ETSI TS 124 237 V10.20.0 (2016-01)

TECHNICAL SPECIFICATION

Universal Mobile Telecommunications System (UMTS);
LTE;
IP Multimedia (IM) Core Network (CN)
subsystem IP Multimedia Subsystem (IMS) service continuity;
Stage 3
(3GPP TS 24.237 version 10.20.0 Release 10)

Reference

RTS/TSGC-0124237vak0

Keywords

LTE,UMTS

ETSI

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

Important notice

The present document can be downloaded from:

http://www.etsi.org/standards-search

The present document may be made available in electronic versions and/or in print. The content of any electronic and/or print versions of the present document shall not be modified without the prior written authorization of ETSI. In case of any existing or perceived difference in contents between such versions and/or in print, the only prevailing document is the print of the Portable Document Format (PDF) version kept on a specific network drive within ETSI Secretariat.

Users of the present document should be aware that the document may be subject to revision or change of status. Information on the current status of this and other ETSI documents is available at

http://portal.etsi.org/tb/status/status.asp

If you find errors in the present document, please send your comment to one of the following services:

https://portal.etsi.org/People/CommiteeSupportStaff.aspx

Copyright Notification

ETSI

Intellectual Property Rights

IPRs essential or potentially essential to the present document may have been declared to ETSI. The information pertaining to these essential IPRs, if any, is publicly available for **ETSI members and non-members**, and can be found in ETSI SR 000 314: *"Intellectual Property Rights (IPRs); Essential, or potentially Essential, IPRs notified to ETSI in respect of ETSI standards"*, which is available from the ETSI Secretariat. Latest updates are available on the ETSI Web server (https://ipr.etsi.org/).

Pursuant to the ETSI IPR Policy, no investigation, including IPR searches, has been carried out by ETSI. No guarantee can be given as to the existence of other IPRs not referenced in ETSI SR 000 314 (or the updates on the ETSI Web server) which are, or may be, or may become, essential to the present document.

Foreword

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

The present document may refer to technical specifications or reports using their 3GPP identities, UMTS identities or GSM identities. These should be interpreted as being references to the corresponding ETSI deliverables.

The cross reference between GSM, UMTS, 3GPP and ETSI identities can be found under http://webapp.etsi.org/key/queryform.asp.

Modal verbs terminology

In the present document "**shall**", "**shall not**", "**should**", "**should not**", "**may**", "**need not**", "**will**", "**will not**", "**can**" and "**cannot**" are to be interpreted as described in clause 3.2 of the ETSI Drafting Rules (Verbal forms for the expression of provisions).

"**must**" and "**must not**" are **NOT** allowed in ETSI deliverables except when used in direct citation.

Contents

Foreword

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

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

Version x.y.z

where:

x the first digit:

1 presented to TSG for information;

2 presented to TSG for approval;

3 or greater indicates TSG approved document under change control.

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

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

1 Scope

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

[69]   3GPP TS 29.079: "Optimal media routeing within the IP Multimedia Subsystem (IMS)".

[70]   3GPP TS 29.280: "3GPP Sv interface (MME to MSC, and SGSN to MSC) for SRVCC".

[71]   3GPP TS 32.260: "Telecommunication management; Charging management".

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 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.

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.

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 or SIP URI representation of tel URI. The static STI is used for CS-PS transfer when dynamic STI is unavailable.

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.

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.

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.

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 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 SRVCC transfer. It is provided in the MIME body as defined in annex D.3.

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)
Local Operating Environment
Remote Leg
Target Access Leg
Source Access Leg
Emergency Session Transfer Number for SR VCC (E-STN-SR)

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

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].

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
SR-VCC   Single Radio VCC
STI   Session Transfer Identifier
STN   Session Transfer Number
STN-SR   Session Transfer Number - Single Radio

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

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 transferare 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;

- procedures for service continuity and MMTEL interactions are specified in clause 20.

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];

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, SR-VCC procedures assume that filter criteria cause all sessions subject to SRVCC to be anchored in an SCC AS as described in 3GPP TS 23.216 [15]).

Configuration of QoS assignment for SR-VCC 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 [15]).

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.

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; and

- the ATU-STI 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; and

- 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.

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 6.2, 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 application server 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.4.1, 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.

5.4 MSC server

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

In order to be compliant with 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 MGCF in annex A of 3GPP TS 24.229 [2] and the role of an MSC server enhanced for SRVCC using SIP interface as described in subclause 12.6; or

- an MSC server shall provide the role of an MSC server enhanced for ICS as specified in subclause 12.4.1

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, subclause 9.4 and subclause 12.4; or

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

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.

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

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 [71] 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.

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 ACTF 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; and

3) if decided to anchor the media in ATGW according to operator policy:

    NOTE: At this point, ATCF interacts with ATGW to provide information needed in the procedures below. The details of interaction between ATCF and ATGW are out of scope of this document.

a. upon receiving an SDP offer or answer included in a SIP message sent by the served UE within the dialog, replace the SDP in the received SIP message with updated SDP provided by ATGW; and

b. upon receiving an SDP offer or answer included in a SIP message sent by remote UE within the dialog, replace the SDP in the received SIP message with updated SDP provided by ATGW.

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

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

6.1 Introduction

Void.

6.2 SC UE

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.

If the SC UE not supporting ICS or with ICS capabilities disabled, and supports the multiple registrations as defined in 3GPP TS 24.229 [2] then the SC UE shall include the g.3gpp.accesstype media feature tag as described in subclause B.3 of 3GPP TS 24.292 [4] in the Contact header field of the SIP REGISTER request.

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: 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. UE 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 static 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.

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 determine that SRVCC is usable for the UE if the UE 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]).

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 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:

- SRVCC was usable and SRVCC is not usable now; or

- SRVCC was not usable and SRVCC is usable now;

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

6.3.3 SCC AS providing the SRVCC related information to the ATCF

In order to provide the 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 annex D.3; and

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.

6.4 MSC server

If the MSC server:

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

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

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 SRVCC related information".

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:

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 1: 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 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]; and

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];

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 2: The list of the exit points can be either obtained as specified in RFC 3263 [68] 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 3: The list of the entry points can be either obtained as specified in RFC 3263 [68] 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 4: The list of the I-CSCFs can be either obtained as specified in RFC 3263 [68] 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 504 (Server Time-Out) response, in accordance with the procedures in RFC 3261 [26].

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; and

NOTE 5: The ATCF Path URI is the URI which the ATCF inserted in the Path header field of the SIP REGISTER request.

NOTE 6: 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) insert a Feature-Caps header field with the g.3gpp.atcf feature-capability indicator containing the STN-SR allocated to ATCF included as described in IETF RFC 6809 [60].

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); and

4) 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.

6A Roles for General Capabilities

6A.1 Introduction

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

6A.2 UE roles

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 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 sessions and session transfer 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 sessions and session transfer procedures;

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 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 according to the home operator's list of preferred access networks contained in the PreferredAccessNetworks node. The UE can evaluate if session transfer 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 in accordance with user preferences if configured in the UE. The UE can evaluate if session transfer 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 based on the SC_non_transferrable_media node value.

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; and

- the C-MSISDN.

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

When sending SIP INVITE request and SIP 1xx or 2xx response to the SIP INVITE request towards the served SC UE and if the session being established is anchored in SCC AS as described in subclause 4.2.2 then the SCC AS shall include the g.3gpp.srvcc feature-capability indicator in a Feature-Caps header field.

The SCC AS shall include into the Feature-Caps header field of any target refresh request and, in each 1xx or 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; or

2) the SIP 1xx or 2xx response to the SIP INVITE request;

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 static 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; and

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 or 2xx response to the SIP INVITE request due to originating filter criteria if the SCC AS applies 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 SRVCC for calls in alerting phase in accordance with subclause 8.3.2.

When sending a SIP 1xx or 2xx response to a SIP INVITE request due to STN-SR, then the SCC AS shall include the saved Contact header field of the remote UE.

6A.4.1 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 [69].

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 3 of 3GPP TS 29.079 [69].

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 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.4.2 Void

6A.4.3 Void

6A.4.3A INVITE responses towards the MSC server

When sending a SIP 1xx or SIP 2xx response to a SIP INVITE request due to STN-SR 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 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 establishment 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 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.

6A.4.4 Void

6A.4.5 Void

6A.4.6 Rejecting malicious 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 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 SIP 403 (Forbidden) response as specified in 3GPP TS 24.229 [2].

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 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 INVITE and an SDP body is present in the response, the MSC server should re-initiate the initial INVITE using the part of the received SDP media description that the MSC server supports.

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]. If the SC UE supports the MSC server assisted mid-call feature, 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 INVITE request. If the

SC UE supports single radio PS to CS access transfer for calls in alerting state, 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 INVITE request.

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.

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 static 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].

If:

1. the SCC AS supports the MSC Server assisted mid-call feature according to operator policy;

2. 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; and

3. the SCC AS is aware that all MSC Servers in the network where the UE is registered which can be involved in the SRVCC procedures support the MSC Server assisted mid-call feature;

   NOTE 1: 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.

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 2xx response to the SIP INVITE request due to originating filter criteria.

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:

1. the SCC AS supports SRVCC for calls in alerting phase according to operator policy;

2. 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 due to originating filter criteria; and

3. the SCC AS is aware that all MSC servers in the network where the UE is registered which can be involved in SRVCC procedures support the SRVCC for calls in alerting phase;

   NOTE 2: 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.

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 any SIP 1xx or 2xx response to the SIP INVITE request due to originating filter criteria as described in IETF RFC 6809 [60].

If the SCC AS supports the 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.

When the SCC AS receives any SIP 1xx or 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 or SIP 2xx response;

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

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

   NOTE 3: If the SCC AS subsequently receives an initial INVITE request due to STN-SR, the SCC AS will include the saved P-Asserted-Identity in the 2xx response to the initial INVITE request due to STN-SR and the saved Contact header field of the remote UE in SIP 1xx or 2xx response to the initial 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.

7.5 Access Transfer Control Function (ATCF)

7.5.1 Distinction of requests

The ATCF needs to distinguish the following initial SIP requests:

- SIP INVITE requests with the ATCF URI for originating requests in the topmost Route header field. In the procedures below, such requests are known as "originating SIP INVITE requests".

- 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".

- 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".

- 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".

7.5.2 Call origination procedures in the ATCF

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 1   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.

Upon receiving the originating SIP INVITE request, the ATCF shall:

NOTE 2:   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; and

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

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

B)   if the originating SIP INVITE request 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 with an updated SDP offer using media parameters provided by the ATGW; and

NOTE 3:   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 1xx or 2xx response to the originating SIP INVITE request, the ATCF shall:

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

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

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

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

NOTE 5:   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 establishment 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 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.

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

Upon receiving a

1. originating SIP request other than INVITE, 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.

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.

1a) If the SC UE supports single radio PS to CS access transfer for calls in alerting state, 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 as described in IETF RFC 6809 [60], 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 180 (Ringing) response to the SIP INVITE request.

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:

      - 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.

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]; and

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

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

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

NOTE: If the SCC AS subsequently receives an initial INVITE request due to STN-SR, the SCC AS will include the saved P-Asserted-Identity in the 2xx response to the initial INVITE request due to STN-SR and the saved Contact header field of the remote UE in SIP 1xx or 2xx response to the initial 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 that all MSC Servers in the network where the UE is registered which can be involved in the 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.

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 SRVCC for calls in alerting phase according to operator policy; and

2. the SCC AS is aware that all MSC Servers in the network where the UE is registered which can be involved in the SRVCC procedures support 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 as described in IETF RFC 6809 [60].

If the SCC AS supports 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 2xx responses to the SIP INVITE request due to terminating filter criteria before forwarding the SIP 1xx and 2xx responses towards the remote UE.

8.4 Access Transfer Control Function (ATCF)

8.4.1 Distinction of requests

The ATCF needs to distinguish the following initial SIP requests:

- 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".

8.4.2 Call termination procedures in the ATCF

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 1 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.

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

NOTE 2: 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 and C-MSISDN:

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

b) if the terminating SIP INVITE request 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; and

NOTE 3: 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;

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

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

NOTE 4: If the ATCF subsequently receives an initial INVITE request due to STN-SR, the ATCF will include the saved the Contact header field of the remote UE in its SIP 200 (OK) response to the initial INVITE request due to STN-SR as describe in subclause 12.7.2.2.

before forwarding the request.

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; and

- 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.

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 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 then if the SC UE is using Gm, then for each 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 according to the ICS UE using Gm procedures for call origination as specified 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 (including the STI of the dialog to be transferred) 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.

- 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 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 full-duplex session with speech media component exists, first initiate the release of all the ongoing full-duplex sessions with speech media component except the full-duplex session with active speech media component that was most recently made active and then the SC UE shall transfer the remaining ongoing full-duplex session with active speech media component.

When transferring the session(s) not using ICS capabilities, the SC UE shall send, a CC SETUP message as specified in 3GPP TS 24.008 [8], to the SCC AS to 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 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 1: If only one session is subject for access transfer the session, CS resources will be released by 3GPP TS 24.008 [2] procedures.

NOTE 2: 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 PS-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.

9.2.1A 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; or

   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.

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; and

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

   C. transfer the ongoing full-duplex session with speech media component.

   NOTE 3: The ongoing full-duplex session with inactive speech media component is transferred to a held CS call.

The SC UE shall associate the additional transferred 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.

Table 9.2.1A-1: held session transaction identifier calculation formula

```
<transaction identifier of the additional transferred session> = (1 + <transaction identifier
    of the CS call established by the SETUP message>) modulo 7
```

If:

1. the SC UE has a subscription as described in subclause 7.2.2 for the ongoing full-duplex session with active speech media component; or

2. 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 the SC UE shall associate the participants extracted in subclause 9.1A with transaction identifiers calculated as in the table 9.2.1A-2 and with TI flag of the session. The offsets 0, 2, 3, 4, 5 are assigned to the participants in their order in the list of the extracted participants.

Table 9.2.1A-2: transaction identifier assignment for participants

```
<transaction identifier of participant> = (<transaction identifier of the conference> +
    <offset of participant>) modulo 7
```

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 the SC UE shall associate the participants extracted in subclause 9.1A with transaction identifiers calculated as in the table 9.2.1A-2 and with TI flag of the additional transferred session. The offsets 0, 1, 2, 3, 4 are assigned to the participants in their order in the list of the extracted participants.

When the transfer is completed, the SC UE shall:

1) if the call is an additional transferred session with inactive speech media, enter the "call 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" 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

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.

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 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 the SC UE shall transfer the remaining ongoing full-duplex session with active speech media component.

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 populate the SIP INVITE request as follows:

   1) the Request-URI set to the static STI; and

   2) 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.

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 the session not using ICS capabilities as follows in addition to the procedures described in subclause 9.2.2:

   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 annex D.1.3; and

   3. include in the Contact header field the g.3gpp.mid-call media feature tag as described in annex C.

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.

Upon receiving a SIP REFER request within the SIP session established by the SIP INVITE request 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; and

5. containing a MIME body of MIME type specified in the annex 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] and IETF RFC 4488 [20]; and

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. 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.mid-call media feature tag as described in annex C; and

    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 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.

        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.

Upon receiving a SIP 2xx response for the SIP INVITE request, 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 static STN, a STN-SR 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 static STN".

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 SIP re-INVITE 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 static STN, the SCC AS shall 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, and send a SIP re-INVITE request towards the remote UE using the existing established dialog. 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. 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 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. The identification of the associated dialog is subject to the following conditions:

1. if only one 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 dialog supporting a session with active speech media component;

2. if no 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 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 dialog that originates from the same device that initiated the received SIP INVITE request due to static STN. If more than one such dialogs exists from the same device, the SCC AS shall proceed with the next step in this list; and

NOTE 1: Whether the dialog originates from the same UE as the received SIP INVITE request 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 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 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; 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 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.

Continuing the session transfer procedures, 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) 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

3) a new SDP offer, including the media characteristics as received in the SIP INVITE request due to static STN, by following the rules of 3GPP TS 24.229 [2].

Upon receiving the SIP ACK request from the IM CN subsystem, then:

   - if the source access leg contains only one speech media component the SCC AS shall initiate release of the source access leg by sending a SIP BYE request toward the S-CSCF for sending to the served SC UE; or

   NOTE: The SC UE will receive this SIP BYE request only the SC UE is using Gm after the PS-CS access transfer is completed

   - If 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.

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 static STN or due to ATU-STI 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 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;

- 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; and

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.

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 static STN.

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 static 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 static 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 static 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 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 static STN as described in subclause 7.3.3.

If the SCC AS applies the MSC Server assisted mid-call feature, two confirmed dialogs supporting a session with speech media component exist for the user identified in the P-Asserted-Identity header field 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 3515 [13] and IETF RFC 4488 [20] in the dialog created by the SIP INVITE request due to static STN; or the SCC AS shall send a SIP REFER request towards the ATCF in accordance with the procedures specified in 3GPP TS 24.229 [2], IETF RFC 3515 [13] and IETF RFC 4488 [20] in the dialog created by the SIP INVITE due to ATU-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 additional transferred session, i.e. session with speech media component other than the session associated with the SIP INVITE request due to static 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"; and

    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:

        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.

4. the Content-Type header field with the value set to MIME type as specified in the annex D.1.3; and

5. a XML body compliant to the XML schema specified in the annex D.1.2. If

    A. the session associated with the SIP INVITE request due to static 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.

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 associate the SIP INVITE request with an ongoing dialog supporting a session. 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. 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. 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 supports the MSC Server assisted mid-call feature, in which case the identification of the associated dialog is subject to the following conditions:

1. if only one dialog supporting a session with active speech media component exists for the user identified in the P-Asserted-Identity header field and a 2xx response has been sent, then continue the session transfer procedures;

2. if no 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, then send a SIP 480 (Temporarily Unavailable) response to reject the SIP INVITE request relating to the session transfer;

3. if more than one 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 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; and

4. 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.

Continuing the session transfer procedures, the SCC AS sends 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 ACK request originated from the UE, the SCC AS shall initiate release of the source access leg by sending a SIP BYE request over the source access leg.

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 specified 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 a 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; and

2. if no dialog supporting a 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.

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 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 3515 [13] and IETF RFC 4488 [20] 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 "body" URI header field populated with an SDP body 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 annex D.1.3; and

5. a XML body compliant to the XML schema specified in the annex D.1.2.

9.4 MSC server enhanced for ICS

If the MSC server enhanced for ICS has registered for the user, it 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, it shall apply the procedures specified in subclause 9.5 and subclause 9.6.

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 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 annex D.1.3;

3. include in the Contact header field the g.3gpp.mid-call media feature tag as described in annex C; 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 SIP INFO request with the Info-Package header field containing the g.3gpp.mid-call package name, if the SIP INVITE request established a session with conference focus, then the MSC server shall associate the participants extracted from the application/vnd.3gpp.mid-call+xml MIME body with transaction identifiers calculated

as in the table 9.2.1A-2 and with TI flag of the session. The offsets 0, 2, 3, 4, 5 are assigned to the participants in their order in the list of the extracted participants.

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 specified 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 annex 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] and IETF RFC 4488 [20]; and

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 "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

   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.

      e. all or 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. 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.

If two sessions are transferred, the MSC server shall:

1. associate the SIP INVITE request for an 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;

2. if the SIP INVITE request for an additional inactive session established a session with conference focus then associate the participants extracted from the application/vnd.3gpp.mid-call+xml MIME body included in the SIP

REFER request with transaction identifiers calculated as in the table 9.2.1A-2 and with TI flag of the session. The offsets 0, 1, 2, 3, 4 are assigned to the participants in their order in the list of the extracted participants.

When the transfer is completed, the MSC server shall:

1) if the call is an additional transferred session with inactive speech media, enter the "call active" (N10) 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" auxiliary state (defined in 3GPP TS 24.083 [43]) for the held call.

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 can not 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 or SIP re-INVITE message to the SCC AS to resume the inactive session.

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.

NOTE: PS-PS access transfer procedures are also used for transfer of additional transferred session when several sessions are transferred by alerting SRVCC or MSC server assisted mid-call feature during PS-CS access transfer.

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 over 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 INVITE request (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 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 over 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 INVITE request (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 then the SC UE performing these actions itself.

10.2.1   Full session transfer

To initiate PS-PS access transfer for a session, the SC UE shall send a SIP INVITE request 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 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. 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";

      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 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

5. if the Source Access Leg is an early dialog and the session was created by a received SIP INVITE request, indicate support of the info package mechanism as specified in IETF RFC 6086 [54].

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.

Upon receiving SIP 2xx response for the SIP INVITE request sent over the Target Access Leg and sending SIP ACK request:

-    if the dialog over the Source Access Leg was a confirmed dialog and if it is still active, the SC UE shall send a SIP BYE request to the SCC AS over the Source Access Leg to terminate the original session; and

-    if the dialog over the Source Access Leg was an early dialog and if it is still active, the SC UE shall send a SIP CANCEL if the transferred session was originated by the UE over the source access, or a SIP 410 (Gone) response if the transferred session was terminated by the UE over the source access.

NOTE 2: 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 SC UE receives any SIP 4xx – 6xx response to the SIP INVITE request 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.

When the session:

-    was created by a received SIP INVITE request; and

-    was transferred using PS-PS access transfer procedures when the session was an early dialog;

then when the session is accepted the SC UE shall send the SIP INFO request inside the Target Access Leg containing:

1. an Info-Package header field as specified in IETF RFC 6086 [54] with g.3gpp.state-and-event info package name; and

2. application/vnd.3gpp.state-and-event-info+xml XML body with the event XML element containing "call-accepted" to indicate that the called party has answered the call.

10.2.1A   Additional procedures for full session transfer when MSC server assisted mid-call feature is supported

In addition to the procedures described in subclause 10.2.1, if the SC UE supports the MSC Server assisted mid-call feature, the SC UE shall include in the Contact header field of the SIP INVITE request the g.3gpp.mid-call media feature tag as described in annex C.

10.2.2   Partial session transfer

To initiate PS-PS access transfer for a session, the SC UE shall send a SIP INVITE request 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 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; and

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 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.

   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.

Upon receiving SIP 2xx response for the SIP INVITE request 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 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   Additional procedures for partial session transfer when MSC server assisted mid-call feature is supported

In addition to the procedures described in subclause 10.2.2, if the SC UE supports the MSC Server assisted mid-call feature, the SC UE shall include in the Contact header field of the SIP INVITE request the g.3gpp.mid-call media feature tag as described in annex C.

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:

- 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".

NOTE 1: If the Request-URI contains the additional transferred session SCC AS URI, the PS-PS access transfer procedure is used to transfer the additional transferred session during PS-CS access transfer with the MSC server assisted mid-call feature.

NOTE 2: 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 on the Target Access Leg due to STI, the SCC AS shall:

- associate the SIP INVITE received on the Target Access Leg with a previously established SIP dialog or a SIP dialog in early state 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 Replaces (see IETF RFC 3891 [10]) or Target Dialog header field (see IETF RFC 4538 [11]) of the SIP INVITE with the previously established SIP dialog or with a dialog in early state. 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. 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 is unable to associate the SIP INVITE with a unique previously established SIP dialog or 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 SIP INVITE request contains a Replaces header field:

  a) follow the procedures defined in IETF RFC 3891 [10] for replacing the Source Access Leg with the SIP request received on the Target Access Leg, including terminating the Source Access Leg by sending a SIP BYE towards the SC UE in accordance with 3GPP TS 24.229 [2] in case the Source Access Leg was an ongoing dialog, or by sending a SIP CANCEL request towards the SC UE in case the Source Access Leg was a dialog in early state and it was terminating for the SC UE, or by sending a SIP 410 (Gone) response towards the SC UE in case the Source Access Leg was a dialog in early state and it was originating for the SC UE; and

  b) either send a SIP re-INVITE request towards the remote UE using the existing established dialog or send SIP UPDATE request(s) towards the remote UE(s) using the existing early dialog(s) which were created by the same 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 existing established 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 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:

           - 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

           - 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,

    c) for a full media transfer, send a SIP BYE towards the SC UE in accordance with 3GPP TS 24.229 [2] in case the Source Access Leg was an ongoing dialog, or send a SIP CANCEL request towards the SC UE in case the Source Access Leg was a dialog in early state and it was terminating for SC UE, or send a SIP 410 (Gone) response towards the SC UE when the Source Access Leg was a dialog in early state and it was originating for SC UE; otherwise, for a partial media transfer, after receiving the SIP ACK request from the SC UE on the Target Access Leg, upon receiving an update (e.g. SIP re-INVITE) from the SC UE on the Source Access Leg, process the update request in accordance with 3GPP TS 24.229 [2].

If the Remote Leg is an early dialog then when receiving 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 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.

If, subsequent to initiating the SIP re-INVITE request or the SIP UPDATE request to the remote UE, and prior to the SIP ACK request being received on the Target Access Leg, the SCC AS decides (for any reason) to reject the access transfer request (e.g. by sending a 4xx response), the SCC AS shall release the Target Access Leg, retain the Source Access Leg, and update the remote leg to match the Source Access Leg.

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 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; and

    2. send SIP 200 (OK) response to the SIP INVITE request over the Target Access Leg.

10.3.3 Additional SCC AS procedures for PS to PS access transfer when MSC server assisted mid-call feature is supported

If:

    1. the SCC AS supports the MSC Server assisted mid-call feature according to operator policy;

    2. 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 STI; and

    3. the SCC AS is aware that all MSC Servers in the network where the UE is registered which can be involved in the 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 2xx response to the SIP INVITE request due to STI in addition to the procedures described in subclause 10.3.2.

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 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 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 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 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; or

    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;

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 static STN or an IMRN in the Request-URI. In the procedures below, such requests are known as "SIP INVITE requests due to static 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 recepetion 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 re-INVITE 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 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 re-INVITE to the remote end.

- The SCC AS shall include a new SDP offer in the 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 static 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;

    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 access transfer in Single Radio VCC. 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 access transfer in SR-VCC 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.

In order to fulfil the requirements for PS-CS access transfer in SR-VCC for calls in alerting state, the SC UE needs to be engaged in a session with speech media component in early dialog state according to the following conditions before SR-VCC access transfer is performed:

- 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:

- 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.

12.2 SC UE procedures for PS to CS access transfer, SR-VCC

12.2.1 General

The SC UE may be engaged in one or more ongoing sessions before SR-VCC 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 SR-VCC 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 an PS bearer with QCI=1 as specified in 3GPP TS 23.203 [65]).

for access transfer. Sessions considered for SR-VCC 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

- SR-VCC 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 SR-VCC the handover is performed on PS level. Due to this, the SIP dialog established over the source PS access network stays the same after SR-VCC procedures, e.g. the IP address of the UE, the Call-ID, the P-CSCF do not change. Therefore in this case a re-INVITE needs to be sent to add ICS-control for the CS bearer.

12.2.3 Not based on ICS

After successful SR-VCC procedures (as described in 3GPP TS 24.008 [8]), 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: 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 retained upon PS handover;

- the SC UE is not using ICS capabilities; and

- SR-VCC procedures (as described in 3GPP TS 24.008 [8]) have been completed;

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.

12.2.3A Not based on ICS with MSC Server assisted mid-call feature

After successful SR-VCC 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 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, the SC UE shall associate the additional transferred session with CS call with transaction identifier 1 and TI flag value as in mobile terminated call.

   NOTE: The session with active speech media component transaction identifier value is described in 3GPP TS 24.008 [8]

If a transferred session is with conference focus then the SC UE shall associate the transaction identifiers to participants as in subclause 9.2.1A.

If single session with inactive speech media component is transferred, the SC UE shall associate the transferred session with CS call with transaction identifier 0 and TI flag value as in mobile terminated call.

12.2.3B Alerting call

12.2.3B.1 General

The SC UE shall apply the procedures in subclauses 12.2.3B.3 for access transfer for calls in alerting state if:

1) the SC UE supports single radio PS to CS access transfer for calls in alerting state; and

2) there are one or more early dialogs created by the same SIP INVITE request with at least one dialog that is an early dialog supporting a session with active speech media component where the SC UE:

   - has sent a Contact header field in a SIP INVITE request or 180 (Ringing) response containing the g.3gpp.srvcc-alerting media feature tag (as described in annex C); and

   - has received a Feature-Caps header field in a SIP INVITE request or 180 (Ringing) response 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 access transfer for calls in alerting state if:

1) the SC UE supports single radio PS to CS access transfer for calls in alerting state;

2) there are several dialogs supporting more than one session where:

   a) there is at least one dialog supporting a session in the confirmed state with active speech media component;

   b) there are one or more early dialogs created by the same SIP INVITE request that has at least one dialog that is an early dialog supporting a session with active speech media component where the SC UE:

      - has sent a Contact header field in a SIP INVITE request or 180 (Ringing) response containing the g.3gpp.srvcc-alerting media feature tag (as described in annex C); and

      - has received a Feature-Caps header field in a SIP INVITE request or 180 (Ringing) response containing the g.3gpp.srvcc-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 access transfer for calls in alerting state 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 there are several dialogs supporting more than one session according to the following conditions:

1) there are no dialogs in the confirmed state supporting a session with active speech media component;

2) there is at least one dialog in the confirmed state supporting a session with inactive speech media component;

3) there is only one session with active speech media component, that has at least one dialog that is an early dialog; and

4) the SC UE:

   - has sent a Contact header field in a SIP INVITE request or 180 (Ringing) response containing the g.3gpp.srvcc-alerting media feature tag (as described in annex C);

   - has sent a Contact header field in a SIP INVITE request or 2xx response containing the g.3gpp.mid-call media feature tag (as described in annex C);

   - has received a Feature-Caps header field in a SIP INVITE request or 180 (Ringing) response containing the g.3gpp.srvcc-alerting feature-capability indicator; and

   - has received a Feature-Caps header field in a SIP INVITE request or 2xx response containing the g.3gpp.mid-call feature-capability indicator.

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 state following access transfer, then 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 state following access transfer, then the SC UE shall associate the transferred session that was in alerting state 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 state

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 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.4 Established call with a session in alerting state

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

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:

- 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).

- 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 if the SC UE the SC UE shall 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.

12.3 SCC AS

12.3.0 General

In the SR-VCC access transfer procedures the SCC AS shall only consider sessions   that have a speech media component and that are subject to SR-VCC as defined in subclause 4.2.2 for access transfer.

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 SR-VCC:

- SIP INVITE request routed to the SCC AS due to a STN-SR belonging to the subscribed user in the Request-URI. 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 re-INVITE request routed to the SCC AS containing one or more already existing media lines for audio indicate a CS bearer. In the procedures below, such requests are known as "SIP re-INVITE requests adding ICS control".

- SIP re-INVITE request routed to the SCC AS containing one or more already existing media lines for audio indicate the port set to "0". In the procedures below, such requests are known as "SIP re-INVITE requests for non-ICS control".

12.3.1 SCC AS procedures for PS to CS access transfer, SR-VCC

When the SCC AS receives a SIP INVITE request due to STN-SR on the Target Access Leg and the SCC AS applies MSC Server assisted mid-call feature, the SCC AS shall, in addition to the procedures in this subclause, follow the procedures in subclause 12.3.2.

When the SCC AS receives a SIP INVITE request due to STN-SR on the Target Access Leg, and the SCC AS does not apply MSC Server assisted mid-call feature as described in subclause 12.3.2, the SCC AS shall:

- follow the PS-CS access transfer procedures specified in subclause 9.3.2 for the session with active speech media component that was most recently made active and the related dialog is in confirmed state. However, the SCC AS does not initiate release for source access leg and does not send a SIP re-INVITE request to update the source access leg. If the SCC AS supports the PS to CS SRVCC of calls in alerting phase, the SCC AS shall 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] in the SIP 200 (OK) response to the SIP INVITE request due to STN-SR; and

- if the SCC AS supports SRVCC for calls in alerting phase than follow the PS-CS access transfer procedures specified in subclause 12.3.4 for the session where the SC UE has completed a reliable offer / answer procedure and with active speech media component and the related dialog is in early state.

If the SCC AS has sent a SIP 480 (Temporarily Unavailable) response to reject a SIP INVITE request due to STN-SR on the Target Access Leg:

1) if the speech media component to be transferred was the only media component in the SIP dialog, the SCC AS shall release the remote leg as specified in 3GPP TS 24.229 [2]; or

2) if the session contains other media components than the active speech media component, the SCC AS shall modify the remote leg and remove the speech media component, as specified in 3GPP TS 24.229 [2].

When the SCC AS receives a SIP re-INVITE request for adding ICS control, the SCC AS shall reply with a SIP 200 (OK) response, treat the ongoing CS call as established using Gm and follow the "SCC AS for service control over Gm" procedures as described in 3GPP TS 24.292 [4] for controlling the CS call.

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.

When the SCC AS receives a SIP re-INVITE request for non-ICS control, the SCC AS shall follow the media removal procedures as specified in 3GPP TS 24.229 [2] for removing PS media.

Unless the MSC Server assisted mid-call feature applies, or the access transfer for calls in alerting phase applies, as only the session with active speech media component which was made active most recently is transferred from PS to CS audio, the SCC AS shall drop all other previously existing speech media components from this UE and indicate them accordingly in the SDP Offer sent within SIP re-INVITE requests towards the remote UE or the SCC AS shall release the remote leg as specified in 3GPP TS 24.229 [2] if the speech media component to be transferred was the only media component in the SIP dialog.

If the SCC AS has executed the procedures for access transfer for calls in alerting phase and has sessions remaining with speech media component which it did not transfer, the SCC AS shall remove the speech media component from these sessions and indicate them accordingly in the SDP Offer sent within SIP UPDATE or SIP re-INVITE requests towards the remote UE or the SCC AS shall release the remote leg as specified in 3GPP TS 24.229 [2] if the speech media component to be transferred was the only media component in the SIP dialog.

If no in-dialog request is received in source access leg of each session with transferred speech media component within an operator defined time after sending of a SIP 2xx or 18x response to the SIP INVITE due to STN-SR the SCC AS shall:

1) release the source access leg of the session; and

2) if the remote leg of the session contains media components other than the speech media component then modify the remote leg of the session and remove all the media components apart from the speech media component.

12.3.2 SCC AS procedures for PS to CS access transfer with MSC server assisted mid-call feature, SR-VCC

If

1. 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;

2. 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

3. 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;

then SCC AS shall apply the MSC Server assisted mid-call feature as described in subclause 9.3.2A with the following differences:

1. the SCC AS shall release all the superfluous sessions with speech media component;

2. the SCC AS does not initiate release for Source Access Leg of the associated SIP dialogs;

3. the SCC AS does not send a SIP re-INVITE request to update the Source Access Leg of the associated SIP dialogs; and

4. if the SCC AS supports the PS to CS SRVCC of calls in alerting phase, the SCC AS shall 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] in the SIP 200 (OK) response to the SIP INVITE request due to STN-SR.

If the SCC AS also supports SRVCC for calls in alerting phase then after finishing the procedures of this subclause, the SCC AS shall perform the procedures in subclause 12.3.4.

12.3.3   SCC AS procedures for SR-VCC, abnormal case

12.3.3.1   SR-VCC 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 then the SCC AS shall wait until this transaction has completed and then continue with the steps described below;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;

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 access leg once the expiration of the timer described in subclause 12.3.1; 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 SR-VCC 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].

12.3.3.1A   SR-VCC 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 access transfer for calls in 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 12.3.1;

> 2) treat the SIP UPDATE request(s) as per procedures for removing and adding media as described in subclause 13.3.1; and

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, SIP INVITE request due to ATU-STI for PS to CS SRVCC, or SIP INVITE request transferring additional session 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 request 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 SIP INVITE request 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, 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 SR-VCC

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 ongoing session

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;

> - the target access leg of an additional transferred session 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 alerting phase,

after having initiated an access transfer that was triggered by a SIP INVITE request due to STN-SR and when the operator specific timer is still running, the SCC AS shall:

1) send the SIP 200 (OK) to the 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.

12.3.3.5 PS to CS SRVCC cancelled by MME/SGSN or release of the target access leg for session in early dialog phase

If the SCC AS applies the procedures for access transfer for calls in alerting phase or calls in pre-alerting phase (as specified in subclause 12.3.4), 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;

- the target access leg of an additional transferred session 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 alerting phase,

after having initiated an access transfer that was triggered by a SIP INVITE request due to STN-SR for a session which is still in early dialog state when the operator specific timer is still running, the SCC AS shall:

1) send the SIP 200 (OK) to the BYE 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, release the call according to procedures in 3GPP TS 24.229 [2].

NOTE 1: All protocols 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 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 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 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.

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.

If the SCC AS applies the procedures for access transfer for calls in alerting phase and when serving a terminating user, then if the SCC AS receives a SIP CANCEL request containing the protocol value "Q.850" and the "cause" header field parameter with a value different then value "31" (normal unspecified) cancelling SIP INVITE request due to STN-SR on:

- the target access leg;

- the target access leg of an additional transferred session 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 alerting phase,

then the SCC AS shall:

1) send a SIP 200 (OK) response to the SIP CANCEL request;

2) send a 486 (Busy) SIP response to the SIP INVITE request received earlier from the remote UE as specified in 3GPP TS 24.229 [2]; and

3) send a SIP 487 (Request Terminated) response to the SIP INVITE request due to STN-SR 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 according to procedures in 3GPP TS 24.229 [2].

12.3.4 SCC AS procedures for PS to CS access transfer when call is in alerting phase

12.3.4.1 General

The SCC AS shall apply the procedures for access transfer for calls in alerting phase as described in subclauses 12.3.4.2 and 12.3.4.3 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 specified in annex C; and

2. one of the following is true:

   A. there are one or more dialogs supporting a session where the SC UE has completed a reliable offer / answer procedure and with active speech media component existing for the served user identified in the P-Asserted-Identity header field such:

   a. all dialogs are early dialogs created by the same SIP INVITE request;

   b. SIP 180 (Ringing) response to SIP INVITE request was received in at least one of those early dialogs;

    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 where the SC UE has completed a reliable offer / answer procedure and 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 created by the same SIP INVITE request 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;

    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; 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.

The SCC AS shall apply the procedures 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;

2. void;

3. void; and

4. one of the following is true:

  A. two or more dialogs supporting sessions where the SC UE has completed a reliable offer / answer procedure and with speech media component exist for the served user identified in the P-Asserted-Identity header field 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;

    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;

    c. one dialog is a confirmed dialog with active speech media component and the remaining dialog(s) are early dialog(s) with active speech media component created by the same SIP INVITE request; and

    d. SIP 180 (Ringing) response to SIP INVITE request was received in at least one of those early dialogs; or

  B. two or more dialogs supporting sessions where the SC UE has completed a reliable offer / answer procedure and with speech media component exist for the served user identified in the P-Asserted-Identity header field 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;

    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;

    c. one dialog is a confirmed dialog with inactive speech media component and the remaining dialog(s) are early dialog(s) with active speech media component created by the same SIP INVITE request;

    d. SIP 180 (Ringing) response to SIP INVITE request was received in at least one of those early dialogs; 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 where the SC UE has completed a reliable offer / answer procedure and with speech media component exist for the served user identified in the P-Asserted-Identity header field 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;

    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;

    c. one dialog is a confirmed dialog with active speech media component, there are one or more dialogs that are confirmed dialogs with inactive speech media component and the remaining dialog(s) are early dialog(s) with active speech media component created by the same SIP INVITE request;

    d. SIP 180 (Ringing) response to SIP INVITE request was received in at least one of those early dialogs; and

    e. the SCC AS does   not apply the MSC server assisted mid-call feature as described in subclause 12.3.2.

12.3.4.2 SCC AS procedures for PS to CS access transfer for terminating call in alerting phase using SRVCC procedure

When the SCC AS applies procedures for access transfer for calls in alerting phase, and receives a SIP INVITE request due to STN-SR on the Target Access leg, the SCC AS shall associate the SIP INVITE request with an early dialog supporting a session for the user identified in the P-Asserted-Identity header field.

If the session is a terminating session in early dialog state available for the served user, 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 the SDP answer received in the SIP 200 (OK) response to the SIP UPDATE request.

Upon receiving the SIP PRACK request from the MSC Server, 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 3gpp.state-and-event info package name; and

2. include an application/vnd. 3gpp.state-and-event-info +xml XML body compliant to the XML schema specified in the annex 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 XML body compliant to the XML schema specified in the annex 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; and

2) a SIP 200 (OK) response to the SIP INVITE request due to STN-SR towards the MSC server to indicate the successful access transfer.

Upon receiving the SIP ACK request from the IM CN subsystem, then

1) if the SCC AS had previously received a SIP 200 (OK) response to the dialog that was previously in early state, from the SC UE, the SCC AS shall send a SIP ACK request to the SC UE; and

NOTE 1: The condition above covers the case where the UE answers the call in the PS domain prior to the completion of the handover to the CS domain, whilst the SCC AS is applying the PS to CS access transfer procedure specified above,

2) if the source access leg contains only one speech media component, the SCC AS shall initiate release of the source access leg by sending a SIP CANCEL request toward the S-CSCF for sending to the served SC UE. The SCC AS shall send the SIP CANCEL request only after an operator specific timer has expired.

NOTE 2: Delaying the SIP CANCEL request 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.

If the SCC AS receives a SIP 200 (OK) response to the dialog that was previously in early state, from the SC UE whilst the SCC AS is applying the PS to CS access transfer procedure specified above, the SCC AS does not confirm reception of the SIP 200 (OK) response with a SIP ACK request and performs no actions on dialogs with the remote party and with the MSC server.

12.3.4.3 SCC AS procedures for PS to CS access transfer for originating call in alerting phase using SRVCC procedure

When the SCC AS applies procedures for access transfer for an originating call in alerting phase, and receives a SIP INVITE request due to STN-SR on the Target Access leg, the SCC AS shall associate the SIP INVITE request with an early dialog or early dialogs related to the originating call for the user identified in the P-Asserted-Identity header field.

The SCC AS shall discard any SIP 1xx provisional responses or the SIP 200 (OK) response to the initial SIP INVITE request received from the remote UE until the SIP 200 (OK) response to the INFO request is received from the MSC server (see later steps in this subclause).

NOTE 1: SIP 1xx responses sent reliably and the SIP 200 (OK) response to the initial SIP INVITE request will be retransmitted by the remote UE if the responses are dropped by the SCC AS.

If there is only one early dialog related to the originating call in alerting phase available for the served user, 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 there are more than one early dialogs related to the originating call in alerting phase available for the served user due to forking as described in 3GPP TS 24.229 [2], the SCC AS shall update the remote legs by sending SIP UPDATE requests simultaneously towards every remote UE using the existing early dialogs 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.

Upon receiving the first SIP PRACK request from the MSC Server, 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 3gpp.state-and-event info package name; and

2. include application/vnd.3gpp.state-and-event-info+xml XML body containing a XML body compliant to the XML schema specified in the annex D.2 with the state-info XML element containing "early" the direction XML element containing "initiator".

Upon receiving the SIP ACK request from the IM CN subsystem, and if the source access leg contains only one speech media component initiate release of the source access leg by sending a 404 (Not Found) response toward the S-CSCF for sending to the served SC UE. The SCC AS shall send the SIP 404 (Not Found) response only after an operator specific timer has expired.

NOTE 2: Delaying the SIP 404 (Not Found) 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.

12.3.4.4 SCC AS procedures for PS to CS access transfer of waiting call

In order to transfer waiting call, the SCC AS shall send a SIP REFER request according to 3GPP TS 24.229 [2], IETF RFC 3515 [13] and IETF RFC 4488 [20] 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] and IETF RFC 4488 [20] in the dialog created by the SIP INVITE request due to ATU-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];

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, 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 early dialog 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"; 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. 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".

12.3.5 SCC AS procedures for PS to CS access transfer: SRVCC enhancement using ATCF

The SCC AS needs to distinguish the following initial SIP request:

- SIP INVITE requests routed to the SCC AS due to ATU-STI in the Request-URI. In the procedures below, such requests are known as "SIP INVITE requests due to ATU-STI".

Upon receiving a SIP INVITE request due to ATU-STI, 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 12.3.1 with the following differences:

      a) 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, then the SCC AS shall:

         i) not send a SIP re-INVITE request towards remote UE; and

         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;

   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; and

  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 12.3.1.

12.4 MSC server enhanced for ICS

12.4.0 MSC server enhanced for ICS supporting SRVCC

When an MSC server enhanced for ICS supporting SRVCC receives an indication for a session transfer as described in 3GPP TS 23.216 [49], then 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; and

  4) include an SDP offer only containing a speech media component .

  NOTE: MSC servers 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.

If the MSC server enhanced for ICS supports the MSC server assisted mid-call feature, it shall apply the procedures defined in subclause 12.4A.

After finishing the access transfer procedures, the MSC and if the access transfer was successful 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.

12.4.1 MSC server enhanced for ICS procedures for PS to CS access transfer for alerting calls

The MSC Server shall perform the procedures described in subclause 12.6.3 for the MSC server enhanced for SRVCC using SIP interface.

12.4.2 Abnormal cases

12.4.2.1 Permanent response codes

When the MSC server enhanced for ICS receives a SIP reject response to the SIP INVITE request due to STN-SR, the MSC server shall regard any of the following SIP reject responses as permanent errors:

- 404 Not found;

- 410 Gone;

- 484 Address Incomplete; and

- 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 as temporary errors.

NOTE: The procedures in 3GPP TS 29.280 [70] requires that the MSC server indicates whether a received SIP reject response to the SIP INVITE request due to STN-SR is temporary or permanent.

12.4.2.2 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 INVITE request due to STN-SR and if the dialog is a dialog with inactive or active speech media component, send a SIP BYE request;

2) if the MSC server applies the MSC server assisted mid-call feature, in the dialog created by the INVITE for the additional transferred session and if the dialog is a dialog with inactive speech media component, send a SIP BYE 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.

12.4A   MSC server assisted mid-call feature

An MSC server supporting the MSC server assisted mid-call feature shall apply procedures as described in subclause 9.5 with the following modifications:

NOTE 1: The MSC server assisted mid-call feature can only be supported by an MSC server enhanced for ICS (subclause 12.4) or an MSC server enhanced for SRVCC using SIP interface (subclause 12.6.1).

0.   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;

1.   if two sessions are transferred, associate the SIP INVITE request for an additional session with inactive speech media component with CS call in the Active (N10) state (as defined in 3GPP TS 24.008 [8]) and in the Call held auxiliary state (as defined in 3GPP TS 24.083 [43]) with transaction identifier 1 and TI flag value as in mobile terminated call; and

NOTE 2: The transaction identifier value for the session with active speech media component is described in 3GPP TS 24.008 [8]

2.   if single session is transferred and all the media in the SDP answer of the SIP 2xx response to the SIP INVITE request have directionality inactive, associate the SIP INVITE request for the session with CS call with transaction identifier 0 and TI flag value as in mobile terminated call.

NOTE 3: For an MSC server enhanced for SRVCC using SIP interface, following access transfer, the procedures for the handling of transferred conference participants are implementation dependent.

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.   associate the SIP INVITE request due to E-STN-SR with a source access leg, i.e. a session with active 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. 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-SR; and

2.   originate session modification as described in 3GPP TS 24.229 [2] towards the remote UE with a new SDP offer with media characteristics as received in 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 shall release the source access leg as described in 3GPP TS 24.229 [2].

NOTE 2: If non-speech media was part of the original emergency call session, the non-speech media will be released.

NOTE 3: 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.

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) to the 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 Reaseon 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.

12.6 MSC server enhanced for SRVCC using SIP interface

12.6.1 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 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 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; and

4) include an SDP offer only containing a speech media component.

If the MSC server enhanced for SRVCC using SIP interface supports the MSC server assisted mid-call feature then in addition to the procedures in this subclause it shall apply the procedures defined in subclause 12.4a.

If the MSC server enhanced for SRVCC using SIP interface supports PS to CS access transfer for alerting calls then in addition to the procedures in this subclause it shall apply the procedures defined in subclause 12.6.3.

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 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; and

4) include an SDP offer with media which the MSC server wishes to use in the session.

12.6.3 MSC server enhanced for SRVCC using SIP interface procedures for PS to CS access transfer for alerting calls

When an MSC server enhanced for SRVCC using SIP interface supports the PS to CS access transfer for calls in alerting state receives an indication for a session transfer as described in 3GPP TS 23.216 [49], then the MSC server

enhanced for SRVCC using SIP interface shall initiate a SIP INVITE request as described in subclause 12.6.1 and in addition it shall populate the INVITE request as follows:

1. an Accept header field containing the MIME type application/vnd.3gpp.state-and-event-info+xml as specified in annex D.2.3;

2. a Contact header field containing the g.3gpp.srvcc-alerting media feature tag as described in annex C;

3. a Recv-Info header field containing the g.3gpp.state-and-event package name; and

4. 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].

NOTE 1: 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 2: If the MSC server supports the MSC server assisted mid-call feature, a Supported header field containing the option-tag "norefersub" is inserted in subclause 9.5.

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 a XML body compliant to the XML schema specified in the annex 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 state (N4) 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].

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 a XML body compliant to the XML schema specified in the annex 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].

Upon receiving a CS Connect request when in Call Received state as specified in 3GPP TS 24.008 [8], the MSC server enhanced for SRVCC using SIP interface shall enter the Active state as specified in 3GPP TS 24.008 [8] and 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 compliant to the XML schema specified in the annex D.2 with the event XML element containing "call-accepted" to indicate that the called party has answered the call.

Upon receiving a CS Connect request after having sent the SIP INVITE request due to STN-SR when not in Call Received (N7) state as specified in 3GPP TS 24.008 [8], the MSC server enhanced for SRVCC using SIP interface will store this event. 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 a XML body compliant to the XML schema specified in the annex D.2 with the state-info XML element containing "early" and direction set to "receiver";

is received, then

1. the MSC server enhanced for SRVCC using SIP interface will enter Active (N10) state as specified in 3GPP TS 24.008 [8];

2. 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:

   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 compliant to the XML schema specified in the annex D.2 with the event XML element containing "call-accepted" to indicate that the called party has answered the call.

   NOTE 3: Procedures in the MSC server enhanced for SRVCC using SIP interface 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";

the MSC server shall:

1. handle the SIP REFER request as specified in 3GPP TS 24.229 [2], IETF RFC 3515 [13] and IETF RFC 4488 [20]; and

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 in the Contact header field the g.3gpp.srvcc-alerting media feature tag as described in annex C; and

   C. 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. 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.

   NOTE 4: 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.

3. if application/vnd.3gpp.state-and-event-info+xml MIME body contains direction XML element containing "initiator", then enter the Call Delivered state (N4) 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; and

4. if application/vnd.3gpp.state-and-event-info+xml MIME body contains 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.

Upon receiving a CS Connect message with transaction identifier 1 and TI flag value as in mobile terminated call when in Call Received 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:

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 compliant to the XML schema specified in the annex D.2 with the event XML element containing "call-accepted" to indicate that the called party has answered the call.

NOTE 5: When the transfer is complete 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 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 12.4.2.1 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 12.4.2.2.

12.7 Access Transfer Control Function (ATCF)

12.7.1 Distinction of requests

The ATCF needs to distinguish the following initial SIP requests:

- SIP INVITE requests containing the STN-SR allocated to the ATCF in the Request-URI and:

  - not containing any 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 STN-SR".

12.7.2 ATCF procedures for PS to CS access transfer, SR-VCC

12.7.2.1 General

Upon receiving the SIP INVITE request due to STN-SR, 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 to the URI in the P-Asserted-Identity header field of the SIP INVITE requests 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.

12.7.2.2 Active session transfer

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; and

- the ATGW anchors the media of 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.

1) send a SIP 200 (OK) response to the received SIP INVITE request due to STN-SR that contains:

a) the saved Contact header field of the remote UE as describe in subclause 7.5.2;

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;

d) the Feature-Caps header field(s) according to IETF RFC 6809 [60], received in the home leg of the session being transferred;

e) the P-Asserted-Identity header field with the identity of the remote UE saved in subclause 7.5.2 or subclause 8.4.2; and

f) if available, the Privacy header field saved in subclause 7.5.2 or subclause 8.4.2; and

NOTE 2: 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 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 previously received from the SCC AS and associated with the session being transferred;

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 3: 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 INVITE request due to STN-SR;

   h) all header fields which are included in the INVITE request due to STN-SR and which contain option tag(s);

   i) if the Recv-Info header field is included in the INVITE request due to STN-SR, the Recv-Info header field that is the same as the Recv-Info header field received in the INVITE request due to STN-SR; and

   j) if the Accept header field is included in the INVITE request due to STN-SR, the Accept header field that is the same as the Accept header field received in the 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 ATGW does not anchor the media of 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 associated with the session being transferred;

before forwarding the request.

12.7.2.3   Abnormal procedures

12.7.2.3.1   P-CSCF releasing the source access leg during SR-VCC

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 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

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 respond with a SIP 480 (Temporarily Unavailable) response.

12.7.2.4   Transfer when only held or alerting session exist

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,   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 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 to transfer as follows:

   a). if

      A) one or more confirmed dialogs supporting a session with 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);

      select the confirmed dialog supporting a session with inactive speech media component that became inactive most recently; and

   b) if no confirmed dialog supporting a session with inactive speech media component exists in the transferable session set but there are one or more dialogs in the transferable session set supporting a session where the SC UE has completed a reliable offer / answer procedure and with an active speech media component such that:

      - all dialogs are early dialogs created by the same SIP INVITE request;

      - the Contact header field provided by the SC UE includes the g.3gpp.srvcc-alerting media feature tag as described in annex C; and

      - 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;

      then select any of the early dialogs;

3) 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 toward the SCC AS populated with:

   a) if

      - only one dialog exists in the session to be 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 to be 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) the Request-URI containing the ATU-STI previously received from the SCC AS and associated with the session being transferred;

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 in-dialog requests from the SCC AS;

NOTE: 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 INVITE request due to STN-SR;

h) all header fields which are included in the INVITE request due to STN-SR and which contain option tag(s);

i) if the Recv-Info header field is included in the INVITE request due to STN-SR, the Recv-Info header field that is the same as the Recv-Info header field received in the INVITE request due to STN-SR; and

j) if the Accept header field is included in the INVITE request due to STN-SR, the Accept header field that is the same as the Accept header field received in the INVITE request due to STN-SR.

Upon receiving an 18x or 2xx response to the SIP INVITE request due to ATU-STI from the SCC AS the ATCF shall before forwarding the SIP response to the SIP INVITE request due STN-SR on the target access leg replace the Record-Route header field with a Record-Route header field that contains only the SIP URI pointing to the ATCF.

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, it 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; and

- 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].

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.

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 the SCC AS shall determine if this SIP INVITE request is for an ongoing call by determining if the received contents of SIP INVITE request's Contact header field is bound to an ongoing CS call session identifier. If the SC UE has an ongoing CS call, the SCC AS shall:

- respond to the SIP INVITE request in accordance with the procedures defined in 3GPP TS 24.292 [4];

- 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

- if the SIP 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].

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 tranfer 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 TIP 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 componets, 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 servive 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.

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, PS to CS access transfer for calls in alerting phase is only supported if a 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

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 indentity of SC UE A.

- tel:+1-237-555-2222 represents the public user indentity of UE B.

- sip:sccas1.home1.net represents the Internet host of SCC AS.

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.
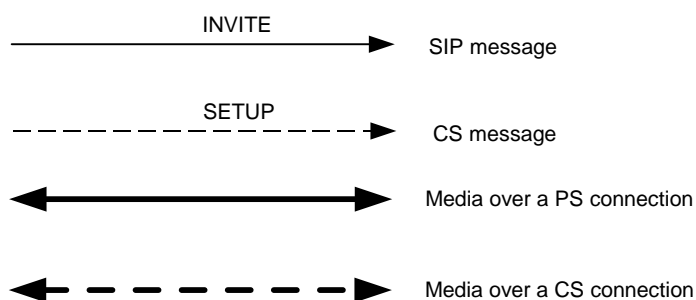
Figure A.2-1: Signalling flow notation

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] subclause 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.
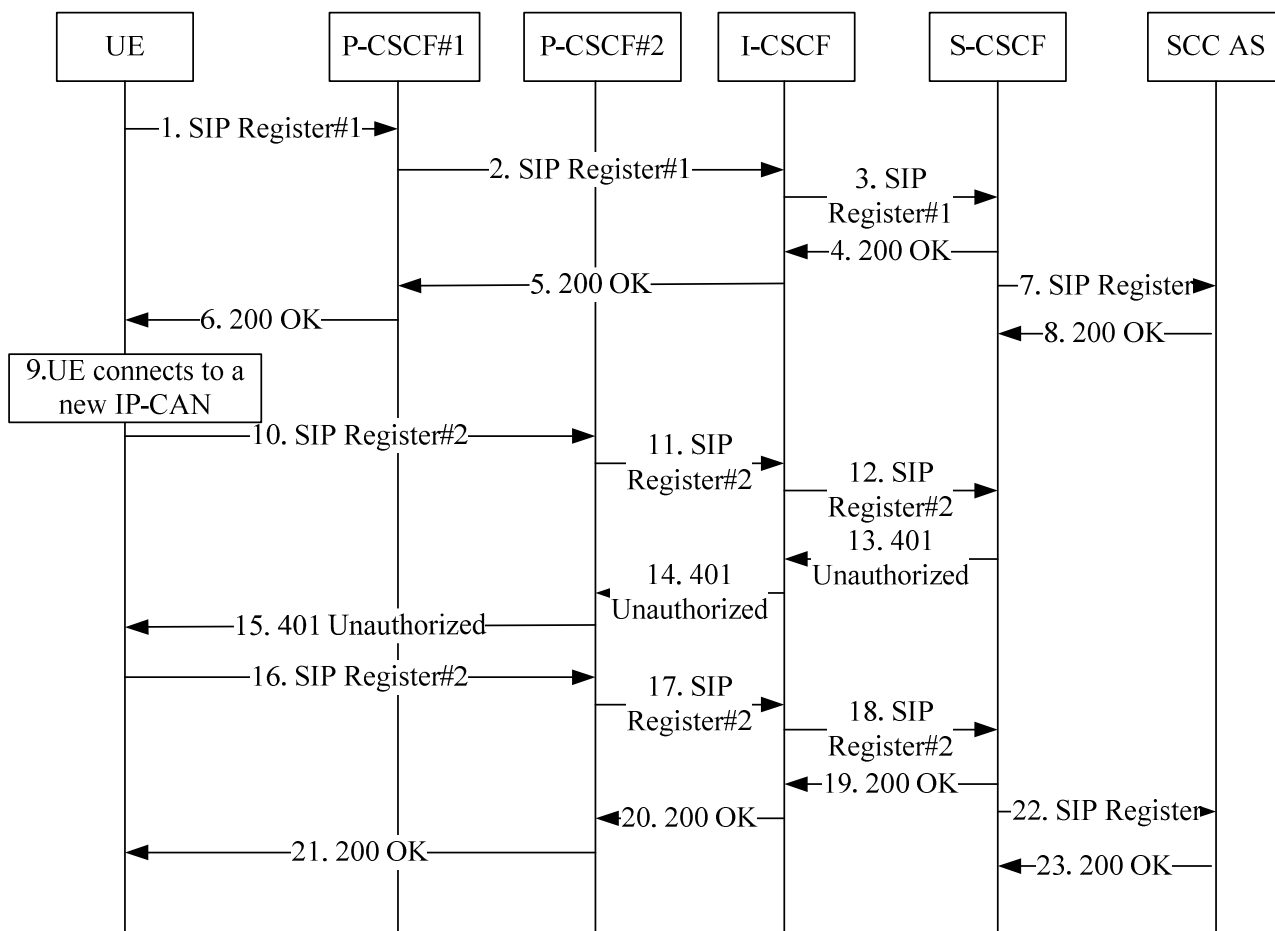
Figure A.3.2-1 Signalling flows for multiple registrations

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%3gpp-
   service.ims.icsi.mmtel";+g.3gpp.ics="principal";+g.3gpp.accesstype="cellular1";expires=6000
   00;+g.3gpp.iut-controller
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: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:
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%3gpp-
    service.ims.icsi.mmtel";+g.3gpp.ics="principal";+g.3gpp.accesstype="cellular1"
    ;expires=600000;+g.3gpp.iut-controller
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%3gpp-
service.ims.icsi.mmtel";+g.3gpp.ics="principal";+g.3gpp.accesstype="cellular1";expires=600000;+g.3gpp.
iut-controller
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%3gpp-
service.ims.icsi.mmtel";+g.3gpp.ics="principal"+g.3gpp.accesstype="cellular1"
;expires=600000;+g.3gpp.iut-controller
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%3gpp-
   service.ims.icsi.mmtel";+g.3gpp.ics="principal";+g.3gpp.accesstype="wlan2";expires=600000;+
   g.3gpp.iut-controller
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%3gpp-
service.ims.icsi.mmtel";+g.3gpp.ics="principal";+g.3gpp.accesstype
="wlan2";expires=600000;+g.3gpp.iut-controller
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%3gpp-
service.ims.icsi.mmtel";+g.3gpp.ics="principal";+g.3gpp.accesstype="wlan2"
;expires=600000;+g.3gpp.iut-controller
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.
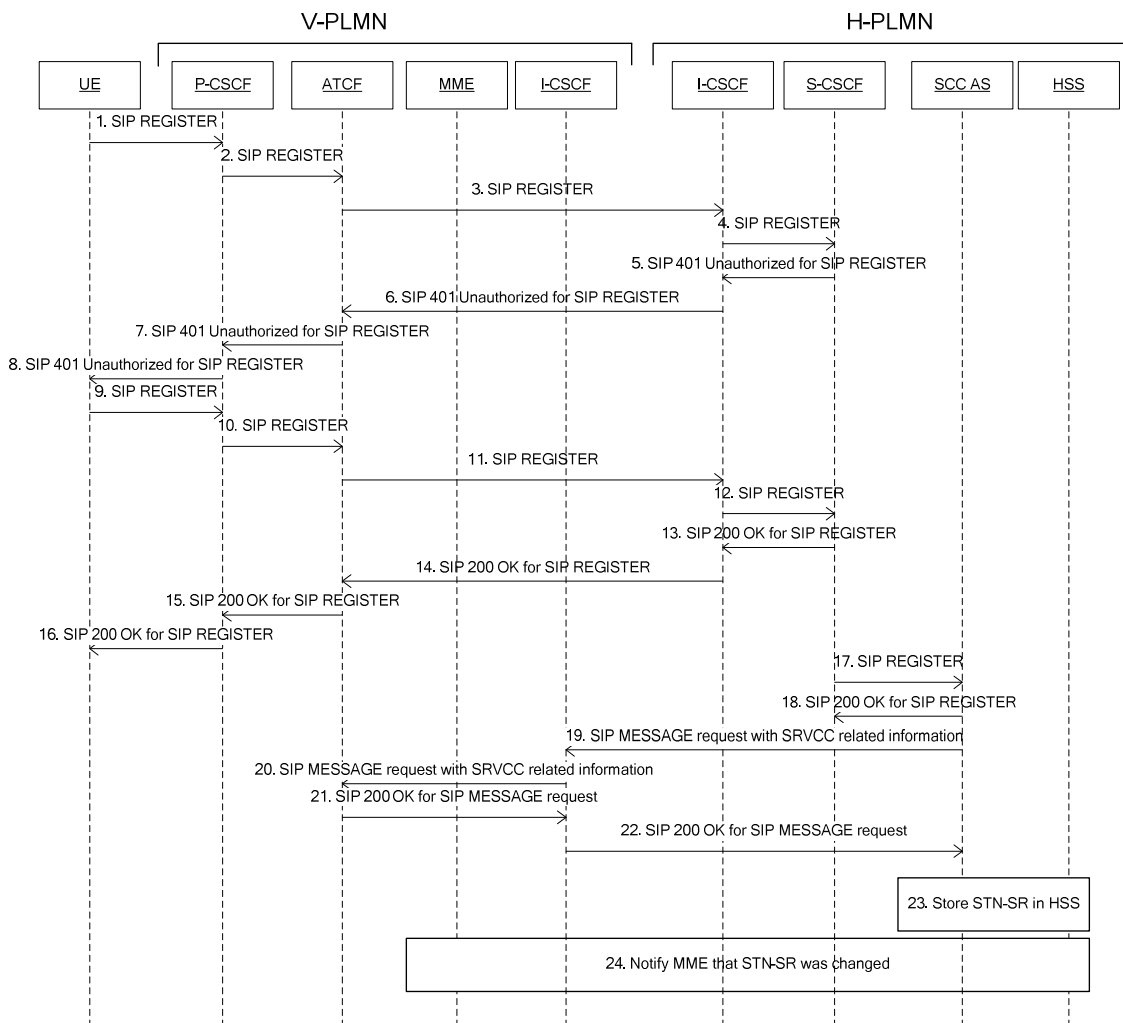
Figure A.3.3-1 registration with SRVCC enhancements

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%3gpp-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=
   "<sip:actf.visited2.net>";+g.3gpp.atcf-path="<sip:termsdgfdfwe@actf.visited2.net>"
Path: <sip:termsdgfdfwe@actf.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 actf.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 (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 and 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.

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=
   "<sip:actf.visited2.net>";+g.3gpp.atcf-path="<sip:termsdgfdfwe@actf.visited2.net>"
Path: <sip:termsdgfdfwe@actf.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 actf.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 (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 and 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.

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: 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@actf.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 actf.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=
   "<sip:actf.visited2.net>";+g.3gpp.atcf-path="<sip:termsdgfdfwe@actf.visited2.net>"
Path: <sip:termsdgfdfwe@actf.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
   actf.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%3gpp-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@actf.visited2.net>,<sip:orig@actf.visited2.net>,<sip:aga2gfgf@pcscf1.visi
   ted2.net:5070;ob>
Via: SIP/2.0/UDP icscf.visited2.net:5060;branch=z9hG4bKnas8866; SIP/2.0/UDP
   actf.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%3gpp-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:actf.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@actf.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-Service: 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.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 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.
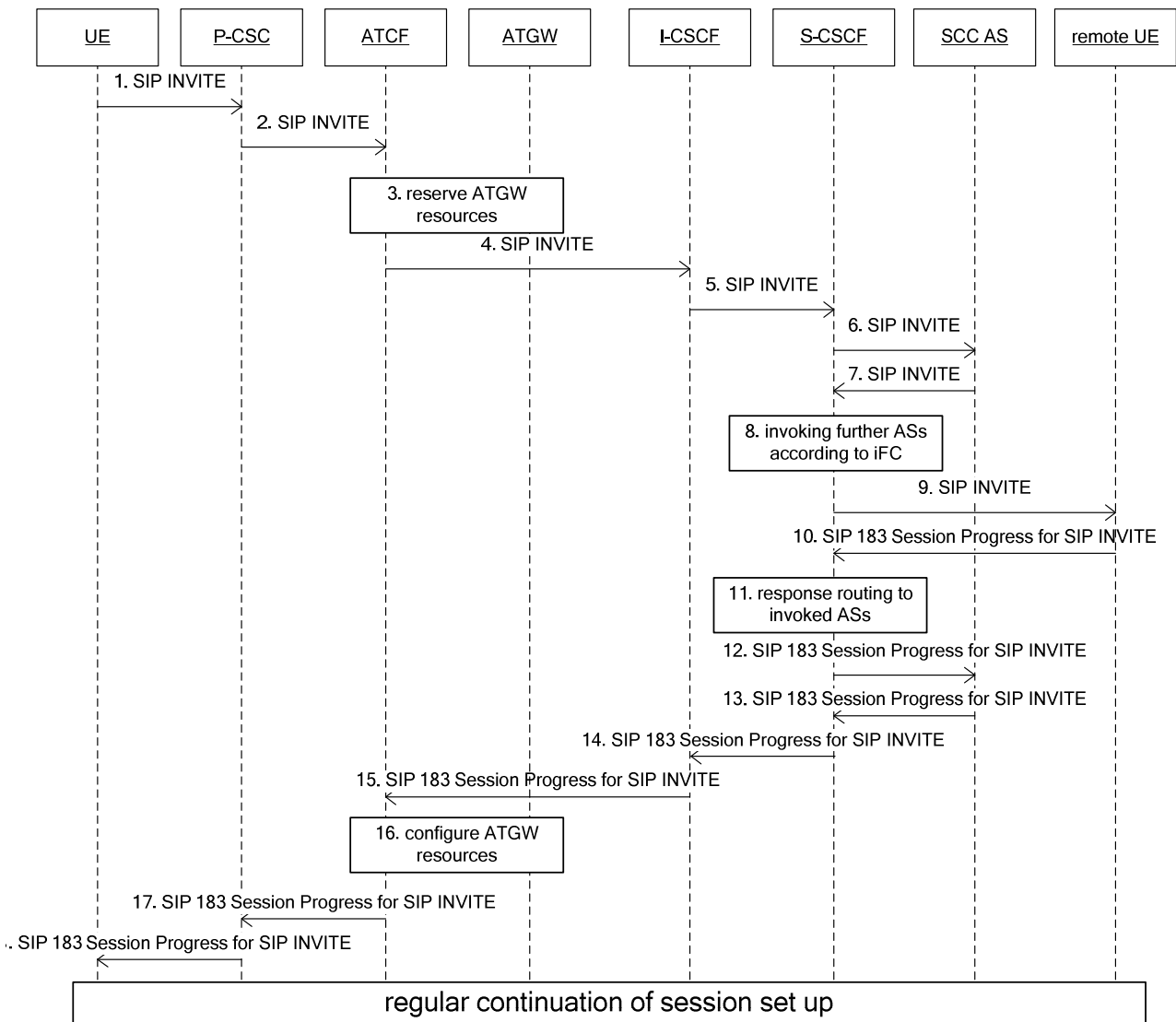
Figure A.4.2-1 Signalling flows for service continuity using SRVCC enhancements

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%3gpp-service.ims.icsi.mmtel"
Accept-Contact: *;+g.3gpp.icsi-ref="urn%3Aurn-7%3gpp-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 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 actf.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
   actf.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%3gpp-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
```

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%3gpp-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.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.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.
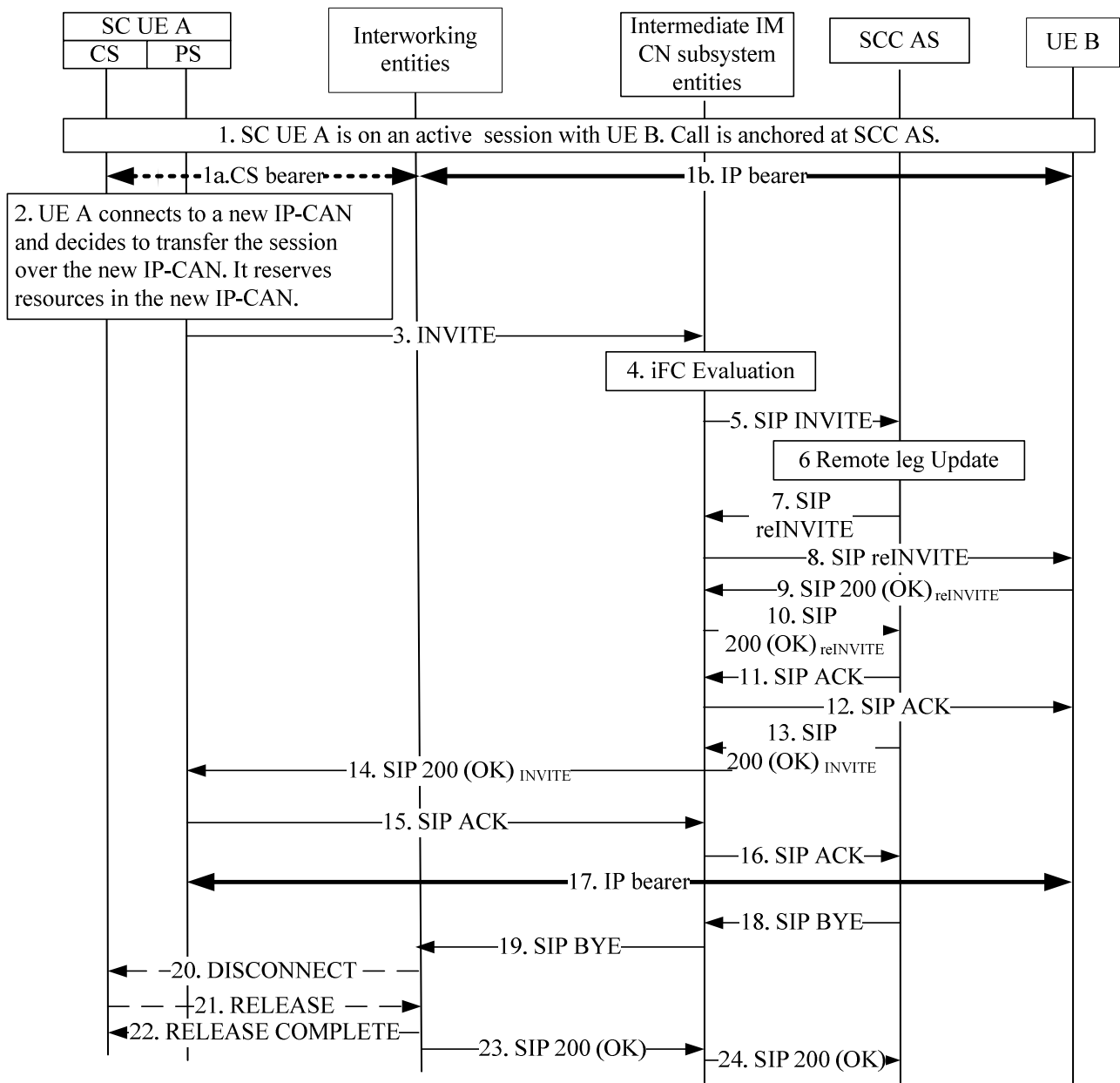
Figure A.6.1-1: Signalling flow for PS-CS Access Transfer: CS to PS

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%3gpp-
   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%3gpp-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 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.
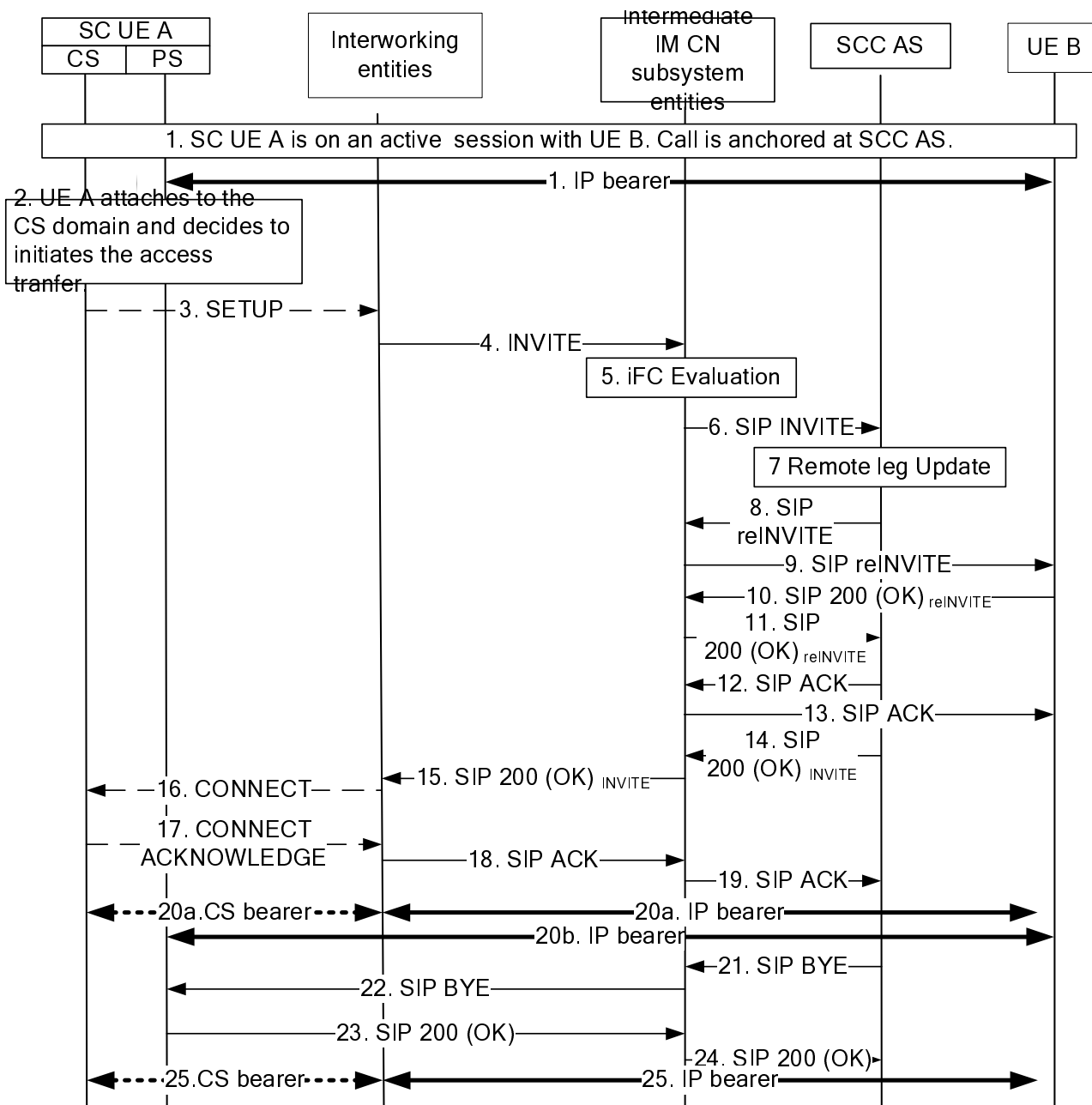
Figure A.6.2-1 Signalling flow for PS-CS access transfer: PS-CS

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 achored 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: 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
Accept-Contact: *;+g.3gpp.icsi-ref="urn%3Aurn-7%3gpp-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%3gpp-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%3gpp-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. **CC CONNECT ACKNOWLEDGE message (SC UE A to interworking entities)**

18-19. **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.

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.
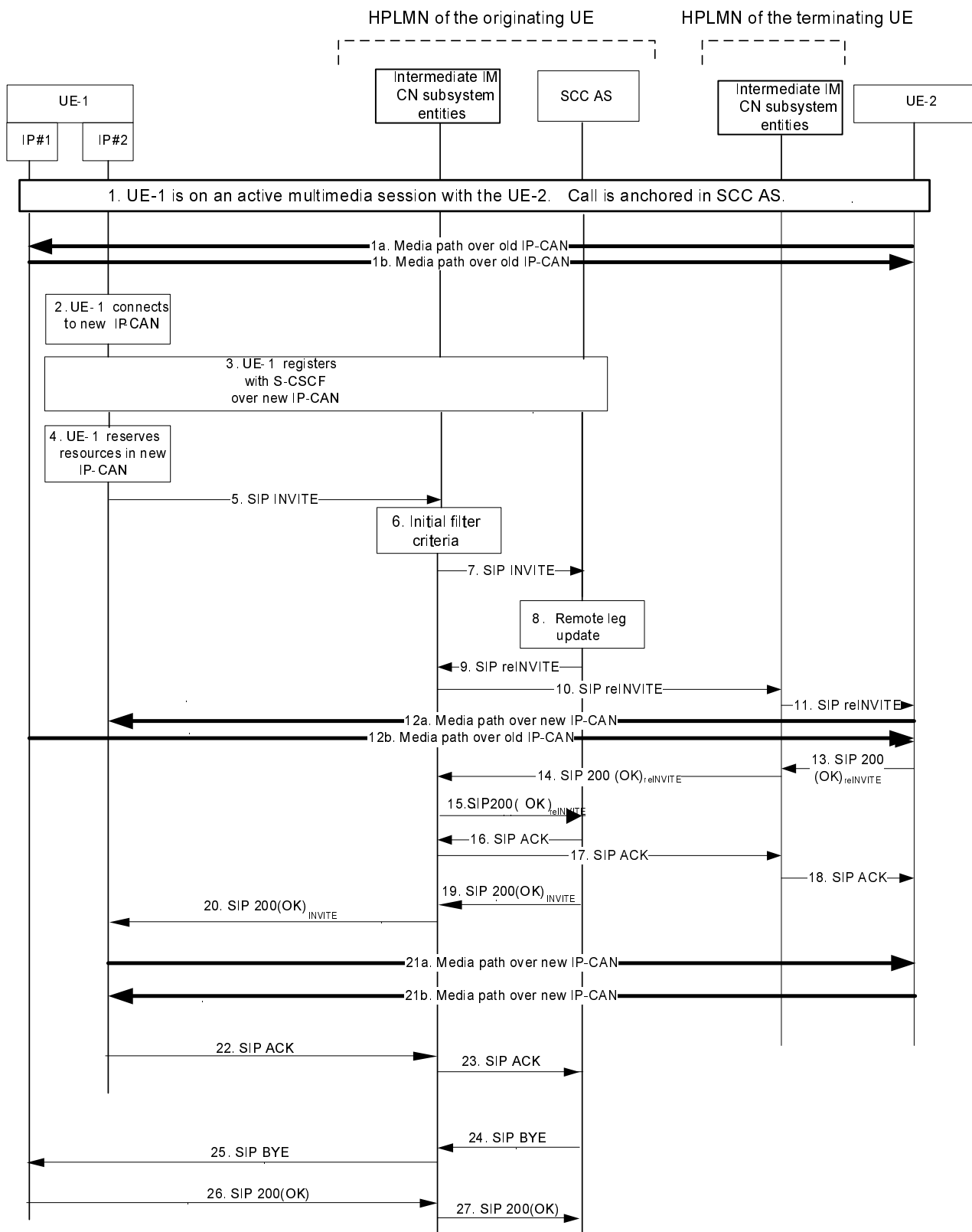
Figure A.7.2-1: Signalling flow for session handover

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 initial SIP INVITE request with 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 tel:+1-212-555-2222 SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Max-Forwards: 70
Route: <sip:pcscf1.home1.net:7531;lr;comp=sigcomp>, <sip:orig@scscf1.home1.net;lr>
P-Preferred-Identity: "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%3gpp-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-2.

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 tel:+1-212-555-2222 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: ccf=[5555::b99:c88:d77:e66]; ccf=[5555::a55:b44:c33:d22];
   ecf=[5555::1ff:2ee:3dd:4ee]; ecf=[5555::6aa:7bb:8cc:9dd]
From: <sip:user1_public1@home1.net>; tag=171828
To: <tel:+1-212-555-2222>
Call-ID:
Cseq:
Supported:
Replaces:
Contact:
Allow:
Accept:
Content-Type:
Content-Length: (…)

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%3gpp-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%3gpp-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.
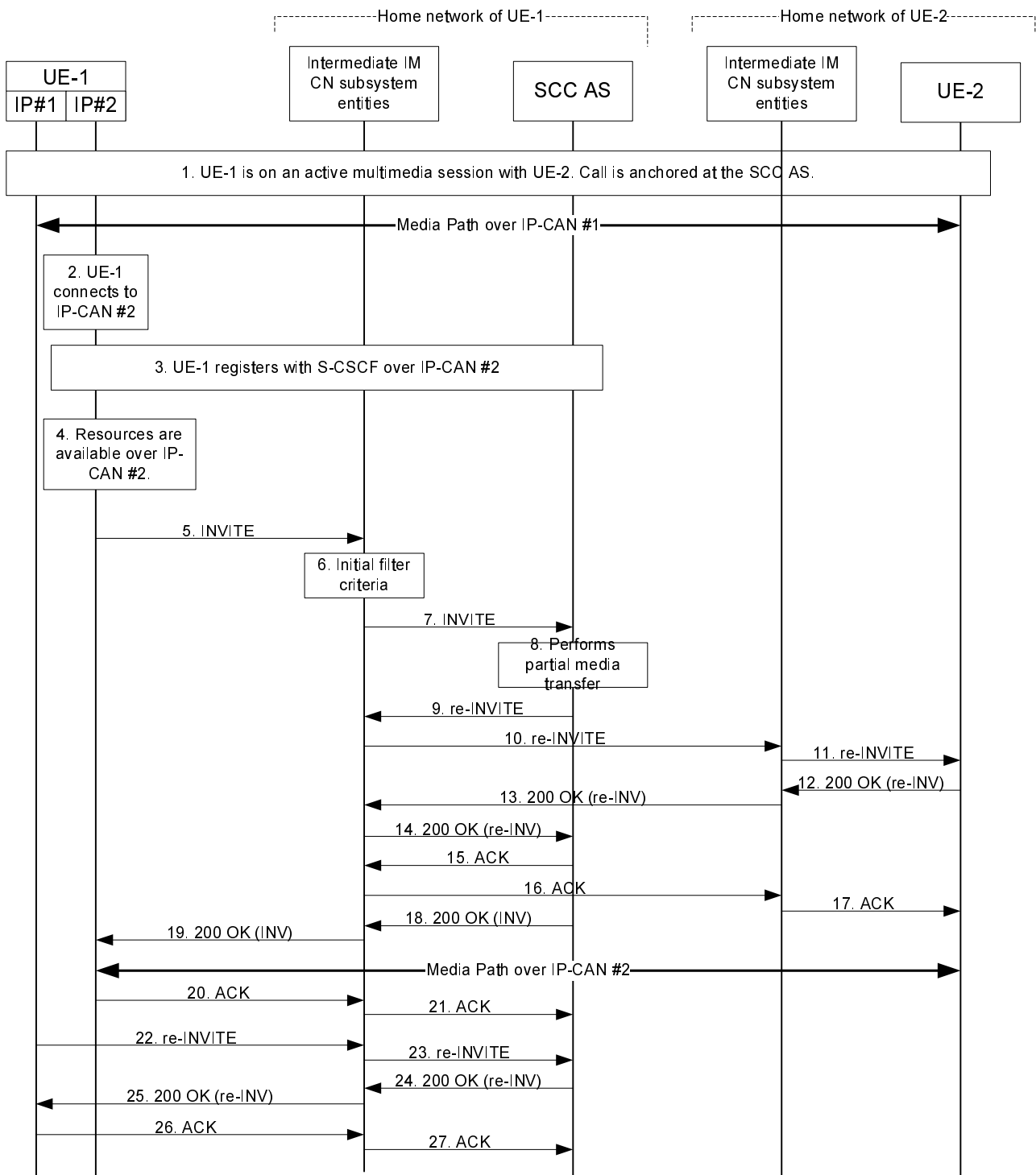
Figure A.7.3-1: Signalling flow for PS-PS session transfer with partial media transfer

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 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: S**IP INVITE request (UE-1 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.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; 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%3gpp-
   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 tel-URI of the destination, i.e. the UE-2.

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%3gpp-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%3gpp-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: <tel:+1-212-555-2222>;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%3gpp-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%3gpp-
   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%3gpp-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.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 tranferred 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.
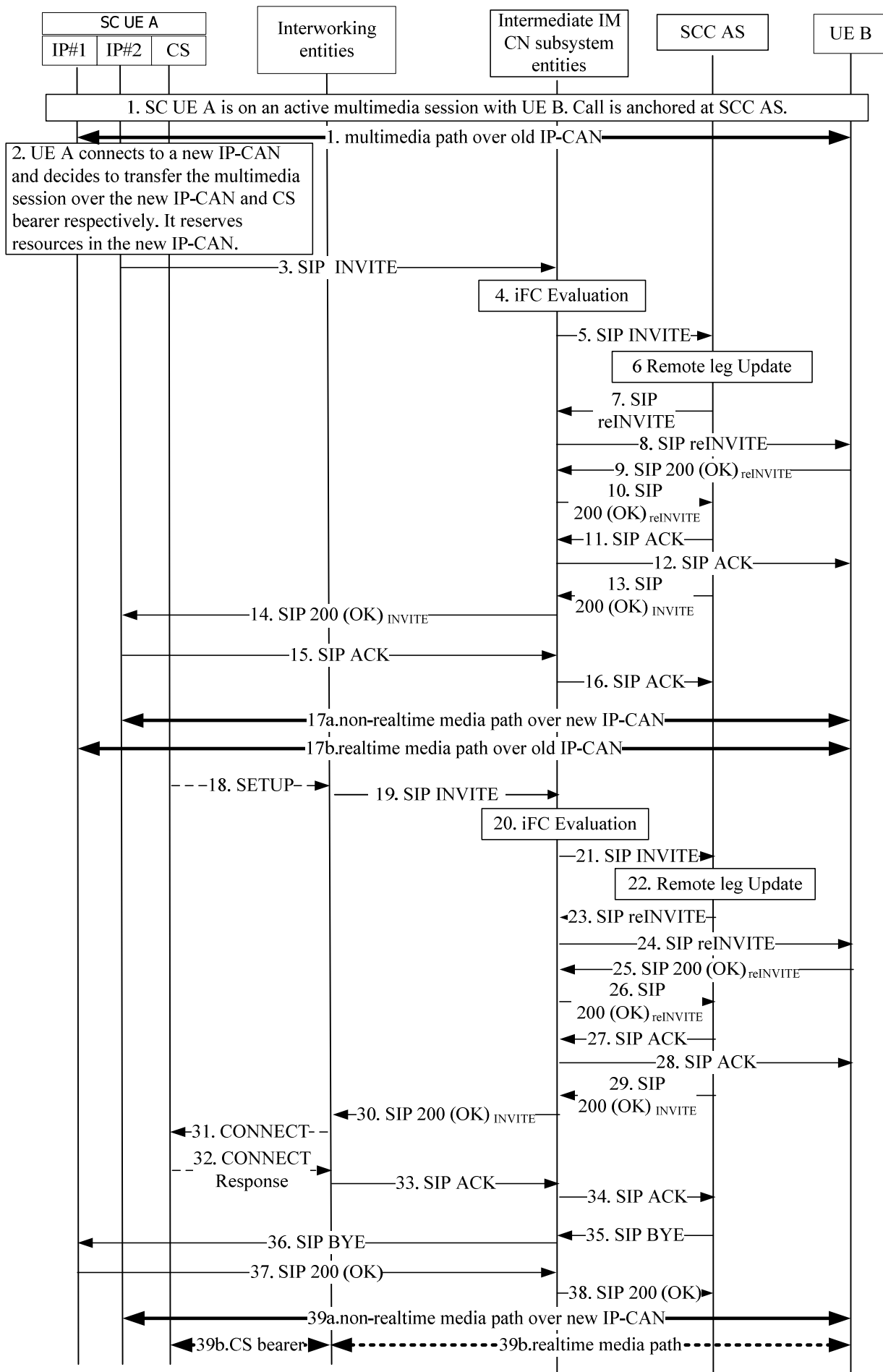
Figure A.8.2-1: Signalling flow for PS - PS in conjunction with PS - CS Access Transfer: PS to CS

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%3gpp-
   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: 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:user1_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-
   00a0c91e6bf6>;+g.3gpp.icsi-ref="urn%3Aurn-7%3gpp-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: icid-value="AyretyU0dm+6O2IrT5tAFrbHLso=023551024"; orig-ioi=home1.net
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%3gpp-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: 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="AyretyU0dm+6O2IrT5tAFrbHLso=023551024"; orig-
    ioi="type3home1.net"
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%3gpp-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. CC CONNECT ACKNOWLEDGEMENT message (SC UE A to interworking entities)

33-34. 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.

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 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 BYE SIP 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.
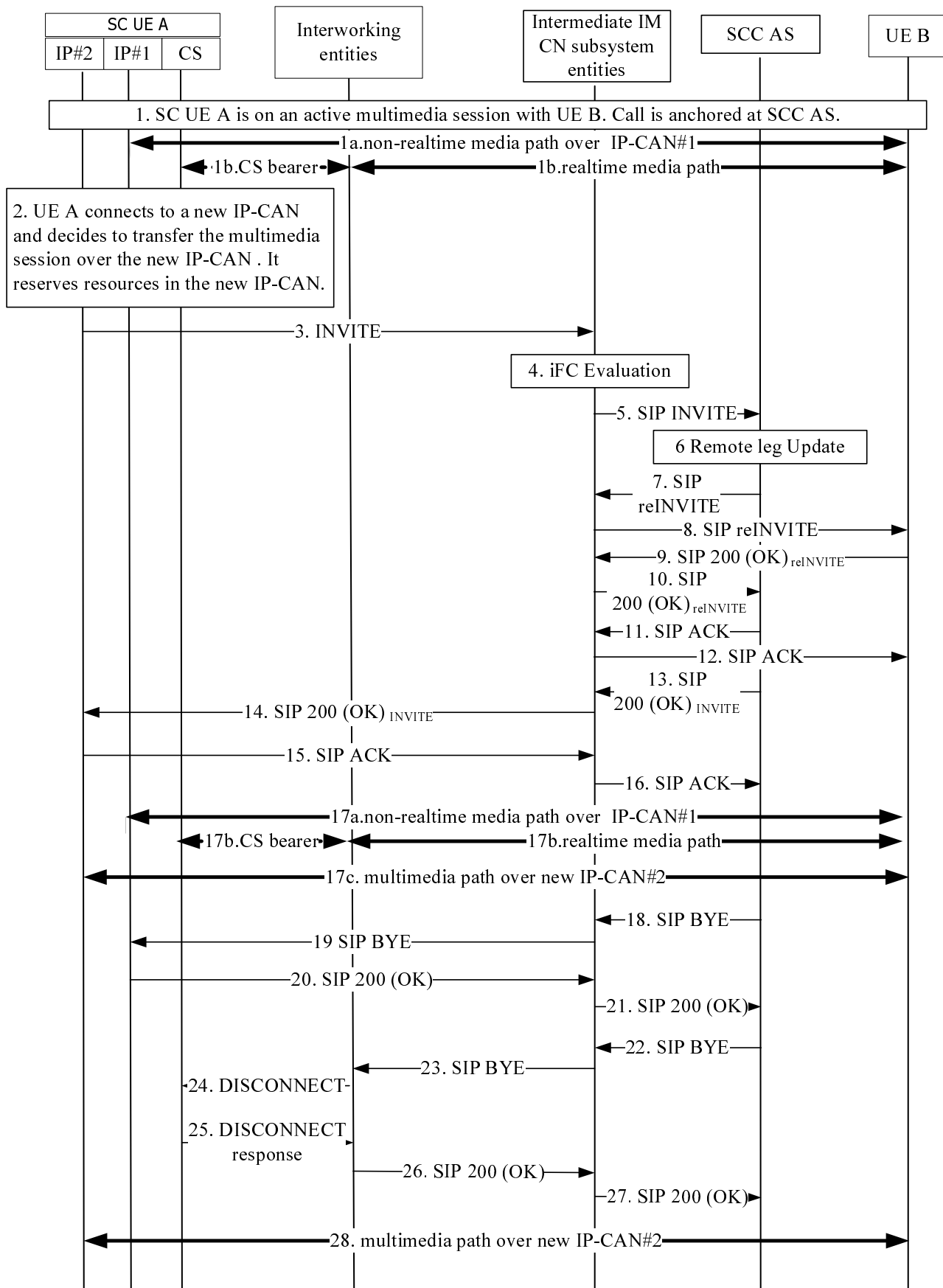
Figure A.8.3-1: Signalling flow for PS - PS in conjunction with PS - CS Access Transfer: CS to PS

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%3gpp-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-xxx%3gpp-
   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%3gpp-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.

24-25. 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.

26-27. **SIP 200 (OK) response (Interworking entities to SCC AS via intermediate IM CN subsystem entities)**

28. 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.
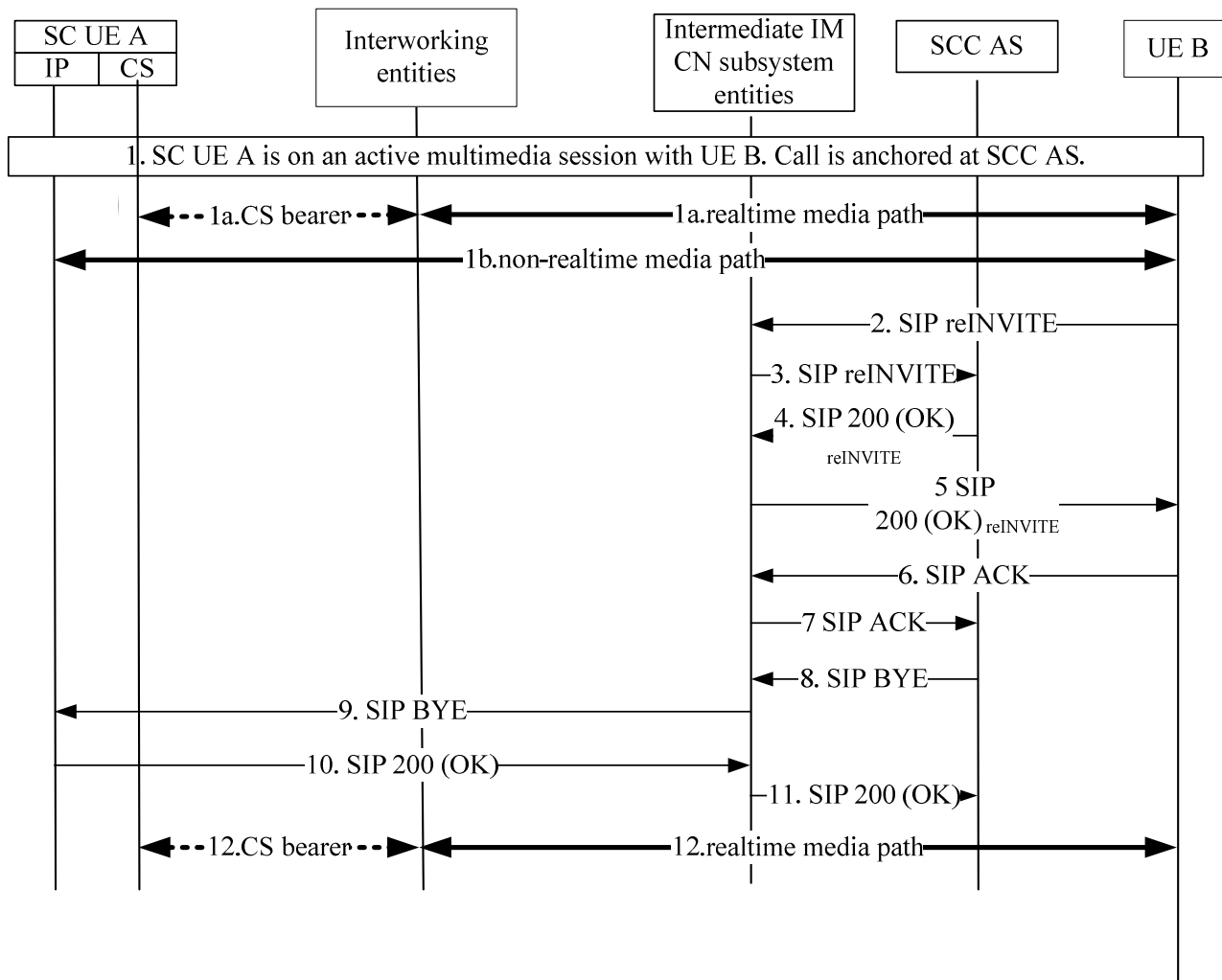
Figure A.9.2-1: Remote End Initiation case – Removing media from split CS and PS sessions

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: 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="AyretyU0dm+6O2IrT5tAFrbHLso=023551024"; orig-
    ioi="type3home1.net"
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%3gpp-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 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%3gpp-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 202 (Accepted) response (UE A to SCC AS via intermediate IM CN subsystem entities)**

   Upon receiving the SIP REFER request, the UE A sends a SIP 202 (Accepted) 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%3Agpp-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.

46-48. **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.

49-50. **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.
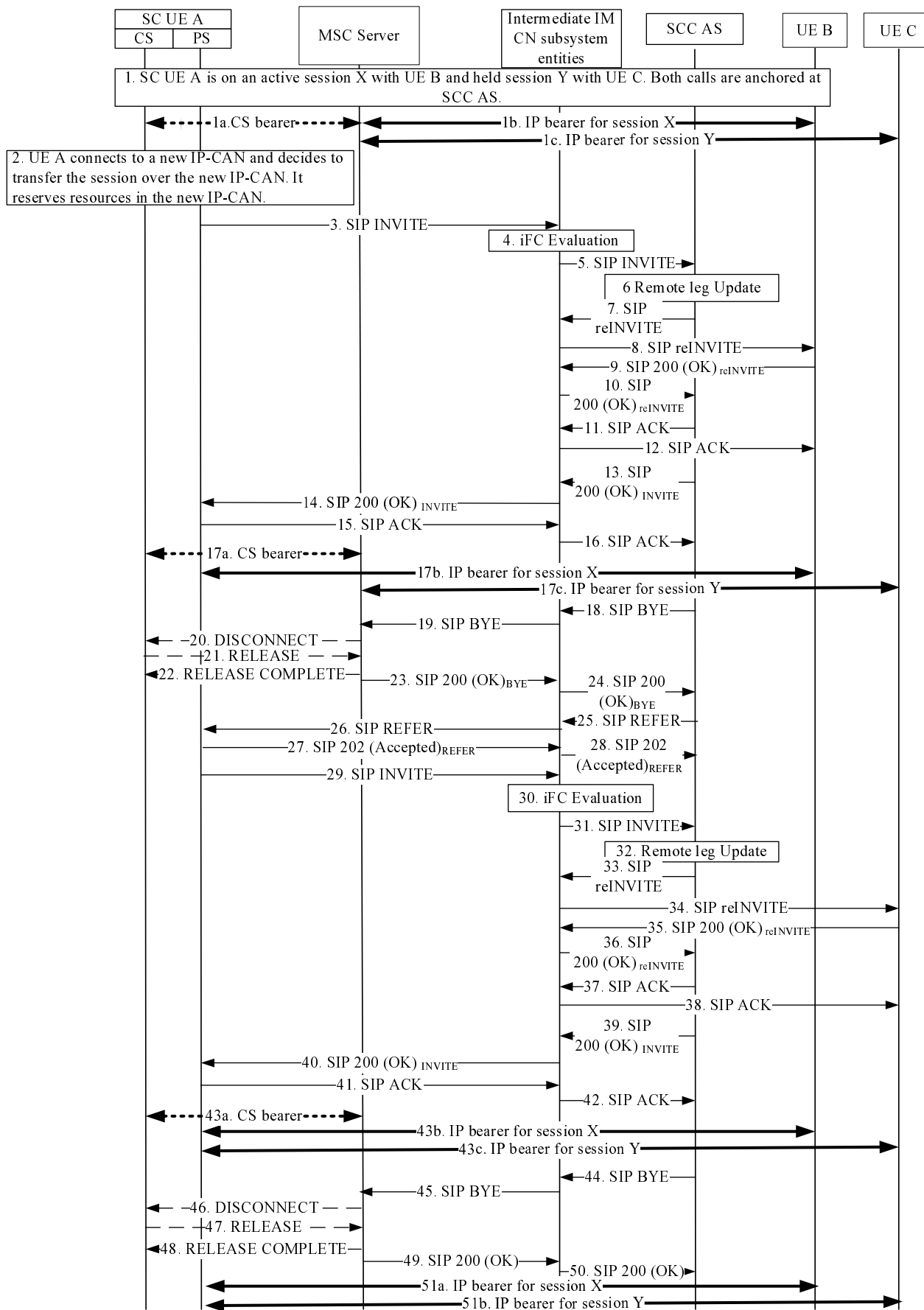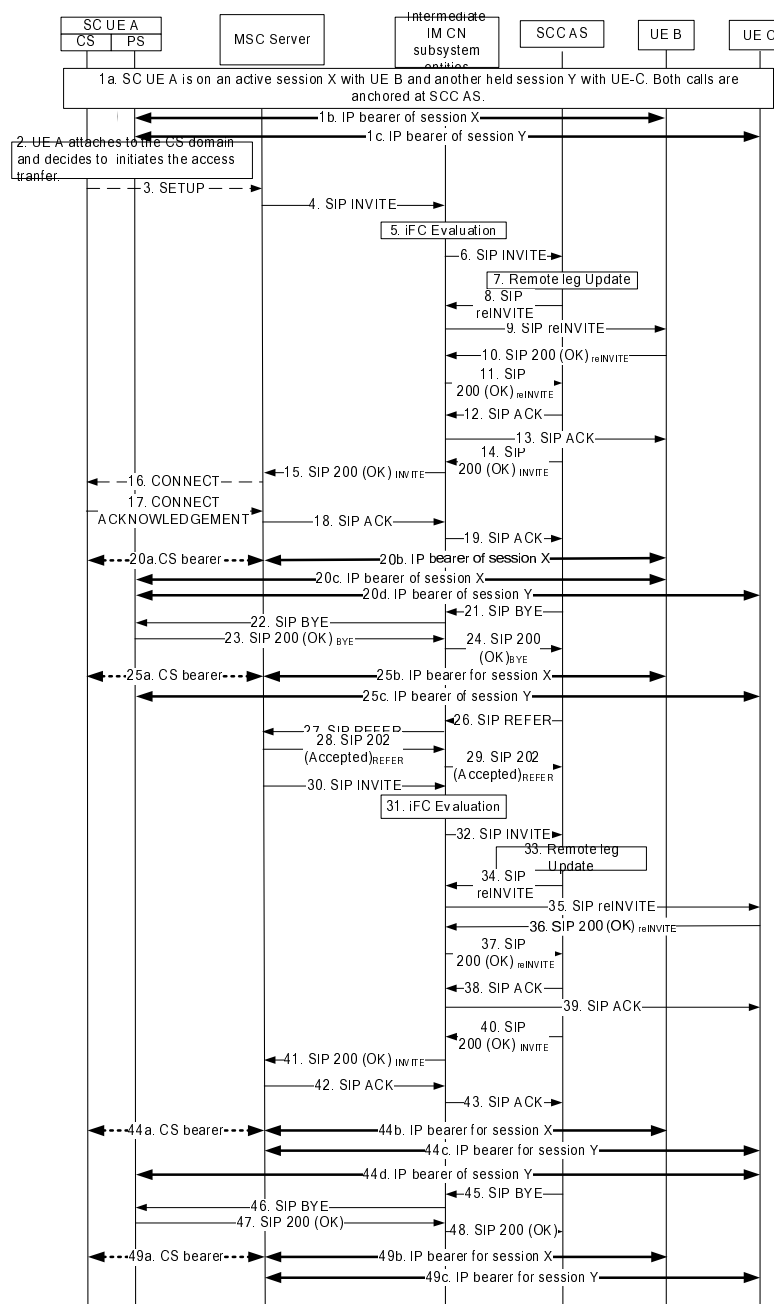
Figure A.15.3-1: Signalling flow for PS-CS access transfer: PS-CS

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%3gpp-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%3gpp-
   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. **CC CONNECT ACKNOWLEDGEMENT message (SC UE A to MSC Server)**

18-19. **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.

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 202 (Accepted) response (MSC Server to SCC AS via intermediate IM CN subsystem entities)

Upon receiving the SIP REFER request, the MSC Server sends a SIP 202 (Accepted) 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%3gpp-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%3gpp-
   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 state. 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 SRVCC procedure to CS access.
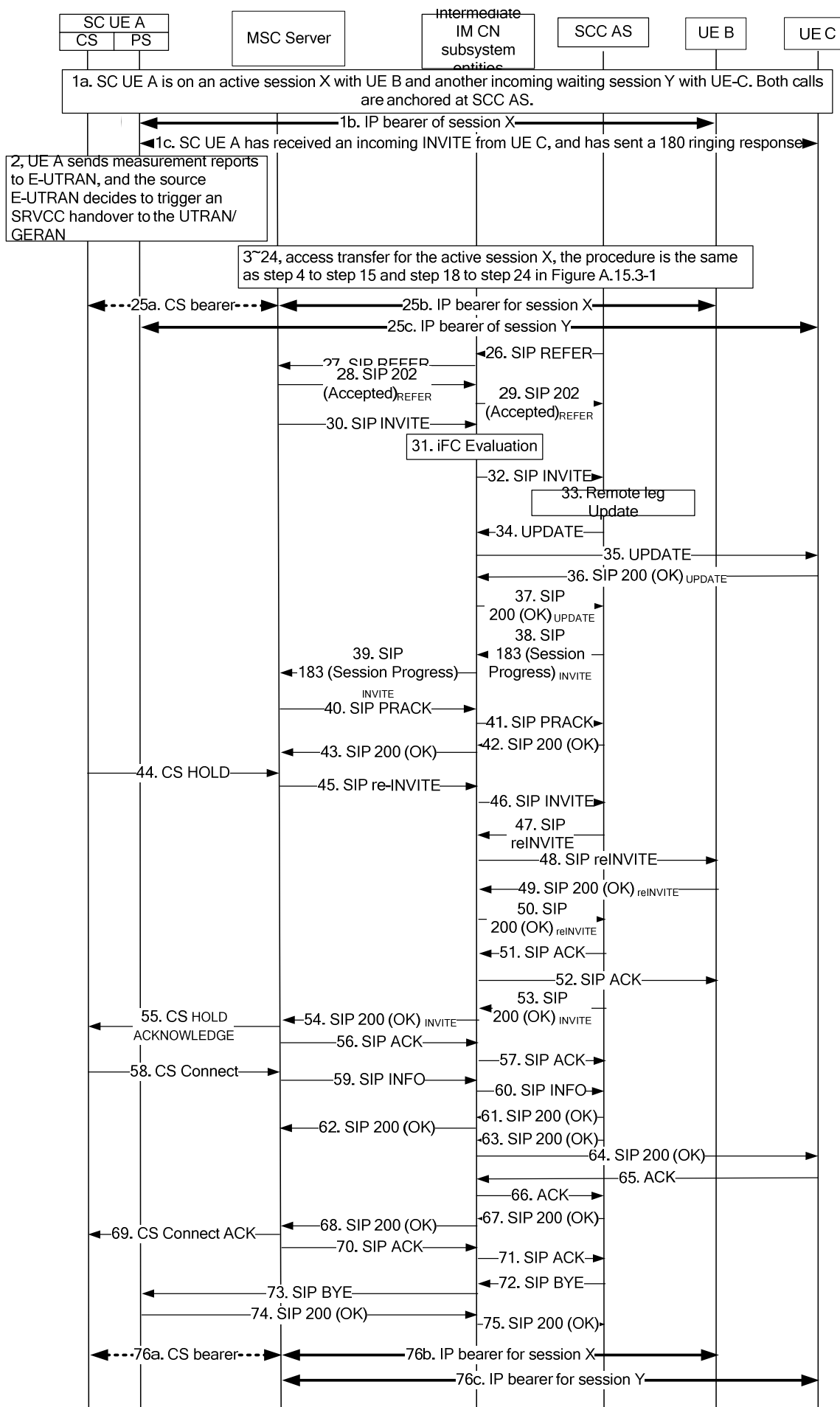
Figure A.15.4-1: Signalling flow for PS to CS access transfer with MSC server assisted mid-call feature with an incoming waiting call in alerting phase

NOTE 1: For clarity, the SIP 100 (Trying) responses are not shown in the signalling flow.

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 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: 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.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>

--boundary--
```

> **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 state.

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 202 (Accepted) response (MSC server to SCC AS via intermediate IM CN subsystem entities)**

Upon receiving the SIP REFER request, the MSC server sends a SIP 202 (Accepted) 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 state, 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%3gpp-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%3gpp-
    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 PRACK with a SIP 200 (OK) response to the MSC server.

44. CS 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 200 (OK) to indicate the succesfull activity to the MSC server that put session X on hold.

55. CS 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 200 OK received from SCC AS.

58. **CS Connect Message from SC UE A to MSC server**

The SC UE A accepts the call and sends CS Connect Message.

59. **SIP INFO request (MSC server to intermediate IM CN subsystem entities) - see example in table A.15.4-59**

A.15.4-59: 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-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.

60. **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.

61. **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

62. **SIP 200 (OK) response (Intermediate IM CN subsystem entities to MSC server)**

The SIP 200 (OK)response is forwarded to the MSC server.

63. **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.

64. **SIP 200 (OK) response (Intermediate IM CN subsystem entities to far end)**

The SIP 200 (OK) response is forwarded to the far end)

65. **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

66. **SIP ACK request (Intermediate IM CN subsystem entities to SCC AS)**

The SIP ACK request is forwarded to the SCC AS.

67. **SIP 200 (OK) response (SCC AS to intermediate IM CN subsystem entities)**

The SCC AS sends 200 (OK) response to indicate the succesfull access transfer to the MSC server.

68. **SIP 200 (OK) response (Intermdeiat IM CN subsystem entities to far end)**

The SIP 200 (OK) response is forwarded to the MSC server.

69. CS Connect Ack (MSC server to SC UE A)

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: 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 SRVCC session transfer for IMS emergency session

A.16.1 Introduction

The signalling flows for SRVCC session transfer for IMS emergency session demonstrate how an IMS emergency session is transferred from PS network to CS network using 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 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.
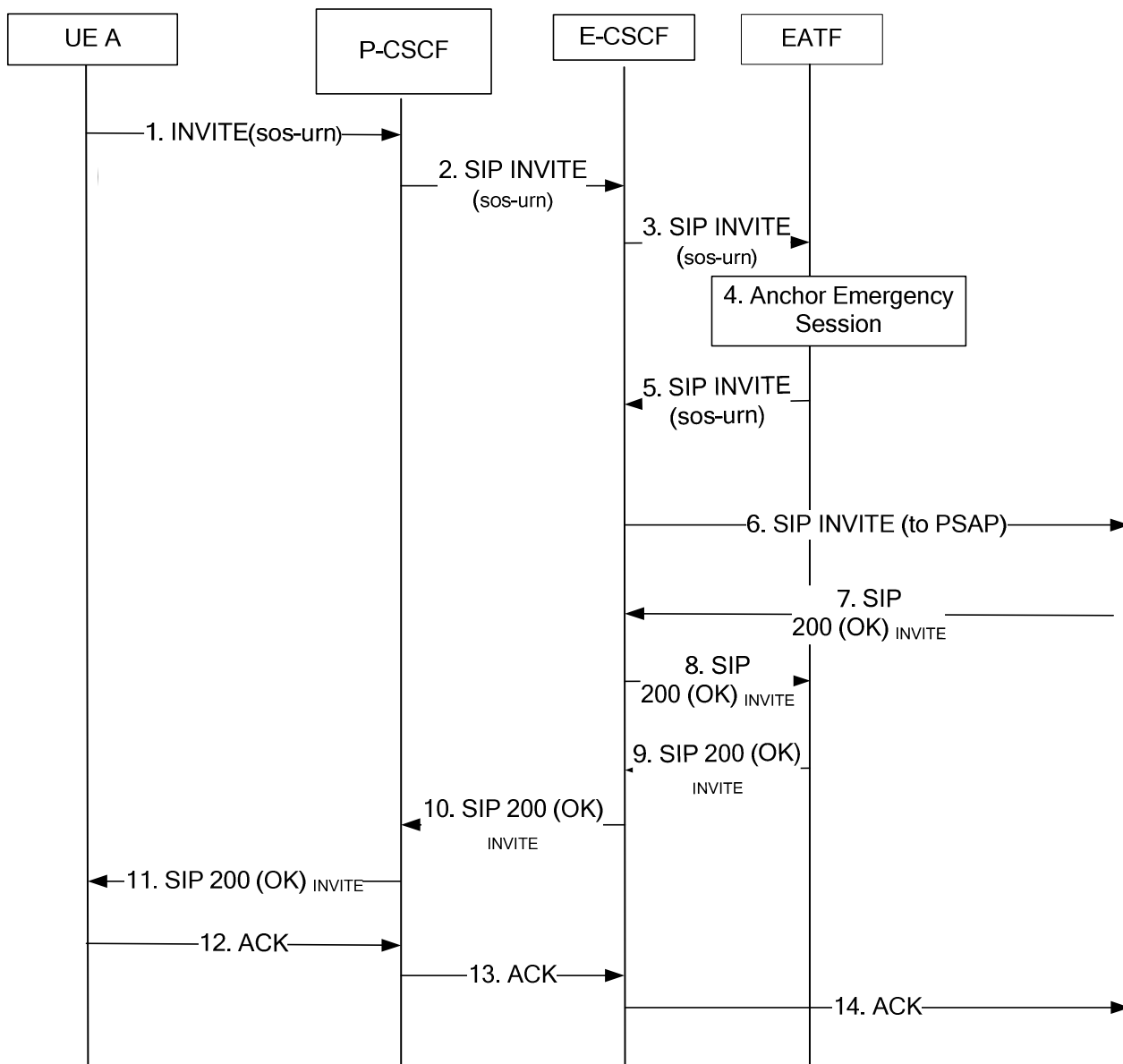
Figure A.16.2-1: Signalling flow for UE initiating an emergency session in IMS

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 (EATF 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

```
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
```

Table A.16.2-4: SIP INVITE request

```
INVITE urn:service:sos.fire SIP/2.0
Via: SIP/2.0/UDP esccas.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:esccas.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 esccas.visit1.net;branch=z9hG4bKnas34r5
Max-Forwards: 67
Route: <sip:ecscf.visit1.net:7531;lr;comp=sigcomp>
Record-Route: <sip:ecscf.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

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP ecscf.visit1.net;branch=z9hG4bKnas34r5
Max-Forwards: 67
Record-Route: <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 200 (OK) response with a SIP ACK request.

A.16.3 Session transfer for emergency session using 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 SRVCC handover to CS access.
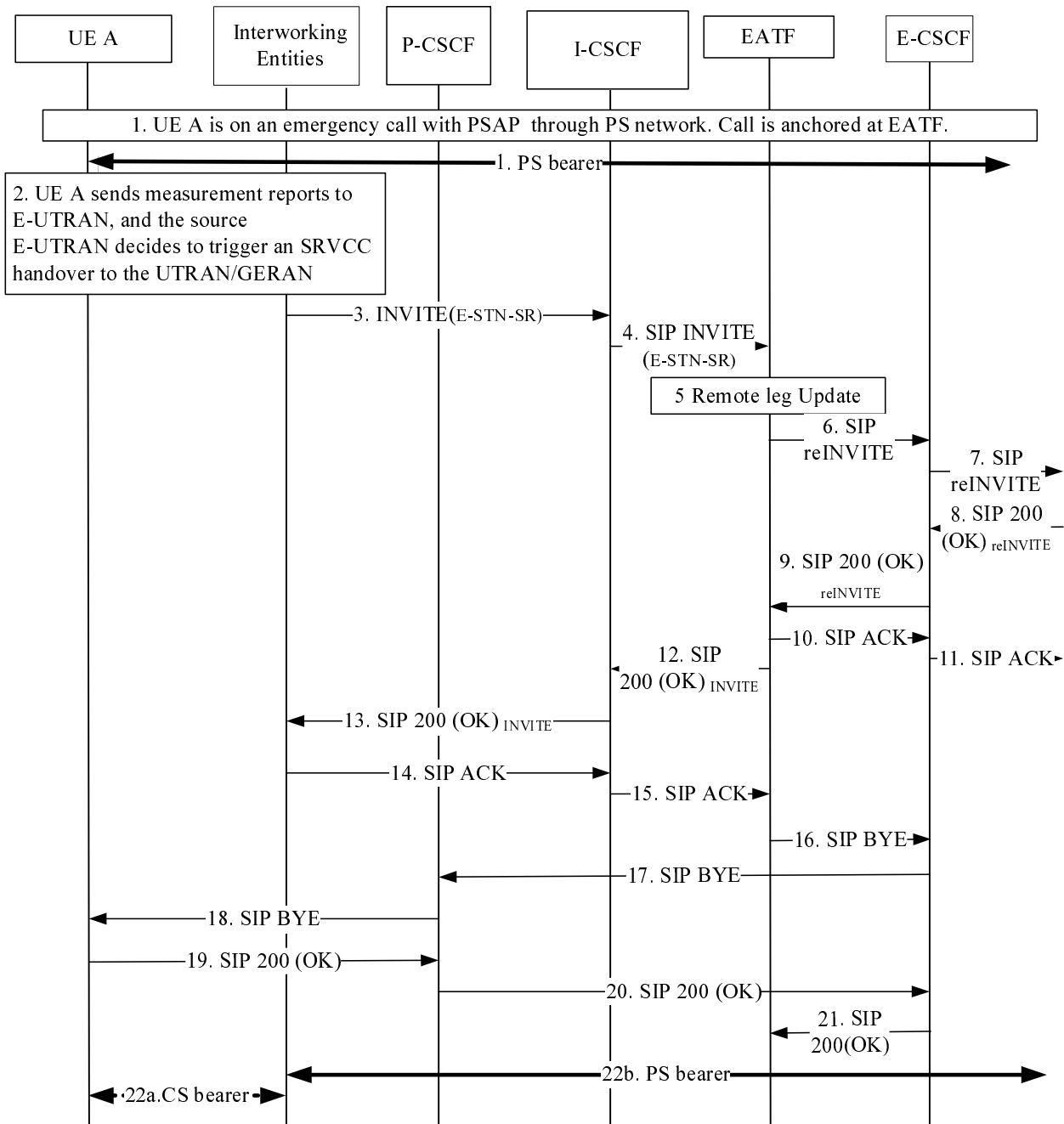
Figure A.16.3-1 Signalling flow for emergency session transfer using SRVCC procedure

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 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: 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%3gpp-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%3gpp-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 (SCC AS to intermediate IM CN subsystem entities)

```
INVITE urn:service:sos.fire SIP/2.0
Via: SIP/2.0/UDP esccas1.home1.net;branch=z9hG4bKnas34r5
```

```
Max-Forwards: 68
Route: <sip:ecscf1.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: <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 SRVCC in Alerting State

A.17.1 Introduction

The signalling flows in the subclause demonstrate how sessions in alerting state can be transferred from PS to CS using 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 SRVCC procedure: PS to CS

In the example flow at the figure A.15.2-1, 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 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 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. 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
Accept-Contact: *;+g.3gpp.icsi-ref="urn%3Aurn-7%3gpp-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%3gpp-service.ims.icsi.mmtel"
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER
Recv-Info: g.3gpp.state-and-event-info
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 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 183 (Session Progress) containing the SDP answer as received from the far end UE. The SDP answer indicates that resources are available. The SIP 183 (Session Progress) will contain a Recv-Info header field set to g.3gpp.state-and-event-info.

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) to the MSC server.

11. **SIP PRACK request (MSC server to Intermediate IM CN subsystem entities)**

    The MSC acknowledges the receipt of the 183 Session Progress..

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 PRACK

14. **SIP 200 (OK) response (Intermediate IM CN subsystem entities to MSC server)**

    The intermediate IM CN subsystem entities forwards the SIP 200 (OK) reponse 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-info
Content-Type: application/vnd.3gpp.state-and-event-info+xmls
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 state.

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 forwards 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 INFO request.

20a. **User answers the call**

20. **CS Connect Message from SC UE A to MSC server**

The SC UE A accepts the call and sends CS Connect Message.

21. **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-info
Content-Type: application/vnd.3gpp.state-and-event-info+xmls
Content-Length:

<?xml version="1.0" encoding="UTF-8"?>
<state-and-event-info>
   <event>call-accepted</event>
</state-and-event-info>
```

22. **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.

23 **SIP 200 (OK)   response (SCC AS to intermediate IM CN subsystem entities)**

The SCC AS acknowledges the receipt of the INFO request indicating that the SC UE A has accepted the call

24 **SIP 200 (OK) response (Intermediate IM CN subsystem entities to MSC server)**

The SIP 200 (OK) response is forwarded to the MSC server.

25 **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.

26 **SIP 200 (OK) response (Intermediate IM CN subsystem entities to far end)**

The SIP 200 (OK) response is forwarded to the far end)

27 **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

28 SIP ACK request (Intermediate IM CN subsystem entities to SCC AS)

The SIP ACK request is forwarded to the SCC AS.

29 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.

30 SIP 200 (OK) response (Intermediate IM CN subsystem entities to far end)

The SIP 200 (OK) response is forwarded to the MSC server.

31 CS Connect Ack (MSC server to SC UE A)

32 SIP ACK request (MSC server to intermediate IM CN subsystem entities)

MSC server acknowledges the 200 OK 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 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 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 SRVCC handover to CS access.
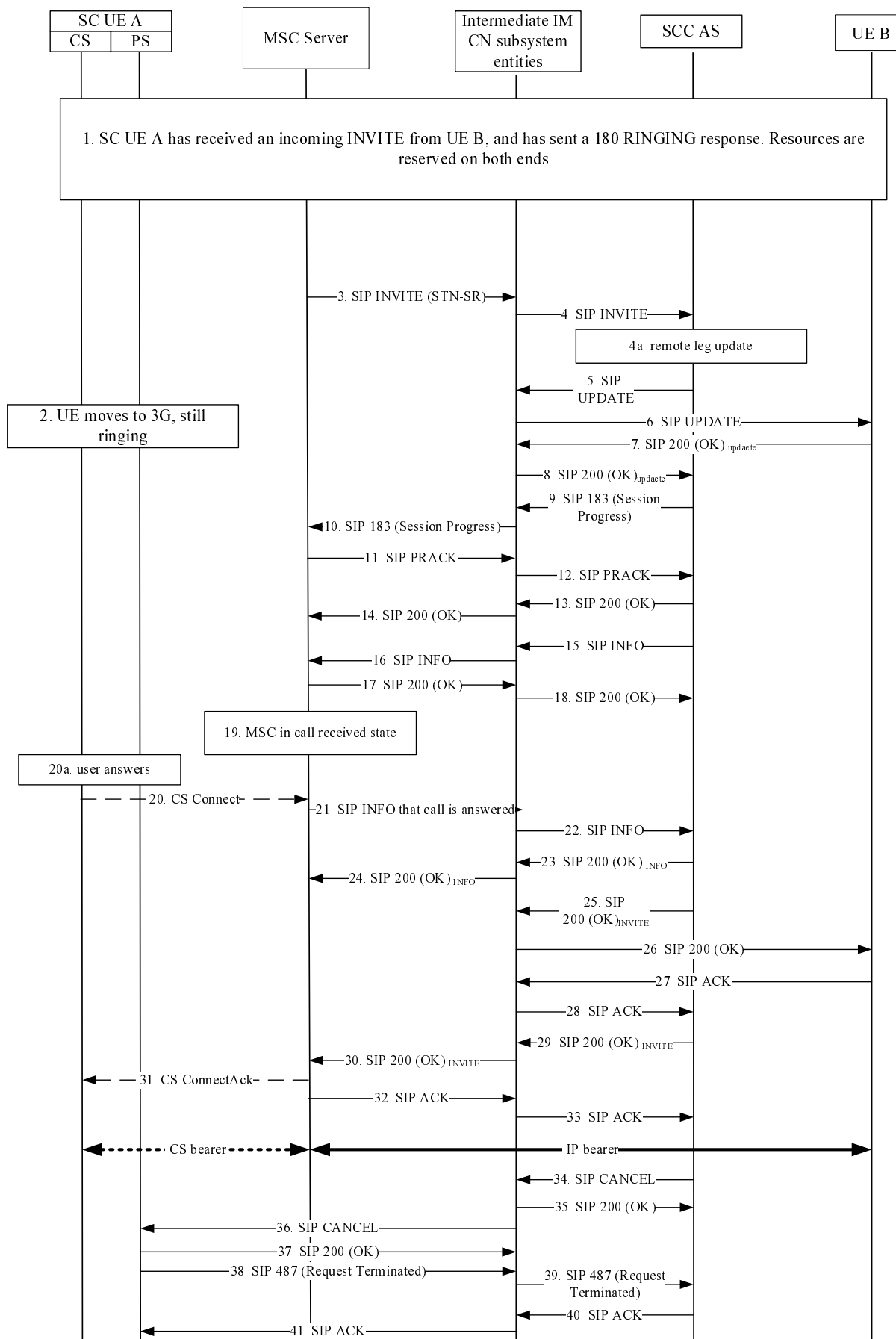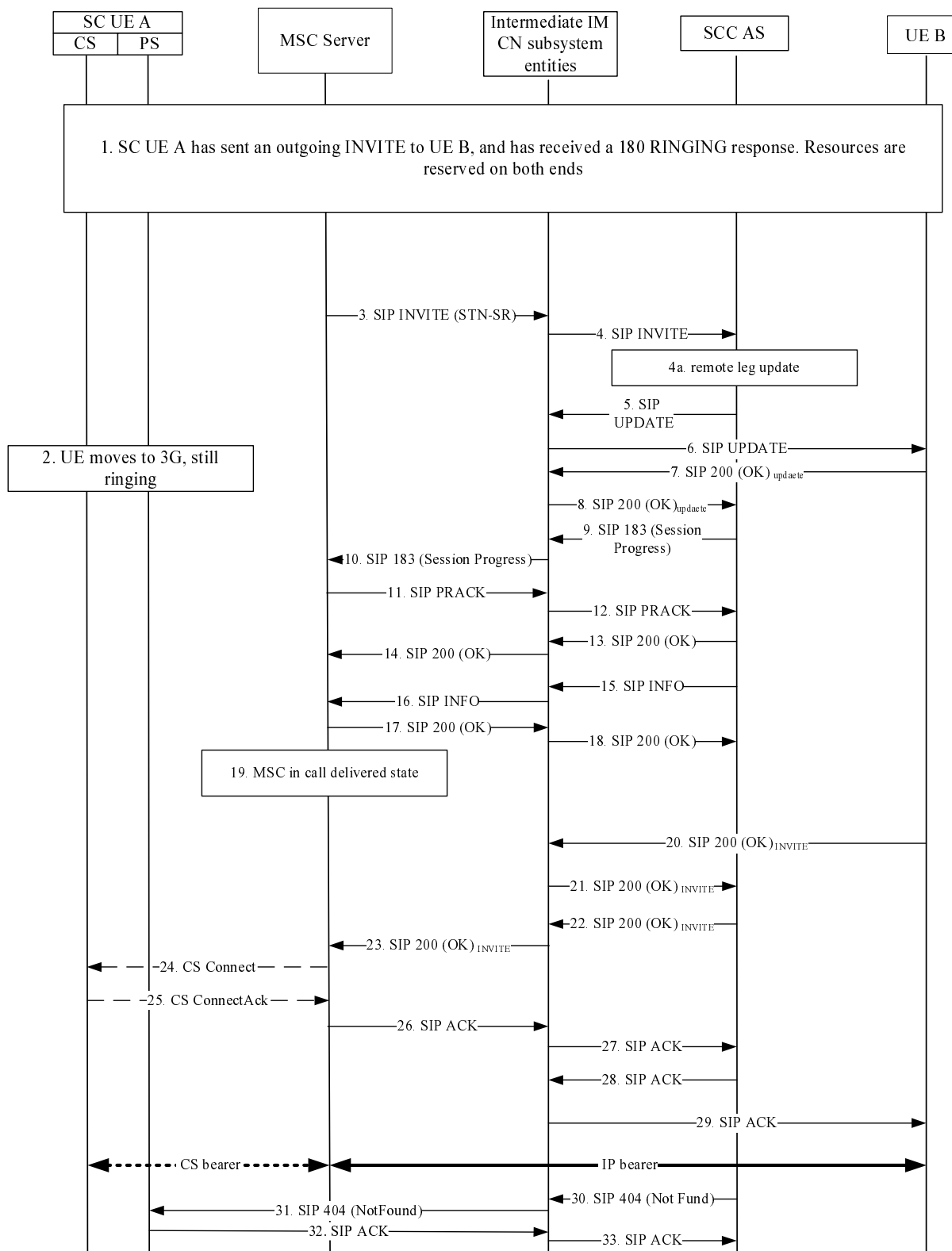
Figure A.17.3-1: PS-CS SRVCC, incoming call in alerting phase

NOTE 1: For clarity, the SIP 100 (Trying) responses are not shown in the signalling flow.

1. SC UE A has 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.

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. 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.17.3-1**

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
Accept-Contact: *;+g.3gpp.icsi-ref="urn%3Aurn-7%3gpp-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%3gpp-service.ims.icsi.mmtel"
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER
Recv-Info: g.3gpp.state-and-event-info
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 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 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 200 OK.

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) 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) to the MSC server.

11. SIP PRACK request (MSC server to Intermediate IM CN subsystem entities)

The MSC acknowledges the receipt of the 183 Session Progress.

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 PRACK

14. SIP 200 (OK) response (Intermediate IM CN subsystem entities to MSC server)

The intermediate IM CN subsystem entities forward the SIP 200 (OK) reponse 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-info
Content-Type: application/ vnd.3gpp.state-and-event-info+xmls
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 state.

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 forwards 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 200 (OK) response.

21. 200 (OK) response (Intermediate IM CN subsystem entities to SCC AS)

The 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 200 (OK) response to indicate that the terminating UE B has accepted the call.

23 200 (OK) response (Intermediate IM CN subsystem entities to MSC server)

The 200 (OK) response is forwarded to the MSC server.

24 CS CONNECT (MSC server to SC UE A)

The MSC server indicates to the SC UA A that the far end has accepted the call.

25 CS CONNECTACK (MSC server to SC UE A)

SC UE A acknowledges the CS CONNECT.

26 SIP ACK request (MSC server to intermediate IM CN subsystem entities)

The MSC server acknowledges the SIP 200 (OK) response   received from SCC AS

27. SIP ACK request (Intermediate IM CN subsystem entities to SCC AS)

The SIP ACK request is forwarded to the SCC AS.

28 SIP ACK request (SCC AS to intermediate IM CN subsystem entities)

The SCC AS acknowledges the SIP 200 (OK) response received towards far end..

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 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.
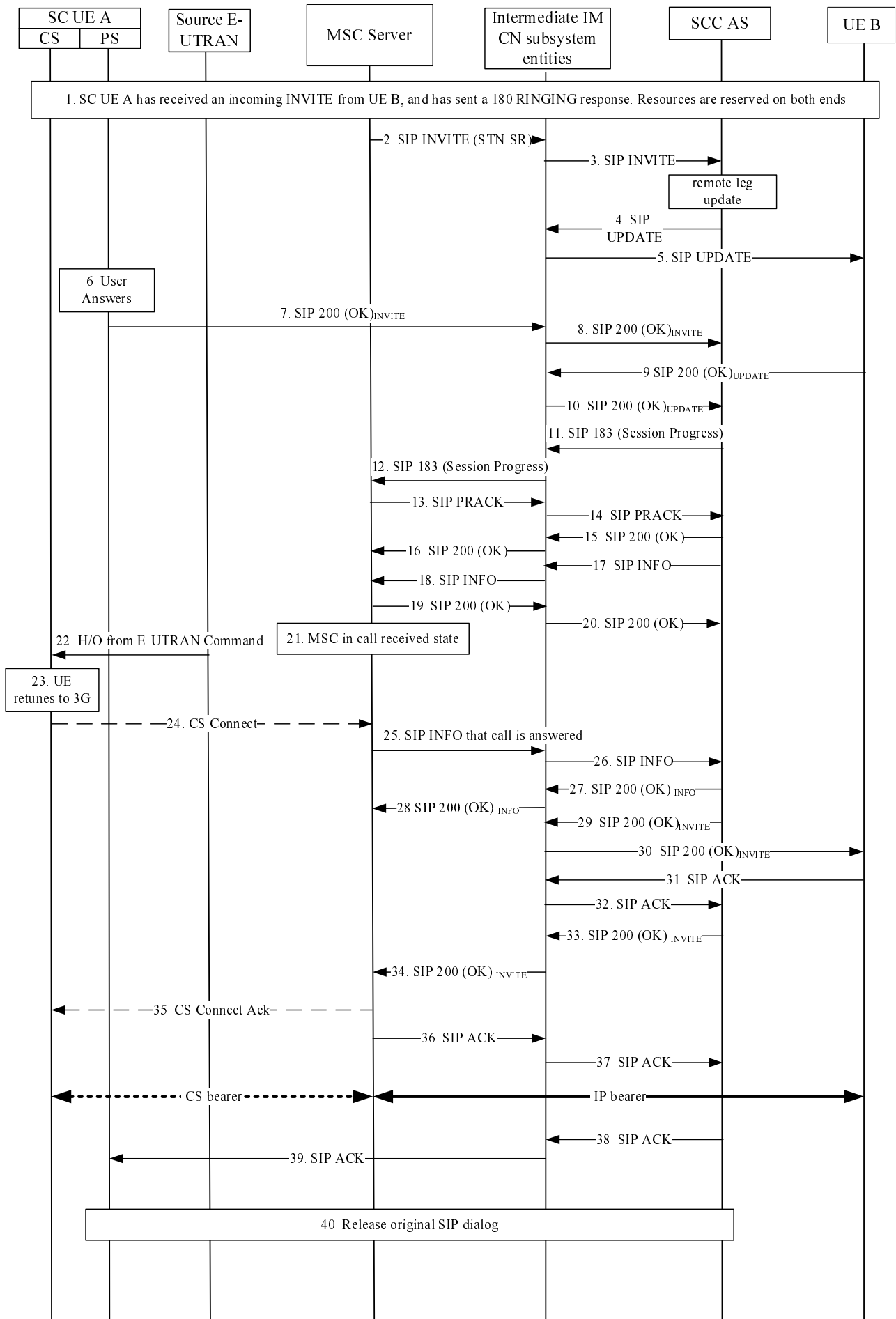
Figure A.17.4-1: SIP 200 OK from SC UE received by SCC AS: Handover to CS successful

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 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 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   CS Connect Message from SC UE A to MSC server

The SC UE A sends CS 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 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 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.
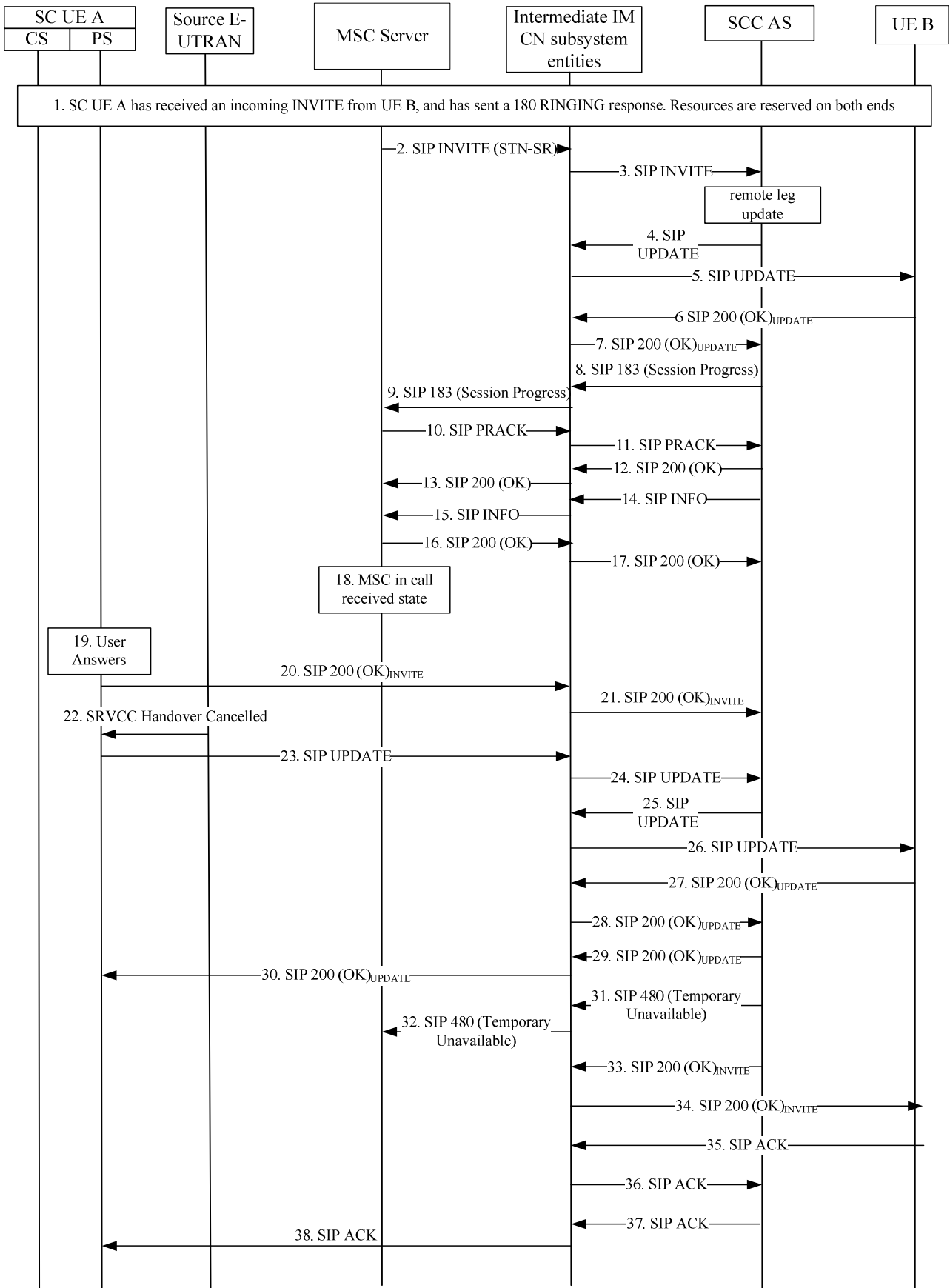
Figure A.17.5-1: SIP 200 OK from SC UE received by SCC AS: Handover cancelled

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 180 (Ringing) response.

2-18   Continuation of procedure for 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 SRVCC in Alerting Phase

These steps are identical to steps 7-19 in subclause A.17.2.

22   SC UE A receives 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 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 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 SRVCC handover to CS access.
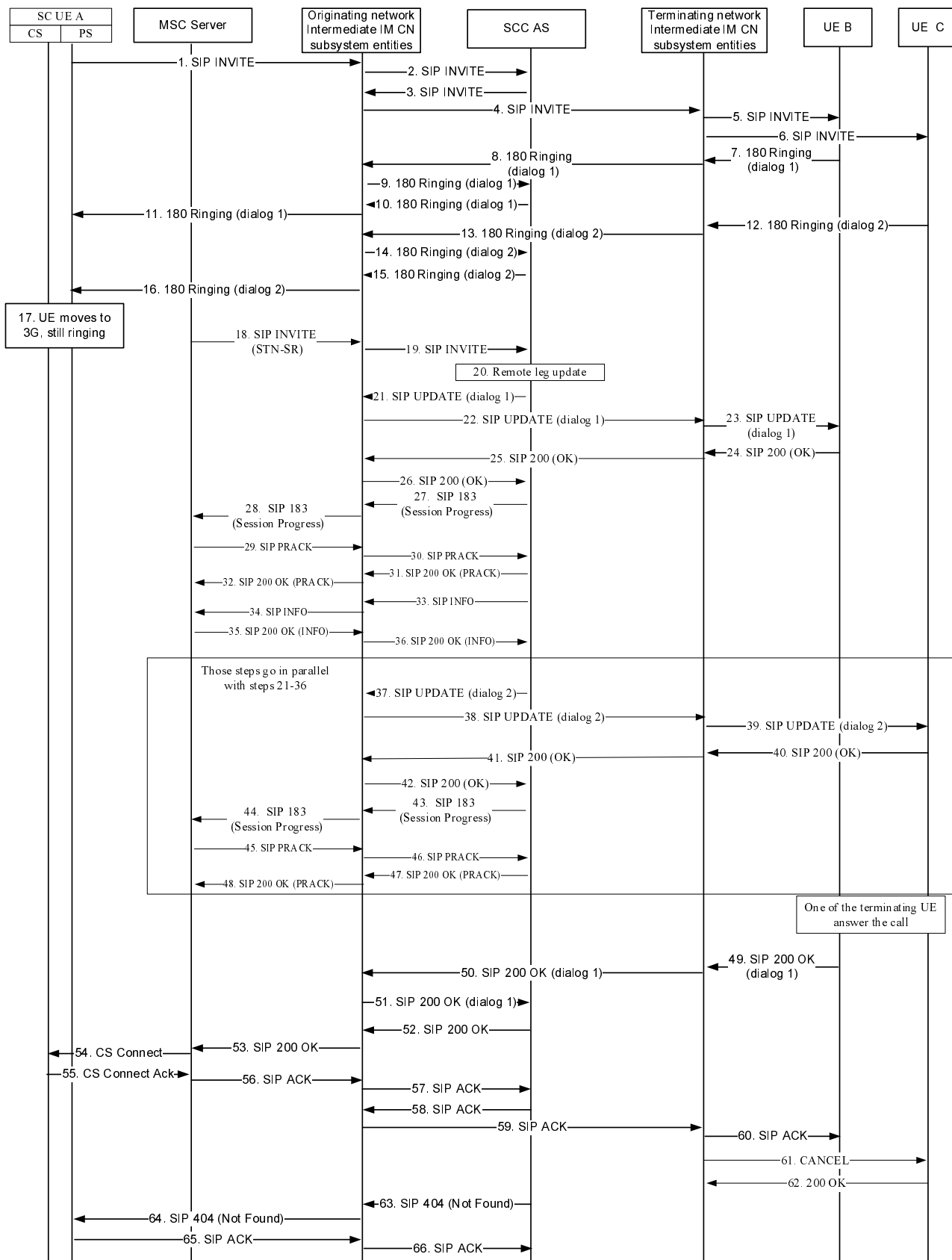
Figure A.17.6-1: PS-CS SRVCC, incoming call in alerting phase with forked responses

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
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%3gpp-service.ims.icsi.mmtel"
Accept-Contact: *;+g.3gpp.icsi-ref="urn%3Aurn-7%3gpp-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%3gpp-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%3gpp-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 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
Accept-Contact: *;+g.3gpp.icsi-ref="urn%3Aurn-7%3gpp-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%3gpp-service.ims.icsi.mmtel"
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER
Recv-Info: g.3gpp.state-and-event-info
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.

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 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 PRACK with the SIP 200 (OK) reponse to the MSC server.

33. **SIP INFO request (SCC AS to Originating network 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-info
Content-Type: application/ vnd.3gpp.state-and-event-info+xmls
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 state.

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 200 OK.

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 PRACK with the SIP 200 (OK) reponse 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 200 (OK) response.

50-51. 200 (OK) response (Intermediate IM CN subsystem entities to SCC AS)

The 200 (OK) response is forwarded to SCC AS.

52-53 200 (OK) response (SCC AS to MSC server through Intermediate IM CN subsystem entities)

The 200 (OK) response is forwarded to the MSC server based on the route established during step 24-28.

54 CS 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.

55 CS CONNECTACK (MSC server to SC UE A)

SC UE A acknowledges the CS CONNECT.

56-60. 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.

61 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.

62   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.

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.18   Signalling flows for PS to CS Access Transfer: SRVCC enhancements using ATCF

A.18.1   Introduction

The siginalling flows in the subclause demonstrate the PS to CS access transfer for SRVCC enhancements using ATCF. The following signalling flows are included:

-      subclause A.18.2 shows an example of PS to CS access transfer for SRVCC enhancements using ATCF and without media anchored.

-      subclause A.18.3 shows an example of PS to CS access transfer for SRVCC enhancements using ATCF and media anchored.

A.18.2   Signalling flows for PS to CS Access Transfer: 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.
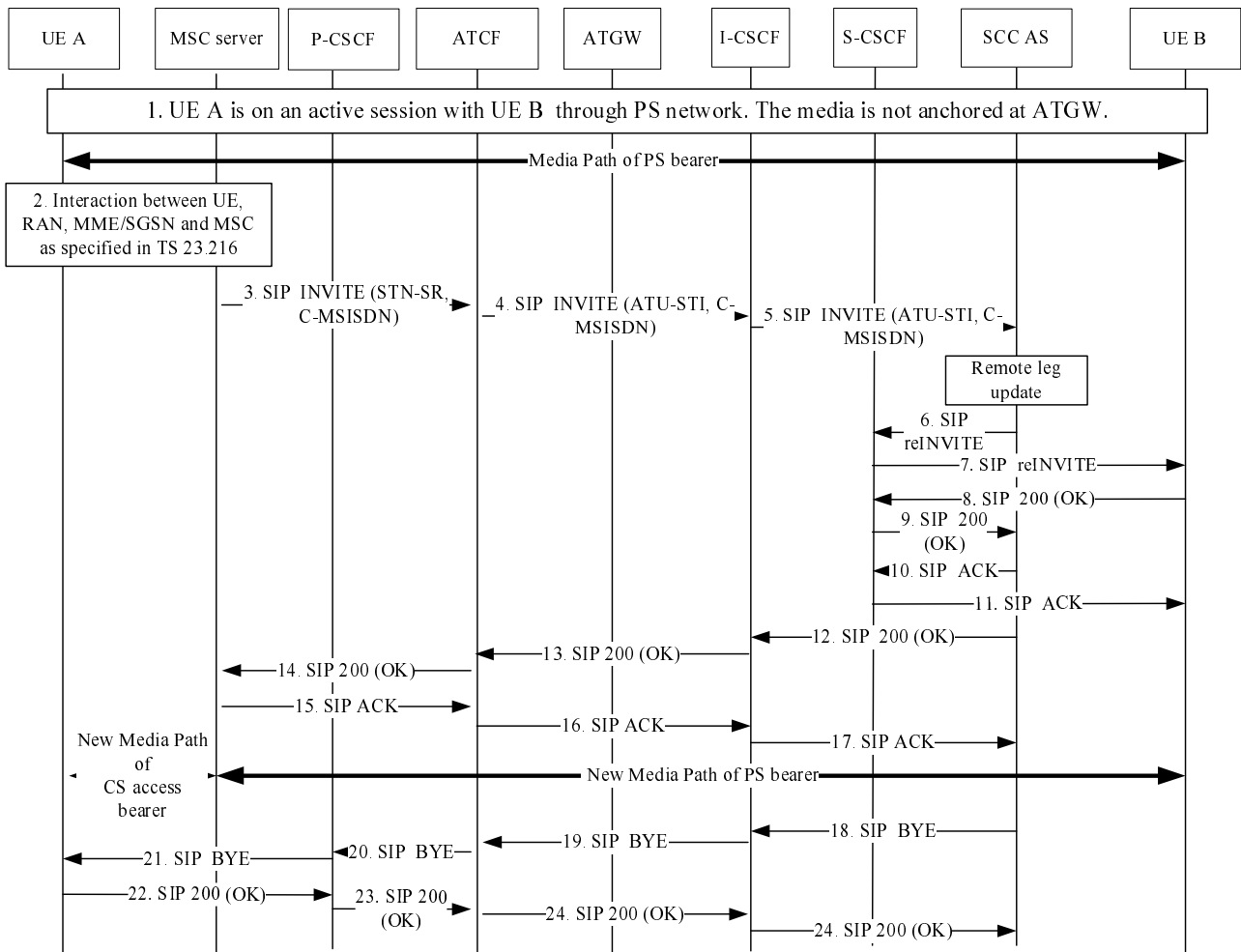
Figure A.18.2-1 Signalling flows for PS to CS access transfer: SRVCC enhancements using ATCF and without media anchored

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 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%3gpp-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%3gpp-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.

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%3gpp-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%3gpp-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: 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 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.
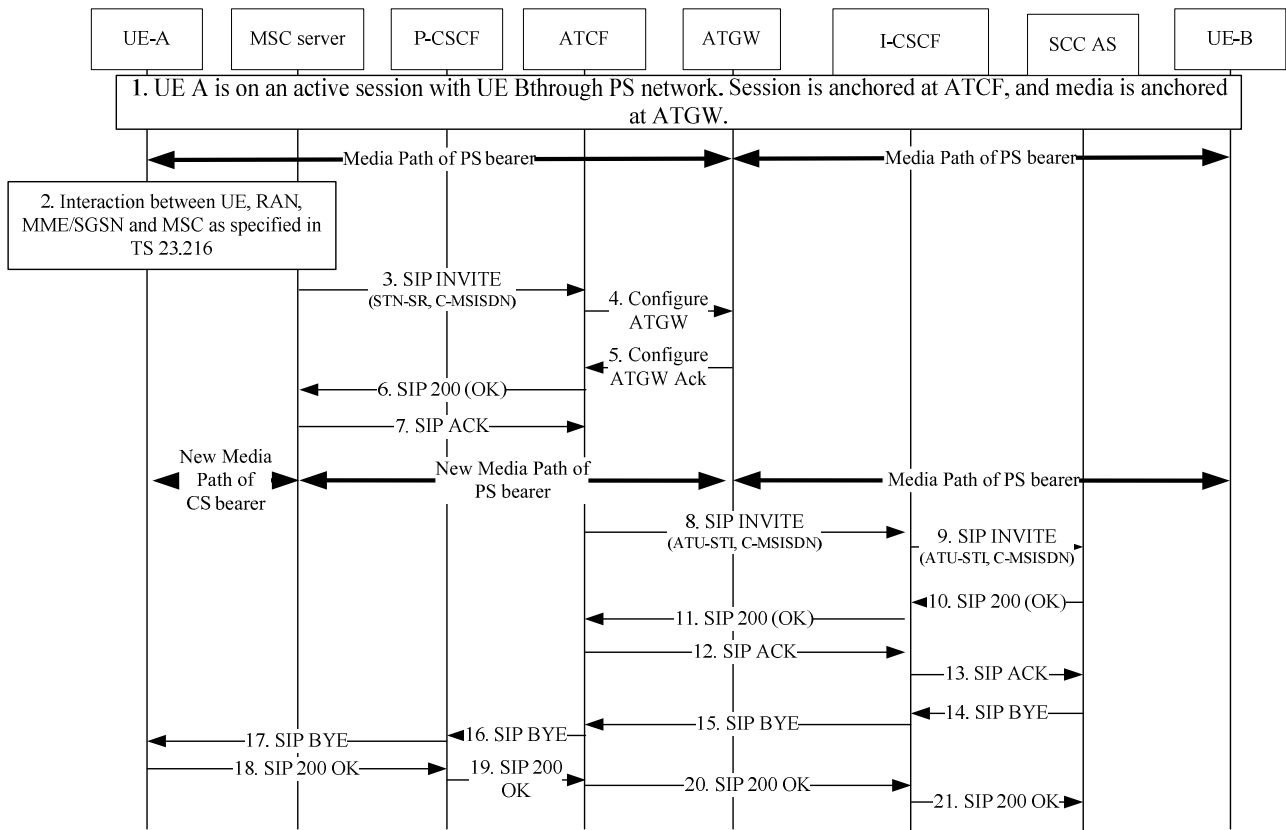
Figure A.18.3-1 Signalling flows for PS to CS access transfer: SRVCC enhancements using ATCF and media anchored

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 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%3gpp-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%3gpp-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.

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 re-establishes the communication with the SCC AS and updates the SCC AS that the transfer has taken place by sending a new SIP INVITE request to the SCC AS using the stored ATU-STI. 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 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:+1-237-555-3333>;tag=171828
To: <tel: +1-237-555-4444>
Call-ID: cb03a0s09a2sdfglkj490334
Cseq: 127 INVITE
Supported: 100rel, precondition, gruu
Target-Dialog: me03a0s09a2sdfgjkl491777; to-tag=774321; from-tag=64727891
Accept-Contact: *;+g.3gpp.icsi-ref="urn%3Aurn-7%3gpp-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%3gpp-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, as routed to the SCC AS.

Target-Dialog: specifies that the existing dialog is related with this request.

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.

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 usesGm 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

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 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 SRVCC access transfer for calls in alerting phase, i.e. for calls whith 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 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.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 access transfer 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 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 that a network entity supports 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 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 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].

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 10.19.0, 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 SRVCC session transfer in alerting phase and for accepting of a call in alerting state transferred by the PS-PS access transfer procedures.

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: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 SRVCC in the alerting state and for UE to communicate acceptance of incoming alerting state call transferred using PS-PS access transfer procedures.

 <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
```

```
early-value = %x65.61.72.6c.79 ; "early"
```

        <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
call-accepted-value = %x63.61.6c.6c.2d.61.63.63.65.70.74.65.64 ; "call-accepted"
```

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 10.19.0, 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:schema>
```

D.3.3 Semantic

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 consists of:

1) if the UE SRVCC Capability (see 3GPP TS 29.328 [6]) has a value UE-SRVCC-CAPABILITY-SUPPORTED and if the private user identity of the UE has an associated STN-SR (see 3GPP TS 29.328 [6]):

   a) <ATU-STI> element containing the ATU-STI of the SCC AS;

   b) <C-MSISDN> element containing the Correlation MSISDN of the UE.

   NOTE: <ATU-STI> element and <C-MSISDN> element are not included if the UE 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]).

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 10.19.0, 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 were discussed 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 (an 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.2.2.7.1General

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.2.2.7.2MIME 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 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 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 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 3gpp.state-and-event info package.

The security of the 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"

Annex F (informative):
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	8.0.0	8.1.0

					comtinuity		
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 clause 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
2011-12	CT#54	CP-110878	0547	3	Resource-Priority handling in ATCF	10.4.0	10.5.0
2011-12	CT#54	CP-110865	0551	1	Transfer failure if configuration tables are changed after the session was established	10.4.0	10.5.0

2011-12	CT#54	CP-110879	0561	3	Reference to Contact changed to reference Feature-Caps	10.4.0	10.5.0
2011-12	CT#54	CP-110879	0569		Answer shall be call-accepted	10.4.0	10.5.0
2011-12	CT#54	CP-110878	0572		Incorrect reference to 24.229	10.4.0	10.5.0
2011-12	CT#54	CP-110878	0577	1	Media offered by ATCF in target access leg during SRVCC	10.4.0	10.5.0
2011-12	CT#54	CP-110878	0579	1	ATCF procedures and "loose route" indication	10.4.0	10.5.0
2011-12	CT#54	CP-110867	0585		Missing SRVCC for IMS emergency procedures for MSC server enhanced for ICS	10.4.0	10.5.0
2011-12	CT#54	CP-110878	0593	2	Cleanup of ATCF procedures	10.4.0	10.5.0
2011-12	CT#54	CP-110873	0601	1	Removing the Accept header field from 183 response	10.4.0	10.5.0
2011-12	CT#54	CP-110878	0603	3	proxy-feature I-D reference update	10.4.0	10.5.0
2011-12	CT#54	CP-110873	0618		Record-Route at ATCF	10.4.0	10.5.0
2012-03	CT#55	CP-120107	0629	1	ATCF passing Recv-Info, Supported, Accept	10.5.0	10.6.0
2012-03	CT#55	CP-120095	0633		Speech media component	10.5.0	10.6.0
2012-03	CT#55	CP-120107	0637	1	ATCF-Path-URI in MESSAGE sent from SCC AS	10.5.0	10.6.0
2012-03	CT#55	CP-120107	0639	2	REGISTER handling in ATCF	10.5.0	10.6.0
2012-03	CT#55	CP-120106	0641	1	Overlapping conditions in MSC on INFO	10.5.0	10.6.0
2012-03	CT#55	CP-120107	0645	1	Reference update: draft-holmberg-sipcore-proxy-feature	10.5.0	10.6.0
2012-06	CT#56	CP-120305	0607	5	MSC server state verification procedure solving an EN	10.6.0	10.7.0
2012-06	CT#56	CP-120305	0675		Incorrect header field for media feature tags	10.6.0	10.7.0
2012-06	CT#56	CP-120304	0680	1	Path header set by ATCF	10.6.0	10.7.0
2012-06	CT#56	CP-120297	0698	1	Pay Load type correction – Additional sessions SRVCC	10.6.0	10.7.0
2012-06	CT#56	CP-120304	0702	1	Removing one EN about charging and ATCF	10.6.0	10.7.0
2012-06	CT#56	CP-120288	0706	3	Payload type correction – media interworking	10.6.0	10.7.0
2012-06	CT#56	CP-120411	0700	1	Pay Load type correction – Single (Held or Alerting) session SRVCC	10.6.0	10.7.0
2012-09	CT#57	CP-120575	0670	3	Use of Feature-Caps header to indicate the support of mid-call feature for SCC AS	10.7.0	10.8.0
2012-09	CT#57	CP-120582	0712	1	Reference update and technical changes: draft-ietf-sipcore-proxy-feature	10.7.0	10.8.0
2012-09	CT#57	CP-120582	0720	1	Replacing the Contact header field with the Feature-Caps header field	10.7.0	10.8.0
2012-09	CT#57	CP-120580	0721	1	Introduction of OMR in SCC AS	10.7.0	10.8.0
2012-09	CT#57	CP-120582	0726	1	Payload type correction – SIP 488 (Not Acceptable Here) response	10.7.0	10.8.0
2012-12	CT#58	CP-120792	0733	1	Handling of 1xx responses in alerting phase	10.8.0	10.9.0
2012-12	CT#58	CP-120790	0738	1	Removing an EN regarding overlaping procedures	10.8.0	10.9.0
2012-12	CT#58	CP-120792	0747	1	SCC AS handling corrections	10.8.0	10.9.0
2012-12	CT#58	CP-120791	0753	2	Feature-Caps header field in target refresh requests and responses	10.8.0	10.9.0
2012-12	CT#58	CP-120792	0755	2	Originating UE behaviours on the ringing tone when the aSRVCC occurs	10.8.0	10.9.0
2012-12	CT#58	CP-120775	0774		PS-PS access transfer correction	10.8.0	10.9.0
2013-03	CT#59	CP-130102	0782	1	404 and 480 from ATCF	10.9.0	10.10.0
2013-03	CT#59	CP-130102	0788		Reference update: RFC 6809	10.9.0	10.10.0
2013-03	CT#59	CP-130100	0800	1	SCC AS capability of MSC server assisted mid-call feature	10.9.0	10.10.0
2013-03	CT#59	CP-130103	0809	2	Alerting phase definition and interaction with CAT	10.9.0	10.10.0
2013-03	CT#59	CP-130097	0817		Feature-Tags with "none" value correction	10.9.0	10.10.0
2013-06	CT#60	CP-130235	0811	4	Completion of offer answer before aSRVCC	10.10.0	10.11.0
2013-06	CT#60	CP-130240	0820	2	Permanent error codes in SIP responses	10.10.0	10.11.0
2013-06	CT#60	CP-130235	0823	6	SRVCC cancellation by the MSC server	10.10.0	10.11.0
2013-06	CT#60	CP-130223	0831		SCC AS acting as an routeing B2BUA	10.10.0	10.11.0
2013-06	CT#60	CP-130235	0842	1	PS to CS SRVCC and malicious REFER transferring additional session	10.10.0	10.11.0
2013-06	CT#60	CP-130223	0855	2	Compliancy statement for SCC AS	10.10.0	10.11.0
2013-06	CT#60	CP-130223	0860	1	Compliancy statement for SC UE	10.10.0	10.11.0
2013-06	CT#60	CP-130236	0867	2	SCC AS providing SRVCC information	10.10.0	10.11.0
2013-06	CT#60	CP-130240	0870	1	Correcting handling of payload types in SC UE	10.10.0	10.11.0
2013-06	CT#60	CP-130223	0875	1	PS to PS - payload types in transfer of session	10.10.0	10.11.0
2013-06	CT#60	CP-130240	0878	1	PS to CS cancellation during an emergency call	10.10.0	10.11.0
2013-06	CT#60	CP-130236	0889		Handover cancellation support in SCC AS	10.10.0	10.11.0
2013-06	CT#60	CP-130232	0897	1	mid-call media feature tag IANA registration	10.10.0	10.11.0
2013-06	CT#60	CP-130236	0900		Completed IANA registrations for 24.237 from R10 on	10.10.0	10.11.0
2013-06	CT#60	CP-130232	0904	2	Resolving IANA comments on XML Schemas in 24.237	10.10.0	10.11.0
2013-09	CT#61	CP-130482	0910	2	Removing the mandate for the SCC AS to have to support PS-PS access transfer	10.11.0	10.12.0
2013-09	CT#61	CP-130492	0917	1	Providing Contact of far end on SRVCC	10.11.0	10.12.0
2013-09	CT#61	CP-130495	0924	2	Dual radio MID-CALL error corrections and clarifications	10.11.0	10.12.0
2013-09	CT#61	CP-130491	0929	1	SRVCC support of RFC 3262	10.11.0	10.12.0
2013-09	CT#61	CP-130495	0937	1	Correction for call termination procedures in the ATCF	10.11.0	10.12.0
2013-09	CT#61	CP-130495	0940	2	SCC AS send REFER to ATCF in case of PS to CS eSRVCC with additional call	10.11.0	10.12.0
2013-12	CT#62	CP-130740	0955	1	The MSC server mid-call support correction	10.12.0	10.13.0

2013-12	CT#62	CP-130740	0959	3	Release of session subject to access transfer in SC UE	10.12.0	10.13.0
2013-12	CT#62	CP-130734	0962	2	ATCF including the P-Asserted-Identity header field towards the MSC server	10.12.0	10.13.0
2013-12	CT#62	CP-130732	0965	3	SCC AS including the P-Asserted-Identity and the Contact header field towards the MSC server	10.12.0	10.13.0
2013-12	CT#62	CP-130732	0975	1	SCC AS receives CANCEL from MSC	10.12.0	10.13.0
2013-12	CT#62	CP-130732	0978	1	norefersub in Require / Supported	10.12.0	10.13.0
2014-03	CT#63	CP-140126	0988	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.	10.13.0	10.14.0
2014-03	CT#63	CP-140123	1008		Correct request for PS to CS cancelled handling	10.13.0	10.14.0
2014-06	CT#64	CP-140302	0995	2	IOI in MESSAGE with SRVCC information	10.14.0	10.15.0
2014-06	CT#64	CP-140298	1033		Incorrect name of the media feature tag carrying instance-id	10.14.0	10.15.0
2014-09	CT#65	CP-140639	1040	2	Removing references in ATCF to TS 24.229 subclause 5.7.5.1	10.15.0	10.16.0
2014-09	CT#65	CP-140683	1048	4	HO cancel for calls in allerting phase	10.15.0	10.16.0
2014-09	CT#65	CP-140637	1052	1	Alerting and forking	10.15.0	10.16.0
2014-12	CT#66	CP-140817	1063	1	E-SRVCC only for sessions with confirmed dialog	10.16.0	10.17.0
2014-12	CT#66	CP-140820	1066	1	SCC AS handling of 5xx responses	10.16.0	10.17.0
2014-12	CT#66	CP-140822	1070		P-Charging-Vector missing in the MSC server	10.16.0	10.17.0
2014-12	CT#66	CP-140821	1076		Removing references ",subclause 5.7.5.1"	10.16.0	10.17.0
2014-12	CT#66	CP-140822	1082	2	P-CSCF releasing the source access leg during PS to CS SRVCC during an emergency call	10.16.0	10.17.0
2015-03	CT#67	CP-150052	1105		MSC server initiate ICS registration after SRVCC	10.17.0	10.18.0
2015-06	CT#68	CP-150296	1135	2	Clarification on speech media component	10.18.0	10.19.0
2015-06	CT#68	CP-150298	1145	1	Addressing IANA Expert Review comments on "g.3gpp.mid-call" INFO package	10.18.0	10.19.0
2015-06	CT#68	CP-150295	1149	1	Addressing IANA Expert Review comments on "state-and-event" INFO package	10.18.0	10.19.0
2015-06	CT#68	CP-150298	1156	1	Addressing IANA Expert Review comments on "vnd.3gpp.mid-call+xml" MIME type	10.18.0	10.19.0
2015-06	CT#68	CP-150296	1163	1	Addressing expected IANA Expert Review comments on "vnd.3gpp.SRVCC-info+xml" MIME type	10.18.0	10.19.0
2015-06	CT#68	CP-150295	1167	1	Addressing IANA Expert Review comments on "vnd.3gpp.state-and-event+xml" MIME type	10.18.0	10.19.0
2015-12	CT#70	CP-150678	1204		Completed Rel-10 IANA Registrations	10.19.0	10.20.0
2015-12	CT#70	CP-150678	1223		Removal of editor's note on g.3gpp.state-and-event INFO package IANA registration	10.19.0	10.20.0
2015-12	CT#70	CP-150679	1227		Include C-MSISDN in the INVITE for transferring additional session	10.19.0	10.20.0

History

Document history		
V10.2.0	April 2011	Publication
V10.3.0	June 2011	Publication
V10.4.0	October 2011	Publication
V10.5.0	January 2012	Publication
V10.6.0	March 2012	Publication
V10.7.0	July 2012	Publication
V10.8.0	October 2012	Publication
V10.9.0	January 2013	Publication
V10.10.0	April 2013	Publication
V10.11.0	July 2013	Publication
V10.12.0	October 2013	Publication
V10.13.0	January 2014	Publication
V10.14.0	March 2014	Publication
V10.15.0	July 2014	Publication
V10.16.0	October 2014	Publication
V10.17.0	January 2015	Publication
V10.18.0	April 2015	Publication
V10.19.0	July 2015	Publication
V10.20.0	January 2016	Publication