits of using digital components. The evolving network is characterized by the integration of digital transmission (cf. Sect. 7) and digital switching. As already discussed in Sects. 1.6, 1.7 and 3.3, the digitized telephone network forms the basis for the ISDN [6.5 through 6.7].

Digital technology not only allows an extended range of user facilities to be provided (see ISDN supplementary services in Sect. 2.3.3 and [6.8]) but also enables all the services to be handled via the same subscriber access. An important principle in ISDN is the separation of service aspects on the one hand and network aspects on the other (Fig. 6.1). The teleservices and bearer services are implemented within the network by means of appropriate *network capabilities* (see CCITT Recs. I.210 [6.9] and I.310 [6.10]) with low layer capabilities (LLCs) in protocol layers 1 to 3 and high layer capabilities (HLCs, relating to teleservices only) in layers 4 to 7. LLCs comprise "basic low layer functions" for setting up and clearing down ISDN connections and "additional low layer functions" for handling supplementary services. Basic low layer functions are determined essentially by the *ISDN connection types* defined in CCITT Rec. I.340 [6.11]. Like the ISDN services, the connection types are described in terms of appropriate attributes (Table 6.1). Dominant connection attributes include the mode (circuit or packet switching) and the bit rate for transferring user information.

The essential functional elements of a public ISDN from the point of view of switching are shown in Fig. 6.2:

- *ISDN local exchanges* with digital subscriber lines (see Fig. 3.1),
- *Common channel signaling capabilities* (CCITT signaling system No. 7) for transferring signaling information between exchanges,
- physical connections with *circuit switching*,
- virtual connections with *packet switching*,
- *specialized equipment* for additional functions.

A large number of ISDN supplementary services can be provided with the aid of stored program control (SPC) exchanges in conjunction with the principle of outslot signaling, i.e. separate channels for transmitting user information and signaling

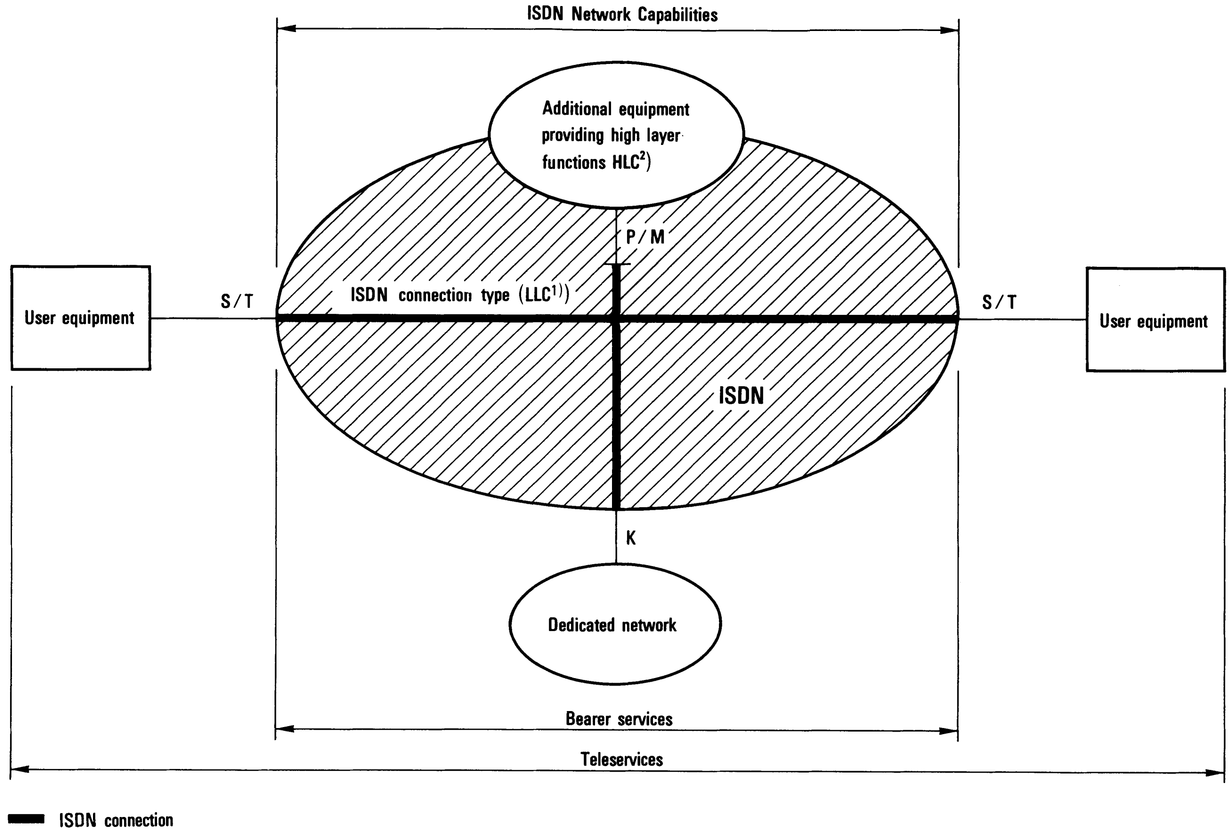


Fig. 6.1. Relationship Between Services and Network Capabilities of an ISDN.
S, T, K, P, M Reference points
[1]) Low Layer Capabilities (relating to layers 1–3)
[2]) High Layer Capabilities (associated with layers 4–7): specialized resource within the ISDN (P) or specialized service provider outside the ISDN (M)

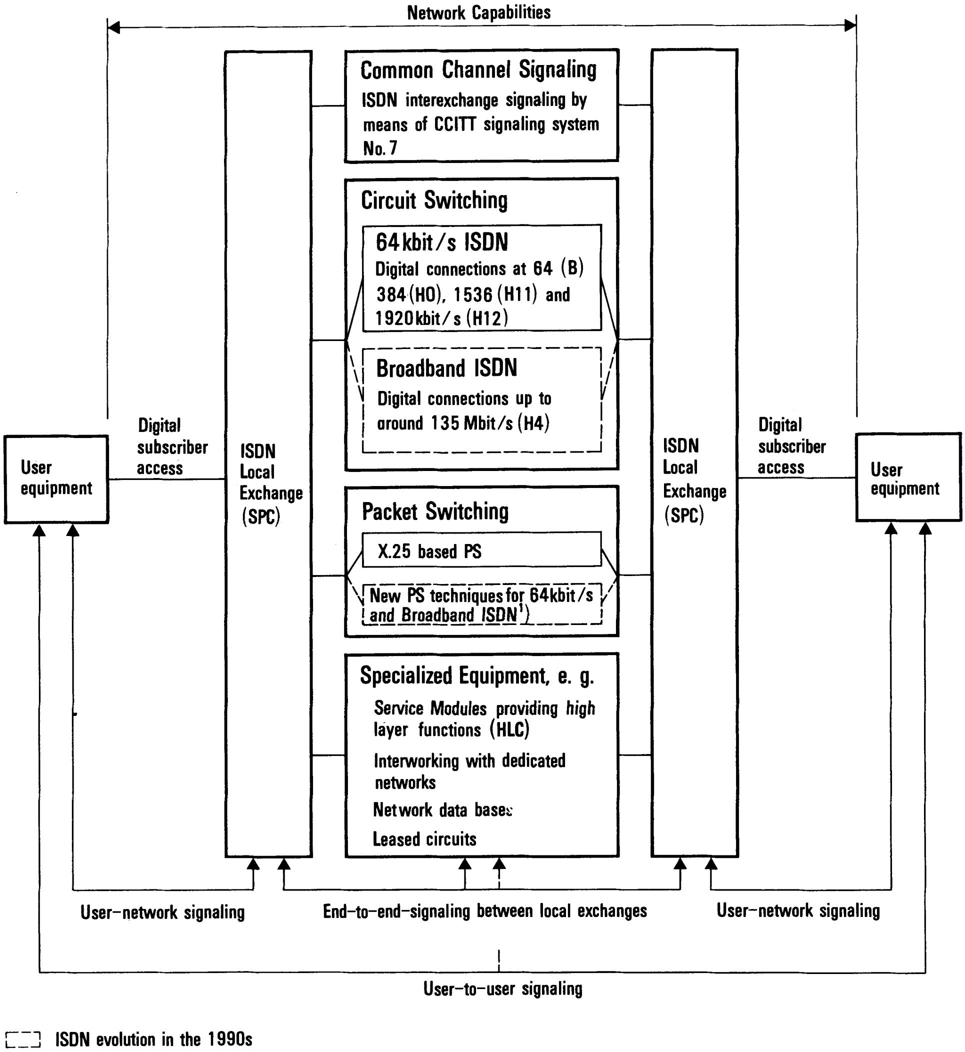


Fig. 6.2. Basic Architectural Model of an ISDN.
LE Local exchange
SPC Stored Program Control
HLC High Layer Capabilities (associated with layers 4–7)
[1]) see Sect. 4.5

information (see Sect. 2.3.3). Control of these ISDN features such as completion of calls to busy subscribers (CCBS) or in-call modification (allowing a user to change from voice to data and back, if necessary, during an established call) resides almost exclusively in the *ISDN local exchanges* (see Fig. 3.1). In the transition from the digital telephone network to the ISDN, therefore, it is the local exchanges which primarily have to be upgraded. This involves not only modifying the hardware to provide digital transmission on subscriber lines (see Cap. 4) but also expanding the (switching) software (see Sect. 6.2).

Table 6.1. Attribute Values for ISDN Connection Types (in accordance with CCITT Rec. I.340)

Attribute	Values[a]
Information transfer mode	– circuit-switched (cs) – packed-switched (ps)
Information transfer rate	– cs: 64 (B), 384 (H0), 1536 (H11), 1920 kbit/s (H12) – ps: packet throughput
Information transfer susceptance	– unrestricted digital information (i.e. any bit sequence) – speech[b] (digital encoding, e.g. A law, μ law) – 3.1 kHz audio[c] (e.g. voice-band data)
Establishment of connection	– switched[d] – semi-permanent[e] – permanent[f]
Symmetry (of information transfer)	– unidirectional – bidirectional symmetric – bidirectional asymmetric[g]
Connection configuration	
– topology	– point-to-point, multipoint
– dynamics	– concurrent, sequential, or/add remove[h]
Strcuture (of user information)	
– layer 1	– 8 kHz integrity[i], unstructured
– layers 2 and 3	– Service data unit integrity, unstructured
Performance	
– error performance	– CCITT Rec. G.821 (see Sect. 7.7.1)
– slip performance	– CCITT Rec. G.822 (see Sect. 7.7.2)

[a] applying to the overall ISDN connection consisting of access and transit connection elements
[b] bit manipulation may be applied (see Sect. 7.2.1 and 7.7.5)
[c] suitable for modem signals
[d] on demand set-up in response to signaling information from the subscriber
[e] connection passes through a switching network (exchange)
[f] connection uses the transmission network only (by-passing of exchanges)
[g] i.e. different information transfer rates in the two directions
[h] refers to the time sequence in which connection elements are set up and released
[i] bits submitted within a demarcated 125 μs interval are delivered within such an interval

A great many ISDN-specific functions, processes and supplementary services have to be handled in the ISDN local exchanges (see Sect. 2.3.3), whereas the transit exchanges are affected by ISDN to a much lesser degree. Consequently, the *common channel signaling* system has to provide the additional facility of signaling between the local ISDN exchanges without the need for the signaling information to be processed in the transit exchanges involved in the connection. In other words, end-to-end signaling between originating and terminating ISDN exchanges is to be provided (see Sect. 6.3). In addition, within CCITT signaling system No. 7 [6.12 to 6.15], the traditional separate signaling procedures for speech (see Telephone User Part TUP [6.16 to 6.20]) and text/data (see Data User Part DUP [6.21]) have to be replaced by a *single* universal integrated services signaling protocol (see ISDN User Part in Sect. 6.3 and [6.22 to 6.26]).

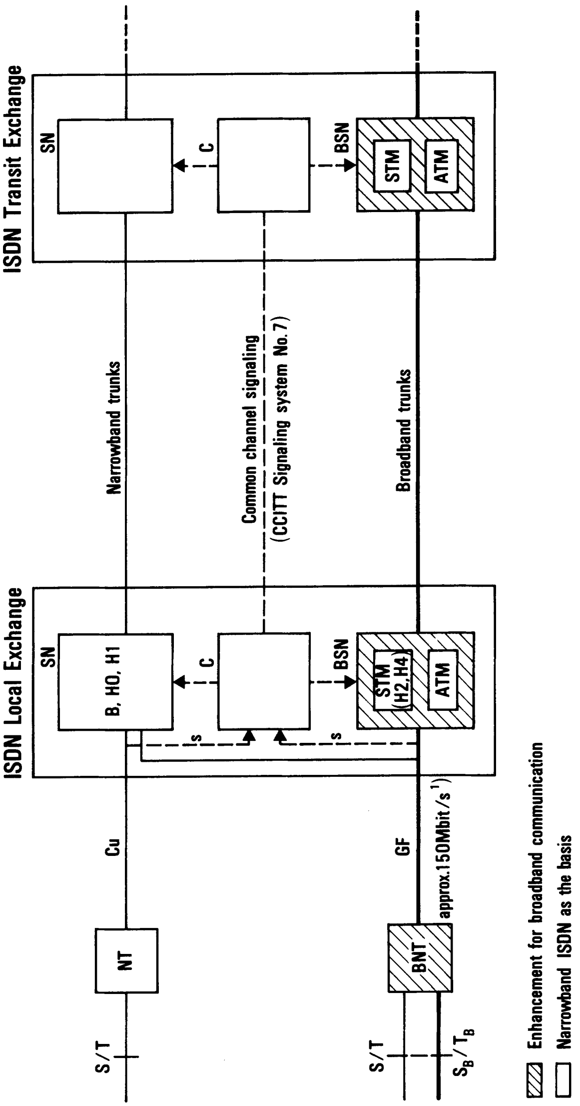


Fig. 6.3. Evolution of ISDN Towards Broadband ISDN.

——	64 kbit/s ISDN
━━	Broadband ISDN (B-ISDN)
- - -	Signaling (call control)
NT	Network termination
BNT	Broadband network termination
Cu	2-wire copper access line
GF	Glass fiber
s	Signaling information
SN	ISDN switching network (connection control)
BSN	Broadband switching network
C	Call control
Broadband channels:	H21/H22 approx. 34/43 Mbit/s H4 approx. 135 Mbit/s
ATM	Asynchronous Transfer Mode (bit rate independent logical channels, see Sect. 4.5)
STM	Synchronous Transfer Mode (equivalent to circuit switching possibly coexisting with ATM in hybrid arrangements)

[1]) or up to 600 Mbit/s for TV program distribution in the direction to the subscriber

The starting point for ISDN is a basic system with circuit switching based on the digital telephone network and featuring the following principal functions (Figs. 6.2 and 6.3):

- switching of digital B channel connections (connection control), and
- signaling functions (call control).

With the transition from the digital telephone network to ISDN, the circuit-switched digital connections of the trunk network, originally provided for transmitting PCM-encoded telephone signals, will become universal transparent 64 kbit/s user-to-user connections which in turn will form the basis for a wide range of text and data services (see Unrestricted Digital Information in Table 6.1). As an interim solution in some countries while ISDN is being introduced, some ISDN connections may be routed via analog network sections. Moreover, ISDN sections may be routed via digital sections which operate with transcoding to 32 kbit/s ADPCM (see Sects. 7.2.1 and 7.7.5). Both measures are only applicable if the relevant service permits the analog transmission or the bit manipulation that they involve; this covers speech and voice-band data as shown in Table 6.1 (3.1 kHz audio, and suitable also for modem signals).

Voice and circuit-switched non-voice services require the same basic ISDN functions for signaling (call control), routing etc. available in all ISDN local and transit exchanges. Accordingly common signaling protocols are employed on the user side (see Sect. 4.3) and on the interexchange side (see Sect. 6.3) for ISDN telephony and for circuit-switched text and data services; this is the principle of *single-step call establishment* (see Sects. 4.4.1 and 4.4.2).

The principle of full integration described above applies also to the expansion of the switching equipment of the 64 kbit/s ISDN for use in the *broadband ISDN* at a later stage (Figs. 6.2 and 6.3). This affects the following aspects of the ISDN exchanges (see Sect. 6.2 and [6.27 to 6.29]):

- the *subscriber access*, which must be equipped for optical fibers (see Sect. 7.3.1) instead of two-wire copper access lines, and
- the *switching network*, which must be supplemented with broadband modules (approx. 34/43 Mbit/s and 135 Mbit/s).

Although full integration of *packet switching* techniques in the ISDN is possible in principle, this must wait for a later phase owing to the considerable effects it would have on existing packet terminals and ISDN exchanges (see below). If *conventional X.25 terminals* are connected to the ISDN (Fig. 6.4a), the rest of the ISDN, i.e. the basic system, merely functions as a transparent access from the local ISDN exchanges via 64 kbit/s trunk connections to special packet switching equipment which may reside either *inside* the ISDN (see the use of the *ISDN virtual circuit bearer service* described in Sect. 4.4.1) or *outside* the ISDN (see the *access to data transmission services provided by separate packet networks*, discussed in Sect. 4.4.1). In both cases, ISDN user-network and interexchange signaling serve merely to establish the access connection so that for packet-switched bearer services additional service-specific protocols have to be handled "in slot", i.e. in the B channel; this is the principle of *two-step call establishment* with separate call control for circuit and packet switching (see Sects. 4.4.1 and 4.4.4).

The key concept of *additional ISDN packet mode bearer services* (see Fig. 6.4b and Sect. 4.5), which are not based on CCITT Recs. X.25/X.75, involves common call control and uniform signaling protocols for circuit *and* packet switching in accordance with the principles of single-step call establishment and outslot signaling – in other words, the use of the extended D channel protocol according to CCITT Rec. I.451 (see Sect. 4.3.5) instead of X.25, and ISDN interexchange signaling ISUP (see Sect. 6.3) instead of signaling based on CCITT Rec. X.75. Two of the advantages of this farther reaching integration of packet switching in ISDN are as follows:

- uniform handling of supplementary services and of operation and maintenance functions for circuit and packet switching;
- possibility of simultaneous use of circuit and packet-switched bearer services in the course of a *single* call (multiservice mode).

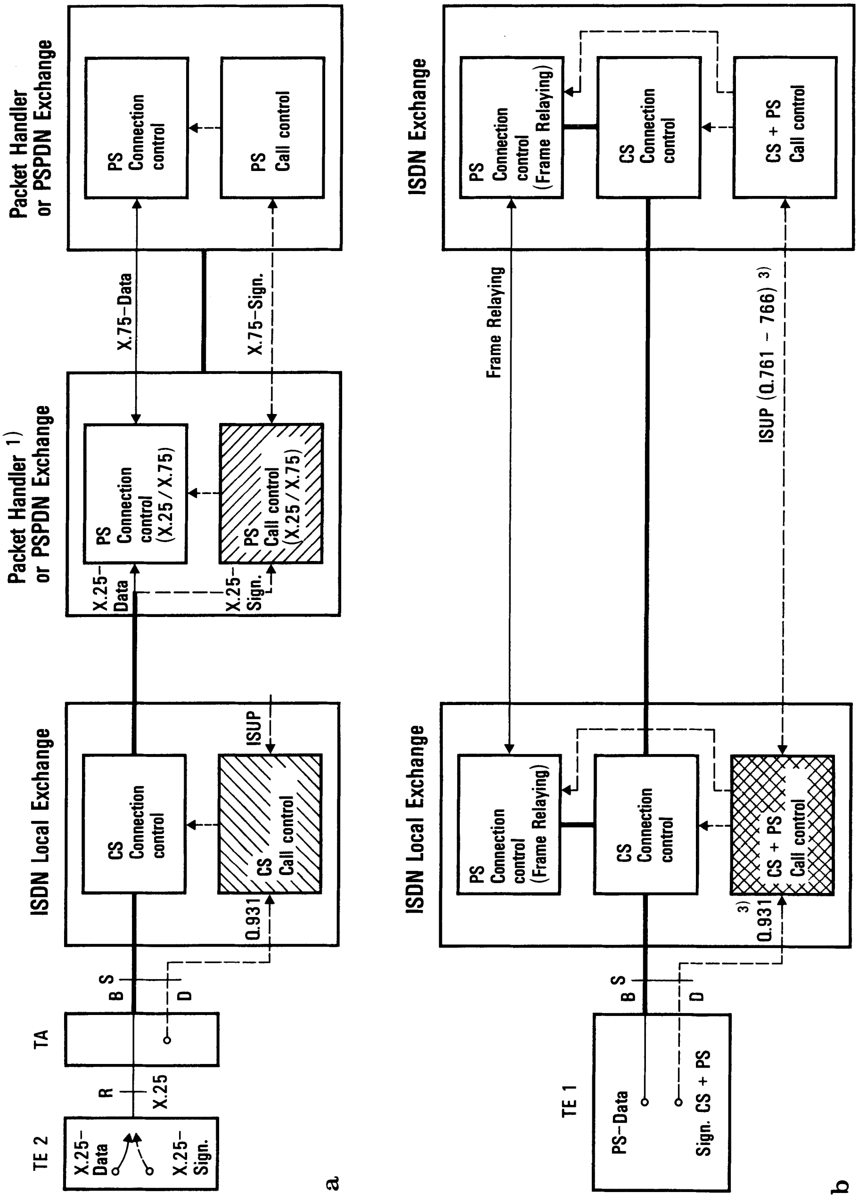


Fig. 6.4 a, b. Stages of Integration for Packet Switching in ISDN.
a X.25 based packet mode bearer services; **b** Additional packet mode bearer services.

CS	Circuit switching	TA	Terminal adaptor
PS	Packet switching	ISUP	ISDN User Part (see Sect. 6.3)
TE 1	ISDN terminal equipment with S interface	PSPDN	Packet-switched public data net
		——— (thick)	64 kbit/s user information channels
TE 2	Terminal equipment with non-ISDN interface	———	Packet data
		- - - -	Signaling (Sign.)/Call Control

[1]) cf. use of ISDN virtual circuit bearer service in Sect. 4.4.1
[2]) cf. access to data transmission services provided by PSPDNs in Sect. 4.4.1
[3]) user-network signaling (Q.931) and interexchange signaling (ISUP) enhanced as compared to case **a**

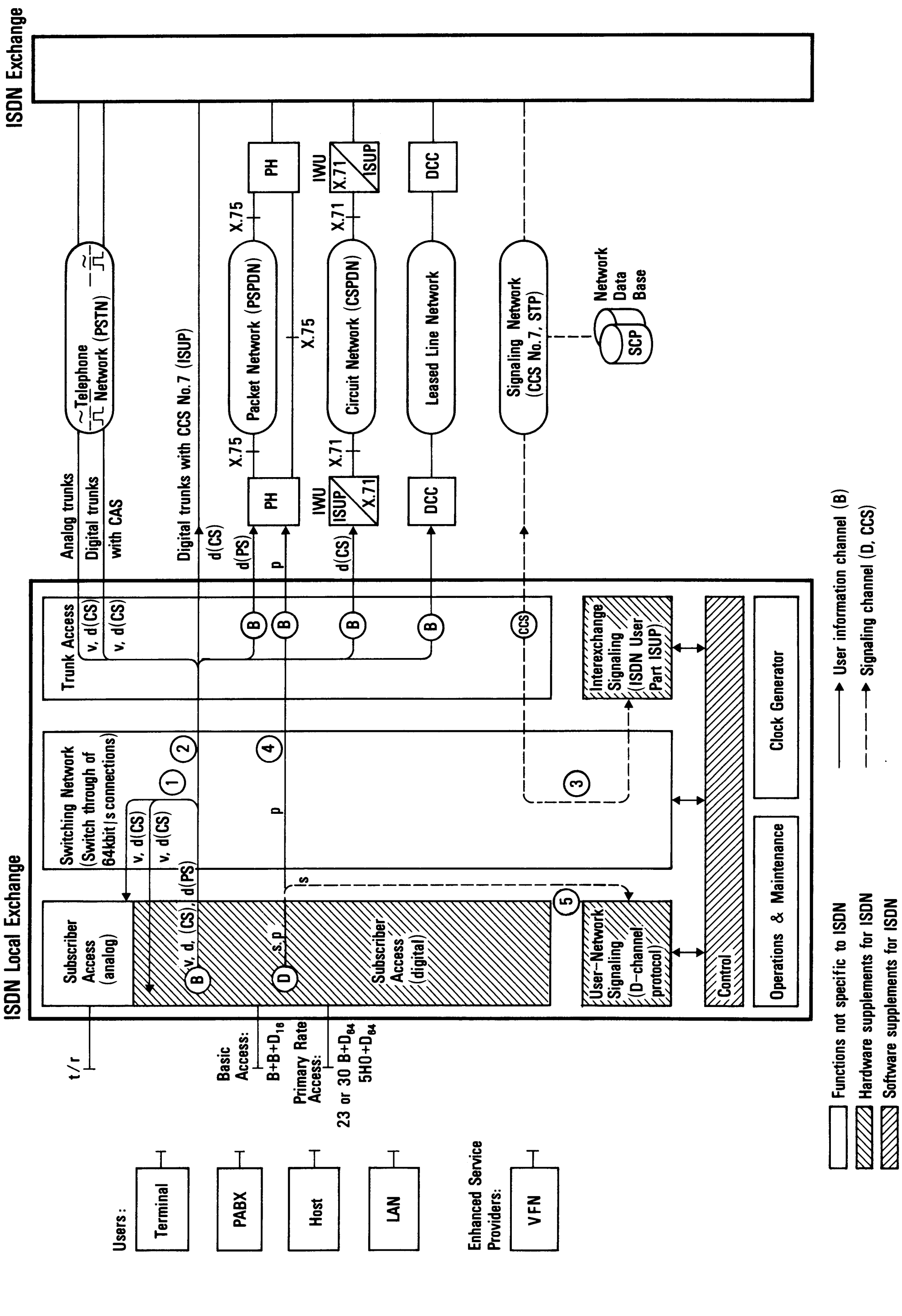
In view of the fact that standardization work has only just started in the CCITT and detailed recommendations which are intended to form the basis for implementation are not due to be published until 1992, we shall probably have to wait for the second phase of ISDN before new ISDN packet switching techniques can be implemented.

For high-layer functions (high layer capabilities) associated with protocol layers 4 to 7 and for certain functions in layers 1, 2 and 3 (low layer capabilities) *specialized equipment* is also required in the ISDN (Fig. 6.2). An example of such a low layer function is *interworking with dedicated text and data networks* (see Sect. 4.4 and Figs. 6.5, 6.8). For reasons of economy and practical implementation, interworking equipment will not be provided in every ISDN exchange but will rather be centralized and assigned to certain exchanges at the higher network levels.

Another example of specialized equipment is a digital cross connect DCC providing for the control of transmission links, e.g. setting up of *leased circuits.* While ISDN exchanges are optimized for establishing circuit-switched connections lasting typically a few minutes, DCC connections are maintained for longer periods of time (from days to years).

Service modules for high layer functions can be implemented as feature nodes FN or as vendor feature nodes VFN, depending on whether value-added services VAS are offered to ISDN subscribers by the network carrier or by independent enhanced service providers (see Figs. 6.5, 6.8). VAS can be characterized as involving storage and processing of user information. Fundamentally, they comprise four main categories: data base access, store-and-forward communication of the message handling type (CCITT X.400 series of Recs.), support services such as electronic on-line directories (CCITT X.500 series of Recs.), and compatibility services (see Sect. 2.3.1.3 and [6.30]).

The feature nodes (FN, VFN) mentioned above add intelligent functionality to the ISDN; this functionality is offered directly to the user, i.e. enhanced services are offered to the subscriber via the basic services. *Network data bases* (see service control point SCP in Figs. 6.5, 6.8), which can be accessed on a real time basis during call control by ISDN exchanges via the common channel signaling network, result in a different kind of internal network intelligence intended for improving the provision of basic services. The main objective of this concept of the *intelligent network* [6.31 – 6.33] is to give administrations a greater amount of flexibility in terms of rapid introduction of new services, even on a trial basis (see Sect. 6.3.7). Since the control logic for handling special calls resides in centralized network data bases, it can be changed without affecting the call processing software in the various ISDN exchanges. Another benefit of the SCP concept is the ability to customize services for unique customer needs including the concept of embedding private networks as logical subnetworks into the resources of the public network; such embedded networks are sometime termed “virtual corporate networks”.


ISDN Exchange
ISDN Local Exchange
Users:
Terminal
PABX
Host
LAN
Enhanced Service Providers:
VFN
t/r
Basic Access:
B+B+D16
Primary Rate Access:
23 or 30 B+D64
5H0+D64
Subscriber Access (analog)
Subscriber Access (digital)
v, d, (CS), d(PS)
s, p
Switching Network (Switch through of 64kbit/s connections)
v, d(CS)
v, d(CS)
p
s
Trunk Access
v, d(CS)
v, d(CS)
User-Network Signaling (D-channel protocol)
Interexchange Signaling (ISDN User Part ISUP)
Control
Operations & Maintenance
Clock Generator
Analog trunks
Digital trunks with CAS
Digital trunks with CCS No.7 (ISUP)
d(CS)
d(PS)
p
d(CS)
Telephone Network (PSTN)
Packet Network (PSPDN)
Circuit Network (CSPDN)
Leased Line Network
Signaling Network (CCS No.7, STP)
Network Data Base
SCP
PH
IWU
ISUP
X.71
X.75
DCC
CCS
Functions not specific to ISDN
Hardware supplements for ISDN
Software supplements for ISDN
User information channel (B)
Signaling channel (D, CCS)

6.2 New Demands Placed on Switching Due to Service Integration in the ISDN

The new demands made on switching are mainly attributable to integration of services. To meet these demands, new ISDN-specific functions, evolving from the functional concept of switching in the digitized telephone network, have had to be introduced. These have resulted in corresponding changes or expansions to the hardware and software. Figure 6.5 shows the functional structure of an ISDN local exchange and its surrounding:

- Subscriber access
- Switching network
- Trunk access
- User-network signaling
- Interexchange signaling
- Control
- Operations, administration and maintenance
- Timing and network synchronization.

The ISDN-specific hardware and software functional units are shown hatched.

6.2.1 Subscriber Access

The functional unit for the subscriber access comprises all the layer 1 interface functions, particularly those for connection of digital ISDN subscriber lines. This includes functions for interoperation with transmission equipment (cf. Sect. 7.4.3) as well as functions associated with the interface structures for the *basic access* ($B+B+D_{16}$) and for the *primary rate access* (primarily $23\,B+D_{64}$ or $30\,B+D_{64}$) already presented in Sect. 4.2, Table 4.3. Figure 6.6 relates to CCITT Rec. Q.512 [6.34] and contains a summary of the digital interfaces of an ISDN local exchange for both the user and the interexchange side (cf. Sect. 6.2.2). It should be noted that in the

Fig. 6.5. Functional Structure and Communication Links of an ISDN Local Exchange.

B	B-channel	PH	Packet Handler (see Sect. 4.4)
CAS	Channel associated signaling	PABX	Private Automatic Branch Exchange
CCS	Common Channel Signaling	PS	Packet switched user information
CS	Circuit switched user information	PSPDN	Packet Switched Public Data Network
CSPDN	Circuit Switched Public Data Network	s	Signaling information
D	D-channel	SCP	Service Control Point (see Sects. 6.1, 6.3.7)
d	Data (non-voice) information	STP	Signaling Transfer Point (see Sect. 6.3)
DCC	Digital Cross Connect	v	digitized voice information
ISUP	ISDN User Part (CCITT Signaling System No. 7)	VFN	Vendor Feature Node
IWU	Interworking Unit: Signaling conversion from ISUP to X.71	1 to 5	Communication links via the switching network (see Sect. 6.2.3)
LAN	Local Area Network		
p	Packet data (on D-channel)		

CCITT Blue Book (1989) the content of the Q.500 series of Recs. will be rearranged and supplemented:

New Rec. Q.512 will describe the subscriber side interface characteristics for local exchanges whereas Rec. Q.511 will deal with interexchange interfaces for local, combined and transit exchanges. Depending on their size, ISDN private automatic branch exchanges (PABXs) can obtain access to the local exchange either via the basic access (see U interface in Fig. 6.6) or the primary rate access (V3).

Interfaces V1 to V4 separate the transmission functions of the ISDN user-network interface from the switching functions. The physical *line termination* (LT) performs transmission functions whereas the *exchange termination* (ET) logically terminates the subscriber access. The V interface is a functional boundary between ET and LT; it does not necessarily exist as a physical interface, since particularly cost-effective solutions can be provided by an integrated LT/ET implementation. This applies to the ISDN basic access for which transmission and frame structure on the subscriber line are not yet defined internationally but (in some countries) nationally (cf. Sect. 7.4.3 and [6.35]). The transmission functions of the LT in the case of the basic access correspond to those of the NT1 (cf. Sect. 4.1).

Subscriber access functions can also be arranged in remote equipment physically separate from the exchange and connected to it by primary rate time-division multiplex systems: such devices include digital concentrators (interface V2 in Fig. 6.6) and digital multiplexers (interface V4, Sect. 7.4.3; to be designated V6 in the Blue Book); this latter option is of importance in areas of low subscriber density and particularly during the introductory phase of ISDN, in which not all local exchanges will yet have been converted to ISDN (cf. Sect. 3.8).

Interfaces V2 to V4 are based on the same digital transmission techniques (cf. Chap. 7) as that used on the interexchange side. The electrical characteristics of these interfaces, i.e. bit rate, pulse shape, etc., are defined in CCITT Rec. G.703. The functional characteristics of V2 through V4, i.e. particularly the frame structure, are described in CCITT Rec. G.704. CCITT Rec. G.705 contains additional definitions for digital connections terminating on digital exchanges.

In addition to the digital interfaces shown in Fig. 6.6, an ISDN local exchange must also have connection facilities for analog subscriber lines (cf. Sect. 2.3.4), since when a digital ISDN exchange replaces a conventional exchange in the analog network it must take over the analog subscriber lines which remain (Fig. 6.5: analog tip/ring interface t/r). Examples of analog subscriber access functions are: Signaling facilities such as tone generators and digit receivers, conversion between analog and digital voice signals (codec) and conversion between two-wire and four-wire operation.

The functional units for digital, i.e. ISDN subscriber access lines and for analog telephone lines are linked to a switching network by time-division multiplexed (TDM) internal paths as are the corresponding interface units on the interexchange side (see Sect. 6.2.2). The function of the switching network (cf. Sect. 6.2.3) is to through-connect the 64 kbit/s B-channels. For this purpose, a time slot of the incoming internal time-division multiplex system, corresponding to an incoming B-channel, is through-connected to the "set" time slot of an outgoing time-division multiplex system. This requires the functional units for user and interexchange access to adapt the different subscriber lines and interexchange trunks of a combined digital/analog local exchange to the internal digital switching paths.

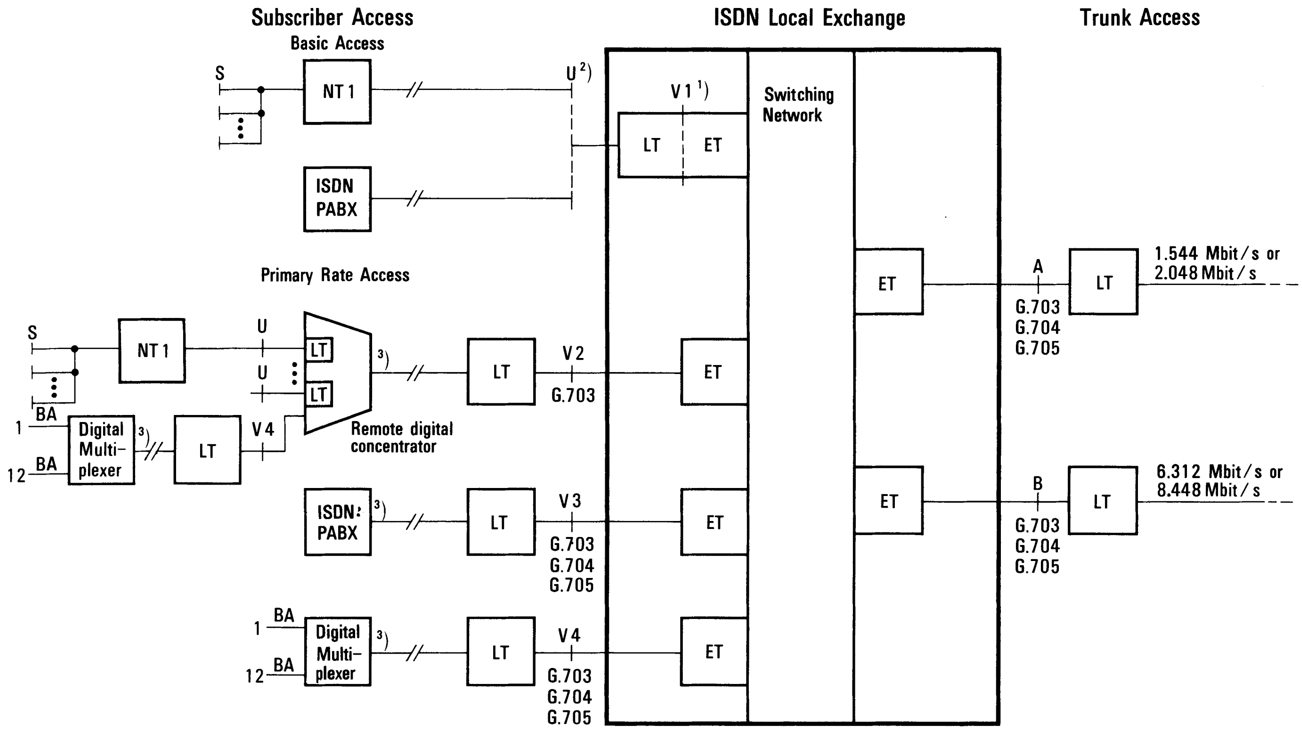


Fig. 6.6. Digital Interface Associated with an ISDN Local Exchange.
BA Basic Access
ET Exchange Termination
LT Line Termination
1) V1 may be an internal interface in the switching system (system dependent, not subject to standardization by CCITT)
2) not yet subject to standardization by CCITT
3) LT not shown

In addition to digitized voice signals v, the 64 kbit/s user information channels (B-channels) can also transmit circuit switched and packet switched non-voice information (d(CS) or d(PS)) at 64 kbit/s. The D_{16} channel of the basic access can also be used to convey packet data p between ISDN users and the ISDN exchange in addition to signaling information s. The s and p-type information must be separated in the exchange (Fig. 6.5). Whereas the signaling information is intended for the user-network signaling functional unit (cf. Sect. 6.2.4), the p information is forwarded to a Packet Handler PH in a "transparent" manner, i.e. without any processing in the ISDN exchanges through which it passes – e.g. via semi-permanent 64 kbit/s connections used for transferring packet data from a number of subscribers on a message interleaving basis (cf. Sect. 4.4.4).

6.2.2 Trunk Access

The functional unit for the trunk access (Fig. 6.5) comprises all the functions for interconnection towards other exchanges at first and second levels of the digital transmission hierarchy (interfaces A and B in Fig. 6.6). Electrical and functional characteristics of the digital interexchange interfaces A and B are defined in CCITT Rec. Q.512 [6.34] for digital local and combined exchanges and in CCITT Rec. Q.502 [6.36] for digital transit exchanges, with references to Recs. G.703, G.704 and G.705 (cf. Chap. 7). As mentioned above, the content of the Q.500 series of Recs. will be rearranged in the CCITT Blue Book: Interexchange interfaces for local, combined and transit exchanges will then be covered by Rec. Q.511.

For a comparatively long transition period while analog multiplex systems are still being used at the interexchange level of the telephone network, it must also be possible, for reasons of compatibility with the telephone network, to connect analog transmission systems to an ISDN exchange (Fig. 6.5). For such circuits, a digital to analog conversion facility must be provided; in practice this is implemented by means of a signaling converter combined with a PCM multiplexer (cf. SC-Mux in Fig. 6.8).

Like the corresponding unit for the subscriber access, the trunk access unit is connected to the switching network via time-division multiplexed (TDM) digital paths (Fig. 6.5). In the switching network, the individual 64 kbit/s channels are through-connected. The common 64 kbit/s signaling channels (CCS) used to transfer the interexchange signaling within CCITT Signaling System No. 7 are also routed to the switching network in the same way. The common signaling channels are subsequently routed to the functional unit for interexchange signaling via dedicated switching network paths (semipermanent through-connection).

6.2.3 Switching Network

In the digitized telephone network with integrated switching and transmission, the 64 kbit/s channels transmitted on a time-division multiplex basis are switched direct from the TDM systems that are connected to the exchange, i.e. without previous splitting up of the TDM signal into individual 64 kbit/s channels. For PCM voice transmission each channel octet contains a PCM codeword (cf. Sect. 7.2.1).

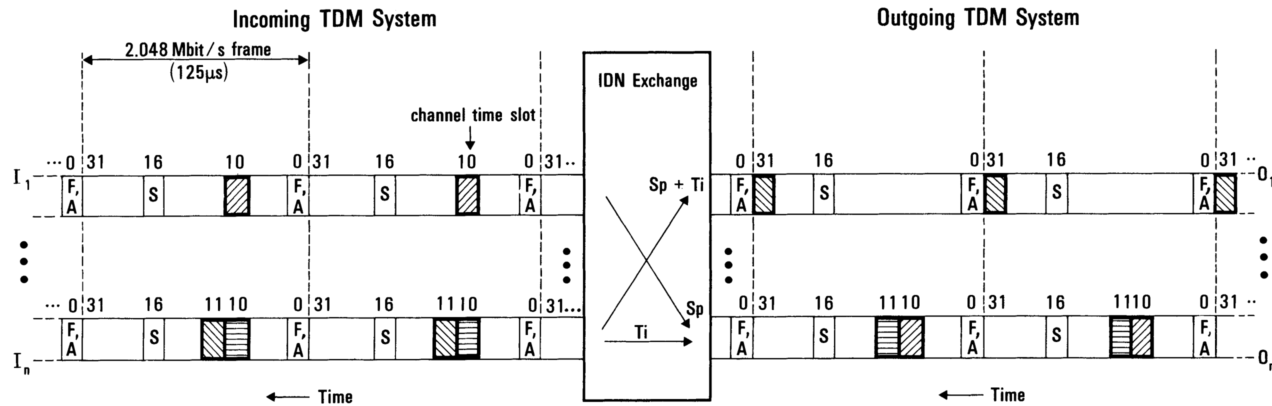


Fig. 6.7. Spatial and Temporal Arrangement of the 64 kbit/s User Information Channels in Time-division Switching.

A	Alarm indication and other messages (channel time slot 0)
F	Frame alignment signal (channel time slot 0): zero point for channel numbering
IDN	Integrated digital network: transmission *and* switching of time-division multiplexed digital channels
$I_1 \ldots I_n$	Digital TDM systems – incoming direction
$O_1 \ldots O_n$	Digital TDM systems – outgoing direction
S	Interexchange signaling information (channel time slot 16)
Sp	Change of position in space (space slot change)
Sp+Ti	Combined change of space *and* time slot
TDM	Time-division multiplex
Ti	Time slot change

Thus, in digital time-division multiplex switching, the switching function consists of connecting octets (cf. Sect. 7.2.3, Fig. 7.3) through the switching network from an incoming TDM signal to the relevant outgoing TDM signal (Fig. 6.7). This involves two basically different through-connection methods:

- *Time stages* through-connect an octet by changing the channel time slot between an input TDM system and the associated output TDM system. (Time slot change Ti in Fig. 6.7: An octet received in time slot 10 of incoming system I_n is transferred to time slot 11 of the corresponding TDM system O_n).
- *Space stages* on the other hand through-connect the octets from an incoming TDM system to another TDM system corresponding to the switching destination while retaining their slot. (Change of position in space, Sp in Fig. 6.7: Octet 10 of input system I_1 retains the same phase when transferred to output system O_n).

The switching process generally requires changing the time slot and the position in space of the octet being through-connected. In practice therefore, time stages and space stages have to be combined, e.g. in the form of "time-space-time" switching network arrangements [6.37].

Depending on the type of information (cf. Sects. 6.2.1 and 6.2.2) which is transferred on the 64 kbit/s channels routed via the switching network, the following "exchange connections", as considered in CCITT Rec. Q.513 [6.38] can occur in an ISDN exchange (cf. Fig. 6.5):

(1) B-channel (B) – B-channel (B)
(2) B-channel (B) – interexchange user information channel (B)
(3) Common signaling channel (CCS) – functional unit for interexchange signaling
(4) Signaling channel (D_{16}) – semipermanent feeder channel (B) to a Packet Handler (PH) in the ISDN (see ISDN virtual circuit bearer service solution in Sect. 4.4.4)
(5) Signaling channel (D_{16} or D_{64}) – functional unit for user-network signaling.

For higher-rate channels (cf. Table 4.1) there is also the option of switching more than one 64 kbit/s channel through the switching network in the course of one connection, e.g. five 64 kbit/s channels in the case of the H0-channel with a total bit rate of 384 kbit/s (cf. Sect. 4.2). The H0-channel occupies five 64 kbit/s time slots per 2.048 Mbit/s pulse frame and all of these have to be through-connected to the same address. This requires special arrangements in the switching network to retain the original octet sequence (octet sequence integrity).

6.2.4 User-Network Signaling

The user-network signaling functional unit – depending on the type of subscriber access – handles ISDN user-network signaling (cf. Sect. 4.3 and Table 6.2) or dialing procedures for analog telephone lines (t/r). This includes receiving and transmitting signaling information as well as conversion of external signaling events into internal messages for the control functional unit and vice versa (Fig. 6.5). The "control" unit assumes the tasks associated with the control of a connection from the incoming side to the outgoing side (cf. Sect. 6.2.5). For example, it determines the path via the switching network between the subscriber access line and the trunk line by evaluating the dial information and sets this path up; in addition, it is responsible for passing on dial information to the interexchange signaling functional unit (cf. Sect. 6.2.6).

As already discussed in Sect. 6.2.1, users can also obtain access via remote concentrators (called Digital Line Units DLU in the EWSD system) connected to the local exchange via primary rate time-division multiplex systems (Fig. 6.6). This system structure allows the EWSD switching system, for example (Fig. 6.8), to exercise decentralized control of the D-channel protocol for the ISDN basic access: the physical and logical functions of layer 1 and the complete layer 2 procedure (HDLC LAPD; see Sect. 4.3.4) for eight ISDN basic accesses are handled on the ISDN "Subscriber Line Module Digital" (SLMD) of the DLU, whereas control of layer 3 of the D channel protocol is undertaken in the "Line/Trunk Groups" LTG of the parent exchange.

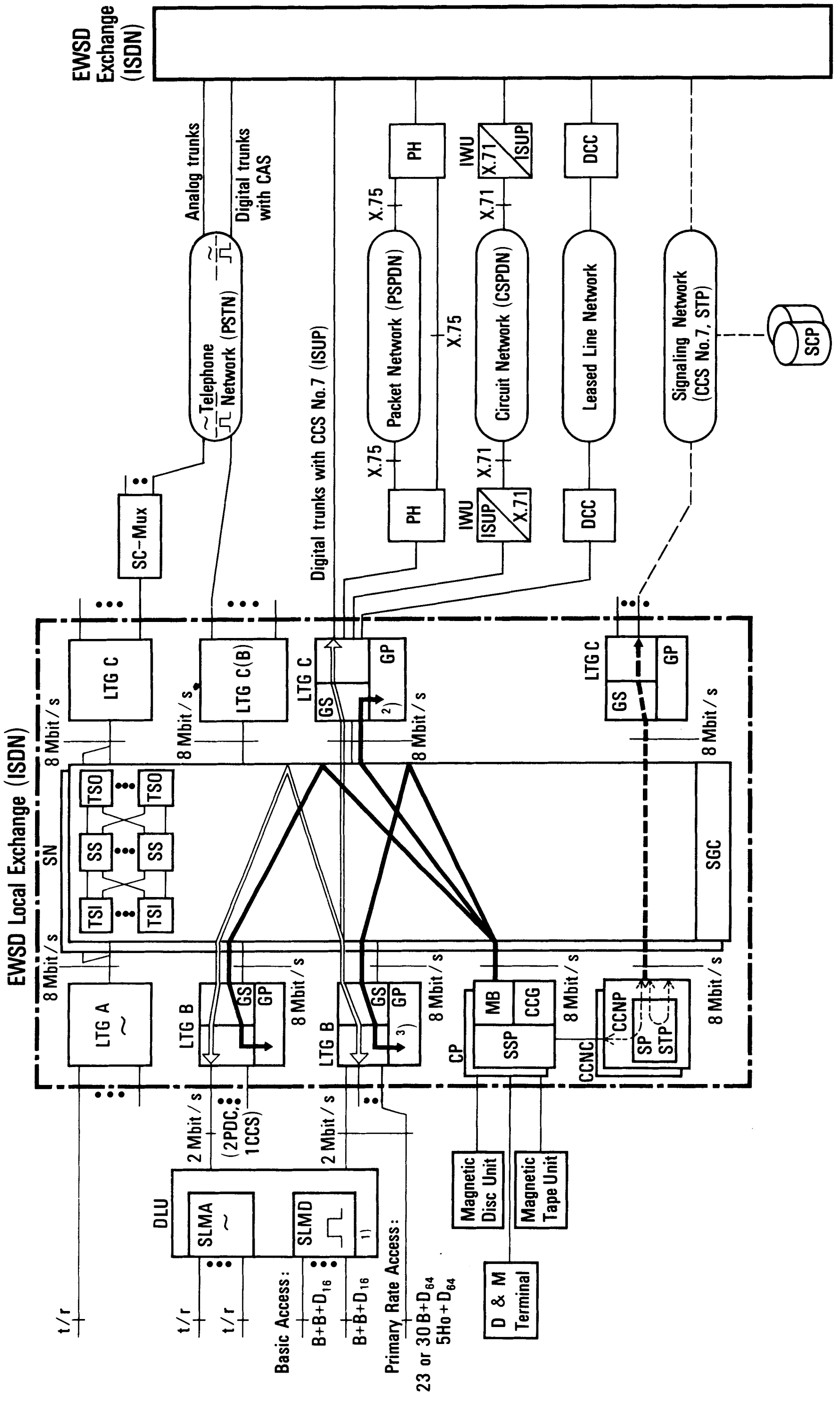
Table 6.2. Switching Functions of an ISDN Local Exchange

(1) *Signaling*
- Transmitting/receiving signaling messages in accordance with D-channel protocol (user side) or ISDN User Part (interexchange side),
- Conversion of external signaling events into internal messages in the exchange for the control functional unit and vice versa.

(2) *Access Control*
- Basic call processing (call establishment and clearing),
- Control of call-related ISDN supplementary services (e.g. completion of calls to busy subscribers, in-call modification, etc.),
- B-channel administration,
- Handling of signaling transactions,
- Conversion of functional signaling into stimulus signaling,
- Subscriber messages for registration and cancellation of user facilities,
- Preprocessing of dial information,
- Authorization check for access to ISDN services,
- Recording of charging and traffic data,
- Overload protection,
- Timing functions.

(3) *Switching*
- Analysis of dial information,
- Group selection and zoning,
- Routing,
- Path search in the switching network from an incoming to an outgoing exchange port,
- Through-connection or release of the switching network paths in the course of setting up or clearing down 64 kbit/s connections by means of switching network commands,
- Calculating charges.

(4) *Interworking with Operations* (e.g. charges, traffic measurement) *and Maintenance* (e.g. fault analysis, recovery)


EWSD Exchange (ISDN)
Analog trunks
Digital trunks with CAS
Telephone Network (PSTN)
Digital trunks with CCS No.7 (ISUP)
Packet Network (PSPDN)
X.75
PH
Circuit Network (CSPDN)
X.71
IWU
ISUP
Leased Line Network
DCC
Signaling Network (CCS No.7, STP)
SCP
SC-Mux
EWSD Local Exchange (ISDN)
8 Mbit/s
LTG C
LTG C(B)
GS
GP
LTG A
LTG B
SN
TSI
SS
TSO
SGC
CP
MB
CCG
SSP
CCNC
CCNP
SP
STP
2 Mbit/s
(2PDC, 1CCS)
DLU
SLMA
SLMD
Magnetic Disc Unit
Magnetic Tape Unit
D & M Terminal
t/r
Basic Access:
B+B+D16
Primary Rate Access:
23 or 30 B+D64
5Ho+D64

6.2.5 Control

The switching functions of an ISDN local exchange may be roughly divided into the following functional areas, as shown in Table 6.2:

(1) Signaling
(2) Access control
(3) Switching between subscriber lines and trunks
(4) Interaction with the Operation and Maintenance functional unit.

The control functional unit comprises functional areas (2) through (4), and in practice these can be implemented as distributed functions. Thus, for instance, the EWSD system (Fig. 6.8) has a modular decentralized control structure to relieve the load on the coordination processor (CP) and this structure is characterized by a high

Fig. 6.8. System Structure and Communication Links of an ISDN Local Exchange (Siemens system EWSD).

═══	64 kbit/s user information channel (switched)
───	64 kbit/s control channel (semipermanent)
- - - -	64 kbit/s signaling channel CCS (semipermanent)
CAS	Channel associated signaling
CCNC	Common Channel Network Control
CCNP	Common Channel Network Processor
CCS	Common Channel Signaling
CP	Coordination Processor
CSPDN	Circuit Switched Public Data Network
DCC	Digital Cross Connect
DLU	Digital Line Unit
GP	Group Processor
GS	Group Switch
ISUP	ISDN User Part: for interexchange signaling within CCITT Signaling System No. 7 (see Sect. 6.3)
IWU	Interworking Unit (see Sect. 4.4.1)
LTG	Line/Trunk Group
MB	Message Buffer
PH	Packet Handler (see Sect. 4.4.4)
PSPDN	Packet Switched Public Data Network
SC-Mux	Signaling Converter – Multiplexer (includes digital to analog conversion)
SCCP	Signaling Connection Control Part (see Sect. 6.3.4)
SCP	Service Control Point (see Sects. 6.1, 6.3.7)
SGC	Switch Group Control
SLMA	Subscriber Line Module Analog
SLMD	Subscriber Line Module Digital
SN	Switching Network
SP	Signaling Point (see Sect. 6.3.1)
SS	Space Stage
SSP	Siemens Switching Processor
STP	Signaling Transfer Point (see Sect. 6.3.1)
TSI	Time Stage Incoming
TSO	Time Stage Outgoing
t/r	analog tip/ring interface

Path of No. 7 messages through the CCNC
[1]) connection of a digital multiplexer (cf. Fig. 6.6) not shown,
[2]) ISUP for link-by-link Signaling (see Sect. 6.3.3),
[3]) ISUP for End-to-end Signaling (see Sect. 6.3.3) and SCCP (see Sect. 6.3.4)

degree of preprocessing in microprocessors of peripheral units – in particular within the group processors (GP) of the line/trunk groups (LTG). The specific control of a subscriber or trunk access line (2) is to a large extent performed independently by the appropriate group processor (GP), whereas switching (3) between the internal ports of different line trunk groups (LTG) is controlled by the coordination processor. For this purpose the line trunk groups, each of which is connected via an internal 8.192 Mbit/s time-division multiplex interface (124 channels each with 64 kbit/s) to the central switching network (SN) that operates at the same rate, use 64 kbit/s channels to exchange not only information between users but also control information with the coordination processor. The 64 kbit/s channels for the control information are switched through the switching network as semipermanent connections, with serial/parallel conversion between the serial 8.192 Mbit/s switching network interface and the parallel coordination processor interface being performed by the message buffer (MB). Through-connection of the 64 kbit/s user information channels via the switching network is controlled by the coordination processor using setting commands to the switch group control (SGC).

6.2.6 Interexchange Signaling

The functional unit for interexchange signaling deals primarily with the CCITT common channel signaling system No. 7 (shown by the abbreviation CCS in Figs. 6.2, 6.5 and 6.8). The system may be used for interworking both with the exchanges in the digital telephone network (Telephone User Part TUP) [6.16 through 6.20] and with other ISDN exchanges. For interexchange signaling in the ISDN, the message transfer part (MTP) already defined [6.39 through 6.46] of Signaling System No. 7 can be used without further modification. To take account of the particular requirements of ISDN interexchange signaling, a new ISDN User Part (ISUP [6.22 through 6.26]) has been defined (see Sect. 6.3).

In addition, to take account of interworking between ISDN and existing analog parts of the telephone network, the appropriate channel-associated signaling (CAS) has to be handled in an ISDN exchange, too.

6.2.7 Operations, Administration and Maintenance

The functions for Operations, Administration and Maintenance (OAM or O&M), as defined in CCITT Recs. Q.516 [6.47] for digital local exchanges and Q.506 [6.48] for transit exchanges, include the following functional areas [6.49]:

- Operation and administration functions associated with the creation, modification and expansion of system data, e.g. exchange, network, trunk group and subscriber access data.
- Cutting over and expanding exchange equipment and service features.
- Repair and maintenance functions to preserve the operability and secure the quality of service of the switching system.

As already mentioned, the content of the Q.500 series of Recs. will be rearranged and supplemented in the CCITT Blue Book: The operations and maintenance *design objectives* applicable to local, combined, and transit exchanges will then be described in

Rec. Q.542 including network management controls, alarm handling and subscriber line maintenance and testing. The *interfaces* associated with operations, administration and maintenance (OAM) for local, combined and transit exchanges will be defined in Rec. Q.513. According to Rec. Q.513 there are two general classes of interfaces for OAM:

- Interfaces for local and remote human-machine functions using the CCITT Man-Machine Language (MML) defined in the Z.300 series of CCITT Recs.
- Interfaces to OAM computers (operations systems, network management centers) and workstations.

Operations and maintenance procedures and protocols associated with the *interexchange area*, i.e. with the common channel signaling network and the exchanges, are described in the Operations and Maintenance Application Part OMAP (see CCITT Rec. Q.795 [6.50] and Sect. 6.3).

OAM functions for the *ISDN subscriber access*, e.g. fault tracing by means of loopbacks (see Sect. 4.1.2), will be contained in the I.600 series of Recs. (cf. CCITT Blue Book). The Q.940 series of Recs. (also in the CCITT Blue Book) will describe user-network interface protocols for the exchange of management information between distributed system management entities (SME) located e.g. in a user terminal on the one hand and in a remote network management center on the other. Management functions include *maintenance* functions (e.g. fault tracing, spontaneous error reporting, error threshold alarm reporting, continuous monitoring, diagnostic testing, resource identification, resource (re)initialization, confidence testing, trouble isolation etc.), *configuration management* (e.g. routing changes, data base changes), *accounting management* (e.g. reporting of billing data), *performance management* (e.g. collecting and reporting of traffic data) and security management.

6.2.8 Timing and Network Synchronization

In synchronous digital networks the timing signals of the primary (2.048 Mbit/s) or secondary (8.448 Mbit/s) multiplex signals produced by the individual exchanges must match very precisely in order to avoid corruption of the user information (see Sect. 7.6). The *central clock generator* (*CCG*) functional unit (cf. Figs. 6.5 and 6.8) has the task of matching the timing signal generated in the exchange to the reference frequency fed in from an external source – e.g. from a reference clock located centrally in the network – and to distribute the resultant timing information within the exchange in such a way that the synchronization of the 64 kbit/s time slots is maintained as they pass through the exchange.

6.2.9 Interworking and Access to Special Equipment

As already explained in Sect. 6.1, specialized equipment for certain additional functions is not provided in every ISDN exchange. Such equipment includes

- service modules performing high layer functions (vendor feature nodes VFN),
- equipment for interworking with dedicated data networks,
- network data bases (service control points SCP, see Sect. 6.3.7)
- equipment for establishing leased line networks (digital cross connects DCC).

For reasons of economy and practical implementation, these functions are centralized. Additional information on the provision of packet switched services to ISDN subscribers as well as on interworking between ISDN and dedicated data networks can be found in Sect. 4.4

Unlike centralized interworking with dedicated data networks, interworking with the conventional telephone network is effected in all ISDN exchanges (cf. Sect. 6.2.2). The significance of this for implementation is that the analog and digital interexchange interfaces and signaling procedures of the telephone network have to be present in every ISDN exchange (Figs. 6.5 and 6.8).

6.3 Interexchange Signaling in the ISDN

6.3.1 Basic Characteristics of Interexchange Signaling Using CCITT Signaling System No. 7

To set up and clear down 64 kbit/s circuit switched network connections and control ISDN services and supplementary services, the ISDN exchanges concerned must be able to exchange signaling information with one another. Interexchange signaling in the ISDN employs CCITT Signaling System No. 7. This section deals with the basic features of this system [6.12 to 6.14, 6.51] insofar as they provide an understanding of the ISDN-specific additions and extensions to the signaling system.

Unlike conventional signaling systems for telephone networks, e.g. CCITT Signaling Systems No. 4 and No. 5 [6.52, 6.53], CCITT No. 7 is a *common channel signaling system* (Fig. 6.8.a). Common channel signaling differs from the signaling systems mentioned above in that the signaling information which relates to the 64 kbit/s user information channels is transferred in separate 64 kbit/s signaling channels which are shared by a number of user information channels. The signaling channels between the exchanges together constitute an independent signaling network (common channel signaling network) based on message interleaving which is completely separate from the user information network (Fig. 6.10).

The advantages of common channel signaling compared to channel-associated signaling include:

- *Signaling simultaneous with transfer of user information* is possible with network connection already set up.
- *Short call setup time* as a result of the high capacity (64 kbit/s) of the signaling channels.
- *Virtually unlimited set of signaling elements* (messages, message parameters) in conjunction with
- *Great flexibility* with respect to new requirements (such as those of ISDN), including the introduction of new services and supplementary services.
- The structure of signaling elements is particularly suitable for *processing in SPC exchanges.*
- *Less expensive than a channel-associated procedure* since centralized signaling equipment of a single 64 kbit/s signaling channel can be used to simultaneously control approximately 1000 user information channels.

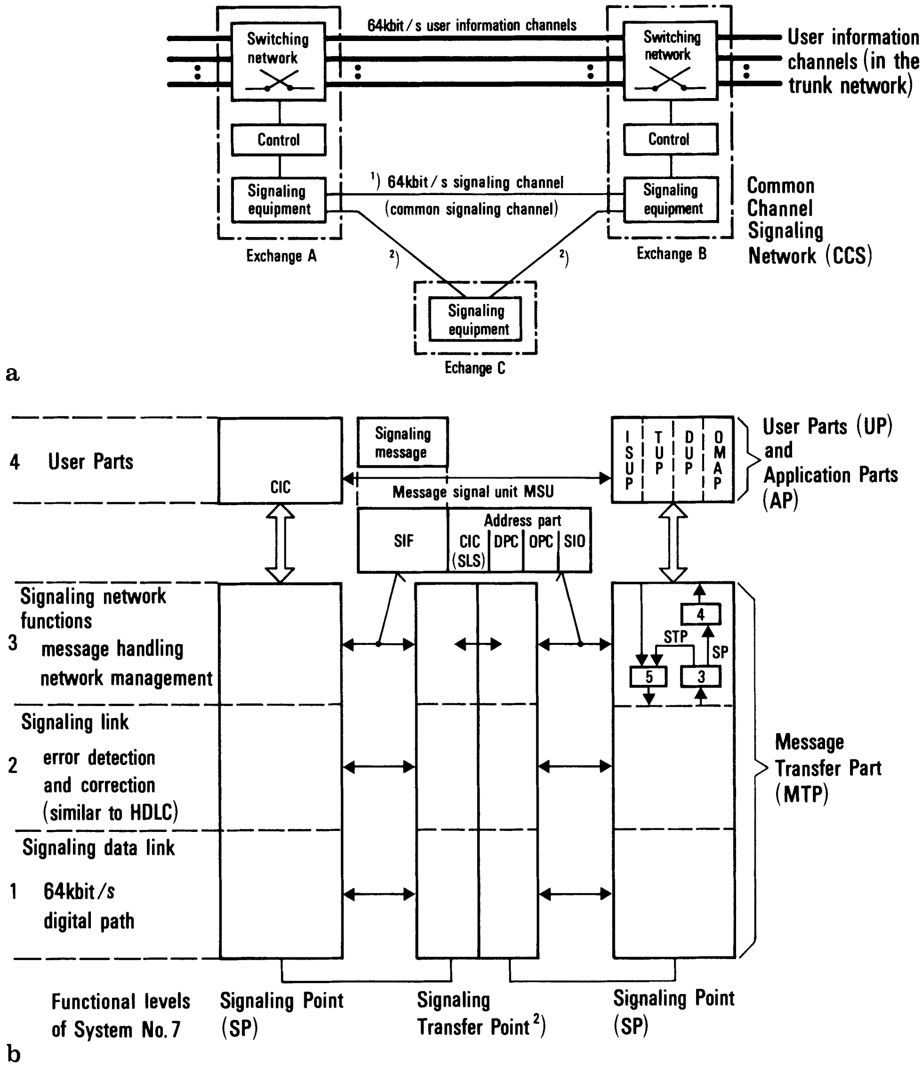


Fig. 6.9 a, b. Common Channel Signaling. **a** Principle; **b** Protocol Architecture of CCITT Signaling System No. 7.

CCS	Common Channel Signaling
ISUP	ISDN User Part
TUP	Telephone User Part
DUP	Data User part
OMAP	Operations and Maintenance Application Part
MSU	Message Signal Unit
SIF	Signaling Information Field
CIC	Circuit Identification Field
SLS	Signaling Link Selection
OPC	Originating Point Code
DPC	Destination Point Code
SIO	Service Information Octet
SI	Service Indicator
SP	Signaling Point: source or sink (with processing) of signaling messages
STP	Signaling Transfer Point: Transfer of signaling messages without processing

[1]) Associated signaling mode
[2]) quasi-associated signaling mode
[3]) Message discrimination
[4]) Message distribution
[5]) Message routing

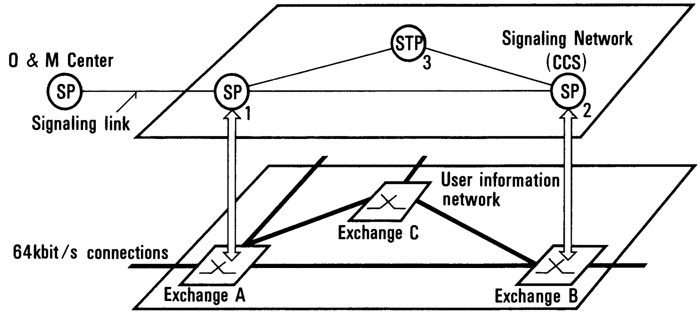


Fig. 6.10. Separate Networks for User Information Transfer and Signaling.

CCS	Common Channel Signaling
O&M	Operations and Maintenance
SP	Signaling Point
STP	Signaling Transfer Point
Signaling modes:	1–2 associated
	1–3–2 quasi-associated
A–B	Signaling relation

- *Possibility of use of the signaling network for applications outside signaling*, e.g. for transfer of operations and maintenance information between the exchanges and O&M centers.
- *Secure transfer* of the signaling elements (i.e. safeguarding against transmission errors).

Interaction between signaling network and user information network occurs during the link-by-link setup of a 64 kbit/s circuit switched network connection only in those exchanges in which the signaling information is passed to the control entity for processing (cf. Sect. 6.2.5). The latter then in turn performs the switching network settings (Fig. 6.9a). Accordingly, those nodes of the signaling network in which there is *application-oriented* processing (or creation) of signaling messages – i.e. the relevant end points (sources, sinks) of a signaling relation between two adjacent exchanges – are referred to as *Signaling Points* (SP see Fig. 6.10). In contrast, the *Signaling Transfer Points* (STP) merely perform the *transport-oriented* signaling functions, i.e. they direct the incoming signaling messages to a destination signaling point without any further processing (routing function of the signaling network). In practice SP and STP functions occur in combination in the same exchange.

The explicit separation of transport-oriented and application-oriented functions described here is reflected in the protocol architecture of CCITT Signaling System No. 7. As shown in Fig. 6.9b, the description of System No. 7 is divided into:

- *A Message Transfer Part (MTP) common to all applications*, and
- *Separate User Parts e.g. for telephony (Telephone User Part TUP* [6.16 to 6.20]), *circuit-switched data services (Data User Part DUP* [6.21] for common channel signaling in data networks) and application parts e.g. for *operations and maintenance tasks (Operations and Maintenance Application Part OMAP* [6.50]).

The special requirements of ISDN interexchange signaling result from service integration – *one* common user part for telephony *and* all non-voice services is needed

– and from the control of innovative ISDN supplementary services. Accordingly, a separate ISDN User Part (ISUP) has been created (cf. [6.15, 6.22 through 6.26] and Sects. 6.3.3 through 6.3.6).

6.3.2 The Message Transfer Part (MTP)

The following paragraphs briefly describe the main MTP functions in SS No. 7 levels 1 through 3. *Level 1* comprises the transmission and access functions of a *physical signaling channel* (signaling data link), while *level 2* ensures *reliable transfer* with protection against transmission errors of the message signal units (messages) over a signaling link to the next node of the signaling network (signaling point SP or signaling transfer point STP) using a procedure similar to HDLC.

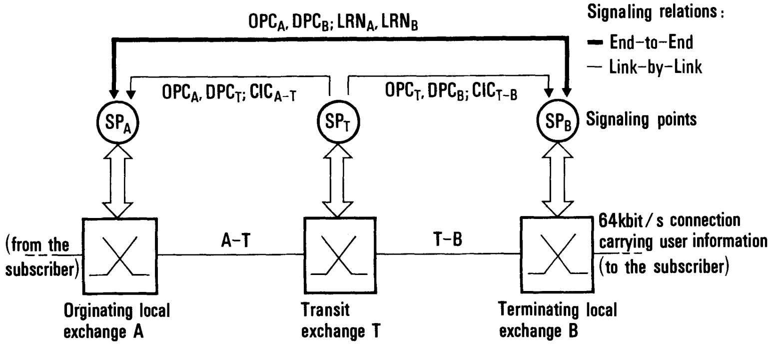
Level 3 mainly contains the signaling network functions necessary for *signaling message handling*, i.e. the functions for the distribution of the messages received at their destination point to the correct user part within the exchange, or for routing of messages destined for another exchange to the correct outgoing signaling link. Level 3 also accommodates the overall control of operations and maintenance functions for the signaling network (*signaling network management*); this includes functions such as distributing the signaling traffic load to a number of signaling links and reconfiguration of signaling routes in case of failures.

Control information used for level 3 functions includes the address of the originating exchange (Originating Point Code OPC), the address of the destination exchange (Destination Point Code DPC), identification of the signaling link chosen (Signaling Link Selection SLS) and the service information octet (SIO). Using the Service Indicator (SI), which is a subfield of the SIO, an incoming message is routed to the correct user part (ISUP, TUP...) within the same exchange (*message distribution function*), provided the preceding evaluation of the DPC indicated that this message was actually destined for that particular signaling point (*message discrimination function*). The transfer messages from other exchanges on incoming signaling links as well as the messages from the user parts within the exchange destined for onward transmission are *routed* to the appropriate outgoing signaling link in the destination direction on the basis of the DPC (connectionless datagram principle).

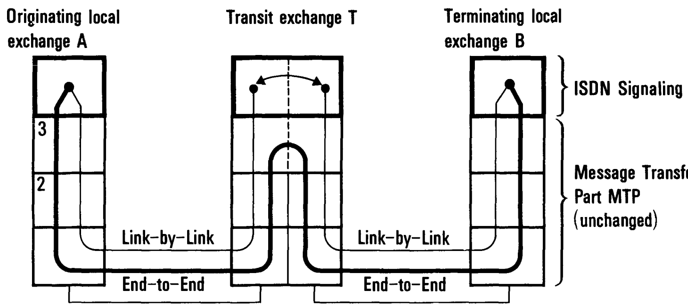
As a consequence of the shared usage of the common signaling channel by a number of network connections, the allocation of each message to a particular user information channel (circuit) has to be uniquely established; this is ensured by the CIC (Circuit Identification Code). Since the signaling link selection code (SLS) mentioned previously is a component of the CIC, this ensures that all messages associated with *one* signaling link take the same path through the common channel signaling network (in-sequence transfer of signaling messages).

6.3.3 Signaling Relations Between ISDN Exchanges

When the ISDN is introduced the circuit-related type of signaling described in Sect. 6.3.2 between pairs of adjacent exchanges throughout the network connection path will no longer be sufficient. This conventional *link-by-link* signaling between adjacent signaling points which is related to the set-up and release of a circuit between



a



b

Fig. 6.11 a, b. Signaling Relations Between ISDN Exchanges. **a** Principle; **b** Layer structure.

OPC	Originating Point Code
DPC	Destination Point Code
CIC	Circuit Identification Code (only in case of link-by-link signaling)
LRN_A, LRN_B	Local reference numbers in exchanges A or B to identify end-to-end signaling relations (logical connections) between exchanges A und B

the respective exchanges, has been supplemented for the ISDN user part (ISUP) by the new function of *end-to-end* signaling between originating and terminating ISDN local exchanges e.g. between signaling points SP_A and SP_B in Fig. 6.11a. Even though it may be related to a call, end-to-end signaling is not directly related to the control of the respective circuit switched connection. Thus, end-to-end signaling can be viewed as the capability to transfer signaling information directly between the endpoints of a circuit-switched connection (transfer of call-related information or of user-to-user information). In addition it may even be employed for exchanging signaling information between signaling points which are not interconnected by a circuit-switched connection (see Sect. 6.3.7).

In case of end-to-end signaling, the intermediate signaling points of the transit exchanges (cf. SP_T in Fig. 6.11a) are by-passed. This can be achieved by the SCCP method described in more detail in Sect. 6.3.4, where the transit exchanges act as signaling transfer points STP (Fig. 6.11b). ISDN exchanges providing interworking with the telephone network and gateway exchanges to foreign ISDNs or to data networks act as originating and terminating exchanges as regards end-to-end signaling.

The main objectives of end-to-end signaling are

- *Relieving the load on the ISUP in the transit exchanges* so that they do not have to process the extra signaling traffic resulting from the control of supplementary services, such as changing from voice to data and back during an established call (see in-call modification in Sect. 6.3.6). These supplementary services are always controlled in the ISDN local exchanges.
- The option of *signaling when a network connection does not exist* or it has already been cleared down, e.g. signaling for a supplementary service such as *completion of calls to busy subscribers.*

The *end-to-end signaling* functions include messages for

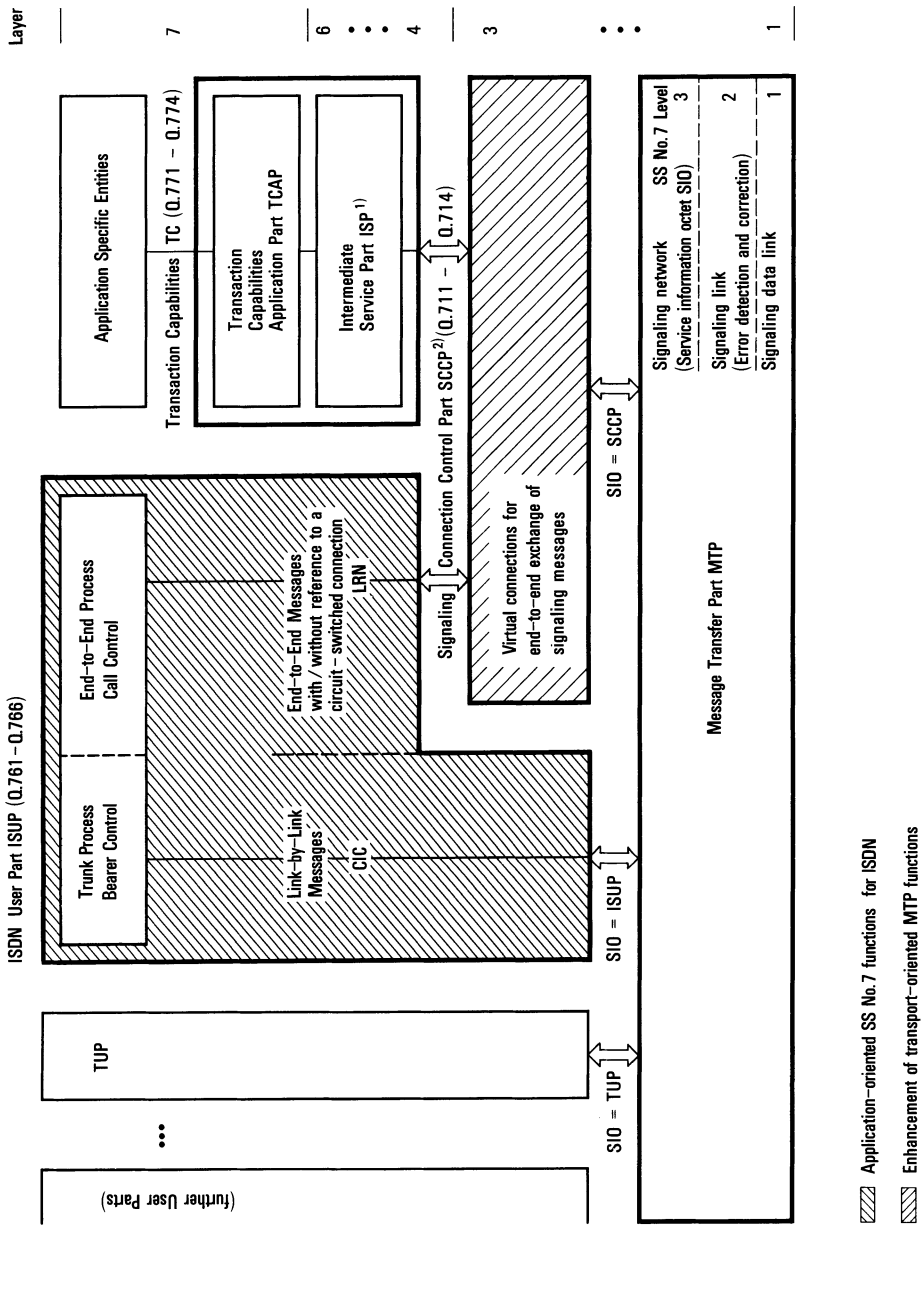
- Requesting a supplementary service:
  - FRQ Facility Request
  - FACD Facility Accepted
  - FRJ Facility Reject
- Transmitting information relevant to supplementary services:
  - FIN Facility Information
- Deactivation of a supplementary service:
  - FDE Facility deactivated
- Information of the called user when the calling user suspends the call (without releasing it) and then resumes it again e.g. for moving a terminal from one socket to another in the case of a bus configuration (see Sect. 4.3.5.4):
  - PAU Pause
  - RES Resume
- Information request from an interworking exchange or gateway exchange
  - IRM Information request
  - INF Information
- Transfer of user-to-user information (cf. Fig. 6.20); the function of the ISUP message USER INFO corresponds to the D-channel message with the same name (see Sects. 4.3.5.5 and 6.4.3):
  - USER INFO User-to-user information

The setup and release of the circuit switched connection (basic call control), on the other hand, is controlled by *link-by-link* messages, as it is in the TUP. These messages can be evaluated by each exchange involved in the path of the overall network connection. A significant difference from the TUP is that call release can be initiated by the called or the calling user, as in data networks. The category of link-by-link transferred ISDN signaling messages includes:

- *Call establishment* messages (cf. Fig. 6.13a):

IAM Initial address

Initializing message sent in the forward direction to initiate seizure of an outgoing circuit and to transmit address information (complete directory number of the called user for en bloc dialing or for overlap (digit-by-digit) dialing insofar as required for routing to the national destination exchange); other information required for call processing, e.g. supplementary services to be taken into account during call setup (e.g. reverse charging).


Layer
7
6
4
3
1
Application Specific Entities
Transaction Capabilities TC (Q.771 – Q.774)
Transaction Capabilities Application Part TCAP
Intermediate Service Part ISP 1)
Signaling Connection Control Part SCCP 2) (Q.711 – Q.714)
Virtual connections for end-to-end exchange of signaling messages
ISDN User Part ISUP (Q.761 – Q.766)
Trunk Process Bearer Control
End-to-End Process Call Control
Link-by-Link Messages
CIC
End-to-End Messages with/without reference to a circuit-switched connection
LRN
TUP
(further User Parts)
SIO = TUP
SIO = ISUP
SIO = SCCP
Message Transfer Part MTP
Signaling network (Service information octet SIO)
SS No.7 Level
3
Signaling link (Error detection and correction)
2
Signaling data link
1
Application-oriented SS No.7 functions for ISDN
Enhancement of transport-oriented MTP functions

SAM Subsequent address
Transports the digits not contained in the IAM if necessary (e.g. in case of overlap dialing).

ACM Address complete
The called user is free (and compatible) and has responded to the call with ALERTing.

ANS Answer
Call accepted by the called user with CONNect.

- *Release* messages (cf. Fig. 6.13b)

REL Release
With REL, one (or both) of the local exchanges initiates the release of the user information channel in the network, as soon as the local subscriber releases with DISConnect. Receipt of a REL message causes the transit exchanges or the released local exchange to disconnect the user information channel and to send an RLC message (see below) in the backward direction as an acknowledgement.

RLC Release Complete
Acknowledgement for REL.

6.3.4 Protocol Architecture of ISDN Interexchange Signaling

To implement the end-to-end signaling between ISDN originating and terminating local exchanges explained in the previous section, the transport-oriented functions implemented by the message transfer part MTP of Signaling System No. 7 must be expanded. In order to avoid affecting the existing user parts which are based directly on the MTP (TUP...), this functional expansion has been undertaken above the otherwise unaltered MTP (Fig. 6.12).

The ISUP signaling procedures in CCITT Rec. Q.764 [6.25] make provision for two different methods of end-to-end signaling:

- With the *SCCP method* the ISDN user part ISUP [6.22 through 6.26] makes use of the services of a new *Signaling Connection Control Part* (SCCP, see CCITT Rec. Q.711 through Q.714 [6.54 through 6.57]) introduced between the Message

Fig. 6.12. Protocol Architecture of CCITT Signaling System No. 7.
ISUP ISDN User Part: ISDN interexchange signaling as per CCITT Rec. Q.761–766 [6.22 through 6.26]
TUP Telephone User Part
SCCP Signaling Connection Control Part as per CCITT Rec. Q.711–714 [6.54 through 6.57]
TC Transaction Capabilities as per CCITT Rec. Q. 771–774 [6.58 through 6.61] provide functions and protocols for a large variety of applications distributed over exchanges and specialized centers in telecommunications networks
CIC Circuit Identification Code
LRN Local Reference Number for identifying end-to-end signaling connections
SIO Service Information Octet
[1]) comprises OSI Transport, Session and Presentation Layers (X.224, X.225; X.208, X.209)
[2]) provides OSI Network Layer Service in conjunction with the MTP: connectionless (classes 0,1) or connection-oriented (classes 2 to 4)

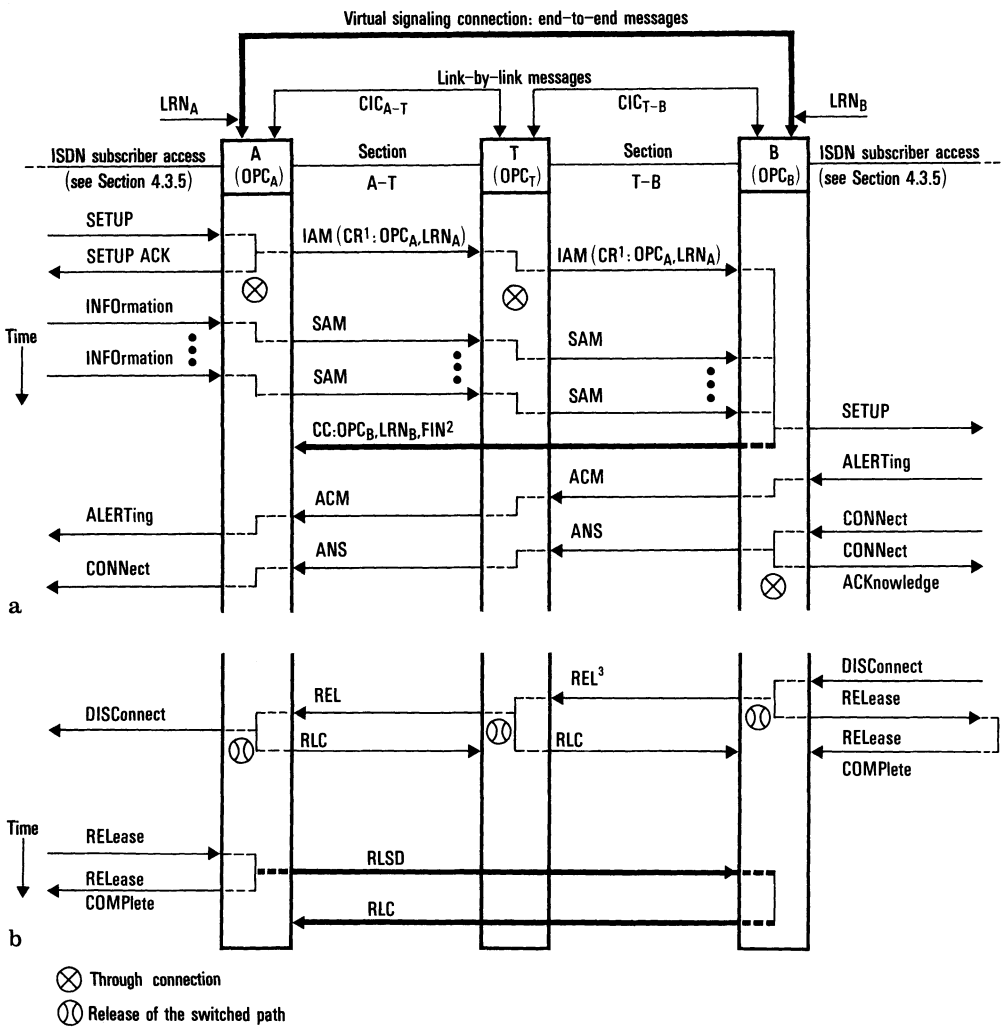


Fig. 6.13 a, b. Set-up and Release of a Circuit Switched Connection in the ISDN. **a** Set-up of the 64 kbit/s connection (with overlap sending) and of the virtual end-to-end signaling connection; **b** Cleardown of the 64 kbit/s connection (initiated by the called terminal) and of the virtual end-to-end signaling connection. IAM, SAM, ACM, ...: ISUP messages to set up and release the 64 kbit/s connection (see Sect. 6.3.3).
CR, CC, RLSD, RLC: SCCP Messages to set up and release the virtual signaling connection between local exchanges A and B (see Sect. 6.3.4).
A Originating local exchange
T Transit exchange
B Terminating local exchange
OPC Originating Point Code
DPC Destination Point Code
CIC Circuit Identification Code
LRN Local Reference Number for identifying the end-to-end signaling connection
[1]) Implicit setup of end-to-end signaling connection by embedding CR in IAM
[2]) Facility information (FIN) message: A-exchange is informed on ISDN subscriber-related service attributes available at B-subscriber
[3]) Release can be initiated either by the calling party (A-subscriber) or the called party (B-subscriber)

Transfer Part and the ISUP. In accordance with the principle of protocol layering this means that ISUP messages for end-to-end signaling are conveyed by SCCP messages (see Data Form 1 below) which in turn are transported in the signaling information field (SIF) of MTP message signal units (MSU). In contrast to the pass-along method (see below), the SCCP method is independent of the presence of a circuit-switched connection between the message originating and terminating local exchanges. The end-to-end signaling connections between the ISDN local exchanges are established in this case exclusively via the MTP function in the transit exchanges (see Fig. 6.11).

- In the case of the *pass-along method* the end-to-end signaling information is passed on a link-by-link basis along the path of transit exchanges at which the sections of the circuit-switched connection are interconnected. Forwarding of incoming end-to-end messages to an outgoing link to the next exchange is effected by the ISUP in the transit exchanges as in the case of link-by-link messages (see Fig. 6.11). End-to-end messages are not processed in the transit exchanges, however, but simply readdressed for subsequent transmission on the outgoing link.

With the SCCP method, the ISUP has two separate layer interfaces for transporting signaling messages, as shown in Fig. 6.12.:

- a direct *interface to the MTP* for *link-by-link* messages and
- an indirect access to the MTP via the *interface to the SCCP* for *end-to-end* messages.

As illustrated in Fig. 6.12, it is the task of the SCCP to raise the level of MTP services up to that available at the OSI layer 3 interface. The OSI network layer service is thus provided by the combination of MTP plus SCCP, also referred to as the Network Service Part NSP (cf. Fig. 6.16). As a consequence, non-circuit related S.S. No. 7 signaling functions built on top of the SCCP in accordance with the OSI reference model (CCITT Rec. X.200) are called application parts such as the Operations and Maintenance Application Part OMAP [6.50] and the Transaction Capabilities Application Part TCAP (CCITT Recs. Q.771 through Q.774 [6.58 through 6.61]). The TCAP is used to access network data bases (see Sects. 6.1 and 6.3.7), e.g. for number translation in conjunction with the freephone service (e.g. service "800" or "130").

End-to-end signaling connections can be set up and released by means of the following SCCP messages (cf. Fig. 6.13):

CR	Connection Request
CC	Connection Confirm
RLSD	Release
RLS	Release Complete

The SCCP message

DT1 Data Form 1 (cf. Fig. 6.20)

is used as a transparent transport container for end-to-end signaling messages of the ISUP via existing end-to-end signaling connections.

As already mentioned in Sect. 6.1, two principal functions of an ISDN exchange are call control and bearer control (i.e. connection control). In the updated CCITT Recs.Q.761 to Q.766 to be published in the Blue Book (1989), call control functions and bearer control functions common to all bearer services will be contained in a single functional block comprising both end-to-end and trunk processes (Fig. 6.12). As compared to discussions in CCITT aiming at a *separated* ISUP consisting of a functional block for call control and separate specialized functional blocks for each bearer service, the approach in the Blue Book and in the Red Book is referred to as the *monolithic* ISUP:

- The *end-to-end process* performs control and coordination of requests for circuit switched connections, packet switched connections (in case of single step call establishment, see Sect. 4.5), virtual signaling connections (SCCP), etc.
- *The trunk process* establishes various types of network connections on a link-by-link basis; in contrast to the end-to-end process it is therefore present in the transit exchanges, too.

6.3.5 Implementation of ISDN Interexchange Signaling in the Exchange

With the Siemens EWSD system, line-type-specific control of subscriber and trunk lines is provided largely by the group processors (GP) in the peripheral line/trunk groups (LTG), in keeping with the concept of a modular, decentralized control structure (Sect. 6.2.5 and [6.4, 6.6]). The *application-oriented No. 7 functions* of a signaling point SP, i.e. those of the ISDN User Part ISUP for example, are thus implemented for all user information channels routed via a particular LTG in the GP of the LTG concerned (Fig. 6.8).

The LTG “C” units on the interexchange side assume the link-by-link signaling functions under these circumstances, whereas the end-to-end signaling is implemented in the LTG “B” unit on the subscriber access side on account of the close connection with user-network signaling. This also applies to the Signaling Connection Control Part SCCP which is another “user” of the message transfer part MTP and expands the MTP functions to include end-to-end transport of ISUP messages.

The No. 7 functional levels 2 (signaling link) and 3 (signaling network) of the *Message Transfer Part MTP*, i.e. functions such as those of a signaling transfer point STP, are handled by a Common Channel Signaling Network Control (CCNC) subsystem designed specially for the common channel signaling network. The 64 kbit/s signaling channels are routed transparently from the LTG “C” units via fixed switching network paths (semipermanent connections) to the CCNC. Incoming ISUP messages intended for a signaling point SP within the same exchange are directed by the CCNC via the CCNC/CP interface and the semipermanent 64 kbit/s control channel between the message buffer MB and the GP (cf. Sect. 6.2.5) to the correct destination LTG for evaluation; outgoing ISUP messages from the local signaling point SP go in the opposite direction. The CCNC undertakes routing of transfer messages as well as outgoing messages from the local SP to a LTG C corresponding to the outgoing signaling link.

6.3.6 Example for ISDN Interexchange Signaling: In-call Modification

A typical ISDN feature is the possibility to modify the characteristics of an established call, e.g. from a voice call to a data call and, if necessary, back to a voice call again. In more general terms, the calling or called user may initiate a change of service during an active call. Such a change of service may imply a modification of the network capabilities (see ISDN connection types in Table 6.1) that have originally been allocated to that call during call establishment.

For this purpose CCITT is currently studying a procedure called *in-call modification.* As an example, Fig. 6.14 depicts changeover from *digital speech* to 64 kbit/s *unrestricted digital information* for non-voice applications. While bit manipulation equipment may be required in the first case, e.g. echo suppressors and/or A-law/μ-law conversion (see Sect. 7.2.1) at international boundaries, a transparent 64 kbit/s connection has to be provided in the latter case. When initiated by the calling or called user by sending a MODify message, the in-call modification procedure causes both the network and the remote user to check that the resources needed for a change of the call mode are available. If the requested call mode can be supported, the initiating user is informed of successful completion by means of a MODify COMPlete message.

It should be noted that CCITT has not yet defined change of service. A procedure for *change of service* will probably be based on the in-call modification mechanism

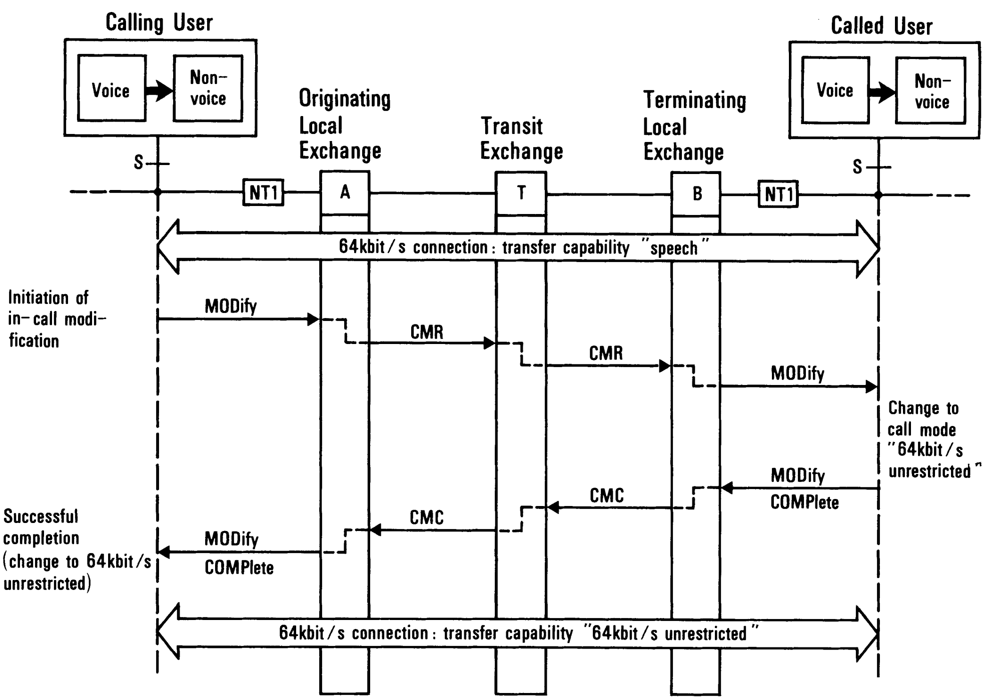


Fig. 6.14. In-call Modification.

CMR Call modification request (ISUP message)
CMC Call modification complete (ISUP message)
MOD Modify (tentative I.451 message to initiate a change of call mode)
MOD COM Modify Complete (tentative I.451 message indicating that the requested call mode can be supported)
speech, 64 kbit/s unrestricted (digital information): examples of call modes (cf. information transfer susceptance in Table 6.1 and bearer capability in Sect. 4.3.3.2)

described above. If service specific terminals are used, an additional requirement is to enable a coordinated transfer of the B-channel from the "old" terminal to the "new" terminal while maintaining the 64 kbit/s connection in the network. A possible solution implemented in the German ISDN and described in Fig. 6.20b in conjunction with PABX extensions is to subdivide the transfer of the B-channel into a call clearing signaling sequence for the "old" terminal and a call establishment sequence for the "new" terminal.

6.3.7 Communication Between ISDN Exchanges and Network Data Bases via CCITT Signaling System No. 7

According to the concept of the *intelligent network* [6.31 through 6.33], as already mentioned in Sect. 6.1, call processing for calls requiring special handling can be distributed between the ISDN exchanges, called *Service Switching Points SSP,* on the one hand and internal network data bases, termed *Service Control Points* SCP on the other (see Fig. 6.15). The overall objective of this partitioning of functions is improved flexibility of call processing in terms of more rapid introduction of new services by administrations as well as to enable customization of services for unique customer needs.

Another important aspect is to give customers control over their service features by allowing direct customer access to the *Service Management System SMS*, e.g. for changing the call routing information in conjunction with service 800 (see below) or even to reconfigure a virtual corporate network which is embedded in the resources of the public ISDN.

The key concept of the intelligent network is to move the control logic and the customer record out of the local exchange (SSP) into the network data base (SCP) or in some cases even into a vendor feature node VFN, as regards the handling of special calls. The corresponding files of the SCP are administered via the SMS. This approach entails the decomposition of the SSP functionality into basic switching primitives which would be controlled by corresponding protocol commands from the SCP. To this aim the SSP must access the network data base on a real time basis during call control, e.g. for translation of a virtual number into the real target number in conjunction with the freephone or green number service (e.g. service "800" or "130"). *Non-circuit-related interactions* (e.g. query/response, i.e. operation requesting an action to be performed by the remote end/reply indicating success or failure) between the SSP and the SCP are effected via the common channel signaling system by means of transaction capabilities (see below). Similar interactions via the D-channel protocol can also be envisaged between the SSP and a VFN or between the SSP and the end user, thus resulting in a three-way interworking among the end user, the SCP and the VFN (see Fig. 6.15 and [6.31]).

Transaction capabilities TC (CCITT Draft Recs. Q.771 through Q.774 [6.58 through 6.61]) provide functions and protocols to support a large variety of applications distributed over exchanges and specialized equipment in telecommunication networks. According to Fig. 6.12, TC are built on top of the SCCP, i.e. are using the layer services of the Signaling Connection Control Part SCCP of SS No. 7. As shown in Fig. 6.16 the detailed protocol architecture of TC consists of the *Transaction*

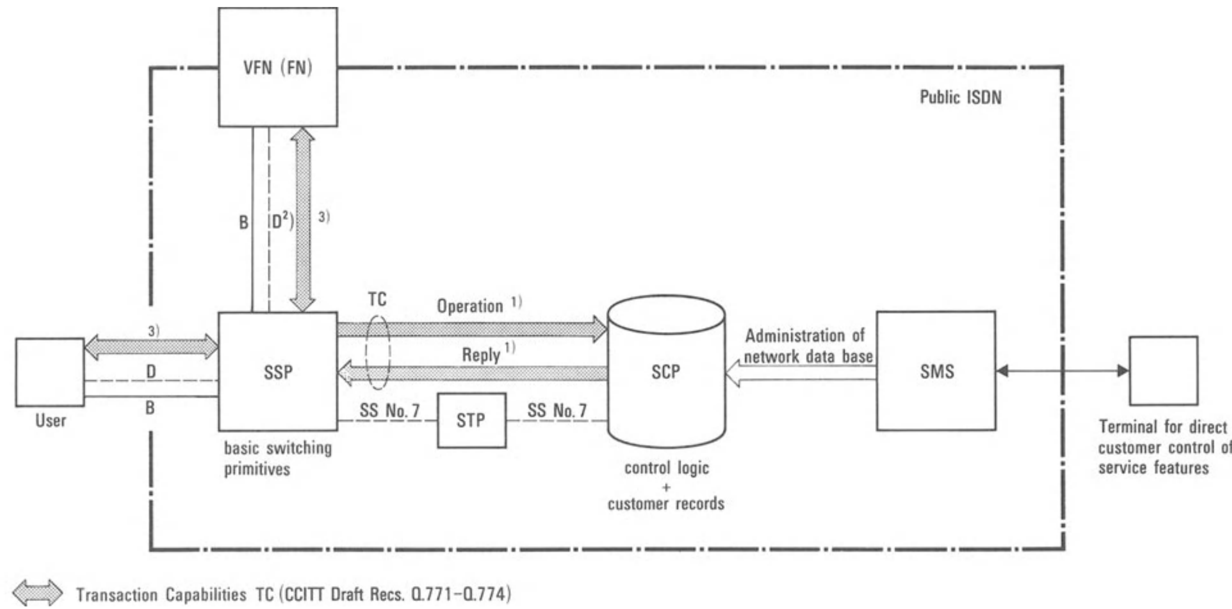


Fig. 6.15. Intelligent Network with Distributed Control.

———	user information (B)
– – – –	signaling (D)
SSP	Service Switching Point (ISDN exchange)
SCP	Service Control Point (network data base)
SMS	Service Management System
(V) FN	(Vendor) Feature Node (see Sect. 6.1)
STP	Signaling Transfer Point (see Sect. 6.3)
S.S. No. 7	CCITT Signaling System No. 7

[1]) cf. component sublayer of TCAP in Fig. 6.16
[2]) VFN (FN) attached via ISDN User-network interface
[3]) Non-circuit related interaction via layer 3 of the D-channel protocol (Rec. I.451)

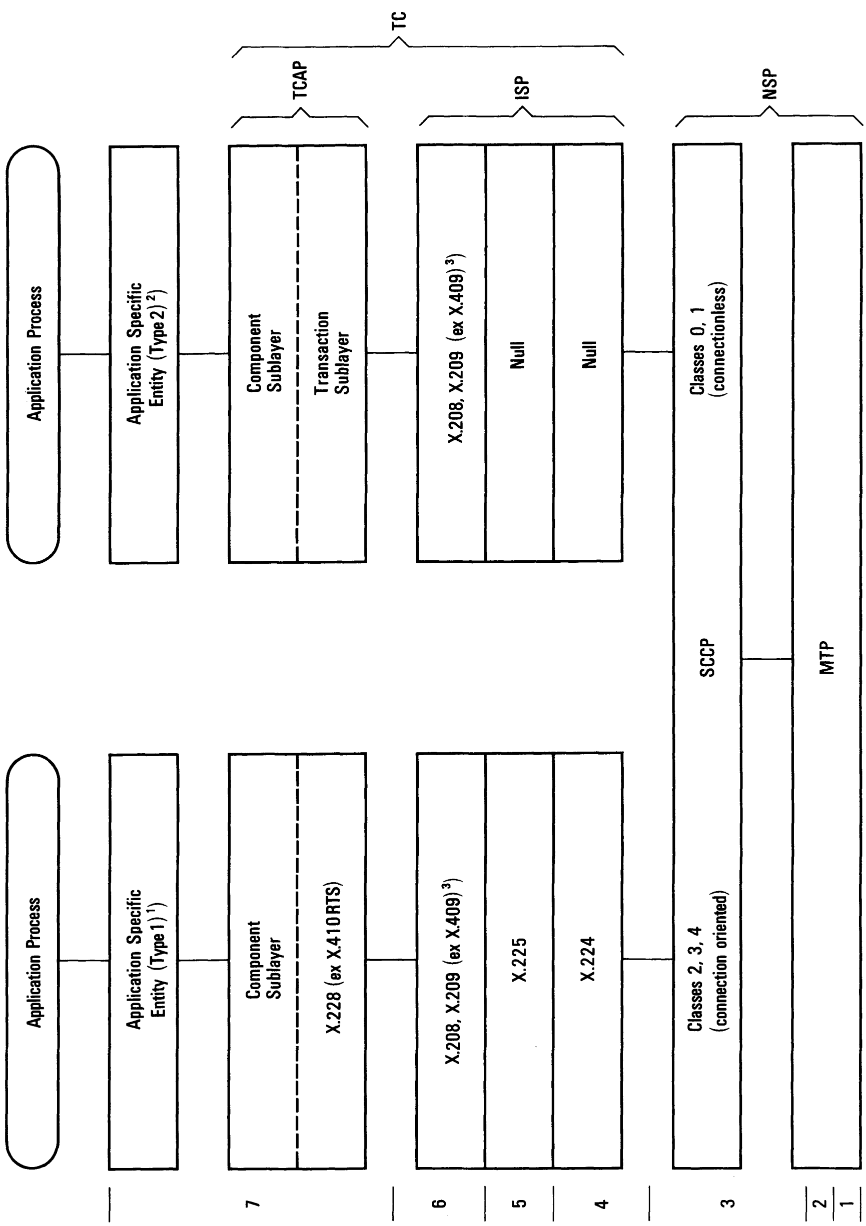


Fig. 6.16. Protocol Architecture of Transaction Capabilities.

TC Transaction Capabilities
TCAP Transaction Capabilities Application Part
ISP Intermediate Service Part
NSP Network Service Part
SCCP Signaling Connection Control Part
MTP Message Transfer Part

[1]) low real time requirements, high data throughput (e.g. OMAP)

[2]) stringent real time requirements, low data throughput (e.g. access to network data base during call control)

[3]) Abstract Syntax Notation One ASN.1 (X.208) and Basic Encoding Rules for ASN.1 (X.209)

Capabilities Application Part TCAP in OSI layer 7 and supporting OSI functionality in layers 4 to 6, called Intermediate Service Part ISP. The remainder of layer 7 deals with application specific entities, i.e. the TC users. While the *component sublayer* of the TCAP is devoted to operations and the corresponding replies from the remote end, the *transaction sublayer* enables setting up end-to-end relations, called transactions.

6.4 Corporate ISDN Networks

6.4.1 Fundamental Solutions

The advantages of integrated voice and data features inherent in the ISDN concept are most useful for business applications. As a consequence, ISDNs can be expected to form the "hub" of current and future business communication and office automation networks.

On the other hand, corporate networks linking multiple company locations have traditionally been characterized by enhanced requirements as compared to public networks. Examples include:

- Wider range of features
- Private facilities, e.g. tie trunks, paging, recorded dictation
- Private numbering plan
- Attendant features, e.g. attendant-assisted calls, directory assistance, message desk service
- Interconnection of multiple customer locations with feature transparency, as if the extensions were on a single switch (e.g. city-wide Centrex)
- Communication management features, e.g. station message detail recording (such as reports on dates, times and length of calls), traffic data to customer premises equipment
- Customer control features, e.g. access to outgoing facilities, customer station rearrangement (with regard to feature and number allocation)

To meet the extended communication requirements of corporate ISDNs, fundamentally different solutions can be utilized. Depending on the regulatory framework and the customer's requirements, these different solutions can in practice complement each other rather than substitute for each other:

- *Private corporate networks* composed of interconnected on-site ISDN private automatic branch exchanges (ISDN PABX or ISPBX) which provide the required features such as those listed above (refer to Fig. 6.17a and [6.5, 6.62, 6.63]); this topic is covered in Sects. 6.4.2 and 6.4.3.
- Corporate networks utilizing the local exchange for providing the required features (such as those listed above). This approach is traditionally known as *Centrex* (*Central Exchange*) where local public ISDN exchanges provide PABX-like features (refer to Fig. 6.17b and [6.64, 6.65]). In practice, a Centrex group may be composed of both PABX and directly connected subscribers. Centrex can provide for both dedicated and virtual private networks. The distinction is that virtual private networks utilize the public trunk network with a private software-defined overlay guaranteeing a specific amount of customer access and control; dedicated private networks, on the other hand, employ private trunks.

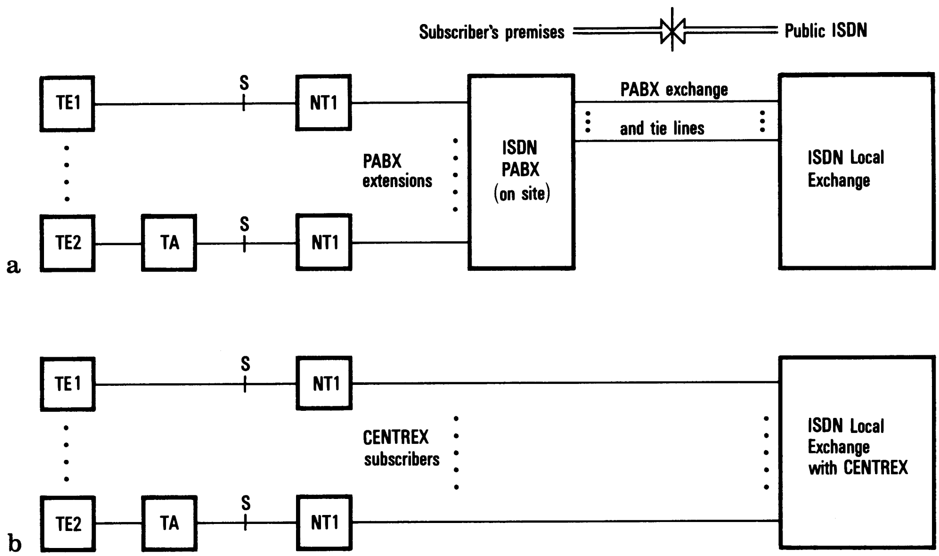


Fig. 6.17 a, b. Basic Solutions for ISDN Corporate Networks. **a** On-site ISDN PABX (private corporate network); **b** Centrex system (corporate network utilizing the local exchange for PABX-like features).

PABX	Private automatic branch exchange
CENTREX	Central exchange
TE1	ISDN terminal equipment with S/T interface
TE2	Terminal with non-ISDN interface
TA	Terminal adaptor (see Sect. 4.4)
NT1	Network termination

6.4.2 Structure and Features of an ISDN PABX

Figure 6.18 shows the structure of an ISDN PABX, which forms the central element of an integrated office communication system (e.g. Siemens System HICOM [6.5, 6.62]. Such a system may comprise the following functional areas:

- Basic circuit-switched system,
- Local area network (LAN),
- Packet Handler providing X.25 data communication services,
- Interconnection of PABXs to form ISDN corporate networks (see Fig. 6.20 and Sect. 6.4.3),
- Special equipment for high layer services and functions beyond information transport, namely
  - *Storage*: Private database systems, inhouse Videotex centers
  - *Processing*: Data processing systems (DPS)
    - subscriber-like connection of a DPS to allow local or remote terminals to access the DPS via the PABX
    - Connection of a DPS via an administration and data server for operations and maintenance linkage between PABX (control) and DPS, e.g. for recording call charge data, centralized operations, remote operations administration and maintenance, and DPS-controlled call setup.

- *Store and forward communication*: Voice mailbox and text mailbox systems (message handling systems)
- *Compatibility*: protocol conversion between terminal and host computer, interworking between different services (e.g. from teletex to facsimile).

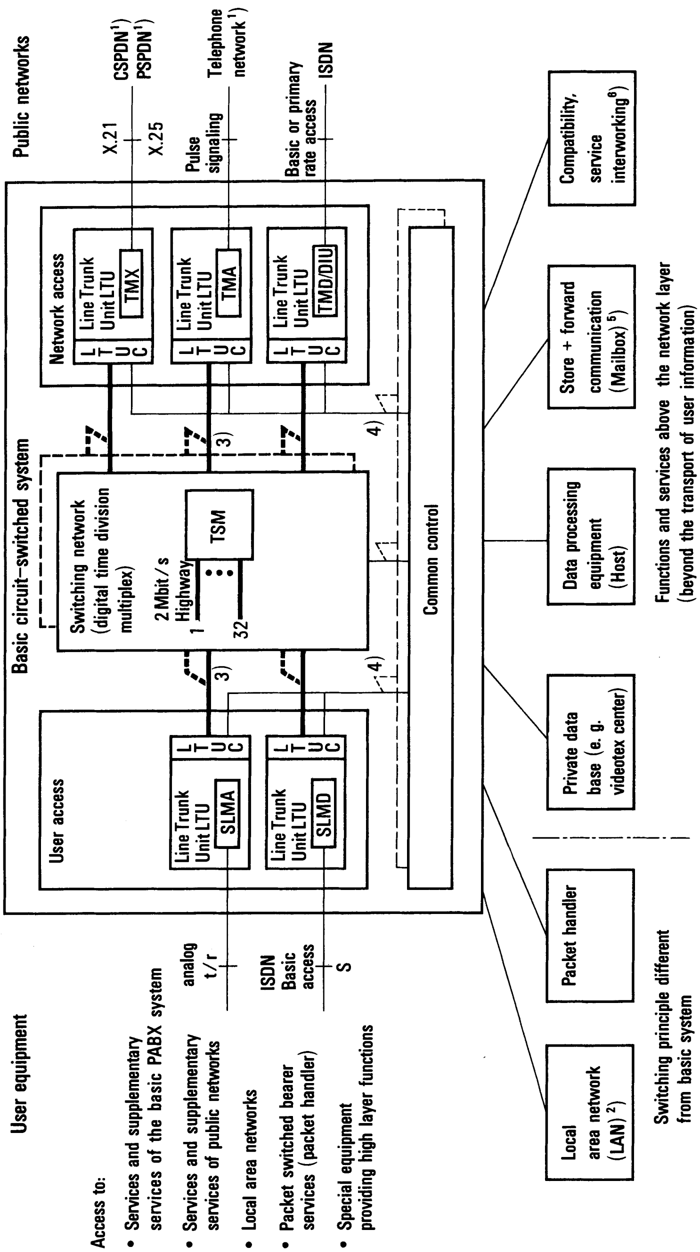
The structure depicted in Fig. 6.18 is based on a digital time-division multiplex switching system; this implicitly defines the following characteristics for a corporate network for office communications:

- *One* network for *all* types of communication, including voice communication,
- *Star* topology with *central* control, i.e., the network structure currently widely used in PABX networks is retained, and the use of the existing inhouse infrastructure of the telephone line network is continued;
- *Circuit switching* supplemented by access to X.25 packet switched services.

The *basic circuit-switched system* (Fig. 6.18) in the ISDN PABX is formed by a digital time-division multiplex switching system. Structure and functions of a private ISDN switching system thus broadly correspond to those described in detail in Fig. 6.5 for public ISDN exchanges. In addition to the peripheral units for user access and for network access, which perform the interface and signaling functions, the PABX, like the public exchange, has central components, duplicated for security reasons, i.e. a non-blocking digital switching network using time-division multiplexing for through-connection of the 64 kbit/s channels and an SPC common control. The functional units (Line Trunk Units LTU) of the periphery accommodate modules for the user access (Subscriber Line Module SLM...) or for the network access (Trunk Module TM...). Typically, the LTUs are linked to the switching network via internal time-division multiplexed 2 Mbit/s highways. Control information is exchanged with common control via separate control paths to relieve the load on the switching network.

In the *user access area* of the ISDN PABX (Fig. 6.19a) the internationally standardized D-channel protocol (S interface) for public ISDN systems can also be used, at least as a basis. The resulting advantage of terminal portability between public and private subscriber lines does not have to be at the cost of reduced flexibility – e.g. with regard to the larger range of features and supplementary services traditionally provided by the PABX as compared to public networks – since in the standard CCITT signaling protocol provision has already been made for the protocol to be tailored to the needs of PABXs, i.e. by additional information elements in the signaling messages (cf. Sect. 4.3.5).

When ISDN PABXs are introduced, it must be ensured that the *existing analog terminal environment* can remain connected, at least for a transitional period: when analog subscriber lines are connected *directly* to a t/r interface, analog/digital conversion of the user information and conversion of the signaling for call establishment and clearing must be performed in peripheral units of the PABX – just as in the case of analog subscriber lines connected to public digital exchanges. For existing non-voice terminals of the analog telephone network it is an advantage to have an *indirect* connection via the ISDN user access with a terminal adaptor (TA) converting either the analog t/r interface (TA – t/r) or the digital V. interface (TA-V.) to the S interface (cf. Fig. 6.19a). In contrast to direct connection, indirect connection allows ISDN supplementary services to be implemented via the S interface using the terminal adaptor.


Public networks
Basic circuit-switched system
User equipment
Network access
User access
Line Trunk Unit LTU
TMX
TMA
TMD/DIU
SLMA
SLMD
LTUC
Switching network (digital time division multiplex)
2 Mbit/s Highway
1
32
TSM
Common control
3)
4)
X.21
X.25
CSPDN[1])
PSPDN[1])
Pulse signaling
Telephone network[1])
Basic or primary rate access
ISDN
analog t/r
ISDN Basic access
S
Access to:
• Services and supplementary services of the basic PABX system
• Services and supplementary services of public networks
• Local area networks
• Packet switched bearer services (packet handler)
• Special equipment providing high layer functions
Local area network (LAN)[2])
Packet handler
Private data base (e. g. videotex center)
Data processing equipment (Host)
Store + forward communication (Mailbox)[5])
Compatibility, service interworking[6])
Switching principle different from basic system
Functions and services above the network layer (beyond the transport of user information)

Even before the corresponding services are offered in a public ISDN or before the necessary interworking facilities are implemented from the public ISDN to the existing dedicated data networks (Fig. 6.19b), an ISDN PABX may provide the following network connections (Fig. 6.18) on the *network side* in addition to the connection to the ISDN:

- to the conventional telephone network,
- to a circuit-switched public data network (CSPDN),
- to a packet-switched public data network (PSPDN).

ISDN PABXs with direct dialing-in capabilities can be regarded as switching equipment similar to the exchanges of the public ISDN since they handle services and supplementary services for their users as the public ISDN local exchange does for public ISDN subscribers. However, the ISDN *inter-exchange signaling protocols* of CCITT Signaling System No. 7 (Message Transfer Part, ISDN User Part ISUP) described in Sect. 6.3 are not used for interfacing ISDN local exchanges, but rather the D-channel protocols designed for the subscriber access side of the public ISDN (see Sect. 4.3); this also applies to the primary rate access. This produces uniform signaling protocols for the user and network access side of the ISDN PABX, provided that on the user access side of the ISDN PABX, as already discussed, the CCITT standard D-channel protocol of the public ISDN is applied (cf. Sect. 6.4.3).

The basic circuit-switched system can be supplemented by *higher layer functions*, i.e. functions which go beyond pure information transport (Fig. 6.18). This extension of the basic system by storage, processing, store-and-forward communication and compatibility capabilities is particularly cost-effective when implemented by modular integration in the form of specialized equipment, commonly referred to as *service modules* or *servers*. Service modules (for voice and text mail, for example; cf. Sect. 2.1)

Fig. 6.18. Basic Structure of an ISDN PABX as the "Hub" of an Office Communication System.

LTU	Line trunk unit
LTUC	LTU control
SLMA	Subscriber line module analog
SLMD	Subscriber line module digital
TSM	Time stage module
TMX	Trunk module with X. interface signaling
TMA	Trunk module analog
TMD	Trunk module digital (ISDN basic access)
DIU	Digital interface unit (ISDN primary rate access)
CSPDN	Circuit-switched public data network
PSPDN	Packet-switched public data network

[1]) as an interim solution prior to the introduction of a public ISDN that provides interworking with dedicated data networks and with the telephone network

[2]) Self-contained network with separate numbering scheme and with non-ISDN compatible access interfaces and protocols

[3]) Exchange of user information via a 2 Mbit/s highway (internal time division multiplex system)

[4]) Exchange of signaling information (control information)

[5]) Servers for voice mail and text mail and store-and-forward features for facsimile and teletex terminals

[6]) Protocol conversion between terminals and data processing equipment (DPE), interworking between common control and DPE or interworking between teletex and facsimile terminals

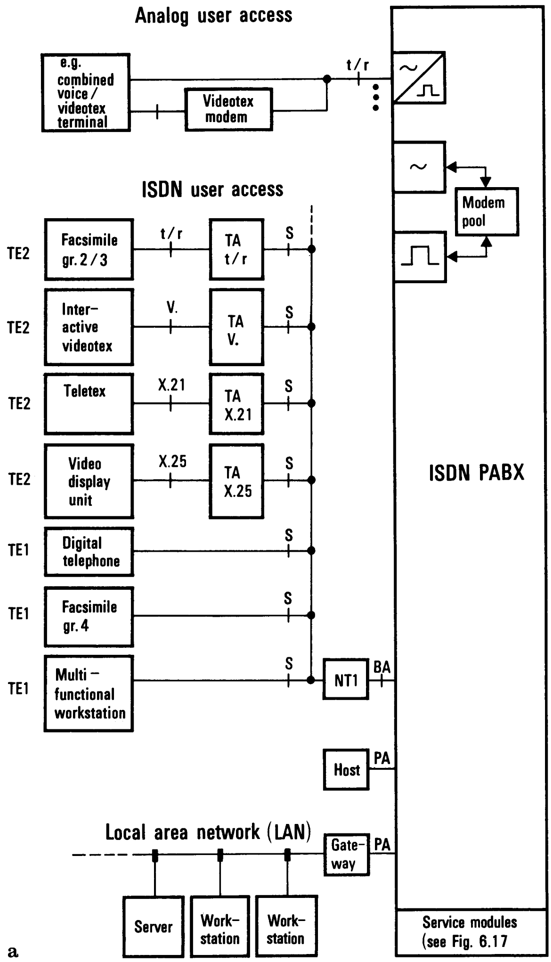


Fig. 6.19 a, b. ISDN Private Automatic Branch Exchange. **a** User access; **b** Access to public and private networks.

TE1	ISDN terminal equipment with S interface
TE2	Terminal with non-ISDN interface
TA	Terminal adaptor (see Sect. 4.4)
NT1	Network termination
BA	ISDN basic access: $B+B+D_{16}$
PA	ISDN primary rate access: $30 \times B+D_{64}$ (2.048 Mbit/s) or $23 \times B+D_{64}$ (1.544 Mbit/s)
PH	Packet Handler (see Sect. 4.4.4)
IWU	Interworking unit (see Sect. 4.4)
DTE	Data terminal equipment
ISUP	ISDN interexchange signaling (ISDN User Part, see Sect. 6.3.3 to 6.3.5)
PABX	Private automatic branch exchange
PSPDN	Packet-switched public data network
CSPDN	Circuit-switched public data network
PSTN	Public switched telephone network

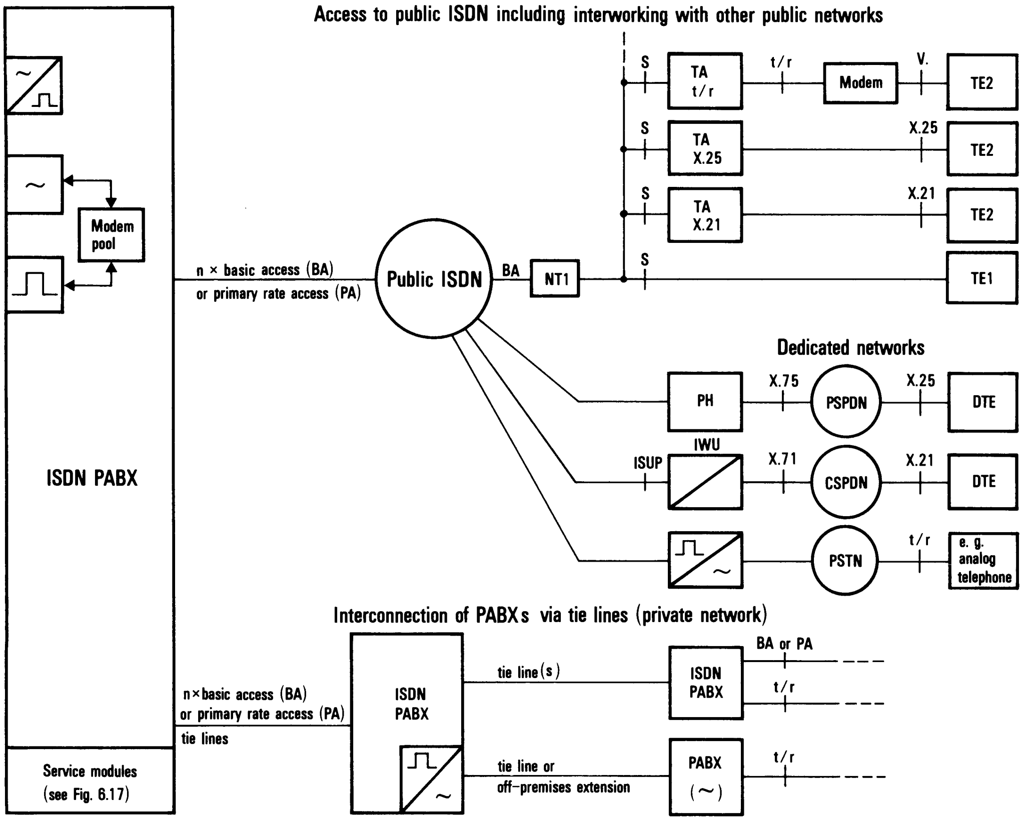
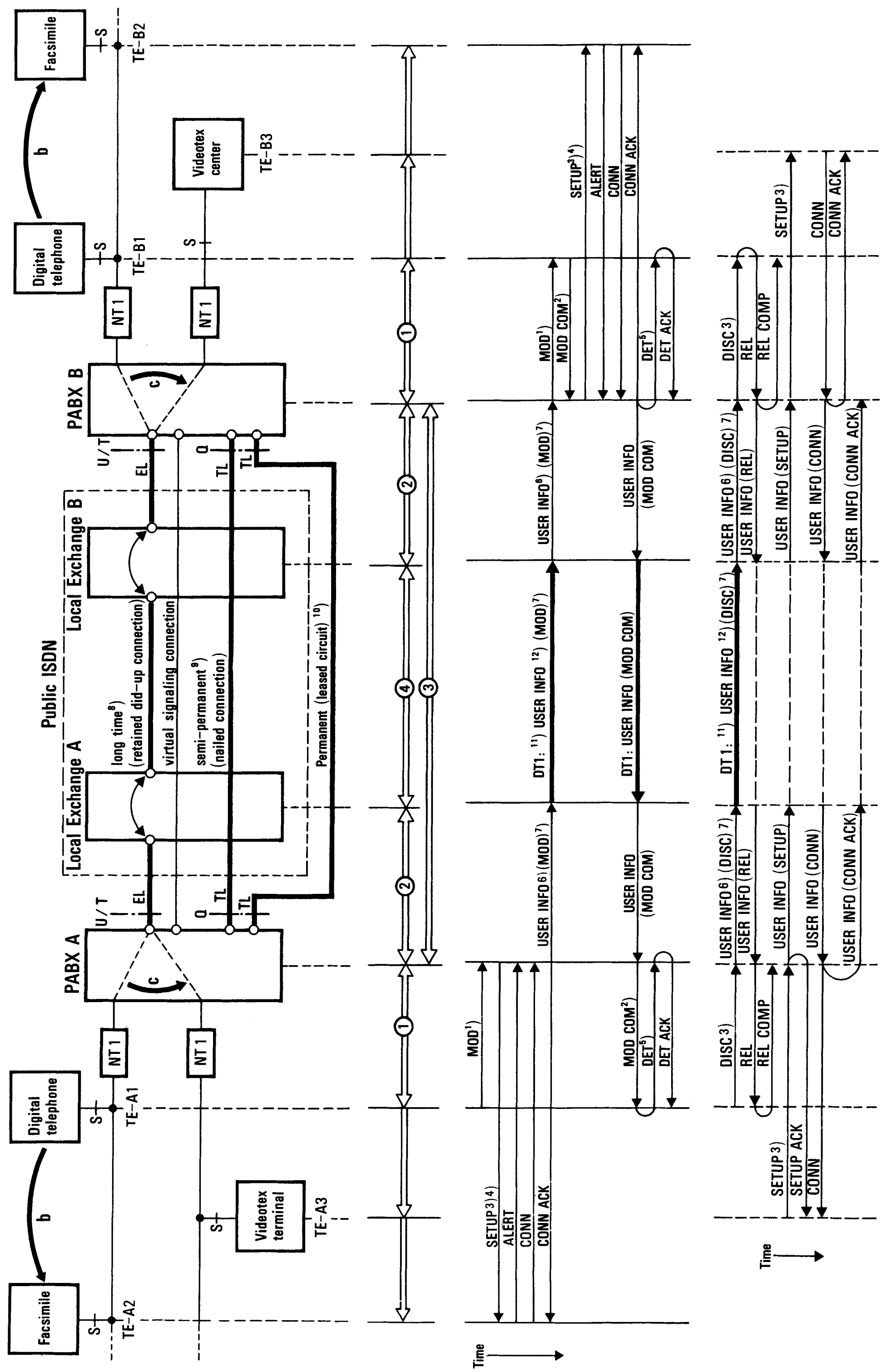


Fig. 6.19b

can be accessed just like private data base systems or data processing systems for local or remote workstations by being connected to the PABX in the same manner as subscribers or other PABXs. They are also connected internally to central control for O&M purposes.

6.4.3 Tie Line Traffic Between ISDN PABXs

Figure 6.20a gives a schematic overview of the networking arrangements and of the interface categories used with PABXs in an ISDN environment. Interface types include interfaces to *user equipment*, interfaces to the *public network* and interfaces to *other PABXs*. As already mentioned in the previous section, ISDN PABXs use, unlike the public ISDN exchanges, the same type of protocol for user access as well as for inter-exchange signaling, i.e. the D-channel protocol defined for public ISDNs (see Sect. 4.3) plus some "upward compatible" enhancements specific to PABX application.


Facsimile
Digital telephone
Videotex terminal
Videotex center
TE-A1
TE-A2
TE-A3
TE-B1
TE-B2
TE-B3
NT1
PABX A
PABX B
Public ISDN
Local Exchange A
Local Exchange B
U/T
EL
Q
TL
long time[8]) (retained did-up connection)
virtual signaling connection
semi-permanent[9]) (nailed connection)
Permanent (leased circuit)[10])
Time
SETUP3)4)
ALERT
CONN
CONN ACK
MOD[1])
MOD COM[2])
DET[5])
DET ACK
USER INFO6)(MOD[7])
DT1: [11]) USER INFO [12]) (MOD[7])
USER INFO[6]) (MOD[7])
USER INFO (MOD COM)
DT1: USER INFO (MOD COM)
DISC3)
REL
REL COMP
SETUP3)
SETUP ACK
USER INFO6)(DISC)7)
USER INFO (REL)
USER INFO (SETUP)
USER INFO (CONN)
USER INFO (CONN ACK)
DT1: [11]) USER INFO [12])(DISC)[7])

a

b

c

Thus, the overall scope of application of the D-channel signaling protocol in the PABX environment includes the following *signaling relations of an ISDN PABX* (see Fig. 6.20a):

(1) Signaling between terminal and ISDN PABX (extension to PABX),
(2) Signaling to set up exchange line (EL) connections (PABX to ISDN local exchange),
(3) Signaling for tie line (TL) connections (PABX to PABX, see reference point Q).

ISDN PABXs, in a basically similar fashion to analog PABXs, make use of *tie lines* TL (for the basic traffic load) and exchange lines EL (essentially for peak traffic load) to

Fig. 6.20a, b. Interfaces and Signaling Interactions for ISDN PABXs. **a** Schematic overview of the ISDN PABX environment; **b** Change of service (in-call modification) for an established 64 kbit/s connection requiring the change of monofunctional terminals, i.e. from digital telephones (TE-A1/TE-B1) to facsimile terminals (TE-A2/TE-B2): the procedure used for extension-to-PABX signaling is in accordance with the respective in-call modification procedure defined for the public ISDN (see Fig. 6.14); **c** Use of the same exchange line (retained dial-up connection) successively for the communication links TE-A1,2/TE-B1,2 *and* TE-A3/TE-B3.

o━━o 64 kbit/s connection
o——o signaling connection
EL Exchange line
TL Tie line
NT1 Network termination
TE Terminal equipment
S, T, U ISDN reference points (see Sect. 4.1.1)
Q Reference point applying to PABX-to-PABX connections (with or without intervention of a public ISDN)
SCCP Signaling Connection Control Part (see Sect. 6.3.4)
ISUP ISDN User Part (see Sect. 6.3.3)
① Extension-to-PABX signaling (based on the user-network signaling procedures defined for the public ISDN with enhancements for use with private networks [6.63])
② User-network signaling (i.e. between an ISDN PABX and a public ISDN)
③ Inter-PABX signaling via the public ISDN (i.e. transparent transport of private ISDN signaling messages (in brackets) without processing in the public ISDN exchanges)
④ ISDN interexchange signaling (see Sect. 6.3)

[1]) D-channel protocol (I.451) message "MODify" requesting in-call modification for an established call (see Sect. 6.3.6)
[2]) I.451 message "MODify COMplete" indicating successful completion of an in-call modification
[3]) I.451 basic call control messages for set-up and clearing of 64 kbit/s ISDN connections (see Sect. 4.3.5 and Fig. 4.23)
[4]) Connection establishment to the "new" terminal (TE-A2, Te-B2)
[5]) Disconnection (release of the B-channel) of the "old" terminal (TE-A1, TE-B1)
[6]) I.451 message "USER INFOrmation" enables the transparent exchange of signaling information between users via the public ISDN (see Sect. 4.3.5.5). Under these circumstances it is used as a transport container for private signaling messages ([7])).
[7]) Private signaling messages between ISDN PABXs in conjunction with tie line connections via virtual signaling connections
[8]) Establishment of the tie line connection via public ISDN exchanges is controlled by the PABX using normal call control procedures, as with regular exchange lines.
[9]) Establishment of the tie line connection via public ISDN exchanges is effected by administrative means in the public ISDN.
[10]) Tie line connection bypassing the public exchanges.
[11]) SCCP message performing end-to-end transport of ISUP signaling messages (see Sect. 6.3.4).
[12]) ISUP message "User-to-user information" (see Sect. 6.3.3).

handle tie line traffic between PABXs (Fig. 6.20a). For tie line traffic between ISDN PABXs via the public network the ISDN related new supplementary services can also be employed on a network-wide basis, i.e. between users connected to different ISDN PABXs without restrictions; this is because in the ISDN signaling information can be exchanged between the ISDN PABXs involved even if there is an active call via a tie line in existence (cf. user-to-user signaling in Sect. 4.3.5.5). This is shown in Fig. 6.20b using the example of *change of service* (*in-call modification*) for an established call from telephony (communication between terminals TE-A1 and TE-B1) to non-voice communication (e.g. facsimile communication between terminals TE-A2 and TE-B2). The signaling sequence employed between the terminals engaged in the change of service and the respective PABX is in accordance with that described in Sect. 6.3.6 for terminals connected to the public ISDN (cf. Fig. 6.14) – since the standard ISDN D-channel protocol can also be used for PABX users.

Similar considerations apply to the network side of ISDN PABXs. Here the ISDN D-channel protocol provided for exchange traffic (PABX to local exchange signaling) can also be used for tie line traffic between ISDN PABXs (PABX to PABX signaling). This is made possible on the one hand by the fact that both signaling messages and – for connection of PABXs only – signaling sequences have been defined symmetrically by the CCITT, i.e. independently of the direction of signaling between calling/releasing or called/released PABX. In addition, the user-to-user signaling already discussed in Sect. 4.3.5.5 prevents the public ISDN from interpreting a signaling message exchanged via the public ISDN as a message in exchange traffic rather than passing it on to the partner PABX: to this end the PABX-to-PABX ISDN signaling messages for transport via the public ISDN are "packed" e.g. in *USER INFOrmation* messages which serve as transparent transport containers.

Tie lines between ISDN PABXs can either be implemented as private leased circuits bypassing the public ISDN exchanges or routed through the public ISDN exchanges as "nailed connections"; nailed connections can be viewed as semipermanent connections which are established by administrative means in the public ISDN (see OAM in Sect. 6.2.7). From the standpoint of the PABX, nailed connections are effectively permanent once they have been established. The virtual signaling connection assigned to the nailed connection can either be established as a *permanent* signaling connection at the same time as the nailed connection, as is assumed in Fig. 6.20, or can be set up or cleared down by the PABX as required (*temporary* signaling connection).

Exchange lines can also be used for tie line traffic between ISDN PABXs to cover peak traffic requirements. In such cases, once an exchange line has been set up using regular call control procedures, it can be used for *a number of* long-distance calls between PABX extensions in succession (Fig. 6.20c), e.g. initially for the connection between TE-A1 and TE-B1 and then for the new connection between TE-A3 and TE-B3. Under such circumstances, the dial-up connection used via the public ISDN is retained for more than one call and is used as a long-time connection between the respective PABXs.

Here the virtual signaling connection assigned to the exchange connection (cf. signaling message *USER INFOrmation*) allows the connection between two PABX extensions (e.g. terminals TE-A1 and TE-A2) to be cleared down using *DISConnect*, without the exchange connection used in the public network being cleared down at the same time.

7 Transmission Methods in the ISDN

7.1 Introductory Remarks

According to its CCITT definition [7.1], the ISDN is based on the digitized telephone network. It can therefore use the same digital transmission systems as the telephone network on the interexchange (trunk) circuits. However, in order to use the subscriber lines for ISDN basic accesses at 144 kbit/s (cf. Sect. 4.2.1.2), new transmission methods must be employed, although existing cables can be retained.

The CCITT and CCIR have defined the basic equipment characteristics (interfaces, bit rates, transmission quality) for transmitting digital signals on cables and other media (see Sect. 7.4), and detailed Recommendations exist for multiplexing equipment (Sect. 7.5).

The CCITT has taken account of ISDN requirements since as early as 1976 when considering transmission quality on digital routes (Sect. 7.7).

7.2 The Hierarchy of Digital Transmission Channels

7.2.1 Basic Building Block: 64 kbit/s

The ISDN is based on the 64 kbit/s channel for the digitized telephone signal. The same applies to the transmission channel hierarchy.

Because of the importance of the 64 kbit/s channel, let us consider its origin in some detail. It derives from the use of pulse code modulation (PCM) in the telephone network. PCM is an analog/digital (A/D) conversion procedure (Fig. 7.1) in which "samples", i.e. discrete instantaneous values, are taken at a rate of 8 000 Hz from the telephone signal, which as in conventional long-distance telephony is limited to a frequency range up to 3 400 Hz. The range of signal values to be transmitted is divided into a certain number of "quantizing intervals" (Fig. 7.1 shows only eight intervals for the sake of clarity). The interval into which each sample falls is ascertained and the number of the interval is transmitted in binary form (3-digit code in Fig. 7.1). The more quantizing intervals are provided, the smaller the "quantizing noise" ($QN = S_r - S_0$ in Fig. 7.1).

When PCM was first introduced (around 1962), the number of quantizing intervals was chosen such that the quantizing distortion is virtually inaudible if in *one* telephone call there are *four* conversions from analog to PCM and vice versa [7.2]. In those days the only conceivable application of PCM was for digital transmission between exchanges using space-division switching. It was found that if a suitable "non-uniform encoding" method [7.3] is used, $128 = 2^7$ quantizing intervals are necessary,

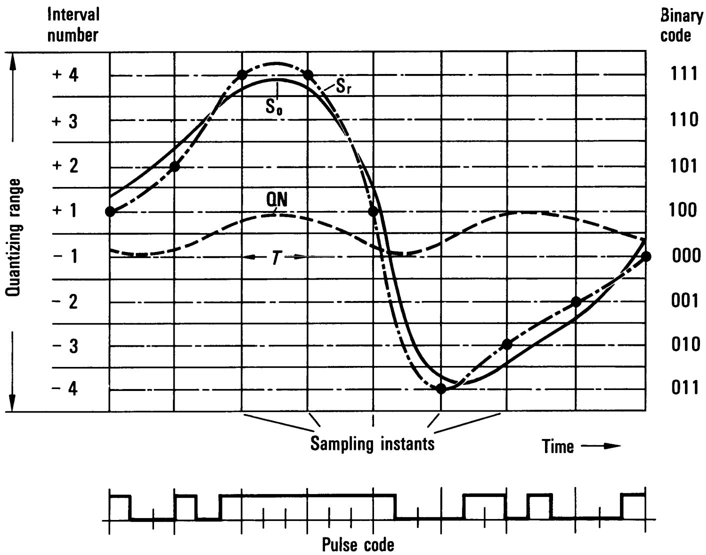


Fig. 7.1. Principle of Pulse Code Modulation (PCM).
S_o original signal, S_r reconstructed signal, QN quantizing noise, T sampling interval (for telephony: 1/8000 Hz = 125 μs). As in CCITT Rec. G.711 the quantizing intervals are numbered from ± 1, while the binary code corresponding to the interval numbers starts from ± 0; the first bit denotes the sign. ● Reconstructed (quantized sample)

i.e. 7 bits must be transmitted for each sample. Later it was realized that in an international connection up to 14 or 15 PCM conversions could be cascaded. The CCITT therefore decided in 1969 to specify 8-bit PCM as the standard; hence the basic unit of 8 bits × 8 000 1/s = 64 kbit/s.

No single internationally accepted standard has been established for 8-bit encoding; instead, two similar encoding laws have evolved: the "A-law" (used in Europe and most non-European countries) and the "μ-law" (North America, Japan) [7.4]. However, it is possible for systems conforming to different laws to interoperate; in principle, every PCM code word of one law is replaced by that word of the other law which yields the best match of the decoded (i.e. analog) sample.

The distribution of the permissible quantizing distortion between the national portions and the international portion of a telephone connection is covered by CCITT Rec. G.113 [7.5; see also 7.2].

In a telephone connection between two subscribers in the ISDN there is only *one* analog/digital and *one* digital/analog conversion. Although there is therefore no necessity in principle to adhere to 8-bit encoding and hence 64 bit/s, this standard can be usefully applied to the ISDN for the following reasons:

- The multiplexed signals introduced (see Sect. 7.2.4) are based on the 64 kbit/s channel.
- Switching networks in digital exchanges (see Sect. 6.2.3) through-connect 64 kbit/s signals.
- Analog/digital and digital/analog PCM converters – also known as codecs (*co*der + *dec*oder) – for PCM at 64 kbit/s are available in the form of LSI devices.

- In future, improved intelligibility and fidelity of reproduction will be desirable for speech transmission. For this purpose a new telephone service providing a voice frequency bandwidth of about 7 kHz may be introduced in the ISDN (see Sect. 2.3.1.2). A subscriber will be able to use the new service instead of PCM telephony if the other party also has a suitable terminal. A/D conversion of a 7-kHz voice signal can be implemented by means of adaptive differential PCM with a sampling frequency of 16 kHz. In differential PCM, the difference between a "predicted" value (estimated value obtained by extrapolation from preceding signal values) and the actual sample is encoded by means of PCM; the "scale of the quantizing intervals" is thus movable within certain limits. Both the predictor used for generating the estimated value, and the scale of quantizing intervals can be matched *adaptively* to the individual characteristics of the signal present (*adaptive* differential PCM, ADPCM); consequently, the signal can be described with fewer bits than in conventional PCM, e.g. with four bits per sample, giving 16×4 kbit/s, i.e. 64 kbit/s for the 7 kHz voice signal – the same as for the conventional telephone signal using PCM.
  The CCITT has elaborated Recommendation G.722 for 7 kHz voice encoding [7.6].
- Terminals for signals other than voice signals (e.g. for facsimile and data transmission) can also use the relatively high bit rate of 64 kbit/s to advantage.

Other bit rates, too, are under consideration, notably 32 kbit/s, again using ADPCM but for conventional telephone signals with a frequency range of 300 to 3 400 Hz. In view of the possibility of a lower bit rate requirement for certain text and data terminals, bit rates of 16 and 8 kbit/s per subscriber channel are also under discussion, and there are proposals – especially in the USA – that these bit rates should also be switched in the digital exchanges ("subrate switching"). However, as far as service integration is concerned, it seems desirable to keep to the uniform bit rate of 64 kbit/s. In the interests of economy it is possible to use a 32-kbit/s method on certain transmission links (e.g. via satellites or transoceanic cables) for voice-frequency signal transmission only. A *non-transparent connection type* (see Sect. 7.7.5) is available for this purpose.

For mobile telephony, an even lower bit rate (16 kbit/s maximum) is desirable since bandwidth is at a premium.

7.2.2 Overview of the Digital Multiplex Hierarchies

The hierarchical structures are given in CCITT Rec. G.702 [7.7]. Fig. 7.2 is adapted from Rec. G.702 and shows the essentials of the two hierarchical systems: the "1.5 Mbit/s hierarchy" which originated in the USA but has a Japanese variant (not shown in the illustration) and the "2 Mbit/s hierarchy" established in Europe and used in most other countries too.

The fixed points in the hierarchies are the *interfaces*, at which equipments operating at the same bit rate can basically be interconnected as required. The pulse shapes and line codes of the interface signals, etc. are defined in CCITT Rec. G.703 [7.8]. The interface circuits are frequently routed to *digital distribution frames* [7.3] providing flexible interconnection.

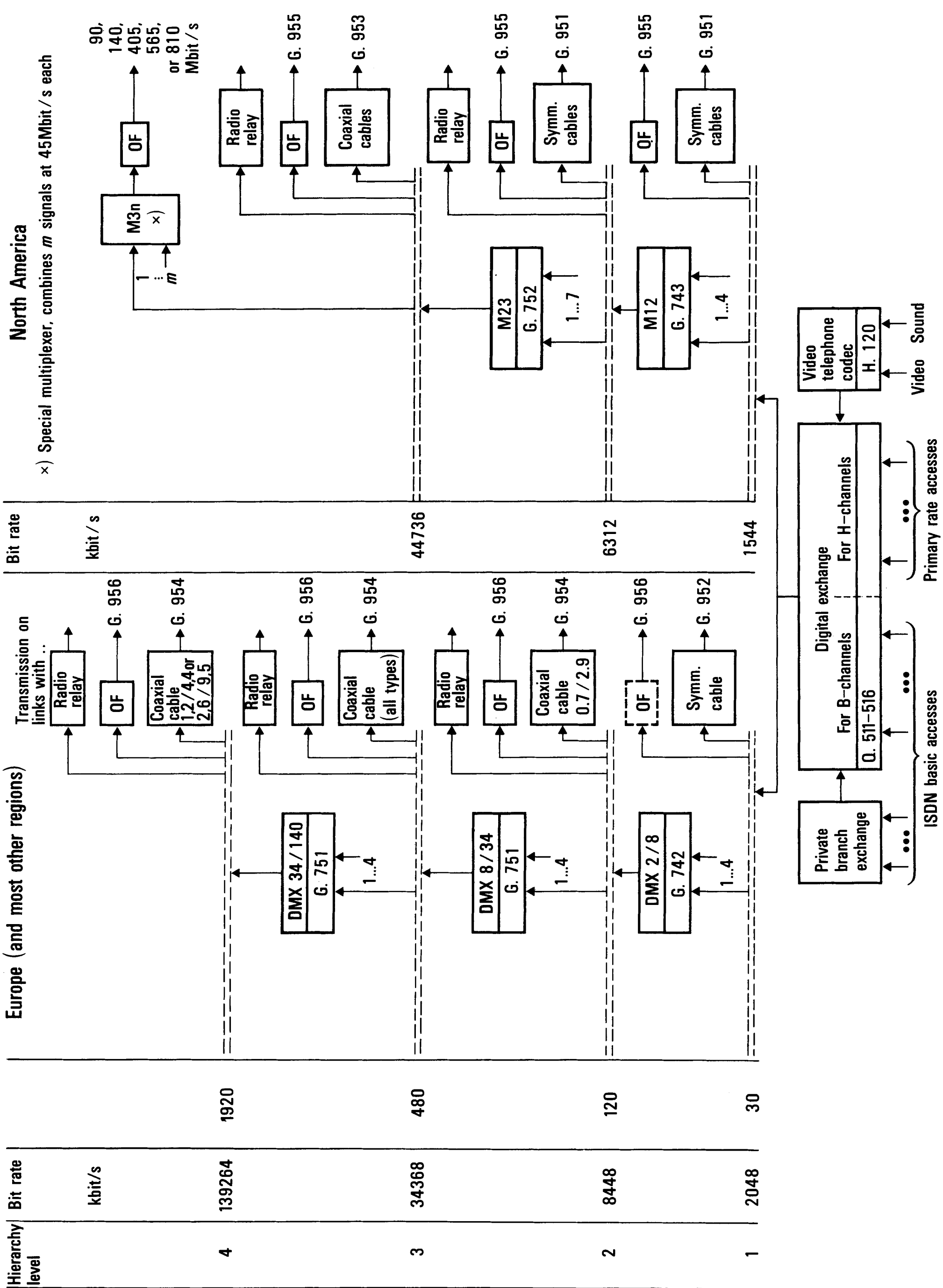


Fig. 7.2. Two Digital Hierarchies, Based on 2048 and 1544 kbit/s Respectively, with Transmission Equipment Applicable to the ISDN.
DMX or M ... digital multiplex equipment, OF optical fiber, G ..., H ..., Q ... designation of the relevant CCITT Recommendation (Red Book); B, D see Sect. 4.2.1.1, = = = interface level (frequently also digital distribution point): interfaces as per CCITT Rec. G.703

7.2.3 Primary Multiplex Signals

A de facto standard (from AT & T) for a PCM multiplex system first came into being in the USA, where in 1962 an equipment was put into service which combined 24 digitized telephone signals each encoded using seven bits; to each 7-bit code word was added an eighth bit for signaling, and for every $24 \times (7+1)$ bits a further bit for frame aligment was provided, giving a total bit rate of $(24 \times 8 + 1)$ bits $\times$ 8 000 1/s = 1 544 kbit/s. This frame structure (as shown in Fig. 7.3a) was retained when 8-bit encoding was introduced (see Sect. 7.2.1) [7.9]. Most existing American 1 544 kbit/s line systems ("T1 systems") do not allow long sequences of zeros to be transmitted; for the time being, therefore, each octet must contain at least *one* 1-bit. In voice transmission this is easily achieved by avoiding the use of the 8×0 octet, but for non-voice signals it is necessary for the time being to fix the eighth bit (the bit transmitted last) at binary "1". The long-term plan is to use the B8ZS line code (cf. Sect. 7.4.2).

Within CEPT, the Europeans agreed in 1969 on a system for combining 30 encoded telephone signals. Signaling was to be completely separate from voice transmission from the outset, and ample provision was to be made for pulse frame alignment, alarm messages, etc. [7.10].

The pulse frame is shown in Fig. 7.3b. Time slots 1 to 15 and 17 to 31 are initially intended for PCM-encoded telephone signals; in the ISDN they can be used for any other 64 kbit/s signals originating from the subscriber and transmitted in B-channels (see Sect. 4.2). Time slot 16 is currently used for channel-associated signaling; in the

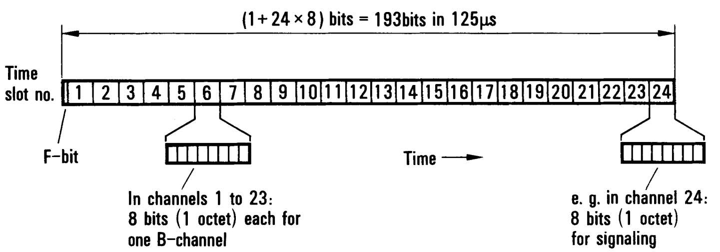


Fig. 7.3a. Pulse Frame of the 1544 kbit/s Signal as Used in the ISDN.
Within a multiframe comprising 24 frames, the F-bit is used for indication of frames and multiframes, for alarm messages and for CRC-6 check bits. These check bits serve to prevent erroneous frame alignment and to measure the bit error ratio

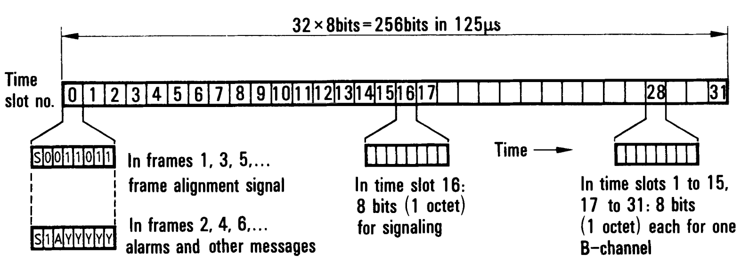


Fig. 7.3b. Pulse Frame of the 2048 kbit/s Multiplex Signal (in the ISDN).
S bit used for multiframing and CRC-4 check, A bit for remote prompt alarm, Y spare

ISDN's primary rate access channel structure (see Sect. 4.2.1.2) e.g. between a large private branch exchange and a local exchange in the public network (see Sect. 6.4), it carries a 64 kbit/s D-channel signal (see Sect. 4.2.3), and between public exchanges the signal of CCITT Common Channel Signaling System No. 7 (see Sect. 6.3).

7.2.4 Higher Bit-Rate Signals

Digital transmission is no different from traditional FDM in that, where large volumes of traffic have to be transmitted, sufficiently large channel blocks must be provided. For this purpose systems of "hierarchical" multiplex levels were developed. In the system which evolved in Europe, *four* lower-level signals are combined at each stage (Fig. 7.2). The US and Japanese systems use multiplexing factors of 3, 4, 5, and 7. The combining of several "tributaries" to obtain a signal at a higher bit rate is effected by *digital multiplexers*, considered in Sect. 7.5.2.

Actual transmission via cable or radio links is very often accomplished at the "hierarchical" bit rates (see Table 7.2). It is also possible first to combine several hierarchical digital signals by additional multiplexing before transmitting them to line. Examples of this are radio relay transmission at $2 \times 8\,448$ kbit/s, transmission on coaxial lines or optical fibers at 565 Mbit/s, and in North America, transmission over cables at 3 152 kbit/s, 90 Mbit/s, or 140 Mbit/s (the latter bit rate being incidentally the same as one of the hierarchical bit rates in Europe). In general there are no internationally standardized equipment interfaces for the resulting non-hierarchical bit rates.

The signals at higher levels of the "digital hierarchy" contain either multiplexed telephone signals, ISDN B-channel signals (cf. Fig. 7.2) or, in future, also broadband signals requiring more than 64 kbit/s from source (Sect. 2.5).

In addition to the existing hierarchical levels, new multiplex structures are evolving, in particular for transmission on optical cables. In the U.S., the key word is SONET (*s*ynchronous *o*ptical *net*work). The new multiplex signals will accommodate the signals of the existing hierarchies. The CCITT has achieved basic agreements on a worldwide standard for a "network node interface" NNI at 155.52 Mbit/s for this purpose.

7.3 Transmission Media

Basically the same transmission media are used in the ISDN as in conventional networks, although conventional copper lines are no longer adequate as subscriber lines for broadband services; optical fibers must be used instead.

7.3.1 Conductors in Cables

Table 7.1 gives an overview of the different media for transmitting digital signals in cables. Some of the basic cable characteristics are briefly described below.

- Symmetrical pairs, i.e. twisted *copper wire pairs* with paper or polyethylene insulation are combined to form cables containing about 20 to 2 000 pairs. These

Table 7.1. Media for Transmitting Digital Signals in Cables

Medium		Characteristic dimensions		CCITT Rec.	Applicability	
					Frequency MHz	Wavelength nm
Copper wire	Symmetrical pair (originally for voice frequency)	Wire diameter	0.4 to 1.2 mm	G.613	Up to 2	
	Symmetrical pair in LOCAP cable	Wire diameter	0.64	G.612	About 5	
	"Mini" coaxial tube	Outer diameter of inner conductor/inner diameter of outer conductor	0.7/2.9 mm	G.621	0.2 to 20	
	"Small" coaxial tube		1.2/4.4 mm	G.622	0.06 to 70	
	"Large" coaxial tube		2.6/9.5 mm	G.623	0.06 to 300	
Optical fiber	Graded-index (multimode)	Core diameter/cladding diameter	50/125 μm	G.651		820 to 900 1270 to 1330
	Single-mode fiber	Mode field diameter[a]/cladding diameter	10/125 μm	G.652		1300 1550

[a] Contains major portion of optical power, is about 10% larger than core diameter, see Fig. 7.5b

are used for transmitting analog voice-frequency signals over subscriber lines and in the local and short-haul network. Since PCM was first introduced, existing cables of this type have also been used for transmitting digital signals. The objective here is to ensure better utilization of cable capacity (for example in the 30-channel PCM system – see Sect. 7.2.3 – instead of 30 wire pairs only two are needed for 30 voice circuits), thus obviating the need to lay new cables to keep pace with increasing traffic volumes. The higher the bit rate of a digital signal, the greater the transmission loss on a wire pair (cf. Fig. 7.8). Crosstalk, i.e. coupling between different wire pairs within a cable also becomes increasingly troublesome. At bit rates of 2 048 or 1 544 kbit/s, in order to provide reliable separation of the wanted signal from interference caused e.g. by crosstalk, it is necessary to install *regenerative repeaters* at intervals of approximately 1.7 to 3.5 km [7.11]. For subscriber lines with a net bit rate of 144 kbit/s, regenerators are only required after 8 km if *echo cancelation* is employed (Sect. 7.4.3).

- *Coaxial cables* consist of pairs whose dimensions are characterized by the external diameter of the inner conductor and the internal diameter of the outer conductor. Coaxial pairs with dimensions of 2.6/9.5 mm (*large tube*) and 1.2/4.4 mm (*small tube*) are used extensively for transmitting carrier frequency signals in the long-haul network. They are also suitable for digital transmission. Often spare capacity is available for this purpose in existing cables. Alternatively, cables may be converted from carrier frequency to digital transmission (retaining the same repeater spacing). It is not likely that new coaxial cables of the two types mentioned will in future be laid specially for digital transmission.
  For economic reasons the 0.7/2.9 mm *mini coaxial tube* has been developed specifically for digital transmission. It is used in some countries for transmitting 2 and 8 Mbit/s signals.
- *Optical fibers*, although a recent development (under discussion since the fifties, but not technically feasible until the seventies), provide nevertheless the most promising medium for digital transmission and hence for future communication

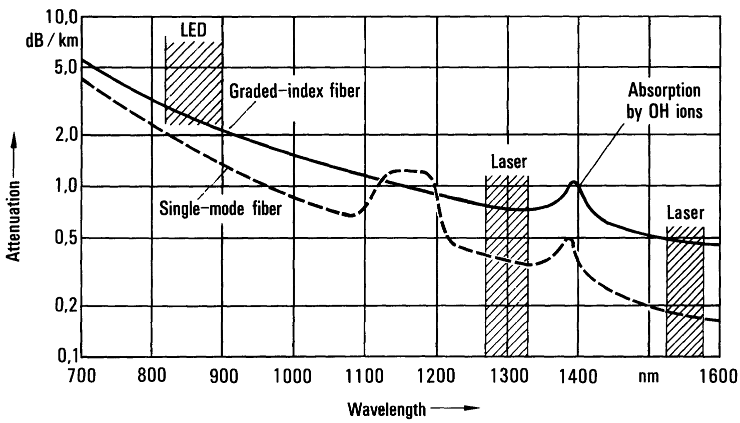


Fig. 7.4. Attenuation (per km) of Typical Optical Fibers as a Function of Wavelength. Hatched areas: wavelength regions of conventional optoelectronic transducers (LEDs and laser diodes)

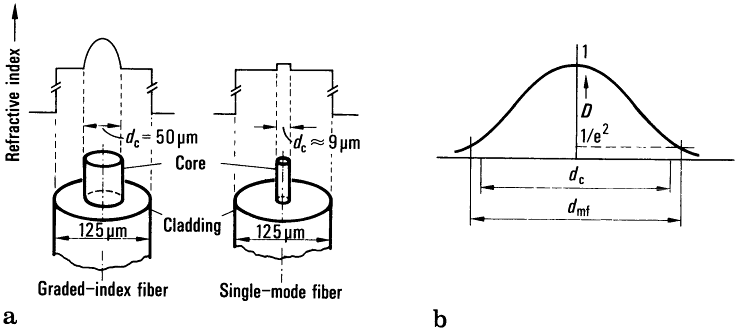


Fig. 7.5. a The Two Types of Optical Fiber. With the single-mode fiber, the mode field diameter rather than the core diameter is specified – see **b**.
b Definition of the Mode Field Diameter d_{mf}. This diameter is about 10% greater than the core diameter d_c. D relative optical power density

systems in general. Optical fibers will certainly predominate in the cables of future transmission systems – at any rate in newly installed cables; this applies to both long-haul and local traffic.

An optical fiber is a silica thread through which light rays are transmitted in the infrared region. Attenuation is a function of the purity of the material. There is however an unavoidable minimum attenuation due to absorption and scattering of the light. Attenuation decreases with increasing wavelength (see Fig. 7.4).

In practice two types of optical fiber are used (Fig. 7.5):

- The *graded-index fiber*: due to its relatively large core diameter (50 µm) it is capable of accepting light from light-emitting diodes (LEDs); however, it is essentially limited to transmitting signals up to 140 Mbit/s.
- The *single-mode fiber*: due to its small core diameter (9 µm) in practice only laser diodes or possibly edge-emitting diodes can be used as light sources.

Details of the technology and application techniques of optical fibers and fiber-optic cables are dealt with more fully in [7.12 and 7.13].

7.3.2 Radio Relay

The transmission medium used by radio relay is free space. According to the laws of radiation, attenuation occurs because there is a limit to the narrowness of any given radio signal beam. The attenuation coefficient (in decibels) does not increase, as in cables, proportionally with the distance but only with its logarithm, so that substantially greater distances can be spanned without amplification than using copper cables. In practice, however, additional attenuation occurs in free space due partly to absorption in the atmosphere and above all to scattering at raindrops. In practice, rain attenuation only becomes a problem at frequencies above about 10 GHz [7.14]. It then necessitates the use of shorter hop lengths (i.e. shorter distances between the relay stations), and for this reason the higher frequencies are in practice only used for short-haul systems (see Sect. 7.4.4).

On terrestrial radio links, interference due to multipath propagation can be very troublesome. This is caused by reflections from level ground, water surfaces and abrupt transitions between tropospheric layers [7.14].

The transmission capacity of radio relay is limited. In the case of terrestrial radio relay this is because the frequency range is limited due to absorption (a few hundred MHz to about 23 GHz) and because the same frequencies can only be used at different locations if the latter are sufficiently far apart.

For radio relay via satellites, the same basic factors apply as to terrestrial radio relay, including attenuation due to rain. In its favour is the fact that radio beams directed at satellites cross the rain layer at a relatively steep angle.

Frequencies up to about 30 GHz are used in satellite systems. However, there is a limit to the number of geostationary satellites which can be positioned in the equatorial plane (at an altitude of some 36 000 km); the present minimum spacing is 3 °, in future it will be 2 °. And so, once again, the total transmission capacity is limited.

A disadvantage of satellite links is the long transmission delay (about 260 ms for a "satellite hop" i.e. ground-satellite-ground). This makes conversation in telephony more difficult (the more so if two satellite links are connected in series) and can provide problems in interactive data communication (see Sect. 3.8.3 and 7.73) [7.15].

Radio relay (including satellite systems) enables communication links to be set up quickly; it is particularly useful if cable transmission systems are not yet available or are ruled out for geographical reasons.

7.4 Equipment for Transmitting Digital Signals on Cable and Radio Links

7.4.1 General

For the user, transmission methods and systems should always meet the same quality requirements irrespective of the transmission medium. This applies especially in the ISDN where every transmission system must be capable of transmitting signals for different services. Network operators, too, are interested in uniform transmission quality, in order to provide flexible interconnection and protection switching of transmission systems. For these reasons, the CCITT has created the notion of the *digital section.*

A digital section is an element in an overall digital link: Fig. 7.6 shows a hypothetical link and its digital sections. According to CCITT definitions [7.3] a digital section includes the whole of the means of digital transmission of a digital signal

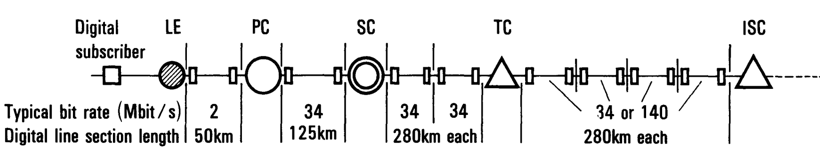


Fig. 7.6. Hypothetical Reference Connection from Subscriber to International Switching Center. LE local exchange, PC primary center, SC secondary center, TC tertiary center, ISC international switching center

of specified bit rate between two consecutive digital distribution frames or equivalent (generally using standardized interfaces – see Sect. 7.2.2). The following basic characteristics have been defined for a digital section:

- Length of the hypothetical digital section to which the performance characteristics relate: for the 2 Mbit/s hierarchy, "hypothetical reference digital sections" have been defined with lengths of 50 km (in practice mainly for 2 Mbit/s and 8 Mbit/s) or 280 km (primarily for higher bit rates).
- Bit sequence independence [7.3], i.e. the capability of the transmission system to transmit any bit sequence, including e.g. all-zeros. This requirement is imposed in the ISDN specifically in the interest of unrestricted transmission of text and data signals. Most 1 544 kbit/s line systems do not meet these requirements at present (see Sects. 7.2.3 and 7.4.2).
- Bit error performance: see Sect. 7.7.1; for example, for a 140 Mbit/s digital section 280 km long, a maximum percentage (0.045 %) of one-minute intervals with a bit error ratio exceeding 10^{-6} is specified.
- Interfaces to other equipment (including the adjacent digital section, if there is any): see Sect. 7.2.2.
- Maximum jitter at input and output. (*Jitter* denotes unintentional but unavoidable phase variations [7.3]; see Sect. 7.7.4).
- Alarm conditions. Alarms must be initiated in the line terminating equipments for multiplex bit rates (from 2 or 1.5 Mbit/s onwards)
  - if the incoming is signal is lost: prompt alarm;
  - if the bit error ratio of 10^{-3} in the signal coming from the transmission link is exceeded: prompt alarm (provided that bit error ratio is monitored).

  In the case of a prompt alarm, the signal traveling onward (downstream) is replaced by the *Alarm Indication Signal* AIS. With most bit rates, this is an all-ones signal. Only at 44 736 kbit/s does the AIS contain framing bits and certain other overhead bits, and the information bits are replaced by a 1010 ... sequence. In the USA the AIS was formerly called a "Blue Signal".

For equipment employed in the 2 Mbit/s hierarchy, the internationally agreed characteristics of the digital sections are detailed in CCITT Rec. G.921 [7.16], which applies to both line systems and radio relay systems.

7.4.2 Transmission on Cables in Trunk Circuits

Table 7.2 gives an overview of typical systems for digital transmission on cables. The table makes no claim to completeness.

As mentioned in Sect. 7.3.1, transmission on copper wires [7.11] is mainly realized where cables are already installed and are either used for analog transmission or are still spare.

The future belongs to optical fibers; this is clear from the media overview in Table 7.1. Optical fibers have the following advantages over copper:

- Low line attenuation, hence widely spaced repeaters; power-fed intermediate repeaters are seldom used, and in the local network virtually not at all (most

Table 7.2. Typical Line Systems for Transmitting Digital Multiplex Signals on Cables. See Sect. 7.4.2 for line codes

Medium			Line code and repeater spacing in systems with							
			1.5 Mbit/s	2 Mbit/s	6 Mbit/s	8 Mbit/s	34 Mbit/s	45 Mbit/s	140 Mbit/s	565 Mbit/s
Copper wire	Symmetr. wire pair (originally for voice frequency)		AMI or B8ZS 1.8 km	HDB3 1.7 to 3.5 km	–	–	–	–	–	–
	Wire pair in LOCAP cable		–	–	B6ZS 4.8 km	–	–	–	–	–
	"Mini" coaxial tube		–	–	–	HBD3 4 km	4B/3T 2 km	–	–	–
	"Small" coaxial tube		–	–	–	–	4B/3T 4 km	–	4B/3T 2 km	–
	"Large" coaxial tube		–	–	–	–	4B/3T 9.3 km	–	4B/3T 4.65 km	AMI 1.55 km
Optical fiber	Graded index fiber with	LED	–	Binary* a) 8 to 12 km b) 20 km	–	Binary* a) 9 to 11 km b) 15 km	5B/6B a) 5.5 km b) 12 to 21 km	Binary** a) 6 km b) 10 km	5B/6B a) 5 km b) 7 km	–
		Laser diode	–	Binary* a) 12 to 16 km b) 30 to 40 km	–	5B/6B a) 10 to 15 km b) 25 to 35 km	5B/6B a) 10 to 13 km b) 29 to 39 km	–	5B/6B b) 20 to 30 km	–
	Single-mode fiber with laser diode	1300 nm 1550 nm	–	–	–	–	5B/6B 30 to 55 km –		5B/6B 30 to 50 km 40 to 70 km	5B/6B 25 to 40 km 30 to 60 km

* e.g. 1B/2B; ** Scrambled
a) Optical wavelength $\lambda = 820$ to 900 nm; b) $\lambda =$ approx. 1300 nm

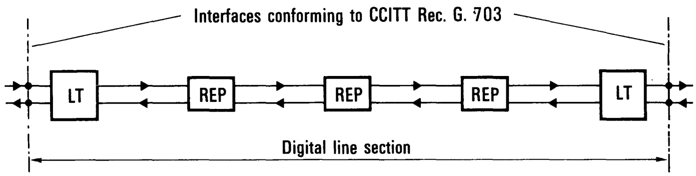


Fig. 7.7. Basic Configuration of a Line System for Digital Transmission on a Cable. LT line terminating equipment, REP repeater

intermediate repeaters can be installed in buildings). For high bit rate systems in the long-haul network – including submarine cables – the single-mode fiber is preferred due to its low attenuation (see Fig. 7.4) and large bandwidth.

- Very high transmission capacity (in theory more than 10 Gbit/s)
- Small size, low weight and high flexibility of cable.
- No electrical conductivity, hence no need for protection against electro-magnetic interference, lightning, etc.

Figure 7.7 shows the basic structure of a line system comprising line terminals, cable wires/fibers and repeaters; in fiber-optic systems, repeaters are frequently not required. The arrangement shown in Fig. 7.7 is the physical implementation of a *digital line section*, i.e. a digital section on cable (see Sect. 7.4.1).

Line terminals have the following functions: converting the digital signal from the standardized interface code to the line code (if necessary) and vice versa, monitoring the bit error ratio, alarm indication and power feeding (if necessary).

Some of the possible line codes are given in Table 7.2:

- AMI (alternate mark inversion) is a *pseudo-ternary* code [7.3], i.e. a code in which binary 1 bits are alternately represented by positive and negative pulses, and 0 bits are represented by the voltage 0. With long strings of zeros, the timing content is insufficient (cf. Sect. 7.2.3).
- B6ZS and B8ZS (bipolar with 6- and 8-zero substitution respectively) are *modified AMI codes* [7.3]: to provide sufficient timing, groups of six (or eight) consecutive zeros are replaced by groups which have additional pulses, recognizable by the fact that some of these pulses violate the alternation rule of AMI.
- HDB3 (high density bipolar of order 3) is another modified AMI code, replacing groups of four zeros by groups containing AMI violations. HDB3 is also used for signals with bit rates of 2, 8 and 34 Mbit/s at hierarchical interfaces as discussed in Sect. 7.2.2.
- 4B/3T is a *redundant ternary* code [7.3]: four consecutive bits of the original signal are replaced by three ternary signal elements, thereby reducing the line digit rate to 3/4, which results in reduced line attenuation. There are several variants of 4B/3T codes, such as the MMS43 code (Modified Monitoring State [7.17]). They are used in transmission systems for 34 and 140 Mbit/s see (Table 7.2), and also in some transmission systems for subscriber lines (see Sect. 7.4.3).
- 5B/6B and 7B/8B are redundant binary codes for optical fibers: to every five or seven information bits a sixth or eighth bit is added for monitoring and to provide bit sequence independence.

7.4.3 Transmission on Subscriber Lines

General

The copper wire pairs of the subscriber lines represent a substantial proportion of the overall capital expenditure on the telephone network. It is intended to continue using them in the ISDN so that no new capital investment in subscriber lines is necessary. As with the balanced pairs of the paper- or polyethylene-insulated trunk curcuits, it must be remembered that subscriber lines were originally intended for transmitting voice frequency signals (cf. Sect. 7.3.1).

Figure 7.8 shows the attenuation-frequency response of typical wire pairs (with 0.4 and 0.6 mm diameter). For instance, the Deutsche Bundespost uses 0.4 mm wires for subscriber lines up to 4.2 km long; where greater lengths are involved, practically the same attenuation can be achieved in the voice frequency range for line lengths between 4.2 and 8 km by the appropriate use of 0.6 mm wires for part or all of the line. However, the attenuation increases somewhat at higher frequencies (e.g. by about 3 % at 60 kHz between 4.2 and 8 km).

Other administrations or operating companies have similar planning guidelines for subscriber lines.

Figure 7.9 shows the frequency distributions of subscriber line lengths in the Deutsche Bundespost area. This distribution can be considered as typical for many countries. It shows that 99 % of the subscriber lines are no more than 8 km long, i.e. they can be implemented using 0.4 and 0.6 mm wires with no additional measures being necessary. For digital transmission on subscriber lines the range should be just as long.

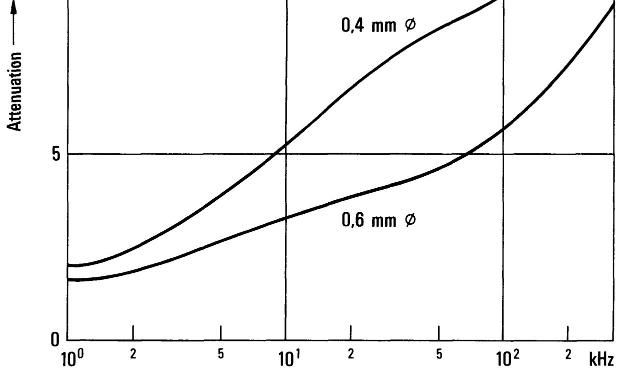


Fig. 7.8. Attenuation Response of Typical Wire Pairs in Local Cables with Polyethylene Insulation

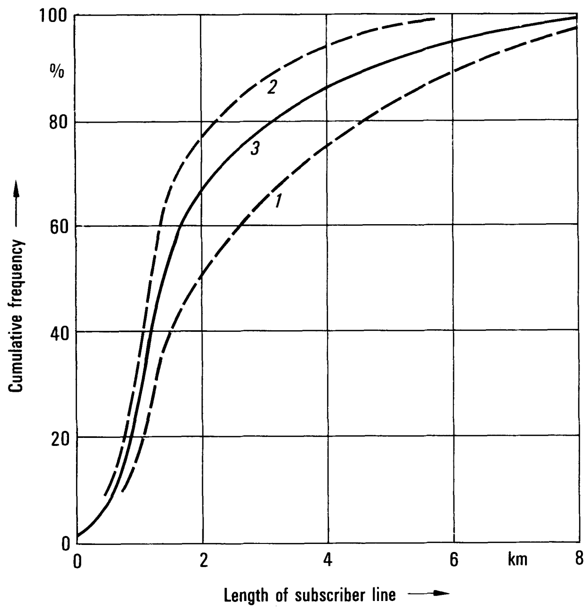


Fig. 7.9. Typical Frequency Distribution of Subscriber Line Lengths (refers to Deutsche Bundespost area).
1 Small local networks (<800 main stations)
2 Large local networks (>10000 main stations)
3 Average of all local networks

For lengths greater than 8 km a higher supply voltage is required in the case of voice frequency use; for digital transmission a regenerative repeater may be necessary, or remote multiplexers can be used (cf. Sect. 7.5.2).

In some countries, very long voice-frequency subscriber lines are fitted with loading coils. The loading coils must be removed for digitization, because loading causes a marked increase in attenuation above the VF range (low-pass characteristic).

Where there are no existing subscriber lines, laying optical fiber cables can be considered. These are also suitable for "broadband" access (above 2 048 kbit/s). In certain rural areas radio relay links may be an option. However, only transmission over copper wire pairs is considered in detail below.

Transmission Method for the Basic Access

So far the CCITT has not standardized the transmission method. It has often been stated that the existing conditions of subscriber line plant in various countries are very different. This is now doubted. It has also been argued that the progress of technology should not be hampered by premature standardization. However, several countries, most notably the USA, aim at national standards, and these may eventually lead to international standards.

For the ISDN basic access (see Sect. 4.2.2) the only certainty at present is that two B-channels and one D-channel are to be used for transmitting information, together with signals for frame alignment and possibly multiframe alignment as well as maintenance information, giving an actual total bit rate of about $(2 \times 64 + 2 \times 16)$ kbit/s $= 160$ kbit/s.

As only one wire pair is available to a subscriber, the digital signals – just like the voice frequency signals in conventional telephony – must be transmitted over this pair in both directions (see Fig. 7.10).

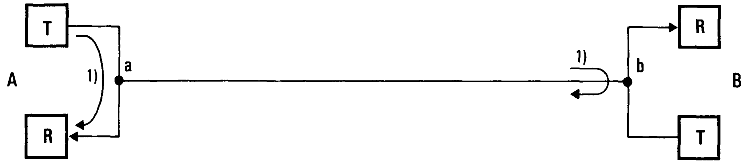


Fig. 7.10. Two-Wire Duplex Transmission: the signals for both transmission directions are carried on the same wire pair.
T transmitter, R receiver, 1) interfering reflections of the signal transmitted in A

The problem here is that the receiver picks up not only the wanted signal from the distant end, but also, due to unavoidable reflections e.g. at the hybrid junctions a and b (Fig. 7.10), the signal which has been transmitted at *the same* end but which causes interference in the receiver. This reflected signal (somewhat imprecisely termed "echo") must be neutralized. In practice there are two possible options:

- The first method studied and implemented was the *time division method* (also known as the burst or pingpong method) whereby information blocks are formed, each containing e.g. two octets from the two B-channels and four bits from the D-channel; these blocks are transmitted *alternately* in both directions as shown in Fig. 7.11. Between the end of transmission of a data block, e.g. at location A, and the start of transmission at the other end of the line (e.g. at location B) there must be an interval T_i in which disturbing reflections can decay. The length of that interval must be somewhat greater than the signal delay T_s. The greater the length of the subscriber line and hence of the signal delay T_s, the less time is available for transmitting the data block, and hence the higher the transmission rate required during actual transmission. Fig. 7.12 shows transmission rate versus line length in the case of four B-octets, etc. being transmitted per block (period $T = 250\,\mu s$ as in Fig. 7.11). It shows that a transmission rate of 550 kbit/s is required for a range of 8 km. Using this high bit rate poses problems due to the high line attenuation, crosstalk between adjacent wire pairs and radio interference. The method can, however, be used for short subscriber lines, e.g. in private branch exchange networks.
- *Echo cancelation* avoids the above-mentioned problems associated with the time division method. It will become the standard in many networks including those of the USA and the Federal Republic of Germany. With this method the transmission rate is independent of the range: transmission takes place continuously. Reflections at hybrid junctions a and b (Fig. 7.10) or at discontinuities on the route cause interference directly. In order to neutralize the reflected signal it must be very precisely simulated and subtracted from the received signal, so that only the wanted signal is left. For example, if a line attenuation of 35 dB is to be bridged (this corresponds to 4.2 km with 0.4 mm wire diameter at 60 kHz, see below), the reflection of the "own" signal must be attenuated by about 55 dB by canceling. The reflected signal must therefore be precisely simulated to within approximately 2 %. This is possible using an *echo canceler* providing automatic, adaptive adjustment to the characteristics of the line [7.18]. A canceler of this type can only be implemented cost-effectively as a digital LSI circuit.

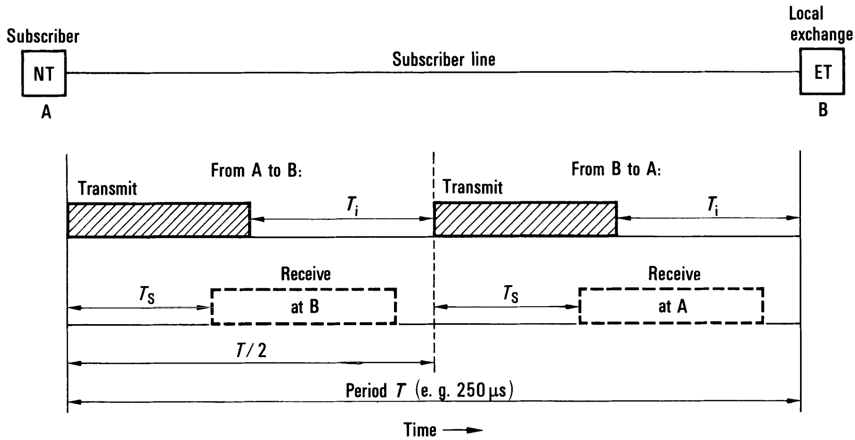


Fig. 7.11. Principle of the Time Division Method.
ET exchange termination, NT network termination, T_i interval, T_s signal delay, LE local exchange

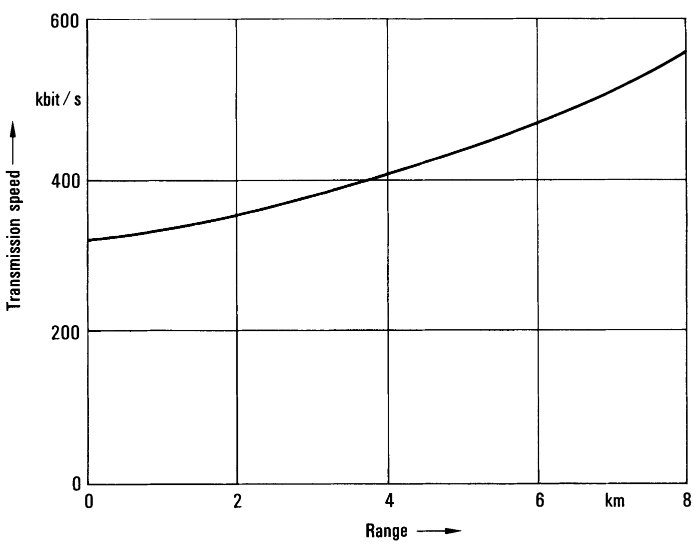


Fig. 7.12. Transmission Rate Required for the Time Division Method as a Function of the Range (applies to $T = 250$ µs)

Figure 7.13 shows the basic setup of a transmission equipment incorporating echo cancelation. The transmitter feeds the signal to the transmission line on the one hand, and to the echo canceler on the other. The latter is designed as a transversal filter [7.19] whose coefficients adjust adaptively such that a replica ER of the echo signal is produced at the output of this filter. This replica is subtracted from the received signal. This function is similar in principle to that of the echo cancelers used in the analog telephone network to neutralize the speaker echo in intercontinental traffic or on satellite links [7.20].

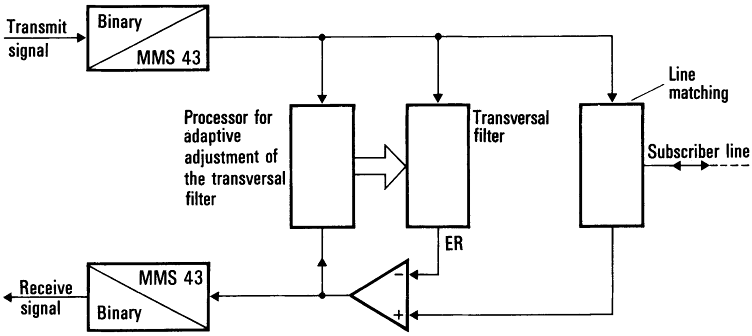


Fig. 7.13. Basic Configuration for Transmitting a 160 kbit/s Signal on Subscriber Lines (echo canceling method). ER output signal from transversal filter (representing echo replica), MMS 43 line code (see Sect. 7.4.2)

Implementation of the digital echo canceler is facilitated by selecting a suitable line code. Several codes are under discussion (and in use). Among these are 4B/3T codes and specifically MMS43 (cf. Sect. 7.4.2). These "redundant ternary codes" provide a relatively small low-frequency portion of the line signal; in addition, the line digit rate is reduced to 3/4 of the 160 kbit/s transmission rate, i.e. to 120 kbauds. The signal power is then concentrated around 60 kHz. The line attenuation is thus reduced and the crosstalk attenuation increased.

In the USA it is intended to use a 2B/1Q code in which two bits of the original signal are replaced by one quaternary (i.e. four-level) signal element. This reduces the line digit rate to 1/2 of the effective bit rate.

As long as copper wires are used for the subscriber lines, it is advisable for the network termination NT1 (or possible NT12, cf. Sect. 4.2.2.6) and perhaps also the terminal equipment (especially a telephone) to be powered from the exchange at least for emergency operation. Power-feeding of the NT1 is mainly advantageous for fault diagnosis using test loops under control of the exchange.

In the idle condition, i.e. while no communication is taking place, the network termination is not fully active, and so the full supply current is not continuously required. However, the network termination NT1 is permanently ready to receive an "activation signal" – a specifically defined pulse train (cf. Sect. 4.2.2.5). Once it is received, NT1 is activated, i.e. all the functional components are switched on.

Multiplexed Signals

Subscribers with *multiple channel access* – mainly large digital private branch exchanges – are normally connected to the exchange via 1 544 or 2 048 kbit/s transmission systems (Table 7.2). The associated pulse frames have already been described in Figs. 7.3a and 7.3b in Sect. 7.2.3. Time slot 24 (or 16) of the multiplexed signal contains a 64 kbit/s D-channel signal.

The transmission method is then essentially the same as on the interexchange trunks (Sect. 7.4.2); regenerators must be used on the copper wire pairs of existing subscriber lines if, for example, in the case of 0.4 mm wires the line length exceeds about 1 km.

In many cases it is advisable to combine the signals of remote subscribers (especially those whose line to the local exchange would be longer than 8 km) using a remote multiplexer. For instance, the signals from 12 basic accesses can be combined to form a 2 048 kbit/s signal, the associated D-channel signals being accommodated separately in specific time slots (which requires 192 kbit/s).

7.4.4 Radio Relay Transmission

Whereas analog radio relay systems have hitherto been used mainly in the toll network, digital radio systems are being deployed increasingly in the *short-haul network* and, in future, in the *local network* also. The present rule of thumb is that for 1.5 to 8 Mbit/s operation a radio relay link is more cost-effective than a cable link if the cable length exceeds about 10 km. The systems used in the short-haul and local network mainly employ radio frequencies of 13 000 to 23 000 MHz [7.21]. This limits hop lengths to 15 to 25 km, at least with the rainfall rate prevalent in Central Europe or regions with a similar climate. This range is adequate for most hops in the short-haul or local networks.

As in cable systems, higher bit rates (namely 34, 140 and $n \times 45$ Mbit/s – cf. Fig. 7.2) are generally used for transmission in the *toll network* due to the larger channel blocks. The existing infrastructure of the analog radio relay network, i.e. primarily the existing system of radio relay towers located between 30 and 70 km apart, can be used by digital systems. In practice, systems in the 1 900, 3 900 and 6 700 MHz bands as well as around 11 GHz are in operation.

7.5 Multiplexed Signals and Multiplexing Equipment

7.5.1 Synchronous Multiplexed Signals

The 2 048 and 1 544 kbit/s signals have already been described in Sect. 7.2.3; their pulse frames are shown in Fig. 7.3. For practical reasons (use of identical components for transmitters, receivers and monitoring equipment with all signal types), CCITT Rec. G.704 [7.22] specifies the following:

- All 2 048 kbit/s signals must have the same frame length (256 bits) and the same bits 1 to 8 in time slot 0 of the frame (see Fig. 7.3b) [1] (the Y bits excepted). This gives a frame repetition frequency of 8 kHz.
- All 1 544 kbit/s signals must have a frame length of 193 bits and the same F bit. The frame repetition frequency is likewise 8 kHz.
- All 2 048 or 1 544 kbit/s signals with octet structure (i.e. signals containing consecutive groups of eight bits) must have a pulse frame with the octet arrangement shown in Figs. 7.3b or 7.3a.
- Similar rules apply to 8 448 and 6 312 kbit/s octet-structured signals.

1 There may be exceptions for "point-to-point" connections. However, these are irrelevant to the ISDN.

Octet-structured signal sources in the ISDN include multiplexers for subscriber signals, concentrators and digital exchanges.

7.5.2 Digital Multiplexers

Digital multiplexing equipment is used to form multiplexed signals with bit rates above 2 048 kbit/s.

As specified in CCITT Recs. G.742 and G.751 [7.23] for the 2 Mbit/s hierarchy, and in G.743 and G.752 [7.24] for the 1.5 Mbit/s hierarchy, priority has been given to high flexibility of use. Multiplexers are therefore so designed that they can combine signals of different origin, i.e. signals with, for instance, different frame structures; these signals may be *plesiochronous*, i.e. with bit rates which are nominally identical but may actually deviate from the nominal value within a certain tolerance range (e.g. $\pm 5 \times 10^{-5}$ at 2 048 and 1 544 kbit/s).

The bit rate of the output signal of a multiplexer is determined by an autonomous crystal-controlled generator and is independent of network synchronization.

Positive justification ("positive stuffing") is employed for the multiplexing process: each input signal is assigned a transmission capacity greater (e.g. by 0.2 percent) than its nominal bit rate. The input signal is written into a buffer store with its own timing signal and read out again by a higher-frequency timing signal corresponding to the transmission capacity. As soon as the phase difference between the write timing signal and the read timing signal exceeds a certain amount, a justifying digit containing no information is inserted in the output signal, thus providing a degree of "breathing space".

Figure 7.14 illustrates the principle of positive justification. In this (simplified) example, if the phase difference between input and output signal has increased by *one* bit interval, a justifying digit is inserted. In practice, if justifying digits are required, they are inserted only at specific positions in the pulse frame. The demultiplexer (receiver) is notified whether or not justification has taken place by means of "justification service digits".

The demultiplexer uses the justification service digits to reconstruct each input signal with no loss of information, i.e. with the original bit rate.

Multilation of the justification service digits causes inadvertent omission or repetition of a bit, i.e. a *slip* (cf. Sects. 7.6.1 and 7.7.2). To avoid this, three or five identical justification bits, distributed over the pulse frame, are transmitted, and "majority decision" is used at the receiver.

The principles described are used to form digital signals at 8 448 and 6 312 kbit/s and higher bit rates which belong to the hierarchies shown in Fig. 7.2. There are also

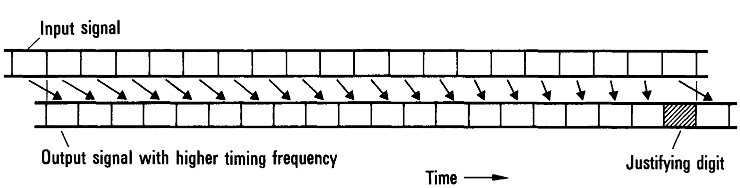


Fig. 7.14. Principle of Positive Justification, as Used in Digital Multiplex Equipment for Timing Alignment

"non-hierarchical" multiplexers which convert e.g. from 45 to 140 Mbit/s ("M3n" in Fig. 7.2). 6312 or 8448 kbit/s signals can also be transmitted without justification – in octet-structured synchronous form – directly from a digital exchange [7.22]. Subscriber signals can also be combined synchronously to form a 2048 or 1544 kbit/s signal in a remote multiplexer (see "Multiplexed Signals" in Sect. 7.4.3). It may be noted that there is a growing tendency to employ synchronous multiplexing even at higher bit rates (cf. Sect. 7.2.4). Standards are being established.

7.6 Network Synchronization

7.6.1 Necessity for Network Synchronization

Two or more digital equipments are said to operate *synchronously* if they have the same timing frequency and hence a fixed phase relationship to each other. The main consideration here is to synchronize the bit timing; once this is achieved, frame alignment is easily established using the *frame alignment signals* (Fig. 7.3). In practice, ideal synchronization is impossible to achieve, especially if equipments are far apart geographically. It is sufficient for the equipments to operate *mesochronously*, i.e. with the same average frequency over time; the phase difference between their respective timing signals can accordingly fluctuate (normally within specified limits). It is customary to describe a network as operating synchronously even if strictly speaking its operation is mesochronous.

If the transmitter and receiver of a digital signal are not synchronous, disturbances occur: the incoming signal is fed at its own timing frequency f_1 into a buffer store and read out at the different local timing frequency f_2. If this is higher than the frequency of the incoming signal, the store is emptied "too quickly", with the result that one or more bits are read out again (i.e. repeated) as soon as the phase difference between the two timing signals has reached a magnitude corresponding to the store capacity.

If the local timing frequency is lower than that of the incoming signal, the buffer memory is emptied "not quickly enough"; as soon as the phase difference reaches the critical magnitude, one or more bits of the incoming signal are skipped, i.e. they are lost.

In both cases the event is known as a *slip*. Slips occur mainly at the inputs to digital exchanges. The multiplex signal inputs of digital exchanges for 1.5, 2, 6 and 8 Mbit/s are equipped accordingly: their buffer stores are so designed that, if a slip occurs, one pulse frame (for 1.5 and 2 Mbit/s: see Fig. 7.3) of the multiplex signal is repeated or is lost, i.e. one octet (= 8 bits) for each 64 kbit/s signal. In PCM systems (see Sect. 7.2.1) this corresponds to a PCM code word.

The pulse frame or PCM code word occurs every 125 µs. Accordingly a slip occurs as soon as the time instant marking the start of the pulse frame of the incoming signal (with timing frequency f_1) has shifted by 125 µs relative to the corresponding time instant of the local timing system operating at f_2. The mean interval between two slips is therefore

$$T_S = \left| \frac{125\ \mu\text{s}}{(f_1 - f_2)/f_2} \right| . \tag{7.1}$$

As the slips produce disturbances of varying magnitude (discussed in Sect. 7.7.2), it is desirable to eliminate them as far as possible. All the clocks in a digital network are therefore at least nominally operated synchronously. Slips cannot be totally prevented in this way, because from time to time a digital exchange may be unable to run synchronously (Sect. 7.6.3), and in digital multiplexers slips can occur as a consequence of bit errors (Sect. 7.5.2). Slips will also occur in traffic between two networks each of which is synchronized independently (e.g. two national networks).

7.6.2 Achieving Network Synchronization

A synchronized network will always be limited geographically, usually to the size of a country such as the Federal Republic of Germany. For practical reasons, a country may also be divided into several regions, each independently synchronized, or several telcom carriers may each have a separate synchronous network. Alternatively, two or more small countries may implement a jointly synchronized network.

In the past, several methods of network synchronization have been discussed and some have been analyzed in great detail; however, in the following we shall only describe the method which in practice is likely to be used virtually exclusively, namely the *master-slave method.* In this method a primary reference clock (the *master*) controls all the exchanges directly or via intermediate stages and thus determines the frequency of all the 64, 1 544 and 2 048 kbit/s signals in the network (as well as the frequency of the 6 312 or 8 448 kbit/s signals originating directly from digital exchanges, Sect. 7.5.2). In the future, with the evolving broadband ISDN and in conjunction with new synchronous multiplex structures, synchronization will be extended to higher bit rates.

The reference clock is a caesium-beam oscillator with a frequency uncertainty of not more than $\pm 10^{-11}$ (cf. Sect. 7.6.3). It is preferably located near the geographical center of the synchronized network, e.g. in the Federal Republic of Germany at Darmstadt, and for the network of the AT & T at Hillsboro (Missouri). The reference clock passes on its frequency to other equipments either as 2 048 kbit/s or 1 544 kbit/s signal timing or, where digital links are not yet established, using FDM *frequency comparison pilots.* (The Hillsboro clock delivers a 2 048 kHz analog signal). Suitable reference timing can also be derived from radio signals of the LORAN-C navigation system.

The synchronizing signals are generally distributed from the top down in accordance with the switching network hierarchy, as shown in Fig. 7.15 (cf. Sect. 3.2.1 and Figs. 3.2, 3.3). This should not be regarded as a rigid principle; the illustration also shows an example of a possible direct path bypassing one hierarchy level (a).

If a primary synchronization path normally used for top-down synchronization fails, if possible a secondary path should be available. It may originate from the same network node or exchange as the normal synchronization path but via a different transmission route (b), or else from a different exchange (c).

A local exchange connected only to *one* primary center does not need a secondary path for synchronization, because if the digital signal from the primary center fails, it can anyway handle only local digital traffic, for which the frequency accuracy of the free-running local exchange (tolerance typically $\pm 10^{-7}$) is adequate.

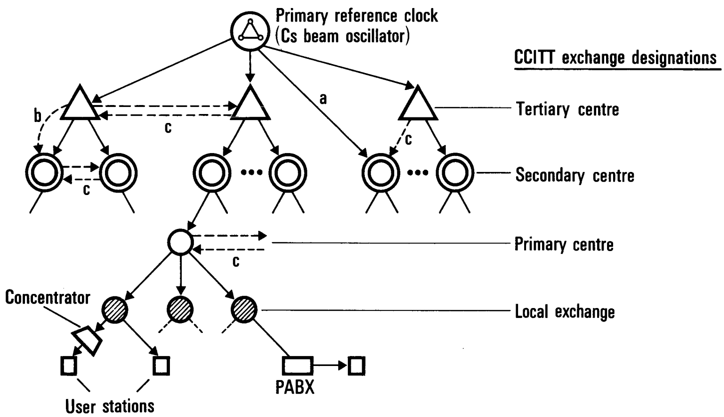


Fig. 7.15. Hierarchical Structure of a Network Synchronization System.
⟶ Normal synchronization path
- - → Secondary synchronization path
a direct path
b,c see text
PABX Private automatic branch exchange

Private branch exchanges as well as digital subscriber stations are synchronized from the public network, i.e. generally from a local exchange.

7.6.3 Clock Supply Requirements

CCITT Rec. G.811 [7.25] gives guidelines for the synchronization of national networks in respect of ISDN requirements, particularly as regards international interworking. The recommendation provides for primary reference clocks as mentioned above with a frequency deviation of up to $\pm 10^{-11}$.

In practice, a generator of this accuracy can only be implemented as a caesium frequency standard. As the clock of every synchronized digital network – for example a national network – is determined by a caesium standard of this kind, the average interval between two slips in international traffic is theoretically not less than $125\,\mu s/2 \times 10^{-11}$ = approximately 70 days (from Eq. 7.1).

CCITT Rec. G.811 also gives maximum values for the phase variation at the primary reference clock output (3 μs) and at the output of a network node (i.e. generally an exchange) handling international traffic (10 μs). These variations are with respect to a hypothetical signal with the mean frequency of the caesium reference clock. In international traffic the phase variations are superimposed on the regular drift of the two national reference clocks; in addition, signal delay variations on the international circuit can occur. The consequence of both effects is that the actual interval between two slips may be greater or less than 70 days; hence 70 days is only a theoretical average.

Finally, CCITT Rec. G.811 also specifies the frequency deviation allowed for free-running timing pulse generators in digital exchanges (and in principle other network nodes also) (Table 7.3).

Table 7.3. Frequency Tolerance of Clocks in Exchanges. Category a corresponds to a slip interval of ≧4.8 hours over the entire connection and of ≧3.5 hours in case of the local exchange, category b to a slip interval of between 2 minutes and 4.8 hours

Category	Maximum frequency deviation		Maximum proportion of time in which the frequency deviation may occur	
	Local exchange	Transit exchange	Local exchange	Transit exchange
a	$\pm 10^{-8}$	$\pm 2\times 10^{-9}$	1%	0.05%
b	$\pm 10^{-6}$	$\pm 5\times 10^{-7}$	0.1%	0.005%

The figures in the table are based on the following assumptions:

- An international connection comprises two national portions (see Fig. 7.6) and an international portion with up to three transit exchanges in third countries, hence in total two local and up to 11 transit exchanges.
- At any given time only one exchange is out of synchronization. (To be more precise: only one may fully exploit the permitted frequency tolerance.)
- During this time the slip rate arising from international plesiochronous operation (on average one slip every 70 days) is negligible.
- Free-running of a local office causes slips at only one point, free-running of a transit exchange at two points, as shown in Fig. 7.16 (the points being those at which a transition from the accurate to the inaccurate timing or vice versa takes place). Therefore, for the same slip frequency, the frequency inaccuracy of a transit exchange may only be half that of a local office.
- The time portions are selected such that slips due to the free-running of exchanges occur in no more than 2.8 % of the time, and hence the requirements of CCITT Rec. G.822 (Sect. 7.7.2) are essentially satisfied.

In the CCITT Blue Book (1989), the specification summarized above will be replaced by one which is more specific to the "nodes" or exchanges (new rec. G.81X), while retaining the basic philosophy. Rec. G.811 will be restricted to primary reference clocks.

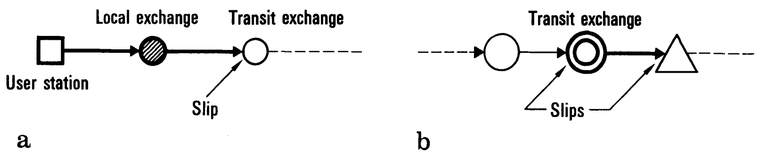


Fig. 7.16a, b. Illustrations of Points (=inputs to exchanges) at which Slips Occur (non-synchronous sections are in heavy lines).
Examples: **a** Local exchange with associated lines is free-running; **b** Transit exchange (here, a secondary exchange) with outgoing lines is free-running

7.7 Disturbances and Transmission Performance

The transmission performance on a digital link is characterized by four variables: bit errors, slips, transmission delay and phase jitter. Their undesirable effects are considered below with particular reference to the ISDN.

7.7.1 Effect of Bit Errors

Just as in the analog telephone network noise interference is unavoidable but can be tolerated if it stays within certain limits, so in the digital network it is inevitable that bits in the digital signal will be multilated by interference (1 becomes 0 or vice versa). This can also be tolerated if it does not occur too frequently.

Bit errors are either caused by external interference or by thermal noise. *External interference* includes:

- Dial pulses or other pulses on wire pairs in the same cable which are not yet operated digitally: these pulses affect the disturbed wire pair through unavoidable electromagnetic coupling.
- Crosstalk between wire pairs transporting similar signals. (In principle this effect can be prevented with careful planning.)
- External electromagnetic effects due e.g. to railways with electrical traction, especially if thyristor control is employed.

These disturbances primarily affect symmetrical pairs in the local network and on subscriber lines. These are therefore assigned a comparatively large proportion of the total bit errors permissible in a connection between two subscribers (see the discussion of CCITT Rec. G.821 below). Bit errors very often occurs in bursts; depending on the circumstances, an error burst can affect between 2 and 50 or in some cases even more consecutive bits.

In coaxial cables, optical fibers and radio relay links, *thermal noise* is a major cause of bit errors. These have a purely random distribution (Poisson distribution) and are amenable to precise planning, which in practice is a compromise between optimum transmission quality and cost considerations.

Bit errors affect individual services in different ways:

- PCM speech transmission: a bit error ratio or error-burst rate of 10^{-5} can be tolerated. Even random (Poisson) errors with a frequency of 10^{-4}, such as may occur briefly in telephony via satellites, only cause slight clicks (not continuous noise).
- Data transmission with error detection (methods using Automatic Repeat Request ARQ [7.26] such as High Level Data Link Control HDLC): if a single bit error or an error burst occurs in a data block, this is detected by the data receiver, which then causes the data block to be repeated. In order to ensure that the effective data throughput is not excessively reduced due to repetitions (i.e. reduced by no more than 10 to 20 %) – and this also applies to long data blocks or a long transmission delay (via satellite) [7.15] – a bit error ratio of less than 10^{-6} is desirable.
- Text transmission: modern text transmission, e.g. teletex, employs protected data blocks; the same comments apply as for data transmission.

- Facsimile transmission: the system mainly considered for ISDN use is ISDN telefax (Sect. 2.3.1.2); here, too, block by block protection is provided by the HDLC method, so that as far as the effect of bit errors and the resulting requirements are concerned, the same comments apply as to data transmission (also applicable to ISDN textfax).
- Common channel signaling according to CCITT Signaling System No. 7 (cf. Sect. 6.3): this is also a special case of block by block data transmission using an HDLC procedure. Owing to this data protection, the probability of an incorrectly established call is several orders of magnitude less than the frequency of a block disturbance.

CCITT Rec. G.821 [7.27] takes into account the described requirements for services. It also specifies the bit error performance with which future new services will have to operate. For an international 64 kbit/s reference connection between two subscribers the recommendation specifies:

- *Degraded minutes*: fewer than 10 % of one-minute intervals should have a bit error ratio worse than 10^{-6}. (In practice this means more than four bit errors.)
- *Severely errored seconds*: fewer than 0.2 % of one-second intervals should have a bit error ratio worse than 10^{-3}.
- *Errored seconds*: fewer than 8 % of one-second intervals should have any errors.

For the degraded minutes and errored seconds objectives, Rec. G.821 gives an allocation to three "circuit classifications" as shown in Fig. 7.17. The classification contains some degrees of freedom to allow for different conditions in countries of different sizes. For "medium-sized" countries such as Germany, it is advisable to allocate the subscriber line the 15 % quoted as an example in the figure, and to extend the "medium grade" quality as far as the primary center (class 4 office) or secondary center (class 3 office).

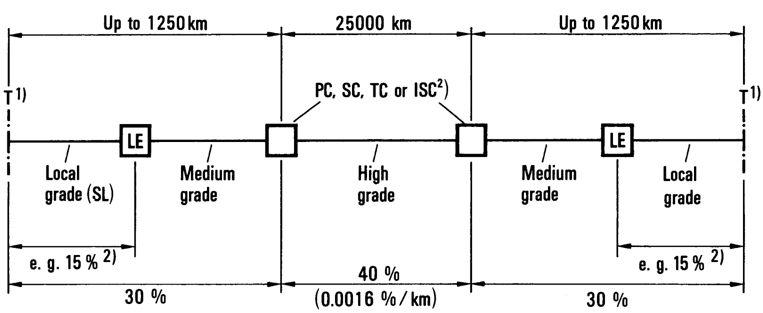


Fig. 7.17. Distribution of Permissible 1-Minute Intervals with >4 Bit Errors and 1-Second Intervals with >0 Bit Errors Among the Portions of a Complete 64 kbit/s Connection.

SL Subscriber line
LE Local exchange
PC Primary center
SC Secondary center
TC Tertiary center
ISC International switching center

1) ISDN reference point T (see Sect. 4.1)
2) depends on choice by administration or carrier

Other relevant details of CCITT Rec. G.821 include the following:

- For "high grade" long-haul systems it is required that *per kilometer* less than 0.00016 % of all 1-minute intervals shall have five or more bit errors and that less than 0.000128 % of all 1-second intervals shall have any bit errors at all.
- A satellite section (within the "high grade" portion) may have more than four bit errors in a maximum of 2 % of all minutes.
- For the "OK" 1-minute intervals (with four or less bit errors) there is no specified distribution of the bit error ratio among sub-sections.

These requirements apply initially to a 64 kbit/s channel.

Additional specifications (but consistent with Rec. G.821) for digital sections (cf. Sect. 7.4.1) in the medium and high grade portions of Fig. 7.17 are given in Rec. G.921 [7.16]. It may be noted that the CCITT has recently agreed to include in G.921 a clause according to which the degraded minutes and severely errored seconds requirements will also be applicable to the line bit rate (e.g. 2 048 kbit/s) – i.e. not only to a 64 kbit/s channel.

7.7.2 Effect of Slips

Slips (see Sects. 7.5.2 and 7.6.1) should be regarded in the same way as other unavoidable disturbances. Every service can be affected by them, and their effect depends on the nature of the signal:

- PCM speech transmission: a slip produces a phase and amplitude shift, which is usually inaudible or at most can be heard as a click. In practice about 20 slips per minute are permissible.
- Data transmission with error detection (ARQ method): just like a bit error, a slip causes a data block to be received incorrectly. The data block must then be repeated. Slips may therefore occur no more frequently than bit errors. Because the occurrence of slips can be kept well under control, it appears reasonable to require that slips occur at no more than one tenth the frequency of bit errors. As CCITT Rec. G.821 (see Sect. 7.7.1) specifies a maximum of four bit errors in 90 % of all 1-minute intervals, about four slips in ten minutes would be permissible.
- Text and facsimile transmission: the remarks on the effect of bit errors (Sect. 7.7.1) similarly apply here: the undesirable effect is in principle the same as for data transmission and gives rise to the same requirements.
- Common channel signaling complying with CCITT Signaling System No. 7 (cf. Sect. 6.3): here, too, slips have the same effect as bit errors. It is only if the data block disturbance is not detected by the receiver – highly unlikely due to the error correction procedure (see Sect. 7.7.1) – that e.g. an unwanted connection could be set up.
- Transmission of data in the form of multiplexed signals (signals conforming to CCITT Recs. X.50 and X.51 [7.28] or X.22 [7.29] comprising e.g. 20 tributary signals each of 2.4 kbit/s): in the ISDN, signals conforming to CCITT Recs. X.50 and X.51 occur on leased lines used specially for data communication. Signals conforming to CCITT Rec. X.22 on the other hand are transferred from subscriber to subscriber. With multiplexed signals, a slip causes the receiver frame alignment

to be lost. The receiver must therefore recover frame alignment. All the individual signals within the multiplexed signal are affected and it is even possible that an entire data switching center may be put on alarm status. Slips may therefore only occur at intervals of at least 15 minutes.

CCITT Rec. G.822 [7.30] takes into account both the known requirements of the services and also what is possible in practice. This Recommendation, like CCITT Rec. G.821 (Sect. 7.7.1) applies to a 64 kbit/s connection in the ISDN, which can be 27 500 km long. For such a connection, Rec. G.822 specifies that in at least 98.9 % of the time the average interval between two slips shall be not less than 4.8 hours and for virtually the rest of the time (about 1 % max.) between 4.8 hours and 2 minutes. (The slip rate may be even higher for up to 0.1 % of the time.) The timing accuracy of digital exchanges must be set accordingly, even under free-running conditions (when clock control "from above" has failed due to a fault) (see Sect. 7.6).

7.7.3 Effect of Signal Delay

Signal delay does not generally receive as much attention as bit errors and slips. However, it is also an important factor, especially in the following cases:

- In *voice communication*, as the signal delay increases, conversation becomes more difficult. The CCITT has therefore laid down in Rec. G.114 [7.31] a maximum signal delay for telephony of 400 ms (for *one* transmission direction); this means that *one* satellite section (signal delay T_D: about 260 ms) is permitted. The 400 ms should only be exceeded in exceptional circumstances, e.g. if no connection at all can otherwise be set up.
- In *data transmission* with ARQ (see Sect. 7.7.1), buffer stores must be available at least in the data transmitter, the store size K corresponding to the number of bits that can be transmitted within $2T_D$, i.e. twice the signal delay time: $K = 64\text{ kbit/s} \times 2T_D$. In the case of data transmission with block-by-block acknowledgement, a long signal delay (e.g. via satellites) can considerably prolong the actual duration of an individual connection.

7.7.4 Effect of Jitter and Wander

As already described in Sect. 7.6, absolutely rigid synchronization of all signals in the digital network is impossible to achieve in practice; frequency variations within a certain range are unavoidable. A frequency variation (i.e. unintentional frequency modulation) can always be described as a phase variation (phase modulation).

Relatively rapid phase variations (with a frequency greater than 20 Hz) are known as *jitter*, slower variations as *wander*. Jitter is mainly attributable to imperfect timing recovery in regenerators of line systems (Sect. 7.4.2), wander to justification processes in digital multiplexers (Sect. 7.5), control errors of phase-locked loops and temperature-dependent signal delay fluctuations in cables.

As jitter and wander are to a certain extent unavoidable, tolerance diagrams are used to specify, for all the interfaces in the digital hierarchy (see Sect. 7.2 and Fig. 7.2) and for the S/T interface (Sect. 4.2.2), how much jitter or wander is permissible at the

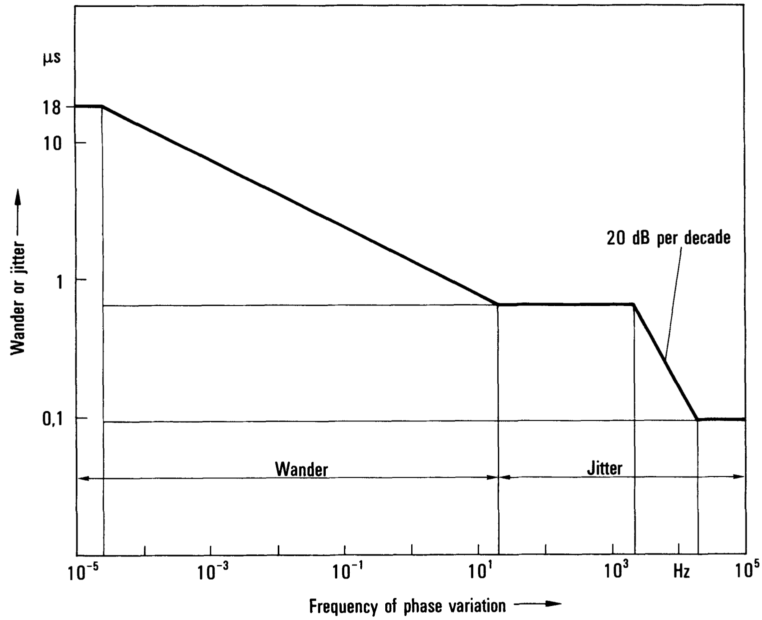


Fig. 7.18. Maximum Permissible Wander and Jitter at the Input to a Network Node (in particular, a digital exchange).
The ordinate values are "peak-to-peak" values for a 2048 kbit/s signal

input of a unit without causing bit errors, slips or other disturbances. The diagram for the 2048 kbit/s interface in Fig. 7.18 is shown as an example [7.32]. A similar requirement applies to the 1544 kbit/s interface. A pseudorandom signal (period $2^{15}-1$ bits, or $2^{20}-1$ bits, respectively) with sinusoidal phase modulation is used as the test signal; a suitable test signal generator (and a jitter measuring set) is specified in CCITT Rec. O.171 [7.33]. If a unit can tolerate the test jitter at the input, then it can be assumed that the jitter present during actual operation (and which naturally has no sinusoidal modulation) will not produce disturbances.

Jitter and wander as defined in the interface specifications [7.32] can always be controlled and hence do not impair the onward signal.

7.7.5 Connection Types in the ISDN

CCITT Rec. I.340 [7.34] specifies *connection types* used to implement network connections for the individual services. Each connection type is described by a number of attributes. The most important connection types in the ISDN will probably be circuit-switched "transparent" and "non-transparent" B-channel connections (see below). The details have not been finalized but the essential attributes will include the transmission quality parameters (cf. Table 6.1).

For the *transparent* connection type, the bit error performance will comply with CCITT Rec. G.821, and the slip frequency with CCITT Rec. G.822; bit integrity is also guaranteed, i.e. no bit or octet in the 64 kbit/s signal is intentionally changed e.g. by recoding from 64 to 32 kbit/s (see below).

Non-transparent connections will probably only exist for telephony. Bit manipulations are possible here, in particular:

- Conversion from the PCM code in accordance with the A-law (Sect. 7.2.1) as used in Europe to the PCM μ-law code used e.g. in the USA, or vice versa; this means replacing one octet (PCM code word) by another.
- Conversion from PCM to ADPCM to reduce the bit rate to 32 kbit/s (see Sect. 7.2.1).

8 ISDN – The User's View

The evaluation of the ISDN from the user's point of view is intended to show the ways in which the ISDN improves communication in terms of meeting user needs and supporting new applications. It also considers those requirements that will for some time have to remain unsatisfied. In general, it should be emphasized that the ISDN improves communication through simpler operation, improved accessibility as well as easier and faster access to information. The three main groups of users, each with their own profile of communication requirements, will be considered separately: people in the office, people at home and people on the move.

8.1 ISDN in the Office

8.1.1 Telephone Communication

Of all the methods of communication used in the office, telephony will continue to be the most important.

Private branch exchanges already provide efficient, high-convenience telephony. Here, the new features offered by the ISDN (see Chap. 2) will improve telephone communication still further [8.1]. It is important to remember that with the implementation of the public ISDN the use of these sophisticated features will not be limited to the use within private branch exchanges, as a considerable proportion of business voice traffic goes beyond the confines of private branch exchanges.

The improvements brought by the ISDN to telephone communication relate particularly to call establishment and the accessibility of call partners. In business traffic especially it is all too often the case that either no-one answers or someone other than the intended person answers, or that the line is busy. A number of studies (including [8.2]) have been conducted on the subject of the accessibility of called parties. These showed that in the business sector only 30 % of all call attempts are successful in the sense that the wanted party is obtained on the first call, and that 10 % of telephoning time is used on wasted calls. Note that it is not the telephoning time which can be saved that is important so much as accelerating the information exchange and hence speeding up office procedures. The problem of party accessibility is likely to become worse rather than better in future because telephone traffic is increasing all the time. Furthermore, in view of the continual increase in business travel it can be assumed that absence from the office will be more and more frequent. The supplementary services offered by the ISDN to improve accessibility, such as *registration of incoming calls, call waiting* (see Sect. 2.3.3 and Table 2.2) are therefore very important, and in this respect the ISDN can be said to provide secretarial services for every subscriber.

It should also be mentioned that the ISDN will enable some special types of telephone communication already available to be improved. The advantages of the ISDN in handsfree speaking have already been mentioned in Sect. 5.3.1. With respect to telephone conferences, the signaling channel provides better control and thus more efficient conferencing. The two B-channels could also be used to implement a type of artificial stereophony to facilitate identification of the speakers in conference calls. Finally, use of the *voice mail* service can be simplified by indicating voice mail system user instructions on the ISDN telephone display, thus avoiding voice output for user prompting.

8.1.2 Non-Voice Communication

8.1.2.1 Significance of Service Integration

Of crucial importance for service integration is the fact that an increasing number of office desks have electronic equipment for local processing and storage of text, data and graphics, in addition to telephones (Fig. 8.1). The introduction of personal computers has substantially contributed to this development. Because of their inexpensiveness, personal computers can be cost-effectively employed for a wide range of applications, and they can also be justified in places where they are less intensively used than in typically procedure-oriented work as for instance financial accounting. Hence personal computers are increasingly being used at general office desks supporting a mixture of office activities such as text editing, running specific user programs or preparing graphic displays, as well as performing personal tasks such as maintaining appointment diaries and private files.

Every desk equipped with text and data processing equipment also needs electronic communication facilities so that processed texts can be forwarded to their destinations, databases accessed, or programs retrieved from a central program memory. Services such as text mail and videotex intensify the need for communication still further. Office desk-top equipment is therefore being provided with communications capability, with the result that text, data and facsimile communication are being increasingly used in addition to voice.

At these desks, where communication in several information types is necessary, the advantages of an integrated services network are fully realized.

- All the terminals of an office desk can be reached under *one* directory number. The dedicated terminal (or the appropriate unit of a multifunction terminal) is accessed automatically. Users do not have to contend with a multiplicity of

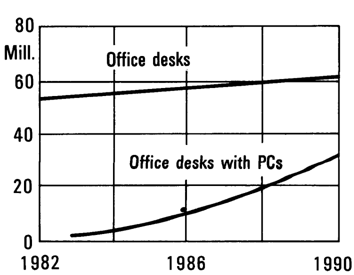


Fig. 8.1. Estimated Penetration of PC-Based Office Desks in the U.S. [8.3]

directory numbers, possibly of different formats (e.g. telephone number and teletex subscriber address with digits and letters).

- All the communication procedures operate to a standard pattern, because only *one* network is involved. Consequently, when additional terminals are added to the office desk, learning periods are shorter and the incidence of user errors while working with the communcation system is minimized.
- The fact that communication with a party is simultaneously possible via both 64 kbit/s channels of the ISDN basic access can be utilized to communicate in more than one information type, e.g. transmitting a facsimile or text during a telephone call. This considerably improves communication especially in the business sector.
- It is also possible to have connections simultaneously to two different destination thanks to the two 64 kbit/s channels. During a telephone call the user could be in contact with a data processing system or a database to retrieve information needed for the telephone call or to record data resulting from the telephone call immediately.
- The advantage of a standard socket for the ISDN terminals of all services should not be underestimated. Quite apart from the fact that there is no need to ensure that the right sockets are installed for the particular terminal configuration, terminals can be flexibly connected to the S-bus to meet changed arrangements within a room. Moreover, the multichannel ISDN access can be implemented using a normal telephone circuit, so that no new wiring is required for ISDN introduction. If a desk is equipped with a second terminal besides the telephone, e.g. a PC with communication capability, it is no longer necessary to provide a second line.

8.1.2.2 Transmission Speed

Bit rate requirements for text, data and facsimile transmission in the business sector depend to a large extent on the specific applications. Fig. 8.2 contains a summary of these application-dependent bit rate requirements and relates them to the 64 kbit/s rate of the B-channel.

The B-channel can generally be regarded as adequate for direct communication between people. It also allows direct personal information exchange in case of facsimile communication, because just a few seconds are required to transmit an A4 page with the quality possible on today's facsimile machines. With the facsimile units specifically envisioned for ISDN use (CCITT group 4), considerably higher resolution of the copy is feasible (e.g. by a factor of 4) and a greater volume of data per A4 page must then be transmitted. Depending on the contents of the page, at 64 kbit/s the transmission time might then be considered excessively long for efficient interpersonal communication.

As soon as workstations and computers become participants in the communication process, i.e. communication is actually taking place between memories, many different applications become possible, with correspondingly varied requirements in terms of transmission speed. If workstations are exchanging files with hosts or if they are remotely loading programs, the capacity of the B-channel is generally adequate. For file transfer between computers, especially in integrated operation (data, load-sharing, functional and safeguarding integration) there are cases where speeds in excess of 64 kbit/s would be desirable. The B-channel will also be too slow where the

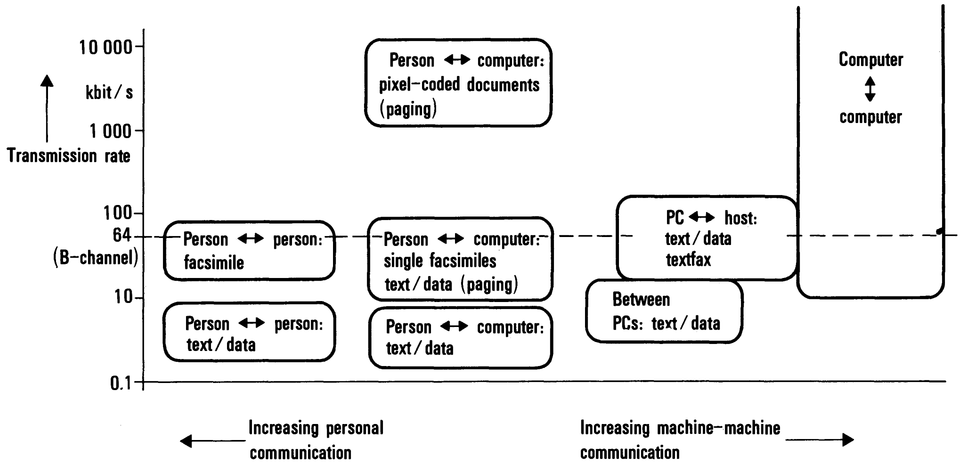


Fig. 8.2. Transmission Rate Requirements for Various Forms of Non-Voice Communication

aim is to page through pixel-coded documents stored in a remote database or archive. This application is gaining in importance with the spread of personal computers with pixel-based screens. Assuming a frequency of two pages per second for efficient paging and taking into account the overhead times that occur in data transmission, the resultant bit rates are in the megabits-per-second range. The same applies if large numbers of facsimiles are transmitted in batch mode in companies' internal electronic mail systems. Channels operating at some tens or hundreds of megabits per second, such as are under discussion for broadband ISDN, are more suitable for applications of this kind. Local area networks (LANs) are also a possibility for local transmission (cf. Sect. 3.6.2).

At this point, some discussion of call set-up times is necessary. The call set-up time for circuit-switched connections in the public ISDN will be between 1 and 3 seconds depending on the number of exchanges involved. In conjunction with pushbutton dialing this is considerably faster than in today's telephone network. The speed of call set-up achieved is sufficient for telephony, for occasional data exchange with hosts and messaging systems and for occasional direct transmission of text and facsimile in the user situations described in Sect. 8.1.2.1. With more intensive workstation applications, e.g. document production using document servers, users are however likely to desire shorter call establishment times (see also 8.1.2.3).

8.1.2.3 Message Structure

As well as circuit-switched 64 kbit/s connections, the ISDN will also provide 64 kbit/s packet mode connections. Details and the advantages of the packet switching principle are given in Chap. 4. The main advantages include efficient use of trunk circuits and computer ports for interactive data communication between workstations and hosts (dialog mode).

Terminals are evolving from simple retrieval devices (such as video data terminals) into workstations having their own processing and storage systems. A change of the structure of messages exchanged between workstation and host is therefore to be expected. Whereas in the case of simple data terminals the entire transaction must be conducted directly with the host, a workstation offers the user the possibility of handling interactive work locally in most cases. Contact with the host is mainly confined to occasional transfer of large quantities of data (e.g. a text prepared locally) or of programs used at the workstation. In this situation it is also conceivable to use a circuit-switched B-channel and to set up the connection for the data transmission phases only. From the point of view of line economy, therefore, the demand for packet-switched B-channels may become less urgent as the above-mentioned terminal evolution proceeds. However, packet mode operation would still have the ability to support several virtual connections simultaneously, permitting among other things the address-multiplex connection which is advantageous for computers.

8.2 ISDN in the Home

An increasing demand for communication facilities in the home, though not as marked as in the office environment, is nevertheless a discernible trend. This is due to a number of factors:

- As the result of increased mobility for professional reasons or to improve the quality of housing, families and circles of acquaintances are more widely scattered.
- Private business transactions (banking, insurance, tax, shopping) are becoming ever more numerous and complex. At the same time there is a growing demand for information for everyday living (public transport, hours of business, events).
- Leisure time is on the increase.
- Security is becoming a growth area.
- Increasingly high-quality, low-cost electronic equipment for home entertainment, information and communication is coming onto the market (video recorders, video disk players, home computers).

Table 8.1 shows how the potential uses of communication and information technology will develop in different spheres of private life. The ISDN may be seen as providing a sound basis for efficient, user-friendly implementation of a number of these potential applications.

The universal, multichannel access to the network is likely to be especially attractive for home users. In the home, as in the office environment, a demand will arise for communication in different information types. Videotex will become established alongside voice communication. As home computer ownership increases, so will the requirement for text and data communication.

As well as service integration, multichannel network access will be welcomed in private households where, as a rule, several people share a single line. For example, it will be possible to use the videotex service without busying out the telephone line.

Messaging services, as available in some videotex systems, will be enhanced by providing an indication at the subscriber terminal that a message has been deposited in the electronic mailbox ("incoming message waiting indication", see Sects. 2.3.1.3 and 2.3.3). This function is particularly important for the private subscriber; although it is

Table 8.1. Use of Communication and Information Technology in the Home

Entertainment, Education
- Television
  offering a wider range of programs via cable TV, video recorders, video disk
- Electronic games
  local, but also with transmission capability
- Sound program reception/reproduction
  enhanced by digital technology
- Use of home computers for hobbies, education

Communication, Information
- Telephony
  greater convenience provided by e.g. pushbutton dialing, abbreviated dialing, name keys, alphanumeric display, home private branch exchanges and cordless telephones
- Information retrieval
  videotex, teletext; in the more distant future via cable: cable text, image retrieval
- Text communication
  e.g. via videotex
- Remote ordering, teleshopping, banking, telesoftware via videotex

Private Office Work Using Home Computers
- Domestic bookkeeping
- Information storage
  addresses, appointments, power consumption
- Correspondence

Security and Control
- Remote control
  heating, kitchen appliances, video recorders, ...
- Monitoring/control of power consumption
- Alarm messages:
  intrusion, fire
- Emergency calls

possible to notify him of received messages each time he uses videotex, this cannot be assumed to be a daily occurrence in the home environment. Moreover, the average private subscriber is likely to receive a message somewhat infrequently and so will not regularly check his mailbox. The D-channel offers the best possibility for signaling "message waiting" to the subscriber, because signals can be exchanged via this channel independently of the use of the B-channels.

The same is true if the signaling channel is used for transmission of emergency signals. This improves the effectiveness of alarm and emergency call systems.

Communication applications in the home also benefit from the high transmission rate in the ISDN access. This applies particularly to videotex. With the 64 kbit/s channel and with pages complying with the current standard [8.4, 8.5], the achievable transmission time of a few seconds for a page allows rapid paging through texts. In addition, the loading of character sets determined by the information provider (DRCS: dynamically redefinable character sets) to prepare the display of subsequent images is then barely noticeable.

The high transmission rate is also indispensable for enhancing the videotex information with photographic images. The transmission time for the latter depends on

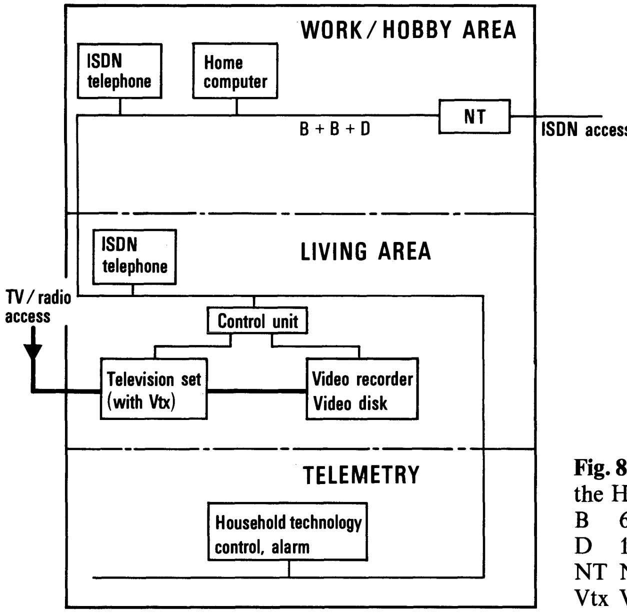


Fig. 8.3. Use of the ISDN Access in the Home.
B 64 kbit/s B channel
D 16 kbit/s signaling channel
NT Network termination
Vtx Videotex

the resolution and coding. Even at 64 kbit/s, about 30 seconds is required to transmit an image corresponding to a color television picture. The same method whereby photographic images are represented and transmitted on videotex could also be used to exchange still images between subscribers (still image transfer service, see Sect. 2.3.1.2). Home video cameras and television sets could be used for this purpose.

The high transmission rate of the ISDN can even be used to advantage for data transmission in the home, e.g. for transmitting programs provided by computer service centers or distributors to the home computer (telesoftware).

The bus design of the ISDN access (see Sect. 4.2) also has benefits for domestic applications. The permissible line length of 100 to 150 m for the passive bus should allow the maximum permissible number of terminals to be wired up as required in a fairly large apartment or average family house (Fig. 8.3). However, if there are substantial internal traffic requirements, e.g. if it is desirable to have internal calls and external traffic at the same time, a network termination incorporating private branch exchange functions must be used. The access network will then have a star or star/bus configuration (see Sect. 4.2).

As well as voice connections, data links within the home may also be required. A typical application is shown in Fig. 8.3: the television set, if equipped for videotex, can also be used as a data monitor in conjunction with the home computer. This would make it possible to use the television set in the living area to access computer programs (e.g. games) running on the home computer set up in the work area. Under control of the home computer, it would even be possible to insert television pictures from a video storage unit (video recorder, video disk player) into the program sequence.

The ISDN bus can also be used for collecting and distributing telemetry data in the home for control and monitoring purposes.

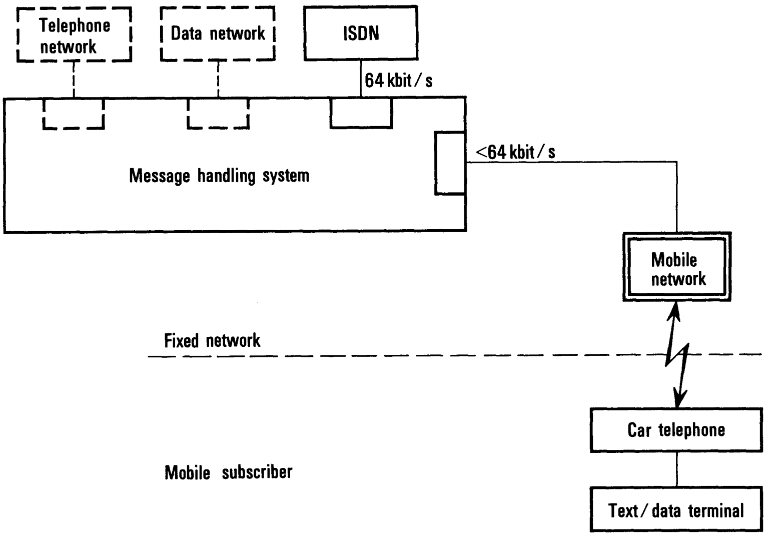


Fig. 8.4. Connection of Mobile Subscribers to the ISDN Text and Data Services via a Message Handling System

8.3 ISDN in Mobile Communication

In Sect. 8.1.1 it was explained how the ISDN supplementary services for call establishment improve subscriber accessibility. These supplementary services allow a degree of subscriber mobility in the ISDN, always provided there is access to a fixed local line. The present section deals with the significance of the ISDN for mobile radio subscribers. Of the various mobile radio systems in operation today, only the *public land mobile network* is considered here, as it alone may generally be regarded as the continuation of the public wired communication network into the mobile domain. New developments in radio systems and mobile telephones [8.6] will provide a very sophisticated and also cost-effective public land mobile system. A rapid increase in the number of subscribers can therefore be expected.

At present the public land mobile network only provides voice communication using car telephones. However, with the spread of text communication in the wired network, the need to send a text to someone traveling by car will also increase. Aside from the general advantages of text communication, receiving a message in textual form has an additional advantage for mobile subscribers: unlike receiving a telephone call, a car driver does not have to divide his attention between driving and communicating. Many sales or service representatives in the field need up-to-date information provided by company databases, e.g. maintenance assignments. Having a data communication facility in the car saves the user the trouble of finding a public telephone. Data entry while *en route* could also be important for purposes of up-to-the-minute data gathering. In conjunction with videotex, the possibility arises of using mobile data communication for retrieving the information required for car journeys: traffic reports, directions in unfamiliar localities (road and city maps), parking facilities and much else besides.

Hence the ISDN services for text and data communication also have a role to play in the public land mobile network. Moreover, the possibility of service integration offered by the ISDN has the advantage that the mobile system does not have to communicate with several different dedicated wired communication networks (e.g. the telephone network and the data network). This means that terminals installed in cars and their operation can be made more uniform.

Only *one* type of interworking facility between mobile and line communication network is required.

However, it is not possible for reasons of cost and of limited frequency bandwidth resources to provide mobile subscribers with 64 kbit/s channels. For instance, recent plans for a European digital mobile system envisage a gross bit rate of 22.8 kbit/s. However, in mobile communication it is necessary to cope with adverse conditions of radio propagation which lead to high values of bit error ratio. Consequently, a large proportion of the gross bit rate is to be assigned to error protection; the remaining net bit rate for voice will be 13 kbit/s, and for non-voice services, 9.6 kbit/s at most. A possible method of realizing non-voice traffic between mobile subscribers and the general ISDN would be to provide mobile subscribers with access to the message handling system described in Sect. 2.3.1.3, which performs the appropriate rate adaption to the ISDN (Fig. 8.4). This will enable mobile subscribers to communicate with fixed subscribers to ISDN text and data communication services while those fixed subscribers maintain their normal data rate, e.g. 64 kbit/s.

ANNEX: CCITT Recommendations and Other Standards Relating to the ISDN

1 Introductory Remarks

Agreement has to be reached in a large number of areas to achieve the ultimate aim of communication between any two ISDN subscribers. This question is receiving the attention of a number of national and international bodies.

The CCITT (Comité Consultatif International Télégraphique et Téléphonique or International Telegraph and Telephone Consultative Committee), the standardizing body of the telecommunication administrations and carriers, defines the ISDN concept as well as interfaces and signaling procedures for the ISDN in a series of recommendations; it deals equally with telecommunication services and terminals, and with transmission. The ISO (International Organization for Standardization) prepares standards for data communication, partly in cooperation with CCITT. These standards include general basic principles of protocol architecture. The IEC (International Electrotechnical Commission) produces agreements on such electrical and electromechanical questions as electrical safety and connectors. The ECMA (European Computer Manufacturers Association) is concerned with aspects of communication in data processing systems, including their connection to ISDN private branch exchanges. In the USA, the Exchange Carriers Standards Committee (ECSA) sponsors the Standards Committee T1-Telecommunications. This committee is open to membership by all interested parties, i.e. exchange carriers, manufacturers, government agencies, etc. T1 develops both national standards (to be promulgated by the American National Standards Institute, ANSI) and U.S. contributions to CCITT. T1 has six technical Subcommittees, each with three to five Working Groups, e.g. Working Group T1D1.1 on "ISDN architectures and services".

This Annex only includes those standards that relate to the ISDN concept and the operation of voice, text and data communication systems via ISDN connections. The designation and title of each standard is given, together with a brief description of its contents.

2 CCITT Recommendations for the ISDN

The CCITT Recommendations for the ISDN listed here have mostly been published in the *Red Book* (Geneva: International Telecommunication Union, 1985). For some recommendations, new versions have been adopted in 1986 by means of the so-called "Accelerated procedure for provisional approval of recommendations". They are marked correspondingly in this Annex and in the reference lists of the individual sections of this book. The revised recommendations are published separately; they can be ordered from the ITU.

The *I. Series* Recommendations contain all those which define the ISDN from the viewpoint of the *user*.

The *Q.* and *G. Series* Recommendations cover specifications for the *network*.

The Q. Series Recommendations relate to digital exchanges, designed in the first instance for telephony but also suitable for the ISDN, and to signaling between user and local exchange as well as between exchanges.

Recommendations relating to transmission performance and to multiplexing and transmission equipment, network synchronization and other general network aspects are covered in the G. Series. The summary below only includes those G. Recommendations directly relating to the ISDN.

Recommendations on text terminals are covered in the *T. Series*.

The *X. Series* contains Recommendations for data communication including basic principles of protocol architecture (conforming to the ISO "Open Systems Interconnection" reference model), and for interworking facilities between the ISDN and dedicated data networks.

Some of the Q. and X. Recommendations dealing with aspects affecting the ISDN user also have I.-Series designations.

Figures A1 and A2 give an overview of the relation of the G., I. and Q. Recommendations to components of the ISDN.

The CCITT *Blue Book* (to be published in 1989) will contain the next issue of a complete set of recommendations; many will be revised, and new ones will be added.

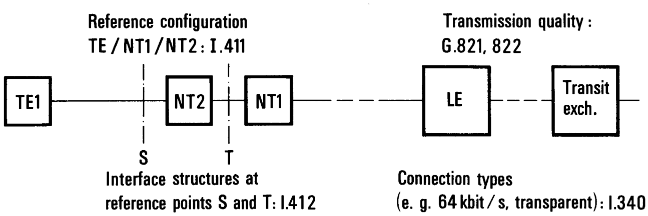


Fig. A1. Configuration of ISDN User Station and Network and Associated CCITT Recommendations with General Specifications.
TE1 ISDN terminal, NT network termination, LE local exchange (end office)

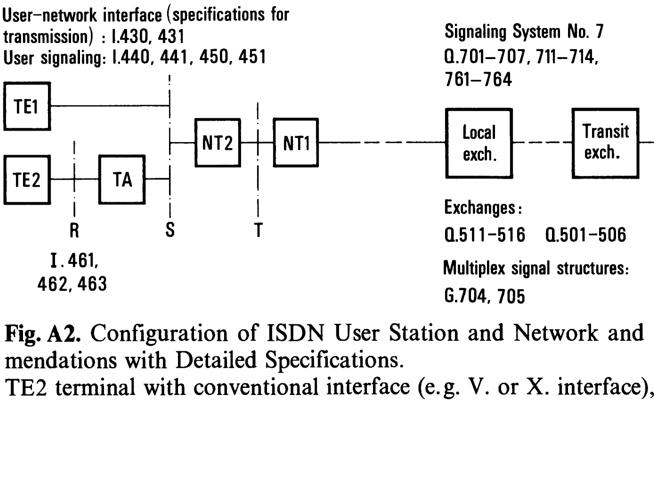


Fig. A2. Configuration of ISDN User Station and Network and Associated CCITT Recommendations with Detailed Specifications.
TE2 terminal with conventional interface (e.g. V. or X. interface), TA terminal adaptor

2.1 ISDN Concepts and Principles (Red Book, Vol. III.5)

2.1.1 General

I.110	General structure of the I-Series recommendations	Overview of the I. Series recommendations
I.111	Relationship with other recommendations relevant to ISDNs	References to recommendations e.g. of the G. Series
I.112	Vocabulary of terms for ISDNs	Definitions
I.130	Attributes for the characterization of telecommunications services supported by an ISDN and network capabilities of an ISDN	Characteristics of services and networks

2.1.2 Services

I.210	Principles of telecommunication services supported by an ISDN	Fundamentals of ISDN services
I.211	Bearer Services supported by an ISDN	Bearer services
I.212	Teleservices supported by an ISDN	Fully defined services (i.e. up to layer 7)

Note: In the CCITT Blue Book (1989), the content of the I.2.. recommendations will be supplemented and rearranged (with modified numbering); recommendations on Supplementary Services will be added.

2.1.3 Network Concept

I.310	ISDN – network functional principles	General information on the network concept
I.320	ISDN protocol reference model	Protocol reference model for ISDN based on the OSI model of the ISO
I.330	ISDN numbering and addressing principles	Fundamentals of subscriber numbering and addressing of terminals
I.331 (same as E.164)	Numbering plan for the ISDN era	Subscriber numbering in the ISDN (based on numbering in the telephone network)
I.340	ISDN connection types	Connection types (e.g. 64 kbits/s, transparent)

2.1.4 User-Network Interfaces

I.410	General aspects and principles relating to recommendations on ISDN user-network interfaces	General information on user-network interfaces
I.411	ISDN user-network interfaces – reference configurations	Arrangement of terminals, network terminations etc.
I.412	ISDN user-network interfaces – interface structures and access capabilities	Multiplex structures at user-network interfaces
I.420	Basic user-network interface	Reference to interface recommendations for basic access (B+B+D)
I.421	Primary rate user-network interface	Reference to interface recommendations for primary rate access (e.g. 30 × B+D)

I.430	Basic user-network interface – Layer 1 specification (Revised version, 1986)	Specifications for layer 1 (physical layer) for basic access
I.431	Primary rate user-network interface – Layer 1 specification	As I.430 but for primary rate access

2.1.5 Adaption of Lower User Bit Rates to 64 kbit/s, and Adaption of Existing Interfaces to the ISDN (cf. Sect. 2.9 of this Annex)

I.460	Multiplexing, rate adaption and support of existing interfaces	Adapting e.g. 8, 9.6, 16 kbit/s to 64 kbit/s
I.461 (same as X.30)	Support of X.21 and X.21bis based data terminal equipments (DTEs) by an Integrated Services Digital Network (ISDN)	Connection of terminals whose interface is designed for circuit-switched data networks
I.462 (same as X.31)	Support of packet mode temrinal equipment by an ISDN	Connection of terminals whose interface is designed for packet-switched data networks
I.463 (same as V.110)	Support of data terminal equipments (DTEs) with V-series type interfaces by an Integrated Services Digital Network (ISDN)	Connection of terminals whose interface is designed for modems (cf. the V. recommendations in Sect. 2.9 of this Annex)
I.464	Multiplexing, rate adaptation and support of existing interfaces for restricted 64 kbit/s transfer capability	Use of 56 kbit/s only (USA)

2.1.6 Signaling Between ISDN Subscriber Station and Local Exchange

I.440 (same as Q.920)	ISDN user-network interface, data link layer – General aspects (Revised version, 1986)	General information on layer 2 of the user access
I.441 (same as Q.921)	ISDN user-network interface, data link layer specification (Revised version, 1986)	Specifications for layer 2
I.450 (same as Q.930)	ISDN user-network interface, layer 3 – General aspects	General information on layer 3 of the user access
I.451 (same as Q.931)	ISDN user-network interface, layer 3 specification	Specifications for layer 3

2.2 Signalling System No. 7 for Interexchange Signalling

2.2.1 General (Red Book, Vol. VI.7)

Q.701 to Q.707	Signalling system No. 7 – Message transfer part (MTP)	Message transfer part
Q.710	Use of signalling system No. 7 for PABX applications	Adapting the message transfer part to I.451

2.2.2 Control Part for Signalling Transactions (Red Book, Vol. VI.7)

Q.711	Functional description of the signalling connection control part (SCCP) of signalling system No. 7	Supplementing the message transfer part by end-to-end transport connections
Q.712	Definitions and functions of signalling connection control part messages	Protocol elements
Q.713	Signalling Connection Control Part (SCCP) formats and codes	Format and coding

Q.714	Signalling connection control part procedures	

2.2.3 ISDN User Part (Red Book, Vol. VI.8)

Q.761	Functional description of the ISDN User Part of signalling system No. 7	Application-oriented signalling functions between ISDN exchanges
Q.762	General function of messages and signals	Protocol elements
Q.763	Formats and codes	
Q.764	Signalling procedures	
Q.766	Performance objectives in the integrated services digital network application	Deals with signalling dependability, delay etc.

2.2.4 Transaction capabilities application part (TCAP). The following recommendations are drafts; the final versions will be published in the CCITT Blue Book (1989).

Q.771	Functional description of transaction capabilities	Objectives and architecture of transaction capabilities (TC): Layer service primitives of the *transaction sublayer* (a transaction specifies an end-to-end relationship between two TC-users) and of the *component sublayer* (remote operations and associated replies) which resides on top of the transaction sublayer.
Q.772	Transaction capabilities message definitions	Functional description of the *transaction sublayer messages* (e.g. to begin or end a transaction and to convey a component sequence) and of the *component types* (used e.g. for invoking or returning the result of a specified remote operation).
Q.773	TCAP formats and coding	Format and encoding of transaction messages and of components.
Q.774	Signalling procedures for the transaction capabilities application part	Transaction control and component sublayer procedures for TC based on a connectionless network.

2.2.5 Operations and maintenance application part (OMAP; in Red Book, Vol. VI.8)

Q.795	Operations and maintenance application Part	

2.3 Digital Exchanges (Red Book, Vol. VI.5)

Recommendations for transit exchanges and (in parentheses) local and combined exchanges.

Q.501 (Q.511)	Introduction, field of application and basic functions	Overview, including application for ISDN
Q.502 (Q.512)	Interfaces	Interfaces to interexchange trunks and in Q.512 also to subscriber lines
Q.503 (Q.513)	Connections, signaling, control, call handling and ancillary functions	Conections through the exchange (especially for ISDN)
Q.504 (Q.514)	Performance and availability design objectives	Delay times in connection setup
Q.505 (Q.515)	Exchange measurements	Traffic measurements
Q.506 (Q.516)	Operation and maintenance functions	Operational functions (including alarms)

Note: In the CCITT Blue Book (1989) the content of the Q.5.. series will be rearranged and supplemented; there will be common recommendations for transit, local and combined exchanges. Numbers will probably be within the range Q.500 to Q.554, with Q.500 providing an overview

2.4 General Aspects of the Digital Network (Red Book, Vol. III.3)

2.4.1 Structure, Interfaces

G.701	Vocabulary of digital transmission and multiplexing, and pulse code modulation terms	Definitions
G.702	Digital hierarchy bit rates	Digital hierarchies, based on 1544 and 2048 kbit/s
G.703	Physical/electrical characteristics of hierarchical digital interfaces	Interfaces at 64 to 139264 kbit/s
G.704	Functional characteristics of interfaces associated with network nodes	Pulse frame structures
G.705	Characteristics required to terminate digital paths on a digital exchange	Digital connections to digital exchanges
G.711	Pulse code modulation of voice frequencies	Encoding per A- and μ-law

2.4.2 Transmission Performance in the Digital Network

G.811	Timing requirements at the outputs of reference clocks and network nodes suitable for plesiochronous operation of international digital links	Requirements relating to synchronization of a network
G.821	Error performance of an international digital connection forming part of an ISDN	Bit errors in an international 64 kbit/s ISDN connection
G.822	Controlled slip rate objectives on an international digital connection	Slips in an international 64 kbit/s ISDN connection
G.823	The control of jitter and wander within digital networks which are based on the 2048 kbit/s hierarchy	Phase jitter and wander at interfaces in the network

G.824	The control of jitter and wander within digital networks which are based on the 1544 kbit/s hierarchy	See G.823
G.921	Digital sections based on the 2048 kbit/s hierarchy	Performance in respect of bit errors, jitter etc.

2.5 Basic Recommendations for Telematic Services Applicable in Principle to the ISDN (Red Book, Vol. II.5)

F.161	International Group 4 facsimile service	Telefax via data networks or ISDN
F.200	Teletex service	Definition of the service
F.300	Videotex service	Definition of the service

2.6 Telematic Terminals (Red Book, Vol. VII.3)

2.6.1 Common Specifications for Different Types of Telematic Terminals

T.51	Coded character sets for Telematic services	Choice of characters for text services
T.62	Control procedures for Teletex and Group 4 facsimile services	Common control procedures (layer 5)
T.70	Network-independent basic transport service for the Telematic services	Specifications for layer 4
T.73	Document interchange protocol for the Telematic services	Specifications for layers 6 and 7

2.6.2 Specifications for Telefax, Teletex and Videotex Terminals (Red Book, Vol. VII.3)

T.0	Classification of facsimile apparatus for document transmission over the public networks	Definitions of fax groups 1 to 4
T.5	General aspects of Group 4 facsimile apparatus	Basic specifications for "fax group 4"
T.6	Facsimile coding schemes and coding control functions for Group 4 facsimile apparatus	Coding for "fax group 4" (layer 6)
T.60	Terminal equipment for use in the Teletex service	Basic characteristics of teletex terminals
T.61	Character repertoire and coded character sets for the international Teletex service	Character representation for teletex (layer 6)
T.72	Terminal capabilities for mixed mode of operation	Combined telefax and teletex
T.100	International information exchange for interactive Videotex	Recommendations T.100 and T.101 relate to basic compatibility requirements
T.101	International interworking for Vidotex service	

2.7 Specifications for Data Traffic within Bearer Services

2.7.1 Speed Classes and Service Attributes (Red Book, Vol. VIII.2)

X.1	International user classes of service in public data networks and integrated services digital networks (ISDNs)	Speed classes up to 64 kbit/s

X.2	International user services and facilities in public data networks	Options for service attributes

2.7.2 Data Transfer within the framework of Open Systems Interconnection (Red Book, Vol. VIII.5)

X.200	Reference model of Open Systems Interconnection for CCITT applications	Essentially the same as ISO 7498
X.210	Open Systems Interconnection (OSI) layer service definition conventions	Definition of the 7 layers
X.213	Network service definition for Open Systems Interconnection (OSI) for CCITT applications	Definitions of the "services" of layer 3
X.214	Transport service definition for Open Systems Interconnection (OSI) for CCITT applications	Definition of the "services" of layer 4
X.215	Session service definition for Open Systems Interconnection for CCITT applications	Definition of the "services" of layer 5
X.224	Transport protocol specification for Open Systems Interconnection for CCITT applications	Specifications for the layer 4 protocol
X.225	Session protocol specification for Open Systems Interconnection for CCITT applications	Specifications for the layer 5 protocol
X.250	Formal description techniques for data communications protocols and services	Description techniques

2.7.3 Message Handling Systems MHS ("Mailbox" Systems: Depositing, Retrieval and Format Conversion; Red Book, Vol. VIII.7)

X.400	System model – service elements	MHS functional model, MHS services, structure of the MHS protocols
X.401	Basic service elements and optional user facilities	Service attributes
X.408	Encoded information type conversion rules	Code and format conversion, e.g. teletex/videotex
X.409	Presentation transfer syntax and notation	Description language for layer 6
X.410	Remote operations and reliable transfer server	Session control (by means of layer 5) between different MHS systems
X.411	Message transfer layer	Transfer of messages between the user and the MHS, and between entities within the MHS
X.420	Interpersonal messaging user agent layer	Message exchange from person to person, including combined text and voice
X.430	Access protocol for Teletex terminals	Access from Teletex to MHS

2.8 Connection of Data Terminal Equipment Designed for Existing Networks (cf. Sect. 2.9 of this Annex) *to the ISDN* (Red Book, Vol. VIII.3 and VIII.4)

X.30 (same as I.461)	Support of X.21 and X.21bis based data terminals equipments (DTEs) by an Integrated Services Digital Network (ISDN)	Connection of terminals whose interface is designed for circuit-switched data networks
X.31 (same as I.462)	Support of packet mode terminal equipment by an ISDN	Connection of terminals whose interface is designed for packet-switched data networks
V.110 (same as I.463)	Support of data terminal equipments (DTEs) with V-series type interfaces by an Integrated Services Digital Network (ISDN)	Connection of terminals whose interface is designed for modems (cf. the V. recommendations in Sect. 2.9 of this Annex)

2.9 Interfaces of Data Terminal Equipment Conforming to Standards Other than those of ISDN (Red Book, Vol. VIII.1 for V. Recommendations and Vol. VIII.3/4 for X. Recommendations)

V.10 (same as X.26)	Electrical characteristics for unbalanced double-current interchange circuits for general use with integrated circuit equipment in the field of data communications	Unbalanced interfaces at up to 100 kbit/s (corresponds to EIA RS-423)
V.11 (same as X.27)	Electrical characteristics for balanced double-current interchange circuits for general use with integrated circuit equipment in the field of data communications	Balanced interfaces at up to 10 Mbit/s (corresponds to EIA RS-442)
V.28	Electrical characteristics for unbalanced double-current interchange circuits	For unbalanced interfaces at up to 20 kbit/s (RS-232-C)
V.24	List of definitions for interchange circuits between data terminal equipment and data circuit-terminating equipment	Basic functions of the interchange circuits (RS-232-C)
V.22 (V.22bis)	1200 bits per second (2400 bits per second) duplex modem (using the frequency division technique) standardized for use in the general switched telephone network and on point-to-point 2-wire leased telephone-type circuits	Recommendations V.22 to V.36 contain interface specifications for terminals that can be connected to ISDN terminal adaptors (TA) as per V.110; modems conforming to recommendations V.22 to V.32 can be connected to a t/r ("tip and ring") interface (i.e. voice-frequency interface of a TA).
V.23	600/1200-baud modem standardized for use in the general switched telephone network	
V.26	2400 bits per second modem standardized for use on 4-wire leased telephone-type circuits	
V.26bis	2400/1200 bits per second modem standardized for use in the general switched telephone network	
V.26ter	2400 bits per second duplex modem using the echo cancellation technique standardized for use on the general switched telephone network and on point-to-point 2-wire leased telephone-type circuits	

V.27	4800 bits per second modem with manual equalizer standardized for use on leased telephone-type circuits	
V.27bis	4800/2400 bits per second modem with automatic equalizer standardized for use on leased telephone-type circuits	
V.27ter	4800/2400 bits per second modem standardized for use in the general switched telephone network	
V.29	9600 bits per second modem standardized for use on point-to-point 4-wire leased telephone-type circuits	
V.32	A family of 2-wire, duplex modems operating at data signaling rates of up to 9600 bits per second for use on the general switched telephone network and on leased telephone-type circuits	
V.35	Data transmission at 48 kilobits per second using 60–108 kHz group band circuits	
V.36	Modems for synchronous data transmission using 60–108 kHz group band circuits	
X.21	Interface between data terminal equipment (DTE) and data circuit-terminating equipment (DCE) for synchronous operations on public data networks	Interface for connection to circuit-switched data networks
X.21bis	Use on public data networks of data terminal equipment (DTE) which is designed for interfacing to synchronous V-series modems	Interface for connection of terminal equipment with V. interface to circuit-switched data networks
X.25	Interface between data terminal equipment (DTE) and data circuit-terminating equipment (DCE) for terminals operating in the packet mode and connected to public data networks by dedicated circuit	Interface for connection to packet-switched data networks
X.32	Interface between data terminal equipment (DTE) and data circuit-terminating equipment (DCE) for terminals operating in the packet mode and accessing a packet switched public data network through a public switched telephone network or a circuit switched public data network. (Revised version, 1986)	Access of a terminal with X.25 interface to a packet network via a circuit-switched network
X.75	Terminal and transit call control procedures and data transfer system on international circuits between packet-switched data networks	Specification of protocols for interworking between packet-switched networks

2.10 Draft Recommendations on ISDN Interworking
(The finalized recommendations will be published in the Blue Book in 1988.)

I.500	General structure of the ISDN interworking recommendations	Organization, scope and objectives of interworking Recommendations
I.510	Definitions and general principles for ISDN interworking	Interworking principles for – network interworking (ISDN/non-ISDN), definition of reference configurations – ISDN–ISDN interworking (to support communication between non-identical ISDN services)
I.511	Layer 1 ISDN interworking	Specification of the layer 1 reference point Q as a common physical interface for logically different reference points K, M, N (see chapter 6) in terms of the relevant G.700–900 series Recommendations for digital networks, transmission systems and multiplexing equipment
I.515	Parameter exchange for ISDN interworking	Out-of-band or in-band parameter exchange in conjuncation with call and service negotiation to establish compatibility between end points, e.g. regarding rate adaption scheme, modem type selection, etc.
I.520	General arrangements for network interworking between ISDNs	– Identifies the recommendations applied to ISDN–ISDN interworking (at reference point N). – Interworking functions and call negotiation (bearer capability, connection type) between two ISDNs in the case where non-identical services are supported
I.530	Network interworking between an ISDN and a Public Switched Telephone Network (PSTN)	– Interworking configurations (interexchange and within an exchange) – ISDN bearer services and connection types suitable for ISDN–PSTN interworking – Functional requirements for ISDN–PSTN interworking – Handling of non-voice calls between ISDN and PSTN subscribers (TA-t/r, TA-V.)
I.540 (same as X.321)	General arrangements for interworking between Circuit Switched Public Data Networks (CSPDNs) and Integrated Services Digital Networks (ISDNs) for the provision of data transmission services	General description of interworking cases: circuit/packet switched bearer service requested on the ISDN

X.81	Interworking between an ISDN and a Circuit Switched Public Data Network (CSPDN)	Detailed interworking procedure, e.g. signaling conversion between CCS No. 7 ISUP and X.71
I.550 (same as X.325)	General arrangements for interworking between Packet Switched Public Data Networks (PSPDNs) and Integrated Services Digital Networks (ISDNs) for the provision of data transmission services	General description of interworking cases: – circuit-switched bearer service on the ISDN: interworking at the network layer (single step call set-up) or by port access (two step call set-up) – packet-switched bearer service on the ISDN.

3 ISO Standards for Data Communication Services Using Open Systems Interconnection (OSI)

The ISO International Standards (ISO . . .) are published by ISO. The standards listed below refer to "Information processing systems", with further titles as given for each standard.

ISO 7498	Open Systems Interconnection – Basic reference model	Cf. Sect. 2.7.2 of this Annex
ISO 3309	Data communication – High-level data link control procedures – Frame structure	ISO 3309, 4335 and 7809 contain specifications for data protection (layer 2) by HDLC
ISO 4335	Data communications – High-level data link control procedures – Consolidation of elements of procedures	
ISO 7809	Data communication – High-level data link control procedures – Consolidation of classes of procedures	
ISO 8348	Data communications – Network service definitions	Definition of the "services" of layer 3 (cf. CCITT Rec. X.213, item 2.7.2 of this Annex)
ISO 8072	Open systems interconnection – Transport service definition	Definition of the "services" of layer 4 (cf. CCITT Rec. X.214)
ISO 8073	Open systems interconnection – Transport protocol specification	Specifications for the layer 4 protocol (cf. CCITT Rec. X.224)
ISO 8326	Basic connection oriented session service definition	Definition of the "services" of layer 5 (cf. CCITT Rec. X.215)
ISO 8327	Basic connection oriented session protocol specification	Specifications for the layer 5 protocol (cf. CCITT Rec. X.225)

4 ISO Draft International Standards for Local Area Networks (LANs)

ISO/DIS 8802/2.2	Logical link control	
ISO/DIS 8802/3	Carrier sense multiple access with collision detection (CSMA/CD)	Access control for LAN according to "Ethernet" principle
ISO/DIS 8802/4	Token-passing bus access method and physical layer specification	Parts 4 and 5 of ISO/DIS 8802 deal with specification for "token

ISO/DIS 8802/5	Token ring access method and physical layer specification	passing" LANs (circulating bit pattern gives authorization to transmit)
ISO/DIS 8802/7	Slotted ring access method and physical layer specification	Using a busy/free indicator, fixed-length "minipackets" are carried in circulating timeslots
ISO/DIS 9314/1 . . 3	Fiber distributed data interface access method and physical layer specifications	Optical high-speed ring system with token passing access

5 ISO Standards for Interface Connectors

ISO 8877	Interface connector and contact assignments for ISDN basic access interface located at reference points S and T

References

Chapter 1

1.1 Armbrüster, H.: Communications Systems – The Current State of the Art and Trends. telcom rep. 7 (1984), pp. 146–151.
1.2 Bocker, P.: Datenübertragung, Bd. I: Grundlagen, 2. Aufl. Berlin, Heidelberg, New York, Tokyo: Springer 1983, pp. 40 et seq.
1.3 CCITT: Recommendation X.1: International user classes of service in public data networks and ISDNs. Red Book, Vol. VIII.2, Geneva: ITU 1985.
1.4 CCITT: Recommendation X.25: Interface between data terminal equipment (DTE) and data circuit-terminating equipment (DCE) for terminals operating in the packet mode and connected to public data networks by dedicated circuit. Red Book, Vol. VIII.3, Geneva: ITU 1985.
1.5 CCITT: Recommendation X.28: DTE/DCE interface for a start-stop mode data terminal equipment accessing the packet assembly/disassembly facility (PAD) in a public data network situated in the same country. Red Book, Vol. VIII.3, Geneva: ITU 1985.
1.6 CCITT: Recommendation T.60: Terminal equipment for use in the Teletex service. Red Book, Vol. VII.3, Geneva: ITU 1985.
1.7 CCITT: Recommendation F.200: Teletex service. Red Book, Vol. II.5, Geneva: ITU 1985.
1.8 Bocker, P.: Datenübertragung, Bd. II: Einrichtungen und Systeme, Berlin, Heidelberg, New York: Springer 1979, pp. 61 et seq.
1.9 Gerke, P.: Neue Kommunikationsnetze. Berlin, Heidelberg, New York: Springer 1982, pp. 32–33.
1.10 Rosenbrock, K.H.: ISDN – A Logical Evolution of the Digital Telephone Network. Special Issue of Jahrbuch der Deutschen Bundespost 1984, Bonn 1984, pp. 478 to 539. Obtainable from the Federal Ministry of Posts and Telecommunications, Section 201, P.O. Box 8001, D-5300 Bonn 1, F.R. Germany.
1.11 Händel, R.: ISDN getting broader – inclusion of wideband capabilities lies ahead. Online Internat. Conf. on Wideband Communications Today and Tomorrow, London: 1–2 October 1986.
1.12 Lutz, K.A.: Implementation of Broadband ISDN with the Siemens Switching System EWSD. IEEE Internat. Conf. on Communications, Toronto 1986. Conf. Rec. Vol. 2, pp. 766–770.
1.13 Huber, J.F.; v. Kienlin, A.: Requirements and Solutions for Future Packet Switching Networks. IEEE Internat. Conf. on Communications, Amsterdam, 1984. Conf. Rec. Vol. 3, pp. 1317–1321.
1.14 Hagen, R.: Teletex – a new text communication service and its impact on network modules. Nat. Telecommun. Conf. 1981, New Orleans. Conf. Rec. Vol. 3, pp. 5.3.1–5.3.5.
1.15 CCITT: Recommendation X.400: Message handling systems: System model – Service elements. Red Book, Vol. VIII.7, Geneva: ITU 1985.
1.16 Villiere, P.: A Framework for Network Solutions. Telephone Engineer & Managment, May 1, 1986, pp. 5–8.

Chapter 2

2.1 CCITT: Recommendation X.200: Reference model of Open Systems Interconnection for CCITT applications. Red Book, Vol. VIII.5, Geneva: ITU 1985.

2.2 CCITT: Recommendation I.430: Basic user-network interface – Layer 1 specification. Published separately by the ITU, Geneva, in 1987.
2.3 CCITT: Recommendation I.431: Primary rate user-network interface – Layer 1 specification. Red Book, Vol. III.5 Geneva: ITU 1985.
2.4 CCITT: Recommendation I.440: ISDN user-network interface, data link layer – General aspects. Published separately by the ITU, Geneva, in 1987.
2.5 CCITT: Recommendation I.441: ISDN user-network interface, data link layer specification. Published separately by the ITU, Geneva, in 1987.
2.6 CCITT: Recommendation I.450: ISDN user-network interface, layer 3 – General aspects. Red Book, Vol. III.5, Geneva: ITU 1985.
2.7 CCITT: Recommendation I.451: ISDN user-network interface, layer 3 specification. Red Book, Vol. III.5, Geneva: ITU 1985.
2.8 CCITT: Recommendation F.200: Teletex service. Red Book, Vol. II.5, Geneva: ITU 1985.
2.9 CCITT: Recommendation T.60: Terminal equipment for use in the Teletex service. Red Book, Vol. VII.3, Geneva: ITU 1985.
2.10 CCITT: Recommendation T.61: Character repertoire and coded character sets for the international Teletex service. Red Book, Vol. VII.3, Geneva: ITU 1985.
2.11 CCITT: Recommendation T.62: Control procedure for Teletex and Group 4 facsimile services. Red Book, Vol. VII.3, Geneva: ITU 1985.
2.12 CCITT: Recommendation T.70: Network-independent basic transport service for the Telematic services. Red Book, Vol. VII.3, Geneva: ITU 1985.
2.13 CCITT: Recommendation X.224: Transport protocol specification for Open Systems Interconnection for CCITT applications. Red Book, Vol. VIII.5, Geneva: ITU 1985.
2.14 CCITT: Recommendation X.225: Session protocol specification for Open System Interconnection for CCITT applications. Red Book, Vol. VIII.5, Geneva: ITU 1985.
2.15 CCITT: Recommendation F.161: International Group 4 facsimile service. Red Book, Vol. II.5, Geneva: ITU 1985.
2.16 CCITT: Recommendation T.5: General aspects of Group 4 facsimile apparatus. Red Book, Vol. VII.3, Geneva: ITU 1985.
2.17 CCITT: Recommendation T.6: Facsimile coding schemes and coding control functions for Group 4 facsimile apparatus. Red Book, Vol. VII.3, Geneva: ITU 1985.
2.18 CCITT: Recommendation T.73: Document interchange protocol for the Telematic services. Red Book, Vol. VII.3, Geneva: ITU 1985.
2.19 CCITT: Recommendation F.200-Annex C: Service requirements unique to the Teletex mixed-mode of operation. Red Book, Vol. II.5, Geneva: ITU 1985.
2.20 CCITT: Recommendation T.72: Terminal capabilities for mixed mode of operation. Red Book, Vol. VII.3, Geneva: ITU 1985.
2.21 CCITT: Recommendation F.300: Videotex service. Red Book, Vol. II.5, Geneva: ITU 1985.
2.22 CCITT: Recommendation T.100: International information exchange for interactive Videotex. Red Book, Vol. VII.3, Geneva: ITU 1985.
2.23 CCITT: Recommendation T.101: International interworking for Videotex services. Red Book, Vol. VII.3, Geneva: ITU 1985.
2.24 CEPT: Recommendation T/CD 06–01: Videotex presentation layer data syntax. Version of June, 1986 (Nice).
2.25 CCITT: Recommendation T.71: LAP B extended for half-duplex physical level facility. Red Book, Vol. VII.3, Geneva: ITU 1985.
2.26 CCITT: Recommendation X.21: Interface between data terminal equipment (DTE) and data circuit-terminating equipment (DCE) for synchronous operation on public data networks. Red Book, Vol. VIII.3, Geneva: ITU 1985.
2.27 CCITT: Recommendation X.21bis: Use on public data networks of data terminal equipment (DTE) which is designed for interfacing to synchronous V-series modems. Red Book, Vol. VIII.3, Geneva: ITU 1985.
2.28 CCITT: Recommendation X.25: Interface between data terminal equipment (DTE) and data circuit-terminating equipment (DCE) for terminals operating in the packet mode and connected to public data networks by dedicated circuit. Red Book, Vol. VIII.3, Geneva: ITU 1985.

2.29 CCITT: Recommendation X.75: Terminal and transit call control procedures and data transfer system on international circuits between packet-switched data networks. Red Book, Vol. VIII.4, Geneva: ITU 1985.
2.30 CCITT: V-Series Recommendations (Interchange circuits between data terminal equipment and data circuit-terminating equipment). Red Book, Vol. VIII.1, Geneva: ITU 1985.
2.31 CCITT: Rec. I.210: Principles of telecommunication services supported by an ISDN. Red Book, Vol. III.5, Geneva: ITU 1985.
2.32 CCITT: Recommendation I.211: Bearer Services supported by an ISDN. Red Book, Vol. III.5, Geneva: ITU 1985.
2.33 CCITT: Recommendation I.212: Teleservices supported by an ISDN. Red Book, Vol. III.5, Geneva: ITU 1985.
2.34 Rosenbrock, K.H.: ISDN – A Logical Evolution of the Digital Telephone Network. Special Issue of Jahrbuch der Deutschen Bundespost 1984, Bonn 1984, pp. 478 to 539. Obtainable from the Federal Ministry of Posts and Telecommunications, Section 201, P.O. Box 8001, D-5300 Bonn 1, F.R. Germany.
2.35 The Council of the European Communities: Council Recommendation on the Coordinated Introduction of the Integrated Services Digital Network (ISDN) in the European Community. EC COM(86) 205, Doc. 7308/86, Brussels, May 1986.
2.36 Rosenbrock, K.H.: Die Einführung des ISDN bei der Deutschen Bundespost. Proceedings of Online '86, 9th European Congress Fair for Technical Communications, Hamburg, Feb. 5 to 8, 1986, Congress 1, pp. 2D–1 to 15.
2.37 Implementing ISDN around the World. 8 contributions of several authors dealing with concepts and plans for the introduction of ISDN in Canada, France, Italy, United Kingdom, USA and F.R. Germany. Conference Record of IEEE International Conference on Communications '86 (Toronto 22–24 June 1986) Vol. 2, Session 33, pp. 33.1.1 to 33.8.4.
2.38 The Federal Minister of Posts and Telecommunications: ISDN – The Deutsche Bundespost's Response to the Telecommunication Requirements of Tomorrow. Bonn 1984. Obtainable from the Federal Ministry of Posts and Telecommunications, Section 247, P.O. Box 8001, D-5300 Bonn.
2.39 Arndt, G.; Rothamel, H.J.: Communication Services in the ISDN. telcom rep. 8 (1985) Special Issue "Integrated Services Digital Netwok ISDN", pp. 10 to 15.
2.40 Schenke, K.: Description of the Status Quo and Strategic Planning for the Future of Text Services. Special print, translated from Z. für das Post- und Fernmeldewesen, Aug. 1984. Obtainable from J. Keller Verlags-K.G., Postfach 1400, D-8130 Starnberg.
2.41 CCITT: X.400-Series Recommendations: Message handlings systems. Red Book, Vol. VIII.7, Geneva: ITU 1985.
2.42 Rothamel, H.J.: Videotex and the Integrated Services Digital Network (ISDN). Proceedings of IEEE International Conference on Communications, 1984, Vol. 2, pp. 836 to 840.
2.43 Rothamel, H.J.: ISDN Broadband Services and Applications. telcom rep. 9 (1986), No. 1, pp. 5 to 10.
2.44 Rothamel, H.J.: A Blueprint for Broadband ISDN services. Telephony, Nov. 17, 1986, pp. 50 to 58.

Chapter 3

3.1 Rey, R.F.: Engineering and Operations in the Bell System. AT & T Bell Laboratories, Murray Hill, N.J. Second Edition 1984.
3.2 Hilz, O.: Digitale Vermittlungstechnik im Fernsprechnetz der Deutschen Bundespost. Nachr. tech. Z. 25 (1982), pp. 626–628.
3.3 Bocker, P.: Datenübertragung, Bd. II: Einrichtungen und Systeme. Berlin, Heidelberg, New York: Springer 1979, pp. 25–28.
3.4 CCITT: Recommendation F.300: Videotex Service. Red Book, Vol. II.5, Geneva: ITU 1985.
3.5 CCITT: Recommendation F.160: General operational provisions for the international public facsimile service. Red Book, Vol. II.5, Geneva: ITU 1985.

3.6 CCITT: X.-Series Recommendations: Data Communication networks. Red Book, Vol. VIII.2, VIII.3 and VIII.4, Geneva: ITU 1985.

3.7 Schollmeier, G.: The user interface in the ISDN. telcom rep. 8 (1985), Special issue "Integrated Services Digital Network ISDN", pp. 22–26

3.8 CCITT: Recommendation E.500: Measurement and recording of traffic. Red Book, Vol. II.3, Geneva: ITU 1985.

3.9 Bear, D.: Principles of telecommunication-traffic engineering. Stevenage, UK, and New York: Peter Peregrinus Ltd. 1980, p. 113.

3.10 Daisenberger, G.; Reger, J.; Wegmann, G.: Traffic measurement and monitoring, an aid for planning and operating telephone exchanges and networks. telcom rep. 4 (1981) pp. 261–269.

3.11 Dietrich, G.: Telephone traffic model for common control investigations. 7th Internat. Teletraffic Congr. Stockholm 1973. Congr. Book, pp. 331/1–331/6.

3.12 Kraus, C.E.: Meeting the public's communications needs. Telecommun. J. 41 (1974), pp. 185–190.

3.13 Gimpelson, L.: Network transition strategies – analog to IDN to ISDN. Commun. internat. 1981 (June), pp. 43–46; (July), pp. 40–47.

3.14 Raab, G.: Private ISDN communications systems and their interoperation with the public ISDN. telcom rep. 8 (1985), Special Issue "Integrated Services Digital Network ISDN", pp. 57–63.

3.15 Fromm, I.: Local area networks – High-speed networks for office communications. telcom rep. 5 (1982), pp. 234–239.

3.16 CCITT: Economic and Technical Aspects of the Choice of Telephone Switching Systems. Handbook, Geneva: ITU 1981, pp. 54–55.

3.17 Rosenbrock, K.H.: Possible integration of telecommunication services in the digital telephone network of the Deutsche Bundespost – ISDN. telcom rep. 5 (1982) pp. 256–260.

3.18 Schramel, F.J.: Trends in digital switching and ISDN. Telecommun. J. 49 (1982), pp. 421–429.

3.19 Bocker, P.: The introduction of satellite routes into existing and future communication networks. Rev. FITCE 23 (1984) No. 2, pp. 47–52.

3.20 Schweizer, L.: Performance of terrestrial and satellite 64 kbit/s paths: requirements of voice and data, and standards of the future Integrated Services Digital Network (ISDN). IEEE Internat. Conf. on Commun., Boston 1983. Conf. Rec. Vol. 1, pp. 23–27.

3.21 Claus, J.: Internationale ISDN Aktivitäten im europäischen Ausland im Vergleich zu den Aktivitäten der Deutschen Bundespost. Online 86, 9th European Congress Fair for Technical Communications, 5.–8.2.1986.

3.22 Orbell, A.G.: British Telecoms plans for ISDN. IEEE Internat. Conf. on Commun., Amsterdam 1984. Conf. Rec. Vol. 2, pp. 576–579.

3.23 Kuwabara, M.: Japan is making INS a reality. telephony, Oct. 24, 1983, pp. 64–81.

3.24 Cheung, J.B.; Fitzgerald, T.J.; Gonsalves, A.A.; O'Reilly, G.P.: ISDN: Evolutionary step to integrated access and transport services. Record, AT & T Bell Laboratories, November 1985, pp. 4–13.

Chapter 4

4.1 CCITT: Recommendation I.410: General aspects and principles relating to recommendations on ISDN user-network interfaces. Red Book, Vol. III.5, Geneva: ITU 1985.

4.2 CCITT: Recommendation I.411: ISDN user-network interfaces – reference configurations. Red Book, Vol. III.5, Geneva: ITU 1985.

4.3 CCITT: Recommendation X.21: Interface between data terminal equipment (DTE) and data circuit-terminating equipment (DCE) for synchronous operation on public data networks. Red Book, Vol. VIII.3, Geneva: 1985.

4.4 CCITT: Recommendation X.25: Interface between data terminal equipment (DTE) and data circuit-terminating equipment (DCE) for terminals operating in the packet mode

and connected to public data networks by dedicated circuit. Red Book, Vol. VIII.3, Geneva: ITU 1985.
4.5 Seidel, H.: The ISDN basic access. telcom rep. 10 (1987), Special Issue "Multiplexing and line transmission", pp. 150–155.
4.6 CCITT: Recommendation I.461: Support of X.21 and X.21bis based data terminal equipments (DTEs) by an ISDN. Red Book, Vol. III.5, Geneva: ITU 1985. (Same as CCITT Rec. X.30. Red Book, Vol. VIII.3, Geneva: ITU 1985.)
4.7 Recommendation I.462: Support of packet mode terminal equipment by an ISDN. Red Book, Vol. III.5, Geneva: ITU 1985. (Same as CCITT Rec. X.31. Red Book, Vol. VIII.3, Geneva: ITU 1985.)
4.8 CCITT: Recommendation I.463: Support of data terminal equipments (DTEs) with V-series type interfaces by an ISDN. Red Book, Vol. III.5, Geneva: ITU 1985. (Same as CCITT Rec. V.110. Red Book, Vol. VIII.1, Geneva: ITU 1985.)
4.9 CCITT: Recommendation I.330: ISDN numbering and addressing principles. Red Book, Vol. III.5, Geneva: ITU 1985.
4.10 ISO Draft International Standards 8802/2, 3, 4, 5 and 7: Local Area Networks.
4.11 CCITT: Recommendation I.412: ISDN user-network interfaces – interface structures and access capabilities. Red Book, Vol. III.5, Geneva: ITU 1985.
4.12 CCITT: Recommendation Q.710: Use of Signalling System No. 7 for PABX application. Red Book, Vol. VI.7, Geneva: ITU 1985.
4.13 CCITT: Recommendation I.460: Multiplexing, rate adaption and support of existing interfaces. Red Book, Vol. III.5, Geneva: ITU 1985.
4.14 CCITT: Recommendation I.464: Multiplexing, rate adaptation and support of existing interfaces for restricted 64 kbit/s transfer capability. Red Book, Vol. III.5, Geneva: ITU 1985.
4.15 CCITT: Recommendation I.430: Basic user-network interface – Layer 1 specification. Published separately by the ITU, Geneva, in 1987.
4.16 Bocker, P.: Datenübertragung, Band I: Grundlagen, 2. Aufl. Berlin, Heidelberg, New York, Tokyo: Springer 1983, pp. 128 et seq.
4.17 CCITT: Recommendation I.431: Primary rate user-network interface – Layer 1 specification. Red Book, Vol. III.5, Geneva: ITU 1985.
4.18 CCITT: Recommendation G.703: Physical/electrical characteristics of hierarchical digital interfaces. Red Book, Vol. III.3, Geneva: ITU 1985.
4.19 CCITT: Recommendation G.704: Functional characteristics of interfaces associated with network nodes. Red Book, Vol. III.3, Geneva: ITU 1985.
4.20 CCITT: Recommendation G.735: Characteristics of primary PCM multiplex equipment operating at 2048 kbit/s and offering digital access at 384 kbit/s and/or synchronous digital access at 64 kbit/s. Red Book, Vol. III.3, Geneva: ITU 1985.
4.21 CCITT: Recommendation G.737: Characteristics of external access equipment operating at 2048 kbit/s offering digital acces at 384 kbit/s and/or synchronous digital access at 64 kbit/s. Red Book, Vol. III.3, Geneva: ITU 1985.
4.22 CCITT: Recommendation I.320: ISDN protocol reference model. Red Book, Vol. III.5, Geneva: ITU 1985.
4.23 CCITT: Recommendation I.420: Basic user-network interface. Red Book, Vol. III.5, Geneva: ITU 1985.
4.24 CCITT: Recommendation I.421: Primary rate user-network interface. Red Book, Vol. III.5, Geneva: ITU 1985.
4.25 CCITT: Recommendation X.200: Reference model of Open Systems Interconnection for CCITT applications. Red Book, Vol. VIII.5, Geneva: ITU 1985.
4.26 ISO International Standard 7498, 1984: Information processing systems – Open Systems Interconnection – Basic reference model.
4.27 CCITT: Recommendation I.440: ISDN user-network interface, data link layer – General aspects. Red Book, Vol. III.5, Geneva: ITU 1985. (Same as CCITT Rec. Q.920. Published separately by the ITU, Geneva, in 1987.)
4.28 CCITT: Recommendation I.441: ISDN user-network interface, data link layer specification. Red Book, Vol. III.5, Geneva: ITU 1985. (Same as CCITT Rec. Q.921. Published separately by the ITU, Geneva, in 1987.)

4.29 CCITT: Recommendation I.450: ISDN user-network interface, layer 3 – General aspects. Red Book, Vol. III.5, Geneva: ITU 1985. (Same as CCITT Rec. Q.930. Red Book, Vol. VI.9, Geneva: ITU 1985.)
4.30 CCITT: Recommendation I.451: ISDN user-network interface, layer 3 specification. Red Book, Vol. III.5, Geneva: ITU 1985. (Same as CCITT Rec. Q.931. Red Book, Vol. VI.9, Geneva: ITU 1985.)
4.31 CCITT: Recommendation I.331: Numbering plan for the ISDN era. Red Book, Vol. III.5, Geneva: ITU 1985. (Same as CCITT Rec. E.164, Red Book, Vol. II.2, Geneva: ITU 1985.)
4.32 CCITT: Recommendation E.163: Numbering plan for the international telephone service. Red Book, Vol. II.2, Geneva: ITU 1985.
4.33 ISO International Standard 3309, 1979: Data communication – High level data link control procedures – Frame structure.
4.34 ISO International Standard 4335, 1979: Data communication – High level data link control procedures – Consolidation of elements of procedures.
4.35 ISO International Standard 7809, 1984: Data communication – Consolidation of classes of procedures.
4.36 CCITT: Recommendation X.75: Terminal and transit call control procedures and data transfer system on international circuits between packet-switched data networks. Red Book, Vol. VIII.4, Geneva: ITU 1985.
4.37 CCITT: Recommendation T.70: Network-independent basic transport service for the Telematic services. Red Book, Vol. VII.3, Geneva: ITU 1985.
4.38 CCITT: Recommendation X.1: International user classes of service in public data networks and integrated services digital networks (ISDNs). Red Book, Vol. VIII.2, Geneva: ITU 1985.
4.39 CCITT: Recommendation V.5: Standardization of data signalling rates for synchronous data transmission in the general switched telephone network. Red Book, Vol. VIII.1, Geneva: ITU 1985.
4.40 CCITT: Recommendation X.121: International numbering plan for public data networks. Red Book, Vol. VIII.4, Geneva: ITU 1985.
4.41 CCITT: Recommendation V.24: List of definitions for interchange circuits between data terminal equipment and data circuit-terminating equipment. Red Book, Vol. VIII.1, Geneva: ITU 1985.
4.42 CCITT: Recommendation V.25: Automatic answering equipment and/or parallel automatic calling equipment on the general switched telephone network including procedures for disabling of echo control devices for both manually and automatically established calls. Red Book, Vol. VIII.1, Geneva: ITU 1985.
4.43 CCITT: Recommendation V.25 bis: Automatic calling and/or answering equipment on the general switched telephone network (GSTN) using the 100-series interchange circuits. Red Book, Vol. VIII.1, Geneva: ITU 1985.
4.44 v. Kienlin, A.; Klunker, J.: Packet switching and ISDN – a powerful alliance. Proc. ISS '87, Phoenix, March 15–20, 1987, pp. A.4.1.1–A.4.1.5.
4.45 Huber, J.F.; Laurens, J.; Ozdamar, M.; Valbonesi, G.: The packet interface between ISDN and public packet switched networks. Proc. ISS '87, Phoenix, March 15–20, 1987, pp. A.2.3.1–A.2.3.6.
4.46 CCITT: Recommendation X.2: International data transmission services and optional user facilities in public data networks. Red Book, Vol. VIII.2, Geneva: ITU 1985.
4.47 CCITT: Recommendation X.32: Interface between data terminal equipment (DTE) and data circuit-terminating equipment (DCE) for terminals operating in the packet-mode and accessing a packet switched public data network through a public switched telephone network or a circuit-switched public data network. Published separately by the ITU, Geneva, in 1986: ISBN 92–61–02911–6.
4.48 CCITT: Recommendation I.211: Bearer services supported by an ISDN. Red Book, Vol. III.5, Genf: ITU 1985.
4.49 Aldermeshian, H.; Chang, R.C.; Neigh, J.L.; Soloway, G.S.: Extending 1984 CCITT Recommendations to support advanced ISDN services. Proc. ISS '87, Phoenix, March 15–20, 1987, pp. A11.2.1–A11.2.7.

4.50 Kirton, P.; Ellershaw, J.; Littlewood, M.: Fast packet switching for integrated network evolution. Proc. ISS '87, Phoenix, March 15–20, 1987, pp. B6.2.1–B6.2.7.

4.51 Beckner, M.Wm.; Lee, T.T.; Minzer, S.E.: A protocol and a prototype for broadband subscriber access to ISDNs. Proc. ISS '87, Phoenix, March 15–20, 1987, pp. B6.3.1–B6.3.8.

4.52 White, P.E.; Holcomb, J.E.: Towards a next generation switching system. Proc. ISS '87, Phoenix, March 15–20, 1987, pp. C5.2.1–C5.2.7.

4.53 Schaffer, B.; Armbrüster, H.: Satisfying Tomorrow's Telecommunications Requirements by Upgrading ISDN to Broadband ISDN. 5th World Telecommunication Forum, Geneva 1987, Part 2, Vol. III, pp. 125–129

4.54 Thomas, A.; Coudreuse, J.P.; Servel, M.: Asynchronous time division techniques: an experimental packet network integrating video communication. Proc. ISS '87, Florence, May 1984.

4.55 CCITT: Draft Recommendation I.122: Framework for additional packet mode bearis services. To be published in the Blue Book. Geneva: ITU 1989.

4.56 CCITT: Draft Recommendation I.121: Broadband aspects of ISDN. To be published in the Blue Book. Geneva: ITU 1989.

Chapter 5

5.1 CCITT: Recommendation I.451: ISDN user-network interface, layer 3 specification. Red Book, Vol. III.5, Geneva: ITU 1985.

5.2 Hirschmann, P.; Wintzer, K.: Basic principles of digital subscriber sets. IEEE Trans. on Commun. 29 (1981), pp. 173–177.

5.3 Ebel, H.; Helmrich, H.: ISDN Terminals – the Basic Facts. telcom rep. 8 (1985) Special Issue "Integrated Services Digital Network ISDN", pp. 64–68.

5.4 Bähr, E.: Lautfernsprechanlagen. Der Ingenieur der Deutschen Bundespost 9 (1959), pp. 106–110.

5.5 Ohmann, F. (Editor): Kommunikations-Endgeräte. Berlin, Heidelberg, New York, Tokyo: Springer 1983, Chapters 5 and 6.

5.6 Kaufmann, P.; Saubert, L.: BITEL – Videotex Telephone T3210. telcom rep. 6 (1983), pp. 283–288.

5.7 Gilch, G.: BITEL – A New Communication Instrument. telcom rep. 6 (1983), pp. 288–291.

Chapter 6

6.1 CCITT: Recommendation G.711: Pulse code modulation (PCM) of voice frequencies. Red Book, Vol. III.3, Geneva: ITU 1985.

6.2 Osterburg, G.D.: The new generation of the service proven digital EDS switching system. telcom rep. 5 (1982) No. 2, pp. 91–96.

6.3 Mair, E.; Hausmann, H.; Naeßl, R.: EWSP – A high performance packet switching system. Proc. ICCC '86, Munich, Sept. 15–19, pp. 359–364.

6.4 telcom rep. 4 (1981) Special issue "EWSD Digital Switching System".

6.5 Raab, G.: Private ISDN communication systems and their interoperation with the public ISDN. telcom rep. 8 (1985), Special issue "Integrated Services Digital Network ISDN", pp. 57–63.

6.6 Mitterer, H.; Steigenberger, H.: EWSD – The ISDN switching system. telcom rep. 10 (1987) No. 5, pp. 235–240.

6.7 Mracek Mitchell, O.M.: Implementing ISDN in the United States. IEEE Journal on selected areas in communications, Vol. SAC–4 (1986), No. 3, pp. 398–406.

6.8 Arndt, G.; Rothamel, H.-J.: Communication services in the ISDN. telcom rep. 8 (1985), Special issue "Integrated Services Digital Network ISDN", pp. 10–15.

6.9 CCITT: Recommendation I.210: Principles of telecommunication services supported by an ISDN. Red Book, Vol. VIII.5, Geneva: ITU 1985.

6.10 CCITT: Recommendation I.310: ISDN – Network functional principles. Red Book, Vol. III.5, Geneva: ITU 1985.

6.11 CCITT: Recommendation I.340: ISDN connection types. Red Book, Vol. VIII.5, Geneva: ITU 1985.
6.12 Schlanger, G.G.: An overview of Signaling System No. 7. IEEE Journal on selected areas in communications, Vol. SAC–4, (1986), No. 3, pp. 360–365.
6.13 Walker, M.G.: Get inside CCITT signaling system No. 7. Telephony, March 10, 1986, pp. 72–77.
6.14 Punsch, H.: Aspects of CCS 7 network configurations. Telecommunications, October 1987, pp. 240–251.
6.15 Lampe, B.; Stoll, A.: Signaling between ISDN exchanges. telcom rep. 8 (1985), Special issue "Integrated Services Digital Network ISDN", pp. 37–41.
6.16 CCITT: Recommendation Q.721: Functional description of the signalling system telephone user part (TUP). Red Book, Vol. VI.8, Geneva: ITU 1985.
6.17 CCITT: Recommendation Q.722: General function of telephone messages and signals. Red Book, Vol. VI.8, Geneva: ITU 1985.
6.18 CCITT: Recommendation Q.723: Formats and codes. Red Book, Vol. VI.8, Geneva: ITU 1985.
6.19 CCITT: Recommendation Q.724: Signalling procedures. Red Book, Vol. VI.8, Geneva: ITU 1985.
6.20 CCITT: Recommendation Q.725: Signalling performance in the telephone application. Red Book, Vol. VI.8, Geneva: ITU 1985.
6.21 CCITT: Recommendation X.61: Signalling System No. 7 – Data user part. Red Book, Vol. VIII.4, Geneva: ITU 1985.
6.22 CCITT: Recommendation Q.761: Functional description of the ISDN user part of Signalling System No. 7. Red Book, Vol. VI.8, Geneva: ITU 1985.
6.23 CCITT: Recommendation Q.762: General function of messages and signals. Red Book, Vol. VI.8, Geneva: ITU 1985.
6.24 CCITT: Recommendation Q.763: Formats and codes. Red Book, Vol. VI.8, Geneva: ITU 1985.
6.25 CCITT: Recommendation Q.764: Signalling procedures. Red Book, Vol. VI.8, Geneva: ITU 1985.
6.26 CCITT: Recommendation Q.766: Performance objectives in the integrated services digital network application. Red Book, Vol. VI.8, Geneva: ITU 1985.
6.27 Händel, R.: Broadband Integrated Services Digital Networks. Telecommunications, April 1987, pp. 40–46.
6.28 Brüninghaus, K.; Kern, P.: Upgrading the EWSD Digital Switching System for Broadband ISDN. telcom rep. 9 (1986), pp. 120–126.
6.29 Armbrüster, H.; Schaffer, B.: Broadband ISDN: Realization aspects of the future telecommunications infrastructure. telcom rep. 10 (1987) No. 5, pp. 206–216.
6.30 Frantzen, V.: Value-added services in public telecommunication networks. Proc. of ICCC '86, Munich, Sept. 15–19, pp. 44–49.
6.31 Eigen, D.J.: Planning for the intelligent network – Siemens' view. 1987 National Communications Forum – NET 15, pp. 536–539.
6.32 Hass, R.J.; Humes, R.W.: Intelligent Network/2: A network architecture concept for the 1990s. Proc. ISS '87, Phoenix, March 15–20, 1987.
6.33 Bloom, P.; Miller, P.: Intelligent Network/2. Telecommunications, January 1987, pp. 66–75.
6.34 CCITT: Recommendation Q.512: Digital local and combined exchanges – Interfaces. Red Book, Vol. VI.5, Geneva: ITU 1985.
6.35 Schollmeyer, G.: The User Interface in the ISDN. telcom rep. 8 (1985), Special issue "Integrated Services Digital Network ISDN", pp. 22–27.
6.36 CCITT: Recommendation Q.502: Digital transit exchanges in integrated digital networks and mixed analogue digital networks – Interfaces. Red Book, Vol. VI.5, Geneva: ITU 1985.
6.37 Neufang, K.: The EWSD digital switching network. telcom rep. 4 (1981), Special issue "EWSD digital switching system", pp. 17–21.
6.38 CCITT: Recommendation Q.513: Digital local and combined exchanges – Connections, signalling, control, call handling and ancillary functions. Red Book, Vol. VI.5, Geneva: ITU 1985.

6.39 CCITT: Recommendation Q.701: Functional description of the signalling system (message transfer part). Red Book, Vol. VI.7, Geneva: ITU 1985.
6.40 CCITT: Recommendation Q.702: Signalling data link (Level 1). Red Book, Vol. VI.7, Geneva: ITU 1985.
6.41 CCITT: Recommendation Q.703: Signalling link (Level 2). Red Book, Vol. VI.7, Geneva: ITU 1985.
6.42 CCITT: Recommendation Q.704: Signalling network functions and messages (Level 3). Red Book, Vol. VI.7, Geneva: ITU 1985.
6.43 CCITT: Recommendation Q.705: Signalling network structure. Red Book, Vol. VI.7, Geneva: ITU 1985.
6.44 CCITT: Recommendation Q.706: Message transfer part signalling performance. Red Book, Vol. VI.7, Geneva: ITU 1985.
6.45 CCITT: Recommendation Q.707: Testing and maintenance. Red Book, Vol. VI.7, Geneva: ITU 1985.
6.46 CCITT: Recommendation Q.708: Numbering of international signalling point codes. Red Book, Vol. VI.7, Geneva: ITU 1985.
6.47 CCITT: Recommendation Q.516: Digital local and combined exchanges – Operations and maintenance functions. Red Book, Vol. VI.5, Geneva: ITU 1985.
6.48 CCITT: Recommendation Q.506: Digital transit exchanges in integrated digital networks and mixed analogue digital networks – Operations and maintenance functions. Red Book, Vol. IV.5, Geneva: ITU 1985.
6.49 Ribbeck, G.: Operation and maintenance of the EWSD system. telcom rep. 4 (1981), Special issue "EWSD digital switching system", pp. 45–50.
6.50 CCITT: Recommendation Q.795: Operations and maintenance application part (OMAP). Red Book, Vol. VI.8, Geneva: ITU 1985.
6.51 Wenzel, G.: CCITT common channel signaling system No. 7 in the EWSD system. telcom rep. 4 (1981), Special issue "EWSD digital switching system", pp. 35–39.
6.52 CCITT: Recommendations Q.120–Q.139: Specifications of Signalling System No. 4. Red Book, Vol. VI.2, Geneva: ITU 1985.
6.53 CCITT: Recommendations Q.140–Q.164: Specifications of Signalling System No. 5. Red Book, Vol. VI.2, Geneva: ITU 1985.
6.54 CCITT: Recommendation Q.711: Functional description of the signalling connection control part of Signalling System No. 7. Red Book, Vol. VI.7, Geneva: ITU 1985.
6.55 CCITT: Recommendation Q.712: Definition and functions of signalling connection control part messages. Red Book, Vol. VI.7, Geneva: ITU 1985.
6.56 CCITT: Recommendation Q.713: Signalling connection control part format and codes. Red Book, Vol. VI.7, Geneva: ITU 1985.
6.57 CCITT: Recommendation Q.714: Signalling connection control part procedures. Red Book, Vol. VI.7, Geneva: ITU 1985.
6.58 CCITT: Recommendation Q.771: Functional description of transaction capabilities. To be published in the Blue Book. Geneva: ITU 1989.
6.59 CCITT: Recommendation Q.772: Transaction capabilities message definitions. To be published in the Blue Book. Geneva: ITU 1989.
6.60 CCITT: Recommendation Q.773: TCAP formats and encoding. To be published in the Blue Book. Geneva: ITU 1989.
6.61 CCITT: Recommendation Q.774: Signalling procedures for the transaction capabilities application part. To be published in the Blue Book. Geneva: ITU 1989.
6.62 Schromm, H.: ISDN solutions to office problems. Communications International, October 1987, Siemens sponsored supplement, pp. 32–33.
6.63 (Anon.) Comprehension: Cornet protocol. COM – Siemens Magazine of Computers and Communication XV (1987) No. 6, p. 37
6.64 Le Minh, T.; Cannon, S.: ISDN-Centrex: The EWSD approach. Proc. of Globecom '86, pp. 19.5.1–19.5.5.
6.65 Pasternak, E.J.; Schulman, S.A.: Customer control of Centrex service. Proc. ISS '87, Phoenix, March 15–20, 1887, pp. C.2.3.1–C.2.3.6.

Chapter 7

7.1 CCITT: Recommendation I.120: Integrated Services Digital Networks (ISDNs). Red Book, Vol. III.5, Geneva: ITU 1985.
7.2 Schweizer, L.: Planning aspects of quantizing distortion in telephone networks. Telecommun. J. 48 (1981), pp. 32–36.
7.3 CCITT: Recommendation G.701: Vocabulary of digital transmission and multiplexing, and pulse code modulation (PCM) terms. Red Book, Vol. III.3, Geneva: ITU 1985.
7.4 CCITT: Recommendation G.711: Pulse code modulation (PCM) of voice frequencies. Red Book, Vol. III.3, Geneva: ITU 1985.
7.5 CCITT: Recommendation G.113: Transmission impairments. Red Book, Vol. III.1, Geneva: ITU 1985.
7.6 CCITT: Recommendation G.722: 7 kHz audio-encoding within 64 kbit/s. To be published in the Blue Book. Geneva: ITU 1989.
7.7 CCITT: Recommendation G.702: Digital hierarchy bit rates. Red Book, Vol. III.3, Geneva: ITU 1985.
7.8 CCITT: Recommendation G.703: Physical/electrical characteristics of hierarchical digital interfaces. Red Book, Vol. III.3, Geneva: ITU 1985.
7.9 CCITT: Recommendation G.733: Characteristics of primary PCM multiplex equipment operating at 1544 kbit/s. Red Book, Vol. III.3, Geneva: ITU 1985.
7.10 CCITT: Recommendation G.732: Characteristics of primary multiplex equipment operating at 2048 kbit/s, and Recommendation G.735: Characteristics of primary multiplex equipment operating at 2048 kbit/s and offering digital access at 384 kbit/s and/or synchronous digital access at 64 kbit/s. Red Book, Vol. III.3, Geneva: ITU 1985.
7.11a Drügh, P.; Senft, R.: 2 Mbit/s digital transmission over local and regional cables. telcom rep., Vol. 2 (1979), Special Issue "Digital Transmission", pp. 85–89.
7.11b Schmidt, V.; v. Winnicki, K.: Digital transmission on balanced copper pairs. telcom rep., Vol. 10 (1987) Special Issue "Multiplexing and line transmission", pp. 137–143.
7.12 ITU Publication: Optical fibres for telecommunications. Geneva: ITU 1984. ISBN 92–61–01841–6.
7.13 telcom rep. Vol. 6 (1983), Special Issue "Optical Communications". 220 pages.
7.14 CCIR: Report 338–5: Propagation data and prediction methods required for line-of-sight radio-relay systems. Recommendations and Reports of the CCIR, Vol. V. Geneva: ITU 1986.
7.15 Schweizer, L.: Performance of terrestrial and satellite 64 kbit/s paths: requirements of voice and data, and standards of the future Integrated Services Digital Network (ISDN). IEEE Internat. Conf. on Commun., Boston, 1983. Conf. Rec. Vol. 1, pp. 23–27.
7.16 CCITT: Recommendation G.921: Digital sections based on the 2048 kbit/s hierarchy. Red Book, Vol. III.3, Geneva: ITU 1985.
7.17 Franaszek, P.A.: Sequential-state coding for digital transmission. Bell Syst. Techn. J. 47 (1968), pp. 143–157.
7.18 Schollmeier, G.: The user interface in the ISDN. telcom rep., Vol. 8 (1985) Special Issue "Integrated Services Digital Network ISDN", pp. 22–27.
7.19 Bocker, P.: Datenübertragung, Band I: Grundlagen, 2. Aufl. Berlin, Heidelberg, New York, Tokyo: Springer 1983, pp. 219–235.
7.20 CCITT: Recommendation G.165: Echo cancellers. Red Book, Vol. III.1, Geneva: ITU 1985.
7.21 Steinkamp, J.: Radio relay systems for modern communication networks – an introduction. telcom rep., Vol. 7 (1984), pp. 252–264.
7.22 CCITT: Recommendation G.704: Functional characteristics of interfaces associated with networks nodes. Red Book, Vol. III.3, Geneva: ITU 1985.
7.23 CCITT: Recommendation G.742: Second order digital multiplex equipment operating at 8448 kbit/s and using positive justification, and Recommendation G.751: Digital multiplex equipments operating at the third order bit rate of 34368 kbit/s and the fourth order bit rate of 139264 kbit/s and using positive justification. Red Book, Vol. III.3, Geneva: ITU 1985.
7.24 CCITT: Recommendation G.743: Second order digital multiplex equipment operating at 6312 kbit/s and using positive justification, and Recommendation G.752: Characteristics of

digital multiplex equipments based on a second order bit rate of 6312 kbit/s and using positive justification. Red Book, Vol. III.3, Geneva: ITU 1985.
7.25 CCITT: Recommendation G.811: Timing requirements at the outputs of reference clocks and network nodes suitable for plesiochronous operation of international digital links. Red Book, Vol. III.3, Geneva: ITU 1985.
7.26 Bocker, P.: Datenübertragung, Band II: Einrichtungen und Systeme. Berlin, Heidelberg, New York: Springer 1979, pp. 191–197.
7.27 CCITT: Recommendation G.821: Error performance of an international connection forming part of an ISDN. Red Book, Vol. III.3, Geneva: ITU 1985.
7.28 CCITT: Recommendation X.50: Fundamental parameters of a multiplexing scheme for the international interface between synchronous data networks, and Recommendation X.51: Fundamental parameters of a multiplexing scheme for the international interface between synchronous data networks using 10-bit envelope structure. Red Book, Vol. VIII.4, Geneva: ITU 1985.
7.29 CCITT: Recommendation X.22: Multiplex DTE/DCE interface for user classes 3–6. Red Book, Vol. VIII.3, Geneva: ITU 1985.
7.30 CCITT: Recommendation G.822: Controlled slip rate objectives on an international digital connection. Red Book, Vol. III.3. Geneva: ITU 1985.
7.31 CCITT: Recommendation G.114: Mean one-way propagation time. Red Book, Vol. III.1, Geneva: ITU 1985.
7.32 CCITT: Recommendation G.823: The control of jitter and wander within digital networks which are based on the 2048 kbit/s hierarchy, and Recommendation G.824: The control of jitter and wander within digital networks which are based on the 1544 kbit/s hierarchy. Red Book, Vol. III.3, Geneva: ITU 1985.
7.33 CCITT: Recommendation O.171: Specification for instrumentation to measure timing jitter on digital equipment. Red Book, Vol. IV.4, Geneva: ITU 1985.
7.34 CCITT: Recommendation I.340: ISDN connection types. Red Book, Vol. III.5, Geneva: ITU 1985.

Chapter 8

8.1 ISDN in the Office – HICOM. Special issue of telcom rep. and Siemens magazine COM, Dec. 1985.
8.2 Kraus, C.E.: Meeting the Public Communications Needs. Telecommun. J. 41 (1974) No. 3, pp. 185–190.
8.3 Information Technology Advisory Program. 1986 Summer Conference, June 24/25, Munich: Proceedings. Gartner Group, Inc., Stamford, CT 06902, USA.
8.4 CCITT: Recommendation T.101: International interworking for Videotex services. Red Book, Vol. VII.3, Geneva: ITU 1985.
8.5 CEPT: Recommendation T/CD 06–01: Videotex presentation layer data syntax. Version of June, 1986 (Nice).
8.6 Kammerlander, K.: C900 – An advanced mobile radio telephone system with optimum frequency utilization. IEEE Trans. VT-33, No. 3 (1984), pp. 205–213.

Subject Index

N. **Kuroyanagi** (Ed.)

Globecom '87

Tokyo Conference Proceedings

IEEE/IEICE Global Telecommunications Conference 1987

1988. Approx. 2200 pages. In cooperation with Omsha Ltd., Tokyo
ISBN 3-540-19510-6

Contents:
Volume 1: ISDN Field Trials: Planning and Initial Results. Motion Video Coding. ISDN Interworking. Security for Future Network Communications. Toward More Flexibility and Intelligent Voice/Data Networks. Asian and Pacific Lightwave and Radio Systems. Radio System for Subscribers. Performance of Distributed Systems. Theoretical Topics in Signal Processing. Modulation and Coding for Satellite Communications. Terminals for the Handicapped. Knowledge-Based Technologies for Communication Systems. Satellite Transmission Systems: Present and Future. High Quality Image Service and Compression Technology. New Generation Digital Signal Processors and Applications. Broadband Network Access Interface. Traffic Engineering for ISDN Design and Planning. Multiple Access Protocols. Spread Spectrum Communications. Specification and Construction Technologies for Telecommunications Software. Coherent Optical Communication Technology.
Volume 2: Multimedia Terminal Systems. Advanced Technologies and Application of Speech Coding. Portable and Mobile Radio Communications. Progress Toward Multi-Gigabit/Second Lightwave Transmission Systems. International and Nationwide Network Operations and Management. Advanced On-board Technologies. Modulation and Coding I. Optical Communication Theory. Systems Engineering Methods in Quality Assurance. Domestic and International Network Planning and Tools. Digital Terminals with Enhanced Capabilities. Reliability Issues in Communications Networks. Multimedia Terminal Applications. Advanced Anti-Multipath Fading Techniques. Telecommunications Operations and Management Networks. Undersea Lightwave Transmission. Impact of Integrated Optical Devices on Fiber-Optic Systems. Modulation and Synchronization. Adaptive Signal Processing in Communication. Efficient Image Coding. Topics in Data Communications.
Volume 3: Optical Transport Systems Architecture and Technology for Broadband ISDN. High Capacity Digital Microwave Radio Systems. Re-use Technologies for Communication Software. New Telecommunication Services and Their Control. Evolving CCITT No. 7 Signalling Network Services. Advances in CSMA Protocols. Field Performance and Support Experience for Packet Switching Services. Mobile Satellite Communication Systems. Modulation and Coding II. VLSI Technologies for ISDN Subscriber Network. Advances in Packet Switching. Expert Systems for Network Operations and Maintenance. New Technologies Toward Packetized Multimedia Communications. Indoor Wireless Communication. Echo Cancellar and Adaptive Filtering in Communication. Broadband Services and Systems. Operations Strategies for Transport Network.

Springer-Verlag
Berlin Heidelberg New York
London Paris Tokyo

Springer