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Telepresence using the IP Multimedia (IM)
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1 Scope

The present document provides the protocol details for telepresence using the IP Multimedia (IM) Core Network (CN) subsystem based on the Session Initiation Protocol (SIP), the Session Description Protocol (SDP), the Binary Floor Control Protocol (BFCP) and the ControlLing mUltiple streams for tElepresence (CLUE) controlling multiple streams for telepresence based on service requirements.

The present document addresses the areas of describing and negotiating IM session with multiple media streams based on the IM CN subsystem, including point to point calls as specified in 3GPP TS 24.229 [2] and multiparty conferences as specified in 3GPP TS 24.147 [3], to facilitate the support of telepresence.

The functionalities for conference policy control and the signalling between a MRFC and a MRFP are not specified in this document.

Where possible, the present document specifies the requirements for this protocol by reference to specifications produced by the IETF within the scope of SIP, SDP, CLUE and BFCP, either directly, or as modified by 3GPP TS 24.229 [2].

The present document is applicable to Application Servers (ASs), Multimedia Resource Function Controllers (MRFCs), Multimedia Resource Function Processors (MRFP) and User Equipment (UE) providing IM session supporting telepresence capabilities.

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] 3GPP TR 21.905: "Vocabulary for 3GPP Specifications".

[2] 3GPP TS 24.229: "Internet Protocol (IP) multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP); Stage 3".

[3] 3GPP TS 24.147: "Conferencing using the IP Multimedia (IM) Core Network (CN) subsystem; Stage 3".

[4] draft-ietf-clue-framework-22 (April 2015): "Framework for Telepresence Multi-Streams".

Editor's note: The above document cannot be formally referenced until it is published as an RFC.

[5] draft-ietf-clue-datachannel-09 (March 2015): "CLUE Protocol data channel".

Editor's note: The above document cannot be formally referenced until it is published as an RFC.

[6] draft-ietf-clue-signaling-05 (March 2015): "CLUE Signaling".

Editor's note: The above document cannot be formally referenced until it is published as an RFC.

[7] draft-ietf-mmusic-sctp-sdp-14 (March 2015): "Stream Control Transmission Protocol (SCTP) – Based Media Transport in the Session Description Protocol (SDP)".

Editor's note: The above document cannot be formally referenced until it is published as an RFC.

[8] IETF RFC 3264 (June 2002): "An Offer/Answer Model with Session Description Protocol (SDP)".

[9] draft-ietf-clue-data-model-schema-09 (April 2015): "An XML Schema for the CLUE data model".

Editor's note: The above document cannot be formally referenced until it is published as an RFC.

[10] 3GPP TS 23.218: "IP Multimedia (IM) session handling; IM call model; Stage 2".

3 Definitions, symbols and abbreviations

3.1 Definitions

For the purposes of the present document, the terms and definitions given in 3GPP TR 21.905 [1] and the following apply. A term defined in the present document takes precedence over the definition of the same term, if any, in 3GPP TR 21.905 [1].

IM session: An IP multimedia (IM) session is a set of multimedia senders and receivers and the data streams flowing from senders to receivers. IP multimedia sessions are supported by the IP multimedia CN Subsystem and are enabled by IP connectivity bearers (e.g. GPRS as a bearer). A user may invoke concurrent IP multimedia sessions.

Telepresence: A conference with interactive audio-visual communications experience between remote locations, where the users enjoy a strong sense of realism and presence between all participants by optimizing a variety of attributes such as audio and video quality, eye contact, body language, spatial audio, coordinated environments and natural image size.

3.2 Symbols

3.3 Abbreviations

For the purposes of the present document, the abbreviations given in 3GPP TR 21.905 [1] and the following apply. An abbreviation defined in the present document takes precedence over the definition of the same abbreviation, if any, in 3GPP TR 21.905 [1].

AS	Application Server
BFCP	Binary Floor Control Protocol
CLUE	Controlling Multiple streams for Telepresence
DTLS	Datagram Transport Layer Security
IMS	IP Multimedia CN subsystem
MCC	Multiple Content Capture
MRF	Multimedia Resource Function
MRFC	Multimedia Resource Function Controller
MRFP	Multimedia Resource Function Processor
SCTP	Stream Control Transmission Protocol
SDP	Session Description Protocol
SIP	Session Initiation Protocol
TP	TelePresence
TP UE	TelePresence User Equipment
UE	User Equipment

4 Telepresence overview

4.1 General

As an architectural framework for provision of IP multimedia services, IMS is capable of delivering various service functionalities and easing to integrate with new kinds of application, such as telepresence. Compared to traditional video conference, telepresence is a communication system with multiple cameras, microphones and screens that has the characteristics of gaze direction, eye contact, spatial audio and scaling images to true size, which in all to achieve the immersive "being there" experience for participants.

IMS is using the IETF defined session control mechanism with the inherited capability to negotiate multiple multimedia streams in one single session, which could be applied as a basis for supporting telepresence in IMS that always has the necessity of producing and rendering various media streams with high qualities among the involved parties, even in the point to point case.

Based on the existing procedures as specified in 3GPP TS 24.229 [2] and 3GPP TS 24.147 [3], this specification introduces updates and enhancements for IMS by incorporating CLUE with SIP, SDP and BFCP to facilitate controlling multiple spatially related media streams in an IM session supporting telepresence.

To provide a "being there" experience for conversational audio and video session between remote locations, a variety of information needs to be coordinated, such as:

- audio and video spatial information;
- information to enable eye contact, gaze awareness, body language and natural image size; and
- information to coordinate the environments;

4.2 Spatial information

A spatial relationship is representative of the arrangement in space of two or more objects in the same capture scene, in contrast to retain in time or other relationships. It involves mainly both video and audio sources in telepresence conferencing system.

For video, a spatial description of source video images sent in video streams, which includes the order of images in the actual captured scene and may be in two or three dimensions, enables a reproduction of the original scene at the receiver side. For audio, a spatial description of source audio including the point of capture, line of capture and capture sensitivity pattern (i.e. omni, shotgun, cardioid, hyper-cardioid) enable a reproduction at the receiver side in a spatially correct manner. Spatial matching may also be needed between audio and video streams coming from a specific party.

When advertising video and audio media captures in an IM session supporting telepresence, a TP UE as well as a TP enabled conference focus sends spatial information, e.g. physical dimensions of the capture area for each video capture, and associated audio captures spatial information. This allows the receiving party to coordinate the capture scenarios and perform a proper rendering. Consider a TP UE of a typical triple-screen/camera system as example, of which each camera can provide one video capture for each 1/3 section of the room. Each capture has spatial information to indicate the scope of view, where a capture showing a zoomed out view of the whole room has the spatial information indicating a full global view.

4.3 Media information

The media information is enhanced in an IM session supporting telepresence by introducing the source components of the media capture, e.g. the original media captures like a camera or a composed media captures indicating a mix of audio streams, a composed or switched media capture indicating the dynamic or most appropriate subset of a "whole".

The enhanced media information enables a sender to describe the sources and a receiver to choose which it wants to see. Taking the above example in subclause 4.2, the TP UE can further provide a single capture representing the active speaker based on voice activity detection, and a single capture representing the active speaker with the other two captures composed as a picture in picture. The media information can be used to distinguish the media captures from each other.

Further media information may also be needed, such as simultaneity constraints. For example, a room camera have two options which are zoomed-in view and zoomed-out view, but there is no way to get them simultaneously.

4.4 Meeting description

Meeting description includes view information, language information, person information and type, as described below which enable the receivers to choose and render different captures.

- View information: indicates a physical or logical region as captured;
- Language information: used in case of multi-lingual and/or accessible conferences;

- Person information: provides specific information about people participating within a multi-media conference; and
- Person type: indicates the type of people participating within a multi-media conference with respect to the meeting agenda.

In addition, there may be some descriptive information which contains a relative priority between different captures, embedded textual information, or additional complementary information.

4.5 Presentation

Presentation indicates resource sharing from one or more specific devices, including slides, video, data and etc. Presentations may have unfixed sources, which varies in placement and can be seen by all the involved parties.

4.6 Information usage

The information detailed above may be used to obtain a better experience during an IM session between involved parties with different capabilities, such as different number of devices, different picture aspect ratios, or different number of media streams for sending and receiving.

The usage of the information depends on the application scenarios. The TP UE described in the example of subclause 4.2 can provide at least 6 video captures, the message for negotiation therefore needs to contain enough parameters to describe the characteristics of each capture in order to allow the receiving party to clearly differentiate the captures and provide a proper rendering, e.g. spatial view, media composition, person information and type, and etc.

Protocols adopted for IM session supporting telepresence, which exchanges above information among involved parties, enables interoperability by handling multiple streams in standardized way.

5 Protocol using SIP for telepresence

5.1 Introduction

SIP, as specified in 3GPP TS 24.229 [2] and 3GPP TS 24.147 [3], is used as the basic session control protocol to create an IM session supporting telepresence.

The usage of SIP for the point to point call supporting telepresence is following the procedures as specified in 3GPP TS 24.229 [2]. The usage of SIP for the multiparty conference supporting telepresence is following the procedures as specified in 3GPP TS 24.147 [3].

To support an IM session supporting telepresence, a "+sip.clue" Contact header field parameter is defined in clause 3 of draft-ietf-clue-signaling [6].

5.2 Functional entities

5.2.1 User Equipment (UE)

For the purpose of IM session supporting telepresence, the TP UE shall implement the role of a participant in both point to point call and multiparty conference as described in subclause 5.3.1.

5.2.2 Media Resource Function Controller (MRFC)

For the purpose of IM session supporting telepresence, the MRFC shall support the procedures as described in subclause 5.2.2 of 3GPP TS 24.147 [3].

5.2.3 Conferencing Application Server (Conferencing AS)

For the purpose of IM session supporting telepresence, the TP enabled conferencing AS shall implement the role of a conference focus, as described in subclause 5.3.2.

5.3 Role

5.3.1 Participant

During registration the TP UE follows the procedures specified in subclause 5.1.1 of 3GPP TS 24.229 [2] as a basis, and it shall include a "+sip.clue" Contact header field parameter in the SIP REGISTER request.

To establish an IM session supporting telepresence, the TP UE shall use the procedures as specified in subclause 5.1 of 3GPP TS 24.229 [2] for point to point call, and follow the conference procedures as specified in subclause 5.3.1 of 3GPP TS 24.147 [3] for multiparty conference, in both the TP UE shall include a "+sip.clue" Contact header field parameter in the SIP INVITE and SIP UPDATE requests and responses.

5.3.2 Conference Focus

The TP conference focus follows the procedures specified in subclause 5.3.2 of 3GPP TS 24.147 [3] appropriate to the functional entity in which the conference focus is implemented.

The TP conference focus shall include the "+sip.clue" Contact header field parameter when generating a SIP request or response for the establishment of an IM conference supporting telepresence.

6 Protocol using SDP for telepresence

6.1 Introduction

SDP, as specified in 3GPP TS 24.229 [2] and 3GPP TS 24.147 [3], is used to establish multimedia streams in an IM session supporting telepresence.

In an IM session supporting telepresence, each party in the session usually sends and receives multiple multimedia streams, which may not be symmetric due to their different capabilities for media production and rendering.

6.2 Functional entities

6.2.1 User Equipment (UE)

For the purpose of IM session supporting telepresence, the TP UE shall implement the role of a participant in both point to point call and multiparty conference as described in subclause 6.3.1.

6.2.2 Media Resource Function Controller (MRFC)

For the purpose of IM session supporting telepresence, the MRFC shall support the procedures described in subclause 6.2.2 of 3GPP TS 24.147 [3].

6.2.3 Conferencing Application Server (Conferencing AS)

For the purpose of IM session supporting telepresence, the TP enabled conferencing AS shall act as a conference focus, as described in subclause 6.3.2. The TP conferencing AS may implement the role of a conference participant as described in subclause 6.3.1.

6.3 Roles

6.3.1 Participant

6.3.1.1 General

In addition to the procedures specified in subclause 6.3.1, the TP UE shall support the point to point call procedures specified in subclause 6.1 of 3GPP TS 24.229 [2], and the TP UE shall support the conference procedures specified in subclause 6.3.1 of 3GPP TS 24.147 [3] appropriate to the functional entity in which the participant is implemented for multiparty conference.

6.3.1.2 Initiation of IM session supporting telepresence

6.3.1.2.1 Origination procedures

To initiate an IM session supporting telepresence, the TP UE shall:

- 1) generate an SDP offer in the SIP request, containing basic media streams and an establishment request for a DTLS/SCTP association used to realize a CLUE data channel. The initial SDP offer message negotiates the port and transport information for setting up the DTLS/SCTP association, via a separate SDP "m=" line with a UDP/DTLS/SCTP or TCP/DTLS/SCTP proto value, together with an SDP "sctp-port" attribute, and an SDP "dcmapp" attribute to indicate "CLUE" as the application protocol running over the data channel. The procedures for establishment of the DTLS/SCTP association via SDP can be found in draft-ietf-mmusic-sctp-sdp [7] and draft-ietf-clue-datachannel [5].
- 2) set up the DTLS/SCTP association used for a CLUE data channel with the remote party, after receiving the SDP answer with information for CLUE data channel establishment (e.g. an SDP "m=" line together with the SDP "sctp-port" attributes to describe a DTLS/SCTP association indicating the use of a data channel).

NOTE 1: If the TP UE receives the SIP response without a "+sip.clue" Contact header field parameter, which means the remote party does not support CLUE protocol, or if the TP UE receives the SDP answer without the support information for DTLS/SCTP establishment (e.g. by setting the port number to zero in the "m=" line describing a DTLS/SCTP association used to realize the CLUE data channel), which means the remote party does not support the DTLS/SCTP association to establish a CLUE data channel, then the session shall fall back to a normal IM session.

NOTE 2: In the point to point cases, the TP UE can initiate CLUE negotiation via the first SDP offer/answer when establishing an IM session or via the update SDP offer/answer during an ongoing IM session depending on local configuration.

As defined in draft-ietf-clue-signaling [6] and draft-ietf-clue-datachannel [5], only a single CLUE data channel is established within the same IM session supporting telepresence. Once the CLUE data channel is connected, the TP UE shall trigger an updated SDP offer/answer exchange to set up media streams for transmission of the media captures as negotiated in subclause 7.3. When constructing the SDP offer in both of point to point call and multiparty conference, the TP UE shall:

- 1) add a corresponding "m=" line for each encoding defined in CLUE messages, using an associated "label" attribute for each "m=" line to identify the each encoding in CLUE messages. In addition, mark the "m=" line(s) as send only with an "a=sendonly" attribute or as inactive with an "a=inactive" attribute used to represent the sender's encode ability and policies; and
- 2) use an SDP "group" session-level attribute to signal that the "m=" line(s) are CLUE-controlled.

When the SDP offer is sent, the following SDP negotiation procedures shall follow normal rules as defined in IETF RFC 3264 [8].

In the conference case, if a TP UE acting as a conference participant joins in the conference, the TP UE shall follow the above steps to set up an IM session supporting telepresence.

6.3.1.2.2 Termination procedures

When receiving a SIP request to initiate an IM session supporting telepresence, the TP UE shall:

- 1) send an SDP answer with information for CLUE data channel establishment (e.g. an SDP "m=" line with a UDP/DTLS/SCTP or TCP/DTLS/SCTP proto value, together with an SDP "sctp-port" attribute, and an SDP "dcmapp" attribute to indicate "CLUE" as the application protocol running over the data channel), if telepresence is supported for the session, i.e. by including a "+sip.clue" Contact header field parameter in the SIP response.

NOTE: If the receiving UE does not support DTLS/SCTP for the request, it shall reject the request for DTLS/SCTP establishment (e.g. by setting the port number to zero in the "m=" line describing the DTLS/SCTP association used to realize to the CLUE data channel). The session shall then proceed as a normal IM session.

Once the CLUE data channel is connected, the TP UE shall perform the updated SDP offer/answer exchange to set up media streams for transmission of the media captures as negotiated in subclause 7.3.

In the conference case, when the TP UE acting as a conference participant responds to a request to join in the conference, it also shall follow the above steps to complete an IM session supporting telepresence.

6.3.1.3 Session update

When an ongoing IM session supporting telepresence is updated, interaction of CLUE protocol and SDP negotiations shall be performed according to clause 5 in draft-ietf-clue-signaling [6].

The update of an ongoing IM session supporting telepresence may result in a session fallback to a normal IM session, by disabling CLUE according to subclause 4.5.4.3 in draft-ietf-clue-signaling [6].

6.3.1.4 End of IM session supporting telepresence

To release an existing IM session supporting telepresence, the TP UE shall generate a SIP BYE request for the session in accordance to the procedures described in 3GPP TS 24.229 [2].

To cancel an IM session supporting telepresence currently being established, the TP UE shall send a SIP CANCEL request to end the session.

6.3.2 Conference Focus

6.3.2.1 General

In addition to the procedures specified in subclause 6.3.2, the conference focus shall support the procedures specified in subclause 6.3.2 of 3GPP TS 24.147 [3] appropriate to the functional entity in which the conference focus is implemented.

6.3.2.2 Initiation of IM session supporting telepresence

6.3.2.2.1 Origination procedures

The TP enabled conference focus shall follow the procedures as described in subclause 6.3.1 to initiate an IM conference supporting telepresence.

Once CLUE data channels between the TP enabled conference focus and TP UEs are connected, the TP enabled conference focus shall:

- 1) trigger an updated SDP offer/answer exchange to set up media streams for transmission of the media captures between individual TP UE and the TP enabled conference focus based on the received CLUE CONFIGURE messages; and
- 2) follow the procedures with TP UEs as described in subclause 6.3.1.2.

6.3.2.2.2 Termination procedures

In the conference case, when the TP UE acting as a conference participant requests to join in the conference, the TP enabled conference focus shall follow the steps related to CLUE data channel establishment as described in subclause 6.3.1.3, to complete an IM session supporting telepresence among the other conference participants.

6.3.2.3 Session update

The conference focus may initiate an update to the ongoing IM session supporting telepresence as specified in subclause 6.3.1.3.

When the conference focus receives an update message to the ongoing IM session supporting telepresence, the conference focus shall:

- 1) finalize the CLUE exchange and possible update of SDP offer/answer according to clause 5 in draft-ietf-clue-signaling [6]; and
- 2) if session update is needed between the TP enabled conference focus and other participants in the same conference, the TP enabled conference focus shall initiate the update procedures accordingly.

6.3.2.4 End of IM session supporting telepresence

The TP enabled conference focus shall use the procedures as specified in subclause 6.3.1.4 to end up an IM session supporting telepresence. The TP enabled conference focus may trigger the procedures upon receiving a REFER request to remove a conference participant from the ongoing IM conference supporting telepresence.

When the TP enabled conference focus is aware that a participant has left an IM conference supporting telepresence, the TP enabled conference focus shall terminate the session with the leaving participant following the procedure specified in subclause 5.3.2.6 of 3GPP TS 24.147 [3], and then:

- 1) initiate session update to remove media streams to those participants who have configured media information from the leaving participant; and
- 2) send CLUE ADVERTISEMENT messages to notify the remaining participants of the latest available media captures as described in subclause 7.3.1.2.

7 Protocol using CLUE for telepresence

7.1 Introduction

CLUE, as specified in draft-ietf-clue-framework [4], is used to advertise and configure audio and video components comprising the media flows in an IM session supporting telepresence.

A data channel for CLUE message, as defined in draft-ietf-clue-datachannel [5], is negotiated via the first INVITE message when creating an IM session supporting telepresence. With the establishment of that channel, the participants have consented to use the CLUE protocol mechanisms to determine the capabilities of the each of the endpoints with respect to multiple streams support. The following exchange of CLUE messages of each participant's "advertisement" and "configure" is to achieve a common view of media components sent and received in the IM session supporting telepresence. A corresponding SDP offer/answer may be needed to establish the media streams based on the user's choice in CLUE messages.

7.2 Functional entities

7.2.1 User Equipment (UE)

The TP UE shall support both of the two roles in CLUE protocol, i.e. media provider and media consumer, as described in subclauses 7.3.1 and 7.3.2.

Typically a TP UE acts as both media provider and media consumer in an IM session supporting telepresence. However, a TP UE may only act as a media provider if it only sends media streams but not to receive, or it may only act as a media consumer if it only receive media streams.

7.2.2 Media Resource Function (MRF)

An MRF is split into the two functional entities, i.e. Multimedia Resource Function Controller (MRFC) and Multimedia Resource Function Processor (MRFP), as specified in clause 8 of 3GPP TS 23.218 [10].

The two functional entities in an IM session supporting telepresence, as part of the TP enabled conference focus, together with a Conferencing AS shall support both of the two roles in CLUE protocol, i.e. media provider and media consumer, as specified in draft-ietf-clue-framework [4].

7.2.3 Conferencing Application Server (Conferencing AS)

A Conferencing AS as part of the TP enabled conference focus, together with an MRF, shall support both of the two roles in CLUE protocol, i.e. media provider and media consumer, as specified in draft-ietf-clue-framework [4].

NOTE: This document only considers the support of CLUE where the AS and the MRF are collocated.

7.3 Roles

7.3.1 Media provider

7.3.1.1 General

As specified in draft-ietf-clue-framework [4], a media provider representing an entity that intends to send CLUE-controlled media streams, shall send CLUE ADVERTISEMENT message to describe the content of the media and the media streams encodings via encoding groups and individual encodings as specified in clause 9 of draft-ietf-clue-framework [4]. A TP UE or a TP enabled conference focus acting as a media provider shall send CLUE ADVERTISEMENT message and receive the corresponding CLUE CONFIGURE message from the remote party. The detailed information and format of CLUE messages can be found in draft-ietf-clue-data-model-schema [9].

7.3.1.2 Media capture advertisement

Once the CLUE data channel is connected during an IM session supporting telepresence, the TP UE, or the TP enabled conference focus, acting as a media provider shall send a CLUE ADVERTISEMENT message with multiple media captures related to the video and audio content it can provide, and carry the identities of these media captures together with information elements to describe characteristics (see clause 4) of these media captures including video capture spatial information in order to allow the remote party to select the source(s) it wants to consume, according to draft-ietf-clue-framework [4].

The media provider may send a new CLUE ADVERTISEMENT message to update the media captures anytime during the ongoing session, as specified in draft-ietf-clue-framework [4]. In the conference case, the TP enabled conference focus acting as a media provider may further construct a new CLUE ADVERTISEMENT message to a TP UE in the conference based on the media capture information received from the other TP UEs.

When the media provider receives the CLUE CONFIGURE message, the media provider shall trigger an updated SDP offer to establish the media streams based on the latest received CLUE CONFIGURE message as described in subclause 6.3.1.

7.3.1.2.1 Multiple content switching and composition

To provide media switching or composition for multiple contents with respect to time and space during an IM session supporting telepresence, the acting media provider, either the TP UE or the TP enabled conference focus, shall include one or more Multiple Content Capture (MCC) that is composed of multiple individual captures in the CLUE ADVERTISEMENT message, as defined in subclause 7.2 of draft-ietf-clue-framework [4].

7.3.1.2.2 Presentation Initiation

To create a presentation during an IM session supporting telepresence, the TP UE or the TP enabled conference focus, acting as a media provider, shall set the presentation attribute (see subclause 4.5) on one or more media captures in the CLUE ADVERTISEMENT message, as defined in subclause 7.1.1.8 of draft-ietf-clue-framework [4].

When receiving the CLUE CONFIGURE message with presentation media captures, the TP UE shall trigger an updated SDP offer as described in subclause 6.3.1 to establish the media stream.

7.3.2 Media consumer

7.3.2.1 General

As specified in draft-ietf-clue-framework [4], a media consumer representing an entity that intends to receive CLUE-controlled media streams, shall send CLUE CONFIGURE message to specify the content and media streams it wants to receive. A TP UE and a TP enabled conference focus acting as a media consumer shall send CLUE CONFIGURE message to select the media captures based on the latest received CLUE ADVERTISEMENT message. The detailed information and format of CLUE messages can be found in draft-ietf-clue-data-model-schema [9].

7.3.2.2 Media capture configuration

Once the CLUE data channel is connected in both point to point cases and conference cases, the TP UE acting as a media consumer shall:

- 1) be prepared to receive a CLUE ADVERTISEMENT message from a remote party;
- 2) select the media captures it wants to receive based on the information elements of the media captures in the received CLUE ADVERTISEMENT message; and
- 3) send the selected media captures within a CLUE CONFIGURE message to the remote party.

In the conference case, the TP enabled conference focus shall select the media captures based on the CLUE CONFIGURE messages received from other TP UEs.

7.3.2.2.1 Multiple content switching and composition

The TP UE or the TP enabled conference focus, acting as a media consumer, may choose to have a switching or composed media by using of Multiple Content Capture (MCC) via CLUE exchange as defined in subclause 7.2 of draft-ietf-clue-framework [4].

EXAMPLE: A TP UE of a typical triple-screen/camera system can produce a composed capture representing the active speaker with the other two captures shown picture in picture, containing attributes to indicate the maximum number of individual captures and their relative positions in a composed image.

7.3.2.2.2 Presentation selection

The TP UE or the TP enabled conference focus, acting as a media consumer, may select a media capture with the attribute of presentation via CLUE exchange as defined in subclause 7.1.1.8 of draft-ietf-clue-framework [4].

8 Protocol for floor control for telepresence

8.1 Introduction

BFCP, as specified in 3GPP TS 24.147 [3], is used to offer floor control of shared resources in an IM session supporting telepresence.

There is no special enhancement to BFCP in this document to support an IM session supporting telepresence.

Annex A (informative): Example signalling flows

A.1 Scope of signalling flows

This annex gives examples of signalling flows for an IM session supporting telepresence based on the Session Initiation Protocol (SIP), SIP Events, the Session Description Protocol (SDP) and the CLUE protocol.

These signalling flows provide detailed flows, which expand on the overview conference information flows provided in 3GPP TS 24.147 [3].

These signalling flows are simplified in that they do not show the interactions between Intermediate IMS networks, nor the interactions between the AS and MRF.

A.2 Key required to interpret signalling flows

The key to interpret signalling flows follows the description as specified in annex A of 3GPP TS 24.147 [3].

A.3 Flows demonstrating the creation of an IM conference supporting telepresence

A.3.1 Introduction

Clause A.3 covers the signalling flows that show how an IM conference supporting telepresence is established in different scenarios.

A.3.2 User creating an IM conference supporting telepresence

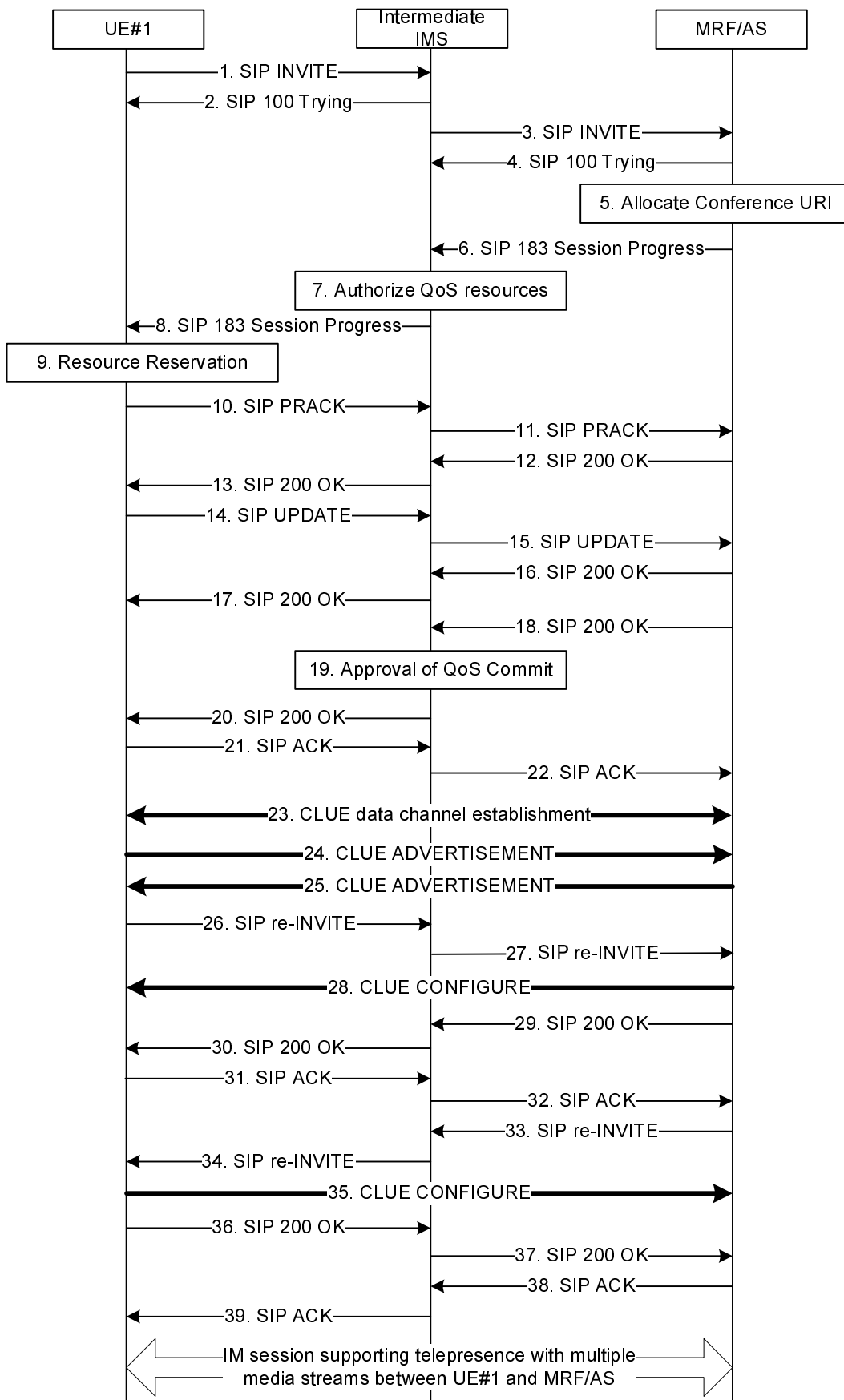


Figure A.3.2-1: User creating an IM conference supporting telepresence

Figure A.3.2-1 shows a user creating an IM conference supporting telepresence by incorporating CLUE with the basic IMS SIP/SDP mechanism.

NOTE: In this example call flows, the CLUE data channel is established after when the SIP 200 OK is received by UE#1, but it can be started earlier if needed by the implementation.

The details of the flows are as follows:

1. SIP INVITE request (UE#1 to Intermediate IMS) – see example in table A.3.2-1

Based on the IMS conference procedures as specified in subclause A.3.2.1 of 3GPP TS 24.147 [3], UE#1 initiates the SIP INVITE request message with a "+sip.clue" Contact header field parameter which indicates the UE supports CLUE protocol, the message further comprises a basic media session and a request for a CLUE data channel (via an SDP "m=" line indicating a UDP/DTLS/SCTP association, together with SDP "sctp-port" attribute indicating the use of data channel and an SDP "dcmapp" attribute to indicate "CLUE" as the application protocol running over the data channel), to create an IM conference supporting telepresence.

The SIP INVITE request is forwarded to Intermediate IMS.

Table A.3.2-1: SIP INVITE request (UE#1 to Intermediate IMS)

```

INVITE sip:conference-factory1@mrfl.home1.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Max-Forwards: 70
Route: <sip:pcscf1.visited1.net:7531;lr;comp=sigcomp>, <sip:orig@scscf1.home1.net;lr>
P-Preferred-Identity: "John Doe" <sip:user1_public1@home1.net>
P-Access-Network-Info: 3GPP-UTRAN-TDD; utran-cell-id-3gpp=234151D0FCE11
Privacy: none
From: <sip:user1_public1@home1.net>; tag=171828
To: <sip:conference-factory1@mrfl.home1.net>
Call-ID: cb03a0s09a2sdfg1kj490333
Cseq: 127 INVITE
Require: sec-agree
Proxy-Require: sec-agree
Supported: precondition, 100rel, gruu, 199
Security-Verify: ipsec-3gpp; q=0.1; alg= hmac-sha-1-96; spi-c=98765432; spi-s=87654321;
port-c=8642; port-s=7531
Contact: <sip:user1_public1@home1.net>;gr=urn:uuid:f81d4fae-7dec-11d0-a765-
00a0c91e6bf6;comp=sigcomp;+sip.clue
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE, SUBSCRIBE, NOTIFY
Accept: application/sdp, application/3gpp-ims+xml
Content-Type: application/sdp
Content-Length: (...)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c=IN IP6 5555::aaa:bbb:ccc:ddd
t=0 0
a=group CLUE 3
m=video 3400 RTP/AVP 98 99
b=AS:75
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
a=rtpmap:99:MPVMP4V-ES
a=mid:1
m=audio 3456 RTP/AVP 97 96
b=AS:25.4
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:96 telephone-event
a=maxtime:20
a=mid:2
m=application 54111 UDP/DTLS/SCTP webrtc-datachannel
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=max-message-size: 100000
a=sctp-port: 5000
a=setup:actpass
a=connection:new
a=fingerprint:SHA-1 \
4A:AD:B9:B1:3F:82:18:3B:54:02:12:DF:3E:5D:49:6B:19:E5:7C:AB
a=dcmap:2 subprotocol="CLUE"
a=mid:3

```

Contact: contains the "+sip.clue" Contact header field parameter.

2. SIP 100 (Trying) response (Intermediate IMS to UE#1)

Intermediate IMS responds to the SIP INVITE request (1) with a 100 (Trying) provisional response.

3. SIP INVITE request (Intermediate IMS to MRF/AS)

Intermediate IMS forwards the SIP INVITE request to the MRF/AS.

4. SIP 100 (Trying) response (MRF/AS to Intermediate IMS)

The MRF/AS responds to the SIP INVITE request (3) with a 100 (Trying) provisional response.

5. Allocate Conference URI

The MRF/AS allocates a conference URI for the request, based on local information and information gained from the conference-factory URI, as well as information gained from other elements of the SIP signalling.

6. SIP 183 (Session Progress) response (MRF/AS to Intermediate IMS) – see example in table A.3.2-2

The MRF determines the set of codecs that it is capable of supporting for this conference. It determines the intersection with those appearing in the SDP in the SIP INVITE request.

The MRF determines the TP UE's support of telepresence by receiving the "+sip.clue" Contact header field parameter of the SIP INVITE request, and it also includes the same parameter in the response message. The media stream capabilities of the destination as well as the connection information for a CLUE data channel are returned in a SIP 183 (Session Progress) provisional response and finally forwarded to UE#1.

Table A.3.2-2: SIP 183 (Session Progress) response (MRF/AS to Intermediate IMS)

```

SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP scscf1.homel.net;branch=z9hG4bK332b23.1, SIP/2.0/UDP
    pcscf1.visited1.net;branch=z9hG4bK240f34.1, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Record-Route: <sip:scscf1.homel.net;lr>, <sip:pcscf1.visited1.net;lr>
P-Asserted-Identity: "Conference Server" <sip:mrfl.homel.net>
P-Charging-Vector: icid-value="AyretyU0dm+602IrT5tAFrbHLso=023551024"; orig-ioi=homel.net;
    term-ioi=homel.net
P-Charging-Function-Addresses: ccf=[5555::b99:c88:d77:e66]; ccf=[5555::a55:b44:c33:d22];
    ecf=[5555::1ff:2ee:3dd:4cc]; ecf=[5555::6aa:7bb:8cc:9dd]
Privacy: none
From:
To: <sip:conference-factory1@mrfl.homel.net>; tag=314159
Call-ID:
CSeq:
Require: precondition, 100rel
Contact: <sip:lmaa234269@mrfl.homel.net;gr>;isfocus;+sip.clue
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE, SUBSCRIBE, NOTIFY, PUBLISH
RSeq: 9021
Content-Type: application/sdp
Content-Length: (...)

v=0
o=- 2987933623 2987933623 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c=IN IP6 5555::eee:fff:aaa:bbb
t=0 0
a=group CLUE 3
m=video 10001 RTP/AVP 98 99
b=AS:75
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos mandatory remote sendrecv
a=conf:qos remote sendrecv
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
a=rtpmap:99 MP4V-ES
a=mid:1
m=audio 6544 RTP/AVP 97 96
b=AS:25.4
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos mandatory remote sendrecv
a=conf:qos remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:96 telephone-event
a=maxptime:20
a=mid:2
m=application 62442 UDP/DTLS/SCTP webrtc-datachannel
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos mandatory remote sendrecv
a=conf:qos remote sendrecv
a= max-message-size: 100000
a=sctp-port: 5100
a=setup:passive
a=connection:new
a=fingerprint:SHA-1 \
    4A:AD:B9:B1:3F:82:18:3B:54:02:12:DF:3E:5D:49:6B:19:E5:7C:AB
a=dcmap:2 subprotocol="CLUE"
a=mid:3

```

Contact: contains the "+sip.clue" Contact header field parameter.

7. Authorize QoS Resources

Intermediate IMS node authorizes the resources necessary for this session. The approval of QoS commitment either happens at this stage or after the 200 (OK) response of SIP INVITE request (26) and SIP INVITE request (34) based on operator local policy.

8. SIP 183 (Session Progress) response (Intermediate IMS to UE#1)

Intermediate IMS forwards the SIP 183 (Session Progress) response to the originating UE#1.

9. Resource reservation

The originating UE#1 sets up the bearer in accordance with the media description received SDP.

10 - 22. Set up of basic media session

The basic media session is processed as per the procedures defined in subclause A.3.2.1 of 3GPP TS 24.147 [3].

23. CLUE channel establishment

With the negotiated connection information in SDP, UE#1 and MRF establishes the CLUE data channel between them.

24-25. CLUE ADVERTISEMENT request (between UE#1 and MRF)

UE#1 and MRF exchange their respective CLUE ADVERTISEMENT messages, each of which contains the available media captures and information elements such as spatial views, media switching, media compositions, and encodings to facilitate capture selection and rendering.

26. SIP re-INVITE request (UE#1 to Intermediate IMS) –see example in table A.3.2-3

UE#1 initiates the SIP re-INVITE request adding new multiple media streams with spatial views that are negotiated via a CLUE data channel.

For this example, it is assumed that UE#1 is a typical triple-screen/camera system which can produce three video media streams.

The SIP re-INVITE request is forwarded to Intermediate IMS.

Table A.3.2-3: SIP re-INVITE request (UE#1 to Intermediate IMS)

```

INVITE sip:lmaa234269@mrfl.home1.net;gr SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Max-Forwards: 70
Route: <sip:pcscf1.visited1.net:7531;lr;comp=sigcomp>, <sip:orig@scscf1.home1.net;lr>
P-Preferred-Identity: "John Doe" <sip:user1_public1@home1.net>
P-Access-Network-Info: 3GPP-UTRAN-TDD; utran-cell-id-3gpp=234151D0FCE11
Privacy: none
From: <sip:user1_public1@home1.net>; tag=171828
To: <sip:lmaa234269@mrfl.home1.net;gr>
Call-ID: cb03a0s09a2sdfglkj490333
Cseq: 127 INVITE
Require: sec-agree
Proxy-Require: sec-agree
Supported: precondition, 100rel, gruu, 199
Security-Verify: ipsec-3gpp; q=0.1; alg= hmac-sha-1-96; spi-c=98765432; spi-s=87654321;
port-c=8642; port-s=7531
Contact: <sip:user1_public1@home1.net>;gr=urn:uuid:f81d4fae-7dec-11d0-a765-
00a0c91e6bf6;comp=sigcomp;+sip.clue
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE, SUBSCRIBE, NOTIFY
Accept: application/sdp, application/3gpp-ims+xml
Content-Type: application/sdp
Content-Length: (...)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c=IN IP6 5555::aaa:bbb:ccc:ddd
t=0 0
a=group CLUE 3 4 5 6 7
m=video 3400 RTP/AVP 98 99
b=AS:75
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
a=rtpmap:99:MPVMP4V-ES
a=mid:1
m=audio 3456 RTP/AVP 97 96
b=AS:25.4
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:96 telephone-event
a=maxptime:20a=mid:2
m=application 54111 UDP/DTLS/SCTP webrtc-datachannel
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=max-message-size: 100000
a=sctp-port: 5000
a=setup:actpass
a=connection:existing
a=fingerprint:SHA-1 \
4A:AD:B9:B1:3F:82:18:3B:54:02:12:DF:3E:5D:49:6B:19:E5:7C:ABa=mid:3
a=dcmap:2 subprotocol="CLUE"
a=mid:3
m=video 3402 RTP/AVP 98 99
b=AS:75
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local send
a=des:qos none remote recv
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
a=rtpmap:99:MPVMP4V-ES
a=label:encl
a=sendonly
a=mid:4
m=video 3404 RTP/AVP 98 99
b=AS:75

```

```

a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local send
a=des:qos none remote recv
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
a=rtpmap:99:MPVMP4V-ES
a=label:enc2
a=sendonly
a=mid:5
m=video 3406 RTP/AVP 98 99
b=AS:75
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local send
a=des:qos none remote recv
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
a=rtpmap:99:MPVMP4V-ES
a=label:enc3
a=sendonly
a=mid:6
m=audio 3458 RTP/AVP 98 99
b=AS:25.4
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local send
a=des:qos none remote recv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; max-change-period=2
a=rtpmap:96 telephone-event
a=maxtime:20
a=label:enc4
a=sendonly
a=mid:7

```

Contact: contains the "+sip.clue" Contact header field parameter.

27. SIP re-INVITE request (Intermediate IMS to MRF/AS)

Intermediate IMS forwards the SIP re-INVITE request to MRF/AS.

28. CLUE CONFIGURE response (MRF to UE#1)

The MRF sends a CLUE CONFIGURE message containing the selected media captures based on the latest received CLUE ADVERTISEMENT message and the SDP offer associated with the advertised media captures in SIP re-INVITE request.

NOTE: When having multiple users in the same meeting, the MRF may also exchange the CLUE messages with other TP UEs based on reception of the CLUE messages and the SDP offer/answer associated with the CLUE messages from UE#1.

29. SIP 200 (OK) response (MRF/AS to Intermediate IMS) – see example in table A.3.2-4

After the success modification of the session, the MRF/AS sends a SIP 200 (OK) final response to the SIP re-INVITE request (27) to Intermediate IMS based on the selected media captures in a CLUE CONFIGURE message.

Table A.3.2-4: SIP 200 (OK) response (MRF/AS to Intermediate IMS)

```

SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf1.home1.net;branch=z9hG4bK332b23.1, SIP/2.0/UDP
    pcscf1.visited1.net;branch=z9hG4bK240f34.1, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Record-Route: <sip:scscf1.home1.net;lr>, <sip:pcscf1.visited1.net;lr>
From:
To:
Call-ID:
CSeq: 127 INVITE
Contact: <sip:lmaa234269@mrfl1.home1.net;gr>;isfocus;+sip.clue
Allow-Events: conference, pending-additions
Content-Length: (...)

v=0
o=- 2987933623 2987933623 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c=IN IP6 5555::eee:fff:aaa:bbb
t=0 0
a=group CLUE 3 4 5 6 7
m=video 10001 RTP/AVP 98 99
b=AS:75
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos mandatory remote sendrecv
a=conf:qos remote sendrecv
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
a=rtpmap:99 MP4V-ES
a=mid:1
m=audio 6544 RTP/AVP 97 96
b=AS:25.4
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos mandatory remote sendrecv
a=conf:qos remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:96 telephone-event
a=maxptime:20
a=mid:2
m=application 62442 UDP/DTLS/SCTP webrtc-datachannel
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos mandatory remote sendrecv
a=conf:qos remote sendrecv
a=connection:new
a=max-message-size: 100000
a=sctp-port: 5100
a=setup:active
a=connection:existing
a=fingerprint:SHA-1 \
    4A:AD:B9:B1:3F:82:18:3B:54:02:12:DF:3E:5D:49:6B:19:E5:7C:ABa=mid:3
a=dcmap:2 subprotocol="CLUE"
a=mid:3
m=video 10003 RTP/AVP 98 99
b=AS:75
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local recv
a=des:qos none remote send
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
a=rtpmap:99 MPVMP4V-ES
a=label:encl
a=recvonly
a=mid:4
m=video 10005 RTP/AVP 98 99
b=AS:75
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local recv
a=des:qos none remote send

```

```

a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
a=rtpmap:99:MPVMP4V-ES
a=label:enc2
a=recvonly
a=mid:5
m=video 10007 RTP/AVP 98 99
b=AS:75
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local recv
a=des:qos none remote send
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
a=rtpmap:99:MPVMP4V-ES
a=label:enc3
a=recvonly
a=mid:6
m=audio 6546 RTP/AVP 97 96
b=AS:25.4
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local recv
a=des:qos mandatory remote send
a=conf:qos remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:96 telephone-event
a=maxptime:20
a=label:enc4
a=recvonly
a=mid:7

```

Contact: contains the "+sip.clue" Contact header field parameter.

30. SIP 200 (OK) response (Intermediate IMS to UE#1)

Intermediate IMS forwards the SIP 200 (OK) response to UE#1.

31-32. ACK request (UE#1 to Intermediate IMS, till MRF/AS)

UE#1 responding to the SIP 200 (OK) response (33) sends a SIP ACK request to Intermediate IMS and till UE#1 as specified in subclause A.3.2 of 3GPP TS 24.147 [3]. Accordingly, the UE#1 starts to send the media streams which are configured by MRF/AS via the exchange of CLUE messages.

33. SIP re-INVITE request (MRF/AS to Intermediate IMS) –see example in table A.3.2-5

MRF/AS sends the SIP re-INVITE request adding new multiple media streams with spatial views that are negotiated via a CLUE data channel.

For this example, it is assumed that MRF/AS can send two media streams based on the negotiation of media captures via the exchange of CLUE messages.

The SIP re-INVITE request is forwarded to Intermediate IMS.

Table A.3.2-5: SIP re-INVITE request (MRF/AS to Intermediate IMS)

```

INVITE sip:user1_public1@home2.net SIP/2.0
Via: SIP/2.0/UDP mrfl.home1.net;branch=z9hG4bK23273846
Max-Forwards: 70
P-Charging-Vector: #####
Privacy: none
From: <sip:conferencel@mrfl.home1.net>;tag=171828
To: <sip:user1_public1@home2.net>
Call-ID: cb03a0s09a2sdfg1kj490333
Cseq: 127 INVITE
Supported: precondition, 100rel
Contact: <sip:lmaa234269@mrfl.home1.net;gr>;isfocus;+sip.clue
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE, SUBSCRIBE, NOTIFY, PUBLISH
Allow-Events: conference, pending-additions
Content-Type: application/sdp
Content-Length: (...)

v=0
o=- 2987933623 2987933623 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c=IN IP6 5555::eee:fff:aaa:bbb
t=0 0
a=group CLUE 3 4 5 6 7 8 9 10
m=video 10001 RTP/AVP 98 99
b=AS:75
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos mandatory remote sendrecv
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
a=rtpmap:99 MP4V-ES
a=mid:1
m=audio 6544 RTP/AVP 97 96
b=AS:25.4
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos mandatory remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:96 telephone-event
a=maxptime:20
a=mid:2
m=application 62442 UDP/DTLS/SCTP webrtc-datachannel
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos mandatory remote sendrecv
a=setup:passive
a=connection:new
a=max-message-size: 100000
a=sctp-port: 5100
a=connection:existing
a=fingerprint:SHA-1 \
    4A:AD:B9:B1:3F:82:18:3B:54:02:12:DF:3E:5D:49:6B:19:E5:7C:ABa=mid:3
a=dcmap:2 subprotocol="CLUE"
a=mid:3
m=video 10003 RTP/AVP 98 99
b=AS:75
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local recv
a=des:qos none remote send
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
a=rtpmap:99:MPVMP4V-ES
a=label:encl
a=recvonly
a=mid:4
m=video 10005 RTP/AVP 98 99
b=AS:75
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local recv
a=des:qos none remote send

```

```

a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
a=rtpmap:99:MPVMP4V-ES
a=label:enc2
a=recvonly
a=mid:5
m=video 10007 RTP/AVP 98 99
b=AS:75
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local recv
a=des:qos none remote send
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
a=rtpmap:99:MPVMP4V-ES
a=label:enc3
a=recvonly
a=mid:6
m=audio 6546 RTP/AVP 97 96
b=AS:25.4
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local recv
a=des:qos mandatory remote send
a=conf:qos remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:96 telephone-event
a=maxptime:20
a=label:enc4
a=recvonly
a=mid:7
m=video 10009 RTP/AVP 98 99
b=AS:75
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local send
a=des:qos none remote recv
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
a=rtpmap:99:MPVMP4V-ES
a=label:enc5
a=sendonly
a=mid:8
m=video 10011 RTP/AVP 98 99
b=AS:75
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local send
a=des:qos none remote recv
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
a=rtpmap:99:MPVMP4V-ES
a=label:enc6
a=sendonly
a=mid:9
m=audio 6548 RTP/AVP 97 96
b=AS:25.4
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local send
a=des:qos mandatory remote recv
a=conf:qos remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:96 telephone-event
a=maxptime:20
a=label:enc7
a=sendonly
a=mid:10

```

Contact: contains the "+sip.clue" Contact header field parameter.

34. SIP re-INVITE request (Intermediate IMS to UE#1)

The SIP re-INVITE request is forwarded to UE#1.

35. CLUE CONFIGURE response (UE#1 to MRF)

UE#1 sends a CLUE CONFIGURE message containing the selected media captures based on the latest received CLUE ADVERTISEMENT message and the SDP offer associated with the advertised media captures in SIP re-INVITE request.

NOTE: When having multiple users in the same meeting, the MRF may also exchange the CLUE messages with other TP UEs based on reception of the CLUE messages and the SDP offer/answer associated with the CLUE messages from UE#1.

36. SIP 200 (OK) response (UE#1 to Intermediate IMS) – see example in table A.3.2-6

After the success modification of the session, UE#1 sends a SIP 200 (OK) final response to the SIP re-INVITE request (34) to Intermediate IMS based on the selected media captures in a CLUE CONFIGURE message. UE#1 further deactivates non-CLUE-controlled media streams according to subclause 4.5.4.1 of draft-ietf-clue-signaling [6].

Table A.3.2-6: SIP 200 (OK) response (UE#1 to Intermediate IMS)

```

SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.visited1.net:5088;comp=sigcomp;branch=z9hG4bK240f34.1 SIP/2.0/UDP
scscf1.home1.net;branch=z9hG4bK332b23.1, SIP/2.0/UDP
icscf1.home1.net;branch=z9hG4bK241d17.2, SIP/2.0/UDP
mrfl1.home1.net;branch=z9hG4bK23273846
Record-Route: <sip:pcscf1.visited1.net:5088;lr;comp=sigcomp>, <sip:scscf1.home1.net;lr>
P-Access-Network-Info: 3GPP-UTRAN-TDD; utran-cell-id-3gpp=234151D0FCE11
From:
To:
Call-ID:
CSeq: 127 INVITE
Contact: <sip:user1\_public1@home2.net>;gr=urn:uuid:2ad8950e-48a5-4a74-8d99-
ad76cc7fc74;comp=sigcomp;+sip.clue
Content-Length: (...)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c=IN IP6 5555::aaa:bbb:ccc:ddd
t=0 0
a=group CLUE 3 4 5 6 7 8 9 10
m=video 0 RTP/AVP 98 99
b=AS:75
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
a=rtpmap:99:MPVMP4V-ES
a=mid:1
m=audio 0 RTP/AVP 97 96
b=AS:25.4
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:96 telephone-event
a=maxptime:20
a=mid:2
m=application 51111 UDP/DTLS/SCTP webrtc-datachannel
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=max-message-size: 100000
a=sctp-port: 5000
a=setup:active
a=connection:existing
a=fingerprint:SHA-1 \
4A:AD:B9:B1:3F:82:18:3B:54:02:12:DF:3E:5D:49:6B:19:E5:7C:ABa=mid:3
a=dcmap:2 subprotocol="CLUE"
a=mid:3
m=video 3402 RTP/AVP 98 99
b=AS:75
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local send
a=des:qos none remote recv
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
a=rtpmap:99:MPVMP4V-ES
a=label:encl
a=sendonly
a=mid:4
m=video 3404 RTP/AVP 98 99
b=AS:75
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local send
a=des:qos none remote recv
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
a=rtpmap:99:MPVMP4V-ES

```

```

a=label:enc2
a=sendonly
a=mid:5
m=video 3406 RTP/AVP 98 99
b=AS:75
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local send
a=des:qos none remote recv
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
a=rtpmap:99:MPVMP4V-ES
a=label:enc3
a=sendonly
a=mid:6
m=audio 3458 RTP/AVP 97 96
b=AS:25.4
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local send
a=des:qos none remote recv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:96 telephone-event
a=maxptime:20
a=label:enc4
a=sendonly
a=mid:7
m=video 3408 RTP/AVP 98 99
b=AS:75
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local recv
a=des:qos none remote send
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
a=rtpmap:99:MPVMP4V-ES
a=label:enc5
a=recvonly
a=mid:8
m=video 3410 RTP/AVP 98 99
b=AS:75
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local recv
a=des:qos none remote send
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
a=rtpmap:99:MPVMP4V-ES
a=label:enc6
a=recvonly
a=mid:9
m=audio 3460 RTP/AVP 97 96
b=AS:25.4
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local recv
a=des:qos none remote send
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:96 telephone-event
a=maxptime:20
a=label:enc7
a=recvonly
a=min:10

```

Contact: contains the "+sip.clue" Contact header field parameter.

37. SIP 200 (OK) response (Intermediate IMS to MRF/AS)

Intermediate IMS forwards the SIP 200 (OK) response to the MRF/AS.

38-39. SIP ACK request (MRF/AS to Intermediate IMS, till UE#1)

The MRF/AS responding to the SIP 200 (OK) response (41) sends a SIP ACK request sent to Intermediate IMS and till UE#1 as specified in subclause A.3.2 of 3GPP TS 24.147 [3]. Accordingly, the UE#1 starts to receive the media streams which are configured by itself via the exchange of CLUE messages.

A.3.3 User invited to an IM conference supporting telepresence

Subclause A.3.3 covers the signalling flows that inviting a TP UE getting invited to an IM conference supporting telepresence, which may be triggered for example after the conference focus receives a REFER request to invite a user join the conference.

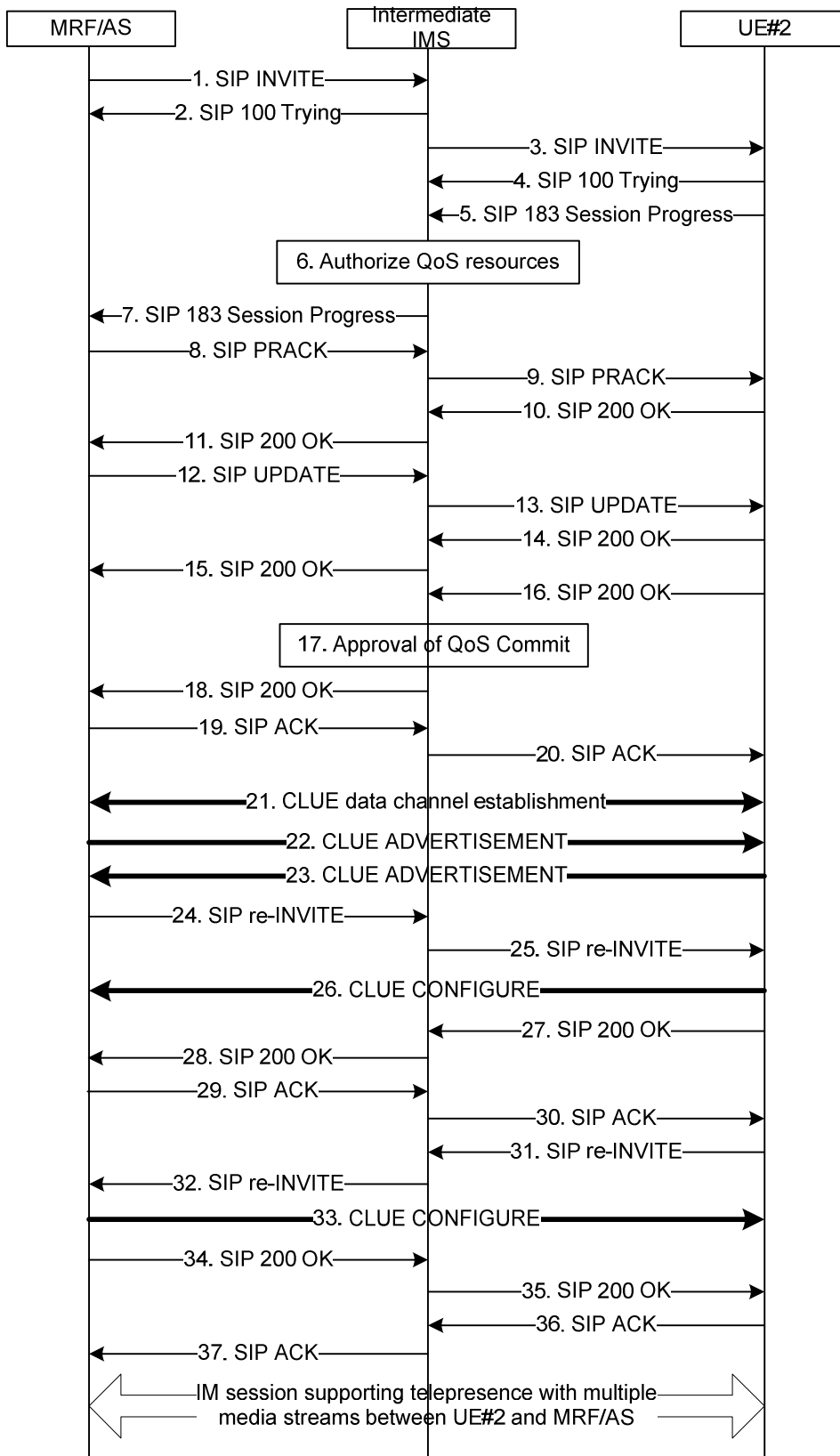


Figure A.3.3-1: A user invited to an IM conference supporting telepresence

Figure A.3.3-1 shows a user invited to an IM conference supporting telepresence by incorporating CLUE tools with the basic IMS SIP/SDP mechanism.

A.4 Flows demonstrating the update of an ongoing IM conference supporting telepresence

Figure A.4-1 shows the example call flow of update during an ongoing IM session supporting telepresence after multiple media streams been established between UE#1 and MRF/AS.

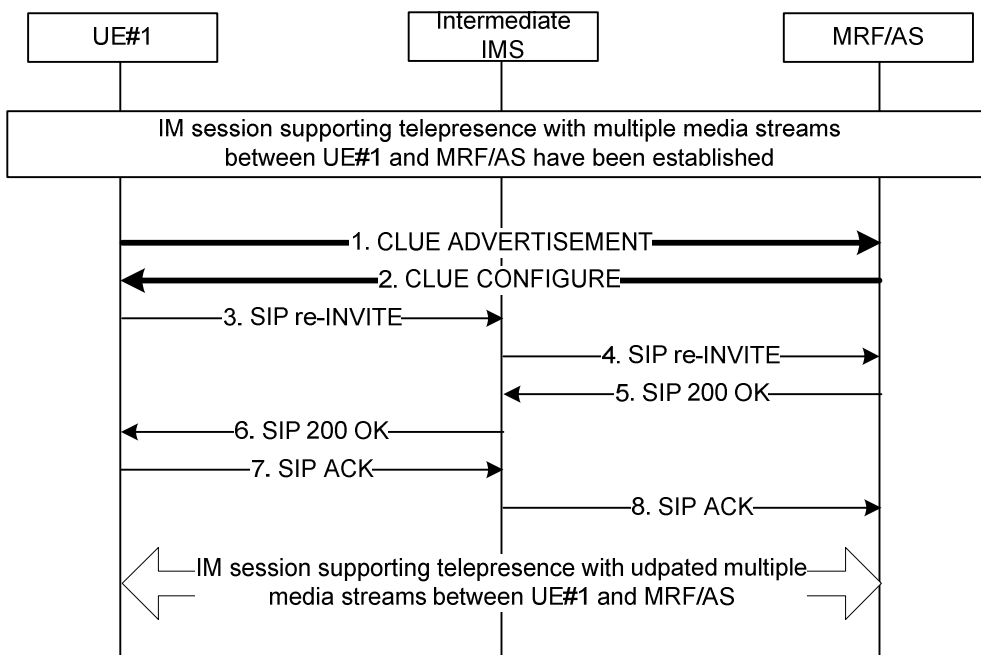


Figure A.4-1: Call flow of update during an ongoing IM conference supporting telepresence

Annex B (informative): Change history

Change history							
Date	TSG #	TSG Doc.	CR	Rev	Subject/Comment	Old	New
2014-03					Initial skeleton provided by rapporteur for CT1#86bis.		0.0.0
2014-04					With agreed skeleton, scope, protocols usage and overview of telepresence after CT1#86bis from C1-141469, C1-141470, C1-141472 and C1-141641	0.0.0	0.1.0
2014-05					Incorporate procedure for protocols using SIP, SDP, CLUE, and message flow for IM conference supporting telepresence based on C1-142344, C1-142345, C1-142346, C1-142347, C1-142348 and C1-142495 at CT1#87.	0.1.0	0.2.0
2014-07					Incorporate procedure for session update and media feature tag alignment based on C1-143201 and C1-143327 at CT1#88.	0.2.0	0.3.0
2014-09	CT-65	CP-140626			Version 1.0.0 presented for information at CT plenary	0.3.0	1.0.0
2014-10					Additions of presentation, MRF roles, protocol alignments and updates of example signalling flows based on agreed C1-143628, C1-144075, C1-144076, C1-144077, C1-144078 and C1-144240 at CT1#88bis.	1.0.0	1.1.0
2014-11					CLUE updates based on agreed C1-144867 at CT1#89.	1.1.0	1.2.0
2014-12	CT-66	CP-140809			Version 2.0.0 presented for approval at CT plenary	1.2.0	2.0.0
2014-12	CT-66				Version 12.0.0 created after approval at CT plenary	2.0.0	12.0.0
2015-03	CT-67	CP-150057	0001	2	IMS Telepresence reference updates	12.0.0	12.1.0
2015-06	CT-68	CP-150306	0004	2	IETF Update on IMS Telepresence	12.1.0	12.2.0

History

Document history		
V12.0.0	January 2015	Publication
V12.1.0	April 2015	Publication
V12.2.0	July 2015	Publication