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Foreword

This European Telecommunication Standard (ETS) has been produced by the Terminal Equipment (TE) Technical Committee of the European Telecommunications Standards Institute (ETSI).

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1 Scope

This European Telecommunication Standard (ETS) defines the stage one of the videoconference teleservice for the pan-European Integrated Services Digital Network (ISDN) as provided by European public telecommunications operators. Stage one is an overall service description from the user's point of view (see CCITT Recommendation I.130 in annex D), but does not deal with the details of the human interface itself.

The videoconference teleservice applies to point-to-point communications established between either two terminals or one terminal and a Multipoint Control Unit (MCU).

The "conference service" provided by a MCU is outside the scope of this ETS. Related information may be found in informative annexes A and B of this ETS, and also in ITU-T Recommendation F.730 (see annex D).

This ETS defines the interworking requirements of private ISDNs with the public ISDN.

In addition, this ETS specifies the base requirements where the service is provided to the user via a private ISDN.

This ETS does not specify the additional requirements where the service is provided to the user via a telecommunications network that is not an ISDN but does include interworking requirements of other networks with the public ISDN.

Charging principles are outside the scope of this ETS.

The values of the general attributes are outside the scope of this ETS.

The videoconference teleservice is a real-time teleservice in which high quality audio, video, control signals and data are interchanged using two or more circuit-mode 64 kbit/s connections.

This ETS is applicable to the stage two and stage three standards for the ISDN videoconference teleservice. The terms "stage two" and "stage three" are also defined in CCITT Recommendation I.130 (see annex D). Where the text indicates the status of a requirement (i.e. as strict command or prohibition, as authorization leaving freedom, or as a capability or possibility), this is reflected in the text of the relevant stage two and stage three standards.

Furthermore, conformance to this ETS is met by conforming to the stage three standard with the field of application appropriate to the equipment being implemented concerning characteristics at the user-network interface, and by conforming to the standard on the end-to-end characteristics appropriate to the equipment being implemented. Therefore, no method of testing is provided for this ETS.

2 Normative references

This ETS incorporates by dated or undated reference, provisions from other publications. These normative references are cited at the appropriate places in the text and the publications are listed hereafter. For dated references, subsequent amendments to, or revisions of any of these publications apply to this ETS only when incorporated in it by amendment or revision. For undated references the latest edition of the publication referred to applies.

[1]   ETS 300 144 (1996): "Integrated Services Digital Network (ISDN); Audiovisual services; Frame structure for a 64 kbit/s to 1 920 kbit/s channel and associated syntax for inband signalling".

[2]   ETS 300 111 (1992): "Integrated Services Digital Network (ISDN); Telephony 3,1 kHz teleservice, Service description".

[3]   ETS 300 263 (1994): "Integrated Services Digital Network (ISDN); Telephony 7 kHz teleservice, Service description".

[4]   CCITT Recommendation G.728 (1992): "Coding of speech at 16 kbit/s using low-delay code excited linear prediction".

[5]   ETS 300 143 (1994): "Integrated Services Digital Network (ISDN); Audiovisual services; inband signalling procedures for audiovisual terminals using digital channels up to 2 048 kbit/s".

[6]   CCITT Recommendation G.711 (1985)88): "Pulse code modulation (PCM) of voice frequencies".

[7]   CCITT Recommendation G.722 (1985)88):   "7 kHz   audio-coding   within 64 kbit/s".

[8]   ITU-T Recommendation I.140 (1985)93):   "Attribute   technique   for   the characterization of telecommunication services supported by an ISDN and network capabilities of an ISDN".

[9]   ETS 300 483 (1996): "Terminal Equipment (TE); Integrated Services Digital Network (ISDN); Multipoint communications for Audiovisual Services; Main functionalities and basic requirements for Multipoint Control Units (MCUs)".

[10]   ITU-T Recommendation H.242 (19985)96):   "System   for   establishing communication between audiovisual terminals using digital channels up to 2 Mbit/s".

[11]   ITU-T Recommendation H.233: "Confidentiality system for audiovisual services".

[12]   ITU-T Recommendation H.243 (1985)93):   "Procedures   for   establishing communication between three or more for audiovisual terminals using digital channels up to 1 920 kbit/s".

[13]   ETS 300 345: "Integrated Services Digital Network (ISDN); Interworking between public ISDNs and private ISDNs for the provision of telecommunication services. General aspects".

[14]   ITU-T Recommendation T.122 (1985)93): "Multipoint communication service for audiographics and audiovisual conferencing service definition".

[15]   ITU-T Recommendation T.123 (1985)94): "Protocol stacks for audiographic and audiovisual teleconference applications".

[16]   ITU-T Recommendation T.124 (1995): "Generic conference control".

[17]   ITU-T Recommendation T.125 (1994):   "Multipoint   communication   service protocol specification".

[18]   ITU-T Recommendation T.126: "Multipoint still image and annotation protocol".

[19]   ITU-T Recommendation T.127: "Multipoint binary file transfer protocol".

[20]   ITU-T Recommendation H.261: "Video codec for audiovisual services at p x 64 kbit/s".

3   Definitions

For the purposes of this ETS, the following definitions apply:

Integrated Services Digital Network (ISDN): An integrated services network that provides digital connections between user-network interfaces (ITU-T Recommendation I.112, subclause 2.3, definition 308).

Channel Aggregation Unit (CAU): A device having both multiple 64 kbit/s accesses and a single access at a higher bitrate. Reference in this ETS to "CAU" or "aggregator" means equipment conforming to ETS 300 481 (cf. bibliography in annex D).

service; telecommunications service: That which is offered by a service provider to its customers in order to satisfy a specific telecommunication requirement [ITU-T Recommendation I.112, subclause 2.2, definition 201].

supplementary service: A supplementary service modifies or supplements a basic telecommunication service. Consequently, it cannot be offered to a customer as a standalone service. It should be offered together with or in association with a basic telecommunication service. The same supplementary service may be common to a number of telecommunication services [ITU-T Recommendation I.210, subclause 2.4].

teleservice: A type of telecommunication service that provides the complete capability, including terminal equipment functions, for communication between users according to protocols established by agreement between service providers [ITU-T Recommendation I.112, subclause 2.2, definition 203].

videoconference terminal: A terminal that supports the videoconference teleservice.

7 khz terminal: A terminal that supports the telephony 7 kHz teleservice.

3,1 khz terminal: A terminal that supports only the telephony 3,1 kHz teleservice.

videotelephone terminal: A terminal that supports the videotelephony teleservice.

fall-back: The mechanism whereby the request for the videoconference teleservice, which includes an indication that an alternative teleservice is acceptable, results in a call using the alternative teleservice. In the case of the videoconference teleservice, the alternative teleservice is the telephony 3,1 kHz teleservice. However, at the called side fall-back might result to the telephony 7 kHz teleservice.

served user: A user to whom the videoconference teleservice is provided.

Network Determined User Busy (NDUB): See description given in subclause 2.1.4 of ITU-T Recommendation I.221.

User Determined User Busy (UDUB): Is specified for the case where the network offers the call to the user and if no compatible terminal responds "positively" but one or more compatible terminals respond "user busy".

restricted network: A network consisting of multiples of 64 kbit/s links, but where only multiples of 56 kbit/s are usable for the terminals (ETS 300 144 [1]).

retention timer: This timer specifies the amount of time that the network retains all the call information supplied by the calling user when the call encounters busy or is terminated. Implementation of this timer is a network option. The value for this timer is greater than 15 seconds.

Multipoint Control Unit (MCU): A piece of equipment located in a node of the network or in a terminal that connects several terminals and, according to certain criterion, processes audiovisual signals and distributes them to the connected terminals. Its main functionalities and applicable requirements are specified in ETS 300 483 [9].

Service Channel (SC): The eighth sub-channel of a 64 kbit/s channel, or the seventh sub-channel when communicating in restricted mode (ETS 300 144 [1]).

multi layer protocols: As defined in the T.120 series of ITU-T recommendations.

ISDN number: A number conforming to the numbering plan and structure specified in CCITT Recommendation E.164.

4   Abbreviations

For the purposes of this ETS, the following abbreviations apply:

BARAS	Baseline Algorithm Recommended for use in Audiovisual Systems
CAU	Channel Aggregation Unit
CIF	Common Intermediate Format
ISDN	Integrated Services Digital Network
MCS	Multipoint Communication Service
MCU	Multipoint Control Unit
MLP	Multi Layer Protocols
MSN	Multiple Subscriber Number
NDUB	Network Determined User Busy
PSTN	Public Switched Telephone Network
QCIF	Quarter Common Interface Format
TE	Terminal Equipment
UDUB	User Determined User Busy

5   Description

The videoconference teleservice is a real-time teleservice in which audio and video signals are transmitted inband together with signals for conference control and data communication, between a terminal and a MCU, or between two terminals by means of two or more (up to 30) 64 kbit/s circuit-mode connections in the ISDN.

> NOTE 1:   The quality of the moving colour pictures of the participants rises quickly with the number of the 64 kbit/s connections.

> NOTE 2:   A videoconference terminal may support, as options, the telephony 3,1 kHz teleservice as described in ETS 300 111 [2] and the telephony 7 kHz teleservice as described in ETS 300 263 [3].

> NOTE 3:   To improve the video and data communication capacity, the audio encoding specified in CCITT Recommendation G.728 [4] may be implemented as an option.

The network provides tones and/or announcements to support this teleservice. Tones and/or announcements can be used to indicate the progress (or lack of progress) of a call.

User information and end-to-end signalling shall be transferred over the B-channel. Signalling for call control shall be provided over the D-channel.

A MCU, to which all locations are connected individually over the ISDN, is required for the interconnection of three or more users in a multipoint configuration. Two or more MCUs may be interconnected to increase the number of participants of a multipoint conference.

6   Procedures

6.1   Provision and withdrawal

The videoconference teleservice shall either be provided after prior arrangement with the service provider, or shall be generally available.

> NOTE 1:   As a service provider option, the videoconference teleservice can be 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 options for the interface are summarized in table 1.

Table 1

Subscription option	Value
Maximum number of information channels available	m, where m is not greater than the number of information channels on the interface
Maximum number of total calls present	n, where n is not greater than the number of information channels on the interface

NOTE 2: The physical interface may be one, several, or a combination of basic and primary rate access(es). A Channel Aggregation Unit (CAU) may be used to allow connection of single access terminals. It should be noted that in this context an interface may consist of a group of physical interfaces.

The user can be identified by an ISDN number, a group of ISDN numbers, or globally for all ISDN numbers on the interface.

More than one ISDN number can be associated with the interface as a part of a supplementary service such as the Multiple Subscriber Number (MSN) supplementary service. In the case of one ISDN number, the option given in table 1 for the number of calls can only exceed the number of information channels in association with a supplementary service (e.g. the call waiting supplementary service). As a network provider option, separate values may be specified for incoming and for outgoing calls for either or both of the limits.

The videoconference teleservice shall be withdrawn by the service provider upon the request of the subscriber, or for service reasons.

6.2 Normal procedures

For the videoconference teleservice, communication is by means of two or more 64 kbit/s connections, and so two or more calls shall be established. For the establishment of the first call, the procedures of subclause 6.2.1 shall apply. The subsequent calls shall be established using the procedures given in subclause 6.2.1, provided that the first call has already been established.

The network shall provide out of band indications to indicate call progress. Network generated tones and/or announcements shall be provided for the videoconference teleservice.

6.2.1 Originating the call (call establishment)

The videoconference teleservice shall be originated by the originating user:

- activating the terminal;
- performing the service selection, if applicable for the originating terminal; and
- terminating selection.

During this process, the originating user shall be given the appropriate indications as to the state of the call.

The end-to-end path is framed according to ETS 300 144 [1]. The inband protocol shall be established according to ETS 300 143 [5].

Call 1 shall be established before initiating any additional call.

NOTE 1: Call 1 is initially devoted to transfer of speech. The transmission audio mode on the end-to-end digital path is defined according to ETS 300 144 [1], using the procedures defined in ETS 300 143 [5].

Audio tones provided to the user during Call 1 shall be as for the telephony 3,1 kHz teleservice, given in ETS 300 111 [2].

The subsequent calls can only be originated by the calling user's terminal after the connection has been established on Call 1 and after the end-to-end capability exchange and mode initialization procedure are completed.

NOTE 2: The subsequent calls are devoted to video and data communication.

6.2.2 Indications during call establishment

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

After initiating Call 1 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 videoconference call. The calling user shall also be given an indication that the call is being offered to the called user. Acceptance of Call 1 by the called user (answer), shall cause the indications to be removed and the bi-directional communication path to be provided.

After initiating each subsequent 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 every additional incoming videoconference call. Originating address (when available) and additional channel identification number shall be used to check compatibility between successive incoming calls.

The acceptance of the subsequent calls by the called user (answer), shall cause the bi-directional communication path to be provided.

6.2.3 Varying the B-channel number during the conference

The user (or MCU) can vary the number of connections during the conference. Set-up and release of additional channels use the standard D-channel procedures. However the change of multiplex structure shall be prepared by preliminary appropriate inband signalling procedures.

6.2.4 Terminating the conference

6.2.4.1 Point-to-point conference

A request to terminate the videoconference teleservice can be generated by either of the users. If one user terminates the service, the other user(s) shall be given an appropriate indication.

6.2.4.2 Multipoint conference

A request to terminate the call(s) can be generated by either the user or the MCU controlled by the MCU director or the user given the Chair control described in ITU-T Recommendation H.243 [12]. If one user terminates his call to the MCU, the other user(s) shall be given an appropriate indication.

The termination of the whole conference shall be controlled by the MCU or by the user given the Chair control.

NOTE: An MCU could be designed to establish the call to a given user if this connection is unintentionally released.

6.2.5 Change of communication mode

Because of end-to-end integrity on the videoconference teleservice, different communication modes can be used according to ETS 300 143 [5].

The audio mode used at call set-up is:

- 3,1 kHz speech (CCITT Recommendation G.711 [6]).

The normal operating mode is:

- 7 kHz speech (CCITT Recommendation G.722 [7]).

However, a video conference terminal should be able to support as an option:

- 3,1 kHz speech (CCITT Recommendation G.728 [4]).

The user (or MCU) can vary the audio mode during the communication.

The available bit rate is used by data and/or video signal for various picture definition (Common Interchange Format (CIF), Quarter Common Interface Format (QCIF)).

6.2.6 Multipoint control

To allow communication between three or more videoconference terminals a MCU shall be used. The MCUs may be cascaded, (multiple MCU configurations) as described in ETS 300 483 [9], when required by a particular conference. The procedures for establishing connections between MCUs can follow the principles for establishing connections between a MCU and a terminal (one MCU plays the role of a simple terminal).

The connections between an MCU and the users participating in the conference shall be established by:

- the MCU calling the user (dial out);

- the user calling the MCU (dial in, also referred as meet-me).

When the dial-in procedure is used, the MCU should provide an optional control system to prevent unauthorized access to the conference.

6.2.7 Data communication

The data transmission facility provides means for transmission of additional control information (e.g. Chair-control functions), transmission of user information, communication between PCs, etc. Protocols for data transmission and conference control are specified in the ITU-T T.120 series of recommendations.

6.3 Exceptional procedures

6.3.1 Situations at the calling user side

When the network receives an improper service request from the user, the network shall give that user the appropriate failure indication and the call establishment shall be ceased.

A user inputting an invalid ISDN number shall be given an appropriate failure indication by the network and the call establishment shall be ceased.

6.3.2 Situations at the called user side

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.

A user attempting to establish a call to a user whose Terminal Equipment (TE) fails to respond shall be given an appropriate failure indication by the network and the call establishment shall be ceased.

On a call to a user whose TE has responded that the called user is being informed of the call, but has failed to establish the connection within a defined time, the calling user attempting to establish the call shall be given an appropriate failure indication by the network and the call establishment shall be ceased.

6.3.3 Situations due to network conditions

A user attempting to establish a call, but meeting problems due to network conditions (e.g. congestion) shall be given the appropriate indication by the network.

6.3.4 Retention of call information

If a user attempts to establish a call, but meets problems due to network conditions (e.g. congestion) or called user state (e.g. NDUB or UDUB) then, according to a network option, the network shall retain all the information supplied by the calling user for the duration of the retention timer.

6.3.5 Failure of additional calls

In a case where the number of additional channels, resulting from the initial capability exchange, cannot be reached, the calling user terminal shall give information to help the decision: either to use a lower channel number, or to clear the call attempt.

7 Intercommunication and interworking

7.1 Normal operation

Intercommunication of videoconference terminals with videotelephony or audiographic ISDN terminals, shall be ensured only in the case where:

- such terminals can (by design and programming) answer videoconference incoming calls;

- the videoconference terminals can initiate, on a call per call basis, videotelephony, or audiographic outgoing calls;

   NOTE 1:   When the call is answered by a videotelephony or an audiographic terminal, the resulting service is limited by the called terminal capabilities.

   NOTE 2:   As a terminal option, videoconference terminals may be pre-programmed to accept only incoming videoconference calls and/or to initiate only outgoing videoconference calls.

7.2 Fall-back procedures

7.2.1 Fall-back to the telephony 3,1 kHz teleservice

Fall-back to telephony 3,1 kHz teleservice shall be an inherent feature of the videoconference teleservice.

If the calling user terminal indicates that fall-back is not allowed when originating a call, fall-back procedures shall not apply.

   NOTE 1:   This situation may lead to an unsuccessful call attempt due to called user terminal capabilities.

If the calling user has allowed fall-back to the telephony 3,1 kHz teleservice, then the following procedure applies:

- the call shall be offered to all terminals connected to the ISDN access(es) identified by the destination number. Any terminal supporting the videoconference teleservice or the telephony 3,1 kHz teleservice shall respond to the incoming call. Priority for alerting shall be given by the network to videoconference terminals. The user may answer the incoming call from any compatible alerting terminal;

- the network and the calling user shall be informed of the resultant telecommunications service, i.e. the videoconference or telephony 3,1 kHz teleservice;

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

   NOTE 2:   Echo cancellation in the network is disabled for videoconference and telephony 7 kHz teleservice calls. If fall-back to telephony 3,1 kHz teleservice occurs, re-enabling of echo cancellers is necessary. However, some networks may not support the corresponding signalling mechanism.

   NOTE 3:   When fall-back is not implemented by the network, fall-back may be performed end-to-end by the calling videoconference terminal, by originating a new call with a different teleservice (e.g. audiographic conference, telephony 7 kHz or telephony 3,1 kHz).

If the calling user has indicated that fall-back to the telephony 3,1 kHz teleservice is allowed and the network does not support the fall-back procedure, the telephony 3,1 kHz teleservice shall be provided to the calling user.

7.2.2 Interworking with non-ISDNs

If the calling user has indicated that fall-back is allowed and interworking with the PSTN occurs, the telephony 3,1 kHz teleservice shall be provided. The calling user shall be informed of this situation.

If fall-back is not allowed for the call, the communication shall not be established.

7.3 Interworking with private ISDNs

General information on the situation where the communicating users are attached to a private ISDN and a public ISDN is given in ETS 300 345 [13].

If the calling user has indicated that fall-back is allowed and the called user is connected to a private ISDN that supports the videoconference teleservice, then, in situations where fall-back applies, the fall-back procedures shall be performed by the private ISDN or the called terminal.

The result of call set-up (the videoconference, or telephony 3,1 kHz teleservice) within the private ISDN shall be indicated to the public ISDN.

In the case where the private ISDN does not support the fall-back procedures, as a result, the telephony 3,1 kHz teleservice is provided by the public network.

If the private network does not support the videoconference teleservice, it should preferably implement fall-back to the telephony 3,1 kHz teleservice.

In both of the last two cases the calling user shall receive an indication from the public ISDN that fall-back to the telephony 3,1 kHz teleservice has occurred.

If fall-back is not allowed for the call, the related mechanism shall not be activated in the private network.

MCUs may be located in a private ISDN.

7.4 Restricted network operation

All terminals for the videoconference teleservice shall be capable of interoperation with terminals connected to restricted networks: that is, networks which provide connections at 56 kbit/s, or at multiples of 64 kbit/s with a ones density restriction, as described in ETS 300 144 [1].

In order to effect such interoperation, the procedures included in clause 13 of ITU-T Recommendation H.242 [10] shall be implemented.

8 Special use of supplementary services

Before invoking supplementary services that interrupt the inband signalling communication (e.g. the call hold supplementary service), the inband communication, managed by the terminals, shall revert to mode 0, as described in ETS 300 143 [5].

If the inband communication is interrupted by the network as a result of one user invoking a supplementary service (e.g. call hold supplementary service, or terminal portability supplementary service), the network shall provide an appropriate indication (e.g. all ones or idle signal) in the B-channels.

Use of Calling Line Identification supplementary service is recommended in order to help sorting incoming calls which might be simultaneous received from different terminals calling an MCU. This will generally speed up conference set-up time.

Line and/or Trunk Hunting supplementary services can be used for call set-up of initial and additional channels where the called terminal (or MCU) is accessed through a group of basic and/or primary rate accesses by a common destination number.

Applicability of other supplementary services is described in annex B.

9 Static description of the service using attributes

The attributes shall be as defined in ITU-T Recommendation I.140 [8], annex A, subclause A.1.1.

The values of the attributes shall be as defined in ITU-T Recommendation I.140 [8], annex A, clause A.2.

9.1 Low layer attributes

The information transfer attributes of this teleservice are specified in tables 2 and 3.

Table 2: Values for information transfer attributes for Call 1

Attribute	Possible values
Information transfer mode	- circuit
Information transfer rate	- 64 kbit/s
Information transfer capability	- unrestricted Digital Information with Tones and Announcement - speech (note)
Structure	- 8 kHz integrity
Establishment of communication	- demand
Symmetry	- bi-directional symmetric
Communication configuration	- point-to-point
NOTE: In the case of fall-back to the telephony 3,1 kHz teleservice, the information transfer capability shall be speech.	

Table 3: Values for information transfer attributes for Call 2 and subsequent calls

Attribute	Possible values
Information transfer mode	- circuit
Information transfer rate	- 64 kbit/s
Information transfer capability	- unrestricted Digital Information
Structure	- 8 kHz integrity
Establishment of communication	- demand
Symmetry	- bi-directional symmetric
Communication configuration	- point-to-point

9.2 Access attributes

The access attributes of this teleservice are specified in tables 4 and 5.

Table 4: Values of access attributes for Call 1

Attribute	Possible values
Access channel and rate	User information: B (64 kbit/s) Signalling: D (16 kbit/s or 64 kbit/s)
Signalling access protocol, information (note)	User information: ETS 300 143 [5], ETS 300 144 [1] CCITT Rec.: G.711 [6], G.722 [7], G.728 [4] ITU-T Rec.: H.243, T.122, T.123, T.124, T.125. Signalling: ETS 300 011, ETS 300 012, ETS 300 402-1, ETS 300 403-1, ETS 300 267-1.

Table 5: Values of access attributes for Call 2 and subsequent calls

Attribute	Possible values
Access channel and rate	User information: B (64 kbit/s) Signalling: D (16 kbit/s or 64 kbit/s)

Signalling access protocol, information (note)	User information: ETS 300 143 [5],
	                                ETS 300 144 [1]
	ITU-T Rec.: H.243, T.122, T.123, T.124, T.125.
	Signalling: ETS 300 011,
	                       ETS 300 012,
	                       ETS 300 402-1,
	                       ETS 300 403-1,
	                       ETS 300 267-1.

9.3 High layer attributes

Values of high layer attributes for Call 1

Type of user information: speech, data, control and indication.

Layer 4 protocol functions: ITU-T Recommendation T.123 [15].

Layer 5 protocol functions: ITU-T Recommendation T.123 [15].

> NOTE 1:   The layer 5 protocol functions is an option in the ISDN extended profile specified in ITU-T Recommendation T.123 [15].

Layer 6 protocol functions: CCITT Recommendations G.711 [6] and G.722 [7] and
CCITT Recommendation G.728 [4], ITU-T Recommendation H.261 [20]
and ITU-T Recommendation T.123 [15].

> NOTE 2:   The layer 6 protocol functions is an option in the ISDN extended profile specified in ITU-T Recommendation T.123 [15].

Layer 7 protocol functions: ITU-T Recommendations T.122 [14], T.124 [16] and T.125 [17].

Values of high layer attributes for Call 2 and subsequent calls

Type of user information: data, graphics, still image control and indication.

Layer 4 protocol functions: ITU-T Recommendation T.123 [15].

Layer 5 protocol functions: ITU-T Recommendation T.123 [15].

> NOTE 3:   The layer 5 protocol functions is an option in the ISDN extended profile specified in ITU-T Recommendation T.123 [15].

Layer 6 protocol functions: ITU-T Recommendation T.123 [15] and
ITU-T Recommendation H.261 [20].

> NOTE 4:   The layer 6 protocol functions is an option in the ISDN extended profile specified in ITU-T Recommendation T.123 [15].

Layer 7 protocol functions: ITU-T Recommendations T.122 [4], T.124 [16] and T.125 [17].

> NOTE 5:   Depending on the application, more high layer attributes may be applicable.

9.4 General attributes

This ETS does not provide values for general attributes.

10   Encryption

This facility is desired by many users, but is optional in the videoconference teleservice. Use shall be made of the Encryption Control Signal channel described in ETS 300 144 [1] according to the procedures of ETS 300 143 [5] and ITU-T Recommendation H.233 [11].

When implemented for general use between any terminals in Europe, it is strongly recommended that the encryption algorithms available should include Baseline Algorithm Recommended for use in Audiovisual Systems (BARAS), in order to ensure that there is at least one algorithm available at both terminals; moreover the key management processes should include either Diffie-Hellman or RSA, according to ITU-T Recommendation H.234, using BARAS for the key encryption.

Other encryption algorithms and key management processes may be appropriate in closed-user groups.

Annex A (informative): Various functionalities of the complete "videoconference service"

A.1 Overall description

The videoconference service provides bi-directional real time transfer of voice and moving colour picture between individual or groups of users located in two or more separate locations.

The videoconference allows communication between:

- two users (e.g. terminals) in a point-to-point configuration via the ISDN over two or several B-channels;

- three or more terminals in a multipoint configuration.

In the case of two locations, the service is supported by two videoconference terminals connected through the ISDN by a point-to-point communication.

When there are more than two locations, the service requires a MCU.

> NOTE 1: In case of terminals located near the MCU or included in the same Private Network one or several point-to-point connections may be permanent and use digital leased lines.

Applications of the videoconference could be:

- business meetings (conducted or not conducted) possibly including interactive viewing of documents, graphics, still and moving videos;
- remote consultations;
- remote support in technical maintenance, medical diagnostics;
- tele-education;
- panel discussion.

> NOTE 2: A special videoconference terminal (with additional keyboard and display) is needed to control MCU operation where a conducted mode is to be provided.

A.2 Videoconference configurations

The configuration can be either point-to-point or multipoint. The latter can be further subdivided in:

- switched multipoint;

- multipoint with image composition.

> NOTE: The videoconference should normally be supported by a multichannel communication, established point-to-point and channel-per-channel, using the videoconference teleservice specified in this ETS; however it may use other telecommunication services (e.g. Multirate n * 64 kbit/s Bearer Service).

A.2.1 Point-to-point configuration

Two videoconference terminals are directly connected via the ISDN (without any MCU). Conference management is by bilateral negotiation between the terminals.

The user should dial a single number on the keyboard or on the pad (the numbering for the various B-channels used in a conference call is perceived to be unique by the user).

> NOTE 1: Some countries may use a numbering scheme for basic accesses where steps of 2, 5 and 10 are defined when multiple basic accesses are used.

> NOTE 2: 56 kbit/s restricted networks for which interworking may be provided use different numbers for the different calls.

A.2.2   Switched video multipoint configuration

This preferred configuration requires at least one MCU (figure A.1). This MCU receives signals from every terminal; it selects according to predetermined rules or to specific commands the image sent out to every terminal; it adds for each terminal the audio signals from all other terminals; it handles the signalling, controls, forwards them when necessary and returns the convenient feedback to the user; it manages the optional data channels and sends the data to the appropriate terminals.
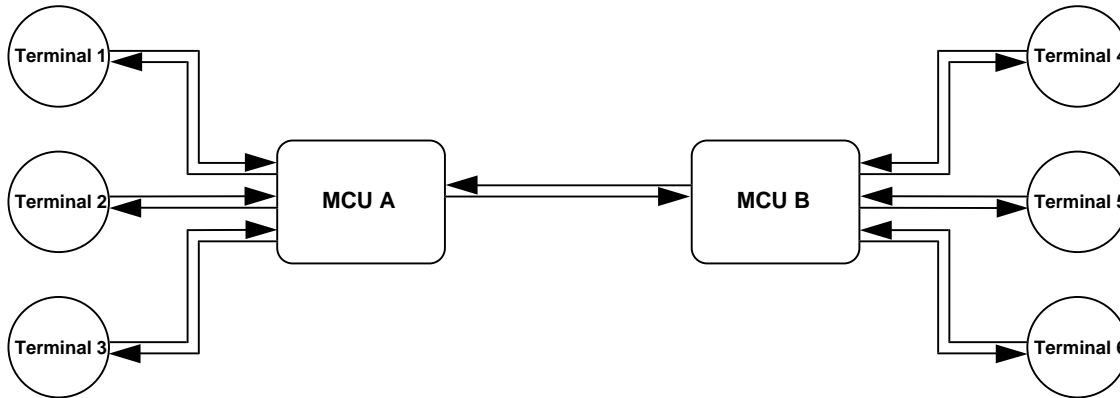

Figure A.1: Switched-video multipoint videoconference with two MCUs

The general requirements applicable to a Multipoint Conference Unit are described in ETS 300 483 [9].

The connections are established through the ISDN using the videoconference teleservice.

For the user (or the MCU) the same dialling procedure as for point-to-point applies.

From the network point of view the connections that need to be established are as follows:

-        several point-to-point connections between each location and the MCU;

-        possibly several point-to-point connections between MCUs.

A.2.3   Multipoint configuration with picture composition

This configuration always requires a MCU. This MCU receives audio and video signals from every terminal and composes combined sound and picture which are transmitted to every videoconference terminal (see figure A.2).

The split-screen technique may be used to permanently display a picture of all participants ("Continuous presence"). The MCU is also used to process the control and indication signals and to transmit data channels when they are present.

In a particular implementation every terminal may transmit a QCIF picture to the MCU, while this one transmits a combined CIF picture using an higher bit rate in the other direction.

For the user (or the MCU) the same dialling procedure as for point-to-point applies.

This configuration relies on teleservices differing from the videoconference teleservice specified in the previous clauses (non-symmetrical bit rates).
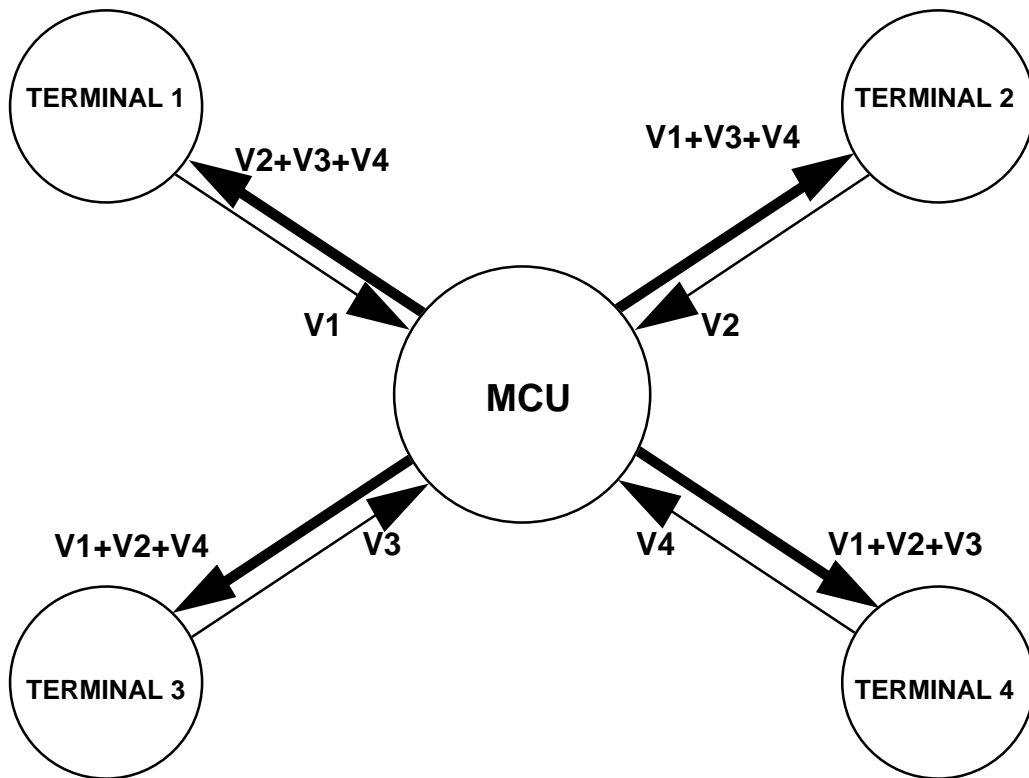
Figure A.2: Shared channel multipoint video conference

A.3   Conference set-up mode

A videoconference may be set-up according to various modes:

- the "Meet me" mode where every participant needs to set-up the communication from his/her own terminal towards the MCU;

- the "Dial-out" mode where the MCU (or MCUs) set-up the communication mode towards videoconference terminals of every participant;

- the "Add-on" mode where one participant (generally the conference controller) sets-up an initial communication towards the MCU and controls dialled-out communication towards other participants, transmitting their destination number through inband signalling.

    NOTE: These conference set-up modes do not require the support of the supplementary service of same name by the terminal or the network.

A reservation system is essential to support the videoconference service. It is normally used to support the dial-out mode. In any case it is used by customers who want reserve, review, change or cancel a videoconference. The reservation parameters include at least: day, time and length of the conference, list of conferees with their ISDN number, set-up mode, and MCU(s) configuration.

Annex B (informative): Applicability of supplementary services to the videoconference teleservice

B.1   Hold, Transfer and Portability

The following supplementary services do not seem useful for the videoconference teleservice; their applicability remains for further study:

- the call hold supplementary service;

- the Explicit Call Transfer supplementary service;

- the Terminal Portability supplementary service.

B.2   Multiparty supplementary services

The following supplementary services, as previously defined for telephony 3,1 kHz teleservice are not applicable for the videoconference teleservice:

- the conference call, add-on supplementary service;

- the meet-me conference supplementary service;

- the three-party (3 pty) supplementary service;

A similar functionality can be obtained using a MCU.

B.3   Supplementary services applicable without restriction

There is no additional requirement for the application of the following supplementary services:

- number identification services:

    a)   Calling Line Identification Presentation (recommended, see clause 8);

    b)   Calling Line Identification Restriction (to be avoided, see clause 8);

    c)   Connected Line Identification Presentation;

    d)   Connected Line Identification Restriction;

- Direct Dialling-In supplementary service;

- Malicious Call Identification supplementary service;

- Multiple Subscriber Number supplementary service;

- Subaddressing supplementary service;

- User-To-User Signalling supplementary service.

B.4   Supplementary services applicable with restriction

The Advice of Charge services:

- Advice of Charge: charging information at call set-up time;

- Advice of Charge: charging information during the call;

- Advice of Charge: charging information at the end of a call;

can be used, however it is recommended to display the total of charges incurred for all B-channels.

The Call Waiting supplementary service is only used for Call 1 (initial channel).

The Call Completion to Busy Subscribers supplementary service is only used for Call 1 (initial channel).

Concerning the Closed User Group supplementary service, in order to ensure the integrity of the Closed User Group supplementary service when fall-back occurs, the Closed User Groups subscribed-to for the videoconference teleservice also apply to the fall-back connection.

For the call diversion supplementary services, the call is diverted as a call using the videoconference teleservice. If the calling user has indicated that fall-back is allowed, the fall-back indication is included when the diverted part of the call is processed. Indication that fall-back has occurred is sent to the calling user independently of the diverting user's subscription option (calling user informed or not that the call has been diverted).

Annex C (informative): Relation between standards and recommendations relevant for the videoconference service

The various applications which may be used during a videoconference are supported by different protocols stacks summarized in the following diagram.
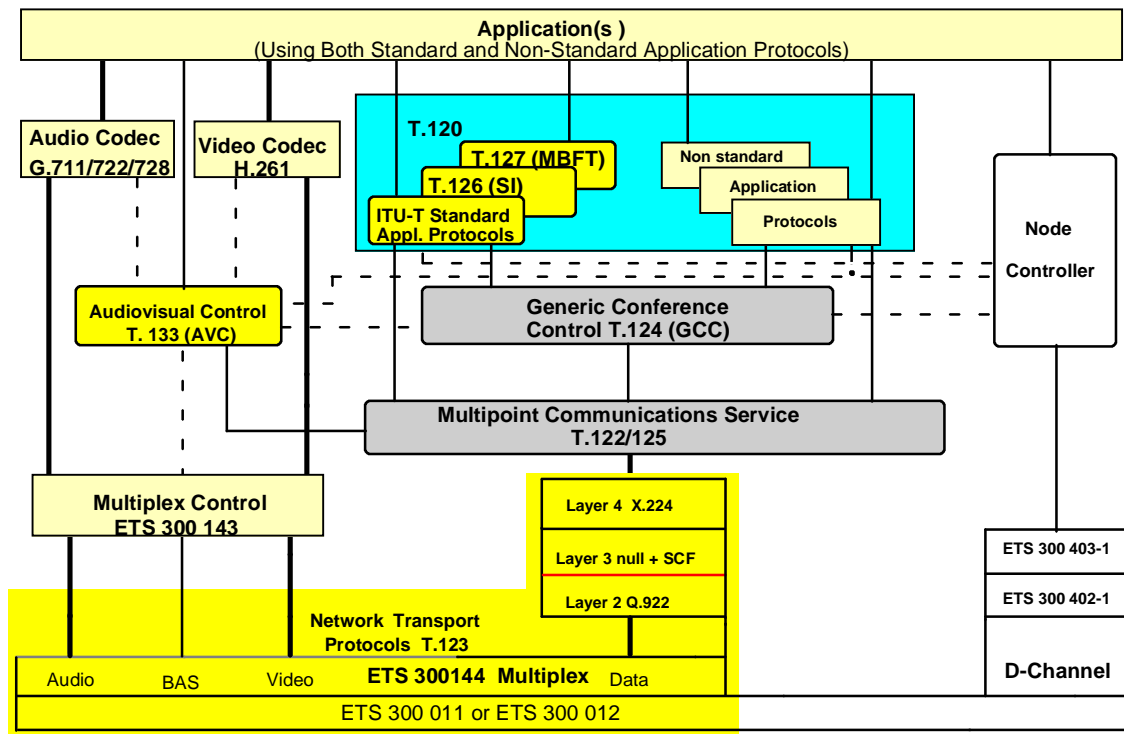

Figure C.1

Some of these protocols are still in their development phase in the ITU-T (cf. bibliography in annex D).

Annex D (informative): Bibliography

The following documents have been introduced during the preparation of this ETS as informative references, which may be useful to improve the understanding of its contents.

- ETS 300 011 (1985)92): "Integrated Services Digital Network (ISDN); Primary rate user-network interface, Layer 1 specification and test principles".

- ETS 300 012 (1985)92): "Integrated Services Digital Network (ISDN); Basic user-network interface, Layer 1 specification and test principles".

- ETS 300 267-1 (1985)94): "Integrated Services Digital Network (ISDN); Telephony 7 kHz and videotelephony teleservices, Digital Subscriber Signalling System No. one (DSS1) protocol, Part 1: Protocol specification".

- ETS 300 402-1: "Integrated Services Digital Network (ISDN); Digital Subscriber Signalling System No. one (DSS1); User-network interface data link layer specifications; Part 1: General aspects [ITU-T Recommendation Q.920 (1985)93) modified]".

- ETS 300 403-1: "Integrated Services Digital Network (ISDN); Digital Subscriber Signalling System No. one (DSS1); User-network interface layer 3 specification for basic call control, Part 1: Protocol specification [ITU-T Recommendation Q.931 (1985)93) modified]".

- ETS 300 481 (1985)96): "Integrated Services Digital Network (ISDN); B-Channel aggregation unit; Procedures and terminal requirements".

- CCITT Recommendation F.730 (1985)92): "Videoconference service - General".

- ITU-T Recommendation H.234 (1985)93): "Encryption key management and authentication system for audiovisual services".

- ITU-T Recommendation T.120: "Data Protocols for Multimedia Conferencing".

- ITU-T Recommendation T.121: "Generic application template".

- Draft ITU-T Recommendation T.130: "Architecture for Real Time Multimedia Conferencing".

- Draft ITU-T Recommendation T.131: "Network Specific Mappings for RTLM".

- Draft ITU-T Recommendation T.132: "Real Time Link Management".

  Draft ITU-T Recommendation T.133: "Audio Video Control Application Protocol".

- CCITT Recommendation E.164 (1985)91): "Numbering plan for the ISDN era".

- CCITT Recommendation I.130 (1985)88): "Method for the characterization of telecommunication services supported by an ISDN and network capabilities of an ISDN".

- ITU-T Recommendation I.112 (1985)93): "Vocabulary of terms for ISDNs".

- ITU-T Recommendation I.210 (1985)93): "Principles of telecommunication services supported by an ISDN and the means to describe them".

- ITU-T Recommendation I.221 (1985)93): "Common specific characteristics of services".

- ITU-T Recommendation X.224 (1985)93): "Protocol for providing the OSI connection-mode transport service".

- ITU-T Recommendation X.225 (1985)94): "Information technology - Open Systems Interconnection - Connection-oriented session protocol: Protocol specification".

- ITU-T Recommendation X.226 (1985)94): "Information technology - Open Systems Interconnection - Connection-oriented presentation protocol: Protocol specification".

- CCITT Recommendation X.227 (1985)92): "Connection-oriented protocol specification for the association control service element".

- CCITT Recommendation Q.922 (1985)92): "ISDN data link layer specification for frame mode bearer services".

History

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