ETSI TS 124 237 V9.2.0 (2010-04)

Technical Specification

Universal Mobile Telecommunications System (UMTS);
LTE;
IP Multimedia Subsystem (IMS) Service Continuity;
Stage 3
(3GPP TS 24.237 version 9.2.0 Release 9)

Reference
RTS/TSGC-0124237v920

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

Individual copies of the present document can be downloaded from:
http://www.etsi.org

The present document may be made available in more than one electronic version or in print. In any case of existing or perceived difference in contents between such versions, the reference version is the Portable Document Format (PDF). In case of dispute, the reference shall be the printing on ETSI printers of the 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:
http://portal.etsi.org/chaircor/ETSI_support.asp

Copyright Notification

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

© European Telecommunications Standards Institute 2010.
All rights reserved.

DECT™, **PLUGTESTS**™, **UMTS**™, **TIPHON**™, the TIPHON logo and the ETSI logo are Trade Marks of ETSI registered for the benefit of its Members.
3GPP™ is a Trade Mark of ETSI registered for the benefit of its Members and of the 3GPP Organizational Partners.
LTE™ is a Trade Mark of ETSI currently being registered
for the benefit of its Members and of the 3GPP Organizational Partners.
GSM® and the GSM logo are Trade Marks registered and owned by the GSM Association.

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 (http://webapp.etsi.org/IPR/home.asp).

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.

Contents

Intellectual Property Rights 2
Foreword 2
Foreword 9
1 Scope 10
2 References 10
3 Definitions and abbreviations 12
3.1 Definitions 12
3.2 Abbreviations 13
4 Overview of IP Multimedia (IM) Core Network (CN) subsystem Service Continuity 13
4.1 General 13
4.2 Underlying network capabilities 14
4.3 URI and address assignments 14
5 Functional entities 15
5.1 Introduction 15
5.2 User Equipment (UE) 15
5.3 Application Server (AS) 15
5.4 MSC Server 15
5.5 EATF 16
6 Roles for registration in the IM CN subsystem for service continuity 16
6.1 Introduction 16
6.2 SC UE 16
6.3 SCC AS 16
6.4 MSC server 17
6A Roles for General Capabilities 17
6A.1 Introduction 17
6A.2 UE roles 17
7 Roles for call origination for service continuity 18
7.1 Introduction 18
7.2 SC UE 18
7.3 SCC AS 18
7.3.1 Distinction of requests sent to the SCC AS 18
7.3.2 Call origination procedures at the SCC AS 18
7.4 EATF 19
7.4.1 Distinction of requests sent to the EATF 19
7.4.2 Call origination procedures at the EATF 19
8 Roles for call termination for service continuity 19
8.1 Introduction 19
8.2 SC UE 19
8.3 SCC AS 20
8.3.1 Distinction of requests sent to the SCC AS 20
8.3.2 Call termination procedures in the SCC AS 20
9 Roles for PS-CS access transfer 20
9.1 Introduction 20
9.2 SC UE 20
9.2.1 SC UE procedures for PS to CS access transfer 20
9.2.1A SC UE procedures for PS to CS access transfer with MSC server assisted mid-call feature 21
9.2.1B SC UE procedures for PS to CS access transfer with MSC server assisted mid-call feature for speech and video session 22
9.2.2 SC UE procedures for CS to PS access transfer 23
9.2.3 SC UE procedures for CS to PS access transfer with MSC server assisted mid-call feature 23

9.3 SCC AS 24
9.3.1 Distinction of requests sent to the SCC AS 24
9.3.2 SCC AS procedures for PS to CS access transfer 25
9.3.2A SCC AS procedures for PS to CS access transfer with MSC server assisted mid-call feature 26
9.3.3 SCC AS procedures for CS to PS access transfer 28
9.3.4 SCC AS procedures for CS to PS access transfer with MSC server assisted mid-call feature 28
9.4 MSC Server enhanced for ICS 30
9.4.1 MSC Server enhanced for ICS procedures for PS to CS session continuity with MSC server assisted mid-call feature 30
9.4.1A MSC Server enhanced for ICS procedures for PS to CS session continuity with MSC server assisted mid-call feature for speech and video session 31

10 Roles for PS-PS access transfer 31
10.1 Introduction 31
10.2 SC UE 31
10.2.0 General 31
10.2.1 Full session transfer 32
10.2.1A Additional procedures for full session transfer when MSC server assisted mid-call feature is supported 33
10.2.2 Partial session transfer 33
10.2.3 Additional procedures for partial session transfer when MSC server assisted mid-call feature is supported 34
10.3 SCC AS 34
10.3.1 Distinction of requests sent to the SCC AS 34
10.3.2 PS to PS access transfer procedures at the SCC AS 34
10.3.3 Additional SCC AS procedures for PS to PS access transfer when MSC server assisted mid-call feature is supported 35

11 Roles for PS-PS access transfer in conjunction with PS-CS access transfer 35
11.1 Introduction 35
11.2 SC UE 36
11.2.1 SC UE procedures for PS to PS+CS access transfer 36
11.2.1.1 General 36
11.2.1.2 SC UE procedures for PS to PS+CS access transfer using ICS 36
11.2.1.3 SC UE procedures for PS to PS+CS access transfer not using ICS 36
11.2.2 SC UE procedures for PS+CS to PS access transfer 37
11.2.2.1 General 37
11.2.2.2 SC UE procedures for PS+CS to PS access transfer using ICS 37
11.2.2.3 SC UE procedures for PS+CS to PS access transfer not using ICS 37
11.3 SCC AS 38
11.3.1 Distinction of requests sent to the SCC AS 38
11.3.2 SCC AS procedures for PS to PS+CS access transfer 38
11.3.3 SCC AS procedures for PS+CS to PS access transfer 39

12 Roles for PS-CS access transfer, Single Radio 39
12.1 Introduction 39
12.2 SC UE procedures for PS to CS access transfer, SR-VCC 40
12.2.1 General 40
12.2.2 ICS-based 40
12.2.3 Not based on ICS 40
12.2.3A Not based on ICS with MSC Server assisted mid-call feature 40
12.2.4 Abnormal cases 41
12.3 SCC AS 41
12.3.1 SCC AS procedures for PS to CS access transfer, SR-VCC 41
12.3.2 SCC AS procedures for PS to CS access transfer with MSC server assisted mid-call feature, SR-VCC 42
12.3.3 SCC AS procedures for SR-VCC, abnormal case 43
12.4 MSC server enhanced for ICS 43
12.5 EATF 44
12.5.1 EATF procedures for PS to CS session continuity, E-SR-VCC 44
MSC server enhanced for SRVCC using SIP interface 44
12.6.1 Session transfer from MSC server enhanced for SRVCC using SIP interface 44
12.6.2 Emergency session transfer from MSC server enhanced for SRVCC using SIP interface 44

13 Roles for media adding/deleting for access transfer 45
13.1 Introduction 45
13.2 SC UE 45
13.2.1 Adding or removing media through Gm 45
13.2.2 Adding Gm control to existing CS session 45
13.3 SCC AS 46
13.3.1 Adding or removing media through Gm 46
13.3.2 Adding Gm control to existing CS session 46

14 Roles for UE discovery for inter-UE transfer 46
14.1 Introduction 46
14.2 SC UE 47
14.3 SCC AS 47

15 Roles for inter-UE transfer without establishment of collaborative session 47
15.1 Introduction 47
15.2 SC UE 48
15.2.1 Transferor SC UE in services defining only originating session set up in UE 48
15.2.2 Transferee SC UE in services defining only originating session set up in UE 48
15.3 SCC AS 48
15.3.1 Distinction of requests sent to the SCC AS 48
15.3.2 Inter UE transfer request authorization in services defining only originating session set up in UE 49
15.3.3 SCC AS procedures for inter UE transfer in services defining only originating session set up in UE 49

16 Roles for collaborative session establishment for inter-UE transfer 50
16.1 Introduction 50
16.2 SC UE 50
16.2.1 SC UE procedures for collaborative session establishment by transferring media used in an existing session 50
16.2.1.1 Controller UE procedures 50
16.2.1.2 Controllee UE procedures 51
16.2.2 SC UE procedures for collaborative session establishment with new media 51
16.2.2.1 Controller UE procedures 51
16.2.2.2 Controllee UE procedures 52
16.2.3 SC UE procedures for subscription to the session description of a collaborative session 52
16.3 SCC AS 52
16.3.1 Distinction of requests sent to the SCC AS 52
16.3.2 SCC AS procedures for collaborative session establishment by transferring media 53
16.3.3 SCC AS procedures for collaborative session establishment with new media 54
16.3.4 Procedures to indicate SCC AS establishing a collaborative session by adding a new media 55
16.3.5 SCC AS procedures for subscription to the session description of a collaborative session 55

17 Roles for media transfer within collaborative session for inter-UE transfer 55
17.1 Introduction 55
17.2 SC UE 55
17.2.1 Procedures for controller UE initiated media transfer from controller UE to controllee UE 55
17.2.1.1 Controller UE procedures 55
17.2.1.2 Controllee UE procedures 56
17.2.2 Procedures for controller UE initiated media transfer from controllee UE to another controllee UE 56
17.2.2.1 Controller UE procedures 56
17.2.2.2 Controllee UE procedures 56
17.3 SCC AS 56
17.3.1 Procedures for controller UE initiated media transfer from controller UE to controllee UE 56
17.3.2 Procedures for controller UE initiated media transfer from controllee UE to another controllee UE 57

18 Roles for release of collaborative session for inter-UE transfer 58
18.1 Introduction 58
18.2 SC UE 58
18.2.1 Procedures for collaborative session release by controller UE 58
18.2.1.1 Controller UE 58
18.2.1.2 Controllee UE 58
18.2.2 Procedures for collaborative session release by remote party 58
18.2.2.1 Controller UE 58

18.2.2.2 Controllee UE 58
18.3 SCC AS 59
18.3.1 Procedures for collaborative session release by controller UE 59
18.3.2 Procedures for collaborative session release by remote party 59

19 Roles for media adding/deleting within collaborative session for inter-UE transfer 59
19.1 Introduction 59
19.2 SC UE 59
19.2.1 Procedures for adding new media on controllee UE by controller UE 59
19.2.2 Procedures for releasing media on controllee UE by controller UE 59
19.2.2A Procedures for releasing media on controller UE by controller UE 60
19.2.2B Procedures for controller UE to remove a controllee UE from the collaborative session 60
19.2.3 Procedures for releasing media component by controllee UE 60
19.2.3.1 Controller UE 60
19.2.3.2 Controllee UE 61
19.2.4 Procedures for modifying media on controllee UE by itself 61
19.2.5 Procedures for adding new media by remote UE when the controller UE does not alert the user 61
19.2.6 Procedures for releasing media by remote UE 62
19.3 SCC AS 62
19.3.0 Distinction of requests at the SCC AS 62
19.3.1 Procedures for adding new media on controllee UE by controller UE 63
19.3.2 Procedures for releasing media on controllee UE by controller UE 63
19.3.2A Procedures for releasing media on controller UE by controller UE 63
19.3.2B Procedures for controller UE removing controllee UE from the collaborative session 64
19.3.3 Procedures for releasing media component initiated by controllee UE 64
19.3.4 Procedures for modifying media on controllee UE by itself 65
19.3.5 Procedures for adding new media by remote UE when the controller UE does not alert the user 65
19.3.6 Procedures for releasing media by remote UE 66

20 Service continuity and MMTEL interactions 67
20.1 Roles for access tranfer and supplementary services interaction 67
20.1.1 Introduction 67
20.1.2 Originating Identification Presentation (OIP) 67
20.1.3 Originating Identification Restriction (OIR) 67
20.1.4 Terminating Identification Presentation (TIP) 67
20.1.5 Terminating Identification Restriction (TIR) 67
20.1.6 Communication Diversion (CDIV) 67
20.1.7 Communication Hold (HOLD) 68
20.1.8 Communication Barring (CB) 68
20.1.9 Message Waiting Indication (MWI) 68
20.1.10 Conference (CONF) 68
20.1.11 Explicit Communication Transfer (ECT) 68
20.1.12 Advice of Charge (AOC) 69
20.1.13 Closed User Groups (CUG) 69
20.1.14 Three-Party (3PTY) 69
20.1.15 Flexible Alerting (FA) 69
20.1.16 Communication Waiting (CW) 69
20.1.17 Completion of Communications to Busy Subscriber (CCBS)/Completion of Communications by No Reply (CCNR) 69
20.1.18 Customized Alerting Tones (CAT) 69
20.1.19 Malicious Communication IDentification (MCID) 69
20.1.20 Reverse Charging 70
20.1.21 Personal Network Management (PNM) 70
20.1.22 Customized Ringing Signal (CRS) 70
20.2 Roles for inter-UE transfer and supplementary services interaction 70
20.2.1 Introduction 70
20.2.2 Originating Identification Presentation (OIP) 70
20.2.3 Originating Identification Restriction (OIR) 70
20.2.4 Terminating Identification Presentation (TIP) 70
20.2.5 Terminating Identification Restriction (TIR) 70
20.2.6 Communication Diversion (CDIV) 70
20.2.7 Communication Hold (HOLD) 70

20.2.8 Communication Barring (CB) 72
20.2.9 Message Waiting Indication (MWI) 72
20.2.10 Conference (CONF) 72
20.2.11 Explicit Communication Transfer (ECT) 72
20.2.12 Advice of Charge (AOC) 72
20.2.13 Closed User Groups (CUG) 72
20.2.14 Three-Party (3PTY) 72
20.2.15 Flexible Alerting (FA) 72
20.2.16 Communication Waiting (CW) 72
20.2.17 Completion of Communications to Busy Subscriber (CCBS)/Completion of Communications by No Reply (CCNR) 73
20.2.18 Customized Alerting Tones (CAT) 73
20.2.19 Malicious Communication IDentification (MCID) 73
20.2.20 Personal Network Management (PNM) 73
20.2.21 Customized Ringing Signal (CRS) 73

Annex A (informative): Example signalling flows 74

A.1 Scope of signalling flows 74

A.2 Introduction 74
A.2.1 General 74
A.2.2 Key required to interpret signalling flows 74

A.3 Signalling flows for registration 75
A.3.1 Introduction 75
A.3.2 Signalling flows for multiple registration for service continuity 75

A.4 Signalling flows for call origination for service continuity 80
A.4.1 Session origination for CS calls 80

A.5 Signalling flows for call termination for service continuity 81
A.5.1 Session termination using CS media 81

A.6 Signalling flows for PS-CS access transfer 81
A.6.1 PS-CS access transfer: CS-PS 81
A.6.2 PS-CS access transfer: PS-CS 85

A.7 Signalling flows for PS-PS access transfer 88
A.7.1 Introduction 88
A.7.2 PS-PS access transfer with full media transfer 88
A.7.3 PS-PS access transfer with partial media transfer 95

A.8 Signalling flows for PS-PS access transfer in conjunction with PS-CS access transfer 103
A.8.1 Introduction 103
A.8.2 PS - PS in conjunction with PS - CS Access Transfer: PS to CS 104
A.8.3 PS - PS in conjunction with PS - CS Access Transfer: CS to PS 111

A.9 Signalling flows for media adding/deleting for access transfer 115

A.10 Signalling flows for Inter-UE Transfer without establishment of Collaborative Session 119
A.10.1 Introduction 119
A.10.2 Complete transfer in services defining only originating session set up in UE 119
A.10.3 Complete transfer in services defining terminating session set up in UE 123

A.11 Signalling flows for establishment of collaborative session for inter-UE transfer 124
A.11.1 Introduction 124
A.11.2 Collaborative session establishment with new media 124

A.12 Signalling flows for media transfer within collaborative session for inter-UE transfer 130
A.12.1 Introduction 130
A.12.2 Controller UE initiated media transfer from controller UE to controllee UE 131
A.12.3 Controller UE initiated media transfer from controllee UE to another controllee UE with subscription to dialog events 136
A.12.3A Controller UE initiated media transfer from controllee UE to another controllee UE 148

A.13 Signalling flows for release of collaborative session for inter-UE transfer 156

A.13.1 Introduction 156
A.13.2 Controller UE releases collaborative session 156
A.13.3 Remote UE releases collaborative session 158

A.14 Signalling flows for media adding/deleting within collaborative session for inter-UE transfer 159
A.14.1 Introduction 159
A.14.2 Controller UE releases media 160
A.14.2.1 Controller UE releases media flow on controller UE 160
A.14.2.2 Controller UE releases media flow on controllee UE 164
A.14.2.2.1 Controller UE removes media at the controllee UE 164
A.14.2.2.2 Controller UE remove the controllee UE from the collaborative session 171
A.14.4 Controllee UE releases media 176
A.14.5 Controllee UE modifies media on itself 179
A.14.6 Remote UE adds new media on controllee UE 182
A.14.7 Remote UE releases media on the controller UE 189
A.15 Signalling flows for MSC server assisted mid-call feature 192
A.15.1 Introduction 192
A.15.2 CS to PS access transfer with MSC server assisted mid-call feature 193
A.15.3 PS to CS access transfer with MSC server assisted mid-call feature 199

A.16 Signalling flows for SRVCC session transfer for IMS emergency session 206
A.16.3 Session transfer for emergency session using SRVCC procedure: PS-CS 212

Annex B (informative): Collaborative session state model 217

Annex C (normative): Media feature tags defined within the current document 220

C.1 General 220

C.2 Definition of media feature tag g.3gpp.mid-call 220

C.3 Definition of media feature tag g.3gpp.iut-controller 220

Annex D (informative): XML schemas 221

D.1 MSC Server assisted mid-call feature XML schema 221
D.1.1 General 221
D.1.2 XML schema 221
D.1.3 IANA registration template 221

Annex E (informative): Change history 223

History 227

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 or across different user equipments (UEs) under the control of the same subscriber.

NOTE: When the communication session is transferred across different UEs, the session can be a collaborative session with controller and controllee UEs. In this version of the document, there can only be one controller UE but several controllee UEs in the collaborative session.

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) and MSC Servers providing IMS Service Continuity capabilities.

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] 3GPP TS 23.167: "IP Multimedia Subsystem (IMS) emergency sessions".

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

[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 Initiation Protocol (SIP)".

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

[50] IETF RFC 3556 (July 2003) "Session Description Protocol (SDP) Bandwidth Modifier".

[51] IETF RFC 792 (September 1981) "INTERNET CONTROL MESSAGE PROTOCOL".

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

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].

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.

Inter UE Transfer SCC AS URI: A SIP URI which is a public service identity hosted by SCC AS and which is used in inter UE transfer procedures.

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.

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

Access Leg
Remote Leg
Target Access Leg

Source Access Leg
Session Transfer Identifier (STI)
Session Transfer Number (STN)
Session Transfer Number for SR-VCC (STN-SR)
Collaborative session
Controllee UE
Controller UE
Inter-UE transfer
Emergency Session Transfer Number for SR VCC(E-STN-SR)
Correlation MSISDN

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 23.003 [12] apply:

IP Multimedia Routing Number (IMRN)

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

Emergency service URN

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
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, SC can be divided into two concepts:

1. providing the capability of transferring 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; and

2. providing the capability of transferring the communication sesions with multiple media across different UEs.

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 transferare specified in clause 9;
- procedures for PS-PS access transferare specified in clause 10;
- procedures for PS-PS access transferin conjunction with PS-CS access transferare specified in clause 11;
- procedures for PS-CS access transferfor Single Radio are specified in clause 12;
- procedures for media adding/deleting for access transfer are specified in clause 13;
- procedures for UE discovery for inter-UE transfer are specified in clause 14;
- procedures for inter-UE transfer without establishment of collaborative session are specified in clause 15;
- procedures for collaborative session establishment for inter-UE transfer are specified in clause 16;
- procedures for media transfer within collaborative session for inter-UE transfer are specified in clause 17;
- procedures for release of collaborative session for inter-UE transfer in clause 18;
- procedure for media adding/deleting within collaborative session for inter-UE transfer are specified in clause 19; and
- procedures for service continuity and MMTEL interactions are specified in clause 20.

For a UE or an AS not supporting ICS procedures, PS-CS service continuity is only possible when the UE is active in a single full-duplex speech or speech/video session i.e. support of access transfer or inter-UE transfer with more than one session or with non voice media is not provided.

4.2 Underlying network capabilities

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];

4.3 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] and an Inter UE Transfer SCC AS URI. 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. The Inter UE Transfer SCC AS URI is used in the inter UE transfer procedures.

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.

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)

UE can be compliant with:

- both the access transfer and the inter-UE transfer;
- the access transfer; or
- the inter-UE transfer

in this document.

If the SC UE supports the Controller UE procedures for IUT transfer then the SC UE may include the g.3gpp.iut-controller media feature tag as described in annex C in the Contact header of SIP requests and SIP responses.

To be compliant with access transfer in this document, a UE shall implement the role of an SC UE according to subclause 6.2, subclause 7.2, subclause 8.2, subclause 9.2, subclause 10.2, subclause 11.2, subclause 12.2 and subclause 13.2.

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

- by following the procedures specified in 3GPP TS 24.229 [2] for registration of the UE in the IM CN subsystem; and
- according to subclause 14.2, subclause 15.2, subclause 16.2, subclause 17.2, subclause 18.2 and subclause 19.2.

NOTE: In the inter-UE transfer, a session can be collaborative session where there are one controller UE and severeal controllee UEs. The controllee UE can be a legacy UE and does not have be compliant with the above subclauses.

5.3 Application Server (AS)

AS can be compliant with:

- both the access transfer and the inter-UE transfer;
- the access transfer; or
- the inter-UE transfer

in this document.

To be compliant with access transfer in this document, an AS shall implement the role of an SCC AS according to subclause 6.3, subclause 7.3, subclause 8.3, subclause 9.3, subclause 10.3, subclause 11.3, subclause 12.3 and subclause 13.3.

To be compliant with inter-UE transfer in this document, an AS shall implement the role of an SCC AS according to subclause 14.3, subclause 15.3, subclause 16.3, subclause 17.3, subclause 18.3 and subclause 19.3.

5.4 MSC Server

An MSC Server can be compliant with SR VCC 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 [11] and the role of an MSC server enhanced for SRVCC using SIP interface as described in subclause 12.6.

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.

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 supports the Controller UE procedures for IUT transfer then the SC UE shall include the g.3gpp.iut-controller media feature tag as described in annex C in the Contact header field of the SIP REGISTER request.

6.3 SCC AS

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).

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.

6.4 MSC server

If MSC server supports the MSC server assisted mid-call feature, the MSC server shall behave as an MSC server enhanced for ICS as specified in 3GPP TS 29.292 [18] and 3GPP TS 24.292 [4] with following additions:

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

6A Roles for General Capabilities

6A.1 Introduction

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

6A.2 UE roles

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.

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 and the EATF.

7.2 SC UE

If SC using ICS is enabled then 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.

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

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 instance-id 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.

7.3 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.

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 [8].

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: 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 media feature tag as described in annex C in the Contact 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.

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 [8].

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.

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 and the SCC AS.

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]and in 3GPP TS 24.292 [4]. If the SC UE supports the MSC server assisted mid-call feature, and the receiving SIP INVITE request includes media feature tag as described in annex C in the Contact 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.

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

The procedures for call termination where the SC UE is receiving calls using CS media are identical to that for ICS UE roles for call termination procedures which are specified in 3GPP TS 24.292 [4].

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 follow the SCC AS roles for call termination 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; 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 media feature tag as described in annex C in the Contact 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.

9 Roles for PS-CS access transfer

9.1 Introduction

For a UE or an AS not supporting ICS procedures, PS-CS access transfer is only possible when the UE is active in a single full-duplex speech or speech/video session i.e. support of session transfer with more than one sessions or with non voice media is not provided.

9.2 SC UE

9.2.1 SC UE procedures for PS to CS access transfer

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 to be transferred and starting with the active session, 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 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, then access transfer is only possible when the UE is active in a single full-duplex speech session. If multiple full-duplex speech sessions exist, 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) first initiate the release of all the ongoing full-duplex speech sessions except the session with active full-duplex speech component that was most recently made active and then the SC UE shall transfer the remaining ongoing active full-duplex speech session.

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;

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 release message to the CC SETUP message sent, then PS-CS acccess transfer has not completed successfully and the call will continue in the Source Access Leg.

9.2.1A SC UE procedures for PS to CS access transfer with MSC server assisted mid-call feature

Editor's note: it is FFS how to ensure that the UE and the MSC Server would use the same transaction identifers for the conferencing sessions

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 active full-duplex speech session and the Contact header field received during the establishment of the ongoing active full-duplex speech session which has been most recently made active includes the g.3gpp.mid-call media feature tag as described in annex C; or

   B. there is no ongoing active full-duplex speech session and the Contact header field received during the establishment of the ongoing inactive full-duplex speech session which became inactive most recently includes the g.3gpp.mid-call media feature tag 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 active full-duplex speech sessions:

   A. initiate the release of all the ongoing full-duplex speech sessions except two that were most recently made active;

   B. initiate the session modification of the ongoing full-duplex speech session that was made active less recently and offer the full-duplex speech component with "sendonly" or "inactive" directionality; and

C. transfer two remaining ongoing full-duplex speech sessions;

NOTE 1: When active and inactive ongoing full-duplex speech sessions exist, one CC SETUP message transfers both sessions.

2. if there are one ongoing active full-duplex speech session and one or more ongoing inactive full-duplex speech session,

   A. initiate the release of all the ongoing inactive full-duplex speech sessions except the one which became inactive most recently; and

   B. transfer two remaining ongoing full-duplex speech sessions;

3. if there is one ongoing active full-duplex speech session and no ongoing inactive full-duplex speech session, transfer the ongoing full-duplex speech session; and

4. if there is no ongoing active full-duplex speech session and there is one or more ongoing inactive full-duplex speech sessions:

   A. initiate the release of all the ongoing inactive full-duplex speech sessions except the one which became inactive most recently; and

   C. transfer the ongoing full-duplex speech session.

NOTE 2: The ongoing inactive full-duplex speech sessions 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

The SC UE shall consider session with speech:

- which has "sendonly" or "inactive" directionality as inactive; and

- which has "recvonly" or "sendrecv" directionality as active;

in this subclause and in the referenced subclauses.

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 to be transferred and starting with the active session, 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, then access transfer is only possible when the SC UE is active in a single full-duplex speech session. If multiple full-duplex speech sessions exist, the SC UE shall first initiate the release of all the ongoing sessions that are currently not active and then the SC UE shall transfer the remaining ongoing active full-duplex speech session.

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.

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 full-duplex speech session 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 media feature tag as specified in annex C was included in the Contact 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; and

      d. media directionality as in the "body" URI header field in the URI in the Refer-To header field of the received SIP REFER request.

NOTE 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.

The SC UE shall consider session with speech:

- which has "sendonly" or "inactive" directionality as inactive; and

- which has "recvonly" or "sendrecv" directionality as active;

in this subclause and in the referenced subclauses.

9.3 SCC AS

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. 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: 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].

9.3.2 SCC AS procedures for PS to CS access transfer

This subclause does not apply to recepetion 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 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 and P-Asserted 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 audio media 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 audio media 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 audio media;

2. if no dialogs supporting a session with active audio media 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 audio media 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 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 Instance-ID 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 audio media 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 audio media and continue the session transfer procedures with the dialog supporting a session with active audio media; 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 audio media 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 audio media 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 audio media 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) 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 audio media components 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

- If the Source Access Leg contains media components other than audio 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 shall apply the MSC Server assisted mid-call feature if:

1. the Contact header field of the SIP INVITE request due to static STN includes the g.3gpp.mid-call media feature tag as specified in annex C;

2. one of the following is true:

   A. at least one dialog supporting a session with active audio media exists for the user identified in the P-Asserted-Identity header field and the Contact header field provided by the SC UE at the establishment of the dialog supporting a session with active audio media which has been most recently made active includes the g.3gpp.mid-call media feature tag as described in annex C; or

   B. no dialog supporting a session with active audio media exists for the user identified in the P-Asserted-Identity header field, one or more dialogs supporting a session with inactive audio media exists for the user and the Contact header field provided by the SC UE at the establishment of the dialog supporting a session with inactive audio media which became inactive most recently includes the g.3gpp.mid-call media feature tag as described in annex C;

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

4. 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.

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 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 audio media 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 audio media, release all dialogs supporting a session with inactive audio media except the one with the audio media which became inactive most recently and continue the session transfer procedures with the dialog supporting a session with active audio media;

2. if more than one dialog supporting a session with active audio media exists for the user identified in the P-Asserted-Identity header field and a SIP 2xx response has been sent for these dialogs, release all dialogs supporting a session with audio media except two with the audio media which became active most recently and continue the session transfer procedures with the dialog supporting a session with the audio media which became active most recently; and

3. if no dialog supporting a session with active audio media exists for the user identified in the P-Asserted-Identity header field, one or more dialogs supporting a session with inactive audio media 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 audio media except the one with the audio media which became inactive most recently and continue the session transfer procedures with the dialog supporting a session with inactive audio media.

Editor's note: transport of the session state information in conferencing sessions is FFS

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 media feature tag as described in annex C in the Contact 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 dialog supporting a session with inactive audio media 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, two SIP dialogs supporting a session with an audio media 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 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. 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 STN, i.e. set to the URI provided by SCC AS in the Contact header field to the SC UE during the session establishment 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.

The SCC AS shall consider dialog supporting a session with audio media:

- which has "sendonly" or "inactive" directionality at the SC UE as inactive; and

- which has "recvonly" or "sendrecv" directionality at the SC UE as active;

in this subclause and in the referenced subclauses.

9.3.3 SCC AS procedures for 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 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 audio media 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 audio media 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 audio media 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 9.3.4, 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 audio media and a SIP 2xx response has been sent for that dialog, then:

   A. if the remaining dialogs support a session with inactive audio media 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 audio media and continue the session transfer procedures with the dialog supporting a session with active audio media; 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;

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

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 audio media 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 audio media, release all dialogs supporting a session with inactive audio media except the one with the audio media which became inactive most recently and continue the session transfer procedures with the dialog supporting a session with active audio media ; and

2. if no dialog supporting a session with active audio media exists for the user identified in the P-Asserted-Identity header field, one or more dialogs supporting a session with inactive audio media 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 audio media except the one with the audio media which became inactive most recently and continue the session transfer procedures with the dialog supporting a session with inactive audio media.

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 media feature tag as described in annex C in the Contact 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 audio media 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 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, two SIP dialogs supporting a session with an audio media 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 URI provided by SCC AS in the Contact header field to the SC UE during the session establishment 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. the To URI header field populated as specified in IETF RFC 3261 [8], 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 [8], 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 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 3GPP IM CN subsystem XML body compliant to the XML schema specified in the annex D.1.2.

The SCC AS shall consider dialog supporting a session with audio media:

- which has "sendonly" or "inactive" directionality at the MSC Server as inactive; and

- which has "recvonly" or "sendrecv" directionality at the MSC Server as active;

in this subclause and in the referenced subclauses.

9.4 MSC Server enhanced for ICS

9.4.1 MSC Server enhanced for ICS procedures for PS to CS session continuity with MSC server assisted mid-call feature

Editor's note: it is FFS how to ensure that the UE and the MSC Server would use the same transaction identifers for the conferencing sessions

In addition to the procedures specified in 3GPP TS 29.292 [18], the MSC Server enhanced for ICS supporting the MSC Server assisted mid-call feature 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; and
3. include in the Contact header field the g.3gpp.mid-call media feature tag as described in annex C.

NOTE 1: Since the MSC Server is not able to distinguish the dual radio access transfer from the regular session set up, the information elements above are added to every SIP INVITE request sent by the MSC Server.

Upon receiving a SIP REFER request

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 media feature tag as specified in annex C was included in the Contact 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 MSC Server enhanced for ICS supports the MSC Server assisted mid-call feature, then the MSC Server enhanced for ICS shall:

1. handle the SIP REFER request as specified in 3GPP TS 29.292 [18], 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 29.292 [18] and IETF RFC 3515 [13]. Additionally, the MSC Server enhanced for ICS 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; 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; and

d. media directionality. as in the "body" URI header field in the URI in the Refer-To header field of the received SIP REFER request

NOTE 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.

If two sessions are transferred, the MSC Server enhanced for ICS shall 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.

9.4.1A MSC Server enhanced for ICS procedures for PS to CS session continuity with MSC server assisted mid-call feature for speech and video session

If the MSC Server enhanced for ICS supports the MSC Server assisted mid-call feature, upon receiving the session state information which indicates an inactive speech and video session, the MSC Server enhanced for ICS shall send a SIP INVITE request for the additional inactive speech and video session as described in subclause 9.4.1A.

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 enhanced for ICS can perform the procedures as specified in 3GPP TS 29.292 [18], with the addition that the MSC Server enhanced for ICS 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.

10.2 SC UE

10.2.0 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.

The SC UE shall follow the procedures specified in subclause 6.1 to perform registration in the IM CN subsystem on the newly selected access network before performing PS-PS access transfer.

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"; and

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 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.

   A.- 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 a port number with value zero; and

   B.- If the SC UE determines to add new media component(s) during the transfer, then the SC UE shall 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, if the dialog over the Source Access Leg 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.

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.

10.2.1A 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 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.

   A. If the SC UE determines to keep the media component on the Source Access Leg, then the SCUE shall set the media line for this media component to a port number with value zero; and

   B. If the SC UE determines to add new media component(s) during the transfer, then the SC UE shall 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. In the procedures below such requests are known as "SIP INVITE requests due to STI".

NOTE: The media streams that need to be transferred are identified using information described in the subsequent sections.

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

10.3.2 PS to PS access transfer procedures at the SCC AS

This subclause applies to reception of a SIP INVITE request due to STI with a PS media only.

When the SCC AS receives a SIP INVITE request on the Target Access Leg due to STI, the SCC AS shall:

- associate the SIP INVITE received on the Target Access Leg with an ongoing SIP dialog i.e. identify the Source Access Leg. 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 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;

- if the SCC AS is unable to associate the SIP INVITE with a unique ongoing dialog, send a SIP 480 (Temporarily Unavailable) response to reject the SIP INVITE request relating to the access transfer and not processes the remaining steps;

- if the 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 [8]; 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 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, 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) 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]; 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, subsequent to initiating the SIP re-INVITE 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.

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 media feature tag as described in annex C in the Contact 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.

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 session with full-duplex speech with the remote end i.e. support of session transfer with more than one session containing full-duplex speech component 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 to be transferred and starting with the session with active full-duplex speech 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 is using CS bearer with its corresponding media description.

- When sending the SIP INVITE request for the sessions with inactive full-duplex speech component and if precondition is used, the SC UE shall indicate that the related local preconditions for the speech component are met.

- For the session with active full-duplex speech 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 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 session with active full-duplex speech 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, then access transfer is only possible when the UE is active in a single session with full-duplex 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 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 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 component is still maintained on the Source Access Leg.

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 session to be transferred and starting with the 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 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. 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 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 full-duplex speech 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 and the SCC AS. For SC UE or SCC AS not supporting ICS procedures, PS-CS access transfer in SR-VCC is only possible when SC is enabled, the UE is active in a single session with full-duplex speech i.e. support of session transfer with more than one session containing full-duplex speech component is not provided.

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.

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 speech session to be transferred, starting with the session with active full-duplex speech component that was most recently made active; and
- within the SDP offer indicate the media line for all active and held audio streams as an audio stream 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 most recently active PS audio session 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 most recently active audio call 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 full-duplex speech media 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 active PS audio session and the Contact header field received by the SC UE at the establishment of the active PS audio session, which has been most recently made active, includes the g.3gpp.mid-call media feature tag as described in annex C; or

   B. there is no active PS audio session and the Contact header field received by the SC UE at the establishment of the inactive PS audio session which became inactive most recently includes the g.3gpp.mid-call media feature tag 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 active PS audio sessions exist, the SC UE shall replace the PS audio components of the two most recently active PS audio sessions with the newly established active and held CS voice calls;

2. if one active PS audio session exists and one or more inactive PS audio sessions exist, the SC UE shall replace the PS audio components of the active PS audio session and of the most recently inactive PS audio session with the newly established active and held CS voice calls;

3. if one active PS audio session exists and no inactive PS audio sessions exist, the SC UE shall replace the PS audio component of the active PS audio session with the newly established active CS voice call; and

4. if no active PS audio session exists and one or more inactive PS audio session exists, the SC UE shall replace the PS audio component of the inactive PS audio session which became inactive most recently with the newly established held CS voice call.

Editor"s Note: It is FFS how to ensure that the UE and MSC Server would use the same transaction identifiers for the conferencing sessions.

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 active session transaction identifier value is described in 3GPP TS 24.008 [8]

The SC UE shall consider session with audio media:

- which has "sendonly" or "inactive" directionality as inactive; and

- which has "recvonly" or "sendrecv" directionality as active;

in this subclause and in the referenced subclauses.

If single inactive session 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.

12.2.4 Abnormal cases

If the SC UE engaged in one or more ongoing IMS sessions 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, then the SC UE shall send a SIP reINVITE request containing an SDP offer, including the media characteristics as used in the existing dialog, by following the rules of 3GPP TS 24.229 [2].

12.3 SCC AS

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

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".

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 full-duplex speech component that was most recently made active. However, the SCC AS does not initiate release for Source Access Leg unless after some specific time defined by the operator policy.

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

2) if the SIP dialog contains other media flows than the active speech flow, the SCC AS shall modify the remote leg and remove the speech media flow, 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 follow the procedures as described for ICS using Gm in subclause 13.3.2.

NOTE: When using the ICS controlled CS bearer, only one audio call can be active at a time. Nevertheless, several calls can be held in parallel. If the user decides to switch to another (previously held) call, the ICS controlled CS bearer is re-used for this call. Therefore no specific procedures for handling of held calls in the case of ICS controlled CS bearer are needed.

When the SCC AS receives a SIP re-INVITE for non-ICS control, the SCC AS shall follow the media removal procedures as specified in subclause 13.3.1.

Unless the MSC Server assisted mid-call feature applies, as only the most recent active audio call is transferred from PS to CS audio, the SCC AS all drop all other previously existing audio session from this UE and indicate them accordingly in the SDP Offer sent within SIP re-INVITE requests towards the remote UE.

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 static STN, 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 inactive session transfer request has not been received within a time given by local policy after the reception of the SIP INVITE request due to static STN;

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 static STN 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 static STN;

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 audio sessions; and

2. the SCC AS does not initiate release for Source Access Leg of the associated SIP dialogs but remove speech media flow for these dialogs.

Removing speech media flow for SIP dialogs, the SCC AS shall:

- if the speech media flow was the only media flow in the SIP dialogs, release the source access leg as specified in 3GPP TS 24.229 [2]; or

- if the SIP dialogs contains other media flows than the speech flow, modify the source access leg and remove the speech media flow, as specified in 3GPP TS 24.229 [2].

The SCC AS shall consider session with speech:

- which has "sendonly" or "inactive" directionality at the SC UE as inactive; and

- which has "recvonly" or "sendrecv" directionality at the SC UE as active;

in this subclause and in the referenced subclauses.

12.3.3 SCC AS procedures for SR-VCC, abnormal case

When the SCC AS receives a SIP reINVITE request on the original source access leg after 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.

When the SCC AS receives a SIP reINVITE request after having performed an access transfer that was triggered by a SIP INVITE request due to STN-SR, then the SCC AS shall:

1) not release the original access leg once the expiration of that timer as described in subclause 12.3.1; and

2) treat the SIP reINVITE request 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 reINVITE indicating success in removing all media components from the dialog that was created due to the SIP INVITE request due to STN-SR then the SCC AS shall send a SIP BYE request on this dialog, by following the rules of 3GPP TS 24.229 [2].

12.4 MSC server enhanced for ICS

If the MSC server supports MSC Server assisted mid-call services feature, the MSC server shall be an MSC server enhanced for ICS as specified in 3GPP TS 29.292 [18] and 3GPP TS 24.292 [4] with following modifications:

1. if two sessions are transferred, associate the SIP INVITE request for an additional inactive session with CS call with transaction identifier 1 and TI flag value as in mobile terminated call;

NOTE: The active session transaction identifier value 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; and

3. apply procedures as described in subclause 9.4.

The MSC server shall consider session with audio media:

- which has "sendonly" or "inactive" directionality as inactive; and

- which has "recvonly" or "sendrecv" directionality as active;

in this subclause and in the referenced subclauses.

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 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: The same E-STN-SR is used for all the emergency session access transfers within one PLMN.

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. an existing SIP session anchored at the EATF with the instance-id media feature tag provided by the SC UE in the Contact header field at session establishment equal to the instance-id media feature tag included in the Contact header field of the received SIP INVITE request. If no source access leg 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, the EATF shall release the source access leg as described in 3GPP TS 24.229 [2].

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 speech session to be transferred; and

2) set the P-Asserted-Identity header field to the Correlation MSISDN.

If the MSC server enhanced for SRVCC using SIP interface applies the MSC Server assisted mid-call features then in addition to the procedures in this subclause it shall support the procedures defined in subclause 12.4.

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 speech session to be transferred;

2) include the instance-id 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; and

3) set the P-Asserted-Identity header field to the Correlation MSISDN if one is available.

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 shall 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 there is a single full-duplex speech session over CS. If there is more than one full-duplex speech session, the SC UE shall release all the ongoing sessions that are not currently active before attempting the procedures described in this section.

If 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 an audio stream 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 the SC UE receives a new SIP INVITE request containing an audio stream 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-duplex speech session 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 an audio stream 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 Roles for UE discovery for inter-UE transfer

14.1 Introduction

This clause specifies the target UE discovery procedures for UEs that are candidate UEs for inter UE transfer. The list of candidate UEs is a contact list such as name of the UE, which is represented in SIP through the use of SIP contact or the instance-id. The subscription of candidate UEs may be configured such that the private user identities associated with the UEs involved in inter UE transfer share the same set of implicitly registered public user identities.

14.2 SC UE

The target UE discovery procedures include the registration status (active, inactive), and the UE capabilities (e.g. support of audio/video formats etc.).

In order to determine a list of UEs sharing the same set of implicitly registered public user identities and their capability information, the SC UE subscribes to the reg-event package as described in 3GPP TS 24.229 [2] in subclause 5.1.1.3.

NOTE 1: In order to allow inter UE transfer to UEs belonging to the same subscription but belonging to a different set of implicitly registered public user identities, a user can have a static list of UEs that is manually administered by the user and stored locally in the user's device (e.g. phone book). Having a static list is an implementation in the UE and has no impact on the standards.

NOTE 2: If the UE is not part of the same set of implicitly registered public user identities as the SC UE, or if the SC UE was unable to obtain the capability information of the UE through the use of reg-event package, the SC UE can send a SIP OPTIONS request to the UE to attempt to retrieve capability information. In order to avoid a lot of transactions, a SIP OPTIONS request is generated based on an action initiated by the user (e.g. after the user has finished adding a new UE in the static list, or the user explicitly asks for getting UE capability information).

14.3 SCC AS

The information of UEs that belong to the same subscription is required at the SCC AS for the purpose of authorizing that the requested inter UE transfer to the UE is allowed, i.e. to prevent the SC UE from performing Inter UE Transfer to a UE under a different subscription.

The SCC AS obtains all the public user identities associated with the user's subscription from the Sh interface as specified in 3GPP TS 29.328 [6] and 3GPP TS 29.329 [7].

NOTE 1: Getting the public user identities over the Sh interface allows the SCC AS to receive information of UEs sharing the same set of implicitly registered public user identities and information of UEs within the same subscription that are not in the same set of implicitly registered public user identities. This is needed to authorize the static list of UEs that is manually administered by the user and stored locally in the user's device.

The SCC AS can obtain the registration information (e.g. GRUU) by the following methods:

1. using the 3rd-party SIP REGISTER request as described in 3GPP TS 24.229 [2] in subclause 5.4.1.7; or
2. the SCC AS subscribes to the reg-event package as described in 3GPP TS 24.229 [2] in subclause 5.4.2.1.1.

NOTE 2: The SCC AS needs to know the public user identity for the authorization of the SIP REFER request for inter UE transfer. To get the public user identity from the public GRUU, the SCC AS can simply remove the "gr" parameter from the public GRUU. Using the 3rd-party registration or subscribing to the reg-event package allows the SCC AS to find the temporary GRUU of the UE and to correlate the GRUU with the public user identities. After correlation, the SCC AS would make a list of GRUUs that are associated with the same subscription and/or with the same set of implicitly registered public user identities.

15 Roles for inter-UE transfer without establishment of collaborative session

15.1 Introduction

This clause specifies the procedures for transferring all media of an existing session from one UE to another UE of the same subscriber. Procedures are specified for the transferor SC UE, the transferee SC UE and the SCC AS.

15.2 SC UE

15.2.1 Transferor SC UE in services defining only originating session set up in UE

In order to transfer all media of an existing session from this UE to another UE of the same subscriber, the SC UE shall send a SIP REFER request as specified in IETF RFC 3515 [13] and in accordance with UE procedures specified in 3GPP TS 24.229 [2]. The SC UE shall populate the SIP REFER request as follows:

1. the Request-URI set to the URI of the UE where the session is to be transferred to;

NOTE: The URI of the UE needs to be a GRUU if several UEs share the same public user identity.

2. the Refer-To header field set to the Inter UE Transfer SCC AS URI and extended with the following URI header fields:

   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 Access Leg between this UE and the SCC AS; 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 Access Leg between this UE and the SCC AS; and

      b. the Require header field populated with the option tag value "tdialog"; and

   C. if the session is established using an IMS communication service that requires the use of an IMS communication service identifier:

      a. optionally the Accept-Contact header field with the g.3gpp.icsi-ref media feature tag containing the IMS communication service identifier of the existing session; and

      b. the P-Preferred-Service header field set to the IMS communication service identifier of the existing session; and

3. include in the Contact header field:

   A. a public GRUU or temporary GRUU as specified in 3GPP TS 24.229 [2].

If the SC UE receives any SIP 4xx – 6xx response to the SIP REFER request or if the SC UE receives a SIP NOTIFY request containing a message/sipfrag body with SIP 4xx – 6xx response, then the inter UE transfer has not completed successfully.

15.2.2 Transferee SC UE in services defining only originating session set up in UE

When sending a SIP INVITE request upon SIP REFER request reception in accordance with UE procedures specified in 3GPP TS 24.229 [2] and IETF RFC 3515 [13], the SC UE shall populate the SIP INVITE request with header fields which were included as URI header fields in the URI in the Refer-To header field of the received SIP REFER request.

15.3 SCC AS

15.3.1 Distinction of requests sent to the SCC AS

The SCC AS needs to distinguish between the following SIP requests to provide specific functionality relating to inter UE transfer:

1. SIP INVITE request routed to the SCC AS upon originating or terminating filter criteria containing the Inter UE Transfer SCC AS URI in the Request-URI and a STI belonging to the same subscriber in:

   A. the Target-Dialog header field; or

   B. in the Replaces header field

   with at least one offered media used in session by a UE other than the UE identified by the Contact header field value. In the procedures below, such request is known as "SIP INVITE request due to inter UE transfer".

2. SIP REFER request routed to the SCC AS due to the originating filter criteria where:

   A. the GRUU in the Contact header field identifies a UE of the user identified in P-Asserted-Identity header field;

   B. the GRUU in the Contact header field identifies a UE of the same subscriber as the SIP URI in the Request-URI; and

   C. the dialog identifier in:

      a. the Replaces URI header field of the URI in the Refer-To header field; or

      b. the Target-Dialog URI header field of the URI in the Refer-To header field

      belongs to a session of the UE identified by the GRUU in the Contact header field.

   In the procedures below, such request is known as "SIP REFER request due to inter UE transfer".

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].

15.3.2 Inter UE transfer request authorization in services defining only originating session set up in UE

Upon receiving a SIP REFER requests due to inter UE transfer, the SCC AS shall:

1. reject the SIP request with a SIP 403 (Forbidden) response and do not process the remaining steps if:

   A. media of the session one leg of which is identified by the dialog identifier in:

      a. the Replaces URI header field of the URI in the Refer-To header field; or

      b. the Target-Dialog URI header field of the URI in the Refer-To header field;

      are delivered to two or more UEs of the subscriber; or

   B. at least one media of the session identified by the dialog identifier in:

      a. the Replaces URI header field of the URI in the Refer-To header field; or

      b. the Target-Dialog URI header field of the URI in the Refer-To header field

      is delivered to a UE other than the UE identified by the GRUU in the Contact header field;

2. insert a Record-Route header field with SCC AS own address; and

3. forward the SIP REFER request in any manner conformant with 3GPP TS 24.229 [2].

The SCC AS shall forward the SIP response to the SIP REFER request, the SIP NOTIFY request, and the SIP response to the NOTIFY request in any manner conformant with 3GPP TS 24.229 [2].

15.3.3 SCC AS procedures for inter UE transfer in services defining only originating session set up in UE

Upon receiving a SIP INVITE request due to inter UE transfer, the SCC AS shall:

1. reject the SIP request with a SIP 403 (Forbidden) response and do not process the remaining steps if:

   A. the SCC AS is not aware of a subscription created by a SIP REFER request with the dialog identifiers:

      a. in the Replaces header field of the received SIP INVITE request and in the Replaces URI header field of the URI in the Refer-To header field of the SIP REFER request are equal; or

      b. in the Target-Dialog header field of the received SIP INVITE request and in the Target-Dialog URI header field of the URI in the Refer-To header field of the SIP REFER request are equal;

2. associate the received SIP INVITE request with an ongoing SIP dialog by matching the dialog identifier in the Replaces header field or the Target-Dialog header field. 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;

3. if dialog identifier is not included in either in the Replaces header field or in the Target-Dialog header field or if the included dialog identifier does not identify an existing ongoing dialog, send a SIP 480 (Temporarily Unavailable) response to reject the SIP INVITE request and not processes the remaining steps;

4. identify the Source Access Leg by the dialog identifier present in the Replaces or the Target-Dialog header field of the SIP INVITE request;

5. if a media used in the Source Access Leg session is not offered in the SDP offer of the SIP INVITE request then reject the SIP request with a SIP 403 (Forbidden) response and do not process the remaining steps;

6. 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]; and

7. 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:

   A. include a new SDP offer including the media characteristics as received in the SIP INVITE request, by following the rules of the 3GPP TS 24.229 [2].

Upon receiving the SIP ACK request originated from the SC UE, the SCC AS shall initiate release of the Source Access Leg by sending a SIP BYE towards the SC UE in accordance with 3GPP TS 24.229 [2].

16 Roles for collaborative session establishment for inter-UE transfer

16.1 Introduction

This clause specifies the roles of controller UE, controllee UE and the SCC AS when controller UE transfers media to a controllee UE or adds a new media to the controllee UE.

16.2 SC UE

16.2.1 SC UE procedures for collaborative session establishment by transferring media used in an existing session

16.2.1.1 Controller UE procedures

To establish a collaborative session by transferring the media component, the controller UE shall send a SIP REFER request outside the existing dialog as specified in IETF RFC 3515 [13] and include:

1) the Request-URI set to the Inter UE Transfer SCC AS URI;

2) the Refer-To header field set as follows:

   a) the SIP URI of the controllee UE;

NOTE: The SIP URI of the controllee UE needs to be a GRUU if the controllee UE and any other UEs share the same public user identity.

   b) the body parameter containing SDP for the media (m=) lines in the session set as follows:

      - media lines that are not being transferred with the port number set to zero

      - media line(s) that are to be transferred containing the port number for the corresponding media received in the media line of the SDP received during the last successful SDPoffer answer exchange;

3) the Accept header field containing the MIME type "message/sipfrag";

4) the Target-Dialog header field containing the dialog parameters for the dialog of the existing session; and

5) the Contact header field containing the g.3gpp.iut-controller media feature tag as described in annex C.

The controller UE shall handle any response to the SIP REFER request and the subsequent SIP NOTIFY requests according to 3GPP TS 24.229 [2] and IETF RFC 3515 [13]. The controller UE shall save the media information (e.g. media line(s) and port number(s)) related to the transferred media(s) received in the sipfrag body of the SIP NOTIFY requests in order to perform further inter-UE transfer operations on the controllee UE. When the controller UE receives a SIP re-INVITE request from the SCC AS to update the status of the transferred media component after a successful transfer, the controller UE shall follow the procedures described in 3GPP TS 24.229 [2], including in the Contact header field of the SIP 200 (OK) response the g.3gpp.iut-controller media feature tag as described in annex C.

If an error response is received for the SIP REFER request or the subsequent SIP NOTIFY requests include a non-2xx final response, the controller UE shall consider the transfer operation failed and continue the existing session.

16.2.1.2 Controllee UE procedures

There are no specific procedures for the controllee UE for the collaborative session establishment by transferring media, besides the procedures described in 3GPP TS 24.229 [2].

16.2.2 SC UE procedures for collaborative session establishment with new media

16.2.2.1 Controller UE procedures

The controller UE may establish a collaborative session with a new media at anytime while it has an ongoing IMS established session according to 3GPP TS 24.229 [2] with a remote UE.

The controller UE shall add the new media by sending a SIP REFER request outside the existing dialog as specified in IETF RFC 3515 [13] and include:

1) the Request-URI set to the Inter UE Transfer SCC AS URI;

2) the Refer-To header field set as follows:

   a) the SIP URI of the controllee UE;

NOTE 1: The SIP URI of the controllee UE needs to be a GRUU if the controllee UE and other UEs share the same public user identity.

   b) the body parameter containing SDP for the media (m=) lines in the session shall be set as follows:

      - media lines that are not being transferred with the port number set to zero

      - media lines that are to be added containing the discard port number "9" for the media line(s) to be transfered

NOTE 2: The discard port number "9" indicates that this port number should be ignored.

3) the Accept header field containing the MIME type "message/sipfrag";

4) the Target-Dialog header field containing the dialog parameters for the dialog of the existing session; and

5) the Contact header field containing the g.3gpp.iut-controller media feature tag as described in annex C.

The controller UE shall handle any response to the SIP REFER request and the subsequent SIP NOTIFY requests according to 3GPP TS 24.229 [2] and IETF RFC 3515 [13]. The controller UE shall save the media information (e.g. media line(s) and port number(s)) received in the sipfrag body of the SIP NOTIFY requests in order to perform further inter-UE transfer operations on the controllee UE.

If error response is received for the SIP REFER request or the subsequent SIP NOTIFY requests include a non-2xx final response, the controller UE shall consider the transfer operation failed and continue the existing session.

16.2.2.2 Controllee UE procedures

There are no specific procedures for the controllee UE for the collaborative session establishment by adding media, besides the procedures described in 3GPP TS 24.229 [2].

16.2.3 SC UE procedures for subscription to the session description of a collaborative session

In order to get the information about the changes of session description of the collaborative session, the controller UE shall subscribe to the dialog event package for the dialog formed between the controller UE and the SCC AS within collaborative session by setting the Target-Dialog header in the SIP SUBSCRIBE request to the associated call-id, from-tag, and to-tag and including the event package parameter "include-session-description" according to IETF RFC 4235 [48]. The controller UE shall set the expire header of the SIP SUBSCRIBE request to either:

- zero to receive one subsequent SIP NOTIFY request; or

- different than zero to receive the subsequent SIP NOTIFY requests.

When the controller UE receives subsequent SIP NOTIFY requests, the controller UE shall update its information about the session description of the collaborative session by the content of that subsequent SIP NOTIFY request. The controller UE shall handle response to the SIP SUBSCRIBE request and the possible subsequent SIP NOTIFY request according to 3GPP TS 24.229 [2] and IETF RFC 4235 [48].

16.3 SCC AS

16.3.1 Distinction of requests sent to the SCC AS

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

1) SIP REFER requests routed to the SCC AS containing:

   a) the SCC AS address in the Request-URI; and

   b) the Refer-To header field including:

      - the address of a UE which is neither the UE which sent the SIP REFER request, nor the remote UE, but which is within the list of UEs which can be involved within an collaborative session with the UE which originated the SIP REFER request;

      - zero or more media lines which indicate the port value 0; and

      - one or more media lines which indicate the discard port value 9.

In the procedures below, such SIP REFER requests are called "SIP REFER requests for establishing new media at controllee UE". 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.

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 [8].

Editor's Note: further types of REFER have to be determined.

16.3.2 SCC AS procedures for collaborative session establishment by transferring media

NOTE: If the controller UE is already involved in a collaborative session then the procedures in subclause 17.13.1 apply.

Editor's Note: The text in this first line will need to be aligned with a future change request that will provide the procedures for determination of when to invoke this subclause.

When the SCC AS establishes a collaborative session by transferring media as a result of receiving a SIP REFER request from the controller UE , the SCC AS shall send:

1) a SIP 202 (Accepted) response to the SIP REFER request and a SIP NOTIFY request containing a sipfrag "SIP 100 Trying" to the controller UE as specified in IETF RFC 3515 [13]; and

2) a SIP INVITE request to controllee UE, containing:

   a) Request-URI with SIP URI from the Refer-To header field of the received SIP REFER request;

   b) the Referred-By header field containing the URI from the P-Asserted-Identity header field of the received SIP REFER request;

   c) the P-Asserted-Identity header field containing the identity of the remote UE as received in the P-Asserted-Identity header field from the remote UE at the original session establishment; and

   d) the SDP information for the media component to be transferred as follows:

      - The media (m=) lines from the body parameter in the Refer-To header field of the received SIP REFER request; and

      - for media lines which have non zero port nunmbers the SDP parameters from the corresponding media lines as received during the last successful SDPoffer-answer exchange from the remote UE.

Upon receiving a SIP final response from the controllee UE, the SCC AS shall send, a SIP NOTIFY request containing the received final response code in the sipfrag body and if the received response was a SIP 200 (OK) response containing an SDP answer then also include in the sipfrag the Content-Type header field from the received 200 (OK) response along with the media (m=) lines from the SDP answer.

If the SIP final response was a non 2xx response then the SCC AS shall consider the transfer operation failed and abort the media transfer and establishment of the collaborative session.

If the SIP final response was a SIP 200 (OK) response containing a SDP answer, the SCC AC shall send a SIP re-INVITE request on the dialog for the remote leg to the remote UE as sepcified in 3GPP TS 24.229 [2]. The SCC AS shall:

1) send a SIP re-INVITE request containing SDP information as follows:

   a) for the transferred media component(s), set the SDP information as from the SDP answer received in the SIP 200 (OK) response from the controllee UE;

   b) for all other media components in the collaborative session, include the SDP information as from the original session to the remote leg.

2) send a SIP re-INVITE request to the controller UE following the procedures described in 3GPP TS 24.229 [2] to remove the media for the transferred media component.

Upon receiving a SIP 200 (OK) response with the SDP answer on the remote leg, the SCC AS shall send a SIP ACK request on the remote leg.

16.3.3 SCC AS procedures for collaborative session establishment with new media

When SCC AS receives a SIP REFER request in a new dialog from the contoller UE for establishing a collaborative session by adding new media to the controllee UE, the SCC AS shall send:

1) a SIP 202 (Accepted) response to the controller UE;

2) a SIP NOTIFY request containing a sipfrag "SIP 100 Trying" as described in IETF RFC 3515 [13] to the controller UE; and

3) SIP INVITE request in accordance to 3GPP TS 24.229 [2] to the controllee UE. The SCC AS shall construct the SIP INVITE request as follows:

   - Request-URI set to the SIP URI from the Refer-To header field of the received SIP REFER request;

   - includes an SDP offer and sets all the m-lines to zero except the m-line of the media component is to be added as the last m-line in the SDP; and

   - if the m-line in the received SDP of the SIP REFER request is set to the discard port number "9", the SCC AS shall include an a-line set to inactive in the SDP offer.

Editor's note: The value of c-line is FFS.

If a SIP non-2xx final response is received from the controllee UE, the SCC AS shall send a SIP NOTIFY request including the SIP final response as a sipfrag body to the controller UE and consider the inter-UE transfer operation failed. Otherwise, the SCC AS continues with the remainder of the steps described in this subclause.

Upon receiving a SIP response from the controllee UE with an SDP answer, the SCC AS shall send a SIP re-INVITE request to the remote UE as specified in 3GPP TS 24.229 [2]. The SCC AS shall construct the SDP offer in the SIP re-INVITE request as follows:

1) Request-URI with SIP URI from the Refer-To header field of the received SIP REFER request;

2) the Referred-By header field containing the URI from the P-Asserted-Identity header field of the received SIP REFER request;

3) the P-Asserted-Identity header field containing the identity of the remote UE as received in the P-Asserted-Identity header field from the remote UE at the original session establishment; and

4) the SDP information from the body parameter in the Refer-To header field of the received SIP REFER request and for media lines which have non zero port nunmbers the SDP parameters forthe corresponding media lines as received during the last successful SDP offer-answer exchange with the remote UE.

If the SIP final response was a non 2xx response then the SCC AS shall consider the transfer operation failed and abort the media transfer and establishment of the collaborative session.

If the SIP final response from the remote UE was a SIP 200 (OK) response with the SDP answer, the SCC AS shall:

1) send to the remote UE a SIP ACK request; and

2) send to the controllee UE a SIP re-INVITE request containing the current port number for the media component to be added.

NOTE 1: Any other changes such as IP address of the remote leg in case remote leg uses different IP addresses for different media components can also be updated in the SIP re-INVITE request.

Upon successful completion of the SDP offer answer exchange using SIP re-INVITE request with the controllee UE, the SCC AS shall:

NOTE 2: This SIP re-INVITE request is triggered by the SIP REFER request. The previous SIP INVITE request was generated by the SCC AS due to third party call control to allow sending this SIP re-INVITE request.

1) send to the controllee UE a SIP ACK request

Editor's note: This is FFS how to inform the controller UE about current port number of the remote UE or any other changes such as IP address.

Upon receiving a SIP final response from the controllee UE, the SCC AS shall send, a SIP NOTIFY request containing the received final response code in the sipfrag body and if the received response was a SIP 200 (OK) response containing an SDP answer then also include in the sipfrag the Content-Type header field from the received 200 (OK) response along with the media descriptions (m=) from the SDP answer.

16.3.4 Procedures to indicate SCC AS establishing a collaborative session by adding a new media

When SCC AS receives a SIP REFER request in a new dialog from the contoller UE with:

1) the Request-URI set to the Inter UE Transfer SCC AS URI; and

2) the Refer-To header field set to GRUU of a controllee UE and URI parameter containing SDP listing:

   - all the media components with associated information in the session; and

   - a new media component with associated port number set to the discard port number "9" as the last media component,

the SCC AS shall follow the procedure in subclause 16.3.2 to establish a collaborative session with the controllee UE.

16.3.5 SCC AS procedures for subscription to the session description of a collaborative session

When the SCC AS receives a SIP SUBSCRIBE request with Target-Dialog header field containing the call-id, from-tag, and to-tag associated to the dialog between the controller UE and the SCC AS and including the event package parameter "include-session-description", the SCC AS shall send session description in SIP NOTIFY requests towards the UE according to IETF RFC 4235 [48].

17 Roles for media transfer within collaborative session for inter-UE transfer

17.1 Introduction

This clause specifies the roles of the controller UE, the controllee UE and the SCC AS when media transfer from the controller UE to a controllee UE or from a controllee UE to another controllee UE within a collaborative session.

17.2 SC UE

17.2.1 Procedures for controller UE initiated media transfer from controller UE to controllee UE

17.2.1.1 Controller UE procedures

The SC UE procedures for media transfer from controller UE to a controllee UE is the same as the procedures described in subclause 16.2.1.1 with exception that the controller UE additionally sets to the port numbers of the media components (in the body header field of the URI in the Refer-To header field of the SIP REFER request) which are being used at the controllee UE to values from the corresponding media lines received during the last successful SDP offer and answer exchange with the remote party.

17.2.1.2 Controllee UE procedures

There are no specific procedures for the controllee UE for media transfer from controller UE to controllee, besides the procedures described in 3GPP TS 24.229 [2].

17.2.2 Procedures for controller UE initiated media transfer from controllee UE to another controllee UE

17.2.2.1 Controller UE procedures

To transfer a media component within a collaborative session from one controllee UE to another controllee UE, the controller UE shall send a SIP REFER request outside the existing dialog as specified in IETF RFC 3515 [13] and include:

1) the Request-URI set to the Inter UE Transfer SCC AS;

2) the Refer-To header field set as follows:

   a) the SIP URI of the controllee UE to which the media m-lines are to be transferred; and

NOTE: The SIP URI of the controllee UE needs to be a GRUU if the controllee UE and any other UEs share the same public user identity.

   b) the body parameter containing SDP for the media m-lines in the session set as follows:

      - media lines which are not served by the target controllee UE and which are not being transferred with the port numbers set to zero;

      - media lines which are already served by the target controllee UE, therefore are not to be transferred containing the port numbers of the remote UE; and

      - media lines which are to be transferred containing the port numbers of the remote UE.

3) the Accept header field containing the MIME type "message/sipfrag"; and

4) the Target-Dialog header field containing the dialog parameters for the dialog of the collaborative session.

The controller UE shall handle the subsequent SIP NOTIFY requests to the SIP REFER request according to 3GPP TS 24.229 [2], IETF RFC 3515 [13].

The controller UE shall save the media information (e.g. media line(s) and port number(s)) related to the media(s) received in the sipfrag body of the SIP NOTIFY requests in order to perform further inter-UE transfer operations on the controllee UE.

If an error response is received for the SIP REFER request or the subsequent SIP NOTIFY requests include a non-2xx final response, the controller UE shall consider the transfer operation failed and continue the existing session.

17.2.2.2 Controllee UE procedures

There are no specific procedures for the controllee UE for transferring media form one controllee UE to another controllee UE, besides the procedures described in 3GPP TS 24.229 [2].

17.3 SCC AS

17.3.1 Procedures for controller UE initiated media transfer from controller UE to controllee UE

The SCC AS procedures for media transfer from the controller UE to a controllee UE is the same as the procedures described in subclause 16.3.2 with the exception that upon receipt of a SIP REFER request from the controller UE the SCC AS sends a SIP re-INVITE request instead of a SIP INVITE request to the controllee UE. The SIP re-INVITE request is within the dialog established when establishing the collaborative session with the controllee UE.

17.3.2 Procedures for controller UE initiated media transfer from controllee UE to another controllee UE

When the SCC AS maintaining a collaborative session and transferring media as a result of receiving a SIP REFER request from one controllee UE to another controllee UE, the SCC AS shall send:

1) a SIP 202 (Accepted) response to the SIP REFER request and a SIP NOTIFY request containing a sipfrag "SIP 100 Trying" to the controller UE as specified in IETF RFC 3515 [13] for the dialog on which the SIP REFER request was received; and

2) a SIP re-INVITE request to controllee UE, to which the media component(s) is to be transferred, containing the SDP information for the media component to be transferred as follows:

   a) the media descriptions (m) from the body parameter in the Refer-To header field of the received SIP REFER request including the associated attributes (a) set to sendonly and the associated bandwidth information (b) with RS set to zero and RR set to zero for those media compontents which are to be transferred to the target controllee UE; and

   b) for media descriptions (m) which have non zero port numbers the SDP parameters from the corresponding media descriptions (m) as received during the last successful SDP offer-answer exchange from the remote UE.

Upon receipt of a 2xx response for the re-INVITE request sent to the controllee UE to which the media component is to be transferred, the SCC AS shall:

1) send a SIP re-INVITE request to controllee UE, from which the media component(s) is to be transferred, containing the SDP information for the media component to be transferred as follows

   a) the media descriptions (m) from the body parameter in the Refer-To header field of the received SIP REFER request including the associated attributes (a) set to sendonly and the associated bandwidth information (b) with RS set to zero and RR set to zero for those media compontents which are to be transferred to the target controllee UE; and

   b) for media descriptions (m) which have non zero port numbers the SDP parameters from the corresponding media descriptions (m) as received during the last successful SDP offer-answer exchange from the remote UE.

If a 2xx response was received to the re-INVITE request sent to the controllee UE from which the media component(s) is to be transferred, the SCC AC shall send a SIP re-INVITE request to the remote UE containing SDP body indicating:

1) for the transferred media component(s), set the SDP information as received from the SDP answer in the SIP 2xx response from the controllee UE to which the media component is to be transferred with the difference that the attributes (a) is set to directionality used by the controlee UE from which the media component was transferred;

2) for all other media components in the collaborative session, include the associated media descriptions (m) as from the original session to the remote leg.

Upon receipt of a 2xx response for the re-INVITE request sent to the remote UE, the SCC AS shall:

1) if the the transferred media component was the only media component active at the controllee UE from which the media component was transferred from, send a SIP BYE request to the controllee UE from which the media was transferred from; or

2) if after the transfer of the media component the controllee UE from which the media was transferred from still has other media components within the collaborative session, send a SIP re-INVITE request to the controllee UE, from which the media component was transferred, following the procedures described in 3GPP TS 24.229 [2] and set the port value of the associated media descriptions (m) for the transferred media to 0.

Upon receipt of a 2xx response for the re-INVITE request or the SIP BYE request sent to the controllee UE from which the media was transferred the SCC AS shall send a SIP re-INVITE request to the controllee UE to which the media was transferred following the procedures described in 3GPP TS 24.229 [2] and setting the attributes (a) associated to the transferred media component to its previous status before the inter-UE Transfer operation for the transferred media components.

Upon receiving a final response to the SIP re-INVITE request which was sent towards the controllee UE to set the attributes (a) associated to the transferred media component to its previous status before the inter-UE Transfer operation of the transferred media components, the SCC AS shall:

1) send a SIP NOTIFY request containing the received final response code in the sipfrag body to the controller UE;

2) if the received response to the SIP re-INVITE is a SIP 2xx response containing an SDP answer, include within the sipfrag body

   a) the Content-Type header field from the received SIP 2xx response; and

   b) the SDP answer.

If any final response to a SIP re-INVITE request (apart from the SIP re-INVITE request which was sent towards the controllee UE to set the attributes (a) associated to the transferred media component to its previous status) was a 3xx or a 6xx response then the SCC AS shall consider the Inter-UE Transfer operation failed and shall send the SIP NOTIFY request to controller UE with the body populated with SIP/2.0 603 Declined.

18 Roles for release of collaborative session for inter-UE transfer

18.1 Introduction

This clause specifies the roles of controller UE, controllee UE and the SCC AS when controller UE or the remote UE releases the collaborative session.

18.2 SC UE

18.2.1 Procedures for collaborative session release by controller UE

18.2.1.1 Controller UE

There are no specific procedures for the controller UE for the collaborative session release besides the procedures described in 3GPP TS 24.229 [2].

18.2.1.2 Controllee UE

There are no specific procedures for the controllee UE for the collaborative session release besides the procedures described in 3GPP TS 24.229 [2].

18.2.2 Procedures for collaborative session release by remote party

18.2.2.1 Controller UE

There are no specific procedures for the controller UE for the collaborative session release besides the procedures described in 3GPP TS 24.229 [2].

18.2.2.2 Controllee UE

There are no specific procedures for the controllee UE for the collaborative session release besides the procedures described in 3GPP TS 24.229 [2].

18.3 SCC AS

18.3.1 Procedures for collaborative session release by controller UE

When the SCC AS receives a SIP BYE request from the controller UE for releasing a collaborative session, the SCC AS shall send a SIP BYE request according to the procedures described in 3GPP TS 24.229 [2] to the remote UE and to all controllee UEs within that collaborative session.

18.3.2 Procedures for collaborative session release by remote party

When the SCC AS receives a SIP BYE request from the remote party for releasing a collaborative session, the SCC AS shall send a SIP BYE request according to the procedures described in 3GPP TS 24.229 [2] to the controller UE and to all controllee UEs within that collaborative session.

19 Roles for media adding/deleting within collaborative session for inter-UE transfer

19.1 Introduction

This clause specifies the roles of the controller UE, the controllee UE and the SCC AS when the controller UE or the remote UE adds or releases media to the collaborative session.

19.2 SC UE

19.2.1 Procedures for adding new media on controllee UE by controller UE

The SC UE procedures for adding new media to a controllee UE by the controller UE is the same as the procedure described in subclause 16.2.2.1 with exception that the controller UE additionally sets the port numbers of the media components (in the body header field of the URI in the Refer-To header field of the SIP REFER request) which are being used at the controllee UE to values from the corresponding media lines received during the last successful SDP offer and answer exchange with the remote party.

19.2.2 Procedures for releasing media on controllee UE by controller UE

The controller UE may release one or more media components on a controllee UE within a collaborative session while it has an ongoing IMS session with a remote UE.

The controller UE shall release the media by sending a SIP REFER request for releasing media component, including:

1) the Request-URI set to the Inter UE Transfer SCC AS URI;

2) the Refer-To header field as follows:

   a) the SIP URI of the controllee UE;

NOTE: The SIP URI of the controllee UE needs to be a GRUU if the controllee UE and any other UEs share the same public user identity.

   b) the body parameter containing SDP for the media (m=) lines in the session shall be set as follows:

      - media lines that are not being released with their port numbers; and

      - media line(s) that are to be released with the port number set to zero;

3) the Accept header field set to "message/sipfrag";

4) the Target-Dialog header field populated as specified in IETF RFC 4538 [11], containing the dialog identifier of the dialog between the SCC AS and the controller UE;and

5) the Contact header field containing the g.3gpp.iut-controller media feature tag as described in annex C.

The controller UE shall handle SIP response to the SIP REFER request and the subsequent SIP NOTIFY requests according to 3GPP TS 24.229 [2]. The controller UE shall save the media information (e.g. media line number) received in the sipfrag body of the SIP NOTIFY request in order to perform further inter-UE transfer operations on the controllee UE.

19.2.2A Procedures for releasing media on controller UE by controller UE

If the controller UE wants to release a media component on the controller UE within a collaborative session, the controller UE shall follow the procedures defined in 3GPP TS 24.229 [2] for removing media with the following differences:

1. include the SDP information for all other media components within the collaborative session in the SIP re-INVITE request;

2. set all the ports number of the media on the controllee UEs with value zero; and

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

19.2.2B Procedures for controller UE to remove a controllee UE from the collaborative session

The controller UE may remove a controllee UE from a collaborative session while it has an ongoing IMS session with a remote UE.

The controller UE shall remove the controllee UE from the collaborative session by sending a SIP REFER request, including:

1) the Request-URI set to the Inter UE Transfer SCC AS URI;

2) the Refer-To header as follows:

   a) the SIP URI of the controllee UE;

NOTE: The SIP URI of the controllee UE needs to be a GRUU if the controllee UE and any other UEs share the same public user identity.

   b) the method parameter set equal to "BYE";

3) the Accept header field set to include "message/sipfrag"; and

4) the Target-Dialog header field populated as specified in IETF RFC 4538 [11], containing the dialog identifier of the dialog between the SCC AS and the controller UE.

The controller UE shall handle response to the SIP REFER request and the subsequent SIP NOTIFY requests according to 3GPP TS 24.229 [2].

19.2.3 Procedures for releasing media component by controllee UE

19.2.3.1 Controller UE

When controller UE receives a SIP re-INVITE request, it can contain an SDP offer with:

- media lines already belonging to the controller UE with non-zero port numbers;

- media line(s) belonging to the controllee UE which is releasing the media conponent(s) with non-zero port number(s); and

- media lines belonging to other controllee UEs, with port numbers set to zero.

Upon receiving such a SIP re-INVITE request, the controller UE shall follow the procedures described in 3GPP TS 24.229 [2] to accept or reject the released media component(s) by the controllee UE.

If the released media component(s) is the only media component(s) used within the collaborative session and the controller UE did not accept that media component, the controller UE shall release the collaborative session following the procedures described in 3GPP TS 24.229 [2].

19.2.3.2 Controllee UE

There are no specific procedures for the controllee UE for release of media component by controllee UE, besides the procedures described in 3GPP TS 24.229 [2].

19.2.4 Procedures for modifying media on controllee UE by itself

If the controllee UE wants to modify the characteristics of a media component on itself within a collaborative session, the controllee UE shall follow the procedures defined in 3GPP TS 24.229 [2] for modifying media.

19.2.5 Procedures for adding new media by remote UE when the controller UE does not alert the user

When controller UE receives a SIP re-INVITE request within an existing dialog from the remote UE to add a new media component on the collaborative session, the controller UE shall decide whether adding the new media on itself or adding it to one of its controllee UE.

If the controller UE decides to add the new media component on itself, the controller UE shall follow the precedure as specified in 3GPP TS 24.229 [2],

If the controller UE decides to add the new media component on one of its controllee UE, the controller UE shall send a SIP REFER request outside the existing dialog as specified in IETF RFC 3515 [13] and include:

1) the Request-URI set to the Inter UE Transfer SCC AS URI;;

2) the Refer-To header field, including:

   a) the SIP URI of the controllee UE where the media stream should be established from; and

NOTE: The SIP URI of the controllee UE needs to be a GRUU if the controllee UE and any other UEs share the same public user identity.

   b) the body parameter containing SDP for the media (m=) lines for the media lines in the session shall be set as follows:

      - media lines that are not being transferred with the port number set to zero;

      - media line(s) that are to be added at the controllee UE the same SDP information as in the SIP re-INVITE request received from the remote UE.

3) the Accept header field containing the MIME types "message/sipfrag";

4) the Target-Dialog header field containing the dialog parameters for the dialog of the existing session; and

5) the Contact header field containing the g.3gpp.iut-controller media feature tag as described in annex C.

The controller UE shall handle a SIP response to the SIP REFER request and the subsequent SIP NOTIFY requests according to 3GPP TS 24.229 [2]. Then the controller UE shall respond to the SIP re-INVITE request with a SIP 200 (OK) response with a SDP answer as specified in 3GPP TS 24.229[2] including in the Contact header field the g.3gpp.iut-controller media feature tag as described in annex C, and construct the SDP information in the SIP 200 (OK) response as follows:

1) set the port number of the new added media component with value zero; and

2) set all the ports number of the media on the controllee UEs with value zero; and

3) for other media components on the controller UE are not changed.

If an error response is received for the SIP REFER request or the subsequent SIP NOTIFY requests include a non-2xx final response, the controller UE shall consider the transfer operation failed and make a decision again whether adding the new media on itself or adding it on other controllee UE.

19.2.6 Procedures for releasing media by remote UE

Upon receipt of a SIP re-INVITE request from the remote party containing an SDP offer indicating one or more media components are to be released on the controller UE, the controller UE shall release the media component following the procedures specified in 3GPP TS 24.229 [2].

If the media component to be released is the last media component between the controller UE and the remote party and if there are more media components within the collaborative session, the controller UE shall not release the dialog with the SCC AS but set the port number(s) associated to the media line(s) to zero.

If the media component to be released is the last media component in the collaborative session, the controller UE shall follow the procedures described in subclause 18.2.1.1.

19.3 SCC AS

19.3.0 Distinction of requests at the SCC AS

When SCC AS receives a SIP REFER request within a new dialog from the controller UE with:

1) the Request-URI set to inter UE transfer SCC AS URI; and
2) the Refer-To header field set to SIP URI of a controllee UE and URI parameter containing SDP listing:
   - all the media components with associated information in the session; and
   - one or more new media components with associated port number set to the discard port number "9";

the SCC AS shall follow the procedure in subclause 19.3.1 to add the media component to the controllee UE.

When the SCC AS receives a SIP REFER request in a new dialog from the contoller UE with:

1) the Request-URI set to inter UE transfer SCC AS URI; and
2) the Target-Dialog header identifies and existing dialog between the SCC AS and the controller UE identified by the SIP URI in the Refer-To header field;
3) the Refer-To header containing the SIP URI of a controllee UE and a URI parameter containing SDP listing:
   - all the media components with associated information in the session; and
   - the media component is to be released with associated port number set to 0.

then the SCC AS shall follow the procedure in subclause 19.3.2 to release the media component from the controllee UE.

When the SCC AS receives a SIP REFER request in a new dialog from the contoller UE with:

1) the Request-URI parameter set to SIP URI of the SCC AS;
2) the Target-Dialog header identifies and existing dialog between the SCC AS and the controler UE identified by the SIP URI in the Refer-To header field; and
3) the Refer-To header containing the SIP URI of a controllee UE and the method parameter set equal to "BYE";

then the SCC AS shall follow the procedure in subclause 19.3.2B to remove the controllee UE.from the collaborative session.

19.3.1 Procedures for adding new media on controllee UE by controller UE

The SCC AS procedures for adding new media on controllee UE by the controller UE is the same as the procedure described in subclause 16.3.3 with exception that upon receipt of SIP REFER request from the controller UE, the SCC AS generates a SIP re-INVITE request within the dialog to the controllee UE instead of SIP INVITE request.

19.3.2 Procedures for releasing media on controllee UE by controller UE

When SCC AS receives a SIP REFER request in a new dialog from the controller UE containing a Refer-To header indicating that a SIP re-INVITE request is to be sent to remove one or more media components on a controllee UE, the SCC AS shall send:

1) a SIP 202 (Accepted) response to the controller UE;

2) SIP NOTIFY request with a sipfrag including SIP 100 (Trying) to the controller UE;

3) if the remote leg is sending media towards the controllee UE, send a SIP UPDATE request or a SIP re-INVITE request towards the remote leg as specified in 3GPP TS 24.229 [2] with the following clarifications:

   - if the controllee UE is sending media, include a "a=sendonly" attribute for the media component to be released;

   - if the controllee UE is only receiving media, include a "a=inactive" attribute for the media component to be released;

   - include b=RR:0 and b=RS:0 bandwidth modifiers as specified in IETF RFC 3556 [50] for the media component to be released; and

   - when the SIP 200 (OK) response to the SIP UPDATE request or the SIP re-INVITE request is received from the remote leg the SCC AS shall continue with the next steps;

NOTE: The steps in 3) are needed to avoid unnecessary ICMP message sending in the underlying IP network due to media sent to closed port that could result in the release of the call. The ICMP message is specified in IETF RFC 792 [51].

5) send a SIP re- INVITE request to the controllee UE, containing an SDP offer changed using the media types present in the body parameter in the URI in the Refer-To header.

Upon receiving a SIP 200 (OK) response from the controllee UE, the SCC AC shall send a SIP re-INVITE request to the remote leg as specified in 3GPP TS 24.229 [2]. The SCC AS shall construct the SDP information in the SIP re-INVITE request as follows:

1) set port number for removed media component to zero; and

2) include the SDP information for all other media in the collaborative session as from the original session to the remote leg.

Upon receiving SIP 200 (OK) response with the SDP answer from the remote leg, the SCC AS shall send:

1) a SIP ACK request to the remote leg;

2) upon successful release of the media component, a SIP NOTIFY request to the controller UEcontaining asipfrag body that shall include the SIP 200 (OK) response of the SIP re-INVITE request and also include the Content-Type header field from the received 200 (OK) response along with the SDP answer received from the controllee UE.

19.3.2A Procedures for releasing media on controller UE by controller UE

When SCC AS receives a SIP re-INVITE request within an existing dialog from the controller UE to remove a media component on itself, the SCC AS shall send a SIP re-INVITE request to the remote UE as specified in 3GPP TS 24.229 [2]. The SCC AS shall construct the SDP information in the SIP re-INVITE request as follows:

1) set port number for the removed media component to zero; and

2) include the SDP information for all other media in the collaborative session as received during the last successful SDP offer-answer exchange from the remote UE.

Upon receiving SIP 200 (OK) response with the SDP answer from the remote UE, the SCC AS shall send:

1) a SIP ACK request to the remote UE;

2) a SIP 200 (OK) response to the controller UE as specified in 3GPP TS 24.229 [2]. The SCC AS shall construct the SDP infromation in the SIP 200 (OK) response as follows:

   - set port number for the removed media component to zero; and

   - set all the ports number of the media on the controllee UEs with value zero.

19.3.2B Procedures for controller UE removing controllee UE from the collaborative session

When SCC AS receives a SIP REFER request in a new dialog from the controller UE containing a Refer-To header indicating that a SIP BYE request is to be sent to a controllee UE, the SCC AS shall send:

1) SIP 202 (Accepted) response to the controller UE;

2) SIP NOTIFY request with sipfrag including SIP 100 Trying to the controller UE; and

3) send a SIP BYE request to the controllee UE to release the controlled session in accordance with 3GPP TS 24.229 [2].

Upon receiving SIP 200 (OK) response from the controllee UE, the SCC AC shall send a SIP re-INVITE request to the remote leg as specified in 3GPP TS 24.229 [2]. The SCC AS shall construct the SDP information in the SIP re-INVITE request as follows:

1) set port number for removed media component to zero; and

2) include the SDP information for all other media in the collaborative session as from the original session to the remote leg.

Upon receiving SIP 200 (OK) response with the SDP answer from the remote leg, the SCC AS shall send:

1) a SIP ACK request to the remote leg;

2) upon successful release of the media component, a SIP NOTIFY request to the controller UE containing a sipfrag body that shall include the SIP 200 (OK) response of the SIP BYE request.

19.3.3 Procedures for releasing media component initiated by controllee UE

When the SCC AS receives a SIP re-INVITE request from a controllee UE containing an SDP offer with one or more media line(s) used on the controllee UE with port number set to zero, where the port number was previously not zero, the SCC AS shall send a SIP re-INVITE request towards the controller UE containing an SDP offer with the following details:

1) for the media lines on the controller UE, set the port numbers to the values received during the last successful SDP offer and answer exchange with the remote party;

2) for the media lines on the controllee UE which were offered with port numbers set to zero in the SDP offer received in the re-INVITE from the controllee UE, set the port numbers to the values received during the last successful SDP offer and answer exchange with the remote party; and

3) set the port number to zero for the remaining media lines.

When the SCC AS receives a SIP BYE request from a controllee UE, the SCC AS shall send a SIP re-INVITE request towards the controller UE containing an SDP offer with the following details:

1) for the media lines on the controller UE, set the port numbers to the values received during the last successful SDP offer and answer exchange with the remote party;

2) for the media lines on the controllee UE, set the port numbers to the values received during the last successful SDP offer and answer exchange with the remote party; and

3) set the port number to zero for the remaining media lines.

Upon receiving the SIP 200 (OK) response to the SIP re-INVITE request from the controller UE and in addition to the procedures of 3GPP TS 24.229 [2], the SCC AS shall send a SIP re-INVITE request towards the remote party containing an SDP offer with:

1) all the media lines which were not released by the controllee UE including their port numbers; and

2) all the media lines which were released by the controllee UE set to the port numbers received in the SDP answer contained in the SIP 200 (OK) response from the controller UE.

Upon receiving the SIP 200 (OK) response from the remote party, and in additon to the procedures of 3GPP TS 24.229 [2], the SCC AS shall:

1) if the transaction was initiated by a SIP re-INVITE request, send a SIP 200 (OK) response for the SIP re-INVITE request towards the controllee UE containing an SDP answer accepting the SDP offer from the controllee UE; or

2) if the transaction was initiated by a SIP BYE request, send a SIP 200 (OK) response for the SIP BYE request towards the controllee UE.

19.3.4 Procedures for modifying media on controllee UE by itself

When SCC AS receives a SIP re-INVITE request within an existing dialog from the controllee UE to modify a media component on itself, the SCC AS shall send a SIP re-INVITE request to the remote UE as specified in 3GPP TS 24.229 [2]. The SCC AS shall construct the SDP information in the SIP re-INVITE request as follows:

1) set the modified media information as the same in the SIP re-INVITE request received from the controllee UE; and

2) include the SDP information for all other media in the collaborative session as received during the last successful SDP offer-answer exchange from the remote UE.

Upon receiving SIP 200 (OK) response with the SDP answer from the remote UE, the SCC AS shall send:

1) a SIP ACK request to the remote UE; and

2) a SIP 200 (OK) response to the controllee UE with the SDP answer only contains the media component on the controllee UE.

19.3.5 Procedures for adding new media by remote UE when the controller UE does not alert the user

When SCC AS receives a SIP REFER request from the controller UE to add a new media component on a controllee UE, the SCC AS shall send:

1) SIP 202 (Accepted) response

2) SIP NOTIFY request containing a sipfig for a SIP 100 (Trying) response to the controller UE as described in IETF RFC 3515 [13];

3) if the target controllee UE has not been involved in the collaborative session, send a initial SIP INVITE request to the controllee UE to add the controlee UE in the collaborative session, containing:

   a) Request-URI with SIP URI from the Refer-To header field of the received SIP REFER request;

   b) the Referred-By header field containing the URI from the P-Asserted-Identity header field of the received SIP REFER request;

c) the P-Asserted-Identity header field containing the identity of the remote UE as received in the P-Asserted-Identity header field from SIP re-INVITE request received from the remote UE; and

d) the SDP information for the media component to be transferred as follows:

- The media (m=) lines from the body parameter in the Refer-To header field of the received SIP REFER request; and

- for media lines which have non zero port nunmbers the SDP parameters from the corresponding media lines as received in the SDP offer from the remote UE in the SIP re-INVITE request.

4) if there are other media component within the collaborative session between the target controllee UE and the remote UE, send a SIP re-INVITE request to the controllee UE, containing:

a) the SDP information for the media component to be transferred as follows:

- The media (m=) lines from the body parameter in the Refer-To header field of the received SIP REFER request; and

- for media lines which have non zero port nunmbers the SDP parameters from the corresponding media lines as received in the SDP offer from the remote UE in the SIP re-INVITE request

- for other media components that have already involved in the collabartive session are not changed.

Upon receiving a SIP final response from the controllee UE, the SCC AS shall send, a SIP NOTIFY request containing the received response code in the sipfrag body and if the received SIP response was a SIP 200 (OK) response containing an SDP answer then also include in the sipfrag the Content-Type header field from the received SIP 200 (OK) response along with the media (m=) lines from the SDP answer.

If the SIP final response was a non 2xx response then the SCC AS shall consider the transfer operation failed and abort the media transfer.

If the SIP final response was a SIP 200 (OK) response containing a SDP answer, the SCC AS shall:

1) send a SIP ACK request to the controllee UE;

2) send a SIP 200 (OK) response to the remote UE as specified in 3GPP TS 24.229 [2]. The SCC AS shall construct the SDP infromation in the SIP 200 (OK) response as follows:

- set the same SDP information for the new added media as in the SIP 200 (OK) response received from the controlee UE; and

- include the SDP information for all other media in the collaborative session as received during the last successful SDP offer-answer exchange from the remote UE.

19.3.6 Procedures for releasing media by remote UE

Upon receipt of a SIP re-INVITE request from the remote party containing an SDP offer indicating one or more media components to be released on the controller UE, the SCC AS shall set the port number on the media line(s) which are not on controller UE to zero and forward the SIP re-INVITE request towards the controller UE following the procedures as specified in 3GPP TS 24.229 [2].

Upon receipt from the remote party of a SIP re-INVITE request containing an SDP offer indicating one or more media components to be released on the controllee UE, the SCC AS shall:

- if there are more media components left after releasing the selected media components, set the port number on the media line(s) which are not on the controllee UE to zero and forward the SIP re-INVITE request; or

- if there are no more media components left after releasing the selected media components, send a SIP BYE request;

towards the controllee UE following the procedures as specified in 3GPP TS 24.229 [2].

Upon receipt of a SIP re-INVITE request from the remote party containing an SDP offer indicating one or more media components to be released on the controller UE and one or more media components to be released on the controllee UE(s), the SCC AS shall:

- send a SIP re-INVITE request to the controller UE containing the SDP received from the remote end with the exception that the port numbers of the media components, not on the controller UE, are set to zero;
- if not all the media component(s) are being released on the controllee UE, send a SIP re-INVITE request to that controllee UE, containing the SDP received from the remote end with the exception that the port numbers of the media components, not on the controllee UE, are set to zero; and
- if all the media component(s) are being released on the controllee UE, send a SIP BYE request to the controllee UE.

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 [45] 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].

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 Roles for inter-UE transfer and supplementary services interaction

20.2.1 Introduction

This subclause describes the SCC AS and SC UE procedures for inter-UE transfer when execution of supplementary service as described in 3GPP TS 22.173 [24].

20.2.2 Originating Identification Presentation (OIP)

There are no specific procedures for the SC UE and the SCC AS for OIP besides the procedures described in 3GPP TS 24.607 [25].

20.2.3 Originating Identification Restriction (OIR)

There are no specific procedures for the SC UE and the SCC AS for OIR besides the procedures described in 3GPP TS 24.607 [25].

20.2.4 Terminating Identification Presentation (TIP)

There are no specific procedures for the SC UE and the SCC AS for TIP besides the procedures described in 3GPP TS 24.608 [26].

20.2.5 Terminating Identification Restriction (TIR)

There are no specific procedures for the SC UE and the SCC AS for TIP besides the procedures described in 3GPP TS 24.608 [26].

20.2.6 Communication Diversion (CDIV)

There are no specific procedures for the SC UE and the SCC AS for TIP besides the procedures described in 3GPP TS 24.604 [27].

20.2.7 Communication Hold (HOLD)

The controller UE may hold an active media component(s) on itself, by following the procedures for HOLD described in 3GPP TS 24.610 [28].

The controller UE may hold or resume an active media component(s) on a controllee UE within a collaborative session while it has an ongoing IMS session with a remote UE, by sending a SIP REFER request and including:

1) the Request-URI set to the Inter UE Transfer SCC AS URI;

2) the Refer-To header as follows:

   a) the SIP URI of the controllee UE;

NOTE: The SIP URI of the controllee UE needs to be a GRUU if the controllee UE and any other UEs share the same public user identity.

   b) the body parameter containing SDP for the media m-lines in the session shall be set as follows:

      - media lines that are not on the controllee UE with port number zero;

      - media lines that are on the controllee UE and are not to be changed with their current directionality and the current port number from the remote end; and

      - media lines that are on the controllee and:

         - if those media lines are to be held, set a-line to inactive or recvonly and the current port number from the remote end; or

         - if those media lines are to be resumed, set a-line to sendonly or sendrecv and the current port number from the remote end.

3) the Accept header field set to "message/sipfrag"; and

4) the Target-Dialog header field populated as specified in IETF RFC 4538 [11], containing the dialog identifier of the collaborative session.

The controller UE shall handle response to the SIP REFER request and the subsequent SIP NOTIFY requests according to 3GPP TS 24.229 [2].

When SCC AS receives a SIP REFER request from the controller UE to hold or resume a media component on a controllee UE, the SCC AS shall send:

1) a SIP 202 (Accepted) response for the SIP REFER request to the controller UE;

2) a SIP NOTIFY request with sipfrag including SIP 100 Trying to the controller UE; and

3) send a SIP re- INVITE request to the controllee UE, containing an SDP offer as received from remote UE in the previous offer/answer exchanged and with directionality as set for the corresponding media in the "body" header of the URI in the Refer-To of the SIP REFER request received from the controller.

Upon receiving SIP 200 (OK) response from the controllee UE, the SCC AC shall send:

1) a SIP NOTIFY request to the controller UE with the sipfrag body including the SIP 200 (OK) response of the SIP re-INVITE request and also include the SDP information received from the controllee UE.

2) a SIP re-INVITE request to the remote leg as specified in 3GPP TS 24.229 [2]. The SCC AS shall construct the SDP information in the SIP re-INVITE request as follows:

   - set the SDP information including the directionality as received in the SIP 200 (OK) response from the controlee UE; and

   - include the SDP information for all other media in the collaborative session as from the original session to the remote leg.

Upon receiving SIP 200 (OK) response with the SDP answer from the remote leg, the SCC AS shall send a SIP ACK request to the remote leg;

When a controllee UE receives a SIP re-INVITE request to hold a media component(s), it shall follow the procedures described in 3GPP TS 24.610 [28].

20.2.8 Communication Barring (CB)

There are no specific procedures for the SC UE and the SCC AS for CB besides the procedures described in 3GPP TS 24.611 [29].

20.2.9 Message Waiting Indication (MWI)

There are no specific procedures for the SC UE and the SCC AS for MWI besides the procedures described in 3GPP TS 24.606 [30].

20.2.10 Conference (CONF)

In a collaborative session, it shall only be possible for the controller UE to invoke the CONF service following the procedures as described in 3GPP TS 24.605 [31].

When the remote UE sends a request for the CONF service to replace an existing collaborative session, the SCC AS shall deliver the request for CONF service to the controller UE, which then sets up new session following the procedures described in 3GPP TS 24.605 [31].

20.2.11 Explicit Communication Transfer (ECT)

In a collaborative session, it shall only be possible for the controller UE to invoke ECT service following the procedures as described in 3GPP TS 24.629 [32].

When the controller UE receives a notification that ECT has been performed successfully, the controller UE shall terminate the previous active session with the transferee UE by terminating all related media control sessions on the controllee UEs.

When the SCC AS receives an ECT transfer request from the remote end to transfer the collaborative session, the SCC AS shall deliver the request to the controller UE.

When the controller UE receives an ECT transfer request to transfer the collaborative session, the controller UE shall establish a new session towards the transfer target following the procedures described in 3GPP TS 24.629 [32].

20.2.12 Advice of Charge (AOC)

When the AOC service specified in 3GPP TS 24.647 [33] is active, the SCC AS shall deliver charging information to the controller UE.

20.2.13 Closed User Groups (CUG)

There are no specific procedures for the SC UE and the SCC AS for CUG besides the procedures described in 3GPP TS 24.654 [34].

20.2.14 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 inter-UE transfer is the same as that specified in subclause 20.2.10 for CONF service.

20.2.15 Flexible Alerting (FA)

There are no specific procedures for the SC UE and the SCC AS for FA besides the procedures described in 3GPP TS 24.239 [35].

20.2.16 Communication Waiting (CW)

There are no specific procedures for the SC UE and the SCC AS for CW besides the procedures described in 3GPP TS 24.615 [36].

20.2.17 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 CCBS/CCNR besides the procedures described in 3GPP TS 24.642 [37].

20.2.18 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].

20.2.19 Malicious Communication IDentification (MCID)

There are no specific procedures for the SC UE and the SCC AS for MCID besides the procedures described in 3GPP TS 24.616 [39].

20.2.20 Personal Network Management (PNM)

There are no specific procedures for the SC UE and the SCC AS for PNM besides the procedures described in 3GPP TS 24.259 [40].

20.2.21 Customized Ringing Signal (CRS)

There are no specific procedures for the SC UE and the SCC AS for CRS besides the procedures described in 3GPP TS 24.183 [41].

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.
- sip:interUEtransfer@sccas1.home1.net represents the Inter UE Transfer SCC AS URI of the UA A.

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.

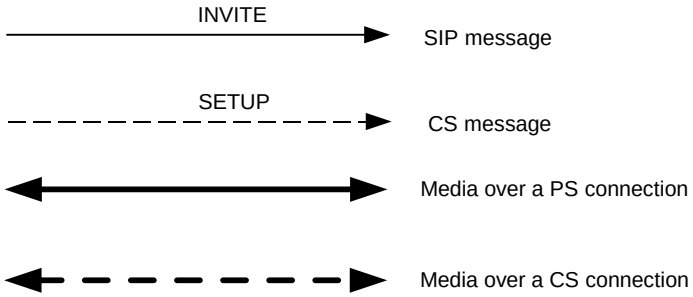


Figure A.2-1: Signalling flow notation

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.

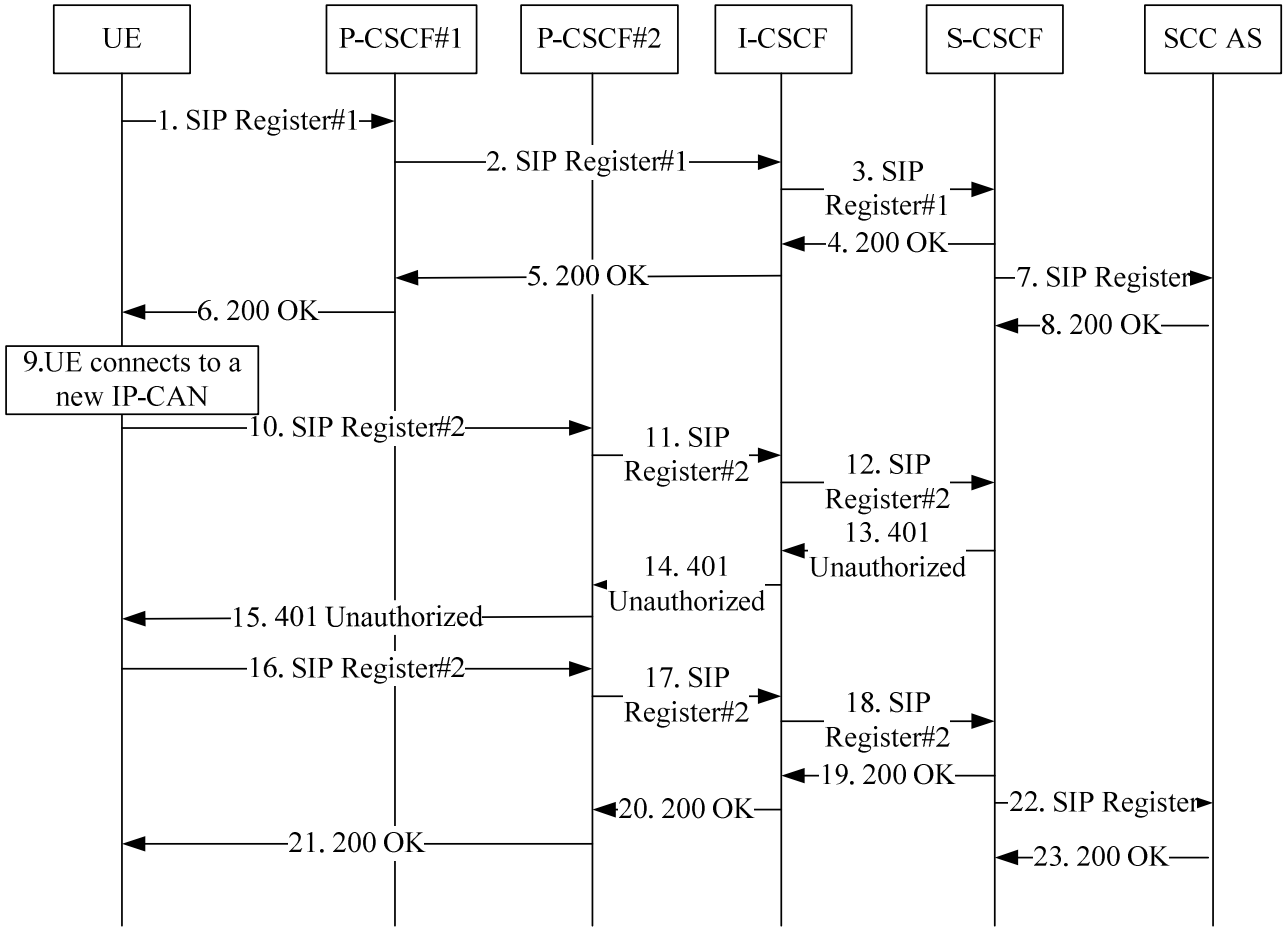


Figure A.3.2-1 Signalling flows for multiple registrations

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
```

1. **SIP REGISTER request (P-CSCF#1 to I-CSCF)-See example in table A.3.2-2**

   After performing the DNS query, the P-CSCF#1 forwards the SIP REGISTER request towards I-CSCF. The P-CSCF adds a Path header field with a flow token and includes the 'ob' parameter

Table A.3.2-2 SIP REGISTER request (P-CSCF#1 to I-CSCF)

```
REGISTER sip:registrar.home1.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.visited1.net;branch=z9hG4bK240f34.1, SIP/2.0/UDP
   [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Max-Forwards: 69
P-Access-Network-Info:
Path: <sip:VsktcQ/S8p4WPbOnHbuyh5iJvJIW3ib@pcscf1.visited1.net;lr;ob>
Require: path
P-Visited-Network-ID: "Visited Network Number 1"
P-Charging-Vector: icid-value="AyretyU0dm+6O2IrT5tAFrbHLso=023551024"
From:
To:
Contact:
Call-ID:
Authorization: Digest username="user1_private@home1.net", realm="registrar.home1.net",
   nonce=base64(RAND + AUTN + server specific data), algorithm=AKAv1-MD5,
   uri="sip:registrar.home1.net", response="6629fae49393a05397450978507c4ef1", integrity-
   protected="yes"
CSeq:
Supported:
Content-Length:
```

2. **SIP REGISTER request (I-CSCF to S-CSCF)**

   The I-CSCF forwards the SIP REGISTER request to the S-CSCF.

3. **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.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.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.

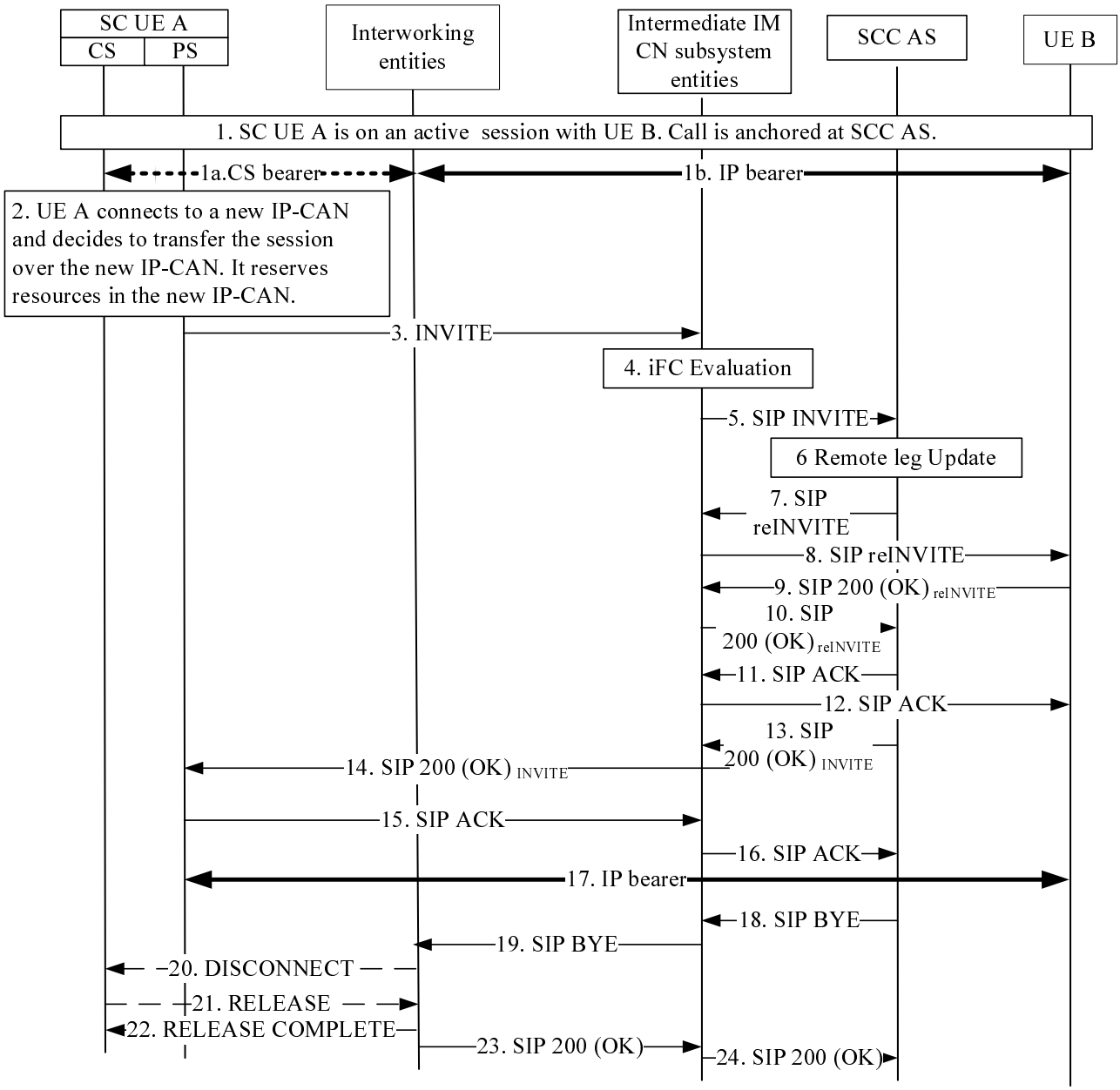


Figure A.6.1-1: Signalling flow for PS-CS Access Transfer: CS to PS

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.

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.

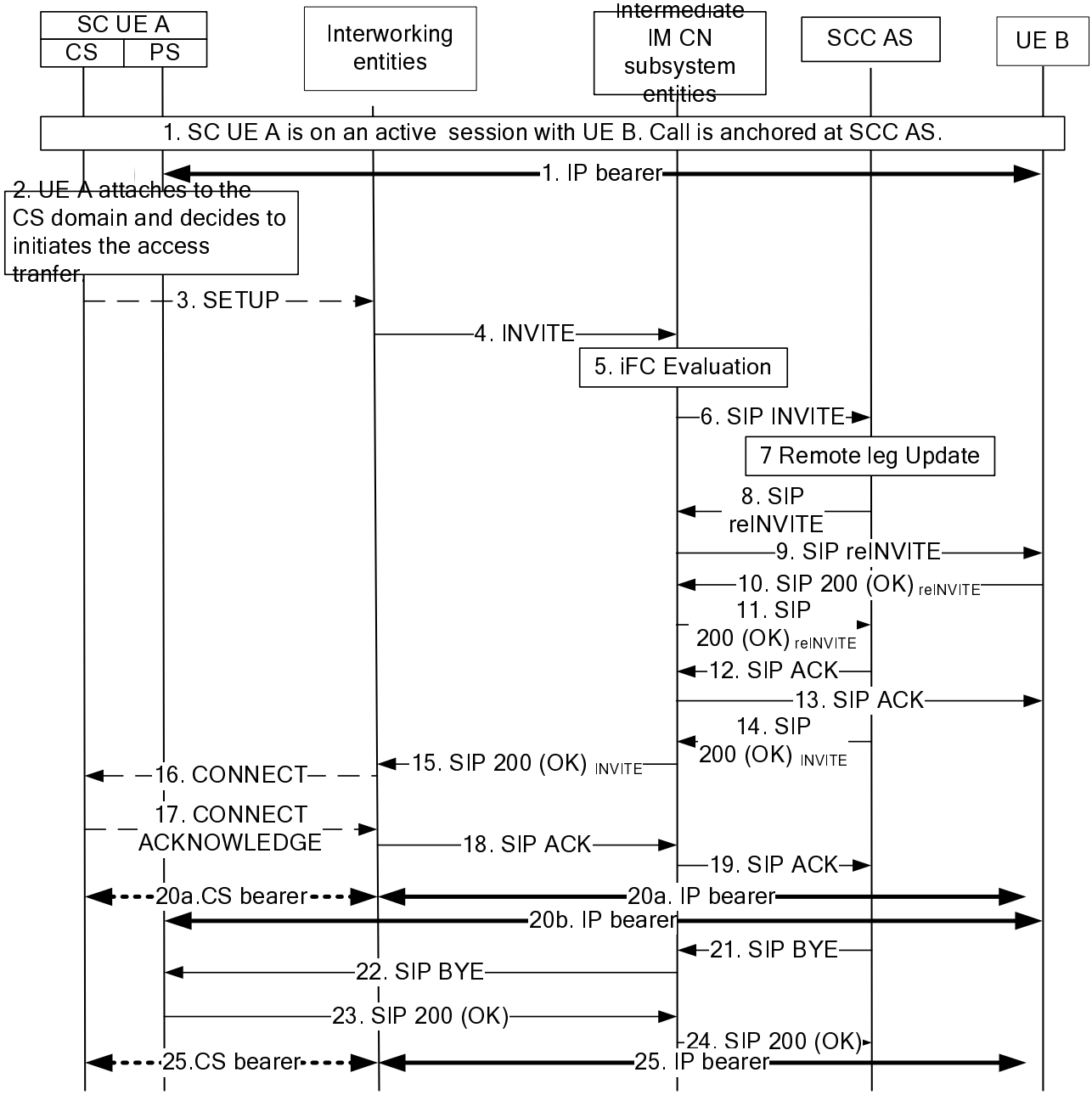


Figure A.6.2-1 Signalling flow for PS-CS access transfer: PS-CS

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.

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.

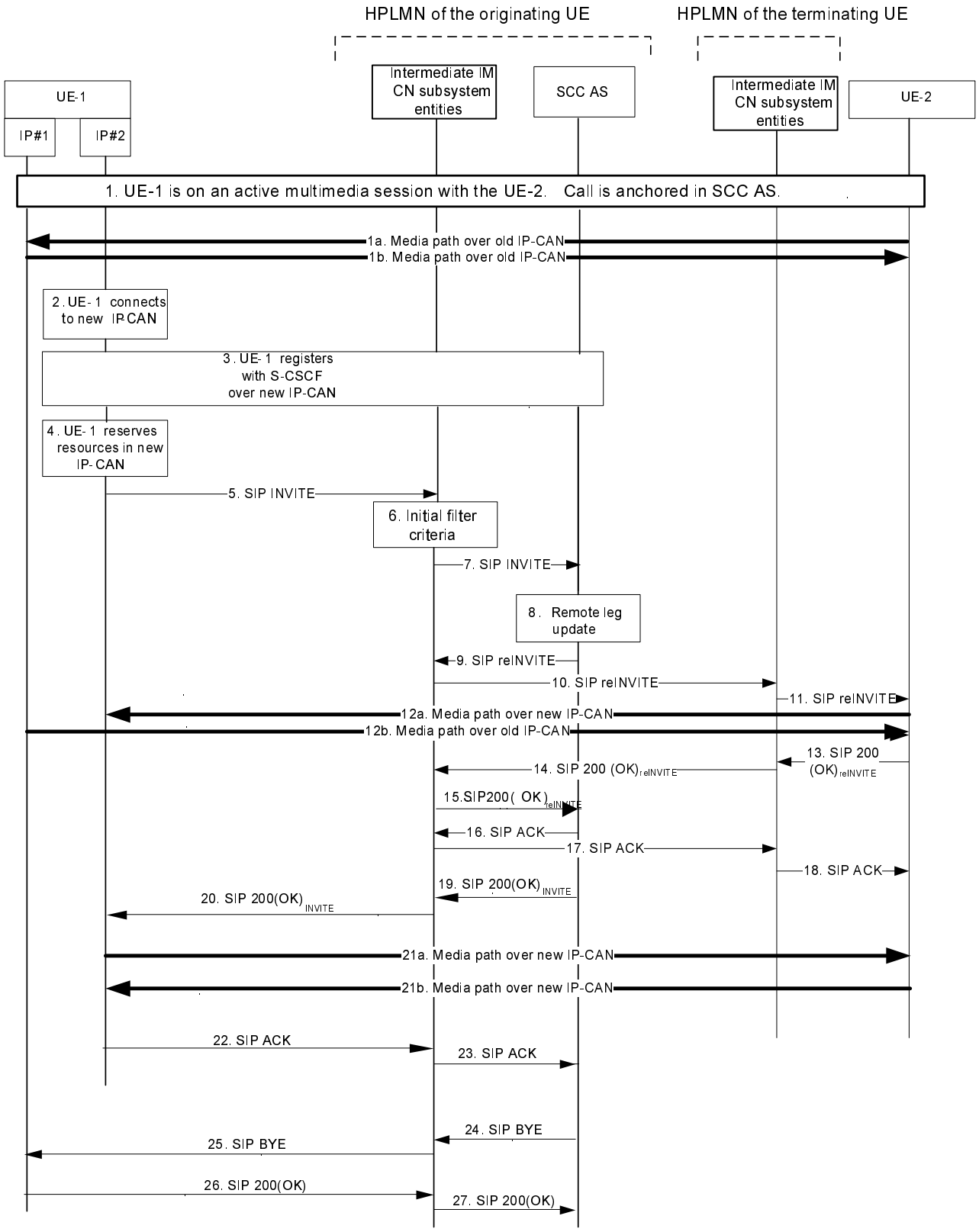


Figure A.7.2-1: Signalling flow for session handover

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 registration 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
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>;+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 SCC AS acts as a routeing B2BUA as specified in 3GPP TS 24.229 [2]. In this example 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>
Record-Route: <sip:scscf1.home1.net;lr>, <sip: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>;+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"
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.

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.

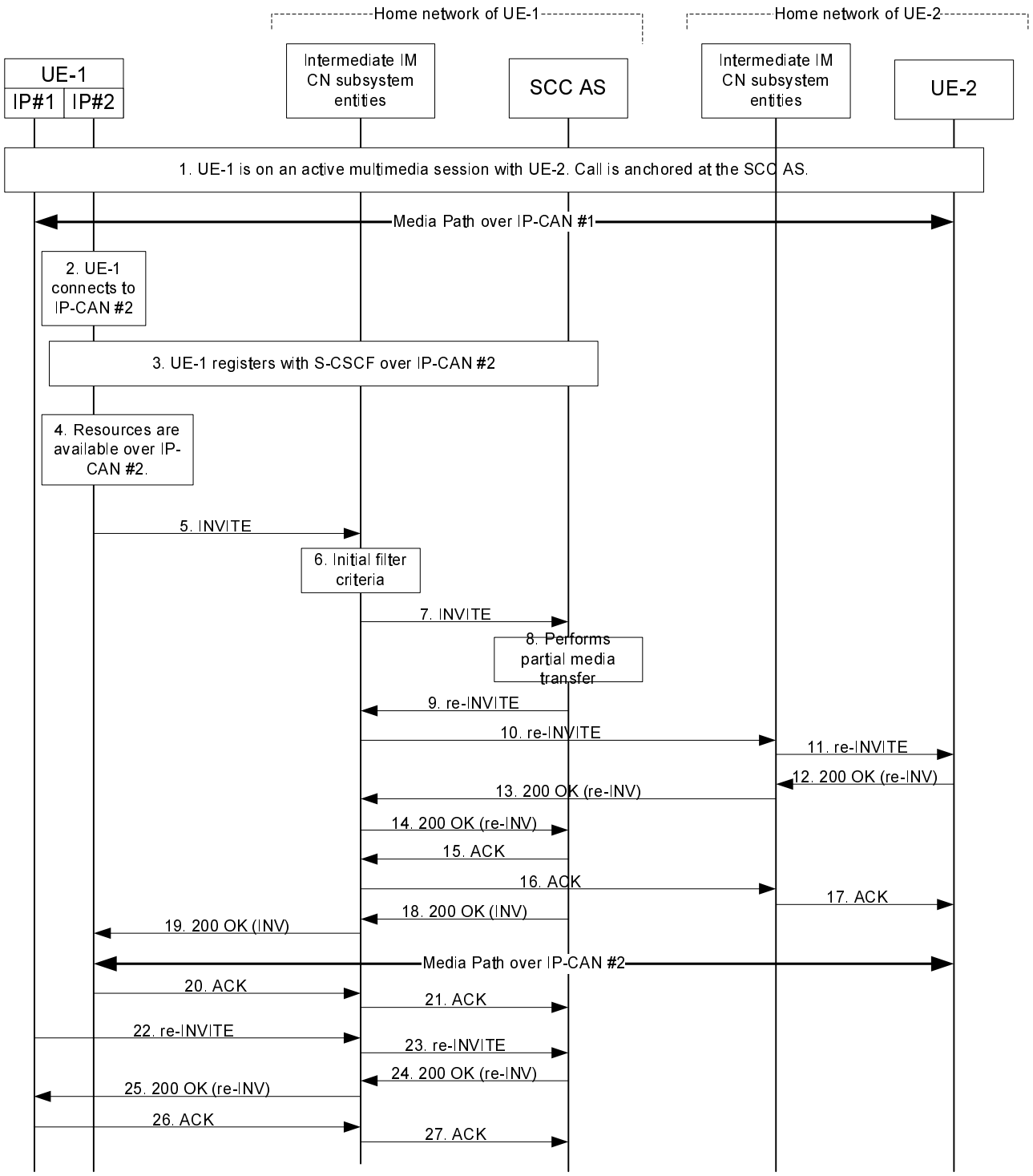


Figure A.7.3-1: Signalling flow for PS-PS session transfer with partial media transfer

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 registration 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: 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
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>;+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 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>;+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: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
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: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>;+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: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.

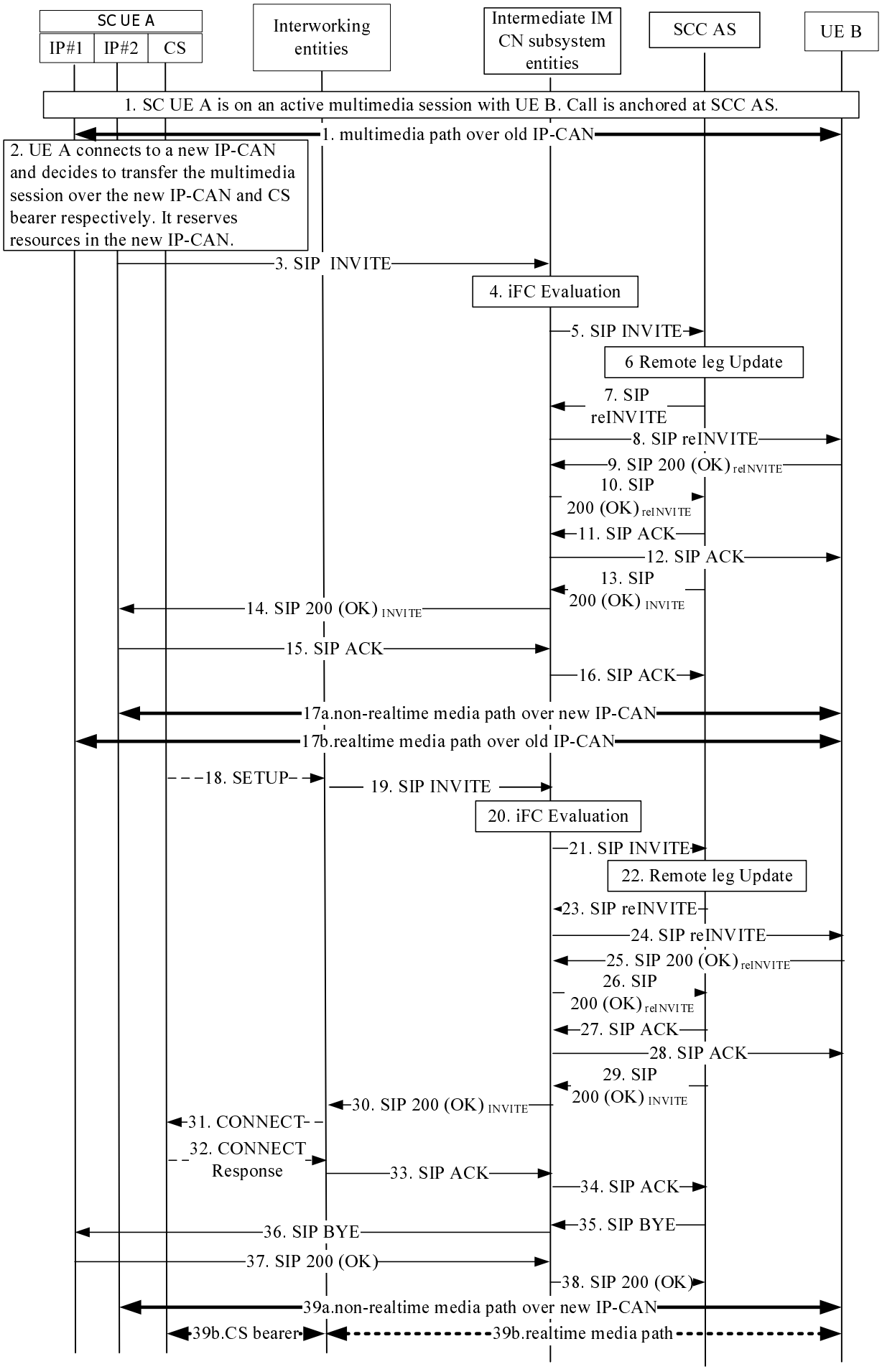

SC UE A
IP#1
IP#2
CS
Interworking entities
Intermediate IM CN subsystem entities
SCC AS
UE B
1. SC UE A is on an active multimedia session with UE B. Call is anchored at SCC AS.
1. multimedia path over old IP-CAN
2. UE A connects to a new IP-CAN and decides to transfer the multimedia session over the new IP-CAN and CS bearer respectively. It reserves resources in the new IP-CAN.
3. SIP INVITE
4. iFC Evaluation
5. SIP INVITE
6 Remote leg Update
7. SIP reINVITE
8. SIP reINVITE
9. SIP 200 (OK) reINVITE
10. SIP 200 (OK) reINVITE
11. SIP ACK
12. SIP ACK
13. SIP 200 (OK) INVITE
14. SIP 200 (OK) INVITE
15. SIP ACK
16. SIP ACK
17a.non-realtime media path over new IP-CAN
17b.realtime media path over old IP-CAN
18. SETUP
19. SIP INVITE
20. iFC Evaluation
21. SIP INVITE
22. Remote leg Update
23. SIP reINVITE
24. SIP reINVITE
25. SIP 200 (OK) reINVITE
26. SIP 200 (OK) reINVITE
27. SIP ACK
28. SIP ACK
29. SIP 200 (OK) INVITE
30. SIP 200 (OK) INVITE
31. CONNECT
32. CONNECT Response
33. SIP ACK
34. SIP ACK
35. SIP BYE
36. SIP BYE
37. SIP 200 (OK)
38. SIP 200 (OK)
39a.non-realtime media path over new IP-CAN
39b.CS bearer
39b.realtime media path

Figure A.8.2-1: Signalling flow for PS - PS in conjunction with PS - CS Access Transfer: PS to CS

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

1. SC UE A has an ongoing multimedia session with remote UE B

The call has been anchored at the SCC AS which is in the HPLMN of originating SC UE A. The call leg over old IP-CAN is identified with "Call-ID= me03a0s09a2sdfgjkl491777", "From tag=64727891", and "To tag=774321". The UE A and UE B exchange media over the old IP-CAN, which is maintained while the SC UE A initiates the handover procedure.

Table A.8.2-1 shows an example of the SDP offer from SC UE A to remote UE B.

NOTE 2: To later show how the media is transferred to the new IP-CAN and CS bearer, only the SDP offer is shown in table A.8.2-1.

Table A.8.2-1: SIP INVITE request (SC UE A to intermediate IM CN subsystem entities)

```
INVITE tel:+1-237-555-2222 SIP/2.0
Via:
Max-Forwards:
Route:
P-Asserted-Identity:
P-Charging-Vector:
P-Access-Network-Info:
Privacy:
From:
To:
Call-ID:
Cseq:
Supported:
Require:
Proxy-Require:
Security-Verify:
Contact:
Allow:
Accept:
Content-Type:
Content-Length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=
c=IN IP6 5555::aaa:bbb:ccc:ddd
t=0 0
m=audio 3456 RTP/AVP 97 96
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:96 telephone-event
a=maxptime:20
m=message 7654 TCP/MSRP 98
a=accept-types:text/plain
```

2. SC UE A connects to a new IP-CAN#2:

The SC UE A decides to transfer the multimedia session over the new IP-CAN and CS bearer respectively. The UE A obtains an IP address that it will use for the signalling and media. It registers with the S-CSCF over the new IP-CAN using multiple registrations procedure. Depending on the UE A configuration, the discovery of the P-CSCF in the new IP-CAN can be needed. Based on the UE A and new IP-CAN capabilities, the UE A decides to use the same codec that was used over the old IP-CAN. The UE A reserves resources (e.g. QoS) in the new IP-CAN that will be needed for the signalling and transferred media, prior to sending the initial SIP INVITE request.

3. SIP INVITE request (SC UE A to intermediate IM CN subsystem entities)- see example in table A.8.2-3

The SC UE A sends an initial SIP INVITE request with a STI and a new SDP offer to the UE B that indicates that the new call replaces the existing call. The initial SIP INVITE request establishes a dialog for signalling and specifies in the SDP a new contact address that will be used for non-realtime media over the new IP-CAN. Upon sending the initial SIP INVITE request, the UE A is ready to receive the RTP packets either over the new IP-CAN or the old IP-CAN. The RTP packets can arrive over the new IP-CAN prior to the SC UE are receiving the SIP 200 (OK) response for the initial SIP INVITE request.

Table A.8.2-3: SIP INVITE request (UE A to intermediate IM CN subsystem entities)

```
INVITE tel:+1-237-555-2222 SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:fff]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Max-Forwards: 70
Route: sip:pcscf1.home1.net:7531;lr;comp=sigcomp>, <sip:orig@scscf1.home1.net;lr>
P-Preferred-Identity: "John Doe" <sip:user1_public1@home1.net>
P-Access-Network-Info: IEEE-802.11b
Privacy: none
From: <sip:user1_public1@home1.net>; tag=171828
To: <tel:+1-237-555-2222>
Call-ID: cb03a0s09a2sdfglkj490237
Cseq: 127 INVITE
Supported: 100rel; precondition
Require: sec-agree
Proxy-Require: sec-agree
Security-Verify: ipsec-3gpp; q=0.1; alg=hmac-sha-1-96; spi=87654321; port1=7531
Contact: <sip:user1_public1@home1.net; gr=urn:uuid:f81d4fae-7dec-11d0-a765-
   00a0c91e6bf6>;+g.3gpp.icsi-ref="urn%3Aurn-7%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.

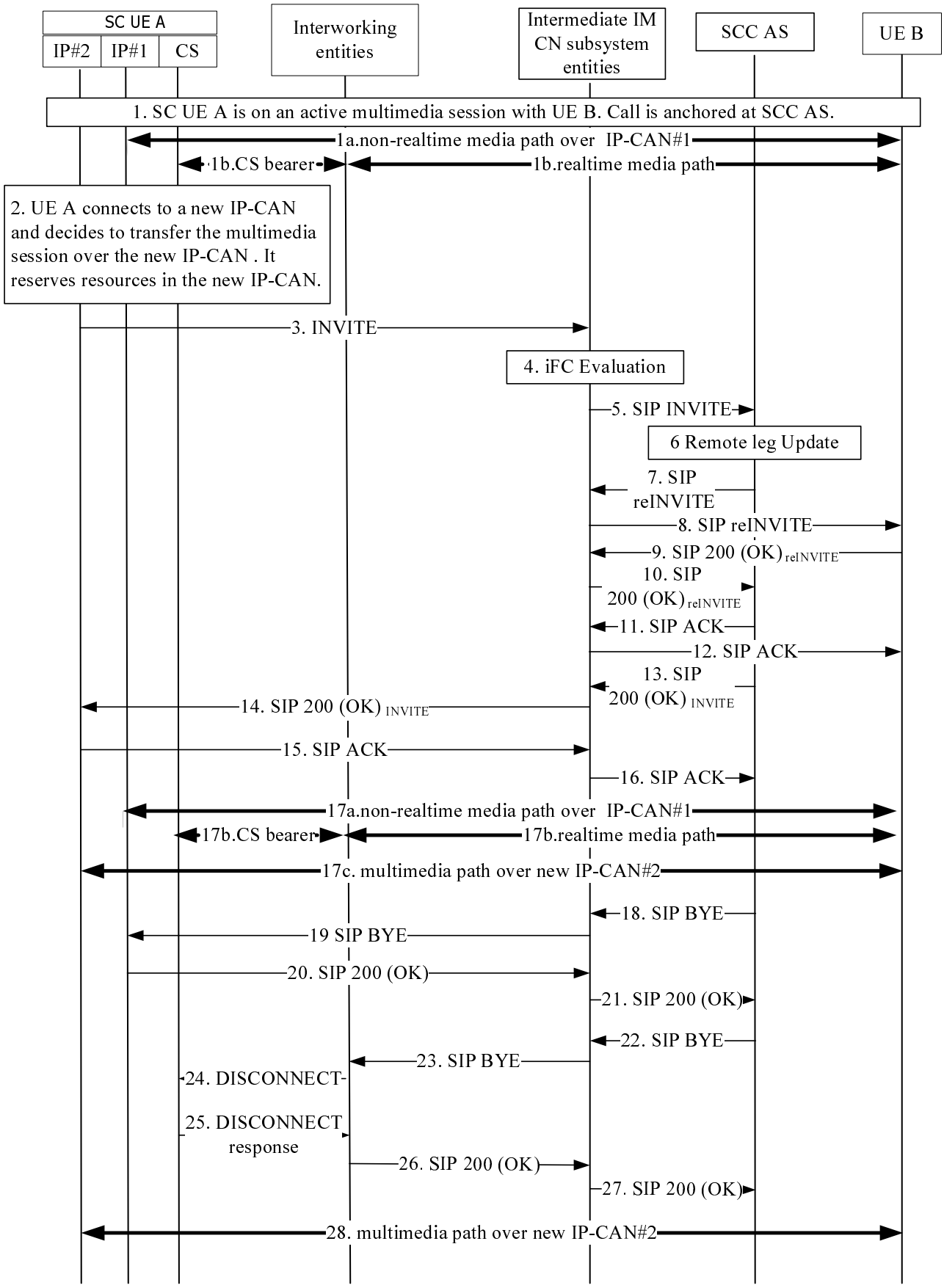


Figure A.8.3-1: Signalling flow for PS - PS in conjunction with PS - CS Access Transfer: CS to PS

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.

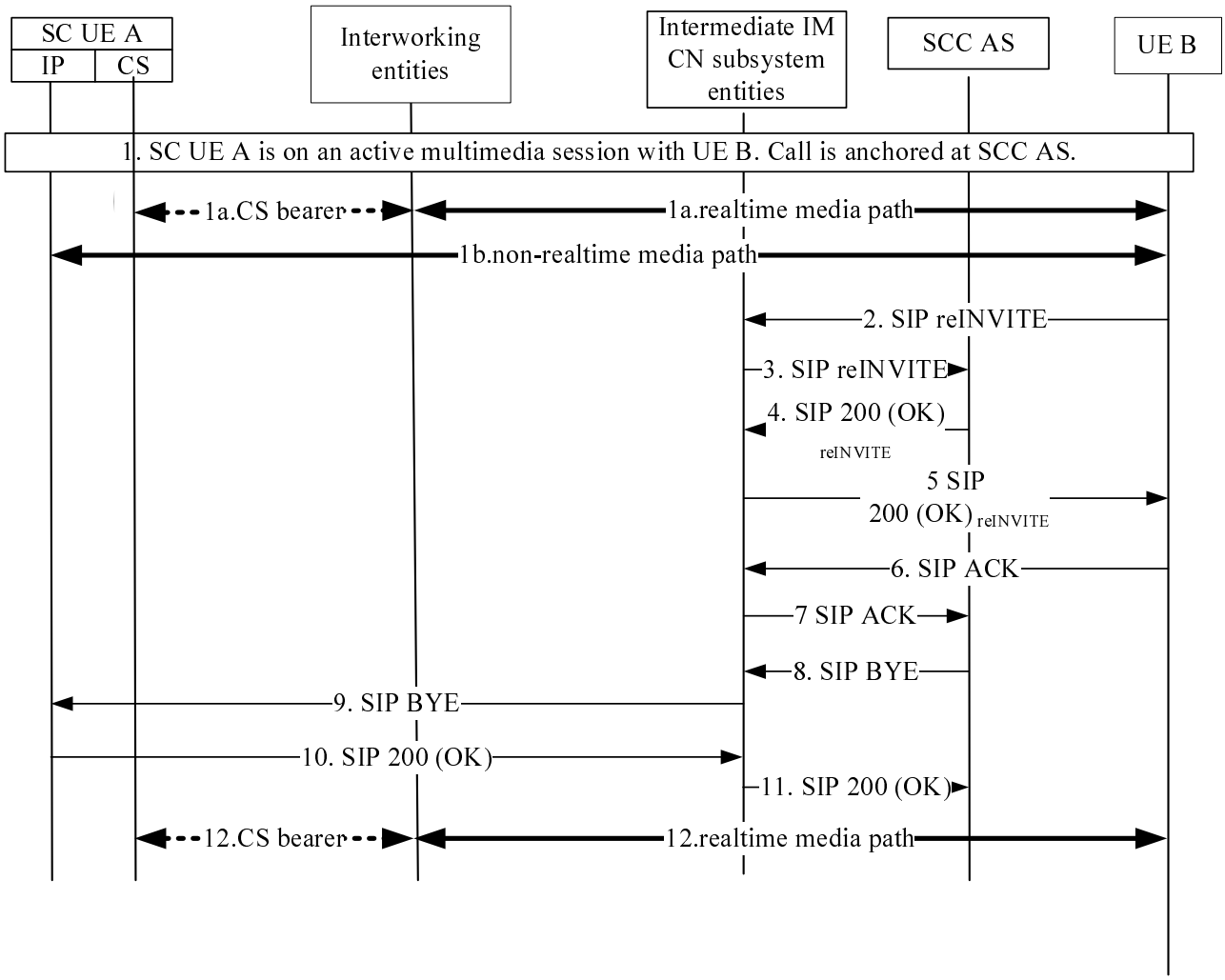


Figure A.9.2-1: Remote End Initiation case – Removing media from split CS and PS sessions

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 Signalling flows for Inter-UE Transfer without establishment of Collaborative Session

A.10.1 Introduction

The signalling flows in the subclause demonstrate how a UE-1 can initiate the inter UE transfer of the complete session without Collaborative Session establishment.

The example assumes that the UE-1 and UE-2 are under the control of the same subscriber.

A.10.2 Complete transfer in services defining only originating session set up in UE

In the example flow at the figure A.10.2-1, UE-1 has an ongoing multimedia session with UE-3 anchored at SCC AS. The session is established using an IMS communication service identified by ICSI urn:urn-7:3gpp-service.ims.icsi.iptv which is an IMS communication service which defines originating session set up in the UE only.

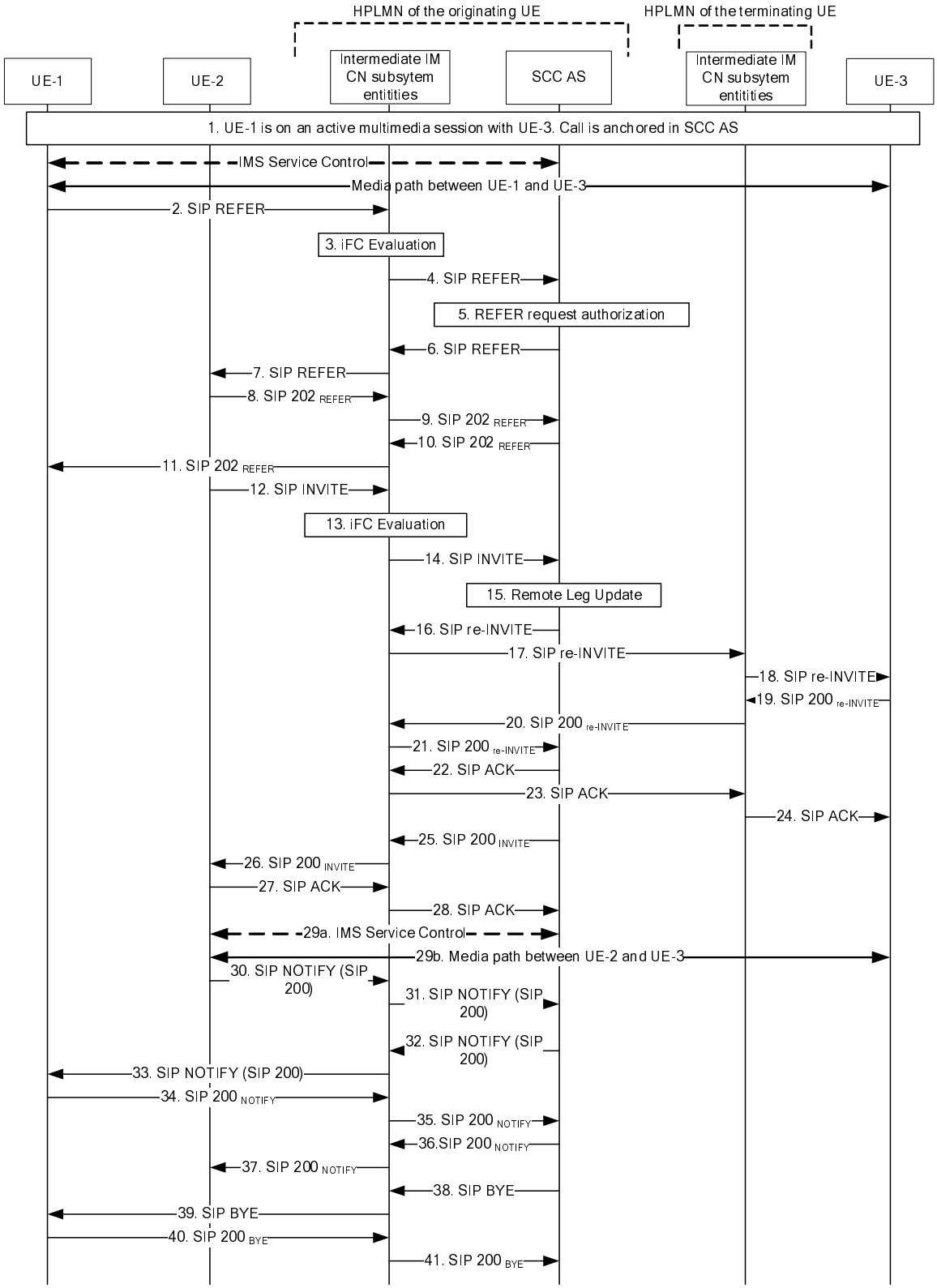


Figure A.10.2-1: Signalling flow for inter UE transfer without Collaborative Session establishment

NOTE 1: For clarity, the SIP 100 (Trying) responses and the SIP NOTIFY requests carring the message/sipfrag with SIP 100 (Trying) response are not shown in the signalling flow.

1. UE-1 is in session with UE-3

There is a multimedia session comprising audio and video media between the UE-1 and the remote UE-3 anchored at SCC AS. The session was established using IMS communication service identified by ICSI urn:urn-7:3gpp-service.ims.icsi.iptv. The dialog identifier of the session is AB03a0s09a2sdfglkj490333, remote-tag=Afgsdfg45, local-tag=U188gg.

2. SIP REFER request initiating the inter UE transfer to UE-2 (UE-1 to Intermediate IM CN subsystem entities) - see example in table A.10.2-2

Table A.10.2-2: SIP REFER request (UE-1 to Intermediate IM CN subsystem entities)

```
REFER sip:user@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-222222222222 SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Max-Forwards: 70
P-Preferred-Identity: <sip:user@home1.net>
From: <sip:user@home1.net>;tag=171828
To: <sip:user@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-222222222222>
Call-ID: Asdasd231233
Cseq: 4127 REFER
Contact: <sip:user@home1.net;gr=gr=urn:uuid:f81d4fae-7dec-11d0-a765-111111111111>
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE
Content-Length: 0
Refer-To: <sip:interUEtransfer@sccas.home1.net?Target-
   Dialog=AB03a0s09a2sdfglkj490333%3Bremote-tag=Afgsdfg45%3Blocal-
   tag=U188gg&Require=tdialog&P-Preferred-Service=urn:urn-7:3gpp-service.ims.icsi.iptv&Accept-
   Contact=*%3b+g.3gpp.icsi-ref%3d%22urn%253Aurn-7%253gpp-service.ims.icsi.iptv%22>
```

Request-URI: contains the GRUU of the UE-2

Refer-To: contains the Inter UE Transfer SCC AS URItogether with Target-Dialog URI header field containing the dialog identifier of the session with UE-3, Require URI header field containing the "tdialog" and P-Preferred-Service and Accept-Contact URI header fields containing the ICSI of the service to be requested by UE-2.

Contact: contains the GRUU of the UE-1

3. Evaluation of initial filter criteria

The S-CSCF evaluates originating initial filter criteria for the served user and as a result routes the SIP REFER request towards the SCC AS.

4. **SIP REFER request** (Intermediate IM CN subsystem entities to SCC AS)

5. The SCC AS authorizes the request and if authorization is passed successfully, the SCC AS forwards the SIP REFER request further

6.-7. SIP REFER request (SCC AS to UE-2)

8.-11. SIP 202 (Accepted) response to the SIP REFER request (UE-2 to UE-1)

12. SIP INVITE request (UE-2 to intermediate IM CN subsystem entities) - see example in table A.10.2-12

Table A.10.2-12: SIP INVITE request (UE-1 to Intermediate IM CN subsystem entities)

```
INVITE sip:interUEtransfer@sccas.home1.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:fff]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Max-Forwards: 70
P-Preferred-Identity: <sip:user@home1.net>
From: <sip:user@home1.net>;tag=171828
To: <sip:interUEtransfer@sccas.home1.net>
Call-ID: tq34gasgaegr
Cseq: 4127 INVITE
Contact: <sip:user@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-222222222222>
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE
Target-Dialog: AB03a0s09a2sdfglkj490333;remote-tag=Afgsdfg45;local-tag=U188gg
Require: tdialog
Content-Type: application/sdp
Content-Length: (…)
Supported: 100rel, precondition
P-Preferred-Service: urn:urn-7:3gpp-service.ims.icsi.iptv
Accept-Contact: *;+g.3gpp.icsi-ref="urn%3Aurn-7%3gpp-service.ims.icsi.iptv"

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 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
m=video 3400 RTP/AVP 98 99
b=AS:75
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
a=rtpmap:99 MP4V-ES
```

Request-URI: set to the URI in the Refer-To of the received SIP REFER request

Contact: contains the GRUU of the UE-2

Target-Dialog: set to the value of the Target-Dialog URI header field of the URI in the Refer-To of the received SIP REFER request

Require: set to the value of the Require URI header field of the URI in the Refer-To of the received SIP REFER request

P-Preferred-Service: set to the value of the P-Preferred-Service URI header field of the URI in the Refer-To of the received SIP REFER request

Accept-Contact: set to the value of the Accept-Contact URI header field of the URI in the Refer-To of the received SIP REFER request

13. Evaluation of initial filter criteria

The S-CSCF evaluates originating initial filter criteria for the served user and as a result routes the SIP INVITE request towards the SCC AS.

14. SIP INVITE request (Intermediate IM CN subsystem entities to SCC AS)

15. Remote Leg Update

Based on the STI in the Target-Dialog header field the SCC AS detects that the inter UE tranfer is being attempted and performs the Remote Leg update by sending the SIP re-INVITE request towards the remote UE.

16-18. SIP re-INVITE request (SCC AS to UE-3 over 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-3 via the intermediate IM CN subsystem entities.

19-21. SIP 200 (OK) response (UE-3 to intermediate IM CN subsystem entities)

Upon receiving the SIP re-INVITE request containing the SDP offer, since the UE-3 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.

22-24. SIP ACK request (SCC AS to UE-3 via intermediate 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-3.

25-26. SIP 200 (OK) response (SCC AS to UE-2 via intermediate 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 UE-2.

27-28. SIP ACK request (UE-2 to SCC AS via intermediate IM CN subsystem entities)

The UE-2 generate the SIP ACK request to the SIP 200 (OK) response, and forward it to the SCC AS.

29. Media and IMS service control paths:

The media path is now established between UE-2 and UE-3 and the IMS service control between UE-2 and SCC AS.

30-33. SIP NOTIFY request (UE-2 to UE-1 over intermediate IM CN subsystem entities and SCC AS)

The UE-2 generate the SIP NOTIFY request carrying the message/sipfrag body and send it towards UE-1.

34-37. SIP 200 OK response to the SIP NOTIFY request (UE-1 to UE-2 over intermediate IM CN subsystem entities and SCC AS)

38-39: SIP BYE request (SCC AS to UE-1 via intermediate IM CN subsystem entities)

The SCC AS terminates the source access leg by sending a BYE request to the UE-1.

40-41. SIP 200 (OK) response (UE-1 to SCC AS via intermediate IM CN subsystem entities)

Upon receiving the SIP BYE request, the UE-1 sends a SIP 200 (OK) response to the SCC AS. Subsequently, the UE-1 relinquishes all resources pertaining to the session.

A.10.3 Complete transfer in services defining terminating session set up in UE

Editor's note: Complete transfer in services defining terminating session set up in UE is FFS

A.11 Signalling flows for establishment of collaborative session for inter-UE transfer

A.11.1 Introduction

The signalling flows for establishing a collaborative session. Two different scenarios have been considered in the clause:

- the first scenario is when the collaborative session established by transferring media component from controller UE to a controllee UE. This scenario is similar to the procedures described in subclause A.13.2 with exception that upon receipt of SIP REFER request from the controller UE, the SCC AS generates a SIP INVITE request instead of a SIP re-INVITE request and send it to the controllee UE; and

- the second scenario is described in subclause A.11.2 and shows an example where the collaborative session established by adding a new media component on a controllee UE.

A.11.2 Collaborative session establishment with new media

There is an existing session with audio between controller UE, UE-1, and remote UE. The controller UE establishes a collaborative session by adding a video media component to the controllee UE, UE-2.

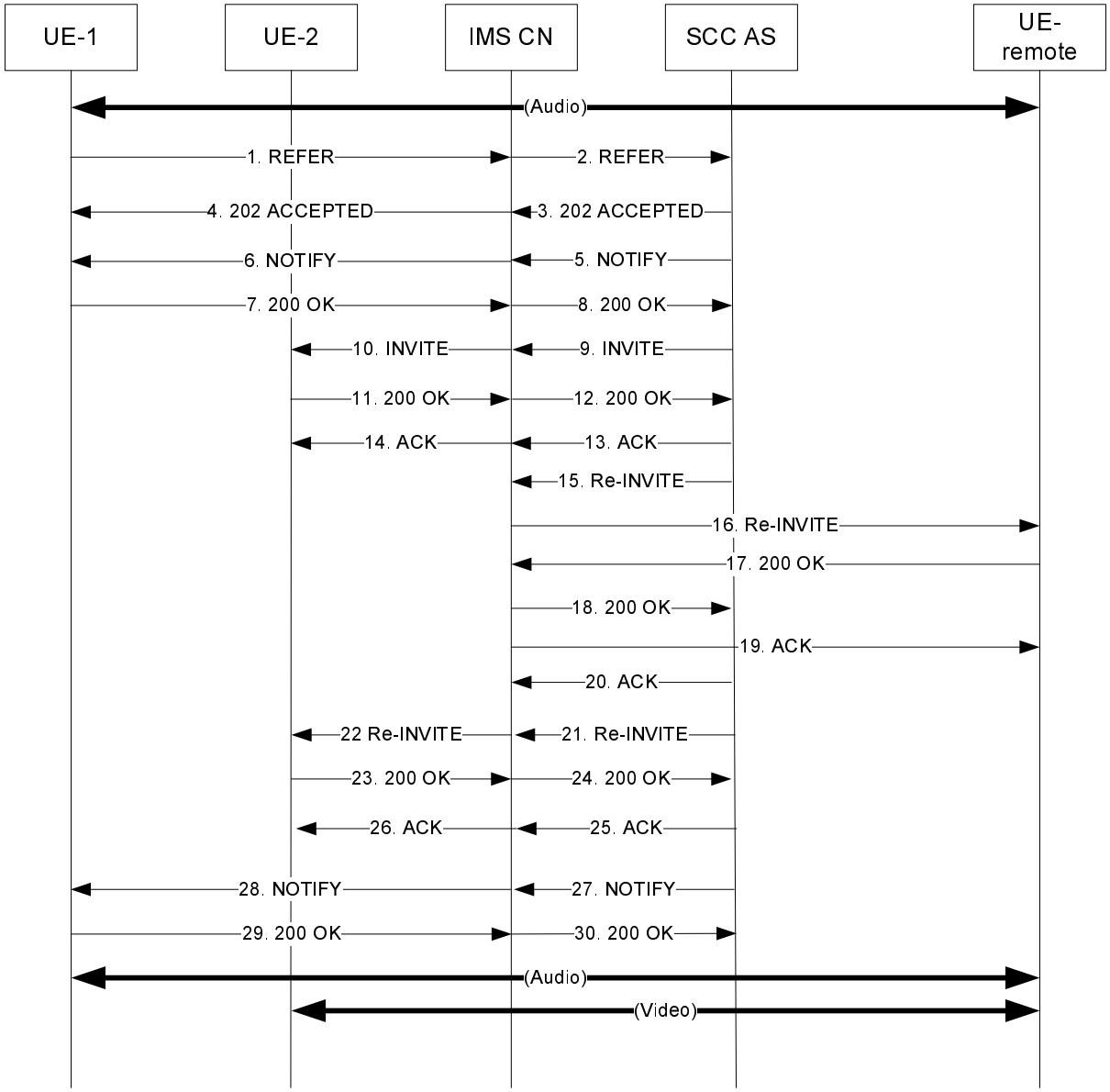


Figure A.11.2: Controller UE establishes a collaborative session by adding a new media on controllee UE

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

1-2. SIP REFER request (from UE-1 to SCC-AS)

The controller UE, UE-1 sends a SIP REFER request to the SCC AS containing a Refer-To header field containing the GRUU of controllee UE, UE-2 and a body parameter containing an m line for audio set to zero and an m line for video with the port number set to the discard port number "9" since the port number is unknown. The SIP REFER request also includes a Target-dialog header field containing the details fo the dialog for the existing session between controller UE, UE-1 and the remote UE.

Table A.11.2-1 SIP REFER request (UE-1 to SCC-AS)

```
REFER sip:interUEtransfer@sccas1.home1.net SIP/2.0
Via:
To: sip:interUEtransfer@sccas1.home1.net
From: sip:user1_public1@home1.net;tag=24680
Call-ID: cb03a0s09a2sdfglkj490333
CSeq: 93809824 REFER
Max-Forwards: 70
P-Preferred-Identity:
Refer-To: <sip:user1_public2@home2.net;gr=urn:uuid:f81d4fae-7dec-11d0-a762-
   00a0c91e6bf6?body=m%3Daudio%200%20RTP%2FAVP%2096%0Dm%3Dvideo%209%20RTP%2FAVP%2098%2099>
Require: target-dialog
Target-dialog: cb03a0s09a2sdfglkj13579; remote-tag=abcdef; local-tag=123456
Contact: <sip:user1_public1@home1.net; gr=urn:uuid:f81d4fae-7dec-11d0-a765-
   00a0c91ewxyz>;+g.3gpp.iut-controller
Allow:
Accept: message/sipfrag
Content-Length: 0
```

3-4. SIP 202 (Accepted) response (from SCC AS to UE-1)

SCC-AS sends SIP 202 (Accepted) response to controller UE-1 as response to the SIP REFER request.

5-6. SIP NOTIFY request (from SCC AS to UE-1)

SCC-AS sends a SIP NOTIFY request to UE-1 notifying implicit subscription to the SIP REFER request.

Table A.11.2-2 SIP NOTIFY request (SCC-AS to UE-1)

```
NOTIFY
Via:
To: sip:user1_public1@home1.net;tag=24680
From: sip:interUEtransfer@sccas1.home1.net;tag=13579
Call-ID:
CSeq:
Max-Forwards:
P-Asserted-Identity:
Require:
Contact: sip: scc-as@home1.net
Allow:
Event: refer
Subscription-State: active;expires=3600
Content-Type: message/sipfrag;version=2.0
Content-Length: (…)

SIP/2.0 100 Trying
```

7-8. SIP 200 (OK) response (from UE-1 AS to SCC-AS)

The controller UE, UE-1, acknowledges the SIP NOTIFY request by sending SIP 200 (OK) response to SCC-AS.

9-10. SIP INVITE request (from SCC-AS to UE-2)

SCC-AS sends SIP INVITE request to the controllee UE, UE-2, adding video media and establishing collaborative session.Since the URI parameters indicates that the port number for video m-line is set to the discard port number "9", the SCC AS realizes that the port number of the remote UE is unknown and therefore adds an a-line to inactive in SDP offer to prevent the controllee UE sending media to the remote UE. The SDP offer contains the audio media component on controller UE, UE-1 set to zero. The SIP INVITE request contains a Referred-By header field containing the identity of UE-1.

Table A.11.2-3 SIP INVITE request (SCC-AS to UE-2)

```
INVITE sip:user2_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91e6bf6 SIP/2.0
Via:
Record-Route: sip:scc-as@home1.net
To: sip:user2_public1@home1.net;
From: sip:user3_public3@home3.net;tag=acegi
Call-ID:
CSeq:
Max-Forwards:
P-Asserted-Identity: "remote user" sip:user3_public3@home3.net
Require:
Referred-By: sip:user1_public1@home1.net
Contact: sip:user3_public3@home3.net;gr=urn:uuid:f81d4fae-17oct-11a1-a678-0054c91eabcd
Allow:
Content-Type: application/sdp
Content-Length:(…)

v=0
o=- 1027933615 1027933615 IN IP4 123.112.67.87
s=-
t=0 0
m=audio 0 RTP/AVP 0
m=video 9 RTP/AVP 98 99
a=inactive
c=123.112.67.87
b=AS:75
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
a=rtpmap:99 MP4V-ES
```

Editor's note: The value of c-line is FFS.

11-12. SIP 200 (OK) response (from UE-2 to SCC-AS)

The controllee UE, UE-2, acknowledges the SIP INVITE request by sending SIP 200 (OK) reqponse to SCC-AS.

Table A.11.2-11 SIP 200 (OK) response (UE-2 to SCC-AS)

```
SIP/2.0 200 OK
Via:
To: sip:user2_public1@home1.net;tag=xyzwv
From: sip:user3_public3@home3.net;tag=acegi
Call-ID:
CSeq:
P-Preferred-Identity:
Contact: sip:user1_public2@home2.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-
   00a0c91e6bf6+g.3gpp.iut-controller
Allow: INVITE, PRACK, UPDATE
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 1027933615 1027933615 IN IP4 145.23.77.88
s=-
t=0 0
m=audio 0 RTP/AVP 0
m=video 9 RTP/AVP 98
a=inactive
c=145.23.77.88
b=AS:75
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
```

13-14. SIP ACK request (from SCC-AS to controllee UE)

SCC-AS sends SIP ACK request to the remote UE.

15-16. SIP re-INVITE request (from SCC-AS to remote UE)

SCC-AS sends SIP re-INVITE request to the remote UE.

Table A.11.2-15 SIP re-INVITE request (SCC-AS to remote UE)

```
INVITE sip:user3_public3@home3.net;gr=urn:uuid:f81d4fae-17oct-11a1-a678-0054c91eabcd SIP/2.0
Via:
To: sip:user3_public3@home3.net;tag=66666
From: sip:user1_public1@home1.net;tag=33333
Call-ID:
CSeq:
Max-Forwards:
P-Asserted-Identity:
Require:
Contact: sip: user1_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91ewxyz
Allow:
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 1027933615 1027933615 IN IP4 123.45.67.89
s=-
t=0 0
m=audio 1300 RTP/AVP 96 97
c=IN IP4 123.45.67.89
b=AS:25.4
a=rtpmap:96 AMR
a=fmtp:96 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:97 telephone-event
a=maxptime:20
m=video 1302 RTP/AVP 98
c=IN IP4 145.23.77.88
b=AS:75
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
```

17-18. SIP 200 (OK) response (from remote UE to SCC-AS)

The remote UE acknowledges the SIP re-INVITE request by sending SIP 200 (OK) response to the SCC-AS.

Table A.11.2-17 SIP 200 (OK) response (remote UE to SCC-AS)

```
SIP/2.0 200 OK
Via:
To:
From:
Call-ID:
CSeq:
P-Asserted-Identity:
Contact: sip:user3_public3@home3.net;gr=urn:uuid:f81d4fae-17oct-11a1-a678-0054c91eabcd
Allow:
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 1027933615 1027933615 IN IP4 123.112.67.87
s=-
c=IN IP4 123.112.67.87
t=0 0
m=audio 3000 RTP/AVP 96 97
b=AS:25.4
a=rtpmap:96 AMR
a=fmtp:96 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:97 telephone-event
a=maxptime:20
m=video 3002 RTP/AVP 98
b=AS:75
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
```

19-20. SIP ACK request (from SCC-AS to remote UE)

SCC-AS sends SIP ACK request to the remote UE.

21-22. SIP re-INVITE request (from SCC-AS to UE-2)

The SCC-AS sends a SIP re-INVITE request to the controllee UE, UE-2 to inform controllee UE about the port number for the video media component of the remote UE. The SCC AS add an a-line set to active in the SDP offer.

NOTE 1: This SIP re-INVITE request is triggered by the SIP REFER request in steps 1-2. The previous SIP INVITE request was generated by the SCC AS due to third party call control to allow sending this SIP re-INVITE request.

NOTE 2: Any other changes such as IP address of the remote UE in case remote UE uses different IP addresses for different media would also be updated in the SIP re-INVITE request.

Table A.11.2-21 SIP re-INVITE request (SCC-AS to UE-2)

```
INVITE sip:user1_public2@home2.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91e6bf6 SIP/2.0
Via:
To:
From:
Call-ID:
CSeq:
Contact: sip:user3_public3@home3.net;gr=urn:uuid:f81d4fae-17oct-11a1-a678-0054c91eabcd
Allow:
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 1027933615 1027933615 IN IP4 123.45.67.87
s=-
c=IN IP4 123.45.67.87
t=0 0
m=audio 0 RTP/AVP 96 97
m=video 3002 RTP/AVP 98
b=AS:75
a=active
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
```

23-24. SIP 200 (OK) response (from controllee UE to SCC-AS)

Table A.11.2-23 SIP 200 (OK) response (UE-2 to SCC-AS)

```
SIP/2.0 200 OK
Via:
To:
From:
Call-ID:
CSeq:
Contact: sip:user1_public2@home2.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91e6bf6
Allow:
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 1027933615 1027933615 IN IP4 145.23.77.88
s=-
t=0 0
m=audio 0 RTP/AVP 0
m=video 1302 RTP/AVP 98
c=145.23.77.88
b=AS:75
a=active
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
```

25-26. SIP ACK request (from SCC-AS to controllee UE)

SCC-AS sends SIP ACK request to controllee UE to acknowledge.

27-28. SIP NOTIFY request (from SCC-AS to conroller UE, UE-1)

When the media component is added to the controllee UE, UE-2, the SCC-AS sends SIP NOTIFY request to controller UE, UE-1 to inform about the success status of adding the media to the UE-2.

Table A.11.2-27 SIP NOTIFY request (SCC-AS to UE-1)

```
NOTIFY
Via:
To: sip:user1_public1@home1.net;tag = 13579
From: sip:scc-as@home1.net;tag=24680
Call-ID:
CSeq:
Max-Forwards:
P-Asserted-Identity:
Require:
Contact: sip: scc-as@home1.net
Allow:
Event: refer
Subscription-State:terminated; reason=noresource
Content-Type: message/sipfrag ;version=2.0
Content-Length: (…)

SIP/2.0 200 OK
Content-Type: application/sdp
m=audio 0 RTP/AVP 0
m=video 1302 RTP/AVP 98 99
c=145.23.77.88
b=AS:75
a=active
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
```

29-30. SIP 200 (OK) response (from controller UE to SCC-AS)

The controller UE acknowledges the NOTIFY request by sending SIP 200 (OK) response to SCC-AS.

A.12 Signalling flows for media transfer within collaborative session for inter-UE transfer

A.12.1 Introduction

This subclause describes signalling flows for media transfer within collaborative session. Two different scenarios are considered in the clause:

- subclause A.12.2 shows an example where media component is transferred from controller UE to controllee UE; and
- subclause A.12.3 shows an example where media component is transferred from one controllee UE to another controllee UE.

A.12.2 Controller UE initiated media transfer from controller UE to controllee UE

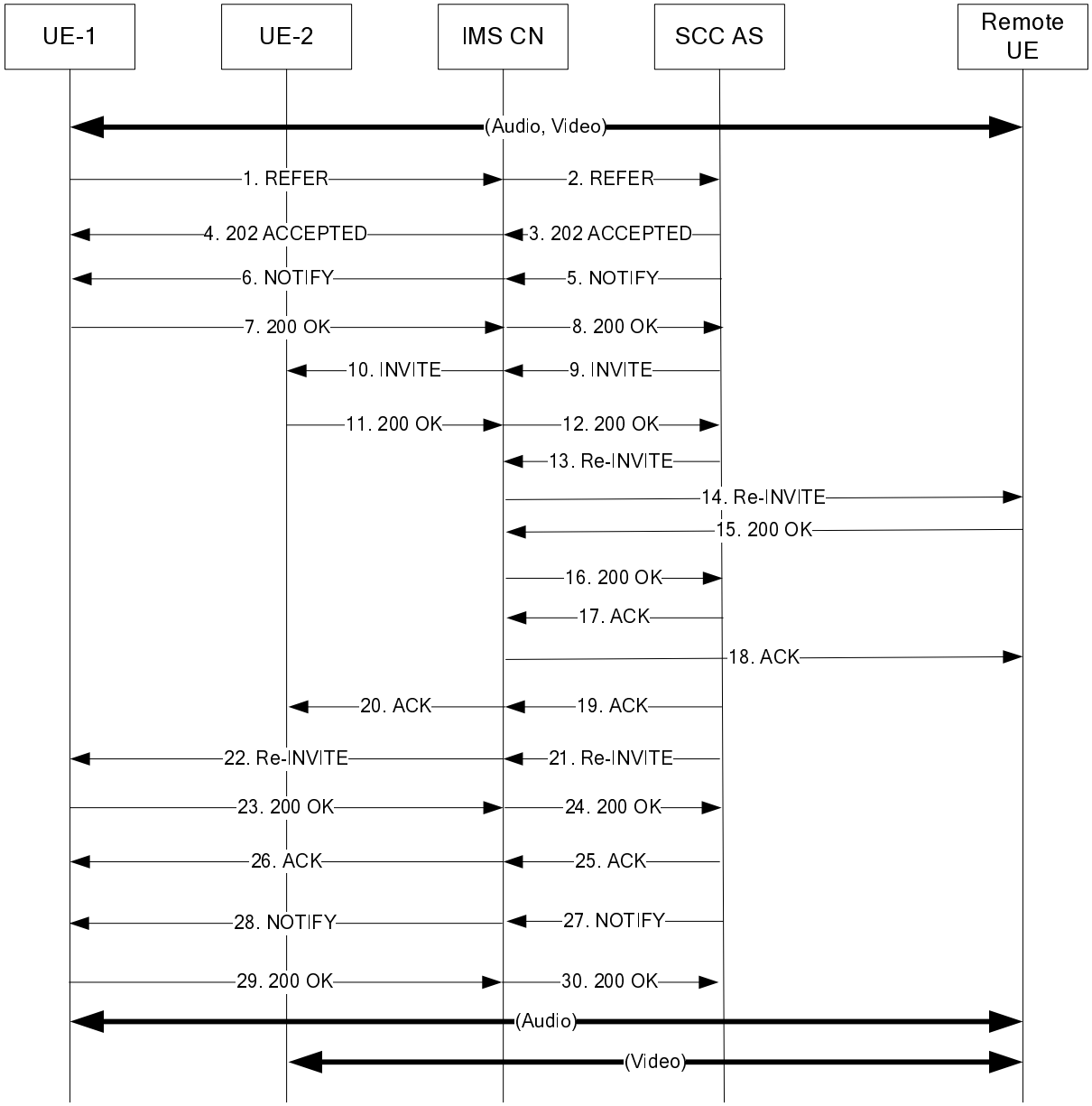


Figure A.12.2: Controller UE transfers a media on controllee UE

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

1-2. SIP REFER request (SIP REFER request from UE-1 to SCC-AS)

There is an existing session with audio and video between controller UE, UE-1 (123.45.67.89), and remote UE (132.54.76.98). The video component is unidirectional from the remote UE to the controller UE, UE1. The Controller UE attempts to transfer the video portion of this session to the controllee UE, UE-2.

Table A.12.2-1 SIP REFER request (UE-1 to SCC-AS)

```
REFER sip:interUEtransfer@sccas1.home1.net SIP/2.0
Via:
To: sip:interUEtransfer@sccas1.home1.net
From: sip:user1_public1@home1.net; tag=13579
Call-ID: cb03a0s09a2sdfglkj490333
CSeq: 93809824 REFER
Max-Forwards: 70
P-Preferred-Identity:
Refer-To: <sip:user1_public2@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91e6bf6?body=
   m%3Daudio%200%20RTP%2FAVP%97%0Dm%3Dvideo%203002%20RTP%2FAVP%2098%2099>
Require: target-dialog
Target-dialog: cb03a0s09a2sdfglkj321576;remote-tag=abcdef;local-tag=123456
Contact: <sip:user1_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-
   00a0c91ewxyz>;+g.3gpp.iut-controller
Accept: application/sdp, message/sipfrag
Content-Length: 0
```

3-4. SIP 202 (Accepted) response (from SCC AS to UE-1)

SCC-AS sends SIP 202 (Accepted) response to controller UE-1 as response to SIP REFER request.

5-6. SIP NOTIFY request (from SCC AS to UE-1)

SCC-AS sends SIP NOTIFY request to UE-1 to notify implicit subscription to the SIP REFER request results.

Table A.12.2-2 SIP NOTIFY request (SCC-AS to UE-1)

```
NOTIFY sip:user1_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91ewxyz SIP/2.0
Via:
To: sip:user1_public1@home1.net;tag=24680
From: sip:interUEtransfer@sccas1.home1.net;tag=13579
Call-ID:
CSeq:
Max-Forwards:
P-Asserted-Identity:
Require:
Contact: sip: scc-as@home1.net
Allow:
Event: refer
Subscription-State: active;expires=3600
Content-Type: message/sipfrag ;version=2.0
Content-Length: (…)

SIP/2.0 100 Trying
```

7-8. SIP 200 (OK) response (from UE-1 AS to SCC-AS)

The controller UE, UE-1, acknowledges the SIP NOTIFY request by sending SIP 200 (OK) response to SCC-AS.

9-10. SIP INVITE request (from SCC-AS to UE-2)

SCC-AS sends SIP INVITE request to the controllee UE, UE-2, to transfer video media.

Editor"s Note: It is ffs how the controller UE signals what media are to be transferred when the SDP media order is different in controller UE and controllee UE.

Table A.12.2-3 SIP INVITE request (SCC-AS to UE-2)

```
INVITE sip:user1_public2@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91e6bf6 SIP/2.0
Via:
To: sip:user1_public2@home1.net;
From: sip:scc-as@home1.net; tag=12486
Call-ID:
CSeq:
Max-Forwards:
P-Asserted-Identity:
Require:
Contact:
Allow:
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 1027933615 1027933615 IN IP4 123.112.67.87
s=-
c=IN IP4 123.112.67.87
t=0 0
m=audio 0 RTP/AVP 97
m=video 3002 RTP/AVP 98 99
b=AS:75
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
a=rtpmap:99 MP4V-ES
```

11-12. SIP 200 (OK) response (from UE-2 to SCC-AS)

The controllee UE, UE-2, acknowledges the SIP INVITE request by sending SIP 200 (OK) response to SCC-AS.

Table A.12.2-4 SIP 200 OK response (UE-2 to SCC-AS)

```
SIP/2.0 200 OK
Via:
To: sip:user1_public2@home1.net; tag = xyzwv
From: sip:scc-as@home1.net; tag = 12486
Call-ID:
CSeq:
P-Preferred-Identity:
Contact: sip:user1_public2@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91e6bf6
Allow:
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 1027933615 1027933615 IN IP4 145.23.77.88
s=-
c=145.23.77.88
t=0 0
m=audio 0 RTP/AVP 97
m=video 1302 RTP/AVP 98 99
b=AS:75
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
a=rtpmap:99 MP4V-ES
```

13-14. SIP re-INVITE request (from SCC-AS to remote UE)

SCC-AS sends SIP re-INVITE request to the remote UE.

Table A.12.2-5 SIP INVITE request (SCC-AS to remote UE)

```
INVITE sip:user3_public3@home3.net SIP/2.0
Via:
To: sip:user3_public3@home3.net;tag = 66666
From: sip:scc-as@home1.net; tag=33333
Call-ID:
CSeq:
Max-Forwards:
P-Asserted-Identity:
Require:
Contact: sip:user1_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91ewxyz
Allow:
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 1027933615 1027933615 IN IP4 123.45.67.89
s=-
t=0 0
m=audio 1300 RTP/AVP 96 97
c=IN IP4 123.45.67.89
b=AS:25.4
a=rtpmap:96 AMR
a=fmtp:96 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:97 telephone-event
a=maxptime:20
m=video 1302 RTP/AVP 98 99
c=IN IP4 145.23.77.88
b=AS:75
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
a=rtpmap:99 MP4V-ES
```

15-16. SIP 200 (OK) response (from remote UE to SCC-AS)

The remote UE acknowledges the SIP re-INVITE request by sending SIP 200 (OK) response to SCC-AS.

Table A.12.2-6 SIP 200 (OK) response (remote UE to SCC-AS)

```
SIP/2.0 200 OK
Via:
To:
From:
Call-ID:
CSeq:
P-Asserted-Identity:
Contact: sip:user3_public3@home3.net
Allow:
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 1027933615 1027933615 IN IP4 123.112.67.87
s=-
c=IN IP4 123.112.67.87
t=0 0
m=audio 3000 RTP/AVP 96 97
b=AS:25.4
a=rtpmap:96 AMR
a=fmtp:96 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:97 telephone-event
a=maxptime:20
m=video 3002 RTP/AVP 98 99
b=AS:75
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
a=rtpmap:99 MP4V-ES
```

17-18. SIP ACK request (from SCC-AS to remote UE)

SCC-AS sends SIP ACK request to the remote UE.

19-20. SIP ACK request (from SCC-AS to controllee UE; UE-2)

SCC-AS sends SIP ACK request to the remote UE.

21-22. SIP re-INVITE request (from SCC-AS to controller UE; UE-1)

SCC-AS sends SIP re-INVITE request to the controller UE.

Table A.12.2-7 SIP INVITE request (SCC-AS to UE-1)

```
INVITE sip:user1_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91ewxyz SIP/2.0
Via:
To:
From:
Call-ID:
CSeq:
Max-Forwards:
P-Asserted-Identity:
Require:
Contact:
Allow:
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 1027933615 1027933615 IN IP4 132.54.76.98
s=-
c=IN IP4 132.54.76.98
t=0 0
m=audio 2000 RTP/AVP 96 97
b=AS:25.4
a=rtpmap:96 AMR
a=fmtp:96 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:97 telephone-event
a=maxptime:20
m=video 0 RTP/AVP 98 99
a=sendonly
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
a=rtpmap:99 MP4V-ES
```

23-24. SIP 200 (OK) response (from UE-1 to SCC-AS)

The controller UE acknowledges the SIP re-INVITE request by sending SIP 200 (OK) response to SCC-AS.

25-26. SIP ACK request (from SCC-AS to UE-1)

SCC-AS sends SIP ACK request to the controlle1 UE, UE-1 in response to the SIP 200 (OK) response.

27-28. SIP NOTIFY request (from SCC-AS to conroller UE, UE-1)

SCC-AS sends SIP NOTIFY request to controller UE, UE-1 to inform about the success status if the inter-UE transfer.

Table A.12.2-8 SIP NOTIFY request (SCC-AS to UE-1)

```
NOTIFY sip:user1_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91ewxyz SIP/2.0
Via:
To: sip:user1_public1@home1.net; tag=24680
From: sip:interUEtransfer@sccas1.home1.net;tag=13579
Call-ID:
CSeq:
Max-Forwards:
P-Asserted-Identity:
Require:
Contact: sip: scc-as@home1.net
Allow:
Event: refer
Subscription-State:terminated; reason=noresource
Content-Type: message/sipfrag ;version=2.0
Content-Length: (…)

SIP/2.0 200 OK
Content-Type: application/sdp

v=0
s=-
m=audio 0 RTP/AVP 97
m=video 1302 RTP/AVP 98 99
b=AS:75
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
a=rtpmap:99 MP4V-ES
```

29-30. SIP 200 (OK) response (from controller UE to SCC-AS)

The remote UE acknowledges the SIP NOTIFY request by sending SIP 200 (OK) response to SCC-AS.

A.12.3 Controller UE initiated media transfer from controllee UE to another controllee UE with subscription to dialog events

The signalling flow in figure A.12.3 describes the procedures for media transfer from one controllee UE to another controllee UE to another controllee with presistant subscription to dialog event.

NOTE 1: For brevity not all SIP NOTIFY requests sent as a result of the subscription to the dialog event package are shown.

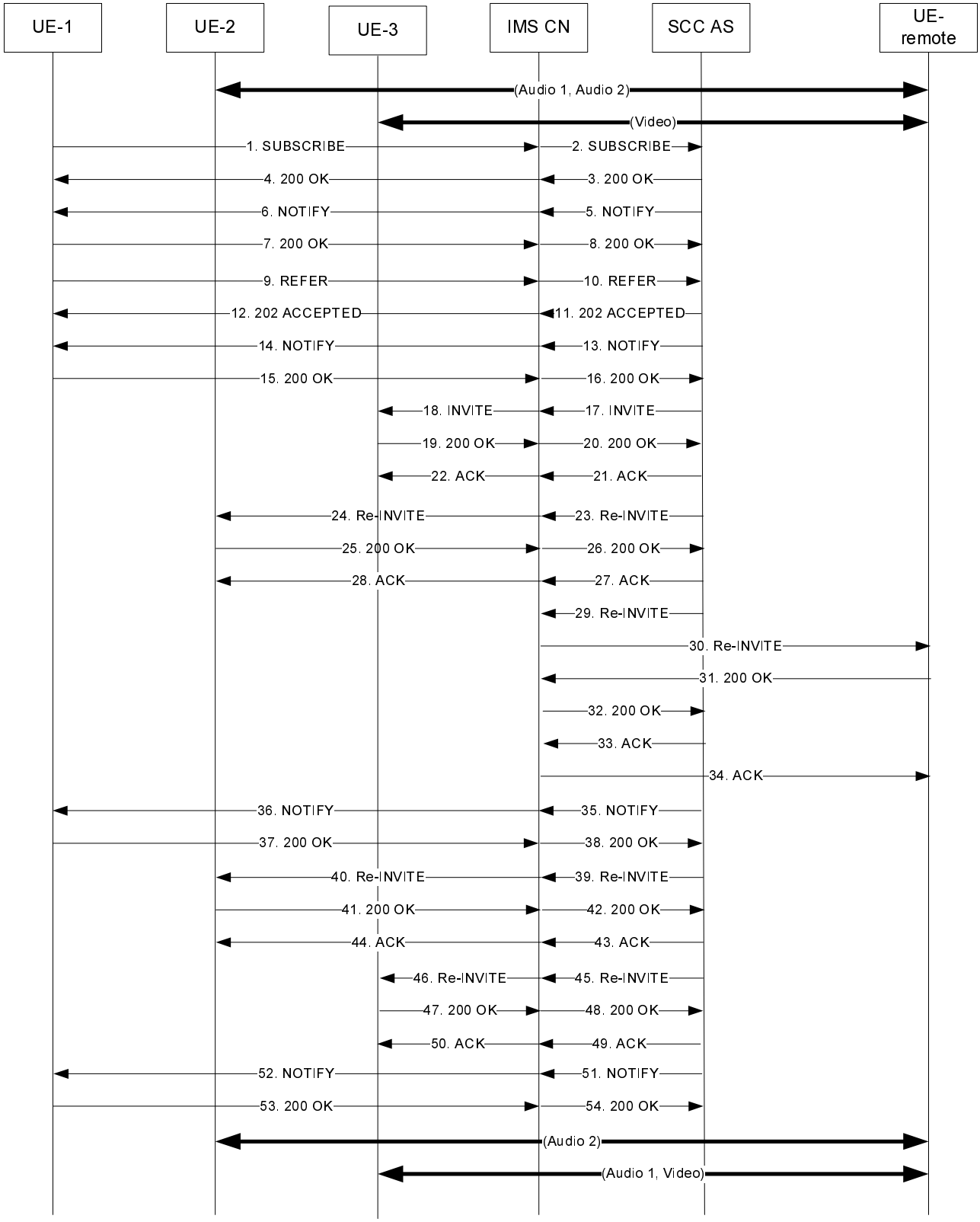


Figure A.12.3: Controller UE transfers a media from one controllee UE to another controllee UE

1-2. SIP SUBSCRIBE request (from controller UE to SCC-AS) - see example in table A.12.3-1

In order that the controller UE fetches the information about the session descriptions of the UEs within the collaborative session, the controller UE subscribes to the existing dialog between the controller UE and the SCC AS.

Table A.12.3-1 SIP SUBSCRIBE request (controller UE to SCC-AS)

```
SUBSCRIBE sip:scc-as@home1.net SIP/2.0
Via:
To: sip:scc-as@home1.net; tag= 24680
From: sip:user1_public1@home1.net; tag=13579
Event: dialog; call-id=" cb03a0s09a2sdfglkj321576"; from-tag="54321"; to-tag="123456";
   include-session-description
Call-ID: cb03a0s09a2sdfglkj490333
CSeq: 1 SUBSCRIBE
Max-Forwards: 70
P-Preferred-Identity:
Require: target-dialog
Expires: 3600
Target-dialog: cb03a0s09a2sdfglkj321576;remote-tag=abcdef;local-tag=123456
Contact: sip:user1_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91ewxyz
CSeq:
Allow:
Accept: application/dialog-info+xml
Content-Length: 0
```

3-4. SIP 200 (OK) response (from SCC-AS to controller UE)

The SCC AS acknowledges the SIP SUBSCRIBE request by sending SIP 200 (OK) response to the controller UE.

5-6. SIP NOTIFY request (from SCC-AS to controller UE) - see example in table A.12.3-5

The SCC AS sends SIP NOTIFY request containing SDP for the remote UE so that the controller UE has the current state of the media for the collaborative session.

Table A.12.3-5 SIP NOTIFY request (SCC-AS to controller UE)

```
NOTIFY sip:user1_public1@home1.net SIP/2.0
Via:
To: sip:user1_public1@home1.net; tag=13579
From: sip:scc-as@home1.net; tag= 24680
Call-ID:
CSeq:
Max-Forwards:
P-Asserted-Identity:
Require:
Contact: sip: scc-as@home1.net
Allow:
Event: dialog
Content-Type: application/dialog-info+xml
Content-Length: (…)

<?xml version="1.0"?>
<dialog-info xmlns="urn:ietf:params:xml:ns:dialog-info"
             version="0"
             state="full"
             entity="sip:scc-as@home1.net">
  <dialog id="xxxx" call-id="ffafa" local-tag="dd" remote-tag="ee">
    <state>confirmed</state>
    <local>
      <identity display="controllee UE">sip:user2_public1@home1.net</identity>
      <target>sip:user2_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-
   00a0c91e6bf6</target>
      <session-description type="application/sdp">
v=0
o=- 1027933615 1027933615 IN IP4 123.112.67.87
s=-
t=0 0
m=audio 49174 RTP/AVP 96 97
c=IN IP4 123.45.67.89
b=AS:25.4
a=rtpmap:96 AMR
a=fmtp:96 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:97 telephone-event
a=maxptime:20
m=audio 44552 RTP/AVP 96 97
c=IN IP4 123.45.67.89
b=AS:25.4
a=rtpmap:96 AMR
a=fmtp:96 mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:97 telephone-event
a=maxptime:20
m=video 1009 RTP/AVP 98 99
c=IN IP4 123.112.67.87
a=sendonly
b=AS:75
a=rtpmap:98 H263
a=fmtp:98 H263
a=fmtp:98 profile-level-id=0
a=rtpmap:99 MP4V-ES
    </session-description>
    </local>
    <remote>
      <identity display="remote UE">tel:+1-237-555-2222</identity>
      <target>sip:user2_public1@home2.net;gr=urn:uuid:2ad8950e-48a5-4a74-8d99-
   ad76cc7fc74</target>
     <session-description type="application/sdp">
v=0
o=- 1027933615 1027933615 IN IP4 123.112.67.87
s=-
t=0 0
m=audio 49174 RTP/AVP 96 97
c= IN IP4 132.54.76.98
b=AS:25.4
a=rtpmap:96 AMR
a=fmtp:96mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:97 telephone-event
a=maxptime:20
m=audio 44552 RTP/AVP 96 97
b=AS:25.4
a=rtpmap:96 AMR
a=fmtp:96mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:97 telephone-event
```

```
a=maxptime:20
c= IN IP4 132.54.76.98
m=video 1009 RTP/AVP 98 99
c= IN IP4 132.54.76.98
a=sendonly
b=AS:75
a=rtpmap:98 H263
a=fmtp:98 H263
a=fmtp:98 profile-level-id=0
a=rtpmap:99 MP4V-ES
      </session-description>
    </remote>
  </dialog>
</dialog-info>
```

7-8. SIP 200 (OK) response (from controllee UE-2 to SCC-AS)

The controller UE-1 acknowledges the SIP NOTIFY request by sending SIP 200 (OK) response to SCC AS.

9-10. SIP REFER request (UE-1 to SCC-AS) - see example in table A.12.3-9

There is an existing session with audio 1 and audio 2 between UE-2 (123.45.67.89), and remote UE (132.54.76.98). The video component is unidirectional from the remote UE to the controllee UE, UE-3 (123.112.67.87). The Controller UE attempts to transfer audio 1 portion of this session to the controllee UE, UE-3.

Table A.12.3-9 SIP REFER request (UE-1 to SCC-AS)

```
REFER sip:scc-as@home1.net SIP/2.0
Via:
To: sip:scc-as@home1.net; tag= 24680
From: sip:user1_public1@home1.net; tag=13579
Call-ID: cb03a0s09a2sdfglkj490333
CSeq: 93809824 REFER
Max-Forwards: 70
P-Preferred-Identity:
Refer-To: <sip:user1_public3@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-
   00a0c91e6bf6body=m%3Daudio%200%20RTP%2FAVP%200%0Dm%3Daudio%20
   49174%20RTP%2FAVP%2096%0Dm%3Dvideo%201009%20RTP%2FAVP%2098%2099>
Require: target-dialog
Target-dialog: cb03a0s09a2sdfglkj321576;remote-tag=abcdef;local-tag=123456
Contact: <sip:user1_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91ewxyz>
Accept: message/sipfrag
Content-Length: 0
```

11-12. SIP 202 (Accepted) response (from SCC AS to UE-1)

SCC-AS sends SIP 202 (Accepted) response to controller UE-1 as response to SIP REFER request.

13-14. SIP NOTIFY request (from SCC AS to UE-1) - see example in table A.13.3-13

SCC-AS sends SIP NOTIFY request to UE-1 to notify implicit subscription to the SIP REFER request results.

Table A.12.3-13 SIP NOTIFY request (SCC-AS to UE-1)

```
NOTIFY
Via:
To: sip:user1_public1@home1.net;tag=24680
From: sip:scc-as@home1.net;tag=13579
Call-ID:
CSeq:
Max-Forwards:
P-Asserted-Identity:
Require:
Contact: sip: scc-as@home1.net
Allow:
Event: refer
Subscription-State: active;expires=3600
Content-Type: message/sipfrag;version=2.0
Content-Length: (…)

SIP/2.0 100 Trying
```

15-16. SIP 200 (OK) response (from UE-1 AS to SCC-AS)

The controller UE, UE-1, acknowledges the SIP NOTIFY request by sending SIP 200 (OK) response to SCC-AS.

17-18. SIP INVITE request (from SCC-AS to UE-3) - see example in table A.12.3-17

Since the message 9-10 contain a Refer-to header field addressed to UE-3 and the URI paramaters, listing an audio line which is not currently supported by another controllee UE than UE-2, the SCC AS realizes the procedures is for transferring the media from that controllee UE (UE-2) to controllee UE (UE-3). SCC-AS sends SIP INVITE request to the controllee UE, UE-3, to transfer the audio media component. The SDP in the SIP INVITE request lists are the media lines within the collaborative session. In order to avoid UE-3 start sending audio to the remote UE, the SCC-AS adds an a-line to inactive in the SDP offer.

Table A.12.3-17 SIP INVITE request (SCC-AS to UE-3)

```
INVITE sip:user1_public3@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91e6bf6 SIP/2.0
Via:
To: sip:user1_public3@home1.net;
From: sip:scc-as@home1.net; tag=12486
Call-ID:
CSeq:
Max-Forwards:
P-Asserted-Identity:
Require:
Contact:
Allow:
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 1027933615 1027933615 IN IP4 132.54.76.98
s=-
c=IN IP4 132.54.76.98
t=0 0
m=audio 49174 RTP/AVP 96 97
a=rtpmap:0 PCMU/8000
a= sendonly
b=RR:0
b=RS:0m=audio 0 RTP/AVP 0
m=video 1009 RTP/AVP 98 99
a=sendonly
b=AS:75
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
a=rtpmap:99 MP4V-ES
```

19-20. SIP 200 (OK) response (from UE-3 to SCC-AS) - see example in table A.12.3-19

The controllee UE, UE-3, acknowledges the SIP INVITE request by sending SIP 200 (OK) response to SCC-AS.

Table A.12.3-19 SIP 200 (OK) response (UE-3 to SCC-AS)

```
SIP/2.0 200 OK
Via:
To: sip:user1_public3@home1.net; tag = xyzwv
From: sip:scc-as@home1.net; tag = 12486
Call-ID:
CSeq:
P-Preferred-Identity:
Contact: sip:user1_public3@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91e6bf6
Allow:
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 1027933615 1027933615 IN IP4 123.112.67.87
s=-
c=123.112.67.87
t=0 0
m=audio 3002 RTP/AVP 96 97
a=rtpmap:0 PCMU/8000
a=recvonly
m=audio 0 RTP/AVP 0
m=video 1302 RTP/AVP 98 99
a=recvonly
b=AS:75
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
a=rtpmap:99 MP4V-ES
```

21-22. SIP ACK request (from SCC-AS to UE-3)

SCC-AS sends a SIP ACK request to UE-3 to acknowlege

23-24. SIP re-INVITE request (from SCC-AS to controllee UE, UE-2) - see example in table A.12.3-23

The SCC AS sends SIP re-INVITE request to controllee UE, UE-2 to put Audio 1 on hold.

Table A.12.3-23 SIP INVITE request (SCC-AS to UE-2)

```
INVITE sip:user1_public2@home3.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91e6bf6 SIP/2.0
Via:
To: sip:user1_public2@home1.net;
From: sip:scc-as@home1.net; tag=12386
Call-ID:
CSeq:
Max-Forwards:
P-Asserted-Identity:
Require:
Contact:
Allow:
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 1027933615 1027933615 IN IP4 123.112.67.87
s=-
c=IN IP4 123.112.67.87
t=0 0
m=audio 49174 RTP/AVP 96 97
a=sendonly
b=RR:0
b=RS:0
m=audio 44552 RTP/AVP 96 97
a=rtpmap:0 PCMU/8000
a=sendonly
b=RR:0
```

25-26. SIP 200 (OK) response (from UE-2 to SCC-AS) - see example in table A.12.3-25

The controllee UE, UE-2, acknowledges the SIP INVITE request by sending SIP 200 (OK) response to SCC-AS.

Table A.12.3-25 SIP 200 (OK) response (UE-2 to SCC-AS)

```
SIP/2.0 200 OK
Via:
To: sip:user1_public2@home1.net; tag = xyzwv
From: sip:scc-as@home1.net; tag = 12486
Call-ID:
CSeq:
P-Preferred-Identity:
Contact: sip:user1_public2@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91e6bf6
Allow:
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 1027933615 1027933615 IN IP4 123.45.67.89
s=-
c=IN IP4 123.45.67.89
t=0 0
m=audio 32324 RTP/AVP 96 97
a=recvonly
m=audio 34002 RTP/AVP 96 97
a=rtpmap:0 PCMU/8000
a=recvonly
```

27-28. SIP ACK request (from SCC-AS to UE-2)

SCC –AS sends a SIP ACK request to UE-2 to acknowlege

29-30 SIP re-INVITE request (from SCC-AS to remote UE) - see example in table A.12.3-29

SCC-AS sends SIP re-INVITE request to the remote UE.

Table A.12.3-29 SIP INVITE request (SCC-AS to remote UE)

```
INVITE sip:user2_public1@home3.net;SIP/2.0
Via:
To: sip:user2_public1@home2.net;tag=66666
From: sip:scc-as@home1.net; tag=33333
Call-ID:
CSeq:
Max-Forwards:
P-Asserted-Identity:
Require:
Contact:sip:user1_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91ewxyz
Allow:
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 1027933615 1027933615 IN IP4 123.112.67.87
s=-
t=0 0
m=audio 3002 RTP/AVP 96 97
c= IN IP4 123.112.67.87
a=rtpmap:0 PCMU/8000
m=audio 34002 RTP/AVP 96 97
C=IN IP4 123.45.67.89
A=rtpmap:0 PCMU/8000
m=video 1302 RTP/AVP 98 99
c= IN IP4 123.112.67.87
b=AS:75
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
a=rtpmap:99 MP4V-ES
a=recvonly
```

31-32. SIP 200 (OK) response (from remote UE to SCC-AS) - see example in table A.12.3-31

The remote UE acknowledges the SIP re-INVITE request by sending SIP 200 (OK) response to SCC-AS.

Table A.12.3-31 SIP 200 (OK) response (remote UE to SCC-AS)

```
SIP/2.0 200 OK
Via:
To:
From:
Call-ID:
CSeq:
P-Asserted-Identity:
Contact: sip:user2_public1@home2.net;
Allow:
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 1027933615 1027933615 IN IP4 132.54.76.98
s=-
c= IN IP4 132.54.76.98
t=0 0
m=audio 49174 RTP/AVP 96 97
b=AS:25.4
a=rtpmap:96 AMR
a=fmtp:96mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:97 telephone-event
a=maxptime:20
m=audio 44552 RTP/AVP 96 97
a=rtpmap:0 PCMU/8000
m=video 1009 RTP/AVP 98 99
a=sendonly
b=AS:75
a=rtpmap:98 H263
a=fmtp:98 H263
a=fmtp:98 profile-level-id=0
a=rtpmap:99 MP4V-ES
a=sendonly
```

33-34. SIP ACK request (from SCC-AS to remote UE)

SCC-AS sends a SIP ACK request to the remote UE.

35-36. SIP NOTIFY request (from SCC-AS to controller UE) - see example in table A.12.3-35

The SCC AS sends SIP NOTIFY request containing SDP for the remote UE so that the controller UE can be aware about the change of state of the media for the collaborative session.

Table A.12.3-35 SIP NOTIFY request (SCC-AS to controller UE)

```
NOTIFY sip:user1_public1@home1.net;
Via:
To: sip:user1_public1@home1.net; tag=13579
From: sip:scc-as@home1.net; tag= 24680
Call-ID:
CSeq:
Max-Forwards:
P-Asserted-Identity:
Require:
Contact: sip: scc-as@home1.net
Allow:
Event: dialog
Content-Type: application/dialog-info+xml
Content-Length: (…)

<?xml version="1.0"?>
<dialog-info xmlns="urn:ietf:params:xml:ns:dialog-info"
             version="1"
             state="full"
             entity="sip:scc-as@home1.net">
  <dialog id="xxxx" call-id="ffafa" local-tag="dd" remote-tag="ee">
    <state>confirmed</state>
    <local>
      <identity display="controllee UE">sip:user2_public1@home1.net</identity>
      <target>sip:user2_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-
   00a0c91e6bf6</target>
      <session-description type="application/sdp">
v=0
o=- 1027933615 1027933615 IN IP4 123.112.67.87
s=-
t=0 0
m=audio 49174 RTP/AVP 96 97
c=IN IP4 123.45.67.89
b=AS:25.4
a=rtpmap:96 AMR
a=fmtp:96mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:97 telephone-event
a=maxptime:20
m=audio 44552 RTP/AVP 96 97
c=IN IP4 123.45.67.89
b=AS:25.4
a=rtpmap:96 AMR
a=fmtp:96mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:97 telephone-event
a=maxptime:20
m=video 1009 RTP/AVP 98 99
c=IN IP4 123.112.67.87
a=sendonly
b=AS:75
a=rtpmap:98 H263
a=fmtp:98 H263
a=fmtp:98 profile-level-id=0
a=rtpmap:99 MP4V-ES
      </session-description>
    </local>
    <remote>
      <identity display="remote UE">tel:+1-237-555-2222</identity>
      <target>sip:user2_public1@home2.net;gr=urn:uuid:2ad8950e-48a5-4a74-8d99-
   ad76cc7fc74</target>
      <session-description type="application/sdp">
v=0
o=- 1027933615 1027933615 IN IP4 123.112.67.87
s=-
t=0 0
c= IN IP4 132.54.76.98
m=audio 49174 RTP/AVP 96 97
b=AS:25.4
a=rtpmap:96 AMR
a=fmtp:96mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:97 telephone-event
a=maxptime:20
m=audio 44552 RTP/AVP 96 97
b=AS:25.4
a=rtpmap:96 AMR
a=fmtp:96mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:97 telephone-event
```

```
a=maxptime:20
m=video 1009 RTP/AVP 98 99
a=sendonly
b=AS:75
a=rtpmap:98 H263
a=fmtp:98 H263
a=fmtp:98 profile-level-id=0
a=rtpmap:99 MP4V-ES
      </session-description>
    </remote>
  </dialog>
</dialog-info>
```

37-38. SIP 200 (OK) response (from controllee UE-2 to SCC-AS)

The controller UE-1 acknowledges the SIP NOTIFY request by sending SIP 200 (OK) response to SCC AS.

39-40. SIP re-INVITE request (from SCC-AS to controllee UE, UE-2) - see example in table A.12.3-39

SCC-AS sends SIP re_INVITE request to UE-2 to remove the audio 1 media component.

Table A.12.3-39 SIP INVITE request (SCC-AS to UE-2)

```
INVITE sip:user1_public2@home3.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91e6bf6 SIP/2.0
Via:
To: sip:user1_public2@home1.net;
From: sip:scc-as@home1.net; tag=12386
Call-ID:
CSeq:
Max-Forwards:
P-Asserted-Identity:
Require:
Contact:
Allow:
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 1027933615 1027933615 IN IP4 123.112.67.87
s=-
c=IN IP4 123.112.67.87
t=0 0
m=audio 0 RTP/AVP 0
m=audio 44552 RTP/AVP 96 97
a=rtpmap:0 PCMU/8000
```

41-42. SIP 200 (OK) response (from UE-2 to SCC-AS) - see example in table A.12.3-41

The controllee UE, UE-2, acknowledges the SIP INVITE request by sending SIP 200 (OK) response to SCC-AS.

Table A.12.3-41 SIP 200 (OK) response (UE-2 to SCC-AS)

```
SIP/2.0 200 OK
Via:
To: sip:user1_public2@home1.net; tag = xyzwv
From: sip:scc-as@home1.net; tag = 12486
Call-ID:
CSeq:
P-Preferred-Identity:
Contact: sip:user1_public2@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91e6bf6
Allow:
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 1027933615 1027933615 IN IP4 123.45.67.89
s=-
c=IN IP4 123.45.67.89
t=0 0
m=audio 0 RTP/AVP 0
m=audio 34002 RTP/AVP 96 97
a=rtpmap:0 PCMU/8000
```

43-44. SIP ACK request (from SCC-AS to UE-2)

SCC –AS sends a SIP ACK request to UE-2 to acknowlege

45-46. SIP re_INVITE request (from SCC-AS to controllee UE; UE-3) - see example in table A.12.3-45

SCC-AS sends SIP re_INVITE request to UE-3 to activate the audio 1 media component.

Table A.12.3-45 SIP re-INVITE request (SCC-AS to UE-3)

```
INVITE sip:user1_public3@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91e6bf6 SIP/2.0
Via:
To: sip:user1_public3@home1.net;
From: sip:scc-as@home1.net; tag=12486
Call-ID:
CSeq:
Max-Forwards:
P-Asserted-Identity:
Require:
Contact:
Allow:
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 1027933615 1027933615 IN IP4 123.112.67.87
s=-
c=IN IP4 123.112.67.87
t=0 0
m=audio 49174 RTP/AVP 96 97
a=rtpmap:0 PCMU/8000
a= sendrecv
m=audio 0 RTP/AVP 0
m=video 1009 RTP/AVP 98 99
b=AS:75
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
a=rtpmap:99 MP4V-ES
a=sendonly
```

NOTE 2: This SIP re-INVITE request is triggered by the SIP REFER request. The previous SIP INVITE request was generated by the SCC AS due to third party call control to allow sending this SIP re-INVITE request.

47-48. SIP 200 (OK) response (from controllee UE; UE-3 to SCC AS)

Controllee UE, UE-3 sends SIP 200 (OK) response to SCC AS.

49-50. SIP ACK request (from SCC-AS to UE-3)

SCC-AS sends a SIP ACK request to the remote UE.

51-52. SIP NOTIFY request (from SCC-AS to controller UE) - see example in table A.12.3-51

The SCC AS sends SIP NOTIFY request containing SDP SIP 200 (OK) response received from the controllee UE; UE-3.

Table A.12.3-51 SIP NOTIFY request (SCC-AS to controller UE)

```
NOTIFY sip:user1_public1@home1.net;
Via:
To: sip:user1_public1@home1.net; tag=13579
From: sip:scc-as@home1.net; tag= 24680
Call-ID:
CSeq:
Max-Forwards:
P-Asserted-Identity:
Require:
Contact: sip: scc-as@home1.net
Allow:
Event: refer
Content-Type: message/sipfrag ;version=2.0 ;
Content-Length: (…)

SIP/2.0 200 OK
To: sip:user1_public3@home1.net; tag = xyzwv
From: sip:scc-as@home1.net; tag = 12486
Call-ID:
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 1027933615 1027933615 IN IP4 123.112.67.87
s=-
c=123.112.67.87
t=0 0
m=audio 3002 RTP/AVP 96 97
a=rtpmap:0 PCMU/8000
m=audio 0 RTP/AVP 0
m=video 1302 RTP/AVP 98 99
a=recvonly
b=AS:75
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
a=rtpmap:99 MP4V-ES
```

53-54. SIP 200 (OK) response (from controller UE to SCC-AS)

The controller UE acknowledges the SIP NOTIFY request by sending SIP 200 (OK) response to SCC AS.

A.12.3A Controller UE initiated media transfer from controllee UE to another controllee UE

The signalling flow in figure A.12.3A describes the procedures for media transfer from one controllee UE to another controllee UE.

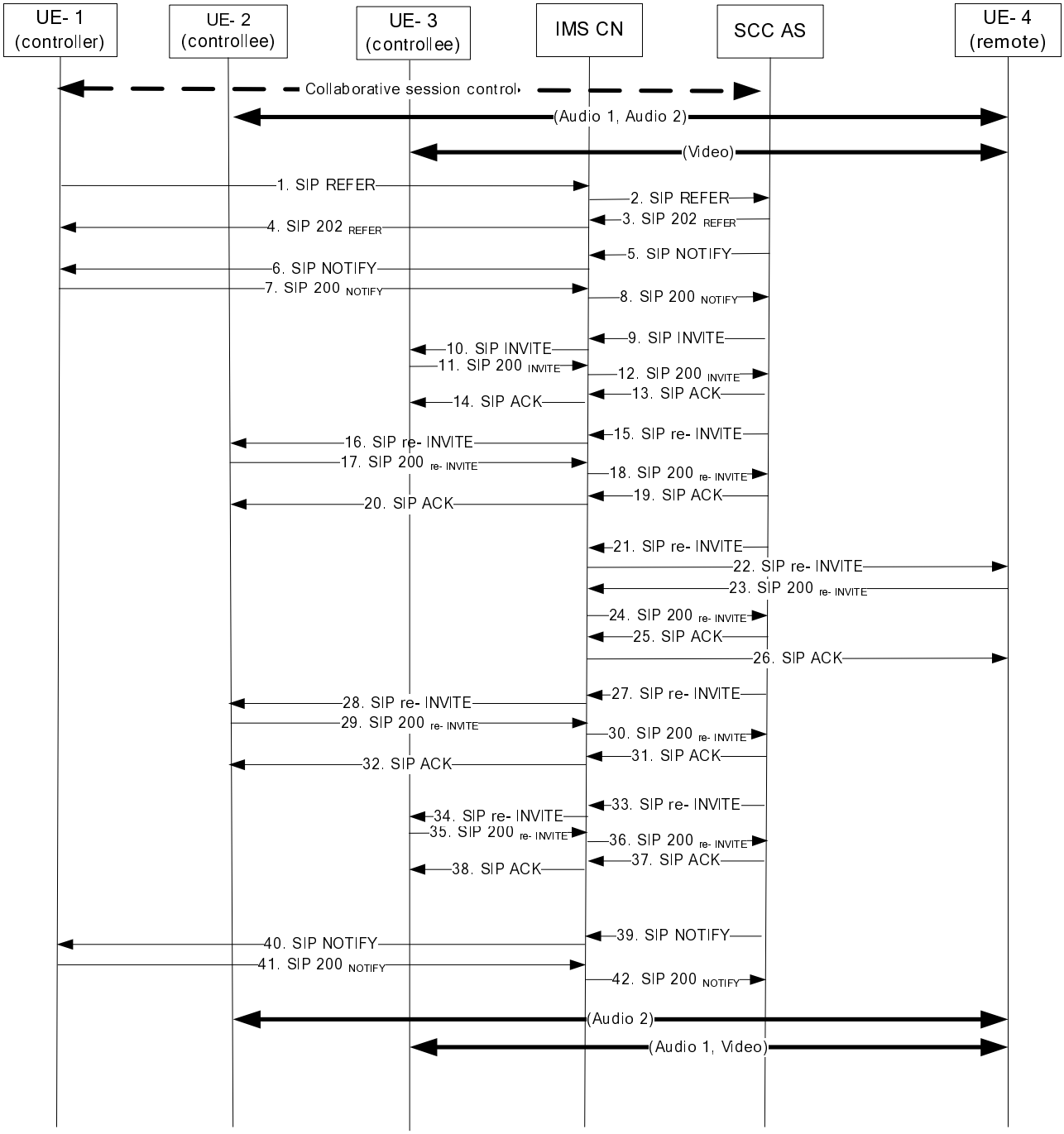


Figure A.12.3A: Controller UE transfers a media from one controllee UE to another controllee UE.

1-2. SIP REFER request (SIP REFER request from UE-1 to SCC-AS)

There is an existing session with audio 1 and audio 2 between UE-2 (123.45.67.89), and remote UE (132.54.76.98). The video component is unidirectional from the remote UE to the controllee UE, UE-3 (123.112.67.87). The Controller UE attempts to transfer audio 1 portion of this session to the controllee UE, UE-3.

Table A.12.3A-1 SIP REFER request (UE-1 to SCC-AS)

```
REFER sip:scc-as@home1.net SIP/2.0
Via:
To: sip:scc-as@home1.net; tag= 24680
From: sip:user1_public1@home1.net; tag=13579
Call-ID: cb03a0s09a2sdfglkj490333
CSeq: 93809824 REFER
Max-Forwards: 70
P-Preferred-Identity:
Refer-To: <sip:user1_public3@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-
   00a0c91e6bf6body=m%3Daudio%200%20RTP%2FAVP%200%0Dm%3Daudio%20
   49174%20RTP%2FAVP%2096%0Dm%3Dvideo%201009%20RTP%2FAVP%2098%2099>
Require: target-dialog
Target-dialog: cb03a0s09a2sdfglkj321576;remote-tag=abcdef;local-tag=123456
Contact: <sip:user1_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91ewxyz>
Accept: message/sipfrag
Content-Length: 0
```

3-4. SIP 202 (Accepted) response (from SCC AS to UE-1)

SCC-AS sends SIP 202 (Accepted) response to controller UE-1 as response to SIP REFER request.

5-6. SIP NOTIFY request (from SCC AS to UE-1)

SCC-AS sends SIP NOTIFY request to UE-1 to notify implicit subscription to the SIP REFER request results.

Table A.12.3A-2 SIP NOTIFY request (SCC-AS to UE-1)

```
NOTIFY

To: sip:user1_public1@home1.net;tag=24680
From: sip:scc-as@home1.net;tag=13579
Call-ID: cb03a0s09a2sdfglkj12345
CSeq:
Max-Forwards:
P-Asserted-Identity:
Require:
Contact: sip:scc-as@home1.net
Allow:
Event: refer
Subscription-State:active;expires=3600
Content-Type: message/sipfrag;version=2.0
Content-Length: (…)

SIP/2.0 100 Trying
```

7-8. SIP 200 (OK) response (from UE-1 AS to SCC-AS)

The controller UE, UE-1, acknowledges the SIP NOTIFY request by sending SIP 200 (OK) response to SCC-AS.

9-10. SIP INVITE request (from SCC-AS to UE-3)

Since the message 1-2 contain a Refer-to header field addressed to UE-3 and the URI paramaters, listing an audio line which is not currently supported by another controllee UE than UE-2, the SCC AS realizes the procedures is for transferring the media from that controllee UE (UE-2) to controllee UE (UE-3). SCC-AS sends SIP INVITE request to the controllee UE, UE-3, to transfer the audio media component. The SDP in the SIP INVITE request lists are the media lines within the collaborative session. In order to avoid UE-3 start sending audio to the remote UE, the SCC-AS adds an a-line to sendonly in the SDP offer.

Table A.12.3A-3 SIP INVITE request (SCC-AS to UE-3)

```
INVITE sip:user1_public3@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91e6bf6 SIP/2.0
Via:
To: sip:user1_public3@home1.net;
From: sip:scc-as@home1.net; tag=12486
Call-ID:
CSeq:
Max-Forwards:
P-Asserted-Identity:
Require:
Contact:
Allow:
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 1027933615 1027933615 IN IP4 132.54.76.98
s=-
c=IN IP4 132.54.76.98
t=0 0
m=audio 49174 RTP/AVP 96 97
a=rtpmap:0 PCMU/8000
a= sendonly
b=RR:0
b=RS:0m=audio 0 RTP/AVP 0
m=video 1009 RTP/AVP 98 99
a=sendonly
b=AS:75
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
a=rtpmap:99 MP4V-ES
```

11-12. SIP 200 (OK) response (from UE-3 to SCC-AS)

The controllee UE, UE-3, acknowledges the SIP INVITE request by sending SIP 200 (OK) response to SCC-AS.

Table A.12.3A-4 SIP 200 OK request (UE-3 to SCC-AS)

```
SIP/2.0 200 OK
Via:
To: sip:user1_public3@home1.net; tag = xyzwv
From: sip:scc-as@home1.net; tag = 12486
Call-ID:
CSeq:
P-Preferred-Identity:
Contact: sip:user1_public3@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91e6bf6
Allow:
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 1027933615 1027933615 IN IP4 123.112.67.87
s=-
c=123.112.67.87
t=0 0
m=audio 3002 RTP/AVP 96 97
a=rtpmap:0 PCMU/8000
a=recvonly
m=audio 0 RTP/AVP 0
m=video 1302 RTP/AVP 98 99
a=recvonly
b=AS:75
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
a=rtpmap:99 MP4V-ES
```

13-14. SIP ACK request(from SCC-AS to UE-3)

SCC-AS sends a SIP ACK request to UE-3 to acknowledge.

15-16. SIP re-INVITE request (from SCC-AS to controllee UE, UE-2)

The SCC AS sends SIP re-INVITE request to controllee UE, UE-2 to put Audio 1 on hold.

Table A.12.3A-5 SIP INVITE request (SCC-AS to controllee UE, UE-2)

```
INVITE sip:user1_public2@home3.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91e6bf6 SIP/2.0
Via:
To: sip:user1_public2@home1.net;
From: sip:scc-as@home1.net; tag=12386
Call-ID:
CSeq:
Max-Forwards:
P-Asserted-Identity:
Require:
Contact:
Allow:
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 1027933615 1027933615 IN IP4 132.54.76.98
s=-
c=IN IP4 132.54.76.98
t=0 0
m=audio 49174 RTP/AVP 96 97
a=sendonly
b=RR:0
b=RS:0
m=audio 44552 RTP/AVP 96 97
a=rtpmap:0 PCMU/8000
a=sendonly
b=RR:0
```

17-18. SIP 200 (OK) response (from UE-2 to SCC-AS)

The controllee UE, UE-2 acknowledges the SIP INVITE request by sending SIP 200 (OK) response to SCC-AS.

Table A.12.3A-6 SIP 200 (OK) response (UE-2 to SCC-AS)

```
SIP/2.0 200 OK
Via:
To: sip:user1_public2@home1.net; tag = xyzwv
From: sip:scc-as@home1.net; tag = 12486
Call-ID:
CSeq:
P-Preferred-Identity:
Contact: sip:user1_public2@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91e6bf6
Allow:
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 1027933615 1027933615 IN IP4 123.45.67.89
s=-
c=IN IP4 123.45.67.89
t=0 0
m=audio 32324 RTP/AVP 96 97
a=recvonly
m=audio 34002 RTP/AVP 96 97
a=rtpmap:0 PCMU/8000
a=recvonly
```

19-20. SIP ACK request (from SCC-AS to UE-2)

SCC-AS send a SIP ACK request to UE-2 to acknowledge.

21-22. SIP re-INVITE request (from SCC-AS to remote UE)

SCC-AS sends SIP re-INVITE request to the remote UE, UE-4.

Table A.12.3A-7 SIP INVITE request (SCC-AS to remote UE, UE-4)

```
INVITE sip:user2_public1@home3.net;SIP/2.0
Via:
To: sip:user2_public1@home2.net;tag=66666
From: sip:scc-as@home1.net; tag=33333
Call-ID:
CSeq:
Max-Forwards:
P-Asserted-Identity:
Require:
Contact:sip:user1_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91ewxyz
Allow:
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 1027933615 1027933615 IN IP4 123.112.67.87
s=-
t=0 0
m=audio 3002 RTP/AVP 96 97
c= IN IP4 123.112.67.87
a=rtpmap:0 PCMU/8000
m=audio 34002 RTP/AVP 96 97
C=IN IP4 123.45.67.89
A=rtpmap:0 PCMU/8000
m=video 1302 RTP/AVP 98 99
c= IN IP4 123.112.67.87
b=AS:75
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
a=rtpmap:99 MP4V-ES
a=recvonly
```

23-24. SIP 200 (OK) response (from remote UE, UE-4 to SCC-AS)

The remote UE acknowledges the SIP re-INVITE request by sending SIP 200 (OK) response to SCC-AS.

Table A.12.3A-8 SIP 200 OK request (UE-4 to SCC-AS)

```
SIP/2.0 200 OK
Via:
To:
From:
Call-ID:
CSeq:
P-Asserted-Identity:
Contact: sip:user2_public1@home2.net;
Allow:
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 1027933615 1027933615 IN IP4 132.54.76.98
s=-
c= IN IP4 132.54.76.98
t=0 0
m=audio 49174 RTP/AVP 96 97
b=AS:25.4
a=rtpmap:96 AMR
a=fmtp:96mode-set=0,2,5,7; mode-change-period=2
a=rtpmap:97 telephone-event
a=maxptime:20
m=audio 44552 RTP/AVP 96 97
a=rtpmap:0 PCMU/8000
m=video 1009 RTP/AVP 98 99
a=sendonly
b=AS:75
a=rtpmap:98 H263
a=fmtp:98 H263
a=fmtp:98 profile-level-id=0
a=rtpmap:99 MP4V-ES
a=sendonly
```

25-26. SIP ACK request (from SCC-AS to remote UE)

SCC-AS sends SIP ACK request to the remote UE.

27-28. SIP re-INVITE request (from SCC-AS to controllee UE, UE-2)

SCC-AS sends SIP re-INVITE request to UE-2 to remove the audio 1 media component.

Table A.12.3A-9 SIP re-INVITE request (SCC-AS to UE-2)

```
INVITE sip:user1_public2@home3.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91e6bf6 SIP/2.0
Via:
To: sip:user1_public2@home1.net;
From: sip:scc-as@home1.net; tag=12386
Call-ID:
CSeq:
Max-Forwards:
P-Asserted-Identity:
Require:
Contact:
Allow:
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 1027933615 1027933615 IN IP4 132.54.76.98
s=-
c=IN IP4 132.54.76.98
t=0 0
m=audio 0 RTP/AVP 0
m=audio 44552 RTP/AVP 96 97
a=rtpmap:0 PCMU/8000
```

29-30, SIP 200 (OK) response (from UE-2 to SCC-AS)

The controllee UE, UE-2, acknowledges the SIP INVITE request by sending SIP 200 (OK) response to SCC-AS.

Table A.12.3A-10 SIP 200 (OK) response (UE-2 to SCC-AS)

```
SIP/2.0 200 OK
Via:
To: sip:user1_public2@home1.net; tag = xyzwv
From: sip:scc-as@home1.net; tag = 12486
Call-ID:
CSeq:
P-Preferred-Identity:
Contact: sip:user1_public2@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91e6bf6
Allow:
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 1027933615 1027933615 IN IP4 123.45.67.89
s=-
c=IN IP4 123.45.67.89
t=0 0
m=audio 0 RTP/AVP 0
m=audio 34002 RTP/AVP 96 97
a=rtpmap:0 PCMU/8000
```

31-32. SIP ACK request (from SCC-AS to UE-2)

SCC-AS sends a SIP ACK request to UE-2 to acknowledge.

33-34. SIP re-INVITE request (from SCC-AS to controllee UE, UE-3)

SCC-AS sends a SIP re-INVITE request to UE-3 to activate the audio 1 media component.

NOTE: The SIP re-INVITE request is triggered by the SIP REFER request. The previous SIP INVITE request was generated by the SCC AS due to third party call cotrol to allow sending this SIP re-INVITE request.

Table A.12.3A-11 SIP re-INVITE request (SCC-AS to UE-3)

```
INVITE sip:user1_public3@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91e6bf6 SIP/2.0
Via:
To: sip:user1_public3@home1.net;
From: sip:scc-as@home1.net; tag=12486
Call-ID:
CSeq:
Max-Forwards:
P-Asserted-Identity:
Require:
Contact:
Allow:
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 1027933615 1027933615 IN IP4 123.112.67.87
s=-
c=IN IP4 123.112.67.87
t=0 0
m=audio 49174 RTP/AVP 96 97
a=rtpmap:0 PCMU/8000
a= sendrecv
m=audio 0 RTP/AVP 0
m=video 1009 RTP/AVP 98 99
b=AS:75
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
a=rtpmap:99 MP4V-ES
a=sendonly
```

35-36. SIP 200 (OK) response (from controllee UE, UE-3 to SCC-AS)

Controllee UE, UE-3 sends SIP 200 (OK) response to SCC AS.

37-38. SIP ACK request (from SCC-AS to UE-3)

SCC-AS sends a SIP ACK request to the remote UE.

39-40. SIP NOTIFY request (from SCC-AS to controller UE, UE-1)

The SCC AS sends SIP NOTIFY request containing SDP SIP 200 (OK) response received from the controllee UE, UE-3.

Table A.12.3A-12 SIP NOTIFY request (SCC-AS to UE-1)

```
NOTIFY sip:user1_public1@home1.net;
Via:
To: sip:user1_public1@home1.net; tag=13579
From: sip:scc-as@home1.net; tag= 24680
Call-ID:
CSeq:
Max-Forwards:
P-Asserted-Identity:
Require:
Contact: sip: scc-as@home1.net
Allow:
Event: refer
Content-Type: message/sipfrag ;version=2.0 ;
Content-Length: (…)

SIP/2.0 200 OK
To: sip:user1_public3@home1.net; tag = xyzwv
From: sip:scc-as@home1.net; tag = 12486
Call-ID:
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 1027933615 1027933615 IN IP4 123.112.67.87
s=-
c=123.112.67.87
t=0 0
m=audio 3002 RTP/AVP 96 97
a=rtpmap:0 PCMU/8000
m=audio 0 RTP/AVP 0
m=video 1302 RTP/AVP 98 99
a=recvonly
b=AS:75
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
a=rtpmap:99 MP4V-ES
```

41-42. SIP 200 (OK) response (from controller UE to SCC-AS)

The controller UE acknowledges the SIP NOTIFY request by sending SIP 200 (OK) response to SCC-AS.

A.13 Signalling flows for release of collaborative session for inter-UE transfer

A.13.1 Introduction

The signalling flows for release of Collaborative Session demonstrate how the session is released by the Controller UE or by the remote party UE. The following signalling flow is included:

- subclause A.13.2 shows an example where the Controller UE initiates release of a Collaborative Session. It demonstrates how the service control signalling and media path between the Controller UE and remote UE and the media path between the Controlee UE and the remote UE are released as a result of the session release; and
- subclause A.13.3 shows an example where the remote UE initiates release of a Collaborative Session.

A.13.2 Controller UE releases collaborative session

In this example, session release is initiated by the Controller UE (UE 1), which is involved in the Collaborative Session with UE 2 and the remote UE (UE 3). The SCC AS ensures that all Controlee UEs involved in the Collaborative Session receive a request for session release in order to completely release the session ongoing with the remote UE.

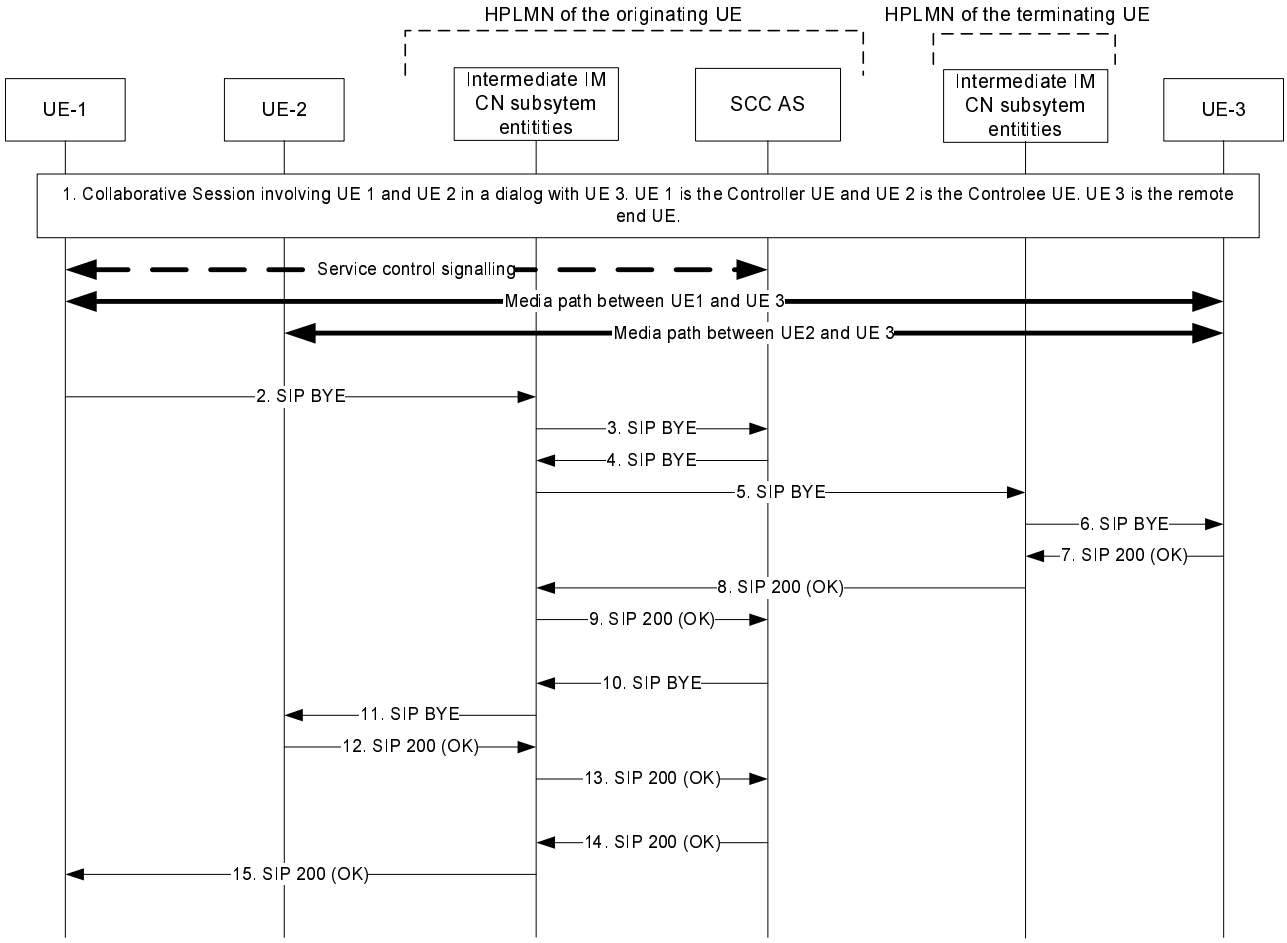


Figure A.13.2-1: Release of Collaborative Session initiated by Controller UE

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

1. Collaborative Session currently exists in a dialog with UE 3

A Collaborative Session involving UE 1 and UE 2 exists in a dialog with UE 3. Media paths exist between UE 1 and UE 3 and between UE 2 and UE 3. In this scenario, UE 1 is the Controller UE in the Collaborative Session and thus maintains service control signalling with the SCC AS.

2-3. SIP BYE request (UE 1 to SCC AS via intermediate IM CN subsystem entities)

UE 1, acting as the Controller UE, initiates session release by sending a SIP BYE request towards the SCC AS. There is no Inter UE transfer specific content in the SIP BYE request.

4-6. SIP BYE request (SCC AS to UE 3 via intermediate IM CN subsystem entities)

The SCC AS routes the SIP BYE request to UE 3 indicating to the remote UE that that the Controller UE requests that the session is to be released.

7-9. SIP 200 (OK) response (UE 3 to SCC AS via intermediate IM CN subsystem entities)

UE 3 responds to the received SIP BYE request with a SIP 200 (OK) response.

10-11. SIP BYE request (SCC AS to UE 2 via intermediate IM CN subsystem entities)

The SCC AS, acting as a routing B2BUA, sends a SIP BYE request towards UE 2 to release the dialog it is involved in with UE 3 via the Collaborative Session. There is no Inter UE transfer specific content in the SIP BYE request.

NOTE 1: The SIP BYE request to UE 2 (step 10) can occur in parallel with the SIP BYE request to UE 3 (step 4). Alternatively, the SCC AS can send the SIP BYE request to UE 2 prior to sending the SIP BYE request to UE 3.

12-13. SIP 200 (OK) response (UE 2 to SCC AS via intermediate IM CN subsystem entities)

UE 2 responds to the received SIP BYE request with a SIP 200 (OK) response.

14-15 SIP 200 (OK) response (SCC AS to UE 1 via intermediate IM CN subsystem entities)

Upon receiving SIP 200 (OK) responses from UE 2 and UE 3, the SCC AS responds to the SIP BYE request from UE 1 with a SIP 200 (OK) response.

NOTE 2: The SIP 200 (OK) response in step 14 can be sent earlier by the SCC AS in response to the SIP BYE request received from UE 1(step 3). For example, the SCC AS can send the SIP 200 (OK) response to UE 1 after receiving the SIP 200 (OK) response from UE 3 (step 9).

A.13.3 Remote UE releases collaborative session

In this example, session release is initiated by the remote UE (UE 3). UE 1 and UE 2 are included in a Collaborative Session with the remote UE. The SCC AS ensures that all Controlee UEs involved in the Collaborative Session receive a request for session release in order to completely release the ongoing session.

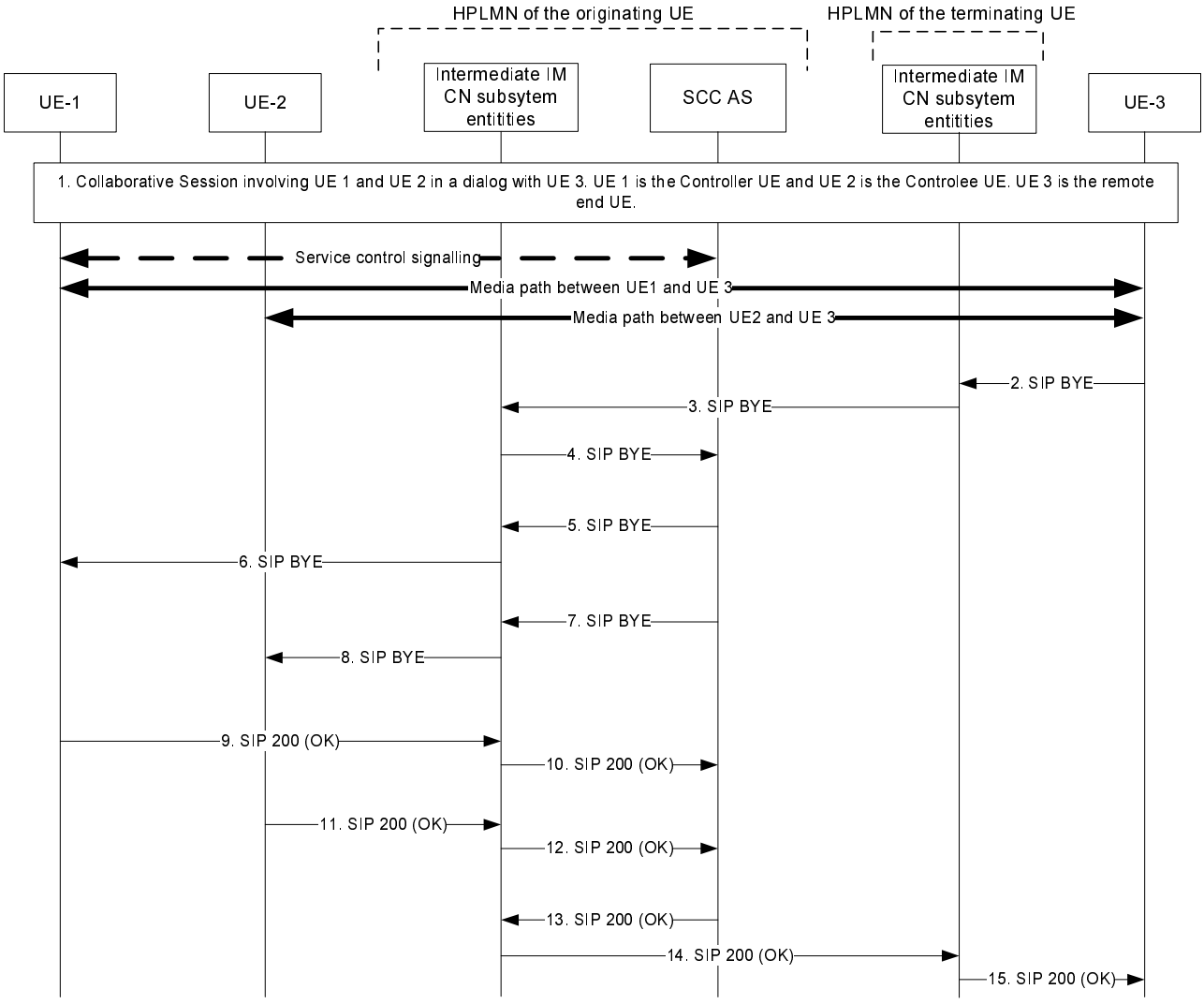


Figure A.13.3-1: Release of Collaborative Session initiated by remote UE

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

1. Collaborative Session currently exists in a dialog with UE 3

A Collaborative Session involving UE 1 and UE 2 exists in a dialog with UE 3. Media paths exist between UE 1 and UE 3 and between UE 2 and UE 3. In this scenario, UE 1 is the Controller UE in the Collaborative Session and thus maintains service control signalling with the SCC AS.

2-4. SIP BYE request (UE 3 to SCC AS via intermediate IM CN subsystem entities)

The remote UE, UE 3, initiates session release by sending a SIP BYE request towards the Controller UE via the SCC AS serving the Controller UE. There is no Inter UE transfer specific content in the SIP BYE request.

5-6. SIP BYE request (SCC AS to UE 1 via intermediate IM CN subsystem entities)

The SCC AS routes the SIP BYE request to UE 1, indicating to UE 1 that the remote UE requests that the session is to be released.

7-8. SIP BYE request (SCC AS to UE 2 via intermediate IM CN subsystem entities)

The SCC AS, acting as a routing B2BUA sends a SIP BYE request to UE 2 to release the dialog it is involved in with UE 3 via the Collaborative Session. There is no Inter UE transfer specific content in the SIP BYE request.

NOTE 1: Step 7 can occur in parallel with step 5. Alternatively, step 7 can occur prior to step 5. The order in which the SCC AS sends the SIP BYE requests is not considered to be important.

9-10. SIP 200 (OK) response (UE 1 to SCC AS via intermediate IM CN subsystems entities)

UE 1 responds to the received SIP BYE request with a SIP 200 (OK) response.

NOTE 2: Step 9 can occur immediately after UE1 has received the SIP BYE request in step 6.

11-12. SIP 200 (OK) response (UE 2 to SCC AS via intermediate IM CN subsystems entities)

UE 2 responds to the received SIP BYE request with a SIP 200 (OK) response.

13-15. SIP 200 (OK) response (SCC AS to UE 3 via intermediate IM CN subsystem entities)

Upon receiving SIP 200 (OK) responses from UE 1 and UE 2, the SCC AS responds to the SIP BYE request from UE 3 with a SIP 200 (OK) response.

NOTE 3: Step 13 can occur after the SCC AS has received the SIP BYE request step 4 since it is not necessary to wait for SIP BYE requests from UE1 and UE2.

A.14 Signalling flows for media adding/deleting within collaborative session for inter-UE transfer

A.14.1 Introduction

This subclause shows signalling flows for adding and releasing of mediacomponent by the controller UE. Four different scenarios are considered in this subclause:

- scenario where the controller UE adds a new media component on a controllee UE. The procedures in this subclause are the same as the procedures described in subclause A.12.3 with the exception that upon the receipt of a SIP REFER request from the controller UE, the SCC AS generates a SIP re-INVITE request within the dialog to the controllee UE instead of a SIP INVITE request;

- scenario where the controller UE releases a media component from the controller UE, see subclause A.14.2.1;

- scenario where the controller UE releases a media component from the controllee UE, see subclause A.14.2.2.1; and

- scenario where the controller UE releases a media component from the controllee UE and that causes the termination of the collaborative session, see subclause A.14.2.2.2.

A.14.2 Controller UE releases media

A.14.2.1 Controller UE releases media flow on controller UE

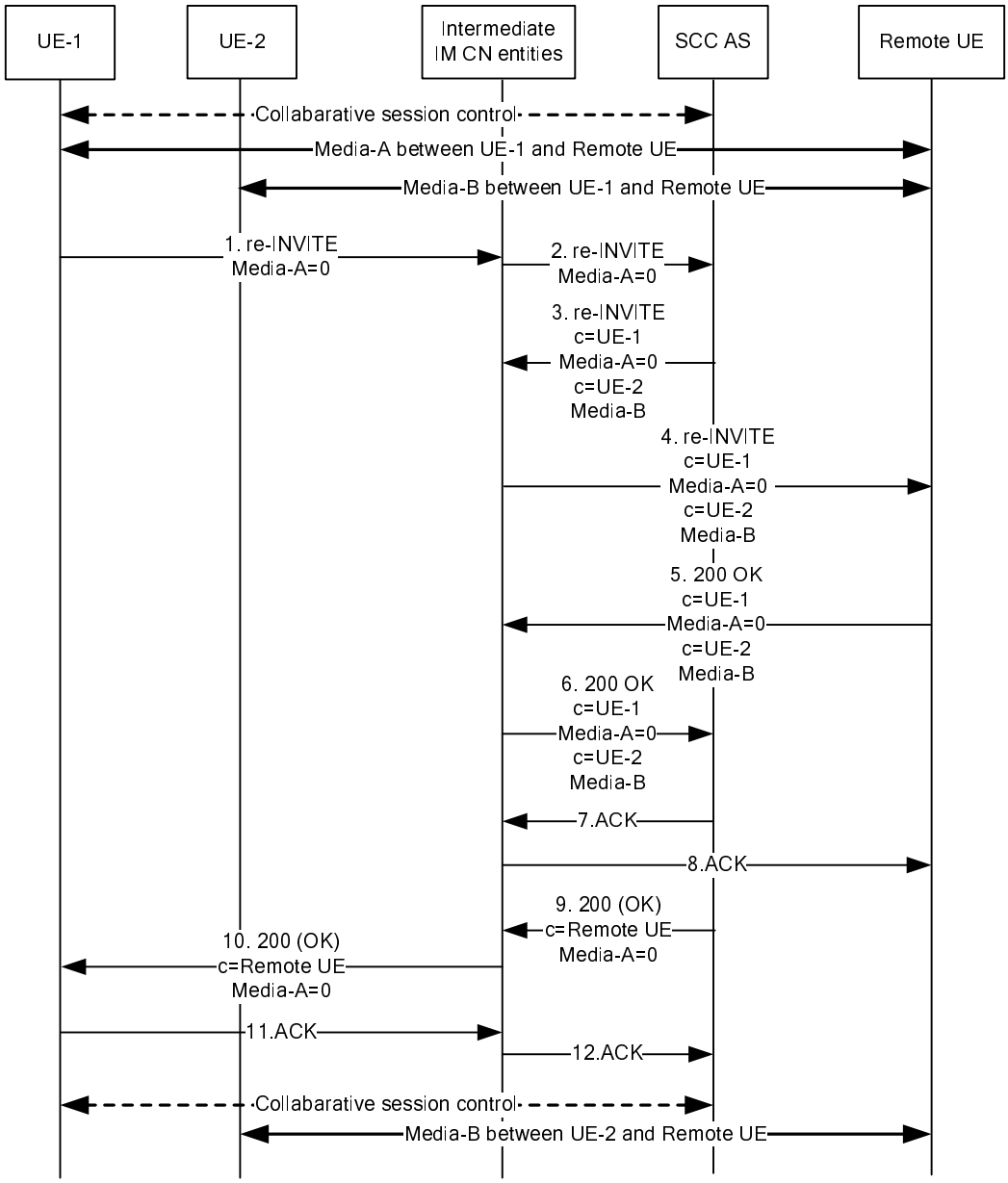


Figure A.14.2.1: Controller UE releases media flow on controller UE

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

It is assumed that UE-1 is the controller UE having collaborative session control. A user has a multimedia session on his device UE-1 with voice (Media A) and UE-2 with video (Media B) media flows. Subsequently, the UE-1 (controller UE) removes the media A flow that is active on the remote UE.

1. **SIP re-INVITE request (UE-1 to intermediate IM CN subsystem entities)- see example in table A.14.2.1-1**

   UE-1 sends a SIP re-INVITE request towards the remote UE indicating Media A is to be removed in the SDP offer.

Table A.14.2.1-1: SIP re-INVITE request (UE-1 to intermediate IM CN subsystem entities)

```
INVITE <sip:userR_public@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91e6bf6> SIP/2.0
Via:SIP/2.0/UDP [3333::eee:fff:aaa:bbb]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Max-Forwards: 70
Route: <sip:pcscf1.visited1.net:7531;lr;comp=sigcomp>, <sip:orig@scscf1.home1.net;lr>
P-Asserted-Identity: "John Doe" <sip:user1_public1@home1.net>
P-Access-Network-Info: 3GPP-UTRAN-TDD; utran-cell-id-3gpp=234151D0FCE11
Privacy: none
From:
To:
Call-ID:
Cseq: 127 INVITE
Require: sec-agree
Proxy-Require: sec-agree
Supported: 100rel, precondition
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_public@home1.net>;gr=urn:uuid:f81d4fae-7dec-11d0-a765-
   00a0c67t6br4>;+g.3gpp.iut-controller
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE, SUBSCRIBE, NOTIFY
Accept: application/sdp, application/3gpp-ims+xml
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 3333::aaa:bbb:ccc:ddd
s=-
t=0 0
m=audio 0 RTP/AVP 97
c=IN IP6 3333::aaa:bbb:ccc:ddd
a=rtpmap:97 PCMU/8000
m=video 0 RTP/AVP 98
c=IN IP6 4444::aaa:bbb:ccc:ddd
a=rtpmap:98 MPV/90000
```

2. **SIP re-INVITE request**

   The intermediate IM CN subsystem entities forward the SIP re-INVITE request to the SCC AS according to standard IMS procedure.

3. **SIP re-INVITE request (SCC AS to intermediate IM CN subsystem entities) - see example in table A.14.2.1-3**

   The SCC AS sends a SIP re-INVITE request with all the media information at the remote UE, sets the port number for media A to 0, and forwards it towards the remote UE through the intermediate IM CN subsystem entities.

Table A.14.2.1-3: SIP re-INVITE request (SCC AS to intermediate IM CN subsystem entities)

```
INVITE <sip:userR_public2@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91e6bf6> IP/2.0
Via: SIP/2.0/UDP sccas1.home1.net;branch=z9hG4bK240f34.12
Max-Forwards: 70
Route: <sip:orig@scscf1.home1.net;lr>
P-Asserted-Identity: "John Doe" <sip:user1_public1@home1.net>
P-Access-Network-Info: 3GPP-UTRAN-TDD; utran-cell-id-3gpp=234151D0FCE11
Privacy: none
From:
To:
Call-ID:
Cseq: 127 INVITE
Require: sec-agree
Proxy-Require: sec-agree
Supported: 100rel, precondition
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:
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE, SUBSCRIBE, NOTIFY
Accept: application/sdp, application/3gpp-ims+xml
Content-Length: (…)
v=0
o=- 2987933615 2987933615 IN IP6 3333::aaa:bbb:ccc:ddd
s=-
t=0 0
m=audio 0 RTP/AVP 97
c=IN IP6 3333::aaa:bbb:ccc:ddd
a=rtpmap:97 MPV/90000
m=video 8888 RTP/AVP 98
c=IN IP6 4444::aaa:bbb:ccc:ddd
a=rtpmap:98 MPV/90000
```

4. SIP re-INVITE request

The intermediate IM CN subsystem entities forward the SIP re-INVITE request to the remote UE according to standard IMS procedure.

5. SIP 200 (OK) response (remote UE to intermediate IM CN subsystem entities) - see example in table A.14.2.1-5

The remote UE sends a SIP 200 (OK) response with an SDP answer.

Table A.14.2.1-5: SIP 200 (OK) response (remote UE to IM CN subsystem entities)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf2.visited2.net;branch=dahtadfz4radgs.12, SIP/2.0/UDP
   scscf2.home1.net;branch=hsdfldf343.12, SIP/2.0/UDP
   scscf1.home1.net;branch=hsdfldf56322cc.13, SIP/2.0/UDP
   sccas1.home1.net;branch=z9hG4bKnas34r2.21
From:
To:
Call-ID:
Cseq: 127 INVITE
Supported: 100rel; precondition
Contact:
Allow:
Accept: application/sdp;
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933300 2987933300 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
a=rtpmap:97 MPV/90000
m=video 6666 RTP/AVP 98
a=rtpmap:98 MPV/90000
```

6. SIP 200 (OK) response

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

The SCC AS sends a SIP ACK request to the remote UE through the intermediate IM CN subsystem entities.

8. SIP ACK request (intermediate IM CN subsystem entities to remote UE)

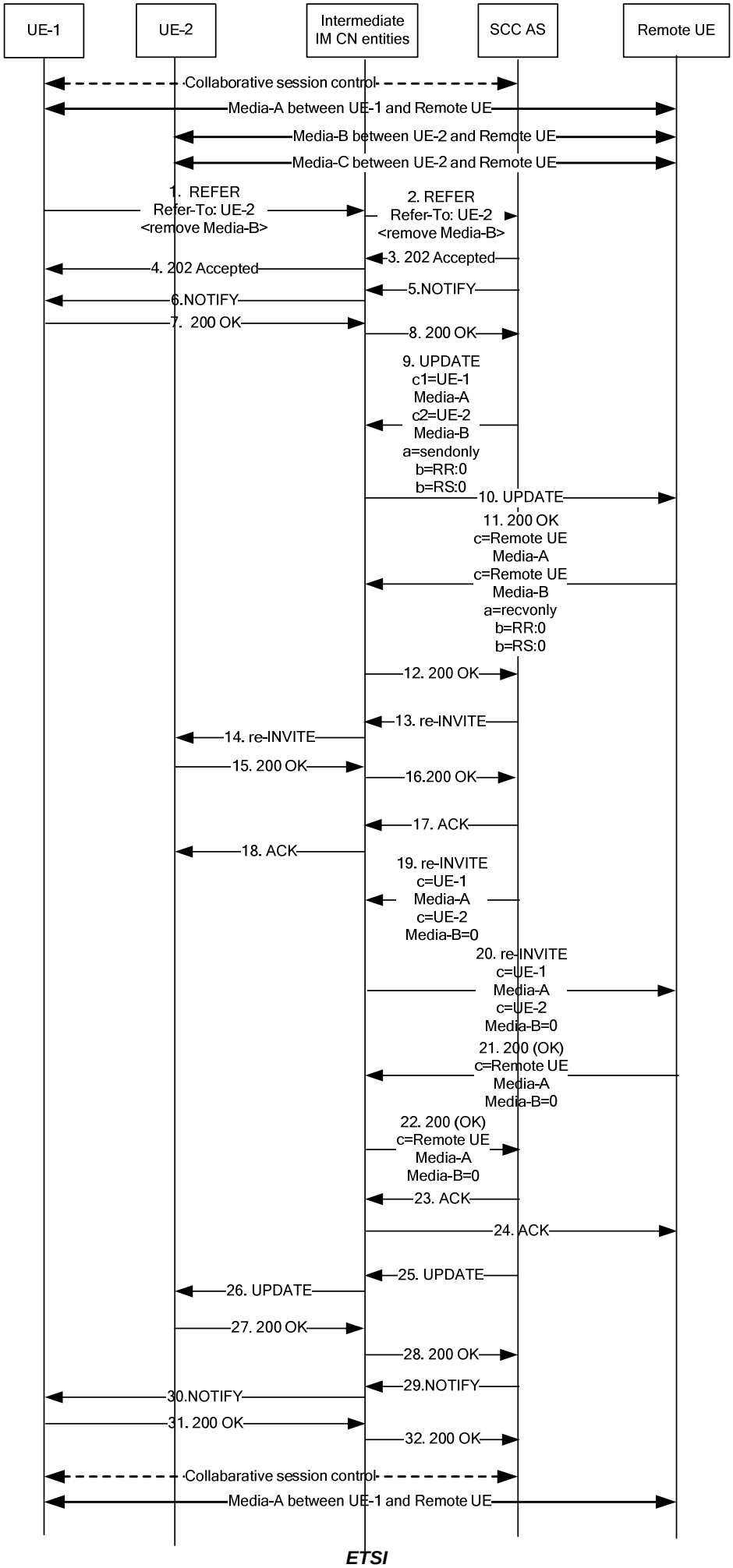
9-10. SIP 200 (OK) response (SCC AS to UE-1 through intermediate IM CN subsystem entities)

The SCC AS sends a SIP 200 (OK) response with an SDP answer indicating that Media-A has been removed

11-12. SIP ACK request (controller UE to SCC AS)

A.14.2.2 Controller UE releases media flow on controllee UE

A.14.2.2.1 Controller UE removes media at the controllee UE


UE-1
UE-2
Intermediate IM CN entities
SCC AS
Remote UE
Collaborative session control
Media-A between UE-1 and Remote UE
Media-B between UE-2 and Remote UE
Media-C between UE-2 and Remote UE
1. REFER
Refer-To: UE-2
<remove Media-B>
2. REFER
Refer-To: UE-2
<remove Media-B>
3. 202 Accepted
4. 202 Accepted
5.NOTIFY
6.NOTIFY
7. 200 OK
8. 200 OK
9. UPDATE
c1=UE-1
Media-A
c2=UE-2
Media-B
a=sendonly
b=RR:0
b=RS:0
10. UPDATE
11. 200 OK
c=Remote UE
Media-A
c=Remote UE
Media-B
a=recvonly
b=RR:0
b=RS:0
12. 200 OK
13. re-INVITE
14. re-INVITE
15. 200 OK
16.200 OK
17. ACK
18. ACK
19. re-INVITE
c=UE-1
Media-A
c=UE-2
Media-B=0
20. re-INVITE
c=UE-1
Media-A
c=UE-2
Media-B=0
21. 200 (OK)
c=Remote UE
Media-A
Media-B=0
22. 200 (OK)
c=Remote UE
Media-A
Media-B=0
23. ACK
24. ACK
25. UPDATE
26. UPDATE
27. 200 OK
28. 200 OK
29.NOTIFY
30.NOTIFY
31. 200 OK
32. 200 OK
Collabarative session control
Media-A between UE-1 and Remote UE

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

Figure A.14.2.2.1:Controller UE remove media at the controllee UE

1-2. SIP REFER request (controller UE to intermediate IM CN subsystem entities) – see example in table A.14.2.1-1

It is assumed that UE-1 is the controller UE having collaborative session control. A user has a multimedia session on his device UE-1 with a voice (Media A) and UE-2(Controllee UE) with a video (Media B) media flows and a voice (Media C) media flow. The controller UE wants to remove the video media (Media B) component on the controllee UE.

Table A.14.2.2.1-1 SIP REFER request (UE-1 to SCC AS)

```
REFER sip:interUEtransfer@sccas1.home1.net SIP/2.0
Via:
To: sip:interUEtransfer@sccas1.home1.net
From: sip:user1_public1@home1.net;tag =13579
Call-ID: cb03a0s09a2sdfglkj490333
CSeq: 93809824 REFER
Max-Forwards: 70
P-Preferred-Identity: "John Doe" <sip:user1_public1@home1.net>
Refer-To: <sip:user2_public2@home2.net;gr=urn:uuid:f81d4fae-7dec-11d0-a762-00a0c91e6bf6?
   body=m%3Daudio%200%20RTP%2FAVP%2097%0Dm%3Daudio%204568%20RTP%2FAVP%2097%0Dm%3Dvideo%200%20R
   TP%2FAVP%2098>
Require: target-dialog
Target-dialog: cb03a0s09a2sdfglkj13579;to-tag=abcdef;from-tag=123456
Contact: <sip:user1_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-
   00a0c91ewxyz>;+g.3gpp.iut-controller
Allow:
Accept: message/sipfrag
Content-Length: 0
```

3-4. SIP 202 (Accepted) response

The SCC-AS sends a SIP 202 (Accepted) response to the controller UE-1 as response to the SIP REFER request.

5-6. SIP NOTIFY request (SCC AS to UE-1 through intermediate IM CN subsystem entities)-see example in table A.14.2.1-5

The SCC-AS sends a SIP NOTIFY request to UE-1 to notify implicit subscription to the SIP REFER request results.

Table A.14.2.2.1-5 SIP NOTIFY request (SCC AS to UE-1)

```
NOTIFY sip:user1_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91ewxyz SIP/2.0
Via:
To: sip:user1_public1@home1.net;tag=13579
From: sip:interUEtransfer@sccas1.home1.net;tag=2255
Call-ID:
CSeq:
Max-Forwards:
P-Asserted-Identity:
Require:
Contact: sip: scc-as@home1.net
Allow:
Event: refer
Subscription-State: active;expires=3600
Content-Type: message/sipfrag ;version=2.0
Content-Length: (…)

SIP/2.0 100 Trying
```

7-8. SIP 200 (OK) response (UE-1 to SCC-AS through intermediate IM CN subsystem entities)

The controller UE, UE-1, acknowledges the SIP NOTIFY request by sending a SIP 200 (OK) response to the SCC AS.

9-10. SIP UPDATE request (SCC AS to remote UE through intermediate IM CN subsystem entities) - see example in table A.14.2.2.2-9.

The SCC AS prepares the removal of the media component received by the controllee UE-2 and sent by the remote UE.

Table A.14.2.2.2-9: SIP UPDATE request (SCC AS to remote UE through intermediate IM CN subsystem entities)

```
UPDATE sip:userR_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a762-00a0c91e6bf6 SIP/2.0
Via:
To:
From:
Call-ID:
Cseq:
Contact: <sip: user1_public@home1.net>; gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c67t6br4>
Allow:
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 3333::aaa:bbb:ccc:ddd
s=-
t=0 0
m=audio 6666 RTP/AVP 97
c=IN IP6 3333::aaa:bbb:ccc:ddd
a=rtpmap:97 PCMU/8000
m=video 6668 RTP/AVP 98
a=sendonly
b=RR:0
b=RS:0
c=145.23.77.88
b=AS:75
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
```

11-12. SIP 200 (OK) response (remote UE to intermediate IM CN subsystem entities) - see example in table A.14.2.2.2-11.

Remote UE send a SIP 200 (OK) response with SDP answer.

Table A.14.2.2.2-11 SIP 200 (OK) response (remote UE to intermediate IM CN subsystem entities)

```
SIP/2.0 200 OK
Via:
To:
From:
Call-ID:
Cseq:
Contact: <sip:userR_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a762-00a0c91e6bf6>
Allow:
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933300 2987933300 IN IP6 5555::aaa:bbb::ccc:ddd
s=-
t=0 0
m=audio 4444 RTP/AVP 97
c=IN IP6 5555::aaa:bbb::ccc:ddd
a=rtpmap:9777 PCMU/8000
m=video 6666 RTP/AVP 98
a=recvonly
b=RR:0
b=RS:0
c=145.23.77.88
b=AS:75
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
```

13. SIP re-INVITE request (SCC AS to intermediate IM CN subsystem entities) - see example in table A.14.2.2.1-13

The SCC AS sends a SIP re-INVITE request towards the Controllee UE (UE-2).

Table A.14.2.2.1-13 SIP re-INVITE request (SCC-AS to IM CN subsystem entities)

```
INVITE sip:user2_public1@home1.net;gr=urn:uuid:f 81d4fae-7dec-11d0-a765-00a0c91e6bf6 SIP/2.0
Via:
Route:
To: sip:user2_public1@home1.net;abcdef
From: sip:user3_public3@home3.net;tag=123456
Call-ID:
CSeq:
Max-Forwards:
Require:
Referred-By: sip:user1_public1@home1.net
Contact: sip:user3_public3@home3.net;gr=urn:uuid:f81d4fae-17oct-11a1-a678-0054c91eabcd
Allow:
Content-Type: application/sdp
Content-Length:(…)

v=0
o=- 1027933615 1027933615 IN IP4 123.112.67.87
s=-
t=0 0
m=audio 0 RTP/AVP 97
m=audio 4568 RTP/AVP 97
c=123.112.67.87
b=AS:75
a=rtpmap:97 PCMU/8000
m=video 0 RTP/AVP 98
a=rtpmap:98 MPV/90000
a=sendonly
```

14. SIP re-INVITE request (intermediate IM CN subsystem entities to UE-2)

15-16. SIP 200 (OK) response (UE-2 to SCC AS through intermediate IM CN subsystem entities)

17-18. SIP ACK request (from SCC-AS to UE-2)

19. SIP re-INVITE request (SCC AS to intermediate IM CN subsystem entities) - see example in table A.14.2.2-19

The SCC AS sends a SIP re-INVITE request with all the media information at the remote UE, set the port number for media B to 0, and forwards it to the remote UE through the intermediate IM CN subsystem entities.

Table A.14.2.2.1-19: SIP re-INVITE request (SCC AS to intermediate IM CN subsystem entities)

```
INVITE <sip:userR_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a762-00a0c91e6bf6> SIP/2.0
Via: SIP/2.0/UDP sccas1.home1.net;branch=z9hG4bKnas34r2.12
Max-Forwards: 70
Route: <sip:orig@scscf1.home1.net;lr>
P-Asserted-Identity: "John Doe" <sip:user1_public1@home1.net>
Privacy: none
From:
To:
Call-ID:
Cseq: 127 INVITE
Supported: 100rel, precondition
Contact: <sip: user1_public@home1.net>; gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c67t6br4>
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE, SUBSCRIBE, NOTIFY
Accept: application/sdp
Content-Length:(..)

v=0
o=- 2987933615 2987933615 IN IP6 3333::aaa:bbb:ccc:ddd
s=-
c=IN IP6 3333::aaa:bbb:ccc:ddd
t=0 0
m=audio 6666 RTP/AVP 97
a=rtpmap:97 PCMU/8000
m=audio 4567 RTP/AVP 97
a=rtpmap:97 PCMU/8000
m=video 0 RTP/AVP 98
a=rtpmap:98 MPV/90000
```

20. SIP re-INVITE request (intermediate IM CN subsystem entities to remote UE)

21. SIP 200 (OK) response (remote UE to intermediate IM CN subsystem entities) - see example in table A.14.2.1-21

The remote UE sends a SIP 200 (OK) response with an SDP answer.

Table A.14.2.2.1-21 SIP 200 (OK) (remote UE to intermediate IM CN subsystem entities)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf2.visited2.net;branch=343asdfaredfatz.12, SIP/2.0/UDP
   scscf2.home1.net;branch=fsc35avhthaz4.22, SIP/2.0/UDP
   scscf1.home1.net;branch=fsc35avhthaz4.12, SIP/2.0/UDP
   sccas1.home1.net;branch=z9hG4bKnas34r2.14
From:
To:
Call-ID:
Cseq: 127 INVITE
Supported: 100rel; precondition
Contact:
Allow:
Accept: application/sdp;
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933300 2987933300 IN IP6 5555::aaa:bbb::ccc:ddd
s=-
c=IN IP6 5555::aaa:bbb::ccc:ddd
t=0 0
m=audio 4444 RTP/AVP 97
a=rtpmap:9777 PCMU/8000
m=audio 4545 RTP/AVP 97
a=rtpmap:97 PCMU/8000
m=video 0 RTP/AVP 98
a=rtpmap:98 MPV/90000
```

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

23-24. SIP ACK request (SCC AS to remote UE)

25-26. SIP UPDATE request (SCC AS to UE-2 through intermediate IM CN subsystem entities)

SCC AS sends a SIP UPDATE request towards Controllee UE (UE-2).

Table A.14.2.2.1-25 SIP UPDATE request (SCC-AS to UE-2)

```
UPDATE sip:user2_public1@home1.net;gr=urn:uuid:f 81d4fae-7dec-11d0-a765-00a0c91e6bf6 SIP/2.0
Via:
Route:
To: sip:user2_public1@home1.net;abcdef
From: sip:user3_public3@home3.net;tag=123456
Call-ID:
CSeq:
Max-Forwards:
Contact: sip:user3_public3@home3.net;gr=urn:uuid:f81d4fae-17oct-11a1-a678-0054c91eabcd
Allow:
Content-Type: application/sdp
Content-Length:(…)

v=0
o=- 1027933615 1027933615 IN IP4 123.112.67.87
s=-
t=0 0
m=audio 0 RTP/AVP 97
m=audio 4568 RTP/AVP 97
c=123.112.67.87
b=AS:75
a=rtpmap:97 PCMU/8000
m=video 0 RTP/AVP 98
```

27-28. SIP 200 (OK) response (UE-2 to SCC AS through intermediate IM CN subsystem entities)

29-30. SIP NOTIFY request (SCC-AS to UE-1)-see example table A.14.2.2.1-21

The SCC AS sends a SIP NOTIFY request to the controller UE, UE-1 to inform about the success status of the inter-UE transfer.

Table A.14.2-2.1-29 SIP NOTIFY request (SCC AS to UE-1)

```
NOTIFY sip:user1_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91ewxyz SIP/2.0
Via:
To: sip:user1_public1@home1.net;tag = 13579
From: sip:interUEtransfer@sccas1.home1.net;tag=2255
Call-ID:
CSeq:
Max-Forwards:
P-Asserted-Identity:
Require:
Contact: sip: scc-as@home1.net
Allow:
Event: refer
Subscription-State:terminated; reason=noresource
Content-Type: message/sipfrag ;version=2.0
Content-Length: (…)

SIP/2.0 200 OK
Content-Type: application/sdp

v=0
s=-
m=audio 3434 RTP/AVP 97
a=rtpmap:97 PCMU/8000
m=video 0 RTP/AVP 98
a=rtpmap:98 MPV/90000
a=sendonly
```

31-32. SIP 200 OK response (UE-1 to SCC AS)

The controller UE,UE-1 acknowledges the SIP NOTIFY request by sending a SIP 200 OK response to the SCC AS.

A.14.2.2.2 Controller UE remove the controllee UE from the collaborative session

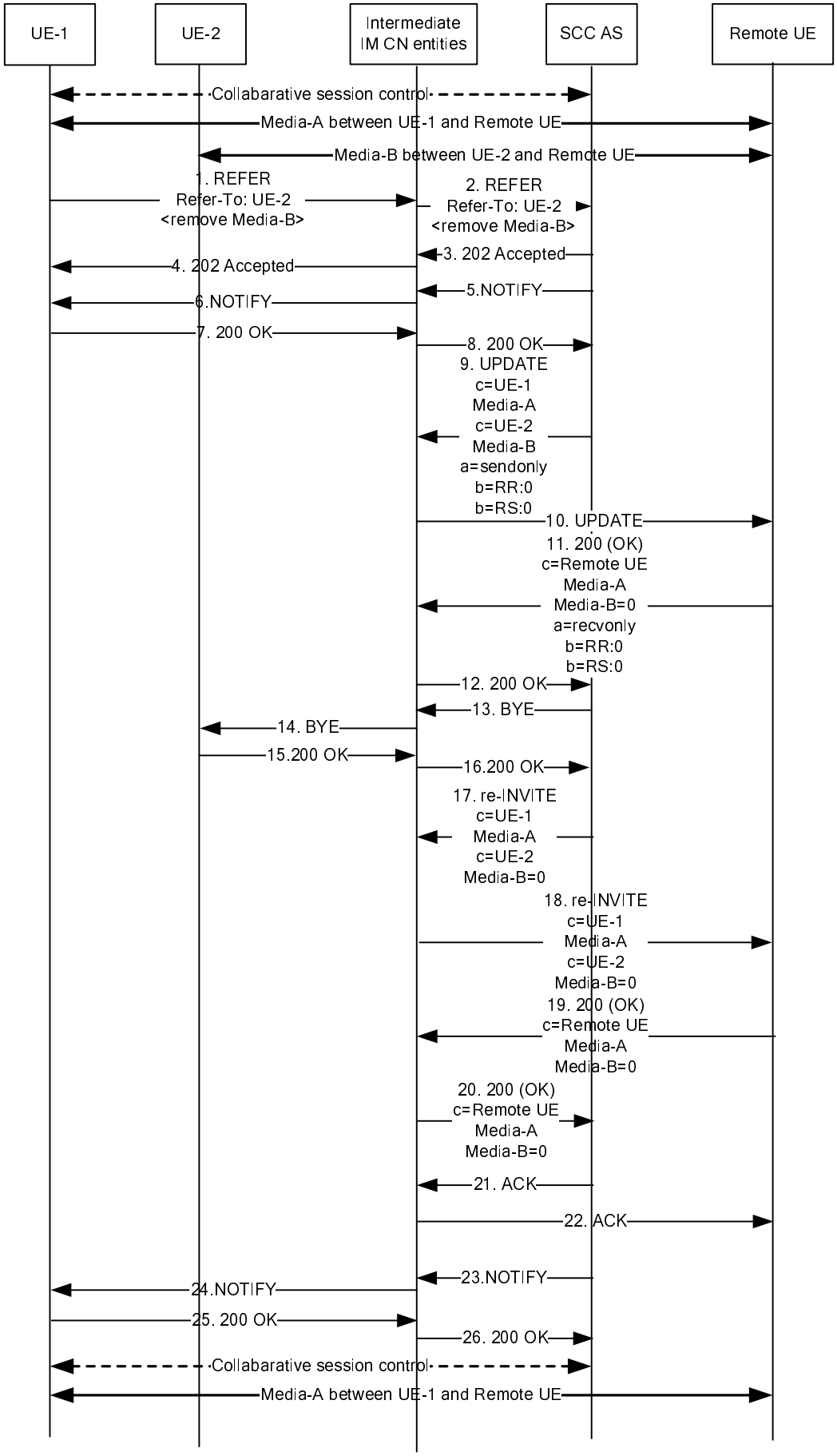


Figure A.14.2.2.2:Controller UE remove the controllee UE from the collaborative session

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

1-2. SIP REFER request (controller UE to intermediate IM CN subsystem entities) – see example in table A.14.2.2.2-1

It is assumed that UE-1 is the controller UE having collaborative session control. A user has a multimedia session on his device UE-1 with a voice (Media A) and UE-2(Controllee UE) with a video (Media B) media flow. The controller UE wants to remove the controllee UE from the collaborative session.

Table A.14.2.2.2-1 SIP REFER request (UE-1 to SCC-AS)

```
REFER sip:interUEtransfer@sccas1.home1.net SIP/2.0
Via:
To: sip:interUEtransfer@sccas1.home1.net
From: sip:user1_public1@home1.net;tag = 13579
Call-ID: cb03a0s09a2sdfglkj490333
CSeq: 93809824 REFER
Max-Forwards: 70
P-Preferred-Identity:
Refer-To: <sip:user2_public2@home2.net;gr=urn:uuid:f81d4fae-7dec-11d0-a762-
   00a0c91e6bf6;method=BYE>
Require: target-dialog
Target-dialog: cb03a0s09a2sdfglkj13579; to-tag=abcdef; from-tag=123456
Contact: sip:user1_public1@home1.net ;+g.3gpp.iut-controller
Allow:
Accept:
Content-Type:
Content-Length:
```

3-4. SIP 202 (Accepted) response

The SCC AS sends a SIP 202 (Accepted) response to the controller UE-1 as response to the SIP REFER request.

5-6. SIP NOTIFY request (SCC AS to UE-1 through intermediate IM CN subsystem entities) – see example in table A.14.2.2.2-5

The SCC AS sends a SIP NOTIFY request to UE-1 to notify implicit subscription to the SIP REFER request results.

Table A.14.2.2.2-5 SIP NOTIFY request (SCC AS to UE-1)

```
NOTIFY
Via:
To: sip:user1_public1@home1.net;tag=13579
From: sip:interUEtransfer@sccas1.home1.net;tag=2255
Call-ID:
CSeq:
Max-Forwards:
P-Asserted-Identity:
Require:
Contact: sip: scc-as@home1.net
Allow:
Event: refer
Subscription-State: active;expires=3600
Content-Type: message/sipfrag ;version=2.0
Content-Length: (…)

SIP/2.0 100 Trying
```

7-8. SIP 200 (OK) response (UE-1 to SCC AS through intermediate IM CN subsystem entities)

The controller UE, UE-1, acknowledges the SIP NOTIFY request by sending a SIP 200 (OK) to the SCC AS.

9-10. SIP UPDATE request (SCC AS to remote UE through intermediate IM CN subsystem entities) - see example in table A.14.2.2.2-9.

The SCC AS prepares the removal of the controllee UE-2 from the collaborative session by stopping the media received by the controllee UE-2 and sent by the remote UE.

Table A.14.2.2.2-9: SIP UPDATE request (SCC AS to Remote UE through intermediate IM CN subsystem entities)

```
UPDATE sip:userR_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a762-00a0c91e6bf6 SIP/2.0
Via:
To:
From:
Call-ID:
Cseq:
Contact: <sip: user1_public@home1.net>; gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c67t6br4>
Allow:
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 3333::aaa:bbb:ccc:ddd
s=-
t=0 0
m=audio 6666 RTP/AVP 97
c=IN IP6 3333::aaa:bbb:ccc:ddd
a=rtpmap:97 PCMU/8000
m=video 6668 RTP/AVP 98
a=sendonly
b=RR:0
b=RS:0
c=145.23.77.88
b=AS:75
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
```

11-12. SIP 200 (OK) response (Remote UE to intermediate IM CN subsystem entities) - see example in table A.14.2.2.2-11.

Remote UE sends a SIP 200 (OK) response with SDP answer.

Table A.14.2.2.2-11 SIP 200 (OK) response (Remote UE to intermediate IM CN subsystem entities)

```
SIP/2.0 200 OK
Via:
To:
From:
Call-ID:
Cseq:
Contact: <sip:userR_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a762-00a0c91e6bf6>
Allow:
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933300 2987933300 IN IP6 5555::aaa:bbb::ccc:ddd
s=-
t=0 0
m=audio 4444 RTP/AVP 97
c=IN IP6 5555::aaa:bbb::ccc:ddd
a=rtpmap:9777 PCMU/8000
m=video 6666 RTP/AVP 98
a=recvonly
b=RR:0
b=RS:0
c=145.23.77.88
b=AS:75
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
```

13-14. SIP BYE request (SCC AS to UE-2 through intermediate IM CN subsystem entities)

The SCC AS sends a SIP BYE request towards the controllee UE (UE-2).

15-16. SIP 200 (OK) response (UE-2 to SCC AS through intermediate IM CN subsystem entities)

17. SIP re-INVITE request (SCC AS to intermediate IM CN subsystem entities) - see example in table A.14.2.2.2-17

The SCC AS sends a SIP re-INVITE request with all the media information at the remote UE, set the port number for media B to 0, and forwards it to the remote UE through the intermediate IM CN subsystem entities.

Table A.14.2.2.2-17: SIP re-INVITE request (SCC AS to intermediate IM CN subsystem entities)

```
INVITE <sip:userR_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a762-00a0c91e6bf6> SIP/2.0
Via: SIP/2.0/UDP sccas1.home1.net;branch=z9hG4bKnas34r2.12
Max-Forwards: 70
Route: <sip:orig@scscf1.home1.net;lr>
P-Asserted-Identity: "John Doe" <sip:user1_public1@home1.net>
P-Access-Network-Info: 3GPP-UTRAN-TDD; utran-cell-id-3gpp=234151D0FCE11
Privacy: none
From:
To:
Call-ID:
Cseq: 127 INVITE
Require: sec-agree
Proxy-Require: sec-agree
Supported: 100rel, precondition
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_public@home1.net>; gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c67t6br4>
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE, SUBSCRIBE, NOTIFY
Accept: application/sdp, application/3gpp-ims+xml
Content-Length:0

v=0
o=- 2987933615 2987933615 IN IP6 3333::aaa:bbb:ccc:ddd
s=-
c=IN IP6 3333::aaa:bbb:ccc:ddd
t=0 0
m=audio 6666 RTP/AVP 97
a=rtpmap:97 PCMU/8000
m=video 0 RTP/AVP 98
a=rtpmap:98 MPV/90000
```

18. SIP re-INVITE request (intermediate IM CN subsystem entities to remote UE)

19. SIP 200 (OK) response (remote UE to intermediate IM CN subsystem entities) - see example in table A.14.2.2.2-19

The remote UE sends a SIP 200 (OK) response with SDP answer.

Table A.14.2.2.2-19 SIP 200 (OK) response (remote UE to intermediate IM CN subsystem entities)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf2.visited2.net;branch=343asdfaredfatz.12, SIP/2.0/UDP
   scscf2.home1.net;branch=fsc35avhthaz4.22, SIP/2.0/UDP
   scscf1.home1.net;branch=fsc35avhthaz4.12, SIP/2.0/UDP
   sccas1.home1.net;branch=z9hG4bKnas34r2.14
From:
To:
Call-ID:
Cseq: 127 INVITE
Supported: 100rel; precondition
Contact:
Allow:
Accept: application/sdp;
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933300 2987933300 IN IP6 5555::aaa:bbb::ccc:ddd
s=-
c=IN IP6 5555::aaa:bbb::ccc:ddd
t=0 0
m=audio 4444 RTP/AVP 97
a=rtpmap:9777 PCMU/8000
m=video 0 RTP/AVP 98
a=rtpmap:98 MPV/90000
```

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

21-22. SIP ACK request (SCC AS to remote party)

23-24. SIP NOTIFY request (SCC-AS to UE-1)-see example in table A.14.2.2.2-25

The SCC-AS sends a SIP NOTIFY request to the controller UE, UE-1 to inform about the success status of the inter-UE transfer.

Table A.14.2.2.2-23 SIP NOTIFY request (SCC-AS to UE-1)

```
NOTIFY
Via:
To: sip:user1_public1@home1.net;tag = 13579
From: sip:interUEtransfer@sccas1.home1.net;tag=2255
Call-ID:
CSeq:
Max-Forwards:
P-Asserted-Identity:
Require:
Contact: sip: scc-as@home1.net
Allow:
Event: refer
Subscription-State:terminated; reason=noresource
Content-Type: message/sipfrag ;version=2.0
Content-Length: (…)

SIP/2.0 200 OK
```

25-26. SIP 200 OK response (UE-1 to SCC-AS)

The controller UE,UE-1 acknowledges the SIP NOTIFY request by sending a SIP 200 (OK) response to the SCC AS.

A.14.4 Controllee UE releases media

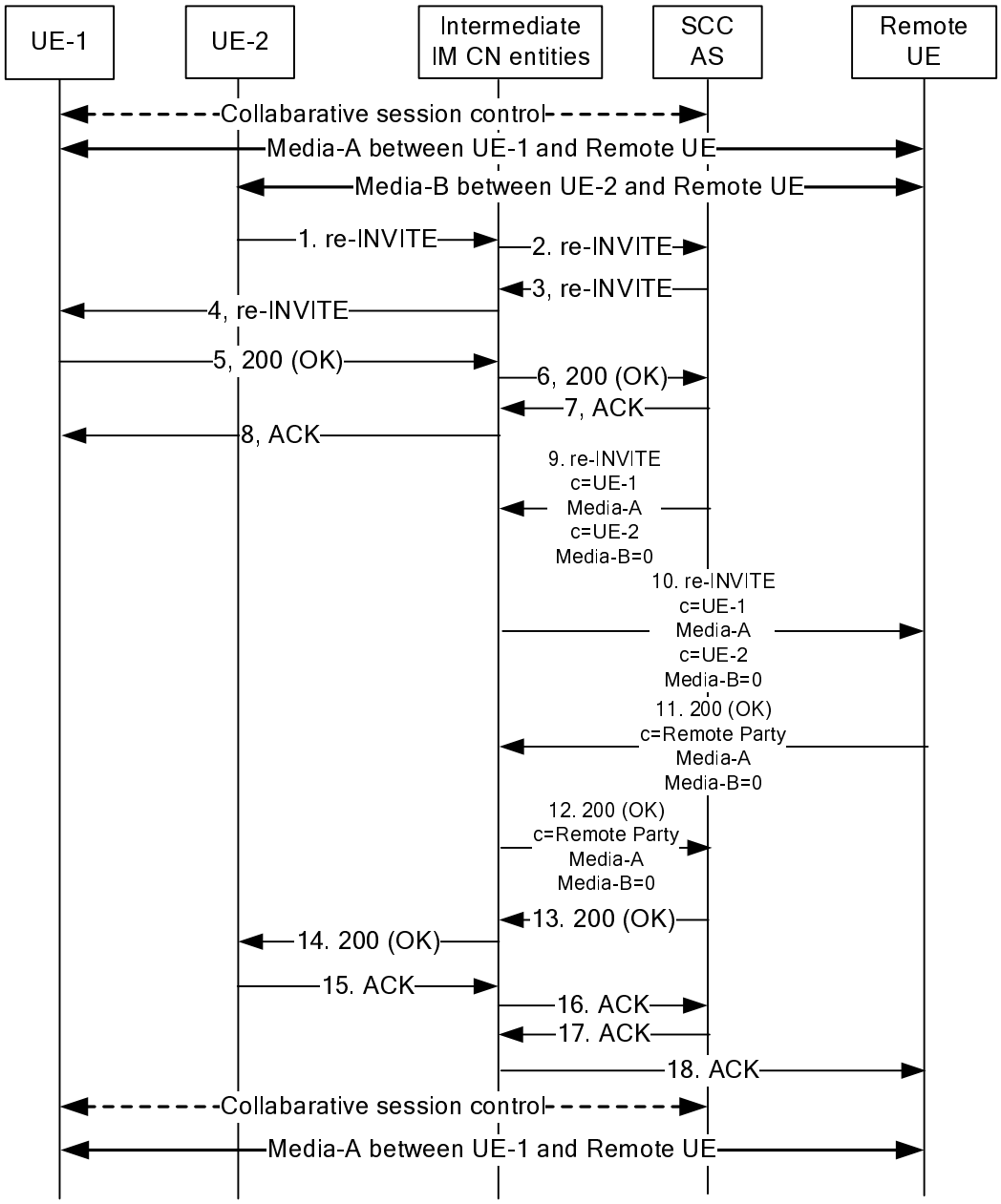


Figure A.14.4: Controllee UE releases media

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

It is assumed that UE-1 is controller UE having collaborative session control. a user has a multimedia session on his device UE-1 with voice (Media A) and UE-2(Controllee UE) video (Media B) media flows. Subsequently, the UE-2 (Controllee UE) removes the media B flow that is active on the remote UE.

1-2. SIP re-INVITE request (UE-2 to SCC AS through IM CN subsystem entities)

A UE-2 wants to release media B active on the remote UE. For this purpose the UE-2 sends a SIP re-INVITE request to the SCC AS through the IM CN subsystem entities.

3. SIP re-INVITE request (from SCC-AS to intermediate IM CN subsystem entities) - see example in table A.14.4-3

The SCC AS sends SIP re-INVITE request to controller UE, UE-1 to inform that the controllee UE wants to release one media, and SCC AS would like to add this media back to the controller UE.

Table A.14.4-3: SIP re-INVITE request (SCC AS to controller UE)

```
INVITE <sip:user1_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a762-00a0c91e6bf6> SIP/2.0
Via: SIP/2.0/UDP sccas1.home1.net;branch=z9hG4bKnas34r2.12
Max-Forwards: 70
Route: <sip:orig@scscf1.home1.net;lr>
P-Asserted-Identity: "John Doe" <sip:user3_public3@home3.net>
Privacy: none
From:
To:
Call-ID:
Cseq: 127 INVITE
Require:
Contact: <sip: user3_public3@home3.net>; gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c67t6br4>
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE, SUBSCRIBE, NOTIFY
Content-type: application/sdp
Content-Length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 3333::aaa:bbb:ccc:ddd
s=-
t=0 0
m=audio 5555 RTP/AVP 97
c=IN IP6 3333::aaa:bbb:ccc:ddd
a=rtpmap:97 PCMU/90000
m=video 3000 RTP/AVP 98
c=IN IP6 4444::aaa:bbb:ccc:ddd
a=rtpmap:98 MPV/90000
```

4. **SIP re-INVITE request (intermediate IM CN subsystem entities to controller UE, UE-1)**

5. **SIP 200 (OK) response (controller UE, UE-1 to intermediate IM CN subsystem entities) - see example in table A.14.4-5**

   In this case, the controller UE does not want to add this media on itself, but like to delete this media within the collaborative session. The controller UE cknowledges the SIP re-INVITE request by sending SIP 200 (OK) response to the SCC AS with the port number set to zero for this media.

Table A.14.4-5: SIP 200 (OK) (controller UE to SCC AS)

```
SIP/2.0 200 OK
Via:
From:
To:
Call-ID:
Cseq:
Contact: <sip: user1_public1@home1.net>; gr=urn:uuid:f81d4fae-7dec-11d0-a765-
   00a0c43t6br4>;+g.3gpp.iut-controller
Allow:
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933300 2987933300 IN IP6 5555::aaa:bbb::ccc:ddd
s=-
c=IN IP6 5555::aaa:bbb::ccc:ddd
t=0 0
m=audio 4444 RTP/AVP 97
a=rtpmap:97 PCMU/8000
m=video 0 RTP/AVP 98
a=rtpmap:98 MPV/9000
```

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

7-8.SIP ACK (SCC AS to controller UE)

9. **SIP re-INVITE request (SCC AS to intermediate IM CN subsystem entities)-see example in table A.14.4-9**

   The SCC AS sends a SIP re-INVITE request to update the remote leg that the media B is released.

Table A.14.4-9: SIP re-INVITE request (SCC AS to intermediate IM CN subsystem entities)

```
INVITE <sip:user3_public3@home3.net;gr=urn:uuid:f81d4fae-7dec-11d0-a762-00a0c91e6bf6> SIP/2.0
Via: SIP/2.0/UDP sccas1.home1.net;branch=z9hG4bKnas34r2.12
Max-Forwards:
Route: <sip:orig@scscf1.home1.net;lr>
P-Asserted-Identity: "Jake" <sip:user1_public1@home1.net>
Privacy: none
From: <sip:user1_public1@home1.net>;tag=171828
To: <sip:user3_public3@home3.net>;tag = 66666
Call-ID: cb03a0s09a2sdfglkj490333
Cseq: 127 INVITE
Supported: 100rel, precondition
Contact: <sip: user1_public1@home1.net>; gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c43t6br4>
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE, SUBSCRIBE, NOTIFY
Accept: application/sdp
ontent-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 3333::aaa:bbb:ccc:ddd
s=-
t=0 0
m=audio 5555 RTP/AVP 97
c=IN IP6 3333::aaa:bbb:ccc:ddd
a=rtpmap:97 PCMU/90000
m=video 0 RTP/AVP 98
c=IN IP6 4444::aaa:bbb:ccc:ddd
a=rtpmap:98 MPV/90000
```

10. SIP re-INVITE request (intermediate IM CN subsystem entities to remote UE)

11. SIP 200 (OK) response (remote UE to intermediate IM CN subsystem entities) - see example in table A.14.4-11

The remote US sends a SIP 200 (OK) with an SDP offer containing Media A and Media B information.

Table A.14.4-11: SIP 200 (OK) (remote UE to intermediate IM CN subsystem entities)

```
SIP/2.0 200 OK
Via:
To:
Call-ID:
Cseq: 127 INVITE
Supported: 100rel; precondition
Contact: <sip: user3_public3@home3.net>; gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c67t6br4>
Allow:
Accept: application/sdp;
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933300 2987933300 IN IP6 5555::aaa:bbb::ccc:ddd
s=-
c=IN IP6 5555::aaa:bbb::ccc:ddd
t=0 0
m=audio 4444 RTP/AVP 97
a=rtpmap:97 PCMU/8000
m=video 0 RTP/AVP 98
a=rtpmap:98 MPV/9000
```

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

13-14. SIP 200 (OK) response (SCC AS to UE-2 through IM CN subsystem entities)

The SCC AS sends a 200 (OK) response.

15-16. SIP ACK request (controllee UE to SCC AS)

17-18. SIP ACK request (SCC AS to remote UE)

A.14.5 Controllee UE modifies media on itself

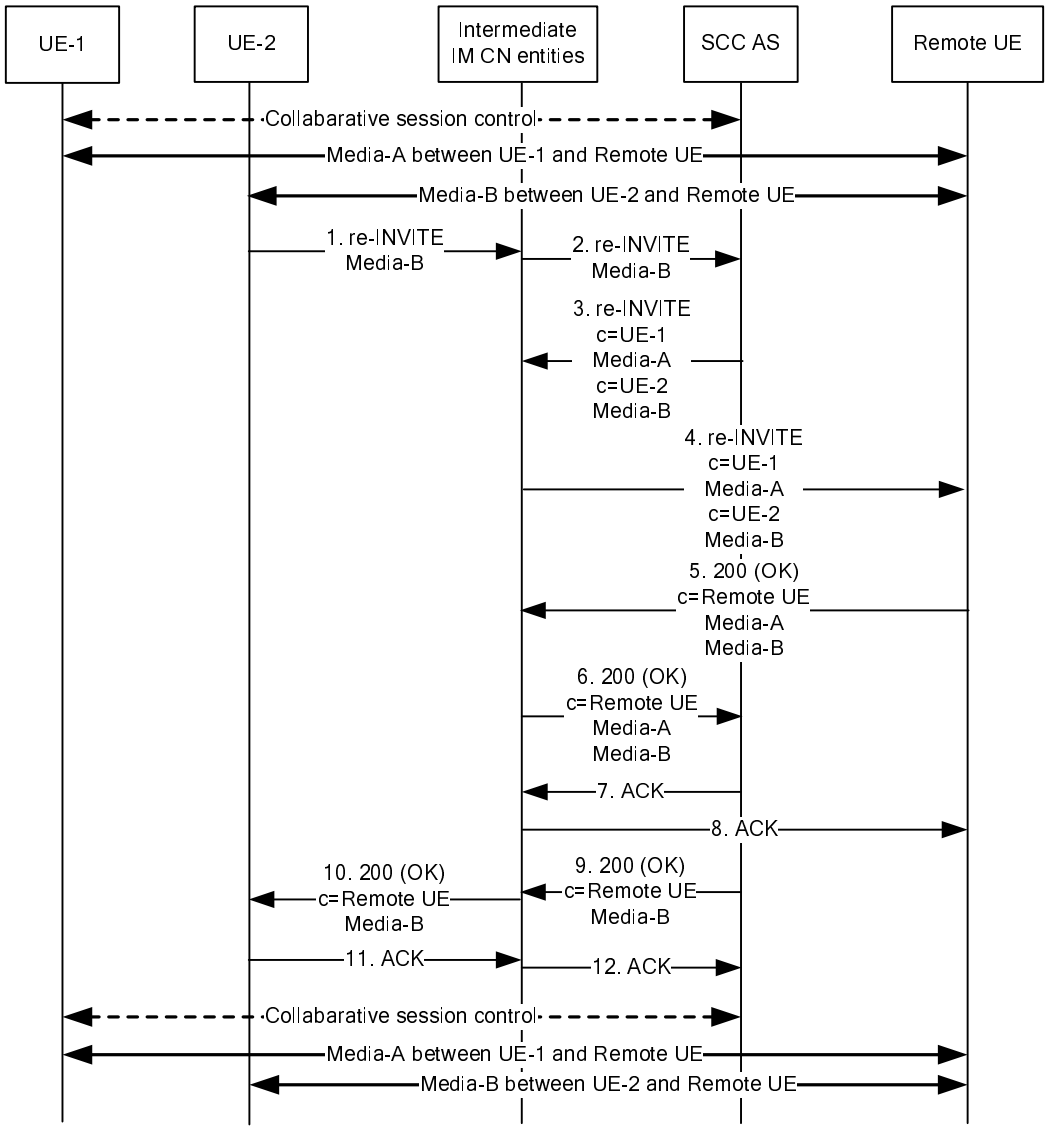


Figure A.14.5: Controllee UE modifies media on itself

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

It is assumed that UE-1 is the controller UE having collaborative session control. A user has a multimedia session on his device UE-1 with voice (Media A) and UE-2 with video (Media B) media flows. Subsequently, the UE-2 (Controllee UE) modifies the media B flow that is active on the remote UE.

1. **SIP re-INVITE request (UE-2 to intermediate IM CN subsystem entities)- see example in table A.14.5-1**

   UE-2 sends a SIP re-INVITE request towards the remote UE containing Media B using SDP offer.

Table A.14.5-1: SIP re-INVITE request (UE-2 to intermediate IM CN subsystem entities)

```
INVITE <sip:userR_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91e6bf6> SIP/2.0
Via:SIP/2.0/UDP [4444::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Max-Forwards: 70
Route: <sip:pcscf1.visited1.net:7531;lr;comp=sigcomp>, <sip:orig@scscf1.home1.net;lr>
P-Asserted-Identity: "John Doe2" <sip:user1_public2@home1.net>
P-Access-Network-Info: 3GPP-UTRAN-TDD; utran-cell-id-3gpp=234151D0FCE11
Privacy: none
From:
To: Call-ID:
Cseq: 127 INVITE
Require: sec-agree
Proxy-Require: sec-agree
Supported: 100rel, precondition
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_public@home1.net>; gr=urn:uuid:f81d4fae-7dec-11d0-a765-
   00a0c67t6br4>Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE, SUBSCRIBE,
   NOTIFY
Accept: application/sdp, application/3gpp-ims+xml
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 4444::aaa:bbb:ccc:ddd
s=-
c=IN IP6 4444::aaa:bbb:ccc:ddd
t=0 0
m=video 4444 RTP/AVP 98
a=rtpmap:98 MPV/90000
```

2. **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 according to standard IMS procedure.

3. **SIP re-INVITE request (SCC AS to intermediate IM CN subsystem entities) - see example in table A.14.5-3**

   The SCC AS sends a SIP re-INVITE request to the remote UE through the intermediate IM CN subsystem entities containing Media A and Media B information in SDP offer.

Table A.14.5-3: SIP re-INVITE request (SCC AS to intermediate IM CN subsystem entities)

```
INVITE <sip:userR_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91e6bf6> SIP/2.0
Via: SIP/2.0/UDP sccas1.home1.net;branch=z9hG4bK240a48.12
Max-Forwards: 70
Route:
P-Asserted-Identity:
P-Access-Network-Info:
Privacy:
From:
To:
Call-ID:
Cseq:
Require:
Proxy-Require:
Supported:
Security-Verify:
Contact:
Allow:
Accept:
Content-Type:
Content-Length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 3333::aaa:bbb:ccc:ddd
s=-
t=0 0
m=audio 2222 RTP/AVP 97
c=IN IP6 3333::aaa:bbb:ccc:ddda=rtpmap:97 MPV/90000
m=video 4444 RTP/AVP 98
c=IN IP6 4444::aaa:bbb:ccc:ddd
a=rtpmap:98 MPV/90000
```

4. SIP re-INVITE request (intermedia IM CN subsystem entities to remote UE)

5. SIP 200 (OK) response (remote UE to intermediate IM CN subsystem entities) - see example in table A.14.5-5

The remote EU sends a an SIP 200 (OK) response with SDP answer.

Table A.14.5-5: SIP 200 (OK) response (remote UE to intermediate IM CN subsystem entities)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf2.visited2.net;branch=z9hG4bK240f26.3, SIP/2.0/UDP
   scscf2.home1.net;branch=z9hG4bK332d25.1, SIP/2.0/UDP
   scscf1.home1.net;branch=z9hG4bK332d25.2, SIP/2.0/UDP
   sccas1.home1.net;branch=z9hG4bK240a48.12
From:
To:
Call-ID:
Cseq: 127 INVITE
Supported: 100rel; precondition
Contact:
Allow:
Accept: application/sdp;
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933300 2987933300 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c=IN IP6 5555::aaa:bbb:ccc:ddd
t=0 0
m=audio 4444 RTP/AVP 97
a=rtpmap:97 PCMU/8000
m=video 6666 RTP/AVP 98
a=rtpmap:98 MPV/90000
```

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

7-8. SIP ACK request (SCC AS to the remote UE through intermediate IM CN subsystem entities)

The SCC AS sends an SIP ACK request to the remote UE through the intermediate IM CN subsystem entities.

9. SIP 200 (OK) response (SCC AS to intermediate IM CN subsystem entities) - see example in table A.14.5-9

The SCC AS sends a SIP 200 (OK) response containing Media B information and send it to UE-2 through the intermediate IM CN subsystem entities.

Table A.14.5-9 SIP 200 (OK) response (SCC AS to intermediate IM CN subsystem entities)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP sccas1.home1.net;branch=z9hG4bK240f42.22,
From:
To:
Call-ID:
Cseq: 127 INVITE
Supported: 100rel; precondition
Contact:
Allow:
Accept: application/sdp;
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933300 2987933300 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c=IN IP6 5555::aaa:bbb:ccc:ddd
t=0 0
m=video 6666 RTP/AVP 98
a=rtpmap:98 MPV/90000
```

10. SIP 200 (OK) response (intermediate IM CN subsystem entities to UE-2)

11-12. SIP ACK request (UE-2 to SCC AS through intermediate IM CN subsystem entities)

UE-2 sends a SIP ACK request to the intermediate IM CN subsystem entities which is terminated by the SCC AS.

A.14.6 Remote UE adds new media on controllee UE

It is assumed that UE-1 is the controller UE having collaborative session control. A user has a multimedia session on his device UE-1 with voice (Media A) and video (Media B) media flows. Subsequently, the remote UE adds the media B flow. In this scenario it is assumed that the controller UE, UE-1 automatically initiates the addition of the new media on UE-2 (Controllee) without first alerting the user and sends a SIP REFER request prior to sending back a SIP 200 (OK) response to the SIP re-INVITE request.

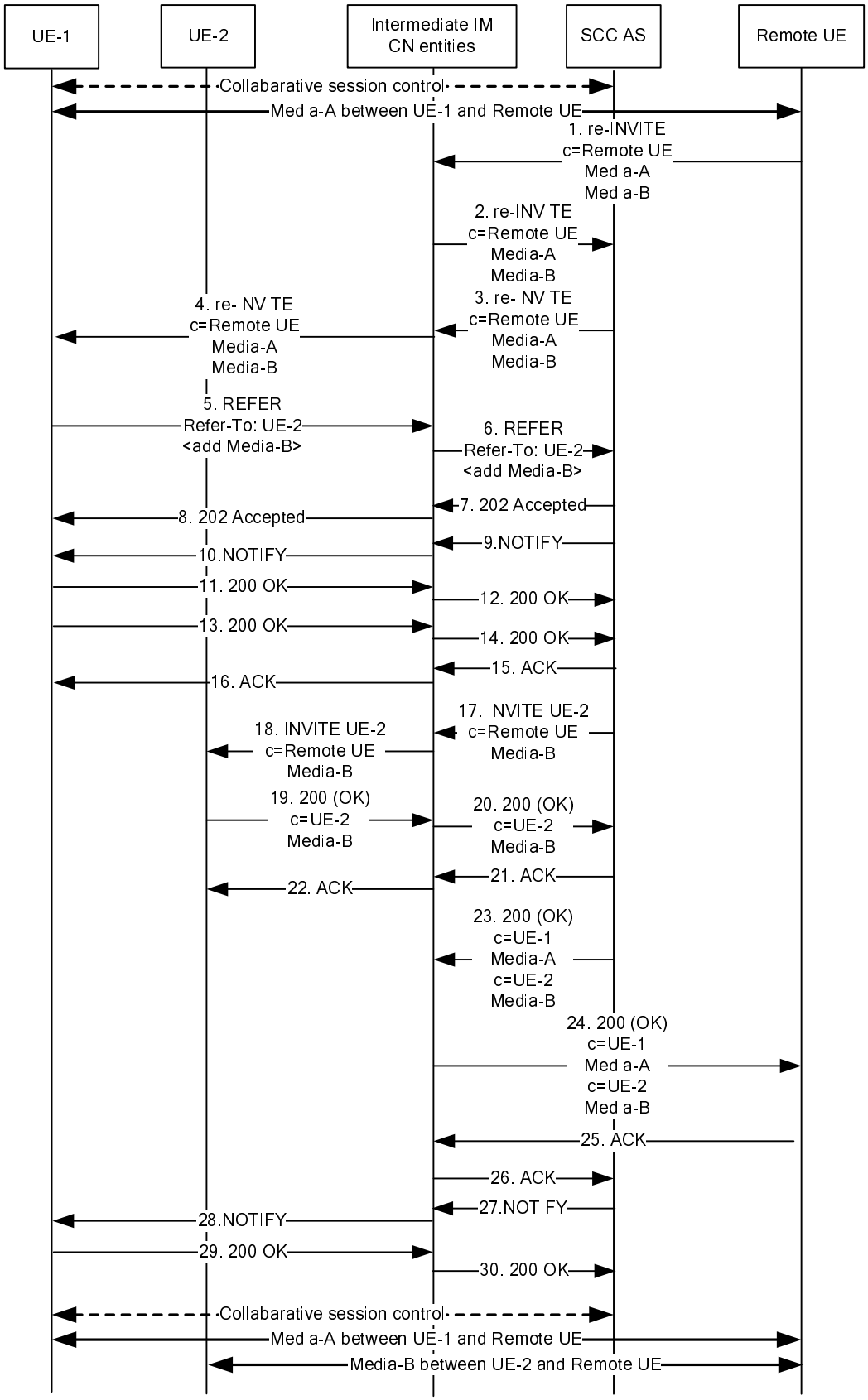


Figure A.14.6: Remote UE add new media on Controllee UE

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

1. **SIP re-INVITE request (remote UE to intermediate IM CN subsystem entities) - see example in table A.14.6-1**

The remote UE sends a SIP re-INVITE request towards the controller UE (UE-1) indicating media B is to be added in SDP offer.

Table A.14.6-1: SIP re-INVITE request (remote UE 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 [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Max-Forwards: 70
Route: <sip:pcscf2.visited2.net:7531;lr;comp=sigcomp>, <sip:orig@scscf2.home1.net;lr>
P-Asserted-Identity: "David Fan" <sip:user3_public3@home3.net>
P-Access-Network-Info: 3GPP-UTRAN-TDD; utran-cell-id-3gpp=234151D0FCE11
Privacy: none
From:
To:
Call-ID:
Cseq: 127 INVITE
Require: sec-agree
Proxy-Require: sec-agree
Supported: 100rel, precondition
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:user3_public3@home3.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c67t6br4>
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE, SUBSCRIBE, NOTIFY
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 6666 RTP/AVP 97
a=rtpmap:97 PCMU/8000
m=video 4444 RTP/AVP 98
a=rtpmap:98 MPV/90000
```

2-4. SIP re-INVITE request

Intermediate IM CN subsystem entities forward the SIP re-INVITE request to UE-1 via the SCC AS according to standard IMS procedure.

5-6. SIP REFER request (UE-1 to SCC AS through intermediate IM CN subsystem entities) - see example in table A.14.6-5

The controller UE determines to add the new media (Media B) on the controllee UE. The controller UE, UE-1 sends a SIP REFER request to the SCC AS containing a Refer-To header field containing the GRUU of controllee UE, UE-2 and a body parameter containing an m line for audio set to 0 and an m line for video with the port number set to the port number of the video media line from the SDP offer in the SIP re-INVITE request from the remote UE. The SIP REFER request also includes a Target-dialog header field containing the details of the dialog for the existing session between the controller UE, UE-1 and the remote UE.

Table A.14.6-5 SIP REFER request (UE-1 to SCC AS)

```
REFER sip:interUEtransfer@sccas1.home1.net SIP/2.0
Via:
To: sip:interUEtransfer@sccas1.home1.net
From: sip:user1_public1@home1.net;tag=34719
Call-ID: cb03a0s09a2sdfglkj490333
CSeq: 93809824 REFER
Max-Forwards: 70
P-Preferred-Identity:
Refer-To: <sip:user1_public2@home2.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-
   00a0c91e6bf6?body=m%3Daudio%200%20RTP%2FAVP%2097%0Dm%3Dvideo%204444%20RTP%2FAVP%2098>
Require: target-dialog
Target-dialog: cb03a0s09a2sdfglkj490333;remote-tag=13579;local-tag=24680
Contact: <sip:user1_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-
   00a0c91e6bf6>;+g.3gpp.iut-controller
Allow:
Accept:message/sipfrag
Content-Length:0
```

7-8. SIP 202 (Accepted) response

The SCC-AS sends a SIP 202 (Accepted) response to the controller UE-1 as response to the SIP REFER request.

9-10. SIP NOTIFY request (SCC AS to UE-1 through intermediate IM CN subsystem entities)-see example in table A.14.3.2-5

The SCC AS sends a SIP NOTIFY request to UE-1 to notify implicit subscription to the SIP REFER request results.

Table A.14.3.2-5 SIP NOTIFY request (SCC-AS to UE-1)

```
NOTIFY sip:user1_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91e6bf6 SIP/2.0
Via:
To: sip:user1_public1@home1.net;tag=34719
From: sip:interUEtransfer@sccas1.home1.net;tag=2255
Call-ID:
CSeq:
Max-Forwards:
P-Asserted-Identity:
Require:
Contact: <sip:user3_public3@home3.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c67t6br4>
Allow:
Event: refer
Subscription-State: active;expires=3600
Content-Type: message/sipfrag;version=2.0
Content-Length: (…)

SIP/2.0 100 Trying
```

11-12. SIP 200 (OK) response (UE-1 to SCC-AS through intermediate IM CN subsystem entities)

The controller UE, UE-1, acknowledges the SIP NOTIFY request by sending a SIP 200 (OK) response to SCC AS.

13-14. SIP 200 (OK) response (UE-1 to SCC-AS through intermediate IM CN subsystem entities)-see example in table A.14.6-13

The controller UE responds to the SIP re-INVITE request in step 4.

Table A.14.6-13: SIP 200 (OK) response (SCC AS to intermediate IM CN subsystem entities)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.visited1.net;branch=z9hG4bK240f34.3, SIP/2.0/UDP
   scscf1.home1.net;branch=z9hG4bK332b23.1, SIP/2.0/UDP
   sccas1.home1.net;branch=z9hG4bKnas34r4.12
From:
To:
Call-ID:
Cseq: 127 INVITE
Supported: 100rel; precondition
Contact:
Allow:
Accept: application/sdp;
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933300 2987933300 IN IP6 3333::aaa:bbb:ccc:ddd
s=-
c=IN IP6 3333::ccc:ddd:aaa:bbb
t=0 0
m=audio 8888 RTP/AVP 97
a=rtpmap:97 PCMU/8000
m=video 0 RTP/AVP 98
```

15-16. SIP ACK (SCC AS to UE-1 through intermediate IM CN subsystem entities)

17. SIP INVITE request (SCC AS to intermediate IM CN subsystem entities)-see example in table A.14.6-15

Editor's Note: This message can be sent as soon as the SIP REFER request is received.

The SCC AS sends a SIP INVITE request towards UE-2 through the intermediate IM CN subsystem entities indicating Media B information in SDP offer.

Table A.14.6-17: SIP INVITE request (SCC AS to intermediate IM CN subsystem entities)

```
INVITE sip:user1_public2@home1.net;gr=urn:uuid:2ad8920e-48a5-4a74-8d99-ad76cc7fc74 SIP/2.0
Via: SIP/2.0/UDP sccas1.home1.net;branch=z9hG4bKnas34r2.12
Max-Forwards: 70
Route: <sip:term@scscf1.home1.net;lr>, <sip:pcscf1.visited1.net:7538;lr;comp=sigcomp>
P-Asserted-Identity: "John Doe" <sip:user3_public3@home3.net>
Privacy: none
From: <sip:user3_public3@home3.net>;tag=171828
To: <sip:user1_public2@home1.net>
Call-ID: cb03a0s09a2sdfglkj490333
Cseq: 127 INVITE
Supported: 100rel, precondition
Referred-By: sip:user1_public1@home1.net
Contact: <sip:user3_public3@home3.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c67t6br4>
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE, SUBSCRIBE, NOTIFY
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 4444::bbb:aaa:ccc:ddd
t=0 0
m=audio 0 RTP/AVP 97
m=video 4444 RTP/AVP 98
a=rtpmap:98 MPV/90000
```

18. SIP INVITE request (intermediate IM CN subsystem entities to UE-2)

19. SIP 200 (OK) response (UE-2 to intermediate IM CN subsystem entities) - see example in table A.14.6-19

UE-2 responds with a SIP 200 (OK) response containing the SDP answer.

Table A.14.6-19: SIP 200 (OK) response (UE-2 to intermediate IM CN subsystem entities)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.visited1.net;branch=z9hG4bK240f34.3, SIP/2.0/UDP
   scscf1.home1.net;branch=z9hG4bK332b23.1, SIP/2.0/UDP
   sccas1.home1.net;branch=z9hG4bKnas34r2.12
From:
To: <sip:user1_public2@home1.net>;tag=237674
Call-ID:
Cseq: 127 INVITE
Supported: 100rel; precondition
Contact: <sip:user1_public2@home1.net;gr=urn:uuid:2ad8920e-48a5-4a74-8d99-
   ad76cc7fc74>;+g.3gpp.icsi-ref="urn%3Aurn-7%3gpp-service.ims.icsi.mmtel"
Allow:
Accept: application/sdp;
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933300 2987933300 IN IP6 4444::aaa:bbb:ccc:ddd
s=-
c=IN IP6 4444::aaa:bbb:ccc:ddd
t=0 0
m=audio 0 RTP/AVP 97
m=video 6666 RTP/AVP 98
a=rtpmap:98 MPV/90000
```

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

21-22. SIP ACK (SCC AS to UE-2 through intermediate IM CN subsystem entities)

23. SIP 200 (OK) response (SCC AS to intermediate IM CN subsystem entities) - see example in table A.14.6-23

In response to the SIP re-INVITE from the remote UE, the SCC AS sends a SIP 200 (OK) response containing the SDP answer towards the remote UE through IM CN subsystem entities, which includes Media A and Media B information.

Table A.14.6-23: 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=z9hG6bKnas34r4, SIP/2.0/UDP
   scscf2.visited2.net;branch=34qtrada3333.22, SIP/2.0/UDP
   pcscf2.visited2.net;branch=34qtrada5454.12, SIP/2.0/UDP
   [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
From:
To:
Call-ID:
Cseq: 127 INVITE
Supported: 100rel; precondition
Contact:
Allow:
Accept: application/sdp;
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933300 2987933300 IN IP6 3333::aaa:bbb:ccc:ddd
s=-
t=0 0
m=audio 8888 RTP/AVP 97
c=IN IP6 3333::ccc:ddd:aaa:bbb
a=rtpmap:97 PCMU/8000
m=video 6666 RTP/AVP 98
c=IN IP6 4444::bbb:aaa:ccc:ddd
a=rtpmap:98 MPV/90000
```

24. SIP 200 (OK) response (intermediate IM CN subsystem entities to remote UE)

25-26. SIP ACK (remote UE to SCC AS through intermediate IM CN subsystem entities)

The remote UEsends a SIP ACK request to the intermediate IM CN subsystem entities which terminated by the SCC AS.

27-28. SIP NOTIFY request (from SCC AS to controller UE, UE-1)

The SCC AS sends a SIP NOTIFY request to the controller UE, UE-1 to inform about the success status if the inter-UE transfer.

Table A.14.6-27: SIP NOTIFY request (SCC AS to UE-1)

```
NOTIFY sip:user1_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91e6bf6 SIP/2.0
Via:
To: sip:user1_public1@home1.net;tag=34719
From: sip:interUEtransfer@sccas1.home1.net;tag=2255
Call-ID:
CSeq:
Max-Forwards:
P-Asserted-Identity:
Require:
Contact: <sip:user3_public3@home3.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c67t6br4>
Allow:
Event: refer
Subscription-State:terminated;reason=noresource
Content-Type: message/sipfrag;version=2.0
Content-Length: (…)

SIP/2.0 200 OK
Content-Type: application/sdp

v=0
s=-
m=audio 0 RTP/AVP 97
m=video 6666 RTP/AVP 98
```

29-30. SIP 200 (OK) response (from controller UE to SCC AS)

The controller UE acknowledges the SIP NOTIFY request by sending a SIP 200 (OK) response to the SCC AS.

A.14.7 Remote UE releases media on the controller UE

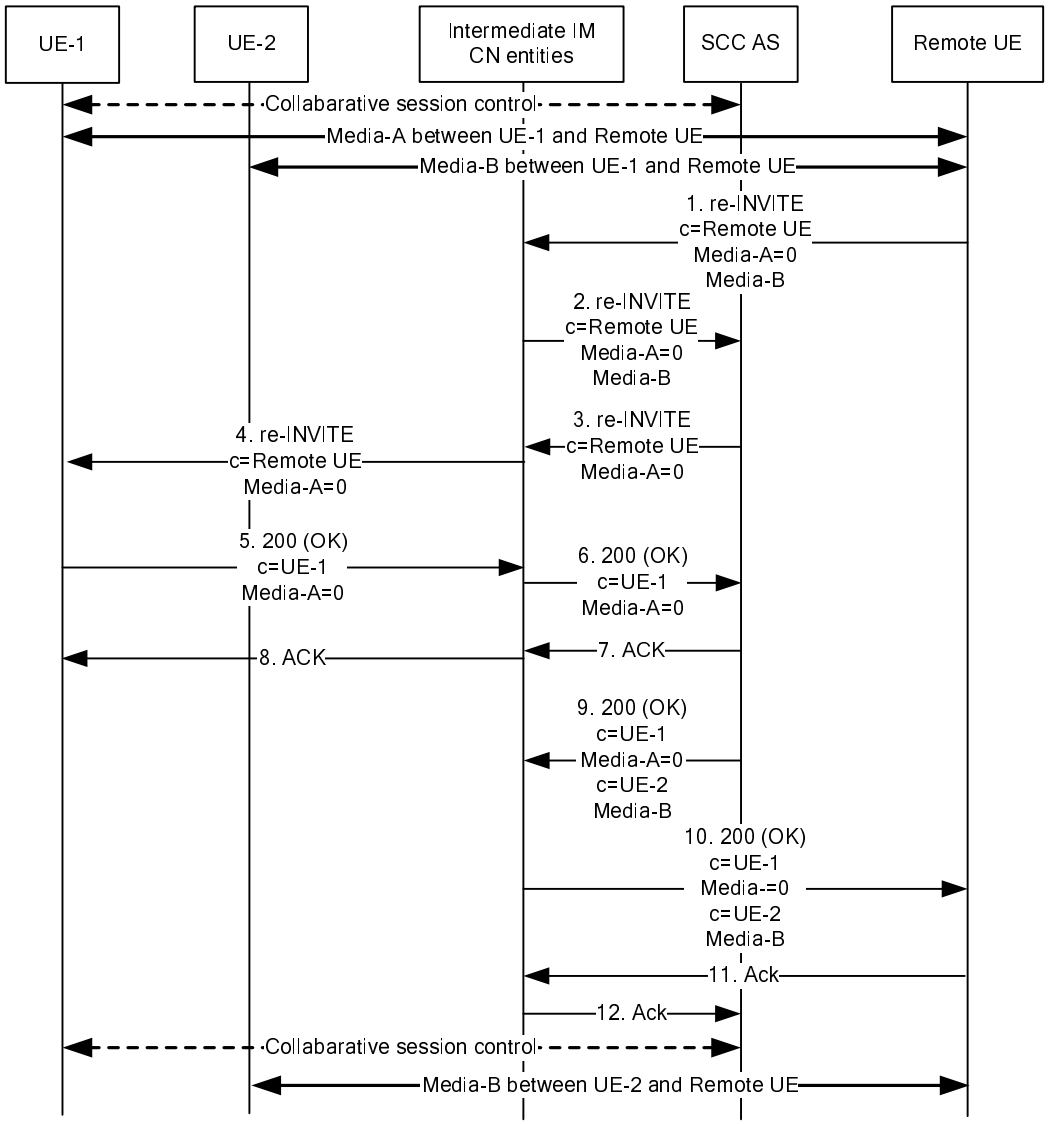


Figure A.14.7: Remote UE releases media on the controller UE

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

1. **SIP re-INVITE request (Remote Party to intermediate IM CN subsystem entities) - see example in table A.14.7-1**

   The remote UE sends a SIP re-INVITE request towards the controller UE (UE-1) indicating Media A is to be removed using SDP offer. If Media B is to be removed, the SIP re-INVITE request will send to controllee UE (UE-2).

Table A.14.7-1: SIP re-INVITE request (Remote UE 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 [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Max-Forwards: 70
Route: <sip:pcscf2.visited2.net:7531;lr;comp=sigcomp>, <sip:orig@scscf2.home1.net;lr>
P-Asserted-Identity: "David Fan" <sip:userR_public1@home1.net>
P-Access-Network-Info: 3GPP-UTRAN-TDD; utran-cell-id-3gpp=234151D0FCE11
Privacy: none
From:
To:
Call-ID:
Cseq: 127 INVITE
Require: sec-agree
Proxy-Require: sec-agree
Supported: 100rel, precondition
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_public@home1.net>; gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c67t6br4>
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE, SUBSCRIBE, NOTIFY
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
a=rtpmap:97 PCMU/8000
m=video 4444 RTP/AVP 98
a=rtpmap:98 MPV/90000
```

2. **SIP re-INVITE request (intermediate IM CN subsystem entities to SCC AS)**

   The intermediate IM CN subsystem entities forward the re-INVITE request to the SCC AS according to standard IMS procedures.

3. **SIP re-INVITE request (SCC AS to intermediate IM CN subsystem entities) - see example in table A.14.7-3**

   The SCC AS removes the Media B information and forwards it towards intermediate IM CN subsystem entities.

Table A.14.7-3: SIP INVITE request (SCC AS to intermediate IM CN subsystem entities)

```
INVITE sip:user1_public1@home1.net SIP/2.0
Via: SIP/2.0/UDP sccas1.home1.net;branch=z9hG4bKnas34r4.12
Max-Forwards: 70
Route: <sip:term@scscf2.home1.net;lr>
P-Asserted-Identity:
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 0 RTP/AVP 97
a=rtpmap: 97 PCMU/8000
```

4. SIP re-INVITE request (intermediate IM CN subsystem entities to UE-1)

The intermediate IM CN subsystem entities forward the SIP re-INVITE request to the controller UE according to standard IMS procedure.

5. SIP 200 (OK) response (UE-1 to intermediate IM CN subsystem entities) - see example in table A.14.7-5

UE-1 sends a SIP 200 (OK) response with an SDP answer.

Table A.14.7-5: SIP 200 (OK) response (UE-1 to intermediate IM CN subsystem entities)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.visited1.net;branch=z9hG4bK240f34.3, SIP/2.0/UDP
   scscf1.home1.net;branch=z9hG4bK332b23.1, SIP/2.0/UDP
   sccas1.home1.net;branch=z9hG4bKnas34r4.12
From:
To:
Call-ID:
Cseq: 127 INVITE
Supported: 100rel; precondition
Contact:
Allow:
Accept: application/sdp;
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933300 2987933300 IN IP6 3333::eee:fff:aaa:bbb
s=-
c=IN IP6 3333::eee:fff:aaa:bbb
t=0 0
m=audio 0 RTP/AVP 97
a=rtpmap:97 PCMU/8000
```

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

7-8. SIP ACK (SCC AS to UE-1 through intermediate IM CN subsystem entities)

The SCC AS sends a SIP ACK request to the controller UE through the intermediate IM CN subsystem entities.

9. SIP 200 (OK) response (intermediate IM CN subsystem entities to SCC AS) – see example in table A.14.7-9

The SCC AS sends a SIP 200 (OK) response with an SDP answer indicating Media A is removed to the intermediate IM CN subsystem entities

Table A.14.7-9: SIP 200 (OK) response (UE-1 to intermediate IM CN subsystem entities)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf1.home1.net;branch=z9hG6bKnas34r4, SIP/2.0/UDP
   scsf2.visited2.net;branch=3q5qefsdr62233.22, SIP/2.0/UDP
   pcscf2.visited2.net;branch=3q5qefsdr62245.12, SIP/2.0/UDP
   [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
From:
To:
Call-ID:
Cseq: 127 INVITE
Supported: 100rel; precondition
Contact: <sip:user1_public1@home1.net;gr=urn:uuid:f81d4fae-7dec-11d0-a765-
   00a0c91e6bf6>;+g.3gpp.iut-controller
Allow:
Accept: application/sdp;
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933300 2987933300 IN IP6 3333::aaa:bbb:ccc:ddd
s=-
c=IN IP6 3333::aaa:bbb:ccc:ddd
t=0 0
m=audio 0 RTP/AVP 97
a=rtpmap:97 PCMU/8000
m=video 6666 RTP/AVP 98
c=IN IP6 4444::aaa:bbb:ccc:ddd
a=rtpmap:98 MPV/90000
```

10. SIP 200 (OK) response (intermediate IM CN subsystem entities to remote UE)

11-12. SIP ACK (remote UE to SCC AS through intermediate IM CN subsystem entities)

The remote UE sends a SIP ACK request to UE-1 through the intermediate IM CN subsystem entities which is terminated by the SCC AS.

Editor's Note: The SDP in this flow is FFS.

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.

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 audio stream. When the SC UE connects to an IP-CAN, it decides to transfer the sessions over the IP-CAN.

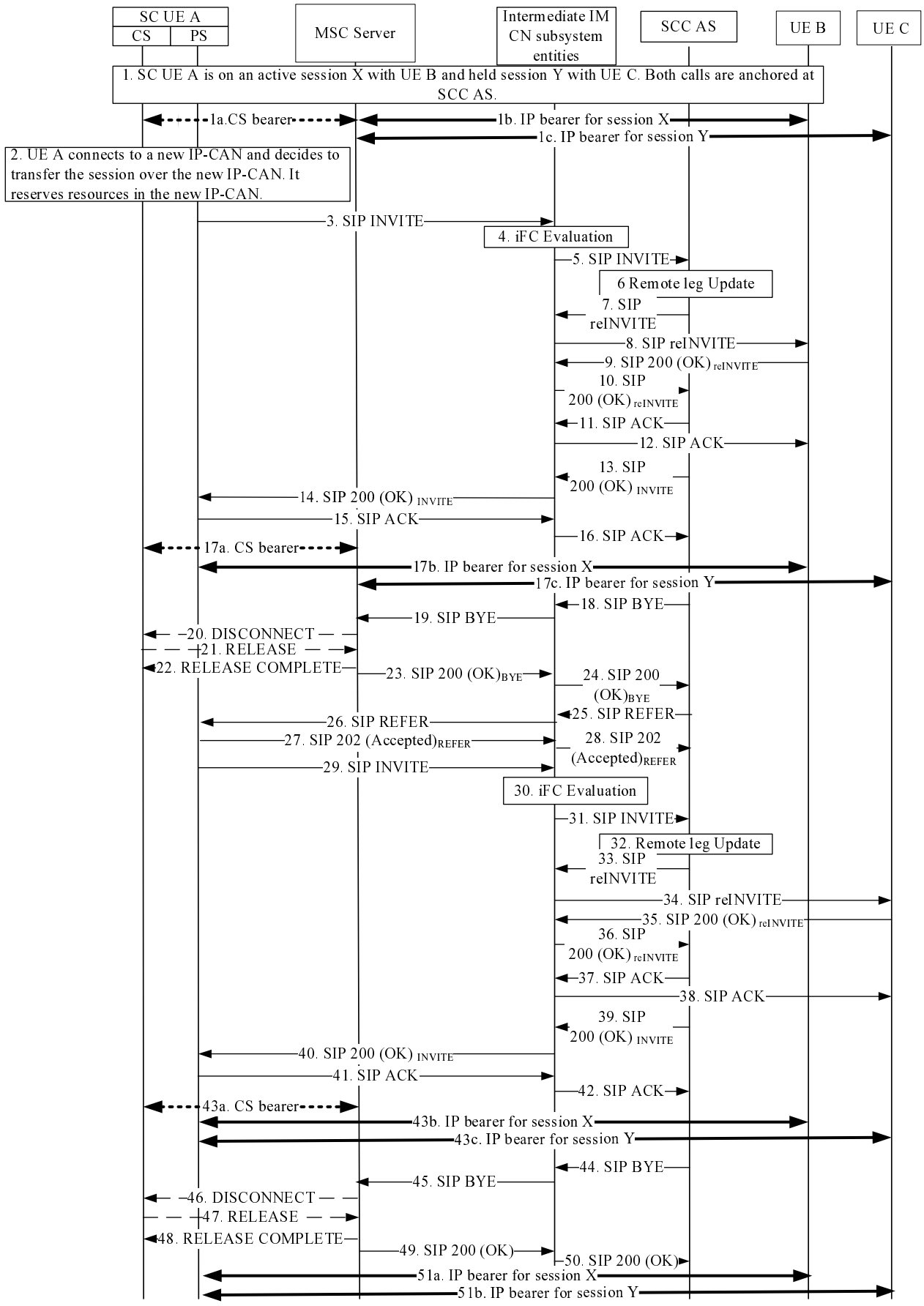


Figure A.15.2-1: Signalling flow for PS-CS Access Transfer: CS to PS

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.

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:user3_public1@home3.net;gr=urn:uuid:2ad8950e-48a5-4a74-8d99-ad76cc7fc74?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 UE C contact address 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:user3_public1@home3.net;gr=urn:uuid:2ad8950e-48a5-4a74-8d99-ad76cc7fc74 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%3gpp-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 remote target 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.

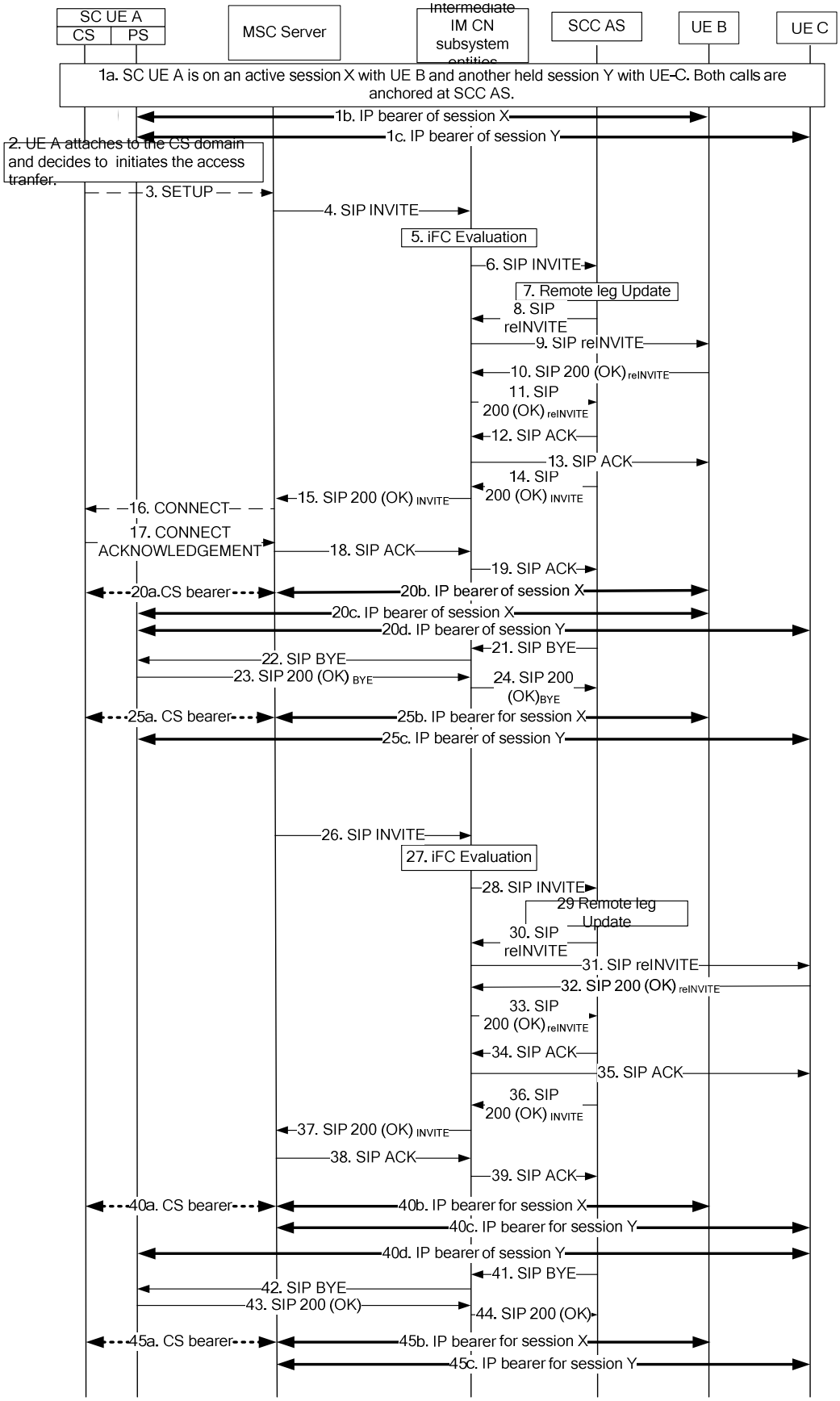
49-50. SIP 200 (OK) response (MSC Server to SCC AS via intermediate IM CN subsystem entities)

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 audio stream. When the SC UE attaches to the CS domain, it decides to transfer the sessions over the CS bearer without using the ICS capability.


SC UE A
CS
PS
MSC Server
Intermediate IM CN subsystem entities
SCC AS
UE B
UE C
1a. SC UE A is on an active session X with UE B and another held session Y with UE-C. Both calls are anchored at SCC AS.
1b. IP bearer of session X
1c. IP bearer of session Y
2. UE A attaches to the CS domain and decides to initiates the access tranfer.
3. SETUP
4. SIP INVITE
5. iFC Evaluation
6. SIP INVITE
7. Remote leg Update
8. SIP reINVITE
9. SIP reINVITE
10. SIP 200 (OK) reINVITE
11. SIP 200 (OK) reINVITE
12. SIP ACK
13. SIP ACK
14. SIP 200 (OK) INVITE
15. SIP 200 (OK) INVITE
16. CONNECT
17. CONNECT ACKNOWLEDGEMENT
18. SIP ACK
19. SIP ACK
20a.CS bearer
20b. IP bearer of session X
20c. IP bearer of session X
20d. IP bearer of session Y
21. SIP BYE
22. SIP BYE
23. SIP 200 (OK) BYE
24. SIP 200 (OK)BYE
25a. CS bearer
25b. IP bearer for session X
25c. IP bearer of session Y
26. SIP INVITE
27. iFC Evaluation
28. SIP INVITE
29 Remote leg Update
30. SIP reINVITE
31. SIP reINVITE
32. SIP 200 (OK) reINVITE
33. SIP 200 (OK) reINVITE
34. SIP ACK
35. SIP ACK
36. SIP 200 (OK) INVITE
37. SIP 200 (OK) INVITE
38. SIP ACK
39. SIP ACK
40a. CS bearer
40b. IP bearer for session X
40c. IP bearer for session Y
40d. IP bearer of session Y
41. SIP BYE
42. SIP BYE
43. SIP 200 (OK)
44. SIP 200 (OK)
45a. CS bearer
45b. IP bearer for session X
45c. IP bearer for session Y

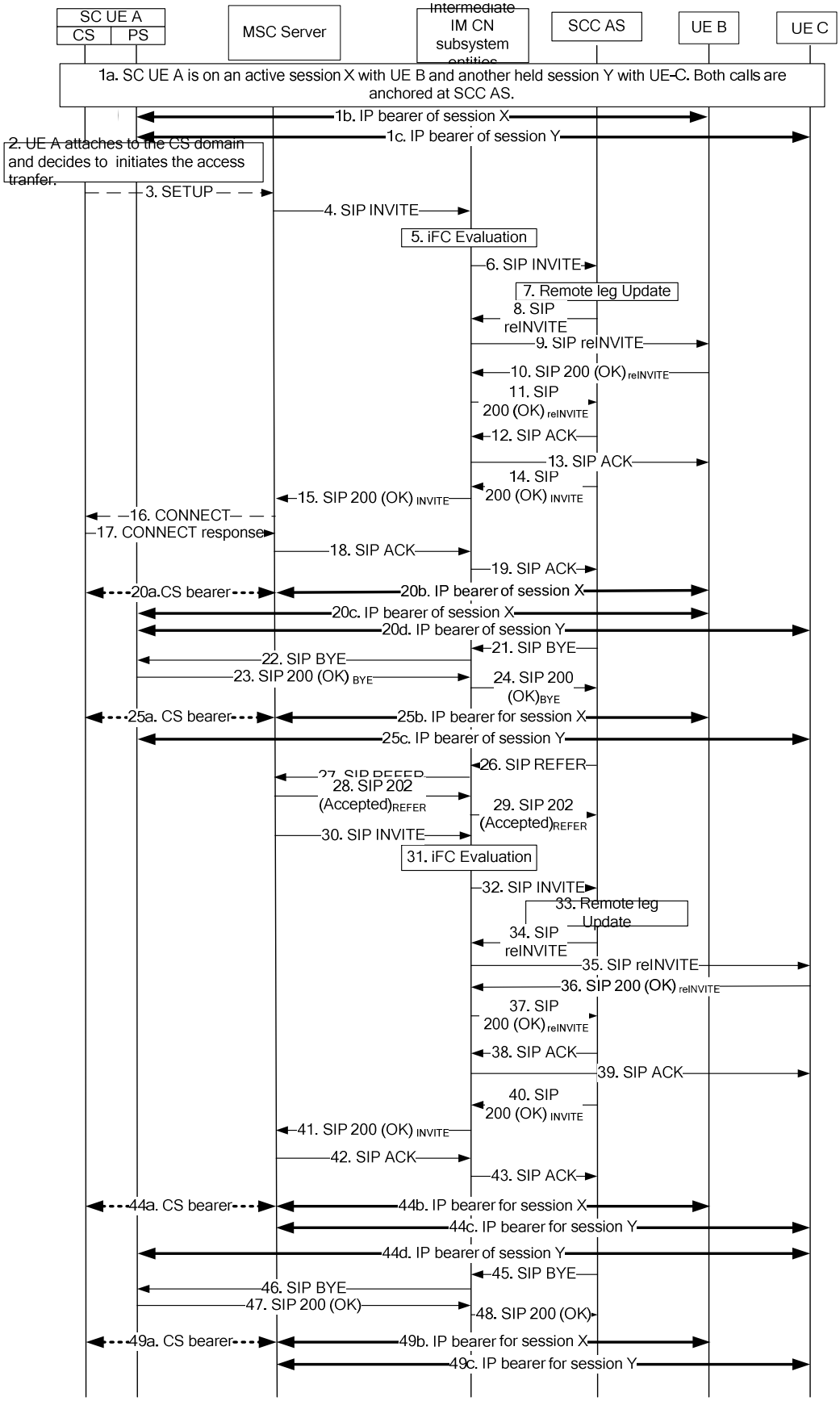


Figure A.15.3-1: Signalling flow for PS-CS access transfer: PS-CS

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


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.

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: <sip:user3_public1@home3.net;gr=urn:uuid:2ad8950e-48a5-4a74-8d99-ad76cc7fc74?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 UE C contact address 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:user3_public1@home3.net;gr=urn:uuid:2ad8950e-48a5-4a74-8d99-ad76cc7fc74 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 remote target 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.

49. 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.

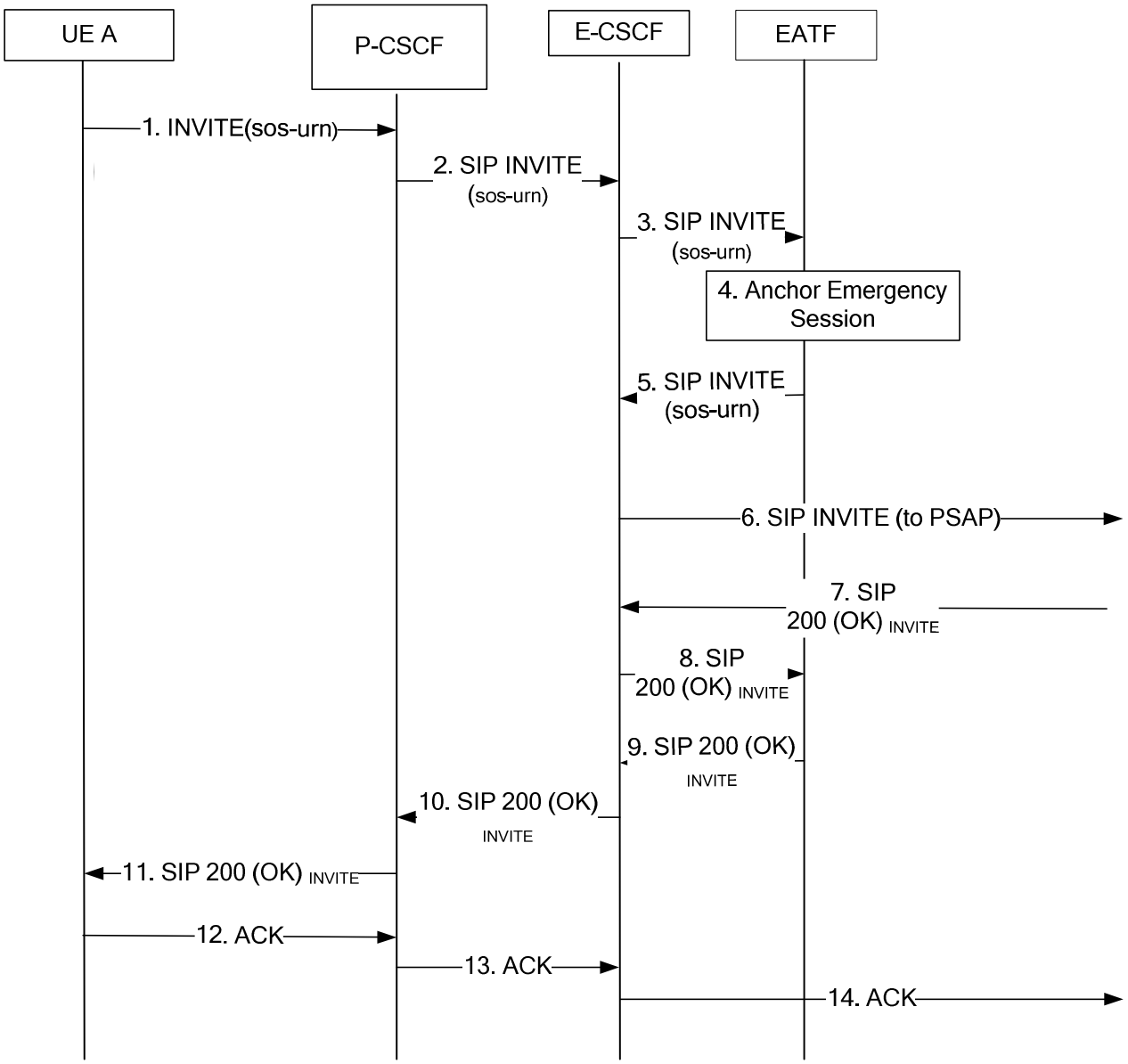


Figure A.16.2-1: Signalling flow for UE initiating an emergency session in IMS

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>
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 "gr" parameter formed from an IMEI URN as specified in 3GPP TS 24.229 [2]

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**

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>
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.

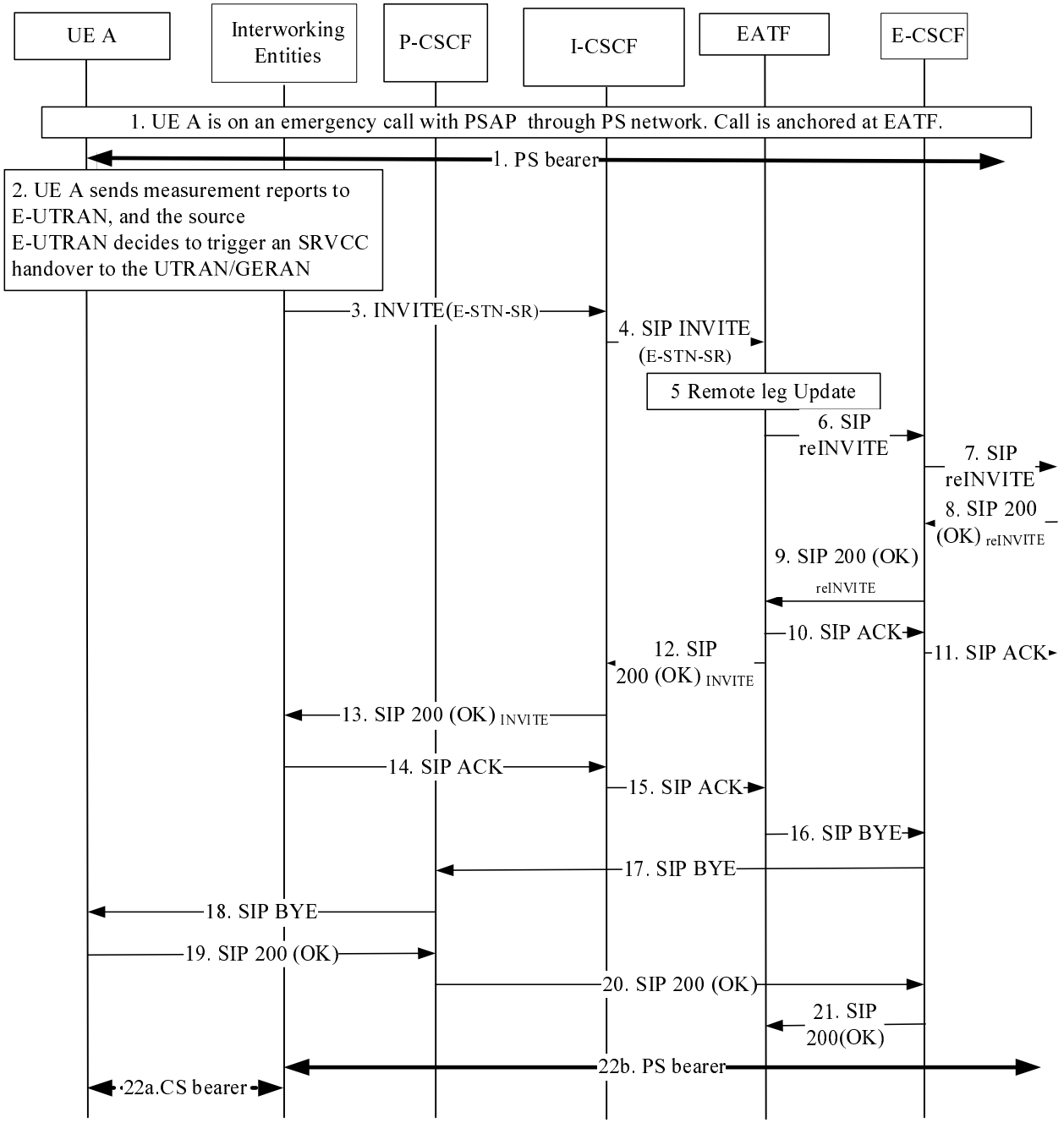


Figure A.16.3-1 Signalling flow for emergency session transfer using SRVCC procedure

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;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
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 "gr" parameter formed from an IMEI URN as specified in 3GPP TS 24.229 [2]

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>
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.

22a. CS bearer establishment (interworking entities to UE A)

22b. **IP bearer establishment (interworking entities to PSAP)**

Annex B (informative): Collaborative session state model

This annex describes a state model applicable for the UEs involved in collaborative session handling for inter-UE transfer, listing possible states and state transitions. Some procedures in collaborative session handling lead to synchronous state transitions of two different UEs.

Editor's note: the state model is not intended directly for implementation; rather it is provided as an abstraction and condensed view of what is specified in procedures in the main body of this specification (clauses 16 to 19). The final status and placement within this specification of contents currently documented in this annex is FFS.

Figure B-1 visualizes the states and state transitions for UEs within a collaborative session for inter-UE transfer.

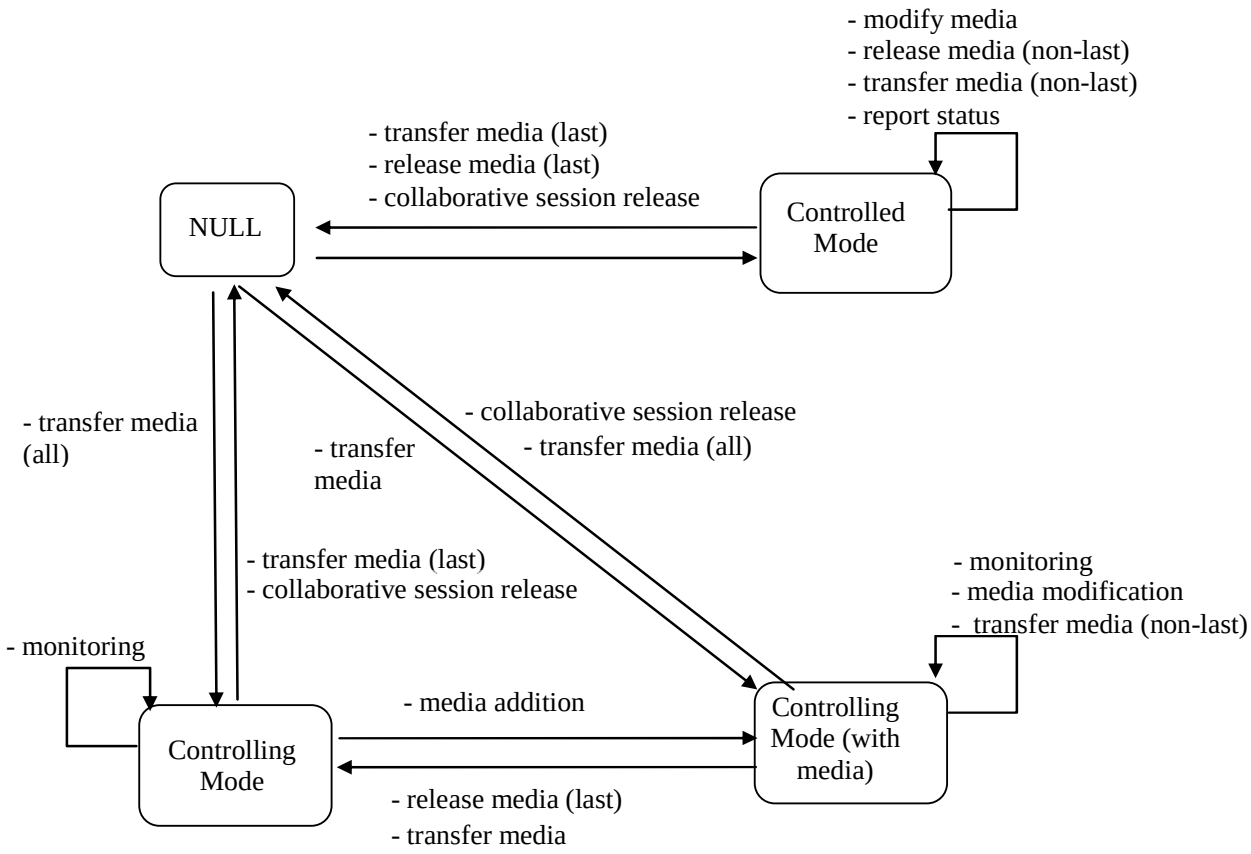


Figure B-1: State model for UEs in collaborative session for inter-UE transfer

The description of the possible states of a UE within a collaborative session is given in table B-1.

Table B-1: States of UEs in collaborative session for inter-UE transfer

State	Description
NULL	No participation in a collaborative session.
Controlling Mode	The UE participates in a collaborative session and performs the role of a controller UE, but has no media with the remote leg established.
Controlling Mode (with media)	The UE participates in a collaborative session and performs the role of a controllling UE; it has at least one media with the remote leg established.
Controlled Mode	The UE participates in a collaborative session and performs the role of a controllee UE; it has at least one media with the remote leg established.

NOTE 1: in this release the state "Controlled Mode" without media is not possible.
NOTE 2: the state "Controlled Mode" does not require any specific functionality for an IMS UE.

The possible state transitions and their triggering procedures are listed in table B-2.

Table B-2: State transitions of UEs within collaborative session for inter-UE transfer

Source state	Target state	State Transition	Nr.	Description	Coupled with
NULL	Controlling Mode	transfer media (outbound, all)	1	The UE transfers all media to a controllee UE, but keeps control of the collaborative session.	3
	Controlling Mode (with media)	transfer media (outbound)	2	The UE transfers (not all) media to a controllee UE, and keeps control of the collaborative session.	3
	Controlled Mode	transfer media (inbound, first)	3	The UE gets media transferred from another UE, which keeps collaborative session control.	1 or 2
Controlling Mode	NULL	transfer media (inbound, last)	4	The UE transfers from another UE the last media of the collaborative session back to itself.	18
		collaborative session release	5	The session with the remote party is released (by the controlling UE, the NW or the remote party).	19
	Controlling Mode	monitoring	6	The controlling UE requests monitoring status of the collaborative session from one or more controllee UEs.	22
	Controlling Mode (with media)	add media	7	The controlling UE adds media (with the remote party) onto itself; other UEs and their media are not affected.	
		transfer media (first)	8	The UE transfers media (its first) from one of the controllee UEs back to itself.	18 or 23
Controlling Mode (with media)	NULL	collaborative session release	9	See nr. 5.	20
		transfer media (last)	10	The controlling UE transfers its last media to a controllee UE and additionally hands over collaborative session control.	
	Controlling Mode	transfer media (outbound, all)	11	The controlling UE transfers (all) media to a controllee and keeps collaborative session control.	23
		release media (last)	12	The controlling UE releases its last media, but other media remain with other UEs within the collaborative session. No change of collaborative session control, which remains with the currently controlling UE.	
	Controlling Mode (with media)	monitoring	13	See nr. 6.	22
		modify media	14	The controlling UE modifies its own media, other UEs and their remain unaffected.	
		transfer media (outbound, non-last)	15	The UE transfers media to one of the controllee UEs at least one media, and thus also collaborative session control remains with it.	
		transfer media (intbound, non-first)	16	The UE transfers media from one of the controllee UEs back to itself.	18 or 23
	void	void	17	void	
Controlled Mode	NULL	transfer media (outbound, last)	18	The controllee UE gets its last media removed by the controlling UE. As a consequence, the UE is no longer participant of the collaborative session.	16 or 8
		release media (last)	19	As nr. 18, but removal of media instead of transfer.	4
		collaborative session release	20	See nr. 5.	5
	void	void	21	void	
	Controlled Mode	report status	22	The UE reports the status of is established media to the controlling UE.	6
		transfer media (inbound or outbound, non-first, non-last)	23	The controllee gets media transferred to or from another UE (controlling or controllee) in the collaborative session, dictated by the controlling UE.	11
		release media (non-last)	24	Media on the UE is released, either by the UE itself or dictated by the controlling UE; at least one media is kept.	
		modify media	25	Media on the UE is modified, either by the UE itself or dictated by the controlling UE; other UEs are not affected.	

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

C.1 General

This subclause describes the media feature tag definitions that are applicable for the 3GPP IM CN Subsystem for the realisation of the MSC server assisted mid-call feature and IUT transfer controller functions.

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.x

Editor's note: The ASN.1 Identifier will need to be updated once the IANA registration is completed.

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: none

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 [34].

C.3 Definition of media feature tag g.3gpp.iut-controller

Media feature-tag name: g.3gpp.iut-controller

ASN.1 Identifier: 1.3.6.1.8.2.x

Editor's note: The ASN.1 Identifier will need to be updated once the IANA registration is completed.

Summary of the media feature indicated by this tag: This media feature-tag when used in a SIP request or a SIP response indicates that the function sending the SIP message supports the IUT Controller functionality.This media feature tag does not imply that the controller UE capabilities are handled in the same protocol manner.

Values appropriate for use with this feature-tag: none

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 IUT controller capability

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].

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: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

Editor"s note: The MIME type "application/vnd.3gpp.mid-call+xml" as defined in this subclause is to be registered in the IANA registry for Application Media Types based upon the following template.

MIME media type name:

application

MIME 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 IETF RFC 3023 [21].

Encoding considerations:

Same as encoding considerations of application/xml as specified in IETF RFC 3023 [21]

Security considerations:

Same as general security considerations for application/xml as specified in section 10 of IETF RFC 3023 [21]. In addition, this content type provides a format for exchanging information in SIP, so the security considerations from IETF RFC 3261 [19] apply.

Interoperability considerations:

Same as interoperability considerations as specified in section 3.1 of IETF RFC 3023 [21].

Published specification:

3GPP TS 24.237 "IP Multimedia Subsystem (IMS) Service Continuity", version 9.1.0, available via http://www.3gpp.org/specs/numbering.htm.

Applications which use this media:

Applications support the service continuity as described in the published specification.

Intended usage:

COMMON

Additional information:

1. Magic number(s): none
2. File extension(s): none
3. Macintosh file type code: none
4. Object Identifiers: none

Annex E (informative): Change history

Change history							
Date	**TSG #**	**TSG Doc.**	**CR**	**Rev**	**Subject/Comment**	**Old**	**New**
2008-05	CT1#53	C1-082007			Skeleton of TS from rapporteur	-	0.0.0
2008-05	CT1#53	C1-082008 C1-082009			Incorporate agreed P-CRs from CT1#53	0.0.0	0.1.0
2008-05	Email Review				Format update	0.1.0	0.1.1
2008-06	CT1#54	C1-082394			Pseudo-CR on Introduction for signalling flows	0.1.1	0.2.0
2008-06	CT1#54	C1-082666			Application level handover	0.1.1	0.2.0
2008-06	CT1#54	C1-082667			Role of UE in registration	0.1.1	0.2.0
2008-06	CT1#54	C1-082668			Role of UE in origination and termination	0.1.1	0.2.0
2008-06	CT1#54	C1-082669			Role of UE in PS-PS session continuity	0.1.1	0.2.0
2008-06	CT1#54	C1-082671			Pseudo-CR on Cleanup of TS 24.237	0.1.1	0.2.0
2008-06	CT1#54	C1-082672			Pseudo-CR on Roles for registration in IMS	0.1.1	0.2.0
2008-06	CT1#54	C1-082766			Pseudo-CR on Signalling flows for CS call origination	0.1.1	0.2.0
2008-06	CT1#54	C1-082767			Pseudo-CR on Signalling flows for CS call termination	0.1.1	0.2.0
2008-08	CT1#55	C1-083376			Clarifications usage of CS and ICS within IMS SC	0.2.0	0.3.0
2008-08	CT1#55	C1-083377			Functional Entities	0.2.0	0.3.0
2008-08	CT1#55	C1-083378			Signalling flows for registration	0.2.0	0.3.0
2008-08	CT1#55	C1-083379			SCC AS procedures for PS-PS session continuity	0.2.0	0.3.0
2008-08	CT1#55	C1-083380			SC UE procedures for PS-PS session continuity	0.2.0	0.3.0
2008-08	CT1#55	C1-083382			Signalling flow for Media adding/deleting	0.2.0	0.3.0
2008-10	CT1#55bis	C1-083783			PS-PS Access transfer with full media transfer	1.0.0	1.1.0
2008-10	CT1#55bis	C1-083903			Editorial Cleanup	1.0.0	1.1.0
2008-10	CT1#55bis	C1-084260			network capabilities and URI assignments for IMS SC	1.0.0	1.1.0
2008-10	CT1#55bis	C1-084261			Procedures for IMS SC call origination	1.0.0	1.1.0
2008-10	CT1#55bis	C1-084262			Procedures for call termination	1.0.0	1.1.0
2008-10	CT1#55bis	C1-084265			Signalling flow for PS-CS session continuity	1.0.0	1.1.0
2008-10	CT1#55bis	C1-084266			Signalling flow for PS-PS session continuity in conjunction with PS- CS session continuity	1.0.0	1.1.0
2008-10	CT1#55bis	C1-084268			Clarification of signalling flow for call termination	1.0.0	1.1.0
2008-10	CT1#55bis	C1-084269			Clarification of signalling flow for call origination	1.0.0	1.1.0
2008-10	CT1#55bis	C1-084433			Signalling Flows for Multiple Registrations	1.0.0	1.1.0
2008-10	CT1#55bis	C1-084447			Procedures for PS-PS session transfer in conjunction with PS-CS session transfer	1.0.0	1.1.0
2008-10	CT1#55bis	C1-084448			Procedures for adding/removing media	1.0.0	1.1.0
2008-10	CT1#55bis	C1-084449			Call flow for PS-PS partial media transfer	1.0.0	1.1.0
2008-10	CT1#55bis	C1-084506			Procedures for PS-CS session transfer	1.0.0	1.1.0
2008-10	CT1#55bis				Change the Keywords to: IMS, Multimedia Session, and Session Continuity.	1.0.0	1.1.0
2008-10	Email Review				Editorial Cleanup	1.1.0	1.1.1
2008-11	CT1#56	C1-084798			Removal of Editor"s Note	1.1.1	1.2.0
2008-11	CT1#56	C1-084866			cleanup for PS-PS session transfer procedures	1.1.1	1.2.0
2008-11	CT1#56	C1-085085			Correction: No usage of CS indication in non-ICS case	1.1.1	1.2.0
2008-11	CT1#56	C1-085242			Scope of signalling flows	1.1.1	1.2.0
2008-11	CT1#56	C1-085243			Cleanup of call origination signalling flows	1.1.1	1.2.0
2008-11	CT1#56	C1-085244			Cleanup of call termination signalling flows	1.1.1	1.2.0
2008-11	CT1#56	C1-085245			Signalling flows for PS to CS session transfer	1.1.1	1.2.0
2008-11	CT1#56	C1-085246			Signalling flows for PS+CS to PS session transfer	1.1.1	1.2.0
2008-11	CT1#56	C1-085247			Clean up of SCC Registration information	1.1.1	1.2.0
2008-11	CT1#56	C1-085251			cleanups to PS-PS in conjunction with PS-CS session transfer procedures	1.1.1	1.2.0
2008-11	CT1#56	C1-085252			cleanups for media adding/removing procedures	1.1.1	1.2.0
2008-11	CT1#56	C1-085449			PS-CS session transfer procedures	1.1.1	1.2.0
2008-11	CT1#56	C1-085464			SR-VCC	1.1.1	1.2.0
2008-11	CT1#56	C1-085481			PS-PS full session transfer using Target-Dialog header	1.1.1	1.2.0
2008-11					Version 2.0.0 created for presentation to CT#42 for approval	1.2.0	2.0.0
2008-12	CT#42				Version 8.0.0 created after approval in CT#42	2.0.0	8.0.0
2009-03	CT#43	CP-090147	0002	3	Cleanup to TS 24.237	8.0.0	8.1.0
2009-03	CT#43	CP-090147	0003	1	Remove void introduction subclauses	8.0.0	8.1.0
2009-03	CT#43	CP-090147	0004	2	UE procedures for operator policy support	8.0.0	8.1.0

2009-03	CT#43	CP-090147	0005	1	Flows for originating and and terminating session in session comtinuity	8.0.0	8.1.0
2009-03	CT#43	CP-090147	0009	1	Correction SR-VCC procedures	8.0.0	8.1.0
2009-03	CT#43	CP-090147	0010	1	Correction to flows for session continuity	8.0.0	8.1.0
2009-03	CT#43	CP-090147	0011	2	Completion of IMRN functionality	8.0.0	8.1.0
2009-03	CT#43	CP-090147	0012	1	SCC AS Transparently passing Contact To and From headers	8.0.0	8.1.0
2009-03	CT#43	CP-090147	0013	3	Conveying ICS registration information using 3rd party Registration and GRUU/ICSI/IARI corrections	8.0.0	8.1.0
2009-03	CT#43	CP-090147	0014	2	Use of GRUU by SC UE	8.0.0	8.1.0
2009-03	CT#43	CP-090147	0015	2	Modification for the SCC AS procedure for CS to PS session transfer	8.0.0	8.1.0
2009-03	CT#43	CP-090147	0018	2	Definitions of Dynamic and Static STI	8.0.0	8.1.0
2009-03	CT#43	CP-090147	0019	2	Miscellaneous corrections to session continuity procedures	8.0.0	8.1.0
2009-03	CT#43	CP-090147	0020	2	SCC – Correlation	8.0.0	8.1.0
2009-06	CT#44	CP-090417	0006	3	Session transfer when PS session exists in target access – terminating case	8.1.0	8.2.0
2009-06	CT#44	CP-090417	0021	1	PS-PS+CS when service control over Gm is retained on source access leg	8.1.0	8.2.0
2009-06	CT#44	CP-090417	0022	1	Service Control Signalling Path transfer for ICS session continuity during PS-PS transfer	8.1.0	8.2.0
2009-06	CT#44	CP-090417	0026	1	Correlation CS anchored call with IMS Registration	8.1.0	8.2.0
2009-06	CT#44	CP-090417	0029	2	Clarification of the identification of the originating user	8.1.0	8.2.0
2009-06	CT#44	CP-090417	0031	3	Clarification of for the BYE request used in PS-PS partial reansfer	8.1.0	8.2.0
2009-06	CT#44				Editorial cleanup by MCC	8.1.0	8.2.0
2009-09	CT#45	CP-090669	0035	1	CS to PS transfer request by non ICS UE	8.2.0	8.3.0
2009-09	CT#45	CP-090669	0036	1	Multiple active sessions, PS to CS transfer	8.2.0	8.3.0
2009-09	CT#45	CP-090669	0052	1	Use of "Target-Dialog" for (PS+CS) to PS session transfer	8.2.0	8.3.0
2009-09	CT#45	CP-090669	0053	2	Source access leg release at the SCC AS for PS to PS session transfer	8.2.0	8.3.0
2009-09	CT#45	CP-090669	0065	2	Directing requests using the appropriate IP-CAN	8.2.0	8.3.0
2009-09	CT#45	CP-090669	0066	1	Flow updates for directing requests using the appropriate IP-CAN	8.2.0	8.3.0
2009-09	CT#45	CP-090669	0073	2	Session transfer when PS session exists in target access	8.2.0	8.3.0
2009-09	CT#45	CP-090669	0079	2	Clarification of Source Access Leg Release in PS-CS transfer procedure	8.2.0	8.3.0
2009-09	CT#45	CP-090688	0037	4	Flows for MSC Server assisted mid-call feature	8.3.0	9.0.0
2009-09	CT#45	CP-090688	0039	1	Flows for inter UE transfer without collaborative session	8.3.0	9.0.0
2009-09	CT#45	CP-090688	0040	3	Inter UE transfer without collaboriative session	8.3.0	9.0.0
2009-09	CT#45	CP-090688	0041	2	Skeleton of additions	8.3.0	9.0.0
2009-09	CT#45	CP-090688	0042	1	Scope change	8.3.0	9.0.0
2009-09	CT#45	CP-090688	0043	1	Changes to definitions and abbreviations	8.3.0	9.0.0
2009-09	CT#45	CP-090688	0044	3	Changes to functional entities	8.3.0	9.0.0
2009-09	CT#45	CP-090688	0045	1	Changes to Overview	8.3.0	9.0.0
2009-09	CT#45	CP-090688	0046	1	Change of Session Continuity to Access Transfer	8.3.0	9.0.0
2009-09	CT#45	CP-090688	0047	2	Call flow for Controller UE releases Collaborative Session	8.3.0	9.0.0
2009-09	CT#45	CP-090688	0049	3	State Model for Collaborative Session handling	8.3.0	9.0.0
2009-09	CT#45	CP-090688	0055	3	Clarification of SR VCC procedure	8.3.0	9.0.0
2009-09	CT#45	CP-090685	0067	6	Call flow for UE initiating an emergency session in IMS using SRVCC	8.3.0	9.0.0
2009-09	CT#45	CP-090685	0068	6	Call flow for EMC using SRVCC procedure	8.3.0	9.0.0
2009-09	CT#45	CP-090688	0072	1	Clarification for the Service Control Signalling Path transfer for ICS session continuity during PS-PS transfer	8.3.0	9.0.0
2009-09	CT#45	CP-090688	0089	2	Add indication of the capability of supporting mid-call feature	8.3.0	9.0.0
2009-09	CT#45	CP-090688	0094	2	MSC Server assisted mid-call feature - SRVCC	8.3.0	9.0.0
2009-09	CT#45	CP-090685	0106	2	E-SCC AS actions for IMS Emergency call	8.3.0	9.0.0
2009-09	CT#45	CP-090688	0109	1	MSC Server assisted mid-call feature - PS to CS	8.3.0	9.0.0
2009-09	CT#45	CP-090688	0111	3	MSC Server assisted mid-call feature - PS to PS	8.3.0	9.0.0
2009-12	CT#46	CP-090929	0048	2	Call flow for Remote party releases Collaborative Session	9.0.0	9.1.0
2009-12	CT#46	CP-090929	0096	3	Signalling flow for Controller UE releases media flow on controller UE	9.0.0	9.1.0
2009-12	CT#46	CP-090929	0097	3	Signalling flow for Controller UE releases media on Controllee UE	9.0.0	9.1.0
2009-12	CT#46	CP-090929	0099	3	Signalling flow for Controllee UE modify media on itself	9.0.0	9.1.0
2009-12	CT#46	CP-090929	0100	6	Signalling flow for Remote party adds new media on controllee UE	9.0.0	9.1.0
2009-12	CT#46	CP-090929	0101	3	Signalling flow for Remote UE releases media	9.0.0	9.1.0
2009-12	CT#46	CP-090929	0110	3	MSC Server assisted mid-call feature - CS to PS - Alt1	9.0.0	9.1.0
2009-12	CT#46	CP-090929	0116	4	Roles for target UE discovery for Inter-UE Transfer	9.0.0	9.1.0
2009-12	CT#46	CP-090929	0117	5	Roles of SCC AS for target UE discovery for Inter-UE Transfer	9.0.0	9.1.0
2009-12	CT#46	CP-090929	0122	1	MSC Server assisted mid-call feature - flow updates - Alt1	9.0.0	9.1.0
2009-12	CT#46	CP-090929	0124	1	MSC Server assisted mid-call feature - capability exchange	9.0.0	9.1.0

					update		
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

History

Document history		
V9.1.0	January 2010	Publication
V9.2.0	April 2010	Publication