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**Universal Mobile Telecommunications System (UMTS);  
LTE;  
Combining Circuit Switched (CS)  
and IP Multimedia Subsystem (IMS) services;  
Stage 3  
(3GPP TS 24.279 version 9.1.0 Release 9)**



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## Foreword

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## Foreword

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# 1 Scope

The present document provides the technical realisation for the combination of Circuit Switched calls and IM sessions when using them simultaneously between the same two users.

The present document describes the use of CS and IM services in combination, using the existing procedures that have been defined for CS and IMS. It includes the necessary function as adding an IM session to an ongoing CS call, adding a CS call to an ongoing IM session, supplementary services as they relate to CSICS and supporting capability exchange.

The present document is applicable to UE and Application Servers providing for the combination of Circuit Switched calls and IM sessions.

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# 2 References

The following documents contain provisions that, 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: "3G Vocabulary".
- [2] 3GPP TS 22.279: "Combined CS Calls and IMS Sessions".
- [3] 3GPP TS 23.279: "Combining CS and IMS services".
- [4] 3GPP TS 24.008: "Mobile radio interface Layer 3 specification; Core network protocols; Stage 3".
- [5] 3GPP TS 24.229: "IP multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP); Stage 3".
- [6] RFC 3840: "Indicating User Agent Capabilities in the Session Initiation Protocol".
- [7] 3GPP TS 22.081: "Line Identification; Service description - Stage 1".
- [8] 3GPP TS 22.087: "User-to-User Signalling (UUS); Service description - Stage 1".
- [9] 3GPP TS 24.087: "User-to-User Signalling (UUS) Supplementary Service - Stage 3".
- [10] 3GPP TS 24.247: "Messaging service using the IP Multimedia (IM) Core Network; Stage 3".
- [11] 3GPP TS 24.228 Release 5: "Signalling flows for the IP multimedia call control based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP) ; Stage 3".
- [12] 3GPP 33.203: "3G security; Access security for IP-based services".
- [13] 3GPP TS 26.141 "IP Multimedia System (IMS) Messaging and Presence; Media formats and codecs".
- [14] RFC 3264: "An Offer/Answer Model with Session Description Protocol (SDP)".
- [15] 3GPP TS 29.162: "Interworking between the IM CN subsystem and IP networks".
- [16] 3GPP TS 29.328: "IP Multimedia (IM) Subsystem Sh interface".
- [17] RFC 3261: " SIP: Session Initiation Protocol".



- [18] draft-ietf-cuss-sip-uu-06 (May 2012): "A Mechanism for Transporting User to User Call Control Information in SIP".

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

- [19] draft-ietf-cuss-sip-uu-isdn-04 (May 2012): "Interworking ISDN Call Control User Information with SIP".

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

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## 3 Definitions, symbols and abbreviations

### 3.1 Definitions

For the purposes of the present document, the following terms and definitions are given in 3GPP TS 22.279 [2].

**Combinational Service**  
**Combinational call**  
**Combinational Session**  
**CSICS capable UE**

For the purposes of the present document, the following terms and definitions are given in 3GPP TS 23.279 [3].

**CSI session**  
**Multimedia IMS session**  
**CSI origination**  
**CSI termination**  
**IMS origination**  
IMS termination

For the purposes of the present document, the following terms and definitions given in 3GPP TR 21.905 [1] apply:

**Universal Subscriber Identity Module (USIM)**  
**User Equipment (UE)**

For the purposes of the present document, the following terms and definitions given in 3GPP TS 33.203 [12] apply:

**IM Subscriber Identity Module (ISIM)**

### 3.2 Abbreviations

For the purposes of the present document, the abbreviations defined in 3GPP TR 21.905 [1] and the following abbreviations apply:

CSICS            Circuit Switched IMS Combinational Service

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## 4 Common capability information and identifiers for CS-first and IMS-first scenarios

### 4.1 UE capability exchange - overview

The terminal capabilities that can be exchanged are:

- a) Media types which can be supported as IMS media streams (i.e. media component definitions of IM sessions);
- b) Media format parameters for supported IMS media types (codecs, media file formats etc.);

- c) MSISDN and preferred SIP URI or tel URI for the UE sending the UE capability information;
- d) Whether the terminal is capable combining an IM session with either CS-Video telephony, CS-Voice, or both;
- e) MMS version that is supported;
- f) Support for other IMS based capabilities and/or services e.g. PoC; and
- g) Personal ME Identifier to identify which of the user's MEs the UE capability information is related to.
- h) UE capability version, which may be used to identify the current capabilities of a terminal to indicate capability update.
- i) The IM Status, which indicates the IMS capability and registration status of the sending UE.

3GPP TS 26.141 [13] specifies the 3GPP recommended media formats and codecs which can be reflected in the SDP used by the CUA.

## 4.2 Personal ME identifier

A specific ME of the user shall be identified by a personal ME identifier, which has the syntax PMI-XXXX. The part "XXXX" shall be a random value, as defined in 3GPP TS 24.008 [4] subclause 2.1.1, in the range from hexadecimal 0000 to hexadecimal FFFF generated by the CUA. The personal ME identifier shall be stored by the UE and can be changed to ensure that two or more of the user's MEs do not have the same personal ME identifier.

At CS call setup, the MSISDN distinguishes a user and the personal ME identifier distinguishes one ME among those belonging to that user. In the IMS, a public user identity distinguishes a user and the personal ME identifier distinguishes one ME among those belonging to that user.

The personal ME identifier may be exchanged during:

- UE capability information exchange;
- IM session set up; and
- CS call set up.

## 4.3 UE capability version

The UE capability version is used to identify the current UE's version of capabilities and has the syntax UCV-XX. The part 'XX' shall be a value in the range from hexadecimal 00 to hexadecimal FF generated by the CUA, and based on the set of capabilities. The UE capability version shall be changed to another value whenever the CUA changes its capabilities.

The UE capability version may be exchanged during:

- UE capability information exchange;
- IM session set up, and
- CS call set up.

## 4.4 Radio environment information

The radio environment information is used to indicate whether the terminal is in a radio environment at CS call setup that supports simultaneous use of CS and PS services.

The radio environment information may be exchanged during CS call setup only.

## 4.5 IM Status

The IM Status is exchanged at CS call setup only. The IM Status is only valid for the duration of the CS call.

The IM Status is an indication of the IMS capability and registration status of the UE that provides this indication. The IM Status, if provided, shall indicate the UE as either:-

- not capable of IM subsystem registration
- IM subsystem registered
- IM subsystem capable and willing to register to IM subsystem
- IM subsystem capable but will not register to IM subsystem

The IM Status of the UE that provides the IM Status can be used by the receiving CUA in its decision to perform an IMS registration if allowed by user's preference (and if the receiving UE is not already IMS registered).

The provision of the IM Status is optional.

---

## 5 Common procedures for CS-first and IMS-first scenarios

### 5.1 Registration of UE capabilities

The CUA may include a media feature tag in the Contact header of a REGISTER request, in accordance with RFC 3840 [6]. The media feature-tags applicable for CSI are g.3gpp.cs-voice, g.3gpp.cs-video, or both values. These media feature tags are further described in annex A.

### 5.2 Criteria for initiating capability exchange

An OPTIONS request should be sent from the UE to the remote UE in the following cases:

- a) when the UE is in a CS call with the remote UE; and
  - the received radio environment information indicates that the remote UE and remote radio environment is capable of handling PS and CS domain simultaneously;
  - the received MSISDN plus personal ME identifier plus UE capability version does not match any combination in use by the UE to cache capabilities; and
  - the received IMS status is "MS is IM subsystem registered" or "MS is IM subsystem capable and willing to register to IM subsystem".
- b) when the UE is in an IM session with the remote UE and the received INVITE request includes UE capability version plus public user identity plus personal ME identifier that does not match any combination in use by the UE to cache capabilities.

NOTE 1: For a CSI call where the CS call is established first, the OPTIONS request can only be sent when the PS domain is available. The PS domain can either be available already when the CS call is set-up or at a later instant due to changed radio condition.

- c) when the UE capabilities have been significantly upgraded; or

NOTE 2: A significant upgrade of UE capabilities is when an UE has been upgraded with e.g. video capability or supports a new service.

- d) when a UE receives an OPTIONS request from a remote UE and there is no ongoing (or recently finished) capability exchange initiated by the UE.

NOTE 3: The OPTIONS request can be sent as a standalone transaction or as a part of a session.

Information specific to capability exchange with a CS-call already established is covered in subclause 7.3.1.2.

Information specific to capability exchange with an IM session already established is covered in subclause 6.3.1.2.

## 5.3 Criteria for responding to a capability exchange request

The end-user or application shall give its approval for the UE capabilities to be included in response to a capability exchange request.

## 5.4 Exchange of radio environment information and IM Status

A UE may send information about its current radio environment and its IM Status during CS Call set-up. If the UE finds that the remote UE's IM Status indicates the remote UE is either IM subsystem registered or IM subsystem capable and willing to register to IM subsystem and its current radio environment supports simultaneous CS and PS services, then if allowed by the user's preference the UE should attempt an IMS registration (if the UE is not already IMS registered).

The information exchanged during the call establishment is valid for the duration of the CS call.

## 5.5 Storage of capabilities in the UE

The terminal capabilities of other UEs are stored by the requesting UE. This stored information shall be valid until ISIM/USIM is removed or until an update of the terminal capabilities occurs according to subclause 5.2. Stored information shall not be deleted when the UE is switched off.

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# 6 Combining a CS call with an existing IM session

## 6.1 Introduction

(void)

## 6.2 Functional entities

### 6.2.1 User equipment

A UE shall implement the role of a CUA as specified in subclause 6.3.1.

### 6.2.2 Application server (AS)

An application server may be included for a CSI call. However, no specific CSI role is assigned to it.

## 6.3 Roles

### 6.3.1 CSI user agent (CUA)

#### 6.3.1.1 General

In addition to the procedures specified in subclause 6.3.1, the CUA shall support the procedures specified in 3GPP TS 24.229 [5] appropriate to the functional entity in which the CUA is implemented.

#### 6.3.1.2 Exchange of UE capability information – IM session first scenario

When a CUA wants to exchange capabilities with the remote party, the CUA originating the OPTIONS request shall apply the procedure as specified in 3GPP TS 24.229 [5] with the following additions:

- a) The CUA shall in the Request URI:

- include the URI received in the P-Asserted-Identity header from the remote CUA; or if that is not available:
  - include a tel URI corresponding to the MSISDN intended for the CS call set up; or
  - include a SIP URI associated with the remote user available in the CUA.

NOTE 1: The MSISDN can only be included as a tel URI if it is in international format.

NOTE 2: If no SIP URI or tel URI in accordance with above is available the CUA cannot initiate the OPTIONS request.

NOTE 3: The OPTIONS request can also be sent as a part of a session. The request URI needs to be set in accordance with TS 24.229 [5] in this case.

- b) The CUA shall in the P-Preferred-Identity header either include:
  - preferably the MSISDN of an assumed registered tel URI of the CUA; or
  - an assumed registered SIP URI associated with the CUA.
- c) The CUA shall in the Accept-Contact header include media feature tag(s) with the value(s) "+g.3gpp.cs-voice" or "+g.3gpp.cs-video" or both, marked as explicit.
- d) The CUA may in the User-Agent header include the personal ME identifier and the UE capability version. Both, the personal ME identifier and the UE capability version, shall be present or neither of them.

The CUA answering with a 200 (OK) response to the OPTIONS request shall apply the procedure as specified in 3GPP TS 24.229 [5] with the following additions:

- a) The CUA shall in the Contact header include media feature tag(s) with the value(s) "+g.3gpp.cs-voice", "+g.3gpp.cs-video" or both. The CUA shall include in the Contact header the SIP-URI and, if available the tel URI that can be used to establish a CSI call.

NOTE 4: The indicated tel URI corresponds to the MSISDN intended for the CS call setup.

- b) The CUA may in the Server header include the personal ME identifier and UE capability version. Both, the personal ME identifier and the UE capability version, shall be present or neither of them.

Upon the receipt of the 200 (OK) response the CUA acts in accordance with 3GPP TS 24.229 [5]. In addition, the CUA may locally update the UE capability information, the URIs associated with the remote CUA and the personal ME identifier of the remote CUA.

### 6.3.1.3 Session set-up – originating case

When the originating CUA wants to establish an IM session in combination with a CS call, the CUA shall apply the session initiation procedure specified in 3GPP TS 24.229 [5] with the following additions:

- a) The CUA shall in the Request-URI in the initial request either include:
  - 1) a tel URI that is the MSISDN intended for CS call set up; or
  - 2) a SIP-URI associated with the remote user available in the CUA.

NOTE: The MSISDN can only be included as a tel URI if it is in the international format.

- b) The CUA shall in the P-Preferred-Identity header in the initial request include an assumed registered tel URI corresponding to the MSISDN of the CUA.
- c) The CUA shall in the Accept-Contact header in the initial request include media feature tag(s) with the values "+g.3gpp.cs-voice", "+g.3gpp.cs-video" or both, marked as explicit.
- d) The Contact header shall include media feature tag(s) with the value(s) "+g.3gpp.cs-voice", "+g.3gpp.cs-video" or both.
- e) The CUA may in the User-Agent header include the personal ME identifier and UE capability version. Both, the personal ME identifier and the UE capability version, shall be present or neither of them.

If the response includes a personal ME identifier and a UE capability version and the combination of public user identity plus personal ME identifier plus UE capability version of the terminating UE is known to the CUA, the CUA may indicate capability information of the terminating UE to the user via the MMI.

#### 6.3.1.4 Session set-up – terminating case

When the terminating CUA receives an initial request, the terminating CUA shall apply the procedures as specified in 3GPP TS 24.229 [5]. The Contact header shall include media feature tag(s) with the value(s) "+g.3gpp.cs-voice", "+g.3gpp.cs-video" or both in the response(s), in accordance with RFC 3840 [6]. The UE may include the personal ME identifier and UE capability version in the Server header in the responses. Both, the personal ME identifier and the UE capability version, shall be present or neither of them.

If the request includes a personal ME identifier and a UE capability version and the combination of public user identity plus personal ME identifier of the originating UE is known to the CUA, the CUA may indicate capability information of the originating UE to the user via the MMI.

#### 6.3.1.5 CS call set-up – originating case

When the originating CUA wants to add the CS part of a CSICS call, it shall set up the call in accordance with 3GPP TS 24.008 [4] with the following additions:

- a) The CUA may include the radio environment information and the IM Status, the personal ME identifier and UE capability version in the user-user information (as defined in 3GPP TS 24.008 [4] Annex O) in the SETUP Message. Both, the personal ME identifier and the UE capability version, shall be present or neither of them. The handling of the user-user information element shall be in accordance with 3GPP TS 24.087 [9].

NOTE 1: If the CUA does not support the user-to-user 1 supplementary service, in accordance with 3GPP TS 22.087 [8] the radio environment information, the IM Status, the personal ME identity, and the UE capability version:

- will not be sent to the remote CUA,
- will be ignored if provided by the remote CUA.

- b) The CUA shall include in the called party BCD number either:

- an E.164 number as included in the P-Asserted-Identity header that was received during IM session establishment; or
- if this is not possible, the MSISDN of the remote user stored in UE.

NOTE 2: It is assumed that the originating CUA uses the COLP supplementary service, in accordance with 3GPP TS 22.081 [7] and UUS service 1, in accordance with 3GPP TS 22.087 [8]. The CLIR supplementary service in accordance with 3GPP TS 22.081 [7] is not in use in the originating CUA.

If the connected number received in the CONNECT message differs from the MSISDN number used in the SETUP message, the actions to be taken by the CUA are implementation dependent.

#### 6.3.1.6 CS call set-up – terminating case

When the terminating CUA receives a CS call, the CUA shall check if there any ongoing IM sessions, which were initiated with a request that included an Accept-Contact header with media feature tag(s) with the value(s) "+g.3gpp.cs-voice", "+g.3gpp.cs-video" or both.

- If there are ongoing IM sessions, which were initiated with a request that included an Accept-Contact header with media feature tag(s) with the values(s) "+g.3gpp.cs-voice", "+g.3gpp.cs-video" or both and if the received Calling Party BCD number corresponds to an entry in the P-Asserted-Identity header in an ongoing IM session, the CUA shall set up the call in accordance with 3GPP TS 24.008 [4]. The CUA may include the user-user information element with the radio environment information, the IM Status, personal ME identifier (as defined in 3GPP TS 24.008 [4] Annex O) and UE capability version in the CONNECT message, in accordance with 3GPP TS 24.087 [9]. Both, the personal ME identifier and the UE capability version, shall be present or neither of them.

- Otherwise, the CUA shall set up the call in accordance with subclause 7.3.1.6.

NOTE 1: If the CUA does not support the user-to-user 1 supplementary service, in accordance with 3GPP TS 22.087 [8] the radio environment information, the IM Status, the personal ME identifier and the UE capability version:

- will not be sent to the remote CUA,
- will be ignored if provided by the remote CUA.

NOTE 2 It is assumed that the terminating CUA uses the CLIP supplementary service, in accordance with 3GPP TS 22.081 [7] and UUS service 1, in accordance with 3GPP TS 22.087 [8] The COLR supplementary service in accordance with 3GPP TS 22.081 [7] is not in use in the terminating CUA.

NOTE 3: If the CUA does not support the user-to-user 1 supplementary service, in accordance with 3GPP TS 22.087 [8] the radio environment information, the IM Status, the personal ME identifier and UE capability version will not be sent to the remote CUA.

### 6.3.1.7 SDP exchange – UE capability information exchange

The CUA shall act in accordance with 3GPP TS 24.229 [5] when the CUA exchanges its capabilities with the remote CUA.

The following information may be included:

- a) Media types which can be supported as IMS media streams (i.e. media component definitions of IM sessions).
- b) Media format parameters for supported IMS media types (codecs, media file formats etc.).

NOTE: The media and codecs used for the CS call are not included in the SDP.

### 6.3.1.8 SDP offer answer – originating case

When a CUA wants to create a CSI communication, the CUA shall populate the SDP as specified in subclause 6.1 in 3GPP TS 24.229 [5].

### 6.3.1.9 SDP offer answer – terminating case

When a CUA wants to participate in a CSI communication, the CUA shall populate the SDP as specified in subclause 6.1 in 3GPP TS 24.229 [5].

---

## 7 Combining an IM session with an existing CS call

### 7.1 Introduction

(void)

### 7.2 Functional entities

#### 7.2.1 User equipment

A UE shall implement the role of a CUA as specified in subclause 7.3.1.

#### 7.2.2 Application server (AS)

An application server may be included for a CSI call. However, no specific CSI role is assigned to it.

## 7.3 Roles

### 7.3.1 CSI user agent (CUA)

#### 7.3.1.1 General

In addition to the procedures specified in subclause 7.3.1 of this document, the CUA shall support the procedures specified in 3GPP TS 24.229 [5] appropriate to the functional entity in which the CUA is implemented.

#### 7.3.1.2 Exchange of UE capability information – CS first scenario

When the CUA wants to exchange capabilities with the remote party, the CUA originating the OPTIONS request shall apply the procedure as specified in 3GPP TS 24.229 [5] with the following additions:

- a) The CUA shall in the Request URI include one of the following:
  - a URI, as received in P-Asserted-Identity header from the remote CUA during the terminal capability information exchange procedure; or, if this is not available;
  - a tel URI consisting of the Connected Number information element or the Calling Party BCD number received during establishment of the existing CS call, or the MSISDN used for the CS call set-up ( the Called Party BCD Number); or
  - a SIP URI associated with the remote user stored in the CUA.

NOTE 1: The MSISDN can only be included as a tel URI if it is in the international format.

NOTE 2: If no SIP URI or tel URI in accordance with above is available the CUA cannot initiate the OPTIONS request.

- b) The CUA shall in the P-Preferred-Identity header either include:
  - the MSISDN of the CUA as an assumed registered tel URI; or
  - an assumed registered SIP URI associated with the CUA.
- c) The CUA shall in the Accept-Contact header include media feature tag(s) with the values(s) "+g.3gpp.cs-voice", "+g.3gpp.cs-video" or both marked as explicit.
- d) The CUA may in the User-Agent header include the personal ME identifier and the UE capability version. Both, the personal ME identifier and the UE capability version, shall be present or neither of them.

The CUA answering with a 200 (OK) response to the OPTIONS request shall apply the procedure as specified in 3GPP TS 24.229 [5] with the following additions:

- a) The CUA shall in the Contact header include media feature tag(s) with the values(s) "+g.3gpp.cs-voice", "+g.3gpp.cs-video" or both. The CUA shall include in the Contact header the SIP-URI and, if available the tel URI that can be used to establish a CSI call.

NOTE 3: The indicated tel URI corresponds to the MSISDN intended for the CS call setup.

- b) The CUA may in the Server header include the personal ME identifier and the UE capability version. Both, the personal ME identifier and the UE capability version, shall be present or neither of them.

Upon the receipt of the 200 (OK) response, the CUA acts in accordance with 3GPP TS 24.229 [5]. In addition, the CUA may locally update the UE capability information for the remote user, the URIs associated with the remote CUA and the personal ME identifier of the remote CUA.

#### 7.3.1.3 Session set-up – originating case

When the originating CUA wants to add an IM session to a CS call, it shall apply the call initiation procedure specified in 3GPP TS 24.229 [5] with the following additions:



- a) The CUA shall in the Request URI in the initial request either include:
  - 1) the URI, as received in P-Asserted Identity header from the terminating CUA during the terminal capability information exchange procedure; or if this is not available
  - 2) a tel URI that either is:
    - the connected number information element, received during establishment of the existing CS call, and if this is not available the MSISDN used for the CS call set-up (the used Called Party BCD Number); or
    - the Calling Party BCD number information element, received during establishment of the existing CS call.
- b) The CUA shall in the P-Preferred Identity header in the initial request include an assumed registered tel URI corresponding to the MSISDN of the CUA.
- c) The CUA shall in the Accept-Contact header in the initial request include media feature tag(s) with the value(s) '+g.3gpp.cs-voice', a '+g.3gpp.cs-video' or both, marked as explicit.
- d) The Contact header shall include media feature tag(s) with the value(s) "+g.3gpp.cs-voice", "+g.3gpp.cs-video" or both.
- e) The CUA may in the User-Agent header include the personal ME identifier and the UE capability version. Both, the personal ME identifier and the UE capability version, shall be present or neither of them.

NOTE 1: If privacy is indicated it is not possible to correlate the CS call and the IM session.

#### 7.3.1.4 Session set-up – terminating case

When the terminating CUA receives an initial request for an IM session, the terminating CUA shall apply the procedures as specified in 3GPP TS 24.229 [5] with the following additions:

- a) If the Accept-Contact header in the INVITE request includes a media feature tag(s) with the value(s) "+g.3gpp.cs-voice", "+g.3gpp.cs-video" or both, the CUA shall check if one of the identities in the P-Asserted-Identity header matches the received Calling Party BCD number information element or Connected number information element received for an ongoing CS call. If there is a match, a CSICS call is established.

Otherwise, the CUA shall not consider this a CSI call/session.

- b) The Contact header shall include media feature tag(s) with the value(s) "+g.3gpp.cs-voice", "+g.3gpp.cs-video" or both in the response(s), in accordance with RFC 3840 [6].
- c) The UE may include the personal ME identifier and the UE capability version in the Server header in the responses. Both, the personal ME identifier and the UE capability version, shall be present or neither of them.

#### 7.3.1.5 CS call set-up – originating case

When the originating CUA wants to set up a CS call in combination with an IM Session, the CUA shall set up the call in accordance with 3GPP TS 24.008 [4] with the following additions:

- a) The SETUP message may include the user-user information element, which can include the radio environment information, the IM Status, the personal ME identifier and the UE capability version as specified in 3GPP TS 24.008 [4] Annex O. Both, the personal ME identifier and the UE capability version, shall be present or neither of them. The handling of the user-user information element shall be in accordance with 3GPP TS 24.087 [9].

If the CONNECT message includes a personal ME identifier and UE capability version and the combination of MSISDN plus personal ME identifier plus UE capability version of the terminating UE is known to the CUA, the CUA may indicate already available capability information of the terminating UE to the user via the MMI. If originating UE finds that terminating UE and its current radio environment supports simultaneous CS and PS services and terminating UE's IM Status indicates the remote UE is either IM subsystem registered or IM subsystem capable and willing to register to IM subsystem, then if allowed by the user's preference originating UE should attempt an IMS registration (if the UE is not already IMS registered).

NOTE 1: If the CUA does not support the user-to-user 1 supplementary service, in accordance with 3GPP TS 22.087 [8] the radio environment information, the IM Status, the personal ME identifier and the UE capability version:

- will not be sent to the remote CUA,
- will be ignored if provided by the remote CUA.

NOTE 2: It is assumed that the originating CUA uses COLP supplementary service, in accordance with 3GPP TS 22.081 [7] and UUS service 1, in accordance with 3GPP TS 22.087 [8] The CLIR supplementary service in accordance with 3GPP TS 22.081 [7] is not in use in the originating CUA.

### 7.3.1.6 CS call set-up – terminating case

When the terminating CUA receives a CS call:

- a) the CUA shall set up the call in accordance with 3GPP TS 24.008 [4] and may include the user to user information element in the CONNECT message with the radio environment information, the IM Status, the personal ME identifier and the UE capability version as defined in 3GPP TS 24.008 [4] Annex O. Both, the personal ME identifier and the UE capability version, shall be present or neither of them. The handling of the user-user information element shall be in accordance with 3GPP TS 22.087 [8].

If the SETUP message includes a personal ME identifier and UE capability version and the combination of MSISDN plus personal ME identifier plus UE capability version of the originating UE is known to the CUA, the CUA may indicate already available capability information of the originating UE to the user via the MMI. If terminating UE finds that originating UE and its current radio environment supports simultaneous CS and PS services and originating UE's IM Status indicates the remote UE is either IM subsystem registered or IM subsystem capable and willing to register to IM subsystem, then if allowed by the user's preference terminating UE should attempt an IMS registration (if the UE is not already IMS registered).

NOTE 1: If the CUA does not support the user-to-user 1 supplementary service, in accordance with 3GPP TS 24.087 [9] the radio environment information, the IM Status, the personal ME identifier and the UE capability version:

- will not be sent to the remote CUA
- will be ignored if provided by the remote CUA.

NOTE 2: It is assumed that the terminating CUA uses the CLIP supplementary service, in accordance with 3GPP TS 22.081 [7] and UUS service 1, in accordance with 3GPP TS 22.087 [8]. The COLR supplementary service in accordance with 3GPP TS 22.081 [7] is not in use in the terminating CUA.

### 7.3.1.7 SDP exchange – UE capability information exchange

The CUA shall act in accordance with subclause 6.3.1.7.

### 7.3.1.8 SDP offer-answer – originating case

The CUA shall act in accordance with subclause 6.3.1.8.

### 7.3.1.9 SDP offer-answer – terminating case

The CUA shall act in accordance with subclause 6.3.1.9.

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## 8 Interaction with supplementary services

### 8.1 Roles

#### 8.1.1 CSI user agent (CUA)

No protocol interactions for supplementary services are identified.

NOTE: Service interactions are specified in 3GPP TS 23.279 [3].

#### 8.1.2 CSI application server (CSI AS)

##### 8.1.2.1 Introduction

Other than the interactions to supplementary services described below, no further interactions to other supplementary services have been identified.

##### 8.1.2.2 Call Forwarding and Call Barring

The determination that Call Forwarding is being applied to terminating CUA or that Call Barring has been applied to the terminating CUA and the subsequent behaviour of the CSI AS is given in subclause 9.3.3.5.

##### 8.2.1.3 Call Completion

If the CSI AS receives indications that the terminating CUA has applied call hold or call retrieval, the CSI AS shall perform corresponding session modification toward the originating UE.

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## 9 IMS origination and CSI termination

### 9.1 Introduction

This subclause describes the detailed procedures for establishing sessions from UEs that use IMS origination to UEs that use CSI termination.

NOTE: The IMS origination and CSI termination are specific session establishment scenarios defined in TS 23.279 [3]. When a UE is considered to use IMS origination and when a UE is considered to use CSI termination is defined in subclause 9.2.1 of TS 23.279 [3].

### 9.2 Functional entities

#### 9.2.1 User equipment

For establishing sessions with IMS origination and CSI termination, two different UEs are involved.

- 1) The UE that uses CSI termination, which terminates sessions by receiving real time media over the CS domain. This UE implements the role of CUA as specified in subclause 6.3.1 and subclause 7.3.1.
- 2) The UE that uses IMS origination, which originates sessions by transmitting real time and non-real time media over an IP-CAN.

## 9.2.2 Application server (AS)

For supporting the control functionality required to handle sessions with IMS origination and CSI termination an application server shall implement the role of CSI AS as defined in subclause 9.3.3.

## 9.2.3 MGCF

No specific role in support of provision of CSI terminating session has been identified for the MGCF. In this version of this document, any CSI capability related information if provided to the MGCF will not be interworked by the MGCF.

# 9.3 Roles

## 9.3.1 Void

## 9.3.2 CSI UA (CUA)

The UE that uses CSI termination implements the role of CUA as specified in subclause 6.3.1 and subclause 7.3.1 with no additional requirements.

NOTE: For proper handling and delivery of CSI terminating sessions, a CUA needs to include the media feature tags applicable for CSI in the Contact header when performing the registration procedure.

## 9.3.3 CSI AS

### 9.3.3.1 General

The CSI AS shall support the procedure for the multimedia session splitting, modifying, combining and releasing.

To support a CSI session between a UE in IMS only mode and a CSI capable UE (CUA), the CSI AS shall support capability exchange with the CUA in the terminating CS domain on behalf of the CUA in the originating IM CN subsystem, when the CSI AS has determined that the network supports the required CSI functionality.

The CSI AS can use the media feature tags (e.g. g.3gpp.cs-voice and g.3gpp.cs-video) obtained from the terminating CUA to perform termination logic.

If the media feature tags indicate that the CUA has both voice and video capabilities and that CSI AS knows that video interworking to the CS domain is supported, then the CSI AS can use the CS domain to terminate any realtime voice and video sessions to the CUA. However the IM CN subsystem can also be used for the video component of the real time conversation.

NOTE 1: The knowledge that the IM CN subsystem supports video interworking to the CS domain is configured in the CSI AS.

The procedure of multimedia session splitting is specified in the subclause 9.3.3.3.

NOTE 2: The CSI AS obtains the media feature tags of the terminating CUA from the reg-event package.

### 9.3.3.2 Retrieval of public user identity information

The CSI AS can obtain information about the public user identities of the CUA by using the reg event package as specified in 3GPP TS 24.229 [5], or via the Sh interface as specified in 3GPP TS 29.328 [16].

The information about the public user identities can also be provisioned in the CSI AS.

NOTE: A third party REGISTER request received by the CSI AS can be triggered by a CUA including in the Contact header of the REGISTER request the media feature tag value(s) of "+g.3gpp.cs-voice" and "+g.3gpp.cs-video" or both.

### 9.3.3.3 Session Splitting towards terminating CUA

#### 9.3.3.3.1 General

When the CSI AS receives at SIP INVITE request including the SDP media descriptions for both voice media and non-voice media(e.g. MSRP) from the originating UE, the CSI AS shall check if the received dialog identifier in the request matches a dialog identifier for an ongoing session in the CSI AS:

- 1) if there is no match, the CSI AS performs the session splitting procedure as follows:
  - a) for the non-voice media, the CSI AS shall initiate the new SIP INVITE request requesting the same non-voice media as received in the incoming INVITE request towards the CUA via IM CN subsystem as per subclause 9.3.3.3.3; and
  - b) for the voice media, the CSI AS shall initiate the new SIP INVITE request requesting the same voice media as received in the incoming INVITE request towards the CUA via CS domain through the MGCF as per subclause 9.3.3.3.2.
- 2) if there is a match, the CSI AS shall act as per subclause 9.3.3.4.3.

When the CSI AS receives the SIP INVITE request including the SDP media description from the originating UE for voice media only, the CSI AS shall check if the dialog identifier in the request matches a dialog identifier for an ongoing session in the CSI AS;

- 1) if there is no match, the CSI AS shall follow the procedure in subclause 9.3.3.3.2; or
- 2) otherwise the CSI AS shall follow the procedures as specified in subcaluse 9.3.3.4.2.

When the CSI AS receives the SIP INVITE request including the SDP media description from the originating UE for non-voice media only, the CSI AS shall check if the dialog identifier in the request matches a dialog identifier for an ongoing session in the CSI AS:

- 1) if there is no match, the CSI AS shall follow the subclause 9.3.3.3.3; or
- 2) otherwise the CSI AS shall follow the subcaluse 9.3.3.4.1.

For the standalone SIP request and the SIP request initiating a new dialog other than the SIP INVITE request, the CSI AS shall follow subclause 5.7.5 of 3GPP TS 24.229[5] without splitting.

#### 9.3.3.3.2 Call Initiation towards the CS domain

When the CSI AS receives an initial SIP INVITE request that requests an IMS session with voice media, the CSI AS shall perform the third party call control, as specified 5.7.5 of 3GPP TS 24.229[5] including the following:

- 1) The CSI AS creates the SIP INVITE request setting the Request-URI and To header to a Tel-URI which is an alias of the SIP-URI which is included in the received SIP INVITE request;

NOTE: The CSI AS maps between SIP-URI and Tel-URI from information provided by the HSS about alias URIs of the registered identity, as specified in 3GPP TS 29.328[16].

- 2) The CSI AS adds the address of BGCF in the Route header to be located behind of the address of S-CSCF;
- 3) The CSI AS populates the SDP with the voice media parameters which were included in the SIP INVITE request from the originating UE; and
- 4) The CSI AS sends the SIP INVITE request towards the S-CSCF for further routeing.

If the CSI AS does not take any action for the received SIP INVITE request, it shall apply the procedure described in subclause 5.7.4 of 3GPP TS 24.229 [5].

#### 9.3.3.3.3 Session Initiation towards the IM CN subsystem

When the CSI AS initiates a session towards the CUA for a non-voice media the CSI AS shall;

- 1) operate as an application server performing 3<sup>rd</sup> party call control acting as a routing B2BUA, as specified in subclause 5.7.5 of 3GPP TS 24.229 [5] for this request and all future requests and in the same dialog;
- 2) set the Request-URI and the To header field to the SIP-URI for the CUA in the outgoing SIP INVITE request;
- 3) populate the SDP with the non-voice media parameters which were included in the SIP INVITE request or a SIP UPDATE request from the originating UE into the outgoing SIP INVITE request;
- 4) include in the Accept-Contact header field the feature tag(s) with the value(s) "+g.3gpp.cs-voice" or "+g.3gpp.cs-video" or both, marked as explicit; and
- 5) send the SIP INVITE request towards the S-CSCF for further routing.

When the CSI AS receives a SIP response from the CUA, the CSI AS shall act in accordance with subclause 9.3.3.5.

When the CSI AS initiates a session towards the originating UE the CSI AS shall:

- 1) operate as an application server performing 3<sup>rd</sup> party call control acting as an initiating B2BUA, specified in subclause 5.7.5 of 3GPP TS 24.229 [5] for this request and all future requests in the same dialog;
- 2) set the Request-URI and the To header field in the outgoing SIP INVITE request to the SIP URI as indicated in the P-Asserted-Identity header field received from the originating UE;
- 3) populate the SDP with the media components which were included in the SDP of a SIP re-INVITE request or a SIP UPDATE request and not possible to use on the existing session to the originating UE in the outgoing SIP INVITE request; and
- 4) send the SIP INVITE request towards the S-CSCF for further routing.

When the CSI AS receives a SIP response from the originating UE the CSI AS shall handle the response by following the rules of the 3GPP TS 24.229 [5].

#### 9.3.3.3.4 Subsequent requests

When the CSI AS receives a subsequent SIP request from the originating UE before it has provided the final SIP response to the originating UE (i.e. in early dialog), the CSI AS shall act as follows:

- 1) The SIP request which is triggered by the subsequent SIP request from the originating UE shall include exactly the same media components as the SIP requests triggered by the initial SIP INVITE request; and
- 2) the CSI AS shall not send a SIP request to the already established call leg if the CSI AS has received the final SIP response for that call leg. Instead the CSI AS shall include the media description indicating the media on the established call leg is ready when responding back to the originating UE.

Upon receipt of the SIP re-INVITE request or the SIP UPDATE request from the originating UE after the session is established, the CSI AS shall follow subclause 9.3.3.4. Handling of other subsequent SIP requests is out of scope of this version of the document.

#### 9.3.3.4 Session Modification

##### 9.3.3.4.1 Adding an IM Session to an existing CS call

When the CSI AS receives a SIP re-INVITE request or a SIP UPDATE request from the originating UE and the received SDP includes other "m=" lines than "m=audio" line that have non-zero port numbers and there do not exist any non voice media session to the CUA, the CSI AS shall initiate a SIP INVITE request requesting the same non-voice media as received in the incoming re-INVITE request or UPDATE request towards the CUA via the IM CN subsystem as specified in the subclause 9.3.3.3.3. If there exist a session the CSI-AS shall send an SIP re-INVITE or SIP UPDATE request to the CUA in accordance with subclause 9.3.3.4.3.

When the CSI AS receives the SIP response message from the CUA, it shall be processed according to subclause 9.3.3.5.

When the CSI AS receives a SIP re-INVITE request or a SIP UPDATE request from the CUA and the CSI AS is required to setup a new session the set up is performed in accordance with subclause 9.3.3.3.3. Otherwise the CSI-AS shall send a SIP re-INVITE request or a SIP UPDATE request to the originating UE including:

- the Request-URI set to the URI received from the originating UE in the Contact header; and
- a new SDP offer, including the media characteristics as received in the incoming SIP re-INVITE or UPDATE request, by following the rules of the 3GPP TS 24.229 [5].

NOTE: Whether the CSI AS uses the existing dialog to the originating UE or uses a new dialog can depend on local policy and received signalling information in the SDP from the originating UE.

When the CSI AS receives a SIP response message from the originating UE, the responses shall be processed according to the rules specified in 3GPP TS 24.229 [5].

#### 9.3.3.4.2 Adding a CS call to an existing IMS Session

When the CSI AS receives a SIP re-INVITE request or a SIP UPDATE request and the received SDP includes an additional "m=audio" line which has non-zero port;

- the CSI AS shall check if there is an ongoing voice call between the CUA and the CSI AS which is associated with the same session in which the SIP re-INVITE request or a SIP UPDATE request is received;
  - a) If a voice call between the CUA and the CSI AS via CS domain does not exist, the CSI AS shall initiate a SIP INVITE requesting the same voice media towards the CUA as specified in the subclause 9.3.3.3.2 ;
  - b) Otherwise, the CSI AS shall reply with 4xx response.

NOTE: The multicall setup is not supported within this specification.

When the CSI AS receives the SIP response from the MGCF, it shall be processed according to the subclause 9.3.3.5.

When the CSI AS receives a SIP re-INVITE request or a SIP UPDATE request and the received SDP includes an additional entry in the existing "m=audio" line which has non-zero port;

- the CSI AS shall check if there is an ongoing voice call between the CUA and the CSI AS which is associated with the same session in which the SIP re-INVITE or a SIP UPDATE request is received;
  - a) If a voice call between the CUA and the CSI AS via CS domain does not exist, the CSI AS shall initiate a SIP INVITE requesting the same voice media towards the CUA as specified in the subclause 9.3.3.3.2;
  - b) Otherwise the CSI AS shall send an SIP re-INVITE or a SIP UPDATE request to the CUA including:
    - the Request-URI set to the URI received from the MGCF in the Contact header; and
    - a new SDP offer, including the media characteristics as received in the SIP re-INVITE or a SIP UPDATE request , by following the rules of the 3GPP TS24.229 [5].

#### 9.3.3.4.3 Modifying an existing CS call and an existing IMS Session

When the CSI AS receives a SIP re-INVITE request or a SIP UPDATE request where the dialog identifier included in the request matches with a dialog identifier of an ongoing session then;

- 1) if the SDP media description for voice in the received SIP re-INVITE request or a SIP UPDATE request is updated;
  - a) the CSI AS shall send a SIP re-INVITE request or a SIP UPDATE request to the MGCF including;
    - the Request-URI and the To header field set to the URI received from the MGCF in the Contact header field; and
    - a new SDP offer, including the media characteristics as received in the SIP re-INVITE or A SIP UPDATE request , by following the rules of 3GPP TS 24.229 [5].
  - b) When the CSI AS receives the SIP response from the MGCF, the SIP response shall be processed according to subclause 9.3.3.5.

- 2) if the SDP media description for non-voice in the received SIP re-INVITE request or SIP UPDATE request is updated;
  - a) the CSI AS shall send a SIP re-INVITE request or SIP UPDATE request to the CUA including;
    - the Request-URI and the To header field set to the URI received from the CUA in the Contact header field; and
    - a new SDP offer, including the media characteristics as received in the SIP INVITE request or a SIP UPDATE request, by following the rules of the 3GPP TS 24.229 [5].
  - b) When the CSI AS receives the SIP response from the CUA, the SIP response shall be processed according to subclause 9.3.3.5.
- 3) if the SDP media description in the received SIP re-INVITE request includes one or more "m=" lines with the port number set to zero, the CSI AS shall act as specified in the subclause 9.3.3.6.

### 9.3.3.5 Session Combining

The CSI AS having established one call leg to the CUA and another call leg to the MGCF shall combine the SIP responses from the CUA and the MGCF as follows:

The CSI AS shall act as a B2BUA as per subclause 5.7.5 of 3GPP TS 24.229 [5] with the following additions.

If the CSI AS receives a SDP answer from both of the two call legs, the CSI AS shall include a SDP answer in a SIP response which includes the SDP from both legs in accordance with 3GPP TS 24.229 [5], i.e. the CSI AS shall respond to the SIP INVITE request from the originating UE when all SDP media descriptions from both of the two split call legs are known to the CSI AS. The SIP response code that is sent to the originating UE by the CSI AS depends on the SIP response received from the two call legs and is specified in table 9.3.3.5A.

If the CSI AS receives only SDP answer from one of the legs, the CSI AS shall include the SDP answer in a SIP response in accordance with 3GPP TS 24.229[5]. The SIP response code that is sent to the originating UE by the CSI AS depends on the SIP response received from the two call legs and is specified in table 9.3.3.5A.

If the CSI AS receives SIP responses without any SDP, the SIP response code that is sent to the originating UE by the CSI AS depends on the SIP response received from the two call legs and is specified in table 9.3.3.5A.

In addition to table 9.3.3.5A, when the CSI AS receives only one SDP answer from one leg while having a SIP 3xx – 6xx error response(s) from the other leg, the CSI AS shall:

- 1) if the SIP error response is received from the MGCF, provide the SIP response to the originating UE as shown in table 9.3.3.5A but with the port of the voice media set to 0 in the SDP answer provided to the originating UE; or
- 2) if the SIP error response is received from the CUA, provide the SIP response to the originating UE as shown in table 9.3.3.5A but with the port of the non-voice media set to 0 in the SDP answer provided to the originating UE.

When the CSI AS receives the SIP responses for the SIP INVITE requests that were sent to the CUA and the MGCF, the CSI AS shall:

- 1) provide the SIP response to the originating UE as per subclause 5.7.5 of 3GPP TS 24.229 [5]. In this SIP response except the final SIP response, the CSI AS shall provide all SDP media description that is received from the CUA and the MGCF in the SDP answer;
- 2) provide the combined final SIP response to the originating UE only if the final SIP responses arriving from each of the two call legs, have the same SIP status code; or
- 3) provide the SIP response as shown in the table 9.3.3.5A below to the originating UE if the SIP responses arrived from the two call legs have different status codes;
- 4) provide a combined SIP 180 (Ringing) response to the originating UE only if the SIP 180 (Ringing) responses arrive from each early dialog of the two call legs. When the precondition mechanism is used, the CSI AS shall not provide a combined SIP 180 (Ringing) response until the required resource reservations are met for each of the two call legs;



- 5) provide - when precondition mechanism is used - a combined SIP 200 (OK) response to the SIP UPDATE request to the originating UE only if both SIP 200 (OK) responses to the SIP UPDATE requests which had been sent in each of the two call legs arrive;
- 6) provide a SIP response to the originating UE including "100rel" in the Require header field if at least one of SIP responses from either call legs include "100rel" in the Required header; and
- 7) if due to forking more than one SIP response is received for the non-voice media, choose one which comes from the CUA and initiate SIP CANCEL request(s) to cancel the other forked dialog(s).

NOTE: The CSI AS recognizes the SIP response from the CUA by regarding the media feature tag or by comparing contact address in the Contact header with the terminal information stored in the CSI AS.

If one of two call legs is established earlier than the other, the CSI AS upon receiving a SIP response from the early dialog of the other call leg shall include the media description showing the media on the established call leg is ready when responding back to the originating UE.

If the CSI AS does not receive any response from one of call legs towards the CUA within an implementation dependent time, the CSI AS shall reject the media of that corresponding leg that has timeout, by indicating such rejection within the SDP body media descriptions when the CSI AS response to the originating UE. For the call leg that has not responded by timeout, the CSI AS may initiate the SIP CANCEL request to cancel the pending session of the corresponding call leg. Any responses received after this timeout are ignored by the CSI AS.

**Table 9.3.3.5A: SIP response (CSI AS to the originating UE)**

Response from MGCF	Response from CUA	Response provided by CSI AS to the originating UE
180	200	180(When preconditions between the UE and CSI AS are met)
200	180	
183	200	183
200	183	
4xx/5xx/6xx	180	180(When preconditions between the UE and CSI AS are met)
180	4xx/5xx/6xx	
4xx/5xx/6xx	183/200	183/200
183/200	4xx/5xx/6xx	
181	Any response	Response will be based on local policy, i.e. keep the successfully established CS call or/and IMS session with the calling party.
Any response	181	
4xx/5xx/6xx	4xx/5xx/6xx	4xx/5xx/6xx (according to the best response procedures as specified in IETF RFC 3261[17])
603	Any response	Either when responding back to the originating UE provide an SDP answer rejecting the media component which is subject to Call Barring and thereby allow the originating UE to renegotiate, or respond to the calling UE with the response indicating that call is barred and initiate a SIP CANCEL request to cancel the other call leg being established toward the CUA.
Any response	603	Either when responding back to the originating UE provide an SDP answer rejecting the media component which is subject to Call Barring and thereby allow the originating UE to renegotiate, or respond to the calling UE with the response indicating that call is barred and initiate a SIP CANCEL request to cancel the other call leg being established toward the CUA.

### 9.3.3.6 Session Release

If the CSI AS receives the SIP re-INVITE request or a SIP UPDATE request which includes "m=" lines where the port number has been changed to zero from that provided in the previous SDP offer from the originating UE, then:

- 1) if the received SDP media description includes the port number set to zero for all "m=audio" lines, the CSI AS shall initiate a SIP BYE request to release the session toward the MGCF which is associated with the session where the SIP re-INVITE or SIP UPDATE request is received;
- 2) if the received SDP media description includes the port number set to zero for a "m=audio" line, the CSI AS shall initiate a SIP re-INVITE or a SIP UPDATE request including a SDP with the received "m=audio" lines in accordance with 3GPP TS 24.229 [5];
- 3) if the received SDP media description includes the port number set to zero for non-"m=audio" lines, the CSI AS initiates a SIP re-INVITE request including SDP media description for non-voice toward the CUA; or
- 4) if the received SDP media description includes the port numbers set to zero for all the "m=" lines other than "m=audio" the CSI AS initiates a SIP BYE request towards the CUA in the IM CN subsystem.

If the CSI AS receives the SIP BYE request from the originating UE, then;

- 1) the CSI AS shall initiate the SIP BYE request to release the session towards the CUA which is associated to the session where the SIP BYE request is received and,
- 2) the CSI AS shall initiate the SIP BYE request to release the session towards the MGCF which is associated to the session where the SIP BYE request is received.

If the CSI AS receives the SIP BYE request from the CUA or the MGCF, then;

- 1) if there is no remaining session other than the session the SIP BYE request belongs to, the CSI AS initiates a SIP BYE request toward the originating UE or,
- 2) otherwise, the CSI AS initiates a SIP re-INVITE request to remove corresponding media towards the originating UE.

### 9.3.3.7 Unregistered Service

To allow for the provision of unregistered service, the CSI AS may keep the UE capability information after the user has been deregistered from the IM CN subsystem. The duration of validity of UE capability information after a UE has deregistered from the IM CN subsystem is an operator policy.

When the CSI AS receives a SIP INVITE request destined for a user who is unregistered to the IM CN subsystem, the CSI AS shall follow 9.3.3.3 with the following addition.

- The CSI AS shall not initiate a session toward the IM CN subsystem and shall in the SDP answer provided to the originating UE, set the port to 0 for all media except the voice media.

NOTE 1: In the case that the CSI terminating service is part of a series of unregistered services for the user, care is taken that the triggering of the CSI AS does not negatively affect the provisioning of other unregistered services for that user in the terminating network, ie. the terminating user's service profile iFC is so set up to ensure orderly triggering of the CSI AS and the other unregistered services for that terminating user.

NOTE 2: Forwarding an incoming call to the CUA via CS domain by using Call Forwarding Supplementary Service is not in the scope of this document.

### 9.3.3.8 Capability exchange with the CSI UE

When the CSI AS has determined that the network supports the functionality required for CSI sessions between UEs in IMS only mode and a CSI capable UE (CUA) and has received an initial INVITE request, then the CSI AS shall perform CSI capability exchange with the terminating CSI UE on behalf of the originating IMS UE. The SIP INVITE request generated by the CSI AS shall include the User-to-User header as defined in draft-ietf-cuss-sip-uu [18]. This User-to-User header shall contain the following CSI capability information in the User-user information element as defined in 3GPP TS 24.008 [4] Annex O:

- 1) The IM Status is configured as IM subsystem registered;
- 2) The radio environment is configured as capable of simultaneous services.

NOTE: 3GPP TS 24.008 [4] specifies that the user-user information element is used to convey information between the mobile station and the remote ISDN user. A package for interworking user-to-user information with the ISDN is defined by draft-ietf-cuss-sip-uu-isdn [19].

---

## Annex A (normative): Media feature tags defined within the current document

### A.1 General

This subclause describes the media feature tag definitions that are applicable for the 3GPP IM CN Subsystem for the realisation of CSI.

---

### A.2 Definition of media feature tag g.3gpp.cs-voice

Media feature-tag name: g.3gpp.cs-voice

ASN.1 Identifier: 1.3.6.1.8.2.1

Summary of the media feature indicated by this tag: This feature-tag indicates that the device supports circuit switched voice when combining circuit switched calls and IM sessions.

Values appropriate for use with this feature-tag: Boolean.

The feature-tag is intended primarily for use in the following applications, protocols, services, or negotiation mechanisms: This feature-tag is most useful in a communications application, for describing the capabilities of a device, such as a phone or PDA.

Examples of typical use: Indicating that a mobile phone can support voice in a circuit switched environment within the context of combining a circuit switched voice call with an IM session.

Related standards or documents: 3GPP TS 24.279: "Combining Circuit Switched (CS) and IP Multimedia Subsystem (IMS) services, stage 3"

Security Considerations: Security considerations for this media feature-tag are discussed in subclause 11.1 of RFC 3840 [6].

---

### A.3 Definition of media feature tag g.3gpp.cs-video

Media feature-tag name: g.3gpp.cs-video

ASN.1 Identifier: 1.3.6.1.8.2.2

Summary of the media feature indicated by this tag: This feature-tag indicates that the device supports circuit switched video when combining circuit switched video calls and IM sessions.

Values appropriate for use with this feature-tag: Boolean.

The feature-tag is intended primarily for use in the following applications, protocols, services, or negotiation mechanisms: This feature-tag is most useful in a communications application, for describing the capabilities of a device, such as a phone or PDA.

Examples of typical use: Indicating that a mobile phone can support video in a circuit switched environment within the context of combining a circuit switched video call with an IM session.

Related standards or documents: 3GPP TS 24.279: "Combining Circuit Switched (CS) and IP Multimedia Subsystem (IMS) services, stage 3"

Security Considerations: Security considerations for this media feature-tag are discussed in subclause 11.1 of RFC 3840 [6].

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# Annex B (informative): Example signalling flows for the combining of CS calls with IM sessions

## B.1 Scope of signalling flows

This annex gives examples of signalling flows for the combination of CS calls with IM sessions within the IP Multimedia (IM) Core Network (CN) subsystem based on the Session Initiation Protocol (SIP).

These detailed signalling flows expand on the overview information flows provided in 3GPP TS 23.279 [3].

---

## B.2 Introduction

### B.2.1 General

The signalling flows provided in this annex follow the methodology developed in 3GPP TS 24.228 [11]. The following additional considerations apply:

- a) 3GPP TS 24.228 [11] shows separate signalling flows with no configuration hiding between networks, and with configuration hiding between networks. There is no combining of CS calls with IM sessions functionality associated with this hiding, and therefore such separate signalling flows are not shown in the present document;
- b) 3GPP TS 24.228 [11] breaks down the functionality of the various CSCFs. Such intervening activity in the CSCFs is in general not relevant to showing the functionality combining of CS calls with IM sessions, and therefore the CSCFs are collapsed into a single entity labelled "Intermediate IM CN subsystem entities".

### B.2.2 Key required to interpret signalling flows

The key to interpret signalling flows specified in 3GPP TS 24.228 [11] subclause 4.1 applies with the additions specified below:

- tel:+12125551111 is the tel URI, PMI-0007 is the personal ME identifier and UCV-04 is the UE capability version relating to CUA#1;
- tel:+12125552222 is the tel URI, PMI-0EA2 is the personal ME identifier and UCV-0D is the UE capability version relating to CUA#2;
- tel:+12125553333 is the tel URI of the UE#1 and
- sip:user3\_public1@home1.net is the SIP URI of the UE#1.

Each signalling flow table contains descriptions for headers where the content of the header is new to that signalling flow, as is already performed in 3GPP TS 24.228 [11].

However, 3GPP TS 24.228 [11] includes extensive descriptions for the contents of various headers following each of the tables representing the contents of the signalling flows. Where the operation of the header is identical to that shown in 3GPP TS 24.228 [11], then such text is not reproduced in the present document.

Additional text can also be found on the contents of headers within 3GPP TS 24.228 [11] in addition to the material shown in the present document.

In order to differentiate between messages for SIP and media, the following notation is used:

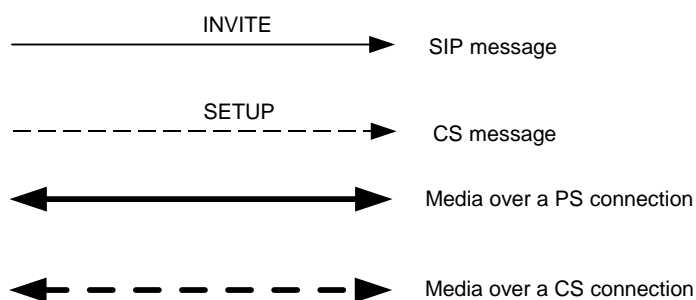


Figure B.2.2-1: Signalling flow notation

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## B.3 Signalling flows demonstrating CSI session setup when no CS call has yet been set up

### B.3.1 Introduction

This subclause provides signalling flows for CSI session setup, using session-based messaging as an example, established before any CS call has been established.

The signalling flows are based on the session establishment flows in subclause A.3 of 3GPP TS 24.247 [10], with some additions related to CSI. The UAC includes the media feature tags `g.3gpp.cs-voice` and `g.3gpp.cs-video` in the Accept-Contact header.

### B.3.2 Establishing a CSI session when no CS call has yet been set up

Figure B.3.2-1 shows the establishment of an MSRP session between two users as well. A CS call is then established after MSRP messages are communicated over the established PS connection.

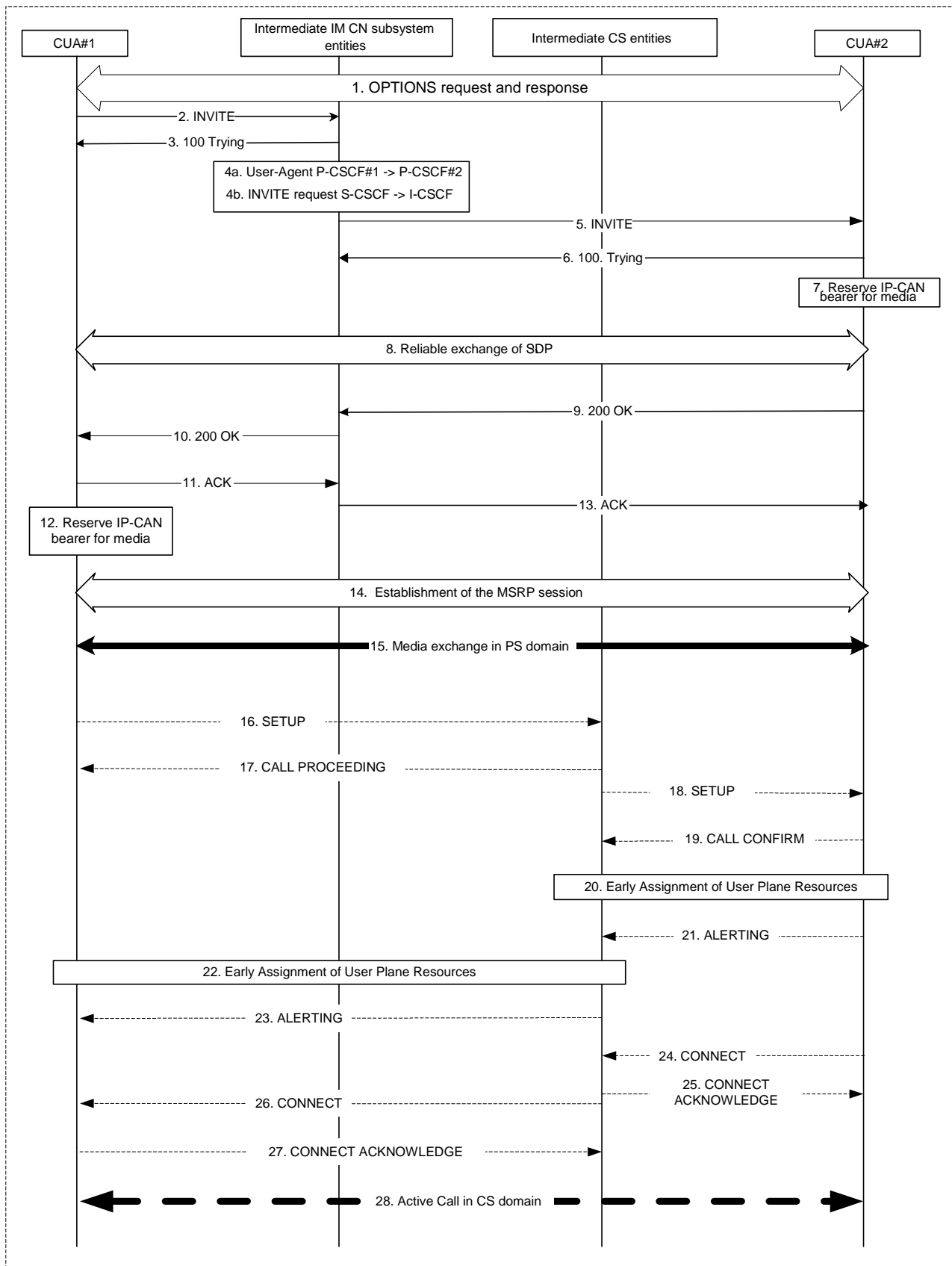


Figure B.3.2-1: Establishment of a session before a CS call is established

The details of the signalling flows are as follows:

1. UE capabilities exchange

Both UEs have performed a capability exchange. In the here shown example both UEs transfer their UE capability version and personal ME identifier as a part of the capability exchange. An example for capability exchange without the usage of UE capability version and personal ME identifier can be found in subclause B.6.

## 2. INVITE request (CUA#1 to P-CSCF#1) - see example in table B.3.2-2

The originating CUA wants to initiate a session with the terminating CUA. In this example, the originating CUA creates a local MSRP URL, which can be used for the communication between the two user agents. It builds a SDP Offer containing the generated MSRP URL and assigns a local port number for the MSRP communication.

The originating CUA declares its capabilities for CS-voice and CS-video, and requests to reach a UE with cs-voice or cs-video capabilities. The originating CUA includes its personal ME identifier and UE capability version in the User-Agent header.

**Table B.3.2-2: INVITE request (CUA#1 to P-CSCF#1)**

```

INVITE tel:+12125552222 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: <tel:+12125551111>
P-Access-Network-Info: 3GPP-UTRAN-TDD; utran-cell-id-3gpp=234151D0FCE11
Privacy: none
From: <sip:user1_public1@home1.net>; tag=171828
To: <tel:+12125552222>
Call-ID: cb03a0s09a2sdfgk490333
Cseq: 127 INVITE
Require: sec-agree
Proxy-Require: sec-agree
Supported: gruu
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>;+g.3gpp.cs-voice;+g.3gpp.cs-video
Accept-Contact: *,+g.3gpp.cs-voice, +g.3gpp.cs-video;explicit
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE, OPTIONS
User-Agent: PMI-0007, UCV-04
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
m=message 3402 TCP/MSRP *
a=accept-types:message/cpim text/plain text/html image/jpeg image/gif video/3gpp
a=path:msrp://[5555::aaa:bbb:ccc:ddd]:3402/s111271;tcp
a=max-size:131072

```

**SDP** The SDP contains a set of content types supported by CUA#1 and desired by the user at CUA#1 for this session in the accept-types attribute and indicates the maximum size message that can be received by CUA#1 in the max-size attribute.

## 3. 100 (Trying) response (P-CSCF#1 to CUA#1)

The P-CSCF responds to the INVITE request with a 100 (Trying) provisional response as described in subclause A.3 of 3GPP TS 24.247 [10].

### 4a. User-Agent (P-CSCF#1 to P-CSCF#2)

The User-Agent header field is transparently passed from P-CSCF#1 to P-CSCF#2.

### 4b. INVITE request (S-CSCF#1 to I-CSCF#2) - see example in table B.3.2-4b

S-CSCF#1 forwards the INVITE request to the I-CSCF#2, after mapping the tel URI to a SIP-URI.



**Table B.3.2-4b: INVITE request (S-CSCF#1 to I-CSCF#2)**

```
INVITE sip:user2_public1@home2.net SIP/2.0
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
Max-Forwards: 68
Record-Route: <sip:scscf1.home1.net;lr>, <sip:pcscf1.visited1.net;lr>
P-Asserted-Identity:
P-Charging-Vector: icid-value="AyretyU0dm+602IrT5tAFrbHLso=023551024"; orig-loi=home1.net
Privacy:
From:
To:
Call-ID:
Cseq:
Supported:
Contact:
Accept-Contact:
Allow:
Accept:
User Agent:
Content-Type:
Content-Length: (...)

v=
o=
s=
c=
t=
m=
a=
a=
a=
```

**5. INVITE request (P-CSCF#2 to CUA#2) – see example in table B.3.2-5**

P-CSCF#2 forwards the INVITE request to the terminating CUA.

**Table B.3.2-5: INVITE request (P-CSCF#2 to CUA#2)**

```

INVITE sip:[5555::eee:fff:aaa:bbb]:8805;comp=sigcomp SIP/2.0
Via: SIP/2.0/UDP pcscf2.visited2.net:5088;comp=sigcomp;branch=z9hG4bK361k21.1, SIP/2.0/UDP
scscf2.home2.net;branch=z9hG4bK764z87.1, SIP/2.0/UDP
icscf2_s.home2.net;branch=z9hG4bK871y12.1, 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
Max-Forwards: 65
Record-Route: <sip:pcscf2.visited2.net:5088;lr;comp=sigcomp>, <sip:scscf2.home2.net;lr>,
<sip:scscf1.home1.net;lr>, <sip:pcscf1.visited1.net;lr>
P-Asserted-Identity:
Privacy:
From:
To:
Call-ID:
Cseq:
Supported:
Contact:
Accept-Contact:
Allow:
Accept:
User-Agent:
P-Called-Party-ID:
Content-Type:
Content-Length: (...)

v=
o=
s=
c=
t=
m=
a=
a=
a=

```

**6. 100 (Trying) response (CUA#2 to P-CSCF#2)**

The terminating CUA sends a 100 (Trying) provisional response to P-CSCF#2 as described in subclause A.3 of 3GPP TS 24.247 [10].

**7. Reserve IP-CAN bearer for media**

The terminating CUA accepts the message session. The terminating CUA reserves an IP-CAN bearer for the message session media component.

**8. Reliable exchange of SDP**

SDP is conveyed between CUA#1 and CUA#2. The mechanism for this is not relevant for the example and is not described.

**9. 200 (OK) response (CUA#2 to P-CSCF#2) – see example in table B.3.2-9**

After reserving an IP-CAN bearer for the message session media component the terminating CUA sends a 200 (OK) response for the INVITE request containing SDP that indicates that the terminating CUA has accepted the message session and listens on the MSRP TCP port returned in the path attribute in the answer for a TCP SETUP from the originating CUA.

CUA#2 declares support only for CS-voice, not CS-video in its Contact header. The terminating CUA includes its personal ME identifier and UE capability version in the Server header.

**Table B.3.2-9: 200 (OK) response (CUA#2 to P-CSCF#2)**

```

SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf2.visited2.net:5088;comp=sigcomp;branch=z9hG4bK361k21.1, SIP/2.0/UDP
scscf2.home2.net;branch=z9hG4bK764z87.1, SIP/2.0/UDP
icscf2_s.home2.net;branch=z9hG4bK871y12.1, 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:pcscf2.visited2.net:5088;lr;comp=sigcomp>>, <sip:scscf2.home2.net;lr>,
<sip:scscf1.home1.net;lr>, <sip:pcscf1.visited1.net;lr>
Privacy: none
P-Access-Network-Info: 3GPP-UTRAN-TDD; utran-cell-id-3gpp=234151D0FCE11
From: <sip:user1_public1@home1.net>; tag=171828
To: <tel:+12125552222>;tag=314159
Call-ID: cb03a0s09a2sdfg1kj490333
Cseq: 127 INVITE
Supported: gruu
Contact: <sip: user2_public1@home2.net;gr=urn:uuid:2ad8950e-48a5-4a74-8d99-
ad76cc7fc74;comp=sigcomp>;+g.3gpp.cs-voice
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE, OPTIONS
Server: PMI-0EA2, UCV-0D
Content-Type: application/sdp
Content-Length: (...)

v=0
o=- 2987933615 2987933617 IN IP6 5555:: eee:fff:aaa:bbb
s=-
c=IN IP6 5555::eee:fff:aaa:bbb
t=0 0
m=message 3402 TCP/MSRP *
a=accept-types:text/plain text/html message/cpim
a=path:msrp://[5555::eee:fff:aaa:bbb]:3402/s234167;tcp
a=max-size:65536

```

**SDP** The SDP contains the set of offered content types supported by CUA#2 and desired by the user at CUA#2 for this session in the accept-types attribute and indicates the maximum size message that can be received by CUA#2 in the max-size attribute.

#### 10. 200 (OK) response (P-CSCF#1 to CUA#1) – see example in table B.3.2-10

P-CSCF#1 forwards the 200 (OK) response to the originating CUA.

**Table B.3.2-10: 200 (OK) response (P-CSCF#1 to CUA#1)**

```

SIP/2.0 200 OK
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Record-Route: <sip:pcscf2.visited2.net;lr>, <sip:scscf2.home2.net;lr>,
<sip:scscf1.home1.net;lr>, <sip:pcscf1.visited1.net:7531;lr;comp=sigcomp>
P-Asserted-Identity:
Privacy:
From:
To:
Call-ID:
CSeq:
Supported:
Contact:
Allow:
Server:
Content-Type:
Content-Length:

v=
o=
s=
c=
t=
m=
a=
a=
a=

```

**P-Asserted-Identity:** This header field is added by the intermediate IM CN subsystem entities in accordance with 3GPP TS 24.229 [5].

#### 11. ACK request (CUA#1 to P-CSCF#1) – see example in table B.3.2-11

The CUA responds to the 200 (OK) response with an ACK request sent to the P-CSCF#1.

**Table B.3.2-11: ACK request (CUA#1 to P-CSCF#1)**

```

ACK sip: user2_public1@home2.net;gr=urn:uuid:2ad8950e-48a5-4a74-8d99-ad76cc7fc74
;comp=sigcomp SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Max-Forwards: 70
P-Access-Network-Info: 3GPP-UTRAN-TDD; utran-cell-id-3gpp=234151D0FCE11
Route: <sip:pcscf1.visited1.net:7531;lr;comp=sigcomp>, <sip:scscf1.home1.net;lr>,
<sip:scscf2.home2.net;lr>, <sip:pcscf2.visited2.net;lr>
From: <sip:user1_public1@home1.net>;tag=171828
To: <sip:user2_public1@home2.net>;tag=314159
Call-ID: cb03a0s09a2sdfg1kj490333
Cseq: 127 ACK
Content-Length: 0

```

#### 12. Reserve IP-CAN bearer for media

The originating CUA reserves an IP-CAN bearer for the message session media component.

#### 13. ACK request (P-CSCF#2 to CUA#2) – see example in table B.3.2-13.

P-CSCF#2 forwards the ACK request to the terminating CUA.

**Table B.3.2-13: ACK request (P-CSCF#2 to CUA#2)**

```

ACK sip:[5555::eee:fff:aaa:bbb]:8805;comp=sigcomp SIP/2.0
Via: SIP/2.0/UDP pscf2.visited2.net:5088;comp=sigcomp;branch=z9hG4bK361k21.1, SIP/2.0/UDP
scscf2.home2.net;branch=z9hG4bK764z87.1, 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
Max-Forwards: 66
From:
To:
Call-ID:
Cseq:
Content-Length:

```

**14. Establishment of the MSRP session**

The CUAs will establish the MSRP session as described in subclause A.3 of 3GPP TS 24.247 [10].

**15. Media exchange in PS domain**

The CUAs will transfer media over the PS domain.

**16. Initiation of CS call establishment with SETUP from originating side (from CUA#1 to MSC#1)**

CUA#1 starts the CS call towards CUA#2 sending out the SETUP.

Specifically for CSI, the SETUP message includes:-

- Called Party Number = [(Numbering plan identifier = ISDN/telephony numbering plan), (type of number = international number), (Number digits = 1212552222)]
- User-User IE = [(Protocol ID =3GPP capability exchange protocol), (Radio Environment = 1), (IM Status = IM subsystem registered), (Personal ME Identifier = 0007), (UE capability version = 04)].

CUA#1 ensures that the Personal ME Identifier within the User-User IE of the SETUP correlates to the Personal ME Identifier used by CUA#1 in the preceding IM session.

**17. CALL PROCEEDING (MSC#1 to CUA#1)**

MSC#1 after call processing checks replies to CUA#1 with Call Proceeding.

MSC#1 progresses the CS Call by contacting MSC#2. Intermediate CS entities could be involved.

The User-User information provided by CUA#1 is forwarded by MSC#1 towards MSC#2.

There is no specific CSI information in CALL\_PROCEEDING.

**18. Initiation of CS terminating call with SETUP towards called party (from MSC#2 to CUA#2)**

MSC#2 starts the terminating call by initiating SETUP to CUA#2.

Specifically for CSI, the SETUP message includes :-

- Called Party Number = [(type of number = international number ), (Numbering plan identifier = ISDN/telephony numbering plan), (Number digits = 1212552222)]
- Calling Party Number = [(presentation indicator = Presentation allowed), (screening indicator = Network provided), (type of number = international number ), (Numbering plan identifier = ISDN/telephony numbering plan), (Number digits = 1212551111)]
- User-User IE = [(Protocol ID =3GPP capability exchange protocol), (Radio Environment = 1), (IM Status = IM subsystem registered), (Personal ME Identifier = 0007), (UE capability version = 04)].

The Personal ME Identifier provided by CUA#1 is used by CUA#2 to relate to the appropriate remote terminal having the parallel IM session. The UE capability version provided by CUA#1 is used by CUA#2 to check whether the terminal capabilities of CUA#1 are already known by CUA#2.

**19. CALL CONFIRM indicating terminating CS Call is being processed (from CUA#2 to MSC#2)**

CUA#2 on accepting the terminating call for further processing respond to MSC#2 with CALL\_CONFIRM. There is no specific CSI information in CALL\_CONFIRM.

## 20. Allocation of user plane resources

MSC#2 initiate provision of user plane resources. In this example call flow, this allocation takes place at this instance in time, but the network can opt to allocate resources after message 24 but certainly completes the allocation before message 25.

There are no CSI specifics.

## 21. ALERTING indicating end user is alerted, (from CUA#2 to MSC#2)

When alerting towards the end user is started (ie. end user ringing) CUA#2 indicates that the call has been delivered by sending ALERTING to MSC#2.

There are no CSI specifics in this message; in particular, User-User Information Element is not provided.

## 22. Originating NW allocates user plane resources

When MSC#2 receives ALERTING (see previous message 21), MSC#2 conveys this through Intermediate CS entities to MSC#1. Knowing that the call has delivered to far end user, MSC#1 initiates allocation of user plane resources to originating CUA#1.

MSC#1 could have opt to allocate user plane resources to CUA#1 much earlier in the messaging sequence (eg. after CALL\_PROCEEDING) or MSC#1 could allocate user plane resource much later eg. just before sending of message 26.

This example flow chooses to indicate user plane resources being allocated at this point in time.

## 23. ALERTING indicating far end alerting has started (from MSC#1 to CUA#1)

MSC#1 sends to CUA#1 ALERTING indicating that the far end user is being alerted. This indication together with available user plane resource allow for ring tone to be provided to the originating user by the network. If the network had opted not to allocate user plane resources, then this ALERTING will allow the CUA#1 to generate local indications to the user that the call has been delivered to the far end user.

## 24. CONNECT - far end user answers the call (from CUA#2 to MSC#2)

When the user answers the call, CUA#2 sends CONNECT to MSC#2.

Specifically for CSI, the CONNECT message includes:-

- User-User IE = [(Protocol ID = 3GPP capability exchange protocol), (Radio Environment = 1), (IM Status = IM subsystem registered), (Personal ME Identifier = 0EA2), (UE capability version = 0D)].

CUA#2 provides its Personal ME Identifier and UE capability version within the User-User IE. This Personal ME Identifier and UE capability version is the same as that provided by CUA#2 in 200 OK (message step 9) of the IM session.

CUA#2 sending CONNECT starts connecting to the user plane resources.

## 25. CONNECT ACKNOWLEDGE (from MSC#2 to CUA#2)

Upon getting CONNECT, MSC#2 starts through connecting the call from the far end through the intermediate CS entities to MSC#1. MSC#2 indicates this by sending CONNECT\_ACKNOWLEDGE to CUA#2 and thus allowing CUA#2 to progress to active call state.

MSC#2 conveys call acceptance to MSC#1 forwarding any User-User information to MSC#1 that is included in the CONNECT from CUA#2.

## 26. CONNECT – far end has answered the call (from MSC#1 to CUA#1)

MSC#1 indicates far end has answered the terminating call by sending CONNECT to CUA#1. MSC#1 having through connected the call on the network side now awaits CUA#1 to connect to the allocated user plane resources.

Specifically for CSI, the CONNECT message includes:

- Connected Number = [(presentation indicator = Presentation allowed), (screening indicator = Network provided), (type of number = international number), (Numbering plan identifier = ISDN/telephony numbering plan), (Number digits = 12125552222)]

- User-User IE = [(Protocol ID = 3GPP capability exchange protocol), (Radio Environment = 1), (IM Status = IM subsystem registered), (Personal ME Identifier = 0EA2), (UE capability version = 0D)].

The Personal ME Identifier provided by CUA#2 is used by CUA#1 to relate to the appropriate remote terminal having the parallel IM session. The UE capability version provided by CUA#2 is used by CUA#1 to check whether the terminal capabilities of CUA#2 are already known by CUA#1.

#### 27. CONNECT ACKNOWLEDGE (from CUA#1 to MSC#1)

CUA#1 receiving CONNECT and with available user plane through connects the call and return CONNECT\_ACKNOWLEDGE to MSC#1. The CONNECT ACKNOWLEDGE allows the network to progress to active call state. There are no CSI specifics in CONNECT ACKNOWLEDGE

#### 28. CS Call takes place

The CS call takes place.

---

## B.4 Signalling flows demonstrating CSI session setup when a CS call is already established

### B.4.1 Introduction

This subclause provides signalling flows for CSI session setup, using video session as an example of the IM session, established after any CS call has been established.

The flows show some additions related to CSI. The CUA includes the media feature tags g.3gpp.cs-voice and g.3gpp.cs-video in the Accept-Contact header and in the Contact header. The User Agent and Server Header carries the personal ME identifier.

### B.4.2 Establishing a CSI session when CS call has been setup

Figure B.4.2-1 shows the establishment of video session between two users. A CS call is established before the video session is established.

It is assumed that both the originating CUA and terminating CUA are using an IP-CAN with a separate bearer for SIP signalling which means that each CUA needs to provide resource reservation before starting the video session.

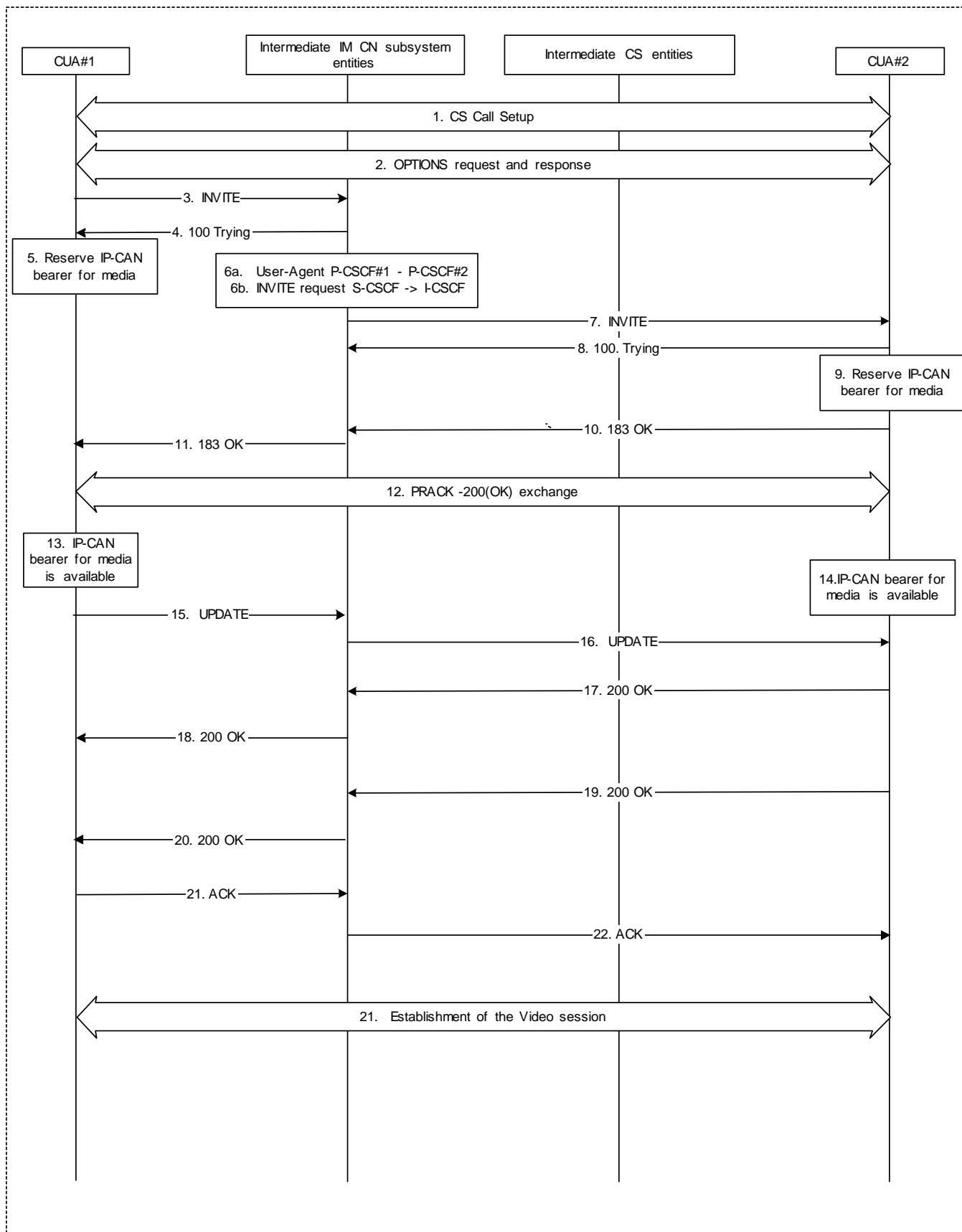


Figure B.4.2-1: Establishment of a video session after a CS call is established



The details of the signalling flows are as follows:

1. CS call

A CS call setup is performed according to subclause B.5.

2. UE capabilities exchange

Both UEs have performed a capability exchange. In the here shown example both UEs transfer their UE capability version and personal ME identifier as a part of the capability exchange. An example for capability exchange without the usage of UE capability version and personal ME identifier can be found in subclause B.6.

3. INVITE request (CUA#1 to P-CSCF#1) - see example in table B.4.2-2

CUA#1 wants to initiate a session with the terminating CUA. In this example, the CUA#1 creates a video session as the IM session. CUA#1 includes the "precondition" and "100rel" options tag in the supported header.

CUA#1 indicates, using the SDP Capability Negotiation mechanism, that it supports and is willing to use AVPF transport for the video stream and the audio stream.

CUA#1 indicates that it supports precondition and it indicates that it supports reliable provisional responses. However, it does not use the "Require" header for these capabilities.

CUA#1 does not have available the resources that are necessary to transport the media.

For this example it is assumed, that signalling encryption was negotiated between CUA#1 and P-CSCF in the security mode set-up procedure during the last successful authentication. This option will only be shown in this example.

CUA#1 declares its capabilities for CS-voice and CS-video, and requests to reach a UE with CS-voice or CS-video capabilities by including these media feature tags in the Accept-contact header and includes its personal ME identifier and UE capability version in User-Agent header.

**Table B.4.2-2: INVITE request (CUA#1 to P-CSCF#1)**

```

INVITE tel:+12125552222 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: <tel:+12125551111>
P-Access-Network-Info: 3GPP-UTRAN-TDD; utran-cell-id-3gpp=234151D0FCE11
Privacy: none
From: <sip:user1_public1@home1.net>; tag=171828
To: <tel:+12125552222>
Call-ID: cb03a0s09a2sdfglkj490333
Cseq: 127 INVITE
Require: sec-agree
Supported: precondition, 100rel, gruu
Proxy-Require: sec-agree
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>;+g.3gpp.cs-voice;+g.3gpp.cs-video
Accept-Contact: *,+g.3gpp.cs-voice, +g.3gpp.cs-video;explicit
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE, OPTIONS
Accept:application/sdp, application/3gpp-ims+xml
User-Agent: PMI-0007, UCV-04
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
m=video 49232 RTP/AVP 107
a=tcap:1 RTP/AVPF
a=pcfg:1 t=1

```

```

b=AS:64.0
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=inactive
a=rtpmap:107 H263/90000
a=fmtp 107 profile=0 level=45

```

**SDP** The SDP contains the video codec supported by CUA#1 and desired by the user at CUA#1 for this session. The precondition attributes are also included.

#### 4. 100 (Trying) response (P-CSCF#1 to CUA#1)

The P-CSCF responds to the INVITE request with a 100 (Trying) provisional response as described in subclause A.3 of 3GPP TS 24.247 [10].

#### 5. Reserve IP CAN bearer for media

CUA#1 starts resource reservation based on the SDP.

#### 6a. User-Agent (P-CSCF#1 to P-CSCF#2)

The User-Agent header field is transparently passed from P-CSCF#1 to P-CSCF#2.

#### 6b. INVITE request (S-CSCF#1 to I-CSCF#2) - see example in table B.4.2-3

S-CSCF#1 forwards the INVITE request to the I-CSCF#2, after replacing the tel URI with a SIP-URI in the Request URI.

**Table B.4.2-3: INVITE request (S-CSCF#1 to I-CSCF#2)**

```

INVITE sip:user2_public1@home2.net SIP/2.0
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
Max-Forwards: 68
Route: <sip:pcscf1.visited1.net:7531;lr;comp=sigcomp>, <sip:orig@scscf1.home1.net;lr>
Record-Route: <sip:scscf1.home1.net;lr>, <sip:pcscf1.visited1.net;lr>
P-Asserted-Identity: <tel:+12125551111>
Privacy:
From:
To:
Call-ID:
Cseq:
Supported:
Contact:
Accept-Contact:
Allow:
Accept:
User-Agent:
Content-Type:
Content-Length:

```

```

v=
o=
s=
c=
t=
m=
a=
a=
b=
a=
a=
a=
a=
a=
a=
a=
a=

```

#### 7. INVITE request (P-CSCF#2 to CUA#2) – see example in table B.4.2-4

P-CSCF#2 forwards the INVITE request to the terminating CUA.

**Table B.4.2-4: INVITE request (P-CSCF#2 to CUA#2)**

```

INVITE sip:[5555::eee:fff:aaa:bbb]:8805;comp=sigcomp>;+g.3gpp.cs-voice
Via: SIP/2.0/UDP pcscf2.visited2.net:5088;comp=sigcomp;branch=z9hG4bK361k21.1, SIP/2.0/UDP
scscf2.home2.net;branch=z9hG4bK764z87.1, SIP/2.0/UDP
icscf2_s.home2.net;branch=z9hG4bK871y12.1, 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
Max-Forwards: 65
Record-Route: <sip:pcscf2.visited2.net:5088;lr;comp=sigcomp>, <sip:scscf2.home2.net;lr>,
<sip:scscf1.home1.net;lr>, <sip:pcscf1.visited1.net;lr>
P-Asserted-Identity:
Privacy:
From:
To:
Call-ID:
Cseq:
Supported:
Contact:
Accept-Contact:
Allow:
Accept:
User-Agent:
P-Called-Party-ID:
Content-Type:
Content-Length:(...)

v=
o=
s=
c=
t=
m=
a=
a=
b=
a=
a=
a=
a=
a=
a=
a=
a=
a=

```

**8. 100 (Trying) response (CUA#2 to P-CSCF#2)**

The terminating CUA sends a 100 (Trying) provisional response to P-CSCF#2 as described in subclause A.3 of 3GPP TS 24.247 [10].

**9. Reserve IP-CAN bearer for media**

The terminating CUA sets up the bearer in accordance with the received SDP..

**10. 183 (session progress) response (CUA#2 to P-CSCF#2) – see example in table B.4.2-5**

CUA#2 declares support for "CS-voice" and "CS-video" in its Contact header. The CUA includes the personal ME identifier and the UE capability version in the Server header. CUA#2 requires resource reservation and supports the precondition mechanism.

**Table B.4.2-5: 183 (session progress) response (CUA#2 to P-CSCF#2)**

```

SIP/2.0 183 Session progress
Via: SIP/2.0/UDP pcscf2.visited2.net:5088;comp=sigcomp;branch=z9hG4bK361k21.1, SIP/2.0/UDP
    scscf2.home2.net;branch=z9hG4bK764z87.1, SIP/2.0/UDP
    icscf2_s.home2.net;branch=z9hG4bK871y12.1, 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:pcscf2.visited2.net:5088;lr;comp=sigcomp>>, <sip:scscf2.home2.net;lr>,
    <sip:scscf1.home1.net;lr>, <sip:pcscf1.visited1.net;lr>
Privacy: none
From: <sip:user1_public1@home1.net>; tag=171828
To: <tel:+12125552222>;tag=314159
Call-ID: cb03a0s09a2sdfgklkj490333
Cseq: 127 INVITE
Contact: <sip: user2_public1@home2.net;gr=urn:uuid:2ad8950e-48a5-4a74-8d99-
    ad76cc7fc74;comp=sigcomp>;+g.3gpp.cs-voice,+g.3gpp.cs-video
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE, OPTIONS
Require: 100rel, precondition
Supported: gruu
Server: PMI-0EA2, UCV-0D
Content-Type: application/sdp
Content-Length: (...)

v=0
o=- 2987933615 2987933617 IN IP6 5555:: eee:fff:aaa:bbb
s=-
c=IN IP6 5555::eee:fff:aaa:bbb
t=0 0
m=video 49234 RTP/AVPF 107
a=acfg:1 t=1
b=AS:64.0
a=curr:qos remote none
a=curr:qos local none
a=des:qos mandatory local sendrecv
a=des:qos mandatory remote sendrecv
a=conf:qos remote sendrecv
a=inactive
a=rtpmap:107 H263/90000
a=fmtp 107 profile=0 level=45

```

11. 183 (session progress) response (P-CSCF#1 to CUA#1) – see example in table B.4.2-6

**Table B.4.2-6: 183 (session progress) response (P-CSCF#1 to CUA#1 )**

```
SIP/2.0 183 Session progress
Via: [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Record-Route: <sip:pcscf2.visited2.net:5088;lr;comp=sigcomp>, <sip:scscf2.home2.net;lr>,
<sip:scscf1.home1.net;lr>, <sip:pcscf1.visited1.net;lr>
Privacy:
From:
To:
Call-ID:
Cseq:
RSeq:
P-Asserted-Identity:< user2_public1@home2.net>, < tel:+12125552222>
Contact:
Allow:
Require:
Supported:
Server:
Content-Type:
Content-Length:

v=0
o=-
s=-
c=
t=
m=
a=
b=
a=
a=
a=
a=
a=
a=
a=
a=
a=
a=
```

**12. PRACK-200 Exchange****13. IP-CAN bearer available for media**

The IP-CAN bearer at CUA#1 is established.

**14. IP-CAN bearer available for media.**

The IP-CAN bearer at CUA#2 is established.

**15. UPDATE request (CUA#1 to P-CSCF#1 ) – see example in table B.4.2-7**

CUA#1 indicates that it can send and receive media.

**Table B.4.2-7: UPDATE request (CUA#1 to P-CSCF#1 )**

```

UPDATE <sip: user2_public1@home2.net;gr=urn:uuid:2ad8950e-48a5-4a74-8d99-ad76cc7fc74
;comp=sigcomp>
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>
From: <sip:user1_public1@home1.net>; tag=171828
To: <tel:+12125552222> tag=314159
Call-ID: cb03a0s09a2sdfg1kj490333
Cseq: 129 UPDATE
Require: sec-agree
Proxy-Require: sec-agree
Security-Verify: ipsec-3gpp; q=0.1; alg=hmac-sha-1-96; spi=87654321; port1=7531
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
m=video 49232 RTP/AVPF 107
b=AS:64.0
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos mandatory remote sendrecv
a=sendrecv
a=rtpmap:107 H263/90000
a=fmtp 107 profile=0 level=45

```

**16. UPDATE request (P-CSCF#2 to CUA#2 ) – see example in table B.4.2-8**

The CUA#2 can start alerting.

**Table B.4.2-8: UPDATE request (P-CSCF#2 to CUA#2)**

```

UPDATE <sip: user2_public1@home2.net;gr=urn:uuid:2ad8950e-48a5-4a74-8d99-
ad76cc7fc74;comp=sigcomp>
Via: : SIP/2.0/UDP pcscf2.visited2.net:5088;comp=sigcomp;branch=z9hG4bK361k21.1,
SIP/2.0/UDP scscf2.home2.net;branch=z9hG4bK764z87.1, SIP/2.0/UDP
icscf2_s.home2.net;branch=z9hG4bK871y12.1, 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
Max-Forwards: 68
From:
To:
Call-ID:
Cseq:
Content-Type:
Content-Length:

v=
o=
s=
c=
t=
m=
b=
a=
a=
a=
a=
a=
a=
a=
a=

```

**17. 200 (OK) response (CUA#2 to P-CSCF#2) – see example in table B.4.2-9**

CUA 2 indicates that media can be received and sent.

**Table B.4.2-9: 200(OK) response (CUA#2 to P-CSCF#2)**

```

SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf2.visited2.net:5088;comp=sigcomp;branch=z9hG4bK361k21.1, SIP/2.0/UDP
scscf2.home2.net;branch=z9hG4bK764z87.1, SIP/2.0/UDP
icscf2_s.home2.net;branch=z9hG4bK871y12.1, 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
From: <sip:user1_public1@home1.net>; tag=171828
To: <tel:+12125552222>;tag=314159
Call-ID: cb03a0s09a2sdfg1kj490333
Cseq: 129 UPDATE
Content-Type: application/sdp
Content-Length: (...)

v=0
o=- 2987933615 2987933617 IN IP6 5555:: eee:fff:aaa:bbb
s=-
c=IN IP6 5555::eee:fff:aaa:bbb
t=0 0
m=video 49234 RTP/AVPF 107
b=AS:64.0
a=curr:qos local sendrecv
a=curr:qos remote sendrecv
a=des:qos mandatory local sendrecv
a=des:qos mandatory remote sendrecv
a=sendrecv
a=rtptime:107 H263/90000
a=fmtp 107 profile=0 level=45

```

**18. 200 (OK) response (P-CSCF#1 to CUA#1) – see example in table B.4.2-10****Table B.4.2-10: 200(OK) response (P-CSCF#1 to CUA#1)**

```

SIP/2.0 200 OK
Via: [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
From:
To:
Call-ID:
Cseq:
Content-Type:
Content-Length:

v=0
o=-
s=-
c=
t=
m=
b=
a=
a=
a=
a=
a=
a=
a=
a=
a=
a=

```

**19. 200 (OK) response (CUA#2 to P-CSCF#2) – see example in table B.4.2-11**

CUA#2 accepts the session.

**Table B.4.2-11: 200 (OK) response (CUA#2 to P-CSCF#2)**

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf2.visited2.net:5088;comp=sigcomp;branch=z9hG4bK361k21.1, SIP/2.0/UDP
scscf2.home2.net;branch=z9hG4bK764z87.1, SIP/2.0/UDP
icscf2_s.home2.net;branch=z9hG4bK871y12.1, 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
Privacy: none
From: <sip:user1_public1@home1.net>; tag=171828
To: <tel:+12125552222>;tag=314159;tag=314159
Call-ID: cb03a0s09a2sdfgkjkj490333
Cseq: 127 INVITE
Contact: <sip:user2_public1@home2.net;gr=urn:uuid:2ad8950e-48a5-4a74-8d99-
ad76cc7fc74;comp=sigcomp>;+g.3gpp.cs-voice, +g.3gpp.cs-video
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE, OPTIONS
Server: PMI-0EA2, UCV-0D
Content-Length: 0
```

**20. 200 (OK) response (P-CSCF#1 to CUA#1) – see example in table B.4.2-12**

P-CSCF#1 forwards the 200 (OK) response to the originating CUA.

**Table B.4.2-12: 200 (OK) response (P-CSCF#1 to CUA#1)**

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
P-Asserted-Identity: < user2_public1@home2.net>, < tel:+12125552222>
Privacy:
From:
To:
Call-ID:
CSeq:
Require:
Contact:
Allow:
Server:
Content-Length:
```

**21. ACK request (CUA#1 to P-CSCF#1) – see example in table B.4.2-13**

The CUA responds to the 200 (OK) response with an ACK request sent to the P-CSCF#1.

**Table B.4.2-13: ACK request (CUA#1 to P-CSCF#1)**

```
ACK sip: <sip:user2_public1@home2.net;gr=urn:uuid:2ad8950e-48a5-4a74-8d99-
ad76cc7fc74;comp=sigcomp>;
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Max-Forwards: 70
P-Access-Network-Info: 3GPP-UTRAN-TDD; utran-cell-id-3gpp=234151D0FCE11
Route: <sip:pcscf1.visited1.net:7531;lr;comp=sigcomp>, <sip:scscf1.home1.net;lr>,
<sip:scscf2.home2.net;lr>, <sip:pcscf2.visited2.net;lr>
From: <sip:user1_public1@home1.net>;tag=171828
To: <sip:user2_public1@home2.net>;tag=314159
Call-ID: cb03a0s09a2sdfgkjkj490333
Cseq: 127 ACK
Content-Length: 0
```

**22. ACK request (P-CSCF#2 to CUA#2) – see example in table B.3.2-14.**

P-CSCF#2 forwards the ACK request to the terminating CUA.



**Table B.3.2-14: ACK request (P-CSCF#2 to CUA#2)**

```
ACK sip: <sip: user2_public1@home2.net;gr=urn:uuid:2ad8950e-48a5-4a74-8d99-  
ad76cc7fc74;comp=sigcomp>  
Via: SIP/2.0/UDP pcscf2.visited2.net:5088;comp=sigcomp;branch=z9hG4bK361k21.1, SIP/2.0/UDP  
scscf2.home2.net;branch=z9hG4bK764z87.1, 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  
Max-Forwards: 66  
From:  
To:  
Call-ID:  
Cseq:  
Content-Length:
```

---

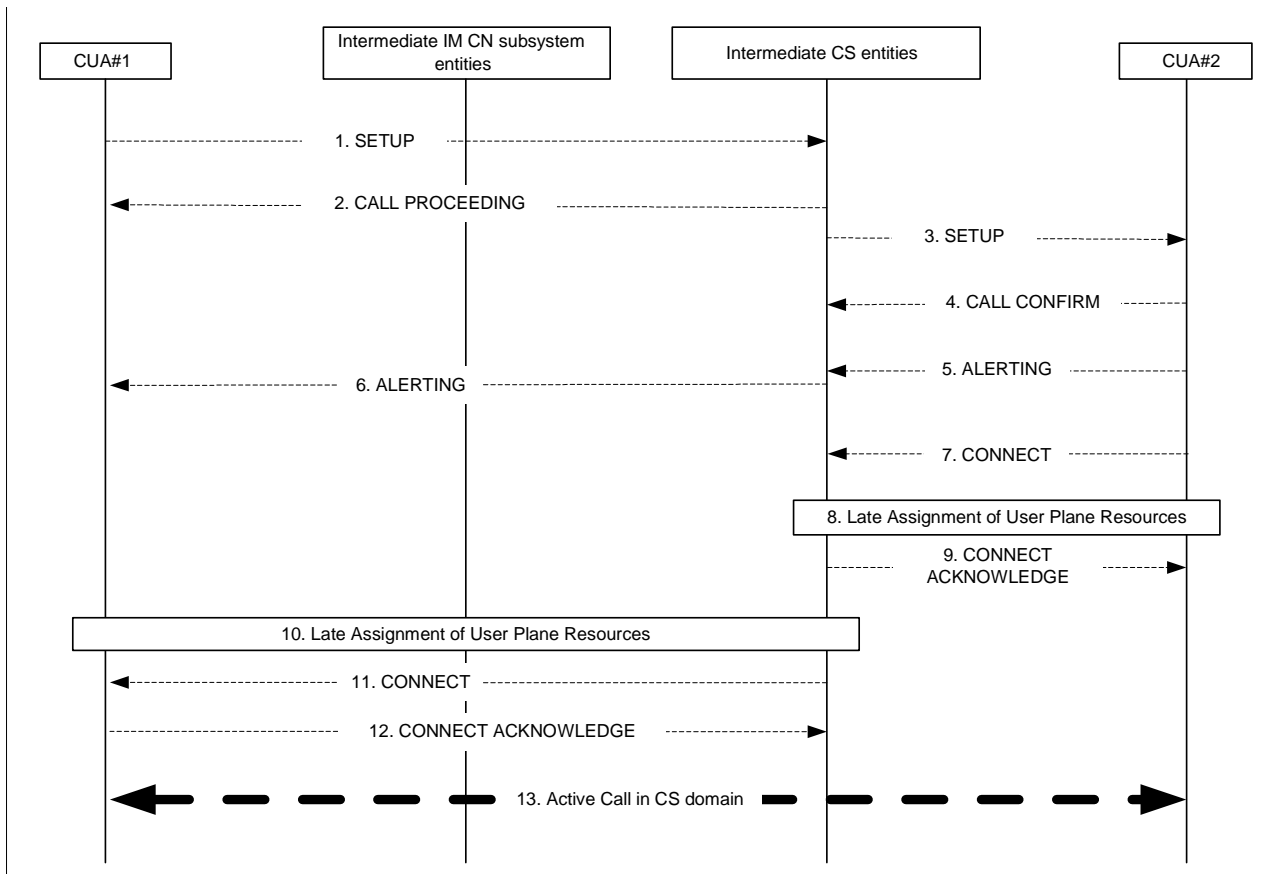
## B.5 Signalling flows demonstrating capability information exchange in a CS call (only)

### B.5.1 Introduction

This subclause provides signalling flows for exchange of information relevant for CSI in a CS call only.

### B.5.2 Establishing a CS call from a CSI capable UE

Figure B.5.2-1 shows the exchange of Radio Environment information, the IM Status, UE capability version and Personal ME Identifier by the CUAs of a CSI capable UE during a normal CS call establishment.



**Figure B.5.2-1: CS call establishment (example of late assignment of user plane resources)**

The details of the signalling flows are as follows:-

**1. Initiation of CS call establishment with SETUP from originating side (from CUA#1 to MSC#1)**

CUA#1 starts the CS call towards CUA#2 sending out the SETUP.

Specifically for CSI, the SETUP message includes:-

- Called Party Number = [(Numbering plan identifier = ISDN/telephony numbering plan), (type of number = international number ), (Number digits = 12125552222)]
- User-User IE = [(Protocol ID =3GPP capability exchange protocol), (Radio Environment = 1), (IM Status = IM subsystem capable and willing to register to IM subsystem), (Personal ME Identifier = 0007), (UE capability version = 04)].

**2. CALL PROCEEDING (MSC#1 to CUA#1)**

MSC#1 after call processing checks replies to CUA#1 with CALL\_PROCEEDING.

MSC#1 progress with the CS Call by contacting MSC#2. Intermediate CS entities could be involved.

The User-User information provided by CUA#1 is forwarded by MSC#1 towards MSC#2.

There is no specific CSI information in CALL\_PROCEEDING.

### 3. Initiation of CS terminating call with SETUP towards called party (from MSC#2 to CUA#2)

MSC#2 starts the terminating call by initiating SETUP to CUA#2.

Specifically for CSI, the SETUP message includes:-

- Called Party Number = [(type of number = international number ),  
(Numbering plan identifier = ISDN/telephony numbering plan), (Number digits = 12125552222)]
- Calling Party Number = [(presentation indicator = Presentation allowed),  
(screening indicator = Network provided), (type of number = international number ),  
(Numbering plan identifier = ISDN/telephony numbering plan), (Number digits = 12125551111)]
- User-User IE = [(Protocol ID =3GPP capability exchange protocol), (Radio Environment = 1),  
(IM Status = IM subsystem capable and willing to register to IM subsystem),  
(Personal ME Identifier = 0007), (UE capability version = 04)].

The Personal ME Identifier provided by CUA#1 is later used by CUA#2 to relate to a particular terminal of CUA#1 that initiated this CS call is later progressed to a CSI call.

### 4. CALL CONFIRM indicating terminating CS Call is being processed (from CUA#2 to MSC#2)

CUA#2 on accepting the terminating call for further processing respond to MSC#2 with CALL\_CONFIRM. There is no specific CSI information in CALL\_CONFIRM.

### 5. ALERTING indicating end user is alerted, (from CUA#2 to MSC#2)

CUA#2 indicates locally to the user that a call has arrived by starting local alerting. With the call delivered, CUA#2 sends ALERTING to MSC#2.

There are no CSI specifics in this message; in particular, there is no inclusion of User-User Information Element.

NOTE: This example flow has taken to illustrate local alerting by CUA#2. This is the case if in the SETUP MSC#2 allows this to be done, as MSC intends to do a late assignment of user plane resources.

### 6. ALERTING indicating far end alerting has started (from MSC#1 to CUA#1)

Through Intermediate CS entities, MSC#2 informs MSC#1 that the call has been delivered.

MSC#1 sends to CUA#1 ALERTING indicating that call has been delivered and far end user is being alerted. Only local indication of call delivery can be provided by CUA#1 to the user (ie. ringing tone) as user plane resources have yet to be assigned by MSC#1.

There are no CSI specifics in this message.

### 7. CONNECT - far end user answers the call (from CUA#2 to MSC#2)

When the user answers the call, CUA#2 sends CONNECT to MSC#2.

Specifically for CSI, the CONNECT message includes:-

- User-User IE = [(Protocol ID =3GPP capability exchange protocol), (Radio Environment = 1),  
(IM Status = IM subsystem capable and willing to register to IM subsystem),  
(Personal ME Identifier = 0EA2), (UE capability version = 0D)].

After sending the CONNECT, CUA#2 is ready to connect to user plane resources and does the instance user plane resources is allocated.

MSC#2 conveys call answer to MSC#1 and forwards any User-User information to MSC#1 that is included in the CONNECT from CUA#2.

The provision of the Personal ME Identifier is to allow CUA#2 and CUA#1 to bind this CS Call answered by a particular terminal of CUA#2 to any eventual associated IM session to progress this CS call to a CSI call.

### 8. Allocation of user plane resources

MSC#2 has to allocate user plane resources to CUA#2 to allow speech to take place. In this example, this is the very latest point in time that MSC#2 can allocate such resources. This is referred to as Late Assignment. With the completion of the user plane resources, CUA#2 connects up and speech begins.

9. **CONNECT ACKNOWLEDGE (from MSC#2 to CUA#2)**

Upon completing Late Assignment, MSC#2 starts through connecting the call all the way from CUA#2 to the far end through the Intermediate CS Entities to MSC#1.

MSC#2 indicates this by sending CONNECT\_ACKNOWLEDGE to CUA#2 and thus allowing CUA#2 to progress to active call state.

10. **Originating NW allocates user plane resources**

Knowing that the call has been answered, MSC#1 initiates allocation of user plane resources to originating CUA#1.

In this example, this is the latest point in time that MSC#1 can assign the user plane resources for speech to take place.

11. **CONNECT (from MSC#1 to CUA#1)**

MSC#1 indicates far end has answered the terminating call by sending CONNECT to CUA#1. MSC#1 through connects the call on the network side and awaits CUA#1 to connect to the allocated user plane resources.

Specifically for CSI, the CONNECT message includes:

- Connected Number = [(presentation indicator = Presentation allowed),  
(screening indicator = Network provided), (type of number = international number),  
(Numbering plan identifier = ISDN/telephony numbering plan), (Number digits = 12125552222)]
- User-User IE = [(Protocol ID = 3GPP capability exchange protocol), (Radio Environment = 1),  
(IM Status = IM subsystem capable and willing to register to IM subsystem),  
(Personal ME Identifier = 0EA2), (UE capability version = 0D)].

12. **CONNECT ACKNOWLEDGE (from CUA#1 to MSC#1)**

CUA#1 receiving CONNECT and with available user plane through connects the call and return CONNECT\_ACKNOWLEDGE to MSC#1. The CONNECT ACKNOWLEDGE is to allow the network to progress to active call state. The CONNECT ACKNOWLEDGE does not have any CSI specifics.

13. **CS Call takes place**

The CS call takes place.

---

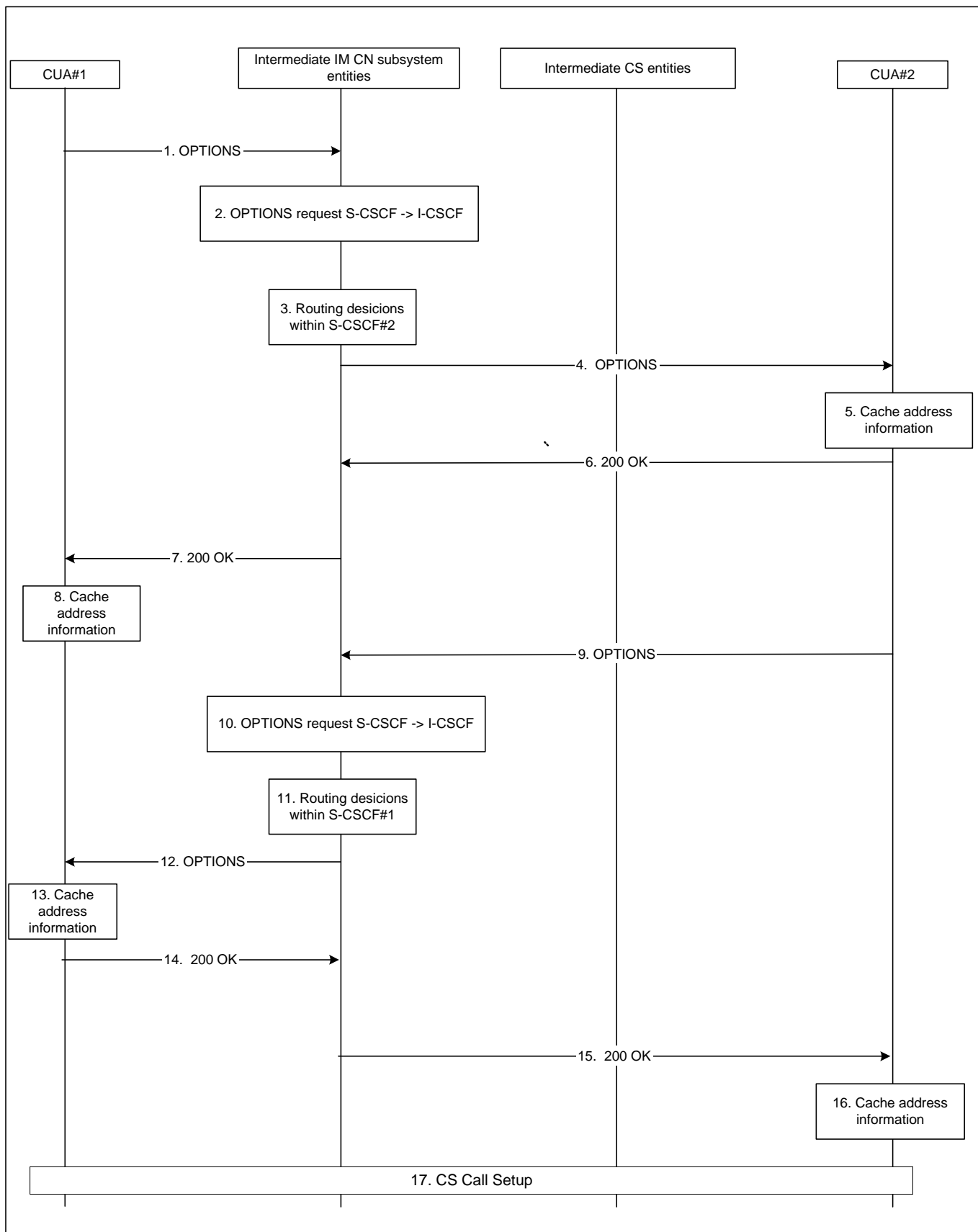
## B.6 Signalling flows demonstrating UE capability exchange with no UE capability version

### B.6.1 Introduction

This subclause provides signalling flows for UE capability exchange with no UE capability version and no personal ME identifier outside a CS call or when a CS call is already in progress.

### B.6.2 UE capability exchange outside a CS call

Figure B.6.2-1 shows the UE capability exchange using the OPTIONS method.



**Figure B.6.2-1: UE capability exchange before any CS call is established**

The details of the signalling flows are as follows:

### 1. OPTIONS request (CUA#1 to P-CSCF#1) - see example in table B.6.2-1

The originating CUA wants to know the capabilities of the terminating UE, preferably a UE that supports CS-voice and CS-video, or one or the other.

**Table B.6.2-1: OPTIONS request (CUA#1 to P-CSCF#1)**

```

OPTIONS tel:+12125552222 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: <tel:+1-212-555-1111>
P-Access-Network-Info: 3GPP-UTRAN-TDD; utran-cell-id-3gpp=234151D0FCE11
Privacy: none
From: <sip:user1_public1@home1.net>; tag=171828
To: <tel:+12125552222>
Call-ID: cb03a0s09a2sdfglkj490333
Cseq: 127 OPTIONS
Require: sec-agree
Proxy-Require: sec-agree
Supported: gruu
Security-Verify: ipsec-3gpp; q=0.1; alg= hmac-sha-1-96; spi-c=98765432; spi-s=87654321;
port-c=8642; port-s=7531
Accept-Contact: *,+g.3gpp.cs-voice,+g.3gpp.cs-video;explicit
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE, OPTIONS
Accept: application/sdp
Content-Length: 0

```

### 2. OPTIONS request (S-CSCF#1 to I-CSCF#2) - see example in table B.6.2-2

S-CSCF#1 forwards the OPTIONS request to the I-CSCF#2, after mapping the tel URI to a SIP-URI.

**Table B.6.2-2: OPTIONS request (S-CSCF#1 to I-CSCF#2)**

```

OPTIONS sip:user2_public1@home2.net SIP/2.0
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
Max-Forwards: 68
Record-Route: <sip:scscf1.home1.net;lr>, <sip:pcscf1.visited1.net;lr>
P-Asserted-Identity:
P-Charging-Vector: icid-value="AyretyU0dm+602Irt5tAFrbHLso=023551024"; orig-icid=home1.net
Privacy:
From:
To:
Call-ID:
Cseq:
Supported:
Accept-Contact:
Allow:
Accept:
Content-Length: 0

```

### 3. Routing decision based on caller preferences and callee capabilities

The S-CSCF for CUA#2 uses caller preferences (information from the Accept-Contact header) and callee capabilities (information stored in the Registrar from the REGISTER Contact header) information to decide how to route the OPTIONS request.

### 4. OPTIONS request (P-CSCF#2 to CUA#2) – see example in table B.6.2-4

P-CSCF#2 forwards the OPTIONS request to the terminating CUA.

**Table B.6.2-4: OPTIONS request (P-CSCF#2 to CUA#2)**

```

OPTIONS sip:[5555::eee:fff:aaa:bbb]:8805;comp=sigcomp SIP/2.0
Via: SIP/2.0/UDP pcscf2.visited2.net:5088;comp=sigcomp;branch=z9hG4bK361k21.1, SIP/2.0/UDP
scscf2.home2.net;branch=z9hG4bK764z87.1, SIP/2.0/UDP
icscf2_s.home2.net;branch=z9hG4bK871y12.1, 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
Max-Forwards: 65
Record-Route: <sip:pcscf2.visited2.net:5088;lr;comp=sigcomp>, <sip:scscf2.home2.net;lr>,
<sip:scscf1.home1.net;lr>, <sip:pcscf1.visited1.net;lr>
P-Asserted-Identity:
Privacy:
From:
To:
Call-ID:
Cseq:
Supported:
Accept-Contact:
Allow:
Accept:
Content-Length: 0
P-Called-Party-ID:

```

#### 5. Save CUA#1's address, prepare 200 OK response

The terminating CUA caches any address related information it learns about CUA#1 from the OPTIONS request, such as its SIP URI. The terminating CUA adds feature parameters to the Contact header field in the OPTIONS response indicating its capabilities. The feature parameters that were included in the registration generated by the terminating CUA are used in the OPTIONS response.

#### 6. 200 (OK) response (CUA#2 to P-CSCF#2) – see example in table B.6.2-6

The terminating CUA sends a 200 (OK) response for the OPTIONS request containing the terminating CUA's capabilities. This information is declared both in the media feature tags listed in the Contact header and in the SDP. In this example, CUA#2 declares capability for cs-voice, but not cs-video, and it declares support for MSRP, RTP video (H.263) and RTP audio (AMR).

**Table B.6.2-6: 200 (OK) response (CUA#2 to P-CSCF#2)**

```

SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf2.visited2.net:5088;comp=sigcomp;branch=z9hG4bK361k21.1, SIP/2.0/UDP
scscf2.home2.net;branch=z9hG4bK764z87.1, SIP/2.0/UDP
icscf2_s.home2.net;branch=z9hG4bK871y12.1, 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:pcscf2.visited2.net:5088;lr;comp=sigcomp>>, <sip:scscf2.home2.net;lr>,
<sip:scscf1.home1.net;lr>, <sip:pcscf1.visited1.net;lr>
Privacy: none
P-Access-Network-Info: 3GPP-UTRAN-TDD; utran-cell-id-3gpp=123451D0FCE11
From: <sip:user1_public1@home1.net>; tag=171828
To: <sip:user2_public1@home2.net>;tag=314159
Call-ID: cb03a0s09a2sdfgkjkj490333
Cseq: 127 OPTIONS
Supported: gruu, preconditions, 100rel
Contact: <sip:user2_public1@home2.net;gr=urn:uuid:2ad8950e-48a5-4a74-8d99-
ad76cc7fc74;comp=sigcomp>;+g.3gpp.cs-voice, <tel:+12125552222>
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE, OPTIONS
Content-Type: application/sdp
Content-Length: (...)

v=0
o=- 2987933615 2987933617 IN IP6 5555:: eee:fff:aaa:bbb
s=-
c=IN IP6 5555::eee:fff:aaa:bbb
t=0 0
m=message 0 TCP/MSRP *
a=accept-types:text/plain text/html message/cpim image/jpeg image/gif video/3gpp
a=max-size:65536
m=video 0 RTP/AVP 96
a=rtpmap:96 H263-2000/90000
m=audio 0 RTP/AVP 97
a=rtpmap:97 AMR/8000
m=video 0 RTP/AVPF 96
a=rtpmap:96 H263-2000/90000
m=audio 0 RTP/AVPF 97
a=rtpmap:97 AMR/8000

```

**SDP** The SDP contains:

- The set of offered content types supported by UE#2 and desired by the user at UE#2 for an MSRP session in the accept-types attribute and indicates the maximum size message that can be received by UE#2 in the max-size attribute (no dynamically allocated attributes are included, for example, the a=path line is not there);
- The supported video codec and relevant parameters (but no dynamically allocated parameters);
- The supported audio codec and relevant parameters (but no dynamically allocated parameters).

No ports are declared since UE capabilities are just being listed.

#### 7. 200 (OK) response (P-CSCF#1 to CUA#1) – see example in table B.6.2-7

P-CSCF#1 forwards the 200 (OK) response to the originating CUA.



**Table B.6.2-7: 200 (OK) response (P-CSCF#1 to CUA#1)**

```

SIP/2.0 200 OK
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Record-Route: <sip:pcscf2.visited2.net;lr>, <sip:scscf2.home2.net;lr>,
  <sip:scscf1.home1.net;lr>, <sip:pcscf1.visited1.net:7531;lr;comp=sigcomp>
P-Asserted-Identity:
Privacy:
From:
To:
Call-ID:
CSeq:
Supported:
Contact:
Allow:
Content-Type:
Content-Length:

v=
o=
s=
c=
t=
m=
a=
a=
m=
a=
m=
a=
m=
a=
m=
a=

```

**8. Cache information on CUA#2's capabilities**

Since no CS call is ongoing, cache the information obtained on CUA#2's capabilities for later use within a CS call.

**9. OPTIONS request (CUA#2 to P-CSCF#2) - see example in table B.6.2-9**

The originating CUA wants to know the capabilities of the terminating UE, preferably a UE that supports cs-voice and cs-video, or one or the other.

**Table B.6.2-9: OPTIONS request (CUA#2 to P-CSCF#2)**

```

OPTIONS tel:+12125551111 SIP/2.0
Via: SIP/2.0/UDP [5555::eee:fff:aaa:bbb]:8805;comp=sigcomp;branch=af45k329jstd43
Max-Forwards: 70
Route: <sip:pcscf2.visited2.net:7743;lr;comp=sigcomp>, <sip:orig@scscf2.home2.net;lr>
P-Preferred-Identity: <tel:+1-212-555-2222>
P-Access-Network-Info: 3GPP-UTRAN-TDD; utran-cell-id-3gpp=123451D0FCE11
Privacy: none
From: <sip:user2_public2@home2.net>; tag=1912031
To: <tel:+12125552222>
Call-ID: ast324c2ho9512go9238ks4b
Cseq: 127 OPTIONS
Require: sec-agree
Proxy-Require: sec-agree
Supported: gruu
Security-Verify: ipsec-3gpp; q=0.1; alg=hmac-sha-1-96; spi-c=23456789; spi-s=12345678;
  port-c=8318; port-s=7743
Accept-Contact: *,+g.3gpp.cs-voice,+g.3gpp.cs-video;explicit
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE, OPTIONS
Accept: application/sdp
Content-Length: 0

```

**10. OPTIONS request (S-CSCF#2 to I-CSCF#1) - see example in table B.6.2-10**

S-CSCF#2 forwards the OPTIONS request to the I-CSCF#1, after mapping the tel URI to a SIP-URI.

**Table B.6.2-10: OPTIONS request (S-CSCF#2 to I-CSCF#1)**

```

OPTIONS sip:user1_public1@home1.net SIP/2.0
Via: SIP/2.0/UDP scscf2.home2.net;branch= af45K329jsg213, SIP/2.0/UDP
    pcscf2.visited2.net;branch= af45K329jsbbb2, SIP/2.0/UDP
    [5555::eee:fff:aaa:bbb]:8805;comp=sigcomp;branch= af45K329jstd43
Max-Forwards: 68
Record-Route: <sip:scscf2.home2.net;lr>, <sip:pcscf2.visited2.net;lr>
P-Asserted-Identity:
P-Charging-Vector: icid-value="H7a2rvdguTd6eYt7br=ac?+g5468HdRdRf54b"; orig-loi=home2.net
Privacy:
From:
To:
Call-ID:
Cseq:
Supported:
Accept-Contact:
Allow:
Accept:
Content-Length: 0

```

**11. Routing decision based on caller preferences and callee capabilities**

The S-CSCF for CUA#1 uses caller preferences (information from the Accept-Contact header) and callee capabilities (information stored in the Registrar from the REGISTER Contact header) information to decide how to route the OPTIONS request.

**12. OPTIONS request (P-CSCF#1 to CUA#1) – see example in table B.6.2-12**

P-CSCF#1 forwards the OPTIONS request to the terminating CUA.

**Table B.6.2-12: OPTIONS request (P-CSCF#1 to CUA#1)**

```

OPTIONS sip:[5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp SIP/2.0
Via: SIP/2.0/UDP pcscf1.visited1.net:6809;comp=sigcomp;branch= af45K329jsgfff, SIP/2.0/UDP
    scscf1.home1.net;branch= af45K329js1234, SIP/2.0/UDP icscf1_s.home1.net;branch=
    af45K329jsskit, SIP/2.0/UDP scscf2.home2.net;branch= af45K329jsg213, SIP/2.0/UDP
    pcscf2.visited2.net;branch= af45K329jsbbb2, SIP/2.0/UDP
    [5555::eee:fff:aaa:bbb]:8805;comp=sigcomp;branch= af45K329jstd43
Max-Forwards: 65
Record-Route: <sip:pcscf1.visited1.net:6809;lr;comp=sigcomp>, <sip:scscf1.home1.net;lr>,
    <sip:scscf2.home2.net;lr>, <sip:pcscf2.visited2.net;lr>
P-Asserted-Identity:
Privacy:
From:
To:
Call-ID:
Cseq:
Supported:
Accept-Contact:
Allow:
Accept:
Content-Length: 0
P-Called-Party-ID:

```

**13. Save CUA#1's address, prepare 200 OK response**

The terminating CUA caches any address related information it learns about CUA#2 from the OPTIONS request, such as its SIP URI. The terminating CUA adds feature parameters to the Contact header field in the OPTIONS response indicating its capabilities. The feature parameters that were included in the registration generated by the terminating CUA are used in the OPTIONS response.

**14. 200 (OK) response (CUA#1 to P-CSCF#1) – see example in table B.6.2-14**

The originating CUA sends a 200 (OK) response for the OPTIONS request containing the originating CUA's capabilities. This information is declared both in the media feature tags listed in the Contact header and in the SDP. In this example, CUA#1 declares capability for cs-voice and cs-video, and it declares support for MSRP, RTP video (H.263) and RTP audio (AMR).

**Table B.6.2-14: 200 (OK) response (CUA#1 to P-CSCF#1)**

```

SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.visited1.net:6809;comp=sigcomp;branch= af45K329jsgfff, SIP/2.0/UDP
scscf1.home1.net;branch= af45K329js1234, SIP/2.0/UDP icscf1_s.home1.net;branch=
af45K329jsskit, SIP/2.0/UDP scscf2.home2.net;branch= af45K329jsg213, SIP/2.0/UDP
pcscf2.visited2.net;branch= af45K329jsbbb2, SIP/2.0/UDP
[5555::eee:fff:aaa:bbb]:8805;comp=sigcomp;branch= af45K329jstd43
Record-Route: <sip:pcscf1.visited1.net:6809;lr;comp=sigcomp>, <sip:scscf1.home1.net;lr>,
<sip:scscf2.home2.net;lr>, <sip:pcscf2.visited2.net;lr>
Privacy: none
P-Access-Network-Info: 3GPP-UTRAN-TDD; utran-cell-id-3gpp=234151D0FCE11
From: <sip:user1_public1@home1.net>; tag=171828
To: <sip:user2_public1@home2.net>; tag=314159
Call-ID: ast324c2ho9512go9238ks4b
Cseq: 127 OPTIONS
Supported: gruu, preconditions, 100rel
Contact: <sip:user1_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91e6bf6;
comp=sigcomp>; +g.3gpp.cs-voice; +g.3gpp.cs-video, <tel:+12125551111>
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE, OPTIONS
Content-Type: application/sdp
Content-Length: (...)

v=0
o=- 2987933615 2987933617 IN IP6 5555:: aaa:bbb:ccc:ddd
s=-
c=IN IP6 5555::aaa:bbb:ccc:ddd
t=0 0
m=message 0 TCP/MSRP *
a=accept-types:text/plain text/html message/cpim image/jpeg image/gif video/3gpp
a=max-size:65536
m=video 0 RTP/AVP 96
a=rtpmap:96 H263-2000/90000
m=audio 0 RTP/AVP 97
a=rtpmap:97 AMR/8000
m=video 0 RTP/AVPF 96
a=rtpmap:96 H263-2000/90000
m=audio 0 RTP/AVPF 97
a=rtpmap:97 AMR/8000

```

**SDP** The SDP contains:

- The set of offered content types supported by UE#1 and desired by the user at UE#1 for an MSRP session in the accept-types attribute and indicates the maximum size message that can be received by UE#1 in the max-size attribute (no dynamically allocated attributes are included, for example, the a=path line is not there);
- The supported video codec and relevant parameters (but no dynamically allocated parameters);
- The supported audio codec and relevant parameters (but no dynamically allocated parameters).

No ports are declared since UE capabilities are just being listed.

#### 15. 200 (OK) response (P-CSCF#2 to CUA#2) – see example in table B.6.2-15

P-CSCF#2 forwards the 200 (OK) response to the originating CUA.

**Table B.6.2-15: 200 (OK) response (P-CSCF#2 to CUA#2)**

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP [5555::eee:fff:aaa:bbb]:8805;comp=sigcomp;branch=af45K329jstd43
Record-Route: <sip:pcscf1.visited1.net:6809;lr;comp=sigcomp>, <sip:scscf1.home1.net;lr>,
<sip:scscf2.home2.net;lr>, <sip:pcscf2.visited2.net;lr>
P-Asserted-Identity:
Privacy:
From:
To:
Call-ID:
CSeq:
Supported:
Contact:
Allow:
Content-Type:
Content-Length:

v=
o=
s=
c=
t=
m=
a=
a=
m=
a=
m=
a=
```

#### 16. Cache information on CUA#1's capabilities

Since no CS call is ongoing, cache the information obtained on CUA#1's capabilities for later use within a CS call.

#### 17. CS call establishment

Either UE initiates the setup of a CS call between CUA#1 and CUA#2, as described in subclause B.5. However, in subclause B.5 UE capability version and personal ME identifier are used.

### B.6.3 UE capability exchange when a CS call is already in progress

Figure B.6.3-1 shows the UE capability exchange using OPTIONS when a CS call is already in progress between CUA#1 and CUA#2.

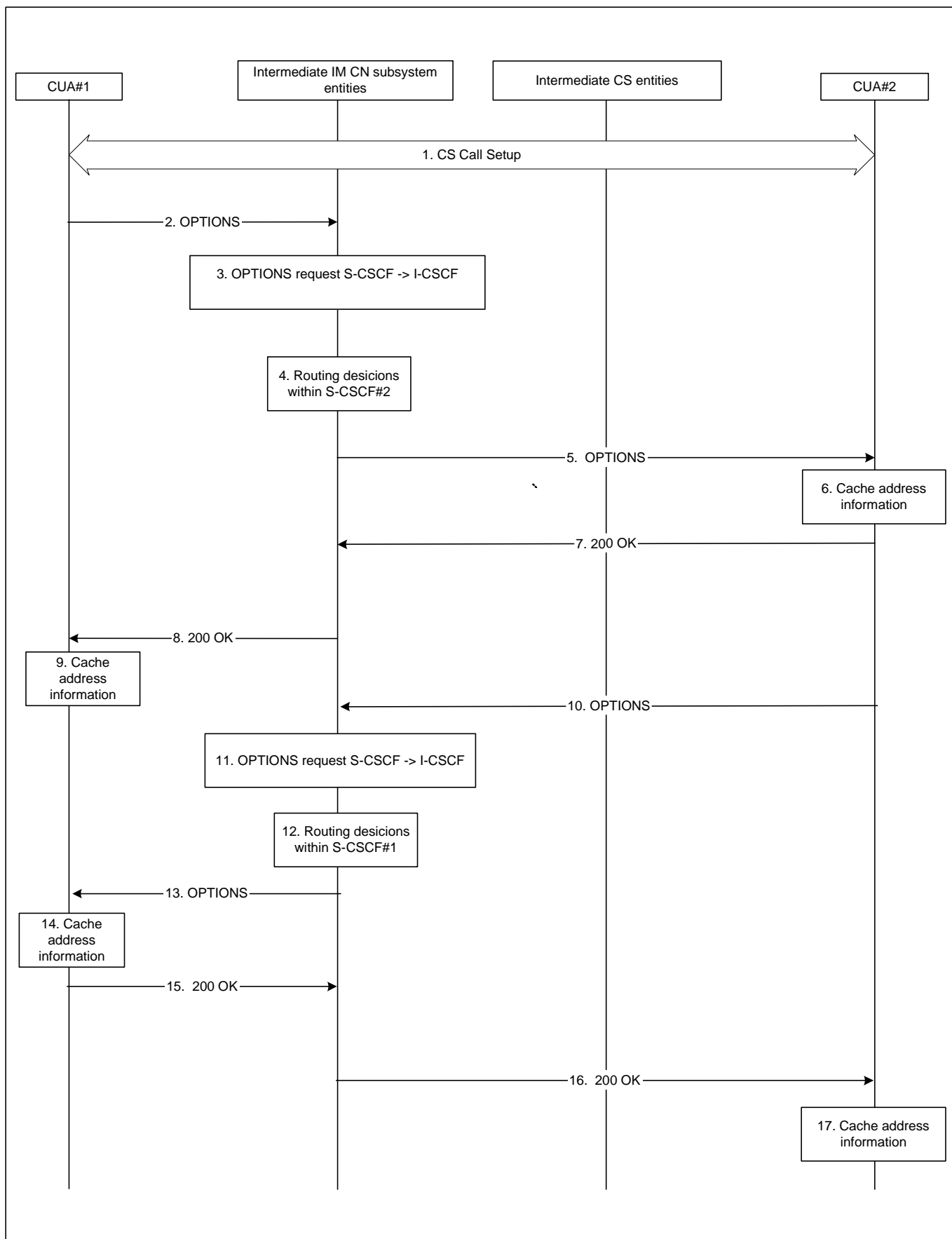


Figure B.6.3-1: UE capability exchange using OPTIONS after a CS call is established

Only the details of the signalling flows are described that are different from the flows where UE capability exchange happens before a CS call is established (see subclause B.6.2).

#### 1. CS call establishment

Either UE initiates the setup of a CS call between CUA#1 and CUA#2, as described in subclause B.5. In this example the UE capability version and personal ME identifier are not transferred as part of the CS call signalling. Therefore, CUA#1 is not aware of the capabilities of CUA#2 and starts a capability exchange.

#### 2. OPTIONS request (CUA#1 to P-CSCF#1)

As the CS call is already established, the OPTIONS request is built using the tel URI that is the same as the connected number received in the Connect Message from the already established CS call. This assumes that CUA#1 originated the CS call.

#### 2. to 5. OPTIONS request (CUA#1 to CUA#2)

Same as steps for OPTIONS when no CS call is established, see subclause B.6.2 steps 1 to 4.

#### 6. Cache information of CUA#1's capabilities and display capabilities available during this CS call

Since a CS call is ongoing, use the information in the OPTIONS request to indicate to the user which capabilities are available during this CS call. Also, cache the information about CUA#1's capabilities for later use.

#### 7. to 8. 200 (OK) response to OPTIONS request (CUA#2 to CUA#1)

Same as steps for OPTIONS when no CS call is established, see subclause B.6.2 steps 6 to 7.

#### 9. Cache information of UE#2's capabilities and display capabilities available during this CS call

Since a CS call is ongoing, use the information in the 200 (OK) response to the OPTIONS request to indicate to the user which capabilities are available during this CS call. Also, cache the information about UE#2's capabilities for later use.

#### 10. OPTIONS request (CUA#2 to P-CSCF#2)

When the CS call is already established, then the OPTIONS request is built using the tel URI that is the same as the calling number received in the SETUP Message from the already established CS call.

#### 10. to 13. OPTIONS request (CUA#2 to CUA#1)

Same as steps for OPTIONS when no CS call is established, see subclause B.6.2 steps 9 to 12.

#### 14. Cache information of CUA#2's capabilities and display capabilities available during this CS call

Since a CS call is ongoing, use the information in the OPTIONS request and if available previously stored capabilities of CUA#2 to indicate to the user which capabilities are available during this CS call. Also, cache the information about CUA#2's capabilities for later use.

#### 15. to 16. 200 (OK) response to OPTIONS request (CUA#1 to CUA#2)

Same as steps for OPTIONS when no CS call is established, see subclause B.6.2 steps 14. to 15.

#### 17. Cache information on UE#1's capabilities and display capabilities available during this CS call

Since a CS call is ongoing, use the information in the 200 (OK) response to the OPTIONS request to indicate to the user which capabilities are available during this CS call. Also, cache the information obtained on UE#1's capabilities for later use.

---

## B.7 Signalling flows demonstrating UE capability exchange with UE capability version

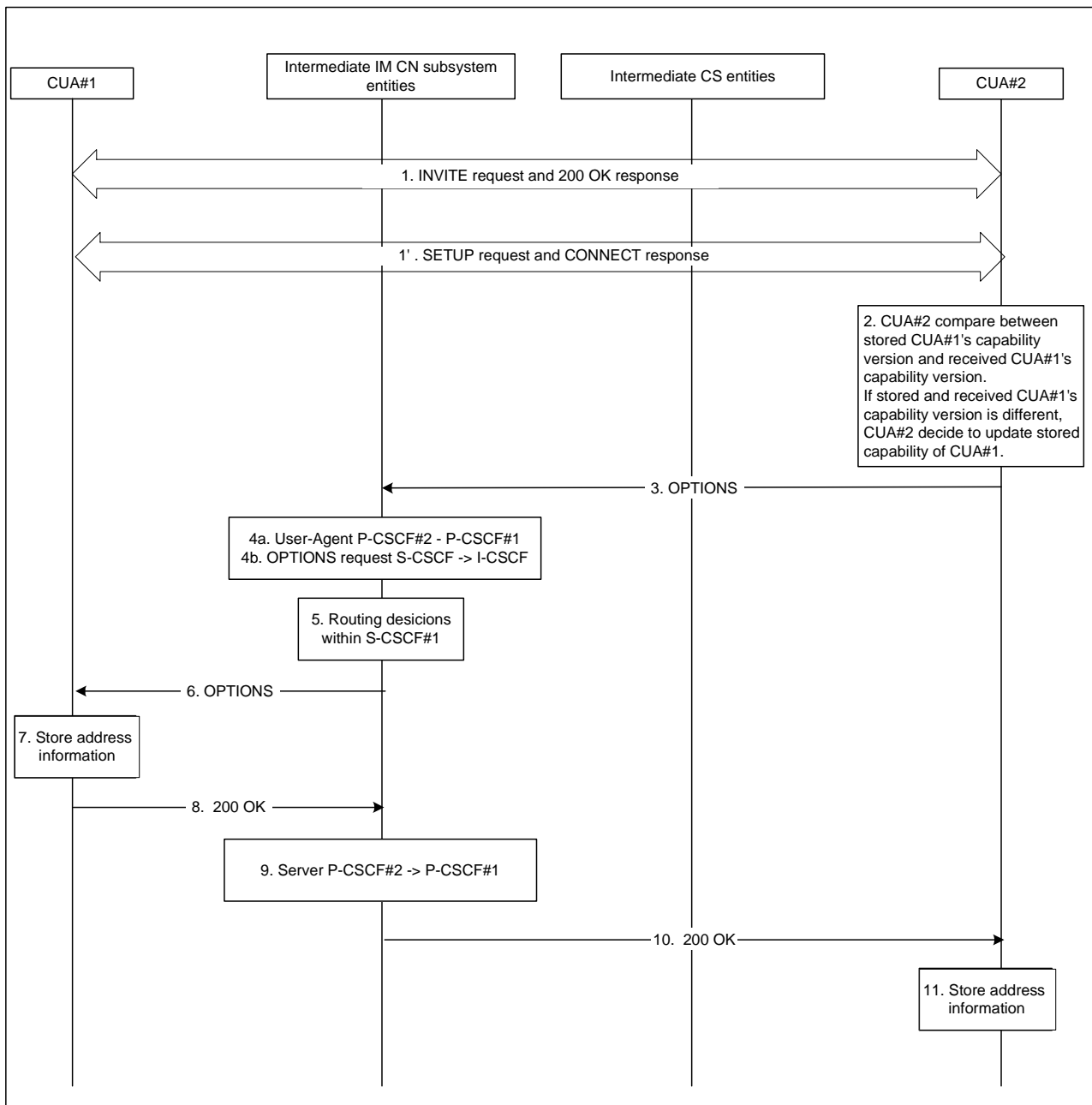
### B.7.1 Introduction

This subclause provides signalling flows for UE capability exchange with UE capability version and personal ME identifier outside a CS call or when a CS call is already in progress.

### B.7.2 UE capability exchange with UE capability version

Figure B.7.2-1 shows the UE capability exchange using OPTIONS when an IM session or a CS call is already in progress between CUA#1 and CUA#2. In this figure, CUA#2 received UE capability version and personal ME identifier, which means for instance that CUA#1's capabilities have been significantly updated.

The difference between with no UE capability version (Figure B.6.2-1, B.6.3-1) and with UE capability version (Figure B.7.2-1) of UE capability exchange is that the frequency of OPTIONS transaction is reduced through exchange of UE capability version (Figure B.7.2-1).



**Figure B.7.2-1: UE capability exchange using OPTIONS after receiving UE capability version during an IM session setup or a CS call setup**

If UE capability exchange performs after receiving UE capability version and personal ME identifier during an IM session setup, initial IM session setup messages are used as follows;

**1. INVITE request and 200 OK response**

**1a. INVITE request (CUA#1 to CUA#2) – see example in table B.7.2-1a**

CUA#1 sends INVITE request with UE#1's capability version to CUA#2. The UE#1's capability version is set to a different value from the previously stored UE#1's capability version, because the CUA#1 has updated its capability.

**Table B.7.2-1a: INVITE request (CUA#1 to CUA#2)**



```

INVITE tel:+12125552222 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: <tel:+1-212-555-1111>
P-Access-Network-Info: 3GPP-UTRAN-TDD; utran-cell-id-3gpp=234151D0FCE11
Privacy: none
From: <sip:user1_public1@home1.net>; tag=171828
To: <tel:+12125552222>
Call-ID: cb03a0s09a2sdfgklkj490333
Cseq: 127 INVITE
Require: sec-agree
Proxy-Require: sec-agree
Supported: gruu
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>
Accept-Contact: *,+g.3gpp.cs-voice,+g.3gpp.cs-video;explicit
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE, OPTIONS
Accept: application/sdp, application/3gpp-ims+xmlUser-Agent: PMI-0007, UCV-04
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
m=message 3402 TCP/MSRP *
a=accept-types:message/cpim text/plain text/html image/jpeg image/gif video/3gpp
a=path:msrp://[5555::aaa:bbb:ccc:ddd]:3402/s111271;tcp
a=max-size:131072

```

### 1b. 200 OK response (CUA#2 to CUA#1) – see example in table B.7.2-1b

CUA#2 accepts the session

**Table B.7.2-1b: 200 OK response (CUA#2 to CUA#1)**

```

SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf2.visited2.net:5088;comp=sigcomp;branch=z9hG4bK361k21.1, SIP/2.0/UDP
scscf2.home2.net;branch=z9hG4bK764z87.1, SIP/2.0/UDP
icscf2_s.home2.net;branch=z9hG4bK871y12.1, 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
Privacy: none
From: <sip:user1_public1@home1.net>; tag=171828
To: <tel:+12125552222>;tag=314159;tag=314159
Call-ID: cb03a0s09a2sdfgklkj490333
Cseq: 127 INVITE
Supported: gruu
Contact: <sip:user2_public1@home2.net;gr=urn:uuid:2ad8950e-48a5-4a74-8d99-
ad76cc7fc74;comp=sigcomp>;+g.3gpp.cs-voice,+g.3gpp.cs-video
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE, OPTIONS
Server: PMI-0EA2, UCV-0D
Content-Length: (...)

v=
o=
s=
c=
t=
m=
a=
a=
a=
a=
a=

```

If UE capability exchange performs after receiving UE capability version during a CS call setup, initial CS call setup messages are used as follows;

#### 1'. SETUP request and CONNECT response

**1'a. SETUP request (CUA#1 to CUA#2)**

CUA#1 send CS call SETUP message to CUA#2. The UE#1's capability version is set to a different value from the previously stored UE#1's capability version, because the CUA#1 has updated its capability. Specifically for CSI with UE capability version, the SETUP message includes;

- Called Party Number = [(Numbering plan identifier = ISDN/telephony numbering plan), (type of number = international number), (Number digits = 12125552222)]
- User-User IE = [(Protocol ID =3GPP capability exchange protocol), (Radio Environment = 1), (Personal ME Identifier = 0007), (UE Capability Version = 04)].

CUA#1 ensures that the Personal ME Identifier and UE capability version within the User-User IE of the SETUP correlates to the Personal ME Identifier and UE capability version used by CUA#1 in the preceding IM session.

**1'b. CONNECT response (CUA#2 to CUA#1)**

CUA#2 send CS call CONNECT message to CUA#1 to establish CS bearer. Specifically for CSI with UE capability version, the CONNECT message includes;

- User-User IE = [(Protocol ID =3GPP capability exchange protocol), (Radio Environment = 1), (Personal ME Identifier = 0EA2), (UE Capability Version = 0D)].

CUA#2 uses the Personal ME Identifier and UE capability version provided by CUA#1 in SETUP to bind the IM session to this CS Call. CUA#2 returns the Personal ME Identifier and UE capability version provided by CUA#2 within the User-User IE.

**2. CUA#2 decide whether to send OPTIONS request according to UE#1's capability version**

CUA#2 decides whether to send SIP OPTIONS request to CUA#1 according to UE capability version received during the IM session setup. If the received UE capability version is set differently from previously stored UE capability version, it means that CUA#1 has updated its capabilities. In this case, CUA#2 sends SIP OPTIONS request to CUA#1 to get CUA#1's updated capability information.

**3 to 6. OPTIONS request (CUA#2 to CUA#1)**

Same as steps for OPTIONS when IM session is established, see subclause B.6.2 steps 9. to 12.

**7. Store CUA#2's address information and display capabilities, which have been updated available during this IM session**

Since an IM session is ongoing, use the information in the SIP OPTIONS request to indicate to the user which capabilities are available during this IM session. Also, store the address information and capabilities from CUA#2 for later use.

**8. to 10. 200 (OK) response to OPTIONS request (CUA#1 to CUA#2)**

Same as steps for OPTIONS when IM session with no UE capability version is established, see subclause B.6.2 steps 14. to 15.

**11. Store information on CUA#1's capabilities**

Since an IM session is ongoing, use the information in the 200 (OK) response to the OPTIONS request to indicate to the user which capabilities are available during this IM session. Also, store the capability information from CUA#1 for later use.

---

## B.8 Signalling flows demonstrating an IMS multimedia session setup with IMS origination and CSI termination.

### B.8.1 Introduction

This subclause provides signalling flows for establishing an IM multimedia session with IMS origination and CSI termination.

The flows shows some additions related to CSI. The B2B CUA of the CSI AS includes the media feature tags `g.3gpp.cs-voice` and `g.3gpp.cs-video` in the Accept-Contact header and in the Contact header.

UE#1 illustrated in the figures is a UE that originates multimedia sessions without involving the CS domain for media transport.

### B.8.2 CSI interworking function establishing a combinational CS call and an IMS session to a terminating UE

Figure B.8.2-1 illustrates an example signalling flow for an IMS originated multimedia session (with real time voice component and a non-real time messaging component) a CSI UE interworked by the CSI interworking function in the IM CN subsystem.

In this example the messaging component is transported by MSRP.

In this example, UE#1 needs to reserve local resources.

In this example, the "100 Trying" in response to an initial INVITE and other SIP message exchanges, eg. the 183 Session Progress from CUA#2 and PRACK going to CUA#2, are not shown for simplicity.

In this example, the originating NW and the terminating NW are shown as two separate NWs. This does not imply that the originating and terminating cannot be just one NW. What this example is meant to illustrate is which part of the NW (originating or terminating) that the CSI AS for terminating session handling is invoked.

In this example the CSI AS in its 183 Session Progress to UE#1 provides an SDP answer wherein the audio component indicates a "c=" line destination pointing to the MGW and for the MSRP component the "c=" line at session level indicates the destination as the terminating CUA itself.

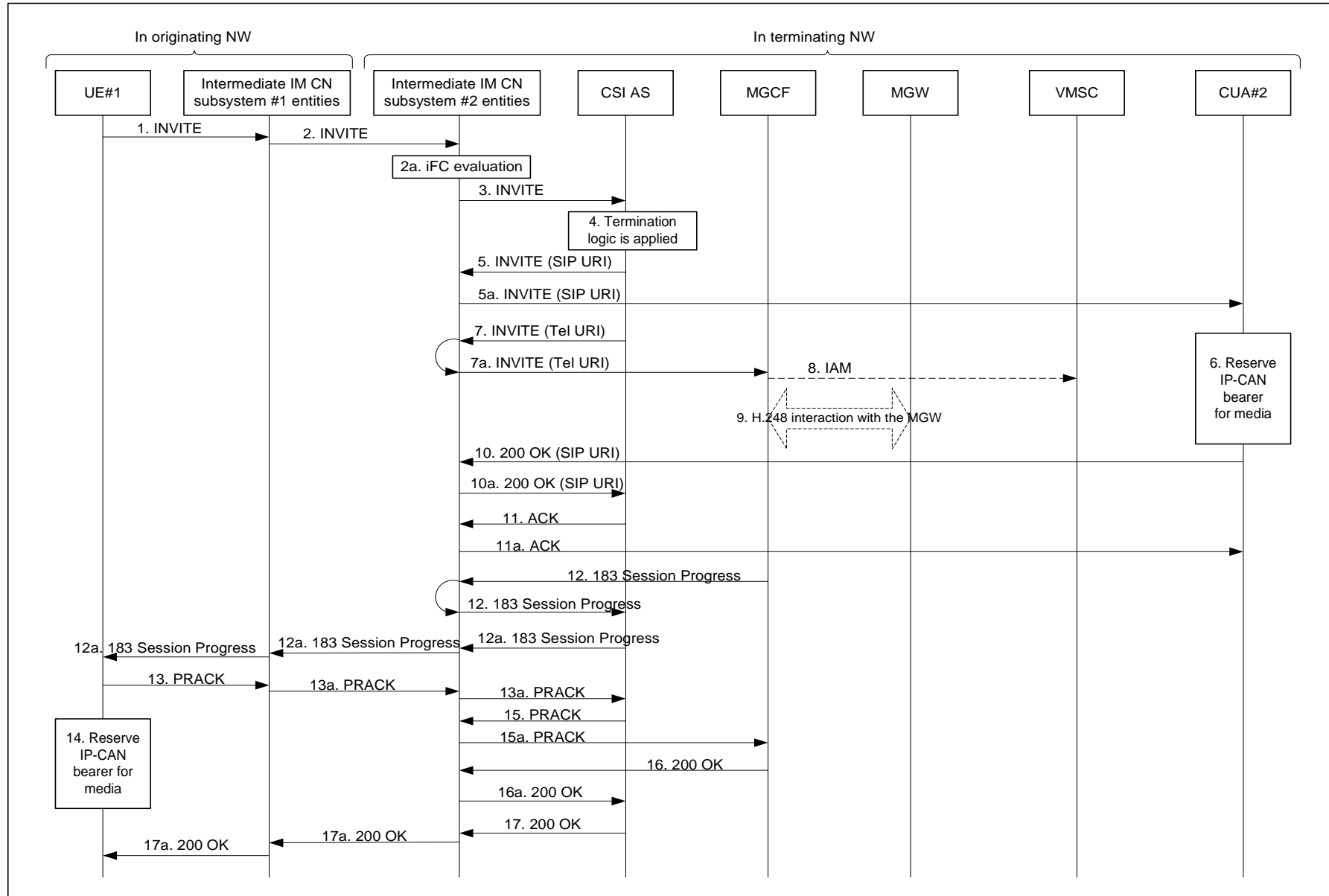


Figure B/8.2-1: Interworking an originating multimedia IMS session to a CSI UE

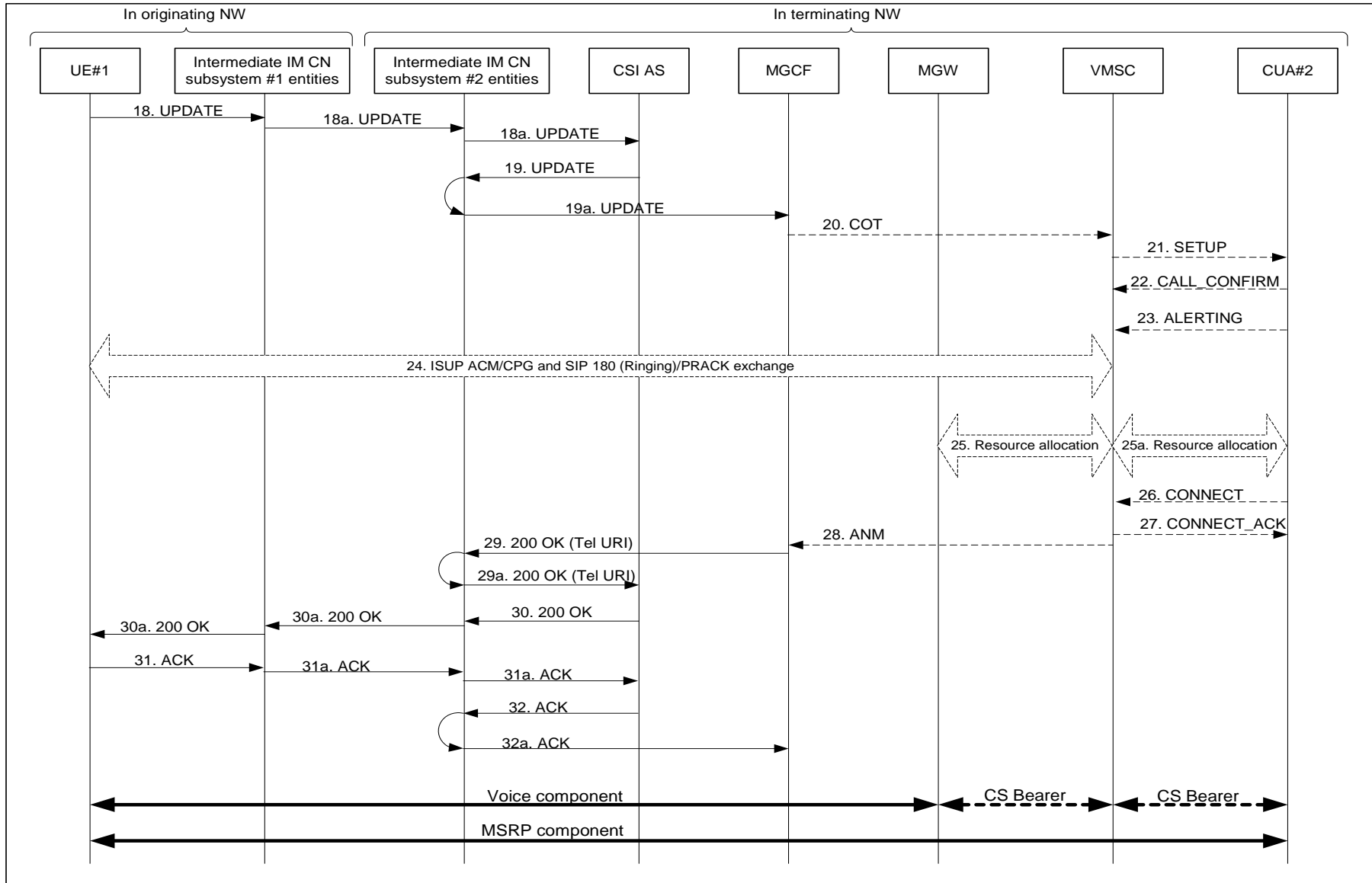


Figure B/8.2-1 (continued): Interworking an originating multimedia IMS session to a CSI UE

## 1. SIP INVITE request (from originating UE#1)

UE#1 wants to initiate a multimedia session with the terminating CUA#2 and sends out an INVITE request. See example in Table B.8.2-1.

**Table B.8.2-1: INVITE request (UE#1 to P-CSCF#1)**

```

INVITE tel:+1-212-555-2222 SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Max-Forwards: 70
Route: sip:pcscf1.home1.net:7531;lr;comp=sigcomp>, <sip:orig@scscf1.home1.net;lr>
P-Preferred-Identity: "Joe Bloke" <sip:user3_public1@home1.net>
P-Access-Network-Info: 3GPP-UTRAN-TDD; utran-cell-id-3gpp=234151D0FCE11
Privacy: none
From: <tel: +1-212-555-3333>; tag=171828
To: <tel:+1-212-555-2222>
Call-ID: cb03a0s09a2sdfg1kj490333
Cseq: 127 INVITE
Supported: 100rel; precondition, gruu
Require: sec-agree
Proxy-Require: sec-agree
Security-Verify: ipsec-3gpp; q=0.1; alg=hmac-sha-1-96; spi=87654321; port1=7531
Contact: < sip: user1_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91e6bf6;comp=sigcomp >
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATEAccept: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
m=audio 3456 RTP/AVP 97 96a=tcap:1 RTP/AVPF
a=pcfg:1 t=1
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv a=inactive
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
m=message 3402 TCP/MSRP
a=accept-types:message/cpim text/plain text/html image/jpeg image/gif video/3gpp
a=path:msrp://[5555::aaa:bbb:ccc:ddd]:3402/s111271;tcp
a=max-size:131072

```

## 2. SIP INVITE request (from originating intermediate IM CN subsystem #1 to the terminating IM CN subsystem #2)

The originating IM CN #1 maps the Tel URI to a SIP URI and forwards the INVITE request towards the terminating IM CN #2. See example in Table B.8.2.-2

**Table B.8.2-2: INVITE request (S-CSCF#1 to S-CSCF#2)**

```

INVITE <sip:user2_public2@home2.net> SIP/2.0
Via: SIP/2.0/UDP scscf1.home1.net;branch=z9hG4bK332b23.1, SIP/2.0/UDP
      pcscf1.home1.net;branch=z9hG4bK240f34.1, SIP/2.0/UDP
      [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Max-Forwards: 67
Route: <sip:as.home1.net;lr>,<sip:cb03a0s09a2sdfglkj490333@scscf1.home1.net;lr>
Record-Route: <sip:scscf1.home1.net>, <sip:pcscf1.home1.net>
P-Asserted-Identity: "Joe Bloke" <sip:user3_public1@home1.net>, <tel: +1-212-555-333>
P-Access-Network-Info:
Privacy: none
P-Charging-Vector:
P-Charging-Function-Addresses: ccf=[5555::b99:c88:d77:e66]; ccf=[5555::a55:b44:c33:d22];
ecf=[5555::1ff:2ee:3dd:4ee]; ecf=[5555::6aa:7bb:8cc:9dd]
From:
To:
Call-ID:
Cseq: 127
Supported:
Contact:
Allow:
Content-Type:
Content-Length: (...)

v=0
o=
s=
c=
t=
m=
a=
a=
b=
a=
a=
a=
a=
a=
a=
a=
a=
a=
a=
a=
a=
a=
m=
a=
a=
a=
    
```

**2a. Evaluation of initial filter criteria (at S-CSCF#2)**

The S-CSCF#2 performs iFC evaluation in order to decide what services to invoke and how to route the INVITE request. In this example, the S-CSCF#2 determines that the INVITE request must be routed to the CSI AS.

**3. SIP INVITE request (to the target CSI application server)**

S-CSCF#2 routes the INVITE to the indicated CSI application server. See example in Table B.8.2-3.

**Table B.8.2-3 INVITE request (S-CSCF#2 to CSI application server)**

```
INVITE <sip:user2_public2@home2.net> SIP/2.0
Via: SIP/2.0/UDP scscf2.home2.net;branch=y2y2y2y2y2y2.2,
     SIP/2.0/UDP scscf1.home1.net;branch=z9hG4bK332b23.1,
     SIP/2.0/UDP pcscf1.home1.net;branch=z9hG4bK240f34.1, SIP/2.0/UDP
     [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Max-Forwards: 66
Route: <sip:as.home1.net;lr>, <sip:cb03a0s09a2sdfglkj490333@scscf1.home1.net;lr>
Record-Route: <sip:scscf2.home2.net>,
              <sip:scscf1.home1.net>,
              <sip:pcscf1.home1.net>
P-Asserted-Identity: "Joe Bloke" <sip:user3_public1@home1.net>
P-Access-Network-Info:
Privacy: none
P-Charging-Vector:
P-Charging-Function-Addresses: ccf=[5555::b99:c88:d77:e66]; ccf=[5555::a55:b44:c33:d22];
ecf=[5555::1ff:2ee:3dd:4ee]; ecf=[5555::6aa:7bb:8cc:9dd]
From:
To:
Call-ID:
Cseq: 127
Supported:
Contact:
Allow:
Content-Type:
Content-Length: (...)

v=0
o=
s=
c=
t=
m=
a=
a=
b=
a=
a=
a=
a=
a=
a=
a=
a=
a=
a=
a=
a=
m=
a=
a=
a=
```

#### 4. Termination logic processing by the CSI AS

The CSI application server invokes its termination logic and based on the requested media components it decides to do a session split, i.e. to terminate the voice component via the CS domain and the session containing MSRP stream via an IP-CAN.

NOTE 1: For splitting the session into two parts, the CSI AS will later initiate two INVITE requests: one in step 5 for the MSRP component and another in step 7 for the voice component. The sequence of these INVITE requests is implementation specific (although this example shows first the INVITE for the voice component).

NOTE 2: The two sessions (for the voice and session containing MSRP stream) are presented to the user as a single composite session.

#### 5. SIP INVITE request (from CSI AS to terminating user CUA#2 in the IM CN subsystem )

The CSI AS (acting as a B2B UA) initiates an INVITE request for the session containing the MRSP stream to the terminating CUA#2 with the Request URI set to the SIP URI received in the initial INVITE request received at step 3. See example in Table B.8.2-4



**Table B.8.2-4: INVITE request (CSI AS as B2BUA to S-CSCF#2 for IMS session)**

```

INVITE sip:user2_public2@home2.net SIP/2.0
Via: SIP/2.0/UDP as.home2.net
Max-Forwards: 70
Route:
Record-Route: <sip:as.home2.net;lr>
P-Asserted-Identity: <tel: +1-212-555-3333>
P-Access-Network-Info:
P-Charging-Vector:
Privacy: none
Supported: gruu
From: <tel: +1-212-555-3333>;tag=171828
To: <tel:+1-212-555-2222>
Call-ID: cb03a0s09a2sdfglkj490333
Cseq: 127 INVITE
Contact: <sip:user1_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91e6bf6;comp=sigcomp>
Accept-Contact: *,+g.3gpp.cs-voice, +g.3gpp.cs-video;explicit
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE
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
m=message 3402 TCP/MSRP
a=accept-types:message/cpim text/plain text/html image/jpeg image/gif video/3gpp
a=path:msrp://[5555::aaa:bbb:ccc:ddd]:3402/s111271;tcp
a=max-size:131072

```

**6. Reserve IP-CAN bearer for Media at terminating side**

The terminating CUA initiates media reservation.

**7. SIP INVITE request (from CSI AS to the terminating user CUA#2 in CS domain)**

Acting again as a B2B UA the CSI AS substitutes the SIP URI in the Request URI (received in step 3) with a Tel URI and initiates a new INVITE request. See example in Table B.8.2-5.

**Table B.8.2-5: INVITE request (CSI AS as B2BUA to S-CSCF#2 for CS call)**

```

INVITE tel:+1-212-555-2222 SIP/2.0
Via: SIP/2.0 as.home.net; branch=z9hG4bKnashdt8t8
Max-Forwards: 65
Route: <sip:scscf2.home2.net>, <sip:bgcf.home2.net>
Record-Route: <sip:as.home2.net>
P-Asserted-Identity: <tel: +1-212-555-3333>
P-Access-Network-Info:
Privacy: none
From: <tel: +1-212-555-3333>; tag=171828
To: <tel:+1-212-555-2222>
Call-ID: cb03a0s09a2sdfglkj490333
Cseq: 127 INVITE
Supported: 100rel, precondition, gruu
Require: sec-agree
Proxy-Require: sec-agree
Security-Verify: ipsec-3gpp; q=0.1; alg= hmac-sha-1-96; spi=87654321; port1=7531
Contact: < sip:user1_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91e6bf6;comp=sigcomp >
Accept-Contact: *,+g.3gpp.cs-voice, +g.3gpp.cs-video;explicit
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE
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
m=audio 3456 RTP/AVP 97 96a=tcap:1 RTP/AVPF
a=pcfg:1 t=1

```

```

b=AS:25.4a=curr:qos local sendrecv=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv a=inactive
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

```

## 8. ISUP IAM (from the MGCF to the VMSC of the terminating CUA#2)

The MGCF performs the IMS/CS interworking functions and initiates an ISUP IAM towards the VMSC. Because the media for the originating side is not yet available, the MGCF indicates "Continuity check performed on previous circuit" in the IAM.

## 9. H248 interactions (between MGCF and MGW)

MGCF interacts with MGW for necessary resource allocation.

## 10. SIP 200 (OK) response (from CUA#2 back to CSI AS through intermediate IM CN subsystem #2)

After reserving an IP-CAN bearer for the message session media component the terminating CUA sends a 200 (OK) response for the INVITE request containing SDP that indicates that the terminating CUA has accepted the message session and listens on the MSRP TCP port returned in the path attribute in the answer for a TCP SETUP from the originating CUA.

CUA#2 declares support only for CS-voice, not CS-video in its Contact header. The terminating CUA includes its personal ME identifier and UE capability version in the service header.

See example in Table B.8.2-6.

**Table B.8.2-6: 200 (OK) response (CUA#2 to P-CSCF#2)**

```

SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf2.visited2.net:5088;comp=sigcomp;branch=z9hG4bK361k21.1, SIP/2.0/UDP
    scscf2.home2.net;branch=z9hG4bK764z87.1, SIP/2.0/UDP
Record-Route: <sip:pcscf2.visited2.net:5088;lr;comp=sigcomp>, <sip:scscf2.home2.net;lr>,
Privacy: none
P-Access-Network-Info: 3GPP-UTRAN-TDD; utran-cell-id-3gpp=234151D0FCE11
From: <tel:+1-212-555-3333>; tag=171828
To: <tel:+1-212-555-2222>; tag=314159
Call-ID: cb03a0s09a2sdfg1kj490333
Cseq: 127 INVITE
Supported: gruu
Contact: <sip:user2_public1@home2.net;gr=urn:uuid:2ad8950e-48a5-4a74-8d99-
    ad76cc7fc74;comp=sigcomp>;+g.3gpp.cs-voice
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE, OPTIONS
Server: PMI-0EA2, UCV-08
Content-Type: application/sdp
Content-Length: (...)

v=0
o=- 2987933615 2987933617 IN IP6 5555:: eee:fff:aaa:bbb
s=-
c=IN IP6 5555:: eee:fff:aaa:bbb
t=0 0
m=message 3402 TCP/MSRP *
a=accept-types:text/plain text/html message/cpim
a=path:msrp://[5555:: eee:fff:aaa:bbb]:3402/s234167;tcp
a=max-size:65536

```

**NOTE:** The SDP contains the set of offered content types supported by CUA#2 and desired by the user at CUA#2 for this session in the accept-types attribute and indicates the maximum size message that can be received by CUA#2 in the max-size attribute.

## 11. SIP ACK request (CSI AS back to CUA#2)

The CSI AS provides SIP ACK request to CUA#2.

## 12. SIP 183 (Session Progress) response (from MGCF to CSI AS through intermediate IM CN subsystem #2)

The MGCF provides a 183 Session Progress request back to the CSI AS through S-CSCF#2.

**12a. SIP 183 (Session Progress) response (from CSI AS to UE#1)**

The CSI AS provides a 183 Session Progress response back to the UE#1.

**12a. SIP 183 (Session Progress) response (from CSI AS to UE#1 through intermediate IM CN subsystems)**

CSI AS in turn provides 183 (Session Progress) response back to UE#1 through the intermediate IM CN subsystems. This 183 (Session progress) response from the CSI AS will include a SDP answer to UE#1's SDP offer in the initial INVITE. See example in Table B.8.2-7.

**Table B.8.2-7: 183 (Session Progress) response (CSI AS to S-CSCF#2)**

```

SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP scscf2.home2.net;branch=z9hG4bK764z87.1,
    SIP/2.0/UDP icscf2_s.home2.net;branch=z9hG4bK871y12.1,
    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:pcscf2.visited2.net:5088;lr;comp=sigcomp>>, <sip:scscf2.home2.net;lr>,
Privacy: none
P-Access-Network-Info: 3GPP-UTRAN-TDD; utran-cell-id-3gpp=234151D0FCE11
From: <tel:+1-212-555-3333>; tag=171828
To: <tel:+1-212-555-2222>
Call-ID: cb03a0s09a2sdfgk490333
Cseq: 127 INVITE
Require: precondition, 100rel
Contact: <sip:user2_public1@home2.net;gr=urn:uuid:2ad8950e-48a5-4a74-8d99-
    ad76cc7fc74;comp=sigcomp>;+g.3gpp.cs-voice
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE, OPTIONS
Server: PMI-0EA2, UCV-08
Content-Type: application/sdp
Content-Length: (...)

v=0
o=- 2987933615 2987933617 IN IP6 5555:: eee:fff:aaa:bbb
s=-
c=IN IP6 5555:: eee:fff:aaa:bbb
t=0 0
m=audio 3456 RTP/AVPF 97 96
a=acfg:1 t=1
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=inactive
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
c=IN IP6 5555:: eee:fff:www:yyy
t=0 0
m=message 3402 TCP/MSRP *
a=accept-types:text/plain text/html message/cpim
a=path:msrp://[5555:: eee:fff:aaa:bbb]:3402/s234167;tcp
a=max-size:65536

```

**13, 13a. SIP PRACK request (from UE#1)**

UE#1 upon receipt of 183 (Session Progress) response replies back with PRACK request. With that UE#1 initiates resource allocation for the session it originated.

**14 Reserve IP-CAN bearer for Media at originating side**

The originating UE starts resource reservation for media bearers.

**15, 15a. SIP PRACK request (from CSI AS to MGCF)**

The PRACK from UE#1 lead the CSI AS to provide PRACK to the MGCF

**16, 16a SIP 200 (OK) response (MGCF to CSI AS )**

MGCF acknowledges the PRACK (from CSI AS) with 200 (OK) (back to CSI AS)

**17, 17a. SIP 200 (OK) response (from CSI AS to UE#1 through intermediate IM CN subsystems)**

CSI AS correspondingly provides the 200 (OK) back to UE#1 through the intermediate IM CN subsystems.

**18, 18a SIP UPDATE request (from UE#1 to CSI AS through intermediate IM CN subsystems)**

UE#1 indicates it has available the necessary resources.

**19, 19a SIP UPDATE request (from CSI AS to MGCF through intermediate IM CN subsystem#2)**

CSI AS provides the indication of available resources at originating side to the MGCF.

**20. ISUP COT (from MGCF to VMSC)**

With the UPDATE indicating that originating side has the necessary media resources and as such conditions are met, the MGCF sends COT to the VMSC indicating "continuity check successful".

**21. SETUP (from VMSC to CUA#2)**

The VMSC now sends the SETUP towards the terminating CUA#2.

**22. CALL CONFIRM (from CUA#2 to VMSC)**

The terminating UE acknowledges and confirms acceptance of the incoming call by sending CONFRIM back to the VMSC.

NOTE: There is no suggestion, by this example or otherwise, that CUA#2 only response with CALL CONFIRM after the INVITE for the IM CN subsystem is received.

**23. ALERTING (from CUA#2 to VMSC)**

For this example, the VMSC allows early local alerting. With the generation of local alerting, ALERTING is sent back from the UE to the VMSC.

**24. ACM, CPG, SIP 180 (Ringing) response and SIP Prack request (VMSC to MGCF through IM CN Subsystem to originating end)**

On receipt of the ALERTING from the terminating UE, the VMSC, MGCF, IM CN subsystem and the originating UE exchange ISUP and SIP signalling to indicate that the called party confirmed the call and did start ringing the end user.

**25, 25a . Resource allocation (VMSC to terminating UE, MGW and VMCS)**

The VMSC starts resource allocation towards the terminating UE.

Between VMSC and the MGW resource allocation is also started.

**26. CONNECT (from CUA#2 to VMSC)**

When the user answers the call, the terminating UE sends CONNECT to the VMSC. After sending the CONNECT, the terminating UE is ready to connect to user plane resources.

**27. CONNECT ACK (from VMCS to CUA#2)**

VMSC connects up the user plane and return CONNECT\_ACKNOWLEDGE to the terminating UE..

Through connect all the way back to originating UE is not initiated.

**28. ANM (from VMSC to MGCF)**

The VMSC having through connected the user plane sends an indication of answer to the MGCF. This is the ISUP ANM message.

**29. SIP 200 (OK) response (from MGCF to the CSI AS via the intermediate IM CN subsystem #2)**

Upon the receipt of the ANM from the VMSC, and after the connection of the media flow, the MGCF sends the SIP 200 (OK) final response to the received SIP INVITE request (in step 5) back to the CSI AS via the intermediate IM CN subsystem #2.

**30, 30a . SIP 200 (OK) response (from CSI AS to intermediate IM CN subsystem #2)**

The SIP 200 (OK) final response is provided by the CSI AS back to the originating UE#1 through the terminating intermediate IM CN subsystem #2.

NOTE: In this example, the CSI AS provides the 200 (OK) response back to the originating UE#1 after both the voice session and the session containing the MRSP have been successful established.

**31, 31a. SIP ACK (from UE#1 to CSI AS through intermediate IM CN subsystems)**

UE#1 responds to the 200 (OK) with ACK

**32, 32a. SIP ACK (CSI AS to MGCF)**

The CSI AS correspondingly provides the ACK to the MGCF.

## Annex C (informative): Change history

Change history							
Date	TSG #	TSG Doc.	CR	R e v	Subject/Comment	Old	New
2006-02					Version 1.0.0 created by shifting text from TR 24.879 v1.2.0, as per C1-060527, after CT1 41.	-	1.0.0
2006-02					Version 2.0.0 created by MCC, editorial changes	1.0.0	2.0.0
2006-03	CT-31	CP-060099			Version 2.0.0 was approved and version 7.0.0 created by MCC	2.0.0	7.0.0
2006-06	CT-32	CP-060275	0001	2	Correction of notation for the signalling flows	7.0.0	7.1.0
2006-06	CT-32	CP-060275	0009		Use of the Contact header in OPTIONS requests for capability exchange	7.0.0	7.1.0
2006-06	CT-32	CP-060275	0011	1	Referencing TS 26.141	7.0.0	7.1.0
2006-09	CT-33	CP-060461	0012		Correction of chapter 4	7.1.0	7.2.0
2006-09	CT-33	CP-060461	0013	3	Correction of criteria for initiating capability exchange	7.1.0	7.2.0
2006-09	CT-33	CP-060461	0015	1	OPTIONS request inside a session	7.1.0	7.2.0
2006-09	CT-33	CP-060461	0016		Usage of UCV and PMI	7.1.0	7.2.0
2006-09	CT-33	CP-060461	0017		tel URI usage during session setup	7.1.0	7.2.0
2006-09	CT-33	CP-060461	0020	1	Merging of the Call procedures for both IMS-first and CS-first with UE capability version	7.1.0	7.2.0
2006-09	CT-33	CP-060472	0019	2	Provision and exchange of IMS status	7.1.0	7.2.0
2006-11	CT-34	CP-060669	0021	2	Signalling flows involving the CSI application server	7.2.0	7.3.0
2006-11	CT-34	CP-060669	0022	1	CSI Termination session handling	7.2.0	7.3.0
2006-11	CT-34	CP-060666	0023	1	Correction of signalling flows	7.2.0	7.3.0
2007-03	CT-35	CP-070137	0028	2	The CSI AS uses the capabilities of the CUA to perform termination logic	7.3.0	7.4.0
2007-03	CT-35	CP-070137	0032	3	Signalling flow for different "c=" in SDP Offer/Answer	7.3.0	7.4.0
2007-03	CT-35	CP-070137	0035	1	SIP URI-Tel URI mapping information in CSI AS	7.3.0	7.4.0
2007-03	CT-35	CP-070137	0036	2	Corrections to CSI termination	7.3.0	7.4.0
2007-03	CT-35	CP-070243	0034	3	Procedures of CSI AS	7.3.0	7.4.0
2007-06	CT-36	CP-070390	0049	1	Correction of references to wrong subclauses	7.4.0	7.5.0
2007-06	CT-36	CP-070390	0038	2	Video interworking alternatives	7.4.0	7.5.0
2007-06	CT-36	CP-070390	0039	3	Retrival of Public user identity information	7.4.0	7.5.0
2007-06	CT-36	CP-070390	0040	3	Session handling	7.4.0	7.5.0
2007-06	CT-36	CP-070390	0044	3	Inclusion of new call cases for IMS session setup	7.4.0	7.5.0
2007-06	CT-36	CP-070390	0045	1	Additions of a call case adding a cs call to an IMS session	7.4.0	7.5.0
2007-06	CT-36	CP-070390	0048	2	Enhancement of session release procedures	7.4.0	7.5.0
2007-06	CT-36	CP-070390	0050	-	Removal of references to MGCF interworking and information for interworking	7.4.0	7.5.0

2007-06	CT-36	CP-070390	0052	2	Further details of combining procedure in CSI AS	7.4.0	7.5.0
2007-06	CT-36	CP-070390	0053	3	CSI AS handling of error response	7.4.0	7.5.0
2007-06	CT-36	CP-070390	0054	1	CSI AS interactions with supplementary services	7.4.0	7.5.0
2007-06	CT-36	CP-070390	0055	3	Unregistered CUA service handling for CSI termination	7.4.0	7.5.0
2007-06	CT-36	CP-070390	0057	4	Detailed procedures about session combining	7.4.0	7.5.0
2007-06	CT-36	CP-070390	0046	4	Clarification on modifying IMS and CS session	7.4.0	7.5.0
2007-06	CT-36	CP-070391	0059	-	g.3gpp.cs-voice and g.3gpp.cs-video registered by IANA	7.4.0	7.5.0
2007-06	CT-36	CP-070481	0051	4	Further details on splitting procedure	7.4.0	7.5.0
2007-06	CT-36	CP-070482	0043	6	Additions to the session initiating cases to the IMS session setup	7.4.0	7.5.0
2007-09	CT-37	CP-070594	0061		Removal of Editor's note in subclause 9.3.3.3.1	7.5.0	7.6.0
2007-09	CT-37	CP-070594	0060	1	Corrections of wrong specifications and requirements of the CSI AS	7.5.0	7.6.0
2007-12	CT-38	CP-070790	0063		ASN.1 not included for feature tags CS-voice and CS-video	7.6.0	7.7.0
2007-12	CT-38	CP-070790	0064		Correction to B.6.2 and B.7.2 signalling flows	7.6.0	7.7.0
2007-12	CT-38	CP-07814	0066	3	Capability Exchange for CSI termination	7.7.0	8.0.0
2008-12	CT-42	CP-080848	0067		Corrections of flows in 24.279	8.0.0	8.1.0
2009-12	CT-46	CP-091041	0068	2	Remove the Editor's Notes in TS 24.279	8.1.0	8.2.0
2009-12	CT-46				Upgrade to Rel-9	8.2.0	9.0.0
2014-03	CT-63	CP-140118	0070		Updating of UUS references	9.0.0	9.1.0



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## History

<b>Document history</b>		
V9.0.0	January 2010	Publication
V9.1.0	March 2014	Publication