3GPP TR 24.879 V7.0.0 (2006-03)

Technical Report

3rd Generation Partnership Project;
Technical Specification Group Core Network and Terminals;
Combining CS calls and IMS sessions;
Stage 3
(Release 7)

The present document has been developed within the 3rd Generation Partnership Project (3GPP TM) and may be further elaborated for the purposes of 3GPP.

The present document has not been subject to any approval process by the 3GPP Organizational Partners and shall not be implemented.
This Specification is provided for future development work within 3GPP only. The Organizational Partners accept no liability for any use of this Specification.
Specifications and reports for implementation of the 3GPP TM system should be obtained via the 3GPP Organizational Partners' Publications Offices.

Keywords

IMS, CSI, SIP

3GPP

Postal address

3GPP support office address

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

Internet

http://www.3gpp.org

Copyright Notification

No part may be reproduced except as authorized by written permission.
The copyright and the foregoing restriction extend to reproduction in all media.

© 2006, 3GPP Organizational Partners (ARIB, ATIS, CCSA, ETSI, TTA, TTC).
All rights reserved.

Contents

Foreword 6

1 Scope 7

2 References....7

3 Definitions, symbols and abbreviations....8
3.1 Definitions 8
3.2 Abbreviations....8

4 Common capability information and identifiers for CS-first and IMS-first scenarios....8
4.1 UE capability exchange - overview....8
4.2 Personal ME identifier....8
4.3 UE capability version....9
4.4 Radio environment information 9

5 Common procedures for CS-first and IMS-first scenarios 9
5.1 Registration of UE capabilities 9
5.2 Criteria for initiating capability exchange....9
5.3 Criteria for responding to a capability exchange request 10
5.4 Exchange of radio environment information 10
5.5 Storage of capabilities in the UE 10

6 Combining a CS call with an existing IM session 10
6.1 Introduction....10
6.2 Functional entities 10
6.2.1 User equipment....10
6.2.2 Application server (AS) 11
6.3 Roles 11
6.3.1 CSI user agent (CUA) 11
6.3.1.1 General....11
6.3.1.2 Exchange of UE capability information – IM session first scenario 11
6.3.1.3 Session set-up – originating case....11
6.3.1.4 Session set-up – terminating case 12
6.3.1.5 CS call set-up – originating case....12
6.3.1.6 CS call set-up – terminating case....13
6.3.1.7 SDP exchange – UE capability information exchange 13
6.3.1.8 SDP offer answer – originating case 13
6.3.1.9 SDP offer answer – terminating case 13

7 Combining an IM session with an existing CS call 14
7.1 Introduction....14
7.2 Functional entities 14
7.2.1 User equipment....14
7.2.2 Application server (AS) 14
7.3 Roles 14
7.3.1 CSI user agent (CUA) 14
7.3.1.1 General....14
7.3.1.2 Exchange of UE capability information – CS first scenario 14
7.3.1.3 Session set-up – originating case....15
7.3.1.4 Session set-up – terminating case 15
7.3.1.5 CS call set-up – originating case....15
7.3.1.6 CS call set-up – terminating case....16
7.3.1.7 SDP exchange – UE capability information exchange....16
7.3.1.8 SDP offer-answer – originating case....16
7.3.1.9 SDP offer-answer – terminating case....16

8 Interaction with supplementary services....17

Annex A (normative): Media feature tags defined within the current document....17

A.1 General....17

A.2 Definition of "+g.3gpp.cs-voice"....17

A.3 Definition of "+g.3gpp.cs-video"....17

Annex B (informative): Example signalling flows for the combining of CS calls with IM sessions....18

B.1 Scope of signalling flows....18

B.2 Introduction....18

B.2.1 General....18
B.2.2 Key required to interpret signalling flows....18
B.3 Signalling flows demonstrating CSI session setup when no CS call has yet been set up....19
B.3.1 Introduction....19
B.3.2 Establishing a CSI session when no CS call has yet been set up....19

B.4 Signalling flows demonstrating CSI session setup when a CS call is already established....28
B.4.1 Introduction....28
B.4.2 Establishing a CSI session when CS call has been setup....28

B.5 Signalling flows demonstrating capability information exchange in a CS call (only)....38
B.5.1 Introduction....38
B.5.2 Establishing a CS call from a CSI capable UE....38

B.6 Signalling flows demonstrating UE capability exchange outside a CS call....41
B.6.1 Introduction....41
B.6.2 UE capability exchange....41
B.6.3 UE capability exchange with UE capability version....49

B.7 Signalling flows demonstrating UE capability exchange when a CS call is already in progress....51
B.7.1 Introduction....51
B.7.2 UE capability exchange with no UE capability version....51
B.7.3 UE capability exchange with UE capability version....53

Annex C (informative): Proposed changes to 3GPP TS 24.008 [4]....55

2 References....55
10.5.4.25 User-user....60

Annex X (normative): 3GPP capability exchange protocol....61

X.1 Scope....61

X.2 User-user protocol contents....62

X.3 Information element identifier....62

X.4 Information elements....63
X.4.1 Personal ME identifier....63
X.4.2 Radio environment capability....64
X.4.3 UE capability version....64

X.5 Handling of unknown, unforeseen, and erroneous protocol data....65
X.5.1 General....65
X.5.2 Not supported IEs, unknown IEIs....66
X.5.3 Repeated IEs....66
X.5.4 Syntactically incorrect IEs....66
X.5.5 Semantically incorrect IEs....66

Annex D (informative): Proposed changes to 3GPP TS 24.229 [5]....66
A.1.3 Roles....66

A.2 Profile definition for the Session Initiation Protocol as used in the present document 68
A.2.1 User agent role 68
A.2.1.1 Introduction 68
A.2.1.2 Major capabilities 69

Annex E (informative): Change history 72

Foreword

This Technical Specification has been produced by the 3rd Generation Partnership Project (3GPP).

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

Version x.y.z

where:

x the first digit:

1 presented to TSG for information;

2 presented to TSG for approval;

3 or greater indicates TSG approved document under change control.

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

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

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.

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".

[5] 3GPP TS 24.229: "IP multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP)".

[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".

[8D] 3GPP TS 24.279: "Combining Circuit Switched (CS) and IP Multimedia Subsystem (IMS) services, stage 3, Release 7".

[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 ".

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 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

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.

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.

At CS call setup or IM session setup, the UE capability version identifies current UE capability information.

The UE capability version may be exchanged during:

- IM session set up, and
- CS call set up.

4.4 Radio environment information

The following information is exchanged:

- the radio environment the terminal is in at CS call setup supports simultaneous CS and PS services.

NOTE: Radio environment information is exchanged at CS call setup only.

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

5.1 Registration of UE capabilities

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

    - requires information of the remote UE capabilities;

        - no radio environment information is received or the received radio environment information indicates that the remote UE and remote radio environment is capable of handling PS and CS domain simultaneously and the received MSIDN or MSISDN plus personal ME identifier is unknown to the UE; or

    - the received remote UE's capability version is different from the stored remote UE's capability version.

b) when the UE is in an IM session with the remote UE, and requires information of the remote UE capabilities and the received INVITE request does not include the UE capability information; and

- the received public user identity or public user identity plus personal ME identifier is unknown to the UE; and

- the received remote UE's capability version is different from the stored remote UE's capability version.

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.

NOTE 2: The method used by UE to determine when it is required to update the remote UE capabilities is implementation dependent.

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

NOTE 3: 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 4: 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

A UE may send information about the current radio environment during CS Call set-up. If the UE finds that the remote UE 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.

6 Combining a CS call with an existing IM session

6.1 Introduction

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.

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 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.

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 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 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 UE capability version..

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 feature tag(s) with the values "+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 UE capability version.

If the response includes a personal ME identifier and the combination of public user identity plus personal ME identifier 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 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.

If the request includes a personal ME identifier 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 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. 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 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 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.

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 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 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, 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].
- 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 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 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.

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 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

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 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.

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 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.

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 should 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 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 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.

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 in a 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 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.

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 personal ME identifier and the UE capability version as specified in 3GPP TS 24.008 [4] Annex O. 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 the combination of MSISDN plus personal ME identifier 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. If originating UE finds that terminating UE and its current radio environment supports simultaneous CS and PS services, 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 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 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.

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 personal ME identifier and the UE capability version as defined in 3GPP TS 24.008 [4] Annex O. 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 the combination of MSISDN 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. If terminating UE finds that originating UE and its current radio environment supports simultaneous CS and PS services, 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 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 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.

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.

8 Interaction with supplementary services

No protocol interactions for supplementary services are identified.

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

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.

Editor's note: The media feature tag values defined in this Annex need IANA registration.

A.2 Definition of "+g.3gpp.cs-voice"

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

ASN.1 Identifier: New assignment by IANA.

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 "+g.3gpp.cs-video"

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

ASN.1 Identifier: New assignment by IANA.

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].

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 show 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 relating to CUA#1; and
- tel:+12125552222 is the tel URI relating to CUA#2.

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:

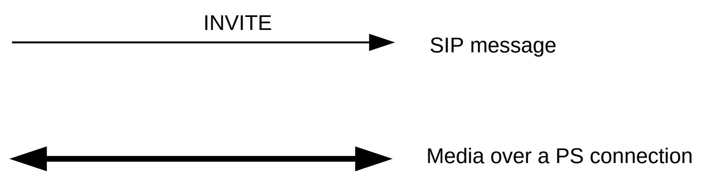

Figure B.2.2-1: Signalling flow notation

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 feature tag "+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.

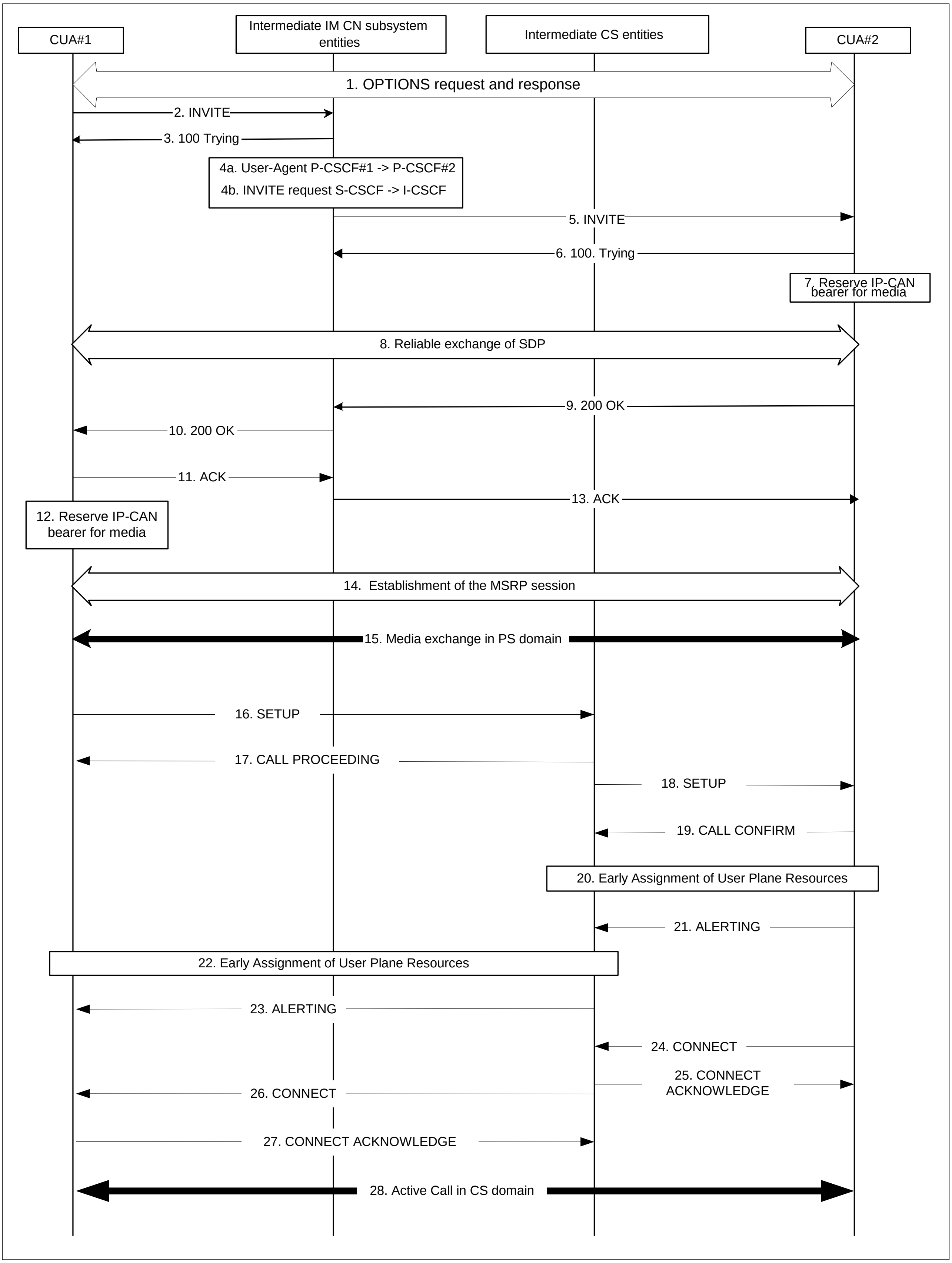

Figure B.3.2-1: Establishment of a session before a CS call is established

The details of the signalling flows are as follows:

1. **UE capabilities exchange**

Both UE's have performed a capability exchangeaccording to 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 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: cb03a0s09a2sdfglkj490333
Cseq: 127 INVITE
Require: sec-agree
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:[5555::aaa:bbb:ccc:ddd]:1357;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
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+6O2IrT5tAFrbHLso=023551024"; orig-ioi=home1.net
Privacy:
From:
To:
Call-ID:
Cseq:
Contact:
Accept-Contact:
Allow:
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:
Contact:
Accept-Contact:
Allow:
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 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: cb03a0s09a2sdfglkj490333
Cseq: 127 INVITE
Contact: <sip:[5555::eee:fff:aaa:bbb]:8805;comp=sigcomp>;+g.3gpp.cs-voice
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE, OPTIONS
Server: PMI-0EA2
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:
Require:
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:[5555::eee:fff:aaa:bbb]:8805;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: cb03a0s09a2sdfglkj490333
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 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:
```

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 = 12125552222)]
- User-User IE = [(Protocol ID =3GPP capability exchange protocol), (Capability Identifier = 0, Radio Environment = 1), (Capability Identifier = 1, Personal ME Identifier = 0007)].

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 IMS 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 = 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), (Capability Identifier = 0, Radio Environment = 1), (Capability Identifier = 1, Personal ME Identifier = 0007)].

The Personal ME Identifier provided by CUA#1 is used by CUA#2 to relate to the appropriate terminal having the parallel IMS Session.

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), (Capability Identifier =0, Radio Environment = 1), (Capability Identifier = 1, Personal ME Identifier = 0EA2)].

CUA#2 uses the Personal ME Identifier provided by CUA#1 in SETUP to bind the IMS session to this CS Call. In return the Personal ME Identifier provided by CUA#2 within the User-User IE. This Personal ME Identifier is the same as that provided by CUA#2 in 200 OK (message step 9) of the IMS 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), (Capability Identifier =0, Radio

Environment = 1),
(Capability Identifier = 1, Personal ME Identifier = 0EA2)].

The Personal ME Identifier in the User-User IE is used by the CUA#1 to bind the IMS session to this CS call.

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 feature tag "+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.

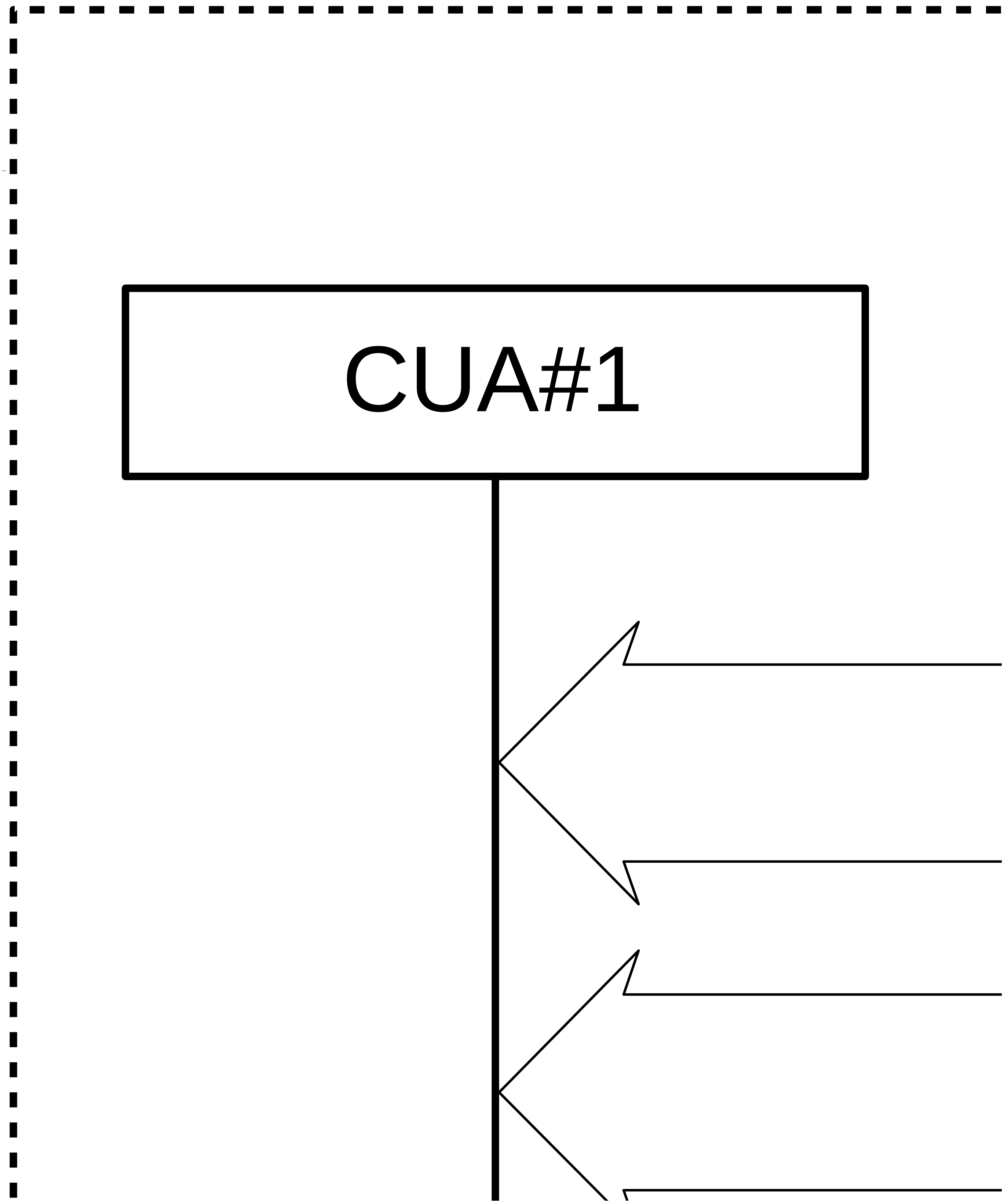

Figure B.4.2-1: Establishment of a video session after a CS call is established

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 UE's have performed a capability exchange according to 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 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 feature tags in the Accept-contact header and includes its personal ME identifier 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
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:[5555::aaa:bbb:ccc:ddd]:1357;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
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
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:
User-Agent:
Content-Type:
Content-Length:

v=
o=
s=
c=
t=
m=
b=
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:
User-Agent:
P-Called-Party-ID:
Content-Type:
Content-Length:(...)

v=
o=
s=
c=
t=
m=
b=
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 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: cb03a0s09a2sdfglkj490333
Cseq: 127 INVITE
Contact: <sip:[5555::eee:fff:aaa:bbb]:8805;comp=sigcomp>;+g.3gpp.cs-voice,+g.3gpp.cs-video
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE, OPTIONS
Require: 100rel, precondition
Server: PMI-0EA2
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/AVP 107
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:
Server:
Content-Type:
Content-Length:

v=0
o=-
s=-
c=
t=
m=
b=
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:[5555::eee:fff:aaa:bbb]:8805;comp=sigcomp>;+g.3gpp.cs-voice,+3gpp.cs-video
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: cb03a0s09a2sdfglkj490333
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
Contact: <sip:[5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp>;+g.3gpp.cs-voice;+g.3gpp.cs-video
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
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:[5555::eee:fff:aaa:bbb]:8805;comp=sigcomp>;+g.3gpp.cs-voice, +g.3gpp.cs-video
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=
```

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: cb03a0s09a2sdfglkj490333
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/AVP 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=rtpmap: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=
```

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: cb03a0s09a2sdfglkj490333
Cseq: 127 INVITE
Contact: <sip:[5555::eee:fff:aaa:bbb]:8805;comp=sigcomp>;+g.3gpp.cs-voice, +g.3gpp.cs-video
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE, OPTIONS
Server: PMI-0EA2
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:
Privacy:
P-Asserted-ID: < user2_public1@home2.net>, < tel:+12125552222>
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:[5555::eee:fff:aaa:bbb]:8805;comp=sigcomp>;+g.3gpp.cs-voice, +g.3gpp.cs-video
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: cb03a0s09a2sdfglkj490333
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:[5555::eee:fff:aaa:bbb]:8805;comp=sigcomp>;+g.3gpp.cs-voice, +g.3gpp.cs-video
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 and Personal ME Identifier by the CUAs of a CSI capable UE during a normal CS call establishment.

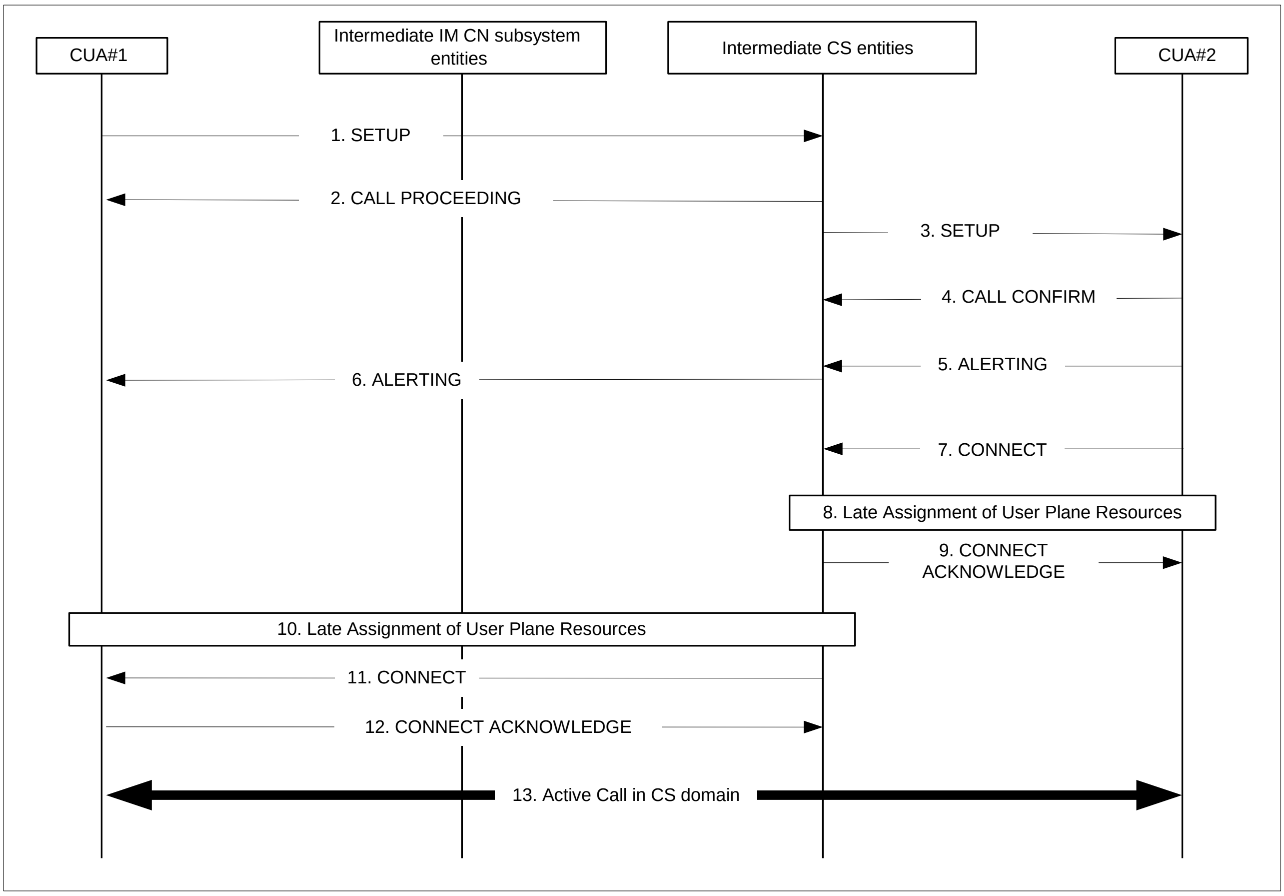

Figure B.5.2-1: CS call establishment (example of late assignment of user plane resources)

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), (Capability Identifier = 0, Radio Environment = 1),
     (Capability Identifier = 1, Personal ME Identifier = 0007)].

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.

Specifcally 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), (Capability Identifier = 0, Radio Environment = 1),
  (Capability Identifier = 1, Personal ME Identifier = 0007)].

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 Internediate 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), (Capability Identifier = 0, Radio Environment = 1),
  (Capability Identifier = 1, Personal ME Identifier = 0EA2)].

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 IMS 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), (Capability Identifier = 0, Radio Environment = 1),
 (Capability Identifier = 1, Personal ME Identifier = 0EA2)].

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 outside a CS call

B.6.1 Introduction

This subclause provides signalling flows for UE capability exchange outside a CS call.

B.6.2 UE capability exchange

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

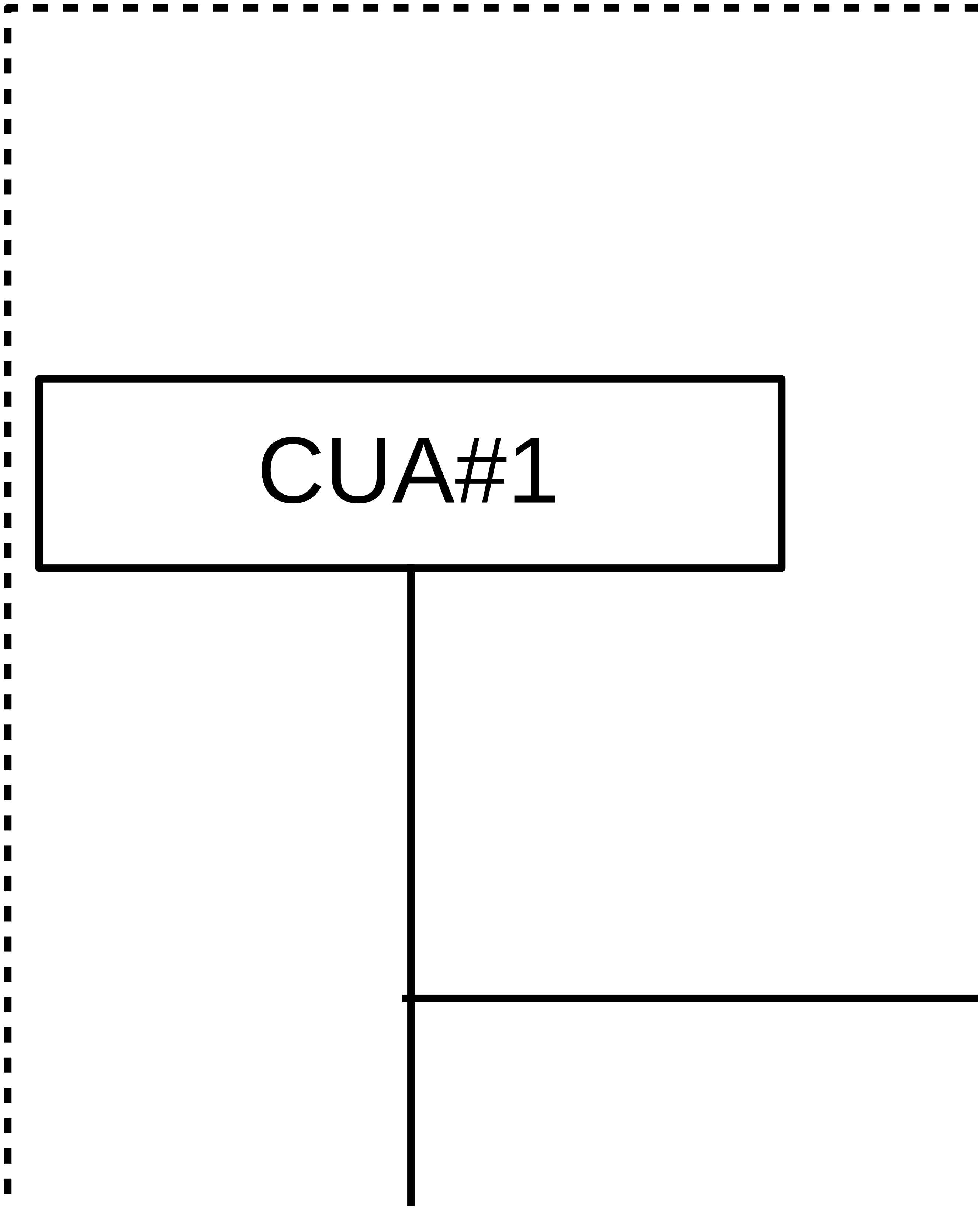

Figure B.6.2-1: UE capability exchange before any CS call is established

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
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:[5555::aaa:bbb:ccc:ddd]:1357;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
User-Agent: PMI-0007
Content-Length: 0
```

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

The User-Agent header field is transparently passed from P-CSCF#1 to P-CSCF#2.2b. **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+6O2IrT5tAFrbHLso=023551024"; orig-ioi=home1.net
Privacy:
From:
To:
Call-ID:
Cseq:
Contact:
Accept-Contact:
Allow:
Accept:
User-Agent:
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:
Contact:
Accept-Contact:
Allow:
Accept:
User-Agent:
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 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). The terminating CUA includes its personal ME identifier in the Server header.

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: cb03a0s09a2sdfglkj490333
Cseq: 127 OPTIONS
Contact: <sip:user2_public1@home2.net >;+g.3gpp.cs-voice, <tel:+12125552222>
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE, OPTIONS
Server: PMI-0EA2
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
```

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.

6b. **Server (P-CSCF#2 to P-CSCF#1)**

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

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:
Require:
Contact:
Allow:
Server:
Content-Type:
Content-Length:

v=
o=
s=
c=
t=
m=
a=
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
Security-Verify: ipsec-3gpp; q=0.1; alg=hmac-sha-1-96; spi-c=23456789; spi-s=12345678;
   port-c=8318; port-s=7743
Contact: <sip:[5555::eee:fff:aaa:bbb]:8805;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
User-Agent: PMI-0EA2
Content-Length: 0
```

10a. **User-Agent (P-CSCF#2 to P-CSCF#1)**

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

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

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

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

```
OPTIONS sip:user2_public1@home2.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-ioi=home2.net
Privacy:
From:
To:
Call-ID:
Cseq:
Contact:
Accept-Contact:
Allow:
Accept:
User-Agent:
Content-Length: 0
```

11. **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.

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

P-CSCF#2 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:
Contact:
Accept-Contact:
Allow:
Accept:
User-Agent:
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#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.

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 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.6.2-14: 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
Contact: <sip:user1_public1@home1.net >; +g.3gpp.cs-voice; +g.3gpp.cs-video,
   <tel:+12125551111>
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE, OPTIONS
Server: PMI-0007
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
```

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.

14b. **Server (P-CSCF#1 to P-CSCF#2)**

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

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

P-CSCF#1 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:
Require:
Contact:
Allow:
Server:
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.

B.6.3 UE capability exchange with UE capability version

Figure B.6.3-1 shows the UE capability exchange using OPTIONS when a IM session is already in progress between CUA#1 and CUA#2. In this figure, CUA#2 received UE capability version, which means CUA#1's capabilities have been significantly updated.

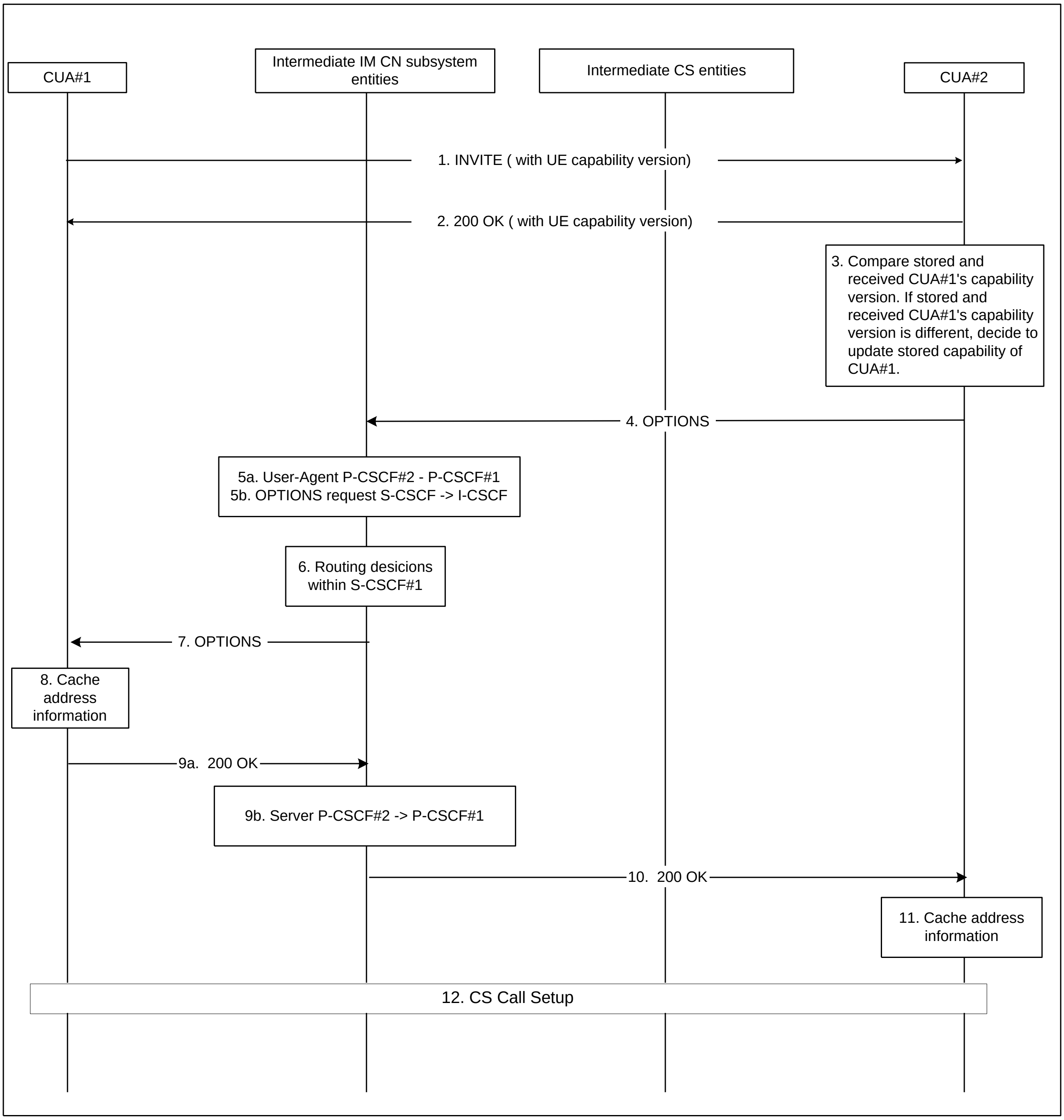

Figure B.6.3-1: UE capability exchange after receiving UE capability version during a IM session setup

The difference between IM session setup with no UE capability version (Figure B.6.2-1) and IM session setup with UE capability version (Figure B.6.3-1) of UE capability exchange is that the frequency of SIP OPTIONS transaction is reduced in case of with UE capability version (Figure B.6.3-1).

1. INVITE request (CUA#1 to CUA#2)

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

2. 200 OK response (CUA#1 to CUA#2)

Same as steps for 200 OK when IM session is established, see TS 24.228.

3. CUA#2 decide whether to send SIP 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.

4 to 7. SIP OPTIONS request (CUA#2 to CUA#1)

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

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

Since a IM session is ongoing, use the information in the SIP OPTIONS response to indicate to the user which capabilities are available during this IM session. Also, store the address information obtained on terminating UE's capabilities for later use.

9. to 10. SIP OPTIONS response (CUA#1 to CUA#2)

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

11. Store information on UE#1's capabilities

Since a IM session is ongoing, use the information in the SIP OPTIONS response to indicate to the user which capabilities are available during this IM session. Also, store the capability information obtained on UE#1's capabilities for later use.

12. CS call establishment

Either UE initiates the setup of a CS call between CUA#1 and CUA#2, as described in subclause B.5.

B.7 Signalling flows demonstrating UE capability exchange when a CS call is already in progress

B.7.1 Introduction

This subclause provides signalling flows for UE capability exchange when a CS call is already established.

B.7.2 UE capability exchange with no UE capability version

Figure B.7.2-1 shows the UE capability exchange using OPTIONS when a CS call is already in progress between CUA#1 and CUA#2.

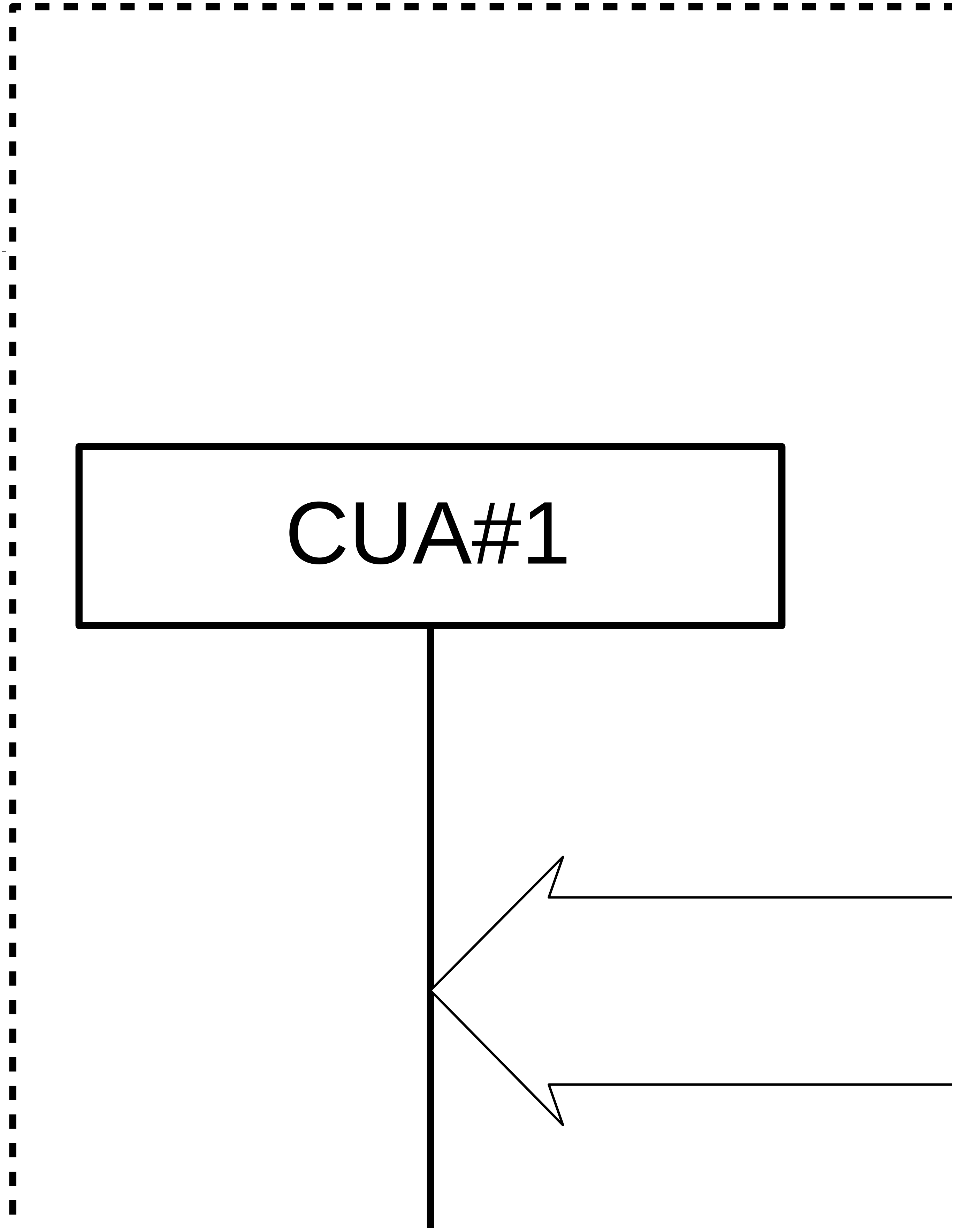

Figure B.7.2-1: UE capability exchange using OPTIONS after a CS call is established

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).

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.

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.

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 on CUA#2's capabilities and display capabilities available during this CS call**

   Since a CS call is ongoing, use the information in the OPTIONS response to indicate to the user which capabilities are available during this CS call. Also, cache the information obtained on CUA#2's capabilities for later use.

6. to 8. **OPTIONS response (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 on UE#1's capabilities and display capabilities available during this CS call**

   Since a CS call is ongoing, use the information in the OPTIONS response 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.

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 connected number received in the Connect 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 on CUA#2's capabilities and display capabilities available during this CS call**

   Since a CS call is ongoing, use the information in the OPTIONS response to indicate to the user which capabilities are available during this CS call. Also, cache the information obtained on CUA#2's capabilities for later use.

15. to 16. **OPTIONS response (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 OPTIONS response 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.7.3 UE capability exchange with UE capability version

Figure B.7.3-1 shows the UE capability exchange using OPTIONS when a CS call is already in progress between CUA#1 and CUA#2. In this figure, CUA#2 received UE capability version which means CUA#1's capabilities have been significantly updated.

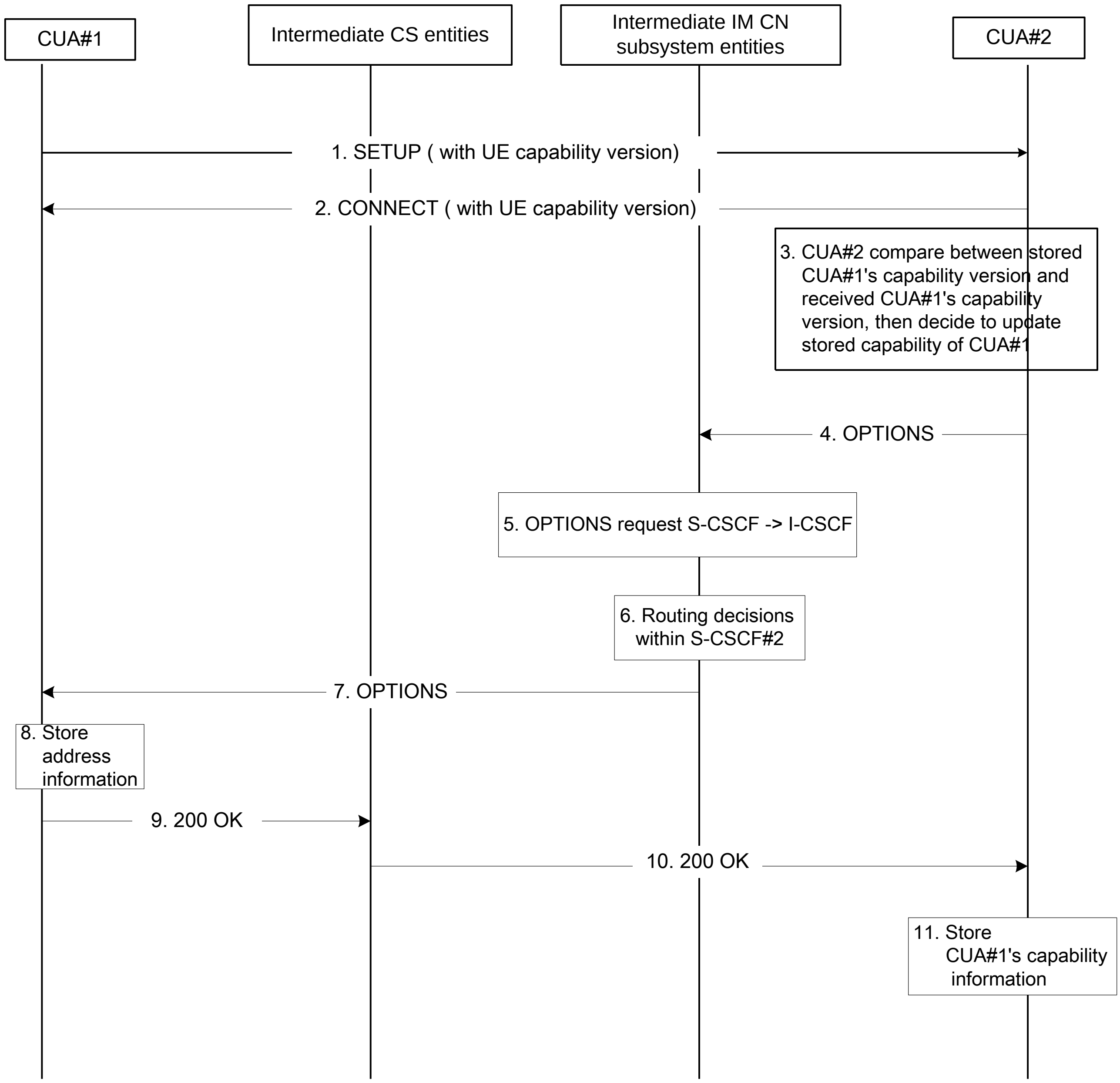

Figure B.7.3-1: UE capability exchange using SIP OPTIONS after received UE capability version during a CS call setup

The difference between CS call setup with no UE capability version (Figure B.7.2-1) and CS call setup with UE capability version (Figure B.7.3-1) of UE capability exchange is that the frequency of SIP OPTIONS transaction is reduced in case of with UE capability version (Figure B.7.3-1).

1. **CS call SETUP**

   CUA#1 send CS call SETUP message to CUA#2. UE capability version field is set differently from previously stored UE capability version, because CUA#1 has updated its capability.

2. **CS call CONNECT**

   CUA#2 send CS call CONNECT message to CUA#1 to establish CS bearer.

3. to 4. **CUA#2 decide whether to send SIP OPTIONS request according to UE capability version**

   CUA#2 decide whether to send SIP OPTIONS request to CUA#1 according to UE capability version received during the CS call 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 send SIP OPTIONS request to CUA#1 to get CUA#1's updated capability information.

5. to 7. **SIP OPTIONS request (CUA#2 to CUA#1)**

Same as steps for SIP OPTIONS when no CS call is established, see subclause B.6 steps 2. to 4.

8. **Store CUA#2's address information and display capabilities, which have been updated available during this CS call**

Since a CS call is ongoing, use the information in the SIP OPTIONS response to indicate to the user which capabilities are available during this CS call. Also, store the address information obtained on UE#2's capabilities for later use.

9. to 10. **SIP OPTIONS response (CUA#1 to CUA#2)**

Same as steps for SIP OPTIONS when no CS call is established, see subclause B.6 steps 5. to 7.

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

Since a CS call is ongoing, use the information in the SIP OPTIONS response to indicate to the user which capabilities are available during this CS call. Also, store the capability information obtained on UE#1's capabilities for later use.

Annex C (informative): Proposed changes to 3GPP TS 24.008 [4]

Editor's Note: This annex captures all possible generic changes to 3GPP TS 24.008 [4]. It will be deleted once the information captured here is moved to 3GPP TS 24.008 [4].

2 References

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

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

[1] Void.

[2] Void.

[2a] 3GPP TR 21.905 "Vocabulary for 3GPP Specifications"

[3] 3GPP TS 22.002: "Circuit Bearer Services (BS) supported by a Public Land Mobile Network (PLMN)".

[4] 3GPP TS 22.003: "Teleservices supported by a Public Land Mobile Network (PLMN)".

[5] 3GPP TS 42.009: "Security aspects".

[5a] 3GPP TS 33.102: "3G security; Security architecture".

[6] 3GPP TS 22.011: "Service accessibility".

[7] 3GPP TS 42.017: "Subscriber Identity Modules (SIM); Functional characteristics".

[8] 3GPP TS 22.101: "Service aspects; Service principles".

[8a] 3GPP TS 22.001: "Principles of circuit telecommunication services supported by a Public Land Mobile Network (PLMN)".

[8b] 3GPP TS 23.038: "Alphabets and language-specific information".

[9] 3GPP TS 23.101: "General UMTS Architecture".

[9a] 3GPP TS 23.108: "Mobile radio interface layer 3 specification core network protocols; Stage 2 (structured procedures)".

[10] 3GPP TS 23.003: "Numbering, addressing and identification".

[11] 3GPP TS 43.013: "Discontinuous Reception (DRX) in the GSM system".

[12] 3GPP TS 23.014: "Support of Dual Tone Multi-Frequency (DTMF) signalling".

[12a] ETSI ES 201 235-2, v1.2.1: "Specification of Dual Tone Multi-Frequency (DTMF); Transmitters and Receivers; Part 2: Transmitters".

[13] 3GPP TS 43.020: "Security-related network functions".

[14] 3GPP TS 23.122: "Non-Access-Stratum functions related to Mobile Station (MS) in idle mode".

[15] 3GPP TS 24.002: "GSM-UMTS Public Land Mobile Network (PLMN) access reference configuration".

[16] 3GPP TS 44.003: "Mobile Station - Base Station System (MS - BSS) interface; Channel structures and access capabilities".

[17] 3GPP TS 44.004: "Layer 1; General requirements".

[18] 3GPP TS 44.005: "Data Link (DL) layer; General aspects".

[19] 3GPP TS 44.006: "Mobile Station - Base Station System (MS - BSS) interface; Data Link (DL) layer specification".

[19a] 3GPP TS 25.321: "Medium Access Control (MAC) protocol specification".

[19b] 3GPP TS 25.322: "Radio Link Control (RLC) protocol specification".

[19c] 3GPP TS 25.413: "UTRAN Iu interface RANAP signalling".

[20] 3GPP TS 24.007: "Mobile radio interface signalling layer 3; General aspects".

[21] 3GPP TS 24.010: "Mobile radio interface layer 3; Supplementary services specification; General aspects".

[22] 3GPP TS 24.011: "Point-to-Point (PP) Short Message Service (SMS) support on mobile radio interface".

[23] 3GPP TS 24.012: "Short Message Service Cell Broadcast (SMSCB) support on the mobile radio interface".

[23a] 3GPP TS 44.071: "Location Services (LCS); Mobile radio interface layer 3 specification."

[23b] 3GPP TS 44.031 "Location Services LCS); Mobile Station (MS) - Serving Mobile Location Centre (SMLC); Radio Resource LCS Protocol (RRLP)".

[23c] 3GPP TS 25.331: "Radio Resource Control (RRC) protocol specification"

[24] 3GPP TS 24.080: "Mobile radio Layer 3 supplementary service specification; Formats and coding".

[25] 3GPP TS 24.081: "Line identification supplementary services; Stage 3".

[26] 3GPP TS 24.082: "Call Forwarding (CF) supplementary services; Stage 3".

[27] 3GPP TS 24.083: "Call Waiting (CW) and Call Hold (HOLD) supplementary services; Stage 3".

[28] 3GPP TS 24.084: "MultiParty (MPTY) supplementary services; Stage 3".

[29] 3GPP TS 24.085: "Closed User Group (CUG) supplementary services; Stage 3".

[30] 3GPP TS 24.086: "Advice of Charge (AoC) supplementary services; Stage 3".

[31] 3GPP TS 24.088: "Call Barring (CB) supplementary services; Stage 3".

[32] 3GPP TS 45.002: "Multiplexing and multiple access on the radio path".

[33] 3GPP TS 45.005: "Radio transmission and reception".

[34] 3GPP TS 45.008: "Radio subsystem link control".

[35] 3GPP TS 45.010: "Radio subsystem synchronization".

[36] 3GPP TS 27.001: "General on Terminal Adaptation Functions (TAF) for Mobile Stations (MS)".

[36a] 3GPP TS 27.060: "Mobile Station (MS) supporting Packet Switched Services ".

[37] 3GPP TS 29.002: "Mobile Application Part (MAP) specification".

[38] 3GPP TS 29.007: "General requirements on interworking between the Public Land Mobile Network (PLMN) and the Integrated Services Digital Network (ISDN) or Public Switched Telephone Network (PSTN)".

[39] 3GPP TS 51.010: "Mobile Station (MS) conformance specification".

[40] 3GPP TS 51.021: "GSM radio aspects base station system equipment specification".

[41] ISO/IEC 646 (1991): "Information technology - ISO 7-bit coded character set for information interchange".

[42] ISO/IEC 6429: "Information technology - Control functions for coded character sets".

[43] ISO 8348 (1987): "Information technology -- Open Systems Interconnection -- Network Service Definition".

[44] ITU-T Recommendation E.163: "Numbering plan for the international telephone service".

[45] ITU-T Recommendation E.164: "The international public telecommunication numbering plan".

[46] ITU-T Recommendation E.212: "The international identification plan for mobile terminals and mobile users".

[47] ITU-T Recommendation F.69 (1993): "The international telex service - Service and operational provisions of telex destination codes and telex network identification codes".

[48] ITU-T Recommendation I.330: "ISDN numbering and addressing principles".

[49] ITU-T Recommendation I.440 (1989): "ISDN user-network interface data link layer - General aspects".

[50] ITU-T Recommendation I.450 (1989): "ISDN user-network interface layer 3 General aspects".

[51] ITU-T Recommendation I.500 (1993): "General structure of the ISDN interworking recommendations".

[52] ITU-T Recommendation T.50: "International Reference Alphabet (IRA) (Formerly International Alphabet No. 5 or IA5) - Information technology - 7-bit coded character set for information interchange".

[53] ITU Recommendation Q.931: ISDN user-network interface layer 3 specification for basic control".

[54] ITU-T Recommendation V.21: "300 bits per second duplex modem standardized for use in the general switched telephone network".

[55] ITU-T Recommendation V.22: "1200 bits per second duplex modem standardized for use in the general switched telephone network and on point-to-point 2-wire leased telephone-type circuits".

[56] ITU-T Recommendation V.22bis: "2400 bits per second duplex modem using the frequency division technique standardized for use on the general switched telephone network and on point-to-point 2-wire leased telephone-type circuits".

[57] Void.

[58] ITU-T Recommendation V.26ter: "2400 bits per second duplex modem using the echo cancellation technique standardized for use on the general switched telephone network and on point-to-point 2-wire leased telephone-type circuits".

[59] ITU-T Recommendation V.32: "A family of 2-wire, duplex modems operating at data signalling rates of up to 9600 bit/s for use on the general switched telephone network and on leased telephone-type circuits".

[60] ITU-T Recommendation V.110: "Support by an ISDN of data terminal equipments with V-Series type interfaces".

[61] ITU-T Recommendation V.120: "Support by an ISDN of data terminal equipment with V-Series type interfaces with provision for statistical multiplexing".

[62] ITU-T Recommendation X.21: "Interface between Data Terminal Equipment (DTE) and Data Circuit-terminating Equipment (DCE) for synchronous operation on public data networks".

[63] Void.

[64] Void.

[65] ITU-T Recommendation X.30: "Support of X.21, X.21 bis and X.20 bis based Data Terminal Equipments (DTEs) by an Integrated Services Digital Network (ISDN)".

[66] ITU-T Recommendation X.31: "Support of packet mode terminal equipment by an ISDN".

[67] Void.

[68] Void.

[69] ITU-T Recommendation X.121: "International numbering plan for public data networks".

[70] ETSI ETS 300 102-1: "Integrated Services Digital Network (ISDN); User-network interface layer 3; Specifications for basic call control".

[71] ETSI ETS 300 102-2: "Integrated Services Digital Network (ISDN); User-network interface layer 3; Specifications for basic call control; Specification Description Language (SDL) diagrams".

[72] ISO/IEC 10646: "Information technology -- Universal Multiple-Octet Coded Character Set (UCS)".

[73] 3GPP TS 22.060: "General Packet Radio Service (GPRS); Service Description; Stage 1".

[74] 3GPP TS 23.060: "General Packet Radio Service (GPRS); Service Description; Stage 2".

[75] 3GPP TS 43.064: "General Packet Radio Service (GPRS); Overall description of the GPRS radio interface; Stage 2".

[76] 3GPP TS 44.060: "General Packet Radio Service (GPRS); Mobile Station (MS) - Base Station System (BSS) interface; Radio Link Control/Medium Access Control (RLC/MAC) protocol".

[77] IETF RFC 1034: "Domain names - concepts and facilities".

[78] 3GPP TS 44.065: "Mobile Station (MS) - Serving GPRS Support Node (SGSN); Subnetwork Dependent Convergence Protocol (SNDCP)".

[78a] 3GPP TS 44.064: "Mobile Station - Serving GPRS Support Node (MS-SGSN) Logical Link Control (LLC) Layer Specification".

[79] ITU Recommendation I.460: "Multiplexing, rate adaption and support of existing interfaces".

[80] 3GPP TS 26.111: "Codec for Circuit Switched Multimedia Telephony Service; Modifications to H.324".

[81] 3GPP TS 23.107: "Quality of Service (QoS) concept and architecture".

[82] 3GPP TS 43.022: "Functions related to Mobile Station (MS) in idle mode and group receive mode".

[83] 3GPP TS 26.103: "Speech Codec List for GSM and UMTS".

[84] 3GPP TS 44.018: "Mobile radio interface layer 3 specification, Radio Resource Control Protocol".

[85] 3GPP TS 48.008: "Mobile-services Switching Centre – Base Station System (MSC – BSS) interface; layer 3 specification".

[86] 3GPP TS 48.018: "General Packet Radio Service (GPRS); Base Station System (BSS) - Serving GPRS Support Node (SGSN); BSS GPRS Protocol (BSSGP)".

[87] 3GPP TS 43.055: "Dual Transfer Mode (DTM); Stage 2".

[88] 3GPP TS 23.067: "enhanced Multi-Level Precedence and Pre-emption service (eMLPP); Stage 2".

[88a] 3GPP TS 23.093: "Technical realization of Completion of Calls to Busy Subscriber (CCBS); Stage 2".

[89] 3GPP TS 22.042: "Network Identity and Time Zone (NITZ), Stage 1".

[90] 3GPP TS 23.040: "Technical realization of Short Message Service (SMS)".

[91] 3GPP TS 44.056: "GSM Cordless Telephony System (CTS), (Phase 1) CTS Radio Interface Layer 3 Specification".

[92] 3GPP TS 23.226: "Global Text Telephony; Stage 2 "

[93] 3GPP TS 26.226: "Cellular Text Telephone Modem (CTM), General Description "

[94] 3GPP TS 23.236: "Intra Domain Connection of RAN Nodes to Multiple CN Nodes"

[95] 3GPP TS 24.229: "IP Multimedia Call Control Protocol based on SIP and SDP"

[96] 3GPP TS 23.205: "Bearer-independent circuit-switched core network; Stage 2".

[97] 3GPP TS 23.172: "UDI/RDI Fallback and Service Modification; Stage 2".

[98] 3GPP TS 25.304: "UE Procedures in Idle Mode and Procedures for Cell Reselection in Connected Mode"

[99] RFC 3513 (April 2003): "Internet Protocol Version 6 (IPv6) Addressing Architecture".

[100] 3GPP TS 29.207: "Policy control over Go interface".

[101] 3GPP TS 21.111: "USIM and IC card requirements".

[102] RFC 1661 (July 1994): "The Point-to-Point Protocol (PPP)".

[103] RFC 3232 (January 2002): "Assigned Numbers: RFC 1700 is Replaced by an On-line Database".

[104] 3GPP TS 23.034: "High Speed Circuit Switched Data (HSCSD) – Stage 2".

[105] 3GPP TS 23.271: "Functional stage 2 description of LCS".

[106] 3GPP TS 23.246: "Multimedia Broadcast/Multicast Service (MBMS); Architecture and Functional Description".

[107] RFC 3376 (October 2002): "Internet Group Management Protocol, Version 3".

[108] RFC 2710 (October 1999): "Multicast Listener Discovery (MLD) for IPv6".

[109] 3GPP TS 23.251: "Network Sharing; Architecture and Functional Description".

[110] 3GPP TS 25.346: "Introduction of the Multimedia Broadcast Multicast Service (MBMS) in the Radio Access Network"

[111] 3GPP TS 44.118: "Radio Resource Control (RRC) protocol; Iu mode".

[112] 3GPP TS 31.102: "Characteristics of the USIM Application".

[113] 3GPP TS 43.129: "Packet-switched handover for GERAN A/Gb mode; Stage 2".

[114] 3GPP TS 23.009: "Handover procedures".

[115] 3GPP TR 23.903: " Redial solution for voice-video switching".

[x] 3GPP TS 24.279: "Combining Circuit Switched (CS) and IP Multimedia Subsystem (IMS) services, stage 3"

10.5.4.25 User-user

The purpose of the user-user information element is to convey information between the mobile station and the remote ISDN user.

The user-user information element is coded as shown in figure 10.5.114/3GPP TS 24.008 and table 10.5.131/ 3GPP TS 24.008. There are no restrictions on the content of the user-user information field.

The user-user is a type 4 information element with a minimum length of 3 octets and a maximum length of either 35 or 131 octets. In the SETUP message the user-user information element has a maximum size of 35 octets in a GSM PLMN. In the USER INFORMATION, ALERTING, CONNECT, DISCONNECT, PROGRESS, RELEASE and RELEASE COMPLETE messages the user-user information element has a maximum size of 131 octets in a GSM PLMN.

In other networks than GSM PLMNs the maximum size of the user-user information element is 35 or 131 octets in the messages mentioned above. The evolution to a single maximum value is the long term objective; the exact maximum value is the subject of further study.

NOTE: The user-user information element is transported transparently through a GSM PLMN.

8	7	6	5	4	3	2	1	
	User-user IEI							octet 1
Length of user-user contents								octet 2
User-user protocol discriminator								octet 3
User-user information								octet 4*
								octet N*

Figure 10.5.114/3GPP TS 24.008 User-user information element

Table 10.5.131/3GPP TS 24.008: User-user information element

User-user protocol discriminator (octet 3)								
Bits								
8	**7**	**6**	**5**	**4**	**3**	**2**	**1**	
0	0	0	0	0	0	0	0	User specific protocol (NOTE 1)
0	0	0	0	0	0	0	1	OSI high layer protocols
0	0	0	0	0	0	1	0	X.244 (NOTE 2)
0	0	0	0	0	0	1	1	Reserved for system management convergence function
0	0	0	0	0	1	0	0	IA5 characters (NOTE 3)
0	0	0	0	0	1	1	1	Rec.V.120 rate adaption
0	0	0	0	1	0	0	0	Q.931 (I.451) user-network call control messages
0	0	0	1	0	0	0	0	Reserved for other network layer or
through								layer 3 protocols
0	0	1	1	1	1	1	1	
0	1	0	0	0	0	0	0	
through								National use
0	1	0	0	1	1	1	0	
0	1	0	0	1	1	1	1	3GPP capability exchange protocol (NOTE 4)
0	1	0	1	0	0	0	0	Reserved for other network
through								layer or layer 3 protocols
1	1	1	1	1	1	1	0	
All other values are reserved.								
NOTE 1: The user information is structured according to user needs.								
NOTE 2: The user information is structured according to Rec.X.244 which specifies the structure of X.25 call user data.								
NOTE 3: The user information consists of IA5 characters.								
NOTE 4: When the user-user protocol discriminator is set to "3GPP capability exchange protocol", the user-user information is coded according to 3GPP TS 24.279[x].								

Editor's Note: The point code chosen for the 3GPP capability exchange protocol identifier must be checked to ensure that it is not used by any other protocol.

Annex X (normative):
3GPP capability exchange protocol

Editor’s Note: This annex is still part of Annex C of 3GPP TR 24.879.

X.1 Scope

This annex specifies the protocol data units used by the 3GPP capability exchange protocol and procedures for the handling of unknown, unforeseen, and erroneous protocol data by the receiving UE.

The 3GPP capability exchange protocol provides services for the end-to-end exchange of capabilities between UEs. It is a separate protocol which uses the user-to-user signalling service 1 of the layer 3 call control protocol as a means of transport.

Functional procedures which use the 3GPP capability exchange protocol in the context of CSI are specified in 3GPP TS 24.279 [X].

X.2 User-user protocol contents

The user-user protocol contents is included in the user-user information element described in subclause 10.5.4.25.

The user-user protocol contents is structured like the non-imperative part of a standard L3 message (see 3GPP TS 24.007 [12], subclause 11.2) and is composed of a variable number of information elements of type 1, 2, 3 and 4. The different formats (TV, TLV) and the categories of information elements (type 1, 2, 3 and 4) are defined in 3GPP TS 24.007 [12].

Within the user-user protocol contents the information elements may occur in an arbitrary order.

All information elements shall be included only once.

8	7	6	5	4	3	2	1	
Information element 1								octet 4* octet 5* octet k*
Information element 2								octet k+1* octet l*
…								octet l+1* octet m*
Information element K								octet m+1* octet n*

Figure X.1/3GPP TS 24.008 User-user information when the user-user protocol indicator is set to "3GPP capability exchange protocol"

X.3 Information element identifier

The information element identifier and its use are defined in 3GPP TS 24.007 [12].

For the information elements defined in subclause X.4, the coding of the information element identifier bits is defined in table X.2/3GPP TS 24.008.

For a method to determine from the information element identifier whether an unknown information element is of type 1 or 2 (i.e. it is an information element of one octet length) or type 4 (i.e. the next octet is the length indicator indicating the length of the remaining of the information element) see 3GPP TS 24.007 [12], subclause 11.2.4.

Table X.1/3GPP TS 24.008: Information element identifier coding for user-user protocol information elements

8	7	6	5	4	3	2	1		Reference clause
								Type 1 information elements:	
1	0	0	0	-	-	-	-	Radio environment capability	X.4.2
								Type 2 information elements:	
								Type 3 and 4 information elements:	
0	0	0	1	0	0	0	1	Personal ME identifier	X.4.1
0	0	1	0	0	0	0	0	UE capability version	X.4.3
	All other values are unused								

X.4 Information elements

X.4.1 Personal ME identifier

The purpose of the *personal ME identifier* is to discriminate between MEs used by the same user (see TS 24.279 [X], subclause 4.2).

NOTE: As the personal ME identifier is generated randomly, it is not guaranteed that it uniquely identifies a specific ME used by the same user.

The *personal ME identifier* has the form PMI-XXXX, where XXXX is a 4-digit hexadecimal number. Only the hexadecimal number XXXX is coded in the *personal ME identifier* information element.

The *personal ME identifier* information element is coded as shown in figure X.2/3GPP TS 24.008 and table X.2/3GPP TS 24.008.

The *personal ME identifier* is a type 3 information element with 3 octets length.

8	7	6	5	4	3	2	1	
Personal ME identifier IEI								octet 1
ME identifier digit 2				ME identifier digit 1				octet 2
ME identifier digit 4				ME identifier digit 3				octet 3

Figure X.2/3GPP TS 24.008 Personal ME identifier

Table X.2/3GPP TS 24.008: Personal ME identifier

ME identifier digits (octets 2, 3)					
Bits 1 to 4 or bits 5 to 8, respectively, contain the binary encoding of a hexadecimal ME identifier digit. Digit 1 is the leftmost digit in the 4-digit hexadecimal number XXXX.					
Bits					ME identifier digit value
4	**3**	**2**	**1**	Or	
8	**7**	**6**	**5**		
0	0	0	0		0
0	0	0	1		1
0	0	1	0		2
0	0	1	1		3
0	1	0	0		4
0	1	0	1		5
0	1	1	0		6
0	1	1	1		7
1	0	0	0		8
1	0	0	1		9
1	0	1	0		A
1	0	1	1		B
1	1	0	0		C
1	1	0	1		D
1	1	1	0		E
1	1	1	1		F

X.4.2 Radio environment capability

The purpose of the *radio environment capability* is to provide information about the current radio environment of the UE.

The *radio environment capability* information element is coded as shown in figure X.3/3GPP TS 24.008 and table X.3/3GPP TS 24.008.

The *radio environment capability* is a type 1 information element with 1 octet length.

8	7	6	5	4	3	2	1	
Radio environment capability IEI				0 spare	0 spare	0 Spare	CS/ PS	octet 1

Figure X.3/3GPP TS 24.008 Radio environment capability contents

Table X.3/3GPP TS 24.008: Radio environment capability contents

CS and PS capability (octet 1, bit 1)	
The CS and PS capability indicates whether the MS is in a radio environment that supports simultaneous use of CS and PS services (see 3GPP TS 24.279 [X]).	
0	simultaneous use of CS and PS services not supported
1	simultaneous use of CS and PS services supported
Bits 2 to 4 of octet 1 are spare and shall be coded as zero.	

X.4.3 UE capability version

The purpose of the *UE capability version* is to inform the receiving UE that the capability of the sending UE has changed since the last UE capability exchange (see 3GPP TS 24.279 [X]).

The UE capability version has the form UCV-XX, where XX is a 2-digit hexadecimal number. Only the hexadecimal number XX is coded in the UE capability version information element.

The *UE capability version* information element is coded as shown in figure X.4/3GPP TS 24.008 and table X.4/3GPP TS 24.008.

The *UE capability update indication* is a type 3 information element with 2 octets length.

8	7	6	5	4	3	2	1	
UE capability version IEI								octet 1
UE capability version digit 2				UE capability version digit 1				octet 2

Figure X.4/3GPP TS 24.008: UE capability version

Table X.4/3GPP TS 24.008: UE capability version

UE capability version digits (octet 2)

Bits 1 to 4 and bits 5 to 8, respectively, contain the binary encoding of a 2-digit hexadecimal UE capability version. Digit 1 is the leftmost digit.

Bits					UE capability version digit value
4	**3**	**2**	**1**	Or	
8	**7**	**6**	**5**		
0	0	0	0		0
0	0	0	1		1
0	0	1	0		2
0	0	1	1		3
0	1	0	0		4
0	1	0	1		5
0	1	1	0		6
0	1	1	1		7
1	0	0	0		8
1	0	0	1		9
1	0	1	0		A
1	0	1	1		B
1	1	0	0		C
1	1	0	1		D
1	1	1	0		E
1	1	1	1		F

X.5 Handling of unknown, unforeseen, and erroneous protocol data

X.5.1 General

The following subclauses specifies procedures for the handling of unknown, unforeseen, and erroneous protocol data by the receiving UE. These procedures are called "error handling procedures", but in addition to providing recovery mechanisms for error situations they define a compatibility mechanism for future extensions of the protocols.

Subclauses X.5.2 to X.5.5 shall be applied in order of precedence.

For the definition of semantical and syntactical errors see 3GPP TS 24.007 [12], subclause 11.4.2.

Where the description of information elements in the present document contains bits defined to be "spare bits", these bits shall set to the indicated value (usually 0) by the sending side, and their value shall be ignored by the receiving side.

X.5.2 Not supported IEs, unknown IEIs

The UE shall ignore all information elements which are not supported and all information elements with unknown IEI.

X.5.3 Repeated IEs

If an information element, for which repetition is not specified in subclause X.2, is repeated in the user-user protocol contents, only the contents of the information element appearing first shall be handled and all subsequent repetitions of the information element shall be ignored. When repetition of information elements is specified, only the contents of specified repeated information elements shall be handled. If the limit on repetition of information elements is exceeded, the contents of information elements appearing first up to the limit of repetitions shall be handled and all subsequent repetitions of the information element shall be ignored.

X.5.4 Syntactically incorrect IEs

The UE shall treat all IEs that are syntactically incorrect as not present in the user-user protocol contents.

X.5.5 Semantically incorrect IEs

When an IE with semantically incorrect contents is received, the foreseen reactions specified for the respective procedure are performed (e.g. in the context of CSI see 3GPP TS 24.279 [X], clauses 5, 6). If however no such reactions are specified, the UE shall ignore the IE.

Annex D (informative): Proposed changes to 3GPP TS 24.229 [5]

Editor's Note: This annex captures all possible generic changes to 3GPP TS 24.229 [5]. It will be deleted once the information captured here is moved to 3GPP TS 24.229 [5]

A.1.3 Roles

Table A.2: Roles

Item	Roles	Reference	RFC status	Profile status
1	User agent	[26]	o.1	o.1
2	Proxy	[26]	o.1	o.1
o.1: It is mandatory to support exactly one of these items.				
NOTE: For the purposes of the present document it has been chosen to keep the specification simple by the tables specifying only one role at a time. This does not preclude implementations providing two roles, but an entirely separate assessment of the tables shall be made for each role.				

Table A.3: Roles specific to this profile

Item	Roles	Reference	RFC status	Profile status
1	UE	5.1	n/a	o.1
2	P-CSCF	5.2	n/a	o.1
3	I-CSCF	5.3	n/a	o.1
3A	I-CSCF (THIG)	5.3	n/a	c1
4	S-CSCF	5.4	n/a	o.1
5	BGCF	5.6	n/a	o.1
6	MGCF	5.5	n/a	o.1
7	AS	5.7	n/a	o.1
7A	AS acting as terminating UA, or redirect server	5.7.2	n/a	c2
7B	AS acting as originating UA	5.7.3	n/a	c2
7C	AS acting as a SIP proxy	5.7.4	n/a	c2
7D	AS performing 3rd party call control	5.7.5	n/a	c2
8	MRFC	5.8	n/a	o.1
9	IMS-ALG	5.9	n/a	o.1

c1: IF A.3/3 THEN o ELSE x - - I-CSCF.
c2: IF A.3/7 THEN o.2 ELSE n/a - - AS.
o.1: It is mandatory to support exactly one of these items.
o.2: It is mandatory to support at least one of these items.

NOTE: For the purposes of the present document it has been chosen to keep the specification simple by the tables specifying only one role at a time. This does not preclude implementations providing two roles, but an entirely separate assessment of the tables shall be made for each role.

Table A.3A: Roles specific to additional capabilities

Item	Roles	Reference	RFC status	Profile status
1	Presence server	3GPP TS 24.141 [8A]	n/a	c1
2	Presence user agent	3GPP TS 24.141 [8A]	n/a	c2
3	Resource list server	3GPP TS 24.141 [8A]	n/a	c3
4	Watcher	3GPP TS 24.141 [8A]	n/a	c4
11	Conference focus	3GPP TS 24.147 [8B]	n/a	c5
12	Conference participant	3GPP TS 24.147 [8B]	n/a	c6
21	CSI User Agent	3GPP TS 24.279 [8D]	n/a	c7

c1: IF A.3/7A AND A.3/7B THEN o ELSE n/a - - AS acting as terminating UA, or redirect server and AS acting as originating UA.
c2: IF A.3/1 THEN o ELSE n/a - - UE.
c3: IF A.3/7A THEN o ELSE n/a - - AS acting as terminating UA, or redirect server.
c4: IF A.3/1 OR A.3/7B THEN o ELSE IF A.3/9 THEN m ELSE n/a - - UE or AS acting as originating UA.
c5: IF A.3/7D AND A.3/4 AND A.3/8 THEN o ELSE n/a - - AS performing 3rd party call control and S-CSCF and MRFC (note 2).
c6: IF A.3/1 OR A.3A/11 THEN o ELSE IF A.3/9 THEN m ELSE n/a - - UE or conference focus.
c7: IF A.3/1 THEN o ELSE n/a – UE.

NOTE 1: For the purposes of the present document it has been chosen to keep the specification simple by the tables specifying only one role at a time. This does not preclude implementations providing two roles, but an entirely separate assessment of the tables shall be made for each role.
NOTE 2: The functional split between the MRFC and the conferencing AS is out of scope of this document and they are assumed to be collocated.

A.2 Profile definition for the Session Initiation Protocol as used in the present document

A.2.1 User agent role

A.2.1.1 Introduction

This subclause contains the ICS proforma tables related to the user role. They need to be completed only for UA implementations:

Prerequisite: A.2/1 - - user agent role.

A.2.1.2 Major capabilities

Table A.4: Major capabilities

Item	Does the implementation support	Reference	RFC status	Profile status
	Capabilities within main protocol			
1	client behaviour for registration?	[26] subclause 10.2	o	c3
2	registrar?	[26] subclause 10.3	o	c4
2A	registration of multiple contacts for a single address of record	[26] 10.2.1.2, 16.6	o	o
2B	initiating a session?	[26] subclause 13	o	o
3	client behaviour for INVITE requests?	[26] subclause 13.2	c18	c18
4	server behaviour for INVITE requests?	[26] subclause 13.3	c18	c18
5	session release?	[26] subclause 15.1	c18	c18
6	timestamping of requests?	[26] subclause 8.2.6.1	o	o
7	authentication between UA and UA?	[26] subclause 22.2	c34	c34
8	authentication between UA and registrar?	[26] subclause 22.2	o	n/a
8A	authentication between UA and proxy?	[26] 20.28, 22.3	o	o
9	server handling of merged requests due to forking?	[26] 8.2.2.2	m	m
10	client handling of multiple responses due to forking?	[26] 13.2.2.4	m	m
11	insertion of date in requests and responses?	[26] subclause 20.17	o	o
12	downloading of alerting information?	[26] subclause 20.4	o	o
	Extensions			
13	the SIP INFO method?	[25]	o	n/a
14	reliability of provisional responses in SIP?	[27]	c19	c18
15	the REFER method?	[36]	o	c33
16	integration of resource management and SIP?	[30] [64]	c19	c18
17	the SIP UPDATE method?	[29]	c5	c18
19	SIP extensions for media authorization?	[31]	o	c14
20	SIP specific event notification?	[28]	o	c13
21	the use of NOTIFY to establish a dialog?	[28] 4.2	o	n/a
22	acting as the notifier of event information?	[28]	c2	c15
23	acting as the subscriber to event information?	[28]	c2	c16
24	session initiation protocol extension header field for registering non-adjacent contacts?	[35]	o	c6
25	private extensions to the Session Initiation Protocol (SIP) for network asserted identity within trusted networks?	[34]	o	m
26	a privacy mechanism for the Session Initiation Protocol (SIP)?	[33]	o	m
26A	request of privacy by the inclusion of a Privacy header indicating any privacy option?	[33]	c9	c11
26B	application of privacy based on the received Privacy header?	[33]	c9	n/a
26C	passing on of the Privacy header transparently?	[33]	c9	c12
26D	application of the privacy option "header" such that those headers which cannot be completely expunged of identifying information without the assistance of intermediaries are obscured?	[33] 5.1	c10	c27
26E	application of the privacy option "session" such that anonymization for	[33] 5.2	c10	c27

	the session(s) initiated by this message occurs?			
26F	application of the privacy option "user" such that user level privacy functions are provided by the network?	[33] 5.3	c10	c27
26G	application of the privacy option "id" such that privacy of the network asserted identity is provided by the network?	[34] 7	c10	n/a
26H	application of the privacy option "history" such that privacy of the History-Info header is provided by the network?	[66] 7.2	c37	c37
27	a messaging mechanism for the Session Initiation Protocol (SIP)?	[50]	o	c7
28	session initiation protocol extension header field for service route discovery during registration?	[38]	o	c17
29	compressing the session initiation protocol?	[55]	o	c8
30	private header extensions to the session initiation protocol for the 3rd-Generation Partnership Project (3GPP)?	[52]	o	m
31	the P-Associated-URI header extension?	[52] 4.1	c21	c22
32	the P-Called-Party-ID header extension?	[52] 4.2	c21	c23
33	the P-Visited-Network-ID header extension?	[52] 4.3	c21	c24
34	the P-Access-Network-Info header extension?	[52] 4.4	c21	c25
35	the P-Charging-Function-Addresses header extension?	[52] 4.5	c21	c26
36	the P-Charging-Vector header extension?	[52] 4.6	c21	c26
37	security mechanism agreement for the session initiation protocol?	[48]	o	c20
38	the Reason header field for the session initiation protocol?	[34A]	o	o (note 1)
39	an extension to the session initiation protocol for symmetric response routeing?	[56A]	o	x
40	caller preferences for the session initiation protocol?	[56B]	C29	c29
40A	the proxy-directive within caller-preferences?	[56B] 9.1	o.5	o.5
40B	the cancel-directive within caller-preferences?	[56B] 9.1	o.5	o.5
40C	the fork-directive within caller-preferences?	[56B] 9.1	o.5	c28
40D	the recurse-directive within caller-preferences?	[56B] 9.1	o.5	o.5
40E	the parallel-directive within caller-preferences?	[56B] 9.1	o.5	c28
40F	the queue-directive within caller-preferences?	[56B] 9.1	o.5	o.5
41	an event state publication extension to the session initiation protocol?	[70]	o	c30
42	SIP session timer?	[58]	c19	c19
43	the SIP Referred-By mechanism?	[59]	o	c33
44	the Session Inititation Protocol (SIP) "Replaces" header?	[60]	c19	c19 (note 1)
45	the Session Inititation Protocol (SIP) "Join" header?	[61]	c19	c19 (note 1)
46	the callee capabilities?	[62]	o	c35
47	an extension to the session initiation	[66]	o	o

	protocol for request history information?			

c2: IF A.4/20 THEN o.1 ELSE n/a - - SIP specific event notification extension.
c3: IF A.3/1 OR A.3/4 THEN m ELSE n/a - - UE or S-CSCF functional entity.
c4: IF A.3/4 THEN m ELSE IF A.3/7 THEN o ELSE n/a - - S-CSCF or AS functional entity.
c5: IF A.4/16 THEN m ELSE o - - integration of resource management and SIP extension.
c6: IF A.3/4 OR A.3/1 THEN m ELSE n/a. - - S-CSCF or UE.
c7: IF A.3/1 OR A.3/4 OR A.3/7A OR A.3/7B OR A.3/7D OR A.3/9 THEN m ELSE n/a - - UA or S-CSCF or AS acting as terminating UA or AS acting as originating UA or AS performing 3rd party call control or IMS-ALG.
c8: IF A.3/1 THEN m ELSE n/a - - UE behaviour.
c9: IF A.4/26 THEN o.2 ELSE n/a - - a privacy mechanism for the Session Initiation Protocol (SIP).
c10: IF A.4/26B THEN o.3 ELSE n/a - - application of privacy based on the received Privacy header.
c11: IF A.3/1 OR A.3/6 THEN o ELSE IF A.3/9 THEN m ELSE n/a - - UE or MGCF, IMS-ALG.
c12: IF A.3/7D THEN m ELSE n/a - - AS performing 3rd-party call control.
c13: IF A.3/1 OR A.3/2 OR A.3/4 OR A.3/9 THEN m ELSE o - - UE or S-CSCF or IMS-ALG.
c14: IF A.3/1 THEN m ELSE IF A.3/2 THEN o ELSE n/a – UE or P-CSCF.
c15: IF A.4/20 AND (A.3/4 OR A.3/9) THEN m ELSE o – SIP specific event notification extensions and S-CSCF, IMS-ALG.
c16: IF A.4/20 AND (A.3/1 OR A.3/2 OR A.3/9) THEN m ELSE o - - SIP specific event notification extension and UE or P-CSCF OR IMS-ALG.
c17: IF A.3/1 or A.3/4 THEN m ELSE n/a - - UE or S-CSCF.
c18: IF A.4/2B THEN m ELSE n/a - - initiating sessions.
c19: IF A.4/2B THEN o ELSE n/a - - initiating sessions.
c20: IF A.3/1 THEN m ELSE n/a - - UE behaviour.
c21: IF A.4/30 THEN o.4 ELSE n/a - - private header extensions to the session initiation protocol for the 3rd-Generation Partnership Project (3GPP).
c22: IF A.4/30 AND (A.3/1 OR A.3/4) THEN m ELSE n/a - - private header extensions to the session initiation protocol for the 3rd-Generation Partnership Project (3GPP) and S-CSCF or UA.
c23: IF A.4/30 AND A.3/1 THEN o ELSE n/a - - private header extensions to the session initiation protocol for the 3rd-Generation Partnership Project (3GPP) and UE.
c24: IF A.4/30 AND A.3/4) THEN m ELSE n/a - - private header extensions to the session initiation protocol for the 3rd-Generation Partnership Project (3GPP) and S-CSCF.
c25: IF A.4/30 AND (A.3/1 OR A.3/4 OR A.3/7A OR A.3/7D OR A.3/9) THEN m ELSE n/a - - private header extensions to the session initiation protocol for the 3rd-Generation Partnership Project (3GPP) and UE, S-CSCF or AS acting as terminating UA or AS acting as third-party call controller, IMS-ALG.
c26: IF A.4/30 AND (A.3/6 OR A.3/7A OR A.3/7B or A.3/7D) THEN m ELSE n/a - - private header extensions to the session initiation protocol for the 3rd-Generation Partnership Project (3GPP) and MGCF, AS acting as a terminating UA, or AS acting as an originating UA, or AS acting as third-party call controller.
c27: IF A.3/7D THEN o ELSE x - - AS performing 3rd party call control.
c28: IF A.3/1 THEN m ELSE o.5 - - UE.
c29: IF A.4/40A OR A.4/40B OR A.4/40C OR A.4/40D OR A.4/40E OR A.4/40F THEN m ELSE n/a - - support of any directives within caller preferences for the session initiation protocol.
c30: IF A.3A/1 OR A.3A/2 THEN m ELSE IF A.3/1 THEN o ELSE n/a - - presence server, presence user agent, UE, AS.
c33: IF A.3/11 OR A.3/12 OR A.3/9 OR A.4/44 THEN m ELSE o - - conference focus or conference participant or IMS-ALG or the Session Inititation Protocol (SIP) "Replaces" header.
c34: IF A.4/44 OR A.4/45 OR A.3/9 THEN m ELSE n/a - - the Session Inititation Protocol (SIP) "Replaces" header or the Session Inititation Protocol (SIP) "Join" header or IMS-ALG.
c35: IF A.3/4 OR A.3/9 OR A.3A/21 THEN m ELSE IF (A.3/1 OR A.3/6 OR A.3/7 OR A.3/8) THEN o ELSE n/a - - S-CSCF or IMS-ALG functional entities, UE or MGCF or AS or MRFC functional entity.
c37 IF A.4/47 THEN o.3 ELSE n/a - - an extension to the session initiation protocol for request history information.
o.1: At least one of these capabilities is supported.
o.2: At least one of these capabilities is supported.
o.3: At least one of these capabilities is supported.
o.4: At least one of these capabilities is supported.
o.5: At least one of these capabilities is supported.

NOTE 1: At the MGCF, the interworking specifications do not support a handling of the header associated with this extension.

Prerequisite A.5/20 - - SIP specific event notification

Table A.4A: Supported event packages

Item	Does the implementation support	Subscriber			Notifier		
		Ref.	RFC status	Profile status	Ref.	RFC status	Profile status
1	reg event package?	[43]	c1	c3	[43]	c2	c4
2	refer package?	[36] 3	c13	c13	[36] 3	c13	c13
3	presence package?	[74] 6	c1	c5	[74] 6	c2	c6
4	eventlist with underlying presence package?	[75], [74] 6	c1	c7	[75], [74] 6	c2	c8
5	presence.winfo template-package?	[72] 4	c1	c9	[72] 4	c2	c10
6	sip-profile package?	[77] 3	c1	c11	[77] 3	c2	c12
7	conference package?	[78] 3	c1	c21	[78] 3	c1	c22
8	message-summary package?	[65]	c1	c23	[65] 3	c2	c24

c1: IF A.4/23 THEN o ELSE n/a - - acting as the subscriber to event information.
c2: IF A.4/22 THEN o ELSE n/a - - acting as the notifier of event information.
c3: IF A.3/1 OR A.3/2 THEN m ELSE IF A.3/7 THEN o ELSE n/a - - UE, P-CSCF, AS.
c4: IF A.3/4 THEN m ELSE n/a - - S-CSCF.
c5: IF A.3A/3 OR A.3A/4 THEN m ELSE IF A.4/23 THEN o ELSE n/a - - resource list server or watcher, acting as the subscriber to event information.
c6: IF A.3A/1 THEN m ELSE IF A.4/22 THEN o ELSE n/a - - presence server, acting as the notifier of event information.
c7: IF A.3A/4 THEN m ELSE IF A.4/23 THEN o ELSE n/a - - watcher, acting as the subscriber to event information.
c8: IF A.3A/3 THEN m ELSE IF A.4/22 THEN o ELSE n/a - - resource list server, acting as the notifier of event information.
c9: IF A.3A/2 THEN m ELSE IF A.4/23 THEN o ELSE n/a - - presence user agent, acting as the subscriber to event information.
c10: IF A.3A/1 THEN m ELSE IF A.4/22 THEN o ELSE n/a - - presence server, acting as the notifier of event information.
c11: IF A.3A/2 OR A.3A/4 THEN o ELSE IF A.4/23 THEN o ELSE n/a - - presence user agent or watcher, acting as the subscriber to event information.
c12: IF A.3A/1 OR A.3A/3 THEN m ELSE IF A.4/22 THEN o ELSE n/a - - presence server or resource list server, acting as the notifier of event information.
c13: IF A.4/15 THEN m ELSE n/a - - the REFER method.
c21: IF A.3A/12 THEN m ELSE IF A.4/23 THEN o ELSE n/a - - conference participant or acting as the subscriber to event information.
c22: IF A.3A/11 THEN m ELSE IF A.4/22 THEN o ELSE n/a - - conference focus or acting as the notifier of event information.
c23: IF (A.3/1 OR A.3/7A OR A.3/7B) AND A.4/23 THEN o ELSE n/a - - UE, AS acting as terminating UA, or redirect server, AS acting as originating UA all as subscriber of event information.
c24: IF (A.3/1 OR A.3/7A OR A.3/7B) AND A.4/22 THEN o ELSE n/a - - UE, AS acting as terminating UA, or redirect server, AS acting as originating UA all as notifier of event information.

Annex E (informative): Change history

Change history								
Date	**TSG #**	**TSG Doc.**	**CR**	**Rev**	**Subject/Comment**	**Old**	**New**	**WG doc**
2005-04					Version 0.0.0 created		0.0.0	C1-050804
2005-07					Version 0.0.1, TR-number added, some formatting cleanup by the rapporteur.	0.0.0.	0.0.1	
2005-09					Version 0.1.0, created after CT1 #39, some formatting cleanup by the rapporteur.	0.0.1	0.1.0	C1-051174 C1-051104 C1-051105 C1-051106 C1-051107 C1-051108 C1-051148

								C1-051168 C1-051169
2005-09					Further formatting corrections and internal alignment by the rapporteur.	0.1.0.	0.1.1	
2005-10					Version 0.2.0, created after CT1 #39bis, some formatting cleanup by the rapporteur.	0.1.1	0.2.0	C1-051259 C1-051260 C1-051276 C1-051277 C1-051263 C1-051266 C1-051267 C1-051264 C1-051265 C1-051268 C1-051270 C1-051278 C1-051272 C1-051273 C1-051279
2005-11					Version 0.3.0, created after CT1 40. Clauses renumbered to align with the future technical specification TS 24.279. Some formatting cleanup by the rapporteur.	0.2.0	0.3.0	C1-051386 C1-051410 C1-051420 C1-051438 C1-051460 C1-051514 C1-051626 C1-051629 C1-051632 C1-051633 C1-051678 C1-051680
2005-11					Editorial corrections. Change history annex moved to the end of the document. Normative annex "Media feature tags defined within the current document" placed before informative ones.	0.3.0	0.3.1	
2005-11					Adjusted the revision marks in Annex C to reflect the correction of wrong codepoint in the user-user information IE in CRs C1-051357, C1-051358, C1-051359 to 3GPP TS 24.008.	0.3.1	0.3.2	
2005-11					Version 1.0.0 created by MCC to present the TR to TSG CT#30 for information	0.3.2	1.0.0	
2006-01					Version 1.1.0 created after CT1 40bis.	1.0.0	1.1.0	C1-060018 C1-060073 C1-060093 C1-060094 C1-060096 C1-060098 C1-060099 C1-060100 C1-060131 C1-060132
2006-02					Version 1.2.0 created after CT1 41.	1.1.0	1.2.0	C1-060241 C1-060521 C1-060523 C1-060524 C1-060525 C1-060591 C1-060604
2006-02					Version 2.0.0 created by MCC to be sent to CT-31 for approval	1.2.0	2.0.0	CP-060098
2006-03	CT-31	CP-060098			Version 2.0.0 was approved and version 7.0.0 created by MCC	2.0.0	7.0.0	