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Technical Specification

3rd Generation Partnership Project;
Technical Specification Group Services and System Aspects;
Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN);
Videotelephony over NGN Service Description
(Release 8)

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3GPP

Postal address

3GPP support office address

650 Route des Lucioles - Sophia Antipolis
Valbonne - FRANCE
Tel.: +33 4 92 94 42 00 Fax: +33 4 93 65 47 16

Internet

http://www.3gpp.org

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Foreword

This Technical Specification (TS) was been produced by ETSI Technical Committee Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN) and originally published as ETSI TS 181 001 [4]. It was transferred to the 3rd Generation Partnership Project (3GPP) in May 2008.

The contents of the present document are subject to continuing work within the TSG and may change following formal TSG approval. Should the TSG modify the contents of the present document, it will be re-released by the TSG with an identifying change of release date and an increase in version number as follows:

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where:

x the first digit:

1 presented to TSG for information;

2 presented to TSG for approval;

3 or greater indicates TSG approved document under change control.

y the second digit is incremented for all changes of substance, i.e. technical enhancements, corrections, updates, etc.

z the third digit is incremented when editorial only changes have been incorporated in the document.

Introduction

TISPAN NGN bases the provision of real-time communication services on the 3GPP specified IMS. This is a flexible tool that can provide a wide range of service interactions between terminals and networks. Videotelephony provides a specific use of the generic capabilities of the IMS. The present document provides requirements to investigate the capabilities of the IMS to provide Videotelephony, and may be used for a "gap analysis" of the features and capabilities provided by the IMS.

The specific services and interactions described in the present document are based on other service specifications. The IMS is an inherently multimedia service control platform, and should be able to provide all of these services. The interactions between the user and the terminal and the terminal and the network are not provided.

The services and capabilities provided by a TISPAN NGN are described in TS 181 005 [2].

The requirements for services, described in the present document, also take account of the interactions required between interconnected networks - both between NGN and the interconnection of NGN with legacy networks.

The aim of the present document is to assist network operators and service providers to deploy NGN multimedia services.

1 Scope

The present document defines the requirements for a videotelephony service which may be supported by a TISPAN NGN. These requirements form the basis for the definition of network capabilities.

The present document provides interoperability service requirements for interconnection between existing networks and a TISPAN NGN, and between TISPAN NGN.

The present document only provides requirements for IP multimedia based networks. Services provided by a TISPAN NGN to support legacy terminals and interfaces (PSTN/ISDN emulation) are defined in existing PSTN/ISDN documents. Requirements for PSTN/ISDN emulation are out of scope of the present document and are described in other documents.

The applicability of PSTN/ISDN simulation services to the videotelephony service requirements are defined in the present document.

The requirements in the present document are described from the user point of view. The requirements do not take into account capabilities of existing protocols defined for the IMS. The evolution or modifications to these protocols are beyond the scope of the present document.

NOTE: The present document uses the term "NGN" only in the context of TISPAN.

2 References

The following documents contain provisions which, through reference in this text, constitute provisions of the present document.

- References are either specific (identified by date of publication, edition number, version number, etc.) or non-specific.
- For a specific reference, subsequent revisions do not apply.
- For a non-specific reference, the latest version applies. In the case of a reference to a 3GPP document (including a GSM document), a non-specific reference implicitly refers to the latest version of that document *in the same Release as the present document*.

[1] ETSI TS 181 002: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); Multimedia Telephony with PSTN/ISDN simulation services".

[2] ETSI TS 181 005: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); Services and Capabilities Requirements".

[3] ETSI TR 180 000: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); NGN Terminology".

[4] ETSI TS 181 001 (V1.1.1): "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); Videotelephony over NGN; Stage 1 service description".

3 Definitions and abbreviations

3.1 Definitions

For the purposes of the present document, the following terms and definitions given in TR 180 000 [3] apply:

- originating party;
- terminating party.

3.2 Abbreviations

For the purposes of the present document, the following abbreviations apply:

ACR	Anonymous Communication Rejection
AoC	Advice of Charge
CB	Communication session Barring
CCBS	Communication Completion on Busy Subscriber
CDIV	Communication DIVersion
CIF	Common Intermediate Format
CONF	CONFerence
CW	Communication Waiting
ECT	Explicit Communication Transfer
HOLD	communication HOLD
IMS	IP Multimedia Subsystem
IP	Internet Protocol
ISDN	Intergrated Services Digital Network
MCID	Malicious Communication IDentification
MWI	Message Waiting Indication
NGN	Next Generation Network
OIP	Originating Identification Presentation
OIR	Originating Identification Restriction
PCM	Pulse Coded Modulation
PIP	PIcture and Picture
PSTN	Public Swithched Telephone Network
QCIF	Quarter Common Intermediate Format
RTP	Real-time Transport Protocol
SMS	Short Message Service
TIP	Terminating Identification Presentation
TIR	Terminating Identification Restriction

4 Service description

Videotelephony is a real time conversational service using video media and audio or other types of media. The service is assumed to be applicable only to dedicated terminal equipment with video capabilities. The videotelephony service may be considered as a specific instance of an IP multimedia service, or as part of the Multimedia Telephony with PSTN/ISDN Simulation Services.

5 Procedures

5.1 General service requirements

The videotelephony service shall conform to the requirements in TS 181 005 [2], clause 5.2.1. In particular, the following capabilities described in TS 181 005 [2] shall be applicable to this service:

- Handling of an incoming session (by the terminating entity).
- Presentation of session originator identity.
- Negotiation of an incoming session.
- Accepting or rejecting an incoming session.
- Handling of an ongoing session.
- User modification of media in an ongoing session.
- Presentation of identity of connected-to party of a session.

- Ending a session.
- Capability Negotiation.
- Redirecting of IP Multimedia sessions.
- Regulatory service requirements.
- Lawful Intercept requirements.
- Emergency service requirements.

NOTE: Emergency Service support for video may be limited.

- Malicious Communication Identity service requirements (MCID).
- Anonymous Communications Rejection service requirements (ACR).

5.2 Registration, provisioning, activation, invocation and withdrawal

The requirements for Registration/Log-out, provisioning, activation, invocation and withdrawal shall be as described in TS 181 005 [2].

5.3 Charging and accounting

The requirements for charging and accounting shall be as described in TS 181 005 [2].

6 Quality of service requirements

6.1 General

The quality of service requirements shall be as described in TS 181 005 [2].

The following parameters may also be taken into account: Overall Delay, Delay Variation, Differential Delay between sound and image, Sound Quality, Image Quality, Echo Cancellation, Sensitivity to Packet Loss, etc.

6.2 Video quality

Four profiles are defined in table 1. Other enhancement or additional functions may be added.

Table 1: Video quality profile

Profile	Description
A	Basic Videotelephony: PCM or equivalent telephony audio, QCIF or CIF video with limited movements capability.
B	Enhanced Videotelephony: wideband audio, CIF video. The service should be usable for sign language and lip-reading and should provide adequate representation of the fluid movements of a person displayed in head and shoulders view. Facial expressions shall be clearly recognized.
C	Television Broadcasting quality.
D	High Definition Television quality.

6.3 Audio quality

Videotelephony does not place any additional requirements for audio quality.

7 Interworking considerations

7.1 General

An NGN shall support the interoperability of the videotelephony service with non NGN network services and vice versa. This includes interworking the audio component to the PSTN/ISDN and vice versa. The video component shall only be interworked if video is supported in the non NGN network.

The scope of this interworking may result in a limited service capability.

7.2 Interworking with emulation services

An NGN shall support the interoperability of the Multimedia Telephony with PSTN/ISDN Simulation Services with the services provided by the NGN PSTN/ISDN Emulation subsystems where both are deployed. The scope of this interworking may result in the same limited service capability as interworking with an existing PSTN/ISDN network.

8 Interactions with PSTN/ISDN simulation services

8.1 General

A network operator may choose to offer the PSTN/ISDN simulation services as described in TS 181 002 [1] in conjunction with this videotelephony service as part of an NGN deployment scenario. The following clauses detail the use of PSTN/ISDN simulation services with this videotelephony service for this scenario.

8.2 Mandatory services

8.2.1 Originating Identification Presentation (OIP)

The principles of the corresponding PSTN/ISDN simulation service defined in TS 181 002 [1] shall apply.

8.2.2 Originating Identification Restriction (OIR)

The principles of the corresponding PSTN/ISDN simulation service defined in TS 181 002 [1] shall apply.

8.2.3 Terminating Identification Presentation (TIP)

The principles of the corresponding PSTN/ISDN simulation service defined in TS 181 002 [1] shall apply.

8.2.4 Terminating Identification Restriction (TIR)

The principles of the corresponding PSTN/ISDN simulation service defined in TS 181 002 [1] shall apply.

8.2.5 Malicious Communication Identification (MCID)

The principles of the corresponding PSTN/ISDN simulation service defined in TS 181 002 [1] shall apply.

8.2.6 Anonymous Communication Rejection (ACR)

The principles of the corresponding PSTN/ISDN simulation service defined in TS 181 002 [1] shall apply.

8.3 Recommended Services

8.3.1 Communication DIVersion (CDIV)

In addition to the principles of the corresponding PSTN/ISDN simulation service defined in TS 181 002 [1] the following shall also apply:

User A requests a communication with user B, who has activated Communication Diversion towards user C. If either user A or C requests a video component to be added to the communication, the request will only be successful if it is a permitted option of all parties (i.e. the service option of user B shall be taken into account for a diverted communication).

8.3.2 Communication Waiting (CW)

The principles of the corresponding PSTN/ISDN simulation service defined in TS 181 002 [1] shall apply.

8.3.3 Communication HOLD (HOLD)

The principles of the corresponding PSTN/ISDN simulation service defined in TS 181 002 [1] shall apply.

8.3.4 Communication Baring (CB)

The principles of the corresponding PSTN/ISDN simulation service defined in TS 181 002 [1] shall apply.

8.3.5 Communication Completion on Busy Subscriber (CCBS)

The principles of the corresponding PSTN/ISDN simulation service defined in TS 181 002 [1] shall apply.

8.3.6 Message Waiting Indication (MWI)

The principles of the corresponding PSTN/ISDN simulation service defined in TS 181 002 [1] shall apply.

8.4 Optional services

8.4.1 CONFerence (CONF)

In addition to the principles of the corresponding PSTN/ISDN simulation service defined in TS 181 002 [1] the following shall also apply.

A videotelephony service shall allow a conference service. If a correspondent's terminal doesn't allow videotelephony service, this conference shall be an audio conference for this correspondent and a video + audio conference for the other correspondents.

A user shall be able to create a videoconference at any time during a communication.

8.4.2 Advice of Charge (AoC)

The principles of the corresponding PSTN/ISDN simulation service defined in TS 181 002 [1] shall apply.

8.4.3 Explicit Communication Transfer (ECT)

In addition to the principles of the corresponding PSTN/ISDN simulation service defined in TS 181 002 [1] the following shall also apply:

When the transferring user A, transfers a communication from user B to user C, which may be:

- just requested;
- in progress of being established; or
- already established.

The resulting communication between user B to user C shall retain the media components common to the original communications. If there are no common components, the ECT will be refused.

8.4.4 Reverse charging

The principles of the corresponding PSTN/ISDN simulation service defined in TS 181 002 [1] shall apply.

Annex A (informative): Guidelines

A.1 Terminal requirements

The Videotelephony service needs to have a friendly user interface allowing an easy use in a family context.

The user shall be able to see the local video and far-end video.

NOTE: LS shall be send to HF and AT concerning terminal requirements before deleting this annex from the present document.

A.1.1 Service options

The following applications may also be supported:

- visual guideline to assist the user to use the videotelephony service;
- transmission of other data such as documents, texts (Instant Messaging, chat, SMS), pictures, music, etc.;
- far-end camera control;
- conference;
- camera with adjustable shooting and zoom (additional cameras and microphones);
- when an external and an internal audio sources are used, the audio signal must be a mix of both and the user should manage it.

A.1.2 Communication set-up

A dedicated information shall indicate to the user the communication status, in particular:

- video-audio mode: both users can see each other;
- mute mode: the user audio signal is not transmitted to the other party;
- video privacy mode: the user video signal is not transmitted to the other party.

The dedicated information can be a visual and/or an audible information.

NOTE: Out of communications, the presence of a camera device can disturb the user. It should be may to hide it physically.

The detailed procedure of the communication establishment depends on the type of terminal used.

Basically the procedure shall be consistent with the existing procedure for a NGN terminal communication and shall be as simple as possible in order to achieve good acceptance:

- the originating party is aware of being connected;
- the originating party indicates the Destination Identification;
- the originating party receives an acknowledgement that the network is able to process the communication;
- the destination party receives an indication of the arrival of an incoming videotelephony communication;
- the originating party receives an indication that the communication has been offered;

- the destination party chooses to be heard or/and to be seen by the originating party (audio mode or/and video mode);
- the destination party answers the communication;
- the bidirectional communication is established (opening of the RTP ports).

Two options shall be possible for the communication establishment:

- the communication is first established in audio mode and then switches to in video mode;
- or either the communication is established directly in video-audio mode.

A.1.2.1 Communication setup in audio mode

In this mode, the communication shall be established first in audio mode and then only will change to video mode upon request of the participants.

A.1.2.2 Communication setup in mute mode

In this mode, the communication shall be established first in video mode and then only will change to video-audio mode upon request of the participants.

A.1.2.3 Communication setup in video-audio mode

In this mode, the communication shall be established directly in video associated with audio: the originating party sees and hears the destination party right from the start of the communication as well as that the destination party sees and hears the originating party right from the start.

Speech and image shall be connected simultaneously and synchronized.

Independently of communication, it should always be possible to display the local image (PIP: PIcture and Picture function).

If the received communication do not support video mode (for example if the communication is received from a legacy terminal), the originating party shall be informed by dedicated information his correspondent receives the call in audio mode.

A.1.3 Communication release

The procedure for terminating a communication shall be the same as for legacy telephony.

A request to terminate the videotelephony communication can be generated by either of the users. If one user terminates the communication, the other shall be given a dedicated indication.

The audio and video connections shall be simultaneously closed when the user hangs up or the network releases the connections.

A.2 Change of communication mode

A.2.1 Mute

Each party shall be able to mute the video and/or audio stream at any time (before or during the communication).

A.2.2 Frozen

Each party shall be able to freeze the picture the terminal is sending to the remote. The remote terminal must display the frozen picture without blinks or impairments, even when the frozen picture is displayed on interlaced monitors such as TV screens.

The frozen image shall not be inversed (i.e. not a mirror image, in order to read a document for example).

A.2.3 Audio/video

The user should be able to switch easily between audio and video communication, either off-line (pre-configuration) or during the communication.

During the modification request, the appropriate media should be selected to allow the right service with full transparency for user. During modification request the previous service should stay active without stopping the communication and the new service selected should be negotiated in background.

Contrary to switch video to audio mode, for switching audio to video mode both parties shall accept the video mode. The switch should not result in a new communication service but the addition or deletion of media components to the existing communication.

A.2.4 External source

The user should be able to change the video source during the call or off-line (before calling) in order to share audio/video with his correspondent.

As soon as the user changes the video source, a dedicated information may warn the user that he has changes the source.

A.2.5 Synchronization

The videotelephony service shall be able to perform synchronization between audio and video to an effect that there is no or acceptable human perceivable lack of lip-synchronization.

A.2.6 Other

The user can choose a screen on his terminal. If no screen is chosen, a default screen is displayed.

A.3 Use cases

Videotelephony scenarios include:

- face-to-face conversation with both audio and video;
- communication of speech and hearing impaired persons using sign language;
- remote video surveillance (e.g. home security; remote baby-monitoring);
- transfer of moving scenes or documents with drawings and text;
- share of documents to work in real time;
- face-to-face conversation with simultaneous transfer of picture data such as images, documents and files of other kinds;
- remote consultation;
- remote diagnosis in telemedicine;
- participation in a videoconferencing.

Annex B (informative): Change history

Change history							
Date	TSG #	TSG Doc.	CR	Rev	Subject/Comment	Old	New
2006-03					Published as ETSI TS 181 001 (from TISPAN)		1.1.1
2008-06					Endorsement as 3GPP TS 22.401	1.0.0	1.1.2

Change history											
TSG SA#	SA Doc.	SA1 Doc	Spec	CR	Rev	Rel	Cat	Subject/Comment	Old	New	WI
SA#40	SP-080312	S1-080733	22.401	-	-	7		Presentation for one-step-approval of TS 22.401	1.1.2	7.0.0	FBI-TIS1
SA#41	SP-080538	-	22.401			8		Raised to Rel-8 to solve TISPAN concerns	7.0.0	8.0.0	TEI8