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650 Route des Lucioles
F-06921 Sophia Antipolis Cedex - FRANCE

Tel.: +33 4 92 94 42 00 Fax: +33 4 93 65 47 16

Siret N° 348 623 562 00017 - NAF 742 C
Association à but non lucratif enregistrée à la
Sous-Préfecture de Grasse (06) N° 7803/88

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Contents

Foreword

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1 Scope

IP Multimedia (IM) Core Network (CN) subsystem Service Continuity (SC) provides the capability of continuing ongoing communication sessions with multiple media across different access networks.

The present document provides the protocol details for enabling IMS SC based on the Session Initiation protocol (SIP) and the Session Description Protocol (SDP) and the protocols of the 3GPP Circuit-Switched (CS) domain (e.g. CAP, MAP, ISUP, BICC and the NAS call control protocol for the CS access).

The present document is applicable to User Equipment (UEs), Application Servers (AS), MSC Servers providing IMS Service Continuity capabilities, Emergency Access Transfer Function (EATF), Access Transfer Control Function (ATCF).

2 References

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

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

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

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

[3] 3GPP TS 24.228 Release 5: "Signalling flows for the IP multimedia call control based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP); Stage 3".

[4] 3GPP TS 24.292: "IP Multimedia (IM) Core Network (CN) subsystem Centralized Services (ICS); Stage 3".

[5] 3GPP TS 24.216: "Communication continuity managed object".

[6] 3GPP TS 29.328: "IP Multimedia Subsystem (IMS) Sh interface; Signalling flows and message contents".

[7] 3GPP TS 29.329: "Sh interface based on the Diameter protocol; Protocol details".

[8] 3GPP TS 24.008: "Mobile radio interface layer 3 specification; Core Network protocols; Stage 3".

[9] 3GPP TS 23.237: "IP Multimedia subsystem (IMS) Service Continuity; Stage 2".

[10] IETF RFC 3891: "The Session Initiation Protocol (SIP) "Replaces" Header".

[11] IETF RFC 4538: "Request Authorization through Dialog Identification in the Session Initiation Protocol (SIP)".

[12] 3GPP TS 23.003: "Numbering, addressing and identification".

[13] IETF RFC 3515: "The Session Initiation Protocol (SIP) Refer Method".

[14] Void.

[15] 3GPP TS 23.228: "IP Multimedia Subsystem (IMS); Stage 2".

[16] IETF RFC 5012 (January 2008): "Requirements for Emergency Context Resolution with Internet Technologies".

[17] IETF RFC 5031 (January 2008): "A Uniform Resource Name (URN) for Services".

[18] 3GPP TS 29.292: "Interworking between the IP Multimedia (IM) Core Network (CN) subsystem and MSC Server for IMS Centralized Services (ICS)".

[19] IETF RFC 3261: "SIP: Session Initiation Protocol".

[20] IETF RFC 4488: "Suppression of Session Initiation Protocol (SIP) REFER Method Implicit Subscription".

[21] IETF RFC 3023: "XML Media Types".

[22] IETF RFC 5626: "Managing Client Initiated Connections in the Session Initiation Protocol (SIP)".

[23] 3GPP TS 24.286: "IP Multimedia (IM) Core Network (CN) subsystem Centralised Services (ICS); Management Object (MO)".

[24] 3GPP TS 22.173: "IP Multimedia Core Network Subsystem (IMS) Multimedia Telephony Service and supplementary services; Stage 1".

[25] 3GPP TS 24.607: "Originating Identification Presentation (OIP) and Originating Identification Restriction (OIR) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol Specification".

[26] 3GPP TS 24.608: "Terminating Identification Presentation (TIP) and Terminating Identification Restriction (TIR) using IP Multimedia (IM)Core Network (CN) subsystem; Protocol Specification".

[27] 3GPP TS 24.604: "Communication Diversion (CDIV) using IP Multimedia (IM)Core Network (CN) subsystem; Protocol specification".

[28] 3GPP TS 24.610: "Communication HOLD (HOLD) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification".

[29] 3GPP TS 24.611: "Anonymous Communication Rejection (ACR) and Communication Barring (CB); using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification".

[30] 3GPP TS 24.606: "Message Waiting Indication (MWI) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification".

[31] 3GPP TS 24.605: "Conference (CONF) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification".

[32] 3GPP TS 24.629: "Explicit Communication Transfer (ECT) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification".

[33] 3GPP TS 24.647: "Advice Of Charge (AOC) using IP Multimedia (IM)Core Network (CN) subsystem; Protocol Specification".

[34] 3GPP TS 24.654: "Closed User Group (CUG) using IP Multimedia (IM) Core Network (CN) subsystem, Protocol Specification".

[35] 3GPP TS 24.239: "Flexible Alerting (FA) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification".

[36] 3GPP TS 24.615: "Communication Waiting (CW) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol Specification".

[37] 3GPP TS 24.642: "Completion of Communications to Busy Subscriber (CCBS) and Completion of Communications by No Reply (CCNR) using IP Multimedia (IM)Core Network (CN) subsystem; Protocol Specification".

[38] 3GPP TS 24.182: "IP Multimedia Subsystem (IMS) Customized Alerting Tones (CAT); Protocol specification".

[39] 3GPP TS 24.616: "Malicious Communication Identification (MCID) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol Specification".

[40] 3GPP TS 24.259: "Personal Network Management (PNM); Stage 3".

[41] 3GPP TS 24.183: "IP Multimedia Subsystem (IMS) Customized Ringing Signal (CRS) service; Stage 3".

[42] 3GPP TS 24.072: "Call Deflection (CD) Supplementary Service; Stage 3".

[43] 3GPP TS 24.083: "Call Waiting (CS) and Call Hold (HOLD) supplementary services; Stage 3".

[44] 3GPP TS 24.294 "IP Multimedia Subsystem (IMS) Centralized Services (ICS) protocol via I1 interface".

[45] Void.

[46] 3GPP TS 24.091: "Explicit Call Transfer (ECT) supplementary service; Stage 3".

[47] 3GPP TS 24.084: "Multi Party (MPTY) supplementary service;Stage 3".

[48] IETF RFC 4235 (November 2005): "An INVITE-Initiated Dialog Event Package for the Session Initiated Protocol (SIP)"..

[49] 3GPP TS 23.216 "Single Radio Voice Call Continuity (SRVCC); Stage 2".

[50] Void.

[51] Void.

[52] 3GPP TS 24.301: "Non-Access-Stratum (NAS) protocol for Evolved Packet System (EPS); Stage 3".

[53] IETF RFC 3840 (August 2004): "Indicating User Agent Capabilities in the Session Initiation Protocol (SIP)".

[54] IETF RFC 6086 (January 2011): "Session Initiation Protocol (SIP) INFO Method and Package Framework".

[55] IETF RFC 4353 (February 2006): "A Framework for Conferencing with the Session Initiation Protocol (SIP)".

[56] Void.

[57] IETF RFC 3326 (December 2002): "The Reason Header Field for the Session Initiation Protocol (SIP)".

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

[59] Void.

[60] IETF RFC 6809 (November 2012): "Mechanism to Indicate Support of Features and Capabilities in the Session Initiation Protocol (SIP)".

[61] 3GPP TS 25.331 "Radio Resource Control (RRC); protocol specification".

[62] 3GPP TS 36.331 "Evolved Universal Terrestrial Radio Access (E-UTRA); Radio Resource Control (RRC); Protocol specification".

[63] 3GPP TS 23.292: "IP Multimedia Subsystem (IMS) Centralized Services; Stage 2".

[64] 3GPP TS 24.337 "IP Multimedia (IM) Core Network (CN) subsystem; inter-UE transfer; Stage 3".

[65] 3GPP TS 23.203: "Policy and charging control architecture".

[66] 3GPP TS 23.107:"Quality of Service (QoS) concept and architecture".

[67] 3GPP TS 23.218: "IP Multimedia (IM) Session Handling; IM call model".

[68] 3GPP TS 26.114: "IP Multimedia Subsystem (IMS); Multimedia Telephony; Media handling and interaction".

[69] 3GPP TS 26.111: "Codec for circuit switched multimedia telephony service; Modifications to H.324".

[70] 3GPP TS 29.274: "Tunnelling Protocol for Control plane (GTPv2-C); Stage 3".

[71] 3GPP TS 29.280: "3GPP Sv interface (MME to MSC, and SGSN to MSC) for SRVCC".

[72] IETF RFC 3263 (June 2002): "Session Initiation Protocol (SIP): Locating SIP Servers".

[73] IETF RFC 7195 (May 2014): "Session Description Protocol (SDP) Extension for Setting Audio and Video Media Streams over Circuit-Switched Bearers in the Public Switched Telephone Network (PSTN)".

[74] IETF RFC 6157 (April 2011): "IPv6 Transition in the Session Initiation Protocol (SIP)".

[75] 3GPP TS 24.007: "Mobile radio interface signalling layer 3; General aspects".

[76] ETSI TS 182 024: "Hosted Enterprise Services; Architecture, functional description and signalling".

[77] 3GPP TS 29.079: "Optimal media routeing within the IP Multimedia Subsystem (IMS)".

[78] IETF RFC 3841 (August 2004): "Caller Preferences for the Session Initiation Protocol (SIP)".

[79] IETF RFC 3966 (December 2004): "The tel URI for Telephone Numbers".

[80] IETF RFC 6228 (May 2011): "Session Initiation Protocol (SIP) Response Code for Indication of Terminated Dialog".

[81] IETF RFC 6665 (July 2012): "SIP-Specific Event Notification".

[82] IETF RFC 7254 (May 2014): "A Uniform Resource Name Namespace for the Global System for Mobile Communications Association (GSMA) and the International Mobile station Equipment Identity (IMEI)".

[83] IETF RFC 7549 (May 2015): "3GPP SIP URI Inter-Operator Traffic Leg Parameter".

[84] 3GPP TS 24.302: "Access to the 3GPP Evolved Packet Core (EPC) via non-3GPP access networks".

[85] 3GPP TS 32.260: "Telecommunication management; Charging management".

[86] IETF RFC 3262 (June 2002): "Reliability of Provisional Responses in the Session Initiation Protocol (SIP)".

[87] IETF RFC 3311 (September 2002): "The Session Initiation Protocol (SIP) UPDATE Method".

[88] IETF RFC 3312 (October 2002): "Integration of Resource Management and Session Initiation Protocol (SIP)".

[89] IETF RFC 4032 (March 2005): "Update to the Session Initiation Protocol (SIP) Preconditions Framework".

[90] IETF RFC 7647 (September 2015): "Clarifications for the Use of REFER with RFC6665".

[91] IETF RFC 3551 (July 2003): "RTP Profile for Audio and Video Conferences with Minimal Control".

[92] IETF RFC 3550 (July 2003): "RTP: A Transport Protocol for Real-Time Applications".

[93] IETF RFC 4566 (July 2006): "SDP: Session Description Protocol".

[94] draft-ietf-insipid-logme-marking-07 (May 2017): "Marking SIP Messages to be Logged".

Editor's note (WI: ISAT, CR#1268): The above document cannot be formally referenced until it is published as an RFC.

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].

Alerting phase: Refers to a SIP session for which all possibly existing dialogs created by the SIP INVITE request initiating the session are early dialogs, for which no final SIP response has been received yet and for which SIP 180 (Ringing) response has already been received in an existing early dialogs.

Pre-alerting phase: Refers to a SIP session for which all possibly existing dialogs created by the SIP INVITE request initiating the session are early dialogs, for which no final SIP response has been received yet and for which SIP 180 (Ringing) response has not been received or sent yet in any existing early dialogs.

Dual radio access transfer for calls in alerting phase: feature enabling dual radio access transfer of a session with speech media component where the session is in an alerting phase.

Dual radio access transfer for originating calls in pre-alerting phase: feature enabling dual radio access transfer of a session with speech media component where the session was originated by the SC UE and the session is in a pre-alerting phase.

Dynamic STI: An STI dynamically assigned by the SCC AS, representing the SIP dialog identifier (Call-ID header field and the values of tags in To and From header fields) and used for session transfer request when Gm service control is available.

Dynamic STN: An STN encoded as an E.164 number in tel URI format dynamically assigned by the SCC AS replacing the static STN during PS to CS dual radio access transfer.

Static STN: An STN configured in the SC UE as an E.164 number. The static STN is used for PS to CS transfer when dynamic STN cannot be used.

Additional transferred session SCC AS URI: A SIP URI which is a public service identity hosted by SCC AS and which is used during PS-CS access transfer with the MSC Server assisted mid-call feature.

Static STI: An STI configured in the SC UE either as a SIP URI or as an E.164 number in tel URI format or SIP URI representation of tel URI. The static STI is used for CS-PS transfer when dynamic STI is unavailable.

PS to PS STI: An STI configured in SC UE either as a SIP URI or as an E.164 number in tel URI format or SIP URI representation of tel URI. The PS to PS STI is used for PS to PS access transfer.

PS to CS STN: An STN that is encoded as an E.164 number. The PS to CS STN is used for PS to CS access transfer.

Speech media component: SDP media component of audio media type with codec suitable for conversational speech. Connection (c=) and attribute (a=) informations at the SDP session level not overridden by information at the SDP media level in the speech media component are considered to be part of the speech media component.

Active speech media component: speech media component which has "recvonly" or "sendrecv" directionality at the SC UE or at the MSC server serving the SC UE.

Inactive speech media component: speech media component which has "sendonly" or "inactive" directionality at the SC UE or at the MSC server serving the SC UE.

Active video media component: video media component which has "recvonly" or "sendrecv" directionality at the SC UE or at the MSC server serving the SC UE.

Inactive video media component: video media component which has "sendonly" or "inactive" directionality at the SC UE or at the MSC server serving the SC UE.

ATCF URI for originating requests: A URI of the ATCF where the ATCF receives requests sent by the served UEs.

ATCF URI for terminating requests: A URI of the ATCF where the ATCF receives requests targeted to the served UEs.

ATCF management URI: A URI hosted by the ATCF where the ATCF performing the role of a UAS receives SIP requests for ATCF management (e.g. SIP MESSAGE requests containing the PS to CS SRVCC related information). The ATCF management URI is routable via the I-CSCF in the network where the ATCF is located using the same routing mechanism as used for Public Service Identities hosted by an AS.

Registration Path: The set of Path header field values and the set of Service-Route header field values created by successful completion of the SIP REGISTER transaction.

SRVCC-related information: Information required by the ATCF to perform PS to CS SRVCC transfer or CS to PS SRVCC transfer or both. It is provided in the MIME body as defined in clause D.3.

UE information for CS to PS SRVCC: Session description containing speech media component that will be used by the ATGW to send media to the SC UE during the CS to PS SRVCC access transfer.

ATGW information for CS to PS SRVCC: Session description containing speech media component that will be used by the SC UE to send media to the ATGW during the CS to PS SRVCC access transfer.

PS to CS SRVCC for calls in alerting phase: feature enabling PS to CS SRVCC of a session with speech media component where the session is in alerting phase.

CS to PS SRVCC for calls in alerting phase: feature enabling CS to PS SRVCC of a session with speech media component where the session is in alerting phase.

PS to CS SRVCC for originating calls in pre-alerting phase: feature enabling PS to CS SRVCC of a session with speech media component where the session was originated by the SC UE and the session is in pre-alerting phase.

PS to CS SRVCC for terminating calls in pre-alerting phase: feature enabling PS to CS SRVCC of a session with speech media component where the session was terminated by the SC UE and the session is in pre-alerting phase.

CS session in an early phase: A CS call for which the CS call setup procedure is not complete, i.e. the CC CONNECT message is not sent or received as described in 3GPP TS 24.008 [8] yet but where either a CC CALL PROCEEDING message as described in 3GPP TS 24.008 [8] has been received or a CC ALERTING message as described in 3GPP TS 24.008 [8] has been sent or received.

Precondition enabled dialog: a dialog (either a confirmed dialog or an early dialog) created by a SIP response containing a Require header field with the precondition option tag.

Precondition enabled initial INVITE request: an initial INVITE request containing a Require header field with the precondition option tag or a Supported header field with the precondition option tag.

RTP payload format: the <encoding name> portion of an "a=rtpmap" attribute according to IETF RFC 4566 [93] for the dynamically assigned RTP payload type numbers or the name of encoding reserved in IETF RFC 3551 [91] table 1 for statically assigned RTP payload type numbers.

RTP payload type number: a number identifying an RTP payload type of a media stream using the RTP based transport protocol. In SDP, the RTP payload type number can be found in a sub-field of an <fmt> portion of an "m=" line, in a <payload type> portion of the "a=rtpmap" attribute and in a <format> portion of an "a=fmtp" attribute, according to IETF RFC 4566 [93]. In RTP, the RTP payload type number is found in the PT field of the RTP header according to IETF RFC 3550 [92].

RTP payload type: an RTP payload type number indicated in a sub-field of an <fmt> portion of an "m=" line and, if included, an "a=rtpmap" attribute and an "a=fmtp" attribute for the RTP payload type number, included in an SDP body, according to IETF RFC 4566 [93].

For the purposes of the present document, the following terms and definitions given in 3GPP TS 23.237 [9] apply:

Access Leg
Access Transfer Control Function (ATCF)

Access Transfer Gateway (ATGW)
Access Transfer Update - Session Transfer Identifier (ATU-STI)
Dual radio
Emergency Session Transfer Number for SR VCC (E-STN-SR)
Home Leg
Local Operating Environment
Remote Leg
Serving Leg Session Transfer Identifier for reverse SRVCC (STI-rSR)
Source Access Leg
Target Access Leg
Emergency Session Transfer Number for DRVCC (E-STN-DRVCC)

For the purposes of the present document, the following terms and definitions given in 3GPP TS 24.292 [4] apply:

CS call
CS media

For the purposes of the present document, the following terms and definitions given in 3GPP TS 23.218 [67] apply:

Initial filter criteria

For the purposes of the present document, the following terms and definitions given in 3GPP TS 23.228 [15] apply:

Policy and Charging Rule Function (PCRF)

For the purposes of the present document, the following terms and definitions given in 3GPP TS 23.003 [12] apply:

Correlation MSISDN
IP Multimedia Routeing Number (IMRN)
Session Transfer Identifier (STI)
Session Transfer Number (STN)
Session Transfer Number for SR-VCC (STN-SR)

For the purposes of the present document, the following terms and definitions given in IETF RFC 5012 [16] apply:

Emergency service URN

For the purposes of the present document, the following terms and definitions given in IETF RFC 4353 [55] apply:

Conference
Conference URI
Focus
Participant

For the purposes of the present document, the following terms and definitions given in IETF RFC 3264 [58] apply:

Directionality

For the purposes of the present document, the following terms and definitions given in 3GPP TS 23.292 [63] apply:

ICS user

For the purposes of the present document, the following terms and definitions given 3GPP TS 24.229 [2] apply:

Authorised Resource-Priority header field
Temporarily Authorised Resource-Priority header field

NOTE: Within the present specification, a Temporarily Authorised Resource-Priority header field can be applied to handling of originating requests in the ATCF.

For the purposes of the present document, the following terms and definitions given in 3GPP TS 24.301 [52] apply:

Persistent EPS bearer context

For the purposes of the present document, the following terms and definitions given 3GPP TS 29.274 [70] apply:

Allocation/Retention Priority (ARP)

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].

1xx	A SIP status-code in the range 101 through 199
18x	A SIP status-code in the range 180 through 189
2xx	A SIP status-code in the range 200 through 299
EATF	Emergency Access Transfer Function
E-STN-SR	Emergency Call Session Transfer Number – Single Radio
E-SR-VCC	Emergency Single Radio Voice Call Continuity
C-MSISDN	Correlation MSISDN
IMRN	IP Multimedia Routing Number
SC	Service Continuity
SCC	Service Centralization and Continuity
SM	Session Management
SRVCC	Single Radio Voice Call Continuity
STI	Session Transfer Identifier
STI-rSR	Session Transfer Identifier for reverse SRVCC
STN	Session Transfer Number
STN-SR	Session Transfer Number - Single Radio
vSRVCC	Single Radio Video Call Continuity

4 Overview of IP Multimedia (IM) Core Network (CN) subsystem Service Continuity

4.1 General

In general, IMS Service Continuity provides the capability of continuing ongoing communication sessions with multiple media across different access networks. The main need for such continuity arises because user equipments (UEs) with multimedia capabilities can move across a multiplicity of different access networks.

NOTE 1: The capability of continuing ongoing communication sessions as a collaboration between different user equipments (UEs) is described in 3GPP TS 24.337 [64].

NOTE 2: In this subclause, the term "PS-CS" is used as a general term to refer to bi-directional access transfer. When required to specify the direction for the access transfer, then the terms "PS to CS" and "CS to PS" are used.

The following procedures are provided within this document:

- procedures for registration in IM CN subsystem are specified in clause 6;
- procedures for call origination are specified in clause 7;
- procedures for call termination are specified in clause 8;
- procedures for PS-CS access transfer are specified in clause 9;

- procedures for PS-PS access transfer are specified in clause 10;
- procedures for PS-PS access transfer in conjunction with PS-CS access transfer are specified in clause 11;
- procedures for PS-CS access transfer for Single Radio are specified in clause 12;
- procedures for media adding/deleting for access transfer are specified in clause 13; and
- procedures for service continuity and MMTEL interactions are specified in clause 20.

In general, the SC UE supports the capabilities of this specification that it needs, see subclause 5.2. Clause G.2 provides a summary of how the capabilities described in this specification are communicated from the SC UE to the network.

Network equipment conforming with this specification is detailed in subclause 5.3 through subclause 5.7, with additional optional procedures specified in clause 7 onwards, but may be summarised as follows:

1) conforming networks support at least one of the following core functionalities:

   a) procedures for PS to CS dual radio access transfer, for a session with an active speech media component;

   b) procedures for PS-PS access transfer, for a session with an active speech media component;

   c) procedures for PS to CS SRVCC for a session with an active speech media component; or

   d) procedures for CS to PS dual radio access transfer, for a session with an active speech media component;

2) for each of the core functionality in 1) above, access transfer for a session with an inactive speech media component may also be supported;

3) for each of the core functionality in 1) above, access transfer for a session with conference control with active speech media component or inactive speech media component may also be supported;

4) for each of the core functionality in 1) above, access transfer in the alerting phase with an active speech media component may also be supported. The alerting phase is applicable only for sessions where a SIP 180 (Ringing) response has been sent or received;

5) if 4) is supported, then PS to CS SRVCC access transfer for originating calls in pre-alerting phase with an active speech media component may also be supported. This applies for an early dialog where a SIP 180 (Ringing) response has not been received;

6) for each of the core functionality in 1) above, access transfer for a session with active speech media component and active video media component may also be supported. If access transfer for a session with active speech media component and active video media component is supported, then:

   a) access transfer in the alerting phase with an active speech media component and active video media component may also be supported. The alerting phase is applicable only for sessions where a SIP 180 (Ringing) response has been sent or received; and

   b) if 6a) is supported, then PS to CS SRVCC for originating calls in pre-alerting phase with an active speech media component and an active video media component may also be supported. This applies for an early dialog where a SIP 180 (Ringing) response has not been received;

7) if 1) c) is supported, then CS to PS SRVCC access transfer for an active session with an active speech media component may also be supported. Within this capability, then:

   a) if 2) is supported, then access transfer for a session with an inactive speech media component may also be supported;

   b) if 3) is supported, then access transfer for a session with conference control with active speech media component or inactive speech media component may also be supported; and

   c) if 4) is supported, then access transfer in the alerting phase with an active speech media component may also be supported; and

8) if 1) a) and 1) d) is supported, then dual radio access transfer for originating calls in pre-alerting phase with an active speech media component may also be supported. This applies for an early dialog where a SIP 180 (Ringing) response has not been received.

In addition to the above, the network can choose to support SRVCC with ATCF and ATGW functionality. For a roaming user, the ATCF allows service continuity to be executed in the visited network as opposed to the home network.

Clause G.3 provides a summary of how the SCC AS and ATCF communicate their support for the access transfer features described in this document, to the SC UE.

For a UE or an AS not supporting ICS procedures, PS-CS access transfer procedures enable transfer of

- one full-duplex session with active speech or speech/video media component; and
- up to one full-duplex session with active speech or speech/video media component and up to one session with inactive speech or speech/video media component when the MSC Server assisted mid-call feature is supported.

4.2 Underlying network capabilities

4.2.1 General

SC assumes the use of a number of underlying network capabilities:

1) provision by the home network operator of SCC AS on the IM CN subsystem, as specified in 3GPP TS 24.229 [2]; and
2) if ICS is used, the network capabilities as specified in 3GPP TS 24.292 [4].

4.2.2 PS-CS session continuity, Single Radio

In order to allow for PS-CS session continuity, Single Radio, SRVCC procedures assume that filter criteria causes all sessions subject to PS to CS SRVCC to be anchored in an SCC AS as described in 3GPP TS 23.216 [5].

Configuration of QoS assignment for PS to CS SRVCC as defined in 3GPP TS 23.203 [65] and 3GPP TS 23.107 [66] need to be aligned with the initial filter criteria and SCC AS determination that a session is subject to SR-VCC as defined in 3GPP TS 23.216 [5].

In order to allow for PS-CS session continuity, Single Radio, vSRVCC procedures assume that filter criteria causes all sessions subject to vSRVCC to be anchored in an SCC AS as described in 3GPP TS 23.216 [5].

Configuration of QoS assignment for vSRVCC as defined in 3GPP TS 23.203 [65] needs to be aligned with the initial filter criteria and SCC AS determination that a session is subject to vSRVCC as defined in 3GPP TS 23.216 [5].

When SRVCC enhanced with ATCF is used, the SRVCC and vSRVCC procedures assume that all sessions subject to SRVCC and vSRVCC are anchored in the same ATCF.

4.2.3 PS to CS and CS to PS session continuity, dual radio access transfer

In order to allow for dual radio access transfer procedures in clauses 9 and 10, the dual radio procedures assume that filter criteria causes all sessions subject to dual radio access transfer to be anchored in an SCC AS as described in 3GPP TS 23.237 [9].

4.3 URI and address assignments

In order to support SC to a subscriber, the following URI and address assignments are assumed:

a) in this version of the document, the SC UE for access transfer will be configured with a static STI, in accordance with subclause 5.11 in 3GPP TS 24.216 [5]; a static STN in accordance with subclause 5.12 in 3GPP TS 24.216 [5]. The static STI is used by the SC UE to perform CS to PS access transfer when no dynamically assigned STI is provided to the UE over the CS domain (e.g. when the SC UE does not support ICS capabilities as defined in 3GPP TS 24.292 [4]). The static STN is used by the SC UE to perform PS to CS access transfer when no service control signalling path as specified in 3GPP TS 24.292 [4] is available; a PS to PS STI URI in accordance with subclause 5.30 in 3GPP TS 24.216 [5].

b) the SC UE will be configured to be reachable in both the IM CN subsystem and the CS domain by one or more public telecommunication numbers which should be correlated between the CS domain and IM CN subsystem. Either:

- this public telecommunication number can be the DN (e.g. MSISDN) used in the CS domain and (in international form) comprise part of the implicit registration set associated with that SC UE in the IM CN subsystem; or
- the SCC AS can be configured to provide a functional relationship between separate numbers providing each of these identities in the CS domain and the IM CN subsystem, respectively.

c) the SCC AS is configured to be reachable using:

- the STN-SR allocated to the SCC AS;
- the additional transferred session SCC AS URI allocated to the SCC AS;
- the additional transferred session SCC AS URI for PS to CS SRVCC allocated to the SCC AS;
- the additional transferred session SCC AS URI for CS to PS SRVCC allocated to the SCC AS;
- the additional transferred session SCC AS URI for PS to CS dual radio allocated to the SCC AS;
- the additional transferred session SCC AS URI for CS to PS dual radio allocated to the SCC AS;
- the ATU-STI for PS to CS SRVCC allocated to the SCC AS;
- the ATU-STI for CS to PS SRVCC allocated to the SCC AS;
- the PS to PS STI for PS to PS access transfer; and
- the dynamic STN allocated to the SCC AS.

d) the ATCF is configured to be reachable using:

- the STN-SR allocated to the ATCF;
- the ATCF URI for originating requests allocated to the ATCF;
- the ATCF URI for terminating requests allocated to the registration path;
- ATCF management URI allocated to the ATCF. The ATCF management URI is included in the g.3gpp.atcf-mgmt-uri feature-capability indicator that the ATCF includes in a Feature-Caps header field in the SIP REGISTER request; and
- ATCF URI for anchoring additionally transferred call in ATCF.

e) the MSC server enhanced for ICS and supporting CS to PS SRVCC is configured to be reachable (in addition to configuration in 3GPP TS 24.292 [4]) using:

- the MSC URI for redirected terminating sessions allocated to the registration path; and
- the MSC server management URI allocated to the MSC server.

4.4 Support of session continuity in enterprise scenarios

Session continuity can be applied where hosted enterprise services are supported as documented in ETSI TS 182 024 [76] the UE registers with the S-CSCF in the normal manner, and the procedures of this document can be used with an SCC AS in the home network.

Where the UE is supported by an application server in the enterprise, any enterprise UE requiring service continuity to be supported by the public network requires an SCC AS in the home network, and therefore registration with an S-CSCF in the home network.

4.5 Guidelines for use of media feature tags or feature capability indicators

NOTE 1: When the values appropriate for use with a media feature tag are of string type, then when included in Contact, Accept-Contact or Reject-Contact header fields, the value of the media feature tag is preceded by "<" and followed by ">" according to IETF RFC 3840 [53] and IETF RFC 3841 [78].

NOTE 2: When the values appropriate for use with feature capability indicators specified in annex C are string, then when the values are included in Feature-Caps header field, the value of the header field is an instance of fcap-string-value of Feature-Caps header field specified in RFC 6809 [60].

5 Functional entities

5.1 Introduction

This clause associates the functional entities with the SC roles described in the stage 2 architecture document (see 3GPP TS 23.237 [9]).

5.2 User Equipment (UE)

To be compliant with access transfer in this document, a UE shall implement the role of an SC UE:

- acting as an UA as defined in 3GPP TS 24.229 [2];
- according to subclause 6.2 for registration of the UE in the IM CN subsystem; and
- dependent on the desired functionality, one or more of the procedures according to subclause 6A.2, subclause 7.2, subclause 8.2, subclause 9.2, subclause 10.2, subclause 11.2, subclause 12.2, subclause 13.2 and subclause 20.1.

5.3 Application Server (AS)

To be compliant with access transfer in this document, an AS shall implement the role of:

1) an AS performing 3rd party call control acting as an routeing B2BUA as defined in 3GPP TS 24.229 [2]; and
2) an SCC AS as follows: dependent on the desired functionality, one or more of the procedures according to subclause 6.3, subclause 6A.4, subclause 7.3, subclause 8.3, subclause 9.3, subclause 10.3, subclause 11.3, subclause 12.3, subclause 13.3 and subclause 20.1.

If the SCC AS receives a SIP INVITE request:

- with either the Replaces header field (see IETF RFC 3891 [10]) or the Target Dialog header field (see IETF RFC 4538 [11]), indicating a dialog identifier of a session belonging to the subscribed user; and
- with the Request-URI not containing the additional transferred session SCC AS URI;

and the SCC AS does not support the procedures for performing PS to PS access transfer specified in subclause 10.3, then the SCC AS shall send a SIP 403 (Forbidden) response to the SIP INVITE request, with a Reason header field containing protocol "SIP" and reason-text set to "PS to PS access transfer not supported".

The SCC AS also handles SDP media description conflicts according to subclause 6A.5.

The SCC AS may also indicate the traffic leg according to subclause 6A.6.

If the SCC AS supports the procedures according to subclause 12.3, the SCC AS shall support procedures according to subclause 22.3.

5.4 MSC server

An MSC server can be compliant with PS to CS SRVCC session transfer procedures as described in this document.

In order to be compliant with PS to CS SRVCC session transfer procedures as described in this document:

- an MSC server using SIP interface to initiate the session transfer shall provide the UA role as defined for a MSC server enhanced for SRVCC using SIP interface in annex A of 3GPP TS 24.229 [2] and the role of an MSC server enhanced for PS to CS SRVCC using SIP interface as described in subclause 12.6.1.1; or

- an MSC server shall provide the role of an MSC server enhanced for ICS as specified in subclause 12.4.0.

If an MSC server is enhanced for ICS and is compliant with PS to CS SRVCC session transfer procedures as described in this document, the MSC server shall also provide the role of an MSC server enhanced for ICS as specified in subclause 22.2.

In order to be compliant with vSRVCC session transfer procedures as described in this document, the MSC server shall be:

- compliant with the PS to CS SRVCC session transfer procedure specified in subclause 12.6.1.1 and additionally provide the functionality to support vSRVCC, as described in subclause 12.6.1.2; or

- compliant with the PS to CS SRVCC session transfer procedure specified in subclauses 12.4.0 and additionally provide the functionality to support vSRVCC, as described in subclause 12.4.0B.

An MSC server can be compliant with the access transfer procedures for the MSC server assisted mid-call feature as described in this document.

In order to be compliant with the access transfer procedures for the MSC server assisted mid-call feature as described in this document, the MSC server shall:

- provide the role of an MSC server enhanced for ICS as described in subclause 6.4 and subclause 9.4 and additionally provide the functionality described in subclause 9.5;

- provide the role of an MSC server enhanced for ICS as described in subclause 12.4.0, and additionally provide the functionality described in subclause 12.4A; or

- provide the role of an MSC server enhanced for PS to CS SRVCC using a SIP interface as described in subclause 12.6.1.1, and additionally provide the functionality described in subclause 12.4A.

In order to enable the UE to remove/add participants from/to an IMS conference call after the access transfer, the MSC Server supporting the MSC server assisted mid-call feature shall provide the role of an MSC server enhanced for ICS.

An MSC server can be compliant with the procedures for the PS to CS SRVCC for calls in alerting phase as described in this document.

In order to be compliant with the procedures for the PS to CS SRVCC for calls in alerting phase as described in this document, the MSC server shall:

- provide the role of an MSC server enhanced for ICS as described in subclause 12.4.0 or subclause 12.4.0B, and additionally provide the functionality described in subclause 12.6.3; or

- provide the role of an MSC server enhanced for SRVCC using a SIP interface as described in subclause 12.6.1 and additionally provide the functionality described in subclause 12.6.3.

The MSC server also handles SDP media description conflicts according to subclause 6A.5.

If the MSC server supports the PS to CS SRVCC for calls in alerting phase, the MSC server may also support the PS to CS SRVCC for originating calls in pre-alerting phase. The procedures for the PS to CS SRVCC for originating calls in pre-alerting phase are described in the subclauses describing the PS to CS SRVCC for calls in alerting phase.

In order to be compliant with PS to CS dual radio access transfer procedures as described in this document an MSC server enhanced for DRVCC using SIP interface to initiate the access transfer shall provide the UA role as defined for an MSC server enhanced for DRVCC using SIP interface in annex A of 3GPP TS 24.229 [2] and the role of an MSC server enhanced for PS to CS dual radio access transfer using SIP interface as described in subclause 9.8.

If a MSC server supports the PS to CS dual radio access transfer for calls in alerting phase, the MSC server shall support UA role procedure defined in IETF RFC 3262 [86] and IETF RFC 3311 [87].

The MSC server may also indicate the traffic leg according to subclause 6A.6.

In all SIP INVITE requests sent by the MSC server, the MSC server shall insert a P-Charging-Vector header field with the "icid-value" header field parameter populated as specified in 3GPP TS 32.260 [85] and a type 1 "orig-ioi" header field parameter. The MSC server shall set the type 1 "orig-ioi" header field parameter to a value that identifies the sending network of the request. The MSC server shall not include the type 1 "term-ioi" header field parameter.

When initiating a failure response to any received request, depending on operator policy, the MSC server may insert a Response-Source header field with an "fe" header field parameter constructed with the URN namespace "urn:3gpp:fe", the fe-id part of the URN set to "msc-server" and optionally an appropriate fe-param part of the URN set in accordance with subclause 7.2.17 of 3GPP TS 24.229 [2].

5.5 EATF

To be compliant with access transfer in this document, the EATF shall act as B2BUA and:

- extract charging information as specified for an AS in 3GPP TS 24.229 [2], subclause 5.7.1.2;
- identify the served user as specified for an AS in 3GPP TS 24.229 [2], subclause 5.7.1.3A.2;
- map the message header fields from a SIP message received in one dialog to related SIP message sent in the correlated dialog managed by EATF as specified for an AS in 3GPP TS 24.229 [2], subclause 5.7.5.1;
- pass signalling elements as specified for an AS in 3GPP TS 24.229 [2], subclause 5.7.5.1;
- handle P-Charging-Vector header as specified for an routeing AS in 3GPP TS 24.229 [2], subclause 5.7.5.1; and
- implement the role of an EATF according to subclause 7.4 and subclause 12.5.

The EATF also handles SDP media description conflicts according to subclause 6A.5.

5.6 Access Transfer Control Function (ATCF)

To be compliant with access transfer in this document, the ATCF shall:

1) provide the proxy role as defined in 3GPP TS 24.229 [2], with the exceptions and additional capabilities as described for the ATCF in subclause 6.5, subclause 6A.3, subclause 7.5, subclause 8.4, and subclause 12.7.2.4;
2) provide the B2BUA functionality with the exceptions and additional capabilities as described for the ATCF in subclause 12.7.2. When providing the B2BUA functionality, the ATCF shall provide the UA role as defined in 3GPP TS 24.229 [2] and additionally shall:
   a. internally map the message header fields from a SIP message received in one dialog to related SIP message sent in the correlated dialog managed by ATCF;
   b. transparently pass supported and unsupported signalling elements (e.g. SIP headers, SIP messages bodies); and
   c. transparently forward received Contact header field, P-Asserted-Identity header field and, if available, the Privacy header field.

The following procedures apply to all procedures at the ATCF:

1) if it has been decided to anchor the media in ATGW according to operator policy, and a SIP message including an SDP offer or answer is received:

NOTE: At this point, ATCF interacts with ATGW to provide information needed in the procedures below, and to request the ATGW to start forwarding the media(s) from the remote UE to the local UE. The details of interaction between ATCF and ATGW are out of scope of this document.

   a. upon the received message with an SDP offer or answer included is sent by the served UE within the dialog, replace the SDP in the received SIP message with updated SDP provided by ATGW, which contains the ATGW IP addresses and ports; and
   b. upon the received message with an SDP offer or answer included is sent by the remote UE within the dialog, replace the SDP in the received SIP message with updated SDP provided by ATGW, which contains the ATGW IP addresses and ports; and

2) the ATCF also handles SDP media description conflicts according to subclause 6A.5.

The ATCF may also indicate the traffic leg according to subclause 6A.6.

The ATCF shall log all SIP requests and responses that contain a "logme" header field parameter, as defined in draft-ietf-insipid-logme-marking [94], in the SIP Session-ID header field if required by local policy.

When initiating a failure response to any received request, depending on operator policy, the ATCF may insert a Response-Source header field with an "fe" header field parameter constructed with the URN namespace "urn:3gpp:fe", the fe-id part of the URN set to "atcf" and optionally an appropriate fe-param part of the URN set in accordance with subclause 7.2.17 of 3GPP TS 24.229 [2].

5.7 Access Transfer Gateway (ATGW)

The functionality of the ATGW is specified in 3GPP TS 23.237 [9].

6 Roles for registration in the IM CN subsystem for service continuity

6.1 Introduction

Void.

6.2 SC UE

6.2.1 Distinction of requests

The SC UE needs to distinguish the following initial SIP requests:

1) SIP MESSAGE requests with the P-Asserted-Identity header field containing the STI-rSR. In the procedures below, such requests are known as "SIP MESSAGE requests with ATGW information for CS to PS SRVCC".

6.2.2 General

Prior to performing IMS registration, if the SC UE supports ICS capabilities as defined in 3GPP TS 24.292 [4], the SC UE shall check that IMS service continuity using ICS is enabled. An indication that SC using ICS is enabled or disabled can be found in the ICS MO ICS_Capabilities_Enabled leaf node (see 3GPP TS 24.286 [23]).

The SC UE shall follow the procedures specified in 3GPP TS 24.229 [2] for registration of the UE in the IM CN subsystem.

If SC using ICS is enabled then prior to making use of ICS procedures, the SC UE shall follow the procedures specified in 3GPP TS 24.292 [4] for registration of the ICS UE in the IM CN subsystem.

The SC UE shall include the g.3gpp.accesstype media feature tag as described in clause B.3 of 3GPP TS 24.292 [4] in the Contact header field of the SIP REGISTER request.

If the SC UE supports the CS to PS SRVCC, the SC UE shall include the g.3gpp.cs2ps-srvcc media feature tag in the Contact header field of the SIP REGISTER request.

Upon receiving a SIP 2xx response to the REGISTER request and if the SIP 2xx response contains a Feature-Caps header field with the g.3gpp.atcf feature-capability indicator and with the g.3gpp.cs2ps-srvcc feature-capability indicator, the SC UE shall:

1) determine STI-rSR as the value of the g.3gpp.cs2ps-srvcc feature-capability indicator in the Feature-Caps header field containing both the g.3gpp.atcf feature-capability indicator and the g.3gpp.cs2ps-srvcc feature-capability indicator; and

2) store the determined STI-rSR.

If the SC UE supports the PS to PS access transfer and the PS to PS STI URI is configured in the SC UE, the SC UE shall include the g.3gpp.pstops-sti media feature tag in the Contact header field of the SIP REGISTER request.

6.2.3 SC UE receiving the ATGW information for CS to PS SRVCC

If the SC UE supports the CS to PS SRVCC, upon receiving a SIP MESSAGE request with ATGW information for CS to PS SRVCC, if the SIP MESSAGE request is acceptable for the UE, in addition to sending a SIP 2xx response to the SIP MESSAGE request, the SC UE shall;

1) determine the ATGW information for CS to PS SRVCC as the application/SDP MIME body of the SIP MESSAGE request;

2) store the determined ATGW information for CS to PS SRVCC;

3) generate the UE information for CS to PS SRVCC as an SDP answer to the determined ATGW information for CS to PS SRVCC according to IETF RFC 3264 [58] and 3GPP TS 24.229 [2];

4) store the generated UE information for CS to PS SRVCC; and

5) send a SIP MESSAGE request according to 3GPP TS 24.229 [2]. The SC UE shall populate the SIP MESSAGE request with:

   A) Request-URI containing the determined STI-rSR;

   B) Content-Disposition header field with value "render"; and

   C) application/sdp MIME body containing the generated UE information for CS to PS SRVCC.

6.3 SCC AS

6.3.1 General

The SCC AS can obtain registration state information that it needs to implement SCC specific requirements from:

a) any received third-party SIP REGISTER request (e.g. including information contained in the body of the third-party SIP REGISTER request) as specified in 3GPP TS 24.229 [2];

b) any received reg event package as specified in 3GPP TS 24.229 [2]; or

c) the Sh interface as specified in 3GPP TS 29.328 [6] and 3GPP TS 29.329 [7].

NOTE 1: Obtaining registration state information from HSS using Sh interface does not allow the SCC AS to know the capabilities supported by the user registered UE(s), including the used IP-CAN(s), other than that is specified in 3GPP TS 29.328 [6], e.g. the UE PS to CS SRVCC capability and 3GPP access networks' information related to T-ADS.

When the SCC AS obtains the registration state information including an Correlation MSISDN using one of the above procedures, the SCC AS shall determine if the registration state information is associated with ongoing CS call by matching the Correlation MSISDN against the:

a) tel URI in the P-Asserted-Identity header field or associated with the received IMRN when the SIP INVITE request was due to PS to CS STN, where the SIP INVITE request was stored according to subclause 7.3.1; or

b) tel URI in the Request-URI when the SIP INVITE request was due to processing unregistered filter criteria, where the SIP INVITE request was stored according to subclause 7.3.1.

If the registration state information is associated with an ongoing call the contents of the registration state information shall be bound to the ongoing CS call session identifier.

NOTE 2: The SCC AS has no responsibility for supervising the registration state of the SCC UE, nor taking any actions resulting from deregistration. If deregistration of the SC UE occurs, then the other functional entities in IMS, e.g. the S-CSCF, will initiate the release of SIP dialogs that are supported in the SCC AS.

6.3.2 Triggers for the SCC AS providing information to ATCF

This subclause applies for a contact address (or a registration flow, if multiple registration mechanism is used) in the registration state information obtained by SCC AS:

1) which is registered by the UE:

   A) in E-UTRAN, UTRAN and GERAN access networks; and

NOTE: The access network where the UE performed registration can be found in the P-Access-Network-Info header field of the SIP REGISTER request.

   B) for a private user identity associated with a C-MSISDN; and

2) where the SIP REGISTER request contained a Feature-Caps header field containing the g.3gpp.atcf feature-capability indicator.

The SCC AS shall identify the ATCF URI for terminating requests of the related ATCF as the URI in the g.3gpp.atcf-path feature-capability indicator included in a Feature-Caps header field of the SIP REGISTER request that created the binding.

The SCC AS shall store the feature-capability indicators indicated in the Feature-Caps header field containing the g.3gpp.atcf feature-capability indicator until the binding is removed.

The SCC AS shall determine that PS to CS SRVCC is usable for the UE if the UE PS to CS SRVCC Capability (see 3GPP TS 29.328 [6]) of the UE has value UE-SRVCC-CAPABILITY-SUPPORTED and if the private user identity of the UE has associated STN-SR (see 3GPP TS 29.328 [6]).

If SCC AS supports CS to PS SRVCC, the SCC AS shall also determine whether the CS to PS SRVCC is usable for the private user identity of the UE as described in subclause 6.3.4.

When the SCC AS becomes aware of a new contact address (or new registration flow, if multiple registration mechanism is used) that fulfils the above criteria and:

- PS to CS SRVCC is usable for the UE; or
- the SCC AS supports CS to PS SRVCC and CS to PS SRVCC is usable for the UE;

the SCC AS shall perform actions as described in subclause 6.3.3 with the related ATCF.

When the SCC AS becomes aware that, for a UE which registered the contact address (or registered the registration flow, if multiple registration mechanism is used) that fulfils the above criteria that:

1) PS to CS SRVCC was usable and PS to CS SRVCC is not usable now;

2) PS to CS SRVCC was not usable and PS to CS SRVCC is usable now; or

3) the SCC AS supports CS to PS SRVCC and:

   A) CS to PS SRVCC was usable and CS to PS SRVCC is not usable now; or

   B) CS to PS SRVCC was not usable and CS to PS SRVCC is usable now;

then the SCC AS shall provide the PS to CS SRVCC related information to the related ATCF as described in subclause 6.3.3.

6.3.3 SCC AS providing the PS to CS SRVCC related information to the ATCF

In order to provide the PS to CS SRVCC related information to the ATCF, the SCC AS shall perform the role of an AS acting as originating UA according to 3GPP TS 24.229 [2] subclause 5.7.3 using the procedure for sending an initial request on behalf of a PSI and shall send a SIP MESSAGE request populated as follows:

1) the Request-URI set to the ATCF management URI of the ATCF associated with the registration path (or registration flow, if multiple registration mechanism is used);

NOTE 1: The ATCF management URI of the ATCF is the URI contained in the g.3gpp.atcf-mgmt-uri feature-capability indicator that is included in a Feature-Caps header field of the SIP REGISTER request which the S-CSCF received from the UE using the method to obtain registration state information described in step a) of subclause 6.3.1.

2) the P-Asserted-Identity header field containing the identity of the SCC AS;

3) the application/vnd.3gpp.SRVCC-info+xml MIME body as defined in clause D.3;

NOTE 2: The ATCF URI for terminating calls of the registration path (or registration flow, if multiple registration mechanism is used) is contained in the g.3gpp.atcf-path feature-capability indicator that is included in a Feature-Caps header field of the SIP REGISTER request which the S-CSCF received from the UE using the method to obtain registration state information described in step a) of subclause 6.3.1.

4) the P-Charging-Vector header field containing a type 1 "orig-ioi" header field parameter. The SCC AS shall set the type 1 "orig-ioi" header field parameter to a value that identifies the sending network of the request. The SCC AS shall not include the type 1 "term-ioi" header field parameter; and

5) if the SCC AS supports indicating the traffic leg associated with a URI as specified in IETF RFC 7549 [83], the UE is roaming and if required by local policy, the SCC AS shall:

   a) append the "iotl" SIP URI parameter to the URI in the Request-URI with a value set to "homeB-visitedB";

   b) if required by local policy, the SCC AS may append an "iotl" SIP URI parameter with a value set to "visitedA-homeA" to:

      - the ATU-STI URI in the the application/vnd.3gpp.SRVCC-info+xml MIME body defined in clause D.3; and

      - the additional transferred session SCC AS URI for PS to CS SRVCC in the Refer-To URI of SIP REFER requests.

NOTE 3: The SCC AS can use the P-Visited-Network-Identity header field in the 3rd party SIP REGISTER request received when the UE registered in PS to determine if the UE is roaming or not.

6.3.4 Triggers for the SCC AS providing information to MSC server

If the SCC AS supports the CS to PS SRVCC, this subclause applies for a contact address in the registration state information obtained by SCC AS:

1) which is registered for a private user identity associated with an MSC server enhanced for ICS according to 3GPP TS 23.003 [12], subclause 20.3.3;

2) which is registered for a private user identity associated with a C-MSISDN; and

3) where the g.3gpp.cs2ps-srvcc media feature tag and the g.3gpp.path media feature tag are associated with the contact address.

The SCC AS shall determine that the CS to PS SRVCC is usable if:

1) a private user identity of a UE (i.e. other than those according to 3GPP TS 23.003 [12], subclause 20.3.3) associated with the same C-MSISDN as the private user identity belonging to the MSC server exists;

2) a binding of a contact address exists for the private user identity of the UE:

   A) such that the g.3gpp.cs2ps-srvcc media feature tag is associated with the contact address of the UE; and

   B) such that SIP REGISTER request which registered the binding contained a Feature-Caps header field with the g.3gpp.atcf feature-capability indicator and with g.3gpp.cs2ps-srvcc media feature tag;

3) the CS to PS SRVCC capability indication is indicated for the private user identity of the UE; and

4) the private user identity of the UE has the CS to PS SRVCC allowed indication in the subscription data.

When the SCC AS becomes aware of a new contact address that fulfils the above criteria and the CS to PS SRVCC is usable, the SCC AS shall perform actions as described in subclause 6.3.5 for the contact address.

When the SCC AS becomes aware that, for a contact address:

1) the CS to PS SRVCC was usable and the CS to PS SRVCC is not usable now; or

2) the CS to PS SRVCC was not usable and the CS to PS SRVCC is usable now;

then the SCC AS shall perform actions as described in subclause 6.3.5 for the contact address.

6.3.5 SCC AS providing the CS to PS SRVCC information to the MSC server

If the SCC AS supports the CS to PS SRVCC, in order to provide the CS to PS SRVCC information to a contact address registered by the MSC server, the SCC AS shall perform the role of an AS acting as originating UA according to 3GPP TS 24.229 [2] subclause 5.7.3 using the procedure for sending an initial request on behalf of a PSI and shall send a SIP MESSAGE request populated as follows:

1) the Request-URI set to an IMS public user identity registered at the contact address;

2) the P-Asserted-Identity header field containing the identity of the SCC AS;

3) the Accept-Contact header field with the g.3gpp.path media feature tag with value of the g.3gpp.path media feature tag associated with the contact address and with "explicit" and "require";

4) the application/vnd.3gpp.srvcc-ext+xml MIME body; and

NOTE: The MSC URI for terminating calls of the contact address is contained in the g.3gpp.path media feature tag that is included in a Contact header field of the SIP REGISTER request which the S-CSCF received from the UE using the method to obtain registration state information described in step a) of subclause 6.3.1.

5) the P-Charging-Vector header field containing a type 1 "orig-ioi" header field parameter. The SCC AS shall set the type 1 "orig-ioi" header field parameter to a value that identifies the sending network of the request. The SCC AS shall not include the type 1 "term-ioi" header field parameter.

6.4 MSC server

6.4.1 Distinction of requests

The MSC server needs to distinguish the following initial SIP requests:

1) SIP MESSAGE requests with the Accept-Contact header field containing the g.3gpp.path media feature tag and with the application/vnd.3gpp.srvcc-ext+xml MIME body. In the procedures below, such requests are known as "SIP MESSAGE requests with MSC information for CS to PS SRVCC".

6.4.2 General

If the MSC server:

- provides the role of an MSC server enhanced for ICS; and

- determines that the served user is an ICS user;

then in addition to the procedures specified in 3GPP TS 29.292 [18] and 3GPP TS 24.292 [4] the MSC server shall:

1) if the MSC server supports the MSC server assisted mid-call feature, include the g.3gpp.mid-call media feature tag (as described in annex C) in the Contact header field of the SIP REGISTER request; and

2) if the MSC server supports the CS to PS SRVCC:

   A) include the g.3gpp.cs2ps-srvcc media feature tag in the Contact header field of the SIP REGISTER request; and

   B) include the g.3gpp.path media feature tag in the Contact header field of the SIP REGISTER request with value uniquely identifying the registration path.

6.4.3 MSC server receiving the MSC information for CS to PS SRVCC

If the MSC server supports the CS to PS SRVCC, upon receiving SIP MESSAGE requests with MSC information for CS to PS SRVCC, the MSC server shall:

1) if the URI in the P-Asserted-Identity header field of the SIP MESSAGE request does not identify an SCC AS authorised to provide the CS to PS SRVCC information, reject the request with SIP 403 (Forbidden) response and do not continue with the remaining steps;

NOTE: in this version of specification, the URIs of SCC ASs authorised to provide PS to CS SRVCC information need to be specified in the roaming agreement.

2) bind the CS to PS SRVCC information received in the application/vnd.3gpp.srvcc-ext+xml MIME body of the SIP MESSAGE request to the contact address ; and

3) send a SIP 200 (OK) response to the MESSAGE request according to 3GPP TS 24.229 [2] and include in the P-Charging-Vector header field the "icid-value" header field parameter set to the value received in the request, the "orig-ioi" header field parameter, if received in the request and a type 1 "term-ioi" header field parameter that identifies the sending network.

6.5 Access Transfer Control Function (ATCF)

6.5.1 Distinction of requests

The ATCF needs to distinguish the following initial SIP requests:

1) SIP REGISTER requests with the ATCF URI for originating requests in the topmost Route header field. In the procedures below, such requests are known as "SIP REGISTER request originated by a UE".

2) SIP MESSAGE requests with the ATCF management URI in the Request-URI and:

   A. not containing any Route header field; or

   B. containing a URI in the topmost Route header field other than the ATCF URI for originating requests and other than the ATCF URI for terminating requests.

   In the procedures below, such requests are known as "SIP MESSAGE requests with the PS to CS SRVCC related information".

3) SIP MESSAGE requests with the STI-rSR allocated by ATCF in the Request-URI and with the ATCF URI for originating requests in the topmost Route header field. In the procedures below, such requests are known as "SIP MESSAGE requests with UE information for CS to PS SRVCC".

6.5.2 Registration related procedures in the ATCF

Upon receiving a SIP REGISTER request originated by a UE, the ATCF shall:

1. if ATCF decides to include itself for access transfer of sessions according to operator policy:

NOTE 1: An example of the operator policy is that the ATCF is included in the signalling path only when the UE registers over the E-UTRAN, UTRAN or GERAN access networks.

   A. generate a unique ATCF URI for terminating requests such that the registration path (or registration flow, if multiple registration mechanism is used) can be determined for terminating requests;

NOTE 1A: One possible construction method is to set the user portion of the ATCF URI for terminating requests to the URI of the most bottom Path header field of the SIP REGISTER request.

   B. insert a Path header field with the generated ATCF URI for terminating requests;

   C. insert a Feature-Caps header field as described in RFC 6809 [60] with:

      a. the g.3gpp.atcf feature-capability indicator containing the STN-SR allocated to ATCF included as described in IETF RFC 6809 [60];

      b. the g.3gpp.atcf-mgmt-uri feature-capability indicator containing the ATCF management URI included as described in IETF RFC 6809 [60];

      c. the g.3gpp.atcf-path feature-capability indicator with value containing the generated ATCF URI for terminating requests as described in IETF RFC 6809 [60];

d. if the ATCF is aware that all MSC servers, which can be involved in the SRVCC procedures and which are in the same network as the ATCF, support the MSC server assisted mid-call feature:

- the g.3gpp.mid-call feature-capability indicator;

e. if the ATCF is aware that all MSC servers, which can be involved in the SRVCC procedures and which are in the same network as the ATCF, support the PS to CS SRVCC for calls in alerting phase:

- the g.3gpp.srvcc-alerting feature-capability indicator; and

- if the ATCF is aware that all MSC servers, which can be involved in the SRVCC procedures and which are in the same network as the ATCF, support the PS to CS SRVCC for originating calls in pre-alerting phase:

  i. the g.3gpp.ps2cs-srvcc-orig-pre-alerting feature-capability indicator as described in annex C; and

- if the ATCF is aware that all MSC servers, which can be involved in the SRVCC procedures and which are in the same network as the ATCF, support the PS to CS SRVCC for terminating calls in pre-alerting phase:

  i. the g.3gpp.ps2cs-srvcc-term-pre-alerting feature-capability indicator as described in annex C; and

f. if the Contact header field of the SIP REGISTER request contains the g.3gpp.cs2ps-srvcc media feature tag and if the ATCF supports the CS to PS SRVCC:

- the g.3gpp.cs2ps-srvcc feature-capability indicator containing the STI-rSR allocated by ATCF;

NOTE 2: Since the ATCF cannot be aware of which MSC server the SC UE can potentially be transferred to and the PS to CS SRVCC access transfer for a call in alerting phase is optional, all MSC servers in the network where the SC UE is attached needs to support the PS to CS SRVCC access transfer for a call in alerting phase before the ATCF can indicate support.

NOTE 3: Since the ATCF cannot be aware of which MSC server the SC UE can potentially be transferred to and the PS to CS SRVCC access transfer for a call in pre-alerting phase is optional, all MSC servers in the network where the SC UE is attached needs to support the PS to CS SRVCC access transfer for a call in pre-alerting phase before the ATCF can indicate support.

NOTE 4: Since the ATCF cannot be aware of which MSC server the SC UE can potentially be transferred to and the MSC server assisted mid-call feature is optional, all MSC servers in the network where the SC UE is attached needs to support the MSC server assisted mid-call feature before the ATCF can indicate support.2. if the ATCF is located in the visited network and local policy requires the application of IBCF capabilities in the visited network towards the home network select an exit point of the visited network and forward the request to that entry point;

NOTE 5: The list of the exit points can be either obtained as specified in RFC 3263 [72] or provisioned in the ATCF.

3. if the ATCF is located in the visited network and local policy does not require the application of IBCF capabilities in the visited network towards the home network select an entry point of the home network and forward the request to that entry point;

NOTE 6: The list of the entry points can be either obtained as specified in RFC 3263 [72] or provisioned in the ATCF. The entry point can be an IBCF or an I-CSCF.

4. if the ATCF is located in the home network select an I-CSCF of the home network and forward the request to that I-CSCF; and

NOTE 7: The list of the I-CSCFs can be either obtained as specified in RFC 3263 [72] or provisioned in the ATCF.

5. if the ATCF fails to forward the SIP REGISTER request to any entry point, the ATCF shall send back a SIP 504 (Server Time-Out) response, in accordance with the procedures in RFC 3261 [19].

Upon receiving a SIP 2xx response to the SIP REGISTER request originated by a served UE and if ATCF decided to include itself for access transfer of sessions according to operator policy, the ATCF shall:

1) update the S-CSCF Service-Route URI bound to the registration path (see subclause 6A.3.1) identified by the ATCF Path URI;

NOTE 8: The ATCF Path URI is the URI which the ATCF inserted in the Path header field of to the SIP REGISTER request.

NOTE 9: The S-CSCF Service-Route URI is the URI in the most bottom Service-Route header field of the SIP 2xx response to the SIP REGISTER request.

2) if the Contact header field of the SIP REGISTER request contains the g.3gpp.cs2ps-srvcc media feature tag and if the ATCF supports the CS to PS SRVCC:

   A) for the registration path, which has the ATCF Path URI matching the URI which the ATCF inserted in the Path header field of to the SIP REGISTER request:

      a) set the route set towards the SC UE bound to the registration path (see subclause 6A.3.1) to the Path header fields in the received SIP 2xx response preceding the ATCF Path URI; and

      b) set the contact address of the SC UE bound to the registration path (see subclause 6A.3.1) to the Contact header field of the SIP REGISTER request; and

3) insert a Feature-Caps header field with:

   A) the g.3gpp.atcf feature-capability indicator containing the STN-SR allocated to ATCF included as described in IETF RFC 6809 [60]; and

   B) if the Contact header field of the SIP REGISTER request contains the g.3gpp.cs2ps-srvcc media feature tag and if the ATCF supports the CS to PS SRVCC:

      a) the g.3gpp.cs2ps-srvcc feature-capability indicator containing the STI-rSR allocated by ATCF.

6.5.3 ATCF receiving the SRVCC-related information

Upon receiving SIP MESSAGE request with the SRVCC-related information, the ATCF shall:

1) if the URI in the P-Asserted-Identity header field of the SIP MESSAGE request does not identify an SCC AS authorised to provide the SRVCC-related information, reject the request with SIP 403 (Forbidden) response and do not continue with the remaining steps;

NOTE: in this version of specification, the URIs of SCC ASs authorised to provide SRVCC-related information need to be specified in the roaming agreement.

2) update the SRVCC-related information bound to the registration path(s) (see subclause 6A.3.1) with information in the application/vnd.3gpp.SRVCC-info+xml MIME body of the SIP MESSAGE request;

3) determine session(s) established using the registration path(s) (see subclause 6A.3.1) whose SRVCC-related information were updated by the SRVCC-related information received in the SIP MESSAGE request and associate those session(s) with the SRVCC-related information bound to the registration path(s);

4) for each registration path in the SRVCC-related information received in the SIP MESSAGE request:

   A) if:

      a) the ATCF indicated the support of the CS to PS SRVCC when handling the SIP REGISTER request establishing the registration path;

      b) the SRVCC-related information for the registration path contains the ATU-STI for CS to PS SRVCC; and

      c) the ATCF does not have the UE information for CS to PS SRVCC bound to the registration path;

      send the ATGW information for CS to PS SRVCC to the SC UE within the registration path using procedure described in subclause 6.5.4; and

5) send a SIP 200 (OK) response to the MESSAGE request according to 3GPP TS 24.229 [2] and include in the P-Charging-Vector header field the "icid-value" header field parameter set to the value received in the request, the

"orig-ioi" header field parameter, if received in the request and a type 1 "term-ioi" header field parameter that identifies the sending network.

6.5.4 ATCF sending the ATGW information for CS to PS SRVCC

If the ATCF supports the CS to PS SRVCC, in order to send the ATGW information for CS to PS SRVCC to the SC UE within a registration path, the ATCF shall:

1) generate the ATGW information for CS to PS SRVCC. When generating the SDP, the ATCF shall:

   A) set c-line to the unspecified address (0.0.0.0), if IPv4, or to a domain name within the ".invalid" DNS top-level domain as described in IETF RFC 6157 [74], if IPv6; and

   B) set port number of the media line to 9;

2) set the ATGW information for CS to PS SRVCC bound to the registration path (see subclause 6A.3.1) to the generated ATGW information for CS to PS SRVCC; and

3) send SIP MESSAGE request according to 3GPP TS 24.229 [2]. The ATCF shall populate the SIP MESSAGE request with:

   A) Request-URI containing the contact address of the SC UE bound to the registration path (see subclause 6A.3.1);

   B) Route header fields containing the route set towards the SC UE of the registration path (see subclause 6A.3.1);

   C) P-Asserted-Identity header field containing the STI-rSR allocated by ATCF;

   D) Content-Disposition header field with value "render"; and

   E) application/sdp MIME body containing the generated ATGW information for CS to PS SRVCC.

6.5.5 ATCF receiving the UE information for CS to PS SRVCC

If the ATCF supports the CS to PS SRVCC, upon receiving SIP MESSAGE request with UE information for CS to PS SRVCC and if the SIP MESSAGE request is acceptable for the ATCF, in addition to sending a SIP 2xx response to the SIP MESSAGE request, the ATCF shall:

1) determine the related registration path, which is a registration path with the ATCF Path URI matching the URI in the top Route header field of the SIP MESSAGE request; and

2) set the UE information for CS to PS SRVCC bound to the determined related registration path (see subclause 6A.3.1) to the application/sdp MIME body of the SIP MESSAGE request.

6A Roles for General Capabilities

6A.1 Introduction

This clause describes the general roles for each functional entity as specified.

6A.2 UE roles

6A.2.1 Operator policy enforcement

The SC UE may receive the operator policy via OMA Device Management, see 3GPP TS 24.216 [5]. When the SC UE receives the operator policy, for each non-emergency session to be transferred, it shall take the operator policy into account when deciding to perform the following:

- selecting the access for initiating the transfer;

- determining whether to transfer full or partial media during PS-PS transfer; or

- determining whether to add or remove media during the PS-PS transfer.

If the SC UE is configured with the operator policy (e.g. via OMA Device Management as described in 3GPP TS 24.216 [5]) then, for each media or group of media contained in the MediaorGroup node, the SC UE shall:

1) restrict originating non-emergency sessions and session transfer of non-emergency sessions towards the access networks contained in the RestrictedAccessNetworkType node;

2) consider the list of access networks contained in the PreferredAccessNetworks node in the order of priority from the access networks such that, when available, the highest priority access network can be used for originating non-emergency sessions and session transfer of non-emergency sessions;

3) if a new access network gets available- transfer media components to a higher priority target network than the current access network based on the value contained in the SC_media_transfer node value. If the SC_media_transfer node value is:

   - "shall" the UE shall start a session transfer of non-emergency sessions according to the home operator' s list of preferred access networks contained in the PreferredAccessNetworks node;

   - "should" the UE is recommended to start session transfer of non-emergency sessions according to the home operator's list of preferred access networks contained in the PreferredAccessNetworks node. The UE can evaluate if session transfer of non-emergency sessions is possible and desirable after having taken into account the Local Operating Environment Information; and

   - "may" the UE can decide whether or not to start session transfer of non-emergency sessions in accordance with user preferences if configured in the UE. The UE can evaluate if session transfer of non-emergency sessions is possible and desirable after having taken into account the Local Operating Environment Information. If user preferences are not configured, the UE can evaluate the home operator's list of preferred access networks contained in the PreferredAccessNetworks node; and

4) decide whether to keep or drop non transferable media components in the case of partial session transfer of non-emergency sessions based on the SC_non_transferrable_media node value.

6A.2.2 Signalling elements

6A.2.2.1 Common SIP message set up procedures

This subclause describes the common procedures for setting up SIP messages sent by SC UE.

6A.2.2.2 SIP INVITE request

When sending a SIP INVITE request regardless of Request-URI, the SC UE shall include the following media feature tags in the Contact header field of the SIP INVITE request according to RFC 3840 [53]:

1) if the SC UE supports the MSC server assisted mid-call feature, include the g.3gpp.mid-call media feature tag as described in annex C;

2) if the SC UE supports the PS to CS SRVCC for calls in alerting phase and is sending an initial SIP INVITE request:

   A) include the g.3gpp.srvcc-alerting media feature tag as described in annex C; and

   B) if the SC UE supports the PS to CS SRVCC for originating calls in pre-alerting phase, include the g.3gpp.ps2cs-srvcc-orig-pre-alerting media feature tag as described in annex C;

3) if the SC UE supports the PS to CS dual radio access transfer for calls in alerting phase and is sending an initial SIP INVITE request:

   A) include the g.3gpp.drvcc-alerting media feature tag as described in annex C; and

   B) if the SC UE supports the PS to CS dual radio access transfer for originating calls in pre-alerting phase, include the g.3gpp.ps2cs-drvcc-orig-pre-alerting media feature tag as described in annex C; and

4) if the SC UE supports the use of dynamic STN, include the g.3gpp.dynamic-stn media feature tag as described in annex C.

Inclusion of the media feature tags of item 2) and 3) in a reINVITE request is optional.

When sending a SIP INVITE request with the Request URI set to an emergency service URN:

1) if the SC UE supports PS to CS DRVCC for emergency session, the SC UE shall include the g.3gpp.dynamic-e-stn-drvcc media feature tag in the Contact header field as described in annex C; and

2) if the SC UE supports the MSC server assisted mid-call feature and supports the PS to CS SRVCC for emergency session in early dialog state with active speech media component when both the emergency session in early dialog state with active speech media component and a non-emergency call in confirmed dialog state with inactive speech media component exists, the SC UE shall include the g.3gpp.ps2cs-srvcc-mid-call-emergency media feature tag in the Contact header field as described in annex C.

6A.3 ATCF

6A.3.1 SRVCC information bound to the registration path

The ATCF shall keep track of existing registrations of the served UEs. Each registration path is identified by the ATCF Path URI.

The ATCF shall bind the following information to the registration path identified by the ATCF Path URI:

- the S-CSCF Service-Route URI;
- the ATU-STI for PS to CS SRVCC; and
- the C-MSISDN.

If the ATCF supports CS to PS SRVCC, the ATCF shall additionally bind the following information to the registration path identified by the ATCF Path URI:

- the ATU-STI for CS to PS SRVCC;
- the contact address of the SC UE;
- the route set towards the SC UE;
- the UE information for CS to PS SRVCC; and
- the ATGW information for CS to PS SRVCC.

When a registration of a served UE expires or is deregistered, the ATCF can remove any SRVCC-related information bound to the registration path.

The ATCF shall determine that a session is established for a specific registration path:

- if the S-CSCF Service-Route URI used during the registration matches the URI in the most bottom Route header field of the originating initial SIP INVITE request; or
- if the ATCF Path URI used during the registration matches the URI in the top Route header field of the terminating initial SIP INVITE request.

6A.4 SCC AS

6A.4.1 Common SIP message set up procedures

This subclause describes the common procedures for setting up SIP messages sent by SCC AS.

6A.4.2 SIP INVITE request

When sending SIP INVITE request towards the served user and if the session being established is anchored in SCC AS as described in subclause 4.2.2 then the SCC AS shall populate the SIP INVITE request with:

1) a Feature-Caps header field according to IETF RFC 6809 [60]:

A) including the g.3gpp.srvcc feature-capability indicator as described in annex C; and

B) including the g.3gpp.remote-leg-info feature-capability indicator as described in annex C;

2) an Accept header field according to IETF RFC 3261 [19] containing the MIME type application/vnd.3gpp.state-and-event-info+xml as specified in subclause D.2.3; and

3) a Recv-Info header field according to IETF RFC 6086 [54] containing the g.3gpp.state-and-event package name.

6A.4.3 SIP INVITE responses towards the SC UE

When sending SIP 1xx response or SIP 2xx response to the SIP INVITE request towards the served user, the SCC AS shall populate the SIP response with a Feature-Caps header field according to IETF RFC 6809 [60] containing:

1) if the session being established is anchored in SCC AS as described in subclause 4.2.2:

A) include the g.3gpp.srvcc feature-capability indicator as described in annex C;

B) if:

a) the SCC AS supports the PS to CS SRVCC with the MSC server assisted mid-call feature according to operator policy;

b) the g.3gpp.mid-call media feature tag as described in annex C is included in the Contact header field of the SIP INVITE request; and

c) the SCC AS is aware:

- by local policy; or

- by ATCF indicating support of the PS to CS SRVCC with the MSC server assisted mid-call feature;

NOTE 1: An ATCF can indicate support of the PS to CS SRVCC with the MSC server assisted mid-call feature by inclusion of the g.3gpp.mid-call feature-capability indicator in the Feature-Caps header field, with the g.3gpp.atcf feature-capability indicator, in the SIP REGISTER request that created the binding of the SC UE.

that all MSC Servers in the network, where the UE is registered, which can be involved in the PS to CS SRVCC procedures, support the PS to CS SRVCC with the MSC server assisted mid-call feature;

NOTE 2: SCC AS can identify the network, where the UE is registered, based on the P-Visited-Network-Id header field and the P-Access-Network-Info header field of the SIP REGISTER request.

include the g.3gpp.mid-call feature-capability indicator as described in annex C;

C) if:

a) the SCC AS supports the PS to CS SRVCC for calls in alerting phase according to operator policy;

b) the g.3gpp.srvcc-alerting feature tag as described in annex C is included in the Contact header field of the SIP INVITE request; and

c) the SCC AS is aware:

- by local policy; or

- by ATCF indicating support of the PS to CS SRVCC for calls in alerting phase;

NOTE 3: An ATCF can indicate support of the PS to CS SRVCC for calls in alerting phase by inclusion of the g.3gpp.srvcc-alerting feature-capability indicator in the Feature-Caps header field, with the g.3gpp.atcf feature-capability indicator, in the SIP REGISTER request that created the binding of the SC UE.

that all MSC Servers in the network, where the UE is registered, which can be involved in the PS to CS SRVCC procedures, support the PS to CS SRVCC for calls in alerting phase;

NOTE 4: SCC AS can identify the network, where the UE is registered, based on the P-Visited-Network-Id header field and the P-Access-Network-Info header field of the SIP REGISTER request.

include the g.3gpp.srvcc-alerting feature-capability indicator as described in annex C;

D) if:

a) the SCC AS supports the PS to CS SRVCC for originating calls in pre-alerting phase and the PS to CS SRVCC for calls in alerting phase according to operator policy;

b) the g.3gpp.ps2cs-srvcc-orig-pre-alerting media feature tag as described in annex C and the g.3gpp.srvcc-alerting media feature tag as described in annex C are included in the Contact header field of the SIP INVITE request due to originating filter criteria;

c) the SCC AS is aware:

- by local policy; or

- by ATCF indicating support of the PS to CS SRVCC for originating calls in pre-alerting phase;

NOTE 5: An ATCF can indicate support of the PS to CS SRVCC for originating calls in pre-alerting phase by inclusion of the g.3gpp.ps2cs-srvcc-orig-pre-alerting feature-capability indicator in the Feature-Caps header field, with the g.3gpp.atcf feature-capability indicator, in the SIP REGISTER request that created the binding of the SC UE.

that all MSC servers in the network where the UE is registered which can be involved in PS to CS SRVCC procedures support the PS to CS SRVCC for originating calls in pre-alerting phase and the PS to CS SRVCC for calls in alerting phase; and

NOTE 6: The SCC AS can identify the network where the UE is registered based on the P-Visited-Network-Id header field and the P-Access-Network-Info header field of the SIP REGISTER request.

d) SIP 180 (Ringing) response to the SIP INVITE request has not been received yet;

include the g.3gpp.ps2cs-srvcc-orig-pre-alerting feature-capability indicator as described in annex C; and

E) include the g.3gpp.remote-leg-info feature-capability indicator as described in annex C;

2) if the SCC AS supports the PS to CS dual radio access transfer for calls in alerting phase according to operator policy, and if the SIP INVITE request included the g.3gpp.drvcc-alerting media feature tag as described in annex C in the Contact header field, include the g.3gpp.drvcc-alerting feature-capability indicator as described in annex C;

3) if:

A) the SCC AS supports the PS to CS dual radio access transfer for originating calls in pre-alerting phase and the PS to CS dual radio access transfer for calls in alerting phase;

B) the g.3gpp.ps2cs-drvcc-orig-pre-alerting media feature tag as described in annex C and the g.3gpp.drvcc-alerting media feature tag as described in annex C are included in the Contact header field of the SIP INVITE request; and

C) SIP 180 (Ringing) response to the SIP INVITE request has not been received yet;

include the g.3gpp.ps2cs-drvcc-orig-pre-alerting feature-capability indicator as described in annex C; and

4) if the SCC AS supports the use of dynamic STN according to operator policy, and if the Contact header field of the SIP INVITE request includes the g.3gpp.dynamic-stn media feature tag as described in annex C, include the g.3gpp.dynamic-stn feature-capability indicator as described in annex C with the dynamic STN.

NOTE 7: Based on implementation the dynamic STN can either be the same or different per call.

Additionally, when sending SIP 1xx response or SIP 2xx response to the SIP INVITE request towards the served user, the SCC AS shall populate the SIP response with:

1) if the session being established is anchored in SCC AS as described in subclause 4.2.2:

A) if the SIP response is a SIP 2xx response, an Accept header field according to IETF RFC 3261 [19] containing the MIME type application/vnd.3gpp.state-and-event-info+xml as specified in subclause D.2.3; and

B) a Recv-Info header field according to IETF RFC 6086 [54] containing the g.3gpp.state-and-event package name.

6A.4.3A SIP INVITE responses towards the MSC server

When sending a SIP 1xx response or SIP 2xx response to a SIP INVITE request due to STN-SR or to a SIP INVITE request due to PS to CS STN and if a Contact header field was saved in subclause 7.3.2 or subclause 8.3.2, the SCC AS shall include the saved Contact header field of the remote UE.

When sending a SIP 2xx response to a SIP INVITE request due to STN-SR or to a SIP INVITE request due to PS to CS STN and if a P-Asserted-Identity header field was saved in subclause 7.3.2 or subclause 8.3.2, the SCC AS shall include the P-Asserted-Identity header field with the identity of the remote user saved in subclause 7.3.2 or subclause 8.3.2 along with the Privacy header field, if available.

When sending a SIP 2xx response to a SIP INVITE request transferring additional session and if a P-Asserted-Identity header field was saved in subclause 7.3.2 or subclause 8.3.2, the SCC AS shall include the P-Asserted-Identity header field with the identity of the remote user saved in subclause 7.3.2 or subclause 8.3.2 along with the Privacy header field, if available.

NOTE: There are situations when the P-Asserted-Identity header field with the public user identity of the remote user can not be saved during the establishement of the communication, e.g. if presentation of the remote user public identity is restricted or if the user does not subscribe to the OIP or TIP service. In those situations the P-Asserted-Identity header field with a public user identity will not be delivered to the MSC server in the SIP 2xx response to the SIP INVITE due to STN-SR, the SIP INVITE request due to PS to CS STN or the SIP INVITE request transferring additional session and can limit the supplementary services that the MSC server can use after SRVCC access transfer is completed.

When sending a SIP 1xx response or SIP 2xx response to a SIP INVITE request due to STN-SR, a SIP INVITE request due to ATU-STI, a SIP INVITE request transferring additional session or a SIP INVITE request due to PS to CS STN, the SCC AS shall include:

1) the g.3gpp.remote-leg-info feature-capability indicator as described in annex C in a Feature-Caps header field according to IETF RFC 6809 [60];

2) if the SIP response is a SIP 2xx response, the Accept header field according to IETF RFC 3261 [19]containing the application/vnd.3gpp.state-and-event-info+xml MIME type; and

3) the Recv-Info header field according to IETF RFC 6086 [54] containing the g.3gpp.state-and-event info package name.

6A.4.4 Handling of OMR specific attributes

When an SDP offer containing OMR specific attributes specified in subclause 7.5.3 of 3GPP TS 24.229 [2] is received from either the source access leg or the target access leg, the SCC AS supporting OMR shall perform the actions specified in subclause 7.2.2 of 3GPP TS 29.079 [77].

When the SCC AS supporting OMR sends an SDP offer towards the remote party and if

- the SDP offer consists of several media lines merged from a source access leg and a target access leg; and

- any of the media lines contains OMR attributes;

then the SCC AS shall recalculate the checksums as specified in subclause 5.6.3 3GPP TS 29.079 [77].

If the SCC AS has not changed the content of a m-line and associated attributes, an SCC AS supporting OMR shall only calculate the session level checksum and replace the new value in each occurrences of the "a=omr-s-cksum" attribute.

The SCC AS supporting OMR shall forward the OMR specific attributes received in the SDP answer.

NOTE: When the SCC AS does not support OMR an optimal media path created before the transfer will not be established again.

6A.4.5 Target refresh request for a dialog and associated responses

The SCC AS shall include into the Feature-Caps header field of any target refresh request and, in each SIP 1xx response or SIP 2xx response to target refresh request sent to the SC UE:

A) the g.3gpp.srvcc feature-capability indicator if the session being established is anchored in the SCC AS as described in subclause 4.2.2 and if the SCC AS inserted the g.3gpp.srvcc feature-capability indicator into the Feature-Caps header field of:

   1) the SIP INVITE request in accordance with subclause 6A.4.2; or

   2) the SIP 1xx response or SIP 2xx response to the SIP INVITE request in accordance with subclause 6A.4.3;

B) the g.3gpp.mid-call feature-capability indicator if the SCC AS inserted the g.3gpp.mid-call feature-capability indicator into the Feature-Caps header field of:

   1) the SIP 2xx response to the SIP INVITE request due to originating filter criteria in accordance with subclause 7.3.2;

   2) the SIP INVITE request due to terminating filter criteria if the SCC AS applies the MSC Server assisted mid-call feature in accordance with subclause 8.3.2;

   3) the SIP 2xx response to the SIP INVITE request due to PS to CS STN if the SCC AS applies the MSC Server assisted mid-call feature in accordance with subclause 9.3.2A;

   4) the SIP 2xx response to the SIP INVITE request due to static STI if the SCC AS applies the MSC Server assisted mid-call feature in accordance with subclause 9.3.4; or

   5) the SIP 2xx response to the SIP INVITE request due to STI if the SCC AS applies the MSC Server assisted mid-call feature in accordance with subclause 10.3.3;

C) the g.3gpp.srvcc-alerting feature-capability indicator if the SCC AS inserted the g.3gpp.srvcc-alerting feature-capability indicator into the Feature-Caps header field of:

   1) any SIP 1xx response or SIP 2xx response to the SIP INVITE request due to originating filter criteria if the SCC AS applies PS to CS SRVCC for calls in alerting phase in accordance with subclause 7.3.2; or

   2) the SIP INVITE request due to terminating filter criteria if the SCC AS applies PS to CS SRVCC for calls in alerting phase in accordance with subclause 8.3.2; and

D) the g.3gpp.ps2cs-srvcc-orig-pre-alerting feature-capability indicator if the SCC AS inserted the g.3gpp.ps2cs-srvcc-orig-pre-alerting indicator into the Feature-Caps header field of any SIP 1xx response or SIP 2xx response to the SIP INVITE request due to originating filter criteria if the SCC AS applies PS to CS SRVCC for calls in alerting phase in accordance with subclause 7.3.2.

If a feature-capability indicator was indicated in a Feature-Caps header field of the initial SIP INVITE request sent to the MSC server or of the SIP 1xx response or SIP 2xx response to the initial SIP INVITE request sent to the MSC server, then the SCC AS shall include the feature-capability indicator into the Feature-Caps header field of any target refresh request and into the Feature-Caps header field of each SIP 1xx response or SIP 2xx response to target refresh request sent to the MSC server.

6A.4.6 Rejecting malicious SIP REFER requests from remote UE

If the SCC AS supports the PS to CS SRVCC of calls in alerting phase, then upon receiving a SIP REFER request:

1. sent inside a SIP dialog on the remote leg;
2. with the Refer-Sub header field containing "false" value;
3. with the Supported header field containing "norefersub" value;
4. with the Refer-To header field containing a SIP URI with the Target-Dialog URI header field; and
5. containing a MIME body of application/vnd.3gpp.state-and-event-info+xml MIME type specified in the subclause D.2.4;

the SCC AS shall reject the SIP REFER request with a SIP 403 (Forbidden) response as specified in 3GPP TS 24.229 [2].

If the SCC AS supports the MSC server assisted mid-call feature, then upon receiving a SIP REFER request:

1. sent inside a SIP dialog on the remote leg;
2. with the Refer-Sub header field containing "false" value;
3. with the Supported header field containing "norefersub" value;
4. with the Refer-To header field containing a SIP URI with the Target-Dialog URI header field; and
5. containing a MIME body of application/vnd.3gpp.mid-call+xml MIME type specified in the subclause D.1.3;

the SCC AS shall reject the SIP REFER request with a SIP 403 (Forbidden) response as specified in 3GPP TS 24.229 [2].

6A.4.7 Protecting from malicious SIP INFO requests with remote leg information from remote UE

The SCC AS shall not include the Accept header field containing the application/vnd.3gpp.state-and-event-info+xml MIME type and shall not include the Recv-Info header field containing the g.3gpp.state-and-event info package name in the SIP INVITE request, SIP re-INVITE request and SIP UDPATE request and related responses sent towards the remote UE.

6A.4.8 Precondition and access transfer

If according to operator policy, the SCC AS:

1) shall support UA role procedures defined in IETF RFC 3262 [86], IETF RFC 3311 [87], IETF RFC 3312 [88] and IETF RFC 4032 [89]; and

2) if the remote leg is a precondition enabled dialog and if the SIP INVITE request on target access leg is not a precondition enabled initial INVITE request, shall interwork precondition related signalling at the remote leg with the target access leg established by SIP response to SIP INVITE request on target access leg. Details of interworking are out of scope of this specification.

6A.5 SDP media description conflict between target and remote access leg

When the SCC AS, the EATF or the ATCF receives an SDP offer on the target access leg, the SDP media descriptions on the target access leg and the remote access leg, can be in conflict. The way how the SCC AS, EATF and ATCF resolve the conflict is implementation dependent.

NOTE 1: Examples of conflicts are when, for a given media type, different IP versions are used on each access leg, or when the same payload type number has been assigned to different codecs on each access leg.

NOTE 2: An example on how to solve a conflict can be that transcoding functionality is enabled by inserting an MRF (in case of SCC AS or EATF) or an ATGW (in case of ATCF). Another example is that a SIP 488 (Not Acceptable Here) response is sent with the correct SDP media description.

When the MSC server receives a SIP 488 (Not Acceptable Here) response to an initial SIP INVITE request and an SDP body is present in the response:

1) if the MSC server supports one or more RTP payload types indicated in the received SDP body, the MSC server should re-initiate the initial SIP INVITE request with an SDP offer containing the RTP payload types (each comprising of an RTP payload type number indicated in a sub-field of an <fmt> portion of an "m=" line and, if included, an "a=rtpmap" attribute and an "a=frmtp" attribute for the RTP payload type number):

   a) copied from the received SDP body (with the RTP payload type number indicated in the received SDP body); and

   b) supported by the MSC server; and

2) if the MSC server does not support any RTP payload type of the received SDP body, the MSC server should re-initiate the initial SIP INVITE request with an SDP offer containing one or more RTP payload types supported by MSC server, associated with RTP payload type number(s) not indicated in the received SDP body.

6A.6 Indicating traffic leg

6A.6.1 The SCC AS server procedure for indicating traffic leg

If the SCC AS supports indicating the traffic leg associated with a URI as specified in IETF RFC 7549 [83] and the UE is roaming, the SCC AS may append the "iotl" SIP URI parameter with the value "visitedA-homeA" to the additional transferred session SCC AS URI for PS to CS SRVCC in the Refer-To URI of SIP REFER requests.

NOTE: The SCC AS can use the P-Visited-Network-Identity header field in the 3rd party SIP REGISTER request received when the UE registered in PS to determine if the UE is roaming or not.

6A.6.2 The MSC server procedure for indicating traffic leg

If the MSC supports indicating the traffic leg associated with a URI as specified in IETF RFC 7549 [83]:

1) the UE is roaming; and

2) the STN-SR does not identify an ATCF in the visited network;

then the MSC server shall, if required by local policy:

1) convert the STN-SR in the Request-URI to the form of a SIP URI with user=phone; and

2) append an "iotl" SIP URI parameter with a value set to "visitedA-homeA" in the Request-URI of the SIP INVITE request due to STN-SR and, if vSRVCC is supported and applied, in the SIP OPTIONS request.

6A.6.3 The ATCF server procedure for indicating traffic leg

If the ATCF supports indicating the traffic leg associated with a URI as specified in IETF RFC 7549 [83]:

1) the UE is roaming; and

2) the ATCF is not in the home network;

then the ATCF may, if required by local policy and if ATCF support the PS to CS SRVCC access transfer, append an "iotl" SIP URI parameter with a value set to "homeB-visitedB" to the ATCF management URI in the g.3gpp.atcf-mgmt-uri feature-capability indicator of the SIP REGISTER request.

6A.7 MSC server

6A.7.1 Precondition and access transfer

Unless local configuration indicates that the network is serving users not supporting SIP preconditions, the following applies for an MSC server supporting PS to CS SRVCC access transfer:

1) the MSC server shall support UA role procedures defined in IETF RFC 3262 [86], IETF RFC 3311 [87], IETF RFC 3312 [88] and IETF RFC 4032 [89];

2) upon generating an initial INVITE request, the MSC server shall include the "precondition" option tag in the Supported header field as defined in IETF RFC 3312 [88] as updated by IETF RFC 4032 [89] and the MSC server shall not indicate the requirement for the precondition mechanism by using the Require header field mechanism; and

3) the MSC server shall indicate the related local preconditions, using the segmented status type, as defined in IETF RFC 3312 [88] and IETF RFC 4032 [89], as well as the strength-tag value "mandatory" for the local segment and the strength-tag value either "optional" or as specified in RFC 3312 [88] and RFC 4032 [89] for the remote segment.

7 Roles for call origination for service continuity

7.1 Introduction

This clause specifies the procedures for call origination, both where the SC UE is generating calls in the CS domain and where the SC UE is generating calls using the IM CN subsystem. Procedures are specified for the SC UE, the SCC AS, the EATF and the ATCF.

Further this clause specifies procedures for cases where the ATCF handles SIP requests that are not related to a call.

7.2 SC UE

7.2.1 General

The SC UE shall support origination of IP multimedia sessions in the IM CN subsystem as specified in 3GPP TS 24.229 [2]. The SC UE shall populate the SIP INVITE request according to subclause 6A.2.2.2.

The SC UE shall support origination of calls in the CS domain as specified in 3GPP TS 24.008 [8].

If SC using ICS is enabled then the procedures for call origination where the SC UE is initiating calls using CS media are identical to that for ICS UE specified in 3GPP TS 24.292 [4].

When originating an emergency call as specified in 3GPP TS 24.229 [2] and if the SC UE has an IMEI, then the SC UE shall include the sip.instance media feature tag as specified in IETF RFC 5626 [22] with value based on the IMEI as defined in 3GPP TS 23.003 [12] in the Contact header field of the SIP INVITE request according to IETF RFC 3840 [53].

7.2.2 Additional procedures with MSC server assisted mid-call feature

Upon receiving a SIP 2xx response to the SIP INVITE request, if:

1. the SC UE supports the MSC server assisted mid-call feature;
2. the g.3gpp.mid-call feature-capability indicator is included in the Feature-Caps header field received during session establishment;
3. the remote UE is a conference focus; and

NOTE: conference focus includes the isfocus media feature tag specified in IETF RFC 3840 [53] in own Contact header field when establishing a session.

4. the session was created as result of the SC UE creating a conference;

then the SC UE shall subscribe to the conference event package as specified in 3GPP TS 24.605 [31] and shall populate the Contact header field of the SUBSCRIBE request with the g.3gpp.mid-call media feature tag.

If the subscription is accepted then the SC UE shall keep one subscription to the conference event package with own Contact header field containing the g.3gpp.mid-call media feature tag for each conference where the SC UE participates using procedures specified in 3GPP TS 24.605 [31].

7.3 SCC AS

7.3.1 Distinction of requests sent to the SCC AS

The SCC AS needs to distinguish between the following initial SIP INVITE requests to provide specific functionality relating to call origination:

- SIP INVITE requests routed to the SCC AS over the ISC interface as a result of processing filter criteria at the S-CSCF according to the origination procedures as specified in 3GPP TS 24.229 [2], and therefore distinguished by the URI relating to this particular filter criteria appearing in the topmost entry in the Route header. In the procedures below, such requests are known as "SIP INVITE requests due to originating filter criteria". It is

assumed that the SCC AS is the first AS that the S-CSCF forwards the request to after receiving the request from the UE.

The SCC AS shall store the SIP INVITE requests due to PS to CS STN (as defined in subclause 9.3.1) and the SIP INVITE requests due to originating filter criteria, at least until their sessions are terminated.

The SCC AS needs to distinguish between the following initial requests to provide specific functionality related to obtaining conference participants:

- SIP SUBSCRIBE requests with an Event header field containing "conference" and with the Contact header field containing the g.3gpp.mid-call media feature tag routed to the SCC AS over the ISC interface as a result of processing initial filter criteria at the S-CSCF according to the originating procedures as specified in 3GPP TS 24.229 [2]. In the procedures below, such requests are known as "SIP SUBSCRIBE requests to conference event package".

Other SIP initial requests for a dialog, and requests for a SIP standalone transaction can be dealt with in any manner conformant with 3GPP TS 24.229 [2].

7.3.2 Call origination procedures at the SCC AS

When the SCC AS receives a SIP INVITE request due to originating filter criteria, the SCC AS shall follow the SCC AS roles for call origination procedures specified in 3GPP TS 24.292 [4].

The SCC AS shall populate the SIP 1xx response or SIP 2xx response to the SIP INVITE request according to subclause 6A.4.3.

If the SCC AS supports the MSC Server assisted mid-call feature according to operator policy, the SCC AS shall remove the g.3gpp.mid-call media feature tag as described in annex C from the SIP INVITE request due to originating filter criteria before forwarding the SIP INVITE request towards the remote UE.

If the SCC AS supports the PS to CS SRVCC for calls in alerting phase according to operator policy, the SCC AS shall remove the g.3gpp.srvcc-alerting media feature tag as described in annex C from the SIP INVITE request due to originating filter criteria before forwarding the SIP INVITE request towards the remote UE.

The SCC AS shall include the "tdialog" option tag and the "replaces" option tag in the Supported header field of SIP 2xx response to the SIP INVITE request due to originating filter criteria.

When the SCC AS receives any SIP 1xx response or SIP 2xx response to a SIP INVITE request due to originating filter criteria, the SCC AS shall:

1) save the Contact header field included in the SIP 1xx response or SIP 2xx response;

2) save the P-Asserted-Identity header field included in the SIP 2xx response; and

3) if included in the SIP response, save the Privacy header field included in the SIP 2xx response.

NOTE: If the SCC AS subsequently receives an initial SIP INVITE request due to STN-SR, the SCC AS will include the saved P-Asserted-Identity in the SIP 2xx response to the initial SIP INVITE request due to STN-SR and the saved Contact header field of the remote UE in SIP 1xx response or SIP 2xx response to the initial SIP INVITE request due to STN-SR.

7.3.3 Subscription related procedures in the SCC AS

When the SCC AS receives a SIP SUBSCRIBE request to conference event package, if the SCC AS supports the MSC Server assisted mid-call feature according to operator policy and if SCC AS determines that the subscription is related to an anchored session then the SCC AS shall ensure that it remains on the path for future requests in the dialog before forwarding the request.

NOTE: ASs acting as Routeing B2BUA and record-routing ASs acting as SIP proxy remain on the path for future requests in the dialog.

When the SCC AS receives SIP 2xx response to the SIP NOTIFY request with conference information, the SCC AS shall update the stored conference information based on the SIP NOTIFY request content and forward the SIP 2xx response in any manner conformant with 3GPP TS 24.229 [2].

The SCC AS shall determine that a subscription to conference event package is related to a session if:

1. the session was originated by served SC UE;
2. remote UE of the session is a conference focus;
3. the P-Asserted-Identity header field of the served SC UE used at the establishment of the session is the same as the P-Asserted-Identity header field of the served SC UE used at the subscription; and
4. the Contact or the P-Asserted-Identity header field provided to the served SC UE at the establishment of the session is the same as the Request-URI used at the subscription.

If multiple such subscriptions exist, the SCC AS shall select the subscription that originates from the same device as the session.

7.4 EATF

7.4.1 Distinction of requests sent to the EATF

The EATF needs to distinguish between the following initial SIP INVITE requests to provide specific functionality relating to call origination:

- SIP INVITE request including a request URI that contains an emergency service URN, i.e. a service URN with a top-level service type of "sos" as specified in IETF RFC 5031 [17]. In the procedures below, such requests are known as "SIP INVITE requests due to emergency service URN".

Other SIP initial requests for a dialog, and requests for a SIP standalone transaction can be dealt with in any manner conformant with 3GPP TS 24.229 [2].

7.4.2 Call origination procedures at the EATF

When the EATF receives a SIP INVITE requests due to emergency service URN, the EATF shall store the SIP INVITE request until the session is terminated, anchor the session and act as specified for a routeing B2BUA in 3GPP TS 24.229 [2], subclause 5.7.5.2.1.

In addition, if:

a) the EATF supports PS to CS DRVCC for emergency session;

b) the SC UE has indicated support for PS to CS DRVCC for emergency session by including the g.3gpp.dynamic-e-stn-drvcc media feature tag; and

c) the P-Access-Network-Info header field in the received SIP INVITE request contains an access-class field set to "3GPP-WLAN" or "untrusted-non-3GPP-VIRTUAL-EPC", and a "network-provided" parameter;

then the EATF shall:

NOTE: The coding of the P-Access-Network-Info header field can be found in 3GPP TS 24.229 [2].

a) generate an E-STN-DR that allows to associate the emergency session on the source access leg with a session transfer INVITE request due to E-STN-DR; and

b) insert a Feature-Caps header field as described in RFC 6809 [60] with the g.3gpp.dynamic-e-stn-drvcc feature-capability indicator containing the E-STN-DR in in a SIP 200 (OK) response to the INVITE request due to emergency service URN as described in annex C.28.

In addition, if:

a) the EATF supports the PS to CS SRVCC for originating emergency sessions in alerting phase according to operator policy;

b) the g.3gpp.srvcc-alerting media feature tag as described in annex C is included in the Contact header field of the SIP INVITE request; and

c) the EATF is aware by local policy that all MSC Servers in the network, where the ATCF is, which can be involved in the PS to CS SRVCC procedures, support the PS to CS SRVCC for originating emergency sessions in alerting phase;

then:

a) the EATF shall insert a Feature-Caps header field as described in RFC 6809 [60] with the g.3gpp.srvcc-alerting feature-capability indicator in the SIP 200 (OK) response to the INVITE request due to emergency service URN as described in annex C; and

b) if:

1) the EATF supports the PS to CS SRVCC for originating emergency sessions in pre-alerting phase according to operator policy;

2) the g.3gpp.ps2cs-srvcc-orig-pre-alerting media feature tag as described in annex C is included in the Contact header field of the SIP INVITE request; and

3) the EATF is aware by local policy that all MSC Servers in the network, where the ATCF is, which can be involved in the PS to CS SRVCC procedures, support the PS to CS SRVCC for originating emergency sessions in pre-alerting phase;

then the EATF shall insert a Feature-Caps header field as described in RFC 6809 [60] with the g.3gpp.ps2cs-srvcc-orig-pre-alerting feature-capability indicator in the SIP 200 (OK) response to the INVITE request due to emergency service URN as described in annex C.

7.5 Access Transfer Control Function (ATCF)

7.5.1 Distinction of requests

The ATCF needs to distinguish the following initial SIP requests:

1) SIP INVITE requests:

A) with the ATCF URI for originating requests in the topmost Route header field; and

B) with the Request-URI containing a URI not matching the STI-rSR allocated to the ATCF.

NOTE: If ATCF does not support the CS to PS SRVCC, the STI-rSR is not allocated to the ATCF.

In the procedures below, such requests are known as "originating SIP INVITE requests from SC UE".

2) SIP requests other than SIP INVITE requests creating a dialog, with the ATCF URI for originating requests in the topmost Route header field. In the procedures below, such requests are known as "originating SIP requests other than INVITE, creating a dialog".

3) SIP requests for a standalone transaction with the ATCF URI for originating requests in the topmost Route header field. In the procedures below, such requests are known as "originating SIP standalone request".

4) SIP request for an unknown method that does not relate to an existing dialog with the ATCF URI for originating requests in the topmost Route header field. In the procedures below, such requests are known as "originating unknown SIP requests".

5) SIP INVITE requests:

A) with the ATCF management URI in the topmost Route header field; and

B) with application/vnd.3gpp.srvcc-ext+xml MIME body containing <srvcc-ext> root element containing <Setup-info> element containing <direction> element with value "initiator".

In the procedures below, such requests are known as "originating SIP INVITE requests from MSC server".

7.5.2 Call origination procedures in the ATCF

7.5.2.1 General

For all SIP transactions identified:

- if priority is supported, as containing an authorised Resource-Priority header field or a temporarily authorised Resource-Priority header field, or, if such an option is supported, relating to a dialog which previously contained an authorised Resource-Priority header field;

the ATCF shall give priority over other transactions or dialogs. This allows special treatment of such transactions or dialogs.

NOTE: The special treatment can include filtering, higher priority processing, routeing, call gapping. The exact meaning of priority is not defined further in this document, but is left to national regulation and network configuration.

7.5.2.2 Sessions originated in PS domain

Upon receiving the originating SIP INVITE request from SC UE, the ATCF shall:

NOTE 1: Since the ATCF acts as proxy, the dialog identifier of the SIP INVITE request is not modified by procedures of the subclause.

0) insert a Record-Route header field containing the SIP URI of the ATCF;

1) if the latest SRVCC-related information received for the registration path which the session being established, contains ATU-STI for PS to CS SRVCC and C-MSISDN:

   A) associate the session being established with the C-MSISDN and the ATU-STI for PS to CS SRVCC bound to the registration path (see subclause 6A.3.1); and

   B) if the originating SIP INVITE request from SC UE contains an SDP offer and if the ATCF decided to anchor the media according to operator policy as specified in 3GPP TS 23.237 [9], replace the SDP offer in the originating SIP INVITE request from SC UE with an updated SDP offer using media parameters provided by the ATGW; and

NOTE 2: ATCF interacts with ATGW to provide the needed media related information. The details of interaction between ATCF and ATGW are out of scope of this document.

2) if the ATCF is located in the visited network, and local policy requires the application of IBCF capabilities in the visited network towards the home network, select an IBCF in the visited network and add the URI of the selected IBCF to the topmost Route header field;

before forwarding the request.

When the ATCF receives any SIP 1xx response or SIP 2xx response to the originating SIP INVITE request from SC UE, the ATCF shall:

1) save the Contact header field included in the SIP 1xx response or SIP 2xx response;

2) save the P-Asserted-Identity header field included in the SIP 2xx response;

3) if included in the response, save the Privacy header field included in the SIP 2xx response;

4) save the P-Charging-Vector header field included in the SIP 1xx response or SIP 2xx response; and

5) save the Feature-Caps header field(s) included in the SIP 1xx response or SIP 2xx response.

NOTE 3: If the ATCF subsequently receives an initial SIP INVITE request due to STN-SR, the ATCF will include the saved P-Asserted-Identity in the SIP 2xx response to the initial SIP INVITE request due to STN-SR and the saved the Contact header field of the remote UE in its SIP 1xx responses and the SIP 200 (OK) response to the initial SIP INVITE request due to STN-SR as describe in subclause 12.7.2.2.

NOTE 4: There are situations when the P-Asserted-Identity header field with the public user identity of the remote user can not be saved during the establishement of the communication, e.g. if presentation of the remote user public identity is restricted or if the user does not subscribe to the OIP or TIP service. In those situations the P-Asserted-Identity header field with a public user identity will not be delivered to the MSC server in the SIP 2xx response to the SIP INVITE due to STN-SR or the SIP INVITE request transferring additional session and can this limit the supplementary services that the MSC server can use after SRVCC access transfer is completed.

7.5.2.3 Sessions originated in CS domain

If the ATCF supports the CS to PS SRVCC, upon receiving the originating SIP INVITE request from MSC server, the ATCF shall act as B2BUA and shall:

1) if ATCF contains an SRVCC-related information (see subclause 6A.3.1) containing C-MSISDN equal to the <C-MSISDN> element of the <Setup-info> element of the value <srvcc-ext> root element of the application/vnd.3gpp.srvcc-ext+xml MIME body of the SIP INVITE request:

   A) associate the session being established with the latest SRVCC-related information (see subclause 6A.3.1) containing C-MSISDN equal to the <C-MSISDN> element of the <Setup-info> element of the value <srvcc-ext> root element of the application/vnd.3gpp.srvcc-ext+xml MIME body of the SIP INVITE request; and

   B) store the value of the g.3gpp.ti media feature tag of the Contact header field of the SIP INVITE request; and

2) send a SIP INVITE request towards the home network according to 3GPP TS 24.229 [2]. The ATCF shall populate the SIP INVITE request towards the home network with:

   A) the Request-URI set to the Request-URI of the originating SIP INVITE request from MSC server;

   B) all Route header fields of the originating SIP INVITE request from MSC server except the topmost Route header field;

   C) the Record-Route header field containing the SIP URI of the ATCF;

   D) the Recv-Info header fields of the originating SIP INVITE request from MSC server except the Recv-Info header field containing the g.3gpp.access-transfer-events info package name;

   E) the Accept header fields of the originating SIP INVITE request from MSC server except the Accept header field containing the application/vnd.3gpp.access-transfer-events+xml MIME type;

   F) if an Accept header field of the originating SIP INVITE request from MSC server contains the application/vnd.3gpp.access-transfer-events+xml with the "et" parameter indicating ability to receive "event-type" attribute with values additional to the value "2":

      a) the Accept header field containing the application/vnd.3gpp.access-transfer-events+xml MIME type with the "et" parameter indicating ability to receive "event-type" attribute with the additional values; and

      b) the Recv-Info header field containing the g.3gpp.access-transfer-events info package name;

   G) if the originating SIP INVITE request from MSC server contains an SDP offer and if the ATCF decided to anchor the media according to operator policy as specified in 3GPP TS 23.237 [9]:

      a) all MIME bodies of the originating SIP INVITE request from MSC server apart from the application/vnd.3gpp.srvcc-ext+xml MIME body and apart from application/sdp MIME body; and

      b) application/sdp MIME body with updated SDP offer using media parameters provided by the ATGW;

NOTE: ATCF interacts with ATGW to provide the needed media related information. The details of interaction between ATCF and ATGW are out of scope of this document.

   H) if the originating SIP INVITE request from MSC server does not contain an SDP offer or if the ATCF decided not to anchor the media according to operator policy as specified in 3GPP TS 23.237 [9]:

      a) all MIME bodies of the originating SIP INVITE request from MSC server apart from the application/vnd.3gpp.srvcc-ext+xml MIME body; and

I) if the ATCF is located in the visited network, and local policy requires the application of IBCF capabilities in the visited network towards the home network, select an IBCF in the visited network and add the URI of the selected IBCF to the topmost Route header field.

When the ATCF receives any SIP 1xx response or SIP 2xx response to the SIP INVITE request towards the home network, the ATCF shall:

1) save the Contact header field included in the SIP response; and

2) generate and send a SIP response to the originating SIP INVITE request from MSC server populated with:

A) the same status code as the received SIP response to the SIP INVITE request towards the home network;

B) the Record-Route header field containing the SIP URI of the ATCF;

C) the Recv-Info header fields of the received SIP response except the Recv-Info header field containing the g.3gpp.access-transfer-events info package name;

D) if the SIP response is a SIP 1xx response:

a) the Recv-Info header field containing the g.3gpp.access-transfer-events info package name with the "et" parameter indicating ability to receive "event-type" attribute with value "1", value "3", value "4" and values, if any, indicated in the "et" parameter of the g.3gpp.access-transfer-events info package name of the Recv-Info header field of the received SIP response; and

E) if the SIP response is a SIP 2xx response:

a) the Recv-Info header field containing the g.3gpp.access-transfer-events info package name;

b) the Accept header fields of the received SIP response except the Accept header field containing the application/vnd.3gpp.access-transfer-events+xml MIME type; and

c) the Accept header field containing the application/vnd.3gpp.access-transfer-events+xml MIME type with the "et" parameter indicating ability to receive "event-type" with value "1", value "3", value "4" and values, if any, indicated in the "et" parameter of the application/vnd.3gpp.access-transfer-events+xml MIME type of the Accept header field of the received SIP response.

7.5.3 Procedures in the ATCF for originating requests not related to a call

Upon receiving a

1. originating SIP request other than SIP INVITE request, creating a dialog;

2. originating SIP standalone request; or

3. originating unknown SIP request;

the ATCF shall:

1) if the ATCF is located in the visited network, and local policy requires the application of IBCF capabilities in the visited network towards the home network, select an IBCF in the visited network and add the URI of the selected IBCF to the topmost Route header field;

before forwarding the request.

7.6 MSC server

7.6.1 Call origination procedures

Upon receipt of a CC SETUP message from the SC UE and if the MSC server:

1) is enhanced for ICS and supports CS to PS SRVCC; and

2) the latest SRVCC information received for the registration path of the SC UE contains the ATCF management URI and the C-MSISDN;

then when sending the SIP INVITE request due to receipt of a CC SETUP message from the SC UE as specified in 3GPP TS 29.292 [18] and 3GPP TS 24.292 [4], then the MSC server shall additionally populate the SIP INVITE request with:

1) topmost Route header field with the ATCF management URI and lr URI parameter;

2) the Accept header field containing application/vnd.3gpp.access-transfer-events+xml MIME type with the "et" parameter indicating ability to receive "event-type" attribute with the value "2";

3) the Recv-Info header field containing the g.3gpp.access-transfer-events info package name;

4) application/vnd.3gpp.srvcc-ext+xml MIME body with the <srvcc-ext> root element containing the <Setup-info> element containing the CS to PS SRVCC information bound to the registration path (see subclause 6A.3.1) and indicating the "initiator" role of the MSC server in the session set up; and

5) the g.3gpp.ti media feature tag with value as described in subclause C.12 in the Contact header field.

8 Roles for call termination for service continuity

8.1 Introduction

This clause specifies the procedures for call termination, both where the SC UE is receiving calls in the CS domain and where the SC UE is receiving calls using the IM CN subsystem. Procedures are specified for the SC UE, the SCC AS and the ATCF.

8.2 SC UE

The SC UE shall support termination of multimedia sessions in the IM CN subsystem as specified in 3GPP TS 24.229 [2] with the following clarifications:

1) If the SC UE supports the MSC server assisted mid-call feature, and the receiving SIP INVITE request includes g.3gpp.mid-call feature-capability indicator, as described in annex C, in the Feature-Caps header field, the SC UE shall include the g.3gpp.mid-call media feature tag as described in annex C in the Contact header field of the SIP 2xx response to the SIP INVITE request according to IETF RFC 3840 [53].

1a) If the SC UE supports the PS to CS SRVCC for calls in alerting phase, and the receiving SIP INVITE request includes the g.3gpp.srvcc-alerting feature-capability indicator as described in annex C in a Feature-Caps header field, the SC UE shall include the g.3gpp.srvcc-alerting media feature tag as described in annex C in the Contact header field of the SIP 1xx and SIP 2xx responses to the SIP INVITE request according to IETF RFC 3840 [53].

1b) If the SC UE supports the PS to CS SRVCC for terminating calls in pre-alerting phase, and the receiving SIP INVITE request includes the g.3gpp.ps2cs-srvcc-term-pre-alerting feature-capability indicator as described in annex C in a Feature-Caps header field, the SC UE shall include the g.3gpp.ps2cs-srvcc-term-pre-alerting media feature tag as described in annex C in the Contact header field of the SIP 1xx and SIP 2xx responses to the SIP INVITE request according to IETF RFC 3840 [53].

2) If the SC UE not supporting ICS or supporting ICS but with ICS capabilities disabled receives a SIP INVITE request containing a SDP offer which includes speech media component transported using an IP bearer, and:

NOTE 1: An indication that an SC UE with ICS capabilities has its ICS capabilities enabled or disabled can be found in the ICS MO ICS_Capabilities_Enabled leaf node (see 3GPP TS 24.286 [23]).

a) if the SC UE sends the response to the SIP INVITE request over GERAN;

b) if the SC UE sends the response to the SIP INVITE request over: and

- E-UTRAN, the IMSVoPS indicator indicates that voice is not supported, and no persistent EPS bearer context exists at the SC UE; or

- UTRAN, and the IMSVoPS indicator indicates that voice is not supported; or

c) if the SC UE sends the response to the SIP INVITE request over an access network other than E-UTRAN, UTRAN and GERAN, and the access network does not support the offered speech media component transported using an IP bearer;

then the SC UE shall send back a SIP 488 (Not Acceptable Here) response without a message body.

The SC UE not supporting ICS or with ICS capabilities disabled shall support termination of calls in the CS domain as specified in 3GPP TS 24.008 [8].

An SC UE that supports ICS and has ICS capabilities enabled shall follow the call termination procedures as specified in 3GPP TS 24.292 [4].

When the SC UE not supporting ICS or with ICS capabilities disabled, and supports multiple registrations receives a SIP INVITE request containing SDP for establishing a session using just an IP bearer, then the SC UE shall establish this session in accordance with 3GPP TS 24.229 [2] with the following clarification:

- if the SIP INVITE request contains a Target-Dialog header field containing dialog parameters that correspond to an existing dialog (or a dialog in the process of being established) between the SC UE and SCC AS, the SC UE shall treat the SIP INVITE request as another dialog that is part of the same session as the dialog identified by the dialog parameters contained in the Target-Dialog header field; and

- if the SIP INVITE request does not contain a Target-Dialog header field but there is an existing dialog (or a dialog in the process of being established) between the SC UE and SCC AS, the SC UE shall check if the dialog parameters for this request correspond to the dialog parameters received in a Target-Dialog header field received on an existing dialog (or a dialog in the process of being established) between the SC UE and SCC AS and if so then the SC UE shall treat the SIP INVITE request as another dialog that is part of the same session as the dialog that the Target-Dialog header field was received on.

NOTE 2: The second case is to cover the possibility that requests can arrive out of the order that they were sent.

If the SC UE supports the use of dynamic STN, the SC UE shall include the g.3gpp.dynamic-stn media feature tag according to annex C in the Contact header field of SIP 1xx and SIP 2xx responses according to IETF RFC 3840 [53].

If the SC UE supports PS to CS dual radio access transfer of calls in alerting phase and if the g.3gpp.drvcc-alerting feature-capability indicator is included in the SIP INVITE request, the SC UE shall include in all SIP 18x responses to the SIP INVITE request, the g.3gpp.drvcc-alerting media feature tag as described in annex C in the Contact header field according to IETF RFC 3840 [53].

8.3 SCC AS

8.3.1 Distinction of requests sent to the SCC AS

The SCC AS needs to distinguish between the following initial SIP INVITE requests to provide specific functionality relating to call termination:

- SIP INVITE requests routed to the SCC AS over the ISC interface as a result of processing filter criteria at the S-CSCF according to the termination procedures as specified in 3GPP TS 24.229 [2], and therefore distinguished by the URI relating to this particular filter criteria appearing in the topmost entry in the Route header field. In the procedures below, such requests are known as "SIP INVITE requests due to terminating filter criteria". It is assumed that the SCC AS is the last AS that the S-CSCF forwards the request to.

Other SIP initial requests for a dialog, and requests for a SIP standalone transaction can be dealt with in any manner conformant with 3GPP TS 24.229 [2].

8.3.2 Call termination procedures in the SCC AS

When the SCC AS receives a SIP INVITE request due to terminating filter criteria, the SCC AS shall:

1) follow the SCC AS roles for call termination procedures specified in 3GPP TS 24.292 [4];

2) save the Contact header field included in the terminating SIP INVITE request;

2) save the P-Asserted-Identity header field included in the terminating SIP INVITE request; and

3) if included in the response, save the Privacy header field included in the terminating SIP INVITE request.

NOTE 1: If the SCC AS subsequently receives an initial SIP INVITE request due to STN-SR, the SCC AS will include the saved P-Asserted-Identity in the SIP 2xx response to the initial SIP INVITE request due to STN-SR and the saved the Contact header field of the remote UE in its SIP 1xx responses and the SIP 200 (OK) response to the initial SIP INVITE request due to STN-SR.

If:

1. the SCC AS supports the MSC Server assisted mid-call feature according to operator policy; and
2. the SCC AS is aware:
   - by local policy; or
   - by ATCF indicating support of the MSC server assisted mid-call feature;

NOTE 2: An ATCF can indicate support of the MSC server assisted mid-call feature by inclusion of the g.3gpp.mid-call feature-capability indicator in the Feature-Caps header field, with the g.3gpp.atcf feature-capability indicator, in the SIP REGISTER request that created the binding of the SC UE.

that all MSC Servers in the network where the UE is registered which can be involved in the PS to CS SRVCC procedures support the MSC Server assisted mid-call feature;

then the SCC AS shall include the g.3gpp.mid-call feature-capability indicator, as described in annex C, in the Feature-Caps header field of the SIP INVITE request due to terminating filter criteria according to IETF RFC 6809 [60].

If the SCC AS supports the MSC Server assisted mid-call feature according to operator policy, the SCC AS shall remove the g.3gpp.mid-call media feature tag as described in annex C from the SIP 2xx response to the SIP INVITE request due to terminating filter criteria before forwarding the SIP 2xx response towards the remote UE.

If:

1. the SCC AS supports the PS to CS SRVCC for calls in alerting phase according to operator policy; and
2. the SCC AS is aware:
   - by local policy; or
   - by ATCF indicating support of the PS to CS SRVCC for calls in alerting phase;

NOTE 3: An ATCF can indicate support of the PS to CS SRVCC for calls in alerting phase by inclusion of the g.3gpp.srvcc-alerting feature-capability indicator in the Feature-Caps header field, with the g.3gpp.atcf feature-capability indicator, in the SIP REGISTER request that created the binding of the SC UE.

that all MSC Servers in the network where the UE is registered which can be involved in the PS to CS SRVCC procedures support the PS to CS SRVCC for calls in alerting phase;

then the SCC AS shall include the g.3gpp.srvcc-alerting feature-capability indicator as described in annex C in the Feature-Caps header field of the SIP INVITE request due to terminating filter criteria according to IETF RFC 6809 [60].

If:

1. the SCC AS supports the PS to CS SRVCC for terminating calls in pre-alerting phase according to operator policy; and
2. the SCC AS is aware:
   - by local policy; or
   - by ATCF indicating support of the PS to CS SRVCC for terminating calls in pre-alerting phase;

NOTE 4: An ATCF can indicate support of the PS to CS SRVCC for terminating calls in pre-alerting phase by inclusion of the g.3gpp.ps2cs-srvcc-term-pre-alerting feature-capability indicator in the Feature-Caps header field, with the g.3gpp.atcf feature-capability indicator, in the SIP REGISTER request that created the binding of the SC UE.

that all MSC Servers in the network where the UE is registered which can be involved in the PS to CS SRVCC procedures support the PS to CS SRVCC for terminating calls in pre-alerting phase;

then the SCC AS shall include the g.3gpp.ps2cs-srvcc-term-pre-alerting feature-capability indicator as described in annex C in the Feature-Caps header field of the SIP INVITE request due to terminating filter criteria according to IETF RFC 6809 [60].

If the SCC AS supports the PS to CS SRVCC for calls in alerting phase according to operator policy, the SCC AS shall remove the g.3gpp.srvcc-alerting media feature tag as described in annex C from SIP 1xx and SIP 2xx responses to the SIP INVITE request due to terminating filter criteria before forwarding the SIP 1xx and SIP 2xx responses towards the remote UE.

If the SCC AS supports the PS to CS SRVCC for terminating calls in pre-alerting phase according to operator policy, the SCC AS shall remove the g.3gpp.ps2cs-srvcc-term-pre-alerting media feature tag as described in annex C from SIP 1xx and SIP 2xx responses to the SIP INVITE request due to terminating filter criteria before forwarding the SIP 1xx and SIP 2xx responses towards the remote UE.

If the SCC AS supports the PS to CS dual radio access transfer for calls in alerting phase according to operator policy, the SCC AS shall include the g.3gpp.drvcc-alerting feature-capability indicator as described in annex C in the Feature-Caps header field of the SIP INVITE request due to terminating filter criteria according to IETF RFC 6809 [60].

If the SCC AS supports the use of dynamic STN according to operator policy, the SCC AS shall include an E.164 number as the dynamic STN for this session and include this dynamic STN in the g.3gpp.dynamic-stn feature-capability indicator with the dynamic STN as described in annex C in a Feature-Caps header field in the SIP INVITE request according to IETF RFC 6809 [60].

NOTE 5: The dynamic STN can either be the same or different per call based on implementation.

The SCC AS shall include the "tdialog" option tag and the "replaces" option tag in the Supported header field of the SIP INVITE request due to terminating filter sent toward the SC UE.

8.4 Access Transfer Control Function (ATCF)

8.4.1 Distinction of requests

The ATCF needs to distinguish the following initial SIP requests:

1) SIP INVITE requests with the ATCF URI for terminating requests in the topmost Route header field. In the procedures below, such requests are known as "terminating SIP INVITE requests for PS".

2) SIP INVITE requests:

   A) with the ATCF management URI in the topmost Route header field; and

   B) with application/vnd.3gpp.srvcc-ext+xml MIME body containing <srvcc-ext> root element containing <Setup-info> element containing <direction> element with value "receiver".

   In the procedures below, such requests are known as "terminating SIP INVITE requests for CS".

8.4.2 Call termination procedures in the ATCF

8.4.2.1 General

For all SIP transactions identified:

- if priority is supported, as containing an authorised Resource-Priority header field or, if such an option is supported, relating to a dialog which previously contained an authorised Resource-Priority header field;

the ATCF shall give priority over other transactions or dialogs. This allows special treatment of such transactions or dialogs.

NOTE: The special treatment can include filtering, higher priority processing, routeing, call gapping. The exact meaning of priority is not defined further in this document, but is left to national regulation and network configuration.

8.4.2.2 Sessions terminated in PS domain

Upon receiving the terminating SIP INVITE request for PS, the ATCF shall:

NOTE 1: Since the ATCF acts as proxy, the dialog identifier of the SIP INVITE request is not modified by procedures of the subclause.

1) if a Feature-Caps header field containing the g.3gpp.srvcc feature-capability indicator is contained in the SIP INVITE request:

   A) insert a Record-Route header field containing the SIP URI of the ATCF; and

   B) if the latest SRVCC-related information received for the registration path which the session being established, is using contains ATU-STI for PS to CS SRVCC and C-MSISDN:

      a) associate the session being established with the C-MSISDN and the ATU-STI for PS to CS SRVCC bound to the registration path (see subclause 6A.3.1); and

      b) if the terminating SIP INVITE request for PS contains an SDP offer and if the ATCF decided to anchor the media according to operator policy as specified in 3GPP TS 23.237 [9], replace the SDP offer in the terminating SIP INVITE request with an updated SDP offer using media parameters provided by ATGW;

NOTE 2: ATCF interacts with ATGW to provide the needed media related information. The details of interaction between ATCF and ATGW are out of scope of this document.

2) save the Contact header field included in the terminating SIP INVITE request for PS;

3) save the P-Asserted-Identity header field included in the terminating SIP INVITE request for PS;

4) if included, save the Privacy header field included in the terminating SIP INVITE request for PS;

5) save the P-Charging-Vector header field included in the terminating SIP INVITE request for PS, and

6) save the Feature-Caps header field(s) included in the terminating SIP INVITE request for PS.

NOTE 3: If the ATCF subsequently receives an initial SIP INVITE request due to STN-SR, the ATCF will include the saved P-Asserted-Identity in the SIP 2xx response to the initial SIP INVITE request due to STN-SR and the saved the Contact header field of the remote UE in its SIP 1xx responses and the SIP 200 (OK) response to the initial SIP INVITE request due to STN-SR as describe in subclause 12.7.2.2.

NOTE 4: There are situations when the P-Asserted-Identity header field with the public user identity of the remote user can not be saved during the establishement of the communication, e.g. if presentation of the remote user public identity is restricted or if the user does not subscribe to the OIP or TIP service. In those situations the P-Asserted-Identity header field with a public user identity will not be delivered to the MSC server in the SIP 2xx response to the SIP INVITE due to STN-SR or the SIP INVITE request transferring additional session and can this limit the supplementary services that the MSC server can use after SRVCC access transfer is completed.

before forwarding the request.

8.4.2.3 Sessions terminated in CS domain

If ATCF supports CS to PS SRVCC then upon receiving the terminating SIP INVITE request for CS, the ATCF shall act as B2BUA and shall:

1) save the Contact header field included in the terminating SIP INVITE request for CS;

2) if ATCF contains an SRVCC-related information (see subclause 6A.3.1) containing C-MSISDN equal to the <C-MSISDN> element of the <Setup-info> element of the value <srvcc-ext> root element of the application/vnd.3gpp.srvcc-ext+xml MIME body of the SIP INVITE request:

   A) associate the session being established with the latest SRVCC-related information (see subclause 6A.3.1) containing C-MSISDN equal to the <C-MSISDN> element of the <Setup-info> element of the value <srvcc-ext> root element of the application/vnd.3gpp.srvcc-ext+xml MIME body of the SIP INVITE request; and

3) send a SIP INVITE request towards the MSC server according to 3GPP TS 24.229 [2]. The ATCF shall populate the SIP INVITE request towards the MSC server with:

   A) the Request-URI set to the Request-URI of the terminating SIP INVITE request for CS;

   B) all Route header fields of the terminating SIP INVITE request for CS except the topmost Route header field;

   C) the Record-Route header field containing the SIP URI of the ATCF;

   D) the Accept header fields of the terminating SIP INVITE request for CS except the Accept header field containing the application/vnd.3gpp.access-transfer-events+xml MIME type;

   E) the Accept header field containing the application/vnd.3gpp.access-transfer-events+xml MIME type with the "et" parameter indicating ability to receive "event-type" attribute with value "1", value "3", value "4" and values, if any, indicated in the "et" parameter of the application/vnd.3gpp.access-transfer-events+xml MIME type of the Accept header field of the terminating SIP INVITE request for CS;

   F) the Recv-Info header fields of the terminating SIP INVITE request for CS;

   G) the Recv-Info header field containing the g.3gpp.access-transfer-events info package name, if not included already;

   H) if the terminating SIP INVITE request for CS contains an SDP offer and if the ATCF decided to anchor the media according to operator policy as specified in 3GPP TS 23.237 [9]:

      a) all MIME bodies of the terminating SIP INVITE request for CS apart from the application/vnd.3gpp.srvcc-ext+xml MIME body and apart from application/sdp MIME body; and

      b) application/sdp MIME body with updated SDP offer using media parameters provided by the ATGW; and

NOTE: ATCF interacts with ATGW to provide the needed media related information. The details of interaction between ATCF and ATGW are out of scope of this document.

   I) if the terminating SIP INVITE request for CS does not contain an SDP offer or if the ATCF decided not to anchor the media according to operator policy as specified in 3GPP TS 23.237 [9]:

      a) all MIME bodies of the terminating SIP INVITE request for CS apart from the application/vnd.3gpp.srvcc-ext+xml MIME body.

When the ATCF receives any SIP 1xx response or SIP 2xx response to the SIP INVITE request towards the MSC server, the ATCF shall:

1) store the value of the g.3gpp.ti media feature tag of the Contact header field of the received SIP response to the SIP INVITE request towards the MSC server;

2) generate and send a SIP response to the terminating SIP INVITE request for CS populated with:

   A) the same status code as the received SIP response to the SIP INVITE request towards the MSC server;

   B) the Record-Route header field containing the SIP URI of the ATCF;

   C) the Recv-Info header fields of the received SIP response except the Recv-Info header field containing the g.3gpp.access-transfer-events info package name;

   D) if the SIP response is SIP 1xx response:

      a) if the SIP response contains an Recv-Info header field containing the g.3gpp.access-transfer-events info package name with the "et" parameter indicating ability to receive "event-type" attribute with values additional to the value "2", then the Recv-Info header field containing the g.3gpp.access-transfer-events info package name with the "et" parameter indicating ability to receive "event-type" attribute with the additional values; and

   E) if the SIP response is SIP 2xx response:

a) if the SIP response contains an Accept header field containing the application/vnd.3gpp.access-transfer-events+xml MIME type with the "et" parameter indicating ability to receive "event-type" attribute with values additional to the value "2":

   i) the Accept header field containing the application/vnd.3gpp.access-transfer-events+xml MIME type with the "et" parameter indicating ability to receive "event-type" attribute with the additional values; and

   ii) the Recv-Info header field containing the g.3gpp.access-transfer-events info package name.

8.5 MSC server

8.5.1 Distinction of requests

The MSC server needs to distinguish the following initial SIP requests:

1) SIP INVITE requests with the topmost Route header field containing the Path header field value inserted by the MSC server in a REGISTER request. In the procedures below, such requests are known as "terminating SIP INVITE requests from home network".

2) SIP INVITE requests with the MSC URI for redirected terminating sessions in the topmost Route header field. In the procedures below, such requests are known as "redirected terminating SIP INVITE requests".

8.5.2 Call termination procedures

8.5.2.1 SIP INVITE request from home network

Upon receiving the terminating SIP INVITE request from home network and if the MSC server:

1) is enhanced for ICS and supports CS to PS SRVCC; and

2) the latest SRVCC information received for the registration path of the SC UE contains the ATCF management URI and the C-MSISDN;

then the MSC server instead of interworking of mobile terminating call setup from SIP to NAS signalling according 3GPP TS 29.292 [18] and 3GPP TS 24.292 [4], the MSC server shall:

1) send a SIP INVITE request towards the ATCF according to 3GPP TS 24.229 [2]. The MSC server shall populate the SIP INVITE request towards the ATCF with:

   A) the Request-URI set to the Request-URI of the terminating SIP INVITE request from home network;

   B) topmost Route header field with the ATCF management URI and lr URI parameter;

   C) all MIME bodies of the terminating SIP INVITE request from home network; and

   D) application/vnd.3gpp.srvcc-ext+xml MIME body with the <srvcc-ext> root element containing the <Setup-info> element containing the CS to PS SRVCC information bound to the registration path (see subclause 6A.3.1) and indicating the "receiver" role of the MSC server in the session set up.

When the MSC server receives any SIP 1xx response or SIP 2xx response to the SIP INVITE request towards the ATCF, the MSC server shall generate and send a SIP response to the terminating SIP INVITE requests from home network populated with the same status code as the received SIP response to the SIP INVITE request towards the ATCF.

8.5.2.2 SIP INVITE request from ATCF

If the MSC server is enhanced for ICS and supports CS to PS SRVCC then upon receiving the redirected terminating SIP INVITE request, the MSC server shall interworking the mobile terminating call setup from SIP to NAS signalling according 3GPP TS 29.292 [18] and 3GPP TS 24.292 [4].

When sending a SIP 1xx response or SIP 2xx response to the redirected terminating SIP INVITE requests, MSC server shall additionally populate the SIP response with:

1) the g.3gpp.ics media feature tag with value "server" in the Contact header field;

2) if the SIP response is SIP 1xx response:

   A) Recv-Info header field containing the g.3gpp.access-transfer-events info package name and with the "et" parameter indicating ability to receive "event-type" attribute with value "2";

3) if the SIP response is SIP 2xx response:

   A) the Recv-Info header field containing the g.3gpp.access-transfer-events info package name; and

   B) the Accept header field containing the application/vnd.3gpp.access-transfer-events+xml MIME type with the "et" parameter indicating ability to receive "event-type" attribute with the value "2"; and

4) the g.3gpp.ti media feature tag with value as described in subclause C.12 in the Contact header field.

9 Roles for PS-CS access transfer

9.1 Introduction

For a UE or an AS not supporting ICS procedures, PS-CS access transfer procedures enable transfer of

- one full-duplex session with active speech or speech/video component;
- up to one full-duplex session with active speech or speech/video media component and up to one full-duplex session with inactive speech or speech/video media component when the MSC Server assisted mid-call feature is supported;
- one full-duplex session in an early dialog phase that can be in a pre-alerting or an alerting phase when dual radio access transfer for originating calls in pre-alerting phase or dual radio access transfer for calls in alerting phase are supported;
- one full-duplex session with active speech media component and one full-duplex session in an early dialog state, that can be in an originating pre-alerting or an alerting phase, when dual radio access transfer for originating calls in pre-alerting phase or dual radio access transfer for calls in alerting phase are supported; and
- one full-duplex session with inactive speech media component and one full-duplex session in an early dialog phase, that can be in a pre-alerting or alerting phase, when the MSC Server assisted mid-call feature and the dual radio access transfer for originating calls in pre-alerting phase or dual radio access transfer for calls in alerting phase are supported.

9.1A Additional procedures with MSC Server assisted mid-call feature

When a conference is transferred to CS domain using MSC Server assisted mid-call feature, the participants are extracted from the stored conference information as follows:

1. at maximum first 5 participants listed in the <user> elements:

   a. included in <users> parent element included in <conference-info> root element of the conference information;

   b. containing at least one <endpoint> child element with <status> child element containing one of the states "connected", "on-hold", "muted-via-focus", "pending", "alerting", "dialing-in" or "dialing-out"; and

   c where "entity" attribute is different than the URI in the P-Asserted-Identity header field of the served SC UE used at the subscription.

9.2 SC UE

9.2.0 General

Void

9.2.1 SC UE not using ICS procedures for PS to CS access transfer

The SC UE may be engaged in one or more ongoing sessions at the time of initiating access transfer. By an ongoing session, it is meant a session for which the SIP 2xx response for the initial SIP INVITE request to establish this session has been sent or received.

Additional to the ongoing sessions, the SC UE can be engaged in one or more sessions in an early dialog phase at the time of initiating access transfer. By a session in an early dialog phase, it is meant a session for which the SIP 18x response for the initial SIP INVITE request to establish this session has been sent or received but the SIP 2xx response has not yet been received or sent.

If the SC UE is not using ICS capabilities and if the SC UE does not apply the MSC Server assisted mid-call feature as specified in subclause 9.2.1A, subject to the SC_non_transferrable_media node value in the Communication Continuity MO (see subclause 5.27 in 3GPP TS 24.216 [5]), the SC UE shall:

a) if more than one ongoing full-duplex session with speech media component exists and if the SC UE does not support PS to CS dual radio access transfer of a session in an early dialog phase or if no session in an early dialog phase exists:

   1) initiate the release of all the full-duplex sessions with speech media component except the full-duplex session with active speech media component that was most recently made active; and

   2) transfer the remaining ongoing full-duplex session with active speech media component by sending a CC SETUP message as described in step A);

b) if one ongoing full-duplex session with active speech media component exist, one or more session in the early dialog phases fulfilling the criteria in subclause 9.2.4 exists and if the SC UE supports PS to CS dual radio access transfer of a session in an early dialog phase:

   1) select a session in an early dialog phase to be transferred as specified in subclause 9.2.4;

   2) initiated the release of the remaining sessions in early dialog phase, except the selected session in an early dialog phase;

   3) if the selected session in the early dialog phase is a session in the terminating alerting phase,

      - transfer the ongoing full-duplex session with active speech media component by sending a CC SETUP message as described in step A) below; and

      - when the transfer of the ongoing full-duplex session with active speech media component is completed, transfer the selected session in the terminating alerting phase by sending a SIP 488 (Not Acceptable Here) response to the SIP INVITE request creating the session in the terminating alerting phase without an SDP body as described in subclause 10.2.4 of 3GPP TS 24.292 [4]; and

   4) if the selected session in the early dialog phase is not a session in the terminating alerting phase, transfer the ongoing full-duplex session with active speech media by sending a CC SETUP message as described in step A) below and when the transfer is completed:

      - if a session in the pre-alerting phase was transferred, assign the TI flag as in the mobile originating case and the TI value as described in the table 9.2.1A-1 in subclause 9.2.1A and continue the session in the pre-alerting phase in the CS domain in the "Mobile originating call proceeding" (U3) call state as described in 3GPP TS 24.008 [8]; and

      - if a session in the originating alerting phase was transferred, assign the TI flag as in the mobile originating case and the TI value as described in the table 9.2.1A-1 in subclause 9.2.1A and continue the session in the originating alerting phase in the CS domain in the "Call delivered" (U4) call state as described in 3GPP TS 24.008 [8]; and

NOTE 1: One CC SETUP message transfers both the ongoing full-duplex session with active speech media component and the session in the early dialog phase.

c) if no ongoing full-duplex session with active speech media component exists, one or more session in the early dialog phases fulfilling the criteria in subclause 9.2.4 exists and if the SC UE supports PS to CS dual radio access transfer of asession in an early dialog phase:

1) select an session in the early dialog phase to be transferred as specified in subclause 9.2.4;

2) initiate the release of remaining sessions in early dialog phase, except the selected session;

3) if the selected session in the early dialog phase is a session in the terminating alerting phase, transfer the selected session by sending a SIP 488 (Not Acceptable Here) response to the SIP INVITE request creating the session in the terminating alerting phase without an SDP body as described in subclause 10.2.4 of 3GPP TS 24.292 [4]; and

4) if the selected session in the early dialog phase is not a session in the terminating alerting phase, transfer the selected session by sending a CC SETUP message as described in step A).

When transferring the session(s) not using ICS capabilities, the SC UE shall:

A) send a CC SETUP message (as specified in 3GPP TS 24.008 [8]) and set up a call over the CS domain. When sending CC SETUP message, the SC UE shall populate the CC SETUP message as follows:

1) the called party BCD number information element set to:

- if the SC UE supports the use of the dynamic STN and if a dynamic STN was received from the SCC AS in the g.3gpp.dynamic-stn feature-capability indicator in a Feature-Caps header field of SIP 1xx responses or the SIP 2xx response to the initial SIP INVITE request as specified in annex C, the E.164 number from the tel URI from the received dynamic STN associated with the call to transfer; or

NOTE 2: The SC UE could have multiple dialogs (e.g active session and a session on hold). The dynamic STN only applies to the dialog for which it is received so the UE needs to store the dynamic STN associated with the dialog on which it was received and upon termination of the dialog (i.e upon sending or receiving a SIP BYE request or SIP CANCEL request) the dynamic STN associated with the dialog is discarded.

- if the use of dynamic STN is not supported by the SC UE or if a dynamic STN was not received from the SCC AS in a g.3gpp.dynamic-stn feature-capability indicator in a Feature-Caps header field of SIP 1xx responses or in the SIP 2xx response to the initial SIP INVITE request, the static STN; and

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

If the SC UE receives a SIP BYE request for a session subject for access transfer and before the access transfer is completed the SC UE shall:

1. send the SIP 200 (OK) response to the SIP BYE request in accordance with 3GPP TS 24.229 [2]; and

2. abort the transfer of the session and if the session is an additional session, internally release any reserved CS resources.

NOTE 3: If only one session is subject for access transfer the session, CS resources will be released by 3GPP TS 24.008 [2] procedures.

NOTE 4: If more than one session is subject for access transfer the remaining session will be transferred by the CC SETUP message.

If the SC UE receives a release message to the CC SETUP message sent, then the PS to CS access transfer has not completed successfully and the call will continue in the Source Access Leg.

After completion of session transfer, if the SC UE is not using Gm, the SC UE shall locally release the resources, if any, that are associated with the source access leg.

9.2.1AAA SC UE not using ICS procedures for PS to CS access transfer of emergency session

The UE may support PS to CS transfer of an emergency session for which the SIP 2xx response for the initial SIP INVITE request to establish this session has been sent or received.

If an E-STN-DR was received from the EATF in g.3gpp.dynamic-e-stn-drvcc feature-capability indicator in a Feature-Caps header field of a SIP 2xx response to the initial SIP emergency INVITE request as specified in annex C, the SC UE, when transferring the emergency session, shall:

A) send a CC SETUP message (as specified in 3GPP TS 24.008 [8]) and set up a call over the CS domain. When sending CC SETUP message, the SC UE shall populate the CC SETUP message as follows:

1) the called party BCD number information element set to the E.164 number of the tel URI from the received E-STN-DR associated with the call to transfer, and

NOTE 1: The E-STN-DR only applies to the dialog for which it is received so the UE needs to store the E-STN associated with the dialog on which it was received and upon termination of the dialog (i.e upon sending or receiving a SIP BYE request or SIP CANCEL request) the E-STN-DR associated with the dialog is discarded.

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

If the SC UE receives a SIP BYE request for a session subject for access transfer and before the access transfer is completed the SC UE shall:

1. send the SIP 200 (OK) response to the SIP BYE request in accordance with 3GPP TS 24.229 [2]; and
2. abort the transfer of the session and if the session is an additional session, internally release any reserved CS resources.

NOTE 2: The access transfer is completed once the SC UE has received CC CONNECT message.

If the SC UE receives a release message to the CC SETUP message sent, then the PS to CS access transfer has not completed successfully and the call will continue in the Source Access Leg.

After successful completion of session transfer, the SC UE shall locally release the resources, if any, that are associated with the speech media component of the emergency session on the source access leg.

9.2.1AA SC UE using ICS procedures for PS to CS access transfer

If SC UE uses ICS capabilities, this subclause applies for IMS sessions containing speech media component only, otherwise subclause 11.2.1.2 applies.

The SC UE may be engaged in one or more ongoing sessions at the time of initiating access transfer. By an ongoing session, it is meant a session for which the SIP 2xx response for the initial SIP INVITE request to establish this session has been sent or received.

If SC using ICS is enabled and if the SC UE is using Gm, then for each session with speech media component to be transferred and starting with the session with the active speech media component, the SC UE shall send a SIP INVITE request to the SCC AS. The SC UE shall populate the SIP INVITE request as follows:

1) the Request-URI set to:

- if the PS to PS STI URI is configured in the SC UE, the configured PS to PS STI URI; and
- if the PS to PS STI URI is not configured in the SC UE, the URI contained in the Contact header field returned at the creation of the dialog on the Source Access Leg;

2) include in the Contact header field according to IETF RFC 3840 [53]:

- a public GRUU or temporary GRUU as specified in 3GPP TS 24.229 [2] if a GRUU was received at registration; and

- the g.3gpp.ics media feature tag set to "principal" as specified in annex B of 3GPP TS 24.292 [4];

3) select one of the following options:

- if usage of SIP Replaces extension is selected:

  a) the Replaces header field populated as specified in IETF RFC 3891 [10], containing the dialog identifier of the session to be transferred; and

  b) the Require header field populated with the option tag value "replaces";

- if usage of SIP Target-Dialog extension is selected:

  a) the Target-Dialog header field populated as specified in IETF RFC 4538 [11], containing the dialog identifier of the session to be transferred; and

  b) the Require header field populated with the option tag value "tdialog";

4) SDP proposing an audio stream over a circuit-switched bearer in accordance with procedures for SDP for ICS UE proposing using a CS audio stream in 3GPP TS 24.292 [4]; and

5) an indication that the related local preconditions for QoS are not met as specified in 3GPP TS 24.229 [11].

Upon the SC UE receiving a reliable SIP 1xx provisional response including a PSI DN from the SCC AS, the SC UE shall follow the procedures for ICS YE setting up a CS call in 3GPP TS 24.292 [4].

When the CS resources are available to the UE, the SC UE shall send an SDP offer including an indication that the related local preconditions for QoS for audio as met as specified in 3GPP TS 24.229 [11].

Upon receiving a SIP 2xx response for the SIP INVITE request, the SC UE shall:

1) send a SIP ACK request; and

2) send a SIP BYE request to the SCC AS on the Source Access Leg to terminate the dialog on the Source Access Leg, if the dialog is still active (e.g. it has not been released by the SCC AS) and no active media streams remain on that dialog on the Source Access Leg.

NOTE: If the contact address used by the dialog over the Source Access Leg was registered using multiple registration procedure, then upon transferring the dialog to the CS domain, the SC UE is still registered on the Source Access Leg and its subscription dialog to its reg-event on the Source Access Leg is intact.

If the SC UE receives any SIP 4xx – 6xx response to the SIP INVITE request, then PS-CS access transfer has not completed successfully and the call will continue on the PS Access Leg.

If the SC UE receives a release message to the CC SETUP message sent, then PS-CS access transfer has not completed successfully and the call will continue in the Source Access Leg.

9.2.1A SC UE procedures for PS to CS access transfer with MSC server assisted mid-call feature

The SC UE shall apply the MSC Server assisted mid-call feature when transferring the session not using ICS capabilities if:

1. the SC UE supports the MSC Server assisted mid-call feature; and

2. one of the following is true:

   A. there is at least one ongoing full-duplex session with active speech media component and the Feature-Caps header field received during the establishment of the ongoing full-duplex session with active speech media component which has been most recently made active includes the g.3gpp.mid-call feature-capability indicator as described in annex C;

   B. there is no ongoing full-duplex session with active speech media component and the Feature-Caps header field received during the establishment of the ongoing full-duplex session with inactive speech media component which became inactive most recently includes the g.3gpp.mid-call feature-capability indicator as described in annex C; or

C. if

- there is one ongoing full-duplex session with active speech media component or one ongoing full-duplex session with inactive speech media component and the Feature-Caps header field received during the establishment of the ongoing full-duplex session includes the g.3gpp.mid-call feature-capability indicator as described in annex C; and

- there is at least one session in the early dialog phase fulfilling the criteria in the subclause 9.2.4 exists.

When the SC UE applies the MSC Server assisted mid-call feature, in addition to the procedures described in subclause 9.2.1, and before sending a message to set up a call over the CS domain, the SC UE shall:

1. if there are two or more ongoing full-duplex sessions with active speech media component:

   A. initiate the release of all the ongoing full-duplex sessions with speech media component except two that were most recently made active;

   B. initiate the session modification of the ongoing full-duplex session with speech media component that was made active less recently and offer the speech media component with "sendonly" or "inactive" directionality; and

   C. transfer two remaining ongoing full-duplex sessions with speech media component;

NOTE 1: When full-duplex session with active speech media component and another session with inactive speech media component exist, one CC SETUP message transfers both sessions.

2. if there are one ongoing full-duplex session with active speech media component and one or more ongoing full-duplex session with inactive speech media component:

   A. initiate the release of all the ongoing full-duplex sessions with inactive speech media component except the one which became inactive most recently; and

   B. transfer two remaining ongoing full-duplex sessions with speech media component;

NOTE 2: When full-duplex session with active speech media component and another session with inactive speech media component exist, one CC SETUP message transfers both sessions.

3. if there is one ongoing full-duplex session with active speech media component and no ongoing full-duplex session with inactive speech media component, transfer the ongoing full-duplex session with the speech media component;

4. if there is no ongoing full-duplex session with active speech media component and there is one or more ongoing full-duplex session with inactive speech media component:

   A. initiate the release of all the ongoing full-duplex sessions with inactive speech media component except the one which became inactive most recently; and

   B. transfer the ongoing full-duplex session with speech media component; and

NOTE 3: The ongoing full-duplex session with inactive speech media component is transferred to a held CS call.

5. if one ongoing full-duplex session with active speech media component or one ongoing full-duplex session with inactive speech media component, one or more session in the early dialog phases fulfilling the criteria in subclause 9.2.4 and if the SC UE supports PS to CS dual radio access transfer of a session in an early dialog phase:

   A) select a session in an early dialog phase to be transferred as specified in subclause 9.2.4;

   B) initiated the release of the remaining sessions in early dialog phase, except the selected session; and

   C) if the selected session in the early dialog phase is a session in the terminating alerting phase,

   - transfer the ongoing full-duplex session with active speech media component by sending a CC SETUP message as described in subclause 9.2.1; and

   - when the transfer of the ongoing full-duplex session with active speech media component is completed, transfer the selected session in the terminating alerting phase by sending a SIP 488 (Not Acceptable Here)

response to the SIP INVITE request creating the session in the terminating alerting phase without an SDP body as described in subclause 10.2.4 of 3GPP TS 24.292 [4]; and

4) if the selected session in the early dialog phase is not a session in the terminating alerting phase, transfer the ongoing full-duplex session with active speech media by sending a CC SETUP message as described in subclause 9.2.1 and when the transfer is completed:

   - if a session in the pre-alerting phase was transferred, continue the session in the pre-alerting phase in the CS domain in the "Mobile originating call proceeding" (U3) call state as described in 3GPP TS 24.008 [8]; and

   - if a session in the originating alerting phase was transferred, continue the session in the originating alerting phase in the CS domain in the "Call delivered" (U4) call state as described in 3GPP TS 24.008 [8].

NOTE 4: One CC SETUP message transfers both the ongoing full-duplex session with active or inactive speech media component and the session in the early dialog phase.

The SC UE shall associate the additional transferred session with a CS call with transaction identifier calculated as in the table 9.2.1A-1 and TI flag value as in mobile originated call.

NOTE 5: If the additional transfer sessions was a session in the terminating alerting phase, the TI flag value and TI value need not to be calculated as in table 9.2.1A-1, instead the TI flag value and the TI value is assigned by the MSC server and the UE using 3GPP TS 24.008 [8] call establishment procedures.

Table 9.2.1A-1: held session transaction identifier calculation formula

```
<transaction identifier of the additional transferred session> = (1 + <transaction identifier
   indicated in the CC SETUP message>) modulo 7
```

If the SC UE has a subscription as described in subclause 7.2.2 for the ongoing full-duplex session with active speech media component then:

1. the SC UE shall associate the ongoing full-duplex session and the participants extracted in subclause 9.1A with CS calls:

   - with transaction identifiers calculated as in the table 9.2.1A-2. The offsets 0, 2, 3, 4, 5 are assigned to the participants in their order in the list of the extracted participants; and

   - with TI flag value as in mobile originated call; and

NOTE 6: The transaction identifier of the CS call established by the CC SETUP message (i.e. transaction identifier of the conference) is the transaction identifier assigned to the first participant (offset 0).

2. the SC UE shall enter the "active" (U10) state (defined in 3GPP TS 24.008 [8]), the "idle" hold auxiliary state (defined in 3GPP TS 24.083 [43]) and the "call in MPTY" multi party auxiliary state (defined in 3GPP TS 24.084 [47]) for the CS calls.

NOTE 7: The "active" (U10) state for the first participant is entered as a result of the CS call being established by the 3GPP TS 24.008 [8] procedures.

Table 9.2.1A-2: transaction identifier assignment for participants

```
<transaction identifier of participant> = (<transaction identifier of transaction identifier
   indicated in the CC SETUP message> + <offset of participant>) modulo 7
```

If the ongoing full-duplex session with active speech media component does not exist and the SC UE has a subscription as described in subclause 7.2.2 for the ongoing full-duplex session with inactive speech media component then:

1. the SC UE shall associate the ongoing full-duplex session and the participants extracted in subclause 9.1A with CS calls:

   - with transaction identifiers calculated as in the table 9.2.1A-2. The offsets 0, 2, 3, 4, 5 are assigned to the participants in their order in the list of the extracted participants; and

- with TI flag value as in mobile originated call; and

NOTE 8: The transaction identifier of the CS call established by the CC SETUP message (i.e. transaction identifier of the conference) is the transaction identifier assigned to the first participant (offset 0).

2. the SC UE shall enter the "active" (U10) state (defined in 3GPP TS 24.008 [8]) , the "call held" hold auxiliary state (defined in 3GPP TS 24.083 [43]) and the "call in MPTY" multi party auxiliary state (defined in 3GPP TS 24.084 [47]) for the CS calls.

NOTE 9: The "active" (U10) state for the first participant is entered as a result of the CS call being established by the 3GPP TS 24.008 [8] procedures.

If

1. the ongoing full-duplex session with active speech media component exists and the SC UE does not have a subscription as described in subclause 7.2.2 for the ongoing full-duplex session with active speech media component; and

2. the SC UE has a subscription as described in subclause 7.2.2 for the additional transferred session;

then:

1. the SC UE shall associate the ongoing full-duplex session and the participants extracted in subclause 9.1A with CS calls:

   - with transaction identifiers calculated as in the table 9.2.1A-2. The offsets 1, 2, 3, 4, 5 are assigned to the participants in their order in the list of the extracted participants; and

   - with TI flag value as in mobile originated call; and

2. the SC UE shall enter the "active" (U10) state (defined in 3GPP TS 24.008 [8]), the "call held" auxiliary state (defined in 3GPP TS 24.083 [43])" and the "call in MPTY" multi party auxiliary state (defined in 3GPP TS 24.084 [47]) for the CS calls.

If the SC UE does not have a subscription as described in subclause 7.2.2 for the transferred session then when the transfer is completed, the SC UE shall:

1) if the call is an additional transferred session with inactive speech media, enter the "active" (U10) state (defined in 3GPP TS 24.008 [8]); and

2) if a call is a session with inactive speech media component, enter the "call held" hold auxiliary state (defined in 3GPP TS 24.083 [43]) for the held call.

9.2.1B SC UE procedures for PS to CS access transfer with MSC server assisted mid-call feature for speech and video session

When PS to CS access transfer occurs, with a speech and video session and another speech session using PS media in the SC UE, the SC UE applies the MSC Server assisted mid-call feature according to the procedures described in subclause 9.2.1A with the following additions:

- if the SC UE supports SCUDIF feature, and the speech and video session is active and speech session is inactive the SC UE shall transfer the active speech and video session as specified in subclause 9.2.1, and indicate the support of SCUDIF in the CC SETUP message as specified in 3GPP TS 24.008 [8], with multimedia bearer capability preferred for the current active session; and

- if the SC UE supports SCUDIF feature, and the speech and video session is inactive and speech session is active, the SC UE shall transfer the speech session as specified in subclause 9.2.1, and indicate the support of SCUDIF in the CC SETUP message as specified in 3GPP TS 24.008 [8], with speech bearer capability preferred for the current active session.

NOTE: After successful transfer of the speech and video session and another speech session from PS to CS, the UE can switch between the two sessions by holding/releasing the active session and resuming the inactive session as specified in 3GPP TS 24.008 [8], with the addition that the UE can initiate the in-call modification or Redial procedures as specified in 3GPP TS 24.008 [8] to change the shared CS bearer of the two sessions from speech to multimedia, or vice versa.

9.2.2 SC UE procedures for CS to PS access transfer

9.2.2.1 Distinction of request

The SC UE needs to distinguish the following SIP requests:

1) SIP REFER request:

   1. with the Refer-Sub header field containing "false" value;

   2. with the Supported header field containing "norefersub" value;

   3. with the Target-Dialog URI header field in the URI of the Refer-To header field;

   4. where the g.3gpp.mid-call feature-capability indicator or the g.3gpp.cs2ps-drvcc-alerting feature-capability indicator as described in annex C was included in the Feature-Caps header field of the SIP 2xx response to the SIP INVITE request due to static STI; and

   5. containing application/vnd.3gpp.mid-call+xml MIME body or the application/vnd.3gpp.state-and-event-info+xml MIME type specified in annex D.

   In the procedures below, such requests are known as "SIP REFER requests for transfer of an additional session".

2) SIP INFO request:

   A) with the Info-Package header field containing the g.3gpp.state-and-event; and

   B) containing an application/vnd.3gpp.state-and-event-info+xml XML body associated with the info package according to IETF RFC 6086 [54] and compliant to the XML schema specified in the clause D.2 with the state-info XML element containing "early" and direction XML element containing "receiver".

   In the procedures below, such requests are known as "SIP INFO requests for transfer of incoming early session".

9.2.2.2 SC UE procedure for transferring the first CS call

The SC UE may be engaged in one or more ongoing sessions before performing access transfer. By an ongoing session, it is meant a CS call for which the CS call setup procedure is complete, e.g. a CC CONNECT message has been sent or received as described in 3GPP TS 24.008 [8] or a call for which the SIP 2xx response for the initial SIP INVITE request to establish this session has been sent or received.

Additional, the SC UE may be involved in one CS session in an early phase before performing access transfer. If not already registered in the IM CN subsystem, the SC UE shall follow the procedures specified in subclause 6.2 to perform registration over the Target Access Leg before performing CS to PS access transfer.

If SC using ICS is enabled then if the original sessions are established using ICS capabilities as defined in 3GPP TS 24.292 [4], then for each session with speech media component to be transferred and starting with the one with active speech media component, the SC UE shall send a SIP INVITE request to the SCC AS in accordance with the UE procedures specified in 3GPP TS 24.229 [2]. The SC UE shall populate the SIP INVITE request as specified for PS-PS access transfer with full media transfer in subclause 10.2.1.

If the original sessions are not established using ICS capabilities and the SC UE does not support the MSC Server assisted mid-call feature as described in subclause 9.2.3, subject to the SC_non_transferrable_media node value in the Communication Continuity MO (see subclause 5.27 in 3GPP TS 24.216 [5]) the SC UE shall:

a) if more than one ongoing full-duplex session with speech media component exists, first initiate the release of all the ongoing sessions that are currently not active with the UE procedures specified in 3GPP TS 24.083 [43] and then transfer the remaining ongoing full-duplex session with active speech media component;

b) if one ongoing full-duplex session with speech media component and one session in a CS session in an early phase exists but the conditions in the subclause 9.2.5.1 for transferring a CS session in an early phase are not fulfilled, release the CS session in an early phase with the UE procedure specified in 3GPP TS 24.008 [8] and then transfer the remaining ongoing full-duplex session with active speech media component; and

c) if no ongoing full-duplex session with speech media component and one CS session in an early phase exists and the conditions in the subclause 9.2.5.1 for transferring a CS session in an early phase are fulfilled, transfer the CS session in an early phase.

When transferring the session(s) not using ICS capabilities, the SC UE shall send a SIP INVITE request to the SCC AS in accordance with the UE procedures specified in 3GPP TS 24.229 [2]. The SC UE shall:

A) populate the SIP INVITE request as follows:

1) the Request-URI set to the static STI;

2) if a GRUU was received at registration, include in the Contact header field a public GRUU or temporary GRUU as specified in 3GPP TS 24.229 [2]; and

3) the signalling elements described in subclause 6A.2.2.2; and

B) if the conditions in the subclause 9.2.5.1 for transferring a CS session in an early phase are fulfilled, additionally apply the procedures defined in subclause 9.2.5.2.

If the SC UE receives any SIP 4xx – 6xx response to the SIP INVITE request, then session transfer has not occurred and the call will continue in the CS domain.

When the SC UE receives a CS call release message, e.g. CC DISCONNECT message as specified in 3GPP TS 24.008 [8], from the network, the SC UE shall comply with network initiated call release procedures to release the CS bearer.

After completion of session transfer, if the SC UE is not using Gm, the SC UE shall locally release the resources, if any, that are associated with the source access leg.

9.2.3 SC UE procedures for CS to PS access transfer with MSC server assisted mid-call feature

When the SC UE supports the MSC Server assisted mid-call feature, the SC UE shall populate the SIP INVITE request for transferring a session not using ICS capabilities as follows in addition to the procedures described in subclause 9.2.2.2:

1. the Supported header field containing the option-tag "norefersub" specified in IETF RFC 4488 [20]; and

2. the Accept header field containing the MIME type as specified in subclause D.1.3.

NOTE 1: If the original sessions are not established using ICS capabilities as defined in 3GPP TS 24.292 [4] and the SCC AS and the SC UE support the MSC Server assisted mid-call feature, up to one active and up to one inactive CS call can be transferred.

NOTE 2: Upon receiving a SIP 2xx response for the SIP INVITE request for transferring a session not using ICS capabilities, the speech media component of the session supported by the dialog is an inactive speech media component, and if the SC UE supports 3GPP TS 24.610 [28], then the SC UE considers the session as being held.

Upon receiving a SIP REFER request within the SIP session established by the SIP INVITE request due to static STI for transferring the session not using ICS capabilities:

1. with the Refer-Sub header field containing "false" value;

2. with the Supported header field containing "norefersub" value;

3. with the Target-Dialog URI header field in the URI of the Refer-To header field;

4. where the g.3gpp.mid-call feature-capability indicator, as specified in annex C, was included in the Feature-Caps header field of the SIP 2xx response to the SIP INVITE request due to static STI; and

5. containing a MIME body of MIME type specified in the subclause D.1.3;

and if the SC UE supports the MSC Server assisted mid-call feature, then the SC UE shall:

1. handle the SIP REFER request as specified in 3GPP TS 24.229 [2], IETF RFC 3515 [13] as updated by IETF RFC 6665 [81], and IETF RFC 4488 [20] without establishing an implicit subscription; and

NOTE 3: In accordance with IETF RFC 4488 [20], the SC UE inserts the Refer-Sub header field containing the value "false" in the SIP 2xx response to the SIP REFER request to indicate that it has not created an implicit subscription.

2. send a SIP INVITE request for an additional inactive session in accordance with the procedures specified in 3GPP TS 24.229 [2] and IETF RFC 3515 [13]. The SC UE shall populate the SIP INVITE request as follows:

   A. header fields which were included as URI header fields in the URI in the Refer-To header field of the received SIP REFER request as specified in IETF RFC 3261 [19] except the "body" URI header field;

   B. include in the Contact header field a public GRUU or temporary GRUU as specified in 3GPP TS 24.229 [2] if a GRUU was received at registration;

   C. the SDP offer with:

      a. the same amount of the media descriptions as in the "body" URI header field in the URI in the Refer-To header field of the received SIP REFER request;

      b. each "m=" line having the same media type as the corresponding "m=" line in the "body" URI header field in the URI in the Refer-To header field of the received SIP REFER request;

      c. port set to zero value in each "m=" line whose corresponding "m=" line in the "body" URI header field in the URI in the Refer-To header field of the received SIP REFER request has port with zero value;

      d. media directionality as in the "body" URI header field in the URI in the Refer-To header field of the received SIP REFER request; and

NOTE 4: port can be sent to zero or non zero value for the offered "m=" line whose corresponding "m=" line in the "body" URI header field in the URI in the Refer-To header field of the received SIP REFER request has port with nonzero value.

      e) all or a subset of payload type numbers and their mapping to codecs and media parameters not conflicting with those in the "body" URI header field in the URI in the Refer-To header field of the received SIP REFER request; and

   D. the signalling elements described in subclause 6A.2.2.2.

Upon receiving a SIP 2xx response for the SIP INVITE request due to additional session transfer, then the SC UE shall proceed as specified in subclause 7.2.2.

9.2.4 SC UE procedures for selecting a session in an early dialog phase to be transferred using PS to CS dual radio access transfer procedure

When transferring a session not using ICS capabilities and when the SC UE supports PS to CS dual radio access transfer for calls in an early phase, the SC UE shall select one session to be transferred.

NOTE 1: There can be several sessions fulfilling the conditions below and it is an UE implementation issue which one to select.

Any session that fulfils one of the following conditions can be selected to be transferred:

1) if there are one or more dialogs supporting a session with active speech media component such that:

   a. all dialogs are early dialogs created by the same SIP INVITE request;

   b. a SIP 180 (Ringing) response to SIP INVITE request was received in at least one of those early dialogs;

   c. a g.3gpp.drvcc-alerting feature-capability indicator as described in annex C was included in a Feature-Caps header field provided by the SCC AS in the SIP 18x responses; and

d. the Contact header field in the SIP INVITE request sent by the SC UE towards the SCC AS included the g.3gpp.drvcc-alerting media feature tag as described in annex C;

2) if there is one dialog supporting a session with active speech media component such that:

a. the dialog is created by a SIP INVITE request;

b. a SIP 180 (Ringing) response to the SIP INVITE request was sent by the SC UE in the early dialog;

c. a g.3gpp.drvcc-alerting feature-capability indicator as described in annex C was included in a Feature-Caps header field provided by the SCC AS in the initial SIP INVITE request; and

d. the Contact header field in the SIP 180 (Ringing) response sent by the SC UE towards the SCC AS included the g.3gpp.drvcc-alerting media feature tag as described in annex C; or

3) if there are zero, one or more dialogs supporting a session with speech media component and a SIP INVITE request was sent by SC UE such that:

a. all dialogs are early dialogs created by a SIP response to the SIP INVITE request;

b. a final SIP response to the SIP INVITE request has not been received yet;

c. a SIP 180 (Ringing) response to the SIP INVITE request has not been received yet in any of the existing dialogs;

d. the SC UE included in the SIP INVITE request a Contact header field containing the g.3gpp.ps2cs-drvcc-orig-pre-alerting media feature tag as described in annex C; and

e. a SIP 18x response to the SIP INVITE request was received where the SIP 18x response contained a Feature-Caps header field with the g.3gpp.ps2cs-drvcc-orig-pre-alerting feature-capability indicator as described in annex C;

NOTE 2: UE can have zero dialogs if all the early dialogs were terminated by the SIP 199 (Early Dialog Terminated) response as described in IETF RFC 6228 [80].

9.2.5 SC UE procedures for CS to PS dual radio access transfer of calls in an early phase

9.2.5.1 Conditions for CS to PS dual radio access transfer of calls in originating pre-alerting or in the alerting phase

A CS call in the pre-alerting phase or in the alerting phase is subject for CS to PS dual radio access transfer when one of the following conditions is fulfilled:

1. the CS call is a CS session in an early phase such that:

a. the SC UE support CS to PS access transfer of a session in the alerting phase;

b. the CS call is in the "call delivered" (U4) call state as defined in 3GPP TS 24.008 [8]; and

c. no CS call in the "active" (U10) call state as defined in 3GPP TS 24.008 [8] exists;

2. if the CS call is a CS session in an early phase such that:

a. the SC UE support CS to PS access transfer of a session in the alerting phase; and

b. the CS call is in the "call received" (U7) call state as defined in 3GPP TS 24.008 [8]; and

3. if the CS call is a CS session an early phase such that:

a. the SC UE support CS to PS access transfer of a session in the alerting phase and in the originating pre-alerting phase;

b. the CS call is in the "mobile originating call proceeding" (U3) call state as defined in 3GPP TS 24.008 [8]; and

c. no CS call in the "active" (U10) call state as defined in 3GPP TS 24.008 [8] exists.

9.2.5.2 SC UE procedure for CS to PS dual radio access transfer of calls in originating pre-alerting or in the alerting phase

When the SC UE supports CS to PS dual radio access transfer for calls in alerting phase or CS to PS dual radio access transfer for originating calls in pre-alerting phase and the conditions for transferring a CS session in an early phase in subclause 9.2.5.1 are fulfilled, the SC UE shall, in addition to the procedures described in subclause 9.2.2, populate the SIP INVITE request for transferring the session not using ICS capabilities as follows:

1. include:

   a) the g.3gpp-cs2ps-drvcc-alerting media feature tag as described in annex C in the Contact header field according to IETF RFC 3840 [34];

   b) if the SC UE supports CS to PS dual radio access transfer for originating calls in pre-alerting phase, the g.3gpp.cs2ps-drvcc-orig-pre-alerting media feature tag as described in annex C in the Contact header field according to IETF RFC 3840 [34]; and

   c) the signalling elements described in subclause 6A.2.2.2.

Upon receiving the SIP INFO request for transfer of an incoming CS session in an early phase inside an early dialog created with the SIP INVITE request due to static STI, the SC UE shall:

1) send SIP 200 (OK) response to the SIP INFO request;

2) consider the SIP dialog to be:

   a) the transferred originating pre-alerting session if the application/vnd.3gpp.state-and-event-info+xml MIME body associated with the info package according to IETF RFC 6086 [54] and compliant to the XML schema specified in the clause D.2 with the state-info XML element containing "pre-alerting" and the direction XML element containing "initiator"; and

   b) the transferred incoming alerting session if the application/vnd.3gpp.state-and-event-info+xml MIME body associated with the info package according to IETF RFC 6086 [54] and compliant to the XML schema specified in the clause D.2 with the state-info XML element containing "early" and the direction XML element containing "receiver".

Upon receiving a SIP 180 (Ringing) response to the SIP INVITE due to static STI the SC UE shall consider the SIP dialog to be the transferred originating alerting session.

When the served user accepts the transferred incoming CS session in an early phase or if the user has accepted it already, the SC UE shall send a SIP INFO request accepting the session inside the early dialog created with the SIP INVITE request due to static STI according to 3GPP TS 24.229 [2] populated with:

1) an Info-Package header field with 3gpp.state-and-event info package name; and

2) application/vnd.3gpp.state-and-event-info+xml XML body associated with the info package according to IETF RFC 6086 [54] and compliant to the XML schema specified in the clause D.2 with the event XML element containing "call-accepted".

When the served user rejects the transferred incoming CS session in an early phase, the SC UE shall send a SIP CANCEL request cancelling the SIP INVITE request due to STI according to 3GPP TS 24.229 [2 populated with a Reason header field containing protocol "SIP" and the "cause" parameter indicating the selected status code and the "text" parameter indicating the selected reason phrase.

9.2.5.3 SC UE procedures for transferring an additional CS session in an early phase

When the SC UE supports CS to PS dual radio access transfer for calls in alerting phase or CS to PS dual radio access transfer for originating calls in pre-alerting phase, the SC UE shall, in addition to the procedures described in subclause 9.2.2, populate the SIP INVITE request for transferring the session not using ICS capabilities as follows:

1. the Supported header field containing the option-tag "norefersub" specified in IETF RFC 4488 [20]; and

2. the Accept header field containing the MIME type as specified in subclause D.2.4.

NOTE 1: If the original sessions are not established using ICS capabilities as defined in 3GPP TS 24.292 [4] and the SCC AS and the SC UE support transferring of an additional CS session in an early phase, one active CS call or one inactive CS call and one CS session in an early phase can be transferred.

Upon receiving a SIP REFER request within the SIP session established by the SIP INVITE request due to static STI for transferring the session not using ICS capabilities:

1. with the Refer-Sub header field containing "false" value;

2. with the Supported header field containing "norefersub" value;

3. with the Target-Dialog URI header field in the URI of the Refer-To header field;

4. where the g.3gpp.cs2ps-drvcc-alerting feature-capability indicator or the g.3gpp.cs2ps-drvcc-originating-pre-alerting feature-capability indicator as described in annex C was included in the Feature-Caps header field of the SIP 2xx response to the SIP INVITE request due to static STI; and

5. containing a MIME body of MIME type specified in the subclause D.2.4,

then the SC UE shall:

1. handle the SIP REFER request as specified in 3GPP TS 24.229 [2], IETF RFC 3515 [13] as updated by IETF RFC 6665 [81], and IETF RFC 4488 [20] without establishing an implicit subscription; and

NOTE 2: In accordance with IETF RFC 4488 [20], the SC UE inserts the Refer-Sub header field containing the value "false" in the SIP 2xx response to the SIP REFER request to indicate that it has not created an implicit subscription.

2. send a SIP INVITE request for transfer of an additional session in accordance with the procedures specified in 3GPP TS 24.229 [2] and IETF RFC 3515 [13]. The SC UE shall populate the SIP INVITE request as follows:

   A. header fields which were included as URI header fields in the URI in the Refer-To header field of the received SIP REFER request as specified in IETF RFC 3261 [19] except the "body" URI header field;

   B. if a GRUU was received at registration, include in the Contact header field a public GRUU or temporary GRUU as specified in 3GPP TS 24.229 [2];

   C) the signalling elements described in subclause 6A.2.2.2; and

   D. the SDP offer with:

      a. the same amount of the media descriptions as in the "body" URI header field in the URI in the Refer-To header field of the received SIP REFER request;

      b. each "m=" line having the same media type as the corresponding "m=" line in the "body" URI header field in the URI in the Refer-To header field of the received SIP REFER request;

      c. port set to zero value in each "m=" line whose corresponding "m=" line in the "body" URI header field in the URI in the Refer-To header field of the received SIP REFER request has port with zero value; and

      d. media directionality as in the "body" URI header field in the URI in the Refer-To header field of the received SIP REFER request.

NOTE 3: port can be set to zero or nonzero value for the offered "m=" line whose corresponding "m=" line in the "body" URI header field in the URI in the Refer-To header field of the received SIP REFER request has port with nonzero value.

Upon receiving a SIP 2xx response for the SIP INVITE request for transfer of an additional session, then the SC UE shall proceed as specified in subclause 7.2.2.

9.3 SCC AS

9.3.0 General

Void

9.3.1 Distinction of requests sent to the SCC AS

The SCC AS needs to distinguish between the following initial SIP INVITE requests to provide specific functionality relating to access transfer:

- SIP INVITE requests routed to the SCC AS containing a STI belonging to the subscribed user in the Replaces header field or Target-Dialog header field and not containing the Inter UE Transfer SCC AS URI defined in 3GPP TS 24.337 [64] in the Request-URI. In the procedures below, such requests are known as "SIP INVITE requests due to STI".

- SIP INVITE requests routed to the SCC AS containing a static STI in the Request-URI. In the procedures below, such requests are known as "SIP INVITE requests due to static STI".

- SIP INVITE requests routed to the SCC AS containing either a dynamic STN, a static STN or an IMRN (as described in 3GPP TS 24.292 [4]) in the Request-URI. In the procedures below, such requests are known as "SIP INVITE requests due to PS to CS STN".

- SIP INVITE requests routed to the SCC AS containing a STI belonging to the subscribed user in Target-Dialog header field and containing additional transferred session SCC AS URI in the Request-URI. In the procedures below such requests are known as "SIP INVITE requests transferring additional session".

NOTE 1: The media streams that need to be transferred are identified using information described in the subsequent sections.

NOTE 2: SIP INVITE requests routed to the SCC AS containing the additional transferred session SCC AS URI in the Request-URI which are used in the PS-CS access transfer with the MSC server assisted mid-call feature are handled by the PS-PS access transfer procedure as described in subclause 10.3.

Other SIP initial requests for a dialog and requests for a SIP standalone transaction can be dealt with in any manner conformant with 3GPP TS 24.229 [2].

9.3.2 SCC AS procedures for PS to CS access transfer

This subclause does not apply to reception of a SIP INVITE request due to STI with CS media and other kind of media or without CS media.

When the SCC AS receives a SIP INVITE request due to STI with CS media only on the Target Access Leg, the SCC AS shall follow the procedures specified in subclause 10.3.2 with the following exceptions:

- As the SIP INVITE request includes an active speech media component using CS bearer, then the SCC AS shall follow the procedures for SCC AS for service control over Gm in 3GPP TS 24.292 [4] to send the PSI DN to the SC UE and wait for the SC UE to set up CS bearer before sending a SIP re-INVITE request to the remote end.

- The SCC AS shall correlate the STI with the allocated PSI DN in order to identify the remote leg to be updated.

When the SCC AS receives SIP INVITE request due to PS to CS STN, the SCC AS shall:

1) associate the SIP INVITE request with an ongoing dialog supporting a session based on information associated with the received IMRN (as described in 3GPP TS 24.292 [4]) or based on information from the SIP History-Info header field or P-Asserted-Identity header field or Contact header field; or

NOTE 1: By an ongoing dialog supporting a session, it is meant a dialog for which a SIP 2xx response to the initial SIP INVITE request has been sent or received.

2) associate the SIP INVITE request with an dialog in an early phase supporting a session based on information associated with the received IMRN (as described in 3GPP TS 24.292 [4]) or based on information from the SIP History-Info header field or P-Asserted-Identity header field or Contact header field.

NOTE 2: By a dialog in an early phase supporting a session is meant a dialog for which SIP 18x responses has been sent or received but no SIP 2xx response has yet been sent or received.

NOTE 3: Multiple dialogs supporting a session associated with the same SC UE can have been anchored when the SCC AS receives a SIP INVITE request due to PS to CS STN. This can occur in the event that the UE does not succeed in releasing all dialogs supporting a session with inactive speech media component or if the UE applies the MSC Server assisted mid-call feature or if the SC UE applies the PS to CS dual radio access transfer of sessions in the pre-alerting or in the alerting phase.

The identification of the associated dialog is subject to the following conditions:

1. if only one ongoing dialog supporting a session with active speech media component exists for the user identified in the P-Asserted-Identity header field and a SIP 2xx response has been sent, then continue the session transfer with the ongoing dialog supporting a session with active speech media component;

2. if no ongoing dialogs supporting a session with active speech media component exist for the user identified in the P-Asserted-Identity header field and a SIP 2xx response has been sent and the SCC AS does not apply the MSC Server assisted mid-call feature as specified in subclause 9.3.2A, then send a SIP 480 (Temporarily Unavailable) response to reject the SIP INVITE request relating to the session transfer;

3. if more than one ongoing dialog supporting a session with active speech media component exists for the user identified in the P-Asserted-Identity header field for which SIP 2xx responses have been sent, then the SCC AS shall perform session transfer procedures for the ongoing dialog that originates from the same device that initiated the received SIP INVITE request due to PS to CS STN. If more than one such dialogs exists from the same device, the SCC AS shall proceed with the next step in this list;

NOTE 4: Whether the dialog originates from the same UE as the received SIP INVITE request due to PS to CS STN is determined based on local information and information related to the correlation MSISDN or the GRUU of the originating user as determined via registration procedures as defined in subclause 6.3.

4. if more than one ongoing dialog supporting a session exists for the user identified in the P-Asserted-Identity header field and exactly one dialog supporting a session with active speech media component exists and a SIP 2xx response has been sent for that dialog, then:

   - if the SCC AS does not apply the MSC Server assisted mid-call feature as specified in subclause 9.3.2A, then the SCC AS may release the ongoing dialogs supporting a session with inactive speech media component and continue the session transfer procedures with the ongoing dialog supporting a session with active speech media component; or

   - if the SCC AS is not able to identify one dialog for session transfer, then the SCC AS shall send a SIP 480 (Temporarily Unavailable) response to reject the SIP INVITE request relating to the session transfer;

5. if more than one ongoing dialog supporting a session with active speech media component exist for the user identified in the P-Asserted-Identity header field and a SIP 2xx response has been sent for that dialog, then:

   - if the SCC AS does not apply the MSC Server assisted mid-call feature as specified in subclause 9.3.2A, the SCC AS may release all dialogs supporting a session with speech media component of the user identified in the P-Asserted-Identity header field for which a SIP 2xx response has been sent except the one with the active speech media component that was most recently made active and continue the session transfer procedures; or

   - if the SCC AS is not able to identify one dialog for session transfer, then the SCC AS shall send a SIP 480 (Temporarily Unavailable) response to reject the SIP INVITE request relating to the session transfer;

6. if one ongoing dialog supporting a session with active speech media component where a SIP 2xx response has been sent for that dialog and if zero, one or more early dialogs created by the same SIP INVITE request exist for the user identified in the P-Asserted-Identity header field, then:

   - if the SCC AS does not apply the MSC Server assisted mid-call feature as specified in subclause 9.3.2A, continue the session transfer procedures with the ongoing dialog supporting a session with active speech media component as described in step A); or

7. if no ongoing dialogs supporting a session with active speech media component where a SIP 2xx response has been sent towards the SC UE and zero, one or more early dialogs created by the same SIP INVITE request exist for the user identified in the P-Asserted-Identity header field, then:

- if the conditions for applying the SCC AS procedures for PS to CS dual radio access transfer of calls in an early dialog phase as specified in subclause 9.3.5.1 are fulfilled, the SCC AS shall continue the session transfer procedure with the early session fulfilling the conditions in subclause 9.3.5.1; or

- if the conditions for applying the SCC AS procedures for PS to CS dual radio access transfer of calls in an early dialog phase as specified in subclause 9.3.5.1 are not fulfilled, the SCC AS shall send a SIP 480 (Temporarily Unavailable) response to reject the SIP INVITE request relating to the session transfer.

NOTE 5: The SC UE can have zero dialogs if all the early dialogs were terminated by the 199 (Early Dialog Terminated) response as described in IETF RFC 6228 [80].

If the session transfer procedure continues, the SCC AS shall:

A) if a dialog in an early phase exists and if the conditions for applying the SCC AS procedures for PS to CS dual radio access transfer for calls in an early phase as specified in subclause 9.3.5.1 are not fulfilled, the SCC AS shall release the dialog in the early phase in accordance with 3GPP TS 24.229 [2];

B) send a SIP re-INVITE request and populate the SIP re-INVITE request as follows:

1) if the dialog to transfer is an ongoing dialog supporting a session with a speech media component:

a. set the Request-URI to the URI contained in the Contact header field returned at the creation of the dialog with the remote UE;

b. set the contact header field to the contact header field provided by the served UE at the creation of the dialog with the remote UE; and

c. if the remote leg is not a precondition enabled dialog, a new SDP offer, including the media characteristics as received in the SIP INVITE request due to PS to CS STN but excluding any precondition mechanism specific SDP attributes, by following the rules of 3GPP TS 24.229 [2];

d. following the rules of 3GPP TS 24.229 [2], include a new SDP offer, including:

- the media characteristics as received in the SIP INVITE request due to PS to CS STN (including any precondition mechanism specific SDP attributes); and

- if the SIP INVITE request due to PS to CS STN is not a precondition enabled initial SIP INVITE request, indicate preconditions as met, using the segmented status type, as defined in IETF RFC 3312 [88] and IETF RFC 4032 [89], as well as the strength-tag value "mandatory" for the local segment and the strength-tag value either "optional" or as specified in RFC 3312 [88] and RFC 4032 [89] for the remote segment; and

2) if the dialog to transfer is an early dialog fulfilling the conditions in subclause 9.3.5.1 and if the SCC AS does not apply the MSC server assisted mid-call feature as specified in subclause 9.3.2A:

a. if the dialog is in the originating alerting phase, perform the actions in the subclause 9.3.5.2 and do not continue with the procedures in this subclause; and

b. if the dialog is in the originating pre-alerting phase, perform the actions in the subclause 9.3.5.3 and do not continue with the procedures in this subclause; and

Upon receiving a SIP reliable provisional response to the SIP re-INVITE request and the reliable provisional SIP response contains an SDP answer, the SCC AS shall:

A. if the SIP INVITE request due to PS to CS STN is a precondition enabled initial SIP INVITE request, forward the SIP response on the target access leg following the rules of 3GPP TS 24.229 [2];

B. if the SIP INVITE request due to PS to CS STN is not a precondition enabled initial SIP INVITE request, forward the SIP 183 (Session Progress) response on the target access leg following the rules of 3GPP TS 24.229 [2] but remove from the SDP answer precondition mechanism specific SDP attributes;

C. if the SCC AS supports the PS to CS dual radio access transfer of calls in alerting phase, include the g.3gpp.drvcc-alerting feature-capability indicator as described in annex C in the Feature-Caps header field according to IETF RFC 6809 [60]; and

D. include the signalling elements described in subclause 6A.4.3A.

Upon receiving a PRACK to the SIP provisional response to the SIP INVITE request due to PS to CS STN, forward the SIP PRACK request towards the remote UE following the rules of 3GPP TS 24.229 [2].

Upon receiving a SIP 200 (OK) SIP response to the SIP PRACK request from the remote UE the SCC AS shall, forward the SIP response on the target access leg following the rules of 3GPP TS 24.229 [2].

Upon receiveing a SIP UPDATE request on the target access leg, the SCC AS shall forward the request towards the remote UE following the rules of 3GPP TS 24.229 [2].

Upon receipt of a SIP 200 (OK) response to the SIP UPDATE request, the SCC AS shall forward the SIP response on the target access leg following the rules of 3GPP TS 24.229 [2].

Upon receiving the SIP 2xx response to the SIP re-INVITE request the SCC AS shall:

1. if a SIP 200 (OK) response to the SIP INVITE due to PS to CS STN has not been sent yet,

   A. send the SIP 200 (OK) response to the SIP INVITE request due to PS to CS STN on the target access leg. If the SIP 2xx response to the SIP re-INVITE request includes an SDP answer:

      a. if the SIP INVITE request due to PS to CS STN is a precondition enabled initial SIP INVITE request, use relevant media parameter of the SDP answer in the received SIP response, by following the rules of 3GPP TS 24.229 [2]; and

      b. if the SIP INVITE request due to PS to CS STN is not a precondition enabled initial SIP INVITE request, remove from the SDP answer precondition mechanism specific SDP attributes;

   B. if the SCC AS supports the PS to CS dual radio access transfer of calls in alerting phase, include the g.3gpp.drvcc-alerting feature-capability indicator as described in annex C in the Feature-Caps header field according to IETF RFC 6809 [60];

   C. include the signalling elements described in subclause 6A.4.3A;

      Upon reception of the SIP ACK request to the the SIP 200 (OK) response to the SIP INVITE request due to PS to CS STN, the SCC AS shall send the SIP ACK request to the SIP 2xx response to the SIP re-INVITE request;

2. if the SIP 200 (OK) response to the SIP INVITE request due to PS to CS STN was already sent, the SCC AS shall send the SIP ACK request to the SIP 2xx response to the SIP re-INVITE request;

3. remove non-transferred audio media components and release the source access leg for the session supporting an active speech media component as specified in subclause 9.3.6; and

4. if the SCC AS supports dual radio access transfer for calls in an early phase and if the conditions specified in subclause 9.3.5.1 are fulfilled for transfer of a session in the pre-alerting phase or in the originating alerting phase, the SCC AS shall follow the procedures in the subclause 9.3.5.4 and do not continue with the procedures in this subclause.

If the SCC AS supports dual radio access transfer for calls in the alerting phase and if the conditions specified in subclause 9.3.5.1 are fulfilled for a dialog in the terminating alerting phase, the SCC AS shall follow the procedures in the subclause 9.3.5.5.

9.3.2A SCC AS procedures for PS to CS access transfer with MSC server assisted mid-call feature

The SCC AS supporting PS to CS access transfer with MSC server assisted mid-call feature shall:

I) if:

   1. the Contact header field of the SIP INVITE request due to PS to CS STN includes the g.3gpp.mid-call media feature tag as specified in annex C; and

   2. one of the following is true:

A. at least one confirmed dialog supporting a session with active speech media component exists for the user identified in the P-Asserted-Identity header field and the following is true for the confirmed dialog supporting a session with active speech media component which has been most recently made active:

- the Contact header field provided by the SC UE at the establishment of the dialog supporting a session with active speech media component which has been most recently made active included the g.3gpp.mid-call media feature tag as described in annex C; and

- the Feature-Caps header field sent by SCC AS towards the SC UE at the establishment of the dialog included the g.3gpp.mid-call feature-capability indicator as described in annex C; or

B. no confirmed dialog supporting a session with active speech media component exists for the user identified in the P-Asserted-Identity header field, one or more confirmed dialogs supporting a session with inactive speech media component exists for the user and the following is true for the confirmed dialog supporting a session with inactive speech media component which has been most recently made inactive:

- the Contact header field provided by the SC UE at the establishment of the dialog included the g.3gpp.mid-call media feature tag as described in annex C; and

- the Feature-Caps header field sent by SCC AS towards the SC UE at the establishment of the dialog included the g.3gpp.mid-call feature-capability indicator as described in annex C;

apply the MSC Server assisted mid-call feature; and

II) if:

1. the Contact header field of the SIP INVITE request due to PS to CS STN does not include the g.3gpp.mid-call media feature tag as specified in annex C; and

2. if a confirmed dialog supporting a session with inactive speech media component exists for the user identified in the P-Asserted-Identity header field and the following is true for the confirmed dialog supporting a session with inactive speech media component:

A. the Contact header field provided by the SC UE at the establishment of the dialog included the g.3gpp.mid-call media feature tag as described in annex C; and

B. the Feature-Caps header field sent by SCC AS towards the SC UE at the establishment of the dialog included the g.3gpp.mid-call feature-capability indicator as described in annex C,

not apply the MSC Server assisted mid-call feature and send a SIP BYE request to the SC UE on the source access leg and towards the remote UE in accordance with 3GPP TS 24.229 [2]; and continue the procedure in subclause 9.3.2 to identify an associated dialog for the SIP INVITE request due to PS to CS STN.

When the SCC AS applies the MSC Server assisted mid-call feature, in addition to the procedures described in subclause 9.3.2, and before determining that the SCC AS is not able to identify one dialog for session transfer, the SCC AS may:

1. if more than one confirmed dialog supporting a session exists for the user identified in the P-Asserted-Identity header field, and exactly one confirmed dialog supporting a session with active speech media component exists and there is at least one remaining confirmed dialog supporting a session with inactive speech media component, then:

- release all dialogs supporting a session with active speech media component for which SIP 2xx responses have not been sent for these dialogs; and

- release all confirmed dialogs supporting a session with inactive speech media component except the one with the speech media component which became inactive most recently and continue the session transfer procedures with the confirmed dialog supporting a session with active speech media component;

2. if more than one confirmed dialog supporting a session with active speech media component exists for the user identified in the P-Asserted-Identity header field, release all confirmed dialogs supporting a session with speech media component except two with the speech media component which became active most recently and continue the session transfer procedures with the confirmed dialog supporting a session with the speech media component which became active most recently;

3. if no confirmed dialog supporting a session with active speech media component exists for the user identified in the P-Asserted-Identity header field, one or more confirmed dialogs supporting a session with inactive speech media component exists for the user then the SCC AS may release all confirmed dialogs supporting a session with speech media component except the one with the speech media component which became inactive most recently and continue the session transfer procedures with the confirmed dialog supporting a session with inactive speech media component; and

4 if one confirmed dialog supporting a session with active or inactive speech media component exists and one or more early dialogs created by the same SIP INVITE request continue with the session with active or inactive speech media component.

When the SCC AS applies the MSC Server assisted mid-call feature, in addition to the procedures described in subclause 9.3.2, the SCC AS shall include the g.3gpp.mid-call feature-capability indicator, as described in annex C, in the Feature-Caps header field of the SIP 2xx response to the SIP INVITE request due to PS to CS STN according to IETF RFC 6809 [60].

When the SCC AS applies the MSC Server assisted mid-call feature and a confirmed dialog supporting a session with inactive speech media component was associated with the SIP INVITE request due to PS to CS STN, in addition to the procedures described in subclause 9.3.2, the SCC AS shall set the directionality of the audio media in the SDP offer as used in the session with remote UE.

If:

- the SCC AS applies the MSC Server assisted mid-call feature;
- the session associated with the SIP INVITE request due to PS to CS STN is related to a subscription as described in subclause 7.3.3; and
- a SIP 2xx response was received to the last SIP NOTIFY request with conference information sent to the UE within the related subscription;

then the SCC AS shall send a SIP INFO request towards the MSC Server as specified in 3GPP TS 24.229 [2] and IETF RFC 6086 [54] in the dialog created by the SIP INVITE request due to PS to CS STN. The SCC AS shall populate the SIP INFO request as follows:

1. include the Info-Package header field as specified in IETF RFC 6086 [54] with g.3gpp.mid-call package name; and
2. include application/vnd.3gpp.mid-call+xml XML body associated with the info package according to IETF RFC 6086 [54] and containing the participants extracted as specified in the subclause 9.1A of the subscription related to the session associated with the SIP INVITE request due to PS to CS STN as described in subclause 7.3.3.

If the SCC AS applies the MSC Server assisted mid-call feature,

1. two confirmed dialogs supporting a session with speech media component exist for the user identified in the P-Asserted-Identity header field;
2. one confirmed dialog supporting a session with speech media component and one early dialog exist for the user identified in the P-Asserted-Identity header field fulfilling the conditions in subclause 9.3.5.1 for transfer of a session in the originating alerting phase and if the SCC AS included the g.3gpp.drvcc-alerting feature-capability indicator as described in annex C in a Feature-Caps header field in a SIP 180 (Ringing) response; and
3. one confirmed dialog supporting a session with speech media component and one early dialog exist for the user identified in the P-Asserted-Identity header field fulfilling the conditions in subclause 9.3.5.1 for transfer of a session in the originating pre-alerting phase and if the SCC AS included the g.3gpp.ps2cs-drvcc-orig-pre-alerting feature-capability indicator as described in annex C in a Feature-Caps header field in a SIP 18x responses,

then the SCC AS shall send a SIP REFER request towards the MSC Server in accordance with the procedures specified in 3GPP TS 24.229 [2], IETF RFC 4488 [20] and IETF RFC 3515 [13] as updated by IETF RFC 6665 [81] and IETF RFC 7647 [90] in the dialog created by the SIP INVITE request due to PS to CS STN.

The SCC AS shall populate the SIP REFER request as follows:

1. the Refer-Sub header field with value "false" as specified in IETF RFC 4488 [20];

2. the Supported header field with value "norefersub" as specified in IETF RFC 4488 [20];

3. the Refer-To header field containing the information related to the additional transferred session, i.e. session with speech media component other than the session associated with the SIP INVITE request due to PS to CS STN, i.e. set to the additional transferred session SCC AS URI and the following URI header fields:

   A. the Target-Dialog URI header field populated as specified in IETF RFC 4538 [11], containing the dialog identifier of the session with the SC UE;

   B. the Require URI header field populated with the option tag value "tdialog";

   C. the To URI header field populated as specified in IETF RFC 3261 [19], containing the P-Asserted-Identity provided by the remote UE during the session establishment;

   D. the From URI header field populated as specified in IETF RFC 3261 [19], containing the public user identity of the SC UE provided during the session establishment;

   E. the Content-Type header field with "application/sdp";

   F. the hname "body" URI header field populated with SDP describing the media streams as negotiated in the session with the remote UE and:

      a. if directionality used by SC UE is "sendrecv" or "sendonly", with the "sendonly" directionality; and

      b. if directionality used by SC UE is "recvonly" or "inactive", with the "inactive" directionality; and

   G. optionally the P-Asserted-Identity URI header field containing value of the P-Asserted-Identity header field of the received SIP INVITE request;

4. the Content-Type header field with the value set to MIME type as specified in the subclause D.1.3; and

5. a XML body compliant to the XML schema specified in the subclause D.1.2. If

   A. the session associated with the SIP INVITE request due to PS to CS STN is not related to any subscription as described in subclause 7.3.3;

   B. the additional transferred session is related to a subscription as described in subclause 7.3.3; and

   C. a SIP 2xx response was received to the last SIP NOTIFY request with conference information sent to the UE within the related subscription;

   then SCC AS shall populate the XML body with the participants extracted as specified in the subclause 9.1A of the subscription related to the additional transferred session as specified in subclause 7.3.3.

When the SCC AS receives a SIP INVITE request transferring additional session for dual radio, the SCC AS shall:

- associate the SIP INVITE request transferring additional session with a previously established SIP dialog i.e. identify the Source Access Leg. The SIP dialog on the Source Access Leg is identified by matching the dialog ID present in the Target-Dialog header field (see IETF RFC 4538 [11]) of the SIP INVITE request transferring additional session with the previously established SIP dialog. By a previously established SIP dialog, it is meant a dialog for which SIP 18x responses or a SIP 2xx response to the initial SIP INVITE request has been sent or received;

- if the SCC AS is unable to associate the SIP INVITE with a unique previously established SIP dialog, send a SIP 480 (Temporarily Unavailable) response to reject the SIP INVITE request relating to the access transfer and not processes the remaining steps;

- if the number of media lines in the Target Access Leg is less than the number of media lines in the Source Access Leg or the media type for the corresponding media lines is not the same as in the original session, send a SIP 4xx response to reject the SIP INVITE request relating to the access transfer and not process the remaining steps;

- if the additional session is a confirmed session, send a SIP re-INVITE request towards the remote UE using the existing established dialog. The SCC AS shall populate the SIP re-INVITE request , following the rules specified in 3GPP TS 24.229 [2] as follows:

A) include a new SDP offer with:

a) if the remote leg is not a precondition enabled dialog, the media characteristics as received in the SIP INVITE request transferring additional session but excluding any precondition mechanism specific SDP attributes for media streams whose port is not set to zero;

b) for the media streams in the SIP INVITE request transferring additional session whose port is set to zero, include the corresponding media characteristics of those streams from the Source Access Leg; and

c) if the remote leg is a precondition enabled dialog, include a new SDP offer, including:

- the media characteristics as received in the SIP INVITE request transferring additional session for PS to CS for dual radio (including any precondition mechanism specific SDP attributes);

- if the SIP INVITE request due to PS to CS STN is not a precondition enabled initial SIP INVITE request, indicate preconditions as met, using the segmented status type, as defined in IETF RFC 3312 [88] and IETF RFC 4032 [89], as well as the strength-tag value "mandatory" for the local segment and the strength-tag value either "optional" or as specified in RFC 3312 [88] and RFC 4032 [89] for the remote segment;

NOTE: If the MSC server is using the precondition mechanism, the local preconditions will always be indicated as met according to subclause 9.5.

- if the additional session is a originating session in pre-alerting phase, perform the actions in the subclause 9.3.5.4; and

- if the additional session is a originating session in the alerting phase perform the actions in the subclause 9.3.5.4.

Upon receiving the SIP 2xx response to the SIP re-INVITE request triggered by the SIP INVITE request transferring additional session for dual radio the SCC AS shall:

1. send the SIP 200 (OK) response to the SIP INVITE request transferring additional session for dual radio on the target access leg.

   a. if the SIP INVITE request transferring additional session for dual radio is a precondition enabled initial SIP INVITE request, use relevant media parameter of the SDP answer in the received SIP response, by following the rules of 3GPP TS 24.229 [2]; and

   b. if the SIP INVITE request transferring additional session for dual radio is not a precondition enabled initial SIP INVITE request, remove from the SDP answer precondition mechanism specific SDP attributes;

2. if the SCC AS supports the PS to CS dual radio access transfer of calls in alerting phase, include the g.3gpp.drvcc-alerting feature-capability indicator as described in annex C in the Feature-Caps header field according to IETF RFC 6809 [60];

3. include the signalling elements described in subclause 6A.4.3A.

If the SCC AS supports dual radio access transfer for calls in the alerting phase and if the conditions specified in subclause 9.3.5.1 are fulfilled for a dialog in the terminating alerting phase, the SCC AS shall follow the procedures in the subclause 9.3.5.5.

The SCC AS shall remove non-transferred audio media components and release source access legs as specified in subclause 9.3.6.

9.3.3 SCC AS procedures for CS to PS access transfer

When the SCC AS receives a SIP INVITE request due to STI on the Target Access Leg offering PS media only, the SCC AS shall follow the procedures specified in subclause 10.3.2.

When the SCC AS receives a SIP INVITE request due to static STI, the SCC AS shall:

1) associate the SIP INVITE request with an ongoing dialog supporting a session; or

NOTE 1: By an ongoing dialog supporting a session, it is meant a dialog for which a SIP 2xx response to the initial SIP INVITE request has been sent or received.

2) if the SCC AS supports CS to PS dual radio access transfer for calls in alerting phase or CS to PS dual radio access transfer for originating calls in pre-alerting phase, associate the SIP INVITE request with a dialog in an early phase.

NOTE 2: By a dialog in an early phase, it is meant a dialog for which a SIP 2xx response to the initial SIP INVITE request has not been sent or received.

Multiple dialogs supporting a session associated with the same SC UE may have been anchored when the SCC AS receives a SIP INVITE request due to static STI.

NOTE 3: This multiple dialogs supporting a session associated with the same SC UE can occur in the event that the UE does not succeed in releasing all dialogs supporting a session with inactive speech media component or if the UE supports the MSC Server assisted mid-call feature or if the SC UE supports the CS to PS dual radio access transfer for calls in alerting phase or CS to PS dual radio access transfer for originating calls in pre-alerting phase, in which case.

The identification of the associated dialog is subject to the following conditions:

1. if only one ongoing dialog supporting a session with active speech media component exists for the user identified in the P-Asserted-Identity header field and a SIP 2xx response has been sent, then continue the session transfer procedures with the ongoing dialog supporting a session with active speech media component;

2. if no ongoing dialogs supporting a session with active speech media component exists for the user identified in the P-Asserted-Identity header field and a SIP 2xx response has been sent and the SCC AS does not apply the MSC Server assisted mid-call feature as specified in subclause 9.3.4 and the condition for transferring dialog(s) in an early dialog phase in subclause 9.3.7.1 are not fulfilled, then send a SIP 480 (Temporarily Unavailable) response to reject the SIP INVITE request relating to the session transfer;

3. if more than one ongoing dialog supporting a session exists for the user identified in the P-Asserted-Identity header field and exactly one ongoing dialog supporting a session with active speech media component and a SIP 2xx response has been sent for that dialog, then:

   A. if the remaining dialogs support a session with inactive speech media component and the SCC AS does not apply the MSC Server assisted mid-call feature as specified in subclause 9.3.4, then the SCC AS may release the dialogs supporting a session with inactive speech media component and continue the session transfer procedures with the dialog supporting a session with active speech media component;

4. if one ongoing dialog with active speech media component and one or more dialogs in an early dialog phase exists and the condition for transferring dialog(s) in an early dialog phase in subclause 9.3.7.1 are fulfilled, continue the session transfer with the ongoing dialog with active speech media component;

5. if one ongoing dialog with active speech media component and one or more dialogs in an early dialog phase exists but the condition for transferring dialog(s) in an early dialog phase in subclause 9.3.7.1 are not fulfilled, then the SCC AS may release the dialogs in an early dialog phase and continue the session transfer with the ongoing dialog with active speech media component;

6. if no ongoing dialogs supporting a session with speech media component and one or more early dialog in an early phase exists and the condition for transferring dialog(s) in an early dialog phase in subclause 9.3.7.1 are fulfilled, proceed with the session transfer as specified in subclause 9.3.7; and

7. if the SCC AS is not able to identify one dialog for session transfer, then the SCC AS shall send a SIP 480 (Temporarily Unavailable) response to reject the SIP INVITE request relating to the session transfer and do not continue with the access transfer.

If the session transfer procedures continues and if the dialog to be transferred is an ongoing dialog supporting a session with speech media component, the SCC AS shall send a SIP re-INVITE request towards the remote UE using the existing established dialog. The SCC AS shall populate the SIP re-INVITE request as follows:

1) set the Request-URI to the URI contained in the Contact header field returned at the creation of the dialog with the remote UE; and

2) a new SDP offer, including the media characteristics as received in the SIP INVITE request due to the static STI, by following the rules of 3GPP TS 24.229 [2].

Upon receiving the SIP 2xx response to the SIP re-INVITE request the SCC AS shall send the SIP 200 (OK) response to the SIP INVITE request due to static STI on the target access leg populated as follows:

1) the relevant media parameter of the SDP answer in the received response, by following the rules of 3GPP TS 24.229 [2];

2) if the SCC AS supports CS to PS dual radio access transfer for calls in alerting phase according to operator policy,

   a) the g.3gpp.cs2ps-drvcc-alerting feature-capability indicator as described in annex C in a Feature-Caps header field according to IETF RFC 6809 [60]; and

   b) if the SCC AS supports PS to CS dual radio access transfer for originating calls in pre-alerting phase the g.3gpp.cs2ps-drvcc-orig-pre-alerting feature-capability indicator as described in annex C in the Feature-Caps header field according to IETF RFC 6809 [60]; and

3) the signalling elements described in subclause 6A.4.3.

Upon receiving the SIP ACK request originated from the SC UE, the SCC AS shall:

1) release the source access leg as specified in subclause 9.3.6.; and

2) if the SCC AS supports the CS to PS dual radio access transfer for a call in an early phase and the condition for transferring dialog(s) in an early dialog phase in subclause 9.3.7.1 are fulfilled, proceed with transferring of the dialog(s) in the early phase as specified in the subclause 9.3.7.4.

If, subsequent to initiating the SIP re-INVITE request to the remote UE, and prior to the SIP ACK request originated from the UE being received from the IM CN subsystem for the source access leg, the SCC AS decides (for any reason) to reject the session transfer request back to the UE (e.g. by sending a SIP 4xx response), the SCC AS shall release the target access leg and maintain the source access leg.

9.3.4 SCC AS procedures for CS to PS access transfer with MSC server assisted mid-call feature

The SCC AS shall apply the MSC Server assisted mid-call feature if:

1. the Contact header field of the SIP INVITE request due to static STI includes the g.3gpp.mid-call media feature tag as described in annex C; and

2. the SCC AS supports the MSC Server assisted mid-call feature according to operator policy.

When the SCC AS applies the MSC Server assisted mid-call feature, in addition to the procedures described in subclause 9.3.3, and before determining that the SCC AS is not able to identify one dialog for session transfer, SCC AS may:

1. if more than one dialog exists for the user identified in the P-Asserted-Identity header field, and exactly one dialog supporting an ongoing session with active speech media component exists, and a SIP 2xx response has been sent for that dialog and there is at least one remaining dialog supporting a session with inactive speech media component, release all dialogs supporting a session with inactive speech media component except the one with the speech media component which became inactive most recently and continue the session transfer procedures with the dialog supporting a session with active speech media component;

2. if no dialog supporting an ongoing session with active speech media component exists for the user identified in the P-Asserted-Identity header field, one or more dialogs supporting a session with inactive speech media component exists for the user and a SIP 2xx response has been sent for these dialogs then the SCC AS may release all dialogs supporting a session with speech media component except the one with the speech media component which became inactive most recently and continue the session transfer procedures with the dialog supporting a session with inactive speech media component; and

3. if more than one dialog exists for the user identified in the P-Asserted-Identity header field, and exactly one dialog supporting an ongoing session with inactive speech media component exists, and a SIP 2xx response has been sent for that dialog and the condition for transferring dialog(s) in an early dialog phase in subclause 9.3.7.1 are fulfilled continue the session transfer procedures with the dialog supporting a session with inactive speech media component;

The SCC AS shall include the signalling elements described in subclause 6A.4.3 in the SIP 1xx response and SIP 2xx response to the SIP INVITE request due to static STI.

When the SCC AS applies the MSC Server assisted mid-call feature and a dialog supporting a session with inactive speech media component was associated with the SIP INVITE request due to static STI, in addition to the procedures described in subclause 9.3.3, the SCC AS shall set the directionality of the speech media component in the SDP offer as used in the session with remote UE.

If the SCC AS applies the MSC Server assisted mid-call feature, two SIP dialogs supporting a session with a speech media component exist for the user identified in the P-Asserted-Identity header field and a SIP 2xx response has been sent for those dialogs then the SCC AS shall send a SIP REFER request towards the SC UE in accordance with the procedures specified in 3GPP TS 24.229 [2], IETF RFC 4488 [20] and IETF RFC 3515 [13] as updated by IETF RFC 6665 [81] and IETF RFC 7647 [90] in the dialog created by the SIP INVITE request due to static STI. The SCC AS shall populate the SIP REFER request as follows:

1. the Refer-Sub header field with value "false" as specified in IETF RFC 4488 [20];

2. the Supported header field with value "norefersub" as specified in IETF RFC 4488 [20];

3. the Refer-To header field containing the information related to the session with an audio media other than the session associated with the SIP INVITE request due to static STI, i.e. set to the additional transferred session SCC AS URI and the following URI header fields:

   A. the Target-Dialog URI header field populated as specified in IETF RFC 4538 [11], containing the dialog identifier of the session with the MSC Server;

   B. the Require URI header field populated with the option tag value "tdialog";

   C. if the remote UE did not request privacy then the To URI header field populated as specified in IETF RFC 3261 [19], containing the P-Asserted-Identity provided by the remote UE during the session establishment;

   D. the From URI header field populated as specified in IETF RFC 3261 [19], containing the public user identity of the SC UE provided during the session establishment;

   E. the Content-Type URI header field with "application/sdp"; and

   F. the hname "body" URI header field populated with SDP describing the media streams as negotiated in the session with the remote UE and with directionality as used by the MSC Server;

4. the Content-Type header field with the value set to MIME type specified in the subclause D.1.3; and

5. a XML body compliant to the XML schema specified in the subclause D.1.2.

When the SCC AS receives a SIP INVITE request transferring additional session for the dialog supporting the ongoing session with a speech media component, the SCC AS shall:

- associate the SIP INVITE request transferring additional session with a previously established SIP dialog i.e. identify the Source Access Leg. The SIP dialog on the Source Access Leg is identified by matching the dialog ID present in the Target-Dialog header field (see IETF RFC 4538 [11]) of the SIP INVITE request transferring additional session with the previously established SIP dialog. By a previously established SIP dialog, it is meant a dialog for which a SIP 2xx response to the initial SIP INVITE request has been sent or received;

- if the SCC AS is unable to associate the SIP INVITE with a unique previously established SIP dialog, send a SIP 480 (Temporarily Unavailable) response to reject the SIP INVITE request relating to the access transfer and not processes the remaining steps;

- if the number of media lines in the Target Access Leg is less than the number of media lines in the Source Access Leg or the media type for the corresponding media lines is not the same as in the original session, send a SIP 4xx response to reject the SIP INVITE request relating to the access transfer and not process the remaining steps; and

- send a SIP re-INVITE request towards the remote UE using the existing established dialog. The SCC AS shall populate the SIP re-INVITE request with a new SDP offer, following the rules specified in 3GPP TS 24.229 [2], containing the following media information:

a) the media characteristics as received in the SIP INVITE request transferring additional session for media streams whose port is not set to zero; and

b) for the media streams in the SIP INVITE request transferring additional session whose port is set to zero, include the corresponding media characteristics of those streams from the Source Access Leg.

When the SCC AS receives the SIP 200 (OK) response to the SIP re-INVITE request, the SCC AS shall send a SIP 200 (OK) response to the SIP INVITE request transferring additional session for the dialog supporting the ongoing session with a speech media component populated as follows:

1) the relevant media parameter of the SDP answer in the received response, by following the rules of 3GPP TS 24.229 [2]; and

2) the signalling elements described in subclause 6A.4.3.

The SCC AS shall release the source access leg as specified in subclause 9.3.6.

9.3.5 SCC AS procedures for PS to CS dual radio access transfer of calls in an early dialog phase

9.3.5.1 Conditions for selecting a sessions in an early dialog phase

An early session is subject for PS to CS dual radio access transfer when one of the following conditions is fulfilled:

1. if there are one or more dialogs in an early phase such that:

   a. all dialogs are dialogs in an early phase are created by the same SIP INVITE request;

   b. a SIP 180 (Ringing) response to SIP INVITE request was received from remote UEs in at least one of those early dialogs;

   c. a g.3gpp.drvcc-alerting feature-capability indicator as described in annex C was included in a Feature-Caps header field provided by the SCC AS in the SIP 180 (Ringing) response;

   d. the Contact header field provided by the SC UE towards the SCC AS in the initial SIP INVITE request included the g.3gpp.drvcc-alerting media feature tag field as described in annex C; and

   e. the SIP INVITE due to PS to CS STN contains the g.3gpp.drvcc-alerting as described in annex C;

2. if there are one or more dialogs in the early phase such that:

   a. all dialogs are dialogs in an early phase are created by the same SIP INVITE request;

   b. a SIP 180 (Ringing) response to SIP INVITE request was received from remote UE in at least one of those early dialogs;

   c. a g.3gpp.drvcc-alerting feature-capability indicator as described in annex C was included in a Feature-Caps header field provided by the SCC AS in the SIP 180 (Ringing) response;

   d. the Contact header field provided by the SC UE towards the SCC AS in the initial SIP INVITE request included the g.3gpp.drvcc-alerting media feature tag field as described in annex C; and

   e. no ongoing dialog supporting a session with active or inactive speech media component where a SIP 2xx response has been sent towards the SC UE exists;

NOTE 1: Transfer of one single dialog in the alerting phase does not require any MSC server support.

3. if there are one or more dialogs in an early phase such that:

   a. all dialogs are in an early phase are created by the same SIP INVITE request;

   b. a SIP 180 (Ringing) response to the SIP INVITE request has not been received from remote UEs yet;

   c. the SCC AS included a g.3gpp.ps2cs-drvcc-orig-pre-alerting feature-capability indicator as described in annex C in a Feature-Caps header field in SIP 18x responses;

d. the Contact header field in the initial SIP INVITE request sent by the SC UE towards the SCC AS included a g.3gpp.ps2cs-drvcc-orig-pre-alerting media feature tag as described in annex C; and

e. the SIP INVITE due to PS to CS STN contains the g.3gpp.drvcc-alerting as described in annex C;

4. if there are one or more dialogs in an early phase such that:

   a. all dialogs are early phase are created by the same SIP INVITE request;

   b. a SIP 180 (Ringing) response to the SIP INVITE request has not been received from remote UEs yet;

   c. the SCC AS included a g.3gpp.ps2cs-drvcc-orig-pre-alerting feature-capability indicator as described in annex C in a Feature-Caps header field in SIP 18x responses;

   d. the Contact header field in the initial SIP INVITE request sent by the SC UE towards the SCC AS included a g.3gpp.ps2cs-drvcc-orig-pre-alerting media feature tag as described in annex C; and

   e. no ongoing dialog supporting a session with active or inactive speech media component where a SIP 2xx response has been sent towards the SC UE exists;

NOTE 2: Transfer of one single dialog in the originating pre-alerting phase does not require any MSC server support.

5. if there is one dialog in an dialog phase such that:

   a. a SIP 180 (Ringing) response to the SIP INVITE request has been received from the SC UE;

   b. the SCC AS included a g.3gpp.drvcc-alerting feature-capability indicator as described in annex C in a Feature-Caps header field in the SIP INVITE request; and

   c. the Contact header field in the SIP 180 (Ringing) response sent by the SC UE towards the SCC AS included a g.3gpp.drvcc-alerting media feature tag as described in annex C.

NOTE 3: Transfer of a dialog in the alerting phase on the terminating side does not require any MSC server support.

9.3.5.2 SCC AS procedures for PS to CS dual radio access transfer of a originating session in the alerting phase

When the SCC AS receives a SIP INVITE request due to PS to CS STN and if there are one or more dialogs in an early dialog phase supporting a session with active speech media component such that:

1) all dialogs are early dialogs created by the same SIP INVITE request;

2) a SIP 180 (Ringing) response to SIP INVITE request was received in at least one of those early dialogs;

3) a g.3gpp.drvcc-alerting feature-capability indicator as described in annex C was included in a Feature-Caps header field by the SCC AS in the SIP 180 (Ringing) response; and

4) the Contact header field in the initial SIP INVITE request sent by the SC UE towards the SCC AS included the g.3gpp.drvcc-alerting media feature tag as described in annex C,

then the SCC AS shall for each early dialog send an SIP UPDATE request towards the remote UE and populate each SIP UPDATE request as follows:

1) set the Request-URI set to the URI contained in the Contact header field returned at the creation of the dialog with the remote UE;

2) the Contact header field set to the Contact header field provided by the served UE at the creation of the dialog with the remote UE; and

3) a new SDP offer, including:

   a) if the remote leg is not a precondition enabled dialog, the media characteristics as received in the SIP INVITE request due to PS to CS STN but excluding any precondition mechanism specific SDP attributes, by following the rules of 3GPP TS 24.229 [2];

b) if the remote leg is a precondition enabled dialog, include a new SDP offer including:

- the media characteristics as received in the SIP INVITE request due to PS to CS STN (including any precondition mechanism specific SDP attributes); and

- if the SIP INVITE request due to PS to CS STN is not a precondition enabled initial SIP INVITE request, indicate preconditions as met, using the segmented status type, as defined in IETF RFC 3312 [88] and IETF RFC 4032 [89], as well as the strength-tag value "mandatory" for the local segment and the strength-tag value either "optional" or as specified in RFC 3312 [88] and RFC 4032 [89] for the remote segment.

For each SIP 200 (OK) response to the SIP UPDATE request (triggered by the SIP INVITE request due to PS to CS STN) from a remote UE the SCC AS shall:

1) if one of the following is true:

   A) if the remote leg is not a precondition enabled dialog;

   B) if the remote leg is a precondition enabled dialog, the SIP INVITE request due to PS to CS STN is a precondition enabled initial SIP INVITE request and both local and remote preconditions are met:

   send a SIP provisional response to the SIP INVITE request due to PS to CS STN following the rules of 3GPP TS 24.229 [2] with the response code corresponding to the actual dialog state populated with:

- an SDP answer based on the SDP answer received from the remote UE; and

- the last received P-Early-Media header field, including the SIP 2xx response to the SIP UPDATE request, if a P-Early-Media has been received from the remote UE.

2) if the remote leg is a precondition enabled dialog but all preconditions are not met, send a SIP 183 (Session Progress) response following the rules of 3GPP TS 24.229 [2] populated with:

   - if SIP INVITE request due to PS to CS STN is a precondition enabled initial SIP INVITE, an SDP answer based on the SDP answer received from the remote UE; and

   - if SIP INVITE request due to PS to CS STN is not a precondition enabled initial SIP INVITE, an SDP answer based on the SDP answer received from the remote UE but excluding the precondition mechanism specific SDP attributes; and

   - the last received P-Early-Media header field, including the SIP 2xx response to the SIP UPDATE request, if a P-Early-Media has been received from the remote UE.

     Upon receipt of an SIP PRACK request on the target access leg send a SIP 200 (OK) response to the SIP PRACK request on the target access leg.

Upon receipt of a provisional SIP response from the remote UE forward the SIP provisional response on the target access leg following the rules of 3GPP TS 24.229 [2]. If the SIP INVITE request due to PS to CS STN is not a precondition enabled initial SIP INVITE remove any precondition mechanism specific SDP attributes from the SDP offer. Upon receipt of the SIP PRACK request on the target access leg, forward the SIP PRACK request towards the remote UE following the rules of 3GPP TS 24.229 [2].

Upon receipt of a SIP UPDATE request on the target access leg the SCC AS shall forward the SIP request to the remote UE following the rules of 3GPP TS 24.229 [2]. Upon receipt of a SIP 200 (OK) response to this SIP UPDATE request from the remote UE:

1) forward the SIP response on the target access leg following the rules of 3GPP TS 24.229 [2]; and

2) if both local and remote preconditions are met and if the remote leg is in the alerting phase, send a SIP 180 (Ringing) response following the rules of 3GPP TS 24.229 [2] with the last received P-Early-Media header field, including the SIP 2xx response to the SIP UPDATE request, if a P-Early-Media header field has been received from the remote UE.

Upon receipt of a SIP UPDATE request from the remote UE, the SCC AS shall forward the SIP request on the target access leg following the rules of 3GPP TS 24.229 [2] but if the SIP INVITE request due to PS to CS STN is not a precondition enabled initial SIP INVITE remove from the SDP answer the precondition mechanism specific SDP

attributes. Upon receipt of a SIP 200 (OK) response to this SIP UPDATE request on the target access leg, the SCC AS shall:

1) forward the SIP response towards the remote UE following the rules of 3GPP TS 24.229 [2]; and

2) if both local and remote preconditions are met and if the remote leg is in the alerting phase, send a SIP 180 (Ringing) response on the target access leg with the last received P-Early-Media header field, including the SIP 2xx response to the SIP UPDATE request, if a P-Early-Media header field has been received from the remote UE.

The SCC AS shall remove non-transferred audio media components and release the source access leg as specified in subclause 9.3.6.

9.3.5.3 SCC AS procedures for PS to CS dual radio, access transfer of a originating session in the pre-alerting phase

When the SCC AS receives a SIP INVITE request due to PS to CS STN and if there are zero, one or more dialogs in an early dialog phase supporting a session with active speech media component such that:

1) all dialogs are early dialogs created by the same SIP INVITE request;

2) a SIP 180 (Ringing) response to the SIP INVITE request has not been received yet in any of the existing dialogs;

3) the SCC AS included a g.3gpp.ps2cs-drvcc-orig-pre-alerting feature-capability indicator as described in annex C in a Feature-Caps header field of SIP 18x responses; and

4) the Contact header field in the initial SIP INVITE request sent by the SC UE towards the SCC AS included a g.3gpp.ps2cs-drvcc-orig-pre-alerting media feature tag as described in annex C,

then the SCC AS shall for each early dialog send a SIP UPDATE request towards the remote UE.

NOTE 1: The SCC AS can have zero dialogs if all the early dialogs were terminated by the 199 (Early Dialog Terminated) response as described in IETF RFC 6228 [80].

Each SIP UPDATE request shall be populated as follows:

1) the Request-URI set to the URI contained in the Contact header field returned at the creation of the dialog with the remote UE;

2) the Contact header field set to the Contact header field provided by the served UE at the creation of the dialog with the remote UE; and

3) an new SDP offer, including

   a) if the remote leg is not a precondition enabled dialog, the media characteristics as received in the SIP INVITE request due to PS to CS STN but excluding any precondition mechanism specific SDP attributes, by following the rules of 3GPP TS 24.229 [2];

   b) if the remote leg is a precondition enabled dialog, include a new SDP offer including:

      - the media characteristics as received in the SIP INVITE request due to PS to CS STN (including any precondition mechanism specific SDP attributes); and

      - if the SIP INVITE request due to PS to CS STN is not a precondition enabled initial SIP INVITE request, indicate preconditions as met, using the segmented status type, as defined in IETF RFC 3312 [88] and IETF RFC 4032 [89], as well as the strength-tag value "mandatory" for the local segment and the strength-tag value either "optional" or as specified in RFC 3312 [88] and RFC 4032 [89] for the remote segment.

For each SIP 200 (OK) response to the SIP UPDATE request (triggered by the SIP INVITE request due to CS to PS STN) from a remote UE the SCC AS shall

1) if one of the following is true:

   A) if the remote leg is not a precondition enabled dialog;

B) if the remote leg is a precondition enabled dialog, the SIP INVITE request due to PS to CS STN is a precondition enabled initial SIP INVITE request and both local and remote preconditions are met:

send a SIP provisional response to the SIP INVITE request due to PS to CS STN with the response code corresponding to the actual dialog state populated with:

- a SDP answer based on the SDP answer received from the remote UE; and
- the last received P-Early-Media header field, including the SIP 2xx response to the SIP UPDATE request, if a P-Early-Media has been received from the remote UE.

2) if the remote leg is a precondition enabled dialog, the SIP INVITE request due to PS to CS STN is a precondition enabled initial SIP INVITE request but all preconditions are not met, send a SIP 183 (Session Progress) response populate with:

- if SIP INVITE request due to PS to CS STN is a precondition enabled initial SIP INVITE, an SDP answer based on the SDP answer received from the remote UE; and
- if SIP INVITE request due to PS to CS STN is not a precondition enabled initial SIP INVITE, an SDP answer based on the SDP answer received from the remote UE but excluding the precondition mechanism specific SDP attributes; and
- the last received P-Early-Media header field, including the SIP 2xx response to the SIP UPDATE request, if a P-Early-Media has been received from the remote UE.

Upon receipt of an SIP PRACK request on the target access leg send a SIP 200 (OK) response to the SIP PRACK request on the target access leg.

Upon receipt of a provisional SIP response from the remote UE forward the SIP provisional response on the target access leg following the rules of 3GPP TS 24.229 [2]. If the SIP INVITE request due to PS to CS STN is not a precondition enabled initial SIP INVITE remove any precondition mechanism specific SDP attributes from the SDP offer. Upon receipt of the SIP PRACK request on the target access leg, forward the SIP PRACK request towards the remote UE following the rules of 3GPP TS 24.229 [2].

Upon receipt of a SIP UPDATE request on the target access leg the SCC AS shall forward the SIP request to the remote UE. Upon receipt of a SIP 200 (OK) response to this SIP UPDATE request from the remote UE, the SCC AS shall forward the SIP response on the target access leg.

Upon receipt of a SIP UPDATE request from the remote UE, the SCC AS shall forward the SIP request on the target access leg. If the SIP INVITE request due to PS to CS STN is not a precondition enabled initial SIP INVITE remove any precondition mechanism specific SDP attributes from the SDP offer. Upon receipt of a SIP 200 (OK) response to this SIP UPDATE request on the target access leg, the SCC AS shall forward the SIP response towards the remote UE.

The SCC AS shall remove non-transferred audio media components and release the source access leg as specified in subclause 9.3.6.

9.3.5.4 SCC AS procedures for PS to CS dual radio access transfer of an additional session in an early dialog phase

In order to transfer an additional session on the originating side that can be in pre-alerting phase or in an alerting phase, the SCC AS shall send a SIP REFER request according to 3GPP TS 24.229 [2], IETF RFC 4488 [20] and IETF RFC 3515 [13] as updated by IETF RFC 6665 [81] and IETF RFC 7647 [90] and IETF RFC 4488 [20] in the dialog created by the SIP INVITE request due to PS to CS STN. The SCC AS shall populate the SIP REFER request as follows:

1. the Refer-Sub header field with value "false" as specified in IETF RFC 4488 [20];
2. the Require header field with value "norefersub" as specified in IETF RFC 4488 [20];
3. the Refer-To header field containing the additional transferred session SCC AS URI for PS to CS dual radio, where the URI also includes the following header fields containing the information related to the additional transferred session:

   A. the Target-Dialog header field populated as specified in IETF RFC 4538 [11], containing the dialog identifier of an dialog in the early phase supporting session of the SC UE;

B. the Require header field populated with the option tag value "tdialog";

C. the To header field populated as specified in IETF RFC 3261 [19], containing the value of the P-Asserted-Identity provided by the remote UE during the session establishment;

D. the From header field populated as specified in IETF RFC 3261 [19], containing the value of the P-Asserted-Identity provided by the SC UE during the session establishment;

E. the Content-Type header field with "application/sdp";

F. the URI header field with the hname "body" populated with SDP describing the media streams as negotiated in the session with the remote UE; and

G. optionally the P-Asserted-Identity URI header field containing value of the P-Asserted-Identity header field of the received SIP INVITE request; and

4. application/vnd.3gpp.state-and-event-info+xml MIME body populated as follows:

A) if a SIP 180 (Ringing) response to the SIP INVITE request has already been received in any of the early dialogs associated with the originating early session not accepted yet, with the state-info XML element containing "early" and the direction XML element containing "initiator"; and

B) if a SIP 180 (Ringing) response to the SIP INVITE request has not been received yet in any of the early dialogs associated with the originating early session not accepted yet, with the state-info XML element containing "pre-alerting" and the direction XML element containing "initiator".

When the SCC AS receives the SIP INVITE request transferring additional session for PS to CS for dual radio, the SCC AS shall:

- associate the SIP INVITE request transferring additional session for PS to CS for dual radio with an SIP dialog in early dialog phase i.e. identify the source access leg;

NOTE 1: The SIP dialog on the source access leg is identified by matching the dialog ID present in Target-Dialog header field (see IETF RFC 4538 [11]) of the SIP INVITE with a dialog in early state.

NOTE 2: By a SIP dialog in early dialog phase, it is meant an early SIP dialog which has been created by a provisional response to the initial SIP INVITE request, but for which the SIP 2xx response has not yet been sent or received;

- if the SCC AS is unable to associate the SIP INVITE with a unique dialog in early dialog phase, send a SIP 480 (Temporarily Unavailable) response to reject the SIP INVITE request relating to the access transfer and not processes the remaining steps;

- if the number of media lines in the target access leg is less than the number of media lines in the source Access leg or the media type for the corresponding media lines is not the same as in the original session, send a SIP 4xx response to reject the SIP INVITE request relating to the access transfer and not process the remaining steps; and

- send a SIP UPDATE request(s) towards the remote UE(s) using the existing early dialog(s) which were created by the same SIP INVITE request as the source access leg. The SCC AS shall populate the SIP UPDATE request(s) following the rules specified in 3GPP TS 24.229 [2], as follows:

A) include a new SDP offer with:

a) the media characteristics as received in the SIP INVITE request transferring additional session for PS to CS for dual radio received on the target access leg for media streams whose port is not set to zero modified as follows:

i) if the remote leg is not a precondition enabled dialog and the SIP INVITE request transferring additional session for PS to CS for dual radio is a precondition enabled initial SIP INVITE request, exclude preconditions specific attributes in the new SDP offer; and

ii) if the remote leg is a precondition enabled dialog and the SIP INVITE request transferring additional session for PS to CS for dual radio is not a precondition enabled initial SIP INVITE request, indicate preconditions as met, using the segmented status type, as defined in IETF RFC 3312 [88] and IETF RFC 4032 [89], as well as the strength-tag value "mandatory" for the local segment and the

strength-tag value either "optional" or as specified in RFC 3312 [88] and RFC 4032 [89] for the remote segment;

NOTE 3: If the MSC server is using the precondition mechanism, the local preconditions will always be indicated as met according to subclause 9.7.

b) for the media streams in the SIP INVITE request transferring additional session for PS to CS for dual radio whose port is set to zero, include the corresponding media characteristics of those streams from the source access leg.

When receiving SIP 2xx response(s) to the SIP UPDATE request(s) triggered by the SIP INVITE request transferring additional session for dual radio, the SCC AS shall send a SIP 18x response with the status code corresponding to the latest SIP 18x response following the rules of 3GPP TS 24.229 [2] received from remote leg in the dialog to the SIP INVITE request transferring additional session for PS to CS for dual radio containing:

1) if

a) the SIP INVITE request transferring additional session for PS to CS for dual radio is a precondition enabled initial SIP INVITE request, an SDP answer with the relevant media parameter of the SDP answer in the received SIP 2xx response;

b) the SIP INVITE request transferring additional session for PS to CS for dual radio is not a precondition enabled initial SIP INVITE request, an SDP answer with the relevant media parameter of the SDP answer in the received SIP 2xx response but excluding the precondition mechanism specific SDP attributes;

If a SIP PRACK request is received on the target access leg, send a SIP 200 (OK) response to the SIP PRACK request on the target access leg; and

2) the last received P-Early-Media header field, including the SIP 2xx response to the SIP UPDATE request, if a P-Early-Media has been received from the remote UE.

The SCC AS shall remove non-transferred audio media components and release the source access leg as specified in subclause 9.3.6.

9.3.5.5 SCC AS procedures for PS to CS dual radio access transfer of a terminating session in the alerting phase

When the SCC AS receives a SIP 488 (Not Acceptable Here) response to the SIP INVITE request creating the session in the terminating alerting phase without an SDP MIME body and if the SCC AS supports PS to CS dual radio access transfer for calls in alerting phase then the SCC AS shall:

1) if a SIP 180 (Ringing) response to the SIP INVITE request has been received from the SC UE;

2) if the SCC AS included a g.3gpp.ps2cs-drvcc-orig-pre-alerting feature-capability indicator as described in annex C in the SIP INVITE request; and

3) if the Contact header field in the SIP 180 (Ringing) response request sent by the SC UE towards the SCC AS included a g.3gpp.drvcc-alerting media feature tag as described in annex C,

terminate the call over CS as follows:

1) perform the actions according to the subclause 10.4.7 in 3GPP TS 24.292 [4] with the following clarifications:

a) the URI in the Request-URI shall be set to C-MSISDN; and

b) the P-Asserted-Identity header field set to:

- if the SIP 180 (Ringing) response contained the g.3gpp.dynamic-stn media feature tag as described in annex C in the Contact header field, the dynamic STN; and

- if the SIP 180 (Ringing) response does not contain the g.3gpp.dynamic-stn media feature tag as described in annex C in the Contact header field, the static STN.

When the SCC AS receives a SIP 1xx response with an SDP answer the SCC AS shall:

a) send a SIP PRACK request towards the CS domain; and

b) send an SIP UPDATE request to the remote UE populated as follows:

- the Request-URI set to the URI contained in the Contact header field returned at the creation of the dialog with the remote UE;
- the Contact header field set to the Contact header field provided by the served UE at the creation of the dialog with the remote UE; and
- anew SDP offer, including the media characteristics as received in the SIP 1xx response with the SDP answer, by following the rules of 3GPP TS 24.229 [2].

Upon receipt of the SIP 200 (OK) response to the SIP UPDATE request, the SC AS shall remove non-transferred audio media components and release the source access leg as specified in subclause 9.3.6.

9.3.6 Removal of non-transferred audio media components and release of source access legs

When the transfer of a session is successfully completed, then the SCC AS shall release the source legs as follows:

If:

1) the source access leg is an ongoing session containing only an active or inactive media component or a session in an early dialog phase on the terminating side, send a SIP BYE request on the source access leg in accordance with 3GPP TS 24.229 [2];

2) the session is dialog in an early dialog phase on the originating side send a SIP 480 (Temporary Unavailable) response on the source access leg in accordance with 3GPP TS 24.229 [2]; and

NOTE: In case of PS to CS dual radio access transfer of a session in an early phase, the SC UE will receive the SIP request or response only if the SC UE is using Gm after the PS-CS access transfer is completed.

3) the source access leg contains media components other than speech media component, the SCC AS should send a SIP re-INVITE request to update the source access leg in accordance with 3GPP TS 24.229 [2].

9.3.7 SCC AS procedures for CS to PS dual radio access transfer for calls in an early phase

9.3.7.1 Conditions for transferring dialog(s) in the originating pre-alerting or the alerting dialog phase

Upon receiving a SIP INVITE request due to static STI and the SCC AS support CS to PS dual radio access transfer for calls in alerting phase or CS to PS dual radio access transfer for originating calls in pre-alerting phase and one of the following conditions are fulfilled:

1) if there are one or more dialog in an early dialog phase such that:

   a) all dialogs are early dialogs created by the same SIP INVITE request;

   b) a SIP 180 (Ringing) response to SIP INVITE request was received from remote UEs in at least one of those early dialogs;

   c) the g.3gpp.cs2ps-drvcc-alerting feature-capability indicator as described annex C in a Feature-Caps header field was included in the SIP INVITE due to static STI; and

   d) the SCC AS supports CS to PS dual radio access transfer for calls in alerting phase;

2) if there are one or more dialog in an early dialog phase such that:

   a) all dialogs are early dialogs created by the same SIP INVITE request;

   b) a SIP 180 (Ringing) response to the SIP INVITE request has not been received from remote UEs yet;

   c) a g.3gpp.cs2to-drvcc-orig-pre-alerting media feature tag as described in annex C in the Contact header field was included in the SIP INVITE request due to static STI; and

d) the SCC AS supports CS to PS dual radio access transfer for originating calls in pre-alerting phase; and

3) if there is one dialog in an early dialog phase such that:

a) a SIP 180 (Ringing) response to the SIP INVITE request has been received from the SC UE;

b) the g.3gpp.cs2ps-drvcc-alerting feature-capability indicator as described annex C in a Feature-Caps header field was included in the SIP INVITE due to static STI; and

c) the SCC AS supports CS to PS dual radio access transfer for calls in alerting phase.

then the SCC AS shall regard the session subject for PS to CS dual radio access transfer.

9.3.7.2 SCC AS procedures for CS to PS dual radio access transfer for originating calls in pre-alerting phase or in alerting phase on the originating side

When the SCC AS receives a SIP INVITE request due to static STI and if the SCC AS supports CS to PS dual radio access transfer for calls in alerting phase or CS to PS dual radio access transfer for originating calls in pre-alerting phase and:

1) if there are one or more dialog in an early dialog phase such that:

a) all dialogs are early dialogs created by the same SIP INVITE request;

b) a SIP 180 (Ringing) response to SIP INVITE request was received from remote UEs in at least one of those early dialogs;

c) the g.3gpp.cs2ps-drvcc-alerting feature-capability indicator in a Feature-Caps header field as described in annex C was included in the SIP INVITE due to static STI; and

d) if the SCC AS supports CS to PS dual radio access transfer for calls in alerting phase; or

2) if there are zero, one or more dialog in an early dialog phase such that:

a) all dialogs are early dialogs created by the same SIP INVITE request;

b) a SIP 180 (Ringing) response to the SIP INVITE request has not been received yet in any of the existing dialogs;

c) a g.3gpp.drvcc-orig-pre-alerting media feature tag as described in annex C in the Contact header field was included in the SIP INVITE due to static STI; and

d) the SCC AS supports CS to PS dual radio access transfer for originating calls in pre-alerting phase,

NOTE: The SCC AS can have zero dialogs if all the early dialogs were terminated by the SIP 199 (Early Dialog Terminated) response as described in RFC 6228 [80].

then the SCC AS shall

A) for each existing early dialog towards remote UEs send an SIP UPDATE request and populate as follows:

a) the Contact header field set to the Contact header field provided on the source leg at the creation of the dialog with the remote UE; and

b) a new SDP offer, including the media characteristics as received in the SIP INVITE request due to static STI, by following the rules of 3GPP TS 24.229 [2].For each SIP 200 (OK) response to the SIP UPDATE request on an existing dialog from a remote UE the SCC AS shall create a new early dialog by sending a SIP provisional response to the SIP INVITE request due to static STI with the response code corresponding to the actual dialog state populated with:

I) an SDP answer based on the SDP answer received from the remote UE;

II) the last received P-Early-Media header field, including the SIP 2xx response to the SIP UPDATE request, if a P-Early-Media has been received from the remote UE;

III) if the SCC AS supports CS to PS dual radio access transfer for calls in alerting phase according to operator policy,

- the g.3gpp.cs2ps-drvcc-alerting feature-capability indicator as described in annex C in a Feature-Caps header field according to IETF 6809 [60]; and
- if the SCC AS supports PS to CS dual radio access transfer for originating calls in pre-alerting phase, the g.3gpp.cs2ps-drvcc-orig-pre-alerting feature-capability indicator as described in annex C in a Feature-Caps header field according to IETF 6809 [60]; and

IV)the signalling elements described in subclause 6A.4.3 in the SIP 1xx response.

Upon receiving the SIP PRACK request from the target access leg, if the early session is an originating early session in the pre-alerting state, the SCC AS shall send a SIP INFO request towards the SC UE as specified in 3GPP TS 24.229 [2] and IETF RFC 6086 [54] in the dialog created by the SIP INVITE request due to static STI populated as follows:

1) include the Info-Package header field as specified in IETF RFC 6086 [54] with 3gpp.state-and-event info package name; and

2) include an application/vnd.3gpp.state-and-event-info+xml MIME body associated with the info package according to IETF RFC 6086 [54] and compliant to the XML schema specified in the clause D.2 with the state-info XML element containing "pre-alerting" and the direction XML element containing "initiator".

If the SCC AS supports the PS to CS dual radio access transfer for originating calls in pre-alerting phase and if a SIP 1xx response or SIP 2xx response establishing a new dialog is received on the remote leg of the additional transferred session where the SIP response is to the SIP INVITE request from the served user, the SCC AS shall:

1) if the SIP 1xx response is received:

   a) send a SIP PRACK request on the remote leg as specified in 3GPP TS 24.229 [2] with the SDP offer received in the SIP INVITE request transferring additional session for PS to CS dual radio as specified in 3GPP TS 24.229 [2];

   b) upon receiving the SIP 200 (OK) response to the SIP PRACK request, send the SIP 1xx response to the SIP INVITE request transferring additional session for PS to CS dual radio as specified in 3GPP TS 24.229 [2] populated as follows:

      i) include the SDP answer received in the SIP 200 (OK) response to the SIP PRACK request as specified in 3GPP TS 24.229 [2]; and

      ii) if the SIP INVITE request transferring additional session for PS to CS dual radio contains a P-Early-Media header field with the "supported" parameter and if the SCC AS has received a P-Early-Media header field in a SIP message in the dialog of the SIP PRACK request, include a P-Early-Media header field containing the value of the last P-Early-Media header field received in a SIP message in the dialog of the SIP PRACK request; and

   c) include the signalling elements described in subclause 6A.4.3 in the SIP 1xx response; and

2) if the SIP 2xx response is received:

   a) send a SIP ACK request on the remote leg as specified in 3GPP TS 24.229 [2];

   b) send a SIP UPDATE request on the remote leg as specified in 3GPP TS 24.229 [2] populated with the SDP offer received in the SIP INVITE request transferring additional session for PS to CS dual radio as specified in 3GPP TS 24.229 [2]; and

   c) upon receiving the SIP 200 (OK) response to the SIP UPDATE request, send a SIP 200 (OK) response to the SIP INVITE request transferring additional session for PS to CS dual radio as specified in 3GPP TS 24.229 [2] populated with:

      i) the SDP answer received in the SIP 200 (OK) response to the SIP UPDATE request as specified in 3GPP TS 24.229 [2]; and

      ii) the signalling elements described in subclause 6A.4.3 in the SIP 1xx response.

The SCC AS shall release the source access leg as specified in subclause 9.3.6.

9.3.7.3 SCC AS procedures for CS to PS dual radio access transfer for a call in the alerting phase on the terminating side

When the SCC AS receives a SIP INVITE request due to static STI and if SCC AS supports CS to PS dual radio access transfer for calls in alerting phase and:

1) if there is one dialog in an early dialog phase such that:

   a) a SIP 180 (Ringing) response to the SIP INVITE request has been received from the SC UE; and

   b) a g.3gpp.drvcc-alerting media feature tag as described in annex C in the Contact header field was included in the SIP INVITE request due to static STI,

   then the SCC AS shall send a SIP UPDATE request towards the remote UE populated as follows:

1) include the Contact header field set to the Contact header field provided on the source leg at the creation of the dialog with the remote UE; and

2) include a new SDP offer, including the media characteristics as received in the SIP INVITE request due to static STI, by following the rules of 3GPP TS 24.229 [2].

Upon receipt of the SIP 200 (OK) response to the SIP UPDATE request from the remote UE the SCC AS shall send a SIP 180 (Ringing) response to the SIP INVITE request due to static STI with populated with:

1) an SDP answer based on the SDP answer received from the remote UE;

2) the last received P-Early-Media header field, including the SIP 2xx response to the SIP UPDATE request, if a P-Early-Media has been received from the remote UE; and

3) the g.3gpp.cs2ps-drvcc-alerting feature-capability indicator in a Feature-Caps header field according to annex C; and

4) if the SCC AS supports PS to CS dual radio access transfer for originating calls in pre-alerting phase, the g.3gpp.cs2ps-drvcc-orig-pre-alerting feature-capability indicator according to annex C in the Feature-Caps header field; and

5) the signalling elements described in subclause 6A.4.3.

Upon receiving the SIP PRACK request from the target access leg, the SCC AS shall send a SIP INFO request towards the SC UE as specified in 3GPP TS 24.229 [2] and IETF RFC 6086 [54] in the dialog created by the SIP INVITE request due to static STI populated as follows:

1) include the Info-Package header field as specified in IETF RFC 6086 [54] with 3gpp.state-and-event info package name; and

2) include an application/vnd.3gpp.state-and-event-info+xml MIME body associated with the info package according to IETF RFC 6086 [54] and compliant to the XML schema specified in the clause D.2 with the state-info XML element containing "early" and the direction XML element containing "receiver".

Upon receiving the SIP INFO request which includes an Info-Package header field containing 3gpp.state-and-event info package name and an application/vnd.3gpp.state-and-event-info+xml MIME body associated with the info package according to IETF RFC 6086 [54] and compliant to the XML schema specified in the clause D.2 from the SC UE with the event XML element containing "call-accepted", the SCC AS shall send as specified in 3GPP TS 24.229 [2]:

1) a SIP 200 (OK) response to the SIP INVITE request received earlier from the remote UE indicating that the called party has answered the call; and

2) a SIP 200 (OK) response to the SIP INVITE request due to static STI towards the SC UE to indicate the successful access transfer.

The SCC AS shall release the source access leg as specified in subclause 9.3.6.

9.3.7.4 SCC AS procedures for PS to CS dual radio access transfer of an additional session in an early dialog phase

In order to transfer of an additional session that can be in originating pre-alerting phase or in an alerting phase, the SCC AS supporting CS to PS dual radio access transfer for calls in an early phase, shall send a SIP REFER request according to 3GPP TS 24.229 [2], IETF RFC 4488 [20] and IETF RFC 3515 [13] as updated by IETF RFC 6665 [81] and IETF RFC 7647 [90] in the dialog created by the SIP INVITE request due to static STI populated as follows:

1) include a Refer-Sub header field with value "false" as specified in IETF RFC 4488 [20];

2) include a Require header field with value "norefersub" as specified in IETF RFC 4488 [20];

3) include a Refer-To header field containing the additional transferred session SCC AS URI for PS to CS dual radio, where the URI also includes the following URI header fields containing the information related to the additional transferred session:

   a) if an early dialog supporting the additional transferred session exists, the Target-Dialog header field populated as specified in IETF RFC 4538 [11], containing the dialog identifier of the dialog in the early phase;

   b) if the SCC AS supports the PS to CS dual radio access transfer for originating calls in pre-alerting phase, if no early dialog supporting the additional transferred session exists, there is a SIP INVITE request from the served user for which a final SIP response has not been received yet and if an early dialog supporting the additional transferred session existed and was terminated, the Target-Dialog header field populated as specified in IETF RFC 4538 [11], containing the dialog identifier on the source access leg of the early dialog supporting the additional transferred session which existed and was terminated;

NOTE 1: Early dialog can be terminated by SIP 199 (Early Dialog Terminated) response.

   c) the Require header field populated with the option tag value "tdialog";

   d) if an early dialog supporting the additional transferred session exists, the To header field populated as specified in IETF RFC 3261 [19], containing the value of the P-Asserted-Identity provided by the remote UE during the session establishment;

   e) the From header field populated as specified in IETF RFC 3261 [19], containing the value of the P-Asserted-Identity provided by the SC UE during the session establishment;

   f) the Content-Type header field with "application/sdp";

   g) if an early dialog supporting the additional transferred session exists, the URI header field with hname "body" populated with SDP describing the media streams as negotiated in the session with the remote UE; and

   h) if the SCC AS supports the PS to CS dual radio access transfer for originating calls in pre-alerting phase, no early dialog supporting the additional transferred session exists, there is a SIP INVITE request from the served user for which a final SIP response has not been received yet, the URI header field with the hname "body" populated with the SDP offer received in the SIP INVITE request from the served user; and

4) application/vnd.3gpp.state-and-event-info+xml MIME with:

   a) if a SIP 180 (Ringing) response to the SIP INVITE request has already been received from the remote UE in any of the early dialogs associated with the originating early session not accepted yet, the state-info XML element containing "early" and the direction XML element containing "initiator";

   b) if a SIP 180 (Ringing) response to the SIP INVITE request has not been received yet from the remote UE in any of the early dialogs associated with the originating early session not accepted yet and the additional transferred session was originated by the SC UE, the state-info XML element containing "pre-alerting" and the direction XML element containing "initiator"; and

   c) if a SIP 180 (Ringing) response to the INVITE request has already been received on the source access leg, the state-info XML element containing "early" and the direction XML element containing "receiver"

When the SCC AS receives the SIP INVITE request transferring additional session for CS to PS for dual radio, the SCC AS shall:

1) if the Target-Dialog of the SIP INVITE request transferring additional session for PS to CS dual radio identifies an existing early dialog, associate the SIP INVITE request transferring additional session for PS to CS for dual radio with the SIP early dialog i.e. identify the source access leg;

NOTE 2: The SIP dialog on the source access leg is identified by matching the dialog ID present in Target-Dialog header field (see IETF RFC 4538 [11]) of the SIP INVITE with a dialog in early state.

NOTE 3: By a SIP dialog in early dialog phase, it is meant an early SIP dialog which has been created by a provisional response to the initial SIP INVITE request, but for which the SIP 2xx response has not yet been sent or received.

2) if the SCC AS supports the PS to CS dual radio access transfer for originating calls in pre-alerting phase, if the Target-Dialog of the SIP INVITE request transferring additional session for PS to CS dual radio identifies an early dialog which has already been terminated, associate the SIP INVITE request transferring additional session for PS to CS dual radio with the early dialog i.e. identify the source access leg;

3) if the SCC AS is unable to associate the SIP INVITE request with a unique dialog in early dialog phase, send a SIP 480 (Temporarily Unavailable) response to reject the SIP INVITE request relating to the access transfer and not processes the remaining steps;

4) if the number of media lines in the target access leg is less than the number of media lines in the source Access leg or the media type for the corresponding media lines is not the same as in the original session, send a SIP 4xx response to reject the SIP INVITE request relating to the access transfer and not process the remaining steps; and

5) if an early dialog exists on the remote leg of the additional transferred session, send a SIP UPDATE request(s) towards the remote UE(s) using the existing early dialog(s) which were created by the same SIP INVITE request as the source access leg populated with a new SDP offer, following the rules specified in 3GPP TS 24.229 [2], containing the following media information:

   a) the media characteristics as received in the SIP INVITE request transferring additional session for CS to PS for dual radio received on the target access leg for media streams whose port is not set to zero; and

   b) for the media streams in the SIP INVITE request transferring additional session for CS to PS for dual radio whose port is set to zero, include the corresponding media characteristics of those streams from the source access leg.

If an early dialog exists on the remote leg then when receiving SIP 2xx response(s) to the SIP UPDATE request(s), the SCC AS shall create a new dialog by sending a SIP 18x response with the status code corresponding to the latest SIP 18x response received from remote leg in the dialog to the SIP INVITE request transferring additional session for CS to PS for dual radio containing:

1) an SDP answer with the relevant media parameter of the SDP answer in the received SIP 2xx response;

2) if the SIP INVITE request transferring additional session for PS to CS dual radio contains a P-Early-Media header field with the "supported" parameter, the last received P-Early-Media header field, including the SIP 2xx response to the SIP UPDATE request, if a P-Early-Media has been received from the remote UE; and

3) the signalling elements described in subclause 6A.4.3.

If the SCC AS supports the PS to CS dual radio access transfer for originating calls in pre-alerting phase and if a SIP 1xx response or SIP 2xx response establishing a new dialog is received on the remote leg of the additional transferred session where the SIP response is to the SIP INVITE request from the served user, the SCC AS shall:

1) if the SIP 1xx response is received:

   a) send SIP PRACK request on the remote leg as specified in 3GPP TS 24.229 [2] populated with the SDP offer received in the SIP INVITE request transferring additional session for PS to CS dual;

   b) upon receiving the SIP 200 (OK) response to the SIP PRACK request, send the SIP 1xx response to the SIP INVITE request transferring additional session for PS to CS dual radio as specified in 3GPP TS 24.229 [2] populated with:

      i) include the SDP answer received in the SIP 200 (OK) response to the SIP PRACK request as specified in 3GPP TS 24.229 [2];

ii) if the SIP INVITE request transferring additional session for PS to CS dual radio contains a P-Early-Media header field with the "supported" parameter and if the SCC AS has received a P-Early-Media header field in a SIP message in the dialog of the SIP PRACK request, include a P-Early-Media header field containing the value of the last P-Early-Media header field received in a SIP message in the dialog of the SIP PRACK request; and

iii) the signalling elements described in subclause 6A.4.3; and

2) if a SIP 2xx response is received:

a) send a SIP ACK request on the remote leg as specified in 3GPP TS 24.229 [2];

b) send a SIP UPDATE request on the remote leg as specified in 3GPP TS 24.229 [2 populated with the SDP offer received in the SIP INVITE request transferring additional session for PS to CS dual radio as specified in 3GPP TS 24.229 [2]; and

c) upon receiving the SIP 200 (OK) response to the SIP UPDATE request, send a SIP 200 (OK) response to the SIP INVITE request transferring additional session for PS to CS dual radio as specified in 3GPP TS 24.229 [2] populated with:

i) the SDP answer received in the SIP 200 (OK) response to the SIP UPDATE request as specified in 3GPP TS 24.229 [2]; and

ii) the signalling elements described in subclause 6A.4.3.

The SCC AS shall release the source access leg as specified in subclause 9.3.6.

Upon receiving the SIP INFO request which includes an Info-Package header field containing 3gpp.state-and-event info package name and an application/vnd.3gpp.state-and-event-info+xml MIME body associated with the info package according to IETF RFC 6086 [54] and compliant to the XML schema specified in the clause D.2 from the SC UE with the event XML element containing "call-accepted", the SCC AS shall send as specified in 3GPP TS 24.229 [2]:

1) a SIP 200 (OK) response to the SIP INVITE request received earlier from the remote UE indicating that the called party has answered the call; and

2) a SIP 200 (OK) response to the SIP INVITE request due to static STI towards the SC UE to indicate the successful access transfer.

9.4 MSC server enhanced for ICS

If the MSC server enhanced for ICS has registered for the user, the MSC server shall apply the procedures as specified in 3GPP TS 29.292 [18].

If the MSC server enhanced for ICS supports the MSC server assisted mid-call feature, the MSC server shall apply the procedures specified in subclause 9.5 and subclause 9.6.

If the MSC server enhanced for ICS supports PS to CS dual radio access transfer for calls in alerting phase, the MSC server shall apply the procedures specified in subclause 9.7.

9.4.1 Void

9.4.1A Void

9.5 PS to CS session continuity with MSC server assisted mid-call feature

This subclause describes the procedures required by an MSC server in order to support the MSC server assisted mid call feature.

The MSC server shall populate the SIP INVITE request due to STN as follows:

1. the Supported header field containing the option-tag "norefersub" specified in IETF RFC 4488 [20];

2. the Accept header field containing the MIME type as specified in subclause D.1.3;

3. include the g.3gpp.mid-call media feature tag as described in annex C in the Contact header field according to IETF RFC 3840 [53]; and

4. the Recv-Info header field containing the g.3gpp.mid-call package name.

NOTE 1: Since the MSC server is not able to distinguish the dual radio access transfer from the regular session set up, the information elements above are added to every SIP INVITE request sent by the MSC server.

Upon receiving a CC CONNECT ACK message related to a CC CONNECT message sent as result of receiving a SIP 2xx response to SIP INVITE request due to STN, if inactive speech media component is negotiated by the SDP answer of the SIP 2xx response to the SIP INVITE request due to STN, after handling the CC CONNECT ACK according to 3GPP TS 24.008 [8], the MSC server shall re-assign the hold auxiliary state (defined in 3GPP TS 24.083 [43]) to "call held" for the transaction identifier and TI flag value as received in the CC SETUP message.

NOTE 2: After handling the CC CONNECT ACK according to 3GPP TS 29.292 [18] and 3GPP TS 24.008 [8], the dialog created by the SIP INVITE request due to STN is associated with a CS call in the "active" (N10) state (defined in 3GPP TS 24.008 [8]), the "idle" hold auxiliary state (defined in 3GPP TS 24.083 [43]) and the "idle" multi party auxiliary state (defined in 3GPP TS 24.084 [47]) with transaction identifier and TI flag value as received in the CC SETUP message.

Upon receiving a SIP INFO request:

- with the Info-Package header field containing the g.3gpp.mid-call package name;

- with the application/vnd.3gpp.mid-call+xml MIME body associated with the info package according to IETF RFC 6086 [54]; and

- with one or more participants included in the application/vnd.3gpp.mid-call+xml MIME body;

if the SIP INFO request is received after a CC CONNECT ACK message related to a CC CONNECT message sent as result of a SIP 2xx response to SIP INVITE request due to STN and if the SIP INVITE request established a session with conference focus, then the MSC server shall:

NOTE 3: If the SIP INFO request is received before the CC CONNECT ACK message, the MSC processes the contents of the SIP INFO request after reception of the CC CONNECT ACK message.

1. if inactive speech media component is negotiated by the SDP answer of the SIP 2xx response to the SIP INVITE request due to STN, associate the session and the participants extracted from the application/vnd.3gpp.mid-call+xml MIME body with CS calls:

   - with transaction identifiers calculated as in the table 9.2.1A-2. The offsets 0, 2, 3, 4, 5 are assigned to the participants in their order in the list of the extracted participants; and

   - with TI flag value as in mobile originated call;

   and enter the "active" (N10) state (defined in 3GPP TS 24.008 [8]), the "call held" hold auxiliary state (defined in 3GPP TS 24.083 [43]) and the "call in MPTY" multi party auxiliary state (defined in 3GPP TS 24.084 [47]) for the CS calls. The MSC server may subscribe to the conference event package as specified in 3GPP TS 24.605 [31]; and

NOTE 4: The transaction identifier that the MSC received in the CC SETUP message is the transaction identifier assigned to the first participant (offset 0).

NOTE 5: After handling the CC CONNECT ACK according to 3GPP TS 29.292 [18] and 3GPP TS 24.008 [8], the dialog created by the SIP INVITE request due to STN is associated with a CS call in the "active" (N10) state (defined in 3GPP TS 24.008 [8]), the "idle" hold auxiliary state (defined in 3GPP TS 24.083 [43]) and the "idle" multi party auxiliary state (defined in 3GPP TS 24.084 [47]) with transaction identifier and TI flag value as received in the CC SETUP message.

NOTE 6: The multi party auxiliary state was initially set to "idle". This state is re-assigned to "call in MPTY" after processing the SIP INFO request to reflect the multi party auxiliary state associated with the first participant.

2. if active speech media component is negotiated by the SDP answer of the SIP 2xx response to the SIP INVITE request due to STN, associate the session and the participants extracted from the application/vnd.3gpp.mid-call+xml MIME body with CS calls:

- with transaction identifiers calculated as in the table 9.2.1A-2; The offsets 0, 2, 3, 4, 5 are assigned to the participants in their order in the list of the extracted participants; and.
- with TI flag value as in mobile originated call;

and enter the "active" (N10) state (defined in 3GPP TS 24.008 [8]), the "idle" hold auxiliary state (defined in 3GPP TS 24.083 [43]) and the "call in MPTY" multi party auxiliary state (defined in 3GPP TS 24.084 [47]) for the CS calls.

Upon receiving a SIP REFER request:

1. with the Refer-Sub header field containing "false" value;
2. with the Supported header field containing "norefersub" value;
3. with the Refer-To header field containing a SIP URI with the Target-Dialog URI header field;
4. sent inside an existing SIP dialog:
   A. which was originated by the MSC server; and
   B. where the g.3gpp.mid-call feature-capability indicator as described in annex C was included in the Feature-Caps header field of the SIP 2xx response to the SIP INVITE request; and
5. containing a MIME body of MIME type specified in the subclause D.1.3;

the MSC server shall:

1. handle the SIP REFER request as specified in 3GPP TS 24.229 [2], IETF RFC 3515 [13] as updated by IETF RFC 6665 [81], and IETF RFC 4488 [20] without establishing an implicit subscription; and

NOTE 7: In accordance with IETF RFC 4488 [20], the MSC server inserts the Refer-Sub header field containing the value "false" in the SIP 2xx response to the SIP REFER request to indicate that it has not created an implicit subscription.

2. send a SIP INVITE request for transfer of an additional inactive session not using ICS capabilities in accordance with the procedures specified in 3GPP TS 24.229 [2] and IETF RFC 3515 [13]. Additionally, the MSC server shall populate the SIP INVITE request as follows:
   A. header fields which were included as URI header fields in the URI in the Refer-To header field of the received SIP REFER request as specified in IETF RFC 3261 [19] except the hname "body" URI header field;
   B. include the g.3gpp.mid-call media feature tag as described in annex C in the Contact header field according to IETF RFC 3840 [53];
   C. the SDP offer with:
      a. the same amount of the media descriptions as in the hname "body" URI header field in the URI in the Refer-To header field of the received SIP REFER request;
      b. each "m=" line having the same media type as the corresponding "m=" line in the hname "body" URI header field in the URI in the Refer-To header field of the received SIP REFER request;
      c. port set to zero value in each "m=" line whose corresponding "m=" line in the hname "body" URI header field in the URI in the Refer-To header field of the received SIP REFER request has port with zero value;
      d. media directionality as in the hname "body" URI header field in the URI in the Refer-To header field of the received SIP REFER request;

NOTE 8: port can be sent to zero or non zero value for the offered "m=" line whose corresponding "m=" line in the hname "body" URI header field in the URI in the Refer-To header field of the received SIP REFER request has port with nonzero value.

      e. all or subset of payload type numbers and their mapping to codecs and media parameters not conflicting with those in the hname "body" URI header field in the URI in the Refer-To header field of the received SIP REFER request; and

f. if local configuration indicates that the network is serving users supporting the precondition mechanims, indicate preconditions as met, using the segmented status type, as defined in IETF RFC 3312 [88] and IETF RFC 4032 [89], as well as the strength-tag value "mandatory" for the local segment and the strength-tag value either "optional" or as specified in RFC 3312 [88] and RFC 4032 [89] for the remote segment;

D. if local configuration indicates that the network is serving users supporting the precondition mechanism, include:

a. a "100rel" option tag as defined in IETF RFC 3262 [86] to indicate the support for reliable provisional responses; and

b. a "precondition" option tag as defined in IETF RFC 3312 [88] to indicate the support for the precondition mechanism; and

E. if a P-Asserted-Identity header field is not included in the headers portion of the URI in the Refer-To header field of the received SIP REFER request as specified in IETF RFC 3261 [19], include a P-Asserted-Identity header field with the value of the P-Asserted-Identity header field of the SIP INVITE requests due to STN which created the dialog in which the REFER request is received.

Upon receiving SIP 2xx response to the SIP INVITE request for transfer of an additional inactive session, the MSC server shall:

1. if:

a) the SIP INVITE request for transfer of the additional inactive session did not established a session with a conference focus; or

b) the application/vnd.3gpp.mid-call+xml MIME body included in the SIP REFER request does not contain one or more participants:

associate the additional inactive session with CS call with transaction identifier calculated as in the table 9.2.1A-1 and TI flag value as in mobile originated call and enter the "active" (N10) state (as defined in 3GPP TS 24.008 [8]), the "call held" hold auxiliary state (as defined in 3GPP TS 24.083 [43]) and the "idle" multi party auxiliary state (defined in 3GPP TS 24.084 [47]) for this CS call; and

2. if the SIP INVITE request for the additional inactive session established a session with conference focus and the application/vnd.3gpp.mid-call+xml MIME body included in the SIP REFER request contained one or more participants:

a) associate the additional inactive session and the participants extracted from the application/vnd.3gpp.mid-call+xml MIME body included in the SIP REFER request with CS calls:

- with transaction identifiers calculated as in the table 9.2.1A-2. The offsets 1, 2, 3, 4, 5 are assigned to the participants in their order in the list of the extracted participants; and

- with TI flag value as in mobile originated call;

and enter the "active" (N10) state (defined in 3GPP TS 24.008 [8]), the "call held" hold auxiliary state (defined in 3GPP TS 24.083 [43]) and the "call in MPTY" multi party auxiliary state (defined in 3GPP TS 24.084 [47]) for the CS calls. The MSC server may subscribe to the conference event package as specified in 3GPP TS 24.605 [31]

9.6 PS to CS session continuity with MSC server assisted mid-call feature for speech and video session

This subclause describes the procedures required by an MSC server in order to support the MSC server assisted mid call feature for speech and video session.

The MSC server , upon receiving the session state information which indicates an inactive speech and video session, shall send a SIP INVITE request for the additional inactive speech and video session as described in subclause 9.5.

NOTE 1: If due to some reason (i.e. the current RAN type not supporting video, lack of resource, etc.) the video media cannot be supported in CS network for the speech and video session, then the MSC server can set the port to zero in the "m=" line for the video media in the SDP offer of the SIP INVITE request for the additional inactive session, so as to inform the SCC AS that the video media is deleted and only the audio media of the speech and video session is transferred to CS.

NOTE 2: After successful transfer of a speech and video session and a speech session from PS to CS, if messages are received from the UE to switch between the two sessions (i.e. HOLD/Release message to hold/release the active session and Retrieve message to retrieve the inactive session), the MSC server can perform the procedures as specified in 3GPP TS 29.292 [18], with the addition that the MSC server can complete the in-call modification or Redial procedures as specified in 3GPP TS 24.008 [8] to change the shared CS bearer of the two sessions from speech to multimedia, or vice versa, before sending a SIP UPDATE request or SIP re-INVITE request to the SCC AS to resume the inactive session.

9.7 MSC procedures for PS to CS dual radio access transfer of calls in an early phase

The MSC server supporting PS to CS dual radio access transfer for calls in alerting phase shall populate a SIP INVITE request due to STN as follows:

1) include the g.3gpp.drvcc-alerting media feature tag as described in annex C in the Contact header field of the SIP INVITE request according to IETF RFC 3840 [53]; and

2) if the MSC server supports the PS to CS dual radio access transfer for originating calls in pre-alerting phase, the MSC server shall include the g.3gpp.ps2cs-drvcc-orig-pre-alerting media feature tag according to annex C in the Contact header field of the SIP INVITE request according to IETF RFC 3840 [53].

NOTE 1: Since the MSC server is not able to distinguish the dual radio access transfer from the regular session set up, the information elements above are added to every SIP INVITE request sent by the MSC server.

Upon receiving a SIP REFER request within the SIP session established by the SIP INVITE request due to STN:

1) with the Refer-Sub header field containing "false" value;

2) with the Supported header field containing "norefersub" value;

3) with the Target-Dialog URI header field in the URI of the Refer-To header field;

4) where the g.3gpp.ps2cs-drvcc-alerting feature-capability indicator or the g.3gpp.ps2cs-drvcc-originating-pre-alerting feature-capability indicator as described in annex C was included in the Feature-Caps header field of the SIP 2xx response to the SIP INVITE request due to STN; and

5) containing a MIME body of MIME type specified in the subclause D.2.4,

NOTE 2: At this point, the MSC server interacts with the MGW to provide information needed in the procedures below and to request MGW to start forwarding the audio media from the remote UE to the MSC server. The details of interaction between the MSC server and the MGC are out of scope of this document.

then the MSC server shall:

1) handle the SIP REFER request as specified in 3GPP TS 24.229 [2], IETF RFC 3515 [13] as updated by IETF RFC 6665 [81], and IETF RFC 4488 [20] without establishing an implicit subscription; and

NOTE 3: In accordance with IETF RFC 4488 [20], the MSC server inserts the Refer-Sub header field containing the value "false" in the SIP 2xx response to the SIP REFER request to indicate that it has not created an implicit subscription.

2) send a SIP INVITE request for transfer of an additional early session in accordance with the procedures specified in 3GPP TS 24.229 [2] and IETF RFC 3515 [13]. The SC UE shall populate the SIP INVITE request as follows:

   A. header fields which were included as URI header fields in the URI in the Refer-To header field of the received SIP REFER request as specified in IETF RFC 3261 [19] except the "body" URI header field;

   B) the SDP offer with:

a) the same amount of the media descriptions as in the "body" URI header field in the URI in the Refer-To header field of the received SIP REFER request;

b) each "m=" line having the same media type as the corresponding "m=" line in the "body" URI header field in the URI in the Refer-To header field of the received SIP REFER request;

c) port set to zero value in each "m=" line whose corresponding "m=" line in the "body" URI header field in the URI in the Refer-To header field of the received SIP REFER request has port with zero value; and

d) media directionality as in the "body" URI header field in the URI in the Refer-To header field of the received SIP REFER request; and

NOTE 4: port can be set to zero or nonzero value for the offered "m=" line whose corresponding "m=" line in the "body" URI header field in the URI in the Refer-To header field of the received SIP REFER request has port with nonzero value.

e) if local configuration indicates that the network is serving users supporting the precondition mechanism, indicate preconditions as met, using the segmented status type, as defined in IETF RFC 3312 [88] and IETF RFC 4032 [rfc4032], as well as the strength-tag value "mandatory" for the local segment and the strength-tag value either "optional" or as specified in RFC 3312 [88] and RFC 4032 [89] for the remote segment;

C) if the MSC server supports the MSC server assisted mid-call feature, include the g.3gpp.mid-call media feature tag as described in annex C in the Contact header field according to IETF RFC 3840 [53];

D) if local configuration indicates that the network is serving users supporting SIP preconditions, include:

a) a "100rel" option tag as defined in IETF RFC 3262 [86] to indicate the support for reliable provisional responses; and

b) a "precondition" option tag as defined in IETF RFC 3312 [88] to indicate the support for the SIP precondition mechanism; and

E) if a P-Asserted-Identity header field is not included in the headers portion of the URI in the Refer-To header field of the received SIP REFER request as specified in IETF RFC 3261 [19], include a P-Asserted-Identity header field with the value of the P-Asserted-Identity header field of the SIP INVITE requests due to STN which created the dialog in which the SIP REFER request is received.

Upon receipt of a SIP 18x response to SIP INVITE request for an additional early session, the MSC server shall:

1) associate the SIP INVITE request for an additional early session with CS call with transaction identifier calculated as in the table 9.2.1A-1 and TI flag value as in originating mobile case;

2) if the received response is a not a SIP 180 (Ringing) response enter the "mobile originating call proceeding" (N3) state as specified in 3GPP TS 24.008 [8]; and

3) if the received response is a SIP 180 (Ringing) response enter the "call delivered" (N4) state as specified in 3GPP TS 24.008 [8].

9.8 MSC server enhanced for dual radio access transfer using a SIP interface

9.8.1 General

When the MSC server enhanced for dual radio access transfer using a SIP interface receives a CC SETUP message containing the Called Party BCD number information element with the STN, the MSC server shall:

1. suppress services provided by the home network as part of the call setup procedures; and

2. not provide announcement or other in-band media for any calls that have been transferred or are in the process of being transferred.

NOTE 1: In the case of roaming the home network operator need to provide the STN as part of the bilateral operator agreement.

An MSC server enhanced for dual radio access transfer using a SIP interface shall interwork CC messages as specified in 3GPP TS 29.292 [18].

If the MSC server enhanced for dual radio access transfer using a SIP interface supports PS to CS dual radio access transfer for calls in alerting phase, the MSC server shall apply the procedures specified in subclause 9.7.

NOTE 2: An MSC server enhanced for dual radio access transfer using a SIP interface that supports PS to CS dual radio access transfer for originating calls in pre-alerting phase also supports PS to CS dual radio access transfer for calls in alerting phase.

NOTE 3: The MSC server enhanced for dual radio access transfer using a SIP interface cannot support PS to CS dual radio access transfer of calls in alerting phase on the terminating side since transfer of a waiting call requires ICS functionality. A waiting call will be transferred using CS access via MGCF when ICS functionality is not supported by the MSC server.

The MSC server enhanced for dual radio access transfer using a SIP interface supporting the MSC server assisted mid-call feature for speech sessions shall apply the procedures specified in subclause 9.5.

9.9 EATF

9.9.1 EATF procedures for PS to CS session continuity, dual radio session transfer of emergency session

The EATF needs to distinguish the following initial SIP INVITE request to provide specific functionality for dual radio session transfer of IMS emergency session:

1. SIP INVITE request routed to the EATF due to E-STN-DRVCC in the Request-URI. In the procedures below, such requests are known as "SIP INVITE requests due to E-STN-DRVCC".

Other initial SIP requests can be dealt with in any manner conformant with 3GPP TS 24.229 [2].

When the EATF receives a SIP INVITE request due to E-STN-DRVCC on the Target Access Leg, the EATF shall:

1. associate the SIP INVITE request due to E-DRVCC-SR with a source access leg, i.e. an emergency session with active speech media component anchored at the EATF.

NOTE 1: The EATF has generated the E-STN-DRVCC when the related emergency SIP INVITE requrest was received and has delivered the E-STN-DRVCC to the UE in the SIP 200 (OK) response to the UE.

If no source access leg exists, i.e. no dialog supporting a session with active speech media component exists or if multiple source access legs exist, then the EATF shall send a SIP 480 (Temporarily Unavailable) response to reject the SIP INVITE request due to E-STN-DRVCC; and

2. originate session modification as described in 3GPP TS 24.229 [2] towards the remote side with a new SDP offer with media characteristics as received in the SIP INVITE request due to E-STN-DRVCC.

Upon reception of the SIP ACK request to the SIP 200 (OK) response to the SIP INVITE request due to E-STN-DRVCC from the target access leg, the EATF shall release the source legs as follows:

1) if the source access leg is an ongoing session containing only an active media component, send a SIP BYE request on the source access leg in accordance with 3GPP TS 24.229 [2]; or

2) if the source access leg contains media components other than speech media component, send a SIP re-INVITE request to update the source access leg in accordance with 3GPP TS 24.229 [2].

NOTE 2: Delaying the release of the source access leg as described above allows an SC UE to reuse the PS dialog in case of PS to CS DRVCC cancellation.

10 Roles for PS-PS access transfer

10.1 Introduction

This clause specifies the procedures for PS-PS access transfer for both full media transfer case and partial media transfer case. Procedures are specified for the SC UE and the SCC AS.

10.2 SC UE

10.2.0 General

The SC UE may be engaged in one or more ongoing sessions or in one or more SIP dialogs in early state before performing access transfer. By an ongoing session, it is meant a session for which the SIP 2xx response for the initial SIP INVITE request to establish this session has been sent or received. By a SIP dialog in early state, it is meant an early SIP dialog which has been created by a provisional response to the initial SIP INVITE request, but for which the SIP 2xx response has not yet been sent or received.

The SC UE shall follow the procedures specified in subclause 6.2 to perform registration in the IM CN subsystem on the newly selected access network before performing PS-PS access transfer. When registering a new contact address, the SC UE may either:

a) not employ the multiple registration mechanism. In this case, upon the registration of the new contact address, all dialogs associated with the old contact address are terminated by the S-CSCF. The terminated dialogs include the dialog on the Source Access Leg and the SC UE's subscription dialog to its reg-event; or

NOTE 1: Since the SCC AS retains the information pertaining to the dialog on the Source Access Leg, as specified in subclause 10.3.4, upon receiving an initial SIP INVITE request due to PS to PS STI (i.e. on the Targer Access Leg) containing the Replaces header field, the SCC AS will be able to identify the dialog toward the the remote UE associated with the dialog on the Source Access Leg being replaced.

b) employ the multiple registration mechanism. In this case, the SC UE may either:

- add new flow that terminates at the new contact address, and leave all dialogs associated with the old flow and old contact address intact; or

- replace the old flow that terminates at the old contact address with a new flow that terminates at the new contact address, resulting in all dialogs associated with the old flow and old contact address being terminated (include the dialog on the Source Access Leg and the SC UE's subscription dialog to its reg-event).

NOTE 2: Since the SCC AS retains the information pertaining to the dialog on the Source Access Leg, as specified in subclause 10.3.4, upon receiving an initial SIP INVITE request due to PS to PS STI (i.e. on the Targer Access Leg) containing the Replaces header field, the SCC AS will be able to identify the dialog toward the the remote UE associated with the dialog on the Source Access Leg being replaced.

NOTE 3: When transferring all media from the Source Access Leg to the Target Access Leg, the SC UE can replace the old flow with a new flow, and let the network terminate all dialogs and the registration associated with the old flow, rather than the SC UE performing these actions itself.

10.2.1 Full session transfer

This subclause specifies a full session transfer applicable to a SC UE that supports dual mode operation and multiple registration procedure.

To initiate PS-PS access transfer for a session, upon acquiring the resources for media on the Target Access Leg, the SC UE shall send a SIP INVITE request due to PS to PS STI on the Target Access Leg in accordance with UE procedures specified in 3GPP TS 24.229 [2]. The SC UE shall populate the SIP INVITE request as follows:

1) the Request-URI set to

   A) if the PS to PS STI URI is configured in the SC UE, the configured PS to PS STI URI; and

   B) if the PS to PS STI URI is not configured in the SC UE, the URI contained in the Contact header field returned at the creation of the dialog on the Source Access Leg;

2) include in the Contact header field:

   A) a public GRUU or temporary GRUU as specified in 3GPP TS 24.229 [2] if a GRUU was received at registration; and

   B) the g.3gpp.ics media feature tag set to "principal" as specified in annex B of 3GPP TS 24.292 [4];

3) select one of the following options:

   A) if usage of SIP Replaces extension is selected:

      a) the Replaces header field populated as specified in IETF RFC 3891 [10], containing the dialog identifier of the session to be transferred; and

      b) the Require header field populated with the option tag value "replaces"; or

   B) if usage of SIP Target-Dialog extension is selected:

      a) the Target-Dialog header field populated as specified in IETF RFC 4538 [11], containing the dialog identifier of the session to be transferred; and

      b) the Require header field populated with the option tag value "tdialog";

4) the SDP payload set for the media component(s) to be transferred, in accordance with the UE SDP origination procedures specified in 3GPP TS 24.229 [2]. The SC UE shall create an SDP offer that contains the same number of media lines in the same order, where each media line corresponds to one of the media components in the original session, unless media components need to be added, and such that:

   A) each media line indicates the same media type as its corresponding media component in the original session and contains at least one codec that was negotiated during the original session;

   B) all or a subset of payload type numbers and their mapping to codecs and media parameters are not conflicting with those negotiated in the original session.; and

   C) if the SC UE determines to:

      a) remove a media component during the transfer, set the media line for this media component to a port number with value zero; and

      b) add new media component(s) during the transfer, include one additional media line with the desired media type and codecs for each new media component at the end of the SDP and indicate that the resources are available;

5) if the Source Access Leg is an early dialog and this early dialog was created by the SC UE receiving a SIP INVITE request, indicate support of the info package mechanism as specified in IETF RFC 6086 [54]; and

6) signalling elements described in subclause 6A.2.2.2.

NOTE 1: If an SC UE is an ICS UE with an ongoing session using CS bearer and Gm reference point for service control signalling, the SC UE can perform an access transfer of the service control signalling from the current IP-CAN to a new IP-CAN with the same capabilities (i.e. supporting CS and PS bearers, simultaneously) while retaining the media component in the CS access network by including the description of audio/video media over a circuit switched bearer in the SDP of the access transfer request, so that service continuity of the session is maintained.

If the dialog on the Source Access Leg is a confirmed dialog, then upon receiving SIP 2xx response for its SIP INVITE request due to PS to PS STI sent on the Target Access Leg, the SC UE shall:

1) send a SIP ACK request;

2) consider the confirmed dialog on the Source Access Leg as being successfully transferred to the Target Access Leg; and

3) send a SIP BYE request to the SCC AS on the Source Access Leg to terminate the confirmed dialog on the Source Access Leg, if the confirmed dialog is still active (e.g. it has not been released by the SCC AS).

NOTE 2: If the dialog on the Source Access Leg is a confirmed dialog, the SC UE upon sending an initial SIP INVITE request due to PS to PS STI on the Target Access Leg will not receive any SIP provisional response from the SCC AS, i.e. the initial SIP INVITE request due to PS to PS STI is either accepted with the SIP 200 (OK) response containing the SDP answer or rejected with an appropriate final SIP response.

NOTE 2A: If the contact address used by the dialog over the Source Access Leg was registered using multiple registration procedure, and the flow over the Target Access Leg did not replace the flow over the Source Access Leg, then upon transferring the dialog to the Target Access Leg, the SC UE is still registered on the Source Access Leg and its subscription dialog to its reg-event the Source Access Leg is intact.

If the dialog on the Source Access Leg is a confirmed dialog and if the SC UE receives any SIP 4xx – 6xx response to the SIP INVITE request due to PS to PS STI sent on the Target Access Leg, then PS-PS access transfer has not completed successfully and the call will continue in the Source Access Leg.

If the dialog on the Source Access Leg is an early dialog, then upon receiving a SIP 183 (Session Progress) response for its SIP INVITE request due to PS to PS STI sent on the Target Access Leg containing the SDP answer, the SC UE shall:

NOTE 3: If the dialog on the Source Access Leg is an early dialog, then the SC UE upon sending an initial SIP INVITE request due to PS to PS STI on the Target Access Leg, receives either a SIP 183 (Session Progress) response containing the SDP answer or the initial SIP INVITE request due to PS to PS STI is rejected with an appropriate final SIP response.

1) respond with a SIP PRACK request; and

2) upon receiving the SIP 200 (OK) response for the SIP PRACK request, consider the early dialog on the Source Access Leg as being successfully transferred to the Target Access Leg and being at the same early dialog stage as the early dialog on the Source Access Leg.

NOTE 4: All subsequent SIP requests or SIP responses originating from the remote UE and destined for the SC UE will be sent to the SC UE over the Target Access Leg. For example, in case of an early dialog originated by the SC UE sending an initial SIP INVITE request to the remote UE and receiving a SIP 183 (Session Progress) response on the Source Access Leg, and subsequently transferring the early dialog to the Target Access Leg, the SIP 180 (Ringing) response from the remote UE will be conveyed to the SC UE on the Target Access Leg rather than on the Source Access Leg.

Since, upon receiving the SIP 200 (OK) response for the SIP PRACK request, the early dialog and the associated media have been transferred from the Source Access Leg to the Target Access Leg (i.e. the resources for media on the Source Access Leg are not used any more), the SC UE may releases the resources on the Source Access Leg by sending a SIP UPDTE request with an appropriate SDP offer on the Source Access Leg. However, in spite of releasing the resources, the dialog on the Source Access Leg is still in the early dialog phase.

If the dialog on the Source Access Leg is an early dialog that was created by the SC UE receiving a SIP INVITE request on the Source Access Leg (i.e. an incoming call), then upon receiving the SIP 200 (OK) response for the SIP PRACK request, the SC UE shall:

1) if the served user accepted the incoming call:

   a) send the SIP INFO request on the Target Access Leg containing:

      A) an Info-Package header field as specified in IETF RFC 6086 [54] with g.3gpp.state-and-event info package name; and

      B) application/vnd.3gpp.state-and-event-info+xml XML body associated with the info package according to IETF RFC 6086 [54] and with the event XML element containing "call-accepted" to indicate that the called party has answered the call; and

   b) upon receiving the SIP 200 (OK) response for the SIP INFO request, and subsequently upon receiving a SIP 200 (OK) response for its SIP INVITE request due to PS to PS STI sent on the Target Access Leg:

      A) consider the early dialog becoming a confirmed dialog and as being successfully transferred to the Target Access Leg; and

      B) release the early dialog on the Source Access Leg, by sending a SIP 410 (Gone) response on the Source Access Leg, if this early dialog is still active (e.g. it has not been previously terminated by the SCC AS);

2) if the incoming call is rejected:

NOTE 5: If, upon the early dialog being transferred to the Target Access Leg, the SC UE rejects the incoming call, the SC UE will terminate the early dialog on the Target Access Leg and the early dialog on the Source Access Leg.

a) send a SIP CANCEL request on the Target Access Leg that pertains to the SIP INVITE request due to PS to PS STI; and

b) send a SIP 410 (Gone) response to the initial SIP INVITE request received on the Source Access Leg; or

3) if the early dialog is transferred back from the Target Access Leg to the Source Access Leg (e.g. the radio is lost while the SC UE is ringing) before the SC UE sends the SIP INFO request on the Target Access Leg:

NOTE 6: If the SC UE transfers back the early dialog from the Target Access Leg to the Source Access Leg, it will re-acquire the resources for media on the Source Access Leg, terminate the early dialog on the Target Access Leg, and accept the incoming call on the Source Access Leg.

a) release the early dialog on the Target Access Leg, by sending a SIP CANCEL request on the Target Access Leg that pertains to the SIP INVITE request due to PS to PS STI;

b) re-acquire the resources for media on the Source Access Leg, if previously released, send a SIP UPDATE request with an appropriate SDP offer on the Source Access Leg; and

c) when the served user ether accepts the call or the call is rejected, send the respective final SIP response on the Source Access Leg, as specified in 3GPP TS 24.229 [2].

If the dialog on the Source Access Leg is an early dialog that was created by the SC UE sending a SIP INVITE request on the Source Access Leg (i.e. an outgoing call), then upon receiving SIP 200 (OK) response for the SIP PRACK request, the SC UE shall:

1) if the SC UE receives a SIP 200 (OK) response for the SIP INVITE request due to PS to PS STI sent on the Target Access Leg (i.e. the outgoing call is accepted by the remote UE):

a) send a SIP ACK request to the received SIP 200 (OK) response;

b) consider the early dialog becoming a confirmed dialog and as being successfully transferred to the Target Access Leg; and

c) terminate the early dialog on the Source Access Leg, by sending a SIP CANCEL request on the Source Access Leg, if this early dialog is still active (e.g. has not been previously terminated by the SCC AS);

2) if the SC UE receives a the SIP 410 (Gone) response to the initial SIP INVITE request on the Source Access Leg, and subsequently any SIP 4xx or 5xx final response to the SIP INVITE request due to PS to PS STI (i.e. the outgoing call is rejected by the remote UE):

a) consider the early dialogs as terminated; or

NOTE 7: If the remote UE rejects the call, the SCC AS will terminate the early dialog on the Source Access Leg prior to terminating the early dialog on the Target Access Leg. This will insure that the SC UE does not un-necessarily transfer the call to the Source Access Leg (e.g. re-acquires the resources for media) prior to the early dialog on the Source Access Leg being terminated.

3) if the early dialog is transferred back from the Target Access Leg to the Source Access Leg (e.g. the radio is lost while the remote UE is ringing) before the SC UE receives any final response on the Target Access Leg:

NOTE 8: If the SC UE transfers back the early dialog from the Target Access Leg to the Source Access Leg, it will re-acquire the resources for media on the Source Access Leg, terminate the early dialog on the Target Access Leg, and wait for the outgoing call to be either accepted or rejected by the remote UE.

a) release the early dialog on the Target Access Leg, by sending a SIP CANCEL request on the Target Access Leg that pertains to the SIP INVITE request due to PS to PS STI;

b) re-acquire the resources for media on the Source Access Leg, if previously released and send a SIP UPDATE request with an appropriate SDP offer on the Source Access Leg; and

c) wait for the final SIP response from the remote UE on the Source Access Leg that will indicate whether the call was accepted or rejected by the remote UE, and proceed as specified in 3GPP TS 24.229 [2].

If the dialog on the Source Access Leg is an early dialog and if the SC UE receives a SIP 4xx – 6xx response to its initial SIP INVITE request due to PS to PS STI sent on the Target Access Leg, (i.e. the access transfer of the early dialog has not completed successfully), the early dialog shall continue on the Source Access Leg, if this early dialog is still active. Hence, the SC UE shall:

NOTE 9: Since the early dialog on the Target Access Leg is terminated by the SCC AS, the SC UE re-acquires the resources on the Source Access Leg.

a) re-acquire the resources for media on the Source Access Leg, if previously released and send a SIP UPDATE request with an appropriate SDP offer on the Source Access Leg; and

b) respond with a SIP ACK request to the SIP 4xx – 6xx response, and consider the early dialog on the Target Access Leg as terminated, and either:

A) wait for the final SIP response from the remote UE on the Source Access Leg that will indicate whether the call was accepted or rejected by the remote UE, if the call is originated by the SC UE; or

B) send the respective final SIP response on the Source Access Leg when the call is accepted or rejected by the user, if the call is terminated at the SC UE.

10.2.1A Void

10.2.2 Partial session transfer

To initiate PS-PS access transfer for a session, the SC UE shall send a SIP INVITE request due to PS to PS STI over the Target Access Leg in accordance with UE procedures specified in 3GPP TS 24.229 [2]. The SC UE shall populate the SIP INVITE request as follows:

1. the Request-URI set to:

   A) if the PS to PS STI URI is configured in the SC UE, the configured PS to PS STI URI; and

   B) if the PS to PS STI URI is not configured in the SC UE, the URI contained in the Contact header field returned at the creation of the dialog over the Source Access Leg;

2. include in the Contact header field:

   A. a public GRUU or temporary GRUU as specified in 3GPP TS 24.229 [2] if a GRUU was received at registration; and

   B. the g.3gpp.ics media feature tag set to "principal" as specified in annex B of 3GPP TS 24.292 [4];

3. the Require header field with the option tag "tdialog" included;

4. the Target-Dialog header field populated as specified in IETF RFC 4538 [11], containing the dialog identifier of the session to be transferred;

5. the SDP payload set for the media component(s) to be transferred, in accordance with the UE SDP origination procedures specified in 3GPP TS 24.229 [2]. The SC UE shall create an SDP offer that contains the same number of media lines in the same order, where each media line corresponds to one of the media components in the original session, unless media components need to be added during the session transfer, and such that:

   A) each media line indicates the same media type as its corresponding media component in the original session and contains at least one codec that was negotiated during the original session;

   B) all or a subset of payload type numbers and their mapping to codecs and media parameters are not conflicting with those negotiated in the original session; and

   C) if the SC UE determines to:

      a. keep the media component on the Source Access Leg, set the media line for this media component to a port number with value zero; and

b. add new media component(s) during the transfer, include one additional media line with the desired media type and codecs for each new media component at the end of the SDP; and

NOTE: If an SC UE is an ICS UE with an ongoing session using CS bearer and Gm reference point for service control signalling, the SC UE can perform an access transfer of the service control signalling from the current IP-CAN to a new IP-CAN with the same capabilities (i.e. supporting CS and PS bearers, simultaneously) while retaining the media component in the CS access network by including the description of audio/video media over a circuit switched bearer in the SDP of the access transfer request, so that service continuity of the session is maintained.

6. signalling elements described in subclause 6A.2.2.2.

Upon receiving SIP 2xx response for the SIP INVITE request due to PS to PS STI sent over the Target Access Leg and sending SIP ACK request, the SC UE shall send a SIP re-INVITE request to the SCC AS over the Source Access Leg to update the original session. The SC UE shall populate the SIP re-INVITE request as follows:

1. the SDP payload set for all the media component(s) within the original session, in accordance with the UE SDP origination procedures specified in 3GPP TS 24.229 [2]. The SC UE shall set the port number for a media component to zero if that media component has been transferred to the Target Access Leg or has to be removed.

If the SC UE receives any SIP 4xx – 6xx response to the SIP INVITE request due to PS to PS STI sent over the Target Access Leg, then PS-PS access transfer has not completed successfully and the call will continue in the Source Access Leg.

10.2.3 Void

10.3 SCC AS

10.3.1 Distinction of requests sent to the SCC AS

The SCC AS needs to distinguish between the following initial SIP INVITE requests to provide specific functionality relating to access transfer:

- If the g.3gpp.pstops-sti media feature tag was included in the Contact header field of the REGISTER request when the SC UE registered, SIP INVITE requests routed to the SCC AS with the Request-URI containing the PS to PS STI URI belonging to the subscribed user are known as "SIP INVITE requests due to STI".

- If the g.3gpp.pstops-sti media feature tag was not included in the Contact header field of the REGISTER request when the SC UE registered, SIP INVITE requests routed to the SCC AS containing a STI belonging to the subscribed user in the Replaces header field or Target-Dialog header field and not containing Inter UE Transfer SCC AS URI in the Request-URI and not containing the additional transferred session SCC AS URI in the Request-URI are known as "SIP INVITE requests due to STI".

NOTE: The media streams that need to be transferred are identified using information described in the subsequent sections.

Other SIP initial requests for a dialog and requests for a SIP standalone transaction can be dealt with in any manner conformant with 3GPP TS 24.229 [2].

10.3.2 PS to PS access transfer procedures at the SCC AS

This subclause applies to reception of a SIP INVITE request due to STI with a PS media only.

When the SCC AS receives a SIP INVITE request due to STI on the Target Access Leg, the SCC AS shall:

- associate the SIP INVITE request on the Target Access Leg with a confirmed dialog or an early dialog on the Source Access Leg by matching the dialog identifier present in either the Replaces header field (see IETF RFC 3891 [10]) or the Target Dialog header field (see IETF RFC 4538 [11]) of the SIP INVITE request with a confirmed dialog or with an early dialog. By a previously established dialog, it is meant a dialog for which a SIP 2xx response to the initial SIP INVITE request has been sent or received. By an early dialog, it is meant an early dialog which has been created by a provisional response to the initial SIP INVITE request, but for which the SIP 2xx response has not yet been sent or received;

- if the SCC AS is unable to associate the SIP INVITE request with a confirmed dialog or an early dialog, send a SIP 480 (Temporarily Unavailable) response to reject the SIP INVITE request due to STI and not processes the remaining steps;

- if the SIP INVITE request contains a Replaces header field:

  a) void; and

  b) send a SIP re-INVITE request towards the remote UE using the confirmed dialog or send SIP UPDATE request(s) towards the remote UE(s) using the existing early dialog(s) which were created by the same SIP INVITE request as the Source Access Leg. The SCC AS shall populate the SIP re-INVITE request or the SIP UPDATE request(s) with a new SDP offer, including the media characteristics as received in the SIP INVITE request due to STI received on the Target Access Leg, by following the rules of 3GPP TS 24.229 [2];

- otherwise, if the SIP INVITE request contains a Target Dialog header field:

  a) if the number of media lines in the Target Access Leg is less than the number of media lines in the Source Access Leg or the media type for the corresponding media lines is not the same as in the original session, send a SIP 4xx response to reject the SIP INVITE request relating to the access transfer and not process the remaining steps;

  b) otherwise, either send a SIP re-INVITE request towards the remote UE using the confirmed dialog or send a SIP UPDATE request(s) towards the remote UE(s) using the existing early dialog(s) which were created by the same SIP INVITE request as the Source Access Leg. The SCC AS shall populate the SIP re-INVITE or the SIP UPDATE request(s) as follows:

    1) void; and

    2) include a new SDP offer, following the rules specified in 3GPP TS 24.229 [2], containing the following media information:

      i) the media characteristics as received in the SIP INVITE request due to STI received on the Target Access Leg for media streams whose port is not set to zero; and

      ii) for the media streams in the SIP INVITE request due to STI whose port is set to zero, include the corresponding media characteristics of those streams from the Source Access Leg.

If the Remote Leg is a confirmed dialog, then upon receiving the SIP 200 (OK) response to the SIP re-INVITE request, the SCC AS shall:

1) send a SIP 200 (OK) response to the initial SIP INVITE request due to STI containing:

   A) a SDP answer constructed from the SDP answer received in the SIP 200 (OK) response to the SIP re-INVITE request; and

   B) the signalling elements described in subclause 6A.4.3;

2) consider the confirmed dialog on the Source Access Leg as being successfully transferred to the Target Access Leg; and

3) if the SIP INVITE request due to STI contains:

   - a Replaces header field, send a SIP BYE request on the Source Access Leg to terminate the dialog on the Source Access Leg, if the dialog is still active (e.g. it has not been previously released by the SC UE); or

   - a Target Dialog header field and SDP of the SIP INVITE request due to STI contains:

     a) no media line whose port is set to zero, send a SIP BYE request on the Source Access Leg to terminate the dialog on the Source Access Leg if the dialog is still active (e.g. it has not been previously released by the SC UE); or

     b) any media line whose port is not zero, receive the SIP BYE request or SIP re-INVITE request from the Source Access Leg in the case of removing media during full transfer or partial access transfer, respectively.

When the SCC AS receives the SIP BYE request on the Source Access Leg, the SCC AS shall:

- if any media are still remaining on the Source Access Leg,

  a) send SIP 200 (OK) response for the SIP BYE request; and

  b) send SIP re-INVITE request to the remote UE to delete the media on the Source Access Leg by following the rules of 3GPP TS 24.229 [2]; and

- if there are no media on the Source Access Leg, send the SIP 200 (OK) respones for the SIP BYE request.

If the SCC AS receives the SIP 200 (OK) response about the SIP re-INVITE request to the remote UE (created by SIP BYE request), the SCC AS sends a SIP ACK request to acknowledge the received SIP 200 (OK) response.

When the SCC AS receives the SIP re-INVITE request on the Source Access Leg, the SCC AS shall send a SIP 200 (OK) response on the Source Access Leg to acknowledge the receipt of the SIP re-INVITE request.

If the Remote Leg is a confirmed dialog, and if subsequent to sending the SIP re-INVITE request to the remote UE and prior to sending any final SIP response on the Target Access Leg, the SCC AS decides (for any reason) to reject the access transfer request, the SCC AS shall release the Target Access Leg (e.g. by sending a SIP 4xx response), retain the Source Access Leg, and update the Remote Leg to match the Source Access Leg.

If the Remote Leg is an early dialog then upon receiving the SIP 2xx response to the SIP UPDATE request, the SCC AS shall send SIP 183 (Session Progress) response to the SIP INVITE request due to STI. The SCC AS shall populate the SIP 183 (Session Progress) response as follows:

a) include a SDP answer constructed from the SDP answer received in the SIP 2xx response to the SIP UPDATE request;

b) if the Remote Leg is an early dialog originated by the remote UE, include a Recv-Info header field containing the g.3gpp.state-and-event package name;

c) if the Remote Leg is an early dialog originated by the SC UE, if the SIP INVITE request due to STI contains a P-Early-Media header field with the "supported" parameter and if the SCC AS has received a P-Early-Media header field in a SIP message in the dialog of the SIP UPDATE request, include a P-Early-Media header field containing the value of the last P-Early-Media header field received in a SIP message in the dialog of the SIP UPDATE request; and

d) signalling elements described in subclause 6A.4.3.

If the dialog on the Source Access Leg is an early dialog, then upon receiving the SIP PRACK request for the SIP 183 (Session Progress) response and responding with a SIP 200 (OK) response, the SCC AS shall consider the early dialog on the Source Access Leg as being successfully transferred to the Target Access Leg and being at the same early dialog stage as the early dialog on the Source Access Leg.

NOTE 1: All subsequent SIP requests or SIP responses originating from the remote UE and destined for the SC UE will be sent to the SC UE over the Target Access Leg. If the SCC AS receives any SIP request on the Source Access Leg, the SCC AS will not convey the received SIP request to the remote UE.

If, upon sending the SIP 200 (OK) response for the SIP PRACK request, the SCC AS receives a SIP UPDATE request on the Source Access Leg that contains an SDP offer that indicates that the SC UE is releasing the resources for media on the Source Access Leg, the SCC AS will respond with a SIP 200 (OK) response containing the appropriate SDP answer, as specified in 3GPP TS 24.229 [2]. However, in spite of the resources being released, the dialog on the Source Access Leg is still active and in the early dialog phase.

If the Remote Leg is an early dialog originated by the remote UE, then upon sending the SIP 200 (OK) response for the SIP PRACK request, and when the SCC AS:

1) receives the SIP INFO request on the Target Access Leg(indicating that the SC UE has accepted the call) containing:

   a) an Info-Package header field as specified in IETF RFC 6086 [54] with 3gpp.state-and-event info package name; and

b) application/vnd.3gpp.state-and-event-info+xml XML body associated with the info package according to IETF RFC 6086 [54] and with the event XML element containing "call-accepted" to indicate that the called party has answered the call;

the SCC AS shall:

a) send a SIP 200 (OK) response on the Target Access Leg to acknowledge the receipt of the SIP INFO request;

b) send SIP 200 (OK) response to the initial SIP INVITE request to the remote UE;

c) upon sending the SIP 200 (OK) response to the SIP INFO request, send another SIP 200 (OK) response on the Target Access Leg that pertains to the SIP INVITE request due to STI received on the Target Access Leg. The SCC AS shall populate the SIP 200 (OK) response to the SIP INVITE request due to STI with signalling elements described in subclause 6A.4.3;

d) terminate the early dialog on the Source Access Leg, if still active (i.e. if not previously terminated by the SC UE) by sending a SIP CANCEL request on the Source Access Leg; and

NOTE 2: The SCC AS may delay the termination of the early dialog on the Source Access Leg to let the SC UE terminate this early dialog.

e) consider the early dialog becoming a confirmed dialog and successfully transferred to the Target Access Leg; or

2) receives both:

NOTE 3: If the SC UE wants to reject the incoming call, upon initiating the transfer of the early dialog to the Target Access Leg, the SC UE will terminate both early dialogs, i.e. the early dialog on the Target Access Leg and the early dialog on the Source Access Leg.

a) a SIP CANCEL request on the Target Access Leg cancelling the SIP INVITE request due to STI; and

b) a SIP 410 (Gone) response to the initial SIP INVITE request sent on the Source Access Leg;

the SCC AS shall:

a) respond to the SIP CANCEL request as specified in 3GPP TS 24.229 [2];

b) respond to the SIP 410 (Gone) response as specified in 3GPP TS 24.229 [2];

c) send the appropriate SIP 4xx response to the initial SIP INVITE request received from the remote UE that indicates to the remote UE that the call has been rejected; and

d) consider the early dialogs as terminated; or

3) receives:

NOTE 4: If the SC UE transfers back the early dialog from the Target Access Leg to the Source Access Leg, it will terminate the early dialog on the Target Access Leg, re-acquire the resources for media on the Source Access Leg, and accept the incoming call on the Source Access Leg.

a) a SIP CANCEL request on the Target Access Leg cancelling the SIP INVITE request due to STI; and

b) a SIP UPDATE request containing a SDP offer on the Source Access Leg, that indicates that the SC UE has re-acquired the resources for media on the Source Access Leg, if previously released;

NOTE 5: If the resources for media on the Source Access Leg have not been previously released, the SCC AS will not receive the SIP UPDATE request containing a SDP offer.

the SCC AS shall:

a) respond to the SIP CANCEL request as specified in 3GPP TS 24.229 [2];

b) if a SIP UPDATE request containing a SDP offer on the Source Access Leg was received:

A) send a SIP UPDATE request to the remote UE containing a SDP offer constructed from the SDP offer included in the SIP UPDATE request received on the Source Access Leg; and

B) when the SIP 2xx response to the SIP UPDATE request containing the SDP answer is received from the remote UE, send a SIP 200 (OK) response to the SIP UPDATE request received on the Source Access Leg that includes a SDP answer constructed from the SDP answer received in the SIP 2xx response to the SIP UPDATE request received from the remote UE; and

c) consider the early dialog as being transferred back to the Source Access Leg.

If the Remote Leg is an early dialog terminated at the remote UE, then upon sending the SIP 200 (OK) response for the SIP PRACK request, if the SCC AS:

1) receives SIP 200 (OK) response to the initial SIP INVITE request from the remote UE indicating that the remote UE has answered the call;

the SCC AS shall:

a) send a SIP 200 (OK) response toward the SC UE on the Target Access Leg that pertains to the SIP INVITE request due to STI. The SCC AS shall populate the SIP 200 (OK) response to the SIP INVITE request due to STI with signalling elements described in subclause 6A.4.3;

b) terminate the early dialog on the Source Access Leg, if still active (i.e. if not previously terminated by the SC UE) by sending the SIP 410 (Gone); and

c) consider the early dialog becoming a confirmed dialog and as successfully transferred to the Target Access Leg;

2) receives any final response (e.g. SIP 4xx response or SIP 5xx response) from the remote UE that indicates that the remote UE has rejected the call, the SCC AS shall:

NOTE 6: If the remote UE rejects the call, the SCC AS will terminate the early dialog on Source Access Leg prior to terminating the early dialog on the Target Access Leg. This will insure that the SC UE does not un-necessarily transfer the call to the Source Access Leg (e.g. re-acquires the resources) prior to the early dialog on the Source Access Leg being terminated.

a) send the SIP 410 (Gone) response to the initial SIP INVITE request received on the Source Access Leg;

b) then send a final response to the SIP INVITE request due to STI that is identical to the final response (e.g. SIP 4xx response or SIP 5xx response) received from the remote UE; and

c) consider the early dialogs as terminated; or

3) receives:

NOTE 7: If the SC UE transfers back the early dialog from the Target Access Leg to the Source Access Leg, before the SC UE receives any final response on the Target Access Leg, the SC UE will terminate the early dialog on the Target Access Leg,, re-acquire the resources for media on the Source Access Leg, and update the early dialog on the Source Access Leg.

a) a SIP CANCEL request on the Target Access Leg cancelling the SIP INVITE request due to STI; and

b) a SIP UPDATE request containing a SDP offer on the Source Access Leg, that indicates that the SC UE has re-acquired the resources for media on the Source Access Leg, if previously released;

NOTE 8: If the resources for media on the Source Access Leg have not been previously released, the SCC AS will not receive an SIP UPDATE request containing a SDP offer.

then the SCC AS shall:

a) respond to the SIP CANCEL request as specified in 3GPP TS 24.229 [2];

b) if a SIP UPDATE request containing a SDP offer on the Source Access Leg was received:

A) send a SIP UPDATE request to the remote UE containing a SDP offer constructed from the SDP offer included in the SIP UPDATE request received on the Source Access Leg; and

B) when the SIP 2xx response to the SIP UPDATE request containing the SDP answer is received from the remote UE, send a SIP 200 (OK) response to the SIP UPDATE request received on the Source Access

Leg that includes a SDP answer constructed from the SDP answer received in the SIP 2xx response to the SIP UPDATE request received from the remote UE; and

c) consider the early dialog as being transferred back to the Source Access Leg.

If the Remote Leg is an early dialog, and if subsequent to sending the SIP UPDATE request to the remote UE, and prior to sending any final SIP response on the Target Access Leg, the SCC AS decides (for any reason) to reject the access transfer request, the SCC AS shall release the Target Access Leg (e.g. by sending a SIP 4xx response), retain the Source Access Leg, and update the remote leg to match the Source Access Leg.

10.3.3 Void

10.3.4 S-CSCF releasing the source access leg during PS to PS access transfer

When SCC AS receives a SIP BYE request on an existing dialog on the Source Access Leg with the status code 480 (Temporarily Unavailable) in a Reason header field indicating that this dialog was released by the S-CSCF, the SCC AS shall delay the release of the dialog toward the the remote UE and retaining the information pertaining to the dialog on the Source Access Leg for a specific time interval. If the SCC AS:

a) receives within this time interval an initial INVITE request (i.e. on the Targer Access Leg) indicating that this dialog is replacing the dialog on the Source Access Leg, then the SCC AS shall not initiate the release of the dialog toward the the remote UE; or

NOTE 1: By retaining the information pertaining to the dialog on the Source Access Leg, and upon receiving an initial SIP INVITE request (i.e. on the Targer Access Leg), the SCC AS will be able to identify the dialog on the Source Access Leg and the associated dialog toward the the remote UE.

b) does not receive within this time interval an initial SIP INVITE request (i.e. on the Target Access Leg) indicating that this dialog is replacing the dialog on the Source Access Leg, then the SCC AS shall initiate the release of the dialog toward the the remote UE and delete the information pertaining to the dialog on the Source Access Leg.

NOTE 2: The time interval is defined by the operator policy. The value of 8 seconds is an appropriate value for the time interval.

NOTE 3: When the UE, prior to sending the initial SIP INVITE request on the Target Access Leg, registers new contact address and either uses the multiple registrations where new flow on the Target Access Leg replaces an old flow on the Source Access Leg or does not uses the multiple registrations, the S-CSCF will terminate all dialogs associated with the old constant address or old flow, as specified in 24.229. By retaining the information pertaining to the dialog on the Source Access Leg, the SCC AS knows which dialog is being replaced.

10.3.5 P-CSCF releasing the source access leg during PS to PS access transfer

The procedures specified in subclause 12.3.3.2 apply.

10.3.6 P-CSCF releasing early dialog during PS to PS access transfer

When the SCC AS that supports PS to PS access transfer for early dialogs, receives either:

1) a SIP BYE request on the Source Access Leg, with the Reason header field containing a SIP 503 (Service Unavailable) response code, that is releasing an early dialog on the Source Access Leg originated by the SC UE;

2) a SIP CANCEL request on the Source Access Leg, with the Reason header field containing a SIP 503 (Service Unavailable) response code, that is releasing an early dialog on the Source Access Leg originated by the SC UE;

3) a SIP 503 (Service Unavailable) response on the Source Access Leg, that is releasing an early dialog on the Source Access Leg terminating at the SC UE;

4) a SIP 500 (Server Internal Error) response on the Source Access Leg, that is releasing an early dialog on the Source Access Leg terminating at the SC UE;

the SCC AS shall delay the release of the associated early dialog toward the the remote UE on the Remote Leg and retaining the information pertaining to the early dialog on the Source Access Leg for a specific time interval. Subsequently, if the SCC AS:

- receives within this time interval an initial SIP INVITE request on the Target Access Leg associated with the early dialog on the Source Access Leg, then the SCC AS shall not initiate the release of the early dialog toward the the remote UE on the Remote Leg; or

- does not receive within this time interval an initial SIP INVITE request on the Target Access Leg associated with the early dialog on the Source Access Leg, then the SCC AS shall initiate the release of the early dialog toward the the remote UE on the Remote Leg and delete the information pertaining to the early dialog on the Source Access Leg.

NOTE: The time interval is defined by the operator policy. The value of 8 seconds is an appropriate value for the time interval.

11 Roles for PS-PS access transfer in conjunction with PS-CS access transfer

11.1 Introduction

This clause specifies the procedures for PS-PS access transfer in conjunction with PS-CS access transfer. Procedures are specified for the SC UE and the SCC AS. For SC UE or SCC AS not supporting ICS procedures, PS-PS access transfer with a remote end in conjunction with PS-CS access transfer with the same remote end is only possible when the UE is active in a single CS call with the remote end i.e. support of session transfer with more than one CS call is not provided.

11.2 SC UE

11.2.1 SC UE procedures for PS to PS+CS access transfer

11.2.1.1 General

The SC UE may be engaged in one or more ongoing sessions before performing access transfer. By an ongoing session, it is meant a session for which the SIP 2xx response for the initial SIP INVITE request to establish this session has been sent or received.

11.2.1.2 SC UE procedures for PS to PS+CS access transfer using ICS

This subclause applies for IMS sessions containing not only speech media component, otherwise subclause 9.2.1 applies.

If SC using ICS is enabled then if the SC UE is using Gm, then for each session with speech media component to be transferred and starting with the full-duplex session with active speech media component, the SC UE shall send a SIP INVITE request to the SCC AS as specified for call origination for ICS UE using Gm in 3GPP TS 24.292 [4]. The SC UE shall populate the SIP INVITE request as specified for PS-PS access transfer with full media transfer in subclause 10.2.1 with the following exceptions:

- The SC UE shall indicate in the SIP INVITE request that the speech media component is using CS bearer with its corresponding media description.

- When sending the SIP INVITE request for the full-duplex sessions with inactive speech media component and if precondition is used, the SC UE shall indicate that the related local preconditions for the speech media component are met.

- For the full-duplex session with active speech media component, upon receiving the PSI DN from the SCC AS, the SC UE shall follow the procedures for call origination for ICS UE using Gm in 3GPP TS 24.292 [4] to set up the CS bearer.

If service control over Gm for the CS bearer is retained on the source access leg, the SC UE shall:

- send an SIP INVITE request as specified for partial session transfer in subclause 10.2.2. indicating transfer of non-speech media to the target access leg; and

- send a SIP re-INVITE request over the source access leg indicating that the speech media component is to be transferred to a CS bearer as described in 3GPP TS 24.292 [4] subclause 8.2.2.2. If other media components are retained or added on the source access leg, then these are included in the SDP offer.

For the full-duplex session with active speech media component, upon receiving the SCC AS PSI DN from the SCC AS, the SC UE shall follow the procedures for call origination for ICS UE using Gm in 3GPP TS 24.292 [4] to set up the CS bearer.

11.2.1.3 SC UE procedures for PS to PS+CS access transfer not using ICS

If the SC UE is not using ICS capabilities and if the SC UE does not apply the MSC server assisted mid-call feature as specified in subclause 9.2.1A, then access transfer is only possible when the UE is active in a single full-duplex session with active speech media component.

For the non-speech components to be transferred to the PS Target Access Leg, the SC UE shall send a SIP INVITE request to the SCC AS as specified for PS-PS access transfer with partial media transfer in subclause 10.2.1. For the speech media component to be transferred to the CS Target Access leg, the SC UE shall send to the SCC AS a CC SETUP message as specified in 3GPP TS 24.008 [8]. When sending the CC SETUP message, the SC UE shall populate the CC SETUP message as follows:

1) the called party BCD number information element set to the STN; and

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

Upon receiving the SIP 2xx response from the SCC AS for the PS Target Access Leg and sending SIP ACK request and upon receiving CS call setup confirmation message, e.g. CC CONNECT message, for the CS Target Access Leg, the SC UE shall send a SIP BYE request to terminate the Source Access Leg, following the procedures specified in 3GPP TS 24.229 [2].

If the SC UE receives any SIP 4xx – 6xx response to the SIP INVITE request for the PS Target Access leg and receives CS call setup failure message for the CS Target Access Leg, then session transfer has not occurred and the call will continue in the original domains.

If the SC UE receives any SIP 4xx – 6xx response to the SIP INVITE request for the PS Target Access Leg and receives CS call setup confirmation message for the CS Target Access Leg, then the session transfer is only successful for part of the media components. The SC UE shall update the Source Access leg by following the procedures specified for PS-PS access transfer with partial media transfer in subclause 10.2.2 to indicate that all media components other than the speech media component are still maintained on the Source Access Leg.

If the SC UE receives CS call setup failure message for the CS Target Access Leg but receives a SIP 2xx response for the PS Target Access Leg, then the session transfer is only successful for part of the media components. Upon sending SIP ACK request, the SC UE shall update the Source Access leg by following the procedures specified for PS-PS access transfer with partial media transfer in subclause 10.2.2 to indicate that the speech media component is still maintained on the Source Access Leg.

11.2.1.4 SC UE procedures for PS to PS+CS access transfer not using ICS with MSC server assisted mid-call feature

In addition to the procedures described in subclause 11.2.1.3 the SC UE shall:

- act as described in subclause 9.2.1A; and

- if the MSC server assisted mid-call feature is applied, transfer the non-speech media components of the additional transferred session to the PS Target Access Leg as specified for PS-PS access transfer with partial media transfer in subclause 10.2.2.

11.2.2 SC UE procedures for PS+CS to PS access transfer

11.2.2.1 General

The SC UE may be engaged in one or more ongoing sessions before performing access transfer. By an ongoing session, it is meant a CS call for which the CC CONNECT message has been sent or received or a call for which the SIP 2xx response for the initial SIP INVITE request to establish this session has been sent or received.

If not already registered over the PS Target Access Leg, the SC UE shall follow the procedures specified in subclause 6.2 to perform IM CN subsystem registration over the Target Access Leg before performing PS/CS to PS access transfer.

11.2.2.2 SC UE procedures for PS+CS to PS access transfer using ICS

If SC using ICS is enabled then if the original sessions are established using ICS capabilities as defined in 3GPP TS 24.292 [4], then for each full-duplex session with speech media component to be transferred and starting with the session with active speech media component, the SC UE shall send a SIP INVITE request to the SCC AS in accordance with the UE procedures specified in 3GPP TS 24.229 [2]. The SC UE shall populate the SIP INVITE request as specified for PS-PS access transfer with full media transfer in subclause 10.2.1. The SC UE shall indicate in the SIP INVITE request that the speech media component is using PS media.

Upon receiving SIP BYE request for the Source Access Leg, the SC UE shall follow the ICS using Gm procedures specified in 3GPP TS 24.292 [4] to release the session. The SC UE also releases the associated CS bearer if no other sessions depend on the CS bearer.

11.2.2.3 SC UE procedures for PS+CS to PS access transfer not using ICS

If the original sessions are not established using ICS capabilities, then access transfer is only possible when the SC UE has a single session with active full-duplex speech media component. The SC UE shall send a SIP INVITE request to the SCC AS in accordance with the UE procedures specified in 3GPP TS 24.229 [2].

The SC UE shall populate the SIP INVITE request as follows:

- the Request-URI set to static STI;
- the Require header field including "replaces" option tag;
- the Replaces header field populated as specified in IETF RFC 3891 [10], containing the dialog identifier of the session to be transferred on the PS Source Access Leg; and
- the SDP payload set for the media component(s) to be transferred, in accordance the UE SDP origination procedures specified in 3GPP TS 24.229 [2]. The SC UE shall create an SDP offer that contains media components in the following order:

    1) the same number of media lines, each corresponding to one of the media components in the session on the PS Source Access Leg; For each media line the SC UE shall indicate the same media type as its corresponding media component in the original session and indicate at least one codec that was negotiated during the original session. If the SC UE determines to remove a media component during the transfer, then the SC UE shall set the media line for this media component to include a port number with value zero;

    2) one speech media component to be transferred, corresponding to the speech media component in the session on the CS Source Access Leg; and

    3) if the SC UE determines to add new media component(s) during the transfer, then one additional media line with the desired media type and codecs each new media component.

If the SC UE receives any SIP 4xx – 6xx response to the SIP INVITE request, then session transfer has not occurred and the call will continue in the original domains.

11.3 SCC AS

11.3.1 Distinction of requests sent to the SCC AS

The SCC AS needs to distinguish between the following initial SIP INVITE requests to provide specific functionality relating to access transfer:

- SIP INVITE requests routed to the SCC AS containing a STI belonging to the subscribed user in the Replaces header field or Target-Dialog header field and not containing Inter UE Transfer SCC AS URI in the Request-URI. In the procedures below, such requests are known as "SIP INVITE requests due to STI".

- SIP INVITE requests routed to the SCC AS containing either a dynamic STN, a static STN or an IMRN in the Request-URI. In the procedures below, such requests are known as "SIP INVITE requests due to PS to CS STN".

- SIP INVITE requests routed to the SCC AS containing a static STI in the Request-URI and a STI in the Replaces or Target-Dialog header field. In the procedures below, such requests are known as "SIP INVITE requests due to two STIs".

NOTE: The media streams that need to be transferred are identified using information described in the subsequent subclauses 11.3.2 and 11.3.3.

Other SIP initial requests for a dialog and requests for a SIP standalone transaction can be dealt with in any manner conformant with 3GPP TS 24.229 [2].

11.3.2 SCC AS procedures for PS to PS+CS access transfer

This subclause does not apply to reception of a SIP INVITE request due to STI with a CS media.

When the SCC AS receives a SIP INVITE request due to STI with PS and CS media on the Target Access Leg, the SCC AS shall follow the PS-PS Access Transfer procedures specified in subclause 10.3.2. with the following exceptions:

If the SIP INVITE request includes an active speech media component using CS bearer, then the SCC AS shall follow the procedures for SCC AS for service control over Gm in 3GPP TS 24.292 [4] to send the PSI DN to the SC UE and wait for the SC UE to set up CS bearer before sending a SIP re-INVITE request to the remote UE.

- The SCC AS shall correlate the STI with the allocated PSI DN in order to identify the remote leg to be updated.

- If service control over Gm is retained on the source access leg, and the SCC AS receives a SIP re-INVITE request indicating CS bearer on an existing session, the SCC AS shall follow procedures as described in 3GPP TS 24.292 [4] subclause 8.4.2 to send the PSI DN to the SC UE and wait for the SC UE to set up CS bearer before sending a SIP re-INVITE request to the remote end.

- The SCC AS shall include a new SDP offer in the SIP re-INVITE request, following the rules specified in 3GPP TS 24.229 [2], containing the following media information:

  - the media characteristics as received in the SIP INVITE request due to STI with PS+CS media received on the Target Access Leg for media streams whose port is not set to zero; and

  - the media characteristics as received in the SIP re-INVITE request for media streams whose port is not set to zero.

When the SCC AS receives a SIP INVITE request due to PS to CS STN on the Target Access Leg, the SCC AS shall follow the PS-CS Access Transfer procedures specified in subclause 9.3.2. However, as the Source Access Leg contains media components other than speech media component, the SCC AS does not initiate release for Source Access Leg.

11.3.3 SCC AS procedures for PS+CS to PS access transfer

This subclause applies to reception of a SIP INVITE request due to STI with a PS media only.

When the SCC AS receives a SIP INVITE request due to STI on the Target Access Leg, the SCC AS shall follow the PS-PS access transfer procedures specified in subclause 10.3.2.

When the SCC AS receives a SIP INVITE request due to two STIs on the Target Access Leg, the SCC AS shall:

- associate the SIP INVITE request received on the Target Access Leg with two ongoing sessions:

  a) an ongoing SIP dialog on the PS Source Access Leg: This is done by matching the dialog ID present in the Replaces header field (see IETF RFC 3891 [10]) or Target-Dialog header field (see IETF RFC 4538 [11]) of the SIP INVITE request with an ongoing dialog. By an ongoing SIP dialog, it is meant a dialog for which a SIP 2xx response to the initial SIP INVITE request has been sent or received; and

  b) a different ongoing SIP dialog with active speech media component;

- if the SCC AS is unable to associate the SIP INVITE request with either one of the above two dialogs, send a SIP 480 (Temporarily Unavailable) response to reject the SIP INVITE request relating to the access transfer and not process the remaining steps; and

- if the session transfer is possible:

  a) follow the procedures defined in IETF RFC 3891 [10] for replacing the two sessions on the Source Access Legs with the SIP request received on the Target Access Leg, including terminating the two Source Access Legs by sending a SIP BYE request on each session towards the SC UE in accordance with 3GPP TS 24.229 [2]; and

  b) send a SIP re-INVITE request towards the remote UE using the existing established dialog. The SCC AS shall populate the SIP re-INVITE request as follows:

     1) set the Request-URI to the URI contained in the Contact header field returned at the creation of the dialog with the remote UE; and

     2) a new SDP offer, including the media characteristics as received in the SIP INVITE request due to two STIs received on the Target Access Leg, by following the rules of 3GPP TS 24.229 [2].

12 Roles for PS-CS access transfer, Single Radio

12.1 Introduction

This clause specifies the procedures for PS-CS SRVCC and vSRVCC access transfer. Procedures are specified for the SC UE, the SCC AS, the EATF, the MSC server enhanced for ICS, the MSC server enhanced for SRVCC and the ATCF.

For SC UE or SCC AS not supporting ICS procedures, PS-CS SR-VCC access transfer enables transfer of

- single session with active speech media component; and

- up to one session with active speech media component and up to one session with inactive speech media component when the MSC Server assisted mid-call feature is supported.

For SC UE or SCC AS, PS-CS access transfer in vSRVCC enables the transfer of a single session with active speech and video media components.

In order to fulfil the requirements for the PS to CS SRVCC for calls in alerting phase or vSRVCC access transfer for calls in alerting phase, the SC UE needs to be:

- engaged in a session with speech media component in early dialog state before PS to CS SRVCC access transfer is performed; or

- engaged in a session with active speech media component and active video media component in early dialog state before vSRVCC access transfer is performed

according to the following conditions:

- a SIP 180 (Ringing) response for the initial SIP INVITE request to establish this session has been sent or received; and

- a SIP final response for the initial SIP INVITE request to establish this session has not been sent or received.

If one of the dialogs meets the above conditions then after successful completion of the PS to CS SRVCC procedures:

- Subclauses 12.2.2, 12.2.3, 12.2.3A and 12.2.4 shall be followed for a SC UE engaged in one or more ongoing sessions.
- Subclauses 12.2.3B and 12.2.4 shall be followed for a SC UE that is engaged in a session in early dialog state.

If one of the dialogs meets the above conditions then after successful completion of the vSRVCC procedures:

- Subclauses 12.2A.2, 12.2A.3, 12.2A.4 and 12.2A.6 shall be followed for a SC UE engaged in one or more ongoing sessions.
- Subclauses 12.2A.5 and 12.2A.6 shall be followed for a SC UE that is engaged in a session in early dialog state.

NOTE: The UE determines from the handover command sent by the eNodeB as specified in 3GPP TS 36.331 [62] that the network intends to perform SRVCC handover or vSRVCC handover, based upon whether the radio resources allocated are for a TS11 bearer or a BS30 bearer.

12.2 SC UE procedures for PS to CS access transfer, PS to CS SRVCC

12.2.1 General

The SC UE may be engaged in one or more ongoing sessions before PS to CS SRVCC access transfer is performed. By an ongoing session, it is meant a session for which the SIP 2xx response for the initial SIP INVITE request to establish this session has been sent or received.

In the PS to CS SRVCC session continuity procedures the SC UE shall consider only sessions where the following applies

1. the SC UE has completed a reliable offer / answer procedure and the session does have a speech media component; and
2. the speech media is carried over PS bearer with traffic-class conversation with source statistics descriptor ="speech" as specified in 3GPP TS 23.107 [66] or over a PS bearer with QCI=1 as specified in 3GPP TS 23.203 [65].

for access transfer. Sessions considered for PS to CS SRVCC procedures are regarded as full-duplex.

12.2.2 ICS-based

If:

- SC using ICS is enabled;
- the Gm reference point is retained upon PS handover procedure;
- the SC UE is using ICS capabilities as defined in 3GPP TS 24.292 [4]; and
- PS to CS SRVCC procedures (as described in 3GPP TS 24.008 [8]) have been completed;

the SC UE, in order to add Gm control for the newly established CS session, shall:

- send a SIP re-INVITE request for each session with speech media component to be transferred, starting with the session with active speech media component that was most recently made active; and
- within the SDP offer indicate the media line for the speech media component (active or held) as an speech media component over circuit switched bearer in accordance with 3GPP TS 24.292 [4]. If the precondition mechanism is used, the SC UE shall indicate the related local preconditions as met.

NOTE: Within PS to CS SRVCC the handover is performed on PS level. Due to this, the SIP dialog established over the source PS access network stays the same after PS to CS SRVCC procedures, e.g. the IP address of the UE, the Call-ID or the P-CSCF do not change. Therefore in this case a SIP re-INVITE request needs to be sent to add ICS-control for the CS bearer.

12.2.3 Not based on ICS

After successful PS to CS SRVCC procedures (as described in 3GPP TS 24.008 [8]) have been completed, if the SC UE is not using ICS capabilities and the SC UE does not apply the MSC Server assisted mid-call feature as specified in subclause 12.2.3A, the SC UE shall replace the ongoing session with active speech media component which was made active most recently with the newly established CS voice call.

NOTE 1: In the case when ICS is not supported or used and the SC UE does not apply the MSC Server assisted mid-call feature, only the ongoing session with active speech media component which was made active most recently is transferred from PS to CS audio.

If the Gm reference point is:

1) retained upon successful PS handover completion;

NOTE 2: The SC UE knows that the Gm reference point is retained upon PS handover if, following handover, the SC UE has a dedicated PDP context for SIP signalling or has a general-purpose PDP context to carry the IM CN subsystem-related signalling, as described in 3GPP TS 24.229 [2] subclause B.2.2.1.

a) and there are one or more remaining non-speech media component(s) in the IMS session other than the speech media component which were transferred to the CS Target Access Leg; the SC UE shall:

- send a SIP re-INVITE request to the SCC AS as specified for media removal in subclause 13.2.1; and
- indicate in the SDP offer the speech media component as removed;

b) and there are no more non-speech media component(s) remaining in the IMS session other than the speech media component which was transferred to the CS Target Access Leg; the SC UE shall either:

- send a SIP re-INVITE request to the SCC AS as specified for media removal in subclause 13.2.1 indicating in the SDP offer the speech media component as removed;
- wait for a period of time for a SIP BYE request to be received before clearing the SIP dialog state internally; or
- clear the SIP dialog state internally; or

2) not retained upon successful PS handover completion the SC UE shall clear the SIP dialog state internally.

NOTE 3: If a SIP BYE request is received after the UE has cleared the SIP dialog state internally the UE will send a SIP 481 (Call/Transaction Does Not Exist) response according to RFC 3261 [19].

12.2.3A Not based on ICS with MSC Server assisted mid-call feature

After successful PS to CS SRVCC procedures (as described in 3GPP TS 24.008 [8]), if:

1. the SC UE is not using ICS capabilities;
2. the SC UE supports the MSC Server assisted mid-call feature; and
3. one of the following is true:

A. there is at least one ongoing session with active speech media component and the Feature-Caps header field received by the SC UE at the establishment of the ongoing session with active speech media component, which has been most recently made active, includes the g.3gpp.mid-call feature-capability indicator as described in annex C; or

B. there is no ongoing session with active speech media component and:

- there is no emergency session in early dialog state with active speech media component; or
- there is an emergency session in early dialog state with active speech media component, unless the UE supports the PS to CS SRVCC for emergency session in early dialog state with active speech media component when both the emergency session in early dialog state with active speech media component and a non-emergency call in confirmed dialog state with inactive speech media component exists;

and the Feature-Caps header field received by the SC UE at the establishment of the ongoing session with inactive speech media component which became inactive most recently includes the g.3gpp.mid-call feature-capability indicator as described in annex C;

then the SC UE shall apply the MSC Server assisted mid-call feature as follows:

1. if two or more ongoing sessions with active speech media component exist, the SC UE shall:

   A) replace the speech media components of the ongoing session with active speech media component which was most recently made active with the newly established active CS voice call; and

   B) replace the speech media component of the ongoing session with active speech media component which was made active second most recently with the newly established held CS voice call;

2. if one ongoing session with active speech media component exists and one or more ongoing sessions with inactive speech media component exist, the SC UE shall:

   A) replace the speech media components of the ongoing session with active speech media component with the newly established active CS voice call; and

   B) replace the speech media component of the ongoing session with inactive speech media component which was most recently made inactive with the newly established held CS voice calls;

3. if one ongoing session with active speech media component exists and no ongoing sessions with inactive speech media component exist, the SC UE shall replace the speech media component of the ongoing session with active speech media component with the newly established active CS voice call; and

4. if no ongoing session with active speech media component exists and one or more ongoing sessions with inactive speech media component exist, the SC UE shall replace the speech media component of the ongoing session with inactive speech media component which became inactive most recently with the newly established held CS voice call.

For each session, the SC UE shall proceed as specified in subclause 12.2.3.

If two sessions are transferred, and the SC UE does not have a subscription as described in subclause 7.2.2 for the additional transferred session the SC UE shall associate the additional transferred session with CS call:

- with transaction identifier 1; and

- with TI flag value as in mobile terminated call;

and the SC UE shall enter the "active" (U10) state (defined in 3GPP TS 24.008 [8]), the "call held" auxiliary state (defined in 3GPP TS 24.083 [43]), and the "idle" multi party auxiliary state (defined in 3GPP TS 24.084 [47]) for the CS call.

NOTE 1: The session with active speech media call state and component transaction identifier value are described in 3GPP TS 24.008 [8].

If single session with inactive speech media component is transferred, and the SC UE does not have a subscription as described in subclause 7.2.2 for the transferred session, then the SC UE shall associate the transferred session with CS call:

- with transaction identifier 0; and

- with TI flag value as in mobile terminated call;

and the SC UE shall enter the "active" (U10) state (defined in 3GPP TS 24.008 [8]), the "call held" auxiliary state (defined in 3GPP TS 24.083 [43]), and the "idle" multi party auxiliary state (defined in 3GPP TS 24.084 [47]) for the CS call.

If a transferred session is a session with a conference focus and the SC UE has a subscription as described in subclause 7.2.2 for the ongoing full-duplex session with active speech media component then:

1. the SC UE shall associate the transferred session and the participants extracted in subclause 9.1A with CS calls:

- with transaction identifiers 0, 2, 3, 4, 5 assigned to the participants in their order in the list of the extracted participants; and
- with TI flag value as in mobile terminating call; and

2. the SC UE shall enter the "active" (U10) state (defined in 3GPP TS 24.008 [8]), the "idle" hold auxiliary state (defined in 3GPP TS 24.083 [43]), and the "call in MPTY" multi party auxiliary state (defined in 3GPP TS 24.084 [47]) for the CS calls.

NOTE 2: The transaction identifier, TI flag value and "active" (U10) state for the first participant are assigned by the call activation procedures for SRVCC in 3GPP TS 24.008 [8].

If a transferred session is a session with a conference focus and the ongoing full-duplex session with active speech media component does not exist and the SC UE has a subscription as described in subclause 7.2.2 for the ongoing full-duplex session with inactive speech media component the SC UE shall associate the transferred session and the participants extracted in subclause 9.1A with CS calls:

- with transaction identifiers 0, 2, 3, 4, 5 assigned to the participants in their order in the list of the extracted participants; and
- with TI flag value as in mobile terminating call;

and the SC UE shall enter the "active" (U10) state (defined in 3GPP TS 24.008 [8]), the "call held" hold auxiliary state (defined in 3GPP TS 24.083 [43]), and the "call in MPTY" multi party auxiliary state (defined in 3GPP TS 24.084 [47]) for the CS calls.

NOTE 3: The transaction identifier, TI flag value and "active" (U10) state for the first participant are assigned by the call activation procedures for SRVCC in 3GPP TS 24.008 [8].

If a transferred session is a session with a conference focus and:

1. the ongoing full-duplex session with active speech media component exists and the SC UE does not have a subscription as described in subclause 7.2.2 for the ongoing full-duplex session with active speech media component; and
2. the SC UE has a subscription as described in subclause 7.2.2 for the additional transferred session;

then:

1. the SC UE shall associate the transferred session and the participants extracted in subclause 9.1A with CS calls:
   - with transaction identifiers 1, 2, 3, 4, 5 assigned to the participants in their order in the list of the extracted participants; and
   - with TI flag value as in mobile terminating call; and
2. the SC UE shall enter the "active" (U10) state (defined in 3GPP TS 24.008 [8]), the "call held" hold auxiliary state (defined in 3GPP TS 24.083 [43]), and the "call in MPTY" multi party auxiliary state (defined in 3GPP TS 24.084 [47]) for the CS calls.

12.2.3B Call in alerting phase

12.2.3B.1 General

The SC UE shall apply the procedures in subclauses 12.2.3B.3.1 and 12.2.3B.3.2 for the PS to CS SRVCC for calls in alerting phase if:

1) the SC UE supports the PS to CS SRVCC for calls in alerting phase; and
2) one of the following is true:
   A. there are one or more dialogs supporting sessions with speech media component, such that:
      - all dialogs are early dialogs;

- SIP 180 (Ringing) response to SIP INVITE request was received in at least one of those early dialogs, all such SIP 180 (Ringing) responses are responses to the same SIP INVITE request and at least one of such SIP 180 (Ringing) responses was received in an early dialog supporting session with active speech media component;
- the SC UE has sent a Contact header field containing the g.3gpp.srvcc-alerting media feature tag (as described in annex C); and
- the SC UE has received a Feature-Caps header field containing the g.3gpp.srvcc-alerting feature-capability indicator (as described in annex C); or

B. there are one or more dialogs supporting a session with speech media component such that:

- there are one or more early dialogs and the remaining dialogs are confirmed dialogs;
- SIP 180 (Ringing) response to SIP INVITE request was received in at least one of those early dialogs, all such SIP 180 (Ringing) responses are responses to the same SIP INVITE request and at least one of such SIP 180 (Ringing) responses was received in an early dialog supporting session with active speech media component;
- all the confirmed dialogs support sessions with inactive speech media component;
- SC UE does not apply the MSC server assisted mid-call feature as described in subclause 12.2.3A;
- in those early dialogs, the SC UE has sent a Contact header field containing the g.3gpp.srvcc-alerting media feature tag (as described in annex C); and
- in those early dialogs, the SC UE has received a Feature-Caps header field containing the g.3gpp.srvcc-alerting feature-capability indicator (as described in annex C); or

C. there is one dialog supporting session with speech media component, such that:

- the dialog is an early dialog;
- SIP 180 (Ringing) response to SIP INVITE request was sent in the early dialog supporting session with active speech media component;
- the SC UE has sent a Contact header field containing the g.3gpp.srvcc-alerting media feature tag (as described in annex C); and
- the SC UE has received a Feature-Caps header field containing the g.3gpp.srvcc-alerting feature-capability indicator (as described in annex C); or

D. there are one or more dialogs supporting a session with speech media component such that:

- there is one early dialog and the remaining dialogs are confirmed dialogs;
- SIP 180 (Ringing) response to SIP INVITE request was sent in the early dialog supporting session with active speech media component;
- all the confirmed dialogs support sessions with inactive speech media component;
- SC UE does not apply the MSC server assisted mid-call feature as described in subclause 12.2.3A;
- in those early dialogs, the SC UE has sent a Contact header field containing the g.3gpp.srvcc-alerting media feature tag (as described in annex C); and
- in those early dialogs, the SC UE has received a Feature-Caps header field containing the g.3gpp.srvcc-alerting feature-capability indicator (as described in annex C).

The SC UE shall apply the procedures in subclauses 12.2.3B.4.1 for the PS to CS SRVCC for calls in alerting phase if:

1) the SC UE supports the PS to CS SRVCC for calls in alerting phase;

2) one of the following is true:

A) there are two or more dialogs supporting more than one session with speech media component, such that:

a) the SC UE has sent a Contact header field containing the g.3gpp.srvcc-alerting media feature tag (as described in annex C);

b) the SC UE has received a Feature-Caps header field containing the g.3gpp.srvcc-alerting feature-capability indicator (as described in annex C).

c) one dialog supporting a session in the confirmed state with active speech media component and remaining dialogs are early dialogs; and

d) SIP 180 (Ringing) response to SIP INVITE request was received in at least one of those early dialogs, all such SIP 180 (Ringing) responses are responses to the same SIP INVITE request and at least one of such SIP 180 (Ringing) responses was received in an early dialog supporting session with active speech media component; or

B) there are two or more dialogs supporting sessions with speech media component, such that:

a) the SC UE has sent a Contact header field containing the g.3gpp.srvcc-alerting media feature tag (as described in annex C); and

b) the SC UE has received a Feature-Caps header field containing the g.3gpp.srvcc-alerting feature-capability indicator (as described in annex C);

c) one or more dialogs are confirmed dialogs supporting sessions with active speech media components, there are one or more dialogs that are confirmed dialogs with inactive speech media component and remaining dialogs are early dialogs;

d) SIP 180 (Ringing) response to SIP INVITE request was received in at least one of those early dialogs, all such SIP 180 (Ringing) responses are responses to the same SIP INVITE request and at least one of such SIP 180 (Ringing) responses was received in an early dialog supporting session with active speech media component; and

e) the SC UE does not apply the MSC server assisted mid-call feature as described in subclause 12.2.3A.

The SC UE shall apply the procedures in subclauses 12.2.3B.3.3 if one of the following is true:

1) there are zero, one or more dialogs supporting a session with speech media component and a SIP INVITE request was sent by SC UE such that:

A) all dialogs are early dialogs created by a SIP response to the SIP INVITE request;

B) a final SIP response to the SIP INVITE request has not been received yet;

C) a SIP 180 (Ringing) response to the SIP INVITE request has not been received yet in any existing early dialog created by a SIP response to the SIP INVITE request;

D) the SC UE included in the SIP INVITE request a Contact header field containing the g.3gpp.ps2cs-srvcc-orig-pre-alerting media feature tag as described in annex C; and

E) a SIP 1xx response to the SIP INVITE request was received where the SIP 1xx response contained a Feature-Caps header field with the g.3gpp.ps2cs-srvcc-orig-pre-alerting feature-capability indicator as described in annex C; or

NOTE: The SC UE can have zero dialogs if all the early dialogs were terminated by 199 (Early Dialog Terminated) as described in RFC 6228 [80].

2) there are one or more dialogs supporting a session with speech media component such that:

A) there are zero, one or more early dialogs and the remaining dialogs are confirmed dialogs;

B) all the confirmed dialogs support sessions with inactive speech media component;

C) the UE does not apply the MSC server assisted mid-call feature according to subclause 12.2.3A;

D) a SIP INVITE request was sent by SC UE such that:

a) all early dialogs are created by a SIP response to the SIP INVITE request;

b) a final SIP response to the SIP INVITE request has not been received yet;

c) a SIP 180 (Ringing) response to the SIP INVITE request has not been received yet in any existing early dialog created by a SIP response to the SIP INVITE request;

d) the SC UE included in the SIP INVITE request a Contact header field containing the g.3gpp.ps2cs-srvcc-orig-pre-alerting media feature tag as described in annex C; and

e) a SIP 1xx response to the SIP INVITE request was received where the SIP 1xx response contained a Feature-Caps header field with the g.3gpp.ps2cs-srvcc-orig-pre-alerting feature-capability indicator as described in annex C.

The SC UE shall apply the procedures in subclauses 12.2.3B.3.4 if it supports PS to CS SRVCC for terminating calls in pre-alerting phase and if one of the following is true:

1) there is one dialog supporting a session with speech media component and a SIP INVITE request was received by SC UE such that:

   A) the dialog is an early dialog created by a SIP response to the SIP INVITE request;

   B) a final SIP response to the SIP INVITE request has not been sent yet;

   C) a SIP 180 (Ringing) response to the SIP INVITE request has not been sent yet;

   D) the SC UE has sent a SIP 1xx response to the SIP INVITE request with a Contact header field containing the g.3gpp.ps2cs-srvcc-term-pre-alerting media feature tag (as described in annex C); and

   E) the SIP INVITE request contained a Feature-Caps header field with the g.3gpp.ps2cs-srvcc-term-pre-alerting feature-capability indicator as described in annex C; or

2) there are one or more dialogs supporting a session with speech media component such that:

   A) there is one early dialog and the remaining dialogs are confirmed dialogs;

   B) all the confirmed dialogs support sessions with inactive speech media component;

   C) the UE does not apply the MSC server assisted mid-call feature according to subclause 12.2.3A;

   D) a SIP INVITE request was received by SC UE such that:

      a) the early dialog is created by a SIP response to the SIP INVITE request;

      b) a final SIP response to the SIP INVITE request has not been sent yet;

      c) a SIP 180 (Ringing) response to the SIP INVITE request has not been sent yet;

      d) the SC UE has sent a SIP 1xx response to the SIP INVITE request with a Contact header field containing the g.3gpp.ps2cs-srvcc-term-pre-alerting media feature tag (as described in annex C); and

      e) the SIP INVITE request contained a Feature-Caps header field with the g.3gpp.ps2cs-srvcc-term-pre-alerting feature-capability indicator as described in annex C.

12.2.3B.1A Considerations for MSC server assisted mid-call feature

If the SC UE supports both the PS to CS SRVCC for calls in alerting phase and the MSC server assisted mid-call feature then in addition to supporting the procedures specified in subclauses 12.2.3B.3 and 12.2.3B.4.1, it shall apply the procedures specified in subclause 12.2.3B.4.2 where one of the following is true:

1) there are two or more dialogs supporting sessions with speech media component, such that:

   a) the SC UE has sent a Contact header field containing the g.3gpp.srvcc-alerting media feature tag (as described in annex C);

   b) the SC UE has received a Feature-Caps header field containing the g.3gpp.srvcc-alerting feature-capability indicator (as described in annex C);

c) one or more dialogs are in the confirmed state supporting a session with inactive speech media component and the remaining dialog(s) are early dialog(s);

d) SIP 180 (Ringing) response to SIP INVITE request was received in at least one of those early dialogs, all such SIP 180 (Ringing) responses are responses to the same SIP INVITE request and at least one of such SIP 180 (Ringing) responses was received in an early dialog supporting session with active speech media component; and

e) the SC UE applies the MSC server assisted mid-call feature according to subclause 12.2.3A;

2) there are one or more dialogs supporting sessions with speech media component according to the following conditions:

A) there are zero, one or more early dialogs and the remaining dialog(s) are confirmed dialog(s);

B) all the confirmed dialogs support sessions with inactive speech media component;

C) the UE applies the MSC server assisted mid-call feature according to subclause 12.2.3A;

D) a SIP INVITE request was sent by SC UE such that:

a) all the early dialogs are created by a SIP response to the SIP INVITE request;

b) a final SIP response to the SIP INVITE request has not been received yet;

c) a SIP 180 (Ringing) response to the SIP INVITE request has not been received yet in any existing early dialog created by a SIP response to the SIP INVITE request;

d) the SC UE included in the SIP INVITE request a Contact header field containing the g.3gpp.ps2cs-srvcc-orig-pre-alerting media feature tag as described in annex C; and

e) a SIP 1xx response to the SIP INVITE request was received where the SIP 1xx response contained a Feature-Caps header field with the g.3gpp.ps2cs-srvcc-orig-pre-alerting feature-capability indicator as described in annex C.

12.2.3B.2 Assignment of Transaction Identifiers to the transferred sessions

If the SC UE applies the procedures in subclause 12.2.3B.3 and the SC UE only has a single call:

- in alerting phase following access transfer;
- in pre-alerting phase and the SC UE supports the PS to CS SRVCC for originating calls in pre-alerting phase; or
- in pre-alerting phase and the SC UE supports the PS to CS SRVCC for terminating calls in pre-alerting phase.

the SC UE shall associate this session with transaction identifier value and TI flag as described in 3GPP TS 24.008 [8].

If the SC UE applies the procedures in subclause 12.2.3B.4 and the SC UE has an established session and an additional session in alerting phase or pre-alerting phase following access transfer, then the SC UE shall associate the transferred session that was in alerting phase or pre-alerting phase with CS call with transaction identifier 1 and TI flag value as in mobile terminated call.

NOTE: For the procedures in subclause 12.2.3B.4.2, the held transaction identifier value is described in subclause 12.2.3A as for single inactive session transfer and the active session transaction identifier value is described in 3GPP TS 24.008 [8].

12.2.3B.3 Single call in alerting phase or pre-alerting phase

12.2.3B.3.1 Terminating call in alerting phase

If the SC UE:

- has received a terminating call which is in the early dialog state according to the conditions in subclauses 12.1 and 12.2.3B.1; and
- successfully performs access transfer to the CS domain;

then the UE continues in Ringing state in CS, i.e. UE moves to Call Received (U7) state as described in 3GPP TS 24.008 [8].

If the SC UE:

- has received a terminating call which is in the early dialog state according to the conditions in subclauses 12.1 and 12.2.3B.1; and

- has sent a SIP 200 (OK) response (i.e. user answers the call when in the PS domain) prior to successfully performing access transfer to the CS domain;

then the UE sends a CC CONNECT message and will finally transitions to Active (U10) state as described in 3GPP TS 24.008 [8].

12.2.3B.3.2 Originating call in alerting phase

If the SC UE has initiated an outgoing call which is in the early dialog state according to the conditions in subclauses 12.1 and 12.2.3B.1 and the SC UE successfully performs access transfer to the CS domain, then the UE continues in Ringing state in CS, i.e. UE moves to Call Delivered (U4) state as described in 3GPP TS 24.008 [8]. If the UE has received a SIP 180 (Ringing) response, depending on the type of the ringing tone, the UE behaves as following:

- if the SC UE is playing the locally generated ringing tone, then the UE keeps playing the locally generated ringing tone; and

- if the SC UE is playing network-generated ringing tone as early media, then the UE attaches the user connection to the MSC server, as specified in 3GPP TS 24.008 [8].

12.2.3B.3.3 PS to CS SRVCC for originating calls in pre-alerting phase

If the SC UE supports the PS to CS SRVCC for originating calls in pre-alerting phase and this subclause is invoked according to the conditions in subclause 12.2.3B.1 and the SC UE successfully performs access transfer to the CS domain, then the UE continues the call in the CS domain in the "Mobile originating call proceeding" (U3) call state as described in 3GPP TS 24.008 [8].

If the SC UE has generated and rendered the locally generated communication progress information before the access transfer to the CS domain, the UE keeps generating and rending the locally generated communication progress information after the access transfer to the CS domain.

If the SC UE has rendered received early media before the access transfer to the CS domain, the UE attaches the user connection, as specified in 3GPP TS 24.008 [8].

12.2.3B.3.4 PS to CS SRVCC for terminating calls in pre-alerting phase

If the SC UE supports the PS to CS SRVCC for terminating calls in pre-alerting phase and this subclause is invoked according to the conditions in subclause 12.2.3B.1 and the SC UE successfully performs access transfer to the CS domain, then the UE continues the call in the CS domain in the "Call present" (U6) call state as described in 3GPP TS 24.008 [8].

12.2.3B.4 Established call with a session in alerting phase or in pre-alerting phase

12.2.3B.4.1 Active session with incoming call in alerting phase

If the SC UE:

- has a session with an active speech media component and has received an incoming call (waiting) which is in the early dialog state according to the conditions in subclauses 12.1 and 12.2.3B.1; and

- successfully performs access transfer to the CS domain;

then the UE moves to Call Received (U7) state (defined in 3GPP TS 24.008 [8]) for the incoming call (waiting) (i.e. continues in Ringing state in CS for the incoming call waiting).

12.2.3B.4.2 Held session with new outgoing call in alerting phase or in pre-alerting phase

If the SC UE:

- has a session with an inactive speech media component and has initiated a new outgoing call which is in the early dialog state according to the conditions in subclauses 12.1 and 12.2.3B.1; and

- successfully performs access transfer to the CS domain;

then:

- if the new outgoing call is in alerting phase, the UE moves to Call Delivered (U4) state (defined in 3GPP TS 24.008 [8]) for the new outgoing call (i.e. UE continues in Ringing state in CS for the outgoing call);

- if the new outgoing call is in pre-alerting phase, the UE moves to Mobile originating call proceeding (U3) state (defined in 3GPP TS 24.008 [8]) for the new outgoing call; and

- the UE moves to Call Active (U10) state (defined in 3GPP TS 24.008 [8]) and Call Held Auxiliary State (defined in 3GPP TS 24.083 [43]) for the held call.

12.2.4 Abnormal cases

12.2.4.1 Confirmed dialog

If the SC UE engaged in one or more ongoing IMS sessions and:

- receives a SM NOTIFICATION message containing an "SRVCC handover cancelled, IMS session re-establishment required" as described in 3GPP TS 24.008 [8] or 3GPP TS 24.301 [52] depending on the access in use; or

- does not successfully retune to the 3GPP UTRAN or 3GPP GERAN after it receives the handover command from the eNodeB (as described in 3GPP TS 36.331 [62]) or from the NodeB (as described in 3GPP TS 25.331 [61]);

then the SC UE shall send a SIP re-INVITE request containing:

1) an SDP offer, including the media characteristics as used in the existing dialog; and

2) a Reason header field containing protocol "SIP" and reason parameter "cause" with value "487" as specified in IETF RFC 3326 [57] and with reason-text text set to either "handover cancelled" or "failure to transition to CS domain";

by following the rules of 3GPP TS 24.229 [2] in each transferred session.

12.2.4.2 Early dialog

If the SC UE is engaged in a session in early dialog state and:

- receives a SM NOTIFICATION message containing an "SRVCC handover cancelled, IMS session re-establishment required" as described in 3GPP TS 24.008 [8] or 3GPP TS 24.301 [52] depending on the access in use; or

- does not successfully retune to the 3GPP UTRAN or 3GPP GERAN after it receives the handover command from the eNodeB (as described in 3GPP TS 36.331 [62]) or from the NodeB (as described in 3GPP TS 25.331 [61]);

then the SC UE shall:

a) send a SIP UPDATE request containing:

   1) an SDP offer, including the media characteristics as used in the existing dialog; and

   2) a Reason header field containing protocol "SIP" and reason parameter "cause" with value "487" as specified in IETF RFC 3326 [57], and with reason-text set to either "handover cancelled" or "failure to transition to CS domain";

   by following the rules of 3GPP TS 24.229 [2] in each transferred session; and

b) if the SC UE is a terminating side UE and has already sent a CC CONNECT on the target access leg, send a SIP 200 (OK) response to the SIP INVITE request received on the source access leg.

12.2.4.3 Moving from a 3GPP access to non-3GPP access colliding with SRVCC access transfer

If the SC UE engaged in one or more IMS sessions receives a handover command from the eNodeB (as described in 3GPP TS 36.331 [62]) or from the NodeB (as described in 3GPP TS 25.331 [61]) when the UE is engaged in moving the PDN connection from an 3GPP access to a non-3GPP access as described in 24.302 [84], the UE shall either:

1) accept the handover command and abort the move to the non-3GPP access; or

2) continue with the move to the non-3GPP access and ignore the handover command.

If the SC UE continues with the move to the non-3GPP access, the SC UE shall for each transferred session send a SIP re-INVITE request containing:

1) an SDP offer, including the media characteristics as used in the existing dialog; and

2) a Reason header field containing protocol "SIP" and reason parameter "cause" with value "487" as specified in IETF RFC 3326 [57];

NOTE: The reason-text text in the Reason header field can be set to either "handover cancelled" or "failure to transition to CS domain".

by following the rules of 3GPP TS 24.229 [2].

12.2A SC UE procedures for PS to CS access transfer, vSRVCC

12.2A.1 General

The SC UE may be engaged in one or more ongoing sessions before vSRVCC access transfer is performed. By an ongoing session, it is meant a session for which the response to the initial SIP INVITE request to establish this session has been sent or received.

In the vSRVCC session continuity procedures the SC UE shall consider only sessions where the following applies:

1. the SIP dialog contains speech and video media components supporting a real time video session synchronized with speech as defined in 3GPP TS 22.173 [24], which includes the codecs for speech and video as specified in 3GPP TS 26.114 [68]; and

2. the speech media is carried over a PS bearer with QCI=1 as specified in 3GPP TS 23.203 [65];

for access transfer. Sessions considered for vSRVCC procedures are regarded as full-duplex.

12.2A.2 ICS-based

If:

- SC using ICS is enabled;

- the Gm reference point is retained upon PS handover procedure;

NOTE 1: The SC UE knows that the Gm reference point is retained upon PS handover if, following handover, the SC UE has a "dedicated PDP context for SIP signalling" or has a "general-purpose PDP context" to carry the IM CN subsystem-related signalling, as described in 3GPP TS 24.229 subclause B.2.2.1.

- the SC UE is using ICS capabilities as defined in 3GPP TS 24.292 [4]; and

- vSRVCC procedures (as described in 3GPP TS 24.008 [8]) have been completed;

then the SC UE, in order to add Gm control for the newly established CS session, shall:

- send a SIP re-INVITE request for each session with only speech media component or with both speech and video media components to be transferred starting with the session with active speech and video media components that was most recently made active;

- within the SDP offer for sessions with speech and video media components, indicate the media lines for the speech and video media components as speech and video media components over circuit switched bearer in accordance with 3GPP TS 24.292 [4]. If the precondition mechanism is used, the SC UE shall indicate the related local preconditions as met; and

- within the SDP offer for sessions with only speech media component, indicate the media lines for the speech media component as speech media component over circuit switched bearer in accordance with 3GPP TS 24.292 [4]. If the precondition mechanism is used, the SC UE shall indicate the related local preconditions as met;

NOTE 2: The SIP dialog established over the source PS access network stays the same after vSRVCC procedures, e.g. the IP address of the UE, the Call-ID or the P-CSCF do not change. Therefore in this case a SIP re-INVITE request needs to be sent to add ICS-control for the CS bearer.

12.2A.3 Not based on ICS

After successful vSRVCC procedures (as described in 3GPP TS 24.008 [8]) have been completed, if the SC UE is not using ICS capabilities, the SC UE shall replace the session with active speech and video media components which was most recently made active with the newly established CS voice/video call.

NOTE 1: In the case when ICS is not supported or used, only the ongoing session with active speech and video media components which was most recently made active is transferred from PS to CS.

In addition, if:

- the Gm reference point is retained upon PS handover; and

NOTE 2: The SC UE knows that the Gm reference point is retained upon PS handover if, following handover, the SC UE has a dedicated PDP context for SIP signalling or has a general-purpose PDP context to carry the IM CN subsystem-related signalling, as described in 3GPP TS 24.229 [2] subclause B.2.2.1.

- there are one or more remaining media component(s) in the IMS session other than the speech and the video media components which were transferred to the CS Target Access Leg;

the SC UE shall:

- send a SIP re-INVITE request to the SCC AS as specified for media removal in subclause 13.2.1; and

- indicate in the SDP offer the speech and video media components as being removed.

NOTE 3: The SC UE can still send a SIP re-INVITE request if there are no remaining media components in the IMS session.

12.2A.4 Void

12.2A.5 Call in alerting phase

If the conditions in subclause 12.2.3B.1 for the application of subclause 12.2.3B.3 are satisfied for a session with active speech media component and active video media component prior to the successful completion of the vSRVCC procedures (as described in 3GPP TS 24.008 [8]), then after the successful completion of the vSRVCC handover procedures, the SC UE shall apply the procedures specified in subclause 12.2.3B.3

12.2A.5A Call in pre-alerting phase

If the SC UE supports the PS to CS SRVCC for originating calls in pre-alerting phase and the conditions in subclause 12.2.3B.1 for the application of subclause 12.2.3B.3.3 are satisfied for a session with active speech media component and active video media component prior to the successful completion of the vSRVCC procedures (as described in 3GPP TS 24.008 [8]), then after the successful completion of the vSRVCC handover procedures, the SC UE shall apply the procedures specified in subclause 12.2.3B.3.3.

If the SC UE supports the PS to CS SRVCC for terminating calls in pre-alerting phase and the conditions in subclause 12.2.3B.1 for the application of subclause 12.2.3B.3.4 are satisfied for a session with active speech media component and active video media component prior to the successful completion of the vSRVCC procedures (as

described in 3GPP TS 24.008 [8]), then after the successful completion of the vSRVCC handover procedures, the SC UE shall apply the procedures specified in subclause 12.2.3B.3.4.

12.2A.6 Abnormal cases

If the SC UE:

- receives a NOTIFICATION message containing an "SRVCC handover cancelled, IMS session re-establishment required" as described in 3GPP TS 24.301 [52]; or
- does not successfully transition to UTRAN after it receives the handover command (as described in 3GPP TS 36.331 [62]);

then:

- if the SC UE is engaged in one or more ongoing IMS sessions, the SC UE shall send a SIP re-INVITE request in accordance with subclause 12.2.4.1; and
- if the SC UE is engaged in a session in early dialog state, the SC UE shall send a SIP UPDATE request, in accordance with subclause 12.2.4.2.

12.2B SC UE procedures for CS to PS SRVCC

12.2B.1 Distinction of requests

The SC UE needs to distinguish the following SIP requests:

1) SIP REFER request:

   A) with the Refer-Sub header field containing "false" value; and

   B) containing application/vnd.3gpp.mid-call+xml MIME body or the application/vnd.3gpp.state-and-event-info+xml MIME type.

   In the procedures below, such requests are known as "SIP REFER requests for transfer of an additional session".

2) SIP INFO request:

   A) with the Info-Package header field containing the g.3gpp.state-and-event; and

   B) containing an application/vnd.3gpp.state-and-event-info+xml XML body associated with the info package according to IETF RFC 6086 [54] and compliant to the XML schema specified in the clause D.2 with the state-info XML element containing "early" and direction XML element containing "receiver".

   In the procedures below, such requests are known as "SIP INFO requests for transfer of incoming early session".

12.2B.2 First call transfer

12.2B.2.1 General

If SC UE supports the CS to PS SRVCC, upon receiving information from the lower layers that the CS to PS SRVCC access transfer is initiated, the SC UE shall:

1) if a CS call in Active (U10) state (defined in 3GPP TS 24.008 [8]) and Idle auxiliary state (defined in 3GPP TS 24.083 [43]) exists and if the ATGW transfer details were received from the lower layers:

   A) determine the active call being transferred as a CS call in Active (U10) state (defined in 3GPP TS 24.008 [8]) and Idle auxiliary state (defined in 3GPP TS 24.083 [43]);

   B) start rendering speech media of the determined active call being transferred received according to the UE information for CS to PS SRVCC sent to the network (see subclause 6.2.3); and

   C) start sending speech media of the determined active call being transferred according to the ATGW information for CS to PS SRVCC received from the network (see subclause 6.2.3) where the address type,

the connection address and the transport port to which the media stream is sent are replaced with the ATGW transfer details received from the lower layers; and

2) send a SIP INVITE request to STI-rSR according to 3GPP TS 24.229 [2]. The SC UE shall populate the SIP INVITE request with:

   A) Request-URI set to the STI-rSR received during registration (see subclause 6.2.1);

   B) SDP offer set to the UE information for CS to PS SRVCC sent to the network (see subclause 6.2.3);

   C) if a GRUU was received at registration, include the public GRUU or temporary GRUU in the Contact header field;

   D) signalling elements described in subclause 6A.2.2.2;

   E) void;

   F) if the SC UE supports the CS to PS SRVCC with the assisted mid-call feature:

      a) the Supported header field containing the option-tag "norefersub" specified in IETF RFC 4488 [20]; and

      b) the Accept header field containing the application/vnd.3gpp.mid-call+xml MIME type; and

   G) if the SC UE supports CS to PS SRVCC for calls in alerting phase:

      a) the Supported header field containing the option-tag "norefersub" specified in IETF RFC 4488 [20], if not inserted already;

      b) an Accept header field containing the application/vnd.3gpp.state-and-event-info+xml MIME type;

      c) a Recv-Info header field containing the g.3gpp.state-and-event package name; and

      d) a Supported header field with "100rel" option tag.

Upon receiving a SIP 1xx response or SIP 2xx response to the SIP INVITE request to STI-rSR, the SC UE shall associate the dialog of the SIP 1xx response or SIP 2xx response with the CS call where the transaction identifier sent by MSC server equals to the value of the g.3gpp.ti feature-capability indicator as described in annex C of a Feature-Caps header field of the SIP response. If the value of the g.3gpp.ti feature-capability indicator as described in annex C of a Feature-Caps header field of the SIP response indicates a CS call of a conference participant of a conference, the SC UE shall associate the dialog of the SIP 1xx response or SIP 2xx response with the CS calls of all the conference participants of the conference.

If the SC UE is not aware of such CS call, or the CS call is the "disconnect request" (U11) call state, the "disconnect indication" (U12) call state, the "release request" (U19) call state or the "null" (U0) call state as described in 3GPP TS 24.008 [8], the SC UE shall release or cancel the dialog established by the SIP 1xx response or SIP 2xx response to the SIP INVITE request to STI-rSR. If the CS call is the "disconnect request" (U11) call state as described in 3GPP TS 24.008 [8], the SC UE shall populate the SIP CANCEL request or the SIP BYE request with a Reason header field with the protocol field set to "SIP", the "cause" header field parameter indicating the selected status code and the "text" header field parameter indicating the selected reason phrase according to IETF RFC 3326 [57].

12.2B.2.2 Transfer of call with active speech media component

If SC UE supports the CS to PS SRVCC, in addition to the procedures in subclause 12.2B.2.1, if the CS call which was associated with the dialog of the SIP 1xx response or SIP 2xx response to the SIP INVITE request to STI-rSR in subclause 12.2B.2.1 is in the "hold request" auxiliary state (as defined in 3GPP TS 24.083 [43]), the speech media component of the session supported by the dialog is an active speech media component, and if the UE supports 3GPP TS 24.610 [28] , then UE shall invoke the hold service on a dialog according to 3GPP TS 24.610 [28] indicating that the speech media component is to be held.

12.2B.2.3 Transfer of call with inactive speech media component

If SC UE supports the CS to PS SRVCC and if the SC UE supports the CS to PS SRVCC with the assisted mid-call feature, in addition to the procedures in subclause 12.2B.2.1, if the CS call which was associated with the dialog of the SIP 1xx response or SIP 2xx response to the SIP INVITE request to STI-rSR in subclause 12.2B.2.1 is in the "retrieve request" auxiliary state (as defined in 3GPP TS 24.083 [43]), the speech media component of the session supported by the dialog is an inactive speech media component, and if the UE supports 3GPP TS 24.610 [28] , then UE shall invoke

the hold service on a dialog according to 3GPP TS 24.610 [28] indicating that the speech media component is to be resumed.

NOTE: If the network associates the SIP INVITE request to STI-rSR with session with inactive speech media component, the SDP answer will contain a=recvonly or a=inactive.

12.2B.2.4 Transfer of originating call in alerting phase

No additional procedures in addition to the procedures in subclause 12.2B.2.1 apply.

12.2B.2.5 Transfer of terminating call in alerting phase

If SC UE supports the CS to PS SRVCC and if the SC UE supports the CS to PS SRVCC for calls in alerting phase, in addition to the procedures in subclause 12.2B.2.1, upon receiving the SIP INFO request for transfer of incoming early session inside an early dialog created with the SIP INVITE request due to STI-rSR, the SC UE shall:

1) send SIP 200 (OK) response to the SIP INFO request; and

2) consider the SIP dialog to be the transferred incoming early session.

When the served user accepts the transferred incoming early session or if the user has accepted it already (i.e. the CS call which was associated with the dialog of the SIP 1xx response or SIP 2xx response to the SIP INVITE request to STI-rSR in subclause 12.2B.2.1 is in the "connect request" (U8) call state or the "active" (U10) call state as described in 3GPP TS 24.008 [8]), the SC UE shall send a SIP INFO request accepting the session inside the early dialog created with the SIP INVITE request due to STI-rSR according to 3GPP TS 24.229 [2]. The SC UE shall populate the SIP INFO request with:

1) an Info-Package header field with 3gpp.state-and-event info package name; and

2) application/vnd.3gpp.state-and-event-info+xml XML body associated with the info package according to IETF RFC 6086 [54] and compliant to the XML schema specified in the clause D.2 with the event XML element containing "call-accepted".

When the served user rejects the transferred incoming early session, the SC UE shall send a SIP CANCEL request cancelling the SIP INVITE request due to STI-rSR according to 3GPP TS 24.229 [2]. The SC UE shall populate the SIP CANCEL request with a Reason header field containing protocol "SIP" and the "cause" parameter indicating the selected status code and the "text" parameter indicating the selected reason phrase.

12.2B.3 Additional call transfer

12.2B.3.1 General

If SC UE supports the CS to PS SRVCC, if the SC UE supports the CS to PS SRVCC with the assisted mid-call feature or the CS to PS SRVCC for calls in alerting phase then upon receiving a SIP REFER request for transfer of an additional session within dialog established by the SIP INVITE request to STI-rSR, the SC UE shall:

1) handle the SIP REFER request as specified in 3GPP TS 24.229 [2], IETF RFC 3515 [13] as updated by IETF RFC 6665 [81], and IETF RFC 4488 [20] without establishing an implicit subscription;

NOTE 1: In accordance with IETF RFC 4488 [20], the SC UE inserts the Refer-Sub header field containing the value "false" in the SIP 2xx response to the SIP REFER request to indicate that it has not created an implicit subscription.

2) send a SIP INVITE request for transfer of an additional session according to 3GPP TS 24.229 [2] and IETF RFC 3515 [13]. The SC UE shall populate the SIP INVITE request as follows:

   A) header fields which were included as URI header fields in the URI in the Refer-To header field of the received SIP REFER request as specified in IETF RFC 3261 [19] except the "body" URI header field;

   B) the SDP offer with:

      a) the same amount of the media descriptions as in the "body" URI header field in the URI in the Refer-To header field of the received SIP REFER request;

b) each "m=" line having the same media type as the corresponding "m=" line in the "body" URI header field in the URI in the Refer-To header field of the received SIP REFER request;

c) port set to zero value in each "m=" line whose corresponding "m=" line in the "body" URI header field in the URI in the Refer-To header field of the received SIP REFER request has port with zero value;

d) media directionality as in the "body" URI header field in the URI in the Refer-To header field of the received SIP REFER request; and

e) payload type numbers and their mapping to codecs and media parameters supported by SC UE, not conflicting with those in the "body" URI header field in the URI in the Refer-To header field of the received SIP REFER request;

NOTE 2: port can be sent to zero or non zero value for the offered "m=" line whose corresponding "m=" line in the "body" URI header field in the URI in the Refer-To header field of the received SIP REFER request has port with nonzero value.

C) if a GRUU was received at registration, include the public GRUU or temporary GRUU in the Contact header field;

D) signalling elements described in subclause 6A.2.2.2;

E) void; and

F) if the SC UE supports the CS to PS SRVCC for calls in alerting phase:

a) a Supported header field with "100rel" option tag; and

3) if the SC UE supports the CS to PS SRVCC for calls in alerting phase and if the SIP REFER request contains a XML body compliant to the XML schema specified in the clause D.2 with the state-info XML element containing "early" and direction set to "receiver" then consider the SIP dialog to be transferred incoming early session.

Upon receiving a SIP 1xx response or SIP 2xx response to the SIP INVITE request for transfer of an additional session, the SC UE shall associate the dialog of the SIP 1xx response or SIP 2xx response with the CS call where the transaction identifier sent by MSC server equals to the value of the g.3gpp.ti feature-capability indicator as described in annex C of a Feature-Caps header field of the SIP response. If the value of the g.3gpp.ti feature-capability indicator as described in annex C of a Feature-Caps header field of the SIP response indicates a CS call of a conference participant of a conference, the SC UE shall associate the dialog of the SIP 1xx response or SIP 2xx response with the CS calls of all the conference participants of the conference.

If the SC UE is not aware of such CS call, or the CS call is the "disconnect request" (U11) call state, the "disconnect indication" (U12) call state, the "release request" (U19) call state or the "null" (U0) call state as described in 3GPP TS 24.008 [8], the SC UE shall release or cancel the dialog established by SIP 1xx response or SIP 2xx response to the SIP INVITE request for transfer of an additional session. If the CS call is the "disconnect request" (U11) call state as described in 3GPP TS 24.008 [8], the SC UE shall populate the SIP CANCEL request or the SIP BYE request with a Reason header field with the protocol field set to "SIP", the "cause" header field parameter indicating the selected status code and the "text" header field parameter indicating the selected reason phrase according to IETF RFC 3326 [57].

12.2B.3.2 Transfer of call with active speech media component

No additional procedures in addition to the procedures in subclause 12.2B.3.1 apply.

12.2B.3.3 Transfer of call with inactive speech media component

If SC UE supports the CS to PS SRVCC and if the SC UE supports the CS to PS SRVCC with the assisted mid-call feature, in addition to the procedures in subclause 12.2B.3.1, if the CS call which was associated with the dialog of the SIP 1xx response or SIP 2xx response to the SIP INVITE request for transfer of an additional session in subclause 12.2B.3.1 is in the "retrieve request" auxiliary state (as defined in 3GPP TS 24.083 [43]), the speech media component of the session supported by the dialog is an inactive speech media component, and if the UE supports 3GPP TS 24.610 [28] , then UE shall invoke the hold service on a dialog according to 3GPP TS 24.610 [28] indicating that the speech media component is to be resumed.

NOTE: If the network associates the SIP INVITE request to STI-rSR with session with inactive speech media component, the SDP answer will contain a=recvonly or a=inactive.

12.2B.3.4 Transfer of originating call in alerting phase

No additional procedures in addition to the procedures in subclause 12.2B.3.1 apply.

12.2B.3.5 Transfer of terminating call in alerting phase

If SC UE supports the CS to PS SRVCC, if the SC UE supports the CS to PS SRVCC for calls in alerting phase, in addition to the procedures in subclause 12.2B.3.1, when the served user accepts the transferred incoming early session or if the user has accepted it already (i.e. the CS call which was associated with the dialog of the SIP 1xx response or SIP 2xx response to the SIP INVITE request for transfer of an additional session in subclause 12.2B.3.1 is in the "connect request" (U8) call state or the "active" (U10) call state as described in 3GPP TS 24.008 [8]), the SC UE shall send a SIP INFO request accepting the session inside the early dialog created with the SIP INVITE request for transfer of an additional session according to 3GPP TS 24.229 [2]. The SC UE shall populate the SIP INFO request with:

1) an Info-Package header field with 3gpp.state-and-event info package name; and

2) application/vnd.3gpp.state-and-event-info+xml XML body associated with the info package according to IETF RFC 6086 [54] and compliant to the XML schema specified in the clause D.2 with the event XML element containing "call-accepted".

If the SC UE supports the CS to PS SRVCC for calls in alerting phase then when the served user rejects the transferred incoming early session, the SC UE shall send a SIP CANCEL request cancelling the SIP INVITE request for transfer of an additional session according to 3GPP TS 24.229 [2]. The SC UE shall populate the SIP CANCEL request with:

1) a Reason header field containing protocol "SIP" and the "cause" parameter indicating the selected status code and the "text" parameter indicating the selected reason phrase.

12.2B.4 Procedures after calls are transferred

If:

1) the multi party auxiliary state (defined in 3GPP TS 24.084 [47]) for the CS call which was associated with the dialog of the SIP 1xx response or SIP 2xx response to the SIP INVITE request to STI-rSR in subclause 12.2B.2.1 is "MPTY request";

2) the multi party auxiliary state (defined in 3GPP TS 24.084 [47]) for the CS call which was associated with the dialog of the SIP 1xx response or SIP 2xx response to the SIP INVITE request for transfer of an additional session in subclause 12.2B.3.1 is "MPTY request";

3) SIP 1xx response or SIP 2xx response to the SIP INVITE request to STI-rSR contains a Contact header field without the isfocus media feature tag specified in IETF RFC 3840 [53]; and

4) SIP 1xx response or SIP 2xx response to the SIP INVITE request for transfer of an additional session contains a Contact header field without the isfocus media feature tag specified in IETF RFC 3840 [53];

and if the UE supports 3GPP TS 24.605 [31], then UE shall create a conference according to 3GPP TS 24.605 [31], subclause 4.5.2.1.4. If the conference creation was successful, the UE shall invite remote user of the dialog of the SIP 1xx response or SIP 2xx response to the SIP INVITE request to STI-rSR and remote user of the dialog of the SIP 1xx response or SIP 2xx response to the SIP INVITE request for transfer of an additional session to the conference according to 3GPP TS 24.605 [31], subclause 4.5.2.1.2.

12.3 SCC AS

12.3.0 General

In the Single Radio access transfer procedures the SCC AS shall only consider sessions that have the necessary media components that meet the criteria for performing Single Radio access transfer as defined in subclause 4.2.2.

On receipt of a SIP INVITE request due to STN-SR or a SIP INVITE request due to ATU-STI the SCC AS shall for all sessions and dialogs in the transferable session set created in subclauses 12.3.0B or 12.7.2 discard any SIP message (i.e. any SIP request or SIP response) from the remote UE to the SC UE that are releated to the call until the procedures in the following subclauses states that forwarding of SIP shall be started again. The SIP 200 (OK) response to the UPDATE request and the SIP 200 (OK) respose to the re-INVITE request related to the ongoing access transfer shall be handled as specified in the following subclauses.

NOTE: SIP responses sent reliably to the initial SIP INVITE request and SIP requests will be retransmitted by the remote UE if SIP responses and SIP requests are dropped by the SCC AS.

All the time during an ongoing PS to CS SRVCC access transfer the SCC AS shall be prepared to receive SIP messages from the SC UE or the MSC server releated to the call (i.e. not related to the ongoing access transfer). When a SIP message releated to the call is received from the SC UE or the MSC server the SCC AS shall forward the SIP message towards the remote UE according to procedures in 3GPP TS 24.229 [2] and the present document.

When a PS to CS SRVCC access transfer is completed, the SCC AS shall perform interactions between the remote UE leg and the target access leg as described in subclause 12.3.11.

12.3.0A Distinction of requests sent to the SCC AS

The SCC AS needs to distinguish between the following SIP INVITE requests to provide specific functionality for PS to CS SRVCC:

- SIP INVITE request routed to the SCC AS due to a STN-SR belonging to the subscribed user in the Request-URI and containing an SDP offer with active speech media component only. These SIP INVITE requests originate from the MSC server. In the procedures below, such requests are known as "SIP INVITE requests due to STN-SR".

- SIP INVITE requests routed to the SCC AS due to ATU-STI for PS to CS SRVCC in the Request-URI. In the procedures below, such requests are known as "SIP INVITE requests due to ATU-STI for PS to CS SRVCC".

- SIP INVITE request routed to the SCC AS contains the additional transferred session SCC AS URI for PS to CS SRVCC in the Request-URI, such a request is in this document known as "SIP INVITE request transferring additional session for PS to CS SRVCC".

The SCC AS needs to distinguish between the following SIP INVITE requests to provide specific functionality for vSRVCC:

- SIP INVITE request routed to the SCC AS due to a STN-SR belonging to the subscribed user in the Request-URI and containing an SDP offer with both active speech and video media components only, which includes the default codecs for speech and video (as specified in 3GPP TS 26.111 [69]). These SIP INVITE requests originate from the MSC server. In the procedures below, such requests are known as "SIP INVITE requests for audio and video due to STN-SR".

12.3.0B Determine the transferable session set

When the SCC AS receives a SIP INVITE request due to STN-SR on the Target Access Leg the SCC AS shall determine the transferable session set.

A session is in the transferable session set when the session:

1) is a session of the SC UE whose private user identity is associated with the C-MSISDN that is contained in the P-Asserted-Identity header field of the SIP INVITE request; and

2) has completed a reliable offer answer procedure and is a session with speech media component.

The SCC AS shall:

1) if the conditions described in subclause 12.3.2.1 are fulfilled, follow the procedures in subclause 12.3.2;

2) if the conditions described in subclause 12.3.4.1 for applying the PS to CS SRVCC for calls in alerting phase in subclauses 12.3.4.2 or 12.3.4.3 are fulfilled, follow the procedures in subclause 12.3.4.2 or 12.3.4.3;

2A) if the conditions described in subclause 12.3.4.1 for applying the PS to CS SRVCC for calls in originating pre-alerting phase in subclause 12.3.4.3 are fulfilled, follow the procedures in subclause 12.3.4.3; and

3) if none of the conditions 1), 2) or 2A) above are fulfilled follow the procedure in subclause 12.3.1.

12.3.1 SCC AS procedures for PS to CS access transfer, PS to CS SRVCC

When the SCC AS receives a SIP INVITE request due to STN-SR on the Target Access Leg the SCC AS shall associate the SIP INVITE request with a session:

- within the transferable session set;
- with active speech media component that was most recently made active; and
- the related dialog is in confirmed state.

If no confirmed dialogs supporting a session with active speech media component exists in the transferable session set the SCC AS shall:

1) send a SIP 480 (Temporarily Unavailable) response to reject the SIP INVITE request due to STN-SR;
2) if the transferable session set contains dialogs supporting sessions with speech media component (inactive speech media component or in an early dialog state):
   a) if the speech media component is the only media component in the dialog then release the remote leg as specified in 3GPP TS 24.229 [2]; and
   b) if the speech media component is not the only media component in the dialog then modify the remote leg and remove the speech media component as specified in 3GPP TS 24.229 [2].

If confirmed dialogs supporting a session with active speech media component exist in the transferable session set the SCC AS shall:

1) send a SIP re-INVITE request towards the remote UE and in a new SDP offer, include the media characteristics as received in the SIP INVITE request due to STN-SR, by following the rules of 3GPP TS 24.229 [2]; or
2) send a SIP re-INVITE request towards the remote UE according to the conditions depicted in subclause 12.3.5 and in a new SDP offer, include the media characteristics as received in the SIP INVITE request due to ATU-STI for PS to CS SRVCC, by following the rules of 3GPP TS 24.229 [2].

Upon receiving the SIP 2xx response to the SIP re-INVITE request the SCC AS shall send the SIP 200 (OK) response to the SIP INVITE request due to STN-SR on the target access leg by following the rules of 3GPP TS 24.229 [2]. The SCC AS shall populate the SIP 200 (OK) response to the SIP INVITE request due to STN-SR as follows:

1) include SDP answer containing the relevant media parameter of the SDP answer in the received response;
2) if the SCC AS supports the PS to CS SRVCC of calls in alerting phase, include the g.3gpp.srvcc-alerting feature-capability indicator as described in annex C in the Feature-Caps header field according to IETF RFC 6809 [60]; and
3) include the signalling elements described in subclause 6A.4.3A.

If the SCC AS supports the PS to CS SRVCC for calls in alerting phase and if the conditions for a terminating call in alerting phase specified in subclause 12.3.4.1 for a session in the transferable session set are fulfilled, the SCC AS shall follow the procedures in the subclause 12.3.4.4 and then continue with the procedures in this subclause.

Upon receipt of the ACK request from the MSC server, start forwarding SIP messages from the remote UE to the MSC server for the session with active speech media component as specified in 3GPP TS 24.229 [2] and the present document.

The SCC AS shall remove non-transferred audio components and superfluous session as specified in subclause 12.3.8.

12.3.2 SCC AS procedures for PS to CS access transfer with MSC server assisted mid-call feature, PS to CS SRVCC

12.3.2.1 General

The SCC AS shall apply the MSC Server assisted mid-call feature as described in subclause 12.3.2.2 if:

1. one of the conditions is true:
   a. the SC UE included the g.3gpp.ics media feature tag as specified in the 3GPP TS 24.292 [4] in the Contact header field during establishment of the session associated with the SIP INVITE request due to STN-SR, the SCC AS local policy requires delaying application of the MSC Server assisted mid-call feature for a time given by local policy and the transfer request for the session with inactive speech media component has not

been received within a time given by local policy after the reception of the SIP INVITE request due to STN-SR;

b. the SC UE included the g.3gpp.ics media feature tag as specified in the 3GPP TS 24.292 [4] in the Contact header field during establishment of the session associated with the SIP INVITE request due to STN-SR and the SCC AS local policy does not require delaying application of the MSC Server assisted mid-call feature for a time given by local policy; or

c. the SC UE did not include the g.3gpp.ics media feature tag as specified in the 3GPP TS 24.292 [4] in the Contact header field during establishment of the session associated with the SIP INVITE request due to STN-SR;

2. the Contact header field of the SIP INVITE request due to STN-SR or SIP INVITE request due to ATU-STI for PS to CS SRVCC includes the g.3gpp.mid-call media feature tag as specified in annex C; and

3. one of the following is true for dialogs in the transferable session set:

A. at least one confirmed dialog supporting a session with active speech media component exists and the following is true for the confirmed dialog supporting a session with the active speech media component which has been most recently made active:

- the Contact header field provided by the SC UE at the establishment of the dialog includes the g.3gpp.mid-call media feature tag as described in annex C; and

- the Feature-Caps header field sent by SCC AS towards the SC UE at the establishment of the dialog included g.3gpp.mid-call feature-capability indicator; or

B. no confirmed dialog supporting a session with active speech media component exists and the following is true for the confirmed dialog supporting a session with inactive speech media component which became inactive most recently:

- the Contact header field provided by the SC UE at the establishment of the dialog includes the g.3gpp.mid-call media feature tag as described in annex C; and

- the Feature-Caps header field sent by SCC AS towards the SC UE at the establishment of the dialog included the g.3gpp.mid-call feature-capability indicator.

12.3.2.2 Transfer of the first session

When the SCC AS applies the MSC Server assisted mid-call feature for transfer of the first session the SCC AS shall select the first session to transfer as follows.

The first session to transfer is a session in the transferable session set such that:

1. if one or more confirmed dialog supporting a session with active speech media component exists in the transferable session set then:

- select the confirmed dialog supporting a session with the active speech media component which became active most recently; and

2. if no confirmed dialog supporting a session with active speech media component exists in the transferable set but one or more confirmed dialogs supporting a session with inactive speech media component exists for the user then:

- select the confirmed dialog supporting a session with inactive speech media component that became inactive most recently.

If the speech media component of the SDP offer in the SIP INVITE request due to ATU-STI is the same as the speech media component of the SDP negotiated by the ATCF in the first session to transfer, the SCC AS shall send a SIP 200 (OK) response to the SIP INVITE request. The SCC AS shall populate the SIP 200 (OK) response to the SIP INVITE request as follows:

1) include SDP answer containing the speech media component of the SDP negotiated by SCC AS towards ATCF in the first session to transfer;

2) include:

- the g.3gpp.mid-call feature-capability indicator as described in annex C; and
- if the SCC AS supports the PS to CS SRVCC of calls in alerting phase, the g.3gpp.srvcc-alerting feature-capability indicator as described in annex C;

in the Feature-Caps header field according to IETF RFC 6809 [60]; and

3) include the signalling elements described in subclause 6A.4.3A.

If the speech media component of the SDP offer in the SIP INVITE request due to ATU-STI is different from the speech media component of the SDP negotiated by the ATCF in the first session to transfer or if the SIP INVITE request due to STN-SR was received, the SCC AS shall send a SIP re-INVITE request towards the remote UE with a new SDP offer, such that:

1) if a session with the confirmed dialog supporting a session with active speech media component was selected, include the media characteristics as received in the SIP INVITE request due to ATU-STI or the SIP INVITE request due to STN-SR, by following the rules of 3GPP TS 24.229 [2] including directionality attributes indicating the directionality used at SC UE; and

2) if a session with the confirmed dialog supporting a session with inactive speech media component was selected then include an SDP offer describing the audio media streams as negotiated in the session with the remote UE and:

   - if directionality used by SC UE is "sendrecv" or "sendonly", with the "sendonly" directionality; and
   - if directionality used by SC UE is "recvonly" or "inactive", with the "inactive" directionality.

Upon receiving the SIP 2xx response to the SIP re-INVITE request the SCC AS shall:

1) send the SIP 200 (OK) response to the SIP INVITE request due to STN-SR on the target access leg populated as follows:

   a) in the SDP answer, use the relevant media parameter of the SDP answer in the received response;

   b) include:

      - the g.3gpp.mid-call feature-capability indicator as described in annex C; and
      - if the SCC AS supports the PS to CS SRVCC of calls in alerting phase, the g.3gpp.srvcc-alerting feature-capability indicator as described in annex C;

      in the Feature-Caps header field according to IETF RFC 6809 [60]; and

   c) include the signalling elements described in subclause 6A.4.3A.

Upon receiving the SIP ACK request related to the SIP 200 (OK) response to the SIP INVITE request, the SCC AS shall start forwarding SIP messages from the remote UE to the MSC server for the session with active or inactive speech media component as specified in 3GPP TS 24.229 [2] and the present specification.

Upon receiving the SIP ACK request related to the SIP 200 (OK) response to the SIP INVITE request and if:

1) the session associated with the SIP INVITE request due to STN-SR is related to a subscription as described in subclause 7.3.3; and

2) a SIP 2xx response was received to the last SIP NOTIFY request with conference information sent to the UE within the related subscription;

then the SCC AS shall send a SIP INFO request towards the MSC Server as specified in 3GPP TS 24.229 [2] and IETF RFC 6086 [54] in the dialog created by the SIP INVITE request due to STN-SR. The SCC AS shall populate the SIP INFO request as follows:

1) include the Info-Package header field as specified in IETF RFC 6086 [54] with g.3gpp.mid-call package name; and

2) include application/vnd.3gpp.mid-call+xml XML body associated with the info package according to IETF RFC 6086 [54] and containing the participants extracted as specified in the subclause 9.1A of the

subscription related to the session associated with the SIP INVITE request due to STN-SR as described in subclause 7.3.3.

Upon receiving the SIP 2xx response to the SIP INFO request or receiving the ACK related to the SIP 200 (OK) response to the SIP INVITE request when the SIP INFO request was not sent, the SCC AS shall:

- if one more confirmed SIP dialogs supporting a session with speech media component exist in the transferable session set transfer the additional second confirmed SIP dialog as described in subclause 12.3.2.3 and then continue with the procedures in this subclause; and

- if no more confirmed SIP dialog supporting a session with speech media component exist in the transferable session set but SCC AS support the PS to CS SRVCC for calls in alerting phase and the conditions in the subclause 12.3.4.1 are fulfilled, perform the actions in subclause 12.3.4.4 and then continue with the procedures in this subclause.

The SCC AS shall remove non-transferred audio components and superfluous session as specified in subclause 12.3.8.

12.3.2.3 Transfer of an additional session

When the SCC AS applies the MSC Server assisted mid-call feature for transfer of the additional session the SCC AS shall select the additional session to transfer as a session in the transferable session set such that:

1. if more than one confirmed dialog supporting a session exists in the transferable session set, and exactly one confirmed dialog supporting a session with active speech media component exists and there is at least one remaining confirmed dialog supporting a session with inactive speech media component then:

   - select the confirmed dialog supporting a session with inactive speech media component that became inactive most recently; and

2. if more than one confirmed dialog supporting a session with active speech media component exists in the transferable session set then:

   - select the confirmed dialog supporting a session with the active speech media component which became active second most recently.

When the SCC AS transfers the selected additional session the SCC AS shall:

A) send a SIP REFER request towards the MSC Server in accordance with the procedures specified in 3GPP TS 24.229 [2], IETF RFC 4488 [20] and IETF RFC 3515 [13] as updated by IETF RFC 6665 [81] and IETF RFC 7647 [90] in the dialog created by the SIP INVITE request due to STN-SR. ; or send a SIP REFER request towards the ATCF in accordance with the procedures specified in 3GPP TS 24.229 [2], IETF RFC 4488 [20] and IETF RFC 3515 [13] as updated by IETF RFC 6665 [81] and IETF RFC 7647 [90] in the dialog created by the SIP INVITE request due to ATU-STI for PS to CS SRVCC. The SCC AS shall populate the SIP REFER request as follows:

   1. the Refer-Sub header field with value "false" as specified in IETF RFC 4488 [20];

   2. the Supported header field with value "norefersub" as specified in IETF RFC 4488 [20];

   3. the Refer-To header field containing the additional transferred session SCC AS URI for PS to CS SRVCC and the following URI header fields containing information related to the additional transferred session:

      a. the Target-Dialog URI header field populated as specified in IETF RFC 4538 [11], containing the dialog identifier of the session with the SC UE;

      b. the Require URI header field populated with the option tag value "tdialog";

      c. the To URI header field populated as specified in IETF RFC 3261 [19], containing the P-Asserted-Identity provided by the remote UE during the session establishment;

      d. the From URI header field populated as specified in IETF RFC 3261 [19], containing the public user identity of the SC UE provided during the session establishment;

      e. the Content-Type header field with "application/sdp";

f. the "body" URI header field populated with an SDP body describing the media streams as negotiated in the session with the remote UE and:

- if directionality used by SC UE is "sendrecv" or "sendonly", with the "sendonly" directionality; and

- if directionality used by SC UE is "recvonly" or "inactive", with the "inactive" directionality; and

g. optionally the P-Asserted-Identity URI header field containing value of the P-Asserted-Identity header field of the received SIP INVITE request;

4. the Content-Type header field with the value set to MIME type as specified in the subclause D.1.3;

5. a XML body compliant to the XML schema specified in the subclause D.1.2;

6. if:

a. the session associated with the SIP INVITE request due to STN-SR, is not related to any subscription as described in subclause 7.3.3;

b. the additional transferred session is related to a subscription as described in subclause 7.3.3; and

c. a SIP 2xx response was received to the last SIP NOTIFY request with conference information sent to the SC UE within the related subscription;

then SCC AS shall populate the XML body with the participants extracted as specified in the subclause 9.1A of the subscription related to the additional transferred session as specified in subclause 7.3.3; and

7. if:

a. SCC AS supports CS to PS SRVCC;

b. the SIP INVITE request due to STN-SR contains Accept header field with application/vnd.3gpp.srvcc-ext+xml MIME type;

c. a private user identity of a UE (i.e. other than those according to 3GPP TS 23.003 [12], subclause 20.3.3) is associated with the C-MSISDN in the P-Asserted-Identity header field of the SIP INVITE request due to STN-SR;

d. a binding of a contact address exists for the private user identity of the UE:

i) where the g.3gpp.cs2ps-srvcc media feature tag is associated with the contact address of the UE; and

ii) such that SIP REGISTER request which registered the binding contained a Feature-Caps header field with the g.3gpp.atcf feature-capability indicator and with g.3gpp.cs2ps-srvcc feature-capability indicator;

e. the CS to PS SRVCC capability indication is indicated for the private user identity of the UE; and

f. the private user identity of the UE has the CS to PS SRVCC allowed indication in the subscription data;

then:

a. a MIME body of application/vnd.3gpp.srvcc-ext+xml MIME type:

i) containing ATU management URI of the ATCF serving the SC UE;

NOTE 1: The ATCF management URI of the ATCF is the URI contained in the g.3gpp.atcf-mgmt-uri feature-capability indicator included in a Feature-Caps header field of the SIP REGISTER request, which registered the binding for the private user identity of the UE.

ii) containing C-MSISDN; and.

iii) not indicating that information relate to a registration of MSC server with IMS.

When the SCC AS receives the SIP INVITE request transferring additional session for PS to CS SRVCC, the SCC AS shall:

1) associate the SIP INVITE request transferring additional session for PS to CS SRVCC with a previously established SIP dialog i.e. identify the source access leg;

NOTE 2: The SIP dialog on the source access leg is identified by matching the dialog present in the Target-Dialog header field (see IETF RFC 4538 [11]) of the SIP INVITE request transferring additional session for PS to CS SRVCC with the previously established SIP dialog.

NOTE 3: By a previously established SIP dialog, it is meant a dialog for which a SIP 2xx response to the initial SIP INVITE request has been sent or received.

2) if the SCC AS is unable to associate the SIP INVITE with a unique previously established SIP dialog, send a SIP 480 (Temporarily Unavailable) response to reject the SIP INVITE request relating to the access transfer and not processes the remaining steps;

3) if the number of media lines in the target access leg is less than the number of media lines in the source access leg or the media type for the corresponding media lines is not the same as in the original session, send a SIP 404 (Not Found) response to reject the SIP INVITE request relating to the access transfer and not process the remaining steps;

4) if the speech media component of the SDP offer in the SIP INVITE request transferring additional session for PS to CS SRVCC is different to the speech media component of the SDP received in the source access leg, send a SIP re-INVITE request towards the remote UE using the existing established dialog. The SCC AS shall populate the SIP re-INVITE request with a new SDP offer, following the rules specified in 3GPP TS 24.229 [2], containing the following media information:

   A) the media characteristics as received in the SIP INVITE request transferring additional session for PS to CS SRVCC for media streams whose port is not set to zero; and

   B) for the media streams in the SIP INVITE request transferring additional session for PS to CS SRVCC whose port is set to zero, include the corresponding media characteristics of those streams from the source access leg; and

5) if the speech media component of the SDP offer in the SIP INVITE request transferring additional session for PS to CS SRVCC is the same as the speech media component of the SDP received in the source access leg, send the SIP 200 (OK) response to the SIP INVITE request transferring additional session for PS to CS SRVCC. In the SIP 200 (OK) response, the SCC AS shall:

   a) include the SDP sent by the SCC AS in the source access leg;

   b) include the g.3gpp.mid-call feature-capability indicator as described in annex C in the Feature-Caps header field; and

   c) include the signalling elements described in subclause 6A.4.3A.

Upon receiving the SIP 2xx response to the SIP re-INVITE request the SCC AS shall send the SIP 200 (OK) response to the SIP INVITE request transferring additional session for PS to CS SRVCC on the target access leg populated as follows:

1) use the relevant media parameter of the SDP answer in the received response;

2) include the g.3gpp.mid-call feature-capability indicator as described in annex C in the Feature-Caps header field; and

3) include the signalling elements described in subclause 6A.4.3A.

Upon receiving the SIP ACK request from the MSC server, the SCC AS shall start forwarding SIP messages from the remote UE to the MSC server for the additional session as specified in 3GPP TS 24.229 [2] and the present specification.

12.3.3 SCC AS procedures for PS to CS SRVCC, abnormal case

12.3.3.1 PS to CS SRVCC cancelled by MME/SGSN or failure by UE to transition to CS domain for ongoing session

When the SCC AS receives a SIP re-INVITE request containing Reason header field containing protocol "SIP" and reason parameter "cause" with value "487" on

- the original source access leg; or
- the original source access leg of the additional transferred session if the SCC AS applies the MSC Server assisted mid-call feature;

after:

a) having initiated an access transfer that was triggered by a SIP INVITE request due to STN-SR and the SIP INVITE request due to STN-SR transaction is not yet completed; or

b) having initiated an access transfer that was triggered by a SIP INVITE request due to ATU-STI for PS to CS SRVCC and the SIP INVITE request due to ATU-STI for PS to CS SRVCC transaction is not yet completed;

then the SCC AS shall wait until this transaction has completed and then continue with the steps 1) and 2) described below.

When the SCC AS receives a SIP re-INVITE request(s) containing protocol "SIP" and reason parameter "cause" with value "487" after:

a) having performed an access transfer that was triggered by a SIP INVITE request due to STN-SR; or

b) having performed an access transfer that was triggered by a SIP INVITE request due to ATU-STI for PS to CS SRVCC;

then the SCC AS shall:

1) not release the original source access leg once the expiration of the timer described in subclause 12.3.8; and

2) treat the SIP re-INVITE request(s) as per procedures for removing and adding media as described in subclause 13.3.1.

NOTE: The SCC AS assigns an operator specific timer to delay the release of the Source Access Leg for PS to CS SRVCC access transfer.

When the SCC AS receives a SIP response to the SIP re-INVITE request indicating success in removing all media components from a dialog that was created:

a) due to the SIP INVITE request due to STN-SR; or

b) due to the SIP INVITE request due to ATU-STI for PS to CS SRVCC;

then the SCC AS shall send a SIP BYE request on this dialog, by following the rules of 3GPP TS 24.229 [2] and start forwarding SIP message from the remote UE to the SC UE for this ongoing session as specified in 3GPP TS 24.229 [2] and the present specification.

12.3.3.1A PS to CS SRVCC cancelled by MME/SGSN or failure by UE to transition to CS domain for session in early dialog state

If the SCC AS applies the procedures for the PS to CS SRVCC for calls in alerting phase or the PS to CS SRVCC for originating calls in pre-alerting phase (as specified in subclause 12.3.4), then when the SCC AS receives a SIP UPDATE request containing Reason header field containing protocol "SIP" and reason parameter "cause" with value "487" on:

- the original source access leg; or
- the original source access leg of the additional transferred session if the SCC AS applies the MSC Server assisted mid-call feature;

after having initiated an access transfer that was triggered by:

a) a SIP INVITE request due to STN-SR; or

b) a SIP INVITE request due to ATU-STI for PS to CS SRVCC;

for a session which is still in early dialog state the SCC AS shall:

1) not release the original access leg after the expiration of the timer described in subclause 12.3.8; and

2) treat the SIP UPDATE request(s) as per procedures for removing and adding media as described in subclause 13.3.1.

The SCC AS shall now start forwarding SIP messages from the remote UE to the SC UE for each dialog created by the SIP INVITE request due to STN-SR, the SIP INVITE request due to ATU-STI or the SIP INVITE request transferring additional session for PS to CS SRVCC as specified in 3GPP TS 24.229 [2] and the present document.

When the SCC AS receives a SIP 200 (OK) response to the SIP UPDATE request, then the SCC AS shall:

1) if the SCC AS has already sent a SIP 200 (OK) response to a SIP INVITE request due to STN-SR or SIP INVITE request due to ATU-STI for PS to CS SRVCC then send a SIP BYE request on this dialog, and

   a) if the SCC AS performs access transfer for an originating session which is in early dialog state , send a SIP 200 (OK) response to the SIP INVITE on the original source access leg; and

   b) if the SCC AS performs access transfer for an additional transferred originating session which is still in early dialog state, send a SIP 200 (OK) response to the INVITE on the original source access leg of the additional transferred session; and

2) if the SCC AS has not sent a SIP 200 (OK) response to a SIP INVITE request due to STN-SR or SIP INVITE request due to ATU-STI for PS to CS SRVCC then send a SIP 480 (Temporarily Unavailable) response to reject the SIP INVITE request due to STN-SR or the SIP INVITE request due to ATU-STI for PS to CS SRVCC.

If the SCC AS has received a SIP 200 (OK) response from the SC UE prior to receiving the SIP UPDATE request from the SC UE, then on receipt of the SIP 200 (OK) response to the SIP UPDATE request sent to the remote UE, the SCC AS shall send a SIP 200 (OK) response to the remote UE. Upon receiving the SIP ACK request from the remote UE, the SCC AS shall send a SIP ACK request to the SC UE.

12.3.3.2 P-CSCF releasing the source access leg during PS to CS SRVCC

When SCC AS receives a SIP BYE request on the Source Access Leg with the Reason header field containing a SIP 503 (Service Unavailable) response code then:

- if the SCC AS receives an initial SIP INVITE request on the Target Access Leg associated with the established dialog on the Source Access Leg, within a time defined by the operator policy after the SIP BYE request reception, then the SCC AS shall not initiate release of the Remote Leg; and

- if the SCC AS does not receive an initial SIP INVITE request on the Target Access Leg associated with the established dialog on the Source Access Leg, within a time defined by the operator policy after the SIP BYE request reception then the SCC AS shall initiate release of the Remote Leg.

NOTE: 8 seconds is an appropriate value for the operator policy.

12.3.3.3 P-CSCF releasing the source access leg when call is in alerting phase

The procedures specified in subclause 10.3.6 apply.

12.3.3.4 PS to CS SRVCC cancelled by MME/SGSN or release of the target access leg for an ongoing session

This subclause describes the procedures for cancelling calls after the SCC AS have initiated an PS to CS SRVCC access transfer that was triggered by a SIP INVITE request due to STN-SR, a SIP INVITE request due to ATU-STI or a SIP INVITE request transferring additional session for PS to CS SRVCC for a session where the dialog is in confirmed state with active or inactive speech media component.

If the SCC AS receives a SIP BYE request containing a Reason header field containing the protocol value "Q.850" and the "cause" header field parameter with the value of "31" (normal unspecified) on:

- the target access leg where the dialog is in confirmed state with active speech media component;
- the target access leg where the dialog is in confirmed state with inactive speech media component, if the SCC AS applies the MSC server assisted mid-call feature;or
- the target access leg of an additional transferred session where the dialog is in confirmed state with an inactive speech media component, if the SCC AS applies the MSC server assisted mid-call feature,

after having initiated an access transfer and when the operator specific timer is still running, the SCC AS shall:

1) send the SIP 200 (OK) response to the SIP BYE request;
2) wait until the operator specific timer expires or until a SIP re-INVITE request from the SC UE containing the protocol value "SIP" and the "cause" header field parameter with the value "487" is received; and
3) if the operator specific timer expires and no SIP re-INVITE request from the SC UE containing the protocol value "SIP" and the "cause" header field parameter with the value "487" is received, release the call according to procedures in 3GPP TS 24.229 [2].

NOTE 1: All protocol values in the Reason header field other than "Q.850" and all values of the "cause" header field parameter other than "31" (normal unspecified) will result in an immediate release of the source access leg and the remote UE leg.

NOTE 2: The SCC AS assigns an operator specific timer to delay the release of the source access leg for PS to CS SRVCC access transfers.

When the SCC AS receives SIP re-INVITE request(s) from the SC UE containing the protocol value "SIP" and the "cause" header field parameter with the value "487" after having performed an access transfer that was triggered by a SIP INVITE request due to STN-SR and after receiving a SIP BYE request containing the Reason header field containing the protocol value "Q.850" and the "cause" header field parameter with the value "31" (normal unspecified) on the target access leg, then the SCC AS shall:

1) not release the original source access leg on expiry of the timer described in subclause 12.3.8; and
2) treat the SIP re-INVITE request(s) as per procedures for removing and adding media as described in subclause 13.3.1.

NOTE 3: In procedures in subclause 13.3.1 the SCC AS starts forwarding SIP messages from the remote UE to the SC UE for the ongoing session as specified in 3GPP TS 24.229 [2] and the present specification.

12.3.3.5 PS to CS SRVCC cancelled by MME/SGSN or release of the target access leg for a session in an early dialog phase

12.3.3.5.1 SCC AS serving an originating user

This subclause describes the procedures for cancelling calls after the SCC AS have initiated an PS to CS SRVCC access transfer that was triggered by a SIP INVITE request due to STN-SR, a SIP INVITE request due to ATU-STI or a SIP INVITE request transferring additional session for PS to CS SRVCC for a session in an early dialog phase as specified in subclause 12.3.4.3 and subclause 12.3.4.4 where the SCC AS is serving a originating user.

If the SCC AS applies the procedures for PS to CS SRVCC access transfer for calls in alerting phase when serving a originating user or if the SCC AS applies the procedures for PS to CS SRVCC access transfer for calls in pre-alerting phase, then if the SCC AS receives a SIP BYE request or a SIP CANCEL request containing a Reason header field containing the protocol value "Q.850" and the "cause" header field parameter with the value "31" (normal unspecified) on:

- the target access leg of a session in the originating pre-alerting phase, if the SCC AS applies PS to CS SRVCC access transfer for calls in pre-alerting phase;
- the target access leg of a session in the alerting phase, if the SCC AS applies PS to CS SRVCC access transfer for calls in alerting phase; and

- the target access leg of an additional transferred session in the pre-alerting phase or in the alerting phase, if the SCC AS applies the MSC server assisted mid-call feature; or

- target of an additional transferred session if the SCC AS applies PS to CS SRVCC for calls in the pre-alerting phase or in alerting phase,

after having initiated an access transfer for a session which is still in early dialog state when the operator specific timer is still running, the SCC AS shall:

1) if a SIP BYE was received, send the SIP 200 (OK) response to the SIP BYE request;

1A) if a SIP CANCEL request was received, send the SIP 200 (OK) response to the SIP CANCEL request;

2) wait until the operator specific timer expires or until a SIP UPDATE request from the SC UE containing the protocol value "SIP" and the "cause" header field parameter with the value "487" is received; and

3) if the operator specific timer expires and no SIP UPDATE request from the SC UE containing the protocol value "SIP" and the "cause" header field parameter with the value "487" is received, cancel the call according to procedures in 3GPP TS 24.229 [2].

NOTE 1: All protocol values in the Reason header field other than "Q.850" and all other values of the "cause" header field parameter other than "31" (normal unspecified) will result in an immediate release of all dialogs associated with the source access leg and cancelling of all dialogs associated with the remote UE leg.

NOTE 2: The SCC AS assigns an operator specific timer to delay the release of the source access leg for PS to CS SRVCC access transfers.

When the SCC AS receives a SIP UPDATE request(s) containing the protocol value "SIP" and the "cause" header field parameter with the value "487" from the SC UE after having performed an access transfer that was triggered by a SIP INVITE request due to STN-SR, a SIP INVITE request due to ATU-STI or a SIP INVITE request transferring additional session for PS to CS SRVCC and after receiving a SIP BYE request or a SIP CANCEL request containing the Reason header field containing the protocol value "Q.850" and the "cause" header field parameter with the value "31" (normal unspecified) on the target access leg, then the SCC AS shall:

1) not release the original source access leg once the expiration of the timer described in subclause 12.3.8; and

2) treat the SIP UPDATE request(s) as per procedures for removing and adding media as described in subclause 13.3.1.

NOTE 3: By removing and adding media in subclause 13.3.1 the SCC AS starts forwarding SIP messages from the remote UE to the SC UE for the ongoing dialogs as specified in 3GPP TS 24.229 [2] and the present specification.

If the SCC AS has received a SIP 200 (OK) response from the SC UE prior to receiving the SIP UPDATE request from the SC UE, then on receipt of the SIP 200 (OK) response to the SIP UPDATE request sent to the remote UE, the SCC AS shall send a SIP 200 (OK) response to the remote UE. Upon receiving the SIP ACK request from the remote UE, the SCC AS shall send a SIP ACK request to the SC UE.

12.3.3.5.2 SCC AS serving a terminating user

This subclause describes the procedures for cancelling calls after the SCC AS have initiated an PS to CS SRVCC access transfer that was triggered by a SIP INVITE request due to STN-SR, a SIP INVITE request due to ATU-STI or a SIP INVITE request transferring additional session for PS to CS SRVCC for a session in an alerting phase as specified in subclause 12.3.4.2 and subclause 12.3.4.4 where the SCC AS is serving a terminating user.

If the SCC AS applies the procedures for PS to CS SRVCC access transfer for calls in alerting phase and when serving a terminating user, then if the SCC AS receives a SIP BYE request or a SIP CANCEL request containing the protocol value "Q.850" and the "cause" header field parameter with the value "31" (normal unspecified) cancelling SIP INVITE request due to STN-SR on:

- the target access leg of a session in the alerting phase, if the SCC AS applies PS to CS SRVCC for calls in alerting phase;

- the target access leg of an additional transferred session in the alerting phase, if the SCC AS applies the MSC server assisted mid-call feature; or
- target of an additional transferred session in the alerting phase, if the SCC AS applies PS to CS SRVCC for calls in alerting phase,

after having initiated an access transfer for a session which is still in the alerting phase when the operator specific timer is still running, then the SCC AS shall:

1) if a SIP BYE was received, send the SIP 200 (OK) response to the BYE request;

1A) if a SIP CANCEL request was received, send a SIP 200 (OK) response to the SIP CANCEL request;

2) wait until the operator specific timer expires or until a SIP UPDATE request from the SC UE containing the protocol value "SIP" and the "cause" header field parameter with the value "487" is received; and

3) if the operator specific timer expires and no SIP UPDATE request from the SC UE containing the protocol value "SIP" and the "cause" header field parameter with the value "487" is received then:

   a) send a SIP 486 (Busy) response to the SIP INVITE request due to to terminating filter criteria from the SC UE towards the remote UE as specified in 3GPP TS 24.229 [2]; and

   b) if a SIP CANCEL request was received, send a SIP 487 (Request Terminated) response to the SIP INVITE request due to STN-SR, the SIP INVITE request due to ATU-STI or the SIP INVITE request transferring additional session for PS to CS SRVCC as specified in 3GPP TS 24.229 [2].

NOTE 1: All protocol values in the Reason header field other than "Q.850" and all other values of the "cause" header field parameter other than "31" (normal unspecified) will result in an immediate release of the source access leg associated with the SC UE and the associated leg towards remote UE. Any other dialogs associated with the same user will remain in the early dialog phase. The procedure for handling a SIP BYE request or SIP CANCEL request with other values of the "cause" header field parameter than "31" (normal unspecified) is described in subclause 12.3.11.

NOTE 2: The SCC AS assigns an operator specific timer to delay the release of the source access leg for PS to CS SRVCC access transfers.

When the SCC AS receives a SIP UPDATE request(s) containing the protocol value "SIP" and the "cause" header field parameter with the value "487" from the SC UE after having performed an access transfer and after receiving a SIP BYE request or a SIP CANCEL request containing the Reason header field containing the protocol value "Q.850" and the "cause" header field parameter with the value "31" (normal unspecified) on the target access leg, then the SCC AS shall:

1) not release the original source access leg once the expiration of the timer as described in subclause 12.3.8;

2) treat the SIP UPDATE request(s) as per procedures for removing and adding media as described in subclause 13.3.1; and

3) start forwarding SIP messages from the remote UE to the SC UE for this dialog triggered by a SIP INVITE request due to STN-SR, a SIP INVITE request due to ATU-STI or a SIP INVITE request transferring additional session for PS to CS SRVCC for a session in an alerting phase as specified in 3GPP TS 24.229 [2] and the present specification.

If the SCC AS has received a SIP 200 (OK) response to the SIP INVITE requests due to terminating filter criteria from the SC UE prior to receiving the SIP UPDATE request from the SC UE, then on receipt of the SIP 200 (OK) response to the SIP UPDATE request sent to the remote UE, the SCC AS shall send a SIP 200 (OK) response to the remote UE. Upon receiving the SIP ACK request from the remote UE, the SCC AS shall send a SIP ACK request to the SC UE.

12.3.4 SCC AS procedures for PS to CS access transfer when call is in alerting phase or pre-alerting phase

12.3.4.1 General

The SCC AS shall apply the procedures for the PS to CS SRVCC for calls in alerting phase as described in subclauses 12.3.4.2 or 12.3.4.3 if:

NOTE 1: The transferable session can contain early dialogs supporting active speech media and video media components if the transferable session set was created due to vSRVCC otherwise the transferable session set can only contain early dialogs supporting active speech media component.

1. the Contact header field of the SIP INVITE request routed to the SCC AS due to a STN-SR includes the g.3gpp.srvcc-alerting media feature tag as specified in annex C; and

2. one of the following is true:

   A. there are one or more dialogs supporting sessions with speech media component or speech media and video media components existing for the served user identified in the transferable set in the P-Asserted-Identity header field, such that:

      a. all dialogs are early dialogs;

      b. SIP 180 (Ringing) response to SIP INVITE request was received in at least one of those early dialogs, all such SIP 180 (Ringing) responses are responses to the same SIP INVITE request and at least one of such SIP 180 (Ringing) responses was received in an early dialog supporting session with active speech media component;

      c. the Contact header field provided by the SC UE includes the g.3gpp.srvcc-alerting media feature tag as described in annex C; and

      d. the Feature-Caps header field provided by the SCC AS towards the SC UE includes the g.3gpp.srvcc-alerting feature-capability indicator as described in annex C; or

   B. there are several dialogs supporting sessions with speech media component for the served user identified in the P-Asserted-Identity header field, such that:

      a. there are one or more early dialogs and the remaining dialogs are confirmed dialogs;

      b. SIP 180 (Ringing) response to SIP INVITE request was received in at least one of those early dialogs, all such SIP 180 (Ringing) responses are responses to the same SIP INVITE request and at least one of such SIP 180 (Ringing) responses was received in an early dialog supporting session with active speech media component;

      c. all the confirmed dialogs support sessions with inactive speech media component;

      d. SCC AS does not apply the MSC server assisted mid-call feature as described in subclause 12.3.2;

      e. the Contact header field provided by the SC UE at the establishment of the early dialog(s) included the g.3gpp.srvcc-alerting media feature tag as described in annex C; and

      f. the Feature-Caps header field provided by the SCC AS towards the SC UE at the establishment of the early dialog(s) includes the g.3gpp.srvcc-alerting feature-capability indicator as described in annex C.

The SCC AS shall apply the procedures for the PS to CS SRVCC of originating call in pre-alerting phase as described in subclauses 12.3.4.3 if:

1) the Contact header field of the SIP INVITE request due to a STN-SR includes the g.3gpp.ps2cs-srvcc-orig-pre-alerting media feature tag as described in annex C; and

2) one of the following is true:

   A) there are zero, one or more dialogs supporting a session with speech media component in the transferable set for the served user identified in the P-Asserted-Identity header field and a SIP INVITE request was received from SC UE of the served user identified in the P-Asserted-Identity header field such that:

      a) all dialogs are early dialogs created by a SIP response to the SIP INVITE request;

      b) a final SIP response to the SIP INVITE request has not been sent yet;

      c) a SIP 180 (Ringing) response to the SIP INVITE request has not been sent yet in any existing early dialog created by a SIP response to the SIP INVITE request;

d) the SIP INVITE request included a Contact header field containing the g.3gpp.ps2cs-srvcc-orig-pre-alerting media feature tag as described in annex C; and

e) a SIP 1xx response to the SIP INVITE request was sent where the SIP 1xx response contained a Feature-Caps header field with the g.3gpp.ps2cs-srvcc-orig-pre-alerting feature-capability indicator as described in annex C; or

NOTE 2: SCC AS can have zero dialogs if all the early dialogs were terminated by 199 (Early Dialog Terminated) as described in RFC 6228 [80].

B) there are one or more dialogs supporting sessions with speech media component in the transferable set for the served user identified in the P-Asserted-Identity header field such that:

a) there are zero, one or more early dialogs and the remaining dialogs are confirmed dialogs;

b) all the confirmed dialogs support sessions with inactive speech media component;

c) SCC AS does not apply the MSC server assisted mid-call feature as described in subclause 12.3.2; and

d) a SIP INVITE request was received from SC UE of the served user identified in the P-Asserted-Identity header field such that:

- all early dialogs are created by a SIP response to the SIP INVITE request;
- a final SIP response to the SIP INVITE request has not been sent yet;
- a SIP 180 (Ringing) response to the SIP INVITE request has not been sent yet in any existing early dialog created by a SIP response to the SIP INVITE request;
- the SIP INVITE request included a Contact header field containing the g.3gpp.ps2cs-srvcc-orig-pre-alerting media feature tag as described in annex C; and
- a SIP 1xx response to the SIP INVITE request was sent where the SIP 1xx response contained a Feature-Caps header field with the g.3gpp.ps2cs-srvcc-orig-pre-alerting feature-capability indicator as described in annex C.

The SCC AS shall apply the procedures for the PS to CS SRVCC of terminating call in pre-alerting phase as described in subclauses 12.3.4.2 if:

1) the Contact header field of the SIP INVITE request due to a STN-SR includes the g.3gpp.ps2cs-srvcc-term-pre-alerting media feature tag as described in annex C; and

2) one of the following is true:

A) there are zero, one or more dialogs supporting a session with speech media component in the transferable set for the served user identified in the P-Asserted-Identity header field and a SIP INVITE request was sent to the SC UE of the served user identified in the P-Asserted-Identity header field such that:

a) all dialogs are early dialogs created by a SIP response to the SIP INVITE request;

b) a final SIP response to the SIP INVITE request has not been sent yet;

c) a SIP 180 (Ringing) response to the SIP INVITE request has not been sent yet in any existing early dialog created by a SIP response to the SIP INVITE request;

d) the SIP INVITE request included a Contact header field containing the g.3gpp.ps2cs-srvcc-term-pre-alerting media feature tag as described in annex C; and

e) a SIP 1xx response to the SIP INVITE request was received where the SIP 1xx response contained a Feature-Caps header field with the g.3gpp.ps2cs-srvcc-term-pre-alerting feature-capability indicator as described in annex C; or

NOTE 3: SCC AS can have zero dialogs if all the early dialogs were terminated by 199 (Early Dialog Terminated) as described in RFC 6228 [80].

B) there are one or more dialogs supporting sessions with speech media component in the transferable set for the served user identified in the P-Asserted-Identity header field such that:

a) there are zero, one or more early dialogs and the remaining dialogs are confirmed dialogs;

b) all the confirmed dialogs support sessions with inactive speech media component;

c) SCC AS does not apply the MSC server assisted mid-call feature as described in subclause 12.3.2; and

d) a SIP INVITE request was sent to the SC UE of the served user identified in the P-Asserted-Identity header field such that:

- all early dialogs are created by a SIP response to the SIP INVITE request;
- a final SIP response to the SIP INVITE request has not been received yet;
- a SIP 180 (Ringing) response to the SIP INVITE request has not been sent yet in any existing early dialog created by a SIP response to the SIP INVITE request;
- the SIP INVITE request included a Contact header field containing the g.3gpp.ps2cs-srvcc-term-pre-alerting media feature tag as described in annex C; and
- a SIP 1xx response to the SIP INVITE request was received where the SIP 1xx response contained a Feature-Caps header field with the g.3gpp.ps2cs-srvcc-term-pre-alerting feature-capability indicator as described in annex C.

The SCC AS shall apply the procedures for the PS to CS SRVCC of call in alerting phase as described in subclauses 12.3.4.4 if:

1. the Contact header field of the SIP INVITE request routed to the SCC AS due to a STN-SR includes the g.3gpp.srvcc-alerting media feature tag as described in annex C;

2. void;

3. void; and

4. one of the following is true:

   A. two or more dialogs supporting sessions with speech media component exist for the served user in the transferable set, such that:

      a. the Contact header fields provided by the SC UE at the establishment of sessions included the g.3gpp.srvcc-alerting media feature tag as described in annex C;

      b. the Feature-Caps header field provided by the SCC AS towards the SC UE at the establishment of sessions included the g.3gpp.srvcc-alerting feature-capability indicator as described in annex C;

      c. one dialog is a confirmed dialog with active speech media component and the remaining dialog(s) are early dialog(s); and

      d. SIP 180 (Ringing) response to SIP INVITE request was received in at least one of those early dialogs, all such SIP 180 (Ringing) responses are responses to the same SIP INVITE request and at least one of such SIP 180 (Ringing) responses was received in an early dialog supporting session with active speech media component;

   B. two or more dialogs supporting sessions with speech media component exist for the served user identified in the transferable session set, such that:

      a. the Contact header fields provided by the SC UE at the establishment of sessions included the g.3gpp.srvcc-alerting media feature tag as described in annex C;

      b. the Feature-Caps header field provided by the SCC AS towards the SC UE at the establishment of sessions included the g.3gpp.srvcc-alerting feature-capability indicator as described in annex C;

      c. one or more dialogs are a confirmed dialog with inactive speech media component and the remaining dialog(s) are early dialog(s);

      d. SIP 180 (Ringing) response to SIP INVITE request was received in at least one of those early dialogs, all such SIP 180 (Ringing) responses are responses to the same SIP INVITE request and at least one of such

SIP 180 (Ringing) responses was received in an early dialog supporting session with active speech media component; and

e. the SCC AS also applies the MSC server assisted mid-call feature as described in subclause 12.3.2; or

C. two or more dialogs supporting the sessions with speech media component exist for the served user identified in the transferable session set, such that:

a. the Contact header fields provided by the SC UE at the establishment of the sessions included the g.3gpp.srvcc-alerting media feature tag as described in annex C;

b. the Feature-Caps header field provided by the SCC AS towards the SC UE at the establishment of sessions included the g.3gpp.srvcc-alerting feature-capability indicator as described in annex C;

c. one or more dialogs are a confirmed dialogs with active speech media components, there are one or more dialogs that are confirmed dialogs with inactive speech media component and the remaining dialog(s) are early dialog(s);

d. SIP 180 (Ringing) response to SIP INVITE request was received in at least one of those early dialogs, all such SIP 180 (Ringing) responses are responses to the same SIP INVITE request and at least one of such SIP 180 (Ringing) responses was received in an early dialog supporting session with active speech media component; and

e. the SCC AS does not apply the MSC server assisted mid-call feature as described in subclause 12.3.2.

The SCC AS shall apply the procedures for the PS to CS SRVCC of originating call in pre-alerting phase as described in subclause 12.3.4.4 if:

1. the Contact header field of the SIP INVITE request routed to the SCC AS due to a STN-SR includes the g.3gpp.ps2cs-srvcc-orig-pre-alerting media feature tag as described in annex C; and

2. there are one or more dialogs supporting sessions with speech media component exist for the served user identified in the transferable session set such that:

A) there are zero, one or more early dialogs and the remaining dialog(s) are confirmed dialog(s);

B) all the confirmed dialogs support sessions with inactive speech media component;

C) the SCC AS also applies the MSC server assisted mid-call feature as described in subclause 12.3.2; and

D) a SIP INVITE request was received from the SC UE such that:

a) all the early dialogs are created by a SIP response to the SIP INVITE request;

b) a final SIP response to the SIP INVITE request has not been received yet;

c) a SIP 180 (Ringing) response to the SIP INVITE request has not been received yet in any existing early dialog created by a SIP response to the SIP INVITE request;

d) the SIP INVITE request received from the SC UE includes a Contact header field containing the g.3gpp.ps2cs-srvcc-orig-pre-alerting media feature tag as described in annex C; and

e) a SIP 1xx response to the SIP INVITE request was sent towards the SC UE where the SIP 1xx response contained a Feature-Caps header field with the g.3gpp.ps2cs-srvcc-orig-pre-alerting feature-capability indicator as described in annex C.

12.3.4.2 SCC AS procedures for PS to CS access transfer for terminating call in alerting phase or pre-alerting phase using PS to CS SRVCC procedure

When the session in the transferable session set is a terminating call not accepted yet the SCC AS shall associate the SIP INVITE request with the early dialog related to the terminating session in the transferable session set.

If the speech media component of the SDP offer in the SIP INVITE request due to STN-SR is the same as the speech media component of the SDP received in the source access leg of the session being transferred, the SCC AS shall send a SIP 183 (Session Progress) response to the SIP INVITE request due to STN-SR containing:

a) the SDP sent by the SCC AS in the source access leg of the session being transferred; and

b) the signalling elements described in subclause 6A.4.3A.

If the speech media component of the SDP offer in the SIP INVITE request due to STN-SR is different to the speech media component of the SDP received in the source access leg of the session being transferred and the remote UE has provided an Allow header field listing the SIP UPDATE method or has not provided an Allow header field, the SCC AS shall update the remote leg by sending a SIP UPDATE request towards the remote UE using the existing established dialog according as specified in 3GPP TS 24.229 [2]. The SCC AS shall populate the SIP UPDATE request with the SDP offer received in the SIP INVITE request due to STN-SR.

Upon receiving the SIP 200 (OK) response to the SIP UPDATE request from the remote UE, the SCC AS shall send a SIP 183 (Session Progress) response in response to the SIP INVITE request due to STN-SR towards the MSC server. The SCC AS shall populate the SIP 183 (Session Progress) response to the SIP INVITE request due to STN-SR with:

a) the SDP answer received in the SIP 200 (OK) response to the SIP UPDATE request; and

b) the signalling elements described in subclause 6A.4.3A.

If:

- the speech media component of the SDP offer in the SIP INVITE request due to STN-SR is different to the speech media component of the SDP received in the source access leg of the session being transferred and the remote UE has provided an Allow header field not listing the SIP UPDATE method; or

- SIP 405 (Method Not Allowed) response was received to the SIP UPDATE sent towards the remote UE;

then the SCC AS shall create a new early dialog by sending a SIP 183 (Session Progress) response to the SIP INVITE request due to STN-SR. The SCC AS shall populate the SIP 183 (Session Progress) response with:

1. an SDP answer:

   A) with c-line set to the unspecified address (0.0.0.0) if IPv4 or to a domain name within the ".invalid" DNS top-level domain in case of IPv6 as described in IETF RFC 6157 [74]; and

   B) including media of media types received in SDP offer of the SIP INVITE request due to STN-SR, which are also offered in the SIP INVITE request from the served user; and

2. the signalling elements described in subclause 6A.4.3A.

Upon receiving the SIP PRACK request from the target access leg, the SCC AS shall send a 200 (OK) response to the PRACK request and then send a SIP INFO request towards the MSC server as specified in 3GPP TS 24.229 [2] and IETF RFC 6086 [54] in the dialog created by the SIP INVITE request due to STN-SR. The SCC AS shall populate the SIP INFO request as follows:

1. include the Info-Package header field as specified in IETF RFC 6086 [54] with 3gpp.state-and-event info package name; and

2. include an application/vnd. 3gpp.state-and-event-info +xml XML body associated with the info package according to IETF RFC 6086 [54] and compliant to the XML schema specified in the clause D.2:

   A) if a SIP 180 (Ringing) response to the SIP INVITE request has already been sent in any of the existing early dialogs associated with the terminating call not accepted yet, with the state-info XML element containing "early" and the direction XML element containing "receiver"; or

   B) if the SCC AS supports the PS to CS SRVCC for terminating calls in pre-alerting phase and if a SIP 180 (Ringing) response to the SIP INVITE request has not been sent yet in any of the existing early dialogs associated with the terminating call not accepted yet, with the state-info XML element containing "pre-alerting" and the direction XML element containing "receiver".

Upon receiving the 200 (OK) to the PRACK request from the MSC server the SCC AS shall start forwarding messages from the remote UE to the MSC server for this early dialog related to the terminating session as specified in 3GPP TS 24.229 [2] and the present specification.

Upon receiving the SIP INFO request which includes an Info-Package header field containing 3gpp.state-and-event info package name and an application/vnd. 3gpp.state-and-event-info +xml XML body associated with the info package

according to IETF RFC 6086 [54] and compliant to the XML schema specified in the clause D.2 from the MSC Server with the event XML element containing "alerting-started", the SCC AS shall send a SIP 180 (Ringing) response to the SIP INVITE request received earlier from the remote UE as specified in 3GPP TS 24.229 [2].

Upon receiving the SIP INFO request which includes an Info-Package header field containing 3gpp.state-and-event info package name and an application/vnd. 3gpp.state-and-event-info +xml XML body associated with the info package according to IETF RFC 6086 [54] and compliant to the XML schema specified in the clause D.2 from the MSC Server with the event XML element containing "call-accepted", the SCC AS shall send as specified in 3GPP TS 24.229 [2]:

1) a SIP 200 (OK) response to the SIP INVITE request received earlier from the remote UE indicating that the called party has answered the call;

2) if the SIP 2xx (OK) response was received to the SIP UPDATE sent towards the remote UE or the speech media component of the SDP offer in the SIP INVITE request due to STN-SR is the same as the speech media component of the SDP received in the source access leg of the session being transferred, a SIP 200 (OK) response to the SIP INVITE request due to STN-SR towards the MSC server to indicate the successful access transfer populated with the signalling elements described in subclause 6A.4.3A; and

3) if:

   - the remote UE has provided an Allow header field not listing the SIP UPDATE method and the speech media component of the SDP offer in the SIP INVITE request due to STN-SR is different to the speech media component of the SDP received in the source access leg of the session being transferred; or

   - a SIP 405 (Method Not Allowed) response was received to the SIP UPDATE sent towards the remote UE;

   then when a SIP ACK request is received on the remote leg, send a SIP re-INVITE request on the remote leg as specified in 3GPP TS 24.229 [2]. The SCC AS shall populate the SIP re-INVITE request with the SDP offer received in the SIP INVITE request due to STN-SR as specified in 3GPP TS 24.229 [2]. Upon receiving the SIP 200 (OK) response to the SIP re-INVITE request, the SCC AS shall send SIP ACK request on the remote leg and shall send a SIP 200 (OK) response to the SIP INVITE request due to STN-SR as specified in 3GPP TS 24.229 [2], using a dialog different to the dialog of SIP 183 (Session Progress) response to the SIP INVITE request due to STN-SR. The SCC AS shall populate the SIP 200 (OK) response with:

   a) the SDP answer received in the SIP 200 (OK) response to the SIP re-INVITE request as specified in 3GPP TS 24.229 [2]; and

   b) the signalling elements described in subclause 6A.4.3A.

The SCC AS shall remove non-transferred audio and video media components and superfluous session as specified in subclause 12.3.8.

12.3.4.3 SCC AS procedures for PS to CS access transfer for originating call in alerting phase or pre-alerting phase using PS to CS SRVCC procedure

When the session in the transferable session set is an originating call not accepted yet the SCC AS shall associate the SIP INVITE request due to STN-SR with an early dialog or early dialogs related to the originating call.

If the SCC AS receives a SIP 18x response on the remote leg after receiving a SIP INVITE request due to STN-SR or a SIP INVITE request due to ATU-STI, and this SIP 18x response does not require use of reliable provisional responses, the SCC AS shall:

- store this SIP 18x response; and

- if a P-Early-Media header field is received in the SIP 18x response, store the P-Early-Media header field.

The SCC AS shall store the received SIP 18x responses separately for each early dialog. If the SCC AS has already stored a SIP 18x response for an early dialog and receives another SIP 18x response for the same early dialog, the SCC AS may remove the stored SIP 18x response for that early dialog and shall store the new SIP 18x response for that early dialog.

The SCC AS shall store the received P-Early-Media header field separately for each early dialog. If the SCC AS has already stored a P-Early-Media header field received in a SIP 18x response for an early dialog, and receives another SIP 18x response for the same early dialog containing a P-Early-Media header field, the SCC AS may remove the stored P-Early-Media header field for that early dialog and shall store the new P-Early-Media header field for that early dialog,

NOTE: The P-Early-Media header field is stored separately to prepare for the case that a subsequent SIP 18x response does not contain a P-Early-Media header field.

If there is only one early dialog related to the originating call not accepted yet available for the served user and if the speech media component of the SDP offer in the SIP INVITE request due to STN-SR is the same as the speech media component of the SDP received in the source access leg of the session being transferred, the SCC AS shall send a SIP 183 (Session Progress) response to the SIP INVITE request due to STN-SR containing:

- the SDP sent by the SCC AS in the source access leg of the session being transferred;
- if the SIP INVITE request due to STN-SR contains a P-Early-Media header field with the "supported" parameter and if the SCC AS has received a P-Early-Media header field in a SIP message in the dialog of the original SIP INVITE request sent to the remote leg, include a P-Early-Media header field containing the value of the last P-Early-Media header field received in a SIP message in the dialog of the original SIP INVITE request sent to the remote leg; and
- the signalling elements described in subclause 6A.4.3A.

If there is only one early dialog related to the originating call not accepted yet available for the served user, the remote UE has provided an Allow header field listing the SIP UPDATE method or has not provided Allow header field, the remote UE has provided the SDP answer, and the speech media component of the SDP offer in the SIP INVITE request due to STN-SR is different with the speech media component of the SDP received in the source access leg of the session being transferred, the SCC AS shall update the remote leg by sending a SIP UPDATE request towards the remote UE using the existing early dialog as specified in 3GPP TS 24.229 [2]. The SCC AS shall populate the SIP UPDATE request with the SDP offer received in the SIP INVITE request due to STN-SR.

Upon receiving the SIP 200 (OK) response to the SIP UPDATE request from the remote UE, the SCC AS shall send a SIP 183 (Session Progress) response in response to the SIP INVITE request due to STN-SR towards the MSC server. The SCC AS shall populate the SIP 183 (Session Progress) response to the SIP INVITE request due to STN-SR with:

- the SDP answer received in the SIP 200 (OK) response to the SIP UPDATE request;
- if the SIP INVITE request due to STN-SR contains a P-Early-Media header field with the "supported" parameter and if the SCC AS has received a P-Early-Media header field in a SIP message in the dialog of the SIP UPDATE request, include a P-Early-Media header field containing the value of the last P-Early-Media header field received in a SIP message in the dialog of the SIP UPDATE request; and
- the signalling elements described in subclause 6A.4.3A.

If there are more than one early dialogs related to the originating call not accepted yet available for the served user due to forking as described in 3GPP TS 24.229 [2], for each existing early dialog where the speech media component of the SDP offer in the SIP INVITE request due to STN-SR is the same as the speech media component of the SDP received in the source access leg of the session being transferred, the SCC AS shall create a new early dialog by sending a SIP 183 (Session Progress) response to the SIP INVITE request due to STN-SR containing:

- the SDP sent by the SCC AS in the source access leg of the session being transferred;
- if the SIP INVITE request due to STN-SR contains a P-Early-Media header field with the "supported" parameter and if the SCC AS has received a P-Early-Media header field in a SIP message in the dialog of the original SIP INVITE request sent to the remote leg, include a P-Early-Media header field containing the value of the last P-Early-Media header field received in a SIP message in the dialog of the original SIP INVITE request sent to the remote leg; and
- the signalling elements described in subclause 6A.4.3A.

If there are more than one early dialogs related to the originating call not accepted yet available for the served user due to forking as described in 3GPP TS 24.229 [2], for each existing early dialog where the speech media component of the SDP offer in the SIP INVITE request due to STN-SR is not the same as the speech media component of the SDP received in the source access leg of the session being transferred, the remote UE has provided an Allow header field listing SIP UPDATE method or has not provided Allow header field, and the remote UE provided SDP answer, the SCC AS shall update the remote leg(s) by sending SIP UPDATE request(s) simultaneously towards remote UE using such early dialog(s) as specified in 3GPP TS 24.229 [2]. The SCC AS shall populate each SIP UPDATE request with the SDP offer received in the SIP INVITE request due to STN-SR. Upon receiving each SIP 200 (OK) response to the SIP UPDATE request from the remote UE, the SCC AS shall create a new early dialog by sending a SIP 183 (Session

Progress) response in response to the SIP INVITE request due to STN-SR towards the MSC server. The SCC AS shall populate the SIP 183 (Session Progress) response to the SIP INVITE request due to STN-SR with:

- the SDP answer received in the SIP 200 (OK) response to the SIP UPDATE request;
- If the SIP INVITE request due to STN-SR contains a P-Early-Media header field with the "supported" parameter and if the SCC AS has received a P-Early-Media header field in a SIP message in the dialog of the SIP UPDATE request, include a P-Early-Media header field containing the value of the last P-Early-Media header field received in a SIP message in the dialog of the SIP UPDATE request; and
- the signalling elements described in subclause 6A.4.3A.

If one or more early dialogs related to the originating call not accepted yet are available for the served user, and in each such early dialog:

- the remote UE of the early dialog has provided an Allow header field not listing the SIP UPDATE method, and the speech media component of the SDP offer in the SIP INVITE request due to STN-SR is not the same as the speech media component of the SDP received in each such source access leg of the session being transferred; or
- SIP 405 (Method Not Allowed) response was received to the SIP UPDATE sent towards the remote UE in the early dialog;

the SCC AS shall create a new dialog by sending a SIP 183 (Session Progress) response to the SIP INVITE request due to STN-SR. The SCC AS shall populate the SIP 183 (Session Progress) response with:

1. an SDP answer:

   A) with c-line set to the unspecified address (0.0.0.0) if IPv4 or to a domain name within the ".invalid" DNS top-level domain in case of IPv6 as described in IETF RFC 6157 [74]; and

   B) including media of media types received in SDP offer of the SIP INVITE request due to STN-SR, which are also offered in the SIP INVITE request from the served user; and

2. the signalling elements described in subclause 6A.4.3A.

If the SCC AS supports the PS to CS SRVCC for originating calls in pre-alerting phase and if there are no early dialogs related to the originating call not accepted yet available for the served user and there is a SIP INVITE request from the served user for which a final SIP response has not been received yet, the SCC AS shall send SIP 183 (Session Progress) response to the SIP INVITE request due to STN-SR. The SCC AS shall populate the SIP 183 (Session Progress) response with:

1. an SDP answer:

   A) with c-line set to the unspecified address (0.0.0.0) if IPv4 or to a domain name within the ".invalid" DNS top-level domain in case of IPv6 as described in IETF RFC 6157 [74]; and

   B) including media of media types received in SDP offer of the SIP INVITE request due to STN-SR, which are also offered in the SIP INVITE request from the served user; and

2. the signalling elements described in subclause 6A.4.3A.

Upon receiving the first SIP PRACK request from the target access leg, the SCC AS shall send a 200 (OK) to the PRACK response and then send a SIP INFO request towards the MSC server as specified in 3GPP TS 24.229 [2] and IETF RFC 6086 [54] in the dialog created by the SIP INVITE request due to STN-SR. The SCC AS shall populate the SIP INFO request as follows:

1. include the Info-Package header field as specified in IETF RFC 6086 [54] with 3gpp.state-and-event info package name; and

2. include application/vnd.3gpp.state-and-event-info+xml XML body associated with the info package according to IETF RFC 6086 [54] and containing a XML body compliant to the XML schema specified in the annex D.2:

   A) if a SIP 180 (Ringing) response to the SIP INVITE request has already been forwarded to the served SC UE before receiving the INVITE due to STN-SR or due to ATU-STI, with the state-info XML element containing "early" the direction XML element containing "initiator"; and

B) if the SCC AS supports the PS to CS SRVCC for originating calls in pre-alerting phase and if a SIP 180 (Ringing) response to the SIP INVITE request has not beenforwarded to the served SC UE before receiving the INVITE due to STN-SR or due to ATU-STI, with the state-info XML element containing "pre-alerting" and the direction XML element containing "initiator".

Upon receiving the 200 (OK) response to the SIP INFO request the SCC AS shall:

1. start forwarding SIP messages from the remote UE to the MSC server as specified in 3GPP TS 24.229 [2] and the present specification for dialogs where a PRACK request is received from the MSC server;

2. if SIP 18x responses were stored after receiving the SIP INVITE request due to STN-SR or SIP INVITE request due to ATU-STI, then for each early dialog where a PRACK request is received from the MSC server:

   a) forward the that was stored most recently IP 18x responses to the MSC server; and

   b) if a P-Early-Media header field is stored, include the P-Early-Media header field that was stored most recently in the SIP 18x response.

If a reliable SIP 1xx response or a SIP 2xx response is received on the remote leg, the SIP response is to the SIP INVITE request from the served user, the SIP response contains an SDP answer, and an SDP answer has not been received from the remote UE on the dialog of the SIP response yet, the SCC AS shall:

1) if the speech media component of the SDP offer in the SIP INVITE request due to STN-SR is the same as the speech media component of the SDP received in the source access leg of the session being transferred, forward the SIP response on the target access leg as a SIP response to the SIP INVITE request due to STN-SR; and

2) if the speech media component of the SDP offer in the SIP INVITE request due to STN-SR is different to the speech media component of the SDP received in the source access leg of the session being transferred:

   A) if the SIP 1xx response is received, send a SIP PRACK request on the remote leg as specified in 3GPP TS 24.229 [2]. The SCC AS shall populate the SIP PRACK request with the SDP offer received in the SIP INVITE request due to STN-SR as specified in 3GPP TS 24.229 [2]. Upon receiving the SIP 200 (OK) response to the SIP PRACK request, the SCC AS shall send a SIP 183 (Session Progress) response to the SIP INVITE request due to STN-SR as specified in 3GPP TS 24.229 [2]. The SCC AS shall populate the SIP 183 (Session Progress) response with:

      a) the SDP answer received in the SIP 200 (OK) response to the SIP PRACK request as specified in 3GPP TS 24.229 [2];

      b) if the SIP INVITE request due to STN-SR contains a P-Early-Media header field with the "supported" parameter and if the SCC AS has received a P-Early-Media header field in a SIP message in the dialog of the SIP PRACK request, a P-Early-Media header field containing the value of the last P-Early-Media header field received in a SIP message in the dialog of the SIP PRACK request; and

      c) the signalling elements described in subclause 6A.4.3A.

   B) if the SIP 2xx response is received, send a SIP ACK request on the remote leg as specified in 3GPP TS 24.229 [2] and send a SIP UPDATE request or SIP re-INVITE request on the remote leg as specified in 3GPP TS 24.229 [2]. The SCC AS shall populate the SIP UPDATE request or SIP re-INVITE request with the SDP offer received in the SIP INVITE request due to STN-SR as specified in 3GPP TS 24.229 [2]. Upon receiving the SIP 200 (OK) response to the SIP UPDATE request or SIP re-INVITE request, the SCC AS shall send a SIP 200 (OK) response to the SIP INVITE request due to STN-SR as specified in 3GPP TS 24.229 [2]. The SCC AS shall populate the SIP 200 (OK) response with:

      a) the SDP answer received in the SIP 200 (OK) response to the SIP UPDATE request or SIP re-INVITE request as specified in 3GPP TS 24.229 [2]; and

      b) the signalling elements described in subclause 6A.4.3A.

If a SIP 2xx response is received on the remote leg, the SIP response is to the SIP INVITE request from the served user, and an SDP answer has already been received from the remote UE on the dialog of the SIP response:

1) if the speech media component of the SDP offer in the SIP INVITE request due to STN-SR is the same as the speech media component of the SDP received in the source access leg of the session being transferred, then

forward the SIP response as specified in 3GPP TS 24.229 [2] on the target access leg as a SIP response to the SIP INVITE request due to STN-SR; and

2) if the speech media component of the SDP offer in the SIP INVITE request due to STN-SR is not the same as the speech media component of the SDP received in the source access leg of the session being transferred:

   A) if the SDP offer received in the SIP INVITE request due to STN-SR has already been sent to the remote UE on the dialog of the SIP 2xx response, accepted with an subsequent SDP answer, the subsequent SDP answer was sent in a 183 (Session Progress) response on a target access leg, and SIP 2xx response does not contain an SDP answer, then forward the SIP 2xx response on the target access leg as a SIP response to the SIP INVITE request due to STN-SR;

   B) if the SDP offer received in the SIP INVITE request due to STN-SR has already been sent to the remote UE on the dialog of the SIP 2xx response, accepted with an subsequent SDP answer, the subsequent SDP answer was sent in a 183 (Session Progress) response on a target access leg, and SIP 2xx response contains an SDP answer, then remove the SDP body from the SIP 2xx response or replace the SDP body in the SIP 2xx response with the subsequent SDP answer, and forward the SIP 2xx response as specified in 3GPP TS 24.229 [2] on the target access leg as a SIP response to the SIP INVITE request due to STN-SR; and

   C) if:

      - the SDP offer received in the SIP INVITE request due to STN-SR has not been sent to the remote UE on the dialog of the SIP response yet; or

      - the SDP offer received in the SIP INVITE request due to STN-SR has already been sent to the remote UE in a SIP UPDATE request within the dialog of the SIP response and the SIP UPDATE request was rejected with SIP 405 (Method Not Allowed) response;

   send a SIP ACK request on the remote leg as specified in 3GPP TS 24.229 [2] and send a SIP re-INVITE request on the remote leg as specified in 3GPP TS 24.229 [2]. The SCC AS shall populate the SIP re-INVITE request with the SDP offer received in the SIP INVITE request due to STN-SR as specified in 3GPP TS 24.229 [2]. Upon receiving the SIP 200 (OK) response to the SIP re-INVITE request, the SCC AS shall send a SIP ACK request on the remote leg as specified in 3GPP TS 24.229 [2] and shall send a SIP 200 (OK) response to the SIP INVITE request due to STN-SR as specified in 3GPP TS 24.229 [2] using a dialog different to the dialog of SIP 183 (Session Progress) response to the SIP INVITE request due to STN-SR. The SCC AS shall populate the SIP 200 (OK) response with:

   a) the SDP answer received in the SIP 200 (OK) response to the SIP re-INVITE request as specified in 3GPP TS 24.229 [2]; and

   b) the signalling elements described in subclause 6A.4.3A.

The SCC AS shall remove non-transferred audio and video media components and superfluous sessions as specified in subclause 12.3.8.

12.3.4.4 SCC AS procedures for PS to CS access transfer of additional call

This section contains procedures related to transfer of additional transferred session which is not accepted yet.

In order to transfer the additional transferred session, the SCC AS shall send a SIP REFER request according to 3GPP TS 24.229 [2], IETF RFC 4488 [20] and IETF RFC 3515 [13] as updated by IETF RFC 6665 [81] and IETF RFC 7647 [90] in the dialog created by the SIP INVITE request due to STN-SR; or the SCC AS shall send a SIP REFER request according to 3GPP TS 24.229 [2], IETF RFC 3515 [13] as updated by IETF RFC 6665 [81] and IETF RFC 7647 [90] in the dialog created by the SIP INVITE request due to ATU-STI for PS to CS SRVCC. The SCC AS shall populate the SIP REFER request as follows:

1. the Refer-Sub header field with value "false" as specified in IETF RFC 4488 [20];

2. the Supported header field with value "norefersub" as specified in IETF RFC 4488 [20];

NOTE 0: IETF RFC 3261 [19] recommends user agent client to include a Supported header field in any SIP request, listing option tags for extensions to SIP supported by the user agent client, that can be applied by the user agent server to the SIP response. In the step above, the SCC AS is mandated to include at least "norefersub" option tag in the Supported header field.

3. the Refer-To header field containing the additional transferred session SCC AS URI for PS to CS SRVCC, where the URI also includes the following header fields containing the information related to the additional transferred session:

   A. if an early dialog supporting the additional transferred session exists, the Target-Dialog header field populated as specified in IETF RFC 4538 [11], containing the dialog identifier of an early dialog supporting session of the SC UE;

   B. if the SCC AS supports the PS to CS SRVCC for originating calls in pre-alerting phase, if no early dialog supporting the additional transferred session exists, there is a SIP INVITE request from the served user for which a final SIP response has not been received yet and if an early dialog supporting the additional transferred session existed and was terminated, the Target-Dialog header field populated as specified in IETF RFC 4538 [11], containing the dialog identifier on the source access leg of the early dialog supporting the additional transferred session which existed and was terminated;

NOTE 1: Early dialog can be terminated by SIP 199 (Early Dialog Terminated) response.

   C. the Require header field populated with the option tag value "tdialog";

   D. if an early dialog supporting the additional transferred session exists, the To header field populated as specified in IETF RFC 3261 [19], containing the value of the P-Asserted-Identity provided by the remote UE during the session establishment;

   E. the From header field populated as specified in IETF RFC 3261 [19], containing the value of the P-Asserted-Identity provided by the SC UE during the session establishment;

   F. the Content-Type header field with "application/sdp";

   G. if an early dialog supporting the additional transferred session exists, the header field with hname "body" populated with an SDP body describing the media streams as negotiated in the session with the remote UE;

   H. if the SCC AS supports the PS to CS SRVCC for originating calls in pre-alerting phase, no early dialog supporting the additional transferred session exists, there is a SIP INVITE request from the served user for which a final SIP response has not been received yet, the header field with hname "body" populated with the SDP offer received in the SIP INVITE request from the served user; and

   I. optionally the P-Asserted-Identity URI header field containing value of the P-Asserted-Identity header field of the received SIP INVITE request;

4. if a SIP 180 (Ringing) response to the SIP INVITE request from the served user has already been received in any of the existing early dialogs associated with the additional transferred session, application/vnd.3gpp.state-and-event-info+xml MIME body with the state-info XML element containing "early" and the direction XML element containing:

   A. if terminating call, the "receiver"; and

   B. if originating call, the "initiator";

5. if the SCC AS supports the PS to CS SRVCC for originating calls in pre-alerting phase, the additional transferred session was originated by the SC UE and if a SIP 180 (Ringing) response to the SIP INVITE request from the served user has not been received yet in any of the existing early dialogs associated with the additional transferred session, application/vnd.3gpp.state-and-event-info+xml MIME body with the state-info XML element containing "pre-alerting" and the direction XML element containing the "initiator"; and

6. if:

   A. SCC AS supports CS to PS SRVCC;

   B. the SIP INVITE request due to STN-SR contains Accept header field with application/vnd.3gpp.srvcc-ext+xml MIME type;

   C. a private user identity of a UE (i.e. other than those according to 3GPP TS 23.003 [12], subclause 20.3.3) is associated with the C-MSISDN in the P-Asserted-Identity header field of the SIP INVITE request due to STN-SR;

   D. a binding of a contact address exists for the private user identity of the UE:

a) where the g.3gpp.cs2ps-srvcc media feature tag as described in annex C is associated with the contact address of the UE; and

b) such that SIP REGISTER request which registered the binding contained a Feature-Caps header field with the g.3gpp.atcf feature-capability indicator as described in annex C and with g.3gpp.cs2ps-srvcc feature-capability indicator as described in annex C;

E. the CS to PS SRVCC capability indication is indicated for the private user identity of the UE; and

F. the private user identity of the UE has the CS to PS SRVCC allowed indication in the subscription data;

then:

A. a MIME body of application/vnd.3gpp.srvcc-ext+xml MIME type:

a) containing ATU management URI of the ATCF serving the SC UE;

NOTE 2: The ATCF management URI of the ATCF is the URI contained in the g.3gpp.atcf-mgmt-uri feature-capability indicator included in a Feature-Caps header field of the SIP REGISTER request, which registered the binding for the private user identity of the UE.

b) containing C-MSISDN; and

c) not indicating that information relate to a registration of MSC server with IMS.

When the SCC AS receives the SIP INVITE request transferring additional session for PS to CS SRVCC, the SCC AS shall:

- if the Target-Dialog header field of the SIP INVITE request transferring additional session for PS to CS SRVCC identifies an existing early dialog, associate the SIP INVITE request transferring additional session for PS to CS SRVCC with the SIP early dialog i.e. identify the Source Access Leg;

NOTE 3: The SIP dialog on the Source Access Leg is identified by matching the dialog ID present in Target-Dialog header field (see IETF RFC 4538 [11]) of the SIP INVITE with a dialog in early state.

NOTE 4: By a SIP dialog in early state, it is meant an early SIP dialog which has been created by a provisional response to the initial SIP INVITE request, but for which the SIP 2xx response has not yet been sent or received;

- if the SCC AS supports the PS to CS SRVCC for originating calls in pre-alerting phase, if the Target-Dialog header field of the SIP INVITE request transferring additional session for PS to CS SRVCC identifies an early dialog which has already been terminated, associate the SIP INVITE request transferring additional session for PS to CS SRVCC with the early dialog i.e. identify the Source Access Leg;

- if the SCC AS is unable to associate the SIP INVITE with a unique dialog in early state, send a SIP 480 (Temporarily Unavailable) response to reject the SIP INVITE request relating to the access transfer and not processes the remaining steps;

- if the number of media lines in the Target Access Leg is less than the number of media lines in the Source Access Leg or the media type for the corresponding media lines is not the same as in the original session, send a SIP 4xx response to reject the SIP INVITE request relating to the access transfer and not process the remaining steps;

- if an early dialog exists on the remote leg of the additional transferred session, for each existing early dialog on the remote leg where the remote UE has provided an Allow header field listing the SIP UPDATE method or has not provided an Allow header field, SDP answer has already been sent or received in a reliable provisional response and the speech media component of the SDP offer in the SIP INVITE request transferring additional session for PS to CS SRVCC is not the same as the speech media component of the SDP received in the source access leg of the early dialog, send a SIP UPDATE request(s) towards the remote UE(s) using such existing early dialog(s) which were created by the same SIP INVITE request as the Source Access Leg. The SCC AS shall populate the SIP UPDATE request(s) with a new SDP offer, following the rules specified in 3GPP TS 24.229 [2], containing the following media information:

a) the media characteristics as received in the SIP INVITE request transferring additional session for PS to CS SRVCC received on the Target Access Leg for media streams whose port is not set to zero; and

b) for the media streams in the SIP INVITE request transferring additional session for PS to CS SRVCC whose port is set to zero, include the corresponding media characteristics of those streams from the Source Access Leg;

- if one or more early dialogs exist on the remote leg of the additional transferred session, for each early dialog where the speech media component of the SDP offer in the SIP INVITE request transferring additional session for PS to CS SRVCC is the same as the speech media component of the SDP received in the source access leg of the early dialog, create a new early dialog by sending a SIP 183 (Session Progress) response to the SIP INVITE request transferring additional session for PS to CS SRVCC. In the SIP 183 (Session Progress) response, the SCC AS shall:

  a) include the SDP sent by the SCC AS in the source access leg of the early dialog;

  b) if the SIP INVITE request transferring additional session for PS to CS SRVCC contains a P-Early-Media header field with the "supported" parameter and the SCC AS has received a P-Early-Media header field in a SIP message in the dialog of the original SIP INVITE request sent to the remote leg, include a P-Early-Media header field containing the value of the last P-Early-Media header field received in a SIP message in the dialog of the original SIP INVITE request sent to the remote leg; and

  c) include the signalling elements described in subclause 6A.4.3A; and

- if one or more early dialogs exist on the remote leg of the additional transferred session and:

  a) the remote UE has provided an Allow header field not listing the SIP UPDATE method; and the speech media component of the SDP offer in the SIP INVITE request transferring additional session for PS to CS SRVCC is not the same as the speech media component of the SDP received in the source access leg of each such early dialog, or

  b) SIP 405 (Method Not Allowed) response was received to the SIP UPDATE sent towards the remote UE;

  create a new early dialog by sending a SIP 183 (Session Progress) response to SIP INVITE request transferring additional session for PS to CS SRVCC. The SCC AS shall populate the SIP 183 (Session Progress) response with:

  a) an SDP answer with c-line set to the unspecified address (0.0.0.0) if IPv4 or to a domain name within the ".invalid" DNS top-level domain in case of IPv6 as described in IETF RFC 6157 [74]; and

  b) the signalling elements described in subclause 6A.4.3A.

If an early dialog exists on the remote leg then when receiving SIP 2xx response to the SIP UPDATE request, the SCC AS shall create a new early dialog by sending SIP 183 (Session Progress) response to the SIP INVITE request transferring additional session for PS to CS SRVCC. The SCC AS shall populate the SIP response as follows:

1. if the Remote Leg is an early dialog originated by the remote UE, include a Recv-Info header field containing the g.3gpp.state-and-event package name;

2. include the SDP answer received in the SIP 200 (OK) response to the SIP UPDATE request;

3. if the SIP INVITE request transferring additional session for PS to CS SRVCC contains a P-Early-Media header field with the "supported" parameter and if the SCC AS has received a P-Early-Media header field in a SIP message in the dialog of the SIP UPDATE request, include a P-Early-Media header field containing the value of the last P-Early-Media header field received in a SIP message in the dialog of the SIP UPDATE request; and

4. the signalling elements described in subclause 6A.4.3A.

For each received SIP PRACK request from the MSC server the SCC AS shall send the 200 (OK) response to the PRACK request according to 3GPP TS 24.229 [2] and start forwarding SIP messages from the remote UE for the associated early dialog as specified in 3GPP TS 24.229 [2] and the present specification.

If a reliable SIP 1xx response or a SIP 2xx response is received on the remote leg of the additional transferred session, the SIP response is to the SIP INVITE request from the served user, the SIP response contains an SDP answer and an SDP answer has not been received from the remote UE on the dialog of the SIP response yet, the SCC AS shall:

1. if the speech media component of the SDP offer in the SIP INVITE request transferring additional session for PS to CS SRVCC is the same as the speech media component of the SDP received in the source access leg of the

additional transferred session, forward the SIP response on the target access leg as a SIP response to the SIP INVITE request transferring additional session for PS to CS SRVCC; and

2. if the speech media component of the SDP offer in the SIP INVITE request transferring additional session for PS to CS SRVCC is not the same as the speech media component of the SDP received in the source access leg of the additional transferred session:

a) if the SIP 1xx response is received, send a SIP PRACK request on the remote leg as specified in 3GPP TS 24.229 [2]. The SCC AS shall populate the SIP PRACK request with the SDP offer received in the SIP INVITE request transferring additional session for PS to CS SRVCC as specified in 3GPP TS 24.229 [2]. Upon receiving the SIP 200 (OK) response to the SIP PRACK request, the SCC AS shall send a SIP 183 (Session Progress) response to the SIP INVITE request transferring additional session for PS to CS SRVCC as specified in 3GPP TS 24.229 [2]. The SCC AS shall populate the SIP 183 (Session Progress) response as follows:

   - include the SDP answer received in the SIP 200 (OK) response to the SIP PRACK request as specified in 3GPP TS 24.229 [2];

   - if the SIP INVITE request transferring additional session for PS to CS SRVCC contains a P-Early-Media header field with the "supported" parameter and if the SCC AS has received a P-Early-Media header field in a SIP message in the dialog of the SIP PRACK request, include a P-Early-Media header field containing the value of the last P-Early-Media header field received in a SIP message in the dialog of the SIP PRACK request; and

   - the signalling elements described in subclause 6A.4.3A; and

b) if the SIP 2xx response is received, send a SIP ACK request on the remote leg as specified in 3GPP TS 24.229 [2] and send a SIP UPDATE request or SIP re-INVITE request on the remote leg as specified in 3GPP TS 24.229 [2]. The SCC AS populate the SIP UPDATE request or SIP re-INVITE request with the SDP offer received in the SIP INVITE request transferring additional session for PS to CS SRVCC as specified in 3GPP TS 24.229 [2]. Upon receiving the SIP 200 (OK) response to the SIP UPDATE request or SIP re-INVITE request, the SCC AS shall send a SIP 200 (OK) response to the SIP INVITE request transferring additional session for PS to CS SRVCC as specified in 3GPP TS 24.229 [2]. The SCC AS shall populate the SIP 200 (OK) response with:

   - the SDP answer received in the SIP 200 (OK) response to the SIP UPDATE request or SIP re-INVITE request as specified in 3GPP TS 24.229 [2]; and

   - the signalling elements described in subclause 6A.4.3A.

If the Remote Leg is an early dialog originated by the remote UE then when receiving the SIP INFO request inside the Target Access Leg containing:

1. an Info-Package header field as specified in IETF RFC 6086 [54] with 3gpp.state-and-event info package name; and

2. application/vnd.3gpp.state-and-event-info+xml XML body associated with the info package according to IETF RFC 6086 [54] and with the event XML element containing "call-accepted" to indicate that the called party has answered the call;

then the SCC AS shall:

1. send SIP 200 (OK) response to the SIP INVITE request to the remote UE;

2. if the SIP 2xx (OK) response was received to the SIP UPDATE sent towards the remote UE or if the speech media component of the SDP offer in the SIP INVITE request transferring additional session for PS to CS SRVCC is the same as the speech media component of the SDP received in the source access leg of the additional transferred session, send SIP 200 (OK) response to the SIP INVITE request over the Target Access Leg;and

3. if:

   - the remote UE has provided an Allow header field not listing the SIP UPDATE method and the speech media component of the SDP offer in the SIP INVITE request transferring additional session for PS to CS SRVCC is different to the speech media component of the SDP received in the source access leg of the additional transferred session; or

- a SIP 405 (Method Not Allowed) response was received to the SIP UPDATE sent towards the remote UE;

and the speech media component of the SDP offer in the SIP INVITE request transferring additional session for PS to CS SRVCC is not the same as the speech media component of the SDP received in the source access leg of the additional transferred session, then when a SIP ACK request is received on the remote leg, send a SIP re-INVITE request on the remote leg as specified in 3GPP TS 24.229 [2]. The SCC AS shall populate the SIP re-INVITE request with the SDP offer received in the SIP INVITE request transferring additional session for PS to CS SRVCC as specified in 3GPP TS 24.229 [2]. Upon receiving the SIP 200 (OK) response to the SIP re-INVITE request, the SCC AS send SIP ACK request on the remote leg and shall shall send a SIP 200 (OK) response to the SIP INVITE request transferring additional session for PS to CS SRVCC as specified in 3GPP TS 24.229 [2], using a dialog different to the dialog of SIP 183 (Session Progress) response to the SIP INVITE request transferring additional session for PS to CS SRVCC. The SCC AS shall populate the SIP 200 (OK) response with:

a) the SDP answer received in the SIP 200 (OK) response to the SIP re-INVITE request as specified in 3GPP TS 24.229 [2]; and

b) the signalling elements described in subclause 6A.4.3A.

If a SIP 2xx response is received on the remote leg, the SIP response is to the SIP INVITE request from the served user, and an SDP answer has already been received from the remote UE on the dialog of the SIP response:

1) if the speech media component of the SDP offer in the SIP INVITE request transferring additional session for PS to CS SRVCC is the same as the speech media component of the SDP received in the source access leg of the additional transferred session, then forward the SIP response as specified in 3GPP TS 24.229 [2] on the target access leg as a SIP response to the SIP INVITE request transferring additional session for PS to CS SRVCC; and

2) if the speech media component of the SDP offer in the SIP INVITE request transferring additional session for PS to CS SRVCC is not the same as the speech media component of the SDP received in the source access leg of the additional transferred session:

A) if:

- the SDP offer received in the SIP INVITE request transferring additional session for PS to CS SRVCC has not been sent to the remote UE on the dialog of the SIP response yet; or

- the SDP offer received in the SIP INVITE request transferring additional session for PS to CS SRVCC has already been sent to the remote UE in a SIP UPDATE request within the dialog of the SIP response and the SIP UPDATE request was rejected with SIP 405 (Method Not Allowed) response;

send a SIP ACK request on the remote leg as specified in 3GPP TS 24.229 [2] and send a SIP re-INVITE request on the remote leg as specified in 3GPP TS 24.229 [2]. The SCC AS shall populate the SIP re-INVITE request with the SDP offer received in the SIP INVITE request transferring additional session for PS to CS SRVCC as specified in 3GPP TS 24.229 [2]. Upon receiving the SIP 200 (OK) response to the SIP re-INVITE request, the SCC AS shall send a SIP ACK request on the remote leg as specified in 3GPP TS 24.229 [2] and shall send a SIP 200 (OK) response to the SIP INVITE request transferring additional session for PS to CS SRVCC as specified in 3GPP TS 24.229 [2] using a dialog different to the dialog of SIP 183 (Session Progress) response to the SIP INVITE request transferring additional session for PS to CS SRVCC. The SCC AS shall populate the SIP 200 (OK) response with:

a) the SDP answer received in the SIP 200 (OK) response to the SIP re-INVITE request as specified in 3GPP TS 24.229 [2]; and

b) the signalling elements described in subclause 6A.4.3A;

B) if the SDP offer received in the SIP INVITE request transferring additional session for PS to CS SRVCC has already been sent to the remote UE on the dialog of the SIP 2xx response, accepted with an subsequent SDP answer, the subsequent SDP answer was sent in a 183 (Session Progress) response on a target access leg, and SIP 2xx response does not contain an SDP answer, then forward the SIP 2xx response on the target access leg as a SIP response to the SIP INVITE request transferring additional session for PS to CS SRVCC; and

C) if the SDP offer received in the SIP INVITE request transferring additional session for PS to CS SRVCC has already been sent to the remote UE on the dialog of the SIP 2xx response, accepted with an subsequent SDP answer, the subsequent SDP answer was sent in a 183 (Session Progress) response on a target access leg, and SIP 2xx response contains an SDP answer, then remove the SDP body from the SIP 2xx response or replace

the SDP body in the SIP 2xx response with the subsequent SDP answer, and forward the SIP 2xx response as specified in 3GPP TS 24.229 [2] on the target access leg as a SIP response to the SIP INVITE request transferring additional session for PS to CS SRVCC.

12.3.5 SCC AS procedures for PS to CS access transfer: PS to CS SRVCC enhancement using ATCF

Upon receiving a SIP INVITE request due to ATU-STI for PS to CS SRVCC, the SCC AS shall:

1) if there is a Target-Dialog header field in the SIP INVITE request:

   A) determine the transferable session set which are all the sessions of the SC UE whose private user identity is associated with Correlation MSISDN that is contained in the P-Asserted-Identity header field of the SIP INVITE request;

   B) determine the session that is to be transferred which is a session:

      a) in the transferable session set;

      b) is in the confirmed dialog state; and

      c) with active speech media component which has been made active most recently; and

   C) if the session that is to be transferred is for the same dialog as the dialog identifier in the Target-Dialog header field in the SIP INVITE request, then perform the procedures described for SIP INVITE request due to STN-SR in subclause 12.3.0B with one of the following options dependent on operator policy:

      a) if:

         - the SDP offer in the SIP INVITE request contains speech media component only and the speech media component of the SDP offer in the SIP INVITE request is the same as the speech media component of the SDP negotiated by the ATCF in the session being transferred; or

         - the SDP offer in the SIP INVITE request contains speech media component and video media component and the speech media component and the video media component of the SDP offer in the SIP INVITE request is the same as the speech media component and the video media component of the SDP negotiated by the ATCF in the session being transferred;

         then the SCC AS shall:

         i) not send a SIP re-INVITE request towards remote UE;

         ii) send a SIP 200 (OK) response to the SIP INVITE request containing:

            - the SDP negotiated by SCC AS towards ATCF in the session being transferred; and

            - the signalling elements described in subclause 6A.4.3A; and

         iii) upon receipt of the ACK request from the ATCF, start forwarding SIP messages from the remote UE to the MSC server for this session with active speech media component; or

      b) if confirmed dialogs supporting a session with active speech media component exist in the transferable session set the SCC AS shall send a SIP re-INVITE request towards the remote UE and in a new SDP offer, include the media characteristics as received in the SIP INVITE request due to ATU-STI, by following the rules of 3GPP TS 24.229 [2];

NOTE: handling when it is determined that there is no session to be transferred or when the dialog identifier in the Target-Dialog header field in the SIP INVITE request identifies a dialog other than the session being transferred is out of scope of this release of this document.

   D) if the session identified by the dialog identifier in the Target-Dialog header field is a session of the SC UE whose private user identity is associated with C-MSISDN that is contained in the P-Asserted-Identity header field of the SIP INVITE request and:

      1) is in an early dialog state; or

2) is in a confirmed dialog state and contains inactive speech media component;

then

1) if the session is in an early dialog state, perform the procedures described for SIP INVITE requests due to STN-SR in subclause 12.3; and

2) if the session is in a confirmed dialog state and contains inactive speech media component, perform the procedures described for SIP INVITE requests due to STN-SR in subclause 12.3.2;

2) if there is no Target-Dialog header field in the SIP INVITE request:

a) perform the procedures described for SIP INVITE requests due to STN-SR in subclause 12.3.0B.

12.3.6 SCC AS procedures for PS to CS access transfer, vSRVCC

12.3.6.0 Determine the transferable session set

When the SCC AS receives a SIP INVITE request for audio and video due to STN-SR on the target access leg the SCC AS shall determine the transferable session set.

A session is in the transferable session set when the session:

1) is a session of the SC UE whose private user identity is associated with the C-MSISDN that is contained in the P-Asserted-Identity header field of the SIP INVITE request; and

2) is a session supporting active speech and video media components.

The SCC AS shall:

1) if the conditions in subclause 12.3.6.2 for applying the PS to CS transfer of a call in an alerting phase feature are fulfilled, follow the procedures in subclause 12.3.6.2; and

2) if the conditions in 1) are not satisfied follow the procedure in subclause 12.3.6.1.

12.3.6.1 General

When the SCC AS receives a SIP INVITE request for audio and video due to STN-SR on the target access leg the SCC AS shall associate the SIP INVITE request with a session:

- within the transferable session set;
- with active speech and video media components that was most recently made active; and
- the related dialog is in confirmed state.

If no confirmed dialogs supporting a session with active speech and video media component exists in the transferable session set the SCC AS shall:

1) send a SIP 480 (Temporarily Unavailable) response to reject the SIP INVITE request due to STN-SR; and

2) if the transferable session set contains dialogs supporting sessions with speech media and/or video media components:

a) if the speech media and/or video media components are the only media component in the dialog then release the remote leg as specified in 3GPP TS 24.229 [2]; and

b) if the speech media and/or video media component are not the only media component in the dialog then modify the remote leg and remove the speech media component as specified in 3GPP TS 24.229 [2].

If confirmed dialogs supporting a session with active speech and video media components exist in the transferable session set the SCC AS shall:

1) send a SIP re-INVITE request towards the remote UE and in a new SDP offer, include the media characteristics as received in the SIP INVITE request due to STN-SR, by following the rules of 3GPP TS 24.229 [2]; or

2) send a SIP re-INVITE request towards the remote UE according to the conditions depicted in subclause 12.3.5 and in a new SDP offer, include the media characteristics as received in the SIP INVITE request due to ATU-STI for PS to CS SRVCC, by following the rules of 3GPP TS 24.229 [2].

Upon receiving the SIP 2xx response to the SIP re-INVITE request the SCC AS shall send the SIP 200 (OK) response to the SIP INVITE request due to STN-SR on the target access leg containing:

1) the relevant media parameter of the SDP answer in the received response, by following the rules of 3GPP TS 24.229 [2]; and

2) the signalling elements described in subclause 6A.4.3A.

The SCC AS shall remove non-transferred audio media and video media components and superfluous session as specified in subclause 12.3.8.

12.3.6.2 SCC AS procedures for PS to CS access transfer when call is in alerting phase, vSRVCC

The SCC AS shall apply the procedures for access transfer for calls in alerting phase in subclauses 12.3.4.2 and 12.3.4.3 according to the conditions specified in subclause 12.3.4.1 with the following differences:

- the SCC AS receives a SIP INVITE request for audio and video due to STN-SR instead of a SIP INVITE request due to STN-SR; and

- one or more early dialogs contain both speech and video media components.

12.3.6.3 SCC AS procedures for PS to CS access transfer: vSRVCC enhancement using ATCF

The SCC AS shall follow the procedures in subclause 12.3.5 with the following difference:

- instead of performing the procedures for SIP INVITE request due to STN-SR in subclause 12.3.1, the SCC AS performs the procedures for SIP INVITE request for audio and video due to STN-SR in subclause 12.3.6.

12.3.6.4 SCC AS procedures for vSR-VCC, abnormal case

The SCC AS shall follow the procedures in subclause 12.3.3 with the following difference:

- access transfer was triggered by the SCC AS receiving a SIP INVITE request for audio and video due to STN-SR instead of a SIP INVITE due to STN-SR.

12.3.7 SCC AS procedures for handling of SIP OPTIONS request

When the SCC AS receives a SIP OPTIONS request on the target access leg and determines for the C-MSISDN in the P-Asserted-Identity header field that the session that was most recently made active is a session with active speech and video media components, the SCC AS shall send a SIP 200 (OK) response to the SIP OPTIONS request with an SDP body containing "m=" lines for audio and video.

When the SCC AS receives a SIP OPTIONS request on the target access leg and determines for the C-MSISDN in the P-Asserted-Identity header field that the session that was most recently made active is a session with an active speech media component but not an active video media component, the SCC AS shall send a SIP 200 (OK) response to the SIP OPTIONS request with an SDP body containing an "m=" line for audio but not video.

If the SCC AS supports the MSC server assisted mid-call feature and:

- has received the g.3gpp.mid-call media feature tag as described in annex C is included in the Contact header field of the SIP INVITE request due to originating filter criteria (as described in subclause 7); or

- has received the g.3gpp.mid-call media feature tag as described in annex C from the SIP 2xx response to the SIP INVITE request due to terminating filter criteria (as described in subclause 8)

then when the SCC AS receives a SIP OPTIONS request on the target access leg and determines for the C-MSISDN in the P-Asserted-Identity header field that there are no sessions with an active speech media component, but there are sessions that contain an inactive speech media component, the SCC AS shall send a SIP 200 (OK) response to the SIP OPTIONS request with an SDP body containing an "m=" line for audio.

NOTE: If the session that is most recently made inactive contains inactive speech and video media components, the SCC AS only returns the "m=" line for audio and not for video.

12.3.8 Removal of non-transferred audio media components and superfluous sessions

Upon receiving the SIP ACK request from target access leg, and after an operator specific timer has expired, the SCC AS shall:

1) for each session where no in-dialog request has been received in the source access leg of the session with transferred media component(s) within the operator defined time:

   a) if the session is a session with an active or inactive media component, send a SIP BYE request on the source access leg;

   b) if the session is an early dialog on originating side send a SIP 404 (Not Found) response on the source access leg; and

   c) if the session is an early dialog on terminating side send a SIP CANCEL request on the source access leg; and

NOTE 1: The SC UE will receive the SIP request or response only if the SC UE is using Gm after the PS-CS access transfer is completed.

NOTE 2: Delaying the SIP request or response as described above allows an ICS UE to add Gm control if needed and an SC UE to reuse the PS dialog in case of SRVCC cancellation.

2) for each session in the transferable session set for which the speech media component, or the speech media and video media component in case of vSRVCC, was not transferred:

   a) if the speech media component or the speech media and video media components is the only media component(s) of the session, release remote leg and source access leg; and

   b) if the speech media component or the speech media and video media components are not the only media components of the session, modify the remote leg and source access leg and remove the media component(s).

NOTE 3: In case of a SIP INVITE request due to STN-SR, video media components are not removed or causing release of the remote leg.

12.3.9 Charging correlation

The SCC AS shall include in SIP 1xx and SIP 2xx responses to the SIP INVITE request due to STN-SR the P-Charging-Vector header field as specified in 3GPP TS 24.229 [2], subclause 5.7.5.1 and include the "related-icid" header field parameter containing the ICID value of the source access leg in the P-Charging-Vector header field. Additionally, if an "icid-generated-at" header field was generated for the source access leg, SCC AS shall include the "related-icid-generated-at" header field parameter containing the host name or IP address included in the "icid-generated-at" header field parameter of the source access leg.

12.3.10 SCC AS procedures for CS to PS SRVCC

12.3.10.1 Distinction of requests

The SCC AS needs to distinguish the following initial SIP requests:

1) SIP INVITE requests routed to the SCC AS due to ATU-STI for CS to PS SRVCC in the Request-URI. In the procedures below, such requests are known as "SIP INVITE requests due to ATU-STI for CS to PS SRVCC".

2) SIP CANCEL requests cancelling the SIP INVITE requests due to ATU-STI for CS to PS SRVCC. In the procedures below, such requests are known as "SIP CANCEL requests cancelling INVITE due to ATU-STI for CS to PS SRVCC".

3) SIP INVITE requests routed to the SCC AS due to additional transferred session SCC AS URI for CS to PS SRVCC in the Request-URI. In the procedures below, such requests are known as "SIP INVITE request transferring additional session".

12.3.10.2 First session transfer

12.3.10.2.1 General

If SCC AS supports CS to PS SRVCC, upon receiving a SIP INVITE request due to ATU-STI for CS to PS SRVCC, the SCC AS shall:

1) determine the transferable dialog set which are all the dialogs (both early and confirmed):

   A) where the g.3gpp.ics media feature tag with value "server" was indicated in Contact header field provided by the served user;

   B) of the served user whose private user identity is associated with C-MSISDN that is contained in the P-Asserted-Identity header field of the SIP INVITE request due to ATU-STI for CS to PS SRVCC; and

   C) supporting a session;

2) if there is a Target-Dialog header field in the SIP INVITE request due to ATU-STI for CS to PS SRVCC:

   A) determine the dialog being transferred as the dialog with the dialog identifier of the Target-Dialog header field in the SIP INVITE request due to ATU-STI for CS to PS SRVCC; and

   B) if the determined dialog being transferred identifies a dialog in the transferable dialog set, continue handling the procedures in the subclause 12.3.10.2.2; and

NOTE: Handling when the dialog identifier in the Target-Dialog header field in the SIP INVITE request due to ATU-STI for CS to PS SRVCC identifies a non existing dialog is out of scope of this release of this document.

3) if there is no Target-Dialog header field in the SIP INVITE request due to ATU-STI for CS to PS SRVCC and if the transferable dialog set is not empty:

   A) if SCC AS supports the CS to PS SRVCC with the assisted mid-call feature according to operator policy, the SIP INVITE request due to ATU-STI for CS to PS SRVCC contains an Accept header field containing the application/vnd.3gpp.mid-call+xml MIME type and if a dialog:

      a) in the transferable dialog set;

      b) which is a confirmed dialog; and

      c) supporting a session with speech media component;

      exists, then continue handling the procedures in the subclause 12.3.10.2.3 for the dialog and do not handle the remaining procedures of this subclause; and

   B) if SCC AS supports the CS to PS SRVCC for calls in alerting phase according to operator policy, the SIP INVITE request due to ATU-STI for CS to PS SRVCC contains an Accept header field containing the application/vnd.3gpp.state-and-event-info+xml MIME type and if a dialog:

      a) in the transferable dialog set;

      b) which is an early dialog;

      c) for which SIP 180 (Ringing) response has been sent or received; and

      d) supporting a session with speech media component;

      exists:

      a) if the dialog was originated by the served user, then continue handling the procedures in the subclause 12.3.10.2.4 for the dialog and do not handle the remaining procedures of this subclause; and

      b) if the dialog was originated by the remote UE, then continue handling the procedures in the subclause 12.3.10.2.5 for the dialog and do not handle the remaining procedures of this subclause.

12.3.10.2.2 Transfer of session with active speech media component

If SCC AS supports CS to PS SRVCC, in order to transfer the determined dialog being transferred, the SCC AS shall:

1) associate the SIP INVITE request due to ATU-STI for CS to PS SRVCC with the remote leg of the determined dialog being transferred;

2) if the speech media component of the SDP offer in the SIP INVITE request due to ATU-STI for CS to PS SRVCC is the same as the speech media component of the SDP negotiated by the ATCF in session supported by the determined dialog being transferred:

   A) send a SIP 200 (OK) response to the SIP INVITE request due to ATU-STI for CS to PS SRVCC according to 3GPP TS 24.229 [2]. The SCC AS shall populate the SIP 200 (OK) response with:

      a) signalling elements described in subclause 6A.4.3; and

      b) the SDP negotiated by SCC AS towards ATCF in the determined dialog being transferred; and

3) if the speech media component of the SDP offer in the SIP INVITE request due to ATU-STI for CS to PS SRVCC differs from the speech media component of the SDP negotiated by the ATCF in the determined dialog being transferred:

   A) send SIP re-INVITE request towards the remote UE inside the remote leg of the determined dialog being transferred according to 3GPP TS 24.229 [2]. The SCC AS shall populate the SIP re-INVITE request with the SDP offer which includes the media characteristics as received in the SIP INVITE request due to ATU-STI for CS to PS SRVCC.

Upon receiving a SIP 2xx response to the SIP re-INVITE request sent towards the remote UE, the SCC AS shall:

1) send a SIP 200 (OK) response to the SIP INVITE request due to ATU-STI for CS to PS SRVCC according to 3GPP TS 24.229 [2]. The SCC AS shall populate the SIP 200 (OK) response with:

   A) signalling elements as described in subclause 6A.4.3; and

   B) the SDP answer received in the SIP 2xx response to the SIP re-INVITE request.

Upon receiving SIP ACK request associated with the SIP 200 (OK) response to the SIP INVITE request due to ATU-STI for CS to PS SRVCC, the SCC AS shall:

1) release the source access leg of the determined dialog being transferred; and

2) continue handling the procedures in the subclause 12.3.10.3.

12.3.10.2.3 Transfer of session with inactive speech media component

If SCC AS supports CS to PS SRVCC, in order to transfer the determined dialog being transferred, the SCC AS shall:

1) associate the SIP INVITE request due to ATU-STI for CS to PS SRVCC with the remote leg of the determined dialog being transferred; and

2) send SIP re-INVITE request towards the remote UE in the remote leg of the determined dialog being transferred according to 3GPP TS 24.229 [2]. The SCC AS shall populate the SIP re-INVITE request with:

   A) the SDP offer which includes the media characteristics as received in the SIP INVITE request due to ATU-STI for CS to PS SRVCC; and

   B) set the directionality of the speech media component in the SDP offer as used in the session with remote UE.

Upon receiving SIP 2xx response to the SIP re-INVITE request sent towards the remote UE, the SCC AS shall send a SIP 200 (OK) response to the SIP INVITE request due to ATU-STI for CS to PS SRVCC according to 3GPP TS 24.229 [2]. The SCC AS shall populate the SIP 200 (OK) response with:

1) signalling elements as described in subclause 6A.4.3;

2) the SDP answer received in the SIP 2xx response to the SIP re-INVITE request; and

3) the Feature-Caps header field containing the g.3gpp.ti feature-capability indicator with value of the g.3gpp.ti media feature tag in the Contact header field received in the source access leg of the determined dialog being transferred.

Upon receiving SIP ACK request associated with the SIP 200 (OK) response to the SIP INVITE request due to ATU-STI for CS to PS SRVCC, the SCC AS shall:

1) release the source access leg of the determined dialog being transferred; and

2) continue handling the procedures in the subclause 12.3.10.3.

12.3.10.2.4 Transfer of originating session in alerting phase

If SCC AS supports CS to PS SRVCC, in order to transfer the determined dialog being transferred, the SCC AS shall:

1) associate the SIP INVITE request due to ATU-STI for CS to PS SRVCC with the remote leg of the determined dialog being transferred; and

2) send SIP UPDATE request towards the remote UE in the remote leg of the determined dialog being transferred according to 3GPP TS 24.229 [2]. The SCC AS shall populate the SIP UPDATE request with the SDP offer which includes the media characteristics as received in the SIP INVITE request due to ATU-STI for CS to PS SRVCC. If several early dialogs on the remote leg were established by the SIP INVITE request establishing the determined dialog being transferred, the SCC AS shall send SIP UPDATE request to each such early dialog.

Upon receiving a SIP 2xx response to the SIP UPDATE request sent towards the remote UE, the SCC AS shall establish a new early dialog by sending a SIP 180 (Ringing) response to the SIP INVITE request due to ATU-STI for CS to PS SRVCC according to 3GPP TS 24.229 [2]. The SCC AS shall populate the SIP 180 (Ringing) response with:

1) signalling elements described in subclause 6A.4.3;

2) the SDP answer received in the SIP 2xx response to the SIP UPDATE request;

3) the Feature-Caps header field containing the g.3gpp.ti feature-capability indicator with value of the g.3gpp.ti media feature tag in the Contact header field received in the source access leg of the determined dialog being transferred; and

4) if the SIP INVITE request due to ATU-STI for CS to PS SRVCC contains a P-Early-Media header field with the "supported" parameter and if the SCC AS has received a P-Early-Media header field in a SIP message in the dialog of the SIP UPDATE request, a P-Early-Media header field containing the value of the last P-Early-Media header field received in a SIP message in the dialog of the SIP UPDATE request.

Upon receiving SIP PRACK request associated with the SIP 180 (Ringing) response to the SIP INVITE request due to ATU-STI for CS to PS SRVCC, the SCC AS shall:

1) reject the source access leg of the determined dialog being transferred with SIP 404 (Not Found) response; and

2) continue handling the procedures in the subclause 12.3.10.3.

12.3.10.2.5 Transfer of terminating alerting session

If SCC AS supports CS to PS SRVCC, in order to transfer the determined dialog being transferred, the SCC AS shall:

1) associate the SIP INVITE request due to ATU-STI for CS to PS SRVCC with the remote leg of the determined dialog being transferred; and

2) send SIP UPDATE request towards the remote UE in the remote leg of the determined dialog being transferred according to 3GPP TS 24.229 [2]. The SCC AS shall populate the SIP UPDATE request with the SDP offer which includes the media characteristics as received in the SIP INVITE request due to ATU-STI for CS to PS SRVCC.

Upon receiving a SIP 2xx response to the SIP UPDATE request sent towards the remote UE, the SCC AS shall send a SIP 183 (Session Progress) response to the SIP INVITE request due to ATU-STI for CS to PS SRVCC according to 3GPP TS 24.229 [2]. The SCC AS shall populate the SIP 183 (Session Progress) with:

1) signalling elements described in subclause 6A.4.3;

2) the SDP answer received in the SIP 2xx response to the SIP UPDATE request;

3) the Recv-Info header field with the 3gpp.state-and-event info package name; and

4) the Feature-Caps header field containing the g.3gpp.ti feature-capability indicator with value of the g.3gpp.ti media feature tag in the Contact header field received in the source access leg of the determined dialog being transferred.

Upon receiving the SIP PRACK request from the SC UE associated with the SIP 183 (Session Progress) response to the SIP INVITE request due to ATU-STI for CS to PS SRVCC, the SCC AS shall:

1) send a SIP INFO request towards the SC UE as specified in 3GPP TS 24.229 [2] and IETF RFC 6086 [54] in the dialog created by SIP INVITE request due to ATU-STI for CS to PS SRVCC. The SCC AS shall populate the SIP INFO request as follows:

   A) include the Info-Package header field with 3gpp.state-and-event info package name; and

   B) include an application/vnd.3gpp.state-and-event-info+xml XML body associated with the info package according to IETF RFC 6086 [54] and compliant to the XML schema specified in the clause D.2 with the state-info XML element containing "early" and the direction XML element containing "receiver"; and

2) cancel the source access leg of the determined dialog being transferred.

Upon receiving the SIP INFO request which includes an Info-Package header field containing 3gpp.state-and-event info package name and an application/vnd.3gpp.state-and-event-info+xml XML body associated with the info package according to IETF RFC 6086 [54] and compliant to the XML schema specified in the clause D.2 from the SC UE with the event XML element containing "call-accepted", the SCC AS shall:

1) send a SIP 200 (OK) response to the SIP INVITE request received earlier from the remote UE as specified in 3GPP TS 24.229 [2]; and

2) send a SIP 200 (OK) response to the SIP INVITE request due to ATU-STI for CS to PS SRVCC as specified in 3GPP TS 24.229 [2] populated as described in subclause 6A.4.3.

Upon receiving the SIP CANCEL request cancelling SIP INVITE request due to ATU-STI for CS to PS SRVCC, the SCC AS shall:

1) send a SIP 200 (OK) response to the SIP CANCEL request;

2) send a SIP response to the SIP INVITE request received earlier from the remote UE as specified in 3GPP TS 24.229 [2]. The SCC AS shall populate the SIP response with:

   A) if the SIP CANCEL request contains a Reason header field with protocol "SIP", then status code and reason text from the Reason header field of the SIP CANCEL request; and

   B) if the SIP CANCEL request does not contain a Reason header field with protocol "SIP", then 486 (Busy) status code and reason text; and

3) send a SIP 487 (Request Terminated) response to the SIP INVITE request due to ATU-STI for CS to PS SRVCC as specified in 3GPP TS 24.229 [2].

Upon receiving the SIP ACK request on the target access leg of the determined dialog being transferred, the SCC AS shall cancel the source access leg of the determined dialog being transferred.

12.3.10.3 Additional session transfer

12.3.10.3.1 General

If SCC AS supports CS to PS SRVCC, if SCC AS supports the CS to PS SRVCC with the assisted mid-call feature according to operator policy and if the SIP INVITE request due to ATU-STI for CS to PS SRVCC contains an Accept header field containing the application/vnd.3gpp.mid-call+xml MIME type then for each dialog:

1) in the transferable dialog set;

2) which is a confirmed dialog;

3) supporting a session with speech media component; and

4) other than the dialog of the source access leg associated with the SIP INVITE request due to ATU-STI for CS to PS SRVCC;

the SCC AS shall perform the procedures in subclause 12.3.10.3.2.

If SCC AS supports the CS to PS SRVCC for calls in alerting phase according to operator policy and if the SIP INVITE request due to ATU-STI for CS to PS SRVCC contains an Accept header field containing the application/vnd.3gpp.state-and-event-info+xml MIME type then for each dialog:

i) in the transferable dialog set;

ii) which is an early dialog;

ii) for which SIP 180 (Ringing) response has been sent or received;

iv) supporting a session with speech media component; and

v) other than the dialog of the source access leg associated with the SIP INVITE request due to ATU-STI for CS to PS SRVCC;

the SCC AS shall perform the procedures in subclause 12.3.10.3.2.

If transfer of any dialog in the transferable dialog set has not been initiated, the SCC AS shall continue handling the procedures in the subclause 12.3.10.4.

12.3.10.3.2 Additional session transfer initiation

If SCC AS supports CS to PS SRVCC, in order to transfer the determined dialog being transferred, the SCC AS shall send a SIP REFER request according to 3GPP TS 24.229 [2], IETF RFC 4488 [20] and IETF RFC 3515 [13] as updated by IETF RFC 6665 [81] and IETF RFC 7647 [90] in the dialog created by the SIP INVITE request due to STN-SR. The SCC AS shall populate the SIP REFER request as follows:

1. the Refer-Sub header field with value "false" as specified in IETF RFC 4488 [20];

2. the Supported header field with value "norefersub" as specified in IETF RFC 4488 [20];

NOTE: IETF RFC 3261 [19] recommends user agent client to include a Supported header field in any SIP request, listing option tags for extensions to SIP supported by the user agent client, that can be applied by the user agent server to the SIP response. In the step above, the SCC AS is mandated to include at least "norefersub" option tag in the Supported header field.

3. the Refer-To header field containing the additional transferred session SCC AS URI for CS to PS SRVCC, where the URI also includes the following header fields containing the information related to the determined dialog being transferred:

   A. the Target-Dialog header field populated as specified in IETF RFC 4538 [11], containing the dialog identifier of the determined dialog being transferred;

   B. the Require header field populated with the option tag value "tdialog";

   C. if the remote UE of the remote leg of the determined dialog being transferred did not request privacy then the To URI header field populated as specified in IETF RFC 3261 [19], containing the value of the P-Asserted-Identity provided by the remote UE during the session establishment;

   D. the From header field populated as specified in IETF RFC 3261 [19], containing the value of the P-Asserted-Identity provided by the SC UE during the session establishment;

   E. the Content-Type header field with "application/sdp"; and

   F. the header field with hname "body" populated with an SDP body describing the media streams as negotiated in the session with the remote UE; and

4. if the determined dialog being transferred is a confirmed dialog, an application/vnd.3gpp.mid-call+xml MIME body; and

5. if the determined dialog being transferred is an early dialog:

   A. application/vnd.3gpp.state-and-event-info+xml MIME body with the state-info XML element containing "early" and the direction XML element containing:

      a. if terminating call, the "receiver"; and

      b. if originating call, the "initiator".

Upon receiving the SIP INVITE request transferring additional session, the SCC AS shall:

1) if the dialog identifier in the Target-Dialog header field of the SIP INVITE request identifies a dialog:

   A) where the asserted identity of the participating served user belongs to the same subscription as the asserted identity of the sender of the SIP INVITE request: and

   B) supporting a session with speech media component:

   then:

   A) determine the additional dialog being transferred as the dialog with the dialog identifier of the Target-Dialog header field in the SIP INVITE request transferring additional session;

   B) associate the SIP INVITE request transferring additional session with the remote leg of the determined additional dialog being transferred;

   C) if the dialog is a confirmed dialog, continue handling the procedures in the subclause 12.3.10.3.3;

   D) if the dialog is an early dialog established by served user, continue handling the procedures in the subclause 12.3.10.3.4; and

   E) if the dialog is an early dialog established by remote UE, continue handling the procedures in the subclause 12.3.10.3.5.

If receiving the SIP 3xx response, 4xx response or 6xx response to the SIP REFER request or if the SIP INVITE request transferring additional session is not received within operator defined time after the SIP REFER request sending, the SCC AS shall release, cancel or reject the remote leg, the source access leg and the target access leg of the determined dialog being transferred.

12.3.10.3.3 Transfer of session with inactive speech media component

If SCC AS supports CS to PS SRVCC, in order to transfer the determined additional dialog being transferred, the SCC AS shall:

1) send SIP re-INVITE request towards the remote UE in the remote leg of the determined additional dialog being transferred according to 3GPP TS 24.229 [2]. The SCC AS shall populate the SIP re-INVITE request with the SDP offer which includes the media characteristics as received in the SIP INVITE request transferring additional session.

Upon receiving a SIP 2xx response to the SIP re-INVITE request sent towards the remote UE, the SCC AS shall:

1) send a SIP 200 (OK) response to the SIP INVITE request transferring additional session according to 3GPP TS 24.229 [2]. The SCC AS shall populate the SIP 200 (OK) response with:

   A) signalling elements described in subclause 6A.4.3;

   B) the SDP answer received in the SIP 2xx response to the SIP re-INVITE request; and

   C) the Feature-Caps header field containing the g.3gpp.ti feature-capability indicator with value of the g.3gpp.ti media feature tag in the Contact header field received in the source access leg of the determined additional dialog being transferred.

Upon receiving SIP ACK request associated with the SIP 200 (OK) response to the SIP INVITE request due to ATU-STI for CS to PS SRVCC, the SCC AS shall:

1) release the source access leg of the determined dialog being transferred; and

2) continue handling the procedures in the subclause 12.3.10.4.

12.3.10.3.4 Transfer of originating session in alerting phase

If SCC AS supports CS to PS SRVCC, in order to transfer the determined additional dialog being transferred, the SCC AS shall:

1) send SIP UPDATE request towards the remote UE in the remote leg of the determined additional dialog being transferred according to 3GPP TS 24.229 [2]. The SCC AS shall populate the SIP UPDATE request with the SDP offer which includes the media characteristics as received in the SIP INVITE request transferring additional session. If several early dialogs on the remote leg were established by the SIP INVITE request establishing the determined additional dialog being transferred, the SCC AS shall send SIP UPDATE request to each such early dialog.

Upon receiving a SIP 2xx response to the SIP UPDATE request sent towards the remote UE, the SCC AS shall:

1) establish a new early dialog by sending a SIP 180 (Ringing) response to the SIP INVITE request transferring additional session according to 3GPP TS 24.229 [2]. The SCC AS shall populate the SIP 180 (Ringing) response with:

   A) signalling elements described in subclause 6A.4.3;

   B) the SDP answer received in the SIP 2xx response to the SIP UPDATE request;

   C) the Feature-Caps header field containing the g.3gpp.ti feature-capability indicator with value of the g.3gpp.ti media feature tag in the Contact header field received in the source access leg of the determined additional dialog being transferred; and

   D) if the SIP INVITE request transferring additional session contains a P-Early-Media header field with the "supported" parameter and if the SCC AS has received a P-Early-Media header field in a SIP message in the dialog of the SIP UPDATE request, a P-Early-Media header field containing the value of the last P-Early-Media header field received in a SIP message in the dialog of the SIP UPDATE request.

Upon receiving SIP PRACK request associated with the SIP 180 (Ringing) response to the SIP INVITE request due to ATU-STI for CS to PS SRVCC, the SCC AS shall:

1) reject the source access leg of the determined dialog being transferred with SIP 404 (Not Found) response; and

2) continue handling the procedures in the subclause 12.3.10.4.

12.3.10.3.5 Transfer of terminating session in alerting phase

If SCC AS supports CS to PS SRVCC, in order to transfer the determined additional dialog being transferred, the SCC AS shall:

1) send SIP UPDATE request towards the remote UE in the remote leg of the determined additional dialog being transferred according to 3GPP TS 24.229 [2]. The SCC AS shall populate the SIP UPDATE request with the SDP offer which includes the media characteristics as received in the SIP INVITE request transferring additional session.

Upon receiving a SIP 2xx response to the SIP UPDATE request sent towards the remote UE, the SCC AS shall:

1) send a SIP 183 (Session Progress) response to the SIP INVITE request transferring additional session according to 3GPP TS 24.229 [2]. The SCC AS shall populate the SIP 183 (Session Progress) with:

   A) signalling elements described in subclause 6A.4.3;

   B) the SDP answer received in the SIP 2xx response to the SIP UPDATE request;

   C) the Recv-Info header field with the 3gpp.state-and-event info package name; and

   D) the Feature-Caps header field containing the g.3gpp.ti feature-capability indicator with value of the g.3gpp.ti media feature tag in the Contact header field received in the source access leg of the determined additional dialog being transferred.

Upon receiving the SIP PRACK request from the SC UE associated with the SIP 183 (Session Progress) response to the SIP INVITE request transferring additional session, the SCC AS shall cancel the source access leg of the determined dialog being transferred.

Upon receiving the SIP INFO request which includes an Info-Package header field containing 3gpp.state-and-event info package name and an application/vnd.3gpp.state-and-event-info+xml XML body associated with the info package according to IETF RFC 6086 [54] and compliant to the XML schema specified in the subclause D.2 from the MSC server with the event XML element containing "call-accepted", the SCC AS shall:

1) send a SIP 200 (OK) response to the SIP INVITE request received earlier from the remote UE as specified in 3GPP TS 24.229 [2]; and

2) send a SIP 200 (OK) response to the SIP INVITE request transferring additional session as specified in 3GPP TS 24.229 [2] populated as described in subclause 6A.4.3.

Upon receiving the SIP CANCEL request cancelling SIP INVITE request transferring additional session, if the SIP CANCEL request is acceptable for the SCC AS, in addition to sending a SIP 2xx response to the SIP CANCEL request, the SCC AS shall:

1) send a SIP response to the SIP INVITE request received from the remote UE as specified in 3GPP TS 24.229 [2]. The SCC AS shall populate the SIP response with:

   A) if the SIP CANCEL request contains a Reason header field with protocol "SIP", then status code and reason text from the Reason header field of the SIP CANCEL request; and

   B) if the SIP CANCEL request does not contain a Reason header field with protocol "SIP", then 486 (Busy) status code and reason text; and

2) send a SIP 487 (Request Terminated) response to the SIP INVITE request transferring additional session as specified in 3GPP TS 24.229 [2].

12.3.10.4 Removal of non-transferred sessions

If SCC AS supports CS to PS SRVCC, in order to remove non-transferred sessions, the SCC AS shall:

1) for each session in the transferable session set for which the speech media component was not transferred:

   a) release, reject or cancel the source access leg;

   b) if the speech media component is the only media component(s) of the session, release, reject or cancel the remote leg; and

   c) if the speech media component is not the only media components of the session, modify the session and remove the speech media component from the remote leg(s).

12.3.11 SCC AS procedures when the access transfer is completed

Once the SCC AS starts forwarding messages between the remote UE and the MSC server the SCC AS shall behave as an AS performing 3rd party call control acting as a routeing B2BUA as defined in 3GPP TS 24.229 [2] with the exceptions described in this subclause.

If SCC AS receives a SIP BYE request or SIP CANCEL request containing a Reason header field with the protocol value "Q.850" and the "cause" header field parameter with a value different than "31" (normal unspecified) on the target access and SCC AS is serving a terminating user in an early dialog phase, the SCC AS shall:

1) send a SIP 200 (OK) response to the SIP BYE request on the target access leg;

2) map the value of the "cause" header field parameter in the Reason header field to a SIP status code as specified in 3GPP TS 29.292 table 5.4.8.1.1 and table 5.4.8.1.2; and

3) send a 4xx, 5xx or 6xx response corresponding to the mapped SIP status code with the Reason header field received in the SIP BYE request included.

NOTE: The procedure when the SCC AS receives a SIP BYE request or CANCEL request containing a Reason header field with the protocol value "Q.850" and the "cause" header field parameter with the value "31" (normal unspecified) is described in subclause 12.3.3.5.2.

When the access transfer is completed and if the received SIP INVITE request due to ATU-STI or due to STN-SR included a P-Access-Network-Info header field, the SCC AS may (according to operator policy):

- for active session transfer or held session transfer, send a SIP reINVITE request towards the remote UE which contains:

  a) the SDP which has recently been negotiated towards the remote UE; and

  b) a P-Access-Network-Info header field which is copied from the SIP INVITE request due to ATU-STI or due to STN-SR; and

- for PS to CS access transfer when call is in alerting phase or pre-alerting phase, send a SIP UPDATE request towards the remote UE which contains:

  a) the SDP which has recently been negotiated towards the remote UE; and

  b) a P-Access-Network-Info header field which is copied from the SIP INVITE request due to ATU-STI or due to STN-SR.

12.4 MSC server enhanced for ICS

12.4.0 MSC server enhanced for ICS supporting PS to CS SRVCC

12.4.0.1 General

The MSC server enhanced for ICS supporting PS to CS SRVCC may support the codec inquiry prior to the PS to CS SRVCC access transfer.

When the MSC server enhanced for ICS supporting SRVCC receives an indication for a PS to CS SRVCC session transfer as described in 3GPP TS 23.216 [49], the MSC server may perform the codec inquiry prior to PS to CS SRVCC access transfer as specified in subclause 12.6.0A according to local policy.

NOTE: The local policy can be based on MSC server configuration of STN-SR(s) owned by ATCF(s) supporting the codec inquiry prior to PS to CS SRVCC access transfer.

If the MSC server enhanced for ICS supporting SRVCC does not support or does not perform the codec inquiry prior to PS to CS SRVCC access transfer as specified in subclause 12.6.0A, the MSC server shall perform the procedures in subclause 12.4.0.2

The MSC server enhanced for ICS supporting PS to CS SRVCC may support the codec re-negotiation after the PS to CS SRVCC access transfer as specified in subclause 12.6.0B.

12.4.0.2 PS to CS SRVCC access transfer

In order to perform the PS to CS SRVCC access transfer, the MSC server enhanced for ICS shall initiate a SIP INVITE request and shall:

1) set the Request-URI to the STN-SR for the session with speech media component to be transferred;

2) set the P-Asserted-Identity header field to the Correlation MSISDN;

3) set the Contact header field to the contact address of the MSC server;

4) include an SDP offer containing only a speech media component. If the MSC server performed procedures in subclause 12.6.0A and received a PS-to-CS-preparation-response, the MSC server should copy into the SDP offer one or more RTP payload types (each comprising of an RTP payload type number indicated in a sub-field of an <fmt> portion of an "m=" line and, if included, an "a=rtpmap" attribute and an "a=frmtp" attribute for the RTP payload type number) indicated in the <IMS-preferred-codec-list> element of the received PS-to-CS-preparation-response described in subclause 12.6.0A;

5) if SRVCC with priority handling (as described in 3GPP TS 23.216 [49]) is supported and a Allocation/Retention priority (ARP) indication is received (as described in 3GPP TS 29.280 [71]), then include an authorised Resource-Priority header field;

NOTE 1: An MSC server enhanced for ICS will use local configuration to map the received ARP value to appropriate values for the authorised Resource-Priority header field.

6) if the MSC server supports the MSC server assisted mid-call feature:

   A) insert the Supported header field containing the option-tag "norefersub" specified in IETF RFC 4488 [20];

   B) insert the Accept header field containing the MIME type as specified in subclause D.1.3;

   C) include in the Contact header field the g.3gpp.mid-call media feature tag as described in annex C; and

   D) insert the Recv-Info header field containing the g.3gpp.mid-call package name;

7) if the MSC server enhanced for ICS supports the PS to CS SRVCC for calls in alerting phase, then include:

   A) void;

   B) a Contact header field containing the g.3gpp.srvcc-alerting media feature tag as described in annex C;

   C) void;

   D) a P-Early-Media header field containing the "supported" parameter;

   E) if the MSC server enhanced for ICS supports the PS to CS SRVCC for originating calls in pre-alerting phase, include the g.3gpp. ps2cs-srvcc-orig-pre-alerting media feature tag as described in annex C into the Contact header field;

   F) if the MSC server does not support the MSC server assisted mid-call feature, a Supported header field containing the option-tag "norefersub" specified in IETF RFC 4488 [20]; and

   G) if the MSC server enhanced for ICS supports the PS to CS SRVCC for terminating calls in pre-alerting phase, include the g.3gpp. ps2cs-srvcc-term-pre-alerting media feature tag as described in annex C into the Contact header field;

NOTE 2: IETF RFC 3261 [19] recommends user agent client to include a Supported header field in any SIP INVITE request, listing option tags for extensions to SIP understood by the user agent client. In the step above, the MSC server is mandated to include at least "norefersub" option tag in the Supported header field.

NOTE 3: If the MSC server supports the MSC server assisted mid-call feature, a Supported header field containing the option-tag "norefersub" was already inserted in preceding steps.

8) if the MSC server supports CS to PS SRVCC:

   A) the Recv-Info header field containing the g.3gpp.access-transfer-events info package name;

   B) the Accept header field containing application/vnd.3gpp.access-transfer-events+xml MIME type with the "et" parameter indicating ability to receive "event-type" attribute with the value "2";

   C) the Accept header field containing application/vnd.3gpp.srvcc-ext+xml MIME type; and

   D) the g.3gpp.ti media feature tag with value as described in subclause C.12 in the Contact header field;

NOTE 4: An MSC server enhanced for ICS does not apply the ICS procedure described in 3GPP TS 29.292 [18] and 3GPP TS 24.292 [4] when sending the SIP INVITE request.

9) an Accept header field according to IETF RFC 3261 [19] containing the MIME type application/vnd.3gpp.state-and-event-info+xml as specified in subclause D.2.3;

10) a Recv-Info header field according to IETF RFC 6086 [54] containing the g.3gpp.state-and-event package name;

11) signalling elements described in subclause 6A.7.1 and shall indicate the related local preconditions as met; and

12) include the P-Access-Network-Info header field in the SIP INVITE request as specified in 3GPP TS 24.229 [2]. The P-Access-Network-Info header field shall include:

a) an access-type field set to "3GPP-GERAN", "3GPP-UTRAN-FDD", "3GPP-UTRAN-TDD", or an access-class field set to"3GPP-GERAN", "3GPP-UTRAN";

b) if available, a "cgi-3gpp" or "utran-sai-3gpp" parameter;

c) if available a "local-time-zone" parameter;

d) a "network-provided" parameter; and

e) if available, a "daylight-saving-time" parameter.

Upon receiving a SIP 200 (OK) response as the first response to the INVITE due to STN-SR (with the exception of the SIP 100 (Trying) response) and if the MSC server does not apply the MSC server assisted mid-call feature procedures in subclause 12.4A, the MSC server shall enter the "active" (N10) state (defined in 3GPP TS 24.008 [8]), regard the access transfer of the session with active speech media component as completed, send an ACK request as specified in 3GPP TS 24.229 [2] and start interworking CC messages as specified in subclause 12.6.5.

If the MSC server enhanced for ICS supports the MSC server assisted mid-call feature, it shall additionally apply the procedures defined in subclause 12.4A.

If the MSC server enhanced for ICS supports the PS to CS SRVCC for calls in alerting phase procedures, it shall additionally apply the procedures defined in subclause 12.6.3.

After finishing the access transfer procedures and if the access transfer was successful, the MSC server enhanced for ICS shall apply the ICS procedure as specified in 3GPP TS 29.292 [18] and 3GPP TS 24.292 [4].

If the access transfer procedure is unsuccessful and if the UE performs CS attachment procedures as specified in 3GPP TS 24.008 [8] after the unsuccessful access transfer procedure, then the MSC server enhanced for ICS shall apply procedures specified in 3GPP TS 29.292 [18] and 3GPP TS 24.292 [4].

12.4.0A MSC server enhanced for ICS procedures for Emergency Session Transfer

The MSC Server enhanced for ICS shall perform the procedures described in subclause 12.6.2 for the MSC server enhanced for SRVCC using SIP interface.

12.4.0B MSC server enhanced for ICS supporting vSRVCC

When the MSC server enhanced for ICS supporting vSRVCC receives an indication for a PS to CS SRVCC session transfer as described in 3GPP TS 23.216 [49], the MSC server enhanced for ICS supporting vSRVCC shall follow the procedures in subclause 12.4.0.

When an MSC server enhanced for ICS supporting vSRVCC receives an indication for a vSRVCC session transfer as described in 3GPP TS 23.216 [49], the MSC server enhanced for ICS shall initiate a SIP OPTIONS request and shall:

1) set the request URI to the STN-SR;

2) set the P-Asserted-Identity header field to the Correlation MSISDN;

3) set the Contact header field to the address of the MSC server; and

4) set the Accept header field to "application/sdp".

When the MSC server enhanced for ICS supporting vSRVCC receives a SIP 200 (OK) response to the SIP OPTIONS request with an SDP body that contains "m=" lines for audio and video, the MSC server enhanced for ICS shall:

1) initiate a SIP INVITE request and shall:

   a) set the request URI to the STN-SR for the session with speech and video media components to be transferred;

   b) set the P-Asserted-Identity header field to the Correlation MSISDN;

   c) set the Contact header field to the address of the MSC server;

   d) include an SDP offer only containing a speech media component and a video media component with default codecs for speech and video (as specified in 3GPP TS 26.111 [69]);

e) if the MSC server enhanced for enhanced for ICS supporting vSRVCC supports PS to CS access transfer for alerting calls, then include:

i) an Accept header field containing the MIME type application/vnd.3gpp.state-and-event-info+xml as specified in subclause D.2.3;

ii) a Contact header field containing the g.3gpp.srvcc-alerting media feature tag as described in annex C; and

iii) a Recv-Info header field containing the g.3gpp.state-and-event package name; and

f) if vSRVCC with priority handling (as described in 3GPP TS 23.216 [49]) is supported and a Allocation/Retention priority (ARP) indication is received (as described in 3GPP TS 29.280 [70]), then include an authorised Resource-Priority header field; and

NOTE: An MSC server enhanced for ICS will use local configuration to map the received ARP value to appropriate values for the authorised Resource-Priority header field.

2) if the MSC server enhanced for ICS supporting vSRVCC supports PS to CS access transfer for alerting calls, then additionally apply the procedures defined in subclause 12.6.3.

When an MSC server enhanced for ICS supporting vSRVCC receives a SIP 200 (OK) response to the SIP OPTIONS request with an SDP body that contains an "m=" line for audio but not video, the MSC server enhanced for ICS supporting vSRVCC shall follow the procedures in subclause 12.4.0.

12.4.1 Void

12.4.2 MSC server enhanced for ICS supporting CS to PS SRVCC

12.4.2.1 Distinction of requests

The MSC server needs to distinguish the following SIP requests:

1) SIP INFO request:

A) with Info-Package header field with value g.3gpp.access-transfer-events; and

B) with application/vnd.3gpp.access-transfer-events+xml MIME body associated with the info package according to IETF RFC 6086 [54] and indicating the session transfer notification response.

In the procedures below, such requests are known as "SIP INFO requests carrying the session transfer notification response".

12.4.2.2 General

If the MSC server supports the CS to PS SRVCC, upon receiving HO required for a UE including an indication that the HO is for CS to PS as described in 3GPP TS 23.216 [15] or if required by procedures in subclause 12.4.2.5, the MSC server shall:

1) determine the transferable dialog set which are all SIP dialogs:

A) interworked with the CS calls of the UE; and

B) supporting a session; and

2) if the determined transferable dialog set is not empty:

NOTE: If the determined transferable dialog set is empty, remaining procedures of this subclause are not invoked.

A) determine the dialog for communication with ATCF as follows:

a) if a CS call in Active (N10) state (defined in 3GPP TS 24.008 [8]) and Idle auxiliary state (defined in 3GPP TS 24.083 [43]) exists, the SIP dialog in the transferable dialog set, which is interworked with the CS call in Active (N10) state (defined in 3GPP TS 24.008 [8]) and Idle auxiliary state (defined in 3GPP TS 24.083 [43]); and

b) if a CS call in Active (N10) state (defined in 3GPP TS 24.008 [8]) and Idle auxiliary state (defined in 3GPP TS 24.083 [43]) does not exist, a SIP dialog in the transferable dialog set, which is interworked with any CS call; and

B) send a SIP INFO request according to 3GPP TS 24.229 [2] within the determined dialog for communication with ATCF. The MSC server shall populate the SIP INFO request with:

a) Info-Package header field with value g.3gpp.access-transfer-events; and

b) application/vnd.3gpp.access-transfer-events+xml MIME body associated with the info package according to IETF RFC 6086 [54] and indicating the session transfer notification request.

Upon receiving SIP INFO request carrying the session transfer notification response within the determined dialog for communication with ATCF, the MSC server shall:

1) send a SIP 200 (OK) response to the SIP INFO request according to 3GPP TS 24.229 [2];

2) if the application/vnd.3gpp.access-transfer-events+xml MIME body indicates that the ATCF does not require the MSC server to redirect the speech media component of the session transferred by the CS to PS SRVCC access transfer, continue handling the procedures in the subclause 12.4.2.3; and

3) if the application/vnd.3gpp.access-transfer-events+xml MIME body indicates that the ATCF requires the MSC server to redirect the speech media component of the session transferred by the CS to PS SRVCC access transfer, continue handling the procedures in the subclause 12.4.2.4.

12.4.2.3 Transfer of session without MSC server redirecting the speech media component

If the MSC server supports the CS to PS SRVCC, if the access transfer is prepared according to 3GPP TS 23.216 [15] and if the ATGW does not require the MSC server to redirect the speech media component of the session transferred by the CS to PS SRVCC access transfer, the MSC server shall:

1) send a SIP INFO request according to 3GPP TS 24.229 [2] within determined dialog for communication with ATCF. The MSC server shall populate the SIP INFO request with:

A) Info-Package header field with value g.3gpp.access-transfer-events; and

B) application/vnd.3gpp.access-transfer-events+xml MIME body associated with the info package according to IETF RFC 6086 [54] and indicating the session transfer preparation.

If the MSC server supports the CS to PS SRVCC, if the ATGW does not require the MSC server to redirect the speech media component of the session transferred by the CS to PS SRVCC access transfer and if the access transfer is cancelled according to 3GPP TS 23.216 [15], the MSC server shall send a SIP INFO request according to 3GPP TS 24.229 [2] within determined dialog for communication with ATCF. The MSC server shall populate the SIP INFO request with:

1) Info-Package header field with value g.3gpp.access-transfer-events; and

2) application/vnd.3gpp.access-transfer-events+xml MIME body associated with the info package according to IETF RFC 6086 [54] and indicating the session transfer cancellation.

12.4.2.4 Transfer of session with MSC server redirecting the speech media component

When the ATGW requires the MSC server to redirect the speech media component of the session transferred by the CS to PS SRVCC access transfer, the MSC server shall:

1) send a SIP INVITE request according to 3GPP TS 24.229 [2]. The MSC server shall populate the SIP INVITE request with:

A) the Request-URI header field set to the ATCF management URI;

B) Contact header field set to the IP address of MSC server;

C) SDP body includes the speech media component of the session transferred by the CS to PS SRVCC access transfer; and

D) the P-Asserted-Identity header field set to the C-MSISDN.

If the access transfer is prepared according to 3GPP TS 23.216 [15] and after the MSC server redirect the speech media component of the session transferred by the CS to PS SRVCC access transfer to the ATGW, the MSC server shall:

1) send a SIP INFO request according to 3GPP TS 24.229 [2] within dialog for communication with ATCF. The MSC server shall populate the SIP INFO request with:

   A) Info-Package header field with value g.3gpp.access-transfer-events; and

   B) application/vnd.3gpp.access-transfer-events+xml MIME body associated with the info package according to IETF RFC 6086 [54] and indicating the session transfer preparation.

If the MSC server supports the CS to PS SRVCC, if the access transfer is cancelled according to 3GPP TS 23.216 [15], the MSC server shall send a SIP INFO request according to 3GPP TS 24.229 [2] within determined dialog for communication with ATCF. The MSC server shall populate the SIP INFO request with:

1) Info-Package header field with value g.3gpp.access-transfer-events; and

2) application/vnd.3gpp.access-transfer-events+xml MIME body associated with the info package according to IETF RFC 6086 [54] and indicating the session transfer cancellation.

12.4.2.5 Abnormal cases

If the dialog for communication with ATCF is released after receiving HO required for a UE including an indication that the HO is for CS to PS as described in 3GPP TS 23.216 [15] and before sending of the CS to PS handover command as described in 3GPP TS 23.216 [15], the MSC shall perform the procedures in subclause 12.4.2.2 again.

12.4.3 Abnormal cases

12.4.3.1 Permanent response codes

When the MSC server enhanced for ICS receives a SIP reject response to the SIP INVITE request due to STN-SR or SIP INVITE request due to E-STN-SR, the MSC server shall regard any of the following SIP reject responses as permanent errors:

- SIP 404 (Not found);
- SIP 410 (Gone);
- SIP 484 (Address Incomplete); and
- SIP 604 (Does not exist anywhere).

The MSC server enhanced for ICS shall regard all other received SIP reject responses to the SIP INVITE request due to STN-SR or SIP INVITE request due to E-STN-SR as temporary errors.

NOTE: The procedures in 3GPP TS 29.280 [71] requires that the MSC server indicates whether a received SIP reject response to the SIP INVITE request due to STN-SR or SIP INVITE request due to E-STN-SR is temporary or permanent.

12.4.3.2 PS to CS SRVCC cancelled by MME/SGSN or failure of the access transfer procedure in the MSC server

If the MSC server enhanced for ICS receives a SRVCC PS to CS Cancel Notification from the MME/SGSN or if the access transfer procedure fails for any other reason in the MSC server enhanced for ICS, the MSC server shall:

1) in the dialog created by the SIP INVITE request due to STN-SR or SIP INVITE request due to E-STN-SR:

   a) if the dialog is a dialog with inactive or active speech media component, send a SIP BYE request; and

   b) if the dialog is in the pre-alerting phase or in the alerting phase, send a SIP BYE request or a SIP CANCEL request;

2) if the MSC server applies the MSC server assisted mid-call feature, in the dialog created by the SIP INVITE for the additional transferred session;

   a) if the dialog is a dialog with inactive speech media component, send a SIP BYE request; and

   b) if the dialog is in the pre-alerting phase or in the alerting phase, send a SIP BYE request or a SIP CANCEL request;

3) if the cancellation is due to SRVCC PS to CS Cancel Notification from the MME/SGSN, include in the SIP request a Reason header field with the protocol value "Q.850" and the "cause" header field parameter with the value "31" (normal unspecified); and

4) if the cancellation is due to any other reason than SRVCC PS to CS Cancel Notification from the MME/SGSN, include in the SIP request a Reason header field with the protocol value "Q.850" and the "cause" header field parameter with a value different from"31", e.g. "41" (temporary failure) or "16" (normal clearing).

NOTE: The inclusion of the protocol value "Q.850" and the "cause" header field parameter with the value "31" (normal unspecified) will result in that the SCC AS delays the release of the source access leg and the remote UE leg allowing the SC UE to continue the call in PS.

If after having sent a SIP CANCEL request:

a) to a SIP INVITE request due to STN-SR; or

b) if the MSC server applies the MSC server assisted mid-call feature, to a SIP INVITE request for the additional transferred session,

and subsequently receiving a SIP 200 (OK) response to such a SIP INVITE request, the MSC server shall send a SIP BYE request containing a Reason header field with the same protocol value and the same "cause" header field as used in the SIP CANCEL request.

12.4.3.3 Guard timer for the CC CONNECT request elapses

NOTE: Timer T313 defined in 3GPP TS 24.008 [8] supervises a CC CONNECT message sent by the MSC server.

If after having sent a CC CONNECT message, the timer that supervises the CC CONNECT message expires before the MSC server has received a CC CONNECT ACK message, the MSC server shall send a SIP BYE request on the SIP dialog created by the SIP INVITE due to STN-SR, by following the rules of 3GPP TS 24.229 [2].

12.4A MSC server assisted mid-call feature

Upon receiving a SIP 2xx response to the SIP INVITE request due to STN-SR, the MSC server shall:

1. if inactive speech media component is negotiated by the SDP answer of the SIP 2xx response to the SIP INVITE request due to STN-SR, associate the dialog created by the SIP INVITE request due to STN-SR with a CS call with transaction identifier 0 and TI flag value as in mobile terminated call and enter the "active" (N10) state (defined in 3GPP TS 24.008 [8]), the "call held" hold auxiliary state (defined in 3GPP TS 24.083 [43]), the "idle" multi party auxiliary state (defined in 3GPP TS 24.084 [47]) for the CS call, regard the access transfer of the session with inactive speech media component as completed and start interworking CC messages as specified in subclause 12.6.5. If the speech media component in SDP answer of the SIP 2xx response to the SIP INVITE request due to STN-SR:

   - has "recvonly" directionality, the MSC server shall determine that the remote UE does not hold the call; and

   - has "inactive" directionality, the MSC server shall determine that the remote UE holds the call; and

2. if active speech media component is negotiated by the SDP answer of the SIP 2xx response to the SIP INVITE request due to STN-SR, associate the dialog created by the SIP INVITE request due to STN-SR with a CS call with the transaction identifier 0 and TI flag value as in mobile terminated call and enter the "active" (N10) state (defined in 3GPP TS 24.008 [8]), the "idle" hold auxiliary state (defined in 3GPP TS 24.083 [43]), the "idle" multi party auxiliary state (defined in 3GPP TS 24.084 [47]) for the CS call, regard the access transfer of the session with active speech media component as completed and start interworking CC messages as specified in subclause 12.6.5. If the speech media component in SDP answer of the SIP 2xx response to the SIP INVITE request due to STN-SR:

- has "sendrecv" directionality, the MSC server shall determine that the remote UE does not hold the call; and
- has "sendonly" directionality, the MSC server shall determine that the remote UE holds the call.

Upon receiving a SIP INFO request:

- sent inside in the dialog created by the SIP INVITE request due to STN-SR;
- with the Info-Package header field containing the g.3gpp.mid-call package name;
- with the application/vnd.3gpp.mid-call+xml MIME body associated with the info package according to IETF RFC 6086 [54]; and
- with one or more participants included in the application/vnd.3gpp.mid-call+xml MIME body;

and if the SIP INVITE request due to STN-SR established a session with conference focus then the MSC server shall:

1. if inactive speech media component is negotiated by the SDP answer of the SIP 2xx response to the SIP INVITE request due to STN, associate the session and the participants extracted from the application/vnd.3gpp.mid-call+xml MIME body with CS calls:

   - with transaction identifiers 0, 2, 3, 4, 5 assigned to the participants in their order in the list of the extracted participants; and
   - with TI flag value as in mobile terminating call;

   and enter the "active" (N10) state (defined in 3GPP TS 24.008 [8]), the "call held" hold auxiliary state (defined in 3GPP TS 24.083 [43]), the "call in MPTY" multi party auxiliary state (defined in 3GPP TS 24.084 [47]) for the CS calls, regard the access transfer of conference participants with transaction identifiers 2,3,4,5 and the TI flag value as in the mobile terminating call as completed and start interworking CC messages as specified in subclause 12.6.5. The MSC server may subscribe to the conference event package as specified in 3GPP TS 24.605 [31]; and

NOTE 1: The transaction identifier and TI flag value for the first participant are assigned by the call activation procedures for SRVCC in 3GPP TS 24.008 [8].

NOTE 2: The multi party auxiliary state was initially set to "idle" on reception of the SIP 2xx response to the SIP INVITE request due to STN-SR. This state is re-assigned to "call in MPTY" after processing the SIP INFO request to reflect the multi party auxiliary state associated with the first participant .

2. if active speech media component is negotiated by the SDP answer of the SIP 2xx response to the SIP INVITE request due to STN, associate the session and the participants extracted from the application/vnd.3gpp.mid-call+xml MIME body with CS calls:

   - with transaction identifiers 0, 2, 3, 4, 5 assigned to the participants in their order in the list of the extracted participants; and
   - with TI flag value as in mobile terminating call;

   and enter the "active" (N10) state (defined in 3GPP TS 24.008 [8]), the "idle" hold auxiliary state (defined in 3GPP TS 24.083 [43]), the "call in MPTY" multi party auxiliary state (defined in 3GPP TS 24.084 [47]) for the CS calls, regard the access transfer of conference participants with transaction identifiers 2,3,4,5 and the TI flag value as in the mobile terminating call as completed and start interworking CC messages as specified in subclause 12.6.5. The MSC server may subscribe to the conference event package as specified in 3GPP TS 24.605 [31].

NOTE 3: For an MSC server enhanced for PS to CS SRVCC using SIP interface, following access transfer, the procedures for the handling of transferred conference participants are implementation dependent.

Upon receiving a SIP REFER request:

1. with the Refer-Sub header field containing "false" value;
2. with the Supported header field containing "norefersub" value;
3. with the Refer-To header field containing a SIP URI with the Target-Dialog URI header field;

4. sent inside a SIP dialog:

   A. created by the SIP INVITE request due to STN-SR; and

   B. where the g.3gpp.mid-call feature-capability indicator as specified in annex C was included in the Feature-Caps header field of the SIP 2xx response to the SIP INVITE request due to STN-SR; and

5. containing a MIME body of MIME type specified in the subclause D.1.3;

the MSC server shall:

1. handle the SIP REFER request as specified in 3GPP TS 24.229 [2], IETF RFC 3515 [13] as updated by IETF RFC 6665 [81], and IETF RFC 4488 [20] without establishing an implicit subscription; and

NOTE 4: In accordance with IETF RFC 4488 [20], the MSC server inserts the Refer-Sub header field containing the value "false" in the SIP 2xx response to the SIP REFER request to indicate that it has not created an implicit subscription.

2. send a SIP INVITE request for transfer of an additional session with inactive speech media component in accordance with the procedures specified in 3GPP TS 24.229 [2] and IETF RFC 3515 [13]. If the MSC server is enhanced for ICS, the MSC server does not apply the ICS procedure described in 3GPP TS 29.292 [18] and 3GPP TS 24.292 [4] when sending the SIP INVITE request for transfer of an additional session with inactive speech media component. Additionally, the MSC server shall populate the SIP INVITE request for transfer of an additional session with inactive speech media component as follows:

   A. header fields which were included as URI header fields in the URI in the Refer-To header field of the received SIP REFER request as specified in IETF RFC 3261 [19] except the "body" URI header field;

   B. include in the Contact header field the g.3gpp.mid-call media feature tag as described in annex C;

   C. the SDP offer with:

      a. the same amount of the media descriptions as in the "body" URI header field in the URI in the Refer-To header field of the received SIP REFER request;

      b. each "m=" line having the same media type as the corresponding "m=" line in the "body" URI header field in the URI in the Refer-To header field of the received SIP REFER request;

      c. port set to zero value in each "m=" line whose corresponding "m=" line in the "body" URI header field in the URI in the Refer-To header field of the received SIP REFER request has port with zero value;

      d. media directionality as in the "body" URI header field in the URI in the Refer-To header field of the received SIP REFER request; and

      e. payload type numbers and their mapping to codecs and media parameters supported by MSC server, not conflicting with those in the "body" URI header field in the URI in the Refer-To header field of the received SIP REFER request;

NOTE 5: port can be sent to zero or non zero value for the offered "m=" line whose corresponding "m=" line in the "body" URI header field in the URI in the Refer-To header field of the received SIP REFER request has port with nonzero value.

   D. if an authorised Resource-Priority header field was included in the SIP INVITE request due to STN-SR, then include an authorised Resource-Priority header field with the same values as used in the SIP INVITE request due to STN-SR;

   E. if the MSC server supports CS to PS SRVCC and the SIP REFER request contains the application/vnd.3gpp.srvcc-ext+xml MIME body:

      a) the topmost Route header field with the ATCF management URI received in the application/vnd.3gpp.srvcc-ext+xml MIME body of the SIP REFER request and "lr" URI parameter;

      b) the Accept header field containing application/vnd.3gpp.access-transfer-events+xml MIME type with the "et" parameter indicating ability to receive "event-type" attribute with the value "2";

      c) the Recv-Info header field containing the g.3gpp.access-transfer-events info package name;

d) the application/vnd.3gpp.srvcc-ext+xml MIME body with the <srvcc-ext> root element containing the <Setup-info> element containing the CS to PS SRVCC information received in the application/vnd.3gpp.srvcc-ext+xml MIME body of the SIP REFER request and indicating the "initiator" role of the MSC server in the session set up; and

e) the g.3gpp.ti media feature tag with value as described in subclause C.12 in the Contact header field;

F. if the MSC server supports procedures in subclause 22.2:

a) an Accept header field according to IETF RFC 3261 [19] containing the MIME type application/vnd.3gpp.state-and-event-info+xml as specified in subclause D.2.3; and

b) a Recv-Info header field according to IETF RFC 6086 [54] containing the g.3gpp.state-and-event package name;

G. signalling elements described in subclause 6A.7.1 and shall indicate the related local preconditions as met;

H. include the P-Access-Network-Info header field in the SIP INVITE request as specified in 3GPP TS 24.229 [11]. The P-Access-Network-Info header field shall include:

a) an access-type field set to "3GPP-GERAN", "3GPP-UTRAN-FDD", "3GPP-UTRAN-TDD", or an access-class field set to"3GPP-GERAN", "3GPP-UTRAN";

b) if available, a "cgi-3gpp" or "utran-sai-3gpp" parameter;

c) if available a "local-time-zone" parameter;

d) a "network-provided" parameter; and

e) if available, a "daylight-saving-time" parameter; and

I. if a P-Asserted-Identity header field is not included in the headers portion of the URI in the Refer-To header field of the received SIP REFER request as specified in IETF RFC 3261 [19], include a P-Asserted-Identity header field with the value of the C-MSISDN contained in the SIP INVITE requests due to STN-SR which created the dialog in which the SIP REFER request is received.

Upon receiving SIP 2xx response to the SIP INVITE request for transfer of an additional session with inactive speech media component, the MSC server shall:

1. if:

a) the SIP INVITE request for transfer of the additional session with inactive speech media component did not established a session with a conference focus; or

b) the application/vnd.3gpp.mid-call+xml MIME body included in the SIP REFER request does not contain one or more participants:

then associate the additional session with inactive speech media component with CS call with transaction identifier 1 and TI flag value as in mobile terminated call and enter the "active" (N10) state (as defined in 3GPP TS 24.008 [8]), the "call held" hold auxiliary state (as defined in 3GPP TS 24.083 [43]) and the "idle" multi party auxiliary state (defined in 3GPP TS 24.084 [47]) for the CS call. If the speech media component in SDP answer of the SIP 2xx response to the SIP INVITE request for transfer of an additional session with inactive speech media component:

- has "recvonly" directionality, the MSC server shall determine that the remote UE does not hold the call; and

- has "inactive" directionality, the MSC server shall determine that the remote UE holds the call;

2. if the SIP INVITE request for transfer of an additional session with inactive speech media component established a session with a conference focus and the application/vnd.3gpp.mid-call+xml MIME body included in the SIP REFER request contained one or more participants:

a) associate the additional session and the participants extracted from the application/vnd.3gpp.mid-call+xml MIME body included in the SIP REFER request with CS calls:

- with transaction identifiers 1, 2, 3, 4, 5 assigned to the participants in their order in the list of the extracted participants; and
- with TI flag value as in mobile terminated call;

and enter the "active" (N10) state (defined in 3GPP TS 24.008 [8]), the "call held" hold auxiliary state (defined in 3GPP TS 24.083 [43]) and the "call in MPTY" multi party auxiliary state (defined in 3GPP TS 24.084 [47]) for the CS calls. The MSC server may subscribe to the conference event package as specified in 3GPP TS 24.605 [31]; and

3. the MSC server shall send the SIP ACK request as specified in 3GPP TS 24.229 [2] and regard the access transfer of the session with inactive speech media component and conference participants (if applicable) as completed and start interwork CC messages as specified in subclause 12.6.5.

NOTE 6: When the access transfer is completed the MSC server can verify the call state of its peer entity using the STATUS ENQUIRY procedure in accordance with procedures in 3GPP TS 24.008 [8] to ensure that SIP requests or SIP responses sent between the SC UE and the SCC AS just before the handover from the PS domain to the CS domain occurred did not result in incompatible call state or auxiliary states. If the call state or auxiliary states are incompatible the transferred session is released.

12.5 EATF

12.5.1 EATF procedures for PS to CS session continuity, E-SR-VCC

The EATF needs to distinguish between the following initial SIP INVITE requests to provide specific functionality for E-SR-VCC:

1. SIP INVITE request routed to the EATF due to E-STN-SR in the Request-URI. In the procedures below, such requests are known as "SIP INVITE requests due to E-STN-SR".

NOTE 1: The same E-STN-SR is used for all the emergency session access transfers within one PLMN.

Other initial SIP requests can be dealt with in any manner conformant with 3GPP TS 24.229 [2].

When the EATF receives a SIP INVITE request due to E-STN-SR on the Target Access Leg, the EATF shall:

1) identify the transferable set which are dialogs supporting a session with speech media component anchored at the EATF with the sip.instance media feature tag provided by the SC UE in the Contact header field at session establishment equal to the sip.instance media feature tag included in the Contact header field of the received SIP INVITE request due to E-STN-SR; and

NOTE 2: When the sip.instance media feature tag contains an IMEI URN as specified in IETF RFC 7254 [82], the spare digit (i.e. the digit matching the spare ABNF rule) is required to be set to zero. If the spare digit is set to non zero value, the spare digit can be ignored.

2) if a confirmed dialog supporting a sessions with active speech media component exists in the transferable set:

   A) associate the SIP INVITE request due to E-STN-SR with the confirmed dialog supporting a session with active speech media component; and

   B) originate session modification as described in 3GPP TS 24.229 [2] towards the remote UE by sending a SIP re-INVITE request with a new SDP offer with media characteristics as received in the SIP INVITE request due to E-STN-SR. If the SIP INVITE request due to E-STN-SR does not contain a Recv-Info header field, the EATF shall include an empty Recv-Info header field in the SIP re-INVITE request as defined in IETF RFC 6086 [54]. If the SIP INVITE request due to E-STN-SR contains a Recv-Info header field, the EATF shall remove the info package names of info packages that terminate at the EATF, if any, and include the remaining info package names in a Recv-Info header field in the SIP re-INVITE request.

12.5.2 EATF procedures for PS to CS SRVCC, abnormal case

12.5.2.1 PS to CS SRVCC cancelled by MME/SGSN or release of the target access leg for ongoing emergency session

If the EATF receives a SIP BYE request containing a Reason header field containing the protocol value "Q.850" and the "cause" header field parameter with the value of "31" (normal unspecified) on the target access leg after having initiated an access transfer that was triggered by a SIP INVITE request due to E-STN-SR and when the operator specific timer is still running, the EATF shall:

1) send the SIP 200 (OK) response to the SIP BYE request;

2) wait until the operator specific timer expires or until a SIP re-INVITE request from the SC UE containing the protocol value "SIP" and the "cause" header field parameter with the value "487" is received; and

3) if the operator specific timer expires and no SIP re-INVITE request from the SC UE containing the protocol value "SIP" and the "cause" header field parameter with the value "487" is received, release the call according to procedures in 3GPP TS 24.229 [2].

NOTE 1: All protocol values in the Reason header field other than "Q.850" and all values of the "cause" header field parameter other than "31" (normal unspecified) will result in an immediate release of the source access leg and the remote UE leg.

NOTE 2: The EATF assigns an operator specific timer to delay the release of the source access leg for PS to CS SRVCC access transfers.

When the EATF receives SIP re-INVITE request(s) from the SC UE containing the Reason header field with the protocol value "SIP" and the "cause" header field parameter with the value "487" after having performed an access transfer that was triggered by a SIP INVITE request due to E-STN-SR and after receiving a SIP BYE request containing the Reason header field containing the protocol value "Q.850" and the "cause" header field parameter with the value "31" (normal unspecified) on the target access leg, then the EATF shall:

1) not release the original source access leg on expiry of the timer described in subclause 12.5.1; and

2) send the SIP re-INVITE request towards the remote leg by following the rules of 3GPP TS 24.229 [2].

When the EATF receives a SIP response to the SIP re-INVITE request, the EATF shall forward the SIP response to the SC UE.

12.5.2.2 PS to CS SRVCC cancelled by MME/SGSN or failure by UE to transition to CS domain for ongoing session

When the EATF receives a SIP re-INVITE request containing the Reason header field with the protocol value "SIP" and the "cause" header field parameter with the value "487" on the original source access leg after having initiated an access transfer that was triggered by a SIP INVITE request due to E-STN-SR and the SIP INVITE request due to E-STN-SR transaction is not yet completed then the EATF shall wait until this transaction has completed and then continue with the steps 1) to 3) described below.

When the EATF receives a SIP re-INVITE request containing the Reason header field with the protocol value "SIP" and header field parameter "cause" with the value "487" on the original source access leg after having performed an access transfer that was triggered by a SIP INVITE request due to E-STN-SR, then the EATF shall:

1) send a SIP BYE request on the target access leg, by following the rules of 3GPP TS 24.229 [2];

2) not release the original access leg on the expiration of the timer described in subclause 12.5.1; and

3) send the SIP re-INVITE request towards the remote leg by following the rules of 3GPP TS 24.229 [2].

NOTE: The EATF assigns an operator specific timer to delay the release of the source access leg for PS to CS SRVCC access transfer.

When the EATF receives a SIP response to the SIP re-INVITE request, the EATF shall forward the SIP response to the SC UE.

12.5.2.3 P-CSCF releasing the source access leg during PS to CS SRVCC

When EATF receives a SIP BYE request on the source access leg with any Reason header field containing protocol "SIP" and reason parameter "cause" with value "503" then:

NOTE 1: The SIP BYE request can contain more than one Reason header field.

- if the EATF receives an initial SIP INVITE request due to E-STN-SR on the target access leg associated with the established dialog on the source access leg, within a time defined by the operator policy after the SIP BYE request reception, then the EATF shall not initiate release of the remote leg; and

- if the EATF does not receive an initial SIP INVITE request due to E-STN-SR on the target access leg associated with the established dialog on the source access leg, within a time defined by the operator policy after the SIP BYE request reception then the EATF shall initiate release of the remote leg.

NOTE 2: 8 seconds is an appropriate value for the operator policy.

12.5.2.3 PS to CS SRVCC cancelled by MME/SGSN or failure by UE to transition to CS domain for session in early dialog state

If the EATF applies the procedures for the PS to CS access transfer for originating call in alerting phase or pre-alerting phase using PS to CS SRVCC procedure (as specified in subclause 12.5.3.2), then when the EATF receives a SIP UPDATE request containing a Reason header field containing protocol "SIP" and reason parameter "cause" with value "487" on the original source access leg after having initiated an access transfer that was triggered by a SIP INVITE request due to E-STN-SR for an emergency session which is still in early dialog state, the EATF shall:

1) stop the operator specific timer and not release the original access leg as described in subclause 12.5.4; and

2) forward the SIP UPDATE request on the remote leg according to 3GPP TS 24.229 [2].

The EATF shall now start forwarding SIP messages from the remote UE to the SC UE on each dialog created by the SIP INVITE on the original source access leg as specified in 3GPP TS 24.229 [2] and the present document.

When the EATF receives a SIP 200 (OK) response to the SIP UPDATE request, then the EATF shall:

1) if the EATF has already sent a SIP 200 (OK) response to a SIP INVITE request due to E-STN-SR then send a SIP BYE request on this dialog, and send a SIP 200 (OK) response to the SIP INVITE on the original source access leg; and

2) if the EATF has not sent a SIP 200 (OK) response to a SIP INVITE request due to E-STN-SR then send a SIP 480 (Temporarily Unavailable) response to reject the SIP INVITE request due to E-STN-SR.

12.5.3 EATF procedures for PS to CS access transfer when emergency session is in alerting phase or pre-alerting phase

12.5.3.1 General

The EATF shall apply the procedures as described in subclause 12.5.3.2 if:

1) the Contact header field of the SIP INVITE request routed to the EATF due to a E-STN-SR includes the g.3gpp.srvcc-alerting media feature tag as specified in annex C; and

2) there are one or more dialogs supporting sessions with speech media component existing in the transferable set, such that:

   A) all dialogs are early dialogs;

   B) SIP 180 (Ringing) response to SIP INVITE request was received in at least one of those early dialogs, all such SIP 180 (Ringing) responses are responses to the same SIP INVITE request and at least one of such SIP 180 (Ringing) responses was received in an early dialog supporting session with active speech media component;

   C) the Contact header field provided by the SC UE includes the g.3gpp.srvcc-alerting media feature tag as described in annex C;

E) the following is true:

- the Contact header field provided by the SC UE does not include the g.3gpp.mid-call media feature tag; or

- the Contact header field provided by the SC UE includes both the g.3gpp.mid-call media feature tag and the g.3gpp.ps2cs-srvcc-mid-call-emergency media feature tag; and

F) the Feature-Caps header field provided by the EATF towards the SC UE includes the g.3gpp.srvcc-alerting feature-capability indicator as described in annex C.

Additionally, the EATF shall apply the procedures as described in subclauses 12.5.3.2 if:

1) the Contact header field of the SIP INVITE request due to a E-STN-SR includes the g.3gpp.ps2cs-srvcc-orig-pre-alerting media feature tag as described in annex C; and

2) there are zero, one or more dialogs supporting a session with speech media component in the transferable set and a SIP INVITE request was received from SC UE with the sip.instance media feature tag in the Contact header field to the sip.instance media feature tag included in the Contact header field of the received SIP INVITE request due to E-STN-SR, such that:

A) all dialogs are early dialogs created by a SIP response to the SIP INVITE request;

B) a final SIP response to the SIP INVITE request has not been sent yet;

C) a SIP 180 (Ringing) response to the SIP INVITE request has not been sent yet in any existing early dialog created by a SIP response to the SIP INVITE request;

D) the SIP INVITE request included a Contact header field containing the g.3gpp.ps2cs-srvcc-orig-pre-alerting media feature tag as described in annex C;

E) the following is true:

- the Contact header field provided by the SC UE does not include the g.3gpp.mid-call media feature tag; or

- the Contact header field provided by the SC UE includes both the g.3gpp.mid-call media feature tag and the g.3gpp.ps2cs-srvcc-mid-call-emergency media feature tag; and

F) a SIP 1xx response to the SIP INVITE request was sent where the SIP 1xx response contained a Feature-Caps header field with the g.3gpp.ps2cs-srvcc-orig-pre-alerting feature-capability indicator as described in annex C.

NOTE: EATF can have zero dialogs if all the early dialogs were terminated by 199 (Early Dialog Terminated) as described in RFC 6228 [80].

12.5.3.2 EATF procedures for PS to CS access transfer for originating call in alerting phase or pre-alerting phase using PS to CS SRVCC procedure

When the session in the transferable session set is an originating call not accepted yet the EATF shall associate the SIP INVITE request due to E-STN-SR with an early dialog or early dialogs related to the originating call.

If the EATF receives a SIP 18x response on the remote leg after receiving a SIP INVITE request due to E-STN-SR, and this SIP 18x response does not require use of reliable provisional responses, the EATF shall:

1) store this SIP 18x response; and

2) if a P-Early-Media header field is received in the SIP 18x response, store the P-Early-Media header field.

The EATF shall store the received SIP 18x responses separately for each early dialog. If the EATF has already stored a SIP 18x response for an early dialog and receives another SIP 18x response for the same early dialog, the EATF may remove the stored SIP 18x response for that early dialog and shall store the new SIP 18x response for that early dialog.

The EATF shall store the received P-Early-Media header field separately for each early dialog. If the EATF has already stored a P-Early-Media header field received in a SIP 18x response for an early dialog, and receives another SIP 18x

response for the same early dialog containing a P-Early-Media header field, the EATF may remove the stored P-Early-Media header field for that early dialog and shall store the new P-Early-Media header field for that early dialog,

NOTE: The P-Early-Media header field is stored separately to prepare for the case that a subsequent SIP 18x response does not contain a P-Early-Media header field.

If there is only one early dialog related to the originating call not accepted yet available for the served user, the remote UE has provided an Allow header field listing the SIP UPDATE method or has not provided Allow header field, the remote UE has provided the SDP answer, the EATF shall update the remote leg by sending a SIP UPDATE request towards the remote UE using the existing early dialog as specified in 3GPP TS 24.229 [2]. The EATF shall populate the SIP UPDATE request with the SDP offer received in the SIP INVITE request due to E-STN-SR.

Upon receiving the SIP 200 (OK) response to the SIP UPDATE request from the remote UE, the EATF shall send a SIP 183 (Session Progress) response in response to the SIP INVITE request due to E-STN-SR towards the MSC server. The EATF shall populate the SIP 183 (Session Progress) response to the SIP INVITE request due to E-STN-SR with:

1) the SDP answer received in the SIP 200 (OK) response to the SIP UPDATE request; and

2) if the SIP INVITE request due to E-STN-SR contains a P-Early-Media header field with the "supported" parameter and if the EATF has received a P-Early-Media header field in a SIP message in the dialog of the SIP UPDATE request, include a P-Early-Media header field containing the value of the last P-Early-Media header field received in a SIP message in the dialog of the SIP UPDATE request.

If there are more than one early dialog related to the originating call not accepted yet available for the served user due to forking as described in 3GPP TS 24.229 [2], the remote UE has provided an Allow header field listing SIP UPDATE method or has not provided Allow header field, and the remote UE provided SDP answer, the EATF shall update the remote leg(s) by sending SIP UPDATE request(s) simultaneously towards remote UE(s) using such early dialog(s) as specified in 3GPP TS 24.229 [2]. The EATF shall populate each SIP UPDATE request with the SDP offer received in the SIP INVITE request due to E-STN-SR. Upon receiving each SIP 200 (OK) response to the SIP UPDATE request from the remote UE, the EATF shall create a new early dialog by sending a SIP 183 (Session Progress) response in response to the SIP INVITE request due to E-STN-SR towards the MSC server. The EATF shall populate the SIP 183 (Session Progress) response to the SIP INVITE request due to E-STN-SR with:

1) the SDP answer received in the SIP 200 (OK) response to the SIP UPDATE request; and

2) If the SIP INVITE request due to E-STN-SR contains a P-Early-Media header field with the "supported" parameter and if the EATF has received a P-Early-Media header field in a SIP message in the dialog of the SIP UPDATE request, include a P-Early-Media header field containing the value of the last P-Early-Media header field received in a SIP message in the dialog of the SIP UPDATE request.

If one or more early dialogs related to the originating call not accepted yet are available for the served user, and in each such early dialog:

1) the remote UE of the early dialog has provided an Allow header field not listing the SIP UPDATE method; or

2) SIP 405 (Method Not Allowed) response was received to the SIP UPDATE sent towards the remote UE in the early dialog;

the EATF shall create a new dialog by sending a SIP 183 (Session Progress) response to the SIP INVITE request due to E-STN-SR. The EATF shall populate the SIP 183 (Session Progress) response with an SDP answer:

1) with c-line set to the unspecified address (0.0.0.0) if IPv4 or to a domain name within the ".invalid" DNS top-level domain in case of IPv6 as described in IETF RFC 6157 [74]; and

2) including media of media types received in SDP offer of the SIP INVITE request due to E-STN-SR, which are also offered in the SIP INVITE request from the served user.

If the EATF supports the PS to CS SRVCC for originating emergency sessions in pre-alerting phase and if there are no early dialogs related to the originating call not accepted yet available for the served user and there is a SIP INVITE request from the served user for which a final SIP response has not been received yet, the EATF shall send a SIP 183 (Session Progress) response to the SIP INVITE request due to E-STN-SR. The EATF shall populate the SIP 183 (Session Progress) response with an SDP answer:

1) with c-line set to the unspecified address (0.0.0.0) if IPv4 or to a domain name within the ".invalid" DNS top-level domain in case of IPv6 as described in IETF RFC 6157 [74]; and

2) including media of media types received in SDP offer of the SIP INVITE request due to E-STN-SR, which are also offered in the SIP INVITE request from the served user.

Upon receiving the first SIP PRACK request from the target access leg, the EATF shall send a 200 (OK) to the PRACK response and then send a SIP INFO request towards the MSC server as specified in 3GPP TS 24.229 [2] and IETF RFC 6086 [54] in the dialog created by the SIP INVITE request due to E-STN-SR. The EATF shall populate the SIP INFO request as follows:

1) include the Info-Package header field as specified in IETF RFC 6086 [54] with 3gpp.state-and-event info package name; and

2) include application/vnd.3gpp.state-and-event-info+xml XML body associated with the info package according to IETF RFC 6086 [54] and containing an XML body compliant to the XML schema specified in clause D.2:

   A) if a SIP 180 (Ringing) response to the SIP INVITE request has already been forwarded to the served SC UE before receiving the INVITE due to E-STN-SR, with the state-info XML element containing "early" the direction XML element containing "initiator"; and

   B) if the EATF supports the PS to CS SRVCC for originating emergency sessions in pre-alerting phase and if a SIP 180 (Ringing) response to the SIP INVITE request has not beenforwarded to the served SC UE before receiving the INVITE due to E-STN-SR, with the state-info XML element containing "pre-alerting" and the direction XML element containing "initiator".

Upon receiving the 200 (OK) response to the SIP INFO request the EATF shall:

1) start forwarding SIP messages from the remote UE to the MSC server as specified in 3GPP TS 24.229 [2] and the present specification for dialogs where a PRACK request is received from the MSC server; and

2) if SIP 18x responses were stored after receiving the SIP INVITE request due to E-STN-SR, then for each early dialog where a PRACK request is received from the MSC server:

   A) forward all stored SIP 18x responses to the MSC server; and

   B) if a P-Early-Media header field is stored, include the P-Early-Media header field that was stored most recently in the SIP 18x response.

If a reliable SIP 1xx response or a SIP 2xx response is received on the remote leg, the SIP response is to the SIP INVITE request from the served user, the SIP response contains an SDP answer, and an SDP answer has not been received from the remote UE on the dialog of the SIP response yet, the EATF shall:

1) if the SIP 1xx response is received, send a SIP PRACK request on the remote leg as specified in 3GPP TS 24.229 [2]. The EATF shall populate the SIP PRACK request with the SDP offer received in the SIP INVITE request due to E-STN-SR as specified in 3GPP TS 24.229 [2]. Upon receiving the SIP 200 (OK) response to the SIP PRACK request, the EATF shall send a SIP 183 (Session Progress) response to the SIP INVITE request due to E-STN-SR as specified in 3GPP TS 24.229 [2]. The EATF shall populate the SIP 183 (Session Progress) response with:

   A) the SDP answer received in the SIP 200 (OK) response to the SIP PRACK request as specified in 3GPP TS 24.229 [2]; and

   B) if the SIP INVITE request due to E-STN-SR contains a P-Early-Media header field with the "supported" parameter and if the EATF has received a P-Early-Media header field in a SIP message in the dialog of the SIP PRACK request, a P-Early-Media header field containing the value of the last P-Early-Media header field received in a SIP message in the dialog of the SIP PRACK request; and

2) if the SIP 2xx response is received, send a SIP ACK request on the remote leg as specified in 3GPP TS 24.229 [2] and send a SIP UPDATE request or SIP re-INVITE request on the remote leg as specified in 3GPP TS 24.229 [2]. The EATF shall populate the SIP UPDATE request or SIP re-INVITE request with the SDP offer received in the SIP INVITE request due to E-STN-SR as specified in 3GPP TS 24.229 [2]. Upon receiving the SIP 200 (OK) response to the SIP UPDATE request or SIP re-INVITE request, the EATF shall send a SIP 200 (OK) response to the SIP INVITE request due to E-STN-SR as specified in 3GPP TS 24.229 [2]. The EATF shall populate the SIP 200 (OK) response with the SDP answer received in the SIP 200 (OK) response to the SIP UPDATE request or SIP re-INVITE request as specified in 3GPP TS 24.229 [2].

If a SIP 2xx response is received on the remote leg, the SIP response is to the SIP INVITE request from the served user, and an SDP answer has already been received from the remote UE on the dialog of the SIP response:

1) if the SDP offer received in the SIP INVITE request due to E-STN-SR has already been sent to the remote UE on the dialog of the SIP 2xx response, accepted with a subsequent SDP answer, the subsequent SDP answer was sent in a 183 (Session Progress) response on a target access leg, and the SIP 2xx response does not contain an SDP answer, then forward the SIP 2xx response on the target access leg as a SIP response to the SIP INVITE request due to E-STN-SR;

2) if the SDP offer received in the SIP INVITE request due to E-STN-SR has already been sent to the remote UE on the dialog of the SIP 2xx response, accepted with a subsequent SDP answer, the subsequent SDP answer was sent in a 183 (Session Progress) response on a target access leg, and the SIP 2xx response contains an SDP answer, then remove the SDP body from the SIP 2xx response or replace the SDP body in the SIP 2xx response with the subsequent SDP answer, and forward the SIP 2xx response as specified in 3GPP TS 24.229 [2] on the target access leg as a SIP response to the SIP INVITE request due to E-STN-SR; and

3) if:

   A) the SDP offer received in the SIP INVITE request due to E-STN-SR has not been sent to the remote UE on the dialog of the SIP response yet; or

   B) the SDP offer received in the SIP INVITE request due to E-STN-SR has already been sent to the remote UE in a SIP UPDATE request within the dialog of the SIP response and the SIP UPDATE request was rejected with SIP 405 (Method Not Allowed) response;

   send a SIP ACK request on the remote leg as specified in 3GPP TS 24.229 [2] and send a SIP re-INVITE request on the remote leg as specified in 3GPP TS 24.229 [2]. The EATF shall populate the SIP re-INVITE request with the SDP offer received in the SIP INVITE request due to E-STN-SR as specified in 3GPP TS 24.229 [2]. Upon receiving the SIP 200 (OK) response to the SIP re-INVITE request, the EATF shall send a SIP ACK request on the remote leg as specified in 3GPP TS 24.229 [2] and shall send a SIP 200 (OK) response to the SIP INVITE request due to E-STN-SR as specified in 3GPP TS 24.229 [2] using a dialog different to the dialog of SIP 183 (Session Progress) response to the SIP INVITE request due to E-STN-SR. The EATF shall populate the SIP 200 (OK) response with the SDP answer received in the SIP 200 (OK) response to the SIP re-INVITE request as specified in 3GPP TS 24.229 [2].

The EATF shall remove non-transferred audio and video media components and superfluous sessions as specified in subclause 12.5.4.

12.5.4 Removal of non-transferred audio media components and superfluous sessions

If no dialog was associated with the SIP INVITE request due to E-STN-SR, the EATF shall send a SIP 480 (Temporarily Unavailable) response to reject the SIP INVITE request due to E-STN-SR.

Upon receiving the SIP ACK request from the Target Access Leg, and after an operator specific timer has expired, the EATF:

1) shall release the source access leg of the session associated with the SIP INVITE request due to E-STN-SR, as described in 3GPP TS 24.229 [2]; and

NOTE 1: If non-speech media was part of the original emergency session, the non-speech media will be released.

NOTE 2: Delaying the release of the source access leg as described above allows an SC UE to reuse the PS dialog in case of PS to CS SRVCC cancellation.

2) for each session in the transferable set which was not transferred, shall release remote leg and source access leg.

12.6 MSC server enhanced for SRVCC using SIP interface

12.6.0 General

The MSC server enhanced for SRVCC using SIP interface may support the codec inquiry prior to the PS to CS SRVCC access transfer.

When the MSC server enhanced for SRVCC using SIP interface receives an indication for a PS to CS SRVCC session transfer as described in 3GPP TS 23.216 [49], the MSC server may perform the codec inquiry prior to PS to CS SRVCC access transfer as specified in subclause 12.6.0A according to local policy.

NOTE: The local policy can be based on MSC server configuration of STN-SR(s) owned by ATCF(s) supporting the codec inquiry prior to PS to CS SRVCC access transfer.

If the MSC server enhanced for SRVCC using SIP interface does not support or does not perform the codec inquiry prior to PS to CS SRVCC access transfer as specified in subclause 12.6.0A, the MSC server shall perform the procedures in subclause 12.6.1.

The MSC server enhanced for SRVCC using SIP interface may support the codec re-negotiation after the PS to CS SRVCC access transfer as specified in subclause 12.6.0B.

12.6.0A Codec inquiry prior to PS to CS SRVCC access transfer

In order to perform the codec inquiry prior to PS to CS SRVCC access transfer, the MSC server shall send a SIP OPTIONS request according to 3GPP TS 24.229 [2]. In the SIP OPTIONS request, the MSC server:

1) shall set the Request-URI to the STN-SR received over Sv interface as described in 3GPP TS 23.216 [49];

2) shall set the P-Asserted-Identity header field to the C-MSISDN received over Sv interface as described in 3GPP TS 23.216 [49]; and

3) shall include an application/vnd.3gpp.PS-to-CS-preparation+xml body specified in subclause D.6 carrying the PS-to-CS-preparation-request. In the PS-to-CS-preparation-request, the MSC server may include an <MSC-server-supported-codec-list> element containing an SDP body with one audio m= line with one or more RTP payload types that are supported commonly by the served SC UE, the target RAN and the target CS-MGW selected by the MSC server. The MSC server shall associate the RTP payload type(s) with the RTP payload type number(s) according to local policy.

If no SIP final response is received within a time defined by local policy to the SIP OPTIONS request, the MSC server shall perform the procedures in subclause 12.6.1 or subclause 12.4.0.2.

Upon receiving a SIP 3xx, 4xx, 5xx or 6xx response to the SIP OPTIONS request, the MSC server shall perform the procedures in subclause 12.6.1 or subclause 12.4.0.2.

Upon reception of a SIP 2xx response to the SIP OPTIONS request:

1) if the SIP 2xx response does not contain an application/vnd.3gpp.PS-to-CS-preparation+xml body specified in subclause D.6 carrying the PS-to-CS-preparation-response, the MSC server shall perform the procedures in subclause 12.6.1 or subclause 12.4.0.2; and

2) if the SIP 2xx response contains an application/vnd.3gpp.PS-to-CS-preparation+xml body specified in subclause D.6 carrying the PS-to-CS-preparation-response, the MSC server:

   a) if the PS-to-CS-preparation-response indicates that the PS to CS SRVCC access transfer is currently possible, shall perform the procedures in subclause 12.6.1 or subclause 12.4.0.2; and

   b) if the PS-to-CS-preparation-response indicates that the PS to CS SRVCC access transfer is currently not possible, can send SRVCC PS to CS Response as specified in 3GPP TS 23.216 [49] with a reject cause or can repeat the procedures of the present subclause after a time defined by local policy.

12.6.0B Codec re-negotiation after session transfer

If a PS to CS SRVCC access transfer has been successfully performed, different RTP payload types were selected during the PS to CS SRVCC access transfer on the CS radio interface and on the dialog in the IMS towards the remote peer, the MSC server may send a SIP re-INVITE request or a SIP UPDATE request on the dialog in the IMS towards the remote peer. In the SIP re-INVITE request or the SIP UPDATE request, the MSC server shall include an SDP offer. In the SDP offer, the MSC server shall indicate a speech media component with RTP payload types supported by the MSC server, with the RTP payload type used at the CS radio interface as the most preferred RTP payload type.

NOTE: If the RTP payload type used at the CS radio interface is compatible with the RTP payload type of the speech media component used on the dialog in the IMS towards the remote peer, then the MGW controlled by the MSC server can influence the encoding of RTP packets by the remote peer using a media plane means, e.g. Codec Mode Request of AMR-WB or EVS sent within the media stream of the speech media component, as described in 3GPP TS 26.114 [68].

12.6.1 Session transfer from MSC server enhanced for SRVCC using SIP interface

12.6.1.1 Session transfer from MSC server enhanced for SRVCC using SIP interface supporting PS to CS SRVCC

In order to perform the PS to CS SRVCC access transfer, the MSC server enhanced for SRVCC using SIP interface shall initiate a SIP INVITE request and shall:

1) set the Request URI to the STN-SR for the session with speech media component to be transferred;

2) set the P-Asserted-Identity header field to the Correlation MSISDN;

3) set the Contact header field to the contact address of the MSC server;

4) include an SDP offer containing only a speech media component. If the MSC server performed procedures in subclause 12.6.0A and received a PS-to-CS-preparation-response, the MSC server should copy into the SDP offer one or more RTP payload types (each comprising of an RTP payload type number indicated in a sub-field of an <fmt> portion of an "m=" line and, if included, an "a=rtpmap" attribute and an "a=frmtp" attribute for the RTP payload type number) indicated in the <IMS-preferred-codec-list> element of the received PS-to-CS-preparation-response described in subclause 12.6.0A; and

5) if SRVCC with priority handling (as described in 3GPP TS 23.216 [49]) is supported and a Allocation/Retention priority (ARP) indication is received (as described in 3GPP TS 29.280 [71]), then include an authorised Resource-Priority header field;

NOTE 1: An MSC server enhanced for SRVCC using a SIP interface will use local configuration to map the received ARP value to appropriate values for the authorised Resource-Priority header field.

6) if the MSC server supports the MSC server assisted mid-call feature:

   A. insert the Supported header field containing the option-tag "norefersub" specified in IETF RFC 4488 [20];

   B. insert the Accept header field containing the MIME type as specified in subclause D.1.3;

   C. include in the Contact header field the g.3gpp.mid-call media feature tag as described in annex C; and

   D. insert the Recv-Info header field containing the g.3gpp.mid-call package name;

7) if the MSC server enhanced for SRVCC using SIP interface supports the PS to CS SRVCC for calls in alerting phase, then include:

   a) an Accept header field containing the MIME type application/vnd.3gpp.state-and-event-info+xml as specified in subclause D.2.3;

   b) a Contact header field containing the g.3gpp.srvcc-alerting media feature tag as described in annex C;

   c) a Recv-Info header field containing the g.3gpp.state-and-event package name;

   d) a P-Early-Media header field containing the "supported" parameter;

   e) if the MSC server enhanced for SRVCC using SIP interface supports the PS to CS SRVCC for originating calls in pre-alerting phase, include the g.3gpp.ps2cs-srvcc-orig-pre-alerting media feature tag as described in annex C into the Contact header field;

   f) if the MSC server does not support the MSC server assisted mid-call feature, a Supported header field containing the option-tag "norefersub" specified in IETF RFC 4488 [20]; and

g) if the MSC server enhanced for SRVCC using SIP interface supports the PS to CS SRVCC for terminating calls in pre-alerting phase, include the g.3gpp.ps2cs-srvcc-term-pre-alerting media feature tag as described in annex C into the Contact header field;

NOTE 2: IETF RFC 3261 [19] recommends user agent client to include a Supported header field in any SIP INVITE request, listing option tags for extensions to SIP understood by the user agent client. In the step above, the MSC server is mandated to include at least "norefersub" option tag in the Supported header field.

NOTE 3: If the MSC server supports the MSC server assisted mid-call feature, a Supported header field containing the option-tag "norefersub" was already inserted in preceding steps.

8) signalling elements described in subclause 6A.7.1 and shall indicate the related local preconditions as met; and

9) include the P-Access-Network-Info header field in the SIP INVITE request as specified in 3GPP TS 24.229 [11]. The P-Access-Network-Info header field shall include:

   a) an access-type field set to "3GPP-GERAN", "3GPP-UTRAN-FDD", "3GPP-UTRAN-TDD", or an access-class field set to"3GPP-GERAN", "3GPP-UTRAN";

   b) if available, a "cgi-3gpp" or "utran-sai-3gpp" parameter;

   c) if available a "local-time-zone" parameter;

   d) a "network-provided" parameter; and

   e) if available, a "daylight-saving-time" parameter.

Upon receiving a SIP 200 (OK) response as the first response to the INVITE due to STN-SR (with the exception of the SIP 100 (Trying) response) and if the MSC server does not apply the MSC server assisted mid-call feature procedures in subclause 12.4A, the MSC server shall enter the "active" (N10) state (defined in 3GPP TS 24.008 [8]), regard the access transfer of the session with active speech media component as completed, send an ACK request as specified in 3GPP TS 24.229 [2] and start interworking CC messages as specified in subclause 12.6.5.If the MSC server enhanced for SRVCC using SIP interface supports the MSC server assisted mid-call feature then it shall additionally apply the procedures defined in subclause 12.4A.

If the MSC server enhanced for SRVCC using SIP interface supports the PS to CS SRVCC for calls in alerting phase then in addition to the procedures in this subclause it shall additionally apply the procedures defined in subclause 12.6.3.

12.6.1.2 Session transfer from MSC server enhanced for SRVCC using SIP interface supporting vSRVCC

When an MSC server enhanced for SRVCC using SIP interface and supporting vSRVCC receives an indication for a vSRVCC session transfer as described in 3GPP TS 23.216 [49], then the MSC server enhanced for SRVCC using SIP interface shall initiate a SIP OPTIONS request and shall:

1) set the request URI to the STN-SR;

2) set the P-Asserted-Identity header field to the Correlation MSISDN;

3) set the Contact header field to the address of the MSC server; and

4) set the Accept header field to "application/sdp".

When an MSC server enhanced for SRVCC using SIP interface and supporting vSRVCC receives a SIP 200 (OK) response to the SIP OPTIONS request with an SDP body that contains "m=" lines for audio and video, the MSC server enhanced for SRVCC using SIP interface shall:

1) initiate a SIP INVITE request and shall:

   a) set the request URI to the STN-SR for the session with speech and video media components to be transferred;

   b) set the P-Asserted-Identity header field to the Correlation MSISDN;

   c) set the Contact header field to the address of the MSC server;

d) include an SDP offer containing only a speech media component and a video media component with default codecs for speech and video (as specified in 3GPP TS 26.111 [69]);

e) if the MSC server enhanced for SRVCC using SIP interface supports the PS to CS SRVCC for calls in alerting phase, then include:

i) an Accept header field containing the MIME type application/vnd.3gpp.state-and-event-info+xml as specified in subclause D.2.3;

ii) a Contact header field containing the g.3gpp.srvcc-alerting media feature tag as described in annex C; and

iii) a Recv-Info header field containing the g.3gpp.state-and-event package name;

f) if vSRVCC with priority handling (as described in 3GPP TS 23.216 [49]) is supported and a Allocation/Retention priority (ARP) indication is received (as described in 3GPP TS 29.280 [70]), then include an authorised Resource-Priority header field; and

NOTE: An MSC server enhanced for SRVCC using a SIP interface will use local configuration to map the received ARP value to appropriate values for the authorised Resource-Priority header field.

2) if the MSC server enhanced for SRVCC using SIP interface supports the PS to CS SRVCC for calls in alerting phase, then additionally apply the procedures defined in subclause 12.6.3.

Upon receiving 200 (OK) response to the INVITE due to STN-SR the MSC server shall regard the access transfer of session with active speech and video media components as completed, send an ACK request as specified in 3GPP TS 24.229 [2], and start interworking CC messages as specified in subclause 12.6.5.When an MSC server enhanced for SRVCC using SIP interface and supporting vSRVCC receives a SIP 200 (OK) response to the SIP OPTIONS request with an SDP body that contains an "m=" line for audio but not video, the MSC server enhanced for SRVCC using SIP interface shall follow the procedures in subclause 12.6.1.1.

12.6.2 Emergency session transfer from MSC server enhanced for SRVCC using SIP interface

When an MSC server enhanced for SRVCC using SIP interface receives an indication for a session transfer for an emergency session as described in 3GPP TS 23.216 [49], then the MSC server enhanced for SRVCC using SIP interface shall initiate a SIP INVITE request and shall:

1) set the request URI to the E-STN-SR for the session with speech media component to be transferred;

2) include the sip.instance media feature tag as specified in IETF RFC 5626 [22] with a value based on the IMEI as defined in 3GPP TS 23.003 [12] in the Contact header field;

3) set the P-Asserted-Identity header field to the Correlation MSISDN if one is available;

4) include an SDP offer with media which the MSC server wishes to use in the session;

5) include the P-Access-Network-Info header field in the SIP INVITE request as specified in 3GPP TS 24.229 [11]. The P-Access-Network-Info header field shall include:

a) an access-type field set to "3GPP-GERAN", "3GPP-UTRAN-FDD", "3GPP-UTRAN-TDD", or an access-class field set to"3GPP-GERAN", "3GPP-UTRAN";

b) if available, a "cgi-3gpp" or "utran-sai-3gpp" parameter;

c) if available a "local-time-zone" parameter;

d) a "network-provided" parameter; and

e) if available, a "daylight-saving-time" parameter; and

6) if the MSC server enhanced for SRVCC using SIP interface supports the PS to CS SRVCC for originating emergency sessions in alerting phase, then include:

a) an Accept header field containing the MIME type application/vnd.3gpp.state-and-event-info+xml as specified in subclause D.2.3;

b) in the Contact header field the g.3gpp.srvcc-alerting media feature tag as described in annex C;

c) a Recv-Info header field containing the g.3gpp.state-and-event package name;

d) a P-Early-Media header field containing the "supported" parameter; and

e) if the MSC server enhanced for SRVCC using SIP interface supports the PS to CS SRVCC for originating emergency sessions in pre-alerting phase, include the g.3gpp.ps2cs-srvcc-orig-pre-alerting media feature tag as described in annex C into the Contact header field.

Upon receiving a SIP 200 (OK) response as the first response to the INVITE due to E-STN-SR (with the exception of the SIP 100 (Trying) response), the MSC server shall enter the "active" (N10) state (defined in 3GPP TS 24.008 [8]), regard the access transfer of the session with active speech media component as completed, send an ACK request as specified in 3GPP TS 24.229 [2] and start interworking CC messages as specified in subclause 12.6.5.

If the MSC server enhanced for SRVCC using SIP interface supports the PS to CS SRVCC for originating emergency sessions in alerting phase then in addition to the procedures in this subclause it shall additionally apply the procedures defined in subclause 12.6.3.

12.6.3 MSC server enhanced for SRVCC using SIP interface procedures for PS to CS access transfer for calls in alerting phase or pre-alerting phase

Upon receiving a SIP 1xx response with P-Early-Media header field authorizing backward early media, unless the CS-MGW has already been through-connected, the MSC server instructs the CS-MGW to through-connect.

Upon receiving a SIP INFO request inside the early dialog created with the SIP INVITE request due to STN-SR or E-STN-SR:

1. with the Info-Package header field containing the g.3gpp.state-and-event; and

2. containing an application/vnd.3gpp.state-and-event-info+xml XML body associated with the info package according to IETF RFC 6086 [54] and compliant to the XML schema specified in the subclause D.2 with the state-info XML element containing "early" and direction XML element containing "initiator";

the MSC server enhanced for SRVCC using SIP interface shall enter the "call delivered" (N4) state as specified in 3GPP TS 24.008 [8]. The MSC server enhanced for SRVCC using SIP interface shall associate this session with transaction identifier value and TI flag as described in 3GPP TS 24.008 [8], regard the access transfer of the session in the alerting phase as completed and start interworking CC messages as specified in subclause 12.6.5.

If the MSC server enhanced for SRVCC using SIP interface supports the PS to CS SRVCC for originating calls in pre-alerting phase then upon receiving a SIP INFO request inside the early dialog created with the SIP INVITE request due to STN-SR or E-STN-SR:

1. with the Info-Package header field containing the g.3gpp.state-and-event; and

2. containing an application/vnd.3gpp.state-and-event-info+xml XML body associated with the info package according to IETF RFC 6086 [54] and compliant to the XML schema specified in the clause D.2 with the state-info XML element containing "pre-alerting" and direction XML element containing "initiator";

the MSC server enhanced for SRVCC using SIP interface shall associate this session with transaction identifier value and TI flag as described in 3GPP TS 24.008 [8] and shall enter the state "Mobile originating call proceeding" (N3) as specified in 3GPP TS 24.008 [8], regard the access transfer of the session in the originating pre-alerting phase as completed and start interworking CC messages as specified in subclause 12.6.5.

NOTE 1: The MSC server enhanced for SRVCC using SIP interface can send CC PROGRESS message to the UE to prevent CS call release early.

If the MSC server enhanced for SRVCC using SIP interface supports the PS to CS SRVCC for terminating calls in pre-alerting phase then upon receiving a SIP INFO request inside the early dialog created with the SIP INVITE request due to STN-SR:

1. with the Info-Package header field containing the g.3gpp.state-and-event; and

2. containing an application/vnd.3gpp.state-and-event-info+xml XML body associated with the info package according to IETF RFC 6086 [54] and compliant to the XML schema specified in the clause D.2 with the state-info XML element containing "pre-alerting" and direction XML element containing "receiver";

the MSC server enhanced for SRVCC using SIP interface shall associate this session with transaction identifier value and TI flag as described in 3GPP TS 24.008 [8] and shall enter the state "Call present" (N6) as specified in 3GPP TS 24.008 [8].

Upon receiving a SIP INFO request inside the early dialog created with the SIP INVITE request due to STN-SR:

1. with the Info-Package header field containing the g.3gpp.state-and-event; and

2. containing an application/vnd.3gpp.state-and-event-info+xml XML body associated with the info package according to IETF RFC 6086 [54] and compliant to the XML schema specified in the subclause D.2 with the state-info XML element containing "early" and direction set to "receiver";

and when a related CC CONNECT has not been received, the MSC server enhanced for SRVCC using SIP interface shall enter the "call received" (N7) state as specified in 3GPP TS 24.008 [8]. The MSC server enhanced for SRVCC using SIP interface will not generate an in-band ring tone towards the calling party. The MSC server enhanced for SRVCC using SIP interface shall associate this session with transaction identifier value and TI flag as described in 3GPP TS 24.008 [8]. If the CS-MGW has already been through-connected, the MSC server instructs the CS-MGW not to through-connect. The MSC server shall regard the access transfer of the session in the alerting phase as completed and start interworking CC messages as specified in subclause 12.6.5.

Upon receiving a CC ALERT message when in "Mobile terminating call confirmed" (N9) state as specified in 3GPP TS 24.008 [8], the MSC server enhanced for SRVCC using SIP interface shall send a SIP INFO request inside the dialog created with the SIP INVITE request due to STN-SR for access transfer containing:

1. an Info-Package header field as specified in IETF RFC 6086 [54] with 3gpp.state-and-event info package name; and

2. include application/vnd.3gpp.state-and-event-info+xml XML body associated with the info package according to IETF RFC 6086 [54] and compliant to the XML schema specified in the subclause D.2 with the event XML element containing "alerting-started" to indicate that the called party is being alerted.

Upon receiving a CC CONNECT message when in "call received" (N7) state as specified in 3GPP TS 24.008 [8], the MSC server enhanced for SRVCC using SIP interface shall send a SIP INFO request inside the dialog created with the SIP INVITE request due to STN-SR for access transfer containing:

1. an Info-Package header field as specified in IETF RFC 6086 [54] with 3gpp.state-and-event info package name; and

2. include application/vnd.3gpp.state-and-event-info+xml XML body associated with the info package according to IETF RFC 6086 [54] and compliant to the XML schema specified in the subclause D.2 with the event XML element containing "call-accepted" to indicate that the called party has answered the call.

NOTE 2: At this point the MSC server enters the "active" (N10) state in accordance with 3GPP TS 24.008 [8] subclause 5.2.2.6 including instructing the CS-GW to through-connect in both directions.

Upon receiving a CC CONNECT message after having sent the SIP INVITE request due to STN-SR when not yet in "call received" (N7) state as specified in 3GPP TS 24.008 [8], the MSC server enhanced for SRVCC using SIP interface shall store this event and proceed with the procedures in 3GPP TS 24.008 [8] subclause 5.2.2.6.

NOTE 3: At this point the MSC server enters the "active" (N10) state in accordance with 3GPP TS 24.008 [8] subclause 5.2.2.6 including instructing the CS-GW to through-connect in both directions.

Once a related SIP INFO request inside the associated early dialog:

1. with the Info-Package header field containing the g.3gpp.state-and-event; and

2. containing an application/vnd.3gpp.state-and-event-info+xml XML body associated with the info package according to IETF RFC 6086 [54] and compliant to the XML schema specified in the subclause D.2 with the state-info XML element containing "early" and direction set to "receiver";

is received, then

1. void; and

2. the MSC server shall send a SIP INFO request inside the dialog created with the SIP INVITE request due to STN-SR for access transfer containing:

   a) an Info-Package header field as specified in IETF RFC 6086 [54] with 3gpp.state-and-event info package name; and

   b) include application/vnd.3gpp.state-and-event-info+xml XML body associated with the info package according to IETF RFC 6086 [54] and compliant to the XML schema specified in the subclause D.2 with the event XML element containing "call-accepted" to indicate that the called party has answered the call.

NOTE 4: Procedures in the MSC server enhanced for SRVCC using SIP interface on how to store and supervise the reception of the INFO request are left implementation specific.

Upon receiving a SIP REFER request:

1. sent inside the dialog created by the SIP INVITE request due to STN-SR where a received Feature-Caps header field contains the g.3gpp.srvcc-alerting feature-capability indicator as described in annex C;

2. with the Refer-Sub header field containing "false" value;

3. with the Supported header field containing "norefersub" value;

4. with the Refer-To header field containing a SIP URI with the Target-Dialog URI header field; and

5. containing application/vnd.3gpp.state-and-event-info+xml MIME body with the state-info XML element containing "early" or "pre-alerting";

the MSC server shall:

1. handle the SIP REFER request as specified in 3GPP TS 24.229 [2], IETF RFC 3515 [13] as updated by IETF RFC 6665 [81], and IETF RFC 4488 [20] without establishing an implicit subscription; and

NOTE 5: In accordance with IETF RFC 4488 [20], the MSC server inserts the Refer-Sub header field containing the value "false" in the SIP 2xx response to the SIP REFER request to indicate that it has not created an implicit subscription.

2. send a SIP INVITE request transferring the additional transferred session according to 3GPP TS 24.229 [2] and IETF RFC 3515 [13]. The MSC server shall populate the SIP INVITE request as follows:

   A. header fields which were included in the URI in the Refer-To header field of the received SIP REFER request as specified in IETF RFC 3261 [19] except the header field with hname "body";

   B. include the g.3gpp.srvcc-alerting media feature tag as described in annex C in the Contact header field according to IETF RFC 3840 [53];

   C. if the MSC server enhanced for SRVCC using SIP interface supports the PS to CS SRVCC for originating calls in pre-alerting phase, include the g.3gpp. ps2cs-srvcc-orig-pre-alerting media feature tag as described in annex C in the Contact header field according to IETF RFC 3840 [53]; and

   D. the SDP offer with:

      a. the same amount of the media descriptions as in the header field with hname "body" in the URI in the Refer-To header field of the received SIP REFER request;

      b. each "m=" line having the same media type as the corresponding "m=" line in the header field with hname "body" in the URI in the Refer-To header field of the received SIP REFER request;

      c. port set to zero value in each "m=" line whose corresponding "m=" line in the header field with hname "body" in the URI in the Refer-To header field of the received SIP REFER request has port with zero value; and

      d. payload type numbers and their mapping to codecs and media parameters supported by MSC server, not conflicting with those in the "body" URI header field in the URI in the Refer-To header field of the received SIP REFER request; and

NOTE 6: Port can be set to zero or non zero value for the offered "m=" line whose corresponding "m=" line in the header field with hname "body" in the URI in the Refer-To header field of the received SIP REFER request has port with nonzero value.

E. if the MSC server supports CS to PS SRVCC and the SIP REFER request contains the application/vnd.3gpp.srvcc-ext+xml MIME body:

a) the topmost Route header field with the ATCF management URI received in the application/vnd.3gpp.srvcc-ext+xml MIME body of the SIP REFER request and "lr" URI parameter;

b) the Accept header field containing application/vnd.3gpp.access-transfer-events+xml MIME type with the "et" parameter indicating ability to receive "event-type" attribute with the value "2";

c) the Recv-Info header field containing the g.3gpp.access-transfer-events info package name;

d) the application/vnd.3gpp.srvcc-ext+xml MIME body with the <srvcc-ext> root element containing the <Setup-info> element containing the CS to PS SRVCC information received in the application/vnd.3gpp.srvcc-ext+xml MIME body of the SIP REFER request and indicating the "initiator" role of the MSC server in the session set up; and

e) the g.3gpp.ti media feature tag with value as described in subclause C.12 in the Contact header field according to IETF RFC 3840 [53];

F. if the MSC server supports procedures in subclause 22.2:

a) an Accept header field according to IETF RFC 3261 [19] containing the MIME type application/vnd.3gpp.state-and-event-info+xml as specified in subclause D.2.3; and

b) a Recv-Info header field according to IETF RFC 6086 [54] containing the g.3gpp.state-and-event package name;

G. signalling elements described in subclause 6A.7.1 and shall indicate the related local preconditions as met;

H. include the P-Access-Network-Info header field in the SIP INVITE request as specified in 3GPP TS 24.229 [11]. The P-Access-Network-Info header field shall include:

a) an access-type field set to "3GPP-GERAN", "3GPP-UTRAN-FDD", "3GPP-UTRAN-TDD", or an access-class field set to"3GPP-GERAN", "3GPP-UTRAN";

b) if available, a "cgi-3gpp" or "utran-sai-3gpp" parameter;

c) if available a "local-time-zone" parameter;

d) a "network-provided" parameter; and

e) if available, a "daylight-saving-time" parameter; and

I. if an authorised Resource-Priority header field was included in the SIP INVITE request due to STN-SR, then include an authorised Resource-Priority header field with the same values as used in the SIP INVITE request due to STN-SR; and

J. if a P-Asserted-Identity header field is not included in the headers portion of the URI in the Refer-To header field of the received SIP REFER request as specified in IETF RFC 3261 [19], include a P-Asserted-Identity header field with the value of the C-MSISDN contained in the SIP INVITE requests due to STN-SR which created the dialog in which the REFER request is received;

3. if application/vnd.3gpp.state-and-event-info+xml MIME body contains the state-info XML element containing "early" and the direction XML element containing "initiator", then enter the "call delivered" (N4) state as specified in 3GPP TS 24.008 [8] for the CS call with transaction identifier 1 and TI flag value as in mobile terminated call;

4. if application/vnd.3gpp.state-and-event-info+xml MIME body contains the state-info XML element containing "early" and the direction XML element containing "receiver", then enter the "call received" (N7) state as specified in 3GPP TS 24.008 [8] for the CS call with transaction identifier 1 and TI flag value as in mobile terminated call. The MSC server will not generate an in-band ring tone towards the calling party; and

5. if application/vnd.3gpp.state-and-event-info+xml MIME body contains the state-info XML element containing "pre-alerting" and the direction XML element containing "initiator", then enter the "Mobile originating call proceeding" (N3) state as specified in 3GPP TS 24.008 [8] for the CS call with transaction identifier 1 and TI flag value as in mobile terminated call.

Upon receiving the SIP 183 (Session Progress) response to the SIP INVITE request transferring the additional transferred session, the MSC server shall regard the access transfer of the additional session as completed, send a SIP PRACK request as specified in 3GPP TS 24.229 [2] and start interworking CC messages as specified in subclause 12.6.5.

Upon receiving a CC CONNECT message with transaction identifier 1 and TI flag value as in mobile terminated call when in "call received" (N7) state as specified in 3GPP TS 24.008 [8], the MSC server shall send a SIP INFO request inside the dialog created by the SIP INVITE request transferring the additional transferred session containing:

NOTE 7: If the SIP INVITE request transferring the additional transferred session has already been sent and an early dialog has not been established by a SIP response to the SIP INVITE request transferring the additional transferred session yet, the MSC server delays sending of the SIP INFO request till after an early dialog is established by a SIP response to the SIP INVITE request transferring the additional transferred session.

1. an Info-Package header field as specified in IETF RFC 6086 [54] with 3gpp.state-and-event info package name; and

2. include application/vnd.3gpp.state-and-event-info+xml XML body associated with the info package according to IETF RFC 6086 [54] and compliant to the XML schema specified in the clause D.2 with the event XML element containing "call-accepted" to indicate that the called party has answered the call.

NOTE 8: At this point the MSC server enters the "active" (N10) state in accordance with 3GPP TS 24.008 [8] subclause 5.2.2.6 including instructing the CS-GW to through-connect in both directions.

NOTE 9: Prior to sending a CC CONNECT, the MSC can send a CC PROGRESS message to attach the user connection (as specified in 3GPP TS 24.008 [8]) to allow the network to provide in-band tones and announcements to the UE, overriding any internally generated alerting indication in the UE.

NOTE 10: The procedure in subclause 5.2.1.6 of 3GPP TS 24.008 [8] results in that the MSC server will instruct the CS-GW to through-connect in both direction and then, after an acknowledgment from the SC UE, the MSC server enters the "active" (N10) state.

NOTE 11: When the access transfer is completed the MSC server can verify the call state of its peer entity using the STATUS ENQUIRY procedure in accordance with procedures in 3GPP TS 24.008 [8] to ensure that SIP requests or SIP responses that have been sent between the SC UE and the SCC AS or the EATF during the handover from the PS domain to the CS domain did not result in incompatible call states. If the call states are incompatible the transferred session are released.

12.6.4 Abnormal cases

12.6.4.1 Permanent response codes

When the MSC server enhanced for SRVCC using SIP interface receives a SIP reject response to the SIP INVITE request due to STN-SR, the MSC server shall regard the SIP response codes listed in subclause 12.4.3.1 as permanent errors.

12.6.4.2 PS to CS SRVCC cancelled by MME/SGSN or failure of the access transfer procedure in the MSC server

If the MSC server enhanced for SRVCC using SIP interface receives a SRVCC PS to CS Cancel Notification from the MME/SGSN or if the access transfer procedure fails for any other reason in the MSC server enhanced for SRVCC using SIP interface, the MSC server shall perform the actions in the subclause 12.4.3.2.

12.6.4.3 Guard timer for the CC CONNECT request elapses

If after having sent a CC CONNECT message, the timer that supervises the CC CONNECT message expires before the MSC server has received a CC CONNECT ACK message, the MSC server shall perform the actions specified in subclause 12.4.3.3.

12.6.5 Interworking of CC messages and SIP messages when PS to CS SRVCC access transfer is completed

When the PS to CS SRVCC access transfer procedure is completed the MSC server shall interwork between the 3GPP profile of SIP as described in 3GPP TS 24.229 [2] and NAS signalling as described in 3GPP TS 24.008 [8] required for the support of IM CN subsystem based multimedia telephony and supplementary services as specified in 3GPP TS 29.292 [18] with the clarifications and exceptions described in this subclause.

When the MSC server is in the "mobile originating call proceeding" (N3) state, the MSC server shall perform interworking as specified in 3GPP TS 29.292 [18] subclauses 5.3.4, 5.3.4a, 5.3.4b, 5.3.6, 5.3.7, 5.3.8 and 5.3.9.When the MSC server is in the "call delivered" (N4) state, the MSC server shall perform interworking as specified in 3GPP TS 29.292 [18] subclauses 5.3.4a, 5.3.4b, 5.3.6, 5.3.7, 5.3.8 and 5.3.9.

When the MSC server is in the "call received" (N7) state receives a CC CONNECT message from the SC UE the MSC server shall send a SIP INFO request as specified in subclause 12.6.3.

When the MSC server is in the "active" (N10) state and in any of the hold and multiparty auxiliary states for a non-emergency session, the MSC server shall perform interworking as specified in 3GPP TS 29.292 [18] subclauses 5.5, 5.6.3 and 5.6.8. When the MSC server is in the "active" (N10) state for an emergency session, the MSC server shall perform interworking as specified in 3GPP TS 29.292 [18] subclauses 5.5.

The MSC server shall interact with CS-GW as specified in 3GPP TS 29.292 [18] subclauses 7.1, 7.3, 7.4, 7.5 and 7.6.

If the MSC server in the "call received" (N7) state:

1) receives a call clearing message from the SC UE, the MSC server shall send a SIP BYE request or a SIP CANCEL request as specified in 3GPP TS 29.292 [18] subclause 5.3.9; or

2) determines due to internal procedures that the call shall be released, the MSC Server shall send a SIP BYE request or a SIP CANCEL request as specified in 3GPP TS 29.292 [18] subclause 5.3.11.

If the MSC server enhanced for SRVCC using SIP interface supports the MSC server assisted mid-call feature and subscribes to the conference event package as described in 3GPP TS 24.147 [7] subclause 5.3.1.2, then after the PS to CS SRVCC access transfer procedure is completed, the MSC server shall apply the interworking procedures as specified in 3GPP TS 29.292 [18] subclauses 5.6.8.2.7.

12.7 Access Transfer Control Function (ATCF)

12.7.1 Distinction of requests

The ATCF needs to distinguish the following initial SIP requests:

1) SIP INVITE requests containing the STN-SR allocated to the ATCF in the Request-URI and:

   A) not containing any Route header field; or

   B) containing a URI in the topmost Route header field other than the ATCF URI for originating requests and other than the ATCF URI for terminating requests.

   In the procedures below, such requests are known as "SIP INVITE requests due to STN-SR".

2) initial SIP INVITE requests containing the STI-rSR allocated to the ATCF in the Request-URI and with the ATCF URI for originating requests in the topmost Route header field. In the procedures below, such requests are known as "SIP INVITE requests due to STI-rSR".

3) SIP INVITE requests containing the ATCF management URI in the Request-URI and:

   - not containing Route header field; or

   - containing a URI in the topmost Route header field other than the ATCF URI for originating requests and other than the ATCF URI for terminating requests.

   In the procedures below, such requests are known as "SIP INVITE requests due to media transfer from MSC Server to ATGW".

4) SIP INVITE requests containing the ATCF URI for anchoring additionally transferred call in ATCF allocated to the ATCF in the Request-URI.

   In the procedures below, such requests are known as "SIP INVITE requests for anchoring call additionally transferred during PS to CS SRVCC in ATCF".

5) SIP OPTIONS requests:

   - containing the STN-SR allocated to the ATCF in the Request-URI; and
   - containing an application/vnd.3gpp.PS-to-CS-preparation+xml body specified in subclause D.6 carrying a PS-to-CS-preparation-request.

   In the procedures below, such requests are known as "SIP OPTIONS requests carrying the PS-to-CS-preparation-request".

The ATCF needs to distinguish the following SIP in-dialog requests:

1) SIP INFO request:

   A) with Info-Package header field with value g.3gpp.access-transfer-events; and

   B) with application/vnd.3gpp.access-transfer-events+xml MIME body associated with the info package according to IETF RFC 6086 [54] and indicating the session transfer notification request.

   In the procedures below, such requests are known as "SIP INFO requests carrying the session transfer notification request".

2) SIP INFO request:

   A) with Info-Package header field with value g.3gpp.access-transfer-events; and

   B) with application/vnd.3gpp.access-transfer-events+xml MIME body associated with the info package according to IETF RFC 6086 [54] and indicating the session transfer preparation.

   In the procedures below, such requests are known as "SIP INFO requests carrying the session transfer preparation".

3) SIP INFO request:

   A) with Info-Package header field with value g.3gpp.access-transfer-events; and

   B) with application/vnd.3gpp.access-transfer-events+xml MIME body associated with the info package according to IETF RFC 6086 [54] and indicating the session transfer cancellation.

   In the procedures below, such requests are known as "SIP INFO requests carrying the session transfer cancellation".

4) SIP INFO request:

   A) with Info-Package header field with value 3gpp.state-and-event; and

   B) with application/vnd.3gpp.state-and-event-info+xml MIME body associated with the info package according to IETF RFC 6086 [54] and with the event XML element containing "call-accepted" as specified in annex C.

   In the procedures below, such requests are known as "SIP INFO requests carrying a "call-accepted" indication"

5) SIP REFER request:

   A) with the Refer-Sub header field containing "false" value;

   B) with the Supported header field containing "norefersub" value;

   C) with the Refer-To header field containing a SIP URI with the Target-Dialog URI header field;

   D) sent inside a SIP dialog:

      a. created by the SIP INVITE request due to STN-SR;

b. where one of the following is true:

- the g.3gpp.mid-call feature-capability indicator as specified in annex C was included in the Feature-Caps header field of the SIP 2xx response to the SIP INVITE request due to STN-SR and the SIP REFER request contains a MIME body of MIME type specified in the subclause D.1.3; or

- the g.3gpp.srvcc-alerting feature-capability indicator as specified in annex C was included in the Feature-Caps header field of the SIP 2xx response to the SIP INVITE request due to STN-SR and the SIP REFER request contains application/vnd.3gpp.state-and-event-info+xml MIME body with the state-info XML element containing "early" or "pre-alerting"; and

E) not containing a MIME body of MIME type specified in the subclause D.4.4

In the procedures below, such requests are known as "SIP REFER requests for transferring additional call".

12.7.2 ATCF procedures for PS to CS access transfer, PS to CS SRVCC

12.7.2.1 General

Upon receiving the SIP INVITE request due to STN-SR or the SIP OPTIONS request carrying the PS-to-CS-preparation-request, the ATCF shall:

1) determine the transferable session set which are all the sessions with a speech media component:

   a) associated with C-MSISDN equal:

      i) to the URI in the P-Asserted-Identity header field of the SIP INVITE request due to STN-SR; or

      ii) to the URI in the P-Asserted-Identity header field of the SIP OPTIONS request due to STN-SR; and

   b) where during establishment of the session a Feature-Caps header field containing the g.3gpp.srvcc feature-capability indicator was received in the initial SIP request or SIP response; and

NOTE: These sessions potentially include recently released sessions for which the ATCF temporarily retains session state according to subclause 12.7.2.3.

2) determine the session being transferred which is a session:

   a) in the transferable session set;

   b) for which the SIP 2xx response for the initial SIP INVITE request to establish this session has been sent or received; and

   c) with active speech media component which has been made active most recently.

12.7.2.1A Determination of session being transferred when only a held session or only a session in originating pre-alerting phase or a session in alerting phase exist

If the transferable session set determined as specified in subclause 12.7.2.1 is not empty and each session in the transferable session set:

1) is in an early dialog state; or

2) is in a confirmed dialog state and contains an inactive speech media component;

the ATCF:

1) if:

   a) one or more confirmed dialogs supporting a session with an inactive speech media component exists in the transferable session set;

   b) the Feature-Caps header field provided by the SCC AS towards the SC UE includes the g.3gpp.mid-call feature-capability indicator as described in annex C; and

c) the Contact header field provided by the SC UE to the SCC AS includes the g.3gpp.mid-call media feature tag (as described in annex C);

shall select the confirmed dialog supporting a session with an inactive speech media component that became inactive most recently as the session being transferred; and

2) if no confirmed dialog supporting a session with an inactive speech media component exists in the transferable session set but there are one or more dialogs in the transferable session set supporting sessions where the SC UE has completed a reliable offer / answer procedure and with an active speech media component such that:

a) a SIP 180 (Ringing) response to a SIP INVITE request was received in at least one of those early dialogs, all such SIP 180 (Ringing) responses are responses to the same SIP INVITE request and at least one of such SIP 180 (Ringing) responses was received in an early dialog supporting session with an active speech media component;

b) the Contact header field provided by the SC UE includes the g.3gpp.srvcc-alerting media feature tag as described in annex C; and

c) the Feature-Caps header field provided by the SCC AS towards the SC UE includes the g.3gpp.srvcc-alerting feature-capability indicator as described in annex C;

shall select any of the early dialogs where a SIP 180 (Ringing) responses was received as the session being transferred; and

3) if no confirmed dialog supporting a session with an inactive speech media component exists in the transferable session set but there are one or more dialogs in the transferable session set supporting one session where the SC UE has completed a reliable offer / answer procedure and with an active speech media component such that:

a) a SIP 180 (Ringing) response to the SIP INVITE request has not been received yet in any of the existing dialogs;

b) all dialogs are early dialogs created by the same SIP INVITE request;

c) the Contact header field provided by the SC UE includes the g.3gpp.ps2cs-srvcc-orig-pre-alerting media feature tag as described in annex C; and

d) the Feature-Caps header field provided by the SCC AS towards the SC UE includes the g.3gpp.ps2cs-srvcc-orig-pre-alerting feature-capability indicator as described in annex C;

shall select any of the early dialogs as the session being transferred.

12.7.2.2 Active session transfer

Upon receiving a SIP INVITE request due to STN-SR, if a session is in the transferable session set as determined in subclause 12.7.2.1 and the following conditions are true:

- the session is a confirmed dialog with an active speech media component which has been made active most recently;
- the ATGW anchors the media of the session being transferred; and
- if the speech media component of the SDP offer in the SIP INVITE request is the same as the speech media component of the SDP negotiated by the ATCF in the session being transferred or if the ATGW can provide media transcoding between the speech media component in the received SDP offer and the speech media component in the session being transferred;

the ATCF shall act as B2BUA as described in subclause 5.6 and shall:

NOTE 1: At this point, ATCF interacts with ATGW to provide information needed in the procedures below and to request ATGW to start forwarding the audio media from the remote UE to the MSC server. The details of interaction between ATCF and ATGW are out of scope of this document.

0) if ATCF supports CS to PS SRVCC:

a) associate the session being established with the latest SRVCC-related information (see subclause 6A.3.1) containing C-MSISDN equal to the URI in the P-Asserted-Identity header field of the SIP INVITE requests due to STN-SR; and

b) store the value of the g.3gpp.ti media feature tag as described in annex C of the Contact header field of the SIP INVITE request due to STN-SR;

1) send a SIP 200 (OK) response to the received SIP INVITE request due to STN-SR that contains:

a) the Contact header field of the remote UE saved in subclause 7.5.2.2 or subclause 8.4.2.2;

b) the Record-Route header field that contains only the SIP URI pointing to the ATCF;

c) the SDP answer that includes the ATGW ports and the IP addresses as provided by the ATGW, and the directionality attribute shall be the same as the SDP negotiated by ATCF towards the UE in the session being transferred;

d) include the P-Charging-Vector header field as specified in 3GPP TS 24.229 [2] populated as follows:

i) the "icid-value" header field parameter containing the same value as received in the SIP INVITE request due to STN-SR;

ii) the "related-icid" header field parameter containing the ICID value of the source access leg in the P-Charging-Vector header field and saved in subclause 7.5.2.2 or subclause 8.4.2.2;

iii) if an "icid-generated-at" header field was generated for the source access leg, include the "related-icid-generated-at" header field parameter containing the host name or IP address included in the "icid-generated-at" header field parameter of the source access leg and saved in subclause 7.5.2.2 or subclause 8.4.2.2;

iv) the "orig-ioi" header field parameter with the value received in the SIP INVITE request due to STN-SR; and

v) a "term-ioi" header field parameter with the value received:

- if the source access leg is terminated by the SC UE, in the "orig-ioi" header field parameter in the P-Charging-Vector header field received in the initial SIP INVITE request from the remote UE and saved in subclause 8.4.2.2; and

- if the source access leg is an originating call initiated by the SC UE, in the "term-ioi" received in the P-Charging-Vector header field in responses from the remote UE to the initial SIP INVITE request and saved in subclause 7.5.2.2;

NOTE 2: The "transit-ioi" header field parameter can not be copied from the source access leg since the SIP INVITE due to ATU-STI can traverse different transit networks compared to the transit networks traversed when the call was initiated.

e) if ATCF supports CS to PS SRVCC:

A) the Recv-Info header field containing the g.3gpp.access-transfer-events info package name;

B) the Accept header fields received in the home leg of the session being transferred by PS to CS SRVCC except the Accept header field containing the application/vnd.3gpp.access-transfer-events+xml MIME type; and

C) the Accept header field containing the application/vnd.3gpp.access-transfer-events+xml MIME type with the "et" parameter indicating ability to receive "event-type" with value "1", value "3", value "4" and values, if any, indicated in the "et" parameter of the application/vnd.3gpp.access-transfer-events+xml MIME type of the Accept header field received in the home leg of the session being transferred by PS to CS SRVCC; and

f) the Feature-Caps header field(s) according to IETF RFC 6809 [60] received in the home leg of the session being transferred and saved in subclause 7.5.2.2 or subclause 8.4.2.2;

g) the P-Asserted-Identity header field with the identity of the remote user saved in subclause 7.5.2.2 or subclause 8.4.2.2; and

h) if available, the Privacy header field saved in subclause 7.5.2.2 or subclause 8.4.2.2; and

NOTE 3: At this point the ATCF requests the ATGW to start forwarding the audio media from the MSC server to the remote UE. The details of interaction between ATCF and ATGW are out of scope of this document.

2) initiate a new dialog toward the SCC AS (i.e. a target access leg) by sending an initial SIP INVITE request due to ATU-STI for PS to CS SRVCC toward the SCC AS populated with:

a) the SDP offer containing the currently used media with ATGW ports and IP addresses towards the remote UE as provided by the ATGW. The ATCF shall include in the SDP offer only the media of the media types offered in the received SIP INVITE request due to STN-SR;

b) the Request-URI containing the ATU-STI for PS to CS SRVCC previously received from the SCC AS and associated with the session being transferred; and

c) the Target-Dialog header field with the dialog identifier of the session being transferred;

d) the Require header field containing the option tag "tdialog";

e) the Contact header field that contains the contact information received in the SIP INVITE request due to STN-SR;

f) the Record-Route header field that includes only the ATCF SIP URI, where the ATCF wants to receive subsequent the in-dialog requests from the SCC AS;

NOTE 4: The ATCF SIP URI included in the Record-Route header field is used by the SCC AS to build a Route header field that the SCC AS will use when sending the in-dialog request toward the ATCF.

g) the P-Asserted-Identity header field that is the same as the P-Asserted-Identity header field received in the SIP INVITE request due to STN-SR;

h) all header fields which are included in the SIP INVITE request due to STN-SR and which contain option tag(s);

i) if the Recv-Info header field is included in the SIP INVITE request due to STN-SR, the Recv-Info header field that is the same as the Recv-Info header field received in the SIP INVITE request due to STN-SR except, if the ATCF supports the CS to PS SRVCC, the Recv-Info header field containing the g.3gpp.access-transfer-events info package name;

j) if the Accept header field is included in the SIP INVITE request due to STN-SR, the Accept header field that is the same as the Accept header field received in the SIP INVITE request due to STN-SR except, if the ATCF supports the CS to PS SRVCC, the Accept header field containing the application/vnd.3gpp.access-transfer-events+xml MIME type;

k) if the ATCF supports the CS to PS SRVCC, if an Accept header field of the SIP INVITE request due to STN-SR contains the application/vnd.3gpp.access-transfer-events+xml with the "et" parameter indicating ability to receive "event-type" attribute with values additional to the value "2":

a) the Accept header field containing the application/vnd.3gpp.access-transfer-events+xml MIME type with the "et" parameter indicating ability to receive "event-type" attribute with the additional values; and

b) the Recv-Info header field containing the g.3gpp.access-transfer-events info package name.

l) if the SIP INVITE request due to STN-SR contains a P-Access-Network-Info header field, a P-Access-Network-Info header field copied from the SIP INVITE request due to STN-SR.

If a session is in the transferable session set as determined in subclause 12.7.2.1, ATCF does not support CS to PS SRVCC and one of the following conditions are true:

- the ATGW does not anchor the media of the session being transferred; or

- if the speech media component of the SDP offer in the SIP INVITE request is not the same as the speech media component of the SDP negotiated by the ATCF in the session being transferred and the ATGW cannot provide media transcoding between the speech media component in the received SDP offer and the speech media component in the session being transferred;

the ATCF shall act as proxy and shall:

1) replace the Request-URI in the received SIP INVITE request due to STN-SR with the ATU-STI for PS to CS SRVCC associated with the session being transferred;

before forwarding the request.

If a session being transferred was determined in subclause 12.7.2.1, ATCF supports CS to PS SRVCC and one of the following conditions are true:

- the ATGW does not anchor the media of the session being transferred; or
- if the speech media component of the SDP offer in the SIP INVITE request is not the same as the speech media component of the SDP negotiated by the ATCF in the session being transferred and the ATGW cannot provide media transcoding between the speech media component in the received SDP offer and the speech media component in the session being transferred;

the ATCF shall act as B2BUA and shall:

1) associate the session being established with the latest SRVCC-related information (see subclause 6A.3.1) containing C-MSISDN equal to the URI in the P-Asserted-Identity header field of the SIP INVITE requests due to STN-SR;

2) store the value of the g.3gpp.ti media feature tag as described in annex C of the Contact header field of the SIP INVITE request due to STN-SR; and

3) send a SIP INVITE request due to ATU-STI for PS to CS SRVCC according to 3GPP TS 24.229 [2]. The ATCF shall populate the SIP INVITE request due to ATU-STI for PS to CS SRVCC with:

   A) the Request-URI set to the ATU-STI for PS to CS SRVCC associated with the session being transferred;

   B) all Route header fields of the SIP INVITE request due to STN-SR except the topmost Route header field;

   C) the Record-Route header field containing the SIP URI of the ATCF;

   D) the Recv-Info header fields of the SIP INVITE request due to STN-SR except the Recv-Info header field containing the g.3gpp.access-transfer-events info package name;

   E) the Accept header fields of the SIP INVITE request due to STN-SR except the Accept header field containing the application/vnd.3gpp.access-transfer-events+xml MIME type;

   F) if an Accept header field of the SIP INVITE request due to STN-SR contains the application/vnd.3gpp.access-transfer-events+xml with the "et" parameter indicating ability to receive "event-type" attribute with values additional to the value "2":

      a) the Accept header field containing the application/vnd.3gpp.access-transfer-events+xml MIME type with the "et" parameter indicating ability to receive "event-type" attribute with the additional values; and

      b) the Recv-Info header field containing the g.3gpp.access-transfer-events info package name;

   G) if the ATCF decided to anchor the media according to operator policy as specified in 3GPP TS 23.237 [9]:

      a) all MIME bodies of the SIP INVITE request due to STN-SR apart from application/sdp MIME body; and

      b) application/sdp MIME body with updated SDP offer using media parameters provided by the ATGW;

NOTE 5: ATCF interacts with ATGW to provide the needed media related information. The details of interaction between ATCF and ATGW are out of scope of this document.

   H) if the ATCF decided not to anchor the media according to operator policy as specified in 3GPP TS 23.237 [9]:

      a) all MIME bodies of the SIP INVITE request due to STN-SR;

   I) if the ATCF is located in the visited network, and local policy requires the application of IBCF capabilities in the visited network towards the home network, select an IBCF in the visited network and add the URI of the selected IBCF to the topmost Route header field.

J) if the SIP INVITE request due to STN-SR contains a P-Access-Network-Info header field, a P-Access-Network-Info header field copied from the SIP INVITE request due to STN-SR.

If the ATCF supports CS to PS SRVCC, when the ATCF receives any SIP 1xx response or SIP 2xx response to the SIP INVITE request due to ATU-STI for PS to CS SRVCC, the ATCF shall:

1) save the Contact header field included in the SIP response; and

NOTE 6: If the ATCF subsequently receives an initial SIP INVITE request due to STI-rSR, the ATCF will include the saved the Contact header field of the remote UE in its SIP 200 (OK) response to the initial SIP INVITE request due to STI-rSR.

2) generate and send a SIP response to the SIP INVITE request due to STN-SR populated with:

A) the same status code as the received SIP response to the SIP INVITE request due to ATU-STI for PS to CS SRVCC; and

B) the Record-Route header field containing the SIP URI of the ATCF;

C) the Recv-Info header fields of the received SIP response except the Recv-Info header field containing the g.3gpp.access-transfer-events info package name;

D) if the SIP response is SIP 1xx response:

a) the Recv-Info header field containing the g.3gpp.access-transfer-events info package name with the "et" parameter indicating ability to receive "event-type" attribute with value "1", value "3", value "4" and values, if any, indicated in the "et" parameter of the g.3gpp.access-transfer-events info package name of the Recv-Info header field of the received SIP response; and

E) if the SIP response is SIP 2xx response:

a) the Recv-Info header field containing the g.3gpp.access-transfer-events info package name;

b) the Accept header fields of the received SIP response except the Accept header field containing the application/vnd.3gpp.access-transfer-events+xml MIME type; and

c) the Accept header field containing the application/vnd.3gpp.access-transfer-events+xml MIME type with the "et" parameter indicating ability to receive "event-type" with value "1", value "3", value "4" and values, if any, indicated in the "et" parameter of the application/vnd.3gpp.access-transfer-events+xml MIME type of the Accept header field of the received SIP response.

12.7.2.3 Abnormal procedures

12.7.2.3.1 P-CSCF releasing the source access leg during PS to CS SRVCC

When the ATCF receives either:

1) a SIP BYE request on the Source Access Leg containing a Reason header field containing a SIP 503 (Service Unavailable) response code, that is terminating an established dialog or an early dialog on the Source Access Leg;

2) a SIP CANCEL request on the Source Access Leg with the Reason header field containing a SIP 503 (Service Unavailable) response code then, that is terminating an early dialog on the Source Access Leg originated by the SC UE; or

3) a SIP 503 (Service Unavailable) response on the Source Access Leg, that is terminating an early dialog on the Source Access Leg terminating at the SC UE;

then:

- the ATCF shall retain session state information and ATGW resources associated with the session until either it receives a SIP INVITE request due to STN-SR or an operator determined period elapses.

NOTE 1: The default value of the operator determined period is 8 seconds.

NOTE 2: The session remains recognizable for PS to CS SRVCC access transfer as shown in subclause 12.7.2.1.

NOTE 3: The SIP BYE request is forwarded to the SCC AS, which also delays release of the session, as described in subclause 12.3.3.2.

12.7.2.3.2 No transferable session exists

Upon receiving a SIP INVITE request due to STN-SR, if the transferable session set determined in subclause 12.7.2.1 does not contain any sessions and the identity in the P-Asserted-Identity header field is a C-MSISDN that is not bound to a registration path in the ATCF, the ATCF shall respond with a SIP 404 (Not Found) response.

If the transferable session set determined in subclause 12.7.2.1 does not contain any sessions and if the identity in the P-Asserted-Identity header field is a C-MSISDN that is bound to a registration path in the ATCF, the ATCF shall:

1) determine whether a transferable SIP INVITE request exists. The transferable SIP INVITE request is a SIP INVITE request sent by SC UE such that:

   A) a final SIP response has not been received yet to the SIP INVITE request;

   B) the session being established by the SIP INVITE request is associated with C-MSISDN equal to the URI in the P-Asserted-Identity header field of the SIP INVITE requests due to STN-SR;

   C) a SIP 1xx response to the SIP INVITE request was received where the SIP 1xx response contained a Feature-Caps header field with the g.3gpp.srvcc feature-capability indicator as described in annex C;

NOTE 0: ATCF can have no dialogs if all the early dialogs were terminated by 199 (Early Dialog Terminated) as described in IETF RFC 6228 [80].

   D) the Contact header field in the SIP INVITE request includes the g.3gpp.ps2cs-srvcc-orig-pre-alerting media feature tag as described in annex C; and

   E) a SIP 1xx response to the SIP INVITE request was received where the SIP 1xx response contained a Feature-Caps header field with the g.3gpp.ps2cs-srvcc-orig-pre-alerting feature-capability indicator as described in annex C;

2) if a transferable SIP INVITE request exists:

   A) if ATCF decides to not anchor media according to local policy and if ATCF does not support CS to PS SRVCC, provide the proxy role as specified in 3GPP TS 24.229 [2] and replace the Request-URI in the received SIP INVITE request due to STN-SR with ATU-STI for PS to CS SRVCC associated with SIP INVITE request before forwarding the request and do not process the remaining steps; and

   B) if ATCF decides to anchor media according to local policy:

      a) if ATCF supports the CS to PS SRVCC:

         - associate the SIP INVITE request with the latest SRVCC-related information (see subclause 6A.3.1) containing C-MSISDN equal to the URI in the P-Asserted-Identity header field of the SIP INVITE requests due to STN-SR; and

         - store the value of the g.3gpp.ti media feature tag as described in annex C of the Contact header field of the SIP INVITE request due to STN-SR; and

      b) provide the role of a B2BUA in accordance with 3GPP TS 24.229 [2] and initiate a new dialog toward the SCC AS (i.e. a target access leg) by sending an initial SIP INVITE request due to ATU-STI for PS to CS SRVCC toward the SCC AS populated with:

         - if ATCF decides to anchor media according to local policy:

            i) the SDP offer containing the media offered in source access leg towards the remote UE, with the currently offered ATGW ports and IP addresses towards the remote UE as provided by the ATGW. The ATCF shall include in the SDP offer only the media of the media types offered in the received SIP INVITE request due to STN-SR; and

            ii) all MIME bodies of the SIP INVITE request due to STN-SR apart from application/sdp MIME body;

- if the ATCF decides not to anchor media according to local policy, all MIME bodies of the SIP INVITE request due to STN-SR;
- the Request-URI containing the ATU-STI for PS to CS SRVCC previously received from the SCC AS and associated with the SIP INVITE request;
- the Contact header field that contains the contact information received in the SIP INVITE request due to STN-SR;
- the Record-Route header field that includes only the ATCF SIP URI, where the ATCF wants to receive subsequent in-dialog requests from the SCC AS;

NOTE 1: The ATCF SIP URI included in the Record-Route header field is used by the SCC AS to build a Route header field that the SCC AS will use when sending the in-dialog request toward the ATCF.

- the P-Asserted-Identity header field that is the same as the P-Asserted-Identity header field received in the SIP INVITE request due to STN-SR;
- all header fields which are included in the SIP INVITE request due to STN-SR and which contain option tag(s);
- if the Recv-Info header field is included in the SIP INVITE request due to STN-SR, the Recv-Info header field that is the same as the Recv-Info header field received in the SIP INVITE request due to STN-SR except, if the ATCF supports the CS to PS SRVCC, the Recv-Info header field containing the g.3gpp.access-transfer-events info package name;
- if the Accept header field is included in the SIP INVITE request due to STN-SR, the Accept header field that is the same as the Accept header field received in the SIP INVITE request due to STN-SR. except, if the ATCF supports the CS to PS SRVCC, the Accept header field containing the application/vnd.3gpp.access-transfer-events+xml MIME type; and
- if the ATCF supports the CS to PS SRVCC and an Accept header field of the SIP INVITE request due to STN-SR contains the application/vnd.3gpp.access-transfer-events+xml with the "et" parameter indicating ability to receive "event-type" attribute with values additional to the value "2":

  i) the Accept header field containing the application/vnd.3gpp.access-transfer-events+xml MIME type with the "et" parameter indicating ability to receive "event-type" attribute with the additional values; and

  ii) the Recv-Info header field containing the g.3gpp.access-transfer-events info package name; and

3) if a transferable SIP INVITE request does not exist, respond with a SIP 480 (Temporarily Unavailable) response.

12.7.2.4 Transfer when only a held session or a session in originating pre-alerting phase or a session in alerting phase exist

Upon receiving a SIP INVITE request due to STN-SR, if the transferable session set determined in subclause 12.7.2.1 is not empty and each session in the transferable session set:

1) is in an early dialog state; or

2) is in a confirmed dialog state and contains inactive speech media component;

then the ATCF shall:

1) if ATCF decides to not anchor media according to local policy and if ATCF does not support CS to PS SRVCC, provide the proxy role as specified in 3GPP TS 24.229 [2] and replace the Request-URI in the received SIP INVITE request due to STN-SR with ATU-STI for PS to CS SRVCC associated with a session in the transferable session set before forwarding the request and do not process the remaining steps;

2) if ATCF decides to anchor media according to local policy, determine the session being transferred as described in subclause 12.7.2.1A;

3) if ATCF supports the CS to PS SRVCC:

a) associate the session being established with the latest SRVCC-related information (see subclause 6A.3.1) containing C-MSISDN equal to the URI in the P-Asserted-Identity header field of the SIP INVITE requests due to STN-SR; and

b) store the value of the g.3gpp.ti media feature tag of the Contact header field of the SIP INVITE request due to STN-SR; and

4) provide the role of a B2BUA in accordance with 3GPP TS 24.229 [2] and initiate a new dialog toward the SCC AS (i.e. a target access leg) by sending an initial SIP INVITE request due to ATU-STI for PS to CS SRVCC toward the SCC AS populated with:

a) if ATCF decides to anchor media according to local policy:

A) if

- only one dialog exists in the session being transferred, the SDP offer containing the currently used media with ATGW ports and IP addresses towards the remote UE as provided by the ATGW. The ATCF shall include in the SDP offer only the media of the media types offered in the received SIP INVITE request due to STN-SR; and

- more than one early dialog exists in a session being transferred, the SDP offer containing the ATGW ports and IP addresses of the selected dialog towards the remote UE as provided by the ATGW and the media types offered in the received SIP INVITE request due to STN-SR;

B) all MIME bodies of the SIP INVITE request due to STN-SR apart from application/sdp MIME body;

C) the Request-URI containing the ATU-STI for PS to CS SRVCC previously received from the SCC AS and associated with the session being transferred;

D) the Target-Dialog header field with the dialog identifier of the session being transferred; and

E) the Require header field containing the option tag "tdialog";

b) if the ATCF supports the CS to PS SRVCC and the ATCF decides not to anchor media according to local policy:

i) all MIME bodies of the SIP INVITE request due to STN-SR; and

ii) the Request-URI containing the ATU-STI for PS to CS SRVCC associated with a session in the transferable session set;

c) the Contact header field that contains the contact information received in the SIP INVITE request due to STN-SR;

d) the Record-Route header field that includes only the ATCF SIP URI, where the ATCF wants to receive subsequent in-dialog requests from the SCC AS;

NOTE 1: The ATCF SIP URI included in the Record-Route header field is used by the SCC AS to build a Route header field that the SCC AS will use when sending the in-dialog request toward the ATCF.

e) the P-Asserted-Identity header field that is the same as the P-Asserted-Identity header field received in the SIP INVITE request due to STN-SR;

f) all header fields which are included in the SIP INVITE request due to STN-SR and which contain option tag(s);

g) if the Recv-Info header field is included in the SIP INVITE request due to STN-SR, the Recv-Info header field that is the same as the Recv-Info header field received in the SIP INVITE request due to STN-SR except, if the ATCF supports the CS to PS SRVCC, the Recv-Info header field containing the g.3gpp.access-transfer-events info package name;

h) if the Accept header field is included in the SIP INVITE request due to STN-SR, the Accept header field that is the same as the Accept header field received in the SIP INVITE request due to STN-SR. except, if the ATCF supports the CS to PS SRVCC, the Accept header field containing the application/vnd.3gpp.access-transfer-events+xml MIME type;

i) if the ATCF supports the CS to PS SRVCC and an Accept header field of the SIP INVITE request due to STN-SR contains the application/vnd.3gpp.access-transfer-events+xml with the "et" parameter indicating ability to receive "event-type" attribute with values additional to the value "2":

A) the Accept header field containing the application/vnd.3gpp.access-transfer-events+xml MIME type with the "et" parameter indicating ability to receive "event-type" attribute with the additional values; and

B) the Recv-Info header field containing the g.3gpp.access-transfer-events info package name.

j) if the SIP INVITE request due to STN-SR contains a P-Access-Network-Info header field, a P-Access-Network-Info header field copied from the SIP INVITE request due to STN-SR.

Upon receiving a SIP 18x response or SIP 2xx response to the SIP INVITE request due to ATU-STI for PS to CS SRVCC from the SCC AS, the ATCF shall:

1) if ATCF supports CS to PS SRVCC, save the Contact header field included in the SIP response; and

NOTE 2: If the ATCF subsequently receives an initial SIP INVITE request due to STI-rSR, the ATCF will include the saved the Contact header field of the remote UE in its SIP 200 (OK) response to the initial SIP INVITE request due to STI-rSR.

2) generate and send a SIP response to the SIP INVITE request due to STN-SR populated with:

a) the Record-Route header field with a Record-Route header field that contains only the SIP URI pointing to the ATCF;

b) the same status code as the received SIP response to the SIP INVITE request due to ATU-STI for PS to CS SRVCC; and

c) if ATCF supports CS to PS SRVCC:

A) the Recv-Info header fields of the received SIP response except the Recv-Info header field containing the g.3gpp.access-transfer-events info package name;

B) if the SIP response is SIP 1xx response, the Recv-Info header field containing the g.3gpp.access-transfer-events info package name with the "et" parameter indicating ability to receive "event-type" attribute with value "1", value "3", value "4" and values, if any, indicated in the "et" parameter of the g.3gpp.access-transfer-events info package name of the Recv-Info header field of the received SIP response; and

C) if the SIP response is SIP 2xx response:

i) the Recv-Info header field containing the g.3gpp.access-transfer-events info package name;

ii) the Accept header fields of the received SIP response except the Accept header field containing the application/vnd.3gpp.access-transfer-events+xml MIME type; and

iii) the Accept header field containing the application/vnd.3gpp.access-transfer-events+xml MIME type with the "et" parameter indicating ability to receive "event-type" with value "1", value "3", value "4" and values, if any, indicated in the "et" parameter of the application/vnd.3gpp.access-transfer-events+xml MIME type of the Accept header field of the received SIP response.

Upon receiving a SIP INFO request carrying a "call-accepted" indication the ATCF shall, if not already done according to operator policy, request the ATGW to through-connect in both directions.

NOTE 3: The details of interaction between ATCF and ATGW are out of scope of this document.

12.7.2.5 Transfer of additional session

Upon receiving a SIP REFER request for transferring additional call, the ATCF shall:

1) store the additional transferred session SCC AS URI contained in the Refer-To header field and associate it with the C-MSISDN contained in the SIP INVITE requests due to STN-SR which created the dialog in which the REFER request is received;

2) replace the additional transferred session SCC AS URI contained in the Refer-To header field with the ATCF URI for anchoring additional transferred call in ATCF allocated to the ATCF extended with any SIP URI headers of the additional transferred session SCC AS URI; and

3) forward the REFER request to the MSC Server.

Upon receiving SIP INVITE request for anchoring call additionally transferred during PS to CS SRVCC in ATCF, the ATCF shall act as B2BUA and shall:

1) determine the transferable session set which are all the sessions with a speech media component:

   A) associated with C-MSISDN equal to the URI in the P-Asserted-Identity header field of the SIP INVITE request; and

   B) where during establishment of the session a Feature-Caps header field containing the g.3gpp.srvcc feature-capability indicator was received in the initial SIP request or SIP response;

NOTE 1: The transferable sessions set potentially includes recently released sessions for which the ATCF temporarily retains session state according to subclause 12.7.2.3.

2) determine the additional session being transferred which is a session in the transferable session set with dialog identifier indicated by the Target-Dialog header field of the SIP INVITE request;

3) if ATCF supports CS to PS SRVCC and if ATCF contains an SRVCC-related information (see subclause 6A.3.1) containing C-MSISDN equal to the <C-MSISDN> element of the <Setup-info> element of the value <srvcc-ext> root element of the application/vnd.3gpp.srvcc-ext+xml MIME body of the SIP INVITE request:

   A) associate the session being established with the latest SRVCC-related information (see subclause 6A.3.1) containing C-MSISDN equal to the URI in the P-Asserted-Identity header field of the SIP INVITE request; and

   B) store the value of the g.3gpp.ti media feature tag of the Contact header field of the SIP INVITE request; and

4) send a SIP INVITE request towards the home network according to 3GPP TS 24.229 [2]. The ATCF shall populate the SIP INVITE request towards the home network with:

   A) the Request-URI set to the additional transferred session SCC AS URI previously stored and associated with the C-MSISDN equal to the URI in the P-Asserted-Identity header field of the SIP INVITE request for anchoring call additionally transferred during PS to CS SRVCC in ATCF;

   B) all MIME bodies of the SIP INVITE request for anchoring call additionally transferred during PS to CS SRVCC in ATCF apart from application/sdp MIME body and apart from the application/vnd.3gpp.srvcc-ext+xml MIME body;

   C) the SDP offer containing the ATGW ports and IP addresses of the selected dialog towards the remote UE as provided by the ATGW and the media types offered in the SIP INVITE request for anchoring call additionally transferred during PS to CS SRVCC in ATCF;

   D) the Target-Dialog header field set to the Target-Dialog header field of the SIP INVITE request for anchoring call additionally transferred during PS to CS SRVCC in ATCF;

   E) the Record-Route header field containing the SIP URI of the ATCF;

   F) the Contact header field that contains the contact information received in the SIP INVITE request for anchoring call additionally transferred during PS to CS SRVCC in ATCF;

   G) the P-Asserted-Identity header field that is the same as the P-Asserted-Identity header field received in the SIP INVITE request for anchoring call additionally transferred during PS to CS SRVCC in ATCF;

   H) all header fields which are included in the SIP INVITE request for anchoring call additionally transferred during PS to CS SRVCC in ATCF and which contain option tag(s);

   I) if the Recv-Info header field is included in the SIP INVITE request for anchoring call additionally transferred during PS to CS SRVCC in ATCF, the Recv-Info header field that is the same as the Recv-Info header field received in the SIP INVITE request for anchoring call additionally transferred during PS to CS SRVCC in

ATCF except, if the ATCF supports the CS to PS SRVCC, the Recv-Info header field containing the g.3gpp.access-transfer-events info package name;

J) if the Accept header field is included in the SIP INVITE request for anchoring call additionally transferred during PS to CS SRVCC in ATCF, the Accept header field that is the same as the Accept header field received in the SIP INVITE request for anchoring call additionally transferred during PS to CS SRVCC in ATCF. except, if the ATCF supports the CS to PS SRVCC, the Accept header field containing the application/vnd.3gpp.access-transfer-events+xml MIME type;

K) if the ATCF supports the CS to PS SRVCC and an Accept header field of the SIP INVITE request for anchoring call additionally transferred during PS to CS SRVCC in ATCF contains the application/vnd.3gpp.access-transfer-events+xml with the "et" parameter indicating ability to receive "event-type" attribute with values additional to the value "2":

a) the Accept header field containing the application/vnd.3gpp.access-transfer-events+xml MIME type with the "et" parameter indicating ability to receive "event-type" attribute with the additional values; and

b) the Recv-Info header field containing the g.3gpp.access-transfer-events info package name;

L) if the ATCF is located in the visited network, and local policy requires the application of IBCF capabilities in the visited network towards the home network, select an IBCF in the visited network and add the URI of the selected IBCF to the topmost Route header field.

M) if the SIP INVITE request due to STN-SR contains a P-Access-Network-Info header field, a P-Access-Network-Info header field copied from the SIP INVITE request due to STN-SR.

When the ATCF receives any SIP 1xx response or a SIP 2xx response to the SIP INVITE request towards the home network, the ATCF shall:

1) if ATCF supports CS to PS SRVCC, save the Contact header field included in the SIP response; and

2) generate and send a SIP response to the SIP INVITE request for anchoring call additionally transferred during PS to CS SRVCC in ATCF populated with:

A) the Record-Route header field with a Record-Route header field that contains only the SIP URI pointing to the ATCF;

B) the same status code as the received SIP response to the SIP INVITE request towards the home network;

C) if ATCF supports CS to PS SRVCC:

a) the Recv-Info header fields of the received SIP response except the Recv-Info header field containing the g.3gpp.access-transfer-events info package name;

b) if the SIP response is a SIP 1xx response, the Recv-Info header field containing the g.3gpp.access-transfer-events info package name with the "et" parameter indicating ability to receive "event-type" attribute with value "1", value "3", value "4" and values, if any, indicated in the "et" parameter of the g.3gpp.access-transfer-events info package name of the Recv-Info header field of the received SIP response; and

c) if the SIP response is a SIP 2xx response:

i) the Recv-Info header field containing the g.3gpp.access-transfer-events info package name;

ii) the Accept header fields of the received SIP response except the Accept header field containing the application/vnd.3gpp.access-transfer-events+xml MIME type; and

iii) the Accept header field containing the application/vnd.3gpp.access-transfer-events+xml MIME type with the "et" parameter indicating ability to receive "event-type" with value "1", value "3", value "4" and values, if any, indicated in the "et" parameter of the application/vnd.3gpp.access-transfer-events+xml MIME type of the Accept header field of the received SIP response;

D) all MIME bodies of the received SIP response to the SIP INVITE request towards the home network apart from application/sdp MIME body; and

E) the SDP answer containing the ATGW ports and IP addresses of the selected dialog towards the MSC server as provided by the ATGW and the media types answered in the the received SIP response to the SIP INVITE request towards the home network.

Upon receiving a SIP INFO request carrying a "call-accepted" indication the ATCF shall, if not already done according to operator policy, request the ATGW to through-connect in both directions.

NOTE 2: The details of interaction between ATCF and ATGW are out of scope of this document.

12.7.2.6 Codec inquiry prior to PS to CS SRVCC access transfer

Upon receiving a SIP OPTIONS request carrying the PS-to-CS-preparation-request, the ATCF shall send a SIP 2xx response to the SIP OPTIONS request according to 3GPP TS 24.229 [2]. In the SIP 2xx response, the ATCF shall include an application/vnd.3gpp.PS-to-CS-preparation+xml specified in subclause D.6 carrying the PS-to-CS-preparation-response. In the PS-to-CS-preparation-response, the ATCF:

1) if:

   a) the session being transferred as described in subclause 12.7.2.1 is determined; or

   b) the session being transferred as described in subclause 12.7.2.1 is not determined and the session being transferred as described in subclause 12.7.2.1A is determined;

   then:

   a) shall include a <currently-possible> XML element; and

   b) shall include an <IMS-preferred-codec-list> element containing an SDP body with one audio m= line. In the m= line, in the following decreasing order of preference, the ATCF:

      - shall include RTP payload type(s) with associated RTP payload type number(s) describing media received by the ATGW in a dialog of the home leg of the session being transferred. If the home leg of the session being transferred consists of several early dialogs, the ATCF shall select one early dialog according to local policy; and

      - may include additional RTP payload type(s) with associated RTP payload type number(s), supported by the ATGW, describing the media which the ATGW is able to send to the MSC server, and selected by local policy.

NOTE 1: If the initial SDP offer of the session being transferred was provided by the remote UE, then the additional RTP payload type(s) can be derived from RTP payload type(s) which were offered in the initial SDP offer provided by the remote UE but which were not accepted by the SC UE. However, the SDP body received from the remote UE describes media which the remote UE wishes to receive while the SDP body in the <IMS-preferred-codec-list> element describes media which the ATGW (and the remote UE if no transcoding occurs) can send to the MSC server, i.e. media in the opposite direction. Therefore, the RTP payload types indicated in the SDP body received from the remote UE need to be adjusted before inclusion in the <IMS-preferred-codec-list> element.

   The ATCF shall associate the RTP payload type(s) with the RTP payload type number(s) in the <IMS-preferred-codec-list> element so that association of the RTP payload type(s) with the RTP payload type number(s) in the <IMS-preferred-codec-list> element do not conflict with association of the RTP payload type(s) with the RTP payload type number(s) describing media received by the ATGW in the selected dialog of the home leg of the session being transferred; and

NOTE 2: RTP payload type number(s) indicated in the <MSC-server-supported-codec-list> element of the PS-to-CS-preparation-request do not influence the RTP payload type number(s) indicated in the <IMS-preferred-codec-list> element.

2) if the session being transferred as described in subclause 12.7.2.1 is not determined and the session being transferred as described n subclause 12.7.2.1A is not determined, shall include a <currently-not-possible> XML element. In the <currently-not-possible> XML element, the ATCF:

   A) shall include the <state-info> element indicating the state of the session;

   B) shall include the <direction> element indicating the direction of the session;

C) shall include the <speech-state> element indicating the state of the speech media component of the session;

D) if the Feature-Caps header field provided by the SCC AS in the session includes the g.3gpp.mid-call feature-capability indicator and the Contact header field provided by the SC UE in the session includes the g.3gpp.mid-call media feature tag, shall include a <feature-tag> element with the "name" attribute set to "g.3gpp.mid-call";

E) if the Feature-Caps header field provided by the SCC AS in the session includes the g.3gpp.srvcc-alerting feature-capability indicator and the Contact header field provided by the SC UE in the session includes the g.3gpp.srvcc-alerting feature tag:

i) shall include a <feature-tag> element with the "name" attribute set to "g.3gpp.srvcc-alerting"; and

ii) if the Feature-Caps header field provided by the SCC AS in the session includes the g.3gpp.ps2cs-srvcc-orig-pre-alerting indicator and the Contact header field provided by the SC UE in the session includes the g.3gpp.ps2cs-srvcc-orig-pre-alerting media feature tag, shall include a <feature-tag> element with the "name" attribute set to "g.3gpp.ps2cs-srvcc-orig-pre-alerting";

for each session in the the transferable session set determined as specified in subclause 12.7.2.1. If the SC UE has several early dialogs created by the same SIP INVITE request, the ATCF shall include the above pieces of information for one of those early dialogs only.

12.7.3 ATCF procedures for CS to PS SRVCC

12.7.3.1 General

If the ATCF supports the CS to PS SRVCC, upon receiving SIP INFO request carrying the session transfer notification request, the ATCF shall:

1) send SIP 200 (OK) response to the SIP INFO request according to 3GPP TS 24.229 [2];

2) determine the session being transferred as the session supported by the dialog of the SIP INFO request;

3) if a SIP 2xx response to the initial SIP INVITE request has been sent in the dialog of the determined session being transferred and if the determined session being transferred includes an active speech media component:

A) if the ATGW anchors the speech media component of the determined session being transferred, continue handling the procedures in the subclause 12.7.3.2; and

B) if the ATGW does not anchor the speech media component of the determined session being transferred, continue handling the procedures in the subclause 12.7.3.3; and

4) if the determined session being transferred does not include an active speech media component, continue handling the procedures in the subclause 12.7.3.4.

12.7.3.2 Transfer of session with active speech media component anchored in ATGW

If the ATCF supports the CS to PS SRVCC, in order to transfer the determined session being transferred with the speech media component anchored in ATGW, the ATCF shall:

NOTE 1: At this point, ATCF interacts with ATGW to reserve resources and provide the information needed in the procedures below. The details of interaction between ATCF and ATGW are out of scope of this document.

1) for each registration path(s), which have the C-MSISDN equal to the C-MSISDN associated with the session of the SIP INFO request carrying the session transfer notification request, set:

A) the CS to PS SRVCC access transfer in progress flag (see subclause 6A.3.1); and

B) the home leg of the session being transferred by CS to PS SRVCC (see subclause 6A.3.1) to the dialog identifier of the home leg of the determined session being transferred; and

2) send a SIP INFO request within the dialog of the SIP INFO request carrying the session transfer notification request according to 3GPP TS 24.229 [2]. The ATCF shall populate the SIP INFO request with:

A) Info-Package header field with value g.3gpp.access-transfer-events; and

B) application/vnd.3gpp.access-transfer-events+xml MIME body associated with the info package according to IETF RFC 6086 [54] and:

a) indicating the session transfer notification response;

b) indicating that the ATCF does not require the MSC server to redirect the speech media component of the session transferred by the CS to PS SRVCC access transfer; and

c) containing the ATGW transfer details.

If receiving the SIP INFO request carrying the session transfer cancellation, the ATCF shall:

NOTE 2: the SIP INFO request carrying the session transfer cancellation is received only when CS to PS SRVCC is cancelled.

1) send SIP 200 (OK) response to the SIP INFO request according to 3GPP TS 24.229 [2]; and

2) for registration path(s), which have the C-MSISDN equal to the C-MSISDN associated with the registration path of the SIP INVITE request due to STI-rSR, reset the CS to PS SRVCC access transfer in progress flag (see subclause 6A.3.1).

NOTE 3: At this point, the ATCF interacts with the ATGW to release any resources reserved on PS serving leg. The details of interaction between the ATCF and the ATGW are out of scope of this document.

NOTE 4: If the SIP INFO request carrying the session transfer cancellation is received, remaining procedures of this subclause are not invoked.

Unless the determined session being transferred is released, upon receiving the SIP INFO request carrying session transfer preparation, the ATCF shall:

1) send SIP 200 (OK) response to the SIP INFO request according to 3GPP TS 24.229 [2].

NOTE 5: At this point, the ATCF interacts with the ATGW to start forwarding the audio media from the remote UE towards the SC UE according to the SC UE information for CS to PS SRVCC (see subclause 6A.3.1). The details of interaction between the ATCF and the ATGW are out of scope of this document.

NOTE 6: At this point, the ATCF interacts with the ATGW to start forwarding the audio media received at the IP address and port provided in the ATGW transfer details according to the ATGW information for CS to PS SRVCC (see subclause 6A.3.1) towards the remote UE. The details of interaction between the ATCF and the ATGW are out of scope of this document.

If receiving the SIP INFO request carrying the session transfer cancellation, the ATCF shall:

NOTE 7: the SIP INFO request carrying the session transfer cancellation is received only when CS to PS SRVCC is cancelled.

1) send SIP 200 (OK) response to the SIP INFO request according to 3GPP TS 24.229 [2]; and

2) for registration path(s), which have the C-MSISDN equal to the C-MSISDN associated with the registration path of the SIP INVITE request due to STI-rSR, reset the CS to PS SRVCC access transfer in progress flag (see subclause 6A.3.1).

NOTE 8: At this point, the ATCF interacts with the ATGW to start forwarding the audio media from the remote UE towards the MSC server according to the SDP negotiated by the MSC server in the CS serving leg. The ATCF also interacts with the ATGW to release any resources reserved on the PS serving leg. The details of interaction between the ATCF and the ATGW are out of scope of this document.

NOTE 9: If the SIP INFO request carrying the session transfer cancellation is received, remaining procedures of this subclause are not invoked.

Upon receiving SIP INVITE request due to STI-rSR and if:

1) the CS to PS SRVCC access transfer in progress flag (see subclause 6A.3.1) of the registration path of the SIP INVITE request due to STI-rSR is set; and

2) the home leg of the session being transferred by CS to PS SRVCC (see subclause 6A.3.1) of the registration path of the SIP INVITE request due to STI-rSR is set and the dialog of the home leg of the session being transferred by CS to PS SRVCC has not been released yet;

the ATCF shall:

1) send a SIP 200 (OK) response to the received SIP INVITE request due to STI-rSR according to 3GPP TS 24.229 [2]. The ATCF shall populate the SIP 200 (OK) response with:

   A) the Contact header field of the remote UE of the home leg of the session being transferred by CS to PS SRVCC;

   B) the Feature-Caps header fields received in the home leg of the session being transferred by CS to PS SRVCC;

   C) the Accept header fields received in the home leg of the session being transferred by CS to PS SRVCC;

   D) the Recv-Info header fields received in the home leg of the session being transferred by CS to PS SRVCC;

   E) all header fields received in the home leg of the session being transferred by CS to PS SRVCC, which contain option tag(s);

   F) the P-Asserted-Identity header field received in the home leg of the session being transferred by CS to PS SRVCC;

   G) the Privacy header fields received in the home leg of the session being transferred by CS to PS SRVCC;

   H) the Record-Route header field that contains only the SIP URI pointing to the ATCF;

   I) the SDP answer that includes the ATGW ports and the IP addresses as provided by the ATGW;

   J) the P-Charging-Vector header field as specified in 3GPP TS 24.229 [2] including the "related-icid" header field parameter containing the ICID value of the source access leg in the P-Charging-Vector header field. Additionally, if an "icid-generated-at" header field was generated for the source access leg, ATCF shall include the "related-icid-generated-at" header field parameter containing the host name or IP address included in the "icid-generated-at" header field parameter of the source access leg; and

   K) the Feature-Caps header field containing the g.3gpp.ti feature-capability indicator with value of the g.3gpp.ti media feature tag in the Contact header field received in the serving leg of the session being transferred by CS to PS SRVCC;

2) initiate a new dialog toward the SCC AS (i.e. a target access leg) by sending an initial SIP INVITE request due to ATU-STI for CS to PS SRVCC. The ATCF shall populate the SIP INVITE request with:

   A) the SDP offer containing the currently used media with the ATGW ports and IP addresses towards the remote UE as provided by the ATGW. The ATCF shall include in the SDP offer only the media of the media types offered in the received SIP INVITE request due to STI-rSR;

   B) the Request-URI containing the ATU-STI for CS to PS SRVCC (see subclause 6A.3.1) previously received from the SCC AS and associated with the registration path of the SIP INVITE request due to STI-rSR;

   C) the Target-Dialog header field containing the dialog identifier of the home leg of the session being transferred by CS to PS SRVCC (see subclause 6A.3.1) of the registration path of the SIP INVITE request due to STI-rSR;

   D) the Require header field containing the option tag "tdialog";

   E) the Contact header field that contains the contact information received in the SIP INVITE request due to STI-rSR;

   F) the Record-Route header field that includes only the ATCF SIP URI, where the ATCF wants to receive subsequent in-dialog requests from the SCC AS; and

NOTE 10: The ATCF SIP URI included in the Record-Route header field is used by the SCC AS to build a Route header field that the SCC AS will use when sending the in-dialog request toward the ATCF.

G) the P-Asserted-Identity header field set to the C-MSISDN (see subclause 6A.3.1) previously received from the SCC AS and associated with the registration path of the SIP INVITE request; and

NOTE 11: Route header field(s) included in the SIP INVITE request due to STI-rSR are not inserted in the SIP INVITE request due to ATU-STI for CS to PS SRVCC.

3) for registration path(s), which have the C-MSISDN equal to the C-MSISDN associated with the registration path of the SIP INVITE request due to STI-rSR, reset the CS to PS SRVCC access transfer in progress flag (see subclause 6A.3.1).

Upon receiving SIP INVITE request due to STI-rSR and if:

1) the CS to PS SRVCC access transfer in progress flag (see subclause 6A.3.1) of the registration path of the SIP INVITE request due to STI-rSR is set; and

2) the home leg of the session being transferred by CS to PS SRVCC (see subclause 6A.3.1) of the registration path of the SIP INVITE request due to STI-rSR is set and the dialog of the home leg of the session being transferred by CS to PS SRVCC has already been released;

the ATCF shall initiate a new dialog toward the SCC AS (i.e. a target access leg) by sending an initial SIP INVITE request due to ATU-STI for CS to PS SRVCC. The ATCF shall populate the SIP INVITE request with:

1) if the ATCF decided to anchor the media according to operator policy as specified in 3GPP TS 23.237 [9]:

A) all MIME bodies of the SIP INVITE request due to STI-rSR apart from the application/sdp MIME body; and

B) application/sdp MIME body with updated SDP offer using media parameters provided by the ATGW;

NOTE 12: ATCF interacts with ATGW to provide the needed media related information. The details of interaction between ATCF and ATGW are out of scope of this document.

2) if the ATCF decided not to anchor the media according to operator policy as specified in 3GPP TS 23.237 [9], all MIME bodies of the originating SIP INVITE request from MSC server;

3) the Request-URI containing the ATU-STI for CS to PS SRVCC (see subclause 6A.3.1) previously received from the SCC AS and associated with the registration path of the SIP INVITE request due to STI-rSR;

4) the Contact header field that contains the contact information received in the SIP INVITE request due to STI-rSR;

5) the Record-Route header field that includes only the ATCF SIP URI, where the ATCF wants to receive subsequent in-dialog requests from the SCC AS; and

NOTE 13: The ATCF SIP URI included in the Record-Route header field is used by the SCC AS to build a Route header field that the SCC AS will use when sending the in-dialog request toward the ATCF.

6) the P-Asserted-Identity header field set to the C-MSISDN (see subclause 6A.3.1) previously received from the SCC AS and associated with the registration path of the SIP INVITE request due to STI-rSR.

Upon receiving SIP 1xx response or SIP 2xx response to the SIP INVITE request due to ATU-STI for CS to PS SRVCC, the ATCF shall:

1) save the Contact header field included in the SIP response; and

2) send a SIP response to the SIP INVITE request due to STI-rSR according to 3GPP TS 24.229 [2]. The ATCF shall populate the SIP response with:

A) the same response code as the received SIP response to the SIP INVITE request due to ATU-STI for CS to PS SRVCC;

B) Record-Route header field containing the SIP URI of the ATCF; and

C) the P-Charging-Vector header field as specified in 3GPP TS 24.229 [2] including the "related-icid" header field parameter containing the ICID value of the source access leg in the P-Charging-Vector header field. Additionally, if an "icid-generated-at" header field was generated for the source access leg, ATCF shall

include the "related-icid-generated-at" header field parameter containing the host name or IP address included in the "icid-generated-at" header field parameter of the source access leg.

12.7.3.3 Transfer of session with active speech media component not anchored in ATGW

In order to transfer the determined session being transferred with the speech media component not anchored in ATGW, the ATCF shall:

NOTE 1: At this point, ATCF interacts with ATGW to reserve resources and provide the information needed in the procedures below. The details of interaction between ATCF and ATGW are out of scope of this document.

1) for each registration path(s), which have the C-MSISDN equal to the C-MSISDN associated with the session of the SIP INFO request carrying the session transfer notification request, set:

   A) the CS to PS SRVCC access transfer in progress flag (see subclause 6A.3.1); and

   B) the home leg of the session being transferred by CS to PS SRVCC (see subclause 6A.3.1) to the dialog identifier of the home leg of the determined session being transferred; and

2) send a SIP INFO request within the dialog of the SIP INFO request carrying the session transfer notification request according to 3GPP TS 24.229 [2]. The ATCF shall populate the SIP INFO request with:

   A) Info-Package header field with value g.3gpp.access-transfer-events; and

   B) application/vnd.3gpp.access-transfer-events+xml MIME body associated with the info package according to IETF RFC 6086 [54] and:

      a) indicating the session transfer notification response;

      b) indicating that the ATGW requires MSC server to redirect the speech media component of the session transferred by the CS to PS SRVCC access transfer; and

      c) containing the ATGW transfer details.

NOTE 2: The ATCF can bind the reserved resources in the ATGW with the C-MSISDN, which will be used to associate the SIP INVITE request due to media transfer from MSC server to ATGW with the session to be transferred.

Upon receiving a SIP INVITE request due to media transfer from MSC server to ATGW, the ATCF shall:

1) use the C-MSISDN in the SIP INVITE request to associate the SIP INVITE request due to media transfer from MSC server to ATGW with the session to be transferred and send a SIP 200 (OK) response to the SIP INVITE request according to 3GPP TS 24.229 [2].

NOTE 3: At this point, ATCF interacts with ATGW to establish the media bearer between MGW and ATGW. The details of interaction between ATCF and ATGW are out of scope of this document.

If receiving the SIP INFO request carrying the session transfer cancellation, the ATCF shall:

NOTE 4: the SIP INFO request carrying the session transfer cancellation is received only when CS to PS SRVCC is cancelled.

1) send SIP 200 (OK) response to the SIP INFO request according to 3GPP TS 24.229 [2]; and

2) for registration path(s), which have the C-MSISDN equal to the C-MSISDN associated with the registration path of the SIP INVITE request due to STI-rSR, reset the CS to PS SRVCC access transfer in progress flag (see subclause 6A.3.1).

NOTE 5: At this point, the ATCF interacts with the ATGW to release any resources reserved on PS serving leg. The details of interaction between the ATCF and the ATGW are out of scope of this document.

NOTE 6: If the SIP INFO request carrying the session transfer cancellation is received, remaining procedures of this subclause are not invoked.

Unless the determined session being transferred is released, upon receiving the SIP INFO request carrying session transfer preparation, the ATCF shall:

1) send SIP 200 (OK) response to the SIP INFO request according to 3GPP TS 24.229 [2].

NOTE 7: At this point, the ATCF interacts with the ATGW to start forwarding the audio media from the remote UE towards the SC UE according to the SC UE information for CS to PS SRVCC (see subclause 6A.3.1). The details of interaction between the ATCF and the ATGW are out of scope of this document.

NOTE 8: At this point, the ATCF interacts with the ATGW to start forwarding the audio media received at the IP address and port provided in the ATGW transfer details according to the ATGW information for CS to PS SRVCC (see subclause 6A.3.1) towards the remote UE. The details of interaction between the ATCF and the ATGW are out of scope of this document.

If receiving the SIP INFO request carrying the session transfer cancellation, the ATCF shall:

NOTE 9: The SIP INFO request carrying the session transfer cancellation is received only when CS to PS SRVCC is cancelled.

1) send SIP 200 (OK) response to the SIP INFO request according to 3GPP TS 24.229 [2]; and

2) for registration path(s), which have the C-MSISDN equal to the C-MSISDN associated with the registration path of the SIP INVITE request due to STI-rSR, reset the CS to PS SRVCC access transfer in progress flag (see subclause 6A.3.1).

NOTE 10: At this point, the ATCF interacts with the ATGW to start forwarding the audio media from the remote UE towards the MSC server according to the SDP negotiated by the MSC server in the CS serving leg. The ATCF also interacts with the ATGW to release any resources reserved on the PS serving leg. The details of interaction between the ATCF and the ATGW are out of scope of this document.

NOTE 11: If the SIP INFO request carrying the session transfer cancellation is received, remaining procedures of this subclause are not invoked.

Upon receiving a SIP INVITE request due to STI-rSR and if:

1) the CS to PS SRVCC access transfer in progress flag (see subclause subclause 6A.3.1) of the registration path of the SIP INVITE request due to STI-rSR is set; and

2) the home leg of the session being transferred by CS to PS SRVCC (see subclause 6A.3.1) of the registration path of the SIP INVITE request due to STI-rSR is set and the dialog of the home leg of the session being transferred by CS to PS SRVCC has not been released yet;

the ATCF shall:

NOTE 12: At this point, the ATCF interacts with the ATGW to provide information needed in the procedures below and to request the ATGW to forward the audio media of the session being transferred from the remote UE to the SC UE. The details of interaction between the ATCF and the ATGW are out of scope of this document.

1) send a SIP 200 (OK) response to the received SIP INVITE request due to STI-rSR according to 3GPP TS 24.229 [2]. The ATCF shall populate the SIP 200 (OK) response with:

   A) the Contact header field of the remote UE of the home leg of the session being transferred by CS to PS SRVCC;

   B) the Feature-Caps header fields received in the home leg of the session being transferred by CS to PS SRVCC;

   C) the Accept header fields received in the home leg of the session being transferred by CS to PS SRVCC;

   D) the Recv-Info header fields received in the home leg of the session being transferred by CS to PS SRVCC;

   E) all header fields received in the home leg of the session being transferred by CS to PS SRVCC, which contain option tag(s);

   F) the P-Asserted-Identity header field received in the home leg of the session being transferred by CS to PS SRVCC;

G) the Privacy header fields received in the home leg of the session being transferred by CS to PS SRVCC;

H) the Record-Route header field that contains only the SIP URI pointing to the ATCF;

I) the SDP answer that includes the ATGW ports and the IP addresses as provided by the ATGW;

J) the P-Charging-Vector header field as specified in 3GPP TS 24.229 [2] including the "related-icid" header field parameter containing the ICID value of the source access leg in the P-Charging-Vector header field. Additionally, if an "icid-generated-at" header field was generated for the source access leg, ATCF shall include the "related-icid-generated-at" header field parameter containing the host name or IP address included in the "icid-generated-at" header field parameter of the source access leg; and

K) the Feature-Caps header field containing the g.3gpp.ti feature-capability indicator with value of the g.3gpp.ti media feature tag in the Contact header field received in the serving leg of the session being transferred by CS to PS SRVCC;

NOTE 13: At this point the ATCF requests the ATGW to forward the audio media from the SC UE to the remote UE. The details of interaction between the ATCF and the ATGW are out of scope of this document.

2) initiate a new dialog toward the SCC AS (i.e. a target access leg) by sending an initial SIP INVITE request due to ATU-STI for CS to PS SRVCC. The ATCF shall populate the SIP INVITE request with:

A) the SDP offer containing the media with the ATGW ports and IP addresses towards the remote UE as provided by the ATGW. The ATCF shall include in the SDP offer only the media of the media types offered in the received SIP INVITE request due to STI-rSR;

B) the Request-URI containing the ATU-STI for CS to PS SRVCC (see subclause 6A.3.1) previously received from the SCC AS and associated with the registration path of the SIP INVITE request due to STI-rSR;

C) the Target-Dialog header field containing the dialog identifier of the home leg of the session being transferred by CS to PS SRVCC (see subclause 6A.3.1) of the registration path of the SIP INVITE request due to STI-rSR;

D) the Require header field containing the option tag "tdialog";

E) the Contact header field that contains the contact information received in the SIP INVITE request due to STI-rSR;

F) the Record-Route header field that includes only the ATCF SIP URI, where the ATCF wants to receive subsequent in-dialog requests from the SCC AS; and

NOTE 14: The ATCF SIP URI included in the Record-Route header field is used by the SCC AS to build a Route header field that the SCC AS will use when sending the in-dialog request toward the ATCF.

G) the P-Asserted-Identity header field set to the C-MSISDN (see subclause 6A.3.1) previously received from the SCC AS and associated with the registration path of the SIP INVITE request; and

3) for registration path(s), which have the C-MSISDN equal to the C-MSISDN associated with the registration path of the SIP INVITE request due to STI-rSR, reset the CS to PS SRVCC access transfer in progress flag (see subclause 6A.3.1).

NOTE 15: Upon receiving a SIP 2xx response to the SIP INVITE request due to ATU-STI for CS to PS SRVCC, the ATCF requests the ATGW to update forwarding of the audio media from the SC UE to the remote UE. The details of interaction between the ATCF and the ATGW are out of scope of this document.

Upon receiving a SIP 2xx response to the SIP INVITE request due to ATU-STI for CS to PS SRVCC, the ATCF shall:

1) send a SIP BYE request to terminate the dialog between MSC Server and ATCF, following the procedures specified in 3GPP TS 24.229 [2].

Upon receiving SIP INVITE request due to STI-rSR and if:

1) the CS to PS SRVCC access transfer in progress flag (see subclause 6A.3.1) of the registration path of the SIP INVITE request due to STI-rSR is set; and

2) the home leg of the session being transferred by CS to PS SRVCC (see subclause 6A.3.1) of the registration path of the SIP INVITE request due to STI-rSR is set and the dialog of the home leg of the session being transferred by CS to PS SRVCC has already been released;

the ATCF shall initiate a new dialog toward the SCC AS (i.e. a target access leg) by sending an initial SIP INVITE request due to ATU-STI for CS to PS SRVCC. The ATCF shall populate the SIP INVITE request with:

1) if the ATCF decided to anchor the media according to operator policy as specified in 3GPP TS 23.237 [9]:

   A) all MIME bodies of the SIP INVITE request due to STI-rSR apart from the application/sdp MIME body; and

   B) application/sdp MIME body with updated SDP offer using media parameters provided by the ATGW;

NOTE 16: ATCF interacts with ATGW to provide the needed media related information. The details of interaction between ATCF and ATGW are out of scope of this document.

2) if the ATCF decided not to anchor the media according to operator policy as specified in 3GPP TS 23.237 [9], all MIME bodies of the originating SIP INVITE request from MSC server;

3) the Request-URI containing the ATU-STI for CS to PS SRVCC (see subclause 6A.3.1) previously received from the SCC AS and associated with the registration path of the SIP INVITE request due to STI-rSR;

4) the Contact header field that contains the contact information received in the SIP INVITE request due to STI-rSR;

5) the Record-Route header field that includes only the ATCF SIP URI, where the ATCF wants to receive subsequent in-dialog requests from the SCC AS; and

NOTE 17: The ATCF SIP URI included in the Record-Route header field is used by the SCC AS to build a Route header field that the SCC AS will use when sending the in-dialog request toward the ATCF.

6) the P-Asserted-Identity header field set to the C-MSISDN (see subclause 6A.3.1) previously received from the SCC AS and associated with the registration path of the SIP INVITE request due to STI-rSR.

Upon receiving a SIP 1xx response or a SIP 2xx response to the SIP INVITE request due to ATU-STI for CS to PS SRVCC, the ATCF shall:

1) save the Contact header field included in the SIP response; and

2) send a SIP response to the SIP INVITE request due to STI-rSR according to 3GPP TS 24.229 [2]. The ATCF shall populate the SIP response with:

   A) the same response code as the received SIP response to the SIP INVITE request due to ATU-STI for CS to PS SRVCC;

   B) Record-Route header field containing the SIP URI of the ATCF; and

   C) the P-Charging-Vector header field as specified in 3GPP TS 24.229 [2] including the "related-icid" header field parameter containing the ICID value of the source access leg in the P-Charging-Vector header field. Additionally, if an "icid-generated-at" header field was generated for the source access leg, ATCF shall include the "related-icid-generated-at" header field parameter containing the host name or IP address included in the "icid-generated-at" header field parameter of the source access leg.

12.7.3.4 Transfer when only held session or session in alerting phase exist

If the ATCF supports the CS to PS SRVCC, in order to transfer the held session or the session in alerting phase, the ATCF shall:

1) for each registration path(s), which have the C-MSISDN equal to the C-MSISDN associated with the session of the SIP INFO request carrying the session transfer notification request:

   A) set the CS to PS SRVCC access transfer in progress flag (see subclause 6A.3.1); and

   B) reset the home leg of the session being transferred by CS to PS SRVCC (see subclause 6A.3.1); and

2) send a SIP INFO request within the dialog of the SIP INFO request carrying the session transfer notification request according to 3GPP TS 24.229 [2]. The ATCF shall populate the SIP INFO request with:

   A) Info-Package header field with value g.3gpp.access-transfer-events; and

   B) application/vnd.3gpp.access-transfer-events+xml MIME body associated with the info package according to IETF RFC 6086 [54]:

      a) indicating the session transfer notification response;

      b) indicating that the ATCF does not require the MSC server to redirect the speech media component of the session transferred by the CS to PS SRVCC access transfer; and

      c) containing the ATGW transfer details indicating that the *ATGW Transfer details content* field is not included.

If receiving the SIP INFO request carrying the session transfer cancellation, the ATCF shall:

NOTE 1: the SIP INFO request carrying the session transfer cancellation is received only when CS to PS SRVCC is cancelled.

1) send SIP 200 (OK) response to the SIP INFO request according to 3GPP TS 24.229 [2]; and

2) for registration path(s), which have the C-MSISDN equal to the C-MSISDN associated with the registration path of the SIP INVITE request due to STI-rSR, reset the CS to PS SRVCC access transfer in progress flag (see subclause 6A.3.1).

NOTE 2: If the SIP INFO request carrying the session transfer cancellation is received, remaining procedures of this subclause are not invoked.

Unless the determined session being transferred is released, upon receiving the SIP INFO request carrying session transfer preparation, the ATCF shall send SIP 200 (OK) response to the SIP INFO request according to 3GPP TS 24.229 [2].

If receiving the SIP INFO request carrying the session transfer cancellation, the ATCF shall:

NOTE 3: the SIP INFO request carrying the session transfer cancellation is received only when CS to PS SRVCC is cancelled.

1) send SIP 200 (OK) response to the SIP INFO request according to 3GPP TS 24.229 [2]; and

2) for registration path(s), which have the C-MSISDN equal to the C-MSISDN associated with the registration path of the SIP INVITE request due to STI-rSR, reset the CS to PS SRVCC access transfer in progress flag (see subclause 6A.3.1).

NOTE 4: If the SIP INFO request carrying the session transfer cancellation is received, remaining procedures of this subclause are not invoked.

Upon receiving SIP INVITE request due to STI-rSR and if:

1) the CS to PS SRVCC access transfer in progress flag (see subclause 6A.3.1) of the registration path of the SIP INVITE request due to STI-rSR is set; and

2) the home leg of the session being transferred by CS to PS SRVCC (see subclause 6A.3.1) of the registration path of the SIP INVITE request due to STI-rSR is not set;

the ATCF shall initiate a new dialog toward the SCC AS (i.e. a target access leg) by sending an initial SIP INVITE request due to ATU-STI for CS to PS SRVCC. The ATCF shall populate the SIP INVITE request with:

1) if the ATCF decided to anchor the media according to operator policy as specified in 3GPP TS 23.237 [9]:

   A) all MIME bodies of the SIP INVITE request due to STI-rSR apart from the application/sdp MIME body; and

   B) application/sdp MIME body with updated SDP offer using media parameters provided by the ATGW;

NOTE 5: ATCF interacts with ATGW to provide the needed media related information. The details of interaction between ATCF and ATGW are out of scope of this document.

2) if the ATCF decided not to anchor the media according to operator policy as specified in 3GPP TS 23.237 [9], all MIME bodies of the originating SIP INVITE request from MSC server;

3) the Request-URI containing the ATU-STI for CS to PS SRVCC (see subclause 6A.3.1) previously received from the SCC AS and associated with the registration path of the SIP INVITE request due to STI-rSR;

4) the Contact header field that contains the contact information received in the SIP INVITE request due to STI-rSR;

5) the Record-Route header field that includes only the ATCF SIP URI, where the ATCF wants to receive subsequent in-dialog requests from the SCC AS; and

NOTE 6: The ATCF SIP URI included in the Record-Route header field is used by the SCC AS to build a Route header field that the SCC AS will use when sending the in-dialog request toward the ATCF.

6) the P-Asserted-Identity header field set to the C-MSISDN (see subclause 6A.3.1) previously received from the SCC AS and associated with the registration path of the SIP INVITE request due to STI-rSR.

Upon receiving a SIP 1xx response or SIP 2xx response to the SIP INVITE request due to ATU-STI for CS to PS SRVCC, the ATCF shall:

1) save the Contact header field included in the SIP response; and

2) send a SIP response to the SIP INVITE request due to STI-rSR according to 3GPP TS 24.229 [2]. The ATCF shall populate the SIP response with:

   A) the same response code as the received SIP response to the SIP INVITE request due to ATU-STI for CS to PS SRVCC;

   B) Record-Route header field containing the SIP URI of the ATCF; and

   C) the P-Charging-Vector header field as specified in 3GPP TS 24.229 [2] including the "related-icid" header field parameter containing the ICID value of the source access leg in the P-Charging-Vector header field. Additionally, if an "icid-generated-at" header field was generated for the source access leg, ATCF shall include the "related-icid-generated-at" header field parameter containing the host name or IP address included in the "icid-generated-at" header field parameter of the source access leg.

13 Roles for media adding/deleting for access transfer

13.1 Introduction

This clause specifies the procedures for adding or deleting media to an existing multimedia session. Procedures are specified for the SC UE and the SCC AS.

13.2 SC UE

13.2.1 Adding or removing media through Gm

If the SC UE wants to add or remove media components to a session that was previously established using Gm reference point, the SC UE shall follow the procedures defined in 3GPP TS 24.229 [2] for adding/removing PS media.

If the SC UE wants to transfer media components from the source access leg to an existing target access leg (i.e. the access legs were previously established due to the partial session transfer) using Gm reference point, the SC UE shall:

1. add the media components to the target access leg; and

2. remove those media components from the source access leg,

by using procedures defined in 3GPP TS 24.229 [2] for adding/removing PS media.

If SC using ICS is enabled then if the SC UE wants to add or remove CS media components to a session, the SC UE shall follow the procedures defined in 3GPP TS 24.292 [4].

If the SC UE receives a SIP re-INVITE request or a SIP UPDATE request from the remote UE to add or remove media components to a session that was previously established using Gm, the SC UE shall:

- follow the procedures defined in 3GPP TS 24.229 [2] for adding or removing PS media; and
- if SC using ICS is enabled, follow the procedures defined in 3GPP TS 24.292 [4] for adding or removing CS media to the session.

13.2.2 Adding Gm control to existing CS session

The SC UE shall add Gm control to an existing CS session only when SC using ICS is enabled and when there is a single full-duplex session with speech media component over CS. If there is more than one full-duplex session with speech media component, the SC UE shall release all the ongoing sessions that are not currently active before attempting the procedures described in this section.

If SC using ICS is enabled and the SC UE wants to add Gm control to an existing CS session that was established without Gm, after registering with the IM CN subsystem, the SC UE shall send an initial SIP INVITE request over the PS access in accordance with 3GPP TS 24.229 [2]. The SC UE shall populate the SIP INVITE request as follows:

- set the Request-URI to the static STI;
- set the SDP payload, in accordance with the procedures defined in 3GPP TS 24.292 [4], proposing a speech media component over a circuit switched bearer. The SC UE can optionally include additional PS media to the SDP in accordance to the procedures defined in 3GPP TS 24.229 [2]; and
- signalling elements described in subclause 6A.2.2.2.

Upon receiving a SIP 200 (OK) response, the SC UE shall treat the ongoing CS call as established using Gm and shall follow the "ICS UE using Gm" procedures defined in 3GPP TS 24.292 [4] for controlling the CS call.

If SC using ICS is enabled and the SC UE receives a new SIP INVITE request containing a speech media component over a circuit-switched bearer in the SDP and the SCC AS PSI DN matches the B-party number of the ongoing CS call that was established without Gm, the SC UE shall:

- respond to the SIP INVITE request in accordance with the procedures defined in 3GPP TS 24.292 [4]; and
- treat the ongoing CS call as established using Gm and shall follow the "ICS UE using Gm" procedures defined in 3GPP TS 24.292 [4] for controlling the CS call.

13.3 SCC AS

13.3.1 Adding or removing media through Gm

If the SCC AS receives a SIP re-INVITE request or a SIP UPDATE request from the SC UE, in which already existing media components of the session are transferred from a source access leg to an already existing target access leg (i.e. the target access leg was already established due to partial session transfer), the SCC AS shall update the remote UE using the session transfer procedures defined in subclause 10.3.2.

NOTE: The SC UE indicates that media is switched from the source access leg to the target access leg by using the procedures defined in 3GPP TS 24.229 [2] for adding / removing PS media, i.e. the related connection and port information of the transferred media component within the SDP is changed from the source access leg to the target access leg.

If the SCC AS receives a SIP re-INVITE request or a SIP UPDATE request from the SC UE or remote UE to add/remove new media components, to an existing access leg of the session established using Gm, the SCC AS shall follow the procedures defined in 3GPP TS 24.229 [2] for adding or removing PS media and shall follow the procedures defined in 3GPP TS 24.292 [4] for adding or removing CS media to the session. If both a PS access leg and a CS access leg exist, the SCC AS shall offer on the PS access leg any new media components offered by the remote UE. If no PS access leg exists and a CS access leg exists, the SCC AS shall reject any new media components offered by the remote UE by setting the port to zero in an SDP answer sent on the remote leg. If a CS access leg exists, the SCC AS shall not offer any new media components offered by the remote UE via a CS access leg.

13.3.2 Adding Gm control to existing CS session

If the SCC AS receives a SIP INVITE request containing the static STI in the Request-URI or receives a SIP re-INVITE request containing one or more already existing media lines in the SDP for audio indicating a CS bearer as defined in IETF RFC 7195 [73] the SCC AS shall:

1) determine if this SIP INVITE request or SIP re-INVITE request is for an ongoing call by determining if the received contents of the Contact header field of the SIP request is bound to an ongoing CS call session identifier;

2) if the SC UE has an ongoing CS call:

   - respond to the SIP INVITE request or SIP re-INVITE request in accordance with the procedures defined in 3GPP TS 24.292 [4]. The SCC AS shall populate the SIP INVITE request with signalling elements described in subclause 6A.4.3; and

   - treat the ongoing CS call as established using Gm and shall follow the "SCC AS for service control over Gm" procedures defined in 3GPP TS 24.292 [4] for controlling the CS call; and

3) if the SIP INVITE request or SIP re-INVITE request contains additional PS media, the SCC AS shall send a SIP re-INVITE request towards the remote UE, including the newly added PS media, in accordance with the procedures defined in 3GPP TS 24.229 [2].

NOTE: When using the ICS controlled CS bearer, only one audio call can be active at a time. Nevertheless, several calls can be held in parallel. If the user decides to switch to another (previously held) call, the ICS controlled CS bearer is re-used for this call. Therefore no specific procedures for handling of held calls in the case of ICS controlled CS bearer are needed.

The SCC AS shall add Gm control to an existing CS session only when there is a single full-session with speech media component over CS. If the SCC AS wants to add Gm control to an existing CS session that was established without Gm, the SCC AS shall send a new SIP INVITE request over the PS access in accordance with 3GPP TS 24.229 [2]. The SCC AS shall populate the SIP INVITE request as follows:

- set the Request-URI to the public user identity of the UE; and

- set the SDP payload, in accordance with the procedures defined in 3GPP TS 24.292 [4], proposing speech media component over a circuit switched bearer.

Upon receiving a SIP 200 (OK) response, the SCC AS shall treat the ongoing CS call as established using Gm and shall follow the "SCC AS for service control over Gm" procedures defined in 3GPP TS 24.292 [4] for controlling the CS call.

14 Void

15 Void

16 Void

17 Void

18 Void

19 Void

20 Service continuity and MMTEL interactions

20.1 Roles for access transfer and supplementary services interaction

20.1.1 Introduction

This subclause describes the SCC AS and SC UE procedures for interaction of access transfer when execution of supplementary service as described in 3GPP TS 22.173 [24].

20.1.2 Originating Identification Presentation (OIP)

There are no specific procedures for the SC UE and the SCC AS for interaction of access transfer and OIP besides the procedures described in 3GPP TS 24.607 [25].

20.1.3 Originating Identification Restriction (OIR)

There are no specific procedures for the SC UE and the SCC AS for interaction of access transfer and OIR besides the procedures described in 3GPP TS 24.607 [25].

20.1.4 Terminating Identification Presentation (TIP)

There are no specific procedures for the SC UE and the SCC AS for interaction of access transfer and TIP besides the procedures described in 3GPP TS 24.608 [26].

20.1.5 Terminating Identification Restriction (TIR)

There are no specific procedures for the SC UE and the SCC AS for interaction of access transfer and TIR besides the procedures described in 3GPP TS 24.608 [26].

20.1.6 Communication Diversion (CDIV)

Upon receiving an incoming session split across multiple access legs, if the SC UE desires to invoke the CDIV, it may choose any of the PS access legs to invoke the call deflection supplementary service following the procedures described in 3GPP TS 24.604 [27] or the CS access leg to invoke the call deflection supplementary service following the procedures described in 3GPP TS 24.072 [42].

NOTE: Communication Forwarding unconditional, Communication forwarding on no reply, Communication Forwarding on Busy, Communication Forwarding Not Logged-in and Communication Diversion Notification supplementary services are invoked by the CDIV AS as described in 3GPP TS 24.604 [27] independent on access type.

When the SCC AS which is dividing an IMS session into multiple access legs, receives a CDIV request from the SC UE on any access leg, the SCC AS shall terminate any other access legs and invoke the CDIV for that access leg according to the procedures described in 3GPP TS 24.604 [27].

20.1.7 Communication Hold (HOLD)

When the SC UE which is dividing an IMS session through multiple access legs, desires to invoke HOLD on one or more media components, it shall proceed according to the procedures described in 3GPP TS 24.610 [28] for PS access legs, 3GPP TS 24.083 [43] for a CS access leg not controlled by the I1 interface or 3GPP TS 24.294 [44] for a CS access leg controlled by the I1 interface which contains the affected media components.

When the SCC AS which dividing an IMS session into multiple access legs, receives a HOLD request from the SC UE or remote end on any access leg, it shall proceed according to the procedures described in 3GPP TS 24.610 [28] for that access leg.

20.1.8 Communication Barring (CB)

There are no specific procedures for the SC UE and the SCC AS for interaction of access transfer and CB besides the procedures described in 3GPP TS 24.611 [29].

20.1.9 Message Waiting Indication (MWI)

There are no specific procedures for the SC UE and the SCC AS for interaction of access transfer and MWI besides the procedures described in 3GPP TS 24.606 [30].

20.1.10 Conference (CONF)

When the SC UE has multiple access legs and if it wants to send any CONF related requests such as SIP SUBSCRIBE request or SIP REFER request, the SC UE may send the request on the PS access leg as describes in 3GPP TS 24.605 [31] or use the procedures described in 3GPP TS 24.294 [44] for a CS access leg controlled by the I1 interface. For a CS access leg without I1 interface control the procedures in 3GPP TS 24.084 [47] shall be used to create and add participants to a conference.

When the SC UE has multiple access legs and if it receives a request on one of the access legs for CONF service to replace an existing session, the SC UE shall:

- if each access les is PS access leg, follow procedures specified in 3GPP TS 24.605 [31] to establish a new session to the conference focus;
- if the CS access leg is not controlled by the I1 interface follow the procedures in 3GPP TS 24.008 [8] for releasing and establishing a new call towards the conference focus; and
- if the CS access leg is controlled by the I1 interface follow the procedures in 3GPP TS 24.294 [44] for establish a new session towards the conference focus.

When the SC UE has multiple access legs and if it receives a request on one the access legs for CONF service to replace an existing session outside the dialog, the SC UE shall follow procedures specified in 3GPP TS 24.605 [31] to establish a new session to the conference focus.

When the SC UE has multiple access legs and if the remote UE sends a request for the CONF service to replace an existing session within the same dialog, the SCC AS shall deliver the request for CONF service on the Gm controlled any of access legs or over the I1 interface if I1 interface control is used or to the CS leg if only a CS leg exists, to the SC UE.

Upon receipt of a CC BuildMPTY request from the SC UE and if the MSC server:

1) is enhanced for ICS and supports CS to PS SRVCC; and
2) the latest SRVCC information received for the registration path of the SC UE contains the ATCF management URI and the C-MSISDN;

then when sending the SIP INVITE request due to receipt of a CC BuildMPTY request from the SC UE as specified in 3GPP TS 29.292 [18] and 3GPP TS 24.292 [4], then the MSC server shall additionally populate the SIP INVITE request with:

1) topmost Route header field with the ATCF management URI and lr URI parameter;

2) the Accept header field containing application/vnd.3gpp.access-transfer-events+xml MIME type with the "et" parameter indicating ability to receive "event-type" attribute with the value "2";

3) the Recv-Info header field containing the g.3gpp.access-transfer-events info package name;

4) application/vnd.3gpp.srvcc-ext+xml MIME body with the <srvcc-ext> root element containing the <Setup-info> element containing the CS to PS SRVCC information bound to the registration path (see subclause 6A.3.1) and indicating the "initiator" role of the MSC server in the session set up; and

5) the g.3gpp.ti media feature tag with value as described in subclause C.12 indicating value of an invited participant in the Contact header field.

If the MSC server:

- is enhanced for ICS and supports CS to PS SRVCC; and

- the latest SRVCC information received for the registration path of the SC UE contains the ATCF management URI and the C-MSISDN;

then upon receipt of a DISCONNECT message from a SC UE with an established conference, with a transaction identifier corresponding to a specific remote party, if:

- not all conference participants are removed when the timer expires as specified in 3GPP TS 29.292 [18]; and

- the g.3gpp.ti media feature tag last provided in the Contact header field towards SCC AS had the value of the transaction identifier corresponding to a specific remote party;

the MSC server shall send UPDATE request within dialog of the established conference. The MSC server shall populate the SIP UPDATE request with the g.3gpp.ti media feature tag with value as described in subclause C.12 indicating value of an another participant of the established conference in the Contact header field.

If the ATCF supports CS to PS SRVCC then upon receipt of a UPDATE request received in dialog established as described in subclause 8.4.2.3 or a described in subclause 12.7.2, if:

- the Contact received from SCC AS indicated isfocus media feature tag; and

- if the UPDATE request contains a Contact header field with the g.3gpp.ti media feature tag;

the ATCF shall store the value of the g.3gpp.ti media feature tag of the Contact header field of the SIP UPDATE request.

If the SCC AS supports CS to PS SRVCC then upon receipt of a UPDATE request received in dialog established as described in subclause 7.3.2 and if:

- the Contact sent towards ATCF indicated isfocus media feature tag; and

- the UPDATE request contains a Contact header field with the g.3gpp.ti media feature tag;

the SCC AS shall store the value of the g.3gpp.ti media feature tag of the Contact header field of the SIP UPDATE request.

20.1.11 Explicit Communication Transfer (ECT)

When the SC UE has multiple access legs and if it acts as the transferor UE, the SC UE may send the request for ECT service on any of the PS legs as specified in 3GPP TS 24.629 [32], or on the CS access leg not controlled by the I1 interface follow the procedures in 3GPP TS 24.091 [46] and on a CS access leg controlled by the I1 interface follow the procedures in 3GPP TS 24.294 [44].

When the SC UE has multiple access legs and if it acts as the transferee UE, the SCC AS may deliver the request for ECT service on any of the access legs.

NOTE: Delivering of the request towards the CS access leg may be controlled by operator policy.

When the SC UE has multiple access legs and if it receives an ECT request on one of the access legs, the SC UE shall follow the procedures specified in 3GPP TS 24.629 [32] to establish a new session to the Transfer Target.

20.1.12 Advice of Charge (AOC)

When the AOC service as specified in 3GPP TS 24.647 [33] is active and if the SC UE has multiple access legs, the SCC AS may deliver charging information during the communication to the SC UE over any of the access legs which accept application/vnd.etsi.aoc+xml MIME type.

20.1.13 Closed User Groups (CUG)

There are no specific procedures for the SC UE and the SCC AS for interaction of access transfer and CUG besides the procedures described in 3GPP TS 24.654 [34].

20.1.14 Three-Party (3PTY)

The 3PTY service is considered as a special case of CONF service in 3GPP TS 24.605 [31] and the interaction with session transfer is the same as that specified in subclause 20.1.10 for CONF service.

20.1.15 Flexible Alerting (FA)

There are no specific procedures for the SC UE and the SCC AS for interaction of access transfer and FA besides the procedures described in 3GPP TS 24.239 [35].

20.1.16 Communication Waiting (CW)

Upon receiving an incoming session split across multiple access legs if the SC UE desires to invoke the CW, it may choose any of the access legs to invoke the CW service following to the procedures defined in 3GPP TS 24.615 [36].

When the SCC AS which is dividing an IMS session into multiple access legs, receives a CW request from the SC UE on any access leg, the SCC AS shall invoke the CW service following the procedures defined in 3GPP TS 24.615 [36].

20.1.17 Completion of Communications to Busy Subscriber (CCBS)/Completion of Communications by No Reply (CCNR)

There are no specific procedures for the SC UE and the SCC AS for interaction of access transfer and CCBS/CCNR besides the procedures described in 3GPP TS 24.642 [37].

20.1.18 Customized Alerting Tones (CAT)

There are no specific procedures for the SC UE and the SCC AS for CAT besides the procedures described in 3GPP TS 24.182 [38].

When the terminating network is providing CAT, the PS to CS SRVCC for calls in alerting phase is only supported if a SIP 180 (Ringing) response is sent. This is not required as part of the CAT service.

20.1.19 Malicious Communication IDentification (MCID)

When invoking the MCID service in temporary subscription mode and there are multiple active access legs for the session, the SC UE may send the SIP re-INVITE request for invoking MCID service as defined in 3GPP TS 24.616 [39] on any of the access legs.

20.1.20 Reverse Charging

The interaction of the Reverse Charging service according to 3GPP TS 24.647 [33] with access transfer is not specified in this version of the specification.

20.1.21 Personal Network Management (PNM)

The interaction of the PNM service according to 3GPP TS 24.259 [40] with access transfer is not specified in this version of the specification.

20.1.22 Customized Ringing Signal (CRS)

The interaction of the CRS service according to 3GPP TS 24.183 [41] with access transfer is not specified in this version of the specification.

20.2 Void

21 Void

22 Roles for detection of remote leg information

22.1 Introduction

This clause specifies the procedures for discovery of remote leg information of a dialog anchored in SCC AS.

Procedures are specified for the MSC server enhanced for ICS and SCC AS.

The procedures can be used by MSC server enhanced for ICS only in a dialog where Feature-Cap header field with g.3gpp.remote-leg-info feature-capability indicator was received.

22.2 MSC server enhanced for ICS

22.2.1 Distinction of requests

The MSC server enhanced for ICS needs to distinguish the following SIP requests:

1) SIP INFO request:

   A) with Info-Package header field with value g.3gpp.state-and-event; and

   B) with application/vnd.3gpp.state-and-event-info+xml MIME body associated with the info package according to IETF RFC 6086 [54] as described in subclause D.2.3, containing a <state-and-event-info> root element, the <state-and-event-info> element containing an <anyExt> element and the <anyExt> element containing a <remoteLegInfoResponse> element with remote leg information.

   In the procedures below, such requests are known as "SIP INFO requests carrying the remote leg information".

22.2.2 Discover remote leg information

In order to discover remote leg information of a dialog where Feature-Cap header field with g.3gpp.remote-leg-info feature-capability indicator was received, the MSC server enhanced for ICS shall send a SIP INFO request according to 3GPP TS 24.229 [2] within the dialog. The MSC server enhanced for ICS shall populate the SIP INFO request with:

a) Info-Package header field with value g.3gpp.state-and-event; and

b) application/vnd.3gpp.state-and-event-info+xml MIME body associated with the info package according to IETF RFC 6086 [54] according to subclause D.2.3, containing a <state-and-event-info> root element, the <state-and-event-info> element containing an <anyExt> element, the <anyExt> element containing a <remoteLegInfoRequest> element and the <remoteLegInfoRequest> element indicating the requested remote leg information.

Upon receiving a SIP INFO request carrying the remote leg information in the dialog dialog where Feature-Cap header field with g.3gpp.remote-leg-info feature-capability indicator was received, the MSC server enhanced for ICS shall associate the information received in the SIP INFO request carrying the remote leg information with the dialog of the SIP INFO request carrying the remote leg information.

NOTE: The MSC server enhanced for ICS uses the remote leg information when performing supplementary services on the session transferred by PS to CS access transfer.

The MSC server enhanced for ICS shall ignore any unknown element and any unknown attribute included in the <remoteLegInfoResponse> element of the SIP INFO request carrying the request for the remote leg information.

22.3 SCC AS

22.3.1 Distinction of requests

The SCC AS needs to distinguish the following SIP requests:

1) SIP INFO request:

   A) with Info-Package header field with value g.3gpp.state-and-event; and

   B) with application/vnd.3gpp.state-and-event-info+xml MIME body associated with the info package according to IETF RFC 6086 [54] as described in subclause D.2.3, containing a <state-and-event-info> root element, the <state-and-event-info> element containing an <anyExt> element and the <anyExt> element containing a <remoteLegInfoRequest> element.

   In the procedures below, such requests are known as "SIP INFO requests carrying the request for the remote leg information".

22.3.2 Providing remote leg information

Upon receiving a SIP INFO request carrying the request for the remote leg information in the dialog where SCC AS included the Feature-Cap header field with g.3gpp.remote-leg-info feature-capability indicator, the SCC AS shall send shall send a SIP INFO request according to 3GPP TS 24.229 [2] within the dialog. The SCC AS shall populate the SIP INFO request with:

a) Info-Package header field with value g.3gpp.state-and-event; and

b) application/vnd.3gpp.state-and-event-info+xml MIME body associated with the info package according to IETF RFC 6086 [54] according to subclause D.2.3, containing a <state-and-event-info> root element, the <state-and-event-info> element containing an <anyExt> element and the <anyExt> element containing a <remoteLegInfoResponse> element with remote leg information requested in the SIP INFO request carrying the request for the remote leg information.

The SCC AS shall ignore any unknown element and any unknown attribute included in the <remoteLegInfoRequest> element of the SIP INFO request carrying the request for the remote leg information.

Annex A (informative): Example signalling flows

A.1 Scope of signalling flows

This annex gives examples of signalling flows for Service Continuity based on the Session Initiation Protocol (SIP) and SIP Events.

These signalling flows provide detailed signalling flows, which expand on the overview information flows provided in 3GPP TS 23.237 [9].

A.2 Introduction

A.2.1 General

The signalling flows provided in this annex follow the methodology developed in 3GPP TS 24.228 [3].

A.2.2 Key required to interpret signalling flows

The key to interpret signalling flows specified in 3GPP TS 24.228 [3] subclauses 4.1 and 4.2 applies with the additions specified below:

- tel:+1-237-555-1111 represents the public user identity of SC UE A.
- tel:+1-237-555-2222 represents the public user identity of UE B.
- sip:sccas1.home1.net represents the Internet host of SCC AS.
- sip:pstops.transfer@sccas1.home1.net represents the PS to PS STI.
- sip:domain.xfer@sccas.home1.net representing the static STI.
- #### represents the contents in the headers are not shown and does not impact the implementation of the flows.
- sip:home1.net represents the home network domain name of the SC UE A.
- sip:additional.session.xfer@sccas.home1.net represents the additional transferred session SCC AS URI.
- sip:additional.session.xfer.pscssrvcc@sccas.home1.net represents the additional transferred session SCC AS URI for PS to CS SRVCC.
- tel:+1-237-888-9999 represents the STN-SR owned by ATCF.
- sip:reg@atcf.visited2.net;lr represents the ATCF URI for originating requests.
- sip:termsdgfdfwe@atcf.visited2.net represents the ATCF URI for terminating requests.
- sip:atcf.visited2.net represents the ATCF management URI.
- sip:sti-sr@atcf.visited2.net represents the STI-rSR owned by ATCF.
- sip:AUT-STI1@sccas.home1.net represents the ATU-STI for PS to CS SRVCC.
- sip:cs2ps@sccas1.home1.net represents the ATU-STI for CS to PS SRVCC.
- sip:termpdfjkghlj@msc123.visited2.net;lr represents the MSC URI for terminating requests.
- sip:msc123.visited2.net represents the MSC management URI.
- sip:eatf1.visit1.net represents the EATF URI.
- sip:ecscf.visit1.net represents the E-CSCF URI.

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

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

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

In order to differentiate between messages for SIP and media, the notation in figure A.2-1 is used.

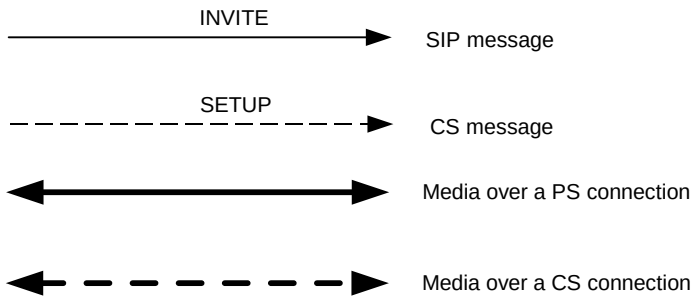


Figure A.2-1: Signalling flow notation

A.3 Signalling flows for registration

A.3.1 Introduction

When using CS access for media and to make use of the ISC procedures, the SC UE is registered in IM CN subsystem and the signalling flows are defined in 3GPP TS 24.292 [4] clause A.2.

When initiating a CS call, the SC UE can be registered in the CS domain as defined in 3GPP TS 24.008 [8].

Whenever the UE acquires IP connectivity via an IP-CAN, the signalling flows for registration in the IM CN subsystem are defined in 3GPP TS 24.228 [3].

A.3.2 Signalling flows for multiple registration for service continuity

The signalling flows shown in figure A.3.2-1 gives an example when a UE connects to different IP-CAN respectively and performs multiple registrations. In this example the UE also supports the Controller UE procedures for IUT transfer. In this example the SCC AS receives the registration state information that it needs to implement SCC specific requirements from the third-party SIP REGISTER request.

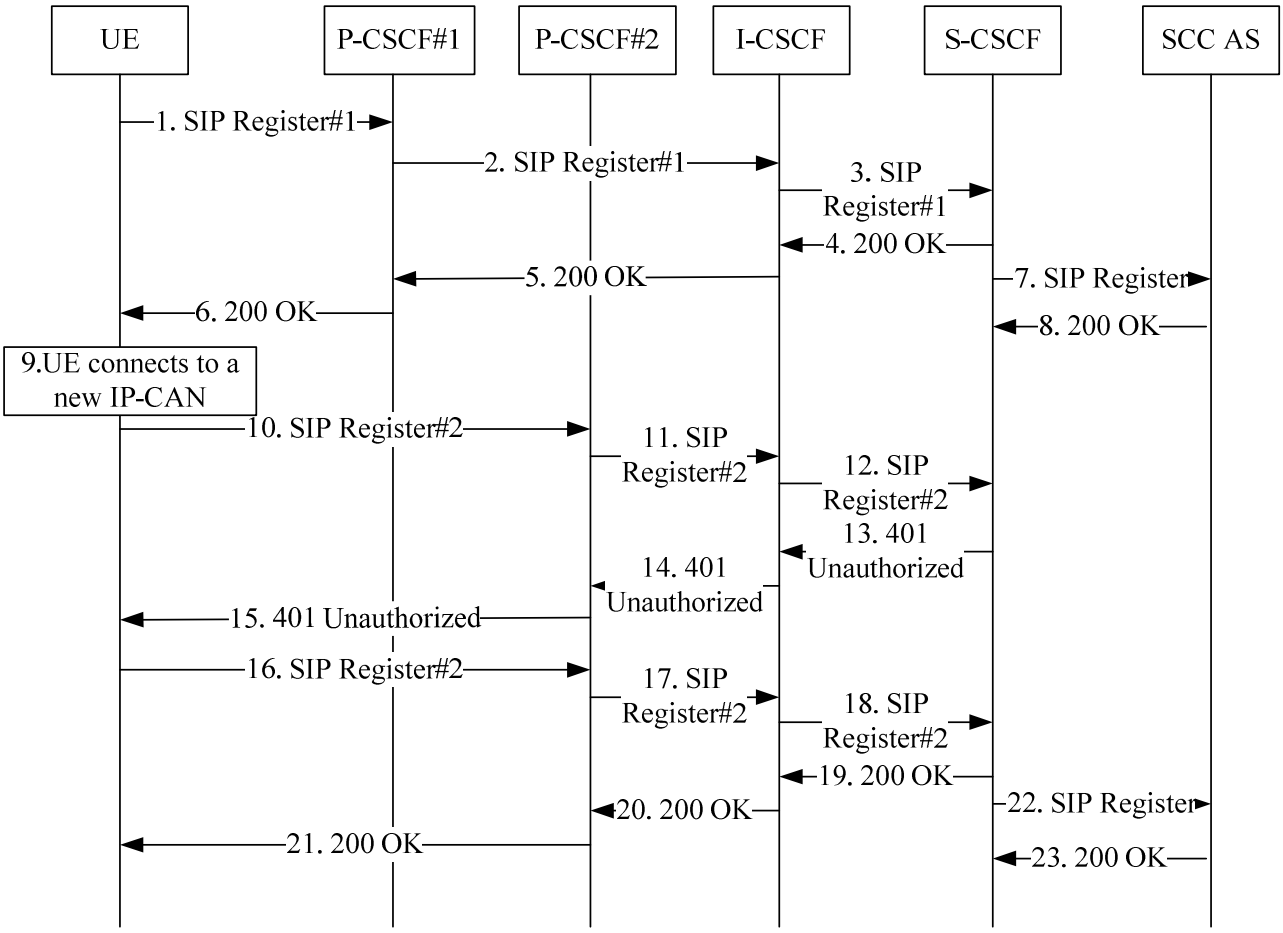


Figure A.3.2-1 Signalling flows for multiple registrations

1. **SIP REGISTER request (UE to P-CSCF#1)-See example in table A.3.2-1**

   UE sends the SIP REGISTER request via the IP-CAN#1.

NOTE 1: For clarity, the unprotected SIP REGISTER request via the IP-CAN#1 is not shown in this example.

Table A.3.2-1SIP REGISTER request (UE to P-CSCF#1)

```
REGISTER sip:registrar.home1.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Max-Forwards: 70
P-Access-Network-Info: 3GPP-UTRAN-TDD; utran-cell-id-3gpp=234151D0FCE11
From: <sip:user1_public1@home1.net>;tag=4fa3
To: <sip:user1_public1@home1.net>
Contact: <sip:[5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp>; reg-id=1; +sip.instance="<
   urn:gsma:imei:90420156-025763-0 >";+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-
   service.ims.icsi.mmtel";+g.3gpp.ics="principal";+g.3gpp.accesstype="cellular1";expires=6000
   00
Call-ID: apb03a0s09dkjdfglkj49111
Authorization: Digest username="user1_private@home1.net", realm="registrar.home1.net",
   nonce=base64(RAND + AUTN + server specific data), algorithm=AKAv1-MD5,
   uri="sip:registrar.home1.net", response="6629fae49393a05397450978507c4ef1"
Security-Client: ipsec-3gpp; alg=hmac-sha-1-96; spi-c=23456789; spi-s=12345678; port-c=2468;
   port-s=1357
Security-Verify: ipsec-3gpp; q=0.1; alg=hmac-sha-1-96; spi-c=98765432; spi-s=87654321; port-
   c=8642; port-s=7531
Require: sec-agree
Proxy-Require: sec-agree
CSeq: 2 REGISTER
Supported: path, outbound, gruu
Content-Length: 0
```

2. **SIP REGISTER request (P-CSCF#1 to I-CSCF)-See example in table A.3.2-2**

After performing the DNS query, the P-CSCF#1 forwards the SIP REGISTER request towards I-CSCF. The P-CSCF adds a Path header field with a flow token and includes the 'ob' parameter.

Table A.3.2-2 SIP REGISTER request (P-CSCF#1 to I-CSCF)

```
REGISTER sip:registrar.home1.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.visited1.net;branch=z9hG4bK240f34.1, SIP/2.0/UDP
   [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Max-Forwards: 69
P-Access-Network-Info:
Path: <sip:VsktcQ/S8p4WPbOnHbuyh5iJvJIW3ib@pcscf1.visited1.net;lr;ob>
Require: path
P-Visited-Network-ID: "Visited Network Number 1"
P-Charging-Vector: icid-value="AyretyU0dm+6O2IrT5tAFrbHLso=023551024"
From:
To:
Contact:
Call-ID:
Authorization: Digest username="user1_private@home1.net", realm="registrar.home1.net",
   nonce=base64(RAND + AUTN + server specific data), algorithm=AKAv1-MD5,
   uri="sip:registrar.home1.net", response="6629fae49393a05397450978507c4ef1", integrity-
   protected=yes
CSeq:
Supported:
Content-Length:
```

3. **SIP REGISTER request (I-CSCF to S-CSCF)**

The I-CSCF forwards the SIP REGISTER request to the S-CSCF.

4. **SIP 200 (OK) response (S-CSCF to I-CSCF)-See example in table A.3.2-4**

The S-CSCF sends a SIP 200 (OK) response to the I-CSCF indicating that Registration was successful. AS the URI in the first Path header field has an "ob" URI parameter, it include a Require header field with the option-tag "outbound".

Table A.3.2-4: SIP 200 (OK) response (S-CSCF to I-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP icscf1_p.home1.net;branch=z9hG4bK351g45.1, SIP/2.0/UDP
     pcscf1.visited1.net;branch=z9hG4bK240f34.1, SIP/2.0/UDP
     [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Path: <sip:term@pcscf1.visited1.net;lr;ob>
Service-Route: <sip:orig@scscf1.home1.net;lr>
From:
To:
Call-ID:
Contact: <sip:[5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp>;
   pub-gruu=" sip:user1_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91e6bf6"
   ;temp-gruu="sip:tgruu.7hs==jd7vnzga5w7fajsc7-ajd6fabz0f8g5@example.com;gr"
   ;+sip.instance="< urn:gsma:imei:90420156-025763-0 >"+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-
   service.ims.icsi.mmtel";+g.3gpp.ics="principal";+g.3gpp.accesstype="cellular1"
   ;expires=600000
CSeq:
Supported: path, outbound
Require: outbound
Date: Wed, 11 July 2001 08:49:37 GMT
P-Associated-URI: <sip:user1_public2@home1.net>, <sip:user1_public3@home1.net>, <sip:+1-212-
   555-1111@home1.net;user=phone>
Content-Length:
```

5-6. **SIP 200 (OK) response (I-CSCF to UE)**

The I-CSCF forwards the SIP 200 (OK) response to the UE via P-CSCF#1.

7. **SIP REGISTER request (S-CSCF to SCC AS)-See example in table A.3.2-7**

After UE successfully registered in the IM CN subsystem, the S-CSCF sends a third party SIP REGISTER request to the SCC AS based on the initial filter criteria it received.

Table A.3.2-7: SIP REGISTER request (S-CSCF to SCC AS)

```
REGISTER sip: sccas.home1.net /2.0
Via: SIP/2.0/UDP scscf1.home1.net;branch=z9hG499ffhy
Max-Forwards: 70
From: <sip:scscf1.home1.net>; tag=538ya
To: <sip:user1_public1@home1.net>
Call-ID: 1asdaddlrfjflslj40a222
P-Access-Network-Info: 3GPP-UTRAN-TDD; utran-cell-id-3gpp=234151D0FCE11
Contact: <sip:scscf1.home1.net>; expires=600000
CSeq: 87 REGISTER
Content-Type: multipart/mixed;boundary="boundary1"
Content-Length: (...)

--boundary1
Content-Type: message/sip

REGISTER sip:registrar.home1.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.visited1.net;branch=z9hG4bK240f34.1, SIP/2.0/UDP
[5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Max-Forwards: 69
P-Access-Network-Info: 3GPP-UTRAN-TDD; utran-cell-id-3gpp=234151D0FCE11
Path: <sip:VskztcQ/S8p4WPbOnHbuyh5iJvJIW3ib@pcscf1.visited1.net;lr;ob>
Require: path
P-Visited-Network-ID: "Visited Network Number 1"
P-Charging-Vector: icid-value="AyretyU0dm+6O2IrT5tAFrbHLso=023551024"
From: <sip:user1_public1@home1.net>;tag=4fa3
To: <sip:user1_public1@home1.net>
Contact: <sip:[5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp>; reg-id=1; +sip.instance="<
urn:gsma:imei:90420156-025763-0>";+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-
service.ims.icsi.mmtel";+g.3gpp.ics="principal";+g.3gpp.accesstype="cellular1";expires=600000
Call-ID: apb03a0s09dkjdfglkj49111
Authorization: Digest username="user1_private@home1.net", realm="registrar.home1.net",
nonce=base64(RAND + AUTN + server specific data), algorithm=AKAv1-MD5,
uri="sip:registrar.home1.net", response="6629fae49393a05397450978507c4ef1"
CSeq: 2 REGISTER
Supported: path, outbound, gruu
Content-Length: 0

--boundary1
Content-Type: message/sip

SIP/2.0 200 OK
Via: SIP/2.0/UDP icscf1_p.home1.net;branch=z9hG4bK351g45.1, SIP/2.0/UDP
pcscf1.visited1.net;branch=z9hG4bK240f34.1, SIP/2.0/UDP
[5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Path: <sip:term@pcscf1.visited1.net;lr;ob>
Service-Route: <sip:orig@scscf1.home1.net;lr>
From: <sip:user1_public1@home1.net>;tag=4fa3
To: <sip:user1_public1@home1.net>;tag=3ec1
Call-ID: apb03a0s09dkjdfglkj49111
Contact: <sip:[5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp>;
pub-gruu="sip:user1_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91e6bf6"
;temp-gruu="sip:tgruu.7hs==jd7vnzga5w7fajsc7-ajd6fabz0f8g5@example.com;gr"
;+sip.instance="< urn:gsma:imei:90420156-025763-0>";+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-
service.ims.icsi.mmtel";+g.3gpp.ics="principal"+g.3gpp.accesstype="cellular1"
;expires=600000
Supported: path, outbound
Require: outbound
Date: Wed, 11 July 2001 08:49:37 GMT
P-Associated-URI: <sip:user1_public2@home1.net>, <sip:user1_public3@home1.net>, <sip:+1-212-555-
1111@home1.net;user=phone>
CSeq: 2 REGISTER
Content-Length: 0

--boundary1--
```

8. SIP 200 (OK) response (SCC AS to S-CSCF)

The SCC AS generates the SIP 200 (OK) response to the third party SIP REGISTER request.

9. UE connects to a new IP-CAN

The UE connects to a new IP-CAN and will perform the registration via the new IP-CAN.

10. SIP REGISTER request (UE to P-CSCF#2)- See example in table A.3.2-10

UE sends the unprotected SIP REGISTER request via the new IP-CAN to P-CSCF+2 which in this example is a different one with previous registration.

Table A.3.2-10: SIP REGISTER request (UE to P-CSCF#2)

```
REGISTER sip:registrar.home1.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:eee];comp=sigcomp;branch=z9hG4bKnasiuen8
Max-Forwards: 70
P-Access-Network-Info: IEEE-802.11b
From: <sip:user1_public1@home1.net>;tag=2hiue
To: <sip:user1_public1@home1.net>
Contact: <sip:[5555::aaa:bbb:ccc:eee];comp=sigcomp>; reg-id=2;   +sip.instance="<
   urn:gsma:imei:90420156-025763-0>;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-
   service.ims.icsi.mmtel";+g.3gpp.ics="principal";+g.3gpp.accesstype="wlan2";expires=600000
Call-ID: E05133BD26DD
Authorization: Digest username="user1_private@home1.net", realm="registrar.home1.net",
   nonce="", uri="sip:registrar.home1.net", response=""
Security-Client: ipsec-3gpp; alg=hmac-sha-1-96; spi-c=23456789; spi-s=12345678; port-c=1234;
   port-s=5678
Require: sec-agree
Proxy-Require: sec-agree
CSeq: 1 REGISTER
Supported: path, outbound, gruu
Content-Length: 0
```

11-12. SIP REGISTER request (P-CSCF#2 to S-CSCF)

The P-CSCF forwards the SIP REGISTER request towards S-CSCF via I-CSCF. Likewise in message #2, P-CSCF#2 adds a Path header field with flow token and 'ob' parameter.

13-15. SIP 401 (Unauthorized) response (S-CSCF to UE)

The authentication challenge is sent in the SIP 401 (Unauthorized) response towards the UE.

16-18. SIP REGISTER request (UE to S-CSCF)

The UE sends the protected SIP REGISTER request towards S-CSCF using contact#2.

19-21. SIP 200 (OK) response (S-CSCF to UE)

The S-CSCF sends a SIP 200 (OK) response towards the UE indicating that registration was successful.

22. SIP REGISTER request (S-CSCF to SCC AS)

The S-CSCF sends a third party SIP REGISTER request to the SCC AS based on the initial filter criteria it received.

Table A.3.2-22: SIP REGISTER request (S-CSCF to SCC AS)

```
REGISTER sip: sccas.home1.net /2.0
Via: SIP/2.0/UDP scscf1.home1.net;branch=z9hG499ffhy
Max-Forwards: 70
From: <sip:scscf1.home1.net>; tag=538ya
To: <sip:user1_public1@home1.net>
P-Access-Network-Info: IEEE-802.11b
Call-ID: 1asdaddlrfjflslj40a222
Contact: <sip:scscf1.home1.net>; expires=600000
CSeq: 87 REGISTER
Content-Type: multipart/mixed;boundary="boundary1"
Content-Length: (...)

--boundary1
Content-Type: message/sip

REGISTER sip:registrar.home1.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.visited1.net;branch=z9hG4bK240f34.1, SIP/2.0/UDP
[5555::aaa:bbb:ccc:eee]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Max-Forwards: 69
P-Access-Network-Info: IEEE-802.11b
Path: <sip:VskztcQ/S8p4WPbOnHbuyh5iJvJIW3ib@pcscf1.visited1.net;lr;ob>
Require: path
```

```
P-Visited-Network-ID: "Visited Network Number 1"
P-Charging-Vector: icid-value="AyretyU0dm+6O2IrT5tAFrbHLso=023551024"
From: <sip:user1_public1@home1.net>;tag=2hiue
To: <sip:user1_public1@home1.net>
Contact: <sip:[5555::aaa:bbb:ccc:eee];comp=sigcomp>;reg-id=2;+sip.instance="<
urn:gsma:imei:90420156-025763-0>;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-
service.ims.icsi.mmtel";+g.3gpp.ics="principal";+g.3gpp.accesstype ="wlan2";expires=600000
Call-ID: apb03a0s09dkjdfglkj49111
Authorization: Digest username="user1_private@home1.net", realm="registrar.home1.net",
nonce=base64(RAND + AUTN + server specific data), algorithm=AKAv1-MD5,
uri="sip:registrar.home1.net", response="6629fae49393a05397450978507c4ef1"
CSeq: 3 REGISTER
Supported: path, outbound, gruu
Content-Length: 0

--boundary1
Content-Type: message/sip

SIP/2.0 200 OK
Via: SIP/2.0/UDP icscf1_p.home1.net;branch=z9hG4bK351g45.1, SIP/2.0/UDP
pcscf1.visited1.net;branch=z9hG4bK240f34.1, SIP/2.0/UDP
[5555::aaa:bbb:ccc:eee]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Path: <sip:term@pcscf1.visited1.net;lr;ob>
Service-Route: <sip:orig@scscf1.home1.net;lr>
From: <sip:user1_public1@home1.net>;tag=2hiue
To: <sip:user1_public1@home1.net>;tag=2da87
Call-ID: apb03a0s09dkjdfglkj49111
Contact: <sip:[5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp>;
pub-gruu="sip:user1_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91e6bf6"
;temp-gruu="sip:tgruu.7hs==jd7vnzga5w7fajsc7-ajd6fabz0f8g5@example.com;gr"
;+sip.instance="<urn:gsma:imei:90420156-025763-0>";+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-
service.ims.icsi.mmtel";+g.3gpp.ics="principal";+g.3gpp.accesstype="wlan2"
;expires=600000
Supported: path, outbound
Require: outbound
Date: Wed, 11 July 2001 08:49:37 GMT
P-Associated-URI: <sip:user1_public2@home1.net>, <sip:user1_public3@home1.net>, <sip:+1-212-555-
1111@home1.net;user=phone>
CSeq: 3 REGISTER
Content-Length: 0

--boundary1--
```

23. SIP 200 (OK) response (SCC AS to S-CSCF)

The SCC AS generates the SIP 200 (OK) response to the third-party SIP REGISTER request.

A.3.3 Signalling flows for registration with SRVCC enhancements

The signalling flows shown in figure A.3.3-1 gives an example flow for UE registration when ATCF is invoked.

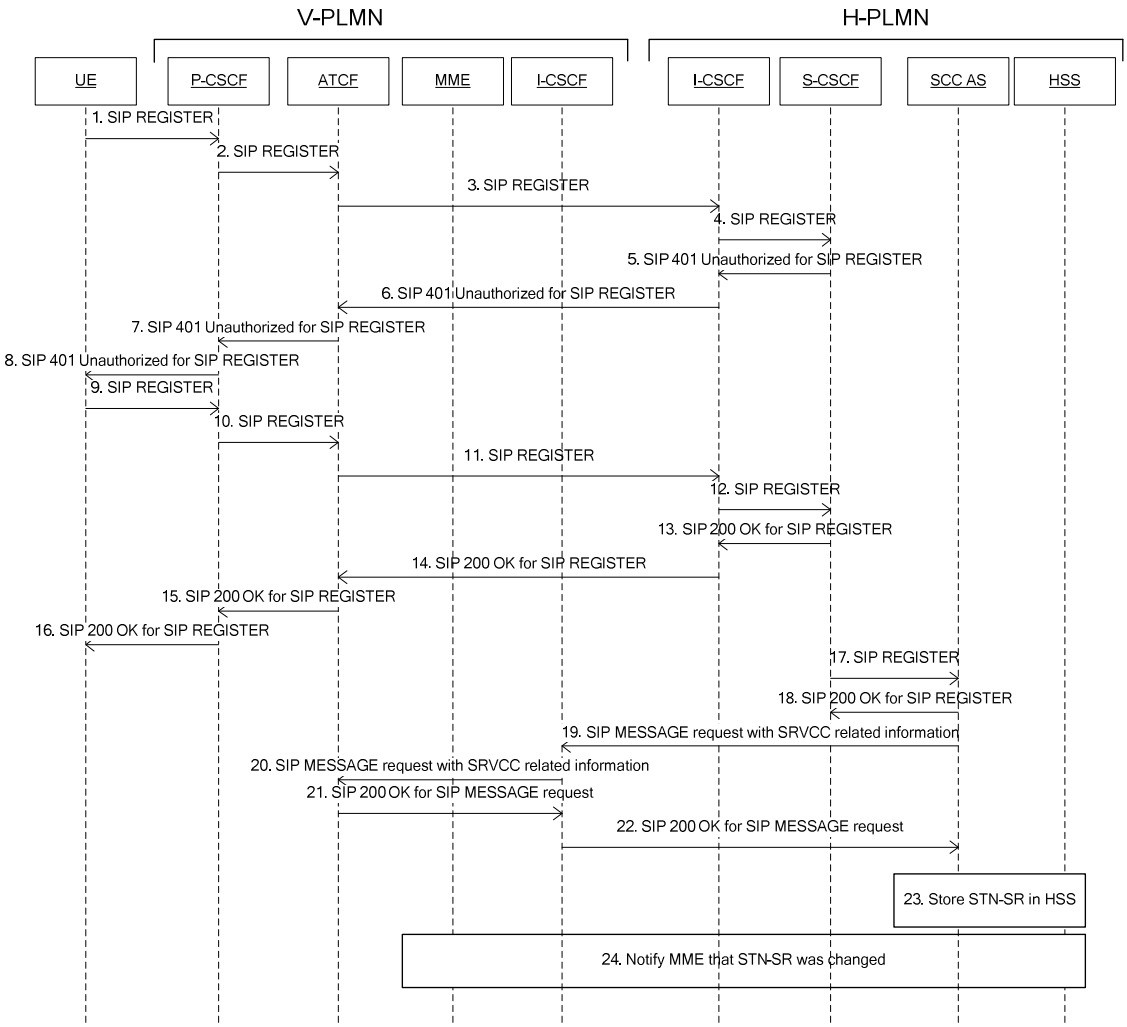


Figure A.3.3-1 registration with SRVCC enhancements

1. SIP REGISTER request (UE to P-CSCF) - see example in table A.3.3-1

UE sends the unprotected SIP REGISTER request to P-CSCF.

Table A.3.3-1: SIP REGISTER request (UE to P-CSCF)

```
REGISTER sip:home1.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:eee];comp=sigcomp;branch=z9hG4bKnasiuen8
Max-Forwards: 70
P-Access-Network-Info: 3GPP-UTRAN-TDD; utran-cell-id-3gpp=234151D0FCE11
From: <sip:user1_public1@home1.net>;tag=2hiue
To: <sip:user1_public1@home1.net>
Contact: <sip:[5555::aaa:bbb:ccc:eee];comp=sigcomp>;+sip.instance="<urn:gsma:imei:90420156-
   025763-0>;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
Call-ID: E05133BD26DD
Authorization: Digest username="user1_private@home1.net", realm="registrar.home1.net",
   nonce="", uri="sip:home1.net", response=""
Security-Client: ipsec-3gpp; alg=hmac-sha-1-96; spi-c=23456789; spi-s=12345678; port-c=1234;
   port-s=5678
Require: sec-agree
Proxy-Require: sec-agree
CSeq: 1 REGISTER
Supported: path, gruu
Content-Length: 0
```

2. SIP REGISTER request (P-CSCF to ATCF) - see example in table A.3.3-2

The P-CSCF forwards the SIP REGISTER request towards ATCF.

Table A.3.3-2: SIP REGISTER request (P-CSCF to ATCF)

```
REGISTER sip:home1.net SIP/2.0
Path:<sip:aga2gfgf@pcscf1.visited2.net:5070;ob>
Route: <sip:reg@atcf.visited2.net;lr>
P-Visited-Network-ID: "Visited Network Number 1"
P-Charging-Vector: icid-value="AyretyU0dm+6O2IrT5tAFrbHLso=023551024";orig-ioi="12345"
Via: SIP/2.0/UDP pcscf1.visited2.net:5060;branch=z9hG4bKnas56565, SIP/2.0/UDP
   [5555::aaa:bbb:ccc:eee];comp=sigcomp;branch=z9hG4bKnasiuen8;rport=5060;received=5555::aaa:b
   bb:ccc:eee
Max-Forwards: 69
P-Access-Network-Info:
From:
To:
Contact:
Call-ID:
Authorization:
Require:
Proxy-Require:
CSeq:
Supported:
Content-Length:
```

Route: ATCF URI for originating requests (as configured in P-CSCF).

3.-4. SIP REGISTER request (ATCF towards S-CSCF) - see example in table A.3.3-3

The ATCF decides to include itself for sessions created using this registration and forwards the SIP REGISTER request along the Route header fields.

Table A.3.3-3: SIP REGISTER request (ATCF towards S-CSCF)

```
REGISTER sip:home1.net SIP/2.0
Feature-Caps: *;+g.3gpp.atcf="<tel:+1-237-888-9999>"; +g.3gpp.atcf-mgmt-uri =
   "<sip:atcf.visited2.net>";+g.3gpp.atcf-
   path="<sip:termsdgfdfwe@atcf.visited2.net>";+g.3gpp.mid-call;+g.3gpp.srvcc-alerting
Path:<sip:termsdgfdfwe@atcf.visited2.net>,<sip:aga2gfgf@pcscf1.visited2.net:5070;ob>
Route: <sip:icscf.home1.net;lr>
P-Visited-Network-ID:
P-Charging-Vector:
Via: SIP/2.0/UDP atcf.visited2.net:5060;branch=z9hG4bKnas5889; SIP/2.0/UDP
   pcscf1.visited2.net:5060;branch=z9hG4bKnas56565, SIP/2.0/UDP
   [5555::aaa:bbb:ccc:eee];comp=sigcomp;branch=z9hG4bKnasiuen8;rport=5060;received=5555::aaa:b
   bb:ccc:eee
Max-Forwards: 68
P-Access-Network-Info:
From:
To:
Contact:
Call-ID:
Authorization:
Require:
Proxy-Require:
CSeq:
Supported:
Content-Length:
```

Path: ATCF URI for terminating requests followed by P-CSCF URI for terminating requests. ATCF URI for terminating requests uniquely identifies registration path (or registration flow, if multiple registration mechanism is used).

Feature-Caps: The header field contains:

- g.3gpp.atcf feature-capability indicator indicating that the ATCF role is supported by this URI;
- g.3gpp.atcf-mgmt-uri feature-capability indicator indicating the management URI of the ATCF for receiving SIP MESSAGE requests containing SRVCC related information and the g.3gpp.atcf-path feature-capability indicator. The value of the g.3gpp.atcf feature-capability indicator contains the STN-SR allocated by ATCF. The value of the g.3gpp.atcf-mgmt-uri feature-capability indicator contains the ATCF management URI. The value of the g.3gpp.atcf-path feature-capability indicator is the ATCF URI for terminating requests;

- g.3gpp.mid-call indicating that all MSC servers, which can be involved in the SRVCC procedures and which are in the same network as the ATCF, support the MSC server assisted mid-call feature; and
- g.3gpp.srvcc-alerting indicating that all MSC servers, which can be involved in the SRVCC procedures and which are in the same network as the ATCF, support the PS to CS SRVCC for calls in alerting phase.

Route: URI of the entry point of the home network of the UE.

5-8. SIP 401 (Unauthorized) response (S-CSCF to UE)

The authentication challenge is sent in the SIP 401 (Unauthorized) response towards the UE.

9. SIP REGISTER request (UE to P-CSCF) - see example in table A.3.3-9

UE sends the protected SIP REGISTER request to P-CSCF.

Table A.3.3-9: SIP REGISTER request (UE to P-CSCF)

```
REGISTER sip:home1.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:eee];comp=sigcomp;branch=z9hG4bKnasiuen8
Max-Forwards: 70
P-Access-Network-Info:
From:
To:
Contact:
Call-ID:
Authorization: Digest username="user1_private@home1.net", realm="registrar.home1.net",
   nonce=base64(RAND + AUTN + server specific data), algorithm=AKAv1-MD5, uri="sip:home1.net",
   response="6629fae49393a05397450978507c4ef1"
Security-Client: ipsec-3gpp; alg=hmac-sha-1-96; spi-c=23456789; spi-s=12345678; port-c=1234;
   port-s=5678
Security-Verify: ipsec-3gpp; q=0.1; alg=hmac-sha-1-96; spi-c=98765432; spi-s=87654321; port-
   c=8642; port-s=7531
Require:
Proxy-Require:
CSeq: 2 REGISTER
Supported:
Content-Length:
```

10. SIP REGISTER request (P-CSCF to ATCF) - see example in table A.3.3-10

The P-CSCF forwards the SIP REGISTER request towards ATCF.

Table A.3.3-10: SIP REGISTER request (P-CSCF to ATCF)

```
REGISTER sip:home1.net SIP/2.0
Path:<sip:aga2gfgf@pcscf1.visited2.net:5070;ob>
Route: <sip:reg@atcf.visited2.net;lr>
P-Visited-Network-ID: "Visited Network Number 1"
P-Charging-Vector: icid-value="AyretyU0dm+6O2IrT5tAFrbHLso=023551024";orig-ioi="12345"
Via: SIP/2.0/UDP pcscf1.visited2.net:5060;branch=z9hG4bKnas56565, SIP/2.0/UDP
   [5555::aaa:bbb:ccc:eee];comp=sigcomp;branch=z9hG4bKnasiuen8;rport=5060;received=5555::aaa:b
   bb:ccc:eee
Max-Forwards: 69
P-Access-Network-Info:
From:
To:
Contact:
Call-ID:
Authorization:
Require:
Proxy-Require:
CSeq:
Supported:
Content-Length:
```

Route: ATCF URI for originating requests (as configured in P-CSCF).

11-12. SIP REGISTER request (ATCF towards S-CSCF) - see example in table A.3.3-11

The ATCF decides to include itself for sessions created using this registration and forwards the SIP REGISTER request.

Table A.3.3-11: SIP REGISTER request (ATCF towards S-CSCF)

```
REGISTER sip:home1.net SIP/2.0
Feature-Caps: *;+g.3gpp.atcf="<tel:+1-237-888-9999>" ;+g.3gpp.atcf-mgmt-uri=
   "<sip:atcf.visited2.net>";+g.3gpp.atcf-
   path="<sip:termsdgfdfwe@atcf.visited2.net>";+g.3gpp.mid-call;+g.3gpp.srvcc-alerting
Path:<sip:termsdgfdfwe@atcf.visited2.net>,<sip:aga2gfgf@pcscf1.visited2.net:5070;ob>
Route: <sip:icscf.home1.net;lr>
P-Visited-Network-ID:
P-Charging-Vector:
Via: SIP/2.0/UDP atcf.visited2.net:5060;branch=z9hG4bKnas5889; SIP/2.0/UDP
   pcscf1.visited2.net:5060;branch=z9hG4bKnas56565, SIP/2.0/UDP
   [5555::aaa:bbb:ccc:eee];comp=sigcomp;branch=z9hG4bKnasiuen8;rport=5060;received=5555::aaa:b
   bb:ccc:eee
Max-Forwards: 68
P-Access-Network-Info:
From:
To:
Contact:
Call-ID:
Authorization:
Require:
Proxy-Require:
CSeq:
Supported:
Content-Length:
```

Path: ATCF URI for terminating requests followed by P-CSCF URI for terminating requests. ATCF URI for terminating requests uniquely identifies registration path (or registration flow, if multiple registration mechanism is used).

Feature-Caps: The header field contains

- g.3gpp.atcf feature-capability indicator indicating that the ATCF role is supported by this URI;
- g.3gpp.atcf-mgmt-uri feature-capability indicator indicating the management URI of the ATCF for receiving SIP MESSAGE requests containing SRVCC related information and the g.3gpp.atcf-path feature-capability indicator. The value of the g.3gpp.atcf feature-capability indicator contains the STN-SR allocated by ATCF. The value of the g.3gpp.atcf-mgmt-uri feature-capability indicator contains the ATCF management URI. The value of the g.3gpp.atcf-path feature-capability indicator is the ATCF URI for terminating requests.
- g.3gpp.mid-call indicating that all MSC servers, which can be involved in the SRVCC procedures and which are in the same network as the ATCF, support the MSC server assisted mid-call feature.
- g.3gpp.srvcc-alerting indicating that all MSC servers, which can be involved in the SRVCC procedures and which are in the same network as the ATCF, support the PS to CS SRVCC for calls in alerting phase.

Route: URI of the entry point of the home network of the UE.

13.-14.SIP 200 (OK) response (S-CSCF to ATCF)

The S-CSCF sends a SIP 200 (OK) response towards the UE indicating that registration was successful.

15.-16.SIP 200 (OK) response (ATCF to UE)

The ATCF sends a SIP 200 (OK) response towards the UE indicating that registration was successful.

Table A.3.3-15: SIP 200 (OK) response to the REGISTER request (ATCF towards UE)

```
SIP/2.0 200 OK
Feature-Caps: *;+g.3gpp.atcf="<tel:+1-237-888-9999>"
Path:<sip:termsdgfdfwe@atcf.visited2.net>,<sip:aga2gfgf@pcscf1.visited2.net:5070;ob>
Service-Route: <sip:orig@scscf1.home1.net;lr>
P-Charging-Vector:
Via: SIP/2.0/UDP atcf.visited2.net:5060;branch=z9hG4bKnas5889; SIP/2.0/UDP
   pcscf1.visited2.net:5060;branch=z9hG4bKnas56565, SIP/2.0/UDP
   [5555::aaa:bbb:ccc:eee];comp=sigcomp;branch=z9hG4bKnasiuen8;rport=5060;received=5555::aaa:b
   bb:ccc:eee
Max-Forwards: 66
From:
To:
Contact:
Call-ID:
Authorization:
CSeq:
Supported:
Content-Length:
```

Feature-Caps: The header field contains g.3gpp.atcf feature-capability indicator indicating that the ATCF role is supported.

17. SIP REGISTER request (S-CSCF to SCC AS) - see example in table A.3.3-17

The S-CSCF sends a third party SIP REGISTER request to the SCC AS based on the initial filter criteria it received.

Table A.3.3-17: SIP REGISTER request (S-CSCF to SCC AS)

```
REGISTER sip: sccas.home1.net /2.0
Via: SIP/2.0/TCP scscf1.home1.net;branch=z9hG499ffhy
Max-Forwards: 70
From: <sip:scscf1.home1.net>; tag=538ya
To: <sip:user1_public1@home1.net>
P-Access-Network-Info: IEEE-802.11b
Call-ID: 1asdaddlrfjflslj40a222
Contact: <sip:scscf1.home1.net>; expires=600000
CSeq: 87 REGISTER
Content-Type: multipart/mixed;boundary="boundary1"
Content-Length: (…)

--boundary1
Content-Type: message/sip

REGISTER sip:home1.net SIP/2.0
Feature-Caps: *;+g.3gpp.atcf="<tel:+1-237-888-9999>" ;+g.3gpp.atcf-mgmt-uri=
   "<sip:atcf.visited2.net>";+g.3gpp.atcf-
   path="<sip:termsdgfdfwe@atcf.visited2.net>";+g.3gpp.mid-call;+g.3gpp.srvcc-alerting
Path:<sip:termsdgfdfwe@atcf.visited2.net>,<sip:aga2gfgf@pcscf1.visited2.net:5070;ob>
P-Visited-Network-ID: "Visited Network Number 1"
P-Charging-Vector: icid-value="AyretyU0dm+6O2IrT5tAFrbHLso=023551024";orig-ioi="12345"
Via: SIP/2.0/UDP icscf.visited2.net:5060;branch=z9hG4bKnas8866; SIP/2.0/UDP
   atcf.visited2.net:5060;branch=z9hG4bKnas5889; SIP/2.0/UDP
   pcscf1.visited2.net:5060;branch=z9hG4bKnas56565, SIP/2.0/UDP
   [5555::aaa:bbb:ccc:eee];comp=sigcomp;branch=z9hG4bKnasiuen8;rport=5060;received=5555::aaa:b
   bb:ccc:eee
Max-Forwards: 66
P-Access-Network-Info: 3GPP-UTRAN-TDD; utran-cell-id-3gpp=234151D0FCE11
From: <sip:user1_public1@home1.net>;tag=2hiue
To: <sip:user1_public1@home1.net>
Contact: <sip:[5555::aaa:bbb:ccc:eee];comp=sigcomp>;+sip.instance="<urn:gsma:imei:90420156-
   025763-0>;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
Call-ID: E05133BD26DD
Authorization: Digest username="user1_private@home1.net", realm="registrar.home1.net",
   nonce="", uri="sip:home1.net", response=""
Require: sec-agree
Proxy-Require: sec-agree
CSeq: 2 REGISTER
Supported: path, gruu
Content-Length: 0

--boundary1
Content-Type: message/sip

SIP/2.0 200 OK
Path:<sip:termsdgfdfwe@atcf.visited2.net>,<sip:aga2gfgf@pcscf1.visited2.net:5070;ob>
Via: SIP/2.0/UDP icscf.visited2.net:5060;branch=z9hG4bKnas8866; SIP/2.0/UDP
   atcf.visited2.net:5060;branch=z9hG4bKnas5889; SIP/2.0/UDP
   pcscf1.visited2.net:5060;branch=z9hG4bKnas56565, SIP/2.0/UDP
   [5555::aaa:bbb:ccc:eee];comp=sigcomp;branch=z9hG4bKnasiuen8;rport=5060;received=5555::aaa:b
   bb:ccc:eee
Service-Route: <sip:orig@scscf1.home1.net;lr>
From: <sip:user1_public1@home1.net>;tag=2hiue
To: <sip:user1_public1@home1.net>;tag=2da87
Call-ID: E05133BD26DD
Contact:
   <sip:[5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp>;+sip.instance="<urn:gsma:imei:90420156-
   025763-0>";+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
 ;pub-gruu="sip:user1_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91e6bf6"
 ;temp-gruu="sip:tgruu.7hs==jd7vnzga5w7fajsc7-ajd6fabz0f8g5@example.com;gr";expires=600000
Supported: path, gruu
P-Associated-URI: <sip:user1_public2@home1.net>, <sip:user1_public3@home1.net>, <sip:+1-212-
   555-1111@home1.net;user=phone>
CSeq: 2 REGISTER
Content-Length: 0

--boundary1--
```

18. SIP 200 (OK) response (SCC AS to S-CSCF)

The SCC AS generates the SIP 200 (OK) response to the third-party SIP REGISTER request.

19.-20.SIP MESSAGE request with SRVCC related information (SCC AS to ATCF)

The SCC AS sends the SIP MESSAGE request with SRVCC related information towards the ATCF serving the registered UE.

Table A.3.3-19: SIP MESSAGE request (SCC AS towards ATCF)

```
MESSAGE sip:atcf.visited2.net SIP/2.0
Via: SIP/2.0/UDP sccas1.home1.net:5060;branch=z9hG4bKnas588339
Max-Forwards: 70
From: <sip:sccas1.home1.net>;tag=aassd
To: sip:atcf.visited2.net
Call-ID: sdvasdfgfasdf
CSeq: 56561 MESSAGE
Content-Length: ...
P-Asserted-Identity: sip:sccas1.home1.net
Content-Type: application/vnd.3gpp.SRVCC-info+xml

<?xml version="1.0" encoding="UTF-8"?>
<SRVCC-infos>
  <SRVCC-info ATCF-Path-URI="sip:termsdgfdfwe@atcf.visited2.net">
    <ATU-STI>sip:sccas1.home1.net</ATU-STI>
    <C-MSISDN>tel:+1-237-555-1111</C-MSISDN>
  </SRVCC-info>
</SRVCC-infos>
```

Request-URI: ATCF management URI

P-Asserted-Identity: SCC AS URI

body: SRVCC related information

21.-22.SIP 200 (OK) response (ATCF to SCC AS)

The ATCF generates the SIP 200 (OK) response to the SIP MESSAGE request.

23. Store STN-SR in HSS (SCC AS to HSS)

SCC AS provides the received STN-SR into the HSS to replace the STN-SR pointing to the SCC AS or the previously stored STN-SR pointing to other ATCF.

NOTE: step 23 can be started in parallel to step 19.

24. Notify MME that STN-SR was changed (HSS to MME)

HSS provides the STN-SR to the MME because of the change of the subscription data.

A.3.4 Signalling flows for registration with SRVCC enhancements from UE supporting CS to PS SRVCC

The signalling flows shown in figure A.3.4-1 gives an example flow for UE registration when ATCF is invoked. UE and ATCF are also enhanced for CS to PS SRVCC.

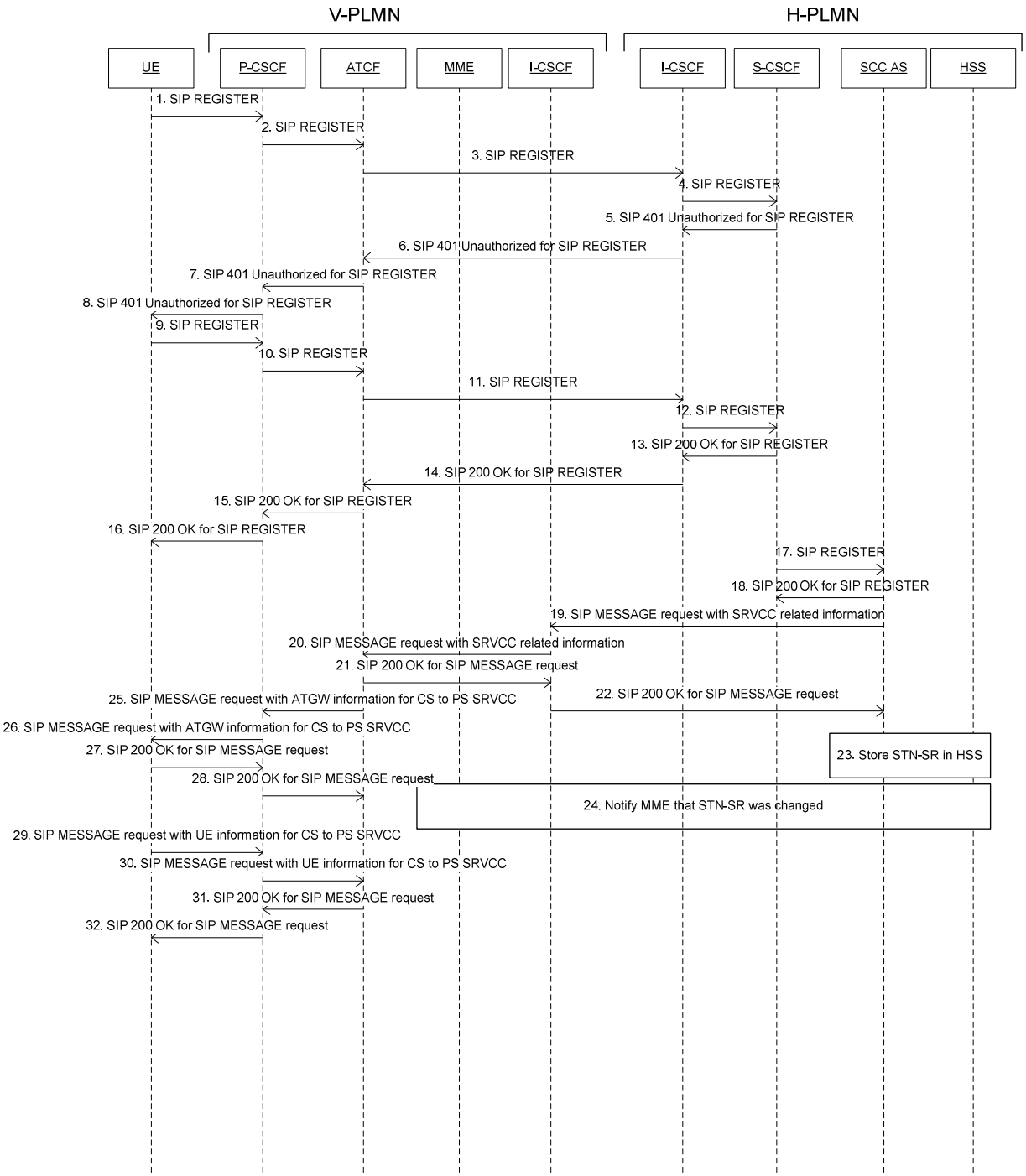


Figure A.3.4-1 registration with CS to PS SRVCC enhancements

1. **SIP REGISTER request (UE to P-CSCF) - see example in table A.3.4-1**

   UE sends the unprotected SIP REGISTER request to P-CSCF.

Table A.3.4-1: SIP REGISTER request (UE to P-CSCF)

```
REGISTER sip:home1.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd];comp=sigcomp;branch=z9hG4bKnasiuen8
Max-Forwards: 70
P-Access-Network-Info: 3GPP-UTRAN-TDD; utran-cell-id-3gpp=234151D0FCE11
From: <sip:user1_public1@home1.net>;tag=2hiue
To: <sip:user1_public1@home1.net>
Contact:
   <sip:[5555::aaa:bbb:ccc:ddd]:5432;comp=sigcomp>;+sip.instance="<urn:gsma:imei:90420156-
   025763-0>;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel";+g.3gpp.cs2ps-
   srvcc;+g.3gpp.cs2ps-srvcc-alerting
Call-ID: E05133BD26DD
Authorization: Digest username="user1_private@home1.net", realm="registrar.home1.net",
   nonce="", uri="sip:home1.net", response=""
Security-Client: ipsec-3gpp; alg=hmac-sha-1-96; spi-c=23456789; spi-s=12345678; port-c=1234;
   port-s=5678
Require: sec-agree
Proxy-Require: sec-agree
CSeq: 1 REGISTER
Supported: path, gruu
Content-Length: 0
```

Contact header field: media feature tag g.3gpp.cs2ps-srvcc indicates support of the CS to PS SRVCC in the UE; and media feature tag +g.3gpp.cs2ps-srvcc-alerting indicates support of the CS to PS SRVCC of calls in alerting phase in the UE.

2. SIP REGISTER request (P-CSCF to ATCF) - see example in table A.3.4-2

The P-CSCF forwards the SIP REGISTER request towards ATCF.

Table A.3.4-2: SIP REGISTER request (P-CSCF to ATCF)

```
REGISTER sip:home1.net SIP/2.0
Path:<sip:aga2gfgf@pcscf1.visited2.net:5070;ob>
Route: <sip:reg@atcf.visited2.net;lr>, <sip:icscf.home1.net;lr>
P-Visited-Network-ID: "Visited Network Number 1"
P-Charging-Vector: ####
Via: SIP/2.0/UDP pcscf1.visited2.net:5060;branch=z9hG4bKnas56565, SIP/2.0/UDP
   [5555::aaa:bbb:ccc:ddd];comp=sigcomp;branch=z9hG4bKnasiuen8;rport=5060;received=5555::aaa:b
   bb:ccc:ddd
Max-Forwards: 69
P-Access-Network-Info:
From:
To:
Contact:
Call-ID:
Authorization:
Require:
Proxy-Require:
CSeq:
Supported:
Content-Length:
```

Route: ATCF URI for originating requests (as configured in P-CSCF) followed by URI of the entry point of the home network of the UE.

3.-4. SIP REGISTER request (ATCF towards S-CSCF) - see example in table A.3.4-3

The ATCF decides to include itself for sessions created using this registration and forwards the SIP REGISTER request along the Route header fields.

Table A.3.4-3: SIP REGISTER request (ATCF towards S-CSCF)

```
REGISTER sip:home1.net SIP/2.0
Feature-Caps: *;+g.3gpp.atcf="<tel:+1-237-888-9999>"; +g.3gpp.atcf-mgmt-uri=
   "<sip:atcf.visited2.net>";+g.3gpp.atcf-path="<sip:termsdgfdfwe@atcf.visited2.net>";
   +g.3gpp.cs2ps-srvcc="<sip:sti-sr@atcf.visited2.net>"
Path:<sip:termsdgfdfwe@atcf.visited2.net>,<sip:aga2gfgf@pcscf1.visited2.net:5070;ob>
Route: <sip:icscf.home1.net;lr>
P-Visited-Network-ID:
P-Charging-Vector:
Via: SIP/2.0/UDP atcf.visited2.net:5060;branch=z9hG4bKnas5889; SIP/2.0/UDP
   pcscf1.visited2.net:5060;branch=z9hG4bKnas56565, SIP/2.0/UDP
   [5555::aaa:bbb:ccc:ddd];comp=sigcomp;branch=z9hG4bKnasiuen8;rport=5060;received=5555::aaa:b
   bb:ccc:ddd
Max-Forwards: 68
P-Access-Network-Info:
From:
To:
Contact:
Call-ID:
Authorization:
Require:
Proxy-Require:
CSeq:
Supported:
Content-Length:
```

Path: ATCF URI for terminating requests followed by P-CSCF URI for terminating requests. ATCF URI for terminating requests uniquely identifies registration (or registration flow, if multiple registration mechanism is used).

Feature-Caps: The header field contains:

- g.3gpp.atcf feature-capability indicator with value containing the STN-SR allocated by ATCF;
- g.3gpp.atcf-mgmt-uri feature-capability indicator with value containing the ATCF management URI;
- g.3gpp.atcf-path feature-capability indicator with value containing the ATCF URI for terminating requests; and
- g.3gpp.cs2ps-srvcc feature-capability indicator with value containing the STI-rSR allocated by ATCF.

Route: URI of the entry point of the home network of the UE.

5-8. SIP 401 (Unauthorized) response (S-CSCF to UE)

The authentication challenge is sent in the SIP 401 (Unauthorized) response towards the UE.

9. SIP REGISTER request (UE to P-CSCF) - see example in table A.3.4-9

UE sends the protected SIP REGISTER request to P-CSCF.

Table A.3.4-9: SIP REGISTER request (UE to P-CSCF)

```
REGISTER sip:home1.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd];comp=sigcomp;branch=z9hG4bKnasiuen8
Max-Forwards: 70
P-Access-Network-Info:
From:
To:
Contact:
   <sip:[5555::aaa:bbb:ccc:ddd]:5432;comp=sigcomp>;+sip.instance="<urn:gsma:imei:90420156-
   025763-0>;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel";+g.3gpp.cs2ps-
   srvcc;+g.3gpp.cs2ps-srvcc-alerting
Call-ID:
Authorization: Digest username="user1_private@home1.net", realm="registrar.home1.net",
   nonce=base64(RAND + AUTN + server specific data), algorithm=AKAv1-MD5, uri="sip:home1.net",
   response="6629fae49393a05397450978507c4ef1"
Security-Client: ipsec-3gpp; alg=hmac-sha-1-96; spi-c=23456789; spi-s=12345678; port-c=1234;
   port-s=5678
Security-Verify: ipsec-3gpp; q=0.1; alg=hmac-sha-1-96; spi-c=98765432; spi-s=87654321; port-
   c=8642; port-s=7531
Require:
Proxy-Require:
CSeq: 2 REGISTER
Supported:
Content-Length: 0
```

Contact header field: media feature tag g.3gpp.cs2ps-srvcc indicates support of the CS to PS SRVCC in the UE; and media feature tag +g.3gpp.cs2ps-srvcc-alerting indicates support of the CS to PS SRVCC of calls in alerting phase in the UE.

10. SIP REGISTER request (P-CSCF to ATCF) - see example in table A.3.4-10

The P-CSCF forwards the SIP REGISTER request towards ATCF.

Table A.3.4-10: SIP REGISTER request (P-CSCF to ATCF)

```
REGISTER sip:home1.net SIP/2.0
Path:<sip:aga2gfgf@pcscf1.visited2.net:5070;ob>
Route: <sip:reg@atcf.visited2.net;lr>, <sip:icscf.home1.net;lr>
P-Visited-Network-ID: "Visited Network Number 1"
P-Charging-Vector: ####
Via: SIP/2.0/UDP pcscf1.visited2.net:5060;branch=z9hG4bKnas56565, SIP/2.0/UDP
   [5555::aaa:bbb:ccc:ddd];comp=sigcomp;branch=z9hG4bKnasiuen8;rport=5060;received=5555::aaa:b
   bb:ccc:ddd
Max-Forwards: 69
P-Access-Network-Info:
From:
To:
Contact:
Call-ID:
Authorization:
Require:
Proxy-Require:
CSeq:
Supported:
Content-Length:
```

Route: ATCF URI for originating requests (as configured in P-CSCF) followed by URI of the entry point of the home network of the UE.

11-12. SIP REGISTER request (ATCF towards S-CSCF) - see example in table A.3.4-11

The ATCF decides to include itself for sessions created using this registration and forwards the SIP REGISTER request.

Table A.3.4-11: SIP REGISTER request (ATCF towards S-CSCF)

```
REGISTER sip:home1.net SIP/2.0
Feature-Caps: *;+g.3gpp.atcf="<tel:+1-237-888-9999>"; +g.3gpp.atcf-mgmt-uri=
   "<sip:atcf.visited2.net>";+g.3gpp.atcf-path="<sip:termsdgfdfwe@atcf.visited2.net>";
   +g.3gpp.cs2ps-srvcc="<sip:sti-sr@atcf.visited2.net>"
Path:<sip:termsdgfdfwe@atcf.visited2.net>,<sip:aga2gfgf@pcscf1.visited2.net:5070;ob>
Route: <sip:sdvfasdgf34t4@pcscf1.visited2.net:5080>
P-Visited-Network-ID:
P-Charging-Vector:
Via: SIP/2.0/UDP atcf.visited2.net:5060;branch=z9hG4bKnas5889; SIP/2.0/UDP
   pcscf1.visited2.net:5060;branch=z9hG4bKnas56565, SIP/2.0/UDP
   [5555::aaa:bbb:ccc:ddd];comp=sigcomp;branch=z9hG4bKnasiuen8;rport=5060;received=5555::aaa:b
   bb:ccc:ddd
Max-Forwards: 68
P-Access-Network-Info:
From:
To:
Contact:
Call-ID:
Authorization:
Require:
Proxy-Require:
CSeq:
Supported:
Content-Length:
```

Path: ATCF URI for terminating requests followed by P-CSCF URI for terminating requests. ATCF URI for terminating requests uniquely identifies registration (or registration flow, if multiple registration mechanism is used).

Feature-Caps: The header field contains:

- g.3gpp.atcf feature-capability indicator with value containing the STN-SR allocated by ATCF;
- g.3gpp.atcf-mgmt-uri feature-capability indicator with value containing the ATCF management URI;
- g.3gpp.atcf-path feature-capability indicator with value containing the ATCF URI for terminating requests; and
- g.3gpp.cs2ps-srvcc feature-capability indicator with value containing the STI-rSR allocated by ATCF.

Route: URI of the entry point of the home network of the UE.

13.-14.SIP 200 (OK) response (S-CSCF towards ATCF)

The S-CSCF sends a SIP 200 (OK) response towards the UE indicating that registration was successful.

15.-16.SIP 200 (OK) response (ATCF towards UE)- see example in table A.3.4-15

The ATCF sends a SIP 200 (OK) response towards the P-CSCF indicating that registration was successful.

Table A.3.4-15: SIP 200 (OK) response to the SIP REGISTER request (ATCF towards UE)

```
SIP/2.0 200 OK
Feature-Caps: *;+g.3gpp.atcf="<tel:+1-237-888-9999>";+g.3gpp.cs2ps-srvcc="<sip:sti-
   sr@atcf.visited2.net>"
Path:<sip:termsdgfdfwe@atcf.visited2.net>,<sip:aga2gfgf@pcscf1.visited2.net:5070;ob>
Service-Route: <sip:orig@scscf1.home1.net;lr>
P-Charging-Vector:
Via: SIP/2.0/UDP atcf.visited2.net:5060;branch=z9hG4bKnas5889; SIP/2.0/UDP
   pcscf1.visited2.net:5060;branch=z9hG4bKnas56565, SIP/2.0/UDP
   [5555::aaa:bbb:ccc:eee];comp=sigcomp;branch=z9hG4bKnasiuen8;rport=5060;received=5555::aaa:b
   bb:ccc:eee
Max-Forwards: 66
From:
To:
Contact:
Call-ID:
Authorization:
CSeq:
Supported:
Content-Length:
```

Feature-Caps: The header field contains:

- g.3gpp.atcf feature-capability indicator with value containing the STN-SR allocated by ATCF; and

- g.3gpp.cs2ps-srvcc feature-capability indicator with value containing the STI-rSR allocated by ATCF.

17. SIP REGISTER request (S-CSCF to SCC AS)

The S-CSCF sends a third party SIP REGISTER request to the SCC AS based on the initial filter criteria it received.

18. SIP 200 (OK) response (SCC AS to S-CSCF)

The SCC AS generates the SIP 200 (OK) response to the third-party SIP REGISTER request.

19.-20.SIP MESSAGE request with SRVCC related information (SCC AS towards ATCF) - see example in table A.3.4-19

The SCC AS sends the SIP MESSAGE request with SRVCC related information towards the ATCF serving the registered UE.

Table A.3.4-19: SIP MESSAGE request (SCC AS towards ATCF)

```
MESSAGE sip:atcf.visited2.net SIP/2.0
Via: SIP/2.0/UDP sccas1.home1.net:5060;branch=z9hG4bKnas588339
Max-Forwards: 70
From: <sip:sccas1.home1.net>;tag=aassd
To: sip:atcf.visited2.net
Call-ID: sdvasdfgfasdf
CSeq: 56561 MESSAGE
Content-Length: ...
P-Asserted-Identity: sip:sccas1.home1.net
Content-Type: application/vnd.3gpp.SRVCC-info+xml

<?xml version="1.0"?>
<SRVCC-infos>
  <SRVCC-info ATCF-Path-URI="sip:termsdgfdfwe@atcf.visited2.net">
    <ATU-STI>sip:sccas1.home1.net</ATU-STI>
    <C-MSISDN>tel:+1-237-555-1111</C-MSISDN>
    <anyExt>
      <CS2PS-ATU-STI>sip:cs2ps@sccas1.home1.net</CS2PS-ATU-STI>
    </anyExt>
  </SRVCC-info>
</SRVCC-infos>
```

Request-URI: ATCF management URI

P-Asserted-Identity: SCC AS URI

body: SRVCC related information. The CS2PS-ATU-STI element contains the ATU-STI to be used in CS to PS SRVCC.

21.-22.SIP 200 (OK) response (ATCF towards SCC AS)

The ATCF generates the SIP 200 (OK) response to the SIP MESSAGE request.

23. Store STN-SR in HSS (SCC AS to HSS)

SCC AS provides the received STN-SR into the HSS to replace the STN-SR pointing to the SCC AS or the previously stored STN-SR pointing to other ATCF.

NOTE: step 23 can be started in parallel to step 19.

24. Notify MME that STN-SR was changed (HSS to MME)

HSS provides the STN-SR to the MME because of the change of the subscription data.

25.-26.SIP MESSAGE request with ATGW information for CS to PS SRVCC (ATCF towards UE) - see example in table A.3.4-25

Table A.3.4-25: SIP MESSAGE request (ATCF towards UE)

```
MESSAGE sip:[5555::aaa:bbb:ccc:ddd]:5432;comp=sigcomp SIP/2.0
Via: SIP/2.0/UDP atcf.visited2.net:5060;branch=z9hG4bKnas66
Max-Forwards: 70
Route: <sip:aga2gfgf@pcscf1.visited2.net:5070;ob>
P-Asserted-Identity: sip:sti-sr@atcf.visited2.net
From: sip:sti-sr@atcf.visited2.net;tag=aaa5234
To: sip:[5555::aaa:bbb:ccc:ddd]:5432
Call-ID: asgag34t34543
CSeq: 1000034
Content-Type:
Content-Length: (…)
Content-Disposition: render
P-Charging-Vector: ####

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c=IN IP6 dfgrrgr.invalid
t=0 0
m=audio 9 RTP/AVP 97 96
b=AS:25.4
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 telephone-event
```

Request-URI: contact address of the UE

Route: P-CSCF path header field

application/sdp MIME body: SDP describing the set of media streams and codecs the ATGW wishes to use receive the media in session transferred in any later CS to PS SRVCC access transfer. The IP addresses and ports can contain any value as the ATGW IP address and port are selected during the later CS to PS SRVCC access transfer.

27.-28.SIP 200 (OK) response (UE towards ATCF)

The UE generates the SIP 200 (OK) response to the SIP MESSAGE request.

29. SIP MESSAGE request with UE information for CS to PS SRVCC (UE to P-CSCF) - see example in table A.3.4-29

The UE sends the SIP MESSAGE request with UE information for CS to PS SRVCC towards the ATCF.

Table A.3.4-29: SIP MESSAGE request (UE to P-CSCF)

```
MESSAGE sip:sti-sr@atcf.visited2.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Max-Forwards: 70
Route: <sip:pcscf1.visited2.net:7531;lr;comp=sigcomp>, <sip:orig@scscf1.home1.net;lr>
P-Preferred-Identity: "John Doe" <sip:user1_public1@home1.net>
P-Access-Network-Info: 3GPP-UTRAN-TDD; utran-cell-id-3gpp=234151D0FCE11
From: <sip:user1_public1@home1.net>;tag=171828
To: sip:sti-sr@atcf.visited2.net
Call-ID: cb03a0s09a2sdfglkj49033333
CSeq: 56561 MESSAGE
Require: sec-agree
Proxy-Require: sec-agree
Security-Verify: ipsec-3gpp; q=0.1; alg=hmac-sha-1-96; spi-c=98765432; spi-s=87654321; port-
   c=8642; port-s=7531
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE
Content-Type: application/sdp
Content-Length: (…)
Content-Disposition: render

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c=IN IP6 5555::aaa:bbb:ccc:ddd
t=0 0
m=audio 3456 RTP/AVP 97 96
b=AS:25.4
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 telephone-event
```

Request-URI: the STI-rSR allocated by ATCF received in message 16

application/sdp MIME body: SDP describing the set of media streams and codecs the UE wishes to use, along with the IP addresses and ports the UE would like to use to receive the media in session transferred in any later CS to PS SRVCC access transfer.

30. SIP MESSAGE request with UE information for CS to PS SRVCC (P-CSCF to ATCF) - see example in table A.3.4-30

Table A.3.4-30: SIP MESSAGE request (P-CSCF to ATCF)

```
MESSAGE sip:sti-sr@atcf.visited2.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.visited2.net:5060;branch=z9hG4bKnas5656544, SIP/2.0/UDP
   [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Max-Forwards:
Route: <sip:orig@atcf.visited2.net;lr>, <sip:orig@scscf1.home1.net;lr>
P-Asserted-Identity: "John Doe" <sip:user1_public1@home1.net>
P-Access-Network-Info:
From:
To:
Call-ID:
CSeq:
Require:
Supported:
Proxy-Require:
Security-Verify:
Allow:
Content-Type:
Content-Length: (…)
Content-Disposition:

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c=IN IP6 5555::aaa:bbb:ccc:ddd
t=0 0
m=audio 3456 RTP/AVP 97 96
b=AS:25.4
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 telephone-event
```

Request-URI: the STI-rSR allocated by ATCF

Route: ATCF URI for originating requests (as configured in P-CSCF).

application/sdp MIME body: SDP describing the set of media streams and codecs the UE wishes to use, along with the IP addresses and ports the UE would like to use to receive the media in session transferred in any later CS to PS SRVCC access transfer.

31.-32.SIP 200 (OK) response (ATCF towards UE)

The ATCF generates the SIP 200 (OK) response to the SIP MESSAGE request.

A.3.5 Signalling flows for UE attaching to CS domain when MSC server is enhanced for ICS and for CS to PS SRVCC and when UE is not registered with IMS in PS access network yet

This signalling flow shown at figure A.3.5-1 describes the scenario of UE attaching to CS domain when the used MSC server is enhanced for ICS and for CS to PS SRVCC and when the UE is not registered with IMS in PS access network yet.

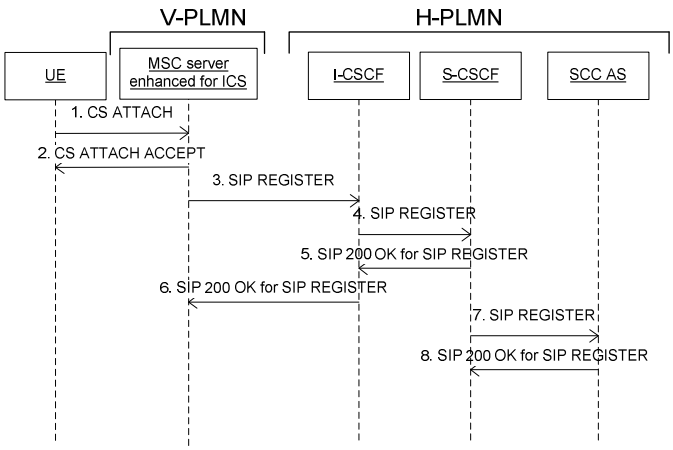


Figure A.3.5-1 MSC Server enhanced for ICS performs registration on behalf of the UE

The details of the signalling flows are as follows:

1. **CS attach (UE to MSC)**

   UE performs CS attachment procedure as specified in 3GPP TS 24.008 [8]. UE indicates support of CS to PS SRVCC in the CS attachment procedure.

3. **CS attach accept (MSC enhanced for ICS to UE)**

   The CS attach request is accepted by the network and an accept message is sent to the MS.

3-4. SIP REGISTER request (MSC Server enhanced for ICS to S-CSCF) – see example in table A.3.5-3

Table A.3.5-3: SIP REGISTER request (MSC Server enhanced for ICS to I-CSCF)

```
REGISTER sip:ics.mnc015.mcc234.3gppnetwork.org SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd];branch=z9hG4bKnashds7
Max-Forwards: 70
P-Access-Network-Info: 3GPP-UTRAN-TDD; utran-cell-id-3gpp=234151D0FCE11
P-Visited-Network-ID: "Visited Network Number 1 for MSC Server"
P-Charging-Vector: ####
Path: <sip:termpdfjkghlj@msc123.visited2.net;lr>
From: <sip:234150999999999@ics.mnc015.mcc234.3gppnetwork.org>;tag=4fa3
To: <sip:234150999999999@ics.mnc015.mcc234.3gppnetwork.org>
Contact: <sip:[5555::aaa:bbb:ccc:ddd]>;expires=600000;+sip.instance="<urn:gsma:imei:90420156-
   025763-0>";+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-
   service.ims.icsi.mmtel";+g.3gpp.ics="server"; +g.3gpp.cs2ps-srvcc;
   +g.3gpp.path="<sip:termpdfjkghlj@msc123.visited2.net;lr>"
Call-ID: apb03a0s09dkjdfglkj49111
Authorization: Digest username="234150999999999@ics.mnc015.mcc234.3gppnetwork.org", realm="
   ics.mnc015.mcc234.3gppnetwork.org", nonce="", integrity-protected=auth-done, uri="sip:
   ics.mnc015.mcc234.3gppnetwork.org", response=""
CSeq: 1 REGISTER
Supported: gruu
Require: path
Content-Length: 0
```

Contact: The header field contains:

- g.3gpp.icsi-ref media feature tag with value of ICSI of IMS multimedia telephony communication service;
- g.3gpp.ics media feature tag with value indicating that the resource is a network node which is ICS capable;
- g.3gpp.cs2ps-srvcc indicating support for CS to PS SRVCC; and
- g.3gpp.path media feature tag with value containing the MSC URI for terminating requests;

5-6. SIP 200 (OK) response (S-CSCF to MSC server enhanced for ICS)

The S-CSCF sends a SIP 200 (OK) response to the MSC server enhanced for ICS.

7. SIP REGISTER request (S-CSCF to SCC AS)

The S-CSCF sends a third party SIP REGISTER request containing in the body the incoming SIP REGISTER request from the PN UE and the SIP 200 (OK) response to the incoming REGISTER request to the SCC AS.

8. SIP 200 (OK) response (SCC AS to S-CSCF)

The SCC AS sends a SIP 200 (OK) response to the S-CSCF indicating the third party SIP REGISTER was successful.

A.3.6 Signalling flows for UE attaching to CS domain when MSC server is enhanced for ICS and for CS to PS SRVCC and when UE is already registered with IMS in PS access network

The signalling flow shown at figure A.3.6-1 describes the scenario of UE attaching to CS domain when MSC server is enhanced for ICS after the UE has already registered with IMS in PS access network. The scenario expects that UE, MSC server enhanced for ICS, ATCF and SCC AS are enhanced for CS to PS SRVCC.

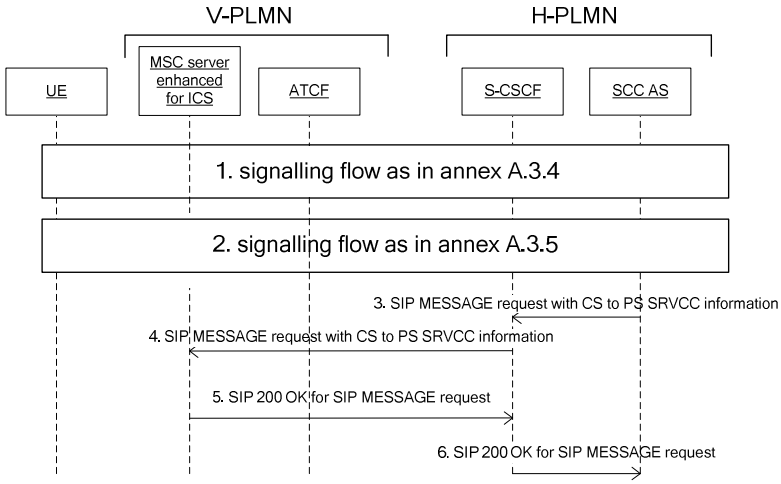


Figure A.3.6-1 registration with SRVCC enhancements

1. SC UE attempts to registers with IMS in PS access network. The signalling flow described in subclause A.3.4 is performed.

2. SC UE attempts to attach to CS domain. MSC server enhanced for ICS registers with IMS without knowing the STN-SR of the ATCF selected during the registration of UE with IMS using PS domain. The signalling flow described in subclause A.3.5 is performed.

3.-4. SIP MESSAGE request with CS to PS SRVCC information (SCC AS to MSC server) - see example in table A.3.6-3

The SCC AS detects that the UE has a registration path over a PS domain where ATCF is included and therefore the SCC AS provides information about the registration path of the UE over a PS domain to the MSC server.

Table A.3.6-3: SIP MESSAGE request (SCC AS towards MSC server)

```
MESSAGE sip:msc123.visited2.net SIP/2.0
Via: SIP/2.0/UDP sccas1.home1.net:5060;branch=z9hG4bKnas588339
Max-Forwards: 70
From: <sip:sccas1.home1.net>;tag=aassd
To: sip:msc123.visited2.net
Call-ID: sdvasdfgfasdfsdfwefw
CSeq: 44561 MESSAGE
Content-Length: ...
P-Asserted-Identity: sip:sccas1.home1.net
Accept-Contact: *;g.3gpp.path="<sip:termpdfjkghlj@msc123.visited2.net;lr>";explicit;require
Content-Type: application/vnd.3gpp.srvcc-ext+xml


<?xml version="1.0"?>
<srvcc-ext>
  <PS-reg-info Path="sip:termpdfjkghlj@msc123.visited2.net;lr">
    <ATCF-Management-URI>sip:atcf.visited2.net</ATCF-Management-URI>
    <C-MSISDN>tel:+1-237-555-1111</C-MSISDN>
    <cs2ps-srvcc-alerting>true</cs2ps-srvcc-alerting>

  </PS-reg-info>
</srvcc-ext>
```

Request-URI: public user identity registered by the MSC server as provided in the step 2

P-Asserted-Identity: SCC AS URI

Accept-Contact: g.3gpp.path media feature tag containing the MSC URI for terminating requests provided in the SIP REGISTER request in the step 2.

body: CS to PS SRVCC information informing the MSC server about the ATCF used in the registration path of the UE over a PS domain. Path attribute contains the MSC URI for terminating requests provided in the SIP REGISTER request in the step 2. ATCF-Management-URI element contains the ATCF management URI of the ATCF, the C-MSISDN element contains the C-MSISDN of the UE, and the cs2ps-srvcc-alerting element shows the support of CS to PS SRVCC in alerting phase.

5.-6. SIP 200 (OK) response (MSC server to SCC AS)

The MSC server generates the SIP 200 (OK) response to the SIP MESSAGE request.

A.4 Signalling flows for call origination for service continuity

A.4.1 Session origination for CS calls

An example flow for session origination for CS calls can be found in 3GPP TS 24.292 [4].

A.4.2 Session origination with PS to CS SRVCC enhancements

The signalling flow shown in figure A.4.2-1 gives an example of originating session set up when ATCF anchors the media of the session. This flow assumes that ATCF was invoked during registration.

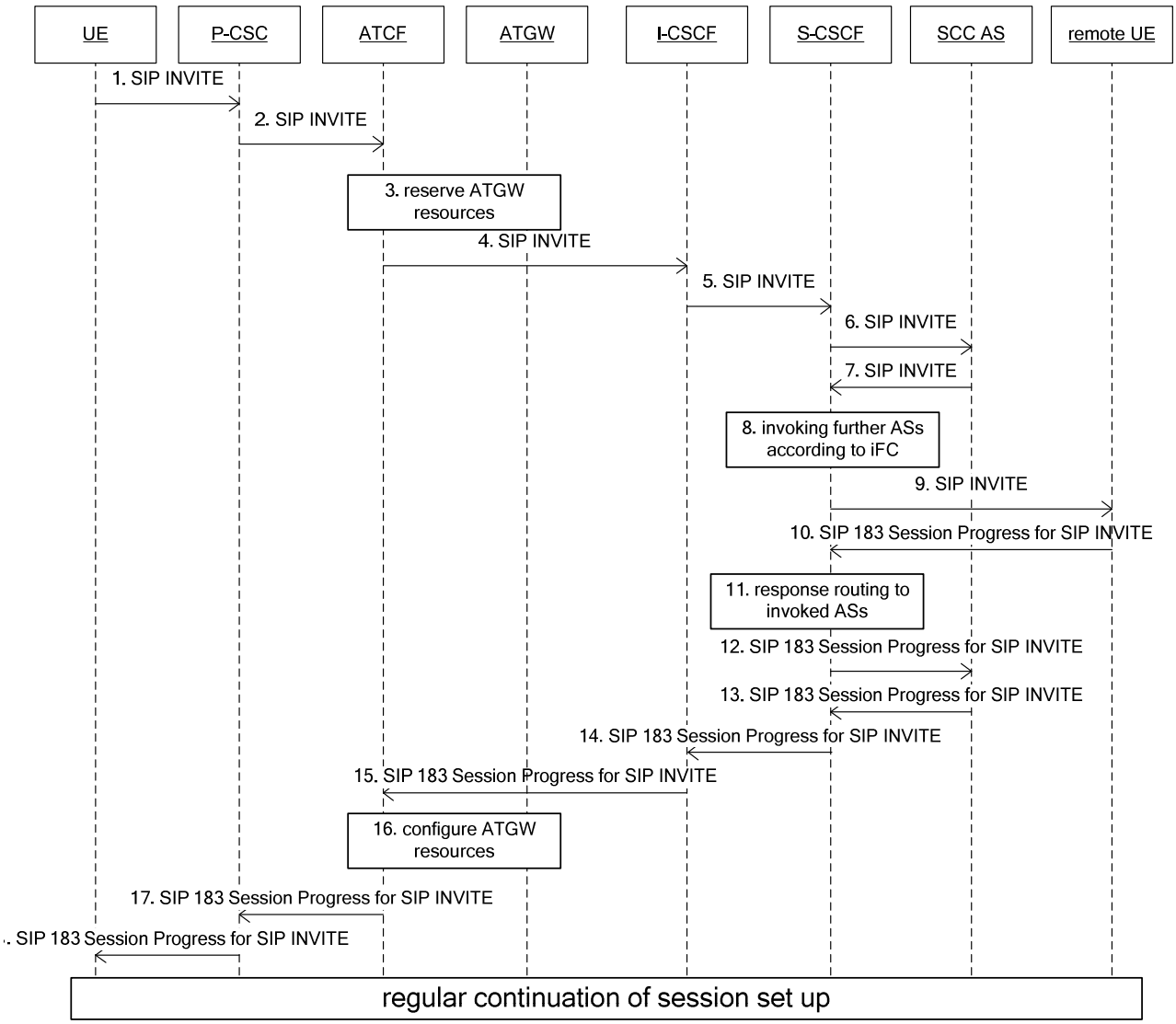


Figure A.4.2-1 Signalling flows for service continuity using PS to CS SRVCC enhancements

1. **SIP INVITE request (UE to P-CSCF) - see example in table A.4.2-1**

Table A.4.2-1: SIP INVITE request (UE to P-CSCF)

```
INVITE tel:+1-212-555-2222 SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Max-Forwards: 70
Route: <sip:pcscf1.visited2.net:7531;lr;comp=sigcomp>, <sip:orig@scscf1.home1.net;lr>
P-Preferred-Identity: "John Doe" <sip:user1_public1@home1.net>
P-Preferred-Service: urn:urn-7:3gpp-service.ims.icsi.mmtel
P-Access-Network-Info: 3GPP-UTRAN-TDD; utran-cell-id-3gpp=234151D0FCE11
Privacy: none
From: <sip:user1_public1@home1.net>;tag=171828
To: <tel:+1-212-555-2222>
Call-ID: cb03a0s09a2sdfglkj490333
Cseq: 127 INVITE
Require: sec-agree
Supported: precondition, 100rel, gruu
Proxy-Require: sec-agree
Security-Verify: ipsec-3gpp; q=0.1; alg=hmac-sha-1-96; spi-c=98765432; spi-s=87654321; port-
   c=8642; port-s=7531
Contact: <sip:user1_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-
   00a0c91e6bf6;comp=sigcomp>;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
Accept-Contact: *;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c=IN IP6 5555::aaa:bbb:ccc:ddd
t=0 0
m=audio 3456 RTP/AVP 97 96
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 telephone-event
```

2. SIP INVITE request (P-CSCF to ATCF) - see example in table A.4.2-2

Since a Feature-Caps header field with the g.3gpp.atcf feature-capability indicator was included in SIP 2xx response to the SIP REGISTER request which created the binding of the contact address using which the SIP INVITE request is sent, the P-CSCF routes the SIP INVITE request to the ATCF.

Table A.4.2-2: SIP INVITE request (P-CSCF to ATCF)

```
INVITE tel:+1-212-555-2222 SIP/2.0
Record-Route: <sip:pcscf1.visited1.net;lr>
Via: SIP/2.0/UDP pcscf1.visited2.net:5060;branch=z9hG4bKnas56565, SIP/2.0/UDP
   [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Max-Forwards: 69
Route: <sip:orig@atcf.visited2.net;lr>, <sip:orig@scscf1.home1.net;lr>
P-Asserted-Identity: "John Doe" <sip:user1_public1@home1.net>
P-Preferred-Service:
P-Access-Network-Info:
Privacy:
From:
To:
Call-ID:
Cseq:
Require:
Supported:
Proxy-Require:
Contact:
Accept-Contact
Allow:
Content-Type:
Content-Length:

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c=IN IP6 5555::aaa:bbb:ccc:ddd
t=0 0
m=audio 3456 RTP/AVP 97 96
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 telephone-event
```

Route: ATCF URI for originating requests (as configured in P-CSCF) followed by the remaining Route header fields determined by P-CSCF.

3. ATGW resource reservation

The ATCF decides to anchor the media of the session and reserves the resources in the ATGW.

4-9. SIP INVITE request (ATCF towards remote UE) - see example in table A.4.2-4

The ATCF modifies the SDP offer without changing the dialog identifier and forwards the SIP INVITE request. The ATCF replaces the IP address, ports, ... with values provided by ATGW.

Table A.4.2-4: SIP INVITE request (ATCF towards remote UE)

```
INVITE tel:+1-212-555-2222 SIP/2.0
Record-Route: <sip:pcscf1.visited1.net;lr>, <sip:atcf.visited.net;lr>
Via: SIP/2.0/UDP atcf.visited2.net:5060;branch=z9hG4bKnas55889, SIP/2.0/UDP
   pcscf1.visited2.net:5060;branch=z9hG4bKnas56565, SIP/2.0/UDP
   [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Max-Forwards: 68
Route: <sip:orig@scscf1.home1.net;lr>
P-Asserted-Identity:
P-Preferred-Service:
P-Access-Network-Info:
Privacy:
From:
To:
Call-ID:
Cseq:
Require:
Supported:
Proxy-Require:
Contact:
Accept-Contact
Allow:
Content-Type:
Content-Length:

v=0
o=- 22 333 IN IP6 8888::111:222:333:444
s=-
c=IN IP6 8888::111:222:333:444
t=0 0
m=audio 8899 RTP/AVP 97 96
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 telephone-event
```

SDP offer: the IP address and ports are updated to contain the values provided by ATGW .

10-12. SIP 183 (Session Progress) response (remote UE towards SCC AS)

The remote UE responds with SIP 183 (Session progress) response.

13.-15.SIP 183 (Session Progress) response (SCC AS towards ATCF) - see example in table A.4.2-13

The SCC AS forwards the SIP 183 (Session progress) response.

Table A.4.2-13: SIP 183 (Session Progress) response (SCC AS towards ATCF)

```
SIP/2.0 183 Session Progress
Feature-Caps: *;+g.3gpp.srvcc
Record-Route: <sip:pcscf1.visited1.net;lr>, <sip:atcf.visited.net;lr>,
   <sip:scscf.home1.net;lr>, <sip:icscf.home1.net;lr>, <sip:sccas.home1.net;lr>
Via: SIP/2.0/UDP sccas.home1.net:5060;branch=z9hG4bKnas522, SIP/2.0/UDP
   scscf.home1.net:5060;branch=z9hG4bKnas889, SIP/2.0/UDP
   icscf.home1.net:5060;branch=z9hG4bKnas225, SIP/2.0/UDP
   atcf.visited2.net:5060;branch=z9hG4bKnas55889, SIP/2.0/UDP
   pcscf1.visited2.net:5060;branch=z9hG4bKnas56565, SIP/2.0/UDP
   [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Max-Forwards: 60
P-Asserted-Identity: <tel:+1-212-555-2222>
Privacy:
From:
To: <tel:+1-212-555-2222>; tag=aaa
Call-ID:
Cseq:
Require:
Supported:
Contact: <sip:user2_public1@home2.net;gr=urn:uuid:2ad8950e-48a5-4a74-8d99-
   ad76cc7fc74>;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
Allow:
Content-Type:
Content-Length:

v=0
o=- 462346 5654 IN IP6 1234::55:66:77:88
s=-
c=IN IP6 1234::55:66:77:88
t=0 0
m=audio 4456 RTP/AVP 97 96
b=AS:25.4
a=curr:qos local none
a=curr:qos remote sendrecv
a=des:qos mandatory local sendrecv
a=des:qos mandatory remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 telephone-event
```

Supported: The SCC AS adds the "tdialog" and the "replaces" option tags in the Supported header field header, if not already included. In this example the "tdialog" and the "replaces" option tags were already included.

16. ATGW resource configuration

The ATCF configures the resources of ATGW.

17. SIP 183 (Session Progress) response (ATCF towards UE) - see example in table A.4.2-17

The ATCF replaces the IP address, ports, ... in SDP answer with values provided by ATGW.

Table A.4.2-17: SIP 183 (Session Progress) response (ATCF towards UE)

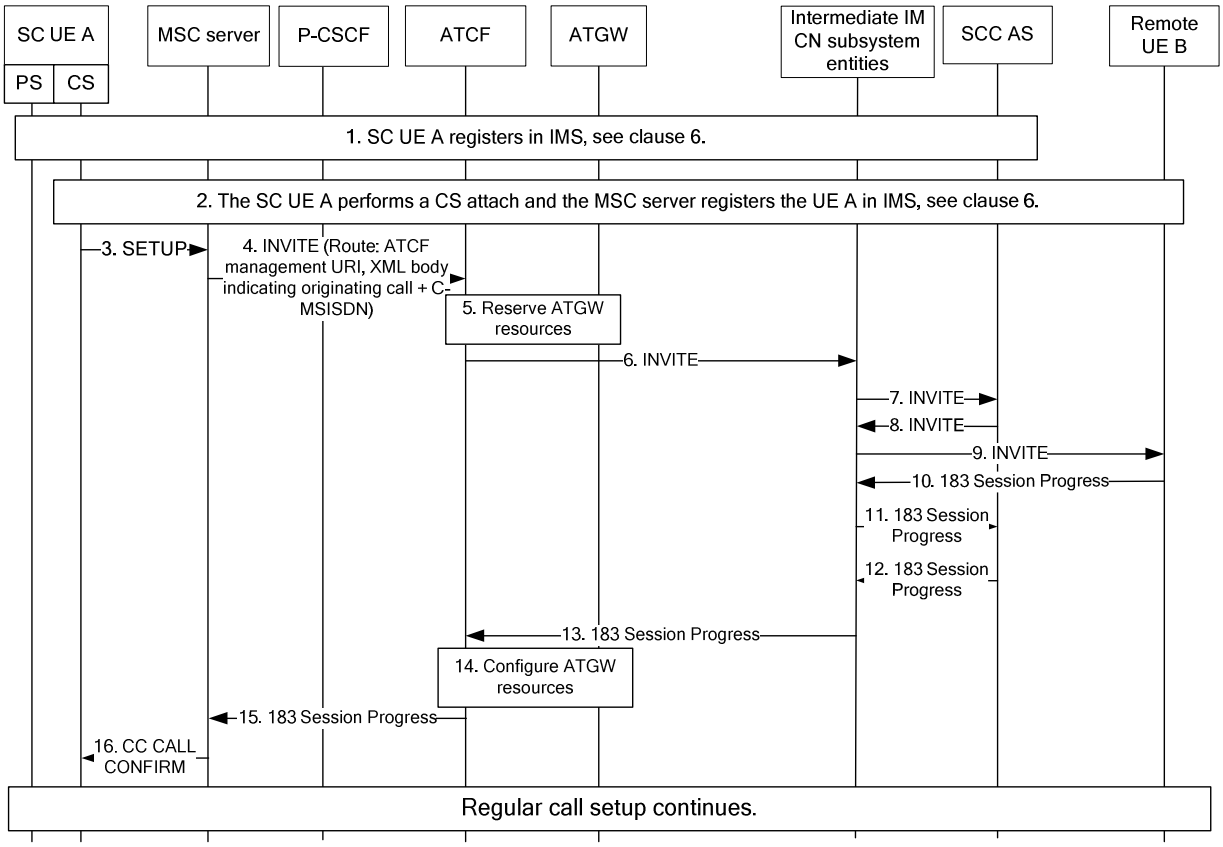
```
SIP/2.0 183 Session Progress
Feature-Caps: *;+g.3gpp.srvcc
Record-Route: <sip:pcscf1.visited1.net;lr>, <sip:atcf.visited.net;lr>,
   <sip:scscf.home1.net;lr>, <sip:icscf.home1.net;lr>, <sip:sccas.home1.net;lr>
Via: SIP/2.0/UDP pcscf1.visited2.net:5060;branch=z9hG4bKnas56565, SIP/2.0/UDP
   [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Max-Forwards: 60
P-Asserted-Identity: <tel:+1-212-555-2222>
Privacy:
From:
To:
Call-ID:
Cseq:
Require:
Supported:
Contact: <sip:user2_public1@home2.net;gr=urn:uuid:2ad8950e-48a5-4a74-8d99-
   ad76cc7fc74>;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
Allow:
Content-Type:
Content-Length:

v=0
o=- 44 555 IN IP6 8888::111:222:333:444
s=-
c=IN IP6 8888::111:222:333:444
t=0 0
m=audio 11234 RTP/AVP 97 96
b=AS:25.4
a=curr:qos local none
a=curr:qos remote sendrecv
a=des:qos mandatory local sendrecv
a=des:qos mandatory remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 telephone-event
```

SDP answer: the IP address and ports are updated to contain the values provided by ATGW.

A.4.3 Call origination prior to CS to PS SRVCC

The signalling flow shown in figure A.4.3-1 gives an example of originating session set up. In this flow, the ATCF decides to anchor the media of the session in ATGW.



NOTE: For clarity, the SIP 100 (Trying) responses are not shown in the signalling flow.

Figure A.4.3-1 Signalling flows for service continuity using CS to PS SRVCC.

1. **SC UE A registers in IMS**

   The SC UE A registers in IMS.

2. **The SC UE A performs a CS attach and the MSC server registers in IMS.**

3. **CC SETUP message (SC UE A to MSC server).**

   The SC UE sends a CC SETUP message according to 3GPP TS 24.008 [8].

4. **SIP INVITE request (MSC server to ATCF) - see example in table A.4.3-4**

   The MSC server enhanced for ICS sends the SIP INVITE request towards the ATCF.

Table A.4.3-4: SIP INVITE request (MSC server to ATCF)

```
INVITE tel:+1-237-555-2222 SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]:1357;branch=z9hG4bKnashds7
Max-Forwards: 70
Route: <sip:atcf2.visited2.net:7531;lr><sip:orig@scscf1.home1.net;lr>
P-Asserted-Identity: tel:+1-212-555-1111
P-Charging-Vector: icid-value="1234bc9876e";icid-generated-at"5555::aaa:bbb:ccc:ddd";orig-
   ioi=visited2.net
Privacy: none
From: <tel:+1-237-555-1111>;tag=171828
To: < tel:+1-237-555-2222 >
P-Access-Network-Info: 3GPP-UTRAN-TDD; utran-cell-id-3gpp=234151D0FCE11
P-Visited-Network-ID: "Visited Network Number 1 for MSC Server"
Call-ID: cb03a0s09a2sdfglkj490333
Cseq: 127 INVITE
Supported: precondition, 100rel
Accept: application/vnd.3gpp.access-transfer-events+xml;et="2"
Recv-Info: g.3gpp.access-transfer-events
Contact: <sip:user1_public1@visited2.net>;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-
   service.ims.icsi.mmtel";+g.3gpp.ti="F0CA"
Accept-Contact: *;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE
Content-Type: multipart/mixed;boundary="boundary1"
Content-Length: (…)

--boundary1
Content-Type: application/sdp

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c=IN IP6 5555::aaa:bbb:ccc:ddd
t=0 0
m=audio 3456 RTP/AVP 97 96
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 telephone-event

--boundary1
Content-Type: application/vnd.3gpp.srvcc-ext+xml

<?xml version="1.0"?>
<srvcc-ext>
<Setup-info>
    <C-MSISDN>tel:+1-212-555-1111</C-MSISDN>
    <direction>initiator</direction>
</Setup-info>
</srvcc-ext>
--boundary1--
```

Route: The ATCF management URI received from SCC AS in the SIP MESSAGE request containing CS to PS SRVCC information during the registration of the user is added at the top of the URIs received in the Service-Route header field of the SIP 200 (OK) response to REGISTER.

application/vnd.3gpp.srvcc-ext+xml: Contains the direction of call and the C-MSISDN of the UE.

Accept: Indicate that the MSC server is able to receive the application/vnd.3gpp.access-transfer-events+xml with the event type 2.

Recv-Info: Indicate the support for g.3gpp.access-transfer-events package.

Contact: g.3gpp.ti media feature tag with value containing the transaction identifier specified in figure 11.9 and table 11.3 of 3GPP TS 24.007 [75] encoded by hexadecimal digit. In this example, the transaction identifier 74 (decimal) and the transaction identifier flag as sent by the MSC server in CS signalling of the originating CS call are shown.

5. **ATCF decides whether to anchor the media in the ATGW. In this flow, the ATCF decides to anchor the media in the ATGW and reserves the resources in the ATGW.**

6-9. SIP INVITE request (ATCF to remote UE B) - see example in table A.4.3-6

Table A.4.3-6: SIP INVITE request (ATCF to remote UE B)

```
INVITE tel:+1-237-555-2222 SIP/2.0
Record-Route: <sip:atcf2.visited2.net;lr>
Via: SIP/2.0/UDP atcf.visited2.net:5060;branch=z9hG4bKnas55889, SIP/2.0/UDP
   [5555::aaa:bbb:ccc:ddd]:1357;;branch=z9hG4bKnashds7
Max-Forwards:
Route: <sip:orig@scscf1.home1.net;lr>
P-Asserted-Identity:
P-Access-Network-Info:
P-Charging-Vector:
Privacy:
From:
To:
Call-ID:
Cseq:
Require:
Supported:
Proxy-Require:
Contact:
Accept-Contact
Allow:
Content-Type:
Content-Length:

v=0
o=- 22 333 IN IP6 8888::111:222:333:444
s=-
c=IN IP6 8888::111:222:333:444
t=0 0
m=audio 8899 RTP/AVP 97 96
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 telephone-event
```

SDP offer: The IP address and ports are updated to contain the values provided by ATGW.

10-11. SIP 183 (Session Progress) response (remote UE B to SCC AS)

The remote UE B responds with SIP 183 (Session progress) response.

12-13. SIP 183 (Session Progress) response (SCC AS to ATCF) - see example in table A.4.3-13

Table A.4.3-13: SIP 183 (Session Progress) response (SCC AS towards ATCF)

```
SIP/2.0 183 Session Progress
Record-Route: <sip:atcf2.visited2.net;lr>,<sip:icscf1.home1.net;lr>,<sip:scscf1.home1.net;lr>,
   <sip:sccas1.home1.net;lr>;
Via: SIP/2.0/UDP scscf1.home1.net:5060;branch=z9hG4bKnas889, SIP/2.0/UDP
   atcf.visited2.net:5060;branch=z9hG4bKnas55889, SIP/2.0/UDP
   [5555::aaa:bbb:ccc:ddd]:1357;branch=z9hG4bKnashds7
Max-Forwards:
P-Asserted-Identity: <tel:+1-237-555-2222>
P-Charging-Vector: icid-value="1234bc9876e@5555::aaa:bbb:ccc:ddd";orig-ioi= visited2.net
Privacy:
From:
To: <tel:+1-237-555-2222>; tag=aaa
Call-ID:
Cseq:
Require:
Supported:
Contact: <sip:user2_public1@home2.net;gr=urn:uuid:2ad8950e-48a5-4a74-8d99-
   ad76cc7fc74>;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
Allow:
Content-Type:
Content-Length:
Feature-Caps: *;+g.3gpp.srvcc

v=0
o=- 462346 5654 IN IP6 1234::55:66:77:88
s=-
c=IN IP6 1234::55:66:77:88
t=0 0
m=audio 4456 RTP/AVP 97 96
b=AS:25.4
a=curr:qos local none
a=curr:qos remote sendrecv
a=des:qos mandatory local sendrecv
a=des:qos mandatory remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 telephone-event
```

14. Configure ATGW resources.

The ATCF configures the resources of ATGW.

15. SIP 183 (Session Progress) response (ATCF to MSC server) - see example in table A.4.3-15

Table A.4.3-15: SIP 183 (Session Progress) response (ATCF to MSC server)

```
SIP/2.0 183 Session Progress
Record-Route: <sip:atcf2.visited2.net;lr>, <sip:scscf1.home1.net;lr>,
   <sip:icscf1.home1.net;lr>, <sip:sccas1.home1.net;lr>
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]:1357;
Max-Forwards:
P-Asserted-Identity:
P-Charging-Vector:
Privacy:
From:
To:
Call-ID:
Cseq:
Require:
Supported:
Recv-Info: g.3gpp.access-transfer-events;et="1,3,4"
Contact: <sip:user2_public1@home2.net;gr=urn:uuid:2ad8950e-48a5-4a74-8d99-
   ad76cc7fc74>;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
Allow:
Content-Type:
Content-Length:

v=0
o=- 44 555 IN IP6 8888::111:222:333:444
s=-
c=IN IP6 8888::111:222:333:444
t=0 0
m=audio 11234 RTP/AVP 97 96
b=AS:25.4
a=curr:qos local none
a=curr:qos remote sendrecv
a=des:qos mandatory local sendrecv
a=des:qos mandatory remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 telephone-event
```

SDP answer: the IP address and ports are updated to contain the values provided by ATGW.

Recv-Info: Indicates the support for the info package g.3gpp.access-transfer-events and is able to receive the event types 1, 3 and 4.

16. CC CALL CONFIRM message (MSC server to SC UE A)

Regular call setup continues.

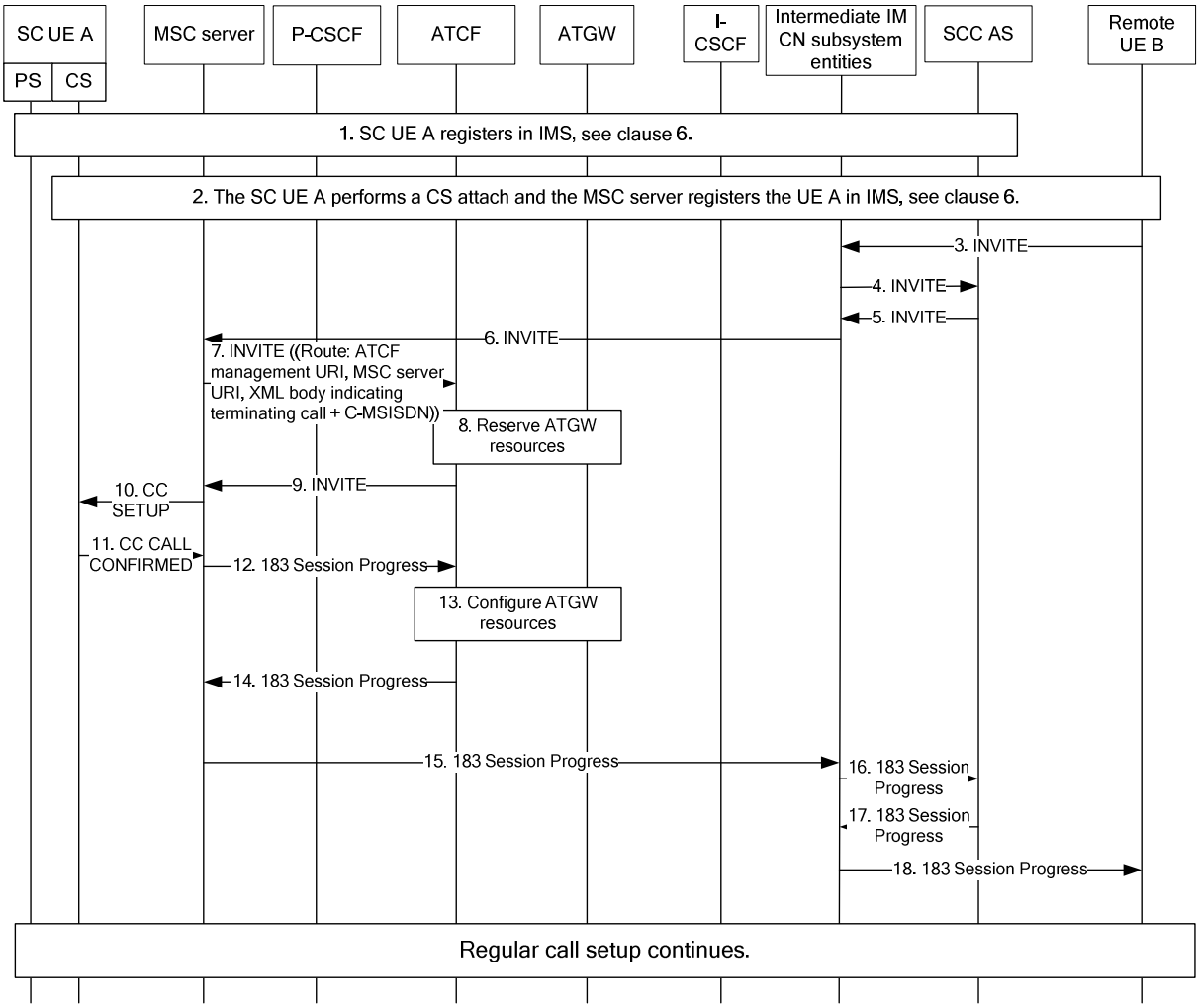
A.5 Signalling flows for call termination for service continuity

A.5.1 Session termination using CS media

An example flow for session termination using CS calls can be found in 3GPP TS 24.292 [4].

A.5.2 Call termination prior to CS to PS SRVCC

The signalling flow shown in figure A.5.2-1 gives an example of a terminating session set up. In this flow, the ATCF anchors the media of the session in ATGW.



NOTE: For clarity, the SIP 100 (Trying) responses are not shown in the signalling flow.

Figure A.5.2-1 Signalling flows for service continuity using CS to PS SRVCC.

1. SC UE A registers in IMS.

2. The SC UE A performs a CS attach and the MSC server registers in IMS.

3-4. SIP INVITE request (Remote UE B to SCC AS)

The remote UE sends a SIP INVITE request towards the user at SC UE A.

5-6. SIP INVITE request (SCC AS to MSC server) - see example in table A.5.2-5

The SCC AS forwards the SIP INVITE request towards the MSC server.

Table A.5.2-5: SIP INVITE request (SCC AS to MSC server)

```
INVITE tel:+1-237-555-1111 SIP/2.0
Via: SIP/2.0/UDP sccas1.home1.net;branch=z9hG4bK871y12.1
Max-Forwards: 70
Route: <sip:scscf1.home1.net;lr>,<sip:msc2.visited2.net;lr>
Record-Route: <sip:sccas1.home1.net;lr>
P-Asserted-Identity: "John Doe" <sip:user1_public2@visited2.net>, <tel:+1-237-555-2222>
P-Asserted-Service: urn:urn-7:3gpp-service.ims.icsi.mmtel
Accept-Contact: *;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
Privacy: none
From: <tel:+1-237-555-2222>;tag=171828
To: tel:+1-237-555-1111
P-Charging-Vector: icid-value="1234bc9876e";icid-generated-at=5555::aaa:bbb:ccc:eeee";orig-
   ioi=visited2.net
Call-ID: cb03a0s09a2sdfglkj490333
Cseq: 127 INVITE
Supported: 100rel, precondition, gruu, 199
Accept: applicatiom/sdp,application/3gpp-ims+xml
Contact: <sip:user2_public1@visited2.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-
   00a0c91e6bf6>;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel">
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE
Feature-Caps: *;+g.3gpp.srvcc
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c=IN IP6 5555::aaa:bbb:ccc:ddd
t=0 0
m=video 3400 RTP/AVP 98 99
b=AS:75
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=inactive
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
a=rtpmap:99 MP4V-ES
m=audio 3456 RTP/AVP 97 0 96
b=AS:25.4
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:96 telephone-event
a=maxptime:20
```

7. **INVITE request (MSC server to ATCF) - see example in table A.5.2-7**

Table A.5.2-7: SIP INVITE request (MSC server to ATCF)

```
INVITE tel:+1-237-555-1111 SIP/2.0
Via: SIP/2.0/UDP msc2.visited2.net;branch=z9hG4bK879l11.1,SIP/2.0/UDP
   scscf1.home1.net;branch=z9hG4bK871z34.1,SIP/2.0/UDP
   sccas1_s.home1.net;branch=z9hG4bK871y12.1
Max-Forwards: 70
Route: <sip:atcf2.visited2.net;lr>,<sip:msc2.visited2.net;lr>
Record-Route:
   <sip:atcf2.visited2.net;lr>,<sip:msc2.visited2.net;lr>,<sip:scscf1.home1.net;lr>,<sip:sccas
   1.home1.net;lr>
P-Asserted-Identity: "John Doe" <sip:user1_public2@visited2.net>, <tel:+1-237-555-2222>
P-Asserted-Service: urn:urn-7:3gpp-service.ims.icsi.mmtel
Accept-Contact: *;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
Privacy: none
From: <sip:user2_public1@visited2.net>;tag=171828
To: tel:+1-237-555-1111
Call-ID: cb03a0s09a2sdfglkj490333
Cseq: 127 INVITE
Supported: 100rel, precondition, gruu, 199
Accept: applicatiom/sdp,application/3gpp-ims+xml
Contact: <sip:user2_public1@visited2.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-
   00a0c91e6bf6>;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel">;g.3gpp.rsrvcc
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE
P-Charging-Vector: icid-value="1234bc9876e";icid-generated-at=5555::aaa:bbb:ccc:eeee";orig-
   ioi=visited2.net
Feature-Caps: *;+g.3gpp.srvcc
Content-Type: multipart/mixed;boundary="boundary1"
Content-Length: (…)

--boundary1
Content-Type: application/sdp

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c=IN IP6 5555::aaa:bbb:ccc:ddd
t=0 0
m=video 3400 RTP/AVP 98 99
b=AS:75
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=inactive
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
a=rtpmap:99 MP4V-ES
m=audio 3456 RTP/AVP 97 0 96
b=AS:25.4
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:96 telephone-event
a=maxptime:20

--boundary1
Content-Type: application/vnd.3gpp.srvcc-ext+xml

<?xml version="1.0"?>
<srvcc-ext>
<Setup-info>
    <C-MSISDN>tel:+1-212-555-1111</C-MSISDN>
    <direction>receiver</direction>
</Setup-info>
</srvcc-ext>
--boundary1--
```

Route: ATCF management URI received from SCC AS in the SIP MESSAGE request containing CS to PS SRVCC information during the registration of the user followed by MSC server URI.

application/vnd.3gpp.srvcc-ext+xml: Contains the direction of call and the C-MSISDN of the UE.

8. **ATCF decides whether to anchor the media in the ATGW. In this flow, the ATCF decides to anchor the media in the ATGW and reserves the resources in the ATGW.**

9. **SIP INVITE request (ATCF to MSC server) - see example in table A.5.2-9**

   The ATCF forwards the SIP INVITE request to MSC server according to the received Route header field.

Table A.5.2-9: SIP INVITE request (ATCF to MSC server)

```
INVITE tel:+1-237-555-1111 SIP/2.0
Via: SIP/2.0/UDP atcf2.visited2.net;branch=z9hG4bK871x99ja,SIP/2.0/UDP
   msc2.visited2.net;branch=z9hG4bK879l11.1,SIP/2.0/UDP
   scscf1.home1.net;branch=z9hG4bK871z34.1,SIP/2.0/UDP sccas1.home1.net;branch=z9hG4bK871y12.1
Max-Forwards: 68
Route:<sip:msc2.visited2.net;lr>
Record-
   Route:<sip:atcf2.visited2.net:7531;lr>,<sip:scscf1.visited2.net;lr>,<sccas1.home1.net;lr>
P-Asserted-Identity:
P-Charging-Vector: icid-value="1234bc9876e";icid-generated-at=5555::aaa:bbb:ccc:eeee";orig-
   ioi=visited2.net
P-Asserted-Service: urn:urn-7:3gpp-service.ims.icsi.mmtel
Accept-Contact: *;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
Privacy:
From:
To:tel:
Call-ID: cb03a0s09a2sdfglkj490333
Cseq: 127 INVITE
Supported:
Accept: applicatiom/sdp,application/3gpp-ims+xml,application/3gpp.access-transfer-
   events+xml;et="1,3,4"
Recv-Info: g.3gpp.access-transfer-events
Contact: sip:<+12375551111 @msc2.visited2.net;user=phone>;+g.3gpp.icsi-ref="urn%3Aurn-
   7%3A3gpp-service.ims.icsi.mmtel">
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE
Feature-Caps: *;+g.3gpp.srvcc
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933615 2987939999 IN IP6 8888::111:222:333:444
s=-
c=IN IP6 8888::111:222:333:444
t=0 0
m=video 3400 RTP/AVP 98 99
b=AS:75
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=inactive
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
a=rtpmap:99 MP4V-ES
m=audio 3456 RTP/AVP 97 0 96
b=AS:25.4
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:96 telephone-event
a=maxptime:20
```

SDP offer: the IP address and ports are updated to contain the values provided by ATGW.

Accept: Indicates that the ATCF is able to receive the application/3gpp.access-transfer-events+xml with event types 1, 3 and 4.

Recv-Info: Indicates that the ATCF support receiving the g.3gpp.access-transfer-events info package.

10. **CC SETUP message (MSC server to UE A)**

The MSC server sends a CC SETUP message. The mapping of the INVITE request to the CC SETUP message is described by 3GPP TS 29.292 [18].

11. CC CALL CONFIRM message (UE A to MSC server)

The UE sends a CC CALL CONFIRM message in accordance to 3GPP TS 24.008 [8].

12. SIP 183 (Session Progress) response (MSC server to ATCF) - see example in table A.5.2-12

The MSC server sends a SIP 183 (Session Progress) response. The CC CONFIRMED message is mapped to the SIP 183 (Session Progress) response as described in 3GPP TS 29.292 [18].

Table A.5.2-12: SIP 183 (Session Progress) response (MSC server to ATCF)

```
SIP/2.0 183 Session Progress
Via:
Route: sip:atcf2.visited2.net:7531;lr;>; <sip:scscf1.visited2.net;lr>>
Record-Route: <sip:scscf1.visited2.net;lr>,<sip:atcf2.visited2.net:7531;lr;>
P-Asserted-Identity: tel:<+12375551111>
P-Called-Party-ID:tel:<+12375551111>
P-Charging-Vector: icid-value="1234bc9876e";icid-generated-at=5555::aaa:bbb:ccc:eeee";orig-
   ioi=visited2.net
From:
To: <tel:+1-237-555-2222>;tag=314159
Call-ID:
Cseq:
Require: 100rel
Contact: <sip:user1_public1@visited2.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-
   00a0c91e6bf6>;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-
   service.ims.icsi.mmtel";+g.3gpp.ti="70D8"
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER
Rseq: 9021
Content-Type: application/sdp
Content-Length: (…)
Recv-Info: g.3gpp.access-transfer-events;et="2"

v=0
o=- 2987933615 2987939999 IN IP6 8888::111:222:333:666
s=-
c=IN IP6 8888::111:222:333:666
t=0 0
m=video 0 RTP/AVP 98 99
m=audio 6544 RTP/AVP 97 96
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des: qos mandatory remote sendrecv
a=conf:qos remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:96 telephone-event
a=maxptime:20
```

Recv-Info: Indicates that the ATCF support receiving the g.3gpp.access-transfer-events info package with event type 2.

Contact: g.3gpp.ti media feature tag with value containing the transaction identifier specified in figure 11.9 and table 11.3 of 3GPP TS 24.007 [75] encoded by hexadecimal digit. In this example, the transaction identifier 88 (decimal) and the transaction identifier flag as sent by the MSC server in CS signalling of the terminating CS call are shown.

13. Configure ATGW resources

The ATCF configures resources in the ATGW.

14. SIP 183 (Session Progress) response (ATCF to MSC server) - see example in table A.5.2-14

Table A.5.2-14: SIP 183 (Session Progress) response (ATCF to MSC server)

```
SIP/2.0 183 Session Progress
Via:
Route: sip:atcf2.visited2.net:7531;lr;>; <sip:scscf1.home1.net;lr>;g.3gpp.srvcc me
Record-Route: <sip:scscf1.home1.net;lr>,<sip:atcf2.visited2.net:7531;lr;>
P-Asserted-Identity: <tel:+1-237-555-1111>
P-Called-Party-ID: <tel:+1-237-555-1111>
P-Charging-Vector: icid-value="1234bc9876e";icid-generated-at=5555::aaa:bbb:ccc:eeee";orig-
   ioi=visited2.net
From:
To: <tel:+1-237-555-2222>;tag=314159
Call-ID:
Cseq:
Require: 100rel
Contact:
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER
Rseq: 9021
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933615 2987332299 IN IP6 8888::111:222:333:446
s=-
c=IN IP6 8888::111:222:333:446
t=0 0
m=video 0 RTP/AVP 98 99
m=audio 53261 RTP/AVP 97 96
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des: qos mandatory remote sendrecv
a=conf:qos remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:96 telephone-event
a=maxptime:20
```

SDP answer: the IP address and ports are updated to contain the values provided by ATGW.

15-16. SIP 183 (Session Progress) response (MSC server to SCC AS) - see example in table A.5.2-15

Table A.5.2-15: SIP 183 (Session Progress) response (ATCF to SCC AS)

```
SIP/2.0 183 Session Progress
Via:
Route: sip:atcf2.visited2.net:7531;lr;>; <sip:scscf1.home1.net;lr>;g.3gpp.srvcc me
Record-Route: <sip:scscf1.home1.net;lr>,<sip:atcf2.visited2.net:7531;lr;>
P-Asserted-Identity: <tel:+1-237-555-1111>
P-Called-Party-ID:tel: <tel:+1-237-555-1111>
P-Charging-Vector: P-Charging-Vector: icid-value="1234bc9876e";icid-generated-
   at=5555::aaa:bbb:ccc:eeee";orig-ioi=visited2.net
From:
To: <tel:+1-237-555-2222>;tag=314159
Call-ID:
Cseq:
Require: 100rel
Contact:
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER
Rseq: 9021
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933615 2987332299 IN IP6 8888::111:222:333:446
s=-
c=IN IP6 8888::111:222:333:446
t=0 0
m=video 0 RTP/AVP 98 99
m=audio 53261 RTP/AVP 97 96
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des: qos mandatory remote sendrecv
a=conf:qos remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:96 telephone-event
a=maxptime:20
```

SDP answer: the IP address and ports are the values provided by ATGW.

16-17. SIP 183 (Session Progress) response (SCC AS to remote UE)

The SCC AS sends the SIP 183 (Session Progress) response towards the remote UE.

Regular call setup continues.

A.6 Signalling flows for PS-CS access transfer

A.6.1 PS-CS access transfer: CS-PS

In this example, SC UE A has an ongoing session with remote UE B over CS bearer before access transfer. When SC UE connects to an IP-CAN, it decides to transfer the session over the new IP-CAN.

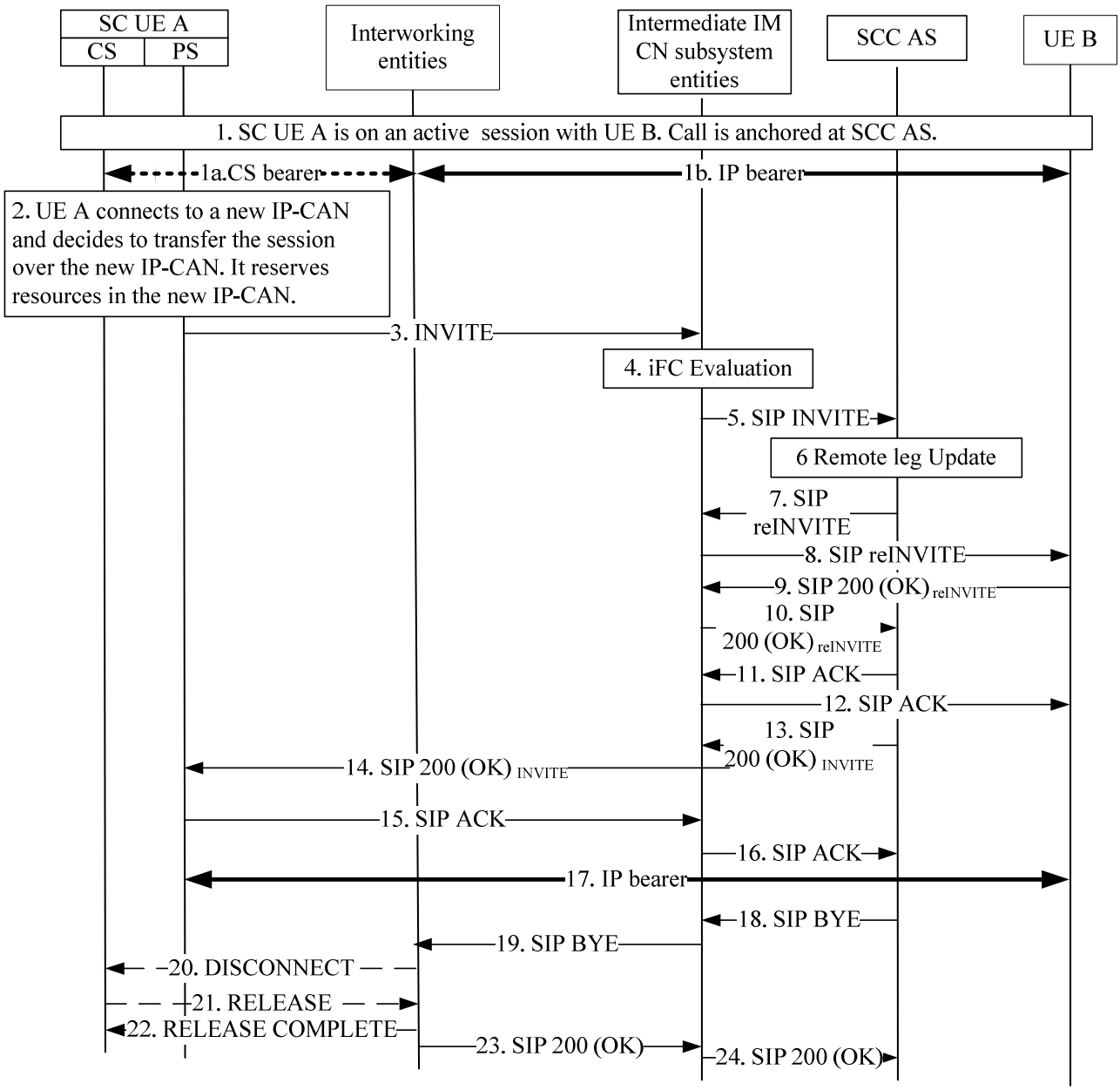


Figure A.6.1-1: Signalling flow for PS-CS Access Transfer: CS to PS

NOTE 1: For clarity, the SIP 100 (Trying) responses are not shown in the signalling flow.

1. **SC UE A has an ongoing session with remote UE B**

   The call has been anchored at the SCC AS which is in the HPLMN of originating SC UE A.

2. **SC UE A connects to a new IP-CAN:**

   The SC UE A decides to transfer the session over the new IP-CAN. The UE A obtains an IP address that it will use for the signalling and media. It registers with the S-CSCF over the new IP-CAN using standard registration procedure and reserves resources in the new IP-CAN.

3. **SIP INVITE request (SC UE A to intermediate IM CN subsystem entities) - see example in table A.6.1-3**

   The SC UE A sends an initial SIP INVITE request to request the new call replaces the existing call.

Table A.6.1-3: SIP INVITE request (UE A to intermediate IM CN subsystem entities)

```
INVITE sip:domain.xfer@sccas.home1.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Max-Forwards: 70
Route: <sip:pcscf1.home1.net:7531;lr >, <sip:orig@scscf1.home1.net;lr>
P-Preferred-Identity: "John Doe" <sip:user1_public1@home1.net>
P-Access-Network-Info: IEEE-802.11b
Privacy: none
From: <sip:user1_public1@home1.net>; tag=171828
To: <tel:+1-237-555-2222>
Call-ID: cb03a0s09a2sdfglkj490237
Cseq: 127 INVITE
Supported: 100rel; precondition
Require: sec-agree
Proxy-Require: sec-agree
Security-Verify: ipsec-3gpp; q=0.1; alg=hmac-sha-1-96; spi=87654321; port1=7531
Contact: <sip:user1_public1@home1.net;gr= urn:uuid:f81d4fae-7dec-11d0-a765-
   00a0c91e6bf6>;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-
   service.ims.icsi.mmtel";+g.3gpp.ics="principal";
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE
Accept: application/sdp; application/3gpp-ims+xml
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=
c=IN IP6 5555::aaa:bbb:ccc:ddd
t=0 0
m=audio 3456 RTP/AVP 97 96
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:96 telephone-event
a=maxptime:20
```

4. **Evaluation of initial filter criteria**

   The S-CSCF evaluates initial filter criteria for the served SC user and as a result routes the SIP INVITE request towards the SCC AS.

5. **SIP INVITE request (intermediate IM CN subsystem entities to SCC AS)**

   The SIP INVITE request is forwarded to the SCC AS as the result of the evaluation of iFC.

6. **Remote Leg Update**

   The SCC AS performs the Remote Leg update by sending the SIP re-INVITE request towards the remote UE.

7. **SIP re-INVITE request (SCC AS to intermediate IM CN subsystem entities)- See example in table A.6.1-7**

   The SCC AS modifies the message in accordance with routeing B2BUA functionality, e.g. mapping of From, To, Cseq and Call-ID header fields from one side of the B2BUA to the other. In this example the SCC AS includes the contents of the Contact header field from the received SIP INVITE request. The SIP re-INVITE request contains the SDP offer that is identical to the SDP offer that the SCC AS received in the initial SIP INVITE request from the UE A (Step 3).

Table A.6.1-7: SIP **re-INVITE request (SCC AS to intermediate IM CN subsystem entities)**

```
INVITE < sip:user1_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91e6bf6> SIP/2.0
Via: SIP/2.0/UDP sccas.home1.net; branch=z9hG4bK332b33.3;
Max-Forwards: 67
Route: <scscf1.home1.net;lr >,<sip:scscf2.home2.net;lr>, <sip:pcscf2.visited2.net;lr>
P-Asserted-Identity: "John Doe" <sip:user1_public1@home1.net>, <tel:+1-237-555-1111>
P-Access-Network-Info: IEEE-802.11b
Privacy: none
P-Charging-Vector: icid-value="AyretyU0dm+6O2IrT5tAFrbHLso=023551024"
P-Charging-Function-Addresses:
From: <sip:user1_public1@home1.net>; tag=1717777
To: <tel:+1-237-555-2222>, tag=4321
Call-ID: dc14b1t10b3teghmlk5013237
Cseq: 111 INVITE
Supported: precondition, 100rel
Contact: <sip:user2_public1@home2.net;gr=urn:uuid:2ad8950e-48a5-4a74-8d99-
   ad76cc7fc74>;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE
Accept: application/sdp
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=
c=IN IP6 5555::aaa:bbb:ccc:ddd
t=0 0
m=audio 3456 RTP/AVP 97 96
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:96 telephone-event
a=maxptime:20
```

8. SIP re-INVITE request (Intermediate IM CN subsystem entities to UE B)

The intermediate IM CN subsystem entities forward the SIP re-INVITE request to remote UE B.

9-10: SIP 200 (OK) response (UE B to SCC AS via Intermediate IM CN subsystem entities)

The UE B generates the SIP 200 (OK) response to the SIP re-INVITE request and forwards it to the SCC AS.

11-12: SIP ACK request (SCC AS to UE B via Intermediate IM CN subsystem entities)

The SCC AS generates the SIP ACK request to the SIP 200 (OK) response and forwards it to the remote UE B.

13-14: SIP 200 (OK) response (SCC AS to UE A via Intermediate IM CN subsystem entities)

The SCC AS generates the SIP 200 (OK) response to the SIP INVITE request and forwards it to the SC UE A.

15-16: SIP ACK request (SC UE A to SCC AS via Intermediate IM CN subsystem entities)

The SC UE A generates the SIP ACK request to the SIP 200 (OK) response and forwards it to the SCC AS

17. Media paths between UE A and UE B

The media path is using the new IP-CAN.

18-19. SIP BYE request (SCC AS to interworking entities via intermediate IM CN subsystem entities)

The SCC AS terminates the replaced call leg, which was using the CS bearer, by sending a SIP BYE request.

20-22. CC DISCONNECT message (interworking entities to SC UE A)

Upon receiving the CC DISCONNECT message, the SC UE A relinquishes all resources pertaining to the CS bearer.

NOTE: Steps 20-21 are performed only if signalling over CS domain is possible after the CS-PS access transfer is completed; otherwise, the SC UE A and the network release the source access leg locally, without any signalling between the SC UE A and the network.

23-24. SIP 200 (OK) response (Interworking entities to SCC AS via intermediate IM CN subsystem entities)

A.6.2 PS-CS access transfer: PS-CS

In this example, SC UE A has an ongoing session with remote UE B over PS bearer before access transfer which is anchored at SCC AS. When the SC UE attaches to the CS domain, it decides to transfer the session over the CS bearer without ICS capability.

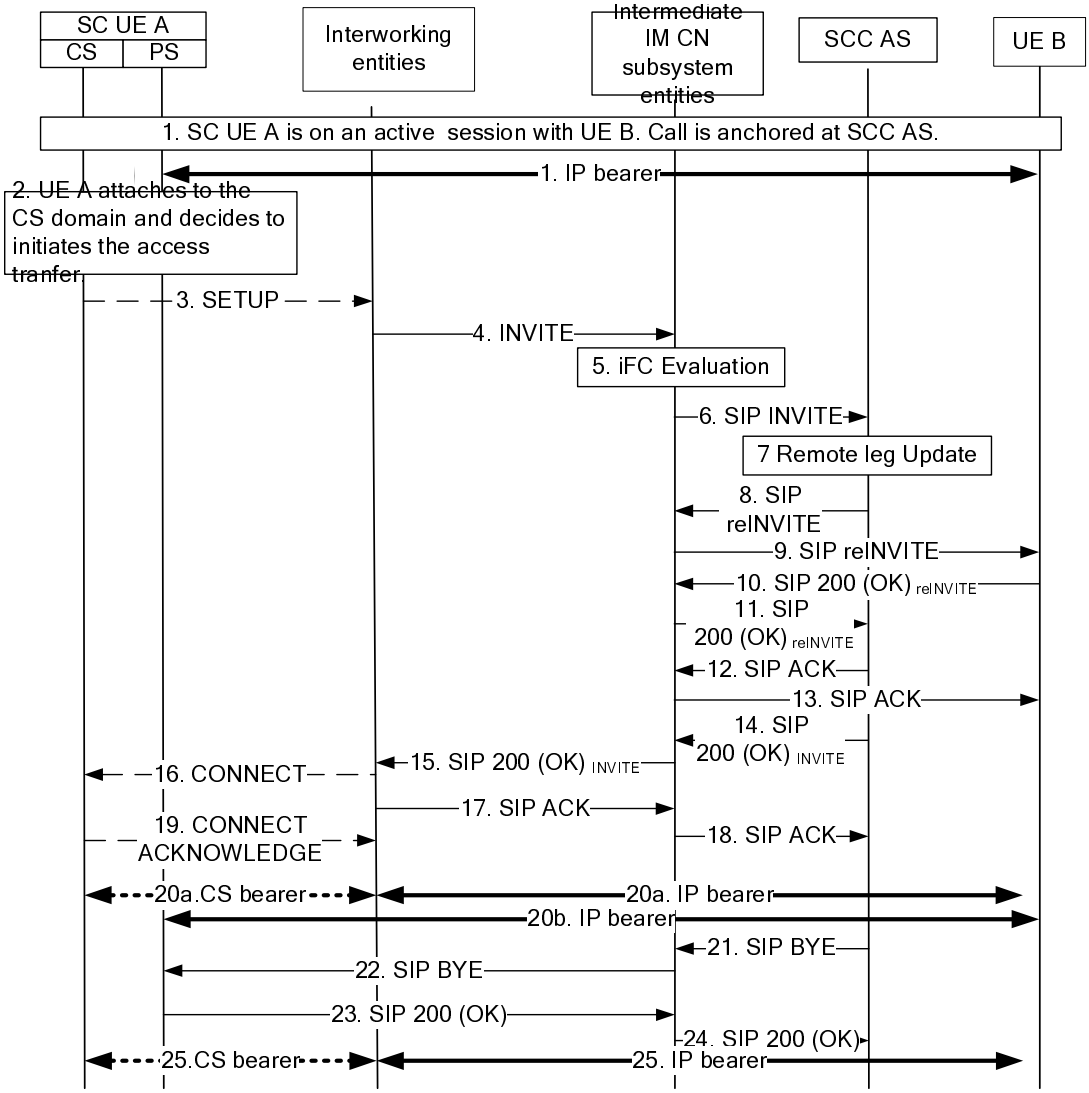


Figure A.6.2-1 Signalling flow for PS-CS access transfer: PS-CS

NOTE 1: For clarity, the SIP 100 (Trying) responses are not shown in the signalling flow.

1. **SC UE A is on an active session with UE B:**

   There is an ongoing IP bearer between the SC UE and the remote end UE B. The call is anchored at SCC AS.

2. **SC UE A attaches to the CS domain**

   The SC UE attaches to the CS domain and decides to transfer the session over the CS bearer.

3. **CC SETUP messages**

   The SC UE sends the CC SETUP message with the static STN as the called party number.

4. **SIP INVITE request (Interworking entities to Intermediate IM CN subsystem entities) -see example in table A.6.2-4**

Table A.6.2-4: SIP INVITE request (interworking entities to intermediate IM CN subsystem entities)

```
INVITE tel: +1-237-555-3333 SIP/2.0
Via: SIP/2.0/UDP mgcf1.home1.net;branch=z9hG4bk731b87
Max-Forwards: 70
Route: <sip:icscf1.home1.net;lr>
P-Asserted-Identity: <tel: +1-237-555-1111>
P-Charging-Vector: ####
Privacy: none
From: <tel: +1-237-555-1111>;tag=171828
To: <tel: +1-237-555-3333>
Call-ID: cb03a0s09a2sdfglkj490333
Cseq: 127 INVITE
Supported: 100rel, precondition
Accept-Contact: *;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
P-Asserted-Service: urn:urn-7:3gpp-service.ims.icsi.mmtel
Contact: <sip:mgcf1.home1.net;gr>;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:eee
s=
c=IN IP6 5555::aaa:bbb:ccc:eee
t=0 0
m=audio 3456 RTP/AVP 97 96
a=tcap:1 RTP/AVPF
a=pcfg:1 t=1
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:96 telephone-event
a=maxptime:20
```

Request-URI: contains the IMRN, as obtained from CS networks signalling.

SDP: The SDP contains preconfigured set of codecs supported by the MGW.

5. **Evaluation of initial filter criteria**

The S-CSCF evaluates initial filter criteria for the served SC user and as a result routes the SIP INVITE request towards the SCC AS.

6. **SIP INVITE request** (Intermediate IM CN subsystem entities to SCC AS)

7. **Remote Leg Update**

The SCC AS performs the Remote Leg update by sending the SIP re-INVITE request towards the remote UE.

8. **SIP re-INVITE request (SCC AS to intermediate IM CN subsystem entities) –see example in table A.6.2-8**

The SCC AS acting as a routing B2BUA generates a SIP INVITE request based upon the received SIP INVITE request and the information previously stored against this session and routes it towards UE B via the intermediate IM CN subsystem entities.

Table A.6.2-8: SIP re-INVITE request (SCC AS to intermediate IM CN subsystem entities)

```
INVITE sip:user2_public1@home2.net;gr=urn:uuid:2ad8950e-48a5-4a74-8d99-ad76cc7fc74 SIP/2.0
Via: SIP/2.0/UDP sccas1.home1.net;branch=z9hG4bKnas34r5
Max-Forwards: 67
Route: <sip:scscf1.home1.net:lr>
P-Asserted-Identity: <tel: +1-237-555-1111>
P-Charging-Function-Addresses: ccf=[5555::b99:c88:d77:e66]; ccf=[5555::a55:b44:c33:d22];
   ecf=[5555::1ff:2ee:3dd:4ee]; ecf=[5555::6aa:7bb:8cc:9dd]
P-Charging-Vector: icid-value="BzyretyU0dm+6O2IrT5tAFrbHLso=023551034"; orig-
   ioi="type3home1.net"
Privacy: none
From: <tel: +1-237-555-1111>;tag=569812
To: <tel:+1-237-555-2222>; tag=26545
Call-ID: dd13a0s09a2sdfglkj490378
Cseq:
Contact: <sip:user1_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-
   00a0c91e6bf6>;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
Allow:
Content-Type: Content-Length:

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:eee
s=
c=IN IP6 5555::aaa:bbb:ccc:eee
t=0 0
m=audio 3456 RTP/AVPF 97 96
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:96 telephone-event
a=maxptime:20
m=message 0 TCP/MSRP 98
a=accept-types:text/plain
```

9. SIP re-INVITE request (Intermediate IM CN subsystem entities to UE B)

Intermediate IM CN subsystem entities forward the SIP re-INVITE request to remote UE B.

10. SIP 200 (OK) response (UE B to intermediate IM CN subsystem entities)

Upon receiving the SIP re-INVITE request containing the SDP offer, since the UE B has all resources available, it sends immediately the SIP 200 (OK) response to the SIP re-INVITE request that contains the SDP answer. The SDP answer indicates that the resources are available.

11. SIP 200 (OK) response (intermediate IM CN subsystem entities to SCC AS)

The intermediate IM CN subsystem entities forward the SIP 200 (OK) response to the SIP re-INVITE request to the SCC AS in the originating network.

12-13. SIP ACK request (SCC AS to UE B via IM CN subsystem entities)

The SCC AS generates the SIP ACK request to the SIP 200 (OK) response, and forwards the SIP ACK request to the remote UE B.

14-15. SIP 200 (OK) response (SCC AS to interworking entities via IM CN subsystem entities)

The SCC AS generates the SIP 200 (OK) response to the SIP INVITE request, and forwards the SIP 200 (OK) response to the interworking entities.

16. CC CONNECT message (interworking entities to SC UE A)

17-18. SIP ACK request (interworking entities to SCC AS via IM CN subsystem entities)

The interworking entities generate the SIP ACK request to the SIP 200 (OK) response, and forward it to the SCC AS.

19. CC CONNECT ACKNOWLEDGE message (SC UE A to interworking entities)

20. Media paths between SC UE A and UE B:

The CS bearer is setup while the PS bearer is still existing.

21-22: SIP BYE request (SCC AS to SC UE A via intermediate IM CN subsystem entities)

The SCC AS terminates the replaced call leg, which was using the old IP-CAN, by sending a SIP BYE request to the UE A.

23-24. SIP 200 (OK) response (SC UE A to SCC AS via intermediate IM CN subsystem entities)

Upon receiving the SIP BYE request over the old IP-CAN, the SC UE A sends a SIP 200 (OK) response over the old IP-CAN to the SCC AS. Subsequently, the SC UE A relinquishes all resources pertaining to the old IP-CAN.

NOTE: Steps 22-23 are performed only if SC UE A is using Gm after the PS-CS access transfer is completed; otherwise, the SC UE A and the network release the source access leg locally, without any signalling between the SC UE A and the network.

25. Media paths between SC UE A and UE B

Finally, the session is transferred from PS bearer to CS bearer.

A.7 Signalling flows for PS-PS access transfer

A.7.1 Introduction

The signalling flows for PS-PS access transfer demonstrate how a multimedia session is transferred from Source Access Leg to the Target Access Leg. The following signalling flows are included:

- subclause A.7.2 shows an example when all media of an ongoing communication session and the associated signalling are transferred from Source Access Leg to the Target Access Leg; and
- subclause A.7.3 shows an example when not all media of an ongoing communication session are transferred from the Source Access Leg to the Target Access Leg.

A.7.2 PS-PS access transfer with full media transfer

The signalling flows shown in figure A.7.2-1 describes the PS-PS access transfer procedure when all media of an ongoing communication session and the associated signalling are transferred from one contact address of an UE to a different contact address of the same UE. No lower-level mechanism to support the access transfer is assumed or needed.

In this example the UE-1 is on an active multimedia session with the UE-2 via one IP-CAN. After changing to a new IP-CAN, obtaining a new IP address, and discovering a P-CSCF, the UE-1 reserves resources in new IP-CAN prior to initiating the PS-PS access transfer procedure. When the PS-PS access transfer procedure is completed, the UE-1 continues the multimedia session with the UE-2 on the new IP-CAN. In this example, when attaching to the new IP-CAN, it is irrelevant whether the UE-1 uses the same P-CSCF or a new P-CSCF.

NOTE 1: This scenario requires that the UE-1 and the IM CN subsystem support simultaneous multiple registrations and requires that the UE-1 supports dual mode operation.

NOTE 2: In this example flow, each call leg is uniquely identified with a respective dialog identifier consisting of the Call-ID, From tag, and To tag.

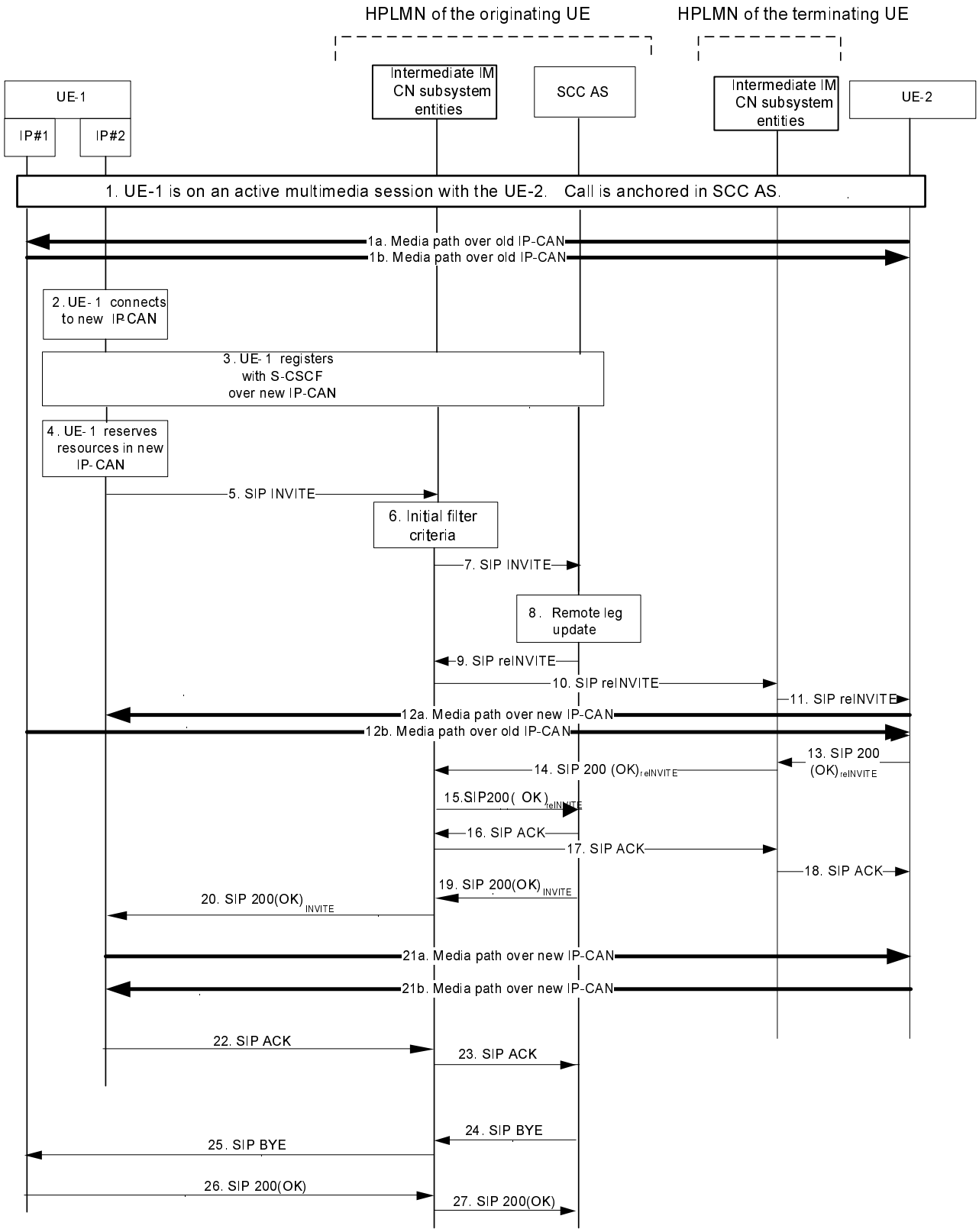


Figure A.7.2-1: Signalling flow for session handover

NOTE 3: For clarity, the SIP 100 (Trying) responses are not shown in the signalling flow.

1. UE-1 is on an active session with UE-2

The UE-1 is in an active session with the UE-2. The call is anchored in the SCC AS. It is irrelevant which endpoint initiated the call. Each call leg is uniquely identified with a respective dialog identifier. The call leg

over old IP-CAN is identified with "Call-ID= me03a0s09a2sdfgjkl491777", "From tag=64727891", and "To tag=774321". The UE-1 and UE-2 exchange media over the old IP-CAN, which is maintained while the UE-1 initiates the handover procedure.

2. **UE-1 connects to new IP-CAN**

   The UE-1 determines that a handover of the session is required. The UE-1 connects to the new IP-CAN. The UE-1 obtains an IP address that it will use for the signalling and media.

3. **UE-1 registers with intermediate IM CN subsystem entities over new IP-CAN**

   The UE-1 registers with the S-CSCF over the new IP-CAN using the standard multiple registrations procedure. Depending on the UE-1 configuration, the discovery of the P-CSCF in the new IP-CAN can precede this.

4. **UE-1 acquires resources in new IP-CAN**

   Based on the UE-1 and new IP-CAN capabilities, the UE-1 decides to use the same codec that was used over the old IP-CAN. The UE-1 reserves resources (e.g. QoS) in the new IP-CAN that will be needed for the signalling and transferred media, prior to sending the initial SIP INVITE request.

5. **SIP INVITE request (UE-1 to intermediate IM CN subsystem entities) - see example in table A.7.2-5**

   The UE-1 sends an initial SIP INVITE request with the PS to PS STI and a new SDP offer to the UE-2 that indicates that the new call replaces the existing call. The initial SIP INVITE request establishes a dialog for signalling and specifies in the SDP the new contact address that will be used for media over the new IP-CAN. Upon sending the initial SIP INVITE request, the UE-1 is ready to receive the RTP packets either over the new IP-CAN or the old IP-CAN. The RTP packets can arrive over the new IP-CAN prior to the UE-1 receiving the SIP 200 (OK) response for the initial SIP INVITE request.

Table A.7.2-5: SIP INVITE request (UE-1 to intermediate IM CN subsystem entities)

```
INVITE sip:pstops.transfer@sccas1.home1.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Max-Forwards: 70
Route: <sip:pcscf1.home1.net:7531;lr;comp=sigcomp>, <sip:orig@scscf1.home1.net;lr>
P-Preferred-Identity: "John Doe" <sip:user1_public1@home1.net>
P-Access-Network-Info:IEEE-802.11b
Privacy: none
From: <sip:user1_public1@home1.net>; tag=171828
To: <sip:pstops.transfer@sccas1.home1.net>
Call-ID: cb03a0s09a2sdfglkj490333
Cseq: 127 INVITE
Supported: 100rel, precondition, gruu, outbound
Require: sec-agree; replaces
Replaces: me03a0s09a2sdfgjkl491777; to-tag=774321; from-tag=64727891
Proxy-Require: sec-agree
Security-Verify: ipsec-3gpp; q=0.1; alg=hmac-sha-1-96; spi=87654321; port1=7531
Contact: <sip:user1_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91e6bf6;
   ob>;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel";+g.3gpp.ics="principal"
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE
Accept: application/sdp, application/3gpp-ims+xml
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c=IN IP6 5555::aaa:bbb:ccc:ddd
t=0 0
m=audio 3456 RTP/AVP 97 96
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:96 telephone-event
a=maxptime:20
```

Request-URI: the PS to PS STI configured in the SC UE or the URI contained in the Contact header field returned at the creation of the dialog on the Source Access Leg.

Require: the "replaces" option tag indicate that the support for Replace header field is required.

Replaces: specifies the existing call that will be replaced with the new call.

SDP: specifies the new IP address that the UE-1 has acquired in the new IP-CAN, and indicates that the resources in the new IP-CAN have been acquired.

6. Evaluation of initial filter criteria

Upon the evaluation of the initial filter criteria, as this is an originating initial SIP INVITE request for a registered user, the S-CSCF routes the initial SIP INVITE request to the SCC AS.

7. SIP INVITE request (intermediate IM CN subsystem entities to SCC AS) - see example in table A.7.2-7

The initial SIP INVITE request is forwarded from intermediate IM CN subsystem entities in the home network to the SCC AS. The P-CSCF added a Record-Route header field with a flow token to ensure that mid-dialog SIP requests are forwarded to the UE-1 over the correct flow. The SCC AS acts as a routeing B2BUA as specified in 3GPP TS 24.229 [2]. The SCC AS includes the contents of the Contact header field from the received SIP INVITE request.

Table A.7.2-7: SIP INVITE request (intermediate IM CN subsystem entities to SCC AS)

```
INVITE sip:pstops.transfer@sccas1.home1.net SIP/2.0
Via: SIP/2.0/UDP scscf1.home1.net;branch=z9hG4bK332b23.1, SIP/2.0/UDP
   pcscf1.home1.net;branch=z9hG4bK240f34.1, SIP/2.0/UDP
   [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Max-Forwards: 67
Route: <sip:sccas.home1.net;lr>; <sip:cb03a0s09a2sdfglkj490333@scscf1.home1.net;lr>;orig-
   dialog-id="O:73935718_92645110-712786jd246395302d-zKE"
Record-Route: <sip:scscf1.home1.net;lr>, <sip:
   GopIKSsn0oGLPXRdV9BAXpT3coNuiGKV@pcscf1.home1.net;lr>
P-Asserted-Identity: "John Doe" <sip:user1_public1@home1.net>, <tel:+1-212-555-1111>
P-Access-Network-Info:Privacy:Require: replaces
P-Charging-Vector: icid-value="AyretyU0dm+6O2IrT5tAFrbHLso=023551024";orig-
   ioi=type3ashome1.net>
P-Charging-Function-Addresses: ####
From: <sip:user1_public1@home1.net>; tag=171828
To: <sip:pstops.transfer@sccas1.home1.net >
Call-ID:
Cseq:
Supported:
Replaces:
Contact:
Allow:
Accept:
Content-Type:
Content-Length: (…)
P-Early-Media: supported

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

8. Remote leg update

The SCC AS based on the content of the Replaces header field correlates the initial SIP INVITE request to the existing local and remote call legs of the existing concatenated end to end session between the UE-1 and UE-2.

The SCC AS updates the remote call leg by sending a SIP re-INVITE request to the UE-2 containing the new SDP offer that it has received from the UE-1.

9. SIP re-INVITE request (SCC AS to intermediate IM CN subsystem entities) - see example in table A.7.2-9

The UE-2 is informed of the change in access leg by the SCC AS sending a SIP re-INVITE request to the S-CSCF.

The SCC AS modifies the message in accordance with routeing B2BUA functionality, e.g. mapping of From, To, Cseq and Call-ID header fields from one side of the B2BUA to the other. In this example the SCC AS includes the contents of the Contact header field from the received SIP INVITE request. The SIP re-INVITE request contains the SDP offer that is identical to the SDP offer that the SCC AS received in the initial SIP INVITE request from the UE-1 (Step 5).

Table A.7.2-9: SIP **re-INVITE request (SCC AS to intermediate IM CN subsystem entities)**

```
INVITE < sip:user2_public1@home2.net;gr=urn:uuid:2ad8950e-48a5-4a74-8d99-ad76cc7fc74> SIP/2.0
Via: SIP/2.0/UDP sccas.home1.net; branch=z9hG4bK332b33.3;
Max-Forwards: 67
Route: <scscf1.home1.net;lr>,<sip:scscf2.home2.net;lr>,<sip:pcscf2.visited2.net;lr>
P-Asserted-Identity:P-Access-Network-Info:Privacy:P-Charging-Vector: icid-
   value="BzyretyU0dm+6O2IrT5tAFrbHLso=023551034 "
P-Charging-Function-Addresses:
From: <sip:user1_public1@home1.net>; tag=1717777
To: <tel:+1-212-555-2222>, tag=4321
Call-ID: dc14b1t10b3teghmlk5013333
Cseq: 111 INVITE
Supported:
Contact: < sip:user1_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-
   00a0c91e6bf6;ob>;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
Allow:
Accept: application/sdp
Content-Type:
Content-Length: (…)

v=0
o=2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
t=0 0
m=audio 3456 RTP/AVP 97 96
b=AS:25.4
a= curr:qos local sendrecv
a= curr:qos remote none
a= des:qos mandatory local sendrecv
a= des:qos none remote sendrecv
a= rtpmap:97 AMR
a= fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a= rtpmap:96 telephone-event
a= maxptime:20
```

Route: The SIP re-INVITE request contains the saved list of Route header fields that the SCC AS has saved for the remote leg of the call.

10. SIP re-INVITE request (intermediate IM CN subsystem entities to intermediate IM CN subsystem entities) - see example in table A.7.2-10

In the originating network, the intermediate IM CN subsystem entities forward the SIP re-INVITE request to the intermediate IM CN subsystem entities in the terminating network.

Table A.7.2-10: SIP re-INVITE request (intermediate IM CN subsystem entities to intermediate IM CN subsystem entities)

```
INVITE < sip:user2_public1@home2.net;gr=urn:uuid:2ad8950e-48a5-4a74-8d99-ad76cc7fc74> SIP/2.0
Via: SIP/2.0/UDP scscf1.home1.net;branch=z9hG4bK240f34.1, SIP/2.0/UDP sccas.home1.net;
   branch=z9hG4bK332b33.3;
Max-Forwards: 66
Route: <sip:scscf2.home2.net;lr>, <sip:pcscf2.visited2.net;lr>
P-Asserted-Identity:
Privacy: none
From:
To:
Call-ID:
Cseq:Supported:Contact: <sip:user1_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-
   00a0c91e6bf6>;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
Contact:
Allow:
Accept:
Content-TypeContent-Length:

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

11. SIP re-INVITE request (intermediate IM CN subsystem entities to UE-2)

In the terminating network, the SIP re-INVITE request is forwarded towards the UE-2 by the intermediate IM CN subsystem entities.

12. Media paths between UE-1 and UE-2

The UE-2 receives the SIP re-INVITE request containing the SDP offer that indicates that the UE-1 is ready to receive the same media on a different contact address. Since the UE-2 has resources already available, it starts to send the media to the UE-1's contact address specified in the SDP offer immediately.

The UE-1 will be receiving the RTP packets over new IP-CAN. However, the UE-1 can receive some out-of-sequence RTP packets over the old IP-CAN. The RTP packets are delivered to the codec in sequence. Once the UE-1 determine that no media will be received over the old IP-CAN (e.g. by examining the sequence numbers in the RTP headers), it can relinquish the resources that it has been using for incoming media on the old IP-CAN.

The UE-1 sends the media to the UE-2 over the old IP-CAN.

Resources used for signalling on the old IP-CAN are not released.

13. SIP 200 (OK) response (UE-2 to intermediate IM CN subsystem entities)

Upon receiving the SIP re-INVITE request containing the SDP offer, since the UE-2 has all resources available, it sends immediately the SIP 200 (OK) response to the SIP re-INVITE request that contains the SDP answer. The SDP answer indicates that the resources are available.

14. SIP 200 (OK) response (intermediate IM CN subsystem entities to intermediate IM CN subsystem entities)

In the terminating network, the intermediate IM CN subsystem entities forward the SIP 200 (OK) response to the SIP re-INVITE request to the intermediate IM CN subsystem entities in the originating network.

15. SIP 200 (OK) response (intermediate IM CN subsystem entities to SCC AS)

The intermediate IM CN subsystem entities in the originating network forward the SIP 200 (OK) response to the SIP re-INVITE request to the SCC AS.

16. SIP ACK request (SCC AS to intermediate IM CN subsystem entities)

The SCC AS acting as a B2BUA acknowledges the receipt of the SIP 200 (OK) response to the SIP re-INVITE request by forwards a SIP ACK request to the intermediate IM CN subsystem entities.

17. SIP ACK request (intermediate IM CN subsystem entities to intermediate IM CN subsystem entities)

In the originating network, the intermediate IM CN subsystem entities forward the SIP ACK request to the intermediate IM CN subsystem entities in the terminating network.

18. SIP ACK request (intermediate IM CN subsystem entities to UE-2)

In the terminating network, the intermediate IM CN subsystem entities forward the SIP ACK request to the UE-2.

19. SIP 200 (OK) response (SCC AS to intermediate IM CN subsystem entities)

The SCC AS forwards the SIP 200 (OK) response to the initial SIP INVITE request to the intermediate IM CN subsystem entities, using the content of the Via header field that was received in the initial SIP INVITE request (step 5).

The SCC AS modifies the message in accordance with routeing B2BUA functionality, e.g. mapping of From, To, Cseq and Call-ID header fields from one side of the B2BUA to the other. The SIP 200 (OK) response to the initial SIP INVITE request contains the SDP answer that is identical to the SDP answer that the SCC AS has received in the SIP 200 (OK) response to SIP re-INVITE request from the UE-2 (Step 13).

20. SIP 200 (OK) response (intermediate IM CN subsystem entities to UE-1)

The intermediate IM CN subsystem entities forward the SIP 200 (OK) response to the UE-1.

21. Media paths between UE-1 and UE-2

The UE-1 receives the SIP 200 (OK) response containing the SDP answer that indicates that the UE-2 is ready to receive media. Since the UE-1 has already resources available, it starts to send media over new IP-CAN to the UE-2's contact address specified in the SDP answer immediately.

The UE-1 can relinquish the resources that it has been using for outgoing media on the old IP-CAN.

Resources used for signalling on the old IP-CAN are not released.

22. SIP ACK request (UE-1 to intermediate IM CN subsystem entities)

The UE-1 completes the new call leg creation with a SIP ACK request sent to the intermediate IM CN subsystem entities.

23. SIP ACK request (-intermediate IM CN subsystem entities to SCC AS)

The intermediate IM CN subsystem entities forward the SIP ACK request to the SCC AS.

24. SIP BYE request (SCC AS to intermediate IM CN subsystem entities)

The SCC AS terminates the replaced call leg- that was using the old IP-CAN, by sending a SIP BYE request to the UE-1.

25. SIP BYE request (intermediate IM CN subsystem entities to UE-1)

The intermediate IM CN subsystem entities forward the SIP BYE request to the UE-1.

26. SIP 200 (OK) response (UE-1 to intermediate IM CN subsystem entities)

Upon receiving the SIP BYE request over the old IP-CAN, the UE-1 sends a SIP 200 (OK) response over the old IP-CAN. Subsequently, the UE-1 relinquishes all resources pertaining to the old IP-CAN.

27. SIP 200 (OK) response (intermediate IM CN subsystem entities to SCC AS)

The intermediate IM CN subsystem entities forward the SIP 200 (OK) response to the SCC AS.

Since both the old contact address and the new contact address were registered using multiple registrations procedure with different reg-id values, then upon transferring the dialog from the old contact address to the new contact address, the UE-1 is still registered with the old contact address and the UE-1 subscription dialog to its reg-event using the old contact address is intact.

A.7.3 PS-PS access transfer with partial media transfer

The signalling flows shown in figure A.7.3-1 describes the PS-PS access transfer procedure when not all media of an ongoing communication session are transferred from the Source Access Leg to the Target Access Leg. No lower-level mechanism to support the access transfer is assumed or needed.

In this example, UE-1 is on an active multimedia session with UE-2 via one IP-CAN. After connecting to an additional IP-CAN, obtaining an additional IP address, discovering a P-CSCF, and performing registration in the IM CN subsystem, UE-1 reserves resources in the new IP-CAN prior to initiating the PS-PS access transfer procedure. When the PS-PS access transfer procedure is completed, UE-1 continues the multimedia session with UE-2 on both the old and the new IP-CANs. In this example, when attaching to the new IP-CAN, it is irrelevant whether the UE-1 uses the same P-CSCF or a new P-CSCF.

NOTE 1: This scenario requires that UE-1 and the IM CN subsystem support simultaneous multiple registrations and requires that UE-1 supports dual mode operation.

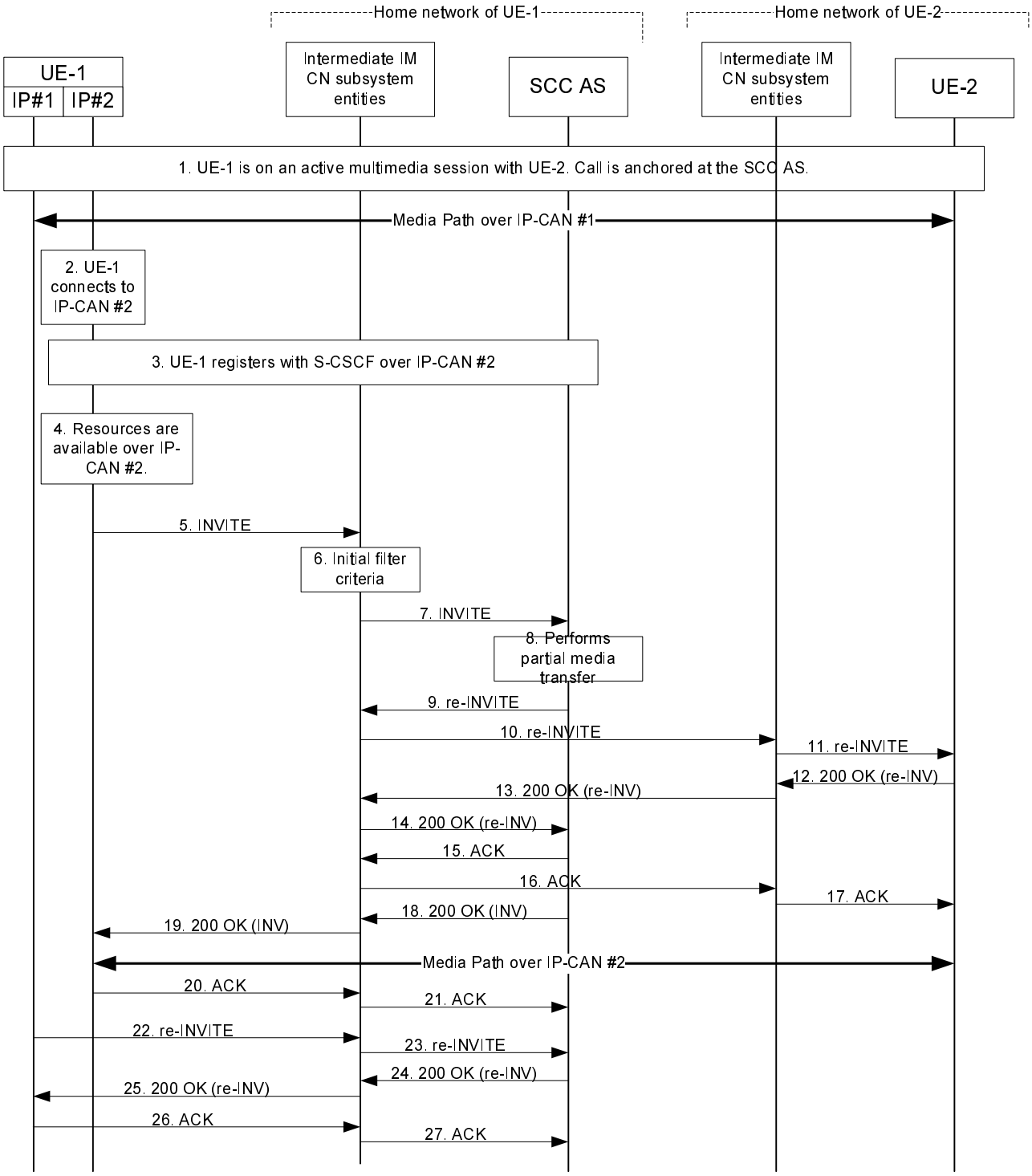


Figure A.7.3-1: Signalling flow for PS-PS session transfer with partial media transfer

NOTE 2: For clarity, the SIP 100 (Trying) responses are not shown in the signalling flow.

1. **UE-1 is on an active session with UE-2**

   UE-1 is in an active session with UE-2. The call is anchored in the SCC AS. It is irrelevant which endpoint initiated the call. Each call leg is uniquely identified with a respective dialog identifier. The call leg over IP-CAN #1 is identified with "Call-ID= me03a0s09a2sdfgjkl491777", "From tag=64727891", and "To tag=774321". UE-1 and UE-2 exchange media over the IP-CAN #1, which is maintained while the UE-1 initiates the session transfer procedure.

2. **UE-1 connects to IP-CAN #2**

UE-1 connects to the new IP-CAN and obtains an IP address that it will use for the signalling and media.

3. **UE-1 registers with intermediate IM CN subsystem entities over IP-CAN #2**

   UE-1 registers with the S-CSCF over the IP-CAN #2 using the standard multiple registrations procedure. The P-CSCF in the signalling path of this registration can be distinct from the one used in the signalling path over IP-CAN #1.

4. **UE-1 acquires resources in IP-CAN #2**

   UE-1 decides to perform partial media transfer to the IP-CAN #2. Based on UE-1 and IP-CAN #2 capabilities, the UE-1 decides to use the same codec that was used over the IP-CAN #1 for the media components to be transferred. UE-1 ensures that the resources (e.g. QoS) in IP-CAN #2 that will be needed for the signalling and transferred media are available, prior to sending the initial SIP INVITE request.

5. **SIP INVITE request (UE-1 to intermediate IM CN subsystem entities) - see example in table A.7.3-5**

   UE-1 sends initial SIP INVITE request with the PS to PS STI and a new SDP offer to UE-2 and indicates that the video component is to be transferred to IP-CAN #2. The initial SIP INVITE request establishes a dialog for signalling and specifies in the SDP new contact address that will be used for media over IP-CAN #2. Upon sending the initial SIP INVITE request, UE-1 is ready to receive the RTP packets over both IP-CAN #1 and IP-CAN #2.

Table A.7.3-5: SIP INVITE request (UE-1 to intermediate IM CN subsystem entities)

```
INVITE sip:pstops.transfer@sccas1.home1.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Max-Forwards: 70
Route: sip:pcscf1.home1.net:7531;lr;comp=sigcomp>, <sip:orig@scscf1.home1.net;lr>
P-Preferred-Identity: "John Doe" <sip:user1_public1@home1.net>
P-Access-Network-Info:IEEE-802.11b
Privacy: none
From: <sip:user1_public1@home1.net>; tag=171828
To: <sip:pstops.transfer@sccas1.home1.net>
Call-ID: cb03a0s09a2sdfglkj490333
Cseq: 127 INVITE
Supported: 100rel, precondition, gruu, outbound
Require: sec-agree; tdialog
Target-Dialog: me03a0s09a2sdfgjkl491777; remote-tag=774321; local-tag=64727891
Proxy-Require: sec-agree
Security-Verify: ipsec-3gpp; q=0.1; alg=hmac-sha-1-96; spi=87654321; port1=7531
Contact: < sip:user1_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-
   00a0c91e6bf6;ob>;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-
   service.ims.icsi.mmtel";+g.3gpp.ics="principal";
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE
Accept: application/sdp, application/3gpp-ims+xml
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c=IN IP6 5555::aaa:bbb:ccc:ddd
t=0 0
m=audio 0 RTP/AVP 97 96
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:96 telephone-event
m=video 3400 RTP/AVP 98 99
b=AS:75
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
a=rtpmap:99 MP4V-ES
```

Request-URI: the PS to PS STI configured in the SC UE or the URI contained in the Contact header field returned at the creation of the dialog on the Source Access Leg .

Require: the "tdialog" option tag indicate that the support for Target-Dialog header field is required.

Target-Dialog: specifies the existing call that will be transferred.

SDP: specifies the new IP address that the UE-1 has acquired in the new IP-CAN, and indicates that only the video component will be transferred and the resources in the new IP-CAN have been reserved.

6. Evaluation of initial filter criteria

Upon the evaluation of the initial filter criteria, as this is an originating initial SIP INVITE request for a registered user, the S-CSCF routes the initial SIP INVITE request to the SCC AS.

7. SIP INVITE request (intermediate IM CN subsystem entities to SCC AS)

The initial SIP INVITE request is forwarded from intermediate IM CN subsystem entities in the home network to the SCC AS. The P-CSCF added a Record-Route header with a flow token to ensure that mid-dialog SIP requests are forwarded to the UE-1 over the correct flow. The SCC AS acts as a routeing B2BUA as specified in 3GPP TS 24.229 [2].

8. Remote leg update

Based on the content of the Target-Dialog header field, the SCC AS correlates the SIP INVITE request for session transfer to the existing local and remote call legs of the existing concatenated end to end session between UE-1 and UE-2. The SCC AS updates the remote call leg by sending a SIP re-INVITE request to the remote UE-2 containing the new SDP offer based on the partial media transfer request received from UE-1 and the negotiated SDP for the original session.

9. SIP re-INVITE request (SCC AS to intermediate IM CN subsystem entities) - see example in table A.7.3-9

UE-2 is informed of the change in access leg by the SCC AS sending a re-INVITE request to the S-CSCF.

The SCC AS modifies the message in accordance with routeing B2BUA functionality, e.g. mapping of From, To, Cseq and Call-ID header fields from one side of the B2BUA to the other. In this example the SCC AS includes the contents of the Contact header field from the received SIP INVITE request. The SIP re-INVITE request contains the SDP offer that is based on original SDP offer and the SDP offer that the SCC AS received in the initial SIP INVITE request from the UE-1 (Step 7).

Table A.7.3-9: SIP **re-INVITE request (SCC AS to intermediate IM CN subsystem entities)**

```
INVITE < sip:user2_public1@home2.net;gr=urn:uuid:2ad8950e-48a5-4a74-8d99-ad76cc7fc74> SIP/2.0>
Via: SIP/2.0/UDP sccas.home1.net; branch=z9hG4bK332b33.3;
Max-Forwards: 70
Route: <scscf1.home1.net;lr>, <sip:scscf2.home2.net;lr>, <sip:pcscf2.visited2.net;lr>
P-Asserted-Identity: "John Doe" <sip:user1_public1@home1.net>, <tel:+1-212-555-1111>
Privacy: none
P-Charging-Vector: icid-value="AyretyU0dm+6O2IrT5tAFrbHLso=023551024"
P-Charging-Function-Addresses:
From: <sip:user1_public1@home1.net>; tag=1717777
To: <tel:+1-212-555-2222>, tag=4321
Call-ID: dc14b1t10b3teghmlk5013333
Cseq: 111 INVITE
Supported: precondition, 100rel
Contact:<sip:user1_public1@home1.net; gr=urn:uuid:f81d4fae-7dec-11d0-a765-
   00a0c91e6bf6;ob>;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE
Accept: application/sdp
Content-Type: application/sdp
Content-Length: (…)

v=0
o=2987933100 2987933101 IN IP6 5555::aaa:bbb:ccc:eee
s=-
t=0 0
m=audio 3456 RTP/AVP 97 96
c=IN IP6 5555::aaa:bbb:ccc:eee
b=AS:25.4
a= curr:qos local sendrecv
a= curr:qos remote none
a= des:qos mandatory local sendrecv
a= des:qos none remote sendrecv
a= rtpmap:97 AMR
a= fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a= rtpmap:96 telephone-event
a= maxptime:20
m=video 3400 RTP/AVP 98 99
c=IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:75
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
a=rtpmap:99 MP4V-ES
```

Route: The SIP re-INVITE request contains the saved list of Route header fields that the SCC AS has saved for the remote leg of the call.

SDP: specifies the new IP address and ports used for the media components. In this case, the audio component is still using the original address and port while the video component is using the new IP address and new port allocated.

10. SIP re-INVITE request (intermediate IM CN subsystem entities to intermediate IM CN subsystem entities)

In the originating network, the intermediate IM CN subsystem entities forward the SIP re-INVITE request to the intermediate IM CN subsystem entities in the terminating network.

11. SIP re-INVITE request (intermediate IM CN subsystem entities to UE-2)

In the terminating network, the SIP re-INVITE request is forwarded towards UE-2 by the intermediate IM CN subsystem entities.

UE-2 receives the SIP re-INVITE request containing the SDP offer that indicates that UE-1 is ready to receive video media on a different contact address. Since UE-2 has resources already available, it starts to send the media to UE-1's contact address specified in the SDP offer immediately.

UE-1 starts receiving the video RTP packets over IP-CAN #2. However, UE-1 can receive some out-of-sequence video RTP packets over IP-CAN #1. The video RTP packets are delivered to the codec in sequence. Once UE-1

determine that no video will be received over IP-CAN #1 (e.g. by examining the sequence numbers in the RTP headers), it can relinquish the resources that it has been using for incoming video media on IP-CAN #1.

At the same time, UE-1 still sends both the audio and video media to UE-2 over IP-CAN #1.

Resources used for signalling on IP-CAN #1 are not released.

12. SIP 200 (OK) response (UE-2 to intermediate IM CN subsystem entities) – see example in table A.7.3-12

Upon receiving the SIP re-INVITE request containing the SDP offer, since UE-2 has all resources available, it sends immediately the SIP 200 (OK) response to the SIP re-INVITE request that contains the SDP answer. The SDP answer indicates that the resources are available.

Table A.7.3-12: SIP **200 (OK) response (UE-2 to intermediate IM CN subsystem entities)**

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf2.visited2.net:5088;comp=sigcomp;branch=z9hG4bK361k21.1,
 SIP/2.0/UDP scscf2.home2.net;branch=z9hG4bK764z87.1,
 SIP/2.0/UDP scscf1.home1.net;branch=z9hG4bK332b23.1,
 SIP/2.0/UDP sccas.home1.net;branch=z9hG4bK332b33.3
Record-Route: <sip:pcscf2.visited2.net:5088;lr;comp=sigcomp>, <sip:scscf2.home2.net;lr>,
   <sip:scscf1.home1.net;lr>
P-Access-Network-Info: 3GPP-UTRAN-TDD; utran-cell-id-3gpp=234151D0FCE11
Privacy: none
From: <sip:user1_public1@home1.net>; tag=1717777
To: <tel:+1-212-555-2222>;tag=4321
Call-ID: dc14b1t10b3teghmlk5013333
CSeq: 111 INVITE
Supported: precondition, 100rel
Contact: <sip:user2_public1@home2.net;gr=urn:uuid:2ad8950e-48a5-4a74-8d99-
   ad76cc7fc74>;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel" >
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933623 2987933624 IN IP6 5555::eee:fff:aaa:bbb
s=-
c=IN IP6 5555::eee:fff:aaa:bbb
t=0 0
m=audio 6544 RTP/AVP 97 96
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote sendrecv
a=des:qos mandatory local sendrecv
a=des:qos mandatory remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:96 telephone-event
a=maxptime:20
m=video 10001 RTP/AVP 98 99
b=AS:75
a=curr:qos local sendrecv
a=curr:qos remote sendrecv
a=des:qos mandatory local sendrecv
a=des:qos mandatory remote sendrecv
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
a=rtpmap:99 MP4V-ES
```

13. SIP 200 (OK) response (intermediate IM CN subsystem entities to intermediate IM CN subsystem entities)

In the terminating network, the intermediate IM CN subsystem entities forward the SIP 200 (OK) response to the SIP re-INVITE request to the intermediate IM CN subsystem entities in the originating network.

14. SIP 200 (OK) response (intermediate IM CN subsystem entities to SCC AS)

The intermediate IM CN subsystem entities in the originating network forward the SIP 200 (OK) response to the SIP re-INVITE request to the SCC AS.

15. SIP ACK request (SCC AS to intermediate IM CN subsystem entities)

The SCC AS acting as a B2BUA acknowledges the receipt of the SIP 200 (OK) response to the SIP re-INVITE request by forwards a SIP ACK request to the intermediate IM CN subsystem entities.

16. SIP ACK request (intermediate IM CN subsystem entities to intermediate IM CN subsystem entities)

In the originating network, the intermediate IM CN subsystem entities forward the SIP ACK request to the intermediate IM CN subsystem entities in the terminating network.

17. SIP ACK request (intermediate IM CN subsystem entities to UE-2)

In the terminating network, the intermediate IM CN subsystem entities forward the SIP ACK request to UE-2.

18. SIP 200 (OK) response (SCC AS to intermediate IM CN subsystem entities) – see example in table A.7.3-18

The SCC AS forwards the SIP 200 (OK) response to the initial SIP INVITE request to the intermediate IM CN subsystem entities, using the content of the Via header field that was received in the initial SIP INVITE request (step 5).

The SCC AS modifies the message in accordance with routeing B2BUA functionality, e.g. mapping of From, To, Cseq and Call-ID header fields from one side of the B2BUA to the other. In this example the SCC AS includes the contents of the Contact header field from the received SIP 200 (OK) response. The SIP 200 (OK) response to the initial SIP INVITE request contains the SDP answer derived from the SDP answer that the SCC AS has received in the SIP 200 (OK) response to SIP re-INVITE request from UE-2 (Step 14).

Table A.7.3-18: SIP **200 (OK) response (SCC AS to intermediate IM CN subsystem entities)**

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf1.home1.net;branch=z9hG4bK332b23.1,
 SIP/2.0/UDP pcscf1.home1.net;branch=z9hG4bK240f34.1,
 SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Record-Route: <sip:sccas.home1.net;lr>,<sip:scscf1.home1.net;lr>, <sip:
   GopIKSsn0oGLPXRdV9BAXpT3coNuiGKV@pcscf1.home1.net;lr>
Privacy: none
From: <sip:user1_public1@home1.net>; tag=171828
To: <sip:pstops.transfer@sccas1.home1.net>;tag=8009
Call-ID: cb03a0s09a2sdfglkj490333
Cseq: 127 INVITE
Supported: 100rel, precondition, gruu, outbound
Contact: < sip:user2_public1@home2.net;gr=urn:uuid:2ad8950e-48a5-4a74-8d99-
   ad76cc7fc74>;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE
Accept: application/sdp
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933300 2987933300 IN IP6 5555::eee:fff:aaa:bbb
s=-
c=IN IP6 5555::eee:fff:aaa:bbb
t=0 0
m=audio 0 RTP/AVP 97 96
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:96 telephone-event
m=video 10001 RTP/AVP 98 99
b=AS:75
a=curr:qos local sendrecv
a=curr:qos remote sendrecv
a=des:qos mandatory local sendrecv
a=des:qos mandatory remote sendrecv
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
a=rtpmap:99 MP4V-ES
```

19. SIP 200 (OK) response (intermediate IM CN subsystem entities to UE-1)

The intermediate IM CN subsystem entities forward the SIP 200 (OK) response to UE-1.

UE-1 receives the SIP 200 (OK) response containing the SDP answer indicating that UE-2 is ready to receive media. Since UE-1 has already resources available, it starts to send video media over IP-CAN #2 to UE-2's contact address specified in the SDP answer immediately.

The UE-1 can relinquish the resources that it has been using for outgoing video media on IP-CAN #1.

Resources used for signalling and audio media on IP-CAN #1 are not released.

20. SIP ACK request (UE-1 to intermediate IM CN subsystem entities)

UE-1 completes the new call leg creation with a SIP ACK request sent to the intermediate IM CN subsystem entities.

21. SIP ACK request (intermediate IM CN subsystem entities to SCC AS)

The intermediate IM CN subsystem entities forward the SIP ACK request to the SCC AS.

22. SIP re-INVITE request (UE-1 to intermediate IM CN subsystem entities) – see example in table A.7.3-22

UE-1 updates the old call leg on IP-CAN #1 by sending a SIP re-INVITE request to the intermediate IM CN subsystem entities.

Table A.7.3-22: SIP re-INVITE **request (UE-1 to intermediate IM CN subsystem entities)**

```
INVITE <sip:user2_public1@home2.net;gr=urn:uuid:2ad8950e-48a5-4a74-8d99-ad76cc7fc74> SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:eee]:2468;comp=sigcomp;branch=z9hG4bKashdns1
Max-Forwards: 70
Route: sip:XopDDDsn0oFFFXRdV9BAXpT3coNuiGKV@pcscf1.home1.net:8765;lr;comp=sigcomp>,
   <sip:orig@scscf1.home1.net;lr>
P-Access-Network-Info: 3GPP-UTRAN-FDD; utran-cell-id-3gpp=123456ABCDE22
Privacy: none
From: <sip:user1_public1@home1.net>; tag=64727891
To: <tel:+1-212-555-2222>; tag=774321
Call-ID: me03a0s09a2sdfgjkl491777
Cseq: 101 INVITE
Supported: 100rel; precondition; tdialog
Require: sec-agree;
Proxy-Require: sec-agree
Security-Verify: ipsec-3gpp; q=0.1; alg=hmac-sha-1-96; spi=12345678; port1=2468
Contact: <sip:user1_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-
   00a0c91e6bf6;ob>;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-
   service.ims.icsi.mmtel";+g.3gpp.ics="principal";
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE
Accept: application/sdp, application/3gpp-ims+xml
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933000 2987933001 IN IP6 5555::aaa:bbb:ccc:eee
s=-
c=IN IP6 5555::aaa:bbb:ccc:eee
t=0 0
m=audio 3456 RTP/AVP 97 96
b=AS:25.4
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:96 telephone-event
m=video 0 RTP/AVP 98 99
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
a=rtpmap:99 MP4V-ES
```

23. SIP re-INVITE request (intermediate IM CN subsystem entities to SCC AS)

The intermediate IM CN subsystem entities forward the SIP re-INVITE request to the SCC AS.

24. SIP 200 (OK) response (SCC AS to intermediate IM CN subsystem entities) – see example in table A.7.3-24

The SCC AS updates the old call leg based on the SIP re-INVITE request and sends the SIP 200 (OK) response to the SIP re-INVITE request to the intermediate IM CN subsystem entities, using the content of the Via header

field that was received in the SIP re-INVITE request (step 23). In this example the SCC AS includes the contents of the Contact header field from the received SIP 200 (OK) response. The SIP 200 (OK) response to the SIP re-INVITE request contains the SDP answer derived from the SDP answer that the SCC AS previously received from UE-2 (Step 14).

Table A.7.3-24: SIP **200 (OK) response (SCC AS to intermediate IM CN subsystem entities)**

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf1.home1.net;branch=z9hG4bK345b32.2,
 SIP/2.0/UDP pcscf1.home1.net;branch=z9hG4bK568f35.1,
 SIP/2.0/UDP [5555::aaa:bbb:ccc:eee]:2468;comp=sigcomp;branch=z9hG4bKashdns1
Record-Route: <sccas.home1.net;lr>,<sip:scscf1.home1.net;lr>, <sip:
   XopDDDsn0oFFFXRdV9BAXpT3coNuiGKV@pcscf1.home1.net;lr>
Privacy: none
From: <sip:user1_public1@home1.net>; tag=64727891
To: <tel:+1-212-555-2222>;tag=774321
Call-ID: me03a0s09a2sdfgjkl491777
Cseq: 101 INVITE
Supported: 100rel; precondition
Contact: < sip:user2_public1@home2.net;gr=urn:uuid:2ad8950e-48a5-4a74-8d99-
   ad76cc7fc74>;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE
Accept: application/sdp
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933800 2987933801 IN IP6 5555::eee:fff:aaa:bbb
s=-
c=IN IP6 5555::eee:fff:aaa:bbb
t=0 0
m=audio 6544 RTP/AVP 97 96
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:96 telephone-event
m=video 0 RTP/AVP 98 99
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
a=rtpmap:99 MP4V-ES
```

25. SIP 200 (OK) response (intermediate IM CN subsystem entities to UE-1)

The intermediate IM CN subsystem entities forward the SIP 200 (OK) response to UE-1.

26. SIP ACK request (UE-1 to intermediate IM CN subsystem entities)

UE-1 completes the old call leg update with a SIP ACK request sent to the intermediate IM CN subsystem entities.

27. SIP ACK request (intermediate IM CN subsystem entities to SCC AS)

The intermediate IM CN subsystem entities forward the SIP ACK request to the SCC AS.

A.7.4 PS-PS Access Transfer with full media transfer for an outgoing call in alerting phase

The signalling flows shown in figure A.7.4-1 describes the PS-to-PS access transfer procedure when an early dialog originated by the SC UE A, is transferred from one contact address of a SC UE A (using an old IP-CAN) to a different contact address of the same SC UE A (using a different IP-CAN). In this example flow, the SC UE A is attached to old IP-CAN, and is in the process of establishing a dialog on its Source Access Leg via this IP-CAN, with the UE B. While the dialog on the Source Access Leg is in the alerting phase, the SC UE A decides (e.g. based on the measurement reports) to transfer this dialog to the Target Access Leg that will be established over the new IP-CAN. Both, the SCC AS and the SC UA A support the PS-to-PS access transfer for the dialogs in early dialog phase.

NOTE 1: This scenario requires that the SC UE A supports dual mode operation and multiple registration procedure.

NOTE 2: In this example flow, each call leg is uniquely identified with a respective dialog identifier consisting of the Call-ID, From tag, and To tag.

NOTE 3: For clarity, the SIP 100 (Trying) responses are not shown in the signalling flow.

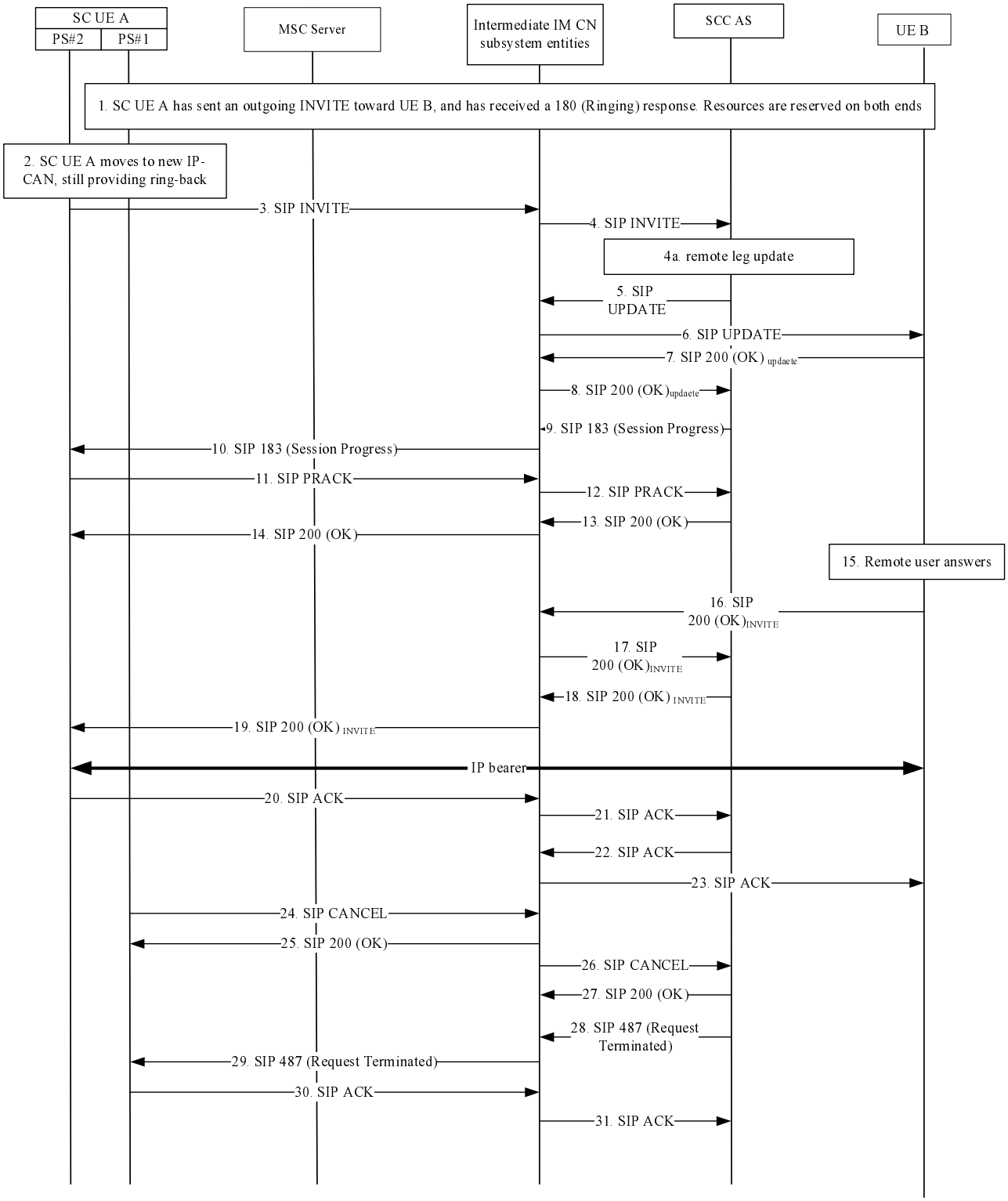


Figure A.7.4-1: Signalling flow for an outgoing call in the alert phase

1. **SC UE A has sent an INVITE request and subsequently has received a SIP 180 (Ringing) response and it is ringing**

   The SC UE A initiated call toward the UE B by sending an initial SIP INVITE request on the Source Access Leg, and subsequently it has received a SIP 180 (Ringing) response, and it is providing ring-back. The call has been anchored at the SCC AS of the SC UE A.

The dialog on the Source Access Leg is identified with "Call-ID= me03a0s09a2sdfgjkl491777", "From tag=64727891", and "To tag=774321".

2. **SC UE A attaches to different IP-CAN**

The SC UE A determines that a handover of the dialog on the Source Access Leg to a Target Access Leg is required while this dialog is in the alerting phase. The SC UE A connects to different IP-CAN and obtains a new IP address that it will use for the subsequent signalling and media. The SC UE A registers with the S-CSCF over the new IP-CAN using the standard multiple registration procedure. If needed, prior to sending the initial SIP INVITE request over the new IP-CAN, the SC UE A reserves resources in the new IP-CAN that will be needed for the signalling and the media.

3. **SIP INVITE request (SC UE A to intermediate IM CN subsystem entities) - see example in table A.7.5-3**

The SC UE A sends an initial SIP INVITE request over the new IP-CAN with a new SDP offer to the UE B that indicates that the new dialog on the Target Access Leg will replace the existing dialog on the Source Access Leg. The SDP offer in the initial SIP INVITE request sent on the Target Access Leg specifies the new contact address on the new IP-CAN that will be used for the media.

Table A.7.5-3: SIP INVITE request (SC UE A to intermediate IM CN subsystem entities)

```
INVITE tel:+1-212-555-2222 SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]:1357; branch=z9hG4bKnashds7
Max-Forwards: 70
Route: <sip:pcscf1.home1.net:7531;lr;comp=sigcomp>, <sip:orig@scscf1.home1.net;lr>
P-Preferred-Identity: "John Doe" <sip:user1_public1@home1.net>
P-Access-Network-Info:IEEE-802.11b
Privacy: none
From: <sip:user1_public1@home1.net>; tag=171828
To: <tel:+1-212-555-2222>
Call-ID: cb03a0s09a2sdfglkj490333
Cseq: 127 INVITE
Supported: 100rel, precondition, gruu, outbound
Require: sec-agree; replaces
Replaces: me03a0s09a2sdfgjkl491777; to-tag=774321; from-tag=64727891
Proxy-Require: sec-agree
Security-Verify: ipsec-3gpp; q=0.1; alg=hmac-sha-1-96; spi=87654321; port1=7531
Contact: <sip:user1_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91e6bf6;
   ob>;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel";+g.3gpp.ics="principal"
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE
Accept: application/sdp, application/3gpp-ims+xml
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c=IN IP6 5555::aaa:bbb:ccc:ddd
t=0 0
m=audio 3456 RTP/AVP 97 96
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:96 telephone-event
a=maxptime:20
```

Request-URI: the tel-URI of the destination, i.e. the UE-B.

Require: the "replaces" option tag indicate that the support for Replace header field is required.

Replaces: identifies the dialog on the Source Access that will be replaced with the new dialog on the Target Access Leg.

SDP: specifies the new IP address for media that the SC UE A has acquired in the new IP-CAN, and also indicates that the resources in the new IP-CAN have been acquired.

4. **SIP INVITE request transferring the session (intermediate IM CN subsystem entities to SCC AS)**

Based on the initial filter criteria in the S-CSCF, the initial SIP INVITE request is routed towards the SCC AS.

4a Remote Leg Update

The SCC AS correlates the initial SIP INVITE request received on the Target Access Leg to the dialog on the Source Access Leg and to the remote call leg. The SCC AS acting as a B2BUA generates a SIP UPDATE request, based on the information in the initial SIP INVITE request received on the Target Access Leg, and the information previously stored against these dialogs. The SIP UPDATE request contains the SDP offer that is identical to the SDP offer that the SCC AS received in the initial SIP INVITE request from the SC UE A on the Target Access Leg.

5. SIP UPDATE request (SCC AS to intermediate IM CN subsystem entities)

The SCC AS performs the remote call leg update by sending the SIP UPDATE request towards the UE B.

6. SIP UPDATE request (Intermediate IM CN subsystem entities to UE B)

The intermediate IM CN subsystem entities forward the SIP UPDATE request to the remote UE B.

7. SIP 200 (OK) response (UE B to Intermediate IM CN subsystem entities)

Upon receiving the SIP UPDATE request containing the SDP offer, the remote UE B sends a SIP 200 (OK) response. The SIP 200 (OK) response contains the SDP answer. The SDP answer indicates that the resources are available.

8. SIP 200 (OK) response (Intermediate IM CN subsystem entities to SCC AS)

The intermediate IM CN subsystem entities forward the SIP 200 (OK) response to the SCC AS.

9. SIP 183 (Session Progress) response (SCC AS to Intermediate IM CN subsystem entities)

The SCC AS sends a 183 (Session Progress) response on the Target Access Leg that contains the SDP answer as received from the remote UE B. The SDP answer indicates that the resources at the UE B are available.

10. SIP 183 (Session Progress) response (Intermediate IM CN subsystem entities to SC UE A)

The intermediate IM CN subsystem entities forward the 183 (Session Progress) response to the SC UE A.

11. SIP PRACK request (SC UE A to Intermediate IM CN subsystem entities)

The SC UE A acknowledges the receipt of the 183 Session Progress response.

12. SIP PRACK request (Intermediate IM CN subsystem entities to SCC AS)

The intermediate IM CN subsystem entities forward the SIP PRACK request to the SCC AS.

13. SIP 200 (OK) response (SCC AS to Intermediate IM CN subsystem entities)

The SCC AS acknowledges the receipt of the SIP PRACK request.

14. SIP 200 (OK) response (Intermediate IM CN subsystem entities to SC UE A)

The intermediate IM CN subsystem entities forward the SIP 200 (OK) response to the SC UE A. Upon successful exchange of the SIP 183 (Session Progress) response and the SIP PRACK request on the Target Access Leg, the early dialog and associated media has been transferred from the Source Access Leg to the Target Access Leg. Since the resources for media on the Source Access Leg are not used any more, the SC UE A releases the resources that the SC UE A was using for media on the Source Access Leg. In spite of releasing the resources, the early dialog on the Source Access Leg is still in the alerting phase.

NOTE 4: For clarity, the exchange of the SIP messages and associated SDPs on the Source Access Leg, to release the resources that the SC UE A was using for media on the Source Access Leg, is not shown in the signalling flow.

15. Remote user answers the call

16. SIP 200 (OK) response (UE B to intermediate IM CN subsystem entities)

The UE B accepts the call and sends a SIP 200 (OK) response to the initial INVITE request received from the SC UE A..

17. SIP 200 (OK) response (Intermediate IM CN subsystem entities to SCC AS)

The SIP 200 (OK) response is forwarded to the SCC AS.

18. SIP 200 (OK) response (SCC AS to intermediate IM CN subsystem entities)

The SCC AS acting as a B2BUA generates the SIP 200 (OK) response to initial SIP INVITE request that it has received on the Target Access Leg , that indicate that the remote UE B has accepted the call.

19. SIP 200 (OK) response (Intermediate IM CN subsystem entities to SC UE A)

The SIP 200 (OK) response is forwarded to the SC UE A.

20. SIP ACK request (SC UE A to intermediate IM CN subsystem entities)

The SC UE A acknowledges the SIP 200 (OK) response received from the SCC AS

21. SIP ACK request (Intermediate IM CN subsystem entities to SCC AS)

The SIP ACK request is forwarded to the SCC AS.

22. SIP ACK request (SCC AS to intermediate IM CN subsystem entities)

The SCC AS acknowledges the SIP 200 (OK) response received from the UE B.

23 SIP ACK request (Intermediate IM CN subsystem entities to UE B)

The SIP ACK request is forwarded towards the UE B.

24-31 SIP CANCEL Processing

The SC UE A cancels the SIP INVITE request sent on the Source Access Leg towards the SCC AS.

A.7.5 PS-PS Access Transfer with full media transfer for an incoming call in alerting phase

The signalling flows shown in the figure A.7.5-1 describes the PS-to-PS access transfer procedure when an incoming dialog that is in alerting phase is transferred from one contact address of the SC UE A (using the old IP-CAN) to a different contact address of the same SC UE A (using a different IP-CAN). In this example flow the SC UE A has an incoming dialog which is anchored at the SCC AS. While the dialog on the Source Access Leg (using the old IP-CAN) is in the alerting phase, the SC UE A decides (e.g. based on the measurement reports) to transfer this dialog to the Target Access Leg that will be established over the new IP-CAN. The the SCC AS and SC UA A support the PS-to-PS access transfer for the dialogs in early dialog phase.

NOTE 1: This scenario requires that the SC UE A supports dual mode operation and multiple registration procedure.

NOTE 2: In this example flow, each call leg is uniquely identified with a respective dialog identifier consisting of the Call-ID, From tag, and To tag.

NOTE 3: For clarity, the SIP 100 (Trying) responses are not shown in the signalling flow.

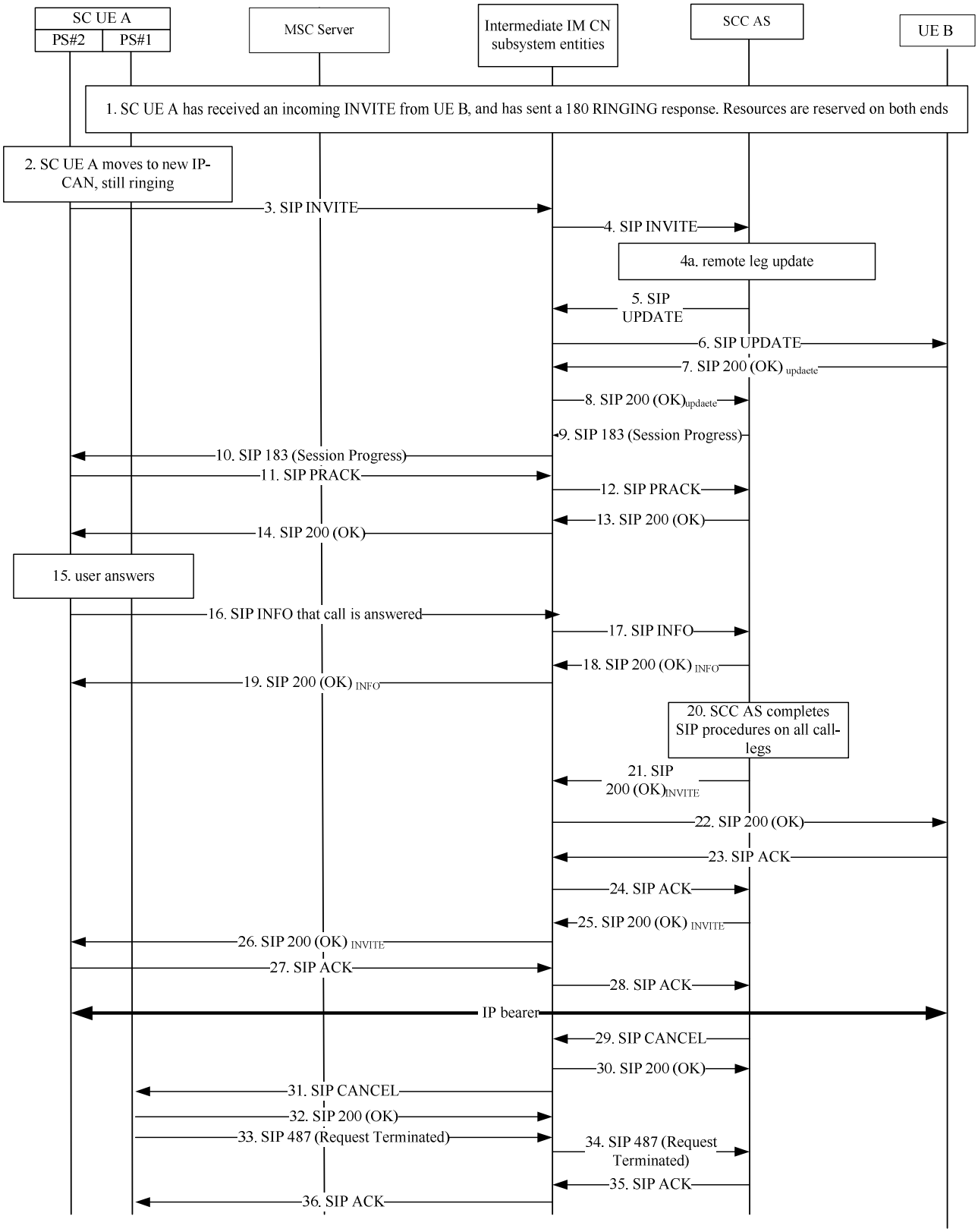


Figure A.7.5-1: Signalling flow for an incoming dialog in the alert phase

1. **SC UE A has received an incoming call and is in Ringing State**

   The incoming call has been anchored at the SCC AS of the SC UE A. If needed, both ends have reserved the resources, and the SC UE A has sent a SIP 180 (Ringing) response to the initial SIP INVITE request received on the Source Access Leg.

The dialog on the Source Access Leg is identified with "Call-ID= me03a0s09a2sdfgjkl491777", "From tag=64727891", and "To tag=774321".

2. **SC UE A attaches to different IP-CAN**

The SC UE A determines that a handover of the dialog on the Source Access Leg to a Target Access Leg is required while this dialog is in the alerting phase. The SC UE A connects to different IP-CAN and obtains new IP address that it will use for the subsequent signalling and media. The SC UE A registers with the S-CSCF over the new IP-CAN using the standard multiple registration procedure. If needed, prior to sending the initial SIP INVITE request over the new IP-CAN, the SC UE A reserves resources in the new IP-CAN that will be needed for the signalling and media.

3. **SIP INVITE request (SC UE A to intermediate IM CN subsystem entities) - see example in table A.7.5-3**

The SC UE A sends an initial SIP INVITE request on the Target Access Leg with a new SDP offer toward the UE B and indicates that the new dialog on the Target Access Leg will replace the existing early dialog on the Source Access Leg. The SDP offer in the initial SIP INVITE request sent on the Target Access Leg specifies the new contact address that will be used for the media over the new IP-CAN.

Table A.7.5-3: SIP INVITE request (SC UE A to intermediate IM CN subsystem entities)

```
INVITE tel:+1-212-555-2222 SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]:1357; branch=z9hG4bKnashds7
Max-Forwards: 70
Route: <sip:pcscf1.home1.net:7531;lr;comp=sigcomp>, <sip:orig@scscf1.home1.net;lr>
P-Preferred-Identity: "John Doe" <sip:user1_public1@home1.net>
P-Access-Network-Info:IEEE-802.11b
Privacy: none
From: <sip:user1_public1@home1.net>; tag=171828
To: <tel:+1-212-555-2222>
Call-ID: cb03a0s09a2sdfglkj490333
Cseq: 127 INVITE
Supported: 100rel, precondition, gruu, outbound
Require: sec-agree; replaces
Replaces: me03a0s09a2sdfgjkl491777; to-tag=774321; from-tag=64727891
Proxy-Require: sec-agree
Security-Verify: ipsec-3gpp; q=0.1; alg=hmac-sha-1-96; spi=87654321; port1=7531
Contact: <sip:user1_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91e6bf6;
   ob>;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel";+g.3gpp.ics="principal"
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE
Accept: application/sdp, application/3gpp-ims+xml
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c=IN IP6 5555::aaa:bbb:ccc:ddd
t=0 0
m=audio 3456 RTP/AVP 97 96
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:96 telephone-event
a=maxptime:20
```

Request-URI: the tel-URI of the destination, i.e. the UE-B.

Require: the "replaces" option tag indicate that the support for Replace header field is required.

Replaces: that identifies the dialog on the Source Access Leg that will be replaced with the new dialog on the Target Access Leg.

SDP: specifies the new IP address for media that the SC UE A has acquired in the new IP-CAN, and also indicates that the resources in the new IP-CAN have been acquired.

4. **SIP INVITE request transferring the session (intermediate IM CN subsystem entities to SCC AS)**

Based on the initial filter criteria in the S-CSCF, the initial SIP INVITE request is routed towards the SCC AS.

4a. Remote Leg Update

The SCC AS correlates the initial SIP INVITE request received on the Target Access Leg to the dialog on the Source Access Leg, and to the remote call leg. The SCC AS acting as a B2BUA generates a SIP UPDATE request, based on the information received in the initial SIP INVITE request on the Target Access Leg, and the information previously stored against these dialogs. The SIP UPDATE request contains the SDP offer that is identical to the SDP offer that the SCC AS received in the initial SIP INVITE request from the SC UE A on the Target Access Leg.

5. SIP UPDATE request (SCC AS to intermediate IM CN subsystem entities)

The SCC AS performs the remote call leg update by sending the SIP UPDATE request towards the UE B.

6. SIP UPDATE request (Intermediate IM CN subsystem entities to UE B)

The intermediate IM CN subsystem entities forward the SIP UPDATE request to the remote UE B.

7. SIP 200 (OK) response (UE B to Intermediate IM CN subsystem entities)

Upon receiving the SIP UPDATE request containing the SDP offer, the remote UE B sends a SIP 200 (OK) response. The SIP 200 (OK) response contains the SDP answer. The SDP answer indicates that the resources are available.

8. SIP 200 (OK) response (Intermediate IM CN subsystem entities to SCC AS)

The intermediate IM CN subsystem entities forward the SIP 200 (OK) response to the SCC AS.

9. SIP 183 (Session Progress) response (SCC AS to Intermediate IM CN subsystem entities)

The SCC AS sends a 183 (Session Progress) response on the Target Access Leg that contains the SDP answer as received from the remote UE B. The SDP answer indicates that resources are available. The SIP 183 (Session Progress) response will contain a Recv-Info header field set to g.3gpp.state-and-event.

10. SIP 183 (Session Progress) response (Intermediate IM CN subsystem entities to SC UE A**)**

The intermediate IM CN subsystem entities forward the 183 (Session Progress) response to the SC UE A.

11. SIP PRACK request (SC UE A to Intermediate IM CN subsystem entities)

The SC UE A acknowledges the receipt of the 183 (Session Progress) response.

12. SIP PRACK request (Intermediate IM CN subsystem entities to SCC AS)

The intermediate IM CN subsystem entities forward the SIP PRACK request to the SCC AS.

13. SIP 200 (OK) response (SCC AS to Intermediate IM CN subsystem entities)

The SCC AS acknowledges the SIP PRACK request.

14. SIP 200 (OK) response (Intermediate IM CN subsystem entities to SC UE A)

The intermediate IM CN subsystem entities forward the SIP 200 (OK) response to the SC UE A. Upon successful exchange of the SIP 183 (Session Progress) response and the SIP PRACK request on the Target Access Leg, the early dialog and associated media has been transferred from the Source Access Leg to the Target Access Leg. Since the resources for media on the Source Access Leg are not used any more, the SC UE A releases the resources that the SC UE A was using for media on the Source Access Leg. In spite of releasing the resources, the early dialog on the Source Access Leg is still in the alerting phase.

NOTE 4: For clarity, the exchange of the SIP messages and associated SDPs on the Source Access Leg, to release the resources that the SC UE A was using for media on the Source Access Leg, is not shown in the signalling flow.

15. User answers the call

16. The SC UE A accepts the call and sends SIP INFO request to intermediate IM CN subsystem entities see example in table A.7.5-16

The SC UE sends a SIP INFO request that indicates that the call has been accepted.

Table A.7.5-16: SIP INFO request (SC UE A to intermediate IM CN subsystem entities)

```
INFO tel:+1-212-555-2222 SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]:1357; branch=z9hG4bKnashds7
Max-Forwards: 70
Route: <sip:pcscf1.home1.net:7531;lr;comp=sigcomp>, <sip:orig@scscf1.home1.net;lr>
From: <sip:user1_public1@home1.net>; tag=171828
To: <tel:+1-212-555-2222>;tag=171828
Call-ID: cb03a0s09a2sdfglkj490333
Cseq: 130 INFO
Info-Package: g.3gpp.state-and-event
Content-Disposition: Info-Package
Content-Type: application/vnd.3gpp.state-and-event-info+xml
Content-Length:

<?xml version="1.0" encoding="UTF-8"?>
<state-and-event-info>
   <event>call-accepted</event>
</state-and-event-info>
```

17. SIP INFO request (Intermediate IM CN subsystem entities to SCC AS)

The intermediate IM CN subsystem entities forward the SIP INFO request to the SCC AS. The SCC AS gets informed that the SC UE A has accepted the call.

18. SIP 200 (OK) response (SCC AS to intermediate IM CN subsystem entities)

The SCC AS acknowledges the receipt of the SIP INFO request indicating that the SC UE A has accepted the call.

19. SIP 200 (OK) response (Intermediate IM CN subsystem entities to SC UE A)

The SIP 200 (OK) response is forwarded to the SC UE A.

20. SCC AS completes the SIP procedure on all three call-legs

The SCC AS generates a SIP 200 (OK) response toward the UE B, a SIP 200 (OK) response toward the UE A on the Target Access Leg, and a SIP CANCEL request toward the UE A on the Source Access Leg.

21. SIP 200 (OK) response (SCC AS to intermediate IM CN subsystem entities)

The SCC AS sends a SIP 200 (OK) response to UE B that indicates that the SC UE A has accepted the call.

22. SIP 200 (OK) response (Intermediate IM CN subsystem entities to the UE B)

The SIP 200 (OK) response is forwarded to the UE B.

23. SIP ACK request (UE B to intermediate IM CN subsystem entities)

The remote UE B acknowledges the SIP 200 (OK) response received from SCC AS by sending a SIP ACK request.

24. SIP ACK request (Intermediate IM CN subsystem entities to SCC AS)

The SIP ACK request is forwarded to the SCC AS.

25. SIP 200 (OK) response (SCC AS to intermediate IM CN subsystem entities)

The SCC AS sends a SIP 200 (OK) response to the SC UE A on the Target Access Leg to indicate the successful access transfer.

26. SIP 200 (OK) response (Intermediate IM CN subsystem entities to SC UE A)

The SIP 200 (OK) response is forwarded to the SC UE A.

27. SIP ACK request (SC UE A to intermediate IM CN subsystem entities)

SC UE A acknowledges the receipt of the 200 (OK) response received on the Target Access Leg from SCC AS by sending a SIP ACK request.

28. SIP ACK request (Intermediate IM CN subsystem entities to SCC AS)

The SIP ACK request is forwarded to the SCC AS**.**

29-36 SIP CANCEL Processing

The SCC AS cancels the early dialog on the Source Access Leg.

A.8 Signalling flows for PS-PS access transfer in conjunction with PS-CS access transfer

A.8.1 Introduction

The signalling flows for PS-PS access transfer conjunction with PS-CS access transfer demonstrate how a multimedia session is transferred from Source Access Leg to the Target Access Leg. The following signalling flows are included:

- subclause A.8.2 shows an example when a multimedia session is transferred from one IP-CAN to a new IP-CAN and the CS bearer respectively ; and
- subclause A.8.3 shows an example when a multimedia session is transferred from one IP-CAN and CS bearer to a new IP-CAN.

A.8.2 PS - PS in conjunction with PS - CS Access Transfer: PS to CS

In this example, SC UE A has an ongoing multimedia session with remote UE B over IP-CAN#1 before access transfer. When SC UE connects to a new IP-CAN#2, it decides to transfer the multimedia session over the new IP-CAN#2 and the CS bearer respectively.

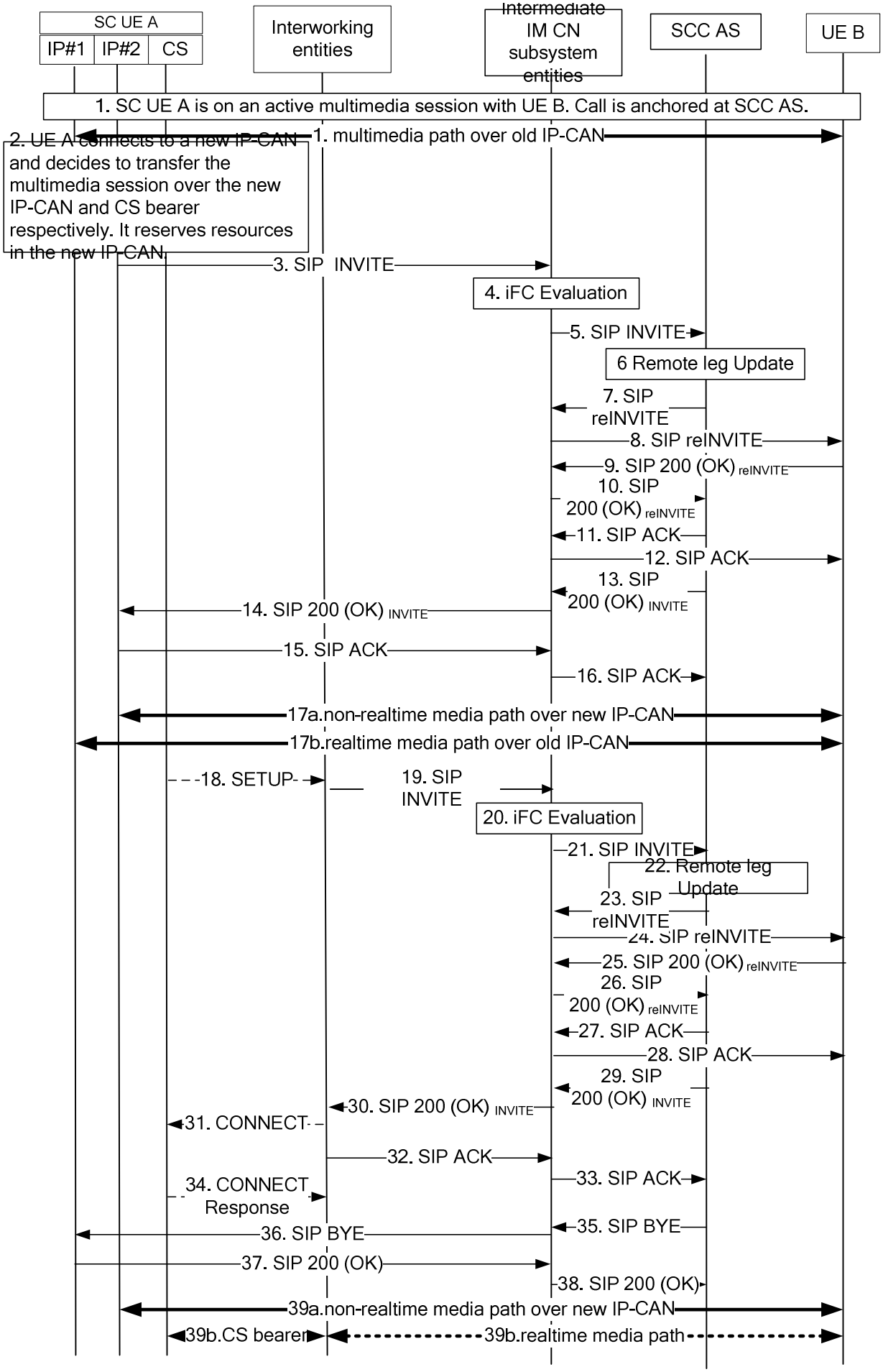

SC UE A
IP#1
IP#2
CS
Interworking entities
Intermediate IM CN subsystem entities
SCC AS
UE B
1. SC UE A is on an active multimedia session with UE B. Call is anchored at SCC AS.
1. multimedia path over old IP-CAN
2. UE A connects to a new IP-CAN and decides to transfer the multimedia session over the new IP-CAN and CS bearer respectively. It reserves resources in the new IP-CAN.
3. SIP INVITE
4. iFC Evaluation
5. SIP INVITE
6 Remote leg Update
7. SIP reINVITE
8. SIP reINVITE
9. SIP 200 (OK) reINVITE
10. SIP 200 (OK) reINVITE
11. SIP ACK
12. SIP ACK
13. SIP 200 (OK) INVITE
14. SIP 200 (OK) INVITE
15. SIP ACK
16. SIP ACK
17a.non-realtime media path over new IP-CAN
17b.realtime media path over old IP-CAN
18. SETUP
19. SIP INVITE
20. iFC Evaluation
21. SIP INVITE
22. Remote leg Update
23. SIP reINVITE
24. SIP reINVITE
25. SIP 200 (OK) reINVITE
26. SIP 200 (OK) reINVITE
27. SIP ACK
28. SIP ACK
29. SIP 200 (OK) INVITE
30. SIP 200 (OK) INVITE
31. CONNECT
32. SIP ACK
33. SIP ACK
34. CONNECT Response
35. SIP BYE
36. SIP BYE
37. SIP 200 (OK)
38. SIP 200 (OK)
39a.non-realtime media path over new IP-CAN
39b.CS bearer
39b.realtime media path

Figure A.8.2-1: Signalling flow for PS - PS in conjunction with PS - CS Access Transfer: PS to CS

NOTE 1: For clarity, the SIP 100 (Trying) responses are not shown in the signalling flow.

1. SC UE A has an ongoing multimedia session with remote UE B

The call has been anchored at the SCC AS which is in the HPLMN of originating SC UE A. The call leg over old IP-CAN is identified with "Call-ID= me03a0s09a2sdfgjkl491777", "From tag=64727891", and "To tag=774321". The UE A and UE B exchange media over the old IP-CAN, which is maintained while the SC UE A initiates the handover procedure.

Table A.8.2-1 shows an example of the SDP offer from SC UE A to remote UE B.

NOTE 2: To later show how the media is transferred to the new IP-CAN and CS bearer, only the SDP offer is shown in table A.8.2-1.

Table A.8.2-1: SIP INVITE request (SC UE A to intermediate IM CN subsystem entities)

```
INVITE tel:+1-237-555-2222 SIP/2.0
Via:
Max-Forwards:
Route:
P-Asserted-Identity:
P-Charging-Vector:
P-Access-Network-Info:
Privacy:
From:
To:
Call-ID:
Cseq:
Supported:
Require:
Proxy-Require:
Security-Verify:
Contact:
Allow:
Accept:
Content-Type:
Content-Length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=
c=IN IP6 5555::aaa:bbb:ccc:ddd
t=0 0
m=audio 3456 RTP/AVP 97 96
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:96 telephone-event
a=maxptime:20
m=message 7654 TCP/MSRP 98
a=accept-types:text/plain
```

2. SC UE A connects to a new IP-CAN#2:

The SC UE A decides to transfer the multimedia session over the new IP-CAN and CS bearer respectively. The UE A obtains an IP address that it will use for the signalling and media. It registers with the S-CSCF over the new IP-CAN using multiple registrations procedure. Depending on the UE A configuration, the discovery of the P-CSCF in the new IP-CAN can be needed. Based on the UE A and new IP-CAN capabilities, the UE A decides to use the same codec that was used over the old IP-CAN. The UE A reserves resources (e.g. QoS) in the new IP-CAN that will be needed for the signalling and transferred media, prior to sending the initial SIP INVITE request.

3. SIP INVITE request (SC UE A to intermediate IM CN subsystem entities)- see example in table A.8.2-3

The SC UE A sends an initial SIP INVITE request with a STI and a new SDP offer to the UE B that indicates that the new call replaces the existing call. The initial SIP INVITE request establishes a dialog for signalling and specifies in the SDP a new contact address that will be used for non-realtime media over the new IP-CAN. Upon sending the initial SIP INVITE request, the UE A is ready to receive the RTP packets either over the new IP-CAN or the old IP-CAN. The RTP packets can arrive over the new IP-CAN prior to the SC UE are receiving the SIP 200 (OK) response for the initial SIP INVITE request.

Table A.8.2-3: SIP INVITE request (UE A to intermediate IM CN subsystem entities)

```
INVITE tel:+1-237-555-2222 SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:fff]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Max-Forwards: 70
Route: sip:pcscf1.home1.net:7531;lr;comp=sigcomp>, <sip:orig@scscf1.home1.net;lr>
P-Preferred-Identity: "John Doe" <sip:user1_public1@home1.net>
P-Access-Network-Info: IEEE-802.11b
Privacy: none
From: <sip:user1_public1@home1.net>; tag=171828
To: <tel:+1-237-555-2222>
Call-ID: cb03a0s09a2sdfglkj490237
Cseq: 127 INVITE
Supported: 100rel; precondition
Require: sec-agree
Proxy-Require: sec-agree
Security-Verify: ipsec-3gpp; q=0.1; alg=hmac-sha-1-96; spi=87654321; port1=7531
Contact: <sip:user1_public1@home1.net; gr=urn:uuid:f81d4fae-7dec-11d0-a765-
   00a0c91e6bf6>;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-
   service.ims.icsi.mmtel";+g.3gpp.ics="principal";
Target-Dialog:me03a0s09a2sdfgjkl491777; to-tag=774321; from-tag=64727891
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE
Accept: application/sdp, application/3gpp-ims+xml
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:fff
s=
t=0 0
m=audio 0 RTP/AVP 97 96
c=IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:96 telephone-event
a=maxptime:20
m=message 7654 TCP/MSRP 98
c=IN IP6 5555::aaa:bbb:ccc:fff
a=accept-types:text/plain
```

4. **Evaluation of initial filter criteria**

   The S-CSCF evaluates initial filter criteria for the served SC user and as a result routes the SIP INVITE request towards the SCC AS.

5. **SIP INVITE request (intermediate IM CN subsystem entities to SCC AS)**

   The SIP INVITE request is forwarded to the SCC AS as the result of the evaluation of iFC.

6. **Remote Leg Update**

   The SCC AS identifies the session to be transferred using the STI. The SCC AS performs the Remote Leg update by sending the SIP re-INVITE request towards the remote UE.

7. **SIP re-INVITE request (SCC AS to intermediate IM CN subsystem entities)- See example in table A.8.2-7**

   The SCC AS modifies the message in accordance with routeing B2BUA functionality, e.g. mapping of From, To, Cseq and Call-ID header fields from one side of the B2BUA to the other. In this example the SCC AS includes the contents of the Contact header field from the received SIP INVITE request. The SIP re-INVITE

request contains the SDP offer that is identical to the SDP offer that the SCC AS received in the initial SIP INVITE request from the UE A (Step 3).

Table A.8.2-7: SIP **re-INVITE request (SCC AS to intermediate IM CN subsystem entities)**

```
INVITE < sip:user2_public1@home2.net;gr=urn:uuid:2ad8950e-48a5-4a74-8d99-ad76cc7fc74>; SIP/2.0
Via: SIP/2.0/UDP sccas.home1.net; branch=z9hG4bK332b33.3;
Max-Forwards: 67
Route: <scscf1.home1.net;lr >,<sip:scscf2.home2.net;lr>, <sip:pcscf2.visited2.net;lr>
P-Asserted-Identity: "John Doe" <sip:user1_public1@home1.net>, <tel:+1-237-555-1111>
P-Access-Network-Info: IEEE-802.11b
Privacy: none
P-Charging-Vector: ####
P-Charging-Function-Addresses:
From: <sip:user1_public1@home1.net>; tag=1717777
To: <tel:+1-237-555-2222>, tag=4321
Call-ID: dc14b1t10b3teghmlk5013237
Cseq: 111 INVITE
Supported: precondition, 100rel
Contact:< sip:user1_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-
   00a0c91e6bf6>;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE
Accept: application/sdp
Content-Type: application/sdp
Content-Length: (…)
v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:fff
s=t=0 0
m=audio 0 RTP/AVP 97 96
c=IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:96 telephone-event
a=maxptime:20
m=message 7654 TCP/MSRP 98
c=IN IP6 5555::aaa:bbb:ccc:fff
a=accept-types:text/plain
```

8. SIP re-INVITE request (Intermediate IM CN subsystem entities to UE B)

The intermediate IM CN subsystem entities forwards the SIP re-INVITE request to remote UE B.

9-10: SIP 200 (OK) response (UE B to SCC AS via Intermediate IM CN subsystem entities)

The UE B generates the SIP 200 (OK) response to the SIP re-INVITE request and forwards it to the SCC AS.

11-12: SIP ACK request (SCC AS to UE B via Intermediate IM CN subsystem entities)

The SCC AS generates the SIP ACK request to the SIP 200 (OK) response and forwards it to the remote UE B.

13-14: SIP 200 (OK) response (SCC AS to UE A via Intermediate IM CN subsystem entities)

The SCC AS generates the SIP 200 (OK) response to the SIP INVITE request and forwards it to the SC UE A.

15-16: SIP ACK request (SC UE A to SCC AS via Intermediate IM CN subsystem entities)

The SC UE A generates the SIP ACK request to the SIP 200 (OK) response and forwards it to the SCC AS

17. Media paths between UE A and UE B

The non-realtime media is using the new IP-CAN while the realtime media path is still over the old IP-CAN.

18. CC SETUP message (SC UE A to Interworking entities)

The SC UE sends the CC SETUP message with the STN as the called party number.

NOTE 3: STN is a PSI DN used by the UE to request a session transfer towards the SCC AS.

19. SIP INVITE request (Interworking entities to Intermediate IM CN subsystem entities) -see example in Table A.8.2-19

Table A.8.2-19: SIP INVITE request (interworking entities to intermediate IM CN subsystem entities)

```
INVITE tel:+1-237-555-3333 SIP/2.0
Via: SIP/2.0/UDP msc1.home1.net; branch=z9hG4bKnashds7
Max-Forwards: 70
Route: <sip:icscf1.home1.net:7531;lr;comp=sigcomp>
P-Asserted-Identity: <tel: +1-237-555-1111>
P-Charging-Vector: ####
Privacy: none
From: <tel: +1-237-555-1111>;tag=171828
To: <tel:+1-237-555-2222>
Call-ID: cb03a0s09a2sdfglkj490333
Cseq: 127 INVITE
Supported: 100rel, precondition
Require: sec-agree
Proxy-Require: sec-agree
Security-Verify: ipsec-3gpp; q=0.1; alg=hmac-sha-1-96; spi=87654321; port=7531
Contact: <sip:mgcf2.home2.net;gr>;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE
Accept: application/sdp, application/3gpp-ims+xml
Content-Type: application/sdp
Content-Length: (…)


v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:eee
s=
c=IN IP6 5555::aaa:bbb:ccc:eee
t=0 0
m=audio 3456 RTP/AVP 97 96
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:96 telephone-event
a=maxptime:20
```

Request-URI: contains the IMRN, as obtained from CS networks signalling.

SDP: The SDP contains preconfigured set of codecs supported by the MGW.

20. Evaluation of initial filter criteria

The S-CSCF evaluates initial filter criteria for the served SC user and as a result routes the SIP INVITE request towards the SCC AS.

21. SIP INVITE request (Intermediate IM CN subsystem entities to SCC AS)

22. Remote Leg Update

The SCC AS performs the Remote Leg update by sending the SIP re-INVITE request towards the remote UE.

23. SIP re-INVITE request (SCC AS to intermediate IM CN subsystem entities) –see example in table A.8.2-23

The SCC AS acting as a routing B2BUA generates a SIP INVITE request based upon the received SIP INVITE request and the information previously stored against this session and routes it towards UE B via the intermediate IM CN subsystem entities. In this example the SCC AS includes the contents of the Contact header field from the received SIP INVITE request.

Table A.8.2-23: SIP re-INVITE request (SCC AS to intermediate IM CN subsystem entities)

```
INVITE < sip:user2_public1@home2.net;gr=urn:uuid:2ad8950e-48a5-4a74-8d99-ad76cc7fc74> SIP/2.0
Via: SIP/2.0/UDP sccas1.home1.net;branch=z9hG4bKnas34r5
Max-Forwards: 67
Route: <sip:scscf1.home1.net:lr>
P-Asserted-Identity: <tel: +1-237-555-1111>
P-Charging-Function-Addresses: ####
P-Charging-Vector: ####
Privacy: none
From: <tel: +1-237-555-1111>;tag=171828
To: <tel:+1-237-555-2222>; tag=26545
Call-ID: cb03a0s09a2sdfglkj490333
Cseq: 127 INVITE
Supported: 100rel, precondition
Require: sec-agree
Proxy-Require: sec-agree
Security-Verify: ipsec-3gpp; q=0.1; alg=hmac-sha-1-96; spi=87654321; port=7531
Contact: < sip:user1_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-
   00a0c91e6bf6>;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE
Accept: application/sdp, application/3gpp-ims+xml
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:eee
s=
t=0 0
m=audio 3456 RTP/AVP 97 96
c=IN IP6 5555::aaa:bbb:ccc:eee
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:96 telephone-event
a=maxptime:20
m=message 7654 TCP/MSRP 98
c=IN IP6 5555::aaa:bbb:ccc:fff
a=accept-types:text/plain
```

24. SIP re-INVITE request (Intermediate IM CN subsystem entities to UE B)

Intermediate IM CN subsystem entities forward the SIP re-INVITE request to remote UE B.

25. SIP 200 (OK) response (UE B to intermediate IM CN subsystem entities)

Upon receiving the SIP re-INVITE request containing the SDP offer, since the UE B has all resources available, it sends immediately the SIP 200 (OK) response to the SIP re-INVITE request that contains the SDP answer. The SDP answer indicates that the resources are available.

26. SIP 200 (OK) response (intermediate IM CN subsystem entities to SCC AS)

The intermediate IM CN subsystem entities forward the SIP 200 (OK) response to the SIP re-INVITE request to the SCC AS in the originating network.

27-28. SIP ACK request (SCC AS to UE B via IM CN subsystem entities)

The SCC AS generates the SIP ACK request to the SIP 200 (OK) response, and forwards the SIP ACK request to the remote UE B.

29-30. SIP 200 (OK) response (SCC AS to interworking entities via IM CN subsystem entities)

The SCC AS generates the SIP 200 (OK) response to the SIP INVITE request, and forwards the SIP 200 (OK) response to the interworking entities.

31. CC CONNECT message (interworking entities to SC UE A)

32-33. SIP ACK request (interworking entities to SCC AS via IM CN subsystem entities)

The interworking entities generate the SIP ACK request to the SIP 200 (OK) response, and forward it to the SCC AS.

34. CC CONNECT ACKNOWLEDGEMENT message (SC UE A to interworking entities)

35-36: SIP BYE request (SCC AS to SC UE A via intermediate IM CN subsystem entities)

The SCC AS terminates the replaced call leg, which was using the old IP-CAN, by sending a SIP BYE request to the UE A.

37-38. SIP 200 (OK) response (SC UE A to SCC AS via intermediate IM CN subsystem entities)

Upon receiving the SIP BYE request over the old IP-CAN, the SC UE A sends a SIP 200 (OK) response over the old IP-CAN to the SCC AS. Subsequently, the SC UE A relinquishes all resources pertaining to the old IP-CAN.

39. Media paths between SC UE A and UE B

Finally, the non-realtime media path is over the new IP-CAN and the realtime media is using the CS bearer.

A.8.3 PS - PS in conjunction with PS - CS Access Transfer: CS to PS

In this example, SC UE A has an ongoing multimedia session with remote UE B over IP-CAN#1 and CS bearer before access transfer. When SC UE connects to a new IP-CAN#2, it decides to transfer all the multimedia session over the new IP-CAN#2.

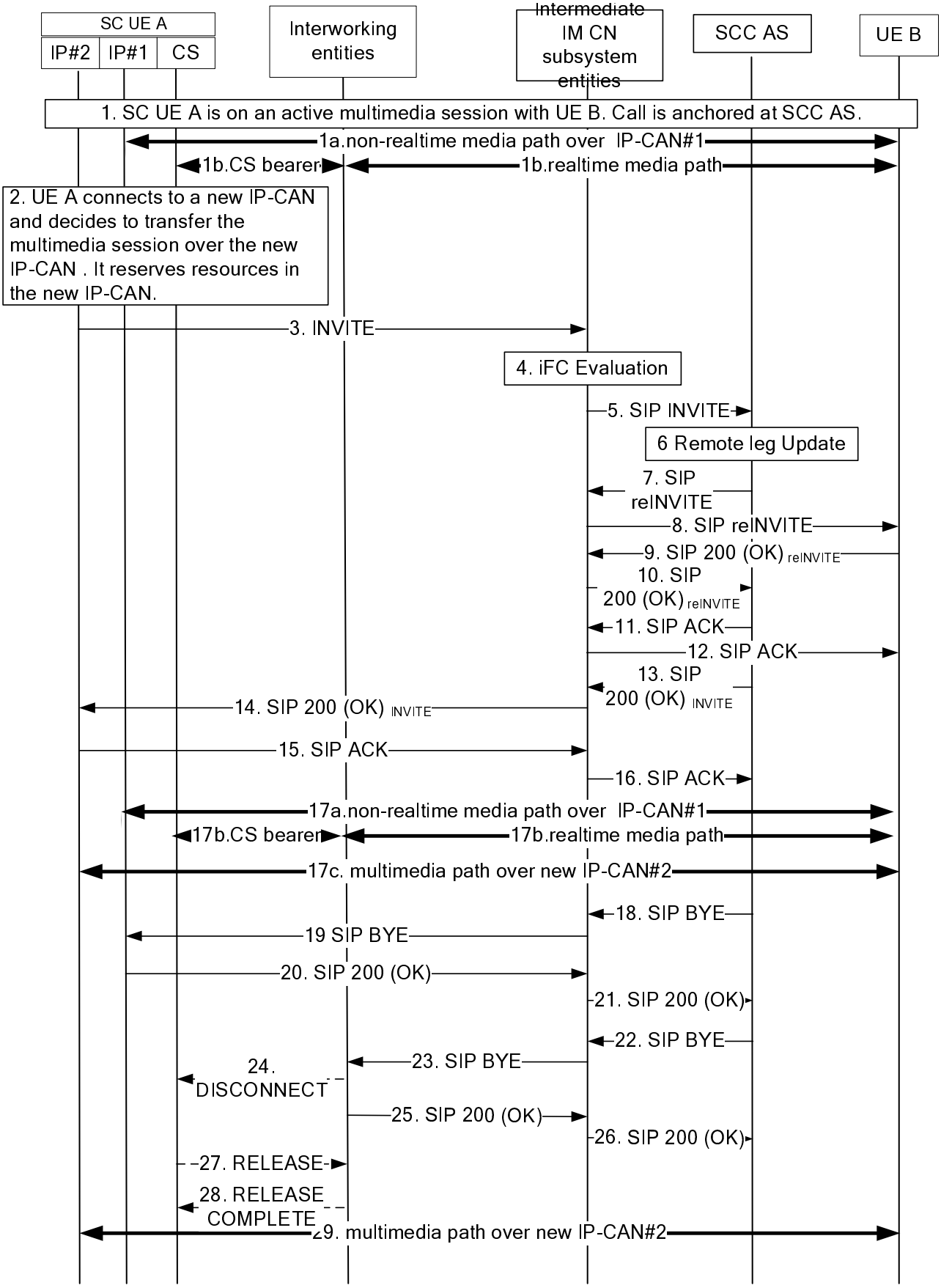


Figure A.8.3-1: Signalling flow for PS - PS in conjunction with PS - CS Access Transfer: CS to PS

NOTE 1: For clarity, the SIP 100 (Trying) responses are not shown in the signalling flow.

1. **SC UE A has an ongoing multimedia session with remote UE B**

   The non realmedia path is over old IP-CAN#1 and the realtime media path is over the CS bearer. The call has been anchored at the SCC AS which is in the HPLMN of originating SC UE A. The call leg over old IP-CAN#1 is identified with "Call-ID= me03a0s09a2sdfgjkl491777", "From tag=64727891", and "To tag=774321". The UE A and UE B exchange media over the old IP-CAN, which is maintained while the SC UE A initiates the handover procedure.

2. **SC UE A connects to a new IP-CAN#2**

   The SC UE A decides to transfer the multimedia session over the new IP-CAN#2. The UE A obtains an IP address that it will use for the signalling and media. It registers with the S-CSCF over the new IP-CAN using multiple registrations procedure. Depending on the UE A configuration, the discovery of the P-CSCF in the new IP-CAN can precede this. Based on the UE A and new IP-CAN capabilities, the UE A decides to use the same codec that was used over the old IP-CAN. The UE A reserves resources (e.g. QoS) in the new IP-CAN that will be needed for the signalling and transferred media, prior to sending the initial SIP INVITE request.

3. **SIP INVITE request (SC UE A to intermediate IM CN subsystem entities)- see example in table A.8.3-3**

   Upon sending the initial SIP INVITE request, the UE A is ready to receive the RTP packets either over the new IP-CAN or the old IP-CAN. The RTP packets can arrive over the new IP-CAN prior to the SC UE are receiving the SIP 200 (OK) response for the initial SIP INVITE request.

Table A.8.3-3: SIP INVITE request (UE A to intermediate IM CN subsystem entities)

```
INVITE tel:+1-237-555-3333 SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:fff]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Max-Forwards: 70
Route: sip:pcscf1.home1.net:7531;lr;comp=sigcomp>, <sip:orig@scscf1.home1.net;lr>
P-Preferred-Identity: "John Doe" <sip:user1_public1@home1.net>
P-Access-Network-Info: IEEE-802.11b
Privacy: none
From: <sip:user1_public1@home1.net>; tag=171828
To: <tel:+1-237-555-2222>
Call-ID: cb03a0s09a2sdfglkj490237
Cseq: 127 INVITE
Supported: 100rel; precondition, gruu, 199
Require: sec-agree, replaces
Proxy-Require: sec-agree
Accept-Contact: *;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
P-Preferred-Service: urn:urn-7:3gpp-service.ims.icsi.mmtel
Security-Verify: ipsec-3gpp; q=0.1; alg=hmac-sha-1-96; spi=87654321; port1=7531
Contact: <sip:user1_public1@home1.net;gr= urn:uuid:f81d4fae-7dec-11d0-a765-
   00a0c91e6bf6>;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-
   service.ims.icsi.mmtel";+g.3gpp.ics="principal";
Replaces: me03a0s09a2sdfgjkl491777; to-tag=774321; from-tag=64727891
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE
Accept: application/sdp, application/3gpp-ims+xml
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:fff
s=
c=IN IP6 5555::aaa:bbb:ccc:fff
t=0 0
m=audio 3456 RTP/AVP 97 96a=tcap:1 RTP/AVPF
a=pcfg:1 t=1
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:96 telephone-event
a=maxptime:20
m=message 7654 TCP/MSRP 98
a=accept-types: text/plain
```

Request-URI: Contains the static STI.

4. Evaluation of initial filter criteria

The S-CSCF evaluates initial filter criteria for the served SC user and as a result routes the SIP INVITE request towards the SCC AS.

5. SIP INVITE request (intermediate IM CN subsystem entities to SCC AS)

The SIP INVITE request is forwarded to the SCC AS as the result of the evaluation of iFC.

6. Remote Leg Update

The SCC AS identifies the session to be transferred using the STI. The SCC AS performs the Remote Leg update by sending the SIP re-INVITE request towards the remote UE.

7. SIP re-INVITE request (SCC AS to intermediate IM CN subsystem entities)- See example in table A.8.3-7

The SCC AS modifies the message in accordance with routeing B2BUA functionality, e.g. mapping of From, To, Cseq and Call-ID header fields from one side of the B2BUA to the other. In this example the SCC AS includes the contents of the Contact header field from the received SIP INVITE request. The SIP re-INVITE request contains the SDP offer that is identical to the SDP offer that the SCC AS received in the initial SIP INVITE request from the UE A (Step 3).

Table A.8.2-7: SIP re-INVITE request (SCC AS to intermediate IM CN subsystem entities)

```
INVITE sip:user2_public1@home2.net;gr=urn:uuid:2ad8950e-48a5-4a74-8d99-ad76cc7fc74 SIP/2.0
Via: SIP/2.0/UDP sccas.home1.net; branch=z9hG4bK332b33.3;
Max-Forwards: 67
Route: <scscf1.home1.net;lr >,<sip:scscf2.home2.net;lr>, <sip:pcscf2.visited2.net;lr>
P-Asserted-Identity: "John Doe" <sip:user1_public1@home1.net>, <tel:+1-237-555-1111>
P-Access-Network-Info: IEEE-802.11b
Privacy: none
P-Charging-Vector: icid-value="AyretyU0dm+6O2IrT5tAFrbHLso=023551024"
P-Charging-Function-Addresses:
From: <sip:user1_public1@home1.net>; tag=569812
To: <tel:+1-237-555-2222>, tag=4321
Call-ID: dc14b1t10b3teghmlk5013237
Cseq: 111 INVITE
Contact:<sip:user1_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-
   00a0c91e6bf6>;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:fff
s=
c=IN IP6 5555::aaa:bbb:ccc:fff
t=0 0
m=audio 3456 RTP/AVPF 97 96
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:96 telephone-event
a=maxptime:20
m=message 7654 TCP/MSRP 98
a=accept-types: text/plain
```

8. SIP re-INVITE request (Intermediate IM CN subsystem entities to UE B)

The intermediate IM CN subsystem entities forwards the SIP re-INVITE request to remote UE B.

9-10: SIP 200 (OK) response (UE B to SCC AS via Intermediate IM CN subsystem entities)

The UE B generates the SIP 200 (OK) response to the SIP re-INVITE request and forwards it to the SCC AS.

11-12: SIP ACK request (SCC AS to UE B via Intermediate IM CN subsystem entities)

The SCC AS generates the SIP ACK request to the SIP 200 (OK) response and forwards it to the remote UE B.

13-14: **SIP 200 (OK) response (SCC AS to UE A via Intermediate IM CN subsystem entities)**

The SCC AS generates the SIP 200 (OK) response to the SIP INVITE request and forwards it to the SC UE A.

15-16: **SIP ACK request (SC UE A to SCC AS via Intermediate IM CN subsystem entities)**

The SC UE A generates the SIP ACK request to the SIP 200 (OK) response and forwards it to the SCC AS

17. Media paths between UE A and UE B

The multimedia is using the new IP-CAN. Resources used for signalling on the old IP-CAN#1 and CS bearer are not released.

18-19. SIP BYE request (SCC AS to SC UE A via intermediate IM CN subsystem entities)

The SCC AS terminates the replaced call leg- that was using the old IP-CAN#1, by sending a SIP BYE request towards the SC UE A.

20-21. SIP 200 (OK) response (SC UE A to SCC AS via intermediate IM CN subsystem entities)

Upon receiving the SIP BYE request over the old IP-CAN#1, the SC UE A sends a SIP 200 (OK) response over the old IP-CAN. Subsequently, the UE-1 relinquishes all resources pertaining to the old IP-CAN.

22-23. **SIP BYE request (SCC AS to interworking entities via intermediate IM CN subsystem entities)**

The SCC AS terminates the replaced call leg, which was using the CS bearer, by sending a SIP BYE request.

24, 27-28. CC DISCONNECT message (interworking entities to SC UE A)

Upon receiving the CC DISCONNECT message, the SC UE A relinquishes all resources pertaining to the CS bearer.

25-26. **SIP 200 (OK) response (Interworking entities to SCC AS via intermediate IM CN subsystem entities)**

29. Media paths between UE A and UE B

The multimedia session is using the new IP-CAN#2.

A.9 Signalling flows for media adding/deleting for access transfer

A.9.1 Introduction

The signalling flows for media adding/deleting demonstrate how the media of a multimedia session is added or deleted. The following signalling flow is included:

- subclause A.9.2 shows an example when the non-realtime media of a multimedia session over the IP-CAN is removed.

A.9.2 Remote End Initiation case – Removing media from split CS and PS sessions

As a precondition the SC UE A has a CS call and IMS multimedia session with the remote UE after session transfer in a manner that more than one session are presented to UE B as one IMS session by the SCC AS.

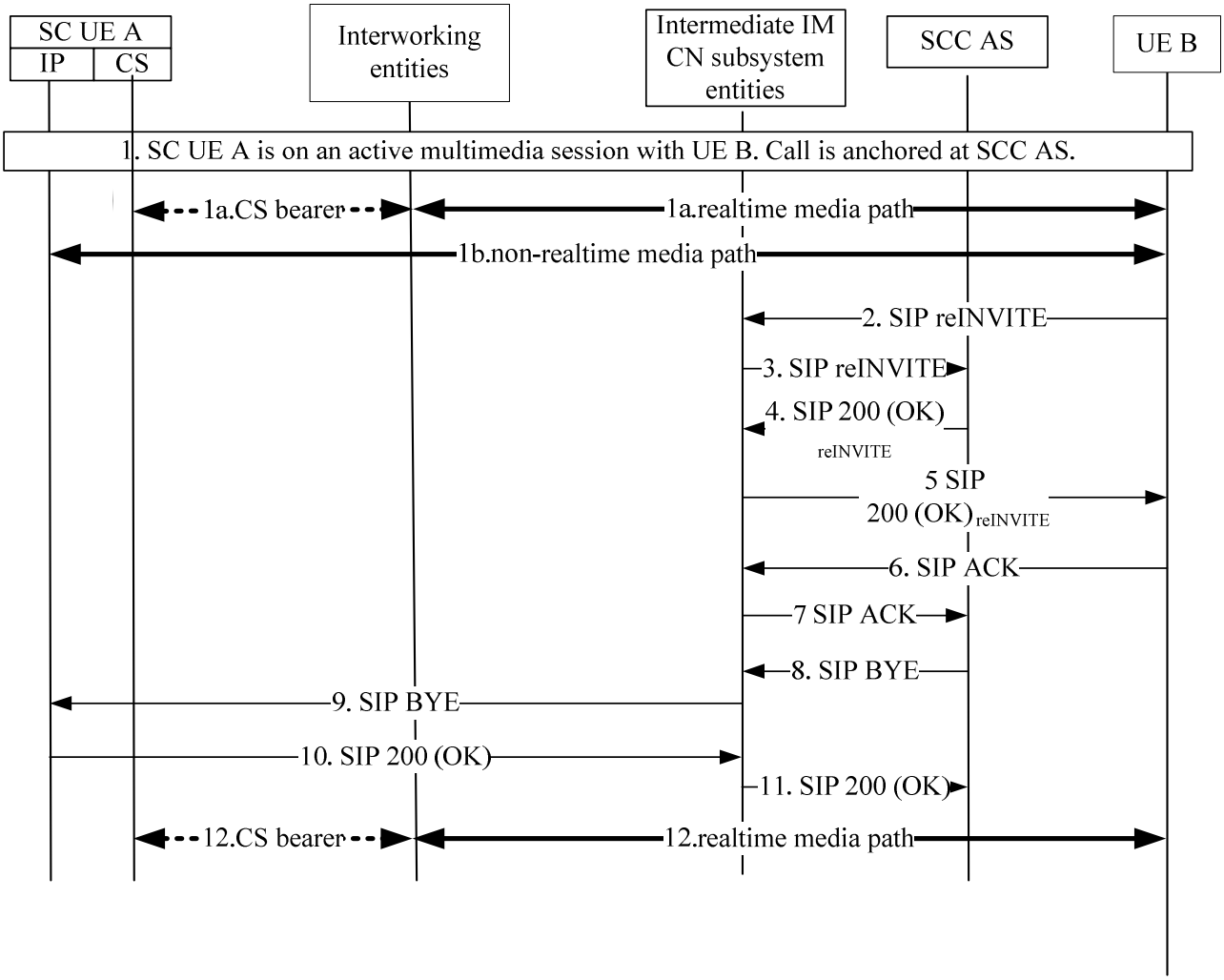


Figure A.9.2-1: Remote End Initiation case – Removing media from split CS and PS sessions

NOTE 1: For clarity, the SIP 100 (Trying) responses are not shown in the signalling flow.

1. SC UE A has an ongoing multimedia session with remote UE B

The call has been anchored at the SCC AS which is in the HPLMN of originating SC UE A.

Table A.9.2-1 shows an example of the SDP offer from SC UE A to remote UE B.

NOTE 2: To show how the media is removed, only the SDP offer is shown in this example.

Table A.9.2-1: SIP INVITE request (SC UE A to intermediate IM CN subsystem entities)

```
INVITE tel:+1-237-555-2222 SIP/2.0
Via:
Max-Forwards:
Route:
P-Asserted-Identity:
P-Charging-Vector:
P-Access-Network-Info:
Privacy:
From:
To:
Call-ID:
Cseq:
Supported:
Require:
Proxy-Require:
Security-Verify:
Contact:
Allow:
Accept:
Content-Type:
Content-Length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=
c=IN IP6 5555::aaa:bbb:ccc:ddd
t=0 0
m=message 7654 TCP/MSRP 98
a=accept-types:text/plain
```

2. **SIP re-INVITE request (UE B to intermediate IM CN subsystem entities)- See example in table A.9.2.-2**

   The remote UE B decides to remove the non-realtime media from the multimedia session. It uses standard IMS procedures to remove one or more PS media from the session.

Table A.9.2-2: SIP re-INVITE request (UE B to intermediate IM CN subsystem entities)

```
INVITE < sip:user1_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91e6bf6> SIP/2.0
Via: SIP/2.0/UDP sccas1.home1.net;branch=z9hG4bKnas34r5
Max-Forwards: 67
Route: <sip:scscf1.home1.net:lr>
P-Asserted-Identity: <tel: +1-237-555-2222>
P-Charging-Function-Addresses: ####
P-Charging-Vector: ####
P-Access-Network-Info:
Privacy: none
From: <tel: +1-237-555-2222; gr=hdg7777ad7aflzig8sf7>;tag=171828
To: <tel:+1-237-555-1111>
Call-ID: cb03a0s09a2sdfglkj490333
Cseq: 127 INVITE
Supported: 100rel, precondition
Require: sec-agree
Proxy-Require: sec-agree
Security-Verify: ipsec-3gpp; q=0.1; alg=hmac-sha-1-96; spi=87654321; port=7531
Contact: < sip:user2_public1@home2.net;gr=urn:uuid:2ad8950e-48a5-4a74-8d99-
   ad76cc7fc74>;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE
Accept: application/sdp, application/3gpp-ims+xml
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=
c=IN IP6 5555::aaa:bbb:ccc:ddd
t=0 0
m=audio 3456 RTP/AVP 97 96
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:96 telephone-event
a=maxptime:20
m=message 0 TCP/MSRP 98
a=accept-types:text/plain
```

3. SIP re-INVITE request (Intermediate IM CN subsystem entities to SCC AS)

4-5. SIP 200 (OK) response (SCC AS to UE B via Intermediate IM CN subsystem entities)

The SCC AS generates the SIP 200 (OK) response to the SIP re-INVITE request and forwards it to the remote UE B.

6-7: SIP ACK request (UE B to SCC AS via Intermediate IM CN subsystem entities)

The UE B generates the SIP ACK request to the SIP 200 (OK) response and forwards it to the SCC AS.

8-9: SIP BYE request (SCC AS to SC UE A via intermediate IM CN subsystem entities)

The SCC AS terminates the replaced call leg, which was using the IP-CAN, by sending a SIP BYE request to the UE A.

10-11. SIP 200 (OK) response (SC UE A to SCC AS via intermediate IM CN subsystem entities)

Upon receiving the SIP BYE request over the IP-CAN, the SC UE A sends a SIP 200 (OK) response over the IP-CAN to the SCC AS. Subsequently, the SC UE A relinquishes all resources pertaining to the IP-CAN.

12. Media paths between SC UE A and UE B

Finally, the non-realtime media path over the IP-CAN is removed.

A.10 Void

A.11 Void

A.12 Void

A.13 Void

A.14 Void

A.15 Signalling flows for MSC server assisted mid-call feature

A.15.1 Introduction

The signalling flows in the subclause demonstrate how full duplex session on hold can be transferred together with active full duplex session when the MSC server assisted mid-call feature is used. The following signalling flows are included:

- subclause A.15.2 shows an example of CS to PS access transfer with the MSC server assisted mid-call feature.
- subclause A.15.3 shows an example of PS to CS access transfer with the MSC server assisted mid-call feature.

  subclause A.15.4 shows an example of PS to CS access transfer with MSC server assisted mid-call feature with an incoming waiting call in alerting phase

The examples assume that:

- the SC UE, the MSC server enhanced for ICS and the SCC AS support the MSC server assisted mid-call feature;
- the SC UE does not use ICS procedures; and
- the SCC AS is allowed to use the MSC server assisted mid-call feature according to operator policy.

A.15.2 CS to PS access transfer with MSC server assisted mid-call feature

In the example flow at the figure A.15.2-1, SC UE A has two ongoing sessions over CS bearer which are anchored at SCC AS. The active session X is with UE B, the held session Y is with UE C. The session X and session Y are two party sessions. The session Y contains rejected video stream and accepted speech media component. When the SC UE connects to an IP-CAN, it decides to transfer the sessions over the IP-CAN.

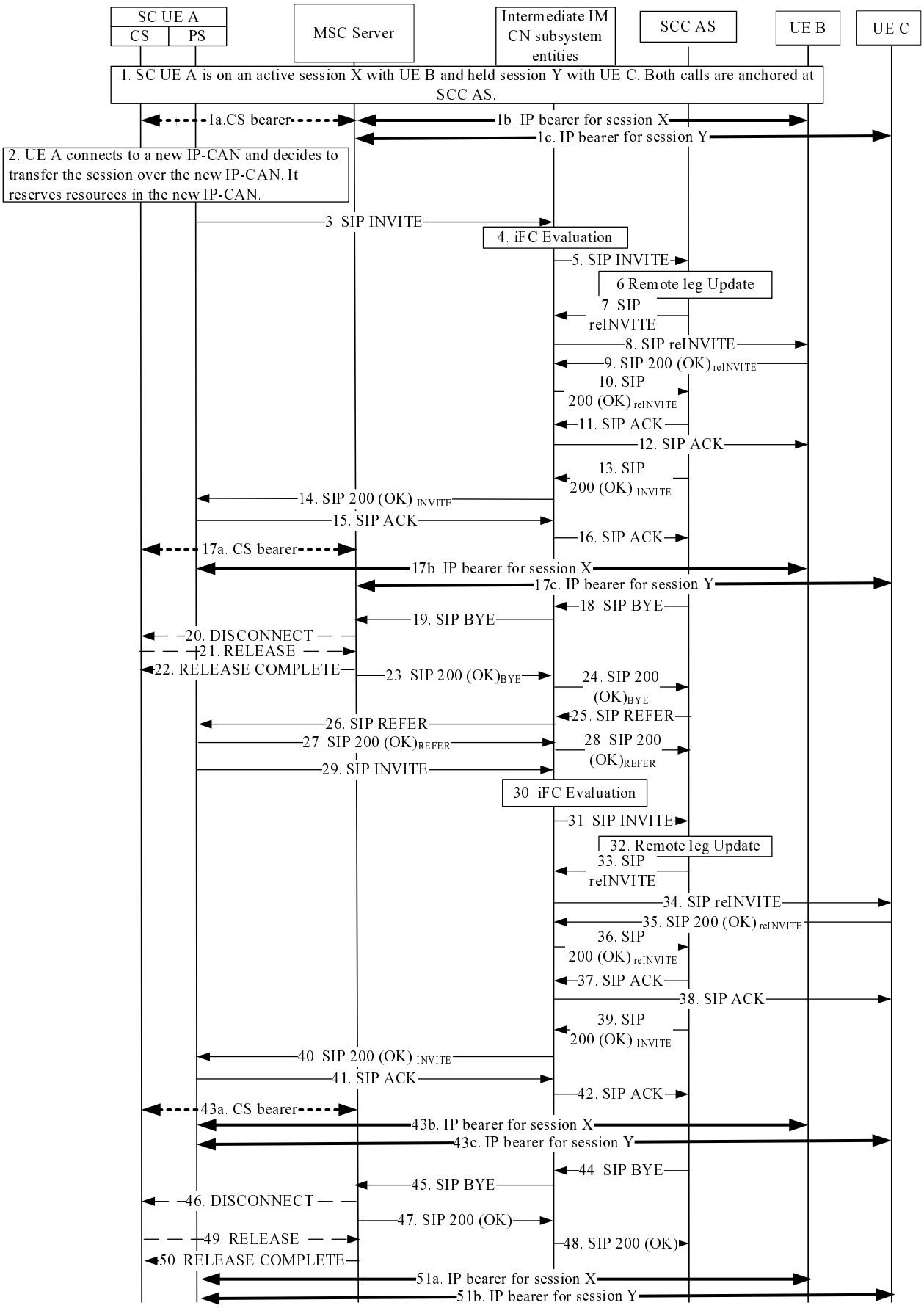


Figure A.15.2-1: Signalling flow for PS-CS Access Transfer: CS to PS

NOTE 1: For clarity, the SIP 100 (Trying) responses are not shown in the signalling flow.

1. **SC UE A has an ongoing active session X with remote UE B and a held session Y with remote UE C**

   The calls have been anchored at the SCC AS which is in the HPLMN of originating SC UE A.

2. **SC UE A connects to a new IP-CAN:**

   The SC UE A decides to transfer the sessions over the new IP-CAN. The UE A obtains an IP address that it will use for the signalling and media. It registers with the S-CSCF over the new IP-CAN using standard registration procedure and reserves resources in the new IP-CAN.

3. **SIP INVITE request transferring the active session X (SC UE A to intermediate IM CN subsystem entities) - see example in table A.15.2-3**

   The SC UE A sends an initial SIP INVITE request to request the new call replaces the existing call X.

Table A.15.2-3: SIP INVITE request (UE A to intermediate IM CN subsystem entities)

```
INVITE sip:domain.xfer@sccas.home1.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Max-Forwards: 70
Route: <sip:pcscf1.home1.net:7531;lr>, <sip:orig@scscf1.home1.net;lr>
P-Preferred-Identity: "John Doe" <sip:user1_public1@home1.net>
P-Access-Network-Info: IEEE-802.11b
Privacy: none
From: <sip:user1_public1@home1.net>; tag=171828
To: <tel:+1-237-555-2222>
Call-ID: cb03a0s09a2sdfglkj490237
Cseq: 127 INVITE
Supported: 100rel, precondition, 199, gruu, norefersub
Require: sec-agree
Proxy-Require: sec-agree
Security-Verify: ipsec-3gpp; q=0.1; alg=hmac-sha-1-96; spi=87654321; port1=7531
Contact: <sip:user1_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-
   00a0c91e6bf6> ;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel";+g.3gpp.mid-
   call
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE
Accept: application/sdp, application/3gpp-ims+xml, application/vnd.3gpp.mid-call+xml
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=
c=IN IP6 5555::aaa:bbb:ccc:ddd
t=0 0
m=audio 3456 RTP/AVP 97 96
a=tcap:1 RTP/AVPF
a=pcfg:1 t=1
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:96 telephone-event
a=maxptime:20
```

Contact: contains the g.3gpp.mid-call media feature tag as defined in annex C indicating the support for the MSC server assisted mid-call feature.

Accept: contains the MSC Server assisted mid-call feature MIME type.

4. **Evaluation of initial filter criteria**

   The S-CSCF evaluates initial filter criteria for the served SC user and as a result routes the SIP INVITE request towards the SCC AS.

5. **SIP INVITE request (intermediate IM CN subsystem entities to SCC AS)**

The SIP INVITE request is forwarded to the SCC AS as the result of the evaluation of iFC.

6. Remote Leg Update

The SCC AS performs the Remote Leg update by sending the SIP re-INVITE request towards the Remote Leg.

7. SIP re-INVITE request (SCC AS to intermediate IM CN subsystem entities)

The SCC AS modifies the message in accordance with routeing B2BUA functionality, e.g. mapping of From, To, Cseq and Call-ID header fields from one side of the B2BUA to the other. The SIP re-INVITE request contains the SDP offer that is identical to the SDP offer that the SCC AS received in the initial SIP INVITE request from the UE A (Step 3).

8. SIP re-INVITE request (Intermediate IM CN subsystem entities to UE B)

The intermediate IM CN subsystem entities forward the SIP re-INVITE request to remote UE B.

9-10: SIP 200 (OK) response (UE B to SCC AS via Intermediate IM CN subsystem entities)

The UE B generates the SIP 200 (OK) response to the SIP re-INVITE request and forwards it to the SCC AS.

11-12: **SIP ACK request (SCC AS to UE B via Intermediate IM CN subsystem entities)**

The SCC AS generates the SIP ACK request to the SIP 200 (OK) response and forwards it to the remote UE B.

13-14: SIP 200 (OK) response (SCC AS to UE A via Intermediate IM CN subsystem entities)

The SCC AS generates the SIP 200 (OK) response to the SIP INVITE request and forwards it to the SC UE A.

15-16: **SIP ACK request (SC UE A to SCC AS via Intermediate IM CN subsystem entities)**

The SC UE A generates the SIP ACK request to the SIP 200 (OK) response and forwards it to the SCC AS

17. Media paths between UE A and UE B

The media path of session X is using the new IP-CAN but the media path of the session Y is still using the CS bearer.

18-19. SIP BYE request (SCC AS to MSC Server via intermediate IM CN subsystem entities)

The SCC AS terminates the replaced call leg, which was using the CS bearer, by sending a SIP BYE request.

20-22. CC DISCONNECT message (interworking entities to SC UE A)

Upon receiving the CC DISCONNECT message, the SC UE A relinquishes all resources pertaining to the CS bearer.

NOTE: Steps 20-22 are performed only if signalling over CS domain is possible after the CS-PS access transfer is completed; otherwise, the SC UE A and the network release the source access leg of session X locally, without any signalling between the SC UE and the network.

23-24. SIP 200 (OK) response (MSC Server to SCC AS via intermediate IM CN subsystem entities)

Upon receiving the SIP BYE request over the old IP-CAN, the MSC Server sends a SIP 200 (OK) response over the old IP-CAN to the SCC AS.

25: **SIP REFER request (SCC AS to Intermediate IM CN subsystem entities) -see example in table A.15.2-25**

The SCC AS sends SIP REFER request towards UE A inside the dialog created by the message 13.

Table A.15.2-25: SIP REFER request (SCC AS to IM CN subsystem entities)

```
REFER sip:user1_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91e6bf6 SIP/2.0
Via: SIP/2.0/UDP sip:sccas1.home1.net;branch=z9hG4bk731b8a
Max-Forwards: 70
P-Charging-Vector: icid-value="AyretyU0dm+6O2IrT5tAFrbHLso=023551024"; orig-ioi=home1.net
From: <tel:+1-237-555-2222>; tag=aasdfgaag
To: <sip:user1_public1@home1.net>; tag=171828
Call-ID: cb03a0s09a2sdfglkj490237
Cseq: 55998 REFER
Content-Length: ...
Route: <sip:scscf1.home1.net;lr>, <sip:pcscf1.home1.net:7531;lr>
Contact: <sip:sccas1.home1.net;gr>
Refer-Sub: false
Supported: norefersub, gruu
Refer-To: <sip:additional.session.xfer@sccas.home1.net?Target-Dialog=a84b4c76e66710%3Bremote-
   tag=654364735%3Blocal-tag=1928301774&Require=tdialog&From=tel:+1-237-555-1111&To=tel:+1-
   987-654-3210&Content-Type=application%2Fsdp&body=v%3D0%0D%0Ao%3D-
   %202987933623%202987933623%20IN%20IP6%205555::ggg:fff:aaa:bbb%0D%0As%3D-
   %0D%0Ac%3DIN%20IP6%205555::ggg:fff:aaa:bbb%0D%0At%3D0%200%0D%0Am%3Dvideo%200%20RTP%2FAVP%20
   98%0D%0Am%3Daudio%203456%20RTP%2FAVP%2097%2096%0D%0Ab%3DAS:25.4%0D%0Aa%3Drtpmap:97%20AMR%0D
   %0Aa%3Dfmtp:97%20mode-set%3D0%2C2%2C5%2C7%3B%20mode-change-
   period%3D2%0D%0Aa%3Dmaxptime:20%0D%0A>
Content-Type: application/vnd.3gpp.mid-call+xml

<?xml version="1.0" encoding="UTF-8"?>
<mid-call/>
```

Refer-To: contains the additional transferred session SCC AS URI and the following URI header fields:

Target-Dialog: the dialog identifier of the source access leg.

Require: containing "tdialog" option tag

From: contains the public user identity of the UE A

To: contains the public user identity of the UE C

Content-Type: containing "application/sdp" MIME type of the "body" URI header field

body: SDP describing the media used in the session

26. SIP REFER request (intermediate IM CN subsystem entities to UE A)

The SIP REFER request is forwarded towards the UE A.

27-28. SIP 200 (OK) response (UE A to SCC AS via intermediate IM CN subsystem entities)

Upon receiving the SIP REFER request, the UE A sends a SIP 200 (OK) response.

29. SIP INVITE request transferring the held session Y (SC UE A to intermediate IM CN subsystem entities) - see example in table A.15.2-29

The SC UE A sends an initial SIP INVITE request to request the new call replacing the existing call Y.

Table A.15.2-29: SIP INVITE request (UE A to intermediate IM CN subsystem entities)

```
INVITE sip:additional.session.xfer@sccas.home1.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Max-Forwards: 70
Route: <sip:pcscf1.home1.net:7531;lr>, <sip:orig@scscf1.home1.net;lr>
P-Preferred-Identity: "John Doe" <sip:user1_public1@home1.net>
P-Access-Network-Info: IEEE-802.11b
Privacy: none
From: <tel:+1-237-555-1111>; tag=171828
To: <tel:+1-987-654-3210>
Call-ID: asdfqweasas
Cseq: 127 INVITE
Supported: 100rel, precondition, 199, gruu
Require: sec-agree
Proxy-Require: sec-agree
Security-Verify: ipsec-3gpp; q=0.1; alg=hmac-sha-1-96; spi=87654321; port1=7531
Contact: <sip:user1_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-
   00a0c91e6bf6> ;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel";+g.3gpp.mid-
   call
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE
Accept: application/sdp, application/3gpp-ims+xml
Target-Dialog: a84b4c76e66710;remote-tag=654364735;local-tag=1928301774
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=
c=IN IP6 5555::aaa:bbb:ccc:ddd
t=0 0
m=video 0 RTP/AVP 98
m=audio 3456 RTP/AVP 97 96
b=AS:25.4
a=curr:qos loca
a=tcap:1 RTP/AVPF
a=pcfg:1 t=1l sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:96 telephone-event
a=maxptime:20
a=sendonly
```

Request-URI: contains the additional transferred session SCC AS URI as received in the Refer-To URI in the SIP REFER request.

Target-Dialog: contains the dialog identifier as received in the Refer-To URI in the SIP REFER request.

Contact: contains the g.3gpp.mid-call media feature tag as defined in annex C indicating the support for the MSC server assisted mid-call feature.

SDP: All the media are offered with the sendonly directionality.

30. Evaluation of initial filter criteria

The S-CSCF evaluates initial filter criteria for the served SC user and as a result routes the SIP INVITE request towards the SCC AS.

31. SIP INVITE request (intermediate IM CN subsystem entities to SCC AS)

The SIP INVITE request is forwarded to the SCC AS as the result of the evaluation of iFC.

32. Remote Leg Update

The SCC AS performs the Remote Leg update by sending the SIP re-INVITE request towards the Remote Leg.

33. SIP re-INVITE request (SCC AS to intermediate IM CN subsystem entities)

The SCC AS modifies the message in accordance with routeing B2BUA functionality, e.g. mapping of From, To, Cseq and Call-ID header fields from one side of the B2BUA to the other. The SIP re-INVITE request contains the SDP offer that is identical to the SDP offer that the SCC AS received in the initial SIP INVITE request from the UE A.

34. SIP re-INVITE request (Intermediate IM CN subsystem entities to UE C)

The intermediate IM CN subsystem entities forward the SIP re-INVITE request to remote UE C.

35-36: SIP 200 (OK) response (UE C to SCC AS via Intermediate IM CN subsystem entities)

The UE C generates the SIP 200 (OK) response to the SIP re-INVITE request and forwards it to the SCC AS.

37-38: SIP ACK request (SCC AS to UE C via Intermediate IM CN subsystem entities)

The SCC AS generates the SIP ACK request to the SIP 200 (OK) response and forwards it to the remote UE C.

39: SIP 200 (OK) response (SCC AS to Intermediate IM CN subsystem entities)

The SCC AS generates the SIP 200 (OK) response to the SIP INVITE request and forwards it to the SC UE A.

40: SIP 200 (OK) response (Intermediate IM CN subsystem entities to UE A)

The SCC AS generates the SIP 200 (OK) response to the SIP INVITE request and forwards it to the SC UE A.

41-42: SIP ACK request (SC UE A to SCC AS via Intermediate IM CN subsystem entities)

The SC UE A generates the SIP ACK request to the SIP 200 (OK) response and forwards it to the SCC AS

43. Media paths between UE A and UE B

The media paths of session X and session Y are using the new IP-CAN but the the CS bearer is still not released.

44-45. SIP BYE request (SCC AS to MSC Server via intermediate IM CN subsystem entities)

The SCC AS terminates the replaced call leg, which was using the CS bearer, by sending a SIP BYE request.

46, 49-50. CC DISCONNECT message (interworking entities to SC UE A)

Upon receiving the CC DISCONNECT message, the SC UE A relinquishes all resources pertaining to the CS bearer.

NOTE: Steps 46-48 are performed only if signalling over CS domain is possible after the CS-PS access transfer is completed; otherwise, the SC UE and the network release the source access leg of session Y locally, without any signalling between the SC UE and the network.

47-48. SIP 200 (OK) response (MSC Server to SCC AS via intermediate IM CN subsystem entities)

51. Media paths between UE A and UE B

The media paths of session X and session Y are using the new IP-CAN.

A.15.3 PS to CS access transfer with MSC server assisted mid-call feature

In the example flow at the figure A.15.3-1, SC UE A has two ongoing sessions over PS bearer which are anchored at SCC AS. When both sessions were established the SC UE and the SCC AS included the g.3gpp.mid-call media feature tag as specified in annex C into the Contact header fields. The active session X is with UE B, the held session Y is with UE C. The session X and session Y are two party sessions. The session Y contains a rejected video stream and an accepted speech media component. When the SC UE attaches to the CS domain, it decides to transfer the sessions over the CS bearer without using the ICS capability.

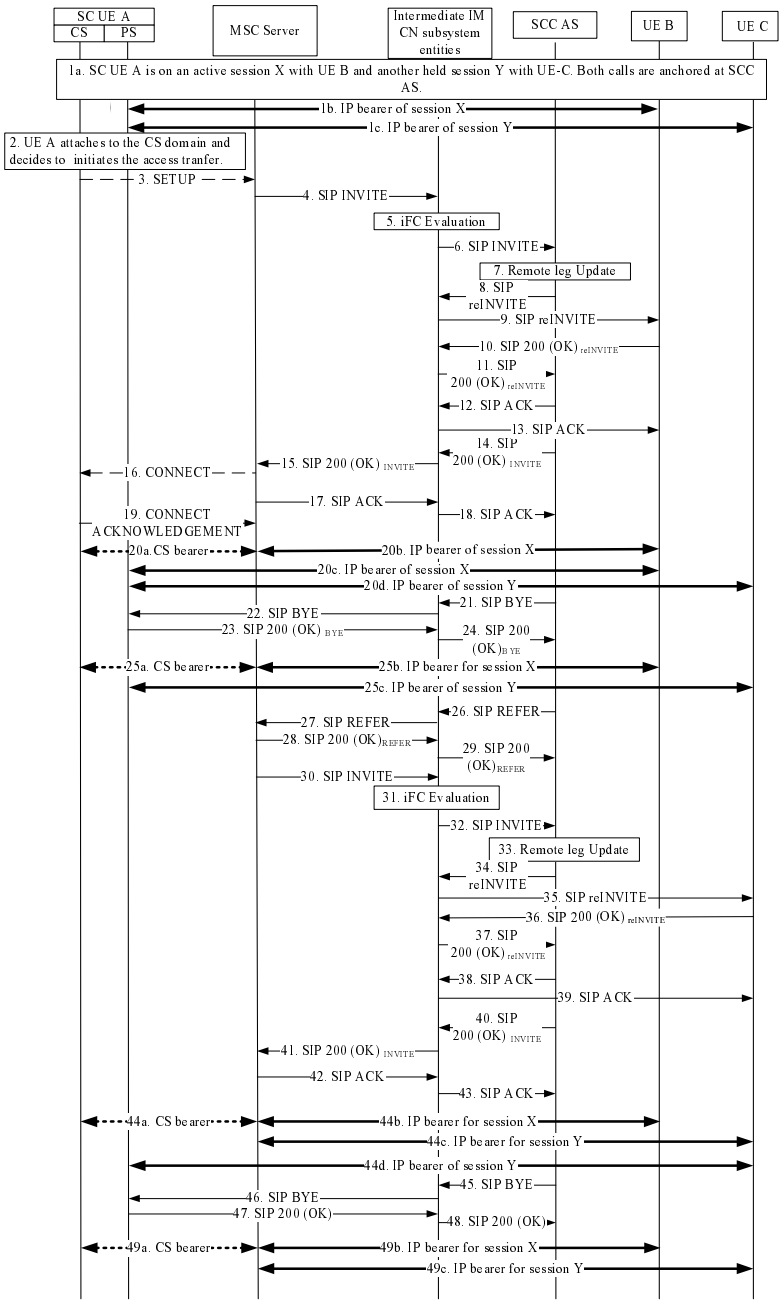


Figure A.15.3-1: Signalling flow for PS-CS access transfer: PS-CS

NOTE 1: For clarity, the SIP 100 (Trying) responses are not shown in the signalling flow.

1. SC UE A is on an active session X with UE B and a held session Y with UE C:

There is an ongoing IP bearer between the SC UE and the remote UE B and another IP bearer between the SC UE and the remote UE C. Both sessions are anchored at SCC AS.

2. SC UE A attaches to the CS domain

The SC UE attaches to the CS domain and decides to transfer the sessions over the CS bearer.

3. CC SETUP messages

Transaction Identifier: 3

4. **SIP INVITE request transferring the active session X (MSC Server to Intermediate IM CN subsystem entities) -see example in table A.15.3-4**

   Upon receiving the CC SETUP message the MSC Server sends a SIP INVITE request and associates the transaction identifier 3 with the SIP INVITE request.

Table A.15.3-4: SIP INVITE request (MSC Server to intermediate IM CN subsystem entities)

```
INVITE tel:+1-237-555-3333 SIP/2.0
Via: SIP/2.0/UDP msc1.home1.net;branch=z9hG4bk731b87
Max-Forwards: 70
P-Asserted-Identity: <tel:+1-237-555-1111>
P-Charging-Vector: icid-value="AyretyU0dm+6O2IrT5tAFrbHLso=023551024"; orig-ioi=home1.net
Privacy: none
From: <tel:+1-237-555-1111>;tag=171828
To: <tel:+1-237-555-3333>
Call-ID: cb03a0s09a2sdfglkj490333
Cseq: 127 INVITE
Supported: 100rel, precondition, gruu, 199, norefersub
Accept-Contact: *;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
P-Asserted-Service: urn:urn-7:3gpp-service.ims.icsi.mmtel
Contact: <sip:user1_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-
   00a0c91e6bf6> ;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-
   service.ims.icsi.mmtel" ;+g.3gpp.ics="server";+g.3gpp.mid-call
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE
Content-Type: application/sdp
Content-Length: (…)
Accept: application/sdp, application/3gpp-ims+xml, application/vnd.3gpp.mid-call+xml
Recv-Info: g.3gpp.mid-call

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:eee
s=
c=IN IP6 5555::aaa:bbb:ccc:eee
t=0 0
m=audio 3456 RTP/AVP 97 96
a=tcap:1 RTP/AVPF
a=pcfg:1 t=1
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:96 telephone-event
a=maxptime:20
```

Request-URI: contains the IMRN, as obtained from CS networks signalling.

SDP: The SDP contains preconfigured set of codecs supported by the MSC Server.

Contact: contains the g.3gpp.mid-call media feature tag as defined in annex C indicating the support for the MSC server assisted mid-call feature.

Accept: contains the MSC Server assisted mid-call feature MIME type.

5. **Evaluation of initial filter criteria**

   The S-CSCF evaluates initial filter criteria for the served SC user and as a result routes the SIP INVITE request towards the SCC AS.

6. **SIP INVITE request** (Intermediate IM CN subsystem entities to SCC AS)

7. **Remote Leg Update**

   The SCC AS performs the Remote Leg update by sending the SIP re-INVITE request towards the Remote Leg.

8. **SIP re-INVITE request (SCC AS to intermediate IM CN subsystem entities)**

The SCC AS acting as a routing B2BUA generates a SIP re-INVITE request based upon the received SIP INVITE request and the information previously stored against this session and routes it towards UE B via the intermediate IM CN subsystem entities.

9. SIP re-INVITE request (Intermediate IM CN subsystem entities to UE B)

Intermediate IM CN subsystem entities forward the SIP re-INVITE request to remote UE B.

10. SIP 200 (OK) response (UE B to intermediate IM CN subsystem entities)

Upon receiving the SIP re-INVITE request containing the SDP offer, since the UE B has all resources available, it sends immediately the SIP 200 (OK) response to the SIP re-INVITE request that contains the SDP answer. The SDP answer indicates that the resources are available.

11. SIP 200 (OK) response (intermediate IM CN subsystem entities to SCC AS)

The intermediate IM CN subsystem entities forward the SIP 200 (OK) response to the SIP re-INVITE request to the SCC AS in the originating network.

12-13. SIP ACK request (SCC AS to UE B via IM CN subsystem entities)

The SCC AS generates the SIP ACK request to the SIP 200 (OK) response, and forwards the SIP ACK request to the remote UE B.

14-15. SIP 200 (OK) response (SCC AS to MSC Server via IM CN subsystem entities)

The SCC AS generates the SIP 200 (OK) response to the SIP INVITE request, and forwards the SIP 200 (OK) response towards the MSC Server.

16. CC CONNECT message (MSC Server to SC UE A)

17-18. SIP ACK request (MSC Server to SCC AS via IM CN subsystem entities)

The MSC Server generates the SIP ACK request to the SIP 200 (OK) response, and forwards it to the SCC AS.

19. CC CONNECT ACKNOWLEDGEMENT message (SC UE A to MSC server)

20. Media paths between SC UE A and UE B:

The CS bearer is setup while the PS bearers are still existing.

21-22: SIP BYE request (SCC AS to SC UE A via intermediate IM CN subsystem entities)

The SCC AS terminates the replaced call leg of the session X, which was using the old IP-CAN, by sending a SIP BYE request to the UE A.

23-24. SIP 200 (OK) response (SC UE A to SCC AS via intermediate IM CN subsystem entities)

Upon receiving the SIP BYE request over the old IP-CAN, the SC UE A sends a SIP 200 (OK) response over the old IP-CAN to the SCC AS. Subsequently, the SC UE A relinquishes all resources pertaining to the old IP-CAN.

NOTE: Steps 22-23 are performed only if SC UE A is using Gm after the PS-CS access transfer is completed; otherwise, the SC UE A and the network release the source access leg of session X locally, without any signalling between the SC UE A and the network.

25. Media paths between SC UE A and UE B

The session X is transferred from PS bearer to CS bearer, but the session Y is still at the PS bearer.

26. SIP REFER request (SCC AS to IM CN subsystem entities) -see example in table A.15.3-26

The SCC AS sends SIP REFER request towards MSC Server inside the dialog created by the the message 14.

Table A.15.3-26: SIP REFER request (SCC AS to IM CN subsystem entities)

```
REFER sip:user1_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91e6bf6 SIP/2.0
Via: SIP/2.0/UDP sip:sccas1.home1.net;branch=z9hG4bk731b8a
Max-Forwards: 70
P-Charging-Vector: icid-value="AyretyU0dm+6O2IrT5tAFrbHLso=023551024"; orig-ioi=home1.net
To: <tel:+1-237-555-1111>;tag=171828
From: <tel:+1-237-555-3333>;tag=sdfsdf
Call-ID: cb03a0s09a2sdfglkj490333
Cseq: 55998 REFER
Content-Length: 125
Route: <sip:scscf1.home1.net;lr>
Refer-Sub: false
Supported: norefersub, gruu
Contact: sip:sccas1.home1.net
Refer-To: <additional.session.xfer@sccas.home1.net?Target-Dialog=ksdjfhwrklf%3Bremote-
   tag=676723565%3Blocal-tag=45418454&Require=tdialog&From=tel:+1-237-555-1111&To=tel:+1-987-
   654-3210&Content-Type=application%2Fsdp&body=v%3D0%0D%0Ao%3D-
   %202987933623%202987933623%20IN%20IP6%205555::ggg:fff:aaa:bbb%0D%0As%3D-
   %0D%0Ac%3DIN%20IP6%205555::ggg:fff:aaa:bbb%0D%0At%3D0%200%0D%0Am%3Dvideo%200%20RTP%2FAVP%20
   98%0D%0Am%3Daudio%203456%20RTP%2FAVP%2097%2096%0D%0Ab%3DAS:25.4%0D%0Aa%3Drtpmap:97%20AMR%0D
   %0Aa%3Dfmtp:97%20mode-set%3D0%2C2%2C5%2C7%3B%20mode-change-
   period%3D2%0D%0Aa%3Dmaxptime:20%0D%0A>
Content-Type: application/vnd.3gpp.mid-call+xml

<?xml version="1.0" encoding="UTF-8"?>
<mid-call/>
```

Refer-To: contains the additional transferred session SCC AS URI and the following URI header fields:

Target-Dialog: the dialog identifier of the source access leg.

Require: containing "tdialog" option tag

From: contains the public user identity of the UE A

To: contains the public user identity of the UE C

Content-Type: containing "application/sdp" MIME type of the "body" URI header field

body: SDP describing the media used in the session

27. SIP REFER request (intermediate IM CN subsystem entities to MSC Server)

The SIP REFER request is forwarded towards the MSC Server.

28-29. SIP 200 (OK) response (MSC Server to SCC AS via intermediate IM CN subsystem entities)

Upon receiving the SIP REFER request, the MSC Server sends a SIP 200 (OK) response.

30. SIP INVITE request for the held session Y (MSC Server to Intermediate IM CN subsystem entities) -see example in table A.15.3-30

Upon receiving the SIP REFER request the MSC Server sends a SIP INVITE request and associates the transaction identifier 4 with the SIP INVITE request.

Table A.15.3-30: SIP INVITE request (MSC Server to intermediate IM CN subsystem entities)

```
INVITE
sip:additional.session.xfer@sccas.home1.net SIP/2.0
Via: SIP/2.0/UDP msc1.home1.net;branch=z9hG4bk731b87
Max-Forwards: 70
P-Asserted-Identity: <tel:+1-237-555-1111>
P-Charging-Vector: icid-value="AyretyU0dm+6O2IrT5tAFrbHLso=023551024"; orig-ioi=home1.net
Privacy: none
From: <tel:+1-237-555-1111>;tag=171828
To: <tel:+1-987-654-3210>
Call-ID: asdfgqwerq
Cseq: 1275 INVITE
Supported: 100rel, precondition, 199, gruu
Accept-Contact: *;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
P-Asserted-Service: urn:urn-7:3gpp-service.ims.icsi.mmtel
Contact: <sip:user1_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-
   00a0c91e6bf6> ;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-
   service.ims.icsi.mmtel" ;+g.3gpp.ics="server";+g.3gpp.mid-call
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE
Content-Type: application/sdp
Target-Dialog: ksdjfhwrklf;remote-tag=676723565;local-tag=45418454
Require: tdialog
Content-Length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:eee
s=
c=IN IP6 5555::aaa:bbb:ccc:eee
t=0 0
m=video 0 RTP/AVP 98
m=audio 3456 RTP/AVP 97 96
a=tcap:1 RTP/AVPF
a=pcfg:1 t=1
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:96 telephone-event
a=maxptime:20
a=sendonly
```

Request-URI: contains the additional transferred session SCC AS URI as received in the Refer-To URI in the SIP REFER request.

Target-Dialog: contains the dialog identifier as received in the Refer-To URI in the SIP REFER request.

Contact: contains the g.3gpp.mid-call media feature tag as defined in annex C indicating the support for the MSC server assisted mid-call feature.

SDP: The SDP contains preconfigured set of codecs supported by the MSC Server. All the media are offered with the sendonly directionality.

31. Evaluation of initial filter criteria

The S-CSCF evaluates initial filter criteria for the served SC user and as a result routes the SIP INVITE request towards the SCC AS.

32. SIP INVITE request (Intermediate IM CN subsystem entities to SCC AS)

33. Remote Leg Update

The SCC AS performs the Remote Leg update by sending the SIP re-INVITE request towards the Remote Leg.

34. SIP re-INVITE request (SCC AS to intermediate IM CN subsystem entities)

The SCC AS acting as a routing B2BUA generates a SIP INVITE request based upon the received SIP INVITE request and the information previously stored against this session and routes it towards UE C via the

intermediate IM CN subsystem entities. The SIP re-INVITE request contains the SDP offer that is identical to the SDP offer that the SCC AS received in the initial SIP INVITE request from the UE A.

35. SIP re-INVITE request (Intermediate IM CN subsystem entities to UE C)

Intermediate IM CN subsystem entities forward the SIP re-INVITE request to remote UE C.

36. SIP 200 (OK) response (UE C to intermediate IM CN subsystem entities)

Upon receiving the SIP re-INVITE request containing the SDP offer, since the UE C has all resources available, it sends immediately the SIP 200 (OK) response to the SIP re-INVITE request that contains the SDP answer. The SDP answer indicates that the resources are available.

37. SIP 200 (OK) response (intermediate IM CN subsystem entities to SCC AS)

The intermediate IM CN subsystem entities forward the SIP 200 (OK) response to the SIP re-INVITE request to the SCC AS in the originating network.

38-39. SIP ACK request (SCC AS to UE C via IM CN subsystem entities)

The SCC AS generates the SIP ACK request to the SIP 200 (OK) response, and forwards the SIP ACK request to the remote UE C.

40. SIP 200 (OK) response (SCC AS to IM CN subsystem entities)

The SCC AS generates the SIP 200 (OK) response to the SIP INVITE request, and forwards the SIP 200 (OK) response towards the MSC Server.

41. SIP 200 (OK) response (Intermediate IM CN subsystem entities to MSC Server)

Intermediate IM CN subsystem entities forward the SIP 200 (OK) response to the SIP INVITE request to MSC Server.

42-43. SIP ACK request (MSC Server to SCC AS via IM CN subsystem entities)

The MSC Server generates the SIP ACK request to the SIP 200 (OK) response, and forwards it to the SCC AS.

44. Media paths between SC UE A and UE B:

The CS bearer and PS bearers for both the sessions are established but there is still the original IP bearer for the held session Y.

45-46: SIP BYE request (SCC AS to SC UE A via intermediate IM CN subsystem entities)

The SCC AS terminates the replaced call leg of the session Y, which was using the old IP-CAN, by sending a SIP BYE request to the UE A.

47-48. SIP 200 (OK) response (SC UE A to SCC AS via intermediate IM CN subsystem entities)

Upon receiving the SIP BYE request over the old IP-CAN, the SC UE A sends a SIP 200 (OK) response over the old IP-CAN to the SCC AS. Subsequently, the SC UE A relinquishes all resources pertaining to the old IP-CAN.

NOTE: Steps 46-47 are performed only if the SC UE A uses Gm after the PS-CS access transfer is completed; otherwise, the SC UE A and the network release the source access leg locally, without any signalling between the SC UE A and the network.

49. Media paths between SC UE A and UE B

Both sessions X and Y are transferred from PS bearer to CS bearer.

A.15.4 PS to CS access transfer with MSC server assisted mid-call feature with an incoming waiting call in alerting phase

In the example flow at the figure A.15.4-1, SC UE A has an ongoing sessions with speech media component and an incoming waiting session with speech media component which are anchored at SCC AS. The incoming waiting call is in alerting phase. The ongoing session X is with UE B, the incoming waiting session Y is with UE C. The session X and

session Y are two party sessions. Based upon measurement reports sent from the UE to E-UTRAN, the source E-UTRAN decides to trigger a PS to CS SRVCC procedure to CS access.

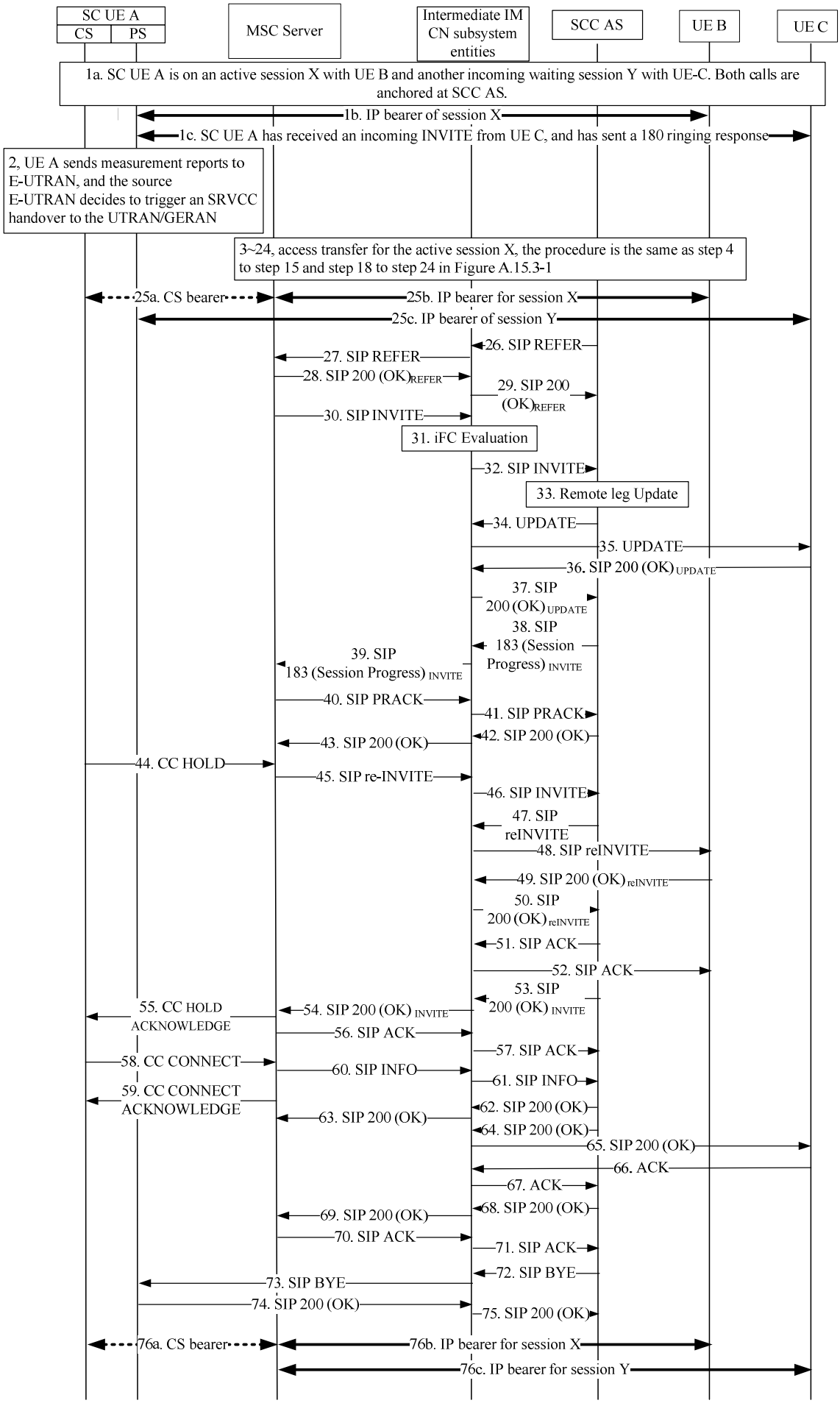

SC UE A
CS
PS
MSC Server
Intermediate IM CN subsystem entities
SCC AS
UE B
UE C
1a. SC UE A is on an active session X with UE B and another incoming waiting session Y with UE-C. Both calls are anchored at SCC AS.
1b. IP bearer of session X
1c. SC UE A has received an incoming INVITE from UE C, and has sent a 180 ringing response
2, UE A sends measurement reports to E-UTRAN, and the source E-UTRAN decides to trigger an SRVCC handover to the UTRAN/GERAN
3~24, access transfer for the active session X, the procedure is the same as step 4 to step 15 and step 18 to step 24 in Figure A.15.3-1
25a. CS bearer
25b. IP bearer for session X
25c. IP bearer of session Y
26. SIP REFER
27. SIP REFER
28. SIP 200 (OK)REFER
29. SIP 200 (OK)REFER
30. SIP INVITE
31. iFC Evaluation
32. SIP INVITE
33. Remote leg Update
34. UPDATE
35. UPDATE
36. SIP 200 (OK) UPDATE
37. SIP 200 (OK) UPDATE
38. SIP 183 (Session Progress) INVITE
39. SIP 183 (Session Progress) INVITE
40. SIP PRACK
41. SIP PRACK
42. SIP 200 (OK)
43. SIP 200 (OK)
44. CC HOLD
45. SIP re-INVITE
46. SIP INVITE
47. SIP reINVITE
48. SIP reINVITE
49. SIP 200 (OK) reINVITE
50. SIP 200 (OK) reINVITE
51. SIP ACK
52. SIP ACK
53. SIP 200 (OK) INVITE
54. SIP 200 (OK) INVITE
55. CC HOLD ACKNOWLEDGE
56. SIP ACK
57. SIP ACK
58. CC CONNECT
59. CC CONNECT ACKNOWLEDGE
60. SIP INFO
61. SIP INFO
62. SIP 200 (OK)
63. SIP 200 (OK)
64. SIP 200 (OK)
65. SIP 200 (OK)
66. ACK
67. ACK
68. SIP 200 (OK)
69. SIP 200 (OK)
70. SIP ACK
71. SIP ACK
72. SIP BYE
73. SIP BYE
74. SIP 200 (OK)
75. SIP 200 (OK)
76a. CS bearer
76b. IP bearer for session X
76c. IP bearer for session Y

Figure A.15.4-1: Signalling flow for PS to CS access transfer with MSC server assisted mid-call feature with an incoming waiting call in alerting phase

NOTE 1: For clarity, the SIP 100 (Trying) responses are not shown in the signalling flow.

NOTE 2: For transferring the incoming waiting call in alerting phase, the feature of PS to CS SRVCC in alerting phase needs to be supported, but the MSC server assisted mid-call feature is not necessary.

1. SC UE A is on an active session X with UE B and an incoming waiting session Y with UE C:

There is an ongoing PS bearer between the SC UE and the remote UE B and another PS bearer between the SC UE and the remote UE C. Both sessions are anchored at SCC AS.

2. SC UE A sends the measurement reports to E-UTRAN

UE A sends the measurement reports to E-UTRAN, and the source E-UTRAN decides to trigger an PS to CS SRVCC handover to CS access. The MSC server initiates the session transfer with the STN-SR, refer to 3GPP TS 23.237 [9].

3-24. Access transfer for the active session X

The procedure for transfering the active session X is the same as step 4 to step 15 and step 18 to step 24 described in subclause A.15.3.

25. Media paths between SC UE A and UE B

The session X is transferred from PS bearer to CS bearer, but the session Y is still at the PS bearer.

26. SIP REFER request (SCC AS to IM CN subsystem entities) -see example in table A.15.4-26

The SCC AS sends SIP REFER request towards MSC server inside the dialog created by the the message 14, and it also contain the state-and-event-info XML body to indicate that the additional session is an incoming session in alerting phase.

Table A.15.4-26: SIP REFER request (SCC AS to IM CN subsystem entities)

```
REFER sip:user1_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91e6bf6 SIP/2.0
Via: SIP/2.0/UDP sip:sccas1.home1.net;branch=z9hG4bk731b8a
Max-Forwards: 70
P-Charging-Vector: icid-value="AyretyU0dm+6O2IrT5tAFrbHLso=023551024"; orig-ioi=home1.net
To: <tel:+1-237-555-1111>;tag=171828
From: <tel:+1-237-555-3333>;tag=sdfsdf
Call-ID: cb03a0s09a2sdfglkj490333
Cseq: 55998 REFER
Content-Length: 125
Route: <sip:scscf1.home1.net;lr>
Refer-Sub: false
Supported: norefersub, gruu
Contact: sip:sccas1.home1.net
Refer-To: <additional.session.xfer@sccas.home1.net?Target-Dialog=ksdjfhwrklf%3Bremote-
   tag=676723565%3Blocal-tag=45418454&Require=tdialog&From=tel:+1-237-555-1111&To=tel:+1-987-
   654-3210&Content-Type=application%2Fsdp&body=v%3D0%0D%0Ao%3D-
   %202987933623%202987933623%20IN%20IP6%205555::ggg:fff:aaa:bbb%0D%0As%3D-
   %0D%0Ac%3DIN%20IP6%205555::ggg:fff:aaa:bbb%0D%0At%3D0%200%0D%0Aaudio%203456%20RTP%2FAVP%209
   7%2096%0D%0Ab%3DAS:25.4%0D%0Aa%3Drtpmap:97%20AMR%0D%0Aa%3Dfmtp:97%20mode-
   set%3D0%2C2%2C5%2C7%3B%20mode-change-period%3D2%0D%0Aa%3Dmaxptime:20%0D%0A>
Content-Type: application/vnd.3gpp.state-and-event-info+xml

<?xml version="1.0" encoding="UTF-8"?>
<state-and-event-info>
    <state-info>early</state-info>
    <direction>receiver</direction>
</state-and-event-info>
```

Refer-To: contains the additional transferred session SCC AS URI and the following URI header fields:

Target-Dialog: the dialog identifier of the source access leg.

Require: containing "tdialog" option tag

From: contains the public user identity of the UE A

To: contains the public user identity of the UE C

Content-Type: containing "application/sdp" MIME type of the "body" URI header field

body: SDP describing the media used in the session.

XML Schema: contain the session state information that the additional session is an incoming session in alerting phase.

27. SIP REFER request (intermediate IM CN subsystem entities to MSC server)

The SIP REFER request is forwarded towards the MSC server.

28-29. SIP 200 (OK) response (MSC server to SCC AS via intermediate IM CN subsystem entities)

Upon receiving the SIP REFER request, the MSC server sends a SIP 200 (OK) response.

30. SIP INVITE request for the held session Y (MSC server to Intermediate IM CN subsystem entities) -see example in table A.15.4-30

Upon receiving the SIP REFER request which contain the session state information to indicate that the additional session in an incoming session in alerting phase, the MSC server moves to Call Received state as described in the SIP REFER request but does not generate an in-band ring tone to the calling party, and sends a SIP INVITE request and associates the transaction identifier with the SIP INVITE request.

Table A.15.4-30: SIP INVITE request (MSC server to intermediate IM CN subsystem entities)

```
INVITE
sip:additional.session.xfer@sccas.home1.net SIP/2.0
Via: SIP/2.0/UDP msc1.home1.net;branch=z9hG4bk731b87
Max-Forwards: 70
P-Asserted-Identity: <tel:+1-237-555-1111>
P-Charging-Vector: icid-value="AyretyU0dm+6O2IrT5tAFrbHLso=023551024"; orig-ioi=home1.net
Privacy: none
From: <tel:+1-237-555-1111>;tag=171828
To: <tel:+1-987-654-3210>
Call-ID: asdfgqwerq
Cseq: 1275 INVITE
Supported: 100rel, precondition, 199, gruu
Accept-Contact: *;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
P-Asserted-Service: urn:urn-7:3gpp-service.ims.icsi.mmtel
Contact: <sip:msc1@home1.net> ;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-
   service.ims.icsi.mmtel" ;+g.3gpp.ics="server";+g.3gpp.mid-call
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE
Content-Type: application/sdp
Target-Dialog: ksdjfhwrklf;remote-tag=676723565;local-tag=45418454
Require: tdialog
Content-Length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:eee
s=
c=IN IP6 5555::aaa:bbb:ccc:eee
t=0 0
m=audio 3456 RTP/AVP 97 96
a=tcap:1 RTP/AVPF
a=pcfg:1 t=1
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:96 telephone-event
a=maxptime:20
```

Request-URI: contains the additional transferred session SCC AS URI as received in the Refer-To URI in the SIP REFER request.

Target-Dialog: contains the dialog identifier as received in the Refer-To URI in the SIP REFER request.

Contact: contains the g.3gpp.mid-call media feature tag as defined in annex C indicating the support for the MSC server assisted mid-call feature.

SDP: The SDP contains preconfigured set of codecs supported by the MSC server.

31. Evaluation of initial filter criteria

The S-CSCF evaluates initial filter criteria for the served SC user and as a result routes the SIP INVITE request towards the SCC AS.

32. SIP INVITE request (Intermediate IM CN subsystem entities to SCC AS)

33. Remote Leg Update

The SCC AS performs the Remote Leg update by sending the SIP re-INVITE request towards the Remote Leg.

34. SIP UPDATE request (SCC AS to intermediate IM CN subsystem entities)

The SCC AS acting as a routing B2BUA generates a SIP UPDATE request based upon the received SIP INVITE request and the information previously stored against this session and routes it towards UE C via the intermediate IM CN subsystem entities. The SIP UPDATE request contains the SDP offer that is identical to the SDP offer that the SCC AS received in the initial SIP INVITE request from the UE A.

35. SIP UPDATE request (Intermediate IM CN subsystem entities to UE C)

Intermediate IM CN subsystem entities forward the SIP UPDATE request to remote UE C.

36. SIP 200 (OK) response (UE C to intermediate IM CN subsystem entities)

Upon receiving the SIP UPDATE request containing the SDP offer, since the UE C has all resources available, it sends immediately the SIP 200 (OK) response to the SIP UPDATE request that contains the SDP answer. The SDP answer indicates that the resources are available.

37. SIP 200 (OK) response (intermediate IM CN subsystem entities to SCC AS)

The intermediate IM CN subsystem entities forward the SIP 200 (OK) response to the SIP UPDATE request to the SCC AS in the originating network.

38-39. SIP 183 (Session Progress) response (SCC AS to MSC server via IM CN subsystem entities)

The SCC AS sends a 183 (Session Progress) containing the SDP answer as received from the UE C. The SDP answer indicates that resources are available

40. SIP PRACK request (MSC Server to Intermediate IM CN subsystem entities)

The MSC server acknowledges the receipt of the 183 Session Progress.

41. SIP PRACK request (Intermediate IM CN subsystem entities to SCC AS)

The intermediate IM CN subsystem forward the SIP PRACK request to the SCC AS

42-43. SIP 200 (OK) response (SCC AS to MSC server via IM CN subsystem entities)

The SCC AS acknowledges the SIP PRACK request with a SIP 200 (OK) response to the MSC server.

44. CC HOLD Message (SC UE to MSC server)

The SC UE A put the active session on hold.

45. SIP re-INVITE request (MSC server to intermediate IM CN subsystem entities)

Upon receiving the CS HOLD Message from the UE, MSC server sends a SIP re-INVITE request towords session X, which put session X on hold.The SDP in this SIP re-INVITE request is based on the last SDP

offer/answer negotiation for the active session transfer form step 3 to 24, but for each media streams set the SDP attribute to "sendonly".

46. SIP re-INVITE request (Intermediate IM CN subsystem entities to SCC AS)

The SIP re-INVITE request is forwarded to the SCC AS.

47-48. SIP re-INVITE request (SCC AS to UE B)

SCC AS sends SIP re-INVITE request to UE B, The SIP re-INVITE request contains the SDP offer that is identical to the SDP offer that the SCC AS received in the SIP re-INVITE request from the MSC server.

49-50. SIP 200 (OK) response (UE B to SCC AS)

Upon receiving the SIP re-INVITE request containing the SDP offer which contain the SDP attribute for each media streams to "sendonly", UE B response the SIP re-INVITE request with a SIP 200 (OK), which set the SDP attribute for each media streams to "receonly".

51-52. SIP ACK request (SCC AS to UE B)

53-54. SIP 200 (OK) response (SCC AS to MSC server via intermediate IM CN subsystem entities)

The SCC AS sends SIP 200 (OK) to indicate the succesful activity to the MSC server that put session X on hold.

55. CC HOLD ACKNOWLEDGE Message (MSC server to SC UE A)

56-57. SIP ACK request (MSC server to SCC AS via intermediate IM CN subsystem entities)

MSC server acknowledges the SIP 200 (OK) received from SCC AS.

58. CC CONNECT message from SC UE A to MSC server

The SC UE A accepts the call and sends CC CONNECT message.

59. CC CONNECT ACKNOWLEDGE (MSC server to SC UE A)

60. SIP INFO request (MSC server to intermediate IM CN subsystem entities) - see example in table A.15.4-60

A.15.4-60: INFO (SCC AS to intermediate IM CN subsystem entities)

```
INFO sip:sccas1.home1.net;gr SIP/2.0
Via: SIP/2.0/UDP msc1.visit1.net;branch=z9hG4bk731b87
Max-Forwards: 68
Route: <sip:scscf1.home1.net;lr>
From: <tel:+1-237-555-1111>;tag=171828
To: <tel: +1-237-555-3333>;tag=171828
Call-ID: cb03a0s09a2sdfglkj490334
Cseq: 130 INFO
Info-Package: g.3gpp.state-and-event
Content-Disposition: Info-Package
Content-Type: application/vnd.3gpp.state-and-event-info+xml
Content-Length:

<?xml version="1.0" encoding="UTF-8"?>
<state-and-event-info>
   <event>call-accepted</event>
</state-and-event-info>
```

XML Schema: contain the session state information indicating that the remote party has answered the call.

61. SIP INFO request (Intermediate IM CN subsystem entities to SCC AS)

The intermediate IM CN subsystem entities forward the SIP INFO request to the SCC AS. The SCC AS gets informed that the SC UE A has accepted the call.

62. SIP 200 (OK) response (SCC AS to intermediate IM CN subsystem entities)

The SCC AS acknowledges the receipt of the SIP INFO request indicating that the SC UE A has accepted the call

63. SIP 200 (OK) response (Intermediate IM CN subsystem entities to MSC server)

The SIP 200 (OK)response is forwarded to the MSC server.

64. SIP 200 (OK) response (SCC AS to intermediate IM CN subsystem entities)

The SCC AS sends SIP 200 (OK) response to indicate to the far end that the SC UE A has accepted the call.

65. SIP 200 (OK) response (Intermediate IM CN subsystem entities to far end)

The SIP 200 (OK) response is forwarded to the far end)

66. SIP ACK request (far end to intermediate IM CN subsystem entities)

The far end UE acknowledges the SIP 200 (OK) response received from the SCC AS

67. SIP ACK request (Intermediate IM CN subsystem entities to SCC AS)

The SIP ACK request is forwarded to the SCC AS.

68. SIP 200 (OK) response (SCC AS to intermediate IM CN subsystem entities)

The SCC AS sends SIP 200 (OK) response to indicate the succesfull access transfer to the MSC server.

69. SIP 200 (OK) response (Intermdiate IM CN subsystem entities to far end)

The SIP 200 (OK) response is forwarded to the MSC server.

70. SIP ACK request (MSC server to intermediate IM CN subsystem entities)

MSC server acknowledges the SIP 200 (OK) response received from SCC AS.

71. SIP ACK request (Intermediate IM CN subsystem entities to SCC AS)

The SIP ACK request is forwarded to the SCC AS.

72-73: SIP BYE request (SCC AS to SC UE A via intermediate IM CN subsystem entities)

The SCC AS terminates the replaced call leg of the session Y, which was using the old IP-CAN, by sending a SIP BYE request to the UE A.

74-75. SIP 200 (OK) response (SC UE A to SCC AS via intermediate IM CN subsystem entities)

Upon receiving the SIP BYE request over the old IP-CAN, the SC UE A sends a SIP 200 (OK) response over the old IP-CAN to the SCC AS. Subsequently, the SC UE A relinquishes all resources pertaining to the old IP-CAN.

NOTE 3: Steps 73-74 are performed only if the SC UE A uses Gm after the PS-CS access transfer is completed; otherwise, the SC UE A and the network release the source access leg of session Y locally, without any signalling between the SC UE A and the network.

76. Media paths between SC UE A and UE B

Both sessions X and Y are transferred from PS bearer to CS bearer.

A.16 Signalling flows for PS to CS SRVCC session transfer for IMS emergency session

A.16.1 Introduction

The signalling flows for PS to CS SRVCC session transfer for IMS emergency session demonstrate how an IMS emergency session is transferred from PS network to CS network using PS to CS SRVCC procedure. The following signalling flow is included:

- subclause A.16.2 shows an example when a UE initiating an emergency session in IMS for the case that the UE is not in limited service mode ;and
- subclause A.16.3 shows an example when the emergency session need to transfer from PS to CS using PS to CS SRVCC procedure for the case that the UE is not in limited service mode.

A.16.2 UE initiating an emergency session in IMS

The signalling flows shown in figure A.16.2-1 describes the UE initiating an IMS emergency session procedure for the case that the UE is not in limited service mode. The flow illustrates the anchoring of the session at the EATF.

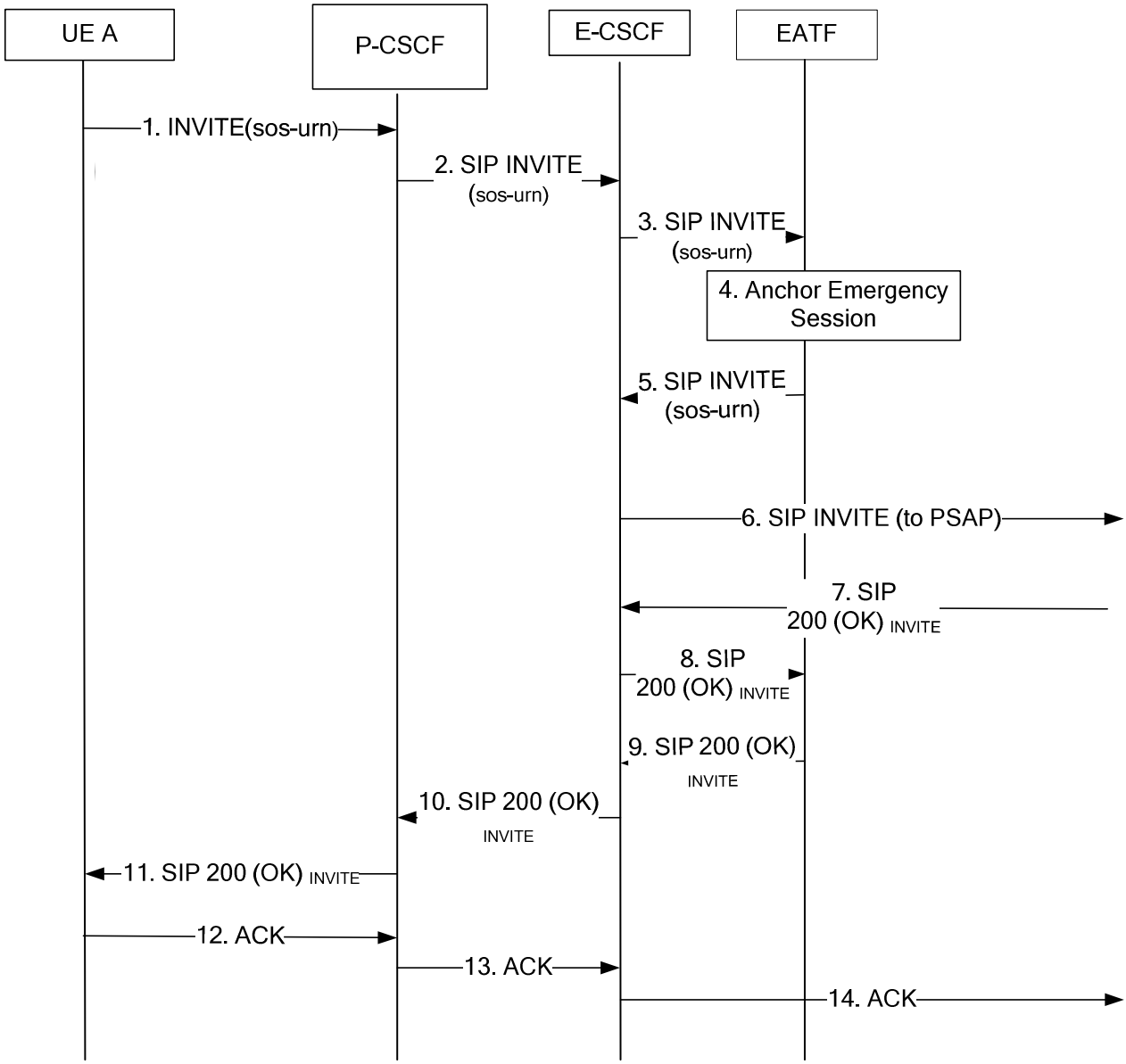


Figure A.16.2-1: Signalling flow for UE initiating an emergency session in IMS

NOTE 1: For clarity, the SIP 100 (Trying) responses are not shown in the signalling flow.

NOTE 2: For clarity, the SIP 180 (Ringing) response is not shown in the signalling flow.

NOTE 3: For clarity, the precondition mechanism is not shown in the signalling flow.

1. SIP INVITE request (UE A to P-CSCF) see example in table A.16.2-2

Table A.16.2-2: SIP INVITE request

```
INVITE urn:service:sos.fire SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Max-Forwards: 70
Route: <sip:pcscf.visit1.net:7531;lr;comp=sigcomp>
P-Preferred-Identity: <sip:user1_public1@home1.net>
P-Access-Network-Info: 3GPP-UTRAN-FDD; utran-cell-id-3gpp=234151D0FCE11
Privacy: none
From: <sip:user1_public1@home1.net>;tag=171828
To: <urn:service:sos.fire>
Call-ID: cb03a0s09a2sdfglkj490333
Cseq: 127 INVITE
Supported: 100rel, precondition, 199, gruu
Accept: application/sdp,application/3gpp-ims+xml
Require: sec-agree
Proxy-Require: sec-agree
Security-Verify: ipsec-3gpp; q=0.1; alg=hmac-sha-1-96; spi=87654321; port=7531
Contact: <sip:user1_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-
   00a0c91e6bf6>;+sip.instance="<urn:gsma:imei:90420156-025763-0>"
Geolocation: <sips:3sdefrhy2jj7@lis.atlanta.example.com>;inserted-
   by="sip:user1_public1@home1.net";routing-allowed="yes"
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=
c= IN IP6 5555::aaa:bbb:ccc:ddd
t=0 0
m=audio 3400 RTP/AVP 98
a=curr: qos local none
a=curr: qos remote none
a=des: qos mandatory local sendrcv
a=des: qos mandatory remote sendrcv
a=inactive
```

Contact: contains the "sip.instance" media feature tag as specified in IETF RFC 5626 [22] with a value formed from an IMEI as defined in 3GPP TS 23.003 [12].

2. SIP INVITE request (P-CSCF to E-CSCF) see example in table A.16.2-3

Table A.16.2-3: SIP INVITE request

```
INVITE urn:service:sos.fire SIP/2.0
Via: SIP/2.0/UDP pcscf.visit1.net;branch=z9hG4bK240f34.1, SIP/2.0/UDP
   [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Max-Forwards: 69
Route: <sip:ecscf.visit1.net;lr;>
Record-Route: <sip:pcscf.visit1.net;lr>
P-Preferred-Identity:
P-Access-Network-Info:
Privacy:
From:
To:
Call-ID:
Cseq:
Supported:
Accept:
Require:
Proxy-Require:
Accept-Contact:
P-Preferred-Service:
Security-Verify:
Contact:
Geolocation:
Allow:
Content-Type:
Content-Length: (…)

v=
o=
s=
c=
t=
m=
a=curr:
a=curr:
a=des:
a=des:
a=
```

3. SIP INVITE request (E-CSCF to EATF) see example in table A.16.2-4

Table A.16.2-4: SIP INVITE request

```
INVITE urn:service:sos.fire SIP/2.0
Via: SIP/2.0/UDP ecscf.visit1.net;branch=z9hG4bK87ly12.1, SIP/2.0/UDP
   pcscf.visit1.net;branch=z9hG4bK240f34.1, SIP/2.0/UDP
   [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Max-Forwards: 68
Route: <sip:eatf1.visit1.net;lr>
Record-Route: <sip:ecscf.visit1.net;lr>,<sip:pcscf.visit1.net;lr>
P-Preferred-Identity:
P-Access-Network-Info:
Privacy:
From:
To:
Call-ID:
Cseq:
Supported:
Accept:
Require:
Proxy-Require:
Accept-Contact:
P-Preferred-Service:
Security-Verify:
Contact:
Geolocation: <sips:3sdefrhy2jj7@lis.atlanta.example.com>;inserted-
   by="sip:user1_public1@home1.net";routing-allowed="yes";used-for-routing
Allow:
Content-Type:
Content-Length: (…)

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

4. **EATF anchors the emergency session**

   The EATF (acting as a routing B2BUA) anchors the emergency session, i.e. the EATF is inserted in the signalling path which invokes a 3pcc for enablement of Access Transfers

5. **SIP INVITE request (EATF to E-CSCF) see example in table A.16.2-5**

   The EATF acting as a routing B2BUA, generates a SIP INVITE request based upon the received SIP INVITE request and the information previously stored against this session and routes it towards PSAP via the intermediate IM CN subsystem entities.

Table A.16.2-5: SIP INVITE request

```
INVITE urn:service:sos.fire SIP/2.0
Via: SIP/2.0/UDP eatf1.visit1.net;branch=z9hG4bKnas34r5
Max-Forwards: 67
Route: <sip:ecscf.visit1.net:7531;lr;comp=sigcomp>
Record-Route: <sip:eatf1.visit1.net;lr>,<sip:ecscf.visit1.net;lr>,<sip:pcscf.visit1.net;lr>
P-Preferred-Identity: <sip:user1_public1@home1.net>
P-Access-Network-Info: 3GPP-UTRAN-FDD; utran-cell-id-3gpp=234151D0FCE11
Privacy: none
From: <sip:user1_public1@home1.net>;tag=171828
To: <urn:service:sos.fire >
Call-ID: cb03a0s09a2sdfglkj490333
Cseq: 127 INVITE
Supported: 100rel, precondition, 199, gruu
Accept: application/sdp,application/3gpp-ims+xml
Require: sec-agree
Proxy-Require: sec-agree
Security-Verify: ipsec-3gpp; q=0.1; alg=hmac-sha-1-96; spi=87654321; port=7531
Contact: <sip:user1_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-
   00a0c91e6bf6>;+sip.instance="<urn:gsma:imei:90420156-025763-0>"
Geolocation: <sips:3sdefrhy2jj7@lis.atlanta.example.com>; inserted-by="
   sip:user1_public1@home1.net"; routing-allowed="yes";used-for-routing
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=
c= IN IP6 5555::aaa:bbb:ccc:ddd
t=0 0
m=audio 3400 RTP/AVP 98
a=curr: qos local none
a=curr: qos remote none
a=des: qos mandatory local sendrcv
a=des: qos mandatory remote sendrcv
a=inactive
```

6. **SIP INVITE request (E-CSCF to PSAP)**

   E-CSCF routes the SIP INVITE request to the PSAP.

7. **SIP 200 (OK) response (PSAP to E-CSCF) see example in table A.16.2-6**

Table A.16.2-6: SIP 200 (OK) response

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP ecscf.visit1.net;branch=z9hG4bKnas34r5
Max-Forwards: 67
Record-Route: <sip:eatf1.visit1.net;lr>;<sip:ecscf.visit1.net;lr>,<sip:pcscf.visit1.net;lr>
Privacy: none
From: <sip:user1_public1@home1.net>;tag=171828
To: < urn:service:sos.fire >;tag=232456
Call-ID:
Cseq:
Require: 100rel, precondition, 199, gruu
Contact: <sip:mgcf.visit1.net>.
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=
c= IN IP6 5555::fff:eee:ccc:ddd
t=0 0
m=audio 3400 RTP/AVP 98
a=curr: qos local none
a=curr: qos remote none
a=des: qos mandatory local sendrcv
a=des: qos mandatory remote sendrcv
a=inactive
```

8-9. SIP 200 (OK) response (E-CSCF to EATF and to E-CSCF)

E-CSCF forwards the SIP 200 (OK) response.

10-11. SIP 200 (OK) response (E-CSCF to UE A) see example in table A.16.2-7

Table A.16.2-7: SIP 200 (OK) response

```
SIP/2.0 200 OK
Via:
Max-Forwards: 65
Record-Route:
Privacy:
From:
To:
P-Asserted-Identity: tel:911;context="+1"
Call-ID:
Cseq:
Require:
Contact:
Allow:
Content-Type:
Content-Length:

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

12. SIP ACK request

UE A responds to the SIP 200 (OK) response with a SIP ACK request.

A.16.3 Session transfer for emergency session using PS to CS SRVCC procedure: PS-CS

In the example in figure A.16.3-1, UE A (which has a valid subscription, is authenticated and authorized for PS service and is normal attached to the network) has an ongoing emergency session with a PSAP using a PS bearer which is anchored at EATF. Based upon measurement reports sent from the UE to E-UTRAN, the source E-UTRAN decides to trigger a PS to CS SRVCC handover to CS access.

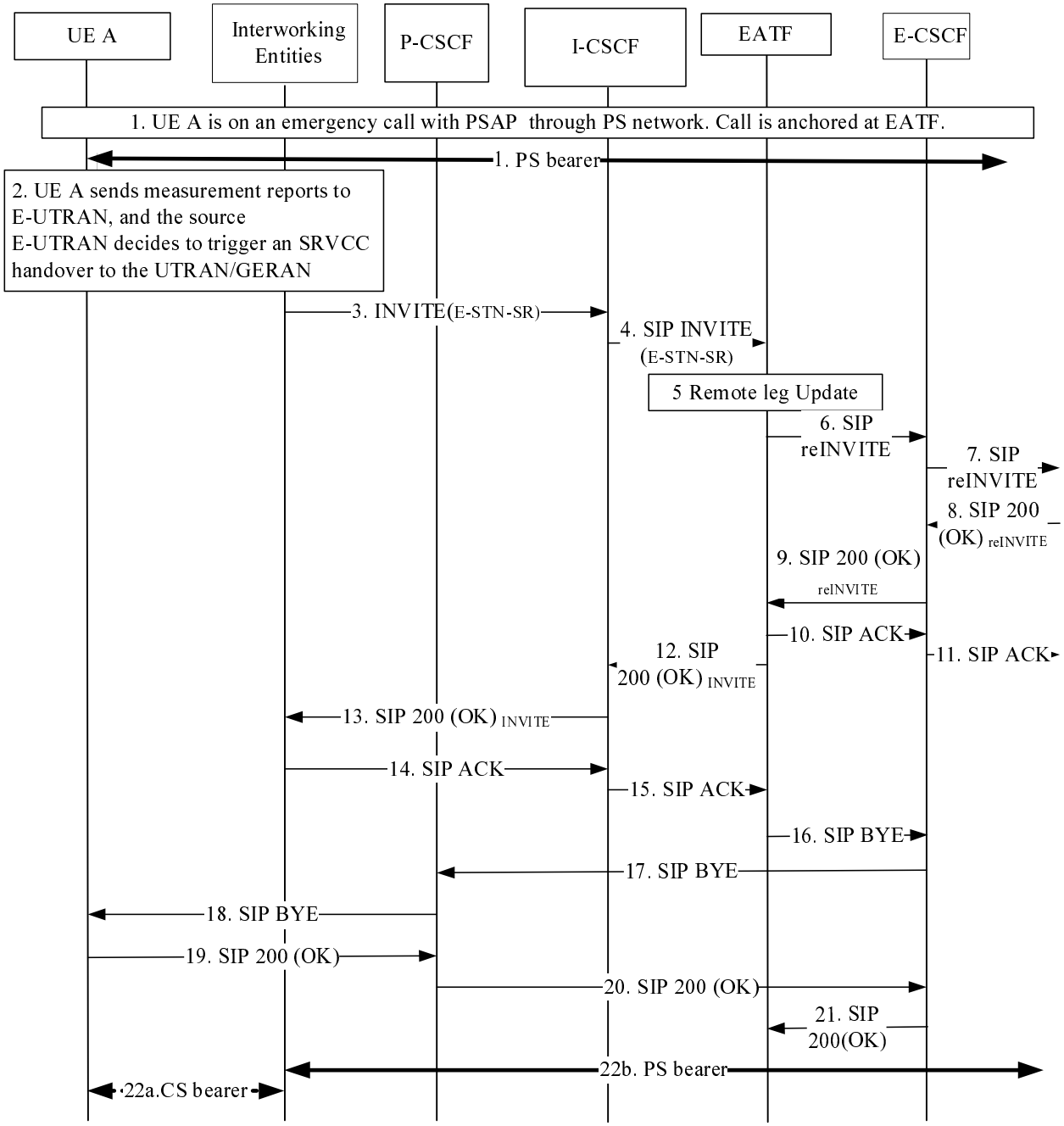


Figure A.16.3-1 Signalling flow for emergency session transfer using PS to CS SRVCC procedure

NOTE 1: For clarity, the SIP 100 (Trying) responses are not shown in the signalling flow.

1. **UE A is on an active emergency session with a PSAP**

   There is an ongoing IP bearer between the UE A and the remote end PSAP. The call is achored at EATF.

2. **SC UE A attaches to the CS domain**

   UE A sends the measurement reports to E-UTRAN, and the source E-UTRAN decides to trigger an PS to CS SRVCC handover to CS access. The MSC Server initiates the session transfer with the E-STN-SR, refer to 3GPP TS 23.237 [9].

3. **SIP INVITE request (Interworking entities to Intermediate IM CN subsystem entities) -see example in table A.16.3-2**

Table A.16.3-2: SIP INVITE request (interworking entities to intermediate IM CN subsystem entities)

```
INVITE tel: +1-237-555-3333 SIP/2.0
Via: SIP/2.0/UDP msc1.visit1.net;branch=z9hG4bk731b87
Max-Forwards: 70
Route: <sip:icscf1.visit1.net;lr>
P-Asserted-Identity: <tel:+1-237-555-1111>
P-Charging-Vector: ####
Privacy: none
From: <tel:+1-237-555-1111>;tag=171828
To: <tel: +1-237-555-3333>
Call-ID: cb03a0s09a2sdfglkj490334
Cseq: 127 INVITE
Supported: 100rel, precondition, gruu
Accept-Contact: *;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
P-Asserted-Service: urn:urn-7:3gpp-service.ims.icsi.mmtel
Contact: <sip:msc1.home1.net>;+sip.instance="<urn:gsma:imei:90420156-025763-0>";+g.3gpp.icsi-
 ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:eee
s=
c=IN IP6 5555::aaa:bbb:ccc:eee
t=0 0
m=audio 3456 RTP/AVP 97 96
a=tcap:1 RTP/AVPF
a=pcfg:1 t=1
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:96 telephone-event
a=maxptime:20
```

Request-URI: contains the E-STN-SR, as routed to the EATF

SDP: The SDP contains preconfigured set of codecs supported by the MGW.

Contact: contains the "sip.instance" media feature tag as specified in IETF RFC 5626 [22] with a value formed from an IMEI as defined in 3GPP TS 23.003 [12].

4. **SIP INVITE request**

The I-CSCF routes the SIP INVITE request directly to the EATF by using the procedure defined in 3GPP TS 23.228 [15] for PSI based application Server termination.

NOTE 2: The use of indirect routing for PSI based Application Server Termination as described in 3GPP TS 23.228 [15] in subclause 5.7.6 cannot be used for routing the SIP INVITE request to the EATF.

5. **Remote Leg Update**

The EATF based on the content of the "gr" parameter in the Contact header field correlates the SIP INVITE request to the local and remote call legs of the existing session between the UE A and the remote end. The EATF performs the Remote Leg update by sending the SIP re-INVITE request towards the Remote Leg.

6. **SIP re-INVITE request (EATF to intermediate IM CN subsystem entities) –see example in table A.16.3-3**

TheEATF acting as a routing B2BUA generates a SIP INVITE request based upon the received SIP INVITE request and the information previously stored against this session and routes it towards PSAP via the intermediate IM CN subsystem entities.

Table A.16.3-3: SIP re-INVITE request (EATF to intermediate IM CN subsystem entities)

```
INVITE #### SIP/2.0
Via: SIP/2.0/UDP eatf1.visit1.net;branch=z9hG4bKnas34r5
Max-Forwards: 68
```

```
Route: <sip:ecscf.visit1.net:lr>
P-Asserted-Identity: <tel: +1-237-555-1111>
P-Charging-Function-Addresses: ####
P-Charging-Vector:####
Privacy: none
From: <sip:user1_public1@home1.net>;tag=171828
To: <urn:service:sos.fire>;tag=232456
Call-ID: cb03a0s09a2sdfglkj490333
Cseq:
Contact: <sip:user1_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-
 00a0c91e6bf6>;+sip.instance="<urn:gsma:imei:90420156-025763-0>"
Allow:
Content-Type: Content-Length:

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:eee
s=
c=IN IP6 5555::aaa:bbb:ccc:eee
t=0 0
m=audio 3456 RTP/AVPF 97 96
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:96 telephone-event
a=maxptime:20
m=message 0 TCP/MSRP 98
a=accept-types:text/plain
```

7. SIP re-INVITE request (E-CSCF to PSAP)

E-CSCF forward the SIP re-INVITE request to the PSAP.

8. SIP 200 (OK) response (PSAP to E-CSCF)

Upon receiving the SIP re-INVITE request containing the SDP offer, since the PSAP has all resources available, it sends immediately the SIP 200 (OK) response to the SIP re-INVITE request that contains the SDP answer. The SDP answer indicates that the resources are available.

9. SIP 200 (OK) response (E-CSCF to EATF)

E-CSCF forward the SIP 200 (OK) response to the SIP re-INVITE request to the EATF in the originating network.

10-11. SIP ACK request (EATF to PSAP via IM CN subsystem entities)

The EATF generates the SIP ACK request to the SIP 200 (OK) response, and forwards the SIP ACK request to the PSAP.

12-13. SIP 200 (OK) response (EATF to interworking entities via IM CN subsystem entities)

The E- SCC AS generates the SIP 200 (OK) response to the SIP INVITE request, and forwards the SIP 200 (OK) response to the interworking entities.

14-15. SIP ACK request (interworking entities to EATF via IM CN subsystem entities)

The interworking entities generate the SIP ACK request to the SIP 200 (OK) response, and forward the SIP ACK request to the EATF.

16-18: SIP BYE request (EATF to UE A via intermediate IM CN subsystem entities)

The EATF terminates the source access leg, which was using the old IP-CAN, by sending a SIP BYE request to the UE A.

19-21. SIP 200 (OK) response (UE A to E- SCC AS via intermediate IM CN subsystem entities)

Upon receiving the SIP BYE request over the old IP-CAN, the UE A sends a SIP 200 (OK) response over the old IP-CAN to the EATF. Subsequently, the UE A relinquishes all resources pertaining to the old IP-CAN.

NOTE: Steps 18-19 are performed only if the UE A uses Gm after the PS-CS access transfer is completed; otherwise, the UE A and the network release the source access leg locally, without any signalling between the UE A and the network.

22a. CS bearer establishment (interworking entities to UE A)

22b. IP bearer establishment (interworking entities to PSAP)

A.17 Signalling flows for the PS to CS SRVCC for calls in alerting phase

A.17.1 Introduction

The signalling flows in the subclause demonstrate how sessions in alerting phase can be transferred from PS to CS using PS to CS SRVCC procedures. The following signalling flows are included:

- subclause A.17.2 shows an example of PS to CS SRVCC transfer where the incoming call is in alerting phase.
- subclause A.17.3 shows an example of PS to CS SRVCC transfer where the outgoing call is in alerting phase.
- subclause A.17.4 shows an example of PS to CS SRVCC transfer where the incoming call is in alerting phase, but the user answers the call in the PS domain prior to the completion of the network handover procedures and the UE retuning to the CS domain.
- subclause A.17.5 shows an example of PS to CS SRVCC transfer where the incoming call is in alerting phase, but the user answers the call in the PS domain prior to the completion of the network handover procedures but the handover to CS does not succeed.
- subclause A.17.6 shows an example of PS to CS SRVCC transfer where the outgoing call is in alerting phase and the UE has received several forked responses prior to the initiation of access transfer.

A.17.2 Session transfer for incoming call is in alerting phase using PS to CS SRVCC procedure: PS to CS

In the example flow at the figure A.17.2-1, SC UE A has an incoming session with speech media component which is anchored at SCC AS. The session is in alerting phase. Based upon measurement reports sent from the UE to E-UTRAN, the source E-UTRAN decides to trigger a PS to CS SRVCC handover to CS access.

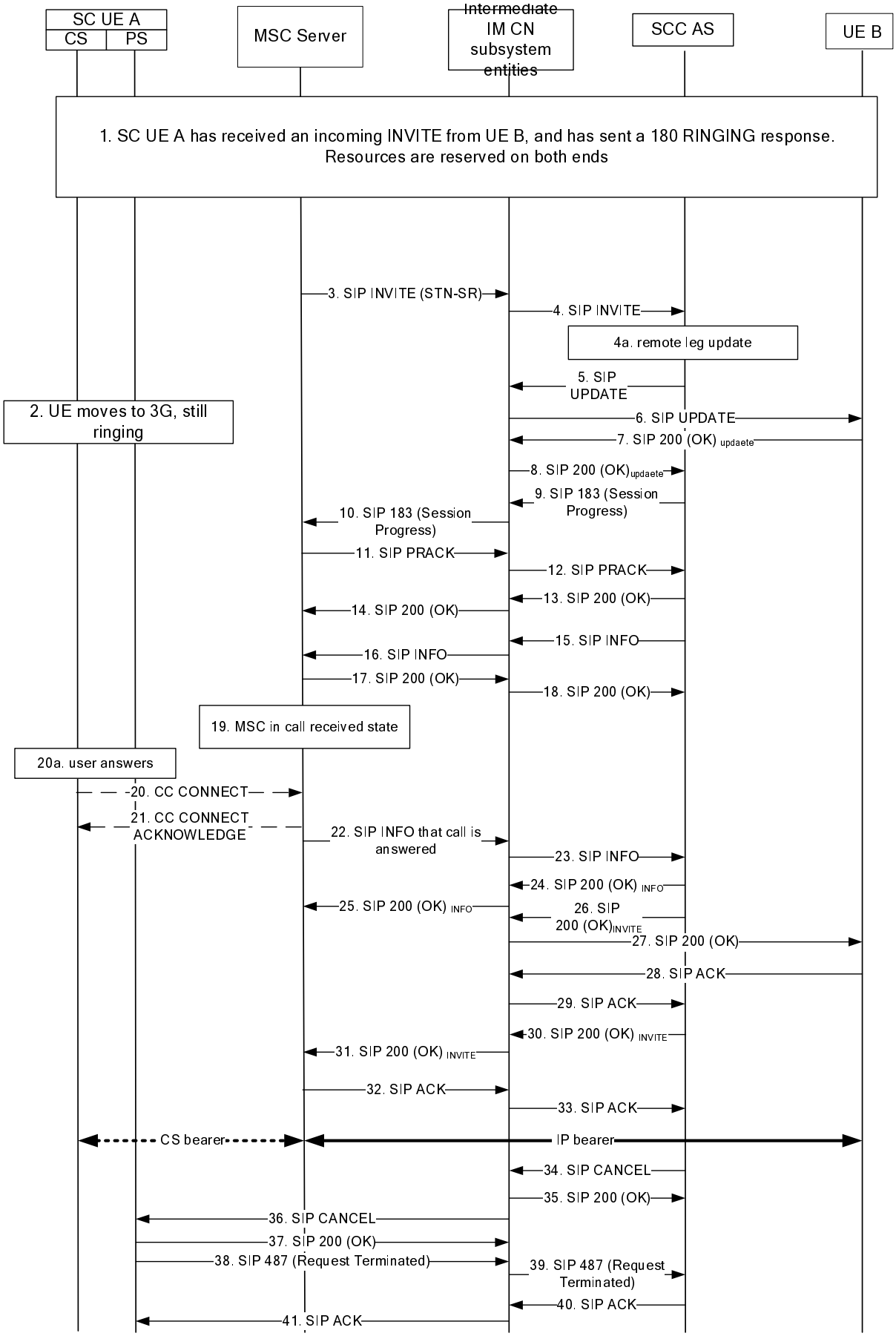


Figure A.17.2-1: PS-CS SRVCC, incoming call in alerting phase

NOTE 1: For clarity, the SIP 100 (Trying) responses are not shown in the signalling flow.

1. **SC UE A has received an incoming call and is in Ringing State**

   The incoming call has been anchored at the SCC AS of SC UE A. Both ends have reserved the resources and SC UE A has sent a SIP 180 (Ringing) response.

2. **SC UE A attaches to the CS domain**

   UE A sends the measurement reports to E-UTRAN, and the source E-UTRAN decides to trigger an PS to CS SRVCC handover to CS access. The MSC server initiates the session transfer with the STN-SR, refer to 3GPP TS 23.237 [9]. The UE continues ringing.

3. **SIP INVITE request transferring the session (MSC server to intermediate IM CN subsystem entities) - see example in table A.17.2-1**

   The MSC server sends an initial SIP INVITE request with STN-SR.

Table A.17.2-1: SIP INVITE request (MSC server to intermediate IM CN subsystem entities)

```
INVITE tel: +1-237-555-3333 SIP/2.0
Via: SIP/2.0/UDP msc1.visit1.net;branch=z9hG4bk731b87
Max-Forwards: 70
Route: <sip:icscf1.visit1.net;lr>
P-Asserted-Identity: <tel:+1-237-555-1111>
P-Charging-Vector: icid-value="AyretyU0dm+6O2IrT5tAFrbHLso=023551024";orig-ioi=visit1.net
Privacy: none
From: <tel:+1-237-555-1111>;tag=171828
To: <tel:+1-237-555-3333>
Call-ID: cb03a0s09a2sdfglkj490334
Cseq: 127 INVITE
Supported: 100rel, precondition, gruu, norefersub
Accept-Contact: *;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
P-Asserted-Service: urn:urn-7:3gpp-service.ims.icsi.mmtel
Contact: <sip:msc1.visit1.net:1357>;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel";
 +g.3gpp.srvcc-alerting
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER
Recv-Info: g.3gpp.state-and-event
Content-Type: application/sdp
Content-Length: (…)
P-Early-Media: supported

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:eee
s=
c=IN IP6 5555::aaa:bbb:ccc:eee
t=0 0
m=audio 3456 RTP/AVP 97 96
a=tcap:1 RTP/AVPF
a=pcfg:1 t=1
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:96 telephone-event
a=maxptime:20
```

Request-URI: contains the STN-SR.

SDP: The SDP contains set of codecs supported by the MGW.

Contact: contains the +g.3gpp.srvcc-alerting feature tag.

4. **SIP INVITE request transferring the session (intermediate IM CN subsystem entities to SCC AS)**

   The SIP INVITE request is routed towards the SCC AS, based on filter criteria in S-CSCF.

4a. Remote Leg Update

The SCC AS correlates SIP INVITE request to the local and remote call legs of the existing session between the UE A and the remote end. The SCC AS performs the Remote Leg update by sending the SIP sending a SIP UPDATE request towards the Remote Leg.

5. **SIP UPDATE request (SCC AS to intermediate IM CN subsystem entities)**

   The SCC AS acting as a B2BUA generates a SIP UPDATE request based upon the received SIP INVITE request and the information previously stored against this session .

6. **SIP UPDATE request (Intermediate IM CN subsystem entities to UE B)**

   The intermediate IM CN subsystem entities forward the SIP UPDATE request to remote UE B.

7. **SIP 200 (OK) response (far end UE to Intermediate IM CN subsystem entities)**

   Upon receiving the SIP UPDATE request containing the SDP offer for the leg to the MSC, the far end sends a SIP 200 (OK) response.

8. **SIP 200 (OK) response (Intermediate IM CN subsystem entities to SCC AS)**

   The intermediate IM CN subsystem entities forward the SIP 200 (OK) response to the SCC AS.

9. **SIP 183 (Session Progress) response (SCC AS to Intermediate IM CN subsystem entities)**

   The SCC AS sends a SIP 183 (Session Progress) response containing the SDP answer as received from the far end UE B. The SDP answer indicates that resources are available. The SIP 183 (Session Progress) response will contain a Recv-Info header field set to g.3gpp.state-and-event.

10. **SIP 183 (Session Progress) response (Intermediate IM CN subsystem entities to MSC server)**

    The intermediate IM CN subsystem entities forward the 183 (Session Progress) response to the MSC server.

11. **SIP PRACK request (MSC server to Intermediate IM CN subsystem entities)**

    The MSC acknowledges the receipt of the SIP 183 (Session Progress) response.

12. **SIP PRACK request (Intermediate IM CN subsystem entities to SCC AS)**

    The intermediate IM CN subsystem forward the SIP PRACK request to the SCC AS.

13. **SIP 200 (OK) response (SCC AS to Intermediate IM CN subsystem entities)**

    The SCC AS acknowledges the SIP PRACK request.

14. **SIP 200 (OK) response (Intermediate IM CN subsystem entities to MSC server)**

    The intermediate IM CN subsystem entities forward the SIP 200 (OK) response to the MSC server.

15. **SIP INFO request (SCC AS to intermediate IM CN subsystem entities) - see example in table A.17.2-2**

Table A.17.2-2: INFO request (SCC AS to intermediate IM CN subsystem entities)

```
INFO sip:msc1.visit1.net:1357 SIP/2.0
Via SIP/2.0/UDP sip:sccas1.home1.net;branch=z9hG4bK332b23.1
Max-Forwards: 68
Route: <sip:scscf1.home1.net;lr>
From: <tel: +1-237-555-3333>;tag=314159
To: <tel:+1-237-555-1111>;tag=171828
Call-ID: cb03a0s09a2sdfglkj490334
Cseq: 129 INFO
Info-Package: g.3gpp.state-and-event
Content-Disposition: Info-Package
Content-Type: application/vnd.3gpp.state-and-event-info+xml
Content-Length:

<?xml version="1.0" encoding="UTF-8"?>
<state-and-event-info>
   <state-info>early</state-info>
   <direction>receiver</direction>
</state-and-event-info>
```

16. SIP INFO request (Intermediate IM CN subsystem entities to MSC server)

The intermediate IM CN subsystem entities forward the SIP INFO request to the MSC server. The MSC server is aware that the call that is transferred is in terminating alerting phase.

17. SIP 200 (OK) response (MSC server to Intermediate IM CN subsystem entities)

The MSC server acknowledges the receipt of the SIP INFO request.

18. SIP 200 (OK) response (Intermediate IM CN subsystem entities to SCC AS)

The intermediate IM CN subsystem entities forward the SIP 200 (OK) response to the SCC AS.

19. MSC goes in Call received state

The MSC enters Call received state due to the information received in the SIP INFO request.

20a. User answers the call

20. CC CONNECT message from SC UE A to MSC server

The SC UE A accepts the call and sends CC CONNECT message.

21 CC CONNECT ACKNOWLEDGE message (MSC server to SC UE A)

22. SIP INFO request (MSC server to intermediate IM CN subsystem entities) - see example in table A.17.2-3

Table A.17.2-3: INFO request (MSC server to intermediate IM CN subsystem entities)

```
INFO sip:sccas1.home1.net;gr SIP/2.0
Via: SIP/2.0/UDP msc1.visit1.net;branch=z9hG4bk731b87
Max-Forwards: 68
Route: <sip:scscf1.home1.net;lr>
From: <tel:+1-237-555-1111>;tag=171828
To: <tel: +1-237-555-3333>;tag=171828
Call-ID: cb03a0s09a2sdfglkj490334
Cseq: 130 INFO
Info-Package: g.3gpp.state-and-event
Content-Disposition: Info-Package
Content-Type: application/vnd.3gpp.state-and-event-info+xml
Content-Length:

<?xml version="1.0" encoding="UTF-8"?>
<state-and-event-info>
   <event>call-accepted</event>
</state-and-event-info>
```

23. SIP INFO request (Intermediate IM CN subsystem entities to SCC AS)

The intermediate IM CN subsystem entities forward the SIP INFO request to the SCC AS. The SCC AS gets informed that the SC UE A has accepted the call.

24 SIP 200 (OK) response (SCC AS to intermediate IM CN subsystem entities)

The SCC AS acknowledges the receipt of the SIP INFO request indicating that the SC UE A has accepted the call

25 SIP 200 (OK) response (Intermediate IM CN subsystem entities to MSC server)

The SIP 200 (OK) response is forwarded to the MSC server.

26 SIP 200 (OK) response (SCC AS to intermediate IM CN subsystem entities)

The SCC AS sends a SIP 200 (OK) response to indicate to the far end that the SC UE A has accepted the call.

27 SIP 200 (OK) response (Intermediate IM CN subsystem entities to far end)

The SIP 200 (OK) response is forwarded to the far end)

28 SIP ACK request (far end to intermediate IM CN subsystem entities)

The far end UE acknowledges the SIP 200 (OK) response received from SCC AS

29 SIP ACK request (Intermediate IM CN subsystem entities to SCC AS)

The SIP ACK request is forwarded to the SCC AS.

30 SIP 200 (OK) response (SCC AS to intermediate IM CN subsystem entities)

The SCC AS sends a SIP 200 (OK) response to indicate the successful access transfer to the MSC server.

31 SIP 200 (OK) response (Intermediate IM CN subsystem entities to far end)

The SIP 200 (OK) response is forwarded to the MSC server.

32 SIP ACK request (MSC server to intermediate IM CN subsystem entities)

MSC server acknowledges the SIP 200 (OK) response received from SCC AS

33. SIP ACK request (Intermediate IM CN subsystem entities to SCC AS)

The SIP ACK request is forwarded to the SCC AS**.**

34-41 SIP CANCEL Processing

The SCC AS cancels the SIP dialog towards the SC UE

NOTE: Steps 36-41 are performed only if the SC UE A usesGm after the PS-CS access transfer in alerting phase is completed; otherwise, the SC UE A and the network release the source access leg locally, without any signalling between the SC UE A and the network

A.17.3 Session transfer for originating call is in alerting phase using PS to CS SRVCC procedure: PS to CS

In the example flow at the figure A.17.3-1, SC UE A has invited for an originating session with speech media component which is anchored at SCC AS. The session is in alerting phase. Based upon measurement reports sent from the UE to E-UTRAN, the source E-UTRAN decides to trigger a PS to CS SRVCC handover to CS access.

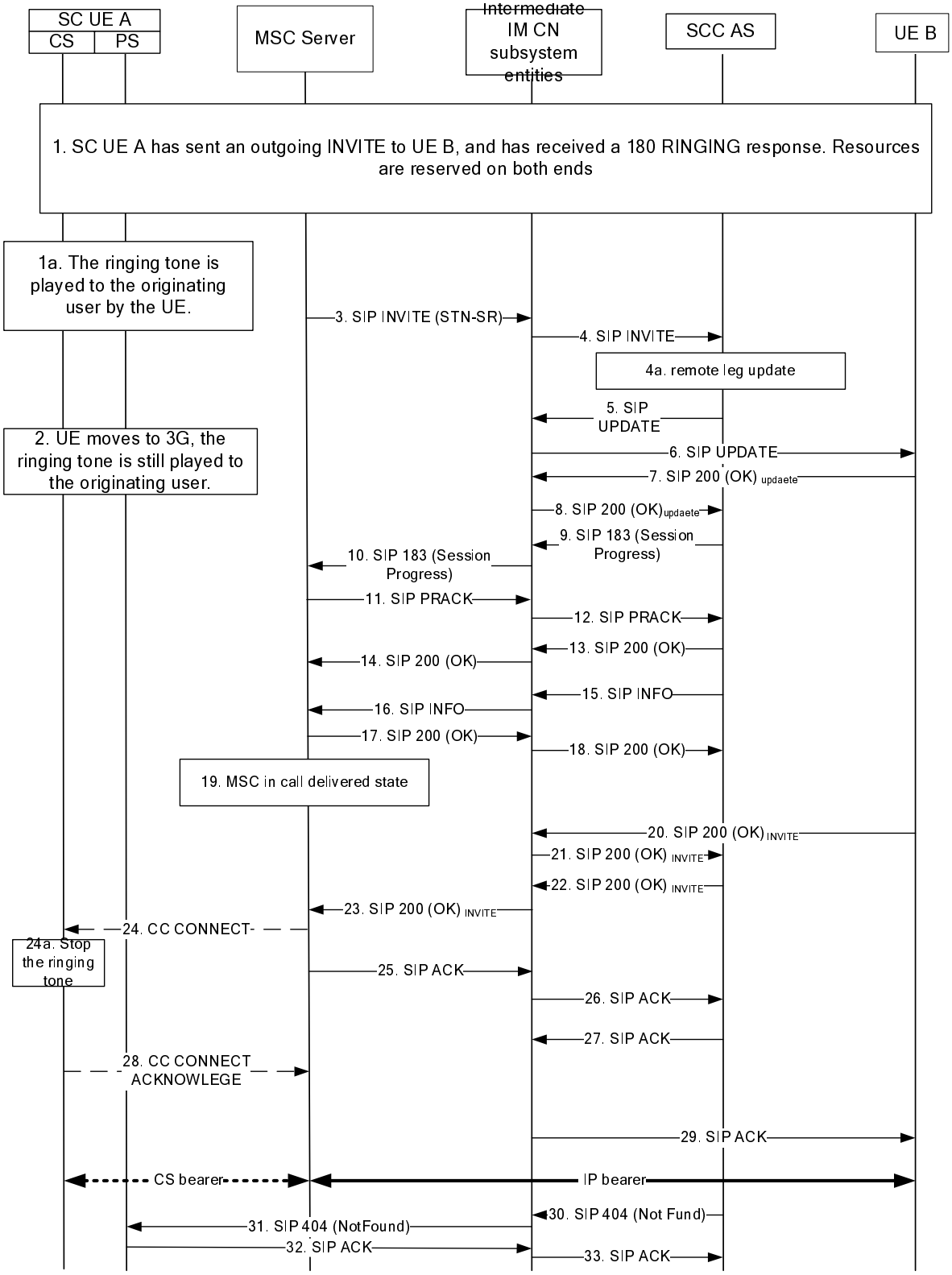


Figure A.17.3-1: PS-CS SRVCC, originating call in alerting phase

NOTE 1: For clarity, the SIP 100 (Trying) responses are not shown in the signalling flow.

1. SC UE A has setup an outgoing call

The outgoing call has been anchored at the SCC AS of SC UE A. Both ends have reserved the resources and SC UE A has received a SIP 180 (Ringing) response.

1a. The ringing tone is played to the originating user

The ringing tone is played by the originating UE as the locally generated ringing tone.

2. SC UE A attaches to the CS domain

UE A sends the measurement reports to E-UTRAN, and the source E-UTRAN decides to trigger an PS to CS SRVCC handover to CS access. The MSC server initiates the session transfer with the STN-SR, refer to 3GPP TS 23.237 [9]. The ringing tone is kept playing to the originating user.

3. SIP INVITE request transferring the session (MSC server to intermediate IM CN subsystem entities) - see example in table A.17.3-1

The MSC server sends an initial SIP INVITE request with STN-SR.

Table A.17.3-1: SIP INVITE request (MSC server to intermediate IM CN subsystem entities)

```
INVITE tel: +1-237-555-3333 SIP/2.0
Via: SIP/2.0/UDP msc1.visit1.net;branch=z9hG4bk731b87
Max-Forwards: 70
Route: <sip:icscf1.visit1.net;lr>
P-Asserted-Identity: <tel:+1-237-555-1111>
P-Charging-Vector: icid-value="AyretyU0dm+6O2IrT5tAFrbHLso=023551024";orig-ioi=visit1.net
Privacy: none
From: <tel:+1-237-555-1111>;tag=171828
To: <tel:+1-237-555-3333>
Call-ID: cb03a0s09a2sdfglkj490334
Cseq: 127 INVITE
Supported: 100rel, precondition, gruu, norefersub
Accept-Contact: *;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
P-Asserted-Service: urn:urn-7:3gpp-service.ims.icsi.mmtel
Contact: <sip: msc1.visit1.net:1357>;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-
 service.ims.icsi.mmtel"; +g.3gpp.srvcc-alerting
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER
Recv-Info: g.3gpp.state-and-event
Content-Type: application/sdp
Content-Length: (…)
P-Early-Media: supported

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:eee
s=
c=IN IP6 5555::aaa:bbb:ccc:eee
t=0 0
m=audio 3456 RTP/AVP 97 96
a=tcap:1 RTP/AVPF
a=pcfg:1 t=1
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:96 telephone-event
a=maxptime:20
```

Request-URI: contains the STN-SR.

SDP: The SDP contains set of codecs supported by the MGW.

Contact: contains the +g.3gpp.srvcc-alerting feature tag.

4. SIP INVITE request transferring the session (intermediate IM CN subsystem entities to SCC AS)

The SIP INVITE is routed towards the SCC AS, based on filter criteria in S-CSCF.

4a. Remote Leg Update

The SCC AS correlates SIP INVITE request to the local and remote call legs of the existing session between the UE A and the remote end. The SCC AS performs the Remote Leg update by sending SIP UPDATE request towards the remote UE B.

5. **SIP UPDATE request (SCC AS to intermediate IM CN subsystem entities)**

   The SCC AS acting as a B2BUA generates a SIP UPDATE request based upon the received SIP INVITE request and the information previously stored against this session.

6. **SIP UPDATE request (Intermediate IM CN subsystem entities to UE B)**

   The intermediate IM CN subsystem entities forward the SIP UPDATE request to remote UE B.

7. **SIP 200 (OK) response (UE B to Intermediate IM CN subsystem entities)**

   Upon receiving the SIP UPDATE request containing the SDP offer for the leg to the MSC, the far end sends a SIP 200 (OK) response.

8. **SIP 200 (OK) response (Intermediate IM CN subsystem entities to SCC AS)**

   The intermediate IM CN subsystem entities forward the SIP 200 (OK) response to the SCC AS.

9. **SIP 183 (Session Progress) response (SCC AS to Intermediate IM CN subsystem entities)**

   The SCC AS sends a SIP 183 (Session Progress) response containing the SDP answer as received from the far end UE. The SDP answer indicates that resources are available

10. **SIP 183 (Session Progress) response (Intermediate IM CN subsystem entities to MSC server)**

    The intermediate IM CN subsystem entities forward the 183 (Session Progress) response to the MSC server.

11. **SIP PRACK request (MSC server to Intermediate IM CN subsystem entities)**

    The MSC acknowledges the receipt of the SIP 183 (Session Progress) response.

12. **SIP PRACK request (Intermediate IM CN subsystem entities to SCC AS)**

    The intermediate IM CN subsystem entities forward the SIP PRACK request to the SCC AS.

13. **SIP 200 (OK) response (SCC AS to Intermediate IM CN subsystem entities)**

    The SCC AS acknowledges the SIP PRACK request.

14. **SIP 200 (OK) response (Intermediate IM CN subsystem entities to MSC server)**

    The intermediate IM CN subsystem entities forward the SIP 200 (OK) response to the MSC server.

15. **SIP INFO request (SCC AS to intermediate IM CN subsystem entities) - see example in table A.17.3-2**

Table A.17.3-2: INFO request (SCC AS to intermediate IM CN subsystem entities)

```
INFO sip: msc1.visit1.net:1357 SIP/2.0
Via SIP/2.0/UDP sip:sccas1.home1.net;branch=z9hG4bK332b23.1
Max-Forwards: 68
Route: <sip:scscf1.home1.net;lr>
From: <tel: +1-237-555-3333>;tag=314159
To: <tel:+1-237-555-1111>;tag=171828
Call-ID: cb03a0s09a2sdfglkj490334
Cseq: 129 INFO
Info-Package: g.3gpp.state-and-event
Content-Disposition: Info-Package
Content-Type: application/vnd.3gpp.state-and-event-info+xml
Content-Length:

<?xml version="1.0" encoding="UTF-8"?>
<state-and-event-info>
   <state-info>early</state-info>
   <direction>initiator</direction>
</state-and-event-info>
```

16. **SIP INFO request (Intermediate IM CN subsystem entities to MSC server)**

The intermediate IM CN subsystem entities forward the SIP INFO request to the MSC server. The MSC server is aware that the call that is transferred is in originating alerting phase.

17. SIP 200 (OK) response (MSC server to Intermediate IM CN subsystem entities)

The MSC Server acknowledges the receipt of the SIP INFO request.

18. SIP 200 (OK) response (Intermediate IM CN subsystem entities to SCC AS)

The intermediate IM CN subsystem entities forward the SIP 200 (OK) response to the SCC AS.

19. MSC goes in Call delivered state

The MSC enters Call delivered state due to the information received in the SIP INFO request.

20. SIP 200 (OK) response (UE B to intermediate IM CN subsystem entities)

The UE B accepts the call and sends a SIP 200 (OK) response.

21. SIP 200 (OK) response (Intermediate IM CN subsystem entities to SCC AS)

The SIP 200 (OK) response is forwarded to SCC AS.

22 SIP 200 (OK) response (SCC AS to intermediate IM CN subsystem entities)

The SCC AS sends the SIP 200 (OK) response to indicate that the terminating UE B has accepted the call.

23 SIP 200 (OK) response (Intermediate IM CN subsystem entities to MSC server)

The SIP 200 (OK) response is forwarded to the MSC server.

24 CC CONNECT message (MSC server to SC UE A)

The MSC server indicates to the SC UA A that the far end has accepted the call.

24a Stop the ringing tone

The UE stops playing the locally generated ringing tone.

25 SIP ACK request (MSC server to intermediate IM CN subsystem entities)

The MSC server acknowledges the SIP 200 (OK) response received from SCC AS

26. SIP ACK request (Intermediate IM CN subsystem entities to SCC AS)

The SIP ACK request is forwarded to the SCC AS.

27 SIP ACK request (SCC AS to intermediate IM CN subsystem entities)

The SCC AS acknowledges the SIP 200 (OK) response received towards far end.

28 CC CONNECTACKNOWLEDGE (MSC server to SC UE A)

SC UE A acknowledges the CS CONNECT message.

29 SIP ACK request (Intermediate IM CN subsystem entities to far end)

The SIP ACK request is forwarded towards the far end.

30 – 33 The SCC AS releases the original source leg towards the SC UE A

The SCC AS sends a SIP 404 (Not Found) response in order to release to original source dialog towards the SC UE A

NOTE: Steps 31-32 are performed only if the SC UE A uses Gm the PS-CS access transfer in alerting phase is completed; otherwise, the SC UE A and the network release the source access leg locally, without any signalling between the SC UE A and the network

A.17.4 User answers in PS domain; Handover to CS successful

In the example flow in figure A.17.4-1, SC UE A has an incoming session with speech media component which is anchored at SCC AS. The session is in alerting phase. Based upon measurement reports sent from the UE to E-UTRAN, the source E-UTRAN decides to trigger a PS to CS SRVCC handover to CS access. However the user answers the call in E-UTRAN and the SC UE sends a SIP 200 (OK) response to the SCC AS. It this scenario the handover to CS is successful.

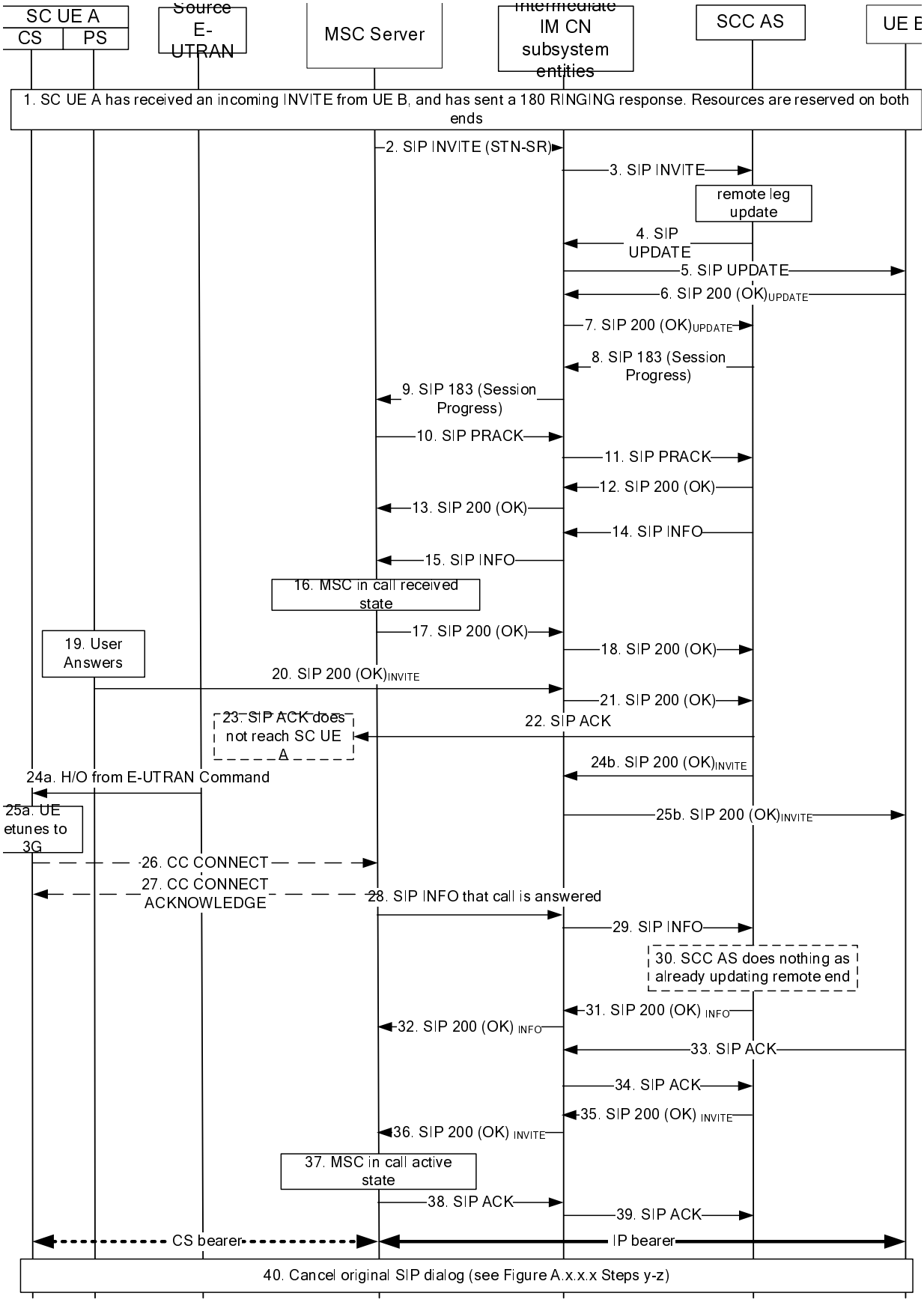


Figure A.17.4-1: SIP 200 (OK) response from SC UE received by SCC AS: Handover to CS successful

NOTE 1: For clarity, the SIP 100 (Trying) responses are not shown in the signalling flow.

1 SC UE A has received an incoming call and is in Ringing State

The incoming call has been anchored at the SCC AS of SC UE A. Both ends have reserved the resources and SC UE A has sent a SIP 180 (Ringing) response.

2-5 MSC server sends session transfer request. SCC AS sends SIP UPDATE to update the remote end

These steps are identical to steps 3-6 in subclause A.17.2.

6 User answers the call when the UE is still in the source E-UTRAN access

7-8 SIP 200 (OK) response (SC UE to intermediate IM CN subsystem entities to SCC AS)

The SCC AS performs no additional actions on receipt of the SIP 200 (OK) i.e. the SCC AS does not confirm reception of the SIP 200 (OK) response with SIP ACK request and performs no actions on dialogs with UE B and with MSC server.

9-21 Continuation of procedure for PS to CS **SRVCC in Alerting Phase**

These steps are identical to steps 7-19 in subclause A.17.2.

22 UE receives H/O command from source E-UTRAN

23 UE retunes to 3G

24 CC CONNECT message from SC UE A to MSC server

The SC UE A sends the CC CONNECT message as it did not receive a SIP ACK to the SIP 200 (OK) sent in step 7.

25-37 Continuation of procedure for PS to CS SRVCC in Alerting Phase

These steps are identical to steps 21-33 in subclause A.17.2.

38-39 SIP ACK request (SCC AS to intermediate IM CN subsystem entities to SC UE)

The SCC AS confirms reception of the SIP 200 (OK) response received in message 8.

40 Release original SIP dialog

The SCC AS releases the SIP dialog towards the SC UE.

NOTE: Step 39 is performed only if the SC UE A uses Gm after the PS-CS access transfer in alerting phase is completed; otherwise, the SC UE A and the network release the source access leg locally, without any signalling between the SC UE A and the network

A.17.5 User answers in PS domain; Handover to CS not successful

In the example flow in figure A.17.5-1, SC UE A has an incoming session with speech media component which is anchored at SCC AS. The session is in alerting phase. Based upon measurement reports sent from the UE to E-UTRAN, the source E-UTRAN decides to trigger a PS to CS SRVCC handover to CS access. However the user answers the call in E-UTRAN and the SC UE sends a SIP 200 (OK) response to the SCC AS. In this scenario the handover to CS is not successful because the source E-UTRAN decides to terminate the handover procedure before its completion. In a similar scenario, the UE can also encounter a failure after it receives the handover command but does not successfully transition to 3GPP UTRAN/GERAN.

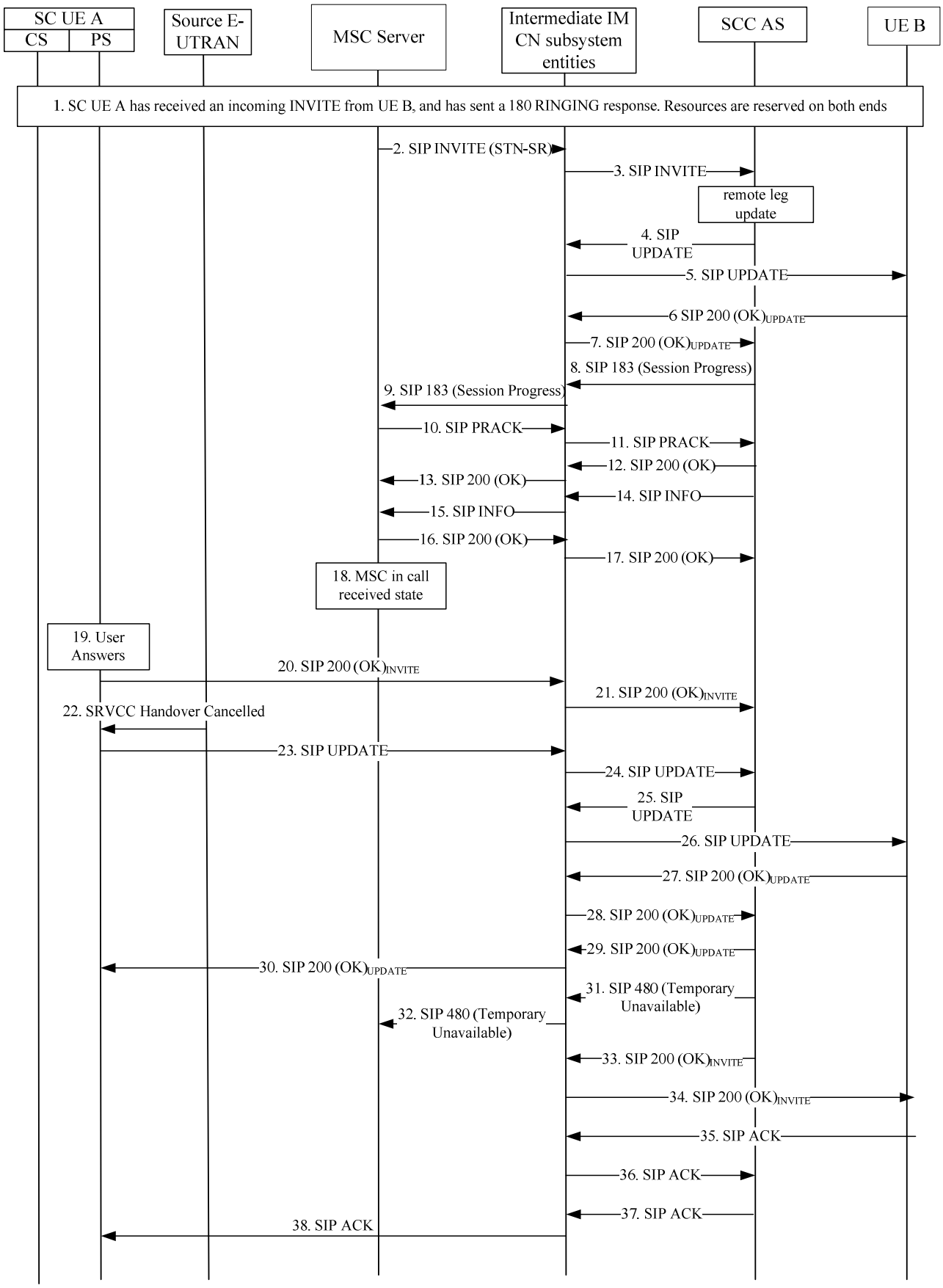


Figure A.17.5-1: SIP 200 (OK) response from SC UE received by SCC AS: Handover cancelled

NOTE 1: For clarity, the SIP 100 (Trying) responses are not shown in the signalling flow.

1 SC UE A has received an incoming call and is in Ringing State

The incoming call has been anchored at the SCC AS of SC UE A. Both ends have reserved the resources and SC UE A has sent a SIP 180 (Ringing) response.

2-18 Continuation of procedure for PS to CS **SRVCC in Alerting Phase**

These steps are identical to steps 3-19 in subclause A.17.2.

19 User answers the call when the UE is still in the source E-UTRAN access

20-21 SIP 200 (OK) response (SC UE to intermediate IM CN subsystem entities to SCC AS)

The SCC AS performs no additional actions on receipt of the SIP 200 (OK) response i.e. the SCC AS does not confirm reception of the SIP 200 (OK) response with SIP ACK request and performs no actions on dialogs with UE B and with MSC server.

9-21 Continuation of procedure for PS to CS **SRVCC in Alerting Phase**

These steps are identical to steps 7-19 in subclause A.17.2.

22 SC UE A receives PS to CS **SRVCC Handover Cancelled command from source E-UTRAN**

23-26 SIP UPDATE request (SC UE to intermediate IM CN subsystem entities to SCC AS to UE B)

SC UE A sends a SIP UPDATE request with a SDP offer, including the media characteristics as used in the existing dialog and with a Reason header field containing protocol "SIP" and reason parameter "cause" with value "487" as specified in IETF RFC 3326 [57] and with reason-text set to "handover cancelled".

NOTE 2: In the case that the handover command was received but the UE did not transition to the CS domain, the UE sends the SIP UPDATE request as described above, but with reason-text set to "failure to transition to CS domain".

27-30 SIP 200 (OK) response to the SIP UPDATE request (UE B to SCC AS to intermediate IM CN subsystem entities to SC UE A)

31-32 SIP 480 (Temporary Unavailable) response (SCC AS to intermediate IM CN subsystem entities to MSC server)

The SCC AS responds to the MSC server with a SIP 480 (Temporary Unavailable) response which indicates that it is unable to go ahead with the session transfer.

33-36 Continuation of procedure for PS to CS SRVCC in Alerting Phase

These steps are identical to steps 25-28 in subclause A.17.2. The SCC AS sends SIP 200 (OK) response to UE B as final confirmation to the original session and UE B sends SIP ACK request back to the SCC AS.

37-38 SIP ACK request (SCC AS to intermediate IM CN subsystem entities to SC UE)

The SCC AS confirms reception of the SIP 200 (OK) response received in message 21.

A.17.6 Session transfer for originating call is in alerting phase with forked responses using PS to CS SRVCC procedure: PS to CS

In the example flow at the figure A.17.6-1, SC UE A initiates an originating session with speech media component which has received several forked responses. The call is anchored at SCC AS and in alerting phase. Based upon measurement reports sent from the UE to E-UTRAN, the source E-UTRAN decides to trigger a PS to CS SRVCC handover to CS access.

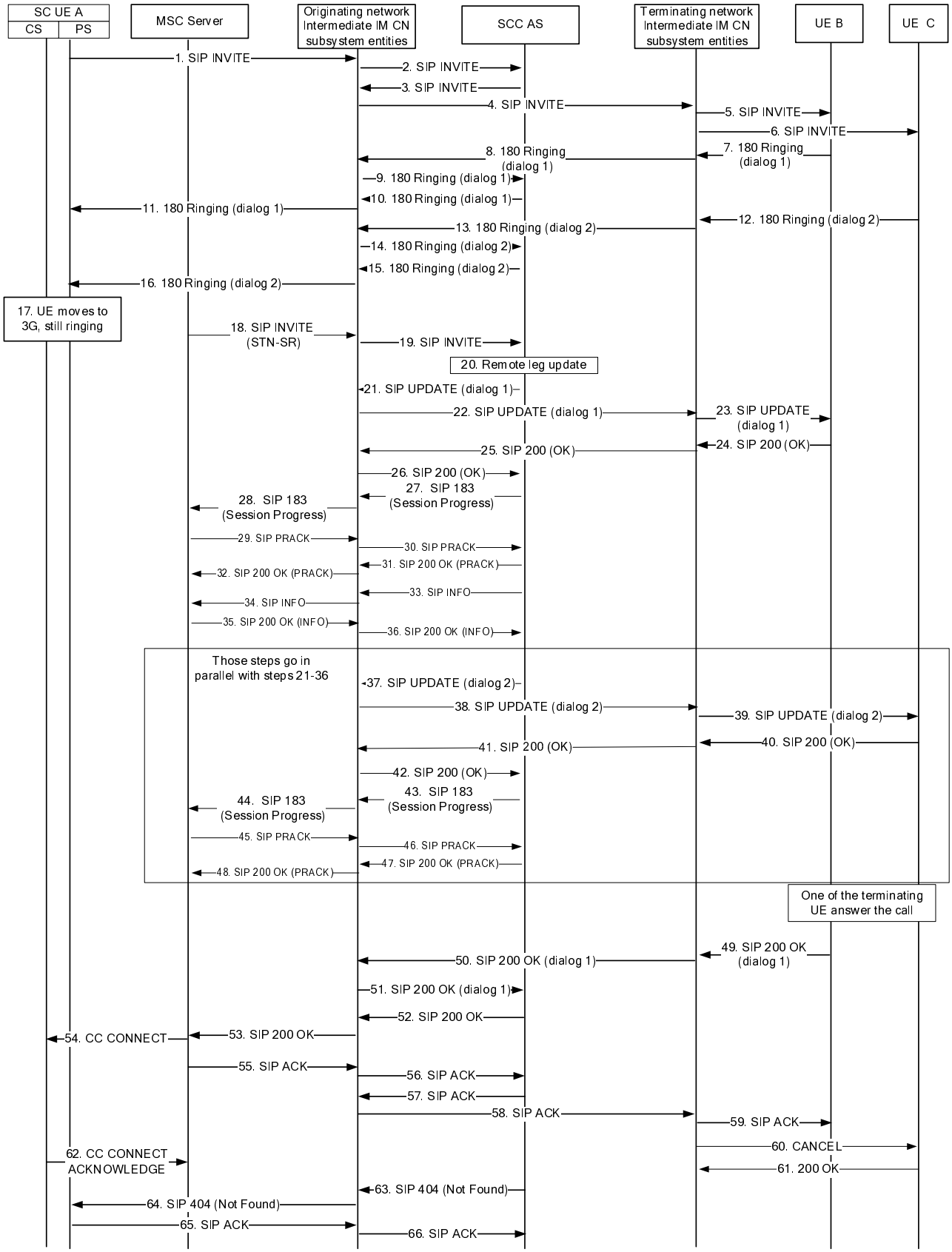


Figure A.17.6-1: PS-CS SRVCC, originating call in alerting phase with forked responses

NOTE 1: For clarity, the SIP 100 (Trying) responses are not shown in the signalling flow.

1-4. SIP INVITE request (SC UE A to Terminating network Intermediate IM CN subsystem entities) - see example in table A.17.6-1

SC UE A sends an outgoing call to the terminating party. The call has been anchored at the SCC AS.

Table A.17.6-1: SIP INVITE request (UE to Intermediate IM CN subsystem entities)

```
INVITE tel:+1-212-555-2222 SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Max-Forwards: 70
Route: <sip:pcscf1.visited2.net:7531;lr;comp=sigcomp>, <sip:orig@scscf1.home1.net;lr>
P-Preferred-Identity: "John Doe" <sip:user1_public1@home1.net>
P-Preferred-Service: urn:urn-7:3gpp-service.ims.icsi.mmtel
P-Access-Network-Info: 3GPP-UTRAN-TDD; utran-cell-id-3gpp=234151D0FCE11
Privacy: none
From: <sip:user1_public1@home1.net>;tag=171828
To: <tel:+1-212-555-2222>
Call-ID: cb03a0s09a2sdfglkj490333
Cseq: 127 INVITE
Require: sec-agree
Supported: precondition, 100rel, gruu, norefersub
Proxy-Require: sec-agree
Security-Verify: ipsec-3gpp; q=0.1; alg=hmac-sha-1-96; spi-c=98765432; spi-s=87654321; port-
   c=8642; port-s=7531
Contact: <sip:user1_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-
   00a0c91e6bf6;comp=sigcomp>;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
Accept-Contact: *;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c=IN IP6 5555::aaa:bbb:ccc:ddd
t=0 0
m=audio 3456 RTP/AVP 97 96
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 telephone-event
```

5. SIP INVITE request (Terminating network Intermediate IM CN subsystem entities to UE B)

The **Terminating network** Intermediate IM CN subsystem entities, i.e. S-CSCF serving for remote UE, determine that the SIP INVITE request should be forked, and send the SIP INVITE request to UE B.

6. SIP INVITE request (Terminating network Intermediate IM CN subsystem entities to UE C)

The **Terminating network** Intermediate IM CN subsystem entities, i.e. S-CSCF serving for remote UE, determine that the SIP INVITE request should be forked, and send the SIP INVITE request to UE C.

7-11. SIP 180 (Ringing) response to SIP INVITE request (UE B to UE A though SCC AS)

The remote UE B responds with SIP 180 (Ringing) response. And a dialog (dialog 1) has been established between UE A and UE B.

Table A.17.6-7: SIP 180 (Ringing) response (UE B to Terminating network Intermediate IM CN subsystem entities)

```
SIP/2.0 180 Ringing
Record-Route: <sip:pcscf1.visited1.net;lr>
Via:
Max-Forwards: 60
P-Asserted-Identity: <tel:+1-212-555-2222>
Privacy:
From:
To: <tel:+1-212-555-2222>; tag=aaa
Call-ID:
Cseq:
Require:
Supported:
Contact: <sip:user2_public1@home2.net;gr=urn:uuid:2ad8950e-48a5-4a74-8d99-
   ad76cc7fc74>;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
Allow:
Content-Type:
Content-Length:

v=0
o=- 462346 5654 IN IP6 1234::55:66:77:88
s=-
c=IN IP6 1234::55:66:77:88
t=0 0
m=audio 4456 RTP/AVP 97 96
b=AS:25.4
a=curr:qos local none
a=curr:qos remote sendrecv
a=des:qos mandatory local sendrecv
a=des:qos mandatory remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 telephone-event
```

12-16. SIP 180 (Ringing) response to SIP INVITE request (UE C to UE A though SCC AS)

The remote UE C responds with SIP 180 (Ringing) response. And a dialog (dialog 2) has been established between UE A and UE B.

Table A.17.6-12: SIP 180 (Ringing) response (UE B to Terminating network Intermediate IM CN subsystem entities)

```
SIP/2.0 180 Ringing
Record-Route: <sip:pcscf1.visited1.net;lr>
Via:
Max-Forwards: 60
P-Asserted-Identity: <tel:+1-212-555-2222>
Privacy:
From:
To: <tel:+1-212-555-2222>; tag=bbb
Call-ID:
Cseq:
Require:
Supported:
Contact: <sip:user2_public1@home2.net;gr=urn:uuid:2ad8950e-48a5-4a74-8d99-
   ad76cc7fc74>;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
Allow:
Content-Type:
Content-Length:

v=0
o=- 462346 5654 IN IP6 1234::55:66:77:88
s=-
c=IN IP6 1234::55:66:77:88
t=0 0
m=audio 4456 RTP/AVP 97 96
b=AS:25.4
a=curr:qos local none
a=curr:qos remote sendrecv
a=des:qos mandatory local sendrecv
a=des:qos mandatory remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 telephone-event
```

17. SC UE A attaches to the CS domain

UE A sends the measurement reports to E-UTRAN, and the source E-UTRAN decides to trigger an PS to CS SRVCC handover to CS access. The MSC server initiates the session transfer with the STN-SR, refer to 3GPP TS 23.237 [9]. The UE continues ringing.

18. SIP INVITE request transferring the session (MSC server to originating network intermediate IM CN subsystem entities) - see example in table A.17.6-18

The MSC server sends an initial SIP INVITE request with STN-SR

Table A.17.6-18: SIP INVITE request (MSC server to intermediate IM CN subsystem entities)

```
INVITE tel: +1-237-555-3333 SIP/2.0
Via: SIP/2.0/UDP msc1.visit1.net;branch=z9hG4bk731b87
Max-Forwards: 70
Route: <sip:icscf1.visit1.net;lr>
P-Asserted-Identity: <tel:+1-237-555-1111>
P-Charging-Vector: icid-value="AyretyU0dm+6O2IrT5tAFrbHLso=023551024";orig-ioi=visit1.net
Privacy: none
From: <tel:+1-237-555-1111>;tag=171828
To: <tel:+1-237-555-3333>
Call-ID: cb03a0s09a2sdfglkj490334
Cseq: 127 INVITE
Supported: 100rel, precondition, gruu, norefersub
Accept-Contact: *;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
P-Asserted-Service: urn:urn-7:3gpp-service.ims.icsi.mmtel
Contact: <sip: msc1.visit1.net:1357>;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER
Recv-Info: g.3gpp.state-and-event
Content-Type: application/sdp
Content-Length: (…)
P-Early-Media: supported

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:eee
s=
c=IN IP6 5555::aaa:bbb:ccc:eee
```

```
t=0 0
m=audio 3456 RTP/AVP 97 96
a=tcap:1 RTP/AVPF
a=pcfg:1 t=1
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:96 telephone-event
a=maxptime:20
```

Request-URI: contains the STN-SR.

SDP: The SDP contains set of codecs supported by the MGW.

19. SIP INVITE request transferring the session (intermediate IM CN subsystem entities to SCC AS)

The SIP INVITE is routed towards the SCC AS, based on filter criteria in S-CSCF.

20. Remote Leg Update

The SCC AS correlates SIP INVITE request to the local and remote call legs of the existing session between the UE A and the remote end. Since the existing session has forked responses, more than one dialog can be correlated to the SIP INVITE due to STN-SR The SCC AS performs the Remote Leg update towards all the correlated dialogs.

21-23. SIP UPDATE request (SCC AS to UE B through Intermediate IM CN subsystem entities)

The SCC AS acting as a B2BUA generates a SIP UPDATE request towards dialog 1 to remote UE B based upon the received SIP INVITE request in step 19.

24-26. SIP 200 (OK) response (Remote UE B to SCC AS through Intermediate IM CN subsystem entities)

Upon receiving the SIP UPDATE request containing the SDP offer for the leg to the MSC, the remote UE B sends SIP 200 (OK).

27-28. SIP 183 (Session Progress) response (SCC AS to MSC server through Intermediate IM CN subsystem entities)

The SCC AS sends a 183 (Session Progress) containing the SDP answer as received from the remote UE B to the MSC server. The SDP answer indicates that resources are available

29-30. SIP PRACK request (MSC server to SCC AS through Intermediate IM CN subsystem entities)

The MSC acknowledges the receipt of the 183 Session Progress by sending SIP PRACK request to the SCC AS.

31-32. SIP 200 (OK) response (SCC AS to MSC server through Intermediate IM CN subsystem entities)

The SCC AS acknowledges the SIP PRACK request with the SIP 200 (OK) response to the MSC server.

33. SIP INFO request (SCC AS to Originating network intermediate IM CN subsystem entities) - see example in table A.17.6-33

Table A.17.6-33: INFO request (SCC AS to intermediate IM CN subsystem entities)

```
INFO sip: msc1.visit1.net:1357 SIP/2.0
Via SIP/2.0/UDP sip:sccas1.home1.net;branch=z9hG4bK332b23.1
Max-Forwards: 68
Route: <sip:scscf1.home1.net;lr>
From: <tel: +1-237-555-3333>;tag=314159
To: <tel:+1-237-555-1111>;tag=171828
Call-ID: cb03a0s09a2sdfglkj490334
Cseq: 129 INFO
Info-Package: g.3gpp.state-and-event
Content-Disposition: Info-Package
Content-Type: application/vnd.3gpp.state-and-event-info+xml
Content-Length:
```

```
<?xml version="1.0" encoding="UTF-8"?>
<state-and-event-info>
   <state-info>early</state-info>
   <direction>initiator</direction>
</state-and-event-info>
```

34. SIP INFO request (Intermediate IM CN subsystem entities to MSC server)

The intermediate IM CN subsystem entities forward the SIP INFO request to the MSC server. The MSC server is aware that the call that is transferred is in originating alerting phase.

35. SIP 200 (OK) response (MSC server to Intermediate IM CN subsystem entities)

The MSC Server acknowledges the receipt of the SIP INFO request.

36. SIP 200 (OK) response (Intermediate IM CN subsystem entities to SCC AS)

The intermediate IM CN subsystem entities forwards the SIP 200 (OK) response to the SCC AS.

37-39. SIP UPDATE request (SCC AS to UE C through Intermediate IM CN subsystem entities)

In parallel with step 21, the SCC AS acting as a B2BUA generates a SIP UPDATE request towards dialog 2 to remote UE C based upon the received SIP INVITE request in step 19.

40-42. SIP 200 (OK) response (Remote UE C to SCC AS through Intermediate IM CN subsystem entities)

Upon receiving the SIP UPDATE request containing the SDP offer for the leg to the MSC, the remote UE C sends a SIP 200 (OK) response.

43-44. SIP 183 (Session Progress) response (SCC AS to MSC server through Intermediate IM CN subsystem entities)

The SCC AS sends a SIP 183 (Session Progress) containing the SDP answer as received from the remote UE C to the MSC server. The SDP answer indicates that resources are available

45-46. SIP PRACK request (MSC server to SCC AS through Intermediate IM CN subsystem entities)

The MSC acknowledges the receipt of the 183 Session Progress by sending SIP PRACK request to the SCC AS.

47-48. SIP 200 (OK) response (SCC AS to MSC server through Intermediate IM CN subsystem entities)

The SCC AS acknowledges the SIP PRACK request with the SIP 200 (OK) response to the MSC server.

49. SIP 200 (OK) response (UE B to intermediate IM CN subsystem entities)

In this example, the remote UE B accepts the call first and sends a SIP 200 (OK) response.

50-51. SIP 200 (OK) response (Intermediate IM CN subsystem entities to SCC AS)

The SIP 200 (OK) response is forwarded to SCC AS.

52-53 SIP 200 (OK) response (SCC AS to MSC server through Intermediate IM CN subsystem entities)

The SIP 200 (OK) response is forwarded to the MSC server based on the route established during step 24-28.

54 CC CONNECT message (MSC server to SC UE A)

The MSC server indicates to the SC UA A that the remote UE B has accepted the call.

55-59. SIP ACK request (MSC server to remote UE B through intermediate IM CN subsystem entities)

The MSC server acknowledges the SIP 200 (OK) response by sending The SIP ACK request to remote UE B.

60 SIP CANCEL request (Terminating network intermediate IM CN subsystem entities to remote UE C)

The intermediate IM CN subsystem entities send the SIP CANCEL request to remote UE C to release the call towards remote UE C.

61 SIP 200 (OK) response to SIP CANCEL request (UE-3 to Intermediate IM CN subsystem entities)

Remote UE C responds SIP 200 (OK) response to the SIP CANCEL request.

62 CC CONNECT ACKNOWLEDGE message (MSC server to SC UE A)

SC UE A acknowledges the CC CONNECT message.

63–66 The SCC AS releases the original source leg towards the SC UE A

The SCC AS sends a SIP 404 (Not Found) response in order to release to original source dialog towards the SC UE A

NOTE: Steps 64-65 are performed only if the SC UE A Gm after the PS-CS access transfer in alerting phase is completed; otherwise, the SC UE A and the network release the source access leg locally, without any signalling between the SC UE A and the network

A.17.7 Session transfer for originating call is in alerting phase using SRVCC procedure with ATCF: PS to CS

In the example flow at the figure A.17.7-1, SC UE A has invited for an originating session with a CAT media component which is anchored at ATGW. The session is in alerting phase. Based upon measurement reports sent from the UE to E-UTRAN, the source E-UTRAN decides to trigger a SRVCC handover to CS access.

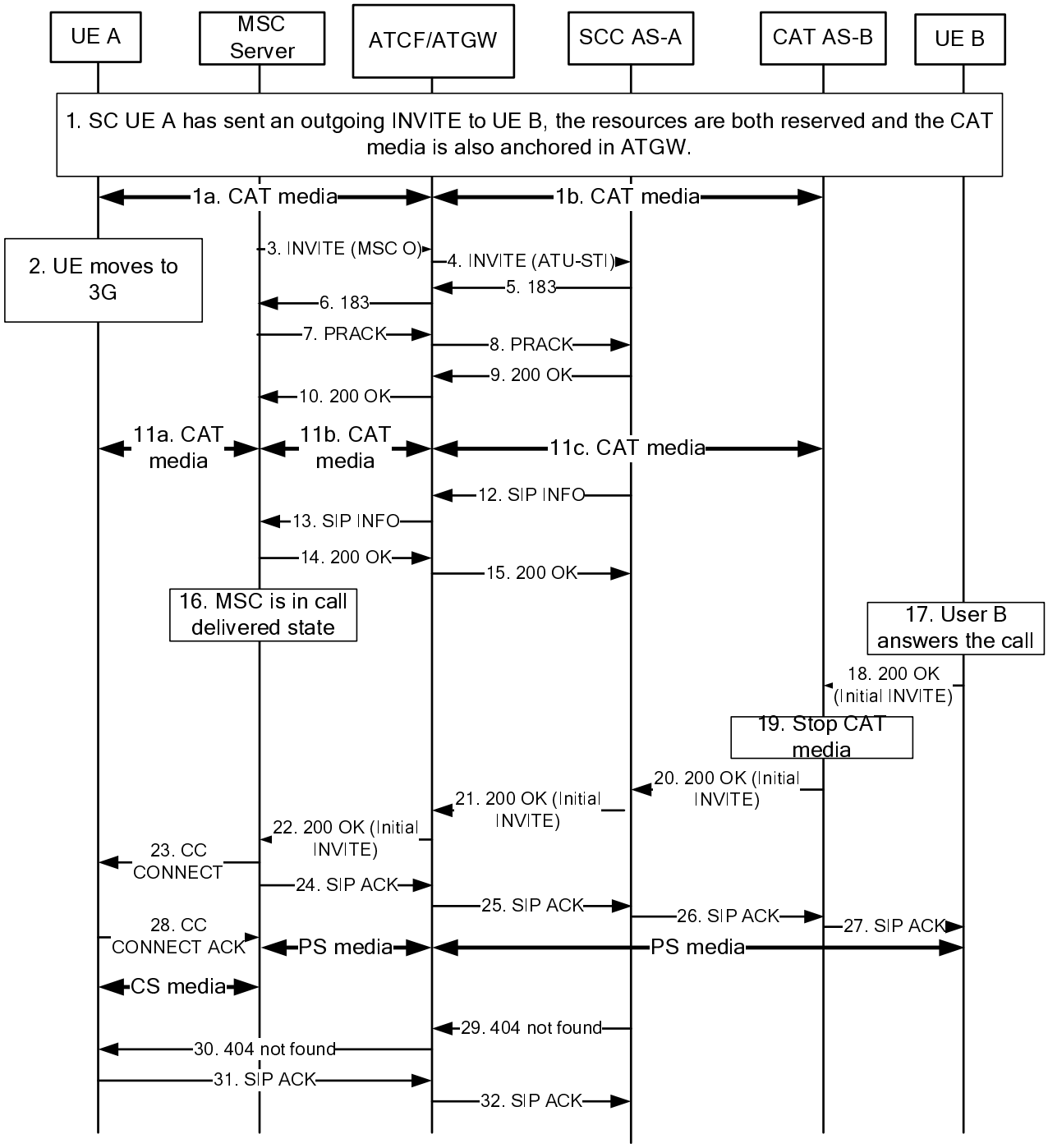


Figure A.17.7-1: PS-CS SRVCC, outgoing call in alerting phase with CAT media anchored at ATGW

NOTE 1: For clarity, the SIP 100 (Trying) responses are not shown in the signalling flow.

1. **SC UE A has setup an outgoing call**

   The outgoing call with the CAT media has been anchored at the ATGW. Both ends have reserved the resources and SC UE A has received a SIP 180 (Ringing) response.

2. **SC UE A attaches to the CS domain**

   UE A sends the measurement reports to E-UTRAN, and the source E-UTRAN decides to trigger an SRVCC handover to CS access.

3. **SIP INVITE request (MSC server to ATCF)-see example in table A.17.7-3**

Table A.17.7-3: SIP INVITE request (MSC server to ATCF)

```
INVITE tel:+1-237-555-3333 SIP/2.0
Via: SIP/2.0/UDP msc1.visit1.net;branch=z9hG4bk731b87
Max-Forwards: 70
P-Asserted-Identity: <tel:+1-237-555-2222>
P-Charging-Vector: icid-value="AyretyU0dm+6O2IrT5tAFrbHLso=023551024";orig-ioi=visit1.net
Privacy: none
P-Early-Media: supported
From: <tel:+1-237-555-1111>;tag=171828
To: <tel:+1-237-555-3333>
Call-ID: cb03a0s09a2sdfglkj490334
Cseq: 127 INVITE
Supported: 100rel, precondition, gruu, norefersub
Accept-Contact: *;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
P-Asserted-Service: urn:urn-7:3gpp-service.ims.icsi.mmtel
Contact: <sip: msc1.visit1.net:1357>;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:eee
s=
c=IN IP6 5555::aaa:bbb:ccc:eee
t=0 0
m=audio 3456 RTP/AVP 97 96
a=tcap:1 RTP/AVPF
a=pcfg:1 t=1
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:96 telephone-event
a=maxptime:20
```

Request-URI: contains the STN-SR, as routed to the ATCF.

SDP: The SDP contains preconfigured set of codecs supported by the MGW.

P-Asserted-Identity: the C-MSISDN of the served UE.

4. **SIP INVITE request (ATCF to SCC AS)-see example in table A.17.7-4**

After receiving the access transfer message, the ATCF establishes a new dialog with the SCC AS by sending a new SIP INVITE request to the SCC AS using the stored ATU-STI for PS to CS SRVCC. The ATCF updates the SCC AS via the new dialog indicating that the transfer has taken place. As there is no update in the SDP information, no remote end update will be performed.

Table A.17.7-4: SIP INVITE request (ATCF to SCC AS)

```
INVITE sip:AUT-STI1@sccas.home1.net SIP/2.0
Via: SIP/2.0/UDP actf.visited2.net:5060;branch=z9hG4bk731b87
Max-Forwards: 70
P-Asserted-Identity: <tel:+1-237-555-2222>
P-Charging-Vector: icid-value="AyretyU0dm+6O2IrT5tAFrbHLso=023551024";orig-ioi=visit1.net
Privacy: none
P-Early-Media: supported
From: <tel:+1-237-555-3333>;tag=1888828
To: <tel:+1-237-555-4444>
Call-ID: cb03a0s09a2sdfglkj490444
Cseq: 127 INVITE
Supported: 100rel, precondition, gruu, norefersub
Require: tdialog,
Record-Route:<sip: actf.visited2.net:5060;lr>
Target-Dialog: me03a0s09a2sdfgjkl491777; remote-tag=774321; local-tag=64727891
Accept-Contact: *;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
P-Asserted-Service: urn:urn-7:3gpp-service.ims.icsi.mmtel
Contact: <sip: msc1.visit1.net:1357>;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER
Content-Type: application/sdp
```

```
Content-Length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ggg
s=
c=IN IP6 5555::aaa:bbb:ccc:ggg
t=0 0
m=audio 3456 RTP/AVP 97 96
a=tcap:1 RTP/AVPF
a=pcfg:1 t=1
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:96 telephone-event
a=maxptime:20
```

Request-URI: contains the ATU-STI for PS to CS SRVCC, that resolves (e.g. via DNS access) to the IP address of the I-CSCF.

Target-Dialog: specifies that the existing dialog is related with this request.

Record-Route: contains the SIP URI of the ATCF, where the ATCF expect to receive the in-dialog request from the SCC AS.

Require: the "tdialog" option tag indicate that the support for Target-Dialog header field is required.

P-Asserted-Identity: the C-MSISDN of the served UE.

SDP: the media information at ATGW.

5-6. SIP 183 (Session Progress) response (SCC AS to MSC Server)

The session achored is in alerting phase, the SCC AS sends the SIP 183Session Progress response to the MSC server.

7-8. SIP PRACK reqeust (MSC Server to SCC AS)

9-10. SIP 200 (OK) response (SCC AS to MSC Server)

11. The CAT media is transferred to the CS access, and kept playing to the originating user.

9. SIP 183 (Session Progress) response (SCC AS to ATCF)

10. SIP PRACK request (ATCF to SCC AS)

11. SIP 200 (OK) response (SCC AS to ATCF)

12-13. SIP INFO request (SCC AS to MSC Server) - see example in table A.17.7-12

Table A.17.7-12: INFO request (SCC AS to MSC Server)

```
INFO sip: msc1.visit1.net:1357 SIP/2.0
Via SIP/2.0/UDP sip:sccas1.home1.net;branch=z9hG4bK332b23.1
Max-Forwards: 68
Route: <sip:scscf1.home1.net;lr>
From: <tel:+1-237-555-3333>;tag=314159
To: <tel:+1-237-555-1111>;tag=171828
Call-ID: cb03a0s09a2sdfglkj490334
Cseq: 129 INFO
Info-Package: g.3gpp.state-and-event-info
Content-Type: application/vnd.3gpp.state-and-event-info+xml
Content-Length:

<?xml version="1.0" encoding="UTF-8"?>
<state-and-event-info>
   <state-info>early</state-info>
   <direction>initiator</direction>
```

```
</state-and-event-info>
```

14-15. SIP 200 (OK) response (MSC server to SCC AS)

The MSC Server acknowledges the receipt of the SIP INFO request.

16. MSC goes in Call delivered state

The MSC enters Call delivered state due to the information received in the SIP INFO request.

17. The User B answers the call

18. SIP 200 (OK) response (UE B to CAT AS B)

The UE B accepts the call and sends a SIP 200 (OK) response.

19. The CAT AS stops the CAT media upon receving the SIP 200 (OK) response.

20-22. SIP 200 (OK) response (CAT AS to MSC Server)

The SIP 200 (OK) response is forwarded to MSC Server.

23. CC CONNECT message (MSC server to SC UE A)

The MSC server indicates to the SC UA A that the far end has accepted the call.

24-27. SIP ACK request (MSC server to UE B)

The MSC server acknowledges the SIP 200 (OK) response.

28. CC CONNECTACKNOWLEDGE (SC UE A to MSC server)

SC UE A acknowledges the CS CONNECT message.

29–32. The SCC AS releases the original source leg towards the SC UE A

The SCC AS sends a SIP 404 (Not Found) response in order to release to original source dialog towards the SC UE A

NOTE: Steps 29-32 are performed only if the SC UE A uses Gm the PS-CS access transfer in alerting phase is completed; otherwise, the SC UE A and the network release the source access leg locally, without any signalling between the SC UE A and the network.

A.17.8 Session transfer for terminating call is in alerting phase using SRVCC procedure with ATCF: PS to CS

In the example flow at the figure A.17.8-1, SC UE B has an incoming session with speech media component which is anchored at ATCF. The session is in alerting phase, and the CAT media is played to the originating UE A. Based upon measurement reports sent from the UE to E-UTRAN, the source E-UTRAN decides to trigger a PS to CS SRVCC handover to CS access.

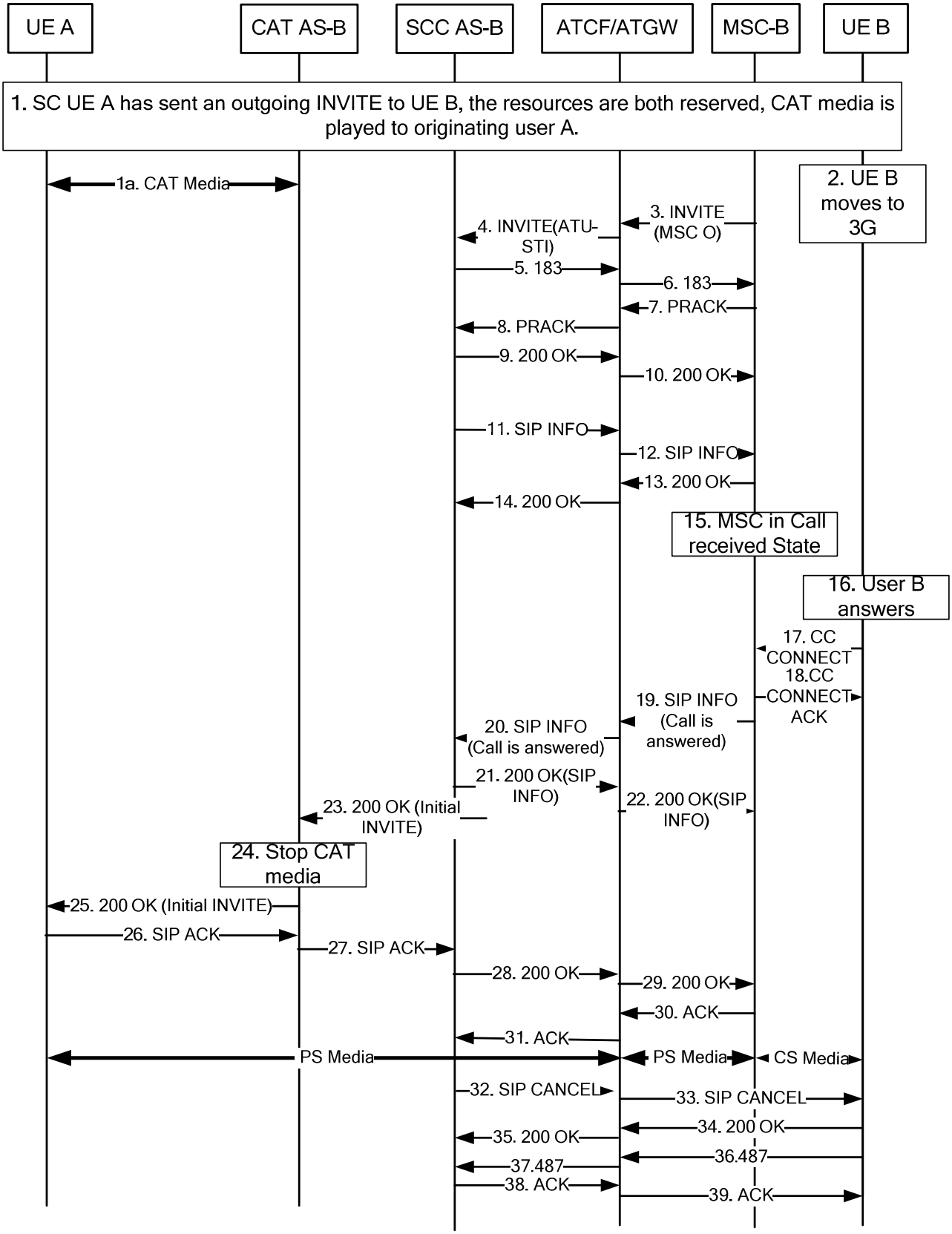


Figure A.17.8-1: PS-CS SRVCC, incoming call in alerting phase with CAT media

NOTE 1: For clarity, the SIP 100 (Trying) responses are not shown in the signalling flow.

1. SC UE B has received an incoming call and is in Ringing State

The incoming call has been anchored at the ATCF of SC UE B. Both ends have reserved the resources and SC UE A has sent a SIP 180 (Ringing) response.

2. **SC UE B attaches to the CS domain**

   UE B sends the measurement reports to E-UTRAN, and the source E-UTRAN decides to trigger an PS to CS SRVCC handover to CS access. The MSC server initiates the session transfer with the STN-SR, refer to 3GPP TS 23.237 [9]. The UE continues ringing.

3. **SIP INVITE request (MSC server to ATCF)-see example in table A.17.8-3**

Table A.17.8-3: SIP INVITE request (MSC server to ATCF)

```
INVITE tel:+1-237-555-3333 SIP/2.0
Via: SIP/2.0/UDP msc1.visit1.net;branch=z9hG4bk731b87
Max-Forwards: 70
P-Asserted-Identity: <tel:+1-237-555-2222>
P-Charging-Vector: icid-value="AyretyU0dm+6O2IrT5tAFrbHLso=023551024";orig-ioi=visit1.net
Privacy: none
P-Early-Media: supported
From: <tel:+1-237-555-1111>;tag=171828
To: <tel:+1-237-555-3333>
Call-ID: cb03a0s09a2sdfglkj490334
Cseq: 127 INVITE
Supported: 100rel, precondition, gruu, norefersub
Accept-Contact: *;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
P-Asserted-Service: urn:urn-7:3gpp-service.ims.icsi.mmtel
Contact: <sip: msc1.visit1.net:1357>;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:eee
s=
c=IN IP6 5555::aaa:bbb:ccc:eee
t=0 0
m=audio 3456 RTP/AVP 97 96
a=tcap:1 RTP/AVPF
a=pcfg:1 t=1
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:96 telephone-event
a=maxptime:20
```

Request-URI: contains the STN-SR, as routed to the ATCF.

SDP: The SDP contains preconfigured set of codecs supported by the MGW.

P-Asserted-Identity: the C-MSISDN of the served UE.

4. **SIP INVITE request (ATCF to SCC AS)-see example in table A.17.8-4**

   After receiving the access transfer message, the ATCF establishes a new dialog with the SCC AS by sending a new SIP INVITE request to the SCC AS using the stored ATU-STI for PS to CS SRVCC. The ATCF updates the SCC AS via the new dialog indicating that the transfer has taken place. As there is no update in the SDP information, no remote end update will be performed.

Table A.17.8-4: SIP INVITE request (ATCF to SCC AS)

```
INVITE sip:AUT-STI1@sccas.home1.net SIP/2.0
Via: SIP/2.0/UDP actf.visited2.net:5060;branch=z9hG4bk731b87
Max-Forwards: 70
P-Asserted-Identity: <tel:+1-237-555-2222>
P-Charging-Vector: icid-value="AyretyU0dm+6O2IrT5tAFrbHLso=023551024";orig-ioi=visit1.net
Privacy: none
P-Early-Media: supported
```

```
From: <tel:+1-237-555-3333>;tag=1888828
To: <tel:+1-237-555-4444>
Call-ID: cb03a0s09a2sdfglkj490444
Cseq: 127 INVITE
Supported: 100rel, precondition, gruu, norefersub
Require: tdialog,
Record-Route:<sip: actf.visited2.net:5060;lr>
Target-Dialog: me03a0s09a2sdfgjkl491777; remote-tag=774321; local-tag=64727891
Accept-Contact: *;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
P-Asserted-Service: urn:urn-7:3gpp-service.ims.icsi.mmtel
Contact: <sip: msc1.visit1.net:1357>;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ggg
s=
c=IN IP6 5555::aaa:bbb:ccc:ggg
t=0 0
m=audio 3456 RTP/AVP 97 96
a=tcap:1 RTP/AVPF
a=pcfg:1 t=1
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:96 telephone-event
a=maxptime:20
```

Request-URI: contains the ATU-STI for PS to CS SRVCC, that resolves (e.g. via DNS access) to the IP address of the I-CSCF.

Target-Dialog: specifies that the existing dialog is related with this request.

Record-Route: contains the SIP URI of the ATCF, where the ATCF expect to receive the in-dialog request from the SCC AS.

Require: the "tdialog" option tag indicate that the support for Target-Dialog header field is required.

P-Asserted-Identity: the C-MSISDN of the served UE.

SDP: the media information at ATGW.

5-6. SIP 183 (Session Progress) response (SCC AS to MSC server)

The session achored is in alerting phase, the SCC AS sends the SIP 183Session Progress response to the MSC server.

7-8. SIP PRACK request (MSC Server to SCC AS)

9-10. SIP 200 (OK) response (ATCF to MSC Server)

11-12. SIP INFO request (SCC AS to MSC Server) - see example in table A.17.8-11

Table A.17.8-11: INFO request (SCC AS to MSC Server)

```
INFO sip:msc1.visit1.net:1357 SIP/2.0
Via SIP/2.0/UDP sip:sccas1.home1.net;branch=z9hG4bK332b23.1
Max-Forwards: 68
Route: <sip:scscf1.home1.net;lr>
From: <tel:+1-237-555-3333>;tag=314159
To: <tel:+1-237-555-1111>;tag=171828
Call-ID: cb03a0s09a2sdfglkj490334
Cseq: 129 INFO
Info-Package: g.3gpp.state-and-event-info
Content-Type: application/vnd.3gpp.state-and-event-info+xml
Content-Length:

<?xml version="1.0" encoding="UTF-8"?>
```

```
<state-and-event-info>
   <state-info>early</state-info>
   <direction>receiver</direction>
</state-and-event-info>
```

13-14. SIP 200 (OK) response (MSC server to ATCF)

The MSC server acknowledges the receipt of the SIP INFO request.

15. MSC goes in Call received state

The MSC enters Call received state due to the information received in the SIP INFO request.

16. The User B answers the call

17. CC CONNECT message from SC UE B to MSC server

The SC UE B accepts the call and sends CC CONNECT message.

18. CC CONNECT ACKNOWLEDGE message (MSC server to SC UE B)

19-20. SIP INFO request (MSC server to SCC AS) - see example in table A.17.8-19

Table A.17.8-19: INFO request (MSC server to SCC AS)

```
INFO sip:sccas1.home1.net;gr SIP/2.0
Via: SIP/2.0/UDP msc1.visit1.net;branch=z9hG4bk731b87
Max-Forwards: 68
Route: <sip:scscf1.home1.net;lr>
From: <tel:+1-237-555-1111>;tag=171828
To: <tel:+1-237-555-3333>;tag=171828
Call-ID: cb03a0s09a2sdfglkj490334
Cseq: 130 INFO
Info-Package: g.3gpp.state-and-event-info
Content-Type: application/vnd.3gpp.state-and-event-info+xml
Content-Length:

<?xml version="1.0" encoding="UTF-8"?>
<state-and-event-info>
   <event>call-accepted</event>
</state-and-event-info>
```

21-22. SIP 200 (OK) response (SCC AS to MSC Server)

The SCC AS acknowledges the receipt of the SIP INFO request indicating that the SC UE B has accepted the call.

23. SIP 200 (OK) response (SCC AS to CAT AS)

The SIP 200 (OK) response is forwarded to the CAT AS by SCC AS.

24. The CAT AS stops the CAT media upon receving the SIP 200 (OK) response.

25. SIP 200 (OK) response (CAT AS to UE A)

The SIP 200 (OK) response is forwarded to UE A.

26-27 SIP ACK request (far end UE A to SCC AS)

The far end UE acknowledges the SIP 200 (OK) response received from SCC AS

28-29 SIP 200 (OK) response (SCC AS to MSC Server)

The SCC AS sends a SIP 200 (OK) response to indicate the successful access transfer to the MSC server.

30-31. SIP ACK request (MSC server to SCC AS)

MSC server acknowledges the SIP 200 (OK) response received from SCC AS.

32-39. SIP CANCEL Processing

The SCC AS cancels the SIP dialog towards the SC UE

NOTE: Steps 32-39 are performed only if the SC UE B uses Gm after the PS-CS access transfer in alerting phase is completed; otherwise, the SC UE B and the network release the source access leg locally, without any signalling between the SC UE B and the network.

A.18 Signalling flows for PS to CS Access Transfer: PS to CS SRVCC enhancements using ATCF

A.18.1 Introduction

The signalling flows in the subclause demonstrate the PS to CS SRVCC enhancements using ATCF. The following signalling flows are included:

- subclause A.18.2 shows an example of PS to CS SRVCC enhancements using ATCF and without media anchored.
- subclause A.18.3 shows an example of PS to CS SRVCC enhancements using ATCF and media anchored.

A.18.2 Signalling flows for PS to CS Access Transfer: PS to CS SRVCC enhancements using ATCF and without media anchored

The signalling flow shown in figure A.18.2-1 gives an example for PS to CS access transfer when using ATCF enhancements and without media anchored. In this case, the ATCF has been included in the path for subsequent transactions created at registration, but media has not been anchored in ATGW.

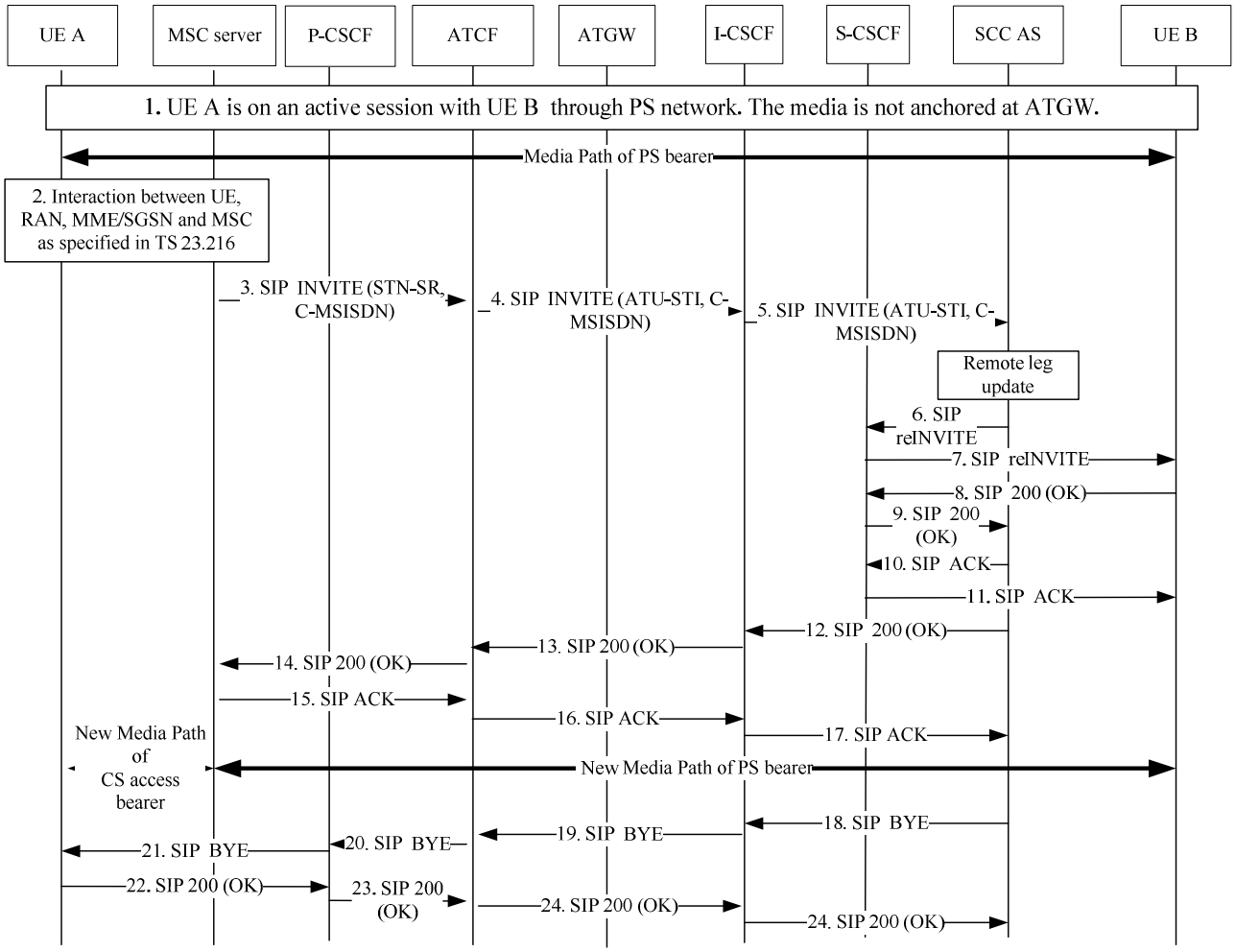


Figure A.18.2-1 Signalling flows for PS to CS access transfer: PS to CS SRVCC enhancements using ATCF and without media anchored

NOTE 1: For clarity, the SIP 100 (Trying) responses are not shown in the signalling flow.

1. **UE A is on an active session with UE B**

   There is an ongoing PS bearer between the UE A and the remote end UE B. The media is not anchored at ATGW.

2. **SC UE A attaches to the CS domain**

   UE A sends the measurement reports to E-UTRAN, and the source E-UTRAN decides to trigger an PS to CS SRVCC handover to CS access. The MSC server initiates the session transfer with the STN-SR, refer to 3GPP TS 23.216 [49].

3. **SIP INVITE request (MSC server to ATCF)-see example in table A.18.2-3**

Table A.18.2-3: SIP INVITE request (MSC server to ATCF)

```
INVITE tel: +1-237-555-3333 SIP/2.0
Via: SIP/2.0/UDP msc1.visit1.net;branch=z9hG4bk731b87
Max-Forwards: 70
P-Asserted-Identity: <tel:+1-237-555-2222>
P-Charging-Vector: icid-value="AyretyU0dm+6O2IrT5tAFrbHLso=023551024";orig-ioi=visit1.net
Privacy: none
From: <tel:+1-237-555-1111>;tag=171828
To: <tel: +1-237-555-3333>
Call-ID: cb03a0s09a2sdfglkj490334
Cseq: 127 INVITE
Supported: 100rel, precondition, gruu
Accept-Contact: *;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
P-Asserted-Service: urn:urn-7:3gpp-service.ims.icsi.mmtel
```

```
Contact: <sip: msc1.visit1.net:1357>;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:eee
s=
c=IN IP6 5555::aaa:bbb:ccc:eee
t=0 0
m=audio 3456 RTP/AVP 97 96
a=tcap:1 RTP/AVPF
a=pcfg:1 t=1
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:96 telephone-event
a=maxptime:20
```

Request-URI: contains the STN-SR, as routed to the ATCF.

SDP: The SDP contains preconfigured set of codecs supported by the MGW.

P-Asserted-Identity: the C-MSISDN of the served UE.

4-5. SIP INVITE request (ATCF to SCC AS via I-CSCF)- see example in table A.18.2-4

Since the media has not been anchored at the ATGW, the ATCF forwards the SIP INVITE request to the SCC AS by replacing the request URI to the stored ATU-STI for PS to CS SRVCC.

Table A.18.2-4: SIP INVITE request (ATCF to I-CSCF)

```
INVITE sip:AUT-STI1@sccas.home1.net SIP/2.0
Via: SIP/2.0/UDP msc1.visit1.net;branch=z9hG4bk731b87
Max-Forwards: 70
P-Asserted-Identity: <tel:+1-237-555-2222>
P-Charging-Vector: icid-value="AyretyU0dm+6O2IrT5tAFrbHLso=023551024";orig-ioi=visit1.net
Privacy: none
From: <tel:+1-237-555-1111>;tag=171828
To: <tel: +1-237-555-4444>
Call-ID: cb03a0s09a2sdfglkj490334
Cseq: 127 INVITE
Supported: 100rel, precondition, gruu
Accept-Contact: *;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
P-Asserted-Service: urn:urn-7:3gpp-service.ims.icsi.mmtel
Contact: <sip: msc1.visit1.net:1357>;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:eee
s=
c=IN IP6 5555::aaa:bbb:ccc:eee
t=0 0
m=audio 3456 RTP/AVP 97 96
a=tcap:1 RTP/AVPF
a=pcfg:1 t=1
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:96 telephone-event
a=maxptime:20
```

6-7. SIP re-INVITE request (SCC AS to UE B via S-CSCF)

The SCC AS based on the content of the C-MSISDN correlates the SIP INVITE request to the local and remote call legs of the existing session between the UE A and the remote end. The SCC AS performs the Remote Leg update by sending the SIP re-INVITE request towards the Remote Leg.

8-9. SIP 200 (OK) response (UE B to SCC AS via S-CSCF)

Upon receiving the SIP re-INVITE request containing the SDP offer, since the UE B has all resources available, it sends immediately the SIP 200 (OK) response to the SIP re-INVITE request that contains the SDP answer. The SDP answer indicates that the resources are available.

10-11. SIP ACK request (SCC AS to UE B via S-CSCF)

The SCC AS generates the SIP ACK request to the SIP 200 (OK) response, and forwards the SIP ACK request to the remote UE B.

12-13. SIP 200 (OK) response (SCC AS to ATCF via I-CSCF)

The SCC AS generates the SIP 200 (OK) response to the SIP INVITE request, and forwards the SIP 200 (OK) response towards the ATCF.

14. SIP 200 (OK) response (ATCF to MSC server)

The ATCF generates the SIP 200 (OK) response to the SIP INVITE request, and forwards the SIP 200 (OK) response towards the MSC server.

15. SIP ACK request (MSC server to ATCF)

The MSC server generates the SIP ACK request to the SIP 200 (OK) response, and forwards it to the ATCF.

16-17. SIP ACK request (ATCF to SCC AS via I-CSCF)

The ATCF generates the SIP ACK request to the SIP 200 (OK) response, and forwards it to the SCC AS.

18-21. SIP BYE request (SCC AS to UE via I-CSCF, ATCF and P-CSCF)

The SCC AS terminates the source access leg, which was using the old IP-CAN, by sending a SIP BYE request to the UE A.

22-24. SIP 200 (OK) response (UE A to SCC AS via P-CSCF, ATCF and I-CSCF)

Upon receiving the SIP BYE request over the old IP-CAN, the UE A sends a SIP 200 (OK) response over the old IP-CAN to the SCC AS. Subsequently, the SC UE A relinquishes all resources pertaining to the old IP-CAN.

NOTE: Steps 21-22 are performed only if the UE A uses Gm after the PS-CS access transfer is completed; otherwise, the UE A and the network release the source access leg locally, without any signalling between the UE A and the network

A.18.3 Signalling flows for PS to CS Access Transfer: PS to CS SRVCC enhancements using ATCF and media anchored

The signalling flow shown in figure A.18.3-1 gives an example for PS to CS access transfer for PS to CS SRVCC enhancements using ATCF and media anchored. In this case, the media is anchored in ATGW and ATCF has been included in the path for subsequent transactions created at registration.

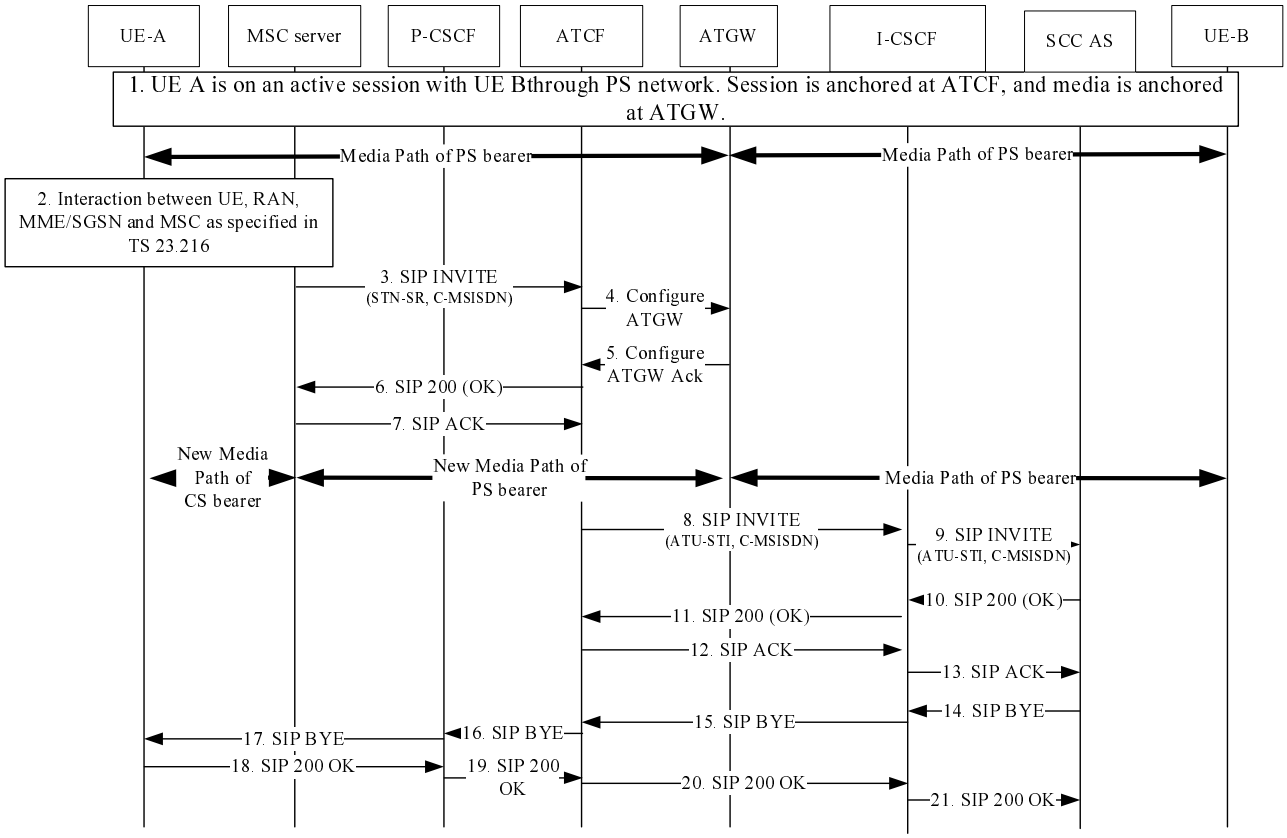


Figure A.18.3-1 Signalling flows for PS to CS access transfer: PS to CS SRVCC enhancements using ATCF and media anchored

NOTE 1: For clarity, the SIP 100 (Trying) responses are not shown in the signalling flow.

1. **UE A is on an active session with UE B**

   There is an ongoing IP bearer between the UE A and the remote end UE B. The media is anchored at ATGW.

2. **SC UE A attaches to the CS domain**

   UE A sends the measurement reports to E-UTRAN, and the source E-UTRAN decides to trigger an PS to CS SRVCC handover to CS access. The MSC server initiates the session transfer with the STN-SR, refer to 3GPP TS 23.216 [49].

3. **SIP INVITE request (MSC server to ATCF)-see example in table A.18.3-3**

Table A.18.3-3: SIP INVITE request (MSC server to ATCF)

```
INVITE tel: +1-237-555-3333 SIP/2.0
Via: SIP/2.0/UDP msc1.visit1.net;branch=z9hG4bk731b87
Max-Forwards: 70
P-Asserted-Identity: <tel:+1-237-555-2222>
P-Charging-Vector: icid-value="AyretyU0dm+6O2IrT5tAFrbHLso=023551024";orig-ioi=visit1.net
Privacy: none
From: <tel:+1-237-555-1111>;tag=171828
To: <tel: +1-237-555-3333>
Call-ID: cb03a0s09a2sdfglkj490334
Cseq: 127 INVITE
Supported: 100rel, precondition, gruu
Accept-Contact: *;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
P-Asserted-Service: urn:urn-7:3gpp-service.ims.icsi.mmtel
Contact: <sip: msc1.visit1.net:1357>;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:eee
```

```
s=
c=IN IP6 5555::aaa:bbb:ccc:eee
t=0 0
m=audio 3456 RTP/AVP 97 96
a=tcap:1 RTP/AVPF
a=pcfg:1 t=1
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:96 telephone-event
a=maxptime:20
```

Request-URI: contains the STN-SR, as routed to the ATCF.

SDP: The SDP contains preconfigured set of codecs supported by the MGW.

P-Asserted-Identity: the C-MSISDN of the served UE.

4. **Configure ATGW (ATCF to ATGW)**

   Upon receiving the access transfer message, the ATCF correlates the transferred session using C-MSISDN. The ATCF updates the ATGW by replacing the existing PS access leg media path information with the new CS access leg media path information, by sending a Configure ATGW message to ATGW.

5. **Configure ATGW ACK (ATGW to ATCF)**

   The ATGW sends Configure ATGW Acknowledgment message back to ATCF.

6. **SIP 200 (OK) response (ATCF to MSC server)**

   The ATCF sends the SIP 200 (OK) response to the MSC server with the media information allocated by the ATGW during session establish procedure. In the SIP 200 (OK) response, the ATCF includes the Record-Route header field containing its SIP URI that indicate where the ATCF expect to receive the indialog request sent by the MSC. In the Contact header field, the ATCF inserts the saved URI of the UE B that the UE A received from the UE B when the IP bearer between the UE A and the UE B was established.

7. **SIP ACK request (MSC server to ATCF)**

8. **SIP INVITE request (ATCF to I-CSCFs)-see example in table A.18.3-8**

   After receiving the access transfer message, the ATCF establishes a new dialog with the SCC AS by sending a new SIP INVITE request to the SCC AS using the stored ATU-STI for PS to CS SRVCC. When resolving the ATU-STI for PS to CS SRVCC (e.g. via DNS access), the ATCF obtains the IP address of the I-CSCF. The ATCF updates the SCC AS via the new dialog indicating that the transfer has taken place. As there is no update in the SDP information, no remote end update will be performed.

Table A.18.3-8: SIP INVITE request (ATCF to I-CSCF)

```
INVITE sip:AUT-STI1@sccas.home1.net SIP/2.0
Via: SIP/2.0/UDP atcf.visited2.net:5060;branch=z9hG4bk731b87
Max-Forwards: 70
P-Asserted-Identity: <tel:+1-237-555-2222>
P-Charging-Vector: icid-value="AyretyU0dm+6O2IrT5tAFrbHLso=023551024";orig-ioi=visit1.net
Privacy: none
From: <tel:+1-237-555-3333>;tag=1888828
To: <tel: +1-237-555-4444>
Call-ID: cb03a0s09a2sdfglkj490444
Cseq: 127 INVITE
Supported: 100rel, precondition, gruu
Require: tdialog,
Record-Route:<sip:atcf.visited2.net:5060;lr>
Target-Dialog: me03a0s09a2sdfgjkl491777; remote-tag=774321; local-tag=64727891
Accept-Contact: *;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
P-Asserted-Service: urn:urn-7:3gpp-service.ims.icsi.mmtel
Contact: <sip: msc1.visit1.net:1357>;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER
```

```
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ggg
s=
c=IN IP6 5555::aaa:bbb:ccc:ggg
t=0 0
m=audio 3456 RTP/AVP 97 96
a=tcap:1 RTP/AVPF
a=pcfg:1 t=1
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:96 telephone-event
a=maxptime:20
```

Request-URI: contains the ATU-STI for PS to CS SRVCC, that resolves (e.g. via DNS access) to the IP address of the I-CSCF.

Target-Dialog: specifies that the existing dialog is related with this request.

Record-Route: contains the SIP URI of the ATCF, where the ATCF expect to receive the in-dialog request from the SCC AS.

Require: the "tdialog" option tag indicate that the support for Target-Dialog header field is required.

P-Asserted-Identity: the C-MSISDN of the served UE.

SDP: the media information at ATGW.

9. SIP INVITE request (I-CSCF to SCC AS)

The I-CSCF forwards the SIP INVITE request to the SCC AS.

10. SIP 200 (OK) response (SCC AS to I-CSCF)

Since there is no update in the session description, no remote end update will be performed. The SCC AS sends confirmation response to the ATCF which contain the SDP answer that the SCC AS stored during the original session establishment procedure. The SIP 200 (OK) response also includes the Record-Route header field(s) that was constructed by the SCC AS adding its SIP URI to the Record-Route header field(s) that was received in the initial SIP INVITE request in step 9. The SIP URI of the SCC AS specifies where the SCC AS expects to receive the in-dialog request from the ATCF.

11. SIP 200 (OK) response (I-CSCF to ATCF)

12-13. SIP ACK request (ATCF to SCC AS via I-CSCF)

14-17. SIP BYE request (SCC AS to UE A via I-CSCF, ATCF and P-CSCF)

The SCC AS terminates the source access leg, which was using the old IP-CAN, by sending a SIP BYE request to the UE A.

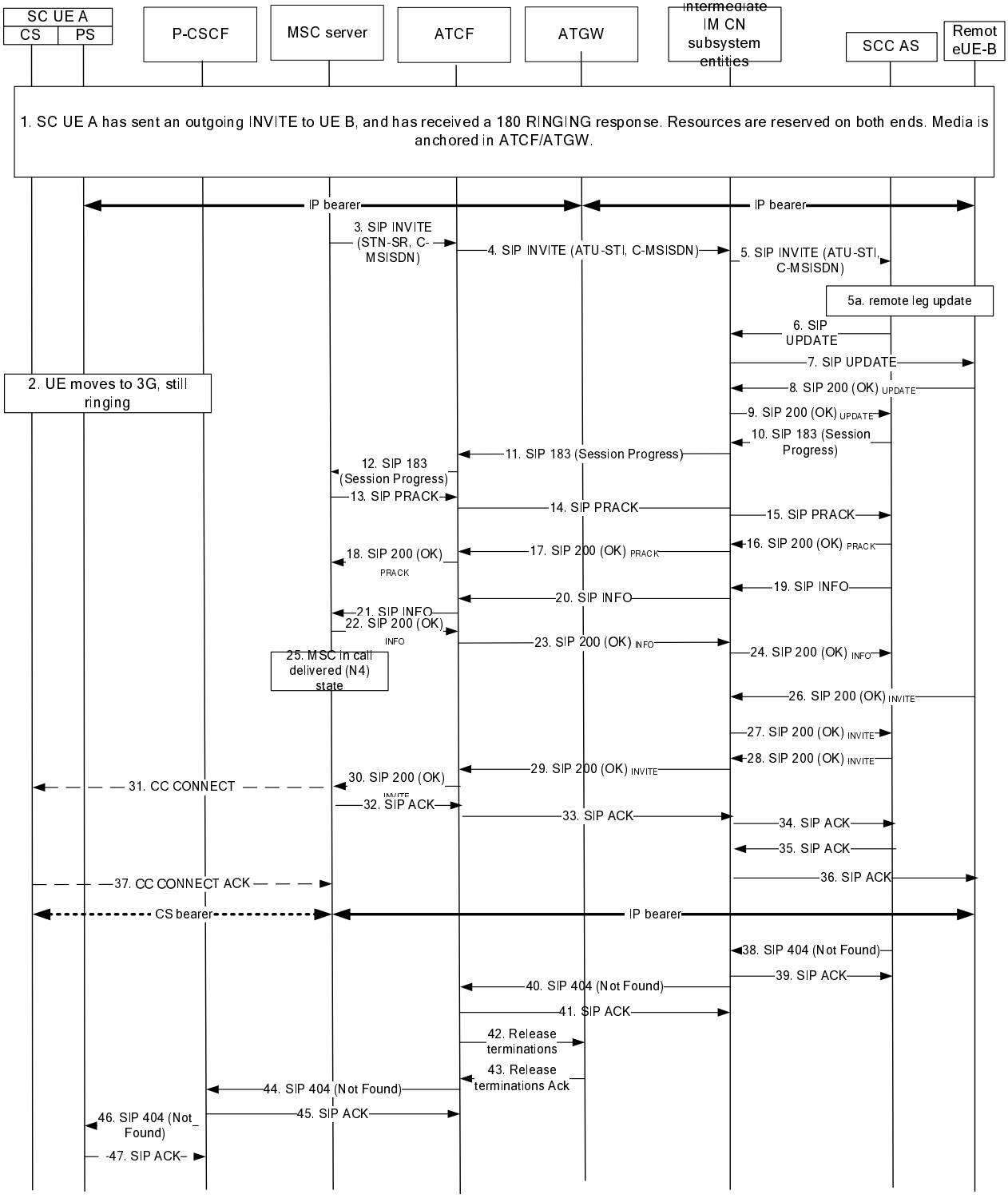
18-21. SIP 200 (OK) response (UE A to SCC AS via P-CSCF, ATCF and I-CSCF)

Upon receiving the SIP BYE request, the UE A sends a SIP 200 (OK) response to the SCC AS. Subsequently, the UE A relinquishes all resources pertaining to the old IP-CAN.

NOTE: Steps 17-18 are performed only if UE A uses Gm after the PS-CS access transfer is completed; otherwise, the UE A and the network release the source access leg locally, without any signalling between the UE A and the network

A.18.4 Session transfer for originating call is in alerting phase using PS to CS SRVCC procedure with ATCF: PS to CS

In the example flow at the figure A.18.4-1, SC UE A has invited for an originating session with speech media component which is anchored at ATCF. The session is in alerting phase. Specifically, this flow illustrates that after successful transfer procedure, a direct media path is established between MSC server and the remote UE and thus any resources in the ATCF and ATGW allocated when the original session was initiated are released.



NOTE: For clarity, the SIP 100 (Trying) responses are not shown in the signalling flow.

Figure A.18.4-1: PS-CS SRVCC, originating call in alerting phase

1. **SC UE A has setup an outgoing call**

   The media of the outgoing call has been anchored at the ATGW. Both ends have reserved the resources and SC UE A has received a SIP 180 (Ringing) response.

2. **SC UE A attaches to the CS domain**

   UE A sends the measurement reports to E-UTRAN, and the source E-UTRAN decides to trigger an PS to CS SRVCC handover to CS access. The MSC server initiates the session transfer with the STN-SR, refer to 3GPP TS 23.237 [9]. The UE continues ringing.

3. **SIP INVITE request (MSC server to ATCF) - see example in table A.18.4-3**

   The MSC server sends an initial SIP INVITE request transferring the session with the recived STN-SR.

Table A.18.4-3: SIP INVITE request (MSC server to ATCF)

```
INVITE tel: +1-237-555-3333 SIP/2.0
Via: SIP/2.0/UDP msc1.visit1.net;branch=z9hG4bk731b87
Max-Forwards: 70
P-Asserted-Identity: <tel:+1-237-555-1111>
P-Charging-Vector: icid-value="AyretyU0dm+6O2IrT5tAFrbHLso=023551024";orig-ioi=visit1.net
Privacy: none
From: <tel:+1-237-555-1111>;tag=171828
To: <tel:+1-237-555-3333>
Call-ID: cb03a0s09a2sdfglkj490334
Cseq: 127 INVITE
Supported: 100rel, precondition, gruu
Accept-Contact: *;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
P-Asserted-Service: urn:urn-7:3gpp-service.ims.icsi.mmtel
Contact: <sip: msc1.visit1.net:1357>;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-
 service.ims.icsi.mmtel"; g.3gpp.srvcc-alerting
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER
Recv-Info: g.3gpp.state-and-event
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:eee
s=
c=IN IP6 5555::aaa:bbb:ccc:eee
t=0 0
m=audio 3456 RTP/AVP 97 96
a=tcap:1 RTP/AVPF
a=pcfg:1 t=1
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:96 telephone-event
a=maxptime:20
```

Request-URI: contains the STN-SR.

SDP: The SDP contains set of codecs supported by the MGW.

4. **SIP INVITE request (ATCF to intermediate IM CN subsystem entities) - see example in table A.18.4-4**

   The ATCF sends the initial SIP INVITE request replacing the STN-SR with an ATU-STI for PS to CS SRVCC associated with a session in the transferable session set to the intermediate IM CN subsystem entities.

Table A.18.4-4: SIP INVITE request (ATCF to intermediate IM CN subsystem entities)

```
INVITE sip:sccas1-atu-sti.home1.net SIP/2.0
Via: SIP/2.0/UDP atcf.visited1.net:5060;branch=z9hG4bKnas56565, SIP/2.0/UDP
msc1.visit1.net;branch=z9hG4bk731b87
Max-Forwards: 69
Route: <sip:icscf1.visit1.net;lr>
P-Asserted-Identity: <tel:+1-237-555-1111>
```

```
P-Charging-Vector: icid-value="AyretyU0dm+6O2IrT5tAFrbHLso=023551024";orig-ioi=visit1.net
Privacy: none
From: <tel:+1-237-555-1111>;tag=171828
To: <tel:+1-237-555-3333>
Call-ID: cb03a0s09a2sdfglkj490334
Cseq: 127 INVITE
Supported: 100rel, precondition, gruu
Accept-Contact: *;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
P-Asserted-Service: urn:urn-7:3gpp-service.ims.icsi.mmtel
Contact: <sip: msc1.visit1.net:1357>;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-
 service.ims.icsi.mmtel"; g.3gpp.srvcc-alerting
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER
Recv-Info: g.3gpp.state-and-event
Record-Route: <atcf.visited1.net;lr>
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:eee
s=
c=IN IP6 5555::aaa:bbb:ccc:eee
t=0 0
m=audio 3456 RTP/AVP 97 96
a=tcap:1 RTP/AVPF
a=pcfg:1 t=1
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:96 telephone-event
a=maxptime:20
```

Request-URI: contains the ATU-STI for PS to CS SRVCC.

Record-Route: contains ATCF URI.

SDP: The SDP contains set of codecs supported by the MGW.

5. SIP INVITE request (intermediate IM CN subsystem entities to SCC AS)

The SIP INVITE is routed towards the SCC AS, based on filter criteria in S-CSCF.

5a. Remote Leg Update

The SCC AS correlates the initial SIP INVITE request to the local and remote call legs of the existing session between the UE A and the remote end. The SCC AS performs the Remote Leg update by sending SIP UPDATE request towards the Remote Leg.

6. SIP UPDATE request (SCC AS to intermediate IM CN subsystem entities)

The SCC AS acting as a B2BUA generates a SIP UPDATE request based upon the received initial SIP INVITE request and the information previously stored against this session.

7. SIP UPDATE request (Intermediate IM CN subsystem entities to remote UE B)

The intermediate IM CN subsystem entities forward the SIP UPDATE request to remote UE B.

8. SIP 200 (OK) response (Remote UE B to Intermediate IM CN subsystem entities)

Upon receiving the SIP UPDATE request containing the SDP offer for the leg to the MSC, the remote UE B sends a SIP 200 (OK) response.

9. SIP 200 (OK) response (Intermediate IM CN subsystem entities to SCC AS)

The intermediate IM CN subsystem entities forward the SIP 200 (OK) response to the SCC AS.

10. SIP 183 (Session Progress) response (SCC AS to intermediate IM CN subsystem entities)

The SCC AS sends a 183 (Session Progress) containing the SDP answer as received from the remote UE B. The SDP answer indicates that resources are available

11. SIP 183 (Session Progress) response (Intermediate IM CN subsystem entities to ATCF)

The intermediate IM CN subsystem entities forward the 183 (Session Progress) response to the ATCF.

12. SIP 183 (Session Progress) response (ATCF to MSC server)

The ATCF forwards the 183 (Session Progress) response to the MSC server.

13. SIP PRACK request (MSC server to ATCF)

The MSC acknowledges the receipt of the 183 (Session Progress) response.

14. SIP PRACK request (ATCF to intermediate IM CN subsystem)

The ATCF forwards the SIP PRACK request to intermediate IM CN subsystem entities.

15. SIP PRACK request (Intermediate IM CN subsystem entities to SCC AS)

The intermediate IM CN subsystem entities forward the SIP PRACK request to the SCC AS.

16. SIP 200 (OK) response (SCC AS to Intermediate IM CN subsystem entities)

The SCC AS acknowledges the SIP PRACK request.

17. SIP 200 (OK) response (Intermediate IM CN subsystem entities to ATCF)

The intermediate IM CN subsystem entities forward the SIP 200 (OK) response to the ATCF.

18. SIP 200 (OK) response (ATCF to MSC server)

The ATCF forwards the SIP 200 (OK) response to the MSC server.

19. SIP INFO request (SCC AS to intermediate IM CN subsystem entities) - see example in table A.18.4-19

The SCC AS sends a SIP INFO request that indicates that the call is an early dialog and that the SC UE was the initiator.

Table A.18.4-19: INFO request (SCC AS to intermediate IM CN subsystem entities)

```
INFO sip: msc1.visit1.net:1357 SIP/2.0
Via SIP/2.0/UDP sip:sccas1.home1.net;branch=z9hG4bK332b23.1
Max-Forwards: 70
Route: <sip:scscf1.home1.net;lr> <atcf.visited1.net;lr>
From: <tel: +1-237-555-3333>;tag=314159
To: <tel:+1-237-555-1111>;tag=171828
Call-ID: cb03a0s09a2sdfglkj490334
Cseq: 129 INFO
Info-Package: g.3gpp.state-and-event
Content-Disposition: Info-Package
Content-Type: application/vnd.3gpp.state-and-event-info+xml
Content-Length:

<?xml version="1.0" encoding="UTF-8"?>
<state-and-event-info>
   <state-info>early</state-info>
   <direction>initiator</direction>
</state-and-event-info>
```

20. SIP INFO request (Intermediate IM CN subsystem entities to ATCF)

The intermediate IM CN subsystem entities forward the SIP INFO request to the ATCF.

21. SIP INFO request (ATCF to MSC server)

The intermediate IM CN subsystem entities forward the SIP INFO request to the ATCF. The MSC server is now aware that the call that is transferred is in originating alerting phase.

22. SIP 200 (OK) response (MSC server to ATCF)

The ATCF forwards the SIP 200 (OK) response to intermediate IM CN subsystem entities.

23. SIP 200 (OK) response (ATCF to intermediate IM CN subsystem entities)

The ATCF forwards the SIP 200 (OK) response to the intermediate IM CN subsystem entities.

24. SIP 200 (OK) response (Intermediate IM CN subsystem entities to SCC AS)

The intermediate IM CN subsystem entities forward the SIP 200 (OK) response to the SCC AS.

25. MSC goes in Call delivered state

The MSC enters Call delivered (N4) state as defined in 3GPP TS 24.008 [8] due to the information received in the SIP INFO request.

26. SIP 200 (OK) response (Remote UE B to intermediate IM CN subsystem entities)

The remote UE B accepts the call and sends a SIP 200 (OK) response.

27. SIP 200 (OK) response (Intermediate IM CN subsystem entities to SCC AS)

The intermediate IM CN subsystem entities forward the SIP 200 (OK) response to SCC AS.

28. SIP 200 (OK) response (SCC AS to intermediate IM CN subsystem entities)

The SCC AS sends the SIP 200 (OK) response to indicate that the remote UE B has accepted the call.

29. SIP 200 (OK) response (Intermediate IM CN subsystem entities to ATCF)

The SIP 200 (OK) response is forwarded to the ATCF.

30. SIP 200 (OK) response (ATCF to MSC server)

The SIP 200 (OK) response is forwarded to the ATCF.

31. CC CONNECT (MSC server to SC UE A)

The MSC server indicates to the SC UA A that the remote UE B has accepted the call in accordance with 3GPP TS 24.008 [8].

32. SIP ACK request (MSC server to ATCF)

The MSC server acknowledges the SIP 200 (OK) response received from SCC AS

33. SIP ACK request (ATCF to intermediate IM CN subsystem entities)

ATCF forwards the SIP ACK request to the intermediate IM CN subsystem entities.

34. SIP ACK request (Intermediate IM CN subsystem entities to SCC AS)

The intermediate IM CN subsystem entities forward the SIP ACK request to the SCC AS. The SCC AS starts a operator specific timer supervising the release of the original source leg.

35. SIP ACK request (SCC AS to intermediate IM CN subsystem entities)

The SCC AS acknowledges the SIP 200 (OK) response received towards the remote UE B.

36. SIP ACK request (Intermediate IM CN subsystem entities to remote UE B)

The SIP ACK request is forwarded towards the remote UE B.

37. CC CONNECT ACK (MSC server to SC UE A)

SC UE A acknowledges the CC CONNECT in accordance with 3GPP TS 24.008 [8].

38. SIP 404 (Not Found) response (SCC AS to intermediate IM CN subsystem entities)

The SCC AS releases the original source leg towards the SC UE A after the operator specific timer has expired by means of a SIP 404 (Not Found) response.

39. SIP ACK (Intermediate IM CN subsystem entities to SCC AS)

The SIP ACK request is sent to SCC AS.

40. SIP 404 (Not Found) response (Intermediate IM CN subsystem entities to ATCF)

Intermediate IM CN subsystem entities send a SIP 404 (Not Found) response in order to release to original source dialog towards the SC UE A.

41. SIP ACK (ATCF to intermediate IM CN subsystem entities)

The SIP ACK request is sent to the intermediate IM CN subsystem entities.

42-43.Media recources reserved in ATGW is released by ATCF.

The ATCF orders the ATGW to release all media terminations (including termination created due to forking on remote end) of the used for media anchoring during call setup in. The ATGW acknowledges the release.

44-47.SIP 404 (Not Found) response (ATCF towards SC UE A)

The ATCF sends a SIP 404 (Not Found) response in order to release to original source dialog towards the SC UE A via P-CSCF.

NOTE : The SC UE A can only receive the SIP 404 (Not Found) response and send the SIP ACK request if the signalling bearer is not suspended.

A.18.5 Signalling flows for PS to CS Access Transfer: SRVCC enhancements using ATCF with MSC server assisted mid-call feature and ATCF anchored

The flow shown in Figure A.18.5-1 applies if the CS to PS SRVCC is supported when the PS to CS SRVCC access transfer takes place.

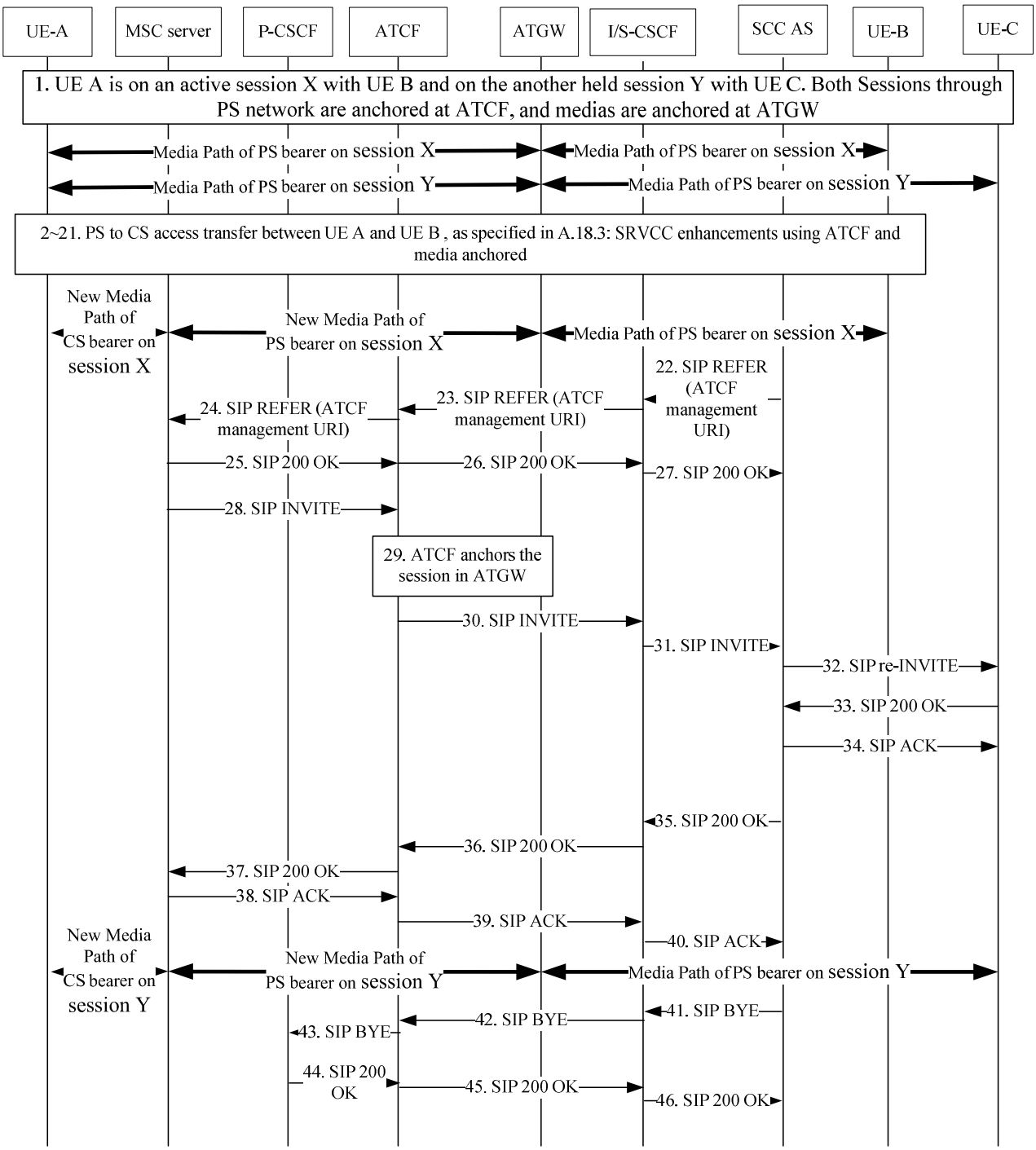


Figure A.18.5-1: Signalling flows for PS to CS Access Transfer: SRVCC enhancements using ATCF with MSC server assisted mid-call feature and ATCF anchored

1. UE A is on an active session X with UE B and a held session Y with UE C

UE A is on an active session X with UE B and on another held session Y with UE C. Both Sessions through PS network are anchored at ATCF, and medias are anchored at ATGW.

2-21. PS to CS access transfer between UE A and UE B

The PS to CS access transfer between UE A and UE B is specified in subclause A.18.3: SRVCC enhancements using ATCF and media anchored.

22. SIP REFER request (SCC AS to I/S-CSCF) - see example in table A.18.5-22

The SCC AS sends a SIP REFER request to the I/S-CSCF inside the dialog created by the message 10.

Table A.18.5-22: SIP REFER request (SCC AS to I/S-CSCF)

```
REFER sip:user1_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91e6bf6 SIP/2.0
Via: SIP/2.0/UDP sip:sccas1.home1.net;branch=z9hG4bk731b8a
Max-Forwards: 70
P-Charging-Vector: icid-value="AyretyU0dm+6O2IrT5tAFrbHLso=023551024"; orig-ioi=home1.net
To: <tel:+1-237-555-1111>;tag=171828
From: <tel:+1-237-555-3333>;tag=sdfsdf
Call-ID:  cb03a0s09a2sdfglkj490444
Cseq: 55998 REFER
Content-Length: 125
Route: <sip:scscf1.home1.net;lr>
Refer-Sub: false
Supported: norefersub, gruu
Contact: sip:sccas1.home1.net
Refer-To: <sip:additional.session.xfer.pscssrvcc@sccas.home1.net?Target-
 Dialog=ksdjfhwrklf%3Bremote-tag=676723565%3Blocal-tag=45418454&Require=tdialog&From=tel:+1-237-
 555-1111&To=tel:+1-987-654-3210&Content-Type=application%2Fsdp&body=v%3D0%0D%0Ao%3D-
 %202987933623%202987933623%20IN%20IP6%205555::ggg:fff:aaa:bbb%0D%0As%3D-
 %0D%0Ac%3DIN%20IP6%205555::ggg:fff:aaa:bbb%0D%0At%3D0%200%0D%0Am%3Dvideo%200%20RTP%2FAVP%2098%0D
 %0Am%3Daudio%203456%20RTP%2FAVP%2097%2096%0D%0Ab%3DAS:25.4%0D%0Aa%3Drtpmap:97%20AMR%0D%0Aa%3Dfmt
 p:97%20mode-set%3D0%2C2%2C5%2C7%3B%20mode-change-period%3D2%0D%0Aa%3Dmaxptime:20%0D%0A>
Content-Type: multipart/mixed;boundary="boundary"

--boundary1
Content-Type: application/vnd.3gpp.mid-call+xml

<?xml version="1.0" encoding="UTF-8"?>
<mid-call/>
--boundary1
Content-Type: application/vnd.3gpp.srvcc-ext+xml

<?xml version="1.0"?>
<srvcc-ext>
  <PS-reg-info>
    <ATCF-Management-URI>sip:atcf.visited2.net</ATCF-Management-URI>
    <C-MSISDN>tel:+1-237-555-1111</C-MSISDN>
  </PS-reg-info>
</srvcc-ext>
--boundary1--
```

Refer-To: contains the additional transferred session SCC AS URI for PS to CS SRVCC and the following URI header fields:

Target-Dialog: the dialog identifier of the source access leg.

Require: containing "tdialog" option tag

From: contains the public user identity of the UE A

To: contains the public user identity of the UE C

Content-Type: containing "application/sdp" MIME type of the "body" URI header field

body: SDP describing the media used in the session

application/vnd.3gpp.mid-call+xml MIME body: indicates that REFER is related to MSC server assisted mid-call feature.

application/vnd.3gpp.srvcc-ext+xml MIME body: provides ATCF management URI and C-MSISDN.

23. SIP REFER request (I/S-CSCF to ATCF)

The I/S-CSCF forwards the SIP REFER request to the ATCF.

24. SIP REFER request (ATCF to MSC server) - see example in table A.18.5-24

The ATCF forwards the SIP REFER within the dialog.

Table A.18.5-24: SIP REFER request (ATCF to MSC server)

```
REFER sip:user1_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91e6bf6 SIP/2.0
Via: SIP/2.0/UDP sip:sccas1.home1.net;branch=z9hG4bk731b8a
Via: SIP/2.0/UDP sip:scscf1.home1.net;branch=z9hG4bk869d11e
Via: SIP/2.0/UDP sip:atcf1.home1.net;branch=z9hG4bk9251re3
Max-Forwards: 70
P-Charging-Vector:
To:
From:
Call-ID:
Cseq:
Content-Length:
Route: <sip:mscserver1.home1.net;lr>
Refer-Sub:
Supported:
Contact:
Refer-To: <sip:additional.session.xfer@sccas.home1.net?Target-Dialog=ksdjfhwrklf%3Bremote-
 tag=676723565%3Blocal-tag=45418454&Require=tdialog&From=tel:+1-237-555-1111&To=tel:+1-987-654-
 3210&Content-Type=application%2Fsdp&body=v%3D0%0D%0Ao%3D-
 %202987933623%202987933623%20IN%20IP6%205555::ggg:fff:aaa:bbb%0D%0As%3D-
 %0D%0Ac%3DIN%20IP6%205555::ggg:fff:aaa:bbb%0D%0At%3D0%200%0D%0Am%3Dvideo%200%20RTP%2FAVP%2098%0D
 %0Am%3Daudio%203456%20RTP%2FAVP%2097%2096%0D%0Ab%3DAS:25.4%0D%0Aa%3Drtpmap:97%20AMR%0D%0Aa%3Dfmt
 p:97%20mode-set%3D0%2C2%2C5%2C7%3B%20mode-change-period%3D2%0D%0Aa%3Dmaxptime:20%0D%0A>
Content-Type: multipart/mixed;boundary="boundary1"

--boundary1
Content-Type: application/vnd.3gpp.mid-call+xml

<?xml version="1.0" encoding="UTF-8"?>
<mid-call/>
--boundary1
Content-Type: application/vnd.3gpp.srvcc-ext+xml

<?xml version="1.0"?>
<srvcc-ext>
  <PS-reg-info>
    <ATCF-Management-URI>sip:atcf2.visited2.net</ATCF-Management-URI>
    <C-MSISDN>tel:+1-237-555-1111</C-MSISDN>
  </PS-reg-info>
</srvcc-ext>
--boundary1--
```

25-27. SIP 200 OK response

Upon receiving the SIP REFER request, the MSC Server sends a SIP 200 (OK) response to ATCF, ATCF forwards it to the SCC AS.

28. SIP INVITE request (MSC Server to ATCF) -see example in table A.18.5-28

Upon receiving the SIP REFER request the MSC Server sends a SIP INVITE request to the ATCF according to the Refer-To header field in the SIP REFER request. MSC server also includes Route header field with the ATCF management URI received in the application/vnd.3gpp.srvcc-ext+xml MIME body of the SIP REFER request.

Table A.18.5-28: SIP INVITE request (MSC Server to ATCF)

```
INVITE sip:additional.session.xfer@sccas.home1.net SIP/2.0
Via: SIP/2.0/UDP msc1.home1.net;branch=z9hG4bk731b87
Max-Forwards: 70
P-Asserted-Identity: <tel:+1-237-555-1111>
P-Charging-Vector: icid-value="AyretyU0dm+6O2IrT5tAFrbHLso=023551024"; orig-ioi=home1.net
Privacy: none
From: <tel:+1-237-555-1111>;tag=171828
To: <tel:+1-987-654-3210>
Call-ID: asdfgqwerq
Cseq: 1275 INVITE
Supported: 100rel, precondition, 199, gruu
Accept-Contact: *;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
P-Asserted-Service: urn:urn-7:3gpp-service.ims.icsi.mmtel
Contact: <sip: msc1.visit1.net:1357>;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE
Target-Dialog: ksdjfhwrklf;remote-tag=676723565;local-tag=45418454
```

```
Require: tdialog
Content-Length: (…)
Content-Type: multipart/mixed;boundary="boundary1"
Route: <sip:atcf2.visited2.net;lr>


--boundary1
Content-Type: application/sdp
v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:eee
s=
c=IN IP6 5555::aaa:bbb:ccc:eee
t=0 0
m=video 0 RTP/AVP 98
m=audio 3456 RTP/AVP 97 96
a=tcap:1 RTP/AVPF
a=pcfg:1 t=1
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:96 telephone-event
a=maxptime:20
a=sendonly
--boundary1
Content-Type: application/vnd.3gpp.srvcc-ext+xml

<?xml version="1.0"?>
<srvcc-ext>
<Setup-info>
    <C-MSISDN>tel:+1-212-555-1111</C-MSISDN>
    <direction>initiator</direction>
</Setup-info>
</srvcc-ext>
--boundary1--
```

Request-URI: contains the additional transferred session SCC AS URI for PS to CS SRVCC as received in the Refer-To header field in the SIP REFER request.

Route: contains the ATCF URI as received in the application/vnd.3gpp.srvcc-ext+xml MIME body in the SIP REFER request and with "lr" parameter.

P-Asserted-Identity: the C-MSISDN of the served UE.

application/vnd.3gpp.srvcc-ext+xml: Contains the direction of call and the C-MSISDN of the UE.

29. ATCF configures the ATGW

Upon receiving the SIP INVITE request to it, the ATCF decides to anchor the ATGW, and configures the ATGW. Then the ATGW return the ACK to complete the configuration.

30-31. SIP INVITE request (ATCF to SCC AS via I/S-CSCF) -see example in table A.18.5-33

The ATCF sends the SIP INVITE request to the I/S-CSCF. The I/S-CSCF forwards the SIP INVITE request to the SCC AS.

NOTE: ATCF uses the same procedure as in subclause A.4.3.

Table A.18.5-33: SIP INVITE request (ATCF to SCC AS via I/S-CSCF)

```
INVITE sip:additional.session.xfer.pscssrvcc@sccas.home1.net SIP/2.0
Via: SIP/2.0/UDP atcf1.home1.net;branch=z9hG4bk731b87
Max-Forwards: 70
P-Asserted-Identity:
P-Charging-Vector:
Privacy: none
From:
To:
Call-ID: asdfgqwerq2
Cseq:
```

```
Supported:
Accept-Contact:
P-Asserted-Service:
Contact:
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE
Content-Type:
Target-Dialog:
Require: tdialog
Content-Length: (…)

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

32. SIP re-INVITE request (SCC AS towards UE C)

33. SIP 200 (OK) response to the SIP re-INVITE request (UE C towards SCC AS)

32. SIP ACK request (SCC AS towards UE C)

35-36. SIP 200 (OK) response (SCC AS to ATCF via I/S-CSCF)

The SCC AS sends the SIP 200 (OK) response to the SIP INVITE to the I/S-CSCF, and the I/S-CSCF forwards it to the ATCF.

37. SIP 200 (OK) response to the SIP INVITE request (ATCF to MSC server)

38-40. SIP ACK request (MSC server to SCC AS via I/S-CSCF)

The MSC server generates the SIP ACK request to the SIP 200 (OK) response, and sends the SIP ACK request to the I/S-CSCF. Then the I/S-CSCF forwards it to the SCC AS.

41-43. SIP BYE request (SCC AS towards SC UE A via I/S-CSCF, ATCF and P-CSCF)

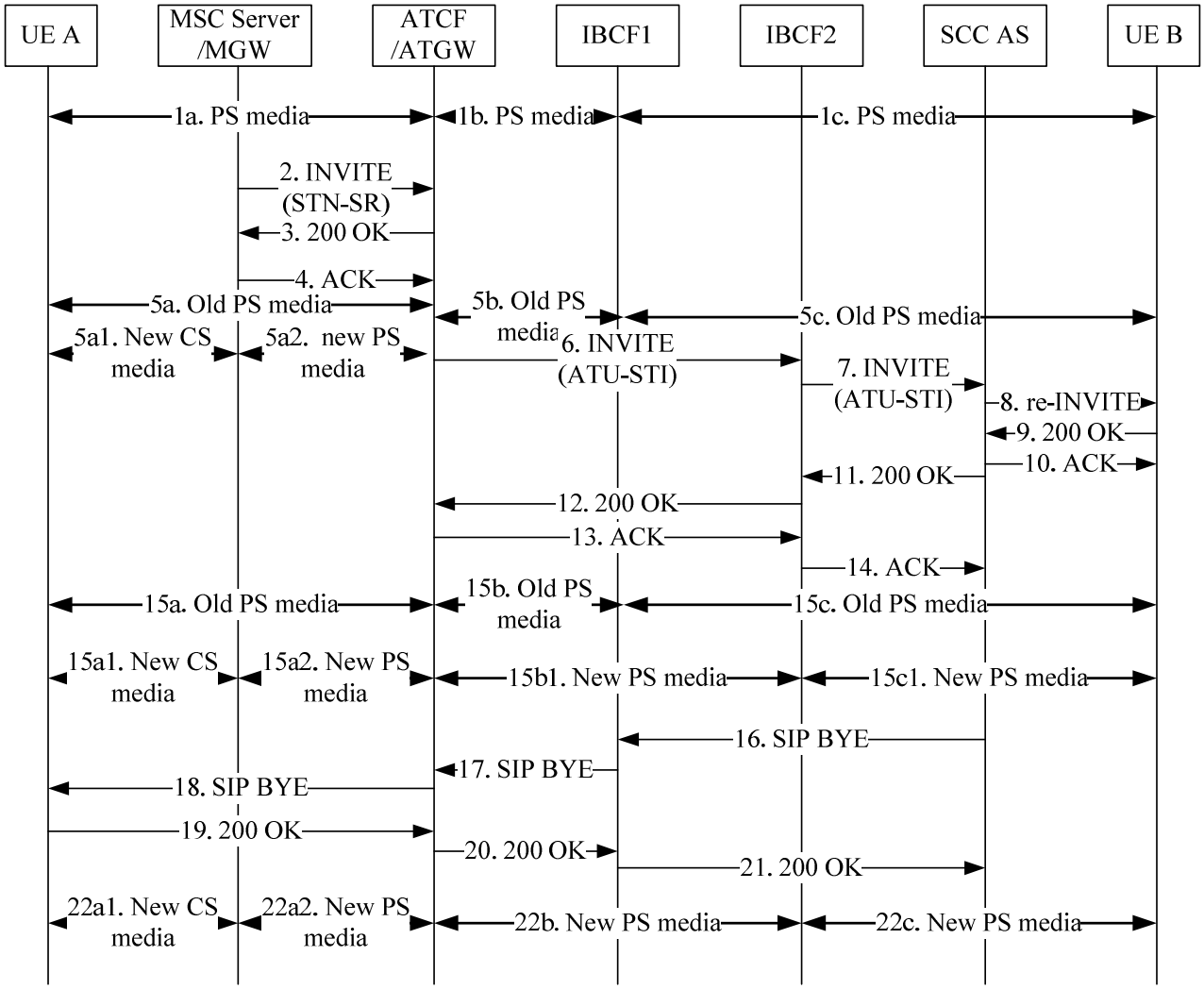
The SCC AS terminates the replaced call leg of the session Y, which was using the old IP-CAN, by sending a SIP BYE request towards the UE A which received by P-CSCF.

44-46. SIP 200 (OK) response (P-CSCF to SCC AS via ATCF and I/S-CSCF)

Upon receiving the SIP BYE request over the old IP-CAN, the P-CSCF sends a SIP 200 (OK) response over the old IP-CAN to the SCC AS.

A.18.6 Signalling flows for PS to CS Access Transfer: PS to CS SRVCC enhancements using ATCF and session traverses IBCF

The signalling flow shown in figure A.18.6-1 gives an example for PS to CS access transfer when using PS to CS SRVCC. The call is established, contains active speech media component and has been anchored in ATGW, and traverses IBCF during the establishment of the call. There are IBCFs between ATCF and SCC AS, e.g. UE A is roaming in another network. When PS to CS SRVCC enhancements using ATCF is triggered, the session trasfer notification message initiated by ATCF using ATU-STI may traverse different IBCF(s) comparing to previous signaling path duing the initial session set up between ATCF and SCC AS.



NOTE: For clarity, the SIP 100 (Trying) responses are not shown in the signalling flow.

Figure A.18.6-1 Signalling flows for CS to PS Access Transfer: CS to PS SRVCC occurs during a call.

1. **The UE A has a session with active speech media component with UE B**

   UE A has an active session with remote UE B, media is anchored in ATGW and the session traverses IBCF1.

2. **SIP INVITE request (MSC server to ATCF)-see example in table A.18.5-2**

Table A.18.6-2: SIP INVITE request (MSC server to ATCF)

```
INVITE tel: +1-237-555-3333 SIP/2.0
Via: SIP/2.0/UDP msc1.visit1.net;branch=z9hG4bk731b87
Max-Forwards: 70
P-Asserted-Identity: <tel:+1-237-555-2222>
P-Charging-Vector: icid-value="AyretyU0dm+6O2IrT5tAFrbHLso=023551024";orig-ioi=visit1.net
Privacy: none
From: <tel:+1-237-555-1111>;tag=171828
To: <tel: +1-237-555-3333>
Call-ID: cb03a0s09a2sdfglkj490334
Cseq: 127 INVITE
Supported: 100rel, precondition, gruu
Accept-Contact: *;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
P-Asserted-Service: urn:urn-7:3gpp-service.ims.icsi.mmtel
Contact: <sip: msc1.visit1.net:1357>;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:eee
```

```
s=
c=IN IP6 5555::aaa:bbb:ccc:eee
t=0 0
m=audio 3456 RTP/AVP 97 96
a=tcap:1 RTP/AVPF
a=pcfg:1 t=1
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:96 telephone-event
a=maxptime:20
```

Request-URI: contains the STN-SR, as routed to the ATCF.

SDP: The SDP contains preconfigured set of codecs supported by the MGW.

P-Asserted-Identity: the C-MSISDN of the served UE.

3. SIP 200 (OK) response (ATCF to MSC server)

The ATCF sends the SIP 200 (OK) response to the MSC server with the media information allocated by the ATGW during session establish procedure. In the SIP 200 (OK) response, the ATCF includes the Record-Route header field containing its SIP URI that indicate where the ATCF expect to receive the indialog request sent by the MSC. In the Contact header field, the ATCF inserts the saved URI of the UE B that the UE A received from the UE B when the IP bearer between the UE A and the UE B was established.

4. SIP ACK request (MSC server to ATCF)

5. The new CS media between UA and MSC Server/MGW is established, and the PS media between MSC Server/MGW and ATCF/ATGW is established.

6-7. SIP INVITE request (ATCF to SCC AS)-see example in table A.18.6-6

After receiving the access transfer message, the ATCF establishes a new dialog with the SCC AS by sending a new SIP INVITE request to the SCC AS using the stored ATU-STI. And the new dialog traverses IBCF2. The ATCF updates the SCC AS via the new dialog indicating that the transfer has taken place.

Table A.18.6-6: SIP INVITE request (ATCF to SCC AS)

```
INVITE sip:AUT-STI1@sccas.home1.net SIP/2.0
Via: SIP/2.0/UDP atcf.visited2.net:5060;branch=z9hG4bk731b87
Max-Forwards: 70
P-Asserted-Identity: <tel:+1-237-555-2222>
P-Charging-Vector: icid-value="AyretyU0dm+6O2IrT5tAFrbHLso=023551024";orig-ioi=visit1.net
Privacy: none
From: <tel:+1-237-555-3333>;tag=1888828
To: <tel: +1-237-555-4444>
Call-ID: cb03a0s09a2sdfglkj490444
Cseq: 127 INVITE
Supported: 100rel, precondition, gruu
Require: tdialog,
Record-Route:<sip: atcf.visited2.net:5060;lr>
Target-Dialog: me03a0s09a2sdfgjkl491777; remote-tag=774321; local-tag=64727891
Accept-Contact: *;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
P-Asserted-Service: urn:urn-7:3gpp-service.ims.icsi.mmtel
Contact: <sip: msc1.visit1.net:1357>;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ggg
s=
c=IN IP6 5555::aaa:bbb:ccc:ggg
t=0 0
m=audio 3456 RTP/AVP 97 96
a=tcap:1 RTP/AVPF
a=pcfg:1 t=1
```

```
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:96 telephone-event
a=maxptime:20
```

Request-URI: contains the ATU-STI, that resolves (e.g. via DNS access) to the IP address of the I-CSCF.

Target-Dialog: specifies that the existing dialog is related with this request.

Record-Route: contains the SIP URI of the ATCF, where the ATCF expect to receive the in-dialog request from the SCC AS.

Require: the "tdialog" option tag indicate that the support for Target-Dialog header field is required.

P-Asserted-Identity: the C-MSISDN of the served UE.

SDP: the media information at ATGW.

8. SIP re-INVITE request (SCC AS to UE-B)

When the SCC AS receives the SIP INVITE from ATCF, since the SDP is different with the one in old session, the SCC AS performs the remote leg update with sending a SIP re-INVITE request to remote UE B.

9. SIP 200 (OK) response (UE B to SCC AS)

10. SIP ACK request (SCC AS to UE B)

11-12. SIP 200 (OK) response (SCC AS to ATCF)

13-14. SIP ACK request (ATCF to SCC AS)

15. There are old PS media and new PS media between ATCF and remote UE B at the same time. Therefore there is no session break due to remote leg update.

16-18. SIP BYE request (SCC AS to UE A)

The SCC AS terminates the source access leg, which was using the old IP-CAN, by sending a SIP BYE request to the UE A. In this case, it's assumed that the SIP BYE request go to UE A.

19-21. SIP 200 (OK) response (UE A to SCC AS)

Upon receiving the SIP BYE request, the UE A sends a SIP 200 (OK) response to the SCC AS. Subsequently, the UE A relinquishes all resources pertaining to the old IP-CAN.

22. After the PS to CS session transfer is completed, a new CS media between UA and MSC Server/MGW is established, and the PS media between MSC Server/MGW and ATCF/ATGW is established, and a new PS media between ATCF/ATGW and UE B is established.

A.19 Signalling flows for vSRVCC

A.19.1 Introduction

The signalling flows in this subclause demonstrate how a session consisting of active speech and video media components is transferred from PS to CS using vSRVCC procedures.

A.19.2 Session transfer for active call with speech and video using vSRVCC procedure: PS to CS

In the example flow in figure A.19.2-1, the SC UE A has an ongoing session anchored at SCC AS, with only active speech and video media components and the SC UE has an ongoing session anchored at the SCC AS with only inactive speech media component. Based upon measurement reports sent from the UE to E-UTRAN, the source E-UTRAN decides to trigger a Single Radio handover from PS to CS access.

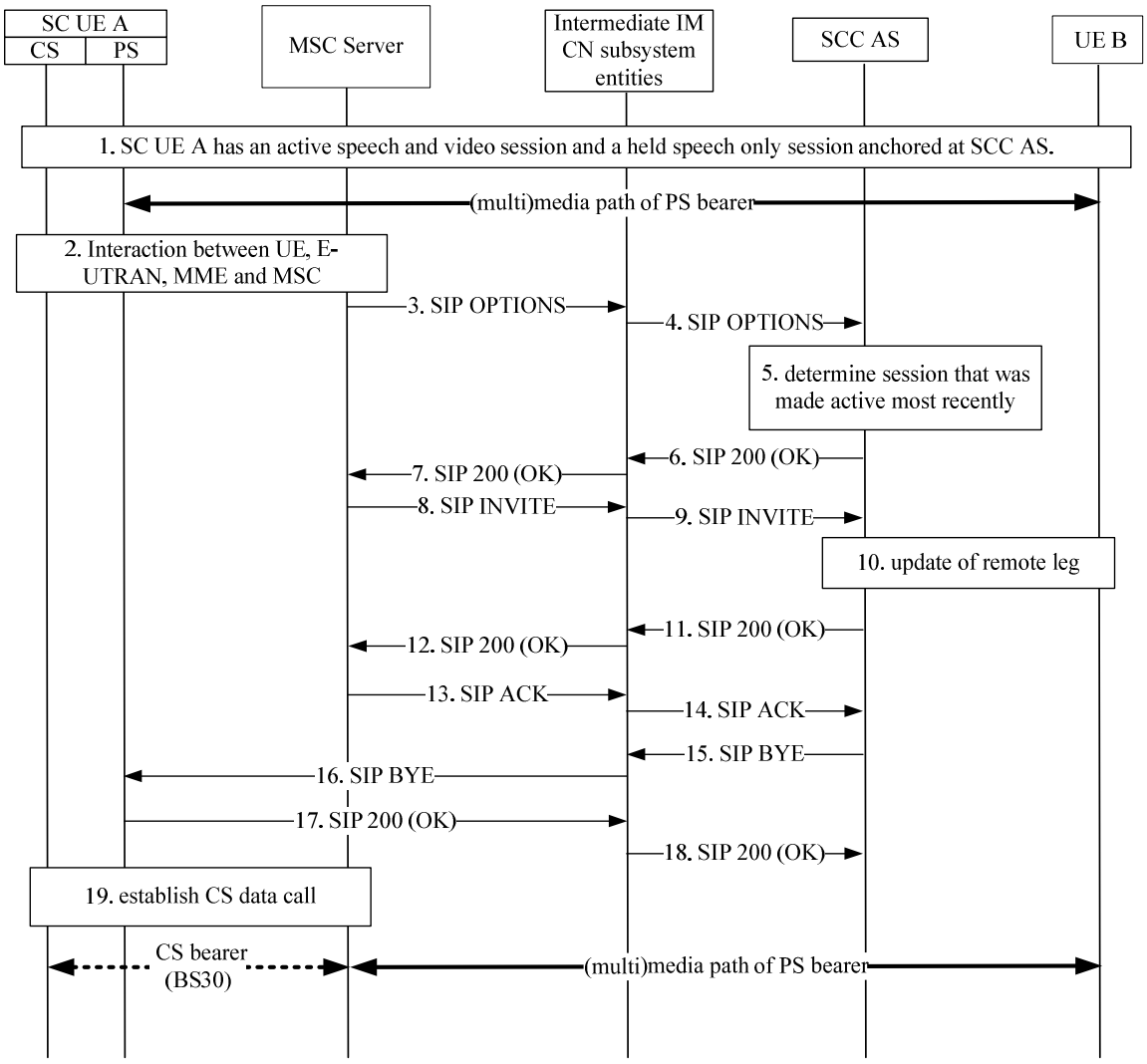


Figure A.19.2-1: PS-CS vSRVCC for an ongoing speech and video session

1. **SC UE A has an active speech and video session and a held speech-only session**

   The UE has an session containing only active speech and video media components, and a session containing only inactive speech media component. The corresponding multimedia PS bearer extends between UE A and UE B via intermediate IM CN entities.

2. **Interaction between UE, E-UTRAN, MME and MSC**

   UE A sends the measurement reports to E-UTRAN, and the source E-UTRAN decides to trigger a Single Radio handover from PS to CS access (as specified in 3GPP TS 23.216 [5]). The MME sends a PS to CS Request message, containing an vSRVCC indication and STN-SR, via Sv to the MSC.

3. **SIP OPTIONS request (MSC server to intermediate IM CN subsystem entities) - see example in table A.19.2-1**

The MSC has no prior knowledge whether the most recently made active session is a session with audio and video media or a session with only audio media. The receipt of the vSRVCC indication over the Sv interface is the trigger for the MSC server to send a SIP OPTIONS request for the purpose of determining the session that was made active most recently.

Table A.19.2-1: SIP OPTIONS request (MSC server to intermediate IM CN subsystem entities)

```
OPTIONS tel: +1-237-555-3333 SIP/2.0
Via: SIP/2.0/UDP mgcf1.visit1.net;branch=z9hG4bk731b87
Max-Forwards: 70
P-Asserted-Identity: <tel:+1-237-555-1111>
From: <tel:+1-237-555-1111>;tag=171828
To: tel:+1-237-555-3333
Call-ID: cb03a0s09a2sdfglkj490334
Contact: sip:msc1.visit1.net
Accept: application/sdp
Content-Length: 0
```

4. **SIP OPTIONS request (intermediate IM CN subsystem entities to SCC AS)**

   The SIP OPTIONS request is routed towards the SCC AS.

5. **Determination of session that was made active most recently**

   The SCC AS determines the session that was made active most recently, based on the C-MSISDN in the P-Asserted-Identity header field.

6. **SIP 200 (OK) response (SCC AS to Intermediate IM CN subsystem entities)**

   The SCC AS sends a SIP 200 (OK) response. with an SDP body containing "m=" lines for audio and video, as the session that was made active most recently is a session with both audio and video media. The contents of the SIP 200 (OK) allow the MSC server to determine how to populate the SDP Offer in the SIP INVITE for STN-SR (see Step 8).

Table A.19.2-2: SIP 200 (OK) response (SCC AS to intermediate IM CN subsystem entities)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP mgcf1.visit1.net;branch=z9hG4bk731b87
From: <tel:+1-237-555-3333>
To: <tel:+1-237-555-1111>;tag=314159
Call-ID: cb03a0s09a2sdfglkj490334
Content-Type: application/sdp
Content-Length: (…)
m=audio 3456 RTP/AVP 97 96
m=video 3400 RTP/AVP 98 99
```

7. **SIP 200 (OK) response (Intermediate IM CN subsystem entities to MSC server)**

   The intermediate IM CN subsystem entities forward the SIP 200 (OK) response to the MSC server.

8. **SIP INVITE (MSC server to Intermediate IM CN subsystem entities)**

   The MSC server sends a SIP INVITE due to STN-SR with the following parameter settings:

   Request-URI: contains the STN-SR

   SDP: contains only a speech media component and a video media component with default codecs for speech and video (as specified in 3GPP TS 26.111 [69])

   Contact: contains the address of the MSC server.

9. **SIP INVITE (Intermediate IM CN subsystem entities to SCC AS)**

   The SIP INVITE is forwarded to the SCC AS with parameters listed in step 8 unchanged.

10. **Update of the remote leg**

The SCC AS performs SIP signaling (re-INVITE) for the purpose of updating the remote leg (UE B) with new contact information (for signaling and media).

11. and 12. SIP 200 (OK) response (SCC AS via Intermediate IM CN subsystem entities to MSC server)

The SCC AS acknowledges the receipt of the INVITE received in step 8. The SIP acknowledgement is forwarded to the MSC server.

13. and 14. SIP ACK (MSC server via Intermediate IM CN subsystem entities to SCC AS)

The MSC server sends a SIP acknowledgement to the Intermediate IM CN subsystem entities; it is forwarded to the SCC AS.

15. and 16. SIP BYE request (SCC AS via intermediate IM CN subsystem entities to UE)

The SCC AS releases the source access.

17. and 18. SIP 200 (OK) response (UE to SCC AS via Intermediate IM CN subsystem entities)

The UE acknowledges the release of the source access leg.

19. UE attaches in CS and sets up a data call

The UE attaches in the CS domain and establishes a data call based on a BS30 bearer, as described in 3GPP TS 24.008 [8]. The UE replaces the session with active speech and video media components with the newly established CS video call.

A.20 Signalling flows for CS to PS Access Transfer: using CS to PS SRVCC

A.20.1 Introduction

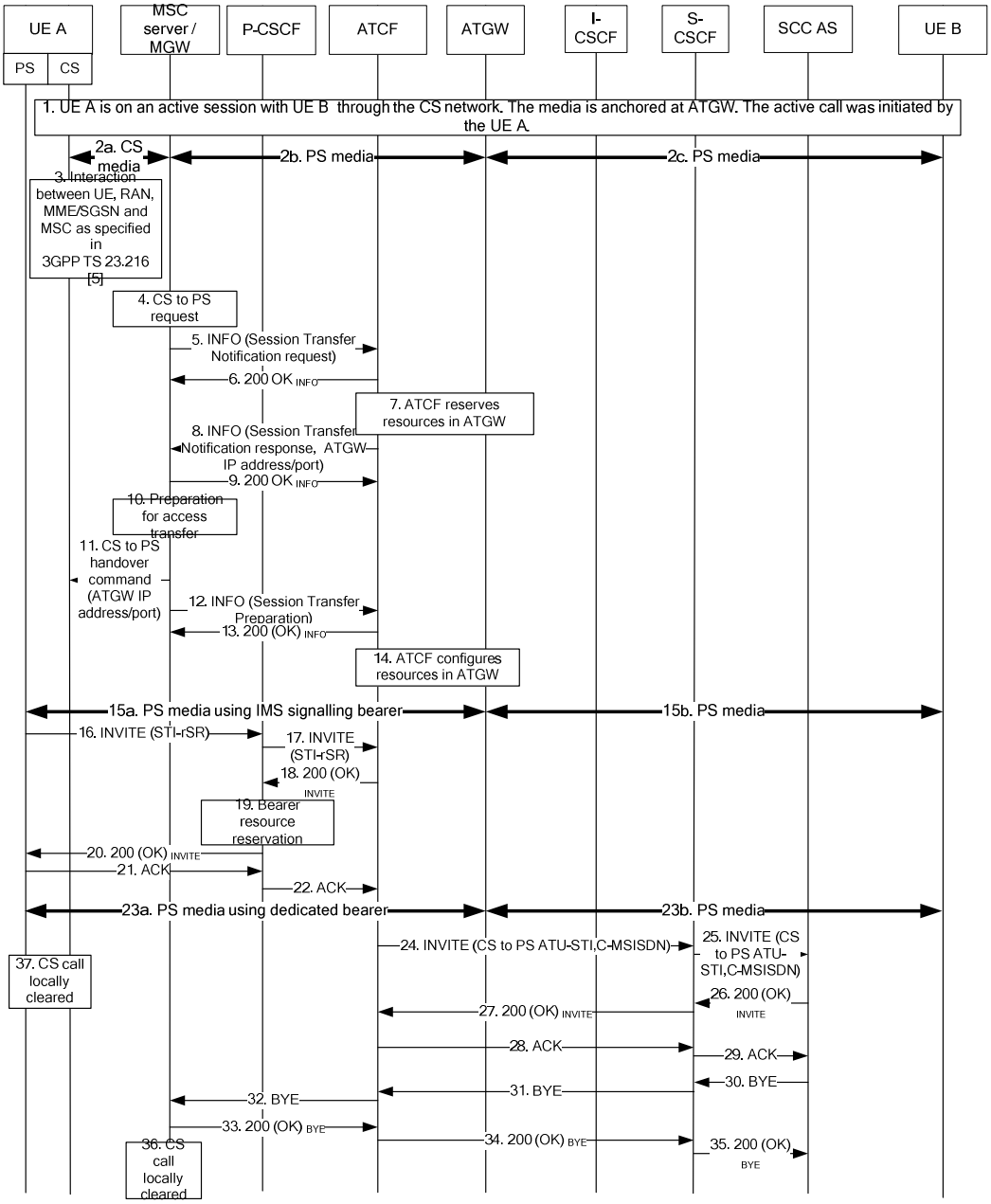
The signalling flows in the subclause demonstrate the CS to PS access SRVCC transfer. The following signalling flows are included:

- subclause A.20.2 shows an example of CS to PS access transfer for SRVCC when CS to PS SRVCC occurs during an ongoing call for which media are anchored in ATGW.
- subclause A.20.3 shows an example of CS to PS access transfer for SRVCC when CS to PS SRVCC occurs during an ongoing call for which media are not anchored in ATGW.

A.20.2 Signalling flows for CS to PS Access Transfer: CS to PS SRVCC occurs during an active call

The signalling flow shown in figure A.20.2-1 gives an example for CS to PS access transfer when using CS to PS SRVCC. The call is established, contains active speech media component and has been anchored in ATGW during the establishment of the call.

The call may have been established either via the MSC server or as the result of the CS to PS SRVCC procedure.



NOTE: For clarity, the SIP 100 (Trying) responses are not shown in the signalling flow.

Figure A.20.2-1 Signalling flows for CS to PS Access Transfer: CS to PS SRVCC occurs during a call.

1+2. The UE A has an session with active speech media component with UE B

There is one CS bearer between the UE A and the MSC server, one PS bearer between the MSC server and the ATGW and one PS bearer between the ATGW and the remote end UE B. The CS call has the transaction identifier 88 (decimal) and was originated by UE B and accepted by UE A.

3. The UE A sends the measurement reports to UTRAN/GERAN that decides to trigger a CS to PS SRVCC handover to the E-UTRAN access.

4. CS to PS request

The MSC server receives a CS to PS request indicating that a CS to PS SRVCC access transfer is initiated.

5. SIP INFO request (MSC server to ATCF) - see example in table A.20.1-5

The MSC Server initiates the CS to PS SRVCC by means of a SIP INFO request sent towards the ATCF. The SIP INFO includes a session transfer notification request. The session transfer notification request is an indication to prepare for the transfer of media to PS.

Table A.20.2-5: SIP INFO request (MSC server to ATCF)

```
INFO sip:user2_public1@home2.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]:1357; branch=z9hG4bKnashds7
Max-Forwards: 70
Route: <sip:atcf.visited.net;lr>, <sip:scscf.home1.net;lr>, <sip:icscf.home1.net;lr>,
   <sip:sccas.home1.net;lr>
From: <tel:+1-212-555-2222>;tag=171828
To: <sip:user1_public1@home1.net>; tag=171828
Call-ID: cb03a0s09a2sdfglkj490333
Cseq: 130 INFO
Info-Package: g.3gpp.access-transfer-events
Content-Disposition: Info-Package
Content-Type: application/vnd.3gpp.access-transfer-events+xml
Content-Length: (…)

<?xml version="1.0"?>
<events>
<event event-type="1"/>
</events>
```

application/vnd.3gpp.access-transfer-events+xml: Contains the value 1 indicating that the MSC server sends the session transfer notification request to the ATCF.

6 SIP 200 (OK) response (ATCF to MSC server)

The ATCF acknowledge the SIP INFO request.

7. ATCF reserves resources in ATGW

The ATCF reserves resources in ATGW towards UE A and the ATGW provides the SDP answer to the SDP which the UE A provided during the registration (see subclase A.3.4). Apart from the IP address and port, the SDP answer contains the same media pararameters as provided to the UE A after PS registration (see subclause A.3.4).

8. SIP INFO request (ATCF to MSC server) - see example in table A.20.1-8

The ATCF sends a SIP INFO request containing the session transfer notification response contains the parameters required for the transfer, including IP address and media port allocated in the ATGW.

Table A.20.2-8: SIP INFO request (ATCF to MSC server)

```
INFO sip:user1_public1@visited2.net SIP/2.0
Via:
Max-Forwards:
Record-Route:
From:
To:
Call-ID:
Cseq:
Content-Disposition: Info-Package
Info-Package: g.3gpp.access-transfer-events
Contact:
Content-Type: application/vnd.3gpp.access-transfer-events+xml
Content-Length: (…)

<?xml version="1.0"?>
<events>
<event event-type="2">
  <STNResp-params>
    <transfer-details>AVL0IrgAAAAAAAAAbwDeAU0BvA==</transfer-details>
    <redirect-speech>false</redirect-speech>
  </STNResp-params>
</event>
</events>
```

application/vnd.3gpp.access-transfer-events+xml: Contains the IPv6 address and port number of the ATGW. With the following <transfer-details>: ATGW-IPv6-address = 8888::111:222:333:444, ATGW-audio-UDP-port = 21236. Also indicates that the ATCF does not require the MSC server to redirect the speech media component of the session transferred by the CS to PS SRVCC access transfer.

Info-Package: Indicates that the SIP INFO request contains the g.3gpp.access-transfer-events info package.

9. SIP 200 (OK) response (MSC server to ATCF)

The MSC server acknowledge the SIP INFO request.

10. The MSC server starts the preparation for the access transfer.

11. When access transfer is prepared, the MSC server sends CS to PS handover command to the UE using access stratum signalling.

12. SIP INFO request (MSC server to ATCF) - see example in table A.20.1-12

The MSC server sends a SIP INFO request containing a session transfer preparation to the ATCF to instruct the ATCF that media should be switched to the target access.

Table A.20.2-12: SIP INFO request (MSC server to ATCF)

```
INFO sip:user2_public1@home2.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]:1357; branch=z9hG4bKnashds7
Max-Forwards: 70
Route: <sip:atcf.visited.net;lr>, <sip:scscf.home1.net;lr>, <sip:icscf.home1.net;lr>,
   <sip:sccas.home1.net;lr>
From: <tel:+1-212-555-2222>;tag=171828
To: <sip:user1_public1@home1.net>; tag=171828
Call-ID: cb03a0s09a2sdfglkj490333
Cseq: 130 INFO
Content-Disposition: Info-Package
Info-Package: g.3gpp.access-transfer-events
Content-Type: application/vnd.3gpp.access-transfer-events+xml
Content-Length: (…)

<?xml version="1.0"?>
<events>
<event event-type="3"/>
</events>
```

application/vnd.3gpp.access-transfer-events+xml: Contains the event 3 indicating that MSC server requests ATCF to perform the CS to PS SRVCC access transfer, i.e. start sending media towards the UE instead of towards the MSC server.

13. SIP 200 (OK) response (ATCF to MSC server)

The ATCF acknowledge the SIP INFO request by means of a SIP 200 (OK) response.

14. ATCF configures resources in ATGW

The ATCF configures resources in the ATGW to start sending and receiving media towards the UE A instead of the MSC server.

15. The media path is now reconfigured. The audio is sent between the UE A and ATGW using IMS signalling bearer.

16-17. SIP INVITE request (UE A to ATCF) - see example in table A.20.1-16

When the UE A receives the CS to PS handover command the UE A sends an SIP INVITE request towards the ATCF.

Table A.20.2-16: SIP INVITE request (UE A to ATCF)

```
INVITE sip:sti-rsr@atcf1.visited2.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Max-Forwards: 70
Route: <sip:pcscf1.visited2.net:7531;lr;comp=sigcomp>,<sip:atcf.visited.net;lr,
   <sip:orig@scscf1.home1.net;lr>
P-Preferred-Identity: "John Doe" <tel:+1-212-555-1111>
P-Preferred-Service: urn:urn-7:3gpp-service.ims.icsi.mmtel
P-Access-Network-Info: 3GPP-UTRAN-TDD; utran-cell-id-3gpp=234151D0FCE11
Privacy: none
From: <tel:+1-212-555-1111>;tag=171828
To: <tel:+1-212-555-2222>
Call-ID: cb03a0s09a2sdfglkj490333
Cseq: 127 INVITE
Require: sec-agree
Supported: 100rel, gruu
Proxy-Require: sec-agree
Security-Verify: ipsec-3gpp; q=0.1; alg=hmac-sha-1-96; spi-c=98765432; spi-s=87654321; port-
   c=8642; port-s=7531
Contact: <sip:user1_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-
   00a0c91e6bf6;comp=sigcomp>;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
Accept-Contact: *;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE
Content-Type: application/sdp
Content-Length: (…)
Accept: application/sdp,application/3gpp-ims+xml

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c=IN IP6 5555::aaa:bbb:ccc:ddd
t=0 0
m=audio 3456 RTP/AVP 97 96
b=AS:25.4
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 telephone-event
```

Request-URI: contains the STI-rSR associated with the transferred call.

SDP offer: The media parameters of the speech media component are the same as the UE A sent to ATCF during registraton (see subclause A.3.4).

18. SIP 200 (OK) response (ATCF to P-CSCF)- see example in table A.20.1-18

The ATCF sends the SIP 200 (OK) response towards the UE A with the media information allocated by the ATGW.

Table A.20.2-18: SIP 200 (OK) response (ATCF to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Max-Forwards: 70
Record-Route: <sip:pcscf1.visited2.net:7531;lr;comp=sigcomp>,<sip:atcf.visited.net;lr>
P-Asserted-Identity: <tel:+1-212-555-2222>
Privacy: none
From: <tel:+1-212-555-1111>;tag=171828
To: <tel:+1-212-555-2222>;tag=aaaa
Call-ID: cb03a0s09a2sdfglkj490333
Cseq: 127 INVITE
Contact: <sip:user1_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-
   00a0c91e6bf6;comp=sigcomp>;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
Feature-Caps: *;+g.3gpp.ti="70D8"
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c=IN IP6 5555::aaa:bbb:ccc:ddd
t=0 0
m=audio 3456 RTP/AVP 97 96
b=AS:25.4
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 telephone-event
```

Feature-Caps: g.3gpp.ti feature-capability indicator with value containing the transaction identifier specified in figure 11.9 and table 11.3 of 3GPP TS 24.007 [75] encoded by hexadecimal digit. In this example, the transaction identifier 88 (decimal) and the transaction identifier flag as sent by the MSC server in CS signalling of the terminating CS call are shown.

19. Bearer resource reservation

P-CSCF initiates bearer resource reservation based on the SDP answer received in the SIP 200 (OK) response.

20. SIP 200 (OK) response (P-CSCF to UE A)

The P-CSCF forwards the SIP 200 (OK) response to the UE A. The UE A associates the dialog established by the SIP 200 (OK) response with the CS call where the transaction identifier sent by MSC server were equal to the value of the g.3gpp.ti feature-capability indicator in the Feature-Caps header field of the SIP 200 (OK) response.

21-22. SIP ACK request (UE A to ATCF)

The UE A acknowledges the reception of the SIP 200 (OK) response.

23. The media path is now reconfigured. The audio is sent between the UE A and ATGW using a dedicated bearer.

24-25. SIP INVITE request (ATCF to SCC AS) - see example in table A.20.1-25

Table A.20.2-25: SIP INVITE request (ATCF to SCC AS)

```
INVITE sip:cs2ps@sccas1.home1.net SIP/2.0
Record-Route: <sip:atcf2.visited2.net;lr>
Via: SIP/2.0/UDP atcf.visited2.net:5060;branch=z9hG4bKnas55889, SIP/2.0/UDP
   [5555::aaa:bbb:ccc:ddd]:1357;;branch=z9hG4bKnashds7
Max-Forwards: 69
Route: <sip:orig@scscf1.home1.net;lr>
P-Asserted-Identity: <tel:+1-212-555-1111>
P-Charging-Vector: icid-value="1234bc9876e";icid-generated-at"5555::aaa:bbb:ccc:ddd";orig-
   ioi=visited2.net
P-Preferred-Service:
P-Access-Network-Info:
Privacy:
From:
To:
Call-ID:
Cseq:
Require:
Supported:
Proxy-Require:
Contact:
Accept-Contact:
Allow:
Content-Type:
Content-Length:
Accept:

v=0
o=- 22 333 IN IP6 8888::111:222:333:444
s=-
c=IN IP6 8888::111:222:333:444
t=0 0
m=audio 8899 RTP/AVP 97 96
b=AS:25.4
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 telephone-event
```

Request-URI: contains the ATU-STI for CS to PS SRVCC associated with the transferred call.

SDP: The SDP contains the SDP used at ATGW towards the remote UE B.

P-Asserted-Identity: the C-MSISDN of the served UE.

26-27. SIP 200 (OK) response (SCC AS to ATCF)

Since there is no update in the session description, no remote end update will be performed. The SCC AS sends confirmation response to the ATCF which contains the SDP answer that the SCC AS stored during the original session establishment procedure.

28-29. SIP ACK request (ATCF to SCC AS)

30-32. SIP BYE request (SCC AS to MSC server)

The SCC AS initiates the release of the source access leg.

33-35. SIP 200 (OK) response (MSC server to UE A)

36. The MSC server clears the call

The MSC server locally clears the call.

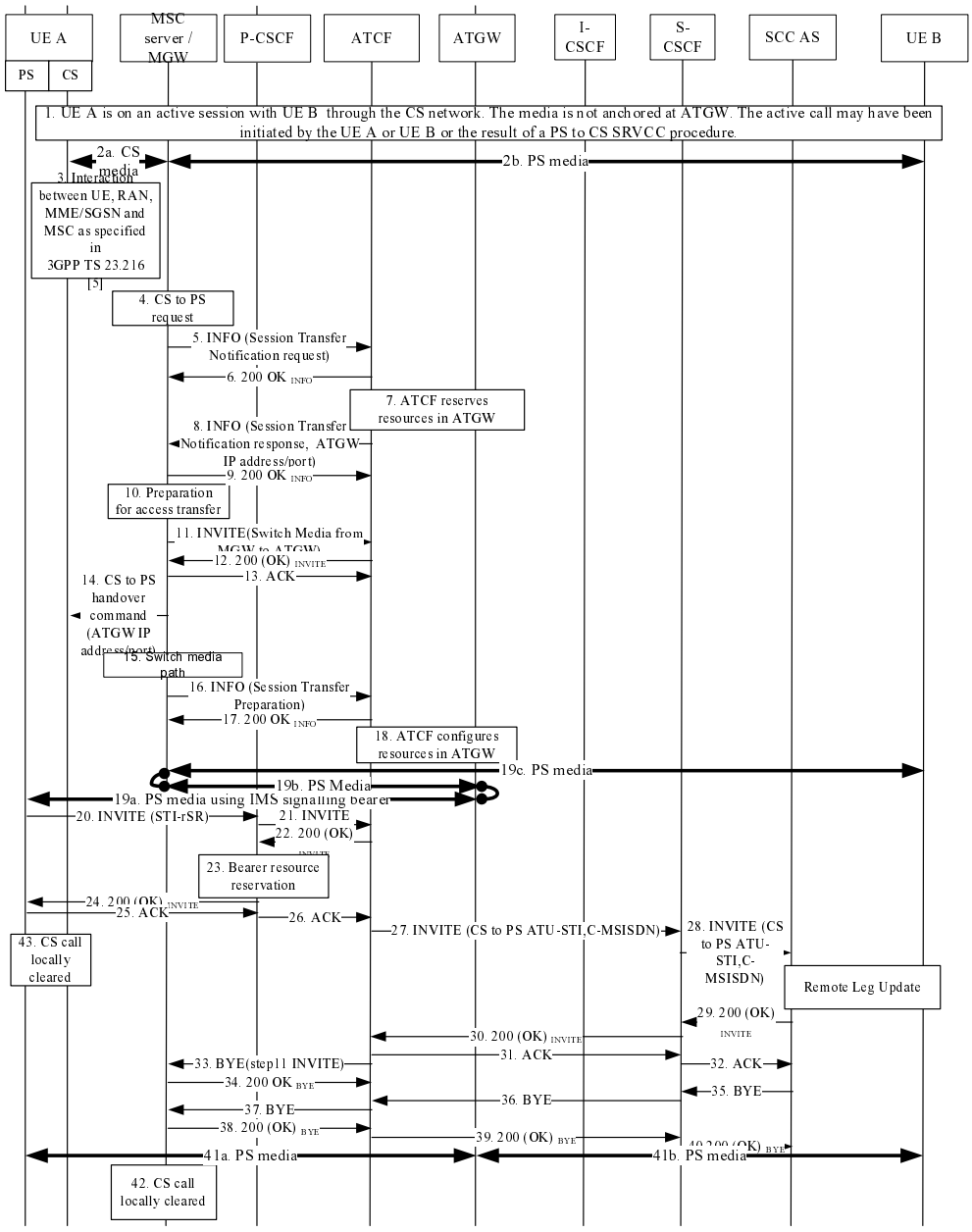
37. The UA A clears the call

The UA locally clears the call.

A.20.3 Signalling flows for CS to PS Access Transfer without CS media anchored in ATGW: CS to PS SRVCC occurs during an active call

The signalling flow shown in figure A.20.3-1 gives an example for CS to PS access transfer when using CS to PS SRVCC. The call is established, contains active speech media component and has not been anchored in ATGW during the establishment of the call.

The call may have been established either via the MSC server or as the result of the CS to PS SRVCC procedure.



NOTE: For clarity, the SIP 100 (Trying) responses are not shown in the signalling flow.

Figure A.20.3-1 Signalling flows for CS to PS Access Transfer: CS to PS SRVCC occurs during a call.

1+2. The UE A has an session with active speech media component with UE B

There is one CS bearer between the UE A and the MSC server, one PS bearer between the MSC server and the remote end UE B. The CS call has the transaction identifier 88 (decimal) and was originated by UE B and accepted by UE A.

3. **The UE A sends the measurement reports to UTRAN/GERAN that decides to trigger a CS to PS SRVCC handover to the E-UTRAN access.**

4. **CS to PS request**

   The MSC server receives a CS to PS request indicating that a CS to PS SRVCC access transfer is initiated.

5. **SIP INFO request (MSC server to ATCF) - see example in table A.20.3-5**

   The MSC Server initiates the CS to PS SRVCC by means of a SIP INFO request sent towards the ATCF. The SIP INFO includes a session transfer notification request. The session transfer notification request is an indication to prepare for the transfer of media to PS.

Table A.20.3-5: SIP INFO request (MSC server to ATCF)

```
INFO sip:user2_public1@home2.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]:1357; branch=z9hG4bKnashds7
Max-Forwards: 70
Route: <sip:atcf.visited.net;lr>, <sip:scscf.home1.net;lr>, <sip:icscf.home1.net;lr>,
   <sip:sccas.home1.net;lr>
From: <tel:+1-212-555-2222>;tag=171828
To: <sip:user1_public1@home1.net>; tag=171828
Call-ID: cb03a0s09a2sdfglkj490333
Cseq: 130 INFO
Content-Disposition: Info-Package
Info-Package: g.3gpp.access-transfer-events
Content-Type: application/vnd.3gpp.access-transfer-events+xml
Content-Length: (…)

<?xml version="1.0"?>
<events>
<event event-type="1"/>
</events>
```

application/vnd.3gpp.access-transfer-events+xml: Contains the value 1 indicating that the MSC server sends the session transfer notification request to the ATCF.

6 **SIP 200 (OK) response (ATCF to MSC server)**

   The ATCF acknowledge the SIP INFO request.

7. **ATCF reserves resources in ATGW**

   The ATCF reserves resources in ATGW towards UE A and the ATGW provides the SDP answer to the SDP which the UE A provided during the registration (see subclase A.3.4). Apart from the IP address and port, the SDP answer contains the same media pararameters as provided to the UE A after PS registration (see subclause A.3.4).

8. **SIP INFO request (ATCF to MSC server) - see example in table A.20.3-8**

   The ATCF sends a SIP INFO request containing the session transfer notification response contains the parameters required for the transfer, including IP address and media port allocated in the ATGW.

Table A.20.3-8: SIP INFO request (ATCF to MSC server)

```
INFO sip:user1_public1@visited2.net SIP/2.0
Via:
Max-Forwards:
Record-Route:
From:
To:
Call-ID:
Cseq:
Content-Disposition: Info-Package
Info-Package: g.3gpp.access-transfer-events
Contact:
Content-Type: application/vnd.3gpp.access-transfer-events+xml
Content-Length: (…)

<?xml version="1.0"?>
<events>
<event event-type="2">
  <STNResp-params>
    <transfer-details>AVL0IrgAAAAAAAAAbwDeAU0BvA==</transfer-details>
    <redirect-speech>true</redirect-speech>
  </STNResp-params>
</event>
</events>
```

application/vnd.3gpp.access-transfer-events+xml: Contains the IPv6 address and port number of the ATGW. With the following <transfer-details>: ATGW-IPv6-address = 8888::111:222:333:444, ATGW-audio-UDP-port = 21236. Also indicates that the ATCF requires the MSC server to redirect the speech media component of the session transferred by the CS to PS SRVCC access transfer.

Info-Package: Indicates that the SIP INFO request contains the g.3gpp.access-transfer-events info package.

9. SIP 200 (OK) response (MSC server to ATCF)

The MSC server acknowledge the SIP INFO request.

10. The MSC server starts the preparation for the access transfer.

11. SIP INVITE request (MSC server to ATCF) - see example in table A.20.3-11

The MSC sends a SIP INVITE request to the ATCF to instruct the ATCF to establish the media bearer between MGW and ATGW.

Table A.20.3-11: SIP INVITE request (MSC server to ATCF)

```
INVITE sip:user2_public1@home2.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]:1357; branch=z9hG4bKnashds7
Max-Forwards: 70
Route: <sip:atcf.visited.net;lr>, <sip:scscf.home1.net;lr>, <sip:icscf.home1.net;lr>,
   <sip:sccas.home1.net;lr>
From: <tel:+1-212-555-2222>;tag=171828
To: <sip:user1_public1@home1.net>;
Call-ID: cb03a0s09a2sdfglkj490333
Contact: <sip:msc1.home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91e6bf6>;+g.3gpp.icsi-
   ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER
Cseq: 1 INVITE
Content-Type: application/sdp
Content-Length: (…)

…
```

12. SIP 200 (OK) response (ATCF to MSC server)

The ATCF acknowledge the SIP INVITE request by means of a SIP 200 (OK) response.

13. SIP ACK request (MSC Server to ATCF)

The MSC Server acknowledges the reception of the SIP 200 (OK) response.

14. When access transfer is prepared, the MSC server sends CS to PS handover command to the UE using access stratum signalling.

15. The MSC Server instructs the MGW to switch the media path from the source access to the target access.

16. SIP INFO request (MSC server to ATCF) - see example in table A.20.3-16

The MSC server sends a SIP INFO request containing a session transfer preparation to the ATCF to instruct the ATCF that media should be switched to the target access.

Table A.20.3-16: SIP INFO request (MSC server to ATCF)

```
INFO sip:user2_public1@home2.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]:1357; branch=z9hG4bKnashds7
Max-Forwards: 70
Route: <sip:atcf.visited.net;lr>, <sip:scscf.home1.net;lr>, <sip:icscf.home1.net;lr>,
   <sip:sccas.home1.net;lr>
From: <tel:+1-212-555-2222>;tag=171828
To: <sip:user1_public1@home1.net>; tag=171828
Call-ID: cb03a0s09a2sdfglkj490333
Cseq: 130 INFO
Content-Disposition: Info-Package
Info-Package: g.3gpp.access-transfer-events
Content-Type: application/vnd.3gpp.access-transfer-events+xml
Content-Length: (…)

<?xml version="1.0"?>
<events>
<event event-type="3"/>
</events>
```

application/vnd.3gpp.access-transfer-events+xml: Contains the event 3 indicating that MSC server requests ATCF to perform the CS to PS SRVCC access transfer, i.e. start sending media towards the UE instead of towards the MSC server.

17. SIP 200 (OK) response (ATCF to MSC server)

The ATCF acknowledge the SIP INFO request by means of a SIP 200 (OK) response.

18. ATCF configures resources in ATGW

The ATCF configures resources in the ATGW to start sending and receiving media towards the UE A instead of the MSC server.

19. The media path is now reconfigured.

20-21. SIP INVITE request (UE A to ATCF) - see example in table A.20.3-20

When the UE A receives the CS to PS handover command the UE A sends an SIP INVITE request towards the ATCF.

Table A.20.3-20: SIP INVITE request (UE A to ATCF)

```
INVITE sip:sti-rsr@atcf1.visited2.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Max-Forwards: 70
Route: <sip:pcscf1.visited2.net:7531;lr;comp=sigcomp>,<sip:atcf.visited.net;lr,
   <sip:orig@scscf1.home1.net;lr>
P-Preferred-Identity: "John Doe" <tel:+1-212-555-1111>
P-Preferred-Service: urn:urn-7:3gpp-service.ims.icsi.mmtel
P-Access-Network-Info: 3GPP-UTRAN-TDD; utran-cell-id-3gpp=234151D0FCE11
Privacy: none
From: <tel:+1-212-555-1111>;tag=171828
To: <tel:+1-212-555-2222>
Call-ID: cb03a0s09a2sdfglkj490333
Cseq: 127 INVITE
Require: sec-agree
Supported: 100rel, gruu
Proxy-Require: sec-agree
Security-Verify: ipsec-3gpp; q=0.1; alg=hmac-sha-1-96; spi-c=98765432; spi-s=87654321; port-
   c=8642; port-s=7531
Contact: <sip:user1_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-
   00a0c91e6bf6;comp=sigcomp>;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
Accept: application/sdp, application/3gpp-ims+xml
Accept-Contact: *;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c=IN IP6 5555::aaa:bbb:ccc:ddd
t=0 0
m=audio 3456 RTP/AVP 97 96
b=AS:25.4
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 telephone-event
```

Request-URI: contains the STI-rSR associated with the transferred call.

SDP offer: The media parameters of the speech media component are the same as the UE A sent to ATCF during registration (see subclause A.3.4).

22. SIP 200 (OK) response (ATCF to P-CSCF) - see example in table A.20.3-22

The ATCF sends the SIP 200 (OK) response towards the UE A with the media information allocated by the ATGW.

Table A.20.3-22: SIP 200 (OK) response (ATCF to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Max-Forwards: 70
Record-Route: <sip:pcscf1.visited2.net:7531;lr;comp=sigcomp>,<sip:atcf.visited.net;lr>
P-Asserted-Identity: <tel:+1-212-555-2222>
Privacy: none
From: <tel:+1-212-555-1111>;tag=171828
To: <tel:+1-212-555-2222>;tag=aaaa
Call-ID: cb03a0s09a2sdfglkj490333
Cseq: 127 INVITE
Contact: <sip:user1_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-
   00a0c91e6bf6;comp=sigcomp>;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
Feature-Caps: *;+g.3gpp.ti="70D8"
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c=IN IP6 5555::aaa:bbb:ccc:ddd
t=0 0
m=audio 3456 RTP/AVP 97 96
b=AS:25.4
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 telephone-event
```

Feature-Caps: g.3gpp.ti feature-capability indicator with value containing the transaction identifier specified in figure 11.9 and table 11.3 of 3GPP TS 24.007 [75] encoded by hexadecimal digit. In this example, the transaction identifier 88 (decimal) and the transaction identifier flag as sent by the MSC server in CS signalling of the terminating CS call are shown.

23. Bearer resource reservation

P-CSCF initiates bearer resource reservation based on the SDP answer received in the SIP 200 (OK) response.

24. SIP 200 (OK) response (P-CSCF to UE A)

The P-CSCF forwards the SIP 200 (OK) response to the UE A. The UE A associates the dialog established by the SIP 200 (OK) response with the CS call where the transaction identifier sent by MSC server were equal to the value of the g.3gpp.ti feature-capability indicator in the Feature-Caps header field of the SIP 200 (OK) response.

25-26. SIP ACK request (UE A to ATCF)

The UE A acknowledges the reception of the SIP 200 (OK) response.

25. The media path is now reconfigured. The audio is sent between the UE A and ATGW using a dedicated bearer.

27-28. SIP INVITE request (ATCF to SCC AS) - see example in table A.20.3-27

Table A.20.3-27: SIP INVITE request (ATCF to SCC AS)

```
INVITE sip:cs2ps@sccas1.home1.net SIP/2.0
Record-Route: <sip:atcf2.visited2.net;lr>
Via: SIP/2.0/UDP atcf.visited2.net:5060;branch=z9hG4bKnas55889, SIP/2.0/UDP
   [5555::aaa:bbb:ccc:ddd]:1357;;branch=z9hG4bKnashds7
Max-Forwards: 69
Route: <sip:orig@scscf1.home1.net;lr>
P-Asserted-Identity: <tel:+1-212-555-1111>
P-Charging-Vector: icid-value="1234bc9876e";icid-generated-at"5555::aaa:bbb:ccc:ddd";orig-
   ioi=visited2.net
P-Preferred-Service:
P-Access-Network-Info:
Privacy:
From:
To:
Call-ID:
Cseq:
Require:
Supported:
Proxy-Require:
Contact:
Accept:
Accept-Contact:
Allow:
Content-Type:
Content-Length:


v=0
o=- 22 333 IN IP6 8888::111:222:333:444
s=-
c=IN IP6 8888::111:222:333:444
t=0 0
m=audio 8899 RTP/AVP 97 96
b=AS:25.4
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 telephone-event
```

Request-URI: contains the ATU-STI for CS to PS SRVCC associated with the transferred call.

SDP: The SDP contains the SDP used at ATGW towards the remote UE B.

P-Asserted-Identity: the C-MSISDN of the served UE.

29-30. SIP 200 (OK) response (SCC AS to ATCF)

Since there is update in the session description, remote end update will be performed. The SCC AS sends confirmation response to the ATCF which contain the SDP answer that the SCC AS recevied during remote end update procedure.

31-32. SIP ACK request (ATCF to SCC AS)

33. SIP BYE request (ATCF to MSC Server)

Upon receiving the SIP 200 (OK) response from SCC AS, the ATCF sends a SIP BYE request to MSC Server to release the session established by SIP INVITE requet in step 11.

34. SIP 200 (OK) response (MSC Server to ATCF)

MSC server sends the SIP 200 (OK) response to the ATCF.

35-37. SIP BYE request (SCC AS to MSC server)

The SCC AS initiates the release of the source access leg.

38-40. SIP 200 (OK) response (MSC server to UE A)

41. The audio is sent between the UE A and ATGW using a dedicated PS bearer.

42. The MSC server clears the call

The MSC server locally clears the call.

43. The UA A clears the call

The UA locally clears the call.

Annex B (informative): Void

Annex C (normative): Media feature tags and feature-capability indicators defined within the current document

C.1 General

This subclause describes the media feature tag definitions and the feature-capability indicators definitions that are applicable for the 3GPP IM CN Subsystem for the realisation of the MSC server assisted mid-call feature, Access Transfer Control Function, and PS to CS SRVCC for calls in alerting phase.

C.2 Definition of media feature tag g.3gpp.mid-call

Media feature-tag name: g.3gpp.mid-call

ASN.1 Identifier: 1.3.6.1.8.2.12

Summary of the media feature indicated by this tag: This feature-tag when used in a SIP request or a SIP response indicates that the function sending the SIP message supports the MSC server assisted mid-call feature.

Values appropriate for use with this feature-tag: Boolean

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

Examples of typical use: Indicating that a mobile phone supports the MSC server assisted mid-call feature

Related standards or documents: 3GPP TS 24.237: "IP Multimedia Subsystem (IMS) Service Continuity; Stage 3"

Security Considerations: Security considerations for this media feature-tag are discussed in subclause 12.1 of IETF RFC 3840 [53].

C.2A Definition of feature-capability indicator g.3gpp.mid-call

Feature-capability indicator name: g.3gpp.mid-call

Summary of the feature indicated by this feature-capability indicator:

This feature-capability indicator when used in a Feature-Caps header field of a SIP request or a SIP response indicates that:

1. the functional entity including the feature-capability indicator in the SIP message supports the MSC server assisted mid-call feature; and
2. all entities of which the functional entity including the feature-capability indicator in the SIP message is aware of being requested to support the feature do support the MSC server assisted mid-call feature.

Feature-capability indicator specification reference:

3GPP TS 24.237, http://www.3gpp.org/ftp/Specs/archive/24_series/24.237/

Values appropriate for use with this feature-capability indicator: none

Examples of typical use: Indicating that a network entity supports the MSC server assisted mid-call feature.

Security Considerations: Security considerations for this feature-capability indicator are discussed in clause 9 of IETF RFC 6809 [60].

C.3 Void

C.4 Definition of feature-capability indicator g.3gpp.atcf

Feature-capability indicator name: g.3gpp.atcf

Summary of the feature indicated by this feature-capability indicator:

This feature-capability indicator when included in a Feature-Caps header field as specified in IETF RFC 6809 [60] in a SIP REGISTER request or a SIP response to the SIP REGISTER request indicates presence and support of a resource which is an Access Transfer Control Function (ATCF) and also the session transfer number allocated to the ATCF.

Feature-capability indicator specification reference:

3GPP TS 24.237, http://www.3gpp.org/ftp/Specs/archive/24_series/24.237/

Values appropriate for use with this feature-capability indicator:

None or string with an equality relationship. When used in a Feature-Caps header field in SIP REGISTER request or response, the value is string containing the session transfer number allocated to the ATCF following the syntax as described in table C.4-1 for g-3gpp-atcf-in-path.

Table C.4-1: ABNF syntax of values of the g.3gpp.atcf feature-capability indicator

```
g-3gpp-atcf-in-path = STN-SR
STN-SR = "<" addr-spec ">"
```

The feature-capability indicator is intended primarily for use in the following applications, protocols, services, or negotiation mechanisms: This feature-capability indicator is used to indicate support of the ATCF.

Examples of typical use: Indicating the presence and support of an ATCF on the routing path of the SIP REGISTER request and SIP response to the SIP REGISTER request and providing the session transfer number allocated to this ATCF.

Security Considerations: Security considerations for this feature-capability indicator are discussed in clause 9 of IETF RFC 6809 [60].

C.5 Definition of media feature tag g.3gpp.srvcc-alerting

Media feature-tag name: g.3gpp.srvcc-alerting

ASN.1 Identifier: 1.3.6.1.8.2.13

Summary of the media feature indicated by this tag: This media feature-tag when used in a Contact header field of a SIP request or a SIP response indicates that the functional entity sending the SIP message supports the PS to CS SRVCC for calls in alerting phase, i.e. for calls with early dialog.

Values appropriate for use with this feature-tag: Boolean

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

Examples of typical use: Indicating that a user equipment supports the PS to CS SRVCC for calls in alerting phase.

Related standards or documents: 3GPP TS 24.237: "IP Multimedia Subsystem (IMS) Service Continuity; Stage 3"

Security Considerations: Security considerations for this media feature-tag are discussed in subclause 12.1 of IETF RFC 3840 [53].

C.5A Definition of feature-capability indicator g.3gpp.srvcc-alerting

Feature-capability indicator name: g.3gpp.srvcc-alerting

Summary of the feature indicated by this feature-capability indicator:

This feature-capability indicator when used in a Feature-Caps header field of a SIP request or a SIP response indicates that:

1. the functional entity including the feature-capability indicator in the SIP message supports the PS to CS SRVCC for calls in alerting phase; and
2. all entities of which the functional entity including the feature-capability indicator in the SIP message is aware of being requested to support the feature do support the PS to CS SRVCC for calls in alerting phase.

Feature-capability indicator specification reference:

3GPP TS 24.237, http://www.3gpp.org/ftp/Specs/archive/24_series/24.237/

Values appropriate for use with this feature-capability indicator: none

Examples of typical use: Indicating that a network entity supports the PS to CS SRVCC for calls in alerting phase.

Security Considerations: Security considerations for this feature-capability indicator are discussed in clause 9 of IETF RFC 6809 [60].

C.6 Definition of feature-capability indicator g.3gpp.atcf-mgmt-uri

Feature-capability indicator name: g.3gpp.atcf-mgmt-uri

Summary of the feature indicated by this feature-capability indicator:

This feature-capability indicator when used in a Feature-Caps header field as specified in IETF RFC 6809 [60] in SIP REGISTER request indicates presence and support of performing as a UAS for SIP requests for ATCF management received at this URI.

Feature-capability indicator specification reference:

3GPP TS 24.237, http://www.3gpp.org/ftp/Specs/archive/24_series/24.237/

Values appropriate for use with this feature-capability indicator:

String with an equality relationship. When used in a Feature-Caps header field, the value is string following the syntax as described in table C.6-1 for g-3gpp-atcf-mgmt-uri-in-path.

Table C.6-1: ABNF syntax of values of the g.3gpp.atcf-mgmt-uri feature-capability indicator

```
g-3gpp-atcf-mgmt-uri-in-path = "<" SIP-URI ">"
```

The feature-capability indicator is intended primarily for use in the following applications, protocols, services, or negotiation mechanisms: This feature-capability indicator is used to indicate the management URI of the ATCF for receiving SIP requests where the ATCF performs the UAS role.

Examples of typical use: Indicating the management URI of the ATCF for SIP requests containing PS to CS SRVCC related information.

Security Considerations: Security considerations for this feature-capability indicator are discussed in clause 9 of IETF RFC 6809 [60].

C.7 Definition of feature-capability indicator g.3gpp.srvcc

Feature-capability indicator name: g.3gpp.srvcc

Summary of the feature indicated by this feature-capability indicator:

This feature-capability indicator when included in a Feature-Caps header field as specified in IETF RFC 6809 [60] of:

- a SIP INVITE request; or
- a SIP INVITE response;

indicates presence and support of a resource capable of performing the PS to CS SRVCC access transfer procedure as specified in 3GPP TS 24.237.

Feature-capability indicator specification reference:

3GPP TS 24.237, http://www.3gpp.org/ftp/Specs/archive/24_series/24.237/

Values appropriate for use with this feature-capability indicator: none

The feature-capability indicator is intended primarily for use in the following applications, protocols, services, or negotiation mechanisms: This feature-capability indicator is most useful in a communications application for indicating that a resource supports single radio voice call continuity.

Examples of typical use: Indicating that a resource supports single radio voice call continuity.

Security Considerations: Security considerations for this feature-capability indicator are discussed in clause 9 of IETF RFC 6809 [60].

C.8 Definition of feature-capability indicator g.3gpp.atcf-path

Feature-capability indicator name: g.3gpp.atcf-path

Summary of the feature indicated by this feature-capability indicator:

This feature-capability indicator when used in a Feature-Caps header field as specified in IETF RFC 6809 [60] in SIP REGISTER request indicates capability of identifying the registration path and binding SRVCC related information to it.

Feature-capability indicator specification reference:

3GPP TS 24.237, http://www.3gpp.org/ftp/Specs/archive/24_series/24.237/

Values appropriate for use with this feature-capability indicator:

String with an equality relationship. When used in a Feature-Caps header field, the value is a SIP URI of ATCF, the ATCF URI for terminating requests, identifying the registration path following the syntax as described in table C.8-1 for g-3gpp-atcf-path.

Table C.8-1: ABNF syntax of values of the g.3gpp.atcf-path feature-capability indicator

```
g-3gpp-atcf-path = "<" SIP-URI ">"
```

The feature-capability indicator is intended primarily for use in the following applications, protocols, services, or negotiation mechanisms: This feature-capability indicator is used in access transfer control function of single radio voice call continuity to identify registration path so that SCC AS can provided the SRVCC related information related to the registration path.

Examples of typical use: Indicating capability of identifying a registration path and binding SRVCC related information to it.

Security Considerations: Security considerations for this feature-capability indicator are discussed in clause 9 of IETF RFC 6809 [60].

C.9 Definition of media feature tag g.3gpp.cs2ps-srvcc

Media feature tag name: g.3gpp.cs2ps-srvcc

ASN.1 Identifier: 1.3.6.1.8.2.16

Summary of the media feature indicated by this tag:

This media feature tag when included in Contact header field indicates support of the CS to PS single radio voice call continuity as specified in 3GPP TS 24.237.

Values appropriate for use with this feature tag: Boolean

The feature tag is intended primarily for use in the following applications, protocols, services, or negotiation mechanisms: This feature tag is most useful in a communications application for indicating that a resource supports the CS to PS single radio voice call continuity.

Examples of typical use: Indicating that a resource supports the CS to PS single radio voice call continuity.

Related standards or documents: 3GPP TS 24.237: "IP Multimedia Subsystem (IMS) Service Continuity; Stage 3"

Security Considerations: Security considerations for this media feature tag are discussed in subclause 12.1 of IETF RFC 3840 [53].

C.9A Definition of feature-capability indicator g.3gpp.cs2ps-srvcc

Feature-capability indicator name: g.3gpp.cs2ps-srvcc

Summary of the feature indicated by this feature-capability indicator:

This feature-capability indicator when included in Feature-Caps header field as specified in IETF RFC 6809 [60] indicates support of the CS to PS single radio voice call continuity as specified in 3GPP TS 24.237.

Feature-capability indicator specification reference:

3GPP TS 24.237, http://www.3gpp.org/ftp/Specs/archive/24_series/24.237/

Values appropriate for use with this feature-capability indicator:

none or string with an equality relationship

When the value is string, the value contains the session transfer identifier for CS to PS SRVCC and follows the syntax as described in table C.9A-1 for g.3gpp.cs2ps-srvcc-value.

Table C.9A-1: ABNF syntax of string values of the g.3gpp.cs2ps-srvcc feature-capability indicator

```
g.3gpp.cs2ps-srvcc-value = STI-rSR
STI-rSR = "<" addr-spec ">"
```

Examples of typical use: Indicating that a resource supports the CS to PS single radio voice call continuity.

Security Considerations: Security considerations for this feature-capability indicator are discussed in clause 9 of IETF RFC 6809 [60].

C.10 Void

C.11 Definition of media feature tag g.3gpp.path

Media feature-tag name: g.3gpp.path

ASN.1 Identifier: 1.3.6.1.8.2.15

Summary of the media feature indicated by this tag:

This media feature tag when used in a Contact header field in SIP REGISTER request or SIP response to the SIP REGISTER request indicates capability of identifying the registration path and binding SRVCC related information to it.

Values appropriate for use with this feature-tag:

String with an equality relationship. The value is the URI for terminating requests, identifying the registration path, and follows the syntax described in table C.11-1 for g-3gpp-path-value.

Table C.11-1: ABNF syntax of string values of the g.3gpp.path media feature tag

```
g-3gpp-path-value = SIP-URI
```

The feature-tag is intended primarily for use in the following applications, protocols, services, or negotiation mechanisms: This feature-tag is used in MSC server to identify registration path so that SCC AS can provided the SRVCC related information related to the registration path.

Examples of typical use: Indicating capability of identifying a registration path and binding SRVCC related information to it.

Related standards or documents: 3GPP TS 24.237: "IP Multimedia Subsystem (IMS) Service Continuity; Stage 3"

Security Considerations: Security considerations for this media feature-tag are discussed in subclause 12.1 of IETF RFC 3840 [53].

C.12 Definition of media feature tag g.3gpp.ti

Media feature-tag name: g.3gpp.ti

ASN.1 Identifier: 1.3.6.1.8.2.18

Summary of the media feature indicated by this tag:

This media feature tag when used in a Contact header field as specified in IETF RFC 3840 [53] in SIP INVITE request or SIP response to the SIP INVITE request indicates the capability of associating a CS call with dialog created by the SIP INVITE request.

Values appropriate for use with this feature-tag:

String with an equality relationship. The value is the transaction identifier specified in figure 11.9 and table 11.3 of 3GPP TS 24.007 [75] encoded by hexadecimal digits. Each octet of the figure 11.9 of 3GPP TS 24.007 [75] is encoded by two hexadecimal digit, first of which represents the bits 8 - 5 of the octet. The transaction identifier flag is set as sent by the MSC server in CS signalling of the associated CS call.

The feature-tag is intended primarily for use in the following applications, protocols, services, or negotiation mechanisms: This feature-tag is used to indicate capability of associating a CS call with dialog created by the SIP INVITE request.

Examples of typical use: Indicating capability of associating a CS call with dialog created by the SIP INVITE request.

Related standards or documents: 3GPP TS 24.237: "IP Multimedia Subsystem (IMS) Service Continuity; Stage 3"

Security Considerations: Security considerations for this media feature tag are discussed in subclause 12.1 of IETF RFC 3840 [53].

C.12A Definition of feature-capability indicator g.3gpp.ti

Feature-capability indicator name: g.3gpp.ti

Summary of the feature indicated by this feature-capability indicator:

This feature-capability indicator when used in a Feature-Caps header field as specified in IETF RFC 6809 [60] in SIP INVITE request or SIP response to the SIP INVITE request indicates the capability of associating a CS call with dialog created by the SIP INVITE request.

Feature-capability indicator specification reference:

3GPP TS 24.237, http://www.3gpp.org/ftp/Specs/archive/24_series/24.237/

Values appropriate for use with this feature-capability indicator:

String with an equality relationship. The value is the transaction identifier specified in figure 11.9 and table 11.3 of 3GPP TS 24.007 [75] encoded by hexadecimal digits following the syntax as described in table C.12A-1 for g.3gpp.ti-value. Each octet of the figure 11.9 of 3GPP TS 24.007 [75] is encoded by two hexadecimal digit, first of which represents the bits 8 - 5 of the octet. The transaction identifier flag is set as sent by the MSC server in CS signalling of the associated CS call.

Table C.12A-1: ABNF syntax of string values of the g.3gpp.ti feature-capability indicator

```
g.3gpp.ti-value = "<" HEXDIG HEXDIG *( HEXDIG HEXDIG ) ">"
```

Examples of typical use: Indicating capability of associating a CS call with dialog created by the SIP INVITE request.

Security Considerations: Security considerations for this feature-capability indicator are discussed in clause 9 of IETF RFC 6809 [60].

C.13 Definition of media feature tag g.3gpp.cs2ps-srvcc-alerting

Media feature-tag name: g.3gpp.cs2ps-srvcc-alerting

ASN.1 Identifier: 1.3.6.1.8.2.14

Summary of the media feature indicated by this tag: This media feature-tag when used in a Contact header field of a SIP request or a SIP response indicates that the functional entity sending the SIP message supports CS to PS SRVCC for calls in alerting phase.

Values appropriate for use with this feature-tag: Boolean

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

Examples of typical use: Indicating that a user equipment supports the CS to PS SRVCC for calls in alerting phase.

Related standards or documents: 3GPP TS 24.237: "IP Multimedia Subsystem (IMS) Service Continuity; Stage 3"

Security Considerations: Security considerations for this media feature-tag are discussed in subclause 12.1 of IETF RFC 3840 [34].

C.14 Definition of media feature tag g.3gpp.pstops-sti

Media feature-tag name: g.3gpp.pstops-sti

ASN.1 Identifier: 1.3.6.1.8.2.17

Summary of the media feature indicated by this tag: This media feature-tag when used in a Contact header field of a SIP request or a SIP response indicates that the functional entity sending the SIP message supports the use of the PS to PS STI URI.

Values appropriate for use with this feature-tag: Boolean

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

Examples of typical use: Indicating that the user equipment supports the use of the PS to PS STI URI for access transfer.

Related standards or documents: 3GPP TS 24.237: "IP Multimedia Subsystem (IMS) Service Continuity; Stage 3"

Security Considerations: Security considerations for this media feature-tag are discussed in subclause 12.1 of IETF RFC 3840 [34].

C.15 Definition of media feature tag g.3gpp.drvcc-alerting

Media feature-tag name: g.3gpp.drvcc-alerting

ASN.1 Identifier: 1.3.6.1.8.2.19

The g.3gpp.drvcc-alerting media feature tag shall be registered in the IANA Media Feature Tags registry "features.global-tree" sub-registry.

Summary of the media feature indicated by this tag: This media feature-tag when used in a Contact header field of a SIP request or a SIP response indicates that the functional entity sending the SIP message supports the PS to CS dual radio access transfer for calls in alerting phase.

Values appropriate for use with this feature-tag: Boolean

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

Examples of typical use: Indicating that the user equipment supports the PS to CS dual radio access transfer for calls in alerting phase.

Related standards or documents: 3GPP TS 24.237: "IP Multimedia Subsystem (IMS) Service Continuity; Stage 3"

Security Considerations: Security considerations for this media feature-tag are discussed in subclause 12.1 of IETF RFC 3840 [34].

C.16 Definition of feature-capability indicator g.3gpp.drvcc-alerting

Feature-capability indicator name: g.3gpp.drvcc-alerting

The g.3gpp.drvcc-alerting feature-capability indicator shall be registered in the IANA Proxy-Feature Feature-Capability Indicator Trees registry "Global Feature-Capability Indicator Registration Tree" sub-registry.

Summary of the feature indicated by this feature-capability indicator:

This feature-capability indicator when included in a Feature-Caps header field as specified in IETF RFC 6809 [60] in a SIP INVITE request or a SIP response to the SIP INVITE request indicates support of PS to CS dual radio access transfer for calls in alerting phase.

Feature-capability indicator specification reference:

3GPP TS 24.237, http://www.3gpp.org/ftp/Specs/archive/24_series/24.237/

Values appropriate for use with this feature-capability indicator: None.

Examples of typical use: Indicating the support of PS to CS dual radio access transfer for calls in alerting phase in the SIP INVITE request and SIP response to the SIP INVITE request.

Security Considerations: Security considerations for this feature-capability indicator are discussed in clause 9 of IETF RFC 6809 [60].

C.17 Definition of feature-capability indicator g.3gpp.dynamic-stn

Feature-capability indicator name: g.3gpp.dynamic-stn

Summary of the feature indicated by this feature-capability indicator:

This feature-capability indicator, when included in a Feature-Caps header field as specified in IETF RFC 6809 [60] in a SIP INVITE request or a SIP response to the SIP INVITE request, indicates support to transfer the session to the circuit switched (CS) domain using the dynamic STN (session transfer number) digit string.

Feature-capability indicator specification reference:

3GPP TS 24.237, http://www.3gpp.org/ftp/Specs/archive/24_series/24.237/

Values appropriate for use with this feature-capability indicator:

String with an equality relationship. When used in a Feature-Caps header field in SIP INVITE request or SIP response, the value is string containing the digit string in tel URI format of the dynamic STN to be used to establish a CS call in order to transfer the session to the CS domain following the syntax as described in table C.17-1 for g-3gpp-dynamic-stn.

Table C.17-1: ABNF syntax of values of the g.3gpp.dynamic-stn feature-capability indicator

```
g-3gpp-dynamic-stn = TEL-URI
TEL-URI = "<tel:" global-number-digits ">" ; global-number-digits as specified in IETF RFC 3966[79]
```

The feature-capability indicator is intended primarily for use in the following applications, protocols, services, or negotiation mechanisms: This feature-capability indicator is used to indicate support to transfer the session to the CS domain using the digit string of the dynamic STN.

Examples of typical use: Indicating the support to transfer the session to the CS domain in the SIP INVITE request and SIP response to the SIP INVITE request and providing the dynamic STN digit string to be used to establish the CS call instead of using the static STN to establish the CS call for the session transfer.

Security Considerations: Security considerations for this feature-capability indicator are discussed in clause 9 of IETF RFC 6809 [60].

C.18 Definition of media feature tag g.3gpp.dynamic-stn

Media feature-tag name: g.3gpp.dynamic-stn

ASN.1 Identifier: 1.3.6.1.8.2.20

The g.3gpp.dynamic-stn media feature tag shall be registered in the IANA Media Feature Tags registry "features.global-tree" sub-registry.

Summary of the media feature indicated by this tag: This media feature-tag when used in a Contact header field of a SIP request or a SIP response indicates that the functional entity sending the SIP message supports the use of dynamic STN.

Values appropriate for use with this feature-tag: Boolean

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

Examples of typical use: Indicating that the user equipment supports the use of dynamic STN.

Related standards or documents: 3GPP TS 24.237: "IP Multimedia Subsystem (IMS) Service Continuity; Stage 3"

Security Considerations: Security considerations for this media feature-tag are discussed in subclause 12.1 of IETF RFC 3840 [34].

C.19 Definition of media feature tag g.3gpp.ps2cs-drvcc-orig-pre-alerting

Media feature-tag name: g.3gpp.ps2cs-drvcc-orig-pre-alerting

ASN.1 Identifier: 1.3.6.1.8.2.21

The g.3gpp.ps2cs-drvcc-orig-pre-alerting media feature tag shall be registered in the IANA Media Feature Tags registry "features.global-tree" sub-registry.

Summary of the media feature indicated by this tag: This media feature-tag when used in a Contact header field of a SIP request or a SIP response indicates that the functional entity sending the SIP message supports the PS to CS dual radio access transfer for originating calls in pre-alerting phase.

Values appropriate for use with this feature-tag: Boolean

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

Examples of typical use: Indicating that the user equipment supports the PS to CS dual radio access transfer for originating calls in pre-alerting phase.

Related standards or documents: 3GPP TS 24.237: "IP Multimedia Subsystem (IMS) Service Continuity; Stage 3"

Security Considerations: Security considerations for this media feature-tag are discussed in subclause 12.1 of IETF RFC 3840 [34].

C.20 Definition of feature-capability indicator g.3gpp.ps2cs-drvcc-orig-pre-alerting

Feature-capability indicator name: g.3gpp.ps2cs-drvcc-orig-pre-alerting

The g.3gpp.ps2cs-drvcc-orig-pre-alerting feature-capability indicator shall be registered in the IANA Proxy-Feature Feature-Capability Indicator Trees registry "Global Feature-Capability Indicator Registration Tree" sub-registry.

Summary of the feature indicated by this feature-capability indicator:

This feature-capability indicator when included in a Feature-Caps header field as specified in IETF RFC 6809 [60] in a SIP INVITE request or a SIP response to the SIP INVITE request indicates support of PS to CS dual radio access transfer for originating calls in pre-alerting phase.

Feature-capability indicator specification reference:

3GPP TS 24.237, http://www.3gpp.org/ftp/Specs/archive/24_series/24.237/

Values appropriate for use with this feature-capability indicator: None.

Examples of typical use: Indicating the support of PS to CS dual radio access transfer for originating calls in pre-alerting phase in the SIP INVITE request and SIP response to the SIP INVITE request.

Security Considerations: Security considerations for this feature-capability indicator are discussed in clause 9 of IETF RFC 6809 [60].

C.21 Definition of media feature tag g.3gpp.ps2cs-srvcc-orig-pre-alerting

Media feature tag name: g.3gpp.ps2cs-srvcc-orig-pre-alerting

ASN.1 Identifier: 1.3.6.1.8.2.22

Summary of the media feature indicated by this tag: This media feature tag when used in a Contact header field of a SIP request or a SIP response indicates that the functional entity sending the SIP message supports the PS to CS SRVCC for originating calls in pre-alerting phase.

Values appropriate for use with this feature-tag: Boolean

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

Examples of typical use: Indicating that a user equipment supports the PS to CS SRVCC for originating calls in pre-alerting phase.

Related standards or documents: 3GPP TS 24.237: "IP Multimedia Subsystem (IMS) Service Continuity; Stage 3"

Security Considerations: Security considerations for this media feature-tag are discussed in subclause 12.1 of IETF RFC 3840 [53].

C.22 Definition of feature-capability indicator g.3gpp.ps2cs-srvcc-orig-pre-alerting

Feature-capability indicator name: g.3gpp.ps2cs-srvcc-orig-pre-alerting

Summary of the feature indicated by this feature-capability indicator:

This feature-capability indicator when used in a Feature-Caps header field of a SIP request or a SIP response indicates that:

1. the functional entity including the feature-capability indicator in the SIP message supports the PS to CS SRVCC for originating calls in pre-alerting phase; and

2. all entities of which the functional entity including the feature-capability indicator in the SIP message is aware of being requested to support the feature do support the PS to CS SRVCC for originating calls in pre-alerting phase.

Feature-capability indicator specification reference:

3GPP TS 24.237, http://www.3gpp.org/ftp/Specs/archive/24_series/24.237/

Values appropriate for use with this feature-capability indicator: None

Examples of typical use: Indicating that an application server supports the PS to CS SRVCC for originating calls in pre-alerting phase.

Security Considerations: Security considerations for this feature-capability indicator are discussed in subclause 9 of RFC 6809 [60].

C.23 Definition of media feature tag g.3gpp.cs2ps-drvcc-alerting

Media feature-tag name: g.3gpp.cs2ps-drvcc-alerting

ASN.1 Identifier: 1.3.6.1.8.2.23

The g.3gpp.cs2ps-drvcc-alerting media feature tag shall be registered in the IANA Media Feature Tags registry "features.global-tree" sub-registry.

Summary of the media feature indicated by this tag: This media feature-tag, when used in a Contact header field of a SIP request or a SIP response, indicates that the functional entity sending the SIP message supports the CS to PS dual radio access transfer for calls in alerting phase.

Values appropriate for use with this feature-tag: Boolean

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

Examples of typical use: Indicating that the user equipment supports the CS to PS dual radio access transfer for calls in alerting phase.

Related standards or documents: 3GPP TS 24.237: "IP Multimedia Subsystem (IMS) Service Continuity; Stage 3"

Security Considerations: Security considerations for this media feature-tag are discussed in subclause 12.1 of IETF RFC 3840 [34].

C.24 Definition of feature-capability indicator g.3gpp.cs2ps-drvcc-alerting

Feature-capability indicator name: g.3gpp.cs2ps-drvcc-alerting

The g.3gpp.cs2ps-drvcc-alerting feature-capability indicator shall be registered in the IANA Proxy-Feature Feature-Capability Indicator Trees registry "Global Feature-Capability Indicator Registration Tree" sub-registry.

Summary of the feature indicated by this feature-capability indicator:

This feature-capability indicator, when included in a Feature-Caps header field as specified in IETF RFC 6809 [60] in a SIP request or a SIP response to the SIP request, indicates support of CS to PS dual radio access transfer for calls in alerting phase.

Feature-capability indicator specification reference:

3GPP TS 24.237, http://www.3gpp.org/ftp/Specs/archive/24_series/24.237/

Values appropriate for use with this feature-capability indicator: None.

Examples of typical use: Indicating the support of CS to PS dual radio access transfer for calls in alerting phase in the SIP INVITE request and SIP response to the SIP INVITE request.

Security Considerations: Security considerations for this feature-capability indicator are discussed in clause 9 of IETF RFC 6809 [60].

C.25 Definition of media feature tag g.3gpp.cs2ps-drvcc-orig-pre-alerting

Media feature-tag name: g.3gpp.cs2ps-drvcc-orig-pre-alerting

ASN.1 Identifier: 1.3.6.1.8.2.24

The g.3gpp.cs2ps-drvcc-orig-pre-alerting media feature tag shall be registered in the IANA Media Feature Tags registry "features.global-tree" sub-registry.

Summary of the media feature indicated by this tag: This media feature-tag when used in a Contact header field of a SIP request or a SIP response indicates that the functional entity sending the SIP message supports the CS to PS dual radio access transfer for originating calls in pre-alerting phase.

Values appropriate for use with this feature-tag: Boolean

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

Examples of typical use: Indicating that the user equipment supports the CS to PS dual radio access transfer access transfer for originating calls in pre-alerting phase.

Related standards or documents: 3GPP TS 24.237: "IP Multimedia Subsystem (IMS) Service Continuity; Stage 3"

Security Considerations: Security considerations for this media feature-tag are discussed in subclause 12.1 of IETF RFC 3840 [34].

C.26 Definition of feature-capability indicator g.3gpp.cs2ps-drvcc-orig-pre-alerting

Feature-capability indicator name: g.3gpp.cs2ps-drvcc-orig-pre-alerting

The g.3gpp.cs2ps-drvcc-orig-pre-alerting feature-capability indicator shall be registered in the IANA Proxy-Feature Feature-Capability Indicator Trees registry "Global Feature-Capability Indicator Registration Tree" sub-registry.

Summary of the feature indicated by this feature-capability indicator:

This feature-capability indicator when included in a Feature-Caps header field as specified in IETF RFC 6809 [60] in a SIP request or a SIP response to the SIP request indicates support of CS to PS dual radio access transfer for originating calls in pre-alerting phase.

Feature-capability indicator specification reference:

3GPP TS 24.237, http://www.3gpp.org/ftp/Specs/archive/24_series/24.237/

Values appropriate for use with this feature-capability indicator: None.

Examples of typical use: Indicating the support of CS to PS dual radio access transfer for originating calls in pre-alerting phase in the SIP INVITE request and SIP response to the SIP INVITE request.

Security Considerations: Security considerations for this feature-capability indicator are discussed in clause 9 of IETF RFC 6809 [60].

C.27 Definition of feature-capability indicator g.3gpp.remote-leg-info

Feature-capability indicator name: g.3gpp.remote-leg-info

Summary of the feature indicated by this feature-capability indicator:

This feature-capability indicator when used in a Feature-Caps header field of a SIP request or a SIP response indicates that the functional entity including the feature-capability indicator in the SIP message supports providing remote leg information within a dialog of the SIP request or SIP response.

Feature-capability indicator specification reference:

3GPP TS 24.237, http://www.3gpp.org/ftp/Specs/archive/24_series/24.237/

Values appropriate for use with this feature-capability indicator: none

Examples of typical use: Indicating that an an application server supports providing remote leg information within a dialog of the SIP request or SIP response.

Security Considerations: Security considerations for this feature-capability indicator are discussed in clause 9 of IETF RFC 6809 [60].

C.28 Definition of feature-capability indicator g.3gpp.dynamic-e-stn-drvcc

Feature-capability indicator name: g.3gpp.dynamic-e-stn-drvcc

Summary of the feature indicated by this feature-capability indicator:

This feature-capability indicator, when included in a Feature-Caps header field as specified in IETF RFC 6809 [60] in a SIP response to a SIP INVITE request, indicates support to transfer the session to the circuit switched (CS) domain using the Emergency Session Transfer Number for DRVCC digit string.

Feature-capability indicator specification reference:

3GPP TS 24.237, http://www.3gpp.org/ftp/Specs/archive/24_series/24.237/

Values appropriate for use with this feature-capability indicator:

String with an equality relationship. When used in a Feature-Caps header field in a SIP response, the value is string containing the digit string in tel URI format of the Emergency Session Transfer Number for DRVCC to be used to establish a CS call in order to transfer the session to the CS domain following the syntax as described in table C.28-1 for g-3gpp-dynamic- e-stn-drvcc.

Table C.28-1: ABNF syntax of values of the g.3gpp. dynamic-e-stn-drvcc feature-capability indicator

```
g-3gpp-dynamic-e-stn-drvcc = TEL-URI
TEL-URI = "<tel:" global-number-digits ">" ; global-number-digits as specified in IETF RFC 3966[79]
```

The feature-capability indicator is intended primarily for use in the following applications, protocols, services, or negotiation mechanisms: This feature-capability indicator is used to indicate support to transfer the session to the CS domain using the digit string of the Emergency Session Transfer Number for DRVCC.

Examples of typical use: Indicating the support to transfer the session to the CS domain in a SIP response to the SIP INVITE request and providing the dynamic Emergency Session Transfer Number for DRVCC digit string to be used to establish the CS call.

Security Considerations: Security considerations for this feature-capability indicator are discussed in clause 9 of IETF RFC 6809 [60].

C.29 Definition of media feature tag g.3gpp.dynamic-e-stn-drvcc

Media feature-tag name: g.3gpp. dynamic-e-stn-drvcc

ASN.1 Identifier: 1.3.6.1.8.2.32

The g.3gpp.dynamic-stn media feature tag shall be registered in the IANA Media Feature Tags registry "features.global-tree" sub-registry.

Summary of the media feature indicated by this tag: This media feature-tag when used in a Contact header field of a SIP request or a SIP response indicates that the functional entity sending the SIP message supports the use of dynamic-e-stn-drvcc.

Values appropriate for use with this feature-tag: Boolean

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

Examples of typical use: Indicating that the user equipment supports the use of dynamic STN.

Related standards or documents: 3GPP TS 24.237: "IP Multimedia Subsystem (IMS) Service Continuity; Stage 3"

Security Considerations: Security considerations for this media feature-tag are discussed in subclause 12.1 of IETF RFC 3840 [34].

C.30 Definition of media feature tag g.3gpp.ps2cs-srvcc-mid-call-emergency

Media feature-tag name: g.3gpp.ps2cs-srvcc-mid-call-emergency

ASN.1 Identifier: 1.3.6.1.8.2.33

The g.3gpp.ps2cs-srvcc-mid-call-emergency media feature tag shall be registered in the IANA Media Feature Tags registry "features.global-tree" sub-registry.

Summary of the media feature indicated by this tag: This media feature-tag, when used in a Contact header field of a SIP request or a SIP response, indicates that the functional entity sending the SIP message supports the PS to CS SRVCC for emergency session in early dialog state with active speech media component when both the emergency session in early dialog state with active speech media component and a non-emergency call in confirmed dialog state with inactive speech media component exists.

Values appropriate for use with this feature-tag: Boolean

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

Examples of typical use: Indicating that the user equipment supports the PS to CS SRVCC for emergency session in early dialog state with active speech media component when both the emergency session in early dialog state with active speech media component and a non-emergency call in confirmed dialog state with inactive speech media component exists.

Related standards or documents: 3GPP TS 24.237: "IP Multimedia Subsystem (IMS) Service Continuity; Stage 3"

Security Considerations: Security considerations for this media feature-tag are discussed in subclause 12.1 of IETF RFC 3840 [34].

C.31 Definition of media feature tag g.3gpp.ps2cs-srvcc-term-pre-alerting

Media feature tag name: g.3gpp.ps2cs-srvcc-term-pre-alerting

ASN.1 Identifier: 1.3.6.1.8.2.34

Summary of the media feature indicated by this tag: This media feature tag when used in a Contact header field of a SIP request or a SIP response indicates that the functional entity sending the SIP message supports the PS to CS SRVCC for terminating calls in pre-alerting phase.

Values appropriate for use with this feature-tag: Boolean

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

Examples of typical use: Indicating that a user equipment supports the PS to CS SRVCC for terminating calls in pre-alerting phase.

Related standards or documents: 3GPP TS 24.237: "IP Multimedia Subsystem (IMS) Service Continuity; Stage 3"

Security Considerations: Security considerations for this media feature-tag are discussed in subclause 12.1 of IETF RFC 3840 [53].

C.32 Definition of feature-capability indicator g.3gpp.ps2cs-srvcc-term-pre-alerting

Feature-capability indicator name: g.3gpp.ps2cs-srvcc-term-pre-alerting

Summary of the feature indicated by this feature-capability indicator:

This feature-capability indicator when used in a Feature-Caps header field of a SIP request or a SIP response indicates that:

1. the functional entity including the feature-capability indicator in the SIP message supports the PS to CS SRVCC for terminating calls in pre-alerting phase; and

2. all entities of which the functional entity including the feature-capability indicator in the SIP message is aware of being requested to support the feature do support the PS to CS SRVCC for terminating calls in pre-alerting phase.

Feature-capability indicator specification reference:

3GPP TS 24.237, http://www.3gpp.org/ftp/Specs/archive/24_series/24.237/

Values appropriate for use with this feature-capability indicator: None

Examples of typical use: Indicating that an application server supports the PS to CS SRVCC for terminating calls in pre-alerting phase.

Security Considerations: Security considerations for this feature-capability indicator are discussed in subclause 9 of RFC 6809 [60].

Annex D (informative): XML schemas

D.1 MSC server assisted mid-call feature XML schema

D.1.1 General

This subclause defines XML schema and MIME type related to the MSC server assisted mid-call feature.

D.1.2 XML schema

```
<?xml version="1.0" encoding="UTF-8"?>
<xs:schema
  xmlns:xs="http://www.w3.org/2001/XMLSchema"
  elementFormDefault="qualified"
  attributeFormDefault="unqualified">

  <xs:element name="mid-call" type="Tmid-call"/>

  <xs:complexType name="Tmid-call">
    <xs:sequence>
      <xs:element name="participant" type="xs:anyURI" minOccurs="0" maxOccurs="unbounded"/>
      <xs:any namespace="##other" processContents="lax" minOccurs="0" maxOccurs="unbounded"/>
    </xs:sequence>
    <xs:anyAttribute namespace="##any" processContents="lax"/>
  </xs:complexType>

</xs:schema>
```

D.1.3 IANA registration template

Your Name:

<MCC name>

Your Email Address:

<MCC email address>

Media Type Name:

Application

Subtype name:

vnd.3gpp.mid-call+xml

Required parameters:

None

Optional parameters:

"charset" the parameter has identical semantics to the charset parameter of the "application/xml" media type as specified in section 9.1 of IETF RFC 7303.

Encoding considerations:

binary.

Security considerations:

Same as general security considerations for application/xml media type as specified in section 9.1 of IETF RFC 7303. In addition, this media type provides a format for exchanging information in SIP, so the security considerations from IETF RFC 3261apply.

The information transported in this media type does not include active or executable content.

Mechanisms for privacy and integrity protection of protocol parameters exist. Those mechanisms as well as authentication and further security mechanisms are described in 3GPP TS 24.229.

This media type does not include provisions for directives that institute actions on a recipient's files or other resources.

This media type does not include provisions for directives that institute actions that, while not directly harmful to the recipient, may result in disclosure of information that either facilitates a subsequent attack or else violates a recipient's privacy in any way.

This media type does not employ compression.

Interoperability considerations:

Same as general interoperability considerations for application/xml media type as specified in section 9.1 of IETF RFC 7303.

Published specification:

3GPP TS 24.237 "IP Multimedia Subsystem (IMS) Service Continuity", version 13.1.0, available via http://www.3gpp.org/specs/numbering.htm.

Applications which use this media type:

Applications supporting the service continuity as described in the published specification.

Fragment identifier considerations:

The handling in section 5 of IETF RFC 7303 applies.

Restrictions on usage:

None

Provisional registration? (standards tree only):

N/A

Additional information:

1. Deprecated alias names for this type: none
2. Magic number(s): none
3. File extension(s): none
4. Macintosh File Type Code(s): none
5. Object Identifier(s) or OID(s): none

Intended usage:

Common

Person to contact for further information:

- Name: <MCC name>
- Email: <MCC email address>
- Author/Change controller:
  - i) Author: 3GPP CT1 Working Group/3GPP_TSG_CT_WG1@LIST.ETSI.ORG
  - ii) Change controller: <MCC name>/<MCC email address>

D.2 state-and-event-info XML schema

D.2.1 General

This subclause defines XML schema and MIME type for session state and event information. It is used in the present document for the PS to CS SRVCC for calls in alerting phase, for accepting of a call in alerting state transferred by the PS-PS access transfer procedures, for the PS to CS SRVCC for originating calls in pre-alerting phase, for PS to CS SRVCC for terminating calls in pre-alerting phase and for requesting and providing remote leg information.

D.2.2 XML schema

```
<?xml version="1.0" encoding="UTF-8"?>
<xs:schema
  xmlns:xs="http://www.w3.org/2001/XMLSchema"
  elementFormDefault="qualified"
  attributeFormDefault="unqualified">


  <xs:simpleType name="directionType">
     <xs:restriction base="xs:string">
        <xs:enumeration value="initiator"/>
        <xs:enumeration value="receiver"/>
     </xs:restriction>
  </xs:simpleType>


  <xs:element name="state-and-event-info" type="Tstate-and-event-info"/>

  <xs:complexType name="Tstate-and-event-info">
    <xs:sequence>
      <xs:element name="state-info" type="xs:string" minOccurs="0" maxOccurs="1"/>
      <xs:element name="direction" type="directionType" minOccurs="0" maxOccurs="1"/>
      <xs:element name="event" type="xs:string" minOccurs="0" maxOccurs="1"/>
      <xs:element name="anyExt" type="anyExtType" minOccurs="0" />
      <xs:any namespace="##other" processContents="lax" minOccurs="0" maxOccurs="unbounded"/>
    </xs:sequence>
    <xs:anyAttribute namespace="##any" processContents="lax"/>
  </xs:complexType>

  <xs:complexType name="anyExtType">
    <xs:sequence>
      <xs:any namespace="##any" processContents="lax" minOccurs="0" maxOccurs="unbounded"/>
    </xs:sequence>
  </xs:complexType>

  <xs:element name="remoteLegInfoRequest" type="anyExtType"/>
  <xs:element name="remoteLegInfoResponse" type="anyExtType"/>

  <xs:complexType name="dialogIdType">
    <xs:sequence>
      <xs:any namespace="##any" processContents="lax" minOccurs="0" maxOccurs="unbounded"/>
    </xs:sequence>
    <xs:attribute name="call-id" type="xs:string"/>
    <xs:attribute name="local-tag" type="xs:string" use="optional"/>
    <xs:attribute name="remote-tag" type="xs:string" use="optional"/>
    <xs:anyAttribute namespace="##any" processContents="lax"/>
  </xs:complexType>

<!-- elements which can be included in remoteLegInfoRequest -->
  <xs:element name="localAssertedIdRequest"/>
  <xs:element name="dialogIdRequest"/>

<!-- elements which can be included in remoteLegInfoResponse-->  <xs:element name="localAssertedId"
type="xs:anyURI"/>
  <xs:element name="dialogId" type="dialogIdType"/>
</xs:schema>
```

D.2.3 XML schema description

This subclause describes the elements of the state-and-info XML Schema.

<state-and-event-info>: The <state-and-event-info> element is used to indicate state and event information related to a specific dialog. In the present document, it is used to communicate information between the SCC AS and the MSC-server for the purpose of PS to CS SRVCC for calls in alerting phase, for UE to communicate acceptance of incoming alerting phase call transferred using PS-PS access transfer procedures and communicate information between the SCC AS and the MSC server for the purpose of the PS to CS SRVCC for originating calls in pre-alerting phase.

<state-info>: The <state-info> element is used to indicate the state of the dialog and is modelled on the FSM described in IETF RFC 4235 [48]. In the present document, it can only have the values specified in table D.2.3-1 for state-info-values.

Table D.2.3-1: ABNF syntax of values of the <state-info> element

```
state-info-values = early-value / pre-alerting-value
early-value = %x65.61.72.6c.79 ; "early"
pre-alerting-value = %x70.72.65.2d.61.6c.65.72.74.69.6e.67 ; "pre-alerting"
```

<direction>: The <direction> element indicates whether the observed user was the initiator of the dialog, or the recipient of the INVITE that created it. It can only have the values specified in table D.2.3-2 for direction-values. In the present document it must be included together with the <state-info> element.

Table D.2.3-2: ABNF syntax of values of the <direction> element

```
direction-values = initiator-value / receiver-value
initiator-value = %x69.6e.69.74.69.61.74.6f.72 ; "initiator"
receiver-value = %x72.65.63.65.69.76.65.72 ; "receiver"
```

<event>: The <event> element is used to communicate an event that causes a dialog state transition. In the present document, the <event> element can only have the values specified in table D.2.3-3 for event-values.

Table D.2.3-3: ABNF syntax of values of the <event> element

```
event-values = call-accepted-value / alerting-started-value
call-accepted-value = %x63.61.6c.6c.2d.61.63.63.65.70.74.65.64 ; "call-accepted"
alerting-started-value = %x61.6c.65.72.74.69.6e.67.2d.73.74.61.72.74.65.64 ; "alerting-started"
```

<anyExt>: The <anyExt> element is used to convey extensions. In the present document, the <anyExt> element contains a <remoteLegInfoRequest> element or a <remoteLegInfo> element.

NOTE 1: The subelements of the <anyExt> are validated by the <xs:any namespace="##any" processContents="lax" minOccurs="0" maxOccurs="unbounded"/> particle of the <anyExt> element.

<remoteLegInfoRequest> The <remoteLegInfoRequest>element indicates request for the remote leg information. The <remoteLegInfoRequest> element contains the optional <localAssertedIdRequest> element and the optional <dialogIdRequest> element.

NOTE 2: The subelements of the <remoteLegInfoRequest> are validated by the <xs:any namespace="##any" processContents="lax" minOccurs="0" maxOccurs="unbounded"/> particle of the <anyExt> element.

<localAssertedIdRequest> The <localAssertedIdRequest> element indicates that SCC AS is to provide <localAssertedId> in the <remoteLegInfoResponse>.

<dialogIdRequest> The <dialogIdRequest> element indicates that SCC AS is to provide <dialogId> in the <remoteLegInfoResponse>.

<remoteLegInfoResponse> The <remoteLegInfoResponse> indicates the remote leg information. The <remoteLegInfoResponse> element contains the optional <localAssertedId> element and the optional <dialogId> element.

NOTE 3: The subelements of the <remoteLegInfoResponse> are validated by the <xs:any namespace="##any" processContents="lax" minOccurs="0" maxOccurs="unbounded"/> particle of the <anyExt> element.

<localAssertedId> The <localAssertedId> element contains URI provided by SCC AS in P-Asserted-Identity header field towards remote UE in remote leg associated with the dialog within which the SIP INFO request is sent.

<dialogId> The <dialogId> element contains dialog identifier of remote leg associated with the dialog within which the SIP INFO request is sent, i.e. the "call-id" attribute containing the call-id of the remote leg, the optional "local-tag" attribute containing tag that would be viewed as the local tag of the remote leg from the perspective of SCC AS and the optional "remote-tag" attribute containing tag that would be viewed as the remote tag of the remote leg from the perspective of SCC AS.

Recipient of the XML ignores any unknown element and any unknown attribute.

D.2.4 IANA registration template

Your Name:

<MCC name>

Your Email Address:

<MCC email address>

Media Type Name:

Application

Subtype name:

vnd.3gpp.state-and-event-info+xml

Required parameters:

None

Optional parameters:

"charset" the parameter has identical semantics to the charset parameter of the "application/xml" media type as specified in section 9.1 of IETF RFC 7303.

Encoding considerations:

binary.

Security considerations:

Same as general security considerations for application/xml media type as specified in section 9.1 of IETF RFC 7303. In addition, this media type provides a format for exchanging information in SIP, so the security considerations from IETF RFC 3261apply.

The information transported in this media type does not include active or executable content.

Mechanisms for privacy and integrity protection of protocol parameters exist. Those mechanisms as well as authentication and further security mechanisms are described in 3GPP TS 24.229.

This media type does not include provisions for directives that institute actions on a recipient's files or other resources.

This media type does not include provisions for directives that institute actions that, while not directly harmful to the recipient, may result in disclosure of information that either facilitates a subsequent attack or else violates a recipient's privacy in any way.

This media type does not employ compression.

Interoperability considerations:

Same as general interoperability considerations for application/xml media type as specified in section 9.1 of IETF RFC 7303.

Published specification:

3GPP TS 24.237 "IP Multimedia Subsystem (IMS) Service Continuity", version 13.1.0, available via http://www.3gpp.org/specs/numbering.htm.

Applications which use this media type:

Applications supporting the service continuity as described in the published specification.

Fragment identifier considerations:

The handling in section 5 of IETF RFC 7303 applies.

Restrictions on usage:

None

Provisional registration? (standards tree only):

N/A

Additional information:

1. Deprecated alias names for this type: none
2. Magic number(s): none
3. File extension(s): none
4. Macintosh File Type Code(s): none
5. Object Identifier(s) or OID(s): none

Intended usage:

Common

Person to contact for further information:

- Name: <MCC name>
- Email: <MCC email address>
- Author/Change controller:
  i) Author: 3GPP CT1 Working Group/3GPP_TSG_CT_WG1@LIST.ETSI.ORG
  ii) Change controller: <MCC name>/<MCC email address>

D.3 SRVCC enhancement related XML schema

D.3.1 General

This subclause defines XML schema and MIME type for transfer of information for SRVCC enhancement.

D.3.2 XML schema

```
<?xml version="1.0" encoding="UTF-8"?>
<xs:schema
  xmlns:xs="http://www.w3.org/2001/XMLSchema"
  elementFormDefault="qualified"
  attributeFormDefault="unqualified">

  <xs:complexType name="SRVCC-infoType">
    <xs:sequence>
      <xs:element name="ATU-STI" type="xs:anyURI" minOccurs="0"/>
      <xs:element name="C-MSISDN" type="xs:anyURI" minOccurs="0"/>
```

```
      <xs:element name="anyExt" type="anyExtType" minOccurs="0"/>
      <xs:any namespace="##other" processContents="lax" minOccurs="0" maxOccurs="unbounded"/>
    </xs:sequence>
    <xs:attribute name="ATCF-Path-URI" type="xs:anyURI"/>
    <xs:anyAttribute namespace="##any" processContents="lax"/>
  </xs:complexType>

  <xs:complexType name="SRVCC-infosType">
    <xs:sequence>
      <xs:element name="SRVCC-info" type="SRVCC-infoType"
    minOccurs="1" maxOccurs="unbounded"/>
      <xs:element name="anyExt" type="anyExtType" minOccurs="0"/>
      <xs:any namespace="##other" processContents="lax" minOccurs="0" maxOccurs="unbounded"/>
    </xs:sequence>
    <xs:anyAttribute namespace="##any" processContents="lax"/>
  </xs:complexType>


  <xs:element name="SRVCC-infos" type="SRVCC-infosType"/>

  <xs:complexType name="anyExtType">
    <xs:sequence>
      <xs:any namespace="##any" processContents="lax" minOccurs="0" maxOccurs="unbounded"/>
    </xs:sequence>
  </xs:complexType>

  <xs:element name="CS2PS-ATU-STI" type="xs:anyURI"/>

</xs:schema>
```

D.3.3 Semantic

The <SRVCC-infos> element is the root element of the XML document and contains one or more <SRVCC-info> elements.

Each <SRVCC-info> element contains SRVCC-related information related to one registration path (or registration flow, if multiple registration mechanism is used) of a UE with IM CN subsystem. The SRVCC-related information in the <SRVCC-info> element consists of:

1) if the UE PS to CS SRVCC Capability (see 3GPP TS 29.328 [6]) has value UE-SRVCC-CAPABILITY-SUPPORTED and if the private user identity of the UE has associated STN-SR (see 3GPP TS 29.328 [6]):

   a) <ATU-STI> element containing the ATU-STI for PS to CS SRVCC of the SCC AS; and

   b) <C-MSISDN> element containing the Correlation MSISDN of the UE.

NOTE 1: <ATU-STI> element and <C-MSISDN> element are not included if the UE PS to CS SRVCC Capability (see 3GPP TS 29.328 [6]) has a value UE-SRVCC-CAPABILITY-NOT-SUPPORTED or if the private user identity of the UE does not have an associated STN-SR (see 3GPP TS 29.328 [6]); and

2) if CS to PS SRVCC is supported by SCC AS and enabled for the UE:

   a) <anyExt> element containing <CS2PS-ATU-STI> element containing the ATU-STI for CS to PS SRVCC of the SCC AS; and

NOTE 2: The subelements of the <anyExt> are validated by the <xs:any namespace="##any" processContents="lax" minOccurs="0" maxOccurs="unbounded"/> particle of the <anyExt> element.

   b) <C-MSISDN> element containing the C-MSISDN of the UE.

NOTE 3: If information for both CS to PS SRVCC and PS to CS SRVCC are included, <C-MSISDN> element is included only once.

The "ATCF-Path-URI" attribute of the <SRVCC-info> element contains the ATCF URI for terminating calls of the registration path (or registration flow, if multiple registration mechanism is used).

<anyExt> element contains optional elements defined by future version of this document.

Recipient of the XML ignores any unknown element and any unknown attribute.

D.3.4 IANA registration template

Your Name:

<MCC name>

Your Email Address:

<MCC email address>

Media Type Name:

Application

Subtype name:

vnd.3gpp.SRVCC-info+xml

Required parameters:

None

Optional parameters:

"charset" the parameter has identical semantics to the charset parameter of the "application/xml" media type as specified in section 9.1 of IETF RFC 7303.

Encoding considerations:

binary.

Security considerations:

Same as general security considerations for application/xml media type as specified in section 9.1 of IETF RFC 7303. In addition, this media type provides a format for exchanging information in SIP, so the security considerations from IETF RFC 3261apply.

The information transported in this media type does not include active or executable content.

Mechanisms for privacy and integrity protection of protocol parameters exist. Those mechanisms as well as authentication and further security mechanisms are described in 3GPP TS 24.229.

This media type does not include provisions for directives that institute actions on a recipient's files or other resources.

This media type does not include provisions for directives that institute actions that, while not directly harmful to the recipient, may result in disclosure of information that either facilitates a subsequent attack or else violates a recipient's privacy in any way.

This media type does not employ compression.

Interoperability considerations:

Same as general interoperability considerations for application/xml media type as specified in section 9.1 of IETF RFC 7303. Any unknown XML elements and any unknown XML attributes are to be ignored by recipient of the MIME body.

Published specification:

3GPP TS 24.237 "IP Multimedia Subsystem (IMS) Service Continuity", version 13.1.0, available via http://www.3gpp.org/specs/numbering.htm.

Applications which use this media type:

Applications supporting the service continuity as described in the published specification.

Fragment identifier considerations:

The handling in section 5 of IETF RFC 7303 applies.

Restrictions on usage:

None

Provisional registration? (standards tree only):

N/A

Additional information:

1. Deprecated alias names for this type: none
2. Magic number(s): none
3. File extension(s): none
4. Macintosh File Type Code(s): none
5. Object Identifier(s) or OID(s): none

Intended usage:

Common

Person to contact for further information:

- Name: <MCC name>
- Email: <MCC email address>
- Author/Change controller:
  i) Author: 3GPP CT1 Working Group/3GPP_TSG_CT_WG1@LIST.ETSI.ORG
  ii) Change controller: <MCC name>/<MCC email address>

D.4 XML schema for CS to PS SRVCC information

D.4.1 General

This subclause defines XML schema and MIME type for transfer of CS to PS SRVCC information.

D.4.2 XML schema

```
<?xml version="1.0"?>
<xs:schema
  xmlns:xs="http://www.w3.org/2001/XMLSchema"
  elementFormDefault="qualified"
  attributeFormDefault="unqualified">

  <xs:element name="srvcc-ext" type="srvcc-extType"/>

  <xs:complexType name="srvcc-extType">
    <xs:sequence>
      <xs:any namespace="##any" processContents="lax" minOccurs="0" maxOccurs="unbounded"/>
    </xs:sequence>
    <xs:anyAttribute namespace="##any" processContents="lax"/>
  </xs:complexType>

  <xs:element name="PS-reg-info" type="PS-reg-infoType"/>

  <xs:complexType name="PS-reg-infoType">
    <xs:sequence>
      <xs:element name="ATCF-Management-URI" type="xs:anyURI" minOccurs="0"/>
      <xs:element name="C-MSISDN" type="xs:anyURI" minOccurs="0"/>
      <xs:element name="cs2ps-srvcc-alerting" type="xs:boolean" minOccurs="0"/>
      <xs:element name="anyExt" type="anyExtType" minOccurs="0"/>
```

```
      <xs:any namespace="##other" processContents="lax" minOccurs="0" maxOccurs="unbounded"/>
    </xs:sequence>
    <xs:attribute name="Path" type="xs:anyURI" use="optional"/>
    <xs:anyAttribute namespace="##any" processContents="lax"/>
  </xs:complexType>

  <xs:element name="Setup-info" type="Setup-infoType"/>

  <xs:complexType name="Setup-infoType">
    <xs:sequence>
      <xs:element name="C-MSISDN" type="xs:anyURI"/>
      <xs:element name="direction" type="directionType"/>
      <xs:any namespace="##any" processContents="lax" minOccurs="0" maxOccurs="unbounded"/>
    </xs:sequence>
    <xs:anyAttribute namespace="##any" processContents="lax"/>
  </xs:complexType>

  <xs:simpleType name="directionType">
     <xs:restriction base="xs:string">
        <xs:enumeration value="initiator"/>
        <xs:enumeration value="receiver"/>
     </xs:restriction>
  </xs:simpleType>

  <xs:complexType name="anyExtType">
    <xs:sequence>
      <xs:any namespace="##any" processContents="lax" minOccurs="0" maxOccurs="unbounded"/>
    </xs:sequence>
  </xs:complexType>

</xs:schema>
```

D.4.3 Semantic

D.4.3.1 General

The <srvcc-ext> element is the root element of the XML document. The <srvcc-ext> element can contain subelements.

NOTE: The subelements of the <srvcc-ext> are validated by the <xs:any namespace="##any" processContents="lax" minOccurs="0" maxOccurs="unbounded"/> particle of the <srvcc-ext> element.

When the <srvcc-ext> element contains the <PS-reg-info> element, then the <PS-reg-info> element contains:

- if the information relates to a registration of MSC server with IMS, then the "Path" attribute identifying the MSC URI for terminating calls used by MSC server when registering the UE;
- optionally the <ATCF-Management-URI> element containing the ATCF management URI of the ATCF used in the registration path of the UE over a PS domain. Absence of this element indicates that the CS to PS SRVCC is disabled;
- optionally the <C-MSISDN> element containing the C-MSISDN of the served SC UE. Absence of this element indicates that the CS to PS SRVCC is disabled;
- the <cs2ps-srvcc-alerting> element indicates whether the CS to PS SRVCC of calls in alerting phase is supported by both the served SC UE and the SCC AS; and
- optionally, <anyExt> element containing elements defined by future version of this document.

When the <srvcc-ext> element contains the <Setup-info> element, then the <Setup-info> element contains:

- the <C-MSISDN> element containing the C-MSISDN of the served SC UE;
- the <direction> element indicating the role of the MSC server in the session set up.

Recipient of the XML ignores any unknown element and any unknown attribute.

D.4.4 IANA registration template

Your Name:

<MCC name>

Your Email Address:

<MCC email address>

Media Type Name:

Application

Subtype name:

vnd.3gpp.srvcc-ext+xml

Required parameters:

None

Optional parameters:

"charset" the parameter has identical semantics to the charset parameter of the "application/xml" media type as specified in section 9.1 of IETF RFC 7303.

Encoding considerations:

binary.

Security considerations:

Same as general security considerations for application/xml media type as specified in section 9.1 of IETF RFC 7303. In addition, this media type provides a format for exchanging information in SIP, so the security considerations from IETF RFC 3261apply.

The information transported in this media type does not include active or executable content.

Mechanisms for privacy and integrity protection of protocol parameters exist. Those mechanisms as well as authentication and further security mechanisms are described in 3GPP TS 24.229.

This media type does not include provisions for directives that institute actions on a recipient's files or other resources.

This media type does not include provisions for directives that institute actions that, while not directly harmful to the recipient, may result in disclosure of information that either facilitates a subsequent attack or else violates a recipient's privacy in any way.

This media type does not employ compression.

Interoperability considerations:

Same as general interoperability considerations for application/xml media type as specified in section 9.1 of IETF RFC 7303. Any unknown XML elements and any unknown XML attributes are to be ignored by recipient of the MIME body.

Published specification:

3GPP TS 24.237 "IP Multimedia Subsystem (IMS) Service Continuity", version 13.1.0, available via http://www.3gpp.org/specs/numbering.htm.

Applications which use this media type:

Applications supporting the service continuity as described in the published specification.

Fragment identifier considerations:

The handling in section 5 of IETF RFC 7303 applies.

Restrictions on usage:

None

Provisional registration? (standards tree only):

N/A

Additional information:

1. Deprecated alias names for this type: none
2. Magic number(s): none
3. File extension(s): none
4. Macintosh File Type Code(s): none
5. Object Identifier(s) or OID(s): none

Intended usage:

Common

Person to contact for further information:

- Name: <MCC name>
- Email: <MCC email address>
- Author/Change controller:
  i) Author: 3GPP CT1 Working Group/3GPP_TSG_CT_WG1@LIST.ETSI.ORG
  ii) Change controller: <MCC name>/<MCC email address>

D.5 XML schema for access transfer events

D.5.1 General

This subclause defines XML schema and MIME type for transport of events related to access transfer of a session.

D.5.2 XML schema

```
<?xml version="1.0"?>
<xs:schema
  xmlns:xs="http://www.w3.org/2001/XMLSchema"
  elementFormDefault="qualified"
  attributeFormDefault="unqualified">

  <xs:element name="events" type="eventsType"/>

  <xs:complexType name="eventsType">
    <xs:sequence>
      <xs:element name="event" type="eventType" maxOccurs="unbounded"/>
    </xs:sequence>
    <xs:anyAttribute namespace="##any" processContents="lax"/>
  </xs:complexType>

  <xs:complexType name="eventType">
    <xs:sequence>
      <xs:any namespace="##any" processContents="lax" minOccurs="0" maxOccurs="unbounded"/>
    </xs:sequence>
    <xs:attribute name="event-type" type="xs:unsignedInt"/>
    <xs:anyAttribute namespace="##any" processContents="lax"/>
  </xs:complexType>

  <xs:element name="STNResp-params" type="STNResp-paramsType"/>

  <xs:complexType name="STNResp-paramsType">
    <xs:sequence>
      <xs:element name="transfer-details" type="xs:base64Binary"/>
      <xs:element name="redirect-speech" type="xs:boolean"/>
      <xs:any namespace="##any" processContents="lax" minOccurs="0" maxOccurs="unbounded"/>
```

```
    </xs:sequence>
    <xs:anyAttribute namespace="##any" processContents="lax"/>
  </xs:complexType>


</xs:schema>
```

D.5.3 Semantic

D.5.3.1 General

The <events> element is the root element of the XML document and contains one or more <event> elements.

Each <event> element describes one event occuring in session and:

1) contains the "event-type" attribute which indicates the event type; and

2) can contain subelements related to the event type indicated by the "event-type" attribute.

NOTE: The subelements of the <event> are validated by the <xs:any namespace="##any" processContents="lax" minOccurs="0" maxOccurs="unbounded"/> particle of the <event> element.

The following applies for the "event-type" attribute of the <event> element:

- sender of the XML does not set the "event-type" attribute to any value other than those described in this section; and

- recipient of the XML ignores the <event> element with "event-type" attribute containing a value other than those described in this section.

Recipient of the XML ignores any unknown element and any unknown attribute.

D.5.3.2 Requirements for individual events

If the "event-type" attribute of the <event> element is 1, then the <event> element indicates the session transfer notification request.

If the "event-type" attribute of the <event> element is 2, then the <event> element indicates the session transfer notification response and the <event> element contains the <STNResp-params> element. The <STNResp-params> element contains <transfer-details> element and the <redirect-speech> element. The <transfer-details> element contains the content according to subclause D.5.3.3. The <redirect-speech> element indicates whether the ATCF requires the MSC server to redirect the speech media component of the session transferred by the CS to PS SRVCC access transfer.

NOTE 1: Binary data are encoded according to the XML type base64Binary.

NOTE 2: MIME body with the "event-type" attribute of the <event> element equal to 2 is provided after receiving the MIME body with the "event-type" attribute of the <event> element equal to 1.

If the "event-type" attribute of the <event> element is 3, then the <event> element indicates the session transfer preparation.

NOTE 3: MIME body with the "event-type" attribute of the <event> element equal to 3 is provided after receiving the MIME body with the "event-type" attribute of the <event> element equal to 2.

If the "event-type" attribute of the <event> element is 4, then the <event> element indicates the session transfer cancellation.

NOTE 4: MIME body with the "event-type" attribute of the <event> element equal to 4 is provided either after receiving the MIME body with the "event-type" attribute of the <event> element equal to 2 or after providing the MIME body with the "event-type" attribute of the <event> element equal to 3.

D.5.3.3 ATGW transfer details

The ATGW transfer details indicate the ATGW media configuration to be used by SC UE when sending the speech media of the session transferred by the CS to PS SRVCC access transfer.

The ATGW transfer details are encoded in the structure shown in the figure D.5.3.3-1 and table D.5.3.3-1.

8	7	6	5	4	3	2	1	
Reserved	Reserved	Reserved	Reserved	Reserved	Reserved	ATGW transfer details content type		octet 1
ATGW transfer details content								octet 2 octet v

Figure D.5.3.3-1: ATGW transfer details structure

Table D.5.3.3-1: ATGW transfer details structure

In this version of specification, the sender sets the *Reserved* field to zero and the recipient ignores the *Reserved* field. The *ATGW transfer details content type* field indicates type of the *ATGW transfer details content*. Bit 2 of the *ATGW transfer details content type* field contains the most significant bit. In this version of the specification, the following *ATGW transfer details content type* values are specified: - 0 (ATGW-IPv4-address-and-port); - 1 (ATGW-IPv6-address-and-port). - 2 (ATGW-not-available). When the *ATGW transfer details content type* field indicates value other than those specified in this version of the specification, the *ATGW Transfer details content* field is ignored by the recipient. The sender sets the *ATGW transfer details content type* field only to a value specified in this version of the specification. When the *ATGW transfer details content type* field indicates ATGW-IPv4-address-and-port, the *ATGW Transfer details content* field is structured as in figure D.5.3.3-2 and table D.5.3.3-2. When the *ATGW transfer details content type* field indicates ATGW-IPv6-address-and-port, the *ATGW Transfer details content* field is structured as in figure D.5.3.3-3 and table D.5.3.3-3. When the *ATGW transfer details content type* field indicates ATGW-not-available, the *ATGW Transfer details content* field is not included.

8	7	6	5	4	3	2	1	
ATGW UDP port								octet 2 octet 3
ATGW IPv4 address								octet 4 : octet 7
Extensions								octet 8 : octet v

Figure D.5.3.3-2: ATGW transfer details content structure when the ATGW transfer details content type field indicates ATGW-IPv4-address-and-port

Table D.5.3.3-2: ATGW transfer details content structure when the ATGW transfer details content type field indicates ATGW-IPv4-address-and-port

The *ATGW UDP port* field indicates the ATGW UDP port to be used by SC UE to send the speech media of the session transferred by the CS to PS SRVCC access transfer. Bit 8 of the first octet of the *ATGW UDP port* field contains the most significant bit and bit 1 of the second octet of the *ATGW UDP port* field contains the least significant bit. The *ATGW IPv4 address* field indicates the ATGW IPv4 address to be used by SC UE to send the speech media of the session transferred by the CS to PS SRVCC access transfer. Bit 8 of the first octet of the *ATGW IPv4 address* field contains the most significant bit and bit 1 of the fourth octet of the *ATGW IPv4 address* field contains the least significant bit. In this version of specification, the sender does not include the *Extensions* field and the recipient ignores the *Extensions* field. See NOTE.
NOTE: Inclusion of *Extensions* field enlarges the access stratum handover command, which can result in longer duration of handover procedure and in message being larger than maximum size of the access stratum messages. Thus, inclusion of *Extensions* field is not recommended.

8	7	6	5	4	3	2	1	
ATGW UDP port								octet 2 octet 3
ATGW IPv6 address								octet 4 : octet 19
Extensions								octet 20 : octet v

Figure D.5.3.3-3: ATGW transfer details content structure when the ATGW transfer details content type field indicates ATGW-IPv6-address-and-port

Table D.5.3.3-3: ATGW transfer details content structure when the ATGW transfer details content type field indicates ATGW-IPv6-address-and-port

The *ATGW UDP port* field indicates the ATGW UDP port to be used by SC UE to send the speech media of the session transferred by the CS to PS SRVCC access transfer. Bit 8 of the first octet of the *ATGW UDP port* field contains the most significant bit and bit 1 of the second octet of the *ATGW UDP port* field contains the least significant bit. The *ATGW IPv6 address* field indicates the ATGW IPv6 address to be used by SC UE to send the speech media of the session transferred by the CS to PS SRVCC access transfer. Bit 8 of the first octet of the *ATGW IPv6 address* field contains the most significant bit and bit 1 of the sixteenth octet of the *ATGW IPv6 address* field contains the least significant bit. In this version of specification, the sender does not include the *Extensions* field and the recipient ignores the *Extensions* field. See NOTE.
NOTE: Inclusion of *Extensions* field enlarges the access stratum handover command, which can result in longer duration of handover procedure and in message being larger than maximum size of the access stratum messages. Thus, inclusion of *Extensions* field is not recommended.

D.5.4 IANA registration template

Your Name:

<MCC name>

Your Email Address:

<MCC email address>

Media Type Name:

Application

Subtype name:

vnd.3gpp.access-transfer-events+xml

Required parameters:

None

Optional parameters:

"charset" the parameter has identical semantics to the charset parameter of the "application/xml" media type as specified in section 9.1 of IETF RFC 7303.

"et" when the MIME type is included in the Accept header field, the "et" parameter value is a comma delimited list of the values of the "event-type" attribute of the <event> element of the <events> root element which the sender of the Accept header field is able to receive. The <events> root element is defined by XML schema described in the Published specification.

Encoding considerations:

binary.

Security considerations:

Same as general security considerations for application/xml media type as specified in section 9.1 of IETF RFC 7303. In addition, this media type provides a format for exchanging information in SIP, so the security considerations from IETF RFC 3261apply.

The information transported in this media type does not include active or executable content.

Mechanisms for privacy and integrity protection of protocol parameters exist. Those mechanisms as well as authentication and further security mechanisms are described in 3GPP TS 24.229.

This media type does not include provisions for directives that institute actions on a recipient's files or other resources.

This media type does not include provisions for directives that institute actions that, while not directly harmful to the recipient, may result in disclosure of information that either facilitates a subsequent attack or else violates a recipient's privacy in any way.

This media type does not employ compression.

Interoperability considerations:

Same as general interoperability considerations for application/xml media type as specified in section 9.1 of IETF RFC 7303. Any unknown XML elements and any unknown XML attributes are to be ignored by recipient of the MIME body.

Published specification:

3GPP TS 24.237 "IP Multimedia Subsystem (IMS) Service Continuity", version 13.1.0, available via http://www.3gpp.org/specs/numbering.htm.

Applications which use this media type:

Applications supporting the service continuity as described in the published specification.

Fragment identifier considerations:

The handling in section 5 of IETF RFC 7303 applies.

Restrictions on usage:

None

Provisional registration? (standards tree only):

N/A

Additional information:

1. Deprecated alias names for this type: none
2. Magic number(s): none
3. File extension(s): none
4. Macintosh File Type Code(s): none
5. Object Identifier(s) or OID(s): none

Intended usage:

Common

Person to contact for further information:

- Name: <MCC name>
- Email: <MCC email address>
- Author/Change controller:
  i) Author: 3GPP CT1 Working Group/3GPP_TSG_CT_WG1@LIST.ETSI.ORG
  ii) Change controller: <MCC name>/<MCC email address>

D.6 PS-to-CS preparation XML schema

D.6.1 General

This subclause defines the XML schema and the MIME type for information exchanged during codec inquiry prior to PS to CS SRVCC access transfer.

D.6.2 XML schema

```
<?xml version="1.0" encoding="UTF-8"?>
<xs:schema
  xmlns:xs="http://www.w3.org/2001/XMLSchema"
  elementFormDefault="qualified"
  attributeFormDefault="unqualified">

  <xs:element name="PS-to-CS-preparation-request" type="PS-to-CS-preparation-requestType"/>
  <xs:element name="PS-to-CS-preparation-response" type="PS-to-CS-preparation-responseType"/>

  <xs:complexType name="PS-to-CS-preparation-requestType">
    <xs:sequence>
      <xs:element name="MSC-server-supported-codec-list" type="xs:base64Binary"/>
      <xs:element name="anyExt" type="anyExtType" minOccurs="0"/>
      <xs:any namespace="##other" processContents="lax" minOccurs="0" maxOccurs="unbounded"/>
    </xs:sequence>
    <xs:anyAttribute namespace="##any" processContents="lax"/>
  </xs:complexType>

  <xs:complexType name="PS-to-CS-preparation-responseType">
    <xs:sequence>
      <xs:choice>
        <xs:element name="currently-possible" type="currently-possibleType" minOccurs="0"/>
        <xs:element name="currently-not-possible" type="currently-not-possibleType" minOccurs="0"/>
      </xs:choice>
      <xs:element name="anyExt" type="anyExtType" minOccurs="0"/>
      <xs:any namespace="##other" processContents="lax" minOccurs="0" maxOccurs="unbounded"/>
```

```
    </xs:sequence>
    <xs:anyAttribute namespace="##any" processContents="lax"/>
  </xs:complexType>

  <xs:complexType name="currently-possibleType">
    <xs:sequence>
      <xs:element name="IMS-preferred-codec-list" type="xs:base64Binary"/>
      <xs:element name="anyExt" type="anyExtType" minOccurs="0"/>
      <xs:any namespace="##other" processContents="lax" minOccurs="0" maxOccurs="unbounded"/>
    </xs:sequence>
    <xs:anyAttribute namespace="##any" processContents="lax"/>
  </xs:complexType>

  <xs:complexType name="currently-not-possibleType">
    <xs:sequence>
      <xs:element name="session" type="sessionType" minOccurs="0" maxOccurs="unbounded"/>
      <xs:element name="anyExt" type="anyExtType" minOccurs="0"/>
      <xs:any namespace="##other" processContents="lax" minOccurs="0" maxOccurs="unbounded"/>
    </xs:sequence>
    <xs:anyAttribute namespace="##any" processContents="lax"/>
  </xs:complexType>

  <xs:complexType name="sessionType">
    <xs:sequence>
      <xs:element name="state-info" type="xs:string"/>
      <xs:element name="direction" type="xs:string"/>
      <xs:element name="speech-state" type="xs:string"/>
      <xs:element name="supported-features" type="supported-featuresType" minOccurs="0"/>
      <xs:element name="anyExt" type="anyExtType" minOccurs="0"/>
      <xs:any namespace="##other" processContents="lax" minOccurs="0" maxOccurs="unbounded"/>
    </xs:sequence>
    <xs:anyAttribute namespace="##any" processContents="lax"/>
  </xs:complexType>

  <xs:complexType name="supported-featuresType">
    <xs:sequence>
      <xs:element name="feature-tag" type="feature-tagType" maxOccurs="unbounded"/>
      <xs:element name="anyExt" type="anyExtType" minOccurs="0"/>
      <xs:any namespace="##other" processContents="lax" minOccurs="0" maxOccurs="unbounded"/>
    </xs:sequence>
    <xs:anyAttribute namespace="##any" processContents="lax"/>
  </xs:complexType>

  <xs:complexType name="feature-tagType">
    <xs:simpleContent>
      <xs:extension base="xs:string">
        <xs:attribute name="name" type="xs:string">
        </xs:attribute>
      </xs:extension>
    </xs:simpleContent>
  </xs:complexType>

  <xs:complexType name="anyExtType">
    <xs:sequence>
      <xs:any namespace="##any" processContents="lax" minOccurs="0" maxOccurs="unbounded"/>
    </xs:sequence>
  </xs:complexType>

</xs:schema>
```

D.6.3 Additional syntactic rules

The root element of the XML text shall be either the <PS-to-CS-preparation-request> element or the <PS-to-CS-preparation-response> element.

The <PS-to-CS-preparation-request> element:

1) shall include the <MSC-server-supported-codec-list> element;

2) may include the <anyExt> element;

3) may include elements specified in other namespaces; and

4) may include attributes specified in this namespace or other namespaces.

The <PS-to-CS-preparation-response> element:

1) shall include one of the following:

   a) the <currently-possible> element; or

   b) the <currently-not-possible> element;

2) may include the <anyExt> element;

3) may include elements specified in other namespaces; and

4) may include attributes specified in this namespace or other namespaces.

The <currently-possible> element:

1) shall include the <IMS-preferred-codec-list> element;

2) may include the <anyExt> element;

3) may include XML elements specified in other XML namespaces; and

4) may include attributes specified in this namespace or other namespaces.

The <currently-not-possible> element:

1) may include one or more <session> elements;

2) may include the <anyExt> element;

3) may include XML elements specified in other XML namespaces; and

4) may include attributes specified in this namespace or other namespaces.

The <session> element:

1) shall include the <state-info> element set to one of the values indicated in table D.6.3-1;

Table D.6.3-1: ABNF syntax of values of the <state-info> element

```
state-info-values = alerting-phase-value / pre-alerting-phase-value / confirmed-dialog-value
alerting-phase-value = %x65.61.72.6c.79 ; "early"
pre-alerting-phase-value = %x70.72.65.2d.61.6c.65.72.74.69.6e.67 ; "pre-alerting"
confirmed-dialog-value = %x63.6f.6e.66.69.72.6d.65.64 ; "confirmed"
```

2) shall include the <direction> element set to one of the values indicated in table D.6.3-2;

Table D.6.3-2: ABNF syntax of values of the <direction> element

```
direction-values = initiator-value / receiver-value
initiator-value = %x69.6e.69.74.69.61.74.6f.72 ; "initiator"
receiver-value = %x72.65.63.65.69.76.65.72 ; "receiver"
```

3) shall include the <speech-state> element set to one of the values indicated in table D.6.3-3; and

Table D.6.3-2: ABNF syntax of values of the <speech-state> element

```
speech-state-values = active-speech-media-component / inactive-speech-media-component
active-speech-media-component = %x61.63.74.69.76.65 ; "active"
inactive-speech-media-component = %x69.6e.61.63.74.69.76.65 ; "inactive"
```

4) may include the <supported-features> element.

The <supported-features> element:

1) shall include zero or more <feature-tag> elements;

2) may include the <anyExt> element; and

3) may include elements specified in other namespaces; and

4) may include attributes specified in this namespace or other namespaces.

The <feature-tag> element shall include the "name" attribute.

D.6.4 Semantic

A entity receiving an XML text compliant to the XML schema specified in subclause D.6.2 shall ignore any unknown element and any unknown attribute.

Presence of the <PS-to-CS-preparation-request> element shall indicate that the XML text carries the PS-to-CS-preparation-request.

Presence of the <PS-to-CS-preparation-response> element shall indicate that the XML text carries the PS-to-CS-preparation-response.

The <MSC-server-supported-codec-list> element shall contain an SDP body with one audio m= line with one or more RTP payload types that are supported commonly by the served SC UE, the target RAN and the target CS-MGW selected by the MSC server.

Presence of the <currently-possible> element shall indicate that the PS to CS SRVCC access transfer is currently possible.

The <IMS-preferred-codec-list> element shall contain an SDP body with one audio m= line with one or more RTP payload types that are preferred by the ATCF to be used by the MSC server.

Presence of the <currently-not-possible> element shall indicate that the PS to CS SRVCC access transfer is currently not possible.

The <session> element describe a session of the served SC UE in PS.

The <state-info> element indicates the state of the session, i.e. whether the session is in the pre-alerting phase, is in the alerting phase or is a confirmed dialog.

The <direction> element indicates the direction of the session, i.e. whether the served SC UE is an initiator of the session or a receiver of the session.

The <speech-state> element indicates the state of the speech media component of the session, i.e. whether the speech media component is an active speech media component or an inactive speech media component.

The "name" attribute of the <feature-tag> element indicates a media feature tag supported by the entity sending the XML text. The value, if included, of the <feature-tag> element indicates a value of the media feature tag supported by the sender of the XML text.

The <anyExt> element shall convey extensions.

D.6.5 IANA registration template

Editors' note (CR#1258, WID: TEI14): MCC is request to register the MIME type when the Rel-14 is completed.

Your Name:

<MCC name>

Your Email Address:

<MCC email address>

Media Type Name:

Application

Subtype name:

vnd.3gpp.PS-to-CS-preparation+xml

Required parameters:

None

Optional parameters:

"charset" the parameter has identical semantics to the charset parameter of the "application/xml" media type as specified in section 9.1 of IETF RFC 7303.

Encoding considerations:

binary.

Security considerations:

Same as general security considerations for application/xml media type as specified in section 9.1 of IETF RFC 7303. In addition, this media type provides a format for exchanging information in SIP, so the security considerations from IETF RFC 3261apply.

The information transported in this media type does not include active or executable content.

Mechanisms for privacy and integrity protection of protocol parameters exist. Those mechanisms as well as authentication and further security mechanisms are described in 3GPP TS 24.229.

This media type does not include provisions for directives that institute actions on a recipient's files or other resources.

This media type does not include provisions for directives that institute actions that, while not directly harmful to the recipient, may result in disclosure of information that either facilitates a subsequent attack or else violates a recipient's privacy in any way.

This media type does not employ compression.

Interoperability considerations:

Same as general interoperability considerations for application/xml media type as specified in section 9.1 of IETF RFC 7303.

Published specification:

3GPP TS 24.237 "IP Multimedia Subsystem (IMS) Service Continuity", version 14.2.0, available via http://www.3gpp.org/specs/numbering.htm.

Applications which use this media type:

Applications supporting the service continuity as described in the published specification.

Fragment identifier considerations:

The handling in section 5 of IETF RFC 7303 applies.

Restrictions on usage:

None

Provisional registration? (standards tree only):

N/A

Additional information:

1. Deprecated alias names for this type: none
2. Magic number(s): none
3. File extension(s): none

4. Macintosh File Type Code(s): none
5. Object Identifier(s) or OID(s): none

Intended usage:

Common

Person to contact for further information:

- Name: <MCC name>
- Email: <MCC email address>
- Author/Change controller:
  i) Author: 3GPP CT1 Working Group/3GPP_TSG_CT_WG1@LIST.ETSI.ORG
  ii) Change controller: <MCC name>/<MCC email address>

Annex E (informative): INFO packages defined in the current document

E.1 Info package for transfer of the conference information

E.1.1 Scope

This subclause contains the information required for the IANA registration of info package g.3gpp.mid-call in accordance with IETF RFC 6086 [54].

E.1.2 g.3gpp.mid-call info package

E.1.2.1 Overall description

When PS to CS access transfer with the MSC Server assisted mid-call feature is applied for a session with conference focus there is a need to deliver participant identities from SCC AS to MSC server.

E.1.2.2 Applicability

This package is used to transport participant identities when the PS to CS access transfer with the MSC server assisted mid-call feature is applied to a session with conference focus.

E.1.2.2A Appropriateness of INFO Package Usage

A number of solutions were discussed for the transportation of identities of up to 5 conference participants from the SCC AS (a B2BUA) to the MSC server (a UA). The solutions were:

1) Use of subscription to the conference event package as specified in IETF RFC 4575.

2) Use of the session related methods (e.g. SIP 200 (OK) response to the SIP INVITE request).

3) Use of the SIP MESSAGE method.

4) Use of media plane mechanisms.

5) Use of the SIP INFO method as decribed in IETF RFC 6086, by defining a new info package.

Furthermore, each of the solutions 1), 2), 3), 4) and 5) was evaluated.

The use of the conference event package was discounted for the following reasons:

1) When an access transfer needs to occur, the MSC server sends a particular SIP INVITE request to the SCC AS to initiate the access transfer of the dialog-to-be-transferred. Upon reception of the particular SIP INVITE request, the SCC AS ensures that the media of the session supported by the dialog-to-be-transferred are redirected from an original UA served by the SCC AS to the MSC server. If the dialog-to-be-transferred happens to be with a conference focus and the conference session happens to be established by the original UA served by the SCC AS, then the access transfer procedure requires the MSC server to be aware of identities of up to 5 conference participants of the conference session. In the majority of cases of access transfer, the dialog-to-be-transferred is not with a conference focus. In even fewer cases of access transfer, the conference session happens to be established by the original UA served by the SCC AS.

2) If the MSC server sent the SIP SUBSCRIBE request before sending the particular SIP INVITE request and the dialog-to-be-transferred does not happen to be with a conference focus or the conference session does not happen to be established by the original UA served by the SCC AS, then two messages (a SIP SUBSCRIBE request, a SIP 4xx response to the SIP SUBSCRIBE request) would be exchanged between the MSC server and the SCC AS. In comparison, zero messages are exchanged when using the SIP INFO method.

3) If the MSC server sent the SIP SUBSCRIBE request before sending the particular SIP INVITE request, the dialog-to-be-transferred happens to be with a conference focus and the conference session happens to be

established by the original UA served by the SCC AS, then four messages (a SIP SUBSCRIBE request, a SIP 2xx response to the SIP SUBSCRIBE request, a SIP NOTIFY request, a SIP 2xx response to the SIP NOTIFY request) would be exchanged between the MSC server and the SCC AS. In comparison, only two messages (a SIP INFO request, a SIP 2xx response to the SIP INFO request, are exchanged when using the SIP INFO method.

4) If the MSC server sent the SIP SUBSCRIBE request after receiving a particular SIP response to the particular SIP INVITE request, the particular SIP response includes a Contact header field with sip.isfocus media feature tag indicating that the dialog-to-be-transferred happens to be with a conference focus, and the conference session happens not to be established by the original UA served by the SCC AS, then two messages (a SIP SUBSCRIBE request, a SIP 4xx response to the SIP SUBSCRIBE request) would be exchanged between the MSC server and the SCC AS. In comparison, zero messages are exchanged when using the SIP INFO method.

5) If the MSC server sent the SIP SUBSCRIBE request after receiving a particular SIP response to the particular SIP INVITE request, the particular SIP response includes a Contact header field with sip.isfocus media feature tag indicating the dialog-to-be-transferred happens to be with a conference focus, and the conference session happens to be established by the original UA served by the SCC AS, then four messages (a SIP SUBSCRIBE request, a SIP 2xx response to the SIP SUBSCRIBE request, a SIP NOTIFY request, a SIP 2xx response to the SIP NOTIFY request) would be exchanged between the MSC server and the SCC AS. In comparison, only two messages (a SIP INFO request, a SIP 2xx response to the SIP INFO request) are exchanged when using the SIP INFO method.

6) In the SIP INFO method based solution, it is simpler to associate the information of identities of up to 5 conference participants of the dialog-to-be-transferred with the particular SIP INVITE request as the SCC AS associates the particular SIP INVITE request with a dialog-to-be-transferred and the SIP INFO request is sent as an in-dialog request in a dialog created as result of the particular SIP INVITE request. In the use of the conference event package, the SIP SUBSCRIBE request and the particular SIP INVITE request are not related at the SIP level and SCC AS might associate the SIP SUBSCRIBE request with a dialog-to-be-transferred different to the one associated with the SIP INVITE request.

The use of session related methods was discounted as it was concluded that bodies should not be included in SIP responses and the SIP UPDATE method is not appropriate since the session is not modified.

Use of the SIP MESSAGE method was discounted since the use of the SIP INFO method enables negotiation of supported event packages in the SIP INVITE transaction while the use of the SIP MESSAGE method does not.

Use of the media plane mechanisms was discounted since the amount of information transferred from the SCC AS to the MSC server is limited and set up of TCP based media stream would generate extra messages.

Based on the above analyses, the SIP INFO method was chosen to transport the identities of up to 5 conference participants from the SCC AS to the MSC server.

E.1.2.3 Info package name

g.3gpp.mid-call

E.1.2.4 Info package parameters

None defined

E.1.2.5 SIP options tags

None defined

E.1.2.6 INFO message body parts

The MIME type of the message body carrying participant identities is application/vnd.3gpp.mid-call+xml. application/vnd.3gpp.mid-call+xml MIME type is defined in 3GPP TS 24.237.

When associated with the g.3gpp.mid-call info package, the Content-Disposition value of the message body carrying participant identities is "info-package".

E.1.2.7 Info package usage restrictions

None defined.

E.1.2.8 Rate of INFO Requests

Single INFO request generated after session set up.

E.1.2.9 Info package security considerations

The security is based on the generic security mechanism provided for the underlying SIP signalling. No additional security mechanism is defined.

E.1.2.10 Implementation details and examples

UAC generation of INFO requests: See 3GPP TS 24.237: "IP Multimedia Subsystem (IMS) Service Continuity; Stage 3"

UAS processing of INFO requests: See 3GPP TS 24.237: "IP Multimedia Subsystem (IMS) Service Continuity; Stage 3"

Examples: See 3GPP TS 24.237: "IP Multimedia Subsystem (IMS) Service Continuity; Stage 3"

E.2 INFO package for transfer of state-and-event info

E.2.1 Scope

This annex defines an info package in accordance with IETF RFC 6086 [54] for sending state and event information during PS to CS SRVCC access transfer using SIP INFO requests.

E.2.2 state-and-event info package

E.2.2.1 General

This subclause contains the information required for the IANA registration of an info package.

E.2.2.2 Overall description

When PS to CS SRVCC access transfer from PS to CS access is applied for a session with an active full duplex speech component and the related dialog is in early state there is a need to deliver state information from an SCC AS to an MSC server. Further it is requested that an MSC server supporting PS to CS SRVCC for calls in alerting phase informs the SCC AS about a UE having accepted a terminating call.

E.2.2.3 Applicability

This package is used to transport session state information and related event information when a session in alerting phase is transferred from PS to CS using SRVCC access transfer procedures.

The mechanism allows that information about the session that is subject to PS to CS SRVCC and related events to be sent inside an existing dialog due to the session transfer SIP INVITE request.

E.2.2.3A Appropriateness of INFO Package Usage

A number of solutions for the transportation of event and state related information between the SCC AS (a B2BUA) and MSC server (a UA). The solutions considered were:

1. Use of session related SIP methods for transporting event and state information, e.g. SIP 183 response, SIP UPDATE request.

2. Use of subscription to the dialog-event package as described in IETF RFC 4325.

3. Use of SIP MESSAGE method.
4. Use of media plane mechanisms.
5. Use of SIP INFO method as decribed in IETF RFC 6086, by defining a new info package.

Furthermore, each of the solutions was evaluated.

The use of session related SIP methods was discounted as it was concluded that bodies should not be included in SIP responses and the SIP UPDATE method is not appropriate since the session is not modified.

The use of the dialog-event package was discounted for the following reasons:

a) When an access transfer needs to occur, the MSC server sends a particular SIP INVITE request to the SCC AS to initiate the access transfer of the dialog-to-be-transferred. Upon reception of the particular SIP INVITE request, the SCC AS ensures that the media of the session supported by the dialog-to-be-transferred are redirected from an original UA served by SCC AS to the MSC server. The access transfer procedure requires the MSC server to be aware of the state related information of the dialog-to-be-transferred. This can be provided implicitly or explicitly to the MSC server. When the dialog-to-be-transferred happens to be a confirmed dialog, the SCC AS implicitly informs the MSC server about the state related information of the dialog-to-be-transferred by accepting the particular SIP INVITE request. When the dialog-to-be-transferred happens to be an early dialog, the access transfer procedure requires the SCC AS to explicitly provide the state related information of the dialog-to-be-transferred to the MSC server. In the majority of cases of access transfer, the dialog-to-be-transferred will be a confirmed dialog. If the MSC server sent the SIP SUBSCRIBE request before sending the particular SIP INVITE request, then the MSC will receive the state related information of the dialog-to-be-transferred explicitly in the majority of cases when such explicit information is not required.

b) If the MSC server sent the SIP SUBSCRIBE request before sending the particular SIP INVITE request, then four messages (a SIP SUBSCRIBE request, a SIP 2xx response to the SIP SUBSCRIBE request, a SIP NOTIFY request, a SIP 2xx response to the SIP NOTIFY request) would be exchanged between the MSC server and the SCC AS. In comparison, zero messages (if the dialog-to-be-transferred happens to be a confirmed dialog) or only two messages (a SIP INFO request, a SIP 2xx response to the SIP INFO request, if the dialog-to-be-transferred happens to be an early dialog) are exchanged when using the SIP INFO method.c) If the MSC server sent the SIP SUBSCRIBE request after receiving a particular provisional response to the particular SIP INVITE request, then four messages (a SIP SUBSCRIBE request, a SIP 2xx response to the SIP SUBSCRIBE request, a SIP NOTIFY request, a SIP 2xx response to the SIP NOTIFY request) would be exchanged between the MSC server and the SCC AS. In comparison, only two messages (a SIP INFO request, a SIP 2xx response to the SIP INFO request) are exchanged when using the SIP INFO method.

d) In the SIP INFO method based solution, it is simpler to associate the event and state related information of the dialog-to-be-transferred with the particular SIP INVITE request as SCC AS associates the particular SIP INVITE request with a dialog-to-be-transferred and the SIP INFO request is sent as an in-dialog request in a dialog created as result of the particular SIP INVITE request. In the use of the dialog-event package, the SIP SUBSCRIBE request and the particular SIP INVITE request are not related on SIP level and SCC AS might associate the SIP SUBSCRIBE request with a dialog-to-be-transferred different than the one associated with the SIP INVITE request.

Use of the SIP MESSAGE method was discounted since the use of the SIP INFO method enables negotiation of supported event packages in the SIP INVITE transaction while the use of the SIP MESSAGE method does not.

Use of the media plane mechanisms was discounted since the amount of information transferred between the SCC AS and the MSC server is limited and set up of TCP based media stream would generate extra messages.

Based on the above analyses, the SIP INFO method was chosen.

E.2.2.4 Info package name

The name of the info package is g.3gpp.state-and-event.

E.2.2.5 Info package parameters

No parameters are defined for the g.3gpp.state-and-event info package.

E.2.2.6 SIP option tags

No SIP option tags are defined for the g.3gpp.state-and-event info package.

E.2.2.7 INFO message body parts

E.2.2.7.1 General

The state-and-event information is carried in the state-and-event-info message body, defined in annex D of 3GPP TS 24.237.

E.2.2.7.2 MIME type

The MIME type value for the message body is "application/vnd.3gpp.state-and-event-info+xml", defined in annex D of 3GPP TS 24.237.

E.2.2.7.3 Content disposition

The Content Disposition value for the message body, when associated with the g.3gpp.state-and-event info package, is "info-package" as defined in IETF RFC 6086.

E.2.2.8 Info package usage restrictions

No usage restrictions are defined for the g.3gpp.state-and-event info package.

E.2.2.9 Rate of INFO requests

No maximum rate or minimum rate is defined for sending SIP INFO requests associated with the g.3gpp.state-and-event info package.

When PS to CS SRVCC for calls in alerting phase is applied, then a single SIP INFO request is generated after the session transfer SIP INVITE request. This can be followed by one more additional SIP INFO request.

E.2.2.10 Info package security considerations

No additional security mechanism is defined for the g.3gpp.state-and-event info package.

The security of the g.3gpp.state-and-event info package is based on the generic security mechanism provided for the underlying SIP signalling.

E.2.2.11 Implementation details and examples

See 3GPP TS 24.237: "IP Multimedia Subsystem (IMS) Service Continuity; Stage 3"

E.3 INFO package for transfer of access transfer events

E.3.1 Scope

This annex contains the information required for the IANA registration of info package g.3gpp.access-transfer-events in accordance with IETF RFC 6086 [54].

E.3.2 g.3gpp.access-transfer-events info package

E.3.2.1 Overall description

This info packages enables SIP entities to inform each other about events occuring during access transfer of session between access networks. E.g. during CS to PS single radio voice call continuity access transfer the MSC server needs to inform ATCF about the session transfer notification request, ATCF needs to inform MSC server about the session transfer notification response and MSC server needs to inform ATCF about the session transfer preparation.

E.3.2.2 Applicability

The info package mechanism has been selected since the transported information cannot be sent in media plane as entities receiving the information are not mandated to always anchor the media and since usage of subscription in order to transport the information has not been seen as feasible as the information can be exchanged bidirectionally.

E.3.2.3 Info package name

g.3gpp.access-transfer-events

E.3.2.4 Info package parameters

"et" when included in the Recv-Info header field and when Accept header field is not present in the SIP message, the "et" parameter value is a comma delimited list of the values of the "event-type" attribute of the <event> element of the <events> root element which the sender of the Recv-Info header field is able to receive. The <events> root element is defined in XML schema of the MIME type application/vnd.3gpp.access-transfer-events+xml. This parameter applies for the Recv-Info header field and does not apply for the Info-Package header field.

E.3.2.5 SIP option tags

None defined

E.3.2.6 INFO message body parts

The MIME type of the message body carrying events information is application/vnd.3gpp.access-transfer-events+xml. The application/vnd.3gpp.access-transfer-events+xml MIME type is defined in 3GPP TS 24.237.

When associated with the g.3gpp.access-transfer-events info package, the Content-Disposition value of the message body carrying participant identities is "info-package".

E.3.2.7 Info package usage restrictions

None defined.

E.3.2.8 Rate of INFO requests

The rate of the INFO requests depends on the supported event types.

E.3.2.9 Info package security considerations

The security is based on the generic security mechanism provided for the underlying SIP signalling. No additional security mechanism is defined.

E.3.2.10 Implementation details and examples

UAC generation of INFO requests: See 3GPP TS 24.237: "IP Multimedia Subsystem (IMS) Service Continuity; Stage 3"

UAS processing of INFO requests: See 3GPP TS 24.237: "IP Multimedia Subsystem (IMS) Service Continuity; Stage 3"

Examples: See 3GPP TS 24.237: "IP Multimedia Subsystem (IMS) Service Continuity; Stage 3"

Annex F (informative): Access transfer scenarios

F.1 Introduction

F.1.1 General

This annex describes the behaviour of

1) the network when determining and performing PS to CS SRVCC access transfer for various types of single and multi session scenarios, based on UE and network support of the following features:

   - MSC server assisted mid-call feature;

   - PS to CS SRVCC for calls in alerting phase; and

   - PS to CS SRVCC for originating calls in pre-alerting phase.

2) the UE when determining and performing PS-CS dual radio access transfer for various types of single and multi session scenarios, based on UE and network support of the following features:

   - MSC server assisted mid-call feature;

   - PS to CS dual radio access transfer for calls in alerting phase and CS to PS dual radio access transfer for calls in alerting phase; and

   - PS to CS dual radio access transfer for originating calls in pre-alerting and CS to PS dual radio access transfer for originating calls in pre-alerting phase.

F.1.2 Void

F.2 PS to CS SRVCC

F.2.1 Key to interpret PS to CS SRVCC access transfer scenarios

The key to interpret the PS to CS SRVCC access transfer scenarios given in tables F.2. 2-1, F.2.2-2 and F.2.3-1 is specified below:

- "MAM" represents support of MSC server assisted mid-call feature
- "ALP" represents support of PS to CS SRVCC for calls in alerting phase
- "PAP" represents support of PS to CS SRVCC for originating calls in pre-alerting phase
- "CAS" represents a session with active speech media component, which is a confirmed dialog
- "CIS" represents a session with inactive speech media component, which is a confirmed dialog
- "EAS" represents a session with active speech media component in alerting phase
- "EAST" represents a terminating session with active speech media component in alerting phase
- "EASO" represents an originating session with active speech media component in alerting phase
- "EPS" represents a session with active speech media component in pre-alerting phase
- "EPST" represents a terminating session with active speech media component in pre-alerting phase
- "EPSO" represents an originating session with active speech media component in pre-alerting phase
- "CACS" represents a session with active speech media component and with conference control, which is a confirmed dialog
- "CICS" represents a session with inactive speech media component and with conference control, which is a confirmed dialog
- "Transfer CACS as CAS" refers to the behaviour of the SCC AS to transfer CACS, but without conference control
- "Transfer CICS as CIS" refers to the behaviour of the SCC AS to transfer CICS, but without conference control
- When "Result in network" is to transfer a session, the Transaction Identifier (TI) assigned to the call is shown in brackets, e.g. Transfer CAS (TI=0) refers to a TI of zero assigned to this call that is transferred.

F.2.2 PS to CS SRVCC access transfer scenarios

Table F.2.2-1 describes PS to CS SRVCC access transfer scenarios considering only cases when only one session in an early phase exists.

Table F.2.2-1: PS to CS SRVCC access transfer scenarios, single early session

UE and Network Support Combination	Access Transfer Scenario	Result in network
UE supports at least MAM and Network supports MAM only or UE supports MAM only and Network supports at least MAM	CAS only	Transfer CAS (TI=0)
	CIS only	Transfer CIS (TI=0)
	EAS only	Send SIP 480 (Temporary Unavailable) to reject SIP INVITE request due to STN-SR
	EPS only	Send SIP 480 (Temporary Unavailable) to reject SIP INVITE request due to STN-SR
	CAS + CIS	Transfer CAS (TI=0) and transfer CIS (TI=1)
	CAS + EAST	Transfer CAS (TI=0) and release EAST
	CAS + EPST	Transfer CAS (TI=0) and release EPST
	CIS + EASO	Transfer CIS (TI=0) and release EASO
	CIS + EAST	Transfer CIS (TI=0) and release EAST
	CIS + EPSO	Transfer CIS (TI=0) and release EPSO

	CAS + CIS + EAST	Transfer CAS (TI=0), transfer CIS (TI=1) and release EAST
	CAS + CIS + EPST	Transfer CAS (TI=0), transfer CIS (TI=1) and release EPST
UE supports ALP only and Network supports at least ALP or UE supports at least ALP and network supports ALP only	CAS only	Transfer CAS (TI=0)
	CIS only	Send SIP 480 (Temporary Unavailable) to reject SIP INVITE request due to STN-SR
	EAS only	Transfer EAS (TI=0)
	EPS only	Send SIP 480 (Temporary Unavailable) to reject SIP INVITE request due to STN-SR
	CAS + CIS	Transfer CAS (TI=0) and release CIS
	CAS + EAST	Transfer CAS (TI=0) and transfer EAST (TI=1)
	CAS + EPST	Transfer CAS (TI=0) and release EPST
	CIS + EASO	Transfer EASO (TI=0) and release CIS
	CIS + EAST	Transfer EAST (TI=0) and release CIS
	CIS + EPSO	Send SIP 480 (Temporary Unavailable) to reject SIP INVITE request due to STN-SR
	CAS + CIS + EAST	Transfer CAS (TI=0), transfer EAST (TI=1) and release CIS
	CAS + CIS + EPST	Transfer CAS (TI=0), release CIS, release EPST
UE supports ALP+PAP only and Network supports at least ALP+PAP or UE supports at least ALP+PAP and Network supports ALP+PAP only	CAS only	Transfer CAS (TI=0)
	CIS only	Send SIP 480 (Temporary Unavailable) to reject SIP INVITE request due to STN-SR
	EAS only	Transfer EAS (TI=0)
	EPSO only	Transfer EPSO (TI=0)
	EPST only	Send SIP 480 (Temporary Unavailable) to reject SIP INVITE request due to STN-SR
	CAS + CIS	Transfer CAS (TI=0) and release CIS
	CAS + EAST	Transfer CAS (TI=0) and transfer EAST (TI=1)
	CAS + EPST	Transfer CAS (TI=0) and release EPST
	CIS + EASO	Transfer EASO (TI=0) and release CIS
	CIS + EAST	Transfer EAST (TI=0) and release CIS
	CIS + EPSO	Transfer EPSO (TI=0) and release CIS
	CAS + CIS + EAST	Transfer CAS (TI=0), transfer EAST (TI=1) and release CIS
	CAS + CIS + EPST	Transfer CAS (TI=0), release CIS, release EPST
UE and Network support MAM+ALP+PAP	CAS only	Transfer CAS (TI=0)
	CIS only	Transfer CIS (TI=0)
	EAS only	Transfer EAS (TI=0)
	EPSO only	Transfer EPSO (TI=0)
	EPST only	Send SIP 480 (Temporary Unavailable) to reject SIP INVITE request due to STN-SR
	CAS + CIS	Transfer CAS (TI=0) and transfer CIS (TI=1)
	CAS + EAST	Transfer CAS (TI=0) and transfer EAST (TI=1)
	CAS + EPST	Transfer CAS (TI=0) and release EPST
	CIS + EASO	Transfer CIS (TI=0) and transfer EASO (TI=1)
	CIS + EAST	Transfer CIS (TI=0) and transfer EAST (TI=1)
	CIS + EPSO	Transfer CIS (TI=0) and transfer EPSO (TI=1)
	CAS + CIS + EAST	Transfer CAS (TI=0), transfer CIS (TI=1) and release EAST NOTE 1: Even though the UE and network can support the transfer of EAST, the network will only transfer CAS and CIS and will release EAST.
	CAS + CIS + EPST	Transfer CAS (TI=0), transfer CIS (TI=1) and release EPST
	CAS only	Transfer CAS (TI=0)
	CIS only	Transfer CIS (TI=0)
	EAS only	Transfer EAS (TI=0)
	EPS only	Send SIP 480 (Temporary Unavailable) to reject SIP INVITE request due to STN-SR
	CAS + CIS	Transfer CAS (TI=0) and transfer CIS (TI=1)
	CAS + EAST	Transfer CAS (TI=0) and transfer EAST (TI=1)
	CAS + EPST	Transfer CAS (TI=0) and release EPST
	CIS + EASO	Transfer CIS (TI=0) and transfer EASO (TI=1)
	CIS + EAST	Transfer CIS (TI=0) and transfer EAST (TI=1)
	CIS + EPSO	Transfer CIS (TI=0) and release EPSO

UE supports MAM+ ALP only and Network supports at least MAM+ALP or UE supports at least MAM+ ALP and Network supports MAM+ALP only	CAS + CIS + EAST	Transfer CAS (TI=0), transfer CIS (TI=1) and release EAST NOTE 2: Even though the UE and network can support the transfer of EAST, the network will only transfer CAS and CIS and will release EAST.
	CAS + CIS + EPST	Transfer CAS (TI=0), Transfer CIS (TI=1), release EPST
UE has no MAM support and Network supports MAM only or UE supports MAM only and Network has no MAM support	CAS only	Transfer CAS (TI=0)
	CIS only	Send SIP 480 (Temporary Unavailable) to reject SIP INVITE request due to STN-SR
	EAS only	Send SIP 480 (Temporary Unavailable) to reject SIP INVITE request due to STN-SR
	EPS only	Send SIP 480 (Temporary Unavailable) to reject SIP INVITE request due to STN-SR
	CAS + CIS	Transfer CAS (TI=0) and release CIS
	CAS + EAST	Transfer CAS (TI=0) and release EAST
	CAS + EPST	Transfer CAS (TI=0) and release EPST
	CIS + EASO	Send SIP 480 (Temporary Unavailable) to reject SIP INVITE request due to STN-SR
	CIS + EAST	Send SIP 480 (Temporary Unavailable) to reject SIP INVITE request due to STN-SR
	CIS + EPSO	Send SIP 480 (Temporary Unavailable) to reject SIP INVITE request due to STN-SR
	CAS + CIS + EAST	Transfer CAS (TI=0), release CIS and release EAST
	CAS + CIS + EPST	Transfer CAS (TI=0), release CIS and release EPST

Table F.2.2-2 below describes PS to CS SRVCC access transfer scenarios when more than one session in an early phase exists and when the UE and the network supports SRVCC access transfer for calls in an early phase.

Table F.2.2-2: PS to CS SRVCC access transfer scenarios, multiple early sessions

UE and Network Support Combination	Access Transfer Scenario	Result in network
UE supports ALP only and Network supports at least ALP or UE supports at least ALP and network supports ALP only	EASO + EAST	Send SIP 480 (Temporarily Unavailable) to reject SIP INVITE request due to STN-SR
	EPSO + EPST	Send SIP 480 (Temporarily Unavailable) to reject SIP INVITE request due to STN-SR
	EAS + EPS	Transfer EAS (TI=0) and release EPS
	CAS + EAS + EPS	Transfer CAS (TI=0), transfer EAS (TEI=1) and release EPS
	CAS + EASO + EAST	Transfer CAS (TI=0), release EASO and release EAST
	CIS + EAS + EPS	Transfer EAS (TI=0), release CIS and release EPS
	CIS + EASO + EAST	Send SIP 480 (Temporarily Unavailable) to reject SIP INVITE request due to STN-SR
	CIS + EPSO + EPST	Send SIP 480 (Temporarily Unavailable) to reject SIP INVITE request due to STN-SR
UE supports ALP+PAP only and Network supports at least ALP+PAP or UE supports at least ALP+PAP and Network supports ALP+PAP only	EASO + EAST	Send SIP 480 (Temporarily Unavailable) to reject SIP INVITE request due to STN-SR
	EPSO + EPST	Send SIP 480 (Temporarily Unavailable) to reject SIP INVITE request due to STN-SR
	EAS + EPS	Transfer EAS (TI=0) and release EPS
	CAS + EAS + EPS	Transfer CAS (TI=0), transfer EAS (TI=1) and release EPS
	CAS + EASO + EAST	Transfer CAS (TI=0), release EASO and release EAST
	CIS + EAS + EPS	Transfer EAS (TI=0), release CIS and release EPS
	CIS + EASO + EAST	Send SIP 480 (Temporarily Unavailable) to reject SIP INVITE request due to STN-SR
	CIS + EPSO + EPST	Send SIP 480 (Temporarily Unavailable) to reject SIP INVITE request due to STN-SR

UE and Network support MAM+ALP+PAP	EASO + EAST	Send SIP 480 (Temporarily Unavailable) to reject SIP INVITE request due to STN-SR
	EPSO + EPST	Send SIP 480 (Temporarily Unavailable) to reject SIP INVITE request due to STN-SR
	EAS + EPS	Transfer EAS (TI=0) and release EPS
	CAS + EAS + EPS	Transfer CAS (TI=0), transfer EAS (TI=1) and release EPS
	CAS + EASO + EAST	Transfer CAS (TI=0), release EASO and release EAST
	CIS + EAS + EPS	Transfer CIS (TI=0), transfer EAS (TI=1) and release EPS
	CIS + EASO + EAST	Transfer CIS (TI=0), release EASO and release EAST
	CIS + EPSO + EPST	Transfer CIS (TI=0), release EPSO and release EPST
UE supports MAM+ ALP only and Network supports at least MAM+ALP or UE supports at least MAM+ ALP and Network supports MAM+ALP only	EASO + EAST	Send SIP 480 (Temporarily Unavailable) to reject SIP INVITE request due to STN-SR
	EPSO + EPST	Send SIP 480 (Temporarily Unavailable) to reject SIP INVITE request due to STN-SR
	EAS + EPS	Transfer EAS (TI=0) and release EPS
	CAS + EAS + EPS	Transfer CAS (TI=0), transfer EAS (TI=1) and release EPS
	CAS + EASO + EAST	Transfer CAS (TI=0), release EASO and release EAST
	CIS + EAS + EPS	Transfer CIS (TI=0), transfer EAS (TI=1) and release EPS
	CIS + EASO + EAST	Transfer CIS (TI=0), release EASO and release EAST
	CIS + EPSO + EPST	Transfer CIS (TI=0), release EPSO and release EPST

F.2.3 PS to CS SRVCC access transfer scenarios with conference control

Table F.2.3-1: PS to CS SRVCC access transfer scenarios with conference control

UE and Network Support Combination	Access Transfer Scenario	Result in network
UE supports at least MAM and Network supports MAM only or UE supports MAM only and Network supports at least MAM	CACS only	Transfer CACS (NOTE 1)
	CICS only	Transfer CICS (NOTE 1)
	CACS + CIS	Transfer CACS (NOTE 1) and transfer CIS (TI=1)
	CAS + CICS	Transfer CAS (TI=0) and transfer CICS (NOTE 2)
	CACS + EAST	Transfer CACS (NOTE 1) and release EAST
	CACS + EPST	Transfer CACS (NOTE 1) and release EPST
	CICS + EASO	Transfer CICS (NOTE 1) and release EASO
	CICS + EAST	Transfer CICS (NOTE 1) and release EAST
	CICS + EPSO	Transfer CICS (NOTE 1) and release EPSO
	CACS + CIS + EAST	Transfer CACS (NOTE 1), transfer CIS (TI=1) and release EAST
	CAS + CICS + EAST	Transfer CAS (TI=0), transfer CICS (NOTE 2) and release EAST
	CACS + CIS + EPST	Transfer CACS (NOTE 1), transfer CIS (TI=1) and release EPST
	CAS + CICS + EPST	Transfer CAS (TI=0), transfer CICS (NOTE 2) and release EPST
	CACS + CICS	Transfer CACS (NOTE 1) and transfer CICS as CIS (TI=1)
	CACS + CICS + EAST	Transfer CACS (NOTE 1), transfer CICS as CIS (TI=1) and release EAST
	CACS + CICS + EPST	Transfer CACS (NOTE 1), transfer CICS as CIS (TI=1) and release EPST
	CACS only	Transfer CACS as CAS (TI=0)
	CICS only	Send SIP 480 (Temporarily Unavailable) to reject SIP INVITE request due to STN-SR
	CACS + CIS	Transfer CACS as CAS (TI=0) and release CIS
	CAS + CICS	Transfer CAS (TI=0) and release CICS
	CACS + EAST	Transfer CACS as CAS (TI=0) and transfer EAST (TI=1)

UE supports ALP only and Network supports at least ALP or UE supports at least ALP and network supports ALP only	CACS + EPST	Transfer CACS as CAS (TI=0) and release EPST
	CICS + EASO	Transfer EASO (TI=0) and release CICS
	CICS + EAST	Transfer EAST (TI=0) and release CICS
	CICS + EPSO	Send SIP 480 (Temporarily Unavailable) to reject SIP INVITE request due to STN-SR
	CACS + CIS + EAST	Transfer CACS as CAS (TI=0), transfer EAST (TI=1) and release CIS
	CAS + CICS + EAST	Transfer CAS (TI=0), transfer EAST (TI=1) and release CICS
	CACS + CIS + EPST	Transfer CACS as CAS (TI=0), release CIS and release EPST
	CAS + CICS + EPST	Transfer CAS (TI=0), release CICS and release EPST
	CACS + CICS	Transfer CACS as CAS (TI=0) and release CICS
	CACS + CICS + EAST	Transfer CACS as CAS (TI=0), transfer EAST (TI=1) and release CICS
	CACS + CICS + EPST	Transfer CACS as CAS (TI=0), release CICS and release EPST
UE supports ALP+PAP only and Network supports at least ALP+PAP or UE supports at least ALP+PAP and Network supports ALP+PAP only	CACS only	Transfer CACS as CAS (TI=0)
	CICS only	Send SIP 480 (Temporarily Unavailable) to reject SIP INVITE request due to STN-SR
	CACS + CIS	Transfer CACS as CAS (TI=0) and release CIS
	CAS + CICS	Transfer CAS (TI=0) and release CICS
	CACS + EAST	Transfer CACS as CAS (TI=0) and transfer EAST (TI=1)
	CACS + EPST	Transfer CACS as CAS (TI=0) and release EPST
	CICS + EASO	Transfer EASO (TI=0) and release CICS
	CICS + EAST	Transfer EAST (TI=0) and release CICS
	CICS + EPSO	Transfer EPSO (TI=0) and release CICS
	CACS + CIS + EAST	Transfer CACS as CAS (TI=0), transfer EAST (TI=1) and release CIS
	CAS + CICS + EAST	Transfer CAS (TI=0), transfer EAST (TI=1) and release CICS
	CACS + CIS + EPST	Transfer CACS as CAS (TI=0), release CIS and release EPST
	CAS + CICS + EPST	Transfer CAS (TI=0), release CICS and release EPST
	CACS + CICS	Transfer CACS as CAS (TI=0) and release CICS
	CACS + CICS + EAST	Transfer CACS as CAS (TI=0), transfer EAST (TI=1) and release CICS
	CACS + CICS + EPST	Transfer CACS as CAS (TI=0), release CICS and release EPST
UE and Network support MAM+ALP+PAP	CACS only	Transfer CACS (NOTE 1)
	CICS only	Transfer CICS (NOTE 1)
	CACS + CIS	Transfer CACS (NOTE 1) and transfer CIS (TI=1)
	CAS + CICS	Transfer CAS (TI=0) and transfer CICS (NOTE 2)
	CACS + EAST	Transfer CACS (NOTE 1) and transfer EAST (TI=1)
	CACS + EPST	Transfer CACS (NOTE 1) and release EPST
	CICS + EASO	Transfer CICS (NOTE 1) and transfer EASO (TI=1)
	CICS + EAST	Transfer CICS (NOTE 1) and transfer EAST (TI=1)
	CICS + EPSO	Transfer CICS (NOTE 1) and transfer EPSO (TI=1)
	CACS + CIS + EAST	Transfer CACS (NOTE 1), transfer CIS (TI=1), release EAST
	CAS + CICS + EAST	Transfer CAS (TI=0), transfer CICS (NOTE 2), release EAST
	CACS + CIS + EPST	Transfer CACS (NOTE 1), transfer CIS (TI=1), release EPST
	CAS + CICS + EPST	Transfer CAS (TI=0), transfer CICS (NOTE 2), release EAST
	CACS + CICS	Transfer CACS (NOTE 1) transfer CICS as CIS (TI=1)
	CACS + CICS + EAST	Transfer CACS (NOTE 1), transfer CICS as CIS (TI=1) and release EAST
	CACS + CICS + EPST	Transfer CACS (NOTE 1), transfer CICS as CIS (TI=1) and release EPST
	CACS only	Transfer CACS (NOTE 1)
	CICS only	Transfer CICS (NOTE 1)

UE supports MAM+ ALP only and Network supports at least MAM+ALP or UE supports at least MAM+ ALP and Network supports MAM+ALP only	CACS + CIS	Transfer CACS (NOTE 1) and transfer CIS (TI=1)
	CAS + CICS	Transfer CAS (TI=0) and transfer CICS (NOTE 2)
	CACS + EAST	Transfer CACS (NOTE 1) and transfer EAST (TI=1)
	CACS + EPST	Transfer CACS (NOTE 1) and release EPST
	CICS + EASO	Transfer CICS (NOTE 1) and transfer EASO (TI=1)
	CICS + EAST	Transfer CICS (NOTE 1) and transfer EAST (TI=1)
	CICS + EPSO	Transfer CICS (NOTE 1) and release EPSO
	CACS + CIS + EAST	Transfer CACS (NOTE 1), transfer CIS (TI=1), release EAST
	CAS + CICS + EAST	Transfer CAS (TI=0), transfer CICS (NOTE 2), release EAST
	CACS + CIS + EPST	Transfer CACS (NOTE 1), transfer CIS (TI=1), release EPST
	CAS + CICS + EPST	Transfer CAS (TI=0), transfer CICS (NOTE 2), release EAST
	CACS + CICS	Transfer CACS (NOTE 1) and transfer CICS as CIS (TI=1)
	CACS + CICS + EAST	Transfer CACS (NOTE 1), transfer CICS as CIS (TI=1) and release EAST
	CACS + CICS + EPST	Transfer CACS (NOTE 1), transfer CICS as CIS (TI=1) and release EPST
UE has no MAM support and Network supports MAM only or UE supports MAM only and Network has no MAM support	CACS only	Transfer CACS as CAS (TI=0)
	CICS only	Send SIP 480 (Temporarily Unavailable) to reject SIP INVITE request due to STN-SR
	CACS + CIS	Transfer CACS as CAS (TI=0) and release CIS
	CAS + CICS	Transfer CAS (TI=0) and release CICS
	CACS + EAST	Transfer CACS as CAS (TI=0) and release EAST
	CACS + EPST	
	CICS + EASO	Send SIP 480 (Temporarily Unavailable) to reject SIP INVITE request due to STN-SR
	CICS + EAST	Send SIP 480 (Temporarily Unavailable) to reject SIP INVITE request due to STN-SR
	CICS + EPSO	Send SIP 480 (Temporarily Unavailable) to reject SIP INVITE request due to STN-SR
	CACS + CIS + EAST	Transfer CACS as CAS (TI=0), release CIS and release EAST
	CAS + CICS + EAST	Transfer CAS (TI=0), release CICS and release EAST
	CACS + CIS + EPST	Transfer CACS as CAS (TI=0), release CIS and release EPST
	CAS + CICS + EPST	Transfer CAS (TI=0), release CICS and release EAST
	CACS + CICS	Transfer CACS as CAS (TI=0) and release CICS
	CACS + CICS + EAST	Transfer CACS as CAS (TI=0), release CICS and release EAST
	CACS + CICS + EPST	Transfer CACS as CAS (TI=0), release CICS and release EPST
NOTE 1: TIs are calculated as in the table 9.2.1A-2 with the offsets 0, 2, 3, 4, 5 assigned to the participants in their order in the list of the extracted participants. NOTE 2: TIs are calculated as in the table 9.2.1A-2 with the offsets 0, 1, 2, 3, 4 assigned to the participants in their order in the list of the extracted participants		

F.2.4 Void

F.3 PS to CS and CS to PS dual radio access transfer

F.3.1 Key to interpret dual radio access transfer scenarios

The key to interpret the PS-CS dual radio access transfer scenarios given in table F.3.1.1 is specified below:

- "MAM" represents support of MSC server assisted mid-call feature

- "ALP" represents support of PS to CS dual radio access transfer for calls in alerting phase and CS to PS dual radio access transfer for calls in alerting phase
- "PAP" represents support of PS to CS dual radio access transfer for originating calls in pre-alerting phase and CS to PS dual radio access transfer for originating calls in pre-alerting phase
- "CAS" represents a session with active speech media component, which is a confirmed dialog
- "CIS" represents a session with inactive speech media component, which is a confirmed dialog
- "EAS" represents a session with active speech media component in alerting phase
- "EAST" represents a terminating session with active speech media component in alerting phase
- "EASO" represents an originating session with active speech media component in alerting phase
- "EPS" represents a session with active speech media component in pre-alerting phase
- "EPST" represents a terminating session with active speech media component in pre-alerting phase
- "EPSO" represents an originating session with active speech media component in pre-alerting phase
- "CACS" represents a session with active speech media component and with conference control, which is a confirmed dialog
- "CICS" represents a session with inactive speech media component and with conference control, which is a confirmed dialog
- "Transfer CACS as CAS" refers to the behaviour of the UE to transfer CACS, but without conference control
- "Transfer CICS as CIS" refers to the behaviour of the UE to transfer CICS, but without conference control
- The TI value in the "Result in UE" column:
  - (TI set by UE) or (TI set by MSC server) indicates that the TI flag and TI value will be set as part of the 3GPP TS 24.008 [8] procedures by either the UE or the MSC server; and
  - TI=<value> indicates that the TI flag is assigned as in the terminating mobile case with the TI value according to the <value>.

NOTE: In the CS to PS dual radio access transfer the TI value is not required.

F.3.2 PS to CS dual radio access transfer scenarios

Table F.3.2.1: PS to CS dual radio access transfer scenarios

UE and Network Support Combination	Access Transfer Scenario	Result in UE
UE supports at least MAM and Network supports MAM only or UE supports MAM only and Network supports at least MAM	CAS only	Transfer CAS (TI set by UE) (NOTE 4)
	CIS only	Transfer CIS (TI set by UE)
	EAS only	Release EAS
	EPS only	Release EPS
	CAS + CIS	Transfer CAS (TI set by UE) and transfer CIS (TI=1)
	CAS + EASO	Transfer CAS (TI set by UE) and release EASO
	CAS + EPSO	Transfer CAS (TI set by UE) and release EPSO
	CAS + EAST	Transfer CAS (TI set by UE) and release EAST
	CAS + EPST	Transfer CAS (TI set by UE) and release EPST
	CIS + EASO	Transfer CIS (TI set by UE) and release EASO
	CIS + EAST	Transfer CIS (TI set by UE) and release EAST
	CIS + EPSO	Transfer CIS (TI set by UE) and release EPSO
	CAS + CIS + EAST	Transfer CAS (TI set by UE), transfer CIS (TI=1) and release EAST
	CAS + CIS + EPST	Transfer CAS (TI set by UE), transfer CIS (TI=1) and release EPST
	CAS only	Transfer CAS (TI set by UE) (NOTE 4)
	CIS only	Release CIS
	EASO only	Transfer EASO (TI set by UE)

UE supports ALP only and Network supports at least ALP or UE supports at least ALP and network supports ALP only	EAST only	Transfer EASO (TI set by MSC server) (NOTE 4)
	EPS only	Release EPS
	CAS + CIS	Transfer CAS (TI set by UE) and release CIS
	CAS + EASO	Transfer CAS (TI set by UE) and release EASO (NOTE 1)
	CAS + EASP	Transfer CAS (TI set by UE) and release EPSO
	CAS + EAST	Transfer CAS (TI set by UE) and transfer EAST (TI set by MSC server) (NOTE 4)
	CAS + EPST	Transfer CAS (TI set by UE) and release EPST
	CIS + EASO	Transfer EASO (TI set by UE) and release CIS
	CIS + EAST	Transfer EAST (TI set by UE) and release CIS
	CIS + EPSO	Release CIS and release EPSO
	CAS + CIS + EAST	Transfer CAS (TI set by UE), transfer EAST (TI set by MSC server) and release CIS
	CAS + CIS + EPST	Transfer CAS (TI set by UE), release CIS released and release EPST
UE supports ALP+PAP only and Network supports at least ALP+PAP or UE supports at least ALP+PAP and Network supports ALP+PAP only	CAS only	Transfer CAS (TI set by UE) (NOTE 4)
	CIS only	Release CIS
	EAS only	Transfer EAS (TI set by UE)
	EPSO only	Transfer EPSO (TI set by UE)
	EPST only	Release EPST
	CAS + CIS	Transfer CAS (TI set by UE) and release CIS
	CAS + EASO	Transfer CAS and release EASO (NOTE 1)
	CAS + EPSO	Transfer CAS and release EPSO (NOTE 2)
	CAS + EAST	Transfer CAS (TI set by UE) and transfer EAST (TI set by MSC server) (NOTE 4)
	CAS + EPST	Transfer CAS (TI set by UE) and release EPST
	CIS + EASO	Transfer EASO (TI set by UE) and release CIS
	CIS + EAST	Transfer EAST (TI set by UE) and release CIS
	CIS + EPSO	Transfer EPSO (TI set by UE) and release CIS
	CAS + CIS + EAST	Transfer CAS (TI set by UE), transfer EAST (TI=1) and release CIS
	CAS + CIS + EPST	Transfer CAS (TI set by UE), release CIS and release EPST
UE and Network support MAM+ALP+PAP	CAS only	Transfer CAS (TI set by UE) (NOTE 4)
	CIS only	Transfer CIS (TI set by UE)
	EAS only	Transfer EAS (TI set by UE)
	EPSO only	Transfer EPSO (TI set by UE) (NOTE 4)
	EPST only	Release EPST
	CAS + CIS	Transfer CAS (TI set by UE) and transfer CIS (TI=1)
	CAS + EASO	Transfer CAS and release EASO (NOTE 1)
	CAS + EPSO	Transfer CAS and release EPSO (NOTE 2)
	CAS + EAST	Transfer CAS (TI set by UE) and transfer EAST (TI set by MSC server)
	CAS + EPST	Transfer CAS (TI set by UE) and release EPST
	CIS + EASO	Transfer CIS (TI set by UE) and transfer EASO (TI=1)
	CIS + EAST	Transfer CIS (TI set by UE) and transfer EAST (TI=1)
	CIS + EPSO	Transfer CIS (TI set by UE) and transfer EPSO (TI=1)
	CAS + CIS + EAST	Transfer CAS (TI set by UE), transfer CIS (TI=1) and release EAST (NOTE 3)
	CAS + CIS + EPST	Transfer CAS (TI set by UE), transfer CIS (TI set by MSC server) and release EPST
UE supports MAM+ ALP only and Network supports at least MAM+ALP or UE supports at least MAM+ ALP and Network supports MAM+ALP only	CAS only	Transfer CAS (TI set by UE) (NOTE 4)
	CIS only	Transfer CIS (TI set by UE)
	EAS only	Transfer EAS (TI set by UE)
	EPS only	Release EPS
	CAS + CIS	Transfer CAS (TI set by UE) and transfer CIS (TI=1)
	CAS + EASO	Transfer CAS and release EASO (NOTE 1)
	CAS + EASP	Transfer CAS and release EPSO
	CAS + EAST	Transfer CAS (TI set by UE) and transfer EAST (TI set by MSC server)
	CAS + EPST	Transfer CAS (TI set by UE) and release EPST
	CIS + EASO	Transfer CIS (TI set by UE) and transfer EASO (TI=1)
	CIS + EAST	Transfer CIS (TI set by UE) and transfer EAST (TI=1)
	CIS + EPSO	Transfer CIS (TI set by UE) and release EPSO
	CAS + CIS + EAST	Transfer CAS (TI set by UE), transfer CIS (TI=1) and release EAST (NOTE 3)
	CAS + CIS + EPST	Transfer CAS (TI set by UE), Transfer CIS (TI=1) and release EPST

UE has no MAM support and Network supports MAM only or UE supports MAM only and Network has no MAM support	CAS only	Transfer CAS (TI set by UE) (NOTE 4)
	CIS only	Release CIS
	EAS only	Release CIS
	EPS only	Release CIS
	CAS + CIS	Transfer CAS (TI set by UE) and release CIS
	CAS + EAST	Transfer CAS (TI set by UE) and release EAST
	CAS + EPST	Transfer CAS (TI set by UE) and release EPST
	CIS + EASO	Release CIS and release EASO
	CIS + EAST	Release CIS and release EAST
	CIS + EPSO	Release CIS and release EPSO
	CAS + CIS + EAST	Transfer CAS (TI set by UE), release CIS and release EAST
	CAS + CIS + EPST	Transfer CAS (TI set by UE), release CIS and release EPST
NOTE 1: Even though the nework and the UE support transfer of EASO, the EASO is released since the combination CAS and EASO cannot occur in CS. NOTE 2: Even though the nework and the UE support transfer of EPSO, the EPSO is released since the combination CAS and EPSO cannot occur in CS. NOTE 3: Even though the UE and network can support the transfer of EAST, the UE will only transfer CAS and CIS and will release EAST since only two calls can be transferred to CS. NOTE 4: Transfer of CAS, EAST or CAS+EAST do not require a MSC server with a direct SIP interface to IMS.		

F.3.3 CS to PS dual radio access transfer scenarios

Table F.3.3.1: CS to PS dual radio access transfer scenarios

UE and Network Support Combination (NOTE)	Access Transfer Scenario	Result in UE
UE supports at least MAM and Network supports MAM only or UE supports MAM only and Network supports at least MAM	CAS only	Transfer CAS
	CIS only	Transfer CIS
	EAS only	Release EAS
	EPS only	Release EPS
	CAS + CIS	Transfer CAS and transfer CIS
	CAS + EAST	Transfer CAS and release EAST
	CAS + EPST	Transfer CAS and transfer EPST
	CIS + EASO	Transfer CIS and release EASO
	CIS + EAST	Transfer CIS and release EAST
	CIS + EPSO	Transfer CIS and release EPSO
UE supports ALP only and Network supports at least ALP or UE supports at least ALP and network supports ALP only	CAS only	Transfer CAS
	CIS only	Release CIS
	EAS only	Transfer EAS
	EPS only	Release EPS
	CAS + CIS	Transfer CAS and release CIS
	CAS + EAST	Transfer CAS and transfer EAST
	CAS + EPST	Transfer CAS and release EPST
	CIS + EASO	Transfer EASO and release CIS
	CIS + EAST	Transfer EAST and release CIS
	CIS + EPSO	CIS released by UE and release EPSO
UE supports ALP+PAP only and Network supports at least ALP+PAP or UE supports at least ALP+PAP and Network supports ALP+PAP only	CAS only	Transfer CAS
	CIS only	Release CIS
	EAS only	Transfer EAS
	EPSO only	Transfer EPSO
	EPST only	Release EPST
	CAS + CIS	Transfer CAS and release CIS
	CAS + EAST	Transfer CAS and transfer EAST
	CAS + EPST	Transfer CAS and release EPST
	CIS + EASO	Transfer EASO and release CIS
	CIS + EAST	Transfer EAST and release CIS
	CIS + EPSO	Transfer EPSO and release CIS
UE and Network support MAM+ALP+PAP	CAS only	Transfer CAS
	CIS only	Transfer CIS
	EAS only	Transfer EAS
	EPSO only	Transfer EPSO
	EPST only	Release EPST

	CAS + CIS	Transfer CAS and transfer CIS
	CAS + EAST	Transfer CAS and transfer EAST
	CAS + EPST	Transfer CAS and release EPST
	CIS + EASO	Transfer CIS and transfer EASO
	CIS + EAST	Transfer CIS and transfer EAST
	CIS + EPSO	Transfer CIS and transfer EPSO
UE supports MAM+ ALP only and Network supports at least MAM+ALP or UE supports at least MAM+ ALP and Network supports MAM+ALP only	CAS only	Transfer CAS
	CIS only	Transfer CIS
	EAS only	Transfer EAS
	EPS only	Release EPS
	CAS + CIS	Transfer CAS and transfer CIS
	CAS + EAST	Transfer CAS and transfer EAST
	CAS + EPST	Transfer CAS and release EPST
	CIS + EASO	Transfer CIS and transfer EASO
	CIS + EAST	Transfer CIS and transfer EAST
	CIS + EPSO	Transfer CIS and release EPSO
UE has no MAM support and Network supports MAM only or UE supports MAM only and Network has no MAM support	CAS only	Transfer CAS
	CIS only	Release CIS
	EAS only	Release CIS
	EPS only	Release CIS
	CAS + CIS	Transfer CAS and release CIS
	CAS + EAST	Transfer CAS and release EAST
	CAS + EPST	Transfer CAS and release EPST
	CIS + EASO	Release CIS and release EASO
	CIS + EAST	Release CIS and release EAST
	CIS + EPSO	Release CIS and release EASO
NOTE: None of the CS to PS dual radio access transfer scenarios requires an MSC server with a direct SIP interface to IMS.		

F.3.4 PS to CS dual access transfer scenarios with conference control

Table F.3.4-1 describes PS to CS dual radio access transfer scenarios considering cases when at least one potential session to transfer is a conference initiated by the UE.

Table F.3.4-1: PS to CS dual radio access transfer scenarios with conference control

UE and Network Support Combination	Access Transfer Scenario	Result in UE
UE supports at least MAM and Network supports MAM only or UE supports MAM only and Network supports at least MAM	CACS only	Transfer CACS (NOTE 1).
	CICS only	Transfer CICS (NOTE 1).
	CACS + CIS	Transfer CACS (NOTE 1) and transfer CIS (TI=1).
	CAS + CICS	Transfer CAS (TI set by UE) and transfer CICS (NOTE 2).
	CACS + EAST	Transfer CACS (NOTE 1) and release EAST.
	CACS + EPST	Transfer CACS (NOTE 1) and release EPST.
	CICS + EASO	Transfer CICS (NOTE 1) and release EASO.
	CICS + EPSO	Transfer CICS (NOTE 1) and release EPSO.
	CACS + CIS + EAST	Transfer CACS (NOTE 1), transfer CIS (TI=1) and release EAST.
	CAS + CICS + EAST	Transfer CAS (TI set by UE), transfer CICS (NOTE 2) and release EAST.
	CACS + CIS + EPST	Transfer CACS (NOTE 1), transfer CIS (TI=1) and release EPST.
	CAS + CICS + EPST	Transfer CAS (TI set by UE) transfer CICS (NOTE 2) and release EPST.
	CACS + CICS	Transfer CACS (NOTE 1) and transfer CICS as CIS (TI=1).
	CACS + CICS + EAST	Transfer CACS (NOTE 1), transfer CICS as CIS (TI=1) and release EAST.
	CACS + CICS + EPST	Transfer CACS (NOTE 1), transfer CICS as CIS (TI=1) and release EPST.

UE supports ALP only and Network supports at least ALP or UE supports at least ALP and network supports ALP only	CACS only	Transfer CACS as CAS (TI set by UE).
	CICS only	Release CICS.
	CACS + CIS	Transfer CACS as CAS (TI set by UE) and release CIS.
	CAS + CICS	Transfer CAS (TI set by UE) and release CICS.
	CACS + EAST	Transfer CACS as CAS (TI set by UE) and release EAST.
	CACS + EPST	Transfer CACS as CAS (TI set by UE) and release EPST.
	CICS + EASO	Transfer EASO (TI set by UE) and release CICS
	CICS + EPSO	Release CICS and release EPSO.
	CACS + CIS + EAST	Transfer CACS as CAS (TI set by UE), transfer EAST (TEI=1) and release CIS.
	CAS + CICS + EAST	Transfer CAS (TI=0), transfer EAST (TI=1) and release CICS.
	CACS + CIS + EPST	Transfer CACS as CAS (TI set by UE), release CIS and release EPST.
	CAS + CICS + EPST	Transfer CAS (TI set by UE), release CICS and release EPST.
	CACS + CICS	Transfer CACS as CAS (TI set by UE) and release CICS.
	CACS + CICS + EAST	Transfer CACS as CAS (TI set by UE), transfer EAST and release CICS.
	CACS + CICS + EPST	Transfer CACS as CAS (TI set by UE), release CICS and release EPST.
UE supports ALP+PAP only and Network supports at least ALP+PAP or UE supports at least ALP+PAP and Network supports ALP+PAP only	CACS only	Transfer CACS as CAS (TI set by UE).
	CICS only	Release CICS.
	CACS + CIS	Transfer CACS as CAS (TI set by UE) and release CIS.
	CAS + CICS	Transfer CAS (TI set by UE) and release CICS.
	CACS + EAST	Transfer CACS as CAS (TI set by UE) and transfer EAST (TI=1).
	CACS + EPST	Transfer CACS as CAS (TI set by UE) and release EPST.
	CICS + EASO	Transfer EASO (TI set by UE) and release CICS.
	CICS + EPSO	Transfer EPSO (TI set by UE) and release CICS.
	CACS + CIS + EAST	Transfer CACS as CAS (TI set by UE), release CIS and release EAST.
	CAS + CICS + EAST	Transfer CAS (TI set by UE), transfer EAST (TI=1) and release CICS.
	CACS + CIS + EPST	Transfer CACS as CAS (TI set by UE), release CIS and release EPST.
	CAS + CICS + EPST	Transfer CAS (TI=0), release CICS and release EPST.
	CACS + CICS	Transfer CACS as CAS (TI set by UE) and release CICS.
	CACS + CICS + EAST	Transfer CACS as CAS (TI set by UE), transfer EAST (TI=1) and release CICS.
	CACS + CICS + EPST	Transfer CACS as CAS (TI set by UE), release CICS and release EPST.
UE and Network support MAM+ALP+PAP	CACS only	Transfer CACS (NOTE 1).
	CICS only	Transfer CICS (NOTE 1).
	CACS + CIS	Transfer CACS (NOTE 1) and transfer CIS (TI=1).
	CAS + CICS	Transfer CAS (TI set by UE) and transfer CICS (NOTE 2).
	CACS + EAST	Transfer CACS (NOTE 1) and transfer EAST (TI=1).
	CACS + EPST	Transfer CACS (NOTE 1) and release EPST.
	CICS + EASO	Transfer CICS (NOTE 1) and transfer EASO (TI=1).
	CICS + EPSO	Transfer CICS (NOTE 1) and transfer EPSO (TI=1).
	CACS + CIS + EAST	Transfer CACS (NOTE 1), transfer CIS (TI=1) and release EAST.
	CAS + CICS + EAST	Transfer CAS (TI set by UE), transfer CICS (NOTE 2) and release EAST.
	CACS + CIS + EPST	Transfer CACS (NOTE 1), transfer CIS (TI=1) and release EPST
	CAS + CICS + EPST	Transfer CAS (TI set by UE), transfer CICS (NOTE 2) and release EPST.
	CACS + CICS	Transfer CACS (NOTE 1), transfer CICS as CIS (TEI=1)
	CACS + CICS + EAST	Transfer CACS (NOTE 1), transfer CICS as CIS (TEI=1) and release EAST.
	CACS + CICS + EPST	Transfer CACS (NOTE 1), transfer CICS as CIS (TEI=1) and release EPST.
	CACS only	Transfer CACS (NOTE 1).
	CICS only	Transfer CICS (NOTE 1).

UE supports MAM+ ALP only and Network supports at least MAM+ALP or UE supports at least MAM+ ALP and Network supports MAM+ALP only	CACS + CIS	Transfer CACS (NOTE 1) and transfer CIS (TI=1).
	CAS + CICS	Transfer CAS (TI set by UE) and transfer CICS (NOTE 2).
	CACS + EAST	Transfer CACS (NOTE 1) and transfer EAST (TI=1).
	CACS + EPST	Transfer CACS (NOTE 1) and release EPST.
	CICS + EASO	Transfer CICS (NOTE 1) and transfer EASO (TI=1).
	CICS + EPSO	Transfer CICS (NOTE 1) and release EPSO.
	CACS + CIS + EAST	Transfer CACS (NOTE 1), transfer CIS (TI=1) and release EAST.
	CAS + CICS + EAST	Transfer CAS (TI=0), transfer CICS (NOTE 2) and release EAST.
	CACS + CIS + EPST	Transfer CACS (NOTE 1), transfer CIS (TI=1) and release EPST.
	CAS + CICS + EPST	Transfer CAS (TI set by UE), transfer CICS (NOTE 2) and release EAST.
	CACS + CICS	Transfer CACS (NOTE 1) and transfer CICS as CIS (TEI=1).
	CACS + CICS + EAST	Transfer CACS (NOTE 1), transfer CICS as CIS (TEI=1) and release EAST.
	CACS + CICS + EPST	Transfer CACS (NOTE 1), transfer CICS as CIS (TEI=1) and release EPST.
UE has no MAM support and Network supports MAM only or UE supports MAM only and Network has no MAM support	CACS only	Transfer CACS as CAS (TI set by UE).
	CICS only	Release CIS.
	CACS + CIS	Transfer CACS as CAS (TI set by UE) and release CIS.
	CAS + CICS	Transfer CAS (TI set by UE) and release CICS.
	CACS + EAST	Transfer CACS as CAS (TI set by UE) and release EAST.
	CACS + EPST	Transfer CACS as CAS (TI set by UE) and release EPST.
	CICS + EASO	Release CICS and release EASO.
	CICS + EPSO	Release CICS and release EPSO.
	CACS + CIS + EAST	Transfer CACS as CAS (TI set by UE) and release EAST.
	CAS + CICS + EAST	Transfer CAS (TI set by UE), release CICS and release EAST.
	CACS + CIS + EPST	Transfer CACS as CAS (TI set by UE), release CIS and release EPST.
	CAS + CICS + EPST	Transfer CAS (TI set by UE), release CICS and release EAST.
	CACS + CICS	Transfer CACS as CAS (TI set by UE) and release CICS.
	CACS + CICS + EAST	Transfer CACS as CAS (TI set by UE), release CICS and release EAST.
	CACS + CICS + EPST	Transfer CACS as CAS (TI set by UE), release CICS and release EAST.

NOTE 1: The TI value for the conference is assigned by 3GPP TS 24.008 [8] procedures. TIs are calculated as in the table 9.2.1A-2 with the offsets 0, 2, 3, 4, 5 assigned to the participants in their order in the list of the extracted participants. The transaction identifier of the conference is identical to the transaction identifier assigned to the first participant (offset 0).

NOTE 2: TIs for participants are calculated as in the table 9.2.1A-2 with the offsets 0, 1, 2, 3, 4 assigned to the participants in their order in the list of the extracted participants. The transaction identifier of the conference is identical to the transaction identifier assigned to the first participant (offset 0).

Annex G (informative): Indication of support for access transfer features

G.1 Introduction

This annex describes how support is indicated for theaccess transfer features described in this document. Specifically:

- How the SC UE indicates support to the network; and
- How the SCC AS and ATCF indicate support to the SC UE.

G.2 SC UE indications of support of access transfer features towards the network

Table G.2-1: SC UE indications of support of access transfer features to the network

Access Transfer Feature (NOTE 1)	**SC UE indication of support to the network**
PS to CS SRVCC (Single Active Call)	The SC UE includes a capability indicator (included by the lower layers) during UE attach and updating procedures (see 3GPP TS 24.301 [52] and 3GPP TS 24.008 [8]).
PS to CS SRVCC with MSC server assisted mid-call feature	The SC UE includes the "g.3gpp.mid-call" media feature tag in the Contact header of the initial SIP INVITE request. Upon receiving a SIP INVITE request including the g.3gpp.mid-call feature-capability indicator in the Feature-Caps header field, the SC UE includes the "g.3gpp.mid-call media feature" tag in the Contact header field of the SIP 2xx response.
PS to CS SRVCC for calls in alerting phase	The SC UE includes the "g.3gpp.srvcc-alerting" media feature tag in the Contact header of the initial SIP INVITE request. Upon receiving a SIP INVITE request including the "g.3gpp.srvcc-alerting" feature-capability indicator in the Feature-Caps header field, the SC UE includes the "g.3gpp.srvcc-alerting" media feature tag in the Contact header field of the SIP1xx and SIP 2xx responses.
PS to CS SRVCC for originating calls in pre-alerting phase	The SC UE includes the "g.3gpp.ps2cs-srvcc-orig-pre-alerting" media feature tag in the Contact header of the initial SIP INVITE request.
CS to PS SRVCC (Single Active Call)	The SC UE includes a capability indicator (included by the lower layers) during UE attach and updating procedures (see 3GPP TS 24.008 [8]) in order to inform the BSS of CS to PS SRVCC support from GERAN. The SC UE includes the "g.3gpp.cs2ps-srvcc" media feature tag in the Contact header field of the SIP REGISTER request.
CS to PS SRVCC with MSC server assisted mid-call feature	The SC UE includes an Accept header in the SIP INVITE request due to STI-rSR containing the application/vnd.3gpp.mid-call+xml MIME type for support of the additional session transfer.
CS to PS SRVCC for calls in the alerting phase	The SC UE includes the "g.3gpp.cs2ps-srvcc-alerting" media feature tag in the Contact header field of the SIP REGISTER request. The SC UE includes an Accept header in the SIP INVITE request due to STI-rSR containing the application/vnd.3gpp.state-and-event-info+xml for support of the alerting session transfer.
PS to CS dual radio access transfer (single active call)	For support of access transfer for an SC UE without ICS capabilities, the SC UE uses a dynamic STN for the access transfer if it supports the use of dynamic STN and receives the dynamic STN from the SCC AS. Otherwise, the SC UE uses a static STN. The SC UE indicates support for the dynamic STN by including the "g.3gpp.dynamic-stn" media feature tag in the Contact header of the initial SIP INVITE request and in the Contact header of SIP 1xx and SIP 2xx responses to the SIP INVITE request. Upon receiving the dynamic STN in the "g.3gpp.dynamic-stn" feature-capability indicator in the Feature-Caps header field of a SIP INVITE request, SIP 1xx or SIP 2xx response, and the SC supports the use of the dynamic STN, the SC UE replaces the static STN with the dynamic STN.
PS to CS dual radio access transfer with MSC server assisted mid-call feature	The SC UE includes the "g.3gpp.mid-call" media feature tag in the Contact header of the initial SIP INVITE request. Upon receiving a SIP INVITE request including the "g.3gpp.mid-call" feature-capability indicator in the Feature-Caps header field, the SC UE includes the "g.3gpp.mid-call media" feature tag in the Contact header field of the SIP 2xx response.

PS to CS dual radio access transfer for calls in alerting phase	The SC UE includes the "g.3gpp.drvcc-alerting" media feature tag in the Contact header of the initial SIP INVITE request. Upon receiving a SIP INVITE request including the "g.3gpp.drvcc-alerting" feature-capability indicator in the Feature-Caps header field, the SC UE includes the "g.3gpp.drvcc-alerting media" feature tag in the Contact header field of all SIP 18x responses to the SIP INVITE request.
PS to CS dual radio access transfer for originating calls in pre-alerting phase	The SC UE includes the "g.3gpp.ps2cs-drvcc-orig-pre-alerting" media feature tag in the Contact header field of the initial SIP INVITE request.
CS to PS dual radio access transfer (single active call)	The SC UE uses a dynamic STI for the access transfer if it is configured to use ICS capabilities. Otherwise, the SC UE uses a static STI.
CS to PS dual radio access transfer with MSC server assisted mid-call feature	The SC UE includes an Accept header in the SIP INVITE request for static STI containing the application/vnd.3gpp.mid-call+xml MIME type, for support of the additional session transfer.
CS to PS dual radio access transfer for calls in the alerting phase	The SC UE (not using ICS capabilities) includes the "g.3gpp.cs2ps-drvcc-alerting" media feature tag in the Contact header field of the SIP INVITE request for static STI.
CS to PS dual radio access transfer for originating calls in the pre-alerting phase	The SC UE (not using ICS capabilities) includes the "g.3gpp.cs2ps-drvcc-orig-pre-alerting" media feature tag in the Contact header field of the SIP INVITE request for static STI.
PS to PS acess transfer	When the "PS to PS STI" is configured, the SC UE includes the "g.3gpp.pstops-sti" media feature tag in the Contact header field of a SIP REGISTER request.
PS to CS dual radio access transfer for emergency session	The SC UE includes the "g.3gpp.dynamic-e-stn-drvcc" media feature tag in the Contact
NOTE 1: ICS capabilities can be used by the SC UE for support of access transfer. Configuration for ICS is specified in 3GPP TS 24.292 [4].	

G.3 Network indications of support of access transfer features towards the SC UE

Table G.3-1: Network indications of support of access transfer features to the SC UE

Access Transfer Feature	SCC AS, ATCF and EATF indication of support to the SC UE
PS to CS SRVCC (Single Active Call)	The SCC AS indicates to the SC UE that a session is anchored in the SCC AS by including the "g.3gpp.srvcc" feature capability indicator in the Feature-Caps header field of an initial SIP INVITE request and in a SIP 1xx response or SIP 2xx response to an initial SIP INVITE request towards the SC UE. When the ATCF decides to include itself for access transfer of sessions, then upon receiving a SIP 2xx response to the SIP REGISTER request sent from the SC UE, the ATCF includes the g.3gpp.atcf feature-capability indicator containing the STN-SR allocated to ATCF in the Feature-Caps header field.
PS to CS SRVCC with MSC server assisted mid-call feature	The SCC AS includes the "g.3gpp.mid-call" feature-capability indicator in the Feature-Caps header field of an initial SIP INVITE request due to terminating filter criteria towards the SC UE. Upon receiving a SIP INVITE request due to originating filter criteria including the "g.3gpp.mid-call" media feature tag in the Contact header field, the SCC AS includes the "g.3gpp.mid-call" capability indicator in the Feature-Caps header field of the SIP 1xx and SIP 2xx responses.
PS to CS SRVCC for calls in alerting phase	The SCC AS includes the "g.3gpp.srvcc-alerting" feature-capability indicator in the Feature-Caps header field of an initial SIP INVITE request due to terminating filter criteria towards the SC UE. Upon receiving a SIP INVITE request due to originating filter criteria including the "g.3gpp.srvcc-alerting" media feature tag in the Contact header field, the SCC AS includes the "g.3gpp.srvcc-alerting" capability indicator in the Feature-Caps header field of the SIP 1xx and SIP 2xx responses.
PS to CS SRVCC for originating calls in pre-alerting phase	Upon receiving a SIP INVITE request due to originating filter criteria including the "g.3gpp.ps2cs-srvcc-orig-pre-alerting" media feature tag in the Contact header field, the SCC AS includes the "g.3gpp.ps2cs-srvcc-orig-pre-alerting" capability indicator in the Feature-Caps header field of the SIP 1xx and SIP 2xx responses.
CS to PS SRVCC (Single Active Call)	Upon receiving a SIP REGISTER request including the "g.3gpp.cs2ps-srvcc" media feature tag in the Contact header field, then on receiving the SIP 2xx response to the SIP REGISTER request, the ATCF includes the "g.3gpp.cs2ps-srvcc" feature-capability indicator in the Feature-Caps header field of the SIP 2xx response.
CS to PS SRVCC with MSC server assisted mid-call feature	None.
CS to PS SRVCC for calls in alerting phase	None.
PS to CS dual radio access transfer (single active call)	For support of access transfer for an SC UE without ICS capabilities the SC UE is configured with a static STN as defined in the Communication Continuity Management Object in 3GPP TS 24.216 [5]. However, the SCC AS can send a dynamic STN to the SC UE to replace the static STN. Upon receiving a SIP INVITE request due to originating filter criteria including the "g.3gpp.dynamic-stn" media feature tag in the Contact header field, and the SCC AS supports use of the dynamic STN, the SCC AS includes the dynamic STN in the "g.3gpp.dynamic-stn" feature-capability indicator in the Feature-Caps header field of SIP 1xx and SIP 2xx responses. Upon receiving a SIP INVITE request due to terminating filter criteria, and the SCC AS supports use of the dynamic STN, the SCC AS includes the dynamic STN in the "g.3gpp.dynamic-stn" feature-capability indicator in the Feature-Caps header field of the SIP INVITE request towards the SC UE.

PS to CS dual radio access transfer with MSC server assisted mid-call feature	The SCC AS includes the "g.3gpp.mid-call" feature-capability indicator in the Feature-Caps header field of an initial SIP INVITE request due to terminating filter criteria towards the SC UE. Upon receiving a SIP INVITE request due to originating filter criteria including the "g.3gpp.mid-call" media feature tag in the Contact header field, the SCC AS includes the "g.3gpp.mid-call" capability indicator in the Feature-Caps header field of the SIP 2xx response.
PS to CS dual radio access transfer for calls in alerting phase	The SCC AS includes the "g.3gpp.drvcc-alerting media" feature-capability indicator in the Feature-Caps header field of an initial SIP INVITE request due to terminating filter criteria towards the SC UE. Upon receiving a SIP INVITE request due to originating filter criteria including the "g.3gpp.drvcc-alerting media" feature tag in the Contact header field, the SCC AS includes the "g.3gpp.drvcc-alerting" feature-capability indicator in the Feature-Caps header field of SIP 1xx and SIP 2xx responses.
PS to CS dual radio access transfer for originating calls in pre-alerting phase	Upon receiving a SIP INVITE request due to originating filter criteria including the "g.3gpp.ps2cs-drvcc-orig-pre-alerting" media feature tag in the Contact header field of the initial SIP INVITE request, the SCC AS includes the "g.3gpp.ps2cs-drvcc-orig-pre-alerting" feature-capability indicator in the Feature-Caps header field of SIP 1xx and SIP 2xx responses.
CS to PS dual radio access transfer (single active call)	For support of access transfer for an SC UE without ICS capabilities, the SC UEis configured with a static STI as defined in the Communication Continuity Management Object in 3GPP TS 24.216 [5]. For support of access transfer for an SC UE with ICS capabilities, the SCC AS assigns a dynamic STI.
CS to PS dual radio access transfer with MSC server assisted mid-call feature	Upon receiving a SIP INVITE request due to static STI containing the"g.3gpp.mid-call" media feature tag and further upon receiving the SIP 2xx response to the SIP re-INVITE request (that was sent to the remote UE) the SCC AS includes the "g.3gpp.mid-call" feature-capability indicator in the Feature-Caps header field of the SIP 2xx response.
CS to PS dual radio access transfer for calls in the alerting phase	Upon receiving a SIP INVITE request due to static STI containing the "g.3gpp.cs2ps-drvcc-alerting" media feature tag and further upon receiving the SIP 2xx response to the SIP re-INVITE request (that was sent to the remote UE), the SCC AS includes the "g.3gpp.cs2ps-drvcc-alerting" feature-capability indicator in the Feature-Caps header field of the SIP 2xx response to the SIP INVITE request due to static STI.
CS to PS dual radio access transfer for originating calls in the pre-alerting phase	Upon receiving a SIP INVITE request due to static STI containing the "g.3gpp.cs2ps-drvcc-orig-pre-alerting " media feature tag and further upon receiving the SIP 2xx response to the SIP re-INVITE request (that was sent to the remote UE), the SCC AS includes the "g.3gpp.cs2ps-drvcc-orig-pre-alerting " feature-capability indicator in the Feature-Caps header field of the SIP 2xx response to the SIP INVITE request due to static STI.
PS to PS acess transfer	The SC UE is configured with a "PS to PS STI" as defined in the Communication Continuity Management Object (MO) in 3GPP TS 24.216 [5].
PS to CS dual radio access transfer for emergency session	Upon receiving a SIP INVITE request with an emergency URN containing the "g.3gpp.dynamic-e-stn-drvcc, the EATF includes the "g.3gpp.dynamic-e-stn-drvcc" feature-capability indicator in the Feture-Caps header field of the SIP 2xx response the the SIP INVITE request with an emergency URN .

Annex H (informative): Change history

Change history							
Date	**TSG #**	**TSG Doc.**	**CR**	**Rev**	**Subject/Comment**	**Old**	**New**
2008-05	CT1#53	C1-082007			Skeleton of TS from rapporteur	-	0.0.0
2008-05	CT1#53	C1-082008 C1-082009			Incorporate agreed P-CRs from CT1#53	0.0.0	0.1.0
2008-05	Email Review				Format update	0.1.0	0.1.1
2008-06	CT1#54	C1-082394			Pseudo-CR on Introduction for signalling flows	0.1.1	0.2.0
2008-06	CT1#54	C1-082666			Application level handover	0.1.1	0.2.0
2008-06	CT1#54	C1-082667			Role of UE in registration	0.1.1	0.2.0
2008-06	CT1#54	C1-082668			Role of UE in origination and termination	0.1.1	0.2.0
2008-06	CT1#54	C1-082669			Role of UE in PS-PS session continuity	0.1.1	0.2.0
2008-06	CT1#54	C1-082671			Pseudo-CR on Cleanup of TS 24.237	0.1.1	0.2.0
2008-06	CT1#54	C1-082672			Pseudo-CR on Roles for registration in IMS	0.1.1	0.2.0
2008-06	CT1#54	C1-082766			Pseudo-CR on Signalling flows for CS call origination	0.1.1	0.2.0
2008-06	CT1#54	C1-082767			Pseudo-CR on Signalling flows for CS call termination	0.1.1	0.2.0
2008-08	CT1#55	C1-083376			Clarifications usage of CS and ICS within IMS SC	0.2.0	0.3.0
2008-08	CT1#55	C1-083377			Functional Entities	0.2.0	0.3.0
2008-08	CT1#55	C1-083378			Signalling flows for registration	0.2.0	0.3.0
2008-08	CT1#55	C1-083379			SCC AS procedures for PS-PS session continuity	0.2.0	0.3.0
2008-08	CT1#55	C1-083380			SC UE procedures for PS-PS session continuity	0.2.0	0.3.0
2008-08	CT1#55	C1-083382			Signalling flow for Media adding/deleting	0.2.0	0.3.0
2008-10	CT1#55bis	C1-083783			PS-PS Access transfer with full media transfer	1.0.0	1.1.0
2008-10	CT1#55bis	C1-083903			Editorial Cleanup	1.0.0	1.1.0
2008-10	CT1#55bis	C1-084260			network capabilities and URI assignments for IMS SC	1.0.0	1.1.0
2008-10	CT1#55bis	C1-084261			Procedures for IMS SC call origination	1.0.0	1.1.0
2008-10	CT1#55bis	C1-084262			Procedures for call termination	1.0.0	1.1.0
2008-10	CT1#55bis	C1-084265			Signalling flow for PS-CS session continuity	1.0.0	1.1.0
2008-10	CT1#55bis	C1-084266			Signalling flow for PS-PS session continuity in conjunction with PS-CS session continuity	1.0.0	1.1.0
2008-10	CT1#55bis	C1-084268			Clarification of signalling flow for call termination	1.0.0	1.1.0
2008-10	CT1#55bis	C1-084269			Clarification of signalling flow for call origination	1.0.0	1.1.0
2008-10	CT1#55bis	C1-084433			Signalling Flows for Multiple Registrations	1.0.0	1.1.0
2008-10	CT1#55bis	C1-084447			Procedures for PS-PS session transfer in conjunction with PS-CS session transfer	1.0.0	1.1.0
2008-10	CT1#55bis	C1-084448			Procedures for adding/removing media	1.0.0	1.1.0
2008-10	CT1#55bis	C1-084449			Call flow for PS-PS partial media transfer	1.0.0	1.1.0
2008-10	CT1#55bis	C1-084506			Procedures for PS-CS session transfer	1.0.0	1.1.0
2008-10	CT1#55bis				Change the Keywords to: IMS, Multimedia Session, and Session Continuity.	1.0.0	1.1.0
2008-10	Email Review				Editorial Cleanup	1.1.0	1.1.1
2008-11	CT1#56	C1-084798			Removal of Editor's Note	1.1.1	1.2.0
2008-11	CT1#56	C1-084866			cleanup for PS-PS session transfer procedures	1.1.1	1.2.0
2008-11	CT1#56	C1-085085			Correction: No usage of CS indication in non-ICS case	1.1.1	1.2.0
2008-11	CT1#56	C1-085242			Scope of signalling flows	1.1.1	1.2.0
2008-11	CT1#56	C1-085243			Cleanup of call origination signalling flows	1.1.1	1.2.0
2008-11	CT1#56	C1-085244			Cleanup of call termination signalling flows	1.1.1	1.2.0
2008-11	CT1#56	C1-085245			Signalling flows for PS to CS session transfer	1.1.1	1.2.0
2008-11	CT1#56	C1-085246			Signalling flows for PS+CS to PS session transfer	1.1.1	1.2.0
2008-11	CT1#56	C1-085247			Clean up of SCC Registration information	1.1.1	1.2.0
2008-11	CT1#56	C1-085251			cleanups to PS-PS in conjunction with PS-CS session transfer procedures	1.1.1	1.2.0
2008-11	CT1#56	C1-085252			cleanups for media adding/removing procedures	1.1.1	1.2.0
2008-11	CT1#56	C1-085449			PS-CS session transfer procedures	1.1.1	1.2.0
2008-11	CT1#56	C1-085464			SR-VCC	1.1.1	1.2.0
2008-11	CT1#56	C1-085481			PS-PS full session transfer using Target-Dialog header	1.1.1	1.2.0
2008-11					Version 2.0.0 created for presentation to CT#42 for approval	1.2.0	2.0.0
2008-12	CT#42				Version 8.0.0 created after approval in CT#42	2.0.0	8.0.0
2009-03	CT#43	CP-090147	0002	3	Cleanup to TS 24.237	8.0.0	8.1.0
2009-03	CT#43	CP-090147	0003	1	Remove void introduction subclauses	8.0.0	8.1.0
2009-03	CT#43	CP-090147	0004	2	UE procedures for operator policy support	8.0.0	8.1.0

2009-03	CT#43	CP-090147	0005	1	Flows for originating and and terminating session in session comtinuity	8.0.0	8.1.0
2009-03	CT#43	CP-090147	0009	1	Correction SR-VCC procedures	8.0.0	8.1.0
2009-03	CT#43	CP-090147	0010	1	Correction to flows for session continuity	8.0.0	8.1.0
2009-03	CT#43	CP-090147	0011	2	Completion of IMRN functionality	8.0.0	8.1.0
2009-03	CT#43	CP-090147	0012	1	SCC AS Transparently passing Contact To and From headers	8.0.0	8.1.0
2009-03	CT#43	CP-090147	0013	3	Conveying ICS registration information using 3rd party Registration and GRUU/ICSI/IARI corrections	8.0.0	8.1.0
2009-03	CT#43	CP-090147	0014	2	Use of GRUU by SC UE	8.0.0	8.1.0
2009-03	CT#43	CP-090147	0015	2	Modification for the SCC AS procedure for CS to PS session transfer	8.0.0	8.1.0
2009-03	CT#43	CP-090147	0018	2	Definitions of Dynamic and Static STI	8.0.0	8.1.0
2009-03	CT#43	CP-090147	0019	2	Miscellaneous corrections to session continuity procedures	8.0.0	8.1.0
2009-03	CT#43	CP-090147	0020	2	SCC – Correlation	8.0.0	8.1.0
2009-06	CT#44	CP-090417	0006	3	Session transfer when PS session exists in target access – terminating case	8.1.0	8.2.0
2009-06	CT#44	CP-090417	0021	1	PS-PS+CS when service control over Gm is retained on source access leg	8.1.0	8.2.0
2009-06	CT#44	CP-090417	0022	1	Service Control Signalling Path transfer for ICS session continuity during PS-PS transfer	8.1.0	8.2.0
2009-06	CT#44	CP-090417	0026	1	Correlation CS anchored call with IMS Registration	8.1.0	8.2.0
2009-06	CT#44	CP-090417	0029	2	Clarification of the identification of the originating user	8.1.0	8.2.0
2009-06	CT#44	CP-090417	0031	3	Clarification of for the BYE request used in PS-PS partial reansfer	8.1.0	8.2.0
2009-06	CT#44				Editorial cleanup by MCC	8.1.0	8.2.0
2009-09	CT#45	CP-090669	0035	1	CS to PS transfer request by non ICS UE	8.2.0	8.3.0
2009-09	CT#45	CP-090669	0036	1	Multiple active sessions, PS to CS transfer	8.2.0	8.3.0
2009-09	CT#45	CP-090669	0052	1	Use of "Target-Dialog" for (PS+CS) to PS session transfer	8.2.0	8.3.0
2009-09	CT#45	CP-090669	0053	2	Source access leg release at the SCC AS for PS to PS session transfer	8.2.0	8.3.0
2009-09	CT#45	CP-090669	0065	2	Directing requests using the appropriate IP-CAN	8.2.0	8.3.0
2009-09	CT#45	CP-090669	0066	1	Flow updates for directing requests using the appropriate IP-CAN	8.2.0	8.3.0
2009-09	CT#45	CP-090669	0073	2	Session transfer when PS session exists in target access	8.2.0	8.3.0
2009-09	CT#45	CP-090669	0079	2	Clarification of Source Access Leg Release in PS-CS transfer procedure	8.2.0	8.3.0
2009-09	CT#45	CP-090688	0037	4	Flows for MSC Server assisted mid-call feature	8.3.0	9.0.0
2009-09	CT#45	CP-090688	0039	1	Flows for inter UE transfer without collaborative session	8.3.0	9.0.0
2009-09	CT#45	CP-090688	0040	3	Inter UE transfer without collaboriative session	8.3.0	9.0.0
2009-09	CT#45	CP-090688	0041	2	Skeleton of additions	8.3.0	9.0.0
2009-09	CT#45	CP-090688	0042	1	Scope change	8.3.0	9.0.0
2009-09	CT#45	CP-090688	0043	1	Changes to definitions and abbreviations	8.3.0	9.0.0
2009-09	CT#45	CP-090688	0044	3	Changes to functional entities	8.3.0	9.0.0
2009-09	CT#45	CP-090688	0045	1	Changes to Overview	8.3.0	9.0.0
2009-09	CT#45	CP-090688	0046	1	Change of Session Continuity to Access Transfer	8.3.0	9.0.0
2009-09	CT#45	CP-090688	0047	2	Call flow for Controller UE releases Collaborative Session	8.3.0	9.0.0
2009-09	CT#45	CP-090688	0049	3	State Model for Collaborative Session handling	8.3.0	9.0.0
2009-09	CT#45	CP-090688	0055	3	Clarification of SR VCC procedure	8.3.0	9.0.0
2009-09	CT#45	CP-090685	0067	6	Call flow for UE initiating an emergency session in IMS using SRVCC	8.3.0	9.0.0
2009-09	CT#45	CP-090685	0068	6	Call flow for EMC using SRVCC procedure	8.3.0	9.0.0
2009-09	CT#45	CP-090688	0072	1	Clarification for the Service Control Signalling Path transfer for ICS session continuity during PS-PS transfer	8.3.0	9.0.0
2009-09	CT#45	CP-090688	0089	2	Add indication of the capability of supporting mid-call feature	8.3.0	9.0.0
2009-09	CT#45	CP-090688	0094	2	MSC Server assisted mid-call feature - SRVCC	8.3.0	9.0.0
2009-09	CT#45	CP-090685	0106	2	E-SCC AS actions for IMS Emergency call	8.3.0	9.0.0
2009-09	CT#45	CP-090688	0109	1	MSC Server assisted mid-call feature - PS to CS	8.3.0	9.0.0
2009-09	CT#45	CP-090688	0111	3	MSC Server assisted mid-call feature - PS to PS	8.3.0	9.0.0
2009-12	CT#46	CP-090929	0048	2	Call flow for Remote party releases Collaborative Session	9.0.0	9.1.0
2009-12	CT#46	CP-090929	0096	3	Signalling flow for Controller UE releases media flow on controller UE	9.0.0	9.1.0
2009-12	CT#46	CP-090929	0097	3	Signalling flow for Controller UE releases media on Controllee UE	9.0.0	9.1.0
2009-12	CT#46	CP-090929	0099	3	Signalling flow for Controllee UE modify media on itself	9.0.0	9.1.0
2009-12	CT#46	CP-090929	0100	6	Signalling flow for Remote party adds new media on controllee UE	9.0.0	9.1.0
2009-12	CT#46	CP-090929	0101	3	Signalling flow for Remote UE releases media	9.0.0	9.1.0
2009-12	CT#46	CP-090929	0110	3	MSC Server assisted mid-call feature - CS to PS - Alt1	9.0.0	9.1.0
2009-12	CT#46	CP-090929	0116	4	Roles for target UE discovery for Inter-UE Transfer	9.0.0	9.1.0
2009-12	CT#46	CP-090929	0117	5	Roles of SCC AS for target UE discovery for Inter-UE Transfer	9.0.0	9.1.0
2009-12	CT#46	CP-090929	0122	1	MSC Server assisted mid-call feature - flow updates - Alt1	9.0.0	9.1.0
2009-12	CT#46	CP-090929	0124	1	MSC Server assisted mid-call feature - capability exchange update	9.0.0	9.1.0
2009-12	CT#46	CP-090929	0125	1	MSC Server assisted mid-call feature - PS to CS and SRVCC updates - Atl1	9.0.0	9.1.0
2009-12	CT#46	CP-090929	0128	1	MSC Server assisted mid-call feature - PS to PS updates	9.0.0	9.1.0
2009-12	CT#46	CP-090929	0131		SCC UE not defined	9.0.0	9.1.0

2009-12	CT#46	CP-090927	0132		Renaming of E-SCC AS to EATF	9.0.0	9.1.0
2009-12	CT#46	CP-090927	0133	2	E-SRVCC session origination	9.0.0	9.1.0
2009-12	CT#46	CP-090927	0134	2	E-SRVCC access transfer	9.0.0	9.1.0
2009-12	CT#46	CP-090929	0136	2	Call flow for transferring a media to a controllee UE	9.0.0	9.1.0
2009-12	CT#46	CP-090928	0146	1	Correction on PS-CS transfer information flow	9.0.0	9.1.0
2009-12	CT#46	CP-090929	0149	1	MSC Server assisted mid-call feature - transaction identifiers	9.0.0	9.1.0
2009-12	CT#46	CP-090911	0153	1	Enabling session continuity functionality	9.0.0	9.1.0
2009-12	CT#46	CP-090929	0155	2	SC UE procedures for collaborative session establishment for transferring media	9.0.0	9.1.0
2009-12	CT#46	CP-090929	0156	2	SC UE procedures for collaborative session establishment with new media	9.0.0	9.1.0
2009-12	CT#46	CP-090929	0157	2	SCC AS procedures for collaborative session establishment for transferring media	9.0.0	9.1.0
2009-12	CT#46	CP-090911	0160	1	Address the Editor's Note in A.3.2	9.0.0	9.1.0
2009-12	CT#46	CP-090911	0162	1	Address the Editor's Note in A.8.2	9.0.0	9.1.0
2009-12	CT#46	CP-090929	0164	2	PS to CS transfer for speech and video session with MSC Server asisted mid-call feature	9.0.0	9.1.0
2009-12	CT#46	CP-090929	0170	2	IUT Procedures	9.0.0	9.1.0
2009-12	CT#46	CP-090929	0171	2	Call flow for adding media to controllee UE	9.0.0	9.1.0
2009-12	CT#46	CP-091047	0173	3	SC AS procedures for collaborative session establishment with new media	9.0.0	9.1.0
2009-12	CT#46	CP-090929	0174	3	Procedures for releasing media on controllee UE by controller UE	9.0.0	9.1.0
2009-12	CT#46	CP-090929	0175	3	Procedure for adding new media by remote party	9.0.0	9.1.0
2009-12	CT#46	CP-090929	0176	3	Procedure for releasing media on ontroller UE by controller UE	9.0.0	9.1.0
2009-12	CT#46	CP-090929	0177	3	Procedure for modifying media on contrllee UE by itself	9.0.0	9.1.0
2009-12	CT#46	CP-090929	0178	2	Signalling flow for Controllee UE releases media	9.0.0	9.1.0
2010-03	CT#47	CP-100123	0144	5	PS to CS or to (PS+CS) session transfer for an SC UE using Gm interface	9.1.0	9.2.0
2010-03	CT#47	CP-100143	0182	1	Procedures for collaborative session release by controller UE	9.1.0	9.2.0
2010-03	CT#47	CP-100143	0183	1	Procedures for controllee UE releases media component	9.1.0	9.2.0
2010-03	CT#47	CP-100143	0184	1	Procedures for collaborative session release by remote party	9.1.0	9.2.0
2010-03	CT#47	CP-100143	0185	1	Procedures for controller UE initiated media transfer from controller UE to controllee UE	9.1.0	9.2.0
2010-03	CT#47	CP-100142	0186	1	Establishment of collaborative session for inter-UE transfer	9.1.0	9.2.0
2010-03	CT#47	CP-100142	0187	1	Media adding/deleting within collaborative session for inter-UE transfer	9.1.0	9.2.0
2010-03	CT#47	CP-100143	0188	2	Procedures for releasing media by remote UE	9.1.0	9.2.0
2010-03	CT#47	CP-100199	0189	4	Controller UE initiated media transfer from controllee UE to another controllee UE	9.1.0	9.2.0
2010-03	CT#47	CP-100143	0191	2	Procedures for controller UE initiated media transfer from controllee UE to another controllee UE	9.1.0	9.2.0
2010-03	CT#47	CP-100143	0192	3	Procedures for controller UE initiated media transfer from controllee UE to another controllee UE—SCC AS behavior	9.1.0	9.2.0
2010-03	CT#47	CP-100143	0193	1	Signalling flows for media transfer within collaborative session for inter-UE transfer	9.1.0	9.2.0
2010-03	CT#47	CP-100142	0194	1	Access tranfer and MMTEL interaction	9.1.0	9.2.0
2010-03	CT#47	CP-100142	0195	2	Inter-UE transfer and MMTEL interaction	9.1.0	9.2.0
2010-03	CT#47	CP-100143	0196	1	Procedures for adding new media on controllee UE by controller UE	9.1.0	9.2.0
2010-03	CT#47	CP-100143	0198	1	Release of collaborative session for inter-UE transfer	9.1.0	9.2.0
2010-03	CT#47	CP-100143	0199	2	Procedures for controllee UE releases media component	9.1.0	9.2.0
2010-03	CT#47	CP-100143	0200	1	Procedures for collaborative session release by remote party	9.1.0	9.2.0
2010-03	CT#47	CP-100143	0201	1	Procedures for controller UE initiated media transfer from controller UE to controllee UE	9.1.0	9.2.0
2010-03	CT#47	CP-100142	0202	3	Adding new media on controllee UE by controller UE	9.1.0	9.2.0
2010-03	CT#47	CP-100143	0203	3	procedures for subscription to the session description	9.1.0	9.2.0
2010-03	CT#47	CP-100143	0204	1	SR VCC from MSC	9.1.0	9.2.0
2010-03	CT#47	CP-100142	0206	1	Deleting the editor note at A.15.3.2.2 for controller UE removing media at controllee UE	9.1.0	9.2.0
2010-03	CT#47	CP-100142	0207	1	Deleting editor's note for clause A.15.3.1	9.1.0	9.2.0
2010-03	CT#47	CP-100142	0208	1	Deleting the editor note at A.15.5 for controllee UE modify meida on iteslf	9.1.0	9.2.0
2010-03	CT#47	CP-100142	0209	2	Complete session transfer routing clarification	9.1.0	9.2.0
2010-03	CT#47	CP-100143	0210	1	MSC Server assisted mid-call feature - single held session	9.1.0	9.2.0
2010-03	CT#47	CP-100143	0211	1	MSC Server assisted mid-call feature - flow clean up	9.1.0	9.2.0
2010-03	CT#47	CP-100143	0212	2	MSC server assisted mid-call feature - sendonly, recvonly	9.1.0	9.2.0
2010-03	CT#47	CP-100142	0213	1	Incorrect reference correction	9.1.0	9.2.0
2010-03	CT#47	CP-100143	0214	1	Removal of ICMP message sending – procedures	9.1.0	9.2.0
2010-03	CT#47	CP-100143	0217	1	Registration requirements for an SC UE that only implements Inter-UE Transfer	9.1.0	9.2.0
2010-03	CT#47	CP-100142	0218	2	Addition of the Content-Type to SIPfrag containing SDP	9.1.0	9.2.0
2010-03	CT#47	CP-100142	0219	3	Separating releasing media on controllee UE and removing controllee UE from collaborative session	9.1.0	9.2.0

2010-03	CT#47	CP-100142	0220	1	Correction of Procedures for adding new media by remote party when the controller UE does not alert the user	9.1.0	9.2.0
2010-03	CT#47	CP-100143	0221	1	Removal of Editor's Note on use of SDP in URI parameters in Refer-To header	9.1.0	9.2.0
2010-03	CT#47	CP-100142	0223	1	Editorial changes to clause 8	9.1.0	9.2.0
2010-03	CT#47	CP-100142	0224	1	Editorial changes to clause 9	9.1.0	9.2.0
2010-03	CT#47	CP-100142	0225	1	Editorial changes to clause 14	9.1.0	9.2.0
2010-03	CT#47	CP-100142	0226	1	Editorial changes to clause 16	9.1.0	9.2.0
2010-03	CT#47	CP-100142	0227	1	Editorial changes to clause 19	9.1.0	9.2.0
2010-03	CT#47	CP-100142	0228	1	Editorial changes to clause A.12.3	9.1.0	9.2.0
2010-03	CT#47	CP-100142	0229	1	Editorial changes to clause A.13.2	9.1.0	9.2.0
2010-03	CT#47	CP-100142	0230	1	Editorial changes to clause A.14	9.1.0	9.2.0
2010-03	CT#47	CP-100142	0232	1	Editorial changes to clause A.16	9.1.0	9.2.0
2010-03	CT#47	CP-100142	0233	1	Editorial changes to clause C.2	9.1.0	9.2.0
2010-03	CT#47	CP-100123	0236	3	Clarification of SC OMA MO use	9.1.0	9.2.0
2010-03	CT#47	CP-100240	0237	1	Removal of IMS communication service from emergency call flows	9.1.0	9.2.0
2010-03	CT#47	CP-100135	0238	1	SR VCC abnormal case	9.1.0	9.2.0
2010-03	CT#47	CP-100123	0240		Correct the definition of Correlation MSISDN	9.1.0	9.2.0
2010-03	CT#47	CP-100143	0241	2	Controller UE initiated media transfer from controllee UE to another controllee UE	9.1.0	9.2.0
2010-03	CT#47	CP-100143	0242		UE remote changed to Remote UE	9.1.0	9.2.0
2010-03	CT#47	CP-100143	0243	1	Merged corrections to A.15.3.2.1, A.15.3.2.2 agreed at the CT1e-meeting	9.1.0	9.2.0
2010-03	CT#47	CP-100140	0244		EATF editor's notes resolution	9.1.0	9.2.0
2010-03	CT#47	CP-100142	0245		EN on SIP URI of SCC AS	9.1.0	9.2.0
2010-03	CT#47	CP-100142	0246	2	Collaborative session establishement of with new media	9.1.0	9.2.0
2010-03	CT#47	CP-100143	0247	1	SCC AS procedures for collaborative session establishment with new media	9.1.0	9.2.0
2010-03	CT#47	CP-100142	0248	4	Addition of media feature tag for indicating IUT Controller capability	9.1.0	9.2.0
2010-03	CT#47	CP-100141	0249		Editorial changes to clause 4	9.1.0	9.2.0
2010-03	CT#47	CP-100141	0250		Editorial changes to clause 6	9.1.0	9.2.0
2010-03	CT#47	CP-100141	0251		Editorial changes to clause 7	9.1.0	9.2.0
2010-03	CT#47	CP-100141	0252	2	Editorial changes to clause 10	9.1.0	9.2.0
2010-03	CT#47	CP-100141	0253	1	Editorial changes to clause 11	9.1.0	9.2.0
2010-03	CT#47	CP-100141	0254		Editorial changes to clause 12	9.1.0	9.2.0
2010-03	CT#47	CP-100141	0255		Editorial changes to clause 13	9.1.0	9.2.0
2010-03	CT#47	CP-100141	0256	1	Editorial changes to clause A.3	9.1.0	9.2.0
2010-03	CT#47	CP-100141	0257	1	Editorial changes to clause A.6	9.1.0	9.2.0
2010-03	CT#47	CP-100141	0258	1	Editorial changes to clause A.7	9.1.0	9.2.0
2010-03	CT#47	CP-100141	0259		Editorial changes to clause A.8	9.1.0	9.2.0
2010-03	CT#47	CP-100141	0260		Editorial changes to clause A.9	9.1.0	9.2.0
2010-03	CT#47	CP-100141	0261		Editorial changes to clause A.11	9.1.0	9.2.0
2010-03	CT#47	CP-100141	0262	1	Editorial changes to clause A.16	9.1.0	9.2.0
2010-03	CT#47	CP-100140	0263	1	Editorial changes to clause A.17	9.1.0	9.2.0
2010-03	CT#47	CP-100141	0264		Inappropriate normative language in relation to registration	9.1.0	9.2.0
2010-03	CT#47	CP-100143	0197	1	Remove of signaling flow for target UE discovery	9.1.0	9.2.0
2010-03	CT#47				Editorial cleanup by MCC	9.1.0	9.2.0
2010-06	CT#48	CP-100359	0130	1	MSC Server assisted mid-call feature - conferencing	9.2.0	9.3.0
2010-06	CT#48	CP-100359	0265	2	Removal of editorial notes	9.2.0	9.3.0
2010-06	CT#48	CP-100359	0266	1	Controllee UE announces controller capabilities	9.2.0	9.3.0
2010-06	CT#48	CP-100359	0268	1	Editorial corrections	9.2.0	9.3.0
2010-06	CT#48	CP-100359	0275		Correction of references	9.2.0	9.3.0
2010-06	CT#48	CP-100359	0276	1	Removing controllee UE procedure correction	9.2.0	9.3.0
2010-06	CT#48	CP-100359	0279	1	Compliance corrections	9.2.0	9.3.0
2010-06	CT#48	CP-100359	0280		PS-CS access transfer corrections	9.2.0	9.3.0
2010-06	CT#48	CP-100359	0282	1	PS to CS+PS access transfer corrections	9.2.0	9.3.0
2010-06	CT#48	CP-100359	0283	2	SRVCC corrections	9.2.0	9.3.0
2010-06	CT#48	CP-100359	0284	1	Race condition during SRVCC	9.2.0	9.3.0
2010-06	CT#48	CP-100359	0285	2	MSC Server assisted mid-call feature and SR VCC abnormal case	9.2.0	9.3.0
2010-06	CT#48	CP-100359	0286	1	Inter UE Transfer corrections - procedure overlap	9.2.0	9.3.0
2010-06	CT#48	CP-100359	0287	2	Inter UE Transfer corrections - collaborative session by media transfer	9.2.0	9.3.0
2010-06	CT#48	CP-100359	0288	2	Inter UE Transfer corrections - collaborative session by media transfer	9.2.0	9.3.0
2010-06	CT#48	CP-100359	0289		Inter UE Transfer corrections - session discovery corrections	9.2.0	9.3.0
2010-06	CT#48	CP-100359	0290	1	Inter UE Transfer corrections - media transfer during col. session	9.2.0	9.3.0
2010-06	CT#48	CP-100359	0291	1	Inter UE Transfer corrections - media adding/deleting during col. session	9.2.0	9.3.0
2010-06	CT#48	CP-100359	0292	2	Correction of Referred-By header usage	9.2.0	9.3.0
2010-06	CT#48	CP-100359	0293	1	Correction of iut-controller feature tag usage	9.2.0	9.3.0
2010-09	CT#49	CP-100505	0298		Editorial Corrections	9.3.0	9.4.0

2010-09	CT#49	CP-100505	0299		Removing unnecessary condition for sending SIP UPDATE or SIP re-INVITE request	9.3.0	9.4.0
2010-09	CT#49	CP-100493	0307	2	Clarifying SC UE procedures for terminations	9.3.0	9.4.0
2010-09	CT#49	CP-100505	0308	1	Inform remote end about new local end	9.3.0	9.4.0
2010-09	CT#49	CP-100493	0310	3	Corrections use of 3GPP TS 24.216 MO leaf	9.3.0	9.4.0
2010-09	CT#49	CP-100505	0314	3	Modification of SCC AS procedure in MSC server assisted mid-call feature	9.3.0	9.4.0
2010-09	CT#49	CP-100505	0315	2	IUT Cleanup	9.3.0	9.4.0
2010-09	CT#49	CP-100493	0319	1	UE compliance	9.3.0	9.4.0
2010-09	CT#49	CP-100493	0327	1	Modification the usage for instance ID in session transfer	9.3.0	9.4.0
2010-09	CT#49	CP-100505	0328	2	Clarification of MSC server apply ICS capability	9.3.0	9.4.0
2010-09	CT#49	CP-100519	0316	1	Insertion of missing requirement to prohibit PS-CS continuity	9.4.0	10.0.0
2010-09	CT#49	CP-100519	0321	1	Error correction and modify signalling flows for controller UE initiated media transfer from controller UE to controllee UE	9.4.0	10.0.0
2010-09	CT#49	CP-100519	0322	1	Add signalling flows for collaborative session establishment with media transfer	9.4.0	10.0.0
2010-12	CT#50	CP-100863	0304	6	Flow for SRVCC in alerting state – terminating case	10.0.0	10.1.0
2010-12	CT#50	CP-100863	0305	6	Flow for SRVCC in alerting state – originating case	10.0.0	10.1.0
2010-12	CT#50	CP-100863	0311	9	Mid-call scenairos for incoming and outgoing call in alerting state	10.0.0	10.1.0
2010-12	CT#50	CP-100863	0325	5	Call flow for transfering an incoming waiting call in alerting phase	10.0.0	10.1.0
2010-12	CT#50	CP-100843	0338	3	Signalling flows for PS-CS access transfer when using ATCF enhancements and without media anchored	10.0.0	10.1.0
2010-12	CT#50	CP-100843	0339	5	Signalling flows for PS-CS access transfer when using ATCF enhancements and media anchored	10.0.0	10.1.0
2010-12	CT#50	CP-100842	0342	2	Corrections of SC UE registaration	10.0.0	10.1.0
2010-12	CT#50	CP-100737	0345	1	ICS UE prevented from Gm control adding when ICS is disabled	10.0.0	10.1.0
2010-12	CT#50	CP-100843	0350	4	SRVCC enhancements - registration flow	10.0.0	10.1.0
2010-12	CT#50	CP-100843	0351	3	SRVCC enhancements - originating session set up flow	10.0.0	10.1.0
2010-12	CT#50	CP-100843	0353	3	SRVCC enhancements - ATCF registration procedures	10.0.0	10.1.0
2010-12	CT#50	CP-100843	0354	1	SRVCC enhancements - scope, definition, compliance	10.0.0	10.1.0
2010-12	CT#50	CP-100842	0359		Correction in SRVCC Emergency Flows.	10.0.0	10.1.0
2010-12	CT#50	CP-100863	0360	3	Flows for SRVCC in alerting state – Race condition when answering in PS	10.0.0	10.1.0
2010-12	CT#50	CP-100863	0362	2	SCC AS procedures for SRVCC alerting state	10.0.0	10.1.0
2010-12	CT#50	CP-100863	0364	3	MSC server procedure for SRVCC in alerting	10.0.0	10.1.0
2010-12	CT#50	CP-100863	0365	3	Definition of INFO package for SRVCC alerting	10.0.0	10.1.0
2010-12	CT#50	CP-100863	0367	2	Further flow for SRVCC in alerting state race condition when answering in PS	10.0.0	10.1.0
2010-12	CT#50	CP-100863	0368	2	UE and SCC AS procedures for abnormal cases of SRVCC when in Alerting Phase	10.0.0	10.1.0
2010-12	CT#50	CP-100863	0369	2	SC UE procedures for SRVCC in Alerting Phase	10.0.0	10.1.0
2010-12	CT#50	CP-100843	0372	3	SCC AS procedure for PS to CS Access Transfer: SRVCC enhancements using ATCF	10.0.0	10.1.0
2010-12	CT#50	CP-100737	0375	2	Correction of the SCC AS association procedure	10.0.0	10.1.0
2010-12	CT#50	CP-100843	0352	2	SRVCC enhancements - ATCF invocation	10.0.0	10.1.0
2010-12	CT#50	CP-100746	0335	1	Editor's note deleting for remote UE releases media on the controller UE	10.0.0	10.1.0
2010-12	CT#50	CP-100746	0356	1	SRVCC clarifications for SDP offer by the MSC.	10.0.0	10.1.0
2010-12	CT#50	CP-100864	0361	5	PS-PS access transfer in early dialog state	10.0.0	10.1.0
2011-03	CT#51	CP-110196	0377	2	Remote leg release	10.1.0	10.2.0
2011-03	CT#51	CP-110196	0379	2	Previously established dialog.	10.1.0	10.2.0
2011-03	CT#51	CP-110196	0381	1	Contact registration	10.1.0	10.2.0
2011-03	CT#51	CP-110199	0382	1	Editor's note delete for aSRVCC on A.17.1	10.1.0	10.2.0
2011-03	CT#51	CP-110198	0383	2	Adding clauce A.18 for eSRVCC signalling flow	10.1.0	10.2.0
2011-03	CT#51	CP-110178	0393		Reference update: RFC 6086	10.1.0	10.2.0
2011-03	CT#51	CP-110199	0394	1	STN-SR missing in procedures	10.1.0	10.2.0
2011-03	CT#51	CP-110199	0395	2	TI assignment for MSC server for srvcc for alerting call	10.1.0	10.2.0
2011-03	CT#51	CP-110196	0409		g.3gpp.access-type media feature tag name corrected	10.1.0	10.2.0
2011-03	CT#51	CP-110178	0411		g.3gpp.mid-call info package description corrected	10.1.0	10.2.0
2011-03	CT#51	CP-110196	0412		PS-PS access transfer, accepting terminating early dialog	10.1.0	10.2.0
2011-03	CT#51	CP-110199	0413	3	Alerting SRVCC when another call exists	10.1.0	10.2.0
2011-03	CT#51	CP-110198	0414	2	SRVCC enhancement, ATCF access transfer procedures	10.1.0	10.2.0
2011-03	CT#51	CP-110198	0415	2	SRVCC enhancement, ATCF compliance update	10.1.0	10.2.0
2011-03	CT#51	CP-110198	0417	5	Format and triggers for SCC AS sending ATU-STI and C-MSISDN to ATCF	10.1.0	10.2.0
2011-03	CT#51	CP-110198	0418	5	SRVCC enhancement, ATCF call set up procedures	10.1.0	10.2.0
2011-03	CT#51	CP-110199	0422		Allowing state-and-event-info XML to be used for other applications in future	10.1.0	10.2.0
2011-03	CT#51	CP-110199	0423	4	Corrections of UE procedures for SRVCC in alerting state	10.1.0	10.2.0
2011-03	CT#51	CP-110178	0426	3	Clarification of MSC server procedures.	10.1.0	10.2.0
2011-03	CT#51	CP-110196	0429	1	PS-PS transfer corrections	10.1.0	10.2.0
2011-03	CT#51	CP-110169	0432	2	Active media component	10.1.0	10.2.0

2011-03	CT#51	CP-110198	0433	3	Transport of ATU-STI and C-MSISDN from SCC AS to ATCF	10.1.0	10.2.0
2011-03	CT#51	CP-110198	0434	1	SRVCC enhancement, ATCF procedures when no active call exists	10.1.0	10.2.0
2011-03	CT#51	CP-110198	0435	2	SRVCC enhancements, SCC AS procedure	10.1.0	10.2.0
2011-03	CT#51	CP-110199	0436	2	Indicating support for SRVCC alerting state	10.1.0	10.2.0
2011-03	CT#51	CP-110198	0453		Referencing 3GPP TS 23.003 for identity definitions	10.1.0	10.2.0
2011-03	CT#51	CP-110199	0455	1	Adding call flow for incoming call in alerting phase with forked responses	10.1.0	10.2.0
2011-03	CT#51	CP-110199	0456	1	Adding the SCC AS procedure for originating call in alerting phase when forked responses received	10.1.0	10.2.0
2011-03	CT#51	CP-110290	0440	3	Shifting Rel-9 IUT material OUT of TS 24.237	10.1.0	10.2.0
2011-06	CT#52	CP-110449	0398	3	Transferable Sessions	10.2.0	10.3.0
2011-06	CT#52	CP-110469	0459		Reference update: draft-ietf-sipcore-proxy-feature	10.2.0	10.3.0
2011-06	CT#52	CP-110469	0460		Correcting erroneous references	10.2.0	10.3.0
2011-06	CT#52	CP-110469	0461		Correcting erroneous subclause references	10.2.0	10.3.0
2011-06	CT#52	CP-110469	0462	1	SC UE checking dialog parameters	10.2.0	10.3.0
2011-06	CT#52	CP-110469	0463		Functional entities missing in some introductions	10.2.0	10.3.0
2011-06	CT#52	CP-110470	0465	1	g.3gpp.mid-call media feature in 2xx response	10.2.0	10.3.0
2011-06	CT#52	CP-110469	0466	1	ATCF adding Record-Route	10.2.0	10.3.0
2011-06	CT#52	CP-110469	0467	2	EN on ATCF URI assignment	10.2.0	10.3.0
2011-06	CT#52	CP-110469	0468		SRVCC related information received when session(s) exist	10.2.0	10.3.0
2011-06	CT#52	CP-110465	0469	1	PS-PS access transfer corrections	10.2.0	10.3.0
2011-06	CT#52	CP-110469	0470	2	SRVCC transferable session	10.2.0	10.3.0
2011-06	CT#52	CP-110470	0471	1	ENs in alerting SRVCC	10.2.0	10.3.0
2011-06	CT#52	CP-110470	0472		Add description of SRVCC alerting with forked responses to introduction clause	10.2.0	10.3.0
2011-06	CT#52	CP-110470	0473	2	Missing aspects for SRVCC due to different permutations of UE and network support	10.2.0	10.3.0
2011-06	CT#52	CP-110449	0476	2	Speech component in session	10.2.0	10.3.0
2011-06	CT#52	CP-110470	0477	1	Handling of error in INFO request	10.2.0	10.3.0
2011-06	CT#52	CP-110469	0479	1	Editor's notes clean up for eSRVCC	10.2.0	10.3.0
2011-06	CT#52	CP-110469	0482	2	Determining eSRVCC transferable session set	10.2.0	10.3.0
2011-09	CT#53	CP-110681	0483		Redundant text removal	10.3.0	10.4.0
2011-09	CT#53	CP-110681	0488	3	P-CSCF releasing dialog during PS-to-PS access transfer	10.3.0	10.4.0
2011-09	CT#53	CP-110681	0489	2	P-CSCF releasing the early dialog during PS-to-PS transfer	10.3.0	10.4.0
2011-09	CT#53	CP-110674	0490	2	ATCF retaining session state information during SRVCC	10.3.0	10.4.0
2011-09	CT#53	CP-110681	0491	2	P-CSCF releasing the early dialog during SR-VCC	10.3.0	10.4.0
2011-09	CT#53	CP-110681	0492		Transferable Session – fixing of incorrect CR	10.3.0	10.4.0
2011-09	CT#53	CP-110674	0493	4	SRVCC related information	10.3.0	10.4.0
2011-09	CT#53	CP-110674	0496	1	Removing confusing "old IP-CAN"	10.3.0	10.4.0
2011-09	CT#53	CP-110670	0497	1	Clarifying the Inter UE Transfer SCC AS URI	10.3.0	10.4.0
2011-09	CT#53	CP-110685	0498	1	Misleading reference in PS to CS transfer incoming call waiting	10.3.0	10.4.0
2011-09	CT#53	CP-110685	0501		SRVCC transferable session not defined	10.3.0	10.4.0
2011-09	CT#53	CP-110681	0502	1	SRVCC and session with full-duplex speech media component	10.3.0	10.4.0
2011-09	CT#53	CP-110655	0505		SRVCC transfers speech media component of established session	10.3.0	10.4.0
2011-09	CT#53	CP-110672	0507	1	Assignment of transferred sessions to CS calls	10.3.0	10.4.0
2011-09	CT#53	CP-110674	0508	2	Cleanup of ATCF procedures	10.3.0	10.4.0
2011-09	CT#53	CP-110685	0509	1	Alerting session may not be transferred when UE has Active + Alerting + Held call	10.3.0	10.4.0
2011-09	CT#53	CP-110674	0510	2	Update of a note in 6.3.1	10.3.0	10.4.0
2011-09	CT#53	CP-110674	0511		Modifying references of ATCF functionality	10.3.0	10.4.0
2011-09	CT#53	CP-110674	0512	1	Correction on ATCF behavior when receiving a SIP INVITE request	10.3.0	10.4.0
2011-09	CT#53	CP-110674	0514	2	MSC contact address	10.3.0	10.4.0
2011-09	CT#53	CP-110674	0515	2	Contact at ATCF	10.3.0	10.4.0
2011-09	CT#53	CP-110655	0529	1	SCC AS procedures in SRVCC	10.3.0	10.4.0
2011-09	CT#53	CP-110672	0534	1	Correcting terminology and references used in subclause 12.3	10.3.0	10.4.0
2011-09	CT#53	CP-110685	0535	1	Missing reference to the PS to CS access transfer for alerting calls	10.3.0	10.4.0
2011-09	CT#53	CP-110685	0536	2	Release of source access leg when signalling path is unavailable between UE and SCC AS	10.3.0	10.4.0
2011-09	CT#53	CP-110693	0484	2	Flow for PS-PS Transfer for outgoing call in alerting phase	10.4.0	11.0.0
2011-09	CT#53	CP-110693	0485	2	Flow for PS-PS Transfer for incoming call in alerting phase	10.4.0	11.0.0
2011-09	CT#53	CP-110693	0495	1	Forking to both CS and PS contact	10.4.0	11.0.0
2011-09	CT#53	CP-110693	0513	2	ATGW performing transcoding	10.4.0	11.0.0
2011-09	CT#53	CP-110693	0516	2	Media path switching at ATGW	10.4.0	11.0.0
2011-09	CT#53	CP-110693	0518	2	ATCF-Note removal	10.4.0	11.0.0
2011-09	CT#53	CP-110693	0519	1	PS to PS transfer at the SC UE	10.4.0	11.0.0
2011-09	CT#53	CP-110693	0520	2	PS to PS transfer at the SCC AS	10.4.0	11.0.0
2011-09	CT#53	CP-110697	0523	2	General IMS procedures for vSRVCC	10.4.0	11.0.0
2011-09	CT#53	CP-110697	0524	2	SC UE procedures for vSRVCC	10.4.0	11.0.0
2011-09	CT#53	CP-110697	0525	2	MSC Server procedures for vSRVCC	10.4.0	11.0.0
2011-09	CT#53	CP-110697	0526	2	SCC AS procedures for vSRVCC	10.4.0	11.0.0
2011-12	CT#54	CP-110881	0545		Media feature tag removed	11.0.0	11.1.0
2011-12	CT#54	CP-110881	0546	1	ATCF selection	11.0.0	11.1.0

2011-12	CT#54	CP-110878	0548	3	Resource-Priority handling in ATCF	11.0.0	11.1.0
2011-12	CT#54	CP-110881	0549	1	eMPS for SRVCC	11.0.0	11.1.0
2011-12	CT#54	CP-110865	0552	1	Transfer failure if configuration tables are changed after the session was established	11.0.0	11.1.0
2011-12	CT#54	CP-110881	0553	1	SRVCC PS to CS transfer of alerting call - FLOW	11.0.0	11.1.0
2011-12	CT#54	CP-110881	0554	1	g.3gpp.srvcc-alerting media feature tag missing in flows	11.0.0	11.1.0
2011-12	CT#54	CP-110879	0562	3	Reference to Contact changed to reference Feature-Caps	11.0.0	11.1.0
2011-12	CT#54	CP-110881	0563	1	Reference to PS to CS access transfer for alerting in MSC server	11.0.0	11.1.0
2011-12	CT#54	CP-110881	0564	2	Single call and call state association missing in MSC server	11.0.0	11.1.0
2011-12	CT#54	CP-110881	0565	2	Readability correction	11.0.0	11.1.0
2011-12	CT#54	CP-110881	0568	2	Action in MSC server when receiving 200 OK is missing	11.0.0	11.1.0
2011-12	CT#54	CP-110879	0570		Answer shall be call-accepted	11.0.0	11.1.0
2011-12	CT#54	CP-110881	0571	1	CS Connect shall be CC CONNECT	11.0.0	11.1.0
2011-12	CT#54	CP-110878	0573		Incorrect reference to 24.229	11.0.0	11.1.0
2011-12	CT#54	CP-110881	0576	3	Serving network capabilities	11.0.0	11.1.0
2011-12	CT#54	CP-110878	0578	1	Media offered by ATCF in target access leg during SRVCC	11.0.0	11.1.0
2011-12	CT#54	CP-110878	0580	1	ATCF procedures and "loose route" indication	11.0.0	11.1.0
2011-12	CT#54	CP-110867	0586	1	Missing SRVCC for IMS emergency procedures for MSC server enhanced for ICS	11.0.0	11.1.0
2011-12	CT#54	CP-110884	0587		Removal of Editor's Note on use of SIP OPTIONS and fixing editorials	11.0.0	11.1.0
2011-12	CT#54	CP-110881	0591	1	Correction of charging headers	11.0.0	11.1.0
2011-12	CT#54	CP-110878	0594	2	Cleanup of ATCF procedures	11.0.0	11.1.0
2011-12	CT#54	CP-110881	0595	1	Flow correction for SRVCC enhancements	11.0.0	11.1.0
2011-12	CT#54	CP-110881	0596		SCC AS includes option tags	11.0.0	11.1.0
2011-12	CT#54	CP-110873	0598	1	Record-Route at ATCF	11.0.0	11.1.0
2011-12	CT#54	CP-110881	0599	2	ATCF on the signalling path	11.0.0	11.1.0
2011-12	CT#54	CP-110873	0602		Removing the Accept header field from 183 response	11.0.0	11.1.0
2011-12	CT#54	CP-110878	0604	3	proxy-feature I-D reference update	11.0.0	11.1.0
2011-12	CT#54	CP-110882	0606	2	PS to CS SRVCC update	11.0.0	11.1.0
2011-12	CT#54	CP-110881	0609		Incompatible codec when transfering a call using ATCF	11.0.0	11.1.0
2011-12	CT#54	CP-110881	0612		Replacing reused procedures for SRVCC and vSRVCC in SCC AS	11.0.0	11.1.0
2011-12	CT#54	CP-110884	0613	1	vSRVCC video bearer identification not required	11.0.0	11.1.0
2011-12	CT#54	CP-110884	0614	2	Determination of vSRVCC handover	11.0.0	11.1.0
2011-12	CT#54	CP-110884	0616	2	Corrections to SC UE procedures for vSRVCC	11.0.0	11.1.0
2011-12	CT#54	CP-110884	0617	1	Example signalling flow for vSRVCC	11.0.0	11.1.0
2012-03	CT#55	CP-120120	0619		CS to PS SRVCC, registration flows in PS	11.1.0	11.2.0
2012-03	CT#55	CP-120120	0620		CS to PS SRVCC, registration flows in CS	11.1.0	11.2.0
2012-03	CT#55	CP-120120	0621		CS to PS SRVCC, originating call	11.1.0	11.2.0
2012-03	CT#55	CP-120120	0622		CS to PS SRVCC, terminating call	11.1.0	11.2.0
2012-03	CT#55	CP-120120	0624		CS to PS SRVCC, MIME type for CS to PS SRVCC information	11.1.0	11.2.0
2012-03	CT#55	CP-120120	0625	1	CS to PS SRVCC, MIME type to be used with INFO	11.1.0	11.2.0
2012-03	CT#55	CP-120120	0626		CS to PS SRVCC, INFO package	11.1.0	11.2.0
2012-03	CT#55	CP-120120	0627		CS to PS SRVCC, feature tags	11.1.0	11.2.0
2012-03	CT#55	CP-120107	0630	1	ATCF passing Recv-Info, Supported, Accept	11.1.0	11.2.0
2012-03	CT#55	CP-120124	0631	1	MSC server assisted mid-call feature - merging MSC server SRVCC procedures	11.1.0	11.2.0
2012-03	CT#55	CP-120095	0634		Speech media component	11.1.0	11.2.0
2012-03	CT#55	CP-120107	0638	1	ATCF-Path-URI in MESSAGE sent from SCC AS	11.1.0	11.2.0
2012-03	CT#55	CP-120107	0640	2	REGISTER handling in ATCF	11.1.0	11.2.0
2012-03	CT#55	CP-120106	0642	1	Overlapping conditions in MSC on INFO	11.1.0	11.2.0
2012-03	CT#55	CP-120124	0643	2	Adding ICS control during SRVCC	11.1.0	11.2.0
2012-03	CT#55	CP-120124	0644	1	SRVCC Charging correlation	11.1.0	11.2.0
2012-03	CT#55	CP-120107	0646	1	Reference update: draft-holmberg-sipcore-proxy-feature	11.1.0	11.2.0
2012-03	CT#55	CP-120126	0647	2	Unclear MSC specification of alerting functionality for vSRVCC and SRVCC.	11.1.0	11.2.0
2012-03	CT#55	CP-120124	0648		Incorrect reference to use of operator specific timer for source access release.	11.1.0	11.2.0
2012-03	CT#55	CP-120120	0649		Signalling flows for eSRVCC assisted mid-call feature including ATCF	11.1.0	11.2.0
2012-03	CT#55	CP-120124	0651	1	Clarifying SCC AS behavior when removing media during full PS-PS access transfer	11.1.0	11.2.0
2012-03	CT#55	CP-120120	0652		CS to PS SRVCC, keys to flows	11.1.0	11.2.0
2012-03	CT#55	CP-120120	0654		CS to PS SRVCC, ATGW transfer details structure	11.1.0	11.2.0
2012-03	CT#55	CP-120120	0623	1	CS to PS SRVCC access transfer occurs during a call	11.1.0	11.2.0
2012-03	CT#55	CP-120120	0650	3	rSRVCC occurs during a call without CS media anchored in ATGW	11.1.0	11.2.0
2012-06	CT#56	CP-120305	0608	5	MSC server state verification procedure solving an EN	11.2.0	11.3.0
2012-06	CT#56	CP-120307	0657		URIs for PS to CS SRVCC	11.2.0	11.3.0
2012-06	CT#56	CP-120317	0658		CS to PS SRVCC - additional URIs of network entities	11.2.0	11.3.0
2012-06	CT#56	CP-120317	0659		ATU-STI for CS to PS SRVCC	11.2.0	11.3.0
2012-06	CT#56	CP-120317	0661	1	CS to PS SRVCC - PS registration	11.2.0	11.3.0
2012-06	CT#56	CP-120317	0662	1	CS to PS SRVCC - originating call	11.2.0	11.3.0

2012-06	CT#56	CP-120317	0663	1	CS to PS SRVCC - terminating call	11.2.0	11.3.0
2012-06	CT#56	CP-120317	0664	1	CS to PS SRVCC, ATGW-anchored element	11.2.0	11.3.0
2012-06	CT#56	CP-120317	0665	1	CS to PS SRVCC - access transfer - MSC server procedures	11.2.0	11.3.0
2012-06	CT#56	CP-120317	0666	1	CS to PS SRVCC - access transfer - ATCF procedures	11.2.0	11.3.0
2012-06	CT#56	CP-120317	0667	2	CS to PS SRVCC - access transfer - SC UE procedures	11.2.0	11.3.0
2012-06	CT#56	CP-120317	0668	2	CS to PS SRVCC - access transfer - SCC AS procedures	11.2.0	11.3.0
2012-06	CT#56	CP-120317	0672	1	Address the Editor's Note in the flows of rSRVCC without CS media anchored in ATGW	11.2.0	11.3.0
2012-06	CT#56	CP-120317	0673	1	CS to PS SRVCC - access transfer - MSC server procedures when media is not anchored in ATGW	11.2.0	11.3.0
2012-06	CT#56	CP-120317	0674	1	CS to PS SRVCC - access transfer - ATCF procedures when media is not anchored in ATGW	11.2.0	11.3.0
2012-06	CT#56	CP-120305	0676		Incorrect header field for media feature tags	11.2.0	11.3.0
2012-06	CT#56	CP-120317	0677	4	CS to PS SRVCC - enhancements of PS to CS SRVCC	11.2.0	11.3.0
2012-06	CT#56	CP-120317	0678	1	CS to PS SRVCC - transaction identifier	11.2.0	11.3.0
2012-06	CT#56	CP-120317	0679	1	CS to PS SRVCC - flows - transaction identifier	11.2.0	11.3.0
2012-06	CT#56	CP-120304	0681	1	Path header set by ATCF	11.2.0	11.3.0
2012-06	CT#56	CP-120307	0682	1	MSC server registration when the access transfer is unsuccessful	11.2.0	11.3.0
2012-06	CT#56	CP-120317	0689	2	Definition of feature tag cs to ps SRVCC in alerting phase	11.2.0	11.3.0
2012-06	CT#56	CP-120317	0690	3	Addition of UE capability for CS to PS SRVCC in alerting phase to PS registration	11.2.0	11.3.0
2012-06	CT#56	CP-120317	0691	2	Addition of capability for CS to PS SRVCC in alerting phase to CS registration	11.2.0	11.3.0
2012-06	CT#56	CP-120317	0692	3	Definition of XML schema for indicating support of CS to PS SRVCC in alerting phase	11.2.0	11.3.0
2012-06	CT#56	CP-120297	0699	1	Pay Load type correction – Additional sessions SRVCC	11.2.0	11.3.0
2012-06	CT#56	CP-120304	0701		Pay Load type correction – Single (Held or Alerting) session SRVCC	11.2.0	11.3.0
2012-06	CT#56	CP-120304	0703	1	Removing one EN about charging and ATCF	11.2.0	11.3.0
2012-06	CT#56	CP-120288	0707	3	Payload type correction – media interworking	11.2.0	11.3.0
2012-06	CT#56	CP-120317	0708	1	CS to PS SRVCC - CS registration	11.2.0	11.3.0
2012-06	CT#56	CP-120307	0709	2	Signalling flows of eSRVCC in case of roaming	11.2.0	11.3.0
2012-06	CT#56	CP-120307	0710		Specification clean up	11.2.0	11.3.0
2012-06	CT#56	CP-120316	0711	1	Service continuity and support of enterprises	11.2.0	11.3.0
2012-09	CT#57	CP-120575	0671	3	Use of Feature-Caps header to indicate the support of mid-call feature for SCC AS	11.3.0	11.4.0
2012-09	CT#57	CP-120582	0713	1	Reference update and technical changes: draft-ietf-sipcore-proxy-feature	11.3.0	11.4.0
2012-09	CT#57	CP-120593	0714		CS to PS SRVCC - session transfer cancellation	11.3.0	11.4.0
2012-09	CT#57	CP-120593	0715		Minor numbering, spelling and styles corrections	11.3.0	11.4.0
2012-09	CT#57	CP-120593	0716		CS to PS SRVCC, alerting and held only	11.3.0	11.4.0
2012-09	CT#57	CP-120593	0717		CS to PS SRVCC - PS to CS SRVCC clean up and EN removal	11.3.0	11.4.0
2012-09	CT#57	CP-120593	0718	1	Incorrect info package name g.3gpp.state-and-event-info	11.3.0	11.4.0
2012-09	CT#57	CP-120580	0722	1	Introduction of OMR in SCC AS	11.3.0	11.4.0
2012-09	CT#57	CP-120582	0727	1	Payload type correction – SIP 488 (Not Acceptable Here) response	11.3.0	11.4.0
2012-09	CT#57	CP-120587	0730		eMPS for vSRVCC	11.3.0	11.4.0
2012-12	CT#58	CP-120792	0734	1	Handling of 1xx responses in alerting phase	11.4.0	11.5.0
2012-12	CT#58	CP-120790	0739	1	Removing an EN regarding overlaping procedures	11.4.0	11.5.0
2012-12	CT#58	CP-120806	0740		Removing EN regarding "related-icid"	11.4.0	11.5.0
2012-12	CT#58	CP-120793	0745	3	Correcting PS to PS access transfer	11.4.0	11.5.0
2012-12	CT#58	CP-120793	0746	1	Adding the static STI address example in A.2.2	11.4.0	11.5.0
2012-12	CT#58	CP-120792	0748	1	SCC AS handling corrections	11.4.0	11.5.0
2012-12	CT#58	CP-120806	0749	1	CS to PS SRVCC, cancellation after preparation	11.4.0	11.5.0
2012-12	CT#58	CP-120806	0750		Content-Disposition in INFO messages	11.4.0	11.5.0
2012-12	CT#58	CP-120806	0751		Editor's notes in A.20.1	11.4.0	11.5.0
2012-12	CT#58	CP-120806	0752	1	Editor's note in 12.7.3.1	11.4.0	11.5.0
2012-12	CT#58	CP-120791	0754	2	Feature-Caps header field in target refresh requests and responses	11.4.0	11.5.0
2012-12	CT#58	CP-120792	0756	2	Originating UE behaviours on the ringing tone when the aSRVCC occurs	11.4.0	11.5.0
2012-12	CT#58	CP-120793	0758	1	Corrections to signalling flows on originating aSRVCC call	11.4.0	11.5.0
2012-12	CT#58	CP-120793	0760	2	Flows for originating call is in alerting phase using SRVCC procedure with ATCF	11.4.0	11.5.0
2012-12	CT#58	CP-120793	0762	2	Flows for terminating call is in alerting phase using SRVCC procedure with ATCF	11.4.0	11.5.0
2012-12	CT#58	CP-120911	0763	2	Clarification to the SRVCC behavior of non-ICS based UE	11.4.0	11.5.0
2012-12	CT#58	CP-120806	0765	1	Editor's note in 6.3.2	11.4.0	11.5.0
2012-12	CT#58	CP-120806	0766	1	Editor's note in 12.7.3.2	11.4.0	11.5.0
2012-12	CT#58	CP-120806	0767	1	Encoding of media feature tag values of string type in Contact	11.4.0	11.5.0
2012-12	CT#58	CP-120806	0768	2	Editor's notes in 12.4.2.2 and A.20.2	11.4.0	11.5.0
2012-12	CT#58	CP-120806	0769		Editor's note in 12.7.3.3	11.4.0	11.5.0
2012-12	CT#58	CP-120793	0770		Authorization of early media during SRVCC	11.4.0	11.5.0
2012-12	CT#58	CP-120793	0771		Authorization of early media during PS-PS access transfer	11.4.0	11.5.0

2012-12	CT#58	CP-120853	0778	2	SCCF AS Procedures for alerting sessions using SRVCC procedure with ATCF	11.4.0	11.5.0
2012-12	CT#58	CP-120806	0779		Incorrect numbering of subclauses	11.4.0	11.5.0
2013-03	CT#59	CP-130106	0781	2	vSRVCC behavior of non-ICS based UE	11.5.0	11.6.0
2013-03	CT#59	CP-130102	0783	1	404 and 480 from ATCF	11.5.0	11.6.0
2013-03	CT#59	CP-130116	0787	3	Correction of PS-CS Access Transfer Procedure	11.5.0	11.6.0
2013-03	CT#59	CP-130102	0789		Reference update: RFC 6809	11.5.0	11.6.0
2013-03	CT#59	CP-130116	0796	1	Correcting charging correlation	11.5.0	11.6.0
2013-03	CT#59	CP-130112	0799		Authorization of early media during CS to PS SRVCC	11.5.0	11.6.0
2013-03	CT#59	CP-130100	0801	1	SCC AS capability of MSC server assisted mid-call feature	11.5.0	11.6.0
2013-03	CT#59	CP-130112	0807		Incorrect name of Table C.12A-1	11.5.0	11.6.0
2013-03	CT#59	CP-130112	0808	1	Fixing ATCF procedures when no active call exists	11.5.0	11.6.0
2013-03	CT#59	CP-130103	0810	2	Alerting phase definition and interaction with CAT	11.5.0	11.6.0
2013-03	CT#59	CP-130097	0818		Feature-Tags with "none" value correction	11.5.0	11.6.0
2013-03	CT#59	CP-130128	0790	2	Dual radio enhancements – Definitions, media feature tags and feature-capability indicators	11.6.0	12.0.0
2013-03	CT#59	CP-130128	0791	2	Dual Radio enhancements – Originating call setup	11.6.0	12.0.0
2013-03	CT#59	CP-130128	0792	2	Dual Radio enhancements – Terminating call setup	11.6.0	12.0.0
2013-03	CT#59	CP-130128	0793		Dual Radio enhancements - Access transfer introduction	11.6.0	12.0.0
2013-03	CT#59	CP-130124	0802	1	PS to CS SRVCC for originating calls in pre-alerting phase - registration	11.6.0	12.0.0
2013-03	CT#59	CP-130124	0803	1	PS to CS SRVCC for originating calls in pre-alerting phase - session setup	11.6.0	12.0.0
2013-03	CT#59	CP-130124	0804	1	PS to CS SRVCC for originating calls in pre-alerting phase - access transfer, single call	11.6.0	12.0.0
2013-03	CT#59	CP-130124	0805	1	PS to CS SRVCC for originating calls in pre-alerting phase - indications	11.6.0	12.0.0
2013-03	CT#59	CP-130124	0806		PS to CS SRVCC for originating calls in pre-alerting phase - XML	11.6.0	12.0.0
2013-03	CT#59	CP-130130	0813		Correction of g.3gpp.atcf-mgmt-uri feature capability indicator examples	11.6.0	12.0.0
2013-03	CT#59	CP-130087	0814	4	Summary of capabilities within service continuity	11.6.0	12.0.0
2013-03	CT#59	CP-130128	0794	2	Dual Radio enhancements - Access transfer impacts in the SC UE	11.6.0	12.0.0
2013-03	CT#59	CP-130181	0795	3	Dual Radio enhancements – Access transfer impacts on SCC AS	11.6.0	12.0.0
2013-06	CT#60	CP-130265	0815	3	Non-dependence of SIP ACK request and CONNECT ACKNOWLEDGEMENT message	12.0.0	12.1.0
2013-06	CT#60	CP-130260	0819	4	Correction of eDRVCC procedures	12.0.0	12.1.0
2013-06	CT#60	CP-130240	0822	2	Permanent error codes in SIP responses	12.0.0	12.1.0
2013-06	CT#60	CP-130235	0825	6	SRVCC cancellation by the MSC server	12.0.0	12.1.0
2013-06	CT#60	CP-130265	0827	1	Removal of STN-SR in clause 9.3.1	12.0.0	12.1.0
2013-06	CT#60	CP-130265	0828	1	PS to CS SRVCC cancelled clarifications	12.0.0	12.1.0
2013-06	CT#60	CP-130223	0833		SCC AS acting as an routeing B2BUA	12.0.0	12.1.0
2013-06	CT#60	CP-130260	0834	1	DRVCC (CS to PS) – general parts	12.0.0	12.1.0
2013-06	CT#60	CP-130260	0835	2	DRVCC (CS to PS) – SC UE access transfer procedures	12.0.0	12.1.0
2013-06	CT#60	CP-130260	0836	2	DRVCC (CS to PS) – SCC AS access transfer procedures	12.0.0	12.1.0
2013-06	CT#60	CP-130260	0837	2	DRVCC (CS to PS) – media feature tags and feature-capability indicators	12.0.0	12.1.0
2013-06	CT#60	CP-130260	0838	3	Small error corrections in PS to CS DRVCC	12.0.0	12.1.0
2013-06	CT#60	CP-130260	0839	2	Completion of summary of capabilities within service continuity for DRVCC	12.0.0	12.1.0
2013-06	CT#60	CP-130256	0840	2	Fixing error created by interaction of CR#0804 and CR#0783	12.0.0	12.1.0
2013-06	CT#60	CP-130265	0841		Inconsistent alerting phase terminology	12.0.0	12.1.0
2013-06	CT#60	CP-130235	0844	1	PS to CS SRVCC and malicious REFER transferring additional session	12.0.0	12.1.0
2013-06	CT#60	CP-130256	0845	2	PS to CS SRVCC for originating calls in pre-alerting phase - access transfer, additional call	12.0.0	12.1.0
2013-06	CT#60	CP-130260	0846	2	DRVCC and bSRVCC indicators in CS to PS SRVCC access transfer	12.0.0	12.1.0
2013-06	CT#60	CP-130243	0849	1	Correct related-ICID handling for SCC AS and ATCF	12.0.0	12.1.0
2013-06	CT#60	CP-130256	0850	1	PS to CS SRVCC for originating calls in pre-alerting phase - minor changes	12.0.0	12.1.0
2013-06	CT#60	CP-130265	0851	1	Network decisions for SRVCC transfer scenarios based on UE and Network support	12.0.0	12.1.0
2013-06	CT#60	CP-130265	0852	1	Completion of summary of capabilities within service continuity	12.0.0	12.1.0
2013-06	CT#60	CP-130223	0857	3	Compliancy statement for SCC AS	12.0.0	12.1.0
2013-06	CT#60	CP-130235	0866	2	Completion of offer answer before aSRVCC	12.0.0	12.1.0
2013-06	CT#60	CP-130236	0869	2	Providing the PS to CS SRVCC information	12.0.0	12.1.0
2013-06	CT#60	CP-130240	0872	1	Correcting handling of payload types in SC UE	12.0.0	12.1.0
2013-06	CT#60	CP-130223	0877	1	PS to PS - payload types in transfer of session	12.0.0	12.1.0
2013-06	CT#60	CP-130240	0880	1	PS to CS cancellation during an emergency call	12.0.0	12.1.0
2013-06	CT#60	CP-130246	0882		CS to PS SRVCC - payload types in transfer of additional session	12.0.0	12.1.0
2013-06	CT#60	CP-130265	0883		Error in flows	12.0.0	12.1.0
2013-06	CT#60	CP-130246	0887	1	CS to PS SRVCC - Extensions field usage	12.0.0	12.1.0

2013-06	CT#60	CP-130260	0888	1	Correcting DRVCC term alerting	12.0.0	12.1.0
2013-06	CT#60	CP-130236	0891		Handover cancellation support in SCC AS	12.0.0	12.1.0
2013-06	CT#60	CP-130265	0893	1	Clarification of UE behaviour for SRVCC	12.0.0	12.1.0
2013-06	CT#60	CP-130251	0895	2	Updating SDP-CS draft reference	12.0.0	12.1.0
2013-06	CT#60	CP-130232	0899	1	mid-call media feature tag IANA registration	12.0.0	12.1.0
2013-06	CT#60	CP-130236	0902		Completed IANA registrations for 24.237 from R10 on	12.0.0	12.1.0
2013-06	CT#60	CP-130232	0906	2	Resolving IANA comments on XML Schemas in 24.237	12.0.0	12.1.0
2013-09	CT#61	CP-130482	0912	2	Removing the mandate for the SCC AS to have to support PS-PS access transfer	12.1.0	12.2.0
2013-09	CT#61	CP-130511	0913		T313 expiry in the MSC server	12.1.0	12.2.0
2013-09	CT#61	CP-130511	0914		PS to CS SRVCC of calls in alerting phase - invocation condition correction	12.1.0	12.2.0
2013-09	CT#61	CP-130508	0915	1	Removing duplicate text on DRVCC and SRVCC indicator inclusion	12.1.0	12.2.0
2013-09	CT#61	CP-130508	0916		DRVCC and SRVCC indicator inclusion in PS to PS access transfer and Gm control adding	12.1.0	12.2.0
2013-09	CT#61	CP-130492	0919	1	Providing Contact of far end on SRVCC	12.1.0	12.2.0
2013-09	CT#61	CP-130511	0920		MSC server trough-connection and alignment with 24.008	12.1.0	12.2.0
2013-09	CT#61	CP-130511	0921	1	ATCF server trough-connection	12.1.0	12.2.0
2013-09	CT#61	CP-130508	0922	1	PS-CS Dual Radio access transfer scenarios	12.1.0	12.2.0
2013-09	CT#61	CP-130508	0923		PS to CS DRVCC – MSC server enhancements	12.1.0	12.2.0
2013-09	CT#61	CP-130495	0926	2	Dual radio MID-CALL error corrections and clarifications	12.1.0	12.2.0
2013-09	CT#61	CP-130508	0928	2	eDRVCC - Minor corrections and clarifications	12.1.0	12.2.0
2013-09	CT#61	CP-130491	0931	1	SRVCC support of RFC 3262	12.1.0	12.2.0
2013-09	CT#61	CP-130508	0932		Inconsistent setting of the TI flag	12.1.0	12.2.0
2013-09	CT#61	CP-130511	0933		Deregistration of the SC UE	12.1.0	12.2.0
2013-09	CT#61	CP-130511	0935	1	SDP handling at ATCF	12.1.0	12.2.0
2013-09	CT#61	CP-130495	0939	1	Correction for call termination procedures in the ATCF	12.1.0	12.2.0
2013-09	CT#61	CP-130495	0942	2	SCC AS send REFER to ATCF in case of PS to CS eSRVCC with additional call	12.1.0	12.2.0
2013-09	CT#61	CP-130511	0947	1	Correction for signalling flows for CS to PS SRVCC	12.1.0	12.2.0
2013-09	CT#61	CP-130511	0948	1	Correction for the functionalities division of the SC UE	12.1.0	12.2.0
2013-12	CT#62	CP-130763	0892	3	Option for eSRVCC for SCC AS procedure	12.1.0	12.2.0
2013-12	CT#62	CP-130763	0950	2	About the decision condition of the initial SIP INVITE request due to ATU-STI at the SCC AS	12.2.0	12.3.0
2013-12	CT#62	CP-130763	0951	3	About the UE and the MSC server call state in case of PS to CS SRVCC with the CAT service	12.1.0	12.2.0
2013-12	CT#62	CP-130763	0952	1	Correcting for the state change in case of aSRVCC	12.2.0	12.3.0
2013-12	CT#62	CP-130763	0953	1	Editorial correction about SRVCC	12.1.0	12.2.0
2013-12	CT#62	CP-130763	0954		Correcting an error in TIR interaction	12.2.0	12.3.0
2013-12	CT#62	CP-130740	0957	1	The MSC server mid-call support correction	12.1.0	12.2.0
2013-12	CT#62	CP-130759	0958	1	The MSC server early session support correction	12.2.0	12.3.0
2013-12	CT#62	CP-130740	0961	2	Release of session subject to access transfer in SC UE	12.1.0	12.2.0
2013-12	CT#62	CP-130734	0964	2	ATCF including the P-Asserted-Identity header field towards the MSC server	12.2.0	12.3.0
2013-12	CT#62	CP-130732	0967	2	SCC AS including the P-Asserted-Identity and the Contact header field towards the MSC server	12.1.0	12.2.0
2013-12	CT#62	CP-130763	0968		SC UE deregistration	12.2.0	12.3.0
2013-12	CT#62	CP-130744	0970		Removing editor's note regarding potential overlap and interactions	12.1.0	12.2.0
2013-12	CT#62	CP-130763	0971		SCC AS not sending re-INVITE to remote UE when PS to CS SRVCC access transfer transfers single held call	12.2.0	12.3.0
2013-12	CT#62	CP-130763	0973	1	Release resources in the ATGW	12.1.0	12.2.0
2013-12	CT#62	CP-130763	0974	1	Indications of support for access transfer features	12.2.0	12.3.0
2013-12	CT#62	CP-130732	0977	1	SCC AS receives CANCEL from MSC	12.1.0	12.2.0
2013-12	CT#62	CP-130732	0980	1	norefersub in Require / Supported	12.2.0	12.3.0
2013-12	CT#62	CP-130752	0981	1	Missing Contact and P-Asserted-Identity header fields	12.1.0	12.2.0
2013-12	CT#62	CP-130759	0982	1	Resolving editor's notes	12.2.0	12.3.0
2013-12	CT#62	CP-130763	0983	1	Correcting ATU-STI to be ATU-STI for PS to CS SRVCC	12.1.0	12.2.0
2013-12	CT#62	CP-130763	0985	1	Correction of call flow for SRVCC procedure with ATCF in alerting phase	12.2.0	12.3.0
2013-12	CT#62	CP-130763	0986		Editorial correction in Annex F (Access Transfer Scenarios)	12.1.0	12.2.0
2013-12	CT#62	CP-130770	0987	1	Update TS 24.237 due to use of RFC 6665 replacing RFC 3265	12.2.0	12.3.0
2014-03	CT#63	CP-140143	1000		Improving the quality of TS 24.237	12.3.0	12.4.0
2014-03	CT#63	CP-140126	0990	2	Allow terminating sessions with voice media in a TA indicating voice over PS is not supported, when there is a session with voice media.	12.3.0	12.4.0
2014-03	CT#63	CP-140152	0991	1	Clarification of conditions for transfer of calls in an early phase	12.3.0	12.4.0
2014-03	CT#63	CP-140143	0992	1	Removing overlapping procedures in SCC AS when MSC server cancels the PS to CS SRVCC procedure	12.3.0	12.4.0
2014-03	CT#63	CP-140143	0993		Explaining 1xx, 18x and 2xx status codes	12.3.0	12.4.0
2014-03	CT#63	CP-140143	0994		Reference to subclause 6A.4.3A	12.3.0	12.4.0
2014-03	CT#63	CP-140152	0998	1	Conditions for applying PS to CS SRVCC access transfer for calls in originating pre-alerting phase added	12.3.0	12.4.0

2014-03	CT#63	CP-140152	0999		Annex F and multiple early sessions	12.3.0	12.4.0
2014-03	CT#63	CP-140143	1002	1	Supported capabilities for transfer scenarios with conference control	12.3.0	12.4.0
2014-03	CT#63	CP-140143	1003	1	SRVCC Transfer scenarios with conference control	12.3.0	12.4.0
2014-03	CT#63	CP-140123	1010		Correct request for PS to CS cancelled handling	12.3.0	12.4.0
2014-03	CT#63	CP-140129	1012	1	Deletion of feature tag g.3gpp.mgmt-uri	12.3.0	12.4.0
2014-06	CT#64	CP-140302	0997	2	IOI in MESSAGE with SRVCC information	12.4.0	12.5.0
2014-06	CT#64	CP-140330	1001	3	Setting the "MPTY auxiliary state" after access transfer of a conference call	12.4.0	12.5.0
2014-06	CT#64	CP-140314	1017	1	Restructuring annex F table and correcting a spelling error	12.4.0	12.5.0
2014-06	CT#64	CP-140330	1019	1	DRVCC Transfer scenarios with conference control	12.4.0	12.5.0
2014-06	CT#64	CP-140330	1022	2	Clarification on usage of values of feature capability indicators	12.4.0	12.5.0
2014-06	CT#64	CP-140330	1025	1	Corrections to Table F.2.3 (SRVCC Transfer scenarios with conference control)	12.4.0	12.5.0
2014-06	CT#64	CP-140311	1026	1	Correction of feature tag name from "g.3gpp.pstop" to "g.3gpp.pstops"	12.4.0	12.5.0
2014-06	CT#64	CP-140330	1027	1	Clarification on requirement for an announcement to the caller after aSRVCC	12.4.0	12.5.0
2014-06	CT#64	CP-140302	1030		EN removal: Feature-capability indicator and media feature tag IANA registration status	12.4.0	12.5.0
2014-06	CT#64	CP-140330	1031	1	Updating PS to CS SRVCC access transfer procedures for transfer of an IMS emergency call to protect from UEs constructing the sip.instance media feature tag value incorrectly	12.4.0	12.5.0
2014-06	CT#64	CP-140298	1035		Incorrect name of the media feature tag carrying instance-id	12.4.0	12.5.0
2014-06	CT#64	CP-140308	1037	1	Resolving TS 24.008 editor's note on handling ongoing signalling when a CS to PS handover occurs	12.4.0	12.5.0
2014-06	CT#64	CP-140324	1038	1	Indicating traffic leg - SRVCC	12.4.0	12.5.0
2014-06	CT#64	CP-140330	1044		Adding SRVCC and DRVCC scenarios for Held Call + Incoming Alerting Call	12.4.0	12.5.0
2014-06	CT#64	CP-140311	1046	1	Updating CS-SDP draft reference to RFC 7195	12.4.0	12.5.0
2014-09	CT#65	CP-140639	1042	2	Removing references in ATCF to TS 24.229 subclause 5.7.5.1	12.5.0	12.6.0
2014-09	CT#65	CP-140685	1050	4	HO cancel for calls in allerting phase	12.5.0	12.6.0
2014-09	CT#65	CP-140658	1051		Reference update: draft-holmberg-dispatch-iotl	12.5.0	12.6.0
2014-09	CT#65	CP-140637	1054	1	Alerting and forking	12.5.0	12.6.0
2014-09	CT#65	CP-140665	1055	3	SRVCC & WiFi HO race condition	12.5.0	12.6.0
2014-09	CT#65	CP-140665	1056	1	Editorial improvements	12.5.0	12.6.0
2014-09	CT#65	CP-140665	1057		SCC AS providing MSC server with dialog identifier of remote leg	12.5.0	12.6.0
2014-09	CT#65	CP-140645	1058	1	Missing references to subclause 6A.4.3A	12.5.0	12.6.0
2014-09	CT#65	CP-140665	1060	1	Updating IMEI URN draft reference to RFC 7254	12.5.0	12.6.0
2014-12	CT#66	CP-140817	1065	1	E-SRVCC only for sessions with confirmed dialog	12.6.0	12.7.0
2014-12	CT#66	CP-140820	1068	1	SCC AS handling of 5xx responses	12.6.0	12.7.0
2014-12	CT#66	CP-140834	1069	2	Preconditions in the MSC server for dual radio	12.6.0	12.7.0
2014-12	CT#66	CP-140822	1072		P-Charging-Vector missing in the MSC server	12.6.0	12.7.0
2014-12	CT#66	CP-140837	1075	1	"term-ioi" missing in 200 OK from ATCF	12.6.0	12.7.0
2014-12	CT#66	CP-140837	1077		norefersub missing in example flows	12.6.0	12.7.0
2014-12	CT#66	CP-140852	1078		Reference update: draft-holmberg-dispatch-iotl	12.6.0	12.7.0
2014-12	CT#66	CP-140830	1079	1	PS to CS SRVCC and precondition - enhanced Alt1	12.6.0	12.7.0
2014-12	CT#66	CP-140837	1080	1	Assignment of CS states when MSC server assisted mid-call feature is used	12.6.0	12.7.0
2014-12	CT#66	CP-140822	1084	2	P-CSCF releasing the source access leg during PS to CS SRVCC during an emergency call	12.6.0	12.7.0
2014-12	CT#66	CP-140830	1085	1	SIP BYE after CANCEL	12.6.0	12.7.0
2014-12	CT#66	CP-140837	1087	1	Correction for message flow for SRVCC	12.6.0	12.7.0
2014-12	CT#66	CP-140837	1088		The dialogID clarification	12.6.0	12.7.0
2015-03	CT#67	CP-150067	1086	1	Update REFER to reflect RFC 6665	12.7.0	12.8.0
2015-03	CT#67	CP-150052	1089	2	MSC server initiate ICS registration after SRVCC	12.7.0	12.8.0
2015-03	CT#67	CP-150059	1090		Correcting incorrect feature-capability indicator names and media feature tag names	12.7.0	12.8.0
2015-03	CT#67	CP-150055	1092		Correcting MIME type of MIME body transferring information for anchoring the 2nd call in ATCF during PS to CS SRVCC when CS to PS SRVCC is supported	12.7.0	12.8.0
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2015-03	CT#67	CP-150074	1097		Reference update: draft-holmberg-dispatch-iotl	12.7.0	12.8.0
2015-03	CT#67	CP-150062	1098	2	DRVCC when only held call exists	12.7.0	12.8.0
2015-03	CT#67	CP-150062	1100		Headings are not visible in print layout mode	12.7.0	12.8.0
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2015-06	CT#68	CP-150328	1107	1	Transfer of additional terminating alerting call clarification	13.0.0	13.1.0
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2015-09	CT#69	CP-150532	1193	1	Adding missing call clearing procedures	13.1.0	13.2.0
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2015-12	CT#70	CP-150682	1221	2	rSRVCC without anchored in ATGW EN deletion	13.2.0	13.3.0
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2016-03	CT#71	CP-160084	1232		Through-connection on receipt of INFO carrying a "call-accepted" indication in ATCF	13.3.0	13.4.0
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2016-03	CT#71	CP-160068	1236	1	Update draft-ietf-sipcore-refer-clarifications reference to reflect RFC 7647	13.3.0	13.4.0
2016-03	CT#71	CP-160084	1237	1	Supression of services and announcements during DRVCC	13.3.0	13.4.0
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Change history							
Date	**Meeting**	**TDoc**	**CR**	**Rev**	**Cat**	**Subject/Comment**	**New version**
2016-09	CT#73	CP-160518	1248		F	Emergency calls not impacted by the policy in /<X>/OperatorPolicy of TS 24.216	14.1.0
2016-09	CT#73	CP-160518	1251	1	F	Correction in detection of remote leg information	14.1.0
2016-12	CT#74	CP-160752	1252	2	B	SCC AS cache 18x	14.2.0
2016-12	CT#74	CP-160752	1253		F	MSC unable to generate offer upon receiving 488 response	14.2.0
2016-12	CT#74	CP-160712	1257		A	Correct incorrect MIME type in CS to PS SRVCC procedures	14.2.0
2016-12	CT#74	CP-160752	1258	3	B	Codec inquiry prior to PS to CS SRVCC access transfer	14.2.0
2016-12	CT#74	CP-160739	1259	2	B	Support for SRVCC of eCall over IMS at the EATF	14.2.0
2016-12	CT#74	CP-160798	1260	1	B	PS to CS session continuity, dual radio session transfer of emergency session	14.2.0
2016-12	CT#74	CP-160752	1261	1	F	SRVCC abnormal cases – emergency call	14.2.0
2017-03	CT#75	CP-170137	1262		F	Corrections of RTP related definitions	14.3.0
2017-03	CT#75	CP-170137	1263	1	B	Codec re-negotiation after session transfer (SRVCC)	14.3.0
2017-03	CT#75	CP-170134	1264	1	F	EATF performing DRVCC dependant on PANI	14.3.0
2017-03	CT#75	CP-170137	1265		F	Editor's note in subclause 12.7.2.6	14.3.0
2017-06	CT#76	CP-171093	1266	2	F	Corrections related to caching and provisional responses and PEM SIP header fields during PS to CS handover in pre-alerting phase at the SCC AS	14.4.0
2017-06	CT#76	CP-171093	1267	1	B	Response-Source header field handling completion	14.4.0
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2017-09	CT#77	CP-172121	1269	1	F	Source Leg Information in session transfer request	15.0.0
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2018-03	CT#79	CP-180154	1276	2	B	bSRVCC-MT XML	15.2.0
2018-03	CT#79	CP-180154	1277	2	B	Bsrvcc-MT session setup	15.2.0
2018-03	CT#79	CP-180090	1278	1	F	PANI addition by MSC server enhanced for ICS	15.2.0
2018-03	CT#79	CP-180066	1280	2	A	Clalrification on usage of pre Alerting SRVCC feature tag	15.2.0
2018-03	CT#79	CP-180154	1282	1	B	Pre-alerting value in state info	15.2.0
2018-06	CT#80	CP-181077	1283	1	F	Correction of text for transfer of additional session	15.3.0
2018-06	CT#80	CP-181077	1284	1	F	Correct format of integrity-protected in flows	15.3.0
2018-12	CT#82	CP-183071	1287		A	Rel-14 completed IANA registrations	15.4.0
2018-12	CT#82	CP-183076	1288		F	Rel-15 completed IANA registrations	15.4.0

History

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