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TELESERVICES TO BE PROVIDED BY AN INTEGRATED SERVICES DIGITAL NETWORK (ISDN)

Recommendation proposed by Working Group T/GT 7 "Services and Facilities" (SF)
Amendments proposed by Project Team Service Descriptions for the ISDN (SDI)

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1. GENERAL

This Recommendation details the teleservices to be provided by an ISDN.
Teleservices are described by a definition and a set of attribute values which fully describe the service from a user's point of view.

2. LIST OF TELESERVICES

The following is an initial list of teleservices that may be available in the ISDN:

	Teleservice	Priority
1.	Telephony 3.1 kHz	E
2.	Telephony 7 kHz	A
3.	Teletex	E
4.	Telefax Group 4	E
5.	Mixed Mode	E
6.	Videotex	E
7.	Telex	A
8.	Videotelephony	A
9.	Videoconferencing	A
10.	Surveillance/Teleaction	A
11.	Picture Mail	A
12.	Film Retrieval	A
13.	Audio Retrieval	A
14.	Message Handling Services	A
15.	Audiographic Teleconferencing	A

Other teleservices than those listed above will be possible in the future.

3. DEFINITIONS OF TELESERVICES

Full descriptions of the teleservices to be provided by an ISDN are shown in Annexes 1 to 8. Definitions are detailed below.

Note: Definitions for the remaining teleservices are not yet available.

3.1. Telephony 3.1 kHz

The Telephony 3.1 kHz teleservice provides users with the ability for real time two-way speech conversation via the public ISDN. User information is provided over a B-channel and signalling is provided over a D-channel.

3.2. Telephony 7 kHz

The Telephony 7 kHz teleservice provides users with the ability for real time two-way high quality speech conversation via the network.

3.3. Teletex

Teletex is an international service enabling subscribers to exchange office correspondence in the form of documents containing teletex coded information on an automatic memory-to-memory basis via the ISDN.

3.4. Telefax Group 4

Telefax Group 4 is an international service enabling subscribers to exchange office correspondence in the form of documents containing facsimile coded information automatically via the ISDN.

3.5. Mixed Mode

The Mixed Mode service provides combined text and facsimile communication for end-to-end transfer of documents containing mixed information of text and fixed images. User information transfer is provided via a B-channel and signalling via the D-channel.

3.6. Videotex

The Videotex teleservice provides through appropriate access the possibility to communicate via telecommunication networks for retrieval and mailbox functions for multimedia information (text, graphics, photographic pictures, sound etc.).

3.7. Telex

The Telex service provides interactive text communication. The digital signal at the S/T reference point follows the internationally agreed Recommendations for Telex above the ISDN physical layer. User information is transferred over circuit or packet-mode bearer channels and signalling is provided over the D-channel.

3.8. Videotelephony

The Videotelephony service is an audiovisual conversational teleservice providing bidirectional symmetric real-time transfer of voice and moving colour pictures between two locations (person-to-person) via the network involved. The minimum requirement is that under normal conditions the picture information transmitted is sufficient for the adequate re-presentation of fluid movements of a person displayed in head and shoulders view.

Annex 1

TELEPHONY 3.1 KHZ

1. DEFINITION

The Telephony 3.1 Khz teleservice provides users with the ability for real time two-way speech conversation via the ISDN network. User information is provided over a B-channel and signalling is provided over a D-channel.

2. DESCRIPTION

2.1. General Description

The Telephony 3.1 Khz teleservice provides speech transmission at an audio bandwidth or 3.1 Khz. The communication is bidirectional with both directions continuously and simultaneously active during the speech phase. The network may use processing techniques appropriate for speech such as analogue transmission, echo cancellation, and low bit rate encoding.

The digital signal at the S/T reference point follows the encoding laws for speech (according to CCITT Recommendation G.711), A-law or μ-law. The network may use digital signal processing techniques. It may also be necessary to use echo cancellation techniques, in particular when interworking with other networks such as the PSTN.

Tones and announcements are provided by the network, encoded according to CCITT Recommendation G.711, although terminals can generate tones or other indications based on the messages received.

2.2. Specific Terminology

Voice Quality:
The required acoustic performance is described in terms of loudness ratings, frequency response, quantising distortion, etc.; overall requirements are given in the P-Series of CCITT Recommendations.

Transmission Delay:
The maximum delay is that specified for the general telephone network (cf. CCITT Recommendation G.114).

Retention Time:
This time specifies the amount of time during which the network retains the call information of the original call upon encountering busy or being released. This time is a network operator

option. The value for this time is greater than 15 seconds.

3. PROCEDURES

3.1. Provision and Withdrawal

Provision of this service will be by arrangement with the network operator.

This Teleservice is offered with several subscription options which apply separately to each ISDN number or groups of ISDN numbers on the interface. For each subscription option, *only one value can be selected. Subscription option for the interface are summarised below:*

Subscription Option	**Value**
Maximum number of information channels available at user B	m, where m is not greater than the number of information channels on the interface
Maximum number of total calls present at user B	n, where n is not greater than the number of information channels on the interface

User B can be an ISDN number or a group of ISDN numbers on the interface.

Note: More than one ISDN number can be associated with the service/interface only as part of a supplementary service such as the Multiple Subscriber Number. In the case of one ISDN number, the option given above for the number of calls can only exceed the number of information channels in association with a supplementary service (e.g. Call Waiting). As a network operator option, separate values may be specified for incoming and for outgoing calls for either or both of the limits.

3.2. Normal Procedures

3.2.1. *Originating the Service (Call Setup)*

The service is originated by the originating user activating the terminal, performing service selection, if applicable for the originating terminal and terminating customer selection. During this process the originating user is given the appropriate indications as to state of the call.

1) A service selection is required on a multi-service terminal.

2) Terminating customer selection is selecting the required termination (user/network interface) by an appropriate means (for example the user of DDI, multiple subscriber number).

3) Indications during call origination may include an indication that the network is ready to receive the network address information (proceed indication) and an indication that the call is progressing through the network. It shall be possible to have audible indications which may be accompanied by other indications.

3.2.2. *Call Acceptance (Answer)*

Selection of the terminating customer is indicated to each user by appropriate indications (call arrival indications and awaiting answer indication). The acceptance of the call by the terminating user (answer) causes the indications to be removed and bidirectional communication paths to be provided. The call is now termed in the speech phase.

3.2.3. *Call Release*

A request to terminate the service may be generated by either user. If one user terminates the service the other user is given an appropriate indication as to the state of the call, provided the call has entered the speech phase.

3.3. User Requirements in case of Unsuccessful Outcome

3.3.1. *Failure Situations due to User Error*

The following failure situations may occur due to user error:

1) User taking too long to input the network address information will be given a failure indication, e.g. during overlap sending (see CCITT Recommendation I.451).

2) User inputting a non-valid network address, e.g. an unallocated address, will be given a failure indication.

3.3.2. *Failure Situations due to Terminating State*

1) User attempting to set up a call to a termination where no free B-channels are available will receive a busy indication unless Call Waiting or another supplementary service is in operation (Note).

Note: In support of some supplementary services (e.g. Call Waiting and Line Hunting) it may optionally be necessary for the subscriber to register some additional parameters (e.g. destination number used to distinguish PSTN telephony calls) with the network to allow the network to know when a channel i busy with telephony.

2) User attempting to set up a call to a termination where the call is not accepted, i.e. no response indicating call acceptance is received, will after a defined period be given a call failure indication (see CCITT Recommendation I.451).

3.3.3. *Failure Situations due to Network Conditions*

User attempting to set up a call but meeting problems in the network (e.g. congestion) will be given a suitable indication.

4. NETWORK CAPABILITIES FOR CHARGING

It shall be possible to charge the subscriber accurately for the service.

5. INTERWORKING REQUIREMENTS

Interworking is required between ISDNs, including private ISDNs, and PSTN.

6. INTERACTION WITH SUPPLEMENTARY SERVICES

Not applicable. Each supplementary service description identifies the applicability to this teleservice.

7. ATTRIBUTES AND VALUES

a) Low Layer Attributes

Information transfer attributes:

1.	Information transfer mode	Circuit
2.	Information transfer rate	64 kbit/s
3.	Information transfer capability	Speech
4.	Structure	8 Khz integrity
5.	Establishment of communication	On Demand
6.	Communication configuration	Point-to-point
7.	Symmetry	Bidirectional Symmetric

Access attributes:

8.	Access channel (and rate)	B(64) for user information, D for signalling (Note)
9.1.	Signalling access protocol layer 1	CCITT Recs I.430/I.431
9.2.	Information access protocol layer 1	CCITT Recs I.430/I.431; G.711
9.3.	Signalling access protocol layer 2	CCITT Recs I.440/I.441
9.4.	Information access protocol layer 2	-
9.5.	Signalling access protocol layer 3	CCITT Recs I.450/I.451
9.6.	Information access protocol layer 3	-

Note: For reserved permanent service the operational, administrative, and maintenance messages related to these services may be conveyed over the D-channel.

b) High Layer Attributes

10.	Type of user information	Speech
11.	Layer 4 protocol functions	-
12.	Layer 5 protocol functions	-
13.	Layer 6 protocol functions	CCITT Rec. G.711
14.	Layer 7 protocol functions	-

c) General Attributes

15.	Supplementary services	See Recs. T/CAC S 10.5 to 10.7 and CCITT Rec. T.250
16.	Quality of service	
17.	Interworking possibilities	See Rec. T/CAC S 10.4
18.	Operational and commercial aspects	See CEPT: User Handbook, ISDN

Note: - = Not Applicable.

8. DYNAMIC DESCRIPTION

The circuit-mode dynamic description appears in CCITT Recommendation I.220

Annex 2

TELEPHONY 7 KHZ

1. DEFINITION

The Telephony 7 kHz teleservice provides users with the ability for real time two-way high quality speech conversation via the network.

2. DESCRIPTION

The Telephone 7 kHz service provides, via the network, the ability to transfer high quality speech allowing users to communicate by interchanging sounds with a high quality. The communication is bidirectional with both directions continuously and simultaneously active during the speech phase.

This service provides high quality speech communication with a frequency range of 50 to 7000 Hz. The digital signal at the S/T reference point follows the internationally agreed encoding laws for high quality speech (CCITT Recommendation G.722). User information is provided over a B-channel, signalling is provided over the D-channel. Tones and announcements are provided by the network. Terminals can, however, generate tones or other indications based on the messages received.

3. OPERATIONAL REQUIREMENTS

3.1. Normal Procedures

3.1.1. *Originating the Service (Call Setup)*

The service is originated by the originating user activating the terminal, performing service selection, if applicable for the originating terminal, and terminating customer selection. During the call setup stage the originating user is given the appropriate indications as to the state of the call.

1) Service selection is required on multi-service terminals and may consist of directly selecting the Telephony 7 kHz teleservice.

2) Terminating customer selection is selecting the required termination (user/network interface) by means of the appropriate network number. In association with Telephony 7 kHz a specific terminal on that interface may be selected by the use of e.g. Direct Dialling In or Sub-Addressing.

3) Indications during call origination may include an indication that the network is ready to receive the network address information (proceed indication) and an indication that the call is progressing through the network. It shall be audible indications but may also be accompanied by other indications.

3.1.2. *Call Acceptance (Answer)*

Selection of the terminating customer is indicated to each user by appropriate indications (call arrival indication and awaiting answer indication). The acceptance of the call by the terminating user (answer) causes the indications to be removed and bidirectional communication paths to be provided. The call is now termed in the speech phase.

3.1.3. *Terminating the Service (Call Release)*

A request to terminate the service may be generated by either user. If one user terminates the service the other user is given an appropriate indication as to the state of the call, provided the call has entered the speech phase.

3.2. User Requirements in case of Unsuccessful Outcome

3.2.1. *Failure Situations due to User Error*

The following failure situations may occur due user error:

1) User taking too long to input the network number information will be given a failure indication where overlap sending applies (see CCITT Recommendation I.451).

2) User inputting a non-valid network number, e.g. an unallocated number, will be given a failure indication.

3.2.2. *Failure Situations due to Terminating Termination State*

1) User attempting to set up a call to termination where no free B-channel is available will receive a busy indication unless Call Waiting or another supplementary service is in operation.

2) User attempting to set up a call to termination where the call is not accepted, i.e. no response indication call acceptance is received will after a defined period be given a call failure indication (see CCITT Recommendation I.451).

3.2.3. *Failure Situations due to Network Conditions*

User attempting to set up a call meeting problems in the network (e.g. congestion) will be given a suitable indication.

4. CHARGING REQUIREMENTS

It shall be possible to charge for the service on a per call basis, based on duration of the call, the numbers of the called and calling parties, and the time of day. Other parameters for use in charging determination may also be available.

5. INTERCOMMUNICATION CONSIDERATIONS

When communication is required with either ISDN telephony terminal incapable of 7 kHz working, or to and from the PSTN, the following applies.

5.1. Non 7 kHz ISDN Terminals

As the calling customer will not necessarily know the capability of the called customer's terminals, it shall be possible for a calling customer to set up a call from his telephony 7 kHz terminal by requesting Telephony 7 kHz. In such situations, where the incoming call is answered by a non 7 kHz telephone, this service will automatically default to the telephony mode and provide communication accordingly.

5.2. Interworking with the PSTN

Where a telephony 7 kHz call setup is attempted to a customer on the PSTN, the calling customer will be given an appropriate indication that interworking with the PSTN has occurred, but the network will continue call setup. If the calling customer does not abandon this call attempt, then this service will automatically default to the telephony mode and provide communication accordingly

6. INTERACTION WITH SUPPLEMENTARY SERVICES

Each supplementary service description identifies the applicability to this teleservice.

If the in-band communication is interrupted by the network as a result of one user invoking a supplementary service (e.g. the Call Hold supplementary service or the Terminal Portability supplementary service) then the network shall provide an appropriate indication in the B-channel.

7. ATTRIBUTES AND VALUES

Attributes	**Values**
a) **Low Layer Attributes**	
Information transfer attributes:	
1. Information transfer mode	Circuit
2. Information transfer rate	64 kbit/s
3. Information transfer capability	Unrestricted digital information
4. Structure	8 kHz integrity
5. Establishment of communication	On demand
6. Communication configuration	Point-to-point
7. Symmetry	Bidirectional symmetric
Access attributes:	
8. Access channel (and rate)	B(64) for user information and for signalling

9. Signalling access protocol

b) **High Layer Attributes**

10. Type of user information	7 kHz Speech
11. Layer 4 protocol functions	-
12. Layer 5 protocol functions	-
13. Layer 6 protocol functions	CCITT Rec. G.722
14. Layer 7 protocol functions	-

c) **General Attributes**

15. Supplementary low layer and high attributes (supplementary services	See CEPT Recs T/CAC S 10.5 to 10.7 and CCITT Rec. I.250
16. Quality of service	
17. Interworking possibilities	See Rec. T/CAC S 10.4
18. Operational and commercial aspects	See CEPT: User Handbook, ISDN

Note: - = Not applicable.

Annex 3

TELETEX

1. DEFINITION

Teletex is an international service enabling subscribers to exchange office correspondence in the form of documents containing teletex coded information on automatic memory-to-memory basis via the ISDN.

2. DESCRIPTION

The Teletex service provides communication between teletex equipments which are used for the preparation, editing, and printing of correspondence containing text information using a standardised character set (Recommendation T.61).

The basic element of correspondence between people using the service is the page as the smallest unit of text treated as an entity. No restrictions shall exist as far as the operator procedures for generation of the text or the positioning of text within the printable area on a page are concerned.

Note 1: This does not necessarily imply that the characters used to construct a graphical symbol are transmitted in the same sequence as that in which they are keyed.

Note 2: This does not necessarily imply that the order in which text on a page is transmitted is the same as that in which it was keyed.

Note 3: An exception to this rule is the application of the processable mode of operation for which the page as a basic element of correspondence cannot be used. The processable mode of operation within the Teletex service is defined in CCITT Recommendation F.220.

The Teletex service in each country and the international interconnection between countries of networks shall use automatic switching so that it is possible for any telex subscriber to reach any other teletex subscriber using fully automatic selection.

It is a requirement to allow the through-connection of a call between a teletex terminal connected to a private automatic branch exchange (or similar systems) and those connected to public exchanges used for the Teletex service.

A virtual dialogue mode of operation, which appears to the subscriber as a conversational mode, should be possible although this is not a basic requirement of the Teletex service.

A virtual dialogue of operation, which appears to the subscriber as a conversational mode, may become possible as a new standardised option within the Teletex service both allowing communications between persons and data base access (refer to CCITT Recommendation I.210).

Processable mode of operation, as a standardised option within the Teletex service, allows the transfer of text containing information for further editing and processing by the recipient (refer to CCITT Recommendation F.220).

Mixed mode of operation using the techniques of Telefax 4 for the transfer of facsimile coded information and of Teletex for transfer of character coded text is described as a standardised option within the Teletex service in CCITT Recommendation F.220.

Two-Way Alternate (TWA) communication is a capability of the Teletex service, which also include One-Way Communication (OWC); the calling subscriber will have full control of the teletex call.

3. PROCEDURES

3.1. Provision and Withdrawal

The national and international facilities of the Teletex service, including the Teletex/Telex conversion facilities, shall be open continuously.

Teletex subscriber equipments for which call numbers are published in the directories shall, in principle, be available to accept calls continuously.

In order to facilitate the 24 hours duration of the service it is permitted to use a centralised storage in the network to realise receiving memory capability of the terminal.

3.2. Call Phases

The operation for each call may be divided into the following three phases:

a) Preparation
   - Preparation of the information in local mode;
   - Loading of the information into a memory.

b) Transmission (in principle, automatic)
   - Call establishment;
   - Pre-information phase (see Note);
   - Information transfer from memory to memory (see Note);
   - Post-information phase (see Note);
   - Call clearing.

Note: During these parts of the transmission phase, the network must be transparent with respect to control procedures.

c) Output
   - Emptying the memory.

Note: The information may consist of one or more teletex documents, each consisting of one or more teletex pages.

The control procedures as specified in CCITT Recommendation T.62 shall be used as the end-to-end communication procedures between any teletex equipment in the basic service.

The lower layer protocols and the network-independent basic transport protocol to be used for Teletex are specified in CCITT Recommendations T.70 and T.90.

The network-dependent control procedures for the Teletex are those that are defined for ISDN.

3.3. Call Identification Line

The teletex procedures include the exchange of reference information prior to sending any document. This reference information includes identification of the parties in the call as well as date and time. Also supplementary reference information is exchanged during a call to allow reference to an individual document or page for error recovery or other purposes.

This reference information, taken together, is defined to be printable on a single line called the call identification line. The use of this information is a local decision except in recovering from an interrupted transmission. The call identification line is composed of the following four fields:

Field 1: Identification of the called Teletex equipment;
Field 2: Identification of the calling Teletex equipment;
Field 3: Date and time;
Field 4: Supplementary reference information.

Field 1	Field 2	Field 3	Field 4
Identification of the called teletex equipment	Identification of the calling teletex equipment	Date and time	Supplementary reference information
24 characters	24 characters	14 characters	7 characters
72 characters			

Figure 1/CCITT I.241.2. Format of the Call Identification Line

Field 1 (identification of the called equipment) contains the identification of the called equipment. It is originated in the control procedures by the called terminal.

Field 2 (identification of the calling equipment) contains the identification of the calling equipment. It is originated in the control procedures by the calling terminal.

Field 3 (date and time) contains the date and time reference information showing the year, month, day, hour and minute in the fixed format of 14 characters, thus YY-MM-DD-HH-MM. This field is originated in the control procedures by the calling equipment which obtains this information from the network. The time represents the local time at the calling equipment and is intended to present the time of call origination.

Field 4 (supplementary reference information) contains a document reference number, a hyphen (coding 2/13) as a separator and a page reference number as defined in CCITT Recommendation T.62. This field has a fixed length of seven character positions and is originated in the control procedures by the teletex equipment sending the associated documents.

3.4. Error Protection

Within the Teletex service a high layer error detection and correction is provided in the session layer for all those errors which are not corrected by the network layers.

To ensure call integrity, error protection will be provided by teletex control procedures (see CCITT Recommendations T.62, T.70, and T.90). The error rate on the pre-information,

information, and post-information phases should not exceed 1 in 10-6 characters.

4. NETWORK CAPABILITIES FOR CHARGING

It shall be possible to charge the subscriber accurately for the service.

5. INTERCOMMUNICATION REQUIREMENTS

5.1. Interworking between Networks

Within the Teletex service interworking between terminals connected to different networks is required. Real time connection between terminals operating at different speeds has to be provided on the basis of at least 2.4 kbit/s.

5.2. Intercommunication with Other Services

The Teletex service will provide the ability to intercommunicate in both directions with the Telex service by means of conversion facilities (refer to CCITT Recommendations F.201, U.201, and T.300).

Intercommunication between basic mode and mixed mode telex terminals and Classes I, II, and III Group 4 facsimile terminals is shown in Table 1 (I.241.2) (refer to CCITT Recommendation F.184).

The Teletex service allow the intercommunication with Interpersonal Massaging Service (IPM) (refer to Recommendation F.422).

6. INTERACTION WITH SUPPLEMENTARY SERVICES

For the ISDN, the international supplementary services which may be used for Teletex in the circuit-mode using a B-channel:

1) Closed User Group
2) Multiple Subscriber Number
3) User-to-User Signalling
4) Calling Line Identification Presentation
5) Calling Line Identification Restriction
6) Called Line Identification Presentation
7) Direct Dialling In

To / From	Facsimile Group 4 Class I	Facsimile Group 4 Class II	Facsimile Group 4 Class III	Teletex basic mode	Teletex mixed mode	Teletex processable mode 1
Facsimile Group 4 Class I	F	F	F			
Facsimile Group 4 Class II	F	F	F			
Facsimile Group 4 Class III	F	T, F, MM	T, F, MM	T	T, F, MM	T
Teletex basic mode		T	T	T	T	T
Teletex mixed mode		T, F, MM	T, F, MM	T	T, F, MM	T
Teletex processable mode 1		T	T	T	T	T, PM1

T: Basic Teletex document with character coded information only.
F: Group 4 Facsimile document with coded information only.
MM: Mixed-mode document with character and facsimile coded information.
PM1: Processable mode document with character coded information only.

Table 1 (I.241.2) Current status of direct intercommunication for Teletex and Group 4 facsimile terminals on the same network.

7. ATTRIBUTES AND VALUES

a) Lower Layer Attributes

Information transfer attributes:

	Circuit-mode Bearer Capability	Packet-mode Bearer Capability
1. Mode	Circuit	Packet
2. Rate	64 kbit/s	Maximum throughput of a given virtual circuit is less than or equal to the maximum bit rate of the user information access channel and the throughput class of virtual circuit.
3. Info transfer cap.	Unrestricted	Unrestricted
4. Structure	Unstructured (See note)	Service data circuit integrity
5. Establishment	Demand	Demand (VC), permanent (PVC)
6. Configuration	Point-to-point	Point-to-point
7. Symmetry	Bidirectional symmetric	Bidirectional symmetric

Access attributes:

	Circuit-mode Bearer Capability	Packet-mode Bearer Capability
8. Access channel	B-channel (B for user info, D for signalling)	User information over virtual circuit within B- or D-channel. When D-channel is used, maximum packet size and quality of service may be restricted. Signalling may be provided via D- and/or virtual circuit within B-channel.
9. Signalling access protocols		

Note: Even if no structure is required, the network may provide 8 kHz integrity.

b) Higher Layer Attributes

10. Type of user info	Teletex
11. Layer 4 protocol	CCITT Rec. T.70
12. Layer 5 protocol	CCITT Rec. T.62
13. Layer 6 protocol	CCITT Rec. T.61
14. Layer 7 protocol	CCITT Rec. T.60

c) General Attributes

15. Supplementary attributes	See sub-clause 6/CCITT Rec. I.241.2
16. Quality of service	
17. Interworking possibilities	See Rec. T/CAC S 10.4
18. Operational and commercial aspects	See CEPT: User Handbook, ISDN

SLP = Single Link Protocol
PLP = Packet Layer Protocol

8. DYNAMIC DESCRIPTION

The circuit-mode dynamic description appears in CCITT Recommendation I.220.

Annex 4

TELEFAX GROUP 4

1. DEFINITION

Telefax Group 4 is an international service enabling subscribers to exchange office correspondence in the form of documents containing facsimile coded information automatically via the ISDN.

2. DESCRIPTION

The Telefax Group 4 service provides a basic level of compatibility between all terminals participating in the service. It offers bidirectional communication between two users via the ISDN using 64 kbit/s digital signals over the B-channel.

There are three classes of Telefax Group 4 terminals:

Class I Minimum requirement terminal is a terminal able to send and receive documents containing facsimile encoded information (in accordance with CCITT Recommendations T.6 and T.400 Series).

Class II Minimum requirement terminal is a terminal able to transmit documents that are facsimile encoded (in accordance with CCITT Recommendations T.6 and T.400 Series). In addition, the terminal must be capable of receiving documents which are facsimile coded (in accordance with CCITT Recommendations T.6 and T.400 Series), Teletex coded (in accordance with the basic coded character repertoire as defined in Recommendation T.61), and also mixed mode documents (in accordance with CCITT Recommendation of the T.400 Series).

Class III Minimum requirement terminal is a terminal that is capable of generating, transmitting, and receiving facsimile coded documents (in accordance with CCITT Recommendations T.6 and T.400 Series), Teletex coded documents (in accordance with the basic coded character as defined in CCITT Recommendation T.61) and mixed mode documents (in accordance with Recommendations of the T.400 Series).

The basic element of the correspondence between people using the service is the page as the smallest unit of text treated as an entity. No restrictions shall exist so far as the operator procedures for generation of the text or the positioning of text within the reproducible area on a page are concerned.

3. PROCEDURES

3.1. Provision and Withdrawal

The national and international Telefax Group 4 service shall be open continuously.

Telefax Group 4 terminals for which call numbers are published in the directories shall, in principle, be available to accept calls continuously.

In order to facilitate the 24 hour duration of the service it is permitted to use a centralised storage in the network to realise receiving memory capability of the terminal.

3.2. General

The Telefax Group 4 service in each country and interconnection between countries or networks shall use automatic switching so that it is possible for a Telefax Group 4 subscriber to reach any other Telefax Group 4 subscriber using fully automatic selection.

It is a requirement to allow the through-connection of a call between Telefax Group 4 terminals connected to a private automatic branch exchange (or similar systems) and those connected to public exchanges used for the Telefax Group 4 service.

Two-Way Alternate (TWA) communication is a capability of the Telefax Group 4 service, which also includes One-Way Communication (OWC); the calling subscriber will have full control of the Telefax Group 4 call.

3.3. Call Phases

The operations for each call may be divided into the following three phases:

a) Preparation:
Preparation of the information to be transmitted.

b) Transmission
- Call establishment (automatic);
- Pre-information phase (see Note);
- Information transfer (see Note);
- Post-information phase (see Note);
- Call clearing.

Note: During these parts of the transmission phase, the network must be transparent with respect to control procedures.

c) Output
Displaying the message either by immediate printing or from a storage medium upon control by the operator.

Note: The information may consist of one or more Telefax Group 4 documents, each consisting of one or more Telefax Group 4 pages.

The control procedures as specified in CCITT RecommendationS of the T.400 Series and T.62 shall be used as the end-to-end communication procedures between terminals in the service.

The low layer protocols and the network-independent basic transport protocol for Telefax Group 4 are specified in CCITT Recommendations T.70 and T.90.

The network-dependent control procedures for the Telefax Group 4 are defined for ISDN.

3.4. Call Identification

The Telefax Group 4 procedures include the exchange of reference information prior to sending any document. This reference information include identification of parties to the call as well as date and time. Also supplementary reference information is exchanged during a call to allow reference to an individual document or page for error recovery or other purposes. Date and time have to be provided by the network and sent to the calling terminal in the call setup phase.

This reference information, taken together, is defined to be printable on a single line called the call identification line. Use of this information is a local decision except in recovering from interrupted transmission.

For the format of the Call Identification Line: see CCITT Recommendation F.200.

3.5. Error Protection

To ensure call integrity, error protection will be provided by Telefax Group 4 control procedures (see CCITT Recommendations T.62, T.70, and T.90). Besides the error detection and correction mechanism in the layer 2 (and 3) an additional error detection and correction mechanism is provided in the session layer. By this mechanism errors of the higher layer functions (e.g. command/response sequence error) and transmission errors, which are not corrected by the lower layers, will be corrected by e.g. retransmission of one or several pages.

The error rate on the pre-information, information, and post-information phases should not exceed $1x10^{-6}$.

4. NETWORK CAPABILITIES FOR CHARGING

It shall be possible to charge the subscriber accurately for the service.

5. INTERCOMMUNICATION REQUIREMENTS

5.1. Interworking between Networks

Within the Telefax Group 4 service interworking between terminals connected to different networks is required.

a) Telefax Group 4 (ISDN) - Telefax Group 4 (CSPDN)
b) Telefax Group 4 (ISDN) - Telefax Group 4 (PSPDN)
c) Telefax Group 4 (ISDN) - Telefax Group 4 (PSTN)

In the case of international interworking between Telefax Group 4 terminals connected to dissimilar networks, CCITT Recommendation X.300 shall apply. For international interworking between PSTN and ISDN, a (separate) Telefax Group 4 interworking unit may be necessary.

International routes between ISDNs for the Telefax Group 4 service shall be capable of supporting user data rates of up to 64 kbit/s.

5.2. Intercommunication with Other Services

Intercommunication between basic mode and mixed mode Teletex terminals and Classes I, II and III Telefax Group 4 terminals connected to the Telefax Group 4 service is shown in Table 1 (CCITT Rec. I.241.3).

To From	Facsimile Group 4 Class I	Facsimile Group 4 Class II	Facsimile Group 4 Class III	Teletex basic mode	Teletex mixed mode	Teletex processable mode 1
Facsimile Group 4 Class I	F	F	F			
Facsimile Group 4 Class II	F	F	F			
Facsimile Group 4 Class III	F	T, F, MM	T, F, MM	T	T, F, MM	T
Teletex basic mode		T	T	T	T	T
Teletex mixed mode		T, F, MM	T, F, MM	T	T, F, MM	T
Teletex processable mode 1		T	T	T	T	T, PM1

T: Basic Teletex document with character coded information only.
E: Telefax Group 4 document with coded information only.
MM: Mixed-mode document with character and facsimile coded information.
PM1: Processable mode document with character coded information only.

Table 1: Current status of direct intercommunication for Teletex and Telefax Group 4 terminals on the same network.

In both the Teletex and Telefax Group 4 services the equipment providing mixed mode should enable a direct exchange of documents in accordance with CCITT Recommendations T.6, T.61, and T.400 Series.

Intercommunication is desirable between terminals of the Telefax Group 4 service and terminals of services other than Telefax Group 4 provided over ISDN and other public switched networks.

Intercommunication possibilities between Telefax Group 4 terminals and Telefax Group 3 terminals have to be provided (see also Recommendation F.180).

a) Telefax Group 4 (ISDN) - Telefax Group 3 (PSTN)
b) Telefax Group 4 (ISDN) - Telefax Group 3 (ISDN, via terminal adapters)

In the case a) Telefax Group 4 terminals use specific service features in ISDN. Intercommunication should be supported by ISDN-PSTN interworking units.

In case b) Telefax Group 3 terminals and Telefax Group 4 terminals which are to be connected in the PSTN can also be connected to the ISDN via terminal adapters.

6. INTERACTION WITH SUPPLEMENTARY SERVICES

International supplementary services for Telefax Group 4 service in the circuit-mode on a B-channel:

1) Closed User Group
2) Multiple Subscriber Number
3) User-to-User Signalling
4) Calling Line Identification Presentation
5) Calling Line Identification Restriction
6) Called Line Identification Presentation
7) Direct Dialling In

7. ATTRIBUTES AND VALUES

a) Lower Layer Attributes

Information transfer attributes:

	Circuit-mode Bearer Capability	Packet-mode Bearer Capability
1. Mode	Circuit	Packet
2. Rate	64 kbit/s 2	Maximum throughput of a given virtual circuit is less than or equal to the maximum bit rate of the user information access channel and the throughput class of virtual circuit.
3. Info transfer cap.	Unrestricted	Unrestricted
4. Structure	Unstructured (See note)	Service data circuit integrity
5. Establishment	Demand	Demand (VC), permanent (PVC)
6. Configuration	Point-to-point	Point-to-point
7. Symmetry	Bidirectional symmetric	Bidirectional symmetric

Access attributes:

	Circuit-mode bearer capability	Packet-mode bearer capability

8. Access channel	B-channel (B for user info, D for signalling)	User information over virtual circuit within B- or D-channel. When D-channel is used, maximum packet size and quality of service may be restricted. Signalling may be provided via D- and/or virtual circuit within B-channel.
9. Signalling access protocol		

Note: Even if no structure is required, the network may provide 8 kHz.

b) Higher Layer Attributes

10. Type of user info	Telefax 4
11. Layer 4 protocol	CCITT Rec. T.70
12. Layer 5 protocol	CCITT Rec. T.62
13. Layer 6 protocol	CCITT Rec. T.400 (Note 4)
13.1 Resolution [ppi]	200x200 standard; 240x240, 300x300, 400x400 optional
14. Layer 7 protocol	T.5-C1.1

c) General Attributes

15. Supplementary attributes	See subclause 6/CCITT Rec. I.241.2
16. Quality of service	
17. Interworking possibilities	See Rec. T/CAC S 10.4
18. Operational and commercial aspects	See CEPT: User Handbook, ISDN

SLP = Single Link Protocol
PLP = Packet Layer Protocol

8. DYNAMIC DESCRIPTION

The circuit-mode dynamic description appears in CCITT Recommendation I.220.

Annex 5

MIXED MODE

1. DEFINITION

This Mixed Mode service provides combined text and facsimile communication for end-to-end transfer of documents containing mixed information of text and fixed images. User information transfer is provided via a B-channel and signalling via the D-channel.

Annex 6

VIDEOTEX

1. DEFINITION

The Videotex teleservice provides through appropriate access the possibility to communicate via telecommunication networks for retrieval and mailbox functions for multimedia information (text, graphics, photographic pictures, sound e.c.t.).

2. DESCRIPTION

2.1. General

The Videotex service is an interactive service which provides facilities, through appropriate access by standardised procedures, for users of Videotex terminals to communicate with data bases via telecommunication networks.

The Videotex service on ISDN is an enhancement of the Videotex service on other networks, e.g. PSTN.

The Videotex service may include some of the following characteristics:

1. Information is multimedia and may include text, graphics, photographic pictures, sound etc.;
2. Information is stored in data bases;
3. Information is transmitted between the data base and the users by telecommunication networks;
4. Displayable information is presented on a suitably modified television receiver or other visual display devices;
5. Access is under the user's direct or indirect control;
6. The service is easily operated by the general public as well as specialist users, i.e. the service is user friendly;
7. The service enables the user to create and modify information in the data bases;
8. The service provides data base management facilities which allow application providers to create, maintain and manage data bases, and to manage closed user groups facilities.

2.2. Videotex Service Profile

The set of functionalities required by a Videotex service. It includes the service, application, and presentation functionalities.

2.3. Videotex Application

Part of a Videotex service which is under the responsibility of only one application provider. The Videotex service provider may also act as an application provider.

3. ATTRIBUTES/VALUES

The following European Telecommunication Standards (ETS) as defined by the European Telecommunications Standard Institute (ETSI) are relevant in the field of Videotex on ISDN:

- ETS 300 072 : Terminal Equipment (TE); Videotex presentation layer protocol; Videotex presentation layer data syntax
- ETS 300 073 : Videotex presentation layer protocol; Geometric Display
- ETS 300 074 : Videotex presentation layer data syntax transparent data
- ETS 300 075 : Terminal Equipment (TE); Videotex processable data
- ETS 300 076 : Terminal Equipment (TE); Videotex: Terminal Facility Identifier (TFI)
- ETS 300 079 : Integrated Services Digital Network (ISDN); Syntax-based Videotex end to end protocols Circuit mode DTE - DTE
- ETS 300 080 : Integrated Services Digital Network (ISDN); Lower layer protocols for telematic terminals
- ETS 300 149 : Terminal Equipment (TE); Videotex: Audio syntax
- ETS 300 177 : Terminal Equipment (TE); Videotex: Photographic syntax
- ETS 300 218 : Integrated Services Digital Network (ISDN); Syntax-based Videotex lower layer protocols for ISDN packet mode (X.31 case A and case B)
- ETS 300 222 : Terminal Equipment (TE); Framework of Videotex terminal protocols
- ETS 300 223 : Integrated Services Digital Network (ISDN); Syntax-based Videotex, end to end protocols

4. DYNAMIC DESCRIPTION

The circuit-mode dynamic description appears in CCITT Recommendation I.220.

Annex 7

TELEX

1. DEFINITION

The Telex service provides interactive text communication. The digital signal at the S/T reference point follows the internationally agreed Recommendations for Telex above the ISDN physical layer. User information is transferred over circuit- or packet-mode bearer channels and signalling is provided over the D-channel.

Annex 8

VIDEOTELEPHONY

1. DEFINITION

Videotelephony service is an audiovisual conversational teleservice providing bidirectional symmetric real-time transfer of voice and moving colour pictures between two locations (person-to-person) via the network involved. The minimum requirement is that under normal conditions the picture information transmitted is sufficient for the adequate representation of fluid movements of a person displayed in head and shoulders view.

2. DESCRIPTION

2.1. General Description

The Videotelephony service is to be used in almost the same way as the ordinary Telephony service for individual communication, the enhancement being in the visibility of the communicating parties which implies a number of possible new applications.

An essential feature of the service is that it is always provided in conjunction with ordinary telephony allowing the user to intercommunicate with all kinds of audiovisual services merely by using the speech communication facility of a videophone terminal. In other words videophone terminals must be capable of supporting all kinds of telephony.

A Videotelephony service may be used also in applications such as communication by speech and hearing impaired persons using sign language and remote surveillance where the speech communication facility is irrelevant.

Videotelephone terminals must be capable of supporting the Telephony teleservice.

An essential feature of the service is that, besides videotelephony, it also provides the user with the possibility to communicate with other ISDN telephone or videotelephone terminals by using only the speech communication facility. It shall be possible to use videotelephone terminals to communicate with 3.1 kHz telephone terminals connected to the PSTN.

2.2. Specific Terminology

Fall-back:
Procedure performed either by the network or by the calling videotelephone terminals to establish calls to 3.1. kHz telephone terminals.

Call 1:
The first call invoked in the Videotelephony teleservice. It identifies the first 64 kbit/s connection between the subscribers. The call is invoked for all the service cases.

Call 2:
The second call invoked in the Videotelephony teleservice. It identifies the second 64 kbit/s connection between the two subscribers. The call is invoked for Case II only (2x64 kbit/s).

Retention time:
This time specifies the amount of time during which the network retains the call information of the original call upon encountering busy or being released. This time is a network operator option. The value of this time is greater than 15 seconds.

Videotelephone terminal:
A terminal that supports the Videotelephony teleservice.

3.1 kHz telephone terminal:
A terminal that supports the Telephony 3.1 kHz teleservice.

7 kHz telephone terminal:
A terminal that supports the Telephony 7 kHz teleservice.

3. PROCEDURES

3.1. General

From the user's point of view the call control procedures should be as simple as for ordinary telephony in order to achieve a high degree of acceptance. The audio tones should have the same meaning as for telephony. Visual guidance with the aid of the videophone terminal display may play an important role in the invocation and operation of the service.

3.2. Provision and Withdrawal

Provision of the service will be by arrangement with the network operator.

3.3. Normal Procedures

3.3.1. *Originating the Call (Call Request)*

Call 1 shall be set up first. After this call has been accepted by the called user, Call 2 can be originated, if necessary.

The characteristics of initial call setup for Case I and Case II shall be identical for Call 1.

Call 1 is devoted to multimedia information transfer (e.g. high quality speech video and data). The transmission audio mode on the end-to-end digital path is defined according to CCITT Recommendations H.221 and H.242.

Call 2 is devoted to video information transfer. The end to end path is framed according to CCITT Recommendation H.221.

Call 2 is invoked by the calling terminal, when Call 1 is in the active phase and after the end-to-end mode initialisation procedure is completed. When the second connection is active, end-to-end alignment procedure occurs and the relative delay between the two connections is adjusted until complete synchronisation is achieved according to CCITT

Recommendation H.221.

The in-band protocol shall be established according to CCITT Recommendation H.242.

The Videotelephony call shall be originated by the originating user activating the terminal performing service selection (if applicable from the originating terminal) and terminating customer selection. During this process, the originating user shall be given the appropriate indications which refer to the state of the call.

From the user's point of view the call request procedure must be available as an operation preferably similar to that for telephony even if two separated calls are established in the network.

Audio tones provided to the user shall be as for the Telephony 3.1 kHz teleservice.

Note: Call 1 should be presented with an alerting phase.
Call 2 should be associated to an automatic answer at the called interface.

If Call 2 cannot be completed due to e.g. remote access conditions or , network congestion, Call 1 can be maintained or released by the calling terminal according to intercommunication requirements. If Call 1 is maintained, the user can re-attempt the establishment of Call 2.

3.3.2. *Indications during Call Setup and Call Acceptance (Answer)*

At the called side, the two calls are accepted after successful checking of compatibility information by the terminal(s) addressed.

After initiating a call, the calling user shall receive an acknowledgement that the network is able to process the call. The called user shall receive an indication of the arrival of an incoming videotelephony call. The calling user shall also be given an indication that the call is being offered to the called user, when an indication is received by the network that the called user is being informed of this call. When the call reaches the called user and the connection is established, an indication shall be sent to the calling user.

The acceptance of the videotelephone call by the terminating user (answer) causes the indication to be removed and bidirectional communication paths to be provided.

The called user controls transmission of his picture to the calling user.

The called user may also provide other information for use by the network in supplementary services provided to the other user (e.g. connected line identity).

Note that in the case where a 3.1 kHz terminal is established first, the acceptance of the call is done according to normal Telephony teleservice procedures.

3.3.3. *Terminating the Call (Call Release)*

A request to terminate the Videotelephony teleservice may be generated by either of the users. If one user terminates the call, the other user is given an appropriate indication.

In general, the release of a videotelephone call should be the same to the release of a telephone call; picture and sound are released simultaneously.

3.3.4. *Change of Terminal Communication Mode*

As a consequence of end-to-end integrity on Videotelephony teleservice, Telephony 7 kHz teleservice and some 3.1 kHz is performed by the network. It will be possible to use the B-channel protocols given in CCITT Recommendations G.725 and H.242.

Depending on the terminal capabilities, it may be possible to change between the following communication modes according to CCITT H.230, tables 2 and 3.

- 3.1 kHz speech (CCITT Rec. G.711),
- 7 kHz speech (CCITT Rec. G.722)
- different videotelephone terminal modes.

Note: The user may be required to establish additional calls in some cases.

Note: As an option, in some circumstances establishment of a videotelephone call can be based on call setup as a 3.1 kHz telephone call and then change to a videotelephone call if a change of service by using an end-to-end procedure is possible. In the case where a 3.1 kHz telephone call is first established on request of the calling user, the calling terminal will try to achieve framing and to exchange terminal capabilities on the existing B-channel. If it succeeds, this channel is used in the same way as a channel which is the result of a Call 1 establishment. The change from the telephone call to a videotelephone call will not cause an interruption of the voice communication. If framing cannot be achieved on the existing B-channel, the calling user has to release the telephone call and to request for Call 1 establishment. This change cannot be performed without an interruption of the existing communication.

4. USER REQUIREMENTS IN CASE OF UNSUCCESSFUL OUTCOME

4.1. Failure Situations due to User Error

1) A user inputting an improper service request shall be given an appropriate failure indication by the network and the call setup shall be ceased.

2) A user inputting a non-valid network number shall be given an appropriate failure indication by the network and the call setup will be ceased.

4.2. Failure Situations due to Called User State

1) A calling user attempting to establish a call to a user who is identified by the network to be busy (either Network Determined User Busy (NDUB) or User Determined User Busy (UDUB)) shall be given an appropriate failure indication by the network and the call setup shall be ceased.

2) A user attempting to establish a call to a user whose terminal equipment fails to respond shall be given an appropriate failure indication by the network and the call setup shall be ceased.

3) On a call to a user whose terminal equipment has responded that the called user is being informed of the call, but has failed to answer within a defined period, the calling user attempting to establish the call shall be given an appropriate failure indication by the network and the call setup shall be ceased.

ISDN and to telephone terminals connected to the PSTN. Optionally it should be able to reach other ISDN audiovisual terminals.

4) A videotelephone terminal shall be able to accept calls from 3.1 kHz and 7 kHz (if 7 kHz capability is supported) telephone terminals connected to the ISDN and from 3.1 kHz telephone terminals connected to the PSTN. Optionally, it should be able to accept calls from other ISDN audiovisual terminals.

   As an option, videotelephone terminals may be pre-programmed to receive incoming videotelephone calls only. This latter function may be requested by users possessing e.g. both a videotelephone terminal and a 3.1 kHz telephone terminal connected to the same access arrangement.

4.3. Fall-back Procedures

4.3.1. *Fall-back in the Destination Network*

Fall-back to 3.1 kHz Telephony shall be an inherent feature of Videotelephony teleservice and shall be provided as a default procedure.
The user shall be offered the possibility of indicating whether interworking/fall-back to the 3.1 kHz Telephony teleservice is required. A request for the Videotelephony teleservice without fall-back (if indicated by the calling terminal) shall be possible.

The following procedure shall apply:

- If the calling user has indicated that fall-back is allowed, the network may offer the call to the called user at all videotelephone and 3.1 kHz telephone terminals, if possible. The called user can accept the call either as a videotelephone or a 3.1 kHz telephone call at any terminal where the call is offered.

  Note: The called terminals may recognise the fall-back situation and indicate it to the user.

- The calling user shall be informed of the resultant telecommunication service, i.e. the Videotelephony or 3.1 kHz Telephony teleservice.

- If no terminal accepts the call, this shall be indicated to the calling user.

- If a busy condition is met at the terminals, supplementary services e.g. Completion of Calls to busy Subscriber shall apply.

  Note: Echo cancellation will be disabled for videotelephone calls. If fall-back occurs there is no current signalling mechanism for re-enabling the echo cancelers.

- When fall-back is not implemented by the network (possible short term situation), fall-back may be performed end-to-end by the calling videotelephone terminal by originating a 3.1 kHz telephone call.

4.3.2. *Fall-back when the ISDN does not offer the Videotelephone Teleservice*

If the calling user has indicated that fall-back is allowed but the destination network does not support the videotelephone capabilities, the calling user shall receive both an indication that fall-back has occurred and an indication of the resultant telecommunication service.
The called user shall be offered the incoming call as a Telephony 3.1 kHz call.

5. INTERWORKING WITH PRIVATE ISDNs

If the called user is on a private ISDN, the fall-back procedures will be performed by the private ISDN.

The result of call presentation (Videotelephony or 3.1 kHz Telephony) within the private ISDN shall be indicated to the public ISDN.

6. ATTRIBUTES AND VALUES

6.1. Application of the Attribute Method

Depending on the case that applies, the Videotelephony teleservice description is based on invocation of one or two calls: Call 1 and Call 2 described according to the attribute method.

6.2. Low Layer Attributes

6.2.1. *Call 1*

1. Transfer mode	Circuit
2. Transfer rate	64 kbit/s
3. Transfer capability	7 kHz

Note: Before 7 kHz audio bearer service is available as an interim solution videotelephones should use unrestricted digital information as the transfer capability when calling other videotelephones.

4. Structure	8 kHz integrity
5. Establishment of communication	Demand
6. Symmetry	Bidirectional symmetric
7. Configuration of communication	Point-to-point, multipoint

Note: In the case where fall-back to the 3.1 kHz Telephony teleservice occurs, values of 3.1 kHz Telephony teleservice bearer capability apply. Also if optionally a 3.1 kHz Telephony teleservice bearer capability apply (telephone call instead of Call 1).

6.2.2. *Call 2*

1. Transfer mode	Circuit
2. Transfer rate	64 kbit/s
3. Transfer capability	Unrestricted digital information
4. Structure	8 kHz integrity
5. Establishment of communication	Demand
6. Symmetry	Bidirectional symmetric
7. Configuration of communication	Point-to-point, multipoint

6.3. High Layer Attributes

6.3.1. *Call 1*

10. Type of user information	Speech (telephony), video, data, audiovisual (information)

11. Layer 4 protocol functions	CCITT Rec. H.221
12. Layer 5 protocol functions	CCITT Rec. H.242
13. Layer 6 protocol functions	CCITT Recs G.722 (option), G.711, H.261
14. Layer 7 protocol functions	

Note: In the case where fall-back to 3.1 kHz Telephony teleservice occurs or, if optionally a 3.1 kHz Telephone call is established first, the value of attribute 10 is speech and the value of attribute 13 is CCITT Rec. G.711.

6.3.2. *Call 2*

10. Type of user information	Video
11. Layer 4 protocol functions	CCITT Rec. H.221
12. Layer 5 protocol functions	CCITT Rec. H.242
13. Layer 6 protocol functions	CCITT Rec. H.261
14. Layer 7 protocol functions	

7. SUPPLEMENTARY SERVICES PROVIDED

Supplementary services should be considered as applicable to the videotelephone communication as a whole, even if two separated calls (i.e. Call 1 and Call 2) are established in the network. Restrictions identified should be applied by the videotelephone terminal.

Application of Telephony supplementary services to the Videotelephony teleservice is described separately.

Note: Only one set of addressing information, i.e. ISDN number address should be allocated to a given videotelephone terminal and the same addressing information should always be used for Call 1 and Call 2 request.

8. QUALITY OF SERVICE

8.1. Synchronism of Speech and Lip Movement (Lip Synchronism)

No subjectively discernible difference in the delay of the speech and video signal.

8.2. Sound Quality

No significant difference compared to the speech quality used in the 64 kbit/s ISDN Telephony services based on bandwiths of 3.1 kHz or 7 kHz.

8.3. Picture Quality

Optimisation of picture quality is for further study including the need for adequate representation of fluid movements (see note).

Note: The urgent need to develop both objective and subjective quality parameters for the received motion picture is widely identified.

8.4. **The Overall Delay**

The overall delay is defined to consist of transmission delay and the characteristic delay of a videotelephone terminal.
Characteristic delay of a videotelephone terminal is the delay introduced by the terminal, when only lips and eyes of the talking user are moving.

The overall effect on quality by the delays introduced by video codecs and transmission facilities needs to be taken into account in the service. Increased delays may impair user acceptability.

Maximum allowable delay including maximum number of satellite hops are left for further study.

For the Videotelephony teleservice Case II it is possible that one 64 kbit/s connection is routed via a terrestrial path while the other is routed via satellite. In this case, the resynchronisation is performed by the terminal.

The quality of service is the same as if both channels were routed via satellite.

9. **INTERCOMMUNICATION/INTERWORKING POSSIBILITIES**

- 7 kHz Telephony

- 3.1 kHz ISDN Telephony

- 3.1 kHz PSTN Telephony

- Audiovisual services