IMS Service Centralization and Continuity Guidelines

Version 4.0

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1 Introduction

1.1 Overview

The 3rd Generation Partnership Project (3GPP) has specified the solution for centralization of services in the IP Multimedia Subsystem (IMS) and of IMS based service continuity in Release 8 onwards. The user shall receive services in a consistent manner when the user accesses IMS either via the Circuit Switched (CS) or the Packet Switched (PS) domain. Service continuity is supported between CS and PS domains.

1.2 Scope

This document provides guidelines for the centralization of IMS services and IMS based service continuity for single radio devices [3] by listing a number of Evolved Packet Core, IMS core, and User Equipment (UE) features on top of the features defined in [11]. The defined guidelines are compliant with 3GPP specifications. Guidelines are provided for the interface between UE and network and for the network architecture. The centralization of IMS services is focused on the network based solution; the UE based IMS Centralized Services (ICS) solution is out of scope of this document and not recommended. IMS based service continuity is focused on Single Radio Voice Call Continuity (SRVCC) from E-UTRAN (Evolved Universal Terrestrial Radio Access Network) to GERAN/UTRAN and from HSPA (High Speed Packet Access) to GERAN/UTRAN; other service continuity scenarios are out of scope.

UE impacts of each feature are described in a separate subsection where applicable.

1.3 Definition of Terms

Term	Description
3GPP	3rd Generation Partnership Project
ATCF	Access Transfer Control Function
ATGW	Access Transfer Gateway
BGCF	Breakout Gateway Control Function
CN	Core Network
CS	Circuit Switched
CSCF	Call Session Control Function
DTM	Dual Transfer Mode
eNB	eNodeB
EPS	Evolved Packet System
EATF	Emergency Access Transfer Function
EDGE	Enhanced Data rates for GSM Evolution
E-CSCF	Emergency CSCF
E-UTRAN	Evolved Universal Terrestrial Radio Access Network
GERAN	GSM/EDGE Radio Access Network
GGSN	Gateway GPRS Support Node
GSM	Global System for Mobile communications
HSPA	High Speed Packet Access
HSS	Home Subscriber Server
ICS	IMS Centralized Services
iFC	Initial Filter Criteria
IMRN	IMS Routing Number
IMS	IP Multimedia Subsystem
IMS AGW	IMS Access Gateway
LTE	Long Term Evolution
MGCF	Media Gateway Control Function
MGW	Media Gateway

MMTel	Multimedia Telephony
MS	Mobile Station
MSC	Mobile Switching Centre
MS-ISDN	Mobile Subscriber ISDN Number
PCC	Policy and Charging Control
PCRF	Policy and Charging Rules Function
P-CSCF	Proxy – CSCF
PS	Packet Switched
QCI	Quality of Service Class Indicator
RAT	Radio Access Technology
RAU	Routing Area Update
RTP	Real Time Protocol
SCC AS	Service Centralization and Continuity Application Server
S-CSCF	Serving CSCF
SGSN	Serving GPRS Support Node
SIP	Session Initiation Protocol
SRVCC	Single Radio Voice Call Continuity
TAS	Telephony Application Server
TAU	Tracking Area Update
TrGW	Transition Gateway
UE	User Equipment
UTRAN	Universal Terrestrial Radio Access Network

1.4 Document Cross-References

Ref	Document Number	Title
1	3GPP TS 23.060	General Packet Radio Service (GPRS); Service description; Stage 2
2	3GPP TS 23.167	IP Multimedia Subsystem (IMS) emergency sessions
3	3GPP TS 23.216	Single Radio Voice Call Continuity (SRVCC); Stage 2
4	3GPP TS 23.237	IP Multimedia Subsystem (IMS) Service Continuity; Stage 2
5	3GPP TS 23.292	IP Multimedia Subsystem (IMS) centralized services, Stage 2
6	3GPP TS 23.401	General Packet Radio Service (GPRS) enhancements for Evolved Universal Terrestrial Radio Access Network (E-UTRAN) access
7	3GPP TS 24.008	Mobile radio interface layer 3 specification; Core Network protocols; Stage 3
8	3GPP TS 24.237	IP Multimedia Subsystem (IMS) Service Continuity; Stage 3
9	3GPP TS 29.292	Interworking between the IP Multimedia (IM) Core Network (CN) subsystem and MSC Server for IMS Centralized Services (ICS)
10	3GPP TS 23.221	Architectural requirements
11	GSMA PRD IR.92	IMS Profile for Voice and SMS
12	3GPP TS 23.203	Policy and charging control architecture
13	3GPP TS 23.228	IP Multimedia Subsystem (IMS); Stage 2
14	3GPP TS 29.205	Application of Q.1900 series to bearer independent Circuit Switched (CS) core network architecture; Stage 3
15	3GPP TS 29.163	Interworking between the IP Multimedia (IM) Core Network (CN) subsystem and Circuit Switched (CS) networks
16	GSMA PRD IR.94	IMS Profile for Conversational Video Service
17	GSMA PRD IR.58	IMS Profile for Voice over HSPA
18	GSMA PRD IR.65	IMS Roaming & Interworking Guidelines

2 Service Centralization in the IMS

2.1 General

3GPP has specified in [4] and [5] the principles for centralization and continuity of services in the IMS in order to provide consistent services to the user regardless of the attached access type. In order to support this principle, originated and terminated sessions via the CS or PS domains need to be anchored in the Service Centralization and Continuity Application Server (SCC AS) in the IMS. The SCC AS must be inserted in the session path using originating and terminating initial Filter Criteria (iFC) [13]; it is configured as the first AS in the originating iFC and as the last AS in the terminating iFC chain.

2.1.1 Anchoring when using PS access

The SCC AS anchors originated and terminated sessions when using the PS access as specified in [4].

2.1.2 Anchoring when using CS access

The SCC AS anchors originated and terminated sessions when using the CS access as specified in [5]. Originated sessions are routed from the Mobile Switching Centre (MSC) Server enhanced for ICS in the VPLMN to the Serving Call Session Control Function (S-CSCF) in the Home Public Land Mobile Network (HPLMN) and the SCC AS and the Telephony Application Server (TAS) are invoked, that is the SCC AS is the first invoked AS (see also Figure 2.1.2-1). Terminating sessions are routed to the S-CSCF and the TAS and the SCC AS are invoked before routing the session to the MSC Server in the VPLMN, that is the SCC AS is the last invoked AS. The interworking between the IMS and the MSC Server enhanced for ICS for originated, terminated and mid-call services is specified in [9].

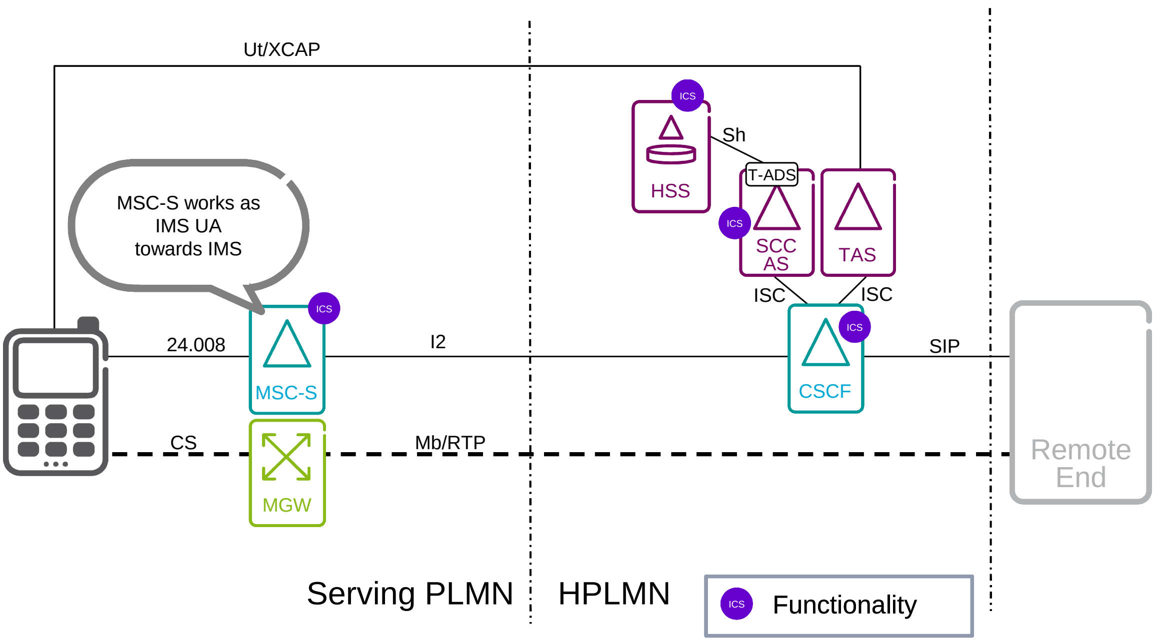

Figure 2.1.2-1: Anchoring in the IMS when using MSC Server enhanced for ICS

In deployments not using the MSC Server enhanced for ICS, originated and terminated services are also provided in the IMS, but mid-call and presentation services must be provided in the subscriber data sent to the MSC such that the MSC can provide these services, see section 7.6.3 of [5]. 3GPP describes in [5] how to route originated sessions from the MSC to the IMS by fetching an IMS Routing Number (IMRN) via CAMEL and how to route terminated session from the IMS to the MSC. Originated sessions are routed from the MSC Server in the VPLMN via a Media Gateway Control Function (MGCF) in the HPLMN to the S-CSCF and the SCC AS and the Telephony Application Server (TAS) are invoked (see also Figure 2.1.2-2). The MSC needs to fetch a routing number from the Service Control Point (SCP).

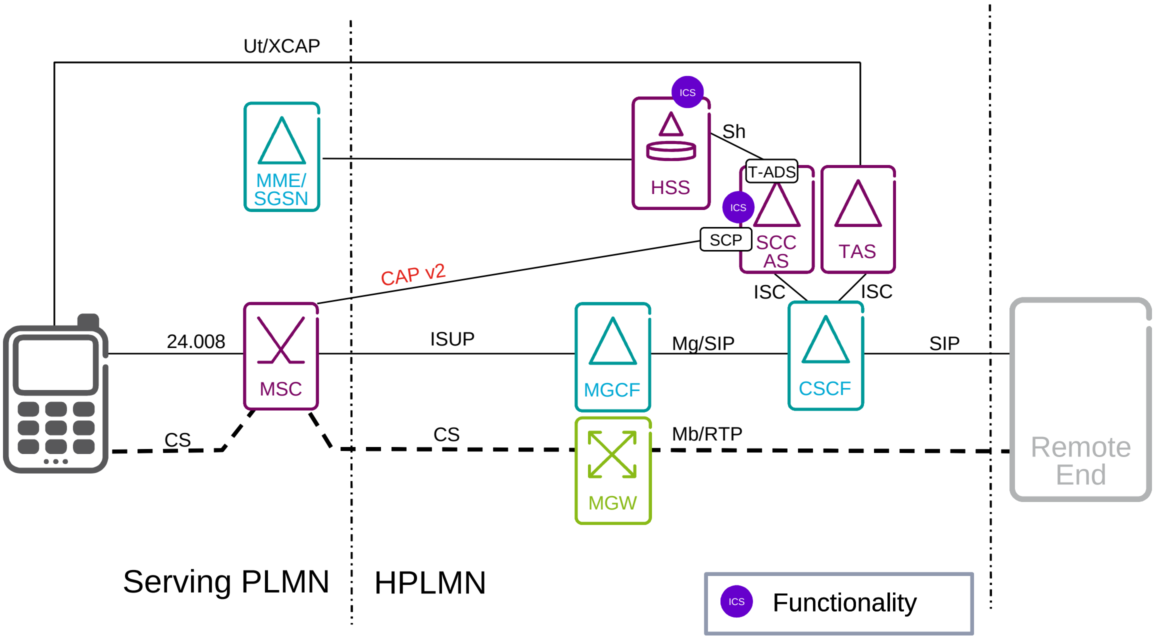

Figure 2.1.2-2: Anchoring in the IMS when using MSC not enhanced for ICS: origination

In deployments where Integrated Services Digital Network User Part (ISUP) does not provide original called number and calling party number, the SCC AS needs to discover this information by interacting with the SCP (for example GSM Service Control Function (gsmSCF)). The interaction between SCP and SCC AS for this purpose is not specified in 3GPP, see section 7.3.2.1.3 of [5].

Terminating sessions are routed to the S-CSCF and the TAS and the SCC AS are invoked before routing the session via Breakout Gateway Control Function (BGCF) (not shown) and Media Gateway Controller Function (MGCF) in the HPLMN to the MSC in the VPLMN (see also Figure 2.1.2-3).

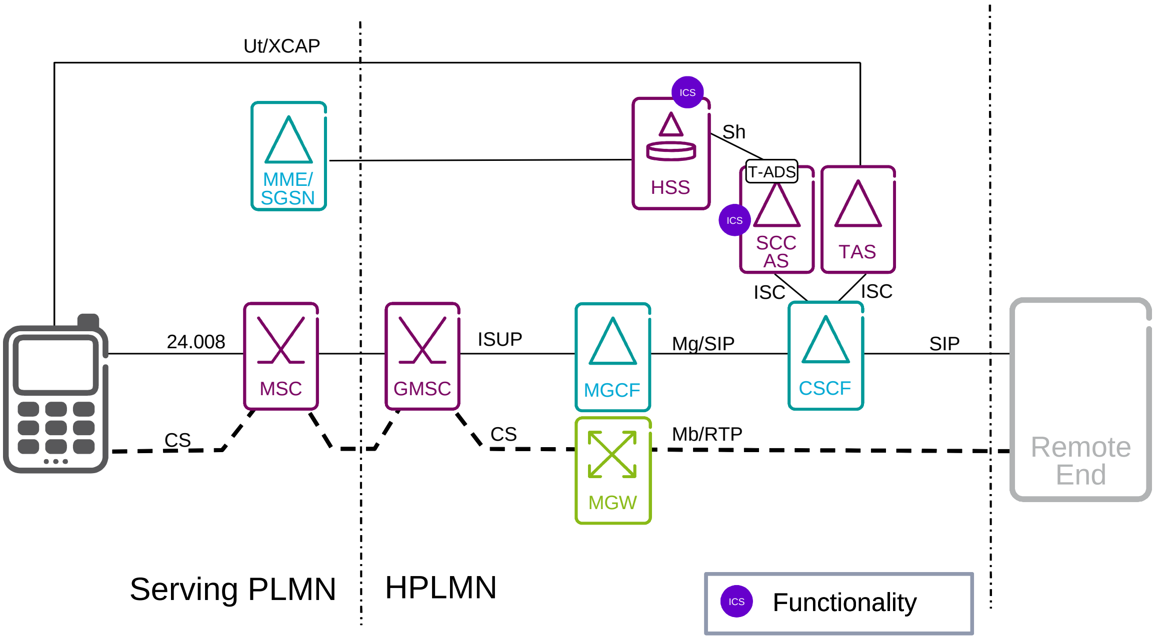

Figure 2.1.2-3: Anchoring in the IMS when using MSC not enhanced for ICS: termination

Solutions for routing a terminated session from the Public Switched Telephone Network (PSTN) domain to the IMS are described in [5], Annex F.

CS emergency calls and SMS are not centralized in the IMS, that is, they are handled as CS service when camping on CS access.

Note: That CS data and fax have been included in Release 10 into [5] but are out of scope of this document.

3 Terminated Access Domain Selection

Terminating Access Domain Selection (T-ADS) selects CS access or PS access network(s) to be used to deliver a terminating voice session to the UE. In 3GPP [4,5], also the case of selecting multiple PS accesses is covered, however, this use case is out scope of this document. T-ADS is a functionality located in the IMS in the SCC AS and specified in [4,5] and must be present in any deployment in which IMS Voice over Internet Protocol (VoIP) capable radio coverage is complemented with the CS radio access for voice.

In order to accurately determine whether to route to CS or PS access, the SCC AS must know whether the UE that is registered in the IMS is currently camping on an IMS Voice over PS capable access (for example on E-UTRAN). For this purpose, network-based T-ADS has been specified in [1,4,6,10] in 3GPP Release 9. It adds the capability to the SCC AS to query the Home Subscriber Server (HSS) for the IMS Voice over PS supported indication. If the subscriber is registered in both Serving GPRS Support Node (SGSN) and Mobility Management Entity (MME) (see also ‘dual registration’ of MME and SGSN as specified in [1,6]) and if the IMS voice over PS support differs between MME and SGSN, then the HSS queries the SGSN and the MME to request the time of the last radio access and the IMS Voice over PS supported indication (see also Figure 3-1).

The procedure requires that the UE performs Routing Area Update (RAU) and Tracking Area Update (TAU) when moving between routing areas and tracking areas with different IMS Voice over PS support (see also section A.2 in [11]).

The T-ADS selects the CS access for termination in the following cases:

- The UE is not registered in the IMS.

- The UE is registered in the IMS but not camping on an IMS Voice over PS capable access.
- The UE is only registered in the IMS by an MSC Server enhanced for ICS as specified in [4].

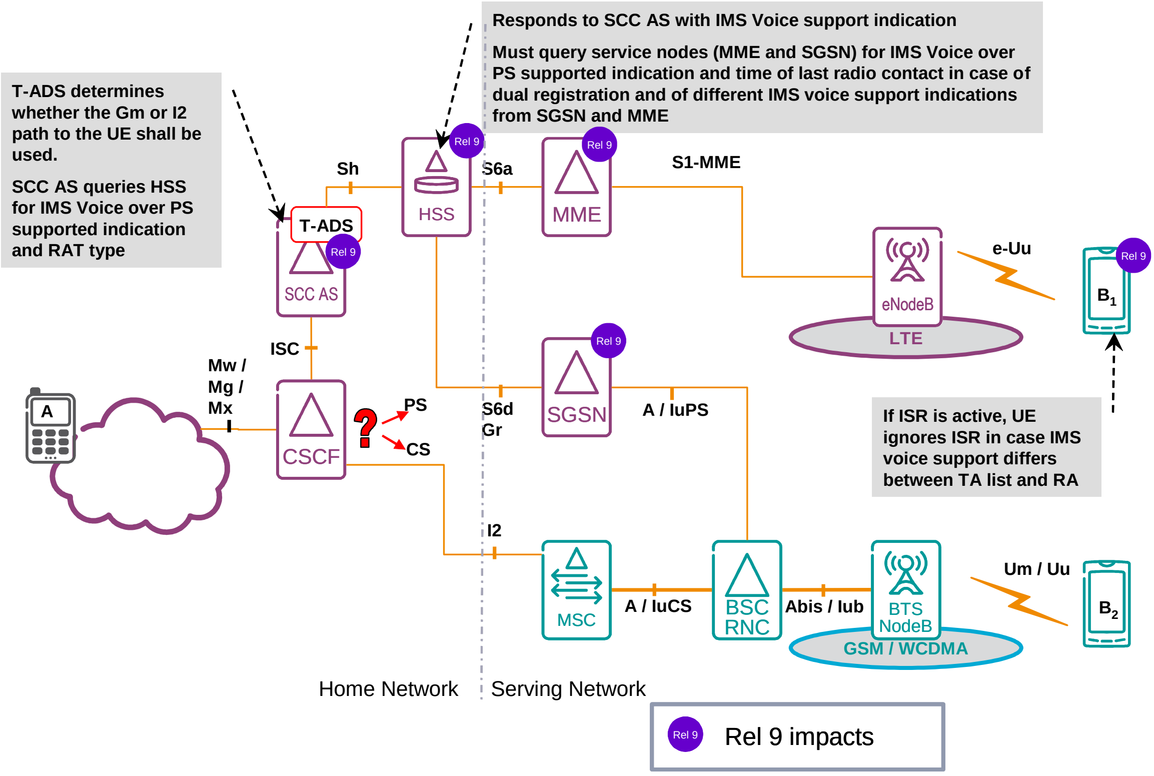

Figure 3-1: Network-based T-ADS

The interface between S-CSCF and MSC in Figure 3-1 can also be “Session Initiation Protocol (SIP) interface” as specified in [8]. The S-CSCF can also use the Mg interface to an MGCF in the HPLMN and between the MGCF and the MSC the ISUP interface is used. The interface between S-CSCF and MSC in the figure can also be Mg in non-roaming scenarios.

4 SRVCC

4.1 General

SRVCC has been specified initially in 3GPP Release 8 in [3,4] to support the transfer of one single active call / session from E-UTRAN to 2G/3G CS and HSPA to 2G/3G CS. Transfer of mid-call state and additional calls using MSC Server assisted mid-call feature was introduced in 3GPP Release 9 (see section 4.4). In Release 10, SRVCC support for calls in alerting state (see section 4.4) and SRVCC enhancements to minimize voice interruption delay (see section 4.3) were specified. SRVCC must be supported in any deployment in which IMS VoIP capable radio coverage is complemented with the CS radio access for voice.

4.2 Impacted entities

SRVCC from E-UTRAN to 2G/3G CS requires support in the UE, eNodeB (eNB), MME, Public Data Network Gateway (PGW) , P-CSCF, Access Transfer Control Function (ATCF), and MSC Server in the VPLMN and in SCC AS and HSS in the HPLMN as shown in Figure 4.3.1-1 and as specified in [3,4,7,8]. In addition, Policy and Charging Control (PCC) is impacted [12].

For an S4 based SGSN, SRVCC from HSPA to 2G/3G CS has the same impacts as SRVCC from E-UTRAN to 2G/3G as depicted in Figure 4.3.1-1, but with the following exceptions:

- Instead of the eNodeB the source UTRAN is impacted
- Instead of the MME the SGSN is impacted.

For an Gn based SGSN, the Gateway GPRS Support Node (GGSN) instead of the PGW is impacted. Figure 4.3.1-2 shows an S4 based SGSN as both source and target serving node, but use of a Gn based SGSN is possible as well either as source, or target or both source and target serving node.

Note: SRVCC from E-UTRAN to 2G/3G has an impact on PGW, P-CSCF and PCC because during SRVCC the MME deletes the PS bearer used for voice media [3]. The PGW is notified of the loss of bearer with a cause code that this is due to SRVCC. PCC [12] is notified of the loss of bearer with the same cause code and in turn informs the P-CSCF that this loss of bearer is due to SRVCC and hence part of the mobility procedure.

4.3 SRVCC Architecture

4.3.1 Overview

To minimize the voice interruption delay, 3GPP has specified in Release 10 the SRVCC architecture as shown in Figure 4.3.1-1 [4.8]. The ATCF and Access Transfer Gateway (ATGW) are deployed in the VPLMN. An ATCF can be co-located with existing nodes, for example, P-CSCF or Interconnection Border Control Function (IBCF). An existing MGW is used in case of ATGW. The interface between ATCF and MSC in the figure can also be “SIP interface” as specified in [8].

The possible cause for the possible long voice interruption in SRVCC Pre-Release 10 is related to the following two procedures:

1. The interaction between MSC Server and SCC AS for transferring the session and the update of the remote end with new media information (Session Description Protocol (SDP)).
2. The interaction between the MSC Server and SCC AS is in roaming cases over the NNI and the remote end update requires an SDP offer/answer exchange towards the remote party.

The later ensures that the remote party voice RTP stream is changed toward the MGW controlled by the MSC Server. Considering that one or both parties involved in a call subject for SRVCC might be roaming, then the voice interruption is negatively influenced by the interaction between the involved VPLMNs and HPLMNs and due to long round-trip time of the remote party update.

Hence enhanced SRVCC Rel 10 architecture introduced the ATCF and the ATGW. The ATCF acts as SIP signalling anchor and is located in the SIP signalling path between P-SCSF and S-CSCF. Both the ATCF and ATGW are located in the serving network. The ATCF controls the ATGW, where the media plane is anchored. For the ATGW, no additional MGW has to be deployed, but one of the existing media gateways can be used, for example, IMS-AGW, Transition Gateway (TrGW), or CS-MGW. It is recommended to always anchor the media in the ATGW during session establishment as specified in section 5.3.4.2 in [4].

During the session transfer, the ATCF establishes a new session with the SCC AS. This new session substitutes the old session between the ATCF and the SCC AS [8].

Both ATCF and ATGW are needed to ensure that:

- The session transfer request send by MSC Server enhanced for SRVCC [3,4] does not need to be routed to the HPLMN.
- Remote end update is not needed.

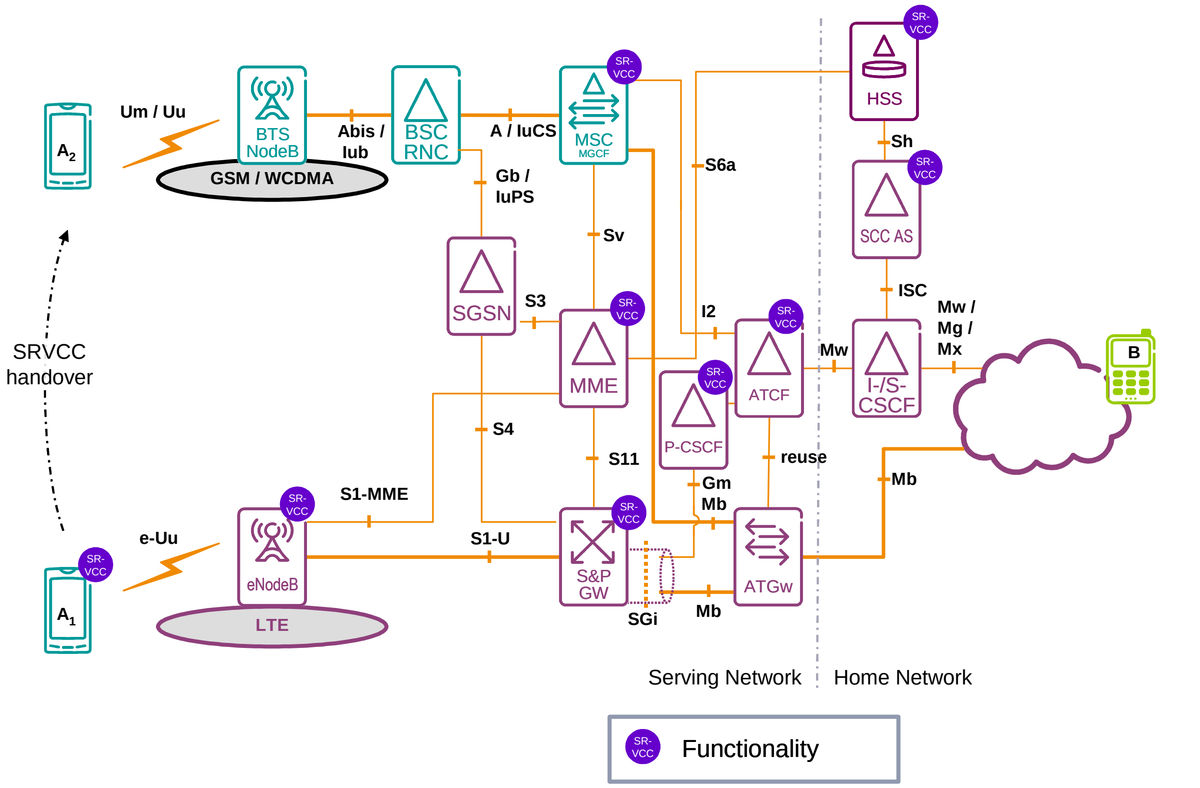

Figure 4.3.1-1: Enhanced SRVCC R10 Architecture - E-UTRAN to GERAN/UTRAN

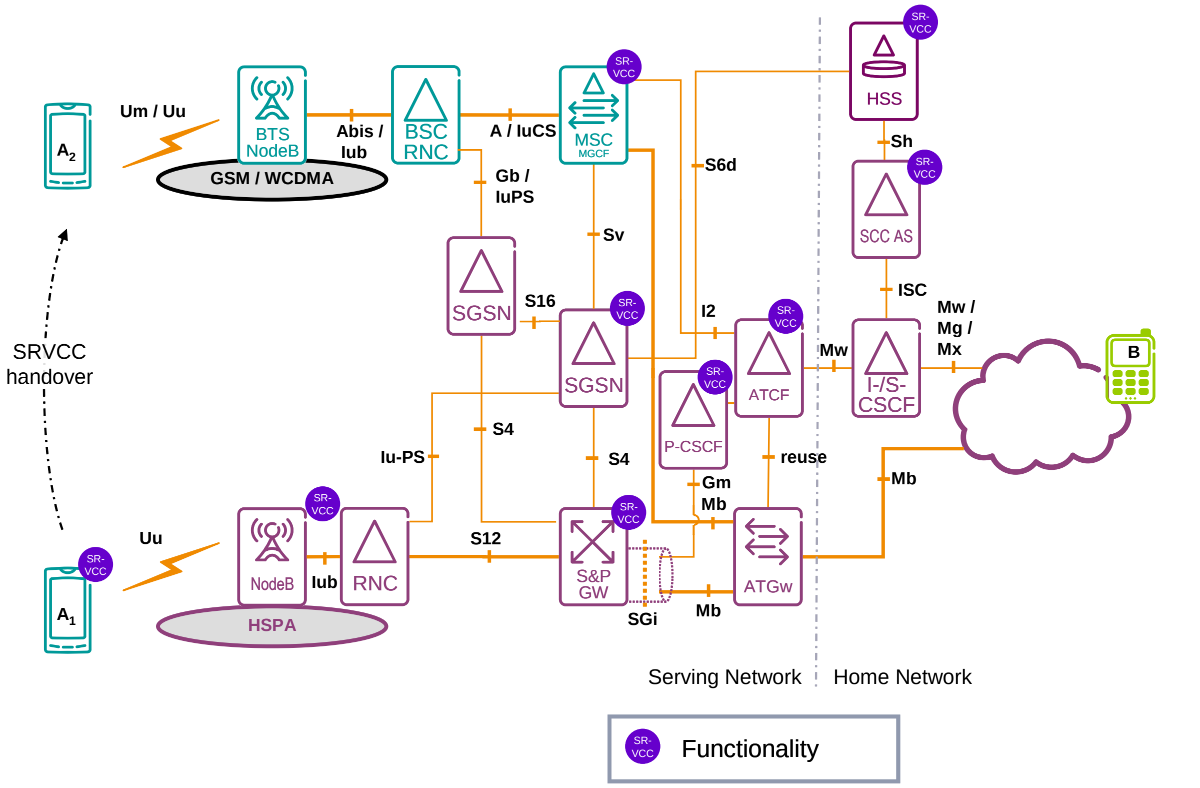

Figure 4.3.1-2: Enhanced SRVCC R10 Architecture - HSPA to GERAN/UTRAN

Prior to performing SRVCC, all originated and terminated sessions on E-UTRAN/HSPA are anchored in the SCC AS, as described in section 2.1.1. After the handover to CS access, the transferred sessions stayed anchored in the SCC AS and newly originated and terminated sessions are anchored in the SCC AS, as described in section 2.1.2. For all terminated sessions, T-ADS is executed as described in section 3.

The SCC AS in the home network provides session state information in the case MSC Server assisted mid-call feature is supported and used, and continues to handle T-ADS related functionality.

When IP interconnect towards the remote end is used also in non-roaming scenario as described in Section 5.2 of IR.65 [17], it is recommended that the home network deploys both ATCF and ATGW. This ensures that the remote end update is not needed.

4.4 Support of mid-call state and calls in alerting state

4.4.1 Overview

MSC Server assisted mid-call feature is specified in [4,8,12] in 3GPP Release 9. It adds to SRVCC the support to transfer:

- Single held call
- One active and one held call
- Conference call state including participants. Conference call can be the active or the held call.

SRVCC for calls in alerting state is specified in [4,8,12] in 3GPP Release 10. It adds to SRVCC the support to transfer:

- Single call in alerting state
- Call in alerting state while there is an established session in place. The established session is an active or held session and can be a conferencing session.

Note: Support of both mid-call feature and alerting state feature is required to support combinations of held, conferencing and alerting state.

4.4.2 UE impacts

The UE impacts required to support during SRVCC mid-call state are specified in [7,8] in Release 9 and calls in alerting state are specified in [7,8] in Release 10.

4.4.3 Network impacts

Support of mid-call state and calls in alerting state during SRVCC requires support in MSC Server, SCC AS and ATCF as specified in [4,7,8].

4.5 SRVCC Architecture for emergency calls

SRVCC for emergency calls is specified in [2,3,4] in 3GPP Release 9. It must be supported in any deployment supporting SRVCC where IMS emergency calls are used over IMS VoIP capable radio coverage and the later is complemented with the CS radio access. The Emergency Access Transfer Function (EATF) is a logical function required in the VPLMN in addition to the functions needed for SRVCC (see also section 4.3) and IMS emergency calls [2] see also Figure 4.5.1-1. The MSC uses the Mw interface to the (Interrogating Call Session Control Function) I-CSCF and includes the equipment identifier into the session transfer request; the equipment identified is used by the EATF to correlate the call legs. SRVCC for emergency calls does not make use of the ATCF and ATGW as described in section 4.3.

The MSC can also use the ISUP interface to an MGCF and then the Mg interface is used between the MGCF and the I-CSCF as specified in [4]. It is assumed that if ISUP is used, the ISUP extension of carrying the International Mobile Equipment Identity (IMEI) to the MGCF is supported [14,15]. The support of SRVCC for emergency calls in deployment where ISUP or SIP does not provide the equipment identifier is implementation and configuration dependent according to section 6c.3 in [4] and hence not recommended.

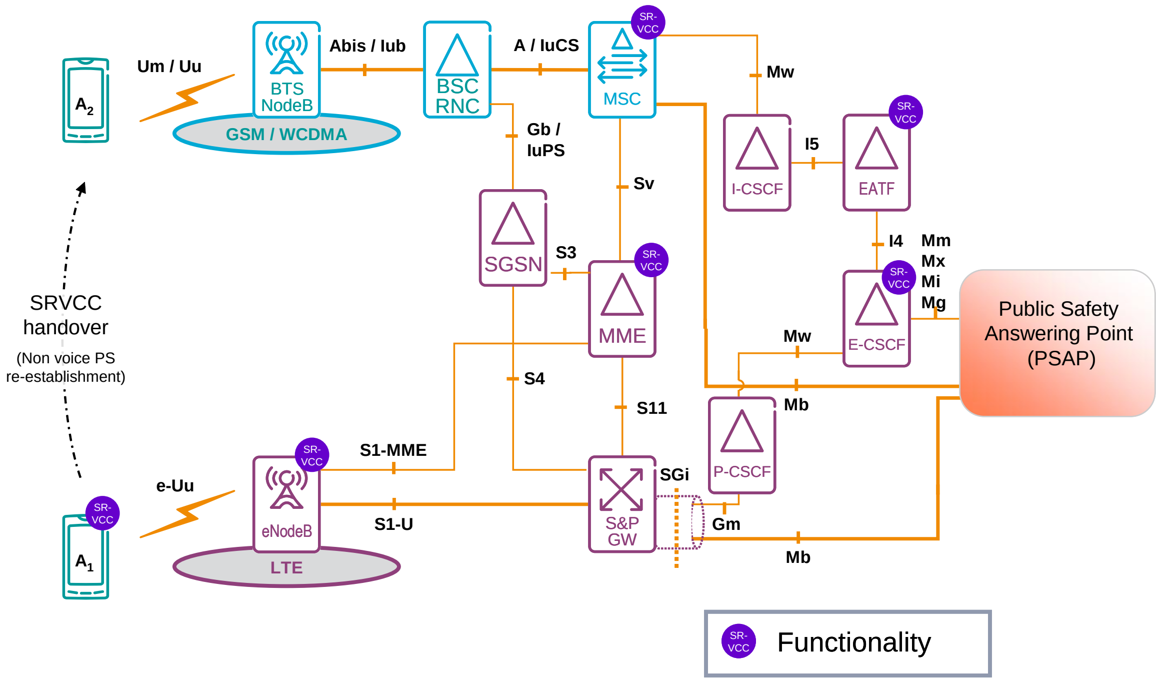

Figure 4.5.1-1: SRVCC for emergency calls Architecture

4.5.1 UE impacts

For UE impacts required to support SRVCC for emergency calls please see [11].

4.5.2 Network impacts

Support of SRVCC for emergency calls requires support in MME, MSC Server, E-CSCF and EATF as specified in [2,3,4,8].

4.6 Handling of non-voice media during SRVCC

In case the IMS session includes besides voice also other media, then during transfer those other media are handled as follows:

- If the target access is GERAN non Dual Transfer Mode (DTM), then all non voice media are removed from the session by the SCC AS as described in [8], clause 12.2.3.
- If the target access is UTRAN or GERAN with DTM (and the UE supports GERAN with DTM), then the UE informs the SCC AS about which non-voice media to keep on PS

access as described in [8], clause 12.2.3. The case of IMS session that includes synchronised video media is described in [16].

Support of non-voice media in emergency calls is out of scope of this document.

Document Management

Document History

Version	Date	Brief Description of Change	Approval Authority	Editor / Company
0.1	26/05/11	New PRD (RILTE Doc 17_011r1).	RILTE #17	Ralf Keller, Ericsson
0.2	08/06/11	Additional input. Reviewed by Jose Antonio Aranda and Donna Mackay (RiLTE Doc 18_008)	RILTE #18	Ralf Keller, Ericsson
0.3	27/06/11	Removal of SRVCC Rel 9 architecture	RILTE #18	Ralf Keller, Ericsson
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0.9	03 October 2011	Submitted to DAG & EMC for approval final approval date 24 October 2011	EMC	Ralf Keller, Ericsson
1.0	18 October 2011	Removal of coversheet after approval	DAG #86 EMC #97	Ralf Keller, Ericsson
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		• MCR 003: SRVCC from HSPA • MCR 004: eSRVCC for Interconnect Scenario • MCR 005: Handling of non-voice media during SRVCC • mCR 006: Correction to diagram - Minor CR	DAG #92, PSMC # 102	Ericsson
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