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Technical Specification

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LTE;
IP Multimedia Subsystem (IMS) Centralized Services (ICS) protocol via I1 interface
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Foreword

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Contents

Foreword

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

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1 Scope

The present document describes the I1 interface between IMS Centralized Services (ICS) UE and Service Centralization and Continuity (SCC) Application Server (AS).

This specification defines a new application layer protocol over I1 interface, specifies the interaction between the ICS UE and the SCC AS including session control procedures and supplementary services control procedures.

The protocol is intended to be independent of the transport-layer protocol used so it can be applied to a number of technologies that need different transport-layer protocols.

The overall ICS architecture is specified in 3GPP TS 23.292 [2].

The procedures for delivery of IMS Service Continuity that do not use the I1 protocol are specified in the document 3GPP TS 24.237 [13].

The present document is applicable to User Equipment (UE) and Application Servers (AS) which are intended to support the IMS centralized services.

2 References

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

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

[1] 3GPP TR 21.905: "Vocabulary for 3GPP Specifications".

[2] 3GPP TS 23.292: "IP Multimedia Subsystem (IMS) Centralized Services; Stage 2".

[3] 3GPP TS 24.008: "Mobile radio interface layer 3 specification; Core Network protocols; Stage 3".

[4] 3GPP TS 24.090: "Unstructured Supplementary Service Data; Stage 3".

[5] 3GPP TS 24.292: "IP Multimedia (IM) Core Network (CN) subsystem Centralized Services (ICS); Stage 3".

[6] RFC 3261 (June 2002): "SIP: Session Initiation Protocol".

[7] 3GPP TS 23.237: "IP Multimedia Subsystem (IMS) Service Continuity; Stage 2".

[8] RFC 3323 (November 2002): "A Privacy Mechanism for the Session Initiation Protocol (SIP)".

[9] RFC 3325 (November 2002): "Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity within Trusted Networks".

[10] 3GPP TS 23.009: "Handover Procedures".

[11] 3GPP TS 25.413: "UTRAN Iu interface Radio Access Network Application Part (RANAP) signalling".

[12] 3GPP TS 24.229: "IP multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP); Stage 3".

[13] 3GPP TS 24.237: "IP Multimedia Subsystem (IMS) Service Continuity; Stage 3".

[14] 3GPP TS 29.002: "Mobile Application Part specification; Stage 3".

[15] 3GPP TS 23.003: "Numbering, addressing and identification".

[16] RFC 3629 (2003): "UTF-8, a transformation format of ISO 10646".

[17] 3GPP TS 23.218: "IP Multimedia (IM) session handling IM Call model, Stage 2".

[18] 3GPP TS 24.173: "IMS Multimedia Telephony Communication Service and Supplementary Services; Stage 3".

[19] 3GPP TS 23.038: "Alphabets and language-specific information".

[20] 3GPP TS 23.078: "Customised Applications for Mobile network Enhanced Logic (CAMEL); Stage 2".

[21] 3GPP TS 24.147: "Conferencing using the IP Multimedia (IM) Core Network (CN) subsystem; Stage 3".

[22] 3GPP TS 24.605: "Conference (CONF) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification".

[23] 3GPP TS 24.615: "Communication Waiting (CW) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification".

3 Definitions and abbreviations

3.1 Definitions

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

For the purposes of the present document, the following terms and definitions given in 3GPP TS 23.292 [2] apply:

ICS UE

SCC AS

For the purposes of the present document, the following terms and definitions given in 3GPP TS 23.237 [7] apply

Access Transfer

Service Control Signalling Path

Session Transfer Identifier (STI)

For the purposes of the present document, the following terms and definitions given in 3GPP TS 24.292 [5] apply:

PSI DN

For the purposes of the present document, the following terms and definitions given in 3GPP TS 24.229 [12] apply:

default public user identity

Stable I1 Session: An session which is setup by the ICS UE using the CS bearer, and for which service control via the I1 interface is applicable.

3.2 Abbreviations

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

ICS	IMS Centralized Services
SCC AS	Service Centralization and Continuity Application Server
STI	Session Transfer Identifier
USSD	Unstructured Supplementary Service Data

4 General description

4.1 General

For the current version of the specification the application layer protocol is run over Unstructured Supplementary Service Data (USSD) transport as defined in 3GPP TS 24.090 [4], however the application layer protocol is not restricted to USSD transport.

4.2 Structure of the protocol

4.2.1 Introduction

The I1 protocol is a message based point to point protocol. The I1 protocol messages are transported within a point-to-point transport-layer protocol and are exchanged between the ICS UE and SCC AS.

The I1 protocol is a transport-independent protocol, i.e. the I1 session control entities can exchange the I1 protocol messages over any transport-layer protocol that connects the ICS UE and the SCC AS.

The I1protocol's notation maintains a format of two parts, i.e. I1 message common part and I1 information elements. The I1 message common part is included in every I1 message. The I1 information elements those are included in an I1 message depend on a type of I1 message being sent.

4.2.2 Application level protocol

Overall descriptions with application level protocol are specified as following:

1) it is used to access IMS services (e.g., IMS session origination);

2) it is a point to point protocol between the ICS UE and the SCC AS;

3) its protocol does not support authentication;

4) it does not support segmentation of messages;

5) its messages are self-identifying; and

6) it runs over any point-to-point transport-layer protocol (e.g. USSD).

4.2.3 Transport-layer protocols

4.2.3.1 General

The transport-layer protocol that is used to transfer the I1 protocol messages is a bi-directional point-to-point connection between the ICS UE and the SCC AS. This transport-layer protocol is a symmetric connection, i.e. the source-point on the transport-layer protocol that is used to send the I1 protocol messages is also a destination-point for the incoming I1 protocol messages.

4.2.3.2 USSD as transport-layer protocol

The USSD provides a point-to-point transport layer connection between the I1 protocol entities. USSD is specified in 3GPP TS 24.090 [4]. When USSD is used for transporting the I1 protocol, the ICS UE and the SCC AS shall set the ussd-DataCodingScheme to "1101", as defined in 3GPP TS 23.038 [19]. For the purposes of this document, USSD is regarded as a real-time transport protocol.

The USSD supports a two-way alternative interactive communication (i.e.semi-duplex communication). At any given time, only one I1 protocol entity (either the ICS UE or the SCC AS) with its turn may send the I1 messages, while at the same time its peer is permitted only to receive the I1 messages. To allow the I1 protocol entity (either the ICS UE or the SCC AS) to send an I1 message to its peer when needed, the initiating entity shall send an I1 message exchange in a single invoke and return result transaction. That is, after the I1 protocol entities have exchanged I1 message(s) in a USSD invoke component and a USSD return result component, the USSD connection are released.

When the USSD is used as the transport layer connection, overall descriptions are specified as following:

1) the initiating entity (either the ICS UE or the SCC AS) shall send a single USSD invoke and the responding entity (either the SCC AS or the ICS UE) shall send a return result;

2) if the responding entity (either the SCC AS or the ICS UE) does not need to send an I1 response in the return result, the responding entity shall send an I1-Dummy message to the peer; and

3) if the I1 session is established, the USSD connection will be released.

For mobile initiated requests, the ICS UE shall use the Process Unstructured SS Request invoke component defined in 3GPP TS 24.090 [4]. The SCC AS shall identify the UE by matching the MSISDN received with the I1 message with the C-MSISDN of the UE. The SCC AS shall send either an I1 response to the I1 message or an I1 Dummy message (if no I1 response is required) in the Process Unstructured SS Request return result.

NOTE: When forwarding the I1 message from the UE to the SCC AS, the HSS includes the optional MSISDN information element along with the mandatory IMSI information element in the Process Unstructured SS Request, as defined in 3GPP TS 23.078 [20].

For network initated requests, the SCC AS shall use either the Unstructured SS Request or the Unstructured SS Notify, as defined in 3GPP TS 23.078 [20] to send an I1 message to the HSS for delivery to the UE. When sending the I1 message to the HSS, the SCC AS shall include within the selected invoke component the MSISDN information element populated with the C-MSISDN for the UE. When the Unstructured SS Notify invoke component is used, no I1 message is included in the return reply because the Unstructured SS Notify return result does not include any information elements. The ICS UE shall include either an I1 response to the I1 message or an I1 Dummy (if no I1 response is required) in the Unstructured SS Request return result.

The I1 Dummy in sent in the USSD return result in response to the following I1 messages in the USSD invoke: I1 Progress, I1 Success, and I1 Failure.

5 Functional entities

5.1 User Equipment (UE)

To be compliant with this specification, a UE shall implement the role of ICS UE capabilities defined in subclauses 6.2.1.2, 6.2.3.2, 6.2.4.2, 6.2.5.2.1, 6.2.5.3.1, 6.4.3, 7.5.2.1.1, 7.5.2.2.1, 7.5.2.3.1, 7.5.3.2.1.1, and 7.5.3.2.2.1.

5.2 Application Server (AS)

To be compliant with this specification, a AS shall implement the role of SCC AS capabilities defined in subclauses 6.2.1.3, 6.2.3.3, 6.2.4.3, 6.2.5.2.2,
6.2.5.3.2, 6.4.4, 7.5.2.1.2, 7.5.2.2.2, 7.5.2.3.2, 7.5.3.2.1.2, and 7.5.3.2.2.2.

6 Communication between ICS UE and SCC AS via I1 interface

6.1 Introduction

The ICS UE and SCC AS use the I1 interface to setup, control, maintain and release an I1 session control channel and associated media over the CS bearer.

If an ICS US capable of using the I1 interface registers with the IM CN Subsystem (IMS), it shall associated keys with public user identities in the format of a SIP URI in accordance with annex A. A public user identity can be derived if a key is associated with the public user identity.

6.2 Session control procedures

6.2.1 Session setup

6.2.1.1 General

The ICS UE setups the I1 session with CS media and the service control signalling via the I1 reference point. I1 is used to control services in the IM CN subsystem.

The I1 sessions can only be created by I1 session setup messages. The I1 Invite message is an I1 session setup message. The I1 sessions can be torn down by I1 session release messages. The I1 Bye message is an I1 session release message.

The following subclauses describe the procedures of the ICS UE and the SCC AS for I1 session setup:

- subclause 6.2.1.2.1 describes the procedures of ICS UE I1 session origination;
- subclause 6.2.1.2.2 describes the procedures of ICS UE I1 session termination without UE assisted T-ADS function;
- subclause 6.2.1.2.3 describes the procedures of ICS UE I1 session termination with UE assisted T-ADS function;
- subclause 6.2.1.2.4 describes the procedures of ICS UE when the I1 session fails;
- subclause 6.2.1.3.1 describes the procedures of SCC AS I1 session origination;
- subclause 6.2.1.3.2 describes the procedures of SCC AS I1 session termination without UE assisted T-ADS function;
- subclause 6.2.1.3.3 describes the procedures of SCC AS I1session termination with UE assisted T-ADS function; and
- subclause 6.2.1.3.4 describes the procedures of SCC AS when the I1 session fails.

6.2.1.2 Detailed behaviour of ICS UE

6.2.1.2.1 ICS UE CS Session Origination

6.2.1.2.1.1 General

The following subclauses describe the procedures at the ICS UE for session origination.

6.2.1.2.1.2 Sending an I1 Invite message

When the ICS UE originates an I1 session using the protocol at the I1reference point, the UE shall:

1) generate an I1 Invite message that includes:

a) a Message Type and a Reason set to indicate the message is a Mobile Originated I1 Invite message, accordance with table 7.3.1;

b) a new value in the Call-Identifier field (Part-1), as specified in subclause 7.2.2.1.4. The Call-Identifier will uniquely identify this I1 session between the ICS UE and the SCC AS;

c) an allocated Sequence-ID;

d) a From-id information element that

- if the UE has previously SIP registered and the public user identity is to be a SIP URI and the public user identity can be derived (see annex A) then:

i) if the public identity indicates the default public user identity, the Code specific field of the From-id information element is set to "Unspecified" (see table 7.4.2.3.1-2) and the length field is set to 0;

ii) if the public identity is not the default public user identity and the public user identity indicated can be derived (see annex A), the Code specific field of the From-id information element is set to "Identifier" (see table 7.4.2.3.1-2) and the length field is set to 4.

- otherwise the Code specific field of the From-id Information information element is set to:

i) a "SIP URI" (see table 7.4.2.3.1-2) if the public user identity is a SIP URI and the Information body (see table 7.3.2.2) containing the SIP URI;

ii) an "International number" (see table 7.4.2.3.1-2), if the public user identity is a Tel URI or SIP URI with URI parameter user=phone and the Information body (see table 7.3.2.2) containing the digit string contained in the URI.

e) a To-id information element that includes either a SIP URI or an E.164 number, and will be used by the SCC AS to determine the identity of the called user;

f) a Privacy information element that indicates the ICS UE's privacy preferences. The SCC AS will apply these preferences to the SIP session that the SCC AS will establish on behalf of the UE;

g) optionally include any feature tags in the:

i) Accept-Contact information element, as specified in subclause 7.3.2.5 if the parameter tag "explicit" or "require" as specified in RFC 3841 [14] are not required;

ii) ERAccept Contact information element, as specified in subclause 7.3.2.6 if the parameter tag "explicit" or "require" as specified in RFC 3841 [14] are required; and

iii) Reject Contact information element as specified in subclause 7.3.2.7; and

h) if using a transport layer protocol that is not a real-time transport layer protocol, a Timestamp information element that includes current local time measured in seconds. The element will be used by the SCC AS to determine the staleness of the message.

2) select the transport-layer protocol (see subclause 4.2.3) depending on the access network type, and forward the I1 Invite message toward the SCC AS.

6.2.1.2.1.3 Receipt of I1 Progress message with Reason Call Progress

When the UE receives an I1 Progress message with Reason set to 183 (Call Progress), the UE shall:

1) save the received Call-Identifier field value and use it for further reference to this session;

2) verify if the message is in sequence according to the value of the Sequence-ID, and save the received Sequence-ID;

3) store the SCC AS PSI DN value (i.e. the E.164 number) received in the SCC-AS-id information element; and

4) store the STI value (i.e. the E.164 number) if received in the Session-identifier information element.

NOTE 1: The STI value uniquely identifies the I1 session being established, and it may be subsequently used to refer to this I1 session, e.g. the SCC AS uses the STI to correlate the access transfer request received via the PS access with the active session established via the I1 interface.

NOTE 2: The UE may indicate the Reason value to the user.

Upon receiving the SCC AS PSI DN (i.e. the E.164 number) conveyed in the I1 Progress message with Reason set to 183 (Call Progress) from the SCC AS, the ICS UE shall initiates the call over the CS domain by sending a CC SETUP message to the MSC Server as specified in 3GPP TS 24.008 [3] as follows:

1) the Called Party BCD Number information element is set to the SCC AS PSI DN (i.e. the E.164 number) received in the I1 Progress message with Reason set to 183 (Call Progress); and

2) Type Of Number is set to "International" and Numbering Plan Indicator set to "E.164".in the Called Party BCD Number information element.

6.2.1.2.1.4 Receipt of I1 Progress message with Reason set to 180

When the ICE UE received an I1 Progress message with Reason set to 180 the UE shall:

1) provide an alerting indication to the user.

6.2.1.2.1.5 Receipt of I1 Success message

When the ICS UE receives an I1 Success message, the UE shall:

1) verify if a I1 session exists for the received Call-Identifier field value;

2) verify if a the message is in sequence according to the value of the Sequence-ID;

2A) verify that a CC CONNECT message as specified in 3GPP TS 24.008 [3] has been received in response to the SETUP message that was sent containing the SCC AS PSI DN; and

3) consider the I1 session to be established, if verification was successful.

6.2.1.2.2 ICS UE CS Session Termination without UE assisted T-ADS

If the ICS UE receives an I1 Invite message from the SCC AS, and the UE determines that no I1 session exists for the received Call-Identifier field value, the ICS UE shall:

0) if using a transport layer protocol that is not a real-time transport layer protocol, retrieve the SCC AS local time value from the Timestamp information element of the I1 Invite message, and validate the staleness of the message by applying the following equation:

 SCC_AS_time_in_the_I1_Invite_message – SCC_AS_time as specified in subclause 6.4.1 item 5)

 >=

 (ICS_UE_local_time - ICS_UE_time as specified in subclause 6.4.3.1 item 1)e) – Deviation

 NOTE: Deviation parameter is 64*T1 seconds.

 If the equation is true the message is not stale and it shall processed by the following sections. Otherwise, the messages is discarded and no response is generated to the I1 Invite message; and

1) store the information contained in the I1 Invite message, including the called party identity included in the To-id information element, the calling user's public user identity included in the From-id information element, the SCC AS PSI DN (i.e., the E.164 number) included in the SCC-AS-id information element, the Sequence-ID, the Call-Identifier field (Part-2), the STI value (i.e. the E.164 number) if received in the Session-identifier information element, Accept-Contact information element, ERAccept-Contact information element and Reject-Contact information element and transport layer information identifying the transport connection over which the I1 Invite message was received; and

1A) if the To-id information element in the I1 Invite message contains a:

i) Code Specific Information element set to "Unspecified" (see table 7.4.2.3A.1) and a length field set to 0 then the Public user identity shall be set to the default public user identity.

ii) Code Specific Information element set to "Identifier" (see table 7.4.2.3A.1-2) and a length field set to 4, then the public user identity can be derived (see Annex A) and shall be set to the identifier value received in the information element body of the To-id information element.

iii) Code Specific Information element set to "International number" (see table 7.4.2.3A.1-2) or "SIP URI" (see table 7.4.2.3.1-2), then the public user identity of the UE shall be set to the Identity in the Information element body of the To-id information element.

NOTE 1: The UE may indicate the public user identity used to address the UE in the incoming session to the user.

2) initiate a call over CS bearer by sending a CC SETUP message to the MSC Server as specified in 3GPP TS 24.008 [3] as follows:

i) the Called Party BCD Number information element is set to the received SCC AS PSI DN (i.e., the E.164 number) received in the I1 Invite message

ii) Type Of Number set to "International" and Numbering Plan Indicator set to "E.164".in the Called Party BCD Number information element.

NOTE 2: When the ICS UE receives an I1 Invite message, the UE may send an I1 Progress message with Reason set to 180 (Call Progress). The I1 Progress message with Reason set to 180 (Call Progress) is identical to the I1 Progress message with Reason set to 180 (Ringing) described below, except the Reason field will be set to Call Progress.

When the ICS UE receives an indication from the CS domain that the media resources are available (i.e. the UE receives a CC ALERTING message as specified in 3GPP TS 24.008 [3]) the UE shall:

1) generate an I1 Progress message with Reason set to 180 (Ringing) containing the following information:

a) a Message Type and a Reason set to that indicate the message is an I1 Progress message, accordance with table 7.3.1;

b) a new value in the Call-Identifier field (Part-1), as specified in subclause 7.2.2. The resulting Call-Identifier value uniquely identifies this I1 session between the UE and SCC AS;

NOTE 3: A new value in the Call-Identifier field (Part-1) is inserted only if this is the first I1 message sent to the SCC AS. Otherwise the previously set Call-Identifier value is used.

c) increment the stored message sequence value, store it, and include it in the Sequence-ID;

d) set the Reason field (per figure 7.3.1) to 180; and

2) send the I1 Progress message with Reason set to 180 (Ringing) towards the SCC AS over the transport layer connection over which the I1 Invite message was received.

If the user accepts the request, the ICS UE shall:

1) generate an I1 Success message containing the following information:

a) a Message Type and a Reason set to indicate the message is an I1 Success message, accordance with table 7.3.1;

b) the stored Call-Identifier value that uniquely identifies this I1 session between the UE and SCC AS;

c) increment the stored message sequence value, store it, and include it in the Sequence-ID; and

2) send the I1 Success message towards the SCC AS over the transport layer connection over which the I1 Invite message was received.

6.2.1.2.3 ICS UE CS Session Termination with UE assisted T-ADS

If the ICS UE receives an I1 Invite (Augmentation) message with a Replaces information element and it is determined that there is a SIP session being established for the Replaces information element value (e.g., the Replaces information element is set to a value identical to (or deduced from) the SIP session identifier in a previously received SIP INVITE), the ICS UE:

1) shall interpret it as session control fallback from Gm to I1; and

2) shall use the Replaces information element value to correlate the I1 Invite message with the SIP INVITE request prevously received, to get SCC AS PSI DN, the called party identity and the calling party identity.

3) shall indicate that the public user identity, the To-id information element, and the SCC AS PSI DN are in the correlated SIP INVITE request, by setting the Code specific field of the To-id information element to "Unspecified" (see table 7.4.2.3A.1) and the length field is set to 1 and octet 3 is set to all "0"s, respectively.

NOTE: In this case, some information element (e.g. Privacy information element) can be omitted from the I1 Invite message, for the information can be obtained by the ICS UE from the correlated SIP INVITE request.

Afterwards, the ICS UE shall behave as specified in subclause 6.2.1.2.2.

6.2.1.2.4 Failure

The ICS UE may receive an I1 Failure message at any time. If the ICS UE receives an I1 Failure message, the ICS UE shall:

1) save the received Call-Identifier field value and use it for further reference to this session;

2) verify if the message is in sequence according to the value of the Sequence-ID, and save the received Sequence-ID;

3) extract the Reason Value as defined in subclause 7.2.2.1.3 from the message; and

4) act in accordance with corresponding equivalent status code value as defined in subclauses 21.3 to 21.6 of RFC 3261[6].

5) release the session as defined in subclause 6.2.3

6.2.1.3 Detailed behaviour of SCC AS

6.2.1.3.1 SCC AS CS Session Origination

6.2.1.3.1.1 General

The following subclauses describe the procedures at the SCC AS for I1 session origination. In this scenario, the SCC AS serves the originating user.

6.2.1.3.1.2 Receipt of I1 Invite message

Upon receiving an initial I1 Invite message from the ICS UE via the I1 reference point, the SCC AS shall:

0) if using a transport layer protocol that is not a real-time transport layer protocol, retrieve the ICS UE local time value from the Timestamp information element of the I1 Invite message, and validate the staleness of the message by applying the following equation:

   ICS_UE_time_in_the_I1_Invite_message – ICS_UE_time_ as specified in subclause 6.4.4.1 item 1)

   >=

   (SCC_AS_local_time - SCC_AS_time_ as specified in subclause 6.4.4.1 item 3)d) – Deviation

NOTE: Deviation parameter is 64*T1 seconds.

   If the equation is true the message is not stale and it shall processed by the following sections. Otherwise, the messages is discarded and no response is generated to the I1 Invite message; and

1) store the information received in the I1 Invite message, including the called party identity included in the To-id information element, From-id information element, the requested privacy type included in the Privacy information element, the Sequence-ID, Accept-Contact information element, ERAccept-Contact information element and Reject-Contact information element, Call-Identifier field (Part-1) (as specified in subclause 7.2.2.1.4), and transport layer information identifying the transport connection over which the I1 Invite message was received; against the IMS private identity of the originating user's UE. The IMS private identity to store the information against is determined by comparing the C-MSISDN associated with the IMS private identity against the:

   i) MAP service ISDN-Address-String as specificed in 3GPP TS 29.002 [14] if USSD is used as the transport layer protocol for the message,

1A) dynamically allocate a STI and bind it to the information stored in step 1. The STI is specified as an E.164 number;

NOTE 1: The STI value uniquely identifies the I1 session being established, and it may be subsequently used to refer to this I1 session, e.g. the SCC AS uses the STI to correlate the access transfer request received via the PS access with the active session established via the I1 interface.

1B) If the From-id information element in the I1 Invite message is

   i) included and the Code Specific information element is set to "Unspecified" (see table 7.4.2.3.1-2) and the length field is set to 0 the default public user identity shall be stored against the I1 Invite message.

   ii) included and the Code Specific information element is set to "Identifier" (see table 7.4.2.3.1-2) and the length field is set to 1, the received identifier as derived in Annex A shall be stored against the I1 Invite message.

   iii) included and the Code Specific information element is set to set to "International number" or "SIP URI" the Identity contained in the information element body of the To information element value shall be stored against the I1 Invite message.

2) Void

6.2.1.3.1.3 Sending an I1 Progress message in response to I1 Invite message

The SCC AS shall:

1) generate an I1 Progress message containing the following information:

   a) a Message Type and a Reason set to indicate the message is an I1 Progress message, accordance with table 7.3.1;

   b) a Call-Identifier field, that was constructed by appending the allocated Call-Identifier (Part-2) subfield to the stored Call-Identifier (Part-1) subfield, as specified in subclause 7.2.2.1.4. The Call-Identifier value uniquely identifies this I1 session between the ICS UE and SCC AS;

   c) add one to the stored message sequence value. Store and include in the Sequence-ID;

   d) include the allocated SCC AS PSI DN (i.e., the E.164 number) in the SCC-AS-id information element;

   e) set the Reason field to 183 (per figure 7.3.1);

   f) include the allocated STI (i.e., the E.164 number) in the Session-identifier information element; and

2) send the I1 Progress message towards the originating ICS UE over the transport layer connection over which the I1 Invite message was received.

6.2.1.3.1.4 Sending an I1 Progress message with reason set to 180

When sending an I1 Progress message with Progress reason set to 180 towards the originating UE, the SCC AS shall:

1) generate an I1 Progress message containing the following information:

   a) the Message Type and a Reason set to indicate the message is an I1 Progress message, accordance with table 7.3.1;

b) the stored Call-Identifier field (as specified in subclause 7.2.2.1.4) that uniquely identifies this I1 session between the ICS UE and SCC AS ; and

c) add one to the stored message sequence value. Store and include in the Sequence-ID; and

2) send the I1 Progress message towards the originating ICS UE over the transport layer connection over which the I1 Invite message was received.

6.2.1.3.1.5 Sending an I1 Success message

When sending an I1 Success message towards the orginating ICS UE, the SCC AS shall:

1) generate an I1 Success message containing the following information:

a) a Message Type and a Reason set to indicate the message is an I1 Success message, accordance with table 7.3.1;

b) a Call-Identifier field containing the Call-Identifier value that uniquely identifies this I1 session between the ICS UE and SCC AS;

2) add one to the stored message sequence value. Store and include in the Sequence-ID; and

3) send the I1 Success message towards the originating ICS UE over the transport layer connection over which the I1 Invite message was received.

6.2.1.3.2 SCC AS CS Session Termination without ICS UE assisted T-ADS

6.2.1.3.2.1 Sending an Initial I1 Invite message

When sending an I1 Invite message towards the ICS UE, the SCC AS shall:

1) perform the procedures per 3GPP TS 24.292 subclause 10.4.4 item 1;

1A) dynamically allocate a STI. The STI is specified as an E.164 number;

NOTE 1: The STI value uniquely identifies the I1 session being established, and it may be subsequently used to refer to this I1 session, e.g. the SCC AS uses the STI to correlate the access transfer request received via the PS access with the active session established via the I1 interface.

2) create an I1 Invite message that includes:

a) a Message Type and a Reason set to indicate the message is a Mobile Terminated I1 Invite message, accordance with table 7.3.1;

b) a Call-Identifier field, that includes an allocated Call-Identifier (Part-2) subfield, (see subclause 7.2.2.1.4). The Call-Identifier field in spite of containing only the Part-2 value uniquely identifies this I1session between the ICS UE and SCC AS;

c) a Sequence-ID;

d) a From-id information element that identifies the remote calling party, if available and if privacy was not requested by the calling party as specified in RFC 3323 [8] and RFC 3325 [9]. If either the E.164 number that identifies the remote calling party is not available or privacy was requested, then the From-id information element shall not be included in the I1 Invite message;

NOTE 2: The SCC AS will include in the From-id information element the remote calling party only if it is an E.164 number and if privacy was not requested.

e) a To-id information element that;

- if the UE has previously SIP registered as specified in 3GPP TS 23.218 [17] and the R-URI is a SIP URI and the R-URI can be derived (see annex A) then if the R-URI in the received SIP INVITE request is;

i) the default public user identity as derived in Annex A for the terminating UE then the Code specific field of the To-id Information information element is set to "Unspecified" (see table 7.4.2.3A.1-2) and the length field is set to 0.

ii) is not the default public user identity for terminating UE but matches one of the public user identities then the Code specific field of the Identity Information information element is set to "Identifier" (see table 7.4.2.3A.1-2) and the length field is set to 1 and the Information Element body of the To-id information element information element shall be the identifier value that was derived (see annex A) and maps to the Public User Identity that was received in the R-URI in the SIP INVITE request.

- otherwise Code specific field of the To-id Information information element is set to

i) a "SIP URI" (see table 7.4.2.3A.1-2) if the public user identity is a SIP URI and the Information body (see table 7.3.2.2) containing the SIP URI.

ii) an "International number" (see table 7.4.2.3A.1-2), if the public user identity is a Tel URI or SIP URI with URI parameter user=phone and the Information body (see table 7.3.2.2) containing the digit string contained in the URI.

f) a Privacy information element set to the value requested by the remote calling party, if available;

g) a SCC-AS-id information element that contains an SCC AS PSI DN set to the E.164 number allocated by the SCC AS itself as per procedures per 3GPP TS 24.292 subclause 10.4.8 item 2;

h) a Session-identifier information element that contains the allocated STI; and

i) if using a transport layer protocol that is not a real-time transport layer protocol, a Timestamp information element that includes current local time measured in seconds. The element will be used by the ICS UE to determine the staleness of the message.

3) store the information sent in the I1 Invite message against the allocated SCC AS PSI DN; and

4) select the transport layer protocol (see subclause 4.2.3) depending on the access network type, and forward the I1 Invite message toward the ICS UE.

Subsequently the SCC AS may receive either an I1 Success message or an I1 Progress message (with Reason field set either to Ringing or Call Progress) from the ICS UE.

6.2.1.3.2.2 Receipt of an I1 Progress message

When the SCC AS receives either an I1 Progress message (with Reason field set either to Ringing or Call progressing), the SCC AS shall:

1) verify if a I1 session exists for the received Call-Identifier field value;

2) verify if the message is in sequence according to the Sequence-ID

3) store the I1 Progress with reason.

NOTE 5: The SCC AS will use the information received in the I1 Progress message (with Reason field set either to Ringing or Call Progress) and the information saved in step 2 when handling a SIP session with the remote party.

6.2.1.3.3 SCC AS CS Session Termination with ICS UE assisted T-ADS

The SCC AS shall generate an I1 Invite message to the terminating ICS UE as specified in subclause 6.2.1.3.2.1 with the following addition:

1) Include a Replaces information element in the I1 (Augmentation) Invite message, which is set to a value identical to (or deduced from) the SIP session identifier in the previous SIP INVITE request, to indicate that it is session control fallback from Gm to I1;

2) Indicate that the public user identity, the To-id information element value and the SCC AS PSI DN are in the correlated SIP INVITE request, by setting the Code specific field of the To-id information element to "Unspecified" (see table 7.4.2.3A.1) and the length field is set to 1 and octet 3 is set to all "0"s.

NOTE: In this case, some information elements (e.g., Privacy information element) can be omitted from the I1 Invite message, for the information can be get by the ICS UE from the correlated SIP INVITE request.

6.2.1.3.4 Failure

The SCC AS shall:

1) create an I1 Failure message that includes:

   a) a Message Type set to the value that indicates that this is an I1 Failure message;

   b) set the reason value (see table 7.3.1) to the same value as received in the status code as specified in subclauses 21.3 to 21.6 of RFC 3261 [6];

   b) generate a Call-Identifier value that identifies the transaction between the ICS UE and SCC AS. Include the Call-Identifier value in the Call-Identifier field in the I1 Failure message;

   c) generate a Sequence-ID. Include the Sequence-ID in the I1 Failure Message;

   e) if a contact header value as specified in subclause 21.3 of RFC 3261 [6] was received in the SIP error response include individual To-id information elements containing the URI contents of each contact header field.

   f) if reason-phrase header as specified in subclauses 21.3 to 21.6 of RFC 3261[6] was received, insert the contents of the reason-phrase header into the Reason field (see table 7.3.8.1).

6.2.2 Void

6.2.3 Session release

6.2.3.1 General

I1 sessions can be torn down by the I1 session release requests and by receipt of a CC DISCONNECT message as specified in 3GPP TS 24.008 [3]. An I1 Bye message is an I1 session release request.

6.2.3.2 Detailed behaviour of ICS UE

When the ICS UE releases an I1 session using the I1 session control channel by sending an I1 Bye message, it shall:

1) set the Call-Identifier field to a value that identifies the I1 session between the ICS UE and SCC AS. Include the Call-ID field in the I1 Bye message;

2) set the Sequence-ID. Include the Sequence-ID in the I1 Bye message;

3) a From-id information element that includes either a SIP URI or an E.164 number, and it will be used by the SCC AS to identify the ICS UE;

4) a To-id information element that includes either a SIP URI or an E.164 number, and will be used by the SCC AS to determine the identity of the called user;

5) a Privacy information element that indicates the ICS UE's privacy preferences. The SCC AS will apply these preferences to the SIP session that the SCC AS will establish on behalf of the UE; and

6) if there are no more I1 service control sessions using the CS bearer, set the CS bearer release timer value.

If the CS bearer release timer expires, the ICS UE shall send a CC DISCONNECT message to the MSC Server as specified in 3GPP TS 24.008 [3], if needed.

Subsequently, if the ICS UE receives an I1 Success message from the SCC AS, it shall:

1) verify if a I1 session exists for the received Call-Identifier field value;

2) verify if a the message is in sequence according to the value of the Sequence-ID; and

3) consider the I1 session to be released, if verification was successful and clear the CS bearer release timer value.

When the ICS UE releases a I1 session using the I1 session control channel by receiving an I1 Bye message, it shall:

1) if there are no more I1 service control sessions using the CS bearer, the ICS UE shall send a CC DISCONNECT message to the MSC Server as specified in 3GPP TS 24.008 [3], if there are no more I1 service control sessions using the CS bearer;

2) if there are more I1 service control sessions using the CS bearer, the ICS UE shall transmit a I1 Success message, containing the following information:

   a) a Message Type set to the value that indicates that is an I1 Success message;

   b) a Call-Identifier field with the stored Call-Identifier value that uniquely identifies this I1 session between the ICS UE and SCC AS; and

   c) increment the stored message sequence value, store it, and include it in the Sequence-ID.

When the ICS UE receives a CC DISCONNECT message to release the CS bearer as specified in 3GPP TS 24.008 [3]:

- the CS bearer release timer expires shall be cleared, if needed;

- if the ICS UE has a SIP REGISTER request associated with the ongoing CS call, the UE shall send a SIP reINVITE request requesting the media over the CS bearer to be deleted.

If the ICS UE receives a SIP reINVITE request requesting the media over the CS bearer to be deleted and a DISCONNECT message for the CS bearer was already received, the ICS UE shall accept the request to delete the media over the CS bearer.

6.2.3.3 Detailed behaviour of SCC AS

6.2.3.3.1 Sending an I1 Bye message to the UE

The SCC AS enhanced for I1 releases a session by generating and sending an I1 Bye message, containing the information:

0) a Message Type and a Reason set to indicate the message is an I1 Bye message, accordance with table 7.3.1;

1) set the Call-Identifier to a value that identifies the I1 session between the ICS UE and SCC AS. Include the Call-Identifier value in the Call-Identifier field in the I1 Bye message;

2) set the Sequence-ID. Include the Sequence-ID in the I1 Invite message.

3) include a From-id information element that identifies the remote calling party, if available;

4) include a To-id information element that includes the E.164 number of the ICS UE;

5) include a Privacy information element set to the value requested by the remote calling party, if available;

6) include a SCC-AS-id information element that contains an SCC AS DN set to the E.164 number allocated by the SCC AS itself; and

7) store the information sent in the I1 Bye message against the allocated SCC AS PSI DN.

6.2.3.3.2 Receipt of I1 Success message from the UE

If the SCC AS receives an I1 Success message from the ICS UE, it shall:

1) verify if a I1 session exists for the received Call-Identifier field value;

2) verify if a the message is in sequence according to the value of the Sequence-ID; and

3) consider the I1 session to be released, if verification was successful.

If any of the operations in 1) or 2) fail the I1 Success message shall be discarded.

6.2.3.3.3 Receipt of I1 Bye message from the UE

The SCC AS shall transmit an I1 Success message using the I1 session control channel, it shall containing the following information:

a) a Message Type and a Reason set to indicate the message is an I1 Success message, accordance with table 7.3.1;

b) a Call-Identifier field set to the stored Call-Identifier value that uniquely identifies this I1 session between the ICS UE and SCC AS; and

c) increment the stored message sequence value, store it, and include it in the Sequence-ID.

6.2.3A I1 session setup when invoking mid-call supplementary service

6.2.3A.1 General

If the ICS UE and SCC AS want to use the I1 protocol that apply to a CS call that was previously established without using the I1 interfaces (e.g. the call was set as specified in 3GPP TS 24.292 [5] subclause 10.4.7 or subclause 7.4.3), the ICS UE and SCC AS may avoid the explicit exchange of the I1 messages to create an I1 session to bind it to the respective CS call (as specified in subclause 6.2.4). In this case, it is assumed that a reserved Call-Identifier value (i.e. all bits set to values "1") will be used to exchange the initial I1 messages (e.g. mid-call I1 messages). Hence, the I1 control of an established CS call is acquired (i.e. an I1 session is automatically created and bound to the respective CS call), upon setting up a transport-layer connection between the ICS UE and the SCC AS, and subsequently the ICS UE and the SCC AS successfully exchanging the I1 messages (e.g. the I1 Mid Call Request message and the I1 Success messages) that use the reserved Call-Identifier value. The ICS UE or SCC AS shall add the I1 control that uses the that a reserved Call-Identifier value to an existing CS call as specified in this subclause only when there is a single CS call (i.e. a CS call that was set up without using the I1) between the ICS UE and the SCC AS over the CS domain. In this case, the information that is already known to the ICS UE and the SCC AS from the existing CS call (e.g. To-id, From-id, Privacy) will be also bound to this I1 session that use the reserved Call-Identifier value and the associated CS call (i.e. a CS call that was set up without using the I1 interfaces). Any I1 session that is subsequently created shall not use the reserved Call-Identifier value.

The automatically established I1 session that uses the reserved Call-Identifier value and associated CS call may be to control the CS call, e.g. to invoke the mid-call supplementary services or to cleared an CS call that is on hold.

6.2.3A.2 Detailed behaviour of ICS UE

6.2.3A.2.1 Mid-call service initiated by ICS UE

If the ICS UE wants to add the I1 control to an existing end-to-end CS call that was previously established without using the I1 protocol, and invoke a mid-call supplementary service to this call, the ICS UE shall send an I1 message to the SCC AS. The ICS UE shall populate the I1 message, as specified in subclause 6.2.1.2.1.2 and subclause 6.3 with the following additions:

1) insert a new value in the Call-Identifier (Part-1) subfield, as specified in subclause 7.2.2.1.4. When inserting a new value in the Call-Identifier (Part-1) subfield, the UE shall set all bits in the Part-1 subfield to values "1" (i.e. "11111111"); and

NOTE 1: In this case some information is already known to the ISC UE from the existing end-to-end call (e.g. To-id, From-id, Privacy)

2) send the I1 message to the SCC AS.

Upon receiving an I1 Success message from the SCC AS with all bits in the Call-Identifier (Part-1 and Part-2) subfield set to values "1", the ICS UE shall treat the I1 session as established and the requested supplementary service (as specified in the subclause 6.3), as applied to the existing end-to-end call (that was set up without using the I1 protocol). The established I1 session may be subsequently used either by the ICS UE or the SCC AS, to exchange the I1 messages that pertain to this end-to-end call (e.g. send an I1 Mid Call Request message to resume the call, if the call was previously placed on hold, as specified in subclause 6.3).

NOTE 2: The Call-Identifier (Part-1 and Part-2) subfield with all bits set to values "1" is a reserved value that identifies the I1 session that was automatically created upon invocation of the mid-call supplementary service.

NOTE 3: The allocated STI is always included in the first I1 message sent by the SCC AS to the ICS UE. If the CS-to-PS access transfer is performed prior to the ICS UE obtaining the allocated STI from the SCC AS, the static STI is used for the CS-to-PS access transfer, as described in 3GPP TS 24.237 [13].

6.2.3A.2.2 Mid-call service initiated by SCC AS

If the ICS UE receives an I1 message from the SCC AS that contains the Call-Identifier field with all bits in the Part-2 subfield set to the values "1", the ICS UE shall identify the call that was previously set without using the I1 protocol.

If the ICS UE identifies an existing end-to-end call (that was previously established without using the I1 protocol), the ICS UE shall generate an I1 Success message containing the information as specified in subclause 6.2.1.2.2 and subclause 6.3 with the following additions:

1) insert a new value in the Call-Identifier (Part-1) subfield with all bits in the Part-1 subfield set to values "1"; and

2) treat the indicated mid-call supplementary service (specified in subclause 6.3) as applied to the existing call, and the I1 session as established.

NOTE: The allocated STI is always included in the first I1 message sent by the SCC AS to the ICS UE. If the CS-to-PS access transfer is performed prior to the ICS UE obtaining the allocated STI from the SCC AS, the static STI is used for the CS-to-PS access transfer, as described in 3GPP TS 24.237 [13].

6.2.3A.3 Detailed behaviour of SCC AS

6.2.3A.3.1 Mid-call service initiated by ICS UE

If the SCC AS receives an I1message from the ICS UE that contains the Call-Identifier field with all bits in the Part-1 subfield set to values "1", the SCC AS shall determine if this I1 message is for an existing end-to-end call (that was previously established without using the I1 protocol) by using the CS domain number (e.g., MSISDN) obtained from the transport layer. If the SCC AS identifies an existing end-to-end call (that was previously established without using the I1 protocol), the SCC AS shall:

1) perform the steps specified in subclause 6.2.1.3.1.2 and subclause 6.3, and invoke the requested supplementary service, as specified in subclause 6.3 (e.g. perform the standard SIP procedures toward the far end and the MGCF).

NOTE 1: In this case, some information elements that are already known to the SCC AS from the existing CS end-to-end call (e.g. To-id, From-id, Privacy) will be bound to the existing (permanent) I1 session and the associated single end-to-end call.

Upon successfully performing the requested supplementary service, the SCC AS shall generate an I1 Success message containing the information as specified in subclause 6.2.1.3.1.3 and subclause 6.3 with the following additions:

1) insert a new value in the Call-Identifier (Part-2) subfield, the UE shall set all bits in the Part-2 subfield to values "1";

2) send the I1 Success message towards the ICS UE; and

3) treat the indicated mid-call supplementary service (specified in subclause 6.3) as applied to the existing end-to-end call, and the I1 session as established.

NOTE 2: The allocated STI is always included in the first I1 message sent by the SCC AS to the ICS UE. If the CS-to-PS access transfer is performed prior to the ICS UE obtaining the allocated STI from the SCC AS, the static STI is used for the CS-to-PS access transfer, as described in 3GPP TS 24.237 [13].

6.2.3A.3.2 Mid-call service initiated by SCC AS

If the SCC AS wants to add the I1 control to an existing end-to-end call that was previously established without using the I1 protocol, and invoke a mid-call supplementary service to this call, the SCC AS shall:

1) invoke the supplementary service, as specified in subclause 6.3 (e.g. perform the standard SIP procedures toward the far end and the MGCF); and

2) send an I1 message to the ICS UE specifying the invoked supplementary service, as specified in subclause 6.3. The SCC AS shall populate this I1 message as specified in subclause 6.2.1.3.2.1 and subclause 6.3 with the following additions:

   a) insert a new value in the Call-Identifier (Part-2) subfield, as specified in subclause 7.2.2.1.4. When inserting a new value in the Call-Identifier (Part-2) subfield, the UE shall set all bits in the Part-2 subfield to values "1"; and

   b) not include the SCC-AS-id information element (that includes the SCC AS PSI DN) in the I1 message.

NOTE 1: In this case some information is already known to the SCC AS from the existing end-to-end call (e.g. To-id, From-id, Privacy).

NOTE 2: The allocated STI is always included in the first I1 message sent by the SCC AS to the ICS UE. If the CS-to-PS access transfer is performed prior to the ICS UE obtaining the allocated STI from the SCC ASI, the static STI is used for the CS-to-PS access transfer, as described in 3GPP TS 24.237 [13].

Upon receiving an I1 Success message from the ICS UE with all bits in the Call-Identifier (Part-1 and Part-2) subfield set to values "1", the SCC AS shall treat the I1 session as established and the requested supplementary service (as specified in the subclause 6.3), as applied to the existing end-to-end call (that was set up without using the I1 protocol). The established I1 session may be subsequently used either by the ICS UE or the SCC AS, to exchange the I1 messages that pertain to this end-to-end call.

6.2.4 Adding I1 control to existing CS session (I1 Augmentation)

6.2.4.1 General

Standard CS procedures can be used to deliver the incoming session to the ICS UE as specified in 3GPP TS 24.292 [5] subclause 10.4.7 (SCC AS for call termination over CS to non-ICS UE) or originate a session as specified in 3GPP TS 24.292 [5] subclause 7.4.3 (ICS UE using CS). Additional IMS parameters or service control can be optionally communicated to the ICS UE using I1 after the call has been setup. The ICS UE or SCC AS shall add I1 control to an existing call only when there is a single session over CS.

6.2.4.2 Detailed behaviour of ICS UE

6.2.4.2.1 Augmentation initiated by ICS UE

If the ICS UE wants to add I1 control to an existing call that was previously established without using either the I1 or the Gm interfaces, the ICS UE shall send an I1 Invite message over I1interface. The ICS UE shall populate the I1 Invite message as specified in subclause 6.2.1.2.1.2 with the following additions:

1) set the Reason field in the I1 Invite message to value hex "002", as specified in Table 7.3.1. This value indicates that augmentation is requested;

NOTE: In this case, some information elements (e.g. From, Privacy) can be omitted from the I1 Invite message, for the information is already known for the ongoing session.

Upon receiving an I1 Success message from the SCC AS, the ICS UE shall treat the ongoing I1 session as established and the existing call being controlled by the respective I1 session.

6.2.4.2.2 Augmentation initiated by SCC AS

If the ICS UE receives a new I1 Invite message from the SCC AS containing the Reason field set to value hex of "002", as specified in Table 7.3.1, the ICS UE shall determine if this I1 Invite message is for an ongoing call that was established without I1 and Gm. If there is an ongoing call, the ICS UE shall:

1) respond to the I1 Invite message with an I1 Success message; and

2) treat the ongoing I1 session as established and the existing call being controlled by the respective I1 session.

6.2.4.3 Detailed behaviour of SCC AS

6.2.4.3.1 Augmentation initiated by ICS UE

If the SCC AS receives an I1 Invite message from the ICS UE containing the Reason field set to value hex "002", as specified in table 7.3.1, the SCC AS shall determine if this I1 Invite message is for an ongoing call using the CS domain number (e.g., MSISDN)obtained from the transport layer. If there is an ongoing call, the SCC AS shall:

1) respond to the I1 Invite message with an I1 Success message; and

2) treat the ongoing I1 session as established and the existing call being controlled by the respective I1 session.

6.2.4.3.2 Augmentation initiated by SCC AS

If the SCC AS wants to add I1 control to an existing call that was previously established without using either the Gm or the I1 interfaces, the SCC AS shall send a new I1 Invite message over the I1 interface. The SCC AS shall populate the I1 Invite message as specified in subclause 6.2.1.3.2.1 with the following addition:

1) set the Reason field in the I1 Invite message to value hex of "002", as specified in Table 7.3.1. This value indicates that augmentation is requested.

NOTE: In this case, some information elements (e.g. From-id, To-id, Privacy) can be omitted from the I1 Invite message, for the information is already known for the ongoing session.

Upon receiving an I1 Success message, the SCC AS shall treat the ongoing I1 session as established and the existing call being controlled by the respective I1 session.

6.2.5 Service control transfer (Gm fallback to I1)

6.2.5.1 General

When the Gm reference point is used for service control signalling, a change of access network due to handover (e.g. as described in 3GPP TS 23.009 [10] and 3GPP TS 25.413 [11]) may result in an inability to use the PS access for the Gm reference point. In this case, if the I1 interface in the target access network is available and supported, the service continuity may be maintained by switching the service control signalling from the Gm reference point to the I1 interface.

If the ICS UE discovers that the Gm reference point is not available for an ongoing session that is using a CS bearer which was established over the respective Gm reference point, the ICS UE can transfer the service control signalling from the Gm reference point to the I1 interface, if the I1 interface is available, while retaining the existing CS bearer (i.e. the existing CS bearer is left intact). However, if prior to the change of the access network, the UE was not attached to the CS domain and a PS bearer was used for either the voice media or voice and video media of the IMS session, then the service continuity may be maintained by switching the service control signalling from the Gm reference point to the I1 interface and transferring the voice media or voice and video media from the PS bearer to the newly-established CS bearer.

6.2.5.2 Service continuity while retaining the use of CS access for media

6.2.5.2.1 Detailed behaviour of ICS UE

When the ICS UE, that has an established IMS session that is using the CS media, originates a service control transfer from the Gm reference point to the I1 reference point while retaining the existing CS bearer and associated media intact, the ICS UE shall behavemessage as specified in the subclause 6.2.1.2.1 with the following additions:

1) include a Replaces information element in the I1 Invite (Augmentation) message that contains a STI. The STI identifies the SIP dialog that was previously established over the Gm reference point on the Source Access Leg and will be transferred to this I1 session on the Target Access Leg.

2) if the To-id information element is included, and the public user identity inserted in the To-id information element is not an E.164 number, then indicate that the public user identity and the To-id information element are in the correlated SIP INVITE request, by setting the To-id Information IE (see table 7.3.2.2) Code Specific

Information element to "Unspecified" (see table 7.4.2.3A.1) and the length IE is set to 1 and octet 3 is set to all "0"s.

NOTE 1: In this case, some I1 information elements (e.g. Privacy) can be omitted from the I1 Invite message, since this information is already known to the SCC AS from the ongoing SIP dialog that was previously established over the Gm reference point on the Source Access Leg. For example, the inclusion of SIP URI into the To-id and From-id information elements is not needed since these information elements may be omitted from the I1 Invite message.

NOTE 2: It is assumed that when the SIP dialog was established over the Gm reference point, the respective STI was used to identify this SIP dialog.

Upon receiving the I1 Progress message from the SCC AS, the UE shall not initiates the call setup over the CS domain by sending a CC SETUP message to the MSC Server, since the I1 session will inherit the existing CS media (i.e. the existing CS bearer is left intact).

When the ICS UE receives an I1 Success message from the SCC AS, the UE shall consider the service control signalling as being transferred from the Gm reference point to the I1interface and the associated CS media as being transferred to the I1 session (i.e. the I1 session is now controlling the inherited CS media). Furthermore, the UE shall considered the SIP dialog on the Source Access Leg that was originally set using the Gm reference point and all remaining PS media associated with this SIP dialog (i.e. the PS media that were not transferred), if any, as terminated.

NOTE 3: If the UE is incapable of simultaneously communicating over the Gm reference point on the Source Access Leg and the I1 interface over the Target Access Leg, the UE will not receive a SIP BYE request from the ICS AS sent over the Gm reference point on the Source Access Leg.

NOTE 4: Irrespective whether the UE receives a SIP BYE request over Gm reference point on the Source Access Leg or not, the UE will consider the SIP dialog on the Source Access Leg and all remaining PS media associated with this SIP dialog (i.e. the PS media that were not transferred), if any, as terminated.

6.2.5.2.2 Detailed behaviour of SCC AS

If the SCC AS, that supports the I1 protocol, receives an initial I1 Invite message with a Replaces information element that contains a STI, the SCC AS shall use the STI to identify an existing SIP dialog that was previously established using the Gm reference point on the Source Access Leg, and will be replaced with the I1 session on the Target Access Leg. If the identified SIP dialog on the Source Access Leg is currently using a CS bearer, the SCC AS shall behave as specified in subclause 6.2.1.3.1 with the following addition:

1) interpret the received I1 Invite message containing the Replaces information element as request for service control transfer from Gm reference point on the Source Access Leg to the I1 interface on the Target Access Leg;

2) correlate the I1 Invite message with the existing SIP dialog that is using a CS bearer, based on the STI included in the Replaces information element;

NOTE 1: In this case, some information elements (e.g. To-id, From-id, Privacy) may not be included in the I1 Invite message. The omitted I1 information elements are already known to the SCC AS from the ongoing SIP dialog that was previously established over the Gm reference point on the Source Access Leg.

3) send the I1 Progress message towards the originating UE that does not include an allocated SCC AS PSI DN;

NOTE 2: Upon sending the I1 Progress message towards the originating UE, the SCC AS will not receive an initial SIP INVITE request from the CS domain, since the existing CS media will be left intact and only the control will be transferred from the SIP dialog identified by the STI in the received Replaces information element to the I1 session being established.

4) examine whether the SIP dialog on the Source Access Leg has a single CS bearer (i.e. no PS bearers) associated with this SIP dialog, or there are additional PS bearers (in addition to the CS bearer) associated with this SIP dialog.

   a) if there is a single CS bearer and no PS bearers associated with this SIP dialog, the SCC AS proceeds with the steps below, starting with the step 5; or

NOTE 3: In spite of the service control being transferred from the Gm reference point to the I1 interface, there is no need to update the remote UE by sending a new SDP offer since the CS media has been left intact.

b) if, in addition to a CS bearer, there are additional PS bearers associated with this SIP dialog, the SCC AS shall proceed as follows:

i) send a SIP re-INVITE request toward the CS domain (e.g. MGCF) that does not contain an SDP offer;

ii) upon receiving an SDP offer from the CS domain (in the response to the SIP re-INVITE request), the SCC AS update the remote UE by sending a SIP re-INVITE request toward the the remote UE. The SDP offer included in the SIP re-INVITE request sent toward the the remote UE contains the information received in the SDP offer from the CS domain and terminates all the PS bearers, as per standard SIP procedures;

iii) upon receiving the SDP answer in the response to the SIP re-INVITE request from the remote UE, the SCC AS sends an SIP ACK toward the CS domain (e.g. MGCF) that contains an SDP anwer. The SDP answer contains the information obtained from the SDP answer conveyed in the response to the SIP re-INVITE request received from the remote UE. In addition, the SCC AS sends a SIP ACK toward the remote UE;

iv) proceeds with the steps below;

5) release the SIP dialog on the Source Access Leg by sending a SIP BYE request via the SIP dialog over the Gm reference point on the Source Access Leg, if the SIP dialog is still active; and

NOTE 4: The SIP dialog may have been released by the IMS core network as specified in 3GPP TS 24.229 [12], subclause 5.2.8.1.2.

NOTE 5: If the UE is incapable of simultaneously communicating over the Gm reference point on the Source Access Leg and the I1 interface over the Target Access Leg, the SCC AS will not receive a 200 (OK) response to a SIP BYE request.

NOTE 6: Irrespective whether the SCC AS receives a 200 (OK) response to the SIP BYE request over the Gm reference point on the Source Access Leg or not, the SCC AS will consider the dialog on the Source Access Leg and all remaining PS media associated with this dialog (i.e. the PS media that were not transferred), if any, as terminated.

6) send an I1 Success message to the UE over the I1 interface. Upon sending the I1 Success message to the UE, the SCC AS shall consider the service control signalling as being transferred from the Gm reference point to the I1 interface and the associated CS media as being transferred to the I1 session (i.e. the I1 protocol is now controlling the inherited CS media).

6.2.5.3 Service continuity when transferring from PS access to CS access

When an UE, that has an established SIP dialog that is using only the PS media (i.e. no CS media) and the Gm reference point for service control signalling (e.g. the UE is not attached to the CS domain), determines that the Gm reference point is not anymore available, the UE may maintain service continuity by switching the service control signalling from the Gm reference point to the I1 reference point and an associated PS bearer to the CS bearer.

The I1 protocol is used to transfer either a single PS voice media or a single PS voice and PS video media session to a single CS bearer. If there are more then one active PS voice media or PS voice media and PS video media associated with the SIP dialog being transferred, then the last-established active PS voice media or PS voice and PS video media will be transferred from the PS domain to the CS domain. If there are only inactive PS voice media or PS voice and PS video media associated with the SIP dialog, then the last-established inactive PS voice media or PS voice and PS video media will be transferred from the PS domain to the CS domain. In either case, the SIP dialog and all associated PS media that were not transferred are terminated.

If the transferred media was active prior to the transfer, it shall stay active upon the completion of the transfer procedure. Likewise, if the transferred media was inactive prior to the transfer, it shall stay inactive upon the completion of the transfer procedure.

6.2.5.3.1 Detailed behaviour of UE

When the UE, that has an established SIP dialog that is using only the PS media (i.e. no CS media), transfers the service control signalling from the Gm reference point to the I1interface and either the voice media or the voice and video media from the PS access to the CS access, the UE behave as specified in subclause 6.2.1.2.1 with the following additions:

1) include a Replaces information element in the I1 Invite message that contains the STI. The STI identifies the SIP dialog that was previously established over the Gm reference point on the Source Access Leg and will be transferred to the I1 session on the Target Access Leg.

NOTE 1: In this case, some I1 information elements (e.g. To-id, From-id, Privacy) can be omitted from the I1 Invite message, since this information is already known to the SCC AS from the ongoing SIP dialog that was previously established over the Gm reference point on the Source Access Leg. For example, the inclusion of SIP URI into the To-id and From-id information elements is not needed since these information elements may be omitted from the I1 Invite message.

NOTE 2: It is assumed that when the SIP dialog was established over the Gm reference point, the respective STI was used to identify this SIP dialog.

When the UE receives an I1 Progress message from the SCC AS that contains an IUA PSI DN, the UE shall initiates a call over the CS domain using the received IUA PSI DN, as specified in subclause 6.2.1.2.1.

When the UE receives an I1 Success message from the SCC AS, the UE shall consider the service control signalling as being transferred from the Gm reference point to the I1interface and the associated voice media or voice and video media as been transferred from the PS domain to the CS domain. Furthermore, the UE shall considered the SIP dialog on the Source Access Leg that was originally set using the Gm interface and all remaining PS media (i.e. the PS media that were not transferred), if any, associated with this SIP dialog as terminated.

NOTE 3: If the UE is incapable of simultaneously communicating over the Gm reference point on the Source Access Leg and the I1 interface over the Target Access Leg, the UE will not receive a SIP BYE request from the SCC AS over the Source Access Leg.

NOTE 4: Irrespective whether the UE receives a SIP BYE request over the Gm reference on the Source Access Leg or not, the UE will consider the dialog on the Source Access Leg and all remaining PS media associated with this SIP dialog that were not transferred, if any, as terminated.

6.2.5.3.2 Detailed behaviour of SCC AS

If the SCC AS that supports the I1 protocol receives an initial I1 Invite message with a Replaces information element that contains a STI, the SCC AS shall use the STI to identify an existing SIP dialog that was previously established using the Gm reference point on the Source Access Leg and will be replaced with the I1 session on the Target Access Leg. If the identified SIP dialog on the Source Access Leg is currently using only PS media, the SCC AS shall behave as specified in subclause 6.2.1.3.1 with the following addition:

1) interpret the received I1 Invite message containing the Replaces information element as request for service control transfer from Gm reference point on the Source Access Leg to the I1 interface on the Target Access Leg;

2) correlate the I1 Invite message to an existing SIP dialog (and is using only PS bearers), based on the STI received in the Replaces information element, and select a PS bearer that is using either a voice media or voice and video media and that will be transferred to the CS bearer;

NOTE 1: If there are more then one active PS voice media or PS voice media and video media associated with the SIP dialog, the SCC AS selects the last-established active PS voice media or PS voice and video media. If there are only inactive PS voice media or PS voice and video media associated with the SIP dialog, then the SCC AS selects the last-established inactive PS voice media or PS voice and video media.

3) send the I1 Progress message towards the originating UE that includes an allocated SCC AS PSI DN, as specified in subclause 6.2.1.3.1;

Upon receiving the initial SIP INVITE request from the CS domain, the SCC AS shall updates the Remote Leg by sending an SIP re-INVITE request toward the remote UE that include an SDP offer. The SDP offer included in the SIP re-INVITE request sent toward the the remote UE specifies which media is being transferred to the CS domain, and which PS media, if any, are being terminated. When generating the SDP offer towards the remote UE, the SCC AS shall use the information received in the SDP offer in the SIP INVITE request received from the CS domain and terminates all the PS bearers that have not been transferred to the CS domain, as per standard SIP procedures.

Upon receiving a SIP 200 (OK) response from the remote UE that contains an SDP answer, the SCC AS shall send the SIP 200 (OK) response towards the CS domain that includes a SDP answer. The SDP answer sent towards the CS domain contains the media information that pertains to the voice media or voice and video that has been received in the SDP answer from the remote UE and is being transferred to the CS domain.

Upon receving a SIP ACK request from the CS domain, the SCC AS shall send a SIP ACK tovard the remote UE, and:

- release the SIP dialog on the Source Access Leg by sending a SIP BYE request via the SIP dialog over the Gm reference point on the Source Access Leg, if the SIP dialog is still active; and
- send an I1 Success message toward the UE over the I1 interface.

NOTE 2: The SIP dialog may have been released by the IMS core network as specified in 3GPP TS 24.229 [12], subclause 5.2.8.1.2.

Upon sending the I1 Success message to the UE, the SCC AS shall consider the service control signalling as being transferred from the Gm interface to the I1interface and the associated CS media as being transferred to the I1 session (i.e. the I1 protocol is now controlling the transferred CS media).

NOTE 3: If the UE is incapable of simultaneously communicating over the Gm reference point on the Source Access Leg and the I1 interface over the Target Access Leg, the SCC AS will not receive a 200 (OK) response to the SIP BYE request from the UE.

NOTE 4: Irrespective whether the SCC AS receives a 200 (OK) response to the SIP BYE request over the Gm reference point on the Source Access Leg or not, the SCC AS will consider the dialog on the Source Access Leg and all non-transferred PS media associated with this dialog, if any, as terminated.

6.2.5.5 Service continuity after transferring multiple calls from PS access to CS access using SRVCC

6.2.5.5.1 General

The ICS UE, that has previously used the Gm reference point to establish multiple active or held calls that are currently using a single CS bearer as a result of completion of SRVCC procedure, shall transfer the service control signalling from the Gm reference point to the I1 interface for each call while retaining the existing single CS bearer.

6.2.5.5.2 Detailed behaviour of ICS UE

The ICS UE shall follow the procedures as specified in subclause 6.2.5.2.1 for each call.

6.2.5.5.3 Detailed behaviour of SCC AS

The SCC AS shall follow the procedures as specified in subclause 6.2.5.2.2 for each call.

6.2.6 Assignment of single CS bearer to I1 sessions

6.2.6.1 General

When the ICS UE wants to set up a new call using the I1 protocol, and if there is an existing CS bearer leg (i.e. between the ICS UE the MSC/MGCF) that has been previously establish either by the ICS UE or the SCC AS, and there are no active I1 sessions using the existing CS bearer leg (all I1 sessions using the existing CS bearer leg are on hold), the ICS UE can use the existing CS bearer leg for the new call. Likewise, for the incoming call destined for the serving ICS UE, the SCC AS can indicate to the ICS UE to use an existing CS bearer leg for the ICS UE terminated call, if there are no active I1 sessions using the existing CS bearer leg (all I1 sessions associated with the existing CS bearer leg are on hold). In either case, once the new I1 session is set up, the existing CS bearer leg (i.e. between the ICS UE the MSC/MGCF) will be concatenated with a new IP bearer leg to form an end-to-end connection for media for the new call between the ISC ICS UE and the remote endpoint.

When either ICS UE or the SCC AS initiate the set up of an I1 session that will use an existing CS bearer leg (i.e. between the ICS UE the MSC/MGCF) for media, it is assumed that there is a single existing CS bearer leg between the ICS UE and the MSC/MGCF. Hence, neither the ICS UE nor the SCC AS will have to explicitly specify which existing CS bearer leg will be used for the call. The ICS UE and the SCC AS implicitly assume that the single existing CS bearer leg is used for the new call being set up. When a call that was on hold is resumed, the SCC AS ensure that the respective end-to-end connection for media associated with the resumed call is also restored, i.e. the existing CS bearer leg is re-allocated to the resumed call and re-concatenated to the IP leg associated with the resumed call.

6.2.6.2 Originating call behaviour

6.2.6.2.1 Detailed behaviour of ICS UE

Prior to initiating the establishment of a new I1 session as described in this subclause, the ICS UE ensures that there is only one existing CS bearer leg between the ICS UE and the MSC/MGCF. If there is either none or more than one existing CS bearer leg between the ICS UE and the MSC/MGCF, the ICS UE shall not initiate the establishment of an I1 session as described in this subclause.

When the ICS UE wants to establish a new I1 session that uses an existing CS bearer leg between the ICS UE and the MSC/MGCF, the ICS UE shall send an I1 Invite message toward the SCC AS, and populate the I1 Invite message as specified in subclause 6.2.1.2.1.2 with the following additions:

1) set the Reason in the I1 Invite message to value hex "003", as specified in table 7.3.1. This value indicates to the SCC AS that the existing CS bearer leg will be used for this call.

If the ICS UE receiving an I1 Progress message with Progress reason set to Call progressing from the SCC AS, the ICS UE shall behave as specified in subclause 6.2.1.2.1.3 with the following addition:

1) ignore the SCC AS PSI DN, if included in the I1 Progress message with Progress reason set to Call progressing received from the SCC AS, i.e. the ICS UE shall not initiate a call toward the CS domain.

NOTE 1: The ICS UE will not initiates the call setup over the CS domain for the purpose of setting up a new CS bearer leg, since the existing CS bearer leg will be used for the call being established.

When the ICS UE receives an I1 Progress message with Progress reason set to 180 the ICS UE shall:

1) provide an alerting indication to the user.

NOTE 2: If in-band alerting is not provided via the CS bearer, the ICS UE will provide local alerting.

When the ICS UE receives an I1 Success message from the SCC AS, the ICS UE shall behave as specified in subclause 6.2.1.2.1.5 with the following addition:

1) the ICS UE shall not expect to receive a CONNECT message from the CS domain as specified in 3GPP TS 24.008 [3] since it did not send a SETUP message toward the CS domain; and

2) consider the I1 session as being established and the existing CS call leg as being assigned to this I1 session.

6.2.6.2.2 Detailed behaviour of SCC AS

Prior to initiating the establishment of an I1 session as described in this subclause, the SCC AS shall insure that there is only one existing CS bearer leg between the ICS UE and the MSC/MGCF. If the SCC AS receives an I1 Invite message from the ICS UE that includes a Bearer information element that contains a Reason field set to value hex "003", as specified in table 7.3.1, and if there is either none or more than one existing CS bearer leg between the ICS UE and the MSC/MGCF, the SCC AS shall reject the request for the I1 session establishment.

If a new call request destined for the ICS UE arrives at the SCC AS during the establishment of an I1 session as described in this subclause, the SCC AS shall reject this request by immediately responding with a busy indication to the new incoming call.

If the SCC AS receives an initial I1 Invite message with the Reason set to value hex "003", as specified in table 7.3.1 that indicates to the SCC AS that the existing CS bearer leg will be used for this call, and there is only one existing CS bearer leg between the ICS UE and the MSC/MGCF, the SCC AS shall behave as specified in subclause 6.2.1.3.1.2 with the following addition:

1) interpret the received I1 Invite message with the Reason set to value hex "003" as a request to use the existing CS bearer leg when setting up an end-to-end connection for media toward the called user.

The SCC AS may send the I1 Progress message with Progress reason set to Call progressing towards the originating ICS UE. If the SCC AS sends the I1 progress message toward the originating ICS UE, the SCC AS shall populate the I1 Progress message with Progress reason set to Call progressing as specified in subclause 6.2.1.3.1.3 with the following addition:

1) the SCC AS shall not include the SCC-AS-id information element that contains an allocated SCC AS PSI DN in the I1 Progress message with Progress reason set to Call progressing.

NOTE 1: The reason for sending the I1 Progress message with Progress reason set to Call progressing, is to increase the retransmissions timer at the ICS UE, if an unreliable transport-layer connection is used (see subclause 7.5.3.2.1.1.2).

Subsequently, the SCC AS shall:

1) send a SIP re-INVITE request toward the CS domain (i.e. the MGCF) that does not contain an SDP offer; and

2) upon receiving a SDP offer from the CS domain (in the response to the SIP re-INVITE request), send an initial SIP INVITE request toward the the remote UE. The SDP offer included in the SIP INVITE request sent toward the the remote UE contains the information in the SDP offer received from the CS domain.

Upon receiving the SIP 180 (Ringing) response from the remote UE, the SCC AS shall send an I1 Progress message with Progress reason set to 180 towards the originating UE, as specified in subclause 6.2.1.3.1.4 with the following addition:

1) if this is the first I1 Progress message (i.e. the SCC AS did not previously sent an I1 Progress message with Progress reason set to Call progressing), the SCC AS shall also include (in the I1 Progress message with Progress reason set to 180) all information elements as specified in subclause 6.2.1.3.1.3, except the SCC AS PSI DN.

Upon receiving a SIP 200 (OK) response from the remote UE, the SCC AS shall:

1) send the SIP ACK request towards the CS domain that includes a SDP answer received from the remote UE. The SDP answer sent towards the CS domain contains the media information that has been received in the SDP answer from the remote UE; and

2) send an I1 Success message toward the ICS UE over the I1 interface as specified in the subclause 6.2.1.3.1.5, and a SIP ACK toward the remote UE.

6.2.6.3 Terminating call behaviour

6.2.6.3.1 Detailed behaviour of ICS UE

Prior to accepting a request for an I1 session as described in this subclause, the ICS UE ensures that there is only one existing CS bearer leg between the ICS UE and the MSC/MGCF. If there is either none or more than one existing CS bearer leg between the ICS UE and the MSC/MGCF, the ICS UE shall reject the request for the I1 session establishment.

If the ICS UE receives an initial I1 Invite message with the Reason set to value hex "003", as specified in table 7.3.1 that indicates to the ICS UE that the existing CS bearer leg will be used for this call, and there is only one existing CS bearer leg between the ICS UE and the MSC/MGCF, the ICS UE shall behave as specified in subclause 6.2.1.2.2 with the following addition:

1) interpret the received I1 Invite message with the Reason set to value hex "003" as a request to use the existing CS bearer leg for this call; and

2) ignore the SCC AS PSI DN, if included in the I1 Invite message received from the SCC AS, i.e. the ICS UE shall not initiate a call toward the CS domain.

NOTE 1: The ICS UE will not initiates the call setup toward the CS domain for the purpose of setting up a new CS bearer leg, since the existing CS bearer leg will be used for the call being established.

Subsequently, the ICS UE shall allert the user and generate an I1 Progress message with Progress reason set to 180 and send it towards the SCC AS, as specified in subclause 6.2.1.2.2.

If the user accepts the call, the ICS UE shall generate an I1 Success message and send it towards the SCC AS, as specified in subclause 6.2.1.2.2.

6.2.6.3.2 Detailed behaviour of SCC AS

Prior to initiating the establishment of an I1 session toward the ICS UE as described in this subclause, the SCC AS ensures that there is only one existing CS bearer leg between the ICS UE and the MSC/MGCF. If there is either none or more than one existing CS bearer leg between the ICS UE and the MSC/MGCF, the SCC AS shall not initiate the establishment of an I1 session toward the ICS UE as described in this subclause.

If a new call request destined for the ICS UE arrives at the SCC AS during the establishment of an I1 session as described in this subclause, the SCC AS shall reject this request by immediately responding with a busy indication to the new incoming call.

When the SCC AS, upon receiving an initial SIP INVITE request from the remote UE destined for the served ICS UE, wants to establish a new I1 session toward the ICS UE that uses an existing CS bearer leg between the ICS UE and the MSC/MGCF, the SCC AS shall send an I1 Invite message toward the ICS UE. The SCC AS shall populate the I1 Invite message as specified in subclause 6.2.1.2.23.2.1 with the following additions:

1) set the Reason in the I1 Invite message to value hex "003", as specified in table 7.3.1. This value indicates to the ICS UE that the existing CS bearer leg will be used for this call;

2) not include the SCC-AS-id information element that contains an allocated SCC AS PSI DN in the I1 Invite message; and

3) send a SIP re-INVITE request toward the CS domain (i.e. MSC/MGCF) that contains an SDP offer received in a SIP INVITE request from remote UE.

When the SCC AS received an I1 Progress with Progress reason set to 180 from the UE, the SCC AS shall send a SIP 180 (Ringing) response to the remote UE.

When the SCC AS receives SIP 200 (OK) response from the CS domain that contains an SDP answer and an I1 Success message from the ICS UE, the SCC AS shall send a SIP 200 (OK) response to the remote UE that contains an SDP answer received from the CS domain. At this stage the SCC AS considers the I1 session as being established and an end-to-end bearer being set up (i.e. the CS call leg as being assigned to this I1 session).

6.3 Supplementary services control procedures

6.3.0 Introduction

Once the UE invokes a mid-call supplementary service from the MSC (via the CS bearer leg) using existing the CS procedures for a call that was previously established without using either the Gm or the I1 interfaces, then for all subsequent calls the UE and SCC AS shall invoke the mid-call supplementary services from the MSC (via the bearer leg) using existing the CS procedures, until all calls have been cleared.

Once the UE and the SCC AS invoke a mid-call supplementary service using the I1 protocol for a call that was previously established without using either the Gm or the I1 interfaces, and the I1 call control is maintained (the control is not transferred to Gm interface), then for all subsequent calls that are established without using either the Gm or the I1 interfaces, the UE and SCC AS shall invoke the mid-call supplementary services using the I1 protocol, until all calls have been cleared.

If the UE and the SCC AS invoke a mid-call supplementary service for a call that was established using the Gm reference point, and if (due to service continuity procedures) the call control is transferred to the I1 control, then the I1 protocol may be used to provide subsequent mid-call supplementary services for this call (e.g. a voice call that was placed on hold using Gm interface may be resumed using the I1 interface).

If the UE and the SCC AS invoke a mid-call supplementary service using the I1 protocol, and if (due to service continuity procedures) the call control is transferred to the Gm interface, then the Gm interface may be used to provide subsequent mid-call supplementary services for this call (e.g. a voice call that was placed on hold using the I1 protocol may be resumed using the I1 interface).

6.3.1 Line ID Services (OIP, OIR, TIP, TIR)

6.3.1.1 Originating Identity Presentation (OIP)

The procedures in subclause 6.2.1.2.1 apply. The From-id information element is used to present the originating identity.

6.3.1.2 Originating Identity Restriction (OIR)

The procedures in subclause 6.2.1.2.1 apply with following addition:

1) a Privacy information element that indicates the ICS UE wants to restrict the presentation of the originating identity.

6.3.1.3 Terminating Identity Presentation (TIP)

The procedures of sending an I1 Success message towards the orginating UE in subclause 6.2.1.3.1 apply with following addition:

1) a To-id information element that includes either a SIP URI or an E.164 number, and will be used to present the terminating identity.

6.3.1.4 Terminating Identity Restriction (TIP)

The procedures of sending an I1 Success message towards the orginating UE in subclause 6.2.1.3.1 apply without a To-id information element.

6.3.2 Communication diversion services (CDIV)

6.3.2.1 Communication Forwarding Unconditional (CFU)

No specific I1 related messages.

6.3.2.2 Communication Forwarding on Not Logged-in (CFNL)

No specific I1 related messages.

6.3.2.3 Communication Forwarding Busy (CFB)

If the ICS UE receives an I1 Invite message from the SCC AS, and the UE determines that the user is busy, the ICS UE shall:

1) generate an I1 Failure message that includes:

   a) a Message Type set to indicate that this is an I1 Failure message;

   b) a new value in the Call-Identifier (Part-1) subfield, as specified in subclause 7.2.2. The Call-Identifier will uniquely identify this I1 session between the ICS UE and the SCC AS;

NOTE 1: A new value in the Call-Identifier (Part-1) subfield is inserted only if this is the first I1 message sent to the SCC AS. Otherwise the previously set Call-Identifier value is used.

   c) increment the stored message sequence value, store it, and include it in the Sequence-ID; and

   d) set the Error-Code information element to 486; and

2) send the I1 Failure message towards the SCC AS over the transport layer connection over which the I1 Invite message was received.

6.3.2.4 Communication Forwarding No Reply (CFNR)

No specific I1 related messages.

6.3.2.5 Communication Forwarding on Subscriber Not Reachable (CFNRc)

If the ICS UE receives an I1 Invite message from the SCC AS, and the UE determines that the user is busy, the ICS UE shall:

1) generate an I1 Failure message that includes:

   a) a Message Type set to indicate that this is an I1 Failure message;

   b) a new value in the Call-Identifier (Part-1) subfield, as specified in subclause 7.2.2. The Call-Identifier will uniquely identify this I1 session between the ICS UE and the SCC AS;

NOTE 1: A new value in the Call-Identifier (Part-1) subfield is inserted only if this is the first I1 message sent to the SCC AS. Otherwise the previously set Call-Identifier value is used.

   c) increment the stored message sequence value, store it, and include it in the Sequence-ID; and

   d) set the Error-Code information element to 480; and

2) send the I1 Failure message towards the SCC AS over the transport layer connection over which the I1 Invite message was received.

6.3.2.6 Communication Deflection (CD)

If the ICS UE receives an I1 Invite message from the SCC AS, and the UE determines that deflect the call, the ICS UE shall:

1) generate an I1 Redirection message that includes:

   a) a Message Type set to indicate that this is an I1 Failure message;

   b) a new value in the Call-Identifier (Part-1) subfield, as specified in subclause 7.2.2. The Call-Identifier will uniquely identify this I1 session between the ICS UE and the SCC AS;

NOTE 1: A new value in the Call-Identifier (Part-1) subfield is inserted only if this is the first I1 message sent to the SCC AS. Otherwise the previously set Call-Identifier value is used.

   c) increment the stored message sequence value, store it, and include it in the Sequence-ID; and

   d) To-id information element set to either a SIP URI or an E.164 of the C-party identity; and

2) send the I1 Redirection message towards the SCC AS over the transport layer connection over which the I1 Invite message was received.

6.3.2.7 Communication Diversion Notification (CDIVN)

If the SCC AS wants to notify the ICS UE that the call was diverted, the SCC AS shall:

1) generate an I1 Notify message that includes:

   a) a Message Type set to indicate that this is an I1 Notify message;

   b) increment the stored message sequence value, store it, and include it in the Sequence-ID; and

   c) a Mid-call information element that indicates that the call was diverted; and

2) send the I1 Notity message towards the ICS UE over the transport layer connection over which other I1 message was received.

6.3.3 Communication Barring

No specific I1 related messages.

6.3.4 Communication Hold (HOLD)/Resume

6.3.4.1 Actions at the ICS UE

When the ICS UE wants to hold/resume an I1 session using an I1reference point, the UE shall:

1) generate an I1 Mid call Request message that includes:

   a) a Message Type set to the value that indicates that this is an I1 Mid call Request message, as specified in table 7.3.1;

   b) increment the stored message sequence value, store it, and include it in the Sequence-ID field; and

   c) set the Mid-call information element that indicates the ICS UE wants to either hold or resume the I1 session as specified in table 7.4.2.12.1; and

2) forward the I1 Mid call Request message toward the SCC AS.

Upon receiving an I1 Success message from the SCC AS, the ICS UE shall consider that the I1 session has been either placed on hold or resumed, as requested.

If the ICS UE receiving an I1Mid call Request message with the Mid-call infromation element, that indicates that the I1 session was either placed on hold or resumed, the ICS UE shall:

1) store the information received in the I1 Mid call Request message;

2) generate an I1 Success message containing the following information:

   a) a Message type field set to the value that indicates that is an I1 Success message;

   b) the stored Call-ID header value;

   c) add one to the stored Sequence-ID field value. Store it, and include it in the Sequence-ID header value; and

3) send the I1 Success message towards the SCC AS over the transport layer connection over which the I1 Mid Call Request message was received.

6.3.4.2 Actions at the SCC AS

Upon receiving an I1Mid call Request message with a Mid-call infromation element, that indicates the I1 session to be either held or resumed, from the ICS UE via the I1 reference point, the SCC AS shall:

1) store the information received in the I1 Mid call Request message;

2) perform the standard SIP procedures toward the far end and the MGCF in order to either inactive or active the RTP media.

Upon remote UE accepting the inactivation or activation of the RTP media, the SCC AS shall:

1) generate an I1 Success message containing the following information:

   a) a Message type field set to the value that indicates that is an I1 Success message;

   b) the stored Call-ID header value;

   c) add one to the stored Sequence-ID field value. Store it, and include it in the Sequence-ID header value; and

2) send the I1 Success message towards the originating ICS UE over the transport layer connection over which the I1 Mid Call Request message was received.

When the SCC AS wants to hold/resume an I1 session using an I1reference point, the SCC AS shall:

1) inactivating or activating the RTP media by performing the standard SIP procedures toward the far end and the MGCF);

2) generate an I1 Mid call Request message that includes:

   a) a Message Type set to the value that indicates that this is an I1 Mid call Request message, as specified in table 7.3.1;

   b) increment the stored message sequence value, store it, and include it in the Sequence-ID field; and

   c) set the Mid-call information element that indicates the I1 session was either placed on hold or resumed;

3) forward the I1 Mid call Request message toward the ICS UE.

Upon receiving an I1 Success message from the ICS UE, the SCC AS shall consider that the I1 session has been either placed on hold or resumed, as requested.

6.3.5 Consultative Explicit Communication Transfer

6.3.5.1 Actions at the ICS UE

When ICS UE A is playing the role of transfer, the ICS UE shall:

1) generate an I1 Refer message that includes:

   a) a Message Type set to indicates that this is an I1 Refer message;

   b) increment the stored message sequence value, store it, and include it in the Sequence-ID; and

   c) a Mid-call information element that indicates that this is a conference invitation; and

2) forward the I1 Mid Call message toward the SCC AS.

6.3.5.2 Actions at the SCC AS

Upon receiving an I1Mid call request message with a Mid call informtion element indicating this is a conference invitation from the ICS UE via the I1 reference point, the SCC AS shall continue session establishment towards the conference AS as specified in 3GPP TS 24.173 [18].

Upon receiving a SIP 200 OK response from conference AS, the SCC AS shall:

1). generate an I1 Success message containing the following information:

   a) a Message Type set to indicate that is an I1 Success message; and

   b) a Call-Identifier field containing the Call-Identifier value that uniquely identifies this I1 session between the UE and SCC AS;

2) add one to the stored message sequence value. Store and include the Sequence-ID; and

3) send the I1 Success message towards the originating UE over the transport layer connection over which the I1 Mid call request message was received.

6.3.6 Conference calling (CONF)

6.3.6.0 General

The conference calling (CONF) service consists of conference creation, joining a conference, inviting others to join a conference and leaving a conference, as described in 3GPP TS 24.147 [21] and 3GPP TS 24.605 [22].

6.3.6.1 Actions at the ICS UE

When the ICS UE is creating a conference, the ICS UE shall send an I1 Invite message addressed to the E.164 identity that corresponds to the Conference factory URI. Upon receipt of the I1 Success message, the ICS UE is considered to have joined the conference, in the same manner as receipt of a SIP 200 (OK) response is treated in the procedures described in 3GPP TS 24.147 [21].

When ICS UE is inviting others to join a conference, the ICS UE shall:

1) generate an I1 Refer message that includes:

    a) a Message type set to indicate that this is an I1 Refer message;

    b) a Sequence-ID information element, that includes a message sequence value, having first added one to the stored value, and stored it again;

    c) a To-id information element that includes to the E.164 identity that corresponds to the Conference factory URI; and

    d) a Replaces information element is optionally included, as described for the equivalent usage of the Replaces header field in subclause 4.5.2.1.2 of 3GPP TS 24.605 [22]; and

2) forward the I1 Refer message toward the SCC AS.

When the ICS UE receives an I1 Success message, the UE shall:

1) verify if a I1 session exists for the received Call-Identifier value;

2) verify if a the message is in sequence according to the message sequence number value contained in the Sequence-ID information element; and

3) consider the invitation for another party to join the conference as successful.

When the ICS user would like to leave the conference, the ICS UE shall generate an I1 Bye message.

6.3.6.2 Actions at the SCC AS

Upon receiving an I1 Invite message from the ICS UE on the respective I1 session for the purposes of conference creation, the SCC AS shall follow the procedures in subclause 6.2.1.3.1of this document for session origination.

NOTE: The conference focus AS will provide the conference creation functions described in subclause 5.3.2.3 of 3GPP TS 24.147 [21].

Upon receiving an I1Refer message from the ICS UE on the respective I1 session, the SCC AS shall:

1) generate a SIP REFER request towards the conference focus AS; and

2) upon receipt of a SIP 200 (OK) response in response to the SIP REFER request sent to the conference focus AS, generate an I1 Success message containing the following information:

    a) a Message Type set to indicate that is an I1 Success message; and

    b) a Call-Identifier information element containing the value that uniquely identifies this I1 session between the UE and SCC AS;

    c) a Sequence-ID information element, having first added one to the stored message sequence value and stored it again; and

4) send the I1 Success message towards the originating UE over the transport layer connection over which the I1 Refer message was received.

Upon receipt of an I1 Bye message from the ICS UE via on the respective I1 session for purposes of leaving a conference, the SCC AS shall follow the procedures in subclause 6.2.3.3.3 of this document for session release.

6.3.7 Communication Waiting

6.3.7.1 Actions at the ICS UE

Upon receipt of an I1 Invite with a Reason Value set to 0x005 (CW), the ICS UE shall follow the procedure described in subclause 4.5.5.3.2 of 3GPP TS 24.615 [23], treating the I1 Invite with a Reason value of 0x005 (CW) in the same way as receipt of a SIP INVITE request with an XML body and a Content-Type header field set to "application/vnd.3gpp.cw+xml".

For Case A of subclause 4.5.5.3.3 of 3GPP TS 24.615 [23], the ICS UE shall send a I1 Success message to indicate an answer of the waiting communication. The ICS UE shall follow explicit procedures to hold or release the active session while doing so. For Case B, of subclause 4.5.5.3.3 of 3GPP TS 24.615 [23], the ICS UE shall send an I1 Failure message with the Error-Code information element set to the equivalent of 480 to indicate that the user has not answered the waiting communication.

6.3.7.2 Actions at the SCC AS

Upon receipt of a SIP INVITE request destined for an ICS UE with an XML body and a Content-Type header field set to "application/vnd.3gpp.cw+xml" as described in subclause 4.5.5.2 of 3GPP TS 24.615 [23], the SCC AS shall send the I1 Invite message as specified in subclause 6.2.1.3.2 or subclause 6.2.1.3.3 of this document with an I1 Invite Reason value set to 0x005 (CW).

6.4 SCC AS and ICS UE Time Synchronization

6.4.1 General

In order to detect stale I1 messages transmitted when non-real time transport layer protocols are used, the ICS UE and SCC AS must be synchronized in time. The staleness of the I1 messages can be determined by the following two steps:

1) The ICS UE originates initial time synchronization procedure with the SCC AS. During this procedure both ICS UE and SCC AS receive an initial time of the peer. The time value is measured in seconds. The initial time value is not important as long as the subsequent time measurements are increased accordingly; and

2) An I1 message receiver (ICS UE or SCC AS) compares the time received in an I1 message i.e. the current time of the peer with the initial time established in step 1, and based on the time difference between the current and initial time it makes a decision about the staleness of the message. If the message is stale it shall be discarded and no response generated to the I1 message.

6.4.2 Generating Time

The initial time value can be initialized using one of the following methods:

i) a local time of the machine or terminal;

ii) a randomly generated time; and

ii) zero.

6.4.3 Detailed behaviour of ICS UE

If time synchronization is supported by the UE (i.e. when the I1 messages are transmitted over non-real time transports) the time synchronization procedure shall be initiated by the ICS UE after the non-real time transport layer connection is established. The time synchronization procedure with the SCC AS may be repeated by the ICS UE, if required and supported.

6.4.3.1 ICS UE Synchronization Origination

When the ICS UE initiates the synchronization procedure using an I1reference point, the UE shall:

1) generate an I1 Notify message that includes:

   a) a Message Type set to indicate that this is an I1 Notify message and Reason field in the I1 Notify message set to value hex "001", that indicates that the I1 Notify message is used for synchronization, as specified in table 7.3.1;

   b) a new value in the Call-Identifier field (Part-1), as specified in subclause 7.2.2. The Call-Identifier value will uniquely identify this I1 session between the ICS UE and the SCC AS;

   c) an allocated Sequence-ID;

   d) a From-id information element that includes either a SIP URI or an E.164 number, and it will be used by the SCC AS to identify the ICS UE; and

   e) a Timestamp information element that includes the initial time generated according to the subclause 6.4.2. The element will be used by the SCC AS to validate the ICS UE I1 messages. The Timestamp value is stored by the ICS UE.

2) select the transport layer protocol (see subclause 4.2.3) depending on the access network type, and forward the I1 Notify message toward the SCC AS.

When the ICS UE receives an I1 Success message, the UE shall:

3) verify if a I1 session exists for the received Call-Identifier field value;

4) verify if a the message is in sequence according to the value of the Sequence-ID; and

5) retrieve and store the Timestamp information element value received in the response.

If the ICS UE does not receive an I1 Success message or an I1 Failure message within 64*T1 seconds, the UE shall consider the I1 Notify message as failed and it may attempt to perform the synchronization procedure again after an interval of time. However, if the ICS UE receives an I1 Failure message (with Reason valus set to 488, as specified in table 7.3.1) indicateing the the SCC AS does not support the synchronization procedure, the ICS UE shall not attempt to perform the synchronization procedure again.

6.4.4 Detailed behaviour of SCC AS

6.4.4.1 SCC AS Synchronization Termination

Upon receiving an I1 Notify message with Reason field in the I1Notify message set to value hex "001", as specified in table 7.3.1, from the ICS UE via the I1 reference point, the SCC AS shall either:

A) respond with an I1 Failure message (with Reason valus set to 488, as specified in table 7.3.1), if the SCC AS does not support the synchronization procedure; or

B) if the SCC AS supports the synchronization procedure:

   1) retrieve the ICS UE time value from the Timestamp information element of the I1 Notify message, and store the value;

   2) save the received Call-Identifier field value and Sequence-ID values and use them for further reference to this session;

   3) generate an I1 Success message containing the following information:

      a) a Message Type set to indicate that is an I1 Success message;

      b) a Call-Identifier field with the stored Call-ID field value;

      c) add one to the stored message sequence value. Store and include the Sequence-ID;

      d) include the Timestamp information element that is generated according to the subclause 6.4.2. The element will be used by the ICS UE to validate the SCC AS I1 messages. The Timestamp value is stored by the SCC AS.

4) send the I1 Success message towards the originating UE over the transport layer connection over which the I1 Notify message was received.

7 Protocol specification and implementation

7.1 Overview of I1 protocol functionality

The I1 protocol includes the procedures for establishing, maintaining, and clearing the I1 sessions between the ICS UE and the SCC AS (see figure 7.1).

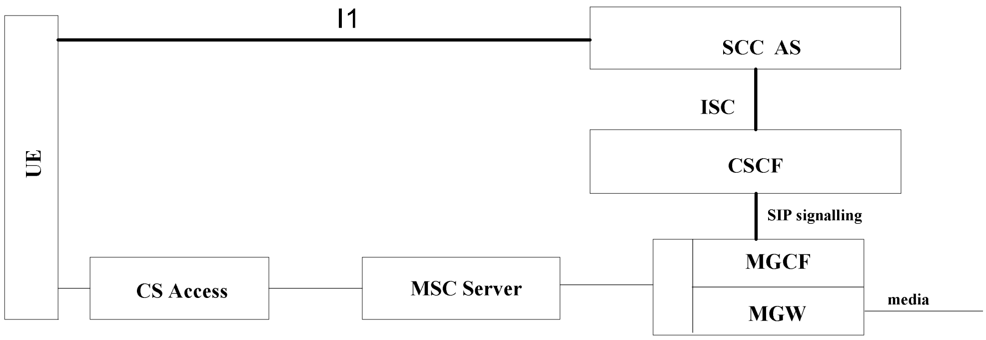

Figure 7.1 UE session signalling and bearer path using I1 interface for Service Control Signalling Path

NOTE 1: Figure 7.1 illustrates an MSC server that is not enhanced for ICS. I1 can also be used when deploying an MSC server enhanced for ICS as specified in 3GPP TS 23.292 [2].

The I1 protocol is a message based point-to-point protocol. The I1 protocol messages are wrapped in a point-to-point transport layer connection protocol (e.g. USSD) and are exchanged between the ICS UE and the SCC AS. Therefore, the I1 protocol does not include any routing capabilities. To address the ICS UE in CS network and establish a transport-layer connection, (the IUA of) the SCC AS shall convert the called party identity (i.e., IMS public user identity) to the CS domain party identity that is required to route the transport layer protocol(i.e., MSISDN, MDN, etc.).

The I1 protocol assumes that there is an associated connection-control protocol that incorporates media negotiation capabilities and provides the setting up and clearing of the connection over which the media will be exchanged. Therefore, any signalling between the UE and the CS access domain (see figure 7.1), as well as the SIP signalling between the MGCF and the SCC AS should be viewed as a procedure to establish a media connection rather then a call control signalling. Obviously, the I1 endpoints will correlate and synchronize the progress of the I1 session establishment and clearing of the I1 session with the associated media-establishing procedures.

NOTE 2: The primitives that are used to communicate between the I1 protocol I1 session entity and the associated connection-control protocol entity, internally in the UE and SCC AS, respectively, are not specified in this document.

The I1 protocol assumes that the application level segmentation of the I1 protocol messages is not supported. The size of the I1 protocol messages is constrained by the limits of the transport-layer message size. For example, USSD allows for a message size of 160 octets. This means that it is not possible to send an I1 protocol message greater than 160 octets, unless message segmentation is designed into the I1 protocol. A USSD dialogue is already segmented by the use of USSD sub-dialogues as a USSD conversation, and this usage of USSD is inappropriate for the I1 protocol.

The I1 protocol is a transport-independent protocol, i.e. the I1 protocol I1 session control entities can exchange the I1 protocol messages over any transport-layer connection that connects the ICS UE and the SCC AS. The ICS UE sends the I1 protocol messages to the SCC AS over a transport-layer connection (e.g. USSD) that the ICS UE knows it will reach the SCC AS. Likewise, The SCC AS sends the I1 protocol messages to the ICS UE over a transport-layer connection (e.g. USSD) that the SCC AS knows that it will reach the ICS UE. For example, the SCC AS forwards the I1 protocol message to the ICS UE over the same transport-layer connection (e.g. USSD) over which it received the previous I1 protocol message from the ICS UE.

The I1 protocol message are self-identifying, i.e. the information contained in the I1 protocol message uniquely identifies the call to which the I1 protocol message pertains to.

The I1 protocol assumes that when the transport-layer connection is established between the UE and SCC, the UE's E.164 number is bound to the respective transport-layer connection at both the UE and SCC AS (e.g. the establishment of an USSD channel can implicitly bind the UE's E.164 number to this transport-layer connection). Subsequently, the request for an I1 session destined for the respective E.164 number will be passed to the UE over the respective transport-layer connection.

NOTE 3: Since the binding between the transport-layer connection and the E.164 number is performed when the transport-layer connection is established, the I1 protocol does not include any registration procedure.

The I1 protocol is a binary-oriented protocol (i.e. the I1 messages are binary encoded). The bit-map tables are used to describe the I1 messages and associated information elements.

In this release of this document, it is assumed that the ICS UE, when establishing a transport-layer connection (e.g. USSD channel) to the SCC AS, will have been authenticated by the CS domain. Due to a relationship between the SCC AS and the CS domain, the ICS UE is not authenticated (e.g. challenged) by the SCC AS when sending any I1 protocol message to the SCC AS. However, the SCC AS will check the UE identity for potential invalid IMS public user identity included by the ICS UE. The CS domain number received from the transport-layer is trustable and will be used by (the IUA of) the SCC AS to check the URI of the UE before the SCC AS provides SIP UA behaviour on behalf of the ICS UE.

7.2 I1-protocol messages and functional definition

7.2.1 I1-protocol messages

7.2.1.1 General

This subclause provides the list of I1-protocol messages (see table 7.2.1) and brief description of each I1-protocol message. Based on their function the I1-protocol messages can be grouped into five categories:

- I1-Session establishment messages;
- I1-Stable Session messages;
- I1-Session clearing messages;
- I1-Error messages;
- I1- Supplementary Service -Invocation messages; and
- I1-Other messages.

Table 7.2.1.1: I1-protocol messages

Message type	Description and content (subclause)
Session establishment messages:	7.2.1.2
I1 Invite message I1 Progress message I1 Success message	
***Stable Session messages*:**	7.2.1.3
I1 Refer message	
***Session clearing messages*:**	7.2.1.4
I1 Bye message I1 Success message	
***Error messages*:**	7.2.1.5
I1 Failure message	
***Supplementary Service Invocation messages*:**	7.2.1.6
I1 Mid Call Request message I1 Redirection message	
Other messages:	7.2.1.7
I1 Notify message	
I1 Dummy message	

7.2.1.2 Session establishment messages

The session establishment I1 messages can be sent either by the ICS UE to the SCC AS or by the SCC AS to the ICS UE.

I1 Invite message

The I1 Invite message is sent either by the calling UE to the SCC AS or by the SCC AS to the called UE to initiate session establishment.

I1 Progress message

The I1 Progress message is a general purpose provisional response, semantically similar to SIP 1xx class responses. The binary Reason field value (per figure 7.3.1) corresponds with the received SIP 1xx response's numeric status-code value.

I1 Success message

The I1 Success message indicates that the action requested in the respective I1 message has been accomplished successfully.

The I1 Success message:

- is transmitted by the SCC AS to the calling UE to indicate that the session has been accepted; or
- is transmitted by the called UE to the SCC AS to indicate that the called UE has accepted the session.

The Reason's corresponding to the I1 Success message are specified in table 7.3.1 and correspond with a SIP 2xx response's numeric status-code.

7.2.1.3 Stable session messages

I1 Refer message

The I1 Refer message is sent either by the ICS UE to the SCC AS or by the SCC AS to the ICS UE to indicate that the recipient of the I1 Refer message should contact the target identified in the I1 Refer message.

7.2.1.4 Session clearing messages

I1 Bye message

The I1 Bye message:

- is transmitted by the SCC AS to the ICS UE to clear the I1 session; or
- is transmitted by the ICS UE to the SCC AS to clear the I1 session.

7.2.1.5 Error messages

I1 Failure message

An I1 Failure response message is sent either by the ICS UE to the SCC AS or by the SCC AS to the ICS UE, to indicate that an error has occurred. The additional parameters included in the I1-Failure message indicate the type of the error that has occurred. The reason value field is a direct one to one mapping to the status code in the status line as specified in subclause 7.2 of RFC 3261 [6].

7.2.1.6 Supplementary Services Invocation related messages

The following section details the messages that are used to request the invocation of a supplementary service.

I1 Mid Call Request message

The I1 Mid Call Message is used for the invocation of mid-call supplementary services. For example: user wishes to hold or resume a call.

I1 Redirection message

The I1 Redirection message is used by the ICS UE to inform the SCC AS of the desire to invoke a supplementary service, in response to incoming signalling. For example: desire for the called user to deflect the call to a third party. The SCC AS interworks the I1 response to the appropriate SIP response to send to the SIP AS.

7.2.1.7 Other messages

I1 Notify message

The I1 Notify message may be used to notifyeither by the ICS UE or the SCC AS to inform its peer about some event that has occurred or to convey some information to its peer, e.g.of events related to the invocation of supplementary services. For example:

- Notification that a call has been forwarded to a third party;
- Notifications related to Explicit Call Transfer; or
- Notifications related to requests to join a Conference.

I1 Dummy message

The I1 Dummy message is only used for those specific transport-layer connections (e.g., USSD) which provide two-way-alternative interactive service. If the party which has the turn hasn't sent an application level protocol message for a specific time, an I1 Dummy message shall be delivered to the counterpart to transfer the turn with the consideration of not delaying its transmission of application protocol message.

7.2.2 I1 message structure and common field encoding

7.2.2.1 General

7.2.2.1.1 Message Header structure

The I1 message structure is shown in figure 7.2.2.1. Each I1 messages consists of two parts, i.e. the first part referred to as the I1 message common part and the second part consisting of zero or more I1 information elements. The I1 message common part is included in every I1 message. The I1 information elements that are included in an I1 message depend on a type of I1 message being sent.

The text in this clause describes the content of the I1 message common part. The octet number 1 (shown at the top of the figure 7.2.2.1) is sent first followed by octet 2, 3, etc. Within each octet, the bit designated "bit 1" is transmitted first, followed by bits 2, 3, 4, etc.

<table>
<tr><td>8</td><td>7</td><td>6</td><td>5</td><td>4</td><td>3</td><td>2</td><td>1</td><td>Octet</td></tr>
<tr><td colspan="4">Protocol version number</td><td colspan="4">Protocol identifier</td><td>1</td></tr>
<tr><td colspan="5">Message type</td><td>R</td><td colspan="2">Reason</td><td>2</td></tr>
<tr><td colspan="8">Reason</td><td>3</td></tr>
<tr><td colspan="8">Call-Identifier (Part-1)</td><td>4</td></tr>
<tr><td colspan="8">Call-Identifier (Part-2)</td><td>5</td></tr>
<tr><td colspan="8">Call-Identifier (Part-2)</td><td>6</td></tr>
<tr><td colspan="8">Sequence-ID</td><td>7</td></tr>
<tr><td colspan="8">Information element #1</td><td rowspan="2"></td></tr>
<tr><td colspan="8">Information element #2</td></tr>
</table>

Figure 7.2.2.1: I1 message structure

7.2.2.1.2 Protocol Version information

The first octet is divided into two four-bit subfields, i.e. the Protocol identifier and the Protocol version number. The Protocol identifier for I1 protocol is "0001" and indicates that the respective message, transported across the transport-layer connection, is an I1 protocol message. The Protocol version number indicates that this is the first version of this specification and the respective value of the Protocol version number subfield is "0001".

7.2.2.1.3 Message Type and Reason

The second octet and third consists of five-bit Message Type field that identifies the type of the I1 message, while the ten-bit Reason fields provide additional information about the respective I1 message, e.g., Progress reason value, as specified in table 7.2.2.1. The bit number 3 in the second octet (marked "R") is reserved for future use.

7.2.2.1.4 Call Identifier

The three octets (i.e. the octet number 4, 5, and 6) that follow the Reason field contain the Call-Identifier field. The Call-Identifier field uniquely specify the I1 session across all I1 interfaces (i.e. between the ICS AS and all ICS UEs connected to the ICS AS). The Call-Identifier field is divided into two subfields, i.e. the part-1 subfield consisting of one octet and the part-2 subfield consisting of two octets. The part 1 subfield is always filled by the UE, while the part-2 subfield is always filled by the ICS AS. The part-1 and part-2 subfields are analogous to the SIP tags inserted in the From and To header fields. The values of all "0" inserted in the octet 3 (i.e. in the Call-Identifier Part-1) indicates that the Call-Identifier (Part-1) subfield is empty (i.e. it has no value). Likewise, values of all "0" inserted in the octet 4 and 5 (i.e., in the Call-Identifier Part-2) indicates that the Call-Identifier Part-2 subfield is empty (i.e., it has no value). When the UE forwards the first I1 message pertaining to an I1 session that is being established (e.g., an I1 Invite or an I1 Progress) to the ICS AS, the UE inserts a new value into the Call-Identifier (Part-1) subfield (i.e., a value that is currently not being unused). Likewise, when the ICS AS forwards the first I1 message pertaining to an I1 session that is being established (e.g., an I1 Invite or an I1 Progress) to the UE, the ICS AS inserts a value into the part-2 subfield. When inserting a value into the Call-Identifier (Part-2) subfield, the ICS AS has to insure that the resulting Call-Identifier field is unique across all I1 interfaces. For example, the ICS AS, upon receiving the first I1 message from the

ICS UE, may insert into the Call-Identifier (Part-2) subfield a value that it is currently using in some other I1 sessions, only if the resulting Call-Identifier field is unique across all I1 interfaces (i.e. between the ICS AS and all ICS UEs).

Some values inserted into the Call-Identifier (Part-1 and Part-2) field are reserved. The Call-Identifier (Part-1 and Part-2) subfield with all bits set to values "1" is a reserved and it is used to identify the I1 session that was automatically bound to a CS bearer that was previously set up without using the I1 protocol. For example, a single CS call that was established using SRVCC may be automatically bound to the I1 session identified with the Call-Identifier (Part-1 and Part-2) subfield with all bits set to values "1" without the ICS UE and the SCC AS exchanging the I1session establishing messages. The usage of the reserved Call-Identifier values is specified in the respective clauses in this document.

7.2.2.1.5 Sequence-ID

The Sequence-ID field value (i.e., the octet number 7) guarantees the proper ordering of the I1 message. The sender of the I1 message increments the Message sequence number value by one for each new I1 message forwarded to its peer. The sequence number value is expressible as an 8-bit unsigned integer. Once the count reaches the value of 2**8, it wraps around back to one.

7.3 Messages

7.3.1 General Messages

Table 7.3.1 lists the I1messages and their encoding. The prefix "0x" indicates that what follows is a bit stream represented as a hexadecimal number.

Table 7.3.1: General Message types

Message	Message Type value (5 bit)	Reason value (10 bit)
I1 Invite (MO)	0x1	0x000
I1 Invite (MT)	0x1	0x001
I1 Invite (Augmentation)	0x1	0x002
I1 Invite (use existing CS bearer)	0x1	0x003
I1 Invite (CW)	0x1	0x005
I1 Bye	0x2	0x000
I1 Notify (used for synchronization)	0x3	0x001
I1 Notify	0x3	0x002 – 0x64
I1 Refer	0x9	0x000
I1 Progress	0x00	0x64 – 0xC7 (NOTE 1)
I1 Success	0x00	0xC8 – 0x12B
I1 Mid Call Request message	0x4	0x001
I1 Notify	0x7	0x001
I1 Failure	0x00	0x12C – 0x25E (NOTE 2)
I1 Dummy	0x00	0x3FF
NOTE 1: The value in the Reason field corresponds to the hex-encoded SIP 1xx values, as specified in subclause 21.1 of RFC 3261[6]. NOTE 2: The value in the Reason field corresponds to the hex-encoded SIP 4xx, 5xx, and 6xx values, as specified in subclauses 21.4, 21.5, and 25.6 of RFC 3261[6].		

7.3.2 I1 INVITE – ICS UE initiated

7.3.2.1 General

This message is sent by the ICE UE to the network to establishment of a session. See table 7.3.2.1.

Message type: I1 INVITE

Direction: ICS UE to SCC AS

Table 7.3.2.1: I1 INVITE message content

Information element	Type/Reference	Presence	Format	Length
Protocol Information	Protocol Information	M	V	1
	7.2.2.1.2			
Message Type	Request Message - INVITE 7.2.2.2.1.2	M	V	2
Call ID	Call-Id 7.2.2.1.4	M	V	2
Message Sequence Number	Sequence-Id 7.2.2.1.5	M	V	1
Timestamp	Timestamp 7.3.2.8	O	V	1
To-id	To-id 7.3.2.3	M	LV	FFS
From-id	From-id 7.3.2.4	M	LV	FFS
Accept Contact	Accept Contact 7.3.2.5	O	TLV	5
ERAccept Contact	ERAccept Contact 7.3.2.6	O	TLV	3-Y
Reject Contact	Reject Contact 7.3.2.7	O	TLV	5

7.3.2.2 Message Type

Identifies that the message is:

i) a Mobile Originated I1 INVITE.

7.3.2.3 To-id

This information element shall be included, it identifies the logical identity of the recipient for the request according to the procedures specified in RFC 3261 [6]. For the coding of this information element please see subclause 7.4.2.3A.

7.3.2.4 From-id

This information element shall be included, it identifies the logical identity that the dialogue originates from according to the procedures specified in RFC 3261 [6]. For the coding of this information element please see subclause 7.4.2.3.

7.3.2.5 Accept Contact

This information element shall be optionally included, if feature tags are indicated.

7.3.2.6 ERAccept Contact

This information element shall be optionally included, if feature tags that have been qualified with the parameter tag "explicit" or "require" are indicated.

7.3.2.7 Reject Contact

This information element shall be optionally included, if feature tags are indicated.

7.3.2.8 Timestamp

This information element shall be included if using a transport layer protocol that is not a real-time transport layer protocol; it provides the SCC AS local time to the ICS UE.

7.3.3 INVITE – SCC AS initiated

7.3.3.1 General

This message is sent by the SCC AS to the ICS UE to establishment of a session. See table 7.3.3.1.

Message type: I1 INVITE.

Direction: SCC AS to ICS UE

Table 7.3.3.1: I1 INVITE message content

Information element	Type/Reference	Presence	Format	Length
Protocol Information	Protocol Information 7.2.2.1.2	M	V	1
Message Type	Request Message - INVITE 7.2.2.2.2.2	M	V	2
CallID	Call-Id 7.2.2.1.4	M	V	2
Message Sequence Number	Sequence-Id 7.2.2.1.5	M	V	1
Timestamp	Timestamp	O	V	1
From-id	From-id 7.3.3.3	M	LV	FFS
SCC AS PSI DN	SCC AS PSI DN 7.3.3.5	M	LV	3-15
To-id	To-id 7.3.3.4	M	LV	FFS

7.3.3.2 Message Type

Identifies that the message is:

i) a Mobile Terminated I1 INVITE.

7.3.3.3 From-id

This information element shall be included; it identifies the logical identity that the dialogue originates from. It is the same as that defined in RFC 3261 [6] however no display name is included. For the coding of this information element please see subclause 7.4.2.3.

7.3.3.4 To-id

This information element shall be included; it identifies the logical identity of the recipient for the request. It is the same as that defined in RFC 3261 [6]. For the coding of this information element please see subclause 7.4.2.3A.

7.3.3.5 SCC AS PSI DN

This information element, as specified in subclause 7.4.2.5, shall be included; it uniquely identifies the SCC AS and session on that AS.

7.3.3.6 Timestamp

This information element,as specified in subclause 7.4.2.14, shall be included if using a transport layer protocol that is not a real-time transport layer protocol, it provides the SCC AS local time to the ICS UE.

7.3.4 BYE – ICS UE initiated

7.3.4.1 General

This message is sent by the ICS UE to the SCC AS to establishment of a session. See table 7.3.4.1.

Message type: I1 BYE

Direction: ICS UE to SCC AS

Table 7.3.4.1: I1 BYE message content

Information element	Type/Reference	Presence	Format	Length
Protocol Information	Protocol Information 7.2.2.1.2	M	V	1
Message Type	Request Message - BYE 7.3.4.2	M	V	2
CallID	Call-Id 7.2.2.1.4	M	V	2

7.3.4.2 Message Type

Identifies that the message is:

i) an I1 BYE.

7.3.5 BYE – SCC AS initiated

7.3.5.1 General

This message is sent by the SCC AS to the ICS UE to establish of a session. See table 7.3.5.1.

Message type: I1 BYE

Direction: SCC AS to ICS UE

Table 7.3.5.1: I1 BYE message content

Information element	Type/Reference	Presence	Format	Length
Protocol Information	Protocol Information 7.2.2.1.2	M	V	1
Message Type	Request Message - BYE 7.3.5.2	M	V	2
CallID	Call-Id 7.2.2.1.4	M	V	2

7.3.5.2 Message Type

Identifies that the message is:

i) an I1 BYE.

7.3.6 I1 PROGRESS – ICS UE initiated

7.3.6.1 General

This message is sent by the ICE UE to the network to establish of a session. See table 7.3.6.1.

Message type: I1 PROGRESS

Direction: ICS UE to SCC AS

Table 7.3.6.1: I1 PROGRESS message content

Information element	Type/Reference	Presence	Format	Length
Protocol Information	Protocol Information	M	V	1
	7.2.2.1.2			
Message Type	Request Message – PROGRESS 7.3.6.2	M	V	2
CallID	Call-Id 7.2.2.1.4	M	V	2
Message Sequence Number	Sequence-Id 7.2.2.1.5	M	V	1

7.3.6.2 Message Type

Identifies that the message is

i) an I1 PROGRESS.

7.3.7 I1 PROGRESS – SCC AS initiated

7.3.7.1 General

This message is sent by the SCC AS to the ICS UE to establish of a session. See table 7.3.3.1.

Message type: I1 PROGRESS

Direction: SCC AS to ICS UE

Table 7.3.3.1: I1 PROGRESS message content

Information element	Type/Reference	Presence	Format	Length
Protocol Information	Protocol Information	M	V	1
	7.2.2.1.2			
Message Type	Request Message - PROGRESS 7.3.7.2	M	V	2
CallID	Call-Id 7.2.2.1.4	M	V	2
Message Sequence Number	Sequence-Id 7.2.2.1.5	M	V	1
SCC AS PSI DN	SCC AS PSI DN TBD	M	LV	??

7.3.7.2 Message Type

Identifies that the message is:

i) an I1 PROGRESS.

7.3.8 I1 FAILURE

7.3.8.1 General

This message is sent by the ICE UE to the network or from the network to the ICS UE to identify that an error has occured. See table 7.3.8.1.

Message type: I1 Failure

Direction: ICS UE to SCC AS and SCC AS to ICS UE

Table 7.3.8.1: I1 Failure message content

Information element	Type/Reference	Presence	Format	Length
Protocol Information	Protocol Information	M	V	1
	7.2.2.1.2			
Message Type	Request Message - INVITE 7.3.8.2	M	V	2
Call ID	Call-Id 7.2.2.1.4	M	V	2
Message Sequence Number	Sequence-Id 7.2.2.1.5	M	V	1
To-id	To-id 7.3.8.3	O	TLV	FFS
Reason Phrase	Phrase	O	TLV	

7.3.8.2 Message Type

Identifies that the message is:

i) an I1 FAILURE.

7.3.8.3 To-id

This information element,as specified in subclause 7.4.2.3, may optionally be included and can appear multiple times. It identifies alternative address"s that the UE should attempt to use It is the same as the contact header field that is defined in sections 21.3 of RFC 3261 [6].

7.3.8.4 Reason Phrase

This information element,as specified in subclause 7.4.2.13, may optionally be included and can appear multiple time. It is the same as the Reason-Phrase header field that is defined in RFC 3261 [6].

7.4 I1 information elements and functional definition

7.4.1 I1 information elements

The list of the I1 information elements is shown in table 7.4.1.

Table 7.4.1 I1-information elements

I1 information element Name	Description and content (subclauses)
Error-code	7.4.2.2
From-id	7.4.2.3
To-id	7.4.2.3A
Privacy	7.4.2.4
SCC-AS-id	7.4.2.5
Session-identifier	7.4.2.6
Replaces	7.4.2.8
Accept Contact	7.3.2.5
ERAccept Contact	7.3.2.6
Reject Contact	7.3.2.7
Mid-Call	7.4.2.12
Reason Phrase	7.4.2.13
Timestamp	7.4.2.14

Error-code

The Error-code information element is included in every I1-Error response message. The Error-code information element is binary encoded SIP failure response. The SIP 4xx request failure responses, the 5xx server failure responses, and the 6xx global failure responses are binary encoded and included in the Error-code information element as specified in subclause 7.4.2.1 and table 7.4.2.1. The interpretation of each binary encoded failure response is analogous to the interpretation of associated SIP failure response.

From-id Information

The From-id Information IE specifies a public user identity of the calling user, e.g., the calling party number, either as an E.164, Identifier or a SIP URI.

The From-id information element may contain either an E.164, a SIP URI or a identifier that identifies a public user identity to be used (see annex A).

To-id Information

The To-id Information IE specifies a public user identity of the called user, e.g., the called party number.

The To-id information element may contain either an E.164, a SIP URI or a identifier that identifies a public user identity to be used (see annex A).

Privacy

The UE uses the Privacy information element to indicate to the SCC AS how to handle the SIP header fields when the SCC AS forwards the SIP requests and responses on behalf of the UE to the far-end UA. The Privacy information element when sent by the UE to the SCC AS contains binary encoded "priv-value" (as specified in the RFC 3323 [8] and RFC 3325 [9]). When the SCC AS, upon receiving a Privacy information element over I1 interface, forwards a SIP request or a response to the far-end UA, the SCC AS behaves as specified in the RFC 3323 [8] and RFC 3325 [9] e.g. the SCC AS inserts a P-Asserted-Identity header field into SIP message as requested by the Privacy information element.

SCC-AS-id

The SCC-AS-id information element contains an International E.164 number representation of the SCC AS PSI DN that points to the SCC AS. When the UE sets up a CS bearer connection by sending a SETUP message to the MSC server, the UE specifies the respective International E.164 number as the called party number. Subsequently the call will be routed to the respective SCC AS via a MGCF where the SCC-AS-PSI-DN will be treated as a wildcard PSI as specified in 3GPP TS 23.003 [15] and procedures as specified in 3GPP TS 24.229 [12] subclause 5.3,2.1 item 3.

Session-identifier

The Session-identifier information element is an identifier used either by the UE or the SCC AS to uniquely and globally identify a I1 session across all interface (i.e. the I1 interface, Gm interface and the IMS). The Session identifier is dynamically allocated by the SCC AS to identify the I1 session that is being established. The SCC AS includes the Session-identifier information element in the first I1 message sent by the SCC AS to the UE. The Session-identifier information element may contain different values, e.g. the Session Transfer Identifier (STI), as specified in subclause 7.4.2.1 and associated subclause 7.4.2.1.

Replaces

The Replaces information element is used by the UE to identify an existing call or a SIP dialog that will be replaced with an I1 session being established over the I1 interface. When the UE wants to replace an existing call or a SIP dialog with a new I1 session, the UE sends an I1 Invite request message to the SCC AS with the Replaces information element that contains the identity of the SIP dialog or a call that will be replaced with a new call being established. In the case of UE assisted T-ADS, the SCC AS may send an I1 Invite request message to the terminating ICS UE with Replaces information element that contains the identity of the SIP dialog to change the service control for the session from Gm to I1.

Accept Contact

The Accept contact information element is used by the UE to identify the SIP feature tags that the UE are included in a SIP Accept Contact header per procedures in RFC 3841 [14]. However if the feature tags are to be appended with "explicit" and or the "require" parameter tags then the SIP feature tag shall not be sent in the Accept Contact but in the ERAccept Contact header information element.

ERAccept Contact

The ERAccept contact information element is used by the UE to identify those SIP feature tags that either "explicit" or "require" parameter tags added per procedures in RFC 3841 [14]. If "explicit" and or "require" parameter tag is required then the feature tag is not included in the Accept Contact information element.

Reject Contact

The Reject contact information element is used by the UE to identify the SIP feature tags that the UE would normally send in a SIP Reject contact header per procedures in RFC 3841 [14].

Mid-Call

The Mid-Call information element is used between ICS UE and the SCC AS to exchange the Mid-call supplementary services control information, e.g. hold, resume, conference.

Timestamp

The Timestamp information element specifies a local time on the I1 message sender. The local time is measured in seconds and it is 32 bits long.

7.4.2 I1 Information elements encoding

7.4.2.1 General

The structure of the I1 information elements is shown in figure 7.4.2.1.

8	7	6	5	4	3	2	1	Octet
Information Element code					Code specific			1
Information Element length (in octets)								2
Information Element body (as required)								3
								etc.

Figure 7.4.2.1: I1 information element format

Each I1 information element contains a common two-octet field followed by a variable-size body. The first octet contains the Information Element code and Code specific values. Each I1 information element is uniquely identified with the respective Information Element code (i.e., encoded with bits numbered 4, 5, to 8 of the first octet). The Code specific value (i.e., encoded with bits numbered 1, 2, and 3 of the first octet) provide additional information about respective I1 information element. For example, if the Information Element code specifies that this is a To-id I1 information element, then the Code specific value will indicate whether the Information Element body contains an E.164 number or SIP URI. The Code specific values for each respective I1 information element are described in the respective subclauses.

The second octet i.e. the Information Element length specifies the length of the I1 information element body (i.e., the number of octets following the Information Element length) in binary format. The bit number 1 of octet number 2 is the list significant bit and bit number 8 of the octet number 2 is the most significant bit. The table 7.4.2.1 specifies the Information Element code for each I1 information element.

Table 7.4.2.1: I1-information element coding

Information Element code	I1 information element Name	Reference subclause
Bits 8 7 6 5 4		
1 0 0 1 1	From-id	7.4.2.3
1 0 1 0 0	Privacy	7.4.2.4
1 0 1 0 1	SCC-AS-id	7.4.2.5
1 0 1 1 0	Session-identifier	7.4.2.6
1 0 0 1 0	Replaces	7.4.2.8
1 1 0 0 0	Mid-Call	7.4.2.12
1 1 0 0 1	Timestamp	7.4.2.14
1 1 1 0 0	To-id	7.4.2.3A

7.4.2.2 Void

7.4.2.3 From-id Information

The purpose of the From-id information element is to transport a public identity of the From Party. The From-id information element may contain either a SIP URI or a telephone number (e.g. international number, national number) or an identifier value that identifies a known public identity. The Code specific field as specified in table 7.4.2.3.1, i.e., the bits 3, 2, and 1 of the octet number 1 specify the type of information contained in the From-id information element.

If the From-id Information is a SIP URI username@domainname then the Code specific field as specified in table 7.4.2.3.1 bits 3,2, and 1 shall be set to "010" and shall be encoded to an octet string according to UTF-8 encoding rules as specified in RFC 3629 [16].

When bits 3, 2, and 1 of the octet number 1 are to be set to "001" to indicate that the Identity Information information element contains an E.164 number (see table 7.4.2.3.1-2). This is deduced when the From-id Information to be used by is a tel URI or a SIP URI with URI parameter User=Phone then the Code specific field and if a tel or SIP URI is identified as being globally unique identified by the presence of "+" character at the start of the digit string.

When bits 3, 2, and 1 of the octet number 1 are to be set to "000" it indicates that the From-id information element contains a number whose type of number is unknown (see table 7.4.2.3.1-2) e.g. local or national number. This is deduced when the Identity Information to be used by is a tel URI or a SIP URI with URI parameter User=Phone and if a tel or SIP URI is not identified as being globally unique identified by the presence of "+" character at the start of the digit string.

When bits 3, 2, and 1 of the octet number 1 are to be set to "000" it indicates that the Identity Information information element contains a public user identity that can be can be derived (see annex A).

When the From-id information element contains an International number (i.e. an E.164 number), the E.164 digit-string is included in the octet 3, octet 4, octet 5, etc. as follows:

- the bits numbers 8, 7, 6, and 5 of octet number 3 are used to binary encode the most significant digit of the E.164 digit-string;

- the bits numbers 4, 3, 2, and 1 of octet number 3 are used binary encode the next significant digit of the E.164 digit-string;

- the bits numbers 8, 7, 6, and 5 of octet number 4 are used binary encode the next significant digit of the E.164 digit-string; and so on until the entire E.164 digit-string is included in the From-id information element; and

- the bit-pattern "1111"iserted either in the bits 8, 7, 6, and 5 or bits 4, 3, 2, and 1 of any octet indicates the end of the E.164 digit-string, i.e. the bit-pattern of "1111" is used as the end-delimiter for the E.164 digit-string.

8	7	6	5	4	3	2	1	Octet
Information Element code					Code specific			1
1	0	0	1	1				
Information Element length (in octets)								2
Information Element body								3-Y

Figure 7.4.2.3.1-1: From-id information element

Table 7.4.2.3.1: From-id information element

(octet 1) Code specific	
Bits	
3 2 1	
0 0 0	Unspecified
0 0 1	International number, i.e. E.164 number (Note 1)
0 1 0	SIP URI
0 1 1	Identifier (See Annex A)
	Other values are reserved for future use
(octet 3)	
SIP URI	
	The URI shall be encoded to an octet string according to UTF-8 encoding rules as specified in RFC 3629 [16]
(octet 3)	
Identifier	
	Contains one octet body coded with identifier value that identifies the public user identity.
(octet 3)	
Bits	
8 7 6 5	the most significant digit of the E.164 digit-string
(octet 3)	
Bits	
4 3 2 1	the next significant digit of the E.164 digit-string (Note 2)
(octet 4)	
Bits	
8 7 6 5	the next significant digit of the E.164 digit-string (Note 2)
(octet 4)	
Bits	
4 3 2 1	the next significant digit of the E.164 digit-string (Note 2)
(next octet)	
Bits (Note 2)	
(Note 3)	
Note 1 – Prefix or escape digits shall not be included. Note 2 – the next significant digits of the E.164 digit-string are included in subsequent bits 8, 7, 6, and 5 or bits 4, 3, 2. Note 3 – The E.164 digit-string terminates with delimiter "1111" in the bits 8, 7, 6, and 5 or bits 4, 3, 2, and 1 of any octet indicating the end of the E.164 digit-string.	

7.4.2.3A To-id Information

The purpose of the To-id information element is to transport a public identity of the To Party. The To-id information element may contain either a SIP URI or a telephone number (e.g. international number, national number) or an identifier value that identifies a known public identity. The Code specific field, i.e., the bits 3, 2, and 1 of the octet number 1 specify the type of information contained in the To-id information element,.

If the To-id Information is a SIP URI username@domainname then the Code specific fields bits 3,2, and 1 shall be set to "010" and shall be encoded to an octet string according to UTF-8 encoding rules as specified in RFC 3629 [16].

When bits 3, 2, and 1 of the octet number 1 are to be set to "001" it indicates that the To-id information element contains an E.164 number (see table 7.4.2.3.1-2). This is deduced when the To-id to be used by is a tel URI or a SIP URI with URI parameter User=Phone then the Code specific field and if a tel or SIP URI is identified as being globally unique identified by the presence of "+" character at the start of the digit string.

When bits 3, 2, and 1 of the octet number 1 are to be set to "000" it indicates that the To-id information element contains a number whose type of number is unknown (see table 7.4.2.3.1-2) e.g. local or national number. This is deduced when the To-id to be used by is a tel URI or a SIP URI with URI parameter User=Phone and if a tel or SIP URI is not identified as being globally unique identified by the presence of "+" character at the start of the digit string.

When bits 3, 2, and 1 of the octet number 1 are to be set to "000" it indicates that the To-id information element contains a public user identity that can be can be derived (see annex A).

When the To-id information element contains an International number (i.e. an E.164 number), the E.164 digit-string is included in the octet 3, octet 4, octet 5, etc. as follows:

- the bits numbers 8, 7, 6, and 5 of octet number 3 are used to binary encode the most significant digit of the E.164 digit-string;
- the bits numbers 4, 3, 2, and 1 of octet number 3 are used binary encode the next significant digit of the E.164 digit-string;
- the bits numbers 8, 7, 6, and 5 of octet number 4 are used binary encode the next significant digit of the E.164 digit-string; and so on until the entire E.164 digit-string is included in the From-id information element; and
- the bit-pattern "1111"iserted either in the bits 8, 7, 6, and 5 or bits 4, 3, 2, and 1 of any octet indicates the end of the E.164 digit-string, i.e. the bit-pattern of "1111" is used as the end-delimiter for the E.164 digit-string.

Table 7.4.2.3A.1-1: To-id information element

<table>
<tr><td>8</td><td>7</td><td>6</td><td>5</td><td>4</td><td>3</td><td>2</td><td>1</td><td>Octet</td></tr>
<tr><td colspan="5">Information Element code
1 1 1 0 0</td><td colspan="3">Code specific</td><td>1</td></tr>
<tr><td colspan="8">Information Element length (in octets)</td><td>2</td></tr>
<tr><td colspan="8" rowspan="2">Information Element body</td><td>3</td></tr>
<tr><td>etc.</td></tr>
</table>

Table 7.4.2.3A.1-2: To-id information element

(octet 1)	Code specific
Bits	
3 2 1	
0 0 0	Unspecified
0 0 1	International number, i.e. E.164 number (Note 1)
0 1 0	SIP URI
0 1 1	Identifier (See Annex A)
	Other values are reserved for future use
(octet 3)	
SIP URI	
	The URI shall be encoded to an octet string according to UTF-8 encoding rules as specified in RFC 3629 [16]
(octet 3)	
Identifier	
	Contains one octet body coded with identifier value that identifies the public user identity.
(octet 3)	
Bits	
8 7 6 5	the most significant digit of the E.164 digit-string
(octet 3)	
Bits	
4 3 2 1	the next significant digit of the E.164 digit-string (Note 2)
(octet 4)	
Bits	
8 7 6 5	the next significant digit of the E.164 digit-string (Note 2)
(octet 4)	
Bits	
4 3 2 1	the next significant digit of the E.164 digit-string (Note 2)
(next octet)	
Bits	(Note 2)
	(Note 3)
Note 1:	Prefix or escape digits shall not be included.
Note 2:	The next significant digits of the E.164 digit-string are included in subsequent bits 8, 7, 6, and 5 or bits 4, 3, 2.
Note 3:	The E.164 digit-string terminates with delimiter "1111" in the bits 8, 7, 6, and 5 or bits 4, 3, 2, and 1 of any octet indicating the end of the E.164 digit-string.

7.4.2.4 Privacy

The ICS UE may include the Privacy information element in the I1 Invite message to indicate its privacy preferences that the SCC AS should apply to the SIP session toward the remote UE. When the SCC AS sets up a SIP session on behalf of the UE, the SCC AS sends a SIP INVITE request that includes the privacy information that the SCC AS received in the Privacy information element.

The Privacy information element when sent by the ICS UE to the SCC AS contains binary encoded "priv-value" (with the same semantics as specified in the RFC 3323 [8] and RFC 3325 [9]). The UE may request multiple types of privacy for the same call (see RFC 3323 [8]). Hence, the UE include all of the requested privacy types in its Privacy information element by setting the respective bits as shown in table 7.4.2.4.1.

8	7	6	5	4	3	2	1	Octet
Information Element code					Code specific			1
1	0	1	0	0				
Information Element length (in octets)								2
Information Element body								3

Figure 7.4.2.4.1: Privacy information element

Table 7.4.2.4.1: Privacy information element

(octet 1) Bits 3 2 1 0 0 1	Code specific (NOTE 1) Other values are reserved for future use
(octet 3) Bit 8 1	 The UE indicates to the SCCAS that "Privacy: id" (as specified in the RFC 3325 [9]) is requested (NOTE 2).
Bit 7 1	 The UE indicates to the SCCAS that "Privacy: header" (as specified in the RFC 3323 [8]) is requested (NOTE 2)
Bit 6 1	 The UE indicates to the SCCAS that "Privacy: session" (as specified in the RFC 3323 [8]) is requested (NOTE 2)
Bit 5 1	 The UE indicates to the SCCAS that "Privacy: user" (as specified in the RFC 3323 [8]) is requested (NOTE 2)
Bit 4 1	 The UE indicates to the SCCAS that "Privacy: none" (as specified in the RFC 3323 [8]) is requested (NOTE 2)
Bit 3 1	 The UE indicates to the SCCAS that "Privacy: critical" (as specified in the RFC 3323 [8]) is requested (NOTE 2)
Bits 2 and 1 reserved for future use	
NOTE 1: If the Code specific value is set to "001" it indicates that the Privacy information element consists of three octets and each bit in octet number 3 is interpreted as specified in this table. NOTE 2: The value of "0" in this bit indicates that corresponding "priv-value' (with the same semantics as specified in the RFC 3323 [8] and RFC 3325 [9]) is not used and respective privacy is not requested.	

7.4.2.5 SCC-AS-id

The SCC-AS-id information element contains an International E.164 Number representation of the SCC AS PSI DN that points to the SCC AS. The SCC AS PSI DN information element has a minimum length of 3 octets and a maximum length of 10 octets.

The SCC-AS-id information element may contain either a SIP URI or an international telephone number (i.e. an E.164 national number). The Code specific field as specified in table 7.4.2.5.1, the bits 3, 2, and 1 of the octet number 1 specify the type of information that is used to identify the SCC AS. When the SCC AS forwards a SCC AS PSI DN associated with the SCC AS to the UE, the SCC AS will include the SCC AS PSI DN in the SCC-AS-id information element. The PSI DN is an E.164 number.

When the Code specific field as specified in table 7.4.2.5.1, bits 3, 2, and 1 of the octet number 1 shall be set to "001" it indicates that the SCC-AS-id information element contains a SCC AS PSI DN (i.e. an E.164 number). When the SCC-AS-id information element contains a PSI DN (i.e. an E.164 number), the E.164 digit-string is included in the octet 3, octet 4, octet 5, etc. as shown in table 7.4.2.5.1.

8	7	6	5	4	3	2	1	Octet
Information Element code					Code specific			1
1	0	1	0	1				
Information Element length (in octets)								2
Information Element body								4

Figure 7.4.2.5.1: SCC-AS-id information element

Table 7.4.2.5.1: SCC-AS-id information element

(octet 1) Bits 3 2 1	Code specific
0 0 0	Unspecified
0 0 1	PSI DN, i.e. E.164 number (Note 1)
	Other values are reserved for future use
(octet 3) Bits 8 7 6 5	the most significant digit of the E.164 digit-string
(octet 3) Bits 4 3 2 1	the next significant digit of the E.164 digit-string (Note 2)
(octet 4) Bits 8 7 6 5	the next significant digit of the E.164 digit-string (Note 2)
(octet 4) Bits 4 3 2 1	the next significant digit of the E.164 digit-string (Note 2)
(next octet) Bits	(Note 3)
Note 1 – Prefix or escape digits shall not be included.	
Note 2 – the next significant digits of the E.164 digit-string are included in subsequent bits 8, 7, 6, and 5 or bits 4, 3, 2.	
Note 3 – The E.164 digit-string terminates with delimiter "1111" in the bits 8, 7, 6, and 5 or bits 4, 3, 2, and 1 of any octet indicating the end of the E.164 digit-string.	

7.4.2.6 Session-identifier

The Session-identifier information element is used either by the ICS UE or the SCC AS to covey the identity of the session being established. The Code specific subfield, i.e., the bits 3, 2, and 1 of the octet number 1 specify the type of information that is used to identify the session across.

When a SIP dialog or an I1 session is identified with an E.164 number (e.g. with a STI), then this identifier is conveyed across the I1 interface in a Session-identifier information element. In this case, the Code specific field, i.e., the bits 3, 2, and 1 of the octet number 1 is set to "001", as shown in figure 7.4.2.6.1 and table 7.4.2.6.1.

8	7	6	5	4	3	2	1	Octet
Information Element code					Code specific			1
1	0	1	1	0				
Information Element length (in octets)								2
Information Element body								3

Figure 7.4.2.6.1: Session-identifier information element

Table 7.4.2.6.1: Session-identifier information element

(octet 1) Bits 3 2 1 0 0 0 0 0 1	Code specific Unspecified Session or a dialog identified with an E.164 number (Note 1) Other values are reserved for future use
(octet 3) Bits 8 7 6 5	 the most significant digit of the E.164 digit-string
(octet 3) Bits 4 3 2 1	 the next significant digit of the E.164 digit-string (Note 2)
(octet 4) Bits 8 7 6 5	 the next significant digit of the E.164 digit-string (Note 2)
(octet 4) Bits 4 3 2 1 *(next octet)* Bits	 the next significant digit of the E.164 digit-string (Note 2) (Note 3)
Note 1 – Prefix or escape digits shall not be included. Note 2 – the next significant digits of the E.164 digit-string are included in subsequent bits 8, 7, 6, and 5 or bits 4, 3, 2. Note 3 – The E.164 digit-string terminates with delimiter "1111" in the bits 8, 7, 6, and 5 or bits 4, 3, 2, and 1 of any octet indicating the end of the E.164 digit-string.	

7.4.2.7 Void

7.4.2.8 Replaces

The Replaces information element is included in the I1 Invite message to indicate to the recipient that the I1 session being established will replace the existing SIP dialog identified by the Replaces information element. The Replaces information element also contains the identity of the dialog that will be replaced.

8	7	6	5	4	3	2	1	Octet
Information Element code 1	0	0	1	0	Code specific			1
Information Element length (in octets)								2
								3
								4

Figure 7.4.2.8.1: Replaces information element

Table 7.4.2.8.1: Replaces information element

(octet 1) Bits 3 2 1 0 0 0 0 0 1	Code specific Unspecified The Code specific value set to "001" Specifies that the SIP dialog that will be replaced is identified with a STI that is an E.164 number. (NOTE 1) Other values are reserved for future use
(octet 3) Bits 8 7 6 5	 the most significant digit of the E.164 digit-string
(octet3) Bits 4 3 2 1	 the next significant digit of the E.164 digit-string (NOTE 2)
(octet 4) Bits 8 7 6 5	 the next significant digit of the E.164 digit-string (NOTE 2)
(octet 4) Bits 4 3 2 1 (next octet) Bits	 the next significant digit of the E.164 digit-string (NOTE 2) (NOTE 3)
NOTE 1: Prefix or escape digits shall not be included. NOTE 2: The next significant digits of the E.164 digit-string are included in subsequent bits 8, 7, 6, and 5 or bits 4, 3, 2. NOTE 3: The E.164 digit-string terminates with delimiter "1111" either in the bits 8, 7, 6, and 5 or bits 4, 3, 2, and 1 of any octet, hence indicating the end of the E.164 digit-string.	

7.4.2.9 Accept Contact

The UE may include the Accept Contact element in the I1 Invite message to indicate any called feature preferences per RFC 3841 [14]. The Code Specific value the bits 3, 2, and 1 of the octet number 1 is set to "001" as specified in table 7.4.2.9.

8	7	6	5	4	3	2	1	Octet
Information Element code					Code specific			1
1	0	0	1	1				
Information Element length (in octets)								2
Information Element body								3_Y

Figure 7.4.2.9: Accept Contact information element

Table 7.4.2.9: Accept Contact information element

(octet 1) Bits 3 2 1 0 0 0 0 0 1	Code specific Unspecified Accept Contact All other values reserved
(octet 3)	Bit Specific
Bits	
1	sip.audio as defined in RFC 3840 [15]
2	sip.application as defined in RFC 3840 [15]
3	sip.data as defined in RFC 3840 [15]
4	sip.control as defined in RFC 3840 [15]
5	sip.video as defined in RFC 3840 [15]
6	sip.text as defined in RFC 3840 [15]
7	sip.automata as defined in RFC 3840 [15]
8	sip.duplex = full as defined in RFC 3840 [15]

(octet 4)	Bit Specific
Bits	
1	sip.duplex = half, as defined in RFC 3840 [15]
2	sip.duplex = receive only as defined in RFC 3840 [15]
3	sip.duplex = send only as defined in RFC 3840 [15]
4	sip.mobility = fixed as defined in RFC 3840 [15]
5	sip.mobility = mobile as defined in RFC 3840 [15]
6	sip.actor =principal, as defined in RFC 3840 [15]
7	sip.actor =attendant, as defined in RFC 3840 [15]
8	sip.actor = msg-taker, as defined in RFC 3840 [15]

(octet 5)	Bit Specific
Bits	
1	sip.actor – information as defined in RFC 3840 [15]
2	sip.isfocus as defined in RFC 3840 [15]
3	sip.byeless as defined in RFC 3840 [15]
4	sip.rendering – yes as defined in RFC 4235 [16]
5	sip.rendering – no as defined in RFC 4235 [16]
6	sip.rendering – unknown as defined in RFC 4235 [16]
7	sip.message as defined in RFC 4569 [17]
8	sip.ice

(octet 6)	Bit Specific
Bits	
1	Reserved
2	Reserved
3	Reserved
4	Reserved
5	Reserved
6	Reserved
7	Reserved

8	Extension

7.4.2.10 ERAccept Contact

The UE may include the ERAccept Contact element in the I1 Invite message to indicate any called feature preferences per RFC 3841 [14] that require the "explicit" and or "require" parameter tag appended to them. The Code Specific value the bits 3, 2, and 1 of the octet number 1 is set to "001" as specified in table 7.4.2.10.

<table>
<tr><td>8</td><td>7</td><td>6</td><td>5</td><td>4</td><td>3</td><td>2</td><td>1</td><td>Octet</td></tr>
<tr><td colspan="5">Information Element code
1 0 1 0 0</td><td colspan="3">Code specific</td><td>1</td></tr>
<tr><td colspan="8">Information Element length (in octets)</td><td>2</td></tr>
<tr><td>E</td><td>R</td><td colspan="6">Feature Tag Value</td><td>3-Y</td></tr>
</table>

Figure 7.4.2.10: ERAccept Contact information element

Table 7.4.2.10: ERAccept Contact information element

(octet 1) Bits 3 2 1 0 0 0 0 0 1	Code specific Unspecified ERAccept All other values reserved
(octet 3-Y)	Feature Tag Value
Bit	
<u>8</u>	Value 1 "explicit' required as defined in RFC 3841 [14]
7	Value 1 "require' required. as defined in RFC 3841 [14]
6-1	Feature Tag

(octet -3-Y)	Bit Specific
Bits	
6 5 4 3 2 1	
0 0 0 0 0 0	sip.audio as defined in RFC 3840 [15]
0 0 0 0 0 1	sip.application as defined in RFC 3840 [15]
0 0 0 0 1 0	sip.data as defined in RFC 3840 [15]
0 0 0 0 1 1	sip.control as defined in RFC 3840 [15]
0 0 0 1 0 0	sip.video as defined in RFC 3840 [15]
0 0 0 1 0 1	sip.text as defined in RFC 3840 [15]
0 0 0 1 1 0	sip.automata as defined in RFC 3840 [15]
0 0 0 1 1 1	sip.duplex = full as defined in RFC 3840 [15]
0 0 1 0 0 0	sip.duplex = half, as defined in RFC 3840 [15]
0 0 1 0 0 1	sip.duplex = receive only as defined in RFC 3840 [15]
0 0 1 0 1 0	sip.duplex = send only as defined in RFC 3840 [15]
0 0 1 0 1 1	sip.mobility = fixed as defined in RFC 3840 [15]
0 0 1 1 0 0	sip.mobility = mobile as defined in RFC 3840 [15]
0 0 1 1 0 1	sip.actor = principal, as defined in RFC 3840 [15]
0 0 1 1 1 0	sip.actor = attendant, as defined in RFC 3840 [15]
0 0 1 1 1 1	sip.actor -= msg-taker, as defined in RFC 3840 [15]
0 1 0 0 0 0	sip.actor = information as defined in RFC 3840 [15]
0 1 0 0 0 1	sip.isfocus as defined in RFC 3840 [15]
0 1 0 0 1 0	sip.byeless as defined in RFC 4235 [16]
0 1 0 0 1 1	sip.rendering = yes as defined in RFC 4235 [16]
0 1 0 1 0 0	sip.rendering =no as defined in RFC 4235 [16]
0 1 0 1 0 1	sip.rendering = unknown as defined in RFC 4235 [16]
0 1 0 1 1 0	sip.message as defined in RFC 4569 [17]
0 1 0 1 1 1	sip.ice
0 1 1 0 0 0 to 1 1 1 1 1 1	Reserved

7.4.2.11 Reject Contact

The UE may include the Reject Contact element in the I1 Invite message to indicate any called feature preferences per RFC 3841 [15]. The Code Specific value the bits 3, 2, and 1 of the octet number 1 is set to "000" as specified in table 7.4.2.11.

8	7	6	5	4	3	2	1	Octet
Information Element code					Code specific			1
1	0	0	1	0				
Information Element length (in octets)								2
Information Element Body								3-6

Figure 7.4.2.11: Reject Contact information element

Table 7.4.2.11: Reject Contact information element

(octet 3) Bit Specific	
Bits	
1	sip.audio as defined in RFC 3840 [15]
2	sip.application as defined in RFC 3840 [15]
3	sip.data as defined in RFC 3840 [15]
4	sip.control as defined in RFC 3840 [15]
5	sip.video as defined in RFC 3840 [15]
6	sip.text as defined in RFC 3840 [15]
7	sip.automata as defined in RFC 3840 [15]
8	sip.duplex = full as defined in RFC 3840 [15]

(octet 4) Bit Specific	
Bits	
1	sip.duplex = half, as defined in RFC 3840 [15]
2	sip.duplex = receive only as defined in RFC 3840 [15]
3	sip.duplex = send only as defined in RFC 3840 [15]
4	sip.mobility = fixed as defined in RFC 3840 [15]
5	sip.mobility = mobile as defined in RFC 3840 [15]
6	sip.actor =principal, as defined in RFC 3840 [15]
7	sip.actor =attendant, as defined in RFC 3840 [15]
8	sip.actor = msg-taker, as defined in RFC 3840 [15]

(octet 5) Bit Specific	
Bits	
1	sip.actor – information as defined in RFC 3840 [15]
2	sip.isfocus as defined in RFC 3840 [15]
3	sip.byeless as defined in RFC 4235 [16]
4	sip.rendering – yes as defined in RFC 4235 [16]
5	sip.rendering – no as defined in RFC 4235 [16]
6	sip.rendering – unknown as defined in RFC 4235 [16]
7	sip.message as defined in RFC 4569 [17]
8	sip.ice

(octet 6) Bit Specific	
Bits	
1	Reserved
2	Reserved
3	Reserved
4	Reserved
5	Reserved
6	Reserved
7	Reserved
8	Extension

7.4.2.12 Mid-Call

The Mid-Call information element is used either by the ICS UE or the SCC AS to covey the supplementary services control information. The Code specific subfield, i.e., the bits 3, 2, and 1 of the octet number 1 specify the type of information that is used to identify the services control type.

8	7	6	5	4	3	2	1	Octet
Information Element code					Code specific			1
1	0	1	1	0				
Information Element length (in octets)								2
Information Element Body								3

Figure 7.4.2.12.1: Mid-Call information element

Table 7.4.2.12.1: Mid-Call information element

(octet 1) Bits 3 2 1	Code specific
0 0 0	Unspecified
0 0 1	Place the call on hold, see subclause 6.3.4 (NOTE 1, NOTE 2)
0 1 0	Resume the call, see subclause 6.3.4 (NOTE 1, NOTE 2)
	Other values are reserved for future use
NOTE 1:	When this Code specific value is used, the Information Element body is empty and the Information Element length value is set to zero.
NOTE 2:	When included by ICS UE, the element indicates that ICS UE puts the call on hold. When included by SCC AS, the element indicates that remote UE puts the call on hold.

7.4.2.13 Reason-Phrase

The purpose of the Reason-Phrase field is as defined in RFC 3261[6].

The information element body as specified in table 7.4.2.13.1 shall be encoded to an octet string according to UTF-8 encoding rules as specified in RFC 3629 [16]. The Code Specific value the bits 3, 2, and 1 of the octet number 1 is set to "001" as specified in table 7.4.2.13.1.

8	7	6	5	4	3	2	1	Octet
Information Element code 1 1 0 0 0					Code specific			1
Information Element length (in octets)								2
Information Element Body								3-Y

Figure 7.4.2.13.1: Reason-Phrase information elementTable 7.4.2.13.1: Reason-Phrase information element

(octet 1) Bits 3 2 1	Code specific
0 0 0	Unspecified
0 0 1	Reason Phrase
	All other values reserved
(octet 3-Y)	Reason encoded as specified in RFC 3629 [16].

7.4.2.14 Time Stamp

The Timestamp information element is used by the ICS UE and the SCC AS to convey local time. The Timestamp information element contains local time measured in seconds.

When the Code specific field as specified in table 7.4.2.14.1, bits 3, 2, and 1 of the octet number 1 shall be set to "001", it indicates that the Timestamp information element contains local time measured in seconds in 32 bits format. The time is conveyed across the I1 interface in a Timestamp information element.

8	7	6	5	4	3	2	1	Octet
Information Element code 1 1 0 0 1					Code specific			1
Information Element length (in octets)								2
Information Element body								3-6

Figure 7.4.2.14.1: Timestamp information element

Table 7.4.2.14.1: Timestamp information element

(octet 1) Bits 3 2 1 0 0 0 0 0 1	Code specific Unspecified Timestamp (Note 1) Other values are reserved for future use
(octet 3) Bits 8 7 6 5 4 3 2 1	 the first (right) 8 bits of the Timestamp
(octet 4) Bits 8 7 6 5 4 3 2 1	 the next 8 bits of the Timestamp
(octet 5) Bits 8 7 6 5 4 3 2 1	 the next 8 bits of the Timestamp
(octet 6) Bits 8 7 6 5 4 3 2 1	 the next 8 bits of the Timestamp
Note 1 – Timestamp in 32 bits format.	

7.5 Session states and Session control procedures

7.5.1 General

This clause defines the basic session control states that an individual session may acquire. Since several sessions may exist simultaneously across the I1 interface and each session may be in a different state, the session states describe the state of a particular session rather then describing the state of the I1 interface. The procedures for session control are given in subclause 7.5.3.

7.5.2 Session states

7.5.2.1 Session originated by the ICS UE

7.5.2.1.1 Session states at ICS UE – ICS UE originated call

This subclause lists the session states that may exist at the UE for a session originated by the ICS UE.

- **null**: No session exists.
- **trying**: This state exists for an UE originated session, when the ICS UE has requested a session establishment by sending an I1Invite message but has not yet received any response.
- **proceeding**: This state exists for an UE originated session when the ICS UE has received an I1 Progress message with Progress reason set to Call progressing from the SCC AS acknowledging that the SCC AS has received the I1 Invite message.
- **alerted**: This state exists for an UE originated session when the calling ICS UE has received an I1 Progress message with Progress reason set to Ringing indicating that remote endpoint alerting has been initiated.
- **confirmed**: This state exists for an UE originated session when the ICS UE has received an I1 Success message indicating that the remote endpoint has accepted the session.

7.5.2.1.2 Session states at SCC AS – ICS UE originated call

This subclause lists the I1 session states that may exist at the SCC AS for a I1 session originated by the ICS UE.

- **null**: No I1 session exists.
- **initiated**: This state exists for an UE originated I1 session when the SCC AS has received an I1 Invite message but has not yet responded.
- **progressing**: This state exists for an UE originated I1 session when the SCC AS has sent an I1 Progress message with Progress reason set to Call progressing acknowledging that the SCC AS has received the I1Invite message.
- **alerting**: This state exists for an UE originated I1 session when the SCC AS has sent an I1 Progress message with Progress reason set to Ringing indicating that remote endpoint alerting has been initiated.
- **confirmed**: This state exists for an UE originated I1 session when the SCC AS has sent an I1 Success message indicating that the I1 session has been accepted.

7.5.2.2 Session terminated at the ICS UE

7.5.2.2.1 Session states at UE – ICS UE terminated call

This subclause lists the I1 session states that may exist in the UE for a I1 session terminated at the ICS UE.

- **null**: No I1 session exists.
- **initiated**: This state exists for a I1 session terminated at the UE when the ICS UE has received an I1Invite message but has not yet responded.
- **progressing**: This state exists for a I1 session terminated at the UE when the ICS UE has sent an I1 Progress message with Progress reason set to Call progressing acknowledging that the ICS UE has received the I1Invite message.
- **alerting**: This state exists for a I1 session terminated at the UE when the ICS UE has sent an I1 Progress message with Progress reason set to Ringing indicating that local alerting has been initiated but the offered call has not yet answered.
- **confirmed**: This state exists for a I1 session terminated at the UE when the ICS UE has sent an I1 Success message indicating that the I1 session has been accepted.

7.5.2.2.2 Session states at SCC AS – ICS UE terminated session

This subclause lists the session states that may exist in the UE for a session terminated at the ICS UE.

- **null**: No session exists.
- **trying**: This state exists for a session terminated at the ICS UE when the SCC AS has requested a session establishment by sending an I1 Invite message but has not yet received a response.
- **proceeding**: This state exists for a session terminated at the ICS UE when the SCC AS has received an I1 Progress message with Progress reason set to Call progressing from the UE acknowledging that the ICS UE has received the I1Invite message.
- **alerted**: This state exists for an UE terminated session when the SCC AS has received an I1 Progress message with Progress reason set to Ringing indicating that the UE has initiated local alerting.
- **confirmed**: This state exists for a session terminated at the UE when the SCC AS has received an I1 Success message indicating that the ICS UE has accepted the session.

7.5.2.3 Session release

7.5.2.3.1 Session states at ICS UE

This subclause lists the session states that may exist at the UE for a session released either by the ICS UE or SCC AS.

- **release-requested**: This state exists when the ICS UE has requested the SCC AS to clear the session by sending an I1 Bye message and the CS bearer has not been cleared using receipt of a DISCONNECT message, in accordance with 3GPP TS 24.008 [3]. Upon determining that the CS bearer has cleared using a DISCONNECT message or determining that an I1 Success message was received, the UE transits to a "null" state. The ICS UE attempts to clear the CS bearer if a retransmission timer fires.
- **release-indication**: This state exists when the ICS UE has received an I1 Bye message from the SCC AS requesting the UE to clear the session. Per subclause 6.2.3.1.2, upon subsequent clearing the CS bearer using a DISCONNECT message, in accordance with 3GPP TS 24.008 [3], or upon subsequent sending of an I1 Success message, the ICS UE transits to a "null" state.

NOTE: The retransmission timer, which is not defined in this specification, is selected appropriately for the transport layer in use.

7.5.2.3.2 Session states at SCC AS

This subclause lists the session states that may exist at the SCC AS for a session released either by the ICS UE or SCC AS.

- **release-requested**: This state exists when the SCC AS has requested the ICS UE to clear the session by sending an I1 Bye message and the CS bearer has not been cleared. Upon determining that the CS bearer has cleared using a DISCONNECT message, in accordance with 3GPP TS 24.008 [3] and 3GPP TS 24.292 [5] or determining that a I1 Success message was received, the SCC AS transits to a "null" state. The SCC AS attempts to clear the CS bearer if a retransmission timer fires.
- **release-indication:** This state exists when the SCC AS has received an I1Bye message from the ICS UE requesting the SCC AS to clear the session. Upon subsequent clearing the CS bearer using a SIP BYE request sent towards the MGCF or upon subsequent sending of an I1 Success message in accordance with 3GPP TS 24.292 [5], the SCC AS transits to a "null" state.

NOTE: The retransmission timer, which is not defined in this specification, is selected appropriately for the transport layer in use.

7.5.3 Session control procedures

7.5.3.1 General

Before the I1 session establishment procedures are invoked, a transport-layer connection must be established between the ICS UE and the SCC AS.

7.5.3.2 Session establishment

7.5.3.2.1 UE-originating case

7.5.3.2.1.1 Procedure at ICS UE

7.5.3.2.1.1.1 Session request

The ICS UE initiates I1 session establishment procedure by sending an I1 Invite message to the SCC AS across the I1 interface. The I1 Invite message shall contain the I1 information elements as specified in subclause 6.2.1.2.1. Following the transmission of the I1 Invite message the I1 session shall transit to the "trying" state. When the I1 session identified by the Call-ID (see subclause 7.2.2.1.4) enters the "trying" state, the ICS UE sets timer F to fire in T3 seconds.

If an unreliable transport-layer connection between the ICS UE and the SCC AS is used, the ICS UE sets timer E to fire in T1 seconds. For reliable transport-layer connection timer E is not used. If timer E fires while the I1 session is still in the "trying" state, the original I1 Invite message (with the same Call-ID and sequence number) is retransmitted and the timer E is reset to value of MIN(2*T1, T2). If the timer E fires again, the original I1 Invite message (with the same sequence number) is retransmitted again and the timer E is reset to a MIN (4*T1, T2). This process continues so that retransmissions occur with an exponentially increasing interval that caps at T2.

NOTE 1: Since the values for the timers T1, T2 and T3 depend on the technology that is used to implement the transport-layer connection (e.g. USSD), the values for the timers T1, T2 and T3 will be specified for each technology.

If timer F fires while the session is still in the "trying" state, the sessioncall establishment has failed, and the ICS UE clears the I1 session, as described in subclause 6.2.3. In addition, if an unreliable transport-layer connection between the UE and the SCC AS is used the, the ICS UE disables timer E.

7.5.3.2.1.1.2 Session proceeding

If an I1 Progress message with Progress reason set to Call progressing and containing an SCC AS PSI DN is received at the ICS UE while the I1 session identified by a valid Call-ID (see subclause 7.2.2.1.4) is in the "trying" state, the I1 session shall transit to the "proceeding" state. If an an unreliable transport-layer connection between the ICS UE and the SCC AS is used timer E shall be stopped and cleared.

If an unreliable transport-layer connection between the ICS UE and the SCC AS is used, when the session enters the "proceeding" state the ICS UE sets the timer E to fire in T2 seconds. If timer E fires while the session is in the "proceeding" state, the original I1 Invite message (with the same sequence number) is retransmitted and the timer E is reset to a value of T2 seconds. This process continues so that retransmissions of the original I1 Invite message occur every T2 seconds.

If timer F fires while the session is in the "proceeding" state, the session establishment has failed, and the ICS UE clears the call, as described in subclause 6.2.3. In addition, if an unreliable transport-layer connection between the ICS UE and the SCC AS is used the, the ICS UE disables timer E.

Upon receiving the I1 Progress message with Progress reason set to Call progressing from the SCC AS, the ICS UE initiates the setting up of the CS bearer connection toward the SCC AS by sending a SETUP message to the MSC Server as specified in subclause 6.2.1.2.1.

NOTE: The request to set up the CS bearer connection arriving at the SCC AS indicates that the I1 Progress message with Progress reason set to Call progressing has been received by the UE. Subsequently, the SCC AS can progress the I1 session toward the far end by sending a SIP INVITE request to the far end.

7.5.3.2.1.1.3 Alerting indication

If an I1 Progress message with Progress reason set to Ringing is received while the I1 session identified by a valid Call-ID (see subclause 7.2.2.1.4) is in the "proceeding" state, the I1 session shall transit to the "alerted" state.

If an unreliable transport-layer connection between the ICS UE and the SCC AS is used, when the session enters the "alerted" state the ICS UE sets timer E to fire in T2 seconds. If timer E fires while the session is in the "alerted" state, the original I1 Invite message (with the same sequence number) is retransmitted and the timer E is reset to a value of T2 seconds. This process continues so that retransmissions of the original I1 Invite request occur every T2 seconds.

If timer F fires while the session is in the "alerted" state, the session establishment has failed, and the ICS UE clears the call, as described in subclause 6.2.3. In addition, if an unreliable transport-layer connection between the ICS UE and the SCC AS is used, the ICS UE disables timer E.

If the ICS UE receives an I1 Progress message with Progress reason set to Ringing, the ICS UE may begin a locally-generated alerting procedure.

7.5.3.2.1.1.4 Session connected

If an I1 Success message is received from the SCC AS while the I1 session at the UE is either in the "proceeding" state or "alerted" state, the I1 session shall transit to the "confirmed" state (i.e., the I1 session has been established) and the timer F is disabled. The ICS UE shall stop any locally generated alerting procedures (if applied).

If an unreliable transport-layer connection between the ICS UE and the SCC AS was used, the timer E is disabled, hence the ICS UE stops retransmitting the I1 Invite message. In addition, the ICS UE discards any subsequent I1 Success message, if it is received over the unreliable transport-layer connection.

7.5.3.2.1.2 Procedure at SCC AS

7.5.3.2.1.2.1 Session request

Upon receiving an I1 Invite message from the ICS UE over the I1 interface, the session at the SCC AS shall transit to the "initiated" state. Once in the "initiated" state, the SCC AS shall immediately respond by sending an I1 Progress message with Progress reason set to Call progressing to the UE and the session enters the "progressing" state, The I1 Progress message with Progress reason set to Call progressing shall contain the I1 information elements as specified in subclause 6.2.1.3.1.

NOTE: The receipt of the I1 Progress message with Progress reason set to Call progressing at the UE, will trigger the UE to set up a CS bearer connection toward the SCC AS by sending a SETUP message to the MSC Server as specified in subclause 6.2.1.2.1.

7.5.3.2.1.2.2 Session progressing

If the SCC AS receives a retransmitted I1 Invite message from the ICS UE, while the I1 session is in the "progressing" state, the SCC AS shall retransmit the previously sent I1 Progress message with Progress reason set to Call progressing to the ICS UE.

NOTE: The SCC AS receives a retransmitted I1 Invite message only if the transport-layer connection between the ICS UE and the SCC AS is an unreliable transport-layer connection. While the I1 session is in the "progressing" state, the SCC AS may send to the ICS UE either an I1 Progress message with Progress reason set to Ringing, an I1 Success message indicating that the I1 session has been accepted, or a new I1 Progress response with Progress reason set to Call progressing.

If timer F fires while the session is in the "progressing" state, the I1 session establishment has failed, and the SCC AS clears the I1 session, as described in subclause 6.2.3.

7.5.3.2.1.2.3 Alerting indication

If the SCC AS sends an I1 Progress message with Progress reason set to Ringing to the ICS UE, the session state at the SCC AS shall transit to the "alerting" state.

If the SCC AS receives a retransmitted I1 Invite message from the ICS UE, while the session is in the "alerting" state, the SCC AS shall retransmit the previously sent I1 Progress message with Progress reason set to Ringing to the ICS UE.

NOTE: The SCC AS receives a retransmitted I1 Invite message only if the transport-layer connection between the UE and the SCC AS is an unreliable transport-layer connection.

If timer F fires while the session is in the "alerting" state, the session establishment has failed, and the SCC AS clears the session, as described in subclause 6.2.3.

7.5.3.2.1.2.4 Session connected

If the SCC AS sends an I1 Success message to the ICS UE indicating that the session has been accepted, the session state at the SCC AS shall transit to the "confirmed" state.

If an unreliable transport-layer connection between the UE and the SCC AS is used, when the session enters the "confirmed" state the SCC AS sets timer G to fire in (n*T2) seconds. For reliable transport-layer connection timer G is not used. If a retransmitted I1 Invite message is received while the timer G is running, the timer G is reset to fire in (n*T2) seconds, and the I1 Success message is retransmitted. The firing of the timer G indicates that the ICS UE has received the I1 Success message and the ICS UE has stopped the retransmission of the I1 Invite message.

If timer G fires while the session is in the "confirmed" state, the timer F is disabled.

If timer F fires while the session is in the "proceeding" state, the session establishment has failed, and the SCC AS resets timer G and clears the session, as described in subclause 6.2.3.

7.5.3.2.2 UE-terminating case

7.5.3.2.2.1 Procedure at ICS UE

7.5.3.2.2.1.1 Session request

Upon receiving an I1 Invite message from the SCC AS over the I1 interface, the session at the ICS UE shall transit to the "initiated" state. Once in the "initiated" state, the ICS UE shall immediately respond by sending an I1 Progress message with Progress reason set to Call progressing to the SCC AS and enter the "progressing" state. The I1 Progress message with Progress reason set to Call progressing shall contain the I1 information elements as specified in subclause 6.2.1.2.2.

NOTE: The receipt of the I1 Invite message at the UE will trigger the UE to set up a CS bearer connection toward the SCC AS by sending a SETUP message to the MSC Server as specified in subclause 6.2.1.2.1.

When the session enters the "initiated" state, the ICS UE also sets timer F to fire in T3 seconds.

7.5.3.2.2.1.2 Session progressing

If the ICS UE receives a retransmitted I1 Invite message from the SCC AS, while the I1 session is in the "progressing" state, the ICS UE shall retransmits the previously sent I1 Progress message with Progress reason set to Call progressing to the UE.

NOTE: The UE receives a retransmitted I1 Invite message only if the transport-layer connection between the UE and the SCC AS is an unreliable transport-layer connection.

While the session is in the "progressing" state, the ICS UE may send to the SCC AS with the same Call-ID, either an I1 Progress message with Progress reason set to Ringing, an I1 Success message indicating that the call has been accepted, or a new I1 Progress message with Progress reason set to Call progressing.

If timer F fires while the session is in the "progressing" state, the session establishment has failed, and the ICS UE clears the call, as described in subclause 6.2.3.

7.5.3.2.2.1.3 Alerting indication

If the ICS UE sends an I1 Progress message with Progress reason set to Ringing, the session state at the ICS UE shall transit to the "alerting" state.

If the ICS UE receives a retransmitted I1 Invite message from the SCC AS with the same Call-ID, while the session is in the "alerting" state, the ICS UE shall retransmit the previously sent I1 Progress message with Progress reason set to Ringing to the SCC AS.

NOTE: The UE receive a retransmitted I1 Invite message only if the transport-layer connection between the UE and the SCC AS is an unreliable transport-layer connection.

If timer F fires while the session is in the "alerting" state, the session establishment has failed, and the UE clears the session, as described in subclause 6.2.3.

7.5.3.2.2.1.4 Session connected

If the ICS UE sends an I1 Success message indicating that the session has been accepted, the session state at the UE transits to the "confirmed" state.

If an unreliable transport-layer connection between the UE and the SCC AS is used, the ICS UE sets timer G to fire in (n*T2) seconds. For reliable transport-layer connection timer G is not used. If an I1 Invite message is received while the timer G is running, the timer G is reset to (n*T2) seconds, and the I1 Success message is retransmitted. The firing of the timer G indicates that the SCC AS has received the Success message and has stopped the retransmission of the I1 Invite message.

If timer G fires while the session is in the "confirmed" state, the timer F is disabled.

If timer F fires while the session is in the "confirmed" state, the session establishment has failed, and the ICS UE clears the session, as described in subclause 6.2.3.

7.5.3.2.2.2 Procedure at SCC AS

7.5.3.2.2.2.1 Session request

The SCC AS initiates a session establishment procedure by sending an I1 Invite message to the UE across the I1 interface. The I1 Invite message shall contain the I1 information elements as specified in subclause 6.2.1.3.2. Following the transmission of the I1 Invite message the session shall transit to the "trying" state. When the session enters the "trying" state, the SCC AS sets timer F to fire in T3 seconds.

If an unreliable transport-layer connection between the UE and the SCC AS is used, the SCC AS sets timer E to fire in T1 seconds. For reliable transport-layer connection timer E is not used. If timer E fires while the session is still in the "trying" state, the original I1 Invite message (with the same sequence number) is retransmitted and the timer E is reset to value of MIN(2*T1, T2). If the timer E fires again, the original I1 Invite message (with the same Call-ID and sequence number) is retransmitted again and the timer E is reset to a MIN (4*T1, T2). This process continues so that retransmissions occur with an exponentially increasing interval that caps at T2.

If timer F fires while the session is still in the "trying" state, the session establishment has failed, and the SCC AS clears the session, as described in subclause 6.2.3. In addition, if an unreliable transport-layer connection between the UE and the SCC AS is used, the SCC AS disables timer E.

7.5.3.2.2.2.2 Call proceeding

If an I1 Progress message with Progress reason set to Call progressing is received at the SCC AS while the session is in the "trying" state, the session shall transit to the "proceeding" state.

If an unreliable transport-layer connection between the UE and the SCC AS is used, when the session enters the "proceeding" state the SCC AS sets timer E to fire in T2 seconds. If timer E fires while the session is in the "proceeding" state, the original I1 Invite message (with the same sequence number) is retransmitted and the timer E is reset to a value of T2 seconds. This process continues so that retransmissions of the original I1 Invite message occur every T2 seconds.

If timer F fires while the session is in the "proceeding" state, the session establishment has failed, and the SCC AS clears the session, as described in subclause 6.2.3. In addition, if an unreliable transport-layer connection between the UE and the SCC AS is used, the SCC AS disables timer E.

NOTE: The request to set up the CS bearer connection arriving at the SCC AS indicates that the I1 Invite message has been received by the UE.

7.5.3.2.2.2.3 Alerting indication

If an I1 Progress message with Progress reason set to Ringing is received while the session is in the "proceeding" state, the session shall transit to the "alerted" state.

If an unreliable transport-layer connection between the UE and the SCC AS is used, when the session enters the "alerted" state the SCC AS sets timer E to fire in T2 seconds. If timer E fires while the session is in the "alerted" state, the original I1 Invite message (with the same sequence number) is retransmitted and the timer E is reset to a value of T2 seconds. This process continues so that retransmissions of the original I1 Invite message occur every T2 seconds.

If timer F fires while the session is in the "alerted" state, the session establishment has failed, and the SCC AS clears the session, as described in subclause 6.2.3. In addition, if an unreliable transport-layer connection between the UE and the SCC AS is used, the SCC AS disables timer E.

7.5.3.2.2.2.4 Session connected

If an I1 Success message is received from the ICS UE while the I1 session at the SCC AS is either in the "proceeding" state or "alerted" state, the I1 session shall transit to the "confirmed" state (i.e., the I1 session has been established) and the timer F is disabled.

If an unreliable transport-layer connection between the UE and the SCC AS was used, the timer E is disabled, hence the SCC AS stops retransmitting the I1 Invite message. In addition, the SCC AS discards any subsequent I1 Success message, if it is received over the unreliable transport-layer connection.

7.5.3.3 I1 service control signalling release

7.5.3.3.1 Initiating release of I1 service control signalling

The ICS UE or the SCC AS can release a I1 service control signalling session at any time irrespective of its state. The ICS UE or the SCC AS releases the I1 service control signalling session by sending an I1 Bye message across the I1 interface. The I1 Bye message shall contain the I1 information elements as specified in subclause 6.2.3.

If an I1 Success message is received while the I1 service control signalling session is in the "release-requested" state, it transits to the "null" state (i.e., the I1 service control signalling session has been released).

7.5.3.3.2 Responding to release of I1 service control signalling

If either the ICS UE or the SCC AS receives an I1 Bye message across the I1 interface, the state of the I1 service control signalling session at the recipient side of the I1 Bye message (i.e. either at the ICS UE or the SCC AS) shall transit to the "release-indication" state. If there are more I1 service control signalling sessions, once in the "release-indication" state, the recipient side of the I1 Bye message shall immediately respond by sending an I1 Success message.

Annex A (normative): Data structure associating keys with values

A.1 General

A UE and a SCC AS maintain a hash table associating keys with values. The keys shall be hashes resulting of applying a hashing function to string values. SHA-1 shall be used as the hash algorithm.

A.2 Associating keys with values

The UE and the SCC AS shall have one or more tables associating keys with values.

A.2.1 Associating keys with public user identities

The UE and the SCC AS shall create a hash table of the SIP URIs present in the P-Associated-URI header field. If the UE and SCC AS also subscriber to the Reg-Event package as documented in 3GPP TS 24.229 [12] the UE and SCC AS shall create a hash table of the GRUU's for URIs received in the Reg-Event package in addition to those received in the P-Associated-URI header field.

NOTE: The 200 (OK) response to a incoming REGISTER request includes a P-Associated-URI header field and is delivered to the SCC AS as part of the third party registration procedures documented in 3GPP TS 24.229 [12].

Annex B (informative): Change history

Change history							
Date	**TSG #**	**TSG Doc.**	**CR**	**Rev**	**Subject/Comment**	**Old**	**New**
2009-04	CT1#58	C1-092099			Initial skeleton from rapporteur	-	0.0.0
2009-04	CT1#58	C1-092097			Scope of TS 24.294	0.0.0	0.1.0
2009-06	CT1#59	C1-092980			I1 messages	0.1.0	0.2.0
2009-06	CT1#59	C1-092981			Text for introduction	0.1.0	0.2.0
2009-06	CT1#59	C1-093056			Procedures for session setup when terminated in ICS UE	0.1.0	0.2.0
2009-06	CT1#59	C1-093067			I1 protocol overview	0.1.0	0.2.0
2009-06	CT1#59	C1-093068			Text for session setup	0.1.0	0.2.0
2009-08	CT1#60	C1-093244			I1 Call States	0.2.0	0.3.0
2009-08	CT1#60	C1-093368			Procedure for adding I1 control to existing CS session (I1 augmentation)	0.2.0	0.3.0
2009-08	CT1#60	C1-093727			Corrections to I1 protocol overview	0.2.0	0.3.0
2009-08	CT1#60	C1-093728			I1 message encoding	0.2.0	0.3.0
2009-08	CT1#60	C1-093729			I1 for Supplementary Service Invocation	0.2.0	0.3.0
2009-08	CT1#60	C1-093734			I1 Call origination at UE	0.2.0	0.3.0
2009-08	CT1#60	C1-093735			I1 Call origination at SCC AS	0.2.0	0.3.0
2009-08	CT1#60	C1-093736			I1 Call terminated at UE	0.2.0	0.3.0
2009-08	CT1#60	C1-093741			Procedure for session termination with UE assisted T-ADS	0.2.0	0.3.0
2009-08	CT1#60	C1-093742			Procedure for Gm fallback to I1	0.2.0	0.3.0
2009-08	CT1#60	C1-093743			I1 protocol functionality	0.2.0	0.3.0
2009-08	CT1#60	C1-093922			I1 Refer	0.2.0	0.3.0
2009-08	CT1#60	C1-093924			I1 information elements	0.2.0	0.3.0
2009-08	CT1#60	C1-093929			I1 information element format	0.2.0	0.3.0
2009-08	CT1#60	C1-093936			I1 message structure	0.2.0	0.3.0
2009-09					Editorial fixes	0.3.0	0.3.1
2009-10	CT1#61	C1-094053			Call origination at UE	0.3.1	0.4.0
2009-10	CT1#61	C1-094056			Call termination at UE	0.3.1	0.4.0
2009-10	CT1#61	C1-094058			From-id and To-id encoding	0.3.1	0.4.0
2009-10	CT1#61	C1-094060			Replaces informat element	0.3.1	0.4.0
2009-10	CT1#61	C1-094352			Cleanup of TS 24.294	0.3.1	0.4.0
2009-10	CT1#61	C1-094502			Message Formats	0.3.1	0.4.0
2009-10	CT1#61	C1-094503			Call origination at SCC AS	0.3.1	0.4.0
2009-10	CT1#61	C1-094504			Call termination at SCC AS	0.3.1	0.4.0
2009-10	CT1#61	C1-094505			Error-code information element	0.3.1	0.4.0
2009-10	CT1#61	C1-094506			Privacy, SCC-AS-id, and Session-identifier encoding	0.3.1	0.4.0
2009-10	CT1#61	C1-094507			I1 Call release	0.3.1	0.4.0
2009-10	CT1#61	C1-094587			Alignment with TS 24.292	0.3.1	0.4.0
2009-10					Editorial fixes	0.4.0	0.4.1
2009-11	CT1#62	C1-094892			Conveying the STI to the UE	0.4.1	0.5.0
2009-11	CT1#62	C1-094893			SCC AS assigning the dynamic STI	0.4.1	0.5.0
2009-11	CT1#62	C1-094894			Conveying the STI for call termination	0.4.1	0.5.0
2009-11	CT1#62	C1-095127			Removal and correction of redundant Editor's Notes	0.4.1	0.5.0
2009-11	CT1#62	C1-095128			Functional entities	0.4.1	0.5.0
2009-11	CT1#62	C1-095133			Correction of tables	0.4.1	0.5.0
2009-11	CT1#62	C1-095135			Resolve Editor's notes with including SDP information	0.4.1	0.5.0
2009-11	CT1#62	C1-095414			Session-identifier	0.4.1	0.5.0
2009-11	CT1#62	C1-095415			Definitions	0.4.1	0.5.0
2009-11	CT1#62	C1-095460			Remove the description of the USSD in 7.1	0.4.1	0.5.0
2009-11	CT1#62	C1-095461			General behaviour of ICS UE and SCC AS	0.4.1	0.5.0
2009-11	CT1#62	C1-095462			Correction of the introduction of I1 protocol	0.4.1	0.5.0
2009-11	CT1#62	C1-095463			Session release	0.4.1	0.5.0
2009-11	CT1#62	C1-095464			On Replaces information element	0.4.1	0.5.0
2009-11	CT1#62	C1-095465			I1 Bye procedures	0.4.1	0.5.0
2009-11					Editorial fixes	0.5.0	0.5.1
2009-12	CT#46				V1.0.0 created by MCC for presentation to CT-46 for information and approval	0.5.1	1.0.0
2009-12	CT#46				V9.0.0 created by MCC after approval at CT-46	1.0.0	9.0.0
2010-03	CT#47	CP-100137	0001	1	No STN	9.0.0	9.1.0
2010-03	CT#47	CP-100137	0002	1	No forking for I1 protocol	9.0.0	9.1.0
2010-03	CT#47	CP-100137	0005	1	Call-Identifier	9.0.0	9.1.0
2010-03	CT#47	CP-100137	0006	1	Clean up of linkage between I1 specifications	9.0.0	9.1.0
2010-03	CT#47	CP-100137	0008	1	Clarification of call set-up procedures	9.0.0	9.1.0

2010-03	CT#47	CP-100137	0009	1	Completion of setting BCD calling parameter for I1 calls	9.0.0	9.1.0
2010-03	CT#47	CP-100137	0012	1	Add missing references	9.0.0	9.1.0
2010-03	CT#47	CP-100137	0013	1	Delete unneeded definitions	9.0.0	9.1.0
2010-03	CT#47	CP-100137	0014	1	Delete unneeded Editor's Notes	9.0.0	9.1.0
2010-03	CT#47	CP-100137	0015		Privacy information element	9.0.0	9.1.0
2010-03	CT#47	CP-100137	0016		Replaces information element	9.0.0	9.1.0
2010-03	CT#47	CP-100137	0017		Gm fallback to I1	9.0.0	9.1.0
2010-03	CT#47	CP-100137	0018	1	Retaining the use of CS access – procedure at UE	9.0.0	9.1.0
2010-03	CT#47	CP-100137	0019	1	Retaining the use of CS access - procedure at SCC AS	9.0.0	9.1.0
2010-03	CT#47	CP-100137	0020	2	Media-transfer from PS to CS access	9.0.0	9.1.0
2010-03	CT#47	CP-100137	0021	2	Transferring the media from PS to CS domain – procedure at UE	9.0.0	9.1.0
2010-03	CT#47	CP-100137	0022	1	Transferring the media from PS to CS domain – procedure at SCC AS	9.0.0	9.1.0
2010-03	CT#47	CP-100137	0023		Removal of editor's note	9.0.0	9.1.0
2010-03	CT#47	CP-100137	0024		Editor's note removal	9.0.0	9.1.0
2010-03	CT#47	CP-100137	0025	1	State machine clarifications	9.0.0	9.1.0
2010-03	CT#47	CP-100137	0026	2	Ability to send reduced SIP/Tel URIs	9.0.0	9.1.0
2010-03	CT#47	CP-100137	0027	1	Inclusion of Accept / Reject contact capabilities	9.0.0	9.1.0
2010-03	CT#47	CP-100137	0028		Remove unneeded Editor's Notes	9.0.0	9.1.0
2010-03	CT#47	CP-100137	0029		Editorial modification	9.0.0	9.1.0
2010-03	CT#47	CP-100137	0030		Session Modification	9.0.0	9.1.0
2010-03	CT#47	CP-100137	0031		Address the I1 dummy message related Editor's Note	9.0.0	9.1.0
2010-03	CT#47	CP-100137	0032		Supplementary services control procedures	9.0.0	9.1.0
2010-03	CT#47	CP-100137	0033		I1 Mid Call information element	9.0.0	9.1.0
2010-03	CT#47	CP-100137	0034		I1 SIP Error cause handling	9.0.0	9.1.0
2010-03	CT#47	CP-100137	0036		Detection of stale message	9.0.0	9.1.0
2010-06	CT#48	CP-100357	0037	1	Removal editors note on inclusion of calling party identity in setup message	9.1.0	9.2.0
2010-06	CT#48	CP-100357	0038	1	Identification how IE"s are encoded.	9.1.0	9.2.0
2010-06	CT#48	CP-100357	0039	2	Clarification of code specific values in IE"s.	9.1.0	9.2.0
2010-06	CT#48	CP-100357	0041		Removal of Editors notes on URI coding	9.1.0	9.2.0
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2010-06	CT#48	CP-100357	0052		Use of short message as a transport layer protocol	9.1.0	9.2.0
2010-06	CT#48	CP-100357	0053	1	Removal of undefined information elements from procedures	9.1.0	9.2.0
2010-06	CT#48	CP-100357	0055	1	Service control transfer (Gm fallback to I1) - Service continuity after transferring multiple calls from PS access to CS access using SRVCC	9.1.0	9.2.0
2010-06	CT#48	CP-100357	0056	2	Privacy requested	9.1.0	9.2.0
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2010-06	CT#48	CP-100357	0068		Editorial corrections to introductory clauses	9.1.0	9.2.0
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