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CCITT

F.721

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TELEMATIC, DATA TRANSMISSION, ISDN BROADBAND, UNIVERSAL, PERSONAL TELECOMMUNICATIONS AND TELECONFERENCE SERVICES

OPERATIONS AND QUALITY OF SERVICE

VIDEOTELEPHONY TELESERVICE FOR ISDN

Recommendation F.721

FOREWORD

The CCITT (the International Telegraph and Telephone Consultative Committee) is a permanent organ of the International Telecommunication Union (ITU). CCITT is responsible for studying technical, operating and tariff questions and issuing Recommendations on them with a view to standardizing telecommunications on a worldwide basis.

The Plenary Assembly of CCITT which meets every four years, establishes the topics for study and approves Recommendations prepared by its Study Groups. The approval of Recommendations by the members of CCITT between Plenary Assemblies is covered by the procedure laid down in CCITT Resolution No. 2 (Melbourne, 1988).

Recommendation F.721 was prepared by Study Group I and was approved under the Resolution No. 2 procedure on the 4th of August 1992.

CCITT NOTES

1) In this Recommendation, the expression “Administration” is used for conciseness to indicate both a telecommunication administration and a recognized private operating agency.

2) A list of abbreviations used in this Recommendation can be found in Annex A.

Recommendation F.721

VIDEOTELEPHONY TELESERVICE FOR ISDN

(1992)

The CCITT,

considering

(a) that considerable efforts have been undertaken worldwide in order to develop videotelephone equipment based on rapid improvements in the quality of video codec algorithms;

(b) that videotelephones of some manufacturers are already available on the market;

(c) that first trials of videotelephony have been performed nationally as well as internationally;

(d) that a number of countries intend to introduce the videotelephone service as soon as possible;

(e) that the ISDN will be an appropriate network for providing the videotelephony teleservice;

(f) that ISDN is offered in a number of countries,

recognizes

the need for a standardized international videotelephony teleservice which will guarantee the compatibility of videotelephones on a worldwide basis and therefore,

recommends

that the videotelephony teleservice, where implemented, respect the requirements stated in this Recommendation.

1 Definition

The **videotelephony teleservice** is a symmetrical, bidirectional, realtime, audiovisual teleservice in which speech and moving pictures are interchanged by means of one or two B-channels, using 64 kbit/s circuit-mode connections in the ISDN. The picture information transmitted is sufficient for adequate representation of fluid movements of a person displayed in head and shoulders view.

2 Description

2.1 *General description*

The videotelephony teleservice is defined as a fully standardized ISDN teleservice following the principles given in Recommendation I.210.

Two cases can be identified for the videotelephony teleservice:

Case I: Videotelephone based on using one circuit-mode 64 kbit/s connection.

Case II: Videotelephone based on using two circuit-mode 64 kbit/s connections.

For case I the 64 kbit/s connection carries both speech and video information; for case II the first connection either carries speech or both speech and some video information and the second connection carries video information.

The basic videotelephony teleservice is characterized by the transmission of moving pictures displayed continuously in colour, simultaneously with the speech of the persons involved in the call (generally two in the case of a point-to-point connection).

The speech quality of this teleservice must be at least as good as that applicable to the telephony teleservice in the 64 kbit/s ISDN based on bandwidths of 3.1 kHz or 7 kHz respectively.

The videotelephony teleservice shall allow the communication between:

– two users (e.g. terminals) in a point-to point configuration via the ISDN over one or two circuit mode 64 kbit/s connection(s);

– three or more users in a multipont configuration as invoked by some supplementary services.

Videotelephone terminals must be capable of supporting the telephony teleservice.

An essential feature of the service is that, besides videotelephony, it also provides to the user 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 public switched telephone network (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 (2 × 64 kbit/s).

Retention timer: This timer specifies the amount of time that the network retains the call information of the original call upon encountering busy or being released. This timer is a network provider option. The value of this timer 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 *Provision/withdrawal*

3.1.1 This teleservice may be provided after prior arrangement with the service provider or be generally available.

3.1.2 As a service provider option, the videotelephony teleservice can be offered with several subscription options which apply separately to each ISDN number, all or a group of ISDN numbers on the interface. For each subscription option, only one value can be selected.

Subscription option	*Value*
Maximum number of information channels available at the called user	*m*, where *m* is not greater than the number of information channels on the interface
Maximum number of total calls presented simultaneously at the called user	*n*, where *n* is not greater than the number of information channels on the interface

The called user is identified by either one ISDN number, all or a group of ISDN numbers on the interface.

3.2 *Normal procedures*

3.2.1 *Activation/deactivation/registration*

Not applicable.

3.2.2 *Invocation and operation*

3.2.2.1 *Originating the service (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 set up 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 Recommendations H.221 and H.242.

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

Call 2 is invoked by the calling terminal when call 1 is in the active phase and after end-to-end mode initialization 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 synchronization is achieved according to Recommendation H.221.

The in-band protocol shall be established according to 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.

The call request procedure from the user’s point of view 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 3.1 kHz telephony 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, 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.

3.2.2.2 *Indications during call set-up 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.

A called user has the control whether his picture is transmitted 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.

3.2.2.3 *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.

3.2.2.4 *Change of terminal communication mode*

As a consequence of end-to-end integrity on videotelephony teleservice, 7 kHz telephony teleservice and some 3.1 kHz telephony teleservice calls (i.e. where no speech processing is performed by the network), it will be possible to use the B-channel protocols given in Recommendations G.725 and H.242.

Depending on the terminal capabilities, it may be possible to change between the following communication modes according to Tables 2/H.320 and 3/H.320:

– 3.1 kHz speech (Recommendation G.711);

– 7 kHz speech (Recommendation G.722);

– different videotelephone terminal modes.

Note 1 – The user may be required to establish additional calls in some cases.

Note 2 – As an option, in some circumstances establishment of a videotelephone call can be based on call set up 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.

3.2.3 *Interrogation*

Not applicable.

3.3 *Exceptional procedures*

3.3.1 *Failure situations due to user error*

i) A user inputting an improper service request shall be given an appropriate failure indication by the network and the call set-up shall be ceased.

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

3.3.2 *Failure situations due to called user state*

i) 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 set-up shall be ceased.

ii) 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 set-up shall be ceased.

iii) 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 set-up shall be ceased.

3.3.3 *Failure situation due to network conditions*

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

3.3.4 *Failure situations due to called user state and/or network condition*

A user attempting to establish a call but meeting call failure situations due to network conditions (e.g. congestion) or called user state (e.g. NDUB or UDUB), can have call information retained for the duration of the retention timer.

3.3.5 *Decreasing of service quality due to fall-back*

In the case of decreasing of service quality due to audio and/or visual fall-back, appropriate indication should be given to both users, even if problems occur in one direction only.

4 Network capabilities for charging

Charging capabilities are outside the scope of this Recommendation.

5 Intercommunication and interworking considerations

5.1 *Intercommunication/interworking with other terminals*

Intercommunication with 3.1 kHz and 7 kHz ISDN terminals or interworking with PSTN shall be offered.

5.1.1 *General principles*

The videotelephony service shall include voice encoding according to Recommendation G.711.

It may include additional speech encoding as an optional feature.

The following fundamental requirements shall be fulfilled:

i) The user of a videotelephone terminal shall be able to establish calls to 3.1 kHz and 7 kHz telephone terminals (if the 7 kHz capability is supported) connected to the ISDN and to telephone terminals connected to the PSTN. Optionally, it should be able to reach other ISDN audiovisual terminals.

ii) 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 functions 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.

5.1.2 *Fall back procedure*

5.1.2.1 *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 be requires interworking/fall-back to the 3.1 kHz telephony teleservice. 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 recognize 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 cancellers.

- 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.

5.1.2.2 *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 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.

5.2 *Interworking with private ISDNs*

If the called user is on 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/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*

a) 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 that 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 telephone call is established first, attributes of 3.1 kHz telephony teleservice bearer capability apply (telephone call instead of call 1).

b) 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 *Access attributes*

a) Call 1

8) Acces channel and rate:

– D(16) or D(64) for signalling, B(64) for user information.

9.1 Signalling access protocol, layer 1:

– Recommendations I.430/I.431.

9.2 Signalling access protocol, layer 2:

– Recommendation Q.921.

9.3 Signalling access protocol, layer 3:

– Recommendation Q.931.

9.4 Information access protocol, layer 1:

– Recommendations H.221, G.711, G.722 (option), H.242, H.261, AV.254.

9.5 Information access protocol, layer 2.

9.6 Information access protocol, layer 3.

Note that in the case where fall-back to the 3.1 kHz telephony teleservice occurs or if optionally a telephone call is established first, attribute 9.4 has the following values: Recommendations I.430/I.431, G.711.

b) Call 2

8) Acces channel and rate:

– D(16/64) for signalling, B(64) for user information.

9.1 Signalling access protocol, layer 1:

– Recommendations I.430/I.431.

9.2 Signalling access protocol, layer 2:

– Recommendation Q.921.

9.3 Signalling access protocol, layer 3:

– Recommendation Q.931.

9.4 Information access protocol, layer 1:

– Recommendations H.221, H.242, H.261, AV.254.

9.5 Information access protocol, layer 2.

9.6 Information access protocol, layer 3.

6.4 *High layer attributes*

a) Call 1

10) Type of user information:

– Speech (telephony), video, data, audiovisual (information).

11) Layer 4 protocol functions:

– Recommendation H.221.

12) Layer 5 protocol functions:

– Recommendation H.242.

13) Layer 6 protocol functions:

– Recommendations G.722 (option), G.711, H.261.

14) Layer 7 protocol functions.

Note that 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 "Recommendation G.711".

b) Call 2

10) Type of user information:

– video.

11) Layer 4 protocol functions:

– Recommendation H.221.

12) Layer 5 protocol functions:

– Recommendation H.242.

13) Layer 6 protocol functions:

– Recommendation H.261.

14) Layer 7 protocol functions.

6.5 *General attributes*

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 bandwidths of 3.1 kHz or 7 kHz.

8.3 *Picture quality*

Optimization 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. A 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 terrestrial path while the other is routed via satellite. In this case, the resynchronization 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;

– other videotelephony modes;

– audiovisual services;

– others for further study.

10 Operational and commercial aspects

For further study.

ANNEXE A

(to Recommendation F.721)

Alphabetical list of abbreviation used
in this Recommendation

ISDN	Integrated services digital network
NDUB	Network determined user busy
PSTN	Public switched telephone nework
UDUB	User detrmined user busy