



MTAS 17B Technical Product Description SCC AS

TECHN PRODUCT DESCR

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1 Introduction

1.1 Scope

This document is part of MTAS TPD document series and focuses on SCC AS. For MTAS common features utilized by SCC AS, see MTAS TPD Common features [7].

NOTE: SCC AS deployed without interface to IMS HSS is not supported in 15A releases.

1.2 Change History

Table 1 - Revision History

Revision	Date	Comments/Changes
A	2015-11-25	Approved for 16A
B	2016-04-28	Updated title, header and front page
C	2016-08-05	Added SRVCC Mid-call for 17A
PD3	2016-11-07	Added ATCF Info restoration via HSS for 17B
PD4	2016-11-28	Added prerequisite for P-Early-Media handling in ch. 2.6.2.2.

2 SCC AS Features

2.1 Solution overview

SCC AS is a central node in IMS to provide IMS Service Centralization and Continuity to ICS/VoLTE Users. That is, support for

- IMS Centralized Services (ICS), a single telephony service engine (IMS) is provided to ICS/VoLTE User terminals when accessing the network both over LTE/WiFi ¹PS and 2G/3G CS, and
- Single Radio Voice Call Continuity (SRVCC), needed for the access transfer of sessions when the mobile moves from LTE PS to 2G/3G CS coverage.

SCC AS is deployed on MTAS, either standalone or co-located with MMTel AS, and in solutions with or without interface to IMS HSS. In the case of no IMS HSS interface, the required subscriber data (IRS and MSISDN) is retrieved from 3pty Registration and the MSRN when breaking out call to CS is retrieved directly from HLR over the MAP SRI interface.

¹ EPC integrated WiFi supported

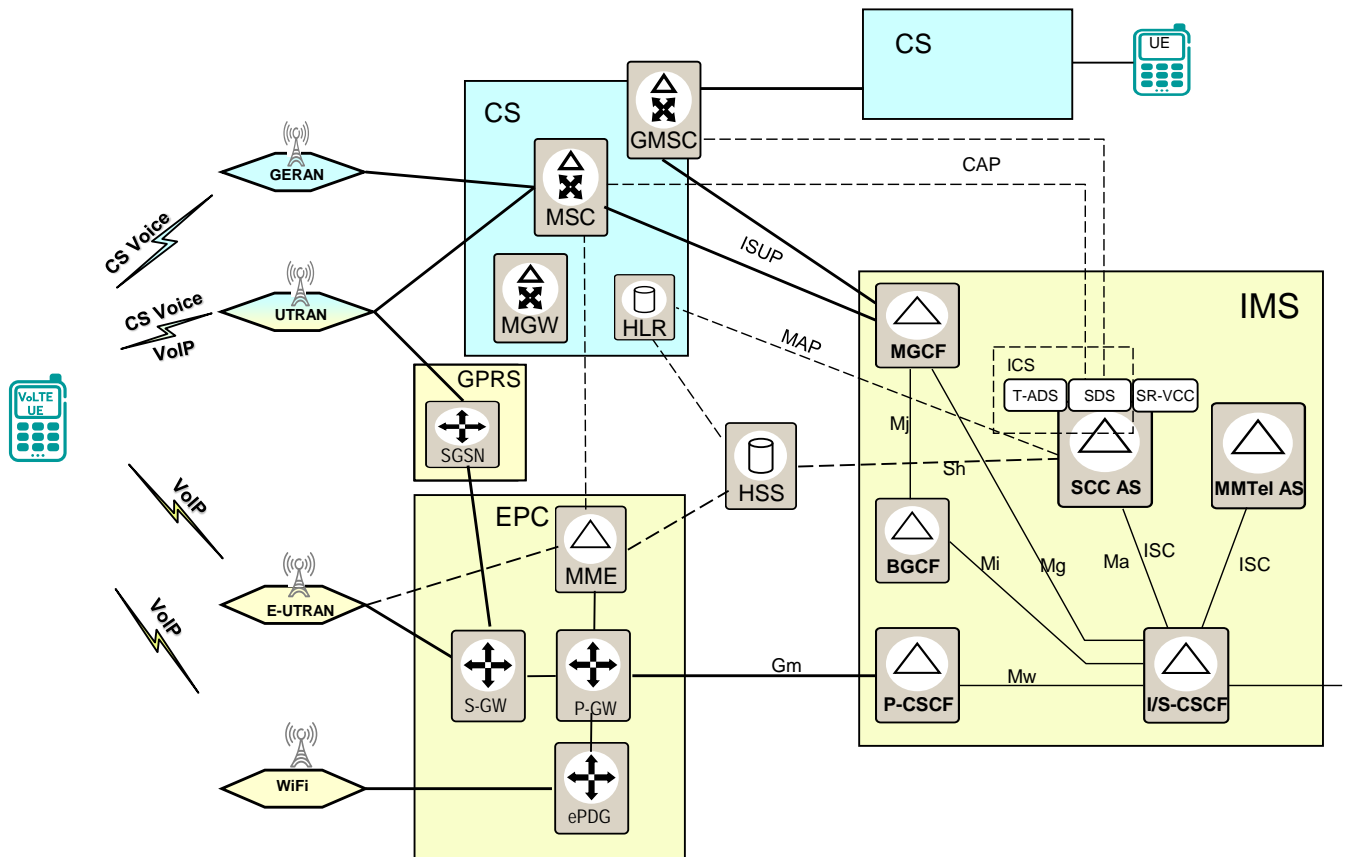


Figure 1 SCC AS in solution

2.2 SCC AS Overview

The SCC AS provides the possibility to offer IMS Centralized Services (ICS) and Single Radio Voice Call Continuity (SR-VCC) according to 3GPP standards (reg [4],[5]) which are key components for a Voice over LTE (VoLTE) solution.

Ericsson MTAS SCC AS features are 3GPP R9 and R10 compliant.

List of features:

- IMS Centralized Services (ICS)
 - Registration procedure
 - Service Domain Selection (SDS)
 - Terminating Access Domain Selection (T-ADS)



- Single Radio Voice Call Continuity (SR-VCC)
- Network Provided Location Information (NPLI)
- Support for geographical redundancy

2.3 Registration procedures

2.3.1 Description

The T-ADS feature relies on UE information being stored from UE registration, for which purpose the 3rd-party REGISTER must be extended with a MIME body containing original REGISTER and 200 OK messages as “message/sip” parts from which the contact’s device type (VoLTE/Fixed), access-type (LTE/WiFi), device instance Id, IMPI are parsed and stored.

In addition, some data for SR-VCC are handled during registration, STN-SR stored and C-MSISDN distributed to ATCF. If HSS based ATCF info storage is enabled in SCC AS (mtasSubsDataSccAtcfInfolnHss=1), ATCF registration information will be stored in HSS transparent service data, which can be used when restoring the registration information at MTAS node failover.

Normally, when SCC AS deployed with IMS HSS interface, some subscriber data like the IRS and MSISDN is fetched over the Sh interface and stored in SCC AS. However, when deployed without IMS HSS interface those data is derived from the 3pty Registration instead.

2.3.2 Example Call Flow

The UE may register from LTE PS domain via P-CSCF, or from CS by MSC enhanced for ICS on behalf of the UE.

The 3rd-party REGISTER must be extended with a MIME body containing original REGISTER and 200 OK messages as “message/sip” parts. Access domain and UE device type is obtained based on information collected from the body. SCC AS caches Contact Data for later use by SRVCC on access transfer and by ICS T-ADS function for terminating calls setup to a VoLTE UE.

SCC AS can be deployed in VoLTE solutions without access to IMS HSS, but only to EPC HSS. In this case the data in IMS HSS must be retrieved from other sources. That is, the IRS is then retrieved from extended third party registration body, C-MSISDN from first tel identity in the IRS or from served user identity when unregistered scenario. Only a deployment with SCC AS standalone is supported in this solution.

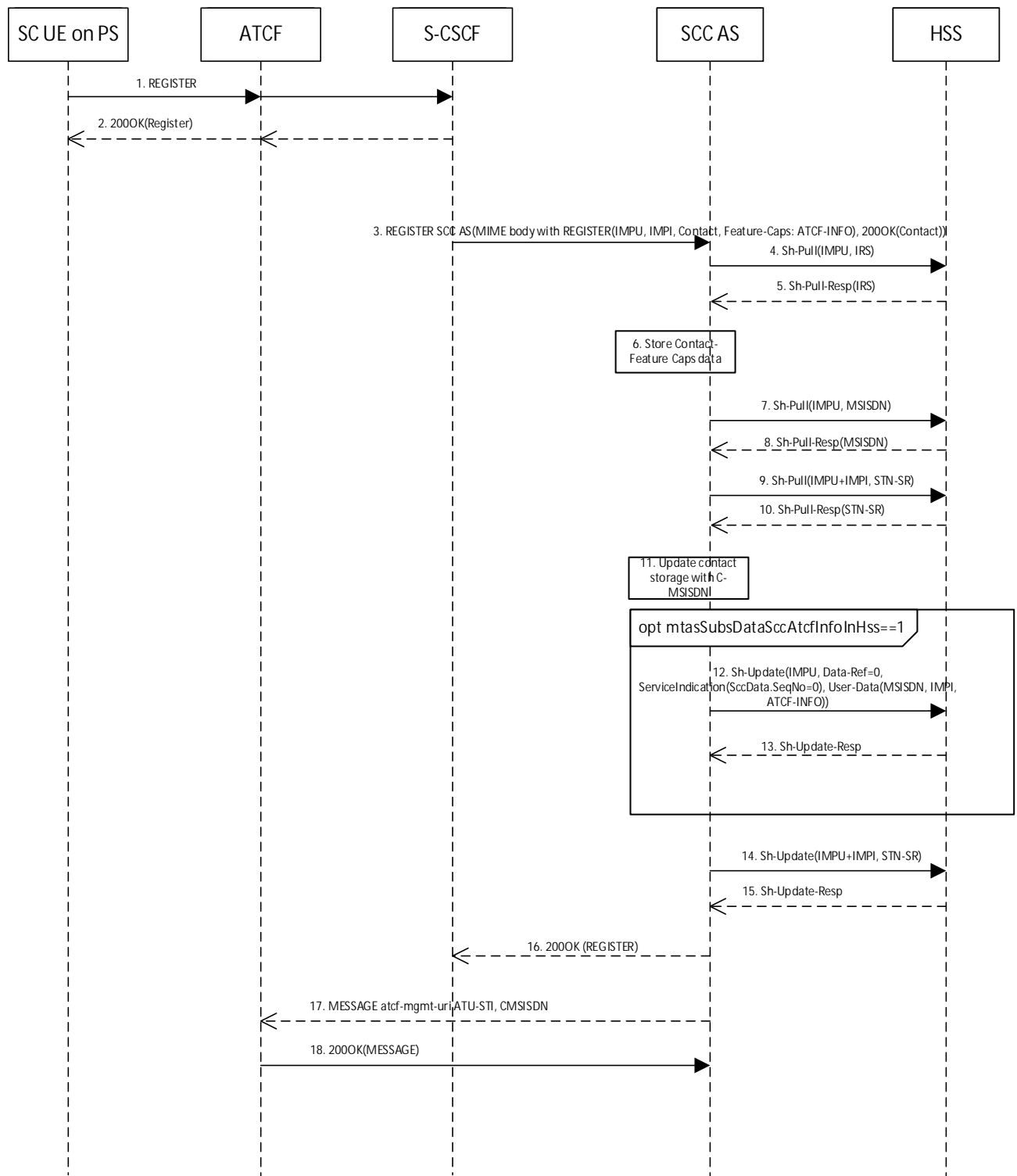


Figure 2 UE registers over LTE PS

1. The UE registers in IMS over LTE PS access. The S-CSCF receives the REGISTER message and processes the registration. If ATCF is in the path then 3GPP R12 SRVCC registration procedures will apply. ATCF inserts its own STN-SR, Management-URI and Feature-Caps header with all supported feature tags.



2. The S-CSCF accepts the REGISTER and sends a 200 OK response.
3. Filter criteria directs the S-CSCF to send a third-party register to SCC AS with the REGISTER request and 200 OK response embedded as “message/sip” parts in the MIME body.
4. For the first registered contact in IRS, an Sh-Pull Request is sent to the HSS to retrieve the IMPUs of the IRS
5. The HSS replies with a successful Sh-Pull response. The response contains the IMPUs of the IRS. The first IMPU is the default IMPU. The mapping of each IMPU to the default IMPU is stored.
6. The contact data is cached based on information parsed from the REGISTER request and 200 OK response embedded in the 3rd-party REGISTER as “message/sip” parts of the MIME body.
The following information for the registered contacts is parsed and cached in MTAS for the IRS/IMPU:

- Contact address
- IMPI (from REGISTER Authorization header)
- sip.instance Contact header feature tag
- g.3gpp.accesstype Contact header feature tag, if present
- g.3gpp.ics Contact header feature tag

The Contact header feature tags are parsed from the 200 OK response embedded in the 3rd-party REGISTER.

UE access domain, UE access node and UE terminal type are derived from the Contact header feature tags and PANI header of the embedded original REGISTER request.

- UE access domain is CS if g.3gpp.ics=“server” or PANI header indicates “3GPP-GERAN” access, else PS.
- UE access node is “SGSN” if PANI indicates “3GPP-UTRAN” access. Else if PANI is present and indicates “3GPP-E-UTRAN” access or if g.3gpp.accesstype=“cellular”, UE access node is classified as “MME”. If none of the above conditions is fulfilled, UE access node remains unknown.



- UE terminal type classification (VoLTE or Fixed) may be based on either registered contact's feature tags or P-Access-Network-Info (PANI) header.
If `mtasSubsDataMobileClassification` CM attribute contains at least one entry, the classification will be based on feature tags listed in this CM attribute only, based on presence of `+g.3gpp.ics="server"` or `+g.3gpp.accesstype="cellular"` or `+g.3gpp.accesstype="wlan"` feature tag.
Otherwise, i.e. if this setting is empty (default value), a device will be classified as "VoLTE" based on the P-Access-Network-Info header indicating 3GPP GERAN, 3GPP UTRAN or 3GPP E-UTRAN access.
If none of the above conditions is fulfilled, a device will be classified as "Fixed" by default.

Classification based on embedded REGISTER request (Contact header feature tags and PANI) takes precedence before classification based on 200 OK response (Contact header feature tags).

There can be only one "mobile/VoLTE" device registered at a time. If more devices are initially classified as "mobile/VoLTE", then the first one is taken and the others are classified as "fixed".

The SRVCC R10 registration procedure parses the Feature-Caps or Path header in the REGISTER in the MIME body searching for the Path URI of Access Transfer Control Function (ATCF) and the STN-SR within the list of the Path headers:

- `g.3gpp.atcf` feature tag includes the STN-SR (E.164 number) pointing to the ATCF node
- `g.3gpp.atcf-mgmt-uri` or `g.3gpp.atcf-psi` feature tag includes the ATCF-Management-URI. It is a PSI-routable URI used for addressing the ATCF within the IMS network.

According to 24.237 version 12.6.0, the ATCF can insert Feature-Caps capabilities indicating that ATCF is aware that all MSC-Servers in the same network as the ATCF support the access transfer of call in alerting and pre-alerting state and MSC assisted mid-call feature support.

The feature tag `g.3gpp.srvcc-alerting` is inserted by ATCF if only alerting capability is being advertised i.e. without pre-alerting capability.

Both feature tags `g.3gpp.srvcc-alerting` and `g.3gpp.ps2cs-srvcc-orig-pre-alerting` are inserted by ATCF if pre-alerting as well as alerting capability are to be advertised. This is because support for alerting is a pre-requisite for support for pre-alerting.

The feature tag `g.3gpp.mid-call` is inserted by ATCF if MSC assisted mid-call feature is supported.

7. For the first registered contact in the IRS, an Sh-Pull Request is sent to the HSS to retrieve C-MSISDN of the IRS. The basic MSISDN is used for C-MSISDN so `Data-Reference=MSISDN` is used in the request. Note that with the limitation of only one VoLTE UE in the IRS, C-MSISDN is only fetched once for the IRS, and not for subsequent contacts.



8. If there is a VoLTE UE in the IRS with SR-VCC capability the HSS replies with the C-MSISDN, else the result set is empty.

In case of SRVCC R9 registration, the steps from 9 to 18 do not apply. SCC AS accepts the registration and sends a 200 OK.

9. SCC AS checks if it had a STN-SR stored for the contact prior to the registration. If not, a Sh-Pull Request is sent to HSS to fetch the STN-SR assigned to the UE.
10. If there is a VoLTE UE in the IRS with SR-VCC capability the HSS replies with the STN-SR, else the result set is empty.
11. SCC AS compares the new STN-SR value (received in the 3rd party REGISTER or SCC AS's static value) with the earlier stored value in HSS and SCC AS. The C-MSISDN and the STN-SR is then stored in SCC AS with the cached contact and HSS is updated in step 15.
12. When `mtasSubsDataSccAtcfInfoInHss==1`, create² service data element `SccData` in HSS transparent repository data (Service Indication=`SccData` and SequenceNumber=0) with registered ATCF info and IMPI for the SC UE(MSISDN).
13. Sh-Update-Response.
14. The SCC AS accepts the registration and sends a 200 OK.
15. If the STN-SR value of the registering contact has changed then the SCC AS updates the HSS with the new STN-SR value.
16. HSS sends an empty Sh-Update-Response.
17. If the 3rd Party REGISTER contains an ATCF-Path-URI then SCC AS sends a SIP MESSAGE to ATCF as a "call out of the blue" request. The MESSAGE informs the ATCF about the Access Transfer Update Session Transfer Identification (ATU-STI) associated with the C-MSISDN of the registering contact. The ATU-STI is a routable URI pointing to SCC AS.
18. The ATCF sends "200 OK" response to MESSAGE request.

2.4 ICS Service Domain Selection (SDS)

2.4.1 Description

ICS Service Domain Selection (SDS) function assists MSC in the IMS service domain selection by implementing the `gsmSCF CAMEL` entity for InitialDP (IDP) requests from MSC/`gsmSSF` over the CAP interface, Originating SDS (O-SDS) for MO calls and Terminating SDS (T-SDS) for MT calls.

² Assumption is that the data element was previously deleted/removed at de-Reg.



The anchoring in IMS is conditional based on a number of factors: the configuration of dynamic SDS in SCC AS; the media type of the call; if subscriber is roaming or not; if the called party number qualifies as a Local number and if the called party number entered with an Escape code.

The result of dynamic SDS is then to either continue the call in CS domain or to anchor it in IMS with an IMRN. The IMRN is returned in CAP Connect response, either allocated from a pool of IMRN:s for O-SDS, or returned as the called party number with a configurable prefix for T-SDS.

2.4.2 Example Call Flow

2.4.2.1 Originating Service Domain Selection (O-SDS)

Call setup is requested from a VoLTE UE attached to the CS domain and MSC via 2G/3G access.

The MSC is not enhanced for ICS and selects IMS as originating service domain based on a CAMEL O-CSI or N-CSI trigger in the subscriber data in HLR. The MSC then uses CAMEL interface to SCC AS/gsmSCF with the aim to get the IMRN routing number and route the call over Mg interface to IMS.

With the dynamic SDS and conditions as described above the result is to either continue the call in CS domain or to anchor in IMS with the IMRN.

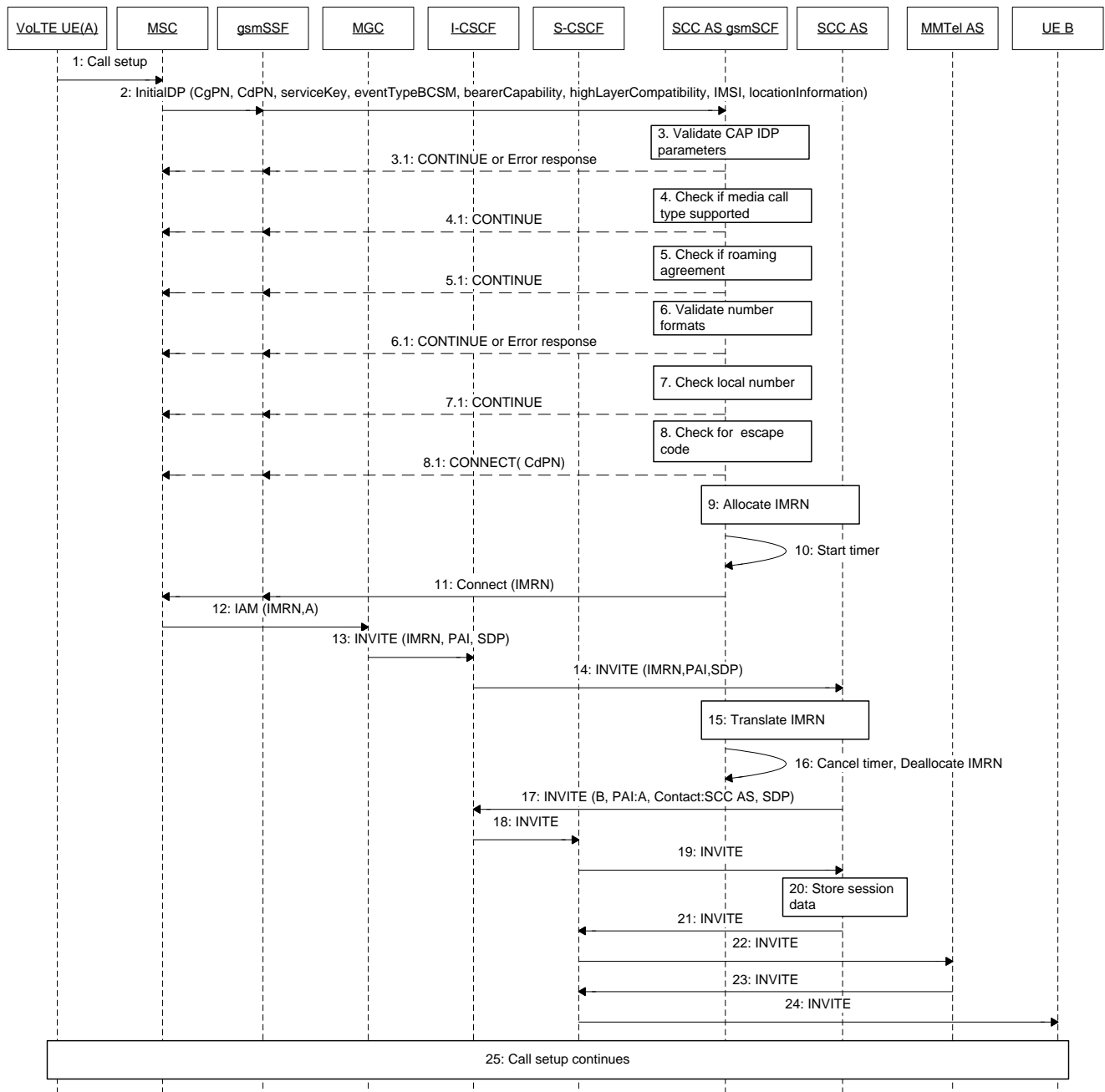


Figure 3 Call flow with originating service domain selection

- 1 The VoLTE UE is attached to the CS network and sends a call setup request to MSC.
- 2 Subscription in HLR indicates that the user is an ICS user and O-CSI or N-CSI triggers MSC/gsmSSF to request routing number from SCC AS/gsmSCF with CAP InitialDP operation.



- 3 Validate the CAP InitialDP parameters. Mandatory parameters as specified in [6] for O-SDS must be present and the serviceKey must be supported by SCC AS.
 - 3.1 If not validated, respond with CAP Error or CAP CONTINUE (depending on configuration).
- 4 If function to check media call type is enabled in SDS, then the media call type is obtained from BearerCapability (BC) and High Layer Compatibility (HLC) parameters in IDP and check with configuration if it is supported.
 - 4.1 If media call type is not supported return a CAP CONTINUE.
- 5 If function to check service profile for roaming user is enabled in SDS, then check if served user is roaming and if IMS anchoring is allowed according to a Service profile for the served user.
 - 5.1 If no IMS anchoring is decided return a CAP CONTINUE.
- 6 Validate the calling and called party numbers received in CAP IDP.
 - 6.1 If the numbers does not validate respond with CAP Error or CAP CONTINUE (depending on configuration).
- 7 If function to check local number is enabled in SDS then check if the dialed number qualifies as a local number.
 - 7.1 If the called party number qualifies as a local number return a CAP CONTINUE.
- 8 If function to check local number is enabled in SDS then check for escape code in called party number.
 - 8.1 If escape code match found, return resulting called party number as destination routing address in CAP CONNECT.
- 9 SCC AS allocates an IMRN from a pool of routing numbers, stores the allocated IMRN with
 - calling party number,
 - calledPartyNumber or calledPartyBCDNumber (depending on mtasSdsCalledPartyNumberPreference)
 - Privacy indication
 - Location information (LAI or CGI and country code)
 - IMSI
- 10 A lifetime timer is started for the allocated IMRN.
- 11 IMRN successfully allocated and returned in CAP Connect(destinationRoutingAddress=IMRN).
- 12 Call setup is routed to MGC over ISUP.
- 13 MGC sends INVITE to I-CSCF with the Request URI set to IMRN as tel URI.
- 14 I-CSCF queries HSS with the Request-URI and HSS finds out that the IMRN is a PSI. I-CSCF then routes the INVITE directly to the SCC AS hosting the PSI.



- 15 SCC AS translates the IMRN and maps the associated parameters to an outgoing INVITE:
 - Contact: SCC AS address
 - Request URI and To header: calledPartyNumber or calledPartyBCDNumber (depending on configured policy) tel URI from stored IMRN association.
 - If over-decadic digits are present, they're mapped according to following schema:
 - calledPartyNumber: Hex'A to 'A', Hex'B to '*', Hex'C to '#', Hex'D to 'D', Hex'E to 'E', Hex'F to 'F'
 - calledPartyBCDNumber: Hex'A to '*', Hex'B to '#', Hex'C to 'C', Hex'D to 'D', Hex'E to 'E', Hex'F used as a filler
 - P-Asserted-Identity: use calling party number from stored IMRN association, in E.164 international format as Tel: URI
 - Privacy header: Privacy indication from stored IMRN association
 - P-Access-Network-Info: LAI or CGI from stored IMRN association, access-type = 3GPP-GERAN and "network-provided" parameter
 - Include History-Info header from incoming INVITE, but only if it contains more than one diversion
 - Set Route header to configured I-CSCF address and append orig parameter
 - SDP from incoming INVITE
- 16 The IMRN lifetime timer is cancelled and the IMRN is deallocated in the central pool and made free for other calls.
- 17 The INVITE is sent to configured I-CSCF address.
- 18 The INVITE is sent to the S-CSCF serving user A.
- 19 SCC Originating Unregistered or SCC Originating session case triggers SCC AS.
- 20 SCC AS obtains access domain and UE terminal type from PANI header and the Contact header feature tags to cache them in Session Data. Access domain is CS and UE terminal type is "mobile" (VoLTE) in this case.
- 21 INVITE sent to S-CSCF.
- 22 Originating Unregistered or Originating session case triggers MMTel AS and MMTel services
- 23 INVITE sent to S-CSCF.
- 24 INVITE sent to domain serving user B.



25 Call setup continues.

2.4.3 Configuration

Examples of node-level configuration parameters related to the SDS feature:

- Administrative state (enable/disable)
- Service profile that defines whether a roaming user on VPLMN is to be anchored in IMS or not.
- HPLMN settings such as mapping from service key in CAP InitialDP request to a Service profile and settings for the HPLMN such as CC, prefixes used in dialing plan, a local number list and an escape code
- VPLMN settings such as CC, prefixes used in dialing plan and a local number list.
- IMRN range for O-SDS.
- IMRN prefix for T-SDS.

2.4.4 Performance management

Example of performance counters provided by SCC AS for the feature:

- Successful CAP InitialDP requests. Incremented by 1 when the SCC AS determines that a CAP InitialDP request is successful.
- Faulty CAP InitialDP requests. Incremented by 1 when the SCC AS sends a failure response to a CAP Initial DP request.
- Percentage of IMRN numbers in use.

2.4.5 Fault management

SCC SDS license absent alarm.

2.5 ICS Terminating Access Domain Selection (T-ADS)

2.5.1 Description

The purpose of Terminating Access Domain Selection (T-ADS) function is to select access domain, LTE/WiFi PS or 2G/3G CS access, for terminating calls to VoLTE or 2G/3G subscribers.

The T-ADS procedure is based on

- The type of served subscriber (VoLTE or 2G/3G),
- cached information on registered contacts (access domain (PS/CS), access type (LTE/WiFi), MSISDN),



- access domain (CS, PS) for ongoing VoLTE session and most recent VoLTE session, and
- the TADS information for VoLTE PS contacts (if the current location radio access network supports IMS Voice over PS) as obtained from HSS.
- MSRN/CSRN when breakout to CS is selected

The CS access is either defined by the on-behalf-of-MSC-registered contact when the MSC is enhanced for ICS or the Mg model with the legacy MSC server. Breakout to CS is selected on the Mg interface with the route defined by either a configured prefix added to the MSISDN and call is routed via GMSC, or by the CSRN/MSRN as obtained from HSS/HLR over Sh or MAP interface and the call is routed directly to the VMSC.

SCC AS can be deployed in VoLTE solutions without access to IMS HSS, but only to EPC HSS. In this case when breakout to CS is selected using MSRN, the MSRN is retrieved over an ETSI MAP interface direct to HLR.

A retry option can be configured in order to retry with a CS breakout when a selected VoLTE PS or CS contact did not answer or responded with an error as matched with a configured list of SIP error codes for the PS contact failure and CS contact failure respectively.

2.5.2 Example Call Flow

2.5.2.1 VoLTE or 2G/3G case

SCC AS is normally used to offer IMS Centralized Services (ICS) to VoLTE users with both LTE and CS access, where T-ADS selection criteria is based on registration status, HSS parameters (MSISDN, TADSinfo) etc. However, SCC AS may also be used for traditional 2G/3G CS only mobiles, in which case terminating calls are always broken out using the Request-URI with a prefix, and no need for HSS parameters regarding this.

SCC AS supports a mix of VoLTE and 2G/3G users. This may be achieved with HSS configuration for the subscriber, ServerName property in the iFC for SCC AS to indicate the type of user. The call case, VoLTE or 2G/3G, is determined and SCC AS can act accordingly for the call. That is, to interface HSS or not for MSISDN and to apply T-ADS or not.

2.5.2.2 Deliver call with TADS

On a VoLTE terminating call setup received by SCC AS, the T-ADS function takes the following knowledge/data into consideration when deciding where to terminate the call:

- 1) Use the access domain, PS or CS, of ongoing or recently terminated session.
- 2) Based on registered contacts information:
 - Access domain (CS or PS)
 - Terminal type (VoLTE, Fixed or Unknown)
 - Access type (LTE or WiFi)



- Voice over PS support for the most recently used LTE radio access retrieved from HSS.

The T-ADS procedure may result in delivering the call:

- to PS contact on LTE
- to PS contact on WiFi
- to CS contact (ICS enhanced MSC)
- to PS & CS contact
- to CS with breakout, based on prefix, or allocated MSRN as retrieved from HSS/HLR

If the call case is 2G/3G, the T-ADS decision is to breakout. MSISDN is in this case defined as the Request-URI in incoming INVITE.

In the case SCC AS is deployed without access to IMS HSS the MSRN for breakout to CS is retrieved directly from HLR over the ETSI MAP SRI interface using procedure supports MTRR and Camel subscribers.

TADS can also be configured to retry with CS breakout in case PS is selected but call fails or times out.

The following sections describes the cases where SCC AS and TADS selection algorithm results in to deliver call to 1) PS LTE contact, 2) PS WiFi contact, 3) CS with breakout based on MSRN, and 4) deliver call to CS with breakout based on MSRN as retrieved from HLR with MAP

2.5.2.3 Deliver call to PS contact on LTE

A terminating session setup request is received by SCC AS. T-ADS procedure result is to deliver the call to LTE PS contact when the recent location PS radio access network supports IMS Voice over PS.

The caller preference is set to reject any CS contact by adding the feature tag +g.3gpp.ics="server" in the Reject-Contact header. If the call is answered by a VoLTE UE, determined by the PANI in 200 OK, the session data is updated with UE type=VoLTE and access domain= PS.

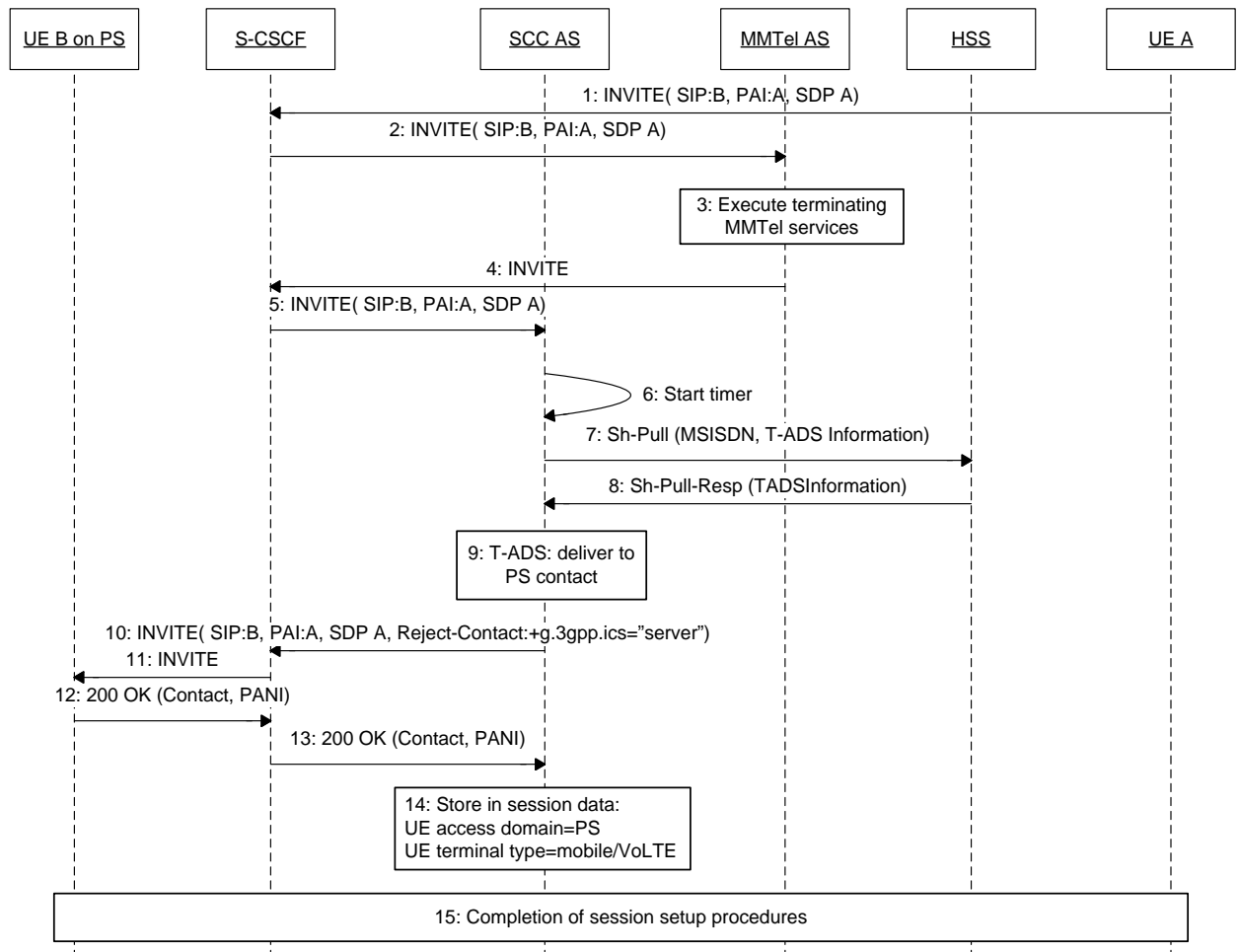


Figure 4 T-ADS deliver call to LTE PS

- 1 A terminating INVITE is received from UE A.
- 2 Terminating filter criteria directs the INVITE to MMTel AS.
- 3 Terminating MMTel services are executed.
- 4 INVITE is passed on to S-CSCF in a new dialog.
- 5 Terminating filter criteria directs the INVITE to the SCC AS.

The SCC AS performs T-ADS, in this case a VoLTE contact is registered in the PS domain.

- 6 A timer (mtasTadsHssTimer) is started for the HSS query for TADS information.
- 7 An Sh-Pull request with the MSISDN for the VoLTE UE is sent to HSS to retrieve the TADS information.
- 8 The HSS replies before timer expires with a successful Sh-Pull-Response containing the TADS information. In this case IMS voice is supported on the current radio access.
- 9 Result of T-ADS procedure is to deliver the call to LTE PS contact.



- 10 The SCC AS creates a new INVITE with caller preferences set in accordance with the result of the T-ADS. That is, if both PS and CS contacts are registered, the CS contact is rejected by adding the feature tag +g.3gpp.ics="server" to the Reject-Contact header. The INVITE is sent to the second URI in the pre-existing route set, which is expected to be S-CSCF.
- 11 The INVITE is routed by S-CSCF based on the registered contact addresses and the caller preferences set by SCC AS, that is, the call may be routed to VoLTE UE B on PS and to other fixed contacts on PS.
- 12 The call is answered and 200 OK responded.
- 13 The 200 OK is routed to SCC AS via S-CSCF, with a PANI header to indicate the type of UE answered the call.
- 14 SCC AS obtains access domain and UE terminal type from PANI header and the Contact header feature tags and caches them in Session Data:
 - UE access domain is PS if g.3gpp.ics feature tag is absent and if PANI access-type is not 3GPP-GERAN.
 - Terminal is classified as "mobile" (VoLTE) if PANI indicates UTRAN or E-UTRAN access, otherwise it is classified as "fixed".

If PANI was absent, UE terminal type would be determined by matching feature tags configured in the mtasSubsDataMobileClassification CM attribute against the Contact header feature tags.

Access domain is PS and UE terminal type is "mobile" (VoLTE) in this case.
- 15 The session setup is completed.

2.5.2.4 Deliver call to PS contact on WiFi

A terminating session setup request is received by SCC AS. T-ADS decision based on registered contacts is to deliver the call to WiFi PS contact.

In this case there is no query for TADS information from HSS, the INVITE is routed to S-CSCF with caller preference for the contact registered on WiFi. The caller preference is set to accept wlan contact by adding the feature tag +g.3gpp.accesstype="wlan"; and +sip.instance=<sip.instance of VoLTE UE> in the Accept-Contact header.

In case the UE has moved out of WiFi (to LTE since the registration is still there), and such re-registration has not reached SCC AS³, S-CSCF will return 480 which will trigger a retry with TADS for LTE contact as described in 2.5.2.3.

2.5.2.5 Deliver call to CS with breakout based on MSRN

Precondition:

³ This can happen in non-Ericsson CSCF case, when iFC is not configured to trigger third party registrations to SCC AS on Re-Registration when on LTE.



- Breakout policy configured to breakout with MSRN

A terminating session setup request is received by SCC AS. The call may originate from IMS, or from CS with T-SDS. The MSC is not enhanced for ICS and T-ADS procedure result is to deliver the call to CS with breakout based on the CSRN as retrieved from HSS which is a temporary allocated MSRN at the visited MSC.

The Request URI is set to tel:'+'CSRN;rn='+'CSRN and To=tel:'+'CSRN.

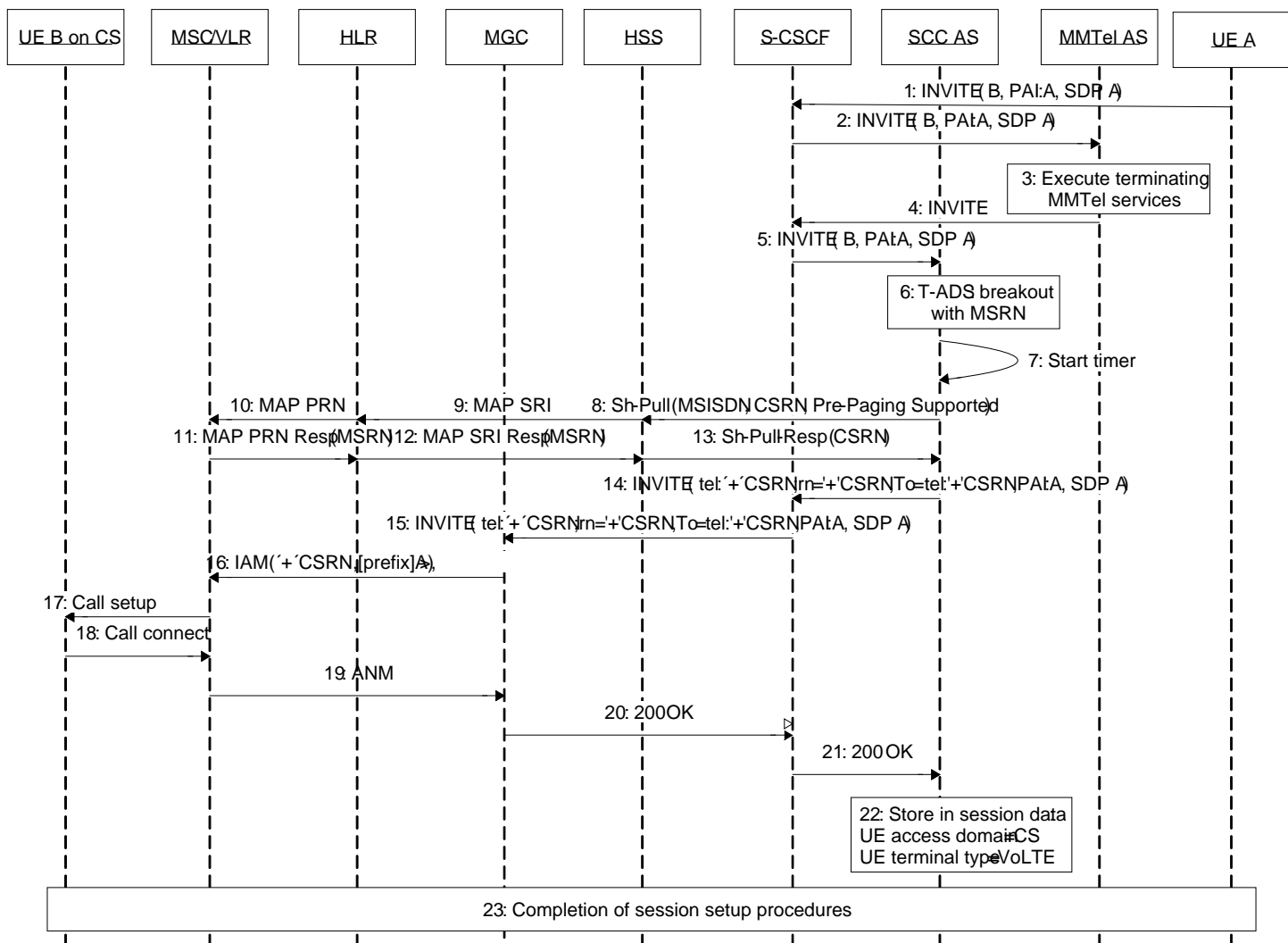


Figure 5 T-ADS and call delivered to CS with MSRN

- 1 A terminating INVITE is received by B's serving S-CSCF.
- 2 Terminating filter criteria directs the INVITE to MMTel AS.
- 3 Terminating MMTel services are executed.
- 4 INVITE is passed on to S-CSCF.



- 5 Terminating filter criteria directs the INVITE to the SCC AS.
- 6 The SCC AS performs T-ADS with the result to breakout to CS with MSRN/CSRN.
- 7 A timer (mtasTadsHssCsrnTimer) is started for the HSS query for CSRN/MSRN.
- 8 An Sh-Pull request with the MSISDN for the VoLTE UE is sent to HSS to retrieve the CSRN/MSRN. Depending on the configured breakout policy, MSRN with or without pre-paging support, the Pre-paging Supported AVP is set accordingly.
- 9 HSS sets a timer and sends a MAP SRI to HLR to request the MSRN, and suppressing any CAMEL trigger and announcement generation at the serving MSC. In addition pre-paging supported is indicated in the request.
- 10 HLR queries the serving network VLR for the roaming number with MAP PRN operation.
- 11 The VLR assigns a temporary MSRN selected from a pool and sends the MSRN back to the HLR. If pre-paging is indicated in the request and the MSC supports it, a connection is established with the UE B before returning the MSRN.
- 12 The HLR passes the MSRN back to HSS.
- 13 The HSS passes the MSRN back to SCC AS as CSRN in Sh-Pull-Resp.
- 14 The SCC AS sets Request URI to tel:+CSRN with rn=+CSRN and To=tel:+CSRN, and sends INVITE to S-CSCF.
- 15 S-CSCF determines that SCC-AS is acting as B2BUA and that the R-URI has been modified by SCC-AS. S-CSCF will therefore stop IFC evaluation for the terminating call and will convert the call into a Call_div case. SPT for the Call_div case is suppressed at this point.
When S-CSCF starts the Call_div case, it will apply B-number normalization through ENUM query if the 'rn' parameter is not set. The presence of the 'rn' parameter and CSCF configured to suppress ENUM before RN routing (CscfExtNetSelOnRnBeforeEnum=True) have the effect that the S-CSCF takes a breakout towards BGCF and MGC without doing further B-number normalization. In any case, B-number normalization in S-CSCF should be configured such that it does not modify this number.
- 16 IAM sent to the MSC where the MSRN was allocated.
- 17 Call setup is completed to VoLTE UE on CS.
- 18 VoLTE UE on CS answers the call and a connect message is sent to the MSC. At this point the allocated MSRN is marked free and can be reassigned for other calls.
- 19 ANM sent to MGC.
- 20 200 OK sent to S-CSCF serving user B.
- 21 200 OK sent to SCC AS.



22 SCC AS stores access domain (CS) and UE terminal type (VoLTE) in session data, based on that the 200 OK is the response on a breakout to CS.

23 The session setup is completed.

2.5.2.6 Deliver call to CS with breakout based on MSRN when no IMS HSS deployment

Precondition:

- Breakout policy configured to breakout with MSRN, with or without pre-paging support in MSC (mtasTadsBreakoutPolicy).
- SCC AS configured without HSS (mtasSccHssDeploymentMode=1)
- MTRR support is enabled (mtasTadsMtrr)
- A valid license for "SCC AS without IMS HSS" exists.
- MTAS Circuit Switch Integration subsystem administrative state is unlocked (mtasCsiAdministrativeState=1).
- ETSI MAP interface on SS7/SCTP stack is enabled

Terminating or terminating unregistered iFC triggers SCC AS. In the unregistered case the MSISDN is obtained from the Request-URI or P-Served-User header depending on configuration (mtasSipSupportPServedUserHeader) and cached for the served user.

The result of the T-ADS procedure is to deliver the call to CS with breakout based on MSRN allocated in CS domain.

The MSRN is allocated with a MAP Send Routing Information (SRI) request over an ETSI MAP interface to HLR with an indication that MTRR is supported, and from the HLR a MAP Provide Roaming Number (PRN) to the visited networks VLR where the MSRN is temporary allocated for the call. The resulting MSRN is returned to HLR and then to SCC AS in the MAP SRI response.

After SCC AS sends SRI with MTRR supported and gets MSRN from HLR, the MTRR procedure is applied when resume call handling is requested from MSC/VLR with a MAP RCH request. The MTRR procedure is to cancel current call setup and request a second MSRN allocation with MAP SRI and with MTRR disabled.

The Request URI is set to tel:'+'MSRN;rn='+'MSRN and To=tel:'+'MSRN.

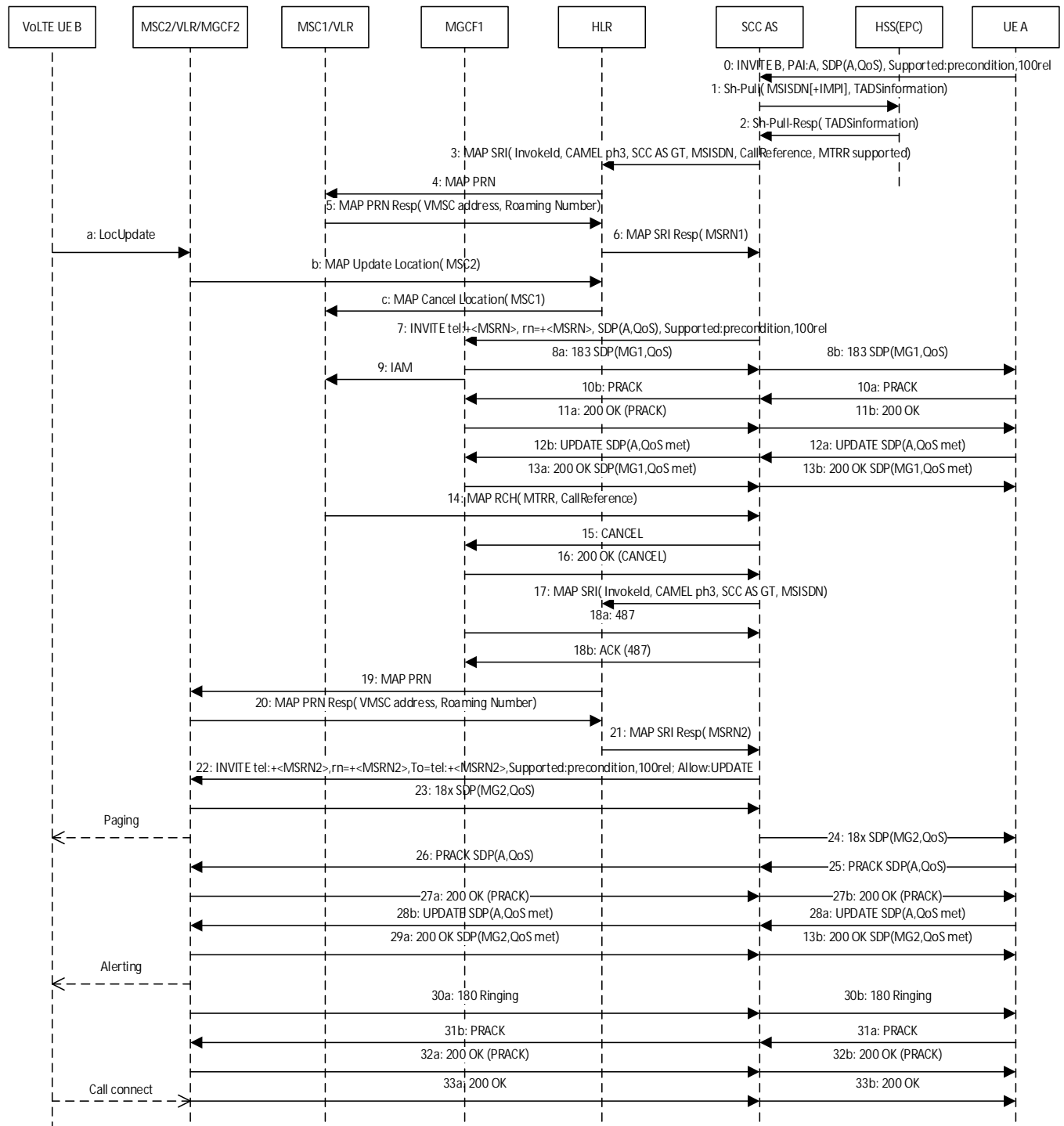


Figure 6 T-ADS MSRN allocation with MAP SRI and MTRR support

24 Terminating INVITE received with SDP including QoS preconditions from originating user A.

25 In this case VoLTE PS registration exist, query for TADSinformation.

26 VoPS is not supported, breakout to CS is selected.



- 27 Breakout policy is set to MSRN, SCC AS deployment mode is without HSS (mtasScchssDeploymentMode=1), a valid license exists, and MTRR is supported in SCC AS: The MAP SRI interface to HLR is selected for obtaining the MSRN, a call reference and MTRR support is indicated.
- 28 In this case MSISDN is a non-CAMEL subscriber, HLR sends MAP PRN to MSC1/VLR for an MSRN.
- 29 MSC1/VLR allocates a temporary MSRN1 and returns to HLR.
- 30 The MSRN1 is returned to SCC AS in a MAP SRI response.

After successful MSRN allocation, a) the mobile moves to a new MSC2, b) the new location is updated in HLR, c) the old location is cancelled in MSC1/VLR.

- 31 Call is setup to MSC1 via MGCF1 with the MSRN1, SDP(A) with precondition on QoS is offered.
- 32 MGCF1 reserves resources in B to A direction and sends 183 provisional response with SDP(MG1,QoS) answer. SCC AS forwards answer to A.
- 33 IAM sent to MSC1.

When the Cancel Location is received by MSC1 as in this case prior to call setup or during paging of the mobile, MSC1 shall instruct the GMSC (SCC AS) to resume terminating call procedure by sending a MAP Resume Call Handling (RCH) message.

- 34 PRACK on 183 received by SCC AS, and forwarded to MGCF1.
- 35 200 OK(PRACK) received by SCC AS, and forwarded to A.
- 36 UPDATE received from A with SDP offer indicating that the preconditions are met, and forwarded to MGCF1.
- 37 Preconditions met at MGCF1 and answers with 200 OK, forwarded by SCC AS to originating side A.
- 38 The MAP RCH with MTRR indication and a call reference is received by SCC AS.
- 39 Following on the MAP RCH, the session to MSC1 is located and the invite is cancelled.
- 40 200 OK on CANCEL.
- 41 Resume call setup in SCC AS by sending a second MAP SRI request to HLR, now without MTRR support indication.
- 42 487 response on INVITE from MGCF1 received by SCC AS, and responded with an ACK.
- 43 HLR knows the updated location of the mobile, and now sends MAP PRN to MSC2/VLR for an MSRN.
- 44 MSC2/VLR allocates a temporary MSRN2 and returns to HLR.
- 45 The MSRN2 is returned to SCC AS in a MAP SRI response.



- 46 Call is setup to MSC2 via MGCF2 with MSRN2, precondition supported but no SDP offer. UE B is paged.
- 47 SCC AS receives 18x response with SDP offer and QoS precondition from MGCF2.
- 48 Early dialog so the SDP offer with QoS precondition is forwarded to A with 18x response in a new early dialog.
- 49 The media resources in A to B direction are reserved at A and answers with SDP answer and preconditions supported.
- 50 PRACK with SDP answer sent to MGCF2.
- 51 MGCF2 reserves media resources in B to A direction and responds with 200 OK on the PRACK.
- 52 UPDATE received from A indicating that the preconditions are met, and forwarded to MGCF2.
- 53 Preconditions met at MGCF2 and answers with 200 OK, forwarded by SCC AS to originating side A.

Paging of VoLTE UE B is successful, and the SDP preconditions are met.

- 54 180 Ringing from B, and forwarded to A.
 - 55 PRACK on the 180 received, and forwarded to B
 - 56 200 OK on PRACK received, and forwarded to A.
- Call is answered by B.
- 57 200 OK to SCC AS, and forwarded to A.

2.5.3 Configuration

Examples of node-level configuration parameters related to the T-ADS feature:

- Administrative state (enable/disable).
- Policy for T-ADS retry with CS breakout when call delivered to PS contact fails or times out.
- Prefix to be added to called party when breakout call to CS with prefix applies.
- Policy for T-ADS CS breakout, either based on configured CSRN prefix or based on MSRN as retrieved from HSS/HLR for the call.

2.5.4 Performance management

Examples of performance counters provided by SCC AS for the feature:

- Current number of VoLTE sessions in progress that are connected via the terminating Circuit Switched (CS) access.



- Current number of VoLTE sessions in progress that are connected via the terminating Packet Switched (PS) access.

2.5.5 Fault management

SCC T-ADS license absent alarm.

2.6 Single Radio Voice Call Continuity (SR-VCC)

2.6.1 Description

The Single Radio Voice Call Continuity (SRVCC) function as specified in 3GPP (ref [5]) allows voice call continuity between IMS over PS access and CS access for calls that are anchored in IMS when the UE is capable of transmitting/receiving on only one of those access networks at a given time.

SRVCC PS to CS access transfer is supported, according to 3GPP release 9,10 and 12 architectures, for active calls as per R9 and R10, for calls in alerting state as per R10 and R12, and for calls in pre-alerting state as per R12. In addition, MSC server assisted Mid-call feature is supported as per R10 for the use cases of single held call, active + additional held call, active + additional incoming call and held call + additional outgoing call transfer.

2.6.2 Example Call Flows

2.6.2.1 Access transfer of call in active state (R9 and R10)

In case of SRVCC there are three different user roles:

- Served UE in PS domain
- Served UE in CS domain
- Remote UE

Served UE in PS domain is a UE that is served by the SCC AS and attached to the PS domain.

Served UE in CS domain is a UE that is served by the SCC AS and attached to the CS domain. The Served UE in the CS domain and the Served UE in the PS domain are the same UEs using different access networks.

Remote UE is the UE in communication with the Served UE (independent of whether the Served UE is attached to PS or CS).

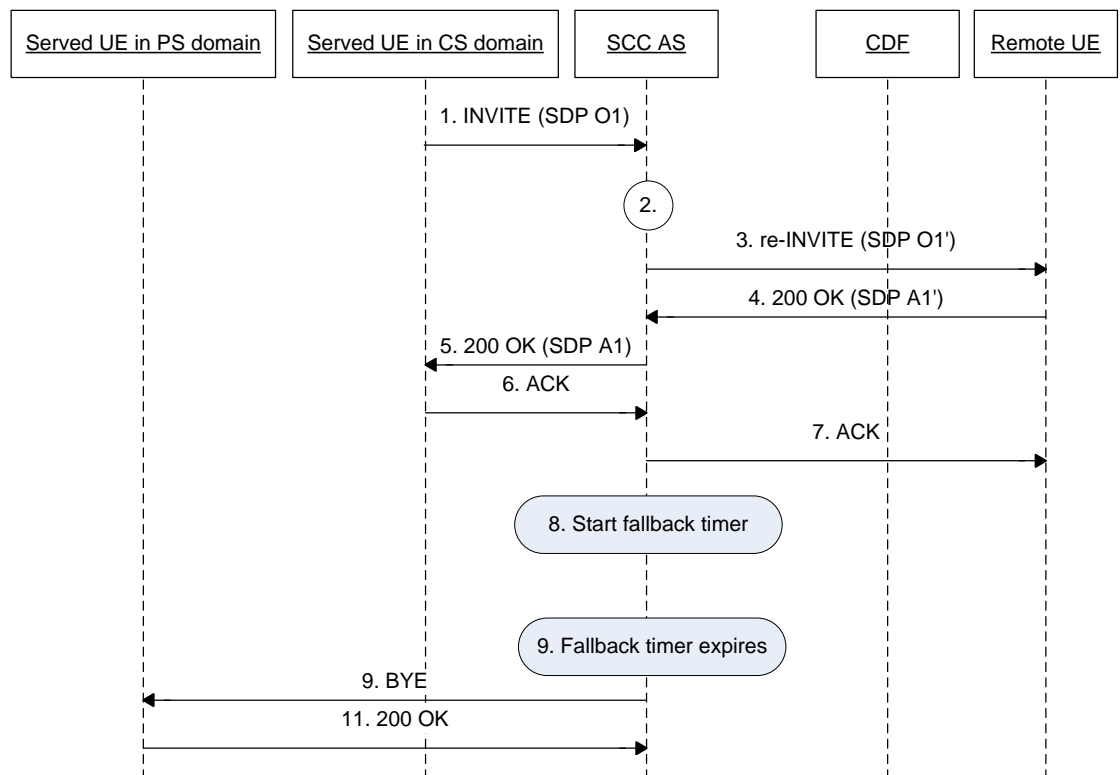


Figure 7 Successful access transfer, SCC AS terminates source access leg

- 1 SCC AS receives an INVITE request from the served UE in CS domain. The Request-URI is set to the configured STN-SR value (mtasSrvccStnSr) in case of SRVCC R9 and is set to the configured STN-SR value (mtasSrvccAtuSti) in case of SRVCC R10. The P-Asserted-Identity header of the request specifies the C-MSISDN value of the Served UE. The request contains an SDP offer describing the parameters of the new media termination, where the session is to be transferred to.
- 2 SCC AS compares the received SDP offer (O1) with the previously negotiated PS media state if SRVCC R10 procedure applies to the transferring contact. If the media states match then there is no need to update the remote UE so the SCC AS answers the INVITE immediately and the steps 3, 4 and 7 will be skipped over.
- 3 If the offered and the old media states do not match or the SRVCC R9 procedure applies to the transferring contact then the SCC AS generates a re-INVITE request. The re-INVITE request contains an SDP offer. The media description of the audio stream to be transferred is updated with the recently received media offer (SDP O1) and the other possible media descriptions are removed by setting the port value to 0. The re-INVITE request is sent towards the remote UE.
- 4 SCC AS receives a 200 OK response from the remote UE.
- 5 The 200 OK response is forwarded to the served UE in CS domain. The forwarded SDP contains only the media description to be transferred.
- 6 SCC AS receives an acknowledgement from the served UE in CS domain.



- 7 SCC AS forwards the acknowledgement towards the remote UE.
- 8 SCC AS starts the fallback timer. The timer is configured by the network operator through the `mtasSrvccFallbackTime` configuration parameter.
- 9 The fallback timer expires.
- 10 SSCC AS terminates the source access leg by sending a BYE request towards the served UE in PS domain.
- 11 SCC AS receives a 200 OK response from the served UE in PS domain.

2.6.2.2 Access transfer for calls in alerting state

A VoLTE UE moves from PS to CS access while the call is in alerting state. This session can be transferred by SCC AS from PS to CS if the UE and network supports the feature (`g.3gpp.srvcc-alerting`) and if operator configuration in SCC AS allows for it.

Similar to 2.6.2.1, there are three different user roles. The served UE in PS domain is registered in IMS. Registration procedures for SCC along with capability advertisements to indicate support for alerting access transfer have been executed, i.e. the handling of STN-SR, ATU-STI, C-MSISDN according to 3GPP release 12.

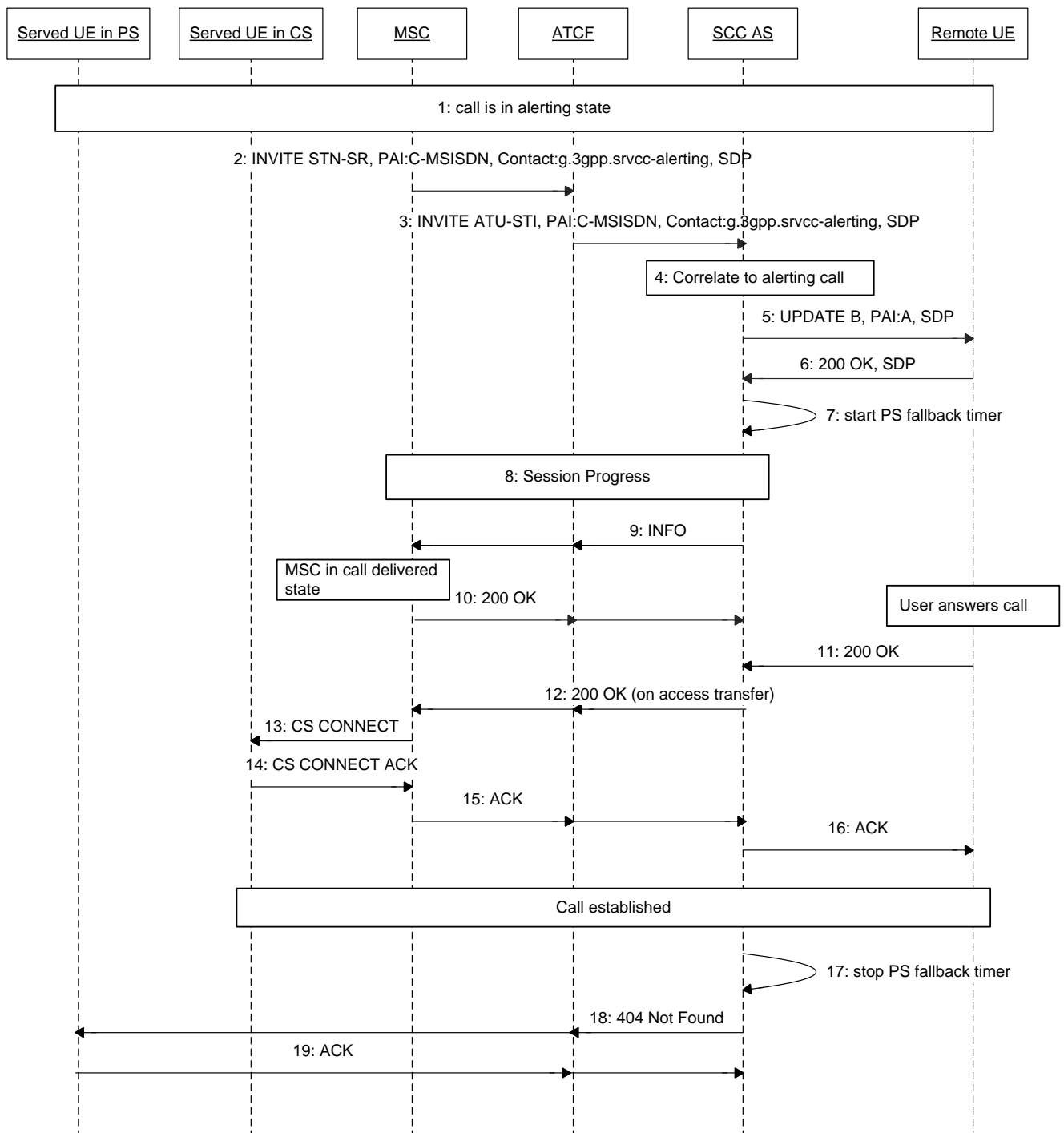


Figure 8: Access transfer for calls in alerting state

1. Served UE in PS originates a call to remote UE and the session is anchored in SCC AS and ATCF. SCC AS stores in session data that the served user supports SRVCC access transfer in alerting state. While the call is in alerting state, served UE moves from PS to CS domain and access transfer procedures are initiated by MSC which is associated with served UE.



SCC AS stores the value of P-Early-Media header from the last 18x response from remote side which creates the last early dialog to SC UE in PS access. SCC AS will choose the last early dialog, where the P-Early-Media header is not inactive for the first audio component in SDP and P-Early-Media does not contain a value of 'gated' at the end of list of values

2. The MSC server initiates transfer INVITE towards IMS by using the STN-SR with
 - P-Asserted-Identity = C-MSISDN
 - SRVCC for calls in alerting state is supported
 - Supported header contains 100rel
 - Recv-Info header contains g.3gpp.state-and-event
 - Accept header contains application/vnd.3gpp.state-and-event-info
3. ATCF shall then provide the proxy role, replace the Request-URI in the received SIP INVITE request due to STN-SR with ATU-STI and forward the INVITE to SCC AS.
4. SRVCC for calls in alerting state is supported by served UE, SCC AS and MSC. SCC AS correlates the received INVITE based on C-MSISDN to the early alerting state between Served UE and remote UE.
5. SCC AS sends an UPDATE with a new SDP offer towards remote UE.
6. Remote UE responses 200 OK with SDP answer on UPDATE.
7. SCC AS starts PS fallback timer (mtasSrvccFallbackTime).
8. SCC AS sends 183 (session progress) to MSC containing the SDP answer as received in 200 OK. MSC responds with PRACK and SCC AS sends 200 OK on PRACK

The value from P-Early-Media header as stored by SCCAS in step 1 will be included in P-Early-Media header of corresponding 183 towards MSC which creates the early dialog towards SC UE in CS access. This will be done only if AT INVITE from MSC contains a P-Early-Media header field with the "supported" parameter and SRVCC R12 functionality is enabled in MTAS. All other 183 responses to MSC will have a value of P-Early-Media header as 'inactive' for the audio component in SDP. Subsequent to this, any value of P-Early-Media header as received in 18x of 2xx responses in any of the early dialogs from remote side UEs, shall all be transparently passed by SCC AS towards MSC

9. For the first received PRACK, SCC AS sends a SIP INFO request to MSC in the dialog created by the access transfer INVITE with application/vnd.3gpp.state-and-event-info/initiator content that the call is in alerting state.
10. MSC acknowledges with 200 OK and enters Call delivered state.
11. Remote UE answers the call.
12. SCC AS sends 200 OK on access transfer to MSC.
13. MSC indicates to the served UE in CS that the far end has accepted the call with CS CONNECT.



14. Served UE in CS acknowledges the CS CONNECT.
15. MSC acknowledges the 200 OK.
16. SCC AS acknowledges the 200 OK towards remote UE, and the call is established successfully.
17. The PS fallback timer is stopped.
18. SCC AS releases the PS access leg by sending SIP 404 (Not Found) response.
19. Served user receives SIP 404 and responds with ACK.

2.6.2.3 Access Transfer of calls in pre-alerting state

Access transfer of calls in pre-alerting state is supported only for originating side according to SRVCC Rel-12 specification.

The Target-Dialog header field is used in INVITE (ATU-STI) request coming from ATCF in order to identify the dialog supporting the PS session to be transferred. If SRVCC for pre-alerting is enabled in SCC AS, and if MSC, ATCF and served UE supports SRVCC for calls in pre-alerting state, the transfer procedures for calls in pre-alerting state applies.

It is assumed that the media has been negotiated in at least one reliable 183 response prior to the access transfer and an early dialog is in place. The session contains one or multiple early dialogs with an active speech media component. Note that 180 Ringing should not have come from any remote UE for pre-alerting access transfer procedure to kick in. For originating call in pre-alerting state, there may be multiple early dialogs due to forking on terminating side.

The remote legs are updated with the SDP from the access transfer INVITE, and the returned SDP answer is then forwarded to MSC in SIP 183. The call direction and state info, which is originating pre-alerting, is sent to MSC in the SIP INFO message. The access transfer is completed when the call is answered, the access transfer INVITE will then be responded with 200 OK including the SDP answer from remote side.

During the access transfer a charging session is created for both the source access leg and the target access leg. For originating pre-alerting, the charging session for the source access leg is indicated with an ACR[event].

2.6.2.3.1 Capability advertisement as per 3GPP release 12 version for calls in pre alerting state

Pre-requisites (other than capability advertisement during registration) by various nodes for access transfer of pre-alerting calls are:

- a Support for access transfer for pre-alerting call is available for the SC UE at the originating side



- b Support for access transfer of alerting calls is a pre-requisite for support of pre-alerting calls.
- c 180 Ringing should not have come from any remote UE yet

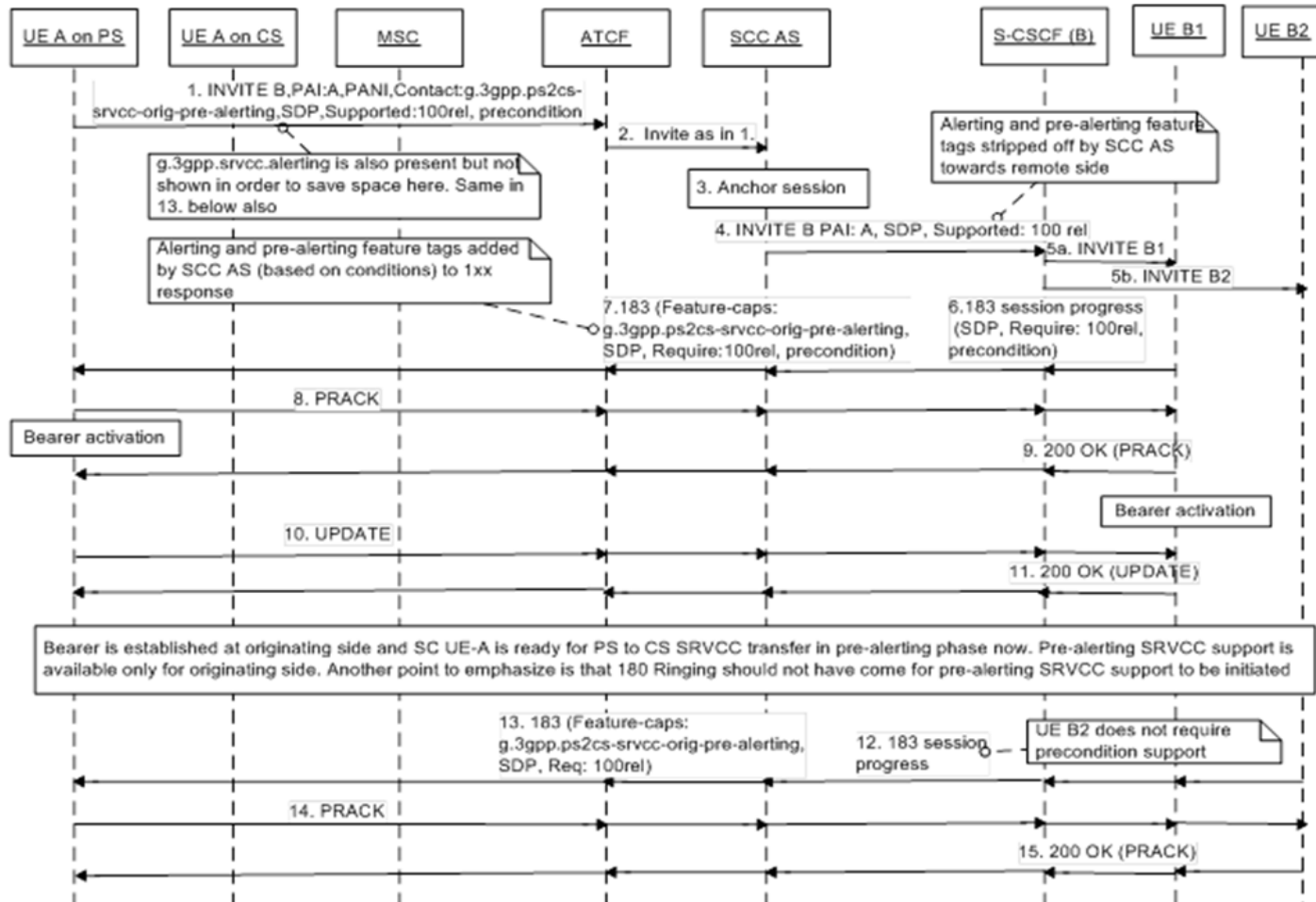


Figure 9 Capability advertisement before access transfer call in pre-alerting state

Explanation of the call flow is as below:

- 1 SC UE A on LTE access originates an IMS call with a speech media component in SDP. Support for SRVCC in pre-alerting (and alerting) states is indicated with Contact header feature tags g.3gpp.ps2cs-srvcc-orig-pre-alerting and g.3gpp.srvcc-alerting. Reliable response support is indicated with 100rel in Supported header. Precondition is also supported by UE A
- 2 Session is anchored in ATCF. INVITE is forwarded to SCC AS by replacing STN-SR with ATU-STI in Request-URI. ATU-STI was received by ATCF earlier from SCC AS in the MESSAGE associated with REGISTER sequence
- 3 Session is anchored in SC AS
- 4 INVITE is sent to terminating side with the feature tags g.3gpp.srvcc-alerting and g.3gpp.ps2cs-srvcc-orig-pre-alerting stripped from Contact header



- 5 S-CSCF at B side forks and INVITE is forwarded to terminating UE B1 and B2
- 6 UE B1 sending 183 session progress to UE A. 100rel and precondition option tags are required
- 7 SCC AS adds Feature-tags in Feature-Caps header towards SC UE.

This is to advertise support for pre-alerting and alerting call SRVCC in the 183 to SC UE. These feature tags are added only if all the conditions below are met:

- a Access transfer of pre-alerting and alerting calls is enabled in SCC AS
- b ATCF has indicated during the earlier UE registration over PS that all MSC servers in its serving network support access transfer of pre-alerting and alerting calls
- c feature tags for alerting and pre-alerting support is received in the initial INVITE over PS from SC UE
- d 180 (Ringing) response has not yet been received yet from any of the remote UEs

Note: If all conditions are satisfied, these feature tags are to be added by SCC AS in all reliable 1xx responses (183,180 etc. including all forking case) as well as to the 2xx final response (successful final response).

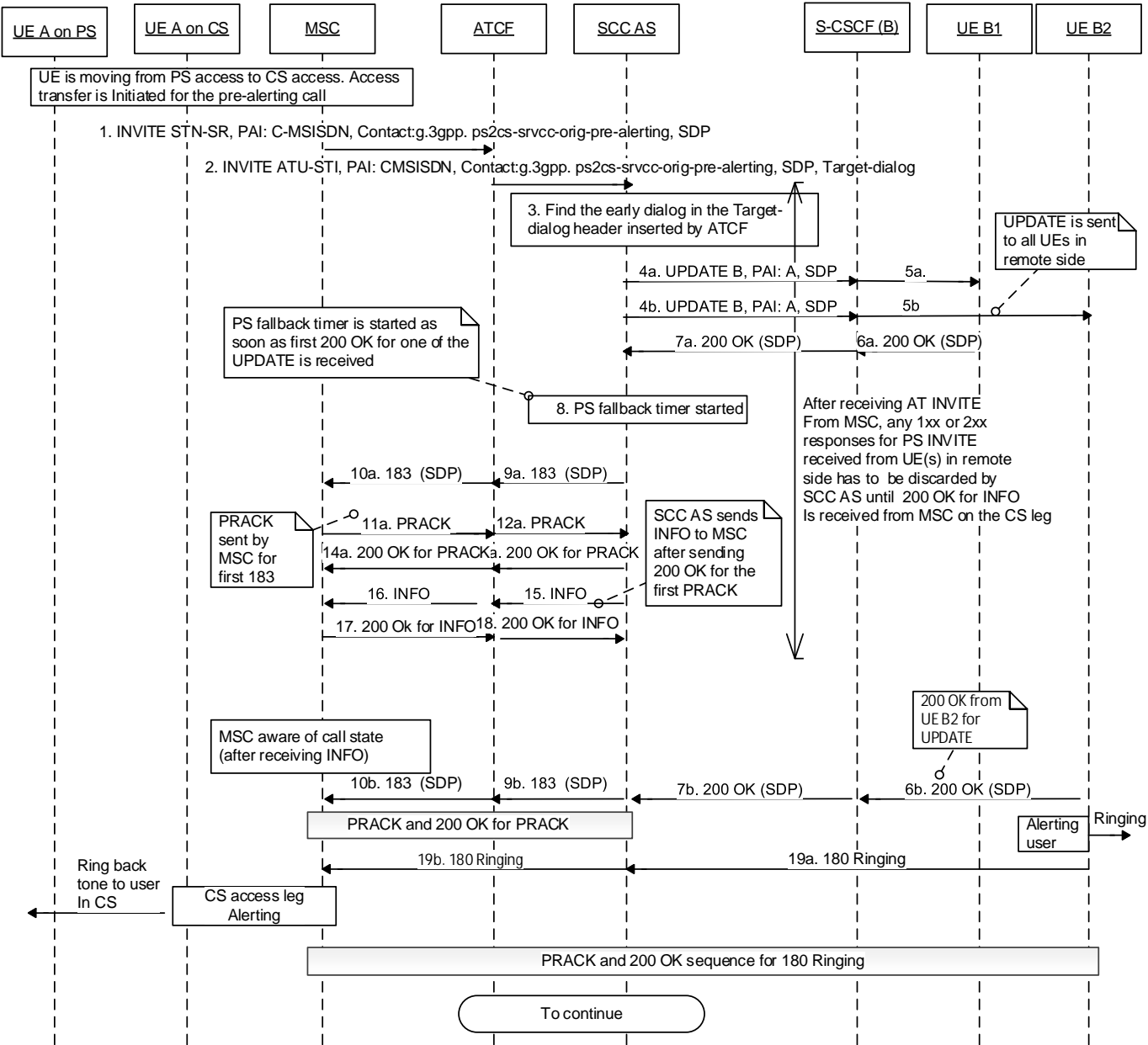
Apart from above feature-tags, the value of last non inactive P-Early-Media header in a 18x response is stored by SCC AS to be used later in the 183 answer to MSC on access transfer INVITE. All other P-Early-Media headers are set "inactive". Since value of P-Early-Media header is stored from the last early dialog this will play the early media from the last UE which handles the call, in case of call diversion scenarios.

- 8 On receiving 183 from B1, UE A sends PRACK response. Bearer activation begins at originating side.
- 9 B1 sends 200 OK for PRACK. On reception of this, bearer activation begins at terminating side
- 10 UE A gets intimated from radio network that bearer is activated now. It sends out UPDATE message which includes QoS reservation confirmation notification

SC UE A is ready for PS to CS SRVCC now as the bearer is established at its end

- 11 UE B1 replies with 200 OK for UPDATE.
- 12 UE B2 will send 183 session progress to UE-A. Here, 183 response by UE B2 is shown after UPDATE-PRACK-200 OK sequence of UE B1 for easy reading
- 13 SCC-AS adds alerting and pre-alerting feature tags to the 183 response.
- 14 PRACK for 183
- 15 and 200 OK for PRACK follows

2.6.2.3.2 SCC AS handling of Access Transfer INVITE from MSC for call in pre-alerting state.



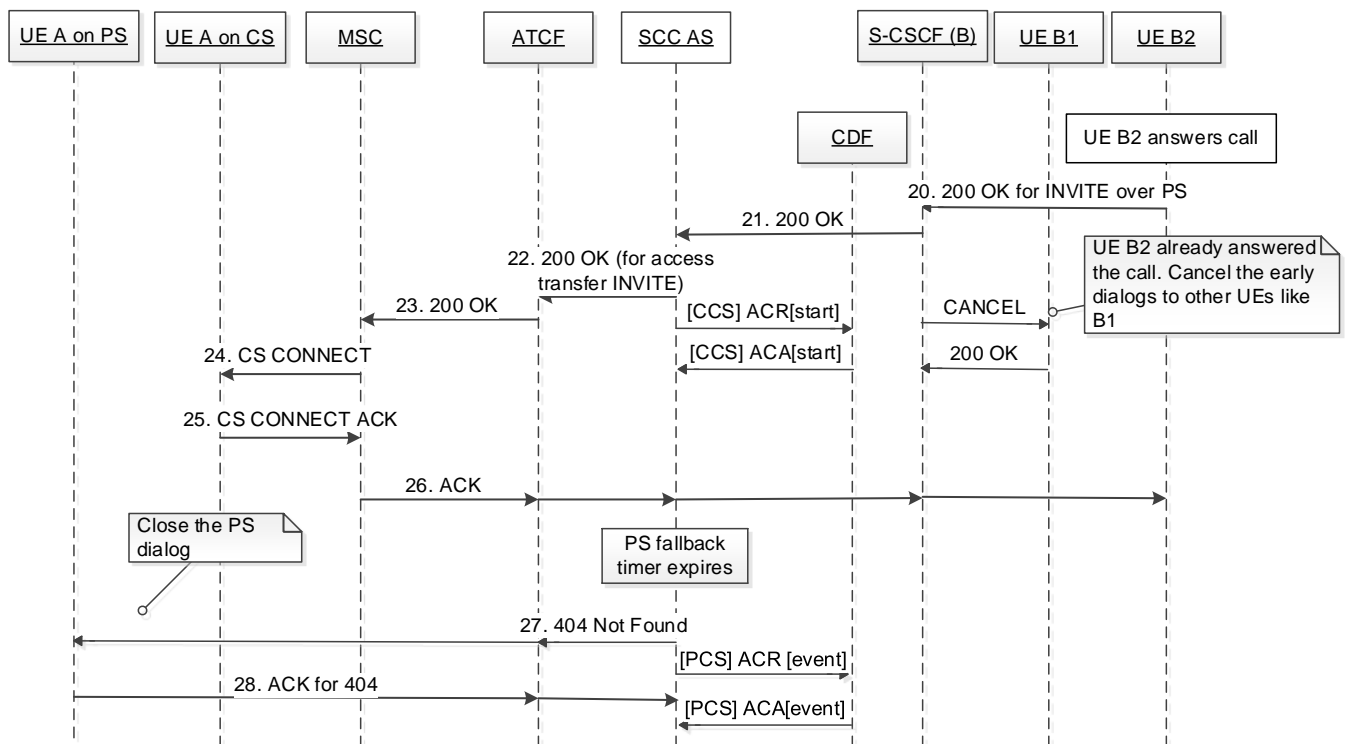


Figure 10 Handling of Access Transfer INVITE from MSC for call in pre-alerting state

- 2 UE moves from PS to CS. MSC send the AT INVITE to ATCF by having Request-URI as STN-SR. STN-SR was received from MME as part of SRVCC trigger. Also,
 - o P-Asserted-Identity = C-MSISDN
 - o SRVCC for calls in pre-alerting and alerting states are supported (g.3gpp.ps2cs-srvcc-orig-pre-alerting and g.3gpp.srvcc-alerting)
 - o Supported header contains 100rel
 - o Recv-Info header contains g.3gpp.state-and-event
 - o Accept header contains application/vnd.3gpp.state-and-event-info
- 2 Transferable session is already in pre-alerting state. ATCF inserts a Target-Dialog header which points to the session that is to be access transferred. Next, it forwards the AT INVITE to SCC AS by overwriting Request-URI to ATU-STI. ATCF received this ATU-STI earlier via MESSAGE from SCC AS as part of UEs REGISTER over PS
- 3 SCC AS finds the session based on Target-Dialog header. Multiple early dialogs may be related to the originating session but Target-Dialog will help uniquely identify the session. If the session cannot be found based on Target-Dialog header, SCC AS would reject the AT INVITE with 404 (Not Found) response



- 4 If pre-alerting and alerting feature tags are not present in the Contact header then SCC AS will reject the AT INVITE with 480 (Temporarily Unavailable). This would get rejected by ATCF before SCC AS anyway
- 5 SIP UPDATE with SDP offer received by B1/B2.
- 6 200 OK for UPDATE with SDP answer from UE
- 7 200 OK received by SCC AS
- 8 The PS fallback timer will be started on the first 200 OK on an UPDATE
- 9 SCC AS sends a 183 (Session Progress) containing the SDP answer as received in 200 OK of UPDATE from remote side and an empty Recv-Info header.

Subsequent to this, any P-Early-Media header as previously stored when the early sessions were anchored in SCC AS, shall all be passed by SCC AS towards MSC.

- 10 183 received by MSC.
- 11 MSC responds with a PRACK for 183
- 12 PRACK received by SCC AS
- 13 200 OK on PRACK sent from SCC AS.
- 14 200 OK on PRACK received by MSC.

For the first received PRACK, SCC AS sends a SIP INFO request to MSC with application/vnd.3gpp.state-and-event-info content to indicate that the call direction and state is originating, pre-alerting. The INFO message informs MSC that the state and direction of the access transferred call is in originating pre-alerting state.

- 15
- 16 SIP INFO received by MSC.
- 17 MSC acknowledges with 200 OK
- 18 200 OK received by SCC AS.
- 19 UE B2 starts ringing. It sends 180 Ringing is send in backward direction to SCC AS. UE A plays ring back tone. PRACK and 200 OK sequence for 180 Ringing follows
- 20 UE B2 answers the call. It sends a 200 OK to the initial INVITE over PS to UE A
- 21 S-CSCF proxies the response in the backward direction to SCC AS. In the case of forking all other outstanding requests will be cancelled, in this case to B1.
- 22 SCC AS sends 200 OK on access transfer to MSC.
- 23 200 OK on access transfer received by MSC.
- 24 MSC indicates to the UE A that the far end has accepted the call with a CS CONNECT.
- 25 UE A acknowledges the CS CONNECT to MSC
- 26 MSC acknowledges 200 OK with SIP ACK request.

The PS fallback timer expires.

- 27 SCC AS releases the source access leg by sending a SIP 404 (Not Found) response to ATCF.
- 28 ACK sent by UE-A for 404



2.6.3 Mid-call transfer of active call and additional held call

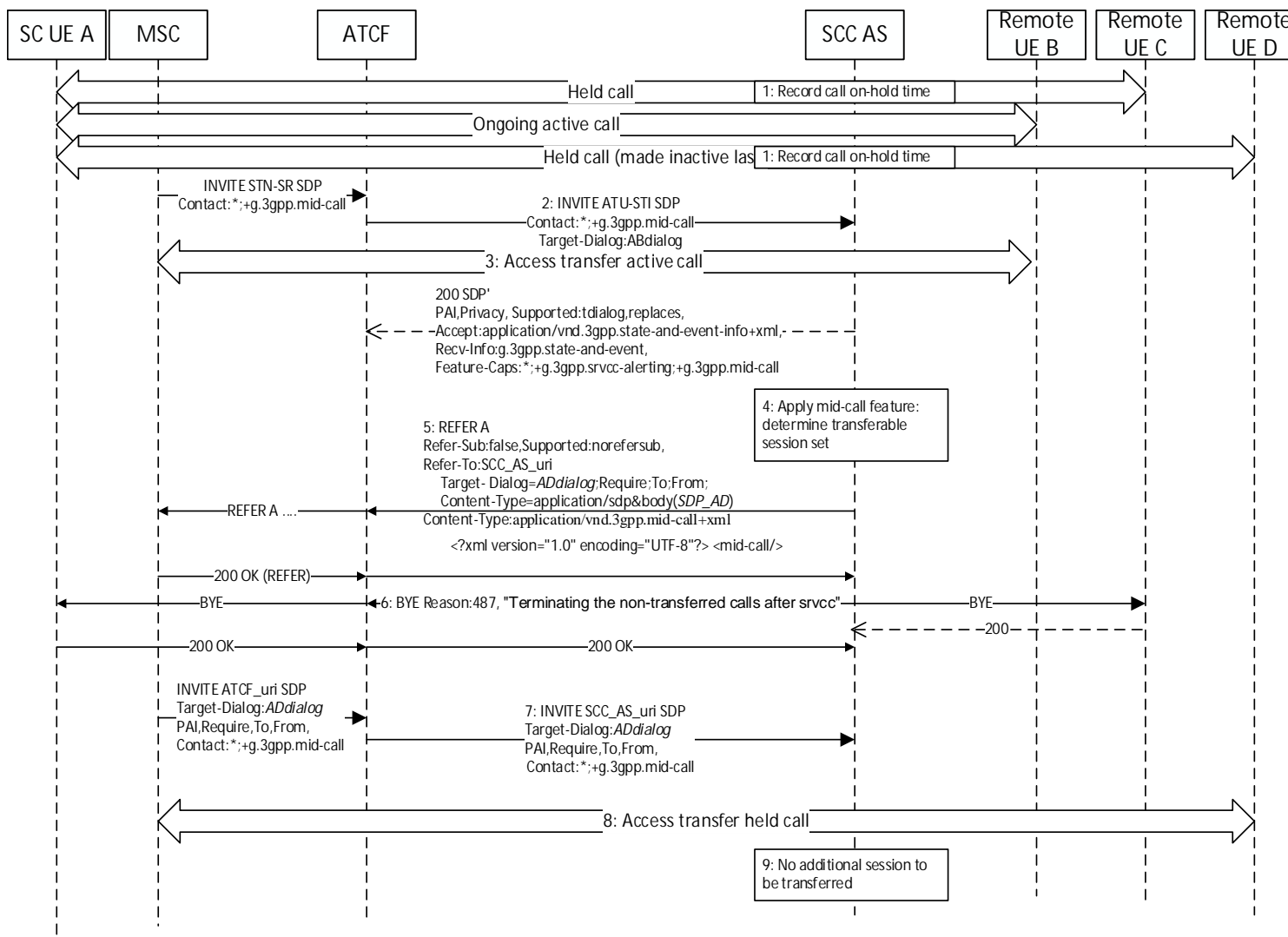


Figure 11 Access transfer active call and additional held call

The use case shows access transfer of an active call with MSC assisted mid-call feature support which in this case means that an additional held call is transferred, initiated by SCC AS with a REFER sent to MSC. Target session for transfer is given by Target-Dialog in the AT INVITE.

- 1 A Held call must be recorded for when it was made in-active in order for SRVCC mid-call to select the additional call for transfer that was made inactive most recently.
- 2 On reception of AT INVITE, SCC AS applies MSC server assisted SRVCC mid-call feature based on
 - a) AT INVITE Contact includes the g.3gpp.mid-call media feature tag, and
 - b) Mid-call feature applies to the targeted session, i.e. all MSCs in serving network supports Mid-call feature as indicated at registration, SC UE



indicates support for Mid-call in session signaling and Mid-call support is enabled in SCC AS.

- 3 The active session given by Target-Dialog is transferred and 200 OK response to AT INVITE including Feature-Caps with g.3gpp.mid-call capability sent to ATCF.
- 4 SRVCC Mid-call feature applies, determine transferable session set; when first session transferred is a session with active speech media, select an additional session 1) as a session on hold or 2) a session in alerting state, in that priority order. In this case the transferable session set indicates to transfer an additional held call. Select the held call that was put on-hold most recently, i.e. A-D session.
- 5 Transfer of the additional held call between A and D is initiated by sending a REFER to MSC in the dialog created by AT INVITE, use Contact URI from AT INVITE as Request URI. The Refer-To header is populated with details of the A to D session as described in 3gpp 24.237, section 12.3.2.3. The Target-Dialog URI header field is populated with the id of the dialog towards SC UE.
- 6 Release any superfluous session that was not in the transferable session set, in this case the Held call between A and C.
- 7 AT INVITE for the additional session is received by SCC AS. SRVCC mid-call feature applies as described in step 2.
- 8 The Target-Id identifies the held session A to D, mid-call feature applies and the session is transferred.
- 9 No additional sessions to be transferred as this was an additional session transfer (Request URI in AT INVITE equal to mtasSrvccSccAsUri).

2.6.4 Configuration

Examples of node-level configuration parameters related to the SR-VCC feature:

- Administrative state (enable/disable)
- STN-SR assigned to this SCC AS for SRVCC R9 access transfer.
- ATU-STI assigned to this SCC AS for SRVCC R10 architecture.
- SCC AS URI assigned to this SCC AS for SRVCC Mid-call transfer.
- Enable/disable SRVCC support for calls in alerting state in MTAS.
- Enable/disable SRVCC support as per 3GPP release 12.
- Enable/disable SRVCC support for calls in pre-alerting state in MTAS.



2.6.5 Performance Management

Example of performance counters provided by SCC AS for the feature:

- The number of attempts to initiate access transfer using Single Radio Voice Call Continuity.
- The number of unsuccessful (due to node external error) access transfers attempts using Single Radio Voice Call Continuity.

2.6.6 Fault Management

SCC SRVCC license absent alarm.

2.7 Network Provided Location Information (NPLI)

2.7.1 Description

Same NPLI feature as in MMTel AS can be enabled in SCC AS. See section 2.33 in TPD for MMTel AS[8] for further description.

2.7.2 Example Call Flows

2.7.2.1 Provide NPLI for terminating call to CS domain

Preconditions:

Terminating NPLI retrieval is enabled in SCC AS.

When 180/200 response from CS access domain does not contain network PANI or CGI in network PANI is not valid, SCC AS retrieves network provided location information from HSS for the served ICS user on CS access and insert in a PANI header of the 180/200 response.

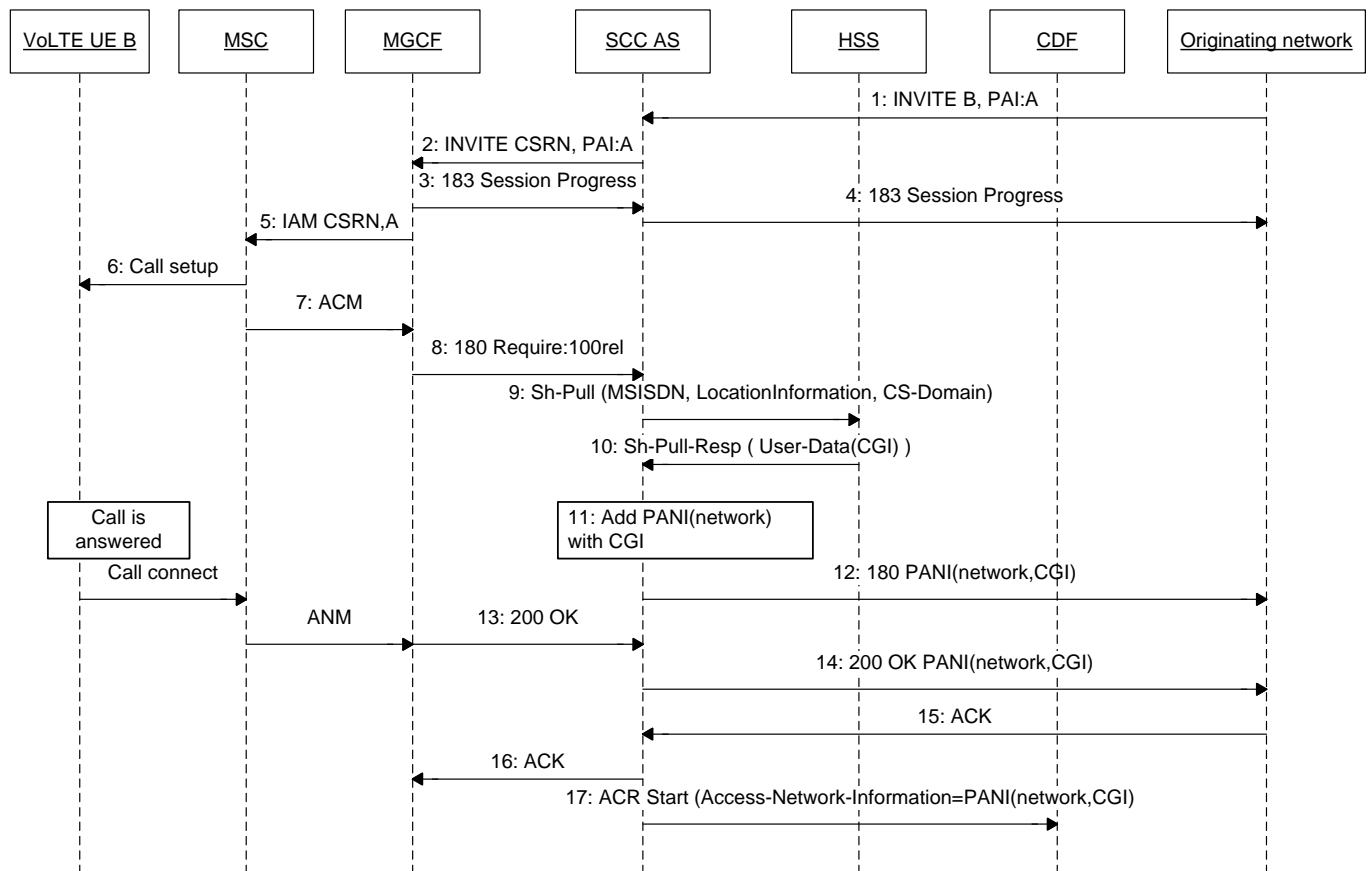


Figure 12 NPLI for terminating CS access

- 1 Terminating INVITE received by SCC AS.
- 2 TADS is executed. The call is broken out to CS domain for example based on CSRN as obtained from HSS.
- 3 Call is routed to MGCF, which might immediately respond with 183 Session Progress.
- 4 183 from MGCF to SCC AS, no PANI is included and SCC AS does not change that, as 183 is not applicable for NPLI retrieval procedure.
- 5 MGCF routes the call over ISUP to the serving MSC given by the CSRN.
- 6 Call delivered to UE B on CS
- 7 ISUP ACM from MSC to MGCF.
- 8 180 from MGCF to SCC AS, no PANI is included.



SCC AS is configured to retrieve NPLI for terminating case when there is no network provided PANI in the 180/200 response or the CGI in network PANI is not valid. In this case there is no location information stored in the session and no PANI in 180, so NPLI will be fetched from HSS.

- 9 SCC AS sends Sh-Pull to HSS for Location Information for UE B (MSISDN) on CS access domain. The access domain is set to 'CS' in this case based on that the 180/200 response is on a CS breakout.
- 10 Successful response from HSS with Location Information including CGI. The location information is stored with the session.
- 11 Create a new PANI header with access-type=3GPP-GERAN and access-info=CGI and with network-provided token. The PANI header is stored with charging session if charging enabled.
- 12 Send 180 to originating network with inserting a new network PANI.
- 13 200 OK without PANI header from UE B.
- 14 The PANI header stored with session is added, and the 200 OK is then sent to the originating network.
- 15 ACK received from originating network.
- 16 ACK sent to terminating network.
- 17 ACR start with the network provided PANI (Access-Network-Information AVP) is sent to the CDF.

2.8 Geographical Redundancy

2.8.1 Description

SCC AS supports geographical redundancy meaning that the IMS system can be configured with two SCC ASs. If the primary SCC AS is restarting or unavailable the secondary SCC AS can be used in the system. The UE contact data is obtained from extended third party registration (see 2.3 or by means of Event Package for Registrations and stored internally in SCC AS.

2.8.2 Example Call Flows

2.8.2.1 Registration based on INVITE for geo-redundancy

This scenario occurs if the SCC AS has not received any registration prior to session setup, or if the IRS and contact database in SCC AS is lost, for example after a node restart. SCC AS retrieves the registration information from the S-CSCF with a one-time subscription for the "reg" event.

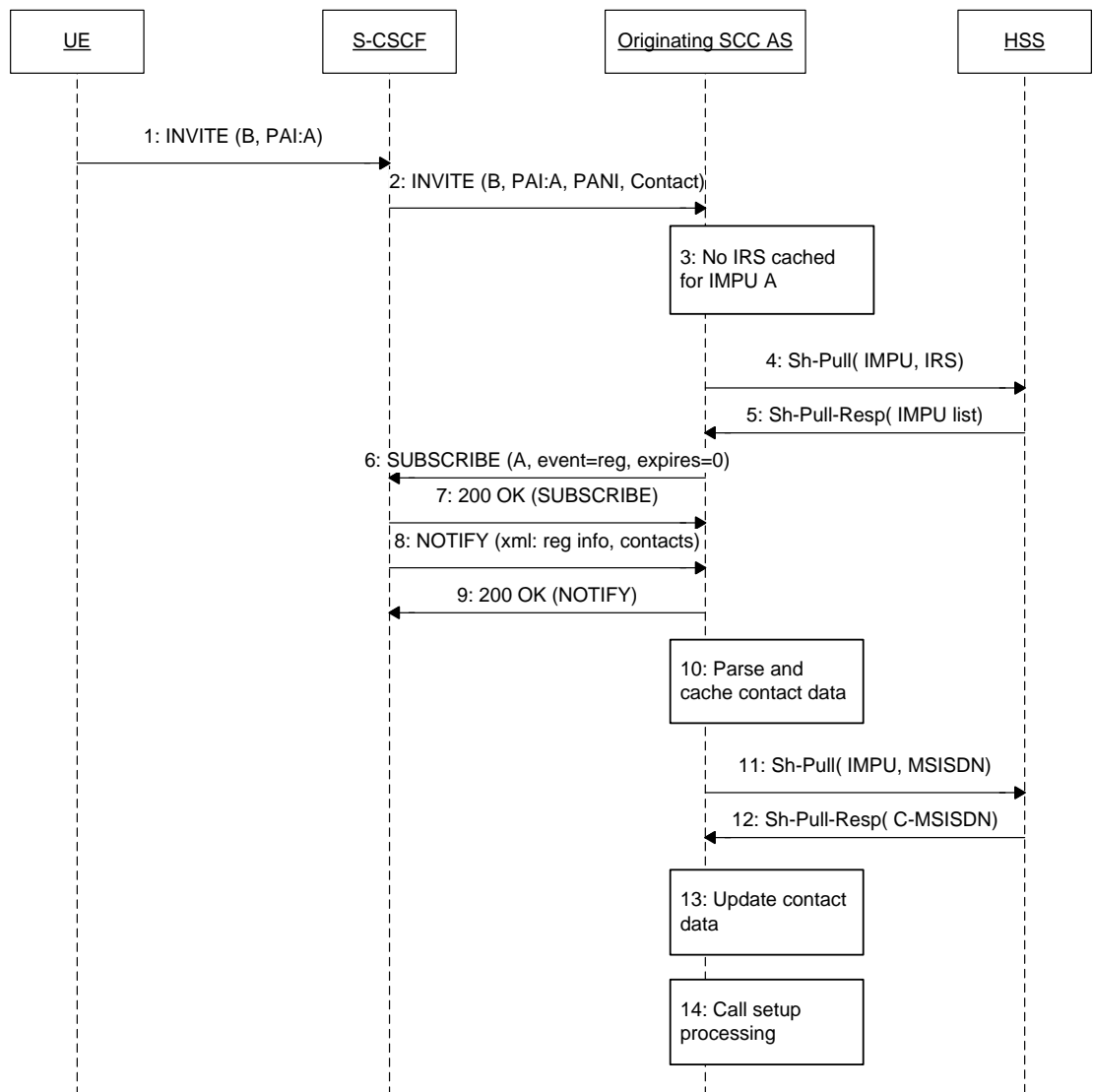


Figure 13 UE registers with INVITE, no ATCF info in S-CSCF

- 1 The UE sends an INVITE to establish a call.
- 2 Originating registered or unregistered iFC triggers S-CSCF to send INVITE to SCC AS.
- 3 The SCC AS detects that no IRS or Contacts data cached for the IMPU.
- 4 An Sh-Pull Request is sent to the HSS to retrieve the IMPUs of the IRS.
- 5 The HSS replies with a successful Sh-Pull response. The response contains the IMPUs of the IRS. The first IMPU is the default IMPU. The mapping of each IMPU to the default IMPU is stored.

In the originating registered case SCC AS subscribes one-time for the “reg” event package and gets information on registered contacts from S-CSCF in NOTIFY (steps 6 to 9):



- 6 SCC AS sends a SIP SUBSCRIBE (Event: reg, Expires: 0, Accept: application/reginfo+xml) request to the S-CSCF.
- 7 S-CSCF accepts the subscription with a 200 OK.
- 8 S-CSCF sends SIP NOTIFY (Content-Type: application/reginfo+xml, Subscription-State: terminated) request to MTAS, containing the contact data..
- 9 SCC AS sends 200 OK in response to the NOTIFY.
- 10 In the originating registered case the following contact information is parsed from the NOTIFY and cached in SCC AS:

- Contact address
- sip.instance Contact header feature tag
- g.3gpp.accesstype Contact header feature tag
- g.3gpp.ics Contact header feature tag

In the originating unregistered case the following information is retrieved from the Contact header of the INVITE and stored in MTAS:

- Contact address
- sip.instance Contact header feature tag
- g.3gpp.ics Contact header feature tag

From this data the UE access domain and terminal type can be derived:

- UE access domain is CS if g.3gpp.ics="server" or PANI header indicates "3GPP-GERAN" access, else PS.
- UE terminal type is "mobile/VoLTE" if PANI header indicates 3GPP-UTRAN, 3GPP-E-UTRAN or 3GPP-GERAN access or if at least one Contact header feature tag matches any of the conditions configured in the mtasSubsDataMobileClassification CM attribute, else "fixed".

- 11 For the first session setup for the IRS, an Sh-Pull Request is sent to the HSS to retrieve C-MSISDN of the IRS. The basic MSISDN is used for C-MSISDN so Data-Reference=MSISDN is used in the request.
- 12 If there is a VoLTE UE in the IRS with SR-VCC capability the HSS replies with the C-MSISDN, else the result set is empty.
- 13 The C-MSISDN is stored with the cached contact.
- 14 The call setup is further processed.

In this scenario no ATCF information was available in the registration information, i.e. SRVCC according to 3GPP R9.



In case of ATCF information in the registration information, 3GPP R10 architecture applies with additional sequence for STN-SR handling and MESSAGE to S-CSCF.

2.9 Service Interaction

2.9.1 Subscriber Data and SR-VCC

During SIP registration procedures the SRVCC service stores the C-MSISDN and the STN-SR values of the served UE. The C-MSISDN value is used during access transfer for identifying the session to be transferred. The HSS is updated with the new STN-SR value if it has changed during the registration. The HSS populates the new C-MSISDN \leftrightarrow STN-SR allocation in turn.

2.9.2 Subscriber Data and T-ADS

During SIP registration procedures Subscriber Data service stores contact data that is used by the T-ADS service to terminate a call.

2.9.3 SR-VCC and T-ADS

After successful access transfer the access type of the served UE is changed from PS to CS. If Terminating Access Domain Selection (T-ADS) procedures are applied after successful access transfer T-ADS service is aware of the new access type.

2.9.4 FCD and T-ADS

In Fixed Mobile Convergence (FMC) scenarios, the MMTel FCD service can be used to distribute a call to the served ICS user's devices by adding a feature tag +sip.instance and mobility selector to the Accept-Contact header for each terminating call leg.

In this case the T-ADS service in SCC AS can distinguish a call targeted to a fixed device and not apply T-ADS on such a call, just pass it on to S-CSCF.

2.9.5 SDS and NPLI

For O-SDS, location information is received in the CAP InitialDP request and then used to populate PANI in the originating SIP INVITE. This means that originating NPLI will not be retrieved if enabled if the location information from CAP is valid.



3 Acronyms and Abbreviations

3GPP	3rd Generation Partnership Project
AS	Application Server
ATCF	Access Transfer Control Function
ATGW	Access Transfer Gateway
ATU-STI	Access Transfer Update Session Transfer Identification
BGCF	Border Gateway Control Function
C-MSISDN	Correlating MSISDN
CAMEL	Customized Application for Mobile Enhanced Logic
CAP	CAMEL Application Protocol
CGI	Cell Global Identification
CgPN	Calling Party Number
CM	Configuration Management
CS	Circuit Switched
CSI	CAMEL Subscription Information
CSRN	CS domain Routing Number
ETSI	European Telecommunication Standards Institute
E-UTRAN	Evolved UTRAN
GERAN	GSM EDGE Radio Access Network
GMSC	Gateway MSC
GPRS	General Packet Radio Service
GSM	Global System for Mobile Communications
gsmSCF	GSM Service Control Function
gsmSSF	GSM Service Switching Function
HLR	Home Location Register
HPLMN	Home PLMN
HSS	Home Subscriber Server
I-CSCF	Interrogating CSCF



ICS	IMS Centralized Services
IETF	Internet Engineering Task Force
iFC	initial Filter Criteria
IMPI	IP Multimedia Private Identity
IMPU	IP Multimedia Public Identity
IMRN	IP Multimedia Routing Number
IMS	IP Multimedia Subsystem
IMSI	International Mobile Subscriber Identity
IRS	Implicit Registration Set
ISC	IP multimedia Service Control
ISDN	Integrated Service Digital Network
ISUP	ISDN User Part
LAI	Location Area Identification
LTE	Long Term Evolution
MAP	Mobile Application Part
MCC	Mobile Country Code
MGCF	Media Gateway Control Function
MME	Mobility Management Entity
MMTel	Multi-Media Telephony
MNC	Mobile Network Code
MO	Mobile Originating call
MSC	Mobile Switching Centre
MSISDN	Mobile Subscriber Integrated Service Digital Network
MSRN	Mobile Station Roaming Number
MT	Mobile Terminating call
MTAS	Multimedia Telephony Application Server
MTRR	Mobile Terminating Roaming Retry
NPLI	Network Provided Location Information



O-SDS	Originating SDS
P-CSCF	Proxy CSCF
PAI	P-Asserted-Identity
PANI	P-Access-Network-Information
PLMN	Public Land Mobile Network
PS	Packet Switched
PSTN	Public Switched Telephony Network
RCH	Resume Call Handling
QoS	Quality of Service
RFC	Request for Comment
S-CSCF	Serving-CSCF
SCP	Service Control Point
SDP	Session Description Protocol
SDS	Service Domain Selection
SGSN	Serving GPRS Support Node
SIP	Session Initiation Protocol
SPT	Service Point Trigger
SCC	Service Centralization and Continuity
SDS	Service Domain Selection
SR-VCC	Single Radio Voice Call Continuity
STN-SR	Session Transfer Number for SRVCC
T-ADS	Terminating Access Domain Selection
T-CSI	Terminating CAMEL Subscription Information
T-SDS	Terminating SDS
TCP	Transport Control Protocol
UA	User Agent
UE	User Equipment
URI	Uniform Resource Identifier



UTRAN	UMTS Terrestrial Radio Access Network
UMTS	Universal Mobile Telecommunications System
VLR	Visited Location Register
VMSC	Visited MSC
VoLTE	Voice over LTE
VoPS	Voice over PS
VPLMN	Visited PLMN
WiFi	Wireless Fidelity
XML	eXtensible Markup Language



4 Reference Documents

- [1] MTAS 16A Feature Description at lighthouse.lmra.ericsson.se/
- [2] GSMA IR.92 – IMS Profile for Voice and SMS
- [3] GSMA IR.94 – IMS Profile for Conversational Video Service
- [4] 3GPP TS 24.292 – IP Multimedia (IM) Core Network (CN) subsystem Centralized Services (ICS); Stage 3
- [5] 3GPP TS 24.237 - IP Multimedia (IM) Core Network (CN) subsystem IP Multimedia Subsystem (IMS) Service Continuity; Stage 3
- [6] 3GPP TS 09.78 v7.1.0, Customized Applications for Mobile network Enhanced Logic (CAMEL); CAMEL Application Part (CAP) specification
- [7] MTAS Technical Product Description ---- Common Features
- [8] MTAS Technical Product Description ---- MMTel AS